

# User's Manual



## VoIP Analog Telephone Adapter

▶ VIP-156 / VIP156PE / VIP-157 / VIP-157S



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## Federal Communication Commission Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experienced radio technician for help.

## FCC Caution:

To assure continued compliance (example-use only shielded interface cables when connecting to computer

or peripheral devices). Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with Part 15 of the FCC Rules. Operation is subject to the Following two conditions: (1) This device may not cause harmful interference, and (2) this Device must accept any interference received, including interference that may cause undesired operation.

## R&TTE Compliance Statement

This equipment complies with all the requirements of DIRECTIVE 1999/5/EC OF THE EUROPEAN PARLIAMENT AND THE COUNCIL OF 9 March 1999 on radio equipment and telecommunication terminal Equipment and the mutual recognition of their conformity (R&TTE) The R&TTE Directive repeals and replaces in the directive 98/13/EEC (Telecommunications Terminal Equipment and Satellite Earth Station Equipment) As of April 8, 2000.

## WEEE Caution



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.

## Safety

This equipment is designed with the utmost care for the safety of those who install and use it. However, special attention must be paid to the dangers of electric shock and static electricity when working with electrical equipment. All guidelines of this and of the computer manufacture must therefore be allowed at all times to ensure the safe use of the equipment.

## Customer Service

For information on customer service and support for the Gigabit SSL VPN Security Router, please refer to the following Website URL:

<http://www.planet.com.tw>

Before contacting customer service, please take a moment to gather the following information:

- ◆ VoIP Analog Telephone Adapter serial number and MAC address
- ◆ Any error messages that displayed when the problem occurred
- ◆ Any software running when the problem occurred
- ◆ Steps you took to resolve the problem on your own

## Revision

User's Manual for PLANET VoIP Analog Telephone Adapter

Model: ATA model

Rev: 1.1 (June, 2013)

**TABLE OF CONTENTS**

---

<b>Chapter 1 Introduction.....</b>	<b>6</b>
Overview.....	6
Package Contents.....	9
Physical Details .....	9
LED Display & Button .....	12
<b>Chapter 2 Preparations &amp; Installation.....</b>	<b>14</b>
Physical Installation Requirements.....	14
LAN IP address configuration via web configuration interface.....	15
<b>Chapter 3 Network Service Configurations.....</b>	<b>20</b>
Configuring and monitoring your ATA from web browser.....	20
Overview of the web interface of ATA.....	20
Manipulation of ATA via web browser .....	20
<b>Chapter 4 VoIP Telephone Adapter Configurations .....</b>	<b>22</b>
Status.....	22
Phone Book.....	23
Call Service.....	25
SNTP setting .....	27
Volume Setting.....	27
Dial Plan Setting .....	28
General.....	32
<b>Chapter 5 Network.....</b>	<b>35</b>
Network Setting .....	35
DDNS Setting .....	36
VLAN Setting .....	36
VPN Setting .....	37
IPV6 Setting.....	38
<b>Chapter 6 NAT Trans .....</b>	<b>39</b>
Stun Setting.....	39
PC Setting .....	39
DMZ and MAC Clone .....	40
Virtual Server .....	40
<b>Chapter 7 SIP Setting.....</b>	<b>42</b>
Service Domain Setting .....	42

Codec Setting .....	43
SIP Advance Setting.....	44
<b>Chapter 8 Advance Setting .....</b>	<b>49</b>
Status Log .....	49
Auto Config .....	49
Management-Advanced Setting.....	50
Tones .....	52
TR-069.....	53
<b>Chapter 9 Other Setting .....</b>	<b>55</b>
System Authority .....	55
Firmware Upgrade .....	55
Auto Update Setting.....	56
Reset to default .....	58
Save and Reboot .....	58
Logout.....	58
<b>Appendix A Voice Communication Samples .....</b>	<b>59</b>
Case 1: ATA to ATA connection via IP address .....	59
Case 2: (Peer-to-Peer mode) VIP-157S Port 1 to Port 2 communications .....	60
Case 3: Call Forward Feature_Example 1.....	60
Case 4: Call Forward Feature_Example 2.....	61
Case 5: Call Forward Feature_Example 3.....	62
Case 6: Call Forward Feature_Example 4.....	63
Case 7: Auto Answer Feature_IP to PSTN.....	64
Case 8: Auto Answer Feature_PSTN to IP.....	66
<b>Appendix B The method of operation guide.....</b>	<b>68</b>
Call Transfer .....	68
3-Way Conference.....	68
Call Waiting .....	68
Switch the Realm (Registration Proxy Server) .....	69
Auto Update firmware manually (Keypad).....	69
<b>Appendix C VIP-156/VIP-156PE/VIP-157/VIP-157S Specifications .....</b>	<b>70</b>
<b>Appendix D Planet DDNS Application .....</b>	<b>71</b>

# Chapter 1

## Introduction

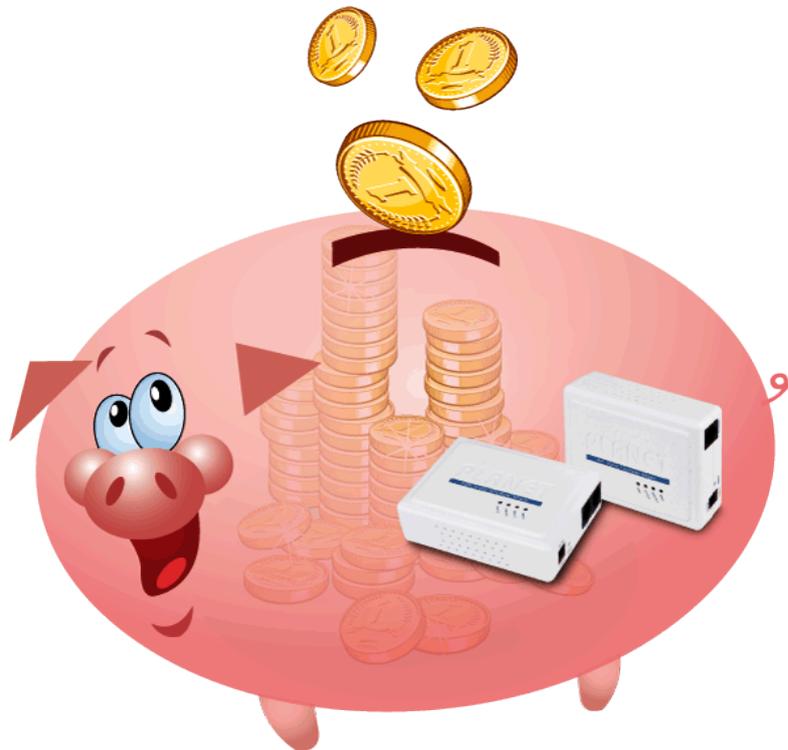
### Overview

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, High-performance**

Cost-effective, easy-to-install and simple-to-use, the 802.3af PoE integration(VIP-156PE) 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 device, home users and companies are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The ATA device can be the bridge between traditional analog systems and IP network with an extremely affordable investment.



### Enhanced, Full-Featured VoIP Adapter

For the new generation communication age, the ATA device supports IPV6 and VPN connection to provide users with more flexible and advantageous communication product. The ATA device and our IP PBX system integration are the ideal combination for your office daily communications. The VIP-156 supports all kinds of SIP based phone features including Phone book, Call forward (busy, no answer, always), Dial plan setting, SNTP setting, DDNS, VLAN, VPN(PPTP&L2TP), IPV6, DMZ, Mac Clone, virtual server... virtual server...



### Standard Compliance

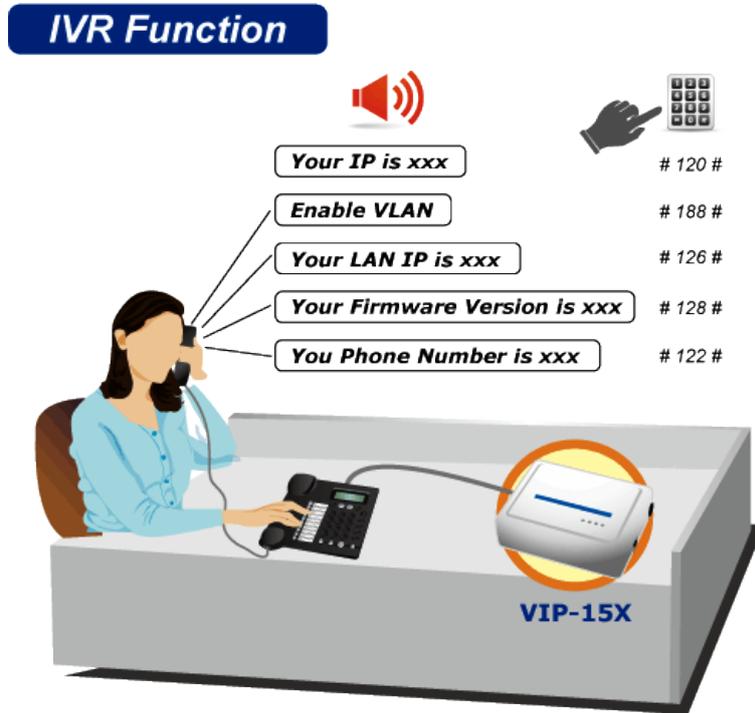
Compliant with the Session Initiation Protocol 2.0 (RFC 3261), the ATA device is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

## Compliant with Standard SIP RFC 3261



**IVR Function to Easily Identify and Manage the ATA**

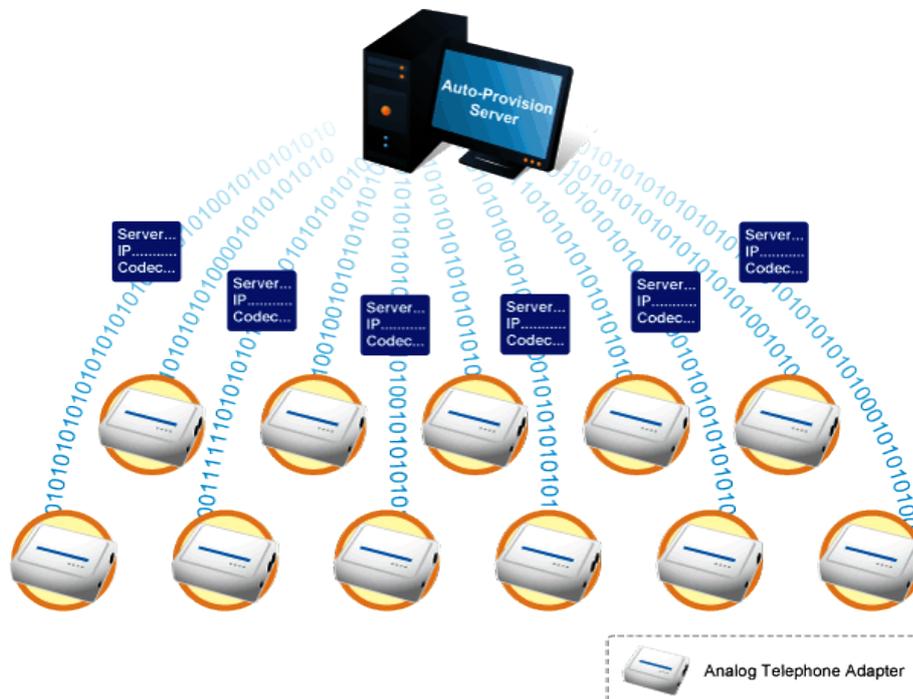
Through the Interactive voice response (IVR) function, user can simply press some function key to search the device information or program the phone feature, e.g #120 to check the LAN IP address, #112 + xxx\*xxx\*xxx\*xxx# to assign the LAN IP address.....



**Through Auto Provision to Synchronize Configuration Parameters**

Through TFTP, FTP or HTTP auto provision function, user can Synchronize Configuration Parameters of the ATA device at the same time. User can easily control the parameters and configuration of thousands of ATA device from the web interface, without having to provision each unit individually.

**Auto Provision - Synchronize Configuration**



## Features

### ➤ **Product features**

- Feature-rich telephone service over home or office Internet/Intranet connection
- Cost effectiveness, field proven compatibility, and stability
- Web-based and telephone keypad machine configuration
- Remote administrator authentication
- Voice prompt for machine configurations
- DMZ and MAC clone

### ➤ **VoIP Feature**

- 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,
- In band and out-of-band DTMF support
- Auto-provision(FTPP, HTTP, FTP)

### ➤ **Other Features**

- IPV6
- VPN connection
- Planet DDNS
- VLAN
- Local Phone book (download/upload)
- QoS
- IVR Function
- 802.3af PoE integration(VIP-156PE)
- FXO integration(VIP-157)

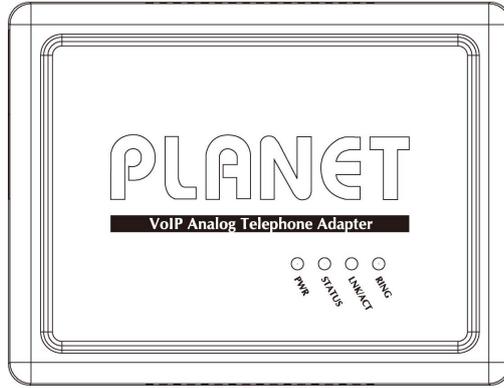
## Package Contents

- 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

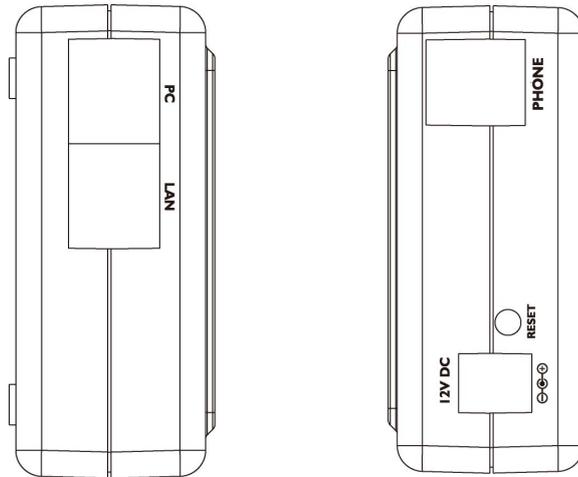
## Physical Details

- The following figure illustrates the front/rear panel of ATA.

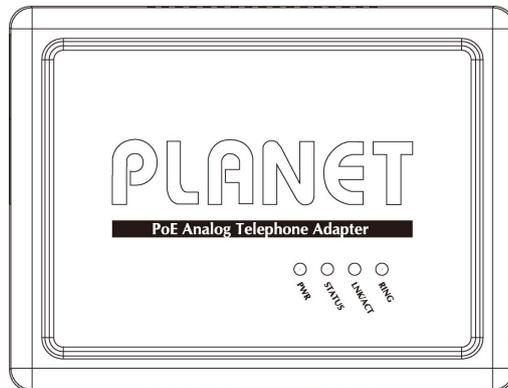
- Respective models/descriptions are shown below:
- VIP-156: SIP Analog Telephone Adapter
- VIP-156PE: 802.3af PoE SIP Analog Telephone Adapter
- VIP-157: 1 FXS/ 1 FXO SIP Analog Telephone Adapter
- VIP-157S: 2-port FXS SIP Analog Telephone Adapter



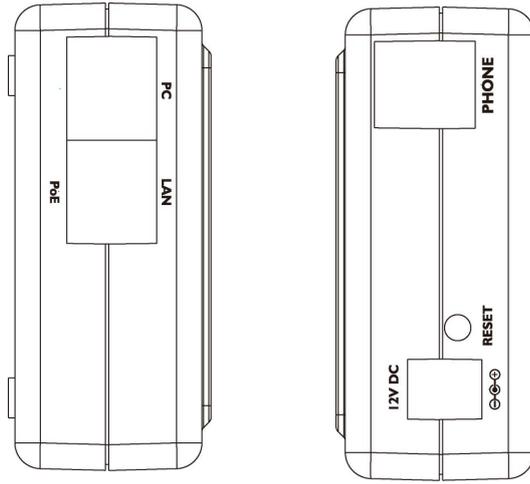
**Front Panel of VIP-156**



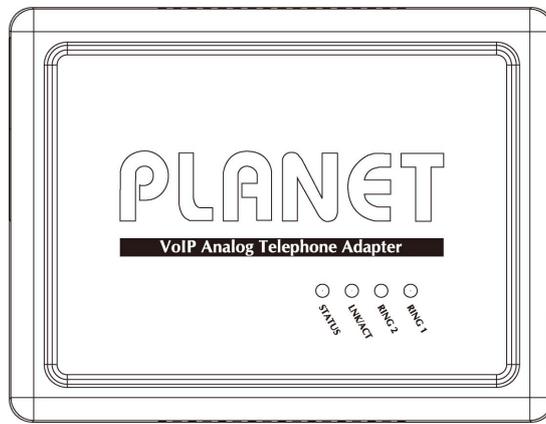
**Left / Right Panel of VIP-156**



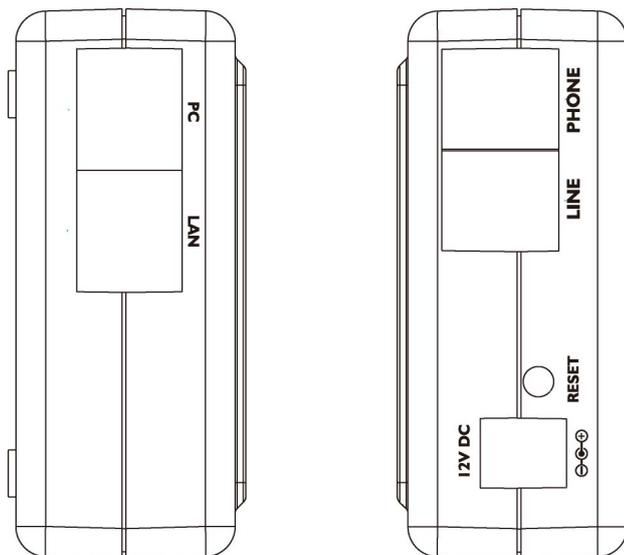
**Front Panel of VIP-156PE**



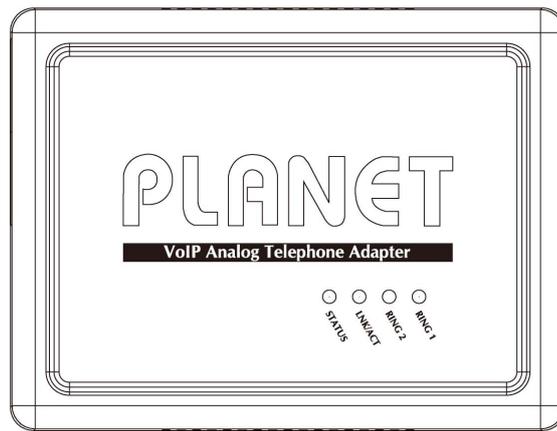
**Left / Right Panel of VIP-156PE**



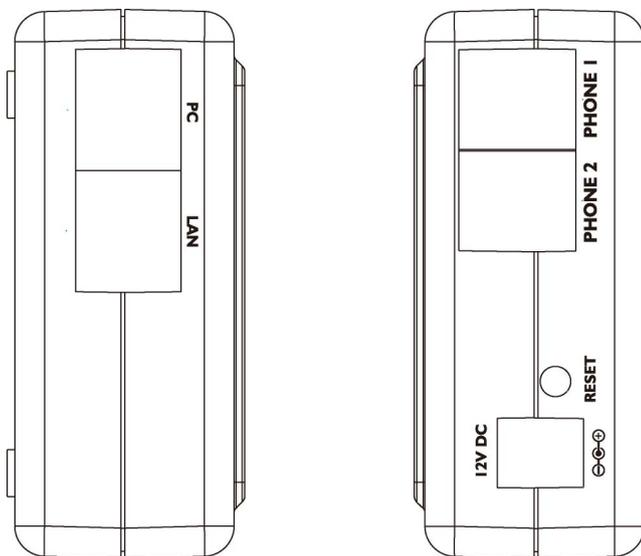
**Front Panel of VIP-157**



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



Front Panel of VIP-157S



Left / Right Panel of VIP-157S

### LED Display & Button

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

4      **Reset**      Reset to the factory default setting

 <b>Note</b>	<p>Machine default IP is http://192.168.0.1. Press <b>RESET</b> button on rear panel for over 5 seconds to reset the VoIP Phone Adapter to factory default value. (Except speed dial and call forward settings)</p>
--	---

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

LED Indicators	Descriptions
<b>PWR</b>	Power is supplied to the device.
<b>STATUS</b>	The Status LED will flash when the machine is operational
<b>LNK/ACT</b>	<b>OFF:</b> the device is connected to LAN at 10Mb/s. <b>ON:</b> the device is connected to LAN at 100Mb/s.
<b>RING</b>	<b>OFF:</b> the phone is idle. <b>ON:</b> the phone is in use (off hook). <b>Blinking:</b> the phone is ringing.

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

LED Indicators	Descriptions
<b>STATUS</b>	The Status LED will flash when the machine is operational
<b>LNK/ACT</b>	<b>OFF:</b> the device is connected to LAN at 10Mb/s. <b>ON:</b> the device is connected to LAN at 100Mb/s.
<b>RING 1</b>	<b>OFF:</b> the phone is idle. <b>ON:</b> the phone is in use (off hook). <b>Blinking:</b> the phone is ringing.
<b>RING 2</b>	<b>OFF:</b> the phone is idle. <b>ON:</b> the phone is in use (off hook). <b>Blinking:</b> the phone is ringing.

# Chapter 2

## Preparations & Installation

### Physical Installation Requirements

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

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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 6.0.0 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.



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

## 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 address bar. After logging on machine with username/password (**default: root / no password**), browse **"Network"** --> **"Network Settings"** configuration menu:

### Network Settings

---

You could configure the Network settings in this page.

Type:	Fixed IP <input type="button" value="v"/>
IP Address:	<input type="text" value="192.168.0.156"/>
Subnet Mask:	<input type="text" value="255.255.255.0"/>
Default Gateway:	<input type="text" value="192.168.0.1"/>
DNS Type:	Fixed <input type="button" value="v"/>
DNS Server 1:	<input type="text" value="168.95.192.1"/>
DNS Server 2:	<input type="text" value="168.95.1.1"/>
MAC ID:	<input type="text" value="00:11:22:33:44:55"/>
Host Name:	<input type="text" value="VOIP_TA1S10"/>

### Parameter Description

---

<b>IP address</b>	LAN IP address of ATA <b>Default:</b> 192.168.0.1
<b>Subnet Mask</b>	LAN mask of ATA <b>Default:</b> 255.255.255.0
<b>Default Gateway</b>	Gateway of ATA <b>Default:</b> 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.

 **Hint**

When you want to run the setup or the start function, it must unlock the protect function **#190#** before setting up network settings and ATA function via keypad.

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.
#190#	Unlock	None	Must unlock the protect function before setting up network settings and ATA function 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

The 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 PC IP Address	IVR will announce the current PC-port IP address of the ATA.
#121#	Check network connection type	IVR will announce if DHCP is 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.
<b>#124#</b>	Check Gateway IP Address	IVR will announce the current gateway IP address of the ATA.
<b>#125#</b>	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
<b>#126#</b>	Check LAN IP Address	IVR will announce the current LAN port IP address of the ATA.
<b>#128#</b>	Check Firmware Version	IVR will announce the version of the firmware running on the ATA.

The following keypad commands can be used to set up the main function .

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
<b>#138#</b>	Enable call waiting	None	Enable Call waiting
<b>#139#</b>	Disable call waiting	None	Disable Call waiting
<b>#160#</b>	Update firmware	None	Update firmware
<b>#510#</b>	Blind Transfer	ATA Bland Transfer	
<b>#511#</b>	Attendant Transfer	ATA Attendant Transfer	
<b>#512#</b>	3-way calling	ATA 3-way calling	
<b>#514#</b>	IP transfer to PSTN	ATA transfer IP call to PSTN side	
<b>#130+[1~8]#</b>	Set Codec	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
<b>#131+[00~15]#</b>	Set Handset Gain	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
<b>#132+[00~12]#</b>	Set Handset Volume	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
<b>#135xxx*xxx*xxx*xxx*xxx#</b>	TFTP Server IP Address	Set Auto config TFTP Server IP Address	You can set the TFTP Server IP address
<b>#136xxx*xxx*xxx*xxx*xxx#</b>	FTP Server IP Address	Set Auto config FTP Server IP Address	You can set the FTP Server IP address
<b>#137+[0~2]#</b>	Auto config mode	0: Disable, 1: TFTP mode, 2: FTP mode	You can set the Auto configuration mode, 0: Disable, 1: use TFTP

			Server, 2: user FTP Server
#145#	Forward function disable	Disable forward function	
#146+Number#	enable forward to FXS Port	Enable forward to FXS Port	
#147+Number#	enable forward to FXO Port	Enable forward to FXO Port	
#116#	Enable PPTP function	None	Enable PPTP function
#117#	Disable PPTP function	None	Disable PPTP function
#118#	Enable VLAN function	None	Enable VLAN function
#119#	Disable VLAN function	None	Disable VLAN function

**i** 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 "**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 client	The ISP will assign IP Address, and related information.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

**i** Hint

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

### Save Modification to Flash Memory

Most of the VoIP router parameters will be effective after modifications, but it is just temporarily stored on RAM only. It will disappear after you reboot or power off the VoIP Phone Adapter. To save the parameters into Flash ROM and let it be effective forever, please remember to press the **Save & Reboot** button after you modify the parameters.

## Save & Reboot

You have to save changes to effect them.

---

Save Changes:

## Chapter 3

# Network Service Configurations

### Configuring and monitoring your ATA 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 of the web interface of ATA

With web graphical user interface, you may have:

- ◆ More comprehensive setting feels than traditional command line interface.
- ◆ Provides user with input data fields, check boxes, and 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

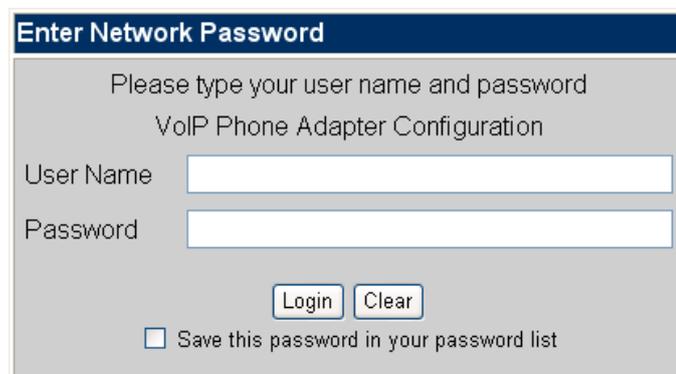
- ◆ Microsoft Internet Explorer 6.0.0 or higher with Java support

#### 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)**



The screenshot shows a web browser window titled "Enter Network Password". The page content includes the text "Please type your user name and password" and "VoIP Phone Adapter Configuration". There are two input fields: "User Name" and "Password". Below the fields are two buttons: "Login" and "Clear". At the bottom, there is a checkbox labeled "Save this password in your password list".

##### ATA log in page

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



**VoIP Phone Adapter  
Configuration Menu**

- Status
- Phone Book
- Phone Settings
- Network
- NAT Trans
- SIP Settings
- Advanced Settings
- System Auth
- System Settings
- Save and Reboot
- Logout

## System Information

This page illustrate the system related information.

Company:	PLANET Technology Corp.
Contact Address:	10F., No.96, Minquan Rd., Xindian Dist., New Taipei City 231, Taiwan (R.O.C.)
Tel:	886-2-22199518
Fax:	888-2-22199528
E-Mail:	<a href="mailto:support_voip@planet.com.tw">support_voip@planet.com.tw</a>
Web Site:	<a href="http://www.planet.com.tw">www.planet.com.tw</a>

*VoIP Phone Adapter main page*

## VoIP Telephone Adapter Configurations

### Status

Status shows all the system information like WAN/LAN IP address, System information, IPV6 connection information, Register status and VPN connection message. (After you set up the VPN line, the status will start to show.)

### Status Information

You could see the information of the VOIP machine.

WAN Port			
Link Status:	UP	Type:	Fixed IP Client
IP Address:	192.168.0.156	Subnet Mask:	255.255.255.0
Default Gateway:	192.168.0.1	DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1	MAC ID:	00:11:22:33:44:55

LAN Port			
IP Address:	192.168.0.1	MAC ID:	00:11:22:33:44:66

System Information			
Firmware Version:	1.1.0-1201180	Update Date:	2005-01-01
DSP Version	NV-1106080		
System Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Current Time:	2012-02-20 15:54		

IPv6 Connection Information			
Link Status:	UP	Type:	Auto
Globe Address:	fe80:0:0:0:211:22ff:fe33:4455		
Gateway:	unknow		
Local Address:	fe80:0:0:0:211:22ff:fe33:4455		

Register Information			
Realm 1 Status:	Registered	Number:	156
Realm 2 Status:	Not Registered	Number:	
Realm 3 Status:	Not Registered	Number:	
Realm 4 Status:	Not Registered	Number:	
Realm 5 Status:	Not Registered	Number:	

## Phone Book

ATA can set up 140 records of Phone Book. User can dial the **Name** records to make calls via **Phone Book** feature.

### Phone Book Setting

You could add/delete items in current phone book.

Page:

Index	Name	Number/URL	Action
1	<input type="text"/>	<input type="text"/>	Delete
2	<input type="text"/>	<input type="text"/>	Delete
3	<input type="text"/>	<input type="text"/>	Delete
4	<input type="text"/>	<input type="text"/>	Delete
5	<input type="text"/>	<input type="text"/>	Delete
6	<input type="text"/>	<input type="text"/>	Delete
7	<input type="text"/>	<input type="text"/>	Delete
8	<input type="text"/>	<input type="text"/>	Delete
9	<input type="text"/>	<input type="text"/>	Delete
10	<input type="text"/>	<input type="text"/>	Delete
11	<input type="text"/>	<input type="text"/>	Delete
12	<input type="text"/>	<input type="text"/>	Delete
13	<input type="text"/>	<input type="text"/>	Delete
14	<input type="text"/>	<input type="text"/>	Delete
15	<input type="text"/>	<input type="text"/>	Delete
16	<input type="text"/>	<input type="text"/>	Delete
17	<input type="text"/>	<input type="text"/>	Delete
18	<input type="text"/>	<input type="text"/>	Delete
19	<input type="text"/>	<input type="text"/>	Delete
20	<input type="text"/>	<input type="text"/>	Delete

Field	Description
<b>Phone Book Page</b>	The default is Page 1. It can select Page1 ~ Page 7 to look through Phone Book records.

<b>Phone</b>	The record number is from 1 ~ 140; it can set up 140 records in total.
<b>Name</b>	The name of Phone Book records; it only can input numerals character.
<b>URL</b>	Fill in the outgoing number (Line Number) or IP address.
<b>Delete</b>	Clean this item's data.
<b>Export csv</b>	Save the phone book data as CSV file.
<b>Upload</b>	Upload the phone book file

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

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

Press "Reset" to erase the data that you didn't save.

**For Example:**

### Phone Book

You could add/delete items in current phone book.

Page: 1

Phone	Name	URL	Action
1	301	301@192.168.1.2	Delete
2	206	17476433364	Delete
3	202	192.168.1.2:5062	Delete
4			Delete
5			Delete
6			Delete
7			Delete
8			Delete
9			Delete
10			Delete
11			Delete

**Ex\_1:**

ATA has added the above phone numbers. User picks up the handset and dial "301" to make the P2P call ([301@192.168.1.2](tel:301@192.168.1.2)).

**Ex\_2:**

User picks up the handset and dial "206" to make the Proxy call (17476433364).

**Ex\_3:**

User picks up the handset and dial "202" to make the P2P call (192.168.1.2:5062).

## Call Service

Forward Type	Forward Number	Rings
Disable <input type="button" value="v"/>		3 <input type="button" value="v"/> Phone 1
Hotline Type	Hotline Number	Hotline Line
Disable <input type="button" value="v"/>		0 <input type="button" value="v"/> Phone 1
DND Type	DND	DND Line
Disable <input type="button" value="v"/>	From <input type="text" value="0"/> : <input type="text" value="0"/> To <input type="text" value="0"/> : <input type="text" value="0"/> (hh:mm)	Phone 1
Alarm Type	Alarm Time	Alarm Line
Disable <input type="button" value="v"/>	<input type="text" value="0"/> : <input type="text" value="0"/> (hh:mm)	Phone 1

### [Call Forward]

This page defines Call Forward function. You can set up the phone number you want to forward on this page. There are three types of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by clicking the icon.

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

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

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

Forward Type	Forward Number	Rings
<input type="button" value="v"/> <ul style="list-style-type: none"> <li>Disable</li> <li style="background-color: #0056b3; color: white;">Disable</li> <li>All</li> <li style="border: 2px solid red;">Busy</li> <li style="border: 2px solid red;">No Answer</li> <li style="border: 2px solid red;">Busy or No Answer</li> </ul>		3 <input type="button" value="v"/> Phone 1
Hotline Type	Hotline Number	Hotline Line
		0 <input type="button" value="v"/> Phone 1

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

**All to PSTN/ No Answer to PSTN (VIP-157):** the 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, then input the name and the PSTN number in the URL/Number field.

Forward Type	Forward Number	Rings
<input type="button" value="Busy or No Answer"/> <input type="button" value="Disable"/> <input type="button" value="All"/> <input type="button" value="Busy"/> <input type="button" value="No Answer"/> <input type="button" value="Busy or No Answer"/> <input type="button" value="All to PSTN"/> <input type="button" value="No Answer of PSTN"/> <input type="button" value="Disable"/>		<input type="button" value="3"/> Phone 1
Hotline Number	Hotline Line	
	<input type="button" value="0"/> Phone 1	
DND	DND Line	
From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (hh:mm)	Phone 1	

*IP Line Forward function for VIP-157*

The IP Line Forward function’s incoming call is IP call type, and the destination is IP or PSTN call type. The FXO Line Forward function’s incoming call is PSTN call type, and the destination is IP call type. The IP / FXO Line Forward functions can be functioned at the same time, and that could separate different incoming call types for fixable applications.

**[Hotline Type]**

Hotline Type	Hotline Number	Hotline Line
<input type="button" value="Disable"/>		<input type="button" value="0"/> Phone 1

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

**Hotline Type:** Click Enable to carry the Hot line function out.

**Hotline number:** The hot line number can input the IP address or registration number.

**Delay time:** If you don’t dial for a period of time, it will automatically dial the hotline number.

**[DND Type]**

DND Type	DND	DND Line
<input type="button" value="Disable"/> <input type="button" value="Disable"/> <input type="button" value="Always"/> <input type="button" value="Period"/>	From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (hh:mm)	Phone 1
Alarm Time	Alarm Line	
<input type="text"/> : <input type="text"/> (hh:mm)	Phone 1	

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

**Always Block:** All incoming calls 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 larger than the “To” time, the Block time will from Day 1 to Day 2.

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

**[Alarm Type]**

Alarm Type	Alarm Time	Alarm Line
Disable <input type="button" value="v"/>	<input type="text" value="0"/> : <input type="text" value="0"/> (hh:mm)	Phone 1

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

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

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

**Alarm Line:** Select the Line for alarm.(only for VIP-157S)

**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, depending on how long you need to synchronize it again. User can also use the “daylight saving” to adjust the daylight time. When you finish the setting, please click the Submit button.

**SNTP Setting**

You could set the configuration of SNTP in this page.

<b>NTP Active:</b>	Auto <input type="button" value="v"/>
<b>Primary NTP:</b>	north-america.pool.ntp.org
<b>Secondary NTP:</b>	asia.pool.ntp.org
<b>Time Zone:</b>	GMT + <input type="button" value="v"/> 08 <input type="button" value="v"/> : 00 <input type="button" value="v"/> (HH:MM)
<b>Update Interval:</b>	6 Hour <input type="button" value="v"/>

<b>Manually Time</b>	<b>(Not use Daylight Saving Time)</b>					
<b>Date &amp; Time</b>	2005 <input type="button" value="v"/> Year	1 <input type="button" value="v"/> Month	1 <input type="button" value="v"/> Date	8 <input type="button" value="v"/> Hour	7 <input type="button" value="v"/> Minute	1 <input type="button" value="v"/> second
	<input type="button" value="Get PC Time"/>					

<b>Daylight Saving Time :</b>	Disable <input type="button" value="v"/>
<b>Offset:</b>	+ 2 Hour <input type="button" value="v"/>
<b>Start Time:</b>	Jan <input type="button" value="v"/> By Day <input type="button" value="v"/> 01 <input type="button" value="v"/> FirstWeek <input type="button" value="v"/> Sun <input type="button" value="v"/> 00 <input type="button" value="v"/>
<b>End Time:</b>	Jan <input type="button" value="v"/> By Day <input type="button" value="v"/> 01 <input type="button" value="v"/> FirstWeek <input type="button" value="v"/> Sun <input type="button" value="v"/> 00 <input type="button" value="v"/>

**Volume Setting**

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

**Handset Volume** is to set the volume you can hear from the handset.

**Handset Gain** is to set the volume sent out to the other side’s handset.

## Volume Setting

You could set the configuration of Volume in this page.

Handset Volume:	10 ▾
Handset Gain:	10 ▾

### *Volume Settings for VIP-156T/VIP-156PT*

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

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

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

## Volume Setting

You could set the configuration of Volume in this page.

Handset Volume:	10 ▾
Handset Gain:	10 ▾
PSTN-Out Volume:	10 ▾
PSTN-In Gain:	10 ▾

(10 representative is 0 dB and every scale is 3 dB)

### *Volume Settings for VIP-157*

## 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.

# Dial Plan Setting

You could set the dial plan in this page.

Index	Drop prefix	Prefix	Replace Rule
1	Disable ▾		
2	Disable ▾		
3	Disable ▾		
4	Disable ▾		

Index	Dial Now Rule
1	
2	
3	
4	
5	
6	
7	
8	

Realm 1 Area Code:	1*
Realm 2 Area Code:	2*
Realm 3 Area Code:	3*
Realm 4 Area Code:	4*
Realm 5 Area Code:	5*

Inter Digit Time:	5 ▾ (seconds)
Key As Send #:	Enable ▾

### Dial Plan Settings for VIP-156

Auto PSTN backup:	Disable ▾
PSTN feature code:	0*
Routing Type:	Disable ▾
Routing Rule:	

*For VIP-157, have four more items.*

Field	Description
<b>Drop Prefix</b>	The rule of add or replace code. If setup as Disable, it will add the prefix number prior to the identification number. If setup as Enable, it will replace the identification number.
<b>Prefix</b>	The prefix number. It only accepts the numeral and the max length is 8.
<b>Rule Rule</b>	The identification number. It can accept the numeral or symbol and the max length is 40. <ul style="list-style-type: none"> <li>- Symbol: It only accepts the [+], [x]</li> <li>- +: It means as "or". For example, [123+456+334+5xx] even if [123 or 456 or 334 or 5xx]</li> </ul>

	- x: It is equal to 0~9. For example, [5xx] even if the number begins with 5.
<b>Dial Now rule</b>	<p>If the dialing number matches this field, it will dial out and need not have to press the “#” key to end the dialing. It accepts the numeral or symbol, and the max length are 124.</p> <div style="border: 1px solid black; padding: 5px;">  <p>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.</p> </div>
<b>Realm 1/2/3/4/5 Area Code</b>	These options can define the switching code for each Realm No.
<b>Inter Digit Time(Auto Dial Time)</b>	Stop dialing after seconds then send dialed number out.
<b>Key as send #</b>	If setup as Yes, the system will stop to receive the dialed number when receiving the [#] key. The system also will determine the Auto Dial Time, It will carry out the calling if there is no dialing after the Auto Dial Time.If setup as No, the system will do according to the Auto Dial Time to determine the end time.

**Descriptions of example:**

## Dial Plan Settings

You could set the dial plan in this page.

Index	Drop prefix	Prefix	Rule
1	Disable	002	8613+8662
2	Enable	006	002+003+004+005+007+009
3	Disable	009	12
4	Disable	007	53+35xx+21xx

Index	Dial Now Rule
1	*xx+#xx+11x+xxxxxxx
2	
3	
4	
5	
6	
7	
8	

**Example\_1:** Drop prefix: **Disable**, Prefix: **002**, Rule: **8613+8662**

1. If the dialing number is “**8613xxxxx**”, it will match the rule [8613], then system will automatically add the prefix [002] in front of [8613].The real dialing number is [**002+8613xxxxx**].
2. If the dialing number is “**8662xxxxx**”, it will match the rule [8662], then system will automatically add the prefix [002] in front of [8662].The real dialing number is [**002+8662xxxxx**].

**Example\_2:** Drop prefix: **Enable**, Prefix: **006**, Rule: **002+003+004+005+007+009**

1. If the dialing number is "002+86xxxx", it will match the rule [002], then system will automatically replace the prefix [002] to the prefix number [006].The real dialing number is [006+8613xxxx].
2. If the dialing number is "003+77xxxx", it will match the rule [003], then system will automatically replace the prefix [003] to the prefix number [006]. The real dialing number is [006+77xxxx].

**Example\_3:** Drop prefix: **Disable**, Prefix: **009**, Rule: **12**

1. If the dialing number is "12xxxxx", it will match the rule [12], then system will automatically add the prefix [009] in front of [12].The real dialing number is [009+12xxxxx].

**Example\_4:** Drop prefix: **Disable**, Prefix: **009**, Rule: **53+35xx+21xx**

1. If the dialing number prefix is [53789], it will match the rule [53], then system will automatically add the prefix [007] in front of [53789].The real dialing number is [007+53789].
2. If the dialing number prefix is [3507], it will match the rule [35xx], then system will automatically add the prefix [007] in front of [3507].The real dialing number is [007+3507].
3. If the dialing number prefix is [2199], it will match the rule [21xx], then system will automatically add the prefix [007] in front of [2199].The real dialing number is [007+2199].

**Example\_5:** Dial Now: **\*xx+#xx+11x+xxxxxx**

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

[For VIP-157 only]

Auto PSTN backup:	Disable ▾
PSTN feature code:	0*
Routing Type:	Disable ▾
Routing Rule:	

Field	Description
Auto PSTN backup	Default is Disable, it's for PSTN backup function, when it "Enable", if SIP regisiter unsuccessful, it will automatically switch to PSTN line to dial out.

	 <p>To enable this function, make sure that the PSTN line is connected to the PSTN port already.</p> <p>Note</p>
<p><b>PSTN feature Code</b></p>	<p>Default is 0*, the code for <b>manually</b> switching to PSTN line, and dial out from PSTN, it can only accept the numeric and *or #, the digital max length is 7.</p>
<p><b>Routing Type</b></p>	<p>Default is Disable, it defines the dialing route, according the [Routing Rule] to define the dialing route which is [IP or FXO].</p>
<p><b>Routing Rule</b></p>	<p>Define the outgoing rule. It can also Add/ Drop prefix number, if you want to increase more then one Routing rule, you can use “+” to accept it by only pressing the numeric or D</p> <p>D: drop</p>

**Example\_5:** Routing: Routing Type: **FXO**, Routing Rule: **D007+009+0800**

1. If the dialing number is “**0800024365**”, it will match the routing rule [0800], then system will automatically dial out from the [**FXO**].
2. If the dialing number is “**00986123456**”, it will match the routing rule [009], then system will automatically dial out from the [**FXO**].
3. If the dialing number is “**00782280220**”, it will match the routing rule [D007], then system will decrease the [007] then dial out from the [**FXO**].The real dialing number is [**82280220**].

## General

This page defines the volume, auto answer, Caller ID, and call waiting caller ID(CID type II),

# General Setting

You could set the general options of your phone in this page.

Call Waiting:	Enable ▾
Ring Timeout:	60 ▾ (seconds)
Caller ID Scheme:	FSK (Bellcore) ▾
CID Type II:	Disable ▾
T.38 (FAX):	Enable ▾
T.38 Pass-Through Codec:	uLaw ▾

VIP-156

# General Setting

You could set the general options of your phone in this page.

Call Waiting:	Enable
Ring Timeout:	60 (seconds)
Caller ID Scheme:	FSK (Bellcore)
CID Type II:	Disable
T.38 (FAX):	Enable
T.38 Pass-Through Codec:	uLaw
Auto Answer Type:	Disable
Auto Answer Counter:	3
PIN Code:	Disable
PIN Code Number:	

## VIP-157

Field	Description
<b>Call Waiting</b>	Default is enable. When you are talking with other people, You can choose If you want to hear the notice when there is a new incoming call. If the call waiting function is On, if there is a new incoming 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.
<b>Ring Timeout</b>	Default is 60(sec). After how long the system will reply the busy(486 busy) message.
<b>Caller ID Scheme</b>	Set the caller ID mode, it supports FSK Bellcore, DTFM, CID-Japan, DTMF-Brazil, DTMF-Denmark. FSK Bellcore: FSK caller ID mode. DTMF: Before first ringing,it will send the DTMF caller ID data. CID-Japan: Japan (Japan) caller ID mode DTMF-Brazil: Brazil (Brazil) caller ID mode DTMF-Denmark: Denmark (Denmark) caller ID mode
<b>CID Type II</b>	To enable the show caller ID function in call waiting. When enable this function system receives a new call in call waiting. It will display the caller ID <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  Your Phone must also support CID Type2.         </div>
<b>T.38 (FAX)</b>	Enable/Disable T.38 FAX function.
<b>T.38 Pass-through codec</b>	Define the T.38 pass-through codec, it can support G.711 u-law/G.711 a-law.

**Auto Answer and PIN (VIP-157)**

<b>Field</b>	<b>Description</b>
<b>Auto Answer Type</b>	Auto Answer: There are different incoming call types for flexible applications. The Trunk Gateway function needs to arrange with the registered Server System. The 3-Party subscribers could make Off-Net call (PSTN) through the FXO port of VIP-157.
<b>AutoAnswer counter</b>	Auto Answer Counter is to set after the ring count met the number you set then the auto answer will enable.
<b>PIN Code</b>	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 to input the PIN Code. Then you can dial out. Please note that the PIN Code function couldn't function with Trunk Gateway function together.
<b>PIN Code Number</b>	

# Chapter 5

## Network

### Network Setting

This page defines the LAN setting on this page.

## WAN Setting

You could configure the Network settings in this page.

<b>WAN Active:</b>	Static IP Address ▾
<b>IP Address:</b>	192.168.0.156
<b>Subnet Mask:</b>	255.255.255.0
<b>Default Gateway:</b>	192.168.0.1
<b>DNS Active:</b>	Static ▾
<b>Primary DNS:</b>	8.8.8.8
<b>Second DNS:</b>	203.248.252.2
<b>MAC Address:</b>	00:30:4f:12:34:59
<b>System Name:</b>	VOIP_TA1S

Field	Description
<b>WAN Active</b>	The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes. <b>Fixed IP:</b> It could set up the IP address manually. <b>DHCP Client:</b> It will acquire the IP address automatically. <b>PPPoE:</b> It will use the PPPoE connection method
<b>IP Address</b>	The IP address
<b>Subnet Mask</b>	The sub net address
<b>Default Gateway</b>	The default gateway address
<b>DNS Active</b>	Static/ Automatically, manually set up the DNS server or automatically accept the DNS server.
<b>Primary DNS</b>	The default is 168.95.192.1, it could set up the first DNS server address.
<b>Second DNS</b>	The default is 168.95.1.1, it could set up the second DNS server address.
<b>MAC Address</b>	The MAC of LAN port
<b>System Name</b>	The product model
<b>PPPoE User Name</b>	The PPPoE connection account name. It could input numeral or character, the maximum date length is 63.
<b>PPPoE Password</b>	The PPPoE connection account password. It could input numeral or

	character, the maximum date length is 63.
<b>PPPoEService name</b>	PPPoE Service provider name
<b>PPPoE AC Name</b>	PPPoE AC name.

## DDNS Setting

This page defines the DDNS setting on this page. You need to have the DDNS account and input the information 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 work with an SIP Proxy Server. When you finish the setting, please click the Submit button.

(For better service, Planet provides **Planet DDNS**, you can apply your DDNS account on web site [www.planetddns.com](http://www.planetddns.com))

## Dynamic DNS Setting

You could set the configuration of DDNS in this page.

<b>DDNS Active:</b>	Disable ▾
<b>Host Name:</b>	<input type="text"/>
<b>User Name:</b>	<input type="text"/>
<b>Password:</b>	<input type="text"/>
<b>E-mail Address:</b>	<input type="text"/>
<b>DDNS Server List:</b>	members.dyndns.org ▾
<b>DDNS Server:</b>	<input type="text"/>
<b>Dynamic DNS Type:</b>	dyndns ▾
<b>Wild Card:</b>	Disable ▾
<b>BACKMX:</b>	Disable ▾
<b>Off Line:</b>	Disable ▾ (Only applies to custom DNS)

### **DDNS Settings for VIP-156/VIP-156PE/VIP-157**

## VLAN Setting

This page defines the VLAN setting on this page. This function needs to co-operate with network devices which have VLAN function, also this page defines the SIP and RTP port number on this page. Each ISP provider will have a different SIP/RTP port setting. Please refer to the ISP to set up the port number correctly. When you finish the setting, please click the Submit button.

# VLAN Setting

You could set the configuration of VLAN in this page.

Network (Both WAN & LAN)	
VLAN Active:	Disable ▾
VID (802.1Q/TAG):	136 (3~4094)
User Priority (802.1P):	0 ▾
SIP & RTP	
SIP VID:	0 (3~4094, 0: "Disable")
SIP User Priority (802.1P):	0 ▾
RTP VID:	0 (3~4094, 0: "Disable")
RTP User Priority (802.1P):	0 ▾

Field	Description
VLAN Activity	If set up as On, it could receive VLAN messages.
VID (802.1Q/TAG)	Dispose VLAN ID is add a Tag header after enabling the VLAN function. The realized voice packets transfer is similar with that of VLAN. The prerequisite is it must be the same as VLAN of upper switch. The value range is 2~4094.
User Priority (802.1P)	To set up the user priority.

Field	Description
SIP VID	Set the SIP VLAN ID, this is an independent function that doesn't need to enable [VLAN Packets: Enable].
SIP User Priority (802.1P)	Set up the SIP Priority.
RTP VID	Set the SIP RTP VID, this is an independent function that doesn't need to enable [VLAN Packets: Enable].
RTP User Priority (802.1P)	Set up the RTP Priority.

## VPN Setting

This page defines the PPTP/L2TP setting on this page. You could set up the PPTP/L2TP Server connection information. When you finish the setting, please click the Submit button.

## VPN Setting

You could set the configuration of VPN in this page.

<b>VPN Active:</b>	Disable ▾
<b>Server Name:</b>	<input type="text"/>
<b>User Name:</b>	<input type="text"/>
<b>Password:</b>	<input type="password"/>
<b>Port:</b>	Default ▾ <input type="text" value="1723"/> (1024-65535, Only Support PPTP)

**Caution:** VIP-156/VIP-157 VPN can't use the encryption or compression for VPN connection.

## IPV6 Setting

### IPv6 Setting

You could set the configuration of IPv6 in this page.

IPv6 Connection	
<b>IPv6 Active:</b>	Static IPv6 Address ▾
WAN IPv6:	
<b>IPv6 Address:</b>	<input type="text" value="2001:b021:47:0:0:0:102"/>
<b>Subnet Prefix Length:</b>	<input type="text" value="64"/>
<b>Default Gateway:</b>	<input type="text" value="0:0:0:0:0:0"/>
LAN IPv6	
<b>LAN IPv6 Address:</b>	<input type="text" value="2001:0:0:0:0:0:1"/> /64
<b>LAN IPv6 Link-Local Address:</b>	<b>fe80:0:0:0:230:4fff:fe12:3458</b>
Address Autoconfiguration	
<b>Autoconfiguration Type:</b>	Stateless ▾

This page defines the IPV6 setting on this page. You can program the IPV6 information.

Field	Description
<b>IPV6 Activity</b>	Support three IPV6 types: Auto, Fixed IPV6, IPV6 in IPV4 Tunnel
<b>IPV6 address</b>	Setting the WAN IPV6 address or display it.(64 bits)
<b>SubnetPrefix Length</b>	Default is 64
<b>Default Gateway</b>	IPV6 gateway address(64 bits)
<b>LAN IPv6 Address:</b>	IP V6 LAN address. (64 bits)
<b>LAN IPv6 Link-Local Address</b>	Link local address information.
<b>Autoconfiguration Type</b>	It supports Statless, stateful(DHCP V6).

# Chapter 6

# 6

## NAT Trans

### Stun Setting

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

## STUN Setting

You could set the IP of STUN server in this page.

STUN Active:	Disable
STUN Server Name:	stun.xten.com
STUN Port:	3478 (80~65535)
Force Active:	Disable
Public IP Address:	
Public Port:	5060 (80~65535)

### PC Setting

This page defines the PC setting on this page.

## PC Setting

You could set the configuration of PC in this page.

Device Active:>	Router
PC IP Address:	192.168.123.11
PC MAC Address:	00:30:4f:12:34:58
Enable DHCP Server:	Enable
IP Address:	150 ~ 200 (Start ~ End, 1~254)
Lease Time:	1440 (10~10080 Minute)

Field	Description
Device Activity	<p>The default is Bridge mode, and it also provides NAT mode.</p> <p><b>Bridge:</b> When set as mode, the LAN and PC ports are in the same network segment.</p> <p><b>NAT:</b> The LAN and PC ports are in a different network segment, and PC port could enable the DHCP Server function to allot the IP</p>

	address.
<b>PC IP address</b>	The IP address of PC port. (In the <b>Birdge</b> mode, the <b>Default IP:</b> 192.168.0.1
<b>PC MAC Address</b>	The MAC of PC port
<b>Enable DHCP Server</b>	It will allot the IP address automatically when enable this function.
<b>IP Address</b>	The range for DHCP IP address.
<b>Lease Time</b>	DHCP server lease time

## DMZ and MAC Clone

This page defines the DMZ and MAC Clone setting on this page.

**DMZ Activity:** If set up as On, all of the packets (except SIP packets) will send to the specific IP address.

**DMZ IP Address:** The DMZ host IP address.

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

## DMZ and MAC Clone Setting

You could set the configuration of DMZ and MAC Clone in this page.

<b>DMZ Active:</b>	Disable ▾
<b>DMZ IP Address:</b>	0.0.0.0
<b>MAC Clone Active:</b>	Disable ▾

## Virtual Server

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

# Virtual Server Setting

You could set the configuration of Virtual Server in this page.

Index	Active	Protocol	Internet Port	Extranet Port	Server IP Address	Action
			Start ~ End	Start ~ End		
1	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
2	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
3	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
4	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
5	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
6	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
7	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
8	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
9	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
10	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
11	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete
12	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	Delete

Field	Description
Index	The serial number. There are totally 12 records from Num 1 to 12.
Activity	The activity status. The default is Disable, this record will activate if enable.
Protocol	The TCP or UDP communication protocol.
Internal Port	For corresponding the internal port.
External Port	For corresponding the external port.
Server IP	To input the Server IP address.
Delete	Delete this item

# Chapter 7

## SIP Setting

### Service Domain Setting

In Service Domain Function, you need to input the account and the related information on this page. Please refer to your ISP provider. You can register five SIP accounts in the ATA. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these five SIP accounts.

## Service Domain Setting

You could set the configuration of Service Domain in this page.

Realm: 1

Realm Active:	Disable
Display Name:	<input type="text"/>
Phone Number:	<input type="text"/>
Authentication ID:	<input type="text"/>
Authentication Password:	<input type="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Subscribe for MWI :	Disable

Field	Description
Realm	Which line you want to use.
Realm Activity	First you need to click Active to enable the Service Domain, then you can input the following items.
Display Name	The serial number. There are totally 24 records from Num 0 to 23.
Phone number	The activate status. The default is Disable, this record will activate if enable.
Authentication ID	you need to input the Register Password obtained from your ISP.
Authentication Password	you need to input the Register Name obtained from your ISP.
Domain Server	you need to input the Domain Server from your ISP.
Proxy Server	you need to input the Proxy Server from your ISP.
Outbound Proxy	you need to input the Outbound Proxy from your ISP. If your ISP does not to provide the information. Then you can skip this item.
Subscribe for MWI	Setting MWI(message-waiting indicator) function, when enable

	<p>system, it will frequently send the MWI message.</p> <div style="border: 1px solid black; padding: 5px;">  <p>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.</p> </div>
--	--

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

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

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

Register Information			
Realm 1 Status:	Not Registered	Number:	156
Realm 2 Status:	Registered	Number:	1005
Realm 3 Status:	Not Registered	Number:	
Realm 4 Status:	Not Registered	Number:	
Realm 5 Status:	Not Registered	Number:	

### Codec Setting

This page defines the Codec priority, RTP packet length, and VAD function on this page. You need to follow the ISP suggestion to set up these items. When you finish the setting, please click the Submit button. Also on this page, it defines the Codec ID. Sometimes two VoIP devices with different Codec IDs will cause the interoperability issue. If you are talking with some people with some problems, you may ask the other one what kind of Codec ID he uses, and then you can change your Codec ID. When you finish the setting, please click the Submit button.

# Codecs Setting

You could set the configuration of Codec in this page.

Disable Codecs		Enable Codecs
G.726 - 16 G.726 - 24 G.726 - 32 G.726 - 40	<input type="button" value=" &gt;&gt;"/>  <input type="button" value=" &lt;&lt;"/>	G.711 u-law G.711 a-law G.723 G.729
	<b>Move</b>	<input type="text" value="[object HTMLInputElement]"/> <input type="text" value="[object HTMLInputElement]"/>

G.711 and G.729:	20 <input type="button" value="v"/> ms
G.723:	30 <input type="button" value="v"/> ms
G.723 5.3K:	Disable <input type="button" value="v"/>
Silence Suppression (VAD):	Disable <input type="button" value="v"/>
Echo Canceller :	Enable <input type="button" value="v"/>

Codec Type	ID Value		
G726-16:	Default <input type="button" value="v"/>	23	(95~127)
G726-24:	Default <input type="button" value="v"/>	22	(95~127)
G726-32:	Default <input type="button" value="v"/>	2	(95~127)
G726-40:	Default <input type="button" value="v"/>	21	(95~127)
RFC 2833:	Default <input type="button" value="v"/>	101	(95~127)

## SIP Advance Setting

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

# SIP - Advanced Setting

You could set the configuration of SIP Common in this page.

SIP Expire Time:	300	(60~86400 Seconds, 0=define by Server)
SIP Expire Time Type:	General	(General: Expire Time - [Expire Time/6])
SIP Registration Retry Timer:	64	(5~250 Seconds)
SIP Session Timer T1:	500	(ms)
SIP Session Timer T2:	4000	(ms)
SIP Session Timer B, F, H:	32000	(ms)
Local SIP Port of Phone 1:	10000 ~ 10999	(1024~40000, Start ~ End)
Local RTP Port of Phone 1:	20000 ~ 21999	(1024~40000, Start ~ End)
Hold Type:	RFC 2543 (0.0.0.0)	
DTMF Type:	RFC 2833	
RPort:>	Disable	
Voice QoS (Diff-Serv):	40	(0~63)
SIP QoS (Diff-Serv):	40	(0~63)
RTP Traffic Class (IPv6):	46	(0~255)
SIP Traffic Class (IPv6):	40	(0~255)
Use DNS SRV:	Disable	
Keep-alive Message:	Disable	
Keep-alive Interval:	60	(15~250 Seconds)
Jitter Buffer:	1 ~ 32	(1~32 Packet)
SIP Server Type:	General	
Use user=phone (Register):	Disable	
Use user=phone (Invite):	Disable	
Send SIP PRACK to Proxy:	Disable	
Only Accept Trusted Certificates:	Disable	

Field	Description
SIP Expire Time	To set up the registration interval time.
SIP Expire Time Type	<p>Default is General; Register interval time setting. Provide items like General (standard), 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10 ◦</p> <div style="border: 1px solid black; padding: 5px; margin: 10px 0;">  <p>Register server need supports this function</p> </div> <p>Register time calculated            General: expiry time-[expiry time/30]*6, when Expiry Time&gt;60 it will start to work, if less than 60 seconds, it will decrease 5 seconds.            1/2: expiry time * 1/2.            2/3: expiry time * 2/3.            3/4: expiry time * 3/4.            4/5: expiry time * 4/5.</p>

	<p>5/6: expiry time * 5/6.</p> <p>6/7: expiry time * 6/7.</p> <p>7/8: expiry time * 7/8.</p> <p>8/9: expiry time * 8/9.</p> <p>9/10: expiry time * 9/10.</p>
<b>SIP Register Retry Timer</b>	If SIP register fails, system will retry interval after this time.
<b>SIP session timer T1</b>	<p>Setting the maximum retransmit interval for non-INVITE requests and INVITE responses.</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <span style="margin-left: 10px;">Register server need supports this function</span> </div>
<b>SIP session timer T2</b>	<p>Setting the maximum retransmit interval for non-INVITE requests and INVITE responses.</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <span style="margin-left: 10px;">Register server need supports this function</span> </div>
<b>SIP session timer Timer B, F, H</b>	<p>Setting the maximum retransmit interval for non-INVITE requests and INVITE responses ◦ ◦</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <span style="margin-left: 10px;">Register server need supports this function</span> </div> <p>B: 64 * SIP T1; INVITE transaction timeout timer ◦</p> <p>F: 64 * SIP T1; non-INVITE transaction timeout timer ◦</p> <p>H: 64 * SIP T1, Wait time for ACK receipt ◦</p>
<b>Local SIP Port of phone 1</b>	Setting the phone 1 SIP start and end port. All the port can't be duplicated
<b>Local RTP Port of phone 1</b>	Setting the phone 1 RTP start and end port. All the port can't be duplicated
<b>Hold type</b>	The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.
<b>DTMF Mode</b>	defines the InBand, RFC2833, SIP Info, RFC2833 + Inband, SIP Info + Inband on this page. To change this setting, please follow your ISP information. When you finish the setting, please click the Submit button.
<b>RPort</b>	To change this setting, please follow your ISP information. When you finish the setting, please click the Submit button.

	 <p>Register server need supports this function</p>
<b>Voice QoS (Diff-Serv)</b>	The Voice QoS feature.
<b>SIP QoS (Diff-Serv)</b>	<p>The SIP QoS feature.</p> <p>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 needs to cooperate with the other Internet devices.</p>
<b>RTP Traffic Class(IPV6)</b>	IPV6 RTP traffic class
<b>SIP Traffic Class(IPV6)</b>	IPV6 SIP traffic class
<b>Use DNS SRV</b>	The default is disable, and use DNS SRV mode. Set enable to use DNS to SRV mode to search the host information.
<b>Send Keep Alive Packet</b>	Always transport the network packets to keep the NAT port
<b>Keep Alive Period</b>	To set up the interval time for transporting packets.
<b>Jitter Buffer</b>	To set up the size for jitter buffer packets.
<b>SIP Server Type</b>	<p>Provides a different register server: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink, for different server systems will adjust some system parameters</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <p>Register server need supports this function</p> </div>
<b>Use user = phone (Register):</b>	<p>When sending the register package, in package Header will add the "user=phone" message ◦</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <p>Register server need supports this function</p> </div>
<b>Use user = phone (Invite):</b>	<p>When sending the dialing package, in package Header will add the "user=phone" message ◦</p> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">  <p>Register server need supports this function</p> </div>

<p><b>Send SIP PRACK to Proxy:</b></p>	<p>When sending the SIP package, in package Header will add the "PRACK" message ◦</p> <table border="1" data-bbox="566 280 1276 436"> <tr> <td data-bbox="566 280 726 436">  Note                 </td> <td data-bbox="726 280 1276 436"> <p>Register server need supports this function</p> </td> </tr> </table>	 Note	<p>Register server need supports this function</p>
 Note	<p>Register server need supports this function</p>		
<p><b>Only Accept Trusted Certificates:</b></p>	<p>Only accept call from proxy, if system receives the IP dialing, system will refuse the call.</p>		

# Chapter 8

## Advance Setting



### Status Log

Display and save systems running status message data. Press “Get Status Log” to back up the status log file.

### View Log

You could get the log of Status in this page.

Page:

Index	Message
0	<2005-01-01 00:00>
1	<2005-01-01 00:00>Enable DHCP_SERVER
2	<2005-01-01 00:00>Init Lan Interface!
3	<2005-01-01 00:00>Iface type : FIXED_IP
4	<2005-01-01 00:00>Init Wan Interface!
5	<2005-01-01 00:00>Application starting ...
6	
7	
8	
9	
10	
11	
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23	
24	

### Auto Config

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.

## Auto Provision Setting

You could set the configuration of Auto Configuration in this page.

Provision Active:	Disable
2 Steps Configuration:	Disable
Server Auto Discovery:	Disable
Scheduling:	Disable

TFTP Server:	
TFTP File Path:	Exp. download/

HTTP Server:	Exp. 60.35.187.30
HTTP File Path:>	Exp. download/

FTP Server:	Exp. 60.35.187.30
FTP User Name:	
FTP Password:	
FTP File Path:	Exp. fileload/

Next Configuration time:

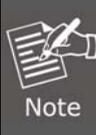
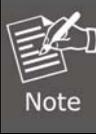
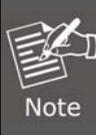
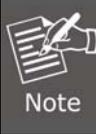
## Management-Advanced Setting

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

## Management - Advanced Setting

You could set the configuration of Management-Advanced in this page.

ICMP Not Echo:	Disable
Anonymous Call:	Disable
Management from WAN:	Enable
Stop Feature Tone:	Disable (MWI, forward, Do Not Disturb....)
Billing Signal:	Disable
CPC Delay:	2 (Seconds)
CPC Duration:	0 (0~120; x 10ms)
IP Dialing Format:	Type 1 (x@x.x.x.x)
Send Flash Event:	Disable
Encryption Type:	Disable
Encryption Key:	
PPPoE Retry Period:	5 (0~250 Seconds)
DHCP Gateway ARP Check Period:	0 (0 or 30~300 Seconds)
Syslog Server IP Address:	
System Log:	Disable
FXS Port Country:	USA
Flash Hook Time (MAX):	60 (4~255; x 10ms)
Flash Hook Time (MIN):	7 (3~12; x 10ms)
NET Bandwidth Limit:	Disable Kbps

Field	Description
<b>ICMP Not Echo</b>	This function can disable echo when someone pings this device. It can avoid hacker trying to attack the device
<b>Anonymous Call</b>	<p>If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information.</p> <div style="border: 1px solid black; padding: 5px;">  <p>Register server need supports this function.</p> </div>
<b>Management form WAN</b>	When [Enable] allow user login from WAN.
<b>Stop Feature Tone</b>	When [Enable] if system sets the function like [Subscribe for MWI, forward, DND], when user picks up the phone, he will hear the remind tone [Do Do Do]
<b>Billing Signal</b>	<p>There are three billing types: Polarity Reversal, Tone_12K and Tone_16K.</p> <div style="border: 1px solid black; padding: 5px;">  <p>Register server need supports this function.</p> </div>
<b>CPC Delay</b>	When receiving the disconnected signal, machine will cut the voltage down to 0V after this time
<b>CPC Duration</b>	When starting to cut the voltage down to 0V, machine will continue this state by this time.
<b>IP Dialing Format</b>	Setting IP dialing format, when [Disable] can't use IP dialing to make call.
<b>Send Flash event</b>	There are two flash formats: DTMF Event and SIP Info.
<b>Encrypt Type</b>	<p>There are seven encrypt formats: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP .</p> <div style="border: 1px solid black; padding: 5px;">  <p>Registered server need supports this function.</p> </div>
<b>Encrypt Key</b>	<p>Some encrypt type must enter the Encrypt Key</p> <div style="border: 1px solid black; padding: 5px;">  <p>Registered server need supports this function.</p> </div>
<b>PPPoE Retry Period</b>	If PPPoE dial-up connection fails, machine will retry the dial-up motion after this time.

<b>DHCP Gateway ARP Check Period</b>	The period to check the DHCP gateway ARP.
<b>Syslog Server IP Address</b>	There are seven Syslog types: Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug and All.
<b>System Log</b>	Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address
<b>PSTN port Country</b>	Set up the FXS Port Coutry
<b>PSTN Silence Timeout</b>	Define the MAX silence time for FXO port. After the time, it will disconnect the line.
<b>PSTN CID forward</b>	It must work with [Phone – General] [Auto Answer] function or [Phone – Caller Service] [Forward] function . When enable this function, the caller ID from FXO can transfer to other devices.
<b>Generate Flash Signal for PSTN</b>	FXO flash time defines would you hold or hang on the phone
<b>FXS Port Coutry</b>	Select the FXO port local country
<b>Flash Hook Time (Max)</b>	Maximum flash time, to detect the call on hold or hang on.
<b>Flash Hook Time (Min)</b>	Minimum Flash time , to detect the call on hold or hang on.
<b>NET Bandwidth Limit</b>	Setting the limitation for LAN Bandwidth

## Tones

This page defines the Tone settings. This function can set up the related parameters of Dial Tone, Ring Back Tone, Busy Tone, Error Tone and other Tone. When you finish the setting, please click the Submit button.

# Tones Setting

You could set the configuration of Tones in this page.

	Dial	Ring Back	Busy	Congestion	Ring	Call Waiting
<b>Cadence On:</b>	<input type="checkbox"/>	<input checked="" type="checkbox"/>				
<b>Hi-Tone Freq.:</b>	440	480	620	620	480	440
<b>Lo-Tone Freq.:</b>	350	440	480	480	440	350
<b>Hi-Tone Gain:</b>	4522	2261	2261	2261	15360	2261
<b>Lo-Tone Gain:</b>	4522	2261	2261	2261	15360	1130
<b>On Time 1:</b>	0 x 10ms	200	50	30	200	30
<b>Off Time 1:</b>	0 x 10ms	400	50	20	400	20
<b>On Time 2:</b>	0 x 10ms	0	0	0	0	30
<b>Off Time 2:</b>	0 x 10ms	0	0	0	0	400
<b>On Time 3:</b>	0 x 10ms	0	0	0	0	0
<b>Off Time 3:</b>	0 x 10ms	0	0	0	0	0

## TR-069

On this page you can program the TR-069 setting.



Note

Different TR-069 server may need to modify some different parameters.

What's TR-069: Technical Report 069 (TR-069) is a customer-premises equipment WAN management protocol (CWMP) technical specification for remote management of end-user devices introduced by the broadband forum (formerly the DSL forum).TR-069 is an integrated framework equipped with safe auto-configuration. It also can take control of other CPE functions .

## TR-069 Setting

You could set the configuration of TR069 in this page.

<b>ACS Service:</b>	Disable ▾
<b>ACS Interval:</b>	60 (30~86400 Seconds)
<b>Provision Code:</b>	provisioningCode
<b>ACS URL:</b>	http://iop.tw.workssys.com/comserver/node1/tr069
<b>ACS Username:</b>	
<b>ACS Password:</b>	
<b>Device S/N :</b>	
<b>Connection Request TCP:</b>	Enable ▾
<b>Connection Request TCP SSL:</b>	Disable ▾
<b>Connection Request TCP Port:</b>	7547 (1~65535)
<b>Connection Request UDP:</b>	Enable ▾
<b>Connection Request UDP SSL:</b>	Disable ▾
<b>Connection Request UDP Port:</b>	7547 (1~65535)
<b>Connection Request Path:</b>	
<b>Connection Request Authority:</b>	0 ▾
<b>Connection Request Username:</b>	
<b>Connection Request Password:</b>	
<b>ISP Username:</b>	
<b>ISP Password:</b>	

# Chapter 9

## Other Setting



### System Authority

In System Authority it can change admin/System/User login password.

## System Authority

You could change the login username/password in this page.

#### Admin

<b>New User Name:</b>	<input type="text"/>
<b>New Password:</b>	<input type="text"/>
<b>Confirmed Password:</b>	<input type="text"/>

#### System

<b>New User Name:</b>	<input type="text"/>
<b>New Password:</b>	<input type="text"/>
<b>Confirmed Password:</b>	<input type="text"/>

#### User

<b>New User Name:</b>	<input type="text"/>
<b>New Password:</b>	<input type="text"/>
<b>Confirmed password:</b>	<input type="text"/>

### Firmware Upgrade

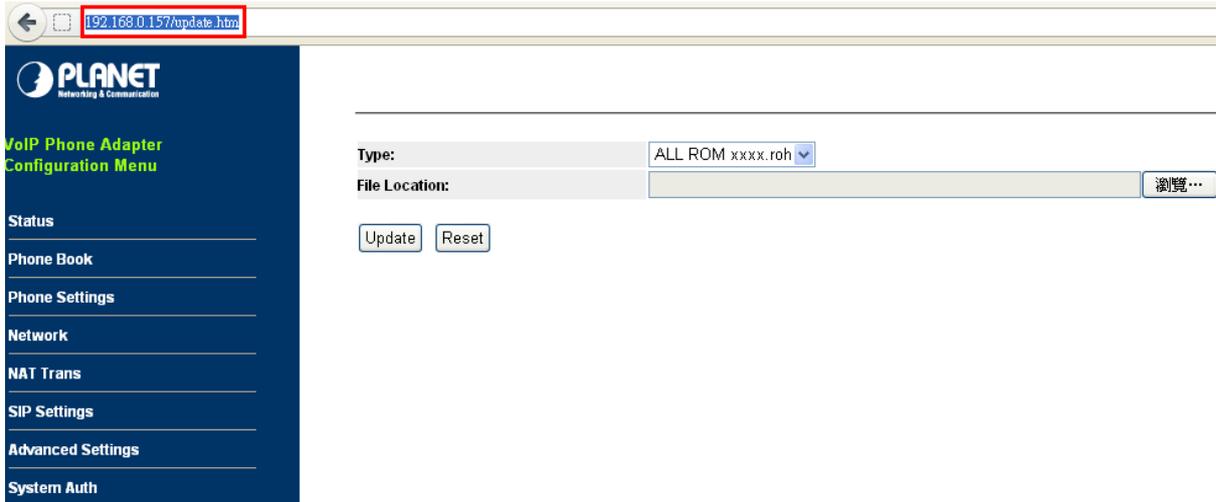
This page defines the SIP and RTP port number on this page. Each ISP provider will have different SIP/RTPport settings, please refer to the ISP to set up the port number correctly. When you finish the setting, please click the Submit button.

## Firmware Upgrade

You could update the newest firmware.

<b>Type:</b>	CPU+DSP xxxx.ssh ▾
<b>File Location:</b>	<input type="text"/> <input type="button" value="瀏覽..."/>

If your update file is xxxx.ROM. you must enter <http://VIP-15X's-IP Address/update.htm>  
 e.g.http://192.168.0.157/update.htm. To upload the ROM file and then update the system.



 <b>Note</b>	<p>For technological consideration, we strongly suggest you refer to the following upgrade methods for updating your device.</p> <p>After firmware is loaded, the unit will reboot, and Default IP address of the customized firmware: <a href="http://192.168.0.1">http://192.168.0.1</a>; login name/password: <b>root/null (no password)</b></p>
--	---

### Auto Update Setting

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

Field	Descriptions
<b>Type</b>	There are TFTP/ FTP and HTTP to provide the auto upgrade function.
<b>TFTP Server</b>	Input the TFTP Server address, and it could input the IP or Domain Name form.
<b>TFTP File Path</b>	Set up the file path.
<b>HTTP Server</b>	Input the HTTP Server address, and it could input the IP or Domain Name form.
<b>HTTP File Path</b>	Set up the file path.
<b>FTP Server</b>	Input the FTP Server address, and it could input the IP or Domain Name form.
<b>FTP Username</b>	The login username.
<b>FTP Password</b>	The login password
<b>FTP File Path</b>	Set up the file path.
<b>Check new firmware</b>	<p>The device will do according to the ways below to check the new firmware.</p> <ul style="list-style-type: none"> <li>- <b>Power On (+ Scheduling)</b>: The machine will check the new firmware when power on and following scheduled date and</li> </ul>

	<p>time.</p> <ul style="list-style-type: none"> <li>- <b>Scheduling:</b> The machine will follow the scheduled date and time to check the new firmware.</li> </ul>
<b>Scheduling (Date)</b>	The machine will check the new firmware between the time range by random.
<b>Automatic Update</b>	<p>There are Notify only and Automatic ways to update.</p> <ul style="list-style-type: none"> <li>- <b>Notify only:</b> If there are new firmware, the ATA will send the "Be Be Be" sound when picking up the handset to prompt there is new firmware.</li> <li>- <b>Automatic:</b> The device will carry out firmware update automatically.</li> </ul>
<b>Firmware File Prefix</b>	It will check the information of model name.
<b>Next update time</b>	It will show the next check date and time.

  
 Note

If the **Check new firmware** field selected to Power On, the machine will check the new firmware according the scheduled time/date and power on. If there is new firmware, it can be upgraded. The machine won't carry out firmware update automatically The machine will send the prompt sounds when picking up the handset, and it needs to update firmware manually.

## Auto Update Settings

You could set auto update settings in this page.

<b>Type:</b>	Disable <input type="button" value="v"/>
<b>TFTP Server:</b>	<input type="text"/>
<b>TFTP File Path:</b>	<input type="text"/> Exp. download/
<b>HTTP Server:</b>	<input type="text"/> Exp. 60.35.187.30
<b>HTTP File Path:</b>	<input type="text"/> Exp. download/
<b>FTP Server:</b>	<input type="text"/> Exp. 60.35.17.1
<b>FTP User Name:</b>	<input type="text"/>
<b>FTP Password:</b>	<input type="text"/>
<b>FTP File Path:</b>	<input type="text"/> Exp. file/load/
<b>Check New Firmware Type:</b>	Scheduling only <input type="button" value="v"/>
<b>Scheduling (Date):</b>	14 (1~30 days)
<b>Scheduling (Time):</b>	AM 00:00- 05:59 <input type="button" value="v"/>
<b>Automatic Update:</b>	Notify only <input type="button" value="v"/>
<b>Firmware File Prefix:</b>	TA1S10
<b>Next Update time:</b>	

## Reset to default

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

# Reset to Default

You could click the restore button to restore the factory settings.

Restore Default Setting:

## Save and 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.

# Save & Reboot

You have to save changes to effect them.

Save Change:

## Logout

Lougout the system, it will return to login page.

# Logout

You could click the logout button to logout.

Are you sure to logout ?

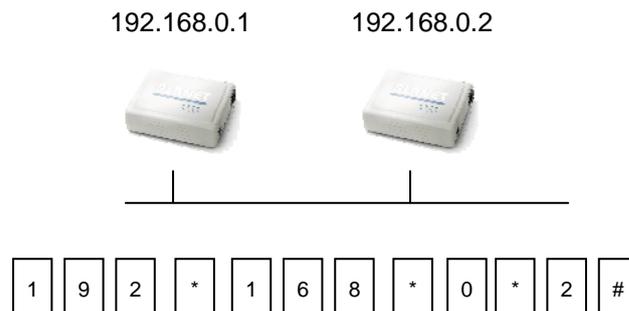
## Appendix A Voice Communication Samples

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 addresses are 192.168.0.1, 192.168.0.2

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



#### Test the scenario:

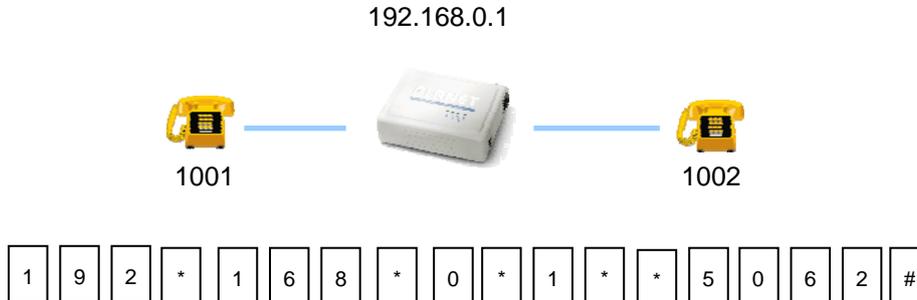
1. Pick up the telephone set on ATA A.
2. Press the keypad: **192\*168\*0\*2#** and will be able to connect to the ATA B.
3. Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

## 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



### Test the scenario:

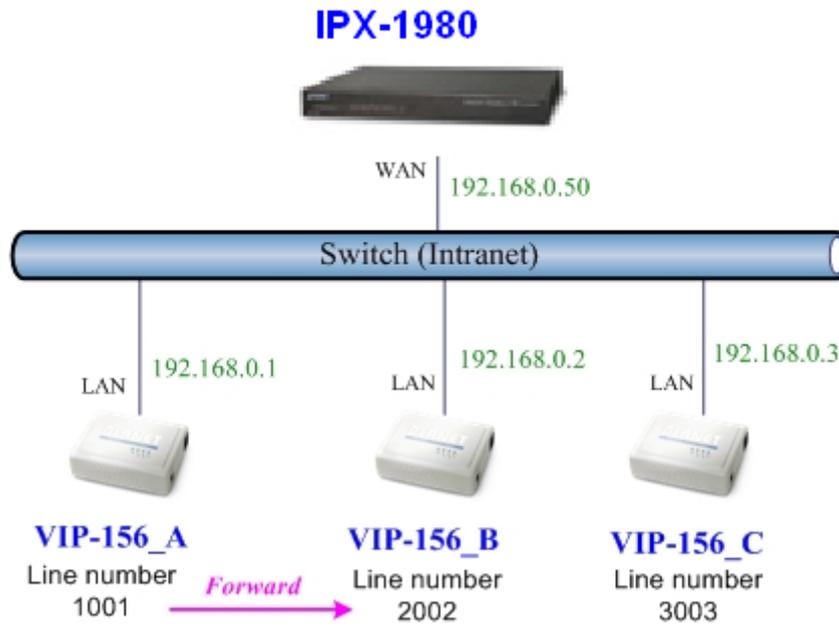
1. Pick up the telephone set on VIP-157S port 1, and you should be able to hear the dial-tone
2. Press the keypad: **192\*168\*0\*1\*\*5062#** and will be able to connect to the VIP-157S port 2
3. Then the telephone set in VIP-157S port 2 should ring. Please repeat the same dialing steps on port 2 to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

 <b>Note</b>	<ul style="list-style-type: none"> <li>• If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with a "#".</li> <li>• 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</li> <li>• [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:  <b>[remote ATA IP address]**5062</b>, for example: <b>192*168*0*2**5062</b></li> </ul>
--	--

## Case 3: Call Forward Feature\_Example 1

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

For this example, there are three VIP-156 register to IPX-1980 and VIP-156\_A has set Call Forward function to VIP-156\_B.



### Machine configuration on the VIP-156:

Please log in VIP-156\_A via web browser and browse the **Phone Settings** menu and select the **Call service** config menu. On the setting page, please enable the **All Forward** function and fill in the **Forward Type and Forward Number** of VIP-156\_B, then the sample configuration screen is shown below:

## Call Service

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

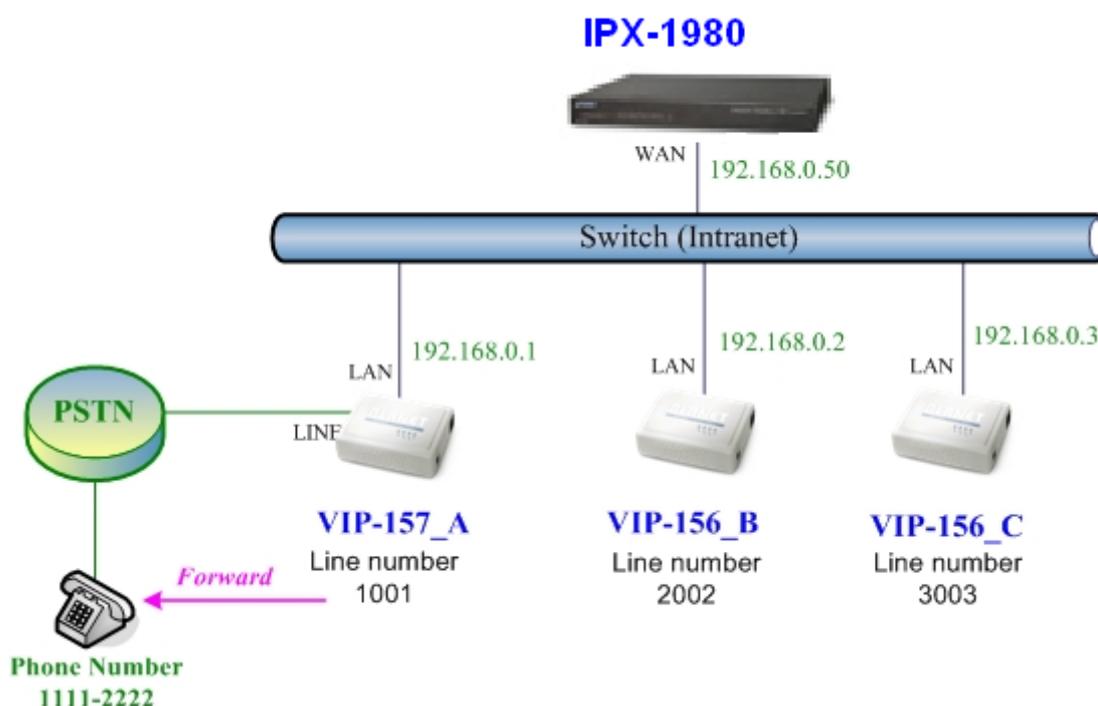
Forward Type	Forward Number	Ring / Phone
Always <input type="button" value="v"/>	2002	3 <input type="button" value="v"/> Phone 1

### Test the scenario:

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

### Case 4: Call Forward Feature\_Example 2

For this example, there are one VIP-157 and two VIP-156 register to IPX-1980. The VIP-157\_A has 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 and browse the **Phone Settings** menu and select the **Call service** config menu. On the setting page, please select the **All Forward** function to **PSTN** choice and fill in the **Forward Type and Forward Number** of PSTN Phone Number 11112222, then the sample configuration screen is shown below:

## Call Service

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

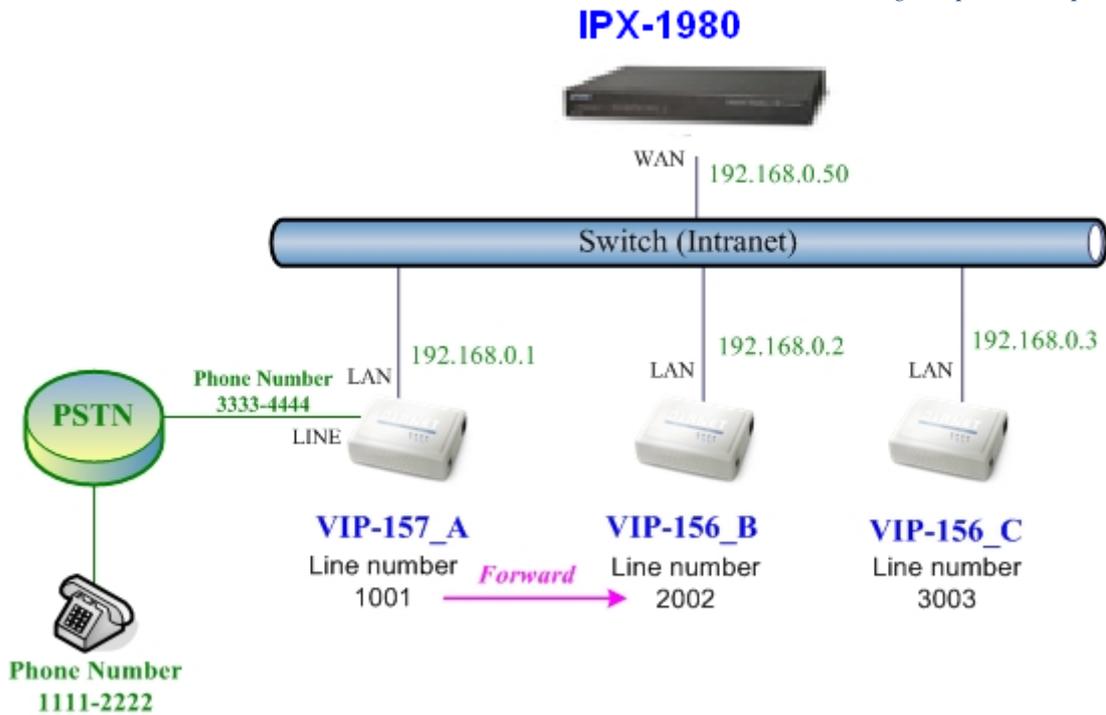
Forward Type	Forward Number	Ring / Phone
Always to PSTN <input type="button" value="v"/>	1111 2222	3 <input type="button" value="v"/> Phone 1

### Test the scenario:

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

### Case 5: Call Forward Feature\_Example 3

For this example, there are one VIP-157 and two VIP-156 register to IPX-1980. The VIP-157\_A has 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 and browse the **Phone Settings** menu and select the **Call service** config menu. On the setting page, please select the **All Forward** function to **IP** choice and fill in the **Forward Type and Forward Number** of of VIP-156\_B, and then the sample configuration screen is shown below:

## Call Service

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

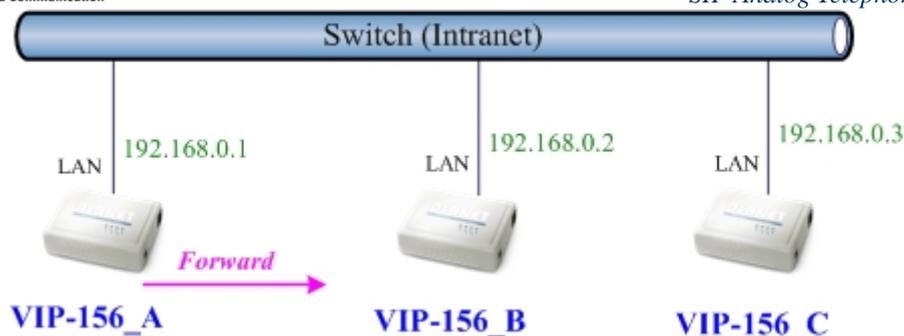
Forward Type	Forward Number	Ring / Phone
Always <input type="button" value="v"/>	2002	3 <input type="button" value="v"/> Phone 1

### 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 picks up the telephone and communication with the PSTN Phone Number 11112222

### Case 6: Call Forward Feature\_Example 4

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



### Machine configuration on the VIP-156:

Please log in VIP-156\_A via web browser and browse the **Phone Settings** menu and select the **Call service** config menu. On the setting page, please enable the **All Forward** function and fill in the **Forward Type and Forward Number** of VIP-156\_B, and then the sample configuration screen is shown below:

## Call Service

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

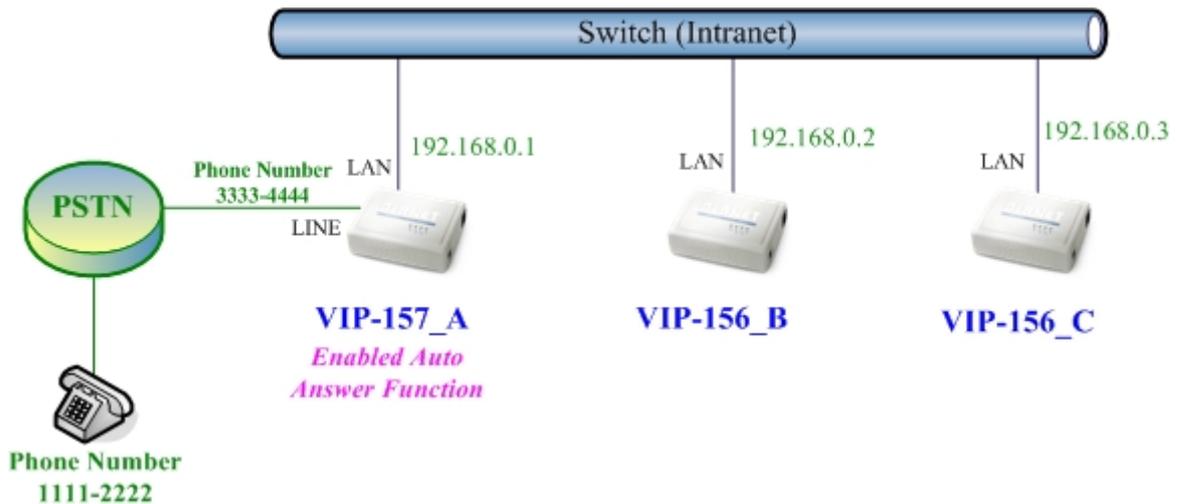
Forward Type	Forward Number	Ring / Phone
Always to PSTN <input type="button" value="v"/>	192.168.0.2	3 <input type="button" value="v"/> Phone 1

### 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 has 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 picks up the telephone and communication with the VIP-156\_C

### Case 7: Auto Answer Feature\_IP to PSTN

For this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157\_A has set Auto Answer function for forwarding calls to arbitrary telephone. If there are 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 and browse the **Phone Settings** menu and select the **Call service** config menu. On the setting page, please disable All **Forward** function, and then the sample configuration screen is shown below:

## Call Service

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

Forward Type	Forward Number	Ring / Phone
Disable <input type="button" value="v"/>	<input type="text"/>	3 <input type="button" value="v"/> Phone 1

### STEP 2:

Please log in VIP-157\_A via web browser and browse the Phone Settings / General setting menu and select the Auto Answer config menu. On the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, then the sample configuration screen is shown below:

Auto Answer Type:	Both <input type="button" value="v"/>
Auto Answer Counter:	3 <input type="button" value="v"/>
PIN Code:	Enable <input type="button" value="v"/>
PIN Code Number:	••• <input type="text"/>

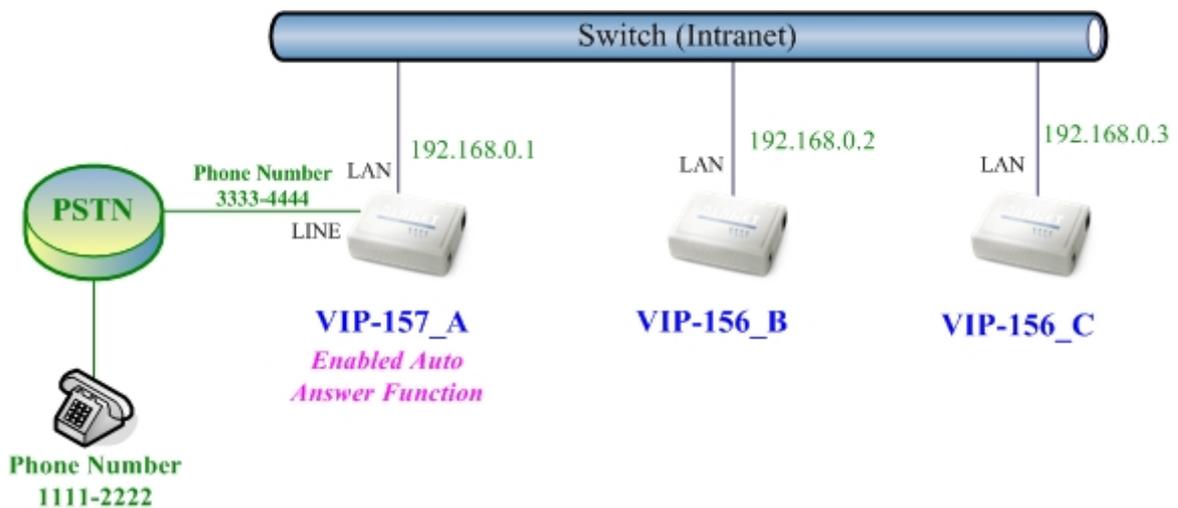
## Test the scenario:

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 and then input the password **123#**
5. VIP-156\_C will hear the dial tone from PSTN line and then input Phone Number 1112222

- The Phone Number 11112222 will ring up; then it picks up the telephone and communication with the VIP-156\_C

### Case 8: Auto Answer Feature\_PSTN to IP

For this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157\_A has set Auto Answer function for forwarding to arbitrary telephone. If there are 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 and browse the Phone Settings / General setting menu and select the Auto Answer config menu. On the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, and then the sample configuration screen is shown below:

<b>Auto Answer Type:</b>	Both <input type="button" value="v"/>
<b>Auto Answer Counter:</b>	3 <input type="button" value="v"/>
<b>PIN Code:</b>	Enable <input type="button" value="v"/>
<b>PIN Code Number:</b>	●●● <input type="text"/>

#### STEP 2:

Please log in VIP-157\_A via web browser and browse the **Phone Book** menu and select the **Speed Dial Settings** config menu. On the setting page, please add a speed dial number for dialing to IP address 192.168.0.2 (VIP-156\_B), and then the sample configuration screen is shown below:

## Phone Book Setting

You could add/delete items in current phone book.

Page: 1

Index	Name	Number/URL	Action
1	<input type="text" value="0"/>	<input type="text" value="192.168.0.2"/>	<input type="button" value="Delete"/>
2	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>

### Test the scenario:

1. The Phone Number 11112222 picks 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 and then input the password **123#**
5. The Phone Number 11112222 will hear the dial tone and then input **0#**
6. The IP address 192.168.0.2 (VIP-156\_B) will ring up and then it picks up the telephone and communication with the Phone Number 11112222

## Appendix B The method of operation guide

In this section, we'll introduce the steps of how to set up some call features of the ATA. Please follow the steps below to utilize those features.

### Call Transfer

#### A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carries out the transfer function (Press “**transfer**” button) to hold the conversation with B.
3. A presses “**#510#**” and hears the dial tone and then input the number of C (Followed by the “**#**” key).
4. C will ring up and A will get the busy tone for prompting to hang up
5. C picks up the handset and has conversation with B.

#### B. Attendant Transfer

1. B calls to A and they are in the process of conversation.
2. A carries out the transfer function to hold the conversation with B.
3. A presses “**#511#**” and hears the dial tone and then input the number of C (Followed by the “**#**” key).
4. C will ring up.
5. C picks up the handset and has conversation with A.
6. A hangs up and C has conversation with B.

### 3-Way Conference

1. A and B are in the process of conversation.
2. A wants to invite C to join their conversation.
3. A presses “**Transfer**” or “**Hold**” button to hold the conversation with B first and then press “**#512#**” and hear the dial tone, and then input the number of C (plus the “**#**” key).
4. C will ring up and pick up the handset to have conversation with A.
5. A presses “**Transfer**” button again, and they will enter the 3-way conference mode.

### Call Waiting

1. A and B are in the process of conversation.
2. C calls to A and A will hear the prompt sounds.
3. A presses “**Hold**” button to hold the conversation with B, and switch to have conversation with C.

### **Switch the Realm (Registration Proxy Server)**

ATA 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 anyone's SIP accounts for making calls through inputting the switch code.

#### **Realm switch code:**

**1\***: Realm 1

**2\***: Realm 2

**3\***: Realm 3

**4\***: Realm 4

**5\***: Realm 5

For example, the default is realm 1, input the **2\*** (Followed 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.

### **Auto Update firmware manually (Keypad)**

If you pick up the handset of ATA, you will hear the "DoDoDo" prompt. If want to carry out the upgrade action, please input "**#190#**" to unlock the device first. Then input "**#160#**" to upgrade the new firmware.

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

Product	SIP Analog Telephone Adapter			
Model	VIP-156	VIP-156PE	VIP-157	VIP-157S
<b>Hardware</b>				
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		---
<b>Protocols and Standard</b>				
Standard	SIP 2.0 (RFC3261)			
Voice codec	G.711a/u, G.723.1 (6.3k/5.3k), G.726, G.729A, G.729B, GSM			
Fax support	T.38			
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer			
Protocols	SIP 2.0 (RFC-3261), TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE			
<b>Network and Configuration</b>				
Access Mode	Static IP, PPPoE, DHCP			
Management	Web, keypad			
Dimensions (W x D x H)	94 x 72 x 30 mm			
Operating Environment	0~40 degrees C, 10~95% humidity			
Power Requirements	12V DC			
EMC/EMI	CE, FCC Class B			

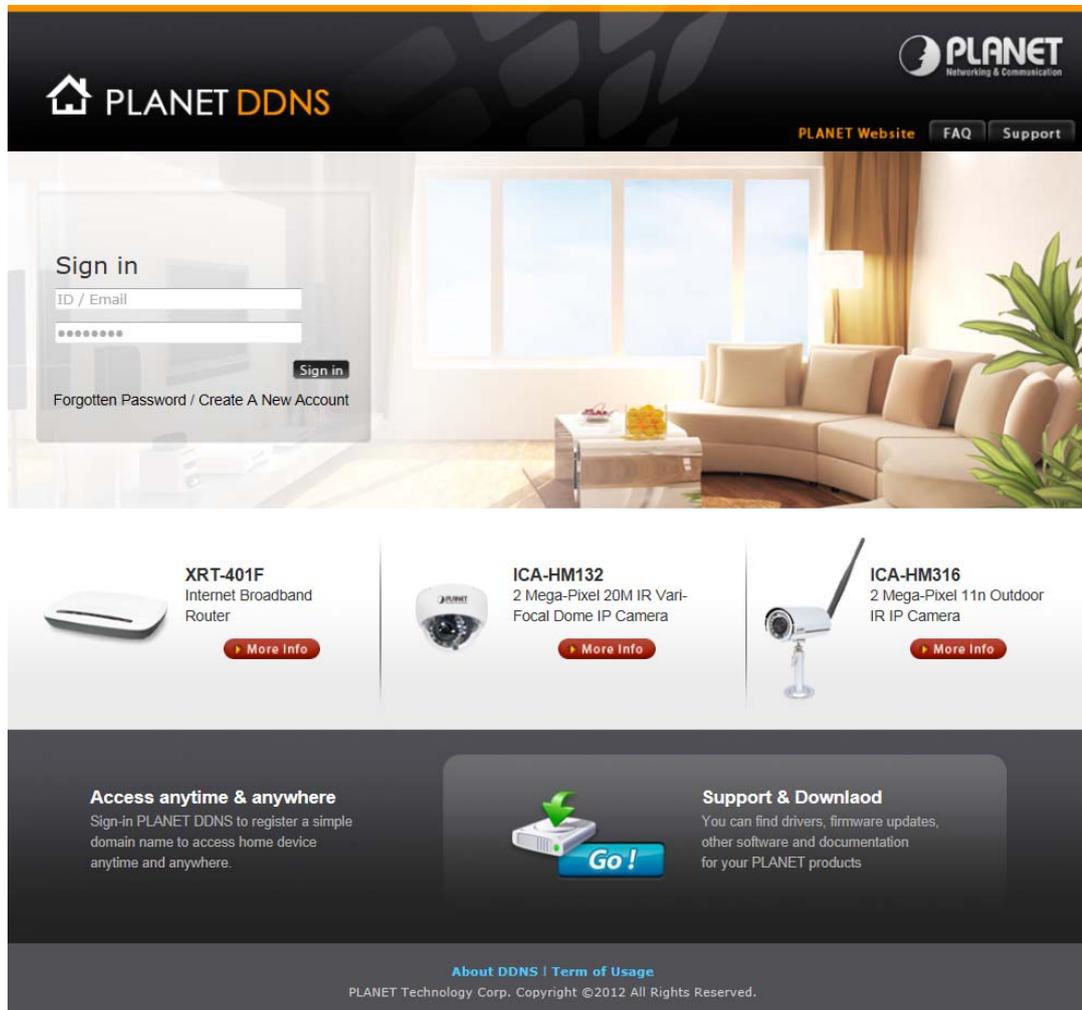
## Appendix D Planet DDNS Application

### Configuring PLANET DDNS steps:

**Step 1** Enable DDNS option through accessing web page of ATA device.

**Step 2** Select DDNS server provided, and register an account if you have not used yet.

Let's take dyndns.org as an example. Register an account at <http://planetddns.com>



## EC Declaration of Conformity

For the following equipment:

\*Type of Product : SIP Telephone Adapter  
\*Model Number : VIP-156

\* Produced by:

Manufacturer's Name : **Planet Technology Corp.**  
Manufacturer's Address: 11F, No 96, Min Chuan Road  
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.

For the evaluation regarding the R&TTE, the following standards were applied:

Emission		
Conducted / Radiated	EN 55022	(1998 + A1:2000 Class B)
Harmonic	EN 61000-3-2	(1995 Class A)
Flicker	EN 61000-3-3	(1995)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995)
RS	EN 61000-4-3	(1995)
EFT/ Burst	EN 61000-4-4	(1995)
Surge Test	EN 61000-4-5	(1995)
CS	EN 61000-4-6	(1996)
Magnetic Field	EN 61000-4-8	(1993)
Voltage Disp	EN 61000-4-11	(1994)
Safety	EN 60950 3rd	(2000)

Responsible for marking this declaration if the:

Manufacturer       Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

Company Address: 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C

Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Taiwan  
Place

7th July, 2005  
Date



Legal Signature

**PLANET TECHNOLOGY CORPORATION**

## EC Declaration of Conformity

For the following equipment:

\*Type of Product : PoE SIP Telephone Adapter  
 \*Model Number : VIP-156PE

\* Produced by:

Manufacturer's Name : **Planet Technology Corp.**  
 Manufacturer's Address: 11F, No 96, Min Chuan Road  
 Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.

For the evaluation regarding the R&TTE, the following standards were applied:

Emission		
Conducted / Radiated	EN 55022	(1998 + A1:2000 Class B)
Harmonic	EN 61000-3-2	(1995 Class A)
Flicker	EN 61000-3-3	(1995)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995)
RS	EN 61000-4-3	(1995)
EFT/ Burst	EN 61000-4-4	(1995)
Surge Test	EN 61000-4-5	(1995)
CS	EN 61000-4-6	(1996)
Magnetic Field	EN 61000-4-8	(1993)
Voltage Disp	EN 61000-4-11	(1994)
Safety	EN 60950 3rd	(2000)

Responsible for marking this declaration if the:

Manufacturer       Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

Company Address: 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C

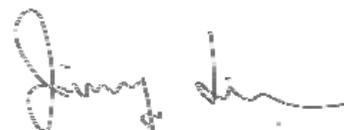
Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Taiwan  
Place

7th July, 2005  
Date

  
 \_\_\_\_\_  
 Legal Signature

### PLANET TECHNOLOGY CORPORATION

## EC Declaration of Conformity

For the following equipment:

\*Type of Product : VoIP Analog Telephone Adapter (1\*FXS + 1\*FXO)  
\*Model Number : VIP-157

\* Produced by:

Manufacturer's Name : **Planet Technology Corp.**  
Manufacturer's Address: 11F, No 96, Min Chuan Road  
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).

For the evaluation regarding the EMC, the following standards were applied:

Conducted / Radiated	EN 55022	(1998 + A1:2000 + A2:2003)
Harmonic	EN 61000-3-2	(2000)
Flicker	EN 61000-3-3	(1995 + A1:2001)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995 + A1:1998 + A2:2000)
RS	EN 61000-4-3	(2002 + A1:2002)
EFT/ Burst	EN 61000-4-4	(1995 + A1:2000 + A2:2001)
Surge Test	EN 61000-4-5	(1995 + A1:2000)
CS	EN 61000-4-6	(1996 + A1:2000)
Magnetic Field	EN 61000-4-8	(1993 + A1:2000)
Voltage Disp	EN 61000-4-11	(1994 + A1:2000)

**Responsible for marking this declaration if the:**

**Manufacturer**       **Authorized representative established within the EU**

**Authorized representative established within the EU (if applicable):**

**Company Name:** Planet Technology Corp.

**Company Address:** 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C

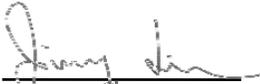
**Person responsible for making this declaration**

**Name, Surname** Jimmy Lin

**Position / Title :** Product Manager

Taiwan  
Place

16 March, 2006  
Date

  
Legal Signature

**PLANET TECHNOLOGY CORPORATION**

e-mail: sales@planet.com.tw      http://www.planet.com.tw

11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C. Tel:886-2-2219-9518 Fax:886-2-2219-9528

## EC Declaration of Conformity

For the following equipment:

\*Type of Product : VoIP Analog Telephone Adapter (2\*FXS)  
\*Model Number : VIP-157S

\* Produced by:

Manufacturer's Name : **Planet Technology Corp.**  
Manufacturer's Address: 11F, No 96, Min Chuan Road  
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).

For the evaluation regarding the EMC, the following standards were applied:

Conducted / Radiated	EN 55022	(1998 + A1:2000)
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Flicker	EN 61000-3-3	(1995 + A1:2001)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995 + A1:2001 + A2:2000)
RS	EN 61000-4-3	(2002 + A1:2002)
EFT/ Burst	EN 61000-4-4	(1995 + A1:2000 + A2:2001)
Surge Test	EN 61000-4-5	(1995 + A1:2000)
CS	EN 61000-4-6	(1996 + A1:2000)
Magnetic Field	EN 61000-4-8	(1993 + A1:2000)
Voltage Disp	EN 61000-4-11	(1994 + A1:2000)

**Responsible for marking this declaration if the:**

**Manufacturer**       **Authorized representative established within the EU**

**Authorized representative established within the EU (if applicable):**

**Company Name:** Planet Technology Corp.

**Company Address:** 11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C

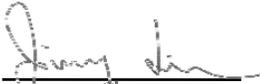
**Person responsible for making this declaration**

**Name, Surname** Jimmy Lin

**Position / Title :** Product Manager

Taiwan  
Place

17 March, 2006  
Date

  
Legal Signature

**PLANET TECHNOLOGY CORPORATION**

e-mail: sales@planet.com.tw      http://www.planet.com.tw

11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C. Tel:886-2-2219-9518 Fax:886-2-2219-9528