Communication Services

VoIP Analog Telephone Adapter

G-157 Land Line Converter

User's manual

Version 1.1

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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.

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Chapter 1

Introduction

Overview

Based on years of VoIP manufacturing experiences, GULFSIP Communications Services provide a varity of VoIP Telecommunication Solutions which meet GULFSIP Customers needs.

Cost-effective, easy-to-install and simple-to-use, the GULFSIP ***G1S, G157 land line converter*** VoIP Phone Adapters ("**ATA**" in the following term) converts standard telephones to IP-based networks. GULFSIP, the service provider which offer users traditional and enhanced telephony communication services via the existing broadband connection to the Internet or corporation network.

With the **G-157 Land Line Converter**, home users and companies are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The **G-157 Land Line Converter** 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, which will help in installation of the device at any networking invironment.

Product Features

- Feature-rich telephone service over home or office Internet/Intranet connection
- 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 **G-157 Land Line Converter**

Power adapter

Quick Installation Guide

User's Manual CD

RJ-11 cable x 1

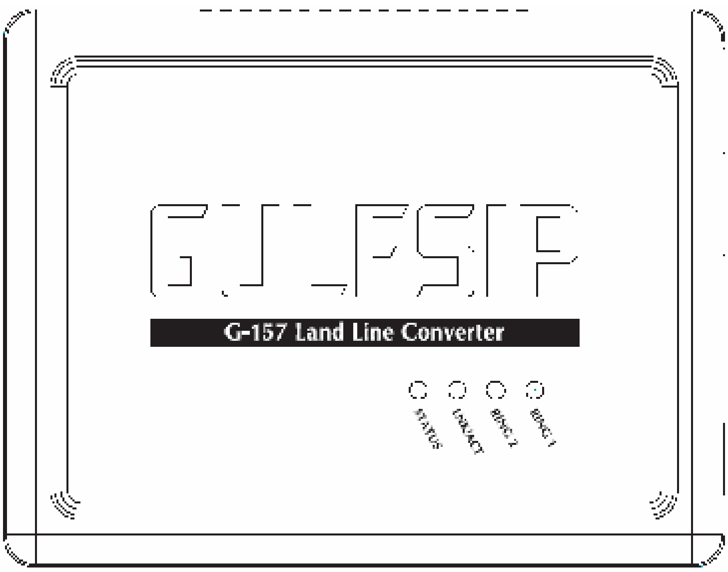
RJ-45 cable x 1

Physical Details

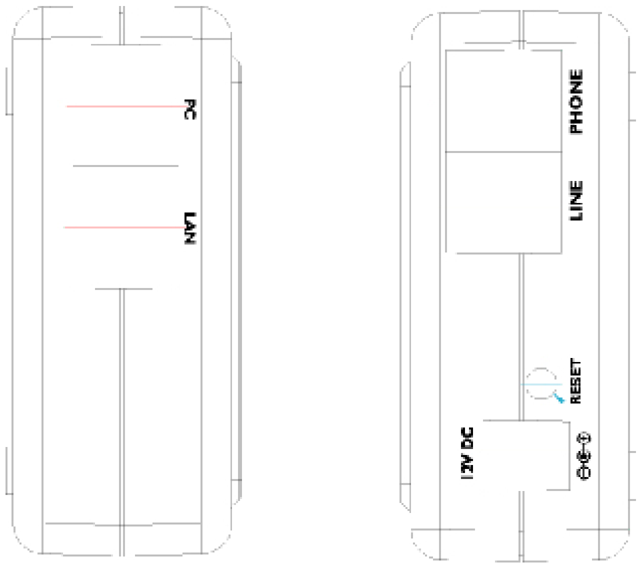
The following figure illustrates the front/rear panel of ATA.

Respective model/descriptions are shown below:

G-157 Land Line Converter SIP Analog Telephone Adapter



Front Panel of G-157 Land Line Converter



Left / Right Panel of `G-157 Land Line Converter`

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.
3	**12V DC**	12V DC Power input outlet
4	**Reset**	Reset to the factory default setting

ë **Note**

Machine default PC Port ip is 192.168.123.1 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 G-157 Land Line Converter

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.

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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

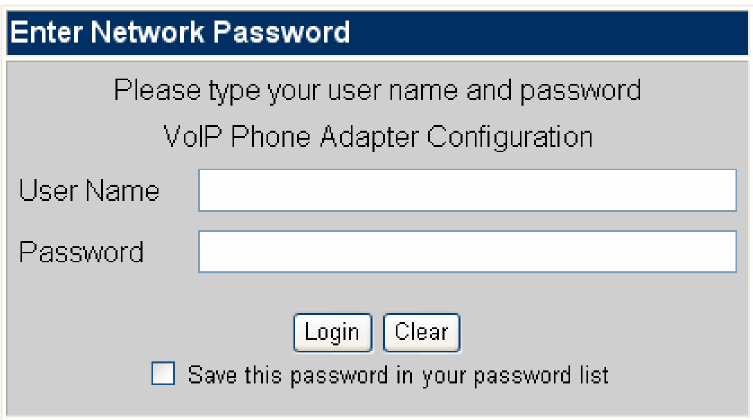
GULFSIP **`G-157 Land Line Converter`** provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access

To start **`G-157 Land Line Converter`** 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 PC interface IP address of **`G-157 Land Line Converter`** is **192.168.123.1**. You may now open your web browser, and insert http://192.168.123.1 in the address bar of web browser to logon ATA web configuration page.



ATA will prompt for logon username/password, please enter: **admin** / **gulfsip** to continue machine administration.

ë **Note**

```
Please make sure that your PC is connected to the PC port
in the G-157 Land Line Converter to start access to the
device. 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 for Static IP or PPPoE:

Static IP configuration Settings:

In case you found that your Internet connection (DSL Router, cable,..) does not provide your device **G-157 Land Line Converter** an IP address from it's DHCP server **(automatic IP address)**, so you need to set a static IP on the LAN port of the device which is connected to the internet connection.

How to insure that your internet router does not provide an IP address?

- Connect a network cable between the LAN port and the Internet router LAN Port.
- Please pick up the handset of the phone which connected to our device in the phone port, and dial #126#, if you hear an IP address from the same range which your internet router provide, for example **192.168.x.x,** so your Router working on DHCP Server mode.

But if you hear **0.0.0.0**, or **169.254.x.x**, so your internet router does not provide IP address, and you need to set a static IP manually.

To set a Static IP, please execute the following Steps:

- Connect a network Cable between your PC and the PC port in our device **G-157 Land Line Converter**, as shown in the Figure above.

You must be sure that your PC is working on the DHCP Client mode, to ensure that please do the following steps:

"**Start** " => "**Control Panel**" => **"Network Connections"**

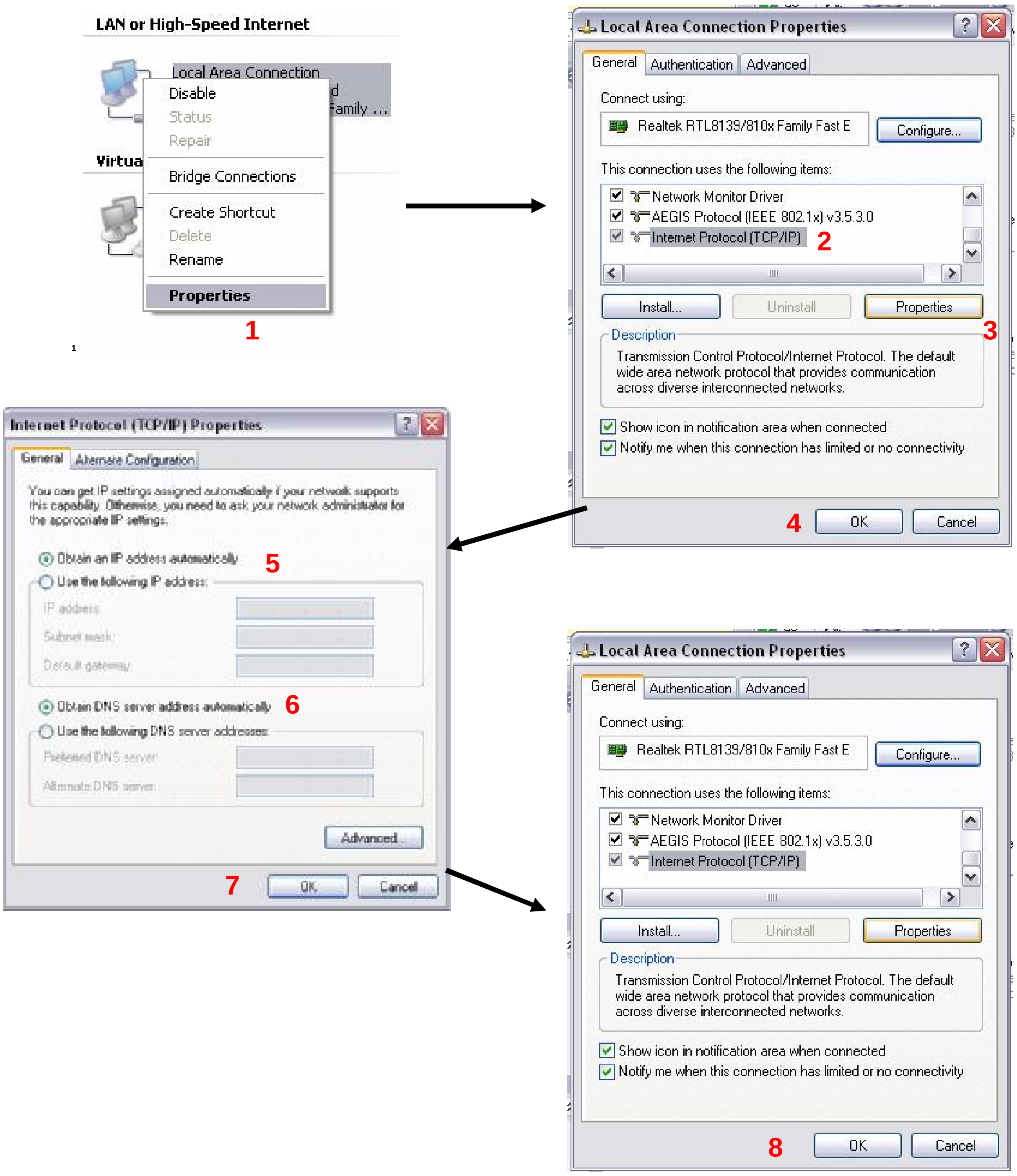


Figure (5)

Execute your web browser, and insert the IP address (default: **192.168.123.1**) of **`G-157 Land Line Converter`** in the adddress bar. After logging on machine with username/password (default: **admin/gulfsip**), browse to “**Network**” --> “**LAN Settings**” configuration menu:

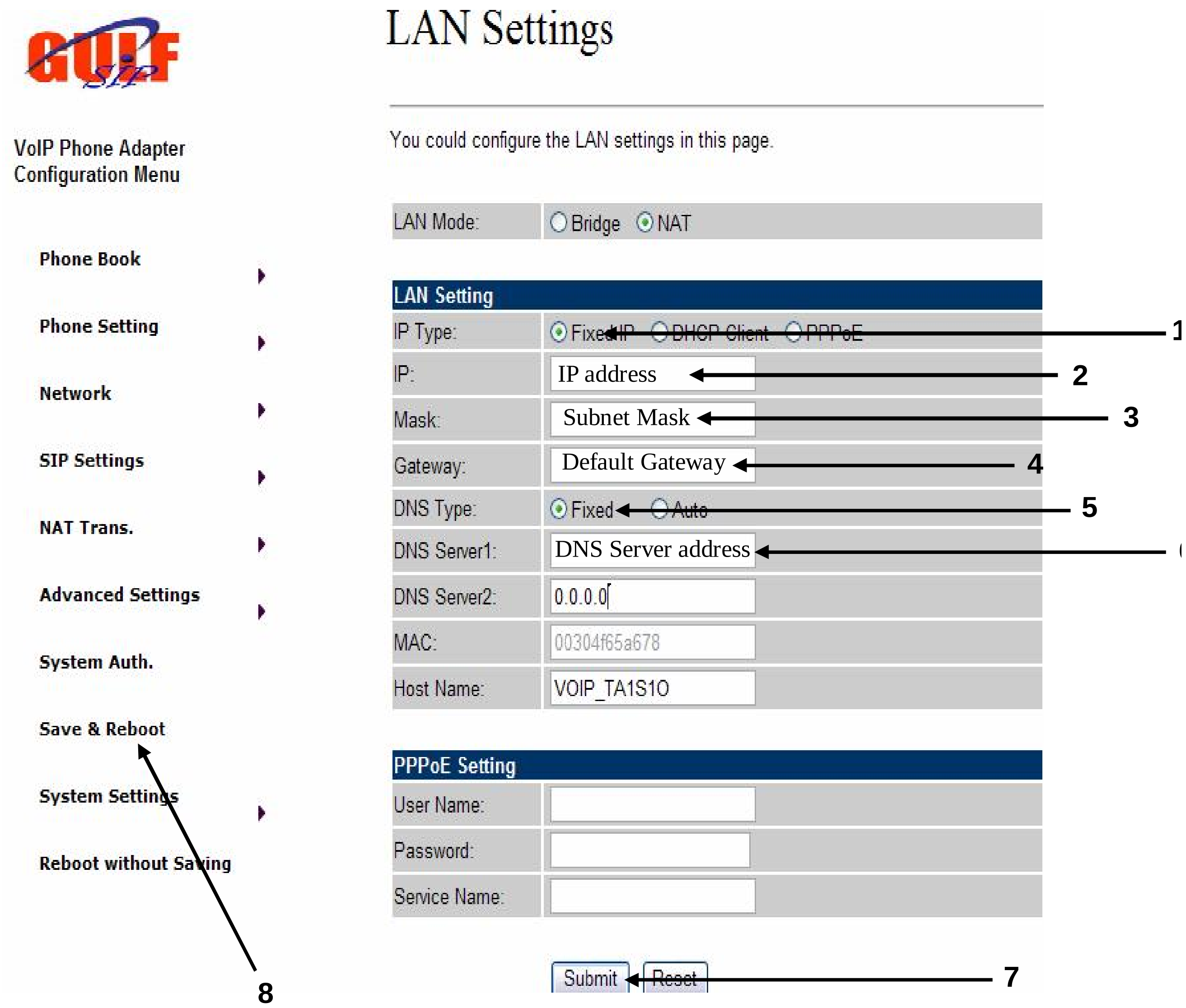


Figure (6)

Parameter Description

IP address LAN IP address of ATA

Default: None

Subnet Mask LAN mask of ATA

Default: None

Default Gateway Gateway of ATA

Default: None

Network settings via Keypad commands

The **`G-157 Land Line Converter`** support telephone keypad configurations, please connect analog telephone set and refer to the following table for machine network configurations.

! Note:

Pick up the handset and press #190# then hang up *before* you do following Settings:

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 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
#134 + [numberof rings] #	Set the number of rings before the Line port open.		To set the number of rings for Auto Answer.
#127 + [port number] #	Set a SIP Port	None	To change the SIP Port.

Following keypad commands can be used to display the network settings enabled on ATA via voice prompt **without requireing to dial #190# before the command**:

IVR Menu Choice	Machine operation	Notes:
#120#	Check PC IP Address	IVR will announce the current PC-port IP address of the ATA.
#126#	Check IP Address of the LAN Port	IVR will announce the current IP address of the `G-157 Land Line Converter` LAN port.
#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.

i Hint

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

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

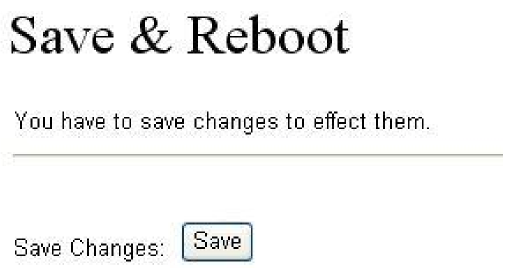
Connection Type	Data required.
Fixed IP	In most circumstances, it is no need to configure the DHCP settings.
DHCP clinet	The ISP will assign IP Address, and related information.
PPPoE	The ISP will assign PPPoE username / password for Internet access,

i Hint

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

Save Modification to Flash Memory

Most of the VoIP router parameters will 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.



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Chapter 3

Network Service Configurations

Configuring and monitoring your ATA from web browser

The **`G-157 Land Line Converter`** 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 **`G-157 Land Line Converter`** web configuration, you must have one of these web browsers installed on computer for management

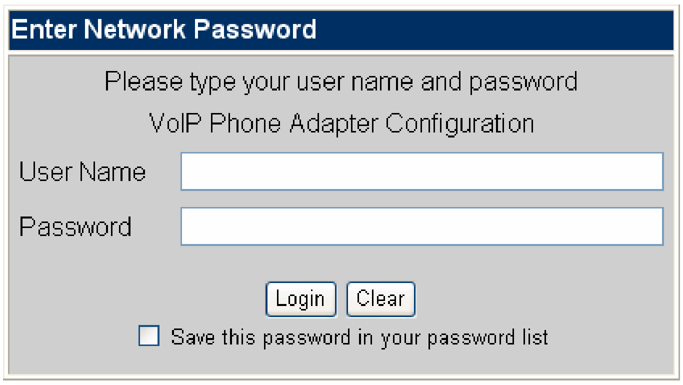
- Microsoft Internet Explorer 6.0.0 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

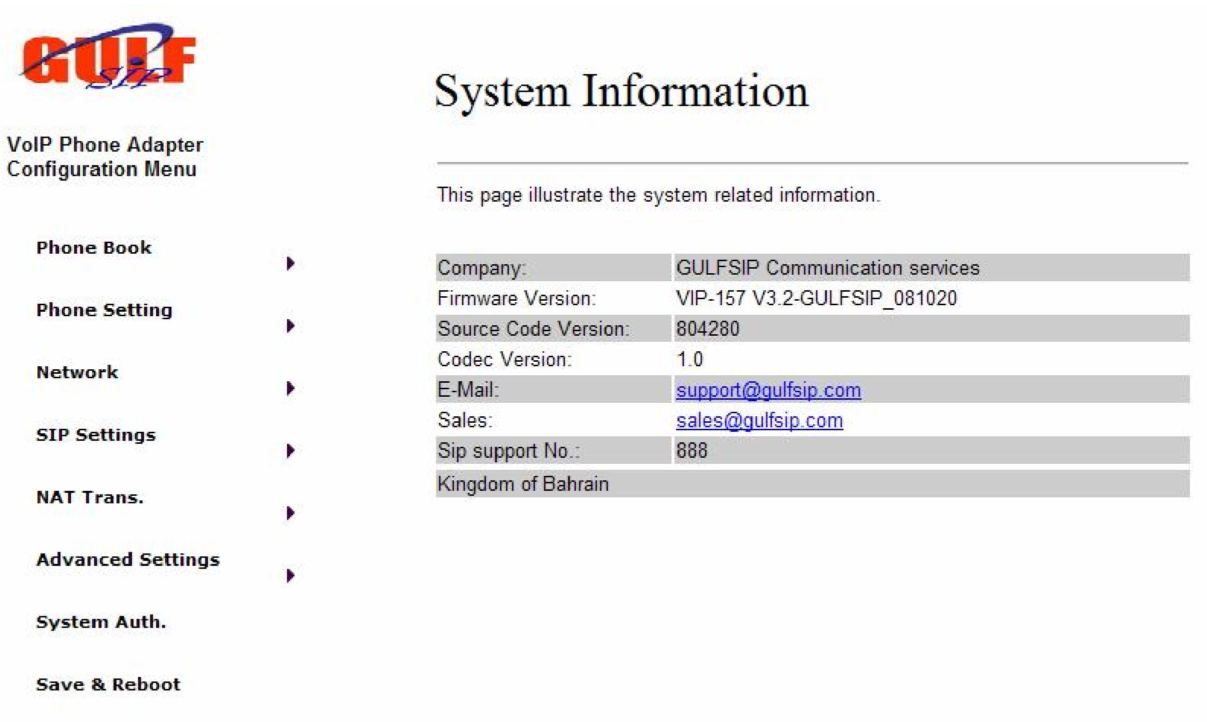
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.123.1 to logon Phone Adapter web configuration page.

Phone Adapter will prompt for logon username/password: ***admin / gulfsip***



G-157 Land Line Converter 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.



G-157 Land Line Converter main page

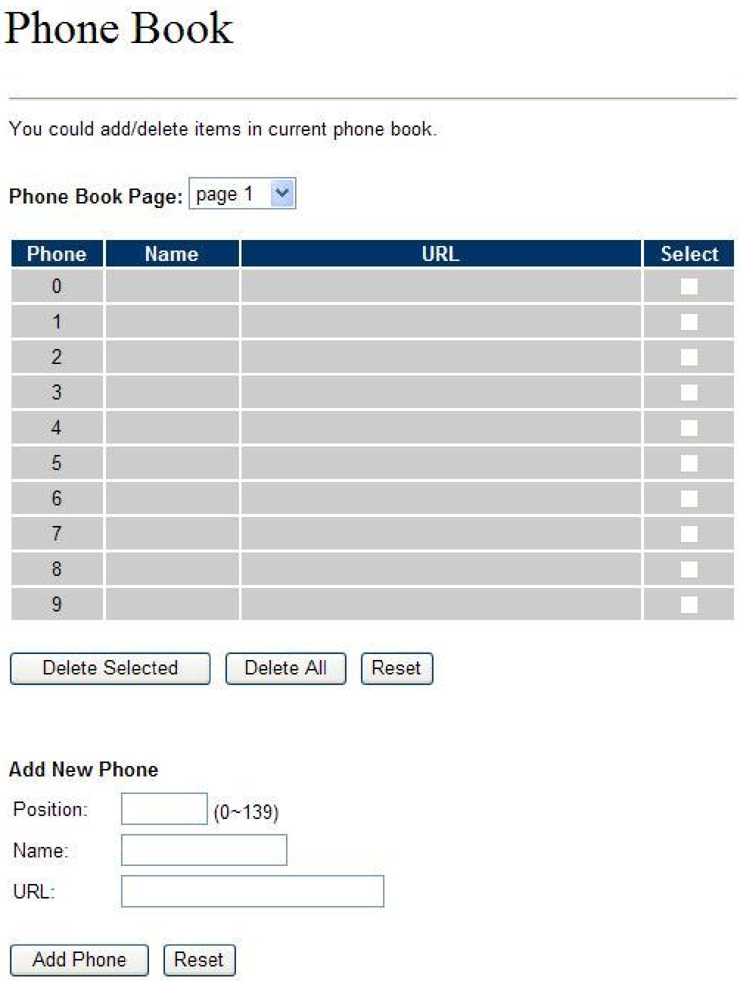
4

Chapter 4

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.



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

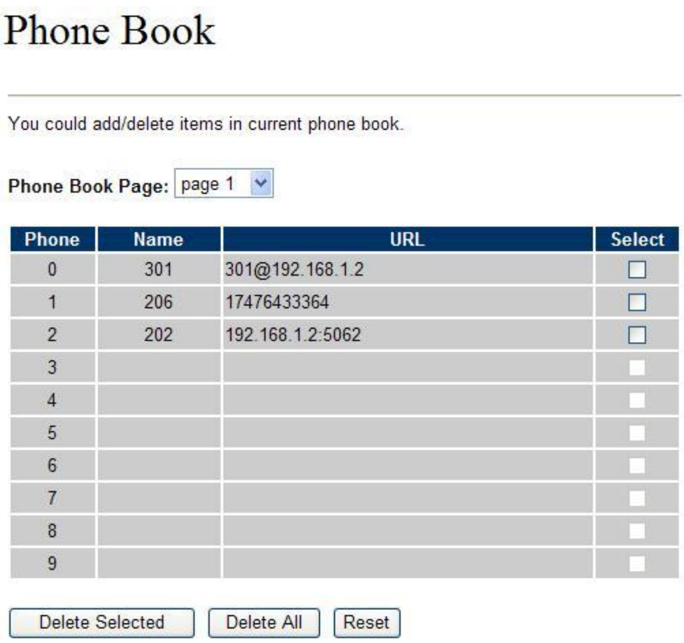
	numerals.
URL	Fill in the outgoing number (Line Number) or IP address.
Select	To select this record.

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:



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

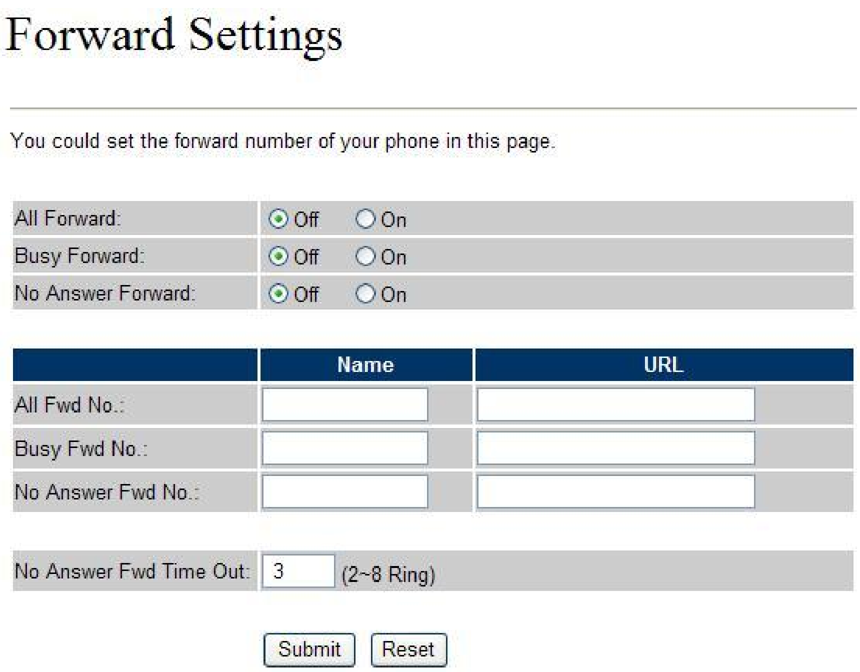
This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

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.

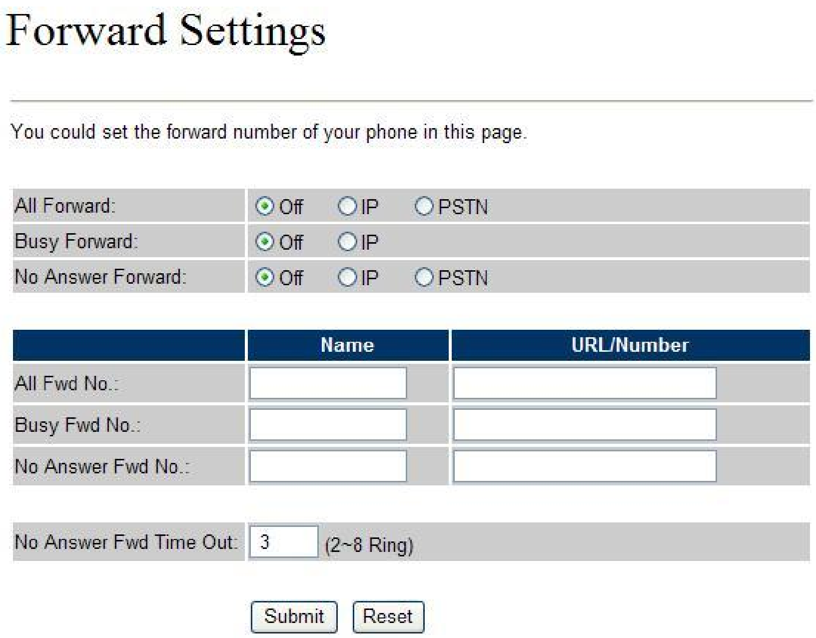
No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

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



Call Forward function for **`G-157 Land Line Converter`**

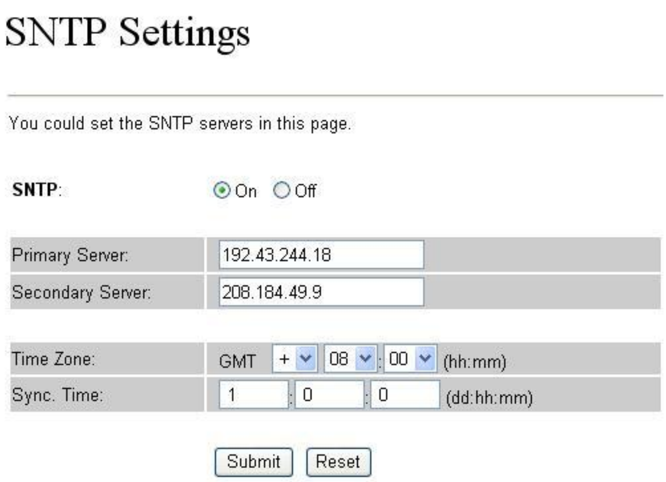
Call Forward to PSTN (`G-157 Land Line Converter`): G-157 not only supports Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, and then input the name and the PSTN number in URL/Number field.



Call Forward function for **`G-157 Land Line Converter`**

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.



Volume Setting

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

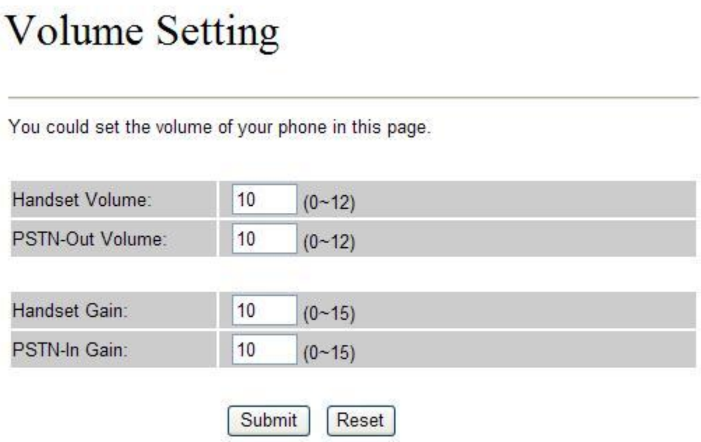
Handset Volume is to set the volume for you can hear from the handset.

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

Beside the above settings, G-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 G-157 land Line Converter

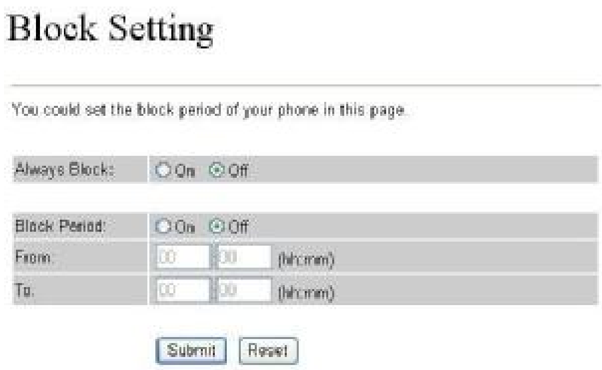
Block Setting

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.

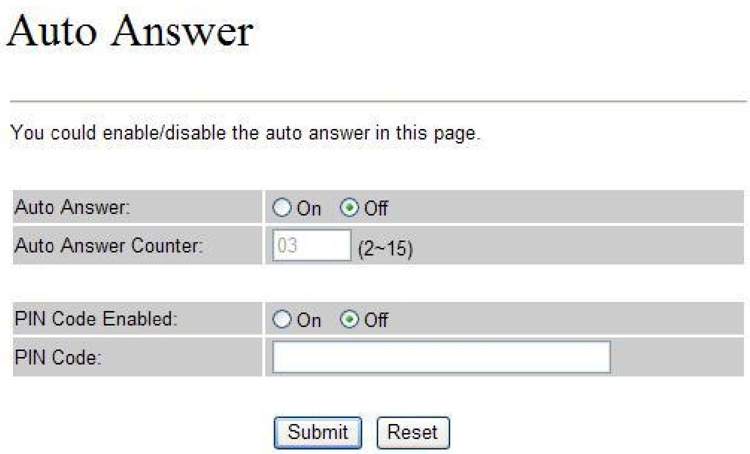
When you finished the setting, please click the Submit button.



Auto Answer (For G-157 Land Line Converter)

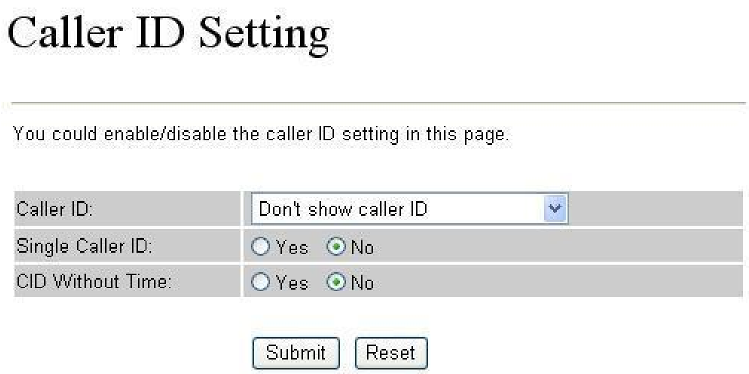
This page defines the Auto Answer function. You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the **`G-157 Land Line Converter`** can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the **`G-157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



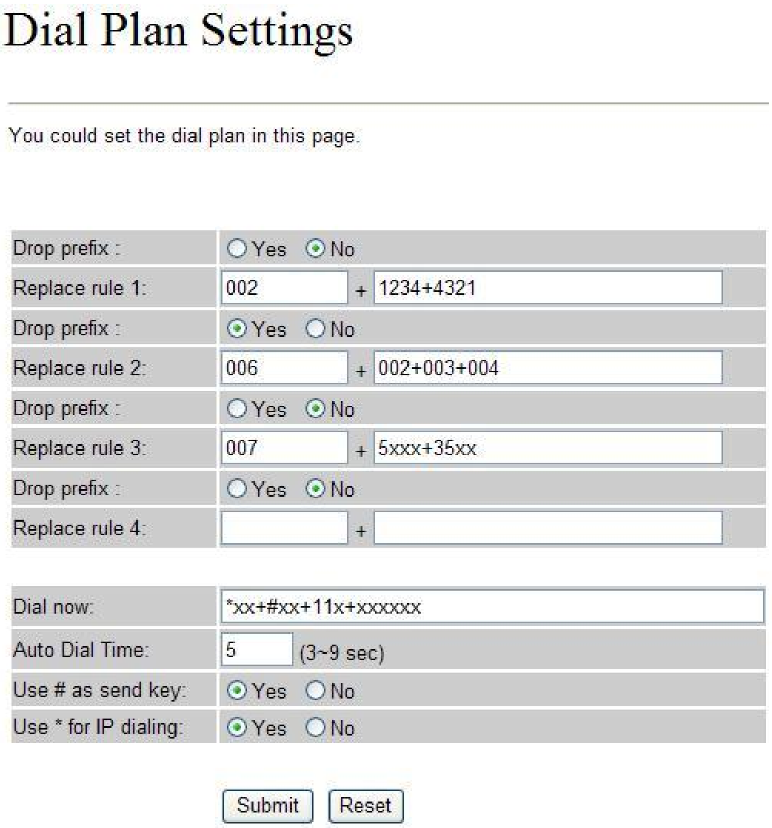
Caller ID settings

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.



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.



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. - 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. i **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 symtem 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.
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)

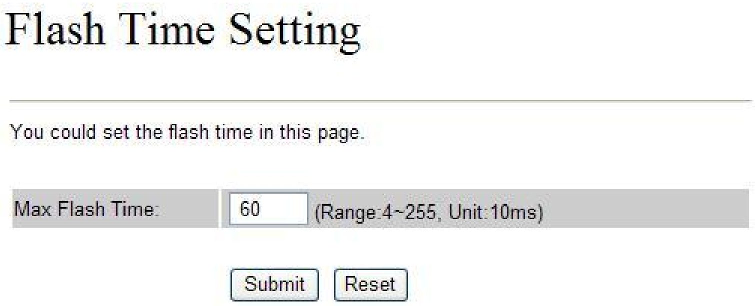
1. If the dialing number start as **"5"** and follow 3 digits, it will add the 007 at begin. The real dialing number is [**0075xxx**].
2. If the dialing number start as **"35"** and follow 2 digits, it will add the 007 at begin. The real dialing number is [**00735xx**].

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**.

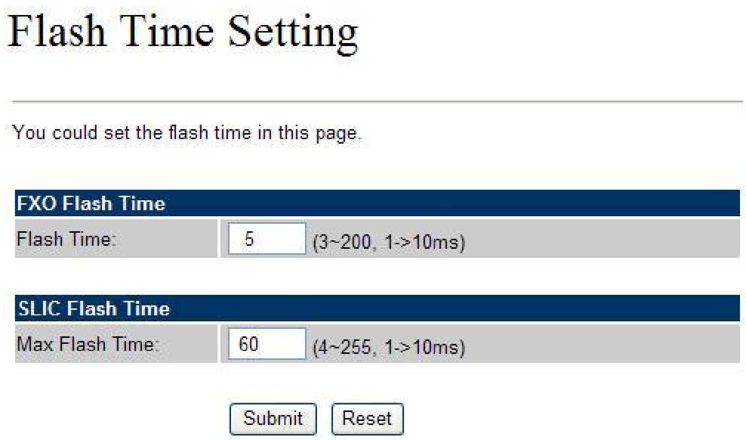
Flash Time Setting

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 Settings for **`G-157 Land Line Converter`**

Beside the above settings, **`G-157 Land Line Converter`** also can set the flash time of FXO port.



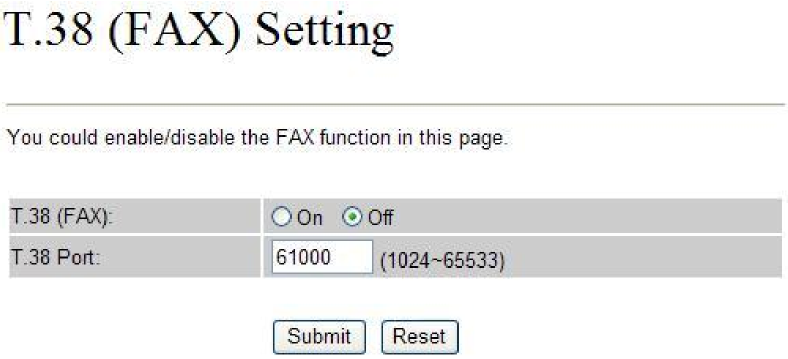
Flash Time Settings for **`G-157 Land Line Converter`**

Call waiting Settings

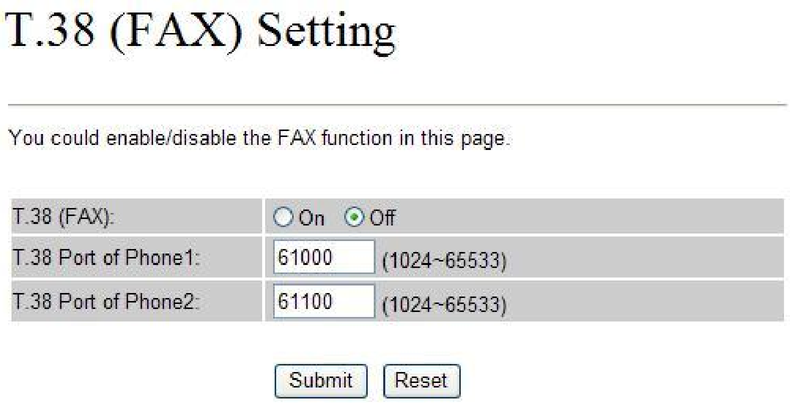
When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear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

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) Settings for **`G-157 Land Line Converter`**



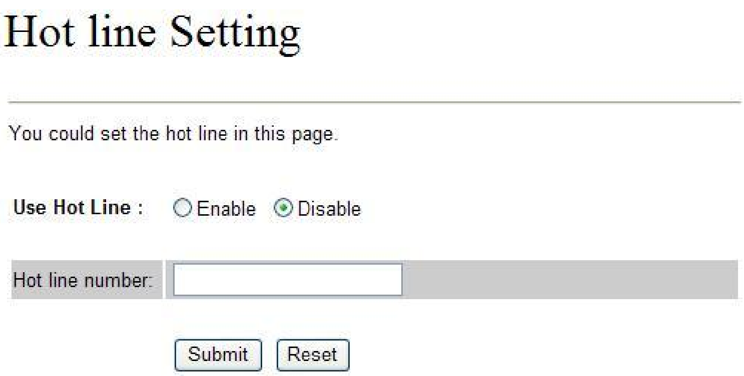
T.38 (FAX) Settings for **`G-157 Land Line Converter`**

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.



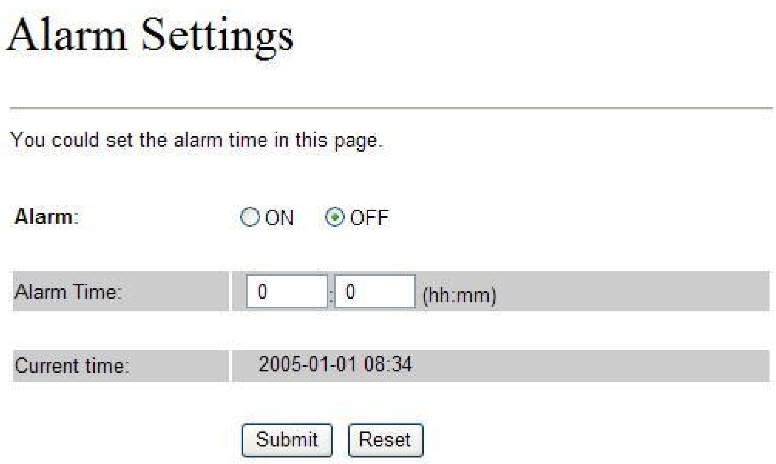
Alarm Settings

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.



LAN Settings

This page defines the LAN setting in this page.

LAN Mode: 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.

IP Type: 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: The IP address

Mask: The sub net address

Gateway: The default gateway address

DNS Server1: The default is 168.95.192.1, it could setup the first DNS server address.

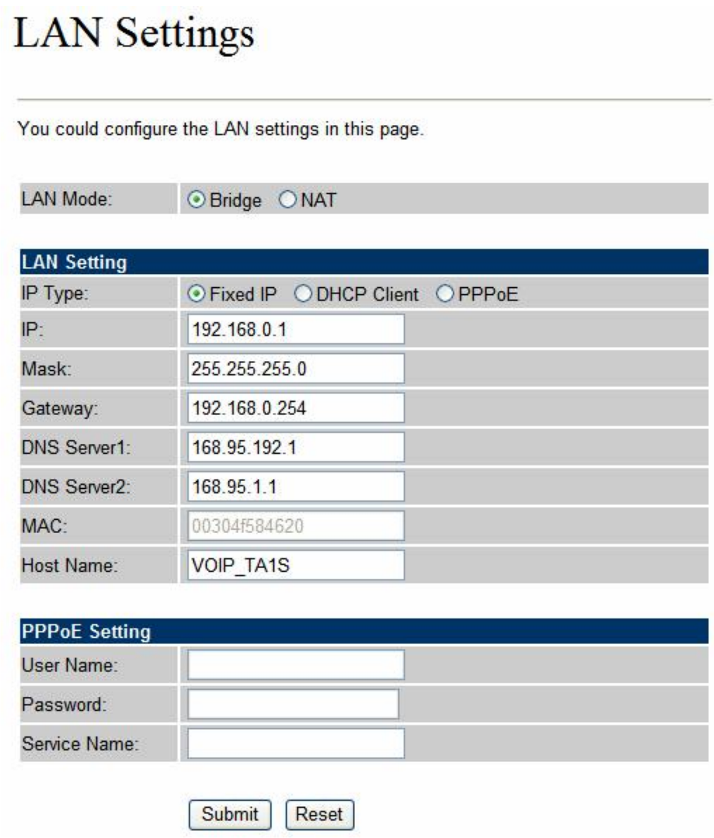
DNS Server2: The default is 168.95.1.1, it could setup the second DNS server address.

MAC: The MAC of LAN port

Host Name: The product model

User Name: The PPPoE connection account name. It could inpout numeral or character, the maximum date length are 63.

Password: The PPPoE connection account password. It could inpout numeral or character, the maximum date length are 63.



PC Settings

This page defines the PC setting in this page.

IP: The IP address of PC port. (**Default**: 192.168.123.1)

Mask: The sub net address. (**Default**: 255.255.255.0)

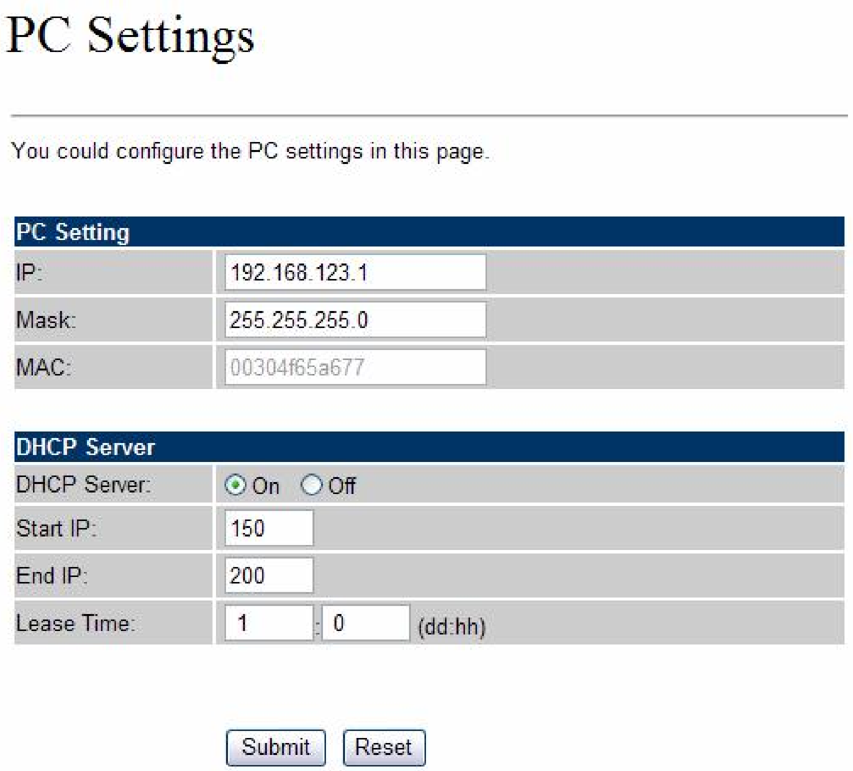
MAC: The MAC of PC port

DHCP Server: It will allot the IP address automatically when enable this function.

Start IP: Start IP of lease table

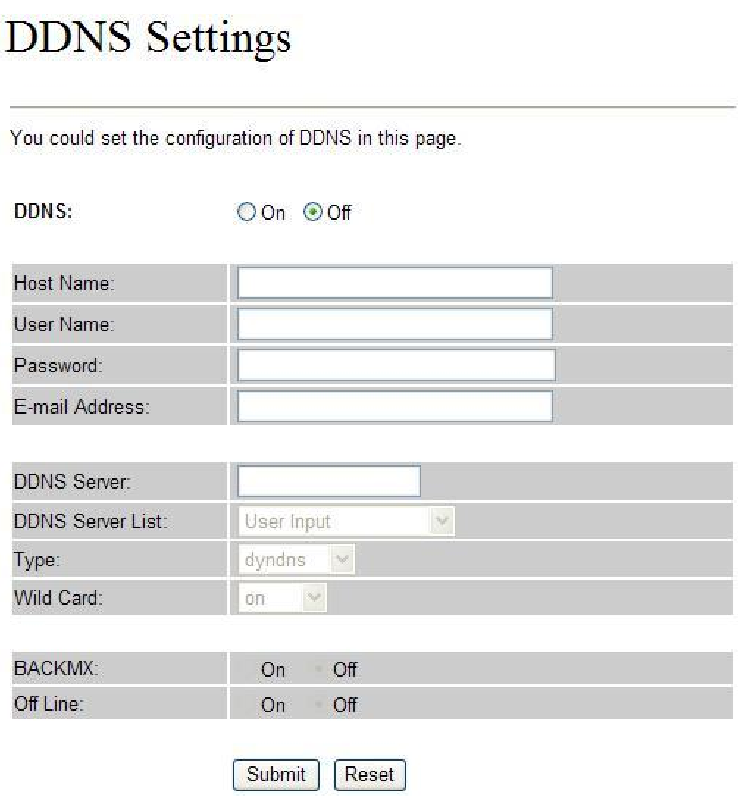
End IP: End IP of lease table. Network device connecting to the PC port can dynamic obtain the IP in the range between start IP and end IP

Lease Time: DHCP server lease time



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.



VLAN Settings

This page defines the VLAN setting in this page. This function needs to co-operate with network devices which have VLAN function.

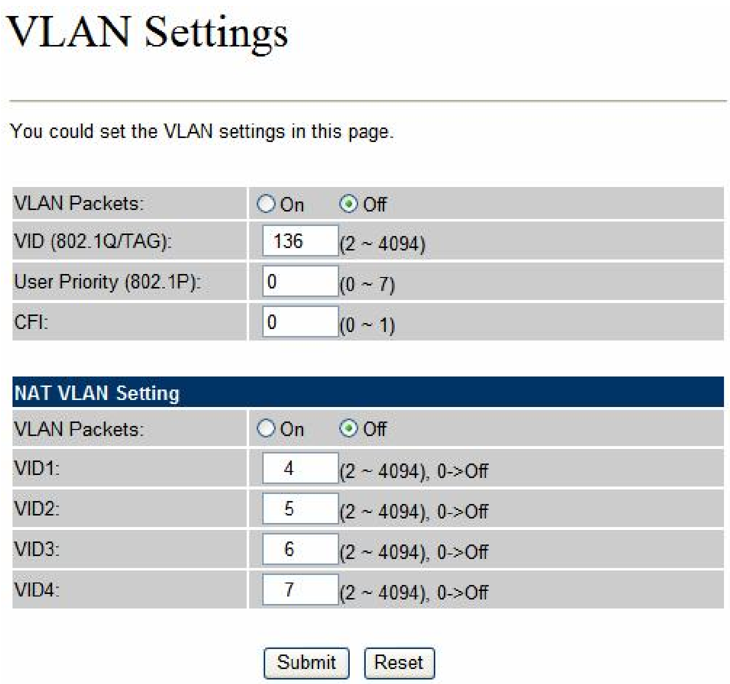
VLAN Packets: If setup as On, it could receive VLAN messages.

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.

CFI: To indicate the Canonical Format.

- If CFI=1, it means the header label include RIF field, and the NCIF flag valus of RIF will to decide the MAC address is Canonical Format or Non-Canonical Format in frame information.
- If CFI=0, it means the header label does not include RIF field, and the MAC address is Canonical Format in frame information.

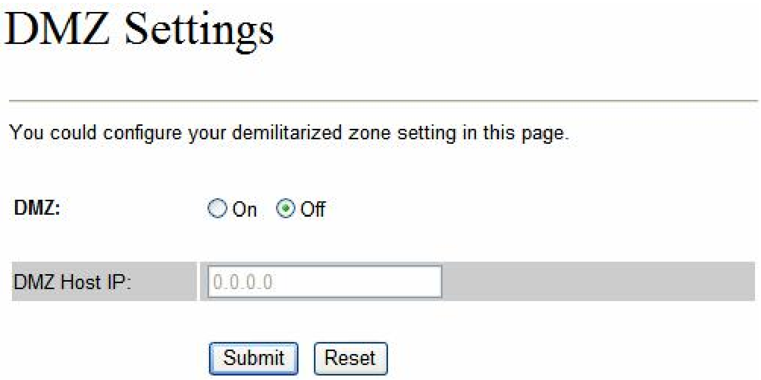


DMZ Settings

This page defines the DMZ setting in this page.

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

DMZ Host IP: The DMZ host IP address.



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.

Virtual Server Page: There are total page1 to page 3. It could choose the page which want to go over.

Num: The serial number. There are total 24 records from Num 0 to 23.

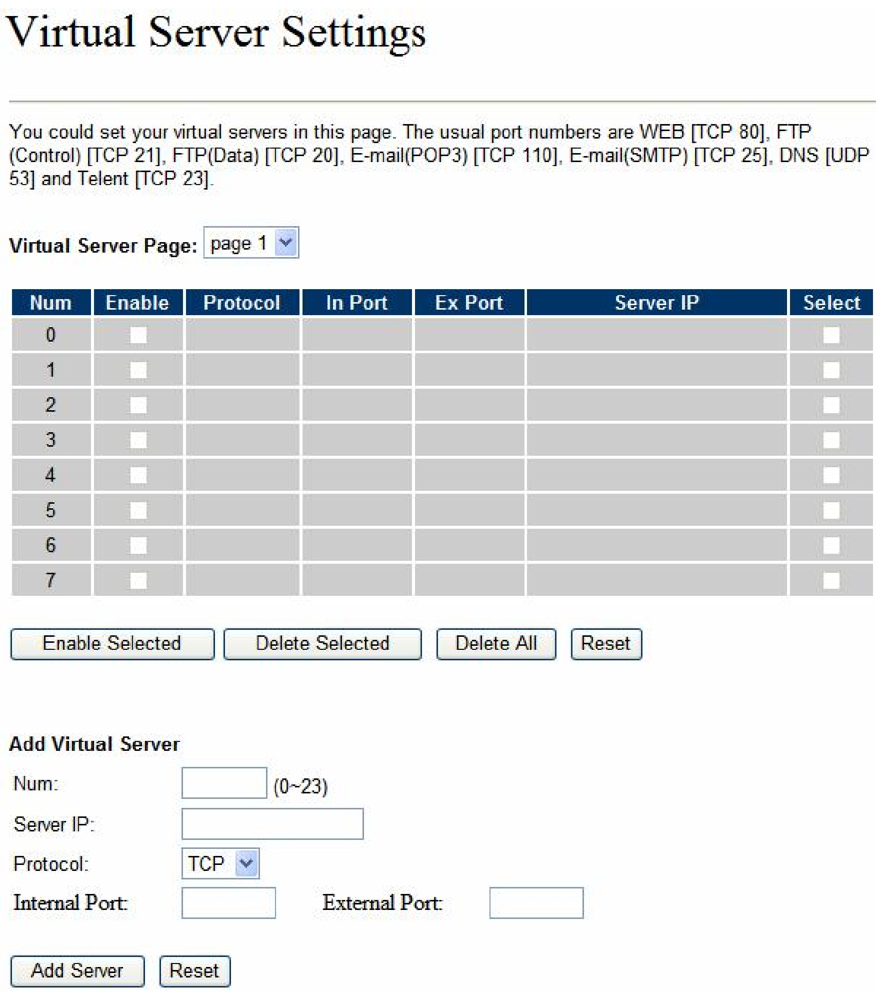
Enable: The activate status. The default is Disable, this record will been activate if enable.

Protocol: The TCP or UDP communication protocol.

Internal Port: For corresponding the internal port.

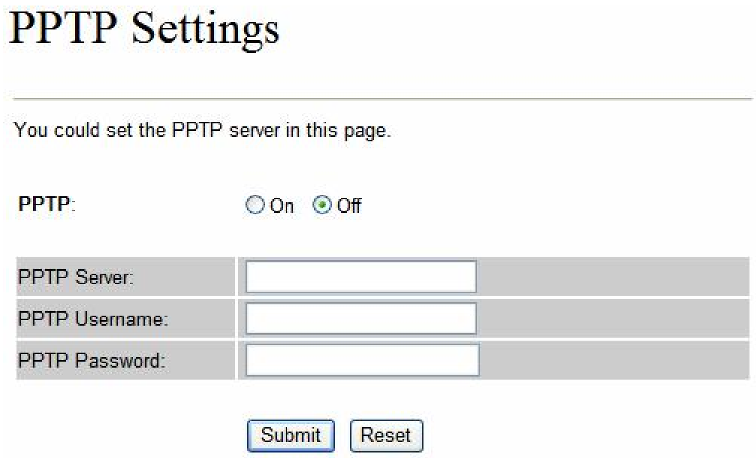
External Port: For corresponding the external port.

Server IP: To input the Server IP address.



PPTP Settings

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



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.

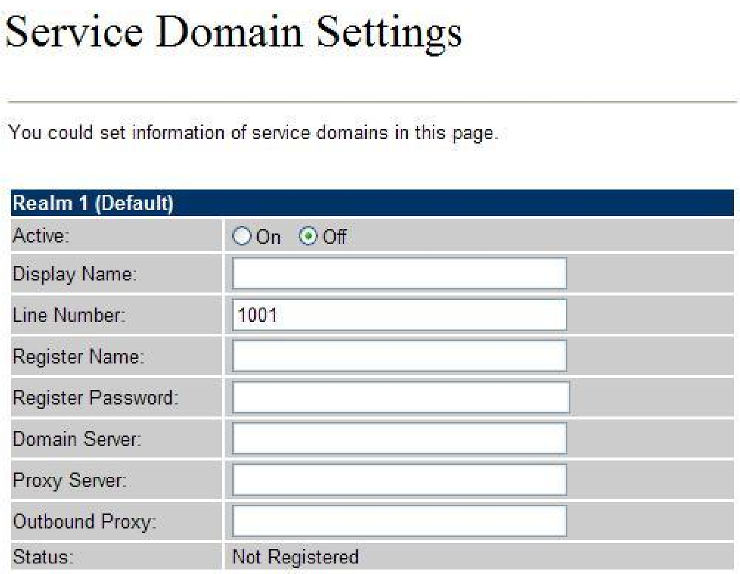
Proxy Server: you need to input the Proxy Server get from your ISP.

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 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.



i **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

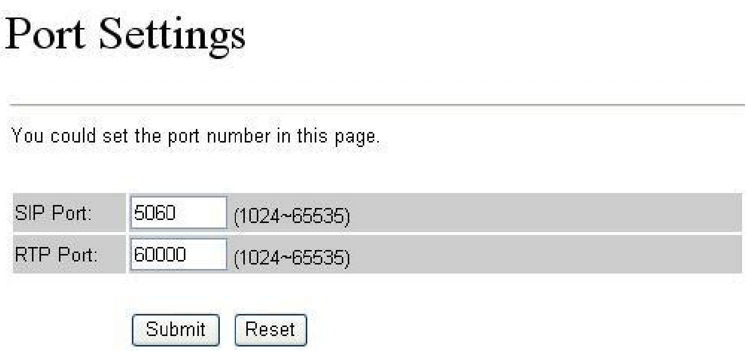
2*: Realm 2

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.

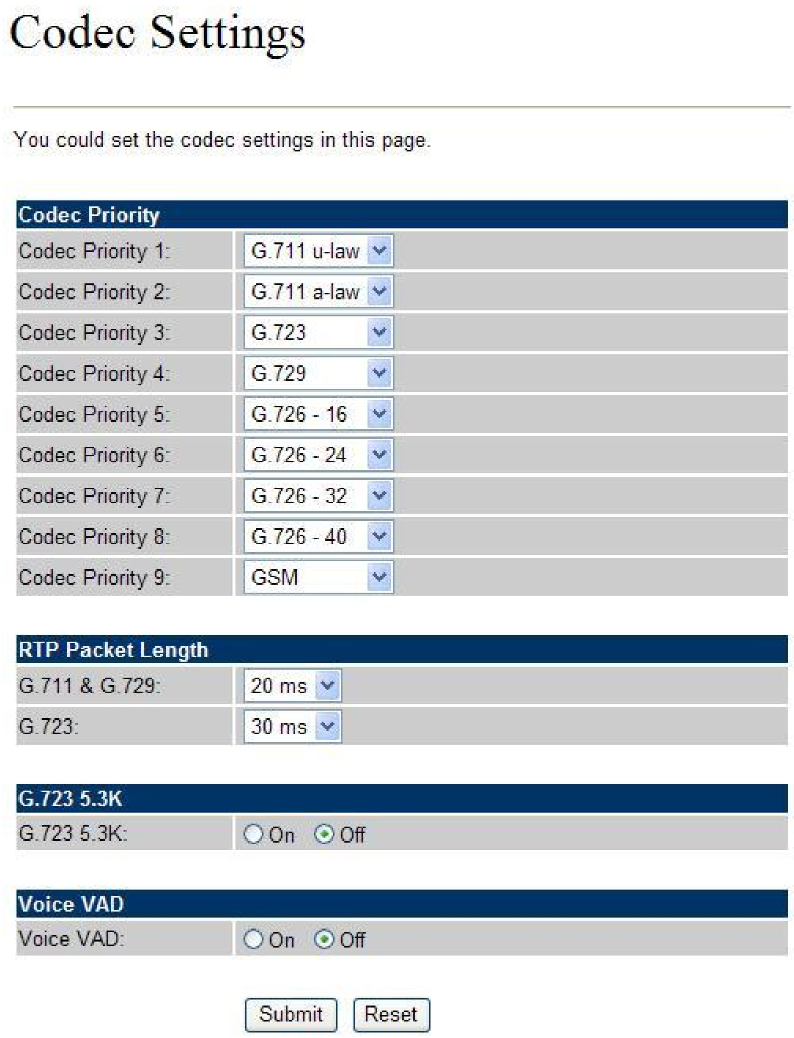
Port Settings

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.



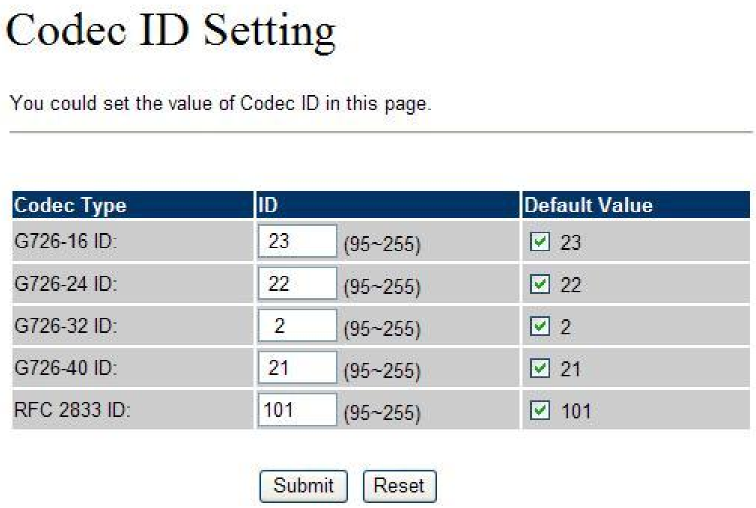
Codec Settings

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 ID Setting

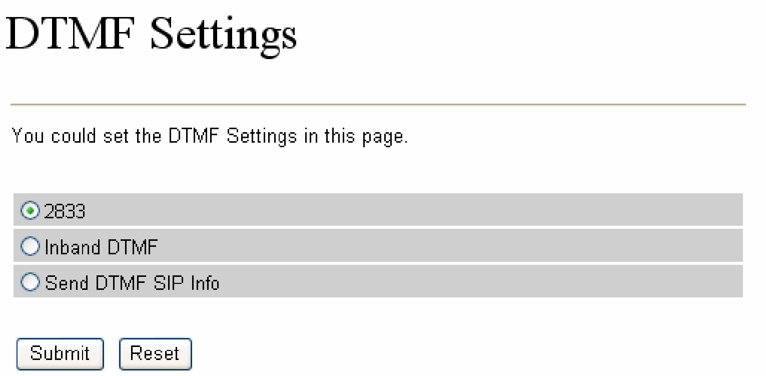
This page defines the Codec ID. Sometimes 2 VoIP devices with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, and then you can change your Codec ID. When you finished the setting, please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



Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	23
G726-24 ID:	22 (95~255)	22
G726-32 ID:	2 (95~255)	2
G726-40 ID:	21 (95~255)	21
RFC 2833 ID:	101 (95~255)	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.



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.

Other Settings

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.

Hold by RFC: The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.

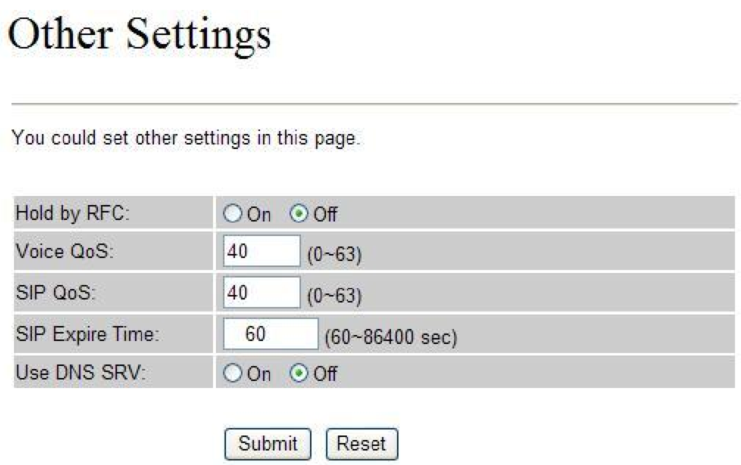
Voice QoS (Diff-Serv): The Voice QoS feature.

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.

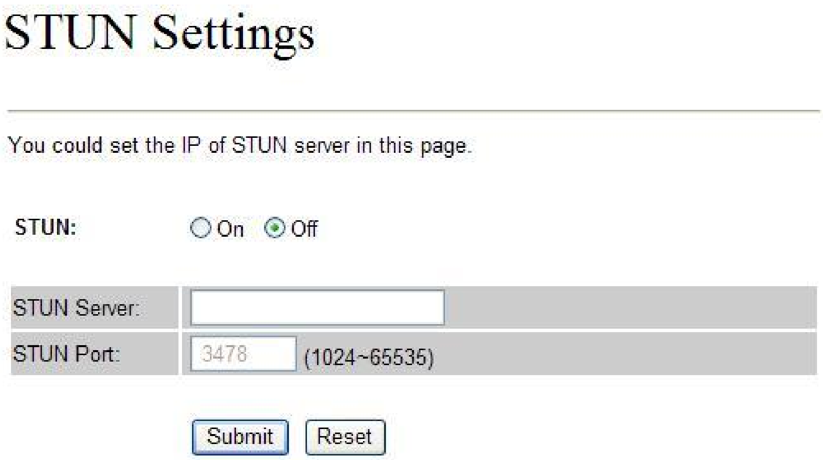
SIP Expire Time: To setup the registration interval time.

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.



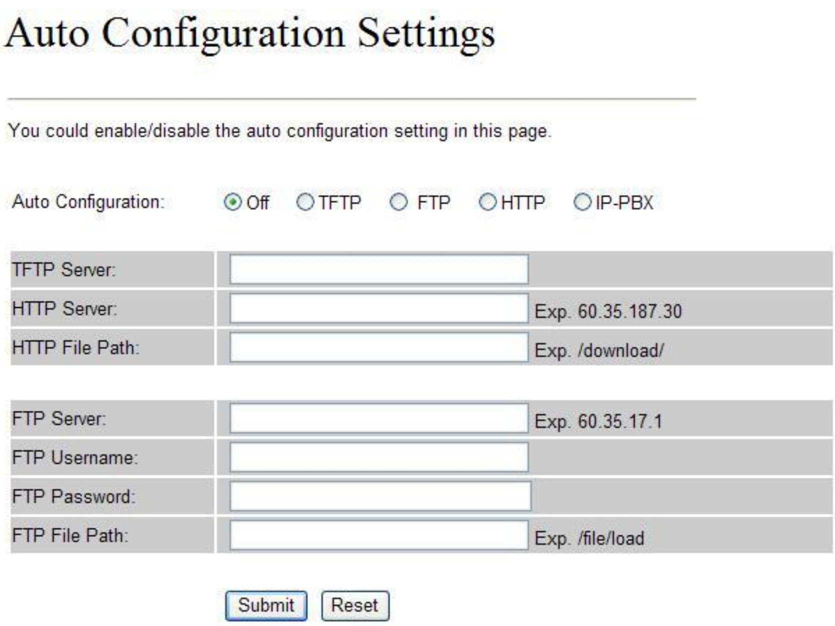
STUN settings

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.



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 boundle with an IP PBX to use in the office.



PTT Settings

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

***PTT Settings for* `G-157 Land Line Converter`**

Beside the above settings, **`G-157 Land Line Converter`** also can set country of FXO port.

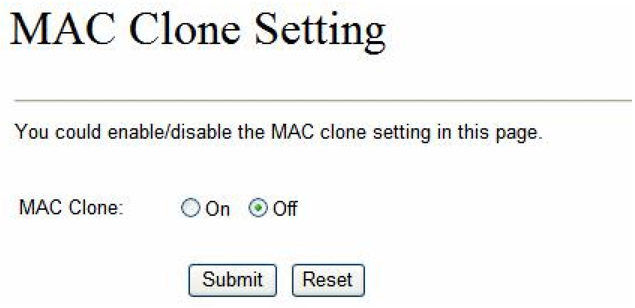
***PTT Settings for* `G-157 Land Line Converter`**

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.

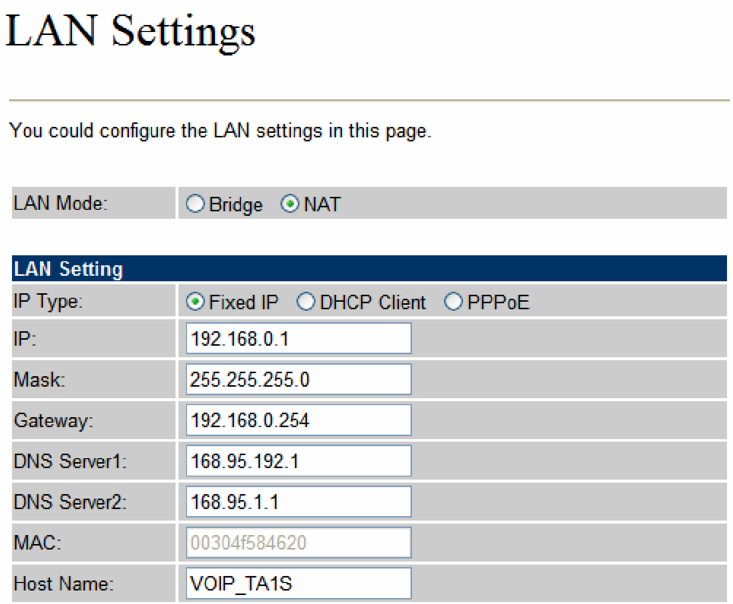
MAC Clone Setting

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.

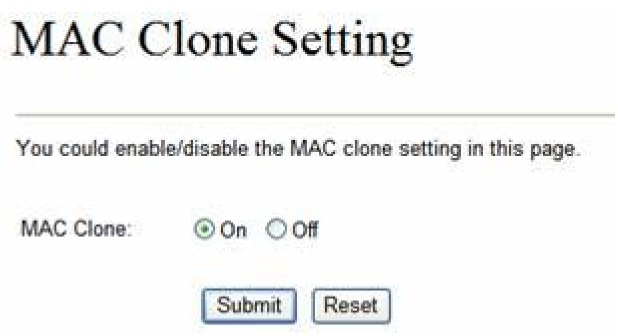


Please refer to the following operate procedures for more understandings to carry out the MAC Clone function.

1. Please login ATA and browse to "**Network -> LAN Settings**" page. To switch the LAN mode to **NAT** mode then press **Save&Reboot** button to save the settings and reboot machine.



2. Please make sure the network cable of your PC directly connect with PC port of ATA, then re-login ATA. (The default IP address of PC port is http://192.168.123.1)
3. Please browse to "**Advanced Settings -> MAC Clone Setting**" page and enable the MAC Clone function.



4. ATA will prompt if sure want to clone the MAC of your PC to the LAN port of ATA.

5. After **Save&Reboot**, the MAC of LAN port will become to PC's original MAC address.

LAN Setting	
IP Type:	⦿ Fixed IP ○ DHCP Client ○ PPPoE
IP:	192.168.0.1
Mask:	255.255.255.0
Gateway:	192.168.0.254
DNS Server1:	168.95.192.1
DNS Server2:	168.95.1.1
MAC:	00304f005743

Tone Settings

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

Tones Settings

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
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:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

Submit Reset

Advanced Settings

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

ICMP Not Echo: This function can disable echo when someone ping this device, it can avoid haker try to attack the device.

Send Anonymous CID: If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information. (The Registration Server also need support this function)

Billing Signal: There are provide three type billing types: Polarity Reversal, Tone_12K and Tone_16K.

(The Registration Server also need support this function)

CPC Delay: When receive the disconnect signal, machine will to cut the voltage 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.

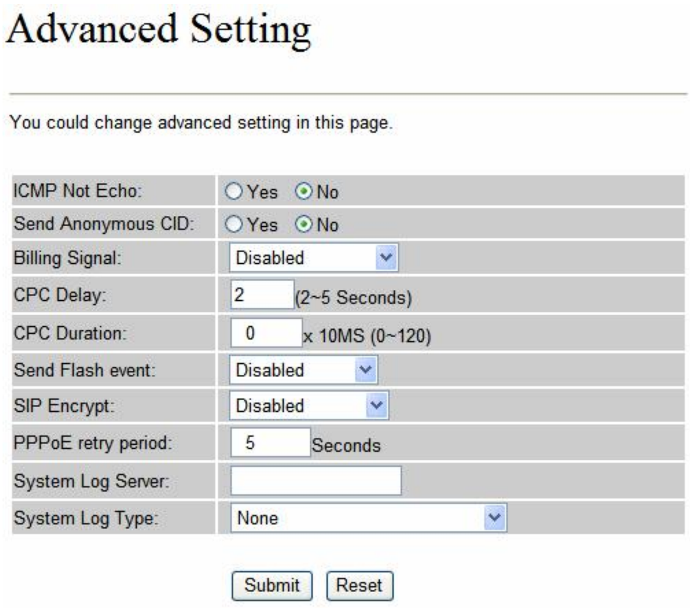
Send Flash event: There are provide two flash formats: DTMF Event and SIP Info.

SIP Encrypt: There are provide seven encrypt formats: INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2 and GX. (The Registration Server also need support this function)

PPPoE retry period: If PPPoE dial-up connection fail, machine will retry the dial-up motion after this time.

System Log Server: Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address.

System Log Type: There are seven Syslog types: Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug and All.



System Authority

In System Authority you can change your login password.

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.

Firmware Upgrade

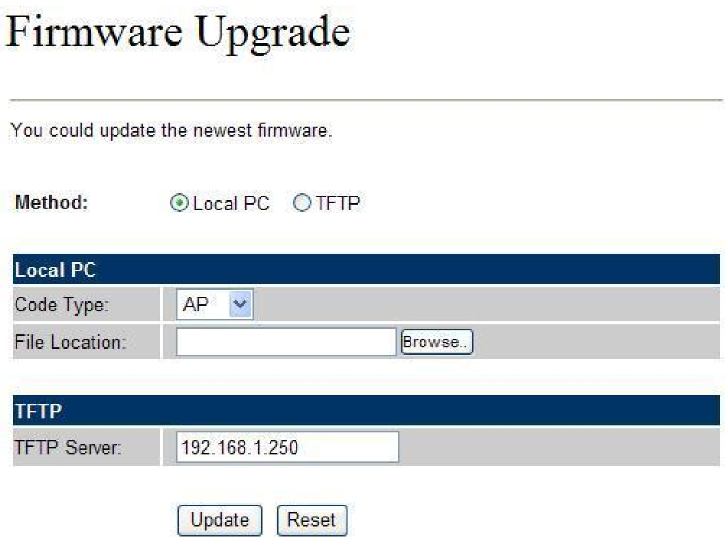
In Firmware Upgrade function you can update new firmware via HTTP or TFTP methods in this page.

You can ugrade 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.



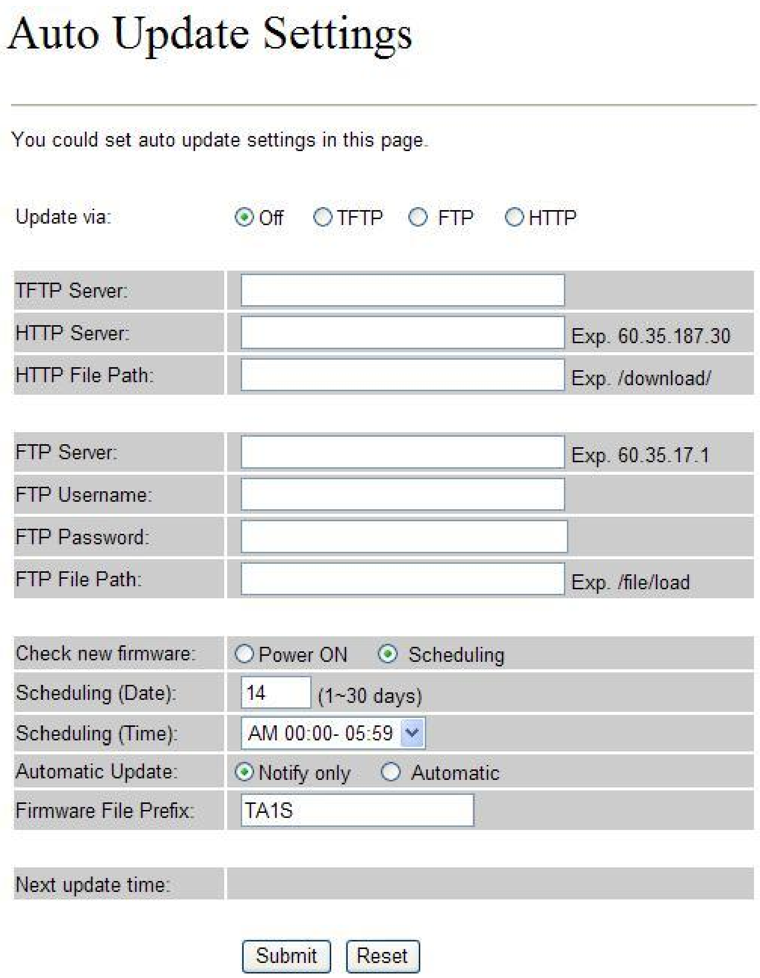
i **Note**:

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://192.168.123.1; login name/password: **admin/Gulfsip**

Auto Upgrade

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



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. **- 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.

i **Note**:

If the **Check new firmware** field selected to Power On, the machine will chck the new firmware accoeding 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.

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.

Appendix A 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 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.