SIP / H.323 PoE IP Phone

VIP-100PT / VIP-150PT

User's Manual

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Chapter 1

Overview of the PoE IP Phone

Meeting the next-generation Internet telephony service demands, PLANET Technology provides feature-rich, toll-quality Internet telephony service solutions. The 802.3af Power over Ethernet (PoE) IP Phone – VIP-100PT / VIP-150PT.   VIP-100PT is H.323 v2/v3/v4 compliant with H.235 security support, the VIP-152T is SIP 2.0 (RFC3261) compliant   with SIP digest authentication supports. The VIP-100PT / VIP-150PT feature high-quality speakerphone technology; also include an easy-to-use speaker on/off button and call hold/transfer buttons for various voice service. These features beyond the conventional voice systems nowadays, and the PoE IP phones are cost-effective solution for Internet Telephony Service Provider (ITSPs) communications and interoperates with major other-vendor VoIP hardware and systems as well as the traditions of PLANET VoIP family.

As feature-rich IP Phones, the VIP-100PT / VIP-150PT fulfill your needs. They are simple to use, and have additional features such as: built-in PPPoE/DHCP clients, password-protected machine management, LCD menu display, hands-free speakerphone, last number redial, incoming message indicator, and user-intuitive web administration system.

The VIP-100PT / VIP-150PT are self-contained, service-integrated IP phones— offers intelligent phone features, and powerful voice procesing power. The VIP-100PT / VIP-150PT can effortlessly deliver toll voice quality equivalent to the regular PSTN connections utilizing cutting-edge Quality of Service, echo cancellation, comfort noise generation and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the VIP-100PT / VIP-150PT allow users to install in an existing network location without interfering with desktop PC network connections..The new VIP-100PT / VIP-150PT deliver more convenience, efficiency, innovation and benefits of VoIP into routine tasks, and dailylife.

PLANET PoE   IP Phones – VIP-100PT / VIP-150PT are ideal IP phone solution for office / home 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 your daily life benefit from the VoIP technologies.

VIP-100PT / VIP-150PT Features

Simple Installation and administration
Configuration of the VIP-100PT / VIP-150PT can be performed in minutes via the keypad, or web interfaces. Using the built-in LCD display, the VIP-150T offers user-friendly configuration guidelines; machine operation status, call status displays, and incoming call identification.

Feature-rich keypad IP Phone
The VIP-100PT / VIP-150PT integrate a high-quality speakerphone with the Call Hold, Forward and Transfer functions and also provides advanced telephone features, such as last number redial, incoming call history indicator in a much more convenient and functional manner than traditional telephone sets.

Dynamic IP address assignment, and voice communication

The PLANET VIP-100PT / VIP-150PT can act as a PPPoE/DHCP client, automatically obtaining an IP address for Internet access.

Various field applications compliant

The VIP-100PT / VIP-150PT are capable of handling both peer-to-peer and H.323GK / SIP proxy registration, authentication, to interact with major H.323/SIP gateway/IP Phone on the market. VIP-150T offers the most flexibility and interoperability with PLANET and 3rd party VoIP vendors, allowing the deployment of both simple and complex VoIP networks such as ITSP, PC-to-Phone/Phone-to-PC or enterprise VoIP environments.

Standards compliant

The VIP-100PT complies with H.323 version 4, and supports H.323 fast start, H.245 tunneling, GateKeeper H.235 security mode for different occasions. The VIP-150PT complies with SIP 2.0 (RFC3261), interoperates with 3rd party SIP voice gateways/terminal/software as well as other PLANET VoIP products. Supported Voice codecs and VoIP technologies are: G.723, G.729ab, G.711mu/ma; Voice Activity Detection (VAD), and the Comfort Noise Generation (CNG).

1.1 Hardware Overview

1.1.1 Front View and Keypad function

PoE IP phone overview

Keypad Description

Menu/OK

When the IP phone is at idle state, this key is used as "Menu" key to bring out the menu selection on the LCD display.

When inside the menu selection/setting on the LCD display, this key is used as "OK" key to enter into a lower layer of menu selection or to accept the edited item's contents.

When the IP phone is on dial mode, the "OK/Menu" key is used as "Dial Out" key.

LEFT

When the IP phone is at idle state or talking on handset or speaker, this key is used to increase the volume of the voice sound.

The volume of speaker, handset and ring are separately adjusted according to the mode of current usage. When in idle, "←" key increase the volume of ring tone, in hand free mode, "←" key increase the volume of speaker, when in handset mode, "←" key increase the volume of handset.

When the IP phone is entered into the menu selection, this key is used to scroll up the menu items.

And when the IP phone is editing some menu item's contents, this key is used as " left delete" to delete a digit per each key press.

When the IP phone is on dial mode, the "←" key is used as "delete" key.

RIGHT

When the IP phone is at idle state or talking on handset or speaker, this key is used to decrease the volume of the voice sound.

The volume of speaker, handset and ring are separately adjusted according to the mode of current usage. When in idle, "→" key decrease the volume of ring tone, in hand free mode, "→" key decrease the volume of speaker, when in handset mode, "→" key decrease the volume of handset.

When the IP phone is entered into the menu selection, this key is used to scroll down the menu items.

And when the IP phone is editing some menu item's contents, this key is used as " right shift" to shift the cursor right a digit per each key press.

CANCEL

When the IP phone is entered into the menu selection, this key is used to escape to an upper layer of the menu selection.
And when the IP phone is editing some menu item's contents, this key is used to cancel current edit and escape to an upper layer of the menu selection.

TRANSFER

This key is pressed to transfer an active call to another IP phone. When the IP phone is active (incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another IP phone's number to transfer the call to another party. If the call transfer is successful, a busy tone will appear to notice the user to on-hook the phone. If the call is failed, the user can press this "Transfer" key again to retrieve back this original call connection.
When a call is incoming and this IP phone is ringing, by pressing this "Transfer" key and then another IP phone's number can transfer the call immediately to another party without answering the call.

RE/DIAL

When the IP phone is off-hooked and dialed some number, this key is pressed to call out. The "#" key has this same function to send out the called number.
When the IP phone is off-hooked and this key is pressed immediately, the last dialed number will be called out right away.

HOLD

To hold the conversation.
This key can also be pressed to do consultant-transfer of an active call to another IP phone. When the IP phone is active (incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another IP phone's number to call to another party and have a conversation with him. Then, by pressing "TRANSFER" key, the call will be transfer to this new party.

SPK

This key is pressed to switch between the usage of the handset and the speaker/microphone devices.

#

The "send " key after phone number is completed during phone dial.
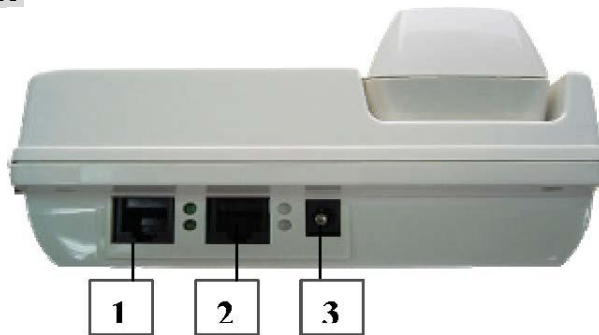
*

Work as symbol "." (dot) when input IP address.

Number 0~9

These keys are used to dial the call out number when the handset is offhooked. But when in the menu item editing mode, these keys can be used to toggle between digits and all English characters by quickly re-pressing the same key.

1	"1" ; " @" ; "-" ; "_"
2	"A" ; " B" ; "C" ; "2"
3	"D" ; "E" ; "F" ; "3"
4	"G" ; "H" ; "I" ; "4"
5	"J" ; "K" ; "L" ; "5"
6	"M" ; "N" ; "O" ; "6"
7	"P" ; "Q" ; "R" ; "S" ; "7"
8	"T" ; "U" ; "V" ; "8"
9	"W" ; "X" ; "Y" ; "Z" ; "9"
0	"0"

1.1.2 Rear View

Rear Panel of VIP-100PT/VIP-150PT

1	PC	RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable
2	WAN	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.
3	12V DC	12V DC Power input outlet

ⓘ Hint

- When VIP-100PT/VIP-150PT are installed with 802.3af PoE device, please make sure the PoE is connected to the machine **WAN** interface to be operational.
- The Power over Ethernet support on PLANET PoE IP phones complies with the 802.3af standards. Using non-802.3af compliant PoE device will burn up the IP Phone permanently.

Chapter 2

Configuring the PoE IP Phone through LCD Phone menus

2.1 Initialize VIP-100PT/VIP-150PT

SETP 1:

When power on the VIP-100PT/VIP-150PT, the LCD screen shows as below.

```
Starting
...
```

SETP 2:

Wait about 10 seconds, the main LCD screen shall be shown as following, and when the Internet access is available, the VIP-100PT/VIP-150PT will connected to the SNTP server. The current time will be displayed on the LCD.

```
Hi (GK off)

                          Date Time
```

LCD screen of VIP-100PT (H.323)

```
SIP (Proxy off)

                          Date Time
```

LCD screen of VIP-150PT (SIP)

1. Press **MENU** to enter configuration mode then press **OK** button to enter sub menus; press **CANCEL** can jump out current menu to previous level.

   1. **View**

   2. **Configure (Password: 123)**

   3. **Advanced Setting (Password: 1234)**

2.2 LCD Menu Configuration

1st Level	2nd Level	3rd Level
View		
	1.1 Network Value	1.1.1 IP Address 1.1.2 Network Mask 1.1.3 Default Route 1.1.4 DNS Server
	1.2 Ping	
	1.3 Restart	
	1.4 Firmware Version	
	1.5 (Yes/No) PKT Trace	
Configure		

	2.1 Network	2.1.1 (Yes/No) Dynamic IP
		2.1.2 (Yes/No) PPPoE
		    If Yes: PPPoE User Name
		            PPPoE Password
		2.1.3 Fixed IP
		       IP Address
		       Network Mask
		       Default Route
		       DNS Server
		2.1.4 GMT

	2.2 H323 (VIP-100PT)	2.2.1 Number 2.2.2 Password 2.2.3 H.323 ID 2.2.4 (Yes/No) Reg to GK 2.2.5 (Yes/No) Full RRQ

	2.2 SIP (VIP-150PT)	2.2.1 Display Name
		2.2.2 Login ID
		2.2.3 Telephone Number
		2.2.4 Password
		2.2.5 (Yes/No) Proxy On
		   If Yes: Proxy Address
		          Proxy Port
		2.2.6 (Yes/No) Outbound Proxy
		   If Yes: Outbound Proxy IP
		          Outbound Proxy Port
		2.2.7 (Yes/No) RTPLowBw

	2.3 Forward Mode	2.3.1 (Yes/No) Immediate
		   If Yes: Immediate Number
		2.3.2 (Yes/No) Busy
		   If Yes: Busy Number
		2.3.3 (Yes/No) No Answer
		   If Yes: No Answer Number
		2.3.4 No Answer Time

"View" menu

In the "**View**" main menu, there are submenus, users may check these items for machine information and also provide some simple functions to check the network status of the IP Phone.

1.1 View current network settings

By selecting "**View**" → "**Network**", you could see the active network parameters in machine.

1.2 Ping

By selecting "**View**" → "**Ping**" and key in the IP address or domain name of another device, the IP Phone can verify the connection between remote device. The "**ping**" command also can be used to verify the Internet connection status via pinging an Internet node

1.3 Restart

By selecting the "**View**" → "**Restart**", the PoE IP Phone will reboot itself immediately. This "warm restart" is different from the power down/up (cold restart).

(i) Hint

- Restarting the IP phone will perform the following tasks:
- Disconnect all the voice communications, and Internet connections
- Registration toward the gatekeeper/proxy server will be torn down.

1.5 Display current software version

By selecting the "**View**" → "**Image Version**", the users can view current software version in the PoE IP Phone.

1.6 PK Trace for signal monitoring

PLANET PoE IP Phones integrate self-analysis packet tracing algorithm, which is able to perform machine signaling monitoring task, By enabling the "**View**" → "**PKT Trace**" to (**Yes**), the LCD screen will display signaling messages received, and this can be used to monitor the signals and identify the machine running status.

Network Configuration

The first thing in using the PoE IP Phone is to set the network configuration to let the IP Phone connect to Internet. Depends on the network environment and phone models, please use the proper method to configure the IP Phone to connect to Internet.

2.1 Dynamic IP Method (DHCP Client)

When the IP phone installed in a DHCP server enabled network environment (for example,

cable modem connection or corporation internal network). Please browse to machine network configuration menu, and enable the DHCP client in **Dynamic IP** selection menu, and the IP phone will obtain the IP parameters dynamically, and connect to Internet automatically.
Configuration steps:

> Configure → Password：123 → Network → (Yes/No) Dynamic IP

2.1.2 PPPoE

Most of the broadband network environment provided now by ISP is the ADSL connection. In the ADSL environment, the IP Phone can directly connect to the ADSL modem by setting the PPPoE account (username and password) provided by ADSL service provider, the configuration steps are shown below: .

> Configure → Password：123 → Network → (Yes/No) PPPoE

Please browse to machine network configuration menu, and enable the PPPoE service in machine, and setup the username/password fields for Internet access.

PPPoE Username – please input the user name of the account given by ADSL ISP.

PPPoE Password – please input the password of the account given by ADSL ISP.

2.1.3 Fixed IP

Under the "**Fixed IP**" submenu, please key in the IP address, network mask and default router settings provided by your ISP or private IP address.

> Configure → Password：123 → Network → Static IP

ⓘ Hint
```
Once all the network settings are completed, please restart your
IP phone. After rebooting, please check Internet connection status
via View -→ Ping, then insert in a public, live IP address (for
example, 168.95.1.1) to ping it. When the response is ok, then the
network configurations are completed.
```

2.1.3 GMT

Via selecting the time zone in the GMT menu, VIP-100PT/ VIP-150PT is able to retrieve time information from timeserver on the net.

> Configure → Password：123 → Network → GMT

2.2 Registration to H.323 gatekeeper / SIP Proxy Server

After setting up the network parameters, the IP Phone is able to perform registration to the H.323 gatekeeper (VIP-100PT) or SIP Proxy server (VIP-150PT) when you've been assigned the account from VoIP vendor/service provider.

2.2.1 Registration to a H.323 gatekeeper (VIP-100PT)

When VIP-100PT is installed in an H.323 GK enabled network, the VIP-100PT registers to the gatekeeper for voice communications. Following descriptions are parameters to complete the registration.

Configure → Password：123 → H.323

Telephone Number:

Telephone phone number used to register toward the gatekeeper.

Password:

Password used to register to the gatekeeper.

This password will be encapsulated in the H.235 message for authentication purpose. Not every gatekeeper demands this value for authentication; please skip this field if your ISP does not provide this information.

H.323 ID:

The H.323 ID will be used to register to the gatekeeper. Not every gatekeeper demands this value for authentication; please skip this field if your ISP does not provide this information.

ⓘNote
```
The telephone number, H.323 ID and password fields are
application-oriented parameters. Please be sure to check with your
service provider to get the proper information before setting up
these fields.
```

GK mode (Yes/No):

Selecting **Yes** will enable machine in GK mode, and register to the gatekeeper. (Machine default mode is Peer-to-Peer (**P2P**))

If you'd like to change the GK settings, it is suggested to log in machine web interface, and browse to H.323 configuration menu to adjust related parameters. When VIP-100PT fails to register with the gatekeeper, the IP phone still able to establish voice calls toward other calling parties Peer-to-Peer via dialing the IP address directly on the keypad.

RTPLowBW (Yes/No):

This ability is to decrease the voice packet bandwidth of the transmit direction in order to improve the voice quality of transmit direction if the Internet bandwidth is not good enough.

2.2.1 Registration to a SIP Proxy Server (VIP-150PT)

When the VoIP vendor/operator is running the SIP system, use the VIP-150PT to register to the proxy server. Configure the following parameters to do the registration.

> **Configure** → **Password:123** → **SIP**

Display Name:

SIP identification

Login ID:

The username used to register to Proxy server. If Proxy server doesn't request specific username, please enter the assigned telephone number here.

Telephone Number:

Please input the phone number to register to the proxy server.

Password:

Please input the password to register to the proxy server.

This password is encapsulated in the SIP Proxy-Authorization field using MD5 digest method for authentication purpose. Not every proxy server needs this field, if not needed, keep it empty.

ⓘNote
```
• The SIP telephone number, login ID and password fields are
  application-oriented parameters. Please be sure to check with
  your service provider to get the proper information before
  inputting parameters.
• VIP-150PT follows the RFC-2617 in the SIP authentication
  applications.
```

Proxy mode (Yes/No):

Please check the application demands and select the appropriate operation mode. Selecting **Yes** to enable SIP proxy mode, and requires additional data input (such as the username, password, server information) to establish the voice communication, SIP proxy is mostly adopted in ITSP (Internet Telephony Service Provider). Selecting **No** will enable machine in **Peer-to-Peer** mode.

After the SIP proxy mode is enabled, two configuration menu will appear: one is

proxy server address, to set the IP address or domain name address of the proxy server. Another is the **proxy server port**, to set the port of the proxy server; Default proxy server port is **5060.** Please do not change the proxy server port number unless your service provider gives these parameters.

If the IP Phone does not register to any proxy server, it still can call to other IP phone by calling the IP address directly.

Outbound Proxy (Yes/No):

Please set this item to **Yes** if the registration needs to pass through the Outbound Proxy server.

Registration / Startup message

When the network and registration configurations are set, please restart the IP Phone, the LCD display on the IP phone will show one of the following message depends on whether the registration is ok or not.

When the VIP-100PT is registering to a H.323 gatekeeper successfully, the LCD screen will display the following message:

Registering (Number)
Date Time

Means that the VIP-100PT is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone 's number.

Hi (GK off)
Date Time

Means that the VIP-100PT is working OK and ready for outgoing/incoming calls. But the "GK mode " flag is "No", i.e. the VIP-100PT not need to register. In this case, can call the IP address of other VIP-100PT directly.

When the IVIP-150PT is registering to a SIP proxy server successfully, the LCD screen will display the following message:

Registering (Number)
Date Time

Means that the VIP-150PT is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone 's number.

```
┌─────────────────────────────────────────┐
│ SIP (Proxy Off)                          │
│                                          │
│                              Date Time   │
└─────────────────────────────────────────┘
```

Means that the VIP-150PT is working OK and ready for outgoing/incoming calls. But the "Proxy On" flag is "No", i.e. the VIP-150PT not need to register. In this case, can call the IP address of other VIP-150PT directly.

When the PoE IP Phone is failed in registering to a H.323 gatekeeper or SIP proxy server, the LCD screen will display the following message:

```
┌─────────────────────────────────────────┐
│ RegFail (Failed message)                 │
│                                          │
│                              Date Time   │
└─────────────────────────────────────────┘
```

The failed message could be one of the followings:

Duplicate: means that the registering number is duplicated with others, or the PoE IP Phone's previous registration information is still kept in the gatekeeper/proxy server if not unregistered last time (this could happened when the PoE IP Phone is powered off instead of restarted from the menu item). If the previous registration information is not cleared, may need to wait about 4 minutes to let PoE IP Phone register successfully again.

Security: means that the account (username/password/H.323 ID) is not correct, please check your account again.

When the **RegFail** message displayed, the IP phone cannot make any calls now.
But the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is failed in registering to a H.323 gatekeeper or SIP

```
┌─────────────────────────────────────────┐
│ DHCPFail (Number)                        │
│                                          │
│                              Date Time   │
└─────────────────────────────────────────┘
```

Means that the PoE IP phone is configured to use DHCP to get IP address, but the DHCP procedure failed (can not find a DHCP server or the DHCP server rejected the IP assignment). The IP phone cannot make any calls and the onhook/offhook

functions cannot work cannot hear any sound on the handset and the speaker. Only the menu selection can work, now.

The PoE IP phone will continuously try to get the IP address from the DHCP server, so, if the DHCP server is responding, the **DHCPFail**() display will change to Hi() display to notify that the PoE IP phone is working OK.

When the PoE IP Phone is set to use the PPPoE method for network connection, but have some problem in finding the PPPoE server (ADSL modem), the LCD screen will display the following message:

PPPoE FindFail (Number)
Date Time

Means that the PoE IP phone is configured to use PPPoE to get IP address, but the PPPoE procedure failed (can not find a PPPoE server or the network connection has some problem). The IP phone cannot make any calls, but the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is set to use the PPPoE method for network connection, but have some problem in the PPPoE account, the LCD screen will display the following message:

PPPoE AuthFail (Number)
Date Time

Means that the PoE IP phone is configured to use PPPoE to get IP address, but the PPPoE server refuse the connection because the username/password is not correct. The PoE IP phone cannot make any calls, but the menu selection and the onhook/offhook function can work OK.

When the PoE IP Phone is enable the "**Immediate Forward**", the LCD screen will display the following message:

```
┌─────────────────────────────────────┐
│ **FWD** (Number)                     │
│                                      │
│                          Date Time   │
└─────────────────────────────────────┘
```

Means that the VIP-100PT is working OK and ready for outgoing/incoming calls. But for any incoming call, it will be forwarded to the "Immediate Forward Number"; this PoE IP Phone will not ring.

2.3   Call Forward Configurations

The PoE IP Phone supports three different kinds of call forward functions, please select "**Configure**" → "**Forward Mode**" for these three kinds of selection:

2.3.1 Immediate Forward

Under "**Forward Mode**" submenu, users are able to setup for immediate forward by selecting Yes on (Yes/No) Immediate and input the number he/she would like the call to be forwarded to. For example, when A calls B, and B has enabled the immediate forward with number 555-5555 on the IP phone. In this circumstance, this phone call from A will be forwarded to number 555-5555.   When the "immediate forward" is enabled, the "busy forward" and "no answer forward" cannot be enabled at the same time.

2.3.2 Busy Forward

Under "**Forward mode**" submenu, users are able to setup for busy forward by selecting **Yes** on "(Yes/No) **Busy**" and input the number he/she would like the call to be forwarded to when this phone is busy (active). For example, when A calls B, and IP phone B is busy and "**busy forward**" has been enabled with number 555-5555, then, this phone call from A will be forwarded to number 555-5555.

2.3.3 No Answer Forward

Under "**Forward mode**" submenu, users are able to setup for no-answer forward by selecting Yes on "(Yes/No) No Answer" and input the number he/she would like the call to be forwarded to when this phone is not answered for the specified seconds. For example, when A calls B, and B's phone is not answered and has Yes on "(Yes/No) No Answer" with number 555-5555 and no-answer time equals 10, then, this phone call from A will be forwarded to number 555-5555 after ringing for 10 seconds no answered.

Chapter 3

Configuring the VIP-100PT/ VIP-150PT through Web Pages

VIP-100PT/VIP-150PT are equipped with 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.

Step 1. Browse the IP Address of your IP phone

After TCP/IP configurations on the phone, you may now open your web browser, and input machine default IP address **192.168.0.1** to logon VIP-100PT/VIP-150PT web configuration page.

Step 2. Input the login name and password

- ■ Login name: **root**
- ■ Password: Null (no password)

PoE IP Phone Web Management

- ■ Username: [   ]
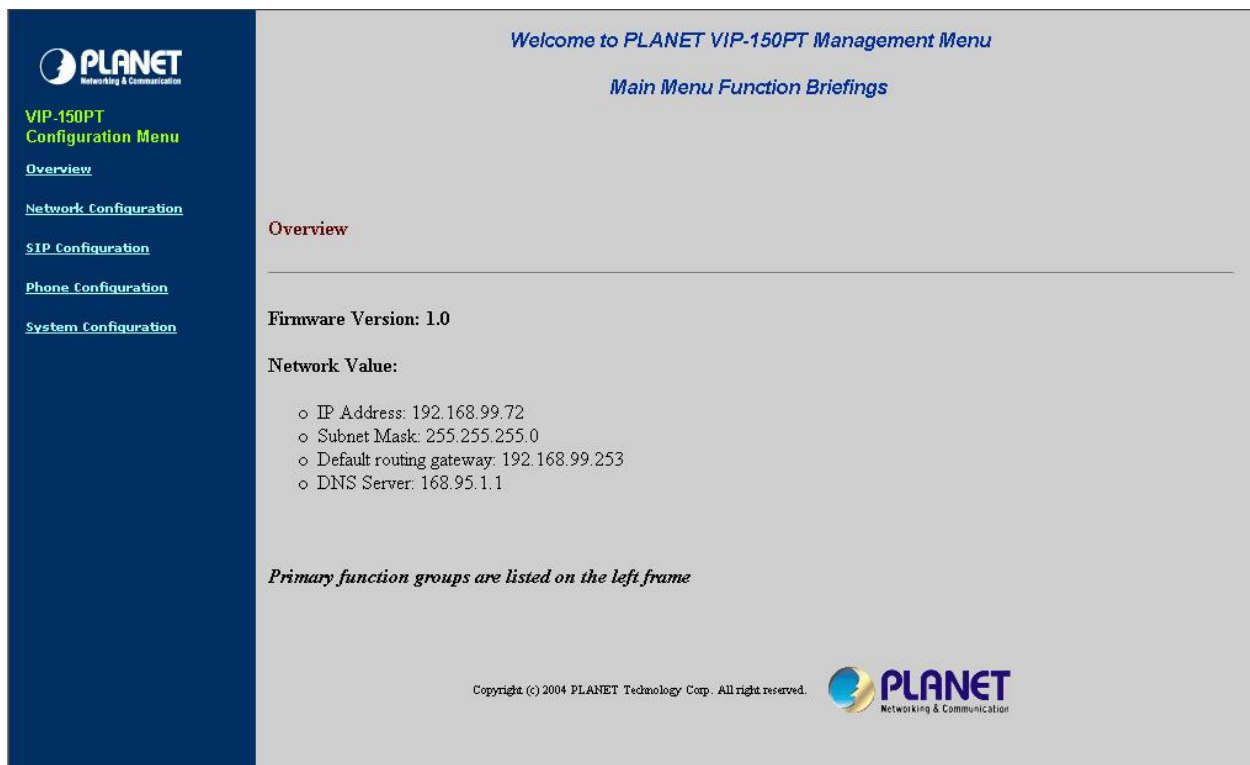- ■ Password: [   ]

[LOGIN]

Step 3. Enter the web interface main screen

After logging in machine, machine main screen is shown below.

1. Overview

Network, and firmware information of the PoE IP Phone will be displayed on machine homepage; more machine configuration menu can be found in the left frame:

- **Firmware version**: machine firmware version information
- **Network value**: (
  IP Address: 192168.0.1
  Subnet Mask: 255.255.255.0
  Default routing gateway: 192.168.0.254
  DNS Server: 168.95.1.1

Step 4. Start machine configurations

Most important items are **Network Configuration** and **H.323/SIP Configuration**. Please remember to configure these commands before start to work with VIP-100PT/VIP-150PT.

ⓘNote

```
After parameter modifications, it is required to browse to the
"System Configuration" page to reboot VIP-100PT/VIP-150PT to make
changes effective.
```

---

2. Network Configuration

Please select the "**Network Configuration**" icon on the left menu; the following page will display to allow you to set all the network related configurations

Network Configuration

- **Network Connect:**
  1. Dynamic IP: ☐
  2. PPPoE: ☐
     1. PPPoE Username: [_____]
     2. PPPoE Password: [_____]
  3. Fixed IP:
     1. IP Address: [192.168.0.1]
     2. Network Mask: [255.255.255.0]
     3. Default Router: [192.168.0.254]
     4. DNS Server: [168.95.1.1]
              [DONE]

- **Others:**
  1. MAC Address: 00304f168888

  2. SNTP Server Address: [195.13.1.153]
  3. GMT: [8]   View
              [DONE]

Network configurations:
- **Dynamic IP**: Enable/disable DHCP Client function.
- **PPPoE**: Enable/disable PPPoE function.
  1. PPPoE Username: Set PPPoE authentication User Name.
  2. PPPoE Password: Set PPPoE authentication password.
- **Fixed IP**
  3. IP Address: IP Address of VIP-100PT/VIP-150PT
  4. Subnet Mask: setup the subnet mask of VIP-100PT/VIP-150PT
  5. Default routing gateway: setup default gateway of VIP-100PT/VIP-150PT
  6. DNS Server: setup domain name server IP address.

Others:
- **MAC Address**: MAC address of the PoE IP Phone
- **SNTP Server Address**: setup SNTP Server Address
- **GMT**: setup time zone for SNTP Server time

ⓘNote
- When SNTP server is available, enable VIP-100PT/VIP-150PT SNTP function to point to SNTP server IP address, so that VIP-100PT/VIP-150PT can obtain proper time information from the SNTP server.
- You may enable different time zone to meet the location of VIP-100PT/VIP-150PT. For example, the time zone in Taiwan should be set as 8,which means GMT+8.

After setting up network parameters, be sure to apply settings (via "**DONE**" button) and browse to "**System Configuration**" menu to reboot the machine to make the settings effective

Connection Type	Data required.
Dynamic IP (DHCP client)	In most circumstances, it is no need to configure the DHCP settings..
PPPoE	The ISP will assign PPPoE username / password for Internet access,
Fixed IP Address	The ISP will assign IP Address, and related information.

(i) Hint

```
When there is difficulty connecting your IP phone to the
Internet, 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.
```

3. H.323 Configuration (VIP-100PT Only)

Please select the "**H.323 Configuration**" icon on the left menu; the following page will display to allow you to set all the H.323 related configurations.

H323 Configuration

- **H323 Parameters:**
  1. Telephone Number: `100`
  2. Password: `[        ]`
  3. H.323 ID: `[            ]`
  4. GK mode: ☐
     1. GK IP: `[         ]`
     2. GK Port: `1719`

  [DONE]

- **Forward Mode:**
  1. Immediate: ☐
  2. Immediate Number: `[          ]`
  3. Busy: ☐
  4. Busy Number: `[          ]`
  5. No Answer: ☐
  6. No Answer Number: `[          ]`
  7. No Answer Time: `0`

  [DONE]

- **Advanced Setting:**
  1. RTP Port (Starting): `10000`
  2. RTP Port (Ending): `10004`
  3. H.323 Fast Connect: ☑
  4. H.245 Mode: ⊙ Tunneling ○ Separate ○ None
  5. H.235 Security: ⊙ Common ○ CAT ○ SHA1
  6. H.450.1 On: ☑
  7. Codec: ⊙ G.723 ○ G.729 ○ G.711u ○ G.711a
  8. Full RRQ: ☐

  [DONE]

H.323 Parameters:
- **Telephone Number**: insert telephone number of VIP-100PT when registering to Gatekeeper.
- **Password**: the authorized password in the gatekeeper application
- **H.323 ID**: setup H.323 ID provided by service provider, the H.323 ID is mostly used as username for authentication when the IP phone is enabled in GK mode.
- **GK mode**: Enable/disable Gatekeeper function.
- **GK IP**: Set Gatekeeper IP address or Domain Name.
- **GK Port**: communication port number of the Gatekeeper for VIP-100PT to send messages.

Forward Mode:

- **Immediate**: enable/disable **Immediate Forward** function.
- **Immediate Number**: setup the immediate forward number here.
- **Busy**: enable /disable **Busy Forward** function.
- **Busy Number**: setup the busy forward number here
- **No Answer**: enable /disable **No Answer Forward** function.
- **No Answer Number**: setup the No Answer Forward number here
- **No Answer Time:** setup the No Answer Forward number here

ⓘ Hint

It is suggested to check section **2.3.1** to **2.3.3** for more information of Call forward feature in machine

Advanced Setting:

- **RTP Port** (**Starting**): the starting port number used in the PoE IP phone voice transmission port.
- **RTP Prot** (**Ending**): the ending port number used in the PoE IP phone voice transmission port.
- **H.323 Fast Connect**: enable/disable the H.323 fast start mode
- **H245 Mode**: enable H.245 tunneling mode or open the H.245 channel after H.323 fast start connection
- **H235 Security:** select the authentication type for the H.235 security
- **H.450.1 On**: enable or disable the H.323 supplementary services in machine
- **Codec**: voice codec used to encode the voice packets
- **Full RRQ**: in the GK application, you may enable or disable sending full RRQ information when registering to the GK (Default: **No**)

3. SIP Configuration (VIP-150PT Only)

Please select the "**SIP Configuration**" icon on the left menu; the following page will display to allow you to set all the SIP related configurations.

SIP Configuration

- **SIP Parameters:**
  1. Display Name: [   ]
  2. Login ID: [   ]
  3. Telephone Number: [   ]
  4. Password: [   ]
  5. Proxy mode: ☐
  6. Proxy Server Address: [   ]
  7. Proxy Port: [5060]
  8. Outbound Proxy: ☐
  9. Outbound Proxy Address: [   ]
  10. Outbound Proxy Port: [   ]
  11. STUN Server: ☐
  12. STUN Address: [   ]
  13. STUN Server Port: [   ]

  [DONE]

- **Forward Mode:**
  1. Immediate: ☐
  2. Immediate Number: [   ]
  3. Busy: ☐
  4. Busy Number: [   ]
  5. No Answer: ☐
  6. No Answer Number: [   ]
  7. No Answer Time: [0]

  [DONE]

- **Advanced Setting:**
  1. Reg From: [   ]
  2. Reg To: [   ]
  3. Reg Expire: [60]
  4. RTP Port: [10000]
  5. Codec: ○ G.711u ○ G.711a ○ G.729 ◉ G.723

  [DONE]

SIP Parameters:
- **Display Name:** SIP identification
- **Login ID**: the username to register to Proxy Server. If Proxy server doesn't request specific username, please enter line number here.
- **Telephone Number**: the telephone number of VIP-150PT to register to SIP proxy.
- **Password**: the password used to register to SIP proxy. This parameter will not be necessary when SIP proxy doesn't request client to set password. In the circumstances, please setup the username same as telephone number for SIP registration.
- **Proxy mode**: enable/disable SIP proxy mode.
- **Proxy Server Address**: SIP proxy IP address or domain name.
- **Proxy Port**: setup proxy port for SIP proxy communication.
- **Outbound Proxy**: Enable/disable SIP outbound proxy function.
- **Outbound Proxy Address**: IP address / domain name.of the SIP outbound proxy
- **Outbound Proxy Port**: communication port number of the SIP outbound proxy
- **STUN server**: Enable/disable SIP STUN function.
- **STUN server Address**: IP address / domain name.of the SIP STUN server
- **STUN server Port**: communication port number of the SIP STUN server

Forward Mode:
- **Immediate**: Enable/disable immediate call forwarding
- **Immediate Number**: the immediate call forwarding number
- **Busy**: Enable/disable busy call forwarding
- **Busy Number**: the busy call forwarding number
- **No Answer**: Enable/disable no-answer call forwarding in machine
- **No Answer Number**: the no-answer call forwarding number
- **No Answer Time**: the waiting time to forward the call to specific number

---

ⓘ Note
- Settings in the SIP configuration menu are mostly application-oriented. Please be sure to check with the service provider to obtain proper information before setting up machine
- Please contact the service provider or VoIP system administrator to have the proper call forwarding settings/information.

Advanced Setting:

- **Reg From**: Modify the "From:" field in the SIP/SDP messages. Should left this empty unless for very specific system.
- **Reg To**: Modify the "To:" field in the SIP/SDP messages. Should left this empty unless for very specific system.
- **Reg Expire**: Modify the value "Expires:" field in the SIP/SDP messages. This will control the re-registration period.
- **RTP Port** (**Starting**): the starting port number used in the PoE IP phone voice transmission port..
- **RTP Port** (**Ending**): the ending port number used in the PoE IP phone voice transmission port.
- **Codec**: voice codec used to encode the voice packets. In SIP communications, the called party determines the type of voice codec used in conversation.

4. Phone Configuration

Please browse to the "**Phone configuration**" on the left menu, following page will be displayed for phone call and voice codec related configurations.

Phone Configuration

- **Prefix Setting:**
  1. International Code:
  2. Country Code:
  3. Area Prefix Code:

    [DONE]

- **Voice:**
  1. Ring Volume(0-9): 2
  2. Handset Volume(0-9): 5
  3. Handfree Volume(0-9): 3
  4. CodecTxGain(0-9): 0
  5. Ring Type: ● Type 1 ○ Type 2 ○ Type 3 ○ Type 4 ○ Type 5 ○ Type 6 ○ Type 7 ○ Type 8 ○ Type 9 ○ Type 10
  6. RTPLowBW: ☑
  7. Jitter Buffer: ☐

    [DONE]

Prefix Setting:

This parameter is used to setup the calling prefix string.

For some specific VoIP system, the IP Phone can call to PSTN/mobile number and the called number must include the country code, e.g. the country code of Taiwan is 886, so

if want to call a Taipei city call 02-12345678, the called number needed to be 886-2-12345678. But this is not consistent with the normal dialing behavior. In order to use the normal dialing behavior, you can set the "My Country Code" value to 886, and set the "Area Prefix Code" value to 0.

As for the "International Code", when you want to dial an international call, the International Code will be removed. E.g., suppose want to make a call to China, the dialed number will be 002-86-2112341234, the IP Phone will remove the "International Code" 002, and send the real number of 86-2112341234.
This prefix code implementation is only needed for some specific VoIP system.
Normally, this is not needed, just let all the prefix values be empty.

Voice:

- **Ring Volume (0-9)**: select the volume of ringing.
- **Handset Volume (0-9)**: select the telephone set volume
- **Handfree Volume (0-9)**: select the volume of handfree speaking.
- **CodecTxGain(0-9)**: select the output voice volume.
- **Ring Type**: select the ring type (ring pattern) of an incoming call.
- **RTPLowBW**: this setting is to decrease the voice packet bandwidth of the transmit direction in order to improve the voice quality of transmit direction if the Internet bandwidth is not good enough.
- **Jitter Buffer**: increase the jitter buffer of receiving voice packet. (from 150ms to 400ms.) When in some specific Internet connection or slow network, the traffic is arriving is with large jitter, i.e. with bad voice quality, you can improve the voice quality via fine tuning the "Jitter Buffering"

Others:

- **VAD**: VAD is short for Voice Activity Detection, the IP Phone will detect if the user is talking or not, and avoid to send voice packet when the user is silent to decrease the bandwidth requirement.
- **BG Noise Level (0-9)**: signal level of back ground noise. This parameter controls the sensitivity of the VAD detection.

5. System Configuration

Please select the "**System Configuration**" on the left menu; you may set up the system related configurations.
In this page, you can perform machine software upgrade, change username/password of the web login, and reboot (restart) machine.

System Configuration

● **Firmware Upgrade:**

  1. FTP Server: [   ]
  2. Image File Name: [   ] [DONE]

     [Upgrade Now]

● **Configuration Backup / Restore:**

  1. FTP Server IP Address: [   ]
  2. Username: [admin   ]
  3. Password: [123   ]
  4. Image File Name: [   ] [DONE]

     [Upload to Server Now]

     [Download from Server Now]

● **Change web Username & Password:**

  1. Username: [root   ]
  2. Password: [   ]
  3. Confirm New Password: [   ]

     [DONE]

● [REBOOT]

Firmware Upgrade:
- **FTP Server**: IP address of the FTP server, which contains the firmware file of the PoE IP phone
- **Image File Name**: firmware file name of VIP-100PT/VIP-150PT.

Configuration Backup/Restore:
- **FTP Server IP Address**: IP address of the FTP server for system configuration backup/restore
- **Username**: **Set** login username for the FTP server
- **Password**: **Set** login password for the FTP server
- **Image File Name**: system configuration file for machine backup/restore

Configuration Backup/Restore: The default username/password for web login is

root / **<no password>**, you may modify the username/password for more security. Please enter the new login name, re-enter the current password, new password and confirm new password to enable new password.

Reboot Button: Press the "**REBOOT**" button will reset the PoE IP Phone

---

①Hint

- The PoE IP phones are optimized for VoIP applications, it will not require firmware upgrade in most cases. It is suggested to check with our local distributor for machine firmware upgrade information.
- While performing firmware upgrade, please allocate the firmware file in the FTP root directory, so that the IP phone can find the firmware file to complete the upgrade process.
- When pressing the "**REBOOT**" button, all the modified data will automatically be saved in machine, and activated in the next power-on.
- If you'd like to load the factory default value into machine, please press "**#*110*** " on the keypad to reset the IP phone.


Chapter 4

Voice communication samples

There are several ways to make calls to desired destination in VIP-100PT/VIP-150PT. In this chapter, we'll lead you step by step to establish your first voice communication via web browsers operations.

Peer to peer (P2P) mode
VIP-100PT to VIP-100PT connection via IP address

Assuming two VIP-100PT are installed in the same network segment, the IP addresses are 192.168.0.1, 192.168.0.2
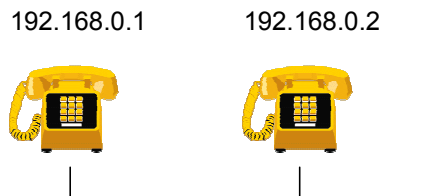
192.168.0.1   192.168.0.2

In peer-to-peer mode, you can direct pickup the handset and dial the IP address of the remote VIP-100PT using the keypad as below:

| 1 | 9 | 2 | * | 1 | 6 | 8 | * | 0 | 0 | 0 | * | 2 | # |

Then you may pick up the ringing VIP-100PT to start conversation. You may repeat the same steps to make calls from another VIP-100PT.

VIP-150PT to VIP-150PT connection via IP address

Assuming two VIP-150PT are installed in the same network segment, the IP addresses are 192.168.0.1, 192.168.0.2
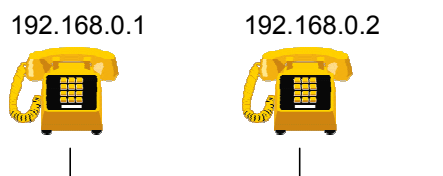
192.168.0.1   192.168.0.2

In peer-to-peer mode, you can direct pickup the handset and dial the number using the keypad as below:

| 1 | 9 | 2 | * | 1 | 6 | 8 | * | 0 | 0 | 0 | * | 2 | # |

Then you may pick up the ringing VIP-150PT to start conversation. You may repeat the same steps to make calls from another VIP-100PT.

ⓘ Hint

- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with an "**#**".
- If the IP phones are installed behind a NAT/firewall/ IP sharing device, please make sure the NAT device support SIP applications before making calls

VIP-150PT to VIP-150PT connection via SIP telephone number

VIP-150PT
WAN IP Address
(192.168.0.1)

Number: 100

Number: 200

VIP-150PT
WAN IP Address
(192.168.0.2)

SETP 1:

Please browse machine "**SIP configuration"** menu, and enable the **Proxy mode**.

Insert IP address of the remote calling party in the "**Proxy Server Address**" field.

Sample configuration screen is shown below:

- **SIP Parameters:**
  1. Display Name: 100
  2. Login ID:
  3. Telephone Number: 100
  4. Password:
  5. Proxy mode: ☑
  6. Proxy Server Address: 192.168.0.2
  7. Proxy Port: 5060
  8. Outbound Proxy: ☐
  9. Outbound Proxy Address:
  10. Outbound Proxy Port:

  DONE

SETP 2:

After these configurations, be sure to click the "**DONE**" button to apply settings and browse to

"**System Configuration**" menu to reboot the machine to make the settings effective After

rebooting, the unit will communicate with the remote calling party, the LCD screen will show

below:

Registering (100)
Date Time
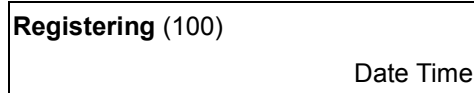
SETP 3:

At this moment, you may pick up the handset and dial "200" to connect the remote VIP-150PT to

start the voice communications.

Appendix A

PoE IP Phone Specifications

Product	H.323 PoE IP Phone	SIP PoE IP Phone
Model	VIP-100PT	VIP-150PT
Hardware		
WAN	1 x 10/100Mbps RJ-45 port (Power Over Ethernet 802.3af function)	
PC	1 x 10/100Mbps RJ-45 port	
LCD display	2 x 16 characters	
Speaker	8 Ohm/0.2 Watt speaker for speakerphone operation	
Protocols and Standard		
Standard	H.323 version 4,H.323 Fast start, H.245 Tunneling and H.245 DTMF relay	SIP 2.0 (RFC3261), SIP digest authentication (MD5),
Voice codec	G.723.1 (6.3k/5.3k), G.729ab, G.711 (a-law/u-law)	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Dynamic Jitter Buffer	
Supplementary services	Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding Call hold/transferring.	
Call history	Record incoming call Outgoing call Missed (not accepted) call history	
Protocols	RFC-3261, H.323, TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE	
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Management	Web, Keypad	
Dimension (W x D x H)	170 mm x 220 mm x 60 mm	
Operating Environment	0~40 degree C, 10~95% humidity	
Power Requirement	12V DC	
EMC/EMI	CE, FCC Class B	