



VoIP Analog Telephone Adapter

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

User's manual

Version 3.01

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CE mark Warning

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

WEEE Warning



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

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Revision

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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/VIP-157S VoIP Phone Adapter (“**ATA**” in the following term) converts standard telephones to IP-based networks. The service providers and enterprises offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

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

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

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

Product Features

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

VoIP Features

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

- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation (CNG)
- In band, out-of-band, and SIP-info DTMF support

Package Content

The contents of your product should contain the following items:

VoIP Telephone Adapter

Power adapter

Quick Installation Guide

User's Manual CD

RJ-11 cable x 1

Physical Details

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

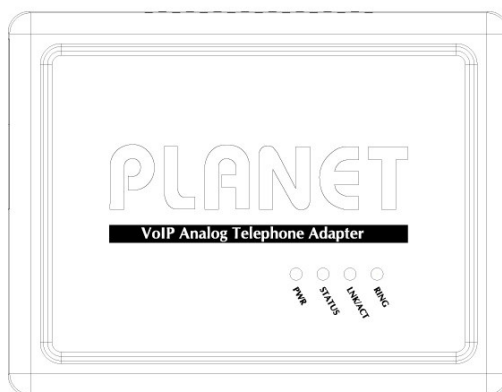
Respective model/descriptions are shown below:

VIP-156: SIP Analog Telephone Adapter

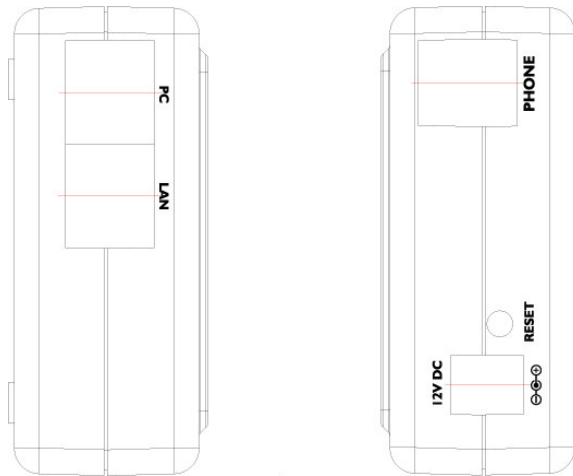
VIP-156PE: 802.3af PoE SIP Analog Telephone Adapter

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

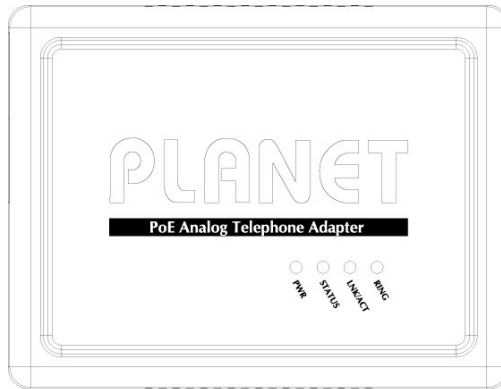
VIP-157S: 2-port FXS SIP Analog Telephone Adapter



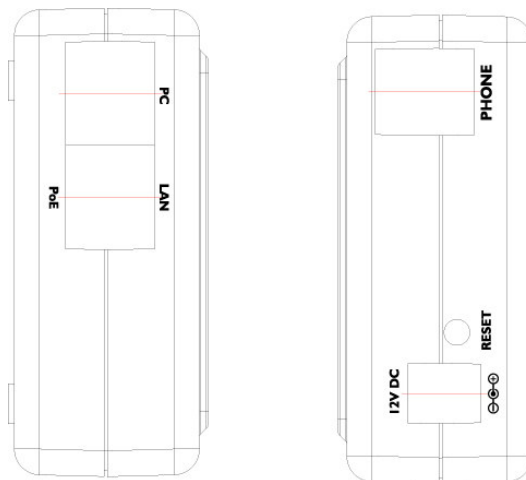
Front Panel of VIP-156



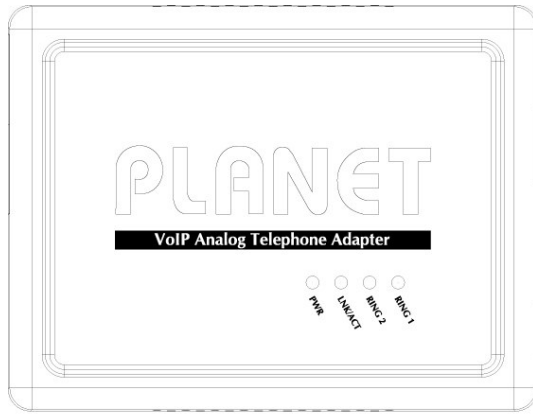
Left / Right Panel of VIP-156



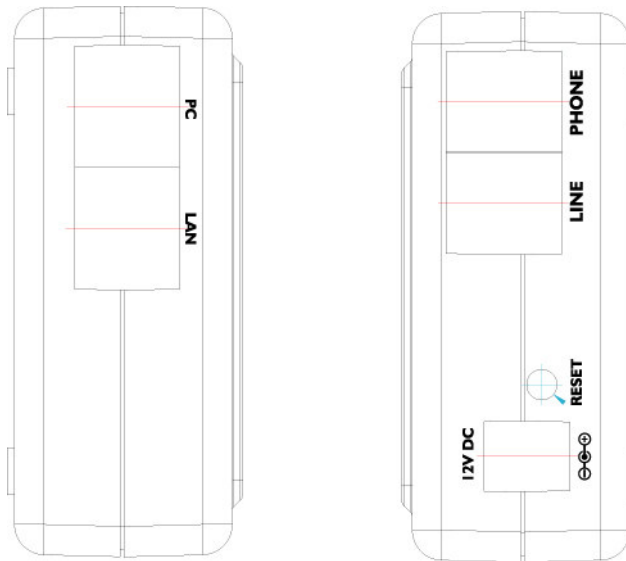
Front Panel of VIP-156PE



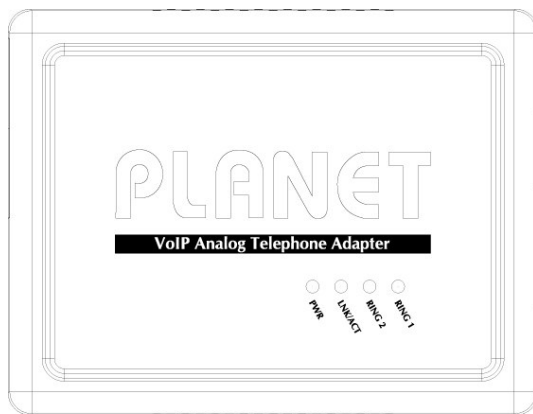
Left / Right Panel of VIP-156PE



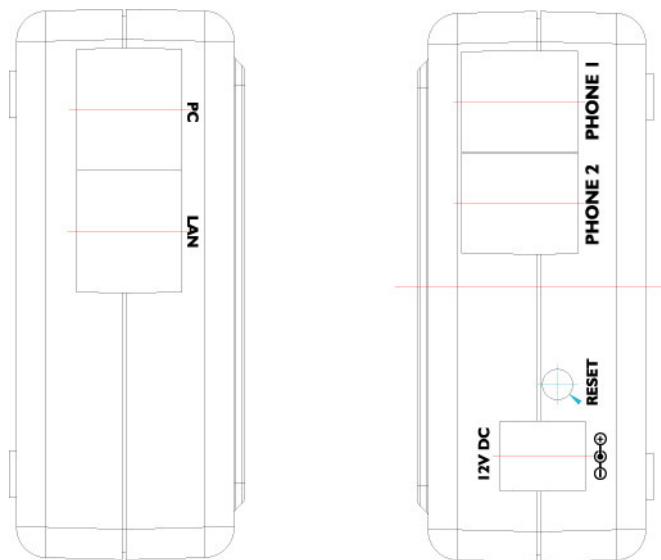
Front Panel of VIP-157



Left / Right Panel of VIP-157



Front Panel of VIP-157S



Left / Right Panel of VIP-157S

LED Display & Button

1	PC	RJ-45 connector, to maintain the existing network structure, connected directly to the PC through straight CAT-5 cable
2	LAN	RJ-45 connector, for Internet access, connected directly to Switch/Hub through straight CAT-5 cable. 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.

Preparations & Installation

Physical Installation Requirement

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

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

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

Administration Interface

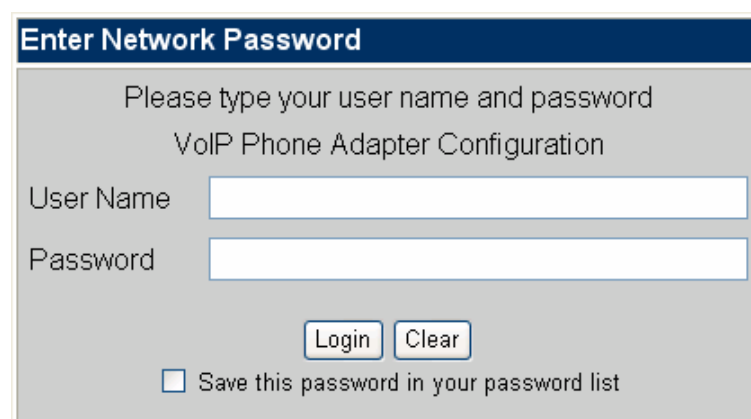
PLANET ATA provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access

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

- Microsoft Internet Explorer 4.01 or higher with Java support

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



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

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

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (default: 192.168.0.1) of VIP in the address bar. After logging on machine with username/password (default: root / no passwd), browse to "Network" --> "Network settings" configuration menu:

Network Settings

You could configure your network settings in this page.

TCP/IP Configuration	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	<input type="text" value="192.168.0.1"/>
Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="192.168.0.254"/>
DNS Server 1:	<input type="text" value="168.95.1.1"/>
DNS Server 2:	<input type="text" value="168.95.192.1"/>
MAC:	<input type="text" value="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.
#190#	Unlock	None	Must unlock the protect function before set up network settings and ATA function via keypad.
#191#	Lock	None	The system will be lock and can't set up network settings via keypad.
#195#	Reboot	None	The system will reboot automatically.
#198#	Factory Reset	None	The system will be reset to factory default value and reboot automatically.
0*	To switch PSTN mode	None	VIP-157 only

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 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.
#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.

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 Bland 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 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

#145#	Forward function disable	Disable forward function	
#146+Number#	enable forward to FXS Port	Enable forward to FXS Port	
#147+Number#	enable forward to FXO Port	Enable forward to FXO Port	

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 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 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:

Network Service Configurations

Configuring and monitoring your ATA from web browser

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

Overview 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

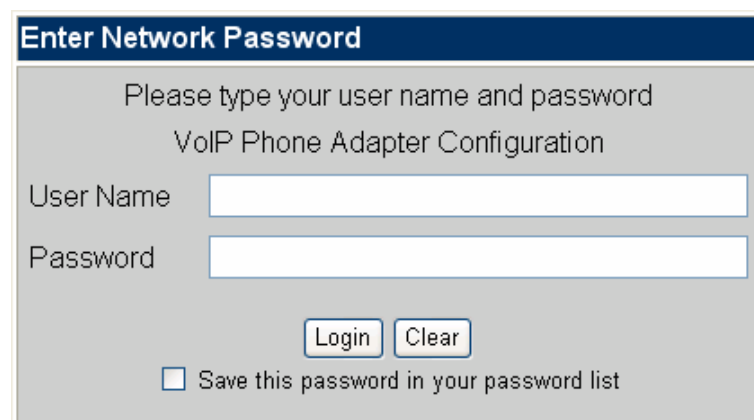
- ◆ Microsoft Internet Explorer 4.01 or higher with Java support

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)**



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

ATA log in page

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

PLANET
Networking & Communication

**VoIP Phone Adapter
Configuration Menu**

- Phone Book
- Phone Setting
- Network
- SIP Settings
- NAT Trans.
- Advanced Settings
- System Auth.
- Save & Reboot
- System Settings

System Information

This page illustrate the system related information.

Company:	PLANET Technology Corp.
Firmware Version:	3.0
Codec Version:	1.0
Contact Address:	11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C
Tel:	886-2-22199518
Fax:	888-2-22199528
E-Mail:	support_voip@planet.com.tw
Web Site:	www.planet.com.tw

VoIP Phone Adatper main page

VoIP Telephone Adapter Configurations

Phone Book

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

Phone Book

You could add/delete items in current phone book.

Phone Book Page:

Phone	Name	URL	Select
0			<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Add New Phone

Position: (0~139)

Name:

URL:

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

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

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

If you want to delete all phone numbers, you can click “Delete All” button.

For Example:

Phone Book

You could add/delete items in current phone book.

Phone Book Page:

Phone	Name	URL	Select
0	301	301@192.168.1.2	<input type="checkbox"/>
1	206	17476433364	<input type="checkbox"/>
2	202	192.168.1.2:5062	<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected

Delete All

Reset

Ex_1:

ATA had added the above phone numbers. User pick up the handset and dial the “**301**” to make the P2P call ([301@192.168.1.2](tel:301@192.168.1.2)).

Ex_2:

Users pick up the handset and dial the “**206**” to make the Proxy call (17476433364).

Ex_3:

Users pick up the handset and dial the “**202**” to make the P2P call (192.168.1.2:5062).

Call forward

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

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

speed dial number you choose.

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

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

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

Forward Settings

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

All Forward:	<input checked="" type="radio"/> Off <input type="radio"/> On
Busy Forward:	<input checked="" type="radio"/> Off <input type="radio"/> On
No Answer Forward:	<input checked="" type="radio"/> Off <input type="radio"/> On

	Name	URL
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out:	<input type="text" value="3"/> (2~8 Ring)
-------------------------	---

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

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

Forward Settings

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

All Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out:	<input type="text" value="3"/> (2~8 Ring)
-------------------------	---

Call Forward function for VIP-157

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:	<input checked="" type="radio"/> On	<input type="radio"/> Off
Primary Server:	<input type="text" value="192.43.244.18"/>	
Secondary Server:	<input type="text" value="208.184.49.9"/>	
Time Zone:	GMT + <input type="text" value="08"/> : <input type="text" value="00"/> (hh:mm)	
Sync. Time:	<input type="text" value="1"/> : <input type="text" value="0"/> : <input type="text" value="0"/> (dd:hh:mm)	

Volume Setting

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

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

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

Volume Setting

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

Handset Volume:	<input type="text" value="10"/>	(0~12)
Handset Gain:	<input type="text" value="10"/>	(0~15)

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

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

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

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

Volume Setting

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

Handset Volume:	<input type="text" value="10"/>	(0~12)
PSTN-Out Volume:	<input type="text" value="10"/>	(0~12)
Handset Gain:	<input type="text" value="10"/>	(0~15)
PSTN-In Gain:	<input type="text" value="10"/>	(0~15)

Volume Settings for VIP-157

Block Setting

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

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

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

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

Block Setting

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

Always Block:	<input type="radio"/> On <input checked="" type="radio"/> Off
Block Period:	<input type="radio"/> On <input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)
To:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)

Auto Answer (For VIP-157)

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

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

For security issue, You'd better to set the PIN Code. If you have set the PIN code, you will hear a tone to inform you input the PIN Code then you can dial out.

Auto Answer

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

Auto Answer:	<input type="radio"/> On <input checked="" type="radio"/> Off
Auto Answer Counter:	<input type="text" value="03"/> (2~15)
PIN Code Enabled:	<input type="radio"/> On <input checked="" type="radio"/> Off
PIN Code:	<input type="text"/>

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:	<input type="text" value="Don't show caller ID"/>
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No

Dial Plan Settings

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

Dial Plan Settings

You could set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	002 + 1234+4321
Drop prefix :	<input checked="" type="radio"/> Yes <input type="radio"/> No
Replace rule 2:	006 + 002+003+004
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	007 + 5xxx+35xx
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	+

Dial now:	*xx+#xx+11x+xxxxxxx
Auto Dial Time:	5 (3~9 sec)
Use # as send key:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Use * for IP dialing:	<input checked="" type="radio"/> Yes <input type="radio"/> No

Field	Description
Drop Prefix	The rule of add or replace code. If setup as No, it will add the prefix number prior to the identification number. If setup as Yes, it will replace the identification number.
Replace rule	The prefix number. It only accept the numeral and the max length is 8.
+	The identification number. It can accept the numeral or symbol and the max length is 40. <ul style="list-style-type: none"> - Symbol: It only accept the [+], [x] - +: It means as "or". For example, [123+456+334+5xx] even if [123 or 456 or 334 or 5xx] - x: It is equal to 0~9. For example, [5xx] even if the number begin 5.
Dial Now	If the dialing number are match with this field, it will dial out and need not to press the "#" key to end the dialing. It accepts the numeral or symbol, and the max length are 124. Note: The starting number can't be the "0". For example, if the number is "0xxxx", because the starting number is "0", so that the

	system will ignore this dial plan.
Auto Dial Time	Stop dialing after seconds then send dial number out.
Use # as send key	If setup as Yes, the system will stop to receive the dialing number when receive the [#] key. The system also will determine the Auto Dial Time, it will carry out the calling if there isn't receive the digit after the Auto Dial Time. If setup as No, the system just according to the Auto Dial Time to determine the end time.
Use * for IP dialing	If setup as Yes, the system will look on [*] as [.]. For example, if dial the "192*168*0*100#", it will dial out as "192.168.0.100#". If setup as No, it just look on [*] as [*]. For example, if dial the "700*#", it will dial out as "700*#".

Descriptions of example:

Example_1: Drop prefix: **No**, Replace rule 1: **002**, +: **1234+4321** (No limit the digit length)

1. If the dialing number is start as "1234", it will add the 002 at begin. The real dialing number is [0021234...].
2. If the dialing number is start as "4321", it will add the 002 at begin. The real dialing number is [0024321...].

Example_2: Drop prefix: **Yes**, Replace rule 2: **006**, +: **002+003+004** (No limit the digit length)

1. If the dialing number is start as "002", it will replace 002 by 006. The real dialing number is [006...].
2. If the dialing number is start as "003", it will replace 003 by 006. The real dialing number is [003...].

Example_3: Drop prefix: **No**, Replace rule 3: **007**, +: **5xxx+35xx** (Has limit the digit length)

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

Example_4: Dial Now: ***xx+#xx+11x+xxxxxx**

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

Flash Time Setting

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 Setting

You could set the flash time in this page.

Max Flash Time:	<input type="text" value="60"/>	(Range:4~255, Unit:10ms)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>		

Flash Time Settings for VIP-156/VIP-156PT/VIP-157S

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

Flash Time Setting

You could set the flash time in this page.

FXO Flash Time		
Flash Time:	<input type="text" value="5"/>	(3~200, 1->10ms)
SLIC Flash Time		
Max Flash Time:	<input type="text" value="60"/>	(4~255, 1->10ms)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>		

Flash Time Settings for VIP-157

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:	<input checked="" type="radio"/> On <input type="radio"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

T.38 (FAX) Settings

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

T.38 (FAX) Setting

You could enable/disable the FAX function in this page.

T.38 (FAX):	<input type="radio"/> On <input checked="" type="radio"/> Off
T.38 Port:	<input type="text" value="61000"/> (1024~65533)

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

T.38 (FAX) Setting

You could enable/disable the FAX function in this page.

T.38 (FAX):	<input type="radio"/> On <input checked="" type="radio"/> Off
T.38 Port of Phone1:	<input type="text" value="61000"/> (1024~65533)
T.38 Port of Phone2:	<input type="text" value="61100"/> (1024~65533)

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

Hot line Settings

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

Use Hot Line: Click Enable to carry the Hot line function out.

Hot line number: The hot line number, it can input the IP address or registration number.

Hot line Setting

You could set the hot line in this page.

Use Hot Line : Enable Disable

Hot line number:

Alarm Settings

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

Alarm: The default is Off. If set up as On, the telephone will ringed up at the specific time.

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

Current time: The next alarm time.

Alarm Settings

You could set the alarm time in this page.

Alarm: ON OFF

Alarm Time: : (hh:mm)

Current time: 2005-01-01 08:34

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:	<input type="text"/>
User Name:	<input type="text"/>
Password:	<input type="text"/>
E-mail Address:	<input type="text"/>

DDNS Server:	<input type="text"/>
DDNS Server List:	User Input <input type="text"/>
Type:	dyndns <input type="text"/>
Wild Card:	on <input type="text"/>

BACKMX:	<input type="radio"/> On <input type="radio"/> Off
Off Line:	<input type="radio"/> On <input type="radio"/> Off

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.

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 directly.

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:	<input type="radio"/> On <input checked="" type="radio"/> Off
Display Name:	<input type="text"/>
Line Number:	<input type="text" value="1001"/>
Register Name:	<input type="text"/>
Register Password:	<input type="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Status:	Not Registered

Note:

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 any one SIP accounts for making calls through input the switch code.

Realm switch code:

1*: Realm 1

2*: Realm 2

3*: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

Port Settings

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

Port Settings

You could set the port number in this page.

SIP Port:	<input type="text" value="5060"/>	(1024~65535)
RTP Port:	<input type="text" value="60000"/>	(1024~65535)

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.711 u-law ▼
Codec Priority 2:	G.711 a-law ▼
Codec Priority 3:	G.723 ▼
Codec Priority 4:	G.729 ▼
Codec Priority 5:	G.726 - 16 ▼
Codec Priority 6:	G.726 - 24 ▼
Codec Priority 7:	G.726 - 32 ▼
Codec Priority 8:	G.726 - 40 ▼
Codec Priority 9:	GSM ▼

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

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

Codec ID Setting

This page defines the Codec ID. Sometimes 2 VoIP device with different Codec ID will cause the interoperability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID. When you finished the setting, please click the Submit button.

Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	<input type="text" value="23"/> (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	<input type="text" value="22"/> (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	<input type="text" value="2"/> (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	<input type="text" value="21"/> (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	<input type="text" value="101"/> (95~255)	<input checked="" type="checkbox"/> 101

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

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

Other Settings

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

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

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

Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS:	<input type="text" value="40"/> (0~63)
SIP QoS:	<input type="text" value="40"/> (0~63)
SIP Expire Time:	<input type="text" value="60"/> (60~86400 sec)
Use DNS SRV:	<input type="radio"/> On <input checked="" type="radio"/> Off

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:	<input type="text"/>
STUN Port:	<input type="text" value="3478"/> (1024~65535)

Auto Configuration

This page defines the Auto Configuration (Auto Provision) setting. ATA supports TFTP, FTP, HTTP and IP PBX auto configuration function in total. In IP PBX Auto Configuration Setting you need to check with your service provider if they have provided this function. Usually this function will be bundle with an IP PBX to use in the office.

Auto Configuration Settings

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

Auto Configuration: Off TFTP FTP HTTP IP-PBX

TFTP Server:	<input type="text"/>
HTTP Server:	<input type="text"/> Exp. 60.35.187.30
HTTP File Path:	<input type="text"/> Exp. /download/

FTP Server:	<input type="text"/> Exp. 60.35.17.1
FTP Username:	<input type="text"/>
FTP Password:	<input type="text"/>
FTP File Path:	<input type="text"/> Exp. /file/load

ICMP Settings

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

ICMP Settings

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

ICMP Not Echo: On Off

Submit

Reset

Cancel without to tag

This function can decide the device if send the cancel tag to Proxy Server. If there has the similar symptom that the caller cancel the call before the called answer the call, the called still continue to ring up even the caller has cancel this call. It could try select this function to **Yes** to avoid the above symptom.

Cancel without to tag

Cancel without to tag.

Cancel without to tag. Yes No

Submit

Reset

PTT Settings

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

PTT Setting

You could select the PTT setting for different country in this page.

SLIC-PTT: USA

Submit

Reset

PTT Settings for VIP-156/VIP-157S

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

PTT Setting

You could select the PTT setting for different country in this page.

PSTN-PTT:	<input type="text" value="USA"/>
SLIC-PTT:	<input type="text" value="USA"/>

PTT Settings for VIP-157

System Authority

In System Authority you can change your login password.

System Authority

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

Username:	<input type="text" value="root"/>
New password:	<input type="text"/>
Confirmed password:	<input type="text"/>

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:

Firmware Upgrade

In Firmware Upgrade function you can update new firmware via HTTP or TFTP methods in this page.

You can upgrade the firmware by the following steps:

Select the upgrade method and the firmware code type, AP or DSP code.

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

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

Firmware Upgrade

You could update the newest firmware.

Method: Local PC TFTP

Local PC	
Code Type:	AP <input type="button" value="v"/>
File Location:	<input type="text"/> <input type="button" value="Browse.."/>

TFTP	
TFTP Server:	<input type="text" value="192.168.1.250"/>

Note:

For technological consideration, we've strongly suggested referring to the following upgrade methods for update your device.

After firmware loaded, the unit will be reboot, and Default IP address of the customized firmware:

<http://192.168.0.1>; login name/password: **root/null (no password)**

Auto Upgrade

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

Auto Update Settings

You could set auto update settings in this page.

Update via: Off TFTP FTP HTTP

TFTP Server:	<input type="text"/>	
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/

FTP Server:	<input type="text"/>	Exp. 60.35.17.1
FTP Username:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. /file/load

Check new firmware:	<input type="radio"/> Power ON <input checked="" type="radio"/> Scheduling
Scheduling (Date):	<input type="text" value="14"/> (1~30 days)
Scheduling (Time):	AM 00:00- 05:59 <input type="button" value="v"/>
Automatic Update:	<input checked="" type="radio"/> Notify only <input type="radio"/> Automatic
Firmware File Prefix:	<input type="text" value="TA1S"/>

Next update time:	<input type="text"/>
-------------------	----------------------

Field	Descriptions
Update via	There are TFTP/ FTP and HTTP three ways to provide the auto upgrade function.
TFTP Server	Input the TFTP Server address, and it could input the IP or Domain Name form.
HTTP Server	Input the HTTP Server address, and it could input the IP or Domain Name form.
HTTP File Path	Set up the file path.
FTP Server	Input the FTP Server address, and it could input the IP or Domain Name form.
FTP Username	The login username.
FTP Password	The login password
FTP File Path	Set up the file path.

Check new firmware	The device will according to the below ways to check the new firmware. <ul style="list-style-type: none"> - Power On (+ Scheduling): The machine will check the new firmware when power on and following the scheduling date and time. - Scheduling: The machine will follow the scheduling date and time to check the new firmware.
Scheduling (Date)	The machine will check the new firmware between the time range by random.
Automatic Update	There are Notify only and Automatic ways to update. <ul style="list-style-type: none"> - Notify only: If there are new firmware, the ATA will send the “Be Be Be” sounds when pick up the handset to prompt there are new firmware. - Automatic: The device will carry firmware update out automatically.
Firmware File Prefix	It will check the information of model name.
Next update time	It will show the next check date and time.

Note:

If the **Check new firmware** field selected to Power On, the machine will check the new firmware according to the scheduling time/date and power on. If there are new firmware can be upgraded, the machine won't carry firmware update out automatic. The machine will send the prompt sounds when pick up the handset, and it needs to update firmware by manual.

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:

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:

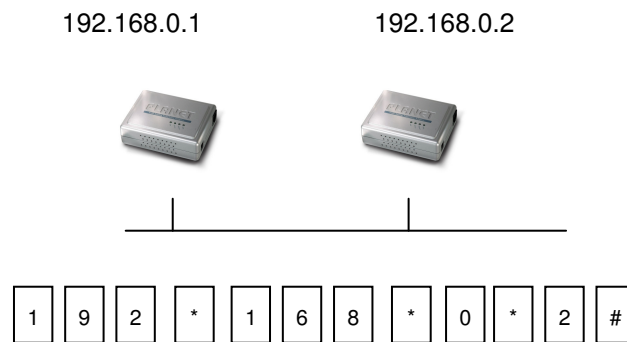
Appendix A Voice Communication Samples

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

Case 1: ATA to ATA connection via IP address

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

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



Test the scenario:

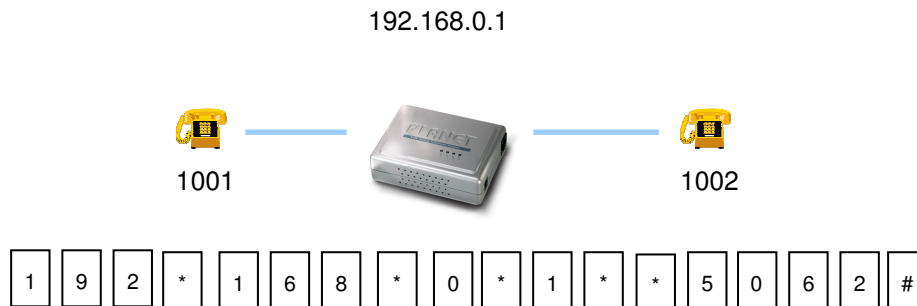
1. Pick up the telephone set on ATA A.
2. Press the keypad: **192*168*0*2#** shall be able to connect to the ATA B.
3. Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

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

Supposing one VIP-157S connects to two telephones, just pick up phone 1 and dial

'192*168*0*1**5062', phone 2 will ring.

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



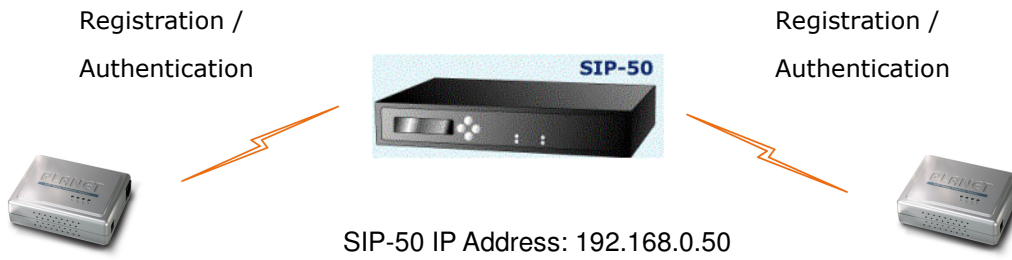
Test the scenario:

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

Hint

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

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



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

Device configurations on the ATA:

STEP 1:

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

STEP 2:

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

Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	1001
Line Number:	1001
Register Name:	planet
Register Password:	●●●
Domain Server:	192.168.0.50
Proxy Server:	192.168.0.50
Outbound Proxy:	
Status:	Registered

STEP 3:

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

Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **2002** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

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

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



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

Device configurations on the IPX-2000:

STEP 1:

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

:: DHCP SERVICE

DHCP POOL
lan

Enable Disable
On-board LAN

Range Single-host
Pool Name: lan
IP: 192.168.1.101 ~ 192.168.1.200
Options: 150,192.168.1.1
Code,Value: 151, http://192.168.0.50/tftpboot/

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

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

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

STEP 2:

Please browse to the **Device → IP Phone** menu and create new device. And press the **EDIT** button for set up the Auto Config configuration.

:: DEVICE PHONE MANAGEMENT

Device ID	Device Administration URL			
<input type="text"/>	<input type="text"/>	<input type="button" value="ADD"/>		
<input type="button" value="DEL"/>				
Device ID	Associated Extension	Device Administration URL	Auto Client Conf	
<input type="checkbox"/> auto_dev_vip156		<input type="text"/> <input type="button" value="LINK"/>	Disabled	<input type="button" value="EDIT"/> <input type="button" value="APPLY"/>
<input type="checkbox"/> auto_dev_vip157		<input type="text"/> <input type="button" value="LINK"/>	Disabled	<input type="button" value="EDIT"/> <input type="button" value="APPLY"/>

STEP 3:

Please fill out the Vendor Prefix code and MAC Address of ATA.

Enable Automatic Client Configuration

Device: auto_dev_vip156

Vendor Prefix: (a-zA-Z0-9_)

MAC Address: : : : : :

Supplementary Configuration:

Codec Preference

1st codec:

1st packet time:

2nd codec:

2nd packet time:

3rd codec:

3rd packet time:

Enable Voice Activity Detection (VAD)

DTMF Mode:

Note:

The following are the Vendor Prefix of devices:

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

STEP 4:

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

:: EXTENSION MANAGEMENT

Extension Number	1001
Associated Device	auto_dev_vip156
Password	•••
User	admin(admin)
Pickup Group	UG_DEF
Line Type	Wired
Language	English
Voicemail	Enable
Voicemail PIN	•••
Unavailable Timeout	10 sec.
<input type="checkbox"/> Allow LAN Use Only	
Try Peer-to-peer RTP	NO
DTMF Mode	inband
	<input type="checkbox"/> Advanced Settings
	<input type="button" value="ADD"/> <input type="button" value="BACK"/>

STEP 5:

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

:: IP PBX SERVICE

Service & Configuration	Advance
IP PBX will reload configuration as soon as possible. Currently active calls will be disconnected in 3 minutes. Do you really want to Continue?	
IP PBX Configuration Reload	<input type="button" value="RELOAD"/>
IP PBX Configuration Backup	<input type="button" value="BACKUP"/> <input type="checkbox"/> PBX Settings Only
IP PBX Configuration Restore	<input type="button" value="RESTORE"/> <input type="button" value="v"/>

Device configurations on the ATA:

STEP 6:

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

Auto Configuration Settings

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

Auto Configuration: Off TFTP FTP HTTP IP-PBX

TFTP Server:	<input type="text"/>	
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/
FTP Server:	<input type="text"/>	Exp. 60.35.17.1
FTP Username:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. /file/load

Submit

Reset

STEP 7:

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

STEP 8:

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

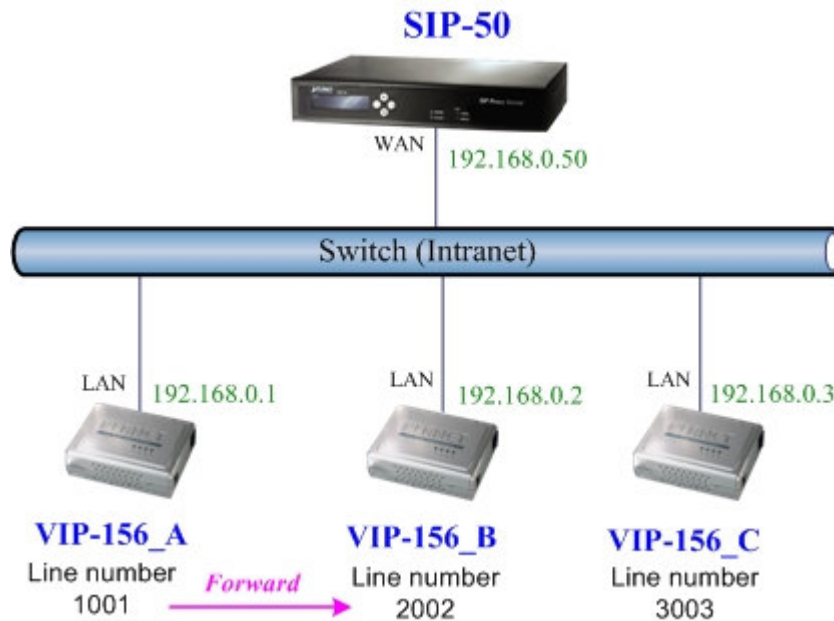
Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **2002** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

Case 5: Call Forward Feature_Example 1

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

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



Machine configuration on the VIP-156:

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

Forward Setting

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

All Forward:	<input type="radio"/> Off <input checked="" type="radio"/> On
Busy Forward:	<input checked="" type="radio"/> Off <input type="radio"/> On
No Answer Forward:	<input checked="" type="radio"/> Off <input type="radio"/> On

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

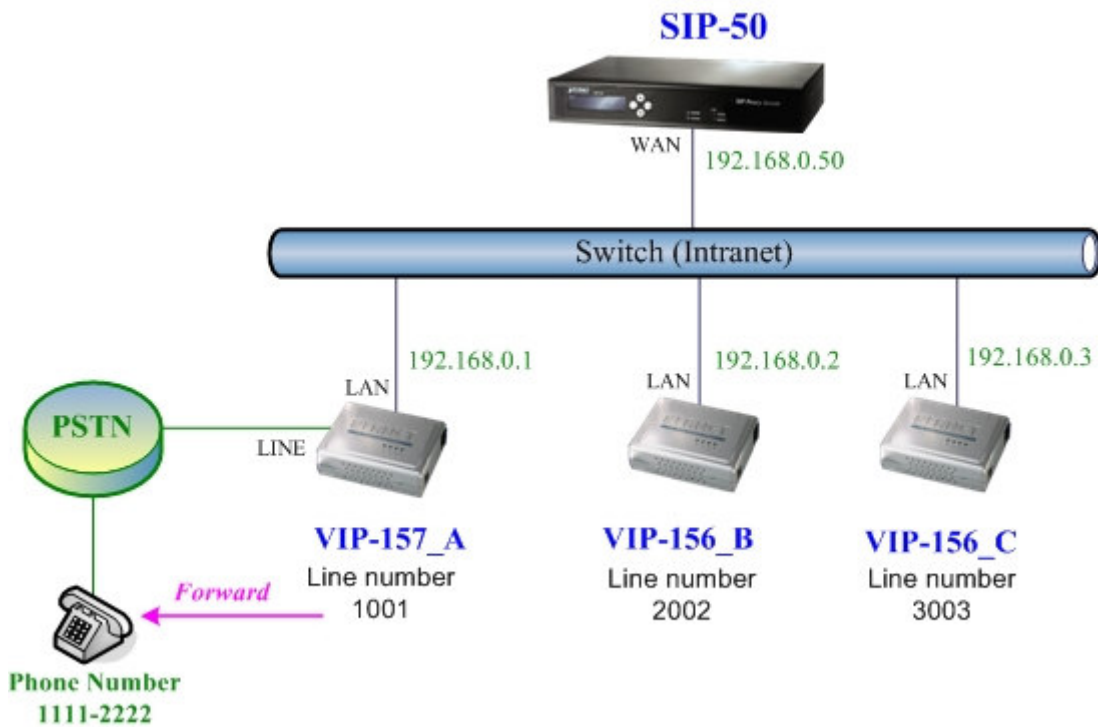
No Answer Fwd Time Out:	3 (2~8 Ring)
-------------------------	--------------

Test the scenario:

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

Case 6: Call Forward Feature_Example 2

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



Machine configuration on the VIP-157:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **PSTN** choice and fill in the **Name** and **URL/Number** of PSTN Phone Number 11112222, then the sample configuration screen is shown below:

Forward Setting

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

All Forward:	<input type="radio"/> Off	<input type="radio"/> IP	<input checked="" type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text" value="PSTN"/>	<input type="text" value="11112222"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

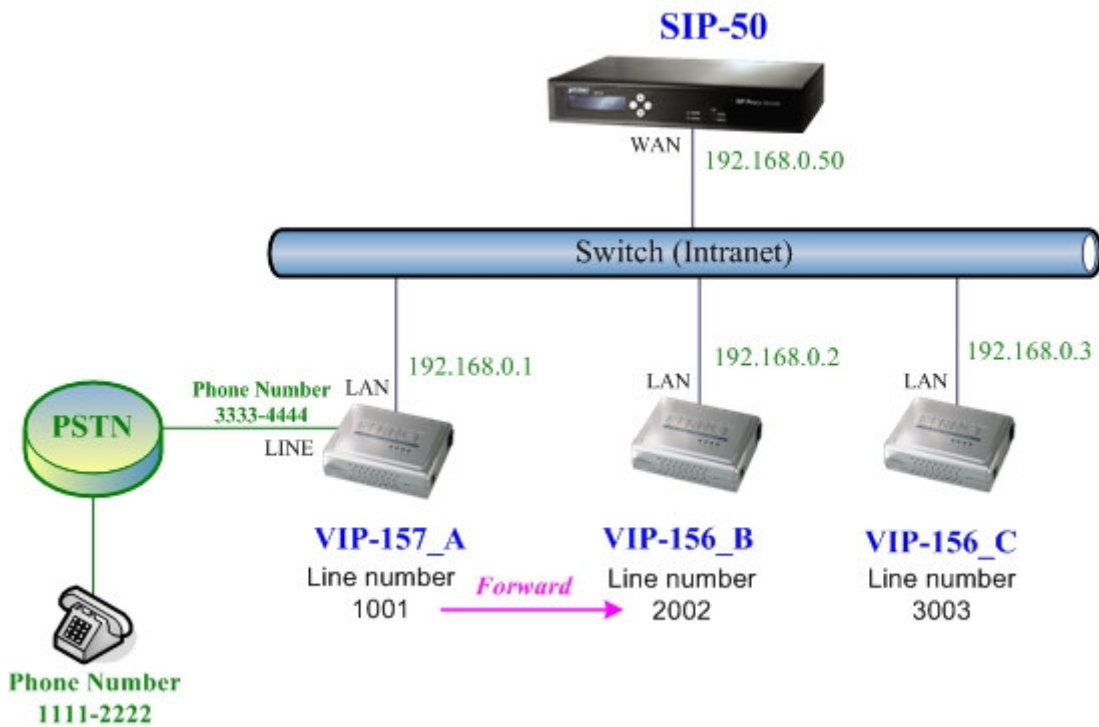
No Answer Fwd Time Out:	<input type="text" value="3"/>	(2~8 Ring)
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Test the scenario:

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

Case 7: Call Forward Feature_Example 3

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



Machine configuration on the VIP-157:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **IP** choice and fill in the **Name** and **URL/Number** of of VIP-156_B, and then the sample configuration screen is shown below:

Forward Setting

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

All Forward:	<input type="radio"/> Off <input checked="" type="radio"/> IP <input type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off <input type="radio"/> IP
No Answer Forward:	<input checked="" type="radio"/> Off <input type="radio"/> IP <input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text" value="VIP-156_B"/>	<input type="text" value="2002"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out:	<input type="text" value="3"/> (2~8 Ring)
-------------------------	---

Test the scenario:

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

Case 8: Call Forward Feature_Example 4

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



Machine configuration on the VIP-156:

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

Forward Setting

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

All Forward:	<input type="radio"/> Off	<input checked="" type="radio"/> On
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> On
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> On

	Name	URL
All Fwd No.:	VIP-156_B	192.168.0.2
Busy Fwd No.:		
No Answer Fwd No.:		

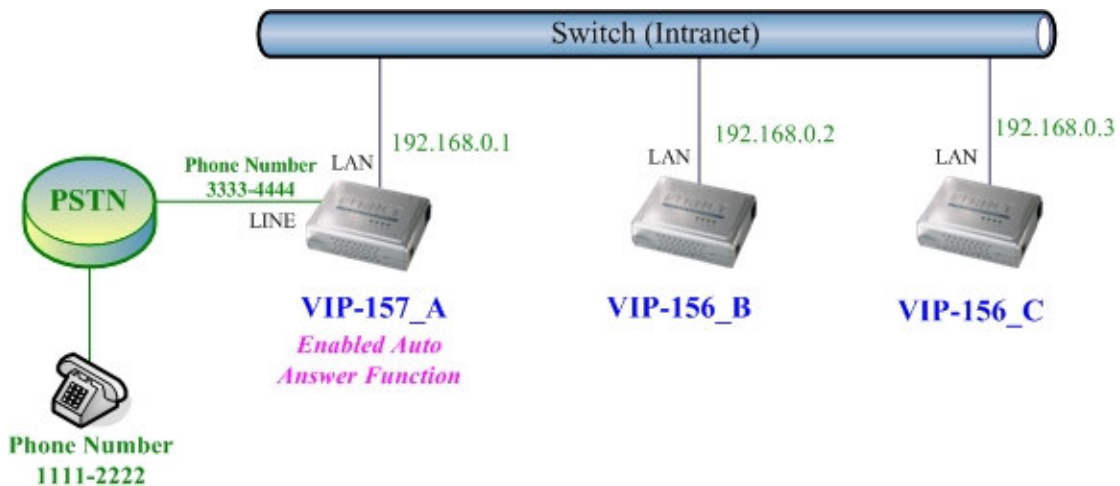
No Answer Fwd Time Out: (2~8 Ring)

Test the scenario:

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

Case 9: Auto Answer Feature_IP to PSTN

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



Machine configuration on the VIP-157:

STEP 1:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please disable All **Forward** function, and then the sample configuration screen is shown below:

Forward Setting

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

All Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out:	<input type="text" value="3"/>	(2~8 Ring)
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STEP 2:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select

the **Auto Answer** config menu. In the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, then the sample configuration screen is shown below:

Auto Answer

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

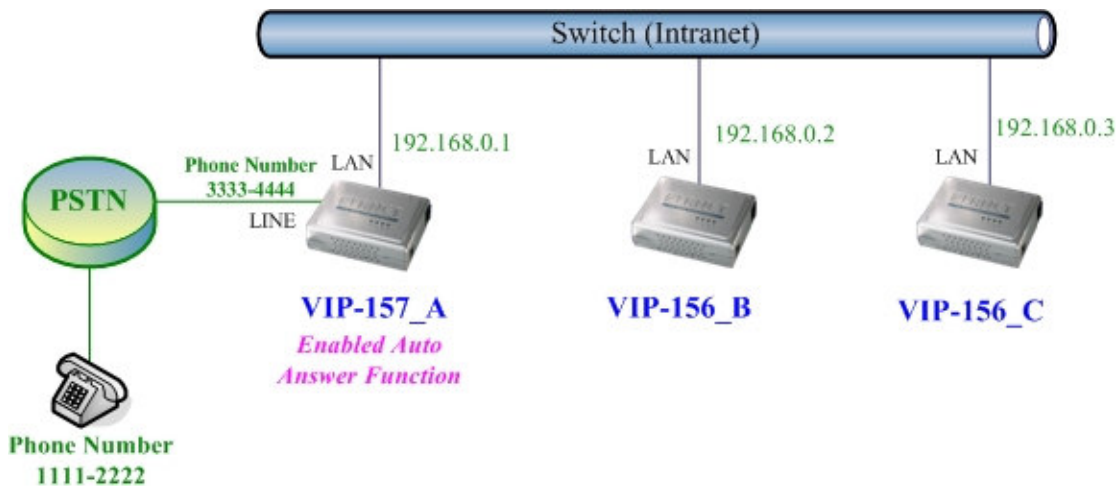
Auto Answer:	<input checked="" type="radio"/> On <input type="radio"/> Off
Auto Answer Counter:	<input type="text" value="03"/> (2~15)
PIN Code Enabled:	<input checked="" type="radio"/> On <input type="radio"/> Off
PIN Code:	<input type="text" value="123"/>

Test the scenario:

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

Case 10: Auto Answer Feature_PSTN to IP

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



Machine configuration on the VIP-157:

STEP 1:

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

Auto Answer

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

Auto Answer:	<input checked="" type="radio"/> On <input type="radio"/> Off
Auto Answer Counter:	<input type="text" value="03"/> (2~15)
PIN Code Enabled:	<input checked="" type="radio"/> On <input type="radio"/> Off
PIN Code:	<input type="text" value="123"/>

STEP 2:

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

Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	URL	Select
0	VIP-156_B	192.168.0.2	<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Test the scenario:

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

Appendix B The method of operation guide

In this section, we'll introduce the features method of operation, and lead you step by step to establish these features.

Call Transfer

A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out (Press “**transfer**” button) to hold the conversation with B.
3. A press “**#510#**” and hear the dial tone, then input the number of C (Follow 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 conversation with B.

B. Attendant Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out to hold the conversation with B.
3. A press “**#511#**” and hear the dial tone, then input the number of C (Follow by the “**#**” key).
4. C will ring up.
5. C picks up the handset and conversation with A.
6. A hang up and C conversation with B.

3-Way Conference

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

Call Waiting

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

Switch the Realm (Registration Proxy Server)

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 any one SIP accounts for making calls through input

the switch code.

Realm switch code:

1*: Realm 1

2*: Realm 2

3*: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

Auto Update firmware by manual (Keypad)

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

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

Product	SIP Analog Telephone Adapter			
Model	VIP-156	VIP-156PE	VIP-157	VIP-157S
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			
Fax support	T.38			
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer			
Protocols	SIP 2.0 (RFC-3261), TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE			
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			