



User Manual

IP0027 VoIP Phone

Version 1.6.81.69



Safety Notices












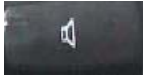

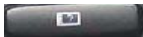




Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- 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°C 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.

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Key	Key name	Function Description
	menu	In idle state, press the MENU key to call up the menu.
	Up/down of navigation	In the IDLE mode, continuously thrice press to display WAN Port IP, gateway's IP and the phone number registered on public server; In the keypad configure mode, go back to previous item/Go forward to sub-item.
	Left/right of navigation	In the keypad configure mode, choose the item on the menu; In the idle mode, show SIP Phone number.
	Directory	Access to phone book.
	Soft1	Select the relevant function on the LCD, pressing group function keys to make up /down /enter /edit /quit /save /record /next /N'pad /dial /delete /IME and so on.
	Soft2	
	Soft3	
	Call records	Check the Income/Outgoing/Missed calls records.
	Exit	During talking, stop record and other operation and return to talking; it also can be used to stop call. In menu configuration mode, press the key to return to stand-by mode. Note: DO NOT press Esc during the configuration process, because phone will not save the configuration modified and return to stand-by status after pressing Esc
	Earphone	Use earphone to receive and make a call
	redial	In the hook off /hands-free /earphone mode, use redial to dial the last call number; In the idle mode, use redial to check the dialed list records
	Handfree	Enter into hands-free mode.
	network	Show WAN IP/gateway IP
	SIP accounts info	Check multi-lines number/server name/IP/status in idle mode. It could be used to enhance volume during talking.
	hold	Temporarily hold the active call during the talking; press the key again might unhold the call. (please refer to 4.4- call hold for more details)
	transfer	Press the key to make the second call out when there have been a active call. If there are two calls made, press the key to transfer one call to the other side. After transfer, phone will be hanged up (please refer to 4.3-call transfer for more details).
	mute	Mute an active call to make the other side can not hear you.
	volume	Adjust voice volume when hookoff or during the talking or adjust ring volume when there is call coming; it can also be modified in the idle mode



Programmable
key

There are 3 kinds of functions configured through web:

- 1、 Memory Key to store number for speed dial.
- 2、 SIP Line key to show SIP account information and status.
- 3、 Command Button key, when SIP1 line is available, there will be sent notify; if SIP1 is unavailable, there will be no notify sent

1.4. Port for connecting



Port name	attribute	description
LAN	Network interface	10/100M Connect it to PC
WAN	Network interface	10/100M Connect it to Network
DC 9V	Power port	5V/1A

IP0027 is provided with two Ethernet cables and a power adaptor. Phone has two interfaces: WAN and LAN. Please refer to safety notes of this manual carefully before power adaptor is connected.

2. Initial connecting and Setting

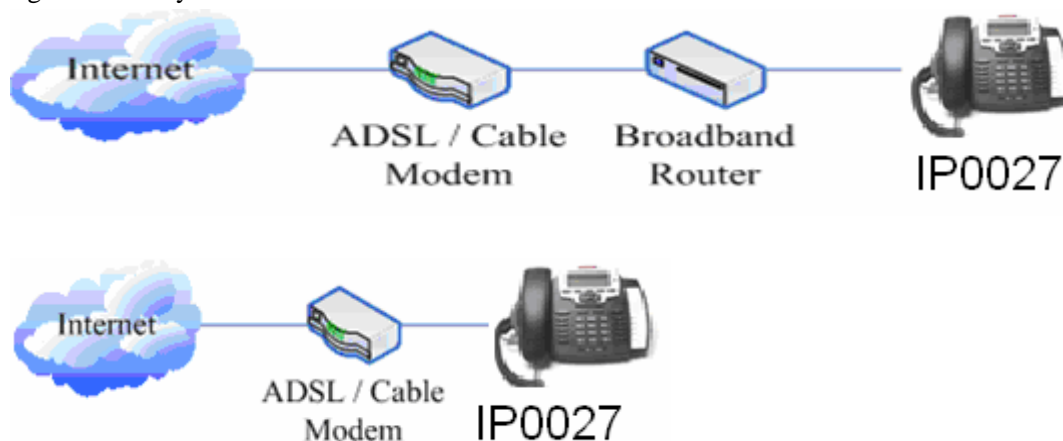
2.1. connect the phone

2.1.1. Connect to network

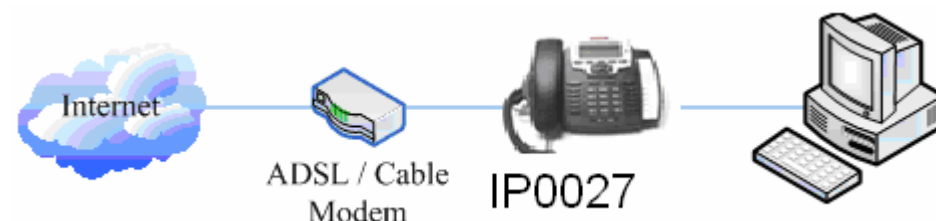
Step 1: Connect the IP Phone to the corporate IP telephony network. Before you connect the phone to the network, please check if your network can work normally.

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 9V 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.

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.

2.1.2. Power adaptor connection

Make sure that the power you use is comply with the parameters of power adaptor.

1. Plug power adaptor to power socket.
2. Plug power adaptor’s DC output to the power port of IP0027 to start up.
3. There will be displayed black line and “wait logon” on the screen. After finishing startup, phone will show greeting, current date and time and so forth.
4. If phone has registered to the server, you can place or answer calls.





2.2. Basic Initialization





IP0027 is provided with a plenty of functions and parameters for configuration. User needs some network and VoIP knowledge so that user could understand the meanings of parameters. In order to make user use the phone more easily and convenient, there are basic configurations introduced which is mandatory to ensure phone calls.

2.2.1. Network settings





Make sure that network is connected already before setting network of phone. IP0027 uses DHCP to get WAN IP configurations. So phone could access to network as long as there is DHCP server in it. If there is no DHCP server available, phone has to be changed WAN network setting to Static IP or PPPoE.


Setting PPPoE mode (for ADSL connection)


1. Get PPPoE account and password first.
2. Pressing  , and pressing  five times, screen shows “**06 Advanced Sublist**”. Now then pressing Soft2(Enter), LCD screen will display “**Enter Password**”.
3. Input password (default is 123) and pressing Soft2(Enter), then pressing  twice, now screen shows “**03 Network Sublist**”.
4. Pressing Soft2(Enter), and pressing  twice, there is **03 PPPoE Set Sublist** shown on the screen. After pressing Soft2(Enter) again, screen displays “**01 Account user123**”; pressing Soft2(Edit), and then pressing Soft1(Del) to delete, you can input PPPoE’s account and press Soft2 (Save). With “saved” displayed, screen will jump to “**01 Account**” afterwards.

5. Pressing  to show “**02 Password**”, and press Soft2 (Edit), input PPPoE’s password and then pressing Soft2(Save), With “Saved” displayed, screen will jump to “**02 Password**” afterwards.
6. Pressing Soft3 (Quit) to quit and pressing  to show “**01 Net Mode**”. Pressing Soft2(Edit)again, and pressing  also, screen jump to “<>**PPPoE**”. With Soft2(Save) pressed again, screen will show “Saved” and then jump to show “**01 Net Mode**”.
7. pressing Soft3 (Quit) thrice to quit to stand-by status and pressing  to shows “**PPPoE**”, phone tries to connect server to get IP. If there is shown “**Negotiating...**”, it shows that the phone is trying to access the PPPoE Server, else it shows that the phone has already get IP with PPPoE.







Setting Static IP mode (static ADSL/Cable, or none PPPoE / DHCP network)

1. Prepare the network’s parameters first. IP Address, Netmask, Default Gateway and DNS server IP address are needed. Please contact the service provider or technician of network.
2. Pressing , and  five times, screen shows “**06 Advanced Sublist**”, then pressing Soft2(Enter), screen will show “**Enter Password:**”.
3. Input password (default is 123) and pressing Soft2(Enter), then pressing  twice, now screen shows “**03 Network Sublist**”.
4. Pressing Soft2 (Enter), then pressing ,screen shows “**02 Static Set Sublist**”. Pressing Soft2(Enter) to make screen show “**01 IP**”, you need press Soft2(Edit), and Soft1(Del) to delete old parameter. Input your IP address and press Soft2 (Save). After “Saved” shown, screen will jump to “**01 IP**”.
5. Press  to show “**02 Netmask**”. Press Soft2(Edit), and then use Soft1(Del) to delete. Input your netmask and press Soft2 (Save). After “Saved” shown, screen will jump to “**02 Netmask**”.
6. Press  to show “**03 Gateway**”. Press Soft2(Edit), and then use Soft1(Del)to delete, Input your gateway and press Soft2(Save). After “Saved” shown, screen will jump to “**03 Gateway**”.
7. Press  to show “**04 DNS**”. Press Soft2(Edit), and use Soft1(Del) to delete. Input your DNS server address and press Soft2 (Save). After “Saved” shown, screen will jump to “**04 DNS**”.
8. Press Soft3 (Quit) quitting. With  pressed, screen shows “**01 Net Mode**”.

Pressing Soft2(Edit) and , screen shows "<>Static"; with Soft2(Save) pressed, screen shows "Saved" and then shows "**01 Net Mode**".

9. Press Soft3 (Quit) thrice to quit to stand-by status. Press  to show "**Static**". If screen shows the IP address and gateway which are set just now, it shows that Static IP mode is taken effect.

Setting DHCP mode

1. Press  and  five times, screen will show "**06 Advanced Sublist**". Then press Soft2(Enter), screen will show "**Enter Password:**".
2. Input password (default is 123) and press Soft2(Enter), and press  twice, then screen displays "**03 Network Sublist**".
3. Press Soft2 (Enter) to show "**01 Net Mode**". After pressing Soft2(Edit), using   to select until screen shows "<>DHCP". Press Soft2(Save), With "saved" displayed, screen will jump to "**01 Net Mode**".
4. Press Soft3 (Quit) thrice quitting to stand-by status. Press  to show "**DHCP**", if there is "**Negotiating...**" shown on screen, it shows that phone is keep trying to search DHCP server or get IP; If there is IP address displayed, it shows that DHCP mode has been taken effect.

3. IP0027's basic operation

3.1. Answer calls

IP0027 will ring to indicate you when there is call incoming, below is ways to answer call:

- Answer with hook off


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

- Answer with handfree

Press  to begin talking. Press  again to finish talk.

- Using handfree instead of handset during a talk

Press  and hook on the handset when you use handset to speak and

want to change to use handfree to speak. Press  again to finish talk.

- Using handset instead of handfree during a talk

Hook off the handset when you want to use handfree to speak and want to change to use handset. Just hook on to finish talk.

3.2. Place calls


- Use handset

Hook off (screen will show the current using line, or you could use programmable key 1 to 5 to select), after getting dialing tone, you could begin to dial number.

After finishing it, press # and IP0027 will send the number and call the number.


When you hear a ringback tone and screen shows the callee's number, it shows that the person you called is ringing. If callee answers the call, you can begin to talk and your phone will keep showing callee's number and counting time. Just hang up to finish talk.




- Use handfree

Press  (screen will show the current using line, or you could use programmable key 1 to 5 to select), after getting dialing tone, you could begin to dial number. After finishing it, press # and IP0027 will send the number and call the number. When you hear a ringback tone and screen shows the callee's number, it shows that the person you called is ringing. If callee answers the call, you can begin to talk and your phone will keep showing callee's number and counting time.

Press  again to finish talk.

- Use directory

press  in stand-by mode, and then press Soft2(Enter), you will access to


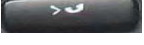

phonebook. If there are many persons records stored in the directory, you can use  to search the person which you want to contact. Press  to forward, and press  to backward. Press Soft2 (Dial) to dial the current number shown on the screen.

- Direct dial

Direct dial means user can make calls directly without hook off or using handfree. User can dial number in stand-by mode and then press Soft2 (Dial) to call. User also can press Soft3 (Save) to save the number to directory. In this way, user can not use #/time out or fix length to collect digits and dial.

- Multi-line calls

IP0027 supports 5 SIP lines max, that is user could use 5 SIP accounts to register and make calls. User could use programmable key as SIP line key. When a line key is pressed, phone will use the server to call. System will use SIP 1 as default line to call.


There are most two calls at the same time. Screen will display the incoming call number when user is keep talking. You can press  or Soft1 (Accept) to accept it, and hold the first one (if you want to use this function, you need enable Call Waiting of the phone first). Use  to switch the two calls to talk. User can also use  to make the second call when there is just a active call.

3.3. End calls

- Hang up with handset hook on


Hook on to finish talking.

- Hang up with handfree

Press  to finish talk when phone is in handfree status.

Note: user can not finish talk by pressing  if phone is used handset to talk.

- Hang up a active call with 2 calls

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

Note: it is no use to press # to finish talk, if there is only one current call.



3.4. Call transfer

- Blind Transfer

During talk, press Soft2 (Transf), and then dial the number that you want to transfer to, and press #. Phone will transfer the current call to the third party. After finishing transfer, the call you talk to will be hanged up. User can not select SIP


line when phone transfers call.

- **Attended Transfer**

During talk, press  and dial the number that you want to transfer to. After that person answers, you can press Soft2 (Transf) to do transfer (you need enable call waiting and call transfer first). If there are two calls, you can just talk to one, and keep hold to the other one. The one who is keep hold can not speak to you or hear from you. In this status, user can press * or Soft3 (Conf) to make calls mode in conference mode. If user wants to stop conference, user can press  (User must enable call waiting and three way call first).

Note: the server that user uses must support RFC3515 or it might not be used.




3.5. Call hold

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





3.6. Call records

IP0027 supports 100 items of missed call, 100 items of incoming call, and 100 items of dialed call. If the records are full, the newest will replace the oldest. If phone's power cut or reboot, call records will be discarded.

- **Missed call**




Press  or , and screen will show “**Missed Call List**”. Press Soft2 (Enter), phone will show the number and time of missed call. User can also use  to browse the missed call records, or use Soft2 (Dial) to dial the number. If there is no missed call, screen will show “**List Is Empty**”.

- **Incoming call**

Press  and use  to select “**Incoming Call**”. Press Soft2 (Enter), phone will show the number and time of incoming call. User can also use  to browse the incoming call records; or user can press Soft2(Dial) to ready dial, and press Soft2(Call) again to send the number. If there is no incoming call, screen will show “**List Is Empty**”. User can press  twice directly in stand-by mode to check “**Incoming Call**” too.

- **Dialed call**

Press , and use  to select to “**Outgoing Call**”. Press



Soft2 (Enter), phone will show the number and time of dialed call. User can also use  to browse the dialed call records; or user can press Soft2 (Dial) to ready dial, and press Soft2 (Call) again to send the number. If there is no dialed call, screen will show “**List Is Empty**”. User can press  thrice or press  to check “**Outgoing Call**”.

3.7. Record during talking

The most difference between IP0027 and other IP phones is the function of voice record. It will be introduced below.

- voice memo

Taking handset and then press Soft3 (Rec) to choose record voice memo. The max record memo time is 50 seconds. It shows as final count down. If record time is over 50 seconds, phone will ask you to save the record or restart a new record.

User can choose user's record as prompt voice: enter menu, press  twice until screen shows “**08 Voice Memo**”, and then press Soft2 (Enter) and use  to select. Press Soft1 (Save) and then quit.

- Voice record during talking

User can make a record during a call. Press Soft3 (Rec) when user is talking, phone will start recording the content of talking. Press Soft3 (Stop) to stop, and there is new record indication on the screen. User may go to check the new record to listen it. Each max time of record is 5 minutes 27 seconds. 15 items record is supported abstractly. If your single record is near 5 minutes, you will not record 15 items max. Maybe less than 15.

- Read record

Press Soft3 (Rec) in stand-by mode to enter voice records, and enter new records to check the newest one. Press Soft2 (Enter) and select the record item you want to get, then press Soft2 (Play) to play.

Note: if user press ESC key during talking and recording, both call and record will be terminated and phone will return to stand-by mode. And the content has been saved and shown on the screen there is new record. While if user press ESC key during voice memo, record has not been saved.

3.8. Special function key

- Programmable key

If programmable key is set as SIP Line key, user can select the key to choose

account to make calls. Note that only the key which is registered is available to be select to call.

- N'Pad

During talking, Soft1 (N'Pad) can be used to record a temporary message. After finishing input, press Soft3 (Save) to save it. After call is over, press Soft1 (N'Pad) in stand-by mode, phone will show the message just saved. In general it is phone number. User can press Soft3 (Clear) to clear message, or use N'pad new message to cover the old one. If it needs to be stored in directory, press Soft2 (Save) to edit and save as directory format.

- Shortcut key of DND

Press Soft2 (DND) in stand-by mode, phone will show "DND" on the screen. And phone will enable do not disturb. If you want to cancel DND, Press Soft2 (DND) in stand-by mode again.

4. Web configuration

4.1. Introduction of configuration

4.1.1. Ways to configure

IP0027 has three different ways to different users.

- Use phone keypad.
- Use web browser (recommendatory way) .
- Use telnet with CLI command.

4.1.2. Password Configuration

There are two levels to access to phone: root level and general level. User with root level can browse and set all configuration parameters, while user with general level can set all configuration parameters except SIP (1-5) or IAX2's that some parameters can not be changed, such as server address and port. User will has different access level with different username and password.

- Default user with general level:
 - ◆ username: guest
 - ◆ password: guest
- Default user with root level:
 - ◆ username: admin
 - ◆ password: admin

The default password of phone screen menu is 123.

4.2. Setting via web browser

When this phone and your PC are connected to your network, open the web browser and then 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/>). After that, browser will show you the logon page. Input username and password with click **Logon** button.



WAN IP may be gotten via pressing  or  to be shown on the screen.

4.3. Configuration via WEB

4.3.1. BASIC

4.3.1.1. Status

BASIC				
STATUS		WIZARD	CALL LOG	MMI SET
Network				
WAN		LAN		
Connect Mode	DHCP	IP Address	192.168.10.1	
MAC Address	00:12:f7:a0:00:54	DHCP Server	ON	
IP Address	192.168.1.14			
Gateway	192.168.1.1			
Phone Number				
SIP LINE 1	@ :5060	Unapplied		
SIP LINE 2	@ :5060	Unapplied		
SIP LINE 3	@ :5060	Unapplied		
SIP LINE 4	@ :5060	Unapplied		
SIP LINE 5	@ :5060	Unapplied		
Version: VOIP PHONE V1.6.81.69 Feb 29 2008 13:28:59				

Status	
Field name	Explanation
Network	Shows the configuration information on WAN and LAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port and LAN port, ON or OFF of DHCP mode of LAN port.
Phone Number	Shows the phone numbers provided by the SIP LINE 1-5 servers. The last line shows the version number and issued date.

4.3.1.2. Wizard

BASIC				
STATUS		WIZARD	CALL LOG	MMI SET
Network Mode Select				
Static IP MODE	<input checked="" type="radio"/>			
DHCP MODE	<input type="radio"/>			
PPPoE MODE	<input type="radio"/>			
BACK		NEXT		

Wizard	
Field Name	Explanation

Static IP MODE	<input checked="" type="radio"/>
DHCP MODE	<input type="radio"/>
PPPoE MODE	<input type="radio"/>

Please select the proper network mode according to the network condition. IP0027 provide three different network settings:

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

You can also refer to 3.2.1 Network setting to speed setting your network.

Choose Static IP MODE, click **【NEXT】** can config the network and SIP(default SIP1) simply, also can browse too. Click **【BACK】** can return to the last page.

Static IP Set	
Static IP Address	192.168.1.179
Netmask	255.255.255.0
Gateway	192.168.1.1
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.68

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

SIMPLE SIP SET	
Display Name	
Server Address	192.168.1.2
Server Port	5060
User Name	2113
Password	••••
Phone Number	2113
Enable Register	<input checked="" type="checkbox"/>

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

WAN	
Connect Mode	Static
Static IP Address	192.168.1.179
Gateway	192.168.1.1
SIP	
Register Server	192.168.1.2
Account/User Name	2113
PhoneNumber	2113
Register	ON
<input type="button" value="BACK"/> <input type="button" value="Finish"/>	

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

PPPOE Set	
PPPOE Server	ANY
Username	user123
Password	*****

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

Notice: Click 【Finish】 button after finished 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 query all the outgoing through this page.

BASIC		
STATUS	WIZARD	CALL LOG
MMI SET		
Call information		
Start Time	Last Time	Called Number
FEB 27 17:14	2	sip:2119@1
FEB 27 16:53	7	sip:2117@1
Call Log		
Field name	explanation	
Start Time	Display the start time of the outgoing record.	
Last Time	Display the conversation time of the outgoing record.	
Called Number	Display the account/protocol/line of the outgoing record.	
Notice: It will cover existing automatically if the call log table has the new record.		

4.3.1.4. MMI SET

BASIC

STATUS
WIZARD
CALL LOG
MMI SET

LANGUAGE SELECTION

Language Set: English

Time Set

Time Zone: (GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi

Daylight:

Greeting Message Set

Greeting Message: VOIP PHONE

APPLY

Version: VOIP PHONE V1.6.81.69 Feb 29 2008 13:28:59

MMI SET	
Field name	explanation
Language Set	Set the language of phone, English is default.
Time Zone	Select the Time zone according to your location.
Daylight	If your time zone supports daylight, you can select it.
Greeting Message	The greeting message will display on lcd when phone is idle. It can support 16 chars. the default chars are VOIP PHONE.

4.3.2. Network

4.3.2.1. WAN Config

NETWORK

WAN
LAN
QOS
SERIVCE PORT
DHCP SERVER
SNTP

WAN Status

Active IP	192.168.1.14
Current Netmask	255.255.255.0
MAC Address	00:12:f7:a0:00:54
Current Gateway	192.168.1.1
Mac Authenticating Code	 Valid MAC

WAN Setting

Static
DHCP
PPPOE

Static IP Address	192.168.1.179
Netmask	255.255.255.0
Gateway	192.168.1.1
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.68

APPLY

WAN Config	
Field Name	explanation
WAN Status	
Active IP	192.168.1.14
Current Netmask	255.255.255.0
MAC Address	00:12:f7:a0:00:54
Current Gateway	192.168.1.1
Mac Authenticating Code	<input type="text" value="Valid MAC"/>
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 hear the busy tone while the network is normal.
WAN Setting	
Static <input checked="" type="radio"/> DHCP <input type="radio"/> PPPOE <input type="radio"/>	
Please select the proper network mode according to the network condition. IP0027 provide three different network settings: <ul style="list-style-type: none"> ● Static: If your ISP server provides you the static IP address, please select this mode, then finish Static Mode setting. If you don't know about parameters of Static Mode setting, please ask your ISP for them. ● DHCP: In this mode, you will get the information from the DHCP server automatically; need not to input this information artificially. ● PPPoE: In this mode, your must input your ADSL account and password. You can also refer to 3.2.1 Network setting to speed setting your network. 	
Static IP Address	<input type="text" value="192.168.1.179"/>
Netmask	<input type="text" value="255.255.255.0"/>
Gateway	<input type="text" value="192.168.1.1"/>
DNS Domain	<input type="text"/>
Primary DNS	<input type="text" value="202.96.134.133"/>
Alter DNS	<input type="text" value="202.96.128.68"/>
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, phone will automatically add this domain to the end of the domain which you input before and parse it again.
Primary DNS	Input your primary DNS server address.
Alter DNS	Input your standby DNS server address.

PPPOE Server	ANY
Username	user123
Password

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

PPPoE Server	It will be provided by ISP.
Username	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 DHCP server distributed is 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

4.3.2.2. LAN Config

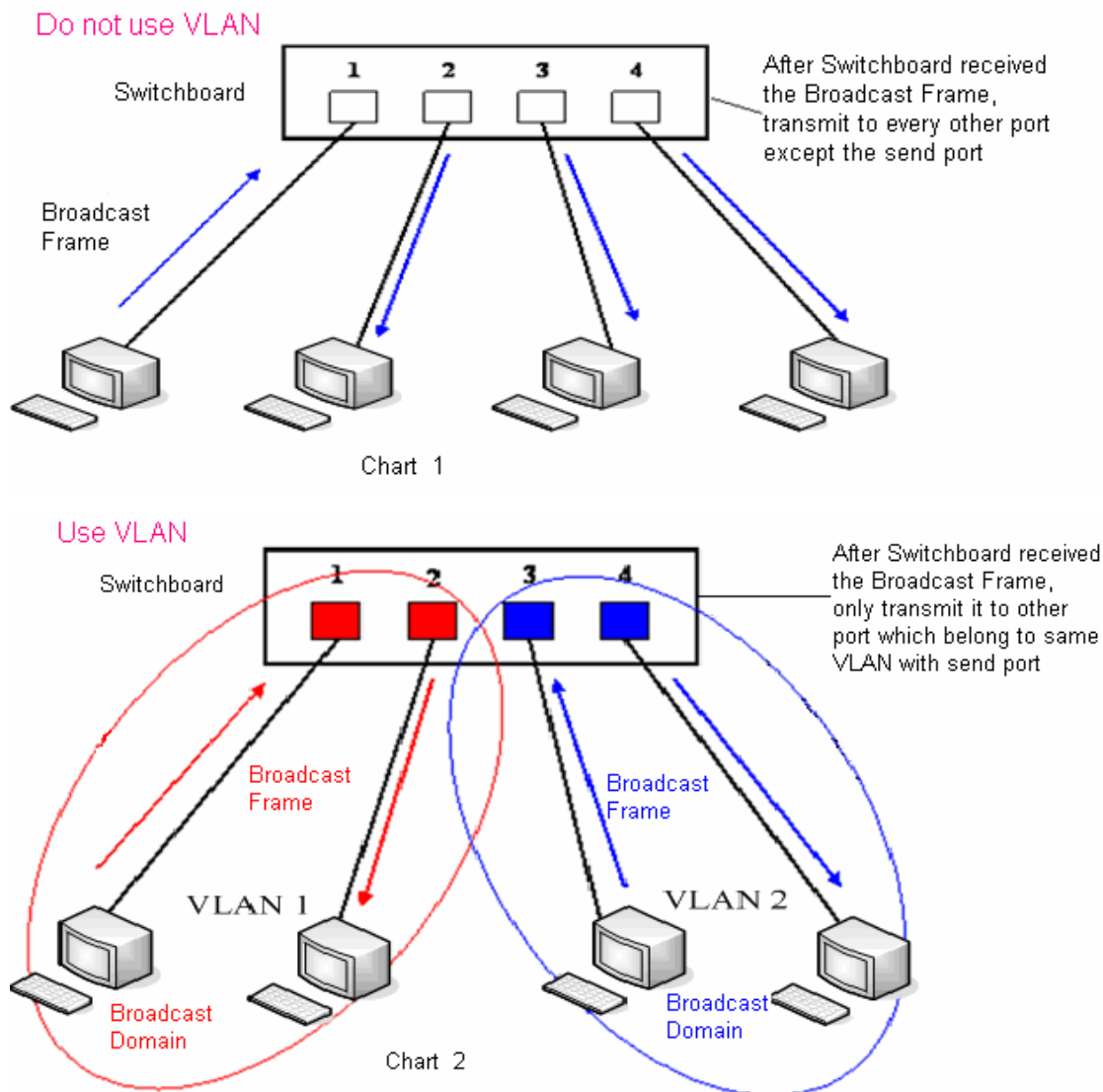
NETWORK	
WAN	LAN
QOS	SERVICE PORT
DHCP SERVER	SNTIP
LAN Setting	
LAN IP	192.168.10.1
Netmask	255.255.255.0
DHCP Service	<input checked="" type="checkbox"/>
NAT	<input checked="" type="checkbox"/>
Bridge Mode	<input type="checkbox"/>
APPLY	

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

Notice: If you choose the bridge mode, the LAN configuration will be disabled.

4.3.2.3. 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.



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, 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 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.

NETWORK			
WAN	LAN	QOS	SERVICE PORT
DHCP SERVER		SNTP	
QoS Set			
<input type="checkbox"/> VLAN Enable			
<input checked="" type="checkbox"/> VLAN ID Check Enable	Voice/Data VLAN differentiated		Undifferentiated
<input type="checkbox"/> DiffServ Enable	DiffServ Value		0x b8
Voice 802.1P Priority	0 (0 - 7)	Data 802.1P Priority	0 (0 - 7)
Voice VLAN ID	256 (0 - 4095)	Data VLAN ID	254 (0 - 4095)
<input type="button" value="APPLY"/>			

QoS Configuration	
Field name	explanation
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 data package is not the same with the phone's or a data package do not have VLAN ID, the data package will be discarded.
Voice/Data VLAN differentiated	After enable VLAN, system will set packets with different type of VLAN ID. Undifferentiated means after using VLAN, both voip packets and other data packets will use the voice VLAN ID; tag differentiated means after using VLAN, voip(signal and voice) packets will add voice VLAN ID, and other data packets will add data VLAN ID; data untagged means after using VLAN, only voip packets will add voice VLAN ID. Other data packets will not use VLAN.
DiffServ Enable	Select it or not to Enable or disable DiffServ.
DiffServ Value	Set DiffServ value, the common value is 0x00.
Voice 802.1P Priority	Specify 802.1P Priority of voice/signal data package.
Data 802.1P Priority	Set 802.1p of data VLAN. Non-voip data (such as http, telnet, ping etc) will use this value to set VLAN package.
Voice VLAN ID	Set VLAN ID of voice/signal data package.
Data VLAN ID	Set 802.1q of data VLAN ID. Non-voip data (such as http, telnet, ping etc) will use this value to set VLAN package.
NOTICE:	
1) Startup VLAN, if set Voice/Data VLAN differentiated as Undifferentiated, all packets will use the Voice VLAN ID as the tag.	
2) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and disable the DiffServ, then system will not distinguish the voice and data, all packets will use the Voice VLAN ID as the tag.	
3) Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and enable the DiffServ, then system will distinguish the voice and data and add the VLAN ID each other.	
4) Startup VLAN, if set Voice/Data VLAN differentiated as data untagged, then the packet of the	

- signal/voice will use the Voice VLAN ID as the tag, but the data packets will not take the VLAN tag.
- 5) If Disable the VLAN, regardless to set the Voice/Data VLAN differentiated or not, all packets will not take the VLAN tag; If enable the DiffServ, all packets will only take the DiffServ value.
 - 6) One must to notice, enable the VLAN ID Check Enable that is default, If enable it, the phone will match the VLAN ID strictly. When others' VLAN ID mismatch with us, the packets will discard. Contrarily, the phone will accept the packets with the distinct VLAN ID.
 - 7) You must gain the IP with the Static mode when you set VLAN, otherwise can't gain the IP in the VLAN and also can not dial with point to point.

4.3.2.4. Service Port

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

NETWORK

WAN
LAN
QOS
SERVICE PORT
DHCP SERVER
SNTP

Service Port

HTTP Port	<input type="text" value="80"/>
Telnet Port	<input type="text" value="23"/>
RTP Initial Port	<input type="text" value="10000"/>
RTP Port Quantity	<input type="text" value="200"/>

If modify HTTP or Telnet port,you'd better set it more than 1024,and the modification need to save and reboot.

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 Initial Port	Set the RTP Initial Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.
Notice:	
<ol style="list-style-type: none"> 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. 	

4.3.2.5. DHCP SERVER

NETWORK

WAN |
 LAN |
 QOS |
 SERIVCE PORT |
 DHCP SERVER |
 SNTP

DHCP Leased Table

Leased IP Address	Client Hardware Address
-------------------	-------------------------

DHCP Lease Table

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

DHCP Lease Table Setting

Lease Table Name	<input type="text"/>
Start IP	<input type="text"/>
End IP	<input type="text"/>
Lease Time	<input type="text"/> (minute)
Netmask	<input type="text"/>
Gateway	<input type="text"/>
DNS	<input type="text"/>
<input type="button" value="Add"/>	

DHCP Lease Table Delete

Lease Table Name	lan ▼	<input type="button" value="Delete"/>
------------------	-------	---------------------------------------

DNS relay Setting

DNS Relay <input checked="" type="checkbox"/>	<input type="button" value="APPLY"/>
---	--------------------------------------

DHCP SERVER

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

DHCP Lease Table

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

DHCP Lease Table Delete

Lease Table Name	lan ▼	<input type="button" value="Delete"/>
------------------	-------	---------------------------------------

Select name of lease table, click the Delete button will delete the selected lease table from DHCP lease table.	
DNS Relay	Select DNS Relay, the default is enable. Click the Apply button to become effective.
<p>Notice:</p> <p>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.</p> <p>2) If you modifies the DHCP lease table, you need save the configuration and reboot.</p>	

4.3.2.6. SNTP

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

NETWORK

WAN	LAN	QOS	SERIVCE PORT	DHCP SERVER	SNTP
SNTP Time Set					
Server	<input type="text" value="209.81.9.7"/>				
Time Zone	<input type="text" value="(GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi"/>				
Time Out	<input type="text" value="60"/> (seconds)				
SNTP	<input checked="" type="checkbox"/>				
Daylight	<input type="checkbox"/>				
<input type="button" value="APPLY"/>					
Manual Timeset					
Year	<input type="text"/>				
Months	<input type="text"/>				
Day	<input type="text"/>				
Hour	<input type="text"/>				
Minute	<input type="text"/>				
<input type="button" value="APPLY"/>					

SNTP	
Field name	explanation
Server	Set SNTP Server IP address.
Time Zone	Select the Time zone according to your location.
Time Out	Set the time out, the default is 60 seconds.
Daylight	If your time zone supports daylight, you can select it.
SNTP	Select the SNTP, and click Apply to make the SNTP Times effective.
Year	<input type="text"/>
Months	<input type="text"/>
Day	<input type="text"/>
Hour	<input type="text"/>
Minute	<input type="text"/>
<input type="button" value="APPLY"/>	
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.

VOIP

SIP
TAX2
STUN
DIAL PEER

SIP Line Select

SIP 1 ▾

Basic Setting

Register Status	Registered	Display Name	<input type="text"/>
Server Name	<input type="text"/>	Proxy Server Address	<input type="text"/>
Server Address	192.168.1.2	Proxy Server Port	<input type="text"/>
Server Port	5060	Proxy Username	<input type="text"/>
Account Name	2113	Proxy Password	<input type="text"/>
Password	••••	Domain Realm	<input type="text"/>
Phone Number	2113	SIP(Default Protocol)	<input checked="" type="checkbox"/>
Enable Register	<input checked="" type="checkbox"/>		

Advanced SIP Setting

Register Expire Time	60 seconds	Forward Type	Off ▾
NAT Keep Alive Interval	60 seconds	Forward Phone Number	<input type="text"/>
User Agent	Voip Phone 1.0	Server Type	common ▾
Signal Key	<input type="text"/>	DTMF Mode	DTMF_RFC2833 ▾
Media Key	<input type="text"/>	RFC Protocol Edition	RFC3261 ▾
Enable PRACK	<input type="checkbox"/>	Subscribe Expire Time	300 seconds
Enable Keep Authentication	<input type="checkbox"/>	Signal Encode	<input type="checkbox"/>
NAT Keep Alive	<input type="checkbox"/>	Rtp Encode	<input type="checkbox"/>
Enable Via rport	<input checked="" type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
Enable Subscribe	<input type="checkbox"/>	Answer With Single Codec	<input type="checkbox"/>
Enable CLIR	<input type="checkbox"/>		

Global SIP Setting

Enable URI Convert	<input checked="" type="checkbox"/>
--------------------	-------------------------------------

SIP Config	
Field name	explanation
SIP Line Select	
SIP 1 ▾ <input type="button" value="Load"/>	
Choose line to set info about SIP, there are 5 lines to choose. You can switch by 【Load】 button.	
Register Status	Shows if the phone has been registered the SIP server or not.

Server Name	Set the server name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
Account Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured.
Display Name	Set the display name.
Proxy Server Address	Set proxy server IP address (Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if your VoIP service provider give different configurations between Register SIP Server and Proxy SIP Server, you need make different settings.)
Proxy Server Port	Set your Proxy SIP server port.
Proxy Username	Input your Proxy SIP server account.
Proxy Password	Input your Proxy SIP server password.
Domain Realm	Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
SIP(Default Protocol)	Use SIP protocol as default dial protocol
Enable Register	Start to register or not by selecting it or not.
Register Expire Time	Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expire time set, the phone will change automatically the time into the time recommended by the server, and register again.
NAT Keep Alive Interval	Set examining interval of the server, default is 60 seconds
User Agent	Set the user agent if have, the default is VoIP Phone 1.0
Signal Key	Set the key for signal encryption
Media Key	Set the key for RTP encryption
Enable PRACK	Enable or disable SIP PRACK function, suggest use the default config.
Enable Keep Authentication	Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.
NAT Keep Alive	Enable/Disable keeps NAT of SIP alive. If some server refuse to register with too short interval time, and has no packets sending to device in private network to keep NAT alive, user could set this function ON. It need set the keep alive interval time less than the NAT server's.
Enable Via rport	Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT.
Enable Subscribe	Can subscribe after registered. Also can Subscribe other's state

	and voicemail.
Enable CLIR	Dial by using the anonymous.
Forward Type	<p>Select call forward mode, the default is Off</p> <ul style="list-style-type: none"> ● Off: Close down calling forward ● Busy: If the phone is busy, incoming calls will be forwarded to the appointed phone. ● No answer: If there is no answer, incoming calls will be forwarded to the appointed phone. ● Always: Incoming calls will be forwarded to the appoint phone directly. <p>The phone will Prompt the incoming while doing forward.</p>
Forward Phone Number	Appoint your forward phone number.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
DTMF Mode	<p>Select DTMF sending mode, there are three modes:</p> <ul style="list-style-type: none"> ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO <p>Different VoIP Service providers may provide different modes.</p>
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.
Subscribe Expire Time	Set expire time of subscribe.
Signal Encode	Enable/Disable Signal Encrypt.
RTP Encode	Enable/Disable RTP Encrypt.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Answer With Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Enable URI Convert	Convert # to %23 when send the URI.

4.3.3.2. IAX2 Config

VOIP			
SIP	IAX2	STUN	DIAL PEER
IAX2			
Register Status	Registered		
IAX2 Server Addr	192.168.1.4		
IAX2 Server Port	4569		
Account Name	4111		
Account Password	••••		
Phone Number	4111		
Local Port	4569		
Voice Mail Number	0		
Voice Mail Text	mail		
Echo Test Number	1		
Echo Test Text	echo		
Refresh Time	60	Seconds	
Enable Register	<input checked="" type="checkbox"/>		
Enable G.729	<input type="checkbox"/>		
IAX2(Default Protocol)	<input checked="" type="checkbox"/>		
<input type="button" value="APPLY"/>			

IAX2 Config	
Field name	explanation
Register Status	Shows if the phone has been registered the IAX2 server or not.
IAX2 Server Addr	Input your IAX2 server address.
IAX2 Server Port	Set your IAX2 server port, the default is 4569.
Account Name	Input your IAX2 register account name.
Account Password	Input your IAX2 register password.
Phone Number	Input your assigned phone number (usually it is same you're your IAX2 account name).
Local Port	Set your local sport, the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specify the voice mail's name.
Echo Test Number	Set echo test number. If IAX2 server supports echo test, and echo test number is non- numeric, system could set an echo test number to replace the echo test text. So user can dial the numeric number to test echo voice test. This function is provided with server to make endpoint to test whether endpoint could talk through server normally.
Echo Test Text	Specify echo test text's name.
Refresh Time	Set expire time of IAX2 server register, you can set it between 60 and 3600 seconds.
Enable Register	Start to register the IAX2 server or not by selecting it or not.
Enable G.729	Enable or disable code G.729 by selecting it or not
	Select it to make all outgoing calls through the IAX2 server by

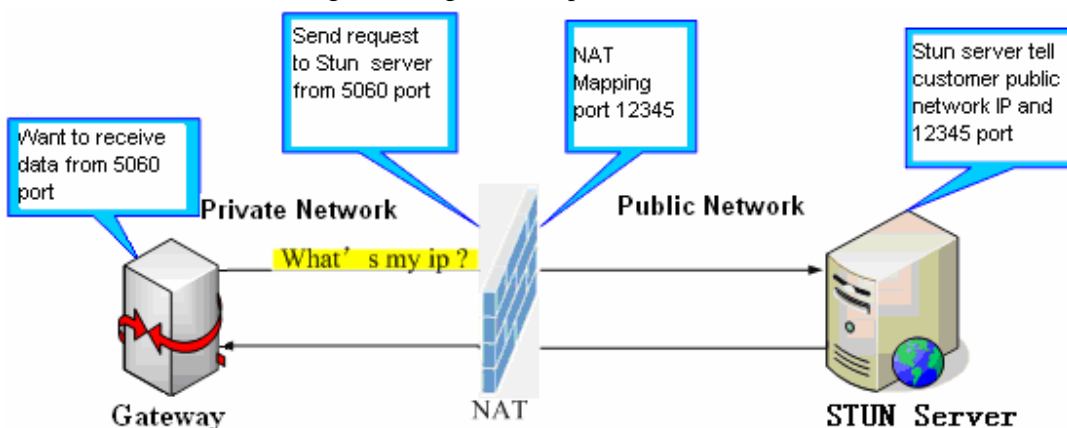
IAX2 (Default Protocol)	default. If you also need make a call through SIP server, you can make prefix in dial peer setting to realize SIP calling. Note: any incoming call can be from both IAX2 and SIP;
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4.3.3.3. Stun Config

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.



VOIP

SIP	IAX2	STUN	DIAL PEER
STUN Set			
STUN NAT Transverse	FALSE		
STUN Server Addr	<input type="text"/>		
STUN Server Port	<input type="text" value="3478"/>		
STUN Effect Time	<input type="text" value="50"/>	Seconds	
Local SIP Port	<input type="text" value="5060"/>		
<input type="button" value="APPLY"/>			
Set Sip Line Enable Stun			
<input type="text" value="SIP 1"/>	<input type="button" value="Load"/>		
Use Stun	<input type="checkbox"/>		
<input type="button" value="APPLY"/>			

STUN	
Field name	explanation
STUN NAT Transverse	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
STUN Server Addr	Set your SIP STUN Server IP address
STUN Server Port	Set your SIP STUN Server Port

STUN Effect Time	Set STUN Effective Time. If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need send a STUN packet to keep the mapping effective and alive.
Local SIP Port	Set the SIP port.
<div style="background-color: #0056b3; color: white; padding: 2px;">Set Sip Line Enable Stun</div> <div style="border: 1px solid #ccc; padding: 5px; margin-top: 5px;"> SIP 1 ▼ Load </div>	
Choose line to set info about SIP, There are 5 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.	

4.3.3.4. 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	Destination	Port	Alias	Suffix	Del Length
156	192.168.1.119	5060	no alias		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.

Number	Destination	Port	Alias	Suffix	Del Length
1T	0.0.0.0	5060	rep:010		1

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

Number	Destination	Port	Alias	Suffix	Del Length
6xxxxxxx	0.0.0.0	5060	add:0755		0
13[5-9]xxxxxxx	0.0.0.0	5060	add:0		0

1、 x Match any single digit that is dialed.

When you want to dial a long distance call to Shenzhen, User dial 8 digit numbers started with 6 can not take the area code, the phone will take 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.

When you want to dial a long distance call to mobile, you needn't add 0 before the phone number, the phone will take automatically.

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

VOIP

SIP
IAX2
STUN
DIAL PEER

Dial Peer Table

Number	Destination	Port	Alias	Suffix	Del Length
9T	0.0.0.0	5060			0
8T	0.0.0.2	5060			0
4xxx	0.0.0.0	5060	rep:1		4
13[0-9]xxx	0.0.0.0	5060	add:0		0

Add Dial Peer

Phone Number	<input style="width: 90%;" type="text"/>
Destination (optional)	<input style="width: 90%;" type="text"/>
Port(optional)	<input style="width: 90%;" type="text"/>
Alias(optional)	<input style="width: 90%;" type="text"/>
Suffix(optional)	<input style="width: 90%;" type="text"/>
Delete Length (optional)	<input style="width: 90%;" type="text"/>
<input type="button" value="Submit"/>	

Dial Peer Option

<input style="width: 90%;" type="text" value="9T"/>	<input type="button" value="Delete"/> <input type="button" value="Modify"/>
---	---

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 in SIP2 line, you need input 255.255.255.255 or 0.0.0.2 in it.
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.
<p>Note: There are four types of aliases.</p> <ol style="list-style-type: none"> 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.

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.

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

Number	Destination	Port	Alias	Suffix	Del Length
9T	0.0.0.0	5060			0
8T	0.0.0.2	5060			0

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

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

Corresponding other lines, like SIP3/SIP4/SIP5, can set Destination as 0.0.0.3/0.0.0.4/0.0.0.5

Examples of different alias application

Set by web	explanation	example												
<table border="1"> <tr> <td>Phone Number</td> <td>9T</td> </tr> <tr> <td>Destination (optional)</td> <td>255.255.255.255</td> </tr> <tr> <td>Port(optional)</td> <td></td> </tr> <tr> <td>Alias(optional)</td> <td>del</td> </tr> <tr> <td>Suffix(optional)</td> <td></td> </tr> <tr> <td>Delete Length (optional)</td> <td>1</td> </tr> </table>	Phone Number	9T	Destination (optional)	255.255.255.255	Port(optional)		Alias(optional)	del	Suffix(optional)		Delete Length (optional)	1	<p>You need set phone number, Destination, Alias and Delete Length.</p> <p>Phone number is XXXT, Destination is 255.255.255.255 and Alias is del.</p> <p>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.</p>	<p>If you dial “93333”, the SIP2 server will receive “3333”</p>
Phone Number	9T													
Destination (optional)	255.255.255.255													
Port(optional)														
Alias(optional)	del													
Suffix(optional)														
Delete Length (optional)	1													

<table border="1"> <tr><td>Phone Number</td><td>2</td></tr> <tr><td>Destination (optional)</td><td></td></tr> <tr><td>Port(optional)</td><td></td></tr> <tr><td>Alias(optional)</td><td>all:33334444</td></tr> <tr><td>Suffix(optional)</td><td></td></tr> <tr><td>Delete Length (optional)</td><td></td></tr> </table>	Phone Number	2	Destination (optional)		Port(optional)		Alias(optional)	all:33334444	Suffix(optional)		Delete Length (optional)		<p>This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.</p>	<p>When you dial “2”, the SIP1 server will receive 33334444</p>
Phone Number	2													
Destination (optional)														
Port(optional)														
Alias(optional)	all:33334444													
Suffix(optional)														
Delete Length (optional)														
<table border="1"> <tr><td>Phone Number</td><td>8T</td></tr> <tr><td>Destination (optional)</td><td></td></tr> <tr><td>Port(optional)</td><td></td></tr> <tr><td>Alias(optional)</td><td>add:0755</td></tr> <tr><td>Suffix(optional)</td><td></td></tr> <tr><td>Delete Length (optional)</td><td></td></tr> </table>	Phone Number	8T	Destination (optional)		Port(optional)		Alias(optional)	add:0755	Suffix(optional)		Delete Length (optional)		<p>The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.</p>	<p>When you dial “8309“, the SIP1 server will receive “07558309”</p>
Phone Number	8T													
Destination (optional)														
Port(optional)														
Alias(optional)	add:0755													
Suffix(optional)														
Delete Length (optional)														
<table border="1"> <tr><td>Phone Number</td><td>010T</td></tr> <tr><td>Destination (optional)</td><td></td></tr> <tr><td>Port(optional)</td><td></td></tr> <tr><td>Alias(optional)</td><td>rep:8610</td></tr> <tr><td>Suffix(optional)</td><td></td></tr> <tr><td>Delete Length (optional)</td><td>3</td></tr> </table>	Phone Number	010T	Destination (optional)		Port(optional)		Alias(optional)	rep:8610	Suffix(optional)		Delete Length (optional)	3	<p>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.</p>	<p>When you dial “0106228”, the SIP1 server will receive “86106228”</p>
Phone Number	010T													
Destination (optional)														
Port(optional)														
Alias(optional)	rep:8610													
Suffix(optional)														
Delete Length (optional)	3													
<table border="1"> <tr><td>Phone Number</td><td>147</td></tr> <tr><td>Destination (optional)</td><td></td></tr> <tr><td>Port(optional)</td><td></td></tr> <tr><td>Alias(optional)</td><td></td></tr> <tr><td>Suffix(optional)</td><td>0011</td></tr> <tr><td>Delete Length (optional)</td><td></td></tr> </table>	Phone Number	147	Destination (optional)		Port(optional)		Alias(optional)		Suffix(optional)	0011	Delete Length (optional)		<p>If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.</p>	<p>When you dial “147”, the SIP1 server will receive “1470011”</p>
Phone Number	147													
Destination (optional)														
Port(optional)														
Alias(optional)														
Suffix(optional)	0011													
Delete Length (optional)														

4.3.4. Phone

4.3.4.1. DSP Config

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

PHONE

DSP	CALL SERVICE	DIGITAL MAP	PHONE BOOK	FUNCTION KEY	
-----	--------------	-------------	------------	--------------	--

DSP Configuration

Coding Rule	g711Ulaw64k ▾	Ring Type	Type 1 ▾
Input Volume	3 (1-9)	Output Volume	7 (1-9)
Handfree Volume	4 (1-9)	Ring Volume	5 (1-9)
G729 Payload Length	20ms ▾	Signal Standard	China ▾
Handdown Time	200 ms	VAD	<input type="checkbox"/>

DSP Configuration	
Field name	explanation
Coding Rule	Select DSP voice coding rule. Such as G.711A/u, G.723 (5.3k/6.3k) , G.729, G.722
Input Volume	Specify Input (MIC) Volume grade.;
Handfree Volume	Specify Handfree Volume grade
G729 Payload Length	Set G729 Payload Length
Handdown Time	Specify the least reflection time of Handdown, the default is 200ms.
Ring Type	Select Ring Type
Output Volume	Specify Output (receiver) Volume grade.
Ring Volume	Specify Ring Volume grade
Signal Standard	Select Signal Standard.
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. Call Service

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

PHONE

DSP
CALL SERVICE
DIGITAL MAP
PHONE BOOK
FUNCTION KEY

Call Service Setting

Hot Line	<input type="text"/>	No Answer Time	20 (seconds)
P2P IP Prefix	<input type="text"/>	Remote Record No	<input type="text"/>
No Disturb	<input type="checkbox"/>	Ban Outgoing	<input type="checkbox"/>
Enable Call Transfer	<input checked="" type="checkbox"/>	Enable Call Waiting	<input checked="" type="checkbox"/>
Enable Three Way Call	<input checked="" type="checkbox"/>	Accept Any Call	<input checked="" type="checkbox"/>
Auto Answer	<input type="checkbox"/>	Enable Voice Record	<input type="checkbox"/>
User-Defined Voice	<input checked="" type="checkbox"/>	Incoming Record Playing	<input checked="" type="checkbox"/>
Use Record Server	<input type="checkbox"/>		

Black List


Black List

▼

Limit List

Limit List

▼

Call Service	
Field name	explanation
Hotline	Specify Hotline number. If you set the number, you can not dial any other numbers.
No Answer Time	Specify No Answer Time
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119. Default is “.”. If there is no “.” Set, it means to disable dialing IP.
Remote Record No	Set Remote Record number. Via dialing this number, you can hear all voice records in your VoIP server.
No Disturb	Select NO Disturb, the phone will reject any incoming call, the callers will be reminded by busy, but any outgoing call from the phone will work well.
Ban Outgoing	If you select Ban Outgoing to enable it, and you can not dial out any number.
Enable Call Transfer	Enable Call Transfer by selecting it.
Enable Call Waiting	Enable Call Waiting by selecting it.
Enable Three Way Call	Enable Three Way Call by selecting it. If you want cancel the 3 ways, you may use the  button.
Accept Any Call	If select it, the phone will accept the call even if the called number

	is not belong to the phone.
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
Use Record Server	Select it or not to Enable or disable Use Record 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, and then you can not dial out any phone number whose prefix is 001.
Notice: Black List and Limit List can record at most 10 items respectively.	

4.3.4.3. 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.

PHONE

DSP
CALL SERVICE
DIGITAL MAP
PHONE BOOK
FUNCTION KEY

Digital Map Set

<input checked="" type="checkbox"/>	End With “#”	
<input type="checkbox"/>	Fixed Length	<input style="width: 100px;" type="text" value="11"/>
<input checked="" type="checkbox"/>	Time Out	<input style="width: 100px;" type="text" value="5"/> (3--30)

Digital Rule table

Rules:

<input style="width: 100%;" type="text"/>	<input type="button" value="Add"/>	<input type="button" value="v"/>	<input type="button" value="Del"/>
---	------------------------------------	----------------------------------	------------------------------------

Digital Map Configuration	
Field name	explanation
End with “#”	Set Enable/Disable the phone ended with “#” dial.
Fixed Length	Specify the Fixed Length of phone ending with.

Time out	Set the timeout of the last dial digit. The call will be sent after timeout.						
Digital Rule table							
Rules:							
<input type="text"/>	<input type="button" value="Add"/> <input type="button" value="Del"/>						
<p>Below is user-defined digital map rule:</p> <p>[] Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits.</p> <p>x Match any single digit that is dialed.</p> <p>. Match any arbitrary number of digits including none.</p> <p>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.</p> <table border="1" style="width: 100%;"> <thead> <tr> <th>RULE</th> </tr> </thead> <tbody> <tr> <td>"[1-8]xxx"</td> </tr> <tr> <td>"9xxxxxxx"</td> </tr> <tr> <td>"911"</td> </tr> <tr> <td>"99T4"</td> </tr> <tr> <td>"9911x.T4"</td> </tr> </tbody> </table> <p>Cause extensions 1000-8999 to be dialed immediately</p> <p>Cause 8 digit numbers started with 9 to be dialed immediately</p> <p>Cause 911 to be dialed immediately after it is entered.</p> <p>Cause 99 to be dialed after 4 seconds.</p> <p>Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.</p>		RULE	"[1-8]xxx"	"9xxxxxxx"	"911"	"99T4"	"9911x.T4"
RULE							
"[1-8]xxx"							
"9xxxxxxx"							
"911"							
"99T4"							
"9911x.T4"							
<p>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.</p>							

4.3.4.4. Phone Book

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

PHONE			
DSP	CALL SERVICE	DIGITAL MAP	PHONE BOOK
Phonebook Table			
Index	Name	Number	Type
1	ad	23	Default
<input type="button" value="1"/>			
Add Phone Book			
Name	<input type="text"/>		<input type="button" value="Add"/>
Number	<input type="text"/>		
Ring Type	Default <input type="button" value="v"/>		
Phone Book Option			
ad <input type="button" value="v"/>		<input type="button" value="Delete"/> <input type="button" value="Modify"/>	

Phone Book			
Field name		explanation	
Index	Name	Number	Type
1	ad	23	Default
<input type="text" value="1"/>			
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			

4.3.4.5. Function Key

PHONE

DSP
CALL SERVICE
DIGITAL MAP
PHONE BOOK
FUNCTION KEY

Functioning Key Type

Memory Button
 PhoneLine Button
 Command Button

Memory Key Setting

F 1	
F 2	
F 3	
F 4	
F 5	
<input type="text" value="F1"/>	<input type="button" value="APPLY"/>

Function Key	
Field name	explanation
Memory Button	Can set number for every memory key, You can dial this numberBy selecting these keys.
PhoneLine Button	Can use the function key to select SIP line.
Command Button	When the SIP1 is available, can use the F1 key to send the NOTIFY, if unavailable, the F1 key will not send the message.
F 1	
F 2	
F 3	
F 4	
F 5	
<input type="text" value="F1"/>	<input type="button" value="APPLY"/>

This table will display the number of the memory key.

4.3.5. Maintenance

4.3.5.1. Auto Provision

MAINTENANCE	
AUTO PROVISION SYSLOG CONFIG UPDATE ACCOUNT REBOOT	
Auto Update Setting	
Current Config Version	2.0002
Server Address	<input type="text" value="0.0.0.0"/>
Username	<input type="text" value="user"/>
Password	<input type="password" value="****"/>
Config File Name	<input type="text"/>
Config Encrypt Key	<input type="text"/>
Protocol Type	<input type="text" value="FTP"/>
Update Interval Time	<input type="text" value="1"/> Hour
Update Mode	<input type="text" value="Disable"/>
<input type="button" value="APPLY"/>	

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

4.3.5.2. Syslog Config

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.

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

MAINTENANCE	
AUTO PROVISION	SYSLOG
CONFIG	UPDATE
ACCOUNT	REBOOT
Syslog Set	
Server IP	<input type="text" value="0.0.0.0"/>
Server Port	<input type="text" value="514"/>
MGR Log Level	<input type="text" value="None"/> ▾
SIP Log Level	<input type="text" value="None"/> ▾
IAX2 Log Level	<input type="text" value="None"/> ▾
Enable Syslog	<input type="checkbox"/>
<input type="button" value="APPLY"/>	

Syslog Configuration	
Field name	explanation
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.

4.3.5.3. Config Setting

MAINTENANCE

AUTO PROVISION
SYSLOG
CONFIG
UPDATE
ACCOUNT
REBOOT

Save Configuration

Press the "Save" button to save the configuration files !

Backup Configuration

Save all Network and VoIP settings.
[Right Click here to Save as Config File \(.txt\)](#)

Clear Configuration

Press the "Clear" button to Clear the configuration files !

Config Setting	
Field name	explanation
Save Config	you can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately. .
Backup Config	Right clicks on "Right click here..." and select "Save Target As...." then you will save the config file in .txt format
Clear Config	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 (SIP1-5 and IAX2) and version number.

4.3.5.4. Update

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

MAINTENANCE

AUTO PROVISION
SYSLOG
CONFIG
UPDATE
ACCOUNT
REBOOT

Web Update

Select file: (*.z,*.txt,*.au)

FTP Update

Server	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
File Name	<input type="text"/>
Type	Application update ▾
Protocol	FTP ▾

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	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
Username	Set the FTP server Username for download/upload.
Password	Set the FTP server password for download/upload.
File name	Set the name of update file or config file. The default name is the MAC of the phone, such as 000102030405.
<p>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.</p>	
Type	<p>Action type that system want to execute:</p> <ol style="list-style-type: none"> 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 file import: Download the config file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset.
Protocol	Select FTP/TFTP server

4.3.5.5. Account Config

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

MAINTENANCE

AUTO PROVISION
SYSLOG
CONFIG
UPDATE
ACCOUNT
REBOOT

Set Keyboard Password

Keyboard Password

User Set

User Name	User Level
admin	Root
guest	General

Add User

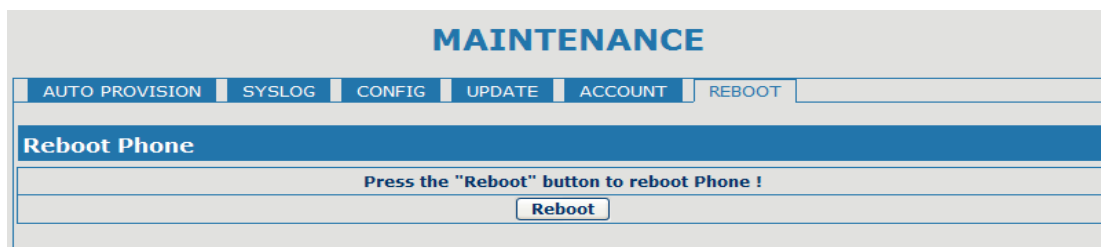
User Name	<input style="width: 60%;" type="text"/>
User Level	Root <input type="button" value="v"/>
Password	<input style="width: 60%;" type="password"/>
Confirm	<input style="width: 60%;" type="password"/>

Account Option

admin <input type="button" value="v"/>	<input type="button" value="Delete"/> <input type="button" value="Modify"/>
--	---

Account Configuration							
Field name	explanation						
Keyboard Password	Set the password for entering the setting menu of the phone by the phone 's key board. The password is digit.						
<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: center;">User Name</th> <th style="text-align: center;">User Level</th> </tr> </thead> <tbody> <tr> <td style="text-align: center;">admin</td> <td style="text-align: center;">Root</td> </tr> <tr> <td style="text-align: center;">guest</td> <td style="text-align: center;">General</td> </tr> </tbody> </table>		User Name	User Level	admin	Root	guest	General
User Name	User Level						
admin	Root						
guest	General						
This table shows the current user existed.							
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.						
<p>Select the account and click the Modify to modify the selected account, and click the Delete to delete the selected account.</p> <p>General user only can add the user whose level is General.</p>							

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

SECURITY

MMI FILTER
FIREWALL
NAT
VPN

MMI Filter Table

Start IP	End IP	Option
<input style="width: 95%;" type="text" value="192.168.1.15"/>	<input style="width: 95%;" type="text" value="192.168.1.20"/>	<input type="button" value="Modify"/> <input type="button" value="Delete"/>

MMI Filter Table Set

Start IP	End IP	
<input style="width: 95%;" type="text"/>	<input style="width: 95%;" type="text"/>	<input type="button" value="Add"/>

MMI Filter Table Set

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						
<div style="background-color: #0056b3; color: white; padding: 2px; font-weight: bold;">MMI Filter Table</div> <table border="1" style="width: 100%; border-collapse: collapse; margin-top: 5px;"> <thead> <tr> <th style="width: 40%;">Start IP</th> <th style="width: 40%;">End IP</th> <th style="width: 20%;">Option</th> </tr> </thead> <tbody> <tr> <td><input style="width: 95%;" type="text" value="192.168.1.15"/></td> <td><input style="width: 95%;" type="text" value="192.168.1.20"/></td> <td style="text-align: center;"> <input type="button" value="Modify"/> <input type="button" value="Delete"/> </td> </tr> </tbody> </table>		Start IP	End IP	Option	<input style="width: 95%;" type="text" value="192.168.1.15"/>	<input style="width: 95%;" type="text" value="192.168.1.20"/>	<input type="button" value="Modify"/> <input type="button" value="Delete"/>
Start IP	End IP	Option					
<input style="width: 95%;" type="text" value="192.168.1.15"/>	<input style="width: 95%;" type="text" value="192.168.1.20"/>	<input type="button" value="Modify"/> <input type="button" value="Delete"/>					
MMI Filter IP Table list:							
<div style="background-color: #0056b3; color: white; padding: 2px; font-weight: bold;">MMI Filter Table Set</div> <table border="1" style="width: 100%; border-collapse: collapse; margin-top: 5px;"> <thead> <tr> <th style="width: 30%;">Start IP</th> <th style="width: 30%;">End IP</th> <th style="width: 40%;"></th> </tr> </thead> <tbody> <tr> <td><input style="width: 95%;" type="text"/></td> <td><input style="width: 95%;" type="text"/></td> <td style="text-align: center;"><input type="button" value="Add"/></td> </tr> </tbody> </table>		Start IP	End IP		<input style="width: 95%;" type="text"/>	<input style="width: 95%;" type="text"/>	<input type="button" value="Add"/>
Start IP	End IP						
<input style="width: 95%;" type="text"/>	<input style="width: 95%;" type="text"/>	<input type="button" value="Add"/>					
Add or delete the IP address segments that access to the phone. Set initial IP address in the Start IP column, Set end IP address in the End IP column, and click Add to add this IP segment. You can also click Delete to delete the selected IP segment.							
MMI Filter	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

SECURITY

MMI FILTER
FIREWALL
NAT
VPN

Firewall Type

In_access Enable
 Out_access Enable

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port

Firewall Output Rule Table

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

Firewall Set

Input/Output	Input	Src Addr	<input type="text"/>	<input type="button" value="Add"/>
Deny/Permit	Deny	Des Addr	<input type="text"/>	
Protocol Type	UDP	Src Mask	<input type="text"/>	
Port Range	more than	Des Mask	<input type="text"/>	

Rule Delete

Input/Output	Input	Index To Be Deleted	<input type="text"/>	<input type="button" value="Delete"/>
--------------	-------	---------------------	----------------------	---------------------------------------

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.

In_access Enable
 Out_access Enable

Input/Output	Input	Src Addr	<input type="text"/>	<input type="button" value="Add"/>
Deny/Permit	Deny	Des Addr	<input type="text"/>	
Protocol Type	UDP	Src Mask	<input type="text"/>	
Port Range	more than	Des Mask	<input type="text"/>	

Field name	explanation
In_access enable	Select it to Enable in_ access rule
out_access enable	Select it to Enable out_ access rule

Input/Output	Specify current adding rule by selecting input rule or output rule.
Deny/Permit	Specify current adding rule by selecting Deny rule or Permit rule.
Protocol Type	Filter protocol type. You can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range
Src Addr	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0
Des Addr	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*
Src Mask	Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Des Mask	Set the destination address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.

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	192.168.1.14	255.255.255.0	192.168.1.118	255.255.255.0	more than	1

Then enable out_access, and click the Apply button.

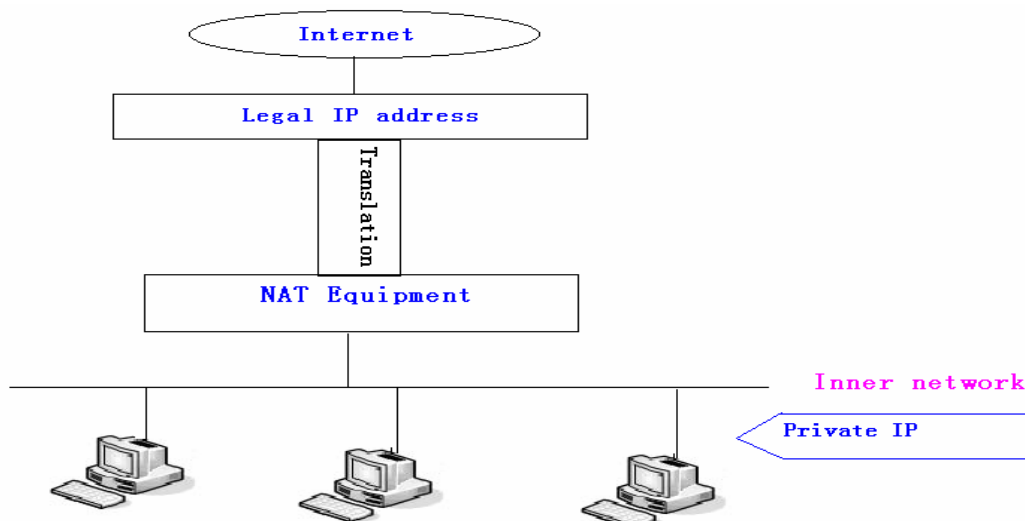
So when devices execute to ping 192.168.1.118, system will deny the request to send icmp request to 192.168.1.118 for the out_access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.

Rule Delete			
Input/Output	Input	Index To Be Deleted	Delete

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

4.3.6.3. NAT Config

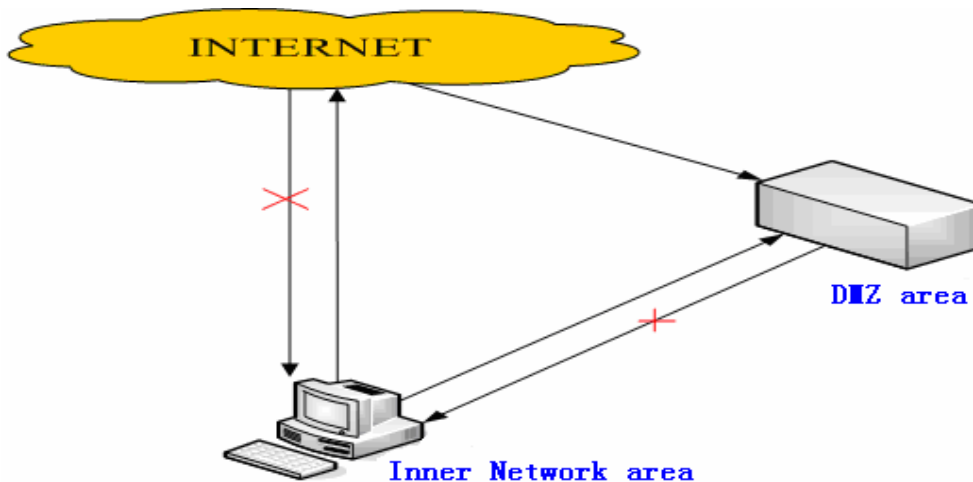
NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.



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

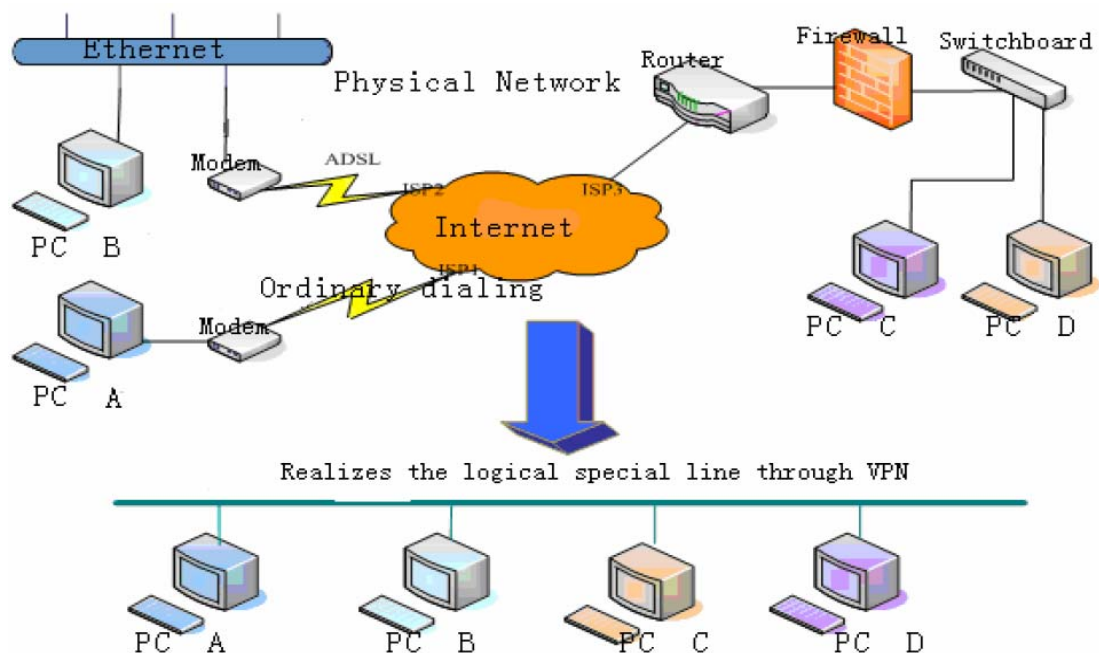


SECURITY			
MMI FILTER	FIREWALL	NAT	VPN
Protocol Set			
<input checked="" type="checkbox"/> IPSec ALG	<input checked="" type="checkbox"/> FTP ALG	<input checked="" type="checkbox"/> PPTP ALG	<input type="button" value="APPLY"/>
NAT Table			
Inside IP	Inside TCP Port	Outside TCP Port	
Inside IP	Inside UDP Port	Outside UDP Port	
NAT Table Option			
Transfer Type	TCP	Outside Port	
Inside Ip		Inside Port	
<input type="button" value="Add"/>		<input type="button" value="Delete"/>	
<input type="button" value="DMZ Config"/>			
DMZ Table			
Outside IP	Inside IP		
DMZ Table Option			
Outside IP			
Inside IP			
Outside IP			
<input type="button" value="Add"/>		<input type="button" value="Delete"/>	

NAT Configuration	
Field name	explanation
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
Inside IP	Inside TCP Port Outside TCP Port
Shows the NAT TCP mapping table	
Inside IP	Inside UDP Port Outside UDP Port
Shows the NAT UDP mapping table	
NAT Table Option	
Transfer Type	TCP ▼ Outside Port
Inside Ip	Inside Port
<input type="button" value="Add"/> <input type="button" value="Delete"/>	
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
<p>Notice: After finish setting, click the Add button to add new mapping table; click the Delete button to delete the selected mapping table.</p>	
DMZ Table	
Outside IP	Inside IP
192.168.1.119	192.168.10.23
Shows the outside WAN port IP address and the inside LAN port IP address.	
Outside IP	Inside IP
Outside IP	192.168.1.119 ▼
<input type="button" value="Add"/> <input type="button" value="Delete"/>	
Outside IP	Set the outside Wan port IP address of DMZ.
Inside IP	Set the inside LAN port IP address of DMZ
<p>Click the Add button to add new table; click the Delete button to delete the selected mapping table.</p>	
<p>Notice: 10M/100M adaptive means the network card, and other equipment physical consultations speed, testing speed under bridge mode near to 100M, in order to ensure the quality of voice and communications real-time performance, we made some sacrifices of NAT under the transmission performance. Transmit with full capability only when system is idle, so can not guarantee that the transmission speed reach to 100M.</p>	

4.3.6.4. 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.



SECURITY			
MMI FILTER		FIREWALL	
NAT		VPN	
VPN IP			
0.0.0.0			
VPN Mode			
<input checked="" type="radio"/> UDP Tunnel		<input type="radio"/> L2TP	
<input type="checkbox"/> Enable VPN			
UDP Tunnel			
VPN Server Addr	0.0.0.0	VPN Server Port	80
Server Group ID	VPN	Server Area Code	12345
L2TP			
VPN Server Addr		VPN User Name	
VPN Password			
<input type="button" value="APPLY"/>			

VPN Configuration	
Field name	explanation
VPN IP	Shows the current VPN IP address
VPN Mode	
<input checked="" type="radio"/> UDP Tunnel <input type="radio"/> L2TP <input type="checkbox"/> Enable VPN	

Select UDP Tunnel (VPN Tunnel) or VPN L2TP. You can choose only one for current state.
After you select it, you'd better save configuration and reboot your phone.

Enable VPN	Select it or not to enable or disable VPN;		
UDP Tunnel			
VPN Server Addr	<input type="text" value="0.0.0.0"/>	VPN Server Port	<input type="text" value="80"/>
Server Group ID	<input type="text" value="VPN"/>	Server Area Code	<input type="text" value="12345"/>
VPN Server Addr	Set VPN Server IP Address		
VPN Server Port	Set VPN Server Port		
L2TP			
VPN Server Addr	<input type="text"/>	VPN User Name	<input type="text"/>
VPN Password	<input type="text"/>		
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		

4.3.7. Logout


System Logout

Logout
<p>Press the "Logout" button to Logout Phone !</p> <div style="text-align: center;"> <input type="button" value="Logout"/> </div>

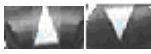

Click **Logout**, and you will exit web page. If you want to enter it next time, you need input user name and password again.

5.4. Configuration via Keypad

5.4.1. Keypad introduction

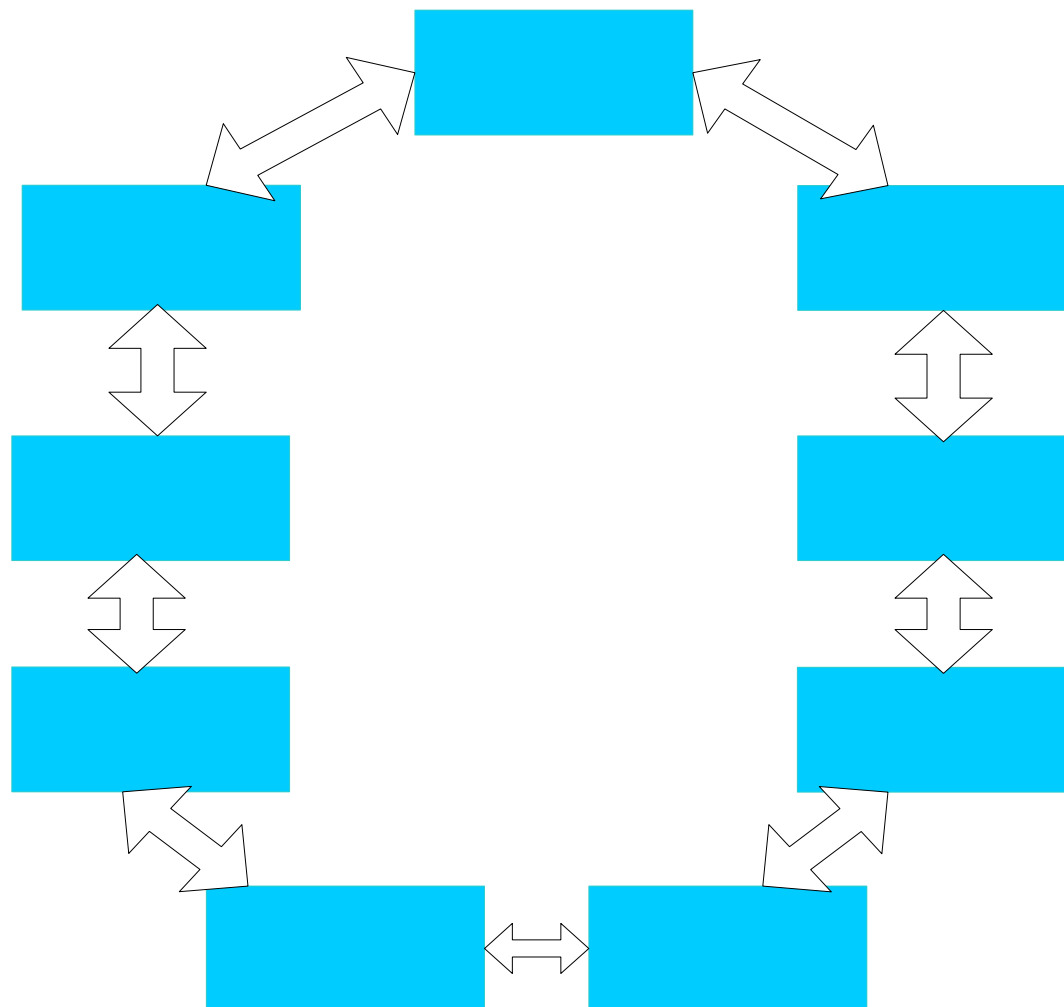
User can do brose, modify or cancel via screen menu by using ,

 and Soft1/Soft2/Soft3

- Use  and Soft1 to select the sub-menu.
- Use  to adjust screen brightness and contrast, ring volume and voice volume.
- Use Soft2 and Soft3 to enter/modify or exit/cancel.

5.4.2. Menu Tree

Menu Tree List:



-- Co
Scre

6. Appendix

6.1. Specification

6.1.1. Hardware

Item		IP0027
Adapter (Input/Output)		Input: 100-240V Output: 5V 1A or PoE(802.3af, optional)
port	WAN	10/100Base-T RJ-45 for LAN
	LAN	10/100Base-T RJ-45 for PC
Power Consumption		Idle: 2.0W/Active: 2.3W
LCD Size		128*64 dot matrix LCD
Operation Temperature		0~40°C
Relative Humidity		10~65%
CPU		INFINEON (MIPS 4Kc@150MHz, DSP@100MHz)
SDRAM		128Mbits
Flash		16Mbits
Dimension(L x W x H)		235x200x81mm
Weight		2.07lb.(0.94kg)

6.1.2. Voice features

- Support SIP 2.0 (RFC3261) and correlative RFCs
- Support IAX2
- SIP supports 5 SIP servers. Can connect to 5 SIP servers at the same time
- Codec: G.711A/u, G.7231 high/low, G.729a/b, G.722
- Echo cancellation: G.168 Compliance in LEC, additional acoustic echo cancellation(AEC) can reach 96ms max filter length in hands-free mode
- Support Voice Gain Setting, VAD, CNG
- Full duplex hands-free speakerphone
- NAT transverse: support STUN client
- SIP support SIP domain, SIP authentication(none basic, MD5), DNS name of server, Peer to Peer/ IP call
- DTMF Relay: support inband, SIP info, RFC2833
- SIP application: support call forward /transfer /holding /waiting /conference /paging and intercom
- Call control features: Flexible dial map, hotline, empty calling No. reject service, black list for reject authenticated call, limit call, no disturb, caller ID
- Support phonebook 500 records
- Incoming calls / outgoing calls / missing calls. Each supports 100 records
- Support voice record on SIP server
- Support voice record in this phone, 20 minutes total, 5 minutes max of 3 records or 15 records max
- 9 kind of ring types and 2 user-defined music rings

6.1.3. Network features

- WAN/LAN: support bridge and router model
- Support PPPoE for xDSL
- Support basic NAT and NAPT
- Support DHCP client on WAN

- Support DHCP server on LAN
- Support main DNS and secondary DNS server
- Support VLAN (optional: voice vlan/ data vlan)
- Support DMZ
- Support DNS Relay, SNTP Client, Firewall
- Support VPN (L2TP/ UDP TUNNEL)
- QoS with DiffServ
- Network tools in telnet server: including ping, trace route, telnet client






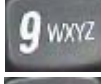






6.1.4. Maintenance and management

- Web ,telnet and keypad management
- Management with different account right
- Upgrade firmware through POST mode
- Upgrade firmware through HTTP, FTP or TFTP Telnet remote management/ upload/download setting file
- Safe mode provide reliability
- Support Auto Provisioning (upgrade firmware or configuration file)
- Support Syslog

6.1.5. Special features

- Support headset
- 5 programmable keys, realizing memory keys or SIP line keys
- Support N'Pad record
- Support desk position and wall-mountable
- Led to indicate missed call or voicemail
- Support 3 softkeys

6.2. digit-character map table

Keypad	Character	Keypad	Character
	1 @		7 P Q R S p q r s
	2 A B C a b c		8 T U V t u v
	3 D E F d e f		9 W X Y Z w x y z
	4 G H I g h i		.
	5 J K L j k l		0
	6 M N O m n o		#