# VoIP Analog Telephone Adapter

# VIP-156/VIP156PE/VIP-157/VIP-157S

# User's manual

**Version 2.0**

## Copyright



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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 VoIP Analog Telephone Adapter:

Model: VIP-156/VIP-156PE/VIP-157/VIP-57S

Rev: 2.0 (November, 2006)

Part No. EM-VIP_ATAV2

# Table of Contents

# Chapter 1

# Introduction

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## Overview

Based on years of VoIP manufacturing experiences, PLANET Technology VoIP total solutions are known for advanced implementation of standards based telephony with mass deployment capability.

Cost-effective, easy-to-install and simple-to-use, the PLANET VIP-156/VIP-157/VIP-157S VoIP Phone Adapter ("**ATA**" in the following term) converts standard telephones to IP-based networks. The service providers and enterprises offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

With the ATA, home users and companies are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The ATA can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

The ATA includes two alternatively Ethernet interface for Internet (PPPoE, DHCP or Fixed IP), or office LAN connection. With adding the auto-provision feature of our IP PBX product - IPX-2000, the ATA can be seamlessly integrated into the telephony network and be used in consumer and business IP telephony service, no expertise required!

The ATA and our IP PBX system integration are the ideal combination for your office daily communications.

### Product Features

- Feature-rich telephone service over home or office Internet/Intranet connection
- Auto-config feature for flexible, ease-of-use IP PBX system integration
- Cost effective, field proven compatibility, and stability
- Web-based and telephone keypad machine configuration
- Remote machine administration authentication
- Voice prompt for machine configurations

### VoIP Features

- SIP 2.0 (RFC3261) compliant
- Peer-to-Peer / SIP proxy calls
- Voice codec support: G.711, G.723.1, G.729A/G.729B
- T.38 FAX transmission over IP network

- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation (CNG)
- In band, out-of-band, and SIP-info DTMF support
- Support Call Hold, Call Forward, Call Transter, Call Waiting, Call ID display and 3-way conference
- Lifeline support for model VIP-157

## Package Content

The contents of your product should contain the following items:
VoIP Telephone Adapter
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 ATA.

Respective model/descriptions are shown below:

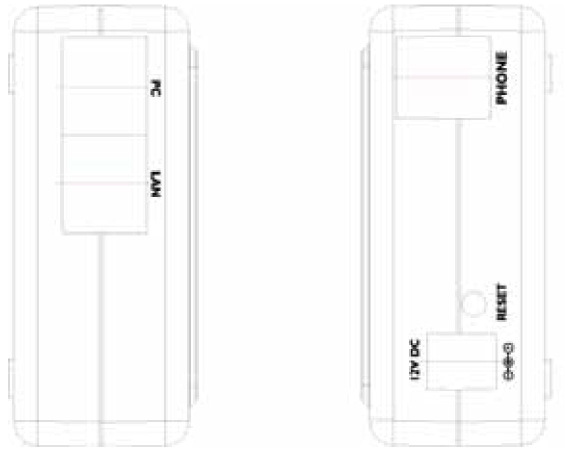
**VIP-156:** `SIP Analog Telephone Adapter`

**VIP-156PE:** `802.3af PoE SIP Analog Telephone Adapter`

**VIP-157:** `1 FXS/ 1 FXO SIP Analog Telephone Adapter`

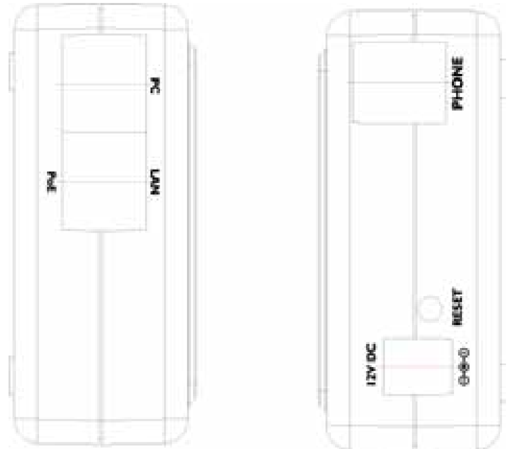
**VIP-157S:** `2-port FXS SIP Analog Telephone Adapter`


**Front Panel of VIP-156**



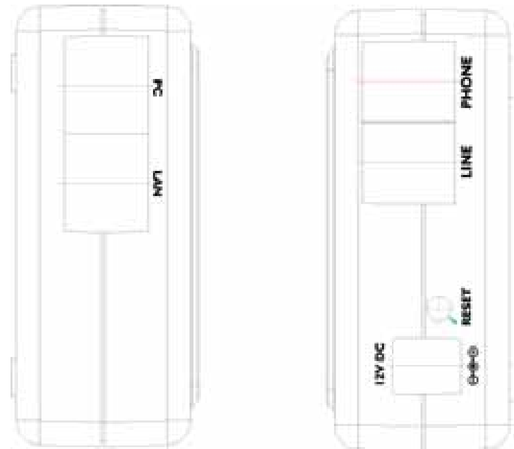
Left / Right Panel of VIP-156

Front Panel of VIP-156PE



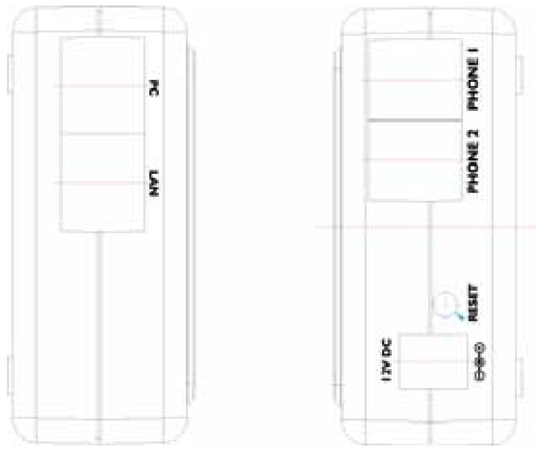
Left / Right Panel of VIP-156PE


**Front Panel of VIP-157**




**Left / Right Panel of VIP-157**


**Front Panel of VIP-157S**




**Left / Right Panel of VIP-157S**

## LED Display & Button

| | | |
|---|---|---|
| **1** | **PC** | RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable |
| **2** | **LAN** | RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.<br><br>The **LAN** interface also can be connected with 802.3af PoE switch or converter for power supply (**VIP-156PE**) |
| **3** | **12V DC** | 12V DC Power input outlet |
| **4** | **Reset** | Reset to the factory default setting |

**Note**

Machine default IP is http://192.168.0.1. Press **RESET** button on rear panel over 5 seconds will reset the VoIP Phone Adapter to factory default value. (Except speed dial and call forward settings)

**LED display of VIP-156 / VIP-156PE**

| LED Indicators | Descriptions |
|---|---|
| PWR | Power is supplied to the device. |
| STATUS | The Status LED will be flashing when the machine is operational |
| LNK/ACT | **OFF**: the device is connected to LAN at 10Mb/s.<br>**ON**: the device is connected to LAN at 100Mb/s. |
| RING | **OFF**: the phone is idle.<br>**ON**: the phone is in use (offhook).<br>**Blinking**: the phone is ringing. |

**LED display of VIP-157 / VIP-157S**

| LED Indicators | Descriptions |
|---|---|
| STATUS | The Status LED will be flashing when the machine is operational |
| LNK/ACT | **OFF**: the device is connected to LAN at 10Mb/s.<br>**ON**: the device is connected to LAN at 100Mb/s. |
| RING 1 | **OFF**: the phone is idle.<br>**ON**: the phone is in use (offhook).<br>**Blinking**: the phone is ringing. |
| RING 2 | **OFF**: the phone is idle.<br>**ON**: the phone is in use (offhook).<br>**Blinking**: the phone is ringing. |

# Chapter 2

# Preparations & Installation

## Physical Installation Requirement

This chapter illustrates basic installation of ATA analog Phone Adapter ("**ATA**" in the following term)

- 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

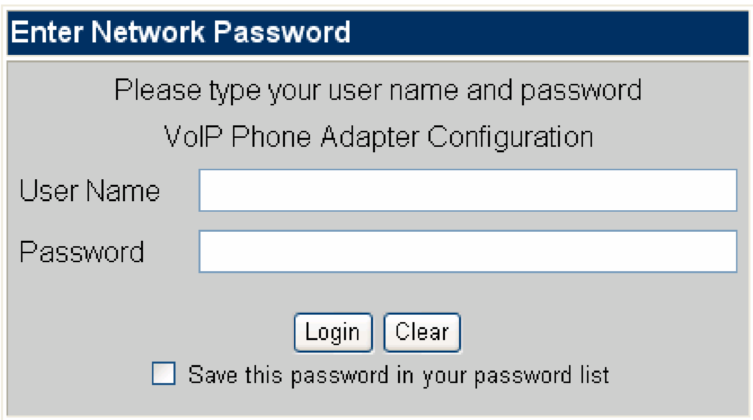
PLANET ATA provides GUI (Web based, Graphical User Interface) for machine management and administration.

### Web configuration access

To start ATA 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 LAN interface IP address of ATA is **192.168.0.1**. You may now open your web browser, and insert http://192.168.0.1 in the address bar of web browser to logon ATA web configuration page.




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

**Note**

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Please locate your PC in the same network segment (**192.168.0.x**) of ATA. 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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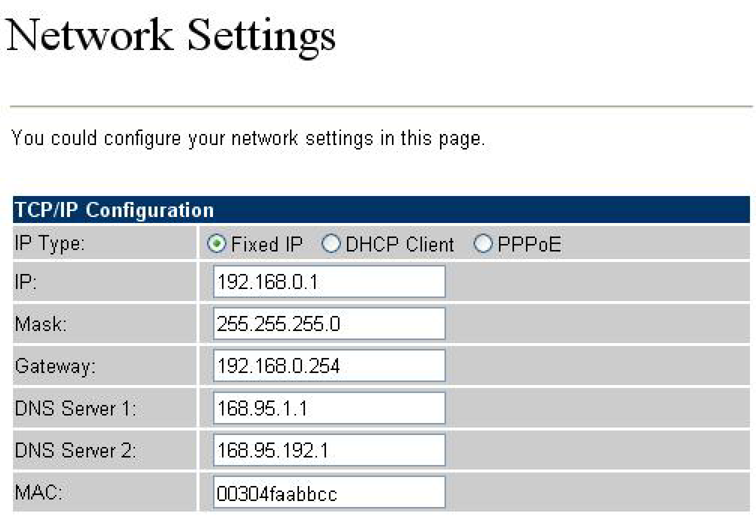
## LAN IP address configuration via web configuration interface

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**" --> "**Network settings**" configuration menu:




### Parameter Description

**IP address** LAN IP address of ATA

**Default**: 192.168.0.1

**Subnet Mask** LAN mask of ATA

**Default**: 255.255.255.0

**Default Gateway** Gateway of ATA

**Default**: 192.168.0.254

### Network settings via Keypad commands

The ATA series phone adapters support telephone keypad configurations, please connect analog telephone set and refer to the following table for machine network configurations.

| IVR Menu Choice | Machine operation | Parameter(s) | Notes: |
|---|---|---|---|
| **#111#** | Set DHCP client | None | ATA will change to DHCP Client |
| **#112xxx*xxx*xxx*xxx#** | Setup Static IP Address | Use the * (star) key when entering a decimal point. | DHCP will be disabled and system will change to the Static IP type. |
| **#113xxx*xxx*xxx*xxx#** | Set Network Mask | Use the * (star) key when entering a decimal point. | Must set Static IP first. |
| **#114xxx*xxx*xxx*xxx#** | Set Gateway IP Address | Use the * (star) key when entering a decimal point. | Must set Static IP first. |
| **#115xxx*xxx*xxx*xxx#** | Set Primary DNS Server | Use the * (star) key when entering a decimal point. | Must set Static IP first. |
| **#136#** | Enable auto-config mode | None | For PLANET IPX -2000 IP PBX System |
| **#137#** | Disable auto-config mode | None | For PLANET IPX -2000 IP PBX System |
| **#190#** | Unlock | None | Must unlock the protect function before set up network settings via keypad. |
| **#191#** | Lock | None | The system will be lock and can't set up network settings via keypad. |
| **#195#** | Reboot | None | The system will reboot automatically. |
| **#198#** | Factory Reset | None | The system will be reset to factory default value and reboot automatically. |
| **0＊** | To switch PSTN mode | None | VIP-157 only |

ⓘ Hint

In initially firmware version: **V1.0**, the VIP-157 default machine operation is at **PSTN** mode. If user want to make a VoIP phone call, please press the **"*"** key to switch to PSTN mode.

After latest firmware version: **V2.0 or higher**, the VIP-157 is at **VoIP** mode. If user want to make a PSTN phone call, please press the **"0*"** key to switch to PSTN mode.

Following keypad commands can be used to display the network settings enabled on ATA via voice prompt.

| IVR Menu Choice | Machine operation | Notes: |
|---|---|---|
| **#120#** | Check IP Address | IVR will announce the current IP address of the ATA. |
| **#121#** | Check network connection Type | IVR will announce if DHCP in enabled or disabled. |
| **#122#** | Check the Phone Number | IVR will announce current enabled VoIP number. |
| **#123#** | Check Network Mask | IVR will announce the current network mask of the ATA. |
| **#124#** | Check Gateway IP Address | IVR will announce the current gateway IP address of the ATA. |
| **#125#** | Check Primary DNS Server Setting | IVR will announce the current setting in the Primary DNS field. |
| **#128#** | Check Firmware Version | IVR will announce the version of the firmware running on the ATA. |

ⓘ Hint

Please contact your Internet service provider to obtain the Internet access type, and select the proper network settings in ATA to establish the network connections.

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.

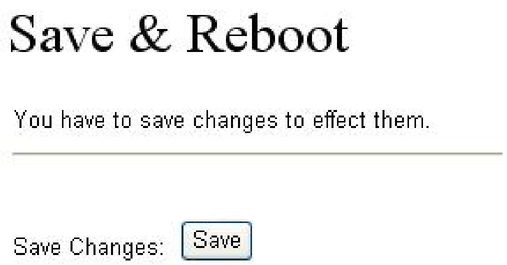
| **Connection Type** | **Data required.** |
|---|---|
| **Fixed IP** | In most circumstances, it is no need to configure the DHCP settings. |
| **DHCP clinet** | The ISP will assign IP Address, and related information. |
| **PPPoE** | The ISP will assign PPPoE username / password for Internet access,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

ⓘ 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 VoIP router parameters will take effective after modifications, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the VoIP Phone Adapter, 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 Phone Adapter from web browser

The ATA 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 ATA

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 ATA web configuration, you must have one of these web browsers installed on computer for management

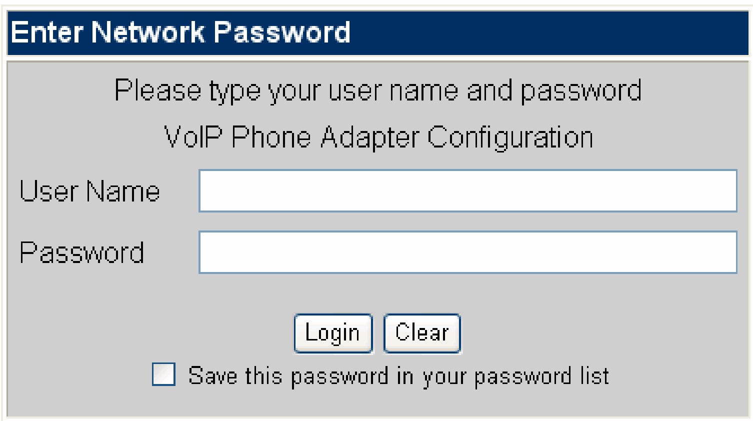
- Microsoft Internet Explorer 4.01 or higher with Java support

## Manipulation of ATA via web browser

**Log on ATA via web browser**

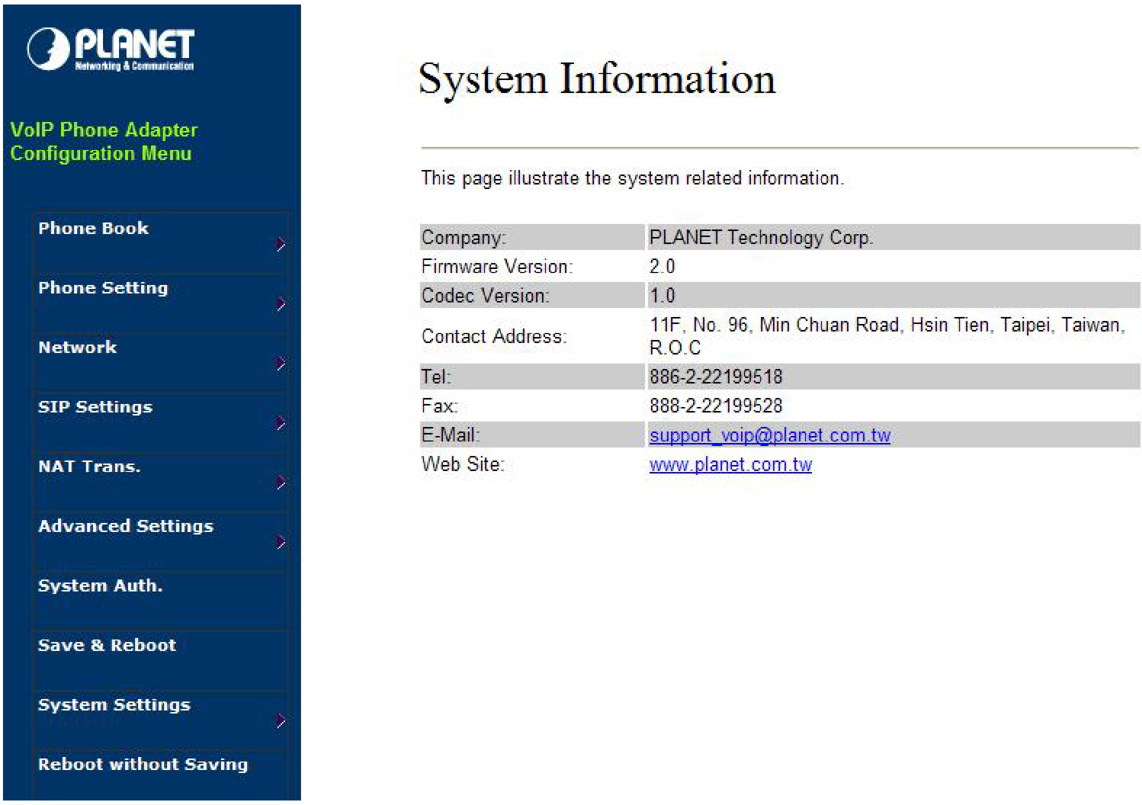
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon Phone Adapter web configuration page.

Phone Adapter will prompt for logon username/password: ***root*** / ***null (no password)***




**ATA log in page**

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




***VoIP Phone Adatper main page***

# Chapter 4

# VoIP Telephone Adapter Configurations

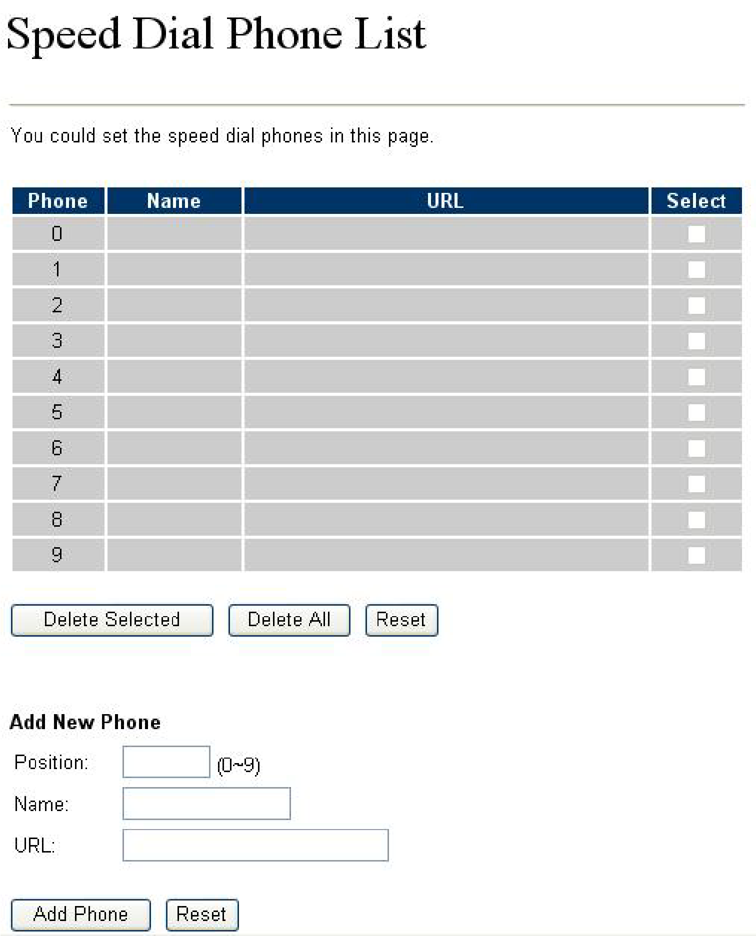
## Speed Dial Setting

In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

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.

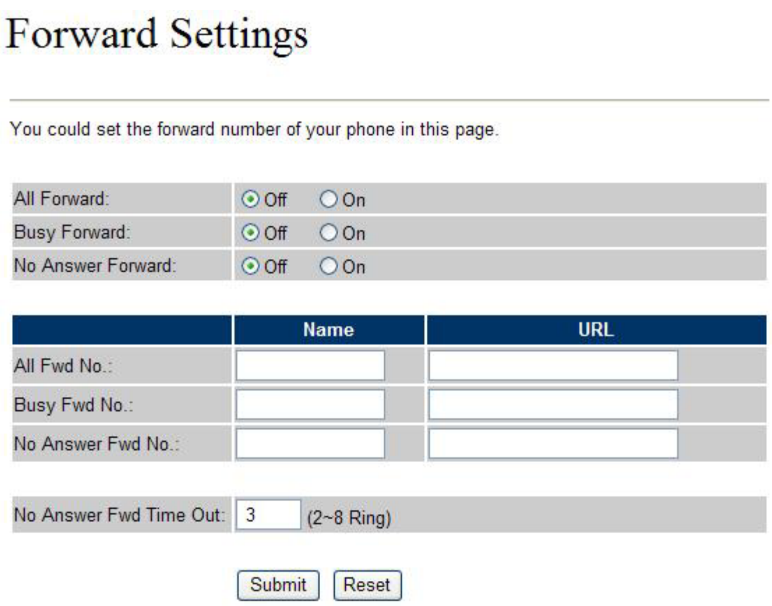


## Call Forward Setting

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.

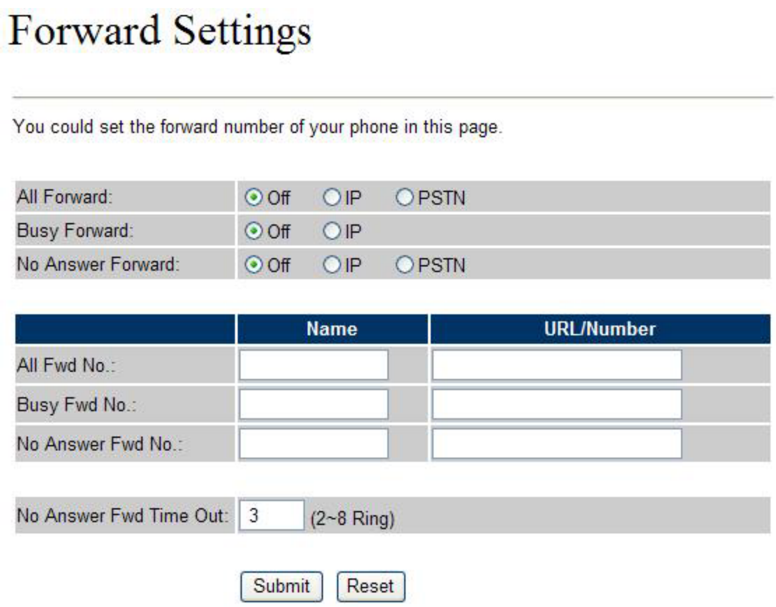
| | |
|---|---|
| **All Forward** | All incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose. |
| **Busy Forward** | If you are on 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.<br>When you finished the setting, please click the Submit button. |

**Note:** When you finished the setting, please click the Submit button.




***Call Forward function for VIP-156/VIP-156PT/VIP-157S***

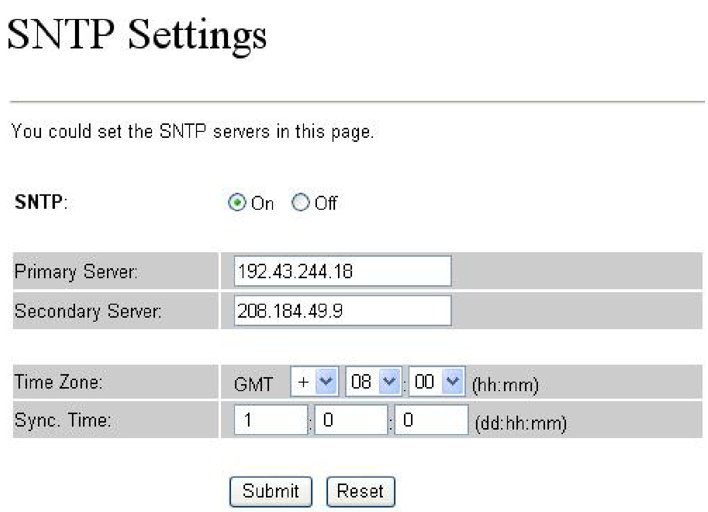
**Call Forward to PSTN (VIP-157):** VIP-157 not only supports Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, and then input the name and the PSTN number in URL/Number field.




***Call Forward function for VIP-157***

## SNTP Setting

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.

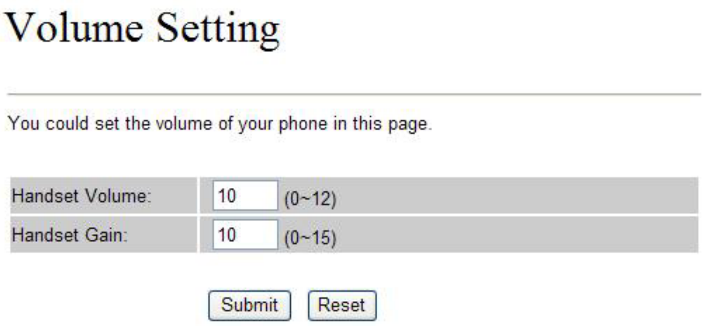



## Volume Setting

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.

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

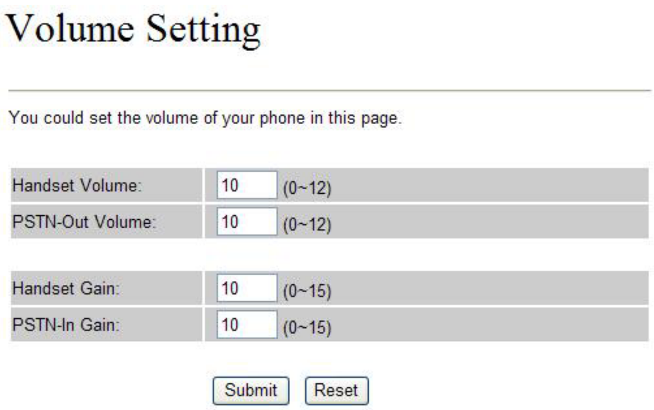



***Volume Settings for VIP-156/VIP-156PE/VIP-157S***

Beside the above settings, VIP-157 also can set the volume of PSTN.

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

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




***Volume Settings for VIP-157***

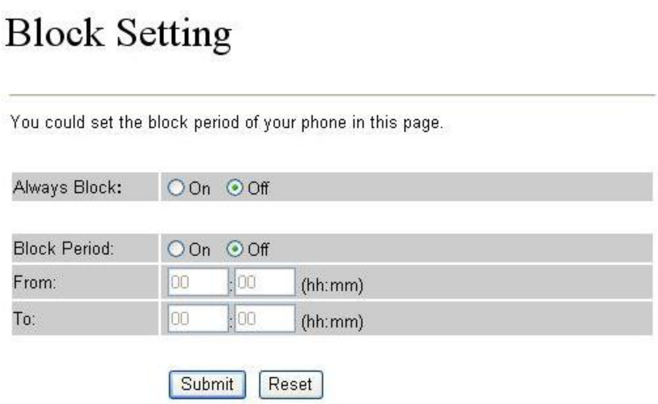
## Block Setting

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.

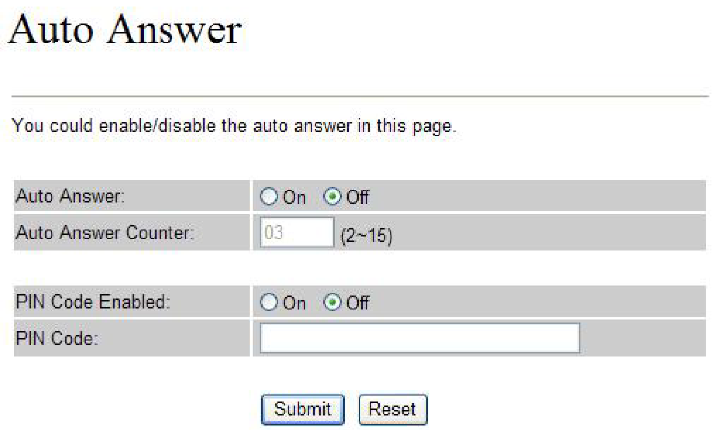


## Auto Answer (For VIP-157)

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-157 can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the VIP-157 can let user to redial to IP Phone number.

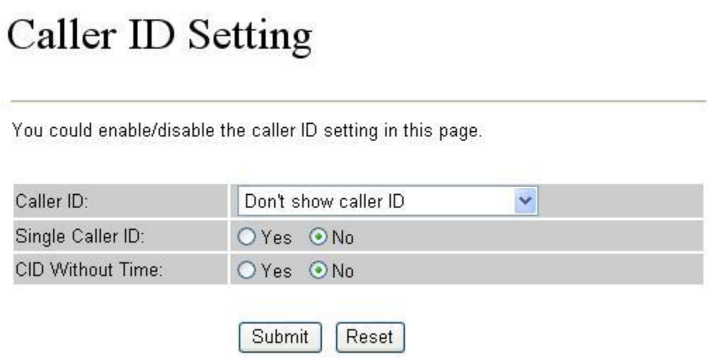
**Auto Answer Counter** is to set after the ring count met the number you set then the auto answer will enable.

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.



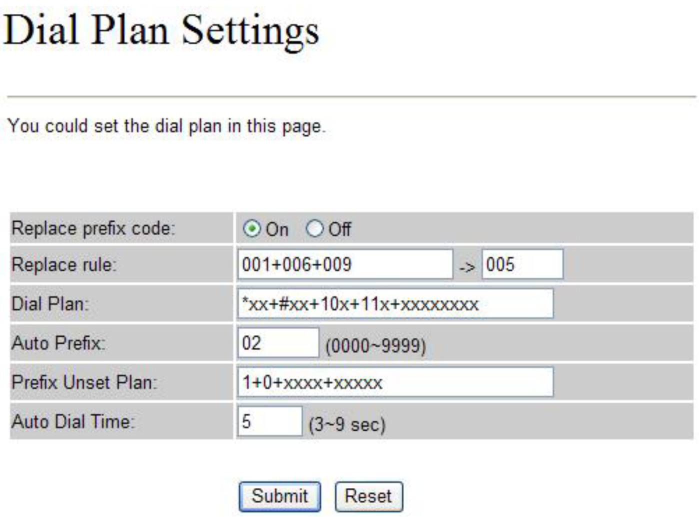

## Caller ID Setting

This page defines the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.



## Dial Plan Setting

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.




**Symbol Explan:**

| Digits | Description |
|---|---|
| X or X | 0, 1, 2, 3, 4, 5, 6, 7, 8, 9 |
| + | or |

**Replace rule**: If replace prefix code function is ON and prefix number is matched with rule then 005 will replace prefix.

**Auto Dial Time**: Stop dialing after seconds then send dial number out.

**Dial Plan**: When match with pattern then send dial number out but if fisrt digit is '0' then dial plan will be ignored.

**For Example:**

| Digits | Description |
|---|---|
| *xx | If matched with one of *00,*01….*99 then will send number out |
| #xx | If matched with one of #00,#01….#99 then will send number out |
| 10x | If matched with one of 100,101….109 then will send number out |
| 11x | If matched with one of 110,111….119 then will send number out |
| Xxxxxxxx | If dial with 8 digits then send number out |

**Auto Prefix**: Number for add before dial number.

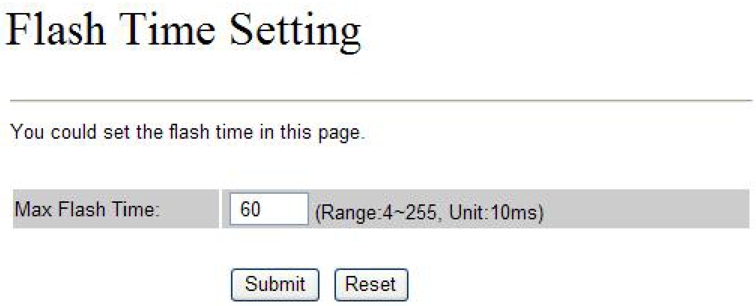
**Prefix Unset Plan**: When first digit or dial numeb match with pattern then ignore auto prefix.

| Digits | Description |
|---|---|
| 0 | Iignore auto prefix if first digit is ‘0’ |
| 1 | Ignore auto prefix if first digit is ‘1’ |
| xxxx | dial numbers are 4 digits ignore auto prefix |
| Xxxxx | dial numbers are 5 digits ignore auto prefix |

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

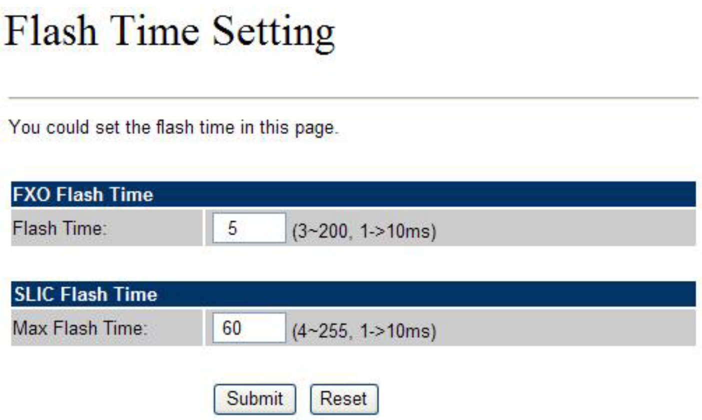
## Flash Time Setting

When you use the PSTN Phone 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 for VIP-156/VIP-156PT/VIP-157S***

Beside the above settings, VIP-157 also can set the flash time of FXO port.



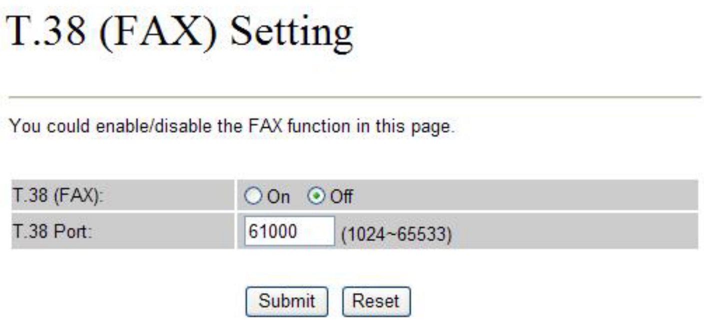

***Flash Time Settings for VIP-157***

## Call Waiting Setting

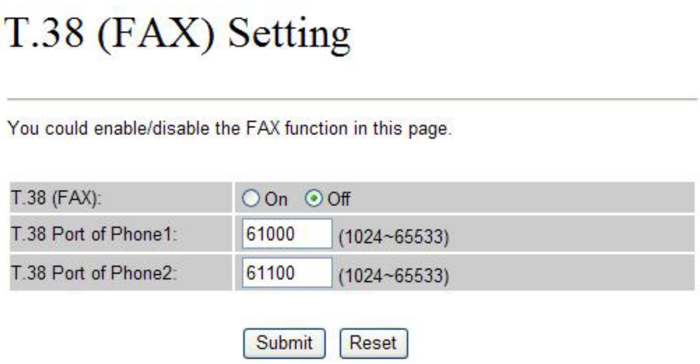
When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear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


## T.38 (FAX) Setting

This page defines the T.38 (FAX) setting function. You can Enable/Disable the T.38 function, and can modify the T.38 transmission port of each FXS port.



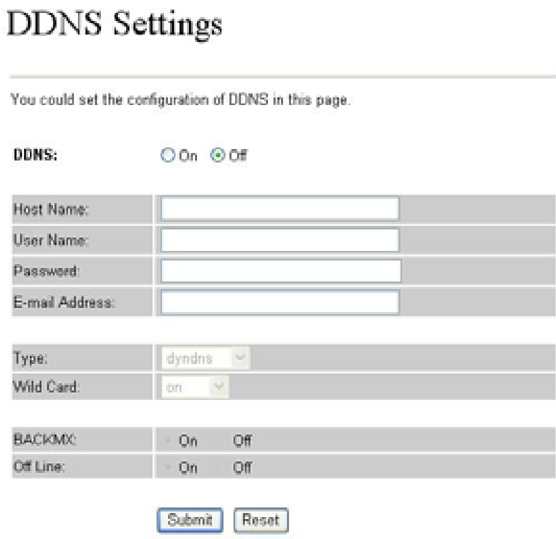

***T.38 (FAX) Settings for VIP-156/VIP-156PE/VIP-157***




***T.38 (FAX) Settings for VIP-157S***

## DDNS Setting

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 Setting

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 ATA. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

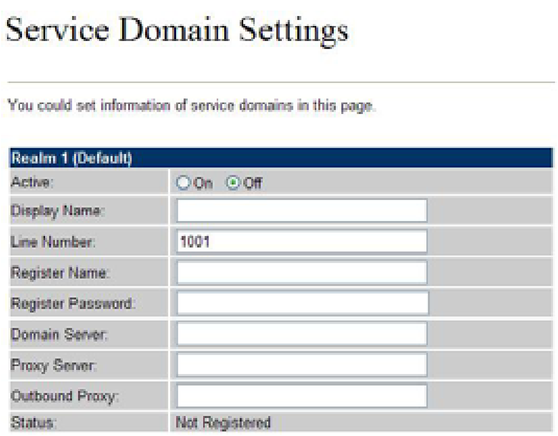
First you need click Active to enable the Service Domain, then you can input the following items:

| | |
|---|---|
| **Display Name** | you can input the name you want to display. |
| **Line number** | 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.

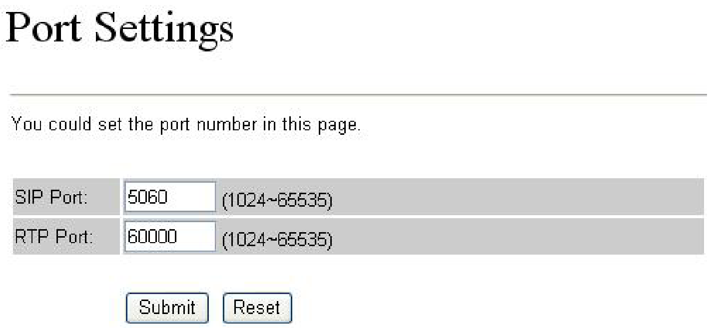
If you have more than one SIP account, you can following the steps to register to the other ISP.

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



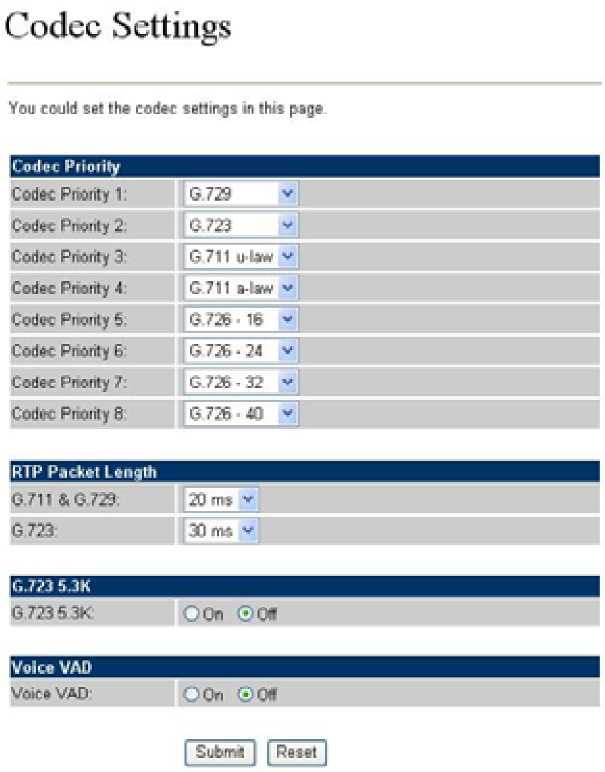
## Port Setting

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.



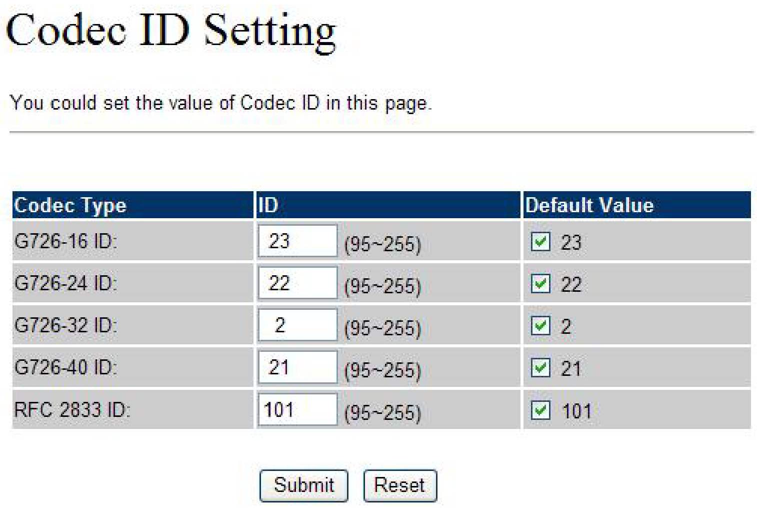

## Codec Settings

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.



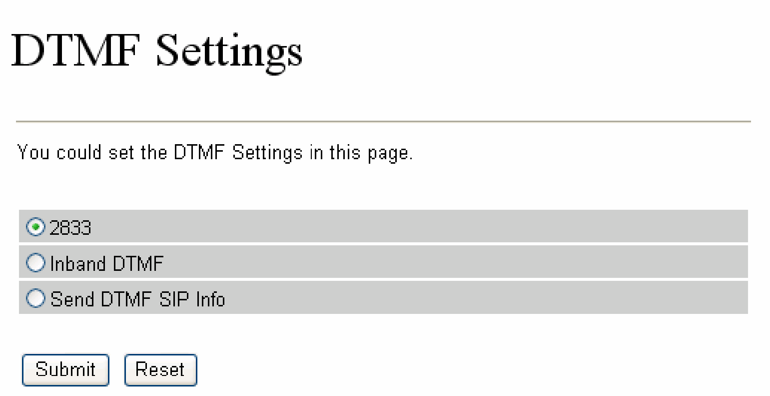

## Codec ID Setting

This page defines the Codec ID. Sometimes 2 VoIP devices 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, and then you can change your Codec ID. When you finished the setting, please click the Submit button.




## DTMF Setting

This page defines the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

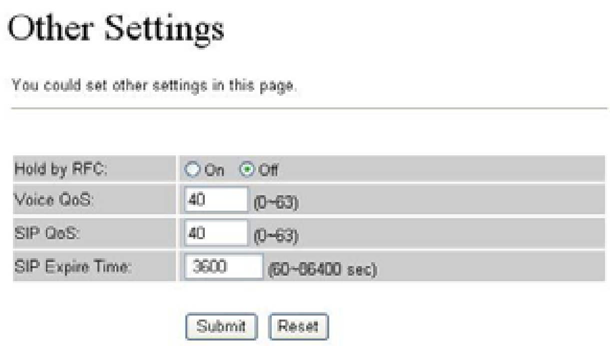



## RPort Setting

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.

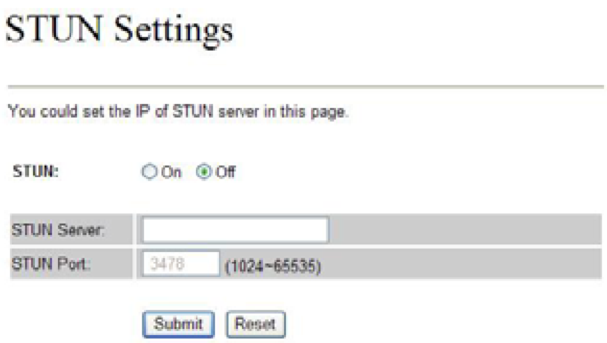
## Other Setting

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.



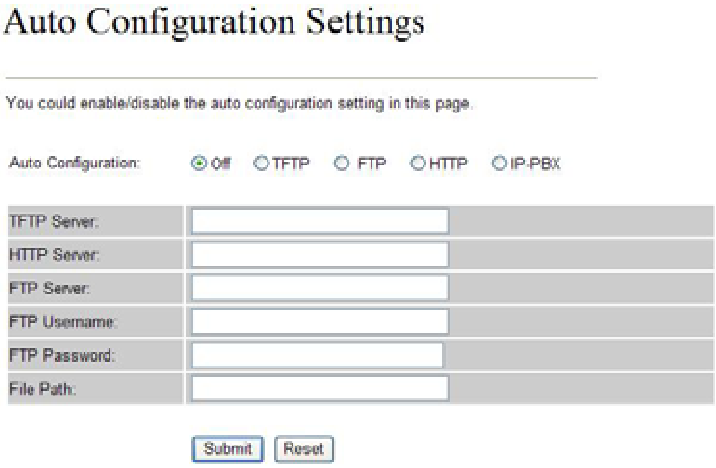

## STUN Setting

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.



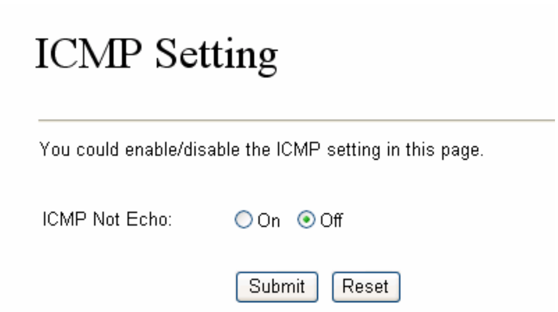

## Auto Configuration

This page defines the Auto Configuration (Auto Provision) setting. ATA 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.




## ICMP Setting

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.




## PTT Setting

In PTT Settings is for you to set the Country, different country will have different settings in FXS inter face.


***PTT Settings for VIP-156/VIP-157S***

Beside the above settings, VIP-157 also can set country of FXO port.


***PTT Settings for VIP-157***

## 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 Phone Adapter, You have to click the Save button. After you click the Save button, the Phone Adapter will automatically restart and the new setting will effect.


## Firmware Upgrade

In Firmware Upgrade function you can update new firmware via HTTP in this page. You can ugrade the firmware by the following steps:

Select 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 Phone Adapter then click the Update button.


**ⓘNote:**

For technological consideration, we've strongly suggested refering to the following upgrade methods for update your IP Phone.

## Firmware Upgrade methods:

Please find the firmware of IP Phone, and be sure to check the firmware upgrade steps to load the firmware into machine properly for revolutions.

a) Log in IP Phone via Microsoft Internet explorer web browser, and insert http://ATA's IP address/update.htm in the address bar.

b) Select update "All ROM", and browse to the firmware location.

c) Please find the firmware (decompress and find the *.rom file for upgrade)

d) After firmware loaded, the unit will be reboot, and loaded with factory default values.

e) Default IP address of the customized firmware: http://192.168.0.1; login name/password: **root/null (no password)**

## Reset to Default

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

## Reboot without saving

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

# Appendix A   Voice Communication Samples

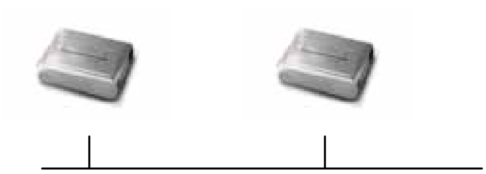
There are several ways to make calls to desired destination in ATA. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

## Case 1: ATA to ATA connection via IP address

Assume there are two ATAs in the network the IP address are 192.168.0.1, 192.168.0.2

Analog telephone sets are connected to the **phone** (RJ-11) port of ATAs respectively

192.168.0.1 192.168.0.2



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

### Test the scenario:

1. Pick up the telephone set on ATA A.
2. Press the keypad: **192*168*0*2#** shall be able to connect to the ATA B.
3. Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

## Case 2: (Peer-to-Peer mode) VIP-157S Port 1 to Port 2 communications

Supposing one VIP-157S connects to two telephones, just pick up phone 1 and dial '**192*168*0*1**5062**', phone 2 will ring.

Analog telephone sets are connected to the phone (RJ-11) ports of VIP-157S respectively


| | | | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 1 | 9 | 2 | * | 1 | 6 | 8 | * | 0 | * | 1 | * | * | 5 | 0 | 6 | 2 | # |

### Test the scenario:

1. Pick up the telephone set on VIP-157S port 1, and you should be able to hear the dial-tone
2. Press the keypad: **192*168*0*1**5062#** shall be able to connect to the VIP-157S port 2
3. Then the telephone set in VIP-157S port 2 should ring. Please repeat the same dialing steps on port 2 to establish the first voice communication from VIP-157S

ⓘ Hint

- In default machine operation, the VIP-157 is VoIP mode. If you want to make a PSTN phone call, press the "**0***" key to switch to PSTN mode.
- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with a "**#**".
- If the ATAs are installed behind a NAT/firewall/IP sharing device for Peer-to-Peer VoIP application, please make sure the NAT device support SIP applications, and suitable settings should be applied to the NAT device to enable the SIP communications before making calls
- [**VIP-157S**] in PLANET ATA series products, to connect to remote ATA, press the keypad in the following sequence to connect to the remote VIP-157S port 2: **[remote ATA IP address]**5062**, for example: **192*168*0*2**5062**

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


- ATA A IP Address: 192.168.0.1
  Line Number: 1001
- ATA B IP Address: 192.168.0.2
  Line Number: 2002

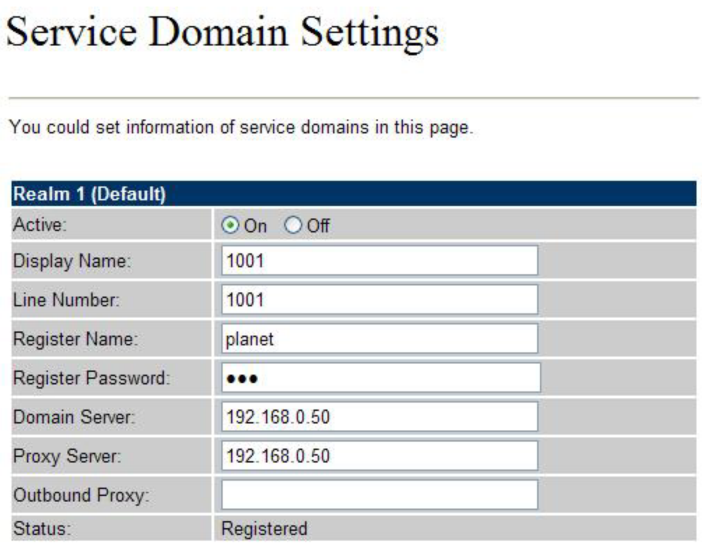
### Device configurations on the ATA:

**STEP 1:**

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

**STEP 2:**

Please log in ATA 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.

| Realm 1 (Default) | |
|---|---|
| Active: | On / Off |
| Display Name: | 1001 |
| Line Number: | 1001 |
| Register Name: | planet |
| Register Password: | ••• |
| Domain Server: | 192.168.0.50 |
| Proxy Server: | 192.168.0.50 |
| Outbound Proxy: | |
| Status: | Registered |

**STEP 3:**

Repeat the same configuration steps on ATA B, and check the machine registration status, make sure the registrations are completed.

## Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **1001** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

## Case 4: Voice communication via IP PBX system – IPX-2000 (Auto-config)

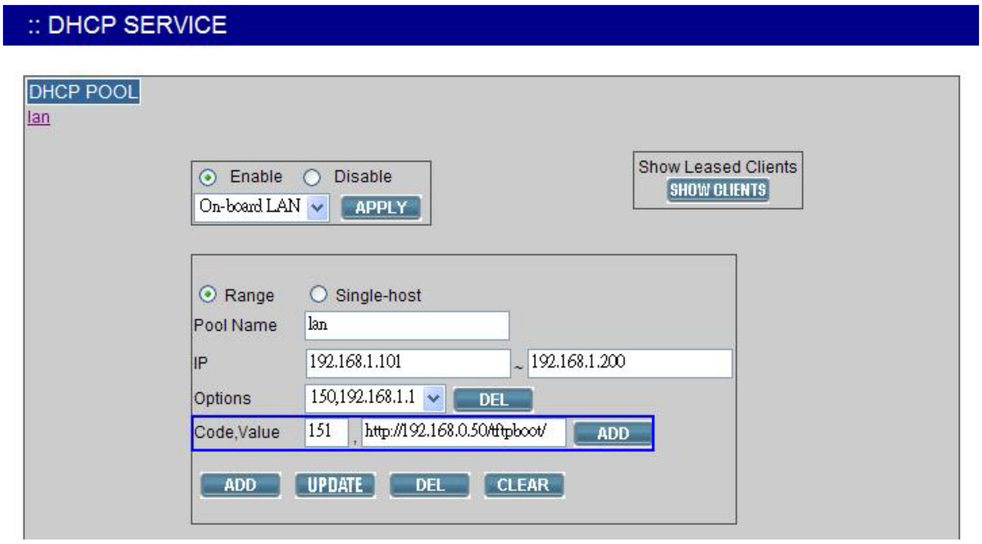
In the following sample, we'll introduce how to integrate the ATA with our IP PBX system IPX-2000 via Auto-config feature.


- ATA A IP Address: 192.168.0.1
  Line Number: 1001
- ATA B IP Address: 192.168.0.2
  Line Number: 2002

### Device configurations on the IPX-2000:

**STEP 1:**

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



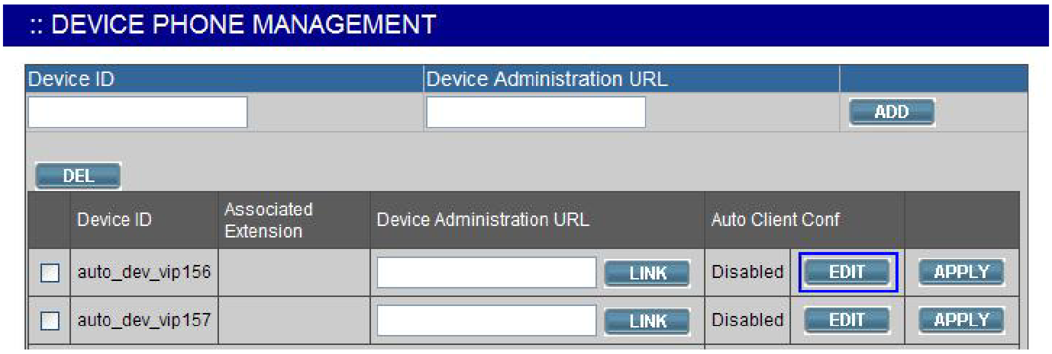

**Code:** please insert **151** as the DHCP server option.

**Value:** http://LAN IP for IPX-2000/tftpboot/

An example option 151 would be option=**151** value= **http://192.168.0.50/tftpboot/**

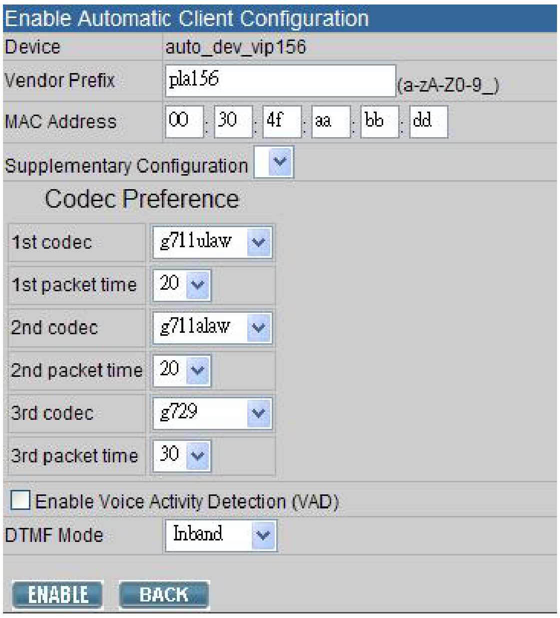
**STEP 2:**

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 ATA.



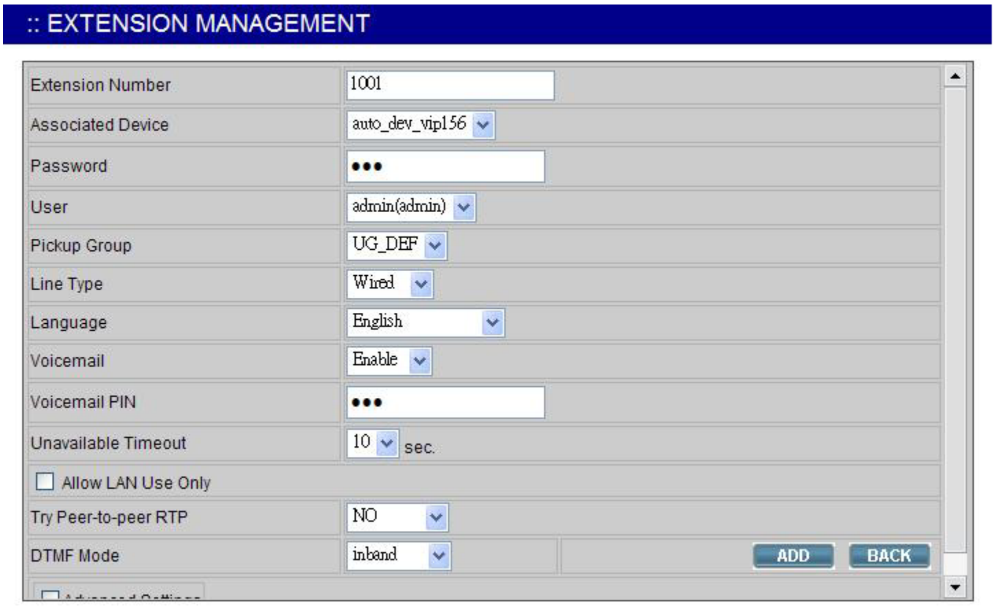

**ⓘNote:**

The following are the Vendor Prefix of devices:

1. VIP-156: **pla156**
2. VIP-157/VIP-157S: **pla157**

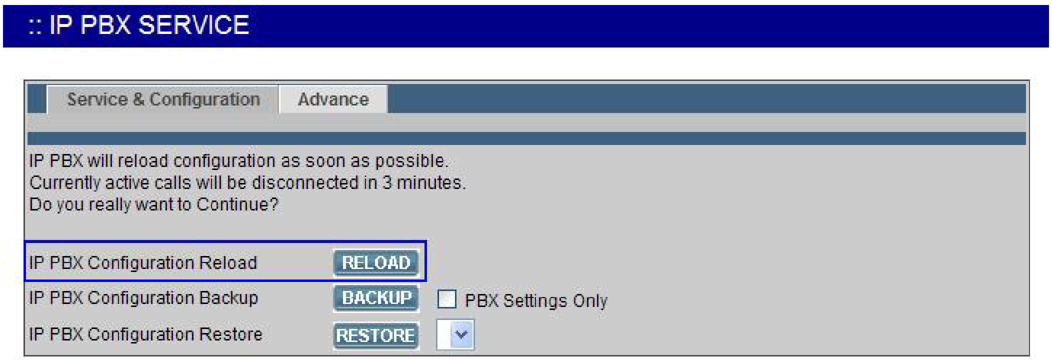
**STEP 4:**

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




**STEP 5:**

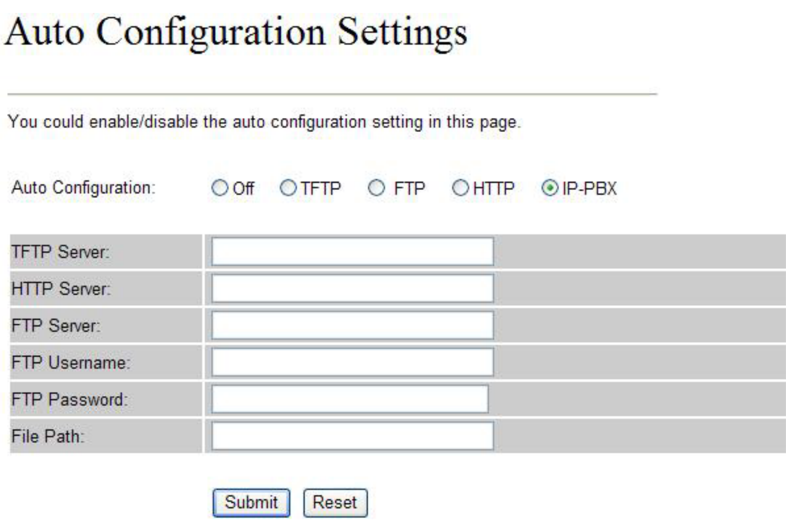
After setting up the parameters, please browse to the **Service → IP PBX service** menu, and press **RELOAD** of IP PBX configuration reload selection for activating the settings.




## Device configurations on the ATA:

**STEP 6:**

Please log in ATA via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please browse to the Auto-config page, and enable the Auto Configuration features for IP PBX system.




**STEP 7:**

After enabling the auto-config feature, the ATA 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 connect telephone set to ATA; press **#120#** to check if the IP address is obtained from IPX-2000. And **#122#** can be used to verify the extension number assigned by IPX-2000.

**STEP 8:**

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

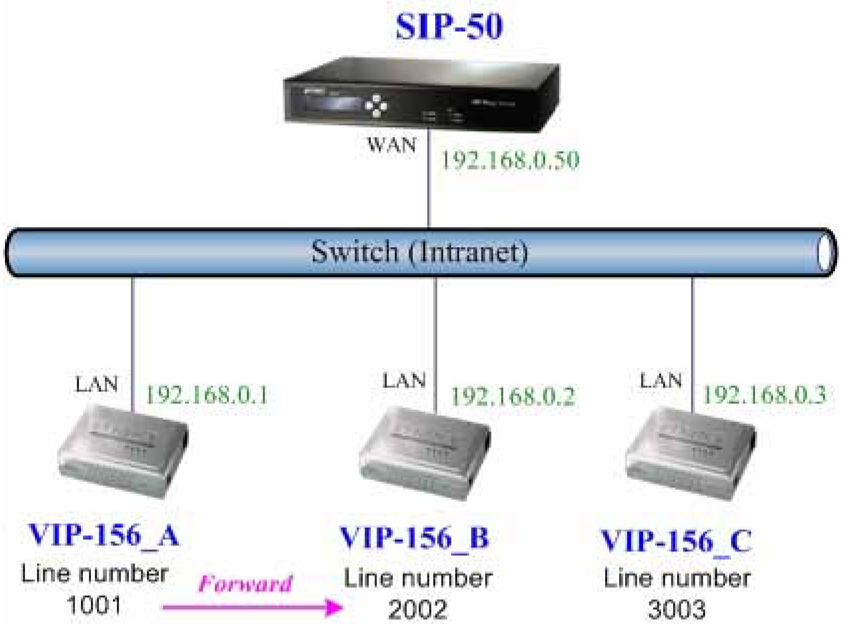
## Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **1001** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

## Case 5: Call Forward Feature_Example 1

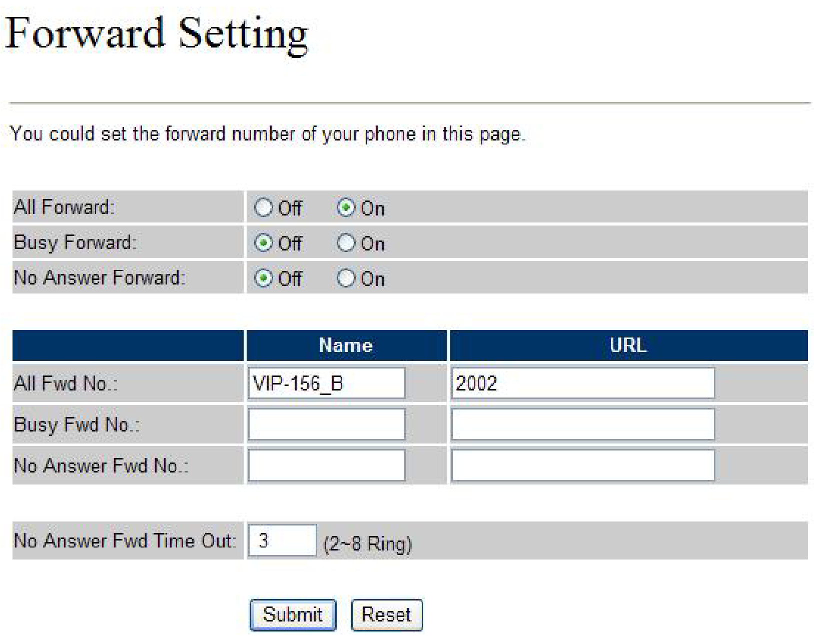
In the following samples, we'll introduce the Call Forward Feature applications.

In this example, there are three VIP-156 register to SIP-50 and VIP-156_A had set Call Forward function to VIP-156_B.




### Machine configuration on the VIP-156:

Please log in VIP-156_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-156_B, 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 ⦿ On |
| Busy Forward: | ⦿ Off ○ On |
| No Answer Forward: | ⦿ Off ○ On |

| | Name | URL |
|---|---|---|
| All Fwd No.: | VIP-156_B | 2002 |
| Busy Fwd No.: | | |
| No Answer Fwd No.: | | |

| | |
|---|---|
| No Answer Fwd Time Out: | 3 (2~8 Ring) |

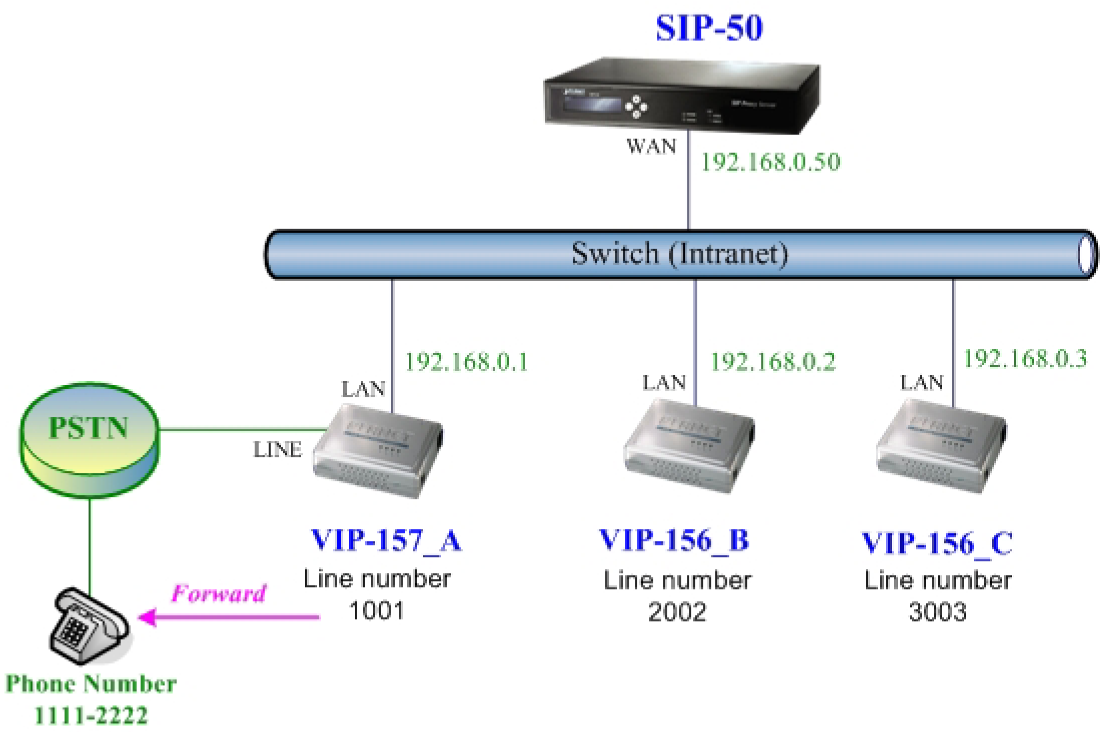
Submit Reset

**Test the scenario:**

1. VIP-156_C pick up the telephone
2. Dial the number 1001(VIP-156_A),
3. Because VIP-156_A had set up **All Forward** function to the number 2002(VIP-156_B)
4. The number 2002(VIP-156_B) will ring up then it pick up the telephone and communication with the number 3003(VIP-156_C)

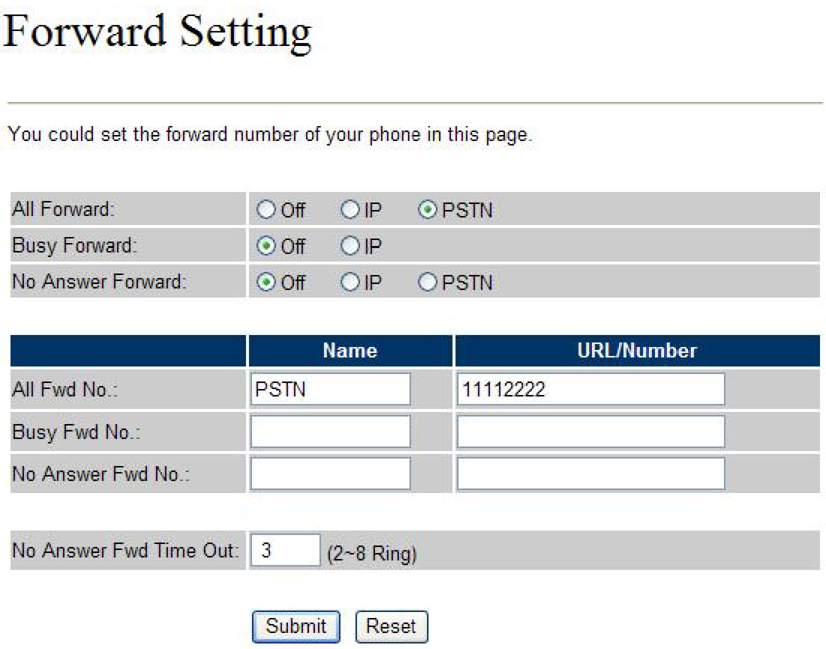
## Case 6: Call Forward Feature_Example 2

In this example, there are one VIP-157 and two VIP-156 register to SIP-50. The VIP-157_A had set Call Forward function to phone number 1111-2222 (PSTN).




### Machine configuration on the VIP-157:

Please log in VIP-157_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:

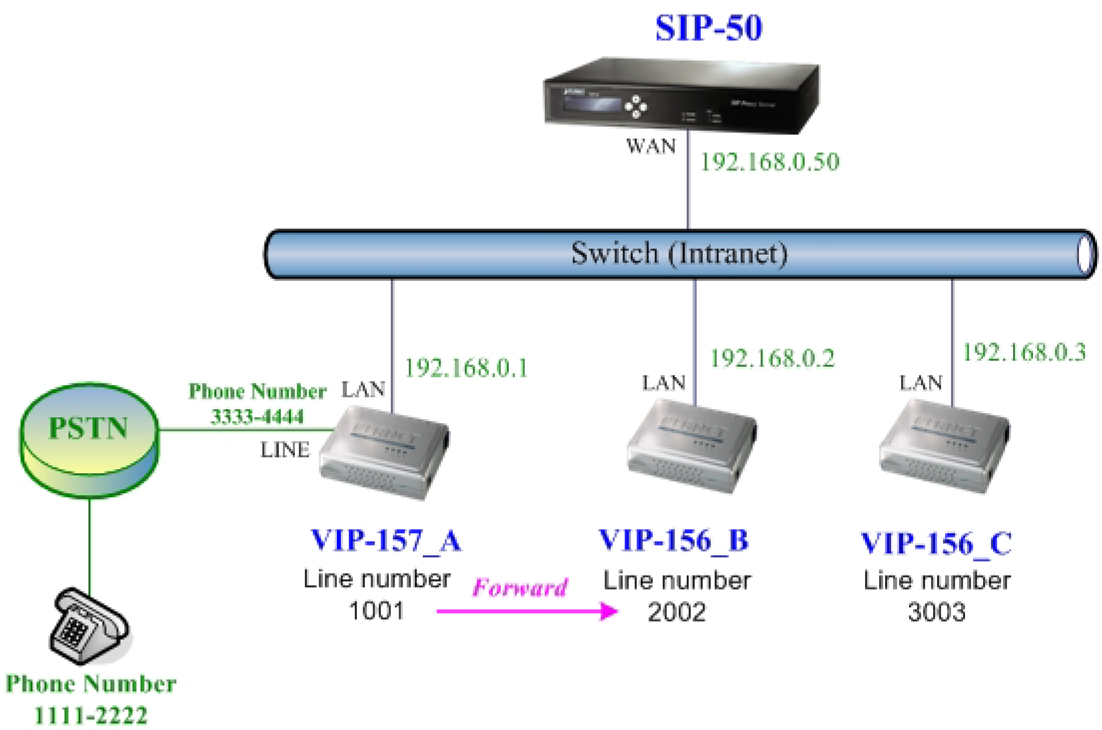



## Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the number 1001(VIP-157_A)
3. Because VIP-157_A had set up **All Forward** function to the PSTN Phone Number 11112222
4. The PSTN Phone Number 11112222 will ring up then it pick up the telephone and communication with the number 3003(VIP-156_C)

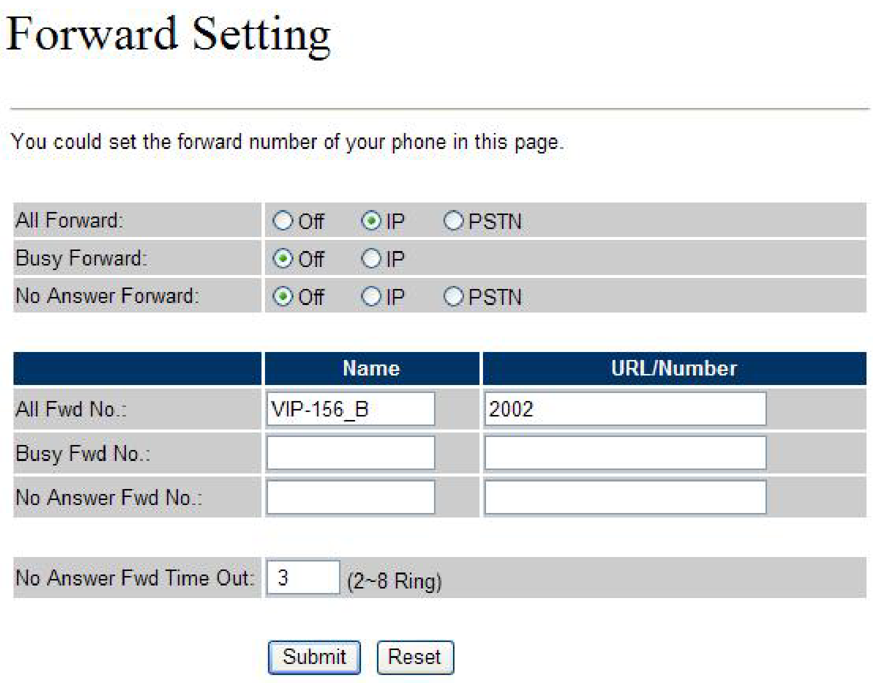
## Case 7: Call Forward Feature_Example 3

In this example, there are one VIP-157 and two VIP-156 register to SIP-50. The VIP-157_A had set Call Forward function to number 2002 (VIP-156_B).




### Machine configuration on the VIP-157:

Please log in VIP-157_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-156_B, and then the sample configuration screen is shown below:

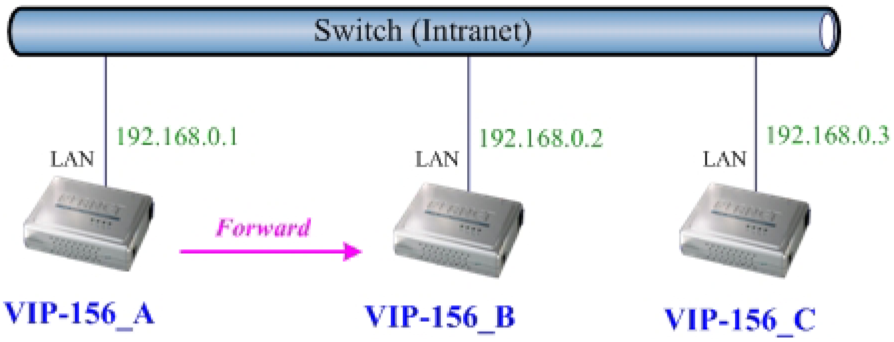



## Test the scenario:

1. PSTN Phone Number 11112222 pick up the telephone
2. Dial the PSTN Phone Number 33334444(VIP-157_A)
3. Because VIP-157_A had set up **All Forward** function to the number 2002(VIP-156_B)
4. The number 2002(VIP-156_B) will ring up then it pick up the telephone and communication with the PSTN Phone Number 11112222

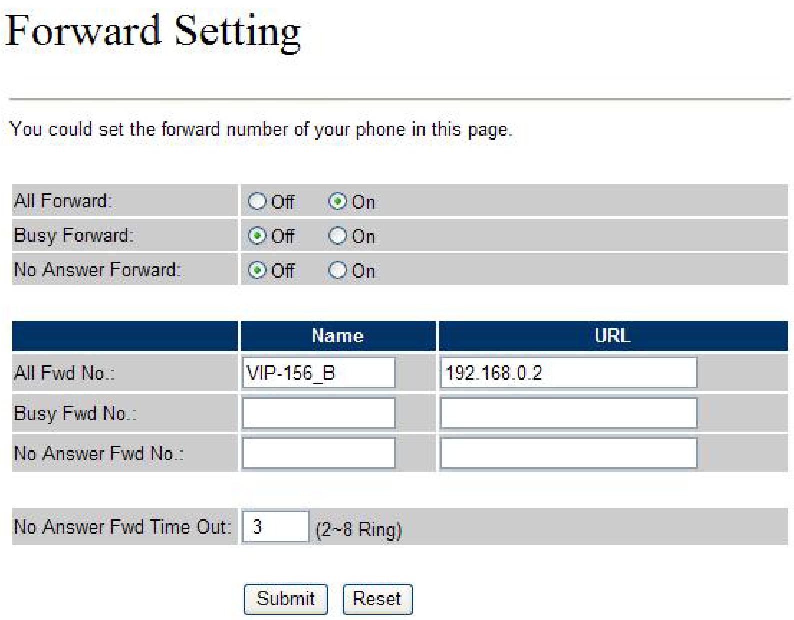
## Case 8: Call Forward Feature_Example 4

In this example, there are three VIP-156 and connect with Peer to Peer mode. VIP-156_A had set Call Forward function to VIP-156_B.




### Machine configuration on the VIP-156:

Please log in VIP-156_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-156_B, and then the sample configuration screen is shown below:

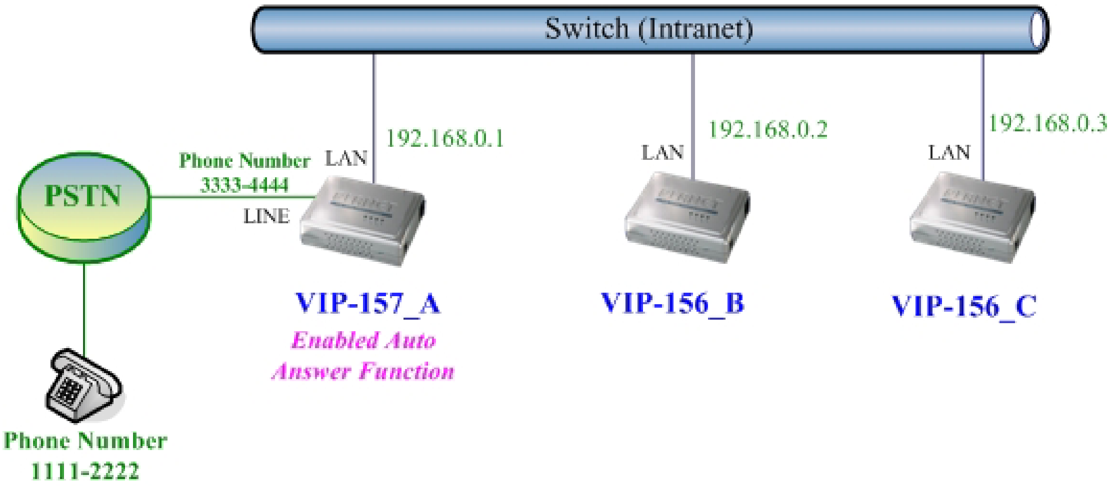



### Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the IP Address 192.168.0.1(VIP-156_A)
3. Because VIP-156_A had set up **All Forward** function to the IP Address 192.168.0.2 (VIP-156_B)
4. The IP Address 192.168.0.2 (VIP-156_B) will ring up then it pick up the telephone and communication with the VIP-156_C

## Case 9: Auto Answer Feature_IP to PSTN

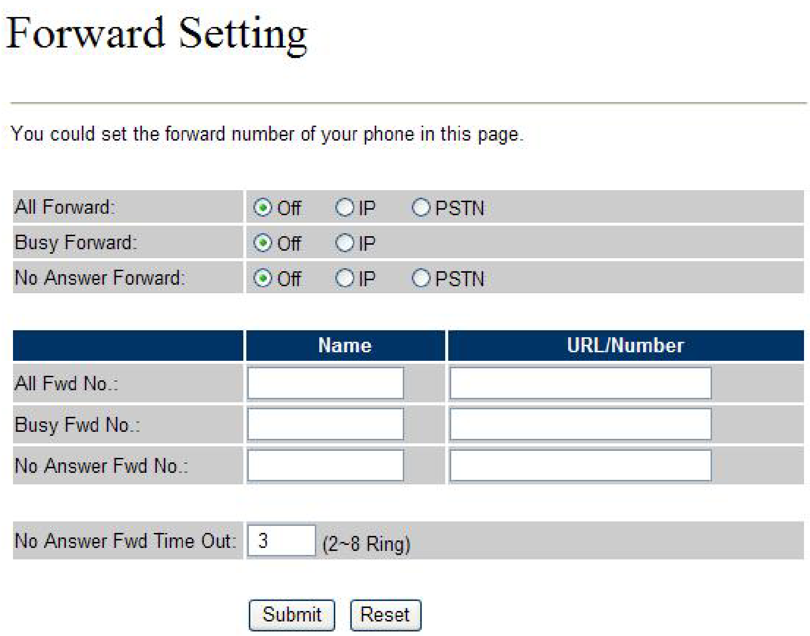
In this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157_A had set Auto Answer function for forwarding calls to arbitrary telephone. If there have incoming IP calls and VIP-157_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password then dial out an arbitrary PSTN telephone.




### Machine configuration on the VIP-157:

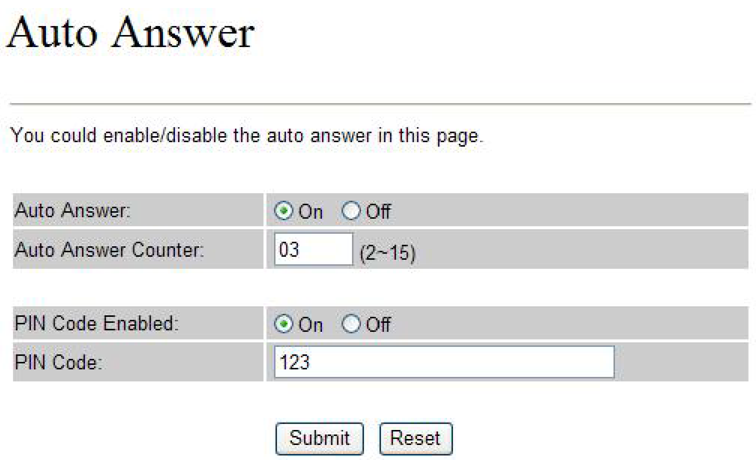
**STEP 1:**

Please log in VIP-157_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-157_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:

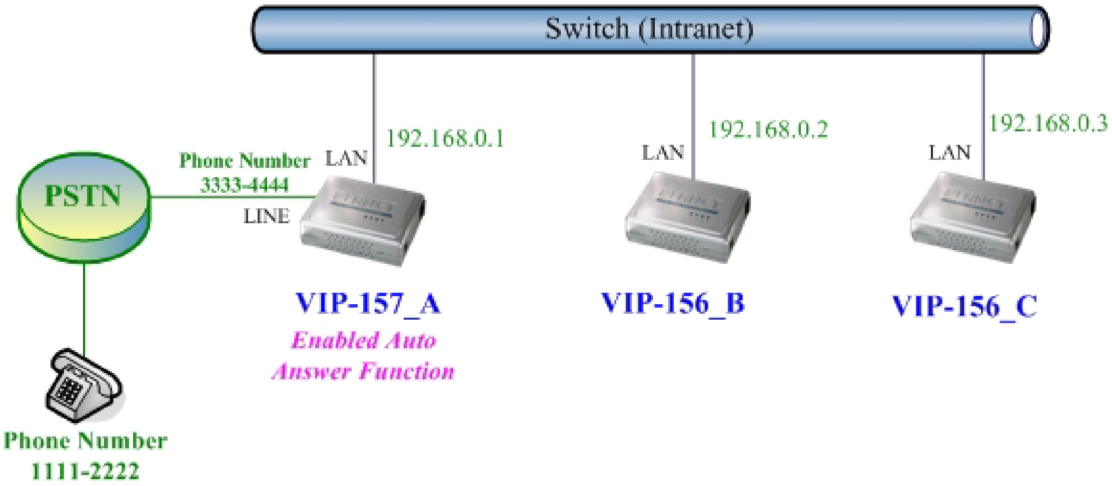



## Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the IP Address 192.168.0.1(VIP-157_A)
3. VIP-157_A will ring up but doesn't answer the call
4. After **3** rings, the VIP-156_C will hear the prompt sounds then input the password **123#**
5. VIP-156_C will hear the dial tone from PSTN line then input Phone Number 11112222
6. The Phone Number 11112222 will ring up then it pick up the telephone and communication with the VIP-156_C

## Case 10: Auto Answer Feature_PSTN to IP

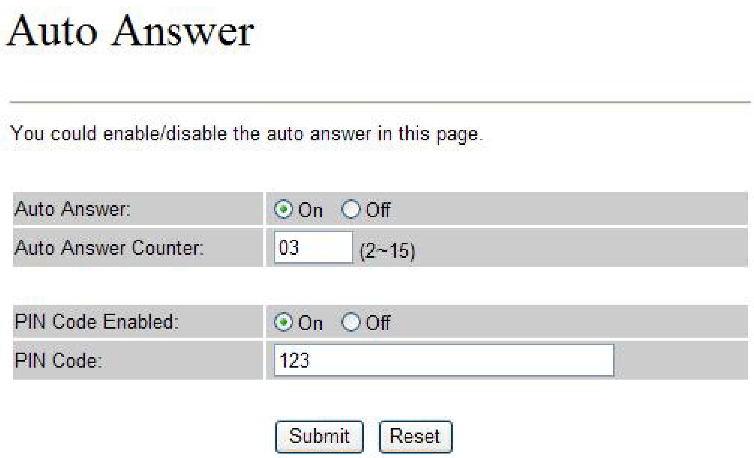
In this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157_A had set Auto Answer function for forwarding to arbitrary telephone. If there have incoming PSTN calls and VIP-157_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password and then dial out an arbitrary IP telephone.




### Machine configuration on the VIP-157:

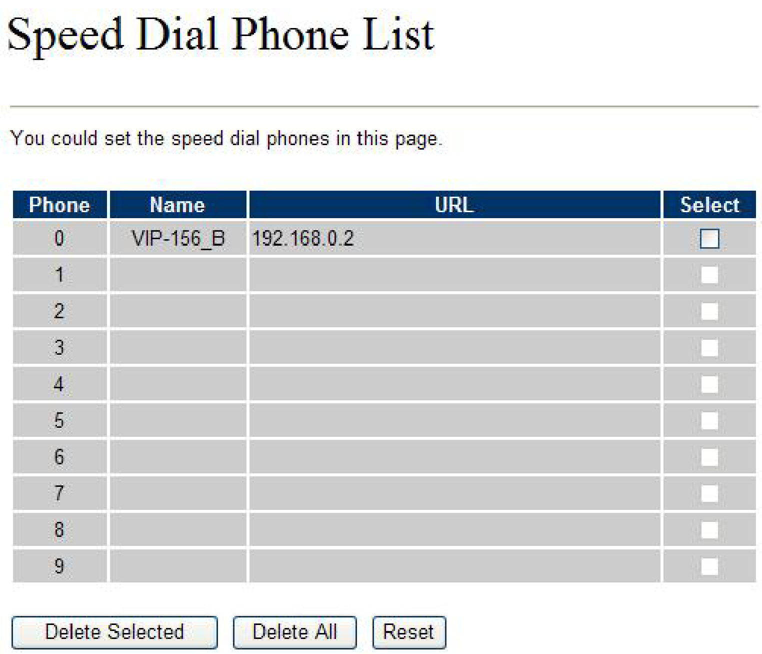
**STEP 1:**

Please log in VIP-157_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-157_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-156_B), and then the sample configuration screen is shown below:




## Test the scenario:

1. The Phone Number 11112222 pick up the telephone
2. Dial the PSTN Phone Number 33334444(VIP-157_A)
3. VIP-157_A will ring up but doesn't answer the call
4. After **3** rings, the Phone Number 11112222 will hear the prompt sounds then input the password **123#**
5. The Phone Number 11112222 will hear the dial tone then input **0#**
6. The IP address 192.168.0.2 (VIP-156_B) will ring up then it pick up the telephone and communication with the Phone Number 11112222

# Appendix B VIP-156/VIP-156PE/VIP-157/VIP-157S Specifications

| | | | | |
|---|---|---|---|---|
| Product | SIP Analog Telephone Adapter | | | |
| Model | VIP-156 | VIP-156PE | VIP-157 | VIP-157S |
| Hardware | | | | |
| LAN | 1 x 10/100Mbps RJ-45 port (802.3af PoE for VIP-156PE) | | | |
| PC | 1 x 10/100Mbps RJ-45 port | | | |
| FXS (for telephone set connection) | 1 x RJ-11 | | | 2 x RJ-11 |
| FXO (PSTN connection) | --- | | 1 x RJ-11 | --- |
| Protocols and Standard | | | | |
| Standard | SIP 2.0 (RFC3261), SIP Outbound proxy, STUN (RFC 3489) | | | |
| Voice codec | G.723.1 (6.3k/5.3k), G.729A, G.729B, G.711 | | | |
| Fax support | T.38 | | | |
| Voice Standard | Voice activity detection (VAD)<br>Comfort noise generation (CNG)<br>Acoustic echo canceller (AEC)<br>G.165: Line echo canceller (LEC)<br>Jitter Buffer | | | |
| Protocols | SIP 2.0 (RFC-3261), TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE | | | |
| Network and Configuration | | | | |
| Access Mode | Static IP, PPPoE, DHCP | | | |
| Management | Web, keypad | | | |
| Dimension (W x D x H) | 94 x 72 x 30 mm | | | |
| Operating Environment | 0~40 degree C, 10~95% humidity | | | |
| Power Requirement | 12V DC | | | |
| EMC/EMI | CE, FCC Class B | | | |