IP Phone User Manual

This Manual provides basic information on how to install and connect IPH303 IP Phone to the network. It also includes features and functions of IPH303 IP phone components, and how to use them.

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Thank you for your purchase of our products ---- CDX-IPH303 enterprise-class IP phones. You make an excellent choice; we hope you will fully enjoy all the features of the product.

CDX-IPH303 is a highly innovative enterprise IP telephone, to provide you the plenty of features and beautiful sound quality. Products fully comply with the SIP protocol standard, and Interconnection with the majority of SIP interoperability of hardware and software equipment of the market.

The content of this manual is subject to change without prior notice.

1 Safety Instructions

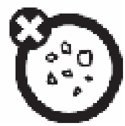
Warning:

Read the safety precautions and the user guide before use.

Explain their contents and the potential hazard associated with using the telephone to your children.

Liquids of any kind

Don’t expose your phone to water, rain, extreme humidity, sweat, or other moisture.



Dust and dirt

Don’t expose your phone to dust, dirt, sand, food, or other inappropriate materials.

Extreme heat or cold

Avoid temperatures below -10°C/14°F or above 45°C/113°F.

Cleaning solutions

To clean your phone, use only a dry soft cloth.

Don’t use alcohol or other cleaning solutions.

Microwaves

Don’t try to dry your phone in a microwave oven.

The ground

Don’t drop your phone.

Safety Standards

CDX-IPH303 follows various safety standards, including FCC / CE. The power adapter of product follows the UL standard, the phone can only be used the power adaptor provided by inner packaging, the damage caused due to the use of other power adapter, which does not belong to the scope of quality assurance of the manufacturer.

2 Before Getting Started

Before you can connect CDX-IPH303 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. CDX-IPH303 IP phone is a stand-alone device, which requires no PC to make Internet calls. CDX-IPH303 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.

3 Package contents

1. CDX-IPH303 IP phone base unit.
2. Handset
3. Coiled handset connecting cord
4. One Straight Ethernet cable
5. One Power supply
6. User guide
7. 2 wall mount bracket
8. 1 table bracket

4 SPEC and Features

4.1 Hardware Spec

- 32-bit 150MHz MIPS CPU
- 16-bit 100MHz DSP
- 4MB flash memory
- 16MB SDARM
- 128x64 pixel graphic LCD with backlight
- 34 KEYS, 1 LED
- WAN: 10/100M RJ45
- LAN: 10/100M RJ45
- RJ9 Headset Jack(option)
- 2.5MM earphone Jack(option)

4.2 DSP Spec

- G.711A/u, G.723.1（5.3k/6.3k）, G.729a/b,G.722,G726
- 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

4.3 Software feature

- Languages: English
- RFC Protocol Edition: RFC3261 and RFC2543
- Compatible with all major SIP Servers: Cisco, Osip, Vocal, ser, Partysip, Simens, Grandstream, etc.
- Server authentication mode: none, basic, MD5
- Peer to Peer SIP call
- Line 1/ Line 2. can support two different SIP servers.

- DTMF Mode: RFC2833, RELAY, SIP INFO
- DNS name of SIP server
- SIP signaling port setting
- NAT traverse, STUN
- NAT traverse, SIP Express router
- 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
- Headset
- Dial Map Table
- 5 Speed dial key.
- 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

4.5 Networking Standards

- WAN/LAN port with Router or Bridge Mode
- NAT ALG
- PPPoE for xDSL, automatically keep alive
- DHCP Client on WAN
- DHCP server on LAN
- DNS client with 2 servers IP
- SNTP
- RTP: RFC3550
- 802.1P QOS

4.6 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

4.7 Physical & Environmental

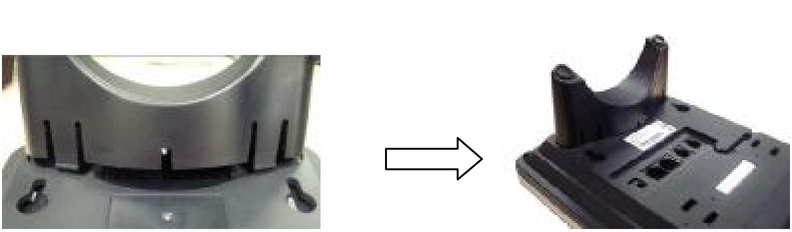
Desktop / Wall mounting
Power Input: 100 to 240 ACV / 50/60Hz Output:7.5 DCV
Dimensions: 210 x 170 x 130 mm
Weight: 680 g (main unit)
Operating Temperature: 0°to +40°C (32°to +104°F)
Storage Temperature:-20°to +70°C (-40°to +158°F)
Humidity: 5% -95% non-condensing

5 Installation

Use the cable for Internet connection; connect the ‘WAN’ port of CDX-IPH303 to router. Get the cable from box and connect the LAN port of phone to your PC. Connect the power adapter in the box to ‘DC’. Then start your phone.

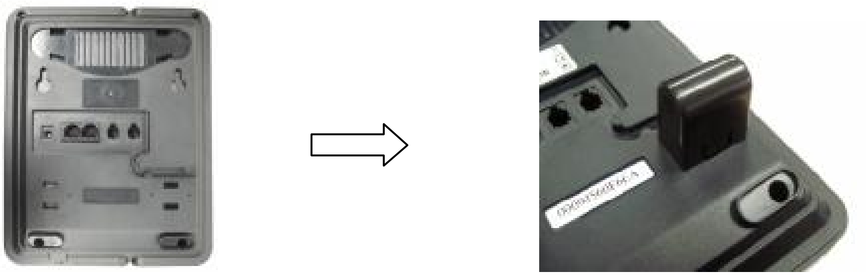
Power Jack	**7.5V DC power connected port**
LAN	**10/100Mbps RJ-45 PC connected port**
WAN	**10/100Mbps RJ-45 Ethernet connected port**
	**RJ9 Headset Jack**
	**Handset Jack**

Desktop installation:



Wall installation

This phone can be installed on the wall, the back of base has two wall -mounted ports:



To install the phone on the wall, put two pothooks on the wall, hang the wall port of the phone back on the wall. The two supporting frames will balance the bottom of the phone and place it to the wall

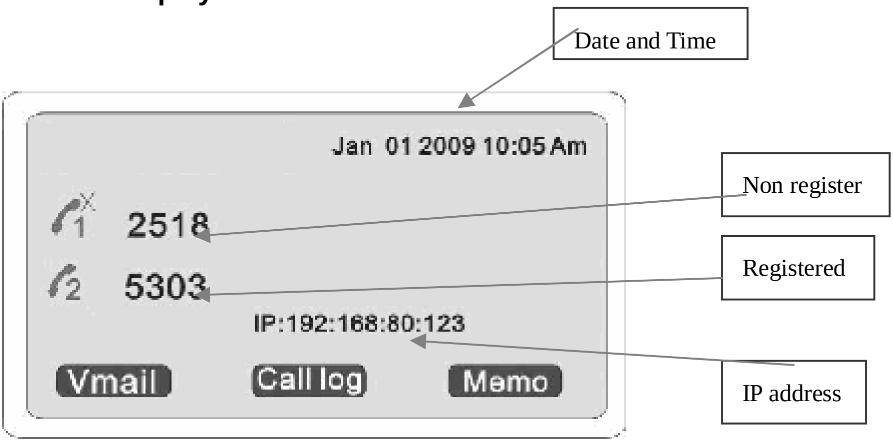
OFF

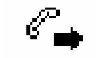
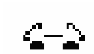
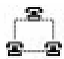
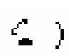
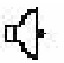
ON

Use the handset, pull out the wall-mounted label (extension of the down) from the handset bracket, rotate the wall label and then insert to wall-mounted label slot, caught by the extension of the handset.

6 General Appearance

6.1 LCD Display

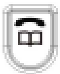
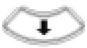
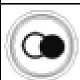


	LCD illustration	Definitions
1		Call forwarding enabled
2		Call on hold
3		Connected call
4		Incoming call
5		Conference call active
6		Conference call on hold
7		Conference call disabled
8		Outgoing call
9		Outgoing call not completed
10		Transferring a call
11		Line 1 (idle)
12		Line 2 (idle)
13		Line disabled
14		Handset in use
15		Speakerphone in use

16		Message waiting
17		Missed call

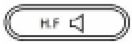
6.2 Key function Definitions

	Key Button	Key Button Definitions
1	**0 - 9, *, #**	Digital, star and pound keys are also used for setting and call process.
2		Soft key 1、2、3 , Display functions as per the LCD menu requested
3		Menu: enter the main menu, or confirm current status

4		LINE1 and LINE2 : It is the transparent key, it will be flash when incoming call, it will be normal on the call, it will be flash on Hold.
5		Hold: hold the active call. The transparent key has red indicator.
6		Call List: browse the call logs
7		Mute key: Used as Mute and quiet on the call, The transparent key has red indicator.
8		Phone Book: Enter to the phonebook to recall and amend the phone numbers.
9		Navigation key used as UP in menu or Increase handset / speakerphone volume
10		Navigation key used as DOWM in menu or reduce handset / speakerphone volume
11		Navigation key used as LEFT in menu or reduce handset/speaker volume
12		Navigation key used as RIGHT in menu or increase handset/speaker volume
13		Headset: activate/deactivate the headset function
14		Speaker: activate/deactivate the hands-free function
15		Redial: Dial a new number or redial the last call.
16	**M1-M5**	Speed dial: make the speed dial call

6.3 Indicator Instruction

	Key light	**Definitions**
1		For each line： l On (steady)-Active call

		l Slow blink-Held call l Fast blink-Incoming call
2		On – Speaker in use
3		On – headset in use
4		On – Phone is muted
5		On – Phone is held
6	**M1~M5**	BLF indicator status： l ON：appointed extension is being used l OFF：appointed extension is on line and on standby FLASH：appointed extension is off line and not registered
7	LCD Backlight	Backlight status： 1. Always light on standby ： missed call or voice message. 2. Always light on using.

7 Phone Operations

7.1 Point to Point Call

Pick up or press the 【H.F】,【】key,【1】、【2】, then input "#Phone IP address#" for example：The IP phone of the other side is：192.168.0.11，it should be when using keyboard input：#192*168*0*11**5060#，* indicates point<.>，** indicates colon<:>

7.2 Two-SIP account and line

CDX-IPH303 support 2 independent SIP accounts. Each account can support independent SIP server, user name and NAT configuration. Line button 【1】、【2】 are match to sole SIP account , when picking, press a free line (【1】 or 【2】) button, and at the same time hear the dial tone. In this state, use UP / Down keys can be cut between the two lines.

7.3 Make a call

1. Use headset, press <SPEAKER> or LINE1/LINE2 key

Step 1: pick up / press 【H.F】 key / use a 【】 or press 【1】or 【2】 key (activated).

Step 2: Telephone dial tone will be sounded and 【1】 LED indicator will be light. You can press 【↑ / ↓】 key to select the other SIP accounts

Step 3: input the number, press 【#】 key to send

2. Use the redial key

Redial the last called number (redial, the phone will use the SIP account of the last call)

Step 1: pick up /press 【H.F】 key / use a 【】 or 【1 or 2】 key, the corresponding LED will light and hear the dial tone (The function is not available now and will be finished some time).

Step 2: Press 【】 Redial key to dial the last dialed number.

Check the dialed record and redial

Step 1: On the standby, press 【】 key to check the dialed record,

Step 2: Use the 【↑ / ↓】 key to select the corresponding call records,

Step 3: Press Redial/pick up/Speaker/LINEx/Headset.

3. Use the phonebook menu to make the call

Press Phone book key to enter the phonebook menu, then select the dialed number, as per the prompt of soft key to process. Use the related SIP account to make a call with hand free.

4. Use the call record to make the call

Press Soft Key 2 to enter the Calllog menu，use 【 / 】 key to check the Missed Call/Incoming Call/Outgoing Call record，select the dialed number, then pick up or press 【H.F】【 】【1 or 2】 to make the call.

5. Standby dial-up call

On standby, dial the called number ,press the soft key 3<Dial>or pick up, press【 H.F 】 or【 】 【1 or 2】，only use LINE 2 key to call from the second line, the other defaults are from LINE 1.

6. Use the phonebook number to make the call

On standby, press the Phone book key to enter the phone booklist , select the dialed number, as per the prompt of Soft Key to process the operation, you can pick up, press 【H.F 】or【 】 【1 or 2】 key to call.

7.4 Phone book number store / edit / delete

On standby .press Phone book key, then as per the prompt of Soft Key to process Store/Edit/Delete.

7.5 Call list check / delete

On standby , press Soft Key 2 to enter Call Log menu，then as per the prompt of Soft Key to process Check / Delete.

7.6 Hold

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

7.7 Mute

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

7.8 Volume adjustment in conversation

During conversation, press navigation 【↑ / ↓】 key to adjust receiving volume, or adjust the volume on menu.

7.9 Rejected call

During new incoming call, as per the prompt of the LCD Soft Key，press soft Key 3 <REJECT> to reject the incoming call.

7.10 Black(White) list setting / edit / delete

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

7.11 Call transfer

During conversation，as per the prompt of LCD Soft key，press Soft Key 3<transfer>, and input transferred number end with 【#】 to transfer the phone to the third part and hang up automatically

7.12 3-party conference

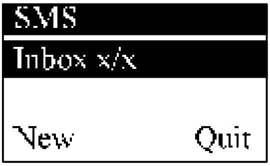
During conversation，as per the prompt of LCD Soft key，press Soft Key 2<Conf>, then dial another number plus # button, while line connected, press Soft Key 2<Conf> to make conference call;

7.13 SMS Function

- Ø The function standard RFC3428（Session Initiation Protocol (SIP) Extension for Instant Messaging）
- Ø This feature must need the support of the SIP server and end phone
- Ø Function operation

7.13.1 Create/New . send message

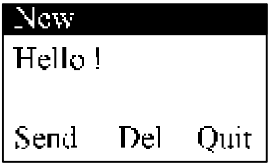
1. On standby state, press Soft Key1 <SMS>to enter SMS menu , see below photo



Inbox x/x：The first letter X means the missed message number, and last letter X means the total message number.

New: Create the new message，Quit: Back to SMS menu

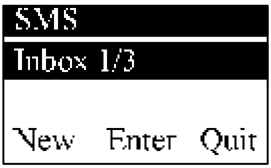
2. New/ Create the new message，see below picture：



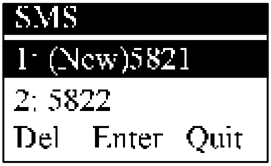
3.Send message: input the extension number and press soft key 2 to send message or select the number on notebook and press soft key 2 to send message.

7.13.2 Read, delete and reply message

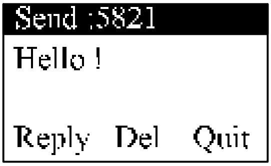
1. On standby , press Soft Key <SMS> to enter to SMS menu，as below picture：



2. Press Soft Key 2<Enter>，you can delete or enter to the message as per the prompt of the Soft Key.



3. Press Soft Key 2 <Enter> to check the messge，you can reply or delete the message as per the prompt of the Soft Key.



4. Press Soft key 1<Reply> to reply the message，enter to the state to input the message, after finished inputting, then press Soft key 1<Reply> to send the replied message.

8 Configuration with keypad and LCD display

8.1 Main menu list operation

In standby, press 【**MENU** 】button till LCD shows

-Configuration

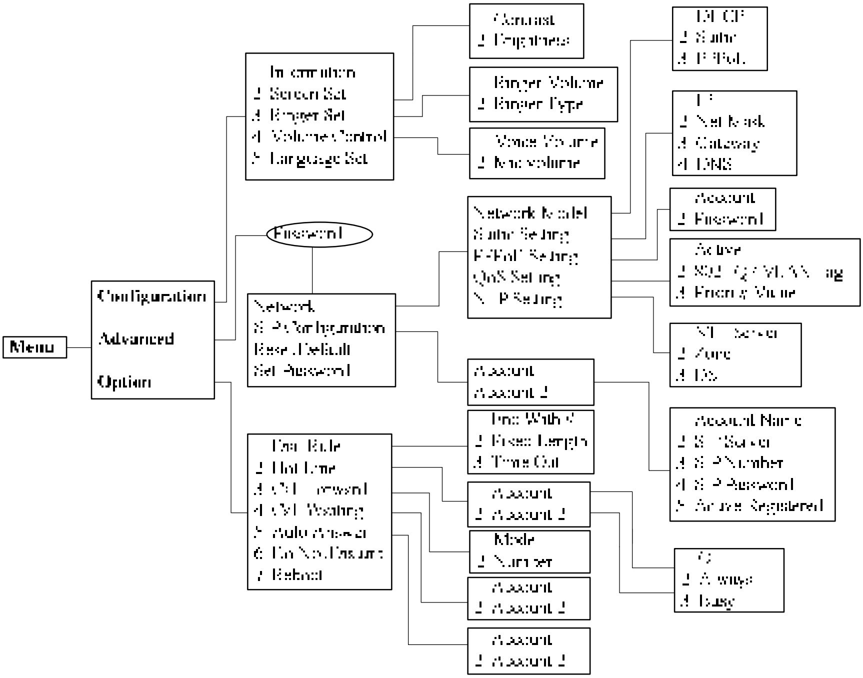
-Advanced

-Option

Three Sub-menu option，“Configuration”is used the configured property parameter，configured the advanced parameter and password protected, the default password is 123，Option the other related functions.

Use 【↑】 and【↓】key to select the related sub item ， and process the operation as per the LCD soft key.

The instruction for the main menu list as below



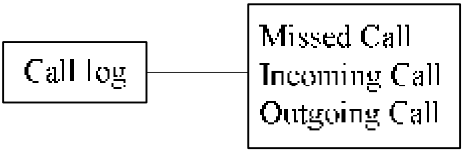
8.2 Phone book operation

On Standby, press Phone Book to enter the phone book menu list, use 【↑】and【↓】key to select the sub item, and process the recall, check, edit, delete etc. as per the LCD soft key

8.3 Call list check

On standby , press Soft Key 2 to enter Call log menu，use【↑】and【↓】key to select the related sub item, and process check, delete, edit ,save etc

as per the LCD soft key.
the instruction for call list as below：



8.4 SMS operation

On standby, press Soft Key 1to enter SMS menu，use【↑】and【↓】key to select the related sub item, and process the related operation as per the LCD soft key.

8.5 Memo operation

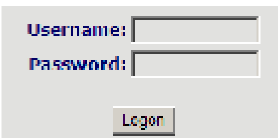
On standby, press Soft Key 3 to enter Memo menu，use【↑】and【↓】key to select the related sub item， and process the related operation as per the LCD soft key.

9. WEB Interface Configuration

The IP Phone Web Configuration Menu can be accessed by the following URI: http://Phone-IP-Address. The default LAN IP address is “192.168.10.1” and WAN IP address is DHCP.
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.

9.1 Logon Web

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



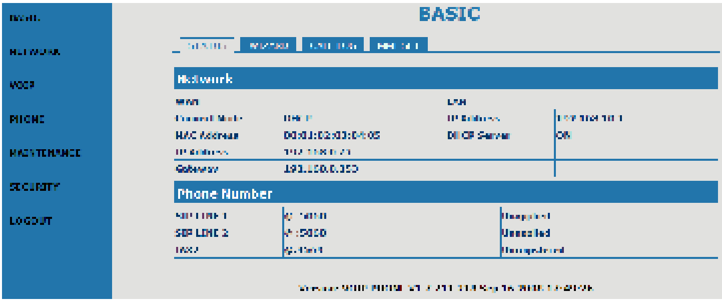
There are two level login as:

-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:



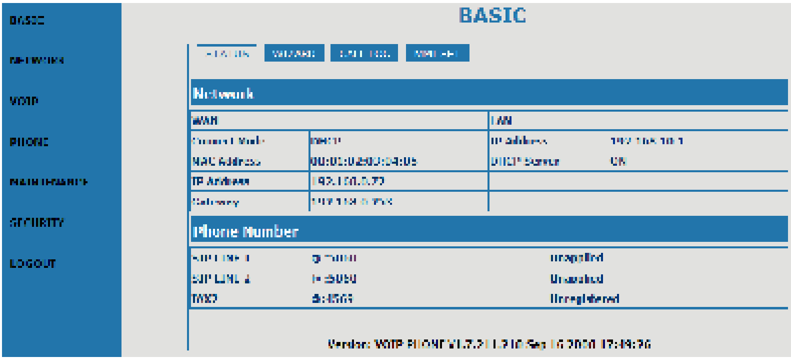
9.2 Current State

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

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

- 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.



9.3 Network Configuration

9.3.1 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.

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

WAN Setting		
Static	DHCP	PPPOE
Obtain DNS server automatically		
Static IP Address	192.168.1.179	
Netmask	255.255.255.0	
Gateway	192.168.1.1	
DNS Domain		
Primary DNS	202.96.134.133	
Alter DNS	202.96.128.68	
	APPLY	

PPPOE Server	ANY
Username	user123
Password	********
	APPLY

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;

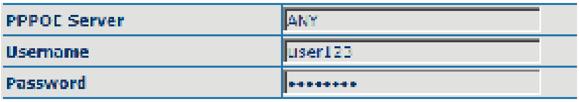
DNS Domain	voip.com

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;

Main DNS server IP address;

The second DNS server IP address;

Configure PPPoE:



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

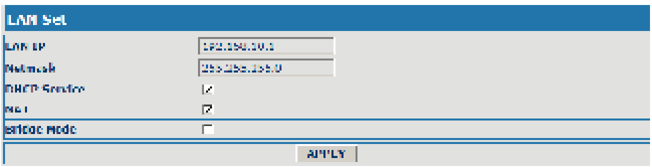
PPPoE account;

PPPoE password;

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

9.3.2 Local area network (LAN)

User can make local area network (LAN) configuration on this page, when bridging mode is selected, the local area network (LAN) configuration will no longer go into effect.



Configuration Explanation:

Use bridge mode（transparent mode）:bridge mode will make the phone no longer set IP address for LAN physical port, LAN and WAN will join in the same network;

Configure LAN static IP;

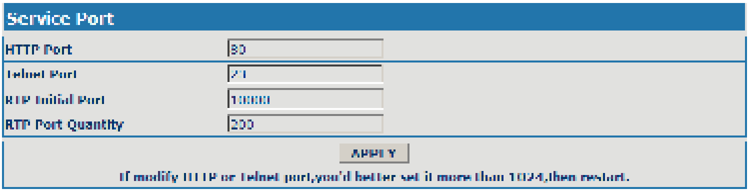
Configure LAN subnet mask;

Enable LAN port DHCP server; after user modify LAN IP, the phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table, user need to restart the phone to make DHCP server configuration go into effect;

Enable NAT.

Service Port configuration

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



Configuration Explanation:

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;

Configure telnet port, the default is 23 port;

Enable RTP initial port configuration. It is dynamic allocation;

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

Leased IP-MAC correspondence table of DHCP;

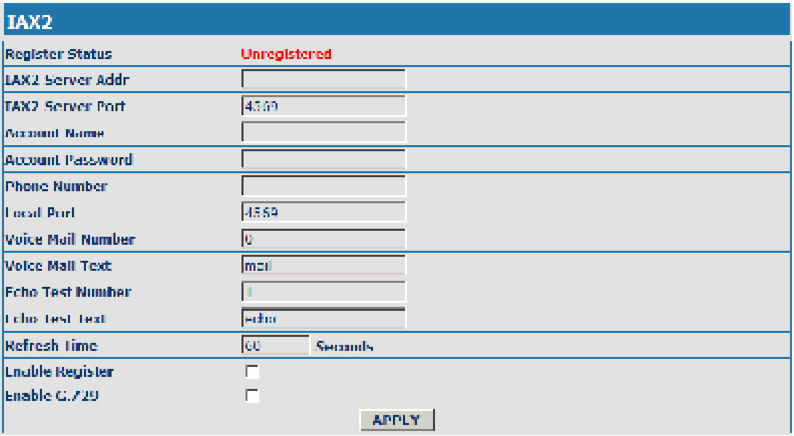
※The configuration on this page needs to be saved after modified and will go into effect after restarting.

※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.

10 VOIP Configuration

10.1 IAX2 Configuration



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;

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;

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

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;

Config echo test text;

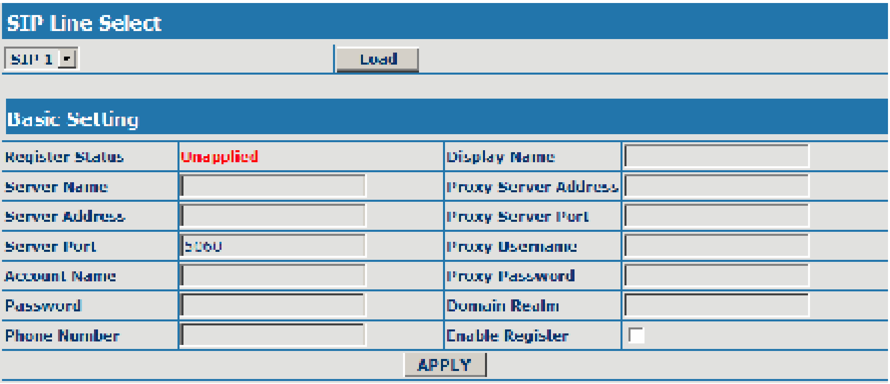
Config IAX2 refresh time, The unit of time for the second, suggested the user in makes the choice 60 to 3600 between;

config the permission/prohibition registers the server;

Config whether supports G.729;

10.2 SIP configuration

Sip register



Configuration Explanation:

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

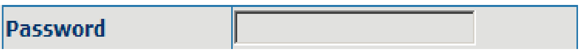
Show SIP register state; if register successfully, there will show Registered in the square bracket, otherwise show Unregistered;

Configure the name of registration server;

Configure SIP register server IP address or Domain Name;

Configure SIP register server signal port;

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）;



Configure password of SIP register account;

Configure the phone number;

Configure display name;

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）;

Configure SIP proxy server signal port;

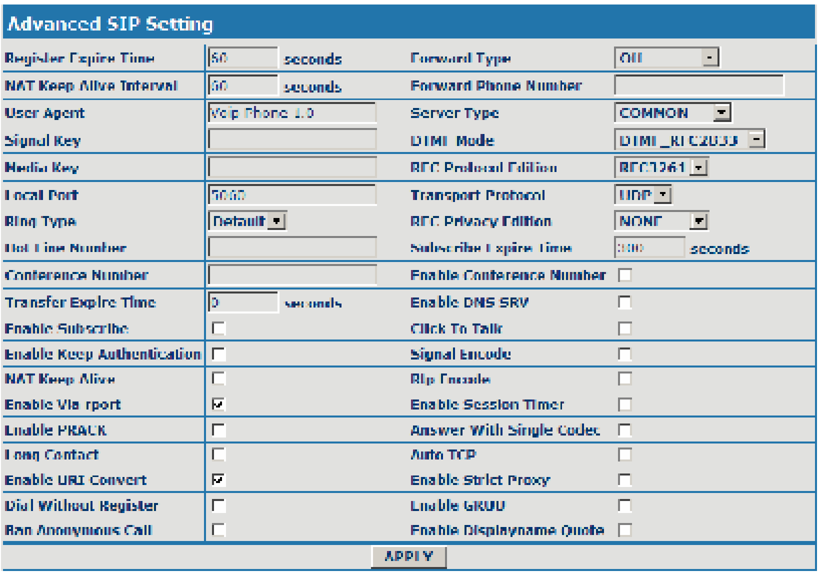
Configure proxy server account;

Configure proxy server password;

Configure domain realm;

Configure enable/disable register;

Advanced sip setting



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 (DTMF_RELAY / DTMF_RFC2833 / DTMF_SIP_INFO)
RFC Protocol Edition	
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	☐

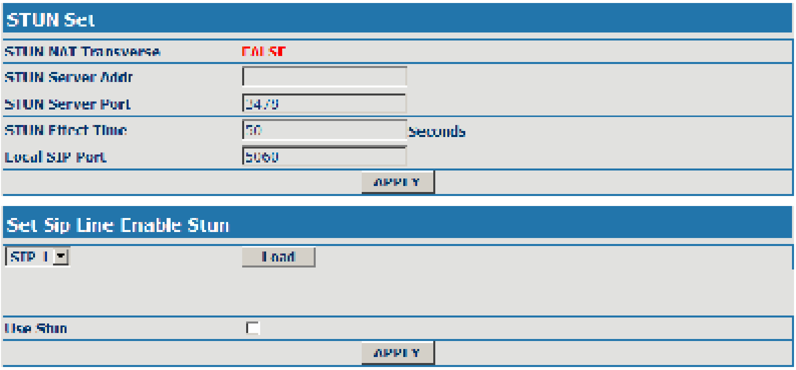
Configure enable/disable GRUU;

Enable Displayname Quote	☐

Configure enable/disable Displayname Quote;

10.3 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.



Display the application status of the STUN NAT;

Configure IP address of SIP STUN server;

Configure port of SIP STUN;

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

Configure Local SIP port;

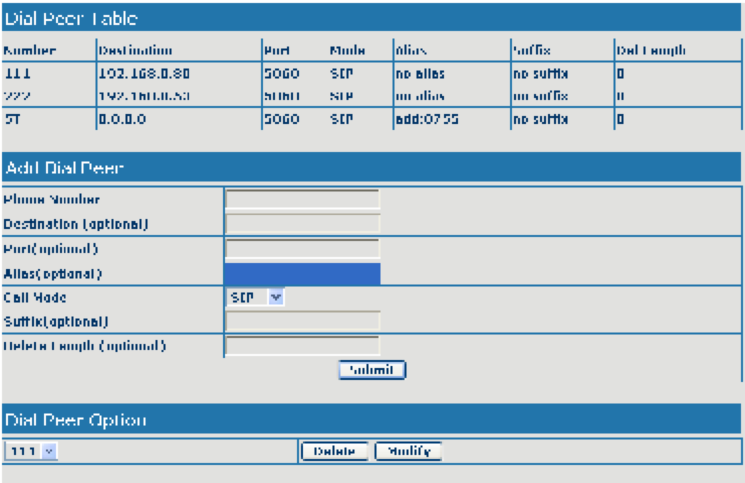
Select the Sip Line;

Configure enable/disable Use STUN;

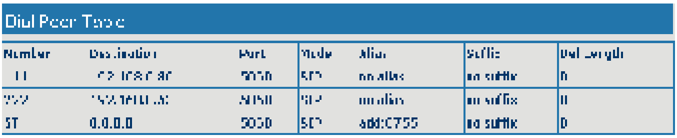
10.4 Dial Peer configuration

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

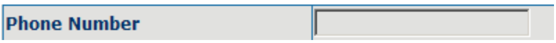
When making deletion or modification, select the number firstly, then click Delete or Modify , and complete the operation.



Configuration Explanation:



Display of calling number IP image list;



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.

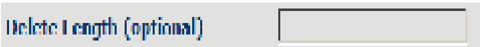
Configure destination address, destination is configured 0.0.0.0, it is SIP1 line.

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;

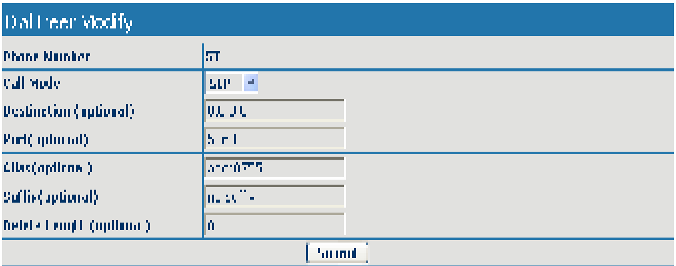
All: xxx, the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all" , the actual calling number will be replaced;

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:



this is the modified number. read-only;

To modify call mode;

To modify destination address; this is optional configuration item;

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

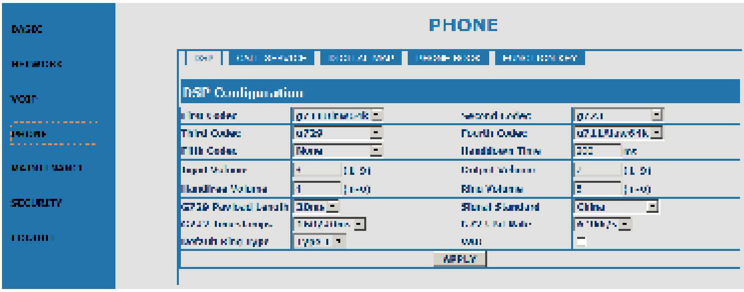
To modify alias; this is optional configuration item;

To modify suffix; this is optional configuration item;

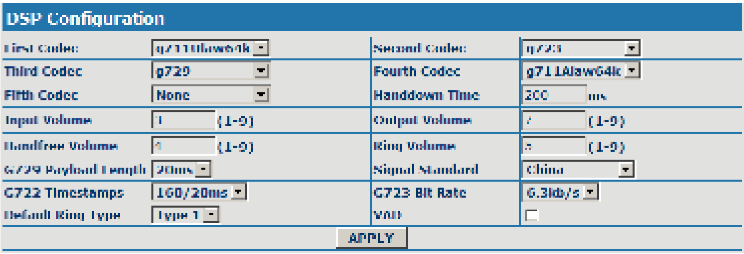
To modify replacing length（if rep and del of alias have been configured）;

Click submit to go into effect

11 Phone Configuration



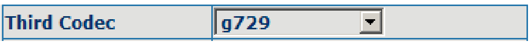
11.1 DSP configuration



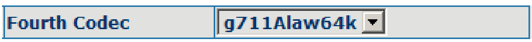
On this page, user can set speech coding, IO volume control, cue tone standard, caller ID standard and so on.

Configure the first codec;

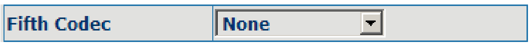
Configure the second codec;



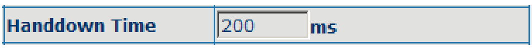
Configure the third codec;



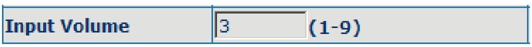
Configure the fourth codec;



Configure the fifth codec;



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



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

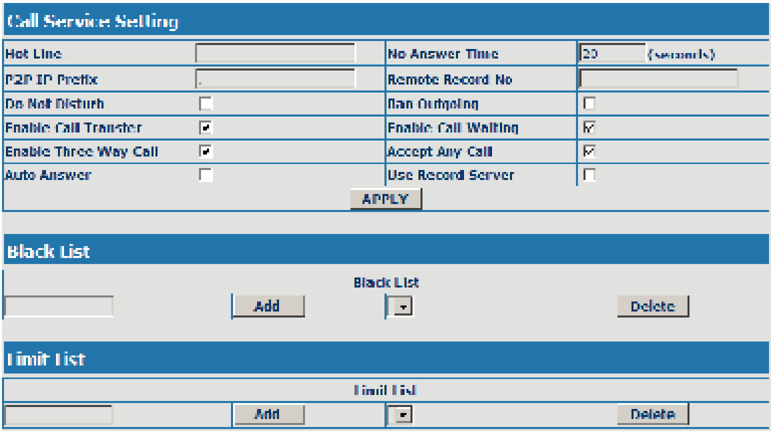
Configure default ring type;

VAD	☐

Configure enable/disable VAD.

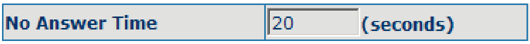
11.2 Call service configuration

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.



Configuration Explanation:

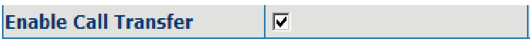
Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number;



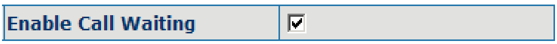
Configure no answer time;

Configure enable/disable Do Not Disturb;

Configure enable/disable Ban outgoing;



Configure enable/disable call transfer (CT); after it is enabled, automatically.



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;

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;

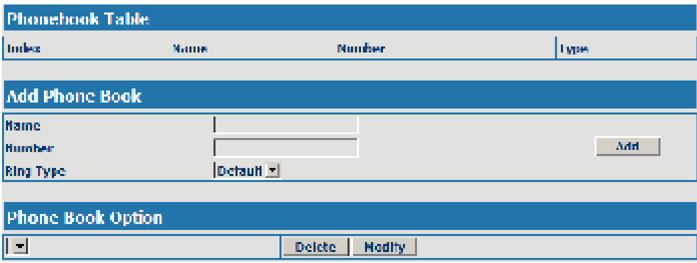
Configure enable/disable Accept Any Call;

Configure enable/disable Auto Answer;

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.

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.

11.3 Phone book configuration

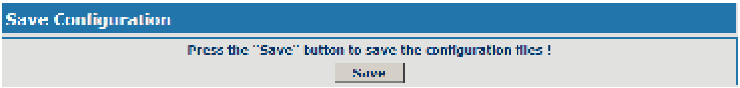


On this page, user can add, delete and modify telephone book.

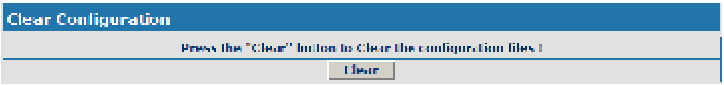
Some ISP internet may inhibit the phone to register and cancel the register in succession, so user had better not apply or register and cancel soon in succession and submit registration repeatedly. Server may stop response of dialogue machine, then the phone receives no certification of register/cancel login request and registration state will show as incorrect!

11.4 Save and Clear Configuration

User can save the current configuration on this page.



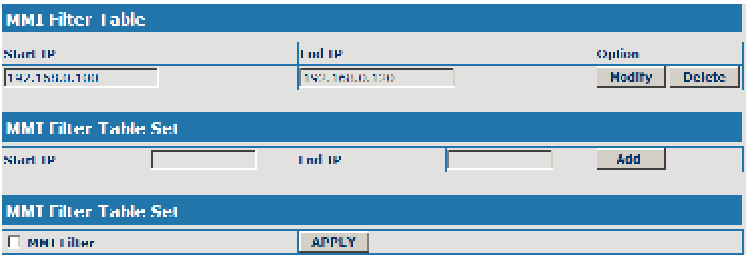
The system configuration can be set as factory default configuration on clear config page and the phone will restart automatically.

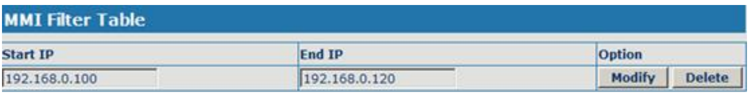


12 Security Configuration

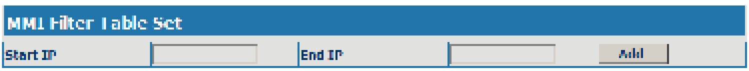
12.1 MMI Filter configuration

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;

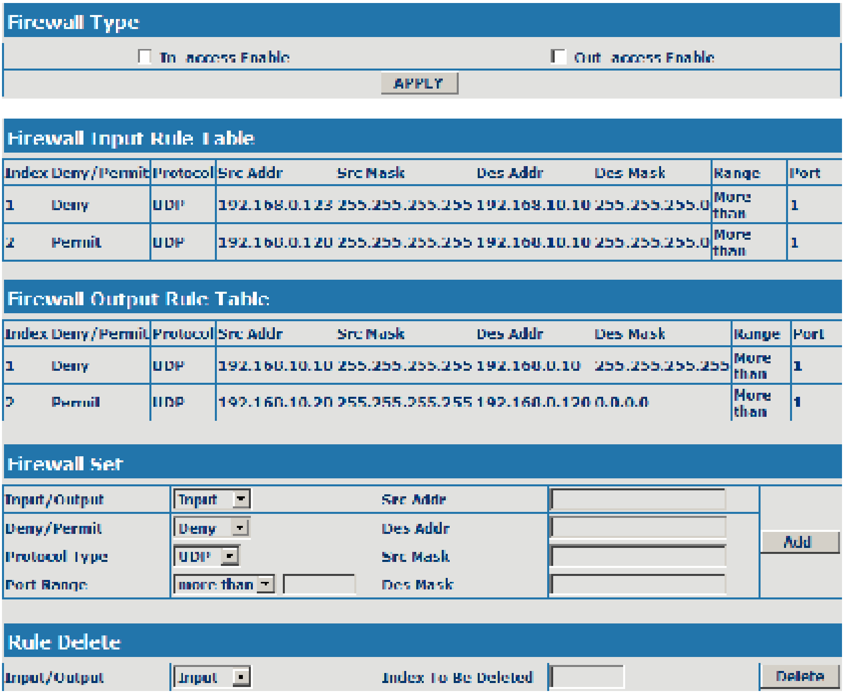


Add MMI filter table;

Configure enable/disable MMI Filter.

12.2 Firewall configuration

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

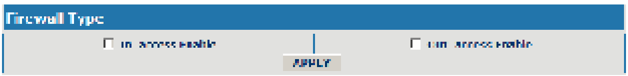


Firewall Input Rule Table

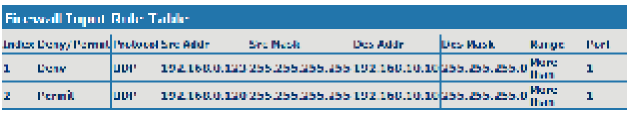
Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
1	Deny	UDP	192.168.0.123	255.255.255.255	192.168.10.10	255.255.255.0	More than	1
2	Permit	UDP	192.168.0.120	255.255.255.255	192.168.10.10	255.255.255.0	More than	1

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



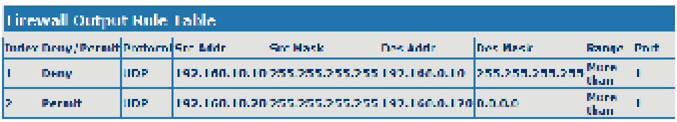
Select firewall type;



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;

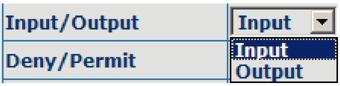


Firewall Output Rule Table

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

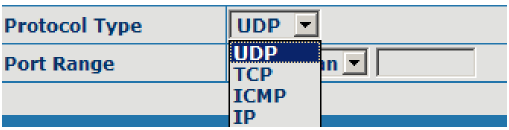
Display firewall output rule table; in the table，Src address and Src Mask confirm the range of the source address, For example：Src addr(192.168.10.10) and Src Mask(255.255.255.255) can confirm that the host is 192.168.10.10；Dse addr(192.168. 0.120) and Des Mask(0.0.0.0) can confirm any host; Src addr(192.168.10.20) 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.

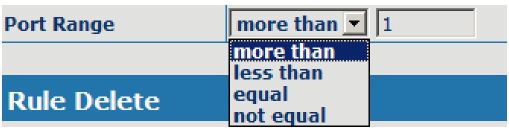


Select the rule of the firewall(input or output);

Select the condition(deny or permit);



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



Configure port range;

Configure source IP address;

Configure destination IP address;

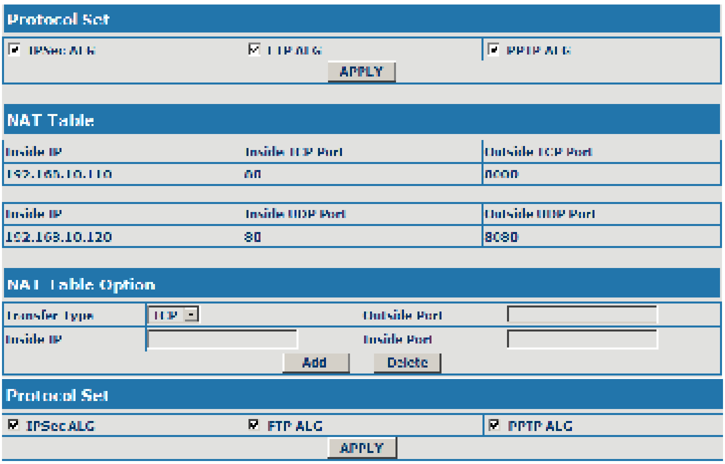
Configure source Mask;

Configure destination Mask;

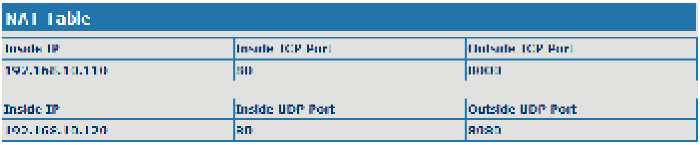
12.3 NAT and DMZ configuration

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.

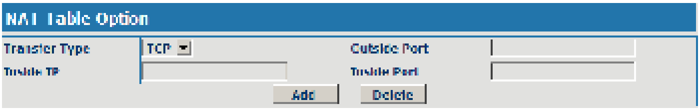
NAT configuration



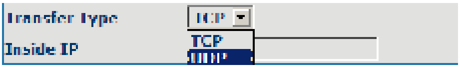
Configure NAT of the Application Layer Gateway(ALG);the protocol includes: IPSec, FTP and PPTP.



Display NAT table;



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



Select transfer type;

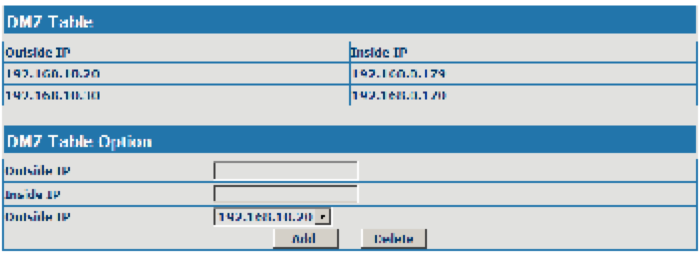
Configure Inside IP;

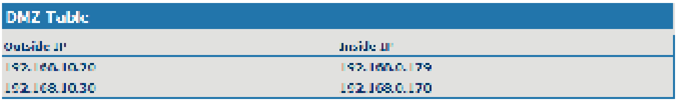
Configure Inside port;

Configure Outside port;

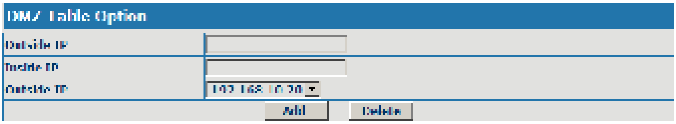
Add or delete NAT table;

DMZ configuration



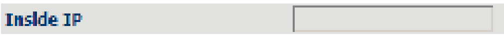


Display DMZ table;



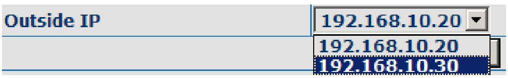
Configure the DMZ rule.

Configure the outside IP of the DMZ;



Configure the inside IP of the DMZ;

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

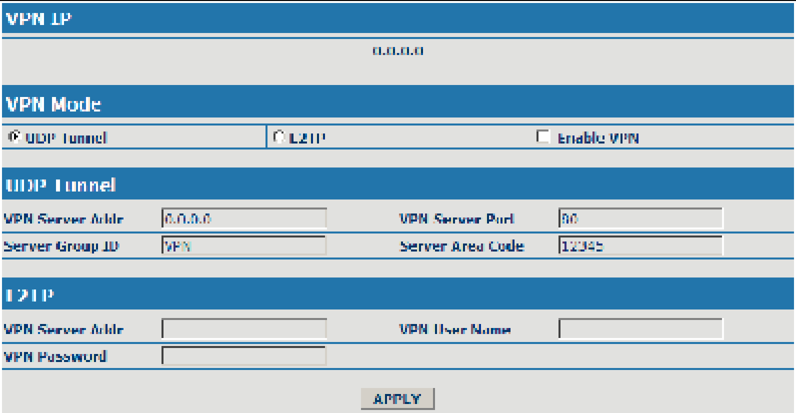


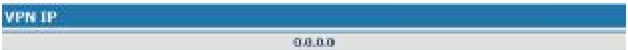
Select Outside IP;

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.

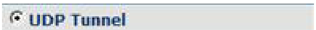




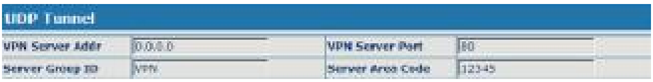
Display the VPN IP of the CDX-IPH303;

Select VPN mode, and configure enable/disable VPN;

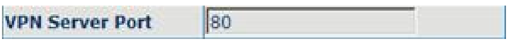
(1) Select UDP tunnel, and configure VPN:



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;

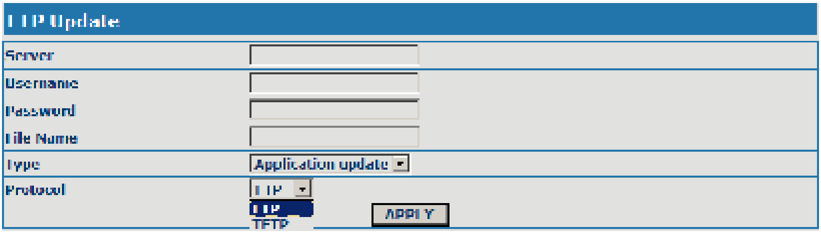
13 Upgrade on-line

13.1 Upload 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.

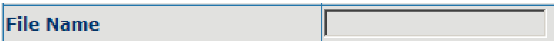


13.3 Configuration Explanation:

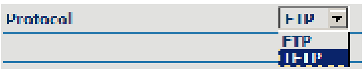
Configure upload or download FTP/ TFTP server IP address;

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

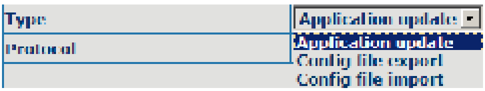
Configure upload or download of FTP server password;



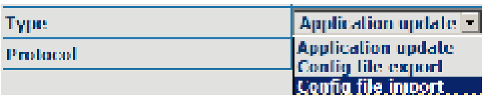
Configure upload or download system upgrade document or system layout file name. It should be noted that system file take .dlf as suffix, configuration files take .cfg as suffix;



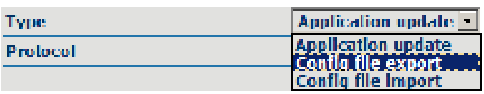
Select server type;



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

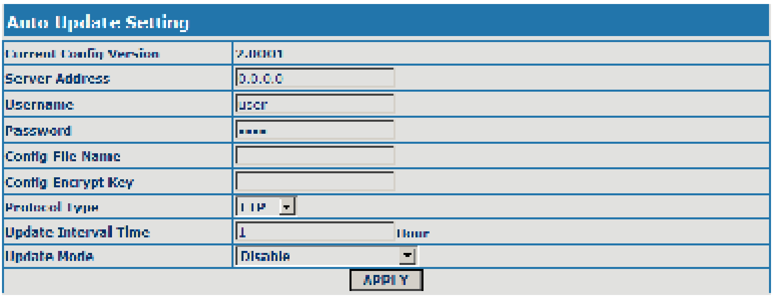
<e/d> e（encrypt）, d (decrypt)

< 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.



Display the current config version;

Configure unit as follow steps:

Input IP add. of desired FTP server.

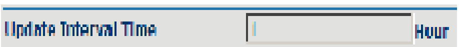
Input user name of desired FTP server.

Input user password of desired FTP server.

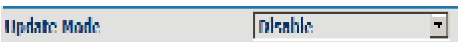
Input name configuration file. Software version must be different for each upgrade file.

If configuration file is encrypted, password needed.

Chose server type as either FTP or TFTP.



Set auto-upgrade interval duration.



Chose auto-upgrade type.

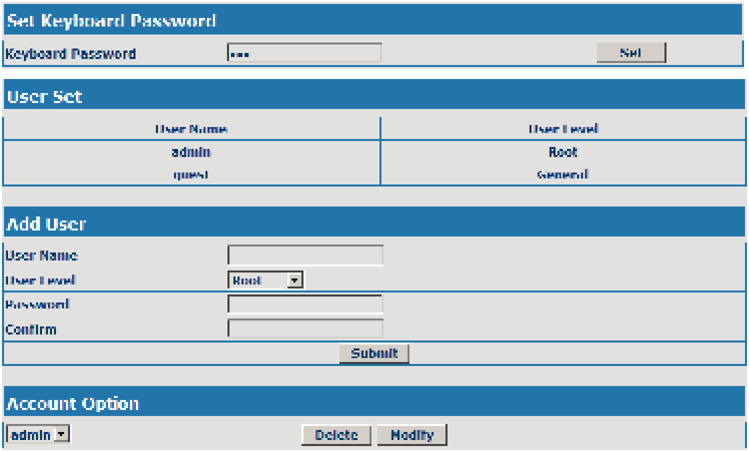
13.6 Configuration files WEB download

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.

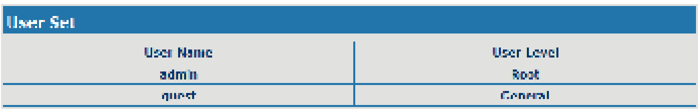
14 System management

14.1 Account management

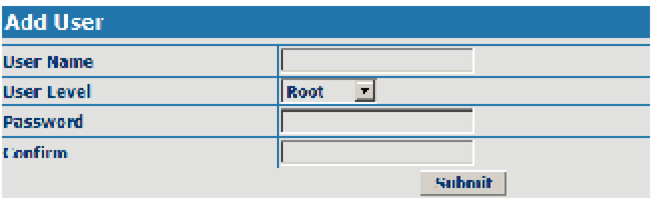
On this page, user can set the keyboard password, the same time, user can add and delete users according to own needs and can modify user's authorities there have been.



14.2 Configuration 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

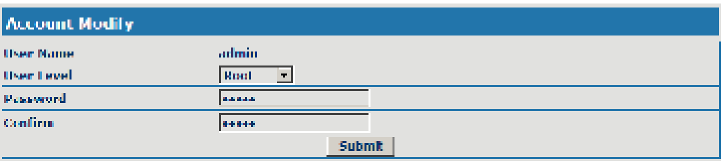
authorities to modify configuration,general possesses read-only authority;as corresponding password of the additive account;

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

Clicks submit to go into effect.

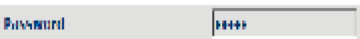
Select users that you want to delete in the drop-down menu, click Delete.

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:

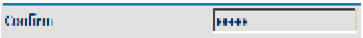


The modified username;

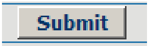
Modify user authorities;



Modify user password;



Make confirmation of the modified user password;



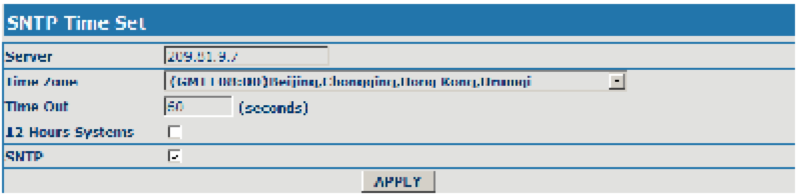
Submit the modification;

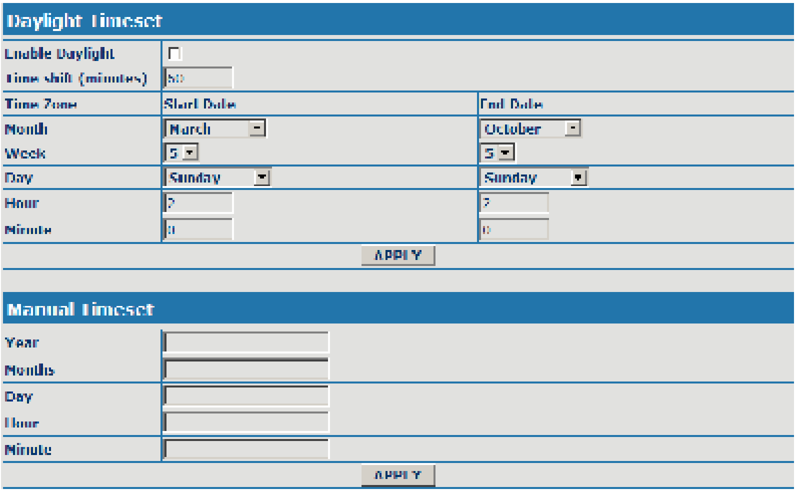
Owing to the phone's default account: accounts of the administrator level-admin and the ordinary level—guest are all weak account and weak password, the username and password will be easily to be guessed on public network, so the user had better modify the administrator and ordinary user.

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.

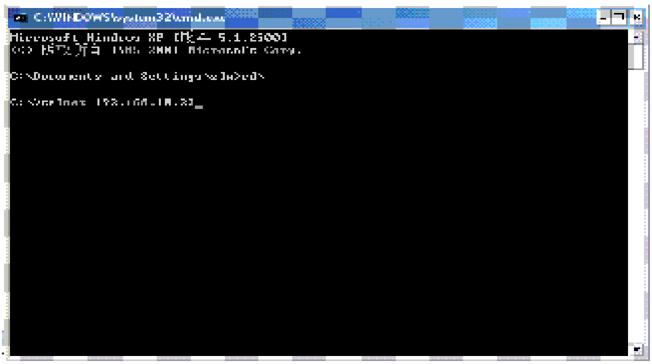




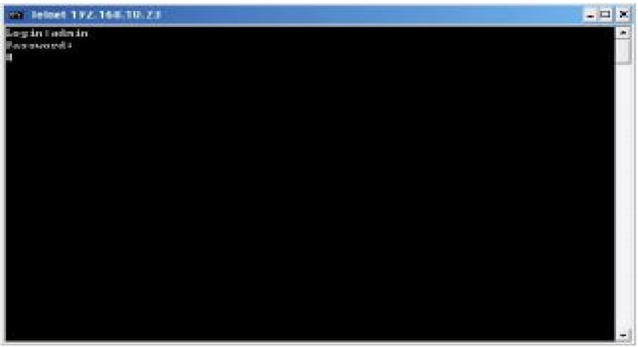
Configure the desired time zone.

15 Configuration via Telnet

In DOS window, input telnet 192.168.10.23, enter:



Then input USER NAME：admin PASWORD：admin



15.1 Basic 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

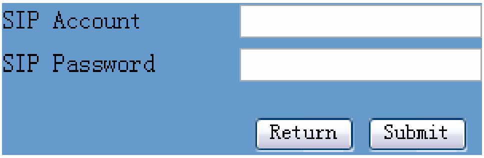
trace ---trancert x.x.x.x

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:

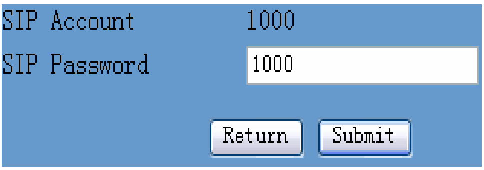


Configure additive passwords

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.