PLANET
Networking & Communication

User's Manual

Professional HD PoE IP Phone (6-Line)

► VIP-5060PT

www.PLANET.com.tw

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CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

Energy Saving Note of the Device

This power required device does not support Stand by mode operation. For energy saving, please remove the DC-plug or push the hardware Power Switch to OFF position to disconnect the device from the power circuit.

Without removing the DC-plug or switching off the device, the device will still consume power from the power circuit. In view of Saving the Energy and reducing the unnecessary power consumption, it is strongly suggested to switch off or remove the DC-plug from the device if this device is not intended to be active.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

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Revision

User’s Manual for PLANET Professional HD PoE IP Phone:
Model: VIP-5060PT
Rev: 1.0 (2013, Oct)
Part No. EM-VIP-5060PT_v1.0

Table of Contents

1 Introduction

Cost-effective, High-performance PoE VoIP Phone

To build high-performance VoIP communications at a low cost, PLANET has launched a new member of its IP Phone family, the VIP-5060PT enterprise-class 6-Line PoE IP Phone. It complies with IEEE 802.3af PoE interface for flexible deployment. The VIP-5060PT makes it simple for the enterprise featuring voice and data system or expanding voice system to new locations. It helps the company to save money on long distance calls; for example, the remote workers can dial in through a Unified VoIP Communication System just like an extension call but no long distance call charge would occur. The VIP-5060PT also allows call to be transferred to anyone at any location within the voice system, which enables the enterprise to communicate more effectively and is helpful to streamline business processes.

High Quality HD VoIP Voice

The VIP-5060PT delivers HD voice (High-Definition Voice) which is the next generation of voice quality for telephony audio, making the quality of voice better than that (toll quality) of the standard digital telephony and even close to that of a room conversation. HD voice is transmitted in the audio frequency range of 50 Hz to 7 kHz or higher over telephone lines, resulting in higher quality voice and clearer communication.

Standard Compliance

The VIP-5060PT supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VIP-5060PT is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

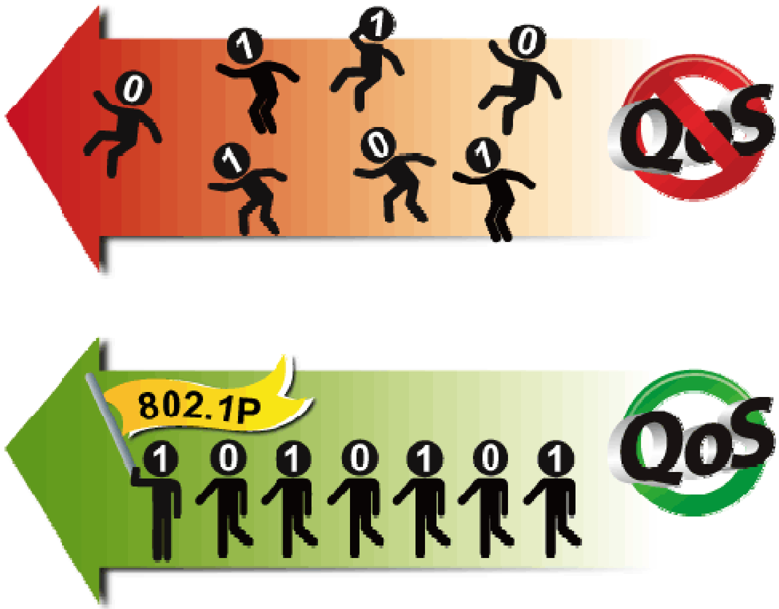
Enhanced, Full-Featured Business IP Phone

The VIP-5060PT is a full-featured enhanced business IP Phone that addresses the communication needs of the enterprises. It provides 6 voice lines and dual 10/100/1000 Mbps Ethernet. Furthermore, the VIP-5060PT delivers user-friendly design containing a 128 x 64 LCD with white backlight, 4 Line keys and 4 soft keys. It supports 5 extension consoles with each consisting of 26 keys.

The VIP-5060PT supports all kinds of SIP based phone features including Call Waiting, Auto Answer, Music on Hold, Caller ID and Call Waiting ID, 3-way Conferencing, Call Hold, Call Forwarding, Black List, DTMF Relay, In-Band, Out-of-Band (RFC 2833) and SIP INFO, among others. Besides office use, the VIP-5060PT is also the ideal solution for VoIP service offered by Internet Telephony Service Provider (ITSP).

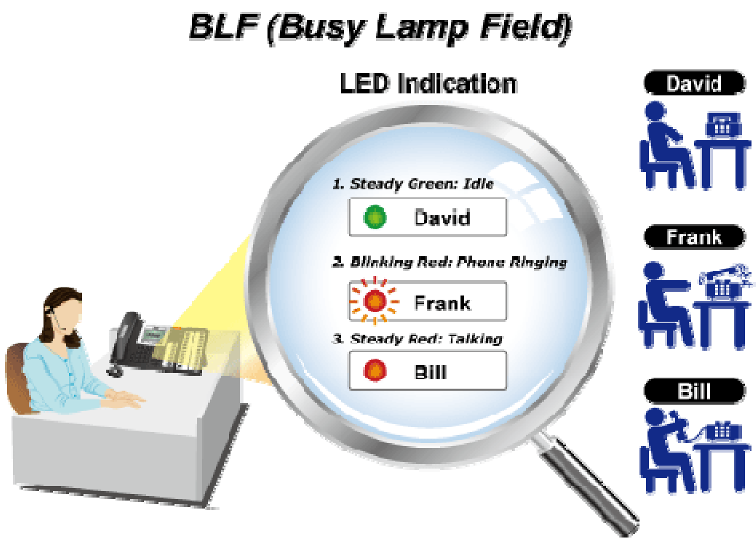
Secure, High-Quality VoIP Communication

The VIP-5060PT can effortlessly deliver secured toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service), 802.1Q VLAN tagging, and IP TOS (Type of Service) technology. Using voice and data VLAN can easily separate the data and voice, thus maintaining the best quality.



Professional Application

The VIP-5060PT supports Busy Lamp Field (BLF) function that, via the lights on the phone, enables users to easily identify the status of other phones which are connected to the same IP PBX, such as busy, idle, ringing, etc. The connected IP PBX must also support BLF feature. The BLF function is helpful for a receptionist on the front desk to route all incoming calls smoothly.



1.1 Features

1.1 Features

- **Highlights**
  - Dual 10/100/1000 Gigabit Ethernet (WAN, LAN)
  - Supports SIP 2.0 (RFC3261)
  - Supports six SIP voice lines
  - IEEE 802.3af Power over Ethernet compliance
  - Supports multiple road calls waiting in line
  - Supports HD voice
  - Supports SRTP and Busy Lamp Field (BLF)
  - Supports 5 extension consoles; max. 130 definable keys

- **Advanced Features**
  - SIP supports SIP domain, SIP authentication (none, basic, MD5), DNS name of server, Peer to Peer/ IP call
  - Inband, SIP info, RFC2833 DTMF Relay
  - 9 kinds of ring types and 3 user-defined music rings
  - Large dot matrix LCD display and soft keys make user easier to use
  - Soft keys and function keys programmable
  - Multilanguage realizes localization
  - Echo cancellation: Supports G.168, and hands-free can support 96ms
  - Full duplex hands-free speaker phone
  - Hands-free headset ringing choice
  - Supports Voice Gain Setting, VAD, CNG
  - Voice codec setting for each SIP line

- **SIP Applications**
  - Call forward / Transfer (blind/attended)
  - Call Holding / Waiting
  - 3-way conference
  - Paging and Intercom
  - Call park / Call pickup / Join call
  - Redial and click to dial
  - Secondary dialing automatically
  - Incoming calls /outgoing calls / missed calls (Each supports 100 records)
  - SMS and Speed Dial
  - Phonebook up to 500 records
  - XML phonebook / browser

- **Call Control Features**
  - Flexible dial map / Hotline / Empty calling no.
  - Reject service / Black list for reject authenticated call
  - White list / Limit cal
  - Do not disturb (DND)
  - Caller ID / CLIR (reject the anonymous call) / CLIP (make a call with anonymous)
  - Dial without register

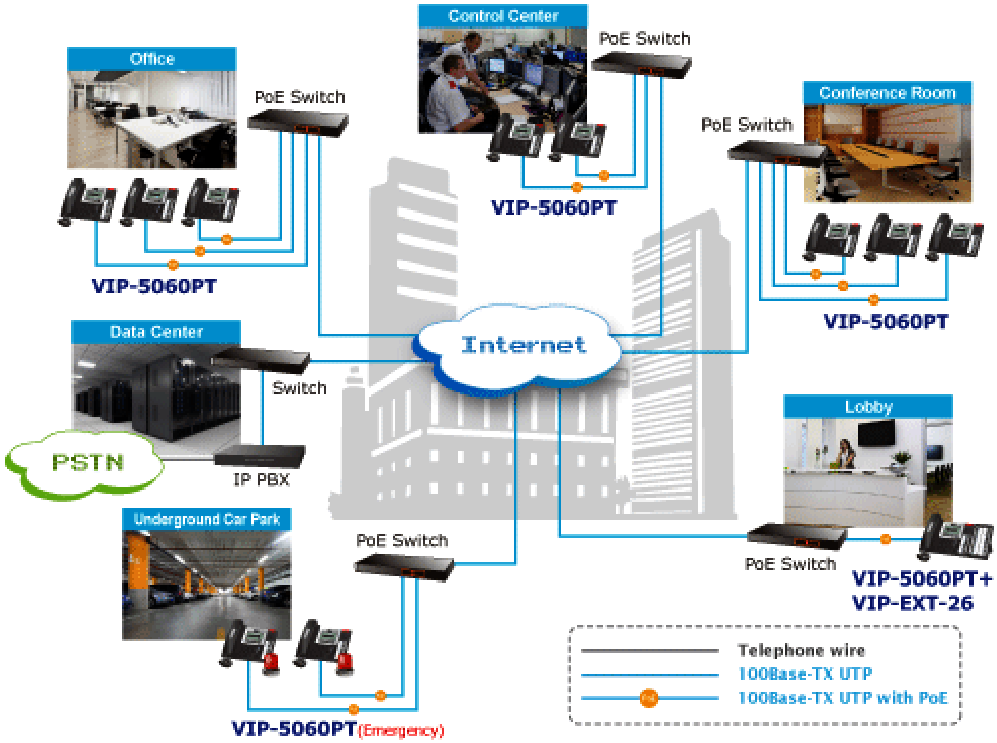
- **Network Features**
  - Route and Bridge modes
  - PPPoE / DHCP client on WAN
  - 802.1 VLAN (voice VLAN / data VLAN)
  - VPN (L2TP) and DMZ
  - Main DNS and secondary DNS server
  - DNS Relay, SNTP Client, Firewall, openVPN

- **Maintenance and Management**
  - Integrated web server provides web-based administration and configuration
  - Telephone keypad configuration via display menu/navigation
  - Automated provisioning and upgrade via HTTPS, HTTP, TFTP
  - User Authentication for configuration pages

- Local and Remote Syslog (RFC 3164)
- SNTP Time Synchronization
- TR069

1.2 Application

1.2 Application



Enterprise IP PBX Deployment of VIP-5060PT

1.3 Product Specifications

1.3 Product Specifications

Product	VIP-5060PT
Hardware	
Lines (Direct Numbers)	6-Line enterprise-class IP phone
Display	80 x 43mm/ 128 x 64 pixel LCD with blue backlight
Feature Keys	4 line keys 8 DSS keys 4 Soft Keys 12 dialing buttons (0~9, *, #) 12 fixed function buttons
WAN	10/100/1000Base-T RJ-45 for WAN
LAN	10/100/1000Base-T RJ-45 for LAN
Protocols and Standard	
Data Networking	MAC Address (IEEE 802.3) IPv4 (RFC 791) Address Resolution Protocol (ARP) DNS: A record (RFC 1706), SRV record (RFC 2782) Dynamic Host Configuration Protocol (DHCP) client (RFC 2131) Internet Control Message Protocol (ICMP) (RFC 792) TCP (RFC 793) User Datagram Protocol UDP (RFC 768) Real Time Protocol RTP (RFC 1889, 1890) Real Time Control Protocol (RTCP) (RFC 1889) Differentiated Services (DiffServ) (RFC 2475) Type of service (ToS) (RFC 791, 1349) VLAN tagging 802.1p Layer 2 quality of service (QoS) Simple Network Time Protocol (SNTP) (RFC 2030) Backward compatible with RFC 2543 Session Timer (RFC 4028) SDP (RFC 2327) NAPTR for SIP URI Lookup (RFC 2915)

Voice Gateway	SIP version 2 (RFC 3261, 3262, 3263, 3264) SIP supported STUN (RFC 3489) Message Waiting Indicator (RFC 3842) Voice algorithms: - G.711 (A-law and μ-law) - G.7231 high/low - G.729a/b - G.722.1 - G.726 Dual-Tone Multi-Frequency (DTMF), In-Band and Out-of-Band (RFC 2833) (SIP INFO) Voice Activity Detection (VAD) with Silence Suppression Adaptive Jitter Buffer Management Comfort Noise Generation Echo Cancellation Message
Provisioning, Administration, and Maintenance	Integrated web server provides web-based administration and configuration Telephone keypad configuration via display menu/navigation Automated provisioning and upgrade via HTTPS, HTTP, TFTP User Authentication for configuration pages Local and Remote Syslog (RFC 3164) SNTP Time Synchronization TR069
Features	
Advantageous Applications	Supports SIP 2.0 (RFC3261) SIP supports 6 SIP lines. IEEE 802.3af Power over Ethernet (PoE) compliant Supports multiple road call waiting in line Supports HD voice Supports SRTP and BLF SIP supports SIP domain, SIP authentication (none, basic, MD5), DNS name of server, Peer to Peer/ IP call DTMF Relay: support inband, SIP info, RFC2833 9 kinds of ring types and 3 user-defined music rings Large dot matrix LCD display and soft keys make user easier to use Supports headset jack- RJ9 4 DSS Key

	Support 5 ext. consoles with each consisting of 26 keys Soft keys programmable; function keys programmable Multilanguage realizes localization Echo cancellation: Supports G.168, and Hands-free can support 96ms, Hands-free Speaker Phone Supports Voice Gain Setting, VAD, CNG Full duplex hands-free speaker phone Hands-free headset ringing choice Voice codec setting for each SIP line
SIP Applications	Call forward Transfer (blind/attended) Holding Waiting 3-way conference Paging and Intercom Call park Call pickup Join call Redial and click to dial Secondary dialing automatically Incoming calls /outgoing calls / missed calls. Each supports 100 records. Support Phonebook 500 records Support SMS and Speed Dial Support XML phonebook/browser
Call Control Features	Flexible dial map Hotline Empty calling no. Reject service Black list for reject authenticated call White list Limit cal Do not disturb Caller ID CLIR (reject the anonymous call) CLIP (make a call with anonymous) Dial without register

Network Features	WAN/LAN: 10/100M Ethernet ports, supports Bridge modes. Supports bridge working as hub Supports PPPoE for xDSL and PoE Supports 802.1 VLAN(voice VLAN/data VLAN) Supports DHCP client on WAN Supports main DNS and secondary DNS server. Supports DNS Relay, SNTP Client, Firewall, openVPN Supports VPN (L2TP) and DMZ Network tools in telnet server: including ping, trace route, telnet client
Maintenance and Management	Web, telnet and keypad management Management with different account right Upgrade firmware through POST mode and HTTP, FTP or TFTP Supports DHCP option66 auto provisioning Telnet remote management/upload/ download setting file Safe mode provide reliability Supports Auto Provisioning to upgrade firmware or configuration file with HTTPS Supports TR-069(optional) and Syslog
Environments	
Power Requirements	5V DC, 1A IEEE 802.3af
Operating Temperature	0 ~ 40 degrees C
Operating Humidity	10 ~ 65% (non-condensing)
Weight	990 g
Dimensions (W x D x H)	290 x 260 x 60 mm
Emission	CE, FCC, RoHS
Connectors	Two 10/100/1000 BASE-T RJ-45 Ethernet ports Handset: RJ-9 connector Headset: RJ-9 connector RJ-11 EXT connector DC power jack Built-in speakerphone and microphone

1.4 Physical specifications and packaging

Physical Specifications

➢ Dimensions

Dimensions	290 (L) x 260 (W) x 60 (H) mm
Net Weight	950g (without package)

BASIC PACKAGING

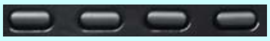
- SIP IP Phone unit
- Power Adapter
- Quick Installation Guide
- CD-ROM containing the on-line manual.
- RJ-45 cable x 1
- Stand x 1

1.5 Keypad

➢ **Keypad, LED, and function key definitions**

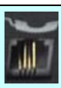
➢ **Keypad Description**

Key	Key Name	Function Description
	**Navigation**	Assists you in selecting an item that you want to process under the menu by pressing the Up, Down, Right or Left button. Press the center button to save.
	**Directory**	Access to phone book by checking the record list, adding new records or revising the record. When checking the phone book record, press this key again to return to idle

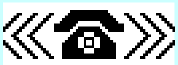
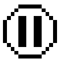
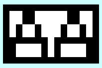
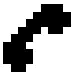
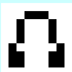
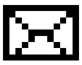
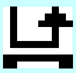
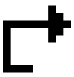
Key	Key Name	Function Description
		mode.
	**Mute**	Press this key in calling mode and you can hear the other side, but the other side cannot hear you.
	**Volume -/+**	Turn down or turn up the volume by pressing the “-“ key or the “+” key.
	**Redial**	1. In the hook off /hands-free mode, use the key to dial the last call number; 2. In stand-by mode, it has a function to check the Outgoing Call.
	**Hands-free**	Make the phone into hands-free mode.
	**Indicator light**	Blinking light indicates there is an incoming call.
 **Soft key 1/2/3/4**		Key combination includes functions such as History/Directory/DND/Menu/Del/Redial/Send/ Quit/Answer/Divert/Reject/Hold/Transfer/Conf/Close and so on.
	**History**	View the Missed Calls, Incoming Calls and Dialed Calls.
	**Digital keyboard**	Inputting the phone number or DTMF.
	**Line Keys**	Switch to different lines
	**DSS keys**	You can configure them on the web page.

- **Rear view and panel descriptions**

- **Keypad Description**

Port	Port name	Description
	**Power switch**	Input: 5V AC, 1A
	**WAN**	10/100/1000M Connect it to Network
	**LAN**	10/100M/1000 Connect it to PC
	**External console interface**	Port type: RJ-11 direct connector
	**Headset**	Port type: RJ-9 connector
	**Handset**	Port type: RJ-9 connector

1.6 Icon introduction

Icon	Description
	Call out
	Call in
	Call hold
	Auto answer
	Call mute
	Contact
	DND(Do not Disturb)
	In hand-free mode
	In handset mode
	In headset mode
	SMS
	Missed call
	Call forward

1.7 LED introduction

Table 1 Programmable Key LED for BLF

LED Status	Description
Steady green	The object is in idle status.
Slow blinking red	The object is ringing.
Steady red	The object is active.
Fast blinking red	The object failed.
Off	No subscription

Table 2 Programmable key LED for Presence

LED Status	Description
Steady green	The object is online.
Slow blinking red	The object is ringing.
Steady red	The object is active.
Fast blinking red	The object failed.
Off	No subscription

Table 3 Programmable key LED for line

LED Status	Description
Steady green	The account is active.
Fast Blinking green	There is an incoming call to the account.
Slow Blinking green	The call is on hold.
Slow Blinking red	Registration is unsuccessful.
Off	The line is not applied or is idle.

Table 4 Programmable key LED for MWI

LED Status	Description
Blinking green	There are new voice mails.
Off	There is no new voice mail.

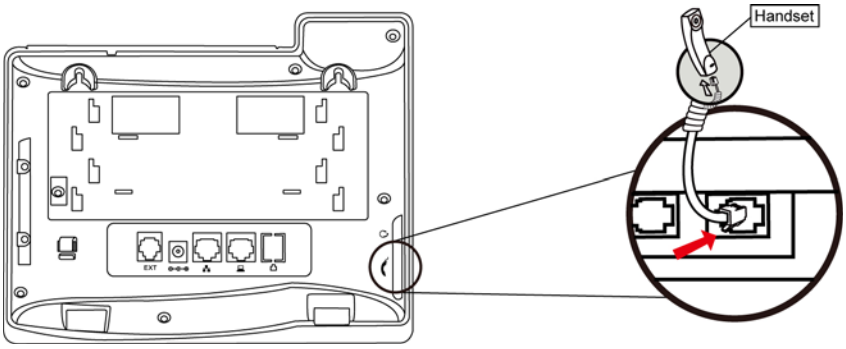
Table 5 Power Indication LED

LED Status	Description
Steady red	Power on.
Fast Blinking red	There is an incoming call.
Off	Power off.

2 Initial Connection and Login

Step 1. Handset Connection

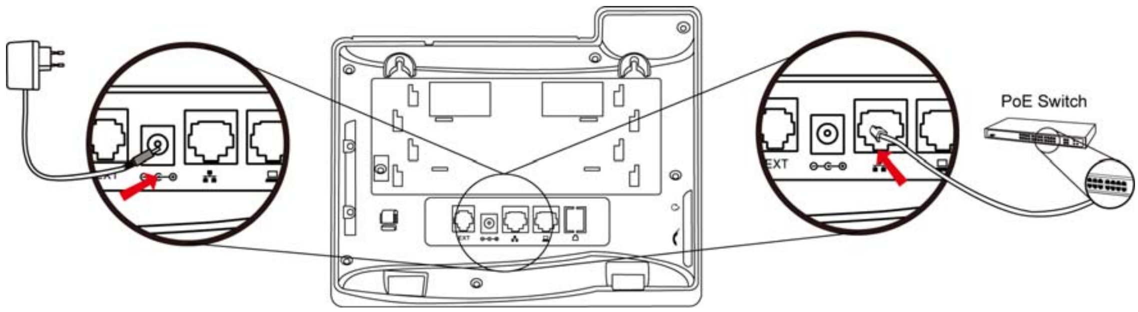
Plug one end of the handset cord into the handset and the other end into the handset jack



Step 2. Connecting Power System

The VIP-5060PT can be powered either by external AC/DC adapter or by connecting to an IEEE802.3af/at PSE device such as 802.3af Injector / Hub or 802.3af/at POE switch.

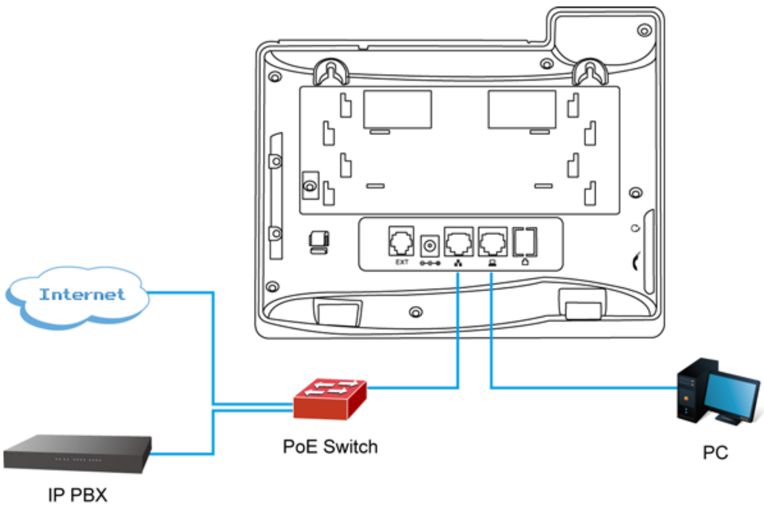
Once the VIP-5060PT is powered, the LCD screen will prompt for POST.



Note1: Use only the power adapter shipped with the unit to ensure correct functionality

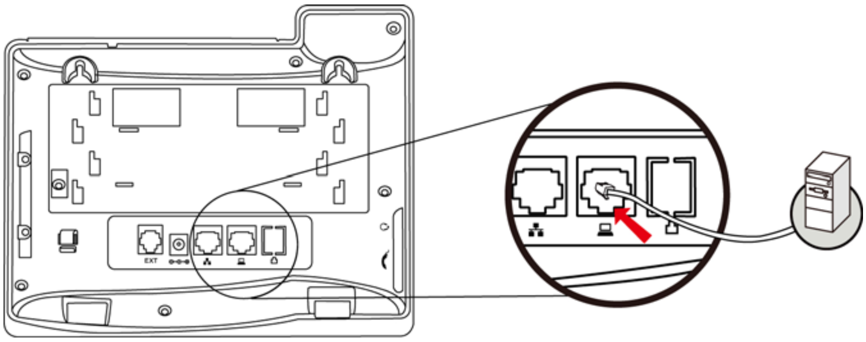
Note2: Only WAN supports POE.

Step 3. Connecting Network



Step 4. Computer Network Setup

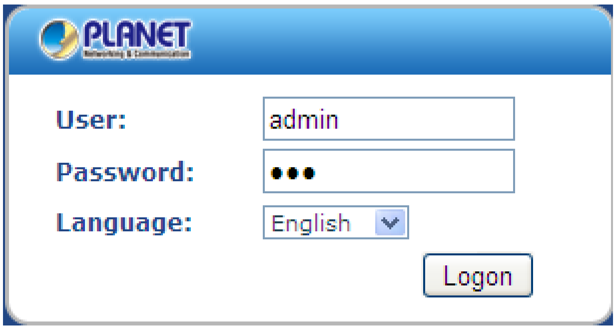
Set your computer's IP address to 192.168.0.x, where x is a number between 2 to 254 (except 1 where is being used for the phone by default). If you don't know how to do this, please ask your network administrator. Connect your PC to VIP-5060PT PC port.



Step 5. Login Prompt

Use web browser (Internet Explorer 6.0 or above) to connect to 192.168.0.1 (type this address in the address bar of web browser).

You'll be prompted to input user name and password: **admin** and **123**



3 Basic Functions

3.1 Making a call

3.1.1 Call Device

User can make a phone call via the following devices:

1. Pick up the handset, icon will be shown on the idle screen.

2. Press the Speaker button, icon will be shown on the idle screen.

3. Press the Headset button if the headset is connected to the Headset Port in advance. The icon will be shown on the idle screen.

User can also dial the number first, and then choose the method user will use to speak to the other party.

3.1.2 Call Methods

User can press an available line button if there is more than one account, then

1. Dial the number User wants to call.

2. Press History softkey. Use the navigation buttons to highlight User choice (press Left/Right button to choose Missed Calls, Incoming Calls and Outgoing Calls.

3. Press the R/SEND button to call the last number called.

4. Press the programmable keys which are set as speed dial button.

Then press the Send button or Dial softkey to make the call if necessary.

3.2 Answering a call

Answering an incoming call

1. If User is not on another phone, lift the handset to use, or press the Speaker button/ Answer softkey to answer using the speaker phone, or press the headset button to answer the headset.
2. If User is on another call, press the answer softkey.

During the conversation, User can alternate between Headset, Handset and Speaker phone by pressing the corresponding buttons or picking up the handset.

3. Then press the Save to save the changes.

3.3 DND

3.7 Mute

Press DND softkey to activate DND Mode. Further incoming calls will be rejected and the display shows: icon. Press DND softkey twice to deactivate DND mode. User can find the incoming call record in the Call History.

Press Mute button during the conversation, icon

3.4 Call Forward

This feature allows User to forward an incoming call to another phone number. The display shows icon.

The following call forwarding events can be configured:
Off: Call forwarding is deactivated by default.
Always: Incoming calls are immediately forwarded.
Busy: Incoming calls are immediately forwarded when the phone is busy.
No Answer: Incoming calls are forwarded when the phone is not answered after a specific period.

To configure Call Forward via Phone interface:

1. **Press Menu → Features → Enter → Call Forwarding → Enter.**
2. There are 4 options: Disabled, Always, Busy, and No Answer.
3. If User chooses one of them (except Disabled), enter the phone number User wants to forward to receiving party. Press Save to save the changes.

3.5 Call Hold

1. Press the Hold button or Hold softkey to put User active call on hold.
2. If there is only one call on hold, press the hold softkey to retrieve the call.
3. If there are more than one call on hold, press the line button, and the Up/Down button to highlight the call, and then press the Un-hold button to retrieve the call.

3.6 Call Waiting

1. **Press Menu → Features → Enter → Call Waiting → Enter.**
2. Use the navigation keys to activate or deactivate call waiting.

will be shown on the LCD. Then the called will not hear User, but User can hear the called. Press it again to get the phone to normal conversation.

3.8 Call transfer

1. Blind Transfer

During talking, press the key “Transf”, and then dial the number that User wants to transfer to, and finish by pressing "#". Phone will transfer the current call to the third party. After finishing transfer, the call User talks to will be hanged up. User cannot select SIP line when phone transfers call.

2. Attended Transfer

During talking, press the key “Transf”, then input the number that User wants to transfer to and press Send. After that third party answers, then press Transfer to complete the transfer. (User needs to enable call waiting and call transfer first). If there are two calls, User can just talk to one, and keep hold to the other one. The one who is keeping hold cannot speak to User or hear from User. In other words, if user wants to invite the third party during the call, they can press Conf to make calls mode in conference mode. If user wants to stop conference, user can press Split. (User must enable call waiting and three way call first).

Note	The server that user uses must support RFC3515 or it might not be used.

3. Alert Transfer

During the talking, press Transf first, and then press Send after inputting the number that User wants to transfer. Users are waiting for connection, now, press Transf and the transfer will be done. (To use this feature, User needs to enable call waiting and call transfer first).

3.9 3-way conference call

1. Press the Conf softkey during an active call.

2. The first call is placed on hold. Then User will hear a dial tone. Dial the number to conference in, and then press Send key.

3. When the call is answered, press Conf and add the first call to the conference.

3.3 DND

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3. If User chooses one of them (except Disabled), enter the phone number User wants to forward to receiving party. Press Save to save the changes.

3.5 Call Hold

1. Press the Hold button or Hold softkey to put User active call on hold.
2. If there is only one call on hold, press the hold softkey to retrieve the call.
3. If there are more than one call on hold, press the line button, and the Up/Down button to highlight the call, and then press the Un-hold button to retrieve the call.

3.6 Call Waiting

1. **Press Menu → Features → Enter → Call Waiting → Enter.**
2. Use the navigation keys to activate or deactivate call waiting.
3. Then press the Save to save the changes.

3.7 Mute

Press Mute button during the conversation, icon will be shown on the LCD. Then the called will not hear User, but User can hear the called. Press it again to get the phone to normal conversation.

3.8 Call transfer

1. Blind Transfer

During talking, press the key “Transf”, and then dial the number that User wants to transfer to, and finish by pressing "#". Phone will transfer the current call to the third party. After finishing transfer, the call User talks to will be hanged up. User cannot select SIP line when phone transfers call.

2. Attended Transfer

During talking, press the key “Transf”, then input the number that User wants to transfer to and press Send. After that third party answers, then press Transfer to complete the transfer. (User needs to enable call waiting and call transfer first). If there are two calls, User can just talk to one, and keep hold to the other one. The one who is keeping hold cannot speak to User or hear from User. In other words, if user wants to invite the third party during the call, they can press Conf to make calls mode in conference mode. If user wants to stop conference, user can press Split. (User must enable call waiting and three way call first).

Note	The server that user uses must support RFC3515 or it might not be used.

3. Alert Transfer

During the talking, press Transf first, and then press Send after inputting the number that User wants to transfer. Users are waiting for connection, now, press Transf and the transfer will be done. (To use this feature, User needs to enable call waiting and call transfer first).

3.9 3-way conference call

1. Press the Conf softkey during an active call.

2. The first call is placed on hold. Then User will hear a dial tone. Dial the number to conference in, and then press Send key.

3. When the call is answered, press Conf and add the first call to the conference.

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 is appointed prefix code. After making the above configuration, A can dial
3 plus B's phone number to make the redial function.
4 is appointed prefix code. After configuration, A can dial *4* to cancel redial function.
User can set prefix at random, in case it does not affect the current dialing rules.

4.4 Click to dial

When user A browses on an appointed Web page, user A can click to call user B via a link (this link to user B), then user A's phone will ring, after A hooks off, the phone will dial to B.

Note	It needs an external software that supports click to dial.

4.5 Call back

This function allows User to dial out the last phone call User receives.

4.6 Auto answer

When there is an incoming call unanswered, the phone will answer the call automatically.

4.7 Hotline

User can set hotline number for every sip, and then enter the dialer interface and after Warm Line Time, the phone will call out the hotline number automatically.

4.8 Applications

4.8.1 SMS

1. **Press Menu → Applications → Enter → SMS → Enter.**

2. Use the navigation keys to highlight the options. User can read the message in the Inbox/Outbox.

3. After viewing the new message, User can press Reply to reply the message, and use the 2aB softkey to change the Input Method. When entering the reply message, press OK, and then use the navigation keys to select the line from which User wants to send, then Send.

4. If User wants to write a message, User can press New and enter message. Use the 2aB softkey to change the Input Method. When User inputs the message User wants to send, press OK, then use the navigation keys to select the line from which User wants to send, then Send.

5. If User wants to delete the message, after viewing the message, press Del, then User has three options to choose from: Yes, All, No.

4.8.2 Memo

User can add some memos to record some important things to remind User.

Press Menu → Application → Memo → Enter → Add.

There are some options to configure: Mode, Date, Time, Text, Ring. When the configuration is completed, press Save.

4.8.3 Ping

1. Input the IP User wants, then User press “start”. User can also press "delete" for modifying IP and change the input method when User inputs errors.
2. User waits till “OK” is shown on LCD, meaning Ping is successful, when User finishes entering the IP. Otherwise, Ping fails.

4.8.4 Voice Mail

1. **Press Menu → Application → Voice Mail → Enter.**

2. Use the navigation keys to highlight the line for which User wants to set, press Edit, and use the navigation key to turn on the mode, and then input the number. Press 2aB softkey to choose the proper input method.

3. Press Save to save the change.

4. To view the new voicemail, press the Voicemail softkey directly. Press Dial, and then User may be prompted to enter the password. User can listen to new and old messages.

4.9 Programmable Key Configuration

The phone has 4 programmable keys which are able to set up many functions. The following list shows the functions User can set on the programmable keys and provides a description for each function. The default configuration for each key is N/A which means the key hasn't been set for any functions.

Set the type as Memory Key

Press Menu → Settings → Basic Settings → Enter → Keyboard → DSS Key Settings

User have two options: Line Key Settings and Function Key Settings. Choose one User wants to make the assignment. Use the navigation key to choose the type as memory key. In the Dial field, User has some options, such as Normal, Speed Dial, Intercom, BLF, Presence, MWI and Call Park.

Speed dial

User can configure the key as a simplified speed dial key. This key function allows User to easily access User most dialed numbers.

Intercom

User can configure the key for Intercom code and it is useful in an office environment as a quick access to connect to the operator or the secretary.

BLF (Busy Lamp Field)

BLF is also called "Busy lamp field", and it is used to prompt the user to pay attention to the state of the object that has been subscribed, and used to cooperate with the server to pick up the phone call. User can configure the key for Busy Lamp Field (BLF) which allows User to monitor the status (idle, ringing, or busy) of other SIP accounts. User can dial out on a BLF configured key. Please refer to "LED Instructions" for more details about the LED status in different situations.

Note	In the Web interface, User can also set the pickup number to activate the pickup function. For example, if User sets the BLF number as 212, and the pickup number is 189, then when there is an incoming call to 212, press the BLF key, it will call out the 189 automatically to pick up the incoming call on 212.

Presence

Presence is called present, and compared to the BLF, it can also check whether object is online.

Note	User can subscribe to the BLF and presence station of the same number at the same time.

MWI (Message-Waiting Indicator)

When the key is configured as MWI, User is allowed to access voicemail quickly by pressing this key.

Call Park

1. User needs to set a server number when User has set what represents Call park. If User has a call but busy to receive the call, User can press the key and hear a number. Then User can choose another phone and input this number, so User can directly recover call.

2. Set the type as Line
   User can set these keys as line keys. When pressing it, it will enter dialer interface.

3. Set the type as Key Event
   User can set these keys as Key Event, and the subtype has many options.
   Choose one and it will have corresponding function.

- None
- Auto Redial Off
- Auto Redial On
- Call Back
- Call Forward
- DND
- Flash
- Headset
- History
- Hold
- Hot Desking: Pressing the key, User can clear all sip information and register your sip information.
- Join
- Lock: Pressing the key, User can lock the keyboard.
- Memo
- MWI
- Phonebook
- Pickup
- Prefix
- Redial
- Release: Pressing the key, User can end the call.
- SMS
- Transfer
- Power Light
- Hot Desking

4. Set the type as DTMF
   User can configure the key as DTMF. This key function allows User to easily dial or edit dial number.

5. Set the type as URL
   User needs to match an XML Phonebook address. By pressing the button, User can directly access the corresponding remote phonebook.

6. Set the type as BLF List Key
   It needs the cooperation with the Broadsoft server. The traditional BLF is that every number will need to be subscribed, so if the numbers that are subscribed are so many, it will cause obstruction. However, BLF List Key will put the numbers that are needed to be subscribed in a group. The phone uses the URL of the group to subscribe and analyze the specific information of each number such as number, name, state and so on according to the notifications from the server. Then set the idle Memory key as BLF List Key, later if the state of an object changes, the corresponding LED will change.

5 Other Functions

5.1 Auto Handdown

1. **Press Menu → Features → Enter → Auto Handdown → Enter.**
2. Set the Mode Enable through the navigation key, then set Time, unit is minute, then press Save.
3. When the call ends, after the time that User has set, the phone will return to the idle mode.

5.2 Ban Anonymous Call

1. **Press Menu → Features → Enter → Ban Anonymous Call → Enter.**
2. Choose which sip User want to enable Ban Anonymous Call, and then press Enter, choose Enabled or disabled through navigation key.
3. If User chooses Enabled, the others can't call the phone by anonymous. If User chooses Disabled, the others can call the phone by anonymous.

5.3 Dial Plan

1. **Press Menu → Features → Enter → Dial Plan → Enter.**
2. The following plans User can set: Press # to Send, Timeout to Send, Timeout, Fixed Length Number, Press # to Do BXFER, BXFER On-hook, AXFER On-hook. User can enable or disable each dial plan.

5.4 Dial Peer

1. **Press Menu → Features → Enter → Dial Peer → Enter.**
2. Press Add to enter the Edit interface, and then input some information. For example, Number: 1T, Dest.: 0.0.0.0, Port: 5060, Mode: SIP, Alisa: all:3333, Suffix: no suffix, Del Len: 0. Then press Save.
3. Input 1+number (1234) in the dial interface, User can dial out 3333. User can refer to 8.3.3.4 DIAL PEER.

5.5 Auto Redial

1. **Press Menu → Features → Enter → Auto Redial → Enter.**

2. Choose Mode Enabled or Disabled through the navigation key. If User chooses Enable, User also needs to set Interval and Times, and then press Save.

3. After enabling auto redial to call out someone, if he is in busy, it will pop up a prompt box whether to auto redial. Press OK and the phone will call out to him according the Interval and Time that User has set.

5.6 Call completion

1. **Press Menu → Features → Enter → Call Completion → Enter.**

2. Enable the function through the navigation key, and then save.

3. Call out others. If he is in busy, it will pop up a prompt Call Completion Waiting number. Press OK, when he is in idle. It will pop up a prompt Call Completion Call number. Press OK and the phone will call out the number automatically.

5.7 Ring From Headset

1. **Press Menu → Features → Enter → Ring From Headset → Enter.**

2. Enable this function through the navigation key. The phone connects to the headset. When the phone has an incoming call, it will ring from the headset.

5.8 Power Light

1. **Press Menu → Features → Enter → Power Light → Enter.**

2. Enable this function through the navigation key.

5.9 Hide DTMF

1. **Press Menu → Features → Enter→ Hide DTMF → Enter.**

2. Through the navigation key, choose: Disabled, All, Delay, Last Show. When User set up a call with

others and need to input the DTMF, the DTMF will show as User has set.

5.10 Ban Outgoing

1. **Press Menu → Features → Ban Outgoing → Enter.**

2. Enable this function; User cannot call any number.

5.11 Pre Dial

1. **Press Menu → Features → Pre Dial → Enter.**

2. Enable this function and User will realize Pre-Dial function.

5.12 Password Dial

1. **Press Menu → Features → Enter → Password Dial → Enter.**
2. Enable this function and User can also set Prefix and Length. For example, User wants to call out 1234567 and User sets Password Dial Prefix 123 and Password Length 3, then enter the dial interface and input 1234567, and then the screen will show 123***7.

5.13 Action URL & Active URI

1. Action URL: The action that the phone carries out. For example, opening DND can produce one URL, and then the phone can send the HTTP to get the URL to PC. The phone can report the action to the PC.

2. Active URI: Enter the web page of the phone, PHONE → FEATURE, input Active URI Limit IP. User can input internet server (e.g. PC'IP), PC can send one URL to the phone. The phone will produce one action; for example, open DND, so PC can control the phone.

5.14 Push XML

Enter the web page of the phone → PHONE → FEATURE, input Push XML Server(e.g. PC'IP), then PC can push text, SMS, phonebook, advertisement, execute, etc. To phone to update the message or the phone makes an action.

6 Basic settings

6.1 Keyboard

1. **Press Menu → Settings → Enter → Basic Settings → Enter → Keyboard → Enter.**

2. There are four items: DSS Key settings, Programmable Keys, Desktop Long Pressed, SoftKey, and User can set up respectively on them. Press the key Enter to the interface, then use the navigation keys to choose the function for the key according to User's requirements.

3. Press the key OK to save.

6.2 Screen Settings

1. **Press Menu → Settings → Enter → Basic Settings → Enter → Screen Settings → Enter.**

2. User can set Contrast, Contrast Calibration and Backlight by pressing Enter and use the navigation keys to set, and then press the key Save.

6.3 Ring Settings

1. **Press Menu → Settings → Enter → Basic Settings → Enter → Ring Settings → Enter.**

2. User can set Ring Volume and Ring Type by pressing Enter and use the navigation keys to set, and then press the key Save. In the Ring Type, the default system rings have nine and the custom ringtones have three that can be set through the web page.

6.4 Voice Volume

1. **Press Menu → Settings → Enter → Basic Setting → Enter → Voice Volume → Enter.**

2. Use the navigation keys to turn down or turn up the voice volume, and then press the key Save.

6.5 Time & Date

1. **Press Menu → Settings → Enter → Basic Settings → Enter → Time & Date → Enter.**

2. User has two options to choose from: Auto and Manual. Use the navigation keys to choose, and then press Save.

6.6 Greeting Words

1. **Press Menu ->Settings → Enter → Basic Settings → Enter → Greeting Words → Enter.**

2. User can enter the message and press Save. It will display on the phone screen when the phone starts up.

6.7 Language

1. **Press Menu → Settings → Enter → Basic Settings → Enter→ Language → Enter.**

2. The VIP-5060PT supports three languages. User can use the navigation keys to choose. The default two languages are English and Chinese.

7 Advanced Settings

7.1 Accounts

Press Menu → Enter → Advanced settings, and then input the password to enter. The default password is **123**. User can set it through the web page. Then choose Account and then press Enter. User can do some sip settings.

7.2 Network

Press Menu → Enter → Advanced settings, and then input the password to enter. Then choose Network and press Enter. User can do network settings by refering to 2.2.1 Network settings.

7.3 Security

Press Menu → Enter → Advanced settings, and then input the password to enter. Then choose Security to configure Menu Password, Key lock Password, Key lock Status and whether to ban Outgoing.

7.4 Maintenance

Press Menu → Enter → Advanced settings, and then input the password to enter the interface. Then choose Maintenance and press Enter. User can configure Auto Provision, Backup, and Upgrade.

7.5 Factory Reset

Press Menu → Enter → Advanced settings, and then input the password to enter the interface. Then choose Factory Reset and press Enter. User can choose Yes or No.

8 Web Configuration

8.1 Introduction of configuration

8.1.1 Ways to configure

The VIP-5060PT has three different ways for different users.

- Use phone keypad.
- Use web browser (recommended way).
- Use telnet with CLI command.

8.1.2 Password Configuration

There are two levels to access to phone: root level and general level. User with root level can browse and set all configuration parameters, while user with general level can set all configuration parameters except SIP (1-2) that some parameters cannot be changed, such as server address and port. User will have a different access level with different user name and password.

- Default user with root level:
  - User Name: **admin**
  - Password: **123**

The default password of phone screen menu is **123**.

8.2 Setting via web browser

When this phone and PC are connected to network, enter the IP address of the WAN or LAN port in this phone as the URL e.g. http:/192.168.0.X/

If User does not know the IP address, User can look it up on the phone's display by pressing Status button.

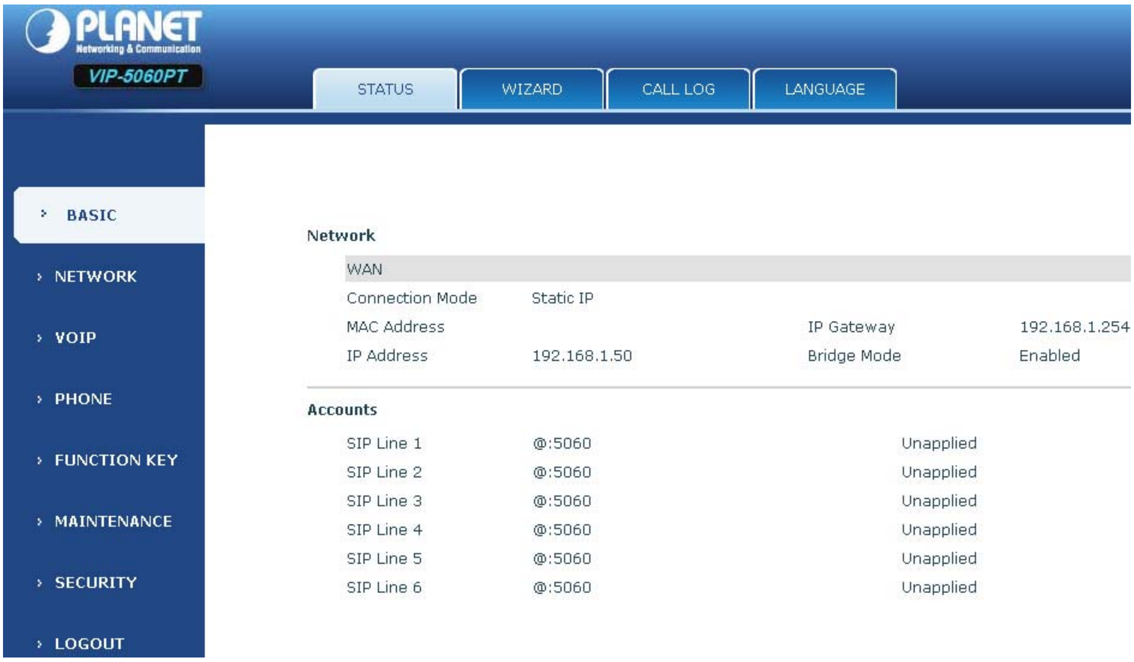
The login page is shown below:

After User configures the IP phone, User needs to click Save button in config under Maintenance on the left side of the screen to save User configuration. Otherwise, the phone will lose User modification after power is off and on.

8.3 Configuration via WEB

8.3.1 BASIC

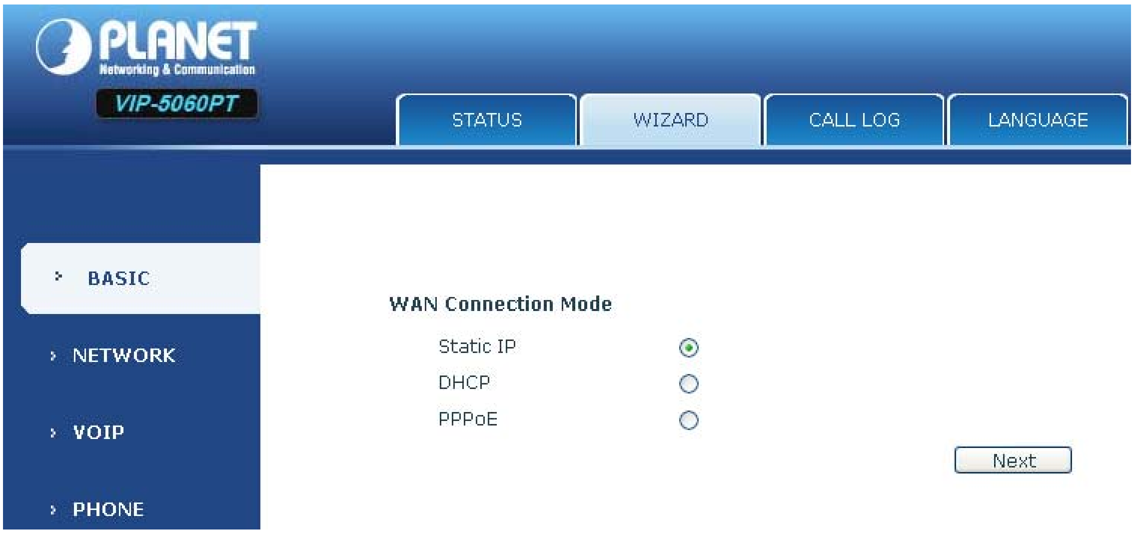
8.3.1.1 STATUS



Status

Field name	Explanation
Network	Shows the configuration information on WAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port
Accounts	Shows the phone numbers provided by the SIP LINE 1-6 servers The last line shows the version number and issued date.

8.3.1.2 WIZARD



Wizard

Please select the proper network mode according to the network condition. The VIP-5060PT provides three different network settings:

- **Static:** If User ISP server provides User with the static IP address, please select this mode, and then finish Static Mode setting. If User doesn't know about parameters of Static Mode setting, please refer to User ISP.

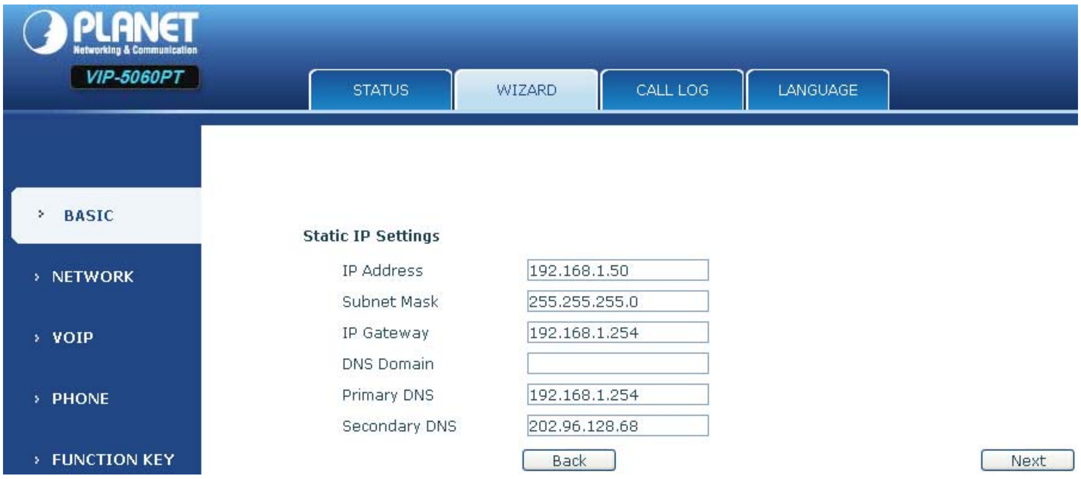
- **DHCP:** In this mode, User will get the information from the DHCP server automatically; need not have to input this information artificially.

- **PPPoE:** In this mode, User must input User ADSL account and password.

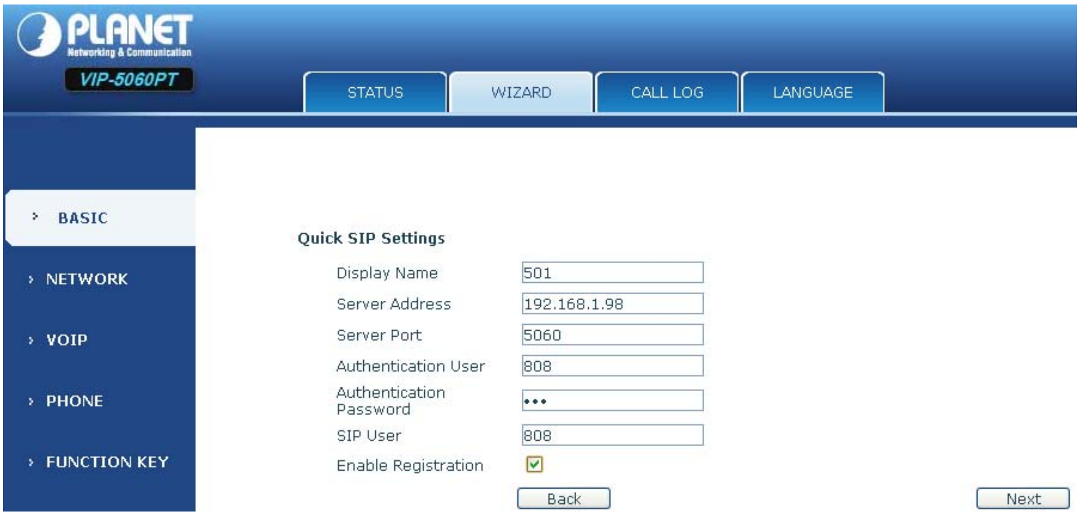
User can also refer to 2.2.1 Network setting to speedily set User network.

Choose Static IP mode and click【**NEXT**】to config the network and SIP (default SIP1)

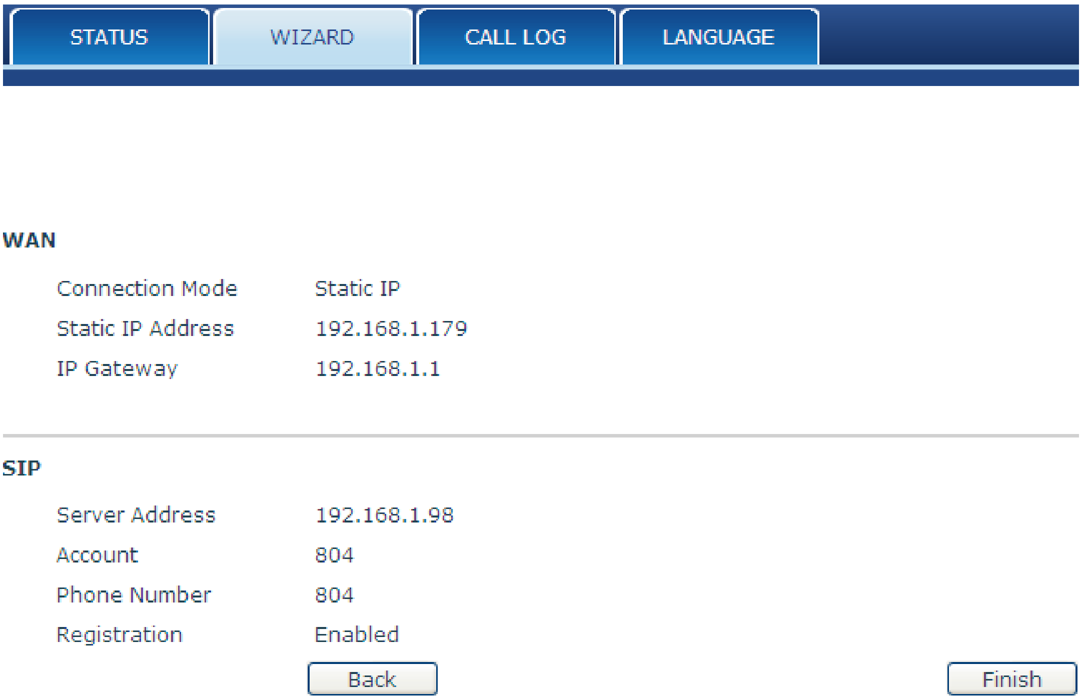
simply. Click【**BACK**】to return to the last page.



IP Address	Input the IP address distributed to User.
Subnet Mask	Input the subnet mask distributed to User.
IP Gateway	Input the Gateway address distributed to User.
DNS Domain	Set DNS domain postfix. When the domain which User input cannot be parsed, phone will automatically add this domain to the end of the domain which User input before and parse it again.
Primary DNS	Input User primary DNS server address.
Secondary DNS	Input User standby DNS server address.



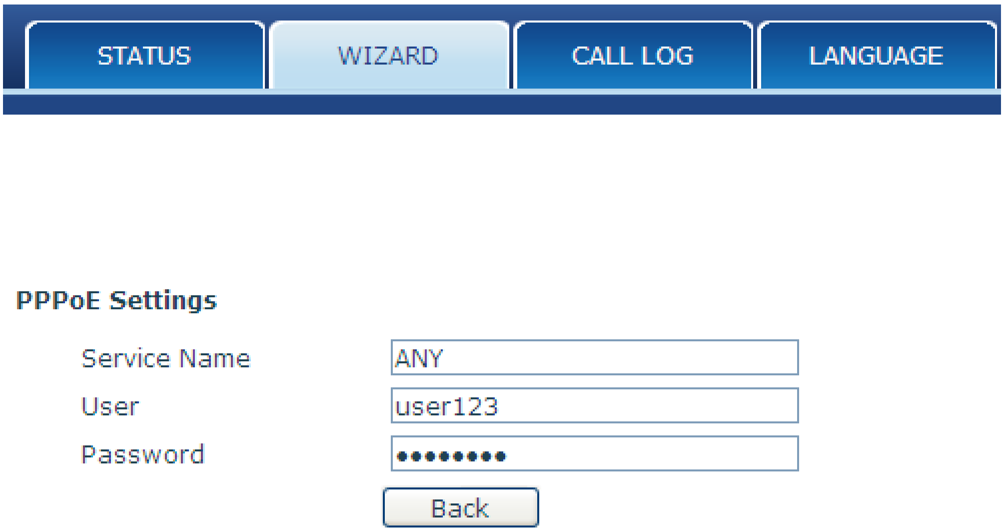
Display Name	Set the display name.
Server Address	Input User SIP server address.
Server Port	Set User SIP server port.
Authentication User	Input User SIP register account name.
Authentication Password	Input User SIP register password.
SIP User	Input the phone number assigned by User VOIP service provider.
Enable Registration	Start to register or not by selecting it or not.



Display detailed information about User manual config.

Choose DHCP mode and click Next to config SIP (default SIP1) simply. Click Back to return to the last page, like static IP mode.

Choose PPPoE mode and click Next to config the PPPoE account/password and SIP (default SIP1) simply. Click Back to return to the last page, like static IP mode.

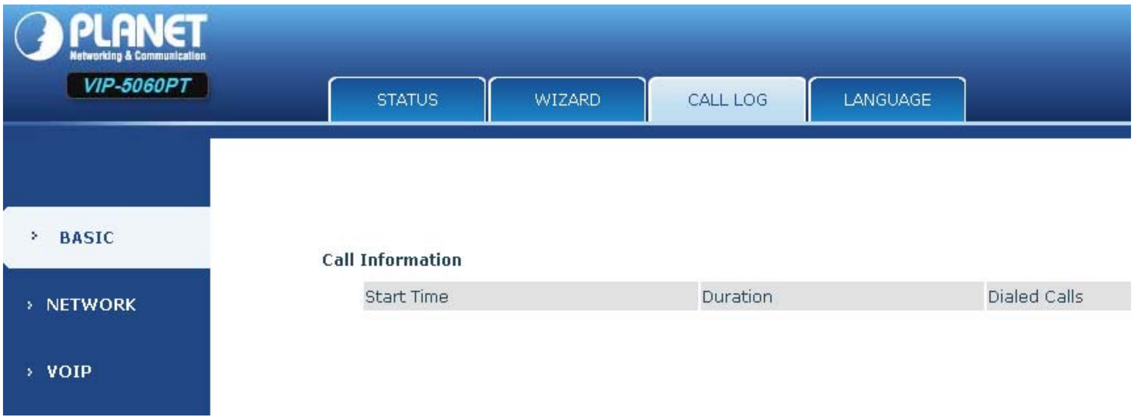


Service Name	It will be provided by ISP.
User	Input User ADSL account.
Password	Input User ADSL password.

Note	Click【**Finish**】button after User setting is done. IP Phone will save the setting automatically and reboot. After reboot, User can dial with the SIP account.

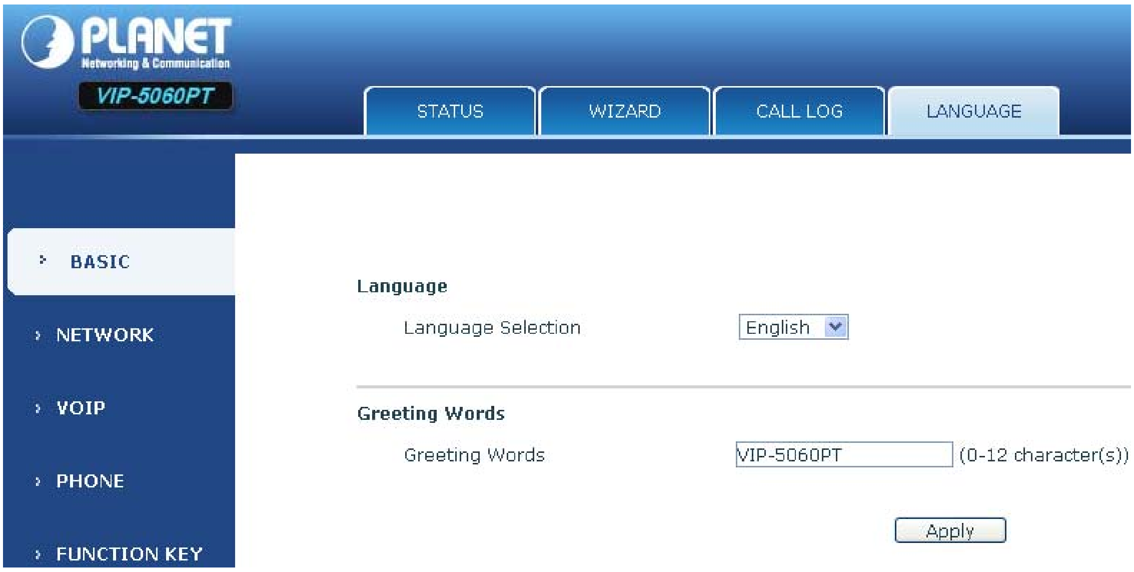
8.3.1.3 CALL LOG

User can check all the outgoing calls on this page shown below:



Call Log	
Field name	**Explanation**
Start Time	Display the start time of the outgoing record.
Duration	Display the conversation time of the outgoing record.
Dialed Calls	Display the account/protocol/line of the outgoing record.

8.3.1.4 LANGUAGE

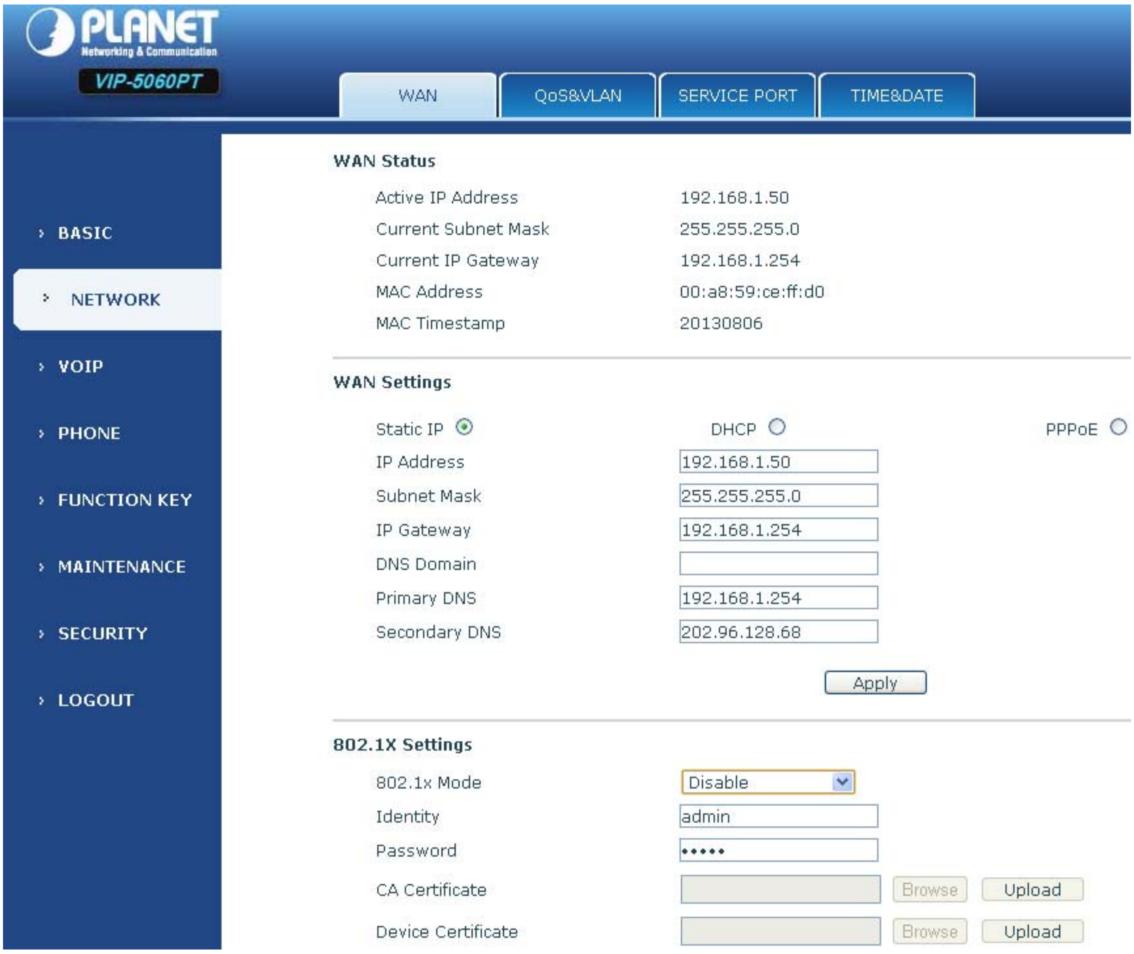


LANGUAGE	
Field name	**Explanation**
Language	Set the language of phone. English is default.
Greeting Words	The greeting words will display on LCD when phone is idle. It can support 12 chars.; the default chars are VOIP PHONE.

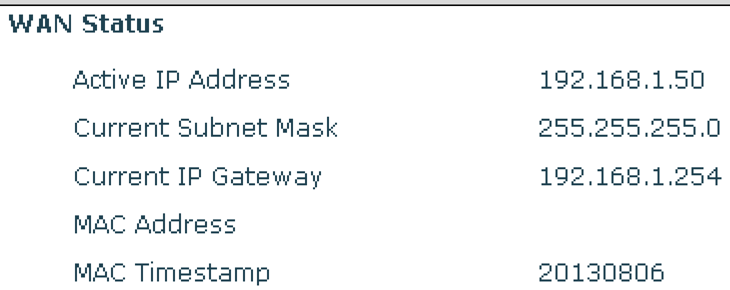
Note	The maximum length of the greeting message is 12 English characters and 5 Chinese characters.

8.3.2 NETWORK

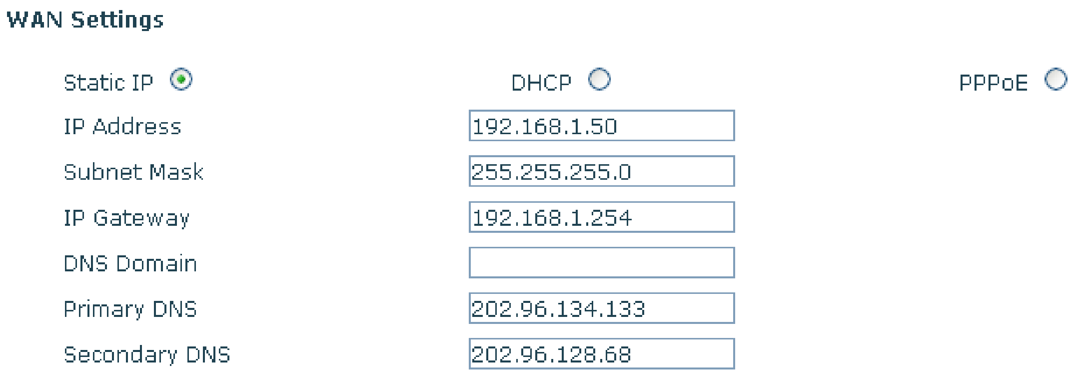
8.3.2.1 WAN



WAN Status



Active IP Address	The current IP address of the phone.
Current Subnet Mask	The current Network mask address.
MAC Address	The current MAC address of the phone.
Current IP Gateway	The current Gateway IP address.
MAC Timestamp	Shows the time of getting MAC address



Please select the proper network mode according to the network condition. The VIP-5060PT provides three different network settings:

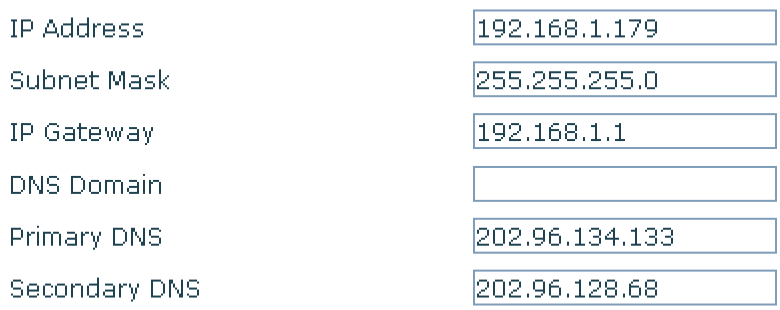
- **Static:** If User ISP server provides User with the static IP address. Please select this mode, and then finish Static Mode setting. If User doesn't know about parameters of Static Mode setting, please refer to User ISP.

- **DHCP:** In this mode, User will get the information from the DHCP server automatically; need not have to input this information artificially.

- **PPPoE:** In this mode, User must input User ADSL account and password.

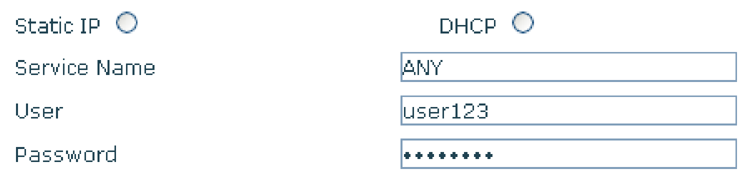
User can also refer to 2.2.1 Network setting to speedily set User network.

Obtain DNS server automatically	Select it to use DHCP mode to get DNS address. If User does not select it, User will use static DNS server. The default is selecting it.



If User uses static mode, User needs to set it.	
IP Address	Input the IP address distributed to User.
Subnet Mask	Input the Network mask distributed to User.
IP Gateway	Input the Gateway address distributed to User.
	Set DNS domain postfix. When the domain which User input

DNS Domain	cannot be parsed, phone will automatically add this domain to the end of the domain which User input before and parse it again.
Primary DNS	Input User primary DNS server address.
Secondary DNS	Input User standby DNS server address.



If User uses PPPoE mode, User need to make the above setting.	
Service Name	It will be provided by ISP.
User	Input User ADSL account.
Password	Input User ADSL password.

Note

1) Click “Apply” button after setting is done. IP Phone will save the setting automatically and new setting will take effect.
2) If User modifies the IP address, the web will not response by the old IP address. User needs to input new IP address in the address column to logon in the phone.

Note	VIP-5060PT LAN is fixed to **bridge mode**, so it doesn’t have programming page.

8.3.2.2 QoS&VLAN

The VOIP phone supports 802.1Q/P protocol and DiffServ configuration. VLAN functionality can use different VLAN IDs by setting signal/voice VLAN and data VLAN. The VLAN application of this phone is very flexible.

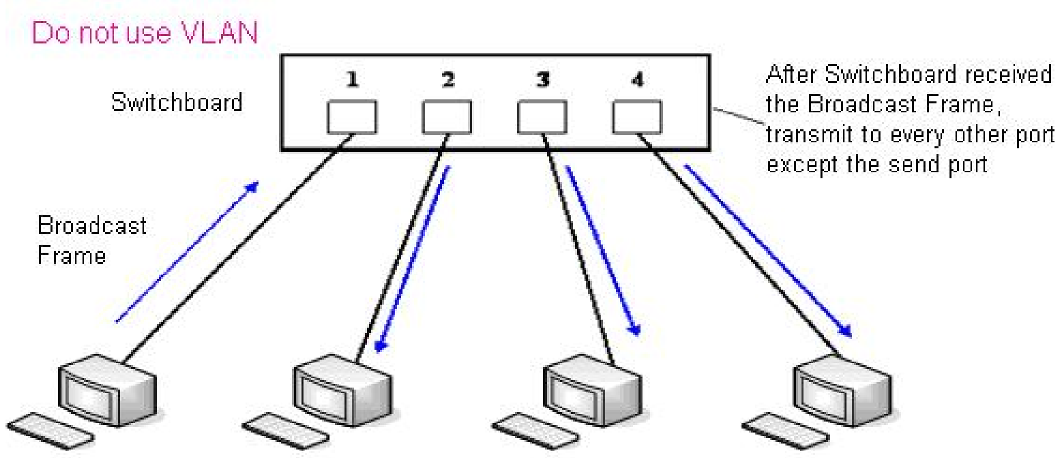


Chart 1

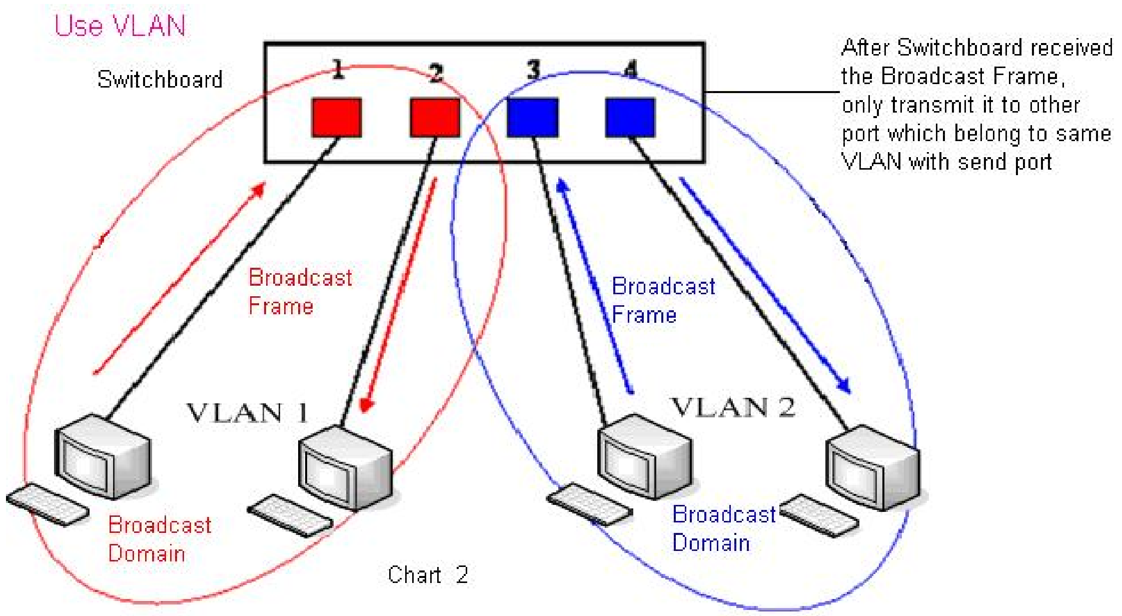
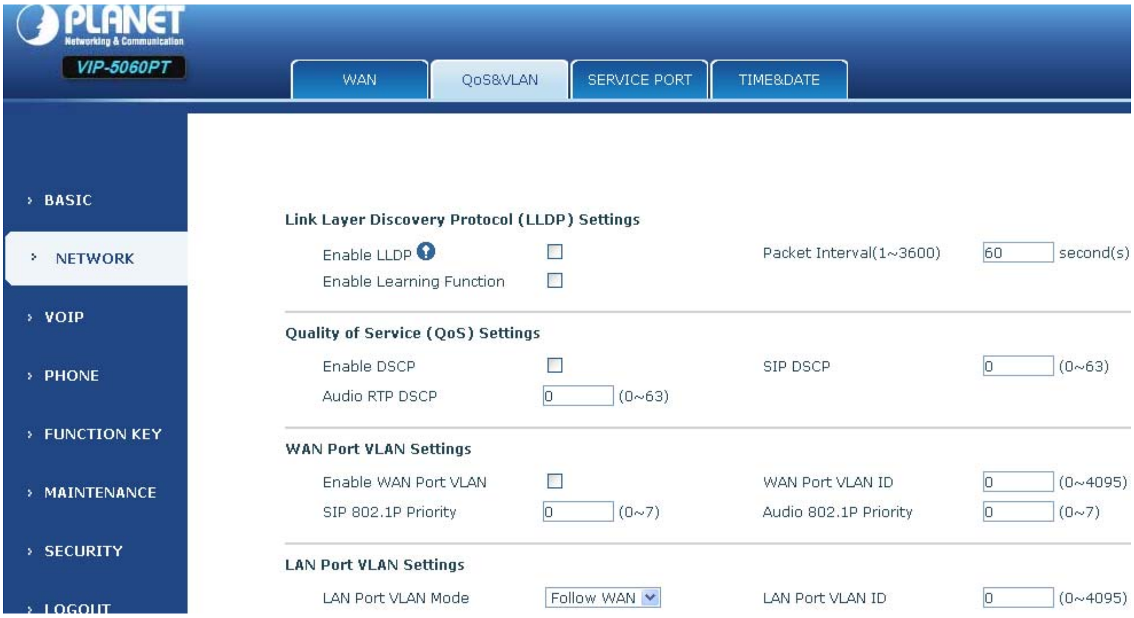


Chart 2

In chart 1, there is a layer 2 that switches go without setting VLAN. Any broadcast frame will be transmitted to the other ports except the send port. For example, a broadcast information is sent out from port 1 then transmitted to ports 2, 3 and 4.

In chart 2, red and blue indicate two different VLANs in the switch, and port 1 and port 2 belong to red VLAN, port 3 and port 4 belong to blue VLAN. If a broadcast frame is sent out from port 1, switch will transmit it to port 2, the other port in the red VLAN and not transmit it to port3 and port 4 in blue VLAN. By this means, VLAN divides the broadcast domain via restricting the range of broadcast frame transition.

Note	Chart 2 uses red and blue to identify the different VLANs, but in practice, VLAN uses different VLAN IDs to identify.



QoS Configuration	
Link Layer Discovery Protocol (LLDP) Settings	
Enable LLDP	Enable LLDP by selecting it.
Enable Learning Function	After enabling LLDP Learn, telephone can automatically learn the data of DSCP, 802.1p, VLAN ID from the switch. If the data is different from the data of the LLDP server, telephone will change its own value as the value of the switch (Synchronous with VLAN in switch).
Package Interval(1-3600)	The time interval of sending LLDP Packet.
Quality of Service (Qos) Settings	
Enable DSCP	Enable DSCP by selecting it.
SIP DSCP	Specify the value of the SIP DSCP.
Audio RTP DSCP	Specify the value of the Audio RTP DSCP.
WAN Port VLAN Settings	
Enable WAN Port VLAN	Enable WAN Port VLAN by selecting it.
WAN Port VLAN ID	Specify the value of the WAN Port VLAN ID, the range of the value is 0-4095.
SIP 802.1p Priority	Specify the value of the sip 8021.p priority, the range of the value is 0-7.
Audio 802.1p Priority	Specify the value of the audio 802.1p priority, the range of the value is 0-7.
LAN Port VLAN Settings	
LAN Port VLAN Mode	Follow WAN: Follow the WAN ID. Disable: Disable Port VALN. Enable: Enable Port VLAN and specify the Port VLAN ID

	different from WAN ID.
LAN Port VLAN ID	Specify the value of the Port VLAN ID different from WAN ID, the range of the value is 0-4095.

8.3.2.3 SERVICE PORT

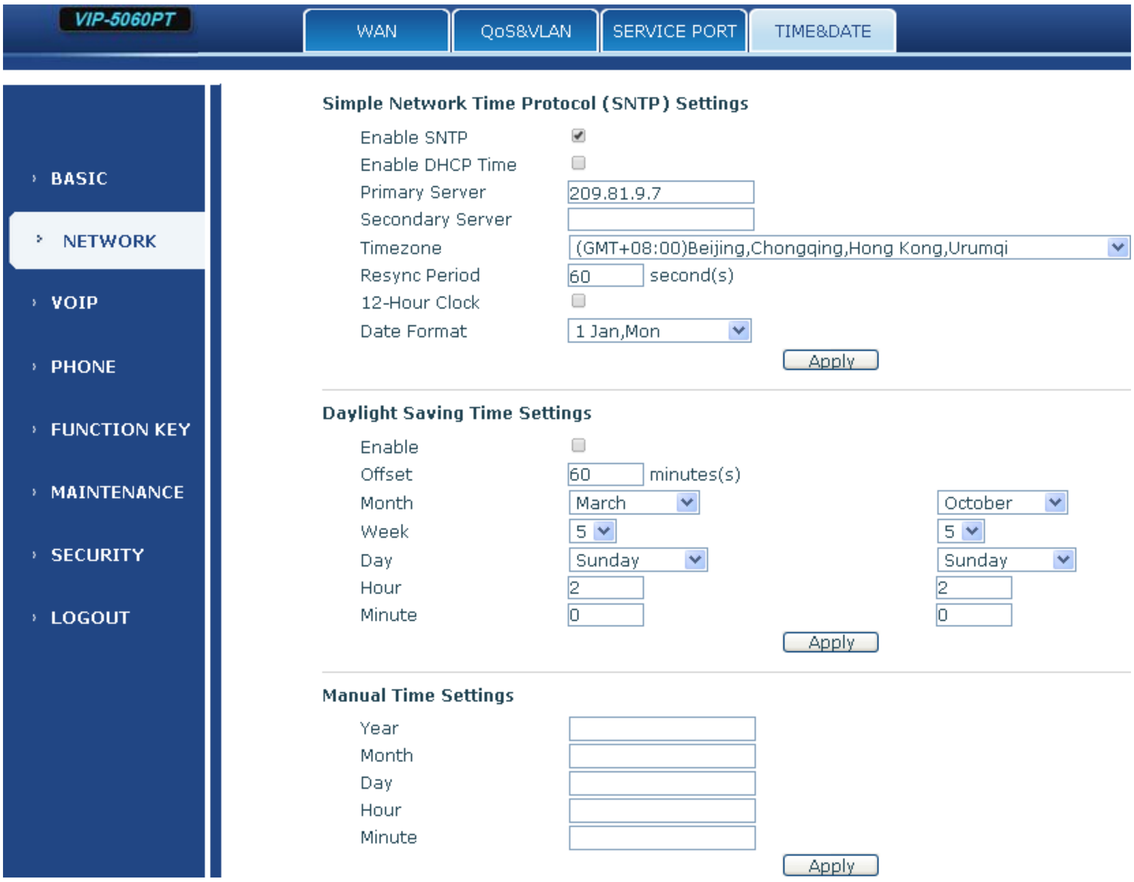
User can set the port of telnet/HTTP/RTP on this page.

SERVICE PORT	
Field name	**Explanation**
Service Port Settings	
Web Server Type	Specify Web Server Type.
HTTP Port	Set web browser port, the default is 80 port, if User want to enhance system safety, User would be better change it into non-80 standard port; Example: The IP address is 192.168.1.70, and the port value is 8090, the accessing address is http://192.168.1.70:8090.
HTTPS Port	Before using the https, User must download https authentication certification into the phone, then Set web browser port, the default is 443 ports; if User want to enhance system safety, User would be better change it into non-443 standard port. User can access to the web in https after rebooting the phone.
Telnet Port	Set Telnet Port, the default is 23. User can change the value into others. Example: The IP address is 192.168.1.70. The telnet port value is 8023; the accessing address is telnet 192.168.1.70 8023.
RTP Port Range Start	Set the RTP Start Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.

Note	
	1) User needs to save the configuration and reboot the phone after setting this page. 2) Please reboot the system if User modifies the HTTP or telnet port number (the new number should be greater than 1024). 3) If User sets 0 for the HTTP port, it will disable HTTP service.

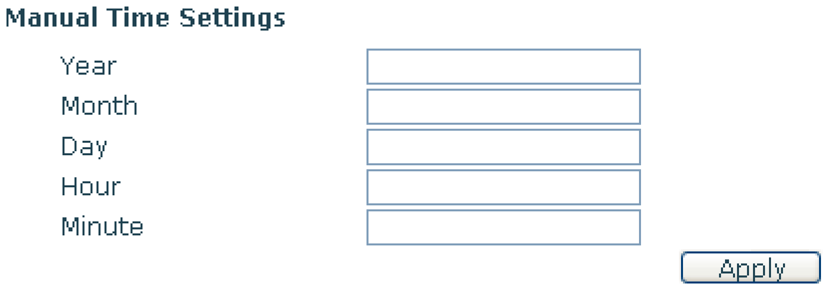
8.3.2.4 TIME&DATE

Setting time zone and SNTP (Simple Network Time Protocol) server according to User location, User can also manually adjust date and time in this web page**.**



TIME&DATE	
Field name	**Explanation**
Simple Network Time Protocol (SNTP) Settings	
Enable SNTP	Enable SNTP by selecting it.
Enable DHCP Time	Enable DHCP Time by selecting it, then the

	phone will automatically synchronize the standard time.
Primary Server	Set SNTP Primary Server IP address.
Secondary Server	Set SNTP Secondary Server IP address.
Time Zone	Select the Time zone according to User location.
Resync Period	Set the time out, the default is 60 seconds.
12 -Hour Clock	Switch the time mechanism between 12 hours and 24 hours. Default is 24 hours mode.
Date format	Specify the date format.
Daylight Saving Time Settings	
Enable	Enable daylight saving time.
Offset(minutes)	Setup the variety length.
Month	Setup start and end month.
Week	Setup start and end week.
Day	Setup start and end day.
Hour	Setup start and end hours.
Minute	Setup start and end minutes.
Manual Time Settings	

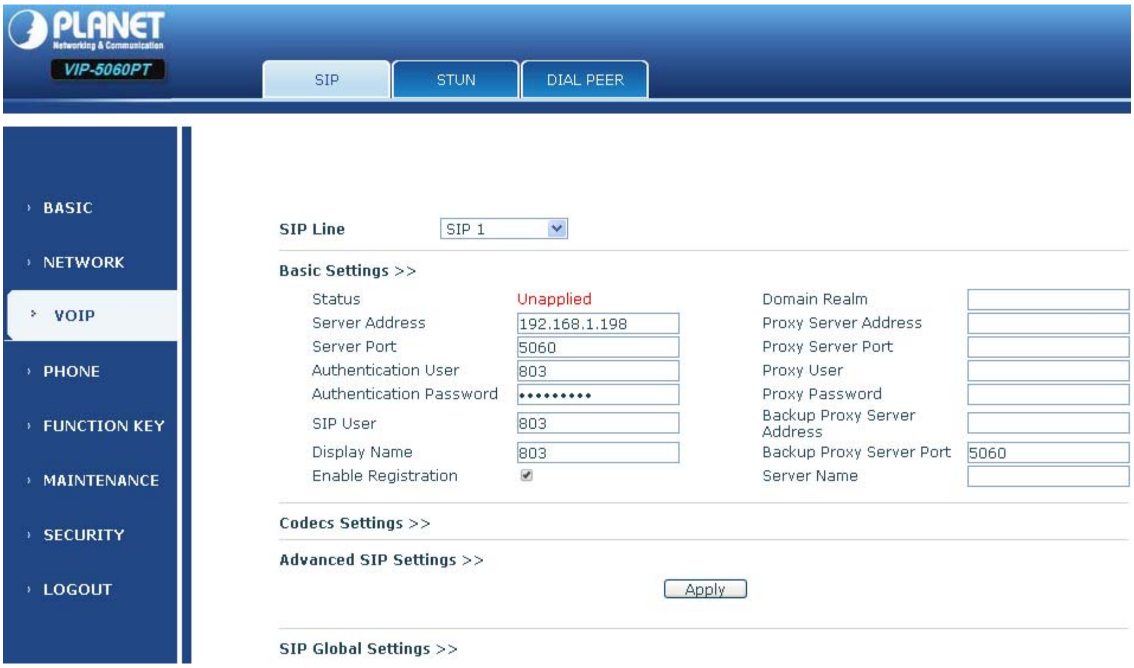


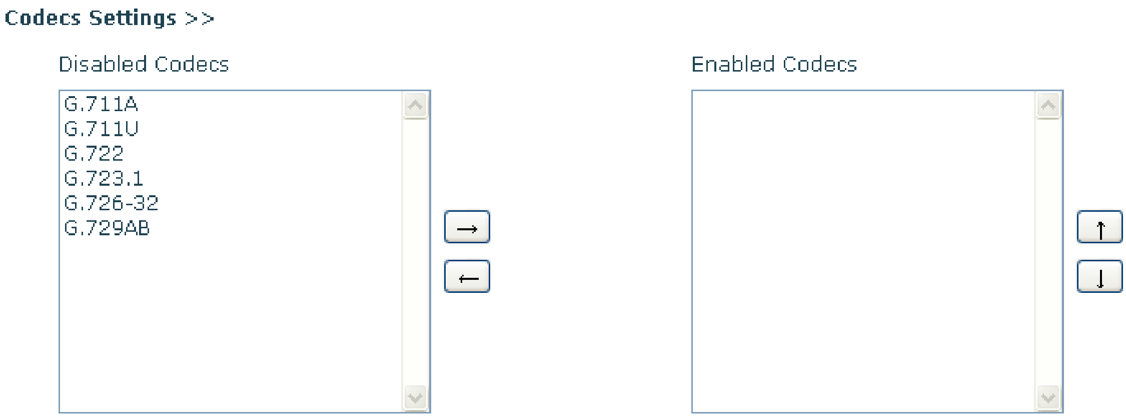
Note	First of all, User needs to disable the SNTP service, and the date hour minute each of which is required to complete and submit to make manual.

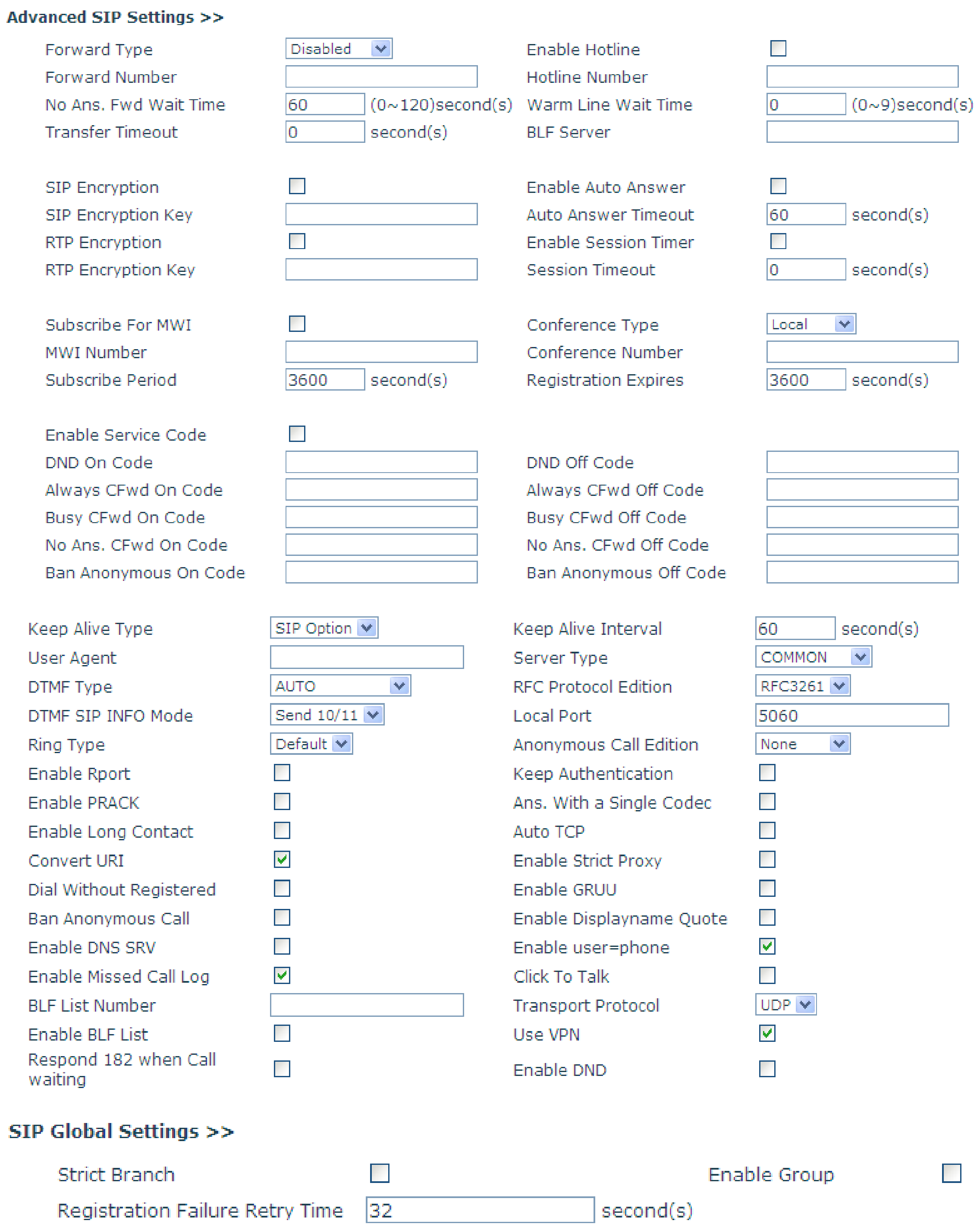
8.3.3 VOIP

8.3.3.1 SIP

Set User SIP server in the following interface.







SIP Config	
Field name	**Explanation**
SIP Line	
Choose line to set info about SIP, there are 4 lines to choose. User can switch by **【Load】** button.	
Basic Settings	
Status	Shows if the phone has been registered the SIP server or not;

	or so, show Unapplied.
Server Address	Input User SIP server address.
Server Port	Set User SIP server port.
Authentication User	Input User SIP register account name.
Authentication Password	Input User SIP register password.
SIP User	Input the phone number assigned by User VoIP service provider. Phone will not register if there is no phone number configured.
Display Name	Set the display name.
Proxy Server Address	Set proxy server IP address (Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if User VoIP service provider gives different configurations between Register SIP Server and Proxy SIP Server, User need make different settings).
Proxy Server Port	Set User Proxy SIP server port.
Proxy User	Input User Proxy SIP server account.
Proxy Password	Input User Proxy SIP server password.
Domain Realm	Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
Backup Server Address	Input the Backup Server Address, if the primary server is unavailable, then the phone will enable the Backup Server Address.
Backup Server Port	Specify the Backup Server Port.
Enable Registration	Start to register or not by selecting it or not.
Codecs Settings	
Disable Codecs/Enable Codecs	Use the navigation keys to highlight the desired one in the Enable/Disable Codecs list, and press the desired to move to the other list.
Advanced SIP Setting	
Forward Type	Select call forward mode, the default is Off. **Off:** Close down calling forward. **Busy:** If the phone is busy, incoming calls will be forwarded to the appointed phone. **No answer:** If there is no answer, incoming calls will be forwarded to the appointed phone after a specific. **Always:** Incoming calls will be forwarded to the appoint phone immediately. The phone will prompt the incoming while doing forward.
Forward Number	Specify the number User want to forward.
No Answer Forward Wait	Specify the No Answer Forward Delay Time, if the Forward

Time	Type is No answer, incoming calls will be forwarded after the no answer forward wait time.
Enable Hot Line	Specify Hot Line by selecting it.
Hot Line Number	Specify Hot Line Number, the phone dial the hot line number automatically at hands-free mode or handset mode after warm line time.
Warm Line Wait Time	Specify the Warm Line Time.
Transfer Timeout	For the phone supports the transfer of certain special features server, set interval time between sending “bye” and hanging up after the phone transfers a call.
BLF Server	The registered server will be gotten subscription package from ordinary application of BLF phone, please enter the BLF server, when the sever dose not support subscription package. then the registered server and subscription server will be separate
SIP Encryption	Enable/Disable SIP Encryption.
SIP Encryption Key	Set the key for sip encryption.
RTP Encryption	Enable/Disable RTP encryption.
RTP Encryption Key	Set the key for RTP encryption.
Enable Auto Answer	Enable Auto Answer by selecting it.
Auto Answer Timeout	Specify Auto Answer Time, the phone auto answers the incoming call after Auto Answer Time.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Session Timeout	Set the session timeout.
Subscribe for MWI	Enable the Subscribe for MWI by selecting it, the phone will send subscribe message for MWI to the SIP Server.
MWI Number	Specify the MWI Number; Please contact User system administrator for the connecting code. Different systems have different codes.
Subscribe Period(s)	Overtime of resending subscribe packet. Suggest using the default configuration.
Conference Type	Specify the Conference Type, if User select the local, User needn’t input the conference number.
Conference Number	Specify the network conference number, please contact User system administrator for the network conference number.
Registration Expire(s)	Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expired time set, the phone will change automatically the time into the time recommended by the server, and register again.
Enable Service Code	If User want to realize the following function by the server,

	please enter the On Code and Off Code option, then when User choose to enable/disable following function on User IP phone, it will send message to the server, and the server will turn on/off the function immediately.
DND On Code	Set the DND On Code, When User press the DND hot key, the phone will send a message to the server, and the server will turn on the DND function. Then any calls to the extension will be rejected by the server automatically. And the incoming call record will not be displayed in the Call History.
DND Off Code	Set the DND Off Code, When User press the DND hot key, the phone will send a message to the server, and the server will turn off the DND function.
Always CFwd On Code	Set the Always CFwd On Code, when User choose to enable the always forward function on User phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will always forward it to the set number automatically. And the IP phone will not show the record in the call history anymore.
Always CFwd Off Code	Set the Always CFwd Off Code, when User choose to disable the always forward function on User phone, it will send message to the server, and the server will turn off the function immediately.
Busy CFwd On Code	Set the Busy CFwd On Code, when User choose to enable the busy forward function v on User phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
Busy CFwd Off Code	Set the Busy CFwd Off Code, when User choose to disable the busy forward function on User phone, it will send message to the server, and the server will turn off the function immediately.
No Answer CFwd On Code	Set the No Answer CFwd On Code, when User choose to enable the on answer forward function on User phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
No Answer CFwd Off Code	Set the No Answer CFwd Off Code, when User choose to disable the busy forward function on User phone, it will send message to the server, and the server will turn off the function immediately.

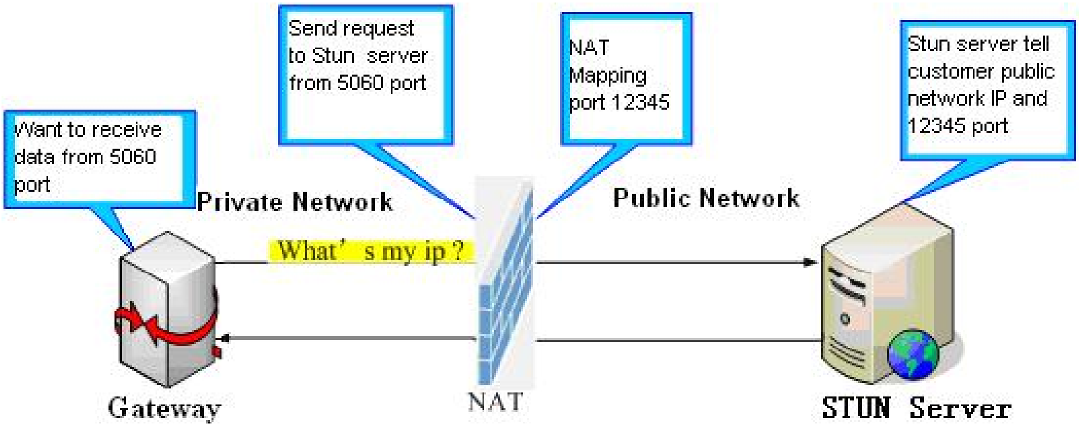
Anonymous On Code	Set the Anonymous On Code, When User choose to enable the anonymous call function on User IP phone, it will send information to the server, and the server will enable the anonymous call function for User IP phone automatically.
Anonymous Off Code	Set the Anonymous Off Code, When User chooses to disable the anonymous call function on User IP phone, it will send information to the server, and the server will disable the anonymous call function for User IP phone automatically.
Keep Alive Type	Specify the keep alive type, if the type is option, the phone will send option sip message to server every NAT Keep Alive Period(s), then the server responses with 200 to keep alive. If the type is UDP, the phone will send UDP message to server to keep alive every NAT Keep Alive Period(s).
Keep Alive Interval	Set examining interval of the server, default is 60 seconds.
User Agent	Set the user agent if have, the default is VoIP Phone 1.0.
DTMF Type	Select DTMF sending mode, there are three modes: ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO Different VoIP Service providers may provide different modes.
Local Port	Set sip port of each line.
Ring Type	Set ring type of each line.
Enable Via Rport	Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT.
Enable PRACK	Enable or disable SIP PRACK function, suggest use the default config.
Enable Long Contact	Set more parameters in contact field; connection with SEM server.
Convert URI	Convert # to %23 when send the URI.
Dial Without Registered	Set call out by proxy without registration.
Ban Anonymous Call	Set to ban Anonymous Call.
Enable DNS SRV	Support DNS looking up with _sip.udp mode.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
RFC Protocol Edition	Select SIP protocol version to adapt for the SIP server which uses the same version as User select. For example, if the server is CISCO5300, User need to change to RFC2543; else phone may not cancel call normally. System uses RFC3261 as default.
Transport Protocol	Set transport protocols, TCP or UDP.
Anonymous call Edition	Set Anonymous call out safely; Support RFC3323and RFC3325.

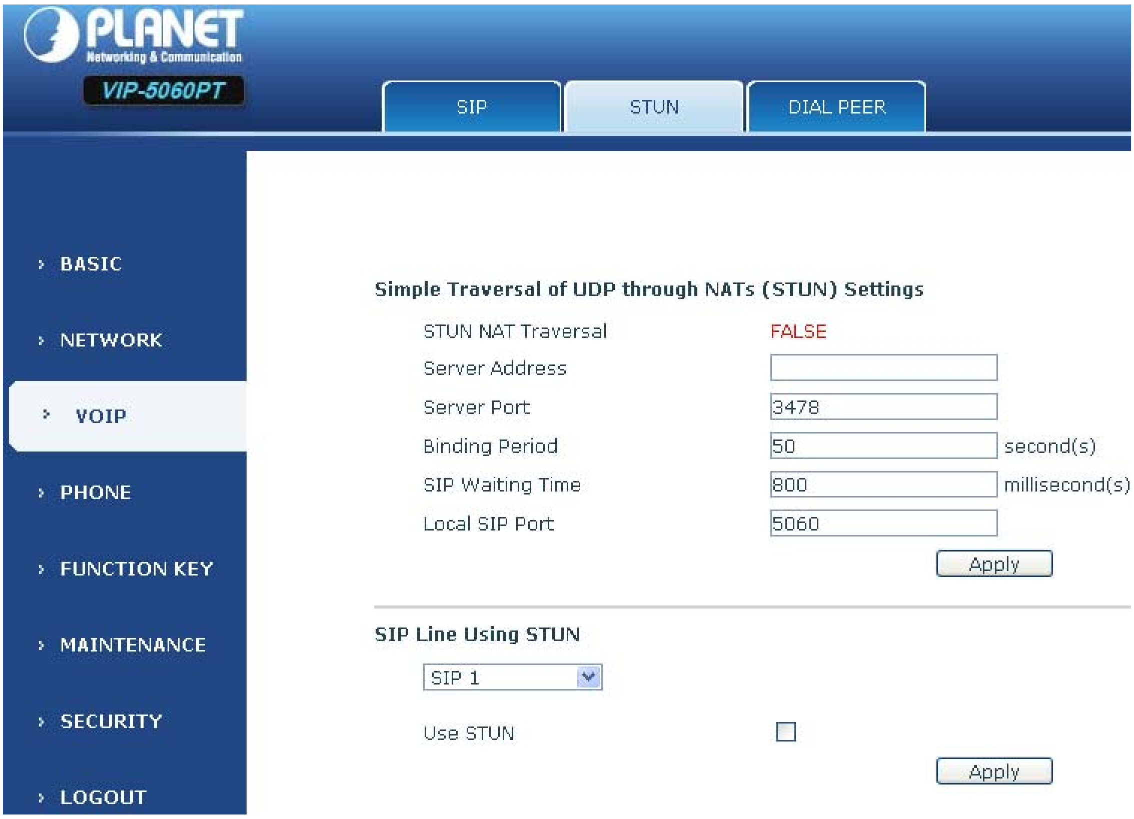
Keep Authentication	Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.
Answer With A Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Auto TCP	Set to use automatically TCP protocol to guarantee usability of transport as message is above 1300 byte
Enable Strict Proxy	Support the special SIP server-when phone receives the packets sent from server, phone will use the source IP address, not the address in via field.
Enable GRUU	Set to support GRUU
Enable Display name Quote	Set to make quotation mark to display name as the phone sends out signal, in order to be compatible with server.
Enable user = phone	Enable user = phone by selecting it, it is contained in the invite sip message, in order to be compatible with server.
Enable Missed Call Log	Enable the missed call log by it, the phone will save the missed call log into the call history record and display the missed calls on the idle screen, or won't save the missed call log into the call history record and display the missed calls on the idle screen.
Click to talk	Set click to Talk (need practical software support).
Enable BLF List	Enable BLF List by selecting it, BLF list is a function which can monitor the group status, it is not one to one monitoring, but the information feedback from the server to decide which BLF list will monitor.
BLF List Number	Specify the BLF List Number.
SIP Global Settings	
Strict Branch	Enable the Strict Branch, the value of the branch must be in the beginning of z9hG4k in via field of the invite sip message received, or the phone won't response to the invite sip message. Notice: the deployment will become effective in all sip lines.
Enable Group	Enable Group by selecting it, then the phone enable the sip group backup function. Notice: the deployment will become effective in all sip lines.
Registration Failure Retry Time	Specify the registration failure retry time, if the phone register failed, the phone will register again after registration failure retry time. Notice: the deployment will become effective in all sip lines.

8.3.3.2 STUN

In this web page, Users can config SIP STUN.

STUN: By STUN server, the phone in private network could know the type of mapping IP and port of SIP. The phone might register itself to SIP server with global IP and port to realize the device both calling and being called in private network.





STUN	
Field name	**Explanation**
Simple Traversal of UDP through STUN Settings	
STUN Traversal	Shows STUN Transverse estimation, true means STUN can

	penetrate NAT, while False means not.
Server Address	Set User SIP STUN Server IP address.
Server Port	Set User SIP STUN Server Port.
Blinding Period(s)	Set STUN blinding period(s). If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need send a STUN packet to keep the mapping effective and alive.
SIP Waiting Time	Specify the sip wait stun time; User can input the time depended on User network condition.
Local SIP Port	Configure the local SIP port, default port is 5060 (the port with immediate effect, after revision, SIP calls will use the modified port.

SIP Line Using STUN

Choose line to set info about SIP, There are 2 lines to choose. User can switch by **【Load】** button.

Use STUN	Enable/Disable SIP STUN.

Note: SIP STUN is used to realize SIP penetration to NAT. If User phone configures STUN Server IP and Port (default is 3478), and enable SIP Stun, User can use the ordinary SIP Server to realize penetration into NAT.

8.3.3.3 DIAL PEER

This functionality offers User more flexible dial rule; User can refer to the following content to know how to use this dial rule. When User wants to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, User can set number 156 to replace 192.168.1.119 here.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

When User want to dial a long distance call to Beijing, User need dial an area code 010 before local phone number, but User can also dial number 1 instead of 010 after we make a setting according to this dial rule. For example, User want to dial 01062213123, but User need dial only 162213123 to realize User long distance call after User make this setting.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
IT	0.0.0.0	5060	SIP	no alias	no suffix	0

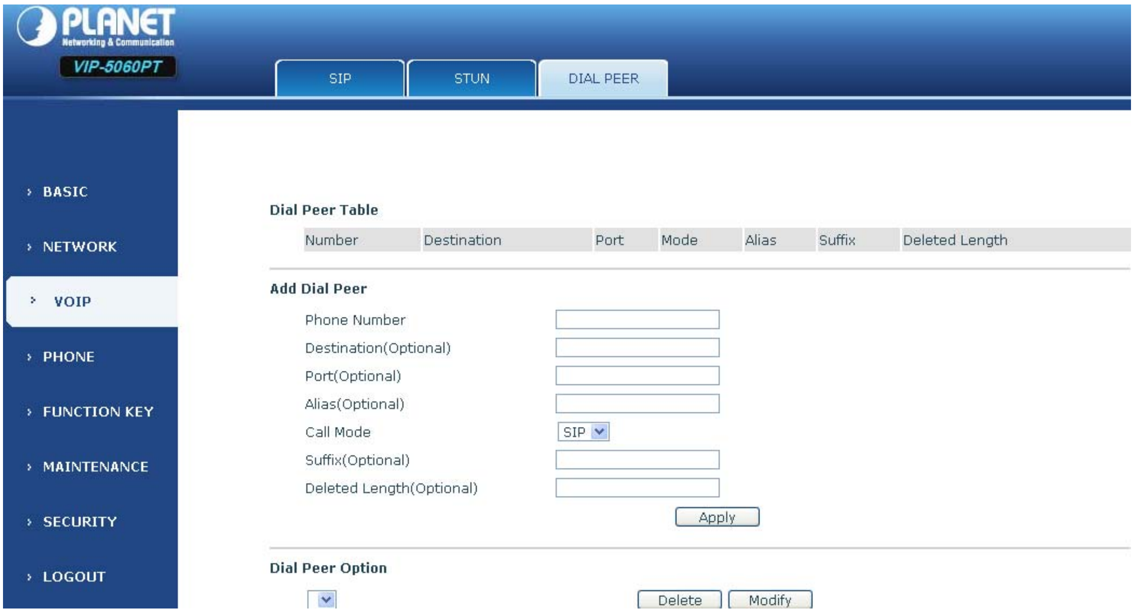
To save the memory and avoid abundant input of user, add the follow functions:

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
IT	0.0.0.0	5060	SIP	no alias	no suffix	0
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
156	192.168.1.119	5060	SIP	no alias	no suffix	0

1.* Match any single digit that is dialed.
If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

2. [ ] Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.
If user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically.
Use this phone User can realize dialing out via different lines without switch in web interface.



DIAL PEER	
Field name	**Explanation**
Phone number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, User need input User desired phone number in this blank, and then User need dial the phone number to realize calling to what the phone number is mapped. In the prefix matching, User need input User

	desired prefix number and T; then dial the prefix and a phone number to realize calling to what User prefix number is mapped. The prefix number supports at most 30 digits.
Destination	Set Destination address. This is optional config item. If User want to set peer to peer call, please input destination IP address or domain name. If User want to use this dial rule on SIP2 line, User need input 255.255.255.255 or 0.0.0.2 in it.SIP3 into 0.0.0.3
Port	Set the Signal port, the default is 5060 for SIP.
Alias	Set alias. This is optional config item. If User don't set Alias, it will show no alias.

Note

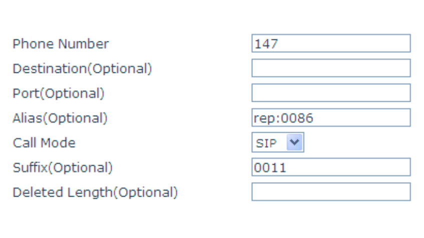
There are four types of aliases.

1) Add: xxx, it means that User need dial xxx in front of phone number, which will reduce dialing number length.
1) All: xxx, it means that xxx will replace some phone number.
2) Del: It means that phone will delete the number with length appointed.
3) Rep: It means that phone will replace the number with length and number appointed.
4) User can refer to the following examples of different alias application to know more how to use different aliases and this dial rule.

Call Mode	Select different signal protocol, SIP
Suffix	Set suffix, this is optional config item. It will show no suffix if User don't set it.
Delete Length	Set delete length. This is optional config item. For example: if the delete length is 3, the phone will delete the first 3 digits then send out the rest digits. User can refer to examples of different alias application to know how to set delete length.

Examples of different alias applications

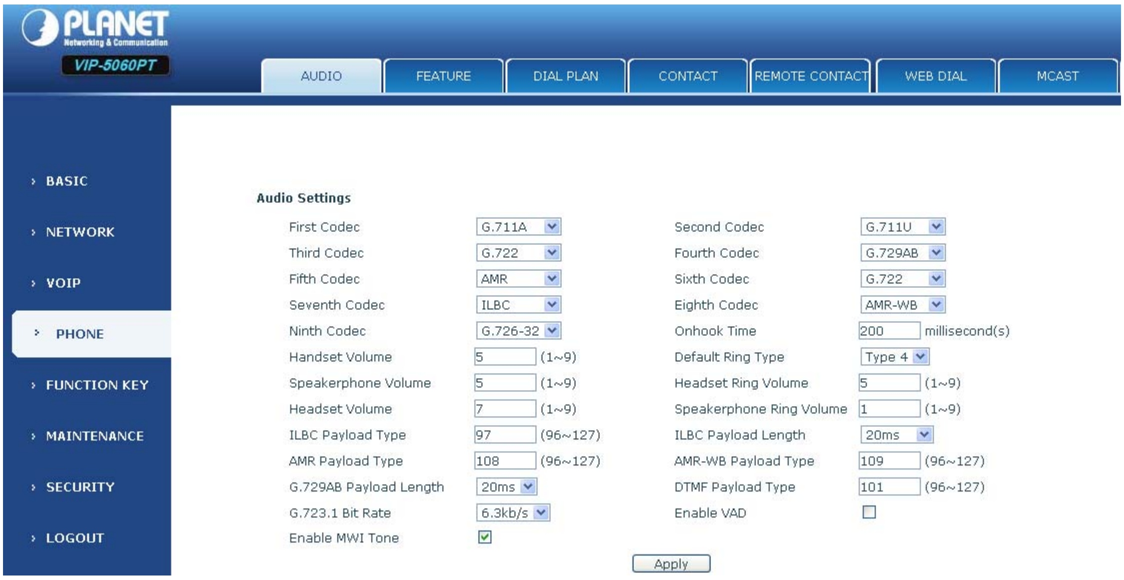
Set by web	Explanation	Example
Add Dial Peer Phone Number: 9T Destination(Optional): 255.255.255.255 Port(Optional): Alias(Optional): del Call Mode: SIP Suffix(Optional): Deleted Length(Optional): 1 Apply	User need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. This means any phone No. that starts with User set phone number will be sent via SIP2 line after the first several digits of User dialed phone number are deleted according to delete length.	If User dials “93333”, the SIP2 server will receive “3333”.
Phone Number: 2 Destination(Optional): Port(Optional): Alias(Optional): all:33334444 Call Mode: SIP Suffix(Optional): Deleted Length(Optional):	This setting will realize speed dial function, after User dialing the numeric key “2”, the number after all will be sent out.	When User dial “2”, the SIP1 server will receive 33334444.
Phone Number: 8T Destination(Optional): Port(Optional): Alias(Optional): add:0755 Call Mode: SIP Suffix(Optional): Deleted Length(Optional):	The phone will automatically send out alias number adding User dialed number, if User dialed number starts with User set phone number.	When User dial “8309“, the SIP1 server will receive “07558309”.
Phone Number: 010T Destination(Optional): Port(Optional): Alias(Optional): rep:0086 Call Mode: SIP Suffix(Optional): Deleted Length(Optional): 3	User need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx If User dialed phone number starts with User set phone number, the first digits same as User set phone number will be replaced by the alias number specified and New phone number will be send out.	When User dial “0106228”, the SIP1 server will receive “86106228”.

	If User dialed phone number starts with User set phone number. The phone will send out User dialed phone number adding suffix number.	When User dial "147", the SIP1 server will receive "1470011".

8.3.4 PHONE

8.3.4.1 AUDIO

On this page, User can configure voice codec, input/output volume and so on.

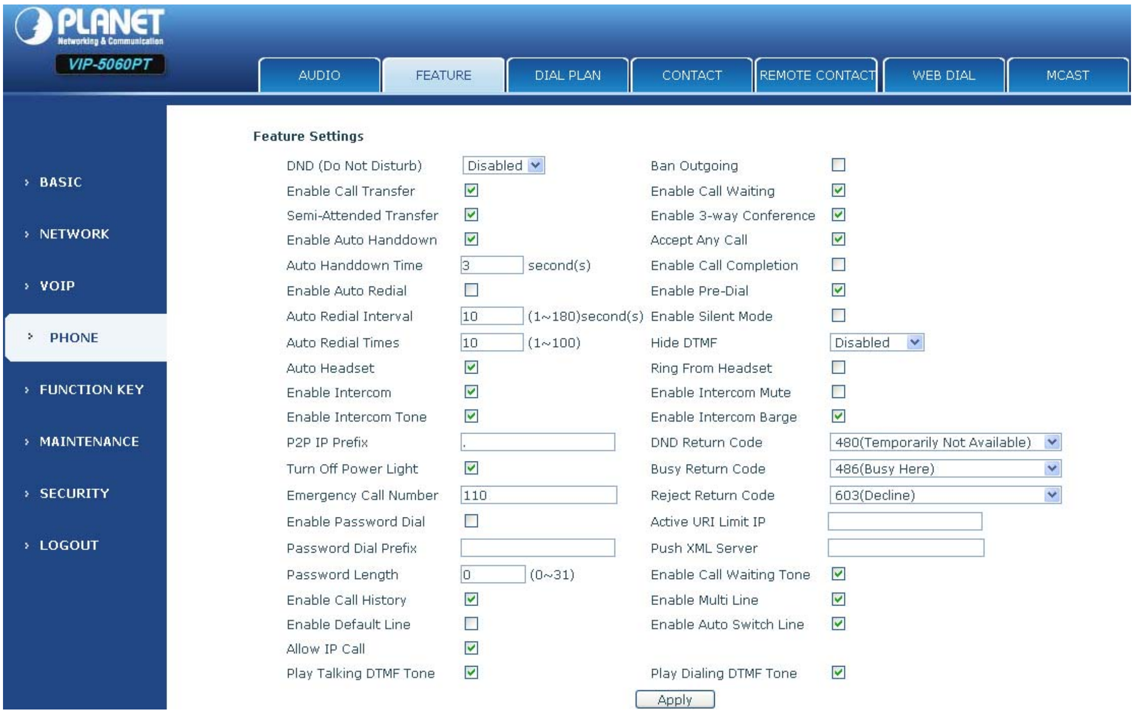


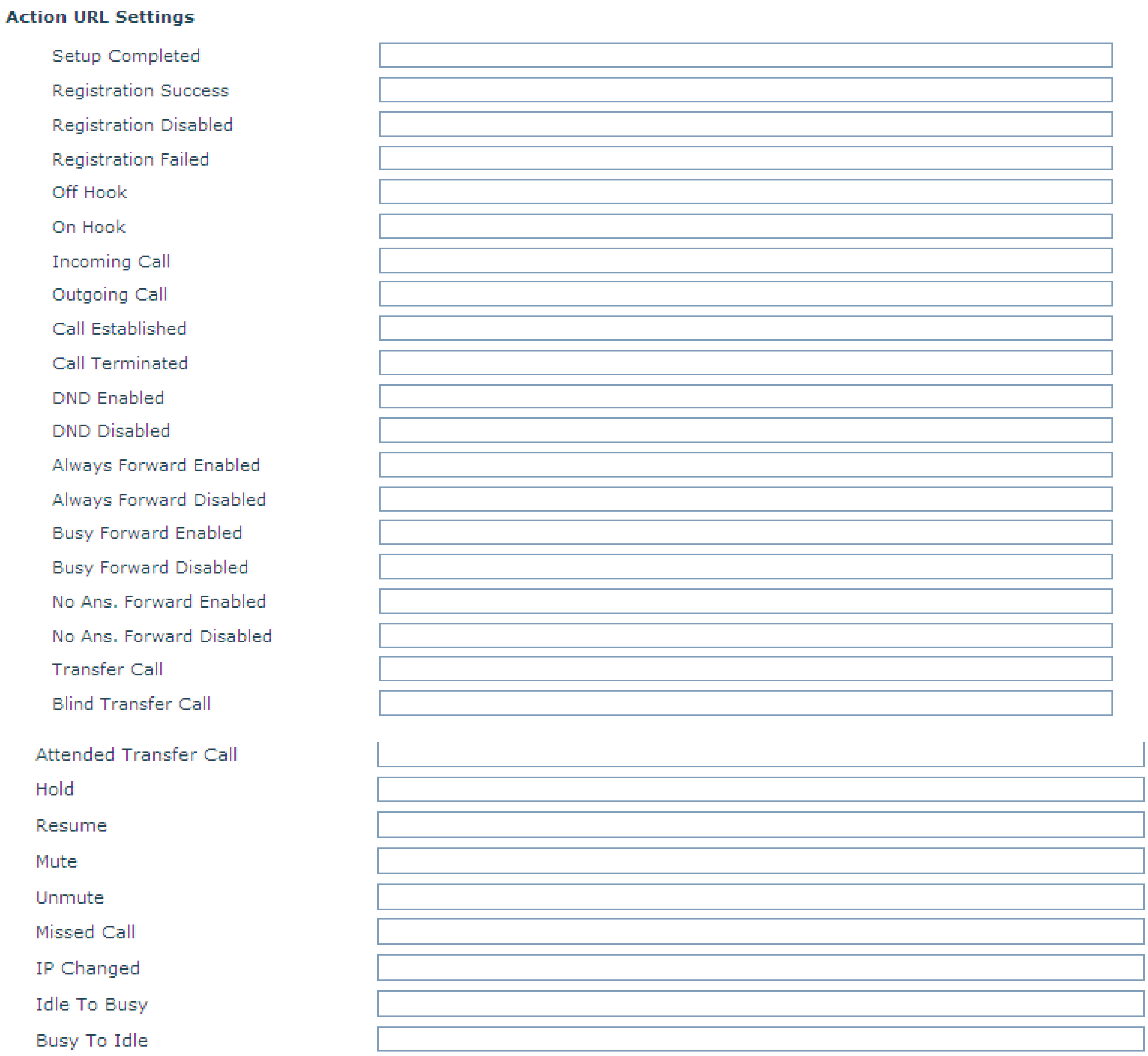
AUDIO Configuration	
Field name	**Explanation**
First Codec	The first preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB,None.
Second Codec	The second preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB,None.
Third Codec	The third preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB,None.
Fourth Codec	The forth preferential DSP codec: G.711A/u, G.722, G.723.1, 726-32 G.729AB, None.
Fifth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723.1, 726-32 G.729AB, None.
Sixth codec	The sixth preferential DSP codec: G.711A/u, G.722, G.723.1, 726-32 G.729AB, None.

Handset Input Volume	Specify Input (MIC) Volume grade.
G729AB Payload Length	Set G729 Payload Length.
Onhook Time	Specify the least reflection time of Hand down, the default is 200ms.
Default Ring Type	Select Ring Type.
Handset Output Volume	Specify Output (receiver) Volume grade.
Speakerphone volume	Specify Speakerphone Volume grade.
Ring Volume	Specify Ring Volume grade.
G722 Timestamps	160/20ms or 320/20ms is available.
G723.1 Bit Rate	5.3 kb/s or 6.3 kb/s is available.
Tone Standard	Select Tone Standard.
Enable VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.
DTMF Payload Type	Set DTMF Payload Type.

8.3.4.2 FEATURE

In this web page, User can configure Hotline, Call Transfer, Call Waiting, 3 Ways Call, Black List, white list Limit List and so on.





FEATURE	
Field name	**Explanation**
Do Not Disturb	Select DND, the phone will reject any incoming call, the callers will be reminded by busy, but any outgoing call from the phone will work well.
Ban Outgoing	If User select Ban Outgoing to enable it, and User cannot dial out any number.
Enable Call Transfer	Enable Call Transfer by selecting it.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it.
Enable Auto Redial	Enable Auto Redial by selecting it, then the phone reminds whether redial, when the caller is busy or rejects.

Auto Redial interval	Specify the Auto Redial interval.
Auto Redial Times	Specify the Auto Redial interval.
Auto Headset	Open this function, if there is a headphones in VIP-5060PT, User can press " answer" key or line key to answer a call with the headset
Enable Call Completion	Enable Call Completion by selecting it.
Enable Pre-Dial	Enable Pre-Dial
Enable Call Waiting	Enable Call Waiting by selecting it. Then the phone reminds whether redial, when the caller is busy or rejects. if it's ok and the phone finds out that the caller is idle by sip message, it will reminds whether redial.
Enable Call Waiting Tone	Turn off this feature, User will not hear issued a " beep" sound with more calls.
Enable 3-way Conference	Enable 3-way conference by selecting it.
Accept Any Call	If select it, the phone will accept the call even if the called number is not belong to the phone.
Enable Auto Hand down	The phone will hang up and return to the idle automatically at hands-free mode.
Auto Hand down Time	Specify Auto Hand down Time, the phone will hang up and return to the idle automatically after Auto Hand down Time at hands-free mode, and play dial tone Auto Hand down Time at handset mode.
Ring From Headset	Enable Ring From Handset by selecting it, the phone plays ring tone from handset.
Enable Intercom	Enable Intercom Mode by selecting it.
Enable Intercom Mute	Enable mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call.
Enable Silent Mode	Enable Silent Mode by selecting it, the phone light will red blink to remind that there is a missed call instead of playing ring tone.
Turn Off Power Light	Enable Turn Off Power Light by selecting it.
Emergency Call Number	Specify the Emergency Call Number. Despite the keyboard is locked, User can dial the emergency call number.
Enable Password Dial	Enable Password Dial by selecting it, When number entered is beginning with the password prefix, the following N numbers

	After the password prefix will be hidden as *, N stand for the value which User enter in the Password Length field. For example: User set the password prefix is 3, enter the Password Length is 2, then User enter the number 34567, it will display 3**67 on the phone.
Password Dial Prefix	Specify the prefix of the password call number.
Password Length	Specify the Password length.
DND Return Code	Specify DND Return code.
Busy Return Code	Specify Busy Return Code.
Reject Return Code	Specify Reject Return Code.
Hide DTMF	Specify the hide DTMF mode.
Push XML Server	Specify the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what User want to dial is 192.168.1.119, If User define P2P IP Prefix as 192.168.1., User dial only #119 to reach 192.168.1.119. Default is “.”. If there is no “.” Set, it means to disable dialing IP.
Active URI Limit IP	Specify the server IP that remote control phone for corresponding operation.
Action URL Settings	
Action URL Settings	Specify the Action URL that Record the operation of phone; send this corresponding information to server, url: http://InternalServer /FileName.xml? (Internal Server is server IP. Filename is name of xml that contains the action message).
Block Out Settings	
Block out	Set Add/Delete Limit List. Please input the prefix of those phone numbers which User forbid the phone to dial out. For example, if User want to forbid those phones of 001 as prefix to be dialed out, User need input 001 in the blank of limit list, and then User cannot dial out any phone number whose prefix is 001. X and are wildcard x means matching any single digit. For example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to dialed out means matching any arbitrary number digit. For example, 6 expresses any number with prefix 6 will be forbidden to dialed out.

Note: Black List and Limit List can record at most 10 items respectively.

8.3.4.3 DIAL PLAN

This system supports 4 dial modes:

1) End with “#”: dial User desired number, and then press #.

2) Fixed Length: the phone will intersect the number according to User specified length.

3) Time Out: After User stop dialing and waiting time out, system will send the number collected.

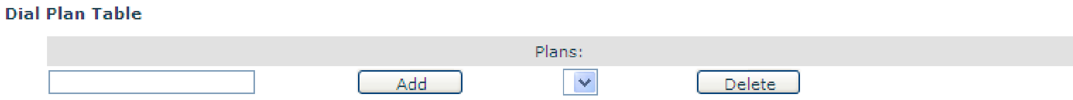
4) User defined: User can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing.

In order to keep some users' secondary dialing manner when dialing the external line with PBX, phone can be added a special rule to realize it. So user can dial a number as external line prefix and get the secondary dial tone to keep dial the external number. After finishing dialing, phone will send the prefix and external number totally to the server.

For example, there is a rule 9, xxxxxxxx in the digital map table. After dialing 9, phone will send the secondary dial tone, user may keep going dialing. After finished, phone will call the number which starts with 9; actually the number sent out is 9-digit with 9.

DIAL PLAN Configuration	
Field name	**Explanation**
Basic Setting	
Press "#" to Send	Set Enable/Disable the phone ended with “#” dial.
Dial Fixed Length	Specify the Fixed Length of phone ending with.

Send after （3-30） seconds	Set the timeout of the last dial digit. The call will be sent after timeout.
Press # to Do Blind Transfer	Enable Blind Transfer On Hook, when executing Blind Transfer End with #, press # after inputting the number that User want to transfer, the phone will transfer the current call to the third party.
Blind Transfer on OnHook	Enable Blind Transfer on On Hook, when executing Blind Transfer, hang up after inputting the number that User want to transfer, the phone will transfer the current call to the third party.
Attend Transfer on OnHook	Enable Attend Transfer on On Hook, when executing Attended Transfer, hang up after the third party answers, the phone will transfer the current call to the third party.



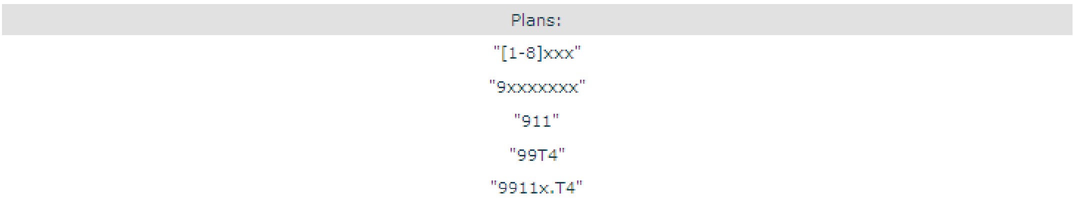
Below is user-defined digital map rule:

[ ] Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits.

* Match any single digit that is dialed.

. Match any arbitrary number of digits including none.

Tn Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans.



Cause extensions 1000-8999 to be dialed immediately.

Cause 8 digit numbers started with 9 to be dialed immediately.

Cause 911 to be dialed immediately after it is entered.

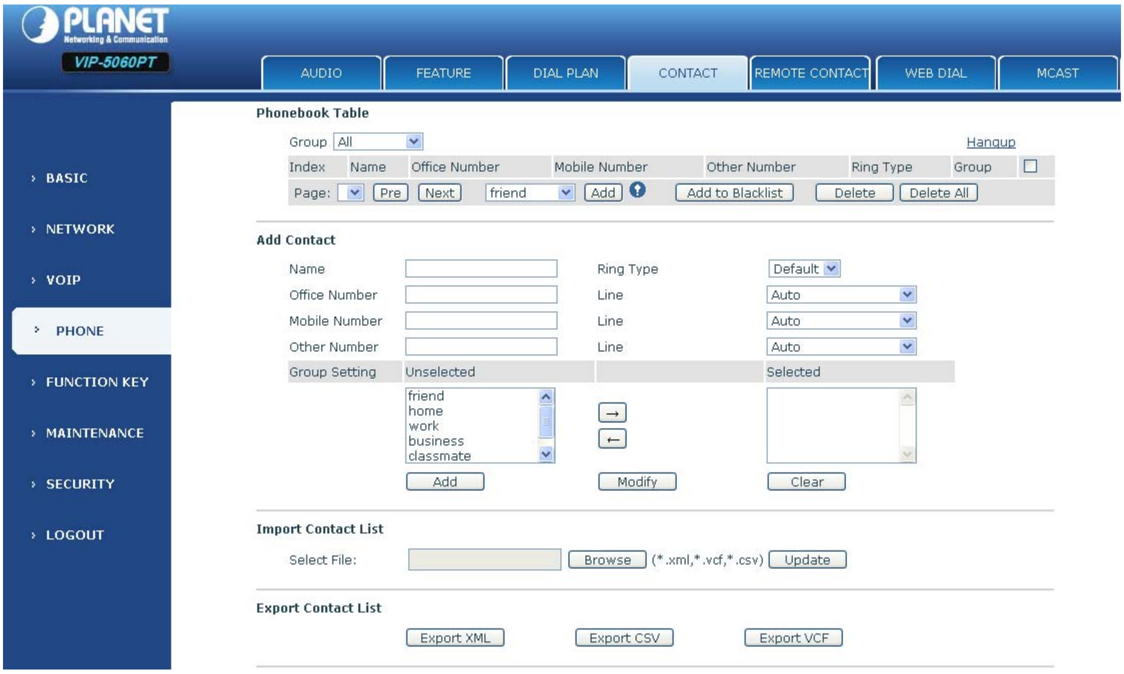
Cause 99 to be dialed after 4 seconds.

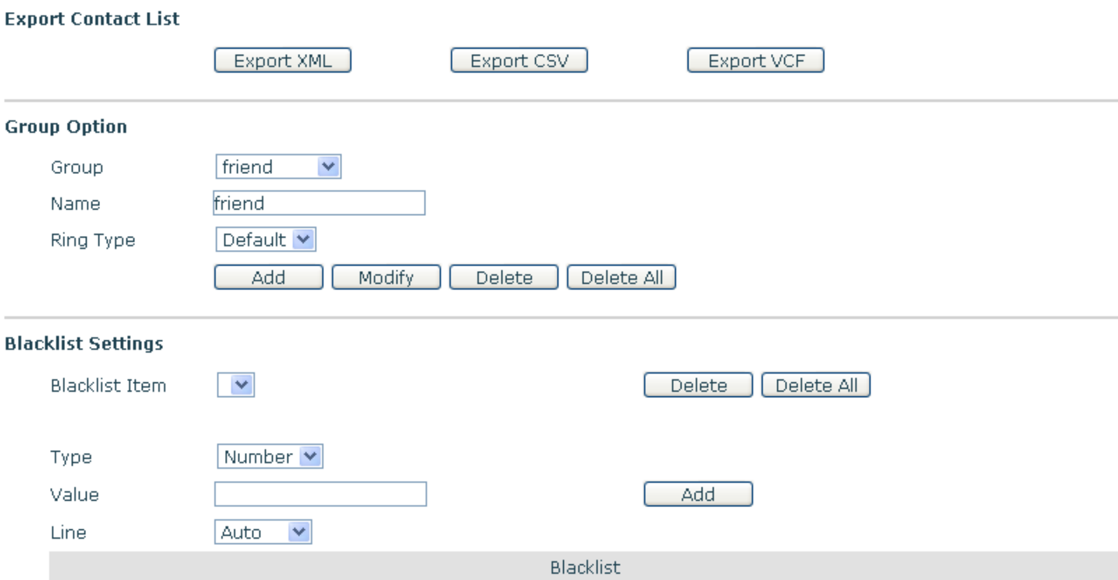
Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.

Note	End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously. System will stop dialing and send number according to User set rules.

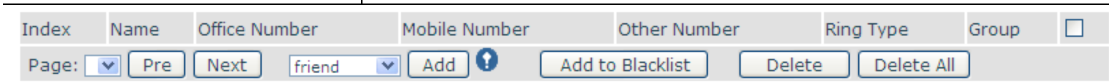
8.3.4.4 CONTACT

User can input the name, phone number and select ring type for each name here.

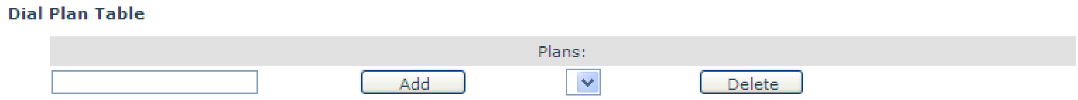




Contact	
Field name	**Explanation**
Phonebook Table	
Name	Shows the name corresponding to the phone number.



Send after （3-30） seconds	Set the timeout of the last dial digit. The call will be sent after timeout.
Press # to Do Blind Transfer	Enable Blind Transfer On Hook, when executing Blind Transfer End with #, press # after inputting the number that User want to transfer, the phone will transfer the current call to the third party.
Blind Transfer on OnHook	Enable Blind Transfer on On Hook, when executing Blind Transfer, hang up after inputting the number that User want to transfer, the phone will transfer the current call to the third party.
Attend Transfer on OnHook	Enable Attend Transfer on On Hook, when executing Attended Transfer, hang up after the third party answers, the phone will transfer the current call to the third party.



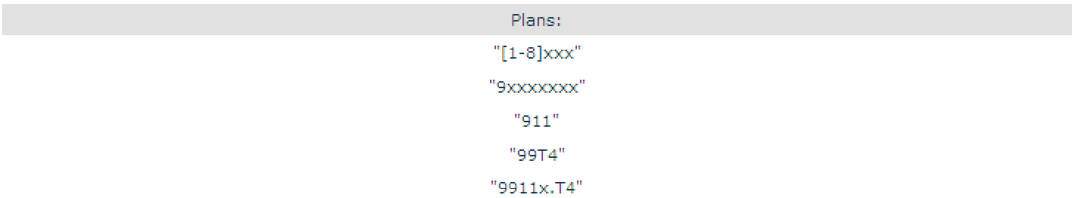
Below is user-defined digital map rule:

[ ] Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits.

* Match any single digit that is dialed.

. Match any arbitrary number of digits including none.

Tn Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans.



Cause extensions 1000-8999 to be dialed immediately.

Cause 8 digit numbers started with 9 to be dialed immediately.

Cause 911 to be dialed immediately after it is entered.

Cause 99 to be dialed after 4 seconds.

Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.

Note	End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously. System will stop dialing and send number according to User set rules.

Note

The add button for adding a new blacklist, the delete button for deleting one item, the delete all button for deleting all items.

If user does not want to answer some phone calls, add these phone numbers to the Black List, and these calls will be rejected x and are wildcard x means matching any single digit. For example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to be responded.

DOT (.) means matching any arbitrary number digit. For example, 6. Expresses any number with prefix 6 will be forbidden to be responded.

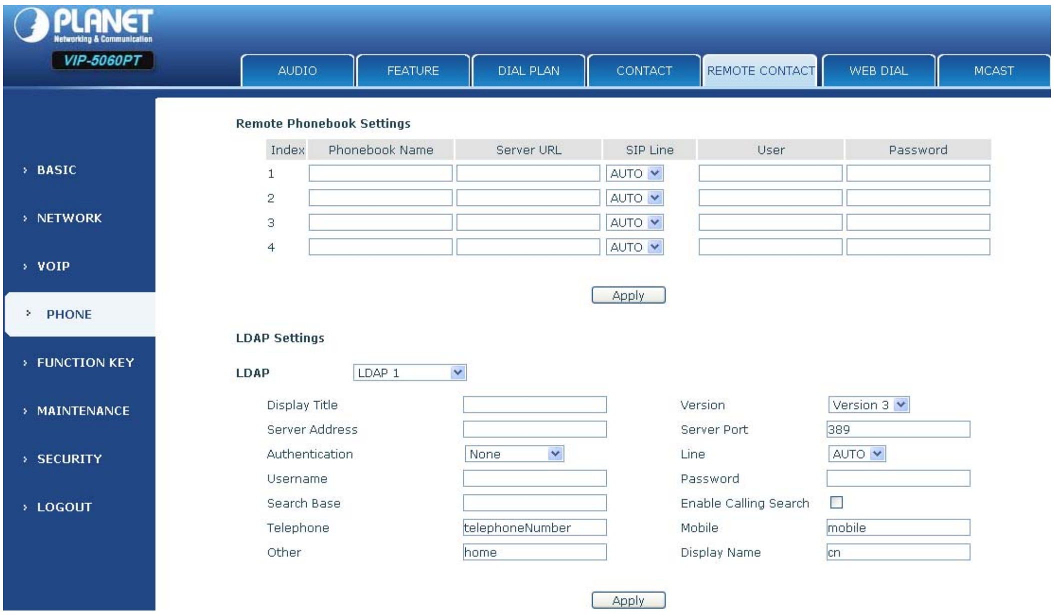
If user wants to allow a number or a series of number incoming, he may add the number(s) to the list as the white list rule. The configuration rule is -number, for example, -123456, or -1234xx.

Blacklist
-4119

Means any incoming number is forbidden except for 4119

Note: End with DOT (.) when set up the white list.

8.3.4.5 REMOTE CONTACT



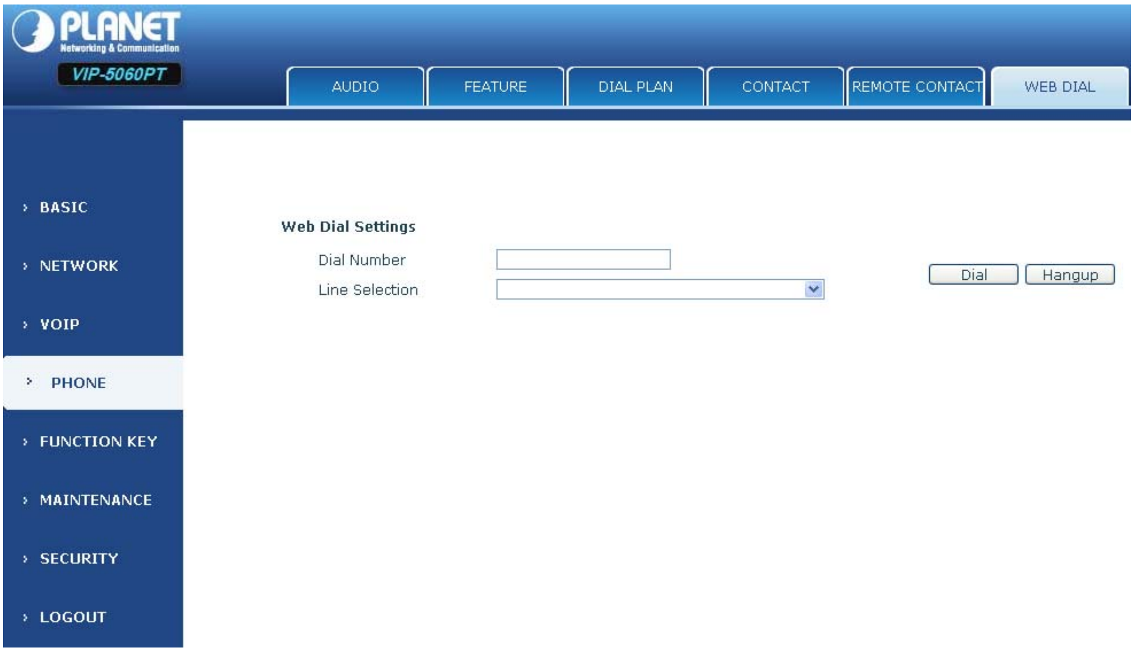
User needs to match a XML Phonebook address and User can directly access to the corresponding remote phonebook on the phone.

For example: Set the Phonebook Name as Planet, Server URL is tftp://192.168.1.3/admin/phonebook/index.xml.

Or Set the Phonebook Name as ldap, Server URL is ldap://192.168.1.3/dc=winline,dc=com.

Remote Phonebook Setting	
Phonebook Name	Custom the phonebook name displayed on the phone.
Server URL	Specify the server url of the remote phonebook.
SIP Line	Specify the sip line for the remote phonebook.
Authentication	Specify the authentication mode for remote phonebook.
User/password	Input the authentication username and password.

8.3.4.6 WEB DIAL



User can make a call through the WEB DIAL, enter the Dial Number then press Dial, if User wants to finish the talk, press Hang-up.

8.3.4.7 MCAST Setting

Use the multicast function to send notice to every member of the multicast is simple and easy. By setting the multicast key on your phone, you can send multicast RTP flow to the pre-configured multicast address. By listening multicast address is configured on the phone, listen and play the multicast address to send the RTP stream.

Send multicast setting

On the phone web page, function key-function key, set a function key，as shown

Value format IP: Port，the IP address of multicast is range from 224.0.0.0 to 239.255.255.255,port is greater than 1024

If multicast codec is G722, the LCD screen will displays "HD", which means the phone is sending high-definition voice stream

Operate steps:

1. When the phone is idle, press multicast key

Multicast RTP stream is send to pre-configured multicast address (IP: Port). The phone which listens to multicast address in the local network can receive the RTP stream. Multicast function key LED lights yellow.

LCD screen displays the following:

2. Press the hold softkey to hold the current multicast session
3. Press the end softkey again or multicast function key, multicast session can be stopped

Notice: RTP stream is one side that is from a sender to a receiver. When the phone initiates a multicast RTP session in a call, the current call is on hold.

Receive multicast setting

You can set up the phone monitoring 10 different multicast addresses to receive these multicast RTP stream.

You have two methods to receive RTP stream of multicast that can be set up through the web page:

Enable priorities of normal calls and Enable page Priority:

Enable priorities of normal call by select it, if the incoming RTP stream priority of multicast lower than the priority of current for normal calls, the phone will ignore the RTP stream of multicast. If the incoming RTP stream priority of multicast higher than the priority of current for normal calls, the phone will receive the RTP stream of multicast, and hold the current call.

Disabled priorities of normal call by select disable, the phone will ignore all local networks RTP stream of multicast.

Options as follows:

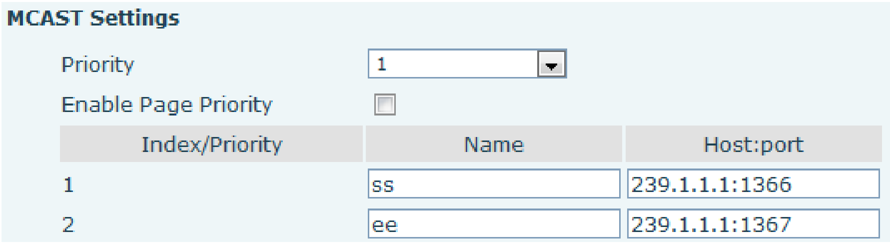
1-10: the priority defined for normal calls, 1 the highest level, 10 the lowest level

Disabled: Ignore all RTP stream of multicast

Enable Page Priority

Page priority determines the phone how to handle the newly received multicast RTP stream when in a multicast session. Enabled page priority, the phone will automatically ignore the low priority multicast RTP stream and receive the high priority multicast RTP stream and hold the current multicast session; If not enabled, the phone will automatically ignore all incoming multicast RTP stream.

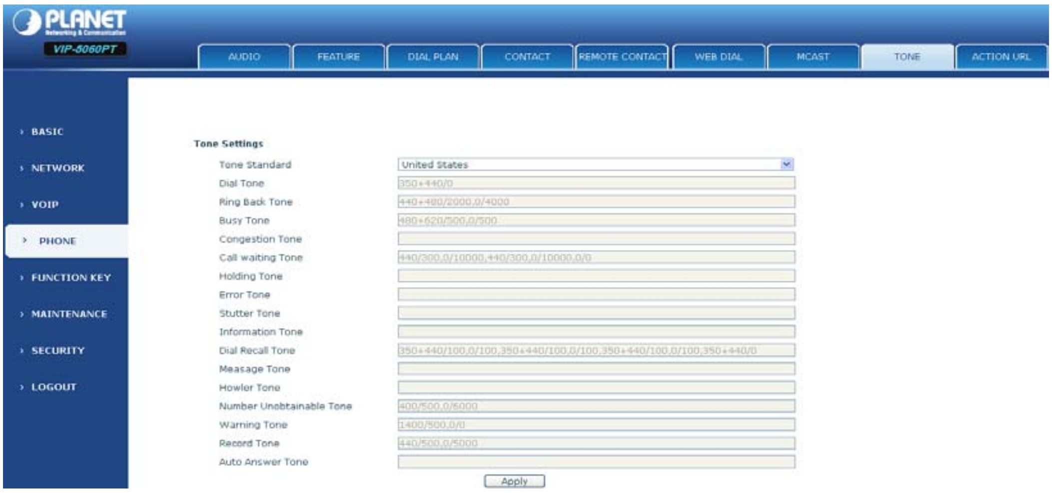
Web page is set as follows:



Now multicast “ss” has higher priority than multicast “ee”, the highest priority is for normal calls

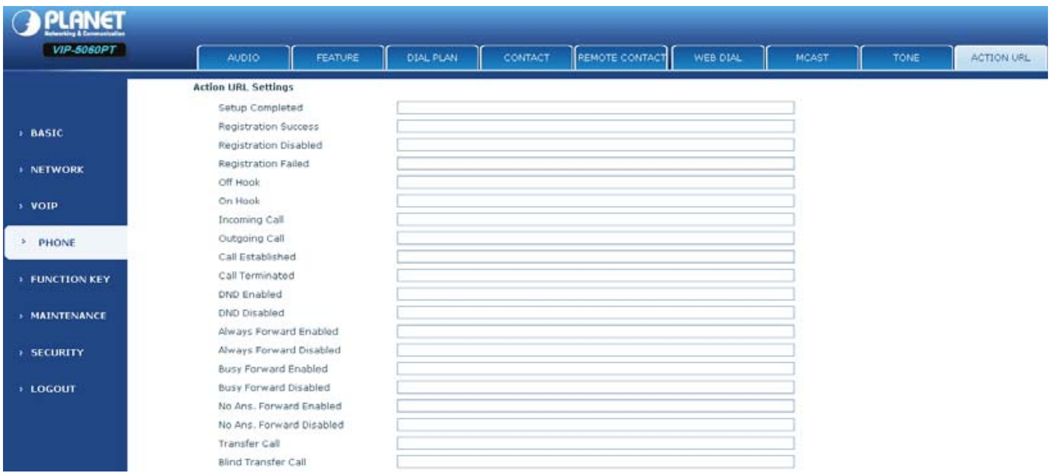
Notice: When a multicast session begins, multicast sender and receiver will beep

8.3.4.8 Tone



User can select the desired tone standard, also can customize the settings

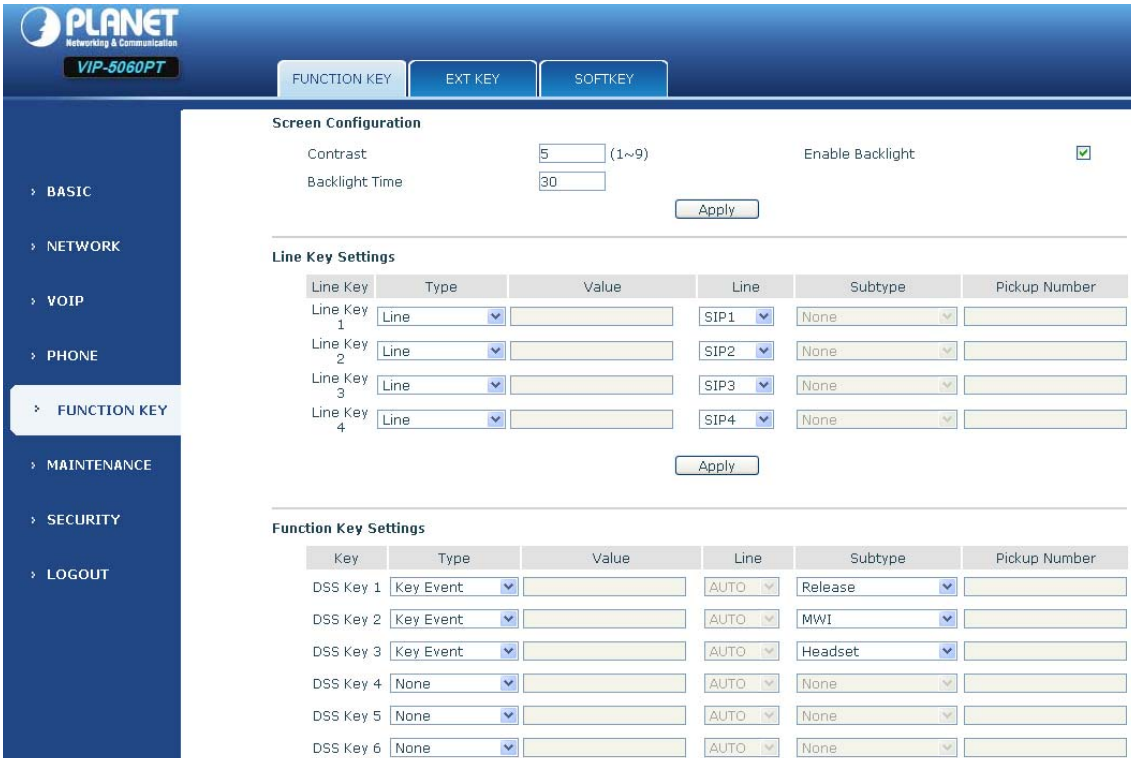
8.3.4.9 Action URL



Specify the Action URL that Record the operation of phone, send these corresponding information to server, url:http://InternalServer /FileName.xml?(Internal Server is server ip, FileName is name of xml that contains the action message)

8.3.5 FUNCTION KEY

8.3.5.1 FUNCTION KEY



Function Key	
Field name	**Explanation**
Contrast	Set contrast of screen.
Enable Backlight	Set enable/disable backlight.
Line Key Settings	
Line: select Auto, SIP1 - SIP6 in function key type. After User set it, User pick up handset or hands-free, press this function key, and then User can use the corresponding SIP line.	
Function Key Settings	

key	Show the function key's serial number.
Type	Memory Key: settings can be stored in key storage for each number, the standby or off-hook, select the function keys on the keyboard can call this number. Line, set the dial mode (Auto, SIP1 to SIP6).Key Event functions, monitor state. DTMF: In the call, send DTMF. URL: User can input remote book url.
Value	Set the type parameter values.
Line	Choose which lines to use this feature.
Subtype	Select the function parameters Key Event and Memory Event.
Pickup Number	Please input the pickup number When SubType is BLF or presence.

NOTICE：

- Memory keys can be configured through the following:

Speed Dial function, through the configuration of the key corresponding to the number of ways as shown below.

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Memory Key	4111	SIP1	Speed Dial	

User can press the F1 key to allocate this number by line1 line.

Intercom function, User can press this key in standby to automatically answer the call and make each other.

Function Key Settings

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Memory Key	4111	SIP1	Intercom	

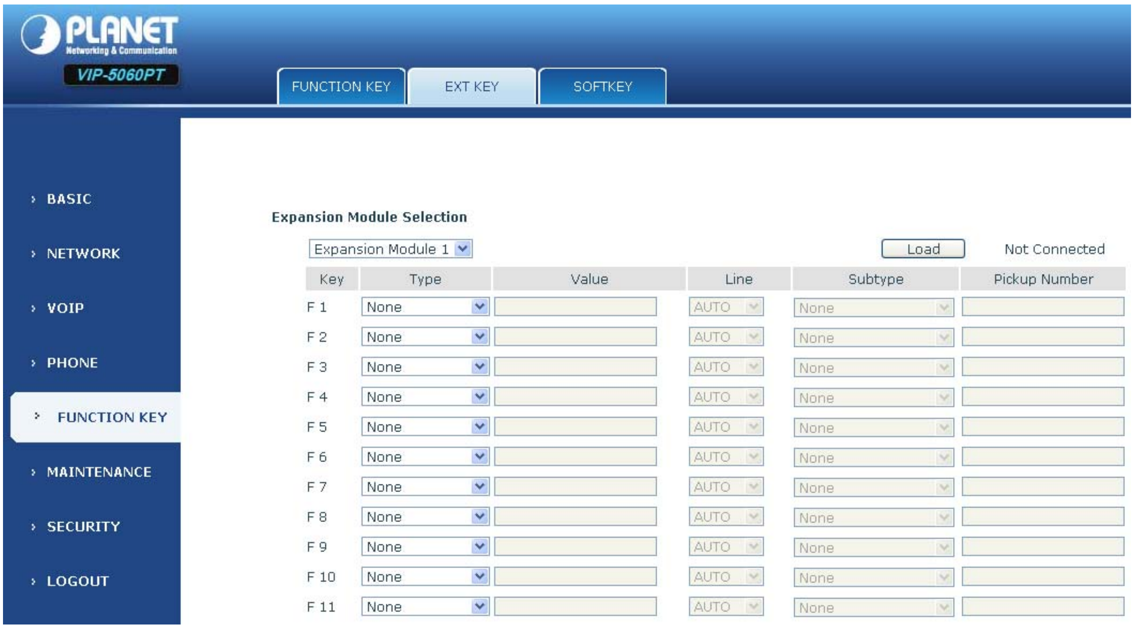
User can be configured in accordance with push to talk function the way: 4116 was the other number; Then press the standby button and make it automatically answer the call 4116.

- key can be configured through the following events:

For example:

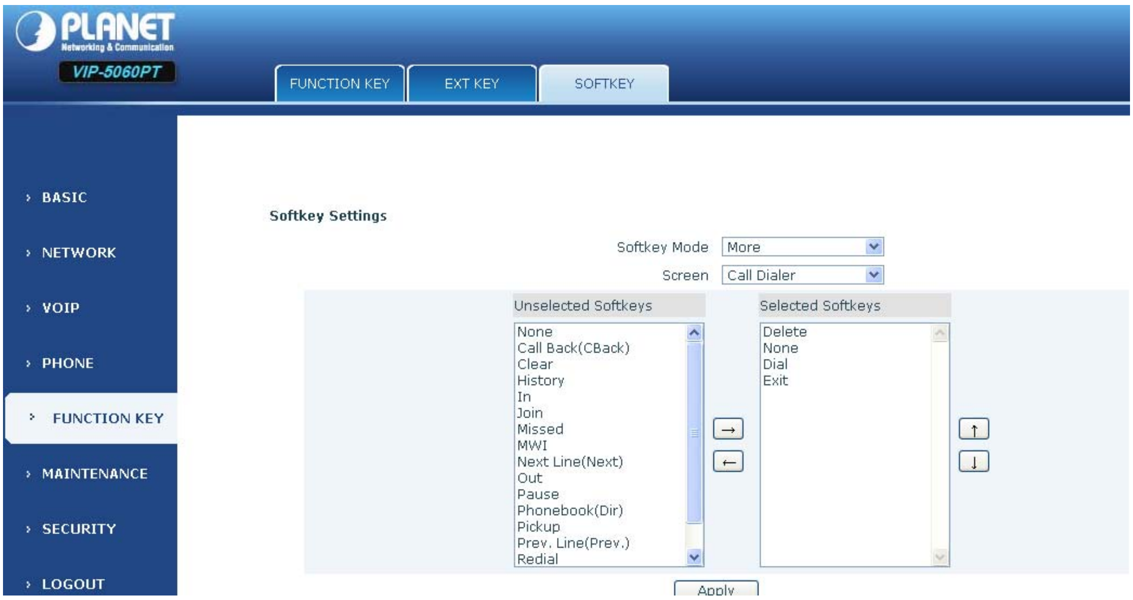
Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Key Event		SIP1	DND	

8.3.5.2 EXIT KEY



EXT KEY has the same usage with the Function key. “In” port connects the phone, “Out” port connects the next one, if there is only, User don’t need for power supply, if there are more than one, User need supply 5V power for the first one, and use RJ-45 direct connector.

8.3.5.3 SOFTKEY

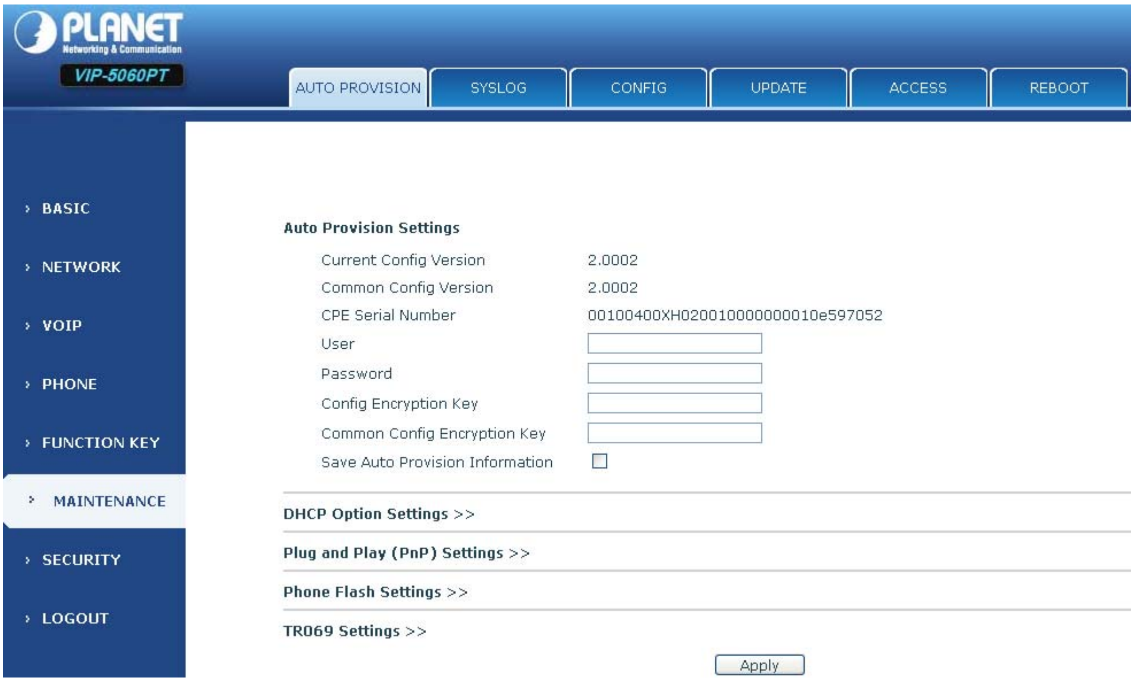


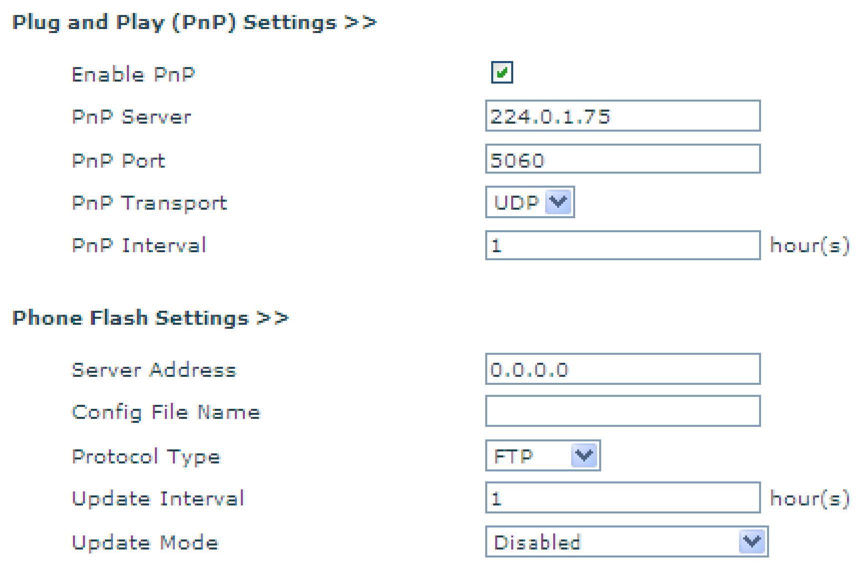
SOFTKEY

User can configure different functions in different screens for every softkey.

8.3.6 Maintenance

8.3.6.1 Auto Provision





Planet endpoint supports PnP and DHCP and Phone Flash to obtain the parameters. The PnP and DHCP and Phone Flash are all deployed, endpoint will go by the following process to try to obtain the server address and other parameters, when it boots up:

DHCP option → PnP server → Phone Flash

Auto Provision

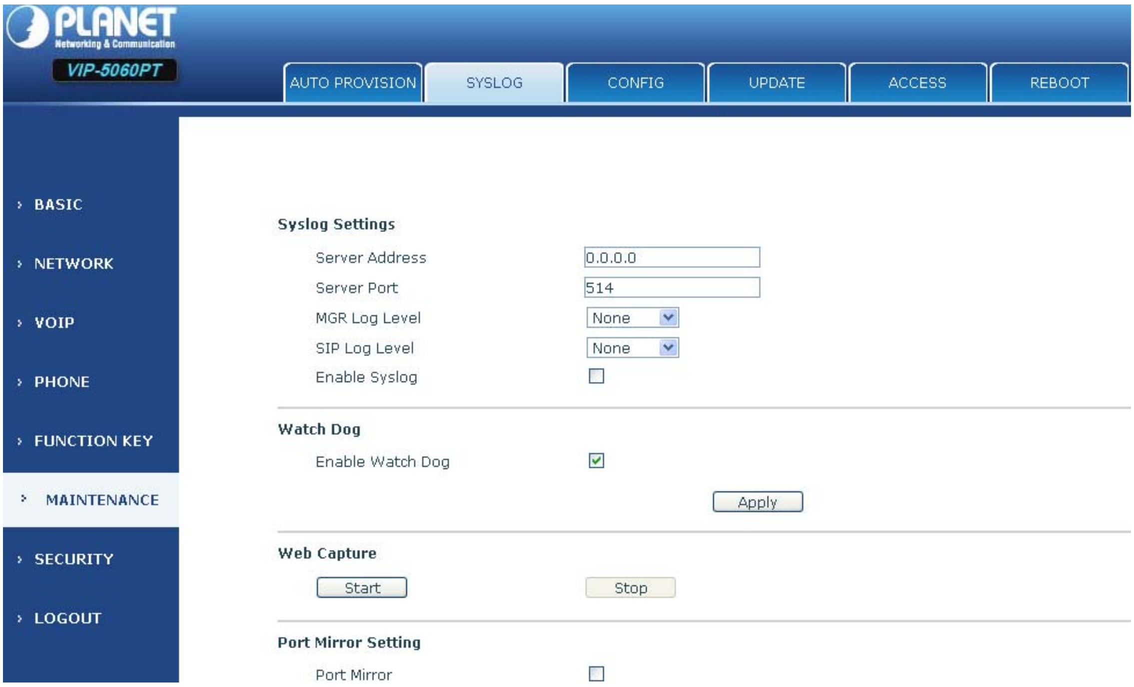
Field name	Explanation
Auto Provision Setting	
Current Config Version	Show the current config file's version. If the version of the configuration downloaded is higher than the version of the running configurations, the auto provision would upgrade, or stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
Common Config Version	Show the common config file's version. If the configuration downloaded and the running configurations are the same, the auto provision would stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
CPE Serial Number	Show CPE Serial Number.
User	Specify FTP/HTTP/HTTPS server Username. System will use anonymous if username keep blank.
Password	Specify FTP/HTTP/HTTPS server Password.
Config Encrypt Key	Input the Encrypt Key, if the configuration file is encrypted.
Common Config Encrypt Key	Input the Common Encrypt Key, if the Common Configuration file is encrypted.
Save Autoprovision Information	Save the username and password authentication message of http/https/ftp and input ID message in the phone until the url in the server changes.
DHCP Option Setting	
DHCP Option Setting	Specify DHCP Option. DHCP option supports DHCP custom option and DHCP option 66 and DHCP option 43 to obtain the parameters. User could choose one method among them; the default is DHCP option disable.
Custom DHCP Option	A valid Custom DHCP Option is from 128 to 254. The Custom DHCP Option must be in accordance with the one defined in the DHCP server.
Plug and Play	
Enable PnP	Enable PnP by selecting it, than the phone will send SIP SUBSCRIBE messages to a multicast address when it boots up. Any SIP server understanding that message will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
PnP Server	Specify the PnP Server.
PnP Port	Specify the PnP Server.
PnP Transport	Specify the PnP Transfer protocol.

PnP Interval	Specify the Interval time, unit is hour.
Phone Flash	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be IP address or Domain name with subdirectory.
Config File Name	Set configuration file's name which need to update. System will use MAC as config file name if config file name keep blank. For example, 000102030405.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify update interval time, unit is hour.
Update Mode	Different update modes: 1. Disable: means no update. 2. Update after reboot: means update after reboot. 3. Update at time interval: means periodic update.
TR069 Settings	
Enable TR069	Enable TR069 by selecting it.
ACS Server Type	Specify the ACS Server Type.
ACS Server URL	Specify the ACS Server URL.
ACS User	Specify ACS User.
ACS Password	Specify ACS Password.
TR069 Auto Login	Enable TR069 Auto Login by selecting it.
"Inform" Sending Period	Specify the "inform" Sending Period, unit is second.

8.3.6.2 SYSLOG

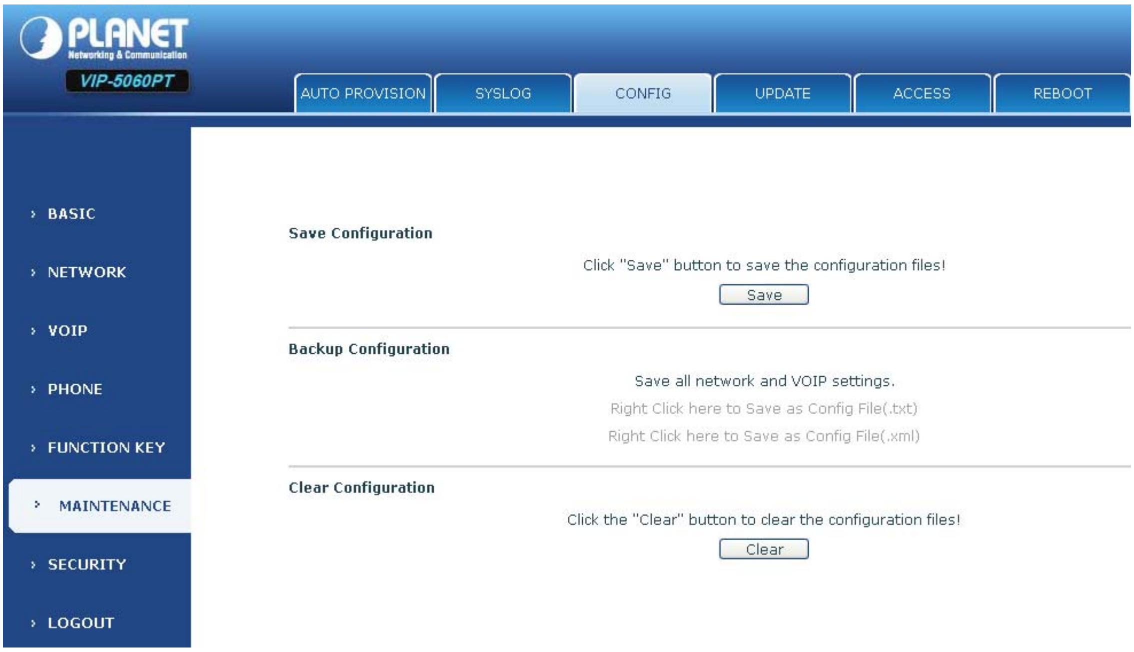
Syslog is a protocol which is used to record the log messages with client/server mechanism. Syslog server receives the messages from clients, and classifies them based on priority and type. Then these messages will be written into log by some rules which administrator can configure. This is a better way for log management.

8 levels in debug information:
Level 0---emergency: This is highest default debug info level. User system cannot work.
Level 1---alert: User system has deadly problem.
Level 2---critical: User system has serious problem.
Level 3---error: The error will affect User system working.
Level 4---warning: There are some potential dangers. But User system can work.
Level 5---notice: User system works well in special condition, but User need to check its working environment and parameter.
Level 6---info: the daily debugging info.
Level 7---debug: the lowest debug info Professional debugging info from R&D person.
At present, the lowest level of debug information is info; debug level only can be displayed on telnet.



Syslog Configuration	
Field name	**Explanation**
Syslog Setting	
Server Address	Set Syslog server IP address.
Server Port	Set Syslog server port.
MGR Log Level	Set the level of MGR log.
SIP Log Level	Set the level of SIP log.
Enable Syslog	Select it or not to enable or disable syslog.
Web Capture	
Start	Click the start button when User need capture the WAN packet stream of the phone, then open or save the file as the interface.
Stop	Click the end button to stop capturing the packet stream.

8.3.6.3 CONFIG

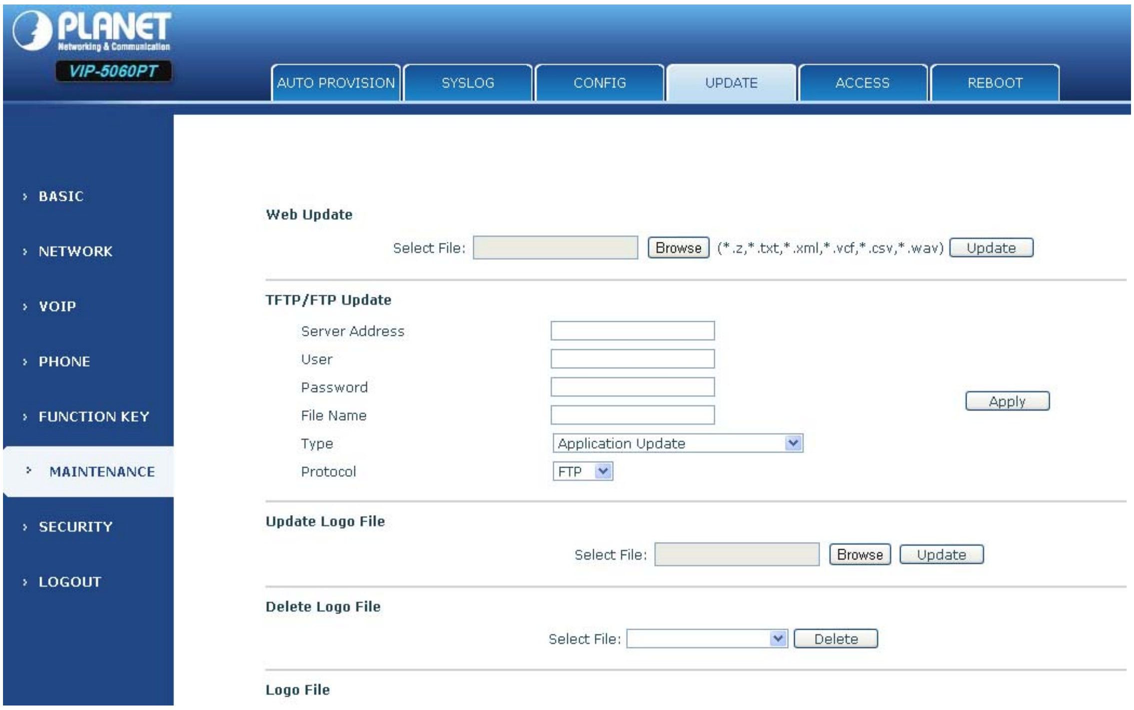


Config Setting

Field name	**Explanation**
Save Configuration	User can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately.
Backup Configuration	Right clicks on “Right click here…” and select “Save Target As config File(.txt)” then User will save the config file in .txt format, or select “Save Target As config File(.xml)” then User will save the config file in .xml format.
Clear Configuration	User can restore factory default configuration and reboot the phone. If User login as Admin, the phone will reset all configurations and restore factory default; if User login as Guest, the phone will reset all configurations except for VoIP accounts (SIP1-6) and version number.

8.3.6.4 UPDATE

User can update User configuration with User config file in this web page.

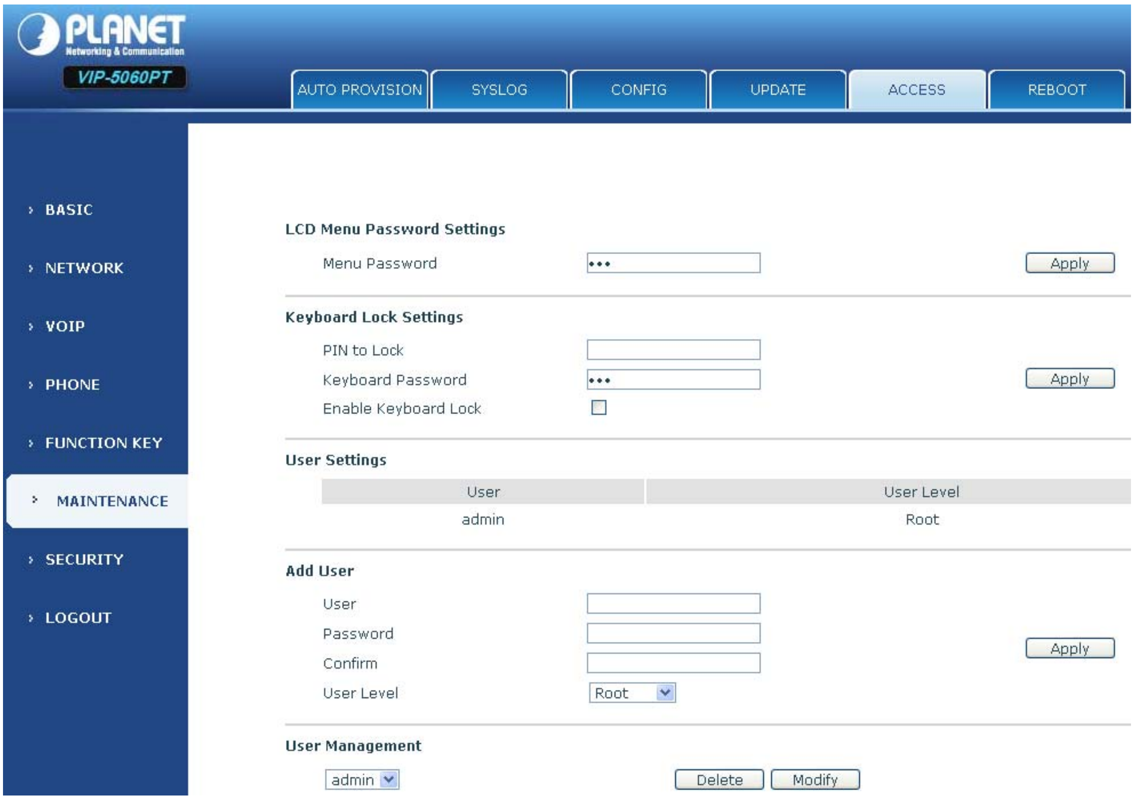


Update	
Field name	**Explanation**
Web Update	
Web Update	Click the browse button, find out the config file saved before or provided by manufacturer, download it to the phone directly, press “Update” to save. User can also update downloaded update file, logo picture, ring, mmiset file by web.
TFTP/FTP Update	
Server Address	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
User	Set the FTP server Username for download/upload.
Password	Set the FTP server password for download/upload.
File name	Set the name of update file or config file. The default name is the MAC of the phone, such as 000102030405.
Note	User can modify the exported config file. And User can also download config file which includes several modules that need to be imported. For example, User can download a config file just to keep with SIP module. After reboot, other modules of system still use the previous setting and are not lost
Type	Action type that system wants to execute:

	1. Application update: download system to update file. 2. Config file export: Upload the config file to FTP/TFTP server, name and save it. 3. Config file import: Download the config file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset. 4. Phone book export (.vcf): Upload the phonebook file to FTP/TFTP server, name and save it. 5. PhoneBook import (.vcf): Download the phonebook file to phone from FTP/TFTP server.
Protocol	Select FTP/TFTP server.
Update Logo File	
Select File	Specify the URL of the logo file.
Delete Logo File	
Select File	Select the logo that User wants to delete.
Logo File	
Logo File	Show the logo file.

8.3.6.5 ACCESS

User can add or delete user account, and change the authority of each user account in this web page.



Access Configuration	
Field name	**Explanation**
Keyboard Password	Set the password for entering the setting menu of the phone by the phone‘s key board. The password is digit.

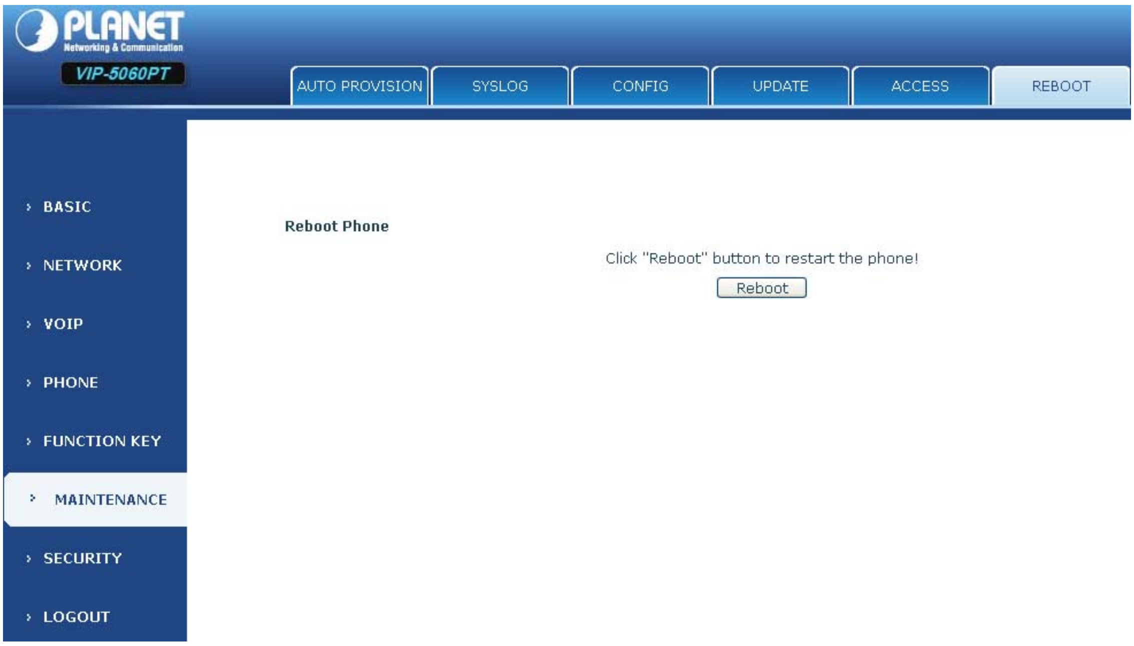
User Settings

User	User Level
admin	Root
root	General

This table shows the current user existed.	
User	Set account user name.
User Level	Set user level, Root user has the right to modify configuration, General can only read.
Password	Set the password.
Confirm	Confirm the password.

Select the account and click the **Modify** to modify the selected account, and click the **Delete** to delete the selected account.
General user only can add the user whose level is General.

8.3.6.6 REBOOT

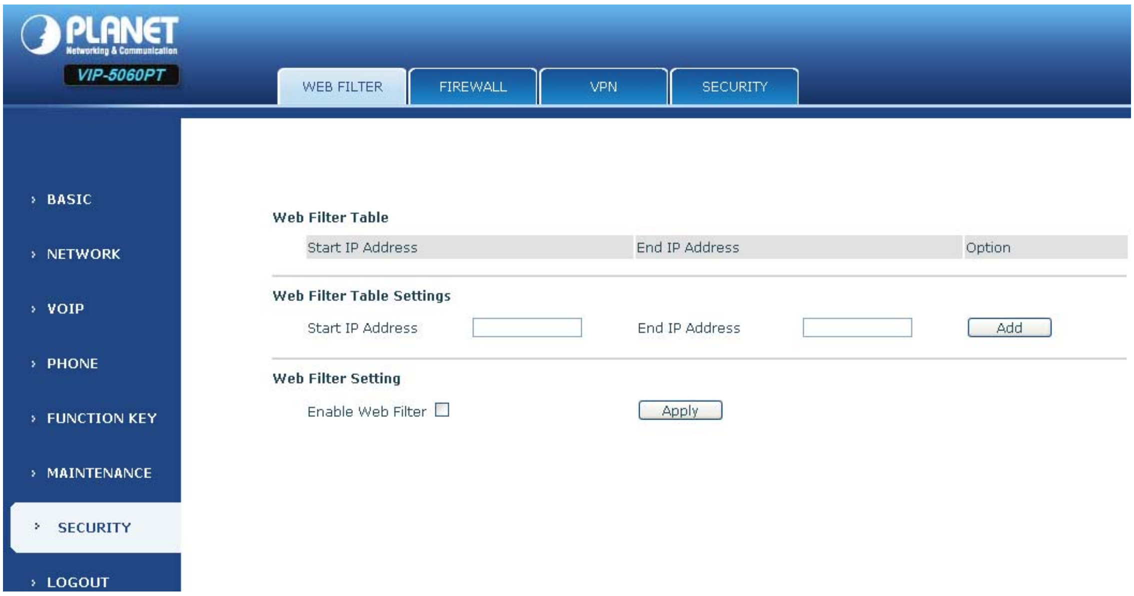


If User modified some configurations which need the phone’s reboot to be effective, User need click the Reboot, then the phone will reboot immediately.

Note	Before reboot, User needs to confirm that User has saved all configurations.

8.3.7 SECURITY

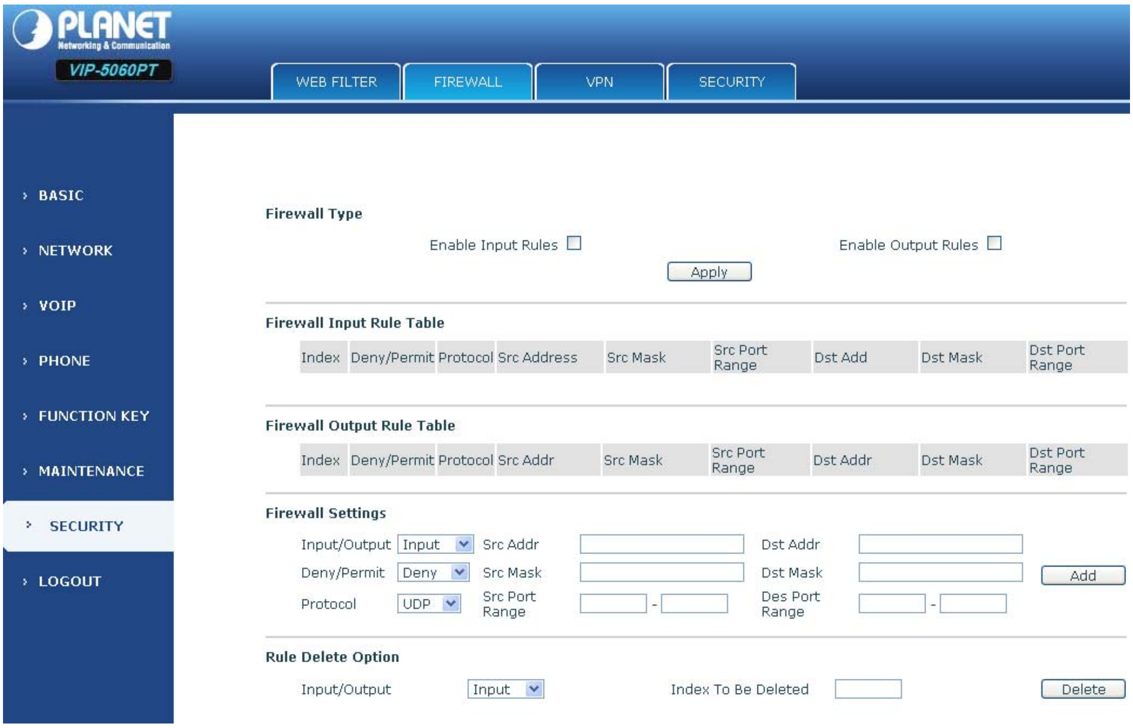
8.3.7.1 WEB FILTER



WEB Filter	
User could make some device own IP, which is pre-specified, access to the MMI of the phone to config and manage the phone.	
Field name	**Explanation**
Web Filter Table Settings:	
Add or delete the IP address segments that access to the phone. Set initial IP address in the Start IP column, Set end IP address in the End IP column, and click Add to add this IP segment. User can also click Delete to delete the selected IP segment.	
Web Filter setting	Select it or not to enable or disable Web Filter. Click **Apply** to make it effective.

Note	Do not set User visiting IP outside the Web filter range; otherwise, User cannot logon to the web.

8.3.7.2 FIREWALL



Firewall Configuration

In this web interface, User can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule).

Firewall supports two types of rules: input access rule and output access rule. Each type supports at most 10 items.

Through this web page, User could set up and enable/disable firewall with input/output rules. System could prevent unauthorized access, or access other networks set in rules for security. Firewall, is also called access list, is a simple implementation of a Cisco-like access list (firewall). It supports two access lists: one for filtering input packets, and the other for filtering output packets. Each kind of list could be added 10 items.

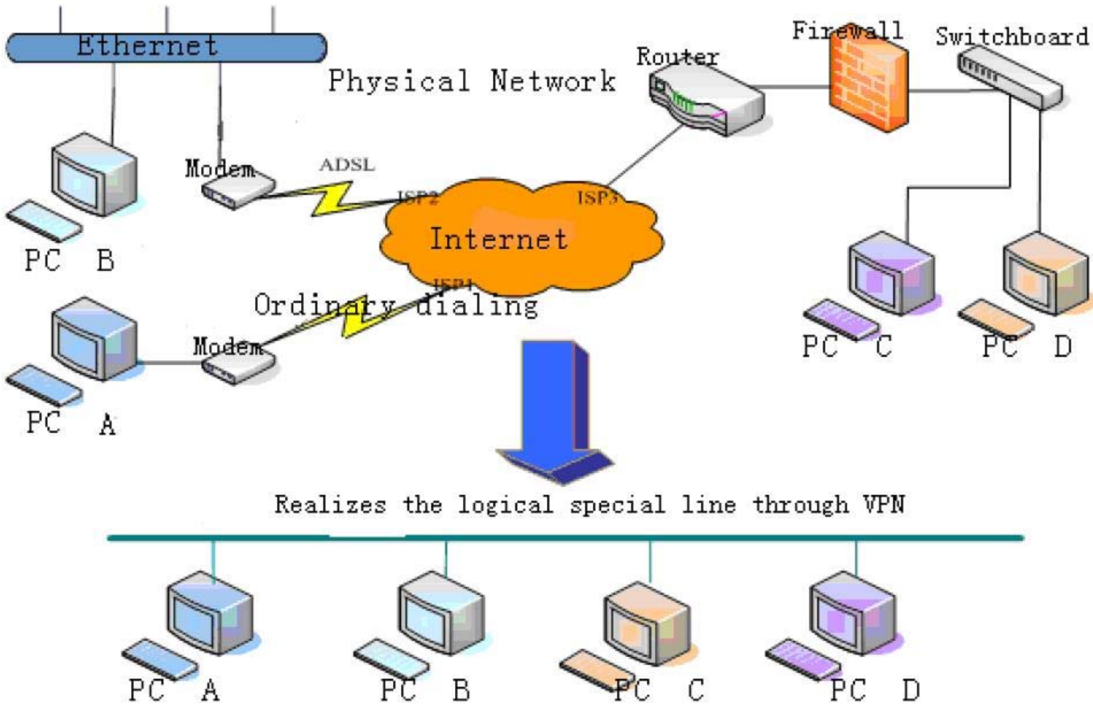
We will give User an instance for User reference.

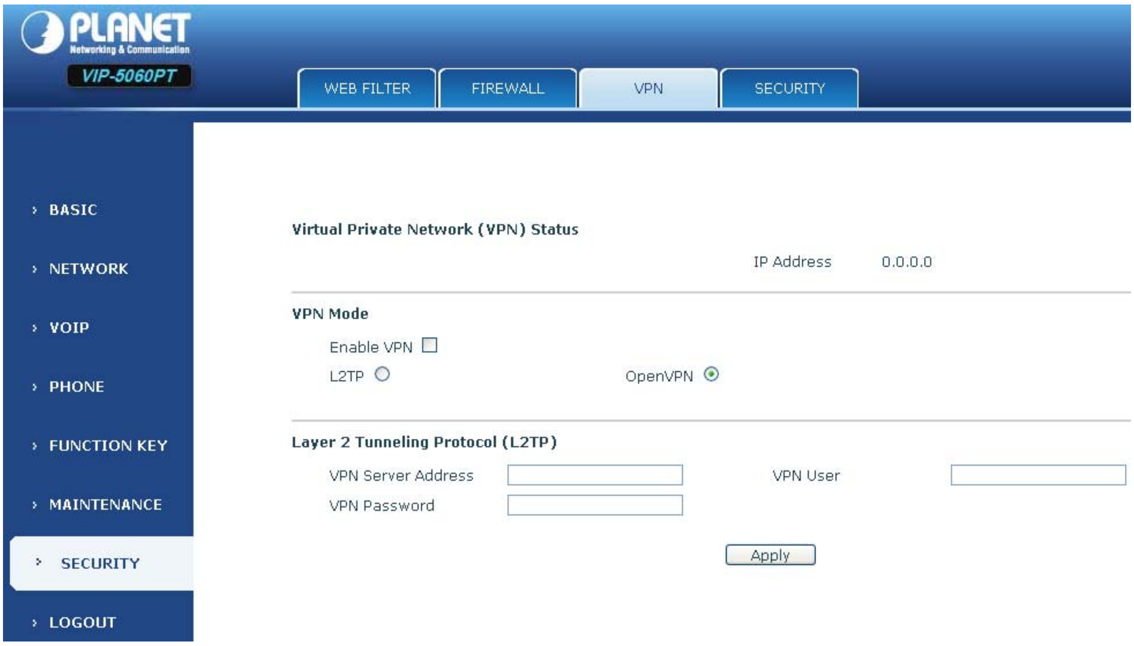
Field name	Explanation
Enable Input Rules	Select it to Enable Input Rules.
Enable Output Rules	Select it to Enable Output Rules.
Input / Output	Specify current adding rule by selecting input rule or output rule.
Deny / Permit	Specify current adding rule by selecting Deny rule or Permit rule.
Protocol	Filter protocol type. User can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range.
Src Address	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0.

Des Address	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*.
Src Mask	Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Dest Mask	Set the destination address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Click the Add button if User wants to add a new output rule. Then enable out access, and click the Apply button. So when devices execute to ping 192.168.1.118, system will deny the request to send icmp request to 192.168.1.118 for the out access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.	
Click the Delete button to delete the selected rule.	

8.3.7.3 VPN

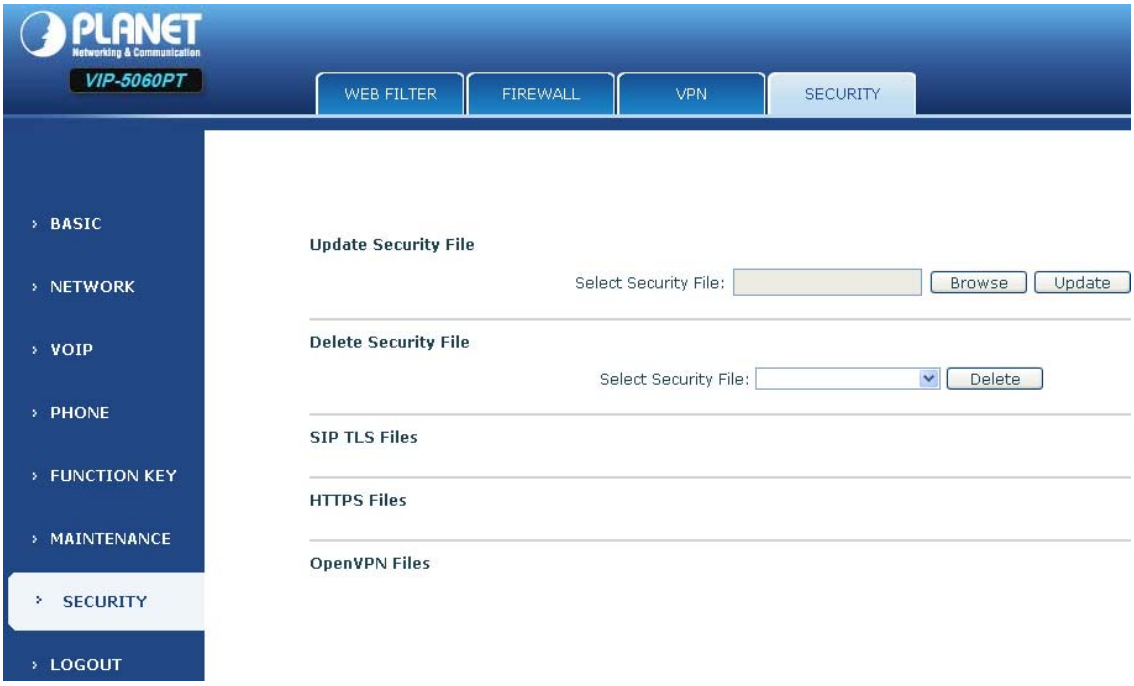
This web page provides us a safe connect mode by which we can make remote access to enterprise inner network from public network. That is to say, User can set it to connect public networks in different areas into inner network via a special tunnel.





VPN Configuration	
Field name	Explanation
VPN IP	Shows the current VPN IP address.
Select L2TP. User can choose only one for current state. After User select it, User's better save configuration and reboot User phone.	
Enable VPN	Select it or not to enable or disable VPN.
VPN Server Address	Set VPN L2TP Server IP address.
VPN User	Set User Name access to VPN L2TP Server.
VPN Password	Set Password access to VPN L2TP Server.

8.3.7.4 SECURITY



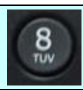
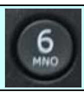
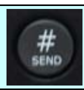
Security	
Field name	**Explanation**
Update Security File	
Select Security File	Select the security file User want to update, then click Update button to update.
Delete Security File	
Select Security File	Select the security file User want to delete, then click Delete button to update.
SIP TLS File	Show SIP TLS authentication certification file.
HTTPS File	Show HTTPS authentication certification file.
Open VPN Files	Show Open VPN File authentication certification file.

8.3.8 LOGOUT

Click **Logout**，and User will exit web page. If User want to enter it next time, User need input user name and password again.

9 Appendix

9.1 Digit-character map table

Keypad	Character	Keypad	Character
	1 @		7 P Q R S p q r s
	2 A B C a b c		8 T U V t u v
	3 D E F d e f		9 W X Y Z w x y z
	4 G H I g h i		*/.
	5 J K L j k l		0
	6 M N O m n o		#/SEND

9.2 Frequently Asked Questions List

Q1: No operation after power on?
A1: Check if the power adapter is properly connected. If applicable, check if the PoE (Power over Ethernet) switch behind the IP phone is set correctly.
Q2: No dial tone?
A2: Check if the handset cord is properly connected.
Q3: Cannot make a call?
A3: Check the status of your SIP registration status or contact your administrator, supplier, or ITSP for more information or assistance.
Q4: Cannot receive any phone call?
A4 : Check the status of your SIP registration status, or contact your administrator, supplier, or ITSP for more information or assistance
Q5: No voice during an active call?
A5: Check if the servers support the current audio codec type, or contact your administrator, supplier, or

ITSP for more information or assistance.

Q6: **Cannot connect to the configuration website?**

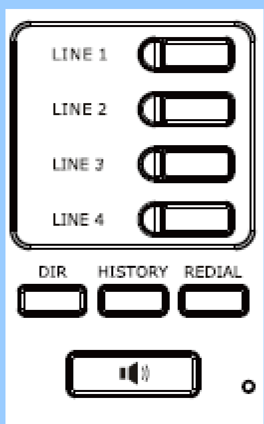
A6: Check if the Ethernet cable is properly connected.
Check if the URL is right; the format of URL is: http:// the Internet port IP address.
Check if your firewall/NAT settings are correct.
Check if the version of IE is IE8, or use other browser such as Firefox or Mozilla, or contact your administrator, supplier, or ITSP for more information or assistance.

Q7: **Forget the password?**

A7: Default password of website and menu is null.
If user changes the password and then forget it, or you cannot access to the configuration website or the menu items need password.
Solution:
Factory default: press Menu button and choose 16Factory Default and then a notice will appear, choose OK by using the corresponding softkey button.
If you choose factory default, you will return the phone to the original factory settings and will erase ALL current settings, including the directory and call logs.

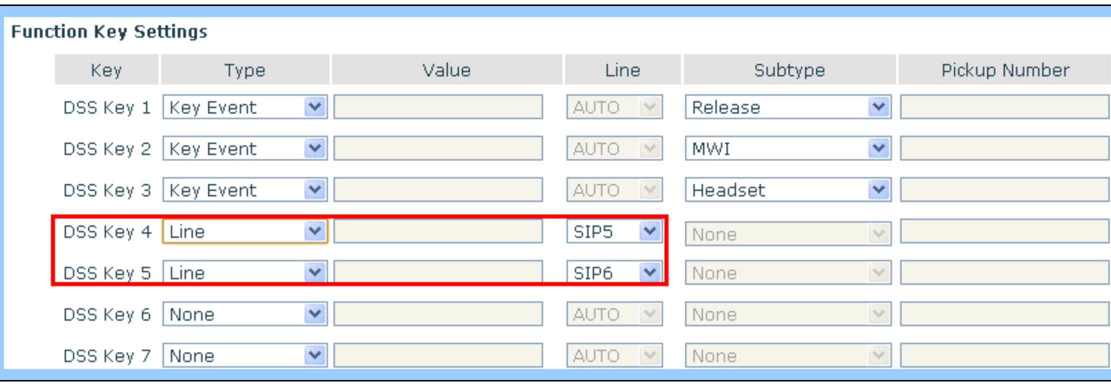
Q7: How to switch to different line to dial out?

A7: Before dialing out, press the correspondence line number you want to use. For example, if User wants to use Line 2 to dial out, please press Line 2.



Note

VIP-5060PT physical line is only 4 lines, the 5th and 6th line must use the **Function Key Settings**, to set it up.



Q8: How to set up the BLF function in the VIP-5060PT?

A8: Before we start, please be reminded your IPPBX must also support BLF function.

In Function key / EXT Key.

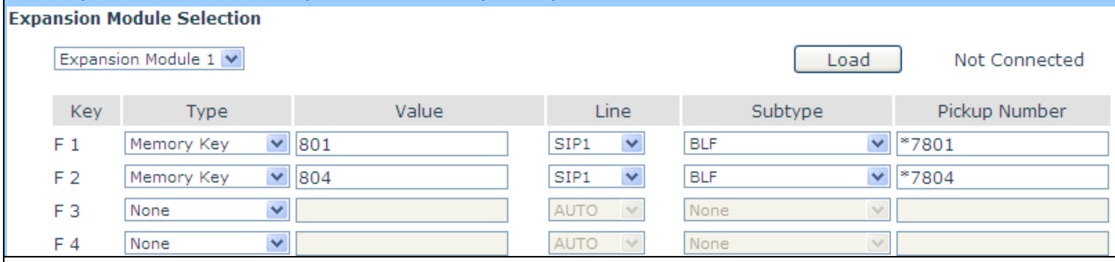
Type: please chose Memory Key

Value: your BLF extension

Line: choose which line you want to use BLF function

Subtype: BLF

Pick up Number: choose your IPPBX to pick up code + Extension number

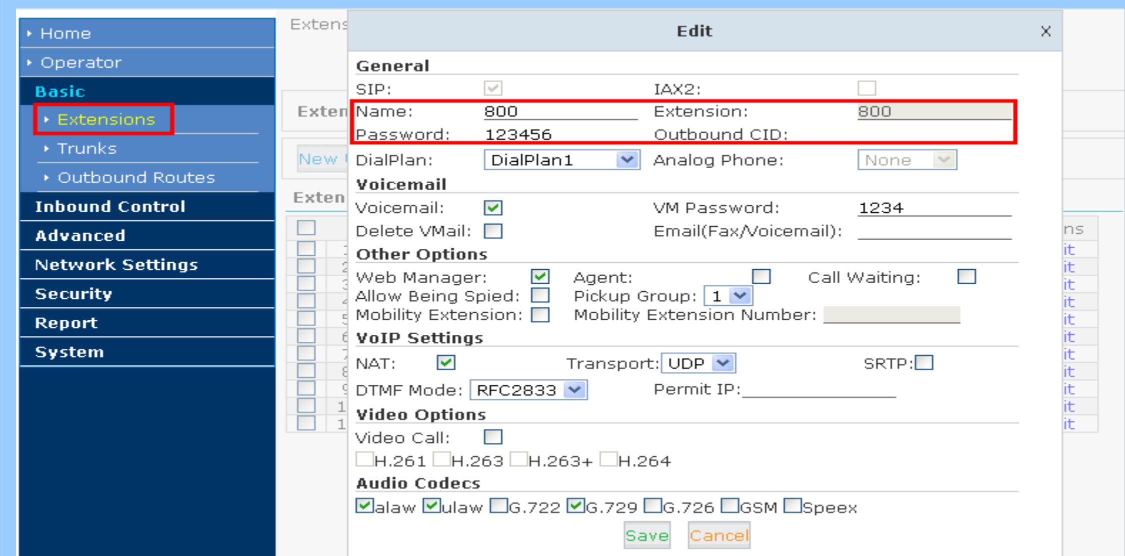


Q9: How to register VIP-5060PT to IPX-2100?

A9:

[In IPX-2100]

For extensions, please create a new account and remember their user name and password.



[In VIP-5060PT]

On VoIP / SIP page, please follow the messages below:
SIP line: choose the line you want to register
Server address: the IPX-2100 IP address
Server port: Server register port default is 5060
Authentication user: 800 (the extension you create in IPX-2100)
SIP user: (the extension you create in IPX-2100)
Display name: the name you want to display on phone screen when pressing the line button.
After saving the modification, the “successfully registered” status will be displayed.