VolP Phone BW210 User Manual





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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1. Introducing BW210 VoIP Phone

1.1. Thank you for your purchasing BW210

Thank you for your purchasing BW210, BW210 is a full-feature telephone that provides voice communication over the same data network that your computer uses. This phone functions not only much like a traditional phone, allowing to place and receive calls, and enjoy other features that traditional phone has, but also it own many data services features which you could not expect from a traditional telephone.

This guide will help you easily use the various features and services available on your phone.



1.2. Delivery Content

Please check whether the delivery contains the following parts:

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

1.3. Keypad



The numeric keypad with the keys 0 to 9, *, and # is used to enter

Digits and letters, additionally, the following

Key mapping:

| Кеу | Key name | Function Description |
|--------------------|--------------------|---|
| | Menu | In idle state, press the MENU key to call up the |
| MENU Phone Book | Phone Book | In idle mode, press the Phone Book key to check the record list and add new records and revise the record. Press this key again will return to idle mode. |
| CALLERS | Callers | In idle/pickup/calling mode, press the Callers key to Check the Income/Outgoing/Missed calls records. Press this key again will return to idle mode |
| | LED | LED blinks to remind user new voicemail. |
| Sysinfo | System Information | In idle mode, press the Sysinfo key to check the phone setting parameters. Such as local phone number, local IP and local Gateway IP address. |
| Enter | Confirm | Use the Enter key to enter next menu, or confirm the setting. |
| Exit | Exit | Use the Exit key to return to previous menu, ,cancel the setting, or reject to answer a call. |
| \odot | Navigation Key | When you pick up the handset or during calling, you can use this key to turn up or turn down the handset volume; when a call comes, you can use this key to adjust ring volume; you also can use this key to choose item in the menu, callers or phone book. |
| \bigcirc | MWI | Use this key to read old or new message. |
| MWI Transfer | Transfer | Use the key to realize blind transfer or attended transfer please refer to 3.1.4. -call transfer for more details). |
| Conference | Conference | Use this key to realize the three party call (pleae refer to 3.1.5-Calling Hold and 3 ways call for more details) |
| Del | Delete | In menu, use this key to modify current setting or delete invalid information. And when you input number do call out, you can use it to delete characters |
| Hold | Hold | press the key again to unhold the call. You also can press this key then input the third party's phone number and end with the # key during calling, you can make a call with the third party and hold the previous calling. (3.1.5-Calling Hold and 3 ways call). |



1.4. Port for connecting



| POWER | Power switch | Select ON/OFF |
|-------|--------------|-----------------------|
| DC | Power port | Output: 5V/1.0A |
| LAN | Network port | Connect it to PC |
| WAN | Network port | Connect it to Network |

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

2.Initial connecting and Setting

2.1. connect the phone

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

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

Direct network connection—by this method, you need at least one available Ethernet port in your workspace. Use the Ethernet cable in the package to connect WAN port on the back of your phone to the Ethernet port in your workspace. 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 port on the back of the phone. Use the power cable to connect the power supply to a standard power outlet in your workspace.

Step 4: push the on/off switch on the back of the phone to the on side, then the phone's LCD screen displays "WAIT LOGON". Later, a ready screen typically displays the date, time and current network mode.

If your LCD screen displays different information from the above, you need refer to the next section "Initial setting" to set your network online mode.

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

2.2. Initial Setting

This VoIP Phone provides you with rich function and parameters setting. If you have enough

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

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

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

2.2.1. PPPoE mode.

1. Press the 3 key for three seconds, then confirm it by the Enter key, your phone network connection mode will switch into PPPoE mode. Prepare your PPPoE account name and password. 2. Press the MENU key, the LCD screen will display "INPUT PASSWORD".

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

| 4. Press the key and LCD screen will display LAN, press the key, enter it by the |
|--|
| key, the LCD screen will display "STATIC NET". Then press the key again, |
| enter it by the LCD screen will display "USER NAME". |
| 5. Press the key and then press the let key, input your PPPOE account number then |
| press the key to confirm. The LCD screen will display the inputted PPPOE account number. |
| 6. Press the key to return to the previous menu, then press the key, the LCD screen will display "PASSWORD". Then press the ENTER key, and the DEL key, input your |
| PPPoE's password and confirm it by the Key, the LCD screen will display the password which you inputted. |
| 7. Press the EXIT key for four times and press the DOWN key, till the LCD screen display "SYSTEM". |
| 8. Press the ENTER key, the screen display "SAVE", then press the key again, the LCD screen will display "ARE YOU SURE". |
| 9. Press the key, the phone will save your setting and the LCD screen will display "SAVING NOW", then return to display "SAVE". |
| 10. Press the key twice, then press numeric key "3" and hold until the screen display "ARE |
| YOU SURE". Press the example key, the screen will display "CHANGING", which means that the |
| phone is trying to switch to PPPoE mode. If the icom "PPPoE" on the top of the screen keeps |

blink, it shows that the phone is trying to access the PPPoE server., and the IP is still static IP if



you press key to display the current IP; if the icon "PPPoE" is showed without blink, it means that the phone has already gotten IP from PPPoE server.

2.2.2. Static IP mode:

1. Press the 1 key for three seconds, then confirm it by the key, your phone network connection mode will switch into Static IP mode. Prepare your phone's network parameters. They are IP Address of this phone, Subnet Mask, Default Gateway/ Router and DNS. You can ask your VoIP service provider for those parameters.

key, the LCD screen will display "INPUT PASSWORD". 2. Press the MENU 3. Input password (default is 123), then press the key, the LCD screen will display" NETWORK". key, and the LCD screen will display "LAN". Press the 4. Press the key, the LCD screen will display "STATIC NET". key again and then 5. Press the key, the LCD screen will display "IP". Press the key, input your desired IP address for your IP phone and confirmed by pressing the the Del key, then the LCD will display the inputted IP address. When inputting IP with keypad, use "*" instead of ".". Exit key, the LCD screen key to return to previous menu, then press the 6. Press the key then the key, input your spare DNS address will display "DNS2". Press the Del and confirm it by pressing the key, and then the LCD will display the inputted DNS address. kev, the LCD screen 7. Press the key to return to previous menu, then press the will display "DNS". Press the key then the Del key, input your DNS address and key, and then the LCD will display the inputted DNS confirm it by pressing the address. key to return to the previous menu, and then press the key, the LCD 8. Press the screen will display "GATEWAY". Press the key again and then the Del key, input your gateway's IP address and confirm it by pressing the key, the LCD screen will display the inputted gateway address.



2.2.3. DHCP mode

Press the numeric key 2 and hold till the LCD screen displays "ARE YOU SURE". Press the

key, the LCD screen will display "CHANGING" and this VoIP phone is trying to switch to DHCP mode. If the icom "DHCP" on the top of the screen keeps blink, it shows that the phone is

trying to access the DHCP server., and the IP is 0.0.0.0 if you press key to display the current IP; if the icon "DHCP" is showed without blink, it means that the phone has already gotten IP from DHCP server.

3. Basic Functions

3.1. Basic operation

3.1.1. Accepting a call

There are four methods to accept an incoming call:

• Pick up handset to accept incoming calls.



- Press the selection
- If you need switch from a hands-free call to handset, please pick up the handset directly.
- If you need switch from a handset call to hands-free, please press the button, and then hang up the handset.

3.1.2. Making a call

• Quick-dialing



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

• Use handset

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

Use hands-free



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



button again to end the call.

Use the phone book

Press the Phone Book button then the

button you will enter into the phone book. Press



Use Callers

Press the CALLERS key, then select your desired phone number in callers by the

key, and next press the button to dial the call.

• Use the R/send key



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

K

3.1.3. Ending a call

• Hangs up by handset onhook



- Hangs up by press when in hands-free
- Hangs up a call in call waiting state.

If you are in call waiting state, you could press **#** key to hang up the current call, and switch to the other call to keep talking.Note: Pressing **#** key will not hang up if there is only one call currently.

3.1.4. Transferring a call

Call transfer has several ways to realize:

1. When A talks to B, B may press the **see and talk** key and dial C phone number. After B talks to C (or

B hear alert from C), B presses the Transfer key, then <u>B hangs</u> up, and A will get through to C.

2. When A is talking with B, C calls B, B may press the key to hold A, and talk to C. Then B

presses the Transfer key, A will get through to C.

3. When A talks to B, B presses the Transfer key, dial C phone number and # key, then hang up and A will get through to C.

1 and 2 are attended transfer; 3 is blind transfer.

Notice to VoIP Phone Carrier: Your VoIP phone server need support FRC3515, or else transferring can not work.

3.1.5. Calling Hold and 3 ways call

There are two modes to enjoy hold function:

1. Press the key during a call, and the call will be on hold. While a call is on hold, you can establish another call by dialing your desired number and confirm it by the # button. Pressing the

key again you will resume the first call. By using hold function, you can talk with only one

party; the other party who is on hold can't talk with you. If you press the * button or Conference key you

will enter into 3 ways call.

2. If the third party calls you during a call, the LCD screen will display the incoming call number.

Press the key to hold the first call, and then you can talk with the third party. By using hold function, you can talk with only one party; the other party who is on hold can't talk with you. If you press # key, phone will hang up the first call, and then accept the new incoming call. Notice: You must enable the calling waiting or else calling hold can't work.

3.1.6. Callers

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

• Missed Calls

Press the CALLERS key, and then the Key, till the LCD screen display "MISSED". Press the key, the LCD screen will display the missed call number and sequence

numbers of the missed call.

You can press the key to dial this phone number, you also press UP/DOWN key to

browse the other missed calls or you can press the **LCD** key again, the LCD screen will display the time of the missed calls. If there is no one missed calls, the LCD will display "LIST IS EMPTY".

Received Calls

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

You can press the key to dial this phone number, you also press key to

browse the other received calls or you can press the **show** key again, the LCD screen will show the time of the received call. If there is no one received call, the LCD will display "**LIST IS EMPTY**".

• Dialed calls

| Press the CALLERS key, and then the NV/N key, till the LCD screen display |
|--|
| Enter |
| "OUTGOING". Press key, the LCD screen will display the phone number and |
| R/Send |
| |
| sequence number of the dialed call. You can press the sequence have to dial this phone number, or |
| |
| press the Name key to browse all record of the dialed calls. If there is on one dialed calls, |

the LCD will display "LIST IS EMPTY".

3.2. The high-level operation

This VoIP Phone provides more advanced functions after setting at the permission scope of SIP server.

3.2.1. Special Keys

• Realize Secondary Dial by Dialing for only one time

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

key to postpone input, and screen display will show--. One --stands for 2 seconds. For example, you input 123--45, the phone will send DTMF(45) 2 seconds after the phone call 123. 123-----45 will make phone send DTMF(45) at 6 seconds interval.

• MWI(Message Waiting Indication)

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

3.2.2. Call pickup

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

The following chart shows how to configure an appointed prefix in dial peer to have call pick up function.

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

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

3.2.3. join call

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

The following chart shows how to configure an appointed prefix in dialpeer to have join call function.

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

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

3.2.4. redial/unredial

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

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

| Number | Destination | Port | Mode | Alias | Suffix | Del Length |
|--------|-------------|------|------|--------------|-----------|------------|
| *3*T | 0.0.0 | 5060 | SIP | rep:redial | no suffix | 3 |
| *4*T | 0.0.0 | 5060 | SIP | rep:unredial | no suffix | 3 |

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

3 plus B'phone number to make the redial function.

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

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

3.2.5. click to dial

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

4. Setting

4.1. Setting methods

VoIP Phone is different from the traditional phone; it need be set to make it active. If your VoIP service provider asks you to set this phone, you can do it easily according to the following methods. This VoIP Phone can be set via three different setting methods:

The phone key. The initial password is 123 for setting via phone key.

The web browser on PC

Telnet

This Manual will tell you about the setting methods via the web browser on PC.

4.2. Setting via Web Browse

When this phone and your PC are connected to your network, enter the IP address of the wan port in this phone as the URL (e.g. http://xxx.xxx.xxx/ or http://xxx.xxx.xxx/).

If you do not know the IP address, you can look it up on the phone's display by pressing the key "SYSINFO".

After you enter the IP address, you will see the following web interface.

| Username: | |
|-----------|------|
| Password: | |
| | ogon |

This phone provides different two privileges for different users to set it.

The two privileges are guest and administrator respectively. In guest privilege, user can see but not modify Register/Proxy Sever Addresses and ports of SIP, advance SIP and Iax2. In administrator privilege, user can see and modify all setting parameters.

Default value in guest privilege

Username: guest Password: guest

Default value in Administrator privilege Username: admin Password: admin

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

4.3. Configuration via WEB

4.3.1. BASIC

4.3.1.1. Status

| BASIC | | | | | | | | |
|--------------|---|--|------------------------------|--|--|--|--|--|
| STATUS WIZAR | STATUS WIZARD CALL LOG MMI SET | | | | | | | |
| Notwork | | | | | | | | |
| NELWOIK | | | | | | | | |
| Connect Mode | | DHCP | | | | | | |
| MAC Address | | 00:0e:10:00:7e:72 | | | | | | |
| IP Address | | 192.168.1.15 | | | | | | |
| Gateway | | 192.168.1.1 | | | | | | |
| Phone Number | | | | | | | | |
| SIP LINE 1 | @ :5060 | | Unapplied | | | | | |
| SIP LINE 2 | @ :5060 | Unapplied | | | | | | |
| | Version: VOIP PI | HONE V1.7.60.47 Mar 3 | 30 2009 16:39:26 | | | | | |
| | | Status | | | | | | |
| Field name | | | Explanation | | | | | |
| Network | Shows the including t PPPoE), N port, ON o | e configuration information on WAN and LAN port, he connect mode of WAN port (Static, DHCP, MAC address, the IP address of WAN port and LAN or OFF of DHCP mode of LAN port | | | | | | |
| Phone Number | Shows the servers. The last lir | phone numbers presences the shows the vers | brovided by the SIP LINE 1-2 | | | | | |

4.3.1.2. Wizard

| | | BASIC |
|----------------|--------------------|-------|
| STATUS WIZ | ARD CALL LOG MMI 9 | SET |
| Network Mode | Select | |
| Static IP MODE | 0 | |
| DHCP MODE | • | |
| PPPoE MODE | 0 | |
| | ВАСК | NEXT |

| Wizard | | |
|---|----------------|---|
| Field Name | | Explanation |
| Static IP MODE | ۲ | • |
| DHCP MODE | 0 | |
| PPPoE MODE | 0 | |
| Please select the proper network mode according to the network condition. FV6030 provide three different network settings: Static: If your ISP server provides you the static IP address, please select this mode, 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 petwork | | |
| Choose Static IP SIP1)easily, also | MODE can br | owse them 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 | 202.96.1 | 28.68 |
| Static IP Addre | ess | 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 inputted can not be parsed, phone will automatically domain to the end of the domain which you inputted and parse it again. | | Set DNS domain postfix. When the domain which you inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again. |
| Primary DNS | S | 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 <mark>Name</mark> | 2113 | |
| Password | •••• | |
| Phone Number | 2113 | |
| Enable Register | | |
| Display Name | | If user set the display name, callee will show this 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 Numb | er | Input the phone number assigned by your VOIP service | | |
| | | provider. | | |
| Enable Regis | ter | 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 | | |
| Sec | | | | |
| SIP | | | | |
| Register Server | 192.168 | .1.2 | | |
| Account/User Name | 2113 | | | |
| PhoneNumber | 2113 | | | |
| Register | ON | | | |
| | BACK | Finish | | |
| Display detailed i | nforma | tion that you manual config. | | |
| Choose DHCP M | IODE, d | click [NEXT] to config simple SIP(default SIP1). You can browse | | |
| it too. Click [BACK] to return to the last page. Like Static IP MODE. | | | | |
| Choose PPPoE N | Choose PPPoE MODE, click [NEXT] to config the PPPoE account/bassword and | | | |
| SIP(default SIP1) |). You c | can browse it too. Click [BACK] to return to the last page. Like | | |
| Static IP MODE | | | | |
| PPPOE Set | | | | |
| PPPOE Server | ANY | | | |
| Username | user123 | | | |
| Password | | | | |
| PPPoE Serve | er | It will be provided by ISP. | | |
| Username | | Input your ADSL account. | | |
| Password | | Input your ADSL password. | | |
| Nation Olick / Firstels 1 | | | | |

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

4.3.1.3. Call Log

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

BASIC

| STATUS WIZARD CALL LOG N | IMI SET | |
|--------------------------|-----------|---------------|
| Call information | | |
| Start Time | Last Time | Called Number |
| SEP 18 14:02 | 0 | sip:123@1 |
| | | |

| Call Log | | |
|---------------|---|--|
| Field name | explanation | |
| Start Time | Display the start time of the outgoing call | |
| Last Time | Display the conversation time of the outgoing call. | |
| Called Number | Display the account/protocol/line of the outgoing call. | |

4.3.1.4. MMI SET

| BASIC | | |
|--|----------------|--|
| STATUS WIZARD CA | LL LOG MMI SET | |
| LANGUAGE SELECTION | | |
| Language Set: | English M | |
| Greeting Message Set | | |
| Greeting Message | VOIP PHONE | |
| APPLY Version: VOIP PHONE V1.7.60.47 Mar 30 2009 16:39:26 | | |
| MMI SET | | |
| Field name | explanation | |

Language Set Set the language of phone, English is default.

4.3.2. Network

4.3.2.1. WAN Config

| | NETWO | ORK |
|-----------------|-------------------|---------|
| WAN QOS SERV | /ICE PORT SNTP | |
| WAN Status | | |
| Active IP | 192.168.1.23 | |
| Current Netmask | 255.255.255.0 | |
| Current Gateway | 192.168.1.1 | |
| MAC Address | 00:0e:10:00:66:10 | |
| WAN Setting | | |
| Static 🔿 | DHCP ④ | РРРОЕ 🔿 |
| | APPLY | |
| | | |

| WAN Config | | |
|-----------------|---------------------------------------|--|
| Field Name | explanation | |
| Active IP | 192.168.1.23 | |
| Current Netmask | 255.255.255.0 | |
| Current Gateway | 192.168.1.1 | |
| MAC Address | 00:0e:10:00:66:10 | |
| | | |
| Active IP | The current IP address of the phone. | |
| Current Netmask | The current Netmask address. | |
| MAC Address | The current MAC address of the phone. | |
| Current Gateway | The current Gateway IP address. | |
| Get MAC Time | Shows the time of getting MAC address | |

| WAN Setting | | |
|--|--|--|
| Static 💿 | DHCP O PPPOE O | |
| Please select the proper provide three different new Static: If your ISP see mode, then finish St Mode setting, pleas DHCP: In this mode automatically; need PPPoE: In this mode You can also refer to the setting the sett | r network mode according to the network condition. FV6030 etwork settings: erver provides you the static IP address, please select this tatic Mode setting. If you don't know about parameters of Static e ask your ISP for them. e, you will get the information from the DHCP server not to input this information artificially. e, your must input your ADSL account and password. o 3.2.1 Network setting to speed setting your network. | |
| 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 | |
| Auto DNS | | |
| If you use static mode, y | rou 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 inputted can not be parsed, phone will automatically add this domain to the end of the domain which you inputted before and parse it again. | |
| Primary DNS | Input your primary DNS server address. | |
| 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 finishe your setting, IP Phone will save the setting automatically and new setting will take effect. 2) If you modify IP address, the web will not response by the old IP address. Your need input new IP address in the address column to logon in the phone. 3) If networks ID which is distributed by DHCP server is same as network ID which is used by LAN of system, phone 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 | | |

networks ID(for example, system will change LAN IP from 192.168.10.1 to 192.168.11.1) when phone uses DHCP client to get IP in startup; if phone uses DHCP client to get IP in running status and network ID is also same as LAN's, phone will refuse to accept the IP to configure WAN.

4.3.2.2. Qos Config

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



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 transmition.

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

| NETWORK | | | | | |
|--------------------------------------|-----------|------------|--------------------------------|-----------|------------|
| WAN QOS SE | RVICE POR | T SNTP | | | |
| QoS Set | | | | | |
| VLAN Enable | | | | | |
| VLAN ID Check Enab | le | | Voice/Data VLAN differentiated | Undiffere | entiated 🕑 |
| 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) |
| | | | 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 untaged 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.
- 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 untaged, 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) user need 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 dismatch 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.

4.3.2.3. Service Port

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

| NETWORK | | |
|--|---------|--|
| WAN QOS SERVICE PO | RT SNTP | |
| Service Port | | |
| HTTP Port | 80 | |
| Telnet Port | 23 | |
| RTP Initial Port | 10000 | |
| RTP Port Quantity | 200 | |
| APPLY | | |
| If modify HTTP or Telnet port,you'd better set it more than 1024,then restart. | | |
| | | |

| 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: | | |

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

- 2) If you modify the port of Telnet and HTTP, you would better set the value more than 1024 because the port value less than 1024 is system port reserved.
- 3) if you set 0 for the HTTP port, it will disable HTTP service.

4.3.2.4. 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 QOS S | ERVICE PORT SNTP | |
| SNTP Time Set | | |
| Server | 209.81.9.7 | |
| Time Zone | (GMT+08:00)Beijing,Chongqing,Hong Kong | Urumqi 🛛 🖂 |
| Time Out | 60 (seconds) | |
| 12 Hours Systems | | |
| SNTP | | |
| | APPLY | |
| | | |
| Daylight Timeset | | |
| Enable Daylight | | |
| Time shift (minutes) | 60 | |
| Time Zone | Start Date | End Date |
| Month | March 🖌 | October 🖌 |
| Week | 5 🛩 | 5 🕶 |
| Day | Sunday 🖌 | Sunday 🖌 |
| Hour | 2 | 2 |
| Minute | 0 | 0 |
| | 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. | | | | |
| 12 Hours Syste | ms | Swich the time mechanism between 12 hours and 24 hours. Default is 24 hours mode | | | | |
| SNTP | | Select the SNTP, and click Apply to make the SNTP Times effective. | | | | |
| Enable Daylig | ht | Enable daylight saving time | | | | |
| Time shift(minut | es) | Setup the variety length | | | | |
| Month | | Setup stat and end month | | | | |
| Week | | Setup start and end week | | | | |
| Day | | Setup start and end day | | | | |
| Hour | | Setup start and end hours | | | | |
| Minute | | Setup start and end minutes | | | | |
| Year | | | | | | |
| Months | | | | | | |
| Day | | | | | | |
| Hour | | | | | | |
| Minute | | | | | | |
| | | APPLY | | | | |
| Notice: You need | specif | v the above all items. | | | | |

4.3.3. VOIP

4.3.3.1. SIP Config Set your SIP server in the following interface.

| SIP STUN DI | AL PEER | | | | | |
|---|---|---|---|---|--|---|
| | | | | | | |
| SIP Line Select | | | | | | |
| SIP 1 ¥ | | Load | | | | |
| | | | | | | |
| Basic Setting | | | | | | |
| Register Status | Registered | | Display Name | | | |
| Server Name | | | Proxy Server Addre | 55 | | |
| Server Address | 192.168.1.2 | | Proxy Server Port | | | |
| Server Port | 5060 | | Proxy Username | | | |
| Account Name | 111 | | Proxy Password | | | |
| Password | ••• | | Domain Realm | | | |
| Phone Number | 111 | | Enable Register | | | |
| | , | [| ΔΡΡΙΥ | | | |
| Advanced SIP Se | tting | Ad | Ivanced Set | | | |
| Advanced SIP Se | tting | Ad | Ivanced Set | | | |
| Advanced SIP Se Register Expire Time | tting 60 | seconds | Forward Type | | Off | × |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva | tting 60 11 60 | seconds seconds | Forward Type | ıber | Off | |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent | tting 60 160 Voip Phor | seconds seconds ne 1.0 | Forward Type Forward Phone Num Server Type | ıber | Off COMMO | |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key | tting 60 Il 60 Voip Phor | seconds seconds ne 1.0 | Forward Type Forward Phone Num Server Type DTMF Mode | ıber | Off COMMO DTMF_R | N V FC2833 V |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent Signal Key Media Key | tting 60 II 60 Voip Phor | seconds seconds ne 1.0 | Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition | iber | Off COMMO DTMF_R RFC326 | N Y FC2833 V 1 V |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent Signal Key Media Key Local Port | tting 60 1 60 Voip Phor 5060 | seconds seconds ne 1.0 | Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol | ıber | Off COMMO DTMF_R RFC326 UDP 💌 | N V FC2833 V 1 V |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent Signal Key Media Key Local Port Ring Type | tting 60 10 60 Voip Phor 5060 Type 10 | seconds seconds ne 1.0 | Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition | iber | Off COMMO DTMF_R RFC326 UDP ♥ NONE | ▼ N ▼ FC2833 ▼ 1 ▼ |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port Ring Type Subscribe Expire Time | tting 60 10 60 Voip Phor 5060 Type 10 300 12121 | seconds seconds ne 1.0 | Manced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time | iber | Off COMMO DTMF_R RFC326 UDP NONE | N V FC2833 V 1 Seconds |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent Signal Key Media Key Local Port Ring Type Subscribe Expire Time Conference Number | tting 60 10 60 Voip Phor 5060 Type 10 300 12121 | seconds seconds ne 1.0 | Manced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N | iber | Off COMMO DTMF_R RFC326 UDP NONE | N V IFC2833 V 1 V seconds |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port Ring Type Subscribe Expire Time Conference Number Stable DNS SRV | tting 60 10 60 Voip Phor 5060 Type 10 300 12121 □ | seconds seconds ne 1.0 | Vanced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N Enable Displayname Click To Talk | iber | Off COMMO DTMF_R RFC326 UDP 💌 NONE 0 | ▼ N ▼ 3FC2833 ▼ 1 ▼ seconds |
| Advanced SIP Se Register Expire Time VAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port Ring Type Subscribe Expire Time Conference Number Enable DNS SRV Enable Subscribe | tting 60 60 Voip Phor 5060 Type 10 300 12121 1 1 1 1 1 1 1 1 1 1 1 1 1 | seconds seconds me 1.0 | Image: Set s | iber e lumber Quote | Off COMMO DTMF_R RFC326 UDP ♥ NONE 0 | N V FC2833 V 1 V seconds |
| Advanced SIP Se Register Expire Time NAT Keep Alive Interva Jser Agent Signal Key Media Key Local Port Ring Type Subscribe Expire Time Conference Number Enable DNS SRV Enable Subscribe Enable Keep Authentica NAT Keep Alive | tting 60 10 60 Voip Phor 5060 Type 10 300 12121 □ 12121 □ 12121 □ 12121 □ 12121 □ 12121 | seconds seconds me 1.0 | Image: Set | aber a a a a a a a a a a a a a a a a a a a | Off COMMO DTMF_R RFC326 UDP ♥ NONE 0 □ □ □ □ □ | N V FC2833 V 1 V seconds |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port Ring Type Subscribe Expire Time Conference Number Sinable DNS SRV Inable Subscribe Sinable Keep Authentica IAT Keep Alive | tting 60 10 60 10 500 5060 7ype 10 300 12121 □ 10 □ 110 10 □ 10 □ | seconds seconds ne 1.0 seconds | Image: Set | e Number Quote | Off COMMO DTMF_R RFC326 UDP NONE 0 | N N FC2833 N 1 Seconds |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port Ring Type Subscribe Expire Time Conference Number Sinable DNS SRV Sinable Subscribe Sinable Subscribe Sinable Keep Authentica IAT Keep Alive Sinable Via rport Sinable PRACK | tting 60 60 60 7000 Phor 5060 7000 Phor 5060 7000 Phor 1000 Phor <td>seconds seconds ne 1.0</td> <td>Image: Set Set Set Set Set Set Set Set Set Set</td> <td>lber lber l l l l l l l l l l l l l l l l l l l</td> <td>Off COMMO DTMF_R RFC326 UDP M NONE 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0</td> <td>N V FC2833 V 1 V seconds</td> | seconds seconds ne 1.0 | Image: Set | lber lber l l l l l l l l l l l l l l l l l l l | Off COMMO DTMF_R RFC326 UDP M NONE 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 | N V FC2833 V 1 V seconds |
| Advanced SIP Se tegister Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Local Port ting Type Subscribe Expire Time Conference Number Sinable DNS SRV Sinable Subscribe Sinable Keep Authentica IAT Keep Alive Sinable Via rport Sinable PRACK Long Contact | tting 60 60 Voip Phor 5060 Type 10 300 12121 0 0 12121 0 0 0 0 0 0 0 0 0 0 0 0 0 | seconds seconds ne 1.0 | Wanced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N Enable Displayname Click To Talk Signal Encode Rtp Encode Enable Session Time Answer With Single Auto TCP | aber aber a a a a a a a a a a a a a a a a a a a | Off COMMO DTMF_R RFC326 UDP NONE 0 | ▼ N IFC2833 1 ✓ seconds |
| Advanced SIP Se Register Expire Time IAT Keep Alive Interva Jser Agent Signal Key Aedia Key Aedia Key Aedia Key Cocal Port Ring Type Subscribe Expire Time Conference Number Subscribe Expire Time Conference Number Subscribe Subscribe Stable DNS SRV Stable Subscribe Stable Subscr | tting 60 60 Voip Phor 5060 Type 10 300 12121 12121 | seconds seconds me 1.0 | Wanced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N Enable Displayname Click To Talk Signal Encode Enable Session Time Answer With Single Auto TCP Enable Strict Proxy | e Aumber Quote er Codec | Off COMMO DTMF_R RFC326 UDP ♥ NONE 0 0 0 0 0 0 0 0 0 0 0 0 0 | ▼ N ▼ EFC2833 ▼ 1 ▼ seconds ● |
| Advanced SIP Se Register Expire Time VAT Keep Alive Interva Jser Agent Signal Key Media Key Local Port Ring Type Subscribe Expire Time Conference Number Enable DNS SRV Enable Subscribe Enable Subscribe Enable Keep Authentica VAT Keep Alive Enable Via rport Enable PRACK Long Contact Enable URI Convert Dial Without Register | tting 60 10 60 10 60 10 7010 10 5060 10 5060 12121 1 12121 <td>seconds seconds me 1.0</td> <td>Wanced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N Enable Displayname Click To Talk Signal Encode Rtp Encode Enable Session Time Auto TCP Enable GRUU</td> <td>lber l l l l l l l l l l l l l l l l l l l</td> <td>Off COMMO DTMF_R RFC326 UDP ♥ NONE 0</td> <td>▼ N ▼ FC2833 ▼ 1 ▼ seconds ●</td> | seconds seconds me 1.0 | Wanced Set Forward Type Forward Phone Num Server Type DTMF Mode RFC Protocol Edition Transport Protocol RFC Privacy Edition Transfer Expire Time Enable Conference N Enable Displayname Click To Talk Signal Encode Rtp Encode Enable Session Time Auto TCP Enable GRUU | lber l l l l l l l l l l l l l l l l l l l | Off COMMO DTMF_R RFC326 UDP ♥ NONE 0 | ▼ N ▼ FC2833 ▼ 1 ▼ seconds ● |

| SIP Config | | | | | | |
|---------------------------------------|---|--|--|--|--|--|
| Field name | explanation | | | | | |
| SIP Line Select | | | | | | |
| SIP 1 💌 | Load | | | | | |
| Choose line to set info | about SIP, there are 2 lines to choose. You can switch by | | | | | |
| [Load] button. | | | | | | |
| Register Status | Shows if the phone has been registered the SIP server or | | | | | |
| | not; or so, show Unapplied; | | | | | |
| Server Name | Set the server name. | | | | | |
| Server Address | Input your SIP server address. | | | | | |
| Server Port Set your SIP server port. | | | | | | |
| Account Name | Input your SIP register account name. | | | | | |
| Password | Input your SIP register password. | | | | | |

| Phone Number | Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured. | | | | |
|-------------------------------|---|--|--|--|--|
| Display Name | Set the display name. | | | | |
| Proxy Server Address | Set proxy server IP address (Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if your VoIP service provider give different configurations between Register SIP Server and Proxy SIP Server, you need make different settings.) | | | | |
| Proxy Server Port | Set your Proxy SIP server port. | | | | |
| Proxy Username | Input your Proxy SIP server account. | | | | |
| Proxy Password | Input your Proxy SIP server password. | | | | |
| Domain Realm | Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address). | | | | |
| Enable Register | Start to register or not by selecting it or not. | | | | |
| Register Expire Time | Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expire time set, the phone will change automatically the time into the time recommended by the server, and register again. | | | | |
| NAT Keep Alive Interval | Set examining interval of the server, default is 60 seconds | | | | |
| User Agent | Set the user agent if have, the default is VoIP Phone 1.0 | | | | |
| Signal Key | Set the key for signal encryption | | | | |
| Media Key | Set the key for RTP encryption | | | | |
| Local port | Set sip port of each line | | | | |
| Ring type | Set ring type of each line | | | | |
| Subscribe Expire Time | Set the interval of Subscribe. | | | | |
| Conference Number | Set the server conference number to jion the the room | | | | |
| Enable DNS SRV | Support DNS looking up with _sip.udp mode | | | | |
| Enable Subscribe | Enable Subscribe. | | | | |
| Enable Keep Authentication | Enable/Disable Keep Authentication. | | | | |
| NAT Keep Alive | Enable/Disable keeps NAT of SIP alive. If some server refuse to register with too short interval time, and has no packets sending to device in private network to keep NAT alive, user could set this function ON. It need set the keep alive interval time less than the NAT server's. | | | | |
| Enable Via rport | Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT. | | | | |
| Enable PRACK | Enable or disable SIP PRACK function, suggest use the default config. | | | | |
| Long Contact | Set more parameters in contact field; connection with SEM server | | | | |
| Enable URI Convert | Convert # to %23 when send the URI. | | | | |
| Dial Without Register | Set call out by proxy without registration; | | | | |
| Ban Anonymous Call | Set to ban Anonymous Call; | | | | |
| Forward Type | Select call forward mode, the default is Off Off: Close down calling forward Busy: If the phone is busy, incoming calls will be forwarded to the appointed phone. | | | | |
| | No answer: If there is no answer, incoming calls will be forwarded to the appointed phone. Always: Incoming calls will be forwarded to the appoint | | | | |

| | phone directly. |
|----------------------|--|
| | The phone will Prompt the incoming while doing forward. |
| Forward Phone Number | Appoint your forward phone number. |
| Server Type | Select the special type of server which is encrypted, or has |
| | some unique requirements or call flows. |
| | Select DTMF sending mode, there are three modes: |
| | DTMF_RELAY |
| DTMF Mode | DTMF_RFC2833 |
| | DTMF_SIP_INFO |
| | Different VoIP Service providers may provide different |
| | modes. |
| | Select SIP protocol version to adapt for the SIP server |
| RFC Protocol Edition | which uses the same version as you select. For example, if |
| | the server is CISCO5300, you need to change to RFC2543, |
| | else phone may not cancel call normally. System uses |
| | RFC3261 as default. |
| Transport Protocol | Set transport protocols, TCP or UDP; |
| RFC Privacy Edition | Set Anonymous call out safely; Support RFC3323and |
| | RFC3325; |
| Transfer Expire Time | The phone send bye and end the call as soon as hang up. |
| Enable Conference | Enable/Disable conference |
| Number | |
| Enable Displayname | Set to make quotation mark to displayname as the phone |
| Quote | sends out signal, in order to be compatible with server. |
| Click to Talk | Set click to Talk (need practical software support). |
| Signal Encode | Enable/Disable Signal Encrypt. |
| RTP Encode | Enable/Disable RTP Encrypt. |
| Enable Session Timer | Set Enable/Disable Session Timer, whether support |
| | RFC4028.It will refresh the SIP sessions. |
| Answer With Single | Enable/Disable the function when call is incoming, phone |
| Codec | replies SIP message with just one codec which phone |
| | supports. |
| Auto TCP | Set to use automatically TCP protocol to guarantee usability |
| | of transport as message is above 1300 byte |
| Enable Strict Proxy | Support the special SIP server-when phone recieves the |
| | patckets sent from server, phone will use the source IP |
| | address, not the address in via field. |
| Enable CRUU | Set to support GRUU |

4.3.3.2. Stun Config

In this web page, you can config SIP STUN.

STUN:

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



VOIP

| SIP STUN DIAL PE | ER | | |
|-----------------------|-------|---------|--|
| STUN Set | | | |
| STUN NAT Transverse | FALSE | | |
| STUN Server Addr | | | |
| STUN Server Port | 3478 | | |
| STUN Effect Time | 50 | Seconds | |
| Local SIP Port | 5060 | | |
| | | APPLY | |
| | | | |
| Set Sip Line Enable S | tun | | |
| SIP 1 🔽 | Load | | |
| | | | |
| | | | |
| Jse Stun | | | |
| | | 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 | Port Set your SIP STUN Server Port | | | | | | |
| Set STUN Effective Time. If NAT server finds that a N | | | | | | | |
| STUN Effect Time mapping is idle after time out, it will release the mappin | | | | | | | |
| the system need send a STUN packet to keep the man | | | | | | | |
| | effective and alive. | | | | | | |
| Local SIP Port | Set the SIP port. | | | | | | |
| Set Sip Line Enable Stun | | | | | | | |
| SIP 1 💌 | Load | | | | | | |
| Choose line to set info abou | t SIP, There are 2 lines to choose. You can switch by 【Load】 | | | | | | |
| button. | | | | | | | |
| Use Stun | Enable/Disable SIP STUN. | | | | | | |
| Notice: SIP STUN is used to | Notice: SIP STUN is used to realize SIP penetration to NAT. If your phone configures STUN | | | | | | |
| Server IP and Port (default is | 3478), and enable SIP Stun, you can use the ordinary SIP | | | | | | |
| Server to realize penetration | to NAT. | | | | | | |

4.3.3.3. DIAL PEER setting

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule. When you want to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, you can set number 156 to replace 192.168.1.119 here.

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

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

| | | | | | | • | |
|--------|-------------|------|------|---------|-----------|------------|--|
| Number | Destination | Port | Mode | Alias | Suffix | Del Length | |
| 1T | 0.0.0 | 5060 | SIP | rep:010 | no suffix | 1 | |

То

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

| Number | Destination | Port | Mode | Alias | Suffix | Del Length |
|----------------|-------------|------|------|-------|-----------|------------|
| 13xxxxxxxxx | 0.0.0 | 5060 | SIP | add:0 | no suffix | 0 |
| 13[5-9]xxxxxxx | 0.0.0 | 5060 | SIP | add:0 | no suffix | 0 |

 $1 \cdot x$ Match any single digit that is dialed.

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

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

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

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

...

| | | | | VOIP | | | |
|---------------|---------------|-------|------|---------|-------|--------|------------|
| SIP ST | TUN DIAL PEER | | | | | | |
| Dial Peer | Table | | | | | | |
| Number | Destination | | Port | Mode | Alias | Suffix | Del Length |
| Add Dial I | Peer | | | | | | |
| Phone Numb | er | | | | | | |
| Destination (| (optional) | | | | | | |
| Port(optiona | il) | | | | | | |
| Alias(option | al) | | | | | | |
| Call Mode | | SIP 🚩 | | | | | |
| Suffix(option | nal) | | | | | | |
| Delete Lengt | th (optional) | | | | | | |
| | | | | Submit | | | |
| | | | | | | | |
| Dial Peer | Option | | | | | | |
| | | | De | lete Mo | lify | | |
| | | | | | | | |
| | | | DI | AL PE | ER | | |

31

| Field name | | | explar | nation | |
|----------------------------|-------------|-------------------------------|--------------------|-----------------|-----------------------|
| | There | are two type | s of matchin | g conditions | : one is full |
| | matchi | ng, the othe | is prefix ma | tching. In th | e Full matching, |
| | you ne | ed input you | r desired ph | one number | in this blank, and |
| Phone number | then yo | bu need dial | the phone nu | umber to rea | lize calling to what |
| | the pho | one number | is mapped. I | n the prefix | matching, you |
| | need II | nput your de | sired prefix r | number and | I; then dial the |
| | prefix a | and a phone | number to re | ealize calling | g to what your |
| | 30 digi | ts | apped. The p | | r supports at most |
| | Set De | estination ad | dress. This | is optional | config item. If you |
| Destination | want t | o set peer | to peer call | l, please in | put destination IP |
| | addres | s or domain | name. If yo | ou want to u | se this dial rule in |
| | SIP2 li | ne, you need | l input 255.2 | 55.255.255 | or 0.0.0.2 in it. |
| Port | Set the | Signal port | the default | is 5060 for S | SIP. |
| Alias | set ali | as. This is o ow no alias. | otional config | g item. If you | u don't set Alias, it |
| Note: There are four ty | pes of ali | ases. | | | |
| 1) add: xxx, it means the | nat you ne | eed dial xxx | in front of ph | one number | , which will reduce |
| dialing number lengt | h. | | | | |
| 2) all: xxx, it means that | t xxx will | replace som | e phone nur | nber. | |
| 3) del: It means that ph | one will o | delete the nu | mber with le | ngth appoin | ted. |
| 4) Rep: It means that p | hone will | replace the | number with | length and r | number appointed. |
| You can refer to the fo | lowing ex | amples of d | ifferent alias | application | to know more how |
| to use different allases | and this | diai ruie. | | | 0 |
| | Select | differenct sig | gnal protocol | , SIP or IAX | <u>Z</u> |
| Suffix | you do | n't set it. | ptional confi | g item. It wi | II SNOW NO SUITIX IT |
| Delete Length | Set de | lete length. | This is optior | nal config ite | m. For example: if |
| _ | the de | lete length is | s 3, the pho | ne will delet | te the first 3 digits |
| | then s | end out the | rest digits. | You can ref | er to examples of |
| | differe | nt alias appli | cation to kno | ow how to se | et delete length. |
| Introduction of how to a | set up dia | I-peer to imp | lement swite | ch between i | multi- SIP lines |
| Number Destination | Port | Mode | Alias | Suffix | Del Length |
| 9T 0.0.0.1 | 506 | 0 SIP | no alias | no suffix | 0 |
| 8T 0.0.0.2 | 506 | 0 SIP | no alias | no suffix | 0 |
| 9T mapping: If you ha | ve regist | ered a SIP1 | server and | set dial-pee | r according to the |
| above table, all calls v | /ill be sen | t via SIP1 se | erver when y | ou press the | numeric key "9" in |
| front of dialing destinat | ion phone | e numbers. | | | |
| 8T mapping: If you hav | e registe | ed a Private | SIP2 server | and set dia | -peer according to |
| the above table, all ca | lls will be | e sent via SII | P2 server wh | nen you pres | s the numeric key |
| "8" in front of dialing | destinatio | n phone nun | nbers. | | |
| Number Destination | P | ort Mode | Alias | Suffix | Del length |
| the rule of 2T means user | need to dis | l the number w | with prefix 2 if h | ne want to dial | via IAX2 server |
| Examples of different | alias ann | lication | | | |
| | | | | | |

| Set by web | explanation | example |
|------------|-------------|---------|
|------------|-------------|---------|

| Phone Number 9T Destination (optional) 255.255.255.255 Port(optional) | You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT, Destination is 255.255.255.255 and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length. | If you dial "93333", the SIP2 server will receive "3333" |
|---|--|--|
| Phone Number 2 Destination (optional) Image: Constraint of the system o | This setting will realize speed dial function, after you dialing the numeric key "2", the number after all will be sent out. | When you dial "2", the SIP1 server will receive 33334444 |
| Phone Number BT Destination (optional) Image: Constraint of the second | The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number. | When you dial "8309", the SIP1 server will receive "07558309" |
| Phone Number 010T Destination (optional) | You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is Rep:xxx If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out. | When you dial "0106228", the SIP1 server will receive "0086106228" |

| Phone Number Destination (optional) Port(optional) Alias(optional) Call Mode Suffix(optional) Delete Length (optional) | 147 SIP v 0011 | If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number. | When you dial "147", the SIP1 server will receive "1470011" |
|--|----------------------------|---|--|
|--|----------------------------|---|--|

4.3.4. Phone

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

PHONE

| DSP CALL SERVICE DIGITAL MAP PHONE BOOK | | | | |
|---|---------|---------------------------------------|-------------------|---------------|
| | | | | |
| DSP Configuration | on | | | |
| First Codec | g711Ula | w64k 🚩 | Second Codec | g711Alaw64k ⊻ |
| Third Codec | g729 | ✓ | Fourth Codec | g723 💌 |
| Fifth Codec | g726-32 | · · · · · · · · · · · · · · · · · · · | Sixth Codec | g722 💌 |
| Handdown Time | 200 | ms | Default Ring Type | Type 1 💌 |
| Input Volume | 3 | (1-9) | Output Volume | 5 (1-9) |
| Handfree Volume | 5 | (1-9) | Ring Volume | 5 (1-9) |
| G729 Payload Length | 20ms 🚩 | | Signal Standard | China 💌 |
| G722 Timestamps | 160/20r | ns 🚩 | G723 Bit Rate | 6.3kb/s 📉 |
| VAD | | | | |
| APPLY | | | | |
| | | | | |

| DSP Configuration | | | |
|---------------------|--|--|--|
| Field name | explanation | | |
| First Codec | The fist preferential DSP codec: G.711A/u, G.722, G.723, G.729, G.726 | | |
| Second Codec | The second preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726 | | |
| Third Codec | The third preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726 | | |
| Forth Codec | The forth preferential DSP codec: G.711A/u, G.722, G.723, G.729,g.726 | | |
| Fifth Codec | The fifth preferential DSP codec: G.711A/u, G.722, G.723, G.729, G.726 | | |
| Sixth Codec | The sixth preferential DSP codec: G.711A/u, G.722, G.723, G.729, G.726 | | |
| 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. | | |
| Output Volume | Specify Output (receiver) Volume grade. | | |
| Ring Volume | Specify Ring Volume grade | | |
| G722 Timestamps | 160/20ms or 320/20ms is available | | |

| G723 Bit Rate | 5.3kb/s or 6.3kb/s is available |
|-------------------|--|
| Default Ring Type | Set up the ring by default |
| 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, white list Limit List and so on.

| PHONE | | | | |
|-----------------------|-------------------|---------------------|--------------|--|
| DSP CALL SERVICE | DIGITAL MAP PHONE | воок | | |
| Call Service Setting | | | | |
| Hot Line | | No Answer Time | 20 (seconds) | |
| P2P IP Prefix | | MWI Number | | |
| Enable Call Transfer | | Enable Call Waiting | | |
| Enable Three Way Call | | Accept Any Call | | |
| Auto Answer | | Ban Outgoing | | |
| Do Not Disturb | | | | |
| APPLY | | | | |
| | | | | |
| Black List | | | | |
| Black List | | | | |
| | Add | | Delete | |
| | | | | |
| Limit List | | | | |
| Limit List | | | | |
| | Add | | Delete | |
| | | | | |

| 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. | | |
| MWI Number | Set the number to listen voice mail in server. | | |
| Enable Call Transfer | Enable Call Transfer by selecting it. | | |
| Enable Call Waiting | Enable Call Waiting by selecting it. | | |
| Enable Three Way Call | Enable Three Way Call | | |
| Accept Any Call | If select it, the phone will accept the call even if the called number is not belong to the phone. | | |
| Auto Answer | If select it, the phone will auto answer when there is an incoming call. | | |
| Ban Outgoing | If you select Ban Outgoing to enable it, and you can not dial out any number. | | |
| Do Not Disturb | Select NO Disturb, the phone will reject any incoming call, the callers will be reminded by busy, but any outgoing call from the phone will work | | |

| | well. |
|--------------------------|---|
| Dia als Lint | Set Add/Delete Black list. If user does not want to answer |
| BIACK LIST | some phone calls, add these phone numbers to the Black List, |
| | and these calls will be rejected. |
| | x and . are wildcard. x means matching any single digit. for |
| | example, 4xxx expresses any number with prefix 4 which |
| | length is 4 will be forbidden to dialed out |
| | DOT (.) means matching any arbitrary number digit. for |
| | example, 6. expresses any number with prefix 6 will be |
| | forbidden to dialed out. |
| | if user wants to allow a number or a series of number |
| | incoming, he may add the number(s) to the list as the white list |
| | rule. the configuration rule is -number, for example, -123456, |
| | or -1234xx |
| | Black List -4119 |
| | · · · · |
| | Means any incoming number is forbidden except for 4119 |
| | Note: End with DOT (.) when set up the white list |
| | Set Add/Delete Limit List. Please input the prefix of those |
| Limit List | phone numbers which you forbid the phone to dial out. For |
| | example, if you want to forbid those phones of 001 as prefix to |
| | be dialed out, you need input 001 in the blank of limit list, and |
| | then you can not dial out any phone number whose prefix is |
| | 001. |
| | x and . are wildcard. x means matching any single digit. for |
| | example, 4xxx expresses any number with prefix 4 which |
| | length is 4 will be forbidden to dialed out |
| | . means matching any arbitrary number digit. for example, 6. |
| | expresses any number with prefix 6 will be forbidden to dialed out. |
| Notice: Black List and L | imit List can record at most10 items respectively. |

4.3.4.3. Digital Map Configuration

This phone supports 4 dial modes:

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

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

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

4). User defined: you can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing.

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

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

| PHONE | | | | |
|---------|-------------------|--------------------|-------|--|
| DSP | CALL SERVICE DIGI | TAL MAP PHONE BOOK | | |
| Digital | Map Set | | | |
| | End With "#" | | | |
| | Fixed Length | 11 | | |
| | Time Out | 5 | (330) | |
| APPLY | | | | |
| | | | | |
| Digital | Rule table | | | |
| | | Rules: | | |
| | | Add | 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. |
| | Set the timeout of the last dial digit. The call will be sent after |
| Time out | timeout. |
| Digital Rule table | |
| | Rules: |
| | Add Del |
| commas, or a list of digit x Match any single digit . Match any arbitrary nur Tn Indicates an addition n is mandatory and ca characters of a dial plan dial plans. | s. that is dialed. mber of digits including none. al time out period before digits are sent of n seconds in length. n have a value of 0 to 9 seconds. Tn must be the last 2 . If Tn is not specified it is assumed to be T0 by default on all |
| "[1-8]xxx" | |
| "9xxxxxx" | |
| "911" | |
| "99T4" | |
| "9911x.T4" | |
| [1-8]xxx: Cause extension 9xxxxxx: Cause 8 digit 911: Cause 911 to be dia 99T4: Cause 99 to be dia 9911x.T4:Cause any nu ceases. | ons 1000-8999 to be dialed immediately numbers started with 9 to be dialed immediately aled immediately after it is entered. aled after 4 seconds. Imber started with 9911 to be dialed 4 seconds after dialing |
| Notice: End with "#", Fix simultaneously, S rules. | ed Length, Time out and Digital Map Table can be used ystem will stop dialing and send number according to your set |

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 | DSP CALL SERVICE DIGITAL MAP PHONE BOOK | | | |
| Phonebook Table | | | | |
| Index | Name | Number | Туре | |
| | | | | |
| Add Phone Book | | | | |
| Name | | | | |
| Number | | | Add | |
| Ring Type | Default 🛩 | | | |
| | | | | |
| Phone Book Option | | | | |
| Delete Modify | | | | |

| Phone Book | | | | | | |
|--|------------------------|--|---|--|---------|--|
| Field name | | | explanation | | | |
| Index | Name | | Number | | Туре | |
| 1 | ad | | 23 | | Default | |
| | | | | | | |
| Shows the detail of current phonebook. | | | | | | |
| Name | Name Shows the | | he name corresponding to the phone number | | | |
| Number | Shows the phone number | | | | | |
| Ring Type Shows the ring type of the incoming call. | | | 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 | | | | | | |

4.3.5. 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 | 0.0.0.0 | | |
| Username | user | | |
| Password | •••• | | |
| Config File Name | | | |
| Config Encrypt Key | | | |
| Protocol Type | FTP ¥ | | |
| Update Interval Time | 1 | Hour | |
| Update Mode | Disable | ▼ | |
| | AP | PLY | |
| | | | |
| Auto Provision | | | |
| Field name | | explanation | |
| Current Config | Show the current | config file's version. | |
| Version | | | |
| Server Address | Set FTP/TFTP/HT | TP 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. | | |
| | Different update modes: | | |
| | 1. Disable: means no update | | |
| Update Mode | 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 | 0.0.0 | |
| Server Port | 514 | |
| MGR Log Level | None 💌 | |
| SIP Log Level | None ¥ | |
| IAX2 Log Level | None ¥ | |
| Enable Syslog | | |
| 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 | | | |
| F | Press the "Save" button to save the configuration files ! | | |
| | Save | | |
| 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 ! | | | |
| Clear | | | |
| | | | |
| | 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 (SIP1-2 and IAX2) and version number. | | |

4.3.5.4. Update

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

| Update | | | |
|------------|--|--|--|
| Field name | explanation | | |
| Web Update | Click the browse button, find out the config file saved before or provided by manufacturer, download it to the phone directly, press "Update" to save. You can also update | | |

| | downloaded update file, logo picture, ring, mmiset file by web. | | | |
|--|---|--|--|--|
| Server | 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. | | | |
| Notice: You can modify the exported config file. And you can also download config file | | | | |
| which includes several | which includes several modules that need to be imported. For example, you can | | | |
| download a config file ju | ust keep with SIP module. After reboot, other modules of system | | | |
| still use previous setting | and are not lost. | | | |
| | Action type that system want to execute: | | | |
| | 1. Application update: download system update file | | | |
| Туре | 2. Config file export: Upload the config file to FTP/TFTP | | | |
| | server, name and save it. | | | |
| | 3. Config fie import: Download the config file to phone from | | | |
| | FTP/TFTP server. The configuration will be effective after | | | |
| | the phone is reset. | | | |
| 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 SYSL | OG CONFIG | UPDATE | ACCOUNT REBOOT | |
| | | | | |
| Set Keyboard Passwoi | rd | | | |
| Keyboard Password | ••• | | | Set |
| | | | | |
| User Set | | | | |
| User | Name | | User Lev | rel |
| adı | admin | | Root | |
| gu | est | | General | |
| | | | | |
| Add User | | | | |
| User Name | | | | |
| User Level | Root | ¥ | | |
| Password | | | | |
| Confirm | | | | |
| Submit | | | | |
| | | | | |
| Account Option | | | | |
| admin 🝸 Delete 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. | |

| User Name | | User Level | |
|--|-----------------------|--|--|
| admin | | Root | |
| guest | | General | |
| This table shows the current user existed. | | | |
| User Name Set account user r | | name. | |
| User Level Set user level, Roc | | Root user has the right to modify configuration, | |
| General can only re | | ead. | |
| Password | Set the password. | | |
| Confirm | Confirm the password. | | |
| Select the account and click the Modify to modify the selected account, and click the | | | |
| Delete to delete the selected account. | | | |
| General user only can add the user whose level is General. | | | |

4.3.5.6. Reboot

MAINTENANCE

| AUTO PROVISION | SYSLOG CONFIG UPDATE ACCOUNT REBOOT |
|----------------|---|
| Reboot Phone | |
| | Press the "Reboot" button to reboot Phone ! |
| | 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 | | | | | | |
|----------------------|-------|--------|--|--------|--|--|
| | EWALL | | | | | |
| MMI Filter Table | | | | | | |
| Start IP | | End IP | | Option | | |
| MMI Filter Table | Set | | | | | |
| Start IP | | End IP | | Add | | |
| MMI Filter Table Set | | | | | | |
| MMI Filter | | APPLY | | | | |

| 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 | Field name explanation | | | | | | |
| MMI Filter Table | MMI Filter Table | | | | | | |
| Start IP Option | | | | | | | |
| 192.168.1.15 192.168.1.20 Modify Delete | | | | | | | |
| MMI Fileter IP Table list: | | | | | | | |

| MMI Filter Table Set | | | | | | |
|---|--------|--|-----|--|--|--|
| Start IP | End IP | | 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 | | | | | | | | |
|-------------|-----------------------------|---------|--------------|---------|--------------------------------|------------|-----------------|--------------|--------|
| MM | MMI FILTER FIREWALL NAT VPN | | | | | | | | |
| | | | | | | | | | |
| Fire | wall Type | | | | | | | | |
| | | 🗌 In_a | ccess Enable | | | | Out_access Enal | ble | |
| | | | | | APPLY | <u> </u> | | | |
| | | | | | | | | | |
| Fire | wall Input | Rule T | able | | | | | | |
| Index | Deny/Permit | Protoco | Src Addr | Src Mas | k | Des Addr | Des Mask | Range | Port |
| | | | | | | | | | |
| Fire | wa <mark>ll</mark> Outpu | t Rule | Table | | | | | | |
| Index | Deny/Permit | Protoco | Src Addr | Src Mas | k | Des Addr | Des Mask | Range | Port |
| 0 | deny | ICMP | 192.168.1.14 | 255.25 | 5.255.0 192.168.1.118 255.255. | | 255.255.255.0 | .0 more than | 1 |
| | | | | | | | | | |
| Fire | wall Set | | | | | | | | |
| Input | /Output | Inp | ut 💌 | | Src Addr | | | | |
| Deny | /Permit | De | ny 👻 | | Des Addı | r 🔽 | | | |
| Proto | col Type | e UDP 👻 | | | Src Mask | د | | | Add |
| Port F | ort Range more than 🖌 | | Des Mask | | | | | | |
| | | | | | | | | | |
| Rule Delete | | | | | | | | | |
| Input | /Output | Inp | ut ⊻ | | Index To | Be Deleted | | | 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

| Input/Output | Input | · | | Src Addr | | | | |
|---|---|--|--------------------------------------|----------------------|----------------------|---------------|----------|--------|
| Deny/Permit | Deny | 2 | | Des Add | • | | | 0.dd |
| Protocol Type | UDP 😽 | | | Src Mask | : | | | Aud |
| Port Range | more tha | an 🖌 | | Des Mas | k | | | |
| Field name | е | | | | explai | nation | | |
| In_access ena | ıble | Select it to Enable in_ access rule | | | | | | |
| out_access ena | able | Select it | Select it to Enable out_ access rule | | | | | |
| Input/Outpu | t | Specify current adding rule by selecting input rule or output rule. | | | | | | |
| Deny/Permi | t | Specify current adding rule by selecting Deny rule or Permit rule. | | | | | | |
| Protocol Typ | e | Filter pro | otoco | ol type. Y | ou can sele | ect TCP, UDP, | ICMP, or | IP. |
| Port Range | • | Set the | ilter | Port ran | ge | | | |
| Src Addr | Src Addr Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address simil | | | network imilar to | | | | |
| Des Addr | Des Addr Set the destination address. It can be IP address, netwo address, complete address 0.0.0.0, or network address similar | | | | network imilar to | | | |
| Src Mask | Set the source address' mask. For example, 255.255.255.2 Src Mask means just point to one host; 255.255.255.0 means point to network ID is C type. | | | | 255.255 pint to a | | | |
| 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 but | tton if y | ou want | to ad | ld a new | output rule | е. | | |
| Firewall Output Rule Table | | | | | | | | |
| Index Deny/Permit Prot | ddr | Src Ma | isk | Des Addr | Des Mask | Range | Port | |
| 0 deny ICM | 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 t | o delete | the s | selected | rule. | | | |

4.3.7. Logout

| System Logout |
|---|
| Logout |
| Press the "Logout" button to Logout Phone ! |
| Logout |

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

4.4. Settings via phone's keyboard.

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

- Press Menu, Up/Down, Enter and exit key to browse, select, and cancel
- Use the Up/Down key to browse the menu and submenu

• Use the ENTER key to enter into submenu and confirm your operation, the EXIT key can be used to back and cancel operation.

4.4.2. Phone menu

Phone main menu:



5. Appendix

5.1. Specification

5.1.1. Device specification

| Item | | this VoIP Phone |
|--------------------------|------------|--|
| Adapter(Input/Output) | | Input:100-240VAC 50~60Hz Output:5V/1A |
| Port | WAN | 10/100Base- T RJ-45 for LAN, Auto MDIX |
| FUIL | LAN | 10/100Base- T RJ-45 for PC, Auto MDIX |
| Power Co | onsumption | Idle:1.5W/Active:1.8W |
| LCI | D size | 74 x 28mm |
| Operation Temperature | | 0∼40 °C |
| Relative Humidity | | 10~65% |
| Main | Chipset | Broadcom |
| SDRAM | | 8Mbits |
| Flash | | 2Mbits |
| Size (W x H x D) | | 11.6×8×3 in.(295×205×75mm) |
| Weight | | 2.07lb.(0.94kg) |

5.1.2. Voice Features

- Support 2 lines SIP, SIP 2.0 (RFC3261)
- Codec: G.711A/u, G.7231 high/low, G.729, G.722, G.726
- Echo cancellation: Support G.168 and hand-free can support 96ms
- Support VAD, CNG
- NAT transverse: support STUN
- Supports full duplex.
- SIP support SIP domain, SIP authentication (none, basic, MD5), DNS name of server, peer to peer
- SIP support 2 servers, user can through each server to calling in and out
- DTMF:SIP info, DTMF Relay, RFC2833
- SIP application: contain SIP call forward/transfer/holding/waiting/3 way conference/Paging and intercom/ click to dial/pickup/ joincall/redial/unredial.
- Call control features: Flexible dial map, support hotline, empty calling no. reject server, black list for reject, authenticated call, no disturb, caller ID and so on.
- Could dial use private server automatically when public server unregistered while private server is resgistered successfully
- Support phonebook 500 records, incoming calls / outgoing calls / missing calls. Each supports 100 records
- Support MWI
- support conference call in server
- Phonebook supports VCard standard
- Support 12/24 time format.
- Support daylight saving time
- Support path, gruu
- Support SIP Privacy.

5.1.3. Network Features

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

5.1.4. Maintenance and Management

- The phone supports post mode, can update firmware by post mode.
- Supports different levels of administration.
- Support Boot Monitor

- Can upgrade firmware through boot monitor access with different authority •

- •
- support auto provisioning Can config through Web, Keypad, Telnet Can upgrade firmware and configuration file through HTTP, FTP, TFTP •
- Support syslog

| Button | Character | Button | Character |
|--------------|---------------|--------------|---------------------------|
| 1 00 | 1 @ | 7PGRS | 7 P Q R S p q r s |
| 2 ABC | 2 A B C a b c | 8 TUV | 8 T U V t u v |
| 3 DEF | 3 D E F d e f | 9 wxyz | 9 W X Y Z w x y z |
| 4 вні | 4 G H I g h i | */= | |
| 5 JKL | 5 J K L j k I | | 0 |
| 6 MNO | 6 M N O m n o | #/= | # |

5.2. Digit-character map table