FANVIL TECHNOLOGY CO., LTD
SMART VOIP

Fanvil Product User Manual
IP Phone
Model: C56/C56P

Version: V.2.2.0.0

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Table of Content

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.

1. Introducing C56/C56PVoIP Phone

1.1. Thank you for your purchasingC56/C56P

Thank you for your purchasing C56/C56P, C56/C56P 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 enjoying 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.

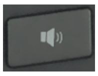
1.2. Delivery Content

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

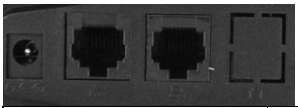
1.3. Keypad

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	Key name	Function Description
	History	In idle/pickup/calling mode, press the Callers key to Check the Income/Outgoing/Missed calls records. Press this key again will return to idle mode
	LED	LED blinks to remind user new voicemail.
	System Information	In idle mode, press the Sysinfo key to check the phone setting parameters. Such as local phone number, local IP and local Gateway IP address.
	Confirm	Use the Enter key to enter next menu, or confirm the setting.
	Exit	Use the Exit key to return to previous menu, cancel the setting, or reject to answer a call.
	Navigation Key	When you pick up the handset or during calling, you can use this key to turn up or turn down the handset volume; when a call comes, you can use this key to adjust ring volume; you also can use this key to choose item in the menu, callers or phone book. Notice: the left has deleting function.
	MWI	Use this key to read old or new message.
	Transfer	Use the key to realize blind transfer or attended transfer please refers to **3.1.4.**-call transfer for more details).
	Conference	Use this key to realize the three party call (please refer to**3.1.5-Calling Hold and 3 ways call** for more details)

	Hold	Temporarily hold the active call during the talking; press the key again to unhold the call. You also can press this key then input the third party's phone number and end with the # key during calling; you can make a call with the third party and hold the previous calling. **(3.1.5-Calling Hold and 3 ways call).**
	mute	Press this key in calling mode, you can hear the other side, and the other side can not hear you
	Redial	In the hook off /hands-free mode, use the key to dial the last call number; use this key to make a quick dial as soon as you select your desired number in phone book or callers.
	Handfree	Enter into hands-free mode.

1.4. Port for connecting



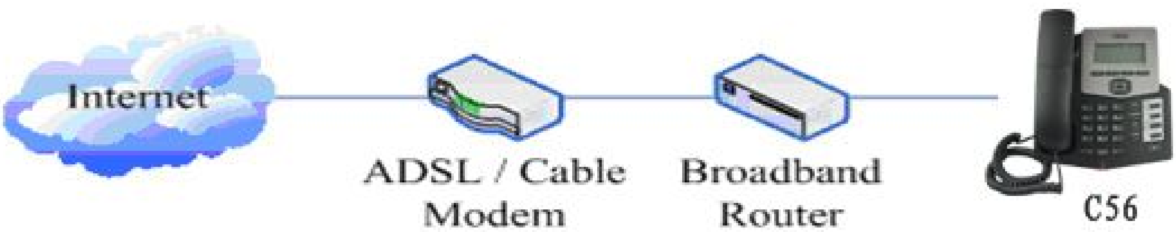
POWER	Power switch	Select ON/OFF
LAN	Network port	Connect it to PC
WAN	Network port	Connect it to Network

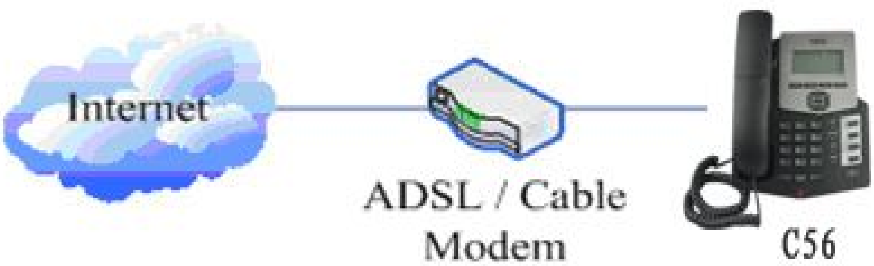
The phone has two Network ports: The WAN port and the LAN port. Before you connect the power source, please carefully read Safety Notices of this user manual.

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.





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 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. Press the 3 key for three seconds, and then confirm it by the Enter key, your phone network connection mode will switch into PPPoE mode. Prepare your PPPoE account name and password.
2. Press the OK 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 

key and LCD screen will display "WAN", press the  key, enter it by the 

key, the LCD screen will display "STATIC NET". Then press the  key again, enter it by the 

key, the LCD screen will display "USER NAME".
5. Press the 

key and then press the  key (the left is also empowered delete function), input your PPPOE account number then press the 

key to confirm. The LCD screen will display the inputted PPPOE account number.
6. Press the 

key to return to the previous menu, and then press the  key, the LCD screen will display "PASSWORD". Then press the ENTER key, and the  key, input your PPPoE's password and confirm it by the 

Key, the LCD screen will display the password 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 

key again, the LCD screen will display "ARE YOU SURE".
9. Press the 

key, the phone will save your setting and the LCD screen will display "SAVING NOW", then return to display "SAVE".

10. Press the  key twice, then press numeric key "3"and hold until the screen display "ARE YOU SURE". Press the  key, the screen will display "CHANGING", which means that the phone is trying to switch to PPPoE mode. If the icon "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  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. Press the 1 key for three seconds, then confirm it by the  key, your phone network connection mode will switch into Static IP mode. 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  key, the LCD screen will display "INPUT PASSWORD".

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

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

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

6. Press the  key to return to previous menu, and then press the  key, the LCD screen will display "DNS2". Press the  key then the  key, input your spare DNS address and confirm it by pressing the  key, and then the LCD will display the inputted DNS address.

7. Press the  key to return to previous menu, and then press the  key, the LCD screen will

display “DNS”. Press the 

key then the  key, input your DNS address and confirm it by pressing the 

key, and then the LCD will display the inputted DNS address.

8. Press the 

key to return to the previous menu, and then press the key, the LCD screen will display “GATEWAY”. Press the 

key and the key, input your gateway’s IP address and confirm it by pressing the 

key, the LCD screen will display the inputted gateway address.

9. Press the 

key to return to the previous menu, and then press the  key, the LCD screen will display “NETMASK”. Press the 

key and the  key, input your netmask and press the 

key to confirm it. The LCD screen will display the inputted netmask.

10. Press the 

key for four times and press the  key, till the LCD Screen displays “SYSTEM”.

11. Press the 

key, the LCD screen will display “save”, then press the 

key again, the LCD screen will display” ARE YOU SURE”.

12. Press the 

key, this phone will display “SAVING NOW”, then return to display “SAVE”.

13. Press the 

key twice to exit the menu, and then press the numeric key 1 till the LCD screen displays “ARE YOU SURE”. Press the 

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 

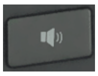
key, the LCD screen will display “CHANGING” and this VoIP phone is trying to switch to DHCP mode. If the icon “DHCP” on the top of the screen keeps blink, it shows that the phone is trying to access the DHCP server, and the IP is 0.0.0.0 if you press  key to display the current IP; if the icon “DHCP” 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  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 thebutton, and then hang up the handset.

3.1.2. Making a call

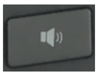
- Quick-dialing

In idle mode, input the called number, and press # key or  button, phone will dial the call and use hands-free automatically.

- 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  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  button again to end the call.

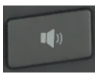
- Use the phone book

Press the  button and input password, then Press the/ button to find phonebook. Press the/ button toselect your desired contact person, and then press the  button to dial the call.

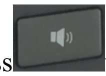
- Use Callers

Press the key, and then select your desired phone number in callers by the/key, and next press the  button to dial the call.

- Use the R/Send key

Please pick up or press the  key. After you hear dialing tone, please press the  key to dial the last phone number. Note: after you reboot the phone, the phone will delete callers and Redial will be invalid.

3.1.3. Ending a call

- Hangs up by handset on hook
- Hangs up by press  when in hands-free
- Hangs up a call 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. Note: 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  key and dial C phone number. After B talks to C (or B hear alert from C), B presses the  key, then B hangs up, and A will get through to C.
2. When A is talking with B, C calls B, B may press the  key to hold A, and talk to C. Then B presses the  key, A will get through to C.
3. When A talks to B, B presses the  key, dial C phone number and # key, then 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  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  key again you 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 or  key 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  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. Callers

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 

 key, and then the  key, till the LCD screen display "**MISSED**". Press the 

 key, the LCD screen will display the missed call number and sequence numbers of the missed call.

  You can press the 

 key to dial this phone number, you also press UP/DOWN key to browse the other missed calls or you can press the 

 key again, 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 

 key, and then the  key, till the LCD screen display "**RECEIVED**". Press ENTER key, the LCD screen will display the received call number and sequence number of the received call.

  You can press the 

 key to dial this phone number, you also press  key to browse the other received calls or you can press the 

 key again, 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 

 key, and then the  key, till the LCD screen display "**OUTGOING**".

  Press 

 key, the LCD screen will display the phone number and sequence number of the dialed call. You can press the 

 key to dial this phone number, or press the  key to browse all record 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.

3.2.1. Special Keys

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

When you make secondary dial in off-hook/handsfree/standby pre-input mode, press

key 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.

- **MWI(Message Waiting Indication)**

When a new voicemail coming, LED on the phone will flash. You can press the **MWI** key to listen new voicemail if you configure mwi number

3.2.2. redial/unredial

If B is in busy line when A calls B, A will get notice: busy, please hang up. If A want to connect Bas soon as B is in idle, he can use redial function at the moment and he can dial 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.

Number	Destination	Port	Mode	Alias	Suffix	Deleted 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.

3.2.3. Click to dial

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

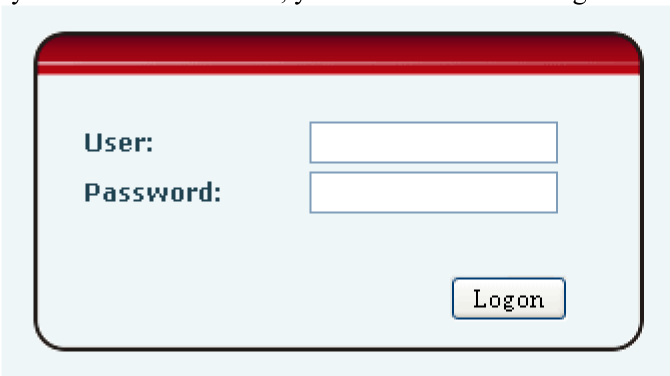
4. Setting

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 initial password is 123 for setting via 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**".
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 Addresses and ports 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

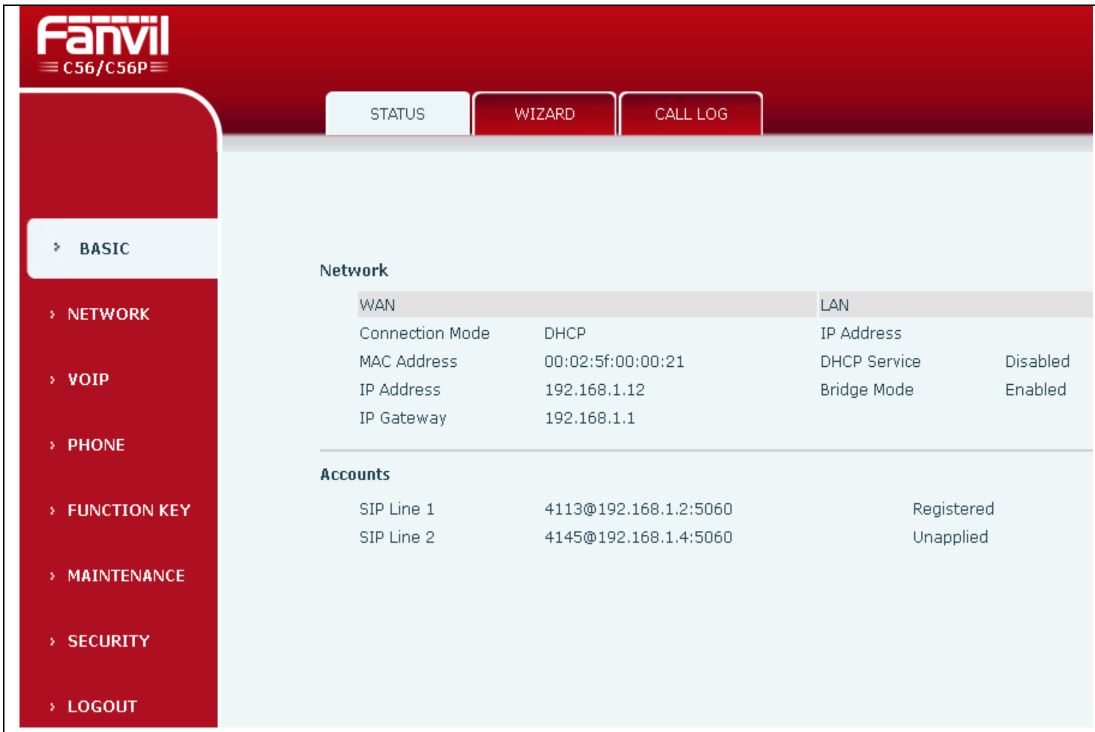
Default value in Administrator privilege
Username: admin
Password: admin

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 "maintenance" and then click the "config" button and the "Save" button on the right field.

4.3. Configuration via WEB

4.3.1. BASIC

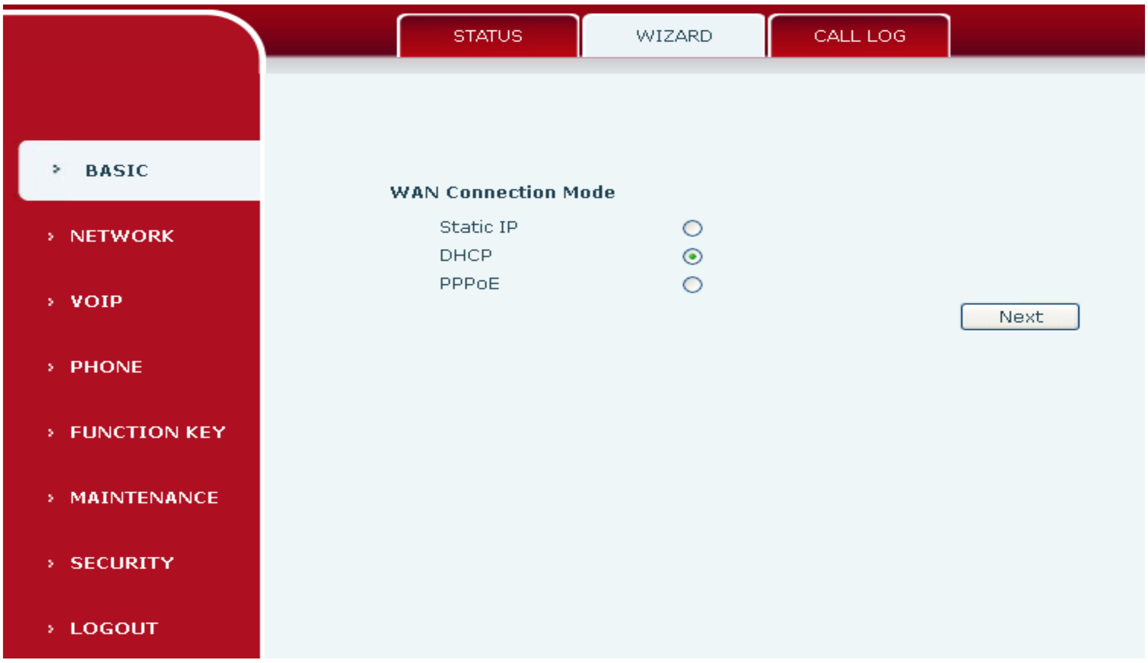
4.3.1.1. Status

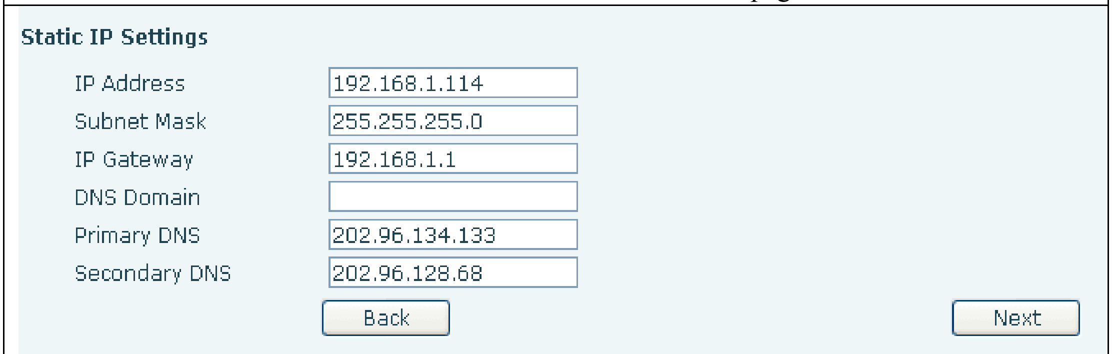
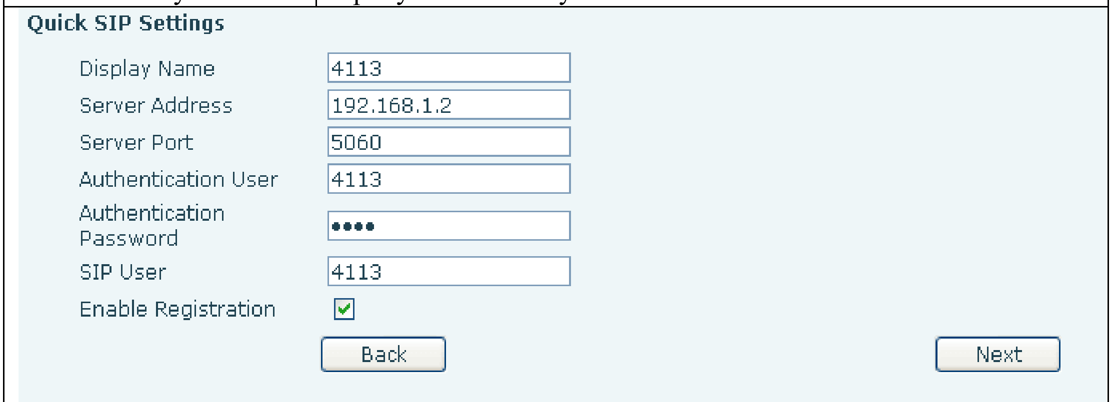


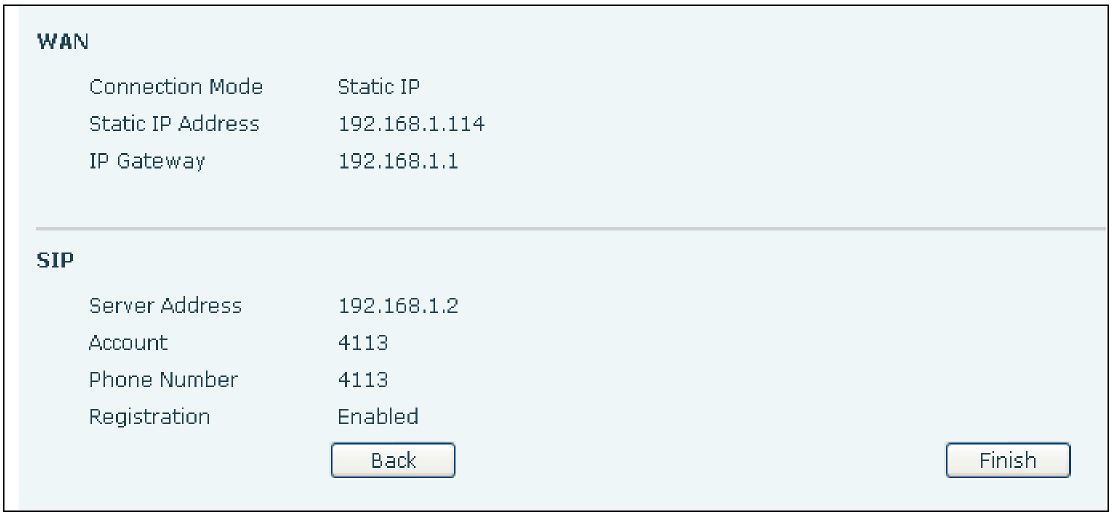
Status

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.
Accounts	Shows the phone numbers provided by the SIP LINE 1-2 servers. The last line shows the version number and issued date.

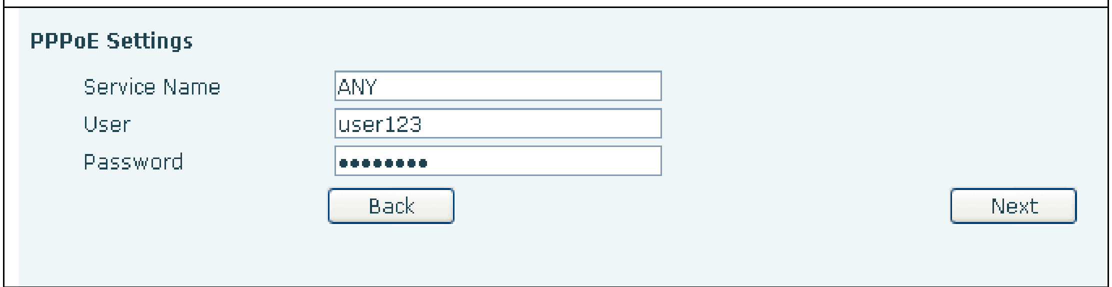
4.3.1.2. Wizard



<table>
<tr><th colspan="2">Wizard</th></tr>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td colspan="2">Please select the proper network mode according to the network condition. FV6030 provide three different network settings:
● Static IP: 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 3.2.1 Network setting to speed setting your network.</td></tr>
<tr><td colspan="2">Choose Static IP MODE， click【NEXT】can config the network and SIP(default SIP1)easily, also can browse them too. Click【BACK】can return to the last page.</td></tr>
<tr><td colspan="2">
</td></tr>
<tr><td>IP Address</td><td>Input the IP address distributed to you.</td></tr>
<tr><td>Subnet Mask</td><td>Input the Subnet Mask distributed to you.</td></tr>
<tr><td>IP Gateway</td><td>Input the Gateway address distributed to you.</td></tr>
<tr><td>DNS Domain</td><td>Set DNS domain postfix. When the domain which you inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again.</td></tr>
<tr><td>Primary DNS</td><td>Input your primary DNS server address.</td></tr>
<tr><td>Secondary DNS</td><td>Input your Secondary DNS server address.</td></tr>
<tr><td colspan="2">
</td></tr>
<tr><td>Display Name</td><td>If user set the display name, caller will show this display name.</td></tr>
<tr><td>Server Address</td><td>Input your SIP server address.</td></tr>
<tr><td>Server Port</td><td>Set your SIP server port.</td></tr>
<tr><td>Authentication User</td><td>Input your SIP registered account name.</td></tr>
<tr><td>Authentication Password</td><td>Input your SIP registered password.</td></tr>
<tr><td>SIP User</td><td>Input the phone number assigned by your VOIP service provider.</td></tr>
<tr><td>Enable Registration</td><td>Start to register or not by selecting it or not.</td></tr>
</table>



Display detailed information that you manual config.
Choose DHCP MODE，click【NEXT】to config simple SIP(default SIP1). You can browse it too. Click【BACK】to return to the last page. Like Static IP MODE。
Choose PPPoE MODE，click【NEXT】to config the PPPoE account/password and SIP(default SIP1). You can browse it too. Click【BACK】to return to the last page. Like Static IP MODE。

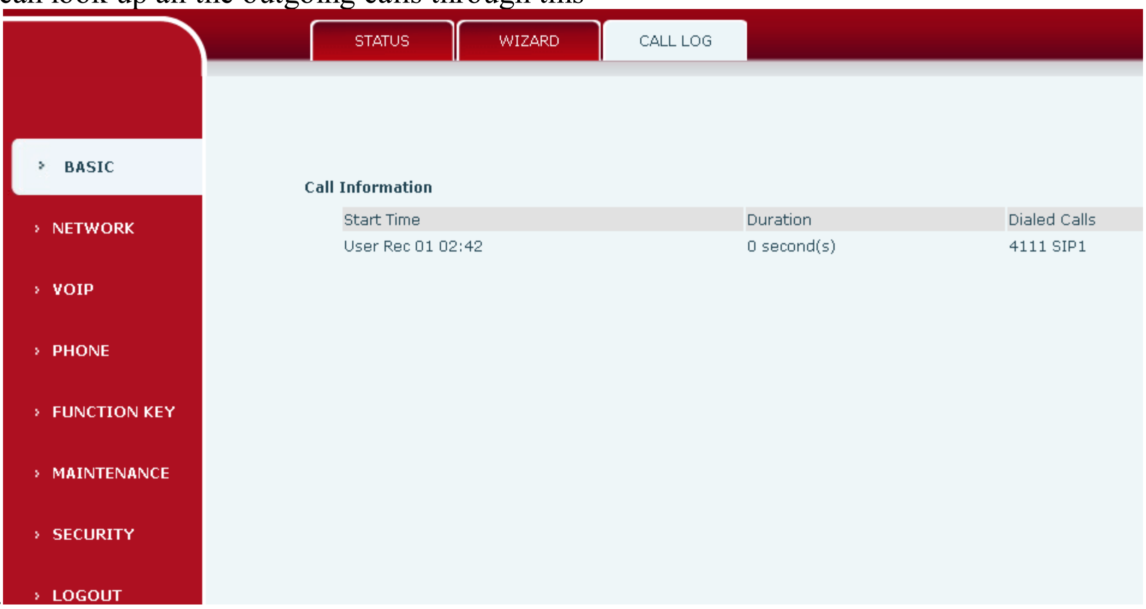


Server Names	It will be provided by ISP.
User	Input your ADSL account.
Password	Input your ADSL password.

Notice: Click【**Finish**】button after finish your setting, IP Phone will save the setting automatically and reboot. After reboot, you can dial by the SIP account.

4.3.1.3. Call Log

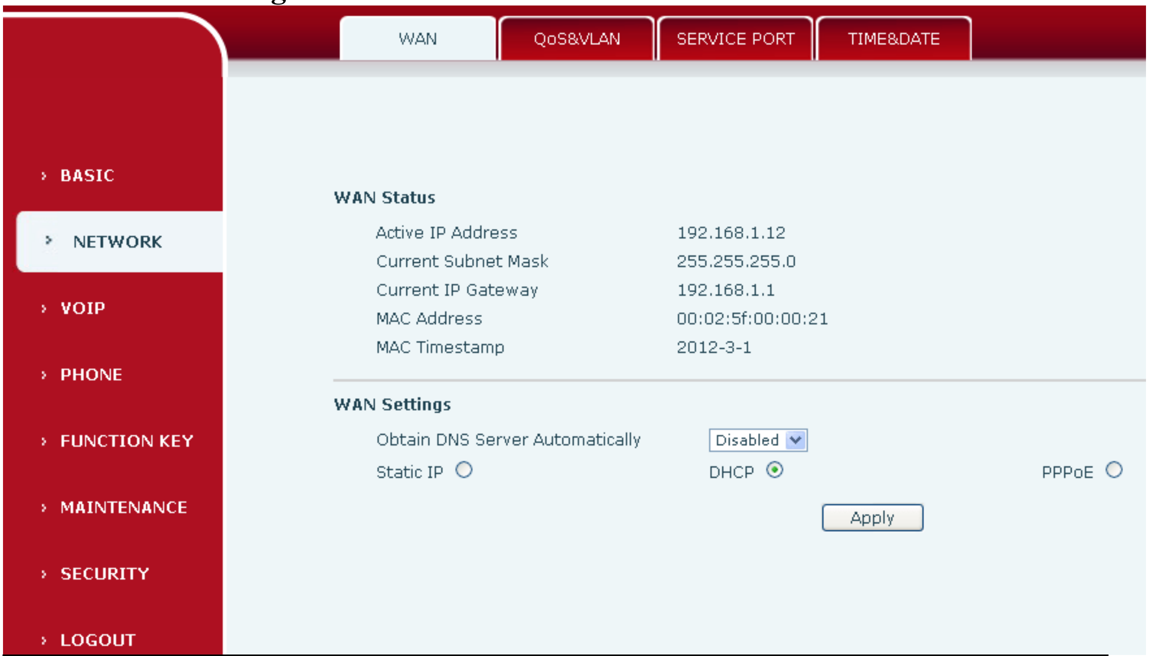
You can look up all the outgoing calls through this page.



Field name	explanation
Start Time	Display the start time of the outgoing call
Duration	Display the conversation time of the outgoing call.
Dialed Calls	Display the account/protocol/line of the outgoing call.

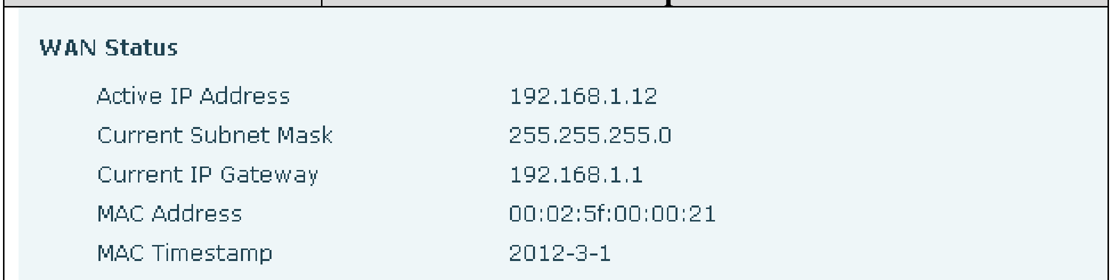
4.3.2. Network

4.3.2.1. WAN Config



WAN Config

Field Name	explanation



Field Name	explanation
Active IP Address	The current IP address of the phone.
Current Subnet Mask	The Current Subnet Mask address.
MAC Address	The current MAC address of the phone.
Current IP Gateway	The current Gateway IP address.
MAC Timestamp	Shows the time of getting MAC address

Please select the proper network mode according to the network condition. FV6030 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 3.2.1 Network setting to speed setting your network.

IP Address	192.168.1.114
Subnet Mask	255.255.255.0
IP Gateway	192.168.1.1
DNS Domain	
Primary DNS	202.96.134.133
Secondary DNS	202.96.128.68

If you use static mode, you need set it.

IP Address	Input the IP address distributed to you.
Subnet Mask	Input the Subnet Mask distributed to you.
IP Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again.
Primary DNS	Input your primary DNS server address.
Secondary DNS	Input your Secondary DNS server address.

Service Name	ANY
User	user123
Password	●●●●●●●●●

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

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

Notice:
1） Click “Apply” button after finish your setting, IP Phone will save the setting automatically and new setting will take effect.
2） If you modify 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.

4.3.2.2. 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.

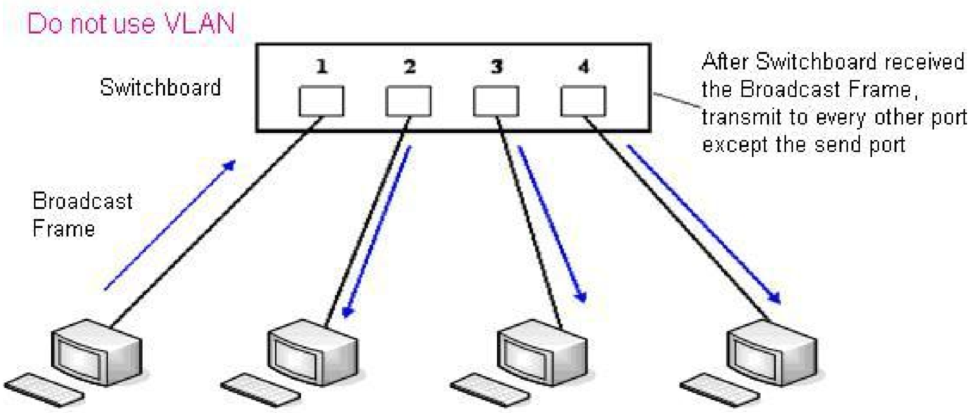


Chart 1

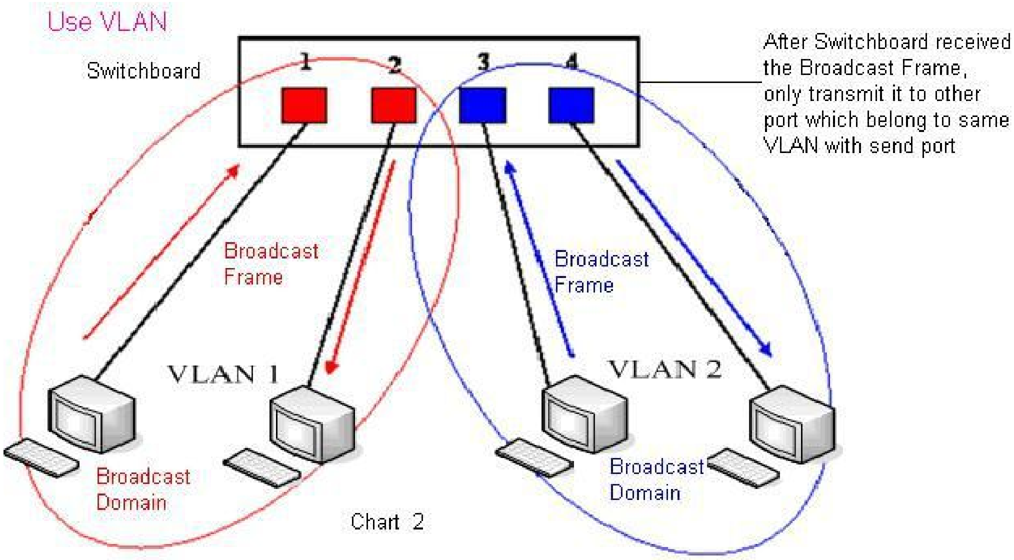
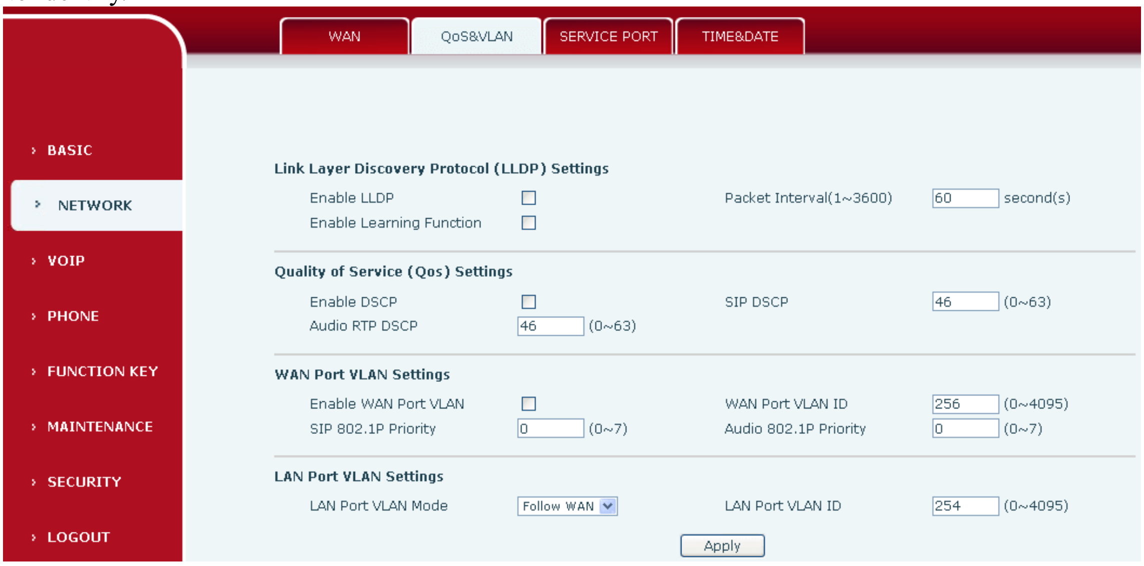


Chart 2

In chart 1, there is a layer 2 switches 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 2,3and 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 viare stricting the range of broadcast frame transmition.

Note: chart 2 use red and blue to identify the different VLAN, but in practice, VLAN uses different VLAN IDs to identify.

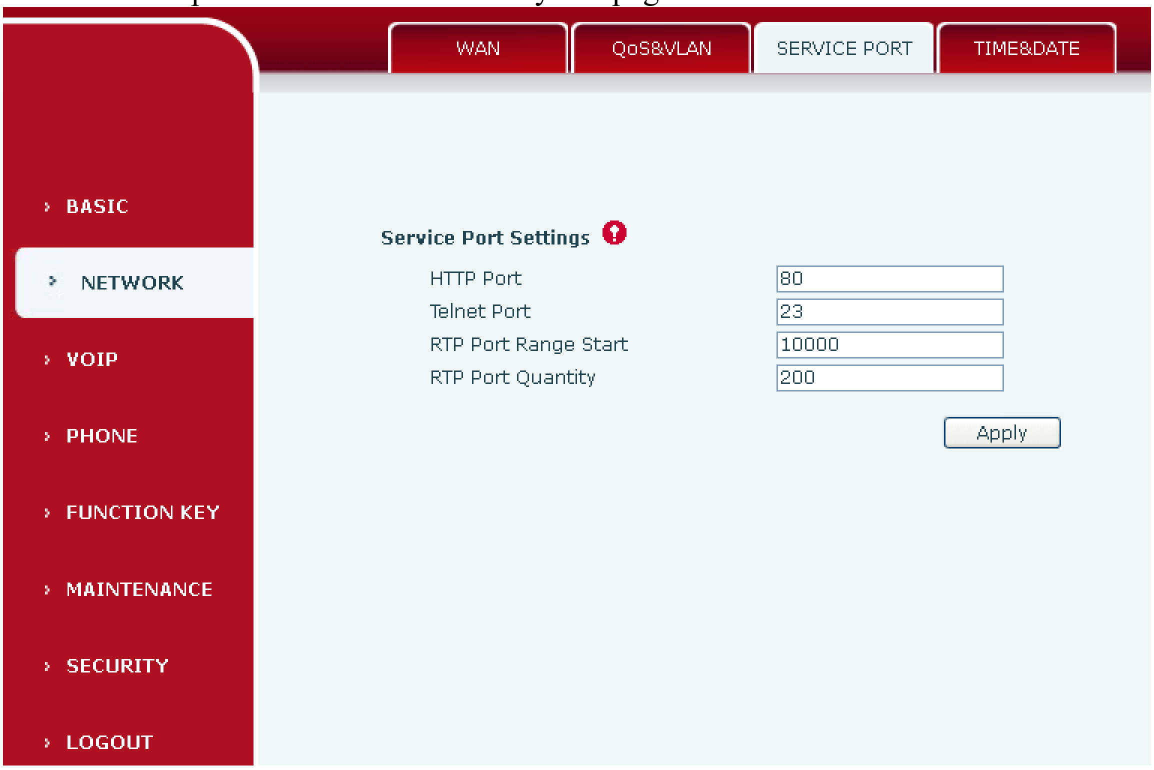


QoS Configuration	
Field name	**explanation**
Enable LLDP	Enable LLDP by selecting it
Packet Interval	The time interval of sending LLDP Packet
Enable Learning Function	After enabling LLDP Learn, telephone can automatically learn the data of DSCP, 802.1p, VLAN ID from the switch. If the data is different from the data of the LLDP server, telephone will change its own value as the value of the switch（Synchronous with VLAN in switch）
Enable DSCP	Enable Dscp by selecting it
SIPDSCP	Specify the value of the SIP DSCP

Audio DSCP	Specify the value of the Audio DSCP
Enable WAN Port VLAN	Enable WAN Port VLAN by selecting it
WAN Port VLAN ID	Specify the value of the WAN Port VLAN ID, the range of the value is 0-4095
SIP 802.1P Priority	Specify the value of the voice 8021.p priority, the range of the value is 0-7
Audio 8021P Priority	Specify the value of the signal 8021.p priority, the range of the value is 0-7
LAN Port VLAN Mode	Follow WAN: Follow the WAN ID Disable: Disable Port VALN Enable: Enable Port VLAN and specify the Port VLAN ID different from WAN ID
LAN Port VLAN ID	Specify the value of the Port VLAN ID different from WAN ID, the range of the value is 0-4095

4.3.2.3. Service Port

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



SERVICE PORT	
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 Port Range Port	Set the RTP Port Range Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.

Notice:

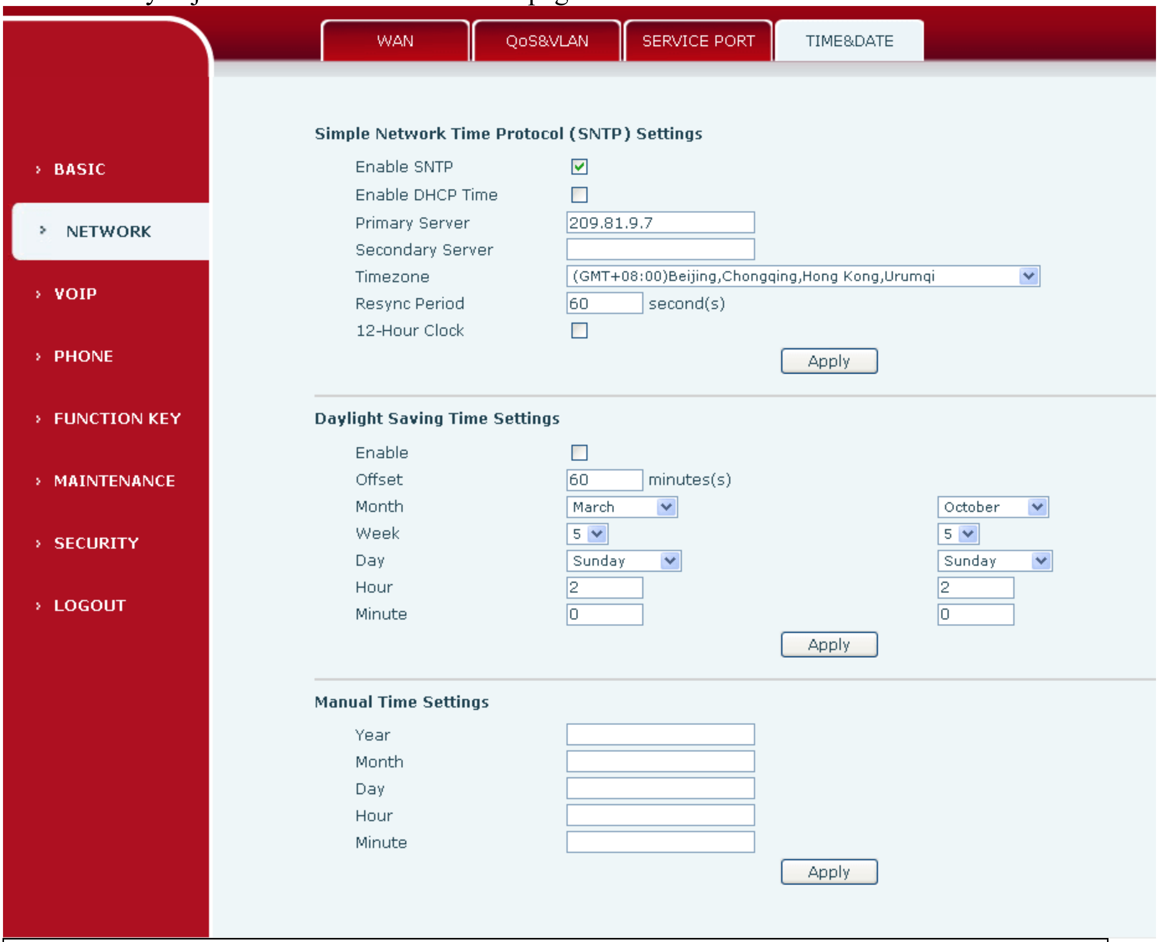
1）You need save the configuration and reboot the phone after set this page.

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.

4.3.2.4. TIME&DATE

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.



SNTP	
Field name	**explanation**
Enable SNTP	Enable SNTP by selecting it
Enable DHCP Time	Enable DHCP Time by selecting it, then the phone will automatically synchronize the standard time.
Primary Server	Set SNTP Primary Server IP address.
Secondary Server	Set SNTP Secondary Server IP address
Timezone	Select the Time zone according to your location.
Resync Period	Set the Resync Period, the default is 60 seconds.
12 –Hour Clock	Switch the time mechanism between 12 hours and 24 hours. Default is 24 hours mode.
Enable	Enable daylight saving time
Offset(minutes)	Setup the variety length
Month	Setup stat and end month
Week	Setup start and end week
Day	Setup start and end day
Hour	Setup start and end hours

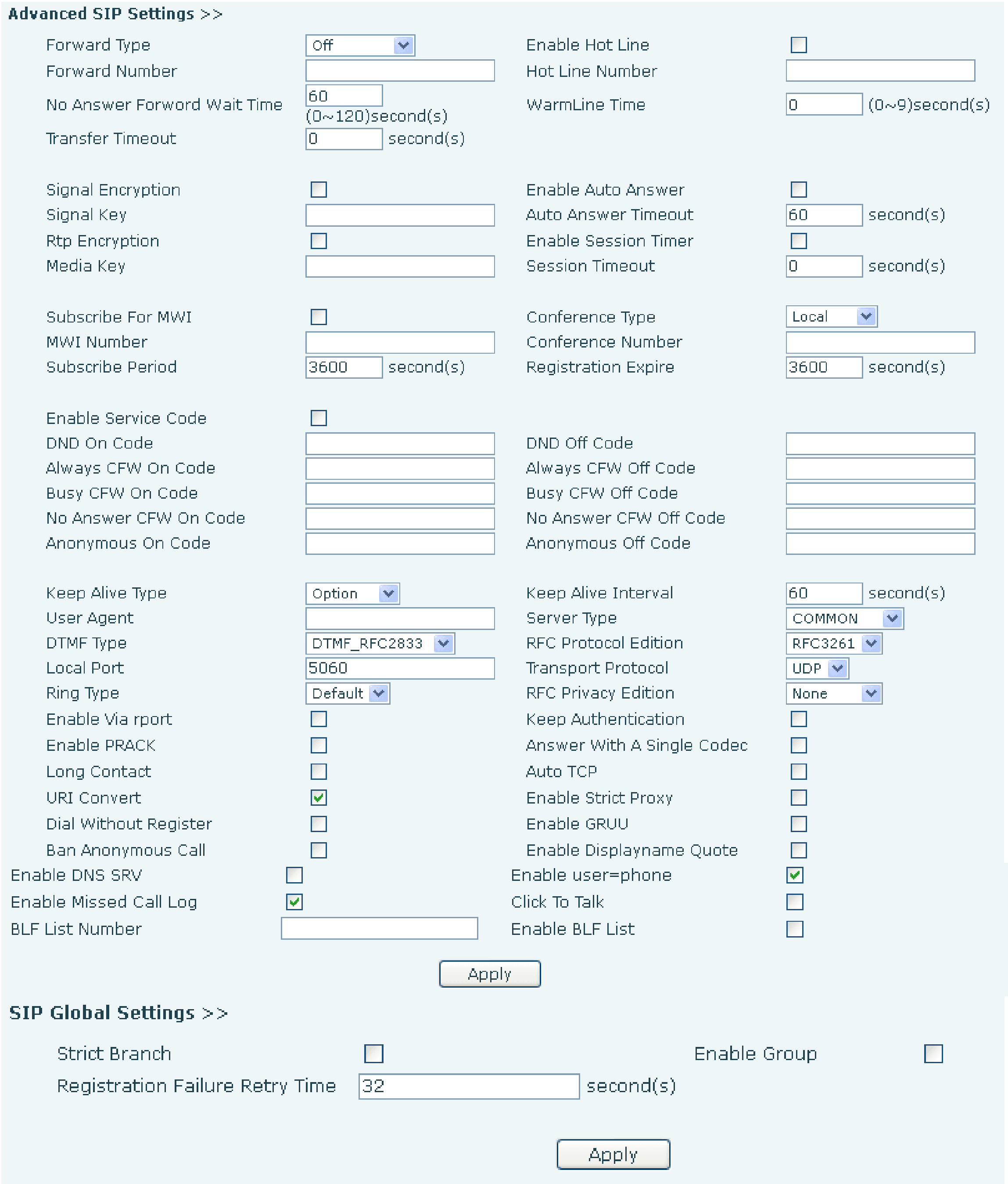
Minute	Setup start and end minutes

Notice: You need specify the above all items.

4.3.3. VOIP

4.3.3.1. SIP Config

Set your SIP server in the following interface.



SIP Config	
Field name	**explanation**
Choose the sip line to set info about SIP;there are 2 lines to choose. You can switch by 【**Load**】 button.	
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.
Authentication User	Input your SIP registered account name.
AuthenticationPassword	Input your SIP registered password.
SIP User	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 User	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).
Backup Server Address	Input the Backup Server Address, if the primary server is unavailable , then the phone will enable the Backup Server Address
Backup Server Port	Specify the Backup Server Port
Enable Registeration	Start to register or not by selecting it or not.
Disable Codecs/Enable Codecs	Use the navigation keys to highlight the desired one in the Enable/Disable Codecs list, and press the desired to move to the other list.
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 after a specific. Always：Incoming calls will be forwarded to the appoint phone immediately. The phone will prompt the incoming while doing forward.
Forward Number	Specify the number you want to forward.
No Ans. Fwd Wait Time	Specify the No Answer Forward Delay Time，if the Forward Type is No answer, incoming calls will be forwarded after the no answer forward wait time
Transfer Timeout	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 Hotline	Specify Hot Line by selecting it
Hotline Number	Specify Hot Line Number, the phone dial the hot line number automatically at hands-free mode or handset mode after warm line time
Warm Line Wait Time	Specify the Warm Line Time
SIP Encryption	Enable/Disable Signal Encrypt.
SIP Encryption Key	Set the key for signal encryption.
RTP Encryption	Enable/Disable RTP Encrypt.
RTP Encryption Key	Set the key for RTP encryption
Enable Auto Answer	Enable Auto Answer by selecting it
Auto Answer Timeout	Specify Auto Answer Time, the phone auto answers the incoming call after Auto Answer Time
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Session Timeout	Set the session timeout
Subscribe For MWI	Enable the Subscribe for MWI by selecting it, the phone will send subscribe message for MWI to the SIP Server
MWI Number	Specify the MWI Number, Please contact your system

	administrator for the connecting code. Different systems have different codes.
Subscribe Period	Overtime of resending subscribe packet. Suggest using the default configuration.
Conference Type	Specify the Conference Type, if you select the local, you needn't input the conference number
Conference Number	Specify the network conference number, please contact your system administrator for the network conference number
Registration Expires	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 expired time set, the phone will change automatically the time into the time recommended by the server, and register again.
Enable Service Code	If you want to realize the following function by the server, please enter the On Code and Off Codeoption, then when you choose to enable/disable following function on your IP phone, it will send message to the server, and the server will turn on/off the function immediately.
DND On Code	Set the DND On Code, When you press the DND hot key, the phone will send a message to the server, and the server will turn on the DND function. Then any calls to the extension will be rejected by the server automatically. And the incoming call record will not be displayed in the Call History.
DND Off Code	Set the DND Off Code, When you press the DND hot key, the phone will send a message to the server, and the server will turn off the DND function.
Always CFwd On Code	Set the Always CFW On Code, when you choose to enable the always forward function on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will always forward it to the set number automatically. And the IP phone will not show the record in the call history anymore.
Always CFwd Off Code	Set the Always CFW Off Code, when you choose to disable the always forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
Busy CFwd On Code	Set the Busy CFW On Code, when you choose to enable the busy forward function v on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
Busy CFwd Off Code	Set the Busy CFW Off Code, when you choose to disable the busy forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
No Ans. CFwd On Code	Set the No Answer CFW On Code, when you choose to enable the on answer forward function on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
No Ans. CFwd Off Code	Set the No Answer CFW Off Code, when you choose to disable the busy forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
Anonymous On Code	Set the Anonymous On Code, When you choose to enable the anonymous call function on your IP phone, it will send information to the server, and the server will enable the anonymous call function for your IP phone automatically.

Anonymous Off Code	Set the Anonymous Off Code, When you choose to disable the anonymous call function on your IP phone, it will send information to the server, and the server will disable the anonymous call function for your IP phone automatically.
Keep Alive Type	Specify the keep alive type, if the type is option, the phone will send option sip message to server every NAT Keep Alive Period(s), then the server responses with 200 to keep alive. If the type is UDP, the phone will send UDP message to server to keep alive every NAT Keep Alive Period(s).
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
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.
Local port	Set sip port of each line
Ring type	Set ring type of each line
Enable 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 config.
Long Long Contact	Set more parameters in contact field; connection with SEM server
Convert URI	Convert # to %23 when send the URI.
Dial Without Registered	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
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
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 RFC 2543, else phone may not cancel call normally. System uses RFC3261 as default.
Transport Protocol	Set transport protocols, TCP or UDP;
Anonymous Call Edition	Set Anonymous call out safely; Support RFC3323and RFC3325;
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.
Ans. With a 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	Enable user=phone by selecting it, it is contained in the invite sip message, in order to be compatible with server
Enable Missed Call Log	Enable the missed call log by it, the phone will save the missed call log into the call history record and display the missed calls on the idle screen, or won't save the missed call log into the call history record and display the missed calls on the idle screen.

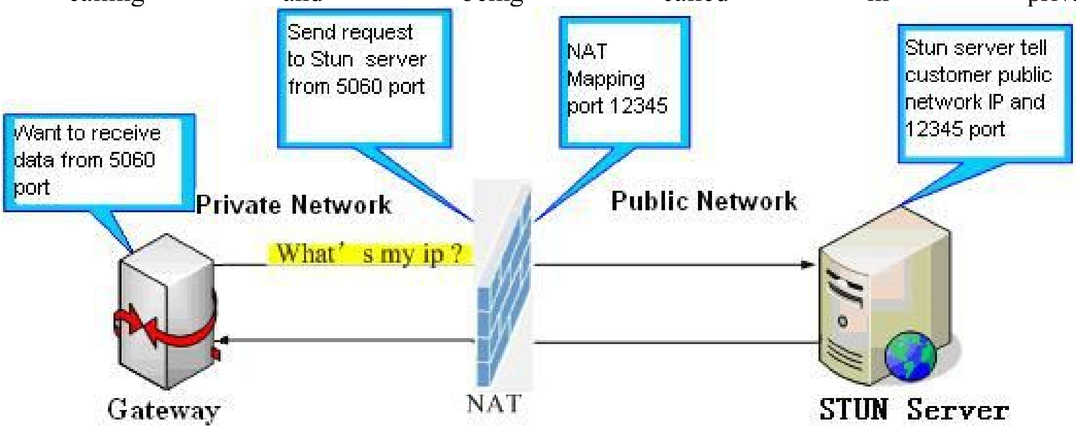
Click to Talk	Set click to Talk (need practical software support).
Enable BLF List	Enable BLF List by selecting it, BLF list is a function which can monitor the group status, it is not one to one monitoring, but the information feedback from the sever to decide which BLF list will monitor
BLF List Number	Specify the BLF List Number
Strict Branch	Enable the Strict Branch, the value of the branch must be in the beginning of z9hG4k in via field of the invite sip message received, or the phone won't response to the invite sip message. Notice: the deployment will become effective in all sip lines
Enable Group	Enable Group by selecting it, then the phone enable the sip group backup function Notice: the deployment will become effective in all sip lines
Registration Failure Retry Time	Specify the registration failure retry time, if the phone register failed, the phone will register again after registration failure retry time. Notice: the deployment will become effective in all sip lines.

4.3.3.2. Stun Config

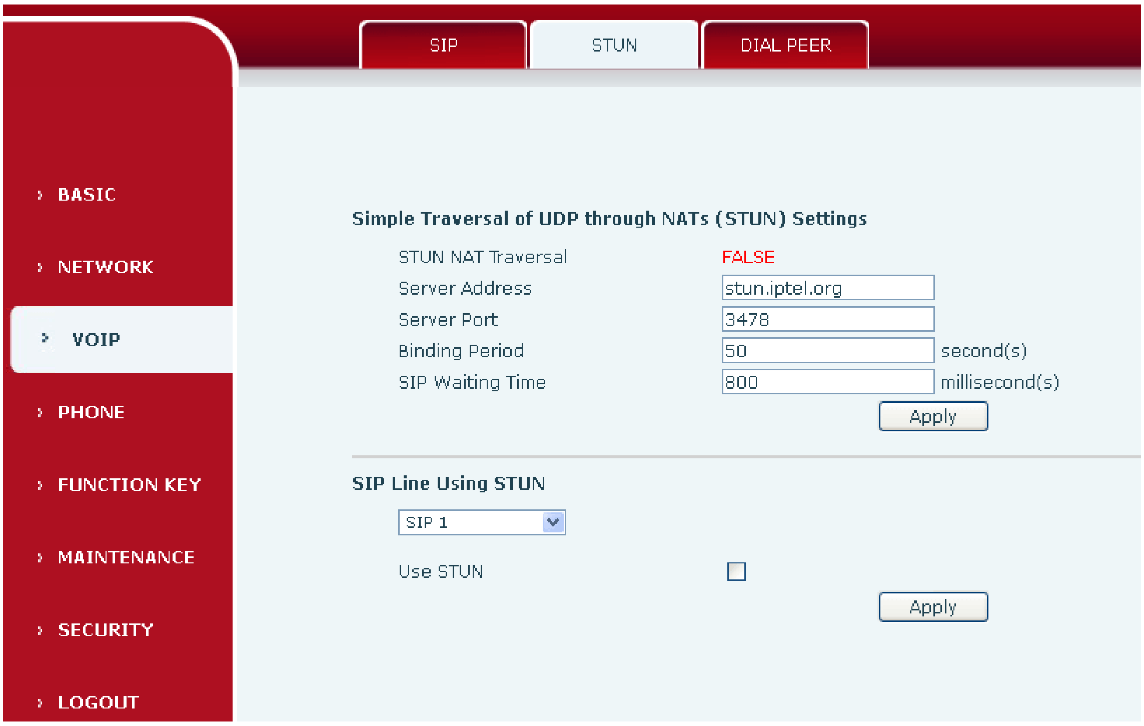
In this web page, you can config SIP STUN.

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.



STUN	
Field name	**explanation**
STUN NAT Transversal	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
Server Address	Set your SIP STUN Server IP address
Server Port	Set your SIP STUN Server Port
Binding Period	Set STUN blinding period(s). 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.
SIP Waiting Time	Specify the sip wait stun time, you can input the time depended on your network condition.
Choose line to set info about SIP, There are 4 lines to choose. You can switch by 【Load】 button.	
Use STUN	Enable/Disable SIP STUN.
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.	

4.3.3.3. 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.

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:

Dial Peer Table

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

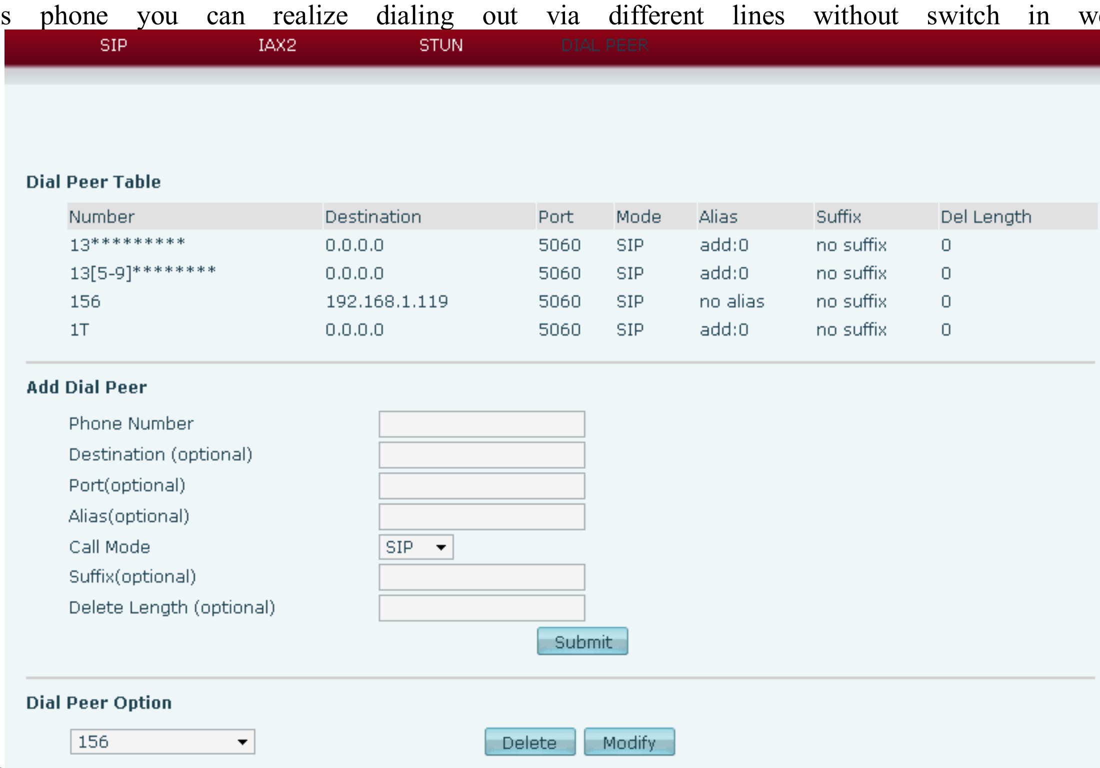
1、 x Match any single digit that is dialed.

If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

2、[] 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 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.

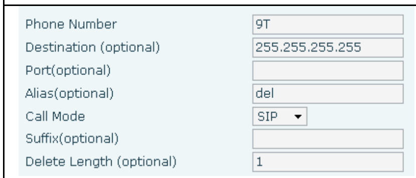
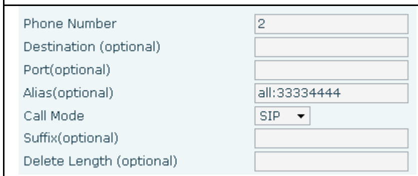
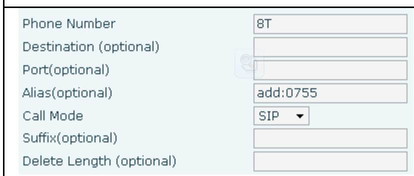
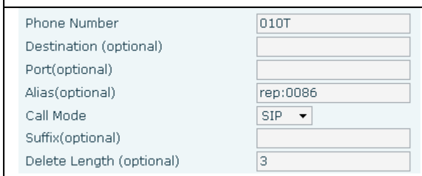
Use this phone you can realize dialing out via different lines without switch in web interface.

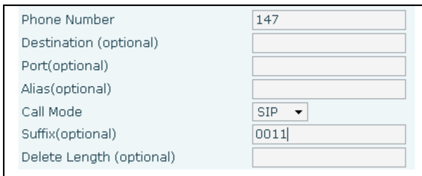


DIAL PEER	
Field name	**explanation**
Phone number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, you need input your desired phone number in this blank, and then you need dial the phone number to realize calling to what the phone 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 config 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 config item. If you don't set Alias, it will show no alias.
Note: 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.	
Call Mode	Select different signal protocol, SIP or IAX2
Suffix	Set suffix, this is optional config item. It will show no suffix if you don't set it.
Delete Length	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.

Examples of different alias application

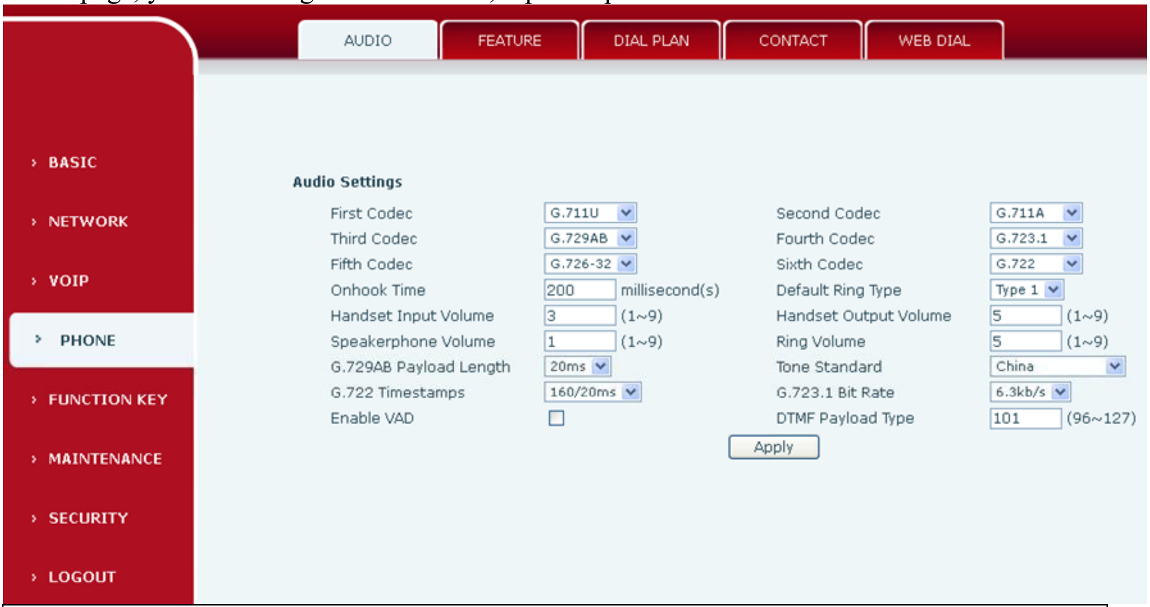
Set by web	explanation	example
	You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of 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”

4.3.4. Phone

4.3.4.1. AUDIO

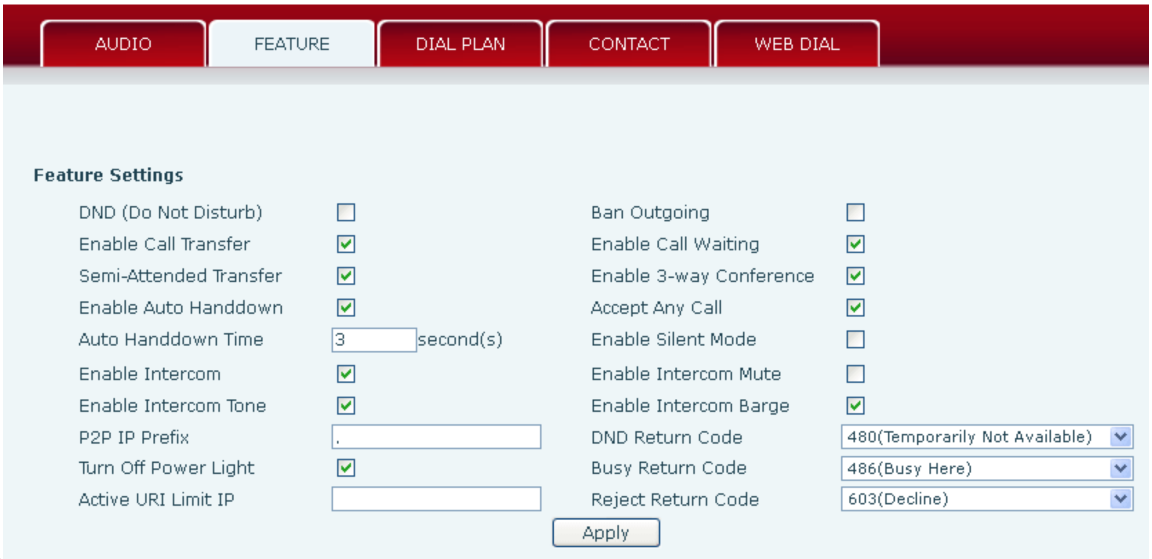
In this page, you can configure voice codec, input/output volume and so on.



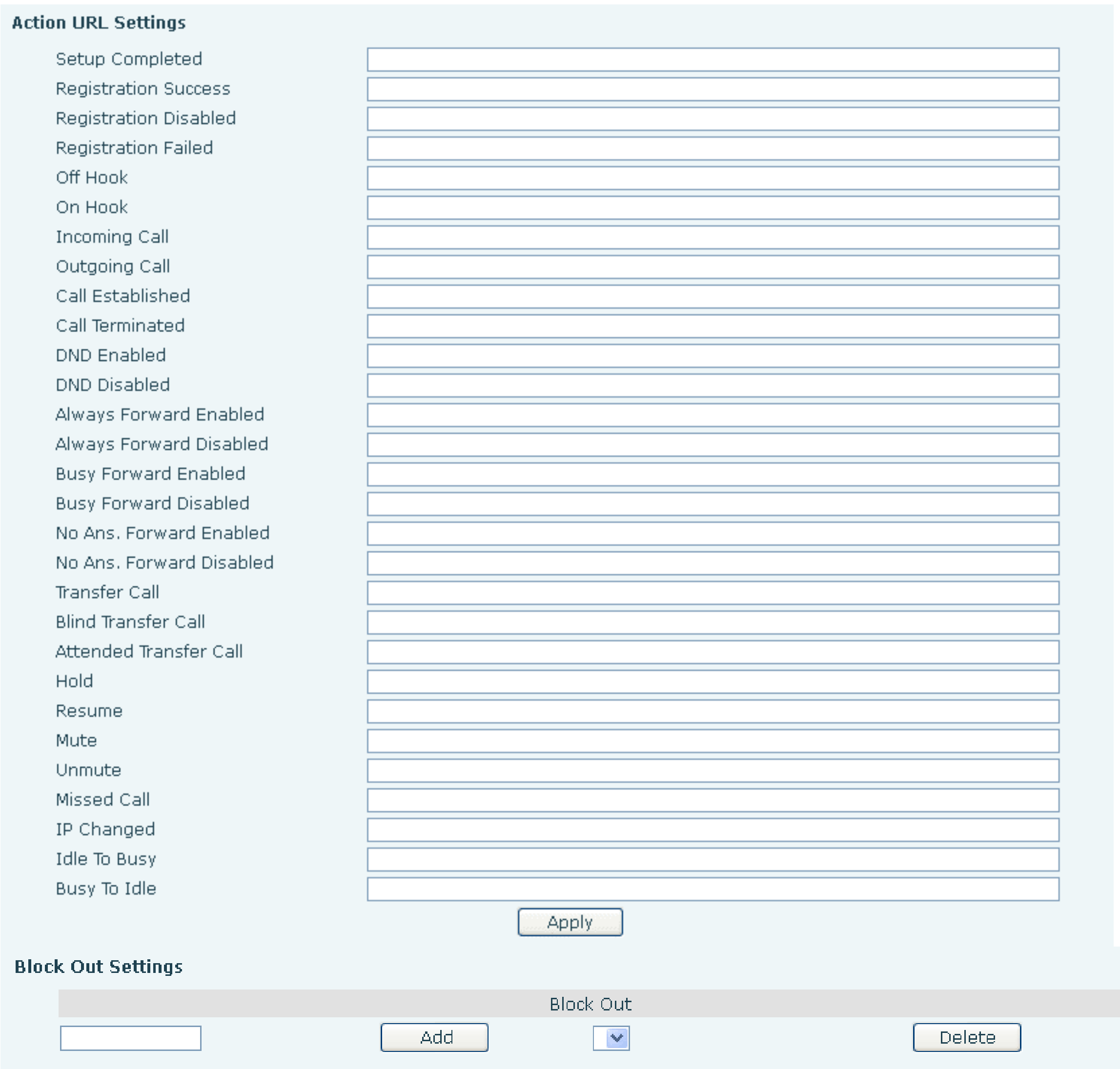
DSP Configuration	
Field name	**explanation**
First Codec	The fist preferential DSP codec: G.711A/u, G.722, G.723, G.729
Second Codec	The second preferential DSP codec: G.711A/u, G.722, G.723, G.729
Third Codec	The third preferential DSP codec: G.711A/u, G.722, G.723, G.729
Fourth Codec	The forth preferential DSP codec: G.711A/u, G.722, G.723, G.729
Fifth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723, G.729
Sixth codec	The sixth preferential DSP codec: G.711A/u, G.722, G.723, G.729
Handset Input Volume	Specify Input (MIC) Volume grade.;
Speakerphone Volume	Specify Hands-free Volume grade
G729 Payload Length	Set G729 Payload Length
Onhook Time	Specify the least reflection time of Hand down, the default is 200ms.
Default Ring Type	Select Ring Type
Handset Output Volume	Specify Output (receiver) Volume grade.
Speakerphone volume	Specify Speakerphone Volume grade.
Ring Volume	Specify Ring Volume grade
G722 Timestamps	160/20ms or 320/20ms is available
G723 Bit Rate	5.3kb/s or 6.3kb/s is available
Default Ring Type	Set up the ring by default
Tone Standard	Select Signal Standard.
Enable VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.

4.3.4.2. FEATURE

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



on.



Call Service	
Field name	**explanation**
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 cannot dial out any number.
Enable Call Transfer	Enable Call Transfer by selecting it.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable Call Waiting	Enable Call Waiting by selecting it. Then the phone reminds whether redial, when the callee is busy or rejects. if it's ok and the phone finds out that the callee is idle by sip message, it will reminds whether redial
Enable 3-way Conference	Enable 3-way conference by selecting it
Accept Any Call	If select it, the phone will accept the call even if the called number is not belong to the phone.
Enable Auto Handdown	The phone will hang up and return to the idle automatically at hands-free mode
Auto Handdown Time	Specify Auto Hand down Time, the phone will hang up and return to the idle automatically after Auto Hand down Time at hands-free mode, and play dial tone Auto Hand down Time at handset mode
Enable Silent Mode	Enable Mute Mode by selecting it, the phone light will red blink to remind that there is a missed call instead of playing ring tone
Enable Intercom	Enable Intercom Mode by selecting it
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call
Turn Off Power Light	Enable Turn Off Power Light by selecting it
DND Return Code	Specify DND Return code
Busy Return Code	Specify Busy Return Code
Reject Return Code	Specify Reject Return Code
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.
Active URI Limit IP	Specify the server IP that remote control phone forcorresponding operation.
Action URL Settings	Specify the Action URL that Record the operation of phone, send these corresponding information to server, url：http://InternalServer /FileName.xml?（InternalServer is server ip，FileName is name of xml that contains the action message ）
Block Out Settings	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 cannot dial out any phone number whose prefix is 001. X and are wildcardx 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.

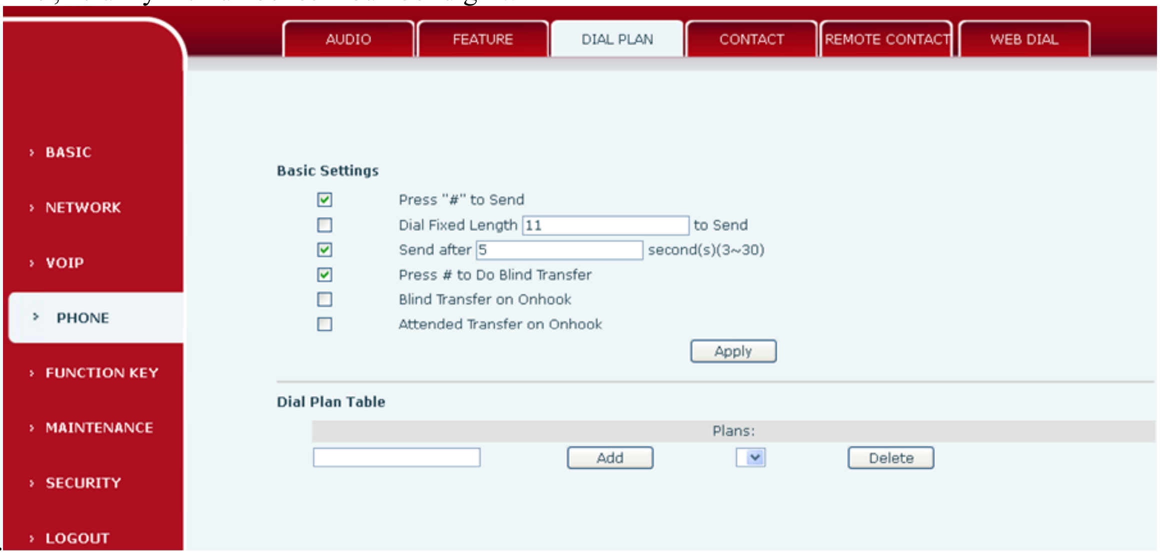
4.3.4.3. DIAL PLAN

This phone 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.

In order to keep some users' secondary dialing manner when dialing the external line with PBX, phone can be added a special rule to realize it. Souser can dial a number as external line prefix and get the secondary dial tone to keep dial the external number. After finishing dialing, phone will send the prefix and external number totally to their server.
For example, there is a rule 9,xxxxxxxx in the digital map table. After dialing 9, phone will send the secondary dial tone, user may keep going dialing. After finished, phone will call the number which starts with 9; actually the number sent out is 9-digit with



9.

Digital Map Configuration	
Field name	**explanation**
Press “#” to Send	Set Enable/Disable the phone ended with “#” dial.
Dial Fixed Length xx to Send	Specify the Fixed Length of phone ending with.
Send after xx second	Set the timeout of the last dial digit. The call will be sent after timeout.
Press # to Do Blind Transfer	Enable Blind Transfer OnHook, when executing Blind Transfer End with #, press # after inputting the number that you want to transfer, the phone will transfer the current call to the third party
Blind Transfer OnHook	Enable Blind Transfer OnHook, when executing Blind Transfer, hang up after inputting the number that you want to transfer, the phone will transfer the current call to the third party
Attend Transfer OnHook	Enable Attend Trans OnHook, when executing Attended Transfer, hang up after the third party answers, the phone will transfer the current call to the third party

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.
* 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.

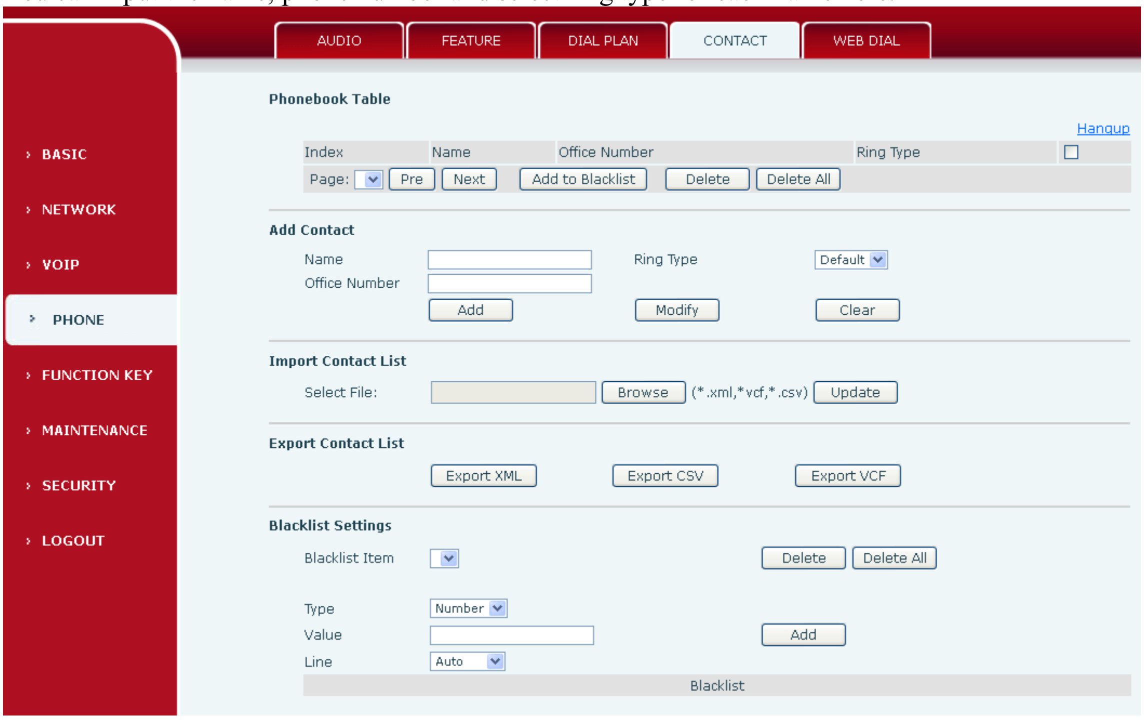
Plans:
"[1-8]XXX"
"9xxxxxxx"
"911"
"99T4"
"9911x.T4"

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.

4.3.4.4. CONTACT

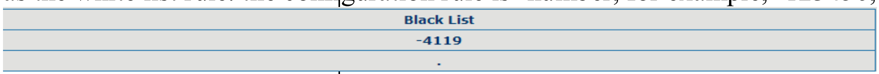
You can input the name, phone number and select ring type for each name here.



Phone Book	
Field name	**explanation**
Phonebook Tables	
Name	Shows the name corresponding to the phone number
Office Number	Shows the phone number
Ring Type	Shows the ring type of the incoming call.
Notice: the maximum capability of the phonebook is 500 items, you can select many or a contact to add to group and add to blacklist, and delete many or a contact, and delete all contacts.	
Add Contact	
Name	Shows the name corresponding to the phone number
Office Number	Shows the phone number
Ring Type	Shows the ring type of the incoming call.
Notice: the add button for adding a new contact, the modify button for modifying the added contact, the clear all button for clear all input information of the contact	
Import Contact List	
Select File	Click the browse button to select the phonebook file that you want to import, than click update button, the phonebook file selected will be

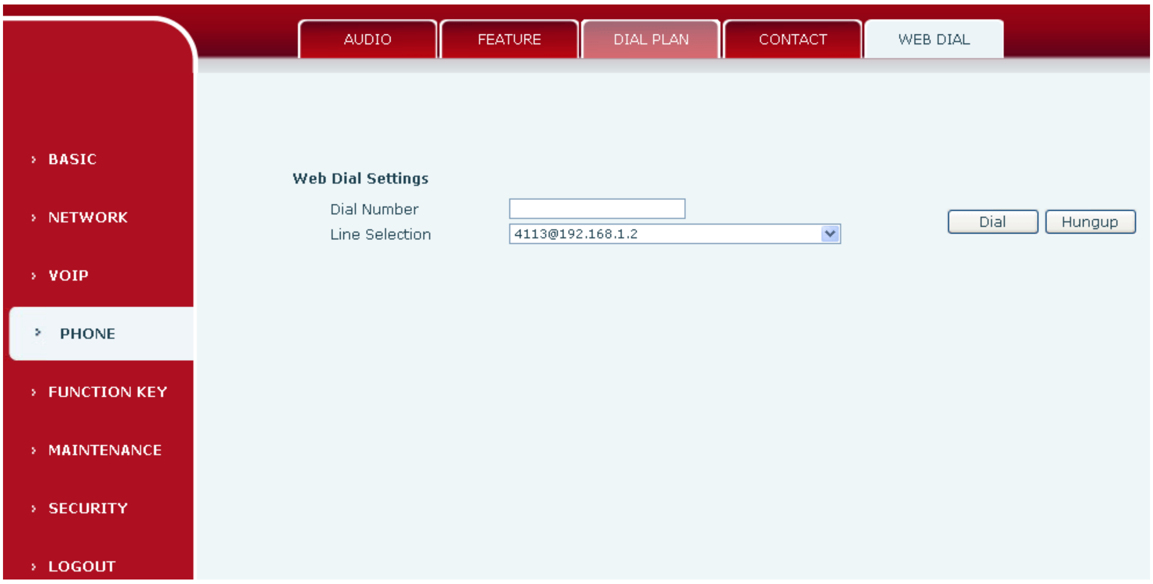
	added to the phone.
Export Contact File	
Export XML	Click export xml button to export phonebook file of xml model
Export CSV	Click export xml button to export phonebook file of csv model
Export VCF	Click export xml button to export phonebook file of vcf model
Blacklist Settings	
Type	Select the blacklist type, you can select number or prefix of number
Value	Input number or prefix of number
Line	Select the sip line

Notice: the add button for adding a new blacklist, the delete button for deleting one item, the delete all button for deleting all items.
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 be responded。
DOT (.) means matching any arbitrary number digit. for example, 6. expresses any number with prefix 6 will be forbidden to be responded。
If user wants to allow a number or a series of number incoming, he may add the number(s) to the list as the white list rule. the configuration rule is -number, for example, -123456, or -1234xx



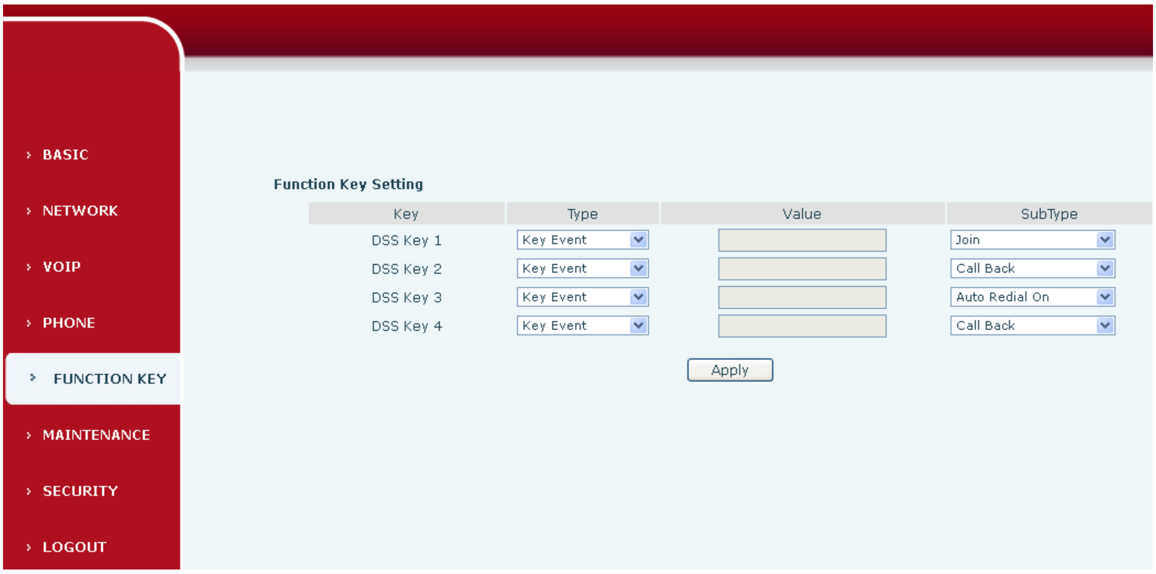
Means any incoming number is forbidden except for 4119
Note: End with DOT (.) when set up the white list

4.3.4.5. WEBDIAL



You can make a call through the WEB DIAL, enter the Dial Num then press Dial, if you want to finish the talk, press Hang-up.

4.3.5. FUNCTION KEY



The phone has 4 programmable keys which are able to set up to many functions per key. The following list shows the functions you can set on the programmable keys and provides a description for each function. The default configuration for each key is none which means the key hasn't been set for any functions.

1. Set the type as Memory Key

When the type is memory key, you can input number in value input, and phone will call the inputted number as pressing DSS key.

2. Set the type as Key Event

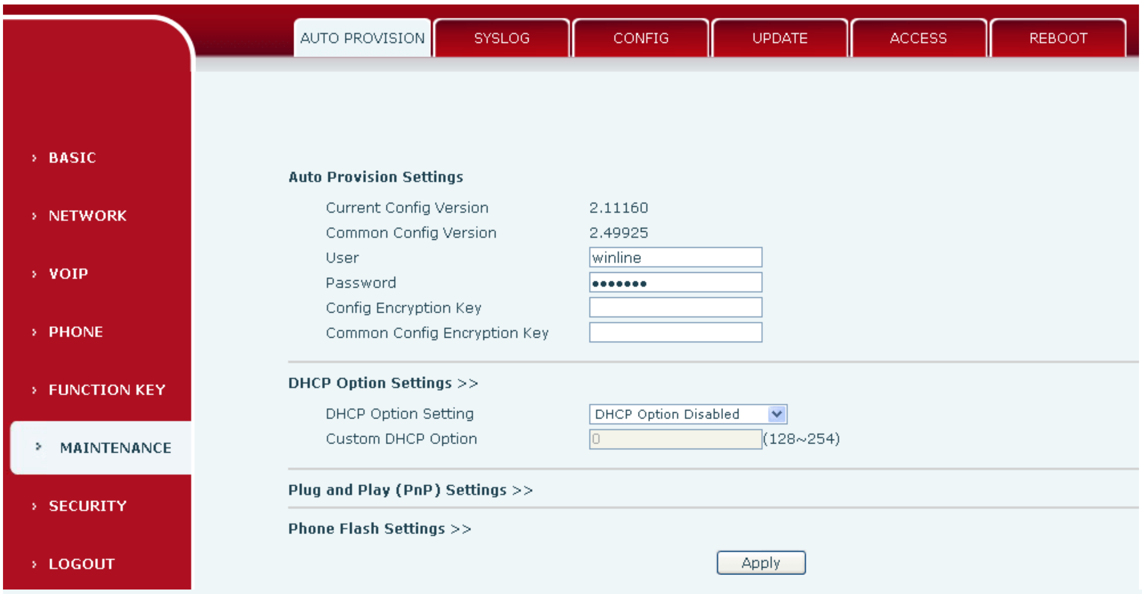
You can set these keys as Key Event, and the subtype have many options.

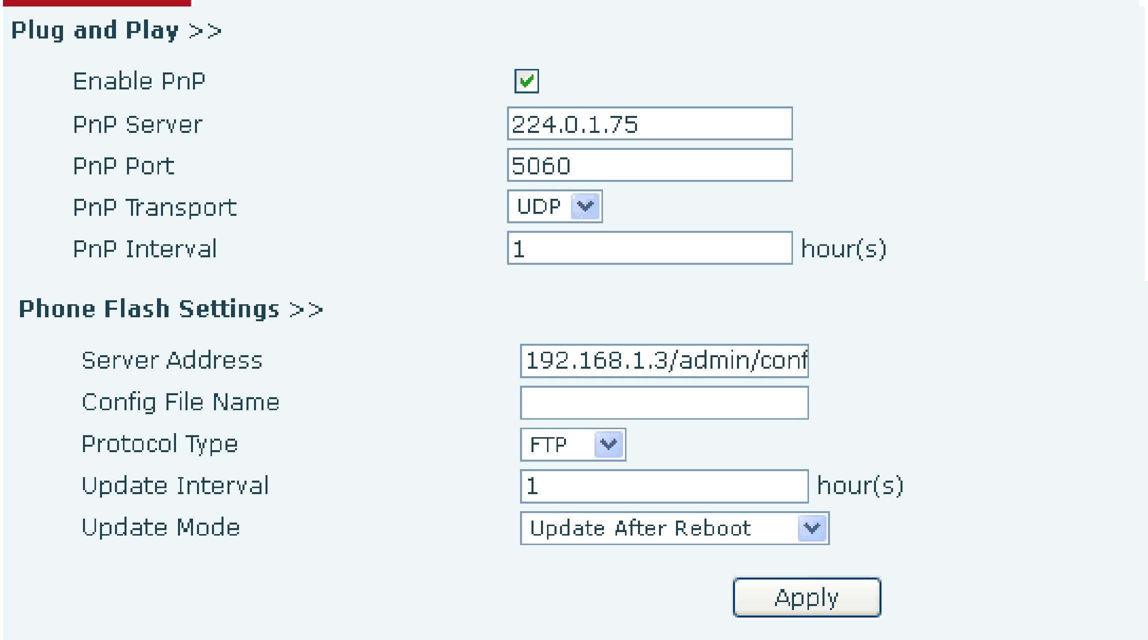
Choose one and it will have corresponding function.

- None
- MWI
- DND (Do Not Disable)
- Hold
- Transfer
- Phone Book
- Redial
- Autoredial on
- Auto redial off
- Call Forward
- History
- Flash
- Headset
- Call Back

4.3.6. Maintenance

4.3.5.1. Auto Provision





Fanvil endpoint supports PnP and DHCP and Phone Flash to obtain the parameters. The PnP and DHCP and Phone Flash are all deployed, endpoint will go by the following process to try to obtain the server address and other parameters, when it boots up:

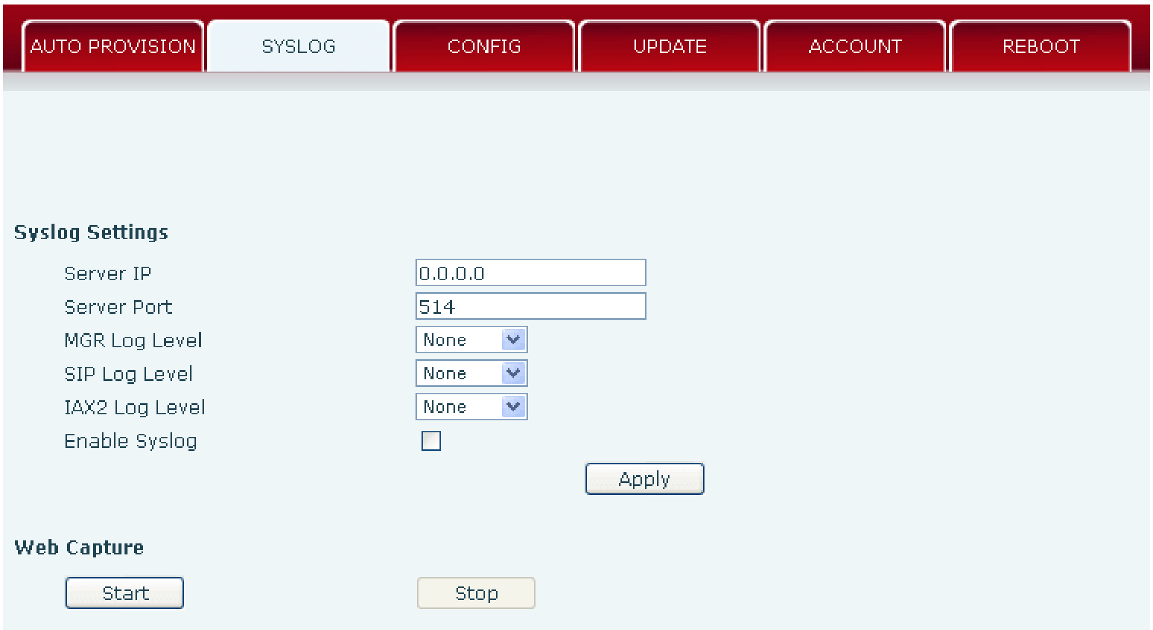
DHCP option→ PnP server →Phone Flash

Auto Provision	
Field name	**explanation**
Auto Provision Setting	
Current Config Version	Show the current config file's version. If the version of the configuration downloaded is higher than the version of the running configurations, the auto provision would upgrade, or stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
Common Config Version	Show the common config file's version. If the configuration downloaded and the running configurations are the same, the auto

	provision would stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
User	Set FTP/HTTP/HTTPS server Username. System will use anonymous if username keep blank.
Password	Set FTP/HTTP/HTTPS server Password.
Config Encrypt Key	Input the Encrypt Key, if the configuration file is encrypted.
Common Config Encrypt Key	Input the Common Encrypt Key, if the Common Configuration file is encrypted
DHCP Option Settings	
DHCP Option Setting	Specify DHCP Option. DHCP option supports DHCP custom option and DHCP option 66 and DHCP option 43 to obtain the parameters. You could choose one method among them，the default is DHCP option disable.
Custom DHCP Option	A valid Custom DHCP Option is from 128 to 254. The Custom DHCP Option must be in accordance with the one defined in the DHCP server.
Plug and Play Settings	
Enable PnP	Enable PnP by selecting it, than the phone will send SIP SUBSCRIBE messages to a multicast address when it boots up. Any SIP server understanding that message will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
PnP Server	Specify the PnP Server
PnP Port	Specify the PnP Server
PnP Transport	Specify the PnP Transfer protocol
PnP Interval	Specify the Interval time, unit is hour
Phone Flash Settings	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be IP address or Domain name with subdirectory.
Protocol Type	Specify the Protocol type FTP、TFTP or HTTP.
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.。
Update Interval	Specify 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.

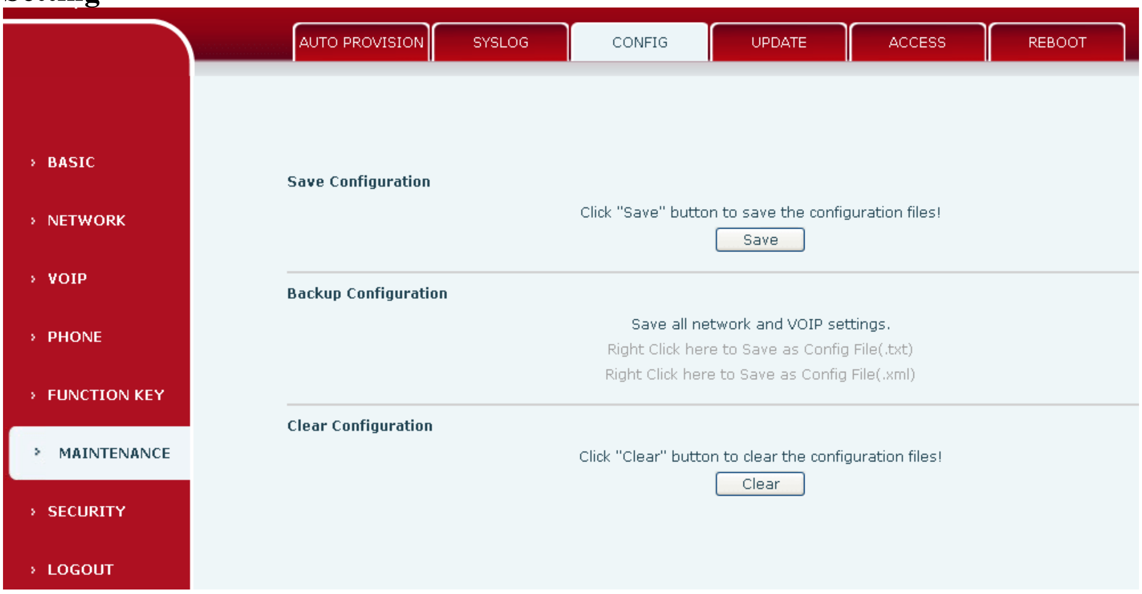
4.3.5.2. Syslog

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.
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. Output debugging information for R&D person.
At present, the lowest level of debug information send to Syslog is info; debug level only can be displayed on telnet.



Syslog Configuration	
Field name	**explanation**
Syslog Settings	
Server IP	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.
Web Capture	
Start	Click the start button when you need capture the WAN packet stream of the phone, then open or save the file as the interface
Stop	Click the end button to stop capturing the packet stream

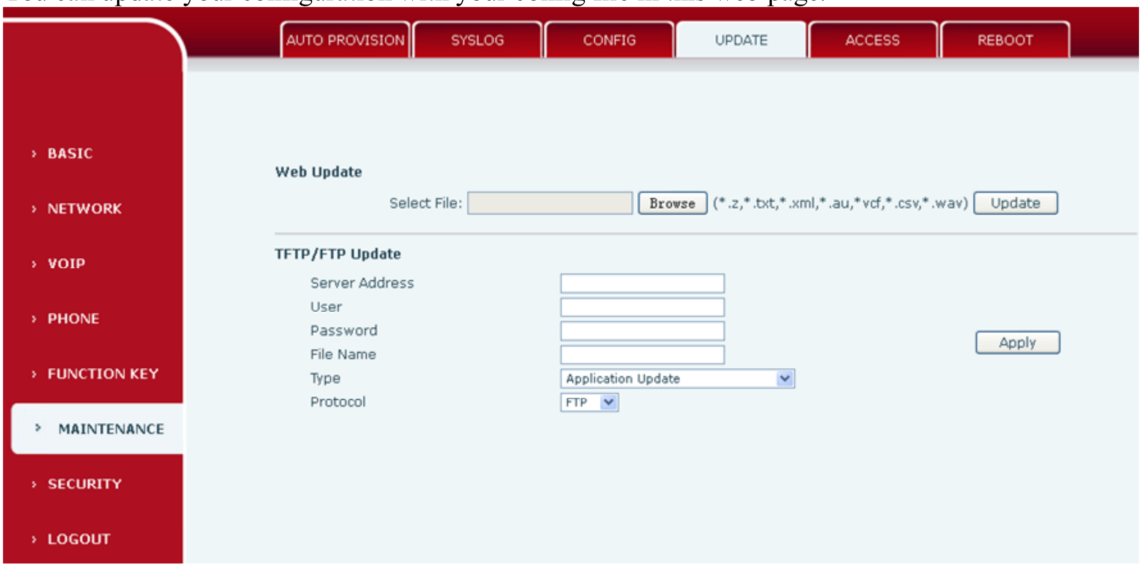
4.3.5.3. Config Setting



Config Setting	
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 Configuration	Right clicks on “Right click here…” and select “Save Target As config File(.txt)” then you will save the config file in .txt format, or select “Save Target As config File(.xml)” then you will save the config file in .xml format
Clear Configuration	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 (SIP1-4 and IAX2) and version number.

4.3.5.4. Update

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

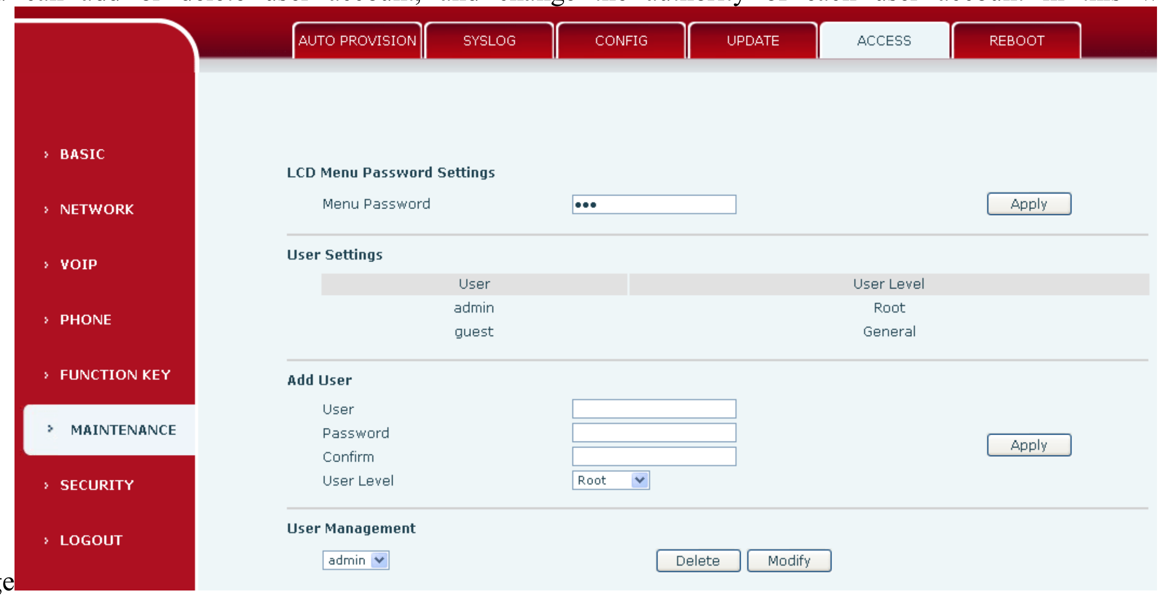


Update	
Field name	**explanation**
Web Update	Click the browse button, find out the config file saved before or provided by manufacturer, download it to the phone directly, press “Update” to save. You can also update downloaded update file, logo picture, ring, mmiset file by web.
Server Address	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
User	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 server. The configuration will be effective after the phone is reset. 4. Phone book export (.vcf, .csv, .xml): Upload the phonebook file to FTP/TFTP server, name and save it. 5. PhoneBook import (.vcf, .csv, .xml): Download the phonebook file to phone from FTP/TFTP server.
Protocol	Select FTP/TFTP server

4.3.5.5. Access

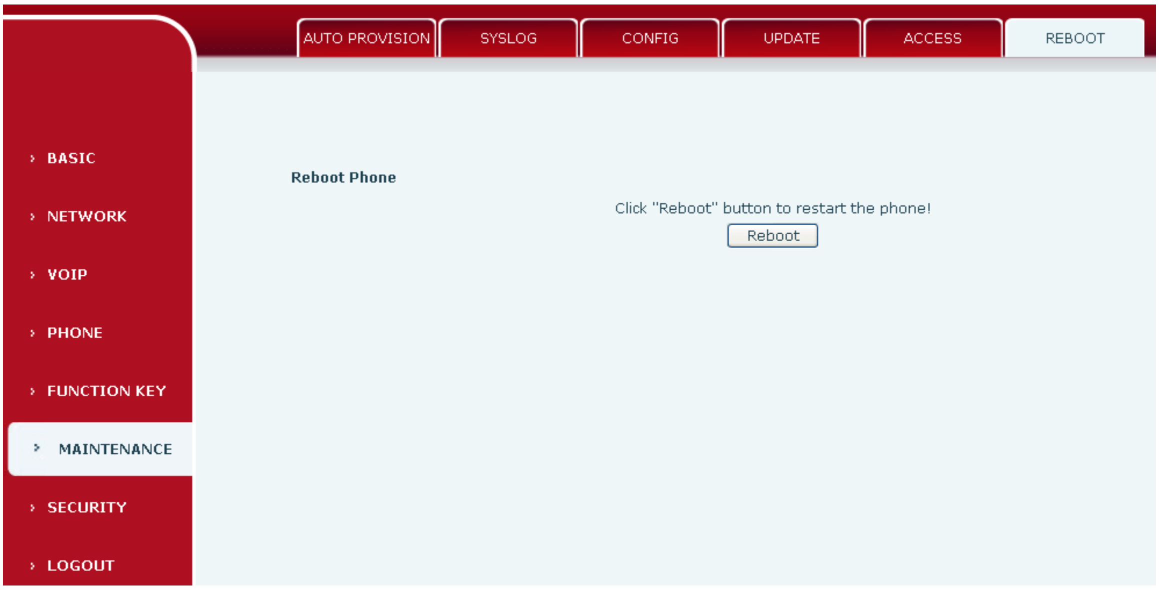
You can add or delete user account, and change the authority of each user account in this web page



Access Configuration	
Field name	**explanation**
LCD Menu Password Settings	Set the password for entering the setting menu of the phone by the phone's key board. The password is digit.

User Set	
User Name	User Level
admin	Root
guest	General
This table shows the current user existed.	
User	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. General user only can add the user whose level is General.	

4.3.5.6. Reboot

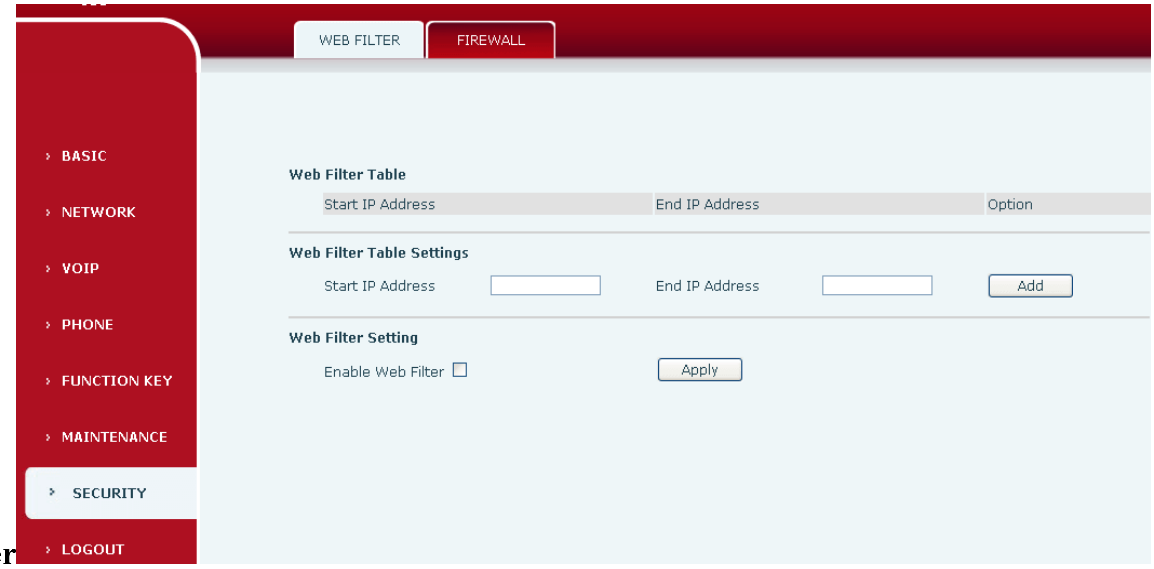


If you modified some configurations which need the phone's reboot to be effective, you need click the Reboot, then the phone will reboot immediately.

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

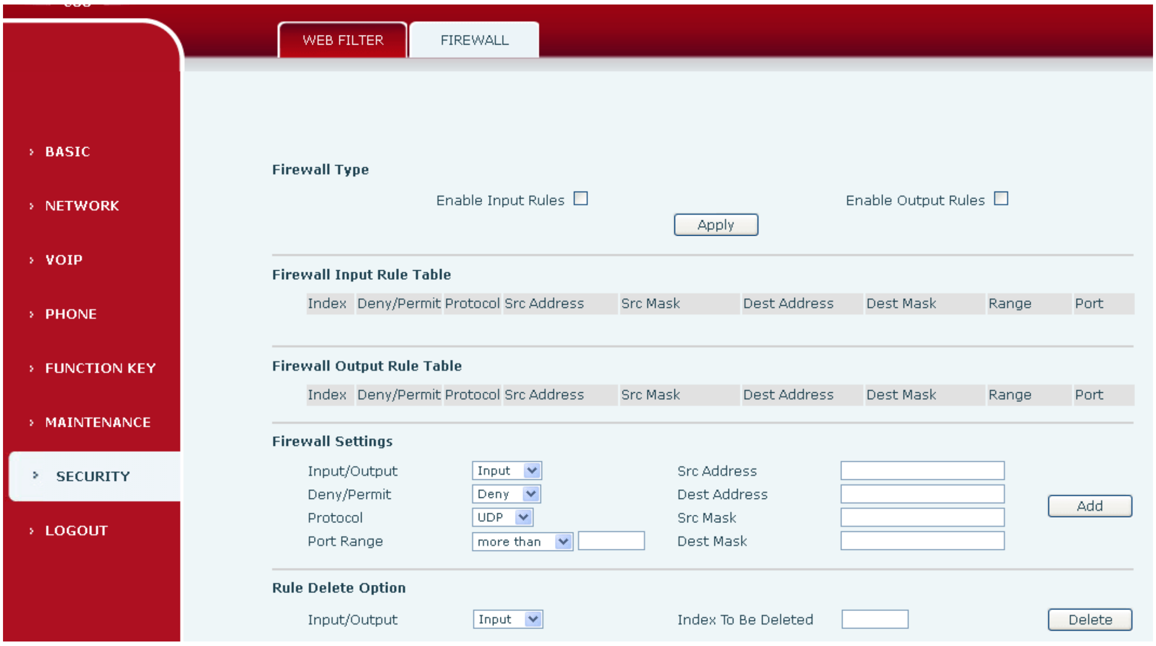
4.3.6. Security

4.3.6.1. MMI Filter



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 IP Table list:	
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.	
Enable Web Filter	Select it or not to enable or disable MMI Filter. Click **Apply** to make it effective.
Notice: Do not set your visiting IP outside the MMI filter range; otherwise, you can not logon through the web.	

4.3.6.2. Firewall

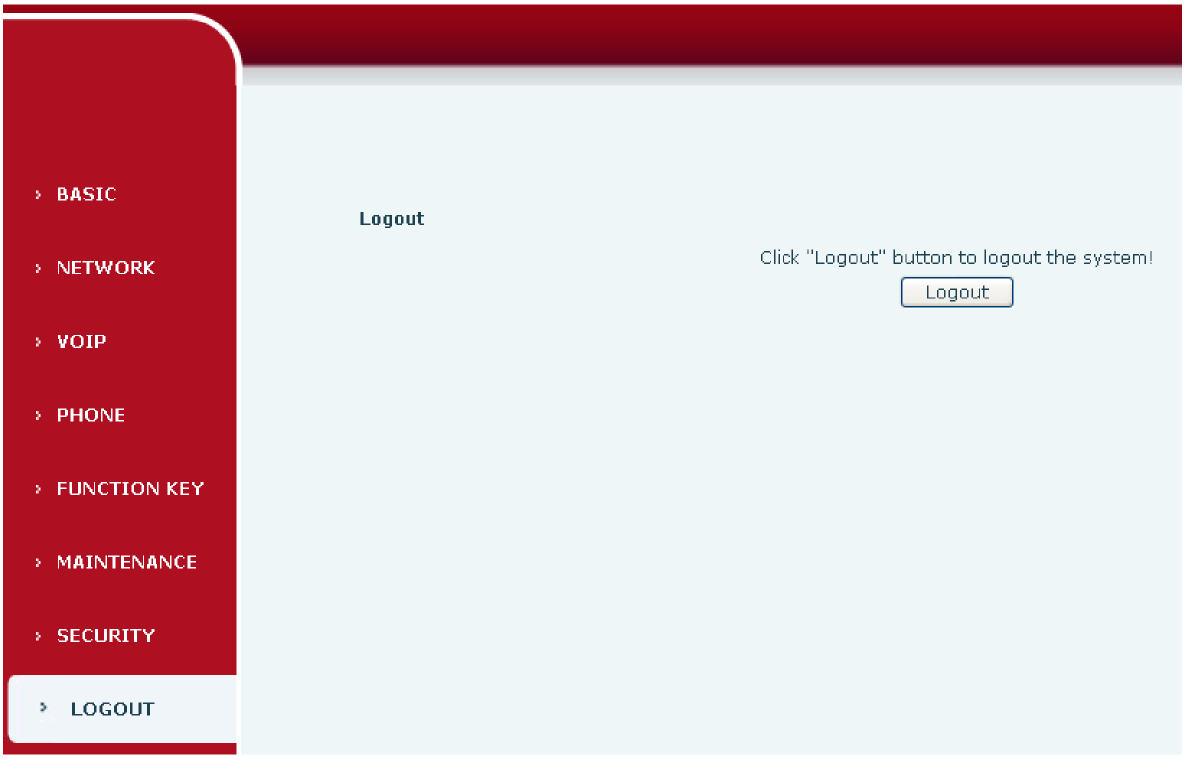


Firewall Configuration	
In this web interface, you can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule). Firewall supports two types 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**
Enable Input Rules	Select it to Enable in_ access rule
Enable Output Rules	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	Filter protocol type. You can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range
Src Address	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0
Dest Address	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

<table>
<tr><td></td><td>which network ID is C type.</td></tr>
<tr><td>Dest Mask</td><td>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.</td></tr>
<tr><td colspan="2">Click the Add button if you want to add a new output rule.

Firewall Input Rule Table
<table><tr><th>Index</th><th>Deny/Permit</th><th>Protocol</th><th>Src Address</th><th>Src Mask</th><th>Dest Address</th><th>Dest Mask</th><th>Range</th><th>Port</th></tr><tr><td>1</td><td>Deny</td><td>UDP</td><td>192.168.1.14</td><td>255.255.255.0</td><td>192.168.1.118</td><td>255.255.255.0</td><td>More than</td><td>1</td></tr></table>
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.</td></tr>
<tr><td colspan="2">Click the Delete button to delete the selected rule.</td></tr>
</table>

4.3.7. 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.

4.4. Settings via phone's keyboard.

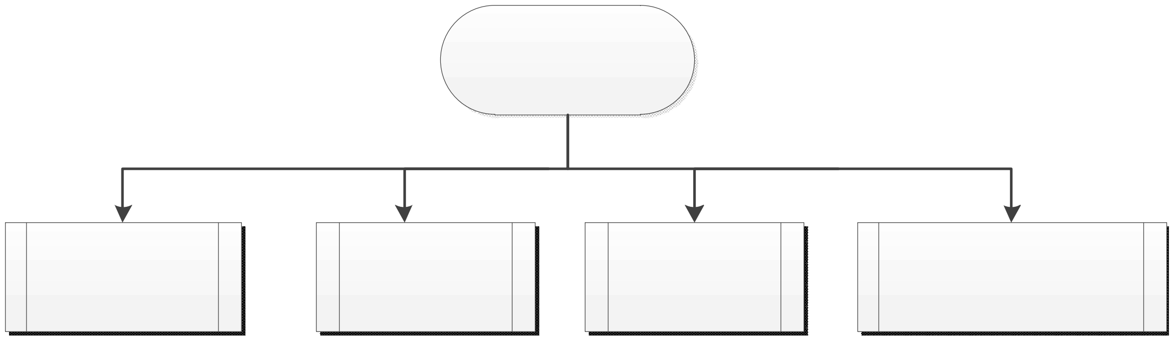
4.4.1. How to set via the phone's keyboard.

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.

4.4.2. 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, Auto MDIX
	LAN	10/100Base- T RJ-45 for PC, Auto MDIX
Power Consumption		Idle：1.5W/Active：1.8W
LCD size		74 x 28mm
Operation Temperature		0～40℃
Relative Humidity		10～65%
Main Chipset		broadcom voip chipset
SDRAM		8MB
Flash		2MB
Size（W x H x D）		20（18.5）x19.3cm
Weight		0.99kg

5.1.2. Voice Features

- Support 2 lines SIP and IAX2, SIP 2.0 (RFC3261)
- Codec：G.711A/u，G.7231 high/low，G.729, G.722,G.726
- Echo cancellation：Support G.168 and hand-free can support 96ms
- Support VAD，CNG
- NAT transverse: support STUN
- Supports full duplex.
- 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/ paging and intercom /redial/unredial.
- Call control features: Flexible dial map, support hotline, empty calling no. reject server, black list for reject, authenticated call, no disturb, caller ID and so on.

- Support phonebook 500 records, incoming calls / outgoing calls / missing calls. Each supports 100 records
- support conference call in server
- Could dial use private server automatically when public server unregistered while private server is registered successfully
- Phonebook supports VCard standard
- Support 12/24 time format.
- 12/24 hours time display
- Support daylight saving time
- Support path, gruu
- Support SIP Privacy.

5.1.3. Network Features

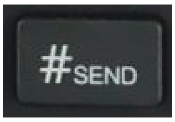
- WAN: support Bridge
- Support PPPoE for xDSL
- support VLAN
- Support Stun penetration
- Support DHCP get IP on WAN port
- Qos supports Diffserv.
- support network tools: contain ping，trace route，telnet client

5.1.4. Maintenance and Management

- The phone supports post mode, can update firmware by post mode.
- Supports different levels of administration.
- 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. Digit-character map table

Button	Character	Button	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		#