



Enterprise PoE IP Phone

VIP-560PT / VIP-560PE

User's manual

Version 1.1

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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 remove the DC-plug or switch off the device, the device wills still consuming power from the power circuit. In the view of Saving the Energy and reduce the unnecessary power consuming, it is strongly suggested to switch off or remove the DC-plug for 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 SIP PoE IP Phone: Model: VIP-560PT / VIP-560PE Rev: 1.1 (2010, August) Part No. EM-VIP560 Series_v1.1

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Chapter 1 1 Introduction

Overview

Combining the cutting edge of Voice over IP and Internet telephony manufacturing experience, PLANET now introduces the latest member of **mainstream enterprise IP phone series**: the VIP-560PT / VIP-560PE, **the 4-line professional PoE IP Phone**.

To bring the most satisfaction to customers, the VIP-560PT / VIP-560PE is the ideal choice for a business to deploy by using IP PBX service. The standard features of the VIP-560PT / VIP-560PE include 4-line, dual 10/100 switched Ethernet ports and integrated IEEE power over Ethernet (802.3af) circuitry for offering a choice of powering and cabling options to help reduce cabling expenses and cord clutter.

To give most flexibility to users, the VIP-560PT / VIP-560PE platform contains a 240 x 160 pixels graphic LCD with Back light, 4 soft-keys, 10 fixed function keys and a 5-position navigation key. The PLANET VIP-560PT / VIP-560PE desktop phone is engineered to make Easy-to-install communications, cost-effective to deploy, self-contained, service-integrated, intelligent phone features offering and powerful voice processing power as possible. The VIP-560PT / VIP-560PE can effortlessly deliver toll voice quality equivalent to the regular VoIP/IP PBX connections utilizing cutting-edge Quality of Service (QoS) capabilities to encompass IP-TOS/DiffServ, 802.1 p/q VLAN tagging, echo cancellation, comfort noise generation (CNG) and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the IP phone allow users to install in an existing network location without interfering with connections of desktop PC networks.

The VIP-560PT / VIP-560PE has streamlined wired IP telephone that provides additional features such as built-in PPPoE/DHCP clients, password-protected machine management, call hold, forwarding, mute, transfer, waiting, pickup, caller ID, peed-dial, 3-way conference, last number redial, incoming message indicator, multiple call appearances and user-intuitive web administration system.

The VIP-560PE could expend the programmable buttons via connected the Expansion Module – VIP-56EXT. the VIP-56EXT features 32 fully programmable buttons (each with dual-color LED) when used with VIP-560PE. The VIP-56EXT creates up to 192 additional programmable extensions when 6 EXT are daisy chained with the VIP-560PE. The VIP-560PE supports speed dialing, BLF (Busy Lamp Field), BLA (Bridged Line Appearance) on each of the programmable buttons on the VIP-56EXT module.

Besides, the VIP-560PT / VIP-560PE are the ideal solution for office use as well as installation for Internet Telephony Service Provider (ITSP) from leading vendors. It's the delivery platform for IP voice services that makes benefits from the VoIP technology in business class communications services.

Product Features

- IEEE 802.3af (Power over Ethernet) compliant
- Full-Featured enterprise SIP Desktop Phone
- Full duplex speakerphone (both Mic and Speaker)
- Large 4-level gray scales LCD (240 x 160) with backlight
- Efficient installation deployment of IP PBX solution
- Reversible base stand and wall mount installation options

VoIP Features

- SIP 2.0 (RFC3261) compliant
- Supports up to 4 service domains
- 8 programmable buttons
- Supports up to 6 Expansion Module (VIP-560PE)
- Interoperability with leading PLANET IP PBX platforms
- Supports BLF (Busy Lamp Field), BLA (Bridged Line Appearance)
- Voice codec support: G.711(A-Law, u-Law), G.723.1, G.722, G.729 A
- In-band, out-of-band DTMF Relay (RFC 2833) and SIP INFO
- 3-Way Conference / Caller ID / Speed Dial
- Call Hold / Mute / Forward / Transfer / Waiting
- Voice processing: VAD, CNG, AEC, Adaptive Jitter Buffer Management

Package Content

The contents of your product should contain the following items: Enterprise PoE IP Phone VIP-560PE unit

Quick Installation Guide

CD-ROM containing the on-line manual.

RJ-45 cable

Physical Details

The following figure illustrates the front/rear panel of IP Phone.

Front View and Keypad function



Keypad Description

1 Headset		Toggles the headset on or off.	
		Red light means the feature is enabled.	
2	Mute	Toggles the Mute feature on or off.	
2	Mute	Red light means the feature is enabled.	
3	Message	Typically auto-dials your voice message service.	
3	Messaye	Red light means have unread voice mail.	
4	Service Open or Close the Services menu. (Reserved for customization purpose)		
5	Directories	tories Use it to access call logs and corporate directories.	
6	MENU Allows you to scroll through menus.		
7	Volume	Controls the volume and other settings.	
8	Conference	Connect calling / called party to the conference.	

9	Redial	To Redial the last number.	
10	Transfer	Transfer redirects a connected.	
11	Hold	Put a call on hold.	
12	Number Keypad	Basic Call Handling: Press "#" sends out a call (default).	
		Toggles the speakerphone on or off.	
13	Speaker	1) Flashing (Red): There is an incoming call.	
		2) Steady (Red): Pick up and enter normal call.	
		Select the phone line (Call or Answer).	
		Different colors for different status:	
14	Line Key	1) Red, flashing: There is an incoming call.	
.4	Line Key	2) Red, steady: Pick up and enter normal call.	
		3) Yellow-green, flashing: Holding call.	
		4) Yellow-green, steady: Active call.	
15	Soft key	Each displays a soft key function. To activate a soft key, press the soft key button.	
		Programmable Buttons can be used to bind in order to achieve speed dial.	
16	Brogrammable Button	Programmable Buttons can be used to bind in order to achieve speed dial. Turn on BLF:	
16	Programmable Button		
16	Programmable Button	Turn on BLF:	
16 17	Programmable Button Back	Turn on BLF: 1) Red, steady: Remote line is busying.	
		Turn on BLF:1) Red, steady: Remote line is busying.2) Yellow-green, steady: Remote line is idle.	
		 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. 	
17	Back	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. 	
17	Back	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. Left: Open "Received Calls" list. 	
17 18	Back Navigation	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. Left: Open "Received Calls" list. Right: Open "Dialed Numbers" list. 	
17 18 19	Back Navigation OK	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. Left: Open "Received Calls" list. Right: Open "Dialed Numbers" list. To confirm the action. 	
17 18 19 20 21	Back Navigation OK Speaker LCD screen	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. Left: Open "Received Calls" list. Right: Open "Dialed Numbers" list. To confirm the action. Hands-free voice of the output. 	
17 18 19 20	Back Navigation OK Speaker	 Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle. Return to the standby interface. Down: Open "Missed Calls" list. Left: Open "Received Calls" list. Right: Open "Dialed Numbers" list. To confirm the action. Hands-free voice of the output. 480*160 pixel Color high-definition display. 	



Keypad Description

AC adapter.

		DIAE composition for intermed access composition directly	
1	LAN	RJ-45 connector, for Internet access, connected directly Switch/Hub through straight CAT-5 cable. The LAN interface also can be connected with 802.3af Persent Source of the sou	
2	PC	RJ-45 connector, to maintain the existing network structur connected directly to the PC through straight CAT-5 cable	re,
3	12V DC	12V DC Power input outlet	
4	Handset Jack	RJ-22 connector, for telephone handset.	
5	Headset Jack (RJ-22)	RJ-22 connector, for ear-phone.	
6	Headset Jack (3.5 mm)	3.5 mm connector, for ear-phone.	
7	EXT	To connect the expansion module (VIP-56EXT) at IN port.	
ote	VIP-560PT/VI effect of cu	IP-560PE at the same time, this may make the arrent pulse and then cause device damage. Be	PI
	2 3 4 5 6 7	2 PC 3 12V DC 4 Handset Jack 5 Headset Jack (RJ-22) 6 Headset Jack (3.5 mm) 7 EXT Please don't VIP-560PT/V effect of cu	1LANThe LAN interface also can be connected with 802.3af Piswitch or converter for power supply2PCRJ-45 connector, to maintain the existing network structure connected directly to the PC through straight CAT-5 cable312V DC12V DC Power input outlet4Handset JackRJ-22 connector, for telephone handset.5Headset Jack (RJ-22)RJ-22 connector, for ear-phone.6Headset Jack (RJ-22)3.5 mm connector, for ear-phone.7EXTTo connect the expansion module (VIP-56EXT) at IN port.bitePlease don't connect POE injector and AC adapter to VIP-560PT/VIP-560PE at the same time, this may make the effect of current pulse and then cause device damage. Be

one

10

Screen Features

This is what your main phone screen might look like with an active call.



Graphic Icon Description

Time and Data	Display current time and data.	
Call activity Area	Displays calls per line, including caller ID, for the highlighted line.	
Missed calls tips	Show the number of missed calls.	
Service status	AA: Auto-answer turned on.Mu: Mute turned on.HS: Headset turned on.	
Line Status / Speed Call	 (1)⁸⁰⁰¹ E Line is successfully registered to a SIP server. (2)⁸⁰⁰¹ E Line is not successfully registered to a SIP server. (3)^{8001:DND}: DND turned on in this line. 	
Softkey labels	Each displays a softkey function. To activate a softkey, press the softkey button.	
	Call activity Area Missed calls tips Service status Line Status / Speed Call	

Chapter 2 Preparations & Installation

Physical Installation

VIP-560PT/560PE: Enterprise PoE SIP IP Phone (2 x RJ-45, 1 x PoE for LAN interface)

Step 1: Handset Connection

Plug Handset Core with Handset and Handset Jack



Step 2a: Connecting Power Adapter and Network





The power adapter not bundles in standard pacage. Please use the DC 12V/1A standard power adapter with the unit to ensure correct functionality.

Step 2b: Connecting Power via PoE interface and Network

In the VIP-560PT/VIP-560PE can be powered without external power, if connecting to an IEEE802.3af PSE devices such as 802.3af POE injector/hub or 802.3af POE Switch.



(i)Note

Please don't connect PoE injector and AC adapter to VIP-560PT/PE at the same time, this may make the effect of current pulse and then cause device damage. Be noted to power the SIP IP phone either from 802.3af PoE or AC adapter.

Step 3: Foot-stand Adjust



(i) Note

Hold the two buttons before adjust the stand angle, bend the stand without holding the button will damage the stand.

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 camera by default). If you don't know how to do this, please ask your network administrator.

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 / 123.

Administration Interface

The IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration. Key pad administration also available for simple configuration.

Web configuration access

To start IP Phone web configuration, you must have one of these web browsers installed on computer for management

• Microsoft Internet Explorer 6.0.0 or higher with Java support

Default IP address of IP Phone is **192.168.0.1**. You may now open your web browser, and insert *http://192.168.0.1* in the address bar of your web browser to logon IP Phone web configuration page. IP Phone will prompt for logon username/password, please enter: *admin* / **123** to continue machine administration.



In order to connect machine for administration, please locate your PC in the same network segment (192.168.0.x) of IP Phone. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

Chapter 3 3

Network Service Configurations

Configuring and monitoring your IP Phone from web browser

The IP Phone integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

Manipulation of IP Phone via web browser

Log on IP Phone via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon IP Phone web configuration page.

IP Phone will prompt for logon username/password: root / null (without password)

Connect to 192.1	168.0.1	? 🛛
password. Warning: This server	68.0.1 requires a usernar is requesting that your us an insecure manner (basic nection).	ername and
User name:	£	~
Password:		
	Remember my passwo	rd
	ОК	Cancel

When users login the web page, users can see the IP Phone system information like firmware version, company...etc in this main page.

	Professional PoE IP Phone
Configure Guide Network SIP Account Programmable Button Expansion Module Audio Phone Book Advance Phone Maintenance Phone Status Advu	∵ Planet IP Telephony Management Interface
中文 English	VIP-560PT/PE
	Software Version: v2.0.1.6 System Upgrade: 2010.06.19 Web Version: 2.8.0.1 Web Upgrade: 2010.06.17

Network configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of IP Phone in the address bar. After logging on machine with username/password (default: **admin / 123**), browse to "**Network**" configuration menu:

etwo	rk		
	● DHCP		
	O Static IP		
	IP Address: 192.168.0.1		
	Netmask: 255, 255, 0		
	Gateway: 192.168.0.1		
	O PPPoE		
	Username:		
	Password:		
	MTU: 1500 (Default: 1500)		
	 Automatic Get DNS 		
	O Manual DNS		
	DNS		
	Primary DNS: 168. 95. 1. 1		
	Secondary DNS: 168. 95. 192. 1		
	MAC Address		
	MAC Address: 00:30:4f:00:0b:5c		
	Port Management		
	HTTP Port. 80		
	Telnet Port: 23		
	Attention: The default HTTP Port is 80, if you change it(for example		
	change it to 88),you must use IP and HTTP Port to login the web page (for example http: //192.168.0.200: 88).It will take effect on next reboot.		
Subn	it]		

Network Parameter Description			
IP address	LAN IP address of IP Phone		
	Default: 192.168.0.1		
Subnet Mask	LAN mask of IP Phone		
	Default: 255.255.255.0		
Default Gateway	Gateway of IP Phone		
	Default: 192.168.0.254		

After confirming the modification you've done, please click on the **Submit** button to apply settings and the machine will be reboot to make the settings effective.

Connection Type	Data required.
Obtain an IP Address Automatically	The ISP will assign IP Address, and related information.
Use the Following IP Address	In most circumstances, it is no need to configure the DHCP settings.
Behind xDSL Modem (PPPoE)	The ISP will assign PPPoE username / password for Internet access,

(i) Hint

Please consult your ISP personnel to obtain proper PPPoE/IP address related information, and input carefully. If Internet connection cannot be established, please check the physical connection or contact the ISP service staff for support information.

Chapter 4

VoIP IP Phone Configurations

General Settings

You can perform basic call-handling tasks using a range of features and services. Feature availability can vary; see your system administrator for more information.

Note: The bold type of the following text and following a "button" in table signifies the phone's button (for example, **OK** button), and the <u>New Call</u> signifies softkey.

Placing a Call

Here are some easy ways to place a call on IP Phone.

Item	Descriptions	
Headset	Toggles the headset on or off.	
	Red light means the feature is enabled.	
Mute	Toggles the Mute feature on or off.	
Mute	Red light means the feature is enabled.	
Massaga	Typically auto-dials your voice message service.	
Message	Red light means have unread voice mail.	
Service	Open or Close the Services menu. (Reserved for customization	
	purpose)	
Directories	Use it to access call logs and corporate directories.	
MENU	Allows you to scroll through menus.	
Volume	Controls the volume and other settings.	
Conference	Connect calling / called party to the conference.	
Redial	To Redial the last number.	
Transfer	Transfer redirects a connected.	
Hold	Put a call on hold.	
Number Keypad	Basic Call Handling: Press "#" sends out a call (default).	
Speaker	Toggles the speakerphone on or off.	

1) Flashing (Red): There is an incoming call.
2) Steady (Red): Pick up and enter normal call.

If you want to		Then
Place a call using the	Pick up the handset;	1) Hear the dial tone.
handset		2) The first line light.
		3) Enter number.4) Press "#" hutten(default)
Place a call using a	Press Speaker button	4) Press "#" button(default) or press Send
Speakerphone	or Line buttons	or wait five seconds (default)
	or New Call	Then send the call.
Place a call using a	Press Headset button.	
headset		
Redial	- Press Redial button to dial the last number or press Navigation	
	button -Right -> "Dialed nu	umber", select a number, and press Dial or
	OK button.	
Dial from the	1) Press MENU or OK button -> "Call history", you can select "Missed	
Directory on your	calls", "Received calls" and "Dialed numbers" or press Navigation	
phone	button (in Standby interface) > select "Missed calls" (Down),	
	"Received calls" (Left) and "Dialed numbers" (Right)	
	2) Then press OK button of	Dial
Place a call while	1) Press Hold button or <mark>Hold</mark>	
Another call is active	2) Enter a number.	
	3) Press '#' button (default)	or press Send to send the number.



You can dial on-hook, without a dial tone (pre-dial). To pre-dial, enter a number, and then go off-hook by lifting the handset or pressing Send , Headset or Speaker button.

If you make a mistake while dialing, press ${\bf C}$ button to erase digits.

Placing a Call

Here are some easy ways to place a call on IP Phone.

If you want to		Then
Place a call using the	Pick up the handset.	1) Hear the dial tone.
handset		2) The first line light.3) Enter number.
Place a call using a	Press Speaker button or	4) Press "#" button (default) or press
Speakerphone	Line buttons or New Call	Send or wait five seconds (default). Then send the call.
Place a call using a	Press Headset button.	
headset		
Redial	Press REDIAL button to d	ial the last number or press Navigation
	<pre>button-Right > "Dialed nu</pre>	mber", select a number, and press
	Dial or OK button.	
Dial from the	1) Press MENU or OK button	> "Call history", you can select "Missed
Directory on your	calls", "Received calls" a	nd "Dialed numbers" or press
phone	Navigation button (in S	tandby interface) > select "Missed calls"
	(down), "Received calls"	' (left) and "Dialed numbers" (right).
	2) Then press OK button or	Dial
Place a call while	1) Press Hold button or Hold	
Another call is active	2) Enter a number	
	3) Press '#' button (default) o	r press <mark>Send</mark> to send the number.

You can dial on-hook, without a dial tone (pre-dial). To pre-dial, enter a number, and then go off-hook by lifting the handset or pressing Send, Headset or Speaker button.
 If you make a mistake while dialing, press C button to erase digits.

Answering a call

You can answer a call by simply lifting the handset [,] or you can use other options if they are available on IP Phone.

If you want to		Then
Answer with a handset	 Your phone ring. Line button of the 	Pick up the handset
Answer with the speakerphone (Non-headset mode)	ringing line is Red and flashing, Light strip is Red	Press Speaker button or press the Line button flashing Red — or press Ans
Answer with the a	and flashing.	Put on headset, press Headset button so
headset		that the status light is Red e and then do as using speakerphone.
Switch from a connected Call to answer a ringing Call	e and flashing.	Red e and flashing, Light strip is Red Line button to answer (at this time, the
Auto-answer	2) Select "Enable".	utton > "Function setting" > "Auto answer". ncoming calls automatically after a few rings.

Ending a Call

To end a call, simply hang up, here are some more details.

If you want to	Then
Hang up while using the Handset	Return the handset to its cradle or press Reject
Hang up while using the	Press Speaker button that is Red 💗 or press Line button
Speakerphone	for the appropriate line or press Reject
Hang up while using the Headset	Press Handset button (Do not keep the headset mode) or
	press Reject (keep the headset mode)
Hang up one call, but preserve	Press Reject or refer to the above three methods

Using Hold and Resume

You can hold and resume calls. You can take a call in one line at anytime, and the other lines would be hold. As a result of that, you can switch different calling line on IP Phone.

If you want to	Then
Put a call on hold	Press HOLD button or press Hold
Hold a line and switch to another line	Press another Line button for the appropriate line
Resume a call on current line	Press HOLD button or press Hold
Release a call on different line	Select the line want to release hold, press the line, so
	recovery.

(i) Tips

Engaging the Hold feature typically generates music or a beeping tone.

A held call is indicated by the Yellow-green flashing Line button.

Transferring Calls

Transfer redirects a connected call. The target is the number to which you want to transfer the call.

If you want to	Then
Talk to the transfer recipient before	1) Press TRANSFER button or press Transf
transferring a call	2) Enter number.
(consult transfer)	3) press "#" (default) or press Send then transfer the call
	or wait five seconds(default)then transfer the call
Transferred to idle lines or other	1) Press TRANSFER button or Transf
numbers without talking to the transfer	2) Press <mark>Blind</mark>
recipient	3) Enter number

and

(Blind transfer)	4) Press " # " (default) or press Send, then transfer the
	call or wait five seconds(default)then transfer the call
Blind transfer to the	1) Press TRANSFER button or press Transf
held line	2) Press the Line button of held line

Using Mute

With Mute enabled, you can hear other parties on a call but they cannot hear you. You can use mute in conjunction with the handset, speakerphone, or a headset.

If you want to	Then
Toggle Mute on	Press Mute button, then the button is Red
Toggle Mute off	Press Mute button, then the button light off.

Do Not Disturb

You can use the Do Not Disturb (DND) feature to block incoming calls on your phone with a busy tone (Can also be set to their voice mail or other extension numbers, etc.).

If you want to	Then
Enable global DND	1) Press DND
	2) All enabled line on the phone would changes to 8001:DND status.
Enable DND on a single line	Press MENU or OK button > "Function setting" > "DND" > (select line)
	"Enable"
Disable DND	1) Global DND enabled, press DND to disable global DND.
	2) Line DND enabled, press twice DND or press MENU or OK button >
	"Function setting" > "DND" >(select line) "Disable"

3-way Conference

You can establish a three-party conference, during the conversation three phone parties can communicate with each other.

If you want to	Then
Invite the transfer recipient into a	1) When the transfer recipient answer the call, press
conference in a transferring	CONFERCENCE button or Confe on your phone.
	2) Then the held one, transfer recipient and you will be into a
	conference.
Invite the third party into a	1) Press CONFERENCE button or Confe in an active call.
conference in a active call	2) Enter the third party number.
	3) After connected the third party, press CONFERENCE
	button or <mark>Confe</mark> again.
establish a conference with held line	1) When one phone line is holding on and the other line is
	busy.
	2) Press CONFERENCE button or Press Confe Soft key
	3) Press the held line's Line button, the 3-way Conference will
	establish.

Advanced Call Handling

Speed Dialing

Speed dialing allows you to enter an index number, press a button, or select a phone screen item to place a call.

If you want to	Then
Set up Speed Dials on	1) Press MENU or OK button > "Function setting" > "Hot line keys"
your phone	2) You can configure twelve speed dial numbers on the IP Phone
	3) Press OK button or Modify to set and modify:
	-Mode:
	-Speed dial: Speed dial mode

-Asterisk BLF: In the Speed dial based on the increase in BLF
(Busy line detection) function
-Account: Speed Dial hot keys using the account
-Name: Description of this hot-key,
-Number: Need to speed dial numbers
4) Press Submit to save the changes

Using the phone book

You can store a large number of contacts in your phone's directory. You can add, edit, delete, dial, or search for a contact in this directory.

If you want to	Then
Add Contacts	1) Press Contac
	-or press MENU button > "Phone book",
	-or press Directories button > "Phone book";
	2) Press Modify
	3) Select "Add contact", press OK button or OK
	4) Use the navigation keys to select content, press OK button or
	Modify to set and modify:
	-Name: set the name of contact,
	-NO.1-5: you can set up 5 contacts' numbers,
	-Group: the contacts be divided into different user's groups
	5) Press Submit soft key to complete
Add group	1) Press Contac soft key or press MENU button > "Phone book" or
	press Directories button > "Phone book".
	2) Press Modify soft key
	3) Select the "add group" then press OK button or <mark>OK</mark>
	4) Use the navigation keys to select content, press OK button or

	Modify to set and modify.
	-Group name: name of the group
	-Description: description of the group
	5) Press Submit soft key to complete
Modify group	1) Press Contac soft key or press MENU button > "Phone book" or
	press Directories button > "Phone book".
	2) Press Modify soft key.
	3) Select the "Modify group" then press OK button or press OK
	4) Select the group you want to modify, press the OK button or
	Modify to set and modify, press Submit to save the change.
Delete group	1) Press Contac soft key or press MENU button > "Phone book" or
	press Directories button > "Phone book".
	2) Press Modify soft key.
	3) Select the "Delete group" or OK button or <mark>OK</mark>
	4) Select a group you want to delete, press OK button or OK
View/Edit Contacts	1) Press Contac soft key or press MENU button > "Phone book" or
	press Directories button > "Phone book".
	2) Select "View ALL" or select a contact that is belong to different
	group.
	3) Select the contact, press the OK button or View (to edit the
	contact's information, press OK button or Modify)
Call from phone book	1) Press Contac soft key or press MENU button > "Phone book" or
	press Directories button > "Phone book".
	2) Select "View ALL" or select a contact that is belong to different
	group.
	3) Select a contact, then press Dial (If there are multiple numbers of

	one contact, press Dial to enter the interface of "call options", select the one you want to call and press Dial)
Modify the relative account	1) Open your web browser; enter the "web" interface. (For details,
of a contact	you can refer to 7. Web Settings.)
	2) Open "Contact" > "Phone book", select the contact who are
	needed to be modified, click
	3) Select the account in the drop-down column of the account; click
	"Submit" to complete it.

Using Call Logs

Your phone maintains records of your missed, placed, and received calls.

If you want to	Then
View your call logs	1) Press MENU button > "Call history" > "Missed Calls", "Received
	Calls", or "Dialed numbers"
	2) Use the navigation keys to view the call record information.
Dial from a call log	Please refer to the previous part 4. Basic call handing – Placing a call.
Erase your call logs	1) If you want to delete a call record, you have to select this record from
	the logs and press Del
	2) If you want to delete an entire call record list, you have to select this
	record list from the logs and press Del All

& Tips

Each call log store up to 20 entries on IP phone.

Keypad Instruction

IP phones can be configured in two ways. The first you can use the phone keypad where you can settings for you IP phones, the other you can log in to User Options web pages where you can settings

for you IP phones.

Use phone keypad to setting. Press **MENU** or **OK** button to the main menu, Use the navigation keys to select menu, press **OK** button to confirm menu selections, press **C** button or **Del** to delete input information.

Language

IP Phone supports Simplified Chinese and English.

If you want to	Then
To change the language via	1) Choose "System setting" > "Phone setting" > "Language";
phone interface	2) Scroll through the list of available languages.
	3) Press OK button or Modify when the desired language is
	highlighted. The language appears on the graphic display will be
	changed to the one you chose.

SIP Account Settings

IP phone make calls based on SIP accounts, IP phone can support 4 independent SIP account, and each account can be configured to different SIP server.

If you want to	Then
Create an sip account	1) Choose "System setting" > "Advanced setting".
	2) Enter the password required (The default is empty).
	3) Choose "SIP" > "Account SIP".
	4) Choose one of the account you want to setting, you can configure the
	following parameters
	-Enable account*: choose Enable
	-Display Name: The name displayed on the screen
	-User Name*: the account matched with the SIP server. (extension
	number)
	-Authen usr: the Authenticated users matched with the SIP server.

	(The default With the same account)
	-user pwd*: the user password matched with the SIP server.
	-Description: description of this account
	-SIP1*: the primary SIP server, By default all calls through the server
	-SIP2: the secondary SIP, When the primary server is
	unavailable ,use the SIP server
	- Refresh time : Registration refresh interval, the minimum value is
	20 The default value is 3600.
	5) Set up the above parameters, press Submit softkey to saves settings,
	complete the account creation.
	* Note : the parameters with the * mark must be set.
Disable sip account	1) Choose "System setting" > "Advanced setting"
	2) Enter the password required (The default is empty)
	3) Choose "SIP" > "Account SIP"
	4) Choose "Enable account" > "Disable"
	5) Press Submit soft key

Network Setting

If you want to	Then
Network Setting	1) Choose "System setting" > "Advanced setting"
	2) Enter the password required (The default is empty)
	3) Choose "Network", you can configure the following parameters:
	- Type : static IP or DHCP
	-IP: enter IP address. Note: Do not duplicate the IP address with other
	devices on the network
	- Mask : enter appropriate sub mask
	-GW: enter appropriate gateway

 -DNS1: enter IP address of the primary DNS server

 -DNS2: enter IP address of the secondary DNS server

 -Web port: the default Web port is 80, if you change it (for example

 change it to 88), you must use IP and Web port to login the web page (for

 example http://192.168.0.200:88). It will take effect on next reboot.

 -Telnet port: the default Telnet port is 23, if you change it (for example

 changes it to 2003), you must use IP and Telnet port to login the manage

 page (for example telnet 192.168.0.200:2003). It will take effect on next

 reboot.

Customizing Rings and Volume

If you want to	This
Change the ring tone	1) Choose "System setting" > "Phone setting" > "Ring type"
	2) Press navigation to choose ring tone.
	3) Press Play softkey to choose a ring tone to play a sample of it.
	Press Stop softkey to Stop Playing
	Press OK or Select softkey to set the ring tone,
	Press Back softkey to return to previous menu.
Adjust the volume level	1) Choose "System setting" > "Phone setting" > "Volume setting"
	2) You can adjust the volume level of following types
	-Ring volume: Phone call ring volume
	-Handset volume: Handle output volume
	-Handset mic volume: Handle input volume
	-Speaker volume: Hands-free speaker output volume
	-Speaker mic volume: Hands-free input volume
	-Headset volume: Headphone output volume



Web Settings

We can configure the IP Phone handier through web setting. Press OK button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example http://192.168.0.200) into the address bar of web browser. The default login name / password are admin / 123.

	Professional PoE IP Phone
Configure Guide Network SIP Account Count Programmable Button Expansion Module Audio Phone Book Advance	∵ Planet IP Telephony Management Interface
 Phone Maintenance Phone Status About 	
	VIP-560PT/PE
	Software Version: v2.0.1.6 System Upgrade: 2010.06.19 Web Version: 2.8.0.1 Web Upgrade: 2010.06.17

Account and Sip server

The phone attempts to register to the SIP server using the account/registrar data provided by the automatic or manual initialization.

Field	Description
Enable	You can choose on/off to enable/disable the line.
Display Name	It is showed as Caller ID when making a phone call.
Username	It is authenticated ID for authentication.

Choose one Account, you will find the following parameters:

Authenticate Name	It is authenticated ID for authentication.
Password	It is provided by administrator for registration.
SIP Server	Server for registration, provided by administrator.
Register Expire Time	IP phone automatically registered every time.

Expansion Module

Expansion module is extended Hotline function; you can believe it support more hotline by using Expansion module.

EXT Mo	odule 1				
Key 1:	Mode:	BLF+Speed Dial 👻	Key 17:	Mode:	BLF+Speed Dial 🗸
	Account:	Account 1 🐱		Account:	Account 1 🐱
	Name:			Name:	
	Number:			Number:	
Key 2:	Mode:	BLF+Speed Dial 💌	Key 18:	Mode:	BLF+Speed Dial 💌
	Account:	Account 1 🐱		Account:	Account 1 💌
	Name:			Name:	
	Number:			Number:	

Expansion module	
Key n	Each Expansion module supports 32 keys.
Mode	Two modes:
	Speed Dial: Enable speed dialing in this key;
	BLF+ Speed Dial
Account	A SIP account relates to this key, another word, you will call this
	hotline by this SIP account.
Name	Description of this hotline.
Number	Number relates to this key.

*Regarding the settings of Expansion module, please confirm the model of your phone is VIP-560PE.

Codec Selection

The IP phone supports the following voice codecs: G.722, G.711A, G.711U, G.723, and G.729A.

You can enable/disable the desired codecs via Web interface. Please contact your system administrator for more details about the codecs.

To enable/disable the codecs:

1) Choose Audio-> Audio Codecs

Audio	
Tone	
Dial Tone: Dial Tone2 💌	Ring Volume(1~9) 3
Output Volume (1~9)	Intput Volume (1~7)
Handset Volume: 3	Handset Mic Volume: 3
SpeakerPhone Volume: 3	SpeakerPhone Mic Volume: 3
Headset Volume: 3	Headset Mic Volume: 3
Voice Codec	
Payload Length: 20 👻 ms	High Rate of G723.1: 💌
Other	
VAD:	Echo Suppression Mode: 📃
Ring	
Ring Type: Ring1 💌	Delete
Uploading Ring Tone	
Brower	
Upload Cancel	
(Please upload a ring tone with G729 audio coding ,and the size must less than 300k))
coung , and the size muscless than 500k)	
G711A	
Audio Codecs : Enable	C >>> Disable
Down G729A	
G722	
Submit	

2) Use the navigation keys to highlight the desired one in the Enabled/Disable Codecs list, and press

the >> / << to move to the other list.

- 3) Choose Submit to save the change.
- Of course, you can control the voice bulk in this choose.

Contact

You can add, edit and delete contact in a phone book on web page of IP Phone.

1) Click "Contact" > "Phone book",

If you want to add a contact, you just ought to click Add Contact .

You can edit an existed contact by click 🧖.

You can delete an existed contact by click \overline{m} , if you want to delete all contacts, you just ought to

click Delete All Contact .

2) When you add a contact or edit an existed contact, you can set several parameter as follow:

Contact	
Serial Number: 2 💌	Description:
First Name:	Family Name:
Mobile Number:	Home Number:
Office Number:	Fax Number:
OtherNumber:	Group: NONE 🗸
Company:	
Position:	
Email:	
Address:	
Account: Account	t1 🗸
Submit Cancel	

Phone book	
Serial number	Serial number of a contact
Description	Description of a contact
Name	Name of a contact
Phone n	You can add 5 different phone number for every contact
Group	You can assign a contact to a specific group. If there isn't any group
	set on the phone, the group is None by default.
Account ID	Select a SIP account relating this contact that is you can dial to the
	contact from this SIP account.

Besides, you can add, edit and delete group by click "**Contact**" > "**Group**". The operation is similar to phone book.

Appendix A Voice communications

There are several ways to make calls to desired destination in IP Phone. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

Case : Voice communication via IP PBX (IPX-300)



Machine configuration on the VIP-560PT:

STEP 1:

Log in IPX-300 and create two testing accounts/password: **100** / **123** (for VIP-560PT-A), and **200** / **123** (for VIP-560PT-B) for the voice calls.

STEP 2:

Please log in VIP-560PT-A via web browser, browse to the "SIP Account -> Account 1" menu. In the setting page, please enable SIP option and insert the Account / Password and SIP Server information obtained from your service provider (in this sample, we're using PLANET IPX-300 as the IP PBX system for SIP account, call authentications), and then the sample configuration screen is shown below:

Account 1		
SIP		
JIP	Enable:	
	Display Name:	100
	Username:	*
	Authenticate Name:	100
	Password:	*
	Label:	100
	SIP Server:	192.168.0.50
	Secondary SIP Server:	
	Outbound Proxy Server:	
	STUN Server:	

STEP 3:

Repeat the same configuration steps on VIP-560PT-B, and check the machine registration status, make sure the registrations are completed.

STEP 4:

To verify the VoIP communication, please pick up the telephone. Dial the destination number to make call between SIP clients. For example, VIP-560PT-A (with number 100) with keypad number 200 to VIP-560PT-B, or reversely makes calls from SIP client (VIP-560PT-B) to the number 100 (VIP-560PT-A).

Appendix B Expansion Module-VIP-56EXT Installation

In this section, we'll introduce how to install the Expansion Module- VIP-56EXT with VIP-560PE

Step 1. Mounting Bracket

Fasten 4 screws to daisy chained with VIP-560PE



Step 2. Connecting RJ-45 cable

Connecting IN port with VIP-560PE's EXT port



Step 3. Extending Module

Fasten 4 screws to daisy chained each other, connect OUT port with next module's IN port via RJ-45 cable



Step 4. Foot-stand Adjust

Pressing Foot-stand button to adjust the same angle with phone base





Hold the two buttons before adjust the stand angle, bend the stand without holding the button will damage the stand.

Appendix C Frequently Asked Questions List

Q1 : [VIP-560PT/VIP-560PE] I can not register to the server?

A1: 1. Check the IP address. If you set your LAN port in DHCP mode, please make sure that your DHCP server is on.

2. Check your gateway.

- 3. Check your DNS server.
- 4. Make sure your account information is the same as you have got from your ISP.
- 5. Check whether the SIP server is on.
- 6. Check the SIP register port, the default value is 5060.

Q2: [VIP-560PT/VIP-560PE] I can't get the IP address?

A2: 1.Make sure you have plugged the Ethernet cable into the LAN port.

- 2. Make sure that the DHCP server is on, and there are available IP addresses in the server.
- 3. Try to set your LAN port to static IP client mode.

Q3: [VIP-560PT/VIP-560PE] During a call, I can not hear any voice?

- **A3:** 1.Make sure Your handset is tightly connected with the phone.
 - 2. Check whether you have muted the conversation or not.
 - 3. Consult the outbound server details with your ISP.
- Q4: [VIP-560PT/VIP-560PE] Have DTMF problem?

A4 : 1.Check which kind of DTMF you are using, and whether it is compatible with the server2.Consult the payload value with your ISP

Q5: [VIP-560PT/VIP-560PE] How to change the time?

A5: Go to "Advance -> Phone Setting" page change to Manual for Set Time mode at first, then to fill in the date and time at fitting fields.

Note: The IP Phone will automatically get the time from the SNTP server for default values.

Q6: [VIP-560PT/VIP-560PE] How to answer the incoming calls during a call?

A6: If a call comes in when you are in a conversation, press the flashing Line key to answer the call, and the original call will been held.

Q7: [VIP-560PT/VIP-560PE] How to send SMS?

A7: You could edit the SMS in the MENU -> Messaging -> Create messages.

Note: Make sure that the SIP server you have registered supports SMS function.

Q8: [VIP-560PT/VIP-560PE] How to update the firmware?

A8: 1. Update the firmware on the webpage "Phone Maintenance -> TFTP/HTTP Upgrade".

2. Select the correct file you want to download to the IP Phone then click the "Upgrade" button.

Q9: [VIP-560PT/VIP-560PE] How to adjust volume?

A9: During a call, press VOL+▲/ VOL-▼ key to adjust the volume of earpiece or speaker.

Q10: [VIP-560PT/VIP-560PE] How to select ring?

A10: 1. There is eight kinds of ring styles to choose.

2. Go to "MENU -> System setting -> Phone setting Ring Type" page, press the ▲/ ▼ key on the phone to choose the ring type.

Q11: [VIP-560PE] Why the IP Phone can't work when connect more than 4 VIP-56EXT?

A11: If the IP Phone be powered by 802.3af POE injector/hub or 802.3af POE Switch, the IP Phone just could connect to 3 VIP-56EXT. If the IP Phone needs to connect more than 4 VIP-56EXT, it needs be powered by external power adapter to make sure all units could work properly.

Appendix D Specifications

Product	Enterprise PoE IP Phone			
Model	VIP-560PT	VIP-560PE		
Hardware				
LAN	1 x 10/100 Base-TX RJ-45 port (802.3af support)			
PC	1 x 10/100 Base-TX RJ-45 port			
Display	240 x 160 pixels LCD with backlight			
Supports Expansion Module	Up to 6 x VIP-56EXT (192			
	N/A	programmable buttons)		
Headset Jack	RJ-11, 3.5 mm (Mic / Speaker separately)			
Function Keys	4 x Line Buttons (Red LED)			
	4 x Soft Buttons			
	5 x Navigation Buttons			
	2 x Volume Buttons			
	8 x Programmable Buttons			
	12 x Fixed Function Buttons (Headphone, MUTE, Voice Mail, Service,			
	Directory, Menu, Conference, Redial, Transfer, Hold, Speaker, Delete)			
Protocols and Standard				
Standard	SIP 2.0 (RFC 3261)			
	RTP(RFC 2833)			
	STUN (RFC 3481)			
	SNTP (RFC 2030)			
	TCP/IP, UDP, HTTP, ICMP, DNS, NTP			
	STUN (RFC 3481)			
	Outbound Proxy			
VPN Network	L2TP, IPSec encryption			
Voice Code	G.711(A-law /µ-law), G.722, G.723.1 (6.3 Kbps / 5.3Kbps), G.729 A			
Voice Standard	Auto negotiation			
	Acoustic echo cancellation for integrate	ed speakerphone operations		
	Voice activity detection			
	Silence suppression			
	Comfort noise generation			
	Dynamic Jitter buffer			
	Call Progress Tone Generation			
	-	DTMF (RFC 2833) / Out-of-band DTMF		
	(SIP Info)	, , , , , , , , , , , , , , , , , , ,		
Security	User Authentication for configuration p	ages		
	Signaling encryption	-		
	Media encryption			
Features				
Call Features	Caller ID display			
	Address Book			
	Missed Calls			
	Received Calls			
	-			

	Placed Calls	
	Date / Time Display	
	Speed dial configuration (mapped to 8 speed dial key)	
	Network Setting	
	Time Zone Setting	
	SIP port configurable	
	RTP port configurable	
	Call Forward: Busy Forward / No Answer Forward / Unconditional Forward	
	Call Waiting	
	Call Transfer	
	Call Hold	
	3-Way Conference	
	DND (Do Not Disturb)	
	Multi-line Appearance / Registration (up to 4)	
	Selectable Ring Tone	
	Distinctive Ring by Group Name	
	NTP Support	
	LCM Contrast Adjustment	
	BLF (Busy Lamp Field), BLA (Bridged Line Appearance)	
Network and Configuration		
Internet Connection Type	Fixed IP, DHCP, PPPoE	
Management	LCD / Keypad UI	
	Web (HTTP)	
	Auto Provision (TFTP)	
Dimension (W x D x H)	254 x 205 x 87 mm	
Operating Environment	0~50 Degree C, 10~90% humidity	
Power Requirement	12V DC, PoE (IEEE 802.3af)	
EMC/EMI	CE, FCC	