

IP Phone User Manual



This Manual provides basic information on how to install and connect IPH30 IP Phone to the network. It also includes features and functions of IPH30 IP phone components, and how to use them.

Contents

1 SAFETY INSTRUCTIONS	5
2 BEFORE GETTING STARTED	7
3 PACKAGE CONTENTS.....	7
4 SPEC AND FEATURES.....	8
4.1 Hardware Spec	8
4.2 DSP Spec	8
4.3 Software feature.....	8
4.5 Networking Standards.....	9
4.6 Others	10
4.7 Physical & Environmental	10
5 INSTALLATION	10
6 GENERAL APPEARANCE.....	12
6.1 LCD Display	12
6.2 Key function Definitions	14
6.3 Indicator Instruction.....	15
7 PHONE OPERATIONS	16
7.1 Point to Point Call	16
7.2 Two-SIP account and line	16
7.3 Make a call	17
7.4 Phone book number store / edit / delete	18
7.5 Call list check / delete	18
7.6 Hold	18
7.7 Mute.....	18
7.8 Volume adjustment in conversation	19
7.9 Rejected call	19
7.10 Black(White) list setting / edit / delete.....	19
7.11 Call transfer	19
7.12 3-party conference	19
7.13 SMS Function	19
7.13.1 Create/New . send message	20
7.13.2 Read, delete and reply message.....	20
8 CONFIGURATION WITH KEYPAD AND LCD DISPLAY	21
8.1 Main menu list operation.....	21

8.2 Phone book operation.....	22
8.3 Call list check	22
9. WEB INTERFACE CONFIGURATION	23
9.1 Logon Web.....	23
9.2 Current State	24
9.3 Network Configuration	25
10 VOIP CONFIGURATION	29
10.1 IAX2 Configuration.....	29
10.2 SIP configuration.....	30
10.3 STUN configuration.....	35
10.4 Dial Peer configuration	36
11 PHONE CONFIGURATION	40
11.1 DSP configuration	40
11.2 Call service configuration.....	41
11.3 Phone book configuration	43
11.4 Save and Clear Configuration.....	44
12 SECURITY CONFIGURATION	44
12.1 MMI Filter configuration	44
12.2 Firewall configuration.....	45
12.3 NAT and DMZ configuration.....	47
12.4 VPN configuration	49
13 UPGRADE ON-LINE	51
13.1 Upload WEB page	51
13.2 FTP/TFTP download.....	51
13.3 Configuration Explanation:.....	52
13.4 Configure file encryption	53
13.5 Auto-update.....	53
13.6 Configuration files WEB download	54
14 SYSTEM MANAGEMENT.....	55
14.1 Account management	55
14.2 Configuration Explanation:.....	55
14.3 Time zone configure.....	57
15 CONFIGURATION VIA TELNET	58
15.1 Basic Command.....	59

15.2 Command structure 59

15.3 Structure of Configuration terminal 60

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

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

4.2 DSP Spec

- I G.711A/u, G.723.1 (5.3k/6.3k) , G.729a/b,G.722,G726
- I G.168 echo cancel
- I Full duplex hand-free
- I VAD (Voice Activity Detection)
- I CNG (Comfortable Noise Generation)
- I AEC (Acoustic Echo Cancellation)
- I Adaptive voice jitter buffer
- I Codec negotiation supported for fixed and dynamic

4.3 Software feature

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

- I DTMF Mode: RFC2833, RELAY, SIP INFO
- I DNS name of SIP server
- I SIP signaling port setting
- I NAT traverse, STUN
- I NAT traverse, SIP Express router
- I Flexible Dial Map: Fix length; End with #; Dial with time out
- I 9 Kinds of ringer able select by number of Phone Box and 2 kinds of ringer user defined
- I Speakerphone
- I Headset
- I Dial Map Table
- I 5 Speed dial key.
- I Black list for reject authenticated call
- I Reject incoming call
- I Limit dialing out No. list
- I No Disturb
- I Caller ID display
- I Call forward, call transfer, call hold, call waiting
- I Call forward with unconditional, busy and no answer
- I 3 party conference
- I 50 entries each for dialed call, received call and missed call

4.5 Networking Standards

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

4.6 Others

- I Boot Monitor
- I Upgrade firmware through POST mode
- I Keyboard Configuration
- I HTTP Web-Based Configuration
- I FTP, TFTP upgrade firmware •
- I WEB upgrade firmware
- I 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

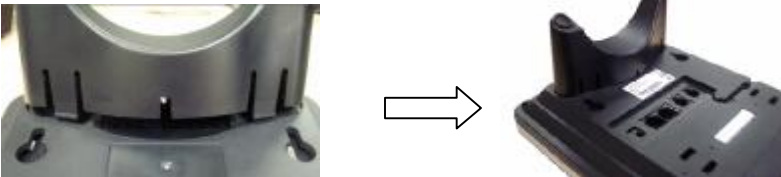
5 Installation

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.



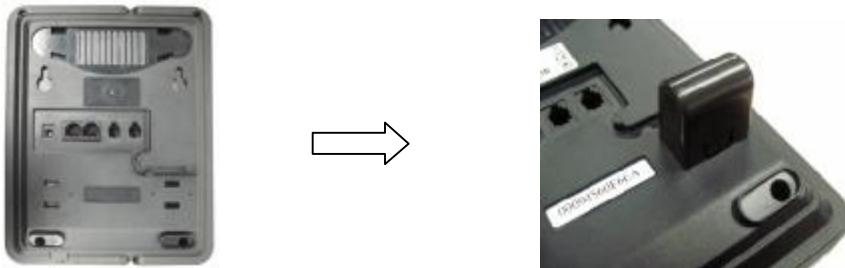
Power Jack	7.5V DC power connected port
LAN	10/100Mbps RJ-45 PC connected port
WAN	10/100Mbps RJ-45 Ethernet connected port
	RJ9 Headset Jack
	Handset Jack

Desktop installation:



Wall installation

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



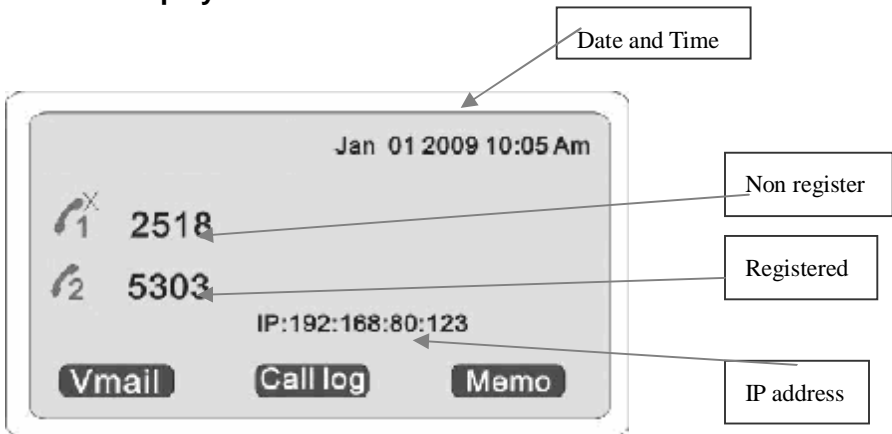
To install the phone on the wall, put two pothooks on the wall, hang the wall port of the phone back on the wall. The two supporting frames will balance the bottom of the phone and place it to the wall




















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: I On (steady)-Active call



		<ul style="list-style-type: none"> Slow blink-Held call 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: <ul style="list-style-type: none"> ON: appointed extension is being used 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, **[**], **[**], 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


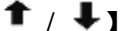
CDX-IPH303 support 2 independent SIP accounts. Each account can support independent SIP server, user name and NAT configuration. Line button **[**], **[**] are match to sole SIP account , when picking, press


a free line ( or ) 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  key / use a  or press  or  key (activated).

Step 2: Telephone dial tone will be sounded and  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  key / use a  or  or  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

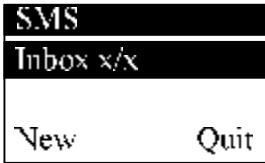
During conversation, as per the prompt of LCD Soft key, press Soft Key 2<Conf>, then dial another number plus # button, while line connected, press Soft Key 2<Conf> to make conference call;

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

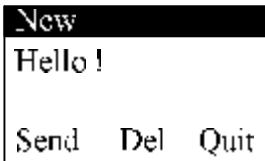
1. On standby state, press Soft Key1 <SMS>to enter SMS menu , see below photo



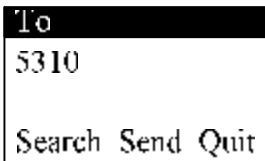
Inbox x/x: The first letter X means the missed message number, and last letter X means the total message number.

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

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

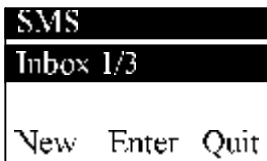


- 3.Send message: input the extension number and press soft key 2 to send message or select the number on notebook and press soft key 2 to send message.

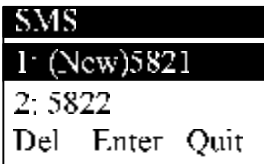


7.13.2 Read, delete and reply message

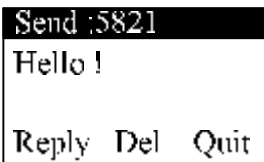
1. On standby , press Soft Key <SMS> to enter to SMS menu, as below picture:



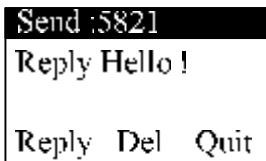
2. Press Soft Key 2<Enter>, you can delete or enter to the message as per the prompt of the Soft Key.



3. Press Soft Key 2 <Enter> to check the message, 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

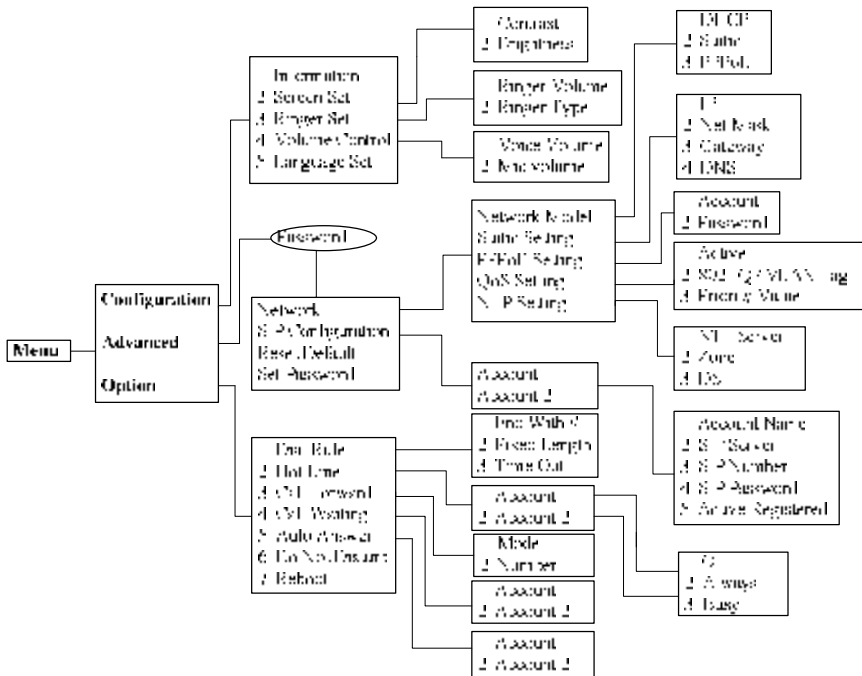
-Advanced

-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



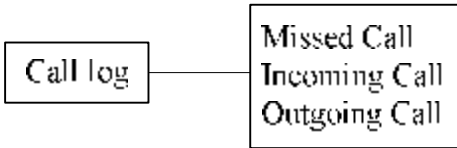
8.2 Phone book operation

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 1 to 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:

Username:
 Password:
 Logon

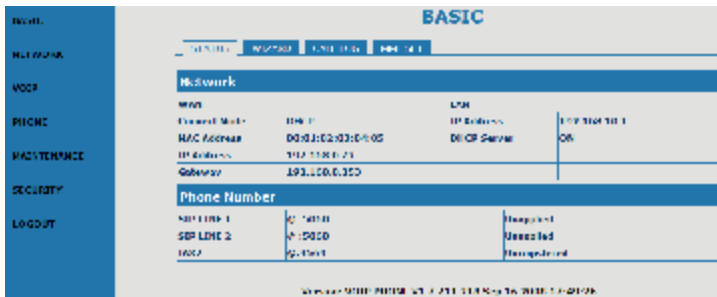
There are two level login as:

-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.

The screenshot displays the 'BASIC' configuration page. On the left is a blue sidebar with menu items: BASIC, NETWORK, VOIP, PHONE, NETWORK, SECURITY, and LOGOUT. The main content area is titled 'BASIC' and contains a 'Network' section. At the top of the Network section are three tabs: 'LAN', 'WAN', and 'PPPOE'. Below these is a table with two columns: 'WAN' and 'LAN'. The WAN table has rows for 'Connect Mode' (DHCP), 'MAC Address' (88:8E:0A:80:04:02), 'IP Address' (192.168.0.72), and 'Gateway' (192.168.0.1). The LAN table has rows for 'IP Address' (192.168.0.1) and 'DHCP Server' (ON). Below the Network section is a 'Phone Number' section with a table for SIP Line 1, 2, and FXS. The SIP Line 1 table has rows for 'SIP Line 1' (1000), 'SIP Line 2' (1000), and 'FXS' (8564). The status for each SIP line is 'Unregistered'. At the bottom of the page, it says 'Ver: VoIP PHONE V1.2.0 | 2.10 | Set | 6 2008 17:49:36'.

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 <input checked="" type="checkbox"/>	DHCP <input type="checkbox"/> PPPOE <input type="checkbox"/>
<input checked="" type="checkbox"/> 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
<input type="button" value="APPLY"/>	

PPPOE Server	ANY
Username	0001001
Password	*****
<input type="button" value="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 <input checked="" type="checkbox"/>	DHCP <input type="checkbox"/>	PPPOE <input type="checkbox"/>
--	-------------------------------	--------------------------------

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.

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.

LAN Set	
LAN IP	192.168.10.1
Netmask	255.255.255.0
DHCP Service	<input checked="" type="checkbox"/>
Net 1	<input checked="" type="checkbox"/>
Bridge mode	<input type="checkbox"/>
APPLY	

Configuration Explanation:

Bridge Mode	<input type="checkbox"/>
-------------	--------------------------

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;

LAN IP	192.168.10.1
--------	--------------

Configure LAN static IP;

Netmask	255.255.255.0
---------	---------------

Configure LAN subnet mask;

DHCP Service	<input checked="" type="checkbox"/>
--------------	-------------------------------------

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;

NAT	<input checked="" type="checkbox"/>
-----	-------------------------------------

Enable NAT.

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
<input type="button" value="Apply"/>	
<small>If modify HTTP or telnet port, you'd better set it more than 1024, then restart.</small>	

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.

10 VOIP Configuration

10.1 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	m21
Echo Test Number	1
Echo Test Text	echo
Refresh time	00 Seconds
Enable Register	<input type="checkbox"/>
Enable G.729	<input type="checkbox"/>
<input type="button" value="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;

Enable Register	<input type="checkbox"/>
------------------------	--------------------------

config the permission/prohibition registers the server;

Enable G.729	<input type="checkbox"/>
---------------------	--------------------------

Config whether supports G.729;

10.2 SIP configuration

Sip register

SIP Line Select			
SIP 1 ▾		Load	
Basic Setting			
Register Status	Unapplied	Display Name	<input type="text"/>
Server Name	<input type="text"/>	Proxy Server Address	<input type="text"/>
Server Address	<input type="text"/>	Proxy Server Port	<input type="text"/>
Server Port	5060	Proxy Username	<input type="text"/>
Account Name	<input type="text"/>	Proxy Password	<input type="text"/>
Password	<input type="text"/>	Domain Realm	<input type="text"/>
Phone Number	<input type="text"/>	Enable Register	<input type="checkbox"/>
APPLY			

Configuration Explanation:

SIP 1 ▾	Load
---------	------

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

Register Status	Unapplied
-----------------	-----------

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

Server Name	<input type="text"/>
-------------	----------------------

Configure the name of registration server;

Server Address	<input type="text"/>
----------------	----------------------

Configure SIP register server IP address or Domain Name;

Server Port	5060
-------------	------

Configure SIP register server signal port;

Account Name	<input type="text"/>
--------------	----------------------

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	<input type="text"/>
----------	----------------------

Configure password of SIP register account;

Phone Number	<input type="text"/>
--------------	----------------------

Configure the phone number;

Display Name	<input type="text"/>
--------------	----------------------

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	<input type="text" value="50"/> seconds	Forward Type	<input type="text" value="Off"/>
NAT Keep Alive Interval	<input type="text" value="30"/> seconds	Forward Phone Number	<input type="text"/>
User Agent	<input type="text" value="Voip Phone 1.0"/>	Server Type	<input type="text" value="COMMON"/>
Signal Key	<input type="text"/>	DTMF Mode	<input type="text" value="DTMF_RI_C2000"/>
Media Key	<input type="text"/>	RFC Protocol Edition	<input type="text" value="RFC1061"/>
Local Port	<input type="text" value="5060"/>	Transport Protocol	<input type="text" value="UDP"/>
Ring Type	<input type="text" value="Default"/>	RFC Privacy Edition	<input type="text" value="NONE"/>
Hot Line Number	<input type="text"/>	Subscribe Expire Time	<input type="text" value="300"/> seconds
Conference Number	<input type="text"/>	Enable Conference Number	<input type="checkbox"/>
Transfer Expire Time	<input type="text" value="0"/> seconds	Enable DNS SRV	<input type="checkbox"/>
Enable Subscribe	<input type="checkbox"/>	Click To Talk	<input type="checkbox"/>
Enable Keep Authentication	<input type="checkbox"/>	Signal Encode	<input type="checkbox"/>
NAT Keep Alive	<input type="checkbox"/>	Rip Encode	<input type="checkbox"/>
Enable Via rport	<input checked="" type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
Enable PRACK	<input type="checkbox"/>	Answer With Single Codes	<input type="checkbox"/>
Ignore Contact	<input type="checkbox"/>	Auto TCP	<input type="checkbox"/>
Enable URI Convert	<input checked="" type="checkbox"/>	Enable Strict Proxy	<input type="checkbox"/>
Dial Without Register	<input type="checkbox"/>	Enable GKUU	<input type="checkbox"/>
Ban Anonymous Call	<input type="checkbox"/>	Enable Displayname Quote	<input type="checkbox"/>
APPLY			

Register Expire Time	<input type="text" value="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	<input type="text" value="60"/>	seconds
--------------------------------	---------------------------------	---------

Configure the NAT keep alive interval;

User Agent	<input type="text" value="Voip Phone 1.0"/>
-------------------	---

Configure the User Agent;

Signal Key	<input type="text"/>
-------------------	----------------------

Configure the Signal Key;

Media Key	<input type="text"/>
------------------	----------------------

Configure the Media Key;

Local Port	<input type="text" value="5060"/>
-------------------	-----------------------------------

Configure the Local Port;

Ring Type	<input type="text" value="Default"/>
------------------	--------------------------------------

Select the Ring type;

Hot Line Number	<input type="text"/>
------------------------	----------------------

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	<input type="text"/>
--------------------------	----------------------

Configure the Conference Number;

Transfer Expire Time	<input type="text" value="0"/>	seconds
-----------------------------	--------------------------------	---------

Configure the Transfer Expire time

Enable Subscribe	<input type="checkbox"/>
-------------------------	--------------------------

Configure enable/disable Subscribe;

Enable Keep Authentication	<input type="checkbox"/>
-----------------------------------	--------------------------

Configure enable/disable Keep Authentication;

NAT Keep Alive	<input type="checkbox"/>
-----------------------	--------------------------

Configure enable/disable NAT Keep Alive

Enable Via rport	<input checked="" type="checkbox"/>
-------------------------	-------------------------------------

Configure enable/disable Via rport

Enable PRACK	<input type="checkbox"/>
---------------------	--------------------------

Configure enable/disable PRACK

Long Contact	<input type="checkbox"/>
--------------	--------------------------

Configure enable/disable Long Contact;

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

Configure enable/disable URI Convert;

Dial Without Register	<input type="checkbox"/>
-----------------------	--------------------------

Configure enable/disable Dial without register;

Ban Anonymous Call	<input type="checkbox"/>
--------------------	--------------------------

Configure enable/disable Ban Anonymous Call;

Forward Type	Off
Forward Phone Number	Off
Server Type	Busy

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
Transport Protocol	DTMF SIP_INFO

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 **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.

STUN Set	
STUN NAT Transverse	FALSE
STUN Server Addr	<input type="text"/>
STUN Server Port	<input type="text" value="3478"/>
STUN Effect Time	<input type="text" value="30"/> Seconds
Local SIP Port	<input type="text" value="5060"/>
<input type="button" value="APPLY"/>	

Set Sip Line Enable Stun	
SIP 1	<input type="button" value="Load"/>
Use Stun	<input type="checkbox"/>
<input type="button" value="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

Configure port of SIP STUN;

STUN Effect Time Seconds

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

Local SIP Port

Configure Local SIP port;

SIP 1

Select the Sip Line;

Use Stun

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.

Dial Peer Table						
Number	Destination	Port	Mode	Alias	Suffix	Dial Length
111	192.168.0.80	5060	SIP	no alias	no suffix	0
222	192.168.0.50	5060	SIP	no alias	no suffix	0
5T	0.0.0.0	5060	SIP	add:0755	no suffix	0

Add Dial Peer	
Phone Number	<input type="text"/>
Destination (optional)	<input type="text"/>
Port(optional)	<input type="text"/>
Alias(optional)	<input type="text"/>
Call Mode	SIP <input type="button" value="v"/>
Suffix(optional)	<input type="text"/>
Dial Length (optional)	<input type="text"/>
<input type="button" value="Submit"/>	

Dial Peer Option	
111 <input type="button" value="v"/>	<input type="button" value="Delete"/> <input type="button" value="Modify"/>

Configuration Explanation:

Dial Peer Table						
Number	Destination	Port	Mode	Alias	Suffix	Dial Length
11	192.168.0.80	5060	SIP	no alias	no suffix	0
222	192.168.0.50	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	<input type="text"/>
--------------	----------------------

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.

Destination (optional)	<input type="text"/>
------------------------	----------------------

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

Port(optional)

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

Alias(optional)

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;

Call Mode

Configure the calling mode:IAX2 and SIP;

Suffix(optional)

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;

Delete Length (optional)

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;

1T	Delete
----	--------

Delete selective number IP image;

1T	Delete	Modify
----	--------	--------

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:

Detail for Modify	
Phone Number	1T
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	1T
--------------	----

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

11 Phone Configuration

PHONE			
DSP Configuration			
First Codec	g711Ulaw6k	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	1 (1-9)	Ring Volume	7 (1-9)
G729 Payload Length	30ms	Signal Standard	China
G729 Timestamps	160/20ms	G723 Bit Rate	6.3kb/s
Default Ring Type	Type 1	VAD	<input type="checkbox"/>
APPLY			

11.1 DSP configuration

DSP Configuration			
First Codec	g711Ulaw6k	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	1 (1-9)	Ring Volume	7 (1-9)
G729 Payload Length	30ms	Signal Standard	China
G729 Timestamps	160/20ms	G723 Bit Rate	6.3kb/s
Default Ring Type	Type 1	VAD	<input type="checkbox"/>
APPLY			

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

First Codec

Configure the first codec;

Second Codec

Configure the second codec;

Third Codec

Configure the third codec;

Fourth Codec

Configure the fourth codec;

Fifth Codec

Configure the fifth codec;

Handdown Time 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 (1-9)

Configure input volume;

Output Volume	<input type="text" value="7"/>	(1-9)
---------------	--------------------------------	-------

Configure output volume;

Handfree Volume	<input type="text" value="4"/>	(1-9)
-----------------	--------------------------------	-------

Configure handfree volume;

Ring Volume	<input type="text" value="5"/>	(1-9)
-------------	--------------------------------	-------

Configure ring volume;

G729 Payload Length	<input type="text" value="20ms"/>
---------------------	-----------------------------------

Configure G729 payload length;

Signal Standard	<input type="text" value="China"/>
-----------------	------------------------------------

Configure signal standard;

G722 Timestamps	<input type="text" value="160/20ms"/>
-----------------	---------------------------------------

Configure G.722 timestamps;

G723 Bit Rate	<input type="text" value="6.3kb/s"/>
---------------	--------------------------------------

Configure G.723 bit rate;

Default Ring Type	<input type="text" value="Type 1"/>
-------------------	-------------------------------------

Configure default ring type;

VAD	<input type="checkbox"/>
-----	--------------------------

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.

Call Service Setting			
Hot Line	<input type="text"/>	No Answer Time	20 (seconds)
P2P IP Prefix	<input type="text"/>	Remote Record No	<input type="text"/>
Do Not Disturb	<input type="checkbox"/>	Ban Outgoing	<input type="checkbox"/>
Enable Call Transfer	<input checked="" type="checkbox"/>	Enable Call Waiting	<input checked="" type="checkbox"/>
Enable Three Way Call	<input checked="" type="checkbox"/>	Accept Any Call	<input checked="" type="checkbox"/>
Auto Answer	<input type="checkbox"/>	Use Record Server	<input type="checkbox"/>
<input type="button" value="APPLY"/>			

Black List	
<input type="text"/>	<input type="button" value="Add"/> Black List ▼ <input type="button" value="Delete"/>

Limit List	
<input type="text"/>	<input type="button" value="Add"/> Limit List ▼ <input type="button" value="Delete"/>

Configuration Explanation:

Hot Line	<input type="text"/>
----------	----------------------

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	<input type="text" value="20"/> (seconds)
----------------	---

Configure no answer time;

Do Not Disturb	<input type="checkbox"/>
----------------	--------------------------

Configure enable/disable Do Not Disturb;

Ban Outgoing	<input type="checkbox"/>
--------------	--------------------------

Configure enable/disable Ban outgoing;

Enable Call Transfer	<input checked="" type="checkbox"/>
----------------------	-------------------------------------

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

Enable Call Waiting	<input checked="" type="checkbox"/>
---------------------	-------------------------------------

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	<input checked="" type="checkbox"/>
-----------------------	-------------------------------------

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	<input checked="" type="checkbox"/>
-----------------	-------------------------------------

Configure enable/disable Accept Any Call;

Auto Answer	<input type="checkbox"/>
-------------	--------------------------

Configure enable/disable Auto Answer;

Black List	
<input type="text"/>	<input type="button" value="Add"/> <input type="button" value="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	
<input type="text"/>	<input type="button" value="Add"/> <input type="button" value="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.

11.3 Phone book configuration

Phonebook Table			
Index	Name	Number	Type
Add Phone Book			
Name	<input type="text"/>		
Number	<input type="text"/>		<input type="button" value="Add"/>
Ring Type	<input type="text" value="Default"/>		
Phone Book Option			
<input type="button" value="Add"/>	<input type="button" value="Delete"/>	<input type="button" value="Modify"/>	

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.

Save Configuration
Press the "Save" button to save the configuration files !
<input type="button" value="Save"/>

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

Clear Configuration
Press the "Clear" button to clear the configuration files !
<input type="button" value="Clear"/>

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.

MMI Filter Table		
Start IP	End IP	Option
<input type="text" value="192.168.0.100"/>	<input type="text" value="192.168.0.120"/>	<input type="button" value="Modify"/> <input type="button" value="Delete"/>
MMI Filter Table Set		
Start IP	<input type="text"/>	End IP <input type="text"/> <input type="button" value="Add"/>
MMI Filter Table Set		
<input type="checkbox"/> MMI Filter	<input type="button" value="APPLY"/>	

MMI Filter Table		
Start IP	End IP	Option
<input type="text" value="192.168.0.100"/>	<input type="text" value="192.168.0.120"/>	<input type="button" value="Modify"/> <input type="button" value="Delete"/>

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

MMI Filter Table Set		
Start IP	<input type="text"/>	End IP <input type="text"/> <input type="button" value="Add"/>

Add MMI filter table;

MMI Filter Table Set

<input type="checkbox"/> MMI Filter	<input type="button" value="APPLY"/>
-------------------------------------	--------------------------------------

Configure enable/disable MMI Filter.

12.2 Firewall configuration

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

Firewall Type

<input type="checkbox"/> In access Enable	<input type="checkbox"/> Out access Enable
<input type="button" value="APPLY"/>	

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	192.168.0.170	0.0.0.0	More than	1

Firewall Set

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

Rule Delete

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

Firewall Type

<input type="checkbox"/> In access enable	<input type="checkbox"/> Out access enable
<input type="button" value="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	192.168.0.170	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.

Firewall Set			
Input/Output	Input	Src Addr	
Deny/Permit	Deny	Des Addr	
Protocol Type	UDP	Src Mask	
Port Range	more than	Des Mask	

Configure the firewall.

Input/Output	Input
Deny/Permit	Input Output

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

Deny/Permit	Deny
Protocol Type	Deny Permit

Select the condition(deny or permit);

Protocol Type	UDP
Port Range	UDP TCP ICMP IP

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

Port Range	more than	1
Rule Delete	more than less than equal not equal	

Configure port range;

Src Addr	
----------	--

Configure source IP address;

Des Addr	<input type="text"/>
-----------------	----------------------

Configure destination IP address;

Src Mask	<input type="text"/>
-----------------	----------------------

Configure source Mask;

Des Mask	<input type="text"/>
-----------------	----------------------

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		
<input checked="" type="checkbox"/> IPsec ALG	<input checked="" type="checkbox"/> FTP ALG	<input checked="" type="checkbox"/> PPTP ALG
<input type="button" value="APPLY"/>		
NAT Table		
Inside IP	Inside TCP Port	Outside TCP Port
152.168.10.110	80	8080
Inside IP	Inside UDP Port	Outside UDP Port
152.168.10.120	80	8080
NAT Table Option		
Transfer type	TCP	Outside Port
Inside IP	<input type="text"/>	Inside Port
<input type="button" value="Add"/> <input type="button" value="Delete"/>		
Protocol Set		
<input checked="" type="checkbox"/> IPsec ALG	<input checked="" type="checkbox"/> FTP ALG	<input checked="" type="checkbox"/> PPTP ALG
<input type="button" value="APPLY"/>		

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
192.168.1.110	80	8080
Inside IP	Inside UDP Port	Outside UDP Port
192.168.1.120	80	8080

Display NAT table;

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

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

Transfer Type	TCP	
Inside IP	TCP	

Select transfer type;

Inside IP	
-----------	--

Configure Inside IP;

Inside Port	
-------------	--

Configure Inside port;

Outside Port	
--------------	--

Configure Outside port;

Add	Delete
-----	--------

Add or delete NAT table;

DMZ configuration

DMZ Table	
Outside IP	Inside IP
192.168.10.20	192.168.1.129
192.168.10.10	192.168.1.120

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	<input type="text"/>
Inside IP	<input type="text"/>
Outside IP	192.168.10.20 ▾
<input type="button" value="Add"/> <input type="button" value="Delete"/>	

Configure the DMZ rule.

Outside IP	<input type="text"/>
------------	----------------------

Configure the outside IP of the DMZ;

Inside IP	<input type="text"/>
-----------	----------------------

Configure the inside IP of the DMZ;

<input type="button" value="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
	192.168.10.30

Select Outside IP;

<input type="button" value="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.

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

VPN IP
0.0.0.0

Display the VPN IP of the CDX-IPH303;

VPN Mode	
<input checked="" type="radio"/> UDP Tunnel	<input type="radio"/> L2TP <input type="checkbox"/> Enable VPN

Select VPN mode, and configure enable/disable VPN;

(1) Select UDP tunnel, and configure VPN:

<input checked="" type="radio"/> UDP Tunnel
--

Select UDP tunnel mode;

UDP Tunnel	
VPN Server Addr	0.0.0.0
VPN Server Port	80
Server Group ID	VPN
Server Area Code	12345

VPN Server Addr	0.0.0.0
------------------------	---------

Configure VPN server address;

VPN Server Port	80
------------------------	----

Configure VPN server port;

Server Group ID	VPN
------------------------	-----

Configure VPN server group ID;

Server Area Code	12345
-------------------------	-------

Configure VPN server area code;

Enable VPN

Configure enable/disable VPN tunnel;
 (2) Select L2TP, and configure VPN:

 L2TP

Select L2TP mode;

L2TP			
VPN Server Addr	<input type="text"/>	VPN User Name	<input type="text"/>
VPN Password	<input type="text"/>		<input type="text"/>

VPN Server Addr

Configure VPN server address;

VPN User Name

Configure VPN User Name;

VPN Password

Configure VPN Password;

 Enable VPN

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.

Web Update	
Select file <input type="text"/>	<input type="button" value="Upload"/>

13.2 FTP/TFTP download

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

FTP Update	
Server	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
File Name	<input type="text"/>
Type	Application update ▾
Protocol	FTP ▾ FTP TFTP
<input type="button" value="APPLY"/>	

13.3 Configuration Explanation:

Server	<input type="text"/>
--------	----------------------

Configure upload or download FTP/ TFTP server IP address;

Username	<input type="text"/>
----------	----------------------

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

Password	<input type="text"/>
----------	----------------------

Configure upload or download of FTP server password;

File Name	<input type="text"/>
-----------	----------------------

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;

Protocol	FTP ▾ FTP TFTP
----------	----------------------

Select server type;

Type	Application update ▾
Protocol	Application update Config file export Config file import

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

Type	Application update ▾
Protocol	Application update Config file export Config file import

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;

Type	Application update ▾
Protocol	Application update Config file export Config file import

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	<input type="text" value="0.0.0.0"/>
Username	<input type="text" value="user"/>
Password	<input type="password" value="****"/>
Config File Name	<input type="text"/>
Config Encrypt Key	<input type="text"/>
Protocol Type	<input type="text" value="FTP"/>
Update Interval Time	<input type="text" value="1"/> Hour
Update Mode	<input type="text" value="Disable"/>
<input type="button" value="APPLY"/>	

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 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.

Set Keyboard Password

Keyboard Password

User Set

User Name	User Level
admin	Root
quest	General

Add User

User Name
 User Level
 Password
 Confirm

Account Option

14.2 Configuration Explanation:

User Name	User Level
admin	Root
quest	General

display of phone user account list;

Add User

User Name
 User Level
 Password
 Confirm

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

figure, of which:

A screenshot of a web form with a label 'User Name' and an empty text input field.

Add new accounts;

A screenshot of a web form with a label 'User Level' and a dropdown menu currently showing 'Root'.

account level; root possesses

A screenshot of a web form with a label 'Password' and an empty text input field.

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

A screenshot of a web form with a label 'Confirm' and an empty text input field.

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

A screenshot of a web form with a 'Submit' button.

Clicks submit to go into effect.

A screenshot of a web form section titled 'Account Option'. It features a dropdown menu with 'admin' selected, and two buttons labeled 'Delete' and 'Modify'.

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:

A screenshot of a web form section titled 'Account Modify'. It contains four input fields: 'User Name' (with 'admin' entered), 'User Level' (with 'Root' selected), 'Password' (with masked characters), and 'Confirm' (with masked characters). A 'Submit' button is located at the bottom right.

A screenshot of a web form with a label 'User Name' and the text 'admin' entered in the input field.

The modified username;

A screenshot of a web form with a label 'User Level' and a dropdown menu currently showing 'Root'.

Modify user authorities;

A screenshot of a web form with a label 'Password' and a text input field containing masked characters (asterisks).

Modify user password;

Confirm:

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.

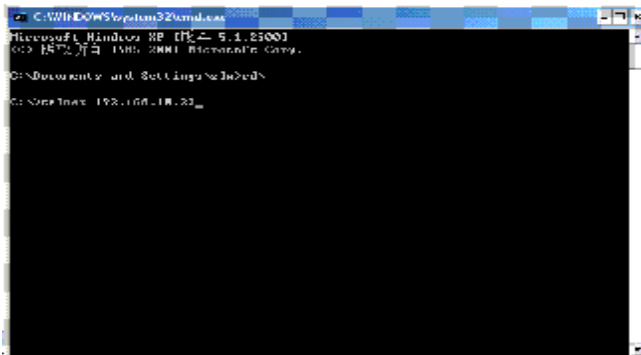
SNTP Time Set	
Server	<input type="text" value="192.168.1.1"/>
Time Zone	{GMT+08:00}Beijing,Chongqing,Hong Kong,Guangdong
Time Out	<input type="text" value="60"/> (seconds)
12 Hours Systems	<input type="checkbox"/>
SNTP	<input checked="" type="checkbox"/>
<input type="button" value="APPLY"/>	

Daylight Timeset		
Enable Daylight	<input type="checkbox"/>	
Time shift (minutes)	30	
Time Zone	Start Date	End Date
Month	March	October
Week	5	5
Day	Sunday	Sunday
Hour	5	7
Minute	0	0
<input type="button" value="APPLY"/>		
Manual Timeset		
Year		
Months		
Day		
Hour		
Minute		
<input type="button" value="APPLY"/>		
Time Zone	GMT+08:00 (Beijing, Chongqing, Hong Kong, Urumqi)	

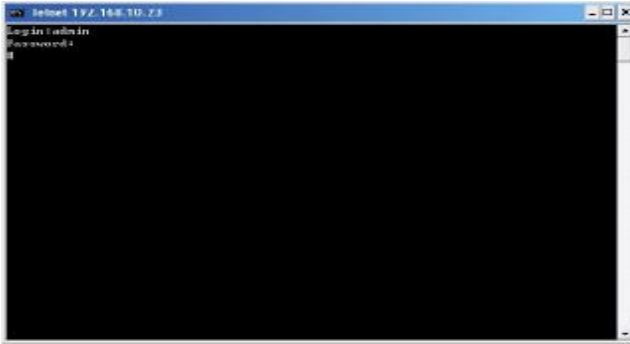
Configure the desired time zone.

15 Configuration via Telnet

In DOS window, input telnet 192.168.10.23, enter:



Then input USER NAME: admin PASSWORD: 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
---tracert
---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 -sradddr 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 Telnet 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.