VoIP Analog Telephone Adapter

VIP-156

VIP-157

User's manual

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CE mark Warning

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

WEEE Warning

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

Trademarks

Revision

User's Manual for PLANET VoIP Analog Telephone Adapter:

Model: ATA

Rev: 1.0 (2006, January)

Part No. EM-VIP_ATAV1

Table of Contents

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

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

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

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

Product Features

- Feature-rich telephone service over home or office Internet/Intranet connection

- Auto-config feature for flexible, ease-of-use IP PBX system integration

- Cost effective, field proven compatibility, and stability

- Web-based and telephone keypad machine configuration

- Remote machine administration authentication

- Voice prompt for machine configurations

VoIP Features

- SIP 2.0 (RFC3261) compliant

- Peer-to-Peer / SIP proxy calls

- Voice codec support: G.711, G.723.1, G.729A/G.729B

- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation (CNG)

- In band, out-of-band, and SIP-info DTMF support

Package Content

The contents of your product should contain the following items:
VoIP Telephone Adapter
Power adapter
Quick Installation Guide
User's Manual CD
RJ-11 cable x 1

Physical Details

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

Respective model/descriptions are shown below:

VIP-156: SIP Analog Telephone Adapter

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

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

VIP-157S: 2-port FXS SIP Analog Telephone Adapter
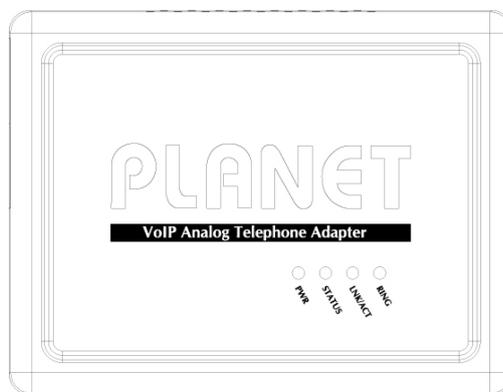
Front Panel of VIP-156
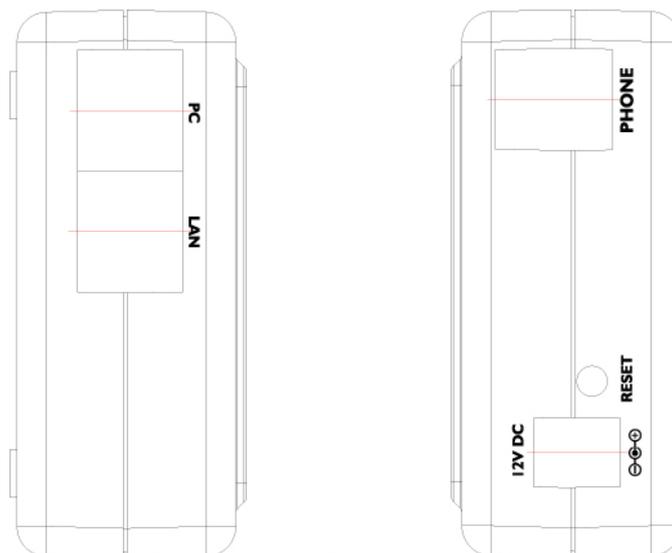
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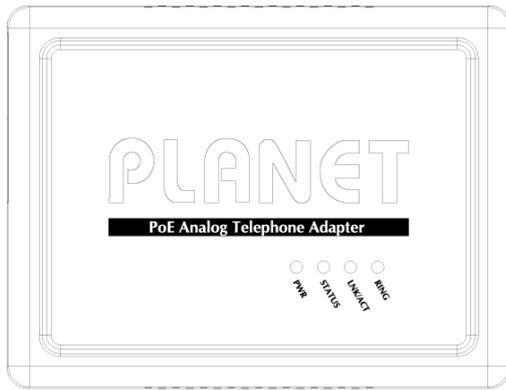
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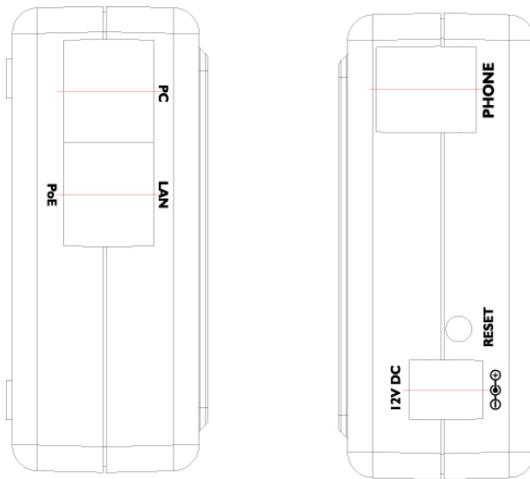
Left / Right Panel of VIP-156PE

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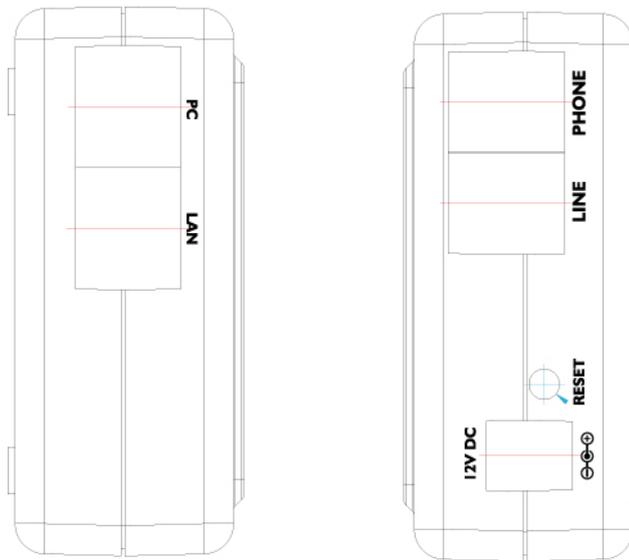
Left / Right Panel of VIP-157

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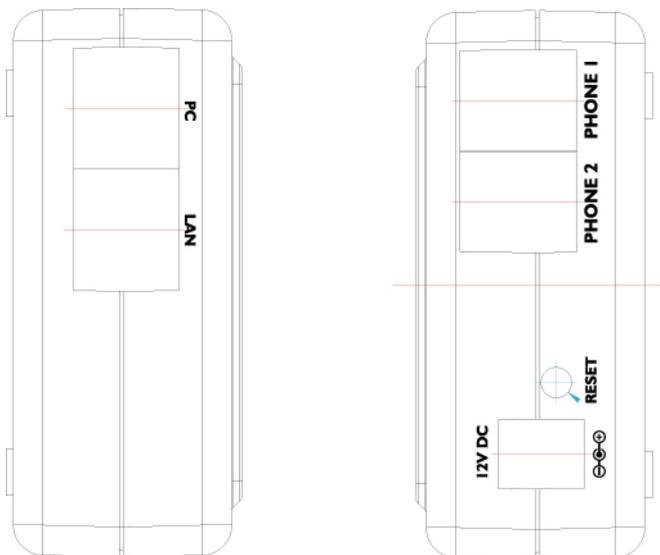
Left / Right Panel of VIP-157S

LED Display & Button

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

🖑 Note

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

LED display of VIP-156 / VIP-156PE

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

LED display of VIP-157 / VIP-157S

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

Chapter 2

Preparations & Installation

2

Physical Installation Requirement

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

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.

- TCP/IP protocol must be installed on all PCs.

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

Administration Interface

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

Web configuration access

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

- Microsoft Internet Explorer 4.01 or higher with Java support

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

Enter Network Password

Please type your user name and password
VoIP Phone Adapter Configuration

User Name [   ]

Password [   ]

[Login] [Clear]

☐ Save this password in your password list

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

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP in the adddress bar.

After logging on machine with username/password (default**: root / no passwrd**), browse to "**Network**"

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

Network Settings

You could configure your network settings in this page.

TCP/IP Configuration	
IP Type:	⊙Fixed IP ○DHCP Client ○PPPoE
IP:	192.168.0.1
Mask:	255.255.255.0
Gateway:	192.168.0.254
DNS Server 1:	168.95.1.1
DNS Server 2:	168.95.192.1
MAC:	00304faabbcc

Parameter Description

IP address   LAN IP address of ATA

                      Default: 192.168.0.1

Subnet Mask   LAN mask of ATA

                      Default: 255.255.255.0

Default Gateway   Gateway of ATA

                      Default: 192.168.0.254

Network settings via Keypad commands

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

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
#111#	Set DHCP client	None	ATA will change to DHCP Client
#112xxx*xxx*xxx* xxx#	Setup Static IP Address	Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
#113xxx*xxx*xxx* xxx#	Set Network Mask	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#114xxx*xxx*xxx* xxx#	Set Gateway IP Address	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#115xxx*xxx*xxx* xxx#	Set Primary DNS Server	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#136#	Enable auto-config mode	None	For PLANET IPX -2000 IP PBX System
#137#	Disable auto-config mode	None	For PLANET IPX -2000 IP PBX System
#195#	Reboot	None	The system will reboot automatically.
#198#	Factory Reset	None	The system will be reset to factory default value and reboot automatically.
＊	To switch IP Phone mode	None	VIP-157 only

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

IVR Menu Choice	Machine operation	Notes:
#120#	Check IP Address	IVR will announce the current IP address of the ATA.
#121#	Check network connection Type	IVR will announce if DHCP in enabled or disabled.
#122#	Check the Phone Number	IVR will announce current enabled VoIP number.

#123#	Check Network Mask	IVR will announce the current network mask of the ATA.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP address of the ATA.
#125#	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
#128#	Check Firmware Version	IVR will announce the version of the firmware running on the ATA.

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

After confirming the modification you've done, Please click on the **Submit** button to apply settings and browse to "**Save & Reboot**" menu to reboot the machine to make the settings effective.

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

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

Save Modification to Flash Memory

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

Save & Reboot

You have to save changes to effect them.

Save Changes: [Save]

Chapter 3

Network Service Configurations

<div style="text-align: right">3</div>

Configuring and monitoring your Phone Adapter from web browser

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

Overview on the web interface of ATA

With web graphical user interface, you may have:

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

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

- ◆ Microsoft Internet Explorer 4.01 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

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

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

ATA log in page

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

VoIP Phone Adatper main page

Chapter 4

VoIP Telephone Adapter Configurations

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Speed Dial settings

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

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

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

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

Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	URL	Select
0			☐
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

[Delete Selected] [Delete All] [Reset]

Add New Phone

Position: [＿＿＿] (0~9)

Name: [＿＿＿＿＿＿]

URL: [＿＿＿＿＿＿＿＿]

[Add Phone] [Reset]

Call forward

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

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

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

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

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

Forward Setting

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

All Forward: ☐   Busy Forward: ☐   No Answer Forward: ☐

	Name	URL
All Fwd No.:		
Busy Fwd No.:		
No Answer Fwd No.:		

Time Out:	20 (10~90 sec)

Submit   Reset

SNTP settings

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

SNTP Settings

You could set the SNTP servers in this page.

SNTP:	● On ○ Off

Primary Server:	192.43.244.18
Secondary Server:	208.184.49.9

Time Zone:	GMT + ▾ 08 ▾ : 00 ▾ (hh:mm)
Sync. Time:	1 : 0 : 0 (dd:hh:mm)

[Submit] [Reset]

[VIP-156] Volume Setting

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

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

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

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

Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	10	(0~12)
Ringer Volume:	10	(0~10)
Handset Gain:	8	(0~15)

[Submit] [Reset]

[VIP-157] Volume Setting

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

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

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

PSTN-Out Volume is to set the PSTN volume for you can hear. [VIP-157 only]

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

PSTN-In Gain is to set the volume send out to the other side's handset. [VIP-157 only]

Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	10	(0~12)
Ringer Volume:	10	(0~10)
PSTN-Out Volume:	10	(0~15)
Handset Gain:	9	(0~15)
PSTN-In Gain:	8	(0~15)

Submit   Reset

Volume Setting of VIP-157

Block Setting

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

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

Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

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

Block Setting

You could set the block period of your phone in this page.

Always Block:	○ On ⊙ Off
Block Period:	○ On ⊙ Off
From:	00 : 00 (hh:mm)
To:	00 : 00 (hh:mm)

Submit   Reset

[VIP-157 only] Auto Answer

This page set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the VIP-157 can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the VIP-157 can let user to redial to IP Phone number. Auto Answer Counter is to set

after the ring count meet the number you set then the auto answer will enable.

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	○ On ◉ Off
Auto Answer Counter:	03 (1~15)

[Submit] [Reset]

Caller ID settings

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

Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:	Don't show caller ID
Single Caller ID:	○ Yes ◉ No
CID Without Time:	○ Yes ◉ No

[Submit] [Reset]

Dial Interval Settings

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 Interval Settings

You could set the dial interval in this page.

Dial Interval:	5 (3~9 sec)

[Submit] [Reset]

Flash Time Setting

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

Flash Time Settings

You could set the flash time in this page.

Flash Time: 65 (Range:1~200, Unit:10ms)

Submit   Reset

Call waiting Settings

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

Call Waiting Settings

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

Call Waiting: ⊙ On   ○ Off

Submit   Reset

DDNS Settings

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

DDNS Settings

You could set the configuration of DDNS in this page.

DDNS: ○ On ◉ Off

Host Name:	
User Name:	
Password:	
E-mail Address:	

Type:	dyndns
Wild Card:	on

BACKMX:	On	Off
Off Line:	On	Off

[Submit] [Reset]

Port Settings

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

Port Settings

You could set the port number in this page.

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

[Submit] [Reset]

Service Domain Settings

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

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

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

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

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

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

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

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

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

Register Period: you need to input the Register Period get from your ISP. This is count in minute.

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

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

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

Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	○ On ⊙ Off
Display Name:	
Line Number:	1001
Register Name:	
Register Password:	
Domain Server:	
Proxy Server:	
Outbound Proxy:	
Register Period:	15 (0~99) [0: 30 sec,1~99 min]
Status:	Not Registered

Codec Settings

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

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.729
Codec Priority 2:	G.723
Codec Priority 3:	G.711 u-law
Codec Priority 4:	G.711 a-law
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	○ On ● Off

Voice VAD	
Voice VAD:	○ On ● Off

[Submit] [Reset]

RTP Setting

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

RTP Settings

You could set the RTP Settings in this page.

Outband DTMF: ● On ○ Off

Send DTMF SIP Info: ○ On ● Off

[Submit] [Reset]

DTMF Setting

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

DTMF Settings

You could set the DTMF Settings in this page.

○ 2833
○ Inband DTMF
○ Send DTMF SIP Info

[Submit] [Reset]

RPort Settings

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

RPort Settings

You could enable/disable the RPort setting in this page.

RPort: ○ On ● Off

[Submit] [Reset]

Other Settings

This page defines the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

Other Settings

You could set other settings in this page.

Hold by RFC:	○ On ● Off	
Voice QoS:	40	(0~63)
SIP QoS:	40	(0~63)
SIP Expire Time:	3600	(60~86400 sec)

[Submit] [Reset]

STUN settings

This page defines the STUN Enable/Disable and STUN Server IP address in this page. This function can help your Phone Adapter working properly behind NAT. To change these settings please following

your ISP information. When you finished the setting, please click the Submit button.

STUN Settings

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

STUN:	○ On ● Off

STUN Server:	
STUN Port:	5060 (1024~65535)

Submit   Reset

Auto Configuration

In Auto Configuration Setting you need to check with your service provider if they have provided this function. Usually this function will be boundle with an IP PBX to use in the office.

Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

Auto Configuration: ○ On ● Off

Submit   Reset

Country Settings

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

Country Setting

You could select your country setting in this page.

Select Country: Argentina ▾

Submit   Reset

ICMP Setting

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

ICMP Setting

You could enable/disable the ICMP setting in this page.

ICMP Not Echo: ○ On ⦿ Off

[Submit] [Reset]

System Authority

In System Authority you can change your login password.

System Authority

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

Username:	root
New password:	
Confirmed password:	

[Submit] [Reset]

Save & Reboot

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

Save & Reboot

You have to save changes to effect them.

Save Changes: [Save]

Firmware Upgrade

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

Select the firmware code type, AP or DSP code.

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

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

Firmware Upgrade

You could update the newest firmware.

Code Type: ⦿ AP   ◯ DSP

File Location: [   ] [Browse...]

[Update] [Reset]

Reset to Default

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

Reset to Default

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

Reset to default: [Restore]

Reboot without saving

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

Reboot without Saving

You could press the reboot button to restart the system.

Reboot without Saving: [Reboot]

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.

ATA to ATA connection via IP address

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

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

192.168.0.1   192.168.0.2
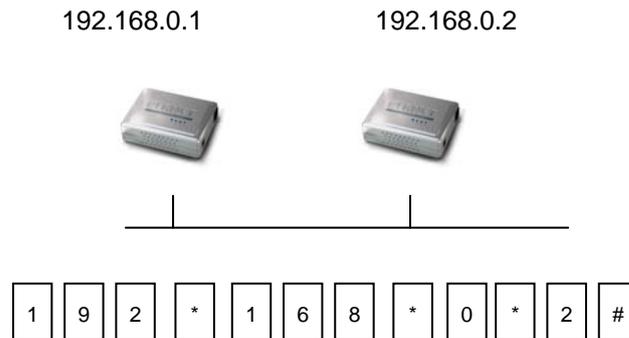
| 1 | 9 | 2 | * | 1 | 6 | 8 | * | 0 | * | 2 | # |

Operation steps:

1. Pick up the telephone set on ATA <u>A</u>.

2. [**VIP-157 only**, for VIP-156, please skip this step] Press the "*****" key to switch to IP phone mode, and you should be able to hear the dial-tone

3. Press the keypad: **192*168*0*2#** shall be able to connect to the ATA <u>B.</u>

4. Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on ATA <u>B</u> to establish the first voice communication from ATA <u>A</u>

(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

192.168.0.1

1001   1002
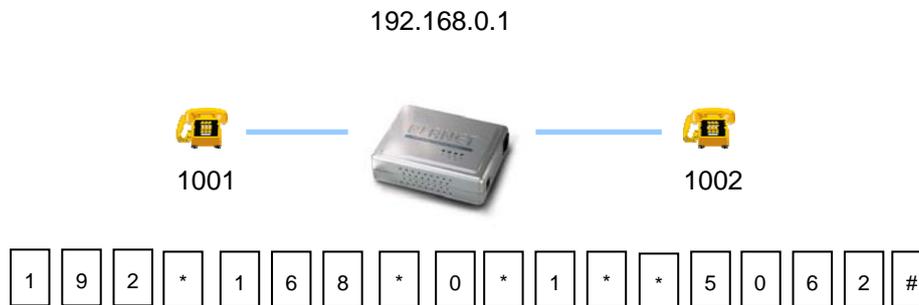
1	9	2	*	1	6	8	*	0	*	1	*	*	5	0	6	2	#

Operation steps:

a) Pick up the telephone set on VIP-157S port 1, and you should be able to hear the dial-tone

b) Press the keypad: **192*168*0*1**5062#** shall be able to connect to the VIP-157S port 2

c) 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

(i) Hint

- In default machine operation, the VIP-157 is at PSTN mode. If you want to make an IP call, press the "*****" key to switch to IP Phone mode. For VIP-156, no need to press the star "*****" key in advance.
- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with a "**#**".
- If the ATAs are installed behind a NAT/firewall/IP sharing device for Peer-to-Peer VoIP application, please make sure the NAT device support SIP applications, and suitable settings should be applied to the NAT device to enable the SIP communications before making calls
- [**VIP-157S**] in PLANET ATA series products, to connect to remote ATA, press the keypad in the following sequence to connect to the remote VIP-157S port 2: **[remote ATA IP address]**5062**, for example: **192*168*0*2**5062**

Voice communication via SIP proxy server – SIP-50

Registration /
Authentication

Registration /
Authentication

SIP-50

SIP-50 IP Address: 192.168.0.50
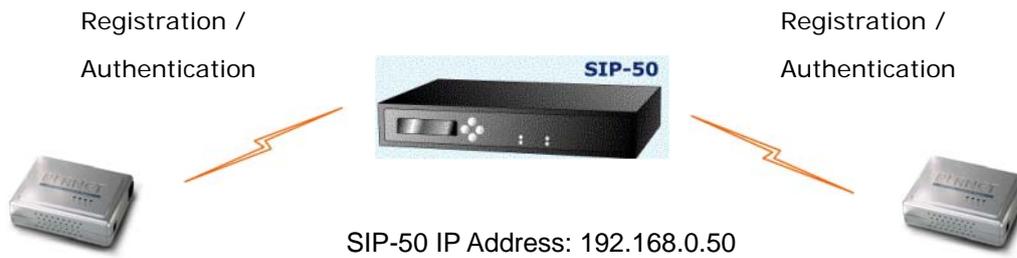
■   ATA <u>A</u> IP Address: 192.168.0.1
Line Number: 1001

■   ATA <u>B</u> IP Address: 192.168.0.2
Line Number: 2002

Device configurations on the ATA:

STEP 1:

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

STEP 2:

Please log in ATA via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:

Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	⊙ On ○ Off
Display Name:	1001
Line Number:	1001
Register Name:	planet
Register Password:	●●●
Domain Server:	192.168.0.50
Proxy Server:	192.168.0.50
Outbound Proxy:	
Register Period:	15 (0~99) [0: 30 sec,1~99 min]
Status:	Registered

STEP 3:

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

Test the scenario:

To verify the VoIP communication, please

Pick up the telephone

[**VIP-157 only**] Press the "*" key to switch to IP phone mode

Dial the destination number to make call between SIP clients. For example, ATA <u>A</u> (with number 1001) with keypad number 1002 to ATA <u>B</u>, or reversely make calls from SIP client (ATA <u>B</u>) to the number 1001 (ATA A)

Voice communication via IP PBX system – IPX-2000 (Auto-config)

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

Registration / Authentication

Registration / Authentication

IPX-2000 LAN IP Address: 192.168.0.50

■ ATA B IP Address: 192.168.0.2
Line Number: 2002

■ ATA A IP Address: 192.168.0.1
Line Number: 1001

Device configurations on the IPX-2000:

STEP 1:

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

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

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

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

STEP 2:

Please browse to the **Device** menu and create new device for the auto configuration.

STEP 3:

Please press the **Show extensions** button to create the two extension accounts/password: **1001/123** (for ATA A), and **1002/123**(for ATA B) for the voice calls.

STEP 4:

After setting up the parameters, please refer to the path to activate the settings: **Service --->** **IP PBX service** ---> **IP PBX configuration reload**

Device configurations on the ATA:

STEP 5:

Please log in ATA via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please browse to the Auto-config page, and enable the Auto Configuration features for IP PBX system. (Your may connect telephone set to ATA, press **#136** to enable the Auto configuration, or press **#137** to disable the Auto Configuration setting.)

Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

Auto Configuration: ⊙ On   ○ Off

[Submit] [Reset]

STEP 6:

After enabling the auto-config feature, the ATA shall be able to obtain IP address and SIP extension information from IP PBX system IPX-2000 information. To verify the auto-config results, you may connect telephone set to ATA; press **#120#** to check if the IP address is obtained from IPX-2000. And **#122#** can be used to verify the extension number assigned by IPX-2000.

STEP 7:

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

Test the scenario:

To verify the VoIP communication, please:

1. Pick up the telephone
2. [**VIP-157 only**] Press the "*" key to switch to IP phone mode
3. Dial the destination number to make calls. For example, ATA <u>A</u> (with number 1001) dial the keypad with number 1002 (ATA <u>B</u>) or reversely make calls from SIP client (ATA <u>B</u>) to the number 1001 (ATA <u>A</u>)

Appendix B VIP-156 / VIP-157 Specifications

Product	SIP Telephone Adapter			
Model	VIP-156	VIP-156PE	VIP-157	VIP-157S
Hardware				
LAN	1 x 10/100Mbps RJ-45 port (802.3af PoE for VIP-156PE)			
PC	1 x 10/100Mbps RJ-45 port			
FXS (for telephone set connection)	1 x RJ-11			2 x RJ-11
FXO (PSTN connection)	---		1 x RJ-11	---
Protocols and Standard				
Standard	SIP 2.0 (RFC3261)			
Voice codec	G.723.1 (6.3k/5.3k), G.729A, G.729B, G.711			
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			
Network and Configuration				
Access Mode	Static IP, PPPoE, DHCP			
Management	Web, keypad			
Dimension (W x D x H)	94 x 72 x 30 mm			
Operating Environment	0~40 degree C, 10~95% humidity			
Power Requirement	12V DC			
EMC/EMI	CE, FCC Class B			