SIP IP Phone

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

User's manual

Version 2.01

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CE mark Warning

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

WEEE Warning

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

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Revision

User’s Manual for PLANET SIP IP Phone:

Model: VIP-154T/VIP-154PT/VIP-154NT

Rev: 2.01 (2007, July)

Part No. EM-VIP154V2.01

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

ⓘNote

1. IP Phone default IP is http://192.168.0.1. 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.
2	**Speed Dial M1~M4**	To make a speed dial call by pressing the speed dial key M1 ~ 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. To page up menu when at configuration mode.
5	**Vol - / ↓**	To decrease the volume of voice when at off-hooked state. To page down menu when at configuration mode.
6	**OK**	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

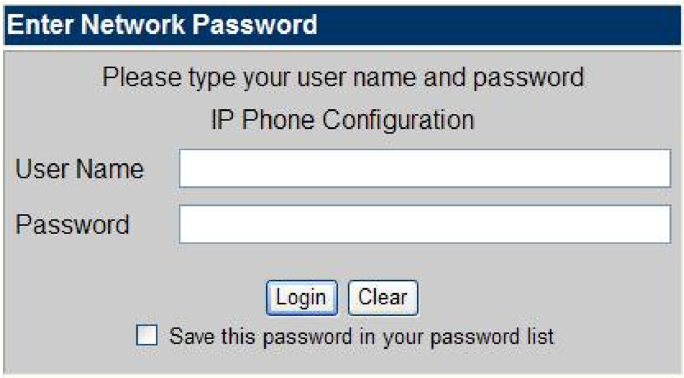
PLANET IP Phone provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access

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

- Netscape Communicator 4.03 or higher
- 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 http://192.168.0.1 in the address bar of your web browser to logon IP Phone web configuration page.



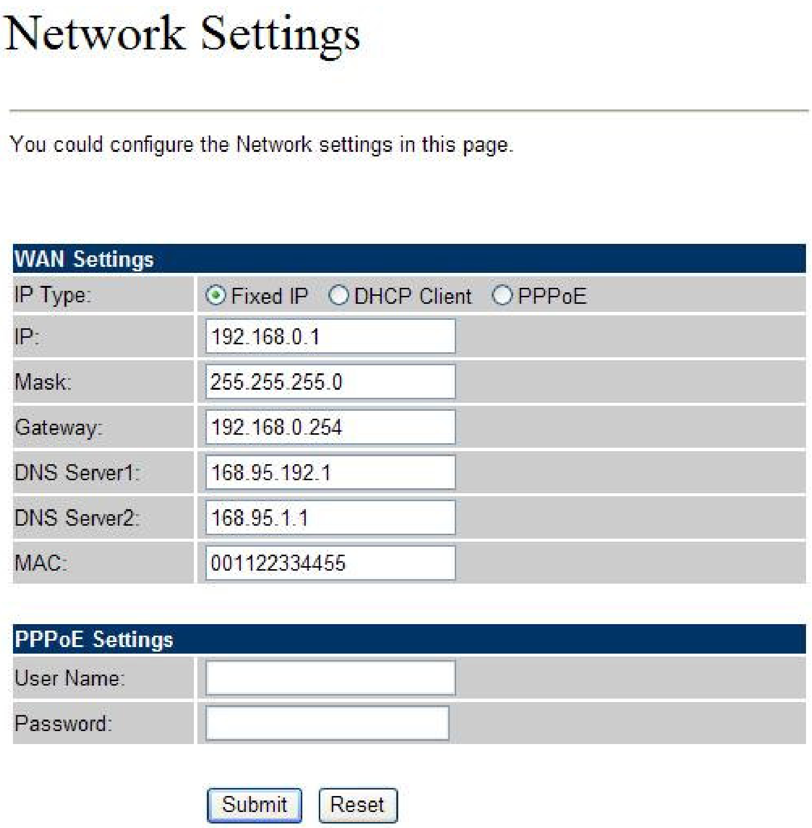
IP Phone will prompt for logon username/password, please enter: ***root*** / **null** (no password) to continue machine administration.

ⓘ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.

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of IP Phone in the adddress bar. After logging on machine with username/password (default**: root** / **not passwrd**), browse to "**Network**" --> "**Network 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.

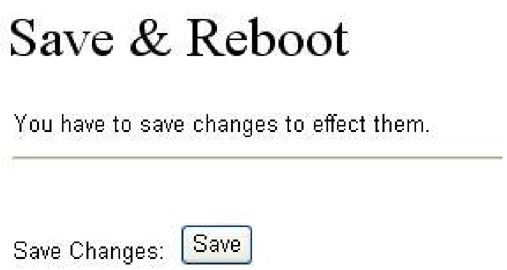
Connection Type	Data required.
Fixed IP	In most circumstances, it is no need to configure the DHCP settings.
DHCP clinet	The ISP will assign IP Address, and related information.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

ⓘ Hint

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

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.



Chapter 3

Network Service Configurations

Configuring and monitoring your IP Phone from web browser

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

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

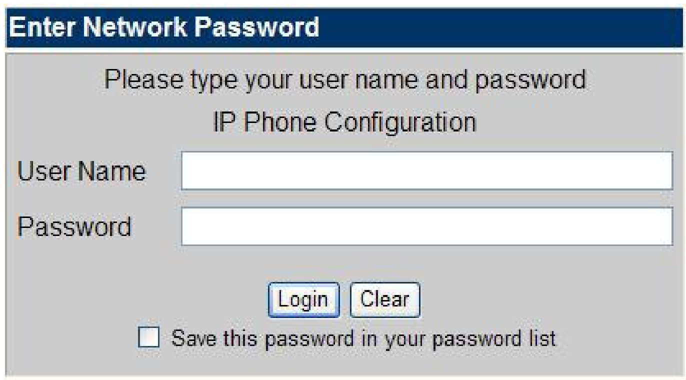
- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Manipulation of IP Phone via web browser

Log on IP Phone via web browser

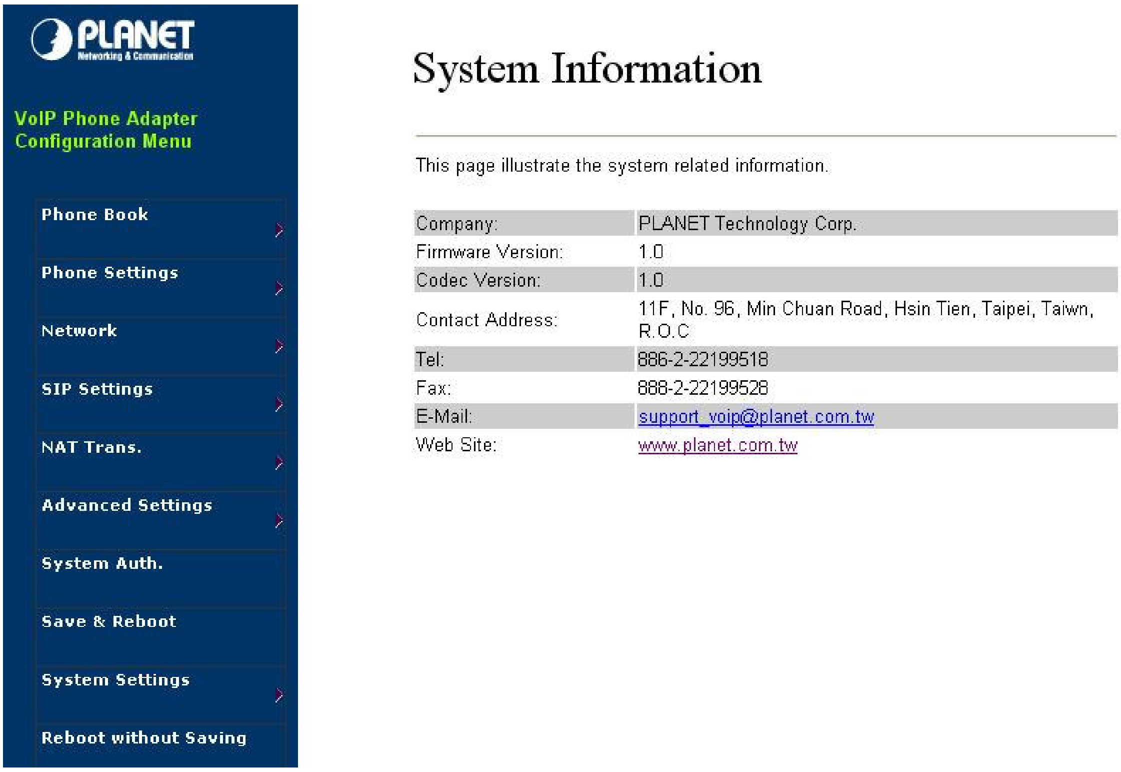
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon IP Phone web configuration page.

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.



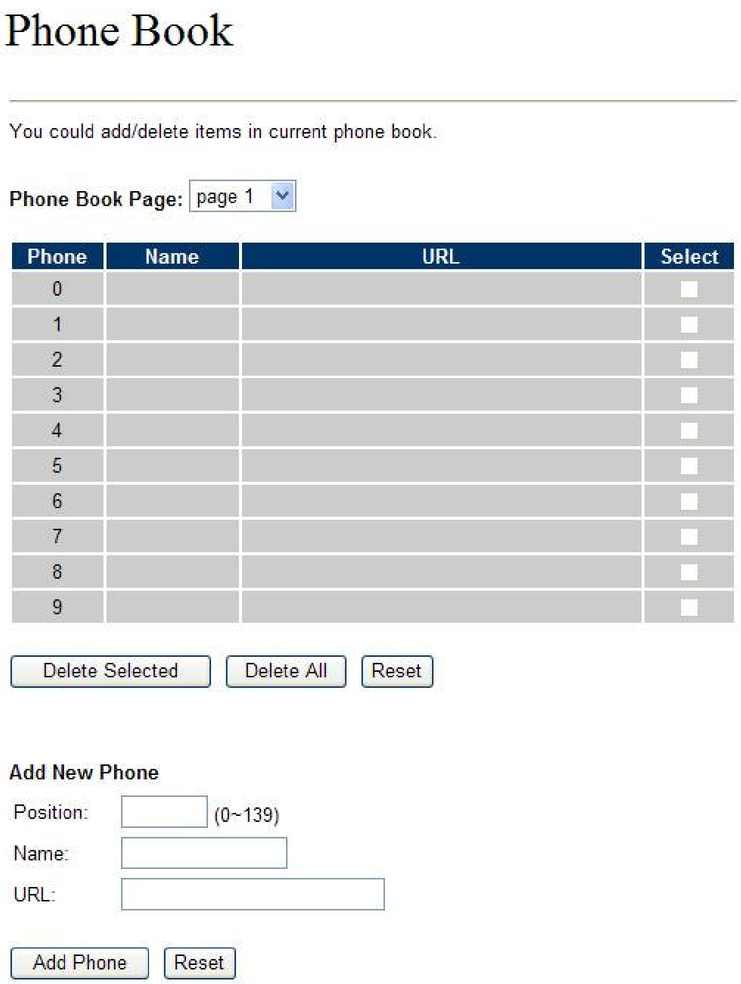
IP Phone main page

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.



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

You could add/delete items in current phone book.

Phone Book Page: page 1

Phone	Name	URL	Select
0	301	301@192.168.1.2	☐
1	206	17476433364	☐
2	202	192.168.1.2:5062	☐
3			
4			
5			
6			
7			
8			
9			

Delete Selected | Delete All | Reset

STEP 1:

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

```
Search:         [   3]
```

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

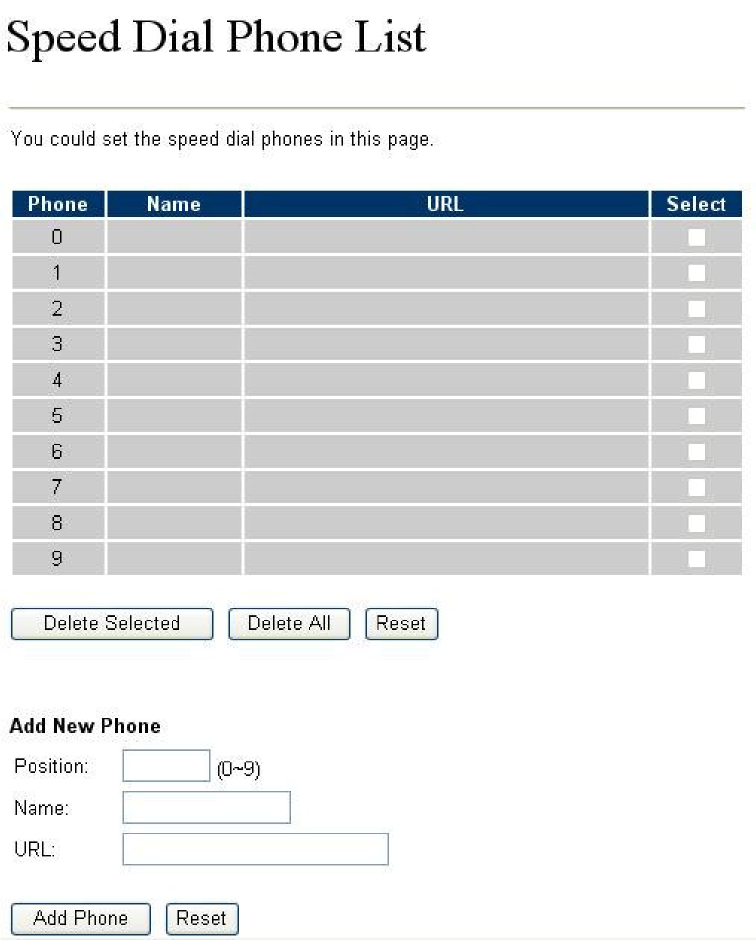
Speed Dial settings

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.



Call forward

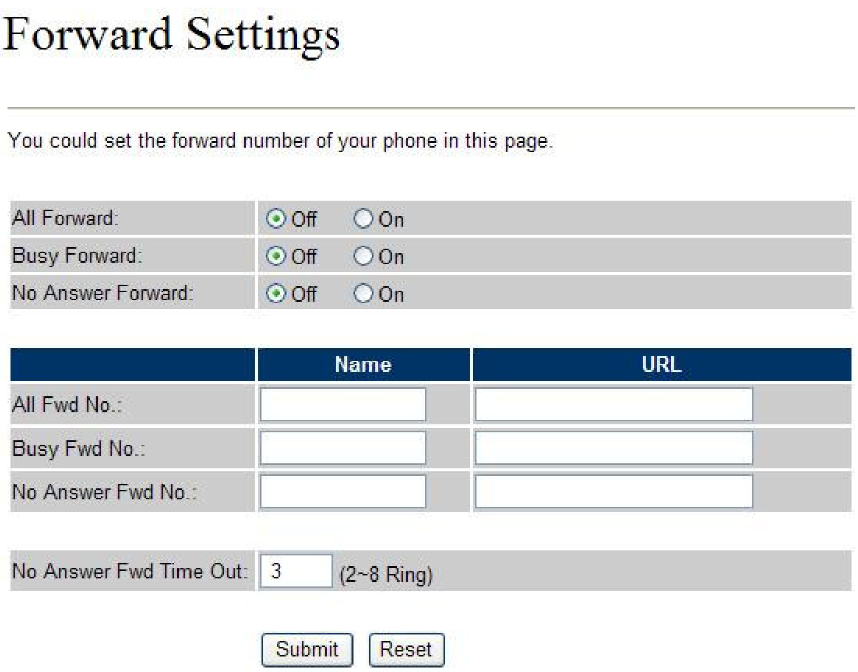
This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

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.

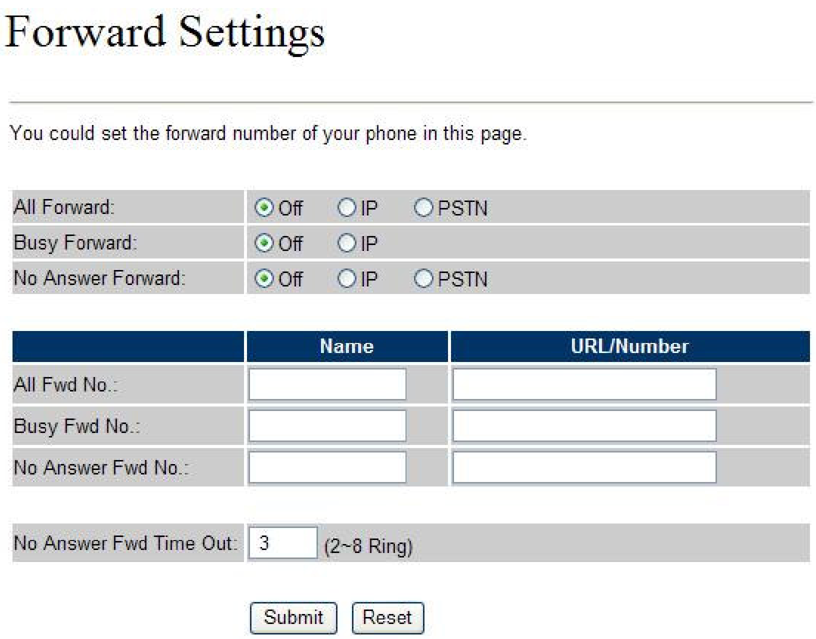
No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

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



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

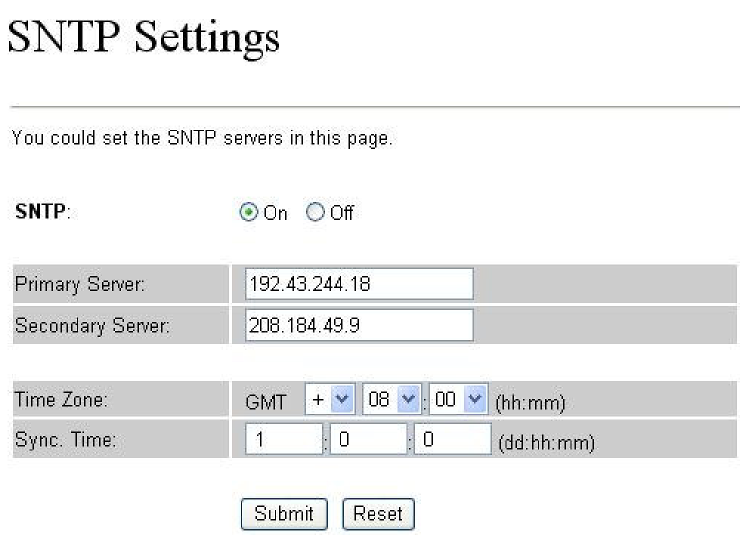
Call Forward to PSTN (VIP-154NT): VIP-154NT not only support Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, then input the name and the PSTN number in URL/Number field.



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.



Volume Setting

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

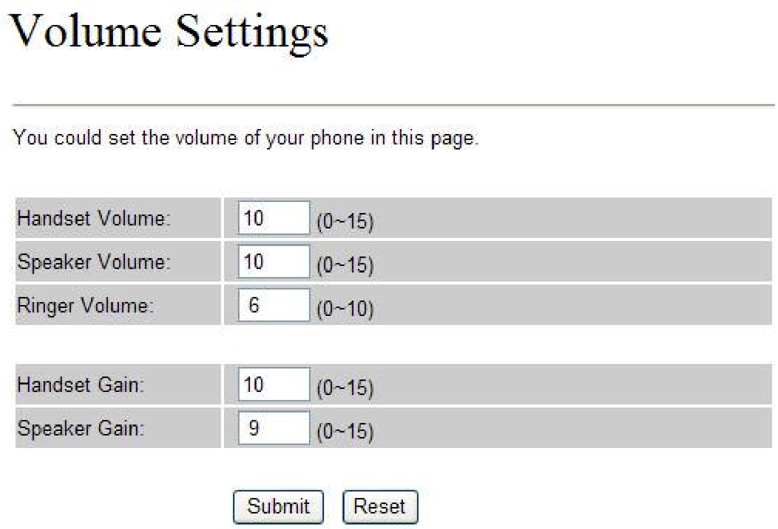
Handset Volume is to set the volume for you can hear from the handset.(Handfree mode)

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)

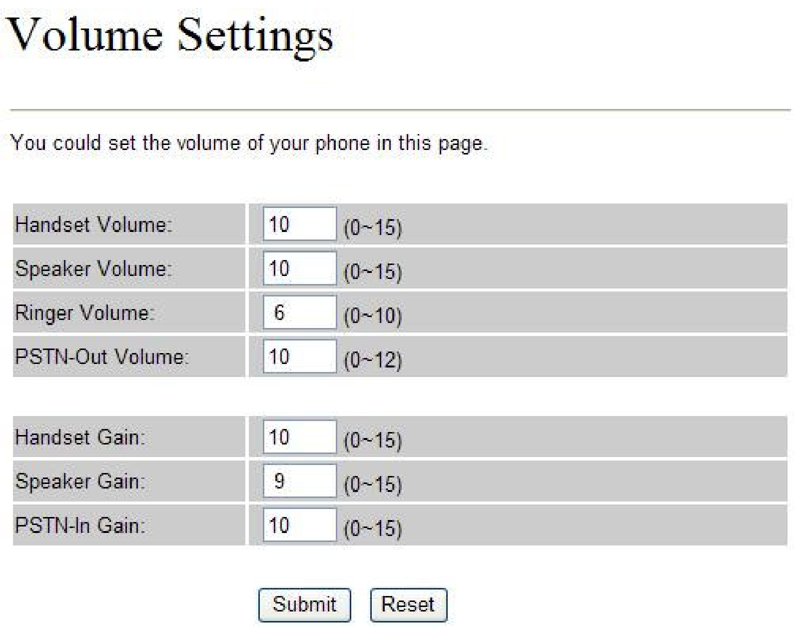


Volume Settings for VIP-154T/VIP-154PT

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.

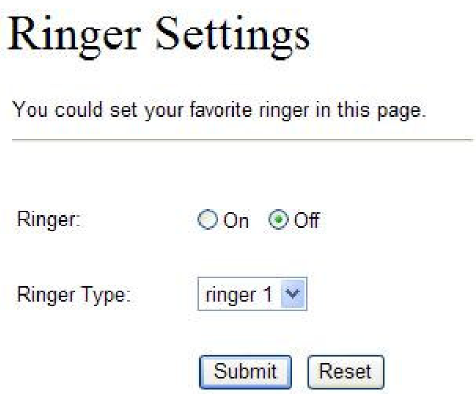
PSTN-In Gain is to set the volume send out to the other PSTN side's haneset.



Volume Settings 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.



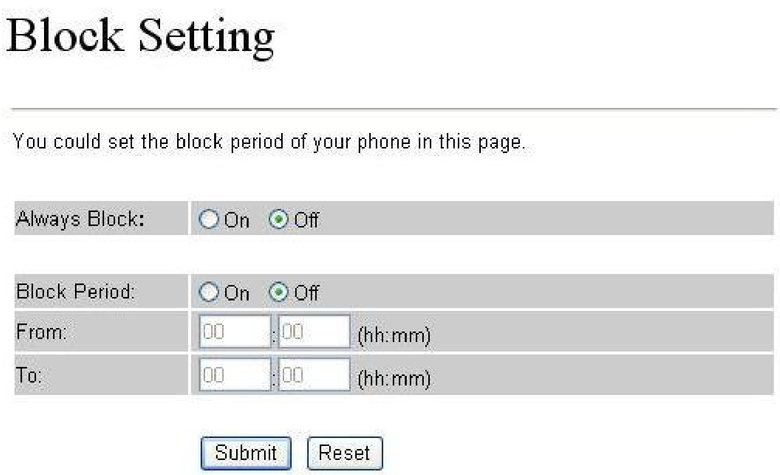
Block Setting

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.

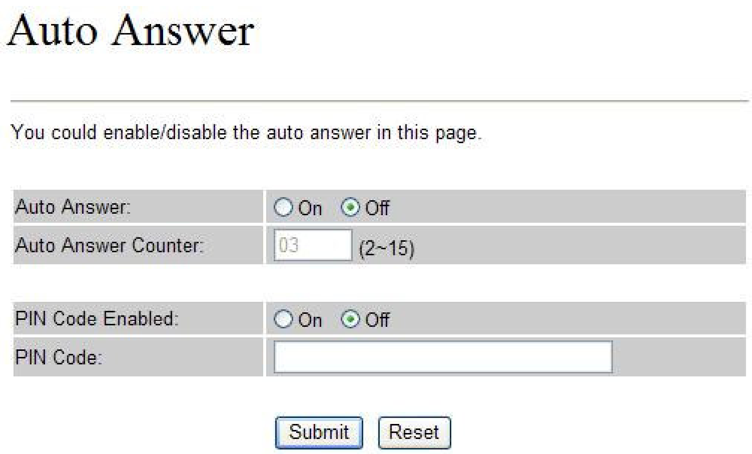


Auto Answer settings (For VIP-154NT)

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.

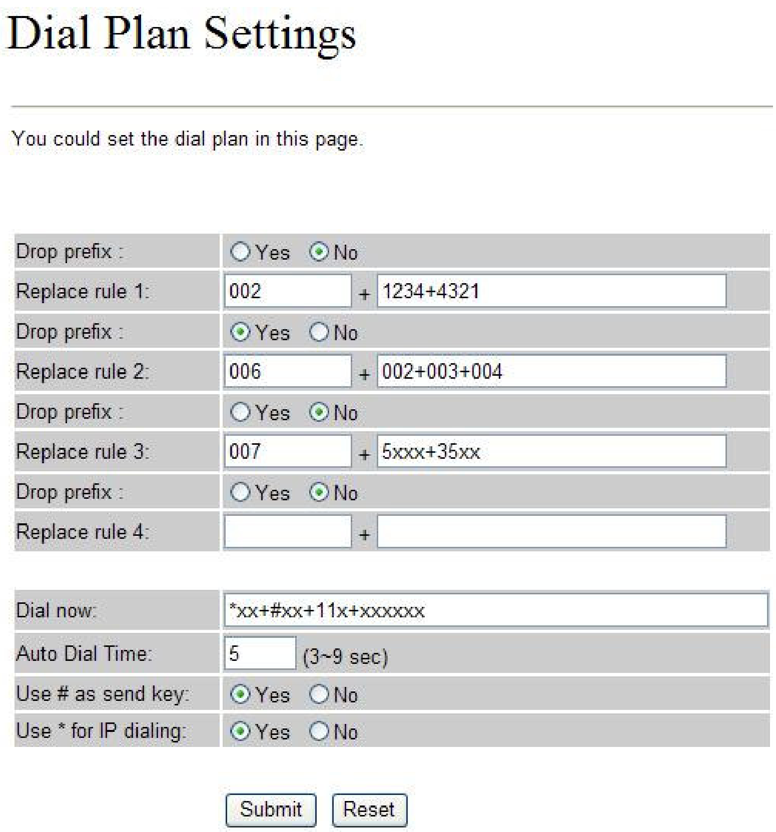
Auto Answer Counter is to set after the ring count met the number you set then the auto answer will enable.

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.



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. ⓘ**Note**: 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)

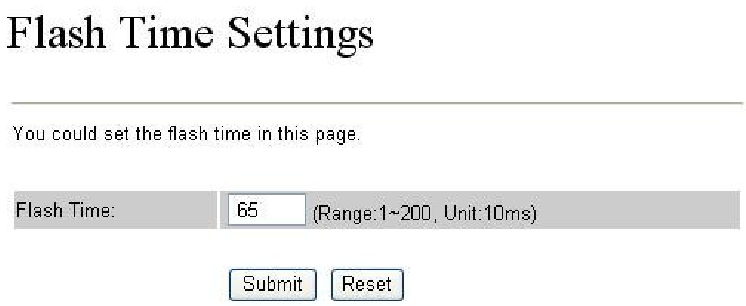
1. If the dialing number start as "**5**" and follow 3 digits, it will add the 007 at begin. The real dialing number is [**0075xxx**].
2. If the dialing number start as "**35**" and follow 2 digits, it will add the 007 at begin. The real dialing number is [**00735xx**].

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**.

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.

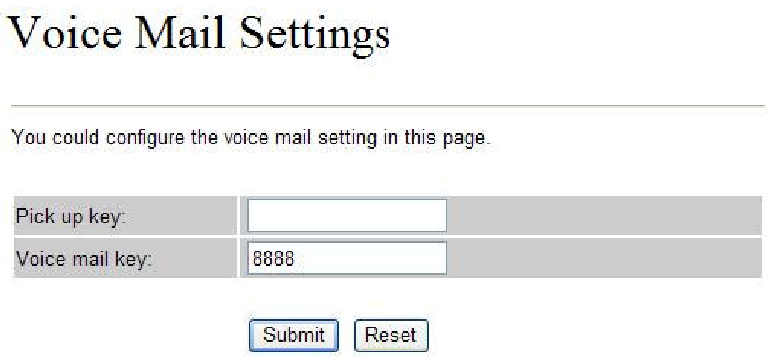


Call waiting Settings

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.

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.

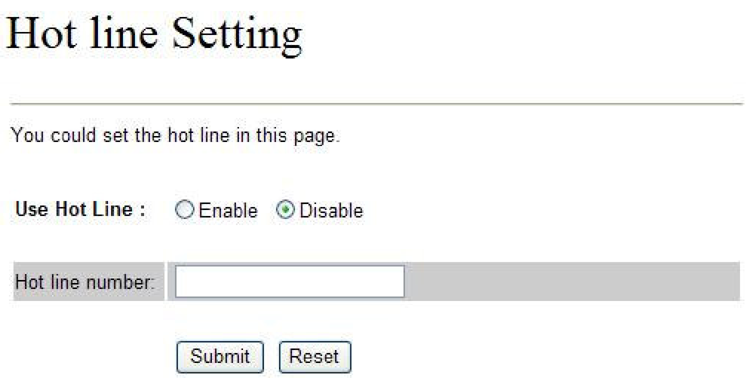


Hot line Settings

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.



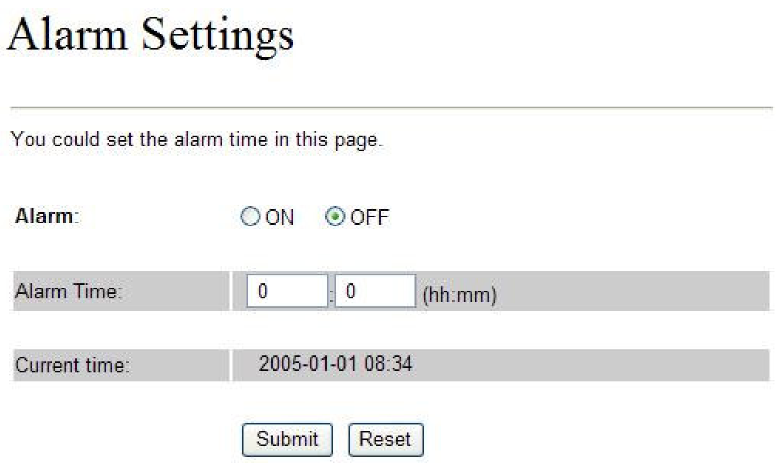
Alarm Settings

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.

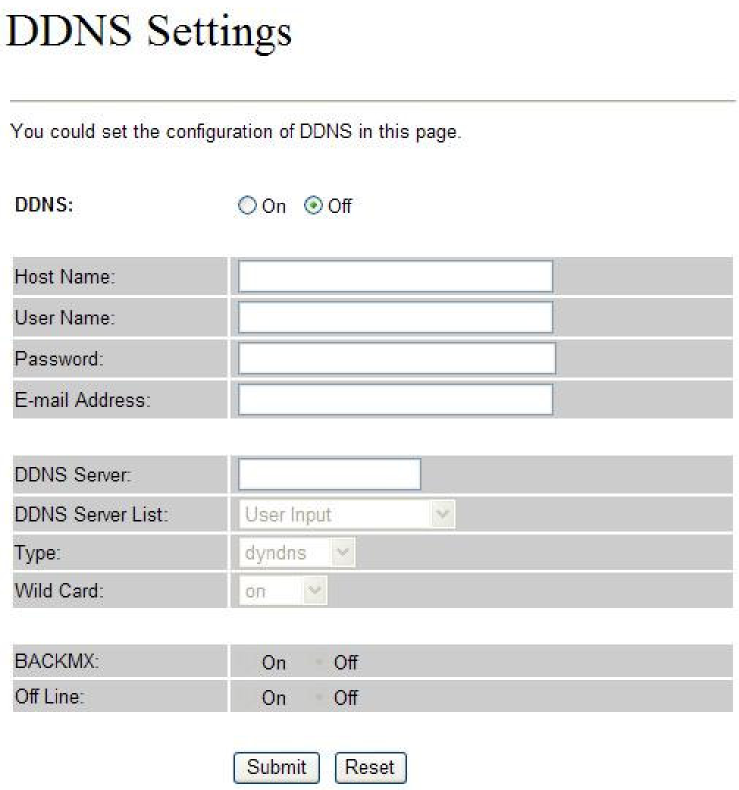
Alarm Time: It can set up the system prompt time with 24 hours.

Current time: The next alarm time.



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.



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.

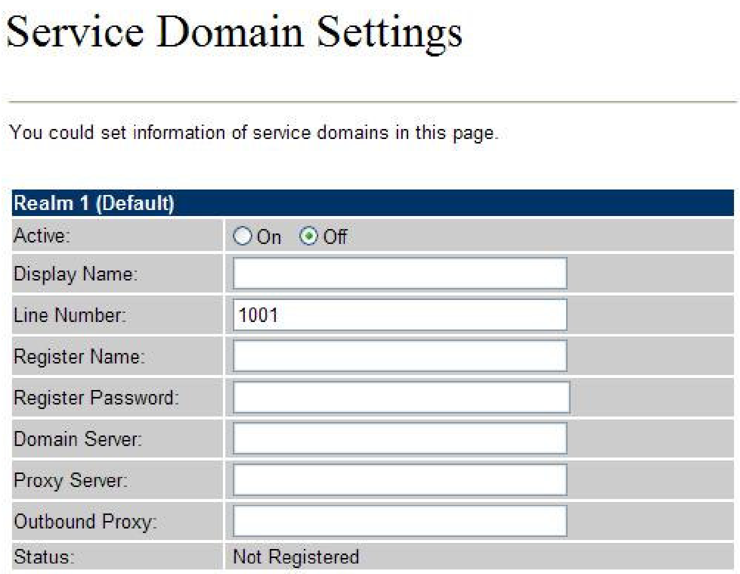
Proxy Server: you need to input the Proxy Server get from your ISP.

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 direcly.

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.



ⓘ**Note**:

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

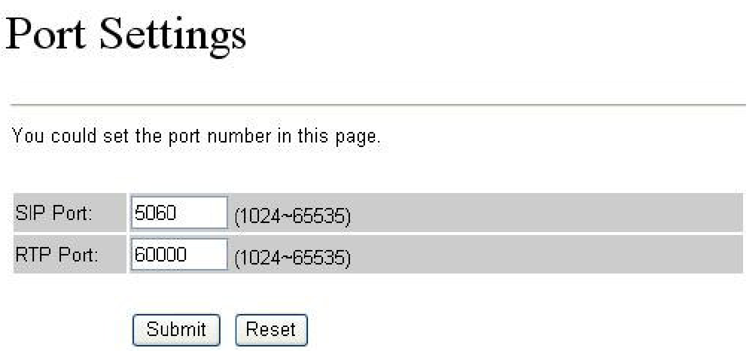
2*: Realm 2

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.

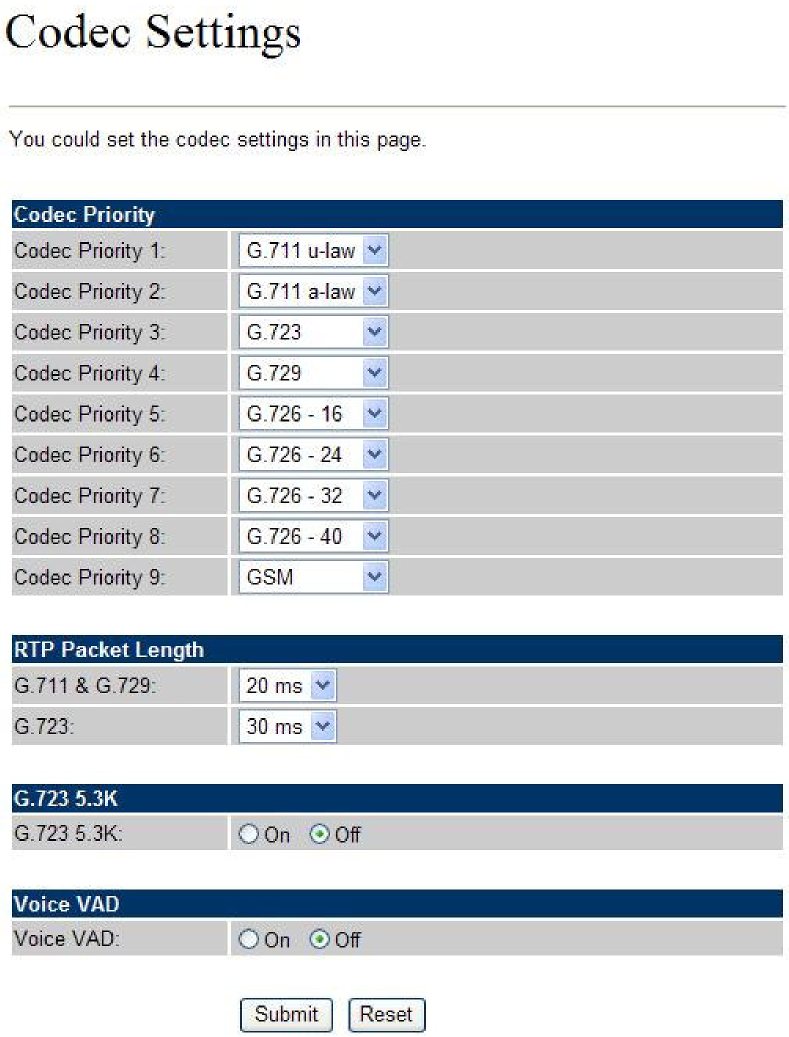
Port Settings

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.



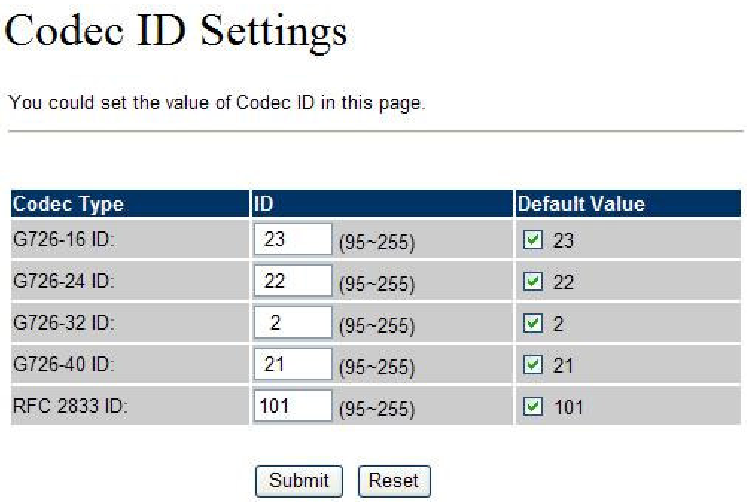
Codec Settings

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.



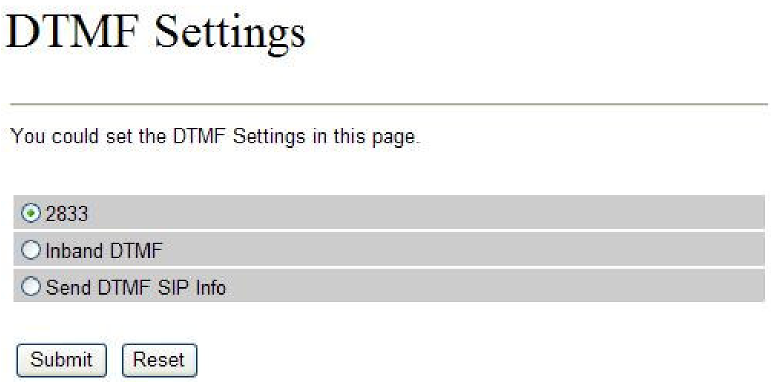
Codec ID Setting

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.



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.



RPort Settings

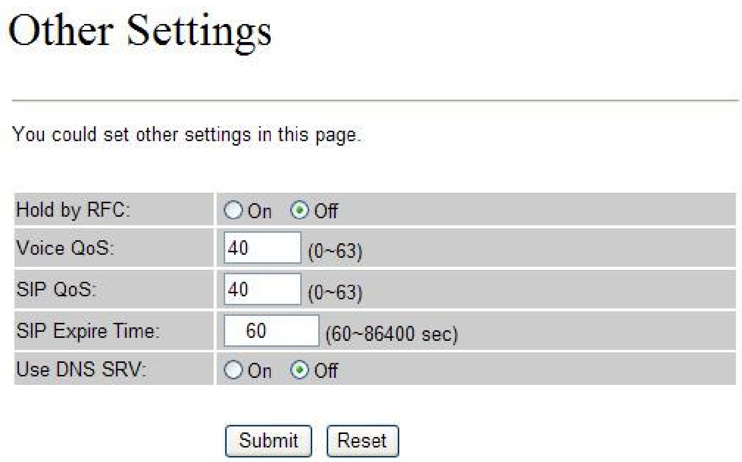
This page defines the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

Other Settings

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

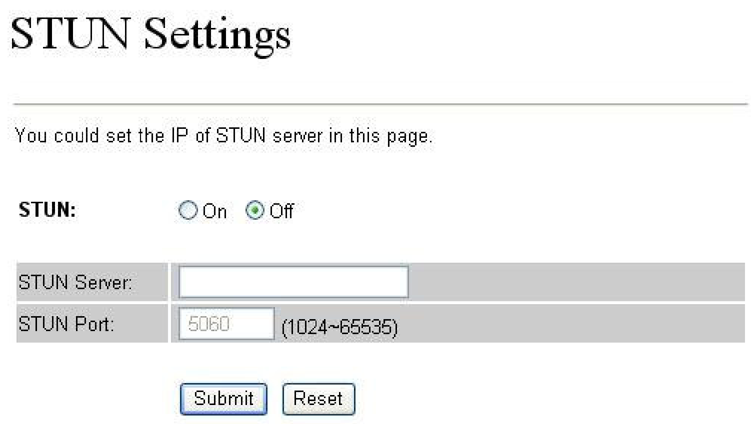
A Domain Name Server (DNS) SRV record helps connecting to a SIP user in a similar way that an MX record helps email delivery. When you send email to "john@example.com", the MX record for example.com might tell the mail transfer agent to deliver email to a completely different machine, such as "zaphod.foobar.com". Similarly, when you want to make a SIP phone call to john@example.com, the SRV record might tell your computer that it should connect to "galaxy.starsystem.tw" to do so.

Dialing by domain names lets a SIP user have a single public "SIP address" which can be redirected at will to their current location. SRV records maintain stability and also opens up the possibility to use your own domain, regardless of the domain of the SIP service you are using.
DNS SRV resource records indicates how to find services for various protocols.



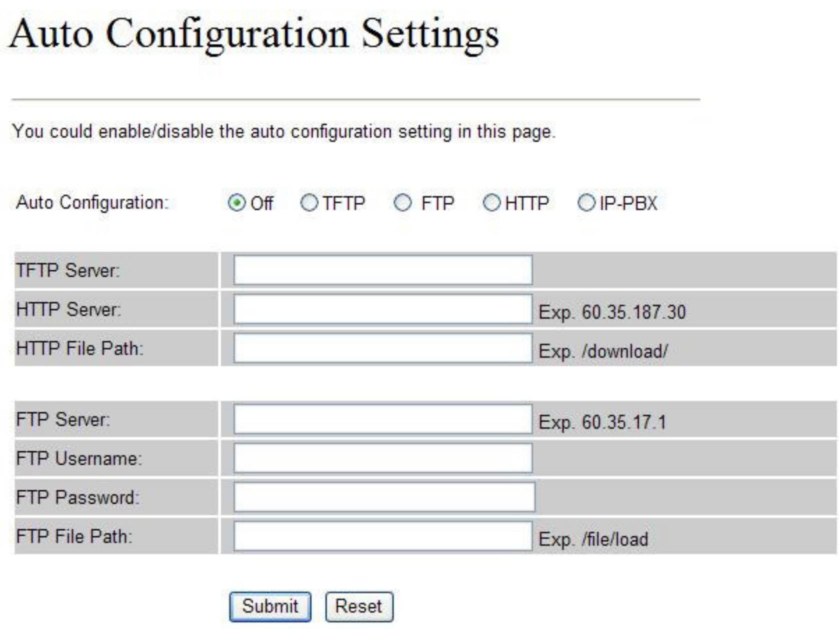
STUN settings

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.



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.

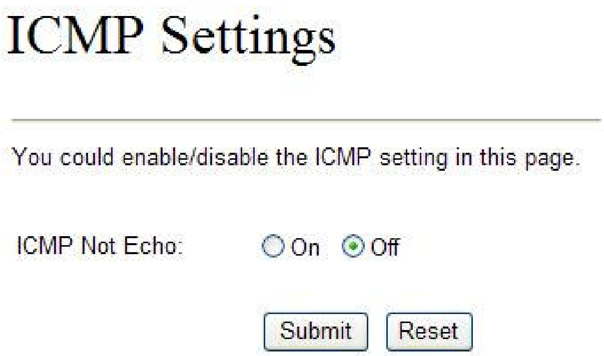


PTT Settings (For VIP-154NT)

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.

ICMP Settings

This page defines the ICMP echo Enable/Disable. This function can disable echo when someone ping this device, it can avoid haker try to attack the device. When you finished the setting, please click the Submit button.

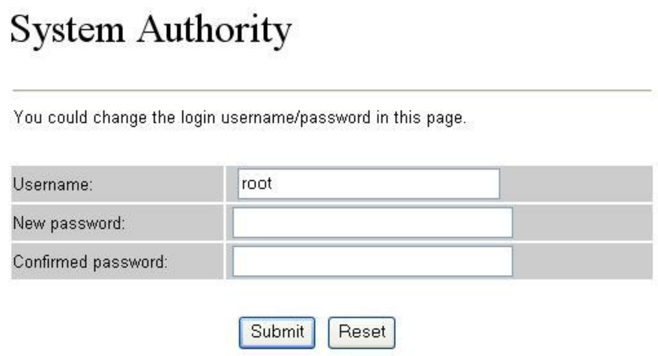


Cancel without to tag

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.

System Authority

In System Authority you can change your login password.



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.

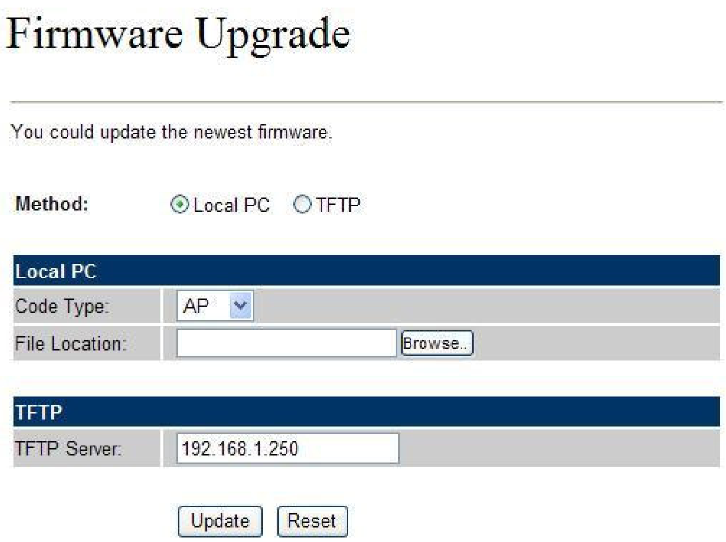
Firmware Upgrade

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.

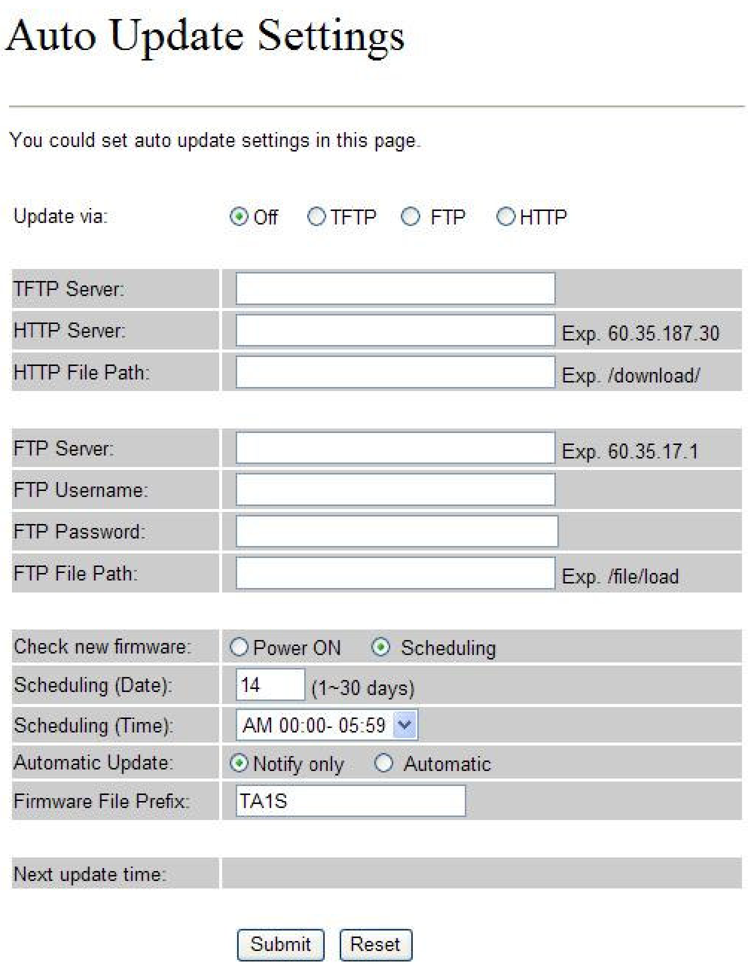


ⓘ**Note**:

After firmware loaded, the unit will be reboot, and Default IP address of the customized firmware: http://192.168.0.1; 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.



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 ATA 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.

ⓘ**Note**:

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.

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.

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)

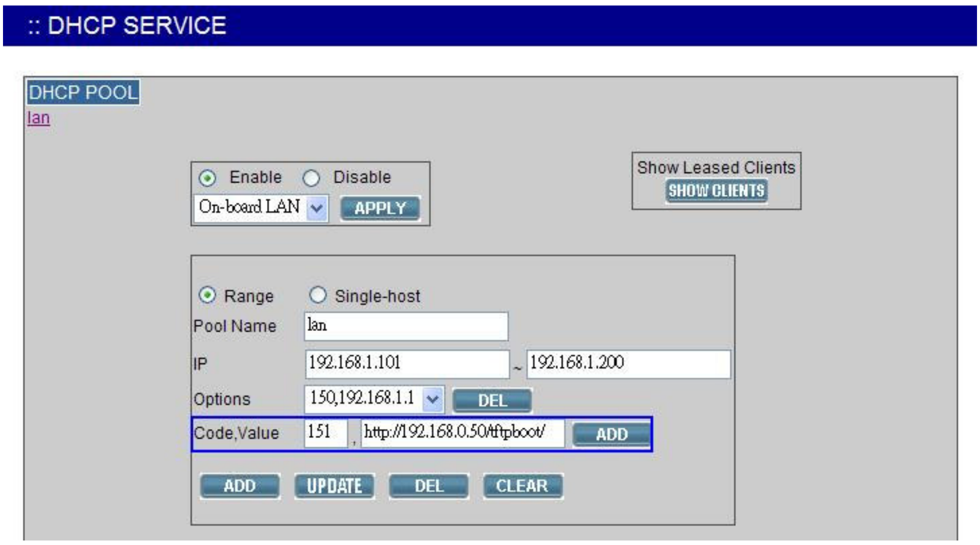
In the following sample, we'll introduce how to integrate the IP Phone with our IP PBX system IPX-2000 via Auto-config feature.

- VIP-154T A IP Address: 192.168.0.1
  Line Number: 1001
- VIP-154T B IP Address: 192.168.0.2
  Line Number: 1002

Machine configuration on the IPX-2000

STEP 1:

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



Code: please insert 151 as the DHCP server option.

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

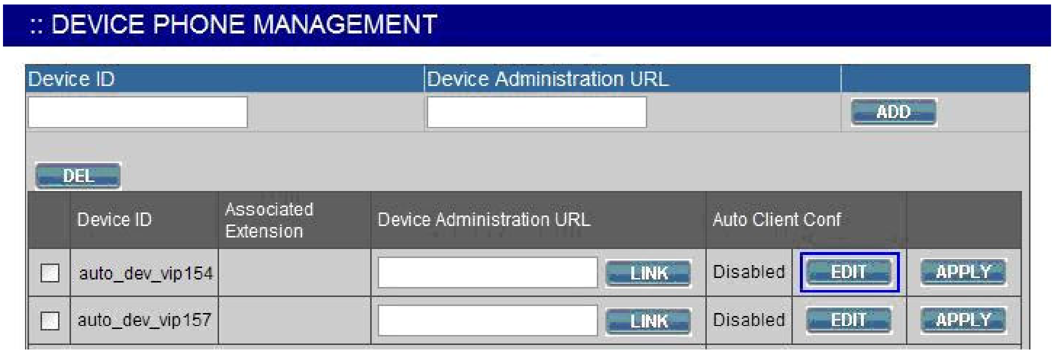
If you'd like to enable auto-config for IP extension features in IPX-2000, please be sure to setup the DHCP option code and the value information.

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

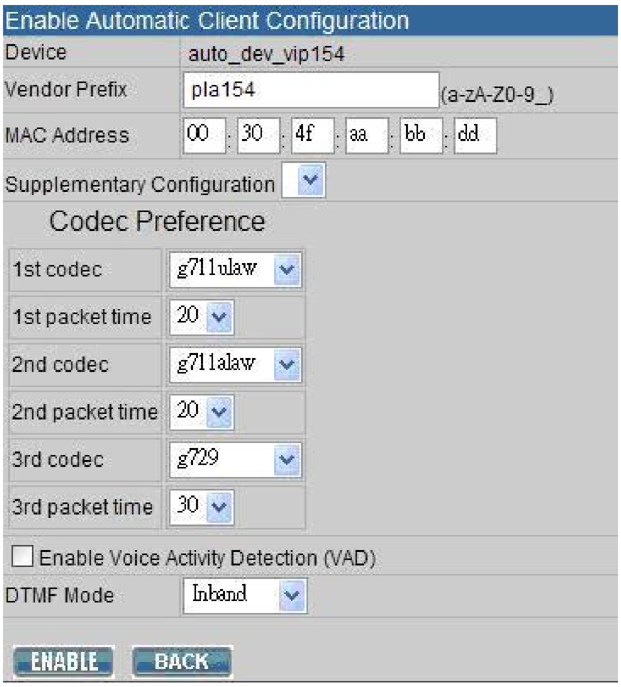
STEP 2:

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.

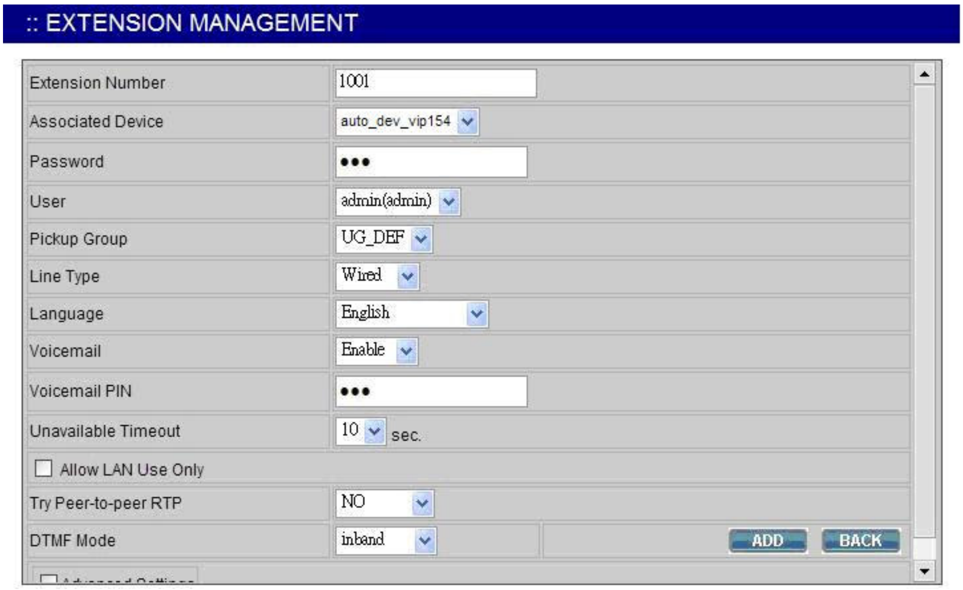


ⓘ**Note**:

The Vendor Prefix of VIP-154 series is **pla154**

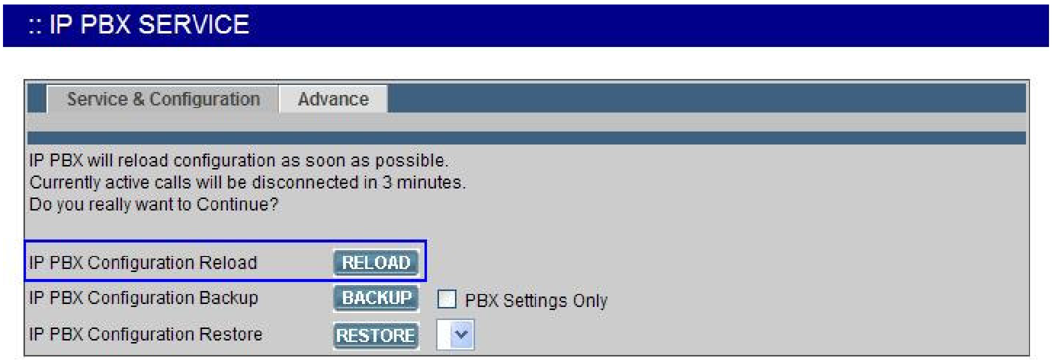
STEP 4:

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



STEP 5:

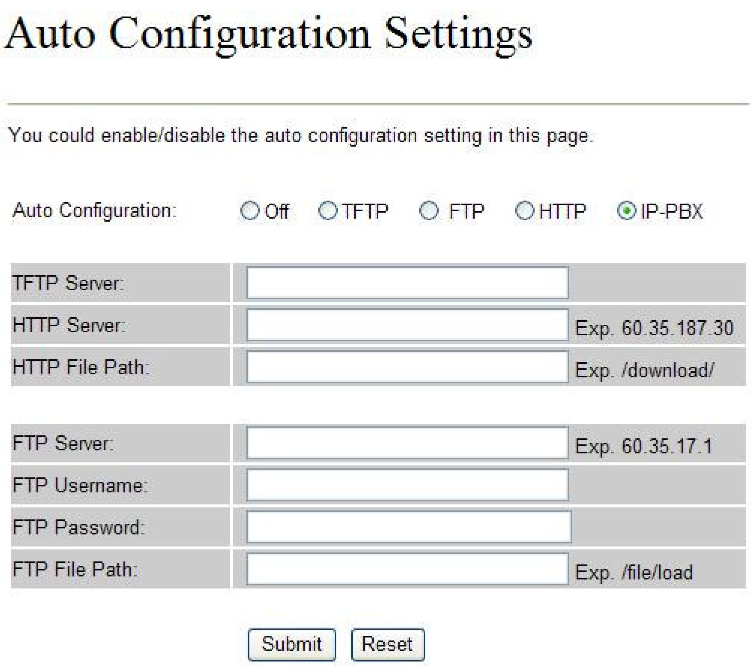
After setting up the parameters, please browse to the **Service → IP PBX service** menu, and press **RELOAD** of IP PBX configuration reload selection for activating the settings.



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.



STEP 7:

After enabled the Auto-config feature, the VIP-154T A 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

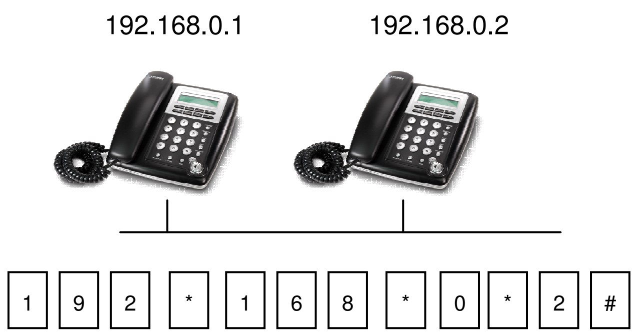
Repeat the same configuration steps on VIP-154T B, and check if the VIP-154T B is successfully registered with the IPX-2000 as one of the IP extensions.

Test the scenario:

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

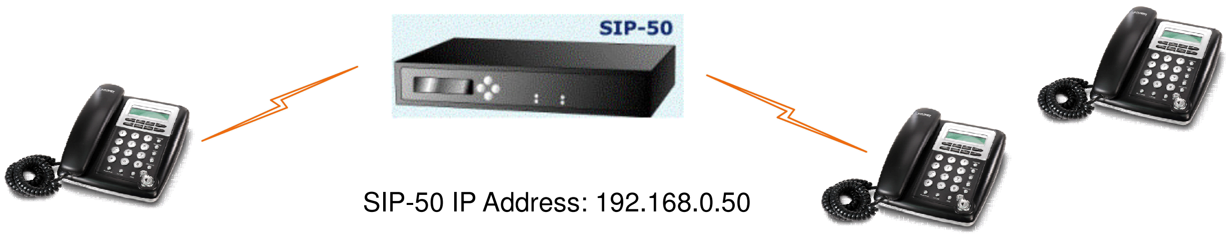
Pick up the VIP-154T A, you should be able to hear the dial tone, press the keypad: 192*168*0*2# shall be able to connect to the VIP-154T B.

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

ⓘ 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

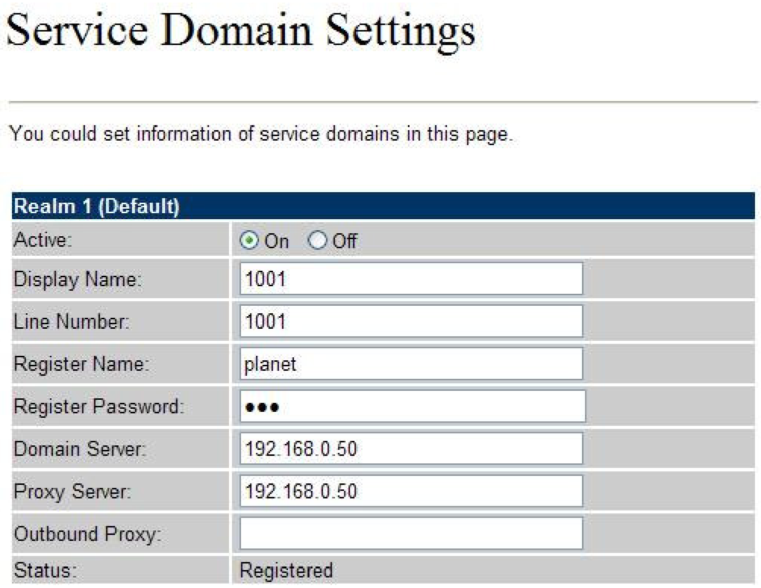
Machine configuration on the VIP-154T:

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:



STEP 3:

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

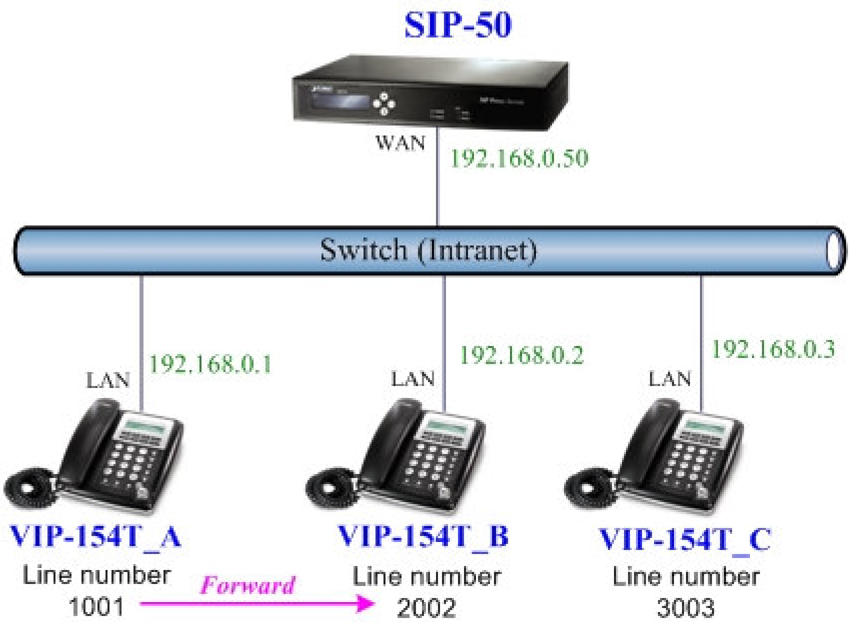
Test the scenario:

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.

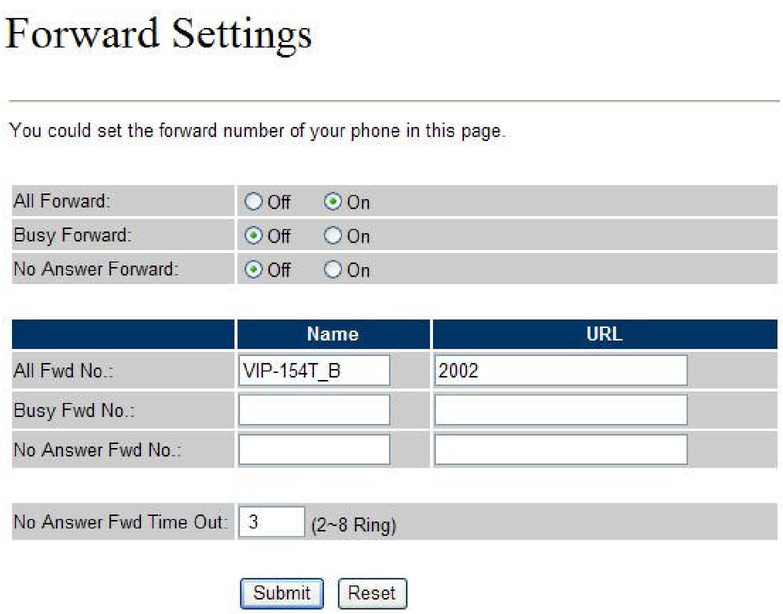
In this example, there are three VIP-154T register to SIP-50 and VIP-154T_A had set Call Forward function to VIP-154T_B. (The detail registration settings of SIP-50 and VIP-154T please refer to the instruction of Case 3)



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

You could set the forward number of your phone in this page.

All Forward:	○ Off ⦿ On
Busy Forward:	⦿ Off ○ On
No Answer Forward:	⦿ Off ○ On

	Name	URL
All Fwd No.:	VIP-154T_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:

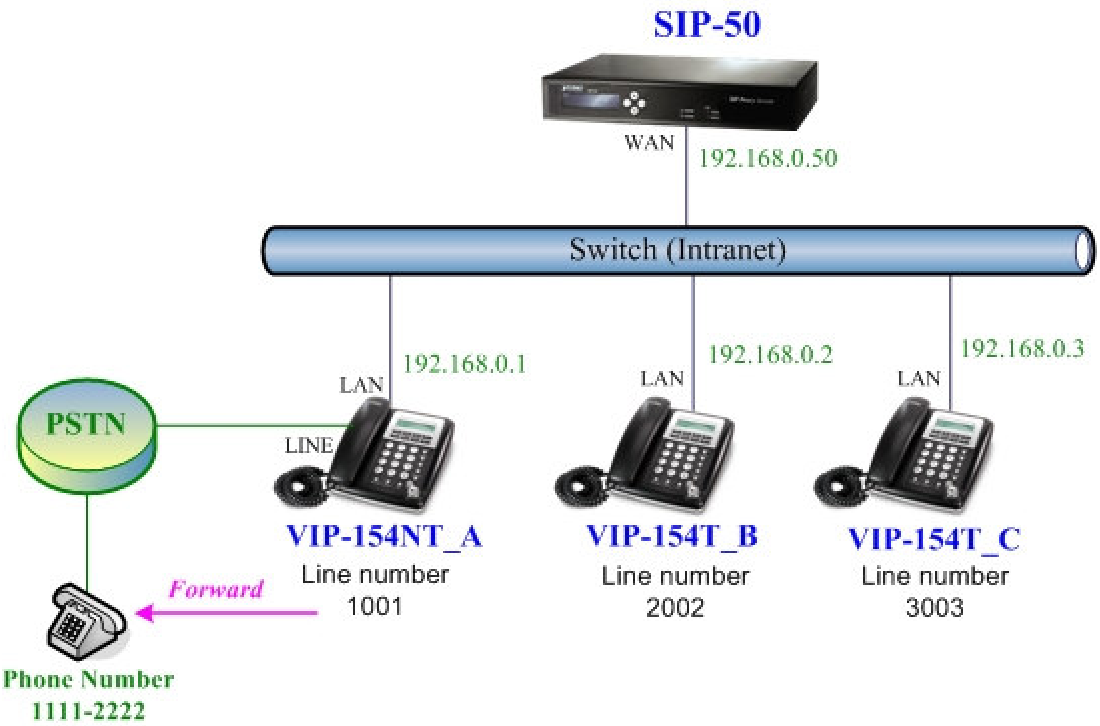
10-19   17:20
AF 2002

Test the scenario:

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

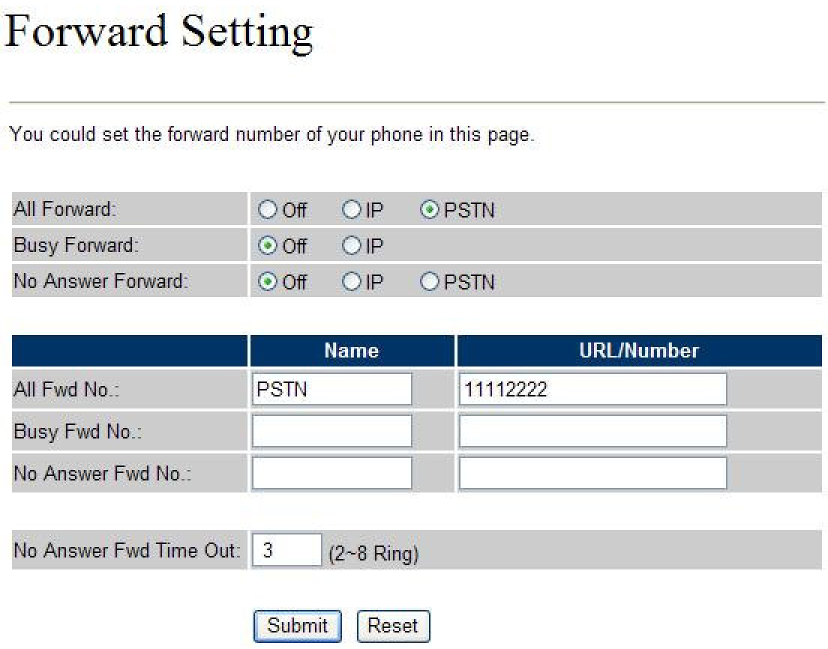
In this example, there are one VIP-154NT and two VIP-154T register to SIP-50. The VIP-154NT_A had set Call Forward function to phone number 1111-2222 (PSTN).



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:



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:

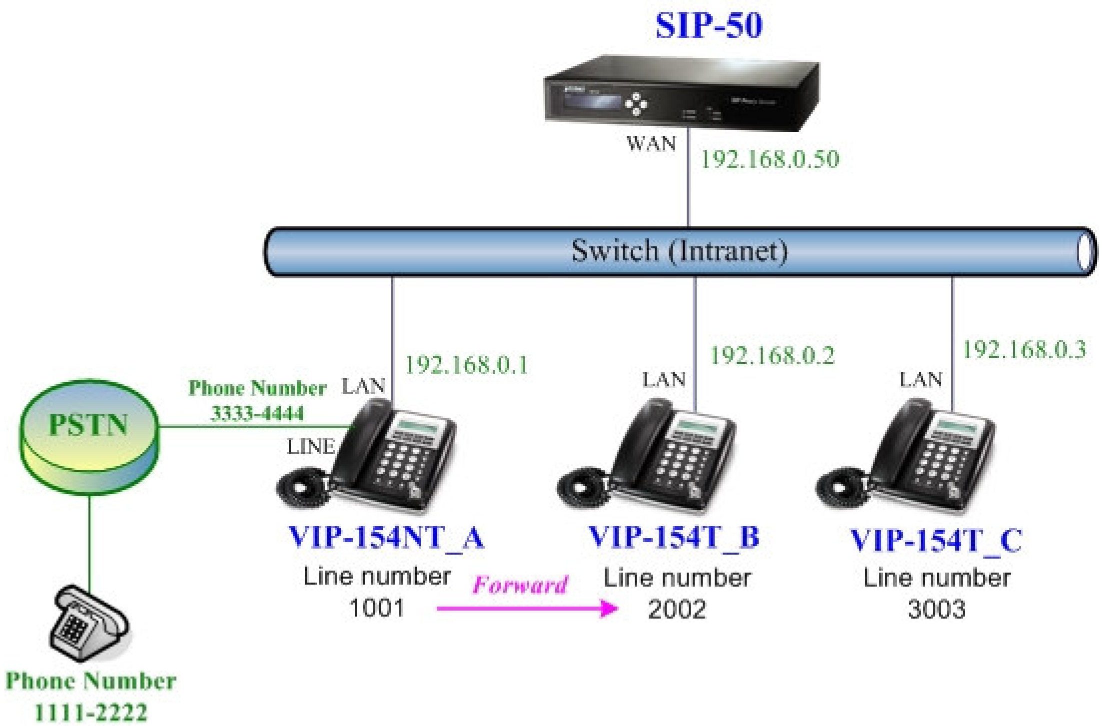
10-19 17:20 AF -11112222

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 SIP-50. 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 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	○ PSTN
Busy Forward:	⊙ Off	○ IP	
No Answer Forward:	⊙ Off	○ IP	○ PSTN

	Name	URL/Number
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:

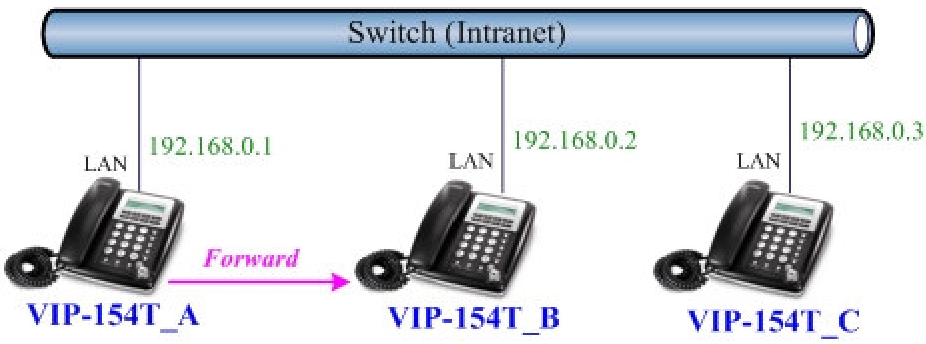
10-19   17:20 AF 2002

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

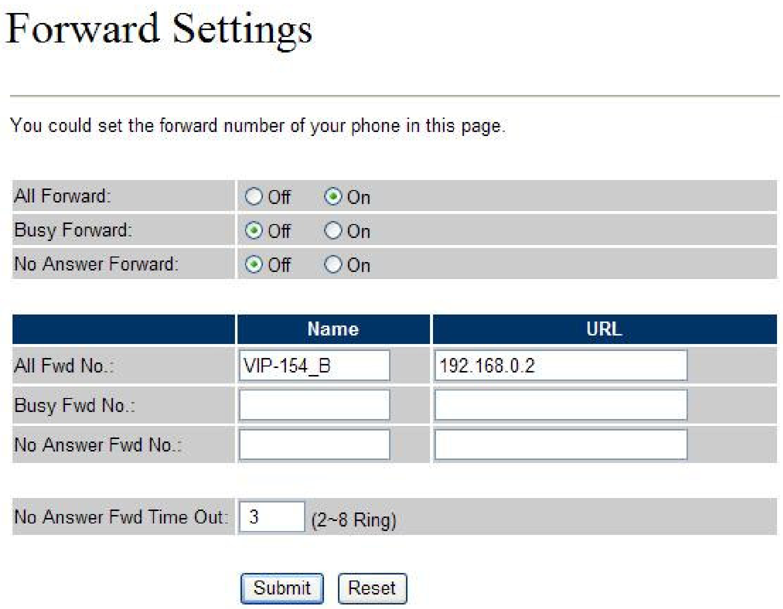
In this example, there are three VIP-154T and connect with Peer to Peer mode. VIP-154T_A had set Call Forward function to VIP-154T_B.



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:



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:

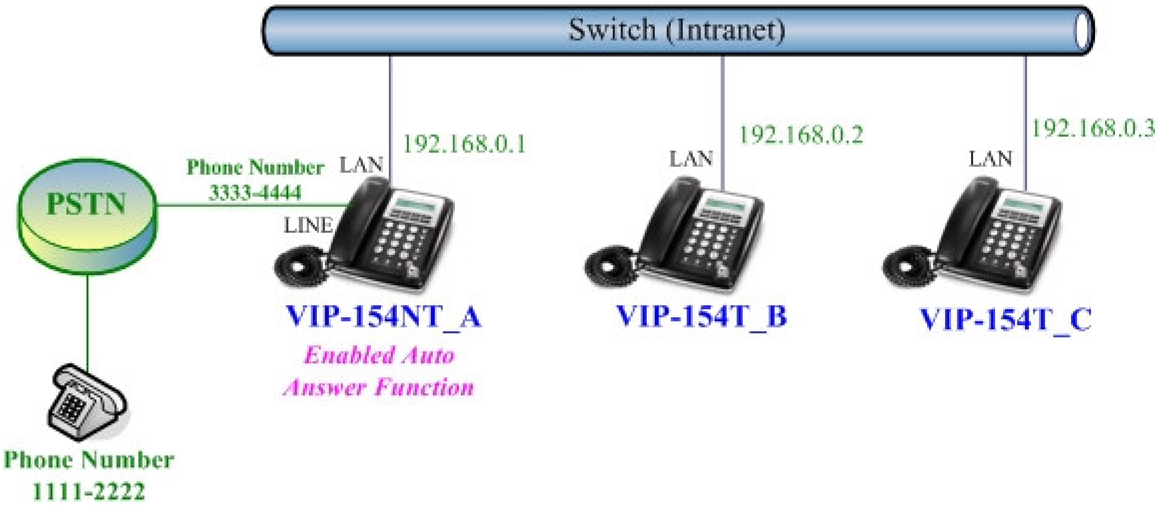
```
10-19   17:20
AF 192.168.0.2
```

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

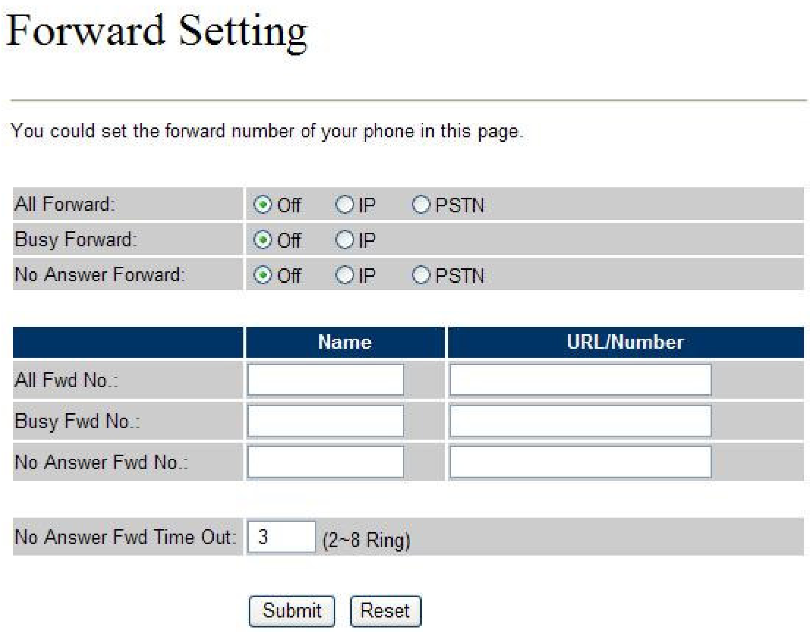
In this example, there are one VIP-154NT and two VIP-154T and connect with Peer to Peer mode. The VIP-154NT_A had set Auto Answer function for forwarding calls to arbitrary telephone. If there have incoming IP calls and VIP-154NT_A doesn’t answer the incoming calls after specific time, the caller will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



Machine configuration on the VIP-154NT:

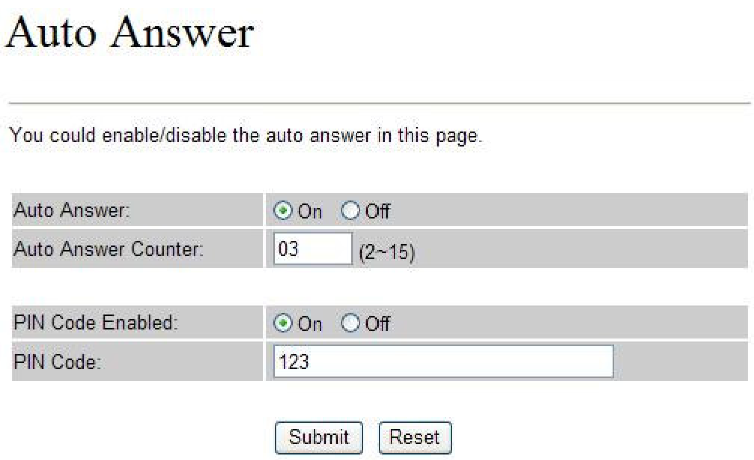
STEP 1:

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:



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:

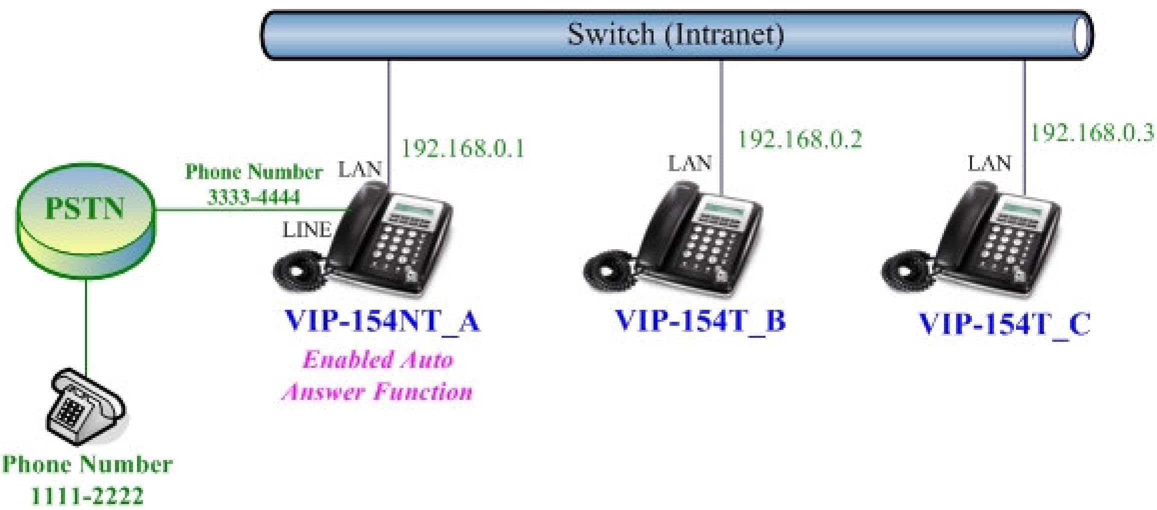


Test the scenario:

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

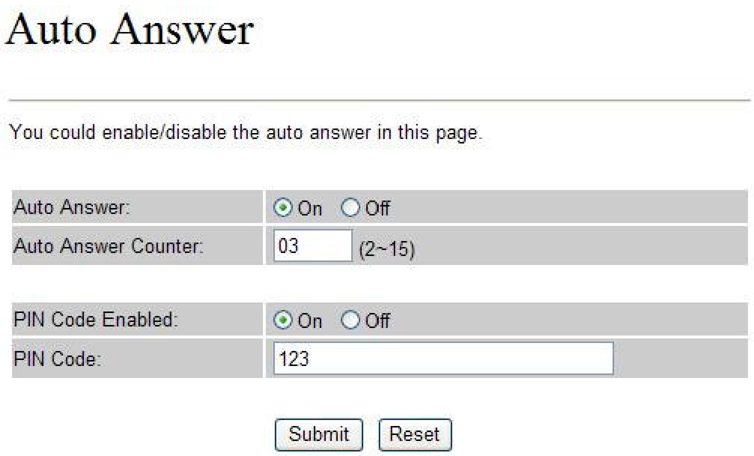
In this example, there are one VIP-154NT and two VIP-154T and connect with Peer to Peer mode. The VIP-154NT_A had set Auto Answer function for forwarding to arbitrary telephone. If there have incoming PSTN calls and VIP-154NT_A doesn't answer the incoming calls after specific time, the caller will hear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



Machine configuration on the VIP-154NT:

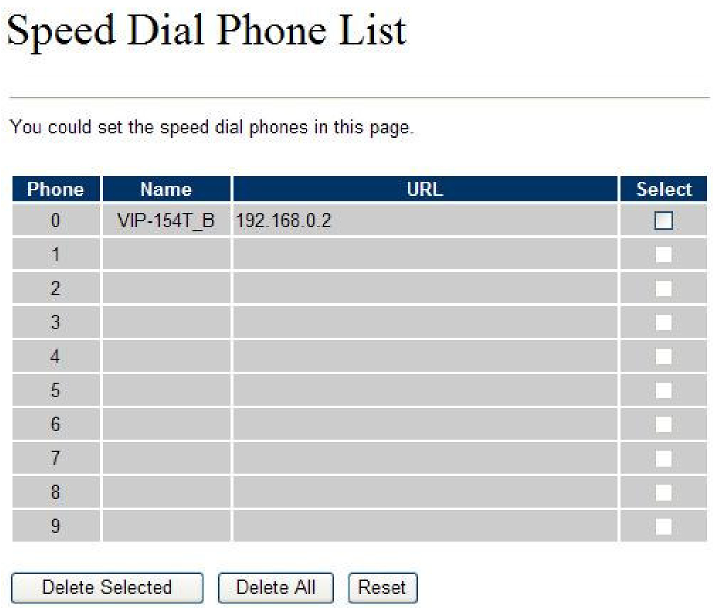
STEP 1:

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:



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:



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).
3. 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 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) G.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) Jitter Buffer		
Supplementary services	Caller ID 3-way conference Immediate (unconditional) call forwarding Busy call forwarding No answer calls forwarding Call Hold/Waiting/Transferring		
Call history	Record incoming call 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		