Networking & Communication

User's Manual

VoIP Analog Telephone Adapter

► **VIP-156/VIP-156PE/VIP-157S**

www.PLANET.com.tw

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This 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.

FCC 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.

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 of PLANET VoIP Analog Telephone Adapter

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

Rev: 6.0 (October, 2017)

TABLE OF CONTENTS

CHAPTER 1. INTRODUCTION

1.1 Overview

PLANET VoIP Adapter

PLANET VIP-156/VIP-156PE/VIP-157S, an easy-to-use-and-install Analog Telephone Adapter, provides the same voice quality and reliability that users have come to expect from PLANET ATAs.

The VIP-156/VIP-156PE/VIP-157S is a 1-/2-port Analog Telephone Adapter which allows you to use your existing analog phones or fax machines to make calls using the internet. The VIP-156/VIP-156PE/VIP-157S supports **T.38 fax** and works with most SIP-based internet service providers (ISPs). The VIP-156/VIP-156PE/VIP-157S is a great choice for voice service providers because it is completely customized **auto-provisioning** and **auto-update**. If you want to have the world's leading VoIP service, you need to have the world's leading VoIP device!

Cost-effective, easy-to-install and simple-to-use, the VIP-156/VIP-156PE/VIP-157S converts standard telephones to IP-based networks. With the **802.3af/at PoE** integration (VIP-156PE only), the service providers and enterprises can offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

Professional Application

The VIP-156/VIP-156PE/VIP-157S includes two Ethernet interfaces for Internet (PPPoE, DHCP or Fixed IP), or office LAN connection. Besides the **IPv6**, **VPN** (PPTP and L2TP), **VLAN** and **PLANET Easy Dynamic**

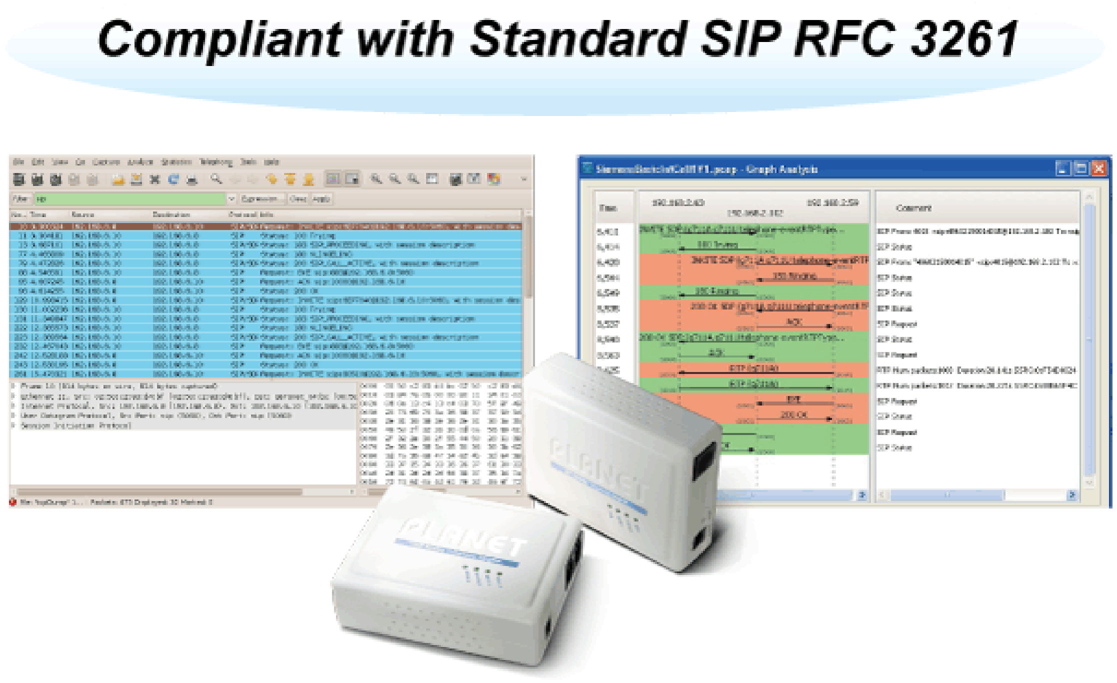
DNS features, the VIP-156/VIP-156PE/VIP-157S is an intelligent low-density Voice over IP (VoIP) gateway that enables carrier-class residential and business IP telephony services delivered over broadband or high-speed Internet connections to two standard RJ11 telephones. The 10/100 LAN port allows internet connectivity to be extended to a second device eliminating the need for a second Ethernet drop.

Calls from your VIP-156/VIP-156PE/VIP-157S to any other device on any VIP-156/VIP-156PE/VIP-157S device anywhere in the world are always free! The VIP-156/VIP-156PE/VIP-157S allows you to use your analog devices with any SIP trucking provider. Simply plug the Analog Telephone Adapter into your broadband router and then plug your analog devices into the ATA. The VIP-156/VIP-156PE/VIP-157S also can connect to your broadband router and to another IP device such as a PC. The 2-port RJ45 router has integrated **QoS** to ensure your voice traffic is prioritized above other types of traffic. It will support up to **5 VoIP services or SIP accounts**.

When used with a SIP service provider, you will enjoy value-added features such as message waiting indication (MWI), hotline, three-way conferencing, caller ID, call forward (always, busy and no answer), call waiting, Do Not Disturb (DND), dial plan and phone book. Using a VIP-156PE with an analog phone as an IP PBX telephone system gives you a special feature at a consumer price; the VIP-156/VIP-156PE/VIP-157S can be seamlessly integrated into the telephony network built by our IP PBX system series. The VIP-156/VIP-156PE/VIP-157S and PLANET IP PBX System (IPX-330/IPX-2100/IPX-2200/IPX-2500) are the ideal combination of your office daily communications.

Compliant with IETFSIP 2.0

SIP service continues to gain popularity among businesses as the preferred protocol for enhancing communication across IP networks. The VIP-156/VIP-156PE/VIP-157S supports **Session Initiation Protocol 2.0 (RFC 3261), STUN** and **Outbound Proxy** for easy integration with general voice over IP system. The VIP-156/VIP-156PE/VIP-157S is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better voice over IP services.

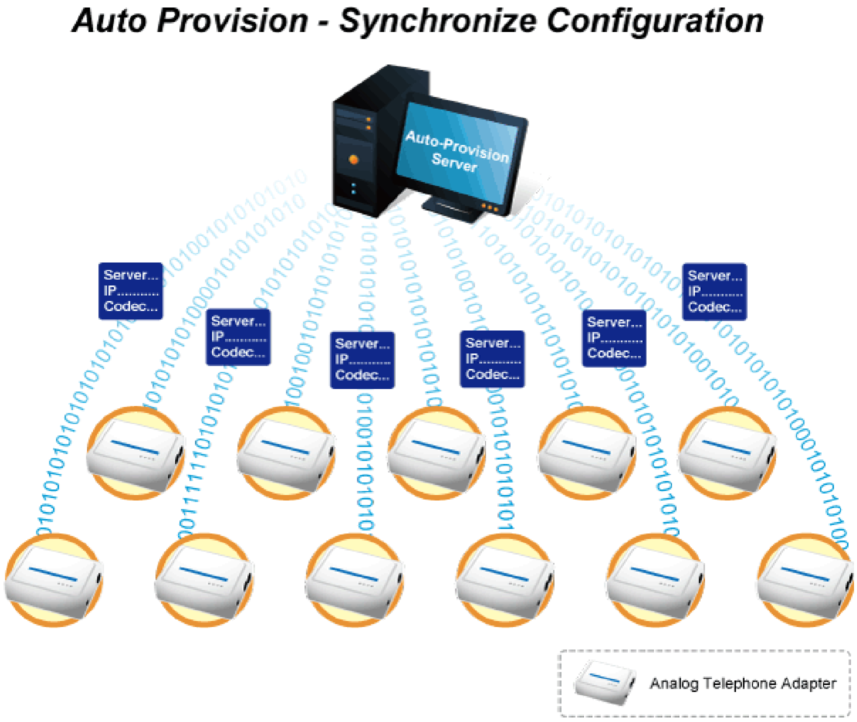


Convenient Design Brings Quality Communication

Based on years of VoIP manufacturing experiences, PLANET VoIP total solutions are known as advanced implementation of standard-based telephony along with mass deployment capability.

The VIP-156/VIP-156PE/VIP-157S is the lowest cost, fastest deployment ready ATA with auto-provision. The ATA can be deployed directly to customers, redirecting automatically to your provisioning server or you can use the **TR069** ITSP portal for target profiles.

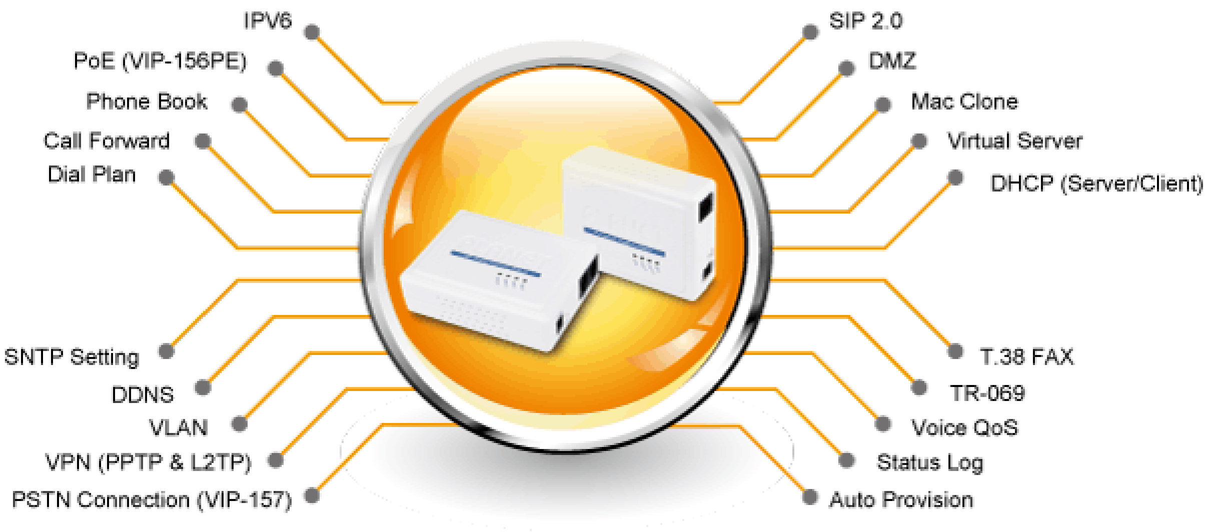
With the VIP-156/VIP-156PE/VIP-157S, enterprises are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The VIP-156/VIP-156PE/VIP-157S can be the bridge between the traditional analog systems and IP network with an extremely affordable investment.


Auto Provision - Synchronize Configuration
Auto-Provision Server
Server...
IP..........
Codec...
Analog Telephone Adapter

1.2 Application

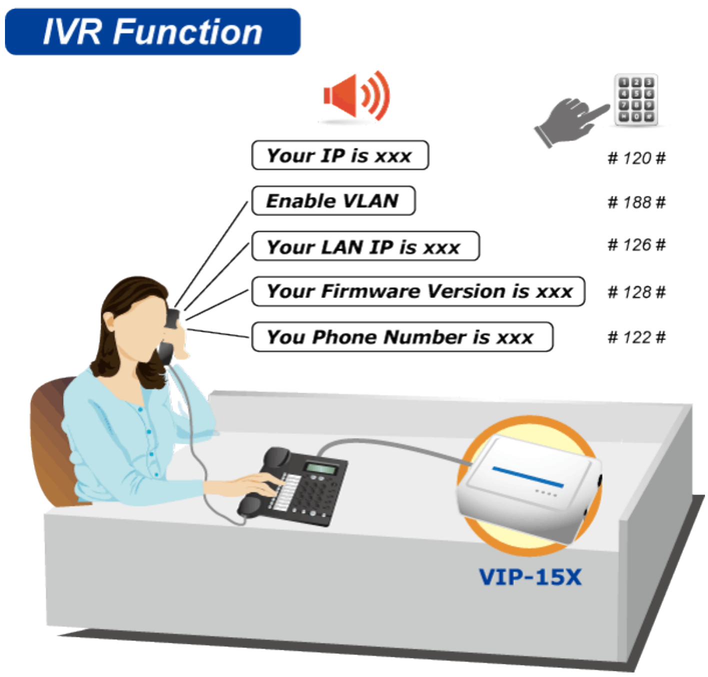
Enhanced, Full-Featured Analog Telephone Adapter

The VIP-156/VIP-156PE/VIP-157S is optimized for executive use for administrative assistants and those working with bandwidth-intensive application on collocated PCs. Four programmable extension keys could be configured as IP PBX features like, MWI, DND, Call Forward, Call Park and many others.



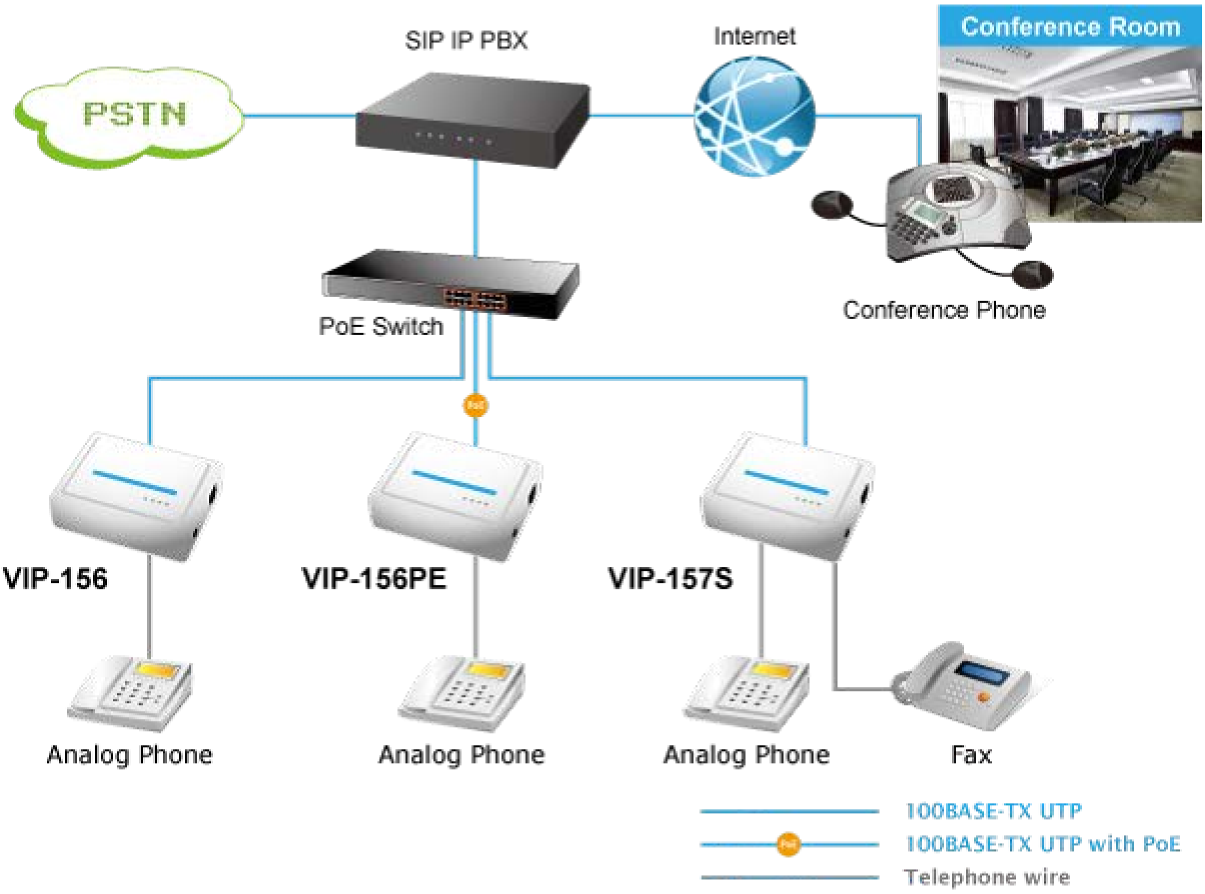
IVR Function to Easily Identify and Manage the ATA

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



Enterprise IP Telephony Deployment of VIP-156PE

The VIP-156PE comes with exceptional audio quality and user-friendly features. The installation and configuration of the IP phone are easier than those of the traditional phone system. Its low cost and high-definition voice quality give you value for money. Based on standard SIP 2.0, it is compatible with all the standard SIP-based servers.



1.3 Features

- **Product features**
  - Feature-rich telephone service over home or office Internet/Intranet connection
  - Auto-provisioning features for flexible, ease-of-use IP PBX system integration
  - IEEE 802.3af/at Power over Ethernet compliant (VIP-156PE)
  - Up to 2 concurrent VoIP calls (VIP-157S)
  - Voice prompt for machine configurations
  - Quick Setup Wizard for easy configuration
  - Easy access with PLANET Dynamic DNS
  - Cost effective, field proven compatibility and stability

- **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)

1.4 Package Contents

The contents of your product should contain the following items:

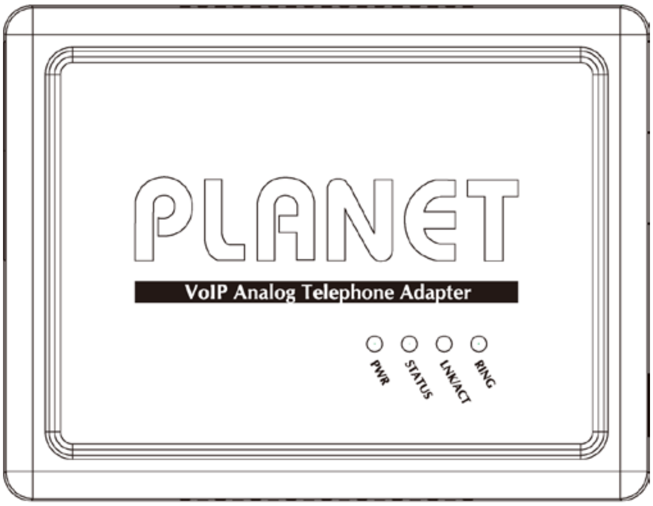
- VoIP Telephone Adapter
- Power Adapter
- Quick Installation Guide
- RJ11 Cable x 1

1.5 Physical Details

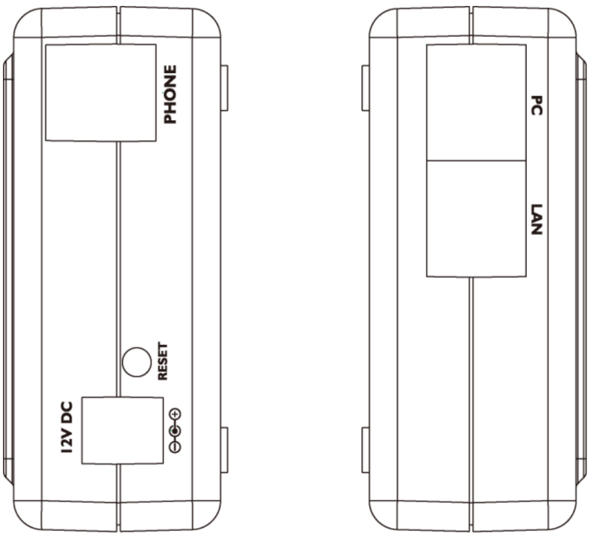
The following figure illustrates the front and rear panels of ATAs.

Respective models/descriptions are shown below:

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

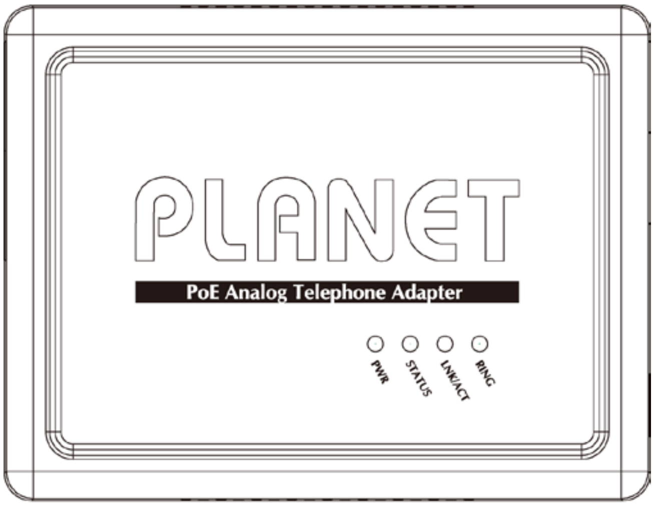


Front Panel of VIP-156

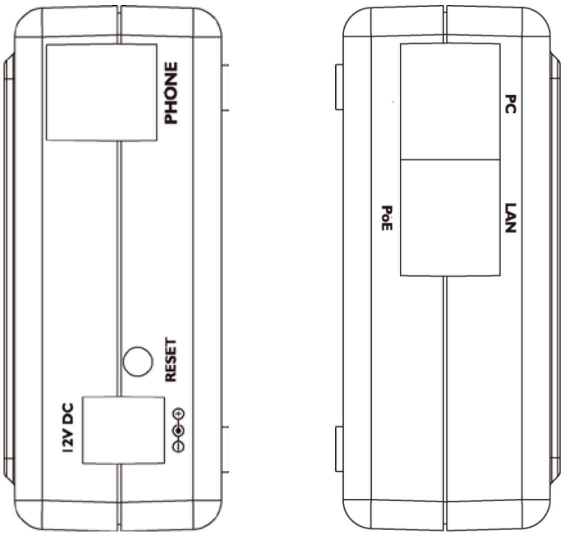


Left and Right Panels of VIP-156

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

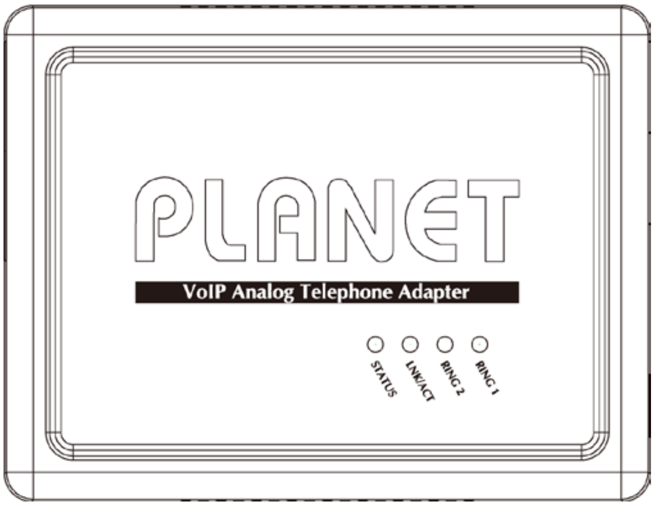


Front Panel of VIP-156PE

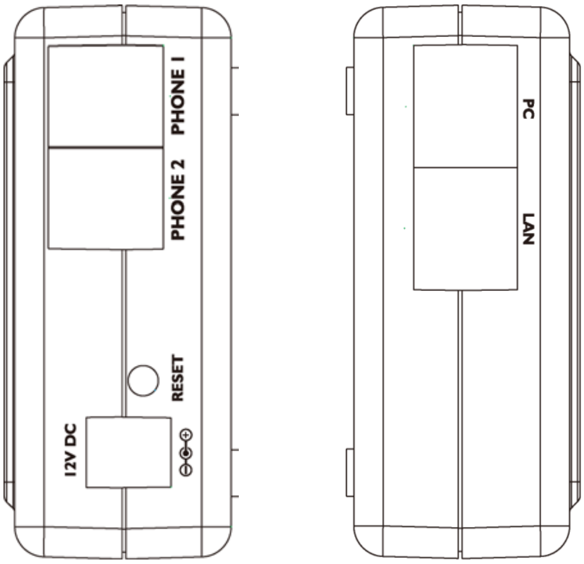


Left and Right Panels of VIP-156PE

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



Front Panel of VIP-157S



Left and Right Panels of VIP-157S

LED Display & Button

1	**PC**	RJ45 connector, to maintain the existing network structure, is connected directly to the **PC** through **straight** Cat5 cable.
2	**LAN**	RJ45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** Cat5 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 the **RESET** button on the rear panel for over 5 seconds to reset the VoIP Phone Adapter to factory default value. (Speed dial and call forward settings are exceptional.)

LED Display of VIP-156/VIP-156PE

LED Indicators	Descriptions
PWR	Power is supplied to the device.
STATUS	The Status LED will flash 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.

LED Indicators	Descriptions
RING	**OFF**: The phone is idle. **ON**: The phone is in use (off hook). **Blinking**: The phone is ringing.

LED Display of VIP-157/VIP-157S

LED Indicators	Descriptions
STATUS	The Status LED will flash 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 (off hook). **Blinking**: The phone is ringing.
RING 2	**OFF**: The phone is idle. **ON**: The phone is in use (off hook). **Blinking**: The phone is ringing.

CHAPTER 2. PREPARATIONS & INSTALLATION

2.1 Physical Installation Requirements

This chapter illustrates basic installation of ATA Analog Phone Adapter (“**ATA**”).

- Network cables. Use standard 10/100BASE-TX 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

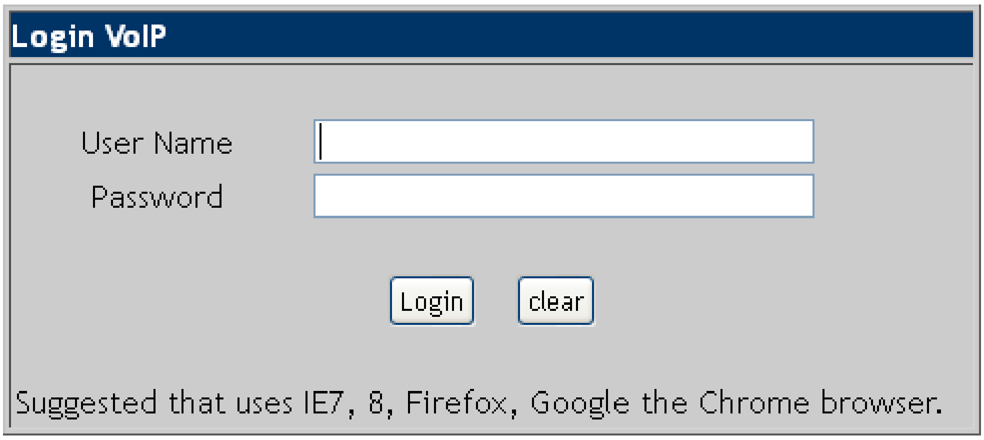
PLANET ATA provides GUI (Web-based graphical user interface) for machine management and administration.

Web configuration access

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

- Microsoft Internet Explorer 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 log on to the ATA web configuration page.

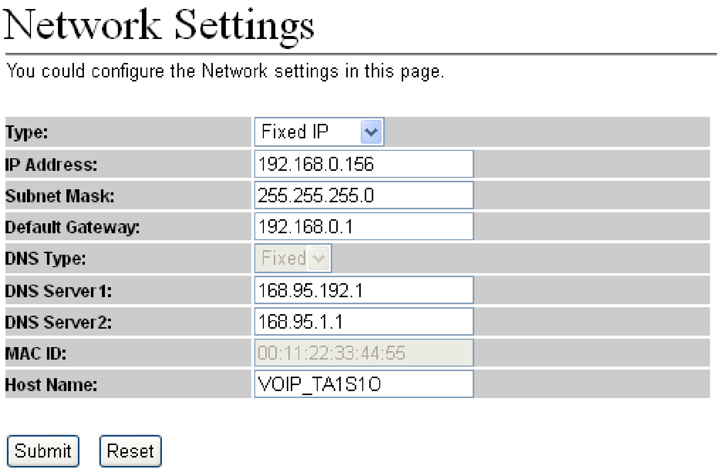


ATA will prompt for logon username/password. Please enter: **root** / **null (no password)** to continue machine administration.

Note	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 the related chapter on user’s manual CD or consult your network administrator for proper network configurations.

2.2 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 to the machine with username and password (default**: root** and **no password**), browse "**Network**" --> "**Network Settings**" configuration menu:



Parameter Description

IP Address	LAN IP address of ATA **Default**: 192.168.0.1
Subnet Mask	LAN mask of ATA **Default**: 255.255.255.0
Default Gateway	Gateway of ATA **Default**: 192.168.0.254

Network settings via Keypad commands

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

ⓘ Hint

When you want to run the setup or the start function, you 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.

IVR Menu Choice	Machine Operation	Parameter(s)	Notes:
#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 locked 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.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP address of the ATA.

IVR Menu Choice	Machine Operation	Notes:
#125#	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
#126#	Check LAN IP Address	IVR will announce the current LAN port IP address of the ATA.
#128#	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:
#138#	Enable call waiting	None	Enable call waiting
#139#	Disable call waiting	None	Disable call waiting
#160#	Update firmware	None	Update firmware
#510#	Blind Transfer	ATA Blind Transfer	
#511#	Attendant Transfer	ATA Attendant Transfer	
#512#	3-way calling	ATA 3-way calling	
#514#	IP transfer to PSTN	ATA transfer IP call to PSTN side	
#130+[1~8]#	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 be the first priority.
#131+[00~15]#	Set Handset Gain	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
#132+[00~12]#	Set Handset Volume	Handset Volume from 0~12	You can set the handset volume to proper value; default is 10
#135xxx*xxx*xxx*xxx#	TFTP Server IP Address	Set Auto config TFTP Server IP Address	You can set the TFTP Server IP address
#136xxx*xxx*xxx*xxx#	FTP Server IP Address	Set Auto config FTP Server IP Address	You can set the FTP Server IP address
#137+[0~2]#	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

IVR Menu Choice	Machine Operation	Parameter(s)	Notes:
#145#	Forward function disable	Disable forward function	
#146+Number#	Enable forward to FXS Port	Eanble forward to FXS Port	
#147+Number#	Enable forward to FXO Port	Eanble 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

ⓘ 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,

ⓘ 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.

CHAPTER 3. NETWORK SERVICE CONFIGURATIONS

3.1 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.

3.1.1 Overview of the web interface of ATA

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Input data fields, check boxes, and changing machine configuration settings
- Machine running configuration

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

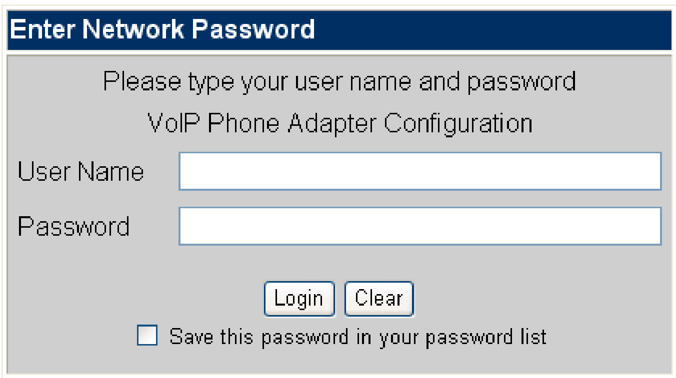
- Microsoft Internet Explorer 6.0.0 or higher with Java support

3.1.2 Manipulation of ATA via web browser

Log on to ATA via web browser

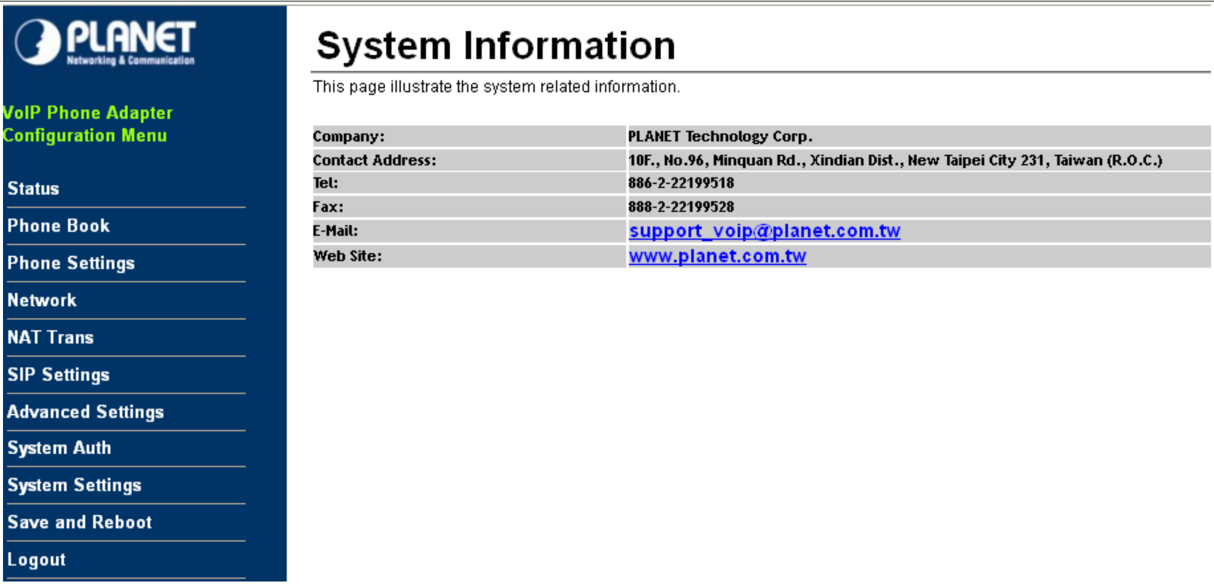
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to log on to the Phone Adapter web configuration page.

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



ATA login page

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



VoIP Phone Adapter main page

CHAPTER 4. VOIP TELEPHONE ADAPTER CONFIGURATIONS

4.1 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

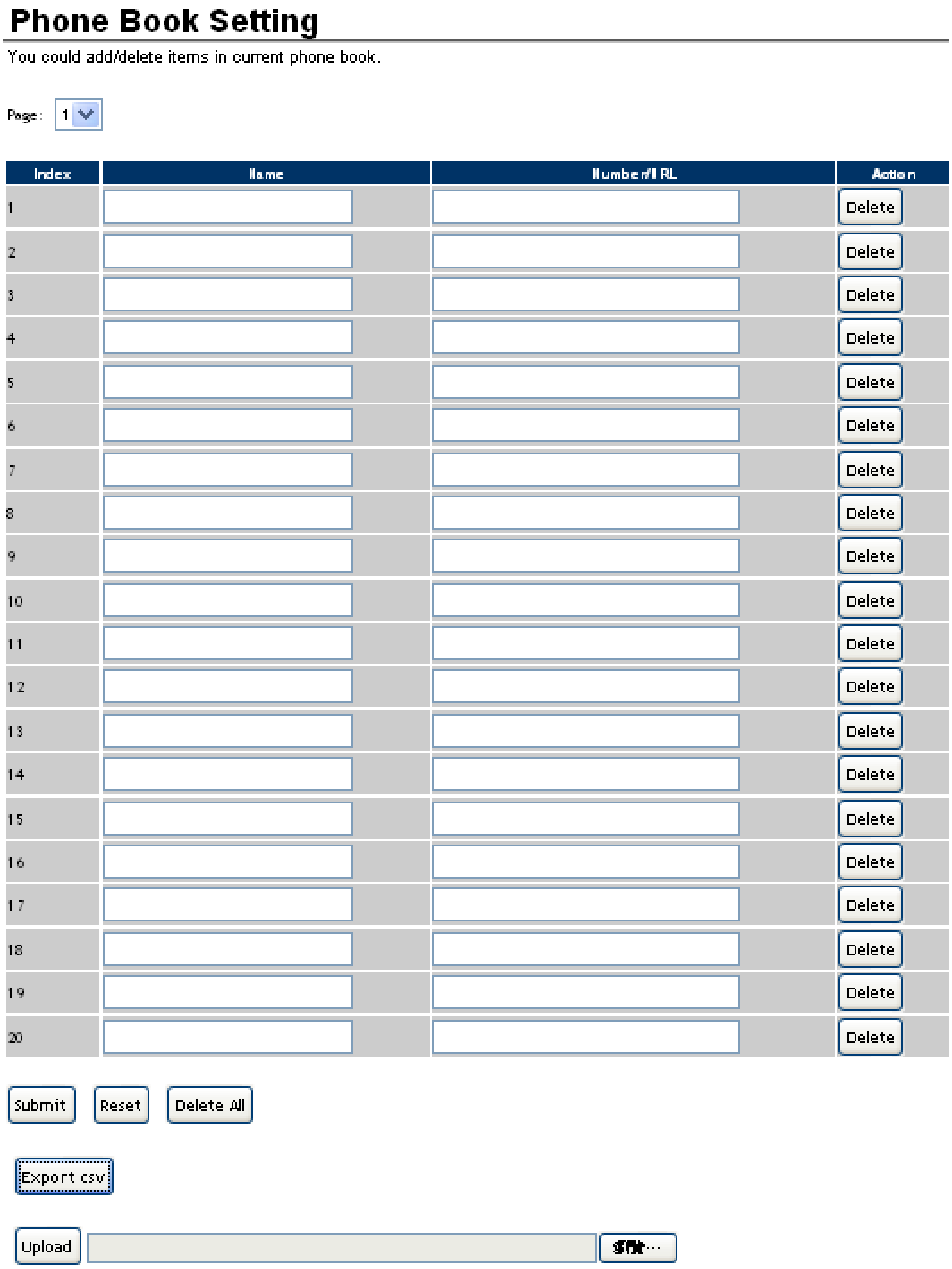
System Information			
Firmware Version:	1.1.0-1201180	Update Date:	2005-01-01
DSP Version	NV-1106080		
System Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Current Time:	2012-02-20 15:54		

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:	

4.2 Phone Book

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



Field	Description
Phone Book Page	The default is Page 1. You can select Page 1 ~ Page 7 to look through Phone Book records.
Phone	The record number is from 1 ~ 140; it can set up 140 records in total.

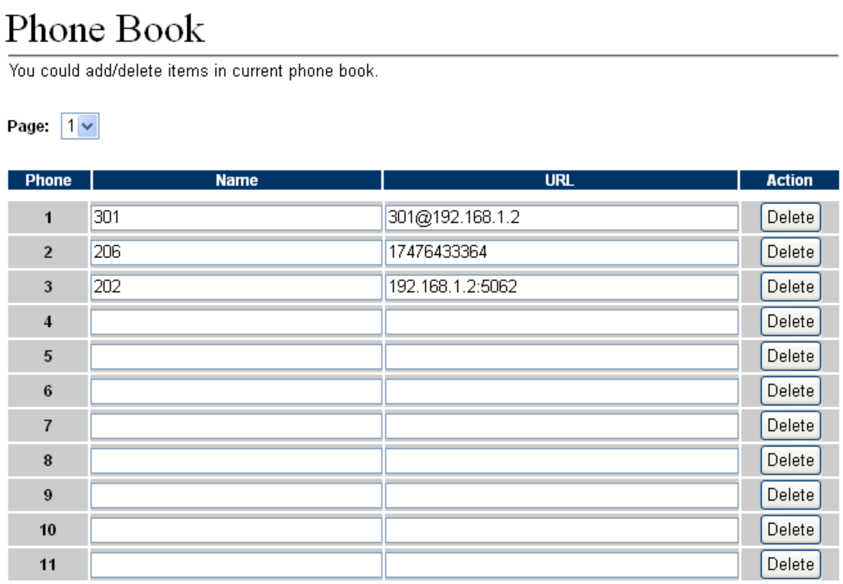
Field	Description
Name	The name of Phone Book records; it only can input numerals character.
URL	Fill in the outgoing number (Line Number) or IP address.
Delete	Clean this item's data.
Export csv	Save the phone book data as CSV file.
Upload	Upload the phone book file

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 the "Delete" button.

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

For Example:



Example_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).

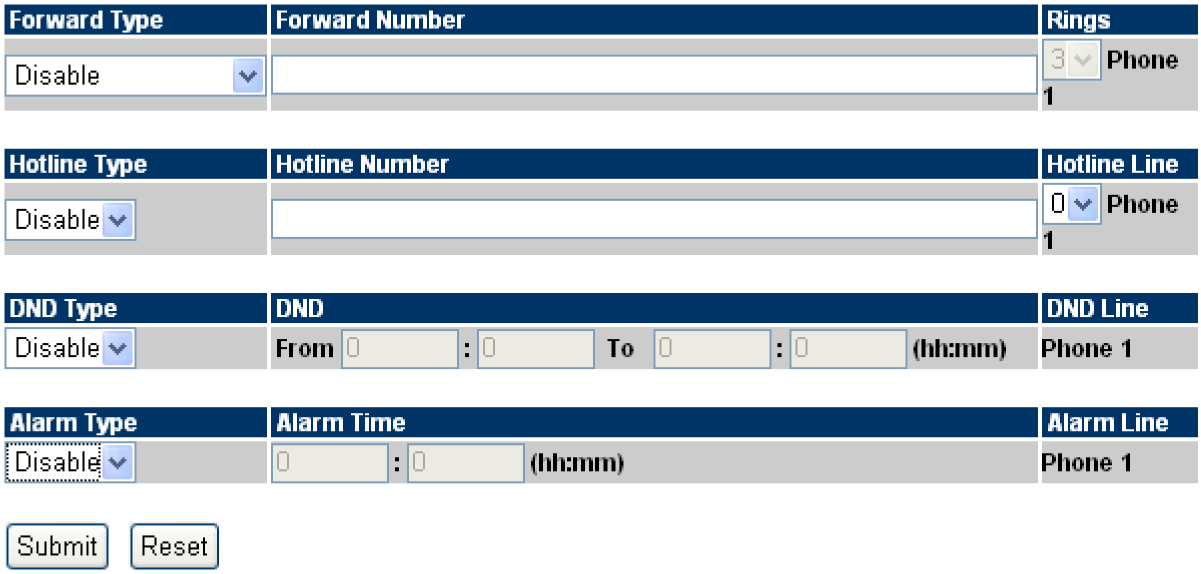
Example_2:

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

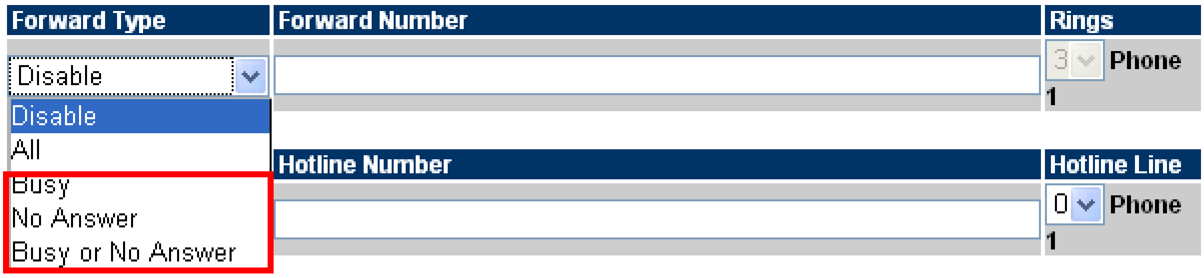
Example_3:

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

4.3 Call Service



[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.

[Hotline Type]

Hotline Type	Hotline Number	Hotline Line
Disable		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
Disable (Disable / Always / Period)	From 0 : 0 To 0 : 0 (hh:mm)	Phone 1
	Alarm Time	**Alarm Line**
	0 : 0 (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 this feature is disabled.

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	0 : 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 rung every day.

Alarm Type:

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

Alarm Time:

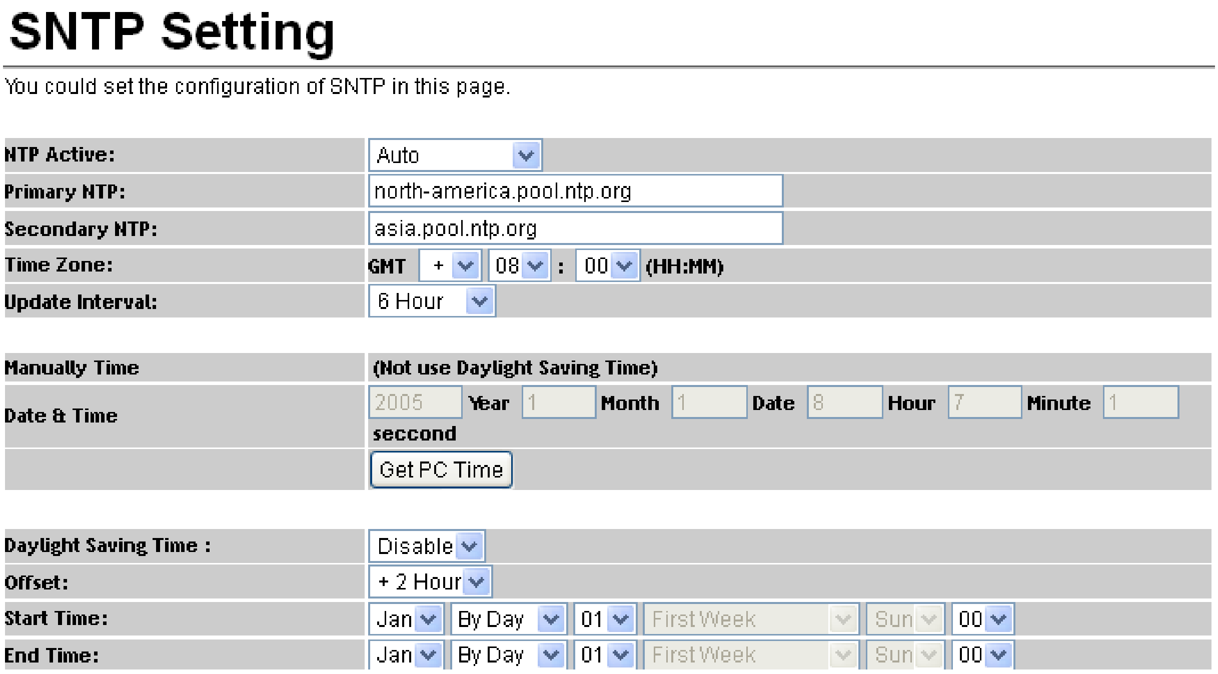
It can set up the system prompt time within 24 hours.

Alarm Line:

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

4.4 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.

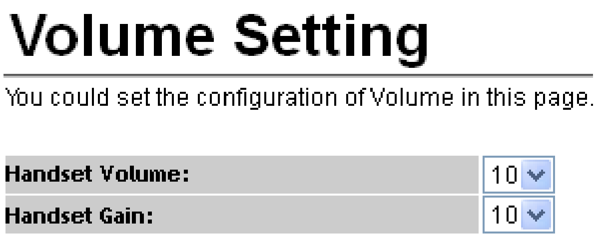


4.5 Volume Setting

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.

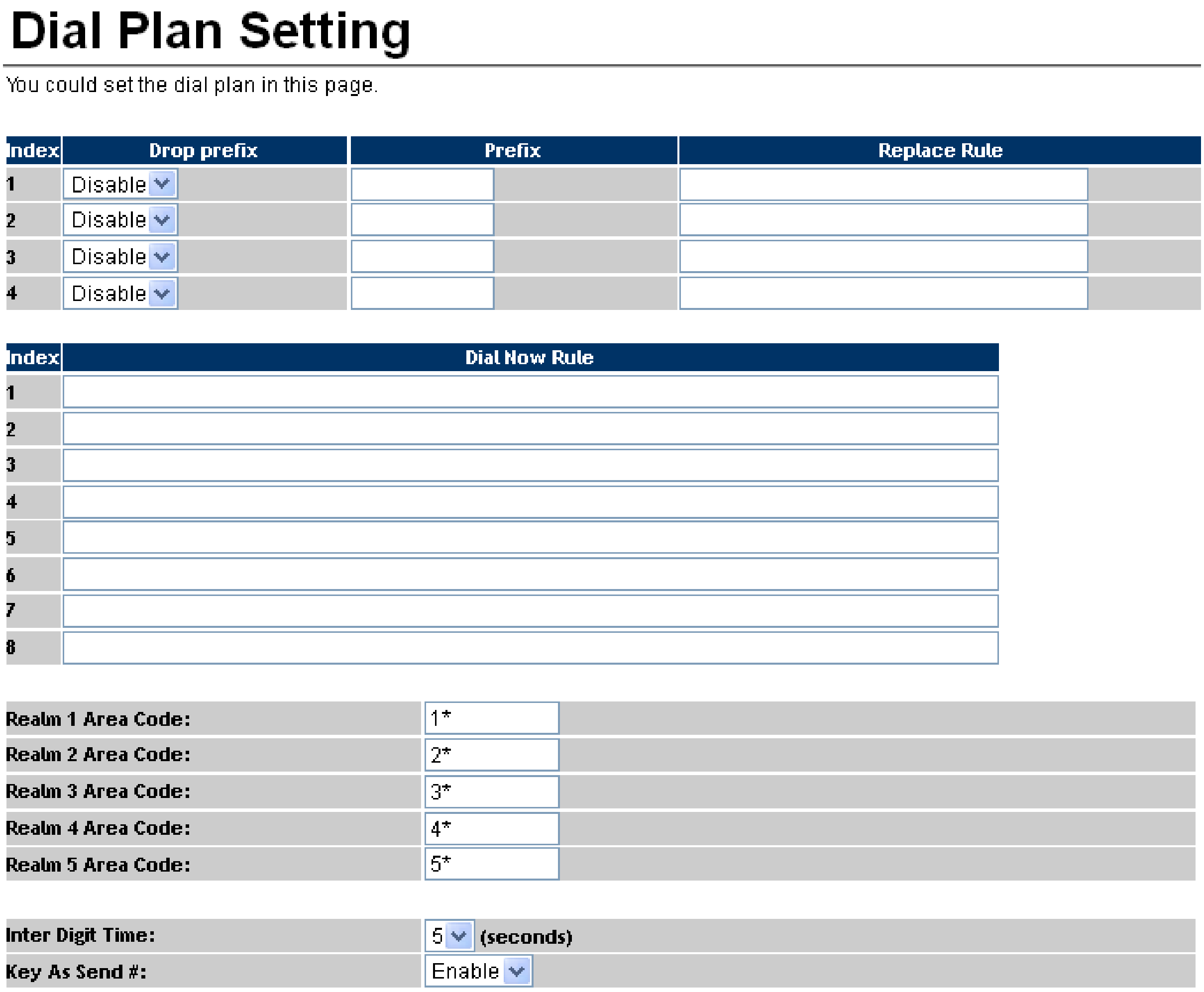
Handset Gain is to set the volume sent out to the other side’s handset.



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

4.6 Dial Plan Setting

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

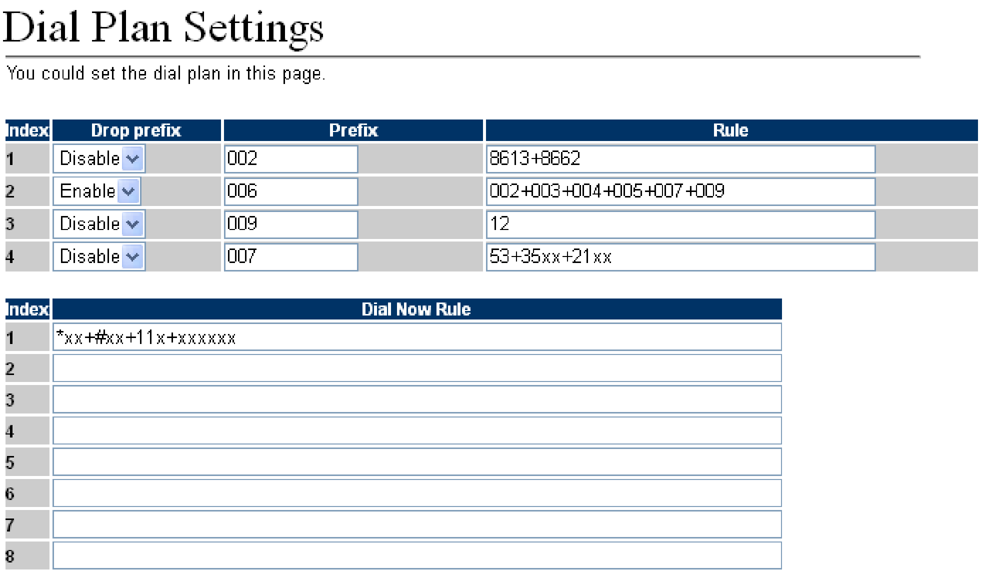


Dial Plan Settings for VIP-156

Field	**Description**
Drop Prefix	The rule of add or replace code -- If Disable is set, it will add the prefix number prior to the identification number. If Enable is set, it will replace the identification number.
Prefix	The prefix number -- It only accepts the numeral and the max. length is 8.
Rule Rule	The identification number -- It can accept the numeral or symbol and the max. length is 40. - Symbol: It only accepts the [+], [x] - +: It means “or”. For example, [123+456+334+5xx] even if [123 or 456 or 334 or 5xx] - x: It is equal to 0~9. For example, [5xx] even if the number begins with 5.

Field	Description
Dial Now Rule	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. Note: The starting number can't be "0". For example, if the number is "0xxxx", the system will ignore this dial plan.
Realm 1/2/3/4/5 Area Code	These options can define the switching code for each Realm No.
Inter Digit Time (Auto Dial Time)	Dialed number is sent out after dialing is stopped.
Key as send #	If Yes is set, the system will stop receiving 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 No is set, the system will do according to the Auto Dial Time to determine the end time.

Descriptions of example:



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+8613xxxxx**].
2. If the dialing number is "003+77**xxxx**", 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**

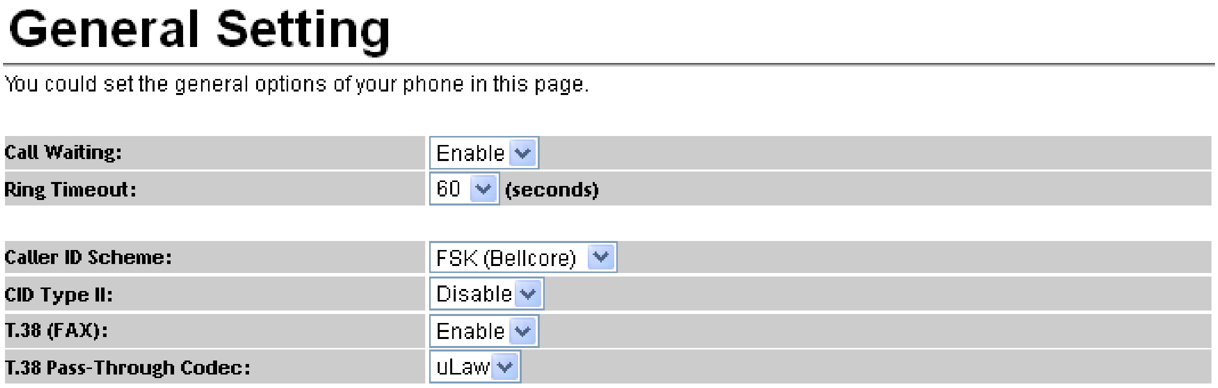
1. If the dialing number prefix is [53789], it will match the rule [53], then system will automatically add the prefix [007] in front of [53789].The real dialing number is [**007+53789**].
2. If the dialing number prefix is [3507], it will match the rule [35xx], then system will automatically add the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

4.7 General

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



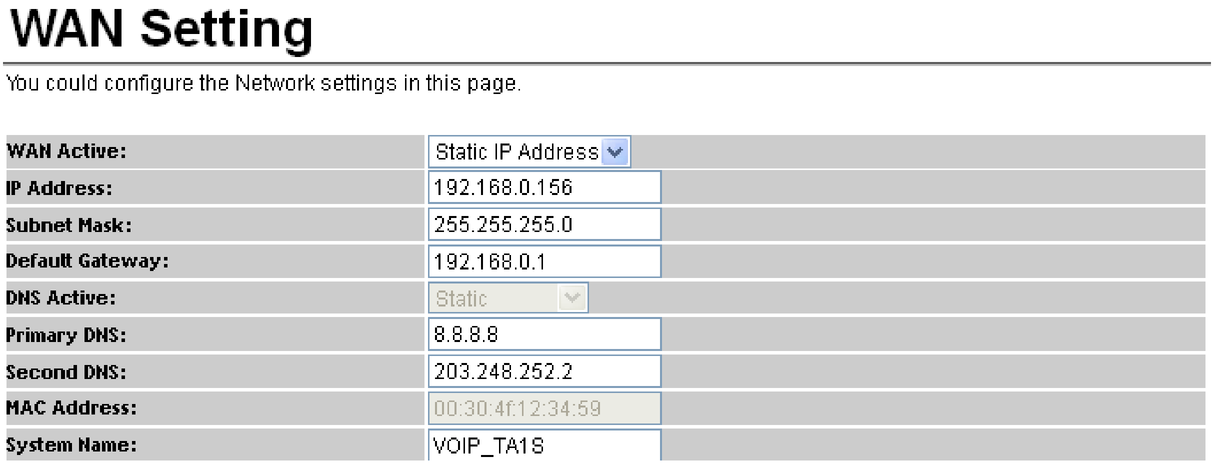
VIP-156

Field	Description
Call Waiting	Default is enable. When you are talking with other people, you can choose if you want to hear the message 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 message in your current call. If you set the function to Off, then you will not hear any message.
Ring Timeout	Default is 60 (sec). After how long the system will reply the busy (486 busy) message.
Caller ID Scheme	Set the caller ID mode, it supports FSK Bellcore, DTFM, CID-Japan, DTMF-Brazil, DTMF-Denmark. FSK Bellcore: FSK caller ID mode. DTMF: Before the first ring, 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
CID Type II	To enable the show caller ID function in call waiting. When enabling this function, system will receive a new call in call waiting. It will display the caller ID Note: Your phone must also support CID Type 2.
T.38 (Fax)	Enable/Disable T.38 Fax function.
T.38 Pass-through Codec	Define the T.38 pass-through codec; it can support G.711 u-law/G.711 a-law.

CHAPTER 5. NETWORK

5.1 Network Setting

This page defines the LAN setting on this page.

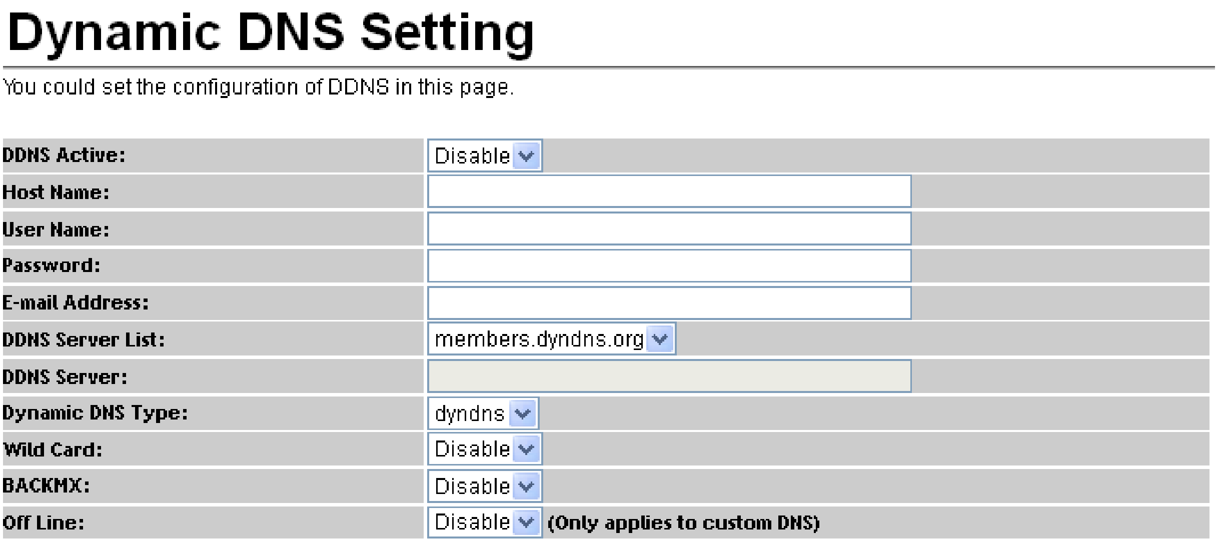


Field	Description
WAN Active	The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes. **Fixed IP:** It could set up the IP address manually. **DHCP Client:** It will acquire the IP address automatically. **PPPoE:** It will use the PPPoE connection method
IP Address	The IP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

5.2 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 its web site www.planetddns.com)



Dynamic DNS Setting

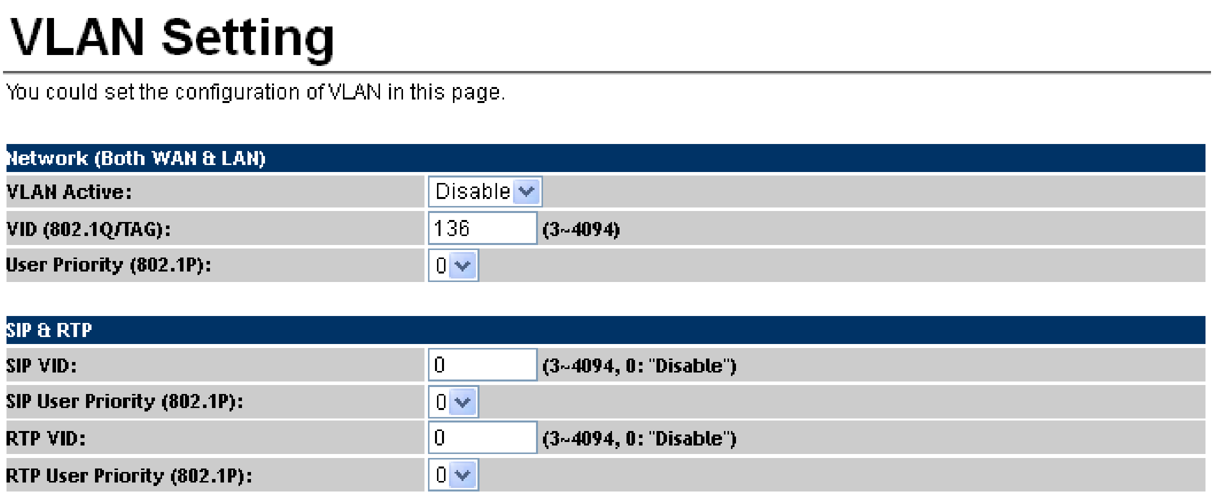
You could set the configuration of DDNS in this page.

DDNS Active:	Disable
Host Name:	
User Name:	
Password:	
E-mail Address:	
DDNS Server List:	members.dyndns.org
DDNS Server:	
Dynamic DNS Type:	dyndns
Wild Card:	Disable
BACKMX:	Disable
Off Line:	Disable (Only applies to custom DNS)

DDNS Settings for VIP-156/VIP-156PE/VIP-157S

5.3 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

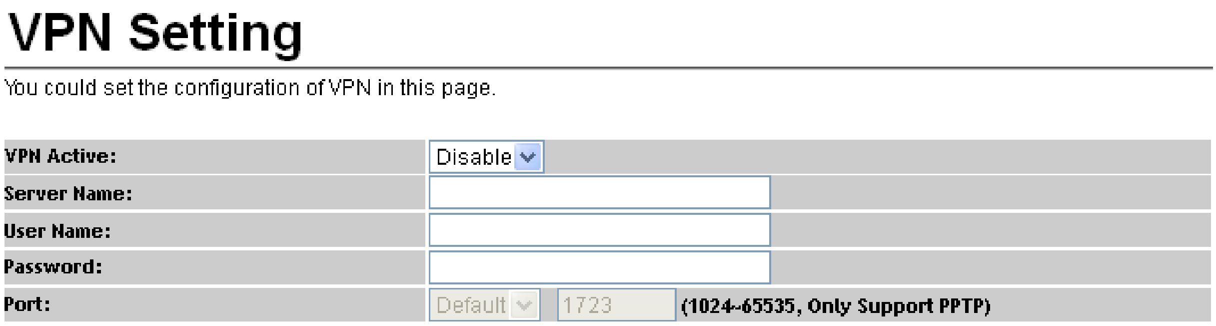
SIP & RTP	
SIP VID:	0 (3~4094, 0: "Disable")
SIP User Priority (802.1P):	0
RTP VID:	0 (3~4094, 0: "Disable")
RTP User Priority (802.1P):	0

Field	Description
VLAN Activity	If On is set, it could receive VLAN messages.
VID (802.1Q/TAG)	Dispose VLAN ID is added as a Tag header after enabling the VLAN function. The realized voice packets transfer is similar to 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.

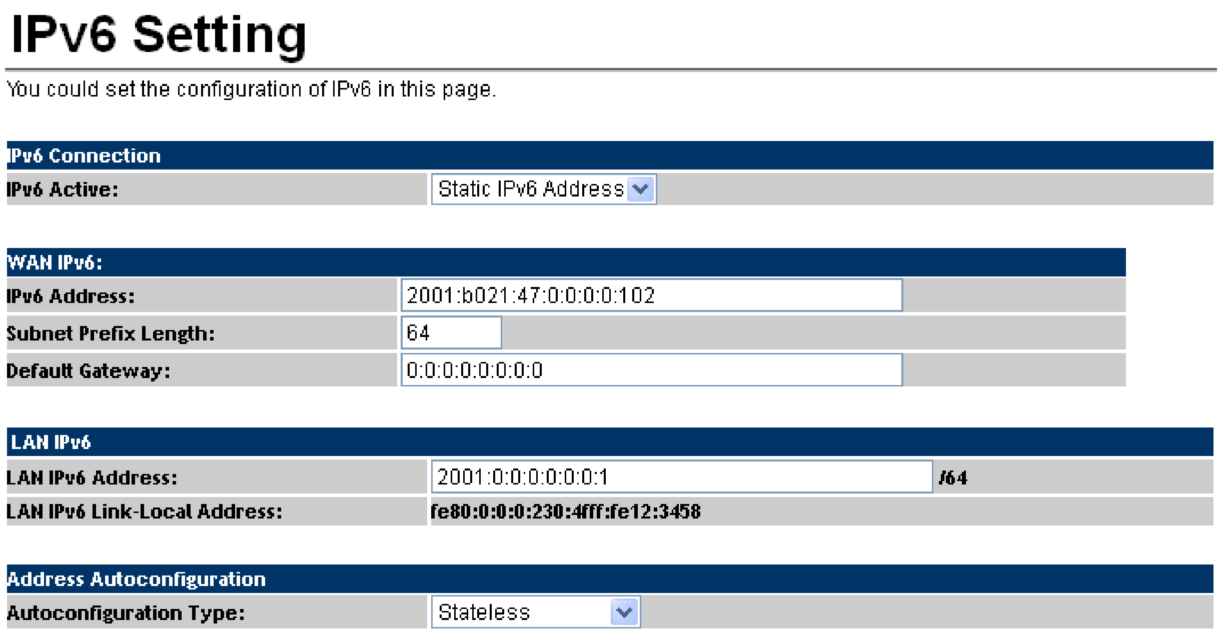
5.4 VPN Setting

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.



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

5.5 IPv6 Setting



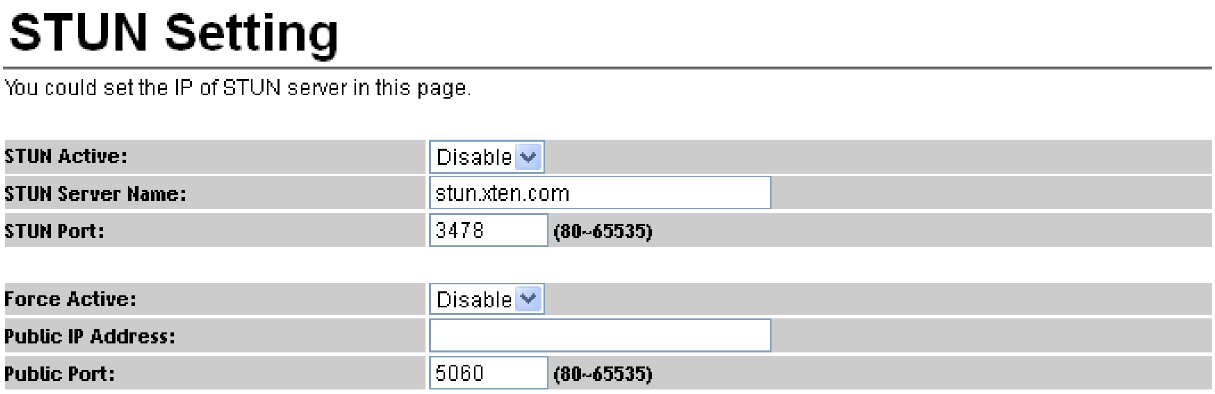
This page defines the IPv6 setting on this page. You can program the IPv6 information.

Field	Description
IPv6 Activity	Support three IPv6 types: Auto, Fixed IPv6, IPv6 in IPv4 Tunnel
IPv6 address	Setting the WAN IPv6 address or display it (64 bits)
SubnetPrefix Length	Default is 64
Default Gateway	IPv6 gateway address (64 bits)
LAN IPv6 Address:	IPv6 LAN address (64 bits)
LAN IPv6 Link-Local Address	Link local address information.
Autoconfiguration Type	It supports stateless or stateful (DHCP V6).

CHAPTER 6. NAT TRANS

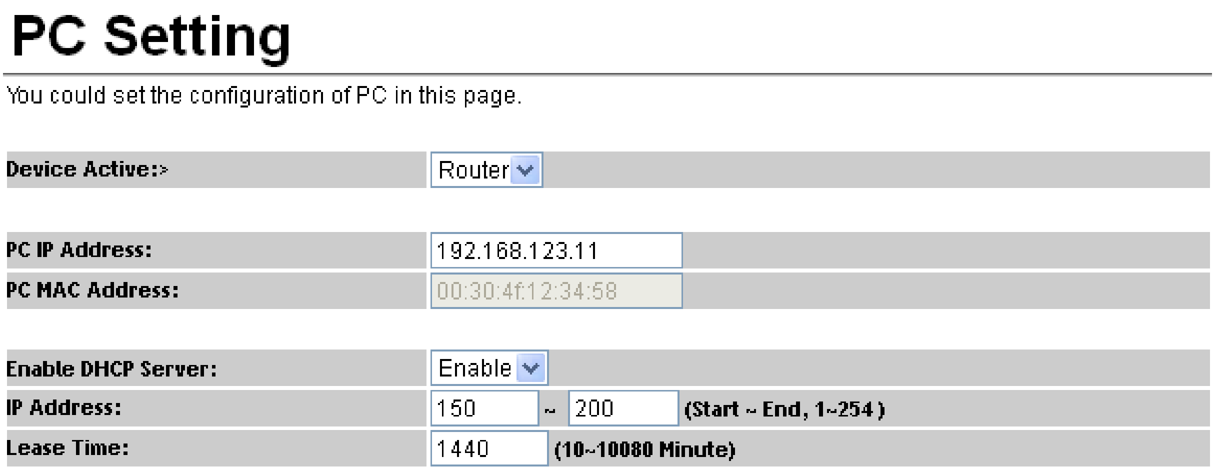
6.1 Stun Setting

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



6.2 PC Setting

This page defines the PC setting on this page.



Field	Description
Device Activity	The default is Bridge mode, and it also provides NAT mode. **Bridge:** When set as mode, the LAN and PC ports are in the same network segment. **NAT:** The LAN and PC ports are in a different network segment, and PC port could enable the DHCP Server function to allot the IP address.
PC IP address	The IP address of PC port. (In the **Birdge** mode, the **Default IP**:

Field	Description
	192.168.0.1
PC MAC Address	The MAC of PC port
Enable DHCP Server	It will allot the IP address automatically when enable this function.
IP Address	The range for DHCP IP address.
Lease Time	DHCP server lease time

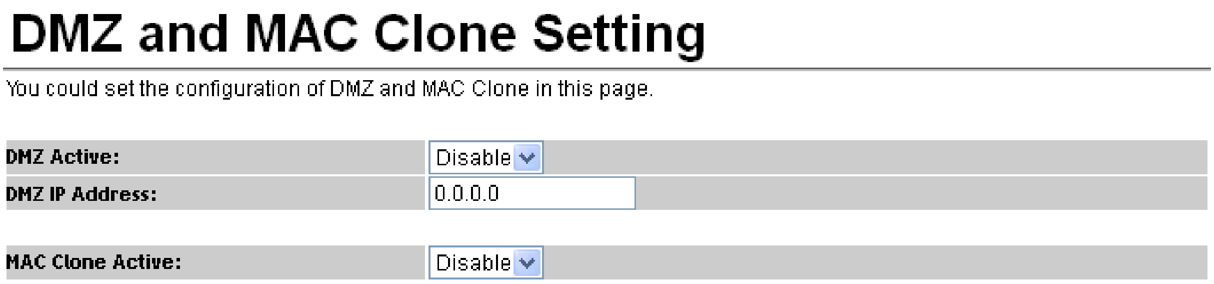
6.3 DMZ and MAC Clone

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 be sent to the specific IP address.

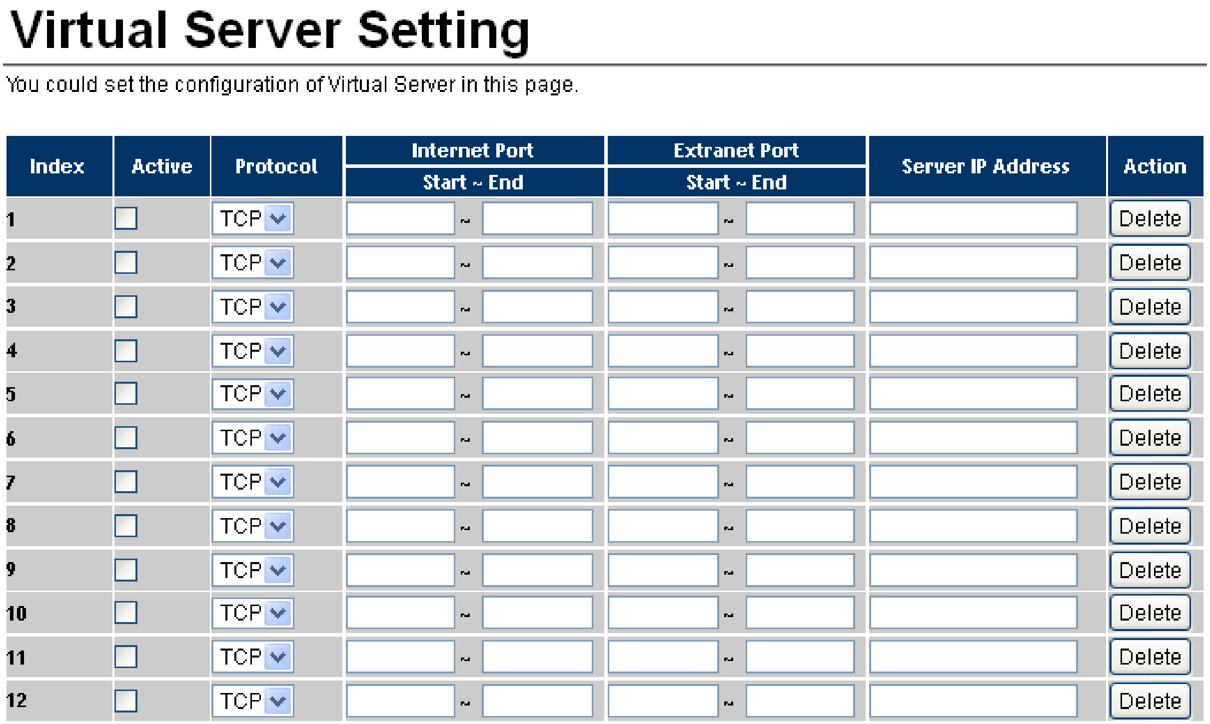
DMZ IP Address: The DMZ host IP address.

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 ISPs will limit the MAC address for PPPoE dial-up connection.



6.4 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.

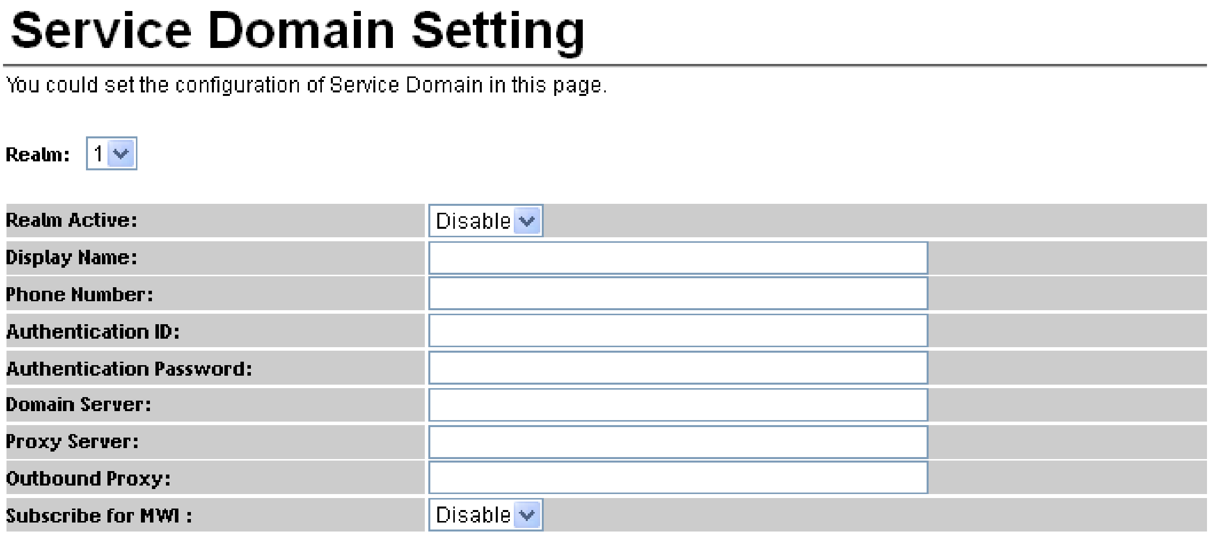


Field	Description
Index	The serial number -- There are totally 12 records from Num 1 to 12.
Activity	The activity status -- The default is Disable; this record will activate if enable.
Protocol	The TCP or UDP communication protocol.
Internal Port	For corresponding the internal port.
External Port	For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

CHAPTER 7. SIP SETTING

7.1 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.



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 enabled.
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 provide the information, then you can skip this item.
Subscribe to MWI	When system is enabled, it will frequently send the MWI message. Note: The starting number can’t be “0”. For example, if the number is “0xxxx”, the system will ignore this dial plan.

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

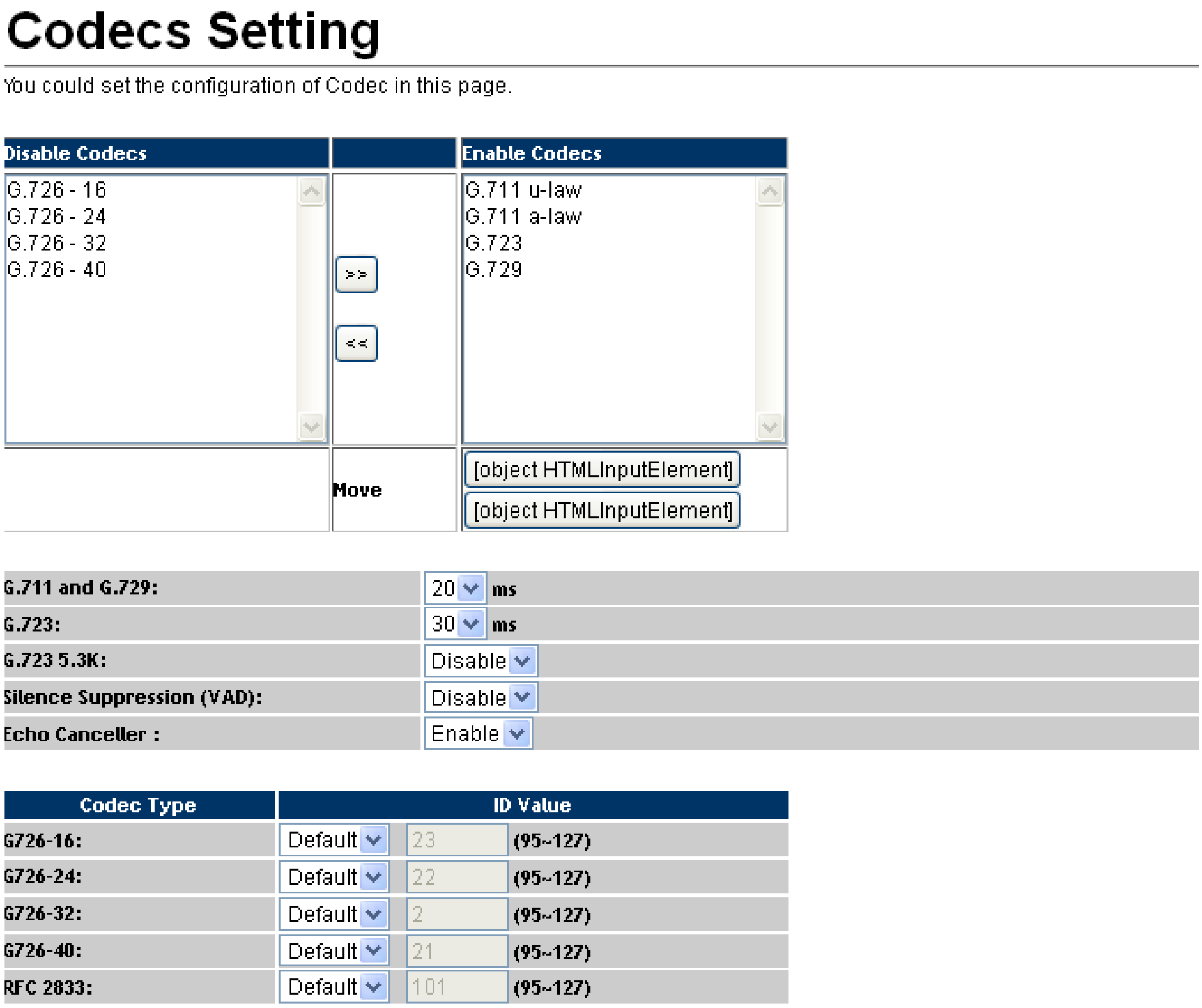
If you have more than one SIP account, you can follow the steps below to register to the other ISP.

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:	

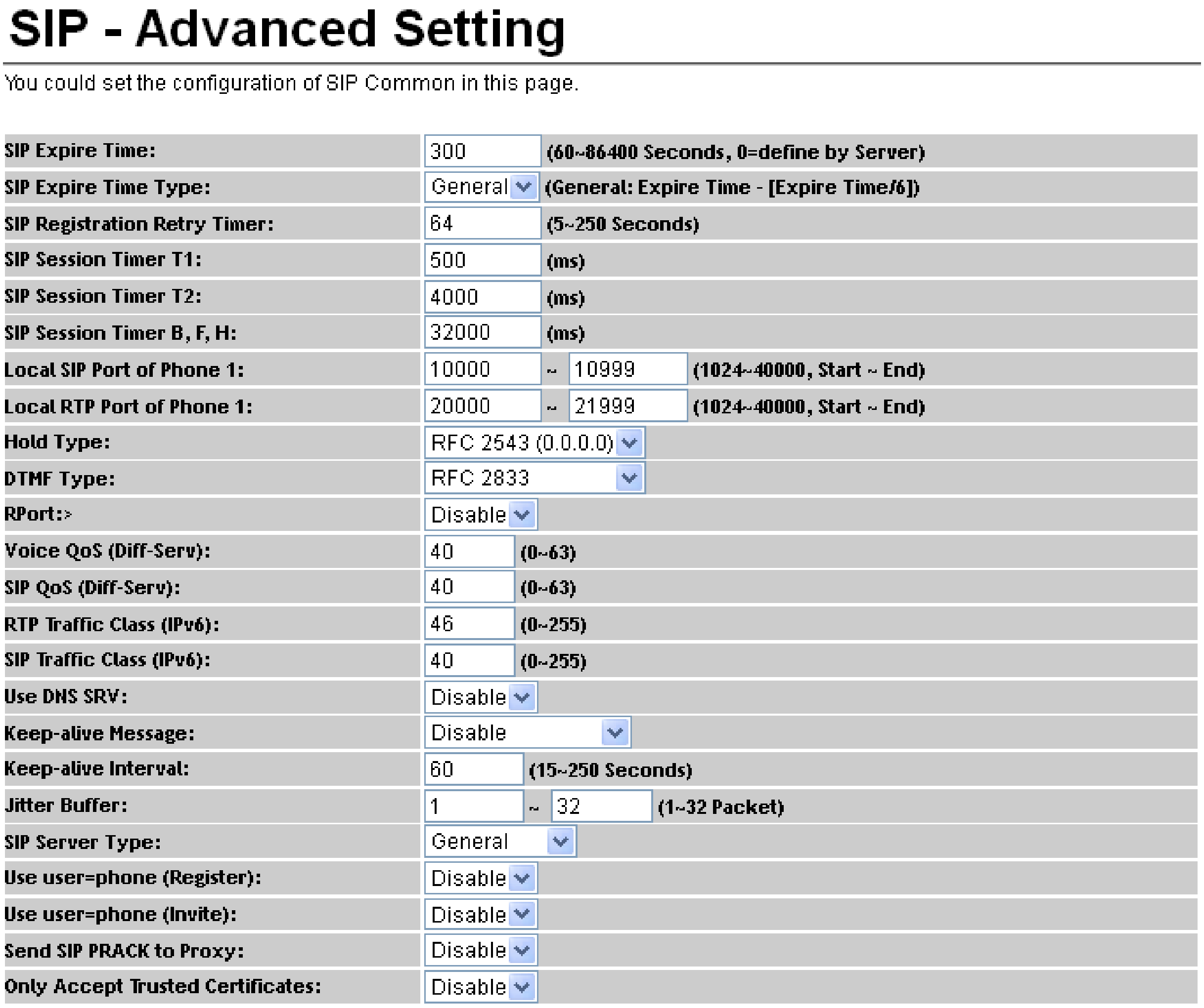
7.2 Codec Setting

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.



7.3 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.



Field	Description
SIP Expire Time	To set up the registration interval time.
SIP Expire Time Type	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. Note: Registered server needs to support this function. Register time calculated General: expiry time-[(expiry time/30)*6], when Expiry Time > 60 it will start to work; if less than 60 seconds, it will decrease 5 seconds. 1/2: expiry time * 1/2.

Field	Description
	2/3: expiry time * 2/3. 3/4: expiry time * 3/4. 4/5: expiry time * 4/5. 5/6: expiry time * 5/6. 6/7: expiry time * 6/7. 7/8: expiry time * 7/8. 8/9: expiry time * 8/9. 9/10: expiry time * 9/10.
SIP Register Retry Timer	If SIP register fails, system will retry interval after this time.
SIP session timer T1	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses. Note: Registered server needs to support this function.
SIP session timer T2	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses. Note: Registered server needs to support this function.
SIP session timer Timer B, F, H	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses. Note: Registered server needs to support this function. B: 64 * SIP T1; INVITE transaction timeout timer. F: 64 * SIP T1; non-INVITE transaction timeout timer. H: 64 * SIP T1, Wait time for ACK receipt.
Local SIP Port of phone 1	Setting the phone 1 SIP start and end port. All the port can't be duplicated
Local RTP Port of phone 1	Setting the phone 1 RTP start and end port. All the port can't be duplicated
Hold type	The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.
DTMF Mode	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.

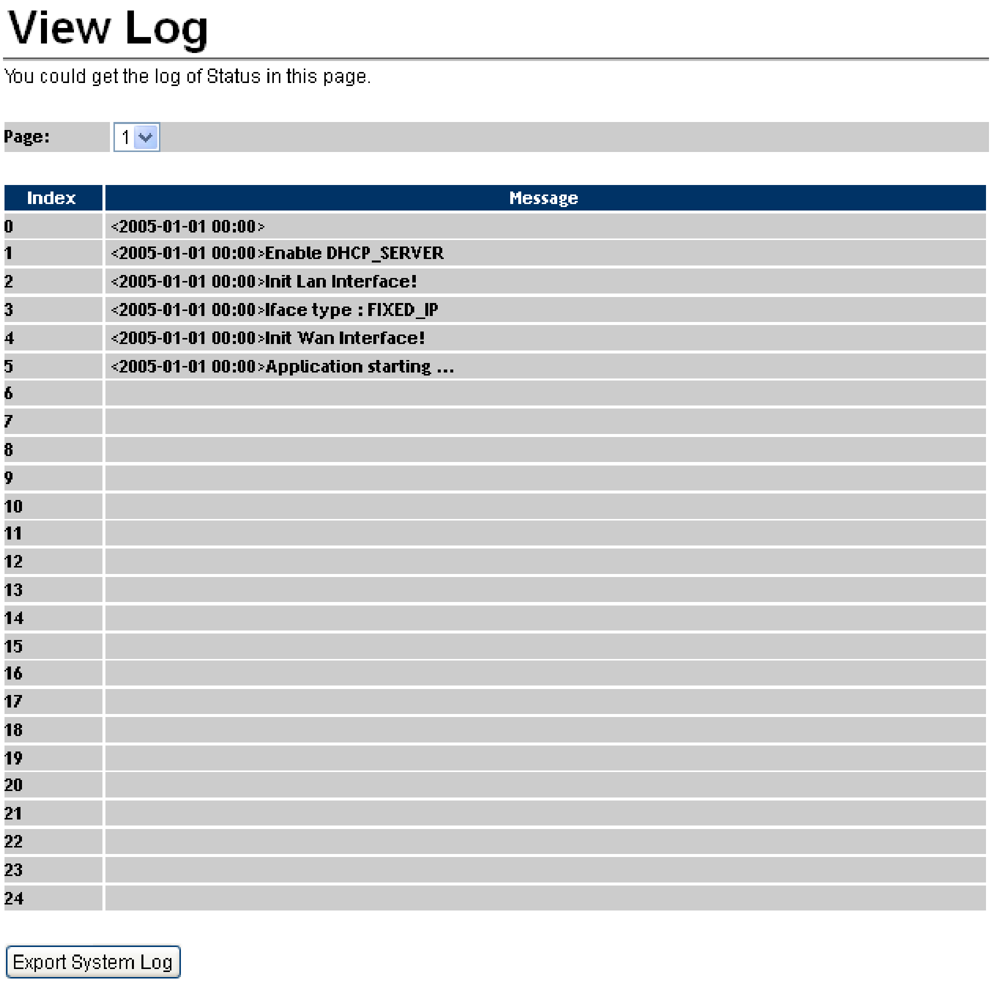
Field	Description
RPort	To change this setting, please follow your ISP information. When you finish the setting, please click the Submit button. Note: Registered server needs to support this function.
Voice QoS (Diff-Serv)	The Voice QoS feature.
SIP QoS (Diff-Serv)	The SIP QoS feature. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still needs to cooperate with the other Internet devices.
RTP Traffic Class (IPv6)	IPv6 RTP traffic class
SIP Traffic Class (IPv6)	IPv6 SIP traffic class
Use DNS SRV	The default is disable, and use DNS SRV mode. Set enable to use DNS to SRV mode to search the host information.
Send Keep Alive Packet	Always transport the network packets to keep the NAT port
Keep Alive Period	To set up the interval time for transporting packets.
Jitter Buffer	To set up the size for jitter buffer packets.
SIP Server Type	Provides a different register server: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink, for different server systems will adjust some system parameters Note: Registered server needs to support this function.
Use user = phone (Register):	When sending the registered package, in package Header will add the "user=phone" message. Note: Registered server needs to support this function.
Use user = phone (Invite):	When sending the dialing package, in package Header will add the "user=phone" message. Note: Registered server needs to support this function.

Field	Description
Send SIP PRACK to Proxy:	When sending the SIP package, in package Header will add the ”PRACK” message. Note: Registered server needs to support this function.
Only Accept Trusted Certificates:	Only accept call from proxy; if system receives the IP dialing, system will refuse the call.

CHAPTER 8. ADVANCE SETTING

8.1 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: 1

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	
12	
13	
14	
15	
16	
17	
18	
19	
20	
21	
22	
23	
24	

Export System Log

8.2 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.

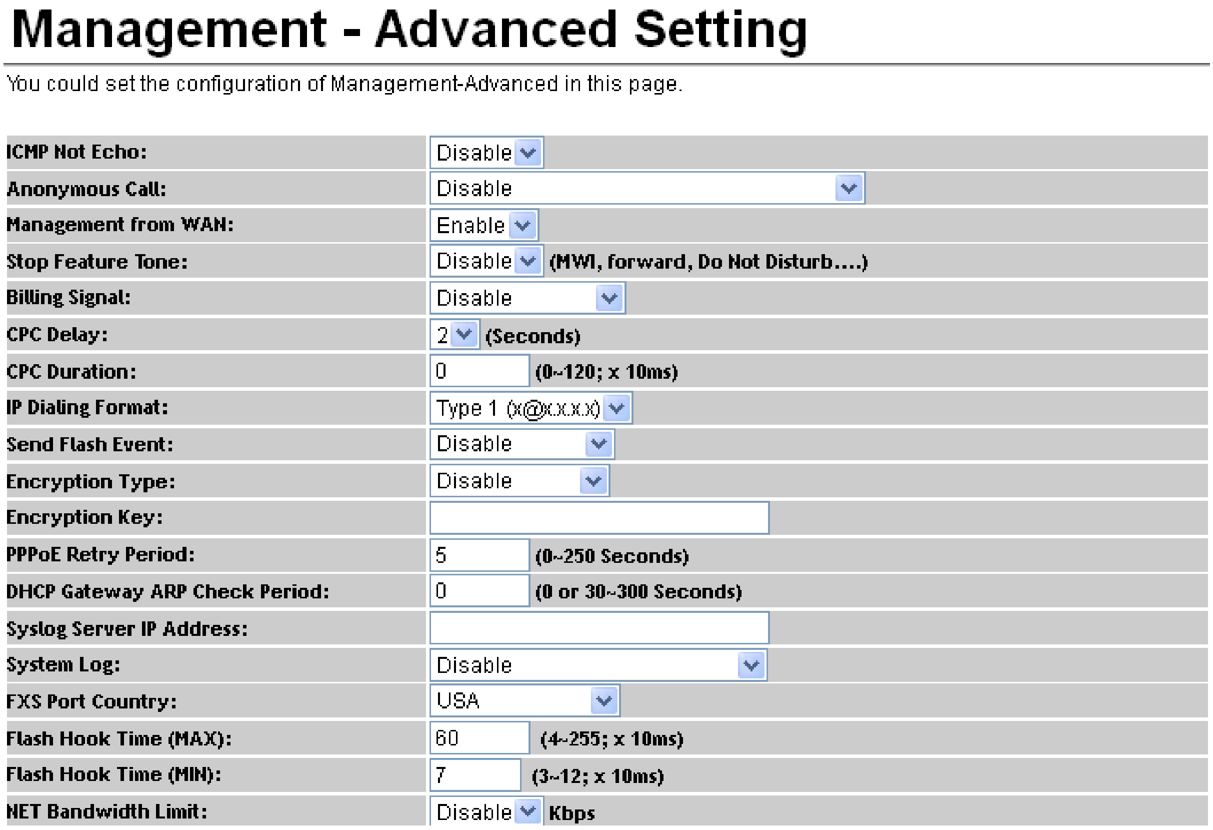
Auto Provision Setting

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. file/load/
Next Configuration time:		

8.3 Management -- Advanced Setting

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



Field	Description
ICMP Not Echo	This function can disable echo when someone pings this device. It can avoid hacker trying to attack the device
Anonymous Call	If this function is enabled, machine will to start the calling hidden function, and it will not send the related Caller information. Note: Registered server needs to support this function.
Management form WAN	When [Enable] allow user login from WAN.
Stop Feature Tone	When [Enable] if system sets the function like [Subscribe to MWI, forward, DND], when user picks up the phone, he will hear the remind tone [Do Do Do]
Billing Signal	There are three billing types: Polarity Reversal, Tone_12K and Tone_16K. Note: Registered server needs to support this function.

Field	Description
CPC Delay	When receiving the disconnected signal, machine will cut the voltage down to 0V after this time.
CPC Duration	When starting to cut the voltage down to 0V, machine will continue this state by this time.
IP Dialing Format	Setting IP dialing format; when [Disable] can't use IP dialing to make a call.
Send Flash event	There are two flash formats: DTMF Event and SIP Info.
Encrypt Type	There are seven encrypt formats: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP. Note: Registered server needs to support this function.
Encrypt Key	Some encrypt types must enter the Encrypt Key Note: Registered server needs to support this function.
PPPoE Retry Period	If PPPoE dial-up connection fails, machine will retry the dial-up motion after this time.
DHCP Gateway ARP Check Period	The period to check the DHCP gateway ARP.
Syslog Server IP Address	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.
System Log	Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address
PSTN port Country	Set up the FXS Port Coutry
PSTN Silence Timeout	Define the maximum silence time for FXO port. After the time, it will disconnect the line.
PSTN CID forward	It must work with [Phone – General] [Auto Answer] function or [Phone – Caller Service] [Forward] function. When this function is enabled, the caller ID from FXO can transfer to other devices.
Generate Flash Signal for PSTN	FXO flash time defines to hold or hang up the phone
FXS Port Coutry	Select the FXO port local country
Flash Hook Time (Max)	Maximum flash time to detect the call on hold or hang up.
Flash Hook Time (Min)	Minimum flash time to detect the call on hold or hang up.

Field	Description
NET Bandwidth Limit	Setting the limitation for LAN bandwidth

8.4 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 Tones. 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
Cadence On:	[ ]		[x]	[x]	[x]	[x]	[x]
Hi-Tone Freq.:	440		480	620	620	480	440
Lo-Tone Freq.:	350		440	480	480	440	350
Hi-Tone Gain:	4522		2261	2261	2261	15360	2261
Lo-Tone Gain:	4522		2261	2261	2261	15360	1130
On Time 1:	0	**x 10ms**	200	50	30	200	30
Off Time 1:	0	**x 10ms**	400	50	20	400	20
On Time 2:	0	**x 10ms**	0	0	0	0	30
Off Time 2:	0	**x 10ms**	0	0	0	0	400
On Time 3:	0	**x 10ms**	0	0	0	0	0
Off Time 3:	0	**x 10ms**	0	0	0	0	0

8.5 TR-069

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

Note	A 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.

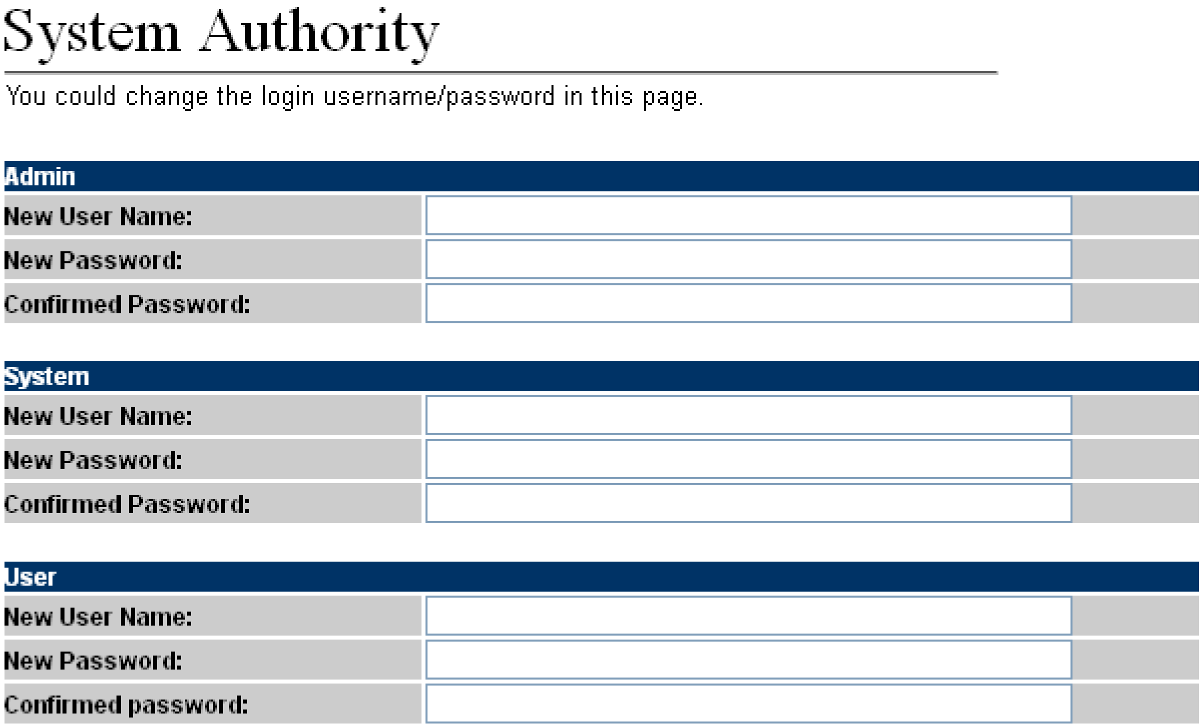
ACS Service:	Disable
ACS Interval:	60 (30~86400 Seconds
Provision Code:	provisioningCode
ACS URL:	http://iop.tw.workssys.com/comserver/node1/tr069
ACS Username:	
ACS Password:	
Device S/N :	

Connection Request TCP:	Enable
Connection Request TCP SSL:	Disable
Connection Request TCP Port:	7547 (1~65535)
Connection Request UDP:	Enable
Connection Request UDP SSL:	Disable
Connection Request UDP Port:	7547 (1~65535)
Connection Request Path:	
Connection Request Authority:	0
Connection Request Username:	
Connection Request Password:	
ISP Username:	
ISP Password:	

CHAPTER 9. OTHER SETTINGS

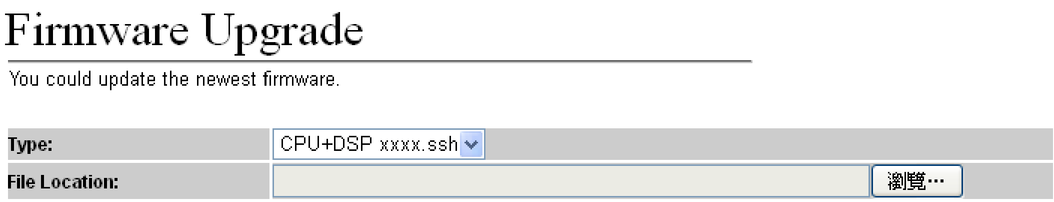
9.1 System Authority

In System Authority, admin/System/User login password can be changed.



9.2 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.



If your updated 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.

Note	For technological consideration, we strongly suggest you refer to the following upgrade methods for updating your device. After firmware is loaded, the unit will reboot, and default IP address of the customized firmware: http://192.168.0.1; login name/password: **root/null (no password)**

9.3 Auto Update Setting

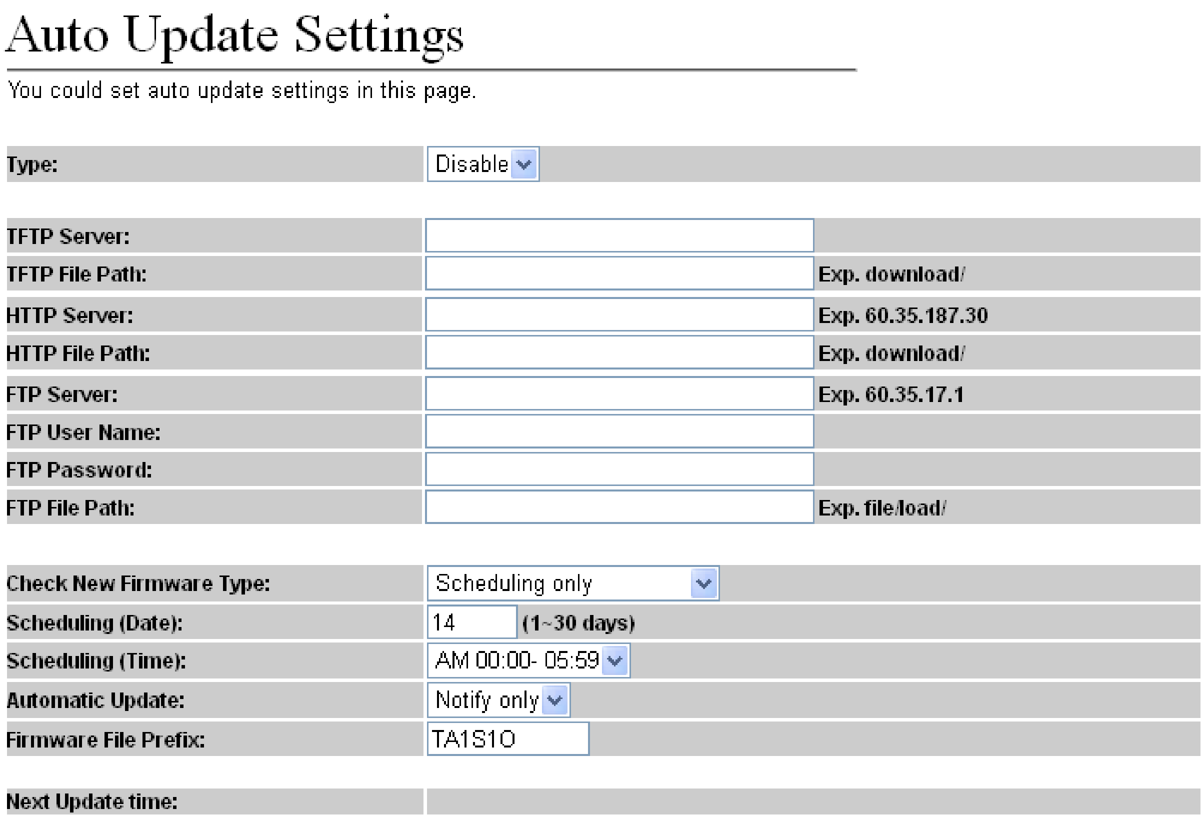
The device can update new firmware with the **gz** or **ds** file format automatically by the Auto Upgrade function.

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

Field	Descriptions
	− **Scheduling**: The machine will follow the scheduled date and time to check the new firmware.
Scheduling (Date)	The machine will check the new firmware between the time range by random.
Automatic Update	There are Notify only and Automatic ways to update. − **Notify only**: If there are new firmware, the ATA will send the “Beep” sound when picking up the handset to prompt you there is a new firmware. − **Automatic**: The device will carry out firmware update automatically.
Firmware File Prefix	It will check the information on model name.
Next update time	It will show the next check date and time.

Note

If the **Check new firmware** field is set to Power On, the machine will check the new firmware according the scheduled time/date and power on. If there is a 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.

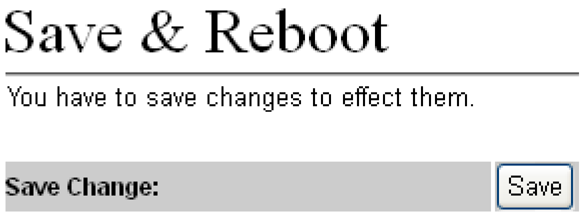


9.4 Reset to default

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.

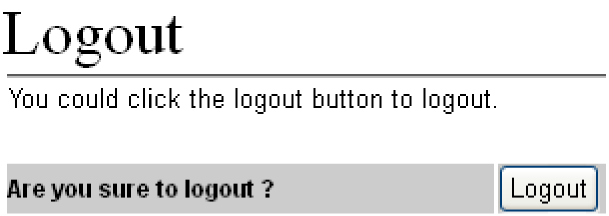
9.5 Save and Reboot

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 take effect.



9.6 Logout

Logging out the system will return you to the login page.



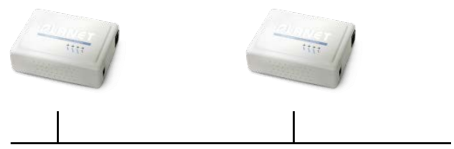
APPENDIX A VOICE COMMUNICATION SAMPLES

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 and 192.168.0.2
Analog telephone sets are connected to the **phone** (RJ11) port of ATAs respectively

192.168.0.1 192.168.0.2



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

Test the scenario:

1. Pick up the telephone set on ATA A.
2. Press the keypad: **192*168*0*2#** 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**', and phone 2 will ring.
Analog telephone sets are connected to the phone (RJ11) ports of VIP-157S respectively

192.168.0.1

1001 1002

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

Test the scenario:

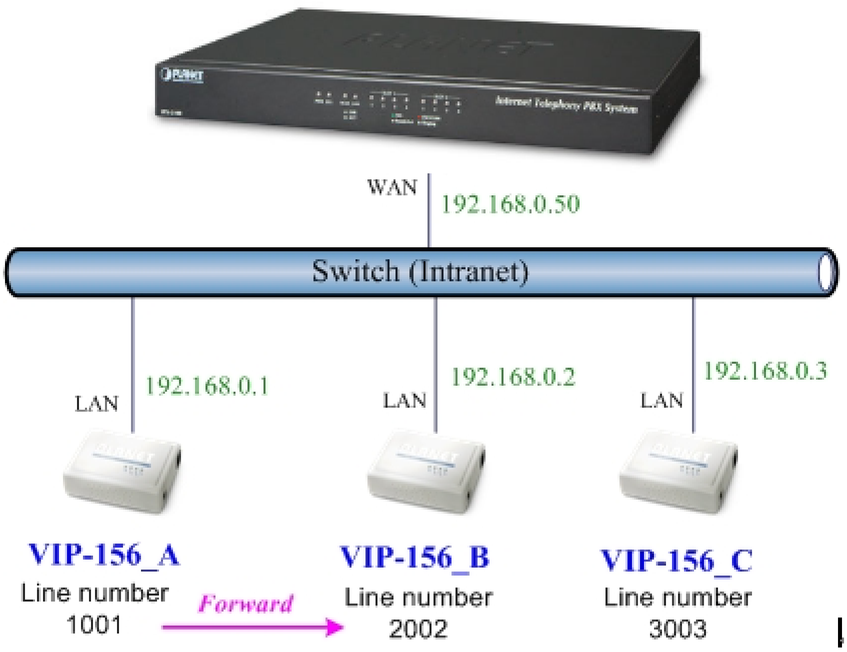
1. Pick up the telephone set on VIP-157S port 1, and you should be able to hear the dial-tone.
2. Press the keypad: **192*168*0*1**5062#** and will be able to connect to the VIP-157S port 2.
3. Then the telephone set in VIP-157S port 2 should ring. Please repeat the same dialing steps on port 2 to establish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

Note	
Note	● 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 supports SIP applications, and suitable settings should be applied to the NAT device to enable the SIP communications before making calls ● [**VIP-157S**] in PLANET ATA series products, to connect to remote ATA, press the keypad in the following sequence to connect to the remote VIP-157S port 2: **[remote ATA IP address]**5062**, for example: **192*168*0*2**5062**

Case 3: 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 registers 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 to 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.

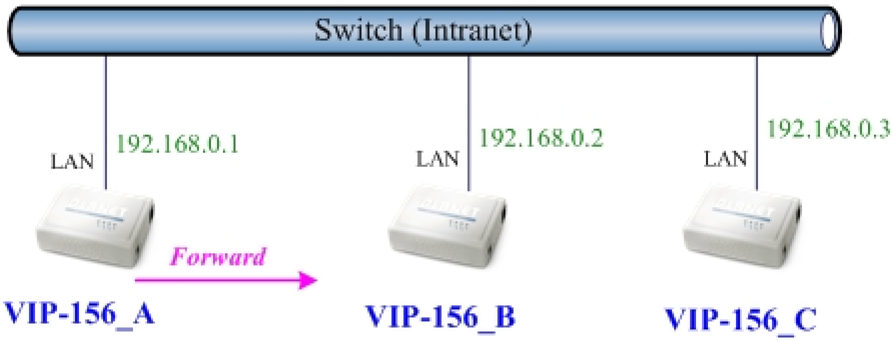
Forward Type	Forward Number	Ring / Phone
Always	2002	3 Phone 1

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 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 to 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	192.168.0.2	3 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

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 called A and they are in the process of conversation.
2. A carries out the transfer function (Press the "**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 called 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 Conferencing

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 the "**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 called A and A will hear the prompt sounds.

3. A presses the "**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-157S SPECIFICATIONS

<table>
<tr><td>Product</td><td colspan="3">SIP Analog Telephone Adapter</td></tr>
<tr><td>Model</td><td>VIP-156</td><td>VIP-156PE</td><td>VIP-157S</td></tr>
<tr><td colspan="4">Hardware</td></tr>
<tr><td>LAN</td><td colspan="3">1 x 10/100Mbps RJ45 port (802.3af PoE for VIP-156PE)</td></tr>
<tr><td>PC</td><td colspan="3">1 x 10/100Mbps RJ45 port</td></tr>
<tr><td>FXS (for telephone set connection)</td><td colspan="2">1 x RJ11</td><td>2 x RJ11</td></tr>
<tr><td colspan="4">Protocols and Standard</td></tr>
<tr><td>Standard</td><td colspan="3">SIP 2.0 (RFC3261)</td></tr>
<tr><td>Voice codec</td><td colspan="3">G.711 u-law/a-law, G.723 (6.3k/5.3k), G.726, G.729</td></tr>
<tr><td>Fax support</td><td colspan="3">T.38, G.711 pass-through</td></tr>
<tr><td>Voice Standard</td><td colspan="3">Voice activity detection (VAD)
Comfort noise generation (CNG)
Acoustic echo canceller (AEC)
G.165: Line echo canceller (LEC)
Jitter buffer</td></tr>
<tr><td>Protocols</td><td colspan="3">SIP 2.0 (RFC-3261), IPv4/IPv6 TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE, VLAN, VPN (PPTP & L2TP), TR069</td></tr>
<tr><td colspan="4">Network and Configuration</td></tr>
<tr><td>Access Mode</td><td colspan="3">Static IP, PPPoE, DHCP</td></tr>
<tr><td>Management</td><td colspan="3">Web, keypad</td></tr>
<tr><td>Dimensions (W x D x H)</td><td colspan="3">94 x 72 x 30 mm</td></tr>
<tr><td>Operating Environment</td><td colspan="3">0~40 degrees C, 10~95% humidity</td></tr>
<tr><td>Power Requirements</td><td colspan="3">12V DC</td></tr>
<tr><td>EMC/EMI</td><td colspan="3">CE, FCC Class B</td></tr>
</table>

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