User's Manual

VoIP Analog Telephone Adapter

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

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CE mark Warning

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

WEEE Warning

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

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Revision

User’s Manual for PLANET VoIP Analog Telephone Adapter:

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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
- IPV6 Support
- TR-069

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.

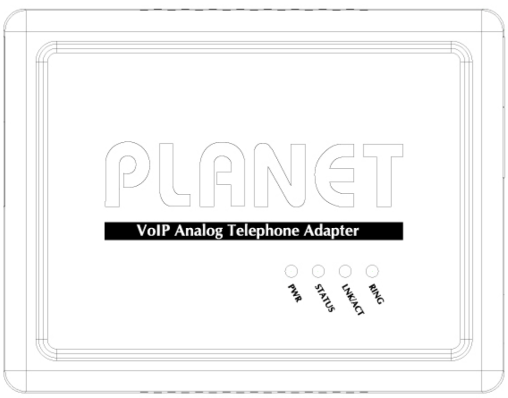
Respective model/descriptions are shown below:

VIP-156: SIP Analog Telephone Adapter

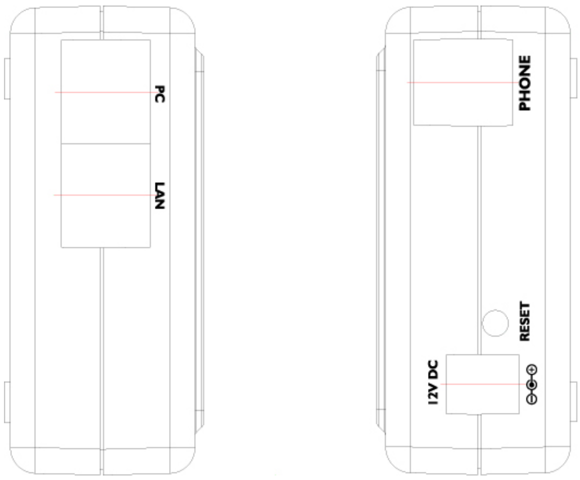
VIP-156PE: 802.3af PoE SIP Analog Telephone Adapter

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

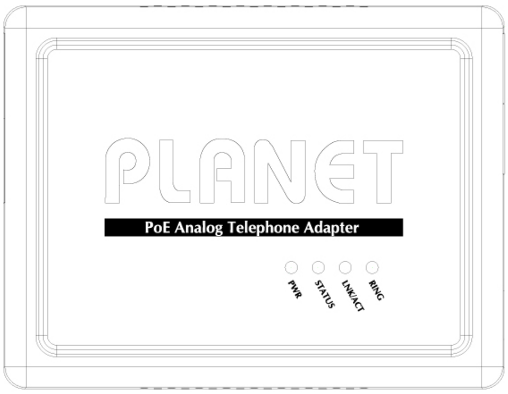
VIP-157S: 2-port FXS SIP Analog Telephone Adapter



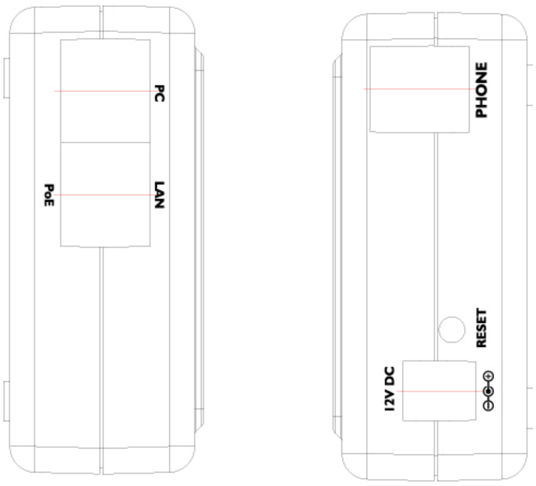
Front Panel of VIP-156



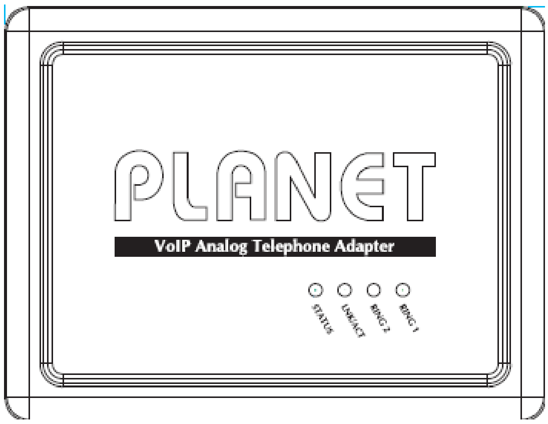
Left / Right Panel of VIP-156



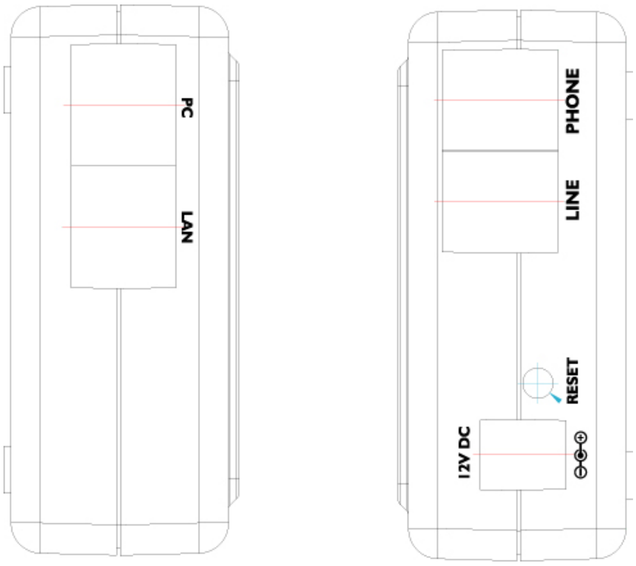
Front Panel of VIP-156PE



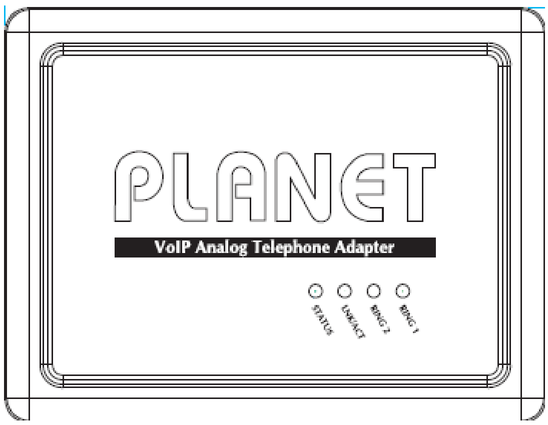
Left / Right Panel of VIP-156PE



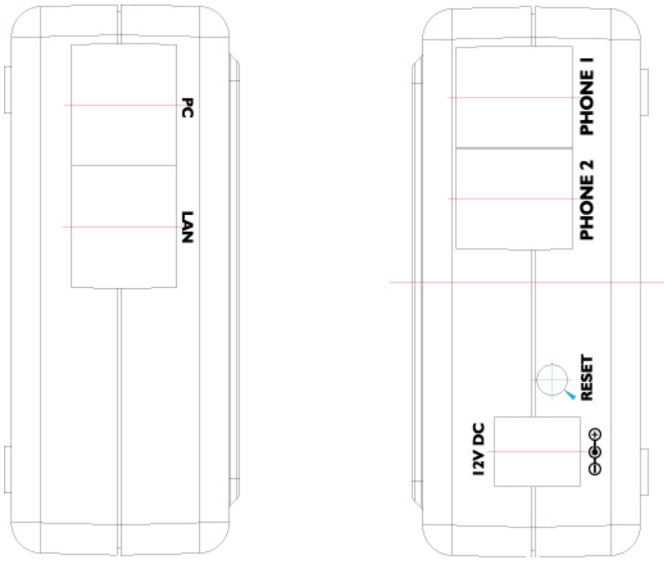
Front Panel of VIP-157



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.

Chapter 2

Preparations & Installation

Physical Installation Requirement

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

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

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

Administration Interface

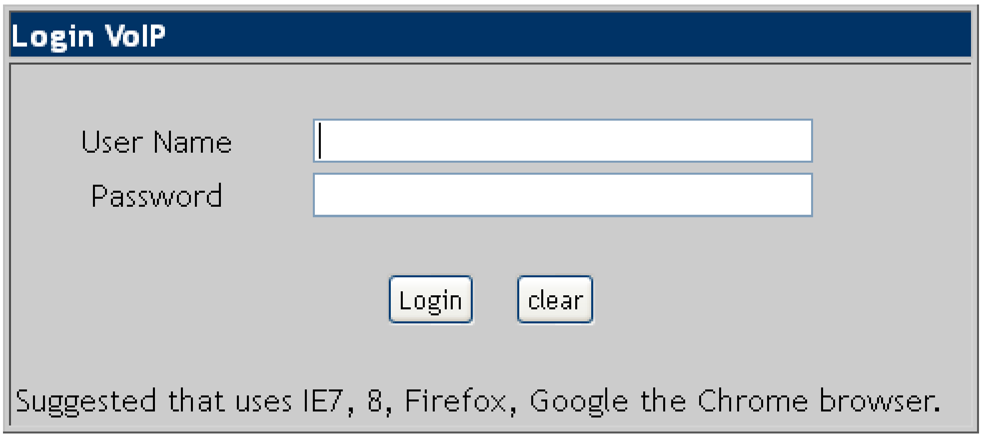
PLANET ATA provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access

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

- Microsoft Internet Explorer 6.0.0 or higher with Java support

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



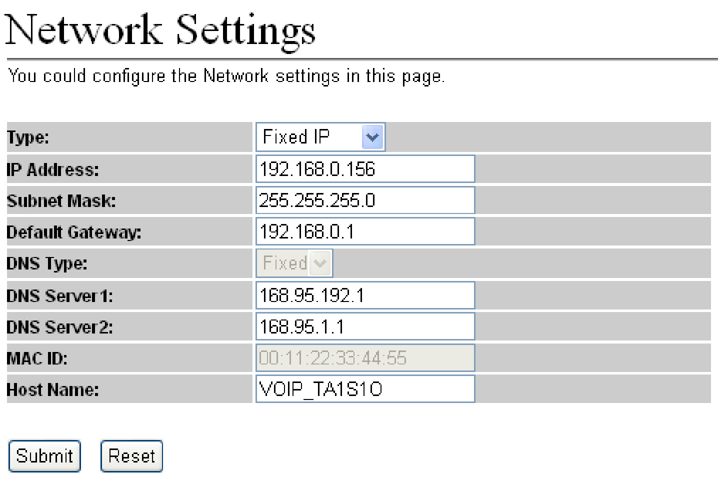
ATA will prompt for logon user name/password, please enter: **root** / **null (no password)** to continue machine administration.

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 password**), browse to "**Network**" --> "**Network Settings**" configuration menu:



Parameter Description

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

Network settings via Keypad commands

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

Hint

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

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
#111#	Set DHCP client	None	ATA will change to DHCP Client
#112xxx*xxx*xxx*xxx#	Setup Static IP Address	Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
#113xxx*xxx*xxx*xxx#	Set Network Mask	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#114xxx*xxx*xxx*xxx#	Set Gateway IP Address	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#115xxx*xxx*xxx*xxx#	Set Primary DNS Server	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#190#	Unlock	None	Must unlock the protect function before 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 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 in enabled or disabled.
#122#	Check the Phone Number	IVR will announce current enabled VoIP number.
#123#	Check Network Mask	IVR will announce the current network mask

		of the ATA.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP address of the ATA.
#125#	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
#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.

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 forwrad funciton	
#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 to "**Save & Reboot**" menu to reboot the machine to make the settings effective.

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

ⓘ Hint

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

Save Modification to Flash Memory

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

Chapter 3

Network Service Configurations

Configuring and monitoring your 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

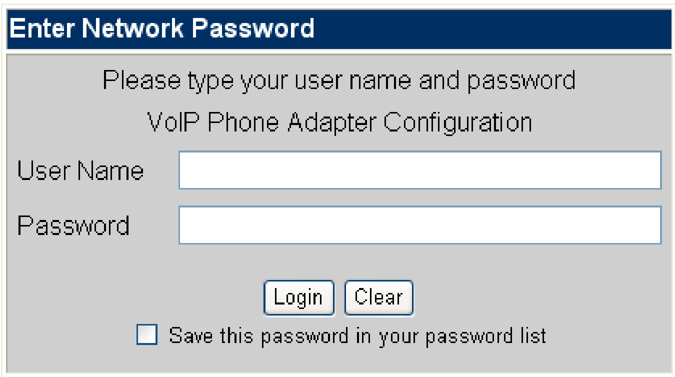
- Microsoft Internet Explorer 6.0.0 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

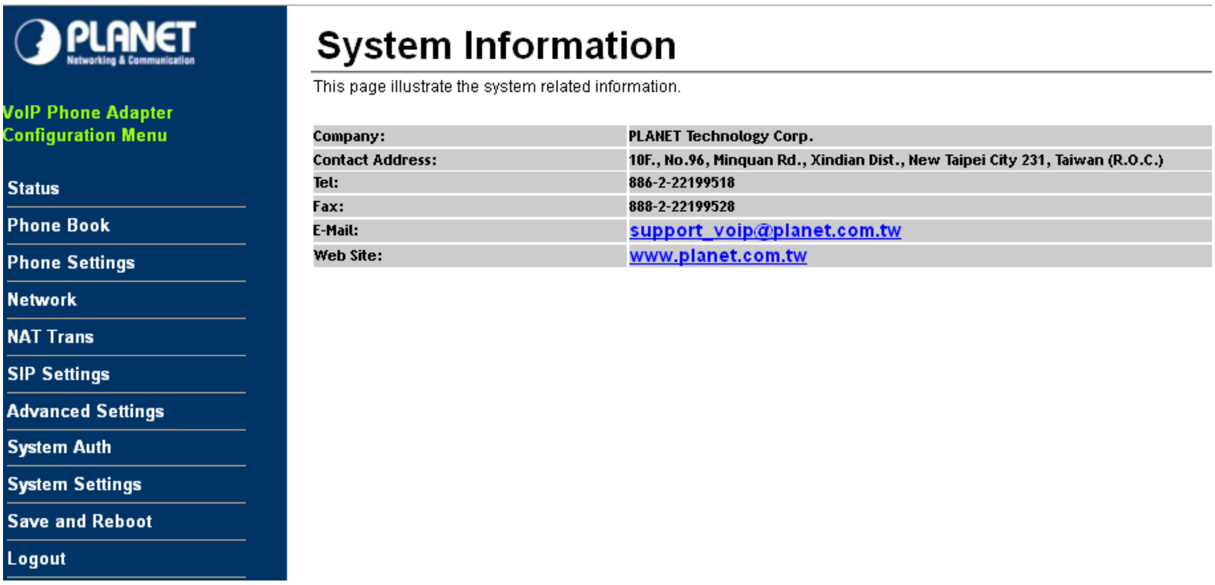
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon Phone Adapter web configuration page.

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



ATA log in page

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



VoIP Phone Adatper main page

Chapter 4

VoIP Telephone Adapter Configurations

Status

Show all the system information, ex: WAN/LAN IP address, System information, IPV6 connection information, Register status and VPN connection message. (After you setup the VPN line then the status will start to show out)

Status Information

You could see the information of the VOIP machine.

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

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

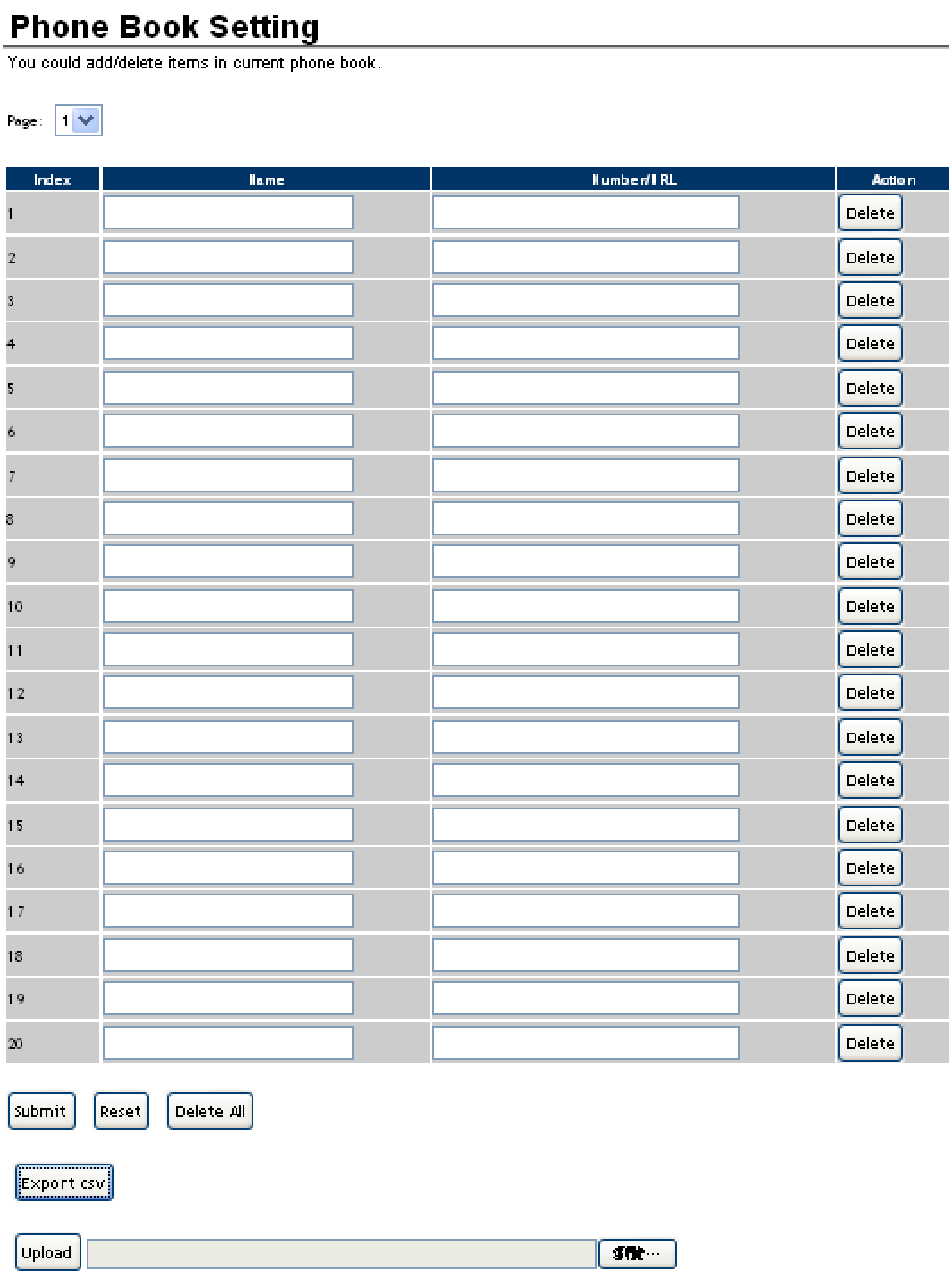
System Information			
Firmware Version:	1.1.0-1201180	Update Date:	2005-01-01
DSP Version	NV-1106080		
System Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 55 minute(s)		
Current Time:	2012-02-20 15:54		

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

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

Phone Book

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



Field	Description
Phone Book Page	The default is Page 1. It can select Page1 ~ Page 7 to look round Phone Book records.
Phone	The record number from 1 ~ 140, it can set up 140

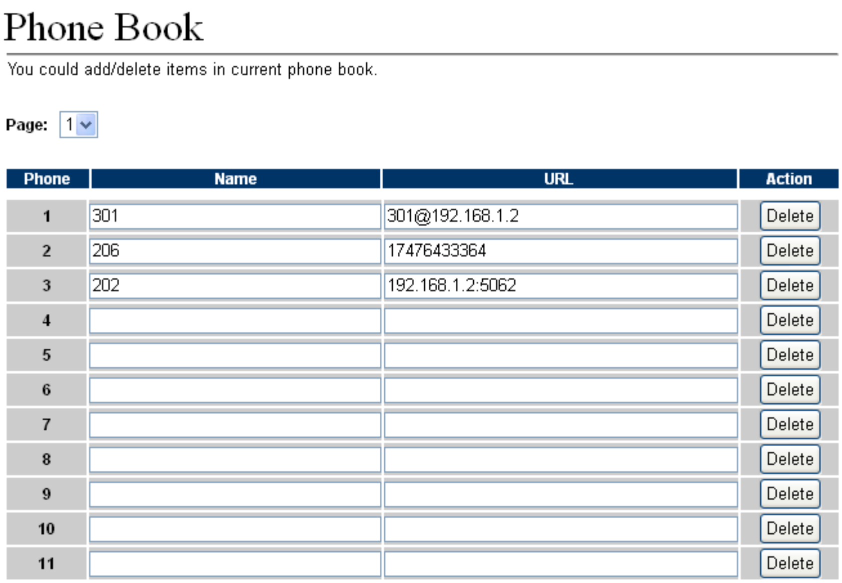
	records in total.
Name	The name of Phone Book records, it only can input numerals character.
URL	Fill in the outgoing number (Line Number) or IP address.
Delete	Clean this item's data.
Export csv	Save the phone book data as CSV file.
Upload	Upload the phone book file

If you need to add a phone number 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 "Submit" button.

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

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

For Example:



Ex_1:

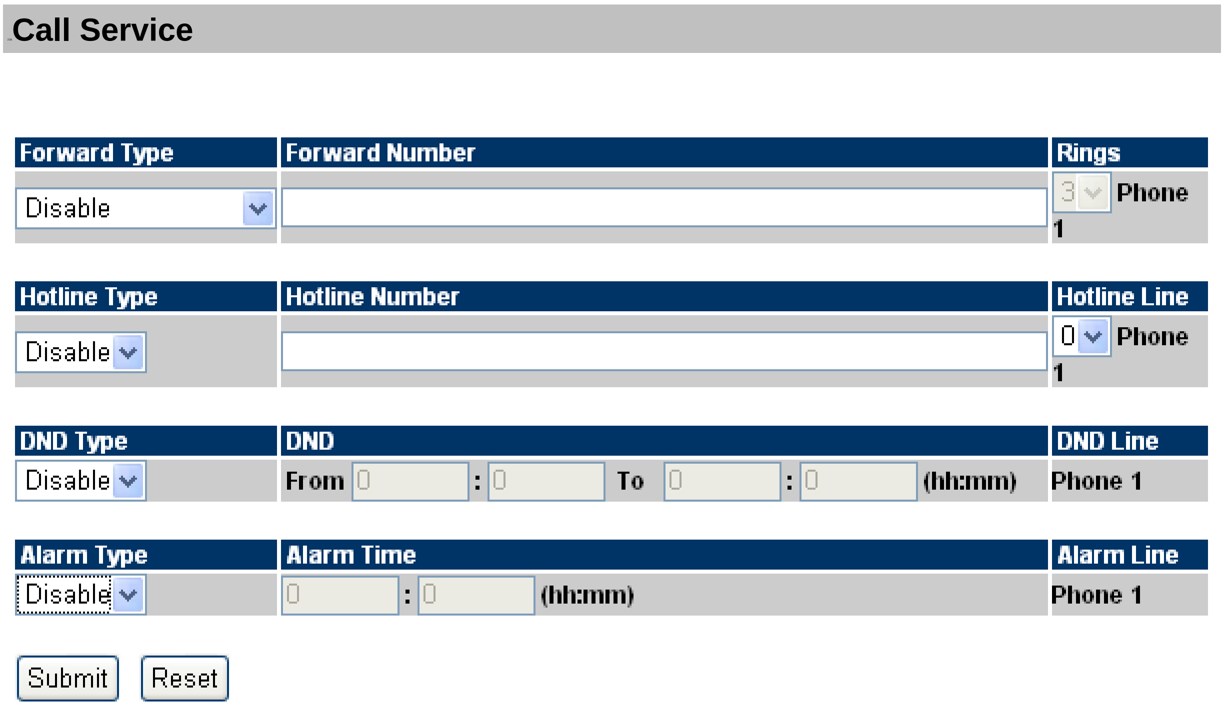
ATA had added the above phone numbers. User pick up the handset and dial the "**301**" to make the P2P call (301@192.168.1.2).

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]

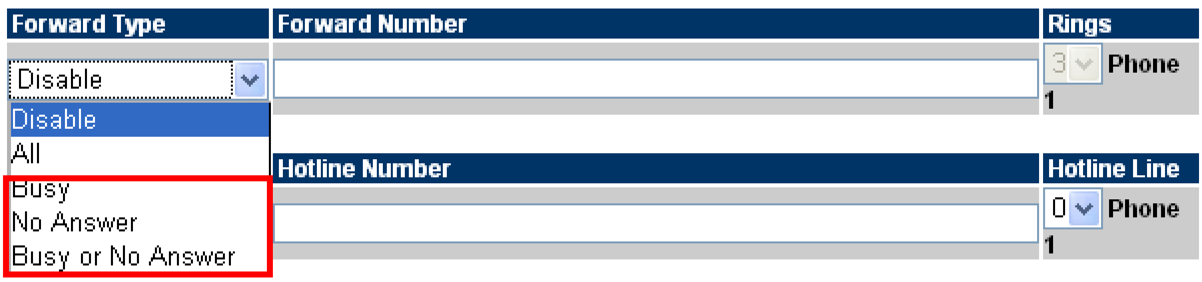
This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

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.

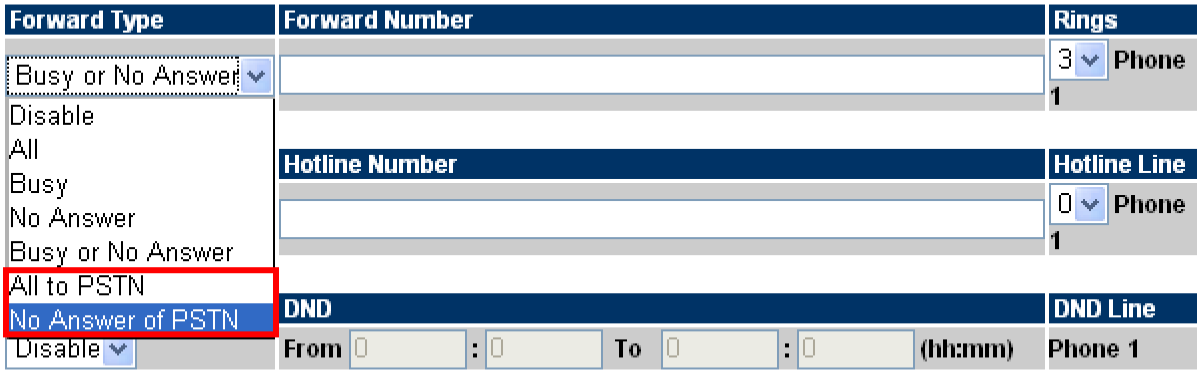
No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

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



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

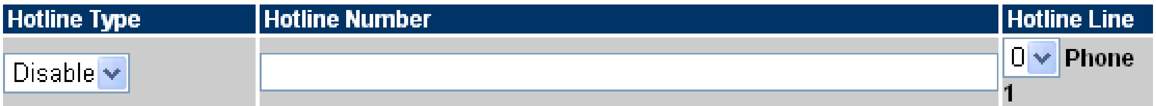
All to PSTN/ No Answer to PSTN (VIP-157): VIP-157 not only support Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, then input the name and the PSTN number in URL/Number field.



IP Line Forward function for VIP-157

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

[Hotline Type]



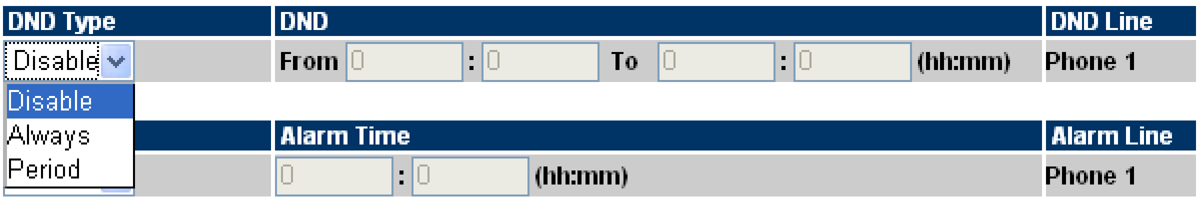
This page defines the Hot line setting in this page. When user pick up the handset, the device will call to the specific number automatically.

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

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

Delay time: After pick up the phone takes how long if not press any digital will start hot line

[DND Type]



This page defines the DND 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.

[Alarm Type]

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.

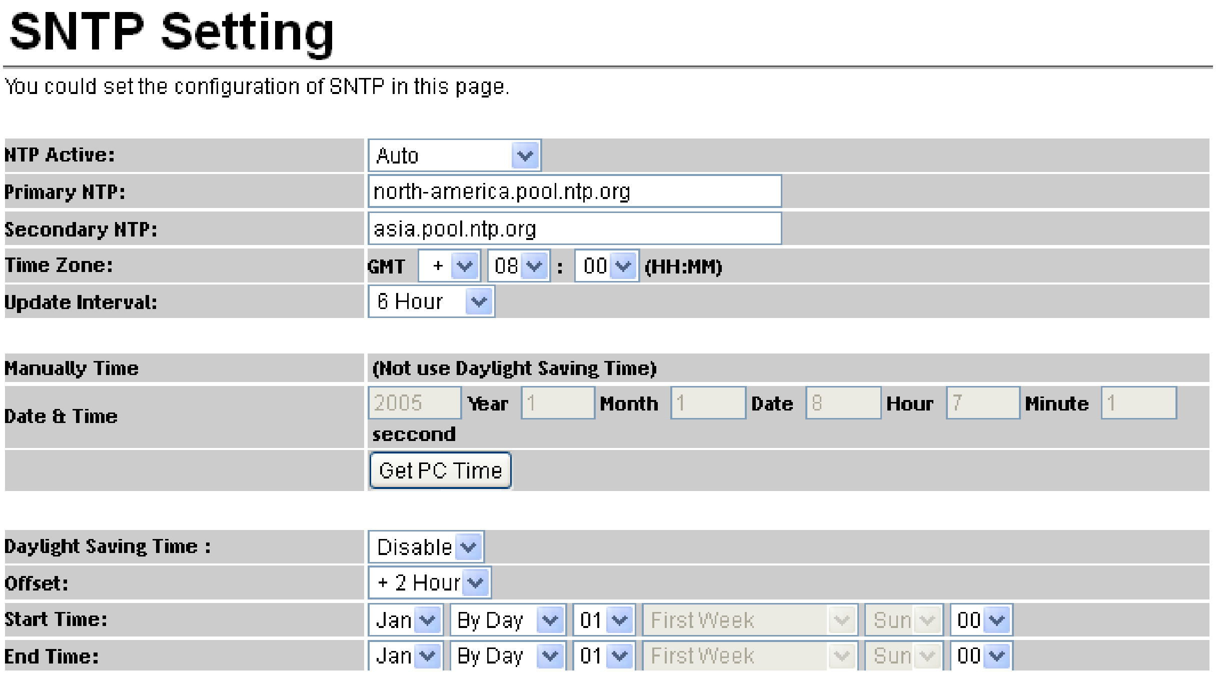
Alarm Type: The default is Off. If set up as On, the telephone will ringed up at the specific time.

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

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

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. User can also use the “daylight saving” to adjust the daylight time. When you finished the setting, please click the Submit button.

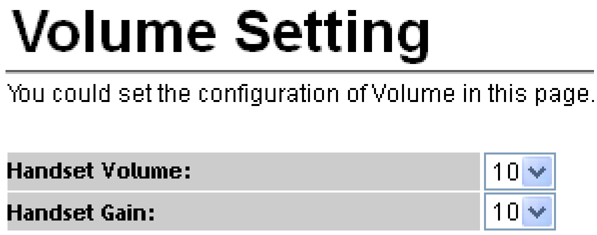


Volume Setting

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

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

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

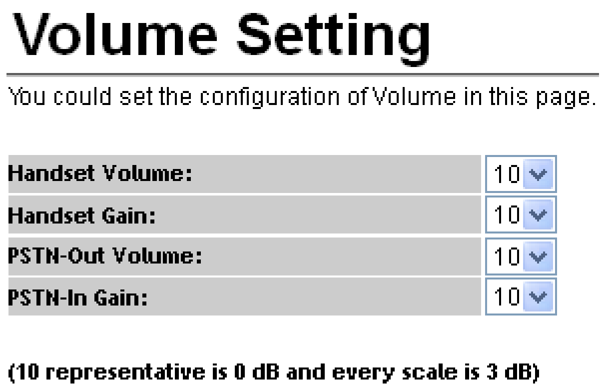


Volume Settings for VIP-156T/VIP-156PT

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

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

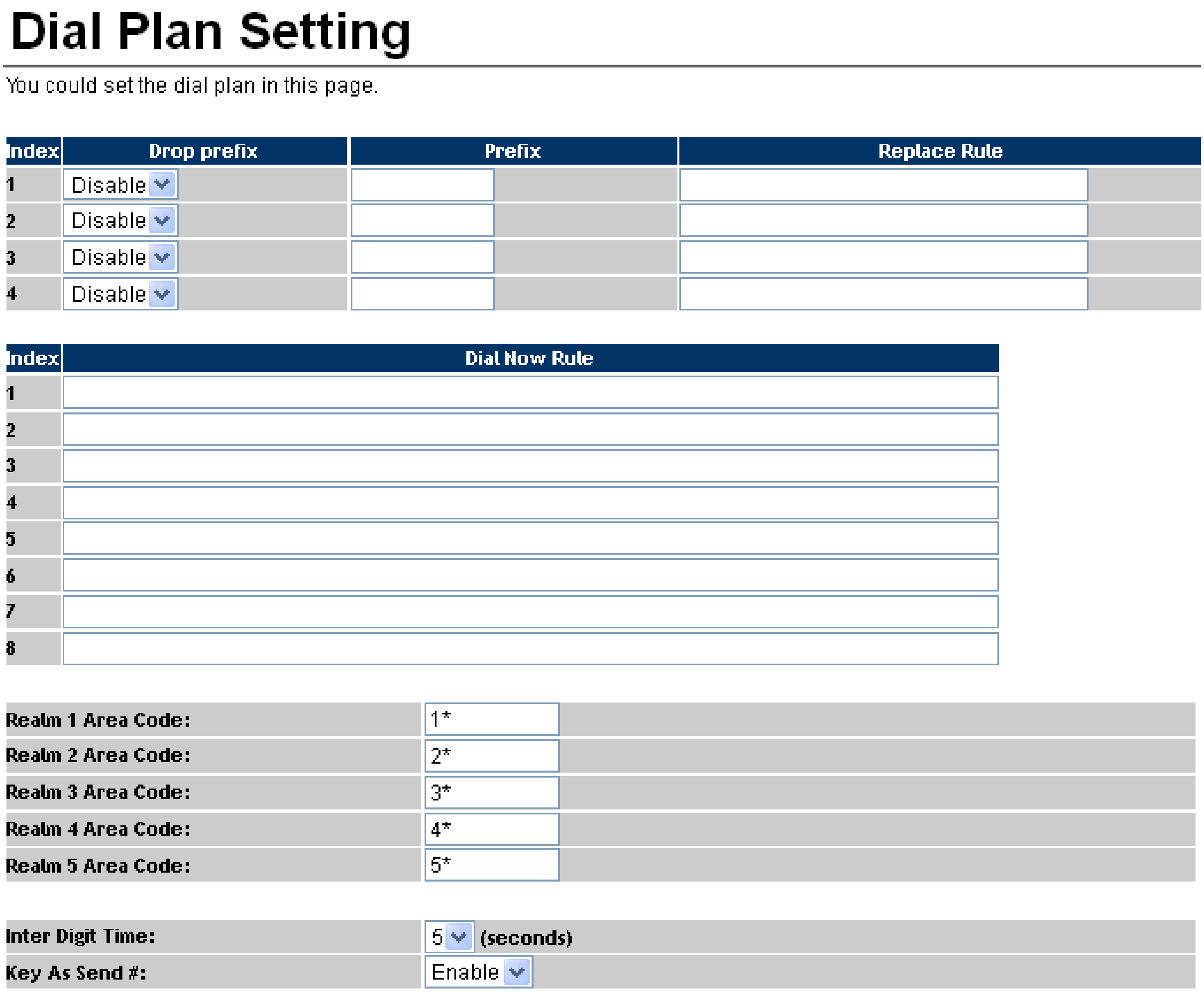
PSTN-In Gain is to set the volume send out to the other side's handset.



Volume Settings for VIP-157

Dail Plan Setting

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



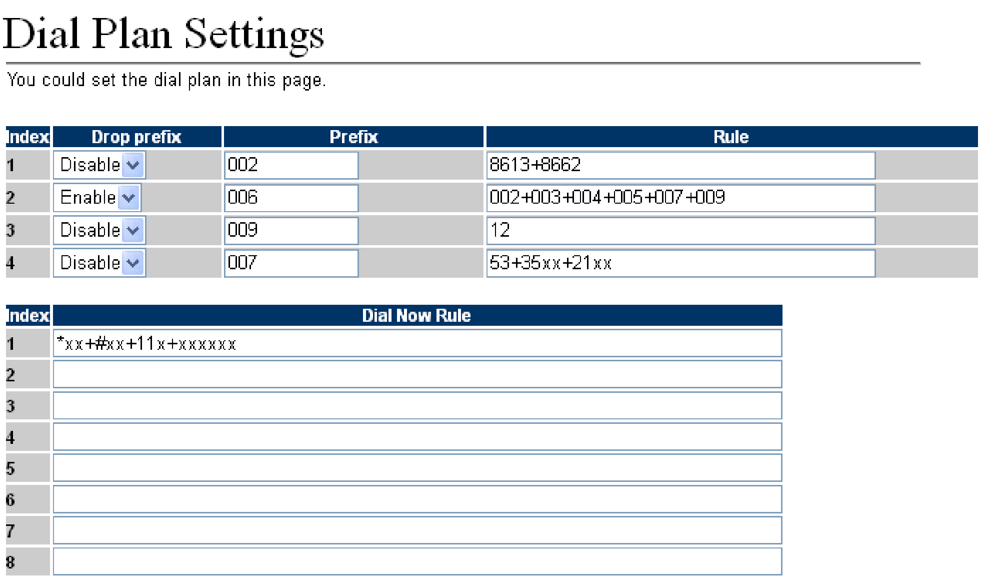
Dial Plan Settings for VIP-156

For VIP-157 have four more items.

Field	Description
Drop Prefix	The rule of add or replace code. If setup as Disable, it will add the prefix number prior to the identification number. If setup as Enable, it will replace the identification number.
Prefix	The prefix number. It only accept 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 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 rule	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.
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)	Stop dialing after seconds then send dial number out.
Key as send #	If setup as Yes, the system sill stop to receive the dialing number when receive the [#] key. The system also will to 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.

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 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**.

[For VIP-157 only]

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

Field	Description
Auto PSTN backup	Deafult is Disable, it's for PSTN backup function, when it "Enable", if SIP regsiter un-successful it will automatically switch to PSTN line to dial out. ⓘ**Note:** To enable this function, make sure that the PSTN line is connected to the PSTN port already.
PSTN feature Code	Deafult is 0*, the code for **<u>manually</u>** switch to PSTN line, and dial out from PSTN, it can only accept the numeric and *or #, the digital max length is 7.

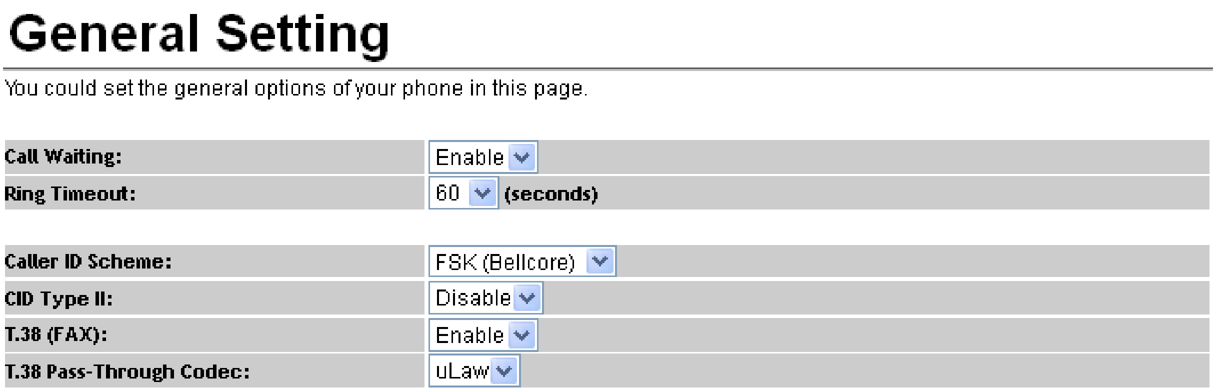
Routing Type	Deafult is Disable, it define the dialing route, according the [Routing Rule] to define the dialing route is [IP or FXO].
Routing Rule	Define the outgong rule. It can also Add/ Drop profix number, if you want to increase more then one Routing rule, you can use "+" to except it can only press the numeric or D D: drop

Example_5: Routing: Routing Type: **FXO**, Routing Rule: **D007+009+0800**

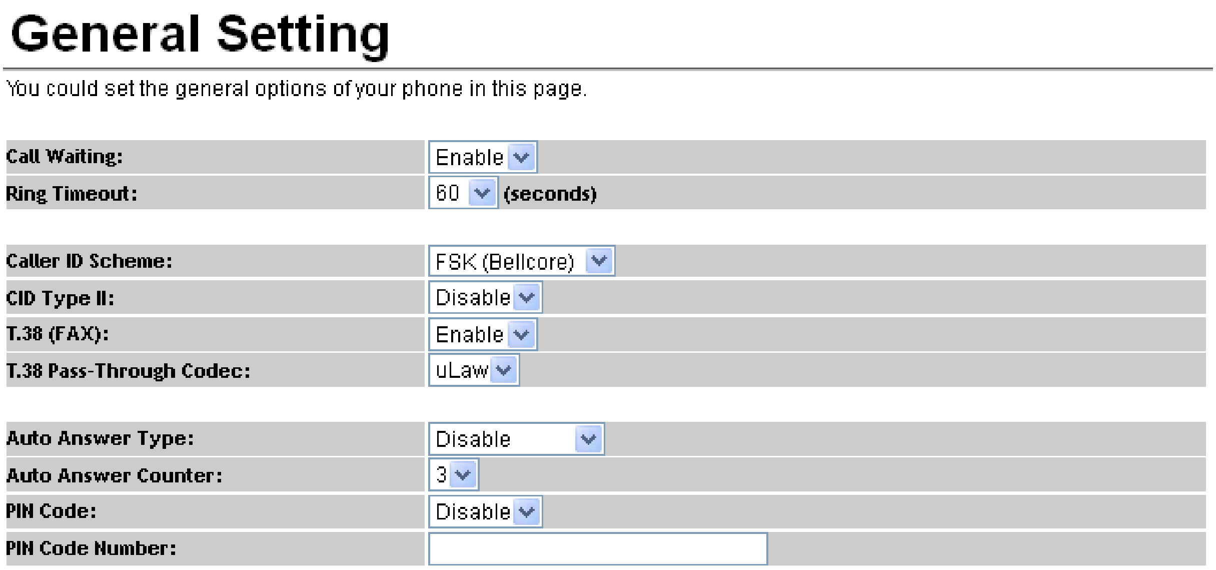
1. If the dialing number is "**0800024365**", it will match the routing rul e [0800], then system will automatically dial out from the [**FXO**].
2. If the dialing number is "**00986123456**", it will match the routing rule [009], the n system will automatically dial out from the [**FXO**].
3. If the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

General

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



VIP-156



VIP-157

Field	Description
Call Waiting	Default is enable. When you are talking with other people, You can choose If you want to hear the notice when there is a new 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.
Ring Timeout	Default is 60(sec). After how long the system will reply the busy(486 busy) message.
Caller ID Scheme	Set thecaller ID mode, it support FSK Bellcore, DTFM, CID-Japnan, DTMF-Brazil, DTMF-Denmark. FSK Bellcore: FSK caller ID mode. DTMF: Before first ring it will send the DTMF caller ID data. CID-Japnan: Janpan (Janpan) 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 enable this function system receive a new call in call waiting, it will display the caller ID ⓘ**Note:** Your Phone must also support CID Type2.
T.38 (FAX)	Enable/Disable T.38 FAX function.
T.38 Pass-trough codec	Define the T.38 pass through codec, it can support G.711 u-law/G.711 a-law.

Auto Answer and PIN (VIP-157)

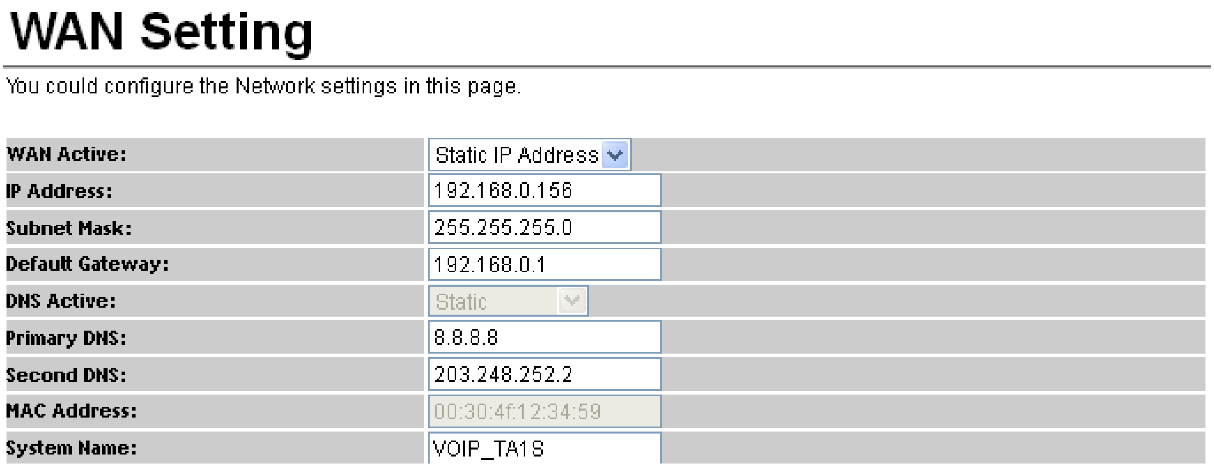
Field	Description
Auto Answer Type	Auto Answer: There are different incoming call types for flexable applications. The Trunk Gateway function needs to arrange in with the registered Server System. The 3-Party subscribers could make Off-Net call (PSTN) through the FXO port of VIP-157.
AutoAnswer cournter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

Chapter 5

Netowrk

Network Settings

This page defines the LAN setting in this page.



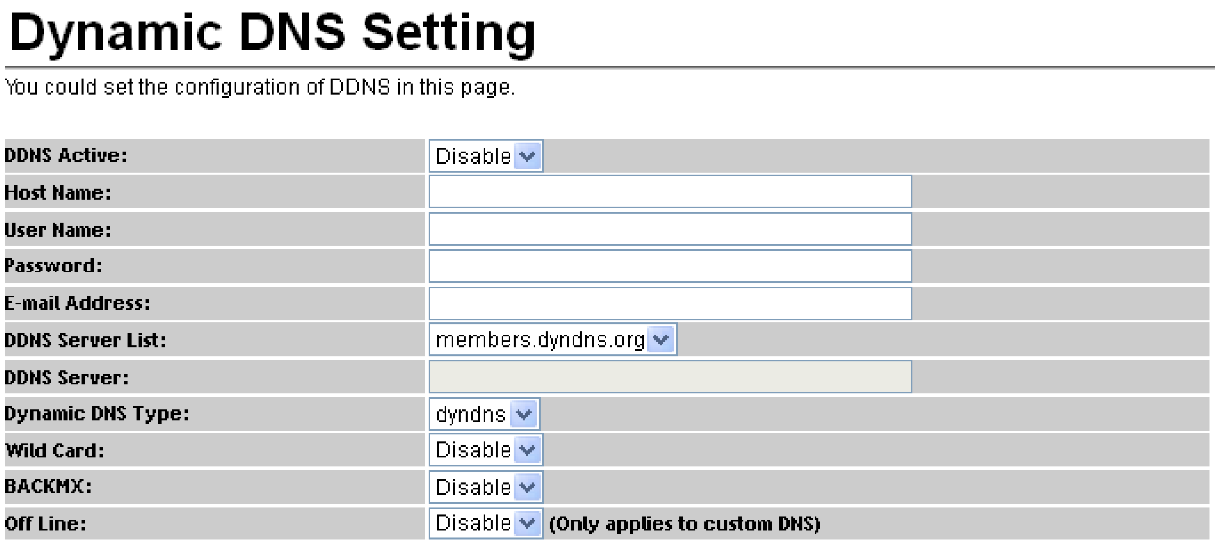
Field	Description
WAN Active	The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes. **Fixed IP:** It could setup the IP address manual. **DHCP Client:** It will acquire the IP address automatically. **PPPoE:** It will use the PPPoE connection method
IP Address	The IP address
Subnet Mask	The sub net address
Default Gateway	The default gateway address
DNS Active	Static/ Automatically, manually setup the DNS server or automatically accept the DNS server.
Primary DNS	The default is 168.95.192.1, it could setup the first DNS server address.
Second DNS	The default is 168.95.1.1, it could setup 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 inpout numeral or character, the maximum date length are 63.
PPPoE Password	The PPPoE connection account password. It could inpout numeral or character, the maximum date length are 63.
PPPoEService name	PPPoE Service provider name

PPPoE AC Name	PPPoE AC name.

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.

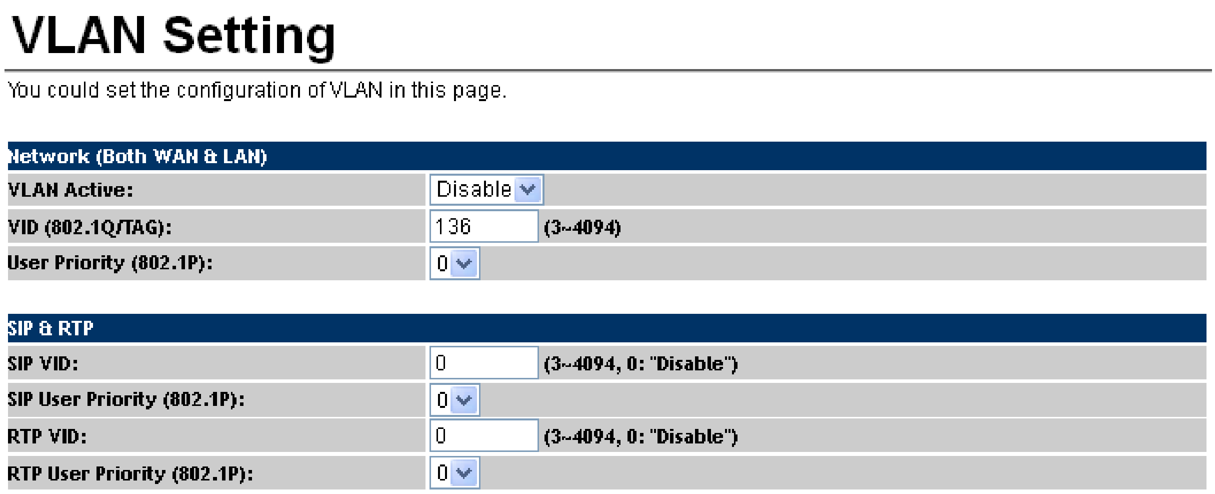
(For better service Planet provide the **Planet DDNS**, you can apply your DDNS account in web site www.planetddns.com)



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

VLAN Settings

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

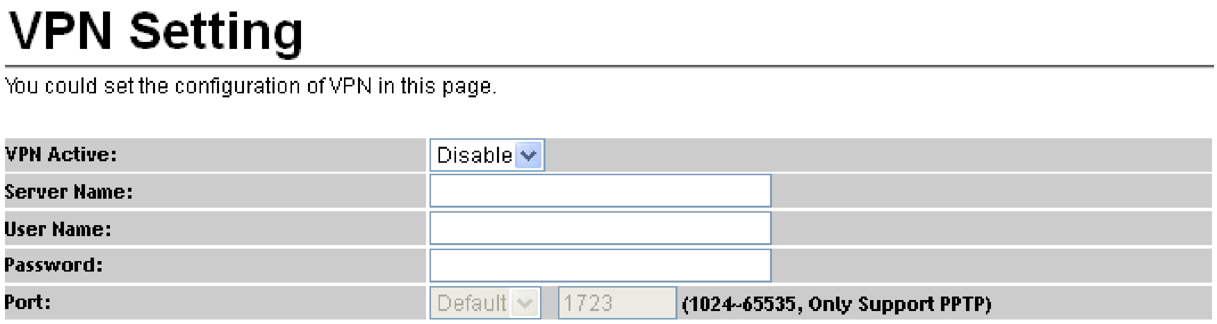


Field	Description
VLAN Active	If setup as On, it could receive VLAN messages.
VID (802.1Q/TAG)	Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 2~4094.
User Priority (802.1P)	To setup the user priority.

Field	Description
SIP VID	Set the SIP VLAN ID, this is the independ en function don't need to enable [VLAN Packets: Enable].
SIP User Priority (802.1P)	Setup the SIP Priority.
RTP VID	Set the SIP RTP VID, this is the inde penden function don't need to enable [VLAN Packets: Enable].
RTP User Priority (802.1P)	Setup the RTP Priority.

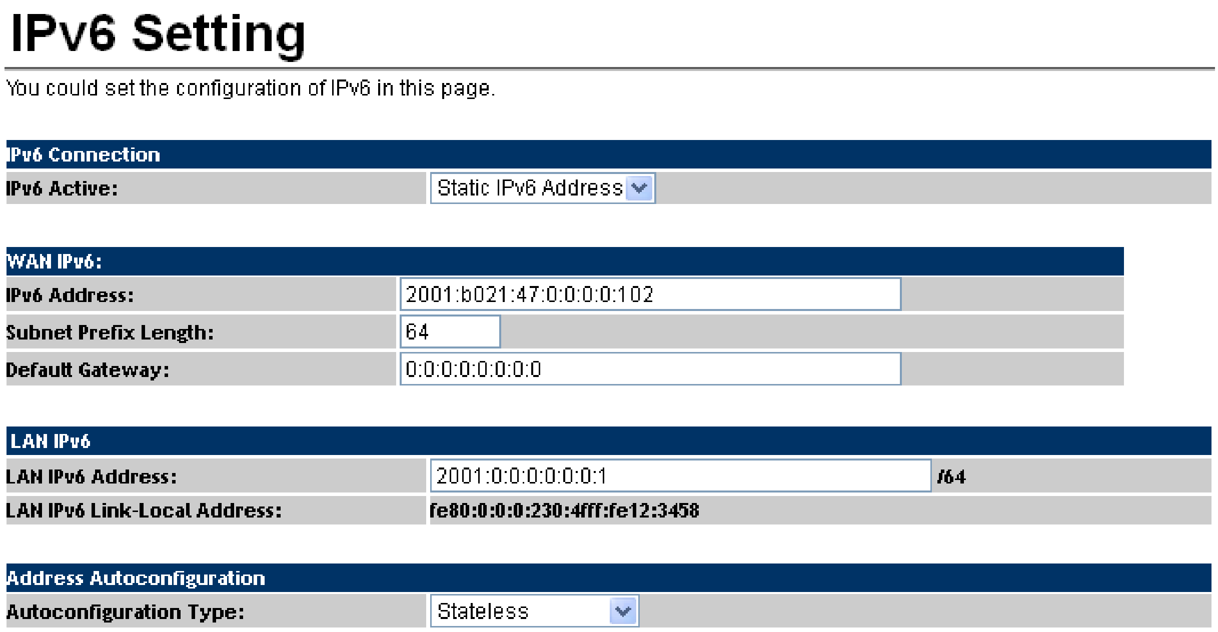
VPN Settings

This page defines the PPTP/L2TP setting in this page. You could setup the PPTP/L2TP Server connection information. When you finished the setting, please click the Submit button.



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

IPV6 Settings



This page defines the IPV6 setting in this page, you can programm the IPV6 information.

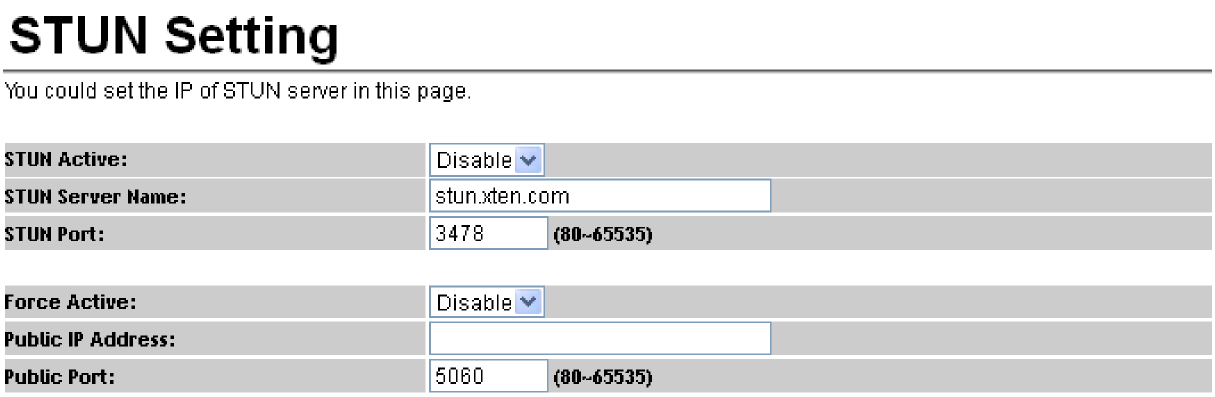
Field	Description
IPV6 Active	Support three IPV6 type: Auto, Fixed IPV6, IPV6 in IPV4 Tunnel
IPV6 address	Setting the WAN IPV6 address or display it.(64 bits)
SubnetPrefix Length	Default is 64, settin the
Default Gateway	IPV6 gateway address(64 bits)
LAN IPv6 Address:	IP V6 LAN address. (64 bits)
LAN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

Chapter 6

NAT Trans

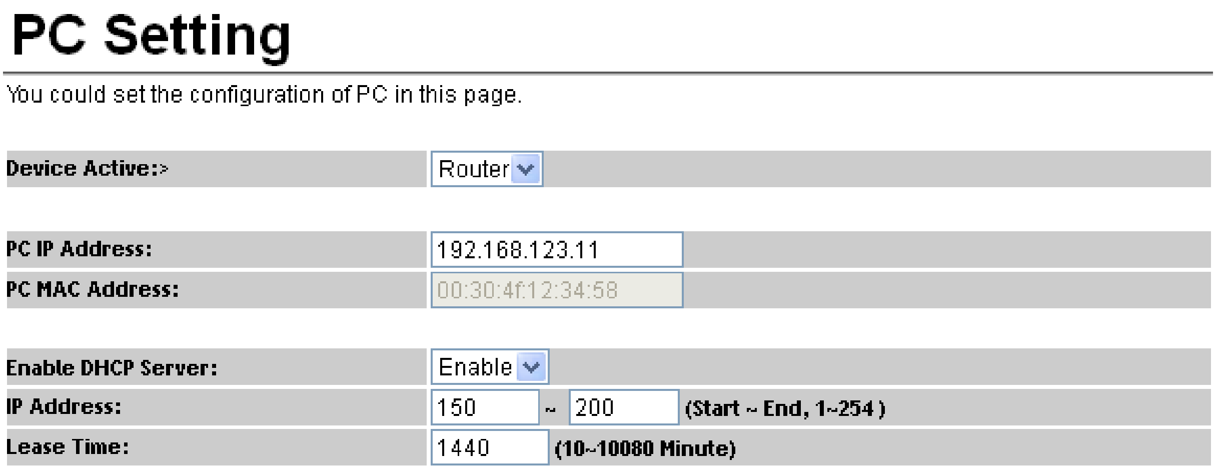
Stun Settings

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.



PC Settings

This page defines the PC setting in this page.



Field	Description
Device Active	The default is Bridge mode, and it also provides NAT mode. **Bridge:** When set as is mode, the LAN and PC ports are in the same network segment. **NAT:** The LAN and PC ports are in the 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**: 192.168.0.1
PC MAC Address	The MAC of PC port
Enable DHCP Server	It will allot the IP address automatically when enabke this function.
IP Address	The range for DHCP IP address.
Lease Time	DHCP server lease time

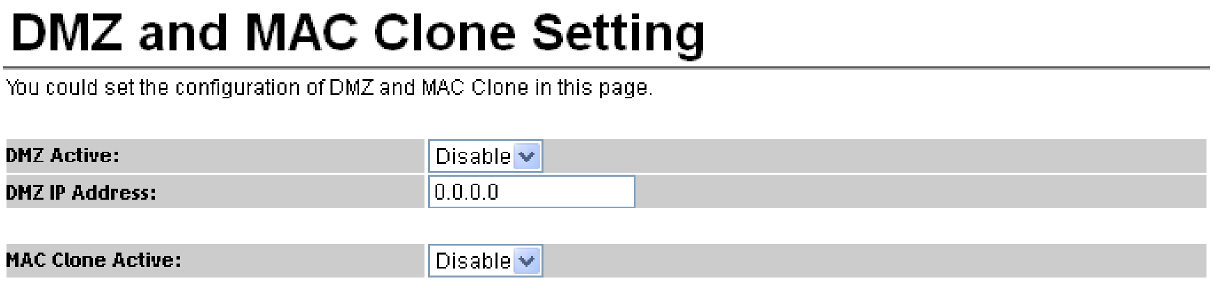
DMZ and MAC Clone

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

DMZ Active: If setup as On, all of packets (expect SIP packets) will send to the specific IP address.

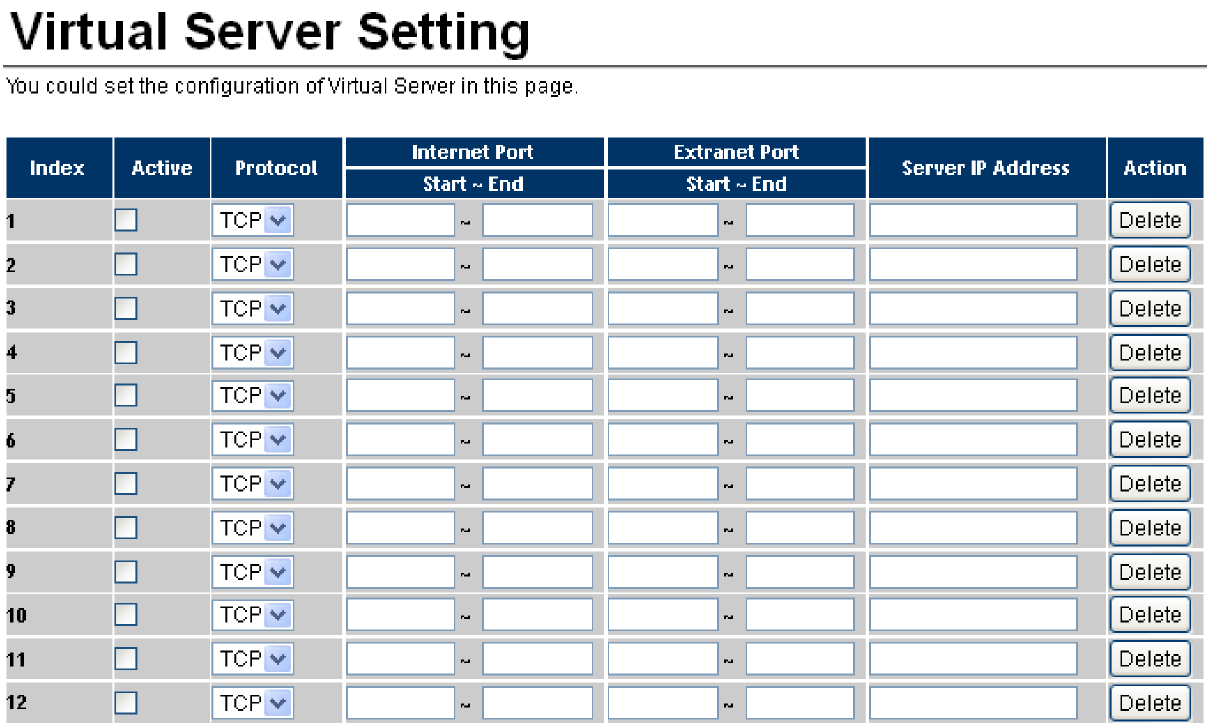
DMZ IP Address: The DMZ host IP address.

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



Virtual Server

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



Field	Description
Index	The serial number. There are total 12 records from Num 1 to 12.
Active	The activate status. The default is Disable, this record will been 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

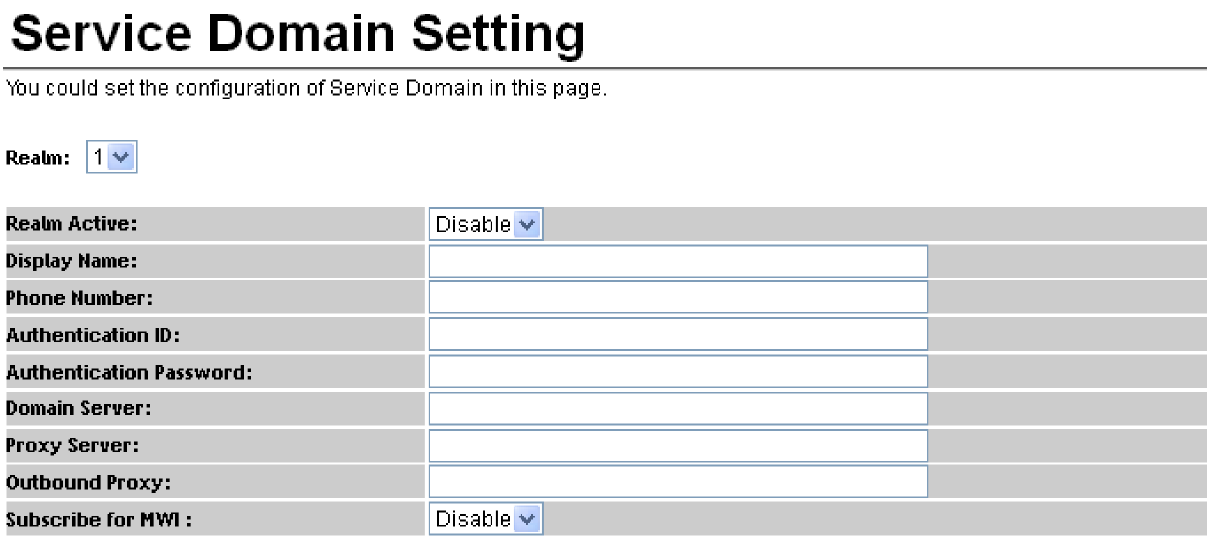
Chapter 7

SIP Setting

7

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 five 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 five SIP accounts.



Field	Description
Realm	Which line you want to use.
Realm Active	First you need click Active to enable the Service Domain, then you can input the following items.
Display Name	The serial number. There are total 24 records from Num 0 to 23.
Phone number	The activate status. The default is Disable, this record will been activate if enable.
Authentication ID	you need to input the Register Password get from your ISP.
Authentication Password	you need to input the Register Name 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.
Subscribe for MWI	Setting MWI(message-waiting indicator) function, when enable system will frequency send the MWI message.

	ⓘ**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.

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

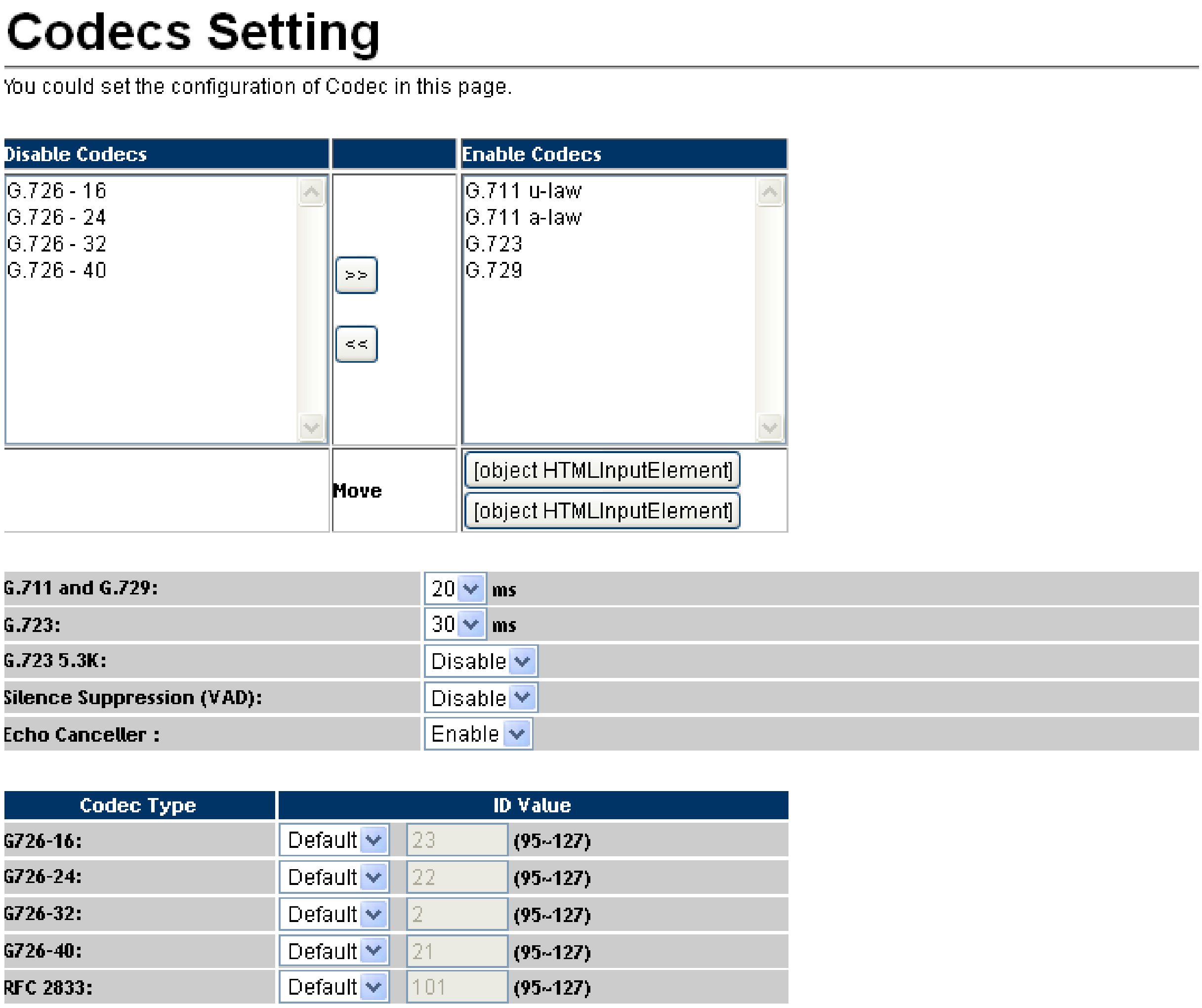
If you have more than one SIP account, you can following the steps to register to the other ISP.

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

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

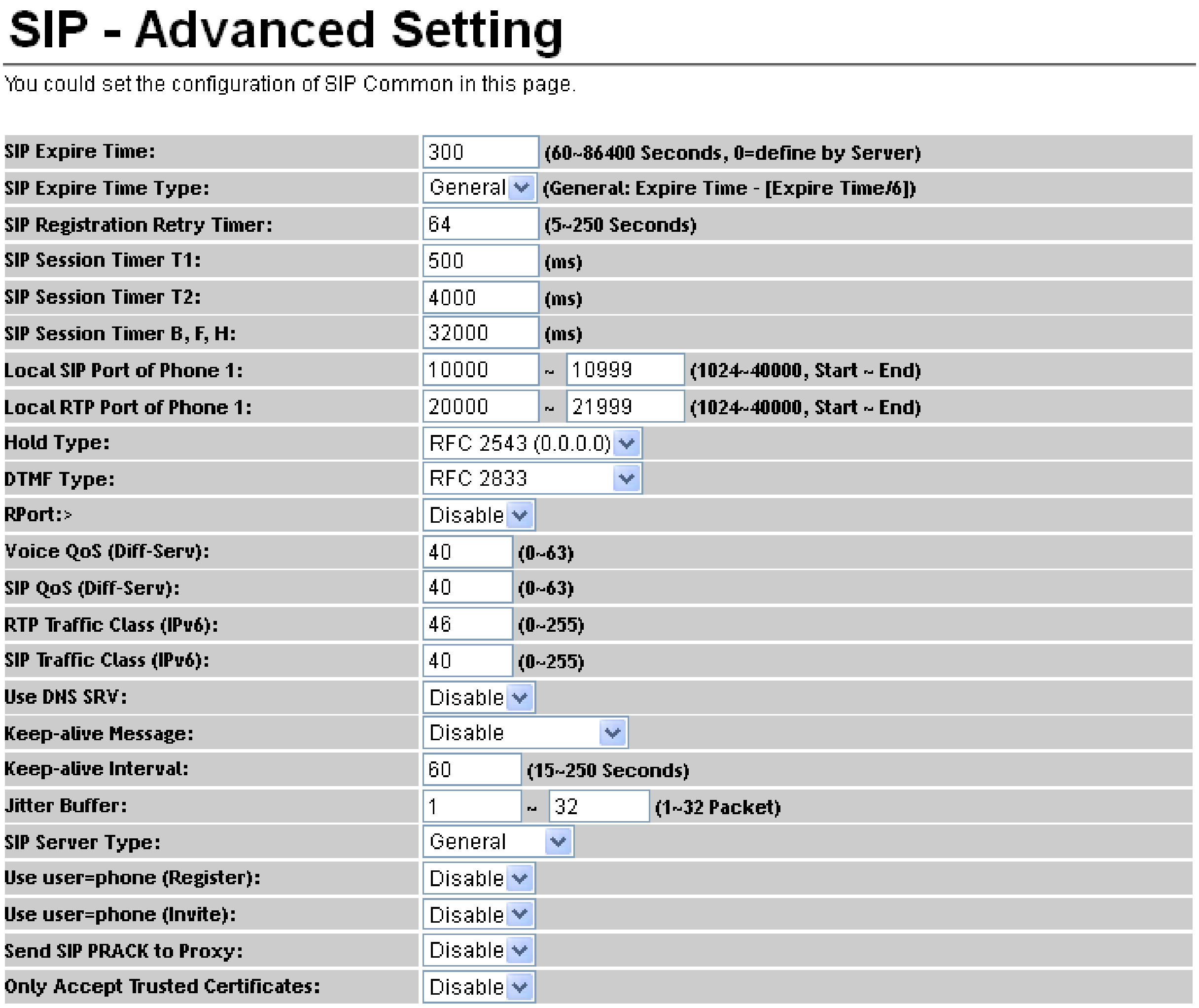
Codec Setting

This page defines the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button. Also in page defines the Codec ID. Sometimes 2 VoIP devices with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, and then you can change your Codec ID. When you finished the setting, please click the Submit button.



SIP Advance Setting

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



SIP - Advanced Setting

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

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

Field	Description
SIP Expire Time	To setup the registration interval time.
SIP Expire Time Type	Default is G eneral; Register interval time setting. Provide items General (standard), 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10。 ⓘ**Note**: register server need support this function. Register time calculated General: expire time-[(expire time/30)*6], when Expire Time>60 it will start to work, if less then 60 seconds, it will decrease 5 seconds. 1/2: expire time * 1/2. 2/3: expire time * 2/3. 3/4: expire time * 3/4. 4/5: expire time * 4/5. 5/6: expire time * 5/6. 6/7: expire time * 6/7. 7/8: expire time * 7/8. 8/9: expire time * 8/9. 9/10: expire time * 9/10.

SIP Register Retry Timer	If SIP register fail, system will retry interval after this time.
SIP session timer T1	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses. ⓘ**Note**: register server need support this function.
SIP session timer T2	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses. ⓘ**Note**: register server need support this function.
SIP session timer Timer B, F, H	Setting the maximum retransmit interval for non-INVITE requests and INVITE responses。。 ⓘ**Note**: register server need 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 duplicate
Local RTP Port of phone 1	Setting the phone 1 RTP start and end port. All the port can't be duplicate
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. in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.
RPort	To change this setting, please following your ISP information. When you finished the setting, please click the Submit button. ⓘ**Note**: register server need 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 need to cooperate with the others Internet devices.
RTP Traffic Class(IPV6)	IPV6 RTP traffic class
SIP Traffic	IPV6 SIP traffic class

Class(IPV6)	
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 Alives Packet	Always to transport the network packets to keep the NAT port could be opened continuous.
Keep Alives Period	To setup the interval time for transporting packets.
Jitter Buffer	To setup the size for jitter buffer packets.
SIP Server Type	Provide different register server: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink, for different server system will adjust some system parameters ⓘNote: register server need support this function.
Use user = phone (Register):	When sending the registe r package, in p ackage Header will add the ”user=phone” message 。 ⓘNote: register server need support this function.
Use user = phone (Invite):	When sending the dialing package, in package Header will add the ”user=phone” message 。 ⓘNote: register server need support this function.
Send SIP PRACK to Proxy:	When sending the SIP package, in package Header will add the ”PRACK” message 。 ⓘNote: register server need support this function.
Only Accept Trusted Certificates:	Only accept call from proxy, if system receive the IP dialing, system will refuse the call.

Chapter 8

Advance Setting

8

Status Log

Display and saving systems running status message data, Press “Get Status Log” can backup 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	
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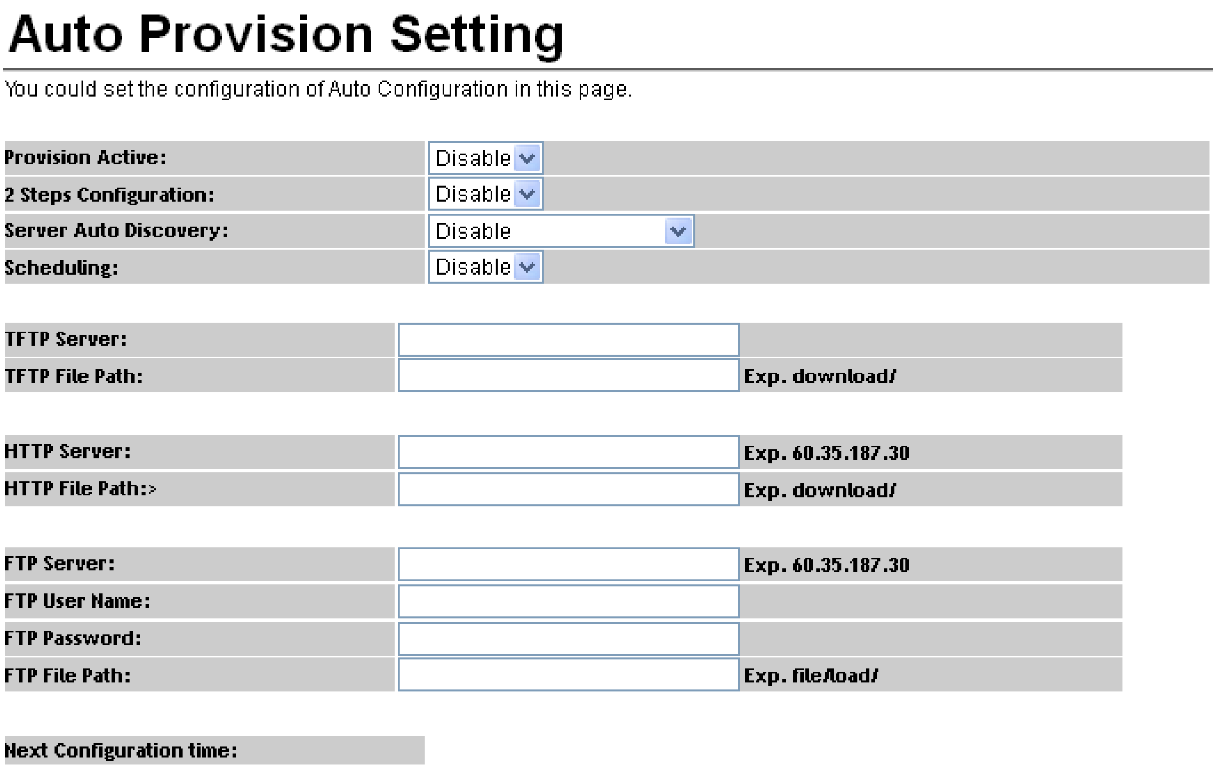
18	
19	
20	
21	
22	
23	
24	

Export System Log

Auto Config

T 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.



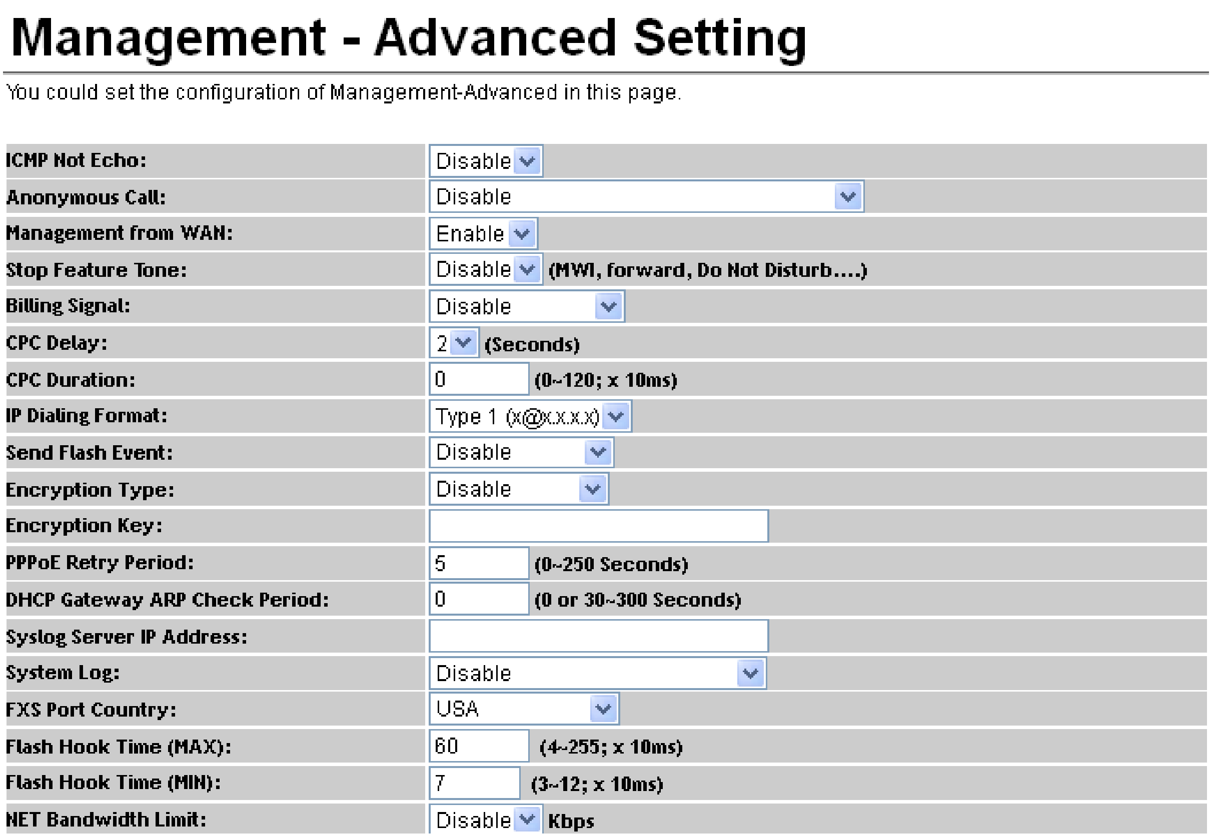
Auto Provision Setting

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:		

Management-Advanced Setting

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



Management - Advanced Setting

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

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

Field	Description
ICMP Not Echo	This function can disable echo when someone ping this device, it can avoid hacker try to attack the device
Anonymous Call	If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information. ⓘNote: register server need support this function.
Management form WAN	When [Enable] allow user login from WAN.
Stop Feature Tone	When [Enable] if system set the function like [Sub scribe for M WI, forward, DND], when user pickup the phone will hear the remind tone [Do Do Do]
Billing Signal	There are provide th ree type billing types: Polarity Reversal, Tone_12K and Tone_16K. ⓘNote: register server need support this function.
CPC Delay	When receive the discon nect signal, machine will to cut the volt age down to 0V after this time
CPC Duration	When starting to cut the voltage down to 0V, machine will to continue this state by this time.
IP Dialing Format	Setting IP dialing formte, when [Disable] can't use IP dialing to ma ke call.
Send Flash event	There are provide two flash formats: DTMF Event and SIP Info.
Encrypt Type	There are provide seven encrypt formats: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP。 ⓘNote: register server need support this function.
Encrypt Key	Some encrypt type must enter the Encrypt Key ⓘNote: register server need support this function.
PPPoE Retry Period	If PPPoE dial-up connection fail, ma chine will retry the dial-up moti on after this time.
DHCP Gateway ARP Check Period	The period to check the DHCP gateway ARP.
Syslog Server IP Address	There are se ven 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	Define the MAX silence time for F XO port. After the time will

Timeout	disconnection the line.
PSTN CID forward	It must work with [Phone – General] [Auto Answer] function or [Phone – Caller Service] [Forward] function。 When enable this function, The caller ID from FXO, can transfer to other device
Generate Flash Signal for PSTN	FXO flash time, define would you hold or hang on 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 on.
Flash Hook Time (Min)	Minimum Flash time , to detect the call on hold or hang on.
NET Bandwidth Limit	Setting the limitation for LAN Bandwidth

Tones

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

Tones Setting

You could set the configuration of Tones in this page.

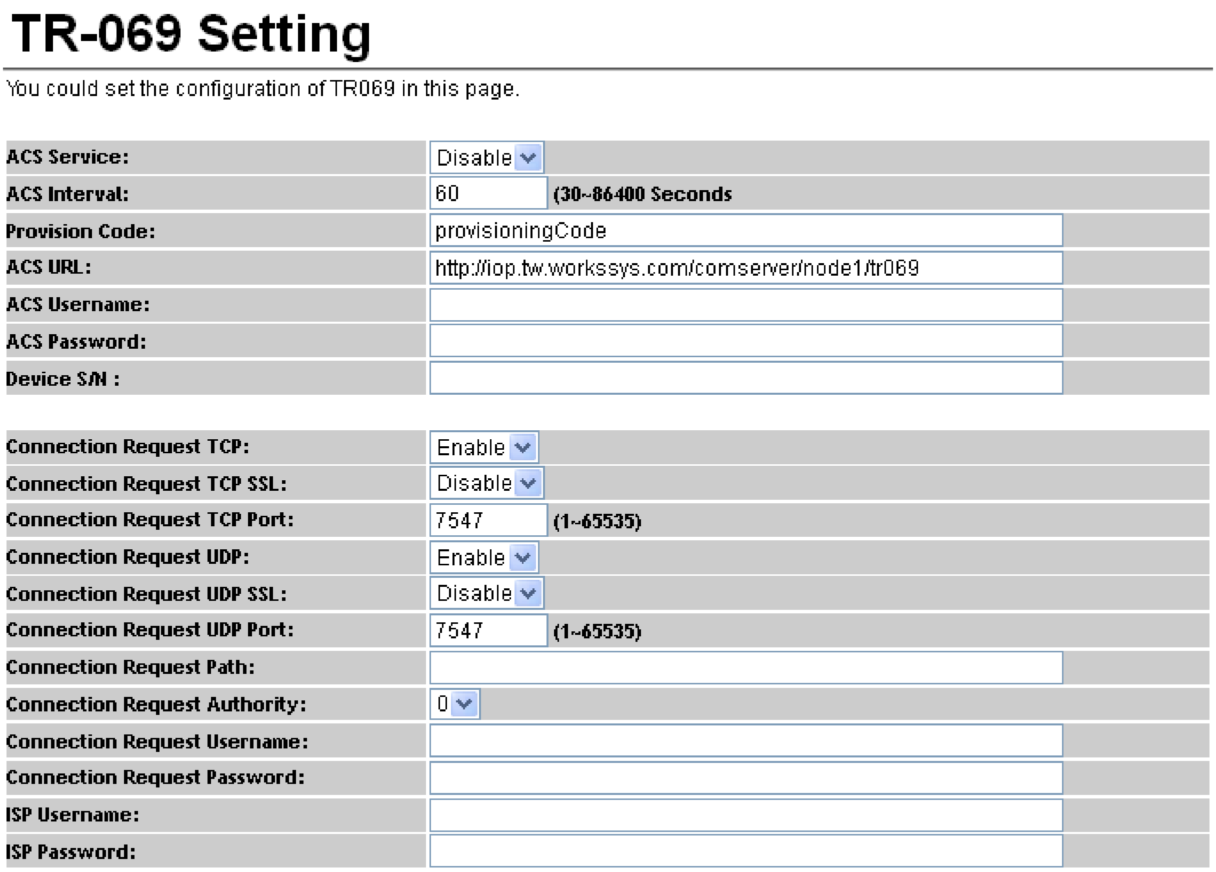
	Dial		Ring Back	Busy	Congestion	Ring	Call Waiting
Cadence On:	☐		☑	☑	☑	☑	☑
Hi-Tone Freq.:	440		480	620	620	480	440
Lo-Tone Freq.:	350		440	480	480	440	350
Hi-Tone Gain:	4522		2261	2261	2261	15360	2261
Lo-Tone Gain:	4522		2261	2261	2261	15360	1130
On Time 1:	0	x 10ms	200	50	30	200	30
Off Time 1:	0	x 10ms	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

TR-069

In this page you can programming the TR-069 setting.

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

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



TR-069 Setting

You could set the configuration of TR069 in this page.

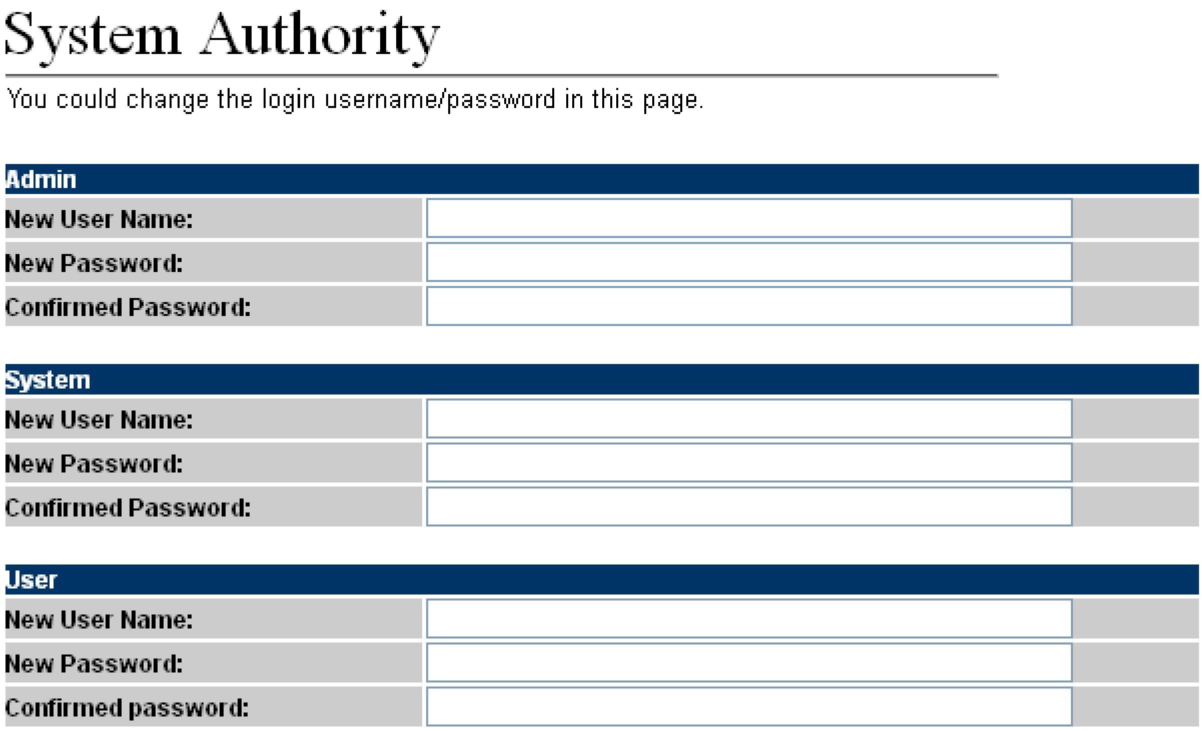
ACS Service:	Disable
ACS Interval:	60 (30~86400 Seconds
Provision Code:	provisioningCode
ACS URL:	http://iop.tw.workssys.com/comserver/node1/tr069
ACS Username:	
ACS Password:	
Device S/N :	
Connection Request TCP:	Enable
Connection Request TCP SSL:	Disable
Connection Request TCP Port:	7547 (1~65535)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

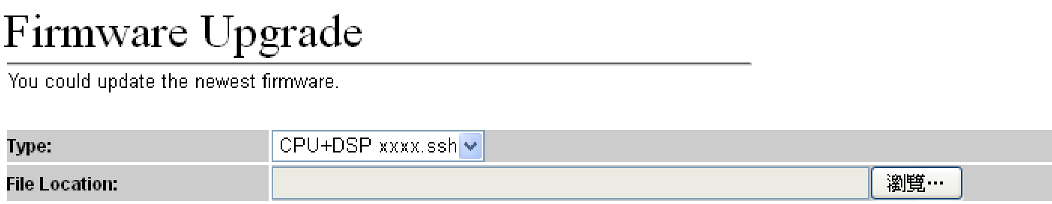
System Authority

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

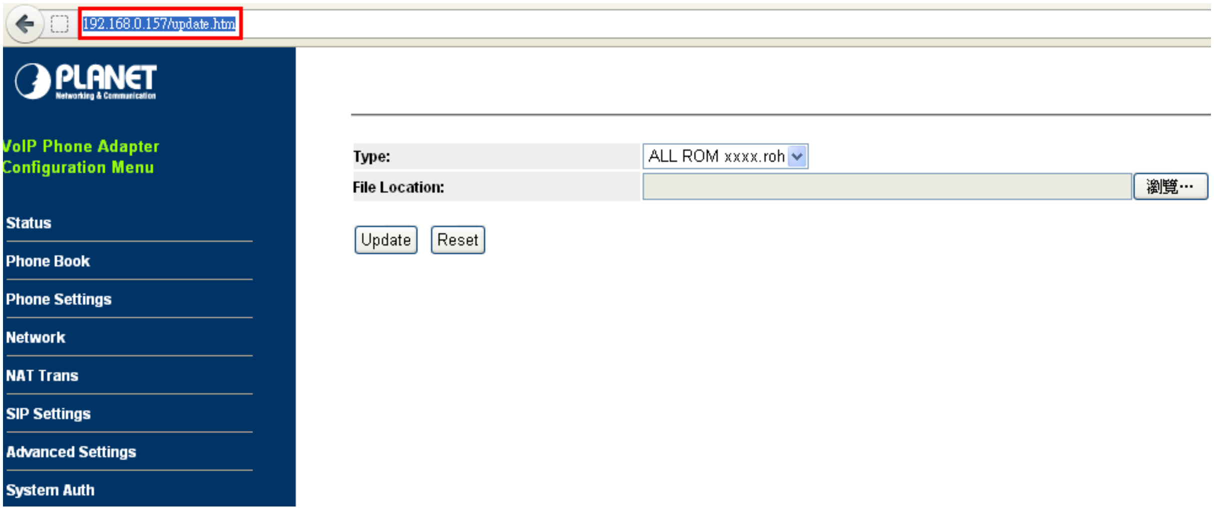


Firmware Upgrade

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.



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



Note:

For technological consideration, we've strongly suggested refering 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 Update Settings

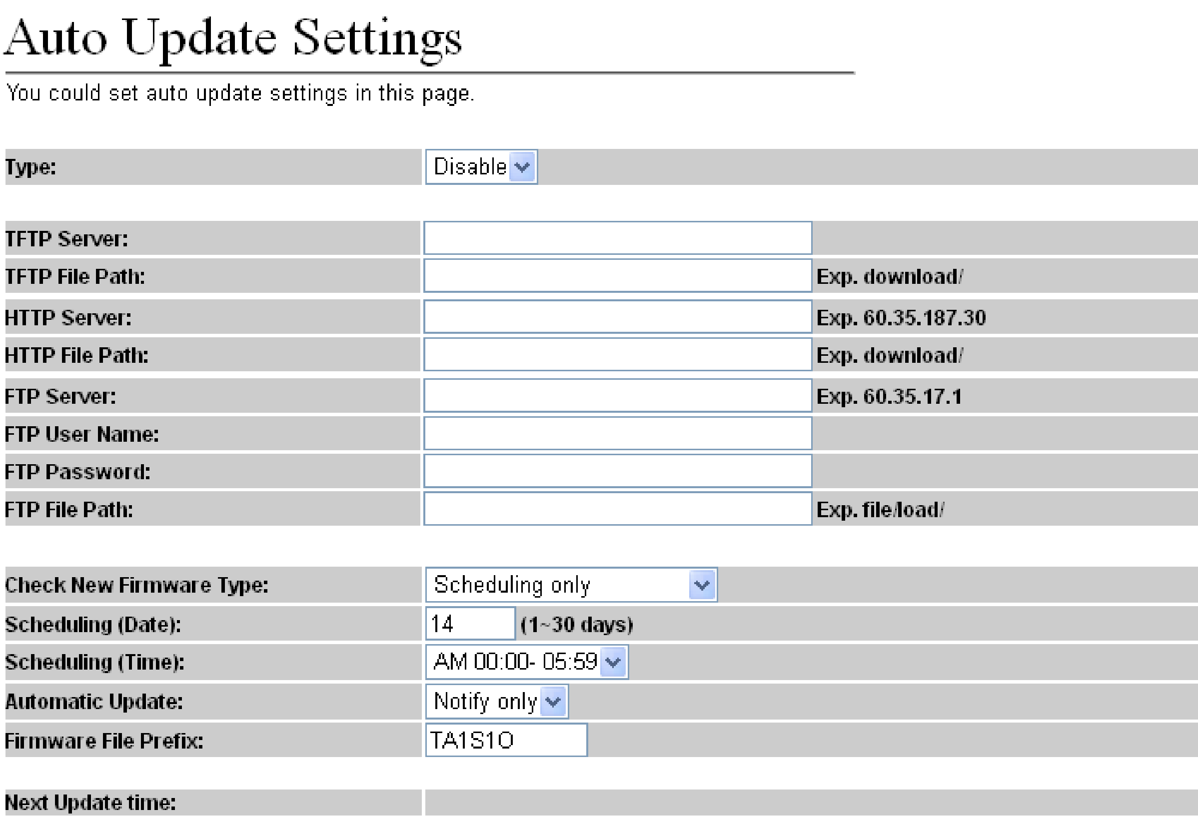
The device can update new firmware with the **gz** or **ds** file format automatically by the Auto Upgrade function.

Field	Descriptions
Type	There are TFTP/ FTP and HTTP 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.
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 according to the below ways to check the new firmware.

	- 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. **- 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 sel ected to Power On, the machine w ill chck the new firmware accoeding the scheduling time/date and power on. If there are ne w 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

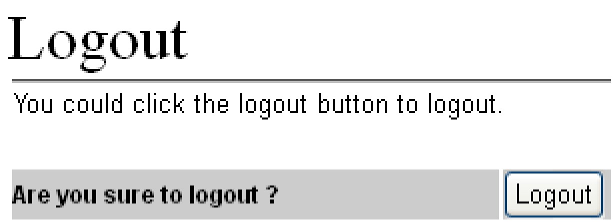
In Default Setting you can restore the Phone Adapter to factory default in this page. You can just click the Restore button, then the Phone Adapter will restore to default and automatically restart again.

Save and Reboot

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

Logout

Lougout the system, it will return to login page.



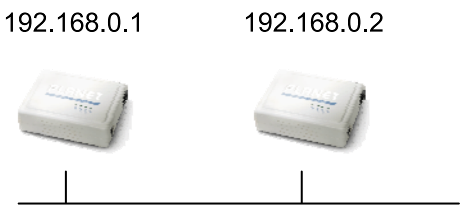
Appendix A Voice Communication Samples

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 address are 192.168.0.1, 192.168.0.2

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



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

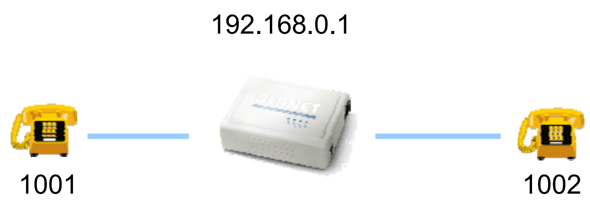
Test the scenario:

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

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

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

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



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

Test the scenario:

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

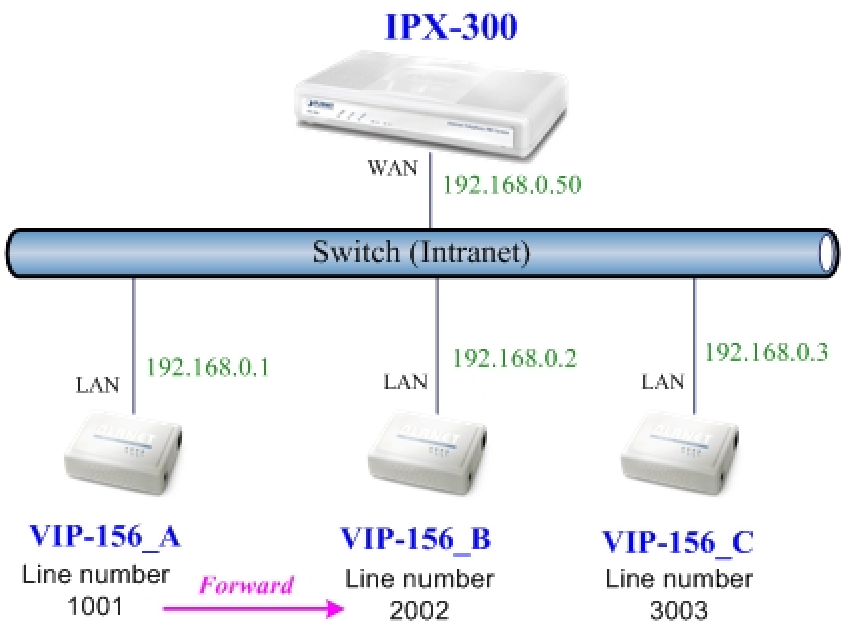
ⓘ Hint

- 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: 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 IPX-300 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 service** config menu. In 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.

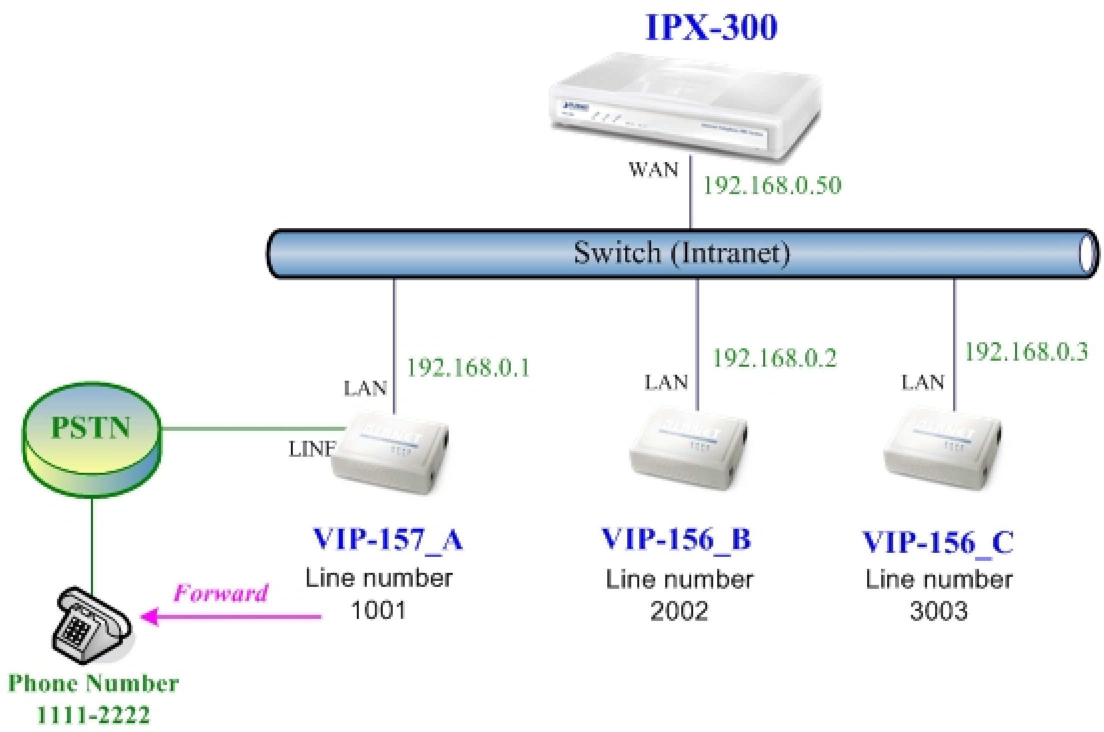
Forward Type	Forward Number	Ring / Phone
Always	2002	3 **Phone 1**

Test the scenario:

1. VIP-156_C 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 4: Call Forward Feature_Example 2

In this example, there are one VIP-157 and two VIP-156 register to IPX-300. 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 service** config menu. In the setting page, please select the **All Forward** function to **PSTN** choice and fill in the **Forward Type and Forward Number** of PSTN Phone Number 11112222, then the sample configuration screen is shown below:

Call Service

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

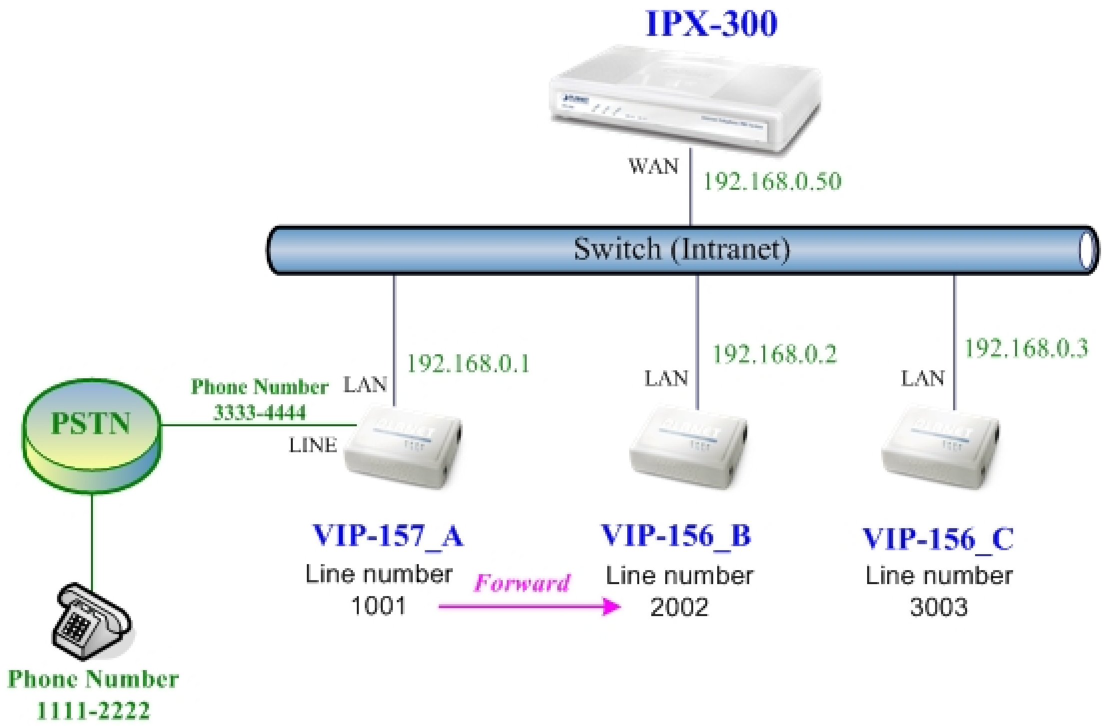
Forward Type	Forward Number	Ring / Phone
Always to PSTN	11112222	3 Phone 1

Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the number 1001(VIP-157_A)
3. Because VIP-157_A 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 5: Call Forward Feature_Example 3

In this example, there are one VIP-157 and two VIP-156 register to IPX-300. 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 service** config menu. In the setting page, please select the **All Forward** function to **IP** choice and fill in the **Forward Type and Forward Number** of of VIP-156_B, and then the sample configuration screen is shown below:

Call Service

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

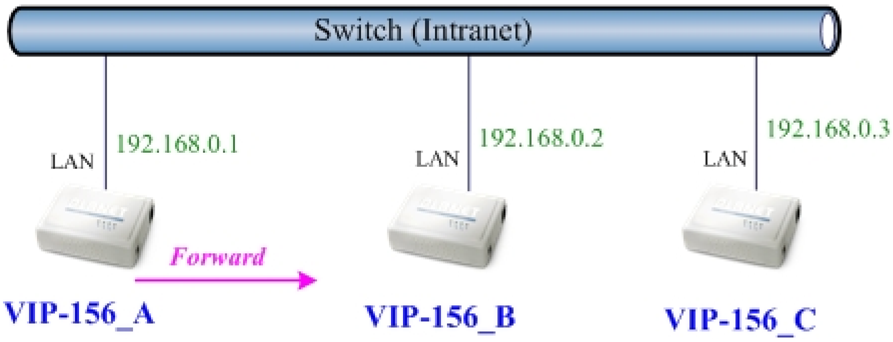
Forward Type	Forward Number	Ring / Phone
Always	2002	3 **Phone 1**

Test the scenario:

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

Case 6: 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 service** config menu. In 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.

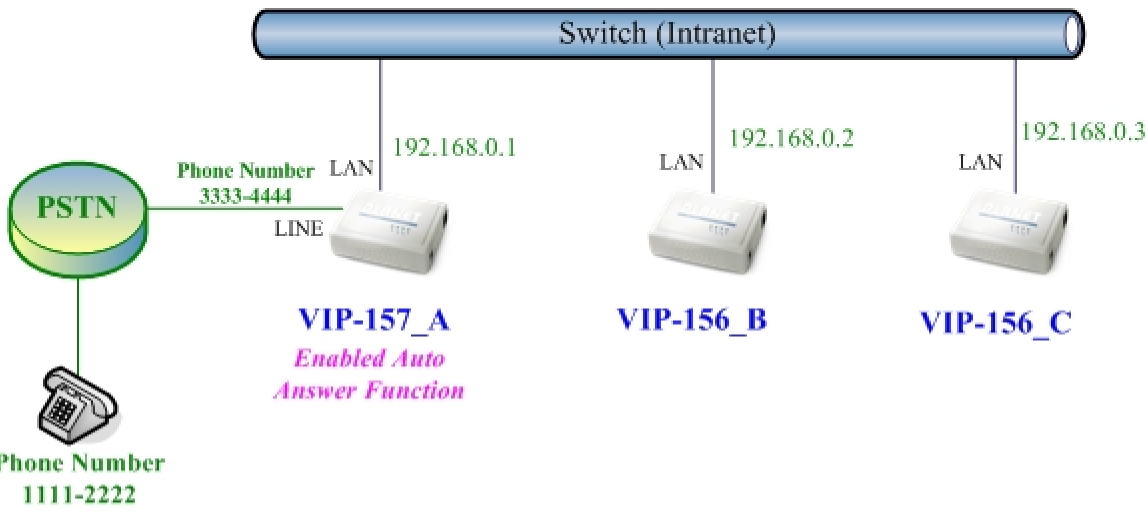
Forward Type	Forward Number	Ring / Phone
Always to PSTN	192.168.0.2	3 **Phone 1**

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 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 7: 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 service** config menu. In the setting page, please disable All **Forward** function, and then the sample configuration screen is shown below:

Call Service

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

Forward Type	Forward Number	Ring / Phone
Disable		3 Phone 1

STEP 2:

Please log in VIP-157_A via web browser, browse to the Phone Settings / General setting 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 Type:	Both
Auto Answer Counter:	3
PIN Code:	Enable
PIN Code Number:	●●●

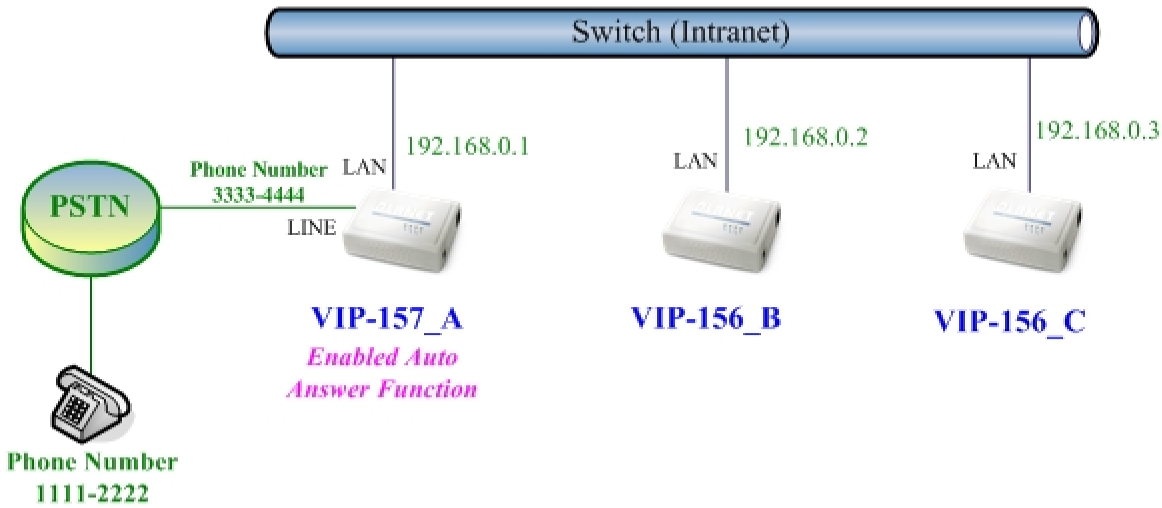
Test the scenario:

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 8: 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:

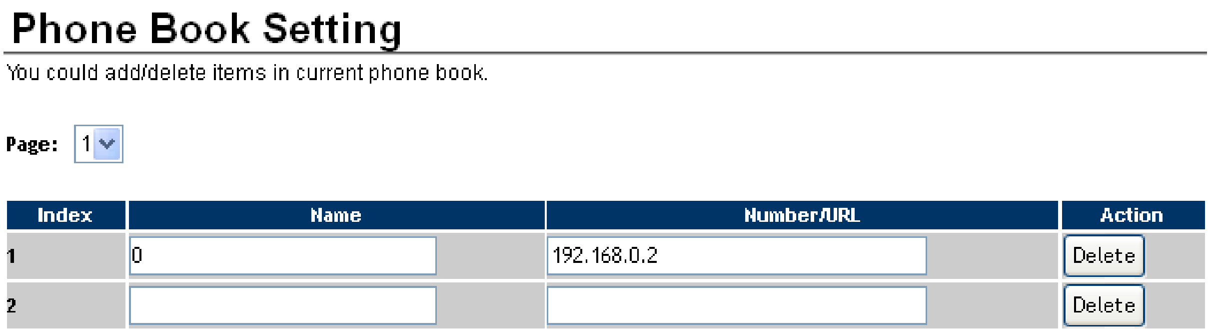
STEP 1:

Please log in VIP-157_A via web browser, browse to the Phone Settings / General setting 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 Type:	Both
Auto Answer Counter:	3
PIN Code:	Enable
PIN Code Number:	•••

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:


Phone Book Setting

You could add/delete items in current phone book.

Page: 1

Index	Name	Number/URL	Action
1	0	192.168.0.2	Delete
2			Delete

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 steps of how to set up some call features of the ATA. Please follow the steps below to utilize those features.

Call Transfer

A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A 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

4*: Realm 4

5*: Realm 5

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 handsetof 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.711a/u, G.723.1 (6.3k/5.3k), G.726, G.729A, G.729B, GSM			
Fax support	T.38			
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer			
Protocols	SIP 2.0 (RFC-3261), TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE			
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			

EC Declaration of Conformity

For the following equipment:

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

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

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.
For the evaluation regarding the R&TTE, the following standards were applied:

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

Responsible for marking this declaration if the:

☒ Manufacturer ☐ Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

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

Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Jimmy Lin

Taiwan — *Place*

7th July, 2005 — *Date*

Legal Singnature

PLANET TECHNOLOGY CORPORATION

e-mail: sales@planet.com.tw http://www.planet.com.tw
11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C. Tel:886-2-2219-9518 Fax:886-2-2219-9528

EC Declaration of Conformity

For the following equipment:

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

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

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.
For the evaluation regarding the R&TTE, the following standards were applied:

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

Responsible for marking this declaration if the:

☒ Manufacturer ☐ Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

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

Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Jimmy Lin

Taiwan — *Place*
7th July, 2005 — *Date*
Legal Singnature

EC Declaration of Conformity

For the following equipment:

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

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

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).
For the evaluation regarding the EMC, the following standards were applied:

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

Responsible for marking this declaration if the:

☒ Manufacturer ☐ Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

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

Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Taiwan	**16 March, 2006**	
Place	*Date*	*Legal Signature*

PLANET TECHNOLOGY CORPORATION

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EC Declaration of Conformity

For the following equipment:

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

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

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).
For the evaluation regarding the EMC, the following standards were applied:

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

Responsible for marking this declaration if the:

☒ Manufacturer ☐ Authorized representative established within the EU

Authorized representative established within the EU (if applicable):

Company Name: Planet Technology Corp.

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

Person responsible for making this declaration

Name, Surname Jimmy Lin

Position / Title : Product Manager

Taiwan	**17 March, 2006**	
Place	*Date*	*Legal Signature*