# IP Phone User Manual


**This Manual provides basic information on how to install and connect IPH301 IP Phone to the network. It also includes features and functions of IPH301 IP phone components, and how to use them.**

# Contents

# 1 Before Getting Started

Before you can connect IPH301 to the network and use it, you must have a high-speed Internet connection installed. A high-speed connection includes such environments as DSL, cable modem, and a leased line. IPH301 IP phone is a stand-alone device, which requires no PC to make Internet calls. IPH301 IP is fully compatible with SIP and IAX2 industry standard and can interoperate with many other SIP or IAX2 compliant devices and software in market.

# 2 Package contents

1. IPH301 IP phone base unit.
2. Handset
3. Coiled handset connecting cord
4. One Straight Ethernet cable
5. One Power supply
6. User guide

# 3 SPEC and Features

## 3.1 Hardware Spec

- 32-bit 150MHz MIPS CPU
- 16-bit 100MHz DSP
- 2MB flash memory
- 32MB SDARM

- 2*16 character LCD
- 34 KEYS, 1 LED
- WAN: 10/100M RJ45
- LAN: 10/100M RJ45 (Option)

### 3.2 DSP Spec

- G.711A/u, G.723.1（5.3k/6.3k）, G.729a/b,G.722,G.726
- G.168 echo cancel
- Full duplex hand-free
- VAD (Voice Activity Detection)
- CNG (Comfortable Noise Generation)
- AEC (Acoustic Echo Cancellation)
- Adaptive voice jitter buffer
- Codec negotiation supported for fixed and dynamic

### 3.3 SIP Protocol Spec

- RFC Protocol Edition: RFC3261 and RFC2543
- Compatible with all major SIP Servers: Cisco, Osip, Vocal, ser, Partysip, Simens, Grandstream, etc.
- DTMF Mode: RFC2833, RELAY, SIP INFO
- Server authentication mode: none, basic, MD5
- DNS name of SIP server
- SIP signaling port setting
- NAT traverse, STUN
- NAT traverse, SIP Express router
- Pubic Server/ Private server. Can connect to ISP and Private SIP server

at the same time9

- Dual back- up servers
- Peer to peer SIP call

### 3.4 Software feature

- Languages: English
- Flexible Dial Map: Fix length; End with #; Dial with time out
- 9 Kinds of ringer able select by number of Phone Box and 2 kinds of ringer user defined
- Speakerphone
- Dial Map Table
- 5 Speed dial key.
- Dual register No. for phone
- Black list for reject authenticated call
- Reject incoming call
- Limit dialing out No. list
- No Disturb
- Caller ID display
- Call forward, call transfer, call hold, call waiting
- Call forward with unconditional, busy and no answer
- 3 party conference
- 50 entries each for dialed call, received call and missed call

### 3.5 Networking Standards

- WAN/LAN port with Router or Bridge Mode
- NAT ALG
- PPPoE for xDSL, automatically keep alive

- DHCP Client on WAN
- DNS client with 2 servers IP
- SNTP
- RTP: RFC3550
- 802.1P QOS

### 3.6 Others

- Boot Monitor
- Upgrade firmware through POST mode
- Keyboard Configuration
- HTTP Web-Based Configuration
- FTP, TFTP upgrade firmware •
- WEB upgrade firmware
- FTP, TFTP, HTTP upload/download configuration file

### 3.7 Physical & Environmental

Desktop / Wall mounting

Power Input: 100 to 240 ACV / 50/60Hz Output:7.5 DCV

Dimensions: 210 x 170 x 130 mm

Weight: 680 g (main unit)

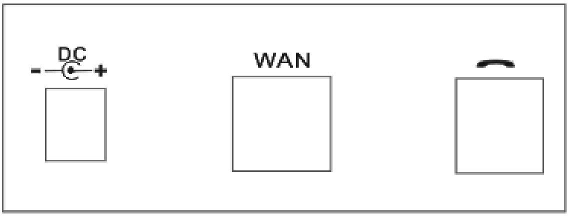
Operating Temperature: 0°to +40°C (32°to +104°F)

Storage Temperature:-20°to +70°C (-40°to +158°F)

Humidity: 5% -95% non-condensing.

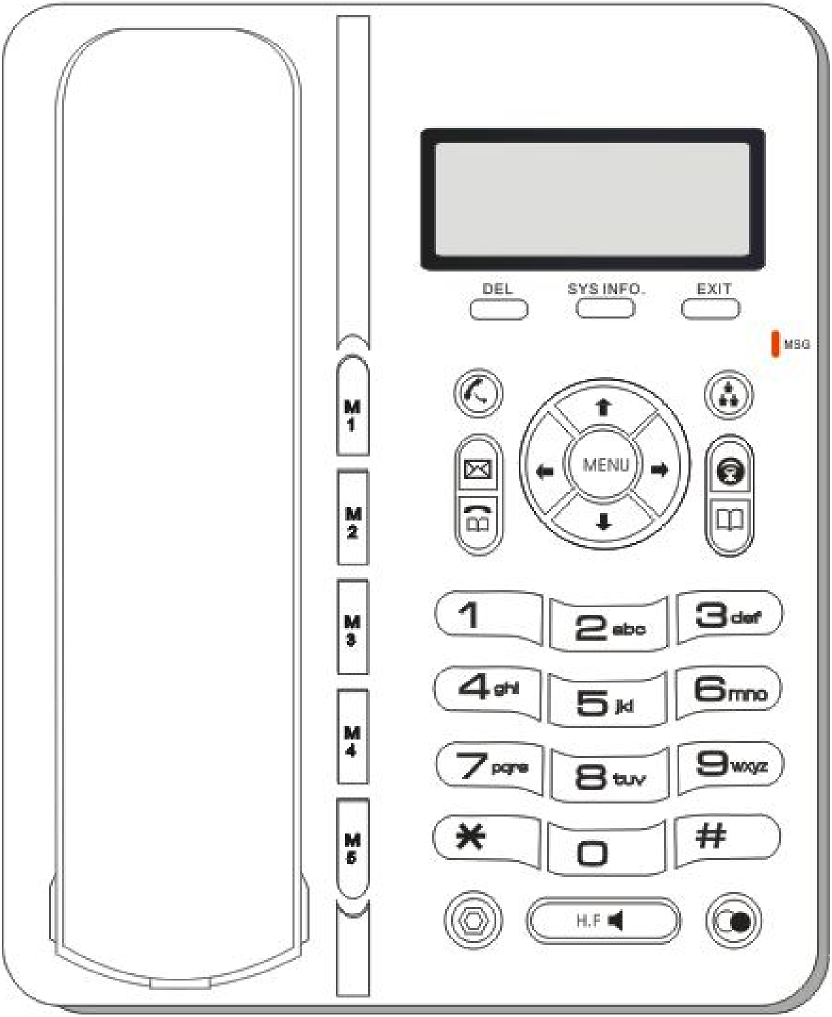
# 4 Installation

Use the cable for Internet connection; connect the 'WAN' port of IPH301 to router. Get the cable from box and connect the WAN port of phone to your PC. Connect the power supply in the box to 'DC'. Then start your phone.

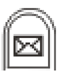
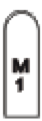



| | | |
|---|---|---|
| 1 | - DC + | Connect to Power Supply |
| 2 | WAN | Connect to Ethernet cable |
| 3 | | Connect to Handset |

# 5 General Appearance



| | Key Button | Key Button Definitions |
|---|---|---|
| 1 | 0 - 9, *, # | Digital, star and pound keys are also used for setting and call process. |
| 2 | Menu/OK   | Menu: enter the main menu, or confirm current status |
| 3 | DEL   | delete the current editing content or delete the incorrect number in pre-dial mode |
| 4 | Sys Info   | Display local IP address on LCD<br>display the current registered account<br>display the register status |
| 5 | Exit   | return the previous menu or exit the main menu |
| 6 |  | Transfer: used as blind transfer, attended transfer and half attended transfer |
| 7 |  | Conference: used as three party conference |
| 8 |  | Hold: hold the active call |
| 9 |  | Call List: browse the call logs |

| | | |
|---|---|---|
| 10 |  | Used as Mute |
| 11 |  | Used as phonebook |
| 12 |  | Navigation key used as UP in menu or Increase handset / speakerphone volume |
| 13 |  | Navigation key used as DOWM in menu or reduce handset / speakerphone volume |
| 14 |  | Navigation key used as LEFT in menu or reduce handset/speaker volume |
| 15 |  | Navigation key used as RIGHT in menu or increase handset/speaker volume |
| 16 |  | Enter to voice mail |
| 17 |   | Speaker: activate/deactivate the hands-free function |
| 18 |  | Redial: Dial a new number or redial the last call. |
| 19 | **M1-M5**   | Speed dial: make the speed dial call |

# 6 Phone Operations

## 6.1 Make a Call

There are some ways to make a call:

1. Pick up handset or press button,then dial the desired numbers.

2. Press the button directly to redial the last call.

3. While checking call logs in menu such as Dialed call / Received call / Missed call, press 【#】or button to dial out displayed number on LCD.

4. When the unit indicates Missed calls, press and button to enter Missed call menu, then press button to review number. Press【#】or button to dial out this number.

5. In standby, input the desired number to make pre-dial, Press 【#】/ button to dial out this number.

6.2 Phone book number store / edit / delete

In standby, -press button, LCD display "Phone Book Current"

-press to check first record, press and to review others.

-press till LCD display " ADD", then press to input name, number, ring types. During the operation, press Local IP/DEL to delete wrong digit.

-press till LCD display "Search", then press , input name of desired number and press to check the item, if you want to edit it, press Local IP/DEL while LCD displays "0=mod 1=del", then press 0/1 to modify/delete.

### 6.3 Call list check / delete / dial out

In standby, press , LCD display "Call Record Dialed",

-Press to check last dialed out number and conversation time, press and to review others; press 【#】to dial out this item; press Local IP/DEL to delete, LCD displays "Are you sure?", press to confirm.

- press till LCD display "Received", then press to check latest received call, press and to review others and press "#" to dial out; press Local IP/DEL to delete.

- press till LCD display "Call Record missed", then press **to** check latest miss call, press and to review other and press “#” to dial out; press Local IP/DEL to delete.

### 6.4 Hold

During conversation, press to keep line, press it again to release.

### 6.5 Volume adjustment in conversation

During conversation, press navigation key to adjust receiving volume.

### 6.6 Block list setting / edit / delete

Please refer to below setting menu and CALL SERVICE setting in web configuration.

### 6.7 Call transfer

TRANSFER: During conversation, press button and input transferred number end with 【#】 to transfer the phone to the third part and hang up automatically

### 6.8 3-party conference

During conversation, press , then dial another number plus # button, while line connected, press  to make conference call;

## 7. Unit Configuration

### 7.1 IP distribution mode selection

Press and hold 【1】 button for 5s, the LCD displays “STATIC MODE”;
Press and hold 【2】 button for 5s, the LCD display “DHCP MODE”;
Press and hold 【3】 button for 5s, the LCD display “PPPOE MODE”.

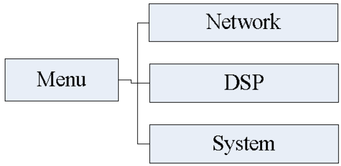
### 7.2 Configuration with keypad and LCD display

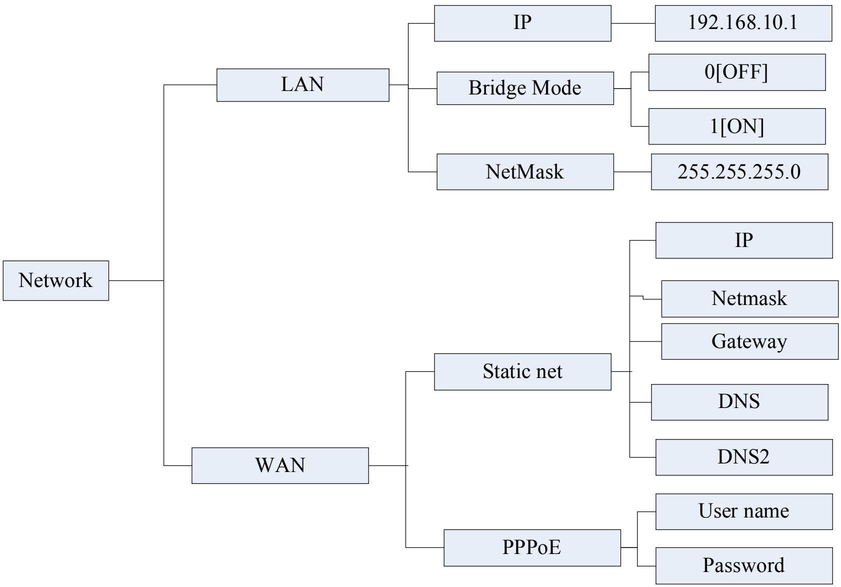
In standby, press  button till LCD shows " Input Password: " input correct password (default is 123), press  key to enter the menu list. Then follow below menu list to set parameters accordingly.

During configuration, operations as follows

-For browse and edit Configuration, press  and  ;

-To change parameter, press【PH No./Edit】firstly, then input desired digit, confirm and save by press buttons .



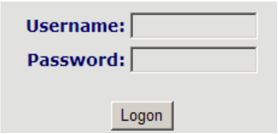

## 8 Web Interface Configuration

The IP Phone Web Configuration Menu can be accessed by the following URI: http://Phone-WAN IP-Address/. The default WAN IP address is dynamic acquisition, press  can access the IP address.

If the web login port of the phone is configured as non-80 standard port, then user need to input http://xxx.xxx.xxx.xxx:yyyy, otherwise the web will show that no server has been found.

## 8.1 Login Web

While input correct IP address as above, logon menu pop out as follows:



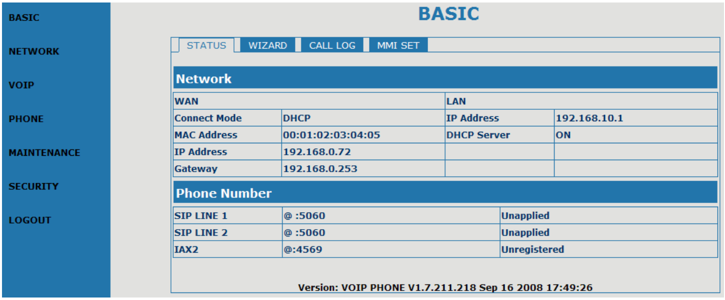

There are two level login as:

-Guest account: the default username and password is "guest", user can have a browse of system.

-Administrator account: the default username and password is "admin", this user can configure the system.

*Note: After inputting username and password, user press carriage return directly to enter the page.*

*While successfully login, web shown as follows:*



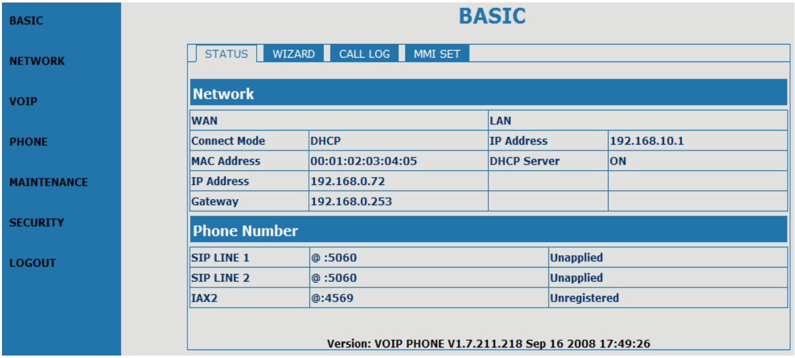
## 8.2 Current state

On this page user can gather information of each normal parameters, as:

-the network section shows the current WAN configurations of the phone, including access way of WAN IP and IP（static state, DHCP, PPPoE), MAC address, WAN IP address of the phone.

-The VoIP section shows the current default signaling protocol, and server parameter; Register server IP of SIP, proxy server IP, whether enables register, whether has registered on register server, whether enables outbound proxy, whether enables STUN server.

-The Phone Number section shows corresponding phone number of each protocol; the version number and date of issue have been shown at the end of the page.




## 8.3 Network configuration

### 8.3.1 Wide area network（WAN）

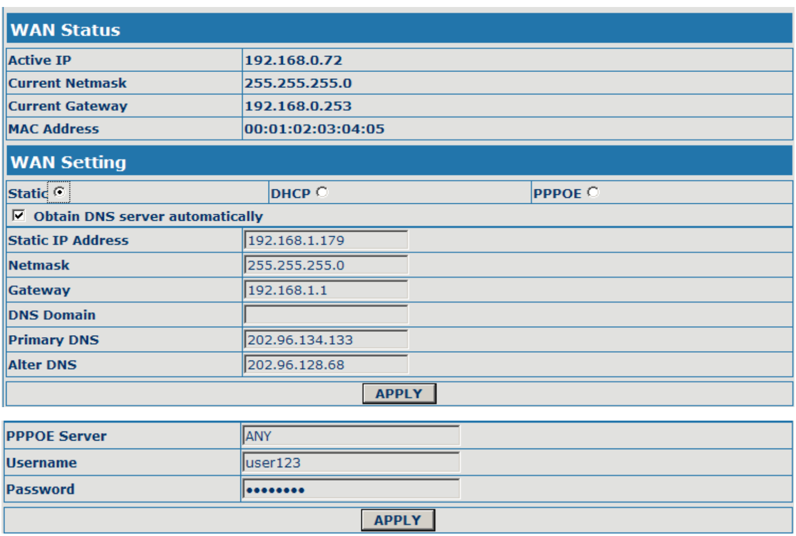
User can view the current network IP linking mode of the system on this page.

User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If the system selects DHCP service in the network which is using DHCP service, IP address will be gained dynamically.

If the system selects PPPOE service in the network which is using the PPPOE service, then the IP address will be gained by the set PPPOE ISP internet and password of the account.

*Note: If IP address has been modified, the web page will no longer respond owing to the modification, so new IP address should be input in the address field now.*




Configuration Explanation:

| WAN Status | |
|---|---|
| Active IP | 192.168.0.72 |
| Current Netmask | 255.255.255.0 |
| Current Gateway | 192.168.0.253 |
| MAC Address | 00:01:02:03:04:05 |

Current phone IP, subnet mask, MAC address and current phone IP;

| | | |
|---|---|---|
| Static ○ | DHCP ○ | PPPOE ◉ |

,

Select acquisition way of IP for WAN; This is single option; Configure static IP parameter for WAN;

| | |
|---|---|
| Static IP Address | 192.168.1.179 |
| Netmask | 255.255.255.0 |
| Gateway | 192.168.1.1 |
| DNS Domain | voip.com |
| Primary DNS | 202.96.134.133 |
| Alter DNS | 202.96.128.68 |

| | |
|---|---|
| Static IP Address | 192.168.1.179 |

Configure static IP address;

| | |
|---|---|
| Netmask | 255.255.255.0 |

Configure subnet mask;

| | |
|---|---|
| Gateway | 192.168.1.1 |

Configure IP address of the phone;

| | |
|---|---|
| DNS Domain | voip.com |

Configure "DNS domain" suffix; if user input "domain" and it can't be resolved, then the phone will add and resolve the "domain" after user has input;

| | |
|---|---|
| Primary DNS | 202.96.134.133 |

Main DNS server IP address;

| | |
|---|---|
| Alter DNS | 202.96.128.68 |

The second DNS server IP address;

Configure PPPoE:

| | |
|---|---|
| PPPOE Server | ANY |
| Username | user123 |
| Password | ●●●●●●●● |

| | |
|---|---|
| PPPOE Server | ANY |

Service name, if PPPoE ISP has no special requirement for this name, generally is the default;

| | |
|---|---|
| Username | user123 |

PPPoE account;

| | |
|---|---|
| Password | ●●●●●●●● |

PPPoE password;

Configure the parameter and then click "apply" to go into effect.

### 8.3.2 Service Port configuration

User can set up Telnet, HTTP, RTP port on this page and view DHCP table.

| 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. | |

Configuration Explanation:

| | |
|---|---|
| HTTP Port | 80 |

Configure 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;

| | |
|---|---|
| Telnet Port | 23 |

Configure telnet port, the default is 23 port;

| | |
|---|---|
| **RTP Initial Port** | 10000 |

Enable RTP initial port configuration. It is dynamic allocation;

| | |
|---|---|
| **RTP Port Quantity** | 200 |

Configure the maximum quantity of RTP port. The default is 200;

| DHCP Leased Table | |
|---|---|
| Leased IP Address | Client Hardware Address |

Leased IP-MAC correspondence table of DHCP;

※The configuration on this page needs to be saved after modified and will go into effect after restarting.

※If the Telnet, HTTP port will be modified, the port is better to be set as greater than 1024, because less than 1024 port system will save ports.

※Set the HTTP port as 0, then the http service will be disabled.

# 9 VOIP Configuration

## 9.1 IAX2 Configuration

| IAX2 | |
|---|---|
| Register Status | Unregistered |
| IAX2 Server Addr | |
| IAX2 Server Port | 4569 |
| Account Name | |
| Account Password | |
| Phone Number | |
| Local Port | 4569 |
| Voice Mail Number | 0 |
| Voice Mail Text | mail |
| Echo Test Number | 1 |
| Echo Test Text | echo |
| Refresh Time | 60 Seconds |
| Enable Register | ☐ |
| Enable G.729 | ☐ |
| APPLY | |

Configuration Explanation: Explanation:

| | |
|---|---|
| Register Status | Unregistered |

IAX2 registration state display； If register successfully, it will display [Registered], otherwise will display [Unregistered];

| | |
|---|---|
| IAX2 Server Addr | |

Config IAX2 the server address, also can use domain name form;

| | |
|---|---|
| IAX2 Server Port | 4569 |

Config IAX2 server port;

| | |
|---|---|
| Account Name | |

Config IAX2 account name；

| | |
|---|---|
| **Account Password** | |

Config IAX2 account password；

| | |
|---|---|
| **Phone Number** | |

Config IAX2 phone number；

| | |
|---|---|
| **Local Port** | 4569 |

Config equipment iax2 monitor port；

| | |
|---|---|
| **Voice Mail Number** | 0 |

Config voice mail number， If the IAX2 support voice mailbox, the voice mailbox is the letter form, the gateway is unable to input the letter, uses this number to replace voice mail the name；

| | |
|---|---|
| **Voice Mail Text** | mail |

Config voice mailbox name; if the IAX2 support voice mailbox, here to config the vocie mailbox the name；

| | |
|---|---|
| **Echo Test Number** | 1 |

Config whether supports echo. If the platform support echo, (echo number is the text format), then the telephone config this echo test number replace echo actual text number. This function is refers through the platform, the terminal may carry on echo the call to test. To see the terminal to the platform converses on the telephone whether normally;

| | |
|---|---|
| **Echo Test Text** | echo |

Config echo test text；

| | |
|---|---|
| **Refresh Time** | 60 **Seconds** |

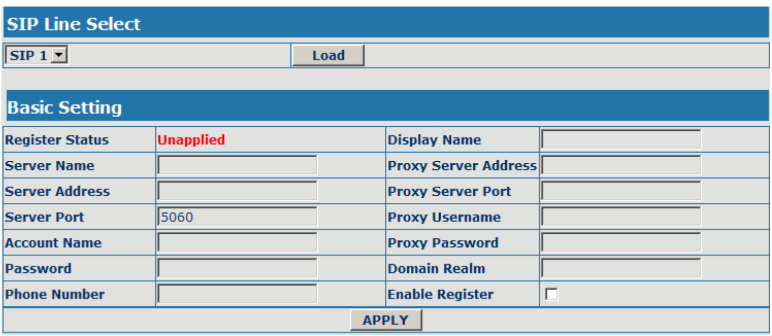
Config IAX2 refresh time，The unit of time for the second, suggested the user in makes the choice 60 to 3600 between；


config the permission/prohibition registers the server；


Config whether supports G.729;

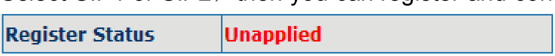
## 9.2 SIP Configuration

Sip register

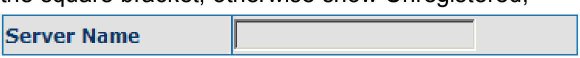



Configuration Explanation:


Select SIP1 or SIP2，then you can register and configure SIP1 or SIP2;




Show SIP register state; if register successfully, there will show Registered in the square bracket, otherwise show Unregistered;



Configure the name of registration server;

| | |
|---|---|
| **Server Address** | |

Configure SIP register server IP address or Domain Name;

| | |
|---|---|
| **Server Port** | 5060 |

Configure SIP register server signal port;

| | |
|---|---|
| **Account Name** | |

Configure SIP register account（usually it is the same with the port number that configured, some special SIP servers will have different port configurations, then the port configuration needs to be configured to be numbers, here the configuration account can be arbitrary character string）;

| | |
|---|---|
| **Password** | |

Configure password of SIP register account;

| | |
|---|---|
| **Phone Number** | |

Configure the phone number;

| | |
|---|---|
| **Display Name** | |

Configure display name;

| | |
|---|---|
| **Proxy Server Address** | |

Configure proxy server IP address or Domain Name（usually SIP will provide user with service of proxy server and register server which have the same configuration, so the configuration of proxy server is usually the same with that of register server, but if the configurations of them are different(such as different IP addresses or Domain Name), then each server's configuration should be modified separately）;

| | |
|---|---|
| **Proxy Server Port** | |

Configure SIP proxy server signal port;

| | |
|---|---|
| Proxy Username | |

Configure proxy server account;

| | |
|---|---|
| Proxy Password | |

Configure proxy server password;

| | |
|---|---|
| Domain Realm | |

Configure domain realm;

| | |
|---|---|
| Enable Register | ☐ |

Configure enable/disable register;

Advanced sip setting

| Advanced SIP Setting | | | |
|---|---|---|---|
| Register Expire Time | 60 seconds | Forward Type | Off |
| NAT Keep Alive Interval | 60 seconds | Forward Phone Number | |
| User Agent | Voip Phone 1.0 | Server Type | COMMON |
| Signal Key | | DTMF Mode | DTMF_RFC2833 |
| Media Key | | RFC Protocol Edition | RFC3261 |
| Local Port | 5060 | Transport Protocol | UDP |
| Ring Type | Default | RFC Privacy Edition | NONE |
| Hot Line Number | | Subscribe Expire Time | 300 seconds |
| Conference Number | | Enable Conference Number | ☐ |
| Transfer Expire Time | 0 seconds | Enable DNS SRV | ☐ |
| Enable Subscribe | ☐ | Click To Talk | ☐ |
| Enable Keep Authentication | ☐ | Signal Encode | ☐ |
| NAT Keep Alive | ☐ | Rtp Encode | ☐ |
| Enable Via rport | ☑ | Enable Session Timer | ☐ |
| Enable PRACK | ☐ | Answer With Single Codec | ☐ |
| Long Contact | ☐ | Auto TCP | ☐ |
| Enable URI Convert | ☑ | Enable Strict Proxy | ☐ |
| Dial Without Register | ☐ | Enable GRUU | ☐ |
| Ban Anonymous Call | ☐ | Enable Displayname Quote | ☐ |
| APPLY | | | |

| | |
|---|---|
| Register Expire Time | 60 seconds |

Configure expire time of SIP server register, the default is 60 seconds. If the expire time that server requires is more or less than that configured by the phone, the phone can automatically modify it to the recommended time limit and register;

| | |
|---|---|
| **NAT Keep Alive Interval** | 60 seconds |

Configure the NAT keep alive interval;

| | |
|---|---|
| **User Agent** | Voip Phone 1.0 |

Configure the User Agent;

| | |
|---|---|
| **Signal Key** | |

Configure the Signal Key;

| | |
|---|---|
| **Media Key** | |

Configure the Media Key;

| | |
|---|---|
| **Local Port** | 5060 |

Configure the Local Port;

| | |
|---|---|
| **Ring Type** | Default |

Select the Ring type;

| | |
|---|---|
| **Hot Line Number** | |

Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number;

| | |
|---|---|
| **Conference Number** | |

Configure the Conference Number;

| | |
|---|---|
| **Transfer Expire Time** | 0 seconds |

Configure the Transfer Expire time

| | |
|---|---|
| **Enable Subscribe** | ☐ |

Configure enable/disable Subscribe；

| | |
|---|---|
| **Enable Keep Authentication** | ☐ |

Configure enable/disable Keep Authentication;

| | |
|---|---|
| NAT Keep Alive | ☐ |

Configure enable/disable NAT Keep Alive

| | |
|---|---|
| Enable Via rport | ☑ |

Configure enable/disable Via rport

| | |
|---|---|
| Enable PRACK | ☐ |

Configure enable/disable PRACK

| | |
|---|---|
| Long Contact | ☐ |

Configure enable/disable Long Contact;

| | |
|---|---|
| Enable URI Convert | ☑ |

Configure enable/disable URI Convert;

| | |
|---|---|
| Dial Without Register | ☐ |

Configure enable/disable Dial without register;

| | |
|---|---|
| Ban Anonymous Call | ☐ |

Configure enable/disable Ban Anonymous Call;

| | |
|---|---|
| Forward Type | Off (Off / Always / Busy / No Answer) |
| Forward Phone Number | |
| Server Type | |

Call forwarding. The default is Disable; when busy is selected, if the number dialed is engaged after the phone has received a call, then it will automatically transfer to the configured number according to the following configuration; when always is selected, then the phone will directly transfer all the numbers that dial to this port to the configured numbers;

| | |
|---|---|
| Forward Phone Number | |

number configuration of call transfer (CT);

| | |
|---|---|
| Server Type | COMMON |

Select the Server type;

| | |
|---|---|
| DTMF Mode | DTMF_RFC2833 ▾ |
| RFC Protocol Edition | DTMF_RELAY<br>DTMF_RFC2833<br>DTMF_SIP_INFO |
| Transport Protocol | |

DTMF sending mode configuration; three kinds: the above are basic configurations of SIP.

| | |
|---|---|
| RFC Protocol Edition | RFC3261 ▾ |

Enable the phone to use protocol edition. When the phone need to communicate with phones which is using SIP1.0 such as CISCO5300 and so on, then it should be configured into RFC2543 to communicate normally. the default is to enable RFC3261;

| | |
|---|---|
| Transport Protocol | UDP ▾ |

Select the Transport Protocol(UDP or TCP);

| | |
|---|---|
| RFC Privacy Edition | RFC3323 ▾ |

Select the RFC Privacy Edition(none,RFC3323 or RFC3325);

| | |
|---|---|
| Subscribe Expire Time | 300 seconds |

Configure Subscribe expire time

| | |
|---|---|
| Enable Conference Number | ☐ |

Configure enable/disable Conference Number;

| | |
|---|---|
| Enable DNS SRV | ☐ |

Configure enable/disable DNS service;

| | |
|---|---|
| Click To Talk | ☐ |

Configure enable/disable Click To Talk;

| | |
|---|---|
| Signal Encode | ☐ |

Configure enable/disable Signal Encode;

| | |
|---|---|
| Rtp Encode | ☐ |

Configure enable/disable RTP Encode;

| | |
|---|---|
| Enable Session Timer | ☐ |

Configure enable/disable Session Timer;

| | |
|---|---|
| Answer With Single Codec | ☐ |

Configure enable/disable answer with single codec;

| | |
|---|---|
| Auto TCP | ☐ |

Configure enable/disable Auto TCP;

| | |
|---|---|
| Enable Strict Proxy | ☐ |

Configure enable/disable Strict Proxy;

| | |
|---|---|
| Enable GRUU | ☐ |

Configure enable/disable GRUU;

| | |
|---|---|
| Enable Displayname Quote | ☐ |

Configure enable/disable Displayname Quote;

### 9.3 STUN configuration

SIP STUN is a kind of server that used to realize the SIP's enablement of NAT, when the STUN server IP of the phone has been configured（generally the default port is 3478）and Enable SIP Stun has been selected, conventional SIP server can be used to realize the phone's penetration of NAT.

| 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 Stun | |
|---|---|
| SIP 1 | Load |
| | |
| Use Stun | ☐ |
| APPLY | |

| | |
|---|---|
| STUN NAT Transverse | FALSE |

Display the application status of the STUN NAT;

| | |
|---|---|
| STUN Server Addr | |

Configure IP address of SIP STUN server;

| | |
|---|---|
| STUN Server Port | 3478 |

Configure port of SIP STUN;

| | | |
|---|---|---|
| STUN Effect Time | 50 | Seconds |

Interval time for STUN's detection on NAT type, the unit is second;

| | |
|---|---|
| Local SIP Port | 5060 |

Configure Local SIP port;

| | |
|---|---|
| SIP 1 | Load |

Select the Sip Line;

| | |
|---|---|
| Use Stun | ☐ |

Configure enable/disable Use STUN;

### 9.4 Dial Peer Configuration

Bases on this configuration, we can make the phone use different accounts and run speed calling without swap.

When making deletion or modification, select the number firstly, then click Delete or Modify, and complete the operation.

**Dial Peer Table**

| Number | Destination | Port | Mode | Alias | Suffix | Del Length |
|---|---|---|---|---|---|---|
| 111 | 192.168.0.80 | 5060 | SIP | no alias | no suffix | 0 |
| 222 | 192.168.0.53 | 5060 | SIP | no alias | no suffix | 0 |
| 5T | 0.0.0.0 | 5060 | SIP | add:0755 | no suffix | 0 |

**Add Dial Peer**

| | |
|---|---|
| Phone Number | |
| Destination (optional) | |
| Port(optional) | |
| Alias(optional) | |
| Call Mode | SIP |
| Suffix(optional) | |
| Delete Length (optional) | |

Submit

**Dial Peer Option**

111 Delete Modify

Configuration Explanation:

**Dial Peer Table**

| Number | Destination | Port | Mode | Alias | Suffix | Del Length |
|---|---|---|---|---|---|---|
| 111 | 192.168.0.80 | 5060 | SIP | no alias | no suffix | 0 |
| 222 | 192.168.0.53 | 5060 | SIP | no alias | no suffix | 0 |
| 5T | 0.0.0.0 | 5060 | SIP | add:0755 | no suffix | 0 |

Display of calling number IP image list;

| | |
|---|---|
| **Phone Number** | |

It is to add outgoing call number, there are two kinds of outgoing call number setup: One is exactitude matching, after this configuration has been done, when the number is totally the same with the user's calling number, the phone will make the call with this number's IP address image or configuration; Another is prefix matching （ be equivalent to PSTN's district number prefix function）,if the previous N bits of this number are the same with that of the user's calling number(the prefix number length),then the phone will use this number's IP address image or configuration to make the call. When configuring the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching; the longest length is 30 bits.

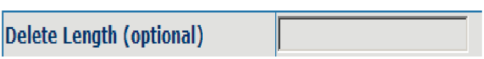

Configure destination address, destination is configured 0.0.0.0, it is SIP1 line.


Configure the protocol signal port, when nothing is input, the default of sip protocol is 5060;


Configure alias, this is optional configuration item: it is the number to be used when the other party's number has prefix; when no configuration has been made, shown as no alias;


Configure the calling mode:IAX2 and SIP;


Configure suffix, this is optional configuration item: it is the additive dial-out number behind the number; when no configuration has been made, shown as no suffix;




Configure the replacing length, replace the number that user input according to this length; this is optional configuration item.

Of which the alias can be divided into four types, it should be combined with replacing length to make the setup:

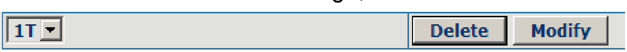
Add: xxx, add xxx before number. in this way it can help user save the dialing length;

All: xxx, the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all", the actual calling number will be replaced;

Del: delete n bit in the front part of the number, n can be decided by the replacing length; this configuration can decide the protocol for appointed number;

Rep: xxx, n bit in the front part of the number will be replaced. n is decided by the replacing length. For example, user want to dial PSTN （0757－86228930） by VoIP's Rec/Finish over service, while actually the called number should be 86757－86228930,then we can configure called number as 0757T,then rep:86757,and then set the replacing length as 3. So that when user make a call with 0757 prefix, the number will be replaced as 86757 plus the number and then sent out. It is a convenient thinking mode for user to make a call;


Delete selective number IP image;




If user want to modify a certain current number image, first select in the drop-down menu and then load the image parameter of the said number, click modify to make modification; of which:

| Dial Peer Modify | |
|---|---|
| Phone Number | 5T |
| Call Mode | SIP |
| Destination (optional) | 0.0.0.0 |
| Port(optional) | 5060 |
| Alias(optional) | add:0755 |
| Suffix(optional) | no suffix |
| Delete Length (optional) | 0 |
| Submit | |

| | |
|---|---|
| Phone Number | 5T |

this is the modified number. read-only;

| | |
|---|---|
| Call Mode | SIP |

To modify call mode;

| | |
|---|---|
| Destination (optional) | 0.0.0.0 |

To modify destination address; this is optional configuration item;

| | |
|---|---|
| Port(optional) | 5060 |

To modify destination phone port; this is optional configuration item;

| | |
|---|---|
| Alias(optional) | add:0755 |

To modify alias; this is optional configuration item;

| | |
|---|---|
| Suffix(optional) | no suffix |

To modify suffix; this is optional configuration item;

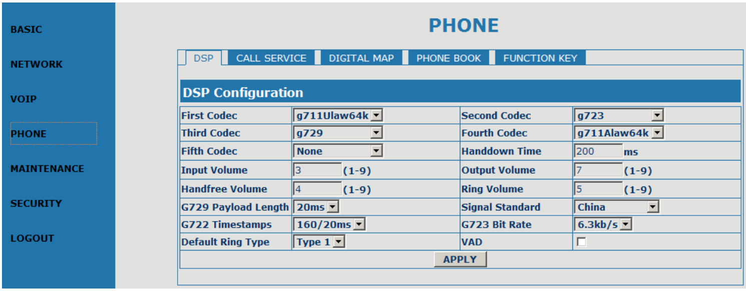
| | |
|---|---|
| Delete Length (optional) | 0 |

To modify replacing length（if rep and del of alias have been configured）;

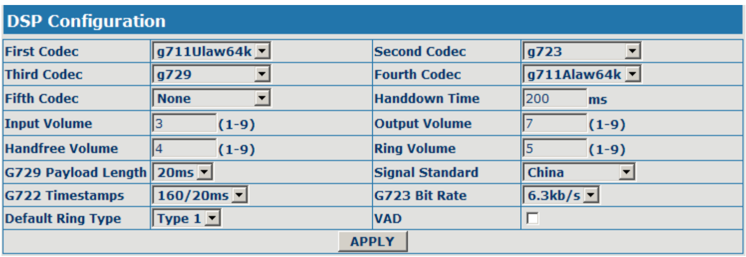
Submit

Click submit to go into effect.

# 10 Phone Configuration

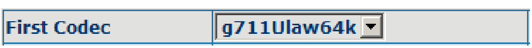



## 10.1 DSP configuration



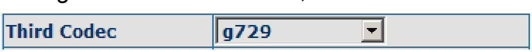
| DSP Configuration | | | |
|---|---|---|---|
| First Codec | g711Ulaw64k | Second Codec | g723 |
| Third Codec | g729 | Fourth Codec | g711Alaw64k |
| Fifth Codec | None | Handdown Time | 200 ms |
| Input Volume | 3 (1-9) | Output Volume | 7 (1-9) |
| Handfree Volume | 4 (1-9) | Ring Volume | 5 (1-9) |
| G729 Payload Length | 20ms | Signal Standard | China |
| G722 Timestamps | 160/20ms | G723 Bit Rate | 6.3kb/s |
| Default Ring Type | Type 1 | VAD | |
| APPLY | | | |

On this page, user can set speech coding, IO volume control, cue tone standard, caller ID standard and so on.

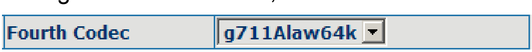



Configure the first codec;

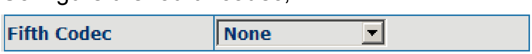

Configure the second codec;




Configure the third codec;




Configure the fourth codec;



Configure the fifth codec;

| | |
|---|---|
| Handdown Time | 200 ms |

Configure handdown time, that is, if the hooking time is shorter than this time, then the gateway will not consider the user has handdown.

| | |
|---|---|
| Input Volume | 3 (1-9) |

Configure input volume;

| | |
|---|---|
| Output Volume | 7 (1-9) |

Configure output volume;

| | |
|---|---|
| Handfree Volume | 4 (1-9) |

Configure handfree volume;

| | |
|---|---|
| Ring Volume | 5 (1-9) |

Configure ring volume;

| | |
|---|---|
| G729 Payload Length | 20ms |

Configure G729 payload length;

| | |
|---|---|
| Signal Standard | China |

Configure signal standard;

| | |
|---|---|
| G722 Timestamps | 160/20ms |

Configure G.722 timestamps;

| | |
|---|---|
| G723 Bit Rate | 6.3kb/s |

Configure G.723 bit rate;

| | |
|---|---|
| Default Ring Type | Type 1 |

Configure default ring type;

| | |
|---|---|
| VAD | |

Configure enable/disable VAD.

## 10.2 Call service configuration

On this page, user can set value added services such as hot-line, call forwarding, call transfer (CT), call-waiting, three way call, blacklist, out-limit list and so on.

**Call Service Setting**

| | | | |
|---|---|---|---|
| Hot Line | | No Answer Time | 20 (seconds) |
| P2P IP Prefix | . | Remote Record No | |
| Do Not Disturb | ☐ | Ban Outgoing | ☐ |
| Enable Call Transfer | ☑ | Enable Call Waiting | ☑ |
| Enable Three Way Call | ☑ | Accept Any Call | ☑ |
| Auto Answer | ☐ | Use Record Server | ☐ |
| | APPLY | | |

**Black List**

| | | | |
|---|---|---|---|
| | Black List | | |
| | Add | | Delete |

**Limit List**

| | | | |
|---|---|---|---|
| | Limit List | | |
| | Add | | Delete |

Configuration Explanation:

| | |
|---|---|
| **Hot Line** | |

Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number;

| | |
|---|---|
| **No Answer Time** | 20 (seconds) |

Configure no answer time;

| | |
|---|---|
| **Do Not Disturb** | ☐ |

Configure enable/disable Do Not Disturb;

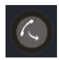
| | |
|---|---|
| **Ban Outgoing** | ☐ |

Configure enable/disable Ban outgoing;

| | |
|---|---|
| **Enable Call Transfer** | ☑ |

Configure enable/disable call transfer (CT); after it is enabled, there are two modes call transfer as below:

UNATTENDED TRANSFER: During conversation, press  button and input transferred number end with 【#】 to transfer the phone to the third part and hang up automatically

HALF ATTENDED TRANSFER: During conversation, press  button to hold this line, and input transferred number end with【 # 】to get through another line. When third part is ringing, press  button to end conversation and transfer the phone to the third part and hang up automatically.

ATTENDED TRANSFER: During conversation, press  button to hold this line, and input transferred number end with【 # 】to get through another line. After conversation with third part, press  button to end conversation and transfer the phone to the third part and hang up automatically.

| | |
|---|---|
| Enable Call Waiting | ☑ |

Configure enable/disable call waiting service; after it is enabled, user can hold calls of the other party by hooking, with hooking again, and the hold call can go on;

| | |
|---|---|
| Enable Three Way Call | ☑ |

Configure enable/disable three way call; user can call the other part as the call origination, after talking, make hooking to hold this part and then press 【Speed Dial/Conference】 key to hear the dialing tone, after call completion to the third party, hooking again to recover the talk with the second part, then the three way call concurrently;

| Accept Any Call | ☑ |
|---|---|

Configure enable/disable Accept Any Call;

| Auto Answer | ☐ |
|---|---|

Configure enable/disable Auto Answer;

| Black List | | | |
|---|---|---|---|
| Black List | | | |
| | Add | | Delete |

Configure add/delete blacklist. If user doesn't want to answer a certain number, please add this number to the list, and then this number will be unable to get through the phone.

| Limit List | | | |
|---|---|---|---|
| Limit List | | | |
| | Add | | Delete |

Configure out-limit list; for example, if user don't want the phone to dial a certain number, please add the number to this table, and the user will be unable to get through this number.

### 10.3 Phone book configuration

On this page, user can add, delete and modify telephone book.

| Phonebook Table | | | |
|---|---|---|---|
| Index | Name | Number | Type |

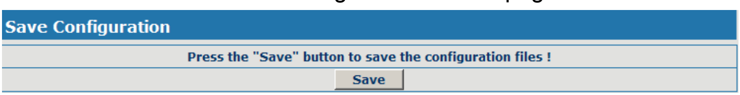
| Add Phone Book | | |
|---|---|---|
| Name | | Add |
| Number | | |
| Ring Type | Default | |

| Phone Book Option | | |
|---|---|---|
| | Delete | Modify |

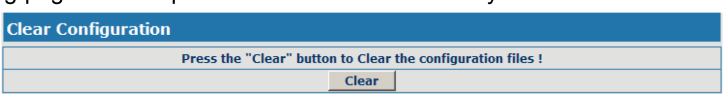
NOTE: SOME ISP INTERNET MAY INHIBIT THE PHONE TO REGISTER AND CANCEL THE REGISTER IN SUCCESSION, SO USER HAD BETTER NOT APPLY OR REGISTER AND CANCEL SOON IN SUCCESSION AND SUBMIT REGISTRATION REPEATEDLY. SERVER MAY STOP RESPONSE OF DIALOGUE MACHINE, THEN THE PHONE RECEIVES NO CERTIFICATION OF REGISTER/CANCEL LOGIN REQUEST AND REGISTRATION STATE WILL SHOW AS INCORRECT!

# 11 Save and Clear Configuration

User can save the current configuration on this page.



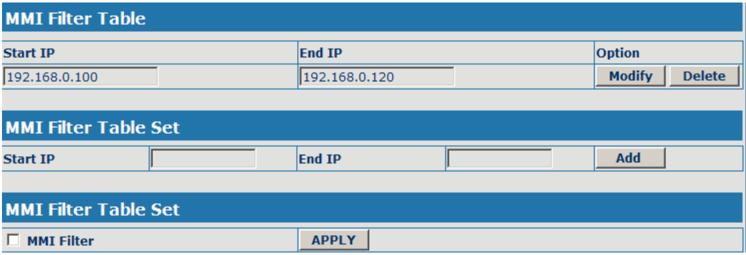

The system configuration can be set as factory default configuration on clear config page and the phone will restart automatically.




# 12 Security Configuration

## 12.1 MMI Filter configuration

On the page, user can configure the function of the MMI Filter. This feature allows only the host within the MMI filter table that they can logon the WEB page.





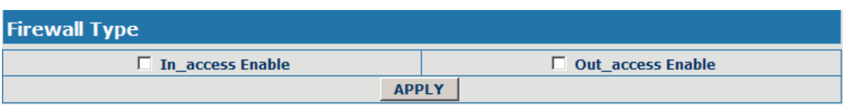
On the MMI filter table, user can modify and delete the MMI filter;


Add MMI filter table;


Configure enable/disable MMI Filter.

## 12.2 Firewall configuration

On the page, user can configure the function of the firewall.



### Firewall Input Rule Table

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|---|---|---|---|---|---|---|---|---|
| 1 | Deny | UDP | 192.168.0.123 | 255.255.255.255 | 192.168.10.10 | 255.255.255.0 | More than | 1 |
| 2 | Permit | UDP | 192.168.0.120 | 255.255.255.255 | 192.168.10.10 | 255.255.255.0 | More than | 1 |

### Firewall Output Rule Table

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|---|---|---|---|---|---|---|---|---|
| 1 | Deny | UDP | 192.168.10.10 | 255.255.255.255 | 192.168.0.10 | 255.255.255.255 | More than | 1 |
| 2 | Permit | UDP | 192.168.10.20 | 255.255.255.255 | 192.168.0.120 | 0.0.0.0 | More than | 1 |

### Firewall Set

| | | | | |
|---|---|---|---|---|
| Input/Output | Input | Src Addr | | |
| Deny/Permit | Deny | Des Addr | | Add |
| Protocol Type | UDP | Src Mask | | |
| Port Range | more than | Des Mask | | |

### Rule Delete

| | | | | |
|---|---|---|---|---|
| Input/Output | Input | Index To Be Deleted | | Delete |

### Firewall Type

| | |
|---|---|
| ☐ In_access Enable | ☐ Out_access Enable |
| APPLY | |

Select firewall type;

### Firewall Input Rule Table

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|---|---|---|---|---|---|---|---|---|
| 1 | Deny | UDP | 192.168.0.123 | 255.255.255.255 | 192.168.10.10 | 255.255.255.0 | More than | 1 |
| 2 | Permit | UDP | 192.168.0.120 | 255.255.255.255 | 192.168.10.10 | 255.255.255.0 | More than | 1 |

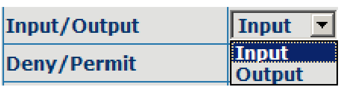
Display firewall input rule table;

### Firewall Output Rule Table

| Index | Deny/Permit | Protocol | Src Addr | Src Mask | Des Addr | Des Mask | Range | Port |
|---|---|---|---|---|---|---|---|---|
| 1 | Deny | UDP | 192.168.10.10 | 255.255.255.255 | 192.168.0.10 | 255.255.255.255 | More than | 1 |
| 2 | Permit | UDP | 192.168.10.20 | 255.255.255.255 | 192.168.0.120 | 0.0.0.0 | More than | 1 |

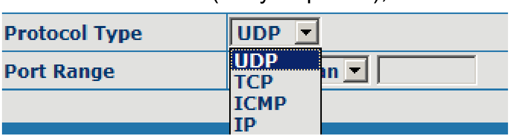
Display firewall output rule table; in the table， Src address and Src Mask confirm the range of the source address, For example ： Src addr(192.168.10.10) and Src Mask(255.255.255.255) can confirm that the host is 192.168.10.10；Dse addr(192.168. 0.120) and Des Mask(0.0.0.0) can confirm any host; Src addr(192.168.10.20) and Src Mask(255.255.255.0) confirm any host within the 192.168.10.0 network segment. So, when the Index is “1” in the output rule table, and the definition of rule is: the UDP package of the host(192.168.10.10) can’t was sent to the host(192.168.10.10)；when the Index is “1” in the output rule table, and the definition of rule is：the host(192.168.10.20) can send the UDP package to any host.


Configure the firewall.

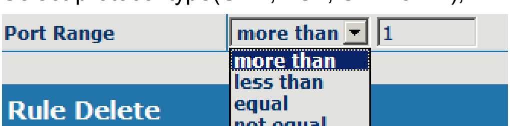



Select the rule of the firewall( input or output);


Select the condition(deny or permit);




Select protocol type(UDP, TCP,ICMP or IP);



Configure port range;

| | |
|---|---|
| Src Addr | |

Configure source IP address;

| | |
|---|---|
| Des Addr | |

Configure destination IP address;

| | |
|---|---|
| Src Mask | |

Configure source Mask;

| | |
|---|---|
| Des Mask | |

Configure destination Mask;

### 12.3 NAT and DMZ configuration

On the page, user can configure NAT and DMZ. T function of the NAT is a network port mapping and the function of the DMZ is a network address mapping.

#### NAT configuration

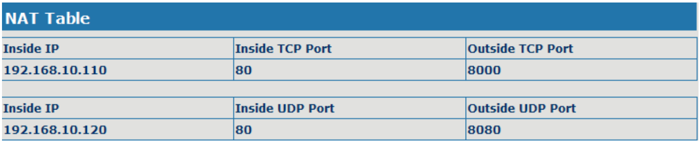
| Protocol Set | | |
|---|---|---|
| ☑ IPSec ALG | ☑ FTP ALG | ☑ PPTP ALG |
| | APPLY | |

| NAT Table | | |
|---|---|---|
| Inside IP | Inside TCP Port | Outside TCP Port |
| 192.168.10.110 | 80 | 8000 |
| Inside IP | Inside UDP Port | Outside UDP Port |
| 192.168.10.120 | 80 | 8080 |

| NAT Table Option | | | |
|---|---|---|---|
| Transfer Type | TCP | Outside Port | |
| Inside IP | | Inside Port | |
| | Add | Delete | |

| Protocol Set | | |
|---|---|---|
| ☑ IPSec ALG | ☑ FTP ALG | ☑ PPTP ALG |
| | APPLY | |

Configure NAT of the Application Layer Gateway(ALG);the protocol includes: IPSec, FTP and PPTP.

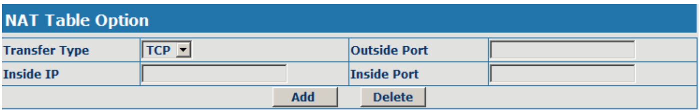


NAT Table

| Inside IP | Inside TCP Port | Outside TCP Port |
|---|---|---|
| 192.168.10.110 | 80 | 8000 |

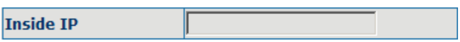
| Inside IP | Inside UDP Port | Outside UDP Port |
|---|---|---|
| 192.168.10.120 | 80 | 8080 |

Display NAT table;

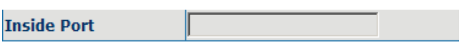



Add and delete NAT table (configure NAT of the Transport Layer, the protocol includes: TCP and UDP);

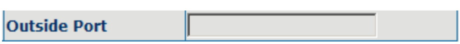

Select transfer type;




Configure Inside IP;




Configure Inside port;




Configure Outside port;


Add or delete NAT table;

**DMZ configuration**

**DMZ Table**

| Outside IP | Inside IP |
|---|---|
| 192.168.10.20 | 192.168.0.179 |
| 192.168.10.30 | 192.168.0.170 |

**DMZ Table Option**

| | |
|---|---|
| Outside IP | |
| Inside IP | |
| Outside IP | 192.168.10.20 |
| | Add Delete |

**DMZ Table**

| Outside IP | Inside IP |
|---|---|
| 192.168.10.20 | 192.168.0.179 |
| 192.168.10.30 | 192.168.0.170 |

Display DMZ table；

**DMZ Table Option**

| | |
|---|---|
| Outside IP | |
| Inside IP | |
| Outside IP | 192.168.10.20 |
| | Add Delete |

Configure the DMZ rule.

**Outside IP**

Configure the outside IP of the DMZ;

**Inside IP**

Configure the inside IP of the DMZ;

**Add**

Configure outside IP and inside IP, then click the Add, user can add the DMZ table.

| | |
|---|---|
| Outside IP | 192.168.10.20 |
| | 192.168.10.20<br>192.168.10.30 |

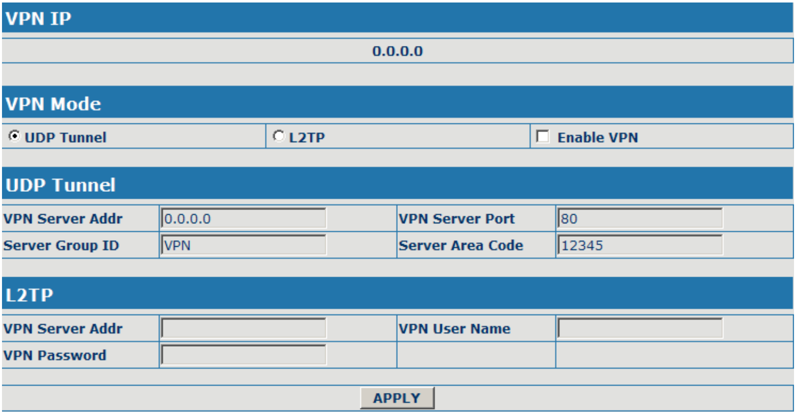
Select Outside IP;

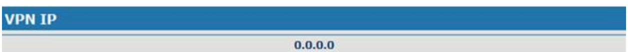
**Delete**

Select outside ip, and click the Delete, user can delete the DMZ table.

## 12.4 VPN configuration

On this page, user can save and configure VPN setting.

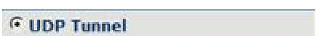


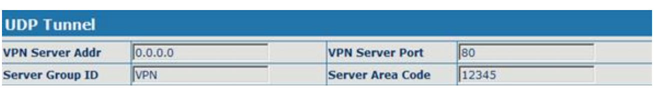

Display the VPN IP of the IPH301;

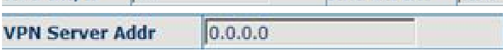

Select VPN mode, and configure enable/disable VPN;

(1) Select UDP tunnel, and configure VPN:




Select UDP tunnel mode;




Configure VPN server address;

Configure VPN server port;

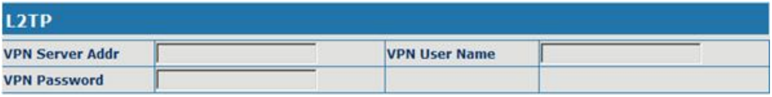

Configure VPN server group ID;


Configure VPN server area code;


Configure enable/disable VPN tunnel;

(2) Select L2TP, and configure VPN:


Select L2TP mode;





Configure VPN server address;


Configure VPN User Name;


Configure VPN Password;


Configure enable/disable VPN;

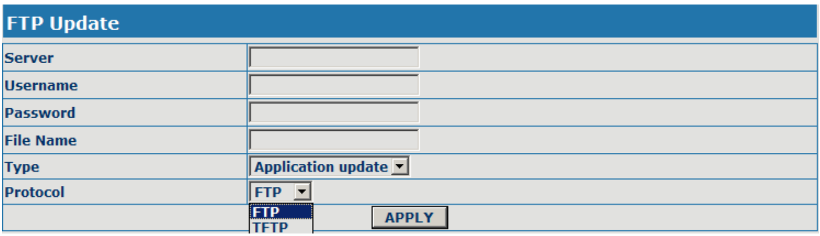
# 13 Upgrade on-line

## 13.1Upload WEB page

On this page, user can select the upgrade document (firmware or config file) on hard disk of the computer directly to run the system upgrade. After the upgrade has been completed, restart the phone and it will be usable at once.


## 13.2 FTP/TFTP download

On this page, user can upgrade system and configure files by FTP or TFTP mode.



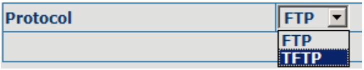

## 13.3 Configuration Explanation:


Configure upload or download FTP/ TFTP server IP address;


Configure username of the upload or download FTP server. If user select TFTP mode, username and password are not required to be configured;

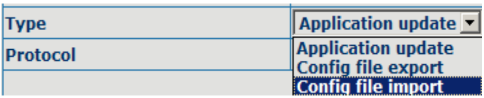

Configure upload or download of FTP server password;


Configure upload or download system upgrade document or system layout file name. It should be noted that system file take .dlf as suffix, configuration files take .cfg as suffix;

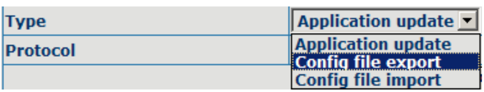



Select server type;


Select Application update type, the phone will upgrade system file;




Select Config file export type and click the Apply, the phone will upload its configuration files to FTP/TFTP server and save with names of user-defined configuration files;




Select Config file import type and click the Apply, the phone will download configuration files of FTP/TFTP server to the phone and the configuration will go into effect after restarting;

Output configure file can be edit, delete, or make comment starting by # on each command. Unit support module upgrade, like if changes made to SIP configure, others in configure file can be deleted and configuration in unit will not be affected.

While upgrade unit with modified configure file, please make sure check each parameter while finished upgrade. In case of anything wrong, please recover configure under POST mode.

## 13.4 Configure file encryption

Configure file can be encryption with DOS command:

dsc.exe <key.txt> <e/d> <old configure> <new configure>.

Dsc.exe-encryption software tool

<key.txt>-user made encryption key file

<e/d> e（encrypt）, d (decrypt)

< Old configure >former configure file name and path,

< New configure >new configure file name, defined by user.

## 13.5 Auto-update

Unit can be set as automatically upgrade from desired FTP or TFTP server.

| Auto Update Setting | |
|---|---|
| Current Config Version | 2.0001 |
| 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 |
| APPLY | |

Display the current config version;

| | |
|---|---|
| Current Config Version | 2.0001 |

Configure unit as follow steps:

| | |
|---|---|
| Server Address | 0.0.0.0 |

Input IP add. of desired FTP server.

| | |
|---|---|
| Username | user |

Input user name of desired FTP server.

| | |
|---|---|
| Password | •••• |

Input user password of desired FTP server.

| | |
|---|---|
| Config File Name | |

Input name configuration file. Software version must be different for each upgrade file.

| | |
|---|---|
| Config Encrypt Key | |

If configuration file is encrypted, password needed.

| | |
|---|---|
| Protocol Type | FTP |

Chose server type as either FTP or TFTP.

| | |
|---|---|
| Update Interval Time | 1 Hour |

Set auto-upgrade interval duration.

| | |
|---|---|
| Update Mode | Disable |

Chose auto-upgrade type.

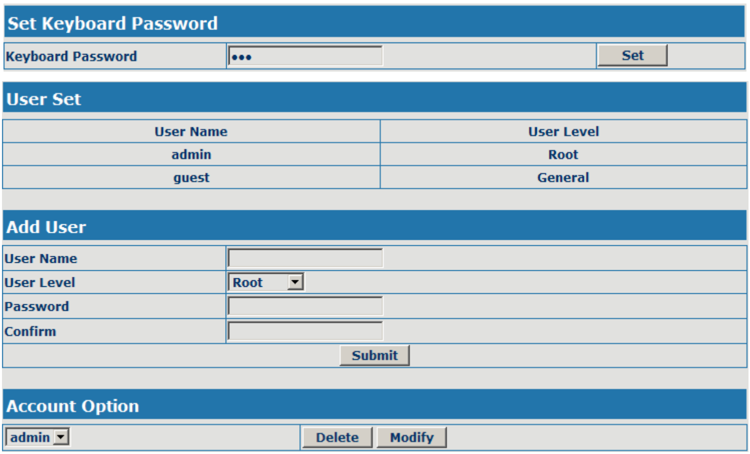
## 13.6 Configuration files WEB download

On this page, user can directly select the configuration files on the hard disk of the computer, and then make modification to the system configuration, after the download, restart the phone and the configuration will go into effect.

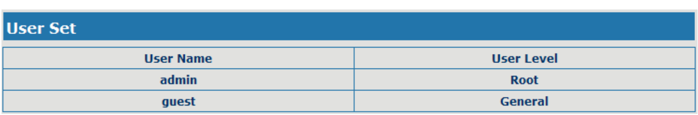
# 14 Syetem management

## 14.1Account management

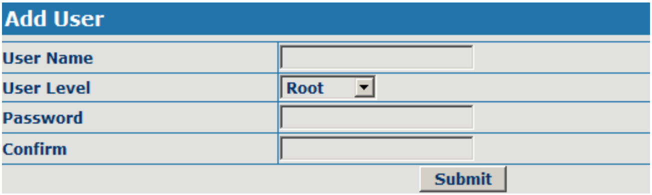
On this page, user can set the keyboard password, the same time, user can add and delete users according to own needs and can modify user's authorities there have been.




## 14.2Configuration Explanation:




display of phone user account list;

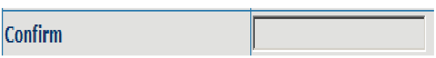



To add phone account; it will be shown at lower part of page as the following figure, of which:

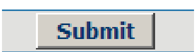

Add new accounts;


account level; root possesses

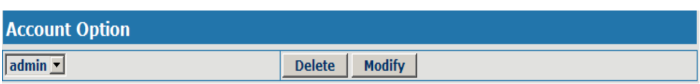

authorities to modify configuration,general possesses read-only authority;as corresponding password of the additive account;




As second confirmation of password, to ensure correct setup of password;

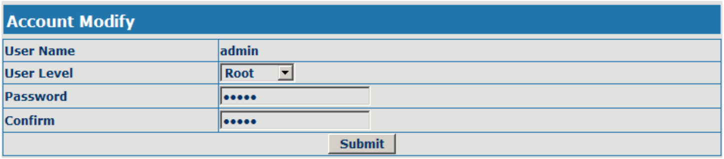



Clicks submit to go into effect.




Select users that you want to delete in the drop-down menu, click Delete.

To modify the chosen accounts, need to select account first, click Modify,it will be shown at lower part of page as the following figure, of which:





The modified username;


Modify user authorities;


Modify user password;


Make confirmation of the modified user password;


Submit the modification;

Owing to the phone's default account: accounts of the administrator level-admin and the ordinary level－guest are all weak account and weak password, the username and password will be easily to be guessed on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification, create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.),set password and then save configuration, entering with new manager account, delete default manager and browse account and save configuration, security will be enhanced!

## 14.3 Time zone configure

On this page, user can save and configure time zone setting.

| 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 | |

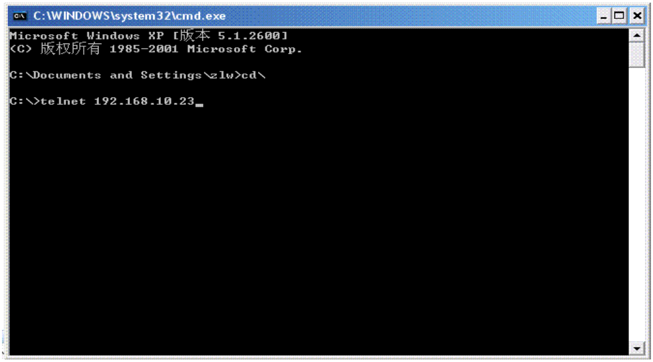
| Manual Timeset | |
|---|---|
| Year | |
| Months | |
| Day | |
| Hour | |
| Minute | |
| | APPLY |

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

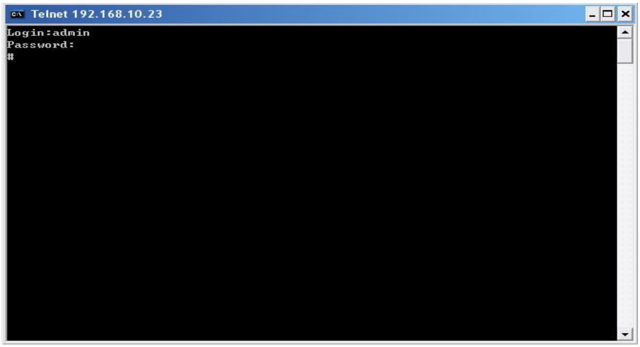
Configure the desired time zone.

# 15 Configuration via Telnet

In DOS window, input telnet 192.168.10.23, enter:




Then input USER NAME：admin PASWORD：admin




## 15.1Basic Command

input “help” or “？”under terminal to check all sub-terminal and globe command; input “help” or “？”under command to check parameters;

input “！” or“exit”to quit former path. 。

while partly input terminal or command, press“TAB”, system will auto finish balance command or list all option.

each help is with comment as <command> or <node> to identify.

parameter of each command including two types:“required” and“optional”:

all “required”start as “-”; all option， start as“_”. Also partly input available like point 3.

after configure, make sure save with command “write”, other wise, all setting lost while re-start.

## 15.2 Command structure

Root terminal with structure as

#

---config

---debug

---download

---language

---password

---setdefault

---show

---telnet

---trancert

---update

---upload

Most command parameters are under terminal “config”, which structures as follows:

<config>#

---accesslist

---dialpeer

---digitalmap

---fastethernet

---mmifilter

---nat

---port

--qos

---syslog

---time

---user

---voip

---vpn

### 15.3 Structure of Configuration terminal

Access list firewall configuration

Path：<config-accesslist>#

[no] entry - <command>Set access list table

[no] in-access - <command>Enable/Disable In-access

[no] out-access - <command>Enable/Disable Out-access

show - <command>Show access list

Ex.: <config-accesslist>#

Add protocol ---entry –I/O xxx –P/D xxx –sraddr x.x.x.x –srcmask x.x.x..x–desaccr x.x.x.x –desmask x.x.x.x –portrange xxx–portnum xxx

Ex.： <config-accesslist>#entry – I/O input – P/D deny – proto udp – straddr 202.112.10.1 – srcmask 255.255.255.0 – desaddr 210.25.132.1 – desmask 255.255.255.0 – protrange neq – portnum 5060

Del protocol ---no entry –I/O xxx –index xxx

Ex： <config-accesslist>#no entry – I/O input – index 1

Fastethernet–Lan configuration

Path： <config–fastethernet–lan>#

[no] bridgemode - <command>Enable/Disable bridge mode

[no] dhcp-server - <command>Enable/Disable DHCP Server

dhcpshow - <command>Show DHCP current leased table

[no] ip - <command>Set lan IP

ipshow - <command>Show LAN interface configuration

[no] nat - <command>Enable/Disable NAT

natshow - <command>Show current NAT status

Ex: <config–fastethernet–lan>#ip –addr 192.168.1.10 –mask 255.255.255.0

Fastethernet–Wan configuration

Path： <config–fastethernet–wan>#

[no] dhcp - <command>Start DHCP client

[no] gateway - <command>Set default gateway

[no] ip - <command>Set WAN IP

[no] pppoe - <command>Enable/Disable PPPoE client

[no] qos - <command>Enable/Disable 802.1p QOS

show - <command>Show WAN interface configuration

Ex: <config-fastethernet-wan>#ip –addr 202.112.241.100 – mask 255.255.255.0

MMI FILTER

Path： <config-mmifilter>#

Add: ---entry –start x.x.x. –end x.x.x.

Ex： <config-mmifilter>#entry –start 202.112.20.1 –end 202.112.20.255

Del ---no entry –start x.x.x.x

Ex： <config-mmifilter>#no entry –start 202.112.20.1

[no] entry - <command>Set IP filter table

modifyfilter - <command> modify mmifilter table

show - <command>Show IP filter table

[no] start-filter - <command>Enable/Disable MMI IP filter

NAT

Path： <config-nat>#

[no] ftpalg - <command>Set NAT FTP application level gateway

[no] ipsecalg - <command>Set NAT IpSec application level gateway

[no] pptpalg - <command>Set NAT Pptp application level gateway

show - <command>Show current NAT state

[no] tcp-entry - <command>Set NAT TCP map table

[no] udp-entry - <command>Set NAT UDP map table

Add TCP tcp-entry–ip x.x.x.x–lanport xxx–wanport xxx

Ex： <config-nat>#tcp-entry –ip 192.168.1.5 –lanport 1720 –wanport 1000

Del TCP ---no entry –ip x.x.x.x –lanport xxx –wanport xxx

Ex： <config-nat>#no tcp-entry –ip 192.168.1.5 –lanport 5060 –wanport 1000

Add UDP ---udp–entry–ip x.x.x.x –lanport xxx –wanport xxx

Del UDP ---no udp–entry –ip x.x.x.x –lanport xxx –wanport xxx

Check NAT ---show

Port configuration

While input PORT under terminal config, the configuration will valid to all ports, if input as PORT X, valid only to port X. Some parameter only valid to some port, then PORT X is needed, otherwise, error report as “Error: Missing parameter”.

Path: <config–port>#

[no] accept-relay - <command>Set accept relay mode

[no] calltransfer - <command>Enable/Disable call transfer

[no] callwaiting - <command>Enable/Disable call waiting

codec - <command>Set Codec

[no] fastcalling - <command>Set fastcalled number

handdown - <command>Hand down delay

[no] in-limit - <command>Set the number which will be not accepted

[no] input - <command>Set Input gain

[no] out-limit - <command>Set the number which can not be dialed

[no] output - <command>Set Output gain

[no] ringvolume - <command> set ring volume

show - <command>Show port configuration

[no] shutdown - <command>Disable/Enable the port

signalmode - <command>Set signal mode

[no] threetalk - <command>Enable/Disable threetalk

QOS

Path: <config-qos>#

[no] 8021p - <command> set 802.1P Priority

[no] diffsevenable - <command> Enable/Disable DiffServ

[no] diffsevvalue - <command> set DiffServ Value

show - <command>Show QOS configuration

[no] vlanid - <command> Set VLAN ID

USER management

Path: <config–user>#

[no] entry - <command>add a user with given level

password - <command>Change password

show - Show all users

Other configuration except Config terminal

TIME

Path: <config-time>#

Manual time setting ---manualset –year xxx –month xxx –day xxx –hour xxx –minute xxx –second xxx

Ex: <config-time>#manulset –year 2004 –month 10 –day 1 –hour 8 –minitute 30 –second 0

manualset - <command>Manual set system time

print - <command>Print SNTP time

sntp - <node>Get current time by using SNTPUpdate

Path: #

Via FTP ---update ftp –user xxx –password –ip x.x.x.x –file x.x.x

Ex: #update ftp –user abc –password 123 –ip 202.112.20.15 –file abc.dlf

Via TFTP ---update tftp –ip x.x.x.x –file xxx

Via FTP uploading file ---upload ftd –user xxx –password xxx –ip x.x.x.x –file xxx

Via TFTP uploading file ---upload tftp –ip x.x.x.x –file xxx

Via FTP download file ---download ftp –user xxx –password xxx –ip x.x.x.x –file xxx

Via TFTP download file ---download tftp –ip x.x.x.x –file xxx

Other commands

Setting module debug message level ---debug all xxx

Setting MGR module debug message level ---debug MGR xxx

Setting SIP module debug message level ---debug sip xxx

Setting IAX2 module debug message level ---debug IAX2 xxx

Remove module debug message level ---debug no all

Remove MGR module debug message level ---debug no MGR

Remove SIP module debug message level ---debug no sip

Remove IAX2 module debug message level ---debug no IAX2

Reset to default ---setdefault

Reset all to default ---setdefault all

Check message of some module ---show xxx

Update present password ---password

Telnet remote login ---telnet x.x.x.x

Use special Telent port ---telnet x.x.x.x –port xxx

Telnet quit ---logout

save ---write

re-start ---reload

get help ---help

quit ---exit

clear screen displa ---clear

PING host ---ping x.x.x.x

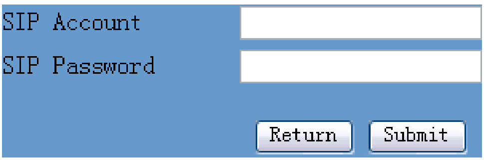
trace ---trancert x.x.x.x

If user has accounts of a certain SIP server and each account has different password, then user should add each account and its corresponding password to the account& password table.

| SIP Account | Password |
|---|---|
| 1000 | 1000 |

Configure display of account & password list;

Click Add to add account and password, it is shown as the following figure:

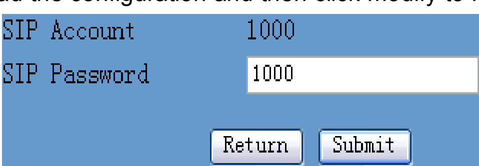



Configure additive passwords

Click submit to submit the configuration, click return to cancel the configuration and return;


Select accounts that you want to delete from the drop-down menu, click delete.


Select drop-down menu to select accounts that want to modify, click load to load the configuration and then click modify to modify;




Accounts to be Modified, read-only;

→Passwords to be modified;

Click submit to submit, click return to cancel the modification and then return.