IP Phone
User Manual

Safety Notices

Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.
- Before using the external power supply in the package, please check with home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature, below 0℃ or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

Table of Content

1. Introduction

1.1. Thank you for your purchasing

Thank you for your purchasing this IP Phone, It is a full-feature telephone that provides voice communication over the same data network that your computer uses. This phone functions not only much like a traditional phone, allowing to place and receive calls, and enjoy other features that traditional phone has, but also it own many data services features which you could not expect from a traditional telephone.
This guide will help you easily use the various features and services available on your phone.

1.2. Delivery Content

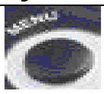
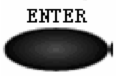
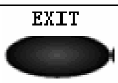
Please check whether the delivery contains the following parts:

The base unit with display and keypad
The handset
The handset cable
The power supply
The Ethernet cable

1.3. Keypad

The numeric keypad with the keys 0 to 9, *, and # is used to enter Digits and letters, additionally, the following keys are available:

Key mapping:

Key	Description
	In idle state，press the MENU key to call up the menu.
	The phone can realize the following features by the UP key or the DOWN key. When you pick up the handset or during calling, use the UP key or the DOWN key to adjust volume; Use the UP key or the DOWN key to browse menu; cancel or confirm action; browse calling list.
	In idle state, press the SYSINFO key for once to look up this VoIP Phone Number, this VoIP Phone local IP address for twice and Local Gateway IP address for three times.
	Press the ENTER key to confirm action, selection, or enter into the next menu in menu mode.
	Use the EXIT key to return to the previous menu in menu mode.
	In idle state, press the IN key or the OUT key to browse missed call, received call or dialed call, and realize dialing by the REDIAL/ SEND key.
	In idle state, press the REC key to look up new, old received Voice record and user-defined voice record, and plays them. During call, press the REC key to record call content.
	In idle state, press the PBOOK key to access phone book, then use the REDIAL/ SEND key to dial. You can browse phone book by the UP key or the DOWN key.
	In menu mode, use the DEL key to delete.
	Mute microphone on/off, during a call.

	Press the HOLD key, input the third party telephone number, then press the # key to realize the third party call. If you want to switch back from the third party call, press the HOLD key again.
	Press the TRANSFER key during call, can realize blind transfer and attended transfer.
	Press the REDIAL/SEND key to dial the last dialed number. Select contact name or telephone number in Phone book, Then press this key to send the number.
	Switch to hand-free mode and back.

1.4. Port for connecting

POWER	Power switch	Select ON/OFF
DC 5V	Power port	Output: 5V/1A
LAN	Network port	Connect it to PC
WAN	Network port	Connect it to Network

The phone has two Network ports: The WAN port and the LAN port. Before you connect the power source, please carefully read Safety Notices of this user manual.

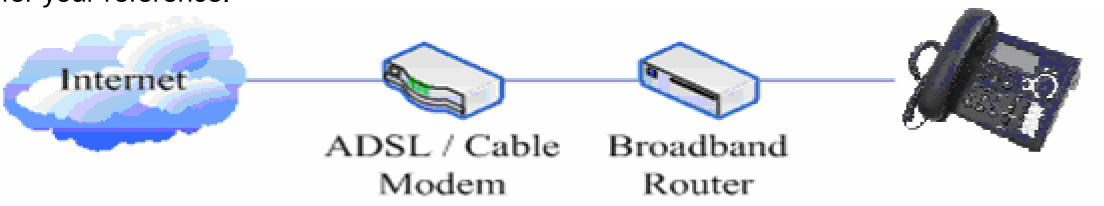
2. Initial connecting and Setting

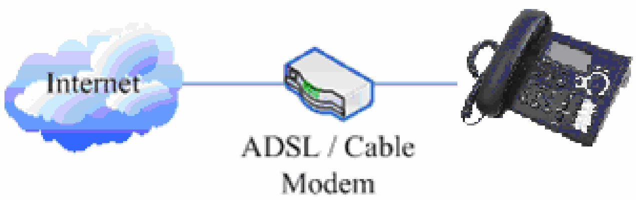
2.1. connect the phone

Step 1: Connect the IP Phone to the corporate IP telephony network. Before you connect the phone to the network, please check if your network can work normally.

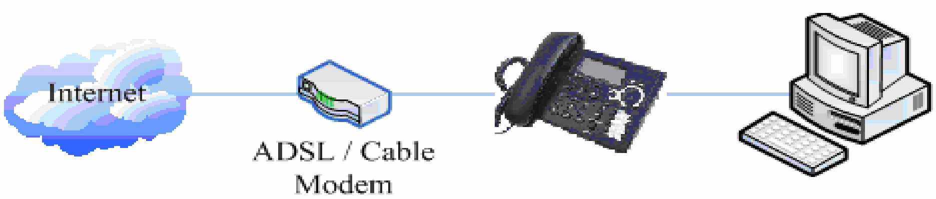
You can do this in one of two ways, depending on how your workspace is set up.

Direct network connection—by this method, you need at least one available Ethernet port in your workspace. Use the Ethernet cable in the package to connect WAN port on the back of your phone to the Ethernet port in your workspace. Since this VoIP Phone has router functionality, whether you have a broadband router or not, you can make direct network connect. The following two figures are for your reference.





Shared network connection—Use this method if you have a single Ethernet port in your workspace with your desktop computer already connected to it. First, disconnect the Ethernet cable from the computer and attach it to the WAN port on the back of your phone. Next, use the Ethernet cable in the package to connect LAN port on the back of your phone to your desktop computer. Your IP Phone now shares a network connection with your computer. The following figure is for your reference.



Step 2: Connect the handset to the handset port by the handset cable in the package.

Step 3: connect the power supply plug to the DC 5V adapter port on the back of the phone. Use the power cable to connect the power supply to a standard power outlet in your workspace.

Step 4: push the on/off switch on the back of the phone to the on side, then the phone's LCD screen displays "WAIT LOGON". Later, a ready screen typically displays the date, time and current network mode.
If your LCD screen displays different information from the above, you need refer to the next section

“Initial setting” to set your network online mode.

If your VoIP phone registers into corporate IP telephony Server, your phone is ready to use.

2.2. Initial Setting

This VoIP Phone provides you with rich function and parameters setting. If you have enough knowledge about network and SIP protocol, it is better for you to understand many parameters. But if you know little about network and SIP protocol, you can also easily make initial setting according to the following steps to enjoy rapidly high quality voice and low cost from this VoIP Phone.

Before make initial setting, please check if your corporate IP telephony network can work normally, and you have finished “connect the phone”.

This VoIP Phone Supports DHCP by default. It will receive an IP address and other network-related settings (Netmask, IP gateway, DNS server) from the DHCP server. If your network supports DHCP, you can connect this VoIP Phone directly to the network. If your network doesn’t support DHCP, you need change this VoIP Phone’s network connection setting. According to the following steps, change this VoIP Phone’s DHCP network connection setting into PPPoE or static IP which your network supports at present.

2.2.1. PPPoE mode.

1. Prepare your PPPoE account name and password.

2. Press the MENU key, the LCD screen will display “INPUT PASSWORD”.

3. Input the password (default value is 123), and press the ENTER key, the LCD screen will display “NETWORK”.

4. Press the ENTER key and LCD screen will display “LAN”, press the DOWN key, enter it by the ENTER key, the LCD screen will display “STATIC NET”. Then press the DOWN key again, enter it by the ENTER key, the LCD screen will display “USER NAME”.

5. Press the DOWN key, the LCD screen will display “PASSWORD”. Then press the ENTER key, and the DEL key, input your PPPoE’s password and confirm it by the ENTER Key, the LCD screen will display the password which you inputted.

6. Press the EXIT key to return to the previous menu, then press the DOWN key, the LCD screen will display “USER NAME”. Press the ENTER key, and the DEL key, input your PPPoE’s account name, then press the ENTER key to confirm it, the LCD screen will display the PPPoE’s account name which you inputted.

7. Press the EXIT key for four times and press the DOWN key, till the LCD screen display “SYSTEM”.

8. Press the ENTER key, the screen display “SAVE”, then press the ENTER key again, the LCD screen will display “ARE YOU SURE”.

9. Press the ENTER key, the phone will save your setting and the LCD screen will display “SAVING”, then return to display “SAVE”.

10. Press the EXIT key twice, then press numeric key “3”and hold until the screen display “ARE YOU SURE”. Press the ENTER key, the screen will display “CHANGING”, which means that the

phone is trying to switch to PPPoE mode. If the icom "PPPoE" on the top of the screen keeps blink, it shows that the phone is trying to access the PPPoE server., and the IP is still static IP if you press SYSINFO key to display the current IP; if the icon "PPPoE" is showed without blink, it means that the phone has already gotten IP from PPPoE server.

2.2.2. Static IP mode:

1. Prepare your phone's network parameters. They are IP Address of this phone, Subnet Mask, Default Gateway/ Router and DNS. You can ask your VoIP service provider for those parameters.

2. Press the MENU key, the LCD screen will display "INPUT PASSWORD".

3. Input password (default is 123), then press the ENTER key, the LCD screen will display" NETWORK".

4. Press the ENTER key, and the LCD screen will display "LAN". Press the DOWN key, then the ENTER key, the LCD screen will display "STATIC NET".

5. Press the ENTER key, the LCD screen will display "IP". Press the ENTER key again and then the DEL key, input your desired IP address for your IP phone and confirmed by pressing the ENTER key, then the LCD will display the input IP address. When inputting IP with keypad, use "*" instead of ".".

6. Press the EXIT key to return to previous menu, then press the DOWN key for twice, the LCD screen will display "DNS". Press the ENTER key then the DEL key, input your DNS address and confirm it by pressing the ENTER key, and then the LCD will display the input DNS address.

7. Press the EXIT key to return to the previous menu, and then press the DOWN key, the LCD screen will display "GATEWAY". Press the ENTER key again and then the DEL key, input your gateway's IP address and confirm it by pressing the ENTER key, the LCD screen will display the input gateway address.

8. Press the EXIT key to return to the previous menu, and then press the DOWN key, the LCD screen will display "NETMASK". Press the ENTER key again and then the DEL key, input your netmask and press the ENTER key to confirm it. The LCD screen will display the input netmask.

9. Press the EXIT key for four times and press the DOWN key, till the LCD Screen displays "SYSTEM".

10. Press the ENTER key, the LCD screen will display "save", then press the ENTER key again, the LCD screen will display" ARE YOU SURE".

11. Press the ENTER key, this phone will display "SAVING", then return to display "SAVE".

12. Press the EXIT key twice to exit the menu, and then press the numeric key 1 till the LCD screen displays "ARE YOU SURE". Press the ENTER key, the LCD screen will display "CHANGING", If the icon "static" on the top of screen shows without blink, it means phone has already used the static IP.

2.2.3. DHCP mode

Press the numeric key 2 and hold till the LCD screen displays "ARE YOU SURE". Press the ENTER key, the LCD screen will display "CHANGING" and this VoIP phone is trying to switch to DHCP mode. If the icom "DHCP" on the top of the screen keeps blink, it shows that the phone is trying to access the DHCP server., and the IP is 0.0.0.0 if you press SYSINFO key to display the current IP;

if the icon “DHCP” is showed without blink, it means that the phone has already gotten IP from DHCP server.

3. Basic Functions

3.1. Basic operation

3.1.1. Accepting a call

There are four methods to accept an incoming call:

- Pick up handset to accept incoming calls.
- Press the **SPEAKER** button
- If you need switch from a hands-free call to handset, please pick up the handset directly.
- If you need switch from a handset call to hands-free, please press the **SPEAKER** button, and then hang up the handset.

3.1.2. Making a call

- Use handset

  Pick up the handset, and the LCD screen will display “**PLEASE DIAL**” and you will hear dialing tone at the same time, then input the phone number and end by the # button. When you hear long ring “du, du…” from handset and the LCD screen display “**CALLING**” the call is through. Hang up the handset to end the call.
- Use hands-free

  Press the **SPEAKER** button and the LCD screen will display “**PLEASE DIAL**” and you will hear dialing tone at the same time, then input the phone number and end by the **#** button. When you hear long ring "du, du…" and the LCD screen display "**CALLING**" the call is through. Press the **SPEAKER** button again to end the call.

- Use the phone book
  Press the **PBOOK** button then the **ENTER** button you will enter into the phone book. Press the **UP/DOWN** button to select your desired contact person, then press the **REDIAL/SEND** button to dial the call.
- Onhook dial
  Input the called number, and press # key or **REDIAL/SEND** button, phone will dial the call and use hands-free automatically.

3.1.3. Ending a call

- Hangs up by handset onhook
- Hangs up by press speaker when in hands-free
- Hangs up a call when in call waiting state. If you are in call waiting state, you could press # key to hang up the current call, and switch to the other call to keep talking.

Pressing # key will not hang up if there is only one call currently.

3.1.4. Transferring a call

Call transfer has several ways to realize:

1. When A talks to B, B may press the HOLD key and dial to C phone number. After B talks to C (or B hear alert from C), B presses the TRANSFER key; B could hang up, and A will get through to C.
2. When A talks to B, there is C call incoming to B; B may press the HOLD key to hold A, and talks to C, pressing the TRANSFER key, so A will get through to C.
3. When A talks to B, B presses the TRANSFER key, dial C phone number and # key, B could hang up and A will get through to C.

1 and 2 are attended transfer; 3 is blind transfer.
Notice to VoIP Phone Carrier: Your VoIP phone server need support FRC3515, or else transferring can not work.

3.1.5. Calling Hold and 3 ways call

There are two modes to enjoy hold function:

1. Press the **HOLD** key during a call, and the call will be on hold. While a call is on hold, you can establish another call by dialing your desired number and confirm it by the # button. Pressing the **HOLD** key again will resume the first call. By using hold function, you can talk with only one party; the other party who is on hold can't talk with you. If you press the * button, you will enter into **3 ways call.**
2. If the third party calls you during a call, the LCD screen will display the incoming call number. Press the hold key or # key to hold the first call, and then you can talk with the third party. By using hold function, you can talk with only one party; the other party who is on hold can't talk with you. If you press # key, phone will hang up the first call, and then accept the new incoming call.

Notice: You must enable the calling waiting or else calling hold can't work.

3.1.6. Calls list

The VoIP phone maintains lists of missed, received, and dialed calls. Each list can contain up to 100 entries. If the call list capacity is full, new call will replace the first call. If you stop power supply or restart the phone, the record will disappear.

- **Missed Calls**
  Press the **IN** key, and then the **UP/DOWN** key, till the LCD screen display "**MISSED**". Press the **ENTER** key, the LCD screen will display the missed call number and sequence numbers of the

missed calls.
You can press the **REDIAL/SEND** key to dial this phone number, or you can press the **ENTER** key, the LCD screen will display the time of the missed calls. If there is no one missed calls, the LCD will display "**LIST IS EMPTY**".

- **Received Calls**
  Press the **IN** key, and then the **UP/DOWN** key, till the LCD screen display "**RECEIVED**". Press ENTER key, the LCD screen will display the received call numbers and sequence numbers of the received calls. You can press the **REDIAL/SEND** key to dial this phone number, or you can Press the **ENTER** key, the LCD screen will show the time of the received call. If there is no one received call, the LCD will display "**LIST IS EMPTY**".
- **Dialed calls**
  Press the **OUT** key, the LCD screen will display the phone numbers and sequence numbers of the dialed calls. You can press the **REDIAL/SEND** key to dial this phone number, or press the **UP/DOWN** key to browse all records of the dialed calls. If there is on one dialed calls, the LCD will display "**LIST IS EMPTY**".

3.2. The high-level operation

This VoIP Phone provides more advanced functions after setting at the permission scope of SIP server. Please refer to next section to operate.

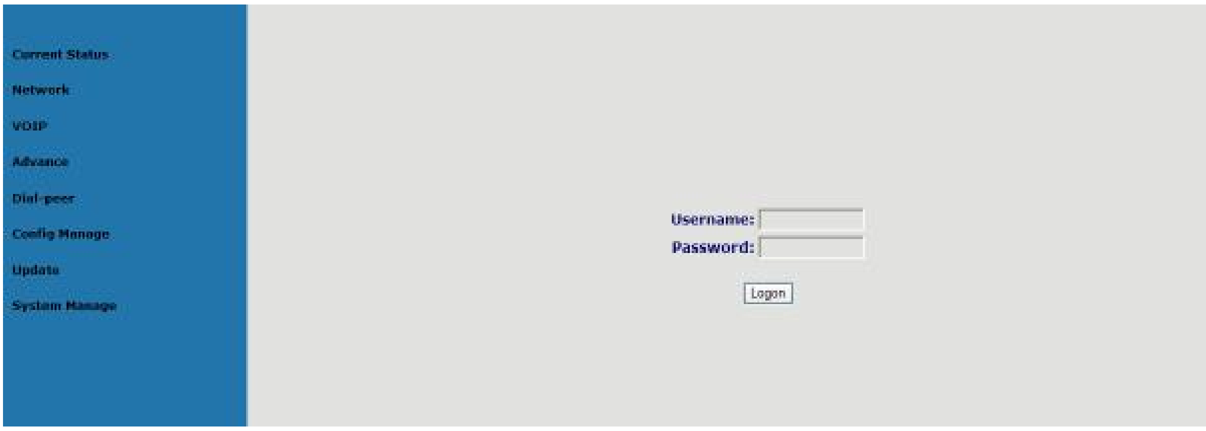
4. Setting

4.1. Setting methods

VoIP Phone is different from the traditional phone; it need be set to make it active. If your VoIP service provider asks you to set this phone, you can do it easily according to the following methods.
This VoIP Phone can be set via three different setting methods:
The phone key
The web browser on PC
Telnet
This Manual will tell you about the setting methods via the web browser on PC.

4.2. Setting via Web Browse

When this phone and your PC are connected to your network, enter the IP address of the wan port in this phone as the URL (e.g. http://xxx.xxx.xxx.xxx/ or http://xxx.xxx.xxx.xxx:xxxx/).
If you do not know the IP address, you can look it up on the phone's display by pressing the key "**SYSINFO**" for at most three times.
After you enter the IP address, you will see the following web interface.



This phone provides different two privileges for different users to set it.
The two privileges are guest and administrator respectively. In guest privilege, user can see but not modify Register/Proxy Sever Address and port of SIP, advance SIP and Iax2. In administrator privilege, user can see and modify all setting parameters.

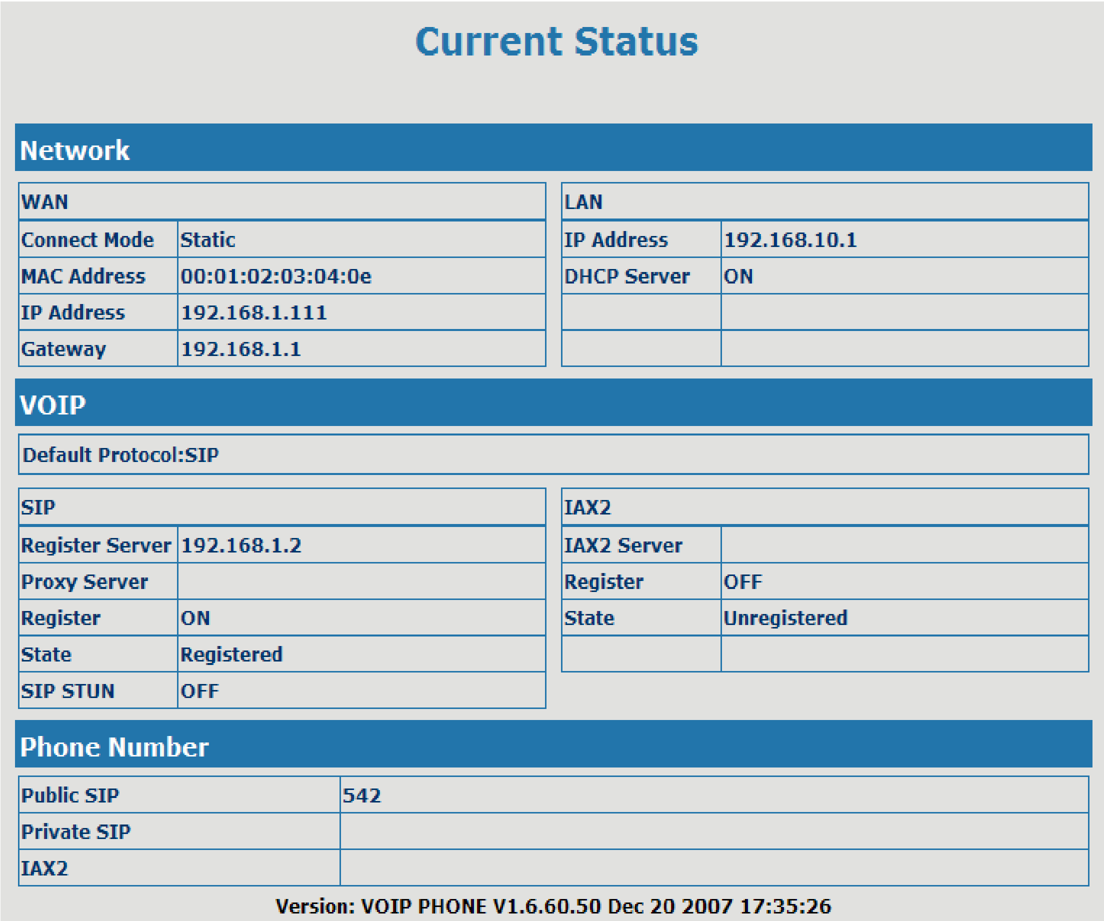
Default value in guest privilege
Username: guest
Password: guest

Default value in Administrator privilege
Username: admin
Password: admin

Input username and password, click "logon", and you will enter setting web interface.
There is a selection menu on the left side of the web interface. Click on the desired submenu; the current settings of this submenu will be displayed in the larger field on the right. You can now modify and store the values by using mouse and keyboard of your PC. To save the changes, click on the submenu of "Save Config" under "Config Manage", then click the "Save" button on the right field.

4.2.1. Current Status

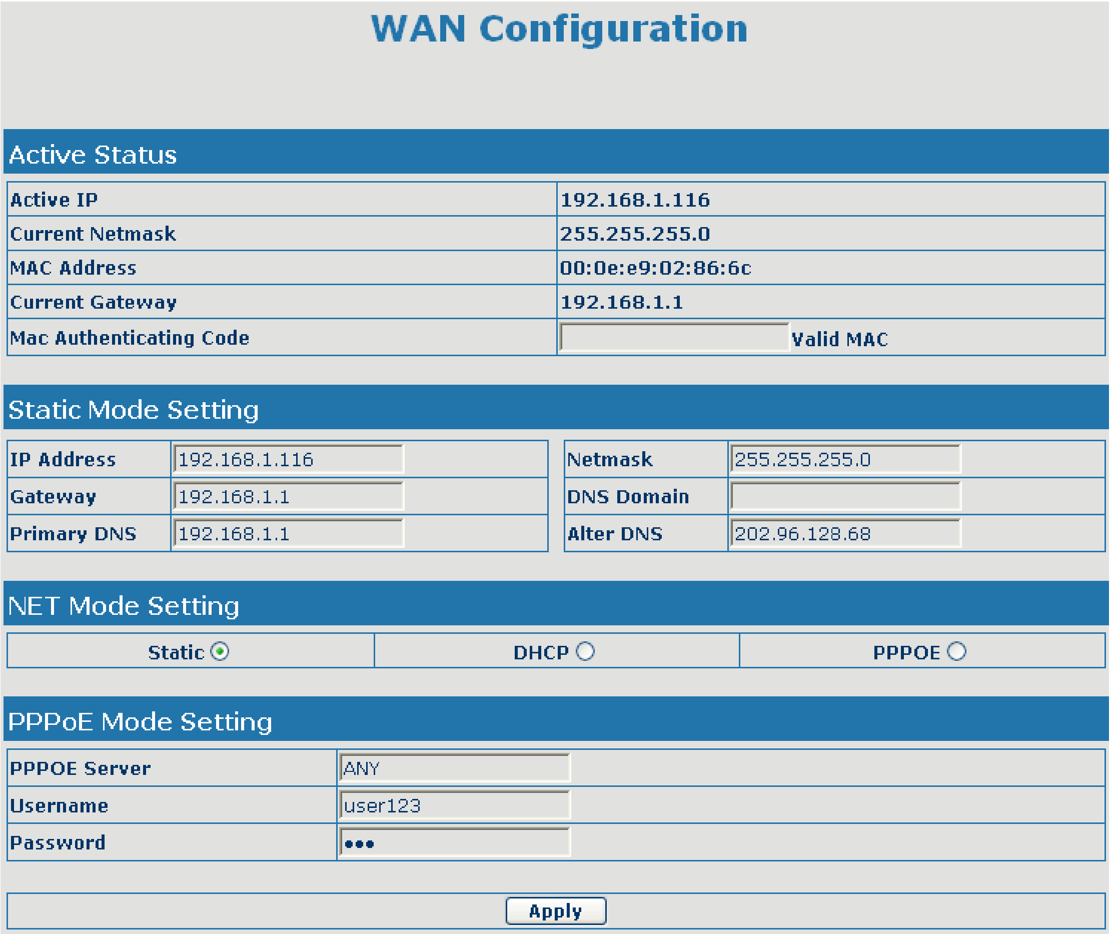
Click on the first submenu "Current status", you will enter in the following web interface. In this web interface, you will see current set parameters, status, and the firmware version.



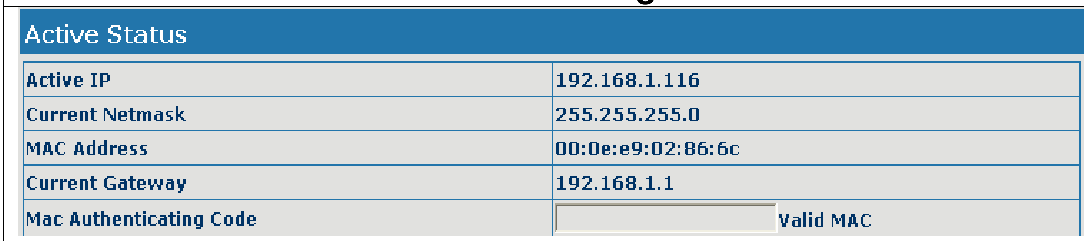
Current Status	
Network	Shows the configuration information on WAN and LAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port and LAN port, ON or OFF of DHCP mode of LAN port.
VoIP	Shows the current protocols of the phone, and some parameters of every protocol. You can know about IP addresses of register servers of both IAX2 and SIP, proxy server IP address, whether start to register the SIP and IAX2 servers or not, whether be registered or unregistered, and whether start to register the STUN server.
Phone Number	Shows the phone numbers provided by the SIP, SIP2 and IAX2 servers. The last line shows the version number and issued date.

4.2.2. Network

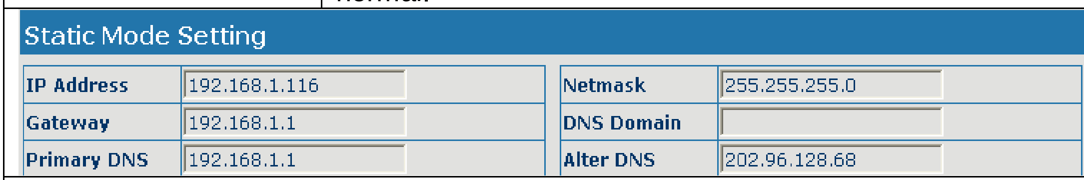
4.2.2.1. WAN Config



WAN Config



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
Mac Authenticating Code	Set the corresponding authenticating code of MAC. If you don't pass the authentication, then it will show "invalid MAC", at this time, phone will have no sound while the network is normal.



If you use static mode, you need set it.

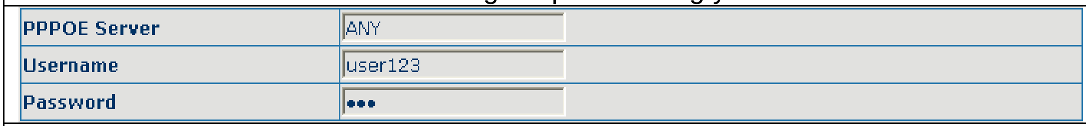
IP Address	Input the IP address distributed to you.
Netmask	Input the Netmask distributed to you.
Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you input

	can not be parsed, gateway 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.

Please select the proper network mode according to the network condition. this VoIP Phone provide three different network settings:

- Static: If your ISP server provides you the static IP address, please select this mode, and 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.

You can also refer to 2.2. Initial Setting to speed setting your network.



If you uses PPPoE mode， you need to make the above setting.

PPPoE Server	It will be provided by ISP.
User	Input your ADSL account.
Password	Input your ADSL password.

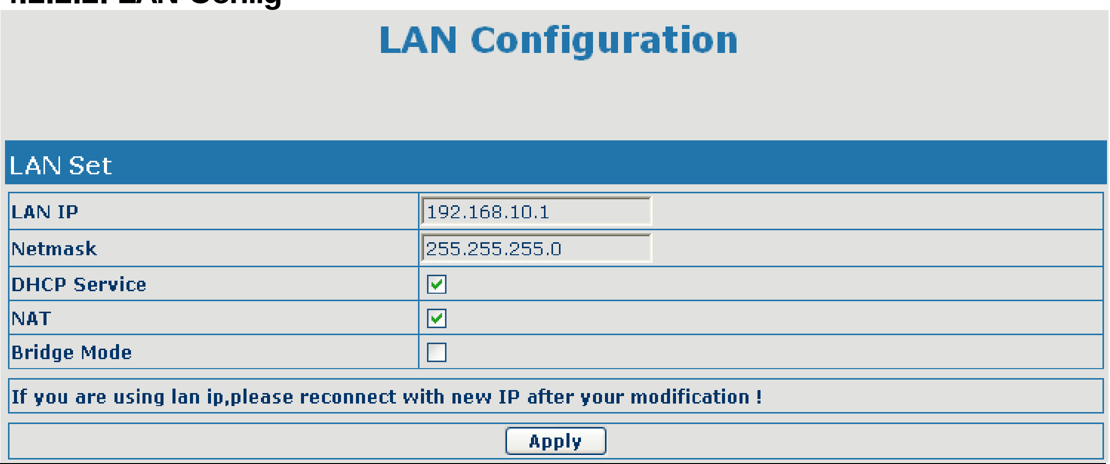
Notice:

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 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 distributed by DHCP server is the same as network ID which is used by LAN of system, 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 system uses DHCP client to get IP in startup; if system uses DHCP client to get IP in running status and network ID is also same as LAN's, system will refuse to accept the IP to configure WAN. So WAN's active IP will be 0.0.0.0

4.2.2.2. LAN Config



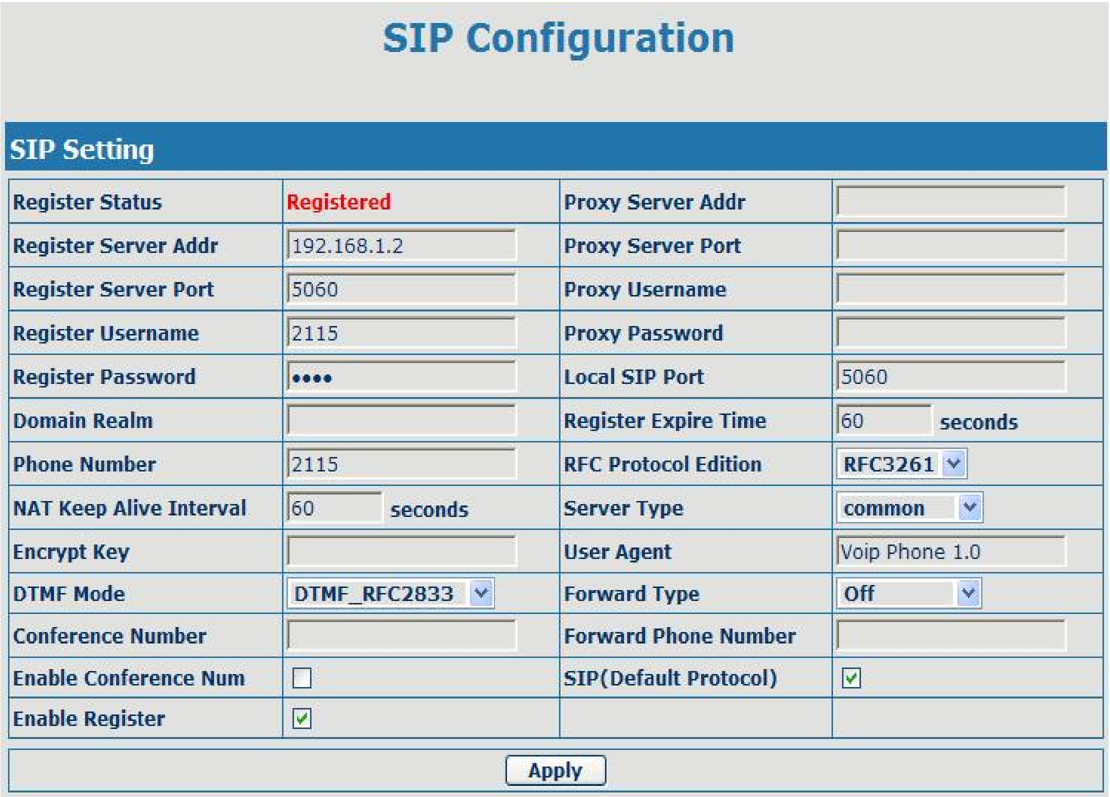
LAN Configuration	
LAN IP	specify LAN static IP
Netmask	specify LAN Netmask

DHCP Service	Select the DHCP server of LAN port or not. After you modify the LAN IP address, gateway 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.
Notice: If you choose the bridge mode, the LAN configuration will be disabled.	

4.2.3. VoIP

4.2.3.1. SIP Config

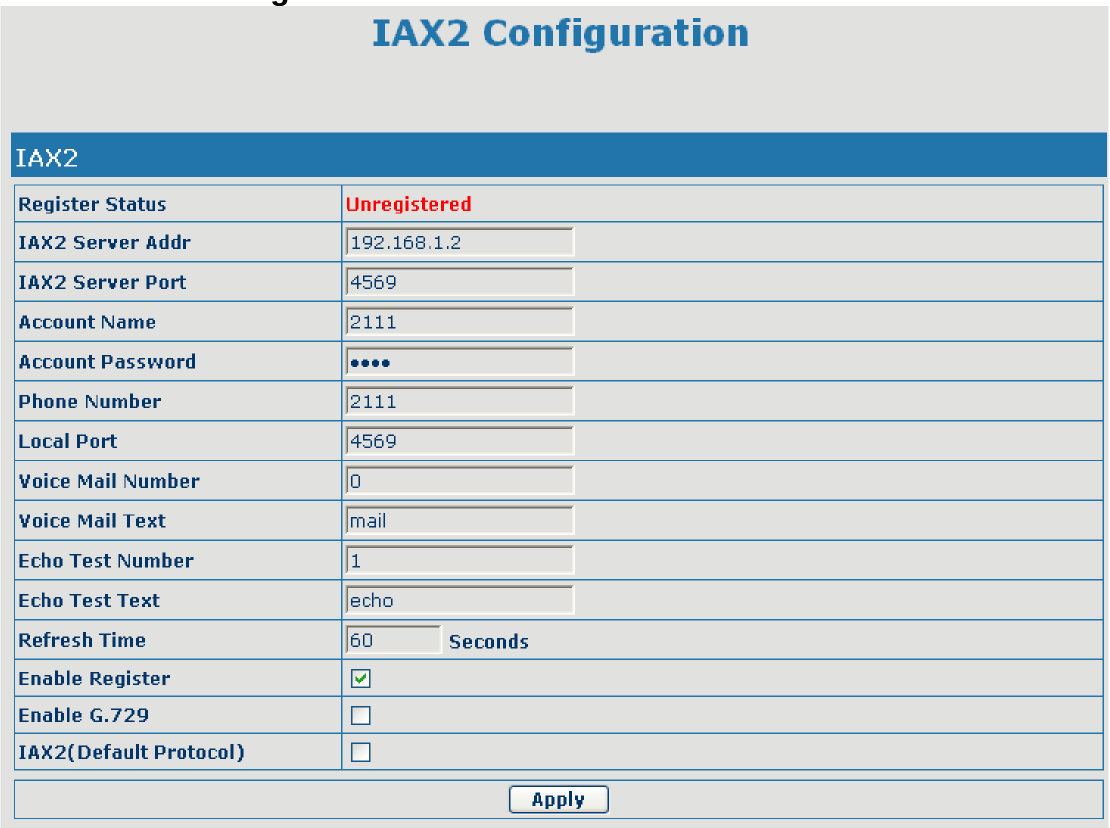
Set your SIP server in the following interface



SIP Config	
Filed name	**Illumination**
Register Status	Shows if the phone has been registered the SIP server or not.
Register Server Addr	Input your SIP server address.
Register Server Port	Set your SIP server port.
Register Username	Input your SIP register account name.
Register Password	Input your SIP register password.
Proxy Server Addr	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.
Local SIP Port	Set your Local SIP port, the default is 5060.
Domain Realm	Set the sip domain if needed, otherwise this VoIP Phone will use the proxy server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
Phone Number	Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured.
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.
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.
DTMF Mode	Select DTMF sending mode, there are three modes: l DTMF_RELAY l DTMF_RFC2833 l DTMF_SIP_INFO Different VoIP Service providers may provide different modes.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
Encrypt Key	Set the key for encryption
User Agent	Set the user agent if have, the default is VoIP Phone 1.0
Forward Type	Select call forward mode, the default is Off l Off：Close down calling forward l Busy：If the phone is busy, incoming calls will be forwarded to the appointed phone. l No answer： If there is no answer, incoming calls will be forwarded to the appointed phone. l Always：Incoming calls will be forwarded to the appoint phone directly, and the phone will not ring.
Conference Number	Set the special phone number of 3 way calling.
Forward Phone Number	Appoint your forward phone number.
Enable Conference Num	Enable/Disable the function which uses SIP server to realize 3 way talking, not realized by our system.
Enable Register	Start to register or not by selecting it or not.
SIP(Default Protocol)	Use SIP protocol as default dial protocol

4.2.3.2. IAX2 Config

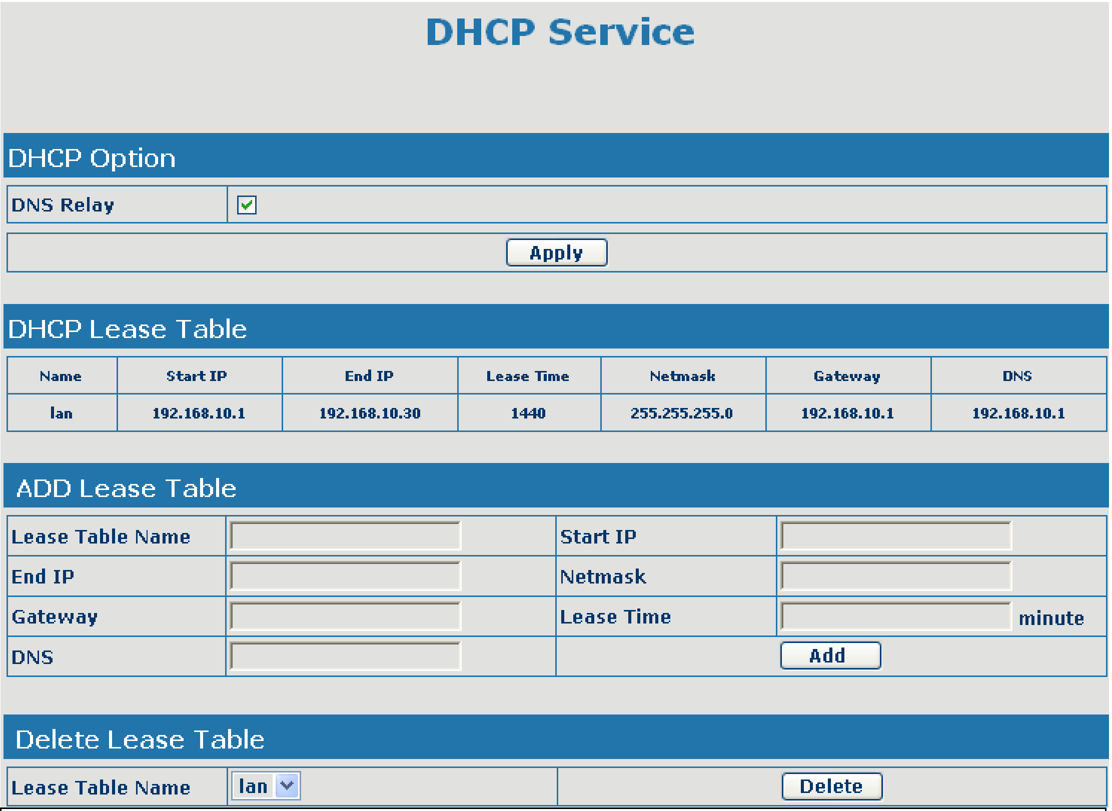


IAX2 Config	
Register Status	Shows if the phone has been registered the IAX2 server or not.
IAX2 Server Addr	Input your IAX2 server address.
IAX2 Server Port	Set your IAX2 server port, the default is 4569.
Account Name	Input your IAX2 register account name.
Account Password	Input your IAX2 register password.
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
IAX2	Select it to make all outgoing calls through the IAX2 server by

(Default Protocol)	default. If you also need make a call through SIP server, you can make prefix in dial peer setting to realize SIP calling. Note: any incoming call can be from both IAX2 and SIP.;

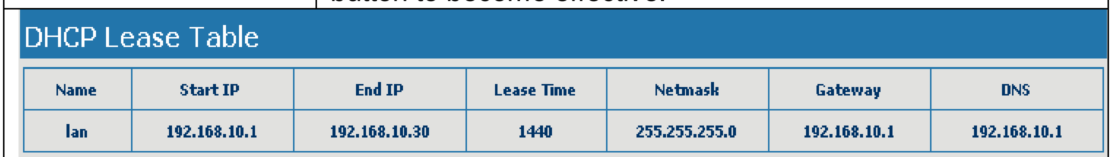
4.2.4. Advance

4.2.4.1. DHCP Service



DHCP Service

DNS Relay	Select DNS Relay, the default is enable. Click the Apply button to become effective.

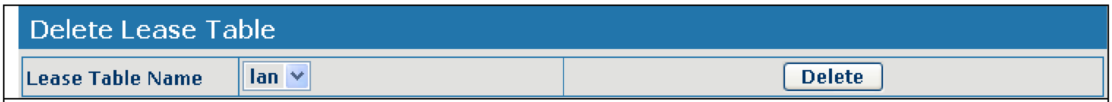


Name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS
lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1

Shows the DHCP Lease Table， the unit of Lease time is Minute.

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



Select name of lease table, click the **Delete** button will delete the selected lease table from DHCP lease table.

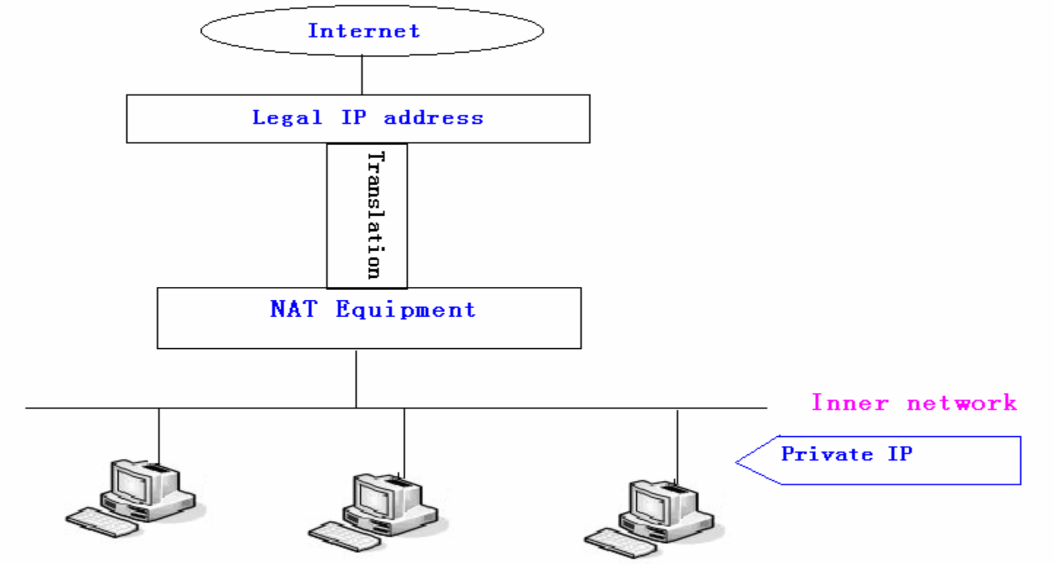
Notice:

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 modifies the DHCP lease table, you need save the configuration and reboot.

4.2.4.2. NAT Configuration

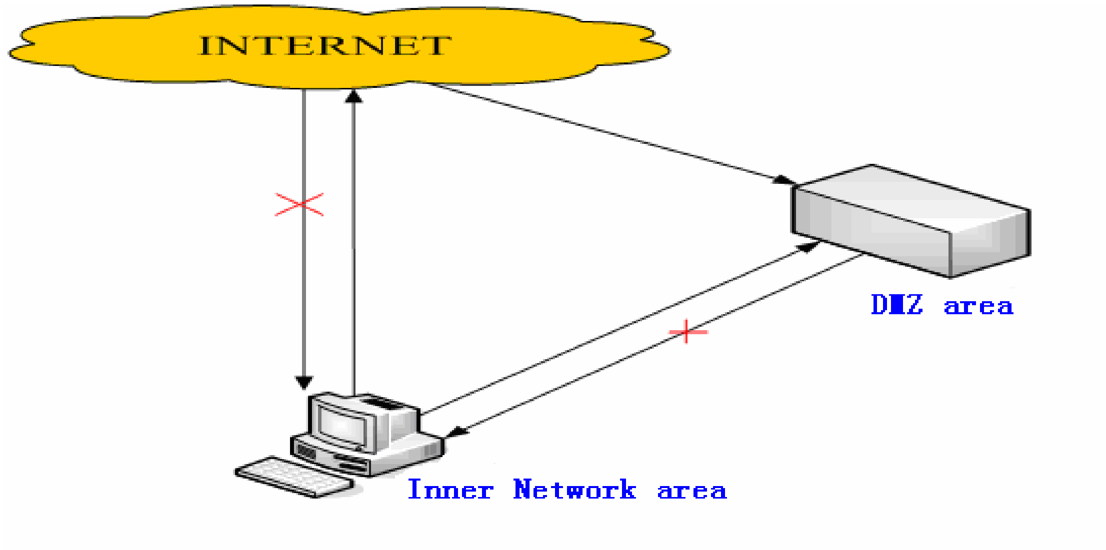
NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.

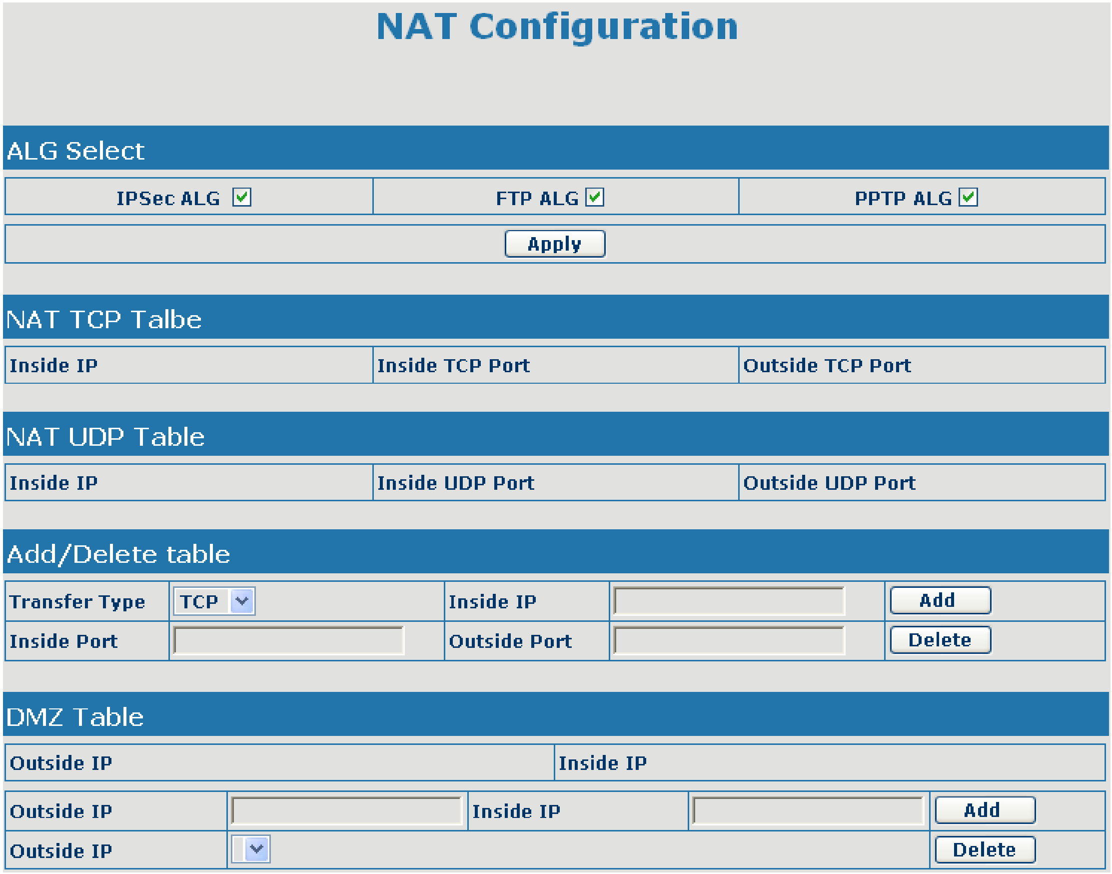


DMZ config：

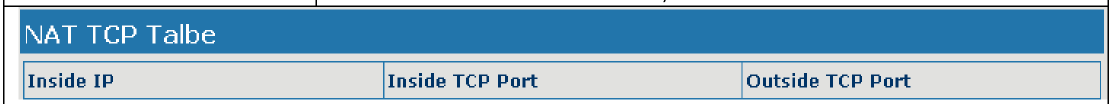
In order to make some intranet equipments support better service for extranet, and make internal network security more effectively, these equipments open to extranet need be separated from the other equipments not open to extranet by the corresponding isolation method according to different demands. 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



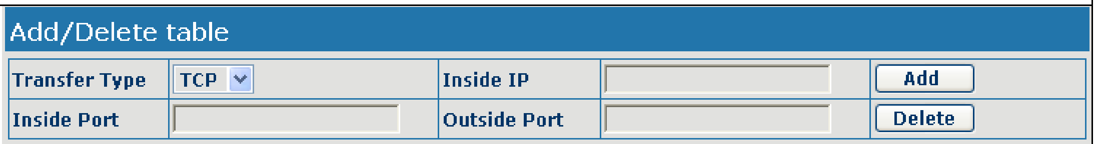


NAT Configuration	
IPSec ALG	It is an encryption technology. Select it to enable IPSec ALG, the default is enable
FTP ALG	FTP is a service of connection layer which can transform intranet IP into extranet IP when intranet IP is sending out packet. Select it to enable FTP ALG, the default is enable
PPTP ALG	Select it enable PPTP ALG, the default is enable



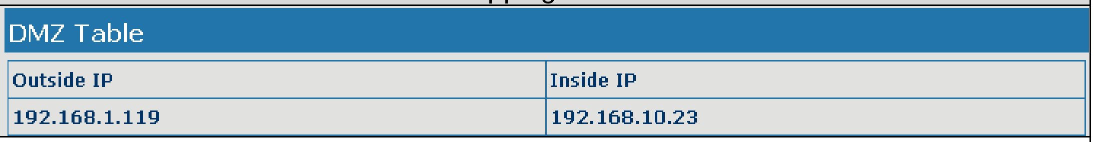
Shows the NAT TCP mapping table

Shows the NAT UDP mapping table



Transfer Type	Select the NAT mapping protocol style, TCP or UDP
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
Outside Port	Set the WAN 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.

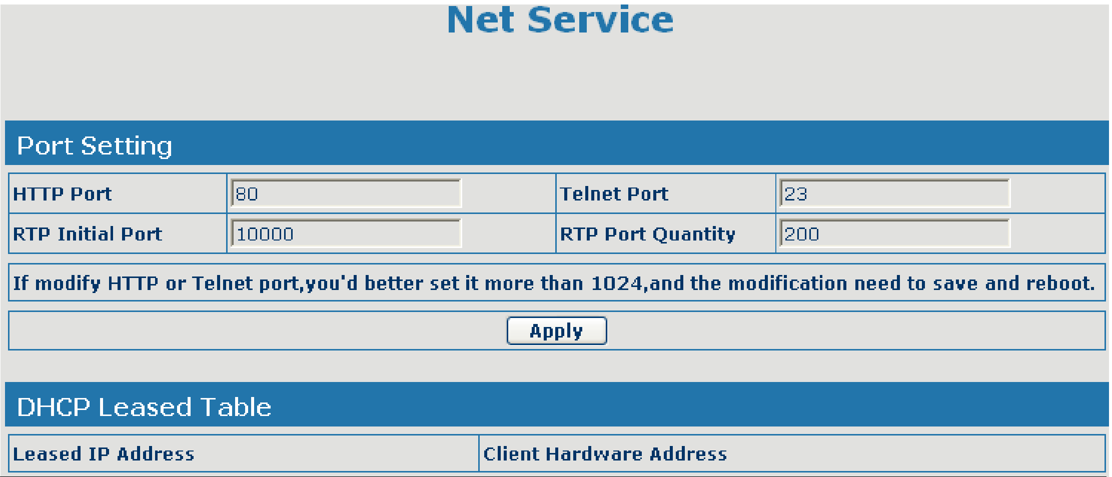


Shows the outside WAN port IP address and the inside LAN port IP address.

Outside IP	Set the outside Wan port IP address of DMZ.
Inside IP	Set the inside LAN pot IP address of DMZ

Click the **Add** button to add new table; click the **Delete** button to delete the selected mapping table.

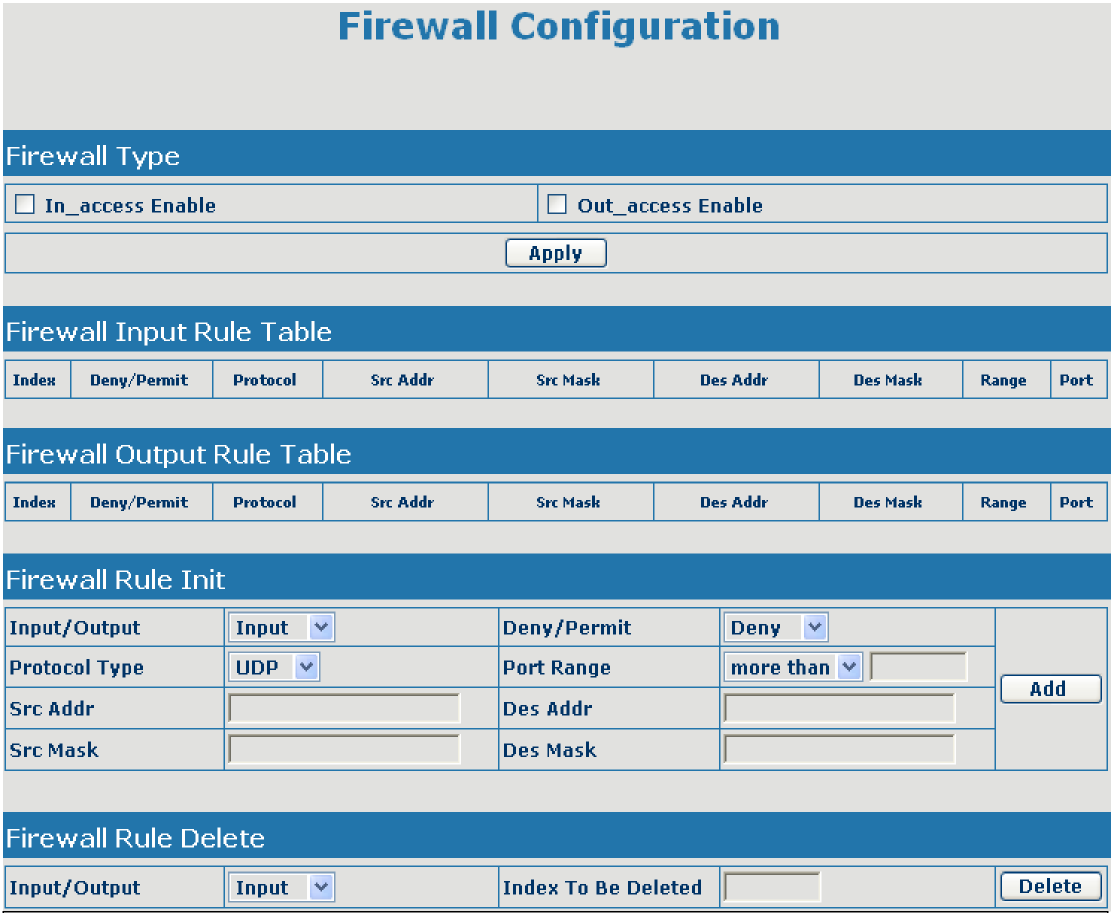
4.2.4.3. Net Service



Net Service	
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,

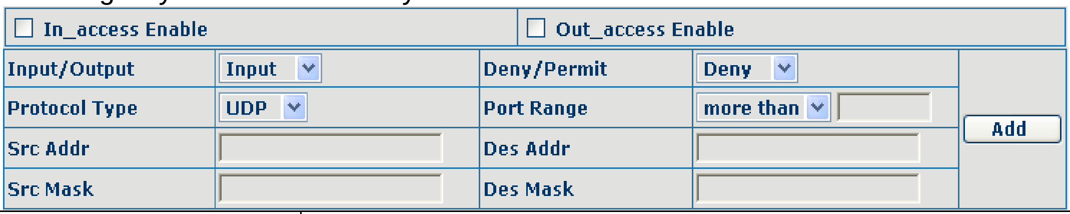
	the accessing address is telnet 192.168.1.70 8023
RTP Initial Port	Set the RTP Initial Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.
DHCP Leased Table	IP-MAC mapping table. If the LAN port of the phone connects to a device, this table will show the IP and MAC address of this device.
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.	

4.2.4.4. Firewall Config



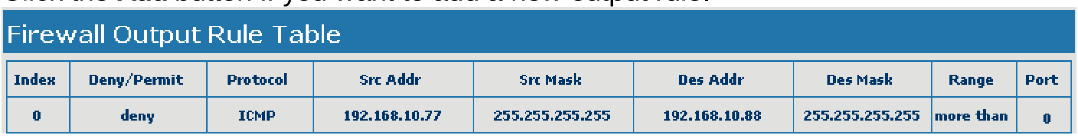
Firewall Configuration
In this web interface, you can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule). Firewall support two type of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



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 rang
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.

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



Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
0	deny	ICMP	192.168.10.77	255.255.255.255	192.168.10.88	255.255.255.255	more than	0

Then enable out_access, and click the Apply button.
So when devices connect to LAN execute to ping 192.168.10.88, system will deny the request to send icmp request to 192.168.10.88 for the out_access rule. But if devices ping other devices which network ID is 192.168.10.0, it will be normal.

Click the **Delete** button to delete the selected rule.

4.2.4.5. QoS Config

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.

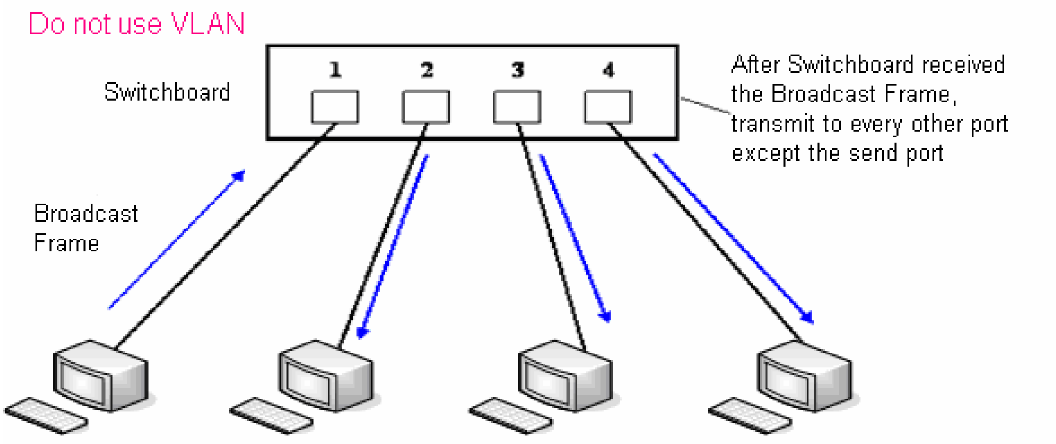
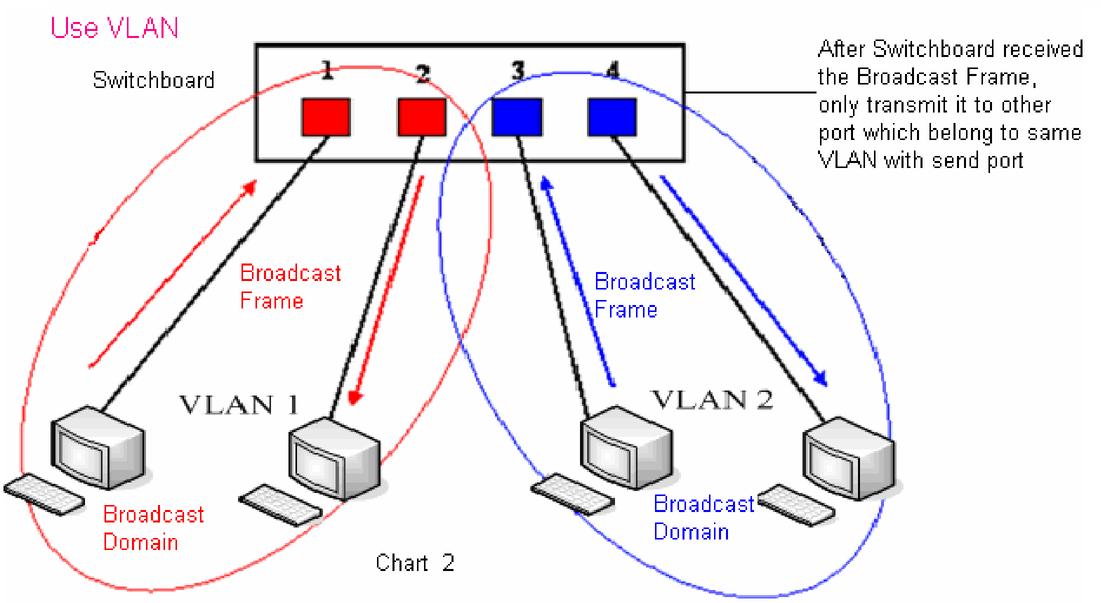
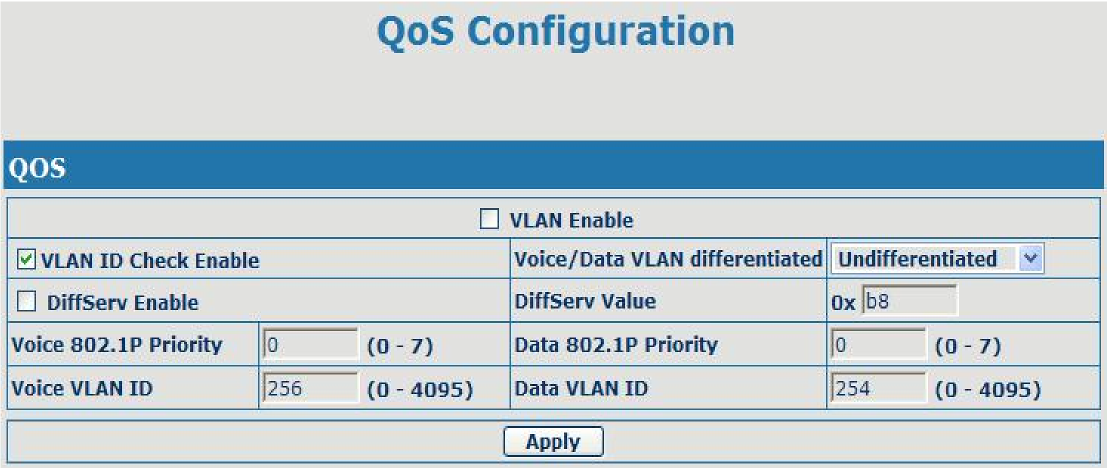


Chart 1



In chart 1, there is a layer 2 switch without setting VLAN. Any broadcast frame will be transmitted to the other ports except the send port. For example, broadcast information is sent out from port 1 then transmitted to port 2, 3 and 4.
In chart 2, red and blue 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, 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 divide the broadcast domain via restricting the range of broadcast frame transmission.
Note: chart 2 use red and blue to identify the different VLAN, but in practice, VLAN uses different VLAN IDs to identify.



QoS Configuration	
VLAN Enable	Before select it to enable VLAN, you need enable Bridge mode in Lan config
VLAN ID Check Enable	Enable VLAN ID check by selecting it. After enable VLAN ID check, if VLAN ID of a packet is not the same with the phone's or a packet do not have VLAN ID, the packet 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. 0xb8, which is the highest priority
Voice 802.1P Priority	Specify 802.1P Priority of voice/signal data packet.
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 patcket.
Voice VLAN ID	Set VLAN ID of voice/signal data packet
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 patcket.

Notice:

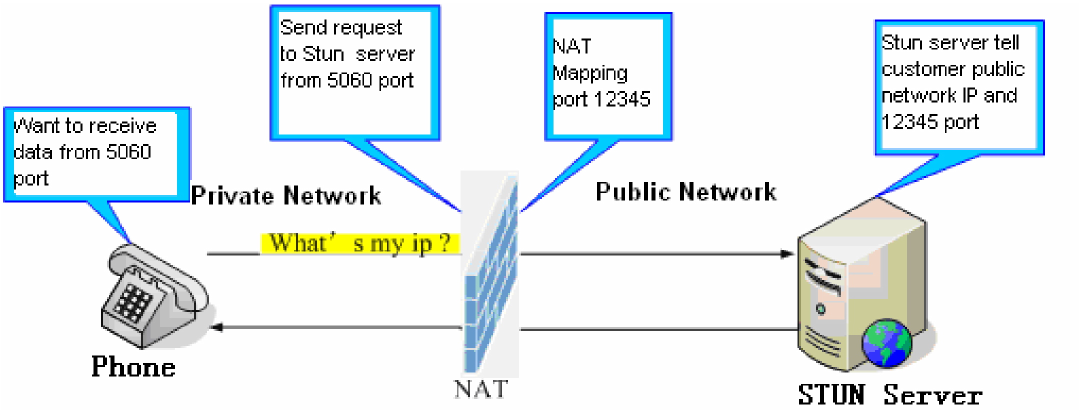
1. If you don't enable diffServ, phone will not set voice/data packets with different VLAN ID even if select tag differentitted.
2. If you disable VLAN, system will not add VLAN ID to all packets, regardless of Data/Voice Diffserv. If you enable diffserv, system will just set the diffserv value to voice/signal packets.
3. VLAN ID Check Enable is on by default. It means system will check VLAN ID strictly; if packets' VLAN ID are not same as value system using or has no VLAN, packets will be lost; if it is off, system might accept packets which VLAN ID are not same as value system using or has no VLAN.

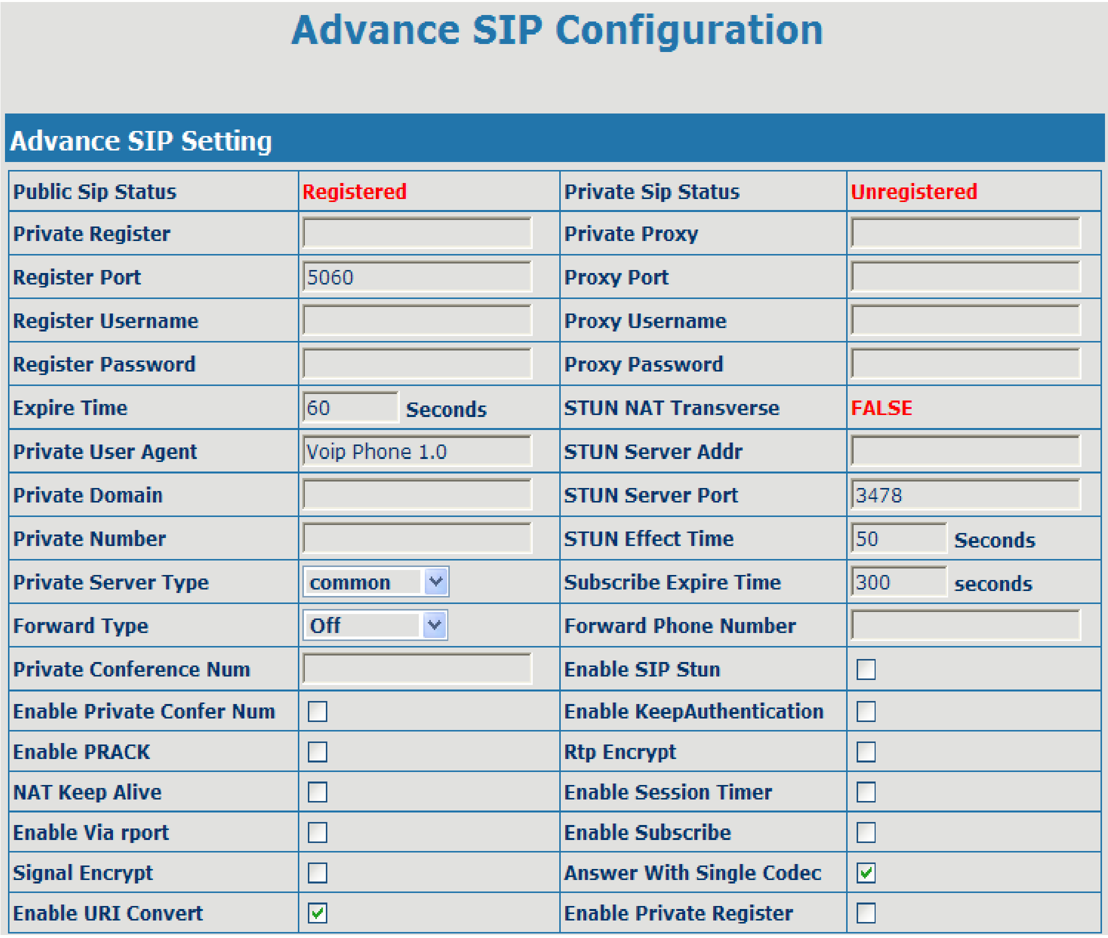
4.2.4.6. Advance SIP Configuration

In this web page, you can config SIP STUN, Private Server and so on.

STUN:

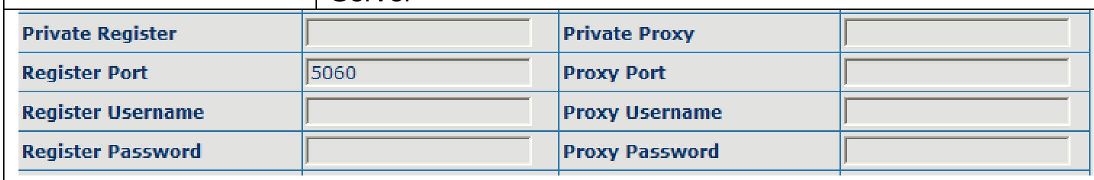
By STUN server, a phone in private network could know the type of NAT and the NAT mapping IP and port of SIP. The phone might register itself to SIP server with global IP and port to realize the device both calling and being called in private network.





Advance SIP Configuration

Advance SIP Setting			
Public Sip Status	Registered	Private Sip Status	Unregistered
Private Register		Private Proxy	
Register Port	5060	Proxy Port	
Register Username		Proxy Username	
Register Password		Proxy Password	
Expire Time	60 Seconds	STUN NAT Transverse	FALSE
Private User Agent	Voip Phone 1.0	STUN Server Addr	
Private Domain		STUN Server Port	3478
Private Number		STUN Effect Time	50 Seconds
Private Server Type	common	Subscribe Expire Time	300 seconds
Forward Type	Off	Forward Phone Number	
Private Conference Num		Enable SIP Stun	☐
Enable Private Confer Num	☐	Enable KeepAuthentication	☐
Enable PRACK	☐	Rtp Encrypt	☐
NAT Keep Alive	☐	Enable Session Timer	☐
Enable Via rport	☐	Enable Subscribe	☐
Signal Encrypt	☐	Answer With Single Codec	☑
Enable URI Convert	☑	Enable Private Register	☐

Advance SIP Configuration	
Public Sip Status	shows that the phone registered or unregistered Public Server
Private Sip Status	shows that the phone registered or unregistered Private Server
 Private Register: ; Register Port: 5060 ; Register Username: ; Register Password: ; Private Proxy: ; Proxy Port: ; Proxy Username: ; Proxy Password: Set Private Server parameters:	
Expire Time	Set the expired time for registering the Private Server.

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.
Subscribe Expire Time	Set the interval time of sending SUBCRIBE message, like register expire time.
Enable SIP STUN	Enable/Disable SIP STUN.
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.
Enable PRACK	Enable/Disable PRACK.
NAT Keep Alive	Enable/Disable keep 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.
Signal Encrypt	Enable/Disable Signal Encrypt.
Enable Private Register	Enable/Disable Private Server Register.
Rtp Encrypt	Enable/Disable Rtp Encrypt.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.
Enable Subscribe	Enable/Disable sending SUBSCRIBE messages to subscribe other phones' status or voice mail after being registered.
Answer With Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Enable URI Convert	Enable/Disable the function when phone sends SIP request, using %23 instead of "#" character in SIP URI.
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.	

4.2.4.7. Digital Map Configuration

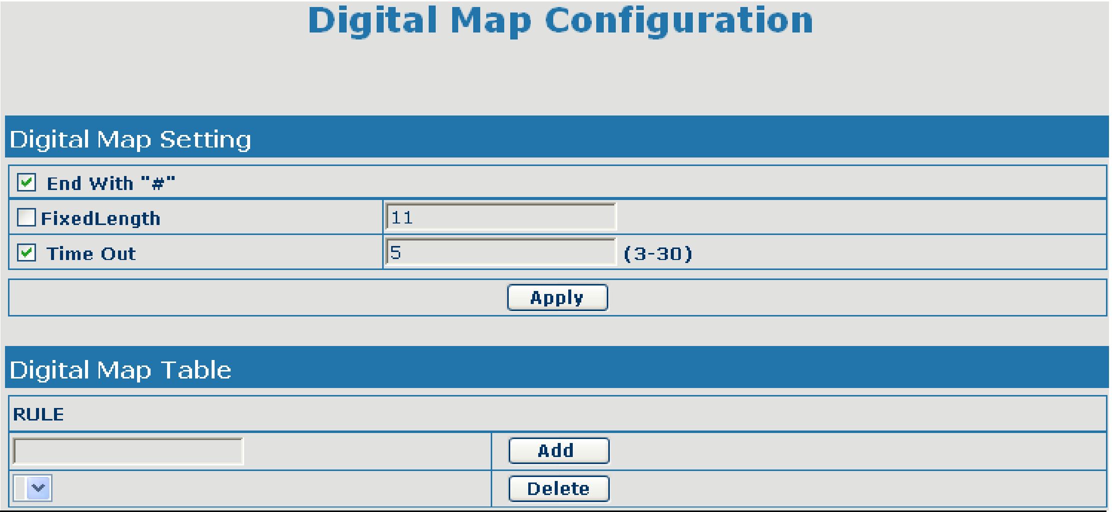
This system supports 4 dial modes:

1). End with "#": dial your desired number, and then press #.

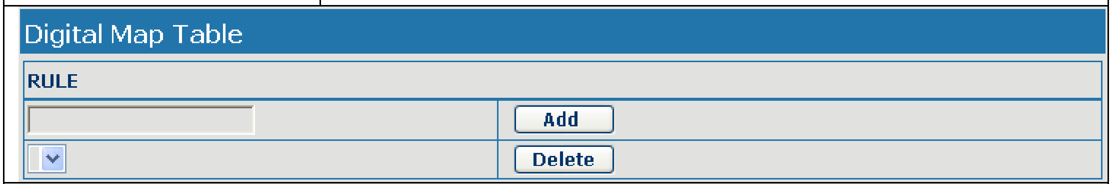
2). Fixed Length: the phone will intersect the number according to your specified length.

3). Time Out: After you stop dialing and waiting time out, system will send the number collected.

4). 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.



Digital Map Configuration	
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.



Below is 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.

RULE
"[1-8]xxx"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

Cause extensions 1000-8999 to be dialed immediately
Cause 8 digit numbers started with 9 to be dialed immediately
Cause 911 to be dialed immediately after it is entered.
Cause 99 to be dialed after 4 seconds.
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.

4.2.4.8. Call Service

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

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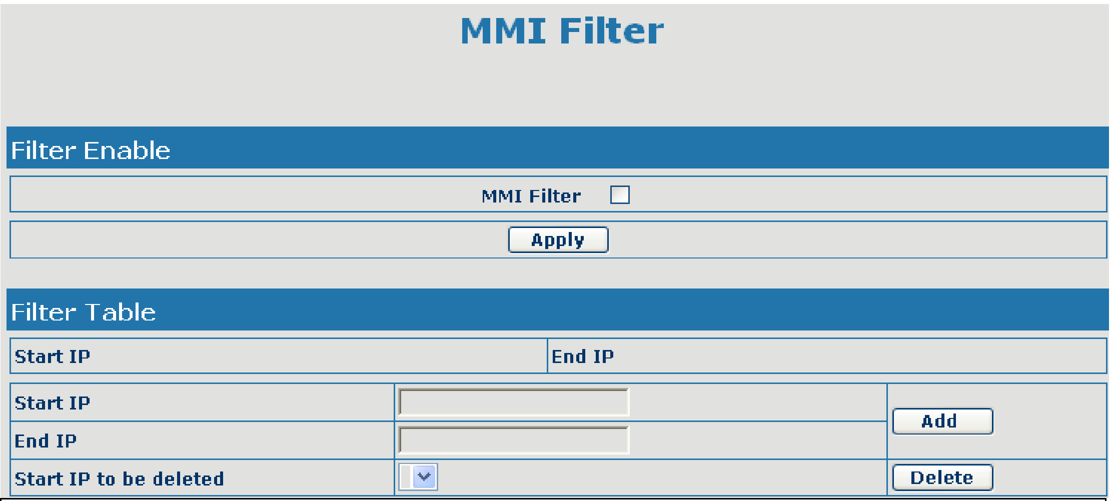
<table>
<tr><th colspan="2">Call Service</th></tr>
<tr><td>Hotline</td><td>Specify Hotline number. If you set the number, you can not dial any other numbers.</td></tr>
<tr><td>No Disturb</td><td>Select NO Disturb, the phone will reject any incoming call, the callers will be reminded by unavailable, but any outgoing call from the phone will work well.</td></tr>
<tr><td>Ban Outgoing</td><td>If you select Ban Outgoing to enable it, and you can not dial out any number.</td></tr>
<tr><td>Enable Call Transfer</td><td>Enable Call Transfer by selecting it.</td></tr>
<tr><td colspan="2">Call transfer has several ways to realize:
1. When A talks to B, B may press the HOLD key and dial to C phone number. After B talks to C (or B hear alert from C), B presses the TRANSFER key; B could hang up, and A will get through to C.
2. When A talks to B, there is C call incoming to B; B may press the HOLD key to hold A, and talks to C, pressing the TRANSFER key, so A will get through to C.
3. When A talks to B, B presses the TRANSFER key, dial C phone number and # key, B could hang up and A will get through to C.

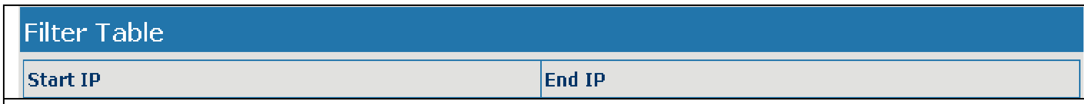
1 and 2 are attended transfer; 3 is blind transfer.
Notice to VoIP Phone Carrier: Your VoIP phone server need support FRC3515, or else transferring can not work.</td></tr>
<tr><td>Enable Call Waiting</td><td>Enable Call Waiting by selecting it.</td></tr>
<tr><td>Enable Three Way Call</td><td>Enable Three Way Call by selecting it.</td></tr>
<tr><td colspan="2">Note: If the party who launched the three way call hangs up, the other two parties can not get through; while if the party who did not launch the three way call hangs up, the other two parties can get through.</td></tr>
<tr><td>Accept Any Call</td><td>If select it, the phone will accept the call even if the called number is not belong to the phone.</td></tr>
</table>

Auto Answer	If select it, the phone will auto answer when there is an incoming call.
Enable Voice Record	If select Enable Voice Record, when no answer time of an incoming call is beyond its set value, the phone will remind the caller to record.
User-Defined Voice	Select it or not to Enable or disable User Defined Voice
Incoming Record Playing	Select it or not to Enable or disable Incoming Record Playing
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.
Use Record Server	Select it or not to Enable or disable Use Record Server.
Remote Record No	Set Remote Record number. Via dialing this number, you can listen all voice records in your VoIP server.
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.
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, then you can not dial out any phone number whose prefix is 001.
Notice: Black List and Limit List can record at most10 items respectively.	

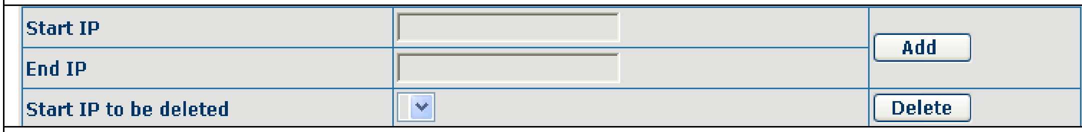
4.2.4.9. MMI Filter



MMI Filter	
User could make some devices own IPs, which are pre-specified, access to phone to config and manage phone.	
MMI Filter	Select it or not to enable or disable MMI Filter. Click **Apply** to make it effective.



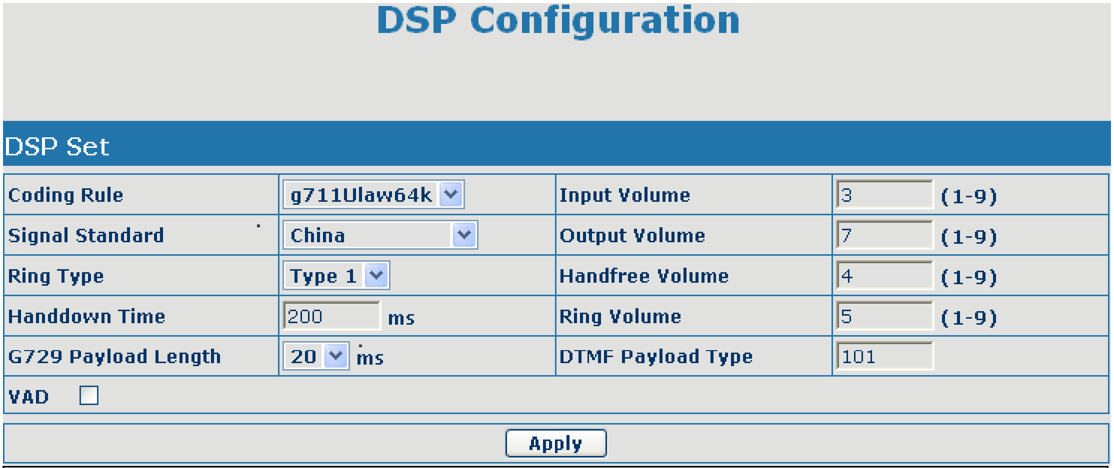
MMI Fileter IPs Table list:



Add or delete the IP address segments that access to 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.

4.2.4.10. DSP Config

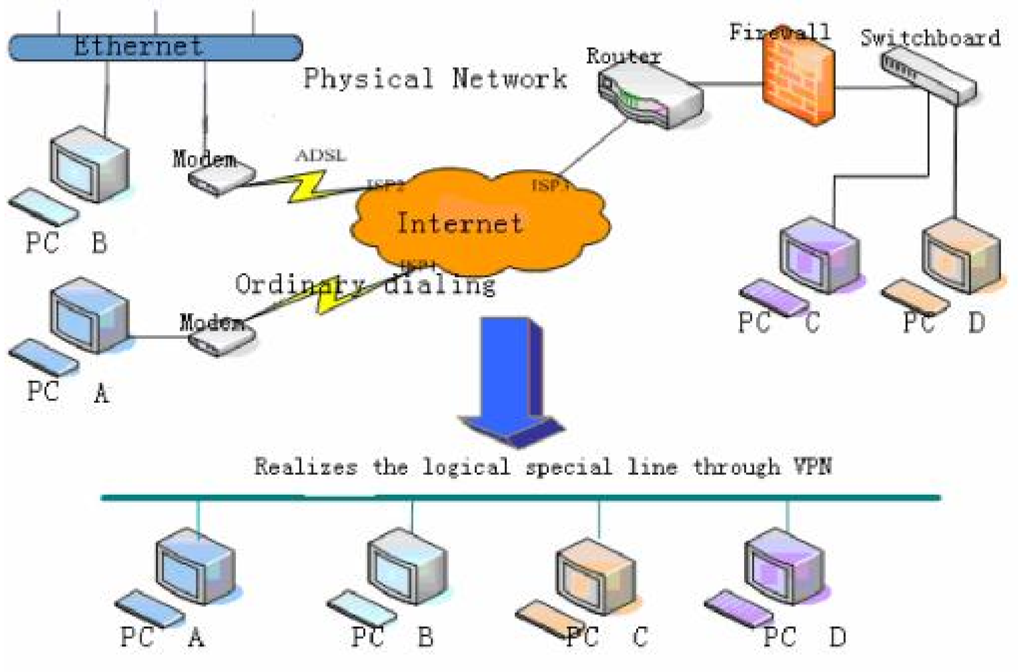
In this page, you can configure voice codec, input/output volume and so on.

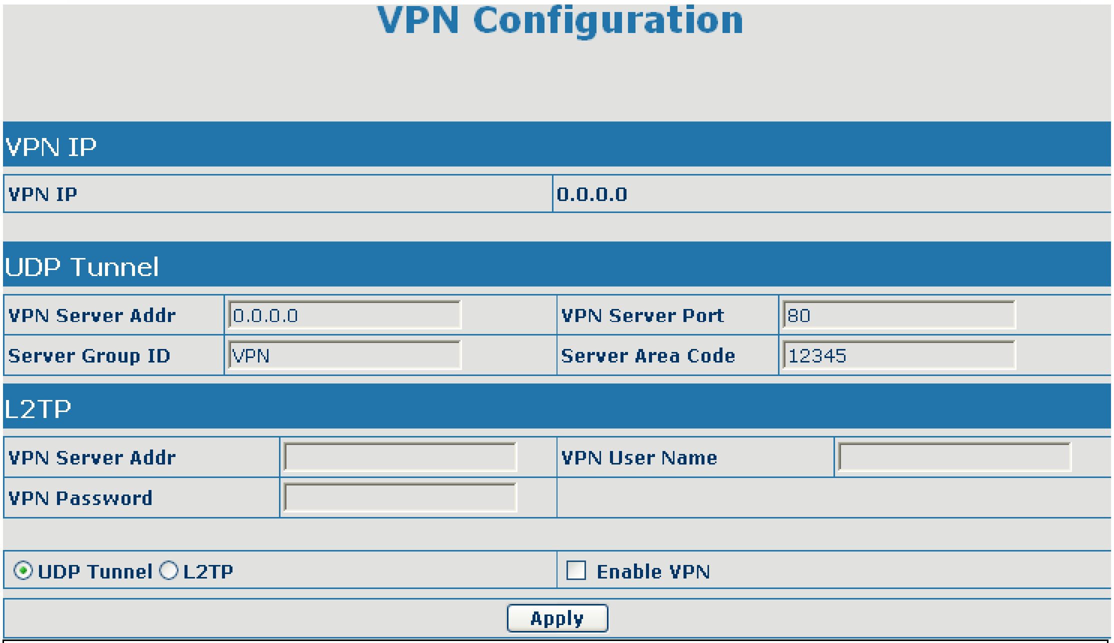


DSP Configuration	
Coding Rule	Select DSP voice coding rule.
Input Volume	Specify Input(MIC) Volume grade.
Signal Standard	Select Signal Standard.
Output Volume	Specify Output(receiver) Volume grade.
Ring Type	Select Ring Type
Handfree Volume	Specify Handfree Volume grade
Handdown Time	Specify the least reflection time of Handdown, the default value is 200ms.
Ring Volume	Specify Ring Volume grade
G729 Payload Length	Set G729 Payload Length
DTMF Payload Type	Set DTMF Payload Type
VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.

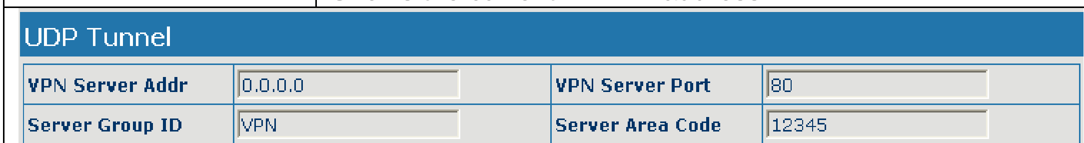
4.2.4.11. VPN Config

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.

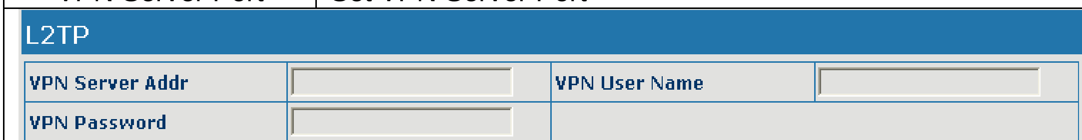




VPN Configuration	
VPN IP	Shows the current VPN IP address



VPN Server Addr	Set VPN Server IP Address
VPN Server Port	Set VPN Server Port



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

◉ UDP Tunnel ◯ L2TP	
Select UDP Tunnel (VPN Tunnel) or VPN L2TP. You can choose only one for current state. After you select it, you'd better save conifguration and reboot your phone.	
Enable VPN	Select it or not to enable or disable VPN;

4.2.5. Dial-Peer Setting

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.

Number	Call Mode	Destination	Port	Alias	Suffix	Del Length
156	sip	192.168.1.119	5060	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 9 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 962213123 to realize your long distance call after you make this setting.

Number	Call Mode	Destination	Port	Alias	Suffix	Del Length
9T	sip	0.0.0.0	5060	rep:010	no suffix	1

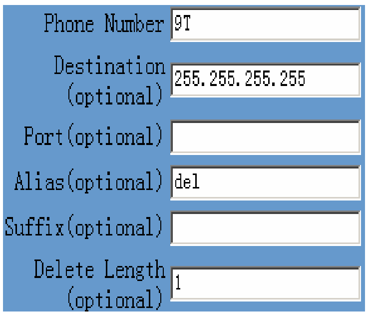
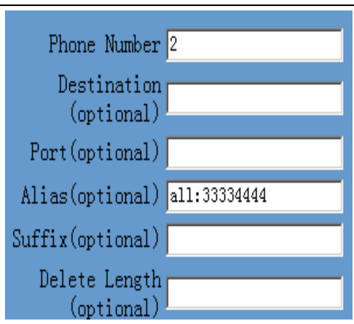
The phone supports two SIP lines and one IAX2 line. After you make a configuration according to this dial rule, you can realize dialing out via different lines without switch in web interface.

Number	Call Mode	Destination	Port	Alias	Suffix	Del Length
9T	sip	0.0.0.0	5060	del	no suffix	1
8T	sip	255.255.255.255	5060	del	no suffix	1
156	sip	192.168.1.119	5060	no alias	no suffix	0

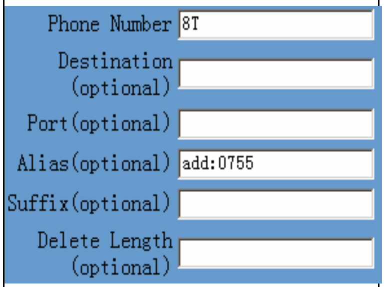
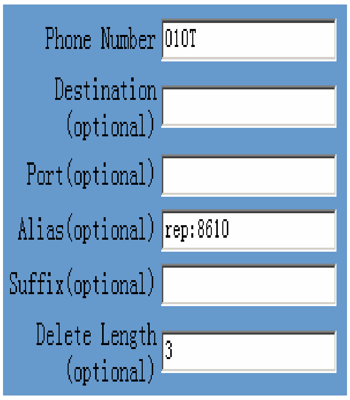
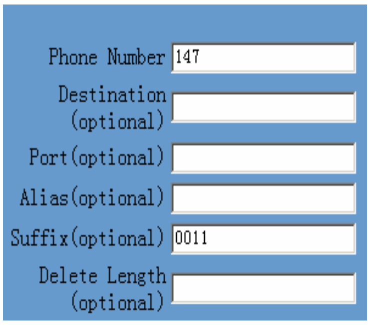
Dial-Peer	
Phone number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, you need input your desired phone number in this blank, and then you need dial the phone number to realize calling to what the phone number is mapped. In the prefix 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
Call Mode	Select SIP or IAX2 protocol
Destination	Set Destination address / phone number. This is optional config item. If you want to set peer to peer call, please input destination IP address or domain name. If you want to use

<table>
<tr><td></td><td>this dial rule in SIP2 line, you need input 255.255.255.255 or 0.0.0.2 in it.</td></tr>
<tr><td>Port</td><td>Set the Signal port, the default is 5060 for SIP</td></tr>
<tr><td>Alias</td><td>Set alias. This is optional config item. If you don’t set Alias, it will show no alias.</td></tr>
<tr><td colspan="2">Please input different alias, there are four types of aliases.
1) add: xxx, it means that you need dial xxx in front of phone number, which will reduce dialing number length.
2) all: xxx, it means that xxx will replace some phone number.
3) del: It means that phone will delete the number with length appointed.
4) 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.</td></tr>
<tr><td>Suffix</td><td>Set suffix, this is optional config item. It will show no suffix if you don’t set it.</td></tr>
<tr><td>Delete Length</td><td>Set delete length. This is optional config 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.</td></tr>
<tr><td colspan="2">Introduction of how to set up dial-peer to implement switch between multi- SIP lines
<table>
<tr><th>Number</th><th>Call Mode</th><th>Destination</th><th>Port</th><th>Alias</th><th>Suffix</th><th>Del Length</th></tr>
<tr><td>9T</td><td>sip</td><td>0.0.0.0</td><td>5060</td><td>del</td><td>no suffix</td><td>1</td></tr>
<tr><td>8T</td><td>sip</td><td>255.255.255.255</td><td>5060</td><td>del</td><td>no suffix</td><td>1</td></tr>
</table>
9T mapping: If you have registered a Public SIP server and set dial-peer according to the above table，all calls will be sent via public server when you press the numeric key “9” in front of dialing destination phone numbers.
8T mapping: If you have registered a Private SIP server and set dial-peer according to the above table，all calls will be sent via private server when you press the numeric key “8” in front of dialing destination phone numbers.</td></tr>
</table>

Examples of different alias application

<table>
<tr><td>
</td><td>You need set phone number, Destination, Alias and Delete Length.
Phone number is XXXT, Destination is 255.255.255.255 and Alias is del.
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.</td><td>If you dial “93333”, the SIP2 server will receive “3333”</td></tr>
<tr><td>
</td><td>You need set Phone number and Alias. Phone number is XXX and Alias is all:xxx

This setting will realize speed dial or memory key functionality.</td><td>When you dial “2”, the SIP1 server will receive 33334444</td></tr>
</table>

	You need set Phone Number and Alias. Phone number is XXXT and Alias is add:xxx 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”
	You need set Phone Number and suffix. Phone number is XXX and Suffix is xxx. 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”

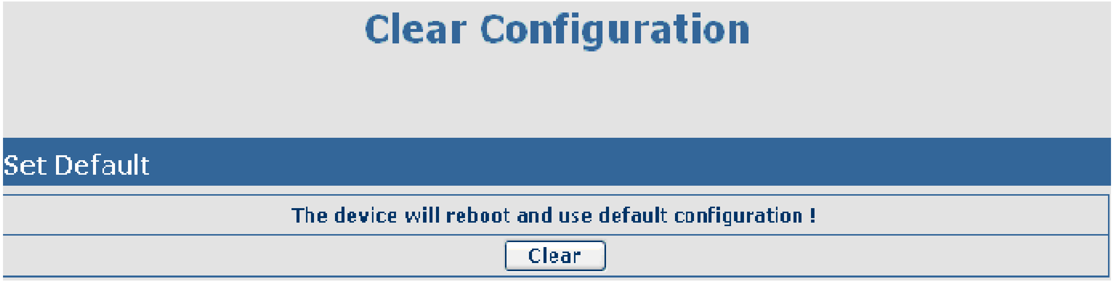
4.2.6. Config Manage

4.2.6.1. Save Config

In this web page, you can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately. .

Notice: If you don’t make a save, some changes of configurations will be discarded after the phone is reset.

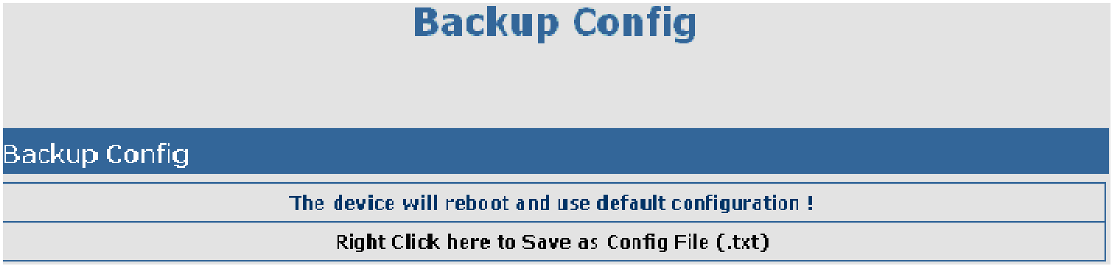
4.2.6.2. Clear Config



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 (SIP, advance SIP and IAX2) and version number.

4.2.6.3. Backup Config

Right click on “Right click here…” and select “Save Target As….” then you will save the config file in .txt format



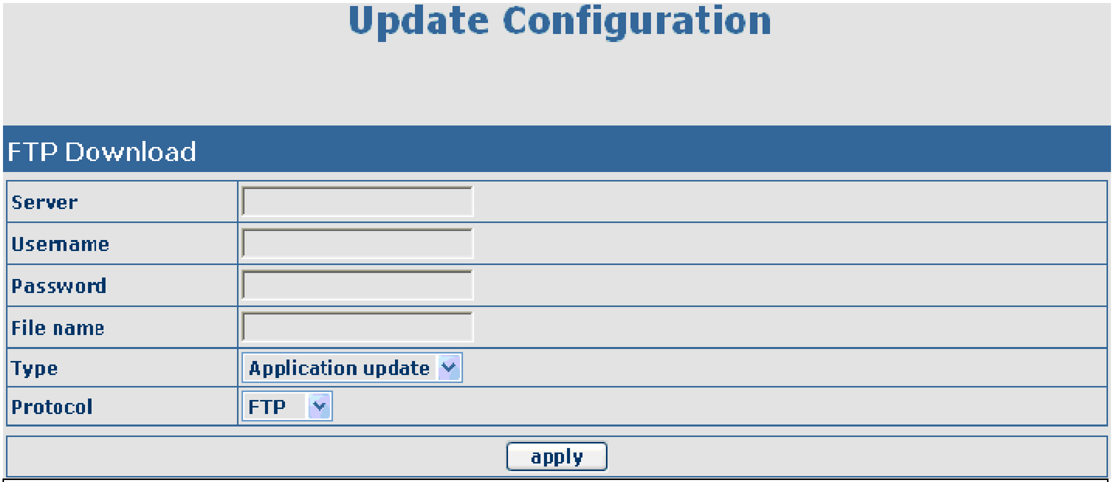
4.2.7. Update

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

4.2.7.1. Web Update

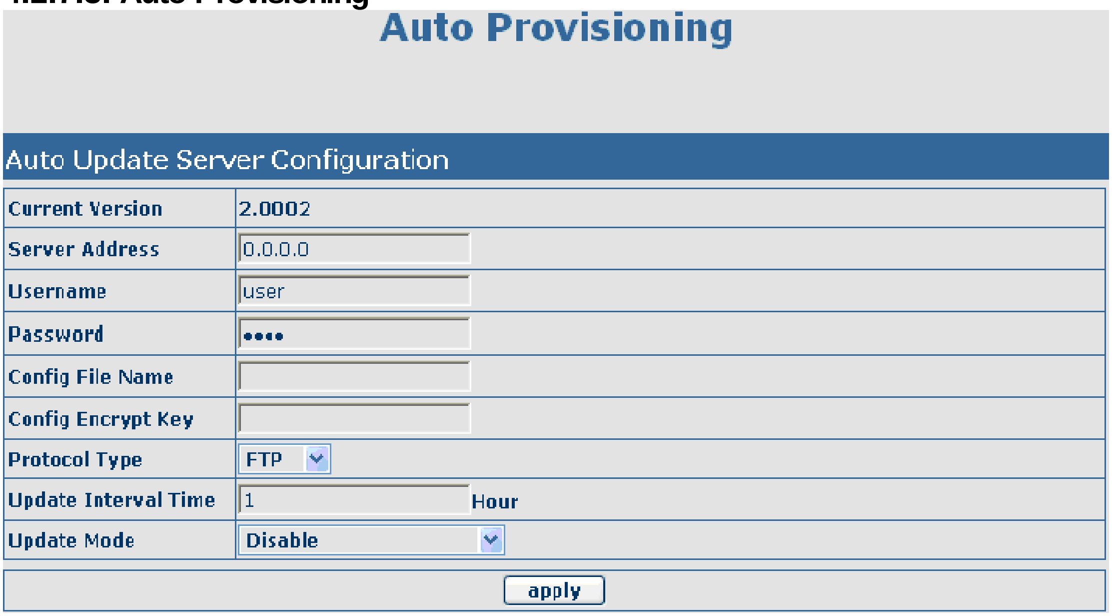
Click the browse button, find out the config file saved before or provided by manufacturer, download it to IP Phone directly, press “Update” to save. You can also update downloaded update file, logo picture, ring, mmiset file by web.

4.2.7.2. FTP/TFTP Update



FTP/TFTP Update	
Server	Set the FTP/TFTP/HTTP 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 config file. The default name is the MAC of the phone, such as 000102030405.
Notice: You can modify the exported config file. And you can also download config file which includes several modules that need to be imported. For example, you can download a config file just keep with SIP module. After reboot, other modules of system still use previous setting and are not lost.	
Type	Action type that system want to execute： 1. Application update: download system update file 2. Config file export: Upload the config file to FTP/TFTP server, name and save it. 3. Config fie import: Download the config file to phone from FTP/TFTP/HTTP server. The configuration will be effective after the phone is reset.
Protocol	Select FTP/TFTP/HTTP server

4.2.7.3. Auto Provisioning

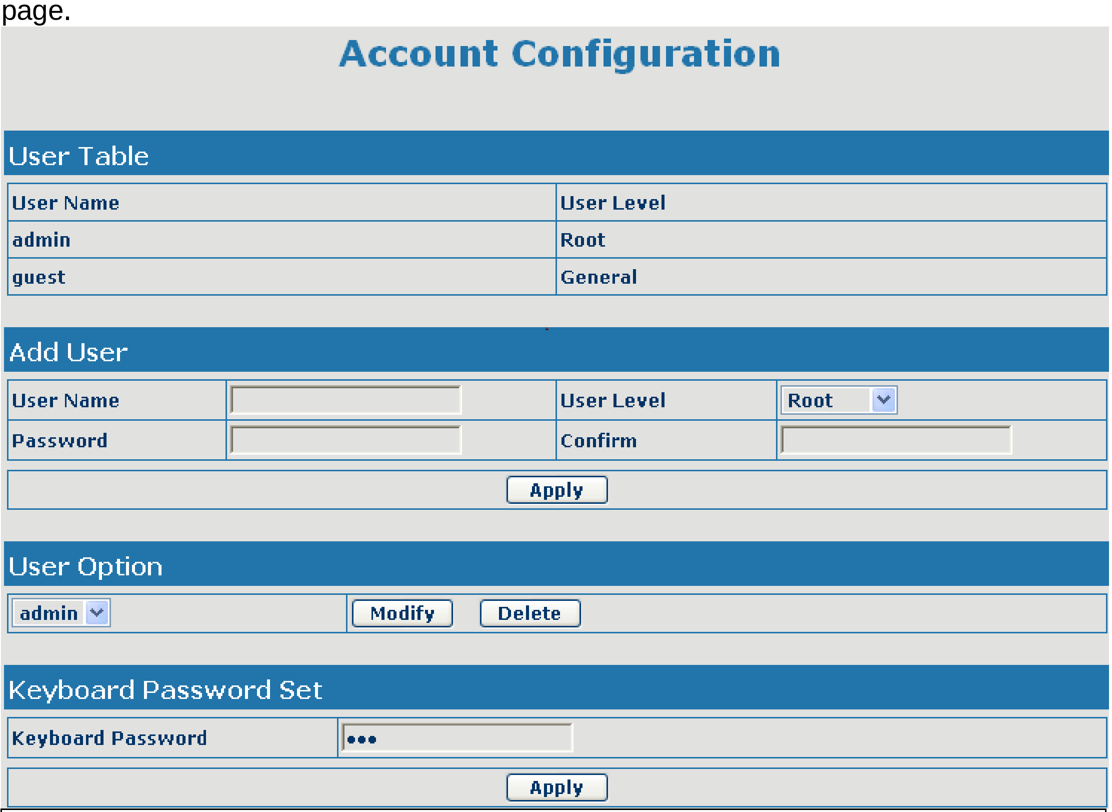


Auto Provisioning	
Current Version	show the current config file's version.
Server Address	Set FTP/TFTP/HTTP server IP address for auto update.
Username	Set FTP server Username. System will use anonymous if username keep blank.
Password	Set FTP server Password.
Config File Name	Set configuration file's 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.

4.2.8. System Manage

4.2.8.1. Account Config

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



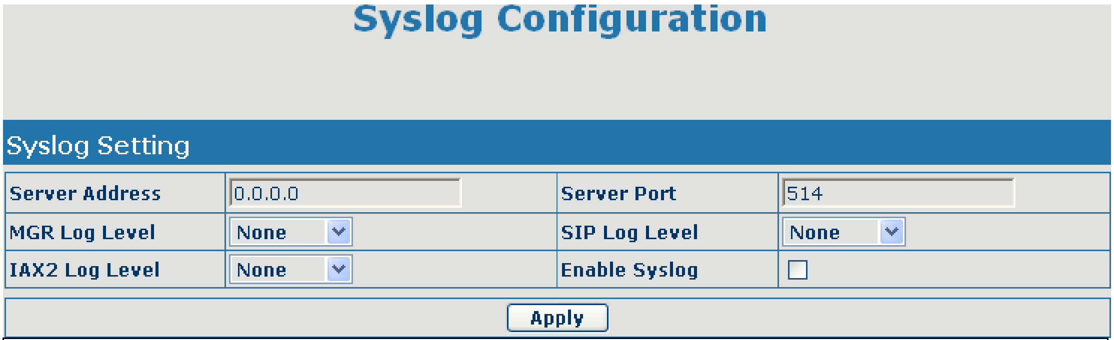
Account Configuration	
User Name	Set account user name.
User Level	Set user level, Root user has the right to modify configuration, General can only read.
Password	Set the password.
Confirm	Confirm the password.
Select the account and click the **Modify** to modify the selected account, and click the **Delete** to delete the selected account.	
Keyboard Password	Set the password for entering the setting menu of the phone

	by the phone’s key board. The password is digit.

4.2.8.2. Syslog Config

You can enable or disable the syslog function and config syslog server IP address & port via this page. 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.

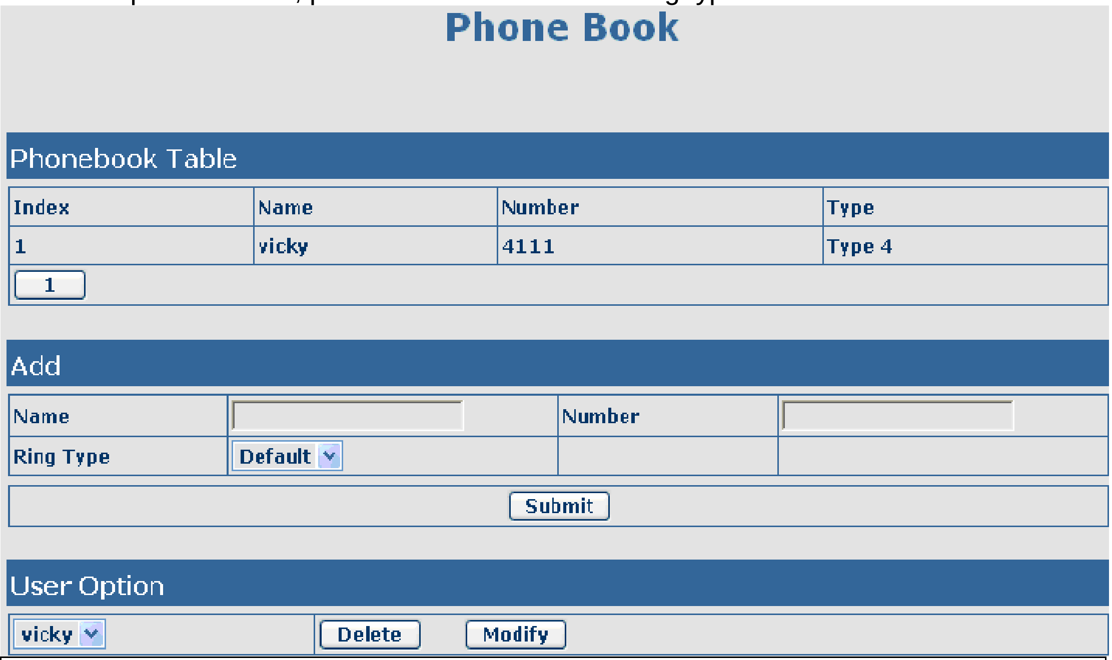
8 levels in debug information:
Level 0---emergency: This is highest default debug info level. You system can not work.
Level 1---alert: Your system has deadly problem.
Level 2---critical: Your system has serious problem.
Level 3---error: The error will affect your system working.
Level 4---warning: There 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.



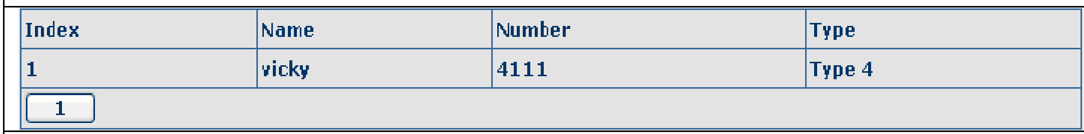
Syslog Configuration	
Server Address	Set Syslog server IP address.
Server Port	Set Syslog server port.
MGR Log Level	Set the level of MGR log.
SIP Log Level	Set the level of SIP log.
IAX2 Log Level	Set the level of IAX2 log.
Enable Syslog	Select it or not to enable or disable syslog.

4.2.8.3. Phone Book

You can input the name, phone number and select ring type for each name here.



Phone Book

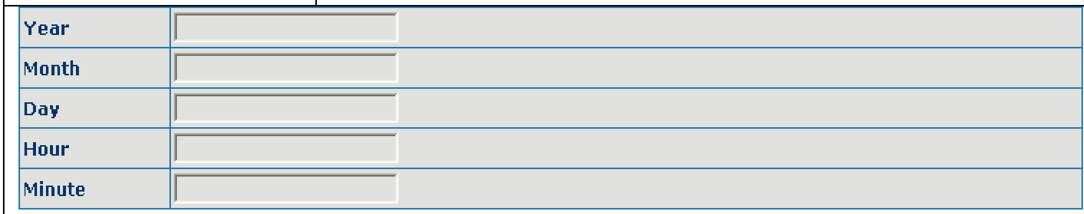


Shows the detail of current phonebook.	
Name	Shows the name corresponding to the phone number.
Number	Shows the phone number.
Ring Type	Shows the ring type of the incoming call.
Click "Modify" to change the selected information and click the "Delete" to delete the selected record.	
Notice: the maximum capability of the phonebook is 500 items	

4.2.8.4. Time Config

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.

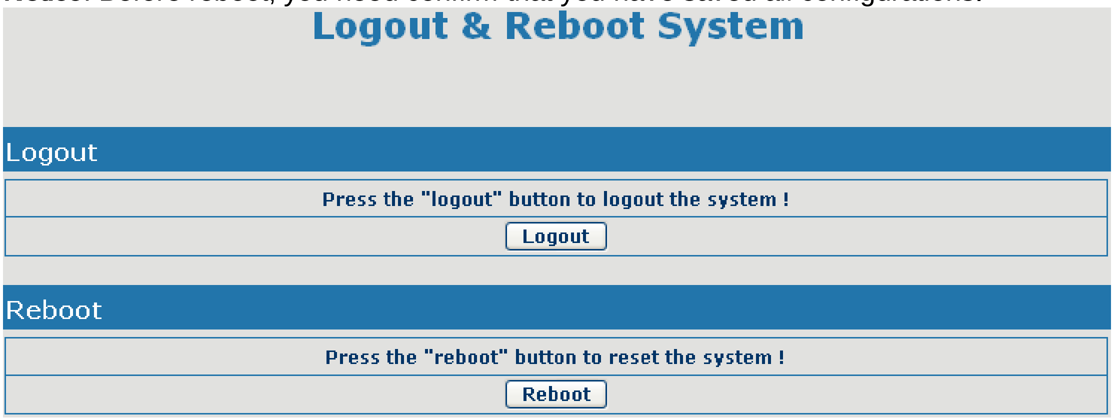
Time Configuration	
Server	Set SNTP Server IP address.
Time Zone	Select the Time zone according to your location.
Time Out	Set the time out, the default is 60 seconds.
Daylight	If your time zone supports daylight, you can select it.
SNTP	Select the SNTP, and click Apply to make the SNTP Times effective.



Notice: You need specify the above all items.

4.2.8.5. Logout &Reboot

Click **Logout**，and you will exit web page. If you want to enter it next time, you need input user name and password again. If you modified some configurations which need the phone's reboot to be effective, you need click the Reboot, then the phone will reboot immediately.
Notice: Before reboot, you need confirm that you have saved all configurations.



4.3. Settings via phone's keyboard.

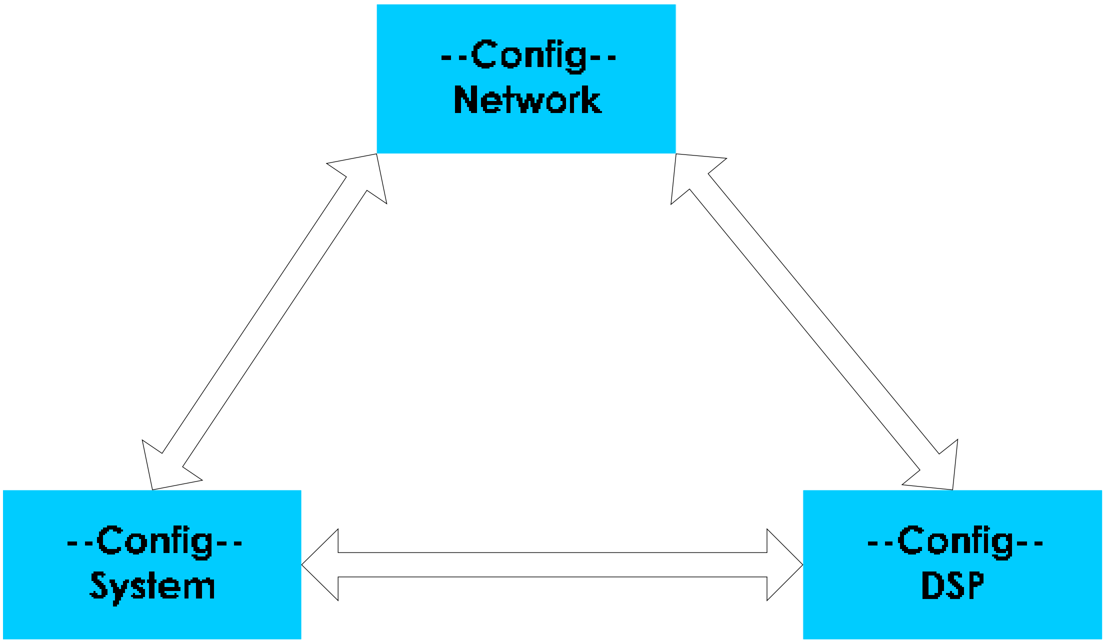
4.3.1. How to set via the phone's keyboard.

Press Menu, Up/Down, Enter and exit key to browse, select, and cancel

- Use the Up/Down key to browse the menu and submenu
- Use the ENTER key to enter into submenu and confirm your operation, the EXIT key can be used to back and cancel operation.

4.3.2. Phone menu

Phone main menu:



5. Appendix

5.1. Specification

5.1.1. Device specification

Item		this VoIP Phone
Adapter(Input/Output)		Input：100-240VAC 50～60Hz　　Output：5V/1A
Port	WAN	10/100Base- T　RJ-45 for LAN
	LAN	10/100Base- T　RJ-45 for PC
Power Consumption		Idle：1.5W/Active：1.8W
LCD size		3in.（74 x 28mm）
Operation Temperature		0～40℃
Relative Humidity		10～65%
Main Chipset		MIPS32(150M)，DSP(100M）
SDRAM		128Mbits
Flash		16Mbits
Size（W x H x D）		11.6×8×3 in.(295×205×75mm)
Weight		2.07lb.(0.94kg)

5.1.2. Voice Features

- Support IAX2 and SIP 2.0 (RFC3261)
- Codec：G.711A/u，G.7231 high/low，G.729, G.722
- Echo cancellation：　Support G.168 and hand-free can support 96ms
- Support VAD，CNG
- NAT transverse: support STUN
- SIP support SIP domain，SIP authentication（none，basic，　MD5），DNS name of server, peer to peer

- SIP support Pubic & Private server, user can through each server to calling in and out
- DTMF：SIP info，DTMF Relay，RFC2833
- SIP application: contain SIP call forward/transfer/holding/waiting/3 way conference
- Call control features: Flexible dial map, support hotline, empty calling no. reject server, black list for reject authenticated call no disturb, caller ID
- support conference call and voice record
- Support English, Spanish and Czechish (optional)
- Could dial use private server automatically when public server unregistered while private server is resgistered successflly
- 8 special ring type
- 500 entry phonebook, Call records: 100 dialed, 100 received, 100 missed calls

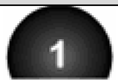
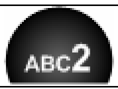
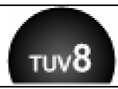
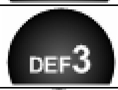
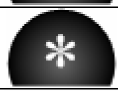
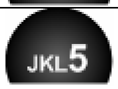
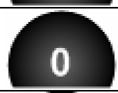
5.1.3. Network Features

- WAN/LAN: support Bridge and Router mode.
- Support basic NAT and NAPT
- Support PPPoE for xDSL
- Support reconnecting automatically when PPPoE(adsl) is disconnected by ISP
- Support DHCP get IP on WAN port
- Support DHCP distribute IP on LAN port
- Support primary DNS server and secondary DNS server
- Support DNS relay，SNTP server，Firewall on WAN port
- support network tools: contain ping，trace route，telnet client
- support VLAN

5.1.4. Maintenance and Management

- Support Boot Monitor
- Can upgrade firmware through boot monitor
- access with different authority
- support auto provisioning
- Can config through Web, Keypad, Telnet
- Can upgrade firmware and configuration file through HTTP, FTP, TFTP
- Support syslog

5.2. Key mapping

Button	Character	Button	Character
	**1 @ - /**		7 P Q R S
	2 A B C		8 T U V
	3 D E F		9 W X Y Z
	4 G H I		.
	5 J K L		0 * #
	6 M N O