



# **SIP IP Phone**

**VIP-154T/VIP-154PT/VIP-154NT** 

# User's manual

Version 2.2

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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.

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#### Revision

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# Chapter 1 Introduction

#### **Overview**

Meeting the next-generation Internet telephony service demands, PLANET Technology provides feature-rich, toll-quality Internet telephony service solutions. The built-in PSTN interface provides user more convenience between IP Phone and PSTN call selections. —VIP-154NT. With 802.3af Power over Ethernet (PoE) IP Phone—VIP-154PT. And the VIP-154T is the cost-effective SIP IP Phone; the VIP-154 series are SIP 2.0 (RFC3261) compliant with SIP digest authentication supports.

The VIP-154T / VIP-154PT / VIP-154NT ("IP Phone" in the following term) features high-quality speakerphone technology, and includes an easy-to-use speaker on/off button and call hold/transfer buttons for various voice services.

The IP Phone has additional features such as built-in PPPoE/DHCP clients, password-protected machine management, LCD menu display, speed-dial 3-way conference keys, hands-free speakerphone, last number redial, incoming message indicator, and user-intuitive web administration system.

The IP Phone is self-contained, service-integrated, intelligent phone features offering, and powerful voice processing power. The IP Phone can effortlessly deliver toll voice quality equivalent to the regular PSTN connections utilizing cutting-edge Quality of Service, 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 desktop PC network connections. When installing the VIP-154T / VIP-154PT / VIP-154NT, SIP IP Phone with IPX-2000 (PLANET IP PBX system), the VIP-154 series IP phones can be easily integrated in your office; via the auto-config support for IPX-2000. No expertise required building up the VoIP network deployment. Besides, the IP Phones are ideal 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 benefit from the VoIP technologies in your daily life.

There are models for VIP-154T/VIP-154PT/VIP-154NT and there are:

VIP-154T: SIP IP Phone

VIP-154PT: 802.3af PoE SIP IP Phone

VIP-154NT: 1 FXO SIP IP Phone

#### **Product Features**

• Built-in PSTN (VIP-154NT)

The built-in PSTN interface provides user more convenience between IP Phone and PSTN call selections easily

• Simple Installation and administration

Configuration of the **IP Phone** can be performed in minutes via the LCD menu keypad, telnet, or web interfaces. Using the built-in LCD display, the **IP Phone** offers user-friendly configuration guidelines, machine operation status, call status displays, and incoming call identification.

#### IP PBX system integration

Via *auto-config* support for IPX-2000, no expertise required to establish your office voice network. VIP-154 series can help you to complete VoIP network deployment in minutes.

#### Feature-rich keypad IP Phone

The **IP Phone** integrates a high-quality speakerphone with the Call Hold, Forward, Transfer and Waiting functions and also provides advanced telephone features, such as 4 speed-dial keys, 3-way conference key, 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 **IP Phone** can act as a PPPoE/DHCP client, automatically obtaining an IP address for Internet access.

#### Various field applications compliant

The **IP Phone** is capable of handling peer-to-peer and SIP proxy / IP PBX registration, authentication to interact with major IP PBX/SIP gateway/IP Phone in the market. The IP Phone 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-154T / VIP-154PT / VIP-154NT 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.711u-law/a-law; Voice Activity Detection (VAD), and the Confort Noise Generation (CNG).

#### **VoIP Features**

- SIP 2.0 (RFC3261) compliant
- Peer-to-Peer / SIP proxy calls
- Voice codec support: G.711, G.723.1, G.726, G.729A, G.729B
- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation
- In band and out-of-band DTMF support

#### Package Content

The contents of your product should contain the following items:

VoIP IP Phone

Power adapter

Quick Installation Guide

User's Manual CD

RJ-11 cable x 1

#### **Physical Details**

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

#### **Rear View**



Rear Panel of VIP-154T



**Rear Panel of VIP-154PT** 



#### **Rear Panel of VIP-154NT**

1	RESET	Reset to the factory default setting
2	PC	RJ-45 connector, to maintain the existing network structure, connected directly to the <b>PC</b> through <b>straight</b> CAT-5 cable
3	LAN	RJ-45 connector, for Internet access, connected directly to <b>Switch/Hub</b> through <b>straight</b> CAT-5 cable.  The <b>LAN</b> interface also can be connected with 802.3af PoE switch or converter for power supply ( <b>VIP-154PT</b> )
4	12V DC	12V DC Power input outlet
5	LINE	FXO interface, for connect with PSTN line. Press <b>0</b> * to switch to PSTN mode. ( <b>VIP-154NT only</b> )

#### (i) Note

- IP Phone default IP is <a href="http://192.168.0.1">http://192.168.0.1</a>. Press RESET button on rear panel over 5 seconds will reset the VoIP IP Phone to factory default value. (Except speed dial and call forward settings)
- 2. For VIP-154PT, either PoE or AC adapter can be deployed at one time  $\,$

#### Front View and Keypad function



#### **Keypad Description**

1	LCD Display	Menu and all status shall be displayed for users.
_	Speed Dial	To make a speed dial call by pressing the speed dial key M1 ~
2	M1~M4	M4.
3	MENU	To bring out the menu selection while IP Phone is in idle state.
4	Vol +/ ↑	To increase the volume of voice when at off-hooked state.
•	4 (0) +/	To page up menu when at configuration mode.
5	Vol - / ↓	To decrease the volume of voice when at off-hooked state.
·	7 C. 7 V	To page down menu when at configuration mode.
6	ОК	To be used as confirm configuration or enter sub-menu.
7	Phone Book	Enter the phone book selection.
8	MESSAGE	Press this button can enter the voicemail service.

9	TRANSFER	To transfer an active call (incoming call answered or outgoing call accepted) to another devices.
10	CONF	Press this button can make conference function.
11	FWD	To carry out forward function.
12	DEL/MUTE	Press to delete digits when at configuration mode or input phone numbers.  Press to mute sounds when at talk mode.
13	Redial	Press to dial the last dialed number when the IP Phone is off-hooked.
14	Handfree	To switch between the usage of the handset and the speaker devices.
15	Hold	To hold the conversation.
16	Call Log	Show the calls history.

# Chapter 2 Preparations & Installation

#### **Physical Installation Requirement**

This chapter illustrates basic installation of IP Phone

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem

#### **Administration Interface**

PLANET IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration.

#### 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 4.01 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 <a href="http://192.168.0.1">http://192.168.0.1</a> 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: **root** / **null** (no password) to continue machine administration.

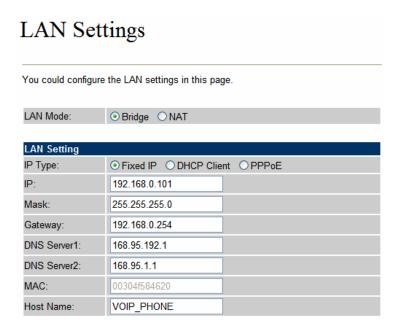
(i) Note

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.

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#### LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default: **root / no passwrd**), browse to "**Network**" --> "**LAN Settings**" configuration menu:



#### **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 browse to "**Save & Reboot**" menu to reboot the machine to make the settings effective.

Connection Type	Data required.
Fixed IP	In most circumstances, it is no need to configure the DHCP
I IXEU II	settings.
DHCP clinet	The ISP will assign IP Address, and related information.
DDD - F	The ISP will assign PPPoE username / password for Internet
PPPoE	access,



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.

#### Save Modification to Flash Memory

Most of the IP Phone parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the IP PHone, to save the parameters into Flash ROM and let it take effective forever, please remember to press the **Save & Reboot** button after you modify the parameters.

# Save & Reboot You have to save changes to effect them. Save Changes: Save

#### 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.

#### Overview on the web interface of IP Phone

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

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

Microsoft Internet Explorer 4.01 or higher with Java support

#### 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 <a href="http://192.168.0.1">http://192.168.0.1</a> to logon IP Phone web configuration page.

IP Phone will prompt for logon username/password: root / null (not password)



IP Phone log in page

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



## System Information

This page illustrate the system related information. PLANET Technology Corp. Company: Firmware Version: VIP-154T/PT V2.2 Codec Version: 11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C Contact Address: Tel: 886-2-22199518 888-2-22199528 Fax: support\_voip@planet.com.tw E-Mail: Web Site: www.planet.com.tw

IP Phone main page

# Chapter 4

# **VoIP Telephone Adapter Configurations**

#### **Phone Book settings**

IP Phone can set up 140 records of Phone Book. User can make calls via **Phone Book** feature of IP Phone.

#### Phone Book You could add/delete items in current phone book. Phone Book Page: page 1 Name URL 1 2 3 4 Delete Selected Delete All Reset Add New Phone Position: (0~139) Name: URL: Add Phone Reset

Field	Description	
Phone Book Bogo	The default is Page 1. It can select Page1 ~ Page 14	
Phone Book Page	to look round Phone Book records.	
Phone	The record number from 0 ~ 139, it can set up 140	
Phone	records in total.	
Name	The name of Phone Book records, it only can input	
Name	numerals.	
URL	Fill in the outgoing number (Line Number) or IP	
UKL	address.	
Select	To select this record.	

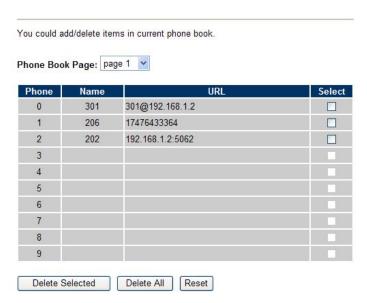
If you need to add a phone number into the Phone Book list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "Add Phone" button.

If you want to delete a phone number, you can select the phone number you want to delete then click "Delete Selected" button.

If you want to delete all phone numbers, you can click "Delete All" button.

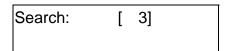
#### For Example:

#### Phone Book



#### STEP 1:

IP Phone had added the above phone numbers. User press **Phone Book** button from keypad then the LCD screen will show below:



#### STEP 2:

Press OK button to enter the Phone Book menu. The LCD screen will show the Phone Book records pervious made.

00	202	
01	206	

#### STEP 3:

Selecting the recorder you want to dial and press OK button. It sill show the detail information as below:

202 192.168.1.2:5062

#### STEP 4:

Pick up the telephone handset or press Handfree button to dial to this telephone.

IP Dialing.. 1 192.168.1.2:5062

#### **Speed Dial settings**

In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) and follow the "#" key.

If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "Add Phone" button.

If you want to delete a phone number, you can select the phone number you want to delete then click "Delete Selected" button.

If you want to delete all phone numbers, you can click "Delete All" button.

#### Speed Dial Phone List

#### **Call Forward**

This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

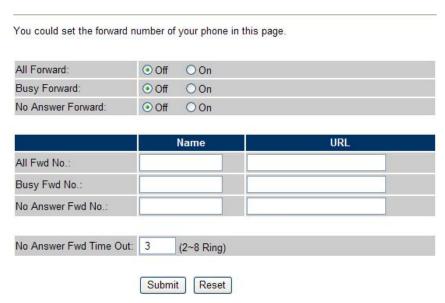
**All Forward**: All incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

**Busy Forward**: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

**No Answer Forward**: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

When you finished the setting, please click the Submit button.

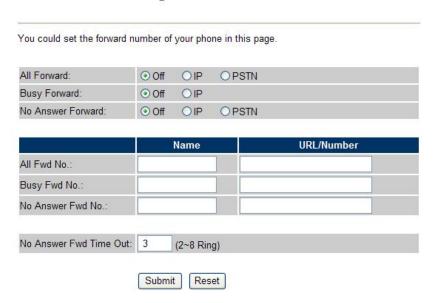
#### Forward Settings



Call Forward function for VIP-154T/VIP-154PT

**Call Forward to PSTN (VIP-154NT):** VIP-154NT not only support Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, then input the name and the PSTN number in URL/Number field.

#### Forward Settings



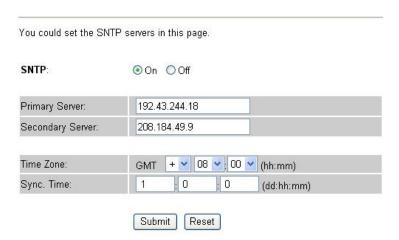
Call Forward function for VIP-154NT

#### **SNTP** settings

This page defines the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again.

When you finished the setting, please click the Submit button.

#### **SNTP Settings**



#### **Volume Setting**

This page defines the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

Handset Volume is to set the volume for you can hear from the handset.(Handfree mode)

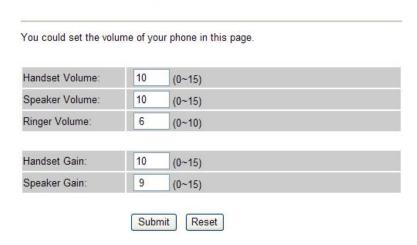
**Speaker Volume** is to set the volume for you can hear from the speaker.

Ringer Volume is to set the ringer volume for you can hear.

Handset Gain is to set the volume send out to the other side's handset.

**Speaker Gain** is to set the volume send out to the other side's handset from the microphone. (Handfree mode)

#### Volume Settings



Volume Settings for VIP-154T/VIP-154PT

Beside the above settings, VIP-154NT also can set the volume of PSTN.

PSTN-Out Volume is to set the volume for you can hear from the PSTN line.

**PSTN-In Gain** is to set the volume send out to the other PSTN side's haneset.

#### Volume Settings

Handset Volume:	10	(0~15)
Speaker Volume:	10	(0~15)
	6	
Ringer Volume:		(0~10)
PSTN-Out Volume:	10	(0~12)
Handset Gain:	10	
nanoset Gain.	10	(0~15)
Speaker Gain:	9	(0~15)
PSTN-In Gain:	10	(0~15)

**Volume Settings for VIP-154NT** 

#### **Ringer Setting**

This page defines the user can set the tinkle of bells when someone ring your IP Phone. If want to set ringer, it need to enable Ringer function and select the Ringer Type you wanted. There are four Ringer Types can be chosen. When you finished the setting, please click the Submit button.

## Ringer Settings

You could set yo	our favorite ringer in this page.
Ringer:	○On
Ringer Type:	ringer 1
	Submit Reset

#### **Block Setting**

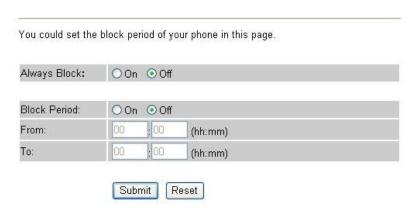
This page defines the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

Always Block: All incoming call will be blocked until disable this feature.

Block Period: Set a time period and the phone will be blocked during the time period. If the "From"

time is large than the "To" time, the Block time will from Day 1 to Day 2. When you finished the setting, please click the Submit button.

#### **Block Setting**

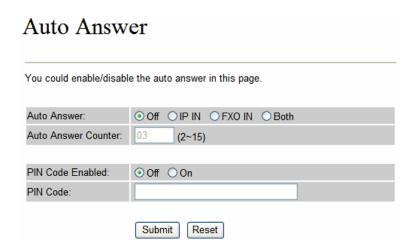


#### **Auto Answer settings (For VIP-154NT)**

This page defines the Auto Answer function. You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the VIP-154NT can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the VIP-154NT can let user to redial to IP Phone number.

**Auto Answer Counter** is to set after the ring count met the number you set then the auto answer will enable.

For security issue, You'd better to set the PIN Code. If you have set the PIN code, you will hear a tone to inform you input the PIN Code then you can dial out.



#### **Dial Plan Settings**

This page defines the Dial Plan Setting function. This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

# Dial Plan Settings

You could set the dial plan in this page.

Drop prefix :	O Yes	⊙ No		
Replace rule 1:	002	+	1234+4321	
Drop prefix :	⊙ Yes (	ONo		
Replace rule 2:	006	+	002+003+004	
Drop prefix :	OYes (	⊙ No		
Replace rule 3:	007	+	5xxx+35xx	
Drop prefix :	O Yes	⊙ No		
Replace rule 4:		+		
Dial now:	*xx+#xx+	11x+x	XXXXX	
Auto Dial Time:	5 (3	3~9 se	c)	
Use # as send key:	⊙ Yes (	○No		
	⊙ Yes (	ONo		

Field	Description			
Drop Prefix	The rule of add or replace code. If setup as No, it will add the prefix			
	number prior to the identification number. If setup as Yes, it will			
	replace the identification number.			
Replace rule	The prefix number. It only accept the numeral and the max length is 8.			
+	The identification number. It can accept the numeral or symbol and			
	the max length is 40.			
	■ Symbol: It only accept the [+], [x]			
	■ +: It means as "or". For example, [123+456+334+5xx] even if			
	[123 or 456 or 334 or 5xx]			
	■ x: It is equal to 0~9. For example, [5xx] even if the number			
	begin 5.			
Dial Now	If the dialing number are match with this field, it will dial out and need			
	not to press the "#" key to end the dialing. It accepts the numeral or			
	symbol, and the max length are 124.			
	<b>①Note</b> : The starting number can't be the "0". For example, if the			
	number is "0xxxx", because the starting number is "0", so that the			
	system will ignore this dial plan.			
Auto Dial Time	Stop dialing after seconds then send dial number out.			
Use # as send key	If setup as Yes, the symtem sill stop to receive the dialing number			

	when receive the [#] key. The system also will to determine the Auto
	Dial Time, it will carry out the calling if there isn't receive the digit after
	the Auto Dial Time.
	If setup as No, the system just according to the Auto Dial Time to
	determine the end time.
Use * for IP dialing	If setup as Yes, the system will look on [*] as [.]. For example, if dial
	the "192*168*0*100#", it will dial out as "192.168.0.100#".
	If setup as No, it just look on [*] as [*]. For example, if dial the "700*#",
	it will dial out as "700*#".

#### **Descriptions of example:**

Example\_1: Drop prefix: No, Replace rule 1: 002, +: 1234+4321 (No limit the digit length)

- 1. If the dialing number is start as "1234", it will add the 002 at begin. The real dialing number is [0021234...].
- 2. If the dialing number is start as "4321", it will add the 002 at begin. The real dialing number is [0024321...].

Example\_2: Drop prefix: Yes, Replace rule 2: 006, +: 002+003+004 (No limit the digit length)

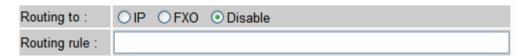
- 1. If the dialing number is start as "**002**", it will replace 002 by 006. The real dialing number is [**006**...].
- 2. If the dialing number is start as "**003**", it will replace 003 by 006. The real dialing number is [**003**...].

Example\_3: Drop prefix: No, Replace rule 3: 007, +: 5xxx+35xx (Has limit the digit length)

- 1. If the dialing number start as "5" and follow 3 digits, it will add the 007 at begin. The real dialing number is [0075xxx].
- 2. If the dialing number start as "**35**" and follow 2 digits, it will add the 007 at begin. The real dialing number is [**00735xx**].

#### Example\_4: Dial Now: \*xx+#xx+11x+xxxxxx

- 1. If the dialing number is match with the rule of "\*xx", it will send out the dialing number directly. For example, \*00/ \*01/ \*02...\*99.
- 2. If the dialing number is match with the rule of "**#xx**", it will send out the dialing number directly. For example, **#00/#01/#02...#99**.
- 3. If the dialing number is match with the rule of "11x", it will send out the dialing number directly. For example, 111/112/113...119.
- 4. If the dialing number is match with the rule of 8 digits, it will send out the dialing number directly. For example, 12345678.



**Routing Settings for VIP-154NT** 

**Routing to:** It can select IP or FXO for the auto routing function. Machine will according to the **Routing** rule to dial out by IP or FXO method.

**Routing rule:** The **Routing rule** feature has drop number function, and it can use "+" to separate for many routing rules.

#### Example: Routing rule: D007+009

- 1. When the dialing numbers are start as "007", for example **00712345678**. It will drop 007 then number become **12345678**, and according to **Routing to** setting to select the routing way.
- When the dialing number is start as "007", for example 00987654321. It needs not drop number, and according to Routing to select the routing way.



If enable the **Routing** function and the dialing number is match with **Routing rule**, machine will carry out the Routing function and to skip over the below **Drop prefix** and **Replace rule** functions.

#### Flash Time Setting (For VIP-154NT)

When you use the VIP-154NT and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function.

#### Flash Time Settings

You could set the	flash time in t	his page.	
Flash Time:	65	(Range:1~200, Unit:10ms)	
	Subr	mit Reset	

#### **Call waiting Settings**

When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear any notice.

#### Call Waiting Settings

You could enable/disable the call waiting setting in this page.

Call Waiting: 

On 
Off

Submit Reset

#### **Voice Mail Settings**

This page defines the voice mail key function. When device register to IP PBX and it support Voice Mail System. It can set up the voice mail number in advanced, and press the "MESSAGE" button from keypad. It will enter for voice mail system.



#### **Hot line Settings**

This page defines the Hot line setting in this page. When user pick up the handset, the device will call to the specific number automatically.

Use Hot Line: Click Enable to carry the Hot line function out.

Hot line number: The hot line number, it can input the IP address or registration number.

#### Hot line Setting

You could set the	hot line in this page.	
Use Hot Line :	○ Enable	
Hot line number:		
	Submit Reset	

#### **Alarm Settings**

This page defines the Alarm setting in this page. It provides the alarm function, and it can set up the Alarm Time to get the telephone ringed up every day.

**Alarm:** The default is Off. If set up as On, the telephone will ringed up at the specific time.

Alarm Time: It can set up the system prompt time with 24 hours.

Current time: The next alarm time.

#### Alarm Settings

You could set the a	alarm time in this page.
Alarm:	○ ON ④ OFF
Alarm Time:	0 (hh:mm)
Current time:	2005-01-01 08:34
	Submit Reset

#### **LAN Settings**

This page defines the LAN setting in this page.

LAN Mode: The default is Bridge mode, and it also provides NAT mode.

- **Bridge:** When set as is mode, the LAN and PC ports are in the same network segment.
- NAT: The LAN and PC ports are in the different network segment, and PC port could enable the DHCP Server function to allot the IP address.

IP Type: The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes.

- Fixed IP: It could setup the IP address manual.
- DHCP Client: It will acquire the IP address automatically.
- PPPoE: It will use the PPPoE connection method.

IP: The IP address

Mask: The sub net address

Gateway: The default gateway address

**DNS Server1:** The default is 168.95.192.1, it could setup the first DNS server address.

**DNS Server2:** The default is 168.95.1.1, it could setup the second DNS server address.

MAC: The MAC of LAN port

**Host Name:** The product model

**User Name:** The PPPoE connection account name. It could inpout numeral or character, the maximum date length are 63.

**Password:** The PPPoE connection account password. It could inpout numeral or character, the maximum date length are 63.

LAN Se	ttings
You could configu	ure the LAN settings in this page.
LAN Mode:	⊙ Bridge ○ NAT
LAN Setting	
IP Type:	● Fixed IP ○ DHCP Client ○ PPPoE
IP:	192.168.0.1
Mask:	255.255.255.0
Gateway:	192.168.0.254
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	00304f584620
Host Name:	VOIP_PHONE
PPPoE Setting	
User Name:	
Password:	
Service Name:	
	Submit Reset

#### **PC Settings**

This page defines the PC setting in this page.

IP: The IP address of PC port. (In the Bridge mode, the Default IP: 192.168.0.1)

Mask: The sub net address. (Default: 255.255.255.0)

MAC: The MAC of PC port

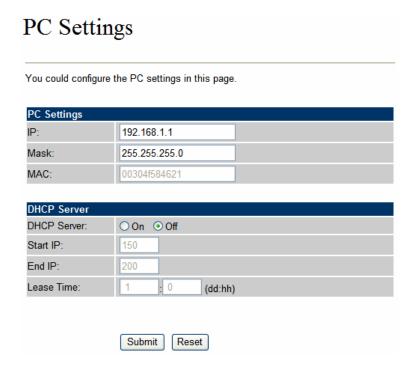
**DHCP Server:** It will allot the IP address automatically when enable this function.

Start IP: Start IP of lease table

End IP: End IP of lease table. Network device connecting to the PC port can dynamic obtain the IP in

the range between start IP and end IP

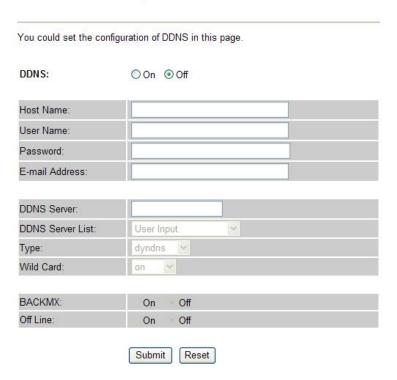
Lease Time: DHCP server lease time



#### **DDNS Settings**

This page defines the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.

#### **DDNS** Settings



#### **VLAN Settings**

This page defines the VLAN setting in this page. This function needs to co-operate with network devices which have VLAN function.

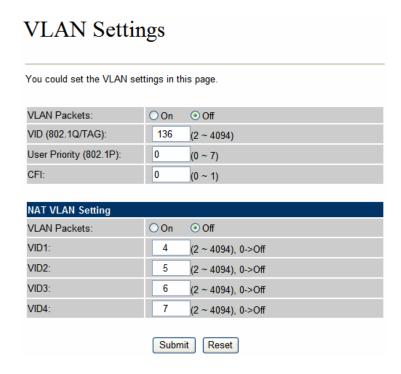
VLAN Packets: If setup as On, it could receive VLAN messages.

VID (802.1Q/TAG): Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 2~4094.

User Priority (802.1P): To setup the user priority.

**CFI:** To indicate the Canonical Format.

- If CFI=1, it means the header label include RIF field, and the NCIF flag valus of RIF will to decide the MAC address is Canonical Format or Non-Canonical Format in frame information.
- If CFI=0, it means the header label does not include RIF field, and the MAC address is Canonical Format in frame information.



#### **DMZ Settings**

This page defines the DMZ setting in this page.

**DMZ:** If setup as On, all of packets (expect SIP packets) will send to the specific IP address.

DMZ Host IP: The DMZ host IP address.



#### **Virtual Server**

This page defines the Virtual Server setting in this page. You could define 24 virtual service information in this page. When you finished the setting, please click the Submit button.

Virtual Server Page: There are total page1 to page 3. It could choose the page which want to go over.

Num: The serial number. There are total 24 records from Num 0 to 23.

Enable: The activate status. The default is Disable, this record will been activate if enable.

**Protocol:** The TCP or UDP communication protocol.

**Internal Port:** For corresponding the internal port.

**External Port:** For corresponding the external port.

Server IP: To input the Server IP address.

### Virtual Server Settings You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23]. Virtual Server Page: page 1 💌 Enable Enable Selected Delete Selected Delete All Reset **Add Virtual Server** Num: (0~23) Server IP: Protocol: TCP V Internal Port: External Port: Add Server Reset

#### **PPTP Settings**

This page defines the PPTP setting in this page. You could setup the PPTP Server connection information. When you finished the setting, please click the Submit button.

PPTP Settin	ngs
You could set the PPTP	server in this page.
PPTP:	○ On
PPTP Server:	
PPTP Username:	
PPTP Password:	
	Submit Reset

#### **Service Domain Settings**

This router comes with the built-in firewall based on the advanced technology of Stateful Packet In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the Phon. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

Display Name: you can input the name you want to display.

User Name: you need to input the User Name get from your ISP.

Register Name: you need to input the Register Name get from your ISP.

Register Password: you need to input the Register Password get from your ISP.

Domain Server: you need to input the Domain Server get from your ISP.

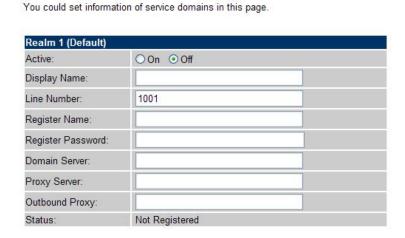
Proxy Server: you need to input the Proxy Server get from your ISP.

**Outbound Proxy**: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

You can see the Register Status in the Status item. If the item shows "Registered", then your Phone Adapter is registered to the ISP, you can make a phone call directly.

If you have more than one SIP account, you can following the steps to register to the other ISP. When you finished the setting, please click the Submit button.

# Service Domain Settings



#### ①Note:

IP Phone can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input the switch code.

#### Realm switch code:

1\*: Realm 1

2\*: Realm 2

3\*: Realm 3

For example: The default is realm 1, input the **2**\* (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

#### **Port Settings**

This page defines the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

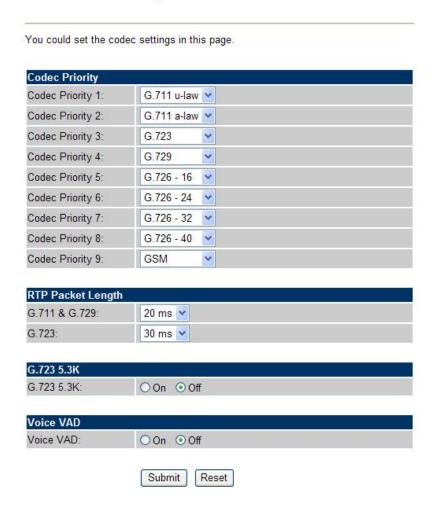
#### Port Settings

SIP Port:	5060	(1024~65535)
RTP Port:	60000	(1024~65535)

#### **Codec Settings**

This page defines the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

# Codec Settings



#### **Codec ID Setting**

This page defines the Codec ID. Sometimes 2 VoIP device with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use then you can change your Codec ID. When you finished the setting, please click the Submit button.

# Codec ID Settings

You could set the value of Codec ID in this page. Codec Type ID Default Value G726-16 ID: 23 (95~255) ☑ 23 G726-24 ID: 22 (95~255) ☑ 22 G726-32 ID: 2 ✓ 2  $(95 \sim 255)$ G726-40 ID: 21 ☑ 21 (95~255) RFC 2833 ID: 101 ☑ 101 (95~255) Submit Reset

#### **DTMF Settings**

This page defines the DTMF parameters. Yyou can setup the InBand DTMF, 2833 Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

# **DTMF** Settings

You could set the DTMF Settings in this page.	
<b>⊙</b> 2833	
O Inband DTMF	
O Send DTMF SIP Info	

#### **RPort Settings**

This page defines the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

# **RPort Settings**

You could en	able/disable the RPort setting in this page.
RPort:	○ On
	Submit Reset

#### **Other Settings**

This page defines the Hold by RFC, Voice/SIP QoS and other settings in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

**Hold by RFC:** The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.

Voice QoS (Diff-Serv): The Voice QoS feature.

SIP QoS (Diff-Serv): The SIP QoS feature.

The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the

others Internet devices.

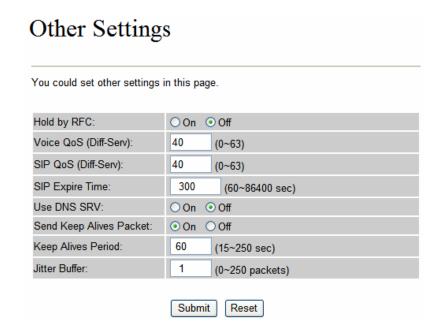
**SIP Expire Time:** To setup the registration interval time.

**Use DNS SRV:** The default is disable, and use DNS SRV mode. Set enable to use DNS to SRV mode to search the host information.

**Send Keep Alives Pcaket:** To deliver the packets on a regular time schedule to keep NAT port could open continued.

Keep Alives Period: To setup the schedule time for delivering the packets.

**Jitter Buffer:** To setup the Jitter Buffer size, and the unit is packet. It needs to refer to the Frame size of Codec.



#### **STUN settings**

This page defines the STUN Enable/Disable and STUN Server IP address in this page. This function can help your Phone Adapter working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

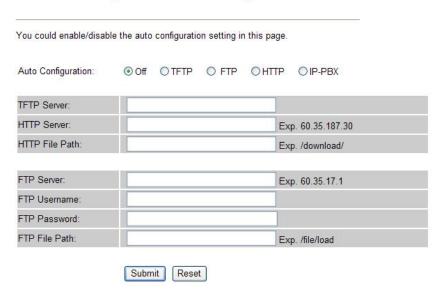
# STUN Settings

You could set the IP of STUN server in this page.  STUN:  On ⊙ Off  STUN Server:  STUN Port:  5060 (1024~65535)
STUN Server:
STUN Port: 5060 (1024~65535)

#### **Auto Configuration**

This page defines the Auto Configuration (Auto Provision) setting. IP Phone supports TFTP, FTP, HTTP and IP PBX auto configuration function in total. In IP PBX Auto Configuration Setting you need to check with your service provider if they have provided this function. Usually this function will be boundle with an IP PBX to use in the office.

# Auto Configuration Settings



#### PTT Settings (For VIP-154NT)

This page defines the PTT settings. When VIP-154NT connected to different country's PSTN Line, you have to set the country's setting to meet the requirement. When you finished the setting, please click the Submit button.

# **PTT Settings**



#### Cancel without to tag

This function can decide the device if send the cancel tag to Proxy Server. If there has the similar symptom that the caller cancel the call before the called answer the call, the called still continue to ring up even the caller has cancel this call. It could try select this function to **Yes** to avoid the above

symptom.

## Cancel without to tag



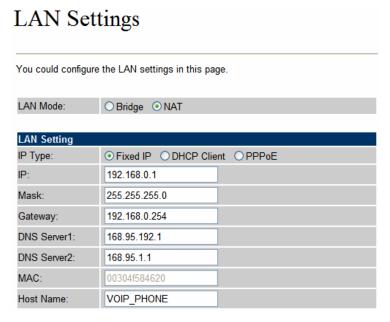
#### **MAC Clone Setting**

This page defines the MAC Clone Enable/Disable. This function will copy the MAC address from NIC (Network Interface Card) which placed in PC to LAN port of IP Phone. That because some ISP will limit the MAC address for PPPoE dial-up connection.

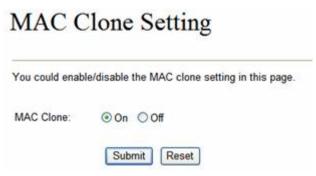


Please refer to the following operate procedures for more understandings to carry out the MAC Clone function.

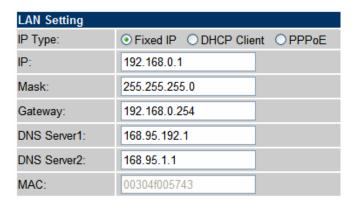
 Please login IP Phone and browse to "Network -> LAN Settings" page. To switch the LAN mode to NAT mode then press Save&Reboot button to save the settings and reboot machine.



- 2. Please make sure the network cable of your PC directly connect with PC port of IP Phone, then re-login IP Phone. (In the **NAT** mode, the default IP address of PC port is <a href="http://192.168.123.1">http://192.168.123.1</a>)
- 3. Please browse to "Advanced Settings -> MAC Clone Setting" page and enable the MAC Clone function.



- 4. IP Phone will prompt if sure want to clone the MAC of your PC to the LAN port of IP Phone.
- 5. After **Save&Reboot**, the MAC of LAN port will become to PC's original MAC address.



#### **Tone Settings**

This page defines the Tone settings. This function can setup the related parameters of Dial Tone, Ring Back Tone, Busy Tone, Error Tone and Inser Tone. When you finished the setting, please click the Submit button.

## **Tones Settings**

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:		✓	✓	✓	<b>✓</b>	✓
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0
	Submit	Reset				

#### **Advanced Settings**

This page defines the advanced functions. When you finished the setting, please click the Submit button.

**ICMP Not Echo:** This function can disable echo when someone ping this device, it can avoid haker try to attack the device.

**Send Anonymous CID:** If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information. (The Registration Server also need support this function)

**Billing Signal:** There are provide three type billing types: Polarity Reversal, Tone\_12K and Tone\_16K. (The Registration Server also need support this function)

**CPC Delay:** When receive the disconnect signal, machine will to cut the voltage down to 0V after this time.

**CPC Duration:** When starting to cut the voltage down to 0V, machine will to continue this state by this time.

**Send Flash event:** There are provide two flash formats: DTMF Event and SIP Info.

SIP Encrypt: There are provide seven encrypt formats: INFINET, AVS, WALKERSUN1,

WALKERSUN2, CSF1, CSF2 and GX. (The Registration Server also need support this function)

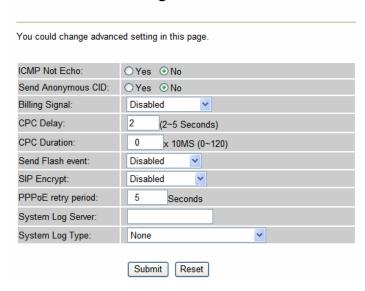
**PPPoE retry period:** If PPPoE dial-up connection fail, machine will retry the dial-up motion after this time.

**System Log Server:** Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address.

System Log Type: There are seven Syslog types: Call Statistics, General Debug, Call Statistics +

General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug and All.

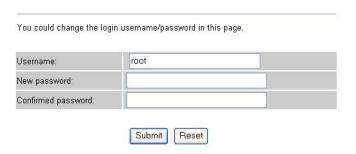
## Advanced Setting



#### **System Authority**

In System Authority you can change your login password.

# System Authority



#### Save & Reboot

In Save & Reboot you can save the changes you have done. If you want to use new setting in the IP Phone, you have to click the Save button. After you click the Save button, the IP Phone will automatically restart and the new setting will effect.

## Save & Reboot

You have to save changes to effect them.

Save Changes: Save

#### **Firmware Upgrade**

In Firmware Upgrade function you can update new firmware via HTTP or TFTP methods in this page.

You can ugrade the firmware by the following steps:

Select the upgrade method and the firmware code type, AP or DSP code.

Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the device then click the Update button.

## 

#### ① Note:

After firmware loaded, the unit will be reboot, and Default IP address of the customized firmware: <a href="http://192.168.0.1">http://192.168.0.1</a>; login name/password: root/null (no password)

#### **Auto Upgrade**

The device can update new firmware with the **gz** or **ds** file format automatically by the Auto Upgrade function.

# Auto Update Settings

You could set auto upda	ate settings in this page.
Update via:	⊙ Off ○TFTP ○ FTP ○ HTTP
TFTP Server:	
HTTP Server:	Exp. 60.35.187.30
HTTP File Path:	Exp. /download/
FTP Server:	Exp. 60.35.17.1
FTP Username:	
FTP Password:	
FTP File Path:	Exp. /file/load
Check new firmware:	O Power ON Scheduling only
Scheduling (Date):	14 (1~30 days)
Scheduling (Time):	AM 00:00- 05:59 🕶
Automatic Update:	Notify only     Automatic
Firmware File Prefix:	PHONE
Next update time:	
	Submit Reset

Field	Descriptions
Update via	There are TFTP/ FTP and HTTP three ways to provide the auto
	upgrade function.
TFTP Server	Input the TFTP Server address, and it could input the IP or Domain
	Name form.
HTTP Server	Input the HTTP Server address, and it could input the IP or Domain
	Name form.
HTTP File Path	Set up the file path.
FTP Server	Input the FTP Server address, and it could input the IP or Domain
	Name form.
FTP Username	The login username.
FTP Password	The login password
FTP File Path	Set up the file path.
Check new firmware	The device will according to the below ways to check the new
	firmware.
	- Power On (+ Scheduling): The machine will check the new
	firmware when power on and following the scheduling date

	and time.			
	- Scheduling: The machine will follow the scheduling date			
	and time to check the new firmware.			
Scheduling (Date)	The machine will check the new firmware between the time range			
	by random.			
Automatic Update	There are Notify only and Automatic ways to update.			
	- Notify only: If there are new firmware, the IP Phone will			
	send the "Be Be Be" sounds when pick up the handset to			
	prompt there are new firmware.			
	- Automatic: The device will carry firmware update out			
	automatically.			
Firmware File Prefix	It will check the information of model name.			
Next update time	It will show the next check date and time.			

#### ①Note:

If the **Check new firmware** field selected to Power On, the machine will chck the new firmware accoeding the scheduling time/date and power on. If there are new firmware can be upgraded, the machine won't carry firmware update out automatic. The machine will show the [Found New s/w] message on LCD. Then press [Menu] button for entrying the main menu and select the [7.Administrator -> 2. Upgrade System -> 1.Upgrade Now] selection to carry out the upgrade firmware action.

#### **Reset to Default**

In Default Setting you can restore the IP Phone to factory default in this page. You can just click the Restore button, then the IP Phone will restore to default and automatically restart again.

# Reset to Default You could click the restore button to restore the factory settings. Reset to default: Restore

#### **Reboot without saving**

Reboot function you can restart the IP Phone. If you want to restart the IP Phone, you can just click the Reboot button, then the IP Phone will reboot automatically.

# Reboot without Saving

You could press the reboot button to restart the system.

Reboot without Saving: Reboot

## **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 1: Voice communication via IP PBX system \_ IPX-2000 (Auto-config)

In the following sample, we'll introduce how to integrate the IP Phone with our IP PBX system IPX-2000 via Auto-config feature.



#### Machine configuration on the IPX-2000

Line Number: 1001

#### STEP 1:

Log in IPX-2000 and browse to the DHCP menu and create new options list for the auto configuration.

:: DHCP SER	VICE	
DHCP POOL lan	● Enable ● Disable On-board LAN ▼ APPLY	
	○ Range ○ Single-host Pool Name lan	
	IP 192.168.1.101 ~ 192.168.1.200 Options 150,192.168.1.1   DEL	
	Code, Value 151 http://192.168.0.50Affpboot/ ADD  ADD UPDATE DEL CLEAR	

Code: please insert 151 as the DHCP server option.

Value: http://LAN IP of IPX-2000/tftpboot

If you'd like to enable auto-config for IP extension features in IPX-2000, please be sure to setup the DHCP option code and the value information.

In most case, insert the optional code 151 and the value=http://192.168.0.50/tftpboot/

Note: the 192.168.0.50 is the IP address of IPX-2000

#### STEP 2:

Please browse to the **Device** → **IP Phone** menu and create new device. And press the **EDIT** button for set up the Auto Config configuration.



#### STEP 3:

Please fill out the Vendor Prefix code and MAC Address of IP Phone.

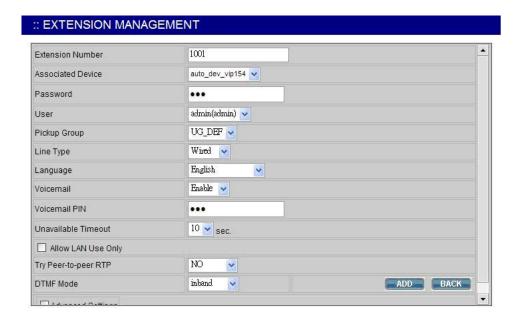
Enable Automa	tic Client Configuration
Device	auto_dev_vip154
Vendor Prefix	pla154 (a-zA-Z0-9_)
MAC Address	00 : 30 : 4f . aa : bb : dd
Supplementary C	onfiguration 💌
Codec Pro	eference
1st codec	g711ulaw 💌
1st packet time	20 💌
2nd codec	g711alaw 💌
2nd packet time	20 💌
3rd codec	g729 💌
3rd packet time	30 💌
Enable Voice	Activity Detection (VAD)
DTMF Mode	Inband 💟

#### ①Note:

The Vendor Prefix of VIP-154 series is pla154

#### STEP 4:

Please browse to the **Device**  $\rightarrow$  **Extension of IP Phone** menu to create the two extension accounts/password: **1001/123** (for VIP-154T <u>A</u>), and **1002/123**(for VIP-154T <u>B</u>) for the voice calls.



#### STEP 5:

After setting up the parameters, please browse to the **Service** → **IP PBX service** menu, and press **RELOAD** of IP PBX configuration reload selection for activating the settings.



#### Machine configuration on the VIP-154T:

#### STEP 6:

Please log in VIP-154T A via web browser, browse to the **Advanced Settings** menu. In the setting page, please browse to the **Auto-config** page, and select to **IP-PBX** choice of the Auto Configuration features for IP PBX system.

# **Auto Configuration Settings**

You could enable/disab	ole the auto	configuration	on setting in	n this page.	
Auto Configuration:	Off	OTFTP	O FTP	ОНПР	● IP-PBX
TFTP Server:					
HTTP Server:				Ex	p. 60.35.187.30
HTTP File Path:				Ex	p. /download/
FTP Server:				Ev	p. 60.35.17.1
FTP Username:	-				p. 00.55.17.1
FTP Password:					
FTP File Path:				Ex	p. /file/load

#### STEP 7:

After enabled the Auto-config feature, the VIP-154T <u>A</u> shall be able to obtain IP address and SIP extension information from IP PBX system IPX-2000 information. To verify the auto-config results, you may check the extension number from LCD display assigned by IPX-2000.

#### STEP 8

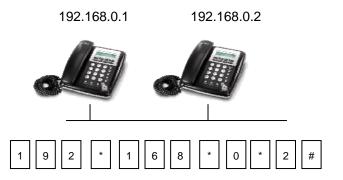
Repeat the same configuration steps on VIP-154T  $\underline{B}$ , and check if the VIP-154T  $\underline{B}$  is successfully registered with the IPX-2000 as one of the IP extensions.

#### Test the scenario:

To verify the VoIP communication, you may make calls from extension side (VIP-154T  $\underline{A}$ ) 1001 to the number 1002 (VIP-154T  $\underline{B}$ ) or reversely make calls from extension client (VIP-154T  $\underline{B}$ ) 1002 to the number 1001 (VIP-154T  $\underline{A}$ )

#### Case 2: VIP-154T to VIP-154T connection via IP address

Assume there are two VIP-154T's in the network the IP address are 192.168.0.1, 192.168.0.2



#### **Operation steps:**

Pick up the VIP-154T A, you should be able to hear the dial tone, press the keypad: 192\*168\*0\*2# shall be able to connect to the VIP-154T B.

Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on VIP-154T B to establish the first voice communication from the second VIP-154T

## (i) 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

#### Case 3: Voice communication via SIP proxy server \_SIP-50



■ VIP-154T IP Address: 192.168.0.1

Line Number: 1001

■ VIP-154T IP Address: 192.168.0.2

Line Number: 2002

#### Machine configuration on the VIP-154T:

#### STEP 1:

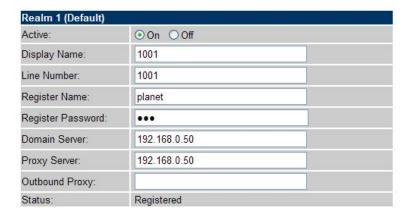
Log in SIP-50 and create two testing accounts/password: 1001/123 (for VIP-154T A), and 1002/123(for VIP-154T B) for the voice calls.

#### STEP 2:

Please log in VIP-154T A via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:

# Service Domain Settings

You could set information of service domains in this page.



#### STEP 3:

Repeat the same configuration steps on VIP-154T B, and check the machine registration status, make sure the registrations are completed.

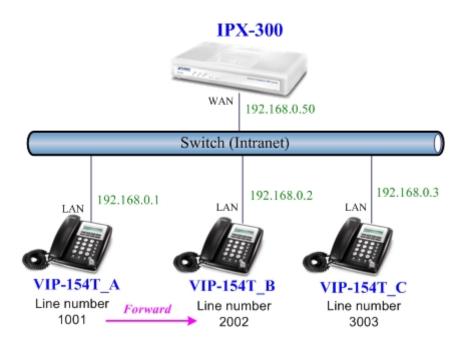
#### Test the scenario:

To verify the VoIP communication, you may make calls from SIP client (VIP-154T A) 1001 to the number 1002 (VIP-154T B) or reversely make calls from SIP client (VIP-154T B) 1002 to the number 1001 (VIP-154T A)

#### Case 4: Call Forward Feature\_Example 1

In the following samples, we'll introduce the Call Forward Feature applications.

In this example, there are three VIP-154T register to IPX-300 and VIP-154T\_A had set Call Forward function to VIP-154T\_B. (The detail registration settings of IPX-300 and VIP-154T please refer to the instruction of Case 3)

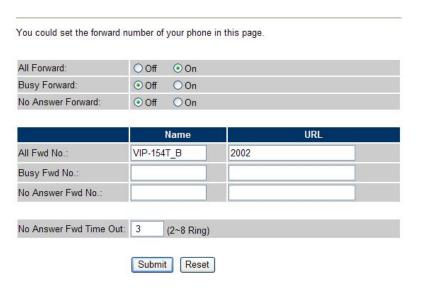


#### Machine configuration on the VIP-154T:

#### STEP 1:

Please log in VIP-154T\_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-154T\_B, then the sample configuration screen is shown below:

# Forward Settings



#### STEP 2:

After set up completed and reboot machine, the LCD screen will show below:

10-19	17:20	
# Forw	ard#	

After 2~3 seconds, the LCD screen will show below:

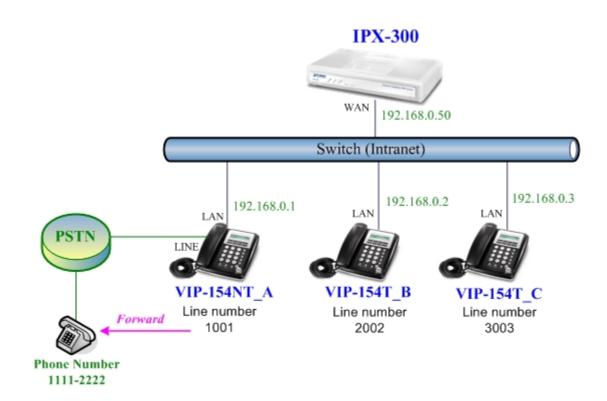
10-19	17:20		
AF 200	2		

#### Test the scenario:

VIP-154T\_C pick up the telephone and dial the number 1001(VIP-154T\_A), because VIP-154T\_A had set up **All Forward** function to the number 2002(VIP-154T\_B), so the number 2002(VIP-154T\_B) will ring up then it pick up the telephone and communication with the number 3003(VIP-154T\_C).

#### Case 5: Call Forward Feature\_Example 2

In this example, there are one VIP-154NT and two VIP-154T register to IPX-300. The VIP-154NT\_A had set Call Forward function to phone number 1111-2222 (PSTN).

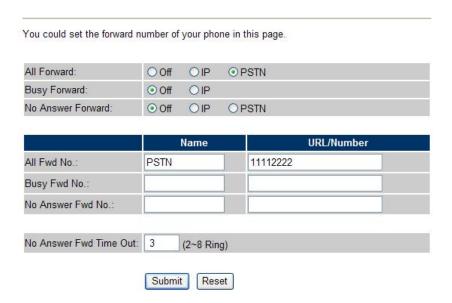


#### Machine configuration on the VIP-154NT:

#### STEP 1:

Please log in VIP-154NT\_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **PSTN** choice and fill in the **Name** and **URL/Number** of PSTN Phone Number 11112222, then the sample configuration screen is shown below:

# Forward Setting



#### STEP 2:

After set up completed and reboot machine, the LCD screen will show below:

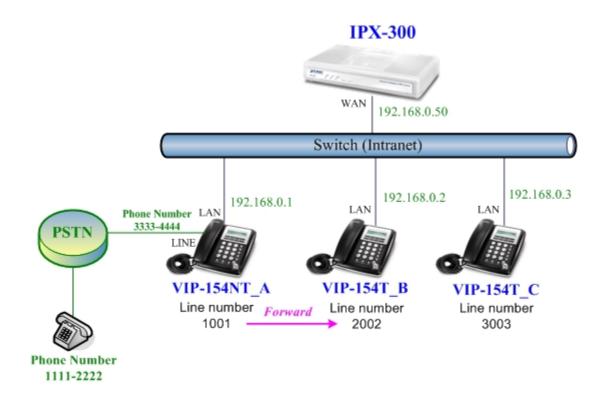
After 2~3 seconds, the LCD screen will show below:

#### Test the scenario:

VIP-154T\_C pick up the telephone and dial the number 1001(VIP-154NT\_A), because VIP-154NT\_A had set up **All Forward** function to the PSTN Phone Number 11112222, so the PSTN Phone Number 11112222 will ring up then it pick up the telephone and communication with the number 3003(VIP-154T\_C).

#### Case 6: Call Forward Feature\_Example 3

In this example, there are one VIP-154NT and two VIP-154T register to IPX-300. The VIP-154NT\_A had set Call Forward function to number 2002 (VIP-154T\_B).



#### Machine configuration on the VIP-154NT:

#### STEP 1:

Please log in VIP-154NT\_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **IP** choice and fill in the **Name** and **URL/Number** of VIP-154T\_B, and then the sample configuration screen is shown below:

#### Forward Settings You could set the forward number of your phone in this page. All Forward: Off ⊙ IP **OPSTN** Busy Forward: ⊙ Off OIP No Answer Forward: Off OIP **OPSTN URL/Number** Name All Fwd No .: VIP154T B 2002 Busy Fwd No .: No Answer Fwd No .: No Answer Fwd Time Out: 3 (2~8 Ring) Submit Reset

#### STEP 2:

After set up completed and reboot machine, the LCD screen will show below:

10-19 17:20 # Forward #

After 2~3 seconds, the LCD screen will show below:

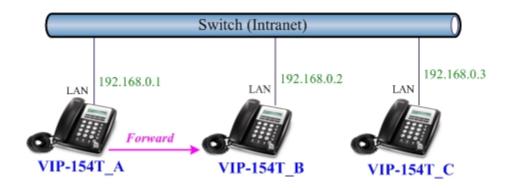
10-19 17:20 AF 2002

#### Test the scenario:

PSTN Phone Number 11112222 pick up the telephone and dial the PSTN Phone Number 33334444(VIP-154NT\_A), because VIP-154NT\_A had set up **All Forward** function to the number 2002(VIP-154T\_B), so the number 2002(VIP-154T\_B) will ring up then it pick up the telephone and communication with the PSTN Phone Number 11112222.

#### Case 7: Call Forward Feature\_Example 4

In this example, there are three VIP-154T and connect with Peer to Peer mode. VIP-154T\_A had set Call Forward function to VIP-154T\_B.

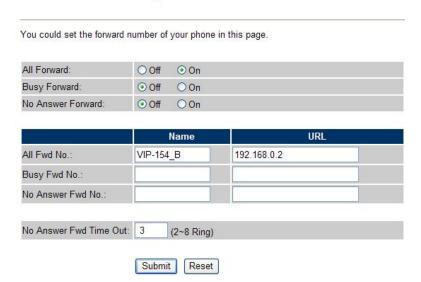


#### Machine configuration on the VIP-154T:

#### STEP 1:

Please log in VIP-154T\_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-154T\_B, and then the sample configuration screen is shown below:

## Forward Settings



#### STEP 2:

After set up completed and reboot machine, the LCD screen will show below:

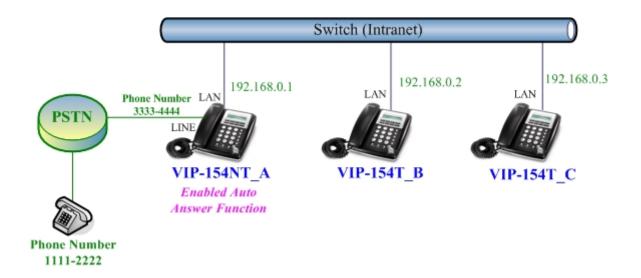
After 2~3 seconds, the LCD screen will show below:

#### Test the scenario:

VIP-154T\_C pick up the telephone and dial the IP Address 192.168.0.1(VIP-154T\_A), because VIP-154T\_A had set up **All Forward** function to the IP Address 192.168.0.2(VIP-154T\_B), so the IP Address 192.168.0.2 (VIP-154T\_B) will ring up then it pick up the telephone and communication with the VIP-154T\_C.

#### Case 8: Auto Answer Feature IP to PSTN

In this example, there are one VIP-154NT and two VIP-154T and connect with Peer to Peer mode. The VIP-154NT\_A had set Auto Answer function for forwarding calls to arbitrary telephone. If there have incoming IP calls and VIP-154NT\_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password then dial out an arbitrary PSTN telephone.



#### Machine configuration on the VIP-154NT:

#### STEP 1:

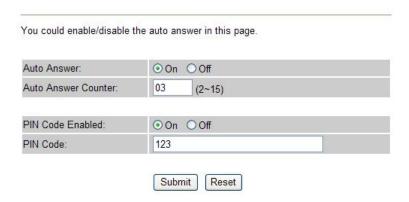
Please log in VIP-154NT\_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please disable All **Forward** function, and then the sample configuration screen is shown below:

#### Forward Setting You could set the forward number of your phone in this page. All Forward: Off OIP **OPSTN** Busy Forward: Off OIP No Answer Forward: OIP ⊙ Off **OPSTN** URL/Number Name All Fwd No .: Busy Fwd No. No Answer Fwd No.: No Answer Fwd Time Out: 3 (2~8 Ring) Submit Reset

#### STEP 2:

Please log in VIP-154NT\_A via web browser, browse to the **Phone Settings** menu and select the **Auto Answer** config menu. In the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, then the sample configuration screen is shown below:

#### Auto Answer

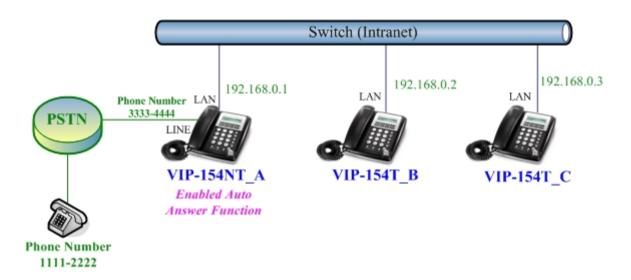


#### Test the scenario:

VIP-154T\_C pick up the telephone and dial the IP Address 192.168.0.1(VIP-154NT\_A), VIP-154NT will ring up but doesn't answer the call. After **3** rings, the VIP-154T\_C will hear the prompt sounds then input the password **123#**. VIP-154T\_C will hear the dial tone from PSTN line then input Phone Number 11112222. The Phone Number 11112222 will ring up then it pick up the telephone and communication with the VIP-154T\_C.

#### Case 9: Auto Answer Feature\_PSTN to IP

In this example, there are one VIP-154NT and two VIP-154T and connect with Peer to Peer mode. The VIP-154NT\_A had set Auto Answer function for forwarding to arbitrary telephone. If there have incoming PSTN calls and VIP-154NT\_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password and then dial out an arbitrary IP telephone.



#### Machine configuration on the VIP-154NT:

#### STEP 1:

Please log in VIP-154NT\_A via web browser, browse to the **Phone Settings** menu and select the **Auto Answer** config menu. In the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, and then the sample configuration screen is shown below:

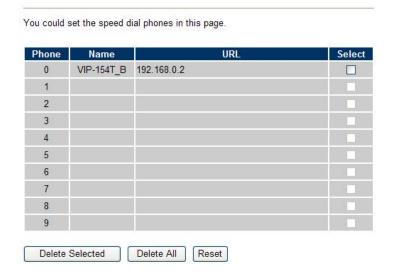
#### Auto Answer

You could enable/disable t	he auto answer in this page.	
Auto Answer:	⊙ On ○ Off	
Auto Answer Counter:	03 (2~15)	
PIN Code Enabled:	⊙ On ○ Off	
PIN Code:	123	

#### STEP 2:

Please log in VIP-154NT\_A via web browser, browse to the **Phone Book** menu and select the **Speed Dial Settings** config menu. In the setting page, please add a speed dial number for dial to IP address 192.168.0.2 (VIP-154T\_B), and then the sample configuration screen is shown below:

## Speed Dial Phone List



#### Test the scenario:

The Phone Number 11112222 pick up the telephone and dial the PSTN Phone Number 33334444(VIP-154NT\_A), VIP-154NT will ring up but doesn't answer the call. After **3** rings, the Phone Number 11112222 will hear the prompt sounds then input the password **123#**. The Phone Number 11112222 will hear the dial tone then input **0#**. The IP address 192.168.0.2 (VIP-154T\_B) will ring up then it pick up the telephone and communication with the Phone Number 11112222.

## Appendix B The method of operation guide

In this section, we'll introduce the features method of operation, and lead you step by step to establish these features.

#### **Call Transfer**

#### A. Blind Transfer

- 1. B call to A and they are in the process of conversation.
- 2. A press "**Transfer**" button to hold the conversation with B, and input the number of C (Follow by the "#" key).
- 3. C will ring up, and A hang up the handset.
- 4. C picks up the handset and conversation with B.

#### **B.** Attendant Transfer

- 1. B call to A and they are in the process of conversation.
- 2. A press "**Transfer**" button to hold the conversation with B, and input the number of C (Follow by the "#" key).
- C will ring up.
- 4. C picks up the handset and conversation with A.
- 5. A hang up and C conversation with B.

#### 3-Way Conference

- 1. A and B are in the process of conversation.
- 2. A want to invite C to join their conversation.
- 3. A press "**Transfer**" button to hold the conversation with B, and input the number of C (Follow by the "#" key).
- 4. C will ring up and pick up the handset to conversation with A.
- 5. A press "CONF" button and they will entry the 3-Way conference mode.

#### **Call Waiting**

- 1. A and B are in the process of conversation.
- 2. C call to A and A will hear the prompt sounds.
- 3. A press "Hold" button to hold the conversation with B, and switch to conversation with C.

## Switch the Realm (Registration Proxy Server)

IP Phone can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input

the switch code.

#### Realm switch code:

- 1\*: Realm 1
- 2\*: Realm 2
- 3\*: Realm 3

For example: The default is realm 1, input the **2**\* (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

# Appendix C VIP-154T / VIP-154PT / VIP-154NT Specifications

Product	SIP IP Phone	SIP PoE IP Phone	SIP IP Phone with PSTN connectivity		
Model	VIP-154T	VIP-154PT	VIP-154NT		
Hardware					
LAN	1 x 10/100Mbps RJ-45 port Power Over Ethernet 802.3af compliant at VIP-154PT				
PC	1 x 10/100Mbps RJ-45 port				
Telephone Interface	1 x RJ-11				
relephone interlace			PSTN connectivity at VIP-154NT		
LCD display	2 x 16 characters				
Speaker	Full duplex hands free speaker phone				
Protocols and Standard					
Standard	SIP 2.0 (RFC3261), MD5 for SIP authentication (RFC2069/ RFC 2617), SIP outbound proxy, SIP NAT Traversal Support STUN (RFC3489)				
Voice codec	G.711: 64k bit/s (PCM)				
	G.723.1: 6.3k / 5.3k bit/s				
	G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)				
		729A: 8k bit/s (CS-ACELP)			
	G.729B: adds VAD & CNG to G.729				
Voice Standard	Voice activity detection (VAD)				
	Comfort noise generation (CNG)				
	Acoustic echo canceller (AEC)				
	G.165: Line echo canceller (LEC)				
0 1	Jitter Buffer				
Supplementary	rvices  Caller ID  3-way conference Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding				
services					
	Call Hold/Waiting/Tran				
Call history	Record incoming call				
oun motory	Outgoing call				
	Missed (not accepted) call history				
Protocols	SIP v1 (RFC2543), v2(RFC3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP,				
	RARP, DNS, DHCP, SNTP, PPPoE				
Network and Configuration					
Access Mode	Static IP, PPPoE, DHCP				
Management	Web, LCD menu keypad, Telnet, auto-config by IPX-2000, auto-provision by				
	TFTP/FTP/HTTP				
Dimension (W x D x H)	170 mm x 220 mm x 60 mm				
Operating Environment	0~50 degree C, 0~90% humidity				
Power Requirement	12V DC				
	Power Over Ethernet 802.3af compliant at VIP-154PT				
EMC/EMI	CE, FCC Class B				