IP Phone User Guide

PH802

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1. Preface

First of all, thanks very much for choose our IP phone---PH802, thank you for your trust in our company. Our IP Phone PH802 completely follows VOIP standard offered by ISO, setting in two Protocols: SIP and IAX2, fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software in the market.

In order to make full use of our IP phone PH802 and getting the best return, please read the user manual carefully before using it and keep the manual for reference.

This manual introduces the IP Phone PH802's Installation, basic function, and special function, we hope this could help you to understand all the function soon and use them proficiently.

Note: since the product update, the manual will change, please visit to our website www.5111soft.com to see about. We may not inform respectively. Thank you for your understanding.

2. Product Overview

2.1 Support protocol

- SIP (RFC3261, RFC2543)
- Support IAX2.
- Support codec: G.711A/u, G.723 high/low, G.729 A/B
- Support G.168 echo cancellation standard, compliant 96ms echo cancellation with speaker mode
- Support voice volume adjustment, including IN/OUT of handset and speaker
- Support Jitter Buffer, VAD, CNG, SIP, Domain name register, point-to-point Call
- Support RTP and RTCP
- Support the Inbound/Outbound transmission; SIP info, DTMF Relay, RFC2833
- Support many countries' standard ring
- Support NAT: Support STUN, CITRON, AVS Mode
- Support SIP domain, SIP Authentication (none, basic, MD5), Domain Name parse
- Support two SIP server synchronously, including Pubic Server/ Private server，can make a call by any proxy. You can back-up and select any above SIP server.
- Support SIP application, including SIP Call forward/transfer/holding/conference/pickup/redial/ unredial/joincall
- Support BLF, server presence and peer to peer presence negotiation

- Support VPN (L2TP) clients and open VPN

2.2 Network features

- Support two models: Bridge and Router, integrate two ports router function.
- Support basic NAT and NAPT
- Support PPPoE for xDSL, and support auto redial when disconnect
- Support DHCP Client for WAN
- Support DHCP server for LAN
- Support DNS relay for LAN and provide DNS service for LAN Network equipment
- Support DNS domain name resolution in WAN port
- Support SNTP Client to get time from internet
- Support advanced DSP tech to ensure high quality voice
- Support advanced jitter buffer tech to prevent the info package delaying and losing
- Support network tool: ping， trace route， telnet client
- Support three modes to configure WAN port IP, they are: static, DHCP, and PPPoE
- Provide firewall for small LAN
- Provide optional priority level for small LAN
- Support second layer QoS (802.1p)
- Support VLAN
- Support VPN, L2TP protocol. (New hardware supports open VPN)

2.3 IP PBX advanced function

- Support 10 group quick dial number, together with IP PBX presence subscribe, the IP phone can display directly the online status of the booking numbers by the indicator. If the indicator is green, means it is online; the indicator is green and twinkle, means it is in the course of the call; if red, means it is offline.
- Support the local voice message, play the message by one key, IVR personality record the message, voice prompt.
- Caller ID display, ban calling out, avoid-disturb setting, auto-answering, auto dial while picking up the telephone, quick dial;
- Call waiting, call transfer, three ways call, and multi-dial forward
- Set the black list and limit numbers
- Support point to point call
- Setting the ended number methods
- Setting the ended number add, delete and substitution

- Setting the fixed calling ways
- Support phone number
- Support Silence Suppression, VAD (Voice Activity Detection)
- Support CNG (Comfort Noise Generation)
- Support Echo Suppression and AGC (Automatic Gain Control)
- Support DIGEST validate and MD5/MD5-sess encapsulation
- Support local/server Message-Waiting Indication
- Support auto long-distance configuration and edition auto-upgrade.
- Support call record checking and management.
- Support call pickup
- Support join Call
- Support redial and unredial
- Support directly dial IP+port to call SIP terminal device

2.4 Management and Maintenance

- Support post mode and upgrade via post mode
- Support web configure, keypad configure, manage and set the phone via Http and Telnet and user IP address filter
- Support update firmware and configure files via HTTP, FTP, TFTP
- Support Syslog
- Support add/delete administrator, changing password
- Support reverses Telnet to pass through NAT/FIREWALL and administrate the phone remotely.

2.5 Hardware Index

Model	PH802
Network Interface	2 * RJ45 10/100 Base-T
Key Appearance	36 buttons、1 LCD、2 POWER led lights
Standard AC Adapter	Input：100-240V—50/60Hz 0.25A Max Output：5V—1A
Dimension	Length: 295mm; Width: 205mm; Height:75mm
Weight	1kg
Operation Temperature	0-40 Centigrade
Humidity	10-65%（No-coagulation）

3. Installation

3.1 Installation

- Insert the power adapter's plug into the phone front Power jack (DC 5V) and the 2-prong plug end of which into grounded power outlet;
- Start up the IP phone by turning the front switch stated 'ON' & 'OFF' to 'ON';
- Connect the internet cable into wan port, and then follow below installation checking way.

If need to set up small LAN network, find the LAN cable in the box and connect between LAN port and your PC (PC is not required to set up for making a call)

3.2 Product Appearance, Name and Function of Each Part

IP phone PH802 is similar with the common telephone in the appearance design; the appearance and keypad buttons name are showed as follows:

PH802 series IP phone has 36 buttons and 2 power lights; the definition is showed as below:

Item	Name	Operation	Function/Display
1	Redial	Enter the dial mode and press this key	Redial the previous number and call out
2	UP	In the off-hook state, press this key	Increase volume (Max configuration is 9)
		Enter the configuration mode, press this key	Option of configuration page (Upwards pages)
3	Down	In the off-hook state, press this key	Debase volume (Min configuration is 1)
		Enter the configuration mode, press this key	Option of configuration page (Downwards pages)
4	Call list	In the on-hook state, press this key	Display the Received calls, Missed calls, Dialed calls
5	REC	In the off-hook state, press this key	Begin to record the call, press again to finish recording the call

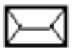
6	DEL	While dialling numbers, press this key	Cancel a number
		Enter the configuration mode, modify the configuration parameters, press this key	Delete the inputted configuration parameters
7	HOLD		If users choose the items “Call waiting” and” three way talk” for the added value service, press “Hold” in the course of the call, can hold the call temporarily , can dial other numbers, press it again then cancel the hold state.
8	PLAY	In the state of on-hook	Play the new voice message.
9	Speaker	In the state of on-hook, press this key	Using the handset, press this key and put down the handset, the telephone can use the speaker function to make a call.
10	0—9	In the mode of dialling number	Common figures “0-9”
		In the mode of configuration	Quick press it once, display the figure, and quick press twice or N times, display the English characters or other characters
11	*	In the mode of dialling number	Except dialing the third numbers in the three way calls, there is no any function
		In the mode of configuration	It is equal to the decimals among the 192.168.10.202
12	#	In the dialing mode (also select the item End with ‘#’ of Digital Map in the WEB page)	It is equal to end the number
13	OK	In the mode of configuration	It is equal to the confirmation key.
14	EXIT	In the mode of configuration	Quit the last menu
15	MENU	In the state of on-hook	Enter the configuration menu, needs to input the password, the default password is 123
16	Sys info	In the state of on-hook	See the telephone numbers in circle, IP, default route
17	M1-M10		Quick dial
18	Transfer	In the state of picking up the telephone	Call transfer

2 Power lights on the faceplate: display power supply, long light.

3.3 LCD Icon Explanations

IP Phone PH802’s LCD is equipped with the backlight, can see all the icons displaying the pictures while turn on the phone.

Icon	LCD Icon Definitions
	Network Status Icon: FLASH in the case of Ethernet link Failure or the phone is not registered properly.
DHCP	Network Status Icon: ON when Phone work on DHCP mode and FLASH when DHCP client is not registered successfully. OFF when Phone is work on another mode.
STATIC	Network Status Icon: ON when Phone work on Static mode and FLASH when IP address is disable. OFF when Phone is work on another mode.

PPPoE	Network Status Icon: ON when Phone work on PPPoE mode and FLASH when PPPoE is not registered successfully. OFF when Phone is work on another mode.
	Message Status Icon: ON and Flash if Phone has new message include text message or voice record
	Missed call display ON and Flash if Phone has missed call and not be read.
SIP1	SIP1 (Public SIP server) register State: Flash when enable register and can not register successfully; ON when enable register and register successfully; OFF when disable register
SIP2	SIP2 (Private SIP server) register Status Icon: Flash when enable register and can not register successfully, ON when enable register and register successfully, OFF when disable register
	Handset Status Icon: ON when off-hook OFF when on-hook
	Hand-free Status Icon: ON when phone work on hand-free mode OFF when IDLE or work on handset mode
SUN MON ...	Weekday Status Icon: Show the correct weekday according to the phone current date
	Numerical Numbers and Characters: 0 - 9 * # @ A, B, C, D, E, F, G, H, I, J, K, L, M, N, O, P, Q, R, S, T, U, V, W, X, Y, Z

4. Register via Web

Login the WEB page via Wan port or LAN port IP after connecting IP PHONE well. The default username and password are both ”admin”

- **Configure via LAN port:**

  A, Connect the PC with IP PHONE LAN Port, ensure LAN port IP is 192.168.10.*

  B, The default IP is 192.168.10.1, PC can login http://192.168.10.1 to configure

- **Configure via WAN Port:**

  A, Check WAN port IP address by the key ”SYSINFO” in the panel of IP PHONE

  B, Input the IP phone’s IP address, login in the WEB page, the default user name and password are all “admin”

4.1 Sip account register

SIP[Registered] Configuration

Register Server Addr	0101hk.com	Proxy Server Addr	
Register Server Port	6058	Proxy Server Port	
Register Username	90601748	Proxy Username	
Register Password	******	Proxy Password	
Domain Realm		Local SIP Port	5060
Phone Number	90601748	Register Expire Time	60 seconds
Detect Interval Time	60 seconds	User Agent	
Encrypt Key		Server Type	common
DTMF Mode	DTMF_RELAY	RFC Protocol Edition	RFC3261
☐ Signal Encode		☐ Rtp Encode	
☑ Enable Register			

Register Server Addr: input the register server address, it can be IP or domain name

Register Server port: Register the server port

Register Username: Register Username

Register Password: Register Password

Phone Number: Register Phone Number

Enable Register: Enable the register service, or register unsuccessfully

Pay more attention to save the configuration after configuring the option by WEB; or it will restore the original reversion when reboot. If register is successful, it will display the pattern"SIP1"on the top right corner on the display screen. If register is in fail, please configure again. You can dial number while others are commonly kept the default. If you have more demand, please look over the advanced configuration manual.

4.2 IAX2 account register

IAX2[Unregistered] Configuration

IAX2 Server Addr	
IAX2 Server Port	4569
Account Name	
Account Password	
Phone Number	
Local Port	4569
voice Mail Number	0
Voice mail text	mail
Echo Test number	1
Echo Test text	echo
Refresh Time	60
☐ Enable Register	☐ Enable G.729
☐ IAX2(Default Protocol)	

IAX2 Server Port IAX2 server port

Account Name Username authentication

Account Password Password authentication.

Phone Number Register phone number

Local Port IAX2 local port in device

voice Mail Number If IAX2 supports voice mail, the voice mail is in the letters form, it can't input letters, so replace the numbers with the name of voice mail.

Voice mail text If IAX2 supports voice mail, hereto configure the name of voice mail

Echo Test number If the platform supports circle loop, the echo test number is text formatting, then replace the echo test number which gateway configured with the echo text number. This function can be tested by platform or terminal. The call be tested it is normal or not from terminal to platform.

Echo Test text The number of echo test text.

Refresh Time The refresh time when IAX is registered.

☐ Enable Register Enable to register to the server.

☐ IAX2(Default Protocol) The iax2 protocol is the default call protocol. That is said, if choose this option, user picks up the handset and dial number, it will be communicated by IAX2 protocol. The default of gateway is SIP protocol, this configuration is aimed at the caller party, the receiver party wouldn't be affected. If user configures this item, it won't be used SIP, but IAX. If user wants to give a call by SIP simultaneously, then can configure the function of prefix substitution in the item of Dial peer and achieve the SIP call.

☐ Enable G.729 Enable to support G729.

5. Phone Operation

5.1 Keypad basic operation guide

Configure the IP phone via keypad, press "MENU" and input default password 123 to enter the configure menu. Use "**UP**", "**DOWN**" **to change the options, "OK" is confirmation, "EXIT" the previous step.**

5.1.1 Voice message and call record

- If the phone has no answer, you can set voice message: **advance→call service→enable voice record –>apply** to set the voice message, you can press "REC" to record the conversation. The maximum voice message and conversation record is 3.

- User can do as the following step to set the voice message IVR: **advance→call service→enable voice record** and user-defined voice and click "apply". You also need to set in the phone: **menu->123—>ok→down→voice record→ok→down→user defined→ok→switch→ok→del →1—>OK** to enable the voice message IVR function.
- If want to hear the another party's real time contents in the situation of opening the sound and words box, needs to set on the web configuration page: **advance→call service→incoming record playing**.
- Record the IVR: **menu->voice record →user-defined→rec**, then can record your voice, press "ok" to begin record, press the key "REC" to end the record.
- **menu->123→ok→down→voice record→ok→down→local→ok→down→rec** Can record one piece of message in the phone, other people can hear your message. Set the phone likes this: **menu->voice record –>local –>rec**

5.1.2 Configure IP phone's IP address

The phone is work in default DHCP mode, if there is no DHCP server in LAN, please do as following to switch to Static IP mode:

- Long press number key 1 to switch to Static IP mode, user can configure the IP phone address, DNS, gateway via IP phone menu button.
- Long press number key 2 to switch DHCP IP mode, default work mode is DHCP.
- Long press number key 3 to switch PPPOE IP mode.

5.1.3 Incoming call answering: three ways to answer the incoming call.

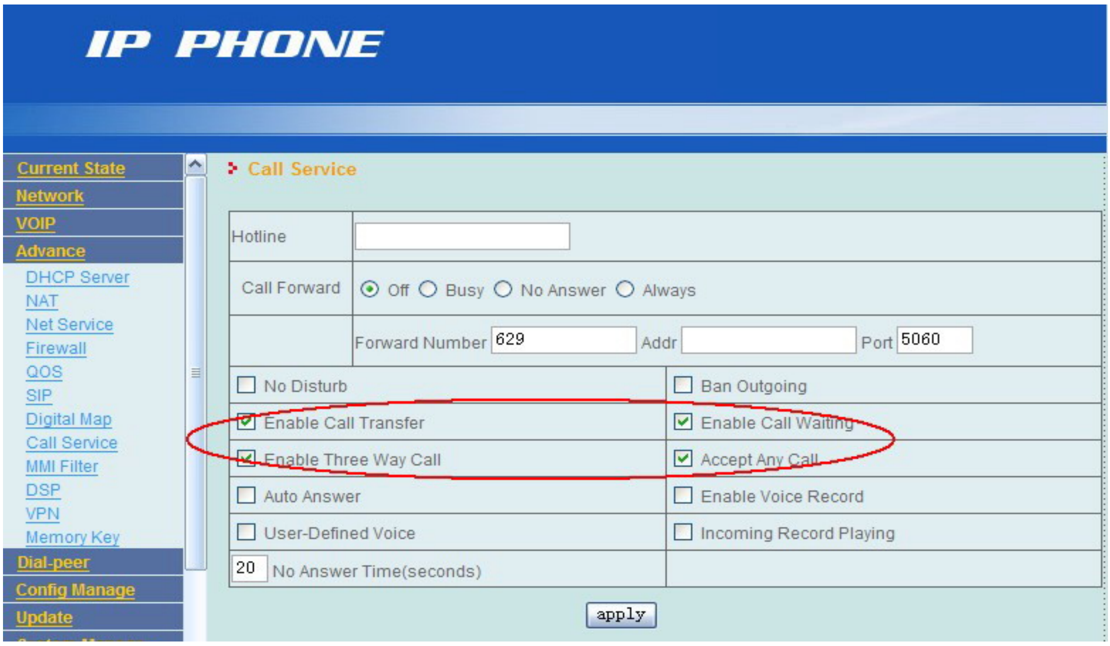
- Pick up handset to receive incoming calls
- Press Speaker button
- Use voice message to record the incoming call and press "play" key to listen. Voice message setting: enter **menu->123->ok->down->voice record->ok->received->old->List No, can play or delete.**

Note: Don't use the handset and the hand-free call at same time. The phone is in "hand-free" mode after press the key" hand-free", at same time, if picking up the handset, the hand-free will close auto.

5.2 Advance Function Explanation

The following four methods can turn on or off with the specific configuration in system (the specific operation methods is introduced by the following introduction）.

Note: Make sure this function is turned on while using this function.



5.2.1 Call Hold

If user press HOLD button while in the calling state, the current talking is locked, at the same time, user can make another call, this can be used with 3-way call.

The operation is: when A and B are talking, if A presses HOLD button, the B will be locked, B can not hear any talk from A. only when A presses HOLD button again, the HOLD state just be released. The talking for A and B could work normally again.

5.2.2 Call Transfer

Transfer has two methods: unattended transfer and attend transfer:

- unattended transfer: press TRANSFER button while in talking state, then the call will be transferred to the third phone directly. The operation is: Advance----Call Service----Enable Call waiting and Advance----Call Service----Enable Call Transfer. When A and B are in the talking state, B can press TRANSFER button directly and make any third number, ended by key # , so, the call will be transferred to the third party.
- attend transfer：when A and B are in talking state, B can press HOLD button to let A hold, and then dial C, ended by key #, at this time, B and C could talk. B presses HOLD button again, C will hold, and B and A could talk. B presses TRANSFER button, A and C could talk, B can put down the handset.

Note: this function need server support.

5.2.3 Three-Way Calling

Press HOLD button while talking state, the current talking is locked, user can make another call, and the 3-way call is working.

The operation is: must Enable Call Waiting and Enable Three Way Call first. A can call B, after talking, A presses HOLD button to keep the talking with B, when hearing the dialing tone, can call C, ended by key #, A and C can talk. A presses HOLD button again to back the talking of A and B, at this time, presses key * to realize the 3-way call.

Note: Once the three-way initiator concludes the three-way calling, the other two sides can not continue the conversation call and hand up automatically.

5.3 Display the Online State of User

There are 10 M keys in the phone，can set the online state of the stored numbers，users can see the online state of the stored numbers. After setting, press a valid memory keys to dial the stored number directly while in the state of handfree or picking up the handset.

Press the key ”M” without any pressing action, can see the number in the corresponding key on the screen, if do any action, will restore within 5 seconds.

If want to modify the definition of the M key, press the key ”DEL” within the displayed 5 seconds, input the new stored number. Press “ok” after confirmation, each memory keys can be set likes this.

After setting, can see clearly the state of each M key and the display of corresponding indicator. When using, press a valid memory keys to dial the stored number directly while in the state of handfree or picking up the handset.

If without any setting (the number is vacant), indicator M isn’t light. After setting, if the number is online, indicator M will be changed to green, if the other side is offline, displays red, if it is green and twinkle, means the other side is in busy line.

5.4 Voice message indication function

Local voice message indicator: enable local voice message function, the phone LCD will have ✉ if there is new voice message, press Play to listen to it.

Server voice message indicator: The phone LCD ✉ will flash with new voice message (need server support)

Note: Presence only supports SIP1 account inquiry.

6. Default Factory Setting

- The port WAN accept IP address in the mode of DHCP, if it is transferred into the static mode, the IP address is 192.168.1.179, the LAN address is 192.168.10.1.The DHCP and NAT will work automatically.
- Use the SIP protocol automatically, the SIP port is 5060
- HTTP port is 80, Telnet port is 23
- Use the key”#” to end the number

◆ The acceptable account names are guest and admin

◆ The phone time is subject to the standard time by using SNTP protocol

◆ Keypad configure password 123.

7. FAQ

7.1 The phone can't assign WAN or LAN IP?

When the phone is in the mode of NAT(no bridge), please don't assign the WAN and LAN IP with the same net domain. e.g., if the LAN IP address is 192.168.1.×, don't set the WAN address in the domain of 192.168.1.×.

7.2 The Memory button Led indicator is red when I configure number in the key?

◆ Please check in Web configure interface: Advanced — SIP— Enable Subscribe

◆ please make sure you SIP account is register on SIP1 and make sure your friends are under the some carrier with you.

7.3 Long press number key 2, still can't get the right IP address via DHCP mode?

◆ Please make sure there is DHCP server in Network.

◆ Make sure the LAN port IP address is in the same range with DHCP server IP address.

7.4 Why can't telnet the IP phone?

It's likely to use the private address in the phone and PC, but the two IP addresses aren't in the same net domain. E.g. the phone uses the address 192.168.1.×, the PC uses the address 192.168.10.180, then add an IP 192.168.1. ××in PC to have a try.

7.5 After configuring as the manual, but can't dial normally?

Check the network, Telnet logs in the phone, then use Ping or tracert command to access the exterior network.

If you meet other problem while configuring, please visit corresponding product FAQ page on www.5111soft.com. Or you may contact our tech engirneer by email or telephone.