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.

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

Contents

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

CDX-IPH303-VoIP User Manual V1.3

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
- WAN: 10/100M RJ45
- 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
- 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
- 1 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
Q	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.



	LCD illustration	Definitions
1	l°⇒	Call forwarding enabled
2	6-3	Call on hold
3	6-2	Connected call
4	= (Incoming call
5	r ≊⊷≊	Conference call active
6	 22	Conference call on hold
7	×	Conference call disabled
8	(ک	Outgoing call
9	×	Outgoing call not completed
10	« ب	Transferring a call
11	C1	Line 1 (idle)
12	C2	Line 2 (idle)
13	\mathcal{C}_1^{\times}	Line disabled
14	C	Handset in use
15	4	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	\bigcap	Soft key 1, 2, 3, Display functions as per the LCD		
		menu requested		
3	Menu	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
	8	key has red indicator.
8	m	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	\sim	Navigation key used as DOWM in menu or reduce
	·	handset / speakerphone volume
11	\sim	Navigation key used as LEFT in menu or reduce
	(+(handset/speaker volume
_	\vee	
12	0	Navigation key used as RIGHT in menu or increase
	0	handset/speaker volume
13	Q	Headset: activate/deactivate the headset function
14	(H.F. d)	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

		I Slow blink-Held call
		I Fast blink-Incoming call
2	(HF C)	On – Speaker in use
3	\bigcirc	On – headset in use
4	0	On – Phone is muted
5	0	On – Phone is held
6	M1~M5	BLF indicator status:
		I ON: appointed extension is being used
		I OFF: appointed extension is on line and on standby
		FLASH: appointed extension is off line and not registered
7	LCD	Backlight status:
	Backlight	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]**, **[**, **k**ey, **[**, **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 [1 /] 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 $[\Omega]$ 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 [COD] 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 [1 / 4] 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 [1] / 4] key to check

5. Standby dial-up call

On standby, dial the called number ,press the soft key 3<Dial>or pick up, press 【 H.F ◀ 】 or 【 Ω 】 【 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 [Ω] [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 (1 + 1) 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

 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:

SMS		
Inbox	1/3	
New	Enter	Quit

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



 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 (1) and (1) 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 [1] and [4] 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 $\begin{bmatrix} \uparrow \end{bmatrix}$ and $\begin{bmatrix} \downarrow \end{bmatrix}$ 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 **1** and **1** 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 <u>http://xxx.xxx.xxx.yyyy</u>, 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:	
Legon	
Password:	

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.

BASIC necen. STATE WARKE CALLOR HE ST HUI WORK Hkdwork vect 1.5.8 100 0.00 PHONE. Convert Marks 112 Robbergs 157 104 10 1 HAC Address 00:01:02:03:04:05 DICE Serve ha RACYTERANCE 19 Kildurys 192,108,073 Gaberray 193.100.0.350 SCOURTY Phone Number SPECIMEN e nom Inspired ьоврит A :5050 SEP LINE 2 Unners lied 9.4564 INCO. Humps In 191 Version WITE PICINE VER 2011 DURING TA 2000 12-2012E

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.

84500			BASIC	
линимски	STATUS W	DANG CALLEGE MMEN	2	
VOIP	Network			
	WAR		LAN	
PHONE	Councel Mode	DHC12	02 Address	192 168 10 1
	NAC Address	00:01:02:00:04:05	DITCP Server	ON
NATH INVESTIGATION	TP Address	192,100,0,72		
	Calmerey	197108-0758		
SECURITY	Phone Numb	er		
LOCOLT	STREET, STREET	0 THE	urapal	ind
	SUPLINE &	14:5050	Unavali	ed .
	TAX2	0:4569	linnegis	tered
		Version: VOIP PHONE	V1.7.211.210 Sep 16 200	0 17:49:76

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		
MAL Address	00:01:02:03:04:05		
WAN Setting			
static⊡	DHCP O	PPPOE C	
P Obtain DNS server au	tomatically		
Static IP Address	192.168.1.179		
Netmask	255,255,255,0		
Galeway	192.160.1.1		
DNS Domain			
Primary DNS	202.96.104.103	-	
Alter DNS	202,96,128,69	-	
	AP	PLY	
PPPOE Server	ANY		
Usemane	user120		
Password			
	AP	4.8	

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 C

DHCP 0

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

PTPOE ®

Static IP Address	192,168,1,179							
Netmask	255,255,255,0							
Galeway	192,160,1,1							
uns uonain	volp.com							
Primary DN5	202.96.134.133							
Atter DNS	202.96.128.68							
Static IP Address		192.16	58.1.17	9				
Configure stat	ic IP addres	s;						
Netmask		255.25	5.255.	0				
Configure sub	net mask;							
Gateway		192.1	68.1.1					
Configure IP a	address of th	ne pho	one;					
DNS Domain		voip.co	om					
Configure "D	NS domain'	" suff	ix; if	user	input	"domain"	and	and it

resolved, then the phone will add and resolve the "domain" after user has input;

202.96.134.133 Primary DNS 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	1

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

Username	user 123	
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	
LON UP	192.150.10.1
Netnesk	255.255.255.0
DRCP Service	12
NG 1	2
BRIGGE MEDE	
	APPLY

П

Configuration Explanation:

Bridge Mode

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 192.160.10.1

CDX-IPH303-VolP	User Manual V1.3
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Configure LAN static IP;

Netmask

Configure LAN subnet mask;

DHCP Service

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

V

255.255.255.0

5

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		
Teloet Port	21		
RTP Initial Port	10000		
RTP Port Quantity	200		
APPLY			
If modify ITTP or Telnet port, you'd better set it more than 1024, then restart.			

Configuration Explanation:

HTTP Port

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

Configure telnet port, the default is 23 port;

23

10000

RTP Initial Port

Enable RTP initial port configuration. It is dynamic allocation;

Client lightwore Address

RTP Port Quantity

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

DHCP Leased Table

Lensed TP 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	Unregiste	ened	
LAX2 Server Addr			
TAX2 Server Port	43.69		
Account Name			
Account Password			
Phone Number			
Local Port	4569		
Voice Mail Number	U .		
Voice Mail Text	mail		
Echo Test Number	1		
t cho test text	echo		
Refresh Time	60	Seconds	
Enable Register	Г		
Enable G.729			
		APPLY	

Configuration Explanation: Explanation:

Register Status

Unregistered

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

IAX2 Server Addr

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

4569

IAX2 Server Port

Config IAX2 server port;

Account Name

Config IAX2 account name;

Account Password

Config IAX2 account password;

U	•	
Phone Number		
Config IAX2 phone	number	5
Local Port		4569

CDX-IPH303-VolP	User	Manual V1.3	

Config equipment iax2 monitor port:

Voice Mail Number

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:

Г

mail

11

Voice Mail Text

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

```
Echo Test Number
```

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;

Seconds

echo Echo Test Text Config echo test text;

60 Config IAX2 refresh time, The unit of time for the second, suggested the user

in makes the choice 60 to 3600 between:

Enable Register

Refresh Time

config the permission/prohibition registers the server;

Enable G.729 Г

Config whether supports G.729;

10.2 SIP configuration

Sip register

SIP Line Select				
S1P 1 •		Load		
Basic Setting				
Register Status	Unapplied		Display Name	
Server Name			Proxy Server Address	
Server Address			Proxy Server Port	
Server Port	5060		Proxy Username	
Account Name			Proxy Password	
Password			Domain Realm	
Phone Number			Enable Register	Г
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

Configure the name of registration server;

Server Address

Configure SIP register server IP address or Domain Name;

Server Port

5060 Configure SIP register server signal port;

Account Name

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

Password

Configure password of SIP register account;

Phone Number

Configure the phone number:

Display Name

Configure display name;

Proxy Server Address

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

 Proxy Server Port

 Configure SIP proxy server signal port;

 Proxy Username

 Configure proxy server account;

 Proxy Password

 Configure proxy server password;

 Domain Realm

 Configure domain realm;

 Fmable Register

 Configure enable/disable register;

Advanced sip setting

Advanced SIP Setting							
Register Expire Time	50	seconds		Forw	ard Type	011	-
NAT Keep Alive Interval	60	seconds	1	Forw	ard Phone Number		
User Agent	Voip Phon	e 1.0] :	Serve	er Type	COMMO	N 🖃
Signal Key] 1	отмі	Mode	DIMI_R	(C2033 🖃
Media Key]	REC P	Protocol Edition	REC126	1 -
Local Port	5060		1.	Trans	port Protocol	UDP -	
Ring Type	Detault •	·		REC P	Privacy Edition	NONE	•
Hot Line Number] :	Subs	ribe Expire Time	300	seconds
Conterence Number				Fnahl	le Conference Number		
Transfer Expire Time	þ	seconds	I	Enabl	e DNS SRV		
Enable Subscribe				Click	To Talk		
Enable Keep Authentication	Г			Signa	il Encode		
NAT Keep Alive			1	RIp F	mesadee		
Enable Via rport	E		1	Enabl	e Session Timer		
Enable PRACK	Г			Альж	er With Single Codec		
Long Contact				Aurto '	TCP		
Enable URI Convert	M			Enabl	e Strict Proxy		
Dial Without Register	E Linable GRUU						
Ban Anonymous Call			1	Fnahl	e Displayname Quote		
			APP	IY			

Ron	ictor	Evniro	Time
ney	DUCI	LAPILC	1 mile

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 F	Port;		
Ring Type Default 💌		ult 💌	
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 Confer	ence N	lumber;	
Transfer Expire Time	0	seconds	
Configure the Transf	er Expi	re time	
Enable Subscribe			
Configure enable/dis	able S	ubscribe;	
Enable Keep Authenticatio	n 🗆		
Configure enable/dis	able Ke	eep Authenticatio	
NAT Keep Alive			
Configure enable/dis	able N	AT Keep Alive	
Enable Via rport			
Configure enable/dis	able Vi	a 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: П

 $\mathbf{\nabla}$

Ban Anonymous Call

Configure enable/disable Ban Anonymous Call;

Forward Type	Off 🔄
Forward Phone Number	Off Always
Server Type	Busy No Answer

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 DTMF_RFC2833
Transport Protocol	DTMF SIP INFO

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

RFC3261 -**RFC Protocol Edition**

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	
Select the Transport P	rotocol(UDP or TCP);
RFC Privacy Edition	RFC3323 💌

Select the RFC Privac	y Editior	n(none,RFC33	23 or RFC3325);
Subscribe Expire Time	300	seconds	
Configure Subscribe e	xpire tin	ne	
Enable Conference Number			
Configure enable/disa	ble Cont	ference Numbe	er;
Enable DNS SRV			
Configure enable/disa	ble DNS	service;	
Click To Talk			
Configure enable/disa	ble Click	k To Talk;	
Signal Encode			
Configure enable/disa	ble Sign	al Encode;	
Rtp Encode			
Configure enable/disa	ble RTP	Encode;	
Enable Session Timer			
Configure enable/disa	ble Sess	sion Timer;	
Answer With Single Codec			
Configure enable/disa	ble ansv	ver with single	codec;
Auto TCP			
Configure enable/disa	ble Auto	TCP;	
Enable Strict Proxy			
Configure enable/disa	ble Stric	t Proxy;	
Enable GRUU			
Configure enable/disa	ble GRL	JU;	
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	TALSE
STUN Server Addr	
STUN Server Port	2478
STUN Effect Time	30 Seconds
Local SIP Port	5060
	APPE T
Set Sip Line Enable Stu	n
STP 1	Load
Use Shin	Г
	APPLY
STUN NAT Transverse	FALSE
Display the applic	ation status of the STUN NAT;
STUN Server Addr	
Configure IP addre	ess of SIP STUN server;
STUN Server Port	3478
Configure port of	SIP STUN;
STUN Effect Time	30 Seconds
Interval time for S	TUN's detection on NAT type, the unit is second
Local SIP Port	5050
Configure Local S	IP port;
SIP 1	Load
Select the Sip Line	e;
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 Pee	r Table					
Number	Destination	Purt	Mode	olias	Soffix	Del Lemph
111	102.168.0.80	506	o ser	no elles	no suttix	0
222	192.160.0050	5116	II SIP	un alias	on soffix	u .
ज	0.0.0.0	506	o str	add:07.55	no suttix	D
Add Dial	Ренг					
Phone Non	duar					
Destination	(aptional)					
Port(option	of)					
Allas(optia	nal)					
Call Made		SEP 💌				
Suttiki,apti	onal)					
Heleta Lau	uth (optional)					
			10	limit		
Dial Pee	r Option					
111 💌			Delete	Mudify		

Configuration Explanation:

Diul Peer Topie						
Number	Destination	Purt	Mide	Alias	Seffic	Del Lergth
11	C2 108 0 80	5050	500	nn allas	an suffic	n
752	152 TELL INC.	8050	21.2	malias	on soffic	н
ST	0.0.0.0	5000	500	add:0755	ao suffic	D

Display of calling number IP image list;

Phone Number

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

Destination (optional)

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;

Allas(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 SIP 💌

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 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:

Delete

Dial treer Viocify		
Phone Namber	ភ	
call mode	sur 🖻	
Destinction (aptional)	or ac	
Port(gdocod)	5 ml	
Cliss(options.)	and:0775	
Suffix(aptional)	0.1075	
helete Lengt (optional)	0	
	Si	imi
Phone Number	51	
this is the modifi	ed number	r. read-only;
Call Mode	SI	
To modify call m	ode;	
Destination (optiona	I)	0.0.0.0
To modify destin	ation addr	ess; this is optional configuration item;
Port(optional)	ļ	5060
To modify destin	ation phon	e port;this is optional configuration item;
Alias(optional)	Ţa	add:0755
To modify alias;	this is option	onal configuration item;
Suffix(optional)	ſ	no suffix
To modify suffix;	this is opt	ional configuration item;
Delete Length (optio	nal) 🛛	
To modify replac	ing length	(if rep and del of alias have been configured);

Submit

Click submit to go into effect

11 Phone Configuration

BASEC	PHONE								
н-гиска	1000 CALL SHALLER COLLAR WAY CHARGE MADE HAVE CALL								
voar	DSP Conligurati								
	cine codec	ga mulawarik 📥	second socied	0774					
PHONE .	Third Oxlec	u729 -	Fourth Codec	071133x64k -					
·····	Fills Codes	Fikane -	Handdown Tine	200 ms					
DATE NAME	Input Volume	4 (1.9)	Colpet Webser	/ (1.9)					
	Handlines Volume	4 (1-9)	Rine Volume	2 11-91					
SECURITY	C7730 Desked Length	30	Shoul Classical	China a					
	CALC Investments	Thild dates in	E CALL MARK	A 199 19 19					
10.00	website sing type	1700.1	WHI	-					
			APPLY						

11.1 DSP configuration

DSP Configuration						
Lirst Codec	g/110law64k -		Seco	nd Codec	q723 ·	
Third Codec	9729 -		Four	th Codec	g71 LAlaw64k 💌	
Fifth Codec	None 📼		Hand	down Time	200 ms	
Input Volume	1 (1-9)		Outp	ut Volume	/ (1-9)	
Handfree Volume	4 (1-9)		Ring	Volume	5 (1-9)	
G729 Payload Length	20ms -		Sign	al Standard	China 💌	
G722 Timestamps	160/20ms -		G723	Bit Rate	6.3kb/s 💌	
Default Ring Type	Type 1		VAD			
		AP	PLY			

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

First Codec	g711Ulaw64k 💌
Configure the first	st codec;
Second Codec	g723 💌
Configure the se	cond codec;
Third Codec	g729 💌
Configure the thi	rd codec;
Fourth Codec	g711Alaw64k 💌
Configure the for	urth codec;
Fifth Codec	None
Configure the fift	h codec;
Handdown Time	200 ms

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

Input Volume 3 (1-9)

Configure input volume;						
Output Volume	7	(1-9)				
Configure output	volume	•				
Handfree Volume	4	(1-9)				
Configure handfre	ee volur	ne;				
Ring Volume	5	(1-9)				
Configure ring volume;						
G729 Payload Length	20ms 💌					
Configure G729 payload length;						
Signal Standard China 💌						
Configure signal	standar	d;				
G722 Timestamps	160/20	ns 💌				
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.

Call Service Setting			
Hot Line		No Answer Time	20 (seconds)
P2P IP Prefix		Remote Record No	
Do Not Disturb		Ran Outgoing	
Enable Call Transfer		Enable Call Waiting	R
Enable Three Way Call	•	Accept Any Call	M
Auto Answer	Г	Use Record Server	
Black List			
	Add	Black List	Delete
Limit List			
		Limit List	
	Add		Delete

Configuration Explanation:

Hot Line

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

 No Answer Time
 20
 (seconds)

 Configure no answer time;

 Do Not Disturb

Configure enable/disable Do Not Disturb;

Г

☑

Ban Outgoing

Configure enable/disable Ban outgoing;

Enable Call Transfer

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

Enable Call Waiting 🔽

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

Enable Three Way Call 🔽

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

second part, then the three way call concurrently; 2

Accept Any Call

Configure enable/disable Accept Any Call;

Auto Answer

Configure enable/disable Auto Answer:

Black List			
		Black List	
	Add		Delete

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

Limit List			
		Limit List	
	hhA		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 Ta	Phonebook Table					
Index	Name	Number	LADer			
Add Phone Bo	ok					
Name Number			Add			
Ring Type	Default -	1				
Phone Book O	ption					
		Delete Hodity				

On this page

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.

Clear Configuration	
	Press the "Clear" Inclose to Clear the configuration files (
	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 Hilter Table					
Shart IP		End 19		Option	
192,158,0,100	_	192,168.0,120		Hodity Delete	
MMT Filter Table	e Set				
Start IP		Find DP		Add	
MMT Filter Table	e Set				
MMI Lilter		APPLY			

MMI Filter Table					
Start IP	End IP	Option			
192.168.0.100	192.168.0.120	Modify Delete			

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

MMI Filter Table Set						
Start IP		End IP		Add		

Add MMI filter table;

MML	Filter	Table Set
	and these	

Configure enable/disable MMI Filter.

12.2 Firewall configuration

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

APPLY

Fire	Firewall Type								
	Thi access Enable				I	Cut. access Enable			
				APPLY					
Hire	Firewall Input Rule Table								
Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port	
1	Deny	יוסט	192.168.0.123	\$ 255,255,255,2	55 192,168,1	0.10 255.255.255.0	More than	1	
2	Permit	иры	192.160.0.120	255,255,255,2	55 192,168,1	0.10 255.255.255.0	More than	1	
Fire	wall Outpu	t Rule	Table						
Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port	
1	Deny	иры	192.160.10.10	255.255.255.2	55 192,168.0	10 255.255.255.2	55 More	1	
2	Permil	IIDP	192.160.10.20	255.255.255.2	55 192,168.0	.120 0.0.0.0	More than	1	
Hire	wall Set								
Input	/Output	Topar	•	Sec Adde	[
Deny,	/Permit	Deny	· •	Des Addr				Add	
Proto	col type	UDP	<u> </u>	Src Mask					
Port F	lange	more	than 🚬	Des Mask					
Pode	a Dadata								
Ruite	Delete				a a ta ta t				
Inpul	/Output	Tube	<u> </u>	Index to	Be Deleted			PIPE	
Firew	all Type								
	E in ar	ross exaltin	APPLY	E on	access enable				
Sel	Select firewall type;								

Firewall Topot Role Table								
Inde	Deny/Pernat	Protocol	Sire Addr	Sire Neok	Des Addr	Des Mask	Range	Port
1	Done	uur	192,168,0,123	255.255.255.255	192.160.10.10	255.255.255.0	More Iban	1
z	Permit	uur	192,168,0,120	255.255.255.255	192.160.10.10	255.255.255.0	More Iban	1

Display firewall input rule table;

Firewall Output Rule Table								
Index	Deny/Permit	Portorni	Src Midr	Sm Mask	Des Addr	Des Masir	Range	Port
1	Deny	UDP	192.140.10.10	255.255.255.255	192,166,0,10	255.255.255.255	Nore than	1
>	Permit	UDP	192,160,10,20	255.255.255.255	192.160.0.120	0.0.0.0	Nore than	1

Display firewall output rule table; in the table, Src address and Src Mask

confirm of the range the source address. For example : Src addr(192.168.10.10) and Src Mask(255.255.255.255) can confirm that the is 192.168.10.10; Dse addr(192.168. 0.120) and Des Mask(0.0.0.0) host Src can confirm anv host: 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/Oxtput	Input 💌	Sirc Addr		_
Deny/Perind	Decy -	Des Addr		
Protocol Type	UDP -	Sirc Mask	-	Add
Port Range	more than 1	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
	16

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

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

Configure port range;

Src Addr

Configure source IP address;

Des Addr

Configure destination IP address;

Src Mask

Configure source Mask;

Des Mask

Configure destination Mask;

12.3 NAT and DMZ configuration

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

NAT configuration

Protocol Set						
IPSec ALK				P PPIP ALC		
NAT Table						
tuside IP		Inside ICP Pur	4		Outside CCP Port	
192,165,10,110		00			ncon	
Inside IP		Inside UDP Port		Outside ODP Part		
192.163.10.120		80			8080	
NAT Table Optio	n					
transfer type	1122 -		Outsi	de Port		
tuside IP		Add	tusidi D	e Port clete		
Protocol Set						
R IPSec ALC		P FTP ALC			PPTP ALC	
			APPLY			

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

NAT Table						
Inside IP Inside ICP Part Uniside ICP Part						
192.168.10.110	50	10003				
	han an a					
Inside IP	Inside UDP Port	Outside UDP Port				
100.168.10.100	80	8080				

Display NAT table;

NAT Table Option						
Transfer Type Tuskle TP	TCP 💌	Outside Port				
	_	Add Delete	,			

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

transfer type	ICP 🔳	
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

DM7 Table	
Outside IP	Inside IP
192.160.10.20	192.100.0.179
142.168.10.10	192.168.0.170
DM7 Table Option	
Dutside IP	
Inside 19	
Outside IP	192.168.10.20 •
	ndd Delete

DMZ Table		
Outside IP	Inside IP	
192.160.10.20	192,160,0,179	
192,168,10,30	192,168.0.170	

Display DMZ table;

DMZ Table Option		
Outside IP		
Inside ID		
Outside IP	102 168 10 20 -	
	Add	Delete

Configure the DMZ rule.

Oulside TP

Configure the outside IP of the DMZ;

Inside IP

Configure the inside IP of the DMZ;

Add

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

Outside IP	192.168.10.20 💌
	192.168.10.20 192.168.10.30

Select Outside IP;

Delete

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

12.4 VPN configuration

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

VPN IP				
			0.0.0.0	
VPN Mode				
@ UDP Tunnel		0 L20P		E Enable VPN
UDP Lunnel				
VPN Server Addr	0.0.0.0		VPN Server Port	80
Server Group ID	VPN		Server Area Code	12345
121P				
VPN Server Addr			VPN User Name	
VPN Password				
			APPLY	
VPN IP				

Display the VPN IP of the CDX-IPH303;

VPN Mode				
R UDP Tanael	C L2TP	T Enable VPN		

Select VPN mode, and configure enable/disable VPN;

(1) Select UDP tunnel, and configure VPN:

• UDP Tunnel

Select UDP tunnel mode;

UDP Tunnel					
VPN Server Addr	0.0.0.0		VPN Server Port	80	
Server Group ID VTV		Server Area Code		12345	1
VPN Server	Addr	0.0.0.0			

Configure VPN server address;

And the second se	Parate
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	VPN User Name	
VPN Password		

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.



13.2 FTP/TFTP download

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

LTP Update	
Server	
Usernanie	
Password	
Lile Name	
Туре	Application update 💌
Protocol	<u> 117 - 117</u>

13.3 Configuration Explanation:

Configure upload or download FTP/ TFTP server IP address;

Username

Server

Configure username of the upload or download FTP server. If user select

TFTP mode, username and password are not required to be configured;

Password

Configure upload or download of FTP server password;

File Name

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	ETN 💌
	FTP
	101000

Select server type;

Туре	Application update 💌
Protocal	Application update
	Config file import

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

Туре	Application update 💌
Protocol	Application update Contin tile errort
	Coufig file inport

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 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	
Correct Coolig Version	2.8001
Server Address	0.0.0.0
Username	user
Password	
Config File Name	
Config Encrypt Key	
Protocol Type	• 411
Update Interval Time	1 Hour
Update Mode	Disable
	APPLY

Display the current config version;

Current Config Version

2.0001

Configure unit as follow steps:

Server Address	0.0.0.0

Input IP add. of desired FTP server.

User User

Input user name of desired FTP server.

Password

ord ----

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.

FTP 🔻

Protocol Type

Chose server type as either FTP or TFTP.

Update Interval Time I 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			Net
User Set			
User Nome		User Leve	4
admin		Root	
quest		General	
Add User			
User Name			
User Level	Root		
Password			
Confirm			
Submit			
Account Option			
admin 💌	Delete	Hodity	

14.2 Configuration Explanation:

User Set	
User Name	User Level
admin	Root
quest	Constal

display of phone user account list;

Add User		
User Name		
User Level	Root 💌	
Password		
Confirm		
	Submit	

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

figure, of which:

User Name

Add new accounts;

User Level Root 💌

account level; root possesses

Password

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

Confirm

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

Submit

Clicks submit to go into effect.

Account Option	
admis 💌	Delete Modity

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:

Account Modily		
User Name User Level	admin Root 💌	
Pessword	*****	_
Confirm	*****	
		Sub
User Name	admin	
The modified username:		
	asemanie,	
User Level	Root	•
Modify user a	authorities.	
modify door d	addiornioo,	
Password		

Confirm	
A DOLLAR DE LA DESERVICIÓN DE LA DESERVIC	

Make confirmation of the modified user password;

Submit

Submit the modification;

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

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

14.3 Time zone configure

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

SNTP Time Set			
Server	209.01.9.7		
time Zane	(GMT+00:00)Beijing,Chongqing,Dong Kong,Druoqi 💽		
Time Out	60 (seconds)		
12 Hours Systems			
SNTP	E		
APPLY			

Daylight Limeset			
Enable Daylight	Π		
time shift (minutes)	50		
Time Zone	Sharl Dale	End Date	
Month	March 🖃	October -	
Week	5 -	5 -	
Day	Sunday 🔳	Sunday 💽	
Hour	2	2	
Ninute	0	D	
APPLY			
Manual Limeset			
Year			
Months			
Day			
Hour			
Minute			
	VIAN A		
Thue Zone IGMT+08:0018eHina Chanaolag Hona Kena, Ununal 🔄			

Configure the desired time zone.

15 Configuration via Telnet

In DOS window, input telnet 192.168.10.23, enter:



Then input USER NAME: admin PA

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>#</config-accesslist>	ŧ
--	---

[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 <config-accesslist>#no entry - I/O input - index 1 Fx٠ Fastethernet-Lan configuration <config-fastethernet-lan># Path: bridgemode - <command>Enable/Disable bridge mode [no] - <command>Enable/Disable DHCP Server [no] dhcp-server - <command>Show DHCP current leased dhcpshow table [no] ip - <command>Set Ian IP - <command>Show LAN interface ipshow 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 <config-fastethernet-wan># Path: [no] dhcp - <command>Start DHCP client [no] gateway - <command>Set default gateway - <command>Set WAN IP [no] ip - <command>Enable/Disable PPPoE client [no] pppoe [no] gos - <command>Enable/Disable 802.1p QOS show - <command>Show WAN interface configuration <config-fastethernet-wan>#ip Ex: –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 <config-mmifilter>#no entry -start 202.112.20.1 Fx٠ - <command>Set IP filter table [no] entry modifyfilter - <command> modify mmifilter table

show	- <command/> Show IP filter table
[no] start-filter	 - <command/>Enable/Disable MMI IP filter
NAT	
Path: <config-< td=""><td>-nat>#</td></config-<>	-nat>#
[no] ftpalg	 - <command/>Set NAT FTP application level gateway
[no] ipsecalg	 - <command/>Set NAT lpSec 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:< td=""><td>>#tcp-entry –ip 192.168.1.5 –lanport 1720 –wanport 1000</td></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 	
signa	almode	 - <command/>Set signal mode 	
[no]	threetalk	- <command/> Enable/Disable threetalk	

QOS

Path:	<config-qos>#</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>#</config–user>	
[no] entry	- <command/> add a user with given level
password	- <command/> Change password
show	- Show all users

Other configuration exce	ept Config terminal		
TIME			
Path: <config-time>#</config-time>			
Manual time setting	manualsetyear xxxmonth xxxday xxxhour		
xxx -minute xxx -secon	d xxx		
Ex: <config-time>#man</config-time>	ulset -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</node>		

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 ftp --user xxx --file xxx

Other commands

---debug all xxx Setting module debug message level 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 ---reload re-start 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

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



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

SIP Account	1000		Accounts to be
SIP Password	1000		Modified, read-only;
	Return Submit	→	Passwords to be
modified;		_	

Click submit to

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

