



# Catalogue

Introduction .....	2
Suitable Uses of YVK1000 IP Phone .....	3
Appearance .....	4
Performance and Features .....	5
Main technical index .....	6
Standard and Protocol .....	7
Electric requirements .....	7
Operating requirements .....	7
Installation .....	8
Configuration .....	9
1, Phone keypad set .....	9
2, Configured by WEB .....	15
3, Configured by PalmTool Usage of the Phone .....	30
Usaje of the Phone .....	30
Upgrade YVK1000 IP Phone .....	32
View Records .....	33

## 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.

YVK1000 series IP net 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 YVK1000 series IP net phone is in generalized inner line mode, it can call another YVK1000 series IP net phone worldwide for free. When YVK1000 series IP net phone is in generalized out line mode, it can places calls to ordinary telephones worldwide at a dramatically low price, because YVK1000 series IP net phone supports using prepaid card supplied by ISP such as Net2phone or eTalk. Moreover, it possesses excellent sound quality just like ordinary phones. YVK1000 IP Phone has 2X16 English LCD with backlit and supports being set by keypad.

YVK1000 series IP Network Phone includes 4 types which called A, B,C and D.

The IP Phone Models is :

Model		RJ45 port	backlight	color
YVK1000 series	YVK1000A(b)	single	no	white
	YVK1000A(h)			black
	YVK1000B(b)		yes	white
	YVK1000B(h)			black
	YVK1000C(b)	double	no	white
	YVK1000C(h)			black
	YVK1000D(b)		yes	white
	YVK1000D(h)			black

## **Suitable Uses of YVK1000 IP Phone**

YVK1000 IP Phone is the ideal choice for those who always place international or long distance calls. If two parties both use YVK1000 serial 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

## **Appearance**

### **1.Front Panel (See Figure 1 please)**

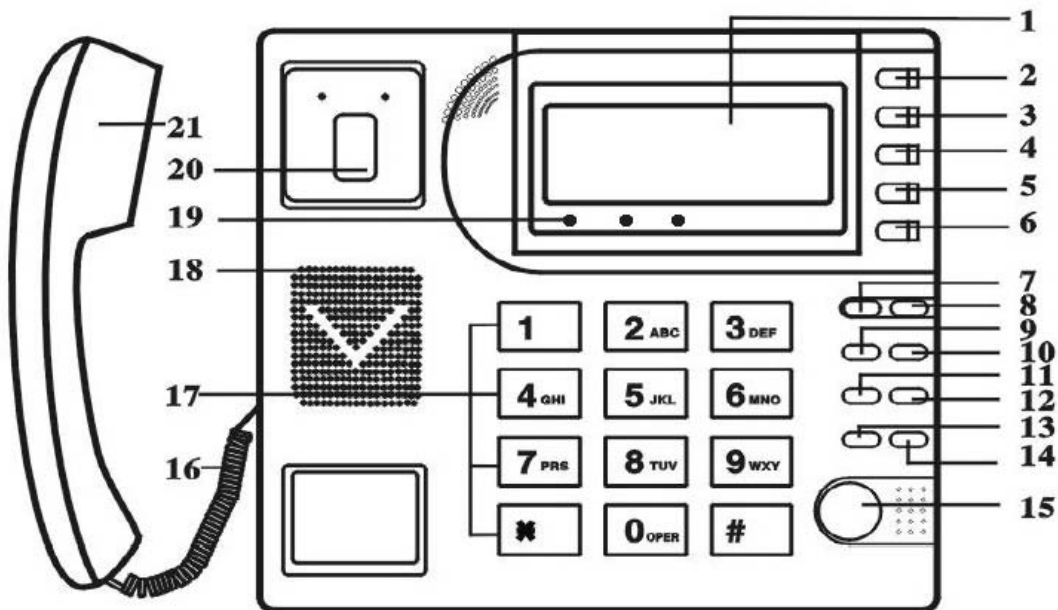


Figure 1 Front Panel

- |               |                 |                   |
|---------------|-----------------|-------------------|
| 1. LCD        | 2. Speed Dial   | 3. Phone Book     |
| 4. Back Space | 5. Volume +     | 6. Volume -       |
| 7. Service IP | 8. Local Number | 9. Missed         |
| 10. Dialed    | 11. Answered    | 12. Call          |
| 13. Local IP  | 14. Redial      | 15. Spk/Hand free |
| 16. Cord      | 17. Number      | 18. Speaker       |
| 19. LED       | 20. Handset     |                   |

**2. Backside view ( see Figure 2)**

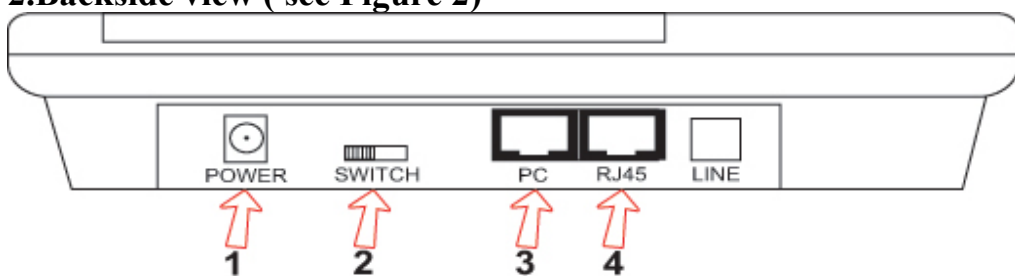


Figure 2 Backside view

- |                       |                 |                   |               |
|-----------------------|-----------------|-------------------|---------------|
| 1. Power adapter port | 2. Power switch | 3. PC(RJ-45) port | 4. RJ-45 port |
|-----------------------|-----------------|-------------------|---------------|

**3, Function Keys of IP Phone Introduction (Normal State)**

<b>Keys</b>	<b>Function</b>
Missed	With handset hung, press this key to review missed number
Answered	With handset hung, press this key to review received number
Local Num	With handset hung, press this key to get phone number
Dialed	With handset hung, press this key to review dialed number
Redial	While reviewing missed, received or dialed number, press this key to dial current number
Speak	Press this key to have a call without lifting the handset
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
Keypad	With handset picked or pressing speaker, press this key to dial number

## **Performance and Features**

- Support H.323 v4, compatible with most H.323 v1-v4 system and devices;
- Built in H.323 proxy support to pass NAT;
- Support MGCP RFC2705
- Support SIP RFC3261
- Support Net2phone private protocol
- Fast start and H.245 tunneling;
- Outband DTMF transmit by H.245 user input or Q.931 keypad;
- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX and gsm610 audio codec
- Caller ID send and receive by Q.931;
- Provide direct IP address to IP address call mode, direct PSTN call by a voice gateway ,E.164 phone number call under the mode of GK;
- Automatically gatekeeper discovery in LAN;
- Call PSTN by ITSP's prepaid card (eTalk, italk, ringtec etc);
- DHCP support for automatically assign IP address and others relevant parameters;
- PPPoE support for ADSL or Cable modem;

- Setting IP Net Phone parameters by standard web browser (such as IE6.0), phone keypad or standard telnet;
- Upgrade program by FTP mode;
- Support G.723.1 5.3k/6.3k, G.729, G.711 A-Law,  $\mu$ -Law audio codec algorithm;
- Dynamic voice detection; Echo cancellation; Comfort noise generation;
- Dynamic voice jitter buffer which minimize effect to the voice caused by the audio delay and jitter and as a result the quality of voice is high;
- Tone generation and Local DTMF generation and detection according with ITU-T;
- E.164 dial plan and customized dial rules;
- 80 entries each for missed call, answered call and dialed call;
- 112 entries for quick dial;
- 16 entries for voice message;
- 2×16 LCD display dial data, caller name, caller number and so on;
- Working status indicating Lamps(red , yellow and green) and keypad jacklight;
- Independently digit adjust the volume of handset, hand free;
- 14 function keys for operating and setting phone besides standard keys 0-9,#;
- Speed dialing;
- Adjustable volume for both handset and speaker;
- 16 function keys, background LED and states indicating lights;
- Settings by HTTP web browser (IE6.0);
- Advanced settings by Telnet;
- Voice prompt;
- Upgrade by TFTP.
- Hotline

## **Main technical index**



PPPoE: Point to Point Protocol Over Ethernet;

DNS: Domain Name Service;

NTS: Network Time service;

Telnet: Remote Host log on Protocol;

FTP: File Transfer Protocol;

HTTP: Hyper Text transfer Protocol;

Build in H.323 proxy support;

Auto search proxy

## **Electric requirements**

- Voltage: 9V DC
- Power: 0.5W (max.)
- Power adapter: AD/DC input 110-230V, 1A 400Ma
- Network interface: 1 or 2 RJ-45 Ethernet Connect

## **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

## **Installation**

## 1, Connect Handset and Phone

Insert Handset cord into handset cord jack of the base. Please see Figure 3.

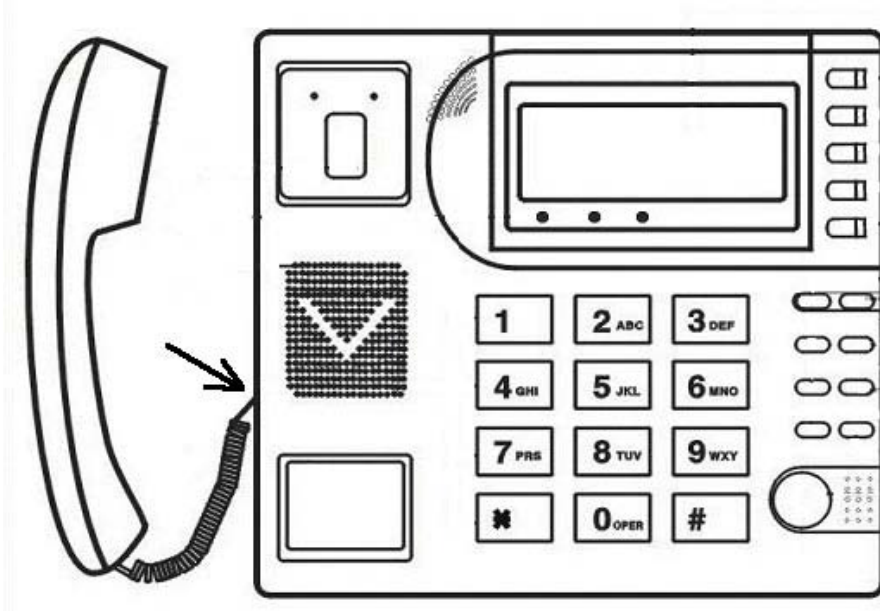


Figure 3 Handset and base of Phone connection

## 2, Connect Phone and Power

Place the phone nearby of Power socket. Plug the power cord adapter into the Power Jack. Then plug the other end of the power cord adapter into the appropriate power socket. Refer to Figure 4.

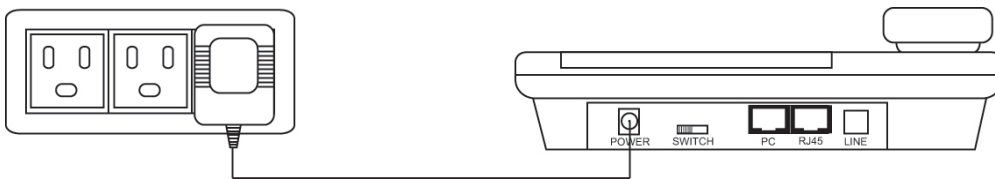


Figure 4 Connect Power

## 3, Connect the phone into the net

LAN users: Plug one end of the direct-connecting cable into RJ-45 jack which is located in the back of phone, and connect the other end of cable to hub. Please see figure 5.

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

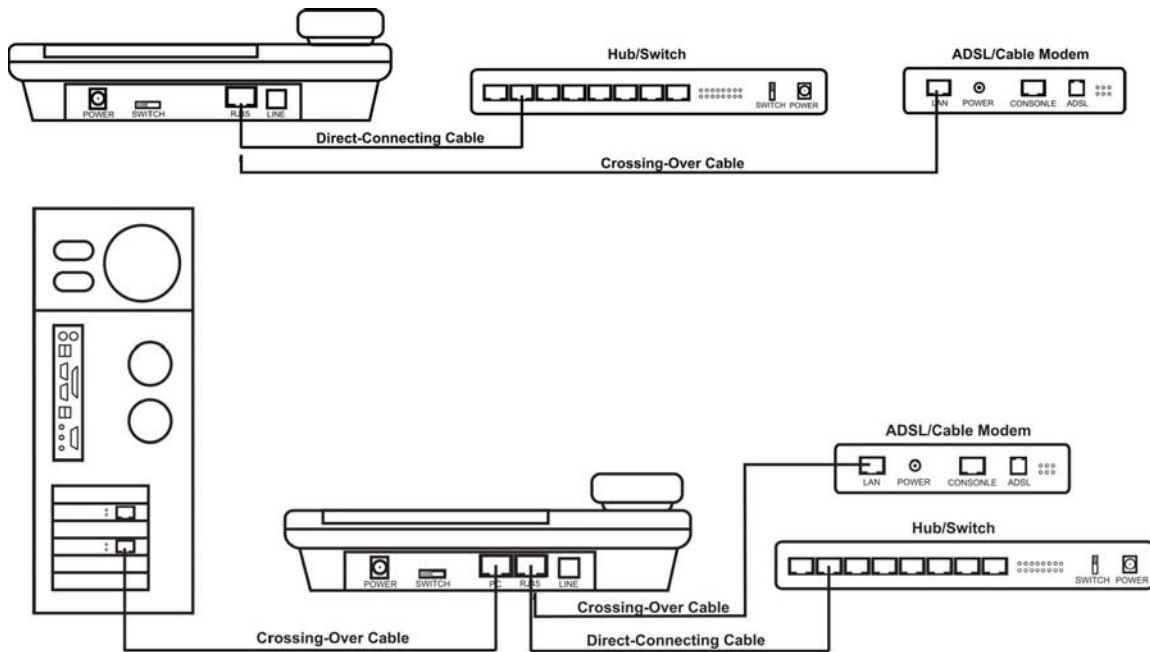


Figure5 LAN Phone connected into net

#### 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.

## Configuration

Four different ways can be used to configure YVK1000 IP phone:Phone keypad set; Web browser, Telnet commands and PalmTool configuration on computer.

### 1, Phone Keypad set

---

**Note: Default password of YVK1000A/B/C/D phone is 1234.**

---

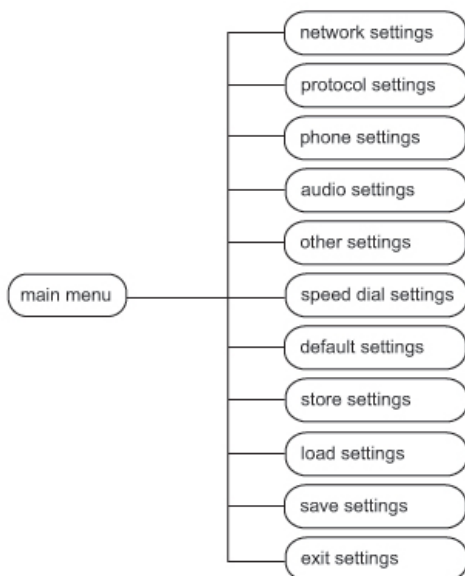
#### (1)Entering into setting mode

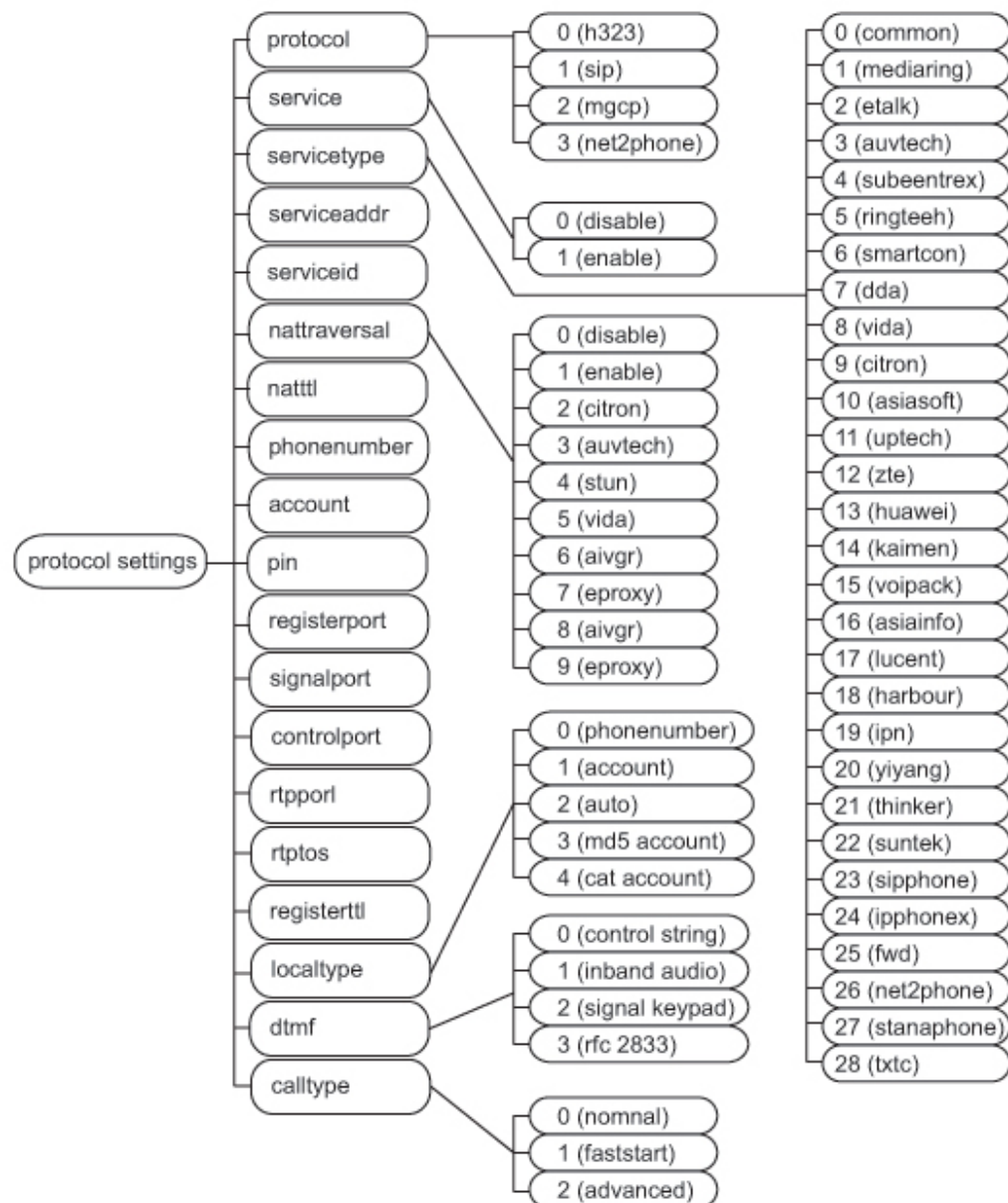
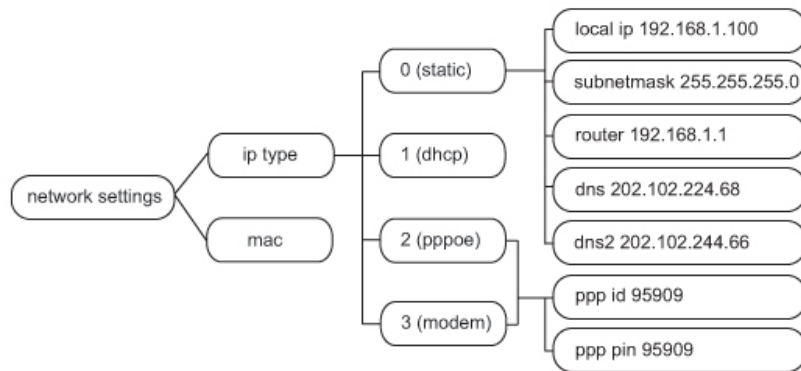
Use the keypad to enter the password of the phone (when debug is not set as 0[disable], default password is 1234; when debug is set as 0[disable], please use super password 19750407), and then press [#], till the **Password:** is displayed. Then enter the password again and press [Spk]to let the phone enter setting mode.

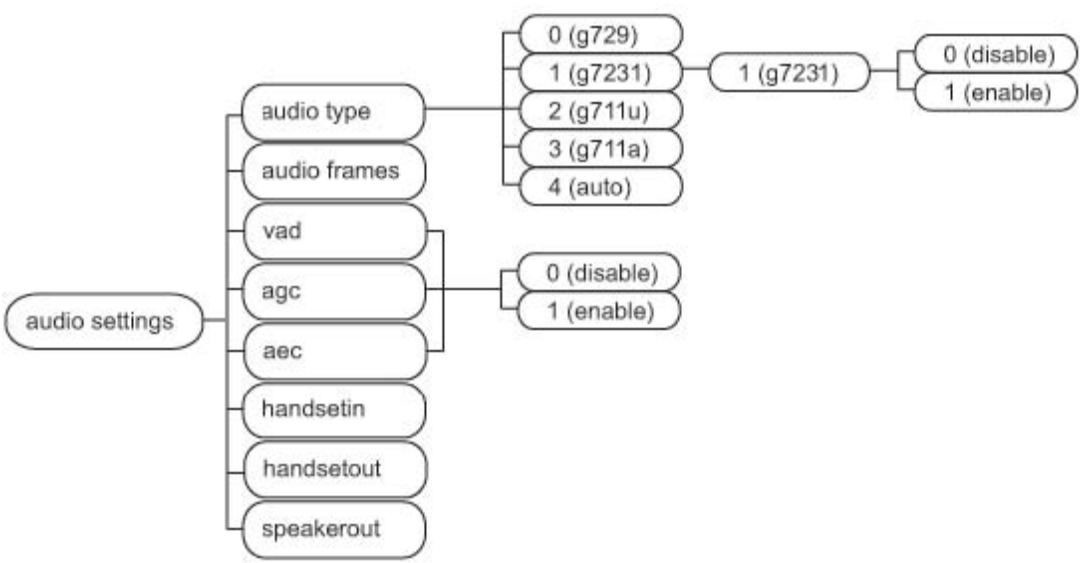
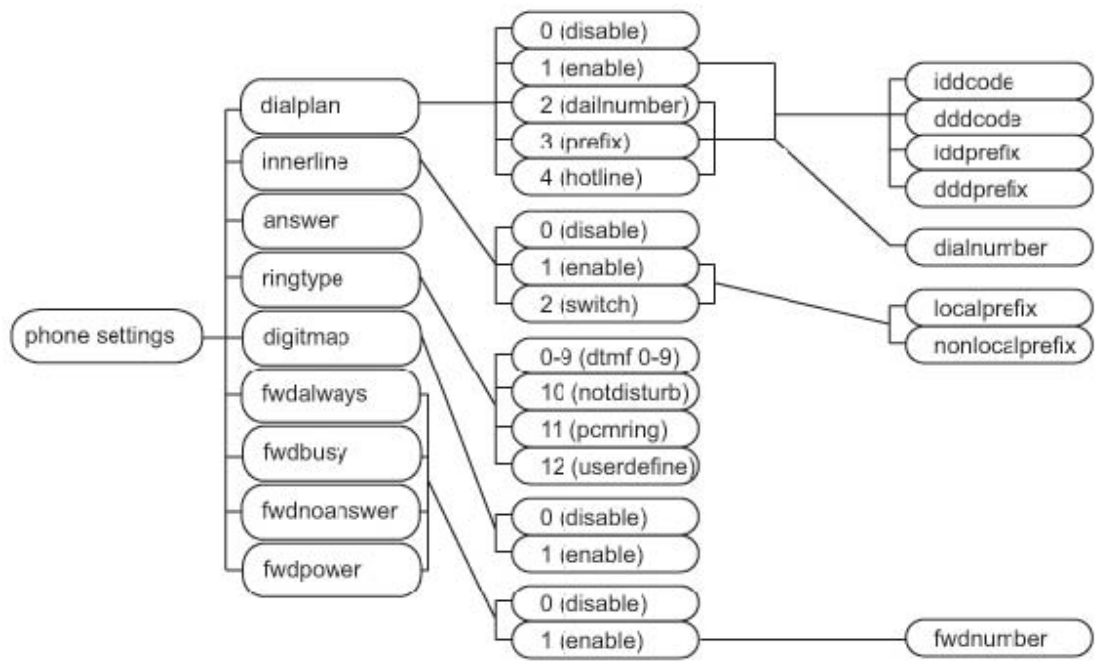
**(2)Introduce of the function of keypad in the keypad setting mode**

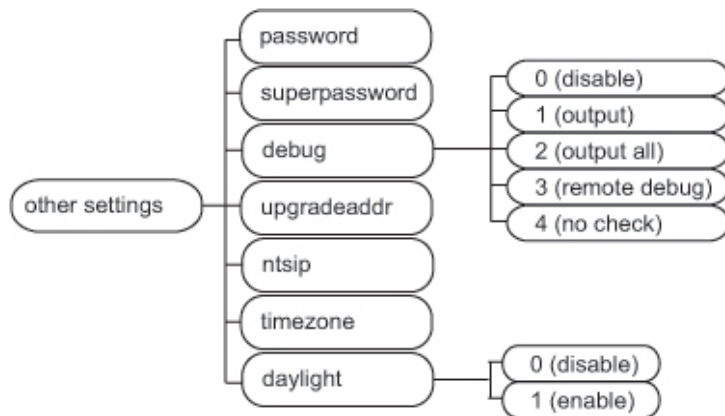
Press key	Function
Spk /Hand free	Enter into submenu of the current menu ;Acknowledge to modification
Volume/+	Scroll menu forward
Volume/-	Scroll menu backward
Local IP	Enter into modification status
Redial	Cancel current setting ; restore to its father catalogue
Back Space	Backspace during the setting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

**(3) Menu Structure**









#### (4)Setting example

We take enabling eTalk prepaid card and setting the ID and pin as an example to explain how to change setting:

##### 4.1. Enter setting mode

Use the keypad to input 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 **Password** is displayed. Then enter the password again and press [Spk] to let the phone enter setting mode.

##### 4.2. Enable eTalk

Once the phone enter setting mode, **network settings** will be seen, then press [Vol/+] to turn over the menu, until **protocol settings** is displayed. Please press [Spk] to enter settings submenu and then continually press [Vol/+] to turn over the menu until **Servicetype 0[normal]** is displayed on the screen. Please press [Local IP] to enter change mode, then you will see a cursor flashing at the beginning of the second line. It indicates that you can change the setting of this item. Then press [2] to enable eTalk and then press [Spk]to confirm **Servicetype 2[etalk]** will be seen.

### **4.3. Set eTalk account (0117ETNS):**

press [Vol/+] to turn over the menu , once *account* is displayed on the screen, please press [Local IP] to enter change mode, then you will see a cursor flashing at the beginning of the second line. It indicates that you can change the setting of this item. Then press 0117 keys once respectively, 3 key thrice, 8 key twice, 6 key thrice and 7 key quintic. Then press [Spk] to confirm. After above operation, the *account 0117ETNS* will be displayed on the LCD.

### **4.4. Set eTalk pin (409466193597)**

press [Vol/+] to turn over the menu, once *pin* is displayed on the screen, please press [Local IP] to enter change mode, then you will see a cursor flashing at the beginning of the second line. It indicates that you can change the setting of this item. Then press 409466193597 keys once respectively to input the password, and then press [Spk] to confirm. After above operation, *pin 409466193597* will be displayed on the LCD.

### **4.5. Save setting and exit setting**

continually press [Redial] to reach save setting menu *save settings*. Then please press [Spk] key to confirm the change. When the menu *save Are you sure ?* is displayed on the LCD, press [Spk] again to confirm. Once the IP phone restart successfully, the new setting is effective.

Refer to above operation; you can reach any menu to modify any value. Please refer to following structure illustrations to learn the values of each menu item. As for the meaning of each item and value, please refer to **Web Browser Setting chapter**.


## **(5)Appendix**

The following chart indicate the digital,letter and mark according to the times the corresponding character key is pressed in the setting mode.

Press key	Press once	Press twice continuously	Press three times continuously	Press four times continuously	Press five times continuously
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	:/@	;/-	'/&
*					
#	Switch between Uppercase and Lowercase				

To make the phone work well, we advice you to restart the phone manually after the configuration.

## 2, Configured by WEB

On a PC connecting with the phone or at the same segment of the phone, 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. From version 1.24, there are two passwords for the IP phone: ordinary password and super password. Default password 1234 is ordinary password and super password is 19750407 (the super pin for some PINGHE phone is not 19750407, please ask for the provider of the IP phone). 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 following configured page will popup. Refer to Fig 7please.

network settings					
iptype	dhcp	ppp id		ppp pin	
local ip	192.168.1.138	subnet mask	255.255.255.0	router ip	192.168.1.1
dns	172.21.212.254	dns2	0.0.0.0	mac	00-09-45-40-41-86
protocol settings					
protocol	h323	use service	<input type="checkbox"/>	register ttl	60
service type	common	service addr		service id	
nat traversal	disable	nat addr		nat ttl	30
phone number	82378008	account		pin	
register port	1720	signal port	1720	control port	1722
jitter size	0	rtp tos	0	rtp port	1722
local type	phonenumber	call type	normal	dtmf	control string
phone settings					
use dialplan	disable	dial number		ddcode	10
iddcode	86	iddprefix	00	dddprefix	0
innerline	disable	local prefix	0	nonlocal prefix	0
answer	30	ring type	dtmf0	use digitmap	<input type="checkbox"/>
forward number	82378009	fwd poweroff	<input type="checkbox"/>	fwd noanswer	<input type="checkbox"/>
fwd always	<input type="checkbox"/>	fwd busy	<input type="checkbox"/>	call waiting	<input type="checkbox"/>
audio settings					
audio type	auto	audio frames	2	g.723.1 high rate	<input checked="" type="checkbox"/>
vad	<input type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
handset in	7	handset out	20	speaker out	20
other settings					
password	1234	super password	19750407	debug	disable
sntp ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	
timezone	(GMT+08:00)Beijing,Hong Kong,Urumqi				
<input type="button" value="Save/Reboot"/>					
<a href="#">Address Book</a>					
<a href="#">Update Firmware, Digitmap and Ring</a>					

Figure 7 Http Setting Page

## (1)Network Setting

network settings					
iptype	dhcp	ppp id		ppp pin	
local ip	192.168.1.138	subnet mask	255.255.255.0	router ip	192.168.1.1
dns	172.21.212.254	dns2	0.0.0.0	mac	00-09-45-40-41-86

Figure 8 Network Setting Page

- **iptype:** Set how IP phone gets relevant network settings parameters by selecting corresponding item from drop down list.
  - static ip: Select this item to authorize users set IP address, subnet mask and router IP address of IP phone manually.

- **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. YVK1000 IP phone is shipped from the factory with DHCP on. So, if your 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.
- **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.
- **modem:** If the IP phone used with modem, please select this item to get relevant network parameters auto. Then please fill ID and pin into ppp id and ppppin 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. YVK1000 phone is shipped from the factory with a unique algorithm MAC address printed on the back of the base.

## (2)protocol settings

protocol settings					
protocol	<input type="text" value="h323"/>	use service	<input type="checkbox"/>	register ttl	<input type="text" value="60"/>
service type	<input type="text" value="common"/>	service addr	<input type="text"/>	service id	<input type="text"/>
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="82378008"/>	account	<input type="text"/>	pin	<input type="text"/>
register port	<input type="text" value="1720"/>	signal port	<input type="text" value="1720"/>	control port	<input type="text" value="1722"/>
jitter size	<input type="text" value="0"/>	rtp tos	<input type="text" value="0"/>	rtp port	<input type="text" value="1722"/>
local type	<input type="text" value="phonenumber"/>	call type	<input type="text" value="normal"/>	dtmf	<input type="text" value="control string"/>

Fig 9 Protocol Setting Page

- **protocol:** Select an item from dropdown list to set the protocol used by the phone.

- h323: Select this item to set the phone use H323 protocol.
- sip: Select this item to set the phone use SIP protocol.
- mgcp: Select this item to set the phone use MGCP protocol.
- n2p: Select this item to set the phone use Net2phone private system.
- iax2: Select this item to set the phone use IAX2 private system.
- wp: Select this item to set the phone use WP private system

- 
- Note With Net2phone selected here, please set other necessary parameters: check use service option, and then fill IP address or domain name of designated server into service addr field; then set service port as 6801; fill account and password of Net2phone card into account and pin fields.
- 

- **use service:** Enable/disable service by checking/unchecking this box. Different service responses different protocol as follows:

- With H323 protocol used, the protocol service refers to the gatekeeper searching the address. To let IP phone call each other by E.164 number, please check this box and then fill the IP address or domain name of corresponding gatekeeper into service addr field. Without this check box being selected, the phone can call by gateway or just by dialing IP address of other IP phone or H323 device (such as Netmeeting) at the same network segment.

- With MGCP protocol used, the protocol service refers to Call Agent. Please check this box according to system, and then fill service addr field with Call Agent IP address or domain name.
- With SIP protocol used, the protocol service refers to SIP Proxy Server. Check this box according to system, and then fill service addr field with SIP Proxy Server IP address or domain name.
- With Net2phone protocol used, the protocol service refers to designated server. Please check this box, and then fill the designated IP address or domain name into service addr field.

- 
- **Note** Designated Net2phone server IP address are: 216.53.3.52; 4.43.114.39; 4.43.114.38 or 205.228.245.8. Domain names are: call1.net2phone.com; call2.net2phone.com; skip1.net2phone.com; skip2.net2phone.com; skip1.f8g9h0.net or skip2.f8g9h0.net.
- 

- **Service type:** This option is used to accommodate the miscellaneous requirements of the system providers. When IP phone is connected to these systems, please select the corresponding service type.

- Common: no special requirements
- Mediating: use Mediating prepaid
- etalk: Use eTalk prepaid card by selecting this item. With eTalk selected here, please select dialnum in dial plan drop down list and then fill 00 into dial number field.

---

- **Note** When eTalk card is used, besides above configuration, please set other parameters as follows: service addr: 202.83.204.213; service id: ETNSGK; account: 0117ETNS.If IP phone can not log in 202.83.204.213, please try to use 202.83.204.213. And if IP address of the phone is a private one, please do port mapping on NAT device. Please refer to build in proxy document to get detailed operation.

---

- auvtech: Use Auvtech's h323 system by selecting this item.

- 
- **Note** When connected with Auvtech's system, please select "auvtech" from the "nat traversal" list box, select "account" from the "local type" list box.
- 

- subcentrex: Use subcentrex service by selecting this item.
- ringtec: Use Ringtec service by selecting this item. With this service used, please select dialnum in use dial plan drop down list and then fill account of Ringtec into dial number field.
- smartcon: Use smartcon service by selecting this item.
- dda: Use dda service by selecting this item.
- vida: Use Vida service by selecting this item.
- citron: Use citron service by selecting this item.
- asiasoft: Use asiasoft service by selecting this item
- Uptech: Use Uptech's MGCP system
- Zte: Use Zte's MGCP system
- Huawei: Use Huawei's MGCP or H.323 system
- Kaimen: Use Kaimen's MGCP system
- Voipack: Use Voipack's MGCP system
- Lucent: Use Lucent's H.323 system
- Harbor: Use Harbor's MGCP system
- IPN: Use IPN's H.323 system
- Yiyang: Use Yiyang's H.323 system
- Thinker: Use Thinker's H.323 system
- Suntek: Use Suntek's H.323 system
- Sipphone: Free SIP service on internet, please visit [www.sipphone.com](http://www.sipphone.com) for more information.
- Inphonex: Free SIP service on internet, please visit [www.inphonex.com](http://www.inphonex.com) for more information.

- Fwd: Free SIP service on internet, please visit [www.freeworldialup.com](http://www.freeworldialup.com) for more information.
- Net2phone: Use Net2phone private protocol system.
- service addr, service id: Please put 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. “service id” is filled according to the protocol used.
- H.323: If “use service” is checked, please put the URI of gatekeeper into “service addr”. The default service port is 1719. If the gatekeeper has a GKID, please put it into “service id”.

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.

- MGCP: Please put the URI of Call Agent into “service addr”. The default service port is 2727. Leave “service id” empty.
- SIP: If “use service” is checked, please put the URI of the SIP proxy server into “service addr”. Put the domain name of the SIP proxy server into “service id” or leave “service id” empty. If the system has an Outbound Proxy, please put the URI of the Outbound proxy into “service addr” and put the domain name of SIP proxy server into “service id”. The default service port is 5060. If “use service” is not checked, please clear “service addr” and “service id”.

Net2phone: Please put the URI of the Net2phone server into “service addr”. The default service port is 6801. Leave “service id” empty.

Net2phone usually use the following servers: [call1.net2phone.com](http://call1.net2phone.com), [call2.net2phone.com](http://call2.net2phone.com), [kip1.net2phone.com](http://kip1.net2phone.com), [skip2.net2phone.com](http://skip2.net2phone.com), [skip1.net2phone.net](http://skip1.net2phone.net), [skip2.net2phone.net](http://skip2.net2phone.net).

- **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 to set the proxy used by the phone.

- **disable:** Select this item when the log in server and IP phone in the same LAN, or the log in system supports the IP phone working behind the LAN.
- **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.
- **citron:** With Citron private protocol used, select this item to fit into the GnuGK system transferring the voice and signal by router.
- **auvtech:** Select this item with Auvtech private system used.
- **stun:** Select this item with SIP protocol used according to requirement of system. With this item selected, nat addr field is activated.
- **vida:** Select this item with Vida private system used.
- **aivgr:** Select this item with aivgr private system used.

• **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](http://www.whatismyip.com).

When “nat traversal” is set to “stun”, please put the URI of the stun server into “nat addr”, in the format as “domain name/IP address : service port”. The default service port for stun is 3478.

• **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 60, default value is 20.


• **phone number:** The local phone number or username of this phone, usually is allocated by system.

• **account:** With H323 protocol used, while calling card is set, please type the account of chosen card into this field; while md5 account item selected in local type dropdown list, enter ID here; while account is selected in local type dropdown list, enter H323 ID here. While prefix item

selected in use dialplan dropdown list, enter language indicating number, card number and # here, such as 14589653185 #. With SIP system which requires authentication, please put the username/account into this field. With MGCP protocol used, please enter local endpoint id (eg., aaln/0) here. With Net2phone system used, enter account of Net2phone card here.

- **pin:** With H323 protocol used, while calling card is set, please type the password of chosen card into this field; while md5 account item selected in local type dropdown list, enter password here. While prefix item selected in use dialplan dropdown list, enter password and # here, such as 3185 #. With SIP system which requires authentication, please put the password into this field. With MGCP protocol used, please enter domain name here. With Net2phone system used, enter password of Net2phone card here.

---

 **Note** When MGCP protocol is used, some system requires adding “[ ]” outside the domain name. So please fill the domain name with “[ ]” into pin fields, such as [voiptest.com] .

---

- **rtp tos:** Fill TOS segment of IP head package in RTP digital follow here.

- **register port:** The local UDP port registered with server to accept incoming handshaking messages. The default port number for MGCP protocol is 2427. The default port number for SIP protocol is 5060. For H.323 or Net2phone, any number between 1024 and 65535 is acceptable.

- **signal port:** With H323 protocol used, signal port is Q.931 port using TCP protocol, can be any number between 1024 and 65535.

- **control port:** With H323 protocol used, this port is H.245 port using TCP protocol, can be any number between 1024 and 65535.

- **register ttl:** With H323 or SIP protocol, IP phone will send a keep-alive registration message to H323 gatekeeper or SIP proxy server every “register ttl” seconds. The minimum value is 10, maximum value is 255. Default is 60.

- **Rtp tos:** Set the TOS field of the IP header of the RTP packets. The bigger this value is, the higher priority the packet is.

- **rtp port:** RTP port is the port transferring and receiving voice packets using UDP protocol. This is an even number between 1024 and 65535, can't be the same as “register port”.

- **local type:** With H323 protocol used, this parameter refers to how IP phone authenticate itself to the gatekeeper. The meaning of each item is as follow:

- phone number: Use phone number as E.164 and H323 ID to login the GK.
- account: Use phone number as E.164 and designated H323 ID filled in account field as H323 ID to login GK.
- auto: Use MD5 or CAT encryption by the mode of auto negotiation, on the condition of H.235 Encryption portocol .
- MD5 account: Use H235 encrypted username and password to login the gatekeeper  
CAT accout: Use Cisco access token  
sha1: Use SHA1(Secure Hash Algorithm v1) encryption mode  
cnc auth: Use CNC(China Network Commuincation) system auth.

- **Call type:** Set call type by selecting the items in drop down list. normal: Call out in normal way by selecting this item. faststart: Call out in faststart way by selecting this item. advanced: Call out in faststart and tunneling way by selecting this item. It is a recommended way with H323 protocol used.

- **dtmf:** Set DTMF signal sending way by selecting control string, inband audio, signal keypad or rfc 2833 from dropdown list.

### (3)phone settings

phone settings					
use dialplan	disable	dial number		ddcode	10
iddcode	86	iddprefix	00	ddprefix	0
innerline	disable	local prefix	0	nonlocal prefix	0
answer	30	ring type	dtmf0	use digitmap	<input type="checkbox"/>
forward number	82378009	fwd poweroff	<input type="checkbox"/>	fwd noanswer	<input type="checkbox"/>
fwd always	<input type="checkbox"/>	fwd busy	<input type="checkbox"/>	call waiting	<input type="checkbox"/>

Fig 10 Phone Setting Page

- **fwd 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 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.
- **fwd noanswer:** Forward calls without replying by checking this box. Please enter receiving forwarded calls phone number into fwd number field.
- **use dialplan:** Set whether use dial plan or use dial number by selecting the corresponding item in drop down list.

- **disable:** Do not use dial plan or dial number by selecting this item.
  - **enable:** Use dial plan by selecting this item.
  - **dialnum:** Use dial number by selecting this item. With this item selected, please enter the dial prefix into dial number field.
  - **prefix:** Use 179XX service by selecting this item.
  - **Hotline:** Use Hotline function by selecting this item. With this item selected, please enter the hotline number into dial number field.
- 

- **Note:** With 179xx service used, please set as follows: fill call prefix into dial number field, such as 17930; type ;language indicating number, card number and # into account field; fill password and # into pin 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 10; Shanghai 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 86; U.S.A .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, idddcode, iddprefix and dddprefix according to requirement of system.
- 

● **innerline:** Enable/disable multi-settings by selecting corresponding items from dropdown list. YVK1000 IP phone allows saving 5 settings totally.

- **disable:** Disable multi-settings by selecting this item, then the phone will call out using current setting.
  - **enable:** Use designated system to place calls by selecting this item.
  - **switch:** Enable multi-settings by selecting this item. Then please fill the prefix switching to backup setting 1 and backup setting2 into local prefix and nonlocal prefix fields.
- 

- **Note:** To modify the parameters of backup settings, please use Telnet commands.
- 

● **local prefix:** With enable or switch selected in innerline dropdown list, please fill the number switching to backup setting 1 here, such as 56.

● **nonlocal prefix:** With enable or switch selected in innerline dropdown list, please fill the number switching to backup setting 2 here, such as 57.

● **use digitmap:** Enable/disable digitmap by checking/unchecking the box.

● **ring type:** Set ring type by selecting corresponding item from drop down list.

- **dtmf 0-9:** Set ring as ordinary rings in different frequency
- **not disturb:** Set the phone do not ring by selecting this item.
- **pcmring:** Set ring as music shipped from factory by selecting this item.
- **user devone:** Set ring as music saved by user by selecting this item.

● **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. To disable auto answer function, please set this parameter as 0.

● **predial time:** Set time limit from picking up the speaker to dialing the first the number.

- **interdial time:** Set time limit between dialing two numbers.
- **postdial time:** Set time limit from dialing the last number to placing a call. If the next number is not dialed within the post dial time limit, then the phone will call the dialed number auto.

#### (4)Audio settings

audio settings					
audio type	auto	audio frames	2	g.723.1 high rate	<input checked="" type="checkbox"/>
vad	<input type="checkbox"/>	agc	<input type="checkbox"/>	aec	<input checked="" type="checkbox"/>
handset in	7	handset out	20	speaker out	20

Fig 11 Audio Setting Page

● **audio type:** Set audio type of the phone by selecting item from drop down list. The options are g729, g7231, g711u, g711a and auto. “auto” is suggested, since when “auto” is selected, IP phone will negotiate with system about which algorithm to use and be able to communicate with more terminals and systems.

● **audio frame:** Set audio frames in RTP package. With G723 audio codec used, set it as 1; with G729 audio codes used, set it as 2. Minimum is 1 and maximum is 8.

● **g.723.1 high rate:** With g.723.1 selected in audio type dropdown list, enable/disable g.723.1 high rate by checking/ unchecking this option.

● **vad:** Enable/disable VAD (voice activity detection) by checking/ unchecking this box.

● **agc:** Enable/disable AGC by checking/unchecking this box.

● **aec:** Enable/disable VEC by checking/unchecking this box.

● **handset in:** Drag the slider to adjust the volume of handset input. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.

● **handset out:** Drag the slider to adjust the volume of handset output. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.

● **speaker out:** Drag the slider to adjust the volume of handfree output. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.

## (5) Other settings

other settings					
password	1234	super password	19750407	debug	disable
sntp ip	210.59.157.10	use daylight	<input type="checkbox"/>	upgrade addr	
timezone	(GMT+08:00)Beijing, Hong Kong, Urumqi				

[Save/Reboot](#)

[Address Book](#)

[Update Firmware, Digitmap and Ring](#)

Fig12 Other Setting Page

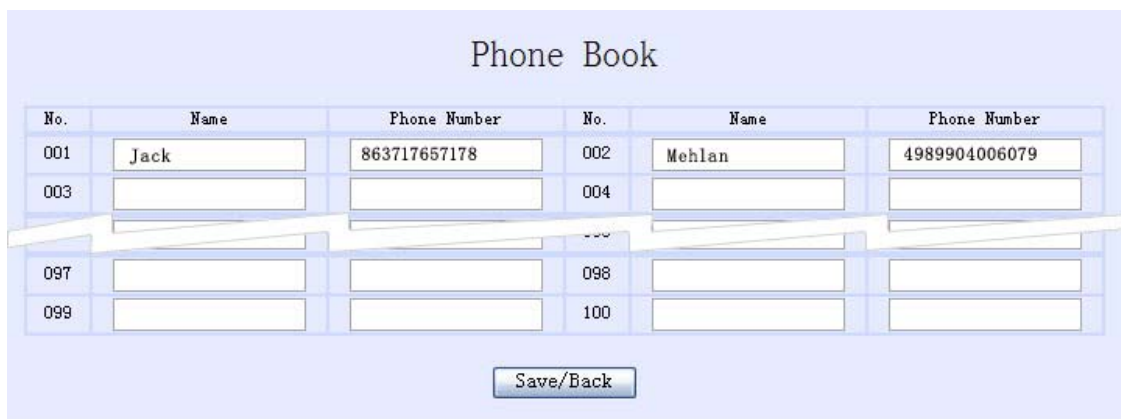
- **password:** Set the password of the phone. (Default password is 1234).
- **super password:** Set the super password of the phone.
- **debug:** Set the debug level of the phone.
  - disable: Disable output the bug message by selecting this item.
  - output: Output the operation information to the window, such as register, input by selecting this item.
  - output all: Output all bug information and data in test window by selecting this item.
  - remote debug: Save the bug information in SDRAM of IP phone by selecting this item.
  - no check: Disable checks the mark by selecting this item.
- **nts ip:** Fill IP address of time server here.
- **use daylight:** Enable/disable daylight by checking/unchecking this box.
- **upgrade addr:** Enter IP address or domain name obtained by ISP of FTP server supplying updated program here.
- **timezone:** Select correct time zone in dropdown list.

## (6) Phone Book

Click this button to open the speed dial settings page. Please refer to Fig 14. 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 863717657178 into Phone number field. Then Jack's number

863717657178 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.

- 
- **Note:** With H323 protocol used, if the phone dose not login Gatekeeper or Gateway; or with SIP protocol used, the phone dose not login Server Proxy, you can save IP address of other phone into phone number field. Please use “\*”instead of “.”. For example, save IP address 192.168.1.221, please input 192\*168\*1\*221.
- 



The screenshot shows a 'Phone Book' interface with a table containing two columns of data. The first column has entries for '001', '003', '097', and '099'. The second column has entries for 'Jack', an empty field, an empty field, and an empty field. The third column has entries for '863717657178', an empty field, an empty field, and an empty field. The fourth column has entries for '002', '004', '098', and '100'. The fifth column has entries for 'Mehlan', an empty field, an empty field, and an empty field. The sixth column has entries for '4989904006079', an empty field, an empty field, and an empty field. A 'Save/Back' button is located at the bottom center of the interface.

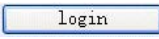
No.	Name	Phone Number	No.	Name	Phone Number
001	Jack	863717657178	002	Mehlan	4989904006079
003			004		
097			098		
099			100		

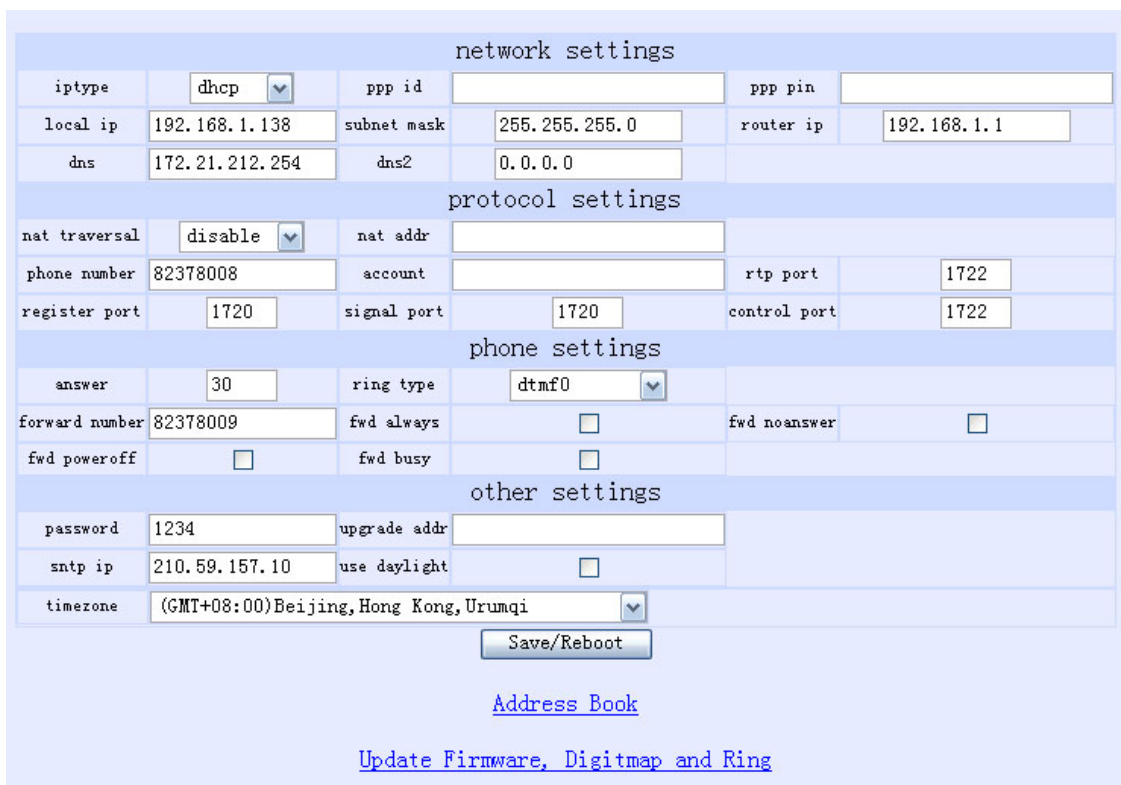
Save/Back

Fig14 Phone Book Illustration

- **Upgrade Program:** Click this button to update the program of IP phone. Before updating, please fill IP address of FTP server into upgrade addr field, and then click this button. Then the phone will read the corresponding bin files from the server and then load into the phone.
- **Update Digitmap:** Click this button to update the digitmap of the phone. Before updating, please fill IP address of FTP server into upgrade addr field, and then click this button. Then the phone will read the corresponding map files from the server and then load into the phone.
- **Upgrade Program:** Click this button to update the program of IP phone. Before updating, please fill IP address of FTP server into upgrade addr field, and then click this button. Then the phone will read the corresponding bin files from the server and Update Digitmap: Click this button to update the digitmap of the phone. Before updating, please fill IP address of FTP server into upgrade addr field, and then click this button. Then the phone will read the corresponding map files from the server .

- 
- **Note:** Please refer to PalmTool User Guide to learn how to write digitmap or just download TXT file from our site. Then please save it as “phone type.map” file, such as YVK1000.map.
- 

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



network settings			
iptype	dhcp	ppp id	ppp pin
local ip	192.168.1.138	subnet mask	255.255.255.0
dns	172.21.212.254	dns2	0.0.0.0
		router ip	192.168.1.1
protocol settings			
nat traversal	disable	nat addr	
phone number	82378008	account	
register port	1720	signal port	1720
		control port	1722
phone settings			
answer	30	ring type	dtmf0
forward number	82378009	fwd always	<input type="checkbox"/>
fwd poweroff	<input type="checkbox"/>	fwd busy	<input type="checkbox"/>
		fwd noanswer	<input type="checkbox"/>
other settings			
password	1234	upgrade addr	
sntp ip	210.59.157.10	use daylight	<input type="checkbox"/>
timezone	(GMT+08:00)Beijing, Hong Kong, Urumqi		

[Save/Reboot](#)

[Address Book](#)

[Update Firmware, Digitmap and Ring](#)


Fig 13 Setting Page using ordinary pin with Debug set as 0 [disable]

- **Update:** Click this button to save the configuration and restart the phone. Once the phone restarts successfully, the new configuration is effective.
- 

- **Note :** After entering set page, if Update 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 Update button to confirm the modification.
-

### 3, Configured by PalmTool

PalmTool is a tool designed especially to configure and upgrade the YVK1000 IP phone. You can visit <http://www.levi.cz> to download the latest version of PalmTool. Then please unzip the downloaded file and save them.

a) 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.

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

From Version 1.24, 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

YVK1000A/B/C/D IP phone can receive incoming calls from other YVK1000A/B/C/D IP phone and devices that support the H.323 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 YVK1000 IP phone under the same Gatekeeper**

**1.1. 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.

**1.2. 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.

**1.3. 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 login the Gatekeeper**

If YVK1000A/B/C/D IP phone does not login the Gatekeeper, 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].

### **(3)Place a call through Gateway**

If YVK1000A/B/C/D IP phone does not login the Gatekeeper, you can place a call through Gateway directly by lifting the handset or pressing [Hand free] and then inputting the IP address of the other party, and then pressing [#] or [Call] .

---

☛ Note When you place a call without Gatekeeper or with Gateway, please log off Gatekeeper. To get the detailed operation please refer to Configuration chapter.

---

## **Upgrade YVK1000 IP Phone**

### **Set FTP server**

FTP server can be supplied by the server provider as well as setup by the users in LAN. Please set the IP address of FTP server.

### **Prepare Updated program**

You can ask the server provider for the latest version of program or visit [www.levi.cz](http://www.levi.cz) to download the latest version.

### **Operation**

If you have got the IP address of the FTP server from ISP, please do as follows:

- a) Use keypad to enter setting mode
- b) Use keypad to input the IP address of FTP server
- c) Press 8 Local IP , then red light will blink twice a second. Once the phone restarts successfully, the new program is effective.

---

☛ Note Please do not change the name of the upgraded program, or the operation will be fail.

---

## View Records

### View missed calls

Click missed call keypad , then you will hear the record of missed call. Click Vol/+ keypad to turn the numbers orderly; click Vol/- to turn the numbers reservedly. If there is no record, you will hear nothing.

### View answered call


Click answered call, then you will hear the record of the received call. Click Vol/+ to turn the number orderly; click Vol/- to turn the numbers reservedly. If there is no record, you will hear nothing.

### View dialed number

Click dialed call , then you will hear the record of the dialed call. Click Vol/+ to turn the number orderly; click Vol/- to turn the numbers reservedly. If there is no record, you will hear nothing.

When you hear the number you want to dial, please press “#” or “call” key to place a call directly.

---

 **Note:** YVK1000 IP phone supports saving 127 entries unanswered call, dialed call and received call ranged from 0-126 at best. When the entries arrives 127, the latest record will cover the first one. The record will be lost when the phone restarts or turned on.

---