
IP Phone

User manual

June 25, 2005

Dear customer:

First thank you to select and purchase our company's network telephone, In order to make it work at the optimum condition, please carefully read this user manual.

Attention:

1. The product specification and the information mentioned in this user manual supply the reference merely. so if has any renewal information or change, we will no longer inform you in advance ! We retains the product performance finally explanation copyright .

2. Although this network telephone already pass the most senior safety precaution test(We passed the CE and the FCC authentication), The safety is your responsibility too!

Therefore,

Please use the power adapter which we provides;

Don't open the product without any correct instruction and our permission;

Please don't remove the power adaptor during the configuration and updating the ip phone!

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Introduction

IP is abbreviation of Internet Protocol. An IP Phone is a telephone transporting voice using grouping data package of IP protocol. It can be used widely for audio communication in the broad band IP Network environment which accord with TCP/IP protocol, such as in the LAN or WAN of Enterprises and Institutions、 Telecom IP Phone services provider's network and broad band INTERNET user, who log on internet through LAN 、 Cable Modem or XDSL and so on.

The mostly significant features of IP Phone is transporting voice message over data communication network at an extremely low price with excellent sound quality. Using IP Phone, you will save dramatically on international calls and long distance calls.

IP Phone uses unique generalized outline and inner line modes. It functions much like an ordinary telephone switching between inner line and out line, so it supplies great conveniences to the users. When IP Phone is in generalized inner line mode, it can call another IP Phone worldwide for free. When IP Phone is in generalized out line mode, it can places calls to ordinary telephones worldwide at a dramatically low price, because IP Phone supports using prepaid card supplied by ISP such as Net2phone or eTalk. Moreover, it possesses excellent sound quality just like ordinary phone.

Suitable Users of IP Phone

IP Phone is the ideal choice for those who always place international or long distance calls. If two parties both use IP Phone, they can communicate with each other even free.

- Telecom Service Provider and Internet Phone Service Provider;
- Foreign capital or joint venture companies; offices, representative offices or agencies of foreign companies in China;
- Abroad hotel (Can be arranged at guest rooms or commercial central) ;
- Large enterprises, multinational enterprise (Used for international call and long distance call)
- Middle and small enterprise with import or export business, abroad travel agencies; study abroad or immigrant mid-agencies;
- Departments relating with international affairs, such as foreign trade department, association for friendship with foreign countries, turnvereins, athenaeums, foreign experts bureau and other departments involving foreign affairs
- Colleges and study institutions, such as dorm for abroad students, professors with close connection with foreign countries;
- Families or persons with close connection with foreign countries, such as foreigners in China and those who prepare the study abroad.

Spec and Install

Appearance

1. 10Model: Front Panel

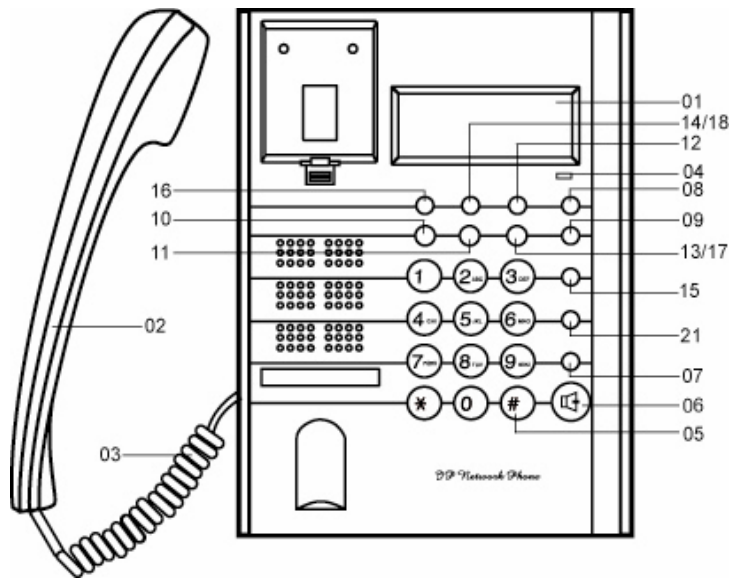


Fig. 1 10 Model Front Panel

2. 100 Model: Front Panel

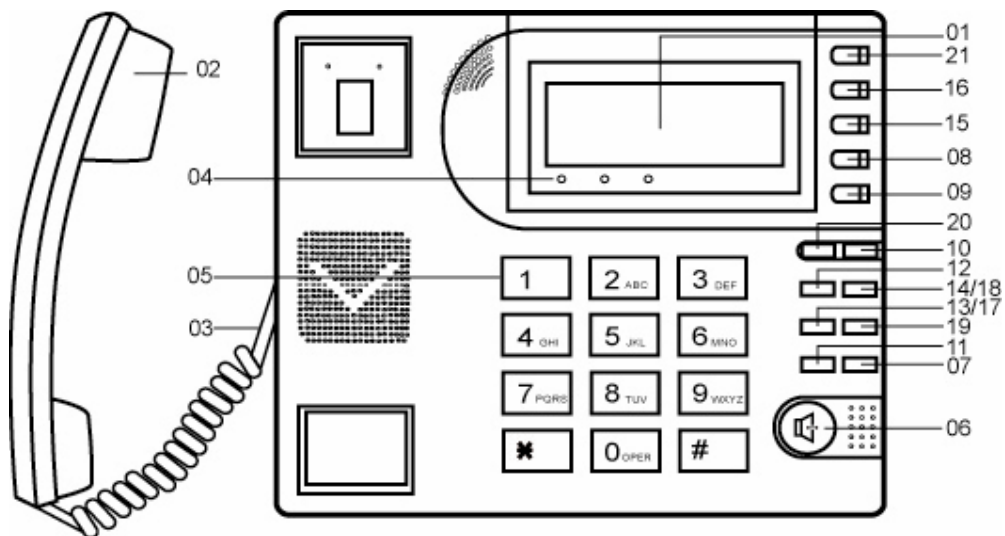


Fig. 2 100 Model Front Panel

3. 200 Model: Front Panel

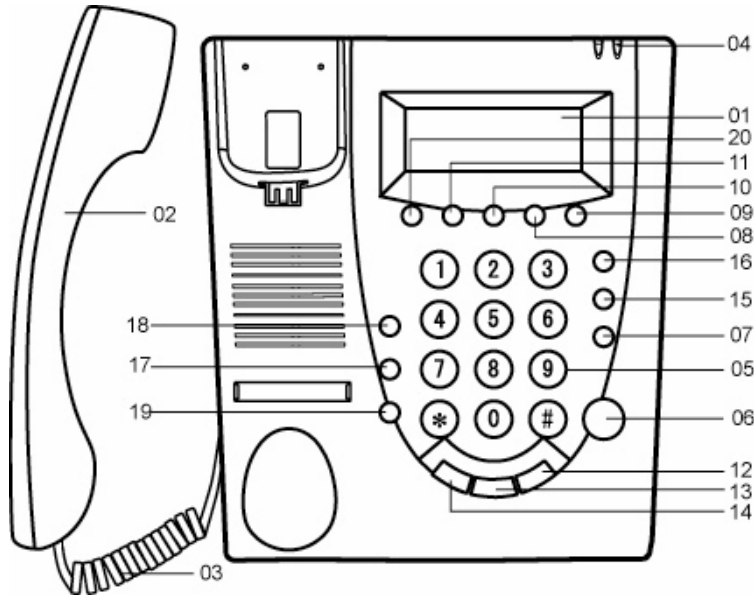


Fig. 3 200 Model Front Panel

4. 500Model: Front Panel

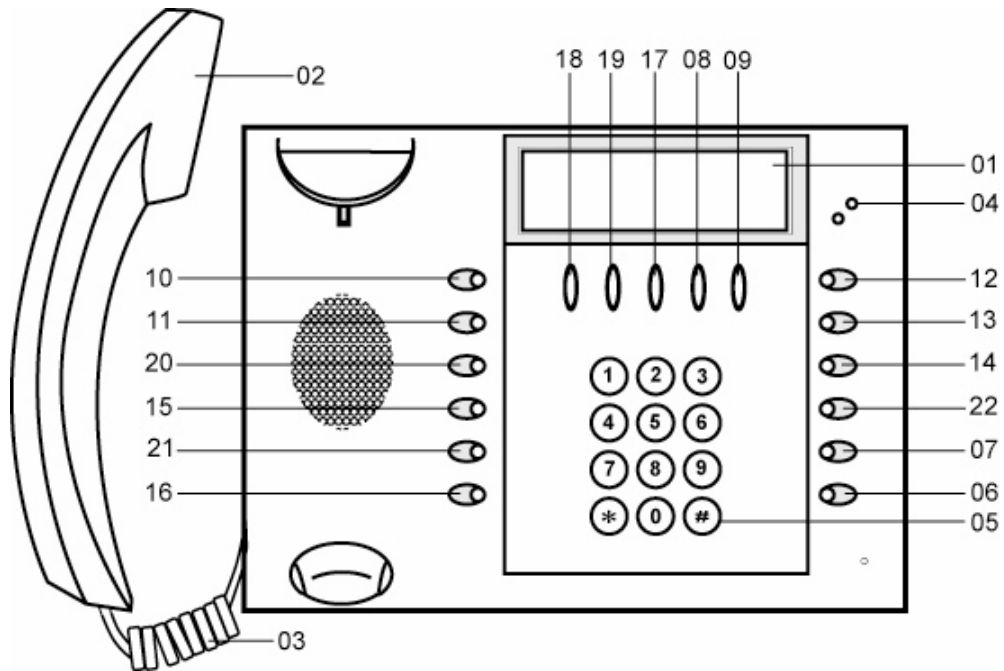


Fig. 4 500 Model Front Panel

- 1. LCD 2. Handset 3. Curve 4. Light 5. Number 6. Speaker 7. Redial 8. Vol/+
- 9. Vol/- 10. Local num 11. Local IP Redial 12 Missed 13. Answered 14. Dialed
- 15. Back Space 16. Phone Book 17. Transfer 18. Hold 19. Flash 20. Service IP

21. Speed dial 22. Call

Backside jack

1, SWITCH: Power On-off; **2, POWER OR DC9V:** Power Adaptor jack; **3, RJ45 :** LAN Interface; **4, PC:** PC Interface; **5,LINE :** PSTN Telephone Line Interface

Performance and Features

Features

Hardware

- Main chip—50MHz
- Data Memory—2MB SDRAM
- Program Memory—1 MB Flash memory
- Ethernet Port—10/100M Connectors
- AC/DC adapter—Input AC 110V or 220V OR 110--- 240V, Output 9V DC, 500mA

Software

- DHCP support for LAN or Cable modem
- PPPoE support for ADSL or Cable modem
- Set phone by HTTP web browser or Telnet
- Upgrade by FTP(TFTP optional)
- Support major G.7XX ,GSM610,iLBC audio codec
- VAD(Voice active detect)
- CNG (Comfort noise generation)
- Dynamic voice jitter buffer
- G..167/165 compliant 16ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- Support the Inband audio; the H245 String; the Q931 Keypad etc DTMF transmission method
- E.164 dial plan and customized dial rules
- 100 entries for speed dial
- 80 entries each for missed calls, answered calls and dialed calls
- Adjustable volume for both handset and speaker
- Voice prompt
- Hotline
- Support adjustable user password and super password

- Support PoE (only for 500 Model)

Standard and Protocol

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX,GSM610,iLBC audio codec
- H.323 V4, MGCP RFC2705, SIP RFC3261, IAX2 Protocol, Net2phone private protocol
- TCP/IP: Internet transfer and control protocol
- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- VAD/CNG save bandwidth
- DHCP: Dynamic Host Configuration Protocol
- PPPoE: PPP Protocol over Ethernet
- DNS: Domain Name Server
- Telnet: Internet's remote login protocol
- FTP: File Transfer protocol
- TFTP: File Transfer protocol (optional)
- HTTP: Hyper Text Transfer protocol
- Build in H.323 proxy

Operating requirements:

- Operation temperature: 0 to 50° C (32° to 122° F)
- Storage temperature: -30° to 65° C (-22° to 149° F)
- Humidity: 10 to 90% no dew

Electric requirements:

- Voltage: 9V DC
- Power: 9W (max.)
- Power adapter: Input AC 110V or 220V OR 110--- 240V, Output 9V DC, 500mA
- Network interface: RJ-45 Ethernet Connectors

Size :

10Model: 202 x 149x 75 mm (L x W x H)

100Model: 188 x 215 x 75 mm (L x W x H)

200 Model: 195 x 175 x 85 mm (L x W x H)

500 Model: 190 x 235 x 90 mm (L x W x H)

Installation

1. Connect Handset and Phone

Insert Handset cord into handset cord jack of the base.

2. Connect Phone and Power

Place the phone nearby of Power socket, Plug the power cord adapter into the **Power** or **DC9V** Jack. Then plug the other end of the power cord adapter into the appropriate power socket.

3. Connect the phone into the network

LAN users: Plug one end of the direct-connecting cable into **RJ45** jack, which is located in the back of phone, and connect the other end of cable to hub.

ADSL/Cable Modem users: Plug the RJ-45 Ethernet crossing-over cable into the **RJ45** Ethernet Jack. Plug the other end of the cable into an ADSL/Cable modem router port. Please see Fig. 5.

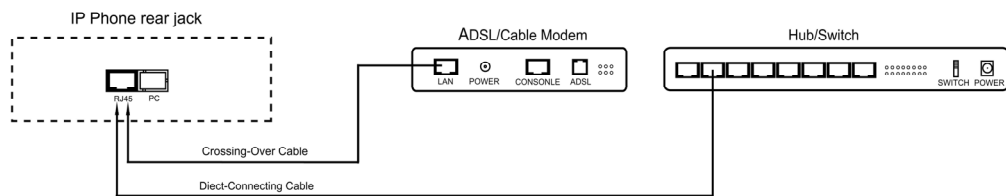


Fig. 5 IP Phone connected into network (for 1-RJ45)

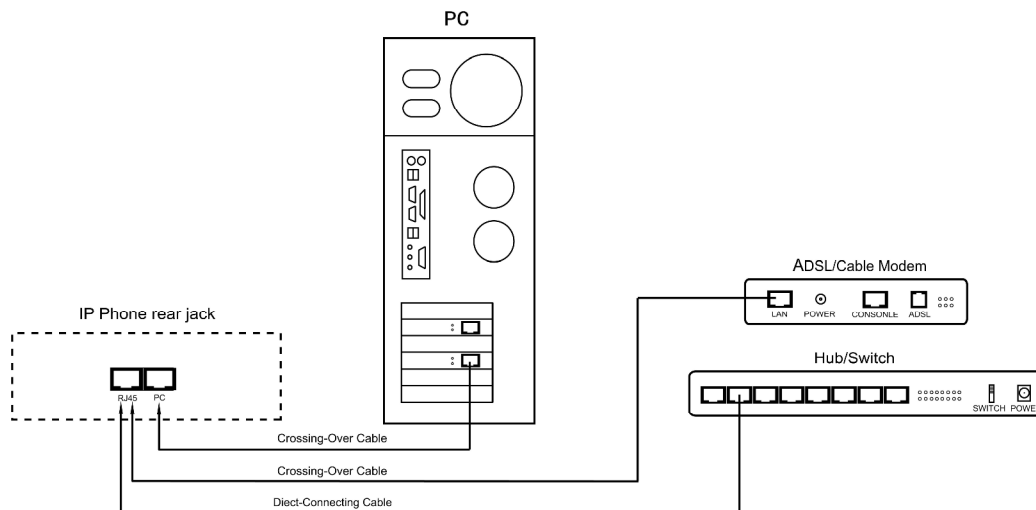


Fig. 6 IP Phone connected into network(for 2-RJ45)

4. Start Phone

Turn on the phone by pulling the switch to ON. Verify that yellow ,green and red lights are on together, and then red light is off; green light blinks or is off ; yellow light blinks or is on. which behalf the success of starting phone and phone enter into normal standby. (only for200 Model and 500 Model)

Setting and configuration

Configured by keypad

When IP Phone has been installed properly, you can enter the menu through the keyboard and LCD, and then set each parameter accordingly.

Function Keys Introduction:

Keys	Function
Service IP	With handset hung, press this key to get the GK IP address
Local IP	With handset hung, press this key to get phone IP address
Local Number	With handset hung, press this key to get phone number
Volume+	Increase the volumes of handset or speaker; turn over the record backward
Volume-	Decrease the volumes of handset or speaker; turn over the record forward.
Redial	While reviewing missed, received or dialed number, press this key to dial current number.
Spk	Press this key to have a call without lifting the handset
Dialed	With handset hung, press this key to review dialed number
Answered	With handset hung, press this key to review received number
Missed	With handset hung, press this key to review missed number

Operate with the keys

- **Enter the menu mode**

Use the keypad to enter the password of the phone (when debug is not set as 0[disable], default password is 1234; when debug is set as 0[disable], please use super password 19750407), and then

press”#”, till this Password : password: is displayed. Then enter the password again and press “Spk” to let the phone enter setting mode.

- **Select the submenu**

After entering the main menu, press “VOL+” or “VOL-” to page the menu up or page the menu down; when you want to set the submenu item to be displayed, press “Spk” to enter the submenu.

- **Enter the EDIT mode**

After entering the submenu, when you want to set the submenu item to be selected, press “Local IP” to enter EDIT mode.

- **Enter the settings**

Once the phone enters EDIT mode, enter the settings by numeric keypad respectively, then press “Spk” key to confirm.

- **Modify enter error**

If the errors come forth when you enter the settings, press “Backspace” key to delete it and enter the settings again.

- **Abort the settings**

If there is no need for the modified settings, press “Redial” key to return to the main menu, and

press “VOL+” or “VOL-” to page the menu up or page the menu down; When “exit settings” submenu item is selected, press “Spk” key to confirm, IP Phone will reboot and be used the current settings.

● **Save the settings**

When all parameters have modified, press “Redial” key to return to the main menu, and press “VOL+” or “VOL-” to page the menu up or page the menu down; When “save settings” submenu item is selected, press “Spk” key to confirm, IP Phone will reboot and use the already modified setting.

Introduce of the function of keypad in the keypad setting mode

Press key	Function
Spk /Hand free	Enter into submenu of the current menu ;Acknowledge to modification
Volume/+	Scroll menu forward
Volume/-	Scroll menu backward
Local IP	Enter into modification status
Redial	Cancel current setting ; restore to its father catalogue
Back Space	Backspace during the setting
Number keypad	Input updating content according to require. Please see appendix for character represented by each key

Set the main menu

● **Main Menu Structure**

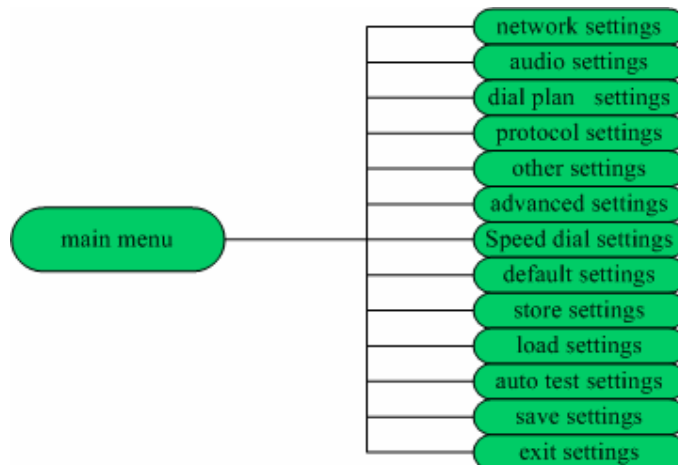


Fig. 7 Main Menu structure Illustration

Set the network submenu

● **Network Submenu Structure**

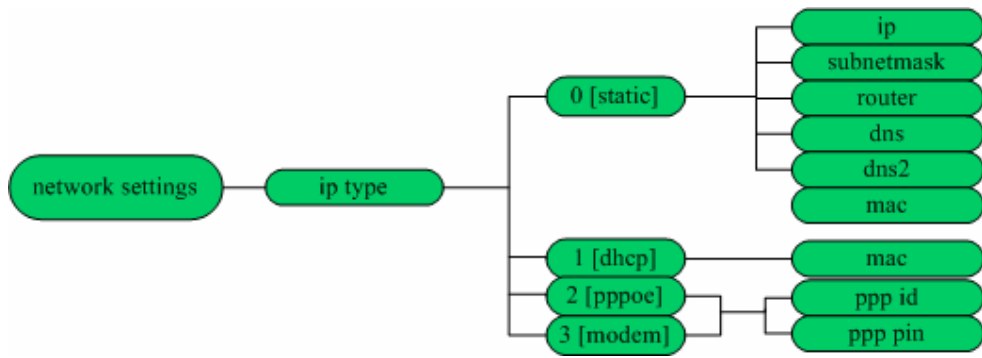


Fig. 8 Network submenu structure Illustration

Set the audio submenu

- **Audio Submenu Structure**

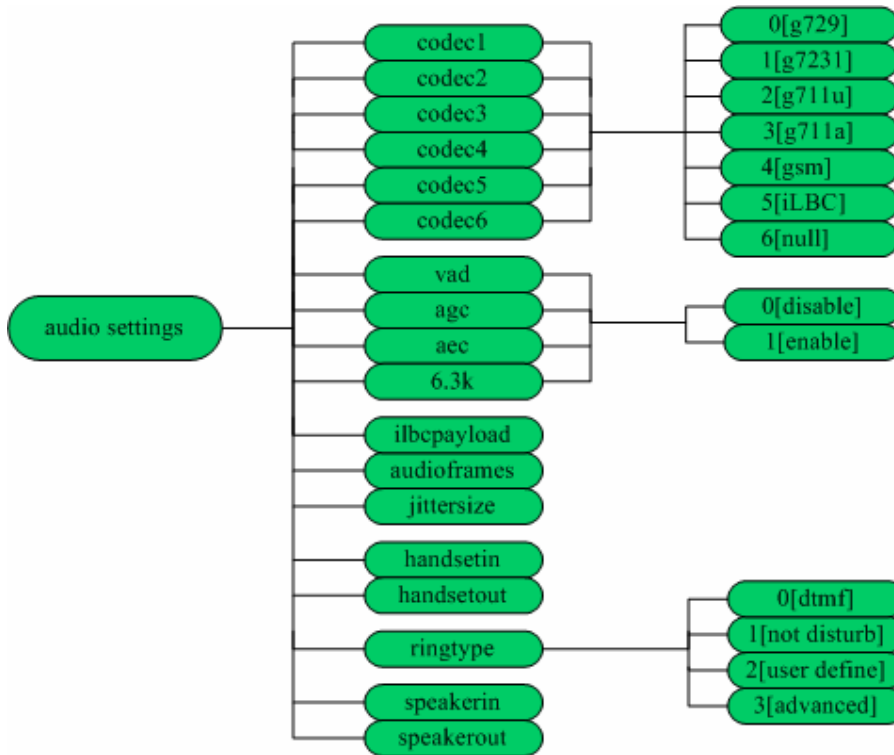


Fig. 9 Audio submenu structure Illustration

Set the dial plan submenu

- **Dial plan submenu Structure**

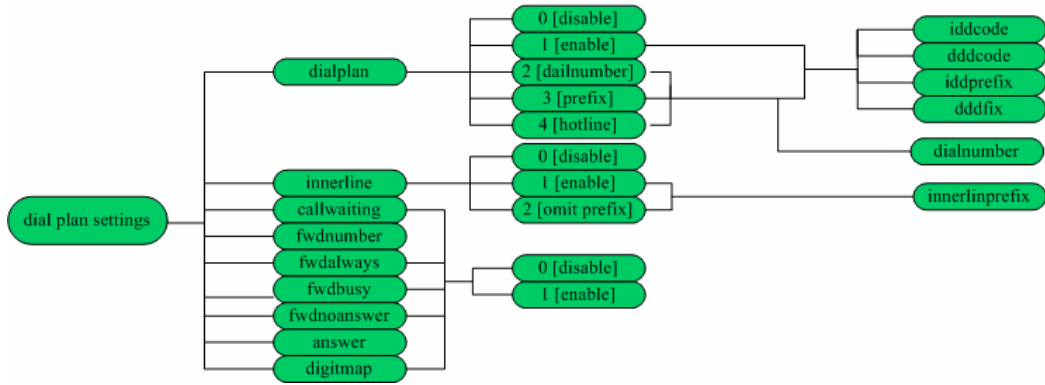


Fig. 10 Dial plan submenu structure Illustration

Set the protocol menu

- Protocol Submenu Structure

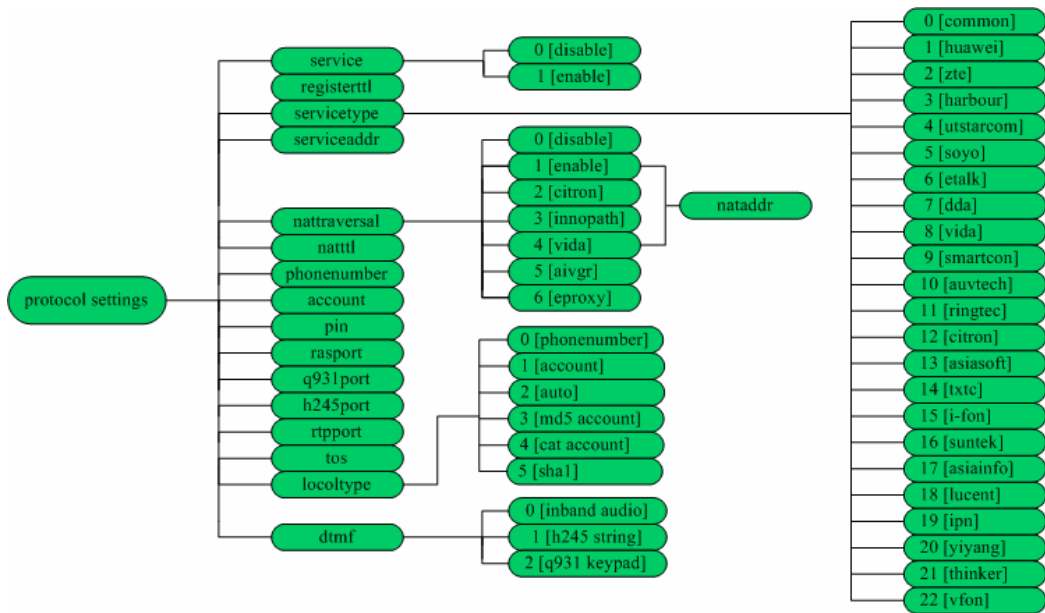


Fig.11-1 H323 Protocol

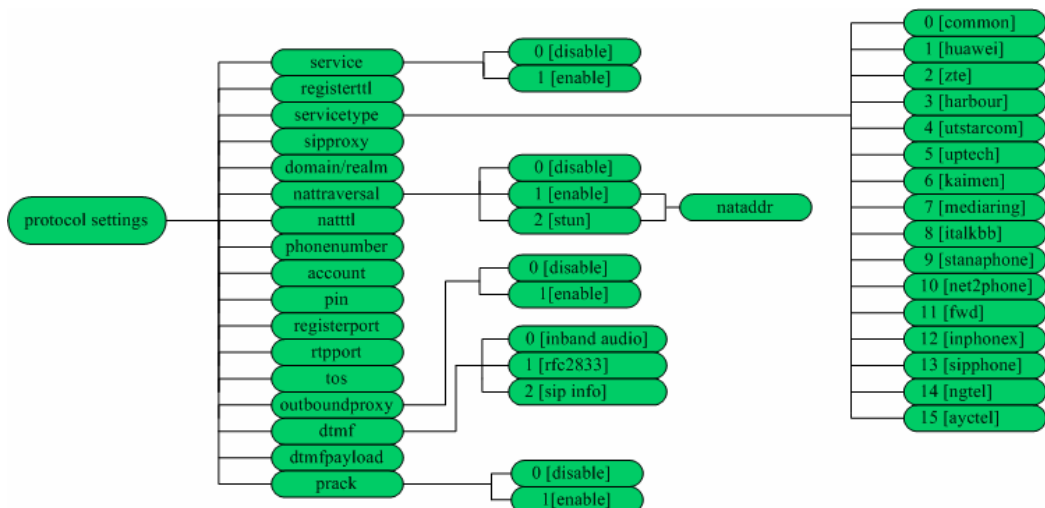


Fig.11-2 SIP Protocol

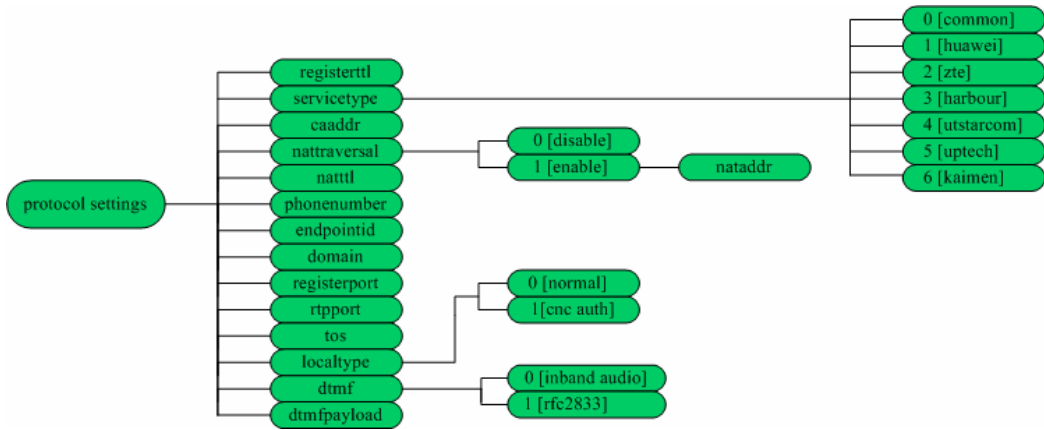


Fig.11-3 MGCP Protocol

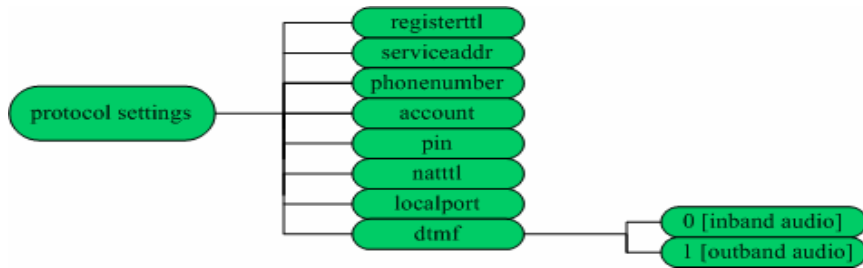


Fig.11-4 IAX2 Protocol

Fig. 11 Protocol submenu structure Illustration

Set the other menu

- Other Submenu Structure

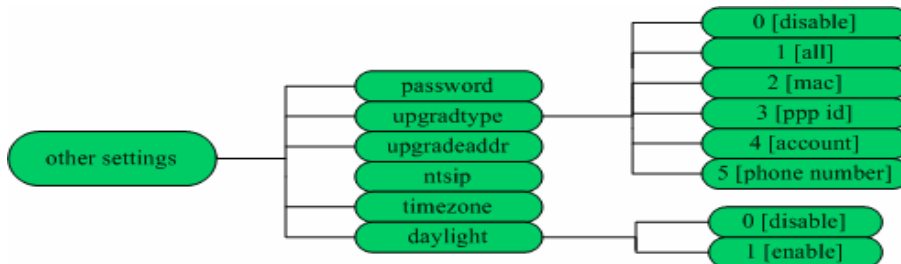


Fig. 12 other submenu structure Illustration

Set the Advanced menu

- Advanced Submenu Structure

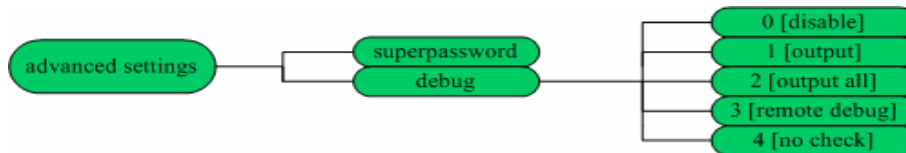



Fig. 13 advanced submenu structure Illustration

Appendix Table :

Keys	Press Once	Press Twice	Press Thrice	Press quartic	Press quintic
1	1	.	,	?/_	!//
2	2	A/a	B/b	C/c	[
3	3	D/d	E/e	F/f]
4	4	G/g	H/h	I/i	*
5	5	J/j	K/k	L/l	
6	6	M/m	N/n	O/o	#
7	7	P/p	Q/q	R/r	S/s
8	8	T/t	U/u	V/v	
9	9	W/w	X/x	Y/y	Z/z
*	.				
0	0	space	:/@	;-	\/&
#	Case change				

Configured by WEB

Double click  icon to open the IE browser. Input the IP address of the phone into address bar

() , and then input password of the phone into the following page.



Fig.14 Login page

Default password 1234 is ordinary password and super password is 19750407 or 12345678. With Debug set 0[disable], please input super password; while Debug is not set as 0[disable], please input ordinary password. Then click button. The Fig.15 configured page will popup (when you use H.323 protocol, the following page will appear, however if you use other protocol, the http setting page is the same, except protocol settings, we will say it more clearly in Fig.20).

Network Settings					
iptype	<input type="text" value="static"/>	ppp id	<input type="text"/>	ppp pin	<input type="text"/>
local ip	<input type="text" value="192.168.1.100"/>	subnet mask	<input type="text" value="255.255.255.0"/>	router ip	<input type="text" value="192.168.1.254"/>
dns	<input type="text" value="202.106.46.151"/>	dns2	<input type="text" value="202.96.128.68"/>	mac	<input type="text" value="00-09-45-0a-45-2e"/>
Audio Settings					
codec1	<input type="text" value="g729"/>	codec2	<input type="text" value="g7231"/>	codec3	<input type="text" value="g711u"/>
codec4	<input type="text" value="g711a"/>	codec5	<input type="text" value="null"/>	codec6	<input type="text" value="null"/>
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
audio frames	<input type="text" value="2"/>	g.723.1 high rate	<input checked="" type="checkbox"/>	ilbc payload	<input type="text" value="98"/>
jitter size	<input type="text" value="0"/>	handset in(0-15)	<input type="text" value="7"/>	handset out(0-31)	<input type="text" value="20"/>
ring type	<input type="text" value="dtmf"/>	speaker out(0-31)	<input type="text" value="20"/>	speaker in(0-15)	<input type="text" value="0"/>
Dial Plan Settings					
use dialplan	<input type="text" value="disable"/>	dial number	<input type="text"/>	ddd code	<input type="text" value="10"/>
idd code	<input type="text" value="86"/>	idd prefix	<input type="text" value="00"/>	ddd prefix	<input type="text" value="0"/>
inner line	<input type="text" value="disable"/>	inner line prefix	<input type="text" value="0"/>	call waiting	<input type="checkbox"/>
forward number	<input type="text" value="82378801"/>	fwd poweroff	<input type="checkbox"/>	fwd noanswer	<input type="checkbox"/>
fwd always	<input type="checkbox"/>	fwd busy	<input type="checkbox"/>	answer	<input type="text" value="30"/>
use digitmap	<input checked="" type="checkbox"/>				
H323 Protocol Settings					
use service	<input checked="" type="checkbox"/>	service type	<input type="text" value="common"/>	service addr	<input type="text" value="203.93.9.57"/>
nat traversal	<input type="text" value="disable"/>	nat addr	<input type="text"/>	nat ttl	<input type="text" value="30"/>
phone number	<input type="text" value="82378808"/>	account	<input type="text"/>	pin	<input type="text"/>
ras port	<input type="text" value="1720"/>	q931 port	<input type="text" value="1720"/>	h245 port	<input type="text" value="1722"/>
rtp port	<input type="text" value="1722"/>	tos	<input type="text" value="0"/>	register ttl	<input type="text" value="60"/>
local type	<input type="text" value="phonenumber"/>	dtmf	<input type="text" value="h245 string"/>		
super password	<input type="text" value="12345678"/>	debug	<input type="text" value="output"/>		
Other Settings					
password	<input type="text" value="1234"/>	upgrade type	<input type="text" value="disable"/>	upgrade addr	<input type="text"/>
sntp ip	<input type="text" value="255.255.255.255"/>	use daylight	<input type="checkbox"/>		
timezone	<input type="text" value="(GMT+08:00)Beijing,Hong Kong,Urumqi"/>				
<input type="button" value="Save Settings"/>		<input type="button" value="Address Book"/>		<input type="button" value="Upgrade Firmware"/>	

Fig. 15 Http Setting

When debug set as 0[disable], and if input ordinary password (default one is 1234), the following page will pop up after clicking . And only those parameters can be modified.

Network Settings					
iptype	static	ppp id		ppp pin	
local ip	192.168.1.100	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-0a-45-2e
Audio Settings					
codec1	g729	codec2	g711a	codec3	g711u
codec4	gsm	codec5	null	codec6	null
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
audio frames	2	g.723.1 high rate	<input checked="" type="checkbox"/>	ilbc payload	98
jitter size	0	handset in (0-15)	15	handset out (0-31)	20
dual mode	pstn first	dual mode prefix	30		
Dial Plan Settings					
use dialplan	enable	dial number	16900	ddd code	10
idd code	86	idd prefix	00	ddd prefix	0
inner line	enable	inner line prefix	0	call waiting	<input checked="" type="checkbox"/>
forward number	95963	fwd poweroff	<input type="checkbox"/>	fwd noanswer	<input checked="" type="checkbox"/>
fwd always	<input checked="" type="checkbox"/>	fwd busy	<input type="checkbox"/>	answer	30
use digitmap	<input type="checkbox"/>				
Other Settings					
password	1234	upgrade type	disable	upgrade addr	
sntp ip	255.255.255.255	use daylight	<input type="checkbox"/>		
timezone	(GMT+08:00)Beijing, Hong Kong, Urumqi				
Save Settings		Address Book		Upgrade Firmware	

Fig .16 Setting Page using ordinary pin with Debug set as 0 [disable]

Network Setting

Network Settings					
iptype	static	ppp id		ppp pin	
local ip	192.168.1.100	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-0d-ea-00-00-03

Fig. 17 Network Setting

- **iptype**: Set how IP Phone gets relevant network parameters by selecting corresponding item from drop down list.
 - n **static ip**: Select this item to authorize users set IP address, subnet mask and router IP address of IP Phone manually.
 - n **dhcp**: Select this item to enable DHCP mode. With this system, your LAN or router automatically assigns all the required network parameters to any device connected to it when the device log on. IP Phone is shipped from factory with DHCP on. So, if you're LAN or router is configured to use DHCP addressing, the IP Phone's LAN parameters will automatically be configured as soon as it is connected to the LAN or router and powered up.
 - n **pppoe** : Those ADSL and Cable Modem users please select this item for it is a protocol especially designed for them. With this system, ADSL ISP automatically assigns all the required IP parameters to any device connected to it when the device log on.
 - n **modem** : If the IP Phone used with modem, please select this item to get relevant network parameters automatically. Then please fill id and pin into ppp id and pppin fields.

- **ppp id:** With **pppoe** or **modem** selected in **iptype** drop down list, please enter the user name here.
- **ppp pin:** With **pppoe** or **modem** selected in **iptype** drop down list, please enter the password here.
- **local ip:** With **static ip** selected in **iptype** drop down list, please enter IP address of IP Phone here.
- **subnet mask:** With **static ip** selected in **iptype** drop down list, please enter subnet mask of IP Phone here.
- **router ip:** With **static ip** selected in **iptype** drop down list, please enter router IP address of IP Phone here.
- **dns:** With **static ip** selected in **iptype** drop down list, please enter IP address of DNS server here.
- **dns 2:** With **static ip** selected in **iptype** drop down list, please enter IP address of backup DNS server here.
- **mac:** MAC address is the physical address supplied by the Ethernet NIC. IP Phone is shipped from the factory with a unique algorithm MAC address printed on the back of the base.

Audio settings

Audio Settings					
codec1	<input type="text" value="g729"/>	codec2	<input type="text" value="g7231"/>	codec3	<input type="text" value="g711u"/>
codec4	<input type="text" value="g711a"/>	codec5	<input type="text" value="null"/>	codec6	<input type="text" value="null"/>
vad	<input checked="" type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
audio frames	<input type="text" value="2"/>	g.723.1 high rate	<input checked="" type="checkbox"/>	ilbc payload	<input type="text" value="98"/>
jitter size	<input type="text" value="0"/>	handset in (0-15)	<input type="text" value="7"/>	handset out (0-31)	<input type="text" value="20"/>
ring type	<input type="text" value="dtmf"/>	speaker out (0-31)	<input type="text" value="20"/>	speaker in (0-15)	<input type="text" value="0"/>

Fig .18 Audio Setting

- **codec1:** Set the priority 1 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.
- **codec2:** Set the priority 2 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.
- **codec3:** Set the priority 3 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.
- **codec4:** Set the priority 4 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.


- **codec5:** Set the priority 5 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.
- **codec6:** Set the priority 6 of the audio compression algorithm. The options are **g729** , **g7231** , **g711u** , **g711a** and **gsm**.
- **vad:** Enable/disable VAD (voice activity detection).
- **agc:** Enable/disable AGC.
- **aec:** Enable/disable VEC.
- **audio frame:** Set audio frames in RTP package. Minimum is 1 and maximum is 8.
- **g.723.1 high rate:** enable/disable g.723.1 high rate. G.723.1 high rate is 6.3kbps, low rate is 5.3kbps.
- **ilbc payload :** Set the payload model of RTP package when ILBC codec is selected to DTMF delay. The value range is 96-255.
- **jitter size :** Set buffer size of RTP package. The value range is 0-32.
- **handset in:** Set the volume of handset input.
- **handset out:** Set the volume of handset output.
- **ring type:** Set ring model by selecting corresponding item from drop down list.
 - n **dtmf :** Set ring as ordinary rings in different frequency
 - n **not disturb:** Set the phone do not ring by selecting this item.
 - n **user define :** Set ring as music saved by user by selecting this item.
 - n **Advanced:** Set ring used the individualized tone provided by system (need system support).
- **speaker out:** Set the volume of handfree output.
- **speaker in:** Set the volume of handfree input.

Dial Plan settings

Dial Plan Settings					
use dialplan	disable ▾	dial number	<input type="text"/>	ddd code	<input type="text" value="10"/>
idd code	<input type="text" value="86"/>	idd prefix	<input type="text" value="00"/>	ddd prefix	<input type="text" value="0"/>
inner line	disable ▾	inner line prefix	<input type="text" value="0"/>	call waiting	<input type="checkbox"/>
forward number	<input type="text" value="82378801"/>	fwd poweroff	<input type="checkbox"/>	fwd noanswer	<input type="checkbox"/>
fwd always	<input type="checkbox"/>	fwd busy	<input type="checkbox"/>	answer	<input type="text" value="30"/>
use digitmap	<input checked="" type="checkbox"/>				

Fig. 19 Dial Plan Setting

- **use dialplan:** Set whether use dial plan or use dial number by selecting the corresponding item in drop down list.
 - n **disable:** Do not use dial plan or dial number by selecting this item.
 - n **enable:** Use dial plan by selecting this item.
 - n **dialnum:** Use dial number by selecting this item. With this item selected, please enter the dial prefix into **dial number** field.
 - n **prefix:** Use 179XX service by selecting this item.
 - n **Hotline:** Use Hotline function by selecting this item. With this item selected, please enter the hotline number into **dial number** field.
- **dial number:** With **dialnum** selected in **use dialplan** drop down list, please enter the dial prefix into this field according to requirement of log in server. For example, with eTalk card used, enter 00 here.
- **ddd code:** With **enable** or **dialnum** selected in **use dialplan** drop down list, set area code according to E.164 dial rule. For example, Beijing is 10; Shanghai is 21.
- **idd code:** With **enable** or **dialnum** selected in **use dialplan** drop down list, set country code according to E.164 dial rule. For example, China is 86; U.S.A is 1.
- **idd prefix:** With **enable** or **dialnum** selected in **use dialplan** drop down list, set international call prefix according to E.164 dial rule, such as 00.
- **ddd prefix:** With **enable** or **dialnum** selected in **use dialplan** drop down list, set long distance call prefix according to E.164 dial rule, such as 0.

 **Note** With **dialnum** selected in **use dialplan** drop down list, you can also set dddcode, iddcode, iddprefix and dddprefix according to requirement of system.

- **innerline:** Enable/disable innerline call by selecting corresponding items from dropdown list.
 - n **disable:** Disable call innerline by selecting this item.
 - n **enable:** enable call innerline by selecting this item.
- **innerlineprefix:** With **enable** selected in **innerline** dropdown list, please fill the number prefix to pick up innerline , such as 0.
- **Call waiting:** Enable/disable call waiting by checking/unchecking the box.

- **forward number:** Enter receiving forwarded calls phone number into this field; If the IP Phone used with modem, with **modem** item selected in **iptype** list box, and then fill ISP number into this field.
- **fwd poweroff:** Forward calls if power off by checking this box. Please enter receiving forwarded calls phone number into **fwd number** field.
- **fwd noanswer:** Forward calls without replying by checking this box. Please enter receiving forwarded calls phone number into **fwd number** field.
- **fwd always:** Forward all calls by checking this box. Please enter receiving forwarded calls phone number into **fwd number** field.
- **fwd busy:** Forward calls if busy by checking this box. Please enter receiving forwarded calls phone number into **fwd number** field.
- **answer:** Enter a number from 0 through 60 to set the entries of the seconds before the phone answer the call auto or forward the calls.
- **Use digitmap:** Enable/disable digit map by checking/unchecking the box.

Protocol Setting

H323 Protocol Settings					
use service	<input checked="" type="checkbox"/>	service type	common	service addr	203.93.9.57
nat traversal	disable	nat addr		nat ttl	30
phone number	82378808	account		pin	
ras port	1720	q931 port	1720	h245 port	1722
rtp port	1722	tos	0	register ttl	60
local type	phonenumber	dtmf	h245 string		
super password	12345678	debug	output		

Fig.20-1 H323 protocol

MGCP Protocol Settings					
service type	common	service addr	203.93.9.57	register ttl	60
nat traversal	disable	nat addr		nat ttl	30
phone number	82378808	endpoint id		domain name	
register port	1720	rtp port	1722	tos	0
local type	normal	dtmf	rfc2833	dtmf payload	101
super password	12345678	debug	output		

Fig.20-2 MGCP protocol

IAX2 Protocol Settings					
service addr	203.93.9.57	register ttl	60		
phone number	82378808	account		pin	
local port	1720	dtmf	outband signal	tos	0
super password	12345678	debug	output		

Fig.20-3 IAX2 protocol

SIP Protocol Settings					
use service	<input checked="" type="checkbox"/>	register ttl	<input type="text" value="60"/>		
service type	<input type="text" value="common"/>	sip proxy	<input type="text" value="203.93.9.57"/>	domain/realm	<input type="text" value="203.93.9.57"/>
nat traversal	<input type="text" value="disable"/>	nat addr	<input type="text"/>	nat ttl	<input type="text" value="30"/>
phone number	<input type="text" value="82378808"/>	account	<input type="text"/>	pin	<input type="text"/>
register port	<input type="text" value="1720"/>	rtp port	<input type="text" value="1722"/>	tos	<input type="text" value="0"/>
outbound proxy	<input type="checkbox"/>	dtmf	<input type="text" value="rfc2833"/>	dtmf payload	<input type="text" value="101"/>
prack	<input type="checkbox"/>	super password	<input type="text" value="12345678"/>	debug	<input type="text" value="output"/>

Fig.20-4 SIP protocol

Fig.20 Protocol Setting

- **use service:** Enable/disable service by checking/clearing this box. To make calls through gatekeeper, please check this box; otherwise, phone can only make IP to IP calls or calls through gateway.
- **service type:** This option is used to accommodate the miscellaneous requirements of the system providers.

§ **common:** no special requirements

§ **huawei:** Use Huawei's system

§ **zte:** Use ZTE's system

§ **harbour:** Use Harbour's system

§ **utstarcom:** Use UTstarcom's system

§ **soyo:** Use Soyo's service

§ **etalk:** use etalk prepaid card. When this is selected, please select “**dialnum**” from the “Use dialplan” list box and put “00” into “**dial number**”.

§ **dda:** Use Dda's system

§ **vida:** Use Vida's system

§ **smartcon:** Use Smartcon's service

§ **auvtech:** use Auvtech's system.

§ **ringtec:** Use Ringtec's system. When this is selected, please select “**dialnum**” from the “Use dialplan” list box and put Ringtec account into “**dial number**”.

§ **citron:** Use Citron's system

§ **asiasoft:** Use Asiasoft's system

§ **txtc:** Use txtc's system

§ **i-fone:** Use i-fone's service

§ **suntek:** Use Suntek's system

§ **asiainfo:** Use Asiainfo's system

§ **lucent:** Use Lucent's system

§ **ipn:** Use IPN's service

§ **yyang:** Use Yiyang's system

§ **thinker:** Use Thinker's system

§ vfone: Use vfone's service

- **service addr** : Please set the URI (domain name/IP address : service port) of the server into “**service addr**”. When the default service port is used, “: service port ” can be omitted. Please note, if you use SIP protocol, this item will not appear!

If “**use service**” is checked, please set the URI of gatekeeper into “**service addr**”. The default service port is 1719.

If “**use service**” is not checked, to make calls through gateway, please put the URI of the gateway into “**service addr**”. To make IP-to-IP call, please clear this field. In both cases, the default service port is 1720.

- **sip proxy** : This item only appear in the SIP protocol. Please input the SIP Proxy Server URI when the **use severice** item is slected, sure on such condition, please input the SIP Proxy Server URI or don't set it anything to **domain/realm** item.
- **domain/realm** : This item only appear in the SIP protocol. In generally, the SIP account is aaa@bbb.ccc format, bbb.ccc is the SIP PROXY SERVER domain, so please input bbb.ccc part to the sip domain item. If this item is not setted, the IP Phone will think sip proxy as behind of @ automaticlly.
- **nat traversal**: When the IP Phone with private IP address need communicate with other IP Phones in a different LAN or on Internet, please select an item from dropdown list. (Fox IAX2 protocol, it will have not this item)
 - n **disable**: Select this item when the login server and IP Phone in the same LAN, or the login system supports the IP Phone working behind the LAN.
 - n **enable**: When the system does not support IP Phone working behind the LAN, please select this item to search public IP address of the NAT device. With this item selected, “**nat addr**” field will be activated. Besides, port mapping (port forwarding) needs to be properly set up on NAT device.
 - n **citron**: With Citron private protocol used, select this item to fit into the GnuGK system transferring the voice and signal by routed. This item only be used in H.323 protocol.
 - n **innopath**: Select this item with Innopath private system used. This item only be used in H.323 protocol.
 - n **vida**: Select this item with Vida private system used. This item only be used in H.323 protocol.
 - n **aivgr**: Select this item with aivgr private system used. This item only be used in H.323 protocol.
 - n **eproxy**: Select this item with eproxy private system used. This item only be used in H.323 protocol.

- n **stun:** Select this item with epoxy private system used. This item only be used in SIP protocol.
- **nat addr:** When “**nat traversal**” is set to “**enable**”, please put the domain name of the servers (These web server helps to find out the public IP of the IP Phone) into “**nat addr**”, such as **www.whatismyip.com**.
- **nat ttl:** When IP Phone sit behind a NAT device, it will send packets to server every “**nat ttl**” seconds to keep the port mapping on the NAT device alive. “**nat ttl**” is an integer between 10 and 65535 .default value is 20.
- **phone number:** The local phone number or username of this phone, usually is allocated by Sever provider. The Max length is 16 digit.
- **account:** when you use calling card(such as etalk,Net2phone) or some special severice, please input the account of chosen card into this field,however ,as for MGCP ,no this item. And when you use H.323 protocol to log on the Gatekeeper,please input H323ID in this column if the **account** item is selected in **local model** dropdown list;however if H235 item is selected in **local model** dropdown list, please enter user name here; While **prefix** item selected in **use dialplan** dropdown list, enter language indicating number, card number and # here, such as 14589653185 # . When use SIP or IAX2 protocl ,please enter account to this column.
- **pin:** While calling card is set, please model the password of chosen card into this field; while **md5 account** item selected in **local model** dropdown list, enter password here. While **prefix** item selected in **use dialplan** dropdown list, enter password and # here, such as 3185 # . For MGCP protocol ,no this item.
- **endpoint id** : this item can only be used in MGCP,please input Local name of EndpointID to this column.
- **domain name** : this item can only be used in MGCP,please input Domain name of EndpintID to this column.
- **ras port:** Set the register port ; can be any number between 1024 and 65535 . This item only be used in H.323 protocol.
- **q931 port:** Set the call signal port; can be any number between 1024 and 65535. This item only be used in H.323 protocol.
- **h245 port:** Set the control port , can be any number between 1024 and 65535. This item only be used in H.323 protocol.
- **rtp port:** RTP port is the port transferring and receiving voice packets using UDP protocol. This

is an even number between 1024 and 65535.

- **register port:**The default port is 2427 when you use MGCP protocol, however it should be 5060 of default port in SIP port.
- **Local port :** This item can only be used in IAX2 protocol,it means the port of IP Phone sending or receiving the register informaion,the default is 4569
- **tos:** Set the TOS field of the IP header of the RTP packets. The bigger this value is 0, the higher priority the packet is 224.
- **register ttl :** IP Phone will send a keep-alive registration message to H323 gatekeeper every “register ttl” seconds. The value range is 10-65535. Default is 60. This setting can only appear in the H.323 protocol
- **local type:** This parameter refers to how IP Phone authenticate itself to the gatekeeper and it will appear in the H.323 protocol and MGCP protocol. The meaning of each item is as follow:
 - n **phone number:** Use phone number as E.164 and H323 ID to login the GK(only to H.323).
 - n **account:** Use phone number as E.164 and designated H323 ID filled in account field as H323 ID to login GK(only to H.323).
 - n **auto:** use support automated negotiate authentication account and pin about MD5 or CAT(only to H.323) .
 - n **md5:** use MD5 authentication account and pin(only to H.323).
 - n **cat:** use CAT (Cisco Access Talk) authentication account and pin(only to H.323).
 - n **sha1:** use SHA1(Secure Hash Algorithm v1) authentication account and pin(only to H.323).
 - n **Ordinary:**use ordinary way to register auth(only to MGCP).
 - n **cnc auth:** use China Network Commuination system auth(only to MGCP).
- **outbound proxy :** It only appear in SIP protocol ,Enable/disable Outbound proxy by checking/clearing this box. If the system has an Outbound Proxy , please set the URI of the Outbound proxy into “sip proxy ” and set the domain name of SIP proxy server into “domain/realm”. The default service port is 5060.
- **prack:** Enable/disable support pre-ack(RFC3262) by checking/clearing this box. Only appear in the SIP protocol, and is be used to set whether the goods support the reliable temporary response.

- **dtmf:** Set DTMF signal sending way by selecting inband audio, h245 string ,q931 keypad , rfc2833, sip info, inband signal, outband signal from list box.
- **dtmf payload:** This item only apprea in SIP and MGCP protocol,it can be used to set the RTP playloya type during using RTP to send DTMF signal.
- **super password:** Set the super password of the phone.(Default super password is 19750407),the max length is not more than 16 bit.
- **debug:** Set the debug level of the phone.
 - n **disable:** Disable output the debug message by selecting this item.
 - n **output:** Output the operation information to the window, such as register, input by selecting this item.
 - n **output all:** Output all debug information and data in test window by selecting this item.
 - n **remote debug:** Save the debug information in SDRAM of IP phone by selecting this item.
 - n **no check:** Disable checking firmware tags when upgrading. This is not suggested, because it will increase the risk of upgrading the wrong firmware into the phone.


Other settings

Other Settings			
password	<input type="text" value="1234"/>	upgrade type	<input type="text" value="disable"/>
sntp ip	<input type="text" value="0.0.0.0"/>	upgrade addr	<input type="text"/>
timezone	<input type="text" value="(GMT+08:00)Beijing, Hong Kong, Urumqi"/>		
use daylight		<input type="checkbox"/>	
Save Settings		Address Book	
		Upgrade Firmware	

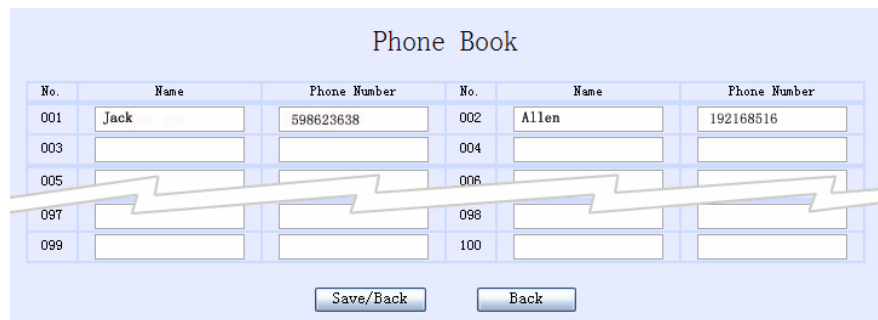
Fig .21 Other Setting

- **password:** Set the password of the phone. (Default password is 1234).
- **Upgrade type:** Set the auto-upgrade model of the phone.
 - n **disable :** Disable auto-upgrade .
 - n **all:** IP Phone will find matching firmware binary file and configuration file of the hardware model only at the FTP server specify by **upgradeaddr** item .
 - n **mac:** IP Phone will find matching firmware binary file and configuration file of the MAC address only at the FTP server specify by **upgradeaddr** item .
 - n **ppp id:** IP Phone will find matching firmware binary file and configuration file of the ppp id only at the FTP server specify by **upgradeaddr** item .
 - n **account:** IP Phone will find matching firmware binary file and configuration file of the account only at the FTP server specify by **upgradeaddr** item .
 - n **phonenumber:** IP Phone will find matching firmware binary file and configuration file of the phone only at the FTP server specify by **upgrade addr** item .
- **upgrade addr:** Put IP address or domain name obtained by ISP of FTP server supplying upgrade program into this field.
- **sntp ip:** Fill IP address of time server here. When network without Internet, Fill special address IP 255.255.255.255 here.

- **use daylight:** Enable/disable daylight.
- **timezone:** Select correct time zone in list box.
- **Save Settings:** Click this button to save the configuration and the phone will reboot. Once the phone reboots successfully, the new configuration is effective.

 **Note** After entering set page, if **save settings** button is not clicked within 5 seconds, then when you click it again, the index page asking for pin will pop up again. Then please input the password again to enter the set page and then click **save settings** button to confirm the modification.

- **Phone Book:** Click this button to open the speed dial settings page. Please refer to Fig.22. In this page, you can set and save the speed dial number by typing the name into the **Name** field and then entering the corresponding number following the name. For example, input Jack in **Name** field following 001, and then input 5989426454 into **Phone number** field. Then Jack's number 5989426454 is saved in phone book. Then please click **Save/Back** button. In normal state, you can use speed dial to call numbers saved in phone book.



No.	Name	Phone Number	No.	Name	Phone Number
001	Jack	598623638	002	Allen	192168516
003			004		
005			006		
097			098		
099			100		

Save/Back Back

Fig.22 Phone Book

- **Upgrade Firmware:** Click this button to update the program, the ring and the digit map of IP phone.



Firmware File Name: 浏览... Update Firmware


Digitmap File Name: 浏览... Update Digitmap

Ring File Name: 浏览... Update Ring

Fig.23 Upgrade firmware

Configured by PalmTool

PalmTool is a tool designed especially to configure and upgrade the IP phone. You can visit <http://www.yntx.com> to download the latest version of PalmTool. Then please unzip the downloaded file and save them.

- 1.) On a PC connecting with the phone or at the same segment of the phone, double click 

icon to open the PalmTool. The index page of PalmTool popup.

2.) Input the IP address of the phone into Local IP field (such as 192.168.1.100), and then click “Phone Settings” button.

From Version1.24, use PalmTool to set the IP phone, please set debug as output or output all firstly, or PalmTool cannot connect IP phone. The parameters of PalmTool is same as the parameters in HTTP, so please refer to HTTP set chapter to learn how to set IP phone.

Usage of the phone

1, Receiving calls

The IP phone can receive incoming calls from other IP phone and devices that support the standard VOIP protocol. It works just like an ordinary phone for incoming calls. When it rings, you can receive the call by following methods:

(1) Use handset

Lift the handset and begin speaking. When the call is over, put the handset back.

(2) Handset to hand free


While receiving call with handset, press “Hand free” on the keypad and then put down the handset. When the call is over, press “Hand free” again.

(3) Use Hand free

Press “Hand free” to speak to the other party. When the call is over , press “Hand free” again.

(4) Hand free to handset

While receiving the call with the “Hand free” pressed, pick up the handset to continue the call. When the call is over, put back the handset.

 **Note** When you communicate with the other party without lifting the handset, please do not exceed 40 CM from speaker.

2, Place a call

(1) Call another IP phone under the same Gatekeeper or server

①. **Handset:** Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press “#” or “Call” to end the dialing. Once the call connection has been

established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, put back the handset. The dialed number has been saved into the buffer.

- ②. **Hand free:** Press “Hand free” and listen for the Internet dial tone. Then input the phone number you wish to call and press “#” or “Call” to end the dialing. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, Press “Hand free” again. The dialed number has been saved into the buffer.
- ③. **Blind dialing:** Use the keypad to enter the phone number you wish to call and then press “#” or “Call” to make the call. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, Press “Hand free” again. The dialed number has been saved into the buffer.

(2) Place a call without use VOIP service (only for H.323 and sip)

The IP phone does not login the VOIP server , you can place a call by lifting the handset or pressing “Hand free” and then inputting the IP address of the other party, and then pressing “#” or “Call” .

FAQ

1.when the program in phone is being upgrading, why read-in data is not acceptable and “invalid update method” is shown?

Answer: open Palmtree and choose not to check for the model of the program as the debug level.

2 . The phone can PING but failed to use Palmtree, what should we do?

Answer: click “start” , choose “run” and input Telnet xxx.xxx.xxx.xxx (local IP) into the blank bar. After entering telnet entry window, input super password: 19750407.

P: \>set debug 4

P:\>write

3. With H323 protocol, your phone could not register for the GK(soft switch) ?

Answer:

A. check confirm the H 323 ID and GK ID (match upper/lower case)

B. take out “use 235” option. If no prepay service exist, select not to use the prepay card column. C. check to see if the service port is 1719. this would sometimes be ignored.

4. If the phone is down as a result of failed upgrading or other reasons, what shall we do?

Answer: enter password on keypad, press “*” and the phone would get into the save mode where we can input #5*5 in turn to drive it reset. After that, it will resume the default factory setting.

Note: the default setting of VoIP phone is on DHCP basis; and the default password is 1234.

5.What’s the difference between Single-end and Double-end VoIP phone?

Answer: Single-end VoIP phone has a single RJ45 interface which would be connected to the internet. And the Double-end one has a RJ45 interface as well as a PC one, which is make it work as a small exchanger that allows us to make a call and visit the website at the same time. However, the chip of this phone recently doesn’t support Router.

6.About the calling rule?

Answer: By getting digitmap , VoIP phone could be directly dialed the same way as traditional phone rather than pressing “#” as ending.

Instruction:

- 1) Write a set of rules and save as txt file;
- 2) Open PamITool, fill in the IP address on the corresponding column.
- 3) Click the Update DigitMap button, and select the txt file saved in step 1)
- 4)Update.

After the set of calling is saved into the phone, the number signal would be sent right after it is dialed

other than press “#” as ending when the number you dial is accord with the rules.

7.when the phone connect to the auto-telephone exchange, failed dialing to the extension occurs. Does it means extension could not be dialed any further after the telephone exchange is connected?

Answer: this problem if related to the DTMF transmission. In phone setting, the way of DTMF transmission is the same as PSTN exchanger and it could be selected in DTMF options.

8.what if TELNET and PING can log on, but IE failed?

Answer: there is edition(s) that don 't support IE configuration. Herewith, phone configuration could be done by TELNET or PalmTool.

9: After updating IP PHONE ,the LCD shows "■■" and stops ,how to deal with it?

Answer:

- 1) Press and hold * while power up, until you see * begin to display on your LCD.
- 2) Power off and do 1 again, after the second time power up, our standard design IP Phone's default ip address is 192.168.1.100.
- 3) Put the IP Phone and a Windows PC on the same LAN.
- 4) Run palmtool.exe on PC. You can try palmtool "Start Debug" button, a debug window will pop us, and if you press key on the IP Phone, the key and the IP Phone's ip address will be displayed on the debug window
- 5) If you PC's address is not 192.168.1.xxx (not on the same LAN of the IP Phone), change its tcp/ip settings to the same LAN.
- 6) Put 192.168.1.100 (or other default ip) in the palmtool "IP Address onChip" field
- 7) Find the correct upgrade file, and use palmtool "Update Program" button to update your IP Phone.

Reserves the right to make changes in technical and product specification without prior notice.

IP Phone User Guide

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