VoIP Analog Telephone Adapter

ATA-150/ATA-150S

User's manual

Version 1.00

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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:

Model: ATA

Rev: 1.00 (2008, November)

Part No. EM-ATA150 Series

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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 ATA-150/ATA-150S VoIP Phone Adapter ("**ATA**" in the following term) converts standard telephones to IP-based networks. The service providers and enterprises can offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

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 equipped with two telephony interfaces, users may register to different SIP proxy servers, IP PBX and establish up to 2 concurrent VoIP calls for more flexibility in the voice communications. ATA can be the bridge between the traditional analog telephones to IP network with an extremely affordable investment.

Product Features

- Feature-rich telephone service over home Internet / Intranet connection
- Up to 2 concurrent VoIP calls
- Cost-effective, easy-to-use solution for Analog Telephone Adapter
- Web-based utility and machine configuration
- Remote administrator authentication
- Voice prompt for machine configurations

VoIP Features

- SIP 2.0 (RFC3261) compliant
- Voice codec: G.711(A-law / μ -law),G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)
- FoIP : T.38 FAX Relay, G.711 Fax pass-through
- QoS : IP TOS (IP Precedence) / DiffServ
- Call Waiting / Hold / Resume / Transfer / Forward /
- 3-Way Conference / Caller ID Generation
- VAD / CNG / Dynamic Jitter Buffer
- SNMP v1/v2, TR-069 and Auto Provision

Package Content

The contents of your product should contain the following items:

VoIP Telephone Adapter

Power adapter

Quick Installation Guide

User's Manual CD

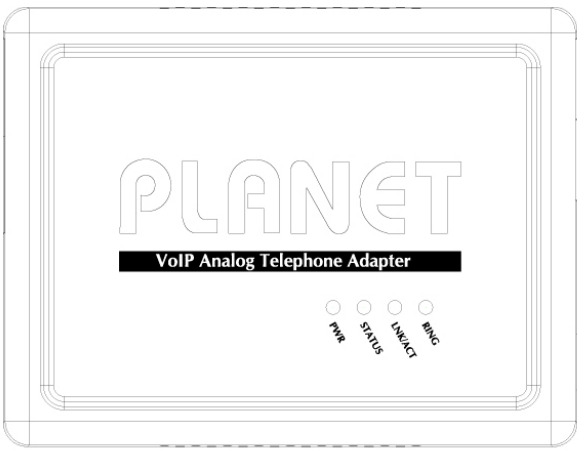
RJ-11 cable x 1

Physical Details

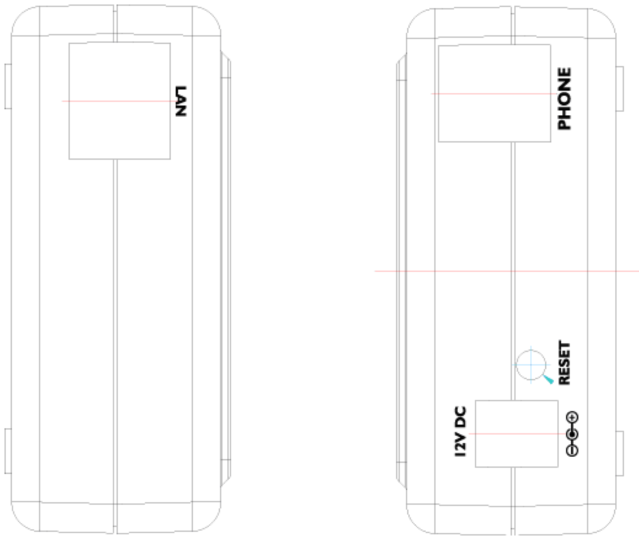
The following figure illustrates the each panel of SIP ATA.

ATA-150: SIP Analog Telephone Adapter (1 x RJ-45, 1 x RJ-11)

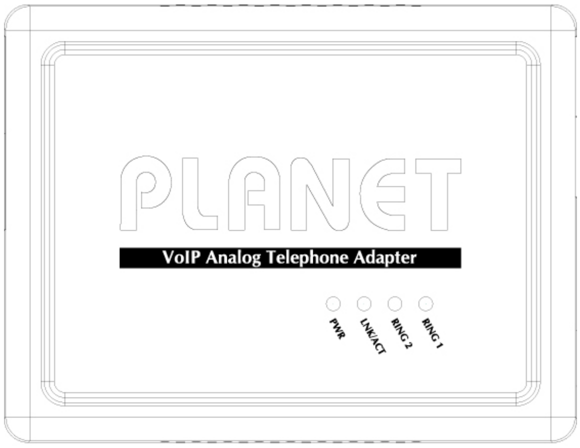
ATA-150S: 2-Port FXS SIP Analog Telephone Adapter (1 x RJ-45, 2 x RJ-11)



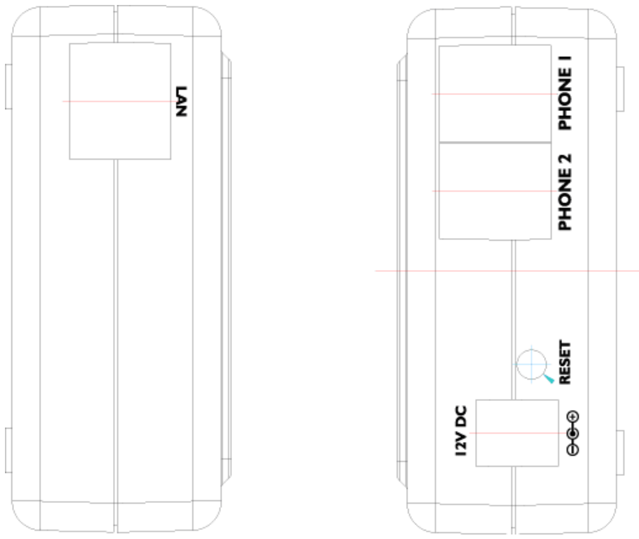
Front Panel of ATA-150



Left / Right Panel of ATA-150



Front Panel of ATA-150S



Left / Right Panel of ATA-150S

Physical Interface & Button

1	**RESET**	Reset to the factory default setting
2	**12V DC**	12V DC Power input outlet
3	**LAN**	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable.
4	**Phone**	RJ-11 connector, connected directly to the analog phone.

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 ATA-150

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 disconnected to LAN. **ON**: the device is connected to LAN.
RING	**OFF**: the phone is idle. **ON**: the phone is in use (off-hook). **Blinking**: the phone is ringing.

LED Display of ATA-150S

PWR	Power is supplied to the device.
LNK/ACT	**OFF**: the device is disconnected to LAN. **ON**: the device is connected to LAN.
RING1	**OFF**: the phone is idle. **ON**: the phone is in use (off-hook). **Blinking**: the phone is ringing.
RING2	**OFF**: the phone is idle. **ON**: the phone is in use (off-hook). **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 10/100Base-TX network (UTP) cables with RJ-45 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

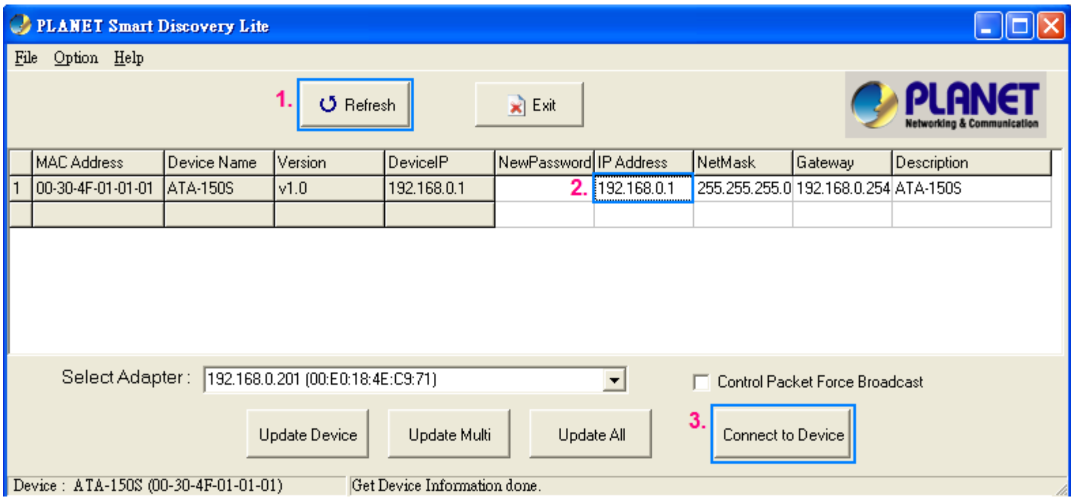
PLANET ATA provides GUI (Web based, Graphical User Interface) and utility for machine management and administration.

Utility quickly search access

Using for soft utility to search SIP ATA from current network. The utility not only easy-to-use and provides user more convenience for configuration access, at the some time If you forget this IP address can also found that via the utility.

Copy this utility tool in your laptop or desktop computer first. And, this utility tool can only be executed in Windows series of operating systems.

Click the icon for windows desktop to start searching ATA in the network.



Select "**Refresh**" and you will get the results as above choose the device you want to configuration, click this IP address of ATA and press the "**Connect to Device**" button to browse the web page.

Web configuration access

You will connect to SIP ATA via your web browser automatically. ATA will prompt for logon username / password, please enter: ***admin*** / **123** to continue machine administration.

ATA will prompt for logon username/password, please enter: ***admin*** / **123** to continue machine administration.

The default IP address of ATA is **192.168.0.1**. You also could open your web browser, and insert ***http://192.168.0.1*** in the address bar of your web browser to logon ATA web configuration page.

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

- Microsoft Internet Explorer 6.00 or higher with Java support

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.

Keypad commands

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

IVR Menu Choice	Machine operation	Parameter(s)	Notes
#111#	Set DHCP client	None	ATA will change to DHCP Client
#112xxx*xxx*xxx*xxx#	Setup Static IP Address	Use the * (star) key when entering a decimal	DHCP will be disabled and system will change to the

		point.	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.
#195#	Save Network Settings	None	Must save network settings after set up network settings via keypad.

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

IVR Menu Choice	Machine operation	Notes
#120#	Check IP Address	IVR will announce the current IP address of the ATA.
#121#	Check network connection Type	IVR will announce if DHCP in enabled or disabled.
#122#	Check the Phone Number	IVR will announce current enabled VoIP number.
#123#	Check Network Mask	IVR will announce the current network mask of the ATA.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP address of the ATA.
#125#	Check DNS Server Setting	IVR will announce the current setting in the DNS field.
#128#	Check Firmware Version	IVR will announce the version of the firmware running on the ATA.

Following keypad commands can be used to set up the main function .

IVR Menu Choice	Machine operation	Parameter(s)	Notes
#130+first priority codec	Set First Priority Codec	01: G.711 u-Law, 02: G.711 a-Law, 03: G.729, 04: G.723 6.3K, 05: G.723 5.3K, 06: G.726 16K, 07: G.726 24K, 08: G.726 32K, 09: G.726 40K, 10: GSM-FR	You can set the codec you want to the first priority. For example: **#13001#** Set G.711 u-Law to the first priority codec
#133#	Set Speaker Voice Gain	00~31, 32: Mute	For example: **#13305#** Mic Voice: 5
#134#	Set Mic Voice Gain	00~31, 32: Mute	For example: **#13410#** Mic Voice: 10
#138#	Enable call waiting	None	Enable Call waiting
#139#	Disable call waiting	None	Disable Call waiting
#140+Forward type+Forward Phone Number#	Forward Settings	Forward Type: 1: Immediate Forward 2: Busy Forward 3: No answer Forward	For example: **#1401101#** Immediate Forward to 101
#141#	Disable Forward Settings	None	
#150#	Select Default Realm	0: Realm 1, 1: Realm 2	For example: **#1501#** Set Default Proxy to Realm 2
#160#	Update firmware	None	Update firmware

Chapter 3

TCP/IP Settings

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

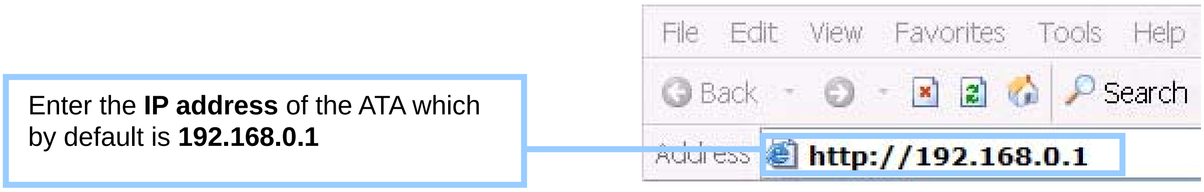
To start ATA web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.00 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

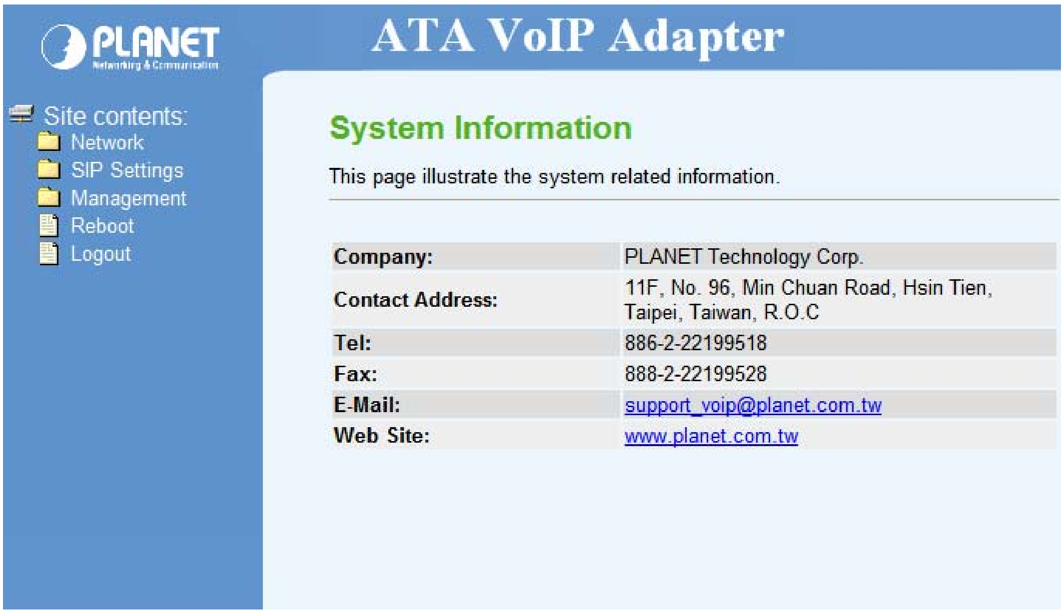
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon Phone Adapter web configuration page.



Phone Adapter will prompt for logon username/password: ***admin*** / ***123***

ATA login prompt screen

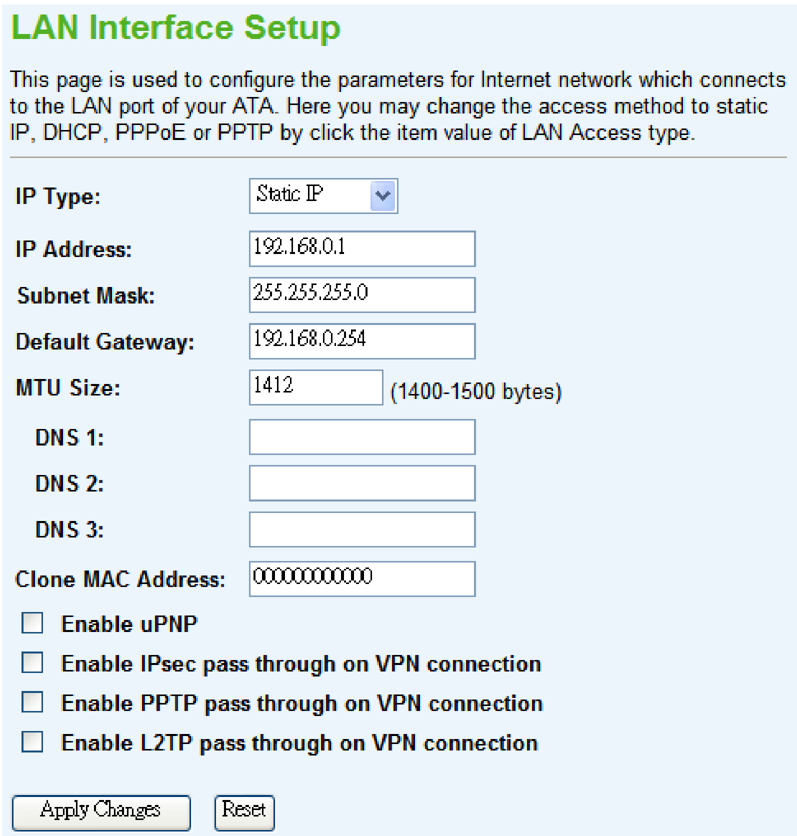
When users login the web page, users can see the Phone Adapter system information like firmware version, company…etc in this main page.



VoIP Phone Adatper main page

LAN IP address configuration via web configuration interface

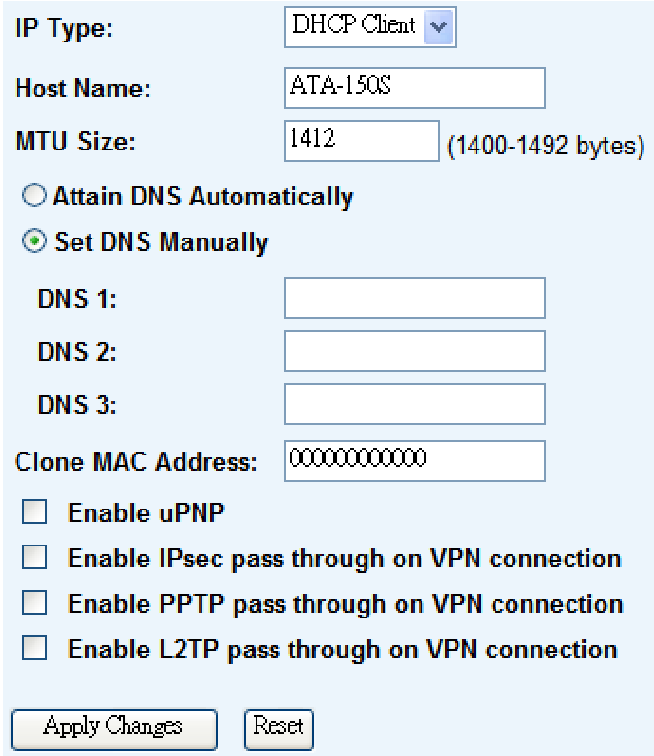
This page is used to configure the parameters for Internet network which connects to the LAN port of your ATA. Here you may change the access method to static IP, DHCP, PPPoE or PPTP by click the item value of LAN Access type.



Connection Type Description – Static IP

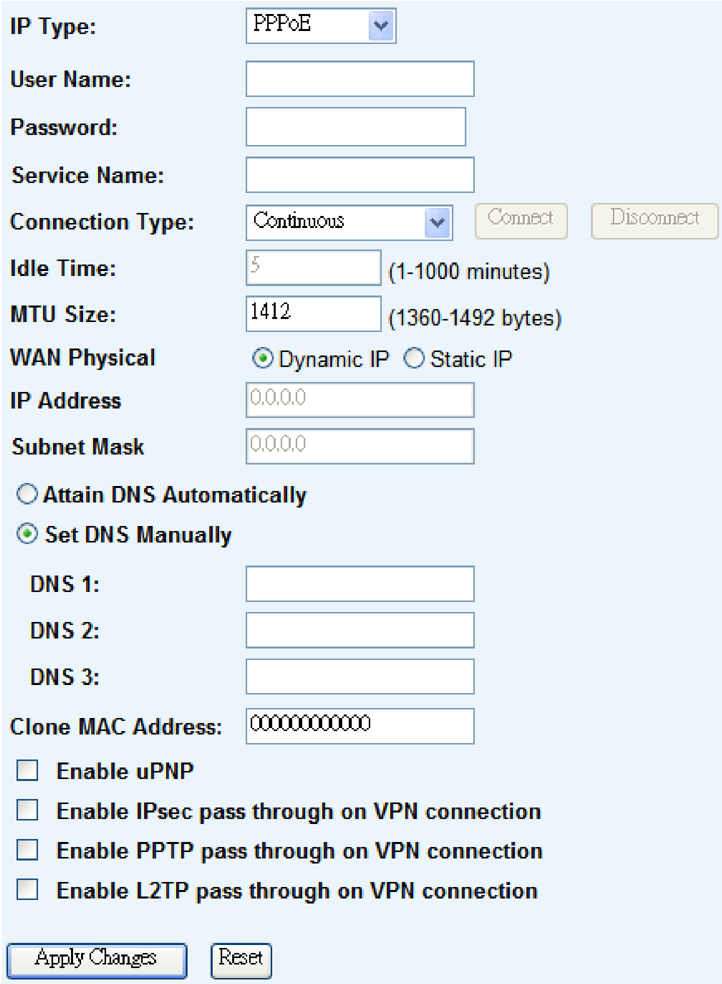
Static IP	Set LAN interface as Static IP mode.
IP Address	LAN IP Address of the ATA **Default : 192.168.0.1**
Subnet Mask	LAN mask of the ATA **Default : 255.255.255.0**
Default Gateway	Gateway of the ATA **Default :** 192.168.0.254
MTU Size	Set MTU (maximum transmission unit) size **Default : 1412**
DNS1/ 2/ 3	Set three alternative Domain Name Server for LAN interface. **Default : Null**
Clone MAC Address	To clone the MAC by manual input. **Default : 000000000000 (Null)**
Enable uPnP	Check to enable UPnP function **Default :** Disable
Enable IPsec pass through on VPN connection	Check to enable IPsec function **Default :** Enable
Enable PPTP pass through on VPN connection	Check to enable PPTP pass through function **Default :** Enable
Enable L2TP pass through on VPN connection	Check to enable L2TP pass through function **Default :** Enable

Connection Type Description – DHCP Client



DHCP Client	Set LAN interface as DHCP mode.
Attain DNS Automatically / Set DNS Manually	Select to attain DNS automatically from server or user wants to set DNS manually. **Default :** Set DNS Manually

Connection Type Description – PPPoE

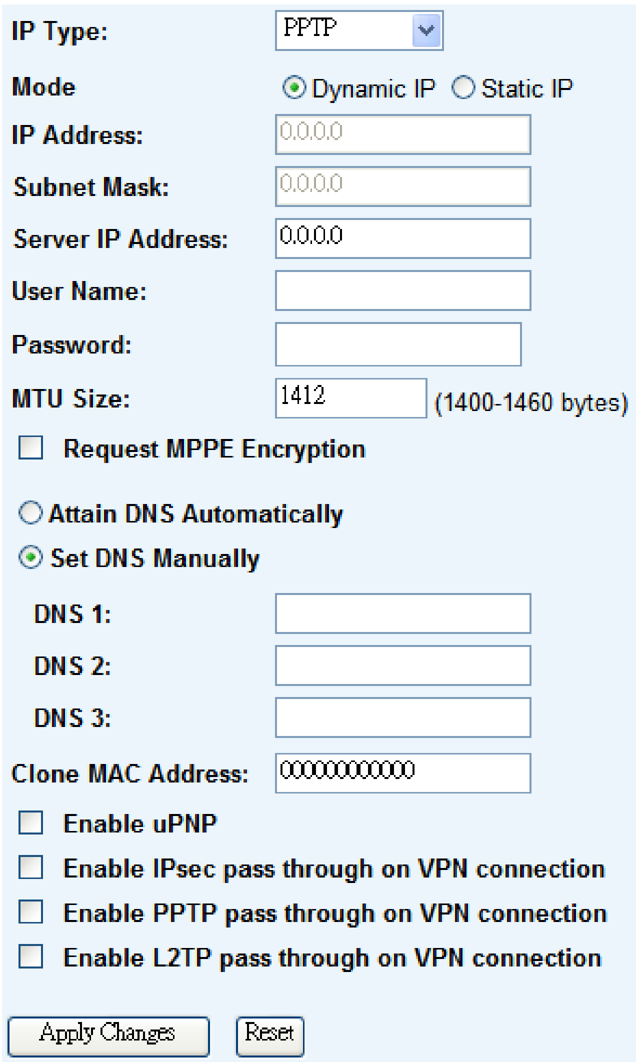


PPPoE	Set LAN interface as PPPoE mode.
User Name	Set user name of PPPoE connection **Default :** Null
Password	Set password of PPPoE connection **Default :** Null
Service Name	Set Service Name of PPPoE for description **Default :** Null
Connection Type	Set PPPoE connection type to be Continuous/ Connect on Demand/ Manual. If user set type as Continuous, ATA will keep trying to connect to server when PPPoE disconnect. If user set type as Connect on Demand, please set following idle time, ATA will check connection after this time. If user set type as Manual, ATA will only connect or disconnect by press Connect or Disconnect manually. **Default :** Continuous

Idle Time	Set PPPoE connection idle time for Connect on Demand.
	Default : 5
LAN Physical	Set IP type if **Dynamic IP** or **Static IP** at PPPoE connection.
	Default : Dynamic IP
IP Address	LAN IP Address of the ATA at **Static IP** type.
	Default : 0.0.0.0
Subnet Mask	LAN Mask of the ATA at **Static IP** type.
	Default : 0.0.0.0

After confirming the modification you've done, please click on the **SUBMIT** button to apply settings effective and the ATA will be reload page automatic by itsely, that you must to afresh enter the final modification IP address for logon web management.

Connection Type Description – PPTP



PPTP	Set LAN interface as PPTP mode.	
Mode	Set IP type if Dynamic IP or Static IP at PPTP connection.	
	Default : Dynamic IP	
IP Address	LAN IP Address of the ATA at **Static IP** type.	
	Default : 0.0.0.0	
Subnet Mask	LAN Mask of the ATA at **Static IP** type.	
	Default : 0.0.0.0	
Server IP Address	Set PPTP Server IP address.	
	Default : 0.0.0.0	
User Name	Set user name of PPTP connection	
	Default : Null	
Password	Set password of PPTP connection	
	Default : Null	

Note

Please be noticed that the Utility Tool is only designed for the LAN environment settgin. If the "**Connect Type**" is "**PPPoE**", the Utility Tool can **NOT** find the device.

Chapter 4

VoIP Settings

Phone 1 / Phone 2 (ATA-150S)

Here is to set VoIP Phone 1 and Phone 2 (ATA-150S) related configurations.

- Default Proxy

Default Proxy	
Select Default Proxy	Realm 1

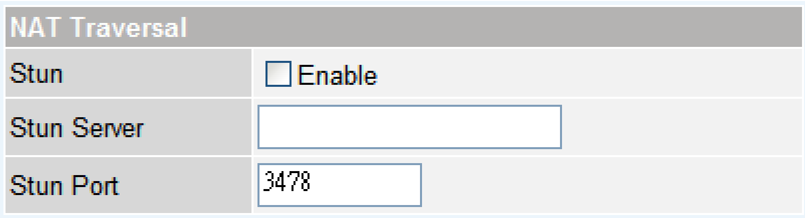
Select Default Proxy	Each Phone port has support register two different Proxy Servers. When select one of Proxy as defult, ATA will use this account for making outgoing call. And ATA could receive incoming calls through both Proxys.
	Default : Realm1

- Realm 1 / Realm 2

Realm 1	
Display Name	
Line Number	
Register Name	
Register Password	
Proxy	☐ Enable
Proxy Server	
Proxy Port	5060
Domain Server	
SIP Expire Time	60
Outbound Proxy	☐ Enable
Outbound Proxy Server	
Outbound Proxy Port	5060
Nortel SoftSwitch	☐ Enable
Register Status	Not Registered

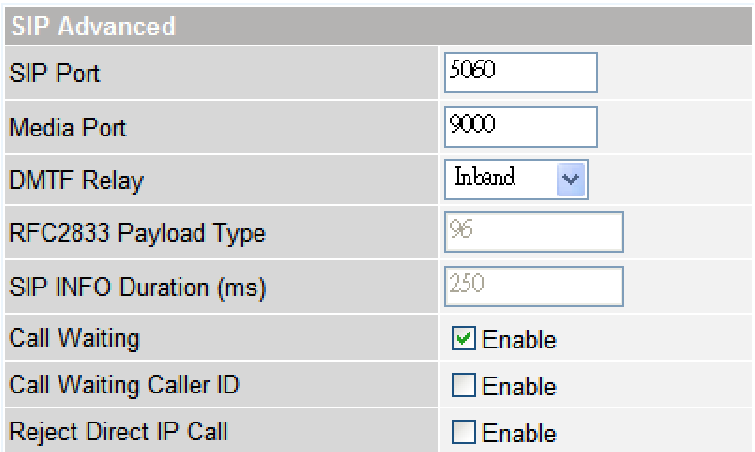
Display Name	Set ATA Phone display name for caller ID information. **Default :** Null
Number	Set registering Phone number. **Default :** Null
Login ID	If Proxy server needs registration authentication please input Login ID here. **Default :** Null
Password	If Proxy server needs registration authentication please input password here. **Default :** Null
Proxy	Check to enable Proxy mode. **Default :** Disable
Proxy Addr	If user enable Proxy mode, please input Proxy address. **Default :** Null
Proxy Port	If user enable Proxy mode, please input Proxy port. **Default :** 5060
SIP Domain	Set SIP domain name for SIP signaling. **Default :** Null
Reg Expire (sec)	Set expire time of registration. ATA will keep re-registering to proxy server before expire timed out. **Default :** 60
Outbound Proxy	Check to enable Outbound Proxy mode. **Default :** Disable
Outbound Proxy Addr	If user enables Outbound Proxy, please input Outbound Proxy address. **Default :** Null
Outbound Proxy Port	If user enables Outbound Proxy, please input Outbound Proxy port. **Default :** 5060
Register Status	Here will display SIP account register status.

- NAT Traversal



Stun	Check to enable STUN function.
	Default : Disable
Stun Server Addr	If user enables STUN function, please input STUN Server address.
	Default : Null
Stun Server Port	If user enables STUN function, please input STUN Server port.
	Default : 3478

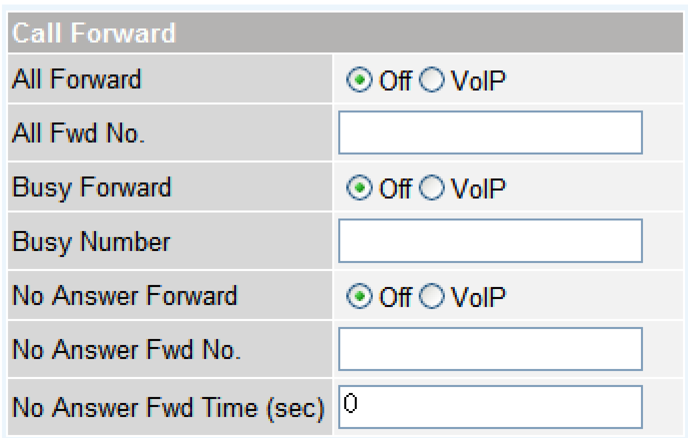
- SIP Advanced



SIP Port	Set local SIP listening port.
	Default : 5060
Media Port	Set RTP port for sending voice data.
	Default : 9000
DMTF Relay	Select DTMF Relay to be In band, RFC 2833, or SIP INFO.
	Default : Inband
RFC2833 Payload Type	If user select DTMF as RFC 2833 type, here can modify RFC 2833 payload type.
	Default : 96

SIP INFO Duration (ms)	If user select DTMF as SIP INFO type, here can modify SIP INFO duration. ATA will send out DTMF as this duration.
	Default : 250
Call Waiting	Check to enable Call Waiting function.
	Default : Enable
Call Waiting Caller ID	Check to enable call waiting caller ID function. If this function is enabled, caller ID will display when having waiting call. Please note that your phone set should also support such function.
	Default : Disable
Reject Direct IP Call	Check to enable Reject Direct IP Call. If this function is enabled, ATA will to reject the incoming peer to peer call.
	Default : Disable

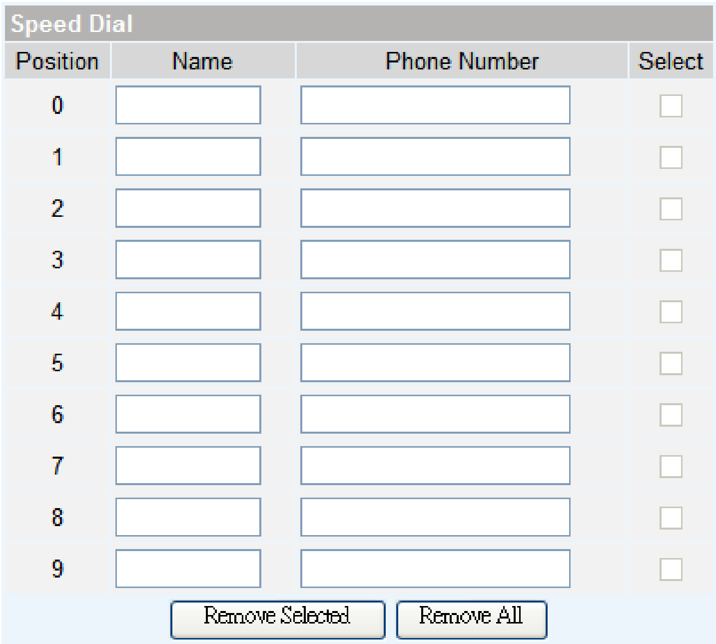
- Forward Mode



Immediate Forward to	This is unconditional forward setting. All incoming call will be forwarded to specified number. Check to enable immediate forward function.
	Default : Off
Immediate Number	Enter the assigned number for Immediate forward.
	Default : Null
Busy Forward to	Check to enable Busy Forward function. When phone is busy, incoming call will be forwarded to assigned number.
	Default : Off
Busy Number	Enter the assigned number for busy forward.
	Default : Null

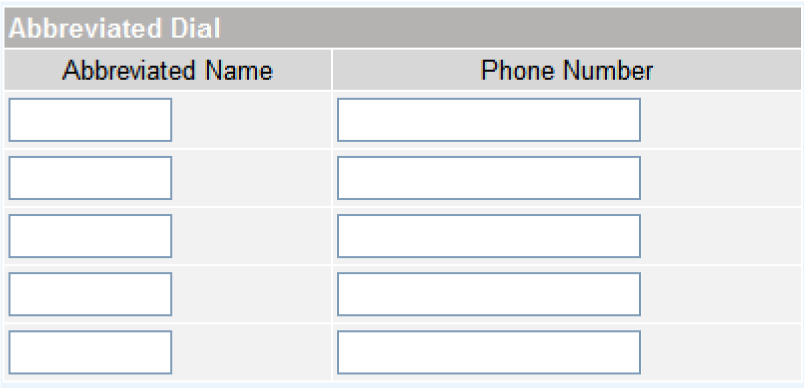
No Answer Forward to	Check to enable no answer forward function. When phone is not answered for a period of time, incoming call will be forwarded to assigned number. **Default :** Off
No Answer Number	Enter assigned number for no answer forward. **Default :** Null
No Answer Time (sec)	Set no answer time. Once phone is not picked up after this time, incoming call be will forwarded to assigned number. **Default :** 0

- Speed Dial



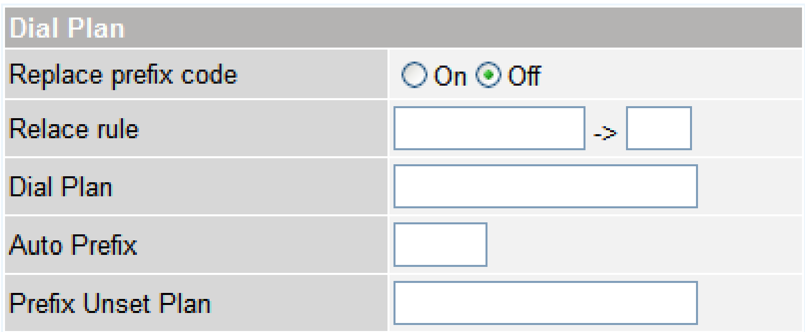
Position	Speed Dial access code. Press this speed dial number and followed by # can dial out assigned phone number.
Name	Name of this speed dial.
Phone Number	Set phone number for ATA to make speed dial.
Select	User can delete selected speed dial data.

- Abbreviated Dial (Phonebook)



Abbreviated Name	Abbreviated Dial (Phonebook) access code. Input this number and followed by # can dial out assigned phone number.
Phone Number	Set phone number for ATA to make speed dial.

- Dial Plan



Replace prefix code	Select to enable (On) or disable (Off) prefix replace function.
	Default : Off
Relace rule	Set prefix replace rule. Once user dial number matched prefix, ATA will replace the number with assigned number. Available parameters are “**0~9**”, “**#**”, “*****”, “**+**”, “**x**”. Symbol “**+**” means “**or**” , “**x**” could be numbers **0~9**. For example, if user set Replace rule as **002+009->005**, which means if user dial 002 87654321 or 009 87654321, these number will be dial out as 005 87654321.
	Default : Null
Dial Plan	User can set how many digits or which number for ATA to dial out immediately. Available parameters are “**0~9”**, “**#”**, “***”**, “**+”**, “**x”**. Symbol “**+**” means “**or**” , “**x**” could be numbers “**0~9”**. For example, user can set Dial Plan as “**911+xxxxxxxx+*xx**, which means if user dial **911**, **87654321**, or ***11**, these number will be dial out immediately without waiting for dial time or pressing # sign.
	Default : Null

Auto Prefix	If user set Auto Prefix number, all number dialed out will be added with this prefix number. Available parameters are “**0~9**”, “**#**”, “*****”.For example, user set Auto Prefix as 02, number 87654321 will be dial out as 02 87654321.
	Default : Null
Prefix Unset Plan	User can set special access code to disable Auto Prefix function in single call. Available parameters are “**0~9**”, “**#**”, “*****”, “**+**”, “**x**”. Symbol “**+**” means “**or**” , “**x**” could be numbers “**0~9**”. For example, if user set Prefix Unset Plan as *1+xxxxxxxxxx. When dialed number as *1 87654321 or 10 digits of number, for this call will not be added with Auto Prefix number.
	Default : Null

- Codec

Codec										
Type	Precedence									Mode
	1	2	3	4	5	6	7	8	9	
G711-ulaw	☑	☐	☐	☐	☐	☐	☐	☐	☐	
G711-alaw	☐	☑	☐	☐	☐	☐	☐	☐	☐	
G729	☐	☐	☑	☐	☐	☐	☐	☐	☐	
G723	☐	☐	☐	☑	☐	☐	☐	☐	☐	6.3k
G726-16k	☐	☐	☐	☐	☑	☐	☐	☐	☐	
G726-24k	☐	☐	☐	☐	☐	☑	☐	☐	☐	
G726-32k	☐	☐	☐	☐	☐	☐	☑	☐	☐	
G726-40k	☐	☐	☐	☐	☐	☐	☐	☑	☐	
GSM-FR	☐	☐	☐	☐	☐	☐	☐	☐	☑	

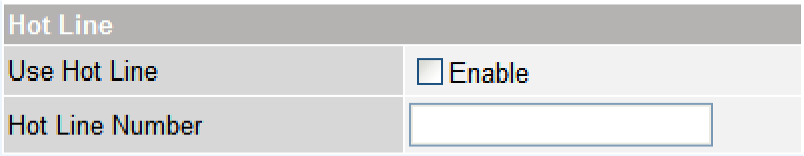
Precedence	Set codec priority sequence.
Rate	Set G.723.1 codec with 5.3 or 6.3k mode.

- T.38 (FAX)

T.38(FAX)	
T.38	☐ Enable
T.38 Port	9008
Fax Modem Detection Mode	AUTO

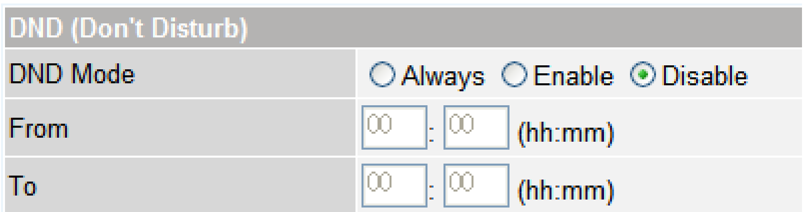
T.38	Check to enable T.38 function.
	Default : Disable
T.38 Port	Set T.38 port for FAX.
	Default : 9008

- Hot Line



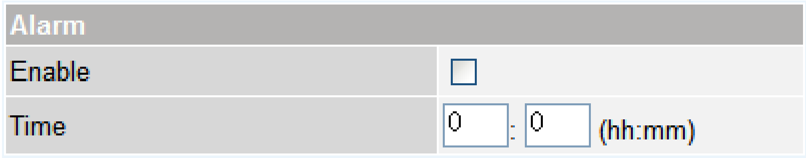
Use Hot Line	Hot Line Number
	Default : Disable
Hot Line Number	Set the destination number for Hot Line function.
	Default : Null

- DND (Don't Disturb)



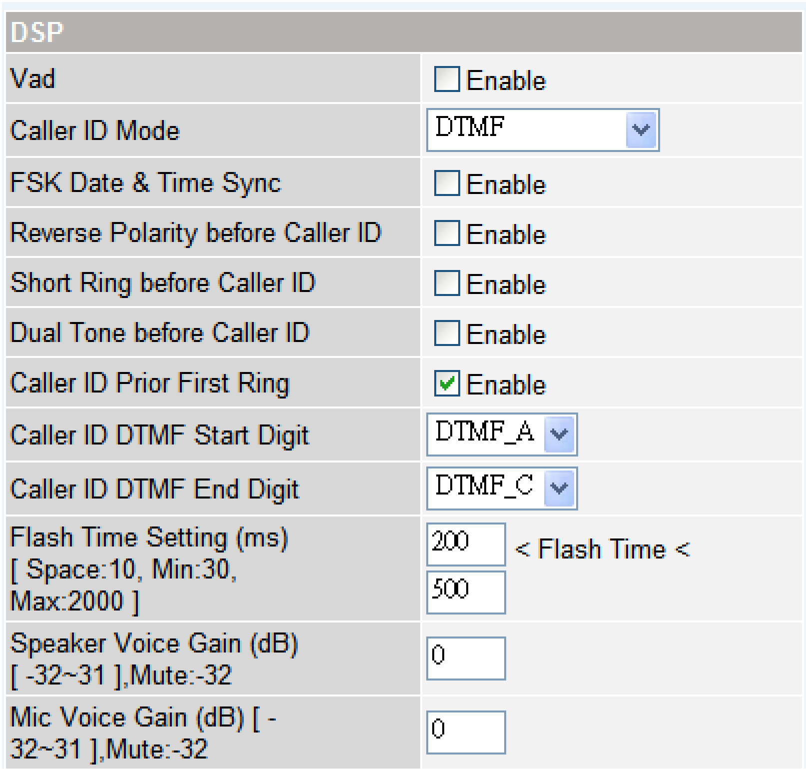
DND Mode	You can select 3 mode of DND. The call will be always rejected if Always is selected. The call will be rejected by below Time setting (From and To) if Enable is selected. The call will be accepted if Disable is selected.
	Default : Disable
From	Set the start time for DND with Enable mode.
	Default : 00:00
To	Set the end time for DND with Enable mode.
	Default : 00:00

- Alarm



Enable	If set up as Enable, the telephone will ringed up at the specific time. **Default :** Disable
Time	It can set up the system prompt time with 24 hours. **Default :** 0:0

- DSP



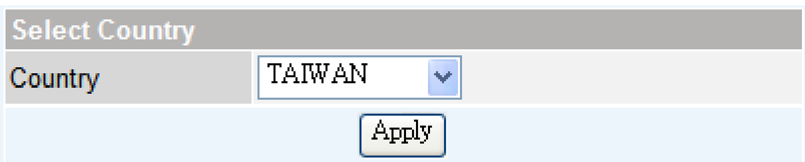
Vad	Check to enable VAD (Voice Activity Function) function. **Default :** Disable
Caller ID Mode	Select caller ID mode as FSK(Bellcore), FSK(ETSI), FSK(BT), FSK(NTT), or DTMF from Phone to send out. **Default :** DTMF
FSK Date & Time Sync	Check to send FSK Date and Time to caller ID display device. **Default :** Disable

Reverse Polarity before Caller ID	Check to send reverse polarity before caller ID. **Default :** Disable
Short Ring before Caller ID	Check to send short ring before caller ID. **Default :** Disable
Dual Tone before Caller ID	Check to send dual tone before caller ID. **Default :** Disable
Caller ID Prior First Ring	Check to send caller ID before first ring. **Default :** Enable
Caller ID DTMF Start Digit	Set caller ID DTMF start digit. **Default :** DTMF_A
Caller ID DTMF End Digit	Set caller ID DTMF end digit. **Default :** DTMF_C
Flash Time Setting (ms) [Space:10, Min:30, Max:2000]	Set Minimum and Maximum Flash time. **Default :** 200 ~ 500
Speaker Voice Gain (dB) [-32~31],Mute:-32	Set Speaker voice volume. **Default :** 0
Mic Voice Gain (dB) [-32~31],Mute:-32	Set microphone voice gain volume. **Default :** 0

Tone

User can adjust the items of the "Call Control" when in VoIP communication. And, basically system will use the following default setting values if user does not want to change them.

- Select Country

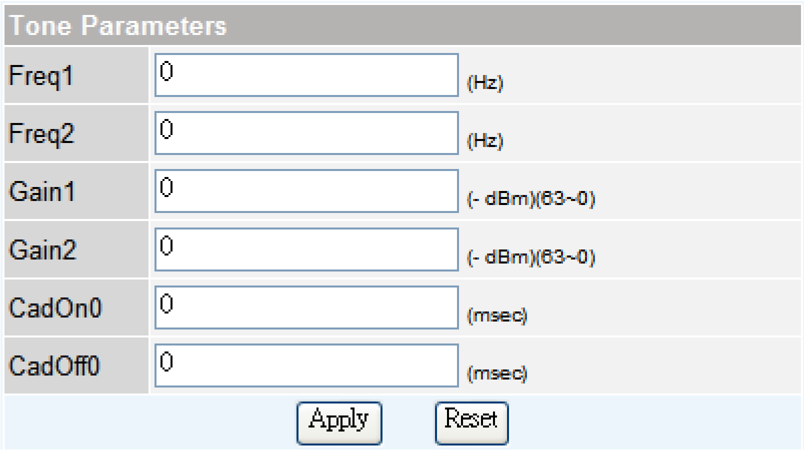


Country	User can select country to specify tone parameters (Dial Tone, Ring Tone, Busy Tone, and Waiting Tone). If user wants to set tone manually, please select CUSTOMER. After selecting CUSTOMER, user can assign Custom 1 to 8 for each tone. **Default :** TAIWAN

- Select Country

Custom Tone	Select Custom tone number to set Tone Parameters.
	Default : Custom1

- Tone Parameters



Freq1	Set first set of tone frequency in Hz.
	Default : 0
Freq2	Set second set of tone frequency in Hz. This frequency is optional.
	Default : 0
Gain1	Set volume level of Freq1 in dB (-7~-10). Please set this parameter under zero and suggested to set between –7 to –10.
	Default : 0
Gain2	Set volume level of Freq2 in dB (-7~-10). Please set this parameter under zero and suggested to set between –7 to –10.
	Default : 0
CanOn	Set cadence time for tone to play in ms. For example, if set CanOn as 100, the tone will be played for 100ms.
	Default : 0
CanOff	Set cadence time for tone not to play in ms. For example, if set CanOff as 100, the tone will stop playing for 100ms.
	Default : 0

Other

- Function Key

Function Key	
Must be * + 0~9	
Call Transfer	*1 (default: *1)

Call Transfer	Set call transfer function key.
	Default : *1

- Dial Option

Dial Option	
Auto Dial Time	5 (3~9 sec, 0 is disable)

Auto Dial Time	Set Auto dial time. When user finish input number after this time, ATA will dial out immediately. If the call is ended by "#", the call will be send innediately and you do not need to wait fot the Auto Dial time.
	Default : 5

- Off-Hook Alarm

Off-Hook Alarm	
Off-Hook Alarm Time	30 (10~60 sec, 0 is disable)

Off-Hook Alarm Time	Set off-hook alarm time. If phone set has been off-hook, after this time, from phone sett will hear alarm.
	Default : 30

- QoS

You can define the DSCP code here for SIP and RTP. Higher DSCP, higher priority.

When DSCP is defined, a DSCP will be added in SIP and RTP packets, and the priority of voice should be higher than data.

QoS	
SIP DSCP	EF (DSCP 0x2e)
RTP DSCP	EF (DSCP 0x2e)

Auto Config

- Auto Config

ATA supports HTTP, TFTP and FTP auto configuration function in total.

Auto Config	
Mode	○ HTTP ○ TFTP ○ FTP ◉ Disable
HTTP Server Address	
HTTP Server Port	80
TFTP Server Address	
FTP Server Address	
FTP Username	
FTP Password	
File Path	
Expire Time	0 days
Apply Changes	Reset

- Auto Firmware Update

The ATA can update new firmware file automatically by the Auto Firmware Update function.

Auto Firmware Update	
Mode	○ TFTP ○ FTP ○ HTTP ◉ Off
TFTP Server Address	
HTTP Server Address	
HTTP File Path	
FTP Server Address	
FTP Username	
FTP Password	
FTP Path	
Check new firmware	○ Power On ◉ Scheduling
Schuduling Day	0 (1~ 30 days)
Schuduling Time	AM 00:00~05:59
Auto Update	○ Automatic ◉ Notify Only
File Prefix	
Next Update Time	Off
Firmware Version	
Apply Changes	Reset

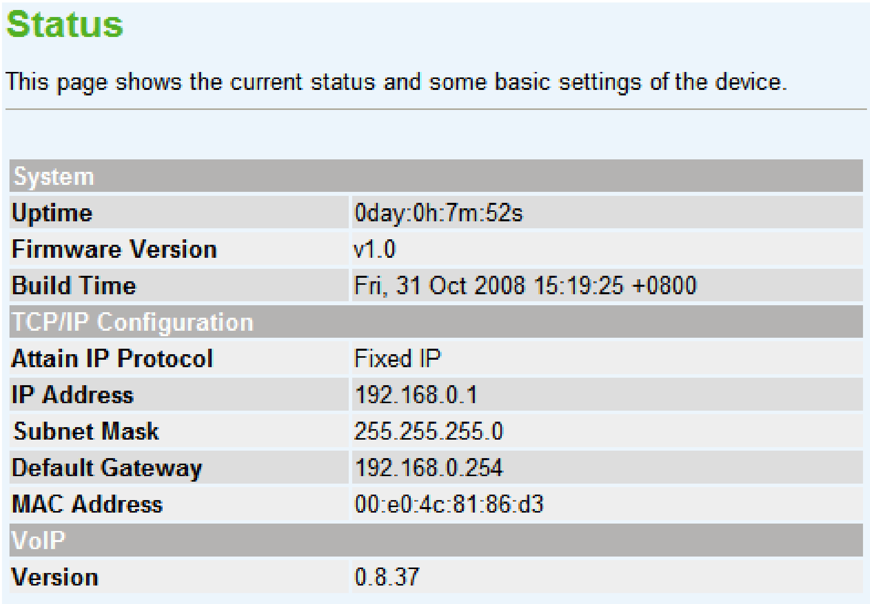
Mode	There are TFTP / FTP and HTTP three ways to provide the auto upgrade function.
TFTP Server Address	Input the TFTP Server address, and it could input the IP or Domain Name form.
HTTP Server Address	Input the HTTP Server address, and it could input the IP or Domain Name form.
HTTP File Path	Set up the file path.
FTP Server Address	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 Path	Set up the file path.
Check new firmware	The ATA will according to the below ways to check the new firmware. **- Power On**: 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 Day	The ATA will check the new firmware every the interval time. The range is 1~30 days.
Scheduling Time	The ATA 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 ATA will carry firmware update out automatically.
File Prefix	It will check the information of model name.
Next update time	It will show the next check date and time.

Chapter 5

Management

Status

In this page can show the current status and some basic settings of the ATA.



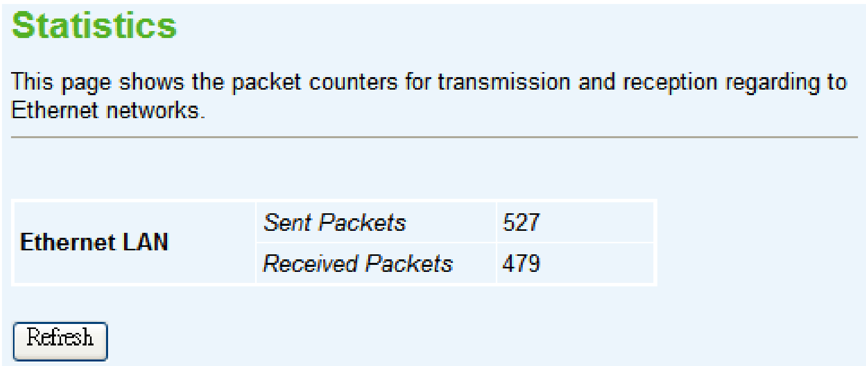
Status

This page shows the current status and some basic settings of the device.

System	
Uptime	0day:0h:7m:52s
Firmware Version	v1.0
Build Time	Fri, 31 Oct 2008 15:19:25 +0800
TCP/IP Configuration	
Attain IP Protocol	Fixed IP
IP Address	192.168.0.1
Subnet Mask	255.255.255.0
Default Gateway	192.168.0.254
MAC Address	00:e0:4c:81:86:d3
VoIP	
Version	0.8.37

Statistics

This page shows the packet counters for transmission and reception regarding to Ethernet networks.



Statistics

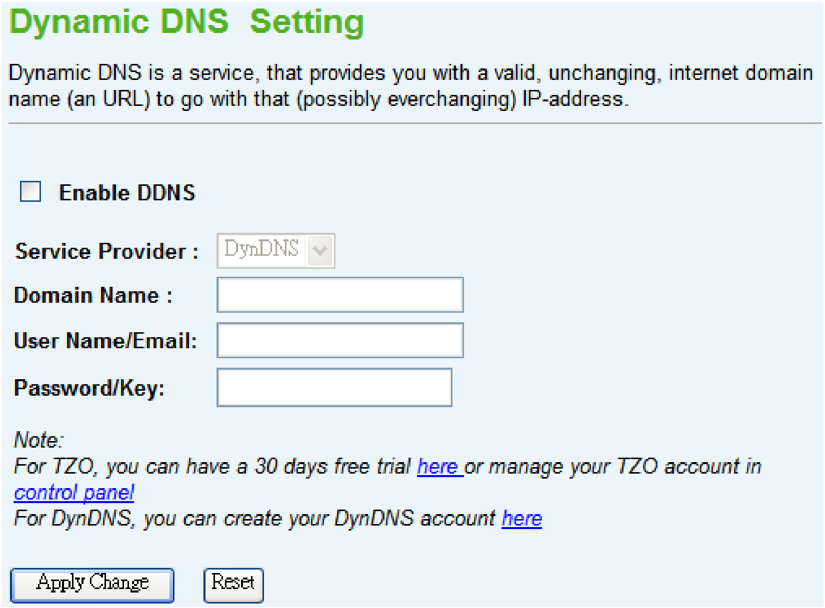
This page shows the packet counters for transmission and reception regarding to Ethernet networks.

Ethernet LAN	*Sent Packets*	527
	Received Packets	479

Refresh

DDNS

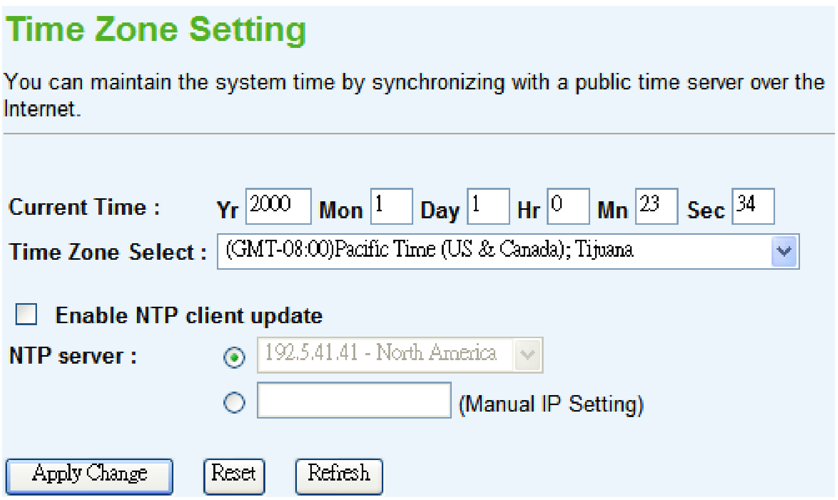
Dynamic DNS is a service, which provides you with a valid, unchanging, internet domain name (an URL) to go with that (possibly ever-changing) IP-address. Before setting this page, you should click below link to DynDNS or TZO to apply an account for DDNS.



Enable DDNS	Check to enable DDNS function. User may register to DDNS server for DDNS function.
Service Provider	Select which server provider to implement DDNS function. For now we provide two servers: DynDNS and TZO.
Domain Name	Input the applied domain name for ATA.
User Name/Email	Input user name for DDNS server login.
Password/Key	Input password for DDNS server login.

Time Zone Setting

You can maintain the system time by synchronizing with a public time server over the Internet.

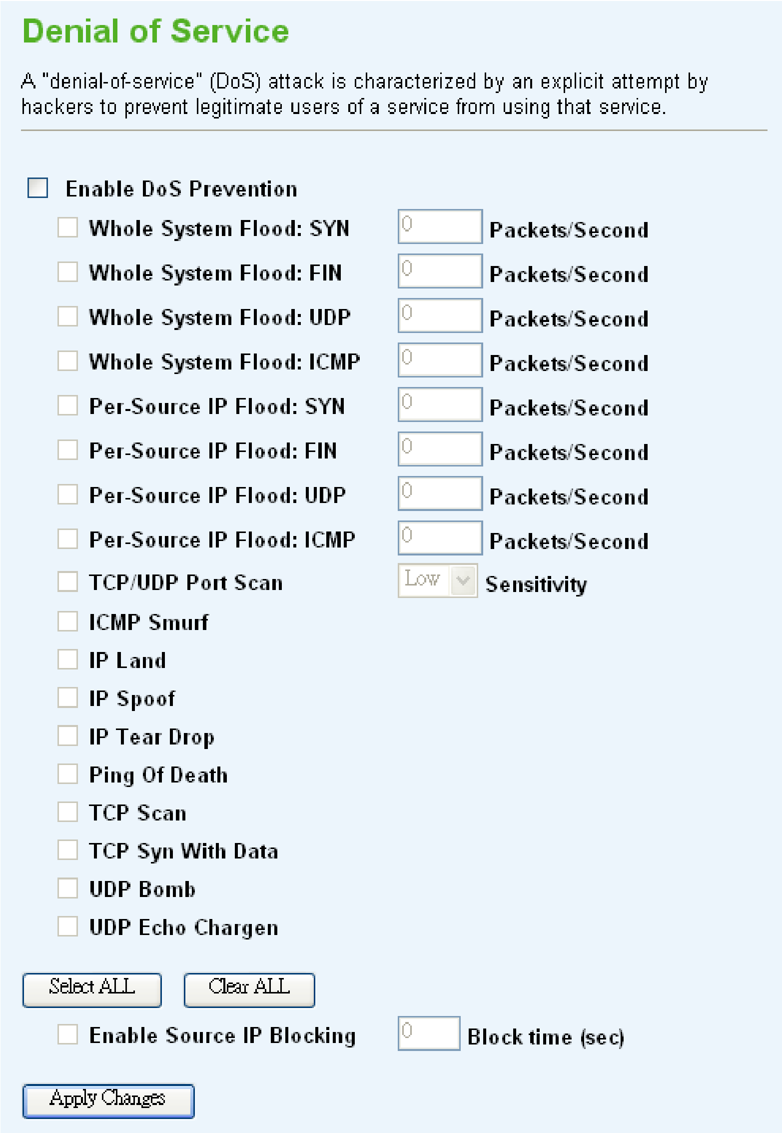


Current Time	Input current time manually.
Time Zone Select	Select local time zone according to location.

Enable NTP client update	Check to enable NTP update. Once this function is enabled, ATA will automatically update current time from NTP server.
NTP server	User may select prefer NTP sever or input address of NTP server manually.

Denial-of-Service

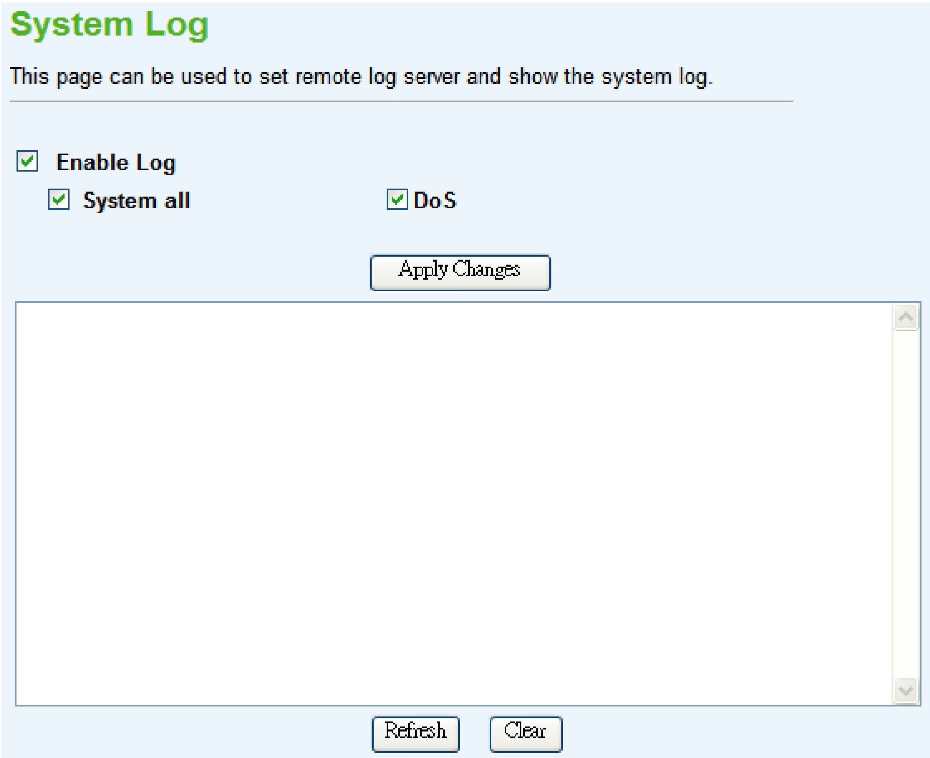
A "denial-of-service" (DoS) attack is characterized by an explicit attempt by hackers to prevent legitimate users of a service from using that service.



Enable DoS Prevention	Check to enable DoS function.
User may set other related configurations about DoS below.	

Log

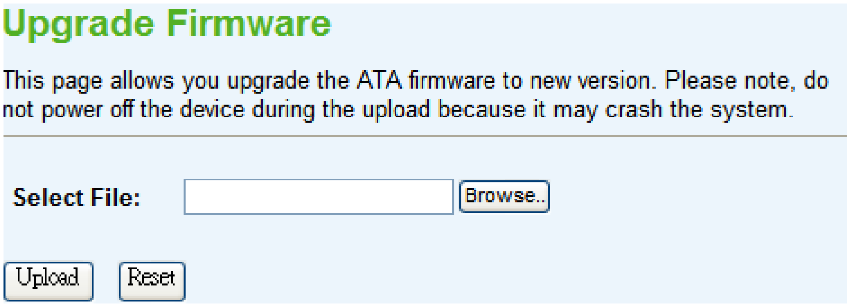
This page can be used to set remote log server and show the system log.



Enable Log	Check to enable log function.
System all/Dos	Select which log you want to check. Related information will be shown at below.

Upgrade Firmware

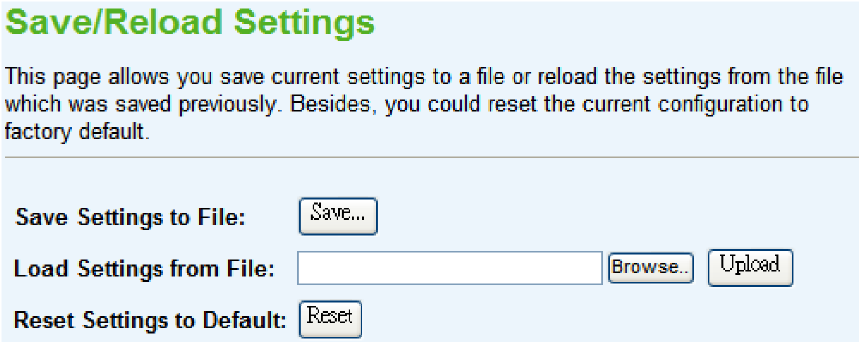
This page allows you upgrade the ATA firmware to new version. Please note, do not power off the device during the upload because it may crash the system.



Select File	Browse and select file you want to upgrade and press Upload to perform upgrade. **Please wait till on screen shows related information after upgrade finished.**

Save / Reload Settings

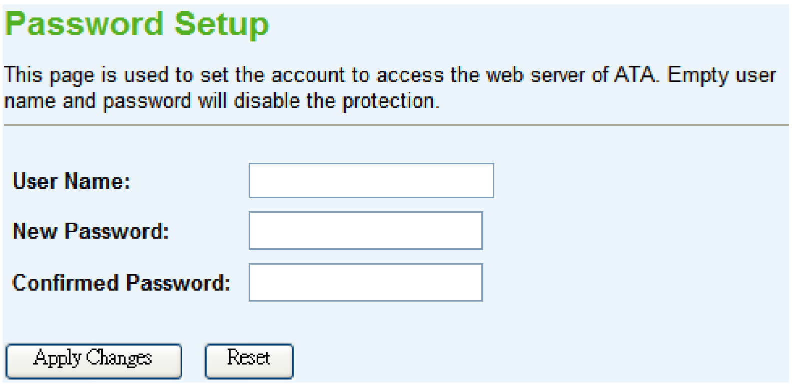
This page allows you save current settings to a file or reload the settings from the file which was saved previously. Besides, you could reset the current configuration to factory default.



Save Settings to File	Save current settings to a file.
Load Settings from File	Browse a file and upload to reload settings.
Reset Settings to Default	Press **Reset** will clean all current configurations and return to default values.

Password Setup

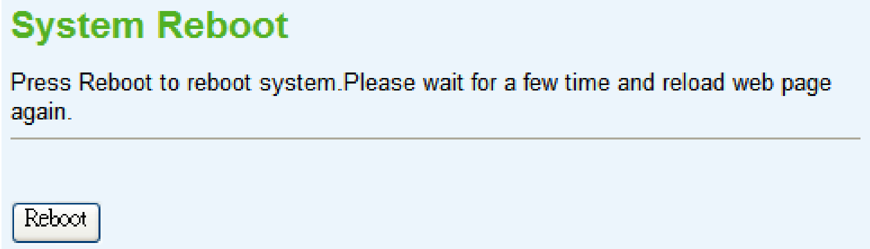
This page is used to set the account to access the web server of ATA. Empty user name and password will disable the protection.



User Name	Enter user name.
New Password	Input password for this user.
Confirmed Password	Confirm password again.

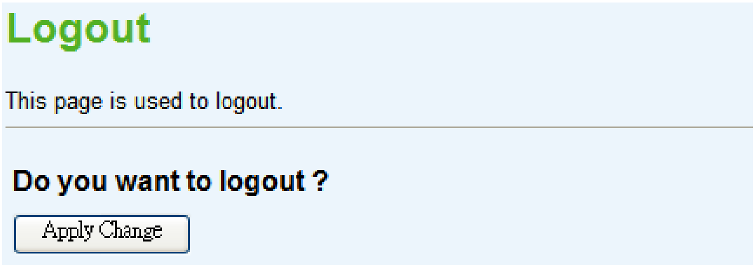
Reboot

Press Reboot to reboot system. Please wait for a few minutes and reload web page again.



Logout

This page is used to logout.



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.

Peer to peer (P2P) mode

Assuming there are two ATA in the network the IP address are 192.168.0.1 and 192.168.0.2

ATA-A
LAN IP address
(192.168.0.1)

ATA-B
LAN IP address
(192.168.0.2)

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

STEP :

Pick up telephone handset of ATA-A and dial "**192.168.0.2#**". Then the phone of ATA-B should ring. You can do the same thing to the ATA-B.

Hint

- If the IP address of the remote calling party is known, you may directly make calls by preset number via its IP address and end with "#".
- If the Telephone Adapter is installed behind a NAT/firewall/ IP sharing device, please make sure the NAT device support SIP applications before making calls.

Case 2: (Peer-to-Peer mode) ATA-150S Port 1 to Port 2 communications

Supposing one ATA-150S connects to two telephones, just pick up phone 1 and dial '**192*168*0*1**5061**', phone 2 will ring.

Analog telephone sets are connected to the phone (RJ-11) ports of ATA-150S respectively

192.168.0.1

1001 1002

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

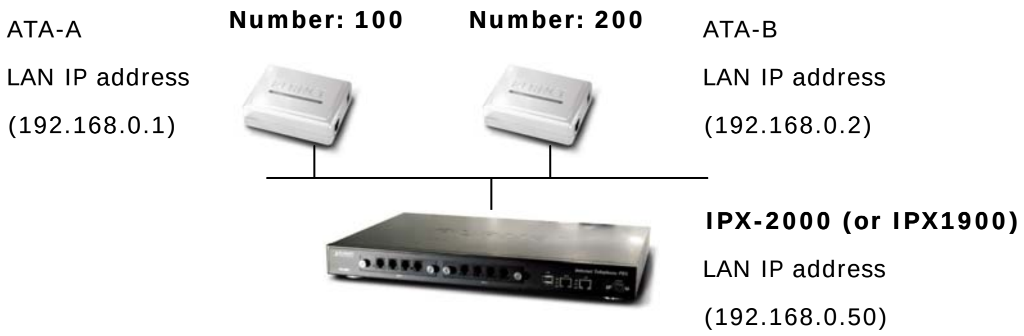
Test the scenario:

1. Pick up the telephone set on ATA-150S port 1, and you should be able to hear the dial-tone
2. Press the keypad: **192*168*0*1**5061#** shall be able to connect to the ATA-150S port 2
3. Then the telephone set in ATA-150S port 2 should ring. Please repeat the same dialing steps on port 2 to establish the first voice communication from ATA-150S

ⓘ 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
- [**ATA-150S**] in PLANET ATA series products, to connect to remote ATA, press the keypad in the following sequence to connect to the remote ATA-150S port 2: **[remote ATA IP address]**5061**, for example: **192*168*0*2**5061**

Case 3: SIP Proxy mode



STEP 1:

Log in IPX-2000 (or IPX-1900) and create two testing accounts/password: **100** / **123** (for ATA-A), and **200** / **123** (for ATA-B) for the voice calls.

STEP 2:

Please log in ATA-A via web browser, find to the **SIP** item. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET IPX-2000 (or IPX-1900) as the IP PBX server for SIP account, call authentications), and then the sample configuration screen is shown below:

Realm 1	
Display Name	100
Line Number	100
Register Name	100
Register Password	●●●
Proxy	☑ Enable
Proxy Server	192.168.0.50
Proxy Port	5060
Domain Server	
SIP Expire Time	60
Outbound Proxy	☐ Enable
Outbound Proxy Server	
Outbound Proxy Port	5060
Register Status	Registered

STEP 3:

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

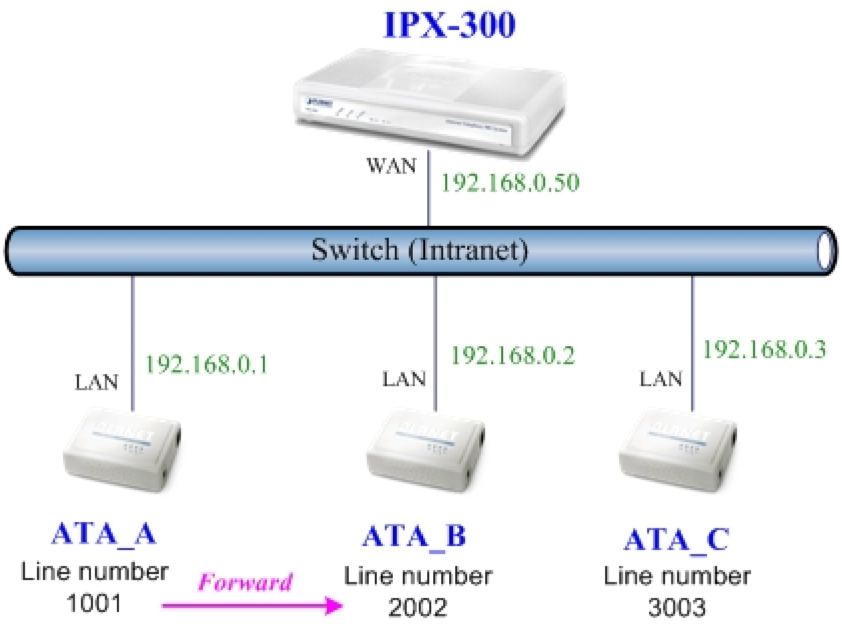
STEP 4:

To verify the VoIP communication, please pick up the telephone. Dial the destination number to make call between SIP clients. For example, ATA-A (with number 100) with keypad number 200 to ATA-B, or reversely makes calls from SIP client (ATA-B) to the number 100 (ATA-A).

Case 4: Call Forward Feature_Example 1

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

In this example, there are three ATA register to IPX-300 and ATA_A had set Call Forward function to ATA_B.



Machine configuration on the ATA:

Please log in ATA_A via web browser, browse to the **Phone 1/2** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the number pf ATA_B in **All Fwd No.** field, then the sample configuration screen is shown below:

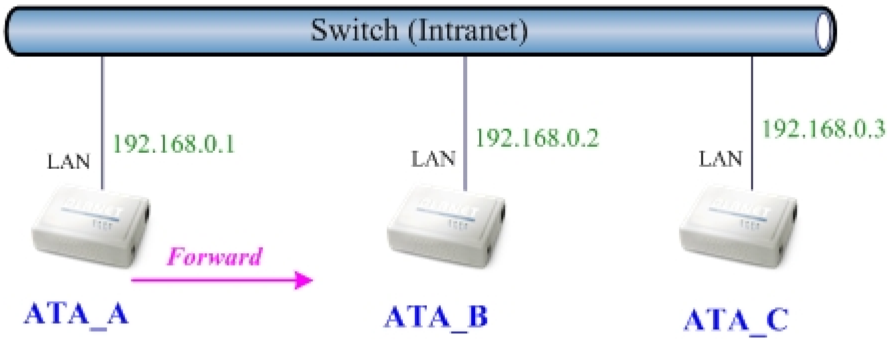
Call Forward	
All Forward	○ Off ◉ VoIP
All Fwd No.	2002
Busy Forward	◉ Off ○ VoIP
Busy Number	
No Answer Forward	◉ Off ○ VoIP
No Answer Fwd No.	
No Answer Fwd Time (sec)	0

Test the scenario:

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

Case 5: Call Forward Feature_Example 2

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



Machine configuration on the ATA:

Please log in ATA_A via web browser, browse to the **Phone 1/2** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the IP address of ATA_B in **All Fwd No.** field, then the sample configuration screen is shown below:

Call Forward	
All Forward	○ Off ◉ VoIP
All Fwd No.	192.168.0.2
Busy Forward	◉ Off ○ VoIP
Busy Number	
No Answer Forward	◉ Off ○ VoIP
No Answer Fwd No.	
No Answer Fwd Time (sec)	0

Test the scenario:

1. ATA_C pick up the telephone
2. Dial the IP Address 192.168.0.1(ATA_A)
3. Because ATA_A had set up **Immediate Forward to** function to the IP Address 192.168.0.2 (ATA_B)
4. The IP Address 192.168.0.2 (ATA_B) will ring up then it pick up the telephone and communication with the ATA_C

Appendix B The method of operation guide

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

Call Transfer

A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out (Press ***1** or "**transfer**" button) to hold the conversation with B.
3. A will hear the dial tone then input the number of C (Follow by the "**#**" key).
4. C will ring up then A hang up the handset.
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 will 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 carry the transfer function out (Press ***1** or "**transfer**" button) to hold the conversation with B at first 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 *1 or "**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 *1 or "**Transfer**" button to hold the conversation with B, and switch to conversation with C.

Switch the Default Proxy

ATA can register to two 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:

#1500#: Realm 1

#1501#: Realm 2

For example: The default is Realm 1, input the **#1501#** from keypad and hang up the telephone set. It will switch to Realm 2 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 Frequently Asked Questions List

If your SIP ATA is not functioning properly, you can refer to this chapter first for sample troubleshooting before contacting your dealer. This can save your time and effort but if the symptoms persist, please consult your dealer.

Q: I forget my ATA login username and / or password

A:

1.) Restore ATA to its factory default settings by pressing the "Reset" button which is at the side panel of the device for 5 seconds or more.

Q: Non of the LEDs are on when I turn on the SIP ATA

A:

1.) Check if power cord is connected properly.

2.) Check if there is proper AC power coming from the power outlet.

Q: Why can't I dial my friend's SIP number?

A:

1.) Check SIP Server Domain Name/IP address. Make sure you have the right Name or IP address.

2.) Check the web browser and access the configuration menu. Make sure that the SIP Server Domain Name/IP Address is correct.

3.) Check the register status under SIP Account Settings in the configuration menu (from web browser). If your status is "Not Registered, it means you do not have a SIP account. Contact your SIP service provider to get an account.

Q: How to know the machine IP address?

A:

1.) To pick up the telephone set, and keyin #120#.

2.) Machine will prompt the current IP address.

Appendix D ATA Specifications

Product	SIP Analog Telephone Adapter	
Model	ATA-150	ATA-150S
Hardware		
LAN	1 x 10/100Mbps RJ-45 port	
FXS	1x RJ-11 connection	2x RJ-11 connection
Protocols and Standard		
Standard	SIP 2.0 (RFC3261), STUN (RFC 3489), UPnP, MD5 for SIP authentication (RFC 2069 / RFC 2617)	
Voice codec	G.711, G.723, G.729	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) G.168: Line echo canceller (LEC) Jitter Buffer DTMF Detection and Generation In-Band and Out-of-Band (RFC 2833), (SIP INFO) QoS : IP TOS (IP Precedence) / DiffServ FAX support : T.38 FAX Relay,G.711 Fax pass-through	
Telephony Features	Call Waiting Call Hold / Resume Call Transfer: Blind Transfer / Attended Transfer Call Forward: On Busy Forward / No Condition forward / No Answer Forward Call Screen: Incoming Call Screen (Reject or Forward Incoming Call) / Outgoing Call Screen (Blocking Outgoing Call) 3-Way Conference	
Protocols	TCP/IP, UDP, DHCP, RTP, HTTP, ICMP, ARP, DNS, TFTP, PPP, PPPoE	
Configuration & Management	Web-based Graphical User Interface Remote management over the IP Network Web-based firmware upgrade Backup and Restore Configuration file SNMP v1/v2 TR-069	
Network and Configuration		
Access Mode	Static IP, DHCP, PPPoE	
Management	Web, Auto-provision, Utility	
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 : VoIP Analog Telephone Adapter (2*FXS)
*Model Number : ATA-150S

* 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 Jonas Yang

Position / Title : Product Manager

Taiwan	**6th November, 2008**	Jonas
Place	*Date*	*Legal Signature*

PLANET TECHNOLOGY CORPORATION

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