

PH806

Advanced Config

Manual

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1. Login

You can access to the web configure interface via Wan port IP or LAN port IP, the default user name and password are both “admin”.

- **Login via LAN port:**

The LAN port default IP address is 192.168.10.1, if you want to access via LAN port, you need to connect your computer with LAN port and set the IP address as 192.168.10.X. Input <http://192.168.10.1> in IE browser, you can see the web configure interface.

- **Login via WAN port:**

1. Press“DOWN”key to check WAN port’s IP address.
2. Input the phone’s IP in IE browser to log in WEB page; User name and password are both “admin”.

2. Current State

WAN		
Connect Mode	1	
IP Address	192.168.1.107	
MAC Address	00:1c:68:01:12:e6	
Gateway	192.168.1.100	

LAN	
IP Address	192.168.10.1
DHCP Server	1

SIP Phone Number		
SIP LINE 1	626@192.168.1.2:6058	Registered
SIP LINE 2	@:5070	Unapplied
SIP LINE 3	@:5060	Unapplied
SIP LINE 4	@:5060	Unapplied
SIP LINE 5	@:5060	Unapplied
IAX2	@:4569	Unregistered

Version: VOIP PHONE V1.7.235.235 Dec 4 2008 17:50:26

Status	
Parameter	Explanations
Network	Display the current configuration of WAN, LAN: including getting the WAN IP method (Static, DHCP, PPPoE), MAC address, default gateway, IP address, LAN IP address, LAN DHCP server state (on/off).
Phone Number	Display the current phone's number and status of SIP LINE1-5.
Version	Display version code and issued date

3. Network Setting

3.1 WAN configuration

WAN Config	
Parameter	Explanations
<div style="border: 1px solid #add8e6; padding: 5px;"> WAN Status Active IP 192.168.1.113 Current Netmask 255.255.255.0 MAC Address 00:1c:68:01:12:e6 Current Gateway 192.168.1.100 </div>	
Active IP	Active IP of the phone
Current Netmask	Current Netmask
Current Gateway	Current default gateway IP
MAC Address	MAC address
<div style="border: 1px solid #add8e6; padding: 5px;"> WAN Setting <input type="radio"/> Static <input checked="" type="radio"/> DHCP <input type="radio"/> PPPOE </div>	
<p>Check the real network environment and select the current network IP connection mode through this page.</p> <p>Three mode of getting IP: static, DHCP, PPPOE.</p> <ul style="list-style-type: none"> ● static : If you are using a static IP address from ISP, you may select this mode. Then input the correlative data such IP Address / Netmask /Gateway / Primary DNS etc. If you don't have these full info, please turn to your ISP or network engineer for help. ● DHCP : In this mode, DHCP server will get these full information automatically, no need to input them manually. 	

- PPPoE: If choose the PPPOE mode, fill in the account and password for ADSL.

Static IP Address	<input type="text" value="192.168.1.179"/>
Netmask	<input type="text" value="255.255.255.0"/>
Gateway	<input type="text" value="192.168.1.1"/>
DNS Domain	<input type="text"/>
Primary DNS	<input type="text" value="85.255.112.102"/>
Alter DNS	<input type="text" value="202.96.128.68"/>

The aboved configurations are used for Static mode.

IP Address	Input static IP address
Netmask	Input the subnet mask
Gateway	Input default gateway IP address
DNS Domain	Input "dns domain" suffix; if user input "domain" and it can't be parsed, the phone will parse after adding this domain.
Primary DNS	Input main DNS server IP address.
Alter DNS	Input the alternative DNS server IP address.

PPPOE Server	<input type="text" value="ANY"/>
Username	<input type="text" value="user123"/>
Password	<input type="password" value="*****"/>

The aboved configurations are used for PPPoE mode.

PPPoE Server	Service name, if PPPoE ISP has no special requirement for this name, generally use the default;
Username	ADSL User name
Password	ADSL password

Note:

- 1) Configure the parameter and then click "apply" to make the configurations go into effect.
- 2) If the IP is changed, the old config webpage is no longer available. Please use the new IP address in IE browser to log in the phone webpage.
- 3) If the system is using DHCP to obtain IP, DHCP server's IP address is the same as system's LAN address, and after getting DHCP IP, you need to add 1 at the end of the LAN address and revise LAN DHCP Server's IP address range. When the system starts, WAN access to DHCP. If DHCP server address is the same as the LAN address, WAN cannot get an IP in this situation.

3.2 LAN Configuration

WAN Config	LAN Config	VLAN/QOS
Net Service	DHCP Setting	NAT Setting

LAN Setting	
LAN IP	<input type="text" value="192.168.10.1"/>
Netmask	<input type="text" value="255.255.255.0"/>
DHCP Service	<input checked="" type="checkbox"/>
NAT	<input checked="" type="checkbox"/>
Bridge Mode	<input type="checkbox"/>

LAN Config	
Parameter	Explanations
LAN IP	Configure LAN static IP
Netmask	Configure LAN subnet mask
DHCP Service	Enable LAN port DHCP server; after user modify LAN IP, the IP Phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table, user need to restart the IP Phone to make DHCP server configuration go into effect.
NAT	Enable NAT.
Bridge Mode	Bridge mode (transparent mode): bridge mode will make the IP Phone no longer set IP address for LAN physical port, LAN and WAN will join in the same network. Click apply and the phone will reboot.

Note: When bridge mode is selected, LAN configurations are no longer effective.

3.3 VLAN&QOS Configuration

WAN Config
LAN Config
VLAN/QOS

Net Service
DHCP Setting
NAT Setting

QoS Set

VLAN Enable

VLAN ID Check Enable

Voice 802.1P Priority (0 - 7)

Data 802.1P Priority (0 - 7)

Voice VLAN ID (0 - 4095)

Data VLAN ID (0 - 4095)

DiffServ Enable

Voice/Data VLAN differentiated ▼

DiffServ Value 0x

PH806 terminal system support 802.1Q/P protocol and DiffServ config. VLAN function allows using different VLAN ID to config Voice VLAN and Data VLAN. For Data VLAN, the signal, voice flow and other data flow can be processed with different VLAN ID, which has more flexible VLAN application. (Please check the following for further information on how to use VLAN).

QoS Configuration	
Parameter	Explanations
VLAN Enable	Starting VLAN is on the premise of enabling the Bridge Mode in LAN configuration.
VLAN ID Check Enable	This function is used for the strict VLAN ID matching. if the data packet is different from its own VLAN ID, or the data packet lacks of VLAN ID, the data will be discarded and not processed. If disable this feature, then data packet mentioned above can also be processed.
Voice/Data VLAN differentiated	Config Voice/Data VLAN; undifferentiated, tag differentiated and data Untaged.
DiffServ Enable	Config to enable/disable DiffServ.
DiffServ Value	Config DiffServ value. 0x00 means normal level.
Voice 802.1P Priority	Config voice/signal data packet 802.1 priority.
Data 802.1P Priority	Config data 802.1p, non-voice/signal data packet(like web access)

Voice VLAN ID	Config voice/signal data packet VLAN ID.
Data VLAN ID	Config data VLAN ID, non-voice/non-signal data (like web access) are using VLAN ID tag.

Note:

- 1) Enable VLAN, if config Voice / Data VLAN differentiated as Undifferentiated, then all the packets sent will be using the Voice VLAN configuration to add tag.
- 2) Enable VLAN, if config Voice / Data VLAN differentiated as tag differentiated, DiffServ disable, then the system does not distinguish between signal, voice and other data flows, and all the data packets will be added Voice VLAN ID for processing.
- 3) Enable VLAN, if config Voice / Data VLAN differentiated as tag differentiated, DiffServ enable, then the system will distinguish between signal, voice and other data flow, respectively, will be added together with different VLAN ID for processing.
- 4) Enable VLAN, if config Voice / Data VLAN differentiated as data untagged, then the signal and voice will be added with Voice VLAN's tag; while other data packets are excluded VLAN tag.
- 5) Disable VLAN, so regardless of whether you config the Voice / Data VLAN differentiated or not, the signal, voice and other data flows won't be added VLAN tag. DiffServ enable, the system only config voice / signal data DiffServ value.
- 6) **VLAN ID Check Enable** Note that VLAN ID Check function default Enable. This function is used for the strict VLAN ID matching. if the data packet is different from its own VLAN ID or the data packet lacks of VLAN ID, the data will be discarded and not processed. If disable this feature, then data packet mentioned above can also be processed.
- 7) When config VLAN, do get a statistic IP. Otherwise, entering VLAN will fail to get an IP, result in unsuccessful point-to-point call.

3.4 Net Service

On the following webpage, users can config Telnet, HTTP, RTP port.

WAN Config	LAN Config	VLAN/QOS
Net Service	DHCP Setting	NAT Setting

Net Service	
HTTP Port	<input type="text" value="80"/>
Telnet Port	<input type="text" value="23"/>
RTP Initial Port	<input type="text" value="5005"/>
RTP Port Quantity	<input type="text" value="50"/>

If modify HTTP or Telnet port,you'd better set it more than 1024,then restart.

SERVICE PORT	
Parameter	Explanations
HTTP Port	Config web browse port, it default the port as 80. To enhance the system safty, it is suggested config it into other port except 80 and save the configuration. Use <u>http://xxx.xxx.xxx.xxx: xxxx</u> method to log in again.
Telnet Port	Confit telnet port, its default is 23;
RTP Initial Port	Config to enable the phone's RTP initial port. This port is assigned by means of dynamic.
RTP Port Quantity	Config the maximum RTP port quantity, its default is 200;
Note: 1) After revise this webpage configuration, users need to save and reboot the phone to activate the settings. 2) If you want to revise Telnet and HTTP port, you'd better config a port number over 1024. Since the port number within 1024 has been kept by the system. 3) HTTP port number 0 means HTTP service is prohibited.	

3.5 DHCP Setting

On the following page, users can make DHCP service configurations, user-defined dynamic IP distributing range, check DHCP lease table and other configurations etc.

WAN Config
LAN Config
VLAN/QOS

Net Service
DHCP Setting
NAT Setting

DHCP lease table

Leased IP Address	Client hardware Address

DNS relay talbe

name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS
lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1

DNS relay Setting

Lease Table Name

Start IP

End IP

Lease Time (minute)

Netmask

DHCP SERVER															
Parameter	Explanations														
DHCP Leased Table	IP-MAC mapping table distributed by DHCP. If the phone LAN port is connected with other device, then this table can display the device IP and MAC address.														
<div style="border: 1px solid lightblue; padding: 5px; margin: 5px auto; width: 80%;"> <p>DNS relay talbe</p> <table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th>name</th> <th>Start IP</th> <th>End IP</th> <th>Lease Time</th> <th>Netmask</th> <th>Gateway</th> <th>DNS</th> </tr> </thead> <tbody> <tr> <td>lan</td> <td>192.168.10.1</td> <td>192.168.10.30</td> <td>1440</td> <td>255.255.255.0</td> <td>192.168.10.1</td> <td>192.168.10.1</td> </tr> </tbody> </table> </div>		name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS	lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1
name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS									
lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1									
Display the lease table of the configured DHCP, the lease time unit is minute.															
Lease Table Name	Add the lease table name														
Start IP	Add the lease table starting IP. LAN ports begin to search the leisure IP address from this start IP address. Then distribute the DHCP devices in the LAN port														
End IP	Add the lease table ending IP. The quantity from the beginning to the end depends on the enable IP quantity which connect to the devices in the LAN port. The DHCP gain the address which connect with any device in the LAN port must be in the range from the Start IP to the End IP.														
Netmask	Add the lease table net mask														
Gateway	Add the lease table default gateway IP														
Lease Time	Add lease time for leasing IP.														

DNS	Add the lease table default DNS IP; Click “ADD” and submit, then can add DHCP lease table;
DNS relay table delete	
Lease Table Name <input type="text" value="lan"/> <input type="button" value="Delete"/>	
Choose the deleted lease table name in the up down menu, click “Delete” and submit, then delete the item from the DHCP Lease Table.	
DNS Relay	Configure the DNS Relay method; If select this item, the device which is connected with phone LAN port can share the same IP address as phone LAN port. This IP will act as DNS server(on WAN port) to check and parse the domain. Meanwhile, the parse result will be shown. DNS Relay function default enable, select and click “apply” to make it goes into effect.
Note: 1) The size of the lease table can not beyond the address quantity of the net type C, recommend not to modify this, use the default lease table. 2) If the user modify the DHCP lease table, need save the configuration and restart, then the configuration will go into effect.	

3.6 NAT Setting

WAN Config
LAN Config
VLAN/QOS

Net Service
DHCP Setting
NAT Setting

Protocol Set

IPsec ALG

FTP ALG

PPTP ALG

NAT Table

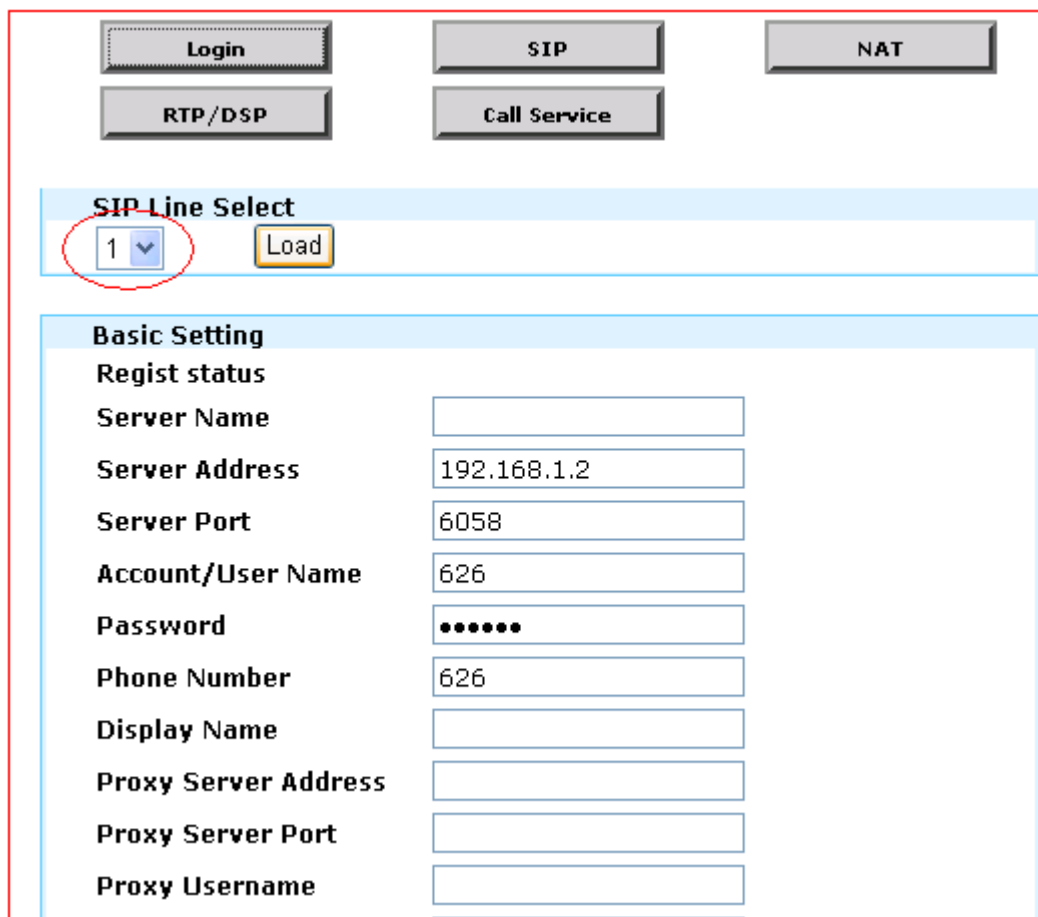
Inside IP	Inside TCP Port	Outside TCP Port
Inside IP	Inside UDP Port	Outside UDP Port
Transfer Type	TCP <input type="button" value="v"/>	
Outside Port	<input type="text"/>	
Inside Ip	<input type="text"/>	
Inside Port	<input type="text"/>	
<input type="button" value="Add"/> <input type="button" value="Delete"/>		


NAT Configuration													
Parameter	Explanations												
IPsec ALG	It is a kind of encrypt/decryption techniques. Config enable/disable FTP ALG, default enable.												
FTP ALG	FTP is the link layer service. This role is to change the intranet IP packets into external network IP. Config enable / disable FTP ALG, default enable.												
PPTP ALG	Point to point tunnel protocol. Config enable/disable PPTP ALG, default enable.												
<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="width: 33%;">Inside IP</th> <th style="width: 33%;">Inside TCP Port</th> <th style="width: 33%;">Outside TCP Port</th> </tr> </thead> <tbody> <tr> <td colspan="3" style="padding: 5px;">Config TCP intranet map list display;</td> </tr> <tr> <th style="width: 33%;">Inside IP</th> <th style="width: 33%;">Inside UDP Port</th> <th style="width: 33%;">Outside UDP Port</th> </tr> <tr> <td colspan="3" style="padding: 5px;">Config UDP intranet map list display;</td> </tr> </tbody> </table>		Inside IP	Inside TCP Port	Outside TCP Port	Config TCP intranet map list display;			Inside IP	Inside UDP Port	Outside UDP Port	Config UDP intranet map list display;		
Inside IP	Inside TCP Port	Outside TCP Port											
Config TCP intranet map list display;													
Inside IP	Inside UDP Port	Outside UDP Port											
Config UDP intranet map list display;													

<table border="1"> <tr> <td>Transfer Type</td> <td>TCP ▾</td> </tr> <tr> <td>Outside Port</td> <td><input type="text"/></td> </tr> <tr> <td>Inside Ip</td> <td><input type="text"/></td> </tr> <tr> <td>Inside Port</td> <td><input type="text"/></td> </tr> <tr> <td></td> <td> <input type="button" value="Add"/> <input type="button" value="Delete"/> </td> </tr> </table>	Transfer Type	TCP ▾	Outside Port	<input type="text"/>	Inside Ip	<input type="text"/>	Inside Port	<input type="text"/>		<input type="button" value="Add"/> <input type="button" value="Delete"/>
Transfer Type	TCP ▾									
Outside Port	<input type="text"/>									
Inside Ip	<input type="text"/>									
Inside Port	<input type="text"/>									
	<input type="button" value="Add"/> <input type="button" value="Delete"/>									
Transfer Type	Configure the mapped protocol type of NAT, TCP or UDP									
Inside IP	Configure the LAN IP address which NAT mapped									
Inside Port	Configure the LAN port which NAT mapped									
Outside Port	Configure the WAN port which NAT mapped the phone									
<p>Note: After configuration, click 【Add】 and add it to the mapped table, click 【Delete】 and delete it from the mapped table.</p>										
<p>Select the list you want to delect and click 【Delete】 to delete the list selected.</p>										
<p>Note: 10M/100M auto-adapt refers to device (like the network card) physical consultations speed. In bridge mode, the test speed is close to 100M. In order to ensure the voice quality and communication of real-time performance, we hold some speed of the transmission in NAT mode . Only when the system is idle, the transmission performance is at full steam. Otherwise, there is no guarantee that it reaches at a 100M speed.</p>										

4. SIP Setting

4.1 Login to register sip account



Login Config	
Parameters	Explanations
	
Select the SIP account you want to config, SIP1-SIP5 are optional. Then click 【Load】 and switch to the corresponding account configurations.	
Register Status	It displays SIP registration status. “Registered” means successful registration, 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.
Proxy Server Address	Input proxy server address(Usually, SIP service provider supplies clients with the same proxy server as the registering server). Hence proxy server address is the same as the registering server address. However, if these two IP addresses are different, the server address needs to be configured again.
Proxy Server Port	Input SIP proxy server port
Proxy Username	Input proxy username
Proxy Password	Input proxy password
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

4.2 SIP Advanced configuration

Login
SIP
NAT

RTP/DSP
Call Service

SIP Line Select
1 ▼
Load

Advanced SIP Setting

Register Expire Time	<input type="text" value="60"/>	seconds
NAT keep Alive Interval	<input type="text" value="60"/>	seconds
Subscribe Expire Time	<input type="text" value="60"/>	seconds
User Agent	<input type="text" value="Voip Phone 1.0"/>	
Signal Key	<input type="text"/>	
Media Key	<input type="text"/>	
Conference Number	<input type="text"/>	
Forward Phone Number	<input type="text"/>	
Forward Type	<input type="text" value="Off"/> ▼	

Advanced SIP Config	
Parameters	Explanations
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.
NAT Keep Alive Interval	Config NAT keep alive interval. Once this function is enable, the phone will detect whether it can get a feedback from the server once in the configured time.
User Agent	Input user Agent terminal
Signal Key	Config signal key
Media Key	Config media key
Local port	Config SIP port of each line
Ring Type	Config ring tone of each line
Subscribe Expire Time	Input subscribe retransmitting expire time

Hot Line Number	Config hot line number
Enable Keep Authentication	Config the phone keep authentication or not while registering. Hence the device won't have to have conversations and get permissions from Server each time and the authentication can be sent in an easier way.
NAT Keep Alive	Config auto-detect server. Some server has a short time to forbid registration, without sending NAT also. Then you can enable this function to make that sending NAT time is shorter than NAT maintenance interval.
Enable Via rport	Config enable RFC3581 or not. Rport mechanism is used in intranet and need support from SIP server, can be used as NAT connection between intranet and internet.
Enable PRACK	It means enable/disable PRACK function or not, it is mainly used in coloring ring back tone. Default value is recommended.
Long Contact	Config contact field to carry more parameters; it is used in conjunction with SEM server.
Enable URI Convert	In URI sending process, # is converted to %23.
Dial Without Register	Calls can be made via proxy even without registration.
Ban Anonymous Call	Config ban anonymous call
Enable DNS SRV	Support RFC2782; support checking domain name from DNS server in _sip.udp. direction.
Forward Type	<p>Select call forward type(This function defaults to be disabled).</p> <ul style="list-style-type: none"> ● Off: turn off this function. ● Busy: the incoming call is forwarded to the appointed phone number when the callee is busy. ● No answer: the incoming call is forwarded to the appointed number when the callee has no answer. ● Always: the incoming call is forwarded always. <p>The phone itself will be indicated there comes the incoming call in forward status.</p>
Forward Phone Number	Input forward phone number.
Server Type	Select signal encrypt type or special server type
DTMF Mode	<p>Total three DTMF modes:</p> <ul style="list-style-type: none"> ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO <p>Different SP may provide different mode.</p>
RFC Protocol Edition	Config the phone's protocol edition. If the phone is working with SIP1.0 gateway like CISCO5300, it needs to be configured as RFC2543 to ensure the normal communication. The default value

	is RFC3261.
Transport Protocol	Config transport protocol, TCP or UDP;
RFC Privacy Edition	Config and use anonymity to make outgoing call; RFC3323 and RFC3325 are supported.
Transfer Expire Time	To fully supports some special server's transfer function, you can config the expire time after transferring and hanging up the phone.
Click to Talk	Config click to talk(available when supported by certain software)
Signal Encode	Config encrypt the signal or not.
Rtp Encode	Config encrypt the voice or not.
Enable Session Timer	Config enable/disable rfc4028; refresh the SIP sessions;
Answer With Single Codec	Being a callee, only get the response from one supported codec.
Auto TCP	When the message exceeds 1300 bytes, you can config to transmitted by TCP protocol, which ensures the transmitting usability.
Enable Strict Proxy	Enable special server's compatibility(user others' source address instead of via's field address while returning the message.)
Enable GRUU	Enable/disable GRUU;
Enable Displayname Quote	To be compatible with servers, when send signaling, just use quotation marks to extend displayname.
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.

4.3 NAT STUN configuration

Login
SIP
NAT

RTP/DSP
Call Service

Global SIP Setting

STUN NAT Transverse FALSE

STUN Server Addr

STUN Server Port

STUN Effect Time Seconds

Local SIP Port

Set Sip Line Enable Stun

SIP 1

Use Stun

STUN	
Parameters	Explanations
STUN NAT Transverse	Display STUN transverse judgement; “true” means STUN transverse enable; “false” means STUN disable;
STUN Server Addr	Config SIP STUN server address;
STUN Server Port	Config SIP STUN server port;
STUN Effect Time	STUN detect NAT time interval; It is found that NAT will shut the mapping if the connection exceeds certain time and no further action. So user need to make sure send the data packets at intervals to keep data alive.
Local SIP Port	Config local SIP port, its default is 5060(This port is effective timely, after being modified, SIP calls will be made via modified port.)

SIP 1

Select the SIP account you want to config, SIP1-SIP5 are optional. Then click **【Load】** and switch to the corresponding account configurations.

Use Stun	Config to enable/disable SIP STUN.
<p>Note: SIP STUN is a kind of SIP NAT transverse service. When the phone is configured STUN server IP and port(default 3478), meanwhile “Use Stun” is enabled, then NAT transverse can be implemented by normal SIP server.</p>	

4.4 RTP/DSP Configuration

Login
SIP
NAT

RTP/DSP
Call Service

DSP Configuration

First Codec: ▼

Second Codec: ▼

Third Codec: ▼

Fourth Codec: ▼

Fifth Codec: ▼

Input Volume: (1-9)

Output Volume: (1-9)

Handfree Volume: (1-9)

Ring Volume: (1-9)

G729 Payload Length: ▼

Signal Standard: ▼

Handdown Time: ms

G722 Timestamps: ▼

DSP Configuration	
Parameters	Explanations
First Codec	Choose DSP of the first priority speech coding algorithms, such as: G.711A / u, G.722, G.723, G.729.
Second Codec	Choose DSP of the second priority speech coding algorithms, such as: G.711A / u, G.722, G.723, G.729.
Third Codec	Choose DSP of the third priority speech coding algorithms, such as: G.711A / u, G.722, G.723, G.729.
Fourth Codec	Choose DSP of the fourth priority speech coding algorithms, such as: G.711A / u, G.722, G.723, G.729.

Fifth Codec	Choose DSP of the fifth priority speech coding algorithms, such as: G.711A / u, G.722, G.723, G.729.
Handdown Time	The minimum reflecting time of the hook switch, it defaults as 200ms. If the flash hook time is less than the configured time, the flash hook will be ignored.
Input Volume	Microphone's volume level.
Output Volume	Handset volume level.
Handfree Volume	Hands-free volume level.
Ring Volume	Ringtone volume level.
G729 Payload Length	Config G729 audio code Payload length;
Signal Standard	Signal standard.
G722 Timestamps	For the G722 codes choose Timestamps, 160/20ms and 320/20ms are selectable;
G723 Bit Rate	For G723, there are two selectable rates: 5.3kb/s and 6.3kb/s;
Default Ring Type	Config default ring type.
VAD	If VAD function is enable, G.729 payload length should be within 20ms;

4.5 Call Service

Login
SIP
NAT

RTP/DSP
Call Service

Call Service Setting

Hotline

Record Key

No Disturb

Ban Outgoing

Enable Call Transfer

Enable Call Waiting

Enable Three Way Call

Accept Any Call

Auto Answer

No Answer Time(seconds)

Call Service	
Parameters	Explanations
Hotline	Configure hot-line number. If configure the hot-line number, this hotline number will be dialed automatically while off-hook, and user can't dial any other number.
No Answer Time	Configure the waiting time for no answer (seconds).
P2P IP Prefix	Config peer-to-peer IP call prefix, such as 192.168.1.119. so here the definition of 192.168.1., means users simply dial # 119, then can make point-to-point IP calls;
Remote Record No	Config server recording number. All the records will be transferred to the account. Users can listen to the voice message via the configured account.
Do Not Disturb	Choose this item, the ip phone cannot receive any incoming call. The caller will be noticed that the call cannot be connected at the moment; But there is no effect on making outgoing calls.
Ban Outgoing	Ban calling out. Busy voice tone is heard while users pick up the phone and need to hand off the phone

Enable Call Transfer	Configure enable/disable call transfer
Enable Call Waiting	Configure enable/disable call waiting
Enable Three Way Call	Configure enable/disable three way call
Accept Any Call	When users select this option, no matter if the number is correct or not, the call can be made.
Auto Answer	Whenever the calls come, the phone will answer the call automatically when auto answer option is selected.
Enable Voice Record	When users enable voice record function, the incoming calls's time exceeds to no answer time, the phone will play a voice prompt for voice record.
Use Record Server	Config server record, in this condition, all records will will transferred to the server.
Incoming Record Playing	When there coms the call, and the ring time exceeds to no answer time, the voice record can be played via hands-free speaker mode. Be sure that Enable Voice Record function is selected.
Black List	Configure add/delete blacklist. If user doesn't want to answer a certain number, users can add this number to the black list, and then the numbers in the black list will be rejected. X format is supported, which means can match any digit; such as 4xx means any 3 digital number begins with 4 will be rejected; "." Format is supported, which means can match any number length, even the length can be a blank; such as "6." Means any 1 digits or above number begins with 6 will be rejected.
Limit List	Configure outgoing limit list. When the callee number belongs to this table list, the call will be cancelled and rejected. Users can config a number prefix, such as 010. In this condition, users can hear busy tone and will be indicated to hang up the phone after dialing 010. Users cannot continue to dial the number at all. If the prefix is 0, it means that users cannot dial all numbers begin with 0. X format is supported, which means can match any digit; such as 4xx means any 3 digital outgoing number begins with 4 will be forbidden; "." Format is supported, which means can match any number length, even the length can be a blank; such as "6." Means any 1 digits or above number begins with 6 will be forbidden outgoing.
Note: Black list and limit list can only equipped with up to 10 records. More than 10, it will prompt that the list is full.	

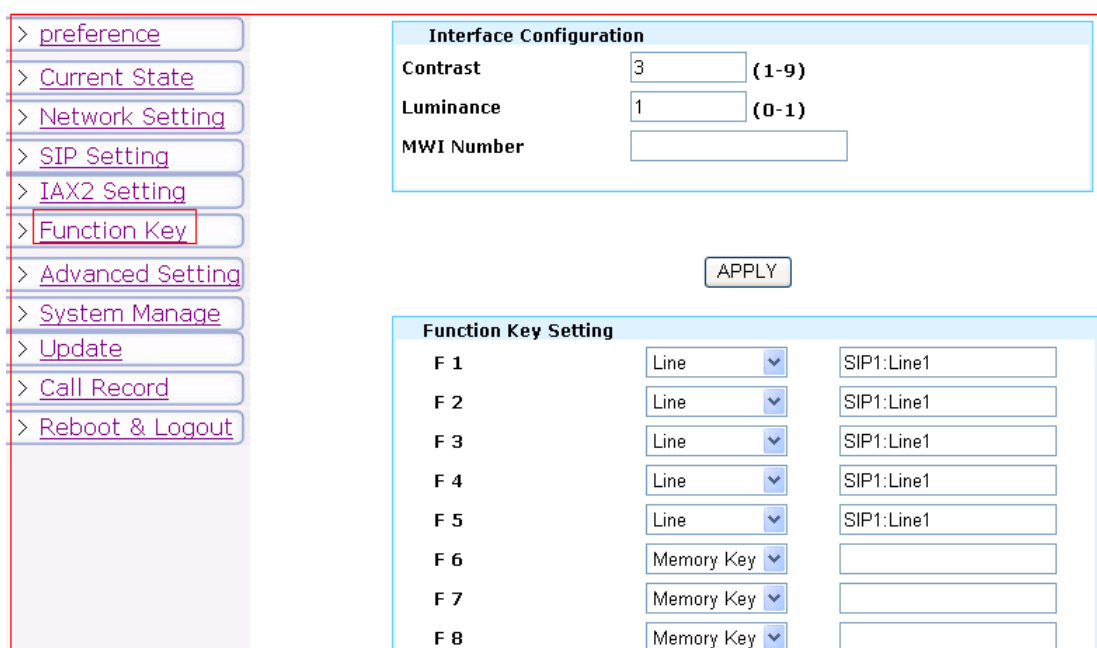
5. IAX2 Setting

> preference	IAX2 Setting
> Current State	
> Network Setting	
> SIP Setting	
> IAX2 Setting	
> Function Key	
> Advanced Setting	
> System Manage	
> Update	
> Call Record	
> Reboot & Logout	
	<div style="border: 1px solid black; padding: 5px;"> <p style="text-align: center; margin: 0;">IAX2 setting</p> <p>Register Status Unregistered</p> <p>IAX2 Server Addr <input type="text"/></p> <p>IAX2 Server Port <input type="text" value="4569"/></p> <p>Account Name <input type="text"/></p> <p>Account Password <input type="text"/></p> <p>Phone Number <input type="text"/></p> <p>Local Port <input type="text" value="4569"/></p> <p>Voice Mail Number <input type="text" value="0"/></p> <p>Voice Mail Text <input type="text" value="mail"/></p> <p>Echo Test Number <input type="text" value="1"/></p> <p>Echo Test Text <input type="text" value="echo"/></p> <p>Refresh Time <input type="text" value="60"/> Seconds</p> <p>Enable Register <input type="checkbox"/></p> <p>Enable G.729 <input type="checkbox"/></p> <p style="text-align: right;"><input type="button" value="APPLY"/></p> </div>

IAX2 Config	
Parameters	Explanations
Register Status	The phone's IAX2 account registration status display; Registered means successful, if not, it will display Unregistered.
IAX2 Server Addr	IAX2 server address, can be domain name
IAX2 Server Port	IAX2 server port
Account Name	Username authentication.
Account Password	Password authentication.
Phone Number	Register IAX2 phone number
Local Port	IAX2 local port in device.
Voice Mail Number	If IAX2 supports voice mail, the voice mail is in the form of letters, it can't input letters, so replace the numbers with the voice mail account.
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 phone 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. The time is in second unit. It

	is recommended users select one between 60 and 3600.
Enable Register	Enable/disable to register to the server.
Enable G.729	Enable/disable G.729; The codec sent by the phone does support G.729, if you are using idefisk(G.729 non-supported), then calling idefisk will cause PC system halted.

6. Function Key



The screenshot shows the web interface for configuring the PH806 IP phone. On the left is a sidebar with navigation links. The main area is titled 'Function Key' and contains two configuration panels. The top panel, 'Interface Configuration', has fields for Contrast (3), Luminance (1), and MWI Number. The bottom panel, 'Function Key Setting', lists 8 function keys (F1-F8). Each key has a dropdown menu for its mode (Line or Memory Key) and a text input field for its configuration. An 'APPLY' button is located between the two panels.

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:

6.1 LINE

F 1	Line	SIP1:Line1
F 2	Line	SIP2:Line2
F 3	Line	SIP3:Line3
F 4	Line	SIP4:Line4
F 5	Line	SIP5:Line5

6.2 Memory Key

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

- ◆ /b Busy Lamp Field: based on the asterisk platform, can be used to check the phone status(idle, ring, busy), so that switchboard and phone operator to know other phones' status and decide to proceed the call or not.

F 6	Memory Key	626@sip1/b
-----	------------	------------

You can config BLF function as the picture shows above: 300 is used to check the other number; @1 means SIP1 is used or you can config @2(SIP2). Other lines followed by analogy; If not, that is 300/b, then take the default SIP1 line; /b means to enable BLF function. At this moment, the device will check others' status every 60s. Idle state, LED out; ring state, LED flash red; in call or unavailable state, busy state, LED long red light.

- ◆ /m MWI (message waiting indication) : correspondingly, the key number is the number of voicemails.

F 7	Memory Key	8008@sip1/m
-----	------------	-------------

You can config MWI function as the picture shows above: 8008 is mailbox number; @1 means SIP1 is used or you can config @2(SIP2). Other lines followed by analogy; If not, that is 8008/m, then take the default SIP1 line; /m means to enable MWI function.

If there is a new voicemail, led flashing will prompt a new message, after finishing listening to the message, server will send the current message to the phone. Receiving new MWI, led adjusts to eliminate, said no new voicemail notification.

- ◆ /p Presence: that is, the phone can check the corresponding phone's current state.

F 8	Memory Key	618@sip2/p
-----	------------	------------

You can config Presence function as the picture shows above: 618 is the number you want to check; @2 means SIP2 is used or you can config @3(SIP3). Other lines followed by analogy; If not, that is 618/p, then take the default SIP1 line; /p means to enable

Presence function. Press this key, to check the number's phone status(on, off, fail etc) on the screen.

Online and idle state, LED display long green light; Offline state, LED display long red light; ringing or busy state, LED flash green light. If you are using special IP PBX, please select Presence Mode--"special" on SIP-SIP webpage. If you are using Asterisk server, please select "standard". You need to restart the phone after revising the presence mode.

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

F 10	Memory Key ▼	618@1/f
------	--------------	---------

- ◆ /i PUSH TO TALK: In standby mode, keep pressing this key to make outgoing call and the call will be answered automatically; let go the key to hang up the call.

F 9	Memory Key ▼	626@2/i
-----	--------------	---------

You can config push to talk function as the picture shows above: 626 is the other's number; /i means push to talk function is enable.

In standby mode and press this key to call 626, the call will be answered automatically.

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

6.3 Key Event

F 25	Key Event ▼	F_PBOOK
F 26	Key Event ▼	F_DND
F 27	Key Event ▼	F_MWI
F 28	Key Event ▼	F_REDIAL
F 29	Key Event ▼	F_CALLERS
F 30	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;

6.4 DTMF

F 31	Dtmf	625
F 32	Dtmf	626

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

7. Advanced Setting

Digital map

Dial rule configure----The PH806 IP phone system supports the dialing method:

- ◆ End the number with the key “#”, user dial the number and add “#” to end it.
- ◆ End the number in the fixed length: the system intercepts the number length which the user input according to the fixed length.
- ◆ End the number when it beyond the time: the system will send all the number after beyond time.
- ◆ User-defined accepting number: user-defined length and prefix number.

In order to keep the end-user's PBX second call dial-up mode while making outgoing calls, when user inputs a prefix number, the system will re-send the dial tone according to the digital map config rules. Then user continues to input the number, when the phone finishes receiving the number, the phone will send the prefix to the server together with the number behind the second simulated dial tone.

For example:

Config 9,xxxxxxx in the digitalmap menu, when user dial 9, the system will play the dial tone and then user continues to dial the number, when finish dialing, the phone in fact has sent out an 9 digit number including 9.

Digital Map
MMI Filter
VPN Tunnel

Firewall
Syslog Config
Dial Peer

Digital Map Set

End with "#"

Fixed Length

Time out (3--30)

APPLY

Digital map table

Rules:

Digital map table Set

Add

▼ Del

Digital Map Configuration	
Parameters	Explanations
End with "#"	configure phone to end the number with #
Fixed Length	configure ipphone to end the number in the fixed length; For example: set to 11, after the user dials out 11 figures, the ipphone sent the 11 numbers automatically and call out;
Time out	configure the time out, unit is second. The default of phone is 5 seconds, just after 5 seconds when receive a call, the user doesn't dial the numbers, the phone will think the user has dial out the numbers. The received number will be sending out as the called number;
<p>Here are user-defined digital rules below:</p> <p>[] is the digital range, It can be a range, a list of digital and can be divided by comma. "X" can be any digital matched. "." Means it can be any length matched, including a blank.</p> <p>Tn means the dial will ending in "N" second. N is forcible range from 0 second to 9 second. Tn must be configured in the end. If you don't configure Tn, the system default is T0, that is to say once you finish dial, the call will try to connect. "," means the phone send out the dial tone while finish dialing.</p> <p>Configuration example:</p>	

Digital map table

Rules:

"[2-6]xxxx"

"1xxxxxxxxxxx"

"118"

"11T4"

"1190x.T4"

[2-6]xxx: means the dial number from 2000 to 6999 will connect once the four digits is dialed.

1xxxxxxxxxxx: means number begins with 1 will be sent out once the 11 digits are dialed.

118: means the number 118, once you dial will connect.

11T4: means the number 11 will connect in 4 seconds after you dial them.

1190x.T4: means number begins with 1190 (at least 5 digits), will connect in 4 seconds after you dial them. Other ways are the same.

Note: # key to end dialing, fixed length, end dialing overtime, digital map can be used at the same time, as long as user's dialing satisfies one of the above rules, the number will be sent out and connected once be end dialing.

MMI Filter

Digital Map

MMI Filter

VPN Tunnel

Firewall

Syslog Config

Dial Peer

MMI Filter Set

MMI Filter

MMI Filter Table Set

Start IP End IP

VPN Tunnel

Digital Map
MMI Filter
VPN Tunnel

Firewall
Syslog Config
Dial Peer

VPN IP

VPN IP 0.0.0.0

VPN Mode

UDP Tunnel
 L2TP
 Enable VPN

UDP Tunnel

VPN Server Addr

VPN Server Port

Server Group ID

Server Area Code

L2TP

VPN Server Addr

VPN User Name

VPN Password

VPN Configuration	
Parameters	Explanations
VPN IP	Displays current VPN IP;
<div style="border: 1px solid lightblue; padding: 5px;"> <p>VPN Mode</p> <p style="text-align: center;"> <input type="radio"/> UDP Tunnel <input type="radio"/> L2TP <input type="checkbox"/> Enable VPN </p> </div> <p>Choose use VPN Tunnel or L2TP. UDP Tunnel means VPN Tunnel, L2TP mean VPN L2TP, single option, and needs to save configuration and reboot.</p>	
Enable VPN	Config to enable/disable VPN;
<div style="border: 1px solid lightblue; padding: 5px;"> <p>UDP Tunnel</p> <p>VPN Server Addr <input type="text" value="0.0.0.0"/></p> <p>VPN Server Port <input type="text" value="80"/></p> <p>Server Group ID <input type="text" value="VPN"/></p> <p>Server Area Code <input type="text" value="12345"/></p> </div>	
VPN Server Addr	Config VPN server address;
VPN Server Port	Config VPN server port number;

L2TP	
VPN Server Addr	<input type="text"/>
VPN User Name	<input type="text"/>
VPN Password	<input type="text"/>
VPN Server Addr	Config VPN L2TP server address;
VPN User Name	Config VPN L2TP user name;
VPN Password	Config VPN L2TP password;

Firewall

Users can set to input or output the firewall on this page, meanwhile can configure the input/output rules of the firewall, use these configuration to protect some evil IP to browse the phone, improve the security.

Firewall Rule is a simple firewall defense module. This function supports two rules: input and output. Each rule will distribute one series number; every kind of rule can allow to configure 10 pieces in max.

In view of the complexity of the firewall, let's take the following for an example.

Digital Map

MMI Filter

VPN Tunnel

Firewall

Syslog Config

Dial Peer

Firewall type

in_access enable out_access enable

Firewall Input/output Rule Table

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port

Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port

Firewall Configuration	
Parameters	Explanations
in_access enable	Enable input rules;
out_access enable	Enable output rules;

Input/Output	Select to add input/output rules;
Deny/Permit	Select to deny/permit current configurations;
Protocol Type	Four filtrate protocol including: TCP, UDP, ICMP, IP;
Port Range	Filtrated port range;
Src Addr	Source address can be phone address, network address; or address like 0.0.0.0, *.*.*.0; Eg: 192.168.1.0.
Des Addr	Destination address can be detailed IP address or can be address like 0.0.0.0, *.*.*.0; Eg: 192.168.1.0.
Src Mask	source mask, when configure it to 255.255.255.255, means it is the specific host, when it is configured in 255.255.255.0 as the net mask, it explains it is the network address.
Des Mask	Destination mask, when configure it to 255.255.255.255, means it is the specific host, when it is configured in 255.255.255.0 as the net mask, it explains it is the network address.

Click **【Add】** after finish configuration, there will be a new option in “output rule table” as the picture shows:

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
1	Deny	TCP	192.168.1.116	255.255.255.255	192.168.1.108	255.255.255.255	More than 1	1

Then select “in_access enable” and click the button **【Apply】** .

In this condition, when PC address 192.168.1.116 visits 192.168.1.108 WEB page, it cannot be visited due to it displaying “deny” in the input rules.

Rule Delete

Input/Output

Index to be deleted

Select the list you want to delect, then click **【Delete】** .

Syslog Config

Digital Map	MMI Filter	VPN Tunnel
Firewall	Syslog Config	Dial Peer

Syslog Config	
Server IP	<input type="text" value="0.0.0.0"/>
Server Port	<input type="text" value="514"/>
MGR Log Level	None <input type="button" value="v"/>
SIP Log Level	None <input type="button" value="v"/>
IAX2 Log Level	None <input type="button" value="v"/>
Enable Syslog	<input type="checkbox"/>

Syslog records the information comes from the system, which provides a mature client-server mechanism. Besides, Syslog receive news from the system, classify the news according to the priorities and the type of information, and then can be configured by the administrator in accordance with the rules which will be written into the log. It is a strong and unified method to manage the log.

Currently the system debug information is divided into 8 grades:

0-emergency, when the system is not available (such as system crashes, need to reboot etc.), the debug message in this condition is the most senior. the system default debug information for 0;

1 level-alert, the info indicate the system have fatal problem

2 level-critical, the info indicates the system have critical problem (e.g.: insufficient system resource, upgrade file error etc)

3 level-error, the info indicate the mistake will influence the system

4 level-warning, the info indicate the system is work normally, but it may contain potential risk.

5 level-notices, the info indicate the system is normal under certain condition, please make sure the environment and parameter is correct.

6 level-info, daily debug output info;

7 level-debug, info for debug, mainly use to output the related debug info for the need of the researcher, lowest but most output debug info

At present, the debug message((the lowest level are info, debug) sent to syslog can be displayed only through the telnet.

Syslog Configuration	
Parameters	Explanations
Server IP	Config Syslog server IP or domain;
Server Port	Config Syslog server port;
MGR Log Level	Config MGR log level;
SIP Log Level	Config SIP log level;
IAX2 Log Level	Config IAX2 log level;
Enable Syslog	Config enable/disable Syslog.

Dial Peer

Digital Map

MMI Filter

VPN Tunnel

Firewall

Syslog Config

Dial Peer

Dial Peer

Number	Destination	Port	Alias	Mode	Suffix	Del length
<div style="display: flex; justify-content: center; gap: 10px;"> <input type="button" value="Add"/> <input type="button" value="Delete"/> <input type="button" value="Modify"/> <input type="button" value="v"/> </div>						

Add Dial Peer

Phone Number	<input type="text"/>
Destination (optional)	<input type="text"/>
Port(optional)	<input type="text"/>
Alias(optional)	<input type="text"/>
Call Mod	SIP <input type="button" value="v"/>
Suffix(optional)	<input type="text"/>
Delete Length (optional)	<input type="text"/>

DIAL PEER	
Parameters	Explanations
Phone number	Outgoing number settings methods can be divided into two kinds: one is the exact match; When user configs to exact match, if this number is exactly the same as the called number, the phone will use this map IP address and corresponding configuration to make

	calls; the other one is the prefix match (the equivalent of the code prefix function PSTN), if this number is exactly the same as the N-bit number (prefix length) user dialed, the phone will use this map IP address and corresponding configuration to make calls. User needs to add T before the prefix number to distinguish it from the exact match numbers; maximum length is 30 digits; X format and progression scope are also supported. Please check the examples.
Destination	Config the destination address; For point to point call, user just need to input the end-to-end IP address or domain name. The phone's DNS server will parse the IP address. If there is no correlative config, it defaults the IP for 0.0.0.0. This is optional configuration item
Port	Config the counterpart protocol signaling port, it defaults for 5060; This is optional configuration item
Alias	Config alias; This is optional configuration item: the other numbers are used to replace the prefix number, If there is no correlative configuration, it shows no alias;
<p>Note: 4 alias sub-types, need the joint settings with the replacement length:</p> <ol style="list-style-type: none"> 1) add: xxx; Add xxx before the number. This can help users save dial-up length; 2) all: xxx; All numbers will be replaced by xxx, which can be used for speed dial. Such as, user config the dial-up number as 1, then through this "all" number, user can transform the actual outgoing number; 3) del; Delete the numbers before the n-bit, n will be replaced by the length settings; 4) rep: xxx; n will be replaced by the length settings if the n-bit numbers are replaced by xxx. For example, users want to call PSTN (0755-25118808) through the landing service by VoIP operators, but the real callee is 0755-25118808. We can configure the callee number as 9T, and rep: 0755, then set 1 in the length settings. Thus all users who dial the number beginning with 9 will be sent out and replaced by "0755 + numbers". It is convenient and easy for user's dialing habits. 	
Call Mode	Config to select the signaling protocol, SIP/IAX2;
Suffix	Config the suffix, this is optional configuration item; It means add the suffix after the dialing number. If there is no correlative configuration, it displays "no suffix";
Delete Length	Config replace/delete length, this is optional configuration item; The dialed number will be replaced/deleted according to delete length.
Here is the introduction of how to config IP number table to achieve using more than one accounts at the same time:	

Number	Destination	Port	Alias	Mode	Suffix	Del length
0T	0.0.0.2	5060	no alias	SIP	no suffix	0
9T	0.0.0.1	5060	no alias	SIP	no suffix	0

9T mapping indicates when user has registered and configured the SIP1 server, then all users should dial 9 before they make every call via SIP1;

0T mapping indicates when user has registered and configured the SIP2 server, then all users should dial 0 before they make every call via SIP2;

For other lines like SIP3/SIP4/SIP5, user can config Destination as 0.0.0.3/0.0.0.4/0.0.0.5

7T	0.0.0.0	4569	no alias	IAX2	no suffix	0
----	---------	------	----------	------	-----------	---

7T mapping indicates when user has registered and configured the IAX2 server, then all users should dial 7 before they make every call via IAX2;

8. System Management

8.1 Save config

Save Config

Reset Default

Backup Config

Phone Book

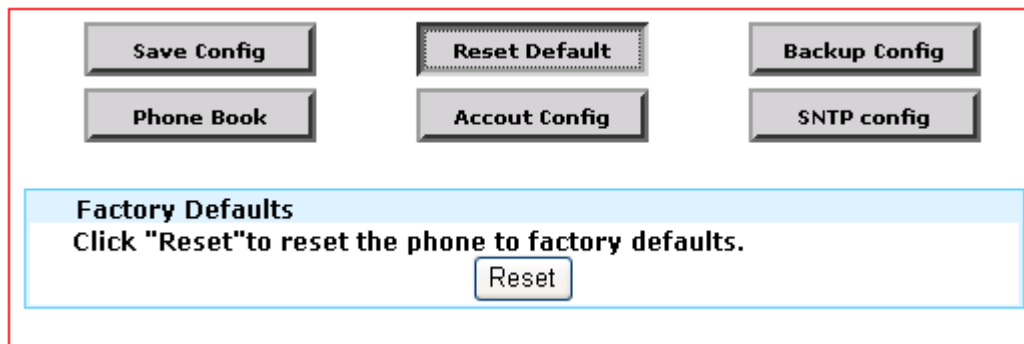
Account Config

SNTP Config

Save Configuration
Press the "Save" button to save the configuration files !

Note: Save current config and then it will go into effect. Otherwise, the settings won't be saved and the revised settings will be lost after the phone reboots.

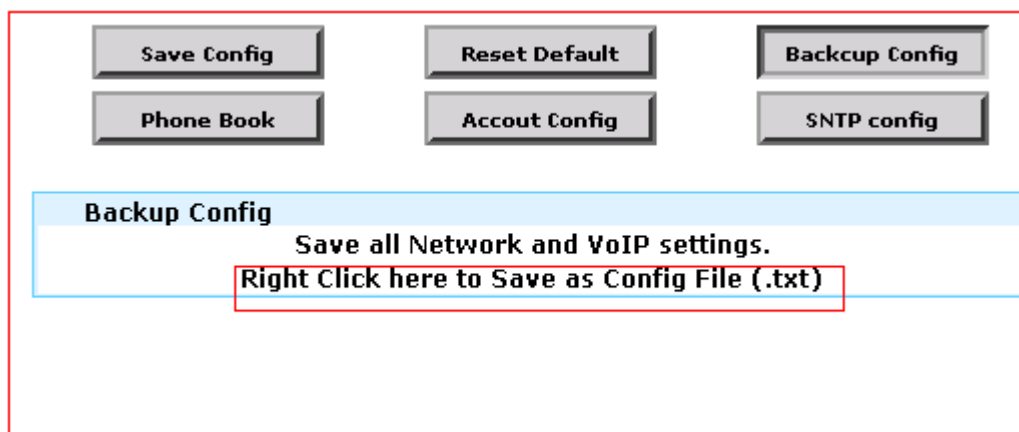
8.2 Restore to Factory Default



Restore the phone to factory default settings and it will reboot automatically.

NOTE: If users log on as the admin, clearing config will make the phone revert to factory default settings; If users log on as the guest, clearing config will clear the configs except the account and current edition version's relevant configuration items (SIP1-SIP5, IAX2).

8.3 Backup Config



Backup the device's all current configuration, right click to select Save As, which is able to download the phone's configuration file (the suffix is .txt.) After the files are exported and the device has been restored to the factory settings, user can directly import the configuration files using WEB update. These files can be imported for other PH806 phone use.

8.4 Phone Book

Save Config
Reset Default
Backup Config

Phone Book
Account Config
SNTP Config

Phone Book			
Index	Name	Number	Ring Type
1	"ouyang"	636	Type 5

Add
Delete
Modify
ouyang ▼

Phone Book													
Parameters	Explanations												
<table border="1" style="width: 100%; border-collapse: collapse; text-align: center;"> <thead> <tr style="background-color: #e1f5fe;"> <th colspan="4">Phone Book</th> </tr> <tr> <th>Index</th> <th>Name</th> <th>Number</th> <th>Ring Type</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>"steven"</td> <td>626</td> <td>Type 9</td> </tr> </tbody> </table>		Phone Book				Index	Name	Number	Ring Type	1	"steven"	626	Type 9
Phone Book													
Index	Name	Number	Ring Type										
1	"steven"	626	Type 9										
Display current detailed info of the phone book.													
Name	Telephone number display name; When comes the incoming call, the phone LCD will display the name corresponding to the number.												
Number	Config the telephone number;												
Ring Type	Config the incoming ring type, 2 user-defined ring types are supported.												
Modify the selected account; Firstly select the account and then click 【Modify】 ; choose one in the drop-down menu and click 【Delete】 to delete the account user.													
Note: Phone book capacity is 500 records maximumly.													

8.5 Account Config

Save Config
Reset Default
Backup Config

Phone Book
Account Config
SNTP Config

Set Keyboard Password

Keyboard password

User Set

User Name	User Level
admin	1
guest	0

admin ▾

Account Configuration							
Parameters	Explanations						
Keyboard Password	Config to input the password via phone keypad. Note: the password should be numbers.						
<div style="border: 1px solid lightblue; padding: 5px; margin: 5px auto; width: 80%;"> <p>User Set</p> <table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">User Name</th> <th style="text-align: left;">User Level</th> </tr> </thead> <tbody> <tr> <td>admin</td> <td>1</td> </tr> <tr> <td>guest</td> <td>0</td> </tr> </tbody> </table> </div>		User Name	User Level	admin	1	guest	0
User Name	User Level						
admin	1						
guest	0						
Display the phone's user name list;							
User Name	Config to add a user name;						
User Level	Config the user level; 1 means users have the modifying authority; 0 means users can read only;						
Password	Config the password;						
Confirm	Comfirm the password to make sure the password is correct;						
Modify the selected account; Firstly select the account and then click 【Modify】 ; choose one in the drop-down menu and click 【Delete】 to delete the account user. General level users can only add the same level users.							

8.6 SNTP Config

Save Config
Reset Default
Backup Config

Phone Book
Account Config
SNTP Config

SNTP Time Set

server

timezone ▼

timeout (seconds)

Sntp

Daylight

SNTP	
Parameters	Explanations
Server	Config SNTP server address;
Time Zone	Config to select the time zone;
Time Out	Time interval to be in-phase with the server, its default time is 60s.
SNTP	Enable/disable SNTP service;
Daylight	Enable daylight saving time; Only partial areas support daylight fuction, such as Alaska.

Manual Timeset

year

months

day

hour

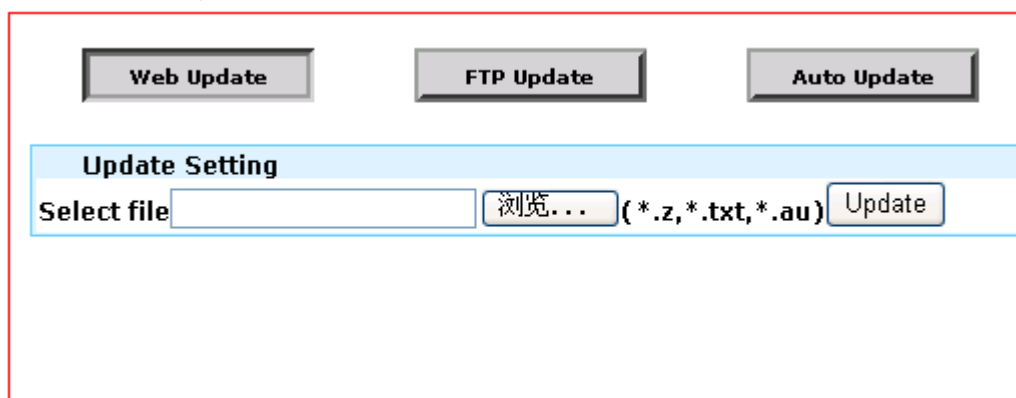
minute

User need to disable SNTP firstly to config the time manually. Only when the manual timesets including year, months, day, hour and minute all are filled in and submitted, then the configuration will go into effect.

9. Update

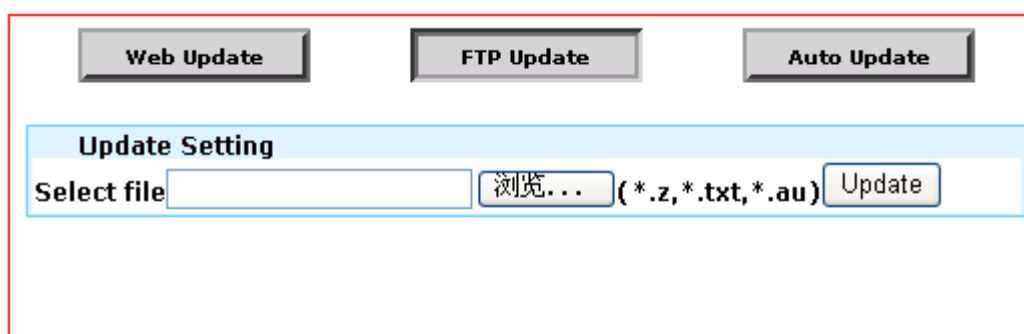
9.1 Web Update

Browse and find the previously-saved config files(or the config files provided by the manufacturer), then download the files to the phone, which save many config troubles. Besides, on this web page, user can download the update file, mmiset files and ring tone(the ring file name should be 1.au and 2.au, correspondingly USER1 and USER2's ring type). Click **【Update】** to activate the config.



Update	
Parameters	Explanations
Update image	.z is the phone's image file format. We provide the updating programme free of charge, please pay attention to our company website update notice.
Update ringer	.au is the ringtone format supported by the phone. The ringtone file cannot be over 256K.
Update config	.txt is the config file format.

9.2 FTP Update



FTP Update	
Parameters	Explanations
Web Update	Browse and find the previously-saved config files(or the config files provided by the manufacturer), then download the files to the phone, which save many config troubles. Besides, on this web page, user can download the update file, mmiset files and ring tone(the ring file name should be 1.au and 2.au, correspondingly USER1 and USER2's ring type). Click 【Update】 to activate the config.
Server	Config to upload or download FTP server address. The server address can be IP format like 192.168.1.1; Or domain format like ftp.domain.com . Additionally, the system supports that server can set up a sub-catalog. Such as, server address can be config as 192.168.1.1/ftp/config/ or ftp.domain.com/ftp/config . It means that the server address is 192.168.1.1 or ftp.domain.com , and the files are saved by path: /ftp/config/. The sub-catalog is optional.
Username	Config to upload or download FTP server user name. For TFTP mode, user don't need to config the user name and password.
Password	Config to upload or download the FTP server password.
File name	Config to upload or download the update files or system config files. If there is no correlative config, the system will use its own MAC address as the file name like 000102030405;
<p>Note: The exported config files can be modified; Besides, user can import the files module by module. Such as, user can only keep SIP module and import this SIP module into the system. Other modules won't be missing due to the import of the partial config module.</p>	
Type	3 config explanations: 1. Application update: download the upgrade files. 2. Config file export: Upload the config files to FTP/TFTP server, and the file names can be defined by users themselves. 3. Config file import: Download the config files from FTP/TFTP server to the phone, reboot and then the settings will go into effect.
Protocol	Select the server type: FTP/TFTP;

9.3 Auto Update

Web Update
FTP Update
Auto Update

Auto Update Setting

Current Version 2.0002

Server Address

Username

Password

Config File Name

Config Encrypt Key

Protocol Type ▼

Update Interval Time Hour

Update Mode ▼

APPLY

Auto Provision	
Parameters	Explanations
Current Config Version	Display current config file's edition version.
Server Address	Config the FTP server address. The server address can be IP format like 192.168.1.1; Or domain format like <u>ftp.domain.com</u> . Additionally, the system supports that server can set up a sub-catalog. Such as, server address can be config as 192.168.1.1/ftp/config/ or <u>ftp.domain.com/ftp/config</u> . It means that the server address is 192.168.1.1 or <u>ftp.domain.com</u> , and the files are saved by path: /ftp/config/. The sub-catalog is optional.
Username	Config user name of FTP server; For TFTP, no need to config; For FTP, leave a blank here; It defaults as a anonymous ftp user.
Password	Config the password of FTP server.
Config File Name	It is left a blank if auto update function is selected, thus the phone device will use its own MAC address as the file name to get the files from the server.
Config Encrypt Key	If the config files are encrypted, user need to input the encrypt key to continue the update.
Protocol Type	3 server types: FTP; TFTP; HTTP.
Update Interval Time	Config update interval time, unit is hour.
	3 update mode:

Update Mode	<ol style="list-style-type: none">1. Disable, means no update.2. Update after reboot3. Update at time interval
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10. Call Record

This is an call record list. User can check all the call record on this page as the picture shows below.

DEC 16 04:38	0	sip:620@1
DEC 16 04:38	0	sip:610@1
DEC 16 04:38	0	sip:618@1
DEC 16 04:38	1	sip:112@1
DEC 16 04:38	4	sip:112@1
DEC 16 04:36	0	sip:112@1
DEC 16 03:55	0	sip:112@2
DEC 16 03:54	0	sip:611@2
DEC 16 03:54	0	sip:112@2
DEC 16 03:54	0	sip:112@2

If you still have questions, please link to our website (www.511soft.com) for more info or contact our technical engineer for further support.