IP Phone User Guide

PH806

Content

Preface

Thanks for purchasing our IP phone---PH806 and thank you for your trust in our company.

Our high-tech IP phone PH806 is designed especially for IP PBX and enterprise level users, it 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 PH806 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 PH806'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 our website www.5111soft.com to have a check. We may not inform respectively. Thank you for your support and understanding.

1. Technical Parameter and Hardware Specifications

1.1 Technical Parameter

Support Protocol:

Support SIP (RFC3261, RFC2543)

Support IAX2

Support Voice codec: G711A/u, G729, and G723.1

Support G.168 echo cancellation standard, compliant 96ms with speaker mode.

Support Jitter Buffer, VAD, CNG, SIP Domain name register, point-to-point Call

Support RTP and RTCP voice communication

Support the Inbound/Outbound transmission; SIP info, DTMF Relay, RFC2833

Support many countries' standard ring

NAT transversal: Support STUN, CITRON, AVS Mode

Support SIP domain, SIP Authentication (none, basic, MD5), Domain Name parse

Support 5 SIP servers and 1 IAX2 account synchronously, can call in and out by either proxy

Support SIP application, including SIP call forward/transfer/holding/waiting

Network Features:

Support two models: Bridge and Router, integrate two ports router function.

Support basic NAT and NAPT.

Support PPPoE for xDSL, and support off hook auto dial.

Support DHCP Client for WAN;

Support DHCP server for LAN;

Support DNS relay on LAN port and can provide DNS service for LAN Network equipment.

Support DNS SRV on WAN port

Support SNTP Client, can auto-obtain time from internet

Use advanced DSP tech to insure high quality voice

Use advanced jitter buffer tech to prevent the delaying and losing for package information

Support Network Tools, including ping, race route, and telnet client.

Support three modes to configure WAN port IP, they are: static, DHCP, and PPPoE.

Provide firewall control for small LAN.

Provide optional communication priority level for small LAN.

Support VPN—L2TP and Openvpn(SSL) protocol

Support Secondly Layer QoS (802.1p)

Advanced Function:

Support headset

Support 128*64 LCD

Support Power over Ethernet (POE) function

3 Interactive soft key, with more humanized operating prompt.

Support 5 SIP servers synchronously.

Support local voice record, message and server message.

Support sending and receiving short message

Support message wait indication.

Support user defined ring tone.

Support L2TP client.

Support call pickup, join call, auto-redial.

Support 5 programmable keys, 5 PSTN keys and 5 SIP keys, and it can be connected with the expansion board which can display more numbers' online status.

Support presence, BLF, Pust to talk

Support dial switchboard and extension number at one time, directly get through the ext. later.

Support phone book, and can set different rings according to different incoming callers.

Call waiting, call transfer, three ways call, and multi-call forward

Caller ID display, ban calling out, setting no-disturb, dial number automatically while picking up the telephone, set VIP numbers.

Set the black name list and confine numbers

Support point-point calling directly.

Support flexible methods of receiving numbers.

Support silence suppression and silence detection.

Support noise background simulation.

Support echoes suppression and auto gain.

1.2 Hardware Specifications

Item		**PH806**
Standard AC Adapter		Input:100-240V Output:5V 1.5A or PoE(802.3af, optional)
Interface	WAN	10/100Base- T RJ-45 for LAN
	LAN	10/100Base- T RJ-45 for PC
LCD size		128 * 64 full-dot matrix LCD
Operation Temperature		0～40℃
Operation Humidity		10～65%
Main Chipset		MIPS32(150M), DSP(100M）
SDRAM		16MB
Flash		4MB

2 Packing

Please check whether the packing contains the following or not.

The basic unit with handset
One cable
One user guide
Standard 5V/1.5A power supply
PH806 extension board (optional for wholesales)

3 Safety Information

You have two options for providing the PH806 with power:

- **Power over the Ethernet (IEE 802.3af compatible)**
- **An external power supply (5V/1.5A)**

Warn: Non-factory power supply may cause the phone damage.

4 Installation

- First connect one end of the handset cable to the handset, and connect the other end to RJ11 port on the phone's left and bottom side.
- Plug the Ethernet (network) cable into the RJ45 connector labeled **WAN**, and the other end is connected

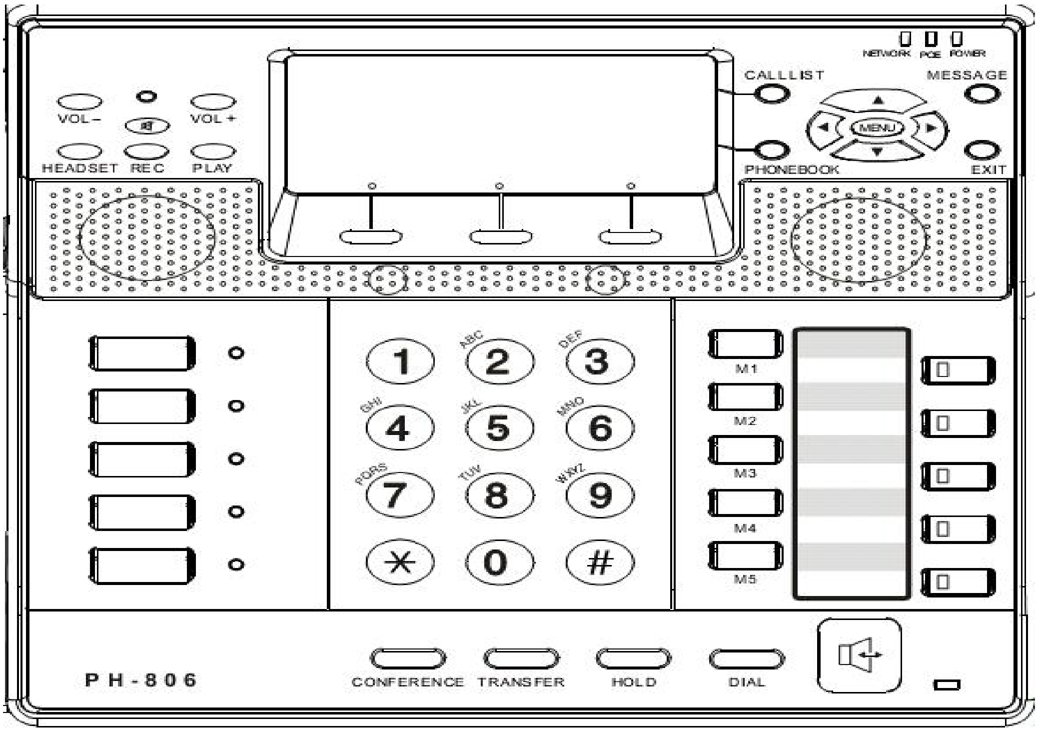
with internet or LAN.

- If you want to use labeled **LAN** port, use another Ethernet cable to connect PC with the phone’s LAN port.
- If you are using an external power supply, get the standard 5V/1.5A power supply and connect it to the phone’s power port; If you are using POE, just connect the Ethernet cable to the WAN port, and the other end of cable is connected with the POE device.
- If you want to use a headset, connect the earphone(excluded in our packing) with the phone. Please be sure the quality of the headset or the voice quality may be badly affected.

5 Keypad and LED Description

5.1 Keypad description

The numeric keypad with the keys 0 to 9, * , and # is used to enter digits and lettters. Depending on the operating mode, different actions can be performed (see the table below):



5.1.1 Number Key Description

Key	Digits	Lower case	Upper case
	**1**	**1 . - ? ! / : @ ‘ ’ ( ) +%**	**1 . - ? ! / : @‘ ’ ( ) +%**
	**2**	abc2	ABC2
	**3**	def3	DEF3

	**4**	ghi4	GHI4
	**5**	jkl5	JKL5
	**6**	mno6	MNO6
	**7**	pqrs7	PQRS7
	**8**	tuv8	TUV8
	**9**	wxyz9	WXYZ9
	*****	“*”“. ” Eg,192*168*1*110	“*”“. ” Eg,192*168*1*110
	**0**	(SPC)_0	(SPC)_0
	**#**	End the number and send; # can also be used to change and switch input method(ABC, abc, 123)	End the number and send; # can also be used to change and switch input method(ABC, abc, 123)

5.1.2 Function Key Description

KEY	Description
	Adjust volume (lower/higher); in standby mode for ring tone volume adjustment, in hands-free mode for speaker volume and in off-hook mode for headphone volume adjustment.
	Mute Microphone (on/off); When Mute function is on, LED light turns red.
	Press this key to answer the call by headset.
	Server’s record function. In the call status, press this button and the phone will send a serial of pre-configed DTMF to registering server. Then server begin to record. Press this button again, server will turn off the record. It needs server’s support.
	When the phone get a new voice message, the MESSAGE light will flashes. Users can press PLAY button and the phone will dial the special number to play the message automatically.
	Callist to display the Received calls, Missed calls, Dialed calls
	Phonebook

	Main menu
	In stand-by mode, press this key to check Missed calls.
	Press this key to check phone’s IP address in standby mode.
	Check dialed call record in standby mode; adjust the volume in off-hook mode and move the cursor to the left side in input mode.
	Press this key to check SIP1-SIP5 registeration status in standby mode, adjust volume in off-hook mode and move the cursor to the right side in input mode.
	LED for MWI (Message waiting indication) ; When the phone get a new voice message, the MESSAGE light will flashes.
	RLS to home state
	Establish 3-way conference.
	In call status, press TRANSFER+number+# can make Blind transfer; Press TRANSFER+number+Send(softkey2) can make Attend transfer.
	Call hold / unhold; In call status, firstly press **HOLD** to hold the call; and press **HOLD** again to unhold the call.
	Press this button to send the number; In standby mode to Redial.
	Speaker; In standby mode, press this button to enter hands-free mode; When there comes the call, press this key to answer the call in hands-free mode.
	Dynamic group function keys; In different situations, 3 softkeys including functions such as enter, save, quit, del , edit, option, send, redial, split, divert and so on.
	F1-F5; User-defined programmable keys; They default as LINE1-LINE5. Right side LED color indicates different status of LINE1to LINE5.

5.1.3 Softkey Description

Softkey is especially designed to improve the phone’s convenient and friendly operations.
For simple prompt keys, users can check it easily; Let’s mainly introduce the function keys below:

Function keys	Descriptions
SMS	In standby mode, press softkey1 to check and send the message.
SDial	In standby mode, press softkey2 to edit and check the speed dial config.
DND	In standby mode, press softkey3 to enable/disable DND(don’t disturb).
Drivert	When there comes the call, press this key and input the number you want to transfer then the call will be transferred directly.
Reject	Press this key to reject the incoming call.
Reset	When LINE1 is in call hold status, press this key to redial the LINE2 so that LINE1 can be holded and user can dial the LINE2 again.

Retrv	When LINE1 is in call hold status, press this key to quit the call hold and turn back to LINE1 talk status.
Conf	Press this key to make 3-way conference call; this key functions the sameas CONFERENCE on the keypad.
Switch	Press this key to switch the line you want to hold or unhold.
Split	In 3-way conference state, press Split to end it, bu can be available to keep the communication with the other two separately.
End	In 3-way conference state, press End to end all communications with the other two at the same time.
Option	While checking phonebook and call record, press Option to enjoy further more operations.

5.2 LED status display explanations

- MUTE LED: when MUTE is on, LED displays red if there is incoming call or in the call status.
- Network LED: LED is dark means no connection on WAN port; LED flashes green light means there exist network flaw.
- POE LED: When Power over Ethernet function turns on, LED is bright with light.
- Power LED: It brights means power supply is available.
- Memory LED: It can display different color and tell the presence status. Red color means the ext. is offline, green means the ext. is free and available, while the flashing green light means the ext. is busy or in the call status. More info please check 7.4.2 Memory Key.
- Message LED: It will flashes when there is voice mail from the server.
- LINE LED: Failed registration, LED is green and flashes occasionally.

  When comes the call and it rings, LED will frequently flash green light.

  In HOLD mode, LED will frequently flash green light.

  In call status, LED is on with green light.

6 Configuration via Keypad

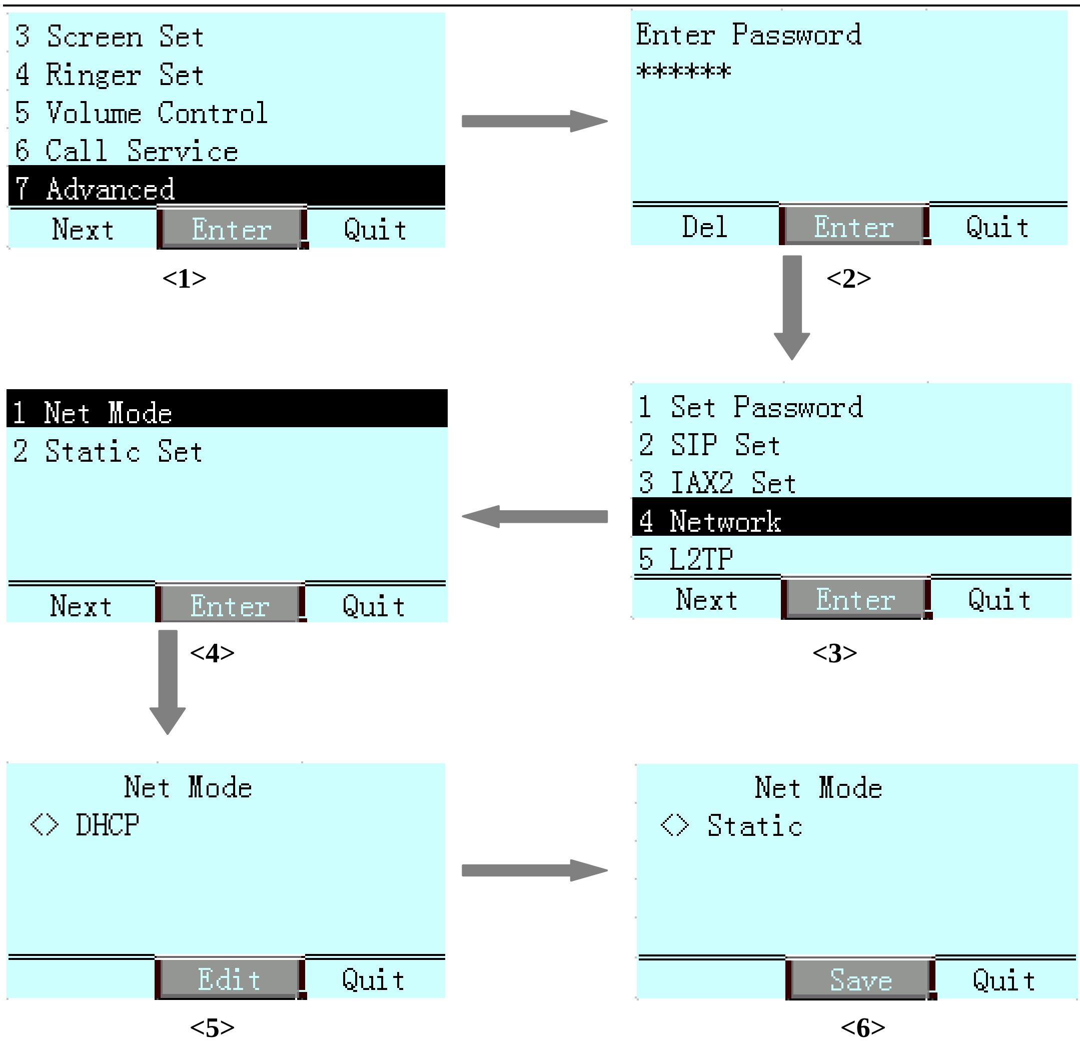
6.1 Network config

The phone defaults to obtain IP address via DHCP.

If DHCP mode isn’t supported from your netwrok environment, please congif static IP address according to the following instructions:

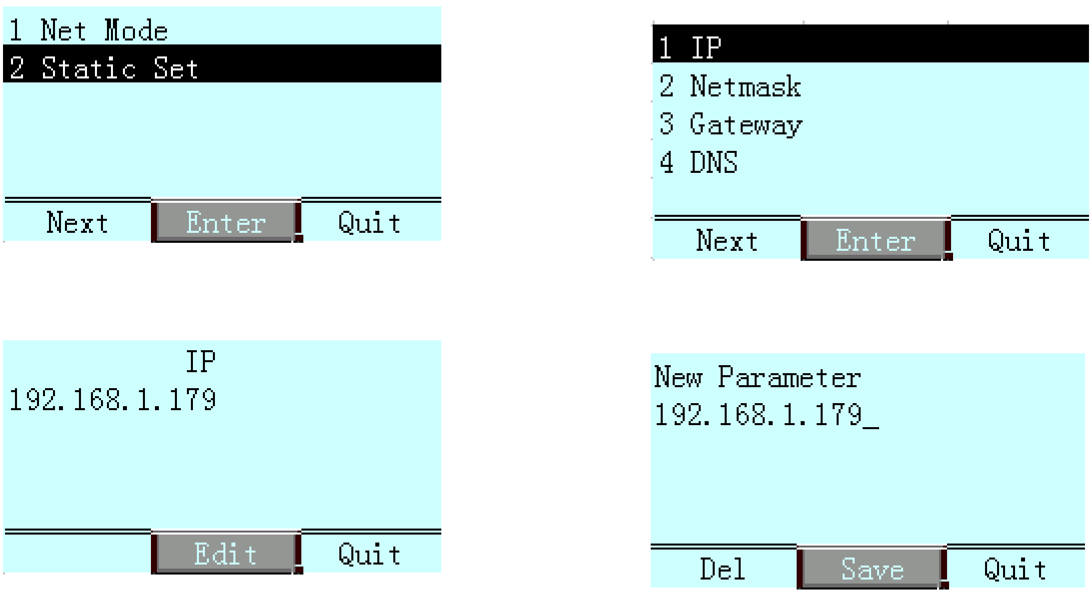
Press MENU key and select “Advanced” as the picture shows.

Enter password 123 and select “Network”. Then you can select the net mode according your network connection model. Let’s take Static IP config for an example.



Press  to select DHCP or Static mode.

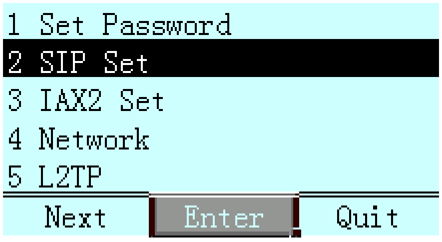
Select “Static”, and you can config IP address as the following instructions shows:

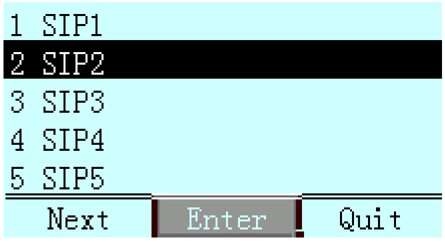


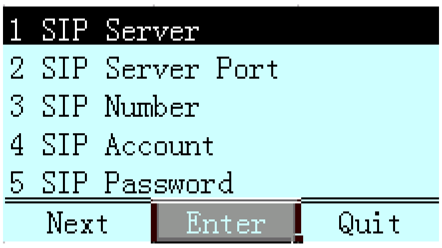
Press “**Edit”** key to edit IP address. You can use left/right key to move the cursor and insert the character to input IP address. Never forget to press “**Save”** key to finish and save the config. Config Netmask, Default Gateway and DNS by following the aboved steps.

After the network is configured, we now begin to config VoIP account.

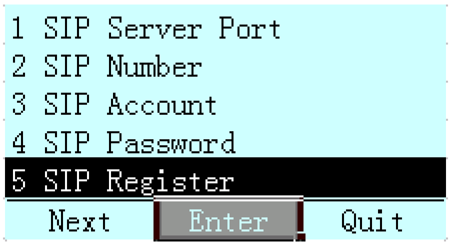
6.2 VOIP config







As the steps shows above, config SIP Number, SIP Account and SIP Password accordingly.



Press “**Edit”** select “ON” to submit SIP registration via  key. When it is registered successfully, you may check each LINE's registration status via key.

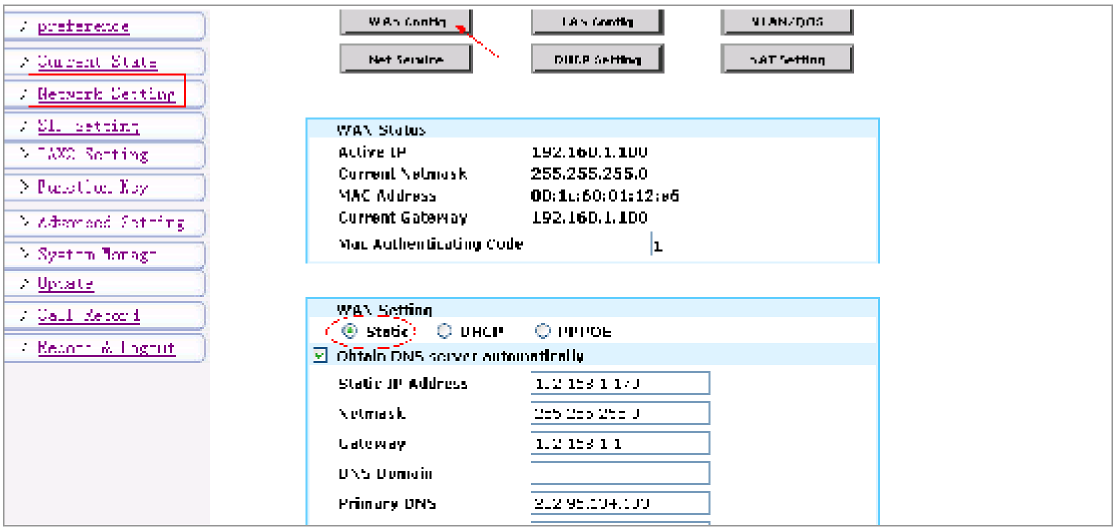
7 Configuration via WEB

Press "DOWN" key to check the phone's IP address, and input the IP address on the browser bar. Eg: **http://192.168.10.1**

7.1 Network Configuration

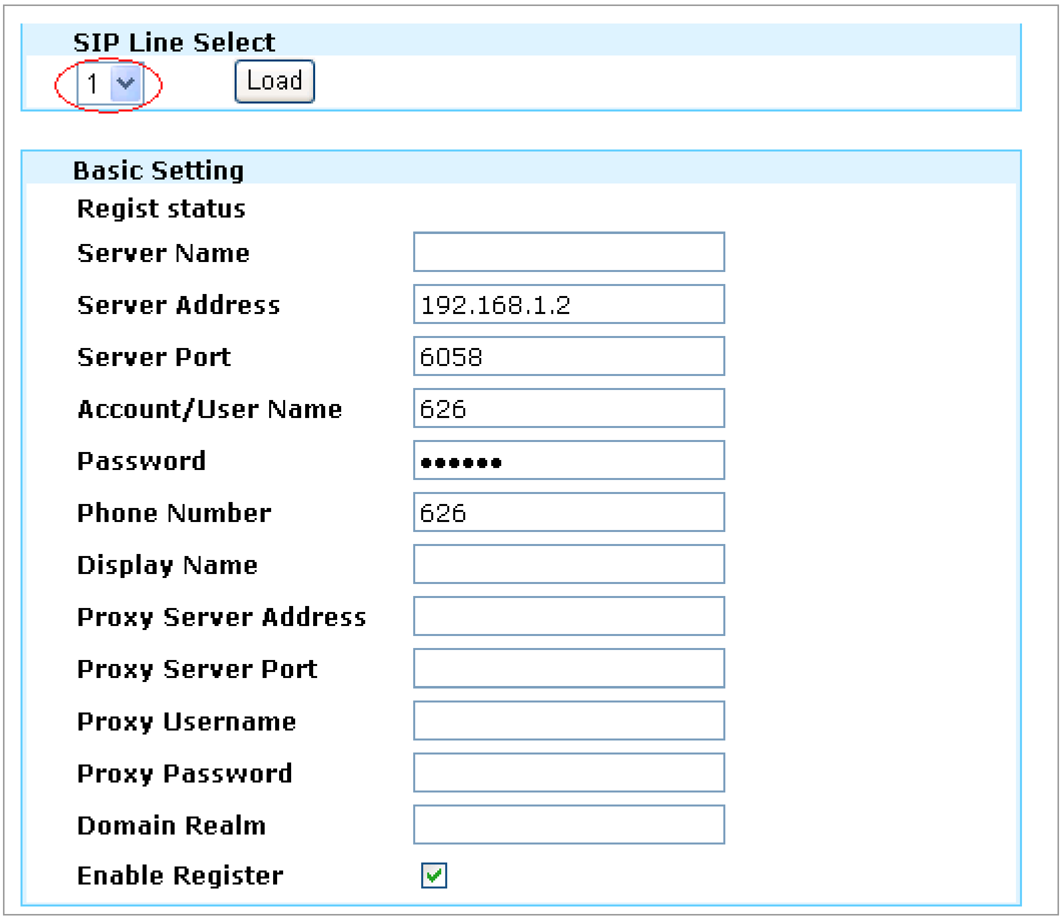
The phone defaults to obtain IP address via DHCP. If there is a DHCP server in your LAN, the phone's WAN port can auto-obtain an IP address, not any network config is needed.

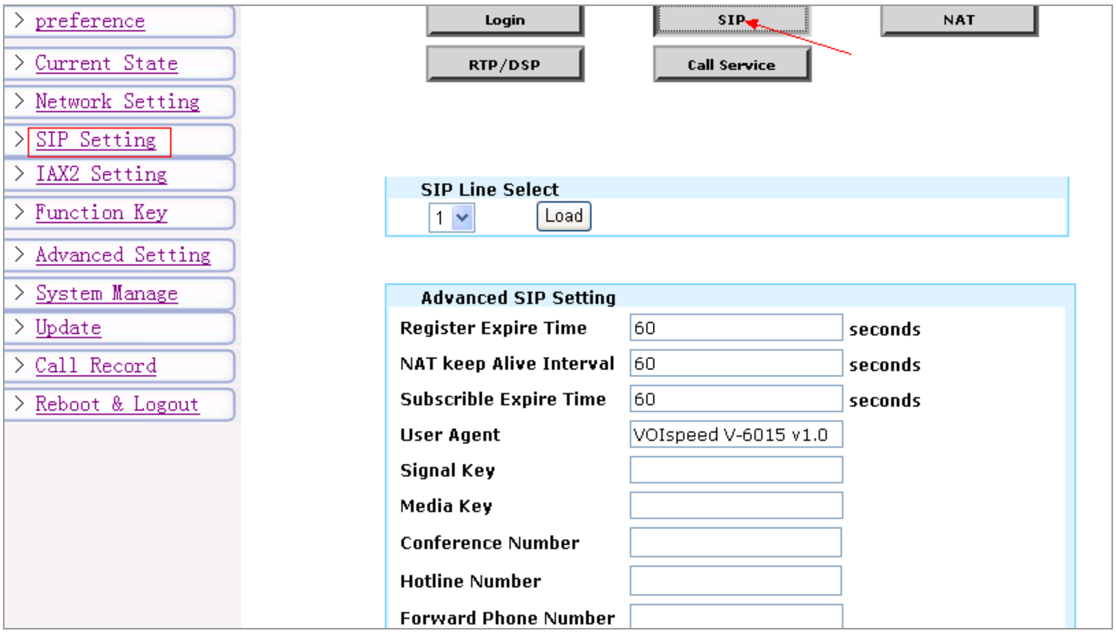
If your LAN router's DHCP function isn't open, then you will have to config static IP address on the phone's WAN port.



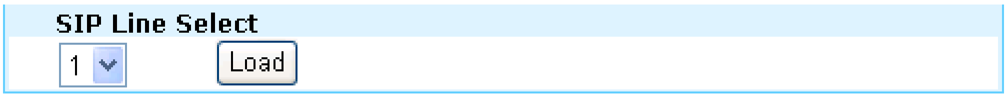
After finishing IP address config, you may login to the phone by telnet, then ping your registration server's address. Eg: Ping regiser.server.com, if ping works, it suggests your network config is correct.

7.2 SIP Configuration

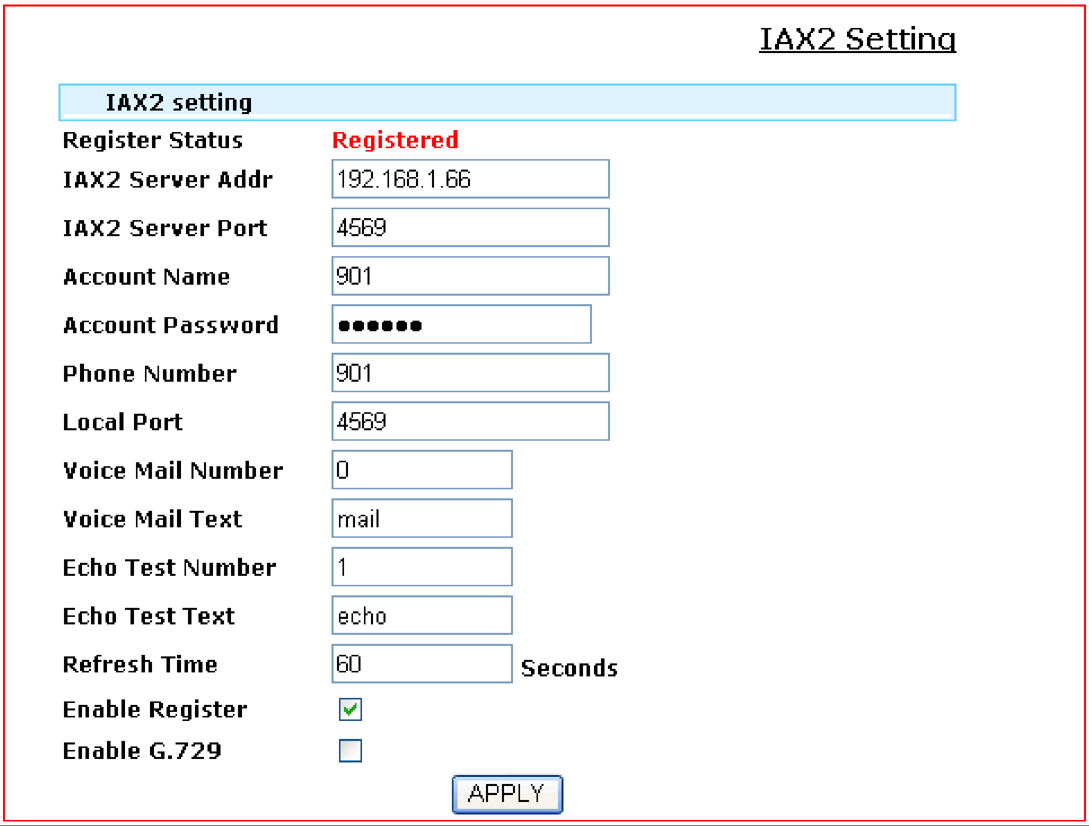




SIP Config

Field name	Explanation
	
Select the SIP account you want to config, SIP1-SIP5 are optional. Then click **【Load】** and switch to the corresponding account config.	
Register Status	It displays SIP registration status. “Registered” means successful registeration, or it will display “Unregistered”. “Unapplied” suggests the register status display function isn’t applied.
Server Name	Input server name
Server Address	SIP registering server address, domain name is supported
Server Port	Input SIP registering server port
Account Name	Input SIP account’s name
Password	Input SIP account’s password
Phone Number	Input SIP server’s phone number. Blank means not to apply for registration.
Display Name	Input the caller’s name you want in the callee’s display, English alphabets input supported.
Domain Realm	Input SIP domain realm. If the server doesn’t request SIP terminal’s local domain as appointed domain, then the local domain can be the same as SIP server. Usually, to simplify user’s input, clients are not necessary to input local domain. Because system will be filled with domain realm in the Register server addr place.
Enable Register	Select enable/disable register
Register Expire Time	Config SIP server register expire time, it defaults as 60s. It can be modified on the phone according to server’s time request and register again.
DTMF Mode	Total three DTMF modes: - DTMF_RELAY - DTMF_RFC2833 - DTMF_SIP_INFO。 Different SP may provide different mode.
Presence Mode	PH806 IP phone supports two standard format Presence definition as follows: Special and Standard. Default for the Standard. You need to restart to activate the phone after revising the mode.
Enable Subscribe	After registered successfully, you can subscribe to others presence status or voice messages etc.

7.3 IAX2 Registration



IAX2 Config

Field name	Explanation
Register Status	It displays SIP registration status. "Registered" means successful registeration, or it will display "Unregistered".
IAX2 Server Addr	Config IAX2 server address, can be in form of domain name.
IAX2 Server Port	Config IAX2 server port;
Account Name	Config IAX2 authenticated account name;
Account Password	Config IAX2 authenticated account password;
Phone Number	Config IAX2 phone number;
Local Port	Config monitoring port of local IAX2;
Echo Test Text	Config echo test text.
Refresh Time	IAX2 registration refresh time, which is in the "second" unit. It is suggested that users make a choice between 60-3600;
Enable Register	Enable/disable registering to the server;
Enable G.729	Enable/disable G.729; the phone supports G.729 codec.If you use the idefisk (G.729 non-supported), then calling idefisk would result in your PC corruption;

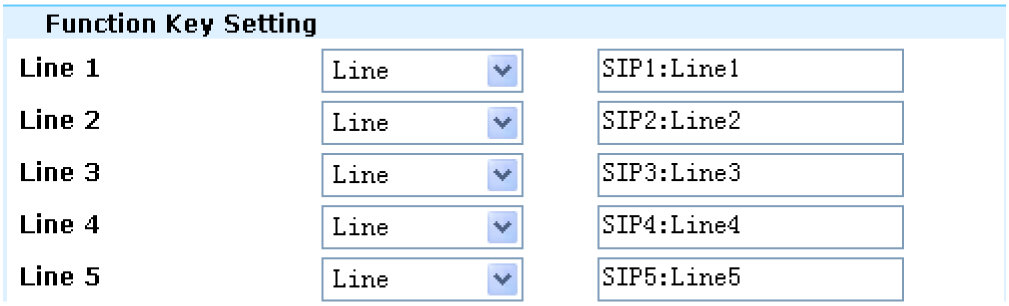
7.4 Function Key

PH806 IP phone's Function key supports LINE, Memory Key, Key Event as well as 4 DTMF models.

- LINE: If you config the function key as SIP LINE via web, then you can select which SIP LINE to call, only successfully registered SIP Line can be selected and available.
- Memory Key: this mode can support Presence, BLF, Push to talk, MWI and other functions.
- Key Event: the user can set shortcut keys as their preferences;
- DTMF: set the number sent by DTMF;

Specific settings are as follows:

7.4.1 LINE



7.4.2 Memory Key

BLF, presence and speed dial functions can be achieved by memory key.

Command mode: number@lineX/option

Number: means the number you want to call or check.

lineX； means the sip line you are now using; Optional: SIP1-SIP5.

Option: mainly refers to the 4 parameters such as /b,/p,/f and /i.

For example: 1800@2/p or 1800@2/b

/b means BLF（Busy Lamp Field）。

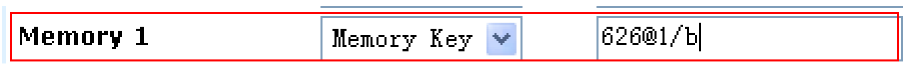
/p means Presence, via this config, users can subscribe other users' presence status.

/f means Speed dial;

/i means Push to talk, via this config on any key, users can achieve push to talk function. Usually, it can default to be the number of a secretary or the receptionist so that they can use this "push to talk" to answer the call, instead of picking up the phone.

7.4.2.1 Busy Lamp Field (BLF)

Please check the config as the following picture shows:



Idle state(successfully registered but not engaged), LED display long green light; ring state, LED flash red; in call or unavailable state, LED long red light.

7.4.2.2 Presence

That is, the phone can check the corresponding phone number's current state.（idle, ring, busy）.Please check the config as the following picture shows:

Memory 2	Memory Key	628@1/p

Online and idle state, LED display long green light; Offline state, LED display long red light; ringing or busy state, LED flash green light.

7.4.2.3 Speed dial

In standby mode, press the key then the phone calls will be put through directly to the speed-dial number.

Memory 1	Memory Key	2006@2/f

7.4.2.4 Push to talk

In standby mode, keep pressing this key to call somebody without his/her answering the phone; let go the key to hang up the call.

Memory 3	Memory Key	1800@1/i

You can config push to talk function as the picture shows above. Just keep pressing this key in standby mode and call 1800, the other side will be put through automatically (sip header takes Call-Info: answer-after=0);

Note: Automatic answering function need supports from callee's phone features. PH806 is supported.

7.4.3 Key Event

Memory 2	Key Event	F_PBOOK
Memory 3	Key Event	F_DND
Memory 4	Key Event	F_MWI
Memory 5	Key Event	F_REDIAL
Memory 6	Key Event	F_CALLERS
Memory 7	Key Event	F_CFWD

- F_PBOOK: shortcut key to config phonebook;
- F_REDIAL: shortcut key to redial;
- F_DND: shortcut key for no disturb;
- F_MWI: MWI shortcut key to check the quantity of old and new message;
- F_CFWD: call forward shortcut key;
- F_CALLERS: shortcut key for call record;

7.4.4 DTMF

After configuration, the phone will send out the number by DTMF mode.

8 Basic Call Functions

8.1 Making calls

1) Select LINEx(LINE 1, 2, 3, 4, LINE5) key, then dial the number and end the dialing by # or dial key to choose the line you want to make outgoing call.
2) Press hands-free key in stand-by mode, then dial the number. It defaults outgoing call from SIP1.
3) Press dial key in stand-by mode, then the screen will display the number you called. Select a number to press Dial or hands-free key to put through the call/
4) Press hands-free key in stand-by mode, and select the kept number from Mrmory Key to make an outgoing call fast.
5) Firstly press Memory Key, select a number you want to dial from the screen then press Dial or hands-free key to put through the call.

6）Press Headset to use the earphone.

8.2 Answering calls

1）Press Answer key（softkey1）to answer the call; press Reject key（softkey3）to reject the call.
2）Press hands-free key to answer the call.
3）Pick up the headphones to answer the call.
4）Press HEADSET to use the earphones to answer the call.
5）Select a LINE key to answer the incoming call accordingly.

8.3 HOLD

8.3.1 Call waiting

For users in the course of call, they may want to handle an emergency, but don't want to get heard from the other on the phone, so they can press 

button to hold the call temporarily and press HOLD button to release the call holding and be back into the call state.

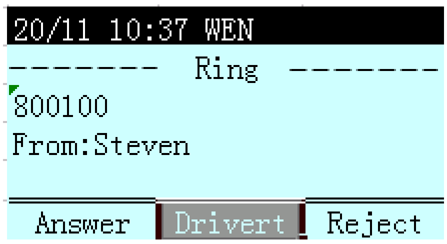
Note: In this mode, users cannot make a new outgoing call. You need to press **HOLD** button to realse the call holding and hang up the phone to continue.

8.3.2 Call holding

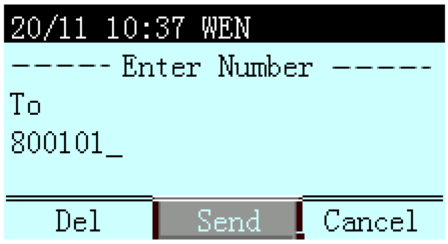
For users in the course of call, they can press CONFERENCE button to hold the current call temporarily to make a new call. When turns back to the first call, the second call will be holded also. Use the Switch button in LCD to switch between these 2 calls. Besides, these 2 calls can be ended separately or at the same time.

8.4 Call transfer

8.4.1 Forward



Step one



Step two

When there comes the incoming call, press Drivert when the phone rings, then enter the number you want to call and press Send. This incoming call will be forwarded automatically. Therefore, if you don't answer a call, use this function to forward the call without caller's awareness.

8.4.2 Blind transfer

When you get a call from A, but A wants to talk with B. Then you can press **Transfer** button, input B's number and end with # button to transfer the call and make A puts through with B.

Note: It can be applied when there comes a new call when you are in the course of call already. In this situation, press **Switch (softkey1)** to switch between the two calls or press **Transfer** to transfer the call.

8.4.3 Attend Transfer

When you get a call from A, and A wants to talk with B. But B might not in the office or busy with something else, you can press **Conference** button to call B to check if B is available. If yes, then you can press **Transfer** to forward the call to make A put through with B; If not, just press Close to end your call with B and the call will be back to A's talking status.

8.5 Three Way Call

When you are talking with A and want to get C in the call also, you can press CONFERENCE button to hold the call with A temporarily and then call C, press CONFERENCE again to make 3-way conference call.

8.6 Advanced paging / intercom

This function works with the SIP server such as asterisk system. When a user call a special number, the system will the invite requests to all users belonging this group and the phone's hands free function will be enabled automatically and turn off the mic but listen to what the caller broadcast.

It is necessary that the SIP Header field from the invite requests contains Alert-Info: answer-after=0. It works with the asterisk server. Such as user can config SIPAddHeader(Alert-Info: answer-after=0) in the dialing rules.

8.7 Instant short message

Press SMS button(softkey1) in stand-by mode, **SMS à Add à** input the message text **à Send à** input the number **à Send**(LINEx）, then the short message will be sent to the inserted number via SIP1. If you press LINEx to send the message, then the message will be sent via your selected SIP LINE.
Note: This function need the server support SMS. Otherwise, the server will send message 405.

8.8 Call Record

In the course of call, if user wants to keep and record the call, just press REC key and the phone will send a string of pre-configured DTMF to the server. Call record begins when the server receives DTMF.

REC: This function requires server's support. Then user can config Record key in the menu SIP Setting->Call Service, (Record key is used for server to receive DTMF), Call record begins when the server receives DTMF.

8.9 MWI and Play

u When receive a voice message, the MWI light will flash to show that a new voice message is received and kept unread.

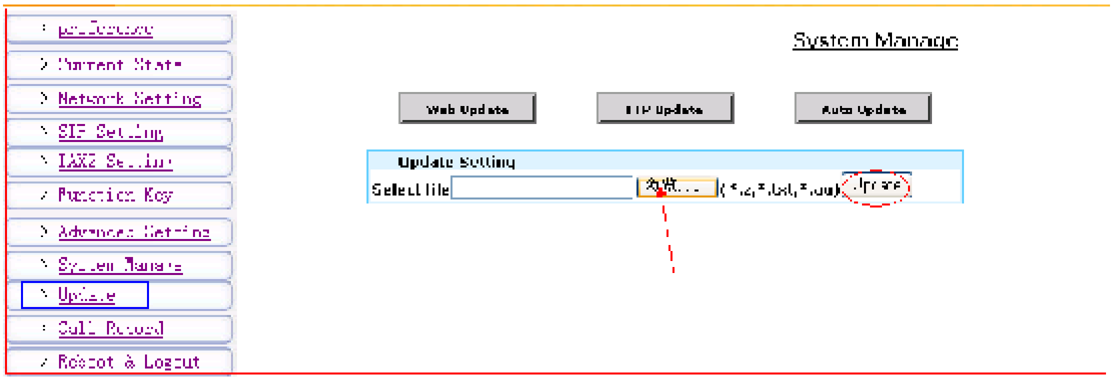
To enable this function, user should Enable Subscribe function on SIP-SIP config page then the phone will subscribe this service from the server. When there comes a new message, the phone can get notice from the server directly. If your server doesn't support this function, just make it unable.

u **PLAY:** To use this function, user should config MWI Number（mailbox number）on SIP-Call service page. When the account receives a voice mail, the MESSAGE LED will flash. And users just press PLAY key in stand by mode to listen to the message since the phone will call the voice mail access number automatically.

8.10 Customized Ringtones

Download your favorite MP3 music, and use the music format converter tool to change the music into au format which can be identified by the phone. One important thing is that the name should be in 1.au, 2. au and should set USER1, USER2's ringtone correspondingly. Then upgrade the ringtone via WEB and config on SIP page's ring type as user1, user2.

9 Software Upgrade



Browser to find previously saved configuration file (or files provided files), download to current phone, which saves the configuration one by one. You can download system upgrade files, ringtone files and mmiset files on this page. Note that for ringtone files, do config the name as 1.au, 2.au, correspondingly USER1, USER2 on ring type menu. Finally click 【**Update**】 activate the configuration.