

VoIP ATA series (ATA171plus, ATA172plus, ATA-171, ATA-172, ATA-171M, ATA-171P)

User Guide

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1. Introduction

This user's manual is for all ATA series VoIP terminal adapter (ATA). This user's manual explains the IVR instruction, web configuration, and command line configuration for the ATA. Before using the ATA, some setup processes are required to make the ATA work properly. Please refer to the "Instruction of Web Environment" for further information.

2. Hardware Overview

The ATA has the following interfaces for Network, telephone interface, LED indication, and power connector.

2.1 Two RJ-45 Network interface.

These two interfaces support 10/100Mbps Fast Ethernet. You can connect WAN RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the LAN port to your computer.

2.2 One or two RJ-11 analog telephone jack and line interfaces.

You can connect one analog telephone to the terminal adapter and one PSTN line (ATA-171P or ATA-171M). Or, two analog telephone sets at ATA172plus and ATA-172. Or, one telephone set at ATA171plus and ATA-171.

2.3 LED Indication.

There are three LED indicators on the ATA to show the Power, Register, and Off-Hook status.

3. Software Overview

Network Protocol	Tone
<ul style="list-style-type: none"> SIP v1 (RFC2543), v2 (RFC3261) IP/TCP/UDP/RTP/RTCP IP/ICMP/ARP/RARP/SNTP TFTP Client/DHCP Client/ PPPoE Client Telnet/HTTP Server DNS Client NAT/DHCP Server 	<ul style="list-style-type: none"> Ring Tone Ring Back Tone Dial Tone Busy Tone Programming Tone
	Phone Function
Codec <ul style="list-style-type: none"> G.711: 64k bit/s (PCM) G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) G.729A: 8k bit/s (CS-ACELP) G.729B: adds VAD & CNG to G.729 G.723.1 	<ul style="list-style-type: none"> Volume Adjustment Speed dial key Phone book Flash
	IP Assignment
	<ul style="list-style-type: none"> Static IP DHCP PPPoE
Voice Quality <ul style="list-style-type: none"> VAD: Voice activity detection CNG: Comfortable noise generator LEC: Line echo canceller Packet Loss Compensation Adaptive Jitter Buffer 	Security
	<ul style="list-style-type: none"> HTTP 1.1 basic/digest authentication for Web setup MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
Call Function <ul style="list-style-type: none"> Call Hold Call Waiting 	<ul style="list-style-type: none"> ToS field
	NAT Traversal
	<ul style="list-style-type: none"> STUN

<ul style="list-style-type: none"> • Call Forward • Caller ID • 3-way conference 	Configuration
DTMF Function <ul style="list-style-type: none"> • In-Band DTMF • Out-of Band DTMF • SIP Info 	
SIP Server <ul style="list-style-type: none"> • Registrar Server (Five SIP accounts) • Outbound Proxy 	<ul style="list-style-type: none"> • Web Browser • Telnet • IVR/Keypad Firmware Upgrade <ul style="list-style-type: none"> • TFTP • HTTP

4. Keypad Interface from analog phone set of ATA

You can use analog phone set's keypad to operate, configure and listen to configuration (IVR play voice messages in English) at ATA without using web interface. The following table is the access code of each feature. Off-Hook analog phone and dial [IVR access code](#) and follow the voice prompts to configure ATA IP address and other features.

Group	IVR Action	IVR access code	Parameter(s)	Notes:
Function	Dial out from PSTN Line	0*	None	Press 0* can route your call to PSTN Line from analog phone set directly, you can dial out from PSTN Line. (For model ATA-171P and ATA-171M only)
Function	Unlock keypad setting	#190#	None	After you unlock keypad setting, you may start to configure ATA from keypad.
Function	Reboot	#195#	None	After you hear "Option Successful" from IVR message, please hang-up. The system will reboot automatically.
Function	Factory Reset	#198#	None	System reboot automatically. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST including network and service provider data.
Function	Enable PPTP client	#116#	None	System will automatically reboot and PPTP client will be enabled
Function	Disable PPTP client	#117#	None	System will automatically reboot and PPTP client will be disabled
Function	Enable VLAN	#118#	None	System will automatically reboot and VLAN will be enabled.
Function	Disable VLAN	#119#	None	System will automatically reboot and VLAN will be disabled
Function	Enable Call Waiting	#138#	None	System will automatically reboot and Call Waiting will be enabled.

Function	Disable Call Waiting	#139#	None	System will automatically reboot and Call Waiting will be disabled.
Function	Enable Anonymous	#140#	None	System will automatically reboot and Send Anonymous CID was enabled.
Function	Disable Anonymous	#141#	None	System will automatically reboot and Send Anonymous CID was disabled.
Function	Blind Transfer	#510#	None	This feature was only performed during a phone call. For ATA-171M, this will transfer the current IP line to another IP line.
Function	Attendant Transfer	#511#	None	Only be performed in a phone call conversation. For ATA-171M, this will transfer the line to IP from PSTN (must be in IP mode to execute this command)
Function	3-way calling (IP Conference)	#512#	None	Only be performed in a phone call conversation.
Function	Attendant Transfer	#514#	None	Only be performed in a phone call conversation. For ATA-171M, this will transfer the call to PSTN from IP (must be in PSTN mode to execute this command)
Info	Check WAN IP Address	#126#	None	IVR will announce the current WAN IP address of the ATA
Info	Check LAN IP Address	#120#	None	IVR will announce the current LAN IP address of the ATA
Info	Check IP Type	#121#	None	IVR will announce if DHCP is enabled or disabled.
Info	Check the Phone Number	#122#	None	IVR will announce current in use VoIP number
Info	Check Network Mask	#123#	None	IVR will announce the current network mask of the ATA.
Info	Check Gateway IP Address	#124#	None	IVR will announce the current gateway IP address of the ATA.
Info	Check Primary DNS Server Setting	#125#	None	IVR will announce the current setting in the Primary DNS field.
Info	Check Firmware Version	#128#	None	IVR will announce the version of the firmware running on the ATA.
Setting	Set DHCP client	#111#	None	The system will change to DHCP Client type
Setting	Set Static IP Address	#112xxx*xxx*xx x*xxx#	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.

Setting	Set Network Mask	#113xxx*xxx*xx x*xxx#	Enter value-using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Gateway IP Address	#114xxx*xxx*xx x*xxx#	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Primary DNS Server	#115xxx*xxx*xx x*xxx#	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Codec	#130+[1-8]#	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.
Setting	Set Handset Gain	#131+[00~15]#	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6
Setting	Set Handset Volume	#132+[00~12]#	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
Setting	Set Auto Configuration Mode	#137X#	Select the auto configuration mode, in the X field, you can press the following; 0:OFF, 1:TFTP, 2:FTP	You can set the auto configuration method you want, default is off
Setting	Set Auto Configuration For TFTP Server	#135xxx*xxx*xx x*xxx#	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set auto configuration method to TFTP first
Setting	Set Auto Configuration For FTP Server	#136xxx*xxx*xx x*xxx#	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set auto configuration method to FTP first

5. Instruction of Web Environment

5.1 Default setting

5.1.1 Default network setting

Network Mode: Default NAT Mode

WAN Port: DHCP Client Mode

LAN Port: DHCP Server Enable, IP Address: 192.168.123.1

5.1.2 Web login

VoIP Web Login default link, <http://192.168.123.1:9999>

➤ Account Login :

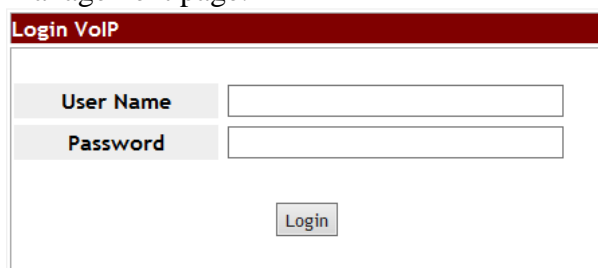
- Administrator: Login Account: root, Password: test
- System: Login Account: system, Password: test
- Normal: Login Account: user, Password: test

5.2 ATA network connection

Please connect PC Ethernet cable to LAN port, and set PC to DHCP mode. Default IP address is 192.168.123.150.

5.3 Login VOIP Web page

Provide login system management page.



Suggested that uses IE7, 8, Firefox, Google the Chrome browser.

User Name	Input user's name, it can be numeral or letters.
Password	Input password, it can be numeral or letters.
Login [button]	Login to the ATA
Clear [button]	Clear all informations

5.4 VoIP main setting page

5.4.1 Function instructions

Provide below function [Information (system information), Phone (phone environment), Network (network environment), NAT (local network), SIP (SIP parameter setting), Management (enhance setting), Save & Reboot , Logout].

5.4.2 Function description

VOIP

ATA172 Plus

- Status
- Phone
- Network
- NAT
- SIP
- Management
- Save and Reboot
- Logout

System Status

WAN Information

Link Status:	Connected	Active:	Fixed IP Client
IP Address:	192.168.22.37	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	164.124.101.2
Second DNS:	203.248.252.2	MAC Address:	00:01:a8:71:02:32

LAN Information

IP Address:	192.168.123.1	MAC Address:	00:01:a8:71:02:32
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System Information

Model Name:	ATA172 Plus	Version:	ATA172_Plus_V3.3
Firmware Version:	2.0.14-1-1210172	DSP Version:	NV-1106080
Current Time:	2015-01-07 10:47		
System Up Time:	0 day(s) 0 hour(s) 1 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 1 minute(s)		

Register Information

Phone 1

Realm 1 Status:	Not in used	Display Name:	
Realm 2 Status:	Not in used	Display Name:	
Realm 3 Status:	Not in used	Display Name:	
Realm 4 Status:	Not in used	Display Name:	
Realm 5 Status:	Not in used	Display Name:	

Phone 2

Realm 1 Status:	Not in used	Display Name:	
Realm 2 Status:	Not in used	Display Name:	
Realm 3 Status:	Not in used	Display Name:	
Realm 4 Status:	Not in used	Display Name:	
Realm 5 Status:	Not in used	Display Name:	

item	Description
Gateway	Device model name for Gateway(ATA) or Phone
Status	Current device information list
Phone	Phone item provide [Phone Book , Dial Plan , Call Service , General setting, Volume setting] function
Network	Network setting provide [WAN , DDNS, VLAN , VPN (PPTP/L2TP), SNTP (time sync)] function
NAT	NAT provide [LAN setting, DMZ & Mac Clone, Virtual Server] function.
SIP	SIP provide [Service (SIP registration), Codec selection, Advanced setting, STUN (STUN & Fource setting)] function.
Management	Management item provide [Status Log , Auto Config , Auto Update , New Firmware , Advanced , Password , Tones), Default (reset to default), Language]function .
Save & Reboot	Save and Reboot function
Logout	Logout system.

5.5 System Information

5.5.1 Function description

There are network information, firmware version and SIP register status.

System Status

WAN Information

Link Status:	Connected	Active:	Fixed IP Client
IP Address:	192.168.22.37	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	164.124.101.2
Second DNS:	203.248.252.2	MAC Address:	00:01:a8:71:02:32

LAN Information

IP Address:	192.168.123.1	MAC Address:	00:01:a8:71:02:32
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System Information

Model Name:	ATA172 Plus	Version:	ATA172_Plus_V3.3
Firmware Version:	2.0.14-1-1210172	DSP Version:	NV-1106080
Current Time:	2015-01-07 10:47		
System Up Time:	0 day(s) 0 hour(s) 1 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 1 minute(s)		

Register Information

Phone 1

Realm 1 Status:	Not in used	Display Name:	
Realm 2 Status:	Not in used	Display Name:	
Realm 3 Status:	Not in used	Display Name:	
Realm 4 Status:	Not in used	Display Name:	
Realm 5 Status:	Not in used	Display Name:	

Item	Description
WAN Information	Shows the current status of WAN Port.
Link Status	Shows the network connected Speed.
Active	Shows the network connected type.
IP Address	Shows IP address of the device.
Subnet Mask	Shows the subnet mask.
Default Gateway	Shows the default gateway.
Primary DNS	Shows the primary DNS server.
Second DNS	Shows the secondary DNS server.
MAC Address	Shows the MAC ID.
LAN Information	Shows the current status of LAN Port
IP Address	Shows IP address of the device.
MAC Address	Shows the MAC ID.
System Information	Shows the status of System.
Model Name	Show device model name.
Version	Show device firmware version.
Firmware Version	Shows the firmware version for software control.

Item	Description
DSP Version	Shows the DSP version. AC: AC97 WM: Winbound LE: Legeevity NV: Nuvoton
Current Time	Shows the current time.
Update Date	Shows the date of updating system.
System Up Time	Shows the system running time.
Netwrk Link Up Time	Shows the network running time.
Register Information	Shows the status of SIP register.
Status	Shows register state.
Display Name	Shows register number.

5.5.2 System Information example

Figure 1: LAN Mode: Bridge (Ethernet Switch mode)

System Status

WAN Information

Link Status:	Connected	Active:	Fixed IP Client
IP Address:	192.168.22.37	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	164.124.101.2
Second DNS:	203.248.252.2	MAC Address:	00:01:a8:71:02:32

System Information

Model Name:	ATA172 Plus	Version:	ATA172_Plus_V3.3
Firmware Version:	2.0.14-1-1210172	DSP Version:	NV-1106080
Current Time:	2015-01-07 11:24		
System Up Time:	0 day(s) 0 hour(s) 0 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 0 minute(s)		

(Figure 1)

Figure 2: LAN Mode: Router (NAT Router)

System Status

WAN Information			
Link Status:	Connected	Active:	Fixed IP Client
IP Address:	192.168.22.37	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	164.124.101.2
Second DNS:	203.248.252.2	MAC Address:	00:01:a8:71:02:32

LAN Information			
IP Address:	192.168.123.1	MAC Address:	00:01:a8:71:02:32

System Information			
Model Name:	ATA172 Plus	Version:	ATA172_Plus_V3.3
Firmware Version:	2.0.14-1-1210172	DSP Version:	NV-1106080
Current Time:	2015-01-07 11:22		
System Up Time:	0 day(s) 0 hour(s) 36 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 36 minute(s)		

(Figure 2)

6. Phone

Provide functions of [[Phone Book](#), [Speed Dial](#), [Dial Plan – Basic](#), [Dial Plan – Advanced](#), [Call Service](#), [General](#), [Volume](#)]

6.1 Phone Book

6.1.1 Function description

Phone Book can provide [140 entries](#); Export/Import feature, the file format is [csv](#).

When user A dials a [Name], Phone Book will check it on Phone Book. If system finds it, it will dial the [Number] of [Name]. If the [Name] is not on Phone Book, system will dial the number you have dialed.

6.1.2 Parameter description

Phone Book Setting

Page:

Index	Name	Number/URL	Action
1	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
2	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
3	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
4	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
5	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
6	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
7	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
8	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
9	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
10	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
11	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
12	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
13	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
14	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
15	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
16	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
17	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
18	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
19	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
20	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>

Item	Description
Page	Default: Page 1. Select the page, from Page1~Page7.
Index	Shows the serial number. 140 entries in total, from Phone 0~139. One page has 20 entries.
Name	Set up the user's name. These columns provide the function of speed dial by only enter numbers; maximum length is 31 bytes.
Number/URL	Set up the user's number. These columns can enter numbers and strings; maximum length is 63 bytes. Ex: 0212345678, 0800024365, www.dyndns.info
Action	Provide [Delete] button to erase the datas.
Submit [Button]	Save the Settings.
Delete All	Reset all data.
Acess Phone Book	Enter "Remote Phone Book Setting" web page.
Export csv [Button]	Export [Phone Book] data, the file format is 『.csv』.
Import csv [Button]	Import [Phone Book] data, the file format is 『.csv』.

Remote Phone Book Setting

Local Book

Export csv

Import csv

瀏覽...

Remote Phone Book

HTTP or TFTP server Address:

Exp: <http://www.voip.com/user/book.xml> (HTTP)
61.62.53.64/user/book.xml (TFTP)

Synchronization period:

(1~24 Hours)

Status:

Submit

Item	Description
Export csv [Button]	Export [Phone Book] data, the file format is 『.csv』.
Import csv [Button]	Import [Phone Book] data, the file format is 『.csv』.
HTTP or TFTP Server Address	Use HTTP or TFTP server to upgrade LP399's phone book data.
Synchronization	The LP399 will reference this time to upgrade phone book data

Item	Description
period	at HTTP or TFTP server by itself.
Submit [Button]	Save the Settings.

6.1.3 Operate Instruction

Example 1: Setup [Phone Book] data

Step 1: On the [Phone Book Setting] page, Setup [Index: 1, Name: 301, Number: 301@192.168.1.2; Index: 2, Name: 206, Number: 1747643364; Index: 3, Name: test, Number: 8123478944566] (See Figure 1).

Index	Name	Number/URL	Action
1	301	301@192.168.1.2	Delete
2	206	1747643364	Delete
3	test	8123478944566	Delete

(Figure 1)

Instruction 1: Dial [301], system find the [301] on Index 1, then system dial Name's Number of Index 1. System will dial [192.168.1.2]

Instruction 2: Dial [206], system find the [206] on Index 2, then system dial Name's Number of Index 2. System will dial [17476433364].

Instruction 3: Because Index 3's Name is a string, so the speed dial function can not be used, you can just check the Index 3' Name and Number.

Example 2: Export / Import [Phone Book] data

◆ Export Feature

Step 1: In [Phone Book Setting] web page, the page have two numbers data. (See Figure 2).

Index	Name	Number/URL	Action
1	May	2206	Delete
2	Rod	2203	Delete
3			Delete

(Figure 2)

Step 2: To perform the export function, please press [Export csv] button (See Figure 3).

Local Book

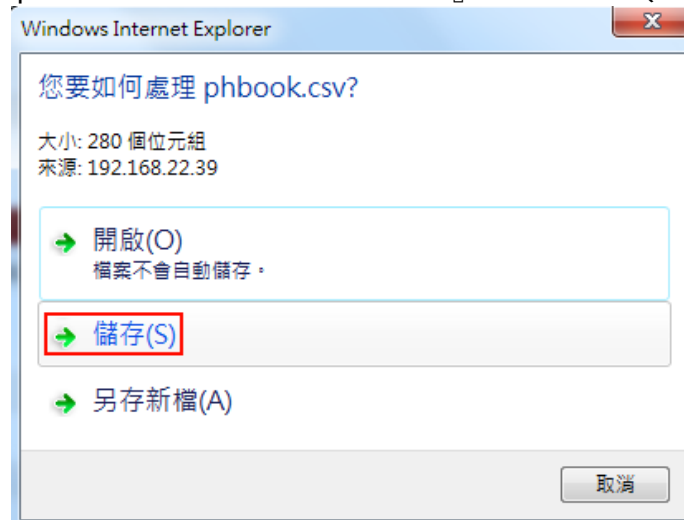
Export csv

Import csv

(Figure 3)

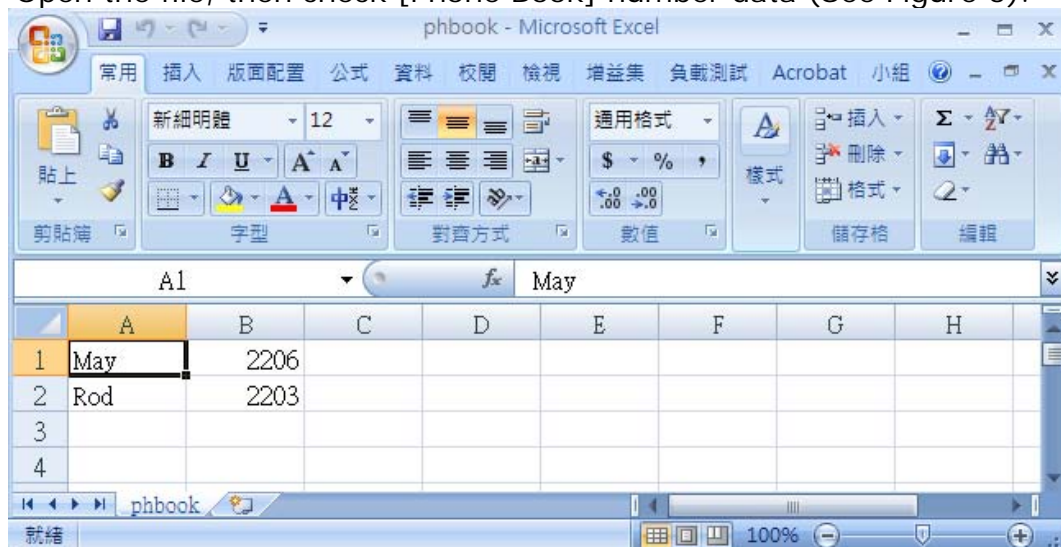
Step 3: Monitor show [File Download] page, default file name is phbook.csv, select [Save]

button, LP399 parameter will save to 『.csv』 file format (See Figure 4).



(Figure 4)

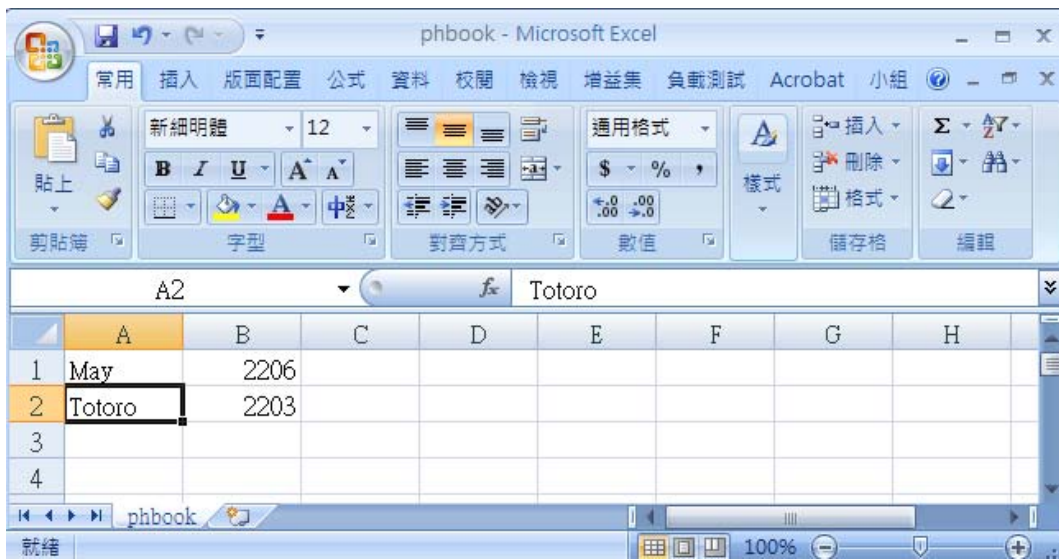
Step 4: Open the file, then check [Phone Book] number data (See Figure 5).



(Figure 5)

◆ Import Feature

Step 1: Set up 『.csv』 format for [Phone Book] data (See Figure 6).



(Figure 6)

Step 2: In [Phone Book Setting] web page, choose the phone book file data to load [example: F:\Test\phbook.csv], press [Import csv] button (See Figure 7).



(Figure 7)

Step 3: Load data finish, [Index: 1 and 2] will show phone book data (See Figure 8).

Index	Name	Number/URL	Action
1	May	2206	Delete
2	Totoro	2203	Delete
3			Delete

(Figure 8)

6.2 Dial Plan setting

6.2.1 Function

Provide dial rule and define proxy server prefix code.

6.2.2 Instructions

Figure 1: Suitable for 1FXS and 2FXS ATA.

Dial Plan Setting - Basic

Use Dial Plan : ▼

Index	Drop prefix	Prefix	Replace Rule
1	Disable ▼	<input type="text"/>	<input type="text"/>
2	Disable ▼	<input type="text"/>	<input type="text"/>
3	Disable ▼	<input type="text"/>	<input type="text"/>
4	Disable ▼	<input type="text"/>	<input type="text"/>

Index	Dial Now Rule
1	<input type="text"/>
2	<input type="text"/>
3	<input type="text"/>
4	<input type="text"/>
5	<input type="text"/>
6	<input type="text"/>
7	<input type="text"/>
8	<input type="text"/>

Realm 1 Area Code:	<input type="text" value="1*"/>
Realm 2 Area Code:	<input type="text" value="2*"/>
Realm 3 Area Code:	<input type="text" value="3*"/>
Realm 4 Area Code:	<input type="text" value="4*"/>
Realm 5 Area Code:	<input type="text" value="5*"/>

Inter Digit Time:	<input type="text" value="5"/> (seconds)
Key As Send #:	Enable ▼
# Format is %23:	Disable ▼

(Figure 1)

Item	Description
Index	Index number. There are 4 rules to support dial rule for Add, drop and replacement features.
Drop Prefix	Default setting is Disable (Add Prefix number feature) . When it was set to "Enable" and the number match the "Rule" number, ATA will replace the "Rule" field number and

Item	Description
	<p>use "Prefix" field number instead.</p> <p>Disable: Add prefix number in front of the called number when called number match "Rule" number</p> <p>Enable: Replace "Rule" number to "Prefix" number when called number match "Rule" number</p>
Prefix	<p>Added or replace number. For numbers only and maximum length is 8 digits.</p>
Rule	<p>Define number manipulation rule.</p> <p>It can be numbers or signs (+, x). The (+) means "Or", (x) means any numbers which are from 0 - 9. Maximum length is 40 digits.</p> <p>NOTE: The first digit can't be 0 if it is 2 digits number.</p>
Index	<p>Index number. There are 8 dial rules to enter.</p>
Dial Now Rule	<p>Automatic dialing Now (immediately). When the dialing rule matches the contents in this column, the automatic dialing function will be executed without waiting for "press #" or "Auto Dial Time" to dial out. Both numbers and symbols can be entered. The number length is 80 digits.</p> <p>Symbols: only allow *, #, + and x.</p> <p>+: represents "or".</p> <p>x: any number between 0 - 9.</p> <p>Note: The 1st digit number can not be set to "0", because "0" will not determine as the Dial Now Rule. If the Dial Now is set to 0xxxx, the system will not follow the dialing rule to dial out.</p>
Realm 1 prefix	<p>Default setting is 1*. When you dial 1* + called number, ATA will switch to the first account and dial out the called number immediately. Maximum length is 7 digits.</p>
Realm 2 prefix	<p>Default setting is 2*. When you dial 2* + called number, ATA will switch to the second account and dial out the called number immediately. Maximum length is 7 digits.</p> <p>PS: If account registers fail, it will not be switched.</p>
Realm 3 prefix	<p>Default setting is 3*. When you dial 3* + called number, ATA will switch to the third account and dial out the called number immediately. Maximum length is 7 digits.</p> <p>PS: If account registers fail, it will not be switched.</p>
Realm 4 prefix	<p>Default setting is 4*. When you dial 4* + called number, ATA will switch to the fourth account and dial out the called number immediately. Maximum length is 7 digits.</p> <p>PS: If account registers fail, it will not be switched.</p>
Realm 5 prefix	<p>Default setting is 5*. When you dial 5* + called number, ATA will switch to the fifth account and dial out the called number immediately. Maximum length is 7 digits.</p> <p>PS: If account registers fail, it will not be switched.</p>
Auto Dial Time	<p>Default is 5 seconds. The configuration range is 3 to 9 seconds. ATA will dial out automatically when user didn't enter any digit within this time.</p>
Use # as send key	<p>Default is Enable. Define [#] for end of dialing key.</p> <p>Enable: when ATA received[#], it will dial out immediately. (no need wait for the auto dial out time)</p>

Item	Description
	Disable: follow [Auto Dial Time] time to dial out.
# Format is %23	Default is Disable; Use [#] word to send %23 singal. Provide drop-down options: Disable, Enable.
Submit	Save the configuration.

Figure 2: 1FXS+1FXO, ATA-171M.

Dial Plan Setting - Basic

Use Dial Plan :

Index	Drop prefix	Prefix	Replace Rule
1	<input type="text" value="Disable"/> <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>
2	<input type="text" value="Disable"/> <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>
3	<input type="text" value="Disable"/> <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>
4	<input type="text" value="Disable"/> <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>

Index	Dial Now Rule
1	<input type="text"/>
2	<input type="text"/>
3	<input type="text"/>
4	<input type="text"/>
5	<input type="text"/>
6	<input type="text"/>
7	<input type="text"/>
8	<input type="text"/>

Realm 1 Area Code: ^{*}
 Realm 2 Area Code: ^{*}
 Realm 3 Area Code: ^{*}
 Realm 4 Area Code: ^{*}
 Realm 5 Area Code: ^{*}

Inter Digit Time: (seconds)
 Key As Send #:
 # Format is %23:

Auto PSTN backup:
 PSTN feature code: ^{*}
 Routing Type:
 Routing Rule:

(Figure 2)

Item	Description
Index	Index number

Item	Description
Drop Prefix	Default setting is Disable. When it was set to "Enable" and the number match the "Rule" number, ATA will replace the "Rule" field number and use "Prefix" field number instead. Disable: Add prefix number in front of the called number when called number match "Rule" number. Enable: Replace "Rule" number with "Prefix" number when called number match "Rule" number
Prefix	Setup Added or Replace number. Only for numbers and maximum length is 8 digits .
Rule	Define dial number manipulation. It allows to enter both numbers and signs (+, x) . The (+) means "Or". The (x) means any numbers which are from 0 to 9 . The maximum length is 40 digits . NOTE: The first digit can't be 0 if it is a 2 digits number.
Index	Index number. There are 8 rule data . Setup Dial Now (dial immediately) rule .
Dial Now Rule	Automatic dialing immediately . When the dialing rule match the contents in this row, the automatic dialing function will be executed without waiting for both "press #" and "Auto Dial Time" to dial out. Both Numbers and symbols can be entered. The number length is 80 digits . Symbols: Only allow to enter *, #, + and x. +: represents "or". x: any number between 0 to 9. Note: The 1st number can not be set to "0", because "0" does not determine the Dial Now standard. If the Dial Now is set to 0xxxx (because it starts with "0"), the system do not follow the dialing rule to dial out.
Realm 1 prefix	Default setting is 1* . When you dial 1* + called number, ATA will switch to the first account and dial out the called number. Maximum data length is 7 digits . It only allows to enter number and #, * .
Realm 2 prefix	Default setting is 2* . When you dial 2* + called number, ATA will switch to the second account and dial out the called number. Maximum data length is 7 digits . It only allows to enter number and #, * . PS: If account registers to SIP Server fail, it will not be switched.
Realm 3 prefix	Default setting is 3* . When you dial 3* + called number, ATA will switch to the third account and dial out the called number. Maximum data length is 7 digits . It only allows to enter number and #, * . PS: If account registers to SIP Server fail, it will not be switched.
Realm 4 prefix	Default setting is 4* . When you dial 4* + called number, ATA will switch to the fourth account and dial out the called number. Maximum data length is 7 bytes . It only allows to enter number and #, * . PS: If account registers to SIP Server fail, it will not be switched.
Realm 5 prefix	Default setting is 5* . When you dial 5* + called number, ATA will switch to the fifth account and dial out the called number. Maximum data length is 7 bytes . It only allows to enter number

Item	Description
	and #, *. PS: If account registers fail, it will not be switched.
Auto Dial Time	Default is 5 seconds, the option range is 3 to 9 seconds. ATA will dial out automatically when user didn't enter any digit within this time period.
Use # as send key	Default is Enable. Define [#] as end of dialing key. Enable: When ATA received [#], it will dial out immediately. (no need wait for the auto dial out time). Disable: follow [Auto Dial Time] time to dial out.
# Format is %23	Default is Disable; Use [#] word to send %23 signal. Provide drop-down options: Disable, Enable.
Auto PSTN backup	Default is Disable. Provide PSTN auto backup function. When SIP account register to SIP Server fail, FXS port was switched to PSTN line automatically if ATA was set to enable. In the meantime, the FXS port's dial tone was generated from PSTN line. * Please make sure the PSTN line connect to ATA correctly if you want to enable this function.
PSTN feature Code	Default is 0*. This code is to switch the route to PSTN port manually. When you dial 0* you will hear dial tone from PSTN line and the call will dial out through PSTN line. Maximum length is 7 digits. It only allows to enter number, * and # digits.
Routing Type	Default is "Disable" (OFF) and provide IP, FXO, and Disable features. The call behavior is based on Routing Rule. According to the routing rule, IP or FXO dial out function can be selected.
Routing Rule	"D" is a dropping prefix function and "+" is used to add multiple routing rule. Example: Routing rule is D007+009. 1. When the entered numbers start with 007, such as 00782280220, the condition is satisfied with D007. The routing rule first drops 007 and replace the numbers with 82280220. Then refer to the "Routing To" setting to select the dialing route. 2. When the entered numbers start with 009, such as 00982280220, the condition is satisfied with 009. The routing rule will not drop any prefix, and then refer to the "Routing To" setting to select the dialing routes.
Submit	Save the configuration.

Figure 3: 1FXS+1PSTN, ATA-171P device.

Dial Plan Setting - Basic

Use Dial Plan : ▼

Index	Drop prefix	Prefix	Replace Rule
1	Disable ▼	<input type="text"/>	<input type="text"/>
2	Disable ▼	<input type="text"/>	<input type="text"/>
3	Disable ▼	<input type="text"/>	<input type="text"/>
4	Disable ▼	<input type="text"/>	<input type="text"/>

Index	Dial Now Rule
1	<input type="text"/>
2	<input type="text"/>
3	<input type="text"/>
4	<input type="text"/>
5	<input type="text"/>
6	<input type="text"/>
7	<input type="text"/>
8	<input type="text"/>

Realm 1 Area Code:

Realm 2 Area Code:

Realm 3 Area Code:

Realm 4 Area Code:

Realm 5 Area Code:

Inter Digit Time: ▼ (seconds)

Key As Send #: ▼

Format is %23: ▼

Auto PSTN backup: ▼

PSTN feature code:

(Figure 3)

Item	Description
Index	Index number. There are 4 entries to configure ADD and Replace dial codes.
Drop Prefix	<p>Default setting is "Disable" (It is also an Add digits feature). When it was set to "Enable" (It is also an Replacement digits feature) and the number match the "Dial Rule" number, ATA will replace the "Rule" field numbers and use "Prefix" field number instead.</p> <p>Disable: Add prefix number in front of the called number when called number match "Rule" number.</p> <p>Enable: Replace "Rule" number with "Prefix" number when</p>

Item	Description
	called number match "Rule" number.
Prefix	Added or Replace number. It only allows to enter number. Maximum data length is 8 digits .
Rule	Define number manipulation rule. It can be numbers or signs (+, x) . The (+) means "Or". The (x) means any numbers which are from 0 to 9. Maximum data length is 40 digits. Note: The first digit can't be 0 if it is 2 digits number length.
Index	Index number. There are 8 entries to configure Dial Out immediately.
Dial Now Rule	Automatic dialing. When the dialing rule match contents in this row, the automatic dialing function will be executed without waiting for "press #" and "Auto Dial Time" to dial out. Both Numbers or symbols can be entered. The number length is 80 digits. Symbols: It was allowed to enter * , # , + and x . +: represents "or". x: any number between 0 to 9. Note: 1st digit can not be set to "0", because "0" does not determine as the Dial Now Rule. If the Dial Now is set to 0xxxx, the system will not follow the dialing rule to dial out.
Realm 1 prefix	Default setting is 1*. When you dial 1* + called number, ATA will switch to the first account and dial out the called number. Maximum data length is 7 digits.
Realm 2 prefix	Default setting is 2*. When you dial 2* + called number, ATA will switch to the second account and dial out the called number. Maximum data length is 7 digits. PS: If account registers to SIP Server fail, it will not be switched.
Realm 3 prefix	Default setting is 3*. When you dial 3* + called number, ATA will switch to the third account and dial out the called number. Maximum data length is 7 digits. PS: If account registers to SIP Server fail, it will not be switched.
Realm 4 prefix	Default setting is 4*. When you dial 4* + called number, ATA will switch to the fourth account and dial out the called number. Maximum data length is 7 digits. PS: If account registers to SIP server fail, it will not be switched.
Realm 5 prefix	Default setting is 5*. When you dial 5* + called number, ATA will switch to the fifth account and dial out the called number. Maximum data length is 7 digits. PS: If account registers to SIP Server fail, it will not be switched.
Auto Dial Time	Default is 5 seconds, the configuration range is 3 to 9 seconds. ATA will dial out automatically when user didn't enter any digit within this time period.
Use # as send key	Default is Enable. Define [#] as end of dialing key. Enable: When ATA received [#], it will dial out immediately. (no need wait for the auto dial out time). Disable: follow [Auto Dial Time] time to dial out.
# Format is %23	Default is Disable; Use [#] word to send %23 signal. Provide drop-down options: Disable, Enable.

Item	Description
Auto PSTN backup	Default is Disable. Provide PSTN auto backup function. When SIP account register to SIP Server fail, FXS port was switched to PSTN line automatically if ATA was set to enable. In the meantime, the FXS port's dial tone was generated from PSTN line. * Please make sure the PSTN line connect to ATA correctly if you want to enable this function.
PSTN feature Code	Default is 0* . This code is to switch the route to PSTN port manually. When you dial 0* you will hear dial tone from PSTN line and the call will dial out through PSTN line. Maximum length is 7 digits. It only allows to enter number, * and # digits.
Submit	Save the configuration.

6.2.3 Operate Instruction

Example 1: Drop Prefix and Dial Now function.

Step 1: In [Dial Plan Setting] page, the configuration is [Index: 1, Drop prefix: Disable, Prefix: 002, Rule: 8613+8662; Index: 2, Drop prefix: Enable, Prefix: 006, Rule: 002+003+004+005+007+009; Index: 3, Drop prefix: Disable, Prefix: Replace: 009, Rule: 12; Index: 4, Drop prefix: Disable, Prefix: 007, Rule: 53+35xx +21xx; Index: 1, Dial Now Rule: *xx+#xx+11x +xxxxxxx] (figure1) .

Index	Drop prefix	Prefix	Replace Rule
1	Disable ▾	002	8613+8662
2	Enable ▾	006	002+003+004+005+007+009
3	Disable ▾	009	12
4	Disable ▾	007	53+35xx+21xx

Index	Dial Now Rule
1	*xx+#xx+11x+xxxxxx
2	

(Figure 1)

Instruction 1:

When the dialing number is like [8613xxxx], it matched [Rule] -> [8613], so ATA will add [prefix] [002] in front of [8613]. The actual dialing number will be [002+8613+xxx].

When the dialing number is like [8662xxxx], it matched [Rule] -> [8662], so ATA will add [prefix] [002] in front of [8662]. The actual dialing number will be [002+8662+xxx].

Instruction 2:

When the dialing number is like [002+86xxxx], it matched [Rule] -> [002], so ATA will replace [002] with [Prefix] [006]. The actual dialing number will be [006+86xxxx].

When the dialing number is like [003+77xxxx], it matched [Rule] -> [003], so ATA will replace [003] with [Prefix] [006]. The actual dialing number will be [006+77xxxx].

Instruction 3:

Drop prefix: Disable, Replace rule: 009, Rule: 12.

When the dialing number is like [12xxxx], it matched [Rule] -> [12], ATA will add [Prefix] [009] in front of [12]. The actual dialing number will be [009+12xxxx].

Instruction 4:

When the dialing number is [53789], it matched [Rule] -> [53], ATA will add [Prefix] [007] in front of [53]. The actual dialing number will be [007+53789].

When the dialing number is [3507], it matched [Rule] -> [35xx], ATA will add [Prefix] [007] in front of [3507]. The actual dialing number will be [007+3507].

When the dialing number is [2199], it matched [Rule] -> [21xx], ATA will add [Prefix] [007] in front of [2199]. The actual dialing number will be [007+2199].

Instruction 5:

When the dialing number is [*00, *01, *02... *99], it matched [Dial Now Rule] -> [*xx]. ATA will dial out immediately.

When the dialing number is [#00, #01, #02... #99], it matched [Dial Now Rule] -> [#xx]. ATA will dial out immediately.

When the dialing number is [110, 111, 112 ... 119], it matched [Dial Now Rule] -> [11x]. ATA will dial out immediately.

When the dialing number is [123456], it matched [Dial Now Rule] -> [xxxxxx]. ATA will dial out immediately.

Example 2: PSTN feature code function.

Step 1: In [Dial Plan Setting] page, the configuration is [Auto PSTN Backup: Enable, PSTN feature Code: *22]. See Figure 2.

Auto PSTN backup:	Disable ▾
PSTN feature code:	*22

(Figure 2)

Description 1:

When ATA registered to SIP Server fail, its FXS phone line was forced to connect with PSTN line when user is going to make a call. The Dial Tone was provided from PSTN line instead.

Description 2:

When ATA registered to SIP Server successfully, press dial code *22 from analog phone set to force ATA switch to PSTN line manually. The Dial Tone was provided from PSTN line instead.

Example 3: Routing function.

Step 1: In [Dial Plan Setting] page, configure Routing Type: FXO, Routing Rule: D007+009+0800]. See Figure 3.

Routing Type:	Disable ▾
Routing Rule:	D007+009+0800

(Figure 3)

Description 1:

When dialing these digits [0800024365], it matched content [0800] of [Routing Rule]. ATA dials these digits from FXO port.

Description 2:

When dialing these digits [00986123456], it matched content [009] of [Routing Rule]. ATA dials these digits from FXO port.

Description 3:

When dialing these digits [00782280220], it matched content [D007] of [Routing Rule]. ATA will drop [007] first and then dial remaining digits [82280220] from [FXO] port.

Example 4: # Format is %23 Function

Step 1: In [Dial Plan Setting] web page, Setup [Key As Send#: Disalbe, # Format is %23: Enable] (See Figure 4).

Inter Digit Time:	5	(seconds)
Key As Send #:	Disable	
# Format is %23:	Enable	

(Figure 4)

Step 2: Pick up the handset, Dial [123#], the correct dialing content is [123%23].

6.3 Call Services

6.3.1 Function

Provide Forward, Hotline, DND, Alarm function.

6.3.2 Instruction

Example 1: 1FXS, 2FXS and 1FXS+1PSTN ATA devices.

Figure 1 : 1FXS for ATA171plus and ATA-171.

Call Service

Forward Type	Forward Number	Ring
Disable ▾	<input type="text"/>	3 ▾
Hotline Type	Hotline Number	Delay
Disable ▾	<input type="text"/>	0 ▾
DND Type	During the DND (Do Not Disturb)	
Disable ▾	From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (HH:MM)	
Alarm Type	Alarm Time	
Disable ▾	<input type="text"/> : <input type="text"/> (HH:MM)	
<input type="button" value="Submit"/>		

(Figure 1)

Item	Description
Forward Type of phone1	Default is Disable. To configure Phone 1 forward type. Here provides 5 options: Disable, All (unconditional), Busy, No Answer, Busy or No Answer. NOTE : Please make sure your service provider support this forward function.
Forward Number of phone1	To configure Phone 1 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 1	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 2 to 8 rings. This mode only supports Forward Type: No Answer.
Hotline Type of phone 1	Phone 1 hotline function, default is disable. Enable: ATA will dial the hotline number immediately when you pick up phone. Note: You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 1	Configure Phone 1 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 1	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
DND Type of phone 1	Configure Phone 1 DND function, default is Disable. When you set to Enable , ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next row).
DND Time of	This command is to configure DND time period at phone 1.

Item	Description
phone 1	Default is From 0:0(start time) To 0:0(end of time) . The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
Alarm Type of phone 1	Default is Disable. Configure Phone 1 alarm function. When you set to enable, phone 1 will Ring according to pre-configured (see next row how to configure alarm time) alarm time . The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 1	Default is 0:0(Hour/Minute). The time format is 24 hours system(hh/mm). Each field allows to enter 2 digits number only.
Submit	Save the configuration.

Figure 2: 2FXS for ATA172plus and ATA-172.

Call Service

Forward Type	Forward Number	Ring
Disable <input type="button" value="v"/>	<input type="text"/>	3 <input type="button" value="v"/> Phone 1
Disable <input type="button" value="v"/>	<input type="text"/>	3 <input type="button" value="v"/> Phone 2

Hotline Type	Hotline Number	Delay
Disable <input type="button" value="v"/>	<input type="text"/>	0 <input type="button" value="v"/> Phone 1
Disable <input type="button" value="v"/>	<input type="text"/>	0 <input type="button" value="v"/> Phone 2

DND Type	During the DND (Do Not Disturb)	
Disable <input type="button" value="v"/>	From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (HH:MM)	Phone 1
Disable <input type="button" value="v"/>	From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (HH:MM)	Phone 2

Alarm Type	Alarm Time	
Disable <input type="button" value="v"/>	<input type="text"/> : <input type="text"/> (HH:MM)	Phone 1
Disable <input type="button" value="v"/>	<input type="text"/> : <input type="text"/> (HH:MM)	Phone 2

(Figure 2)

Item	Description
Forward Type of phone1	Default is Disable. To configure Phone 1 forward type. Here provides 5 options: Disable, All (unconditional), Busy, No Answer, Busy or No Answer. NOTE : Please make sure your service provider support this forward function.
Forward Number of phone1	To configure Phone 1 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 1	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 1 to 6 rings. This mode only supports Forward Type: No Answer.
Forward Type of	Default is Disable. To configure Phone 2 forward type.

Item	Description
phone 2	Here provides 5 options: Disable, All (unconditional), Busy, No Answer, Busy or No Answer. NOTE : Please make sure your service provider support this forward function.
Forward Number of phone 2	To configure Phone 2 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 2	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 1 to 6 rings. This mode only supports Forward Type: No Answer.
Hotline Type of phone 1	Phone 1 hotline function, default is disable. Enable: ATA will dial the hotline number immediately when you pick up phone. Note: You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 1	Configure Phone 1 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 1	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default delay time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
Hotline Type of phone 2	Phone 2 hotline function, default is disable. Enable: ATA will dial the hotline number immediately when you pick up phone. Note: You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 2	Configure Phone 2 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 2	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default delay time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
DND Type of phone 1	Configure Phone 1 DND function, default is Disable. When you set to Enable , ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next row).
DND Time of phone 1	This command is to configure DND time period at phone 1. Default is From 0:0(start time) To 0:0(end of time) . The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
DND Type of phone 2	Configure Phone 2 DND function, default is Disable. When you set to Enable , ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next

Item	Decription
	row).
DND Time of phone 2	This command is to configure DND time period at phone 2. Default is From 0:0(start time) To 0:0(end of time) . The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
Alarm Type of phone 1	Default is Disable. Configure Phone 1 alarm function. When you set to enable, phone 1 will Ring according to pre-configured (see next row how to configure alarm time) alarm time . The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 1	Default is 0:0(Hour/Minute). The time format is 24 hours system(hh/mm). Each field allows to enter 2 digits number only.
Alarm Type of phone 2	Default is Disable. Configure Phone 2 alarm function. When you set to enable, phone 2 will Ring according to pre-configured (see next row how to configure alarm time) alarm time . The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 2	Default is 0:0(Hour/Minute). The time format is 24 hours system(hh/mm). Each field allows to enter 2 digits number only.
Submit	Save the configuration.

Figure 3: FXS+FXO, ATA-171M device.

Call Service

Forward Type	Forward Number	Ring
Disable ▾	<input type="text"/>	3 ▾
Hotline Type	Hotline Number	Delay
Disable ▾	<input type="text"/>	0 ▾
DND Type	During the DND (Do Not Disturb)	
Disable ▾	From <input type="text"/> : <input type="text"/> To <input type="text"/> : <input type="text"/> (HH:MM)	
Alarm Type	Alarm Time	
Disable ▾	<input type="text"/> : <input type="text"/> (HH:MM)	
<input type="button" value="Submit"/>		

(Figure 3)

Item	Decription
Forward Type of phone1	Default is Disable. To configure Phone 1 forward type. Here provides 5 options: Disable, All (unconditional), Busy, No Answer, Busy or No Answer. NOTE : Please make sure your service provider support this forward function.
Forward Number	To configure Phone 1 forward number, simply dial [number or

Item	Description
of phone1	digit string]. The maximum digit length is 63.
Rings of phone 1	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 2 to 8 rings. This mode only supports Forward Type: No Answer.
Hotline Type of phone 1	Phone 1 hotline function, default is disable. Enable: ATA will dial the hotline number immediately when you pick up phone. Note: You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 1	Configure Phone 1 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 1	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
DND Type of phone 1	Configure Phone 1 DND function, default is Disable. When you set to Enable , ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable , Always , Period (DND enable according pre-defined time period, refer to time setting at next row) .
DND Time of phone 1	This command is to configure DND time period at phone 1. Default is From 0:0(start time) To 0:0(end of time) . The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
Alarm Type of phone 1	Default is Disable. Configure Phone 1 alarm function. When you set to enable, phone 1 will Ring according to pre-configured (see next row how to configure alarm time) alarm time . The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 1	Default is 0:0(Hour/Minute). The time format is 24 hours system(hh/mm). Each field allows to enter 2 digits number only.
Submit	Save the configuration.

6.3.3 Operational Description

Example 1: Forward Feature

◆ All (Unconditional Forward)

Step 1: In [Call Service Setting] web page, Setup [Forward Type: All, Forward Number: 812345678] (See Figure 1).

Forward Type	Forward Number	Ring
Always ▼	812345678	3 ▼

(Figure 1)

Step 2: When receiving a new incoming call, LP399 will forward to this number [Forward Number: 812345678] automatically.

◆ Busy (Busy Forward)

Step 1: In [Call Service Setting] web page, Setup [Forward Type: Busy, Forward Number: 405] (See Figure 2).

Forward Type	Forward Number	Ring
Busy	405	3

(Figure 2)

Step 2: When LP399 is busy, it will forward to [Forward Number: 405] automatically.

◆ No Answer (No Answer Forward)

Step 1: In [Call Service Setting] web page, Setup [Forward Type: No Answer, Forward Number: 031237788, Rings: 3] (See Figure 3).

Forward Type	Forward Number	Ring
No Answer	031237788	3

(Figure 3)

Step 2: When LP399 rings 3 times and nobody answer the phone, it will forward to [Forward Number: 031237788].

◆ Busy or No Answer (Busy Forward & No Answer Forward)

Step 1: In [Call Service Setting] web page, Setup [Forward Type: Busy or No Answer, Forward Number: 0800024365, Rings: 3] (See Figure 4).

Forward Type	Forward Number	Ring
Busy or No Answer	0800024365	3

(Figure 4)

Step 2: When LP399 rings 3 times and if nobody answer the phone or the phone is busy, it will forward to [Forward Number: 0800024365]

Example 2: Hotline Feature

Example 2: Hotline Feature

◆ Dial SIP Account

Step 1: In [Call Service Setting] web page, Setup [Hotline Type: Enable, Hot Line number: 82341234, Delay: 3] (See Figure 5).

Hotline Type	Hotline Number	Delay
Enable	82341234	3

(Figure 5)

Step 2: When picking up LP399 phone and wait for 3 seconds delay time, it will dial to [Hot Line number: 82341234] automatically.

◆ Dial IP Address

Step 1: In [Call Service Setting] web page, Setup [Hotline Type: Enable, Hot Line number: 192.168.50.4, Delay: 3] (See Figure 6).

Hotline Type	Hotline Number	Delay
Enable ▾	192.168.50.4	3 ▾

(Figure 6)

Step 2: When picking up LP399 phone and wait for 3 seconds delay time, it will dial to [Hot Line number: 192.168.50.4] automatically.

Example 3: DND Feature

◆ Period

Step 1: In [Call Service Setting] web page, Setup [DND Type: Period, During the DND From: 18:15, To: 22:20] (See Figure 7).

DND Type	During the DND (Do Not Disturb)
Period ▾	From 18 : 15 To 22 : 20 (HH:MM)

(Figure 7)

Step 2: When receiving a new call during DND time period, the caller will hear “busy tone”.

◆ Always

Step 1: In [Call Service Setting] web page, Setup [DND Type: Always] (See Figure 8).

DND Type	During the DND (Do Not Disturb)
Always ▾	From 18 : 15 To 22 : 20 (HH:MM)

(Figure 8)

Step 2: When receiving a new call, the caller will hear “busy tone”.

Example 4: Alarm Feature

Step 1: In [Call Service Setting] web page, Setup [Alarm Type: Enable, Alarm Time: 21:00] (See Figure 9).

Alarm Type	Alarm Time
Enable ▾	21 : 0 (HH:MM)

(Figure 9)

Step 2: At 21:00 everyday, the alarm will start to work and last for 1 minute. After 1 minute, the alarm will stop. During ringing and pick up the phone, the alarm will stop automatically.

6.4 General

6.4.1 Function

Provide Caller ID, Call waiting, auto answer and T.38 FAX transmission.

6.4.2 Instruction

Figure 1: 1FXS(ATA171plus, ATA-171), 2FXS(ATA172plus, ATA-172) and 1FXS+1PSTN (ATA-171P)

General Setting

Call Waiting:	Enable	▼
Ring Timeout:	60	▼ (seconds)
Caller ID Scheme:	FSK (Bellcore)	▼
CID Type II:	Enable	▼
T.38 (FAX):	Enable	▼
FAX Pass-Through Codec:	uLaw	▼

(Figure 1)

Item	Description
Call Waiting	Enable/Disable call waiting function. This feature allows you to answer the incoming call when you are on line at another call. When you are on line to talk, an “Du Du” sound was heard to remind you there is an incoming call. To answer this incoming call, simple activate HOLD feature to hold existing call and answer incoming call.
Ring Timeout	Default setting is 60 seconds. ATA responses Busy tone (SIP command 486) to caller when nobody answer incoming call once configured time was expired. The configured timeout option is : 20, 40, 60, 80, 120, 180 and 240 seconds.
Caller ID Scheme	Default is Disable. The Caller ID supports the following protocol: FSK Bellcore, DTMF (Caller ID before first Ring), CID-Japan, DTMF-Brazil and DTMF-Denmark mode. Note: Your analog telephone set MUST support proper Caller ID mode to show CID number.
CID Type II	Default is Disable. If ATA has enabled both CID Type II and Call Waiting function, ATA will show the incoming call Caller ID when you are on line(busy). Note: Your analog telephone set MUST support proper Caller ID Type II mode to show CID number.
T.38 (FAX)	Default is Enable to support T.38 FAX transmission function.
T.38 Pass-through codec	Default codec is G.711 u-Law (enable) to support T.38 FAX pass through. ATA only uses codec either G.711 u-law or G.711 a-law to transmit FAX over T.38 protocol.
Submit [button]	Save the configuration.

Figure2: 1FXS+1FXO, ATA-171M

General Setting

Call Waiting:	Enable ▼
Ring Timeout:	60 ▼ (seconds)
Caller ID Scheme:	FSK (Bellcore) ▼
CID Type II:	Disable ▼
T.38 (FAX):	Enable ▼
FAX Pass-Through Codec:	uLaw ▼
Auto Answer Type:	Trunk Gateway ▼
Auto Answer Counter:	3 ▼
PIN Code:	Disable ▼
PIN Code Number:	<input type="text"/>

(Figure 2)

Item	Decription
Call Waiting	Enable/Disable call waiting function. This feature allows you to answer the incoming call when you are on line at another call. When you are on line to talk, an “Du Du” sound was heard to remind you there is an incoming call. To answer this incoming call, simple activate HOLD feature to hold existing call and answer incoming call.
Ring Timeout	Default setting is 60 seconds. ATA responses Busy tone (SIP command 486) to caller when nobody answer incoming call once configured time was expired. The configured timeout option is : 20, 40, 60, 80, 120, 180 and 240 seconds.
Caller ID Scheme	Default is Disable. The Caller ID supports the following protocol: FSK Bellcore, DTMF (Caller ID before first Ring), CID-Japan, DTMF-Brazil and DTMF-Denmark mode. Note: Your analog telephone set MUST support proper Caller ID mode to show CID number.
CID Type II	Default is Disable. If ATA has enabled both CID Type II and Call Waiting function, ATA will show the incoming call Caller ID when you are on line(busy). Note: Your analog telephone set MUST support proper Caller ID Type II mode to show CID number.
T.38 (FAX)	Default is Enable to support T.38 FAX transmission function.
T.38 Pass-through codec	Default codec is G.711 u-Law (enable) to support T.38 FAX pass through. ATA only uses codec either G.711 u-law or G.711 a-law to transmit FAX over T.38 protocol.
FXO Setting	FXO interface configuration.
Auto Answer	Default is Disable. Define an incoming call at Auto Answer method. There are configuration option: Disable , IP In , FXO In , Both and Trunk Gateway . IP In: IP incoming call auto switch to FXO port after pre-configured Ring Cycles was expired. FXO In: PSTN incoming call auto switch to IP SIP call after pre-configured Ring Cycles was expired.

Item	Description
	<p>Both: IP or FXO incoming call auto switch to FXO or IP correspondant after pre-configured Ring Cycles was expired.</p> <p>Trunk Gateway: ATA forwards an IP incoming call from SIP Proxy to FXO port directly. This is an VoIP termination to local PSTN feature.</p> <p>NOTE: Trunk Gateway function doesn't work with PIN Code authentication function when terminate an call to FXO port. Both SIP Server and ATA-171M MUST support this feature to implement it.</p>
Auto Answer Counter	Default is 3 rings. ATA will switch to SIP IP port or FXO port automatically and provide second dial tone after pre-configured ring cycles count arrived. Ring count option is 0 to 8.
PIN Code	<p>Default is Disable. This feature provides Password (PIN Code) authorization when ATA receive an incoming call. ATA will require PIN code authorization when call is coming to ATA. The calling user has to enter PIN code for ATA to verify before call was established .</p> <p>Note:</p> <ol style="list-style-type: none"> 1. This function Only work with [Auto Answer] function. 2. When an incoming call from FXO port, ATA only accepts PIN code DTMF via In-Band and RFC2833. However, an incoming call from SIP IP trunk PIN code DTMF, ATA only accepts RFC2833 (Not support In-Band DTMF).
PIN Code Number	Configure PIN code password. Allow number only and Maximum length is 31 digits. When ATA answers an incoming call, the caller has to enter pre-configured PIN code number for ATA to verify. If password is correct, caller will hear second dial tone and continue to dial.
Submit [button]	Save the configuration.

6.5 Volume

6.5.1 Function

This function is to adjust volume of microphone and speaker at analog phone set, and FXO port's volume as well.

6.5.2 Instruction

Figure1: ATA171plus, ATA-171, ATA172plus, ATA-172, ATA-171P

Volume Setting

Handset Volume:	10 ▼
Handset Gain:	10 ▼

(10 representative is 0 dB and every scale is 3 dB)

(Figure 1.)

Item	Decription
Handset Volume	Default is 10. Control the volume of the Handset receiver from (0 to 14). Maximum length is 2 digits.
Handset Gain	Default is 10. Control the handset gain (microphone volume to send to remote site) from (0 to 15). Maximum length is 2 digits.
Submit [button]	Save the configuration.

Figure 2: ATA171plus, ATA-171 and ATA-171M

Volume Setting

Handset Volume:	10 ▼
Handset Gain:	10 ▼
PSTN-Out Volume:	10 ▼
PSTN-In Gain:	10 ▼

(10 representative is 0 dB and every scale is 3 dB)

(Figure 2.)

Item	Decription
Handset Volume	Default is 10. Control the volume of the Handset receiver from (0 to 14). Maximum length is 2 digits.
Handset Gain	Default is 10. Control the handset gain (microphone volume to send to remote site) from (0 to 15). Maximum length is 2 digits.

PSTN-Out Volume	Default is 10. Adjust the volume from FXO to IP port (0 to 12).
PSTN-In Gain	Default is 10. Adjust the volume from IP to FXO port (0 to 12).
Submit [button]	Save the configuration.

7. Network environment

Provide [WAN, DDNS, VLAN, VPN, SNTP] function setting.

7.1 WAN (network setting)

7.1.1 Function

WAN provides function to set up WAN port network IP address with fixed IP, DHCP Client and PPPoE.

7.1.2 Instruction

WAN Setting

WAN Active:	DHCP
IP Address:	192.168.23.21
Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254
DNS Active:	Static
Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1
MAC Address:	00:01:a9:39:90:00
System Name:	VOIP_PHONE

Submit

Item	Decription
WAN Active	Default: DHCP Client. Setup the network connecting type including fixed IP, DHCP Client and PPPoE. Fixed IP: Enter a static IP address. DHCP Client: Get IP address from DHCP server. PPPoE: Uses PPPoE to connect IP network. Provide options: Fixed IP, DHCP Client, and PPPoE.
IP Address	Show the current IP address, enter type is xxx.xxx.xxx.xxx of 15 bytes. *If you want to setup the IP address, please Setup [TYPE] to [Fixed IP] first. Then you can enter IP address.
Subnet Mask	Shows the current Subnet Mask IP Address, the enter type is xxx.xxx.xxx.xxx of 15 bytes.
Default Gateway	Shows current Default Gateway IP Address, the enter type is xxx.xxx.xxx.xxx of 15 bytes.
DNS Active	Default: Auto. Fixed: Setup DNS Server address. Auto: Get DNS Server address from DHCP Server and this option only supports DHCP Client and PPPoE. Provide options: Fixed, Auto.

Primary DNS	Default: 168.95.192.1. The enter type is xxx.xxx.xxx.xxx of 15 bytes.
Secondsond DNS	Default: 168.95.1.1. The enter type is xxx.xxx.xxx.xxx of 15 bytes.
MAC Address	Shows the MAC ID address.
System Name	Default: VOIP_Phone. This column can enter numbers and strings; maximum length is 15 bytes.
PPPoE User	This column can enter numbers and strings; maximum length is 32 bytes.
PPPoE Password	This column can enter numbers and strings; maximum length is 32 bytes.
PPPoE Service Name	This column can enter numbers and strings; maximum length is 32 bytes. <i>*The information of this column is provided by ISP, if you don't known what is the data, do not Setup it.</i>
PPPoE AC Name	This column can enter numbers and strings; maximum length is 32 bytes. <i>*The data of this column is provided by ISP, if you don't known what is the data, do not setup it.</i>
Submit [Button]	Save the settings.

7.1.3 Operate Instruction

Example 1: Check Host Name

Step 1: In [WAN Setting] web page, Setup [WAN Active: DHCP, System Name: VOIP_PHONE] (See Figure 1).

WAN Setting

WAN Active:	DHCP
IP Address:	192.168.23.21
Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254
DNS Active:	Static
Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1
MAC Address:	00:01:a9:39:90:00
System Name:	VOIP_PHONE

Submit

(Figure 1)

Step 2: In [System Status] web page, User can view [WAN] port network status (See Figure 2).

System Status

WAN Information			
Link Status:	Connected	Active:	DHCP Client
IP Address:	192.168.23.21	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1	MAC Address:	00:01:a9:39:90:00

(Figure 2)

Example 2: Check PPPoE Service Name & AC Name

Step 1: In [WAN Setting] web page, Setup [WAN Active: PPPoE, PPPoE User: test@hinet.net, PPPoE Password: test] (See Figure 3).

WAN Setting

WAN Active:	PPPoE
IP Address:	192.168.23.21
Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254
DNS Active:	Static
Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1
MAC Address:	00:01:a9:39:90:00
System Name:	VOIP_PHONE
PPPoE User:	test@hinet.net
PPPoE Password:	●●●●●●
PPPoE Service Name:	
PPPoE AC Name:	

(Figure 3)

Step 2: In [System Status] web page, User can view [WAN] port network status [Type: PPPoE Client] (See Figure 4).

System Status

WAN Information			
Link Status:	Connected	Active:	PPPoE Client
IP Address:	0.0.0.0	Subnet Mask:	0.0.0.0
Default Gateway:	0.0.0.0	Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1	MAC Address:	00:01:a9:39:90:00

(Figure 4)

7.2 DDNS (Dynamic DNS Settings)

7.2.1 Function

Dynamic DNS provides a residential user's Internet gateway that has a variable, often changing IP address with a well known hostname resolvable through standard DNS queries.

7.2.2 Instruction

Dynamic DNS Setting

DDNS Active:	Disable ▾
Host Name:	<input type="text"/>
User Name:	<input type="text"/>
Password:	<input type="text"/>
E-mail Address:	<input type="text"/>
DDNS Server List:	members.dyndns.org ▾
DDNS Server:	<input type="text"/>
Dynamic DNS Type:	dyndns ▾
Wild Card:	Disable ▾
BACKMX:	Disable ▾
Off Line:	Disable ▾ (Only applies to custom DNS)

Item	Decription
DDNS Active	Default is Disable; The DDNS function will be enabled when you set to Enable.
Host name	Enter Host name which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Length is 63 digits.
User Name	Enter user's name for registering to DDNS Server.
Password	Enter the password. Maximum length is 63 digits.
E-mail address	Enter E-mail address. Maximum length is 63 digits.
DDNS Server List	Default is Disable; Configure your service provider here. Provide option: User input , members.dyndns.rog and www.dtdns.com mode.
DDNS Server	Enter DDNS Server which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits.
Type	Default is dyndns. Provide 3 options: dyndns , statdns and customer mode .
Wild Card	Default is Enable. Provides 3 options: Enable, Disable and Nochg. NOTE: Please make sure your DDNS provider supports this feature when you set it to enable.
BACKMX	Default is Disable; The backup MX function will was activate when you set it to enable. It provides two options: Disable and Enable. NOTE: Please make sure your DDNS provider supports this feature when you set it to enable.

Item	Description
Off Line	Default is Disable. The Off Line function will be activate when you set it to enable. It provides two options: Disable and Enable. NOTE: Please make sure your DDNS provider supports this feature when you set it to enable.
Submit [button]	Save the configuration.

- *: 1. Not all DNS providers can support this function. If you want to use this function, please contact with your provider.
2. Query DNS data did not update in real time because of the DNS server updated cycle time. Please make sure the DNS server updated correct time or login DNS server to check if the IP address is correct or not.

7.3 VLAN

7.3.1 Function

Provide Network, SIP and RTP VLAN function. **This feature needs to work with VLAN Router.**

7.3.2 Instruction

VLAN Setting

Network (Both WAN & LAN)

VLAN Active:	<input type="text" value="Disable"/>	
VID (802.1Q/TAG):	<input type="text" value="136"/>	(3~4094)
User Priority (802.1P):	<input type="text" value="7"/>	

SIP & RTP

SIP VID:	<input type="text" value="0"/>	(3~4094, 0: Disable)
SIP User Priority (802.1P):	<input type="text" value="0"/>	
RTP VID:	<input type="text" value="0"/>	(3~4094, 0: Disable)
RTP User Priority (802.1P):	<input type="text" value="0"/>	

Item	Description
Network (Both WAN & LAN)	Configure VLAN function of both WAN and LAN ports.
VLAN Packets	Default is Disable. ATA supports VLAN function (accept VLAN packets) when you set it to enable.
VID (802.1Q/TAG)	Default: 136. Provide Virtual LAN ID (VLAN or VID) for VLAN Server. Data range: 3~4097. Maximum length is 4 digits.
User Priority (802.1P)	Default: 0. Set the user's priority. Data range: 0 to 7.
SIP & RTP	Define SIP & RTP VLAN feature.
SIP VID	Default is 0 (disable). This feature is to define SIP VLAN ID. Range is 3~4094. This feature is an independent one which do not need to Enable [VLAN Packets] if you want to enable this feature. Data length is 4 digits.
User Priority (802.1P)	Default is 0 (disable). Define SIP package priority. Range is 0~7.
RTP VID	Default is 0 (disable). Define SIP VLAN ID. Range is 3~4094. This feature is an independent one which do not need to Enable [VLAN Packets] if you want to enable this feature. Data length is 4 digits.
User Priority (802.1P)	Default is 0 (disable); Define RTP package priority. Range: 0~7
Submit [button]	Save the configuration.

7.4 VPN (PPTP/L2TP connection)

7.4.1 Function

Provide [PPTP or L2TP] configuration. **Note:** After you have enabled the VPN functions, you could login in LP399 to configure via LAN port.

7.4.2 Instruction

VPN Setting

VPN Active:	<input type="text" value="PPTP"/>
Server Name:	<input type="text" value="192.168.22.120"/>
User Name:	<input type="text" value="test"/>
Password:	<input type="password" value="••••"/>
Port:	<input type="text" value="Default"/> <input type="text" value="1723"/> (1024~65535, Only Support PPTP)

Item	Decription
Type	Default is Disable. Provide PPTP/L2TP connection mode.
Server Name	Enter PPTP/L2TP Server information which can be IP Address or Domain Name. Format is : xxx.xxx.xxx.xxx. Maximum length is 63 digits.
User Name	Enter PPTP/L2TP Server user's name or IP address which can be number or strings. Maximum length is 63 digits.
Password	Enter PPTP/L2TP password which can be numbers or strings. Maximum length is 63 digits.
Port Number	Default PPTP Port is 1723. Prodive 2 options: Default, Customer (User define port) and the range is 1024~65535. Only number and length is 5 digits.
Submit [button]	Save the configuration.

7.5 NTP (Time Environment)

7.5.1 Function

Provide time synchronization and daylight saving function.

7.5.2 Instruction

NTP Setting

NTP Active:	Auto ▼
Primary NTP:	north-america.pool.ntp.org
Secondary NTP:	asia.pool.ntp.org
Time Zone:	GMT + ▼ 08 ▼ : 00 ▼ (HH:MM)
Update Interval:	6 Hour ▼

Manually Time	(Not use Daylight Saving Time)
Date & Time	2005 Year 1 Month 1 Date 8 Hour 9 Minute 48 second
	<input type="button" value="Get PC Time"/>

Daylight Saving Time :	Disable ▼
Offset:	+ 1 Hour ▼
Start Time:	Jan ▼ By Day ▼ 01 ▼ First Week ▼ Sun ▼ 00 ▼
End Time:	Jan ▼ By Day ▼ 01 ▼ First Week ▼ Sun ▼ 00 ▼

Item	Decription
NTP Active	Default: Enable; When Enable this function, the SNTP is on. Provide options: Disable, Enable.
Primary NTP	Default: north-america.pool.ntp.org. This column can enter IP or Domain Name with the format of xxx.xxx.xxx.xxx; maximum length is 63 bytes.
Secondsondary NTP	Default: asia.pool.ntp.org. This column can enter IP or Domain Name with the format of xxx.xxx.xxx.xxx; maximum length is 63 bytes.
Time Zone	Default: GMT + 08:00 (hh:mm) and the format is (+/-, hh:mm). Provide options: +/-, 0~13(hh), 00, 15, 30, 45(mm).
Update Interval	Default: 6; Sync. Time will be checked with NTP Server according to the interval you have setup. Provide options: 1 min., 5 min., 30 min., 1 hour, 3 hour, 6 hour, 12 hour, 24 hour.
Manually Time	When enable this time function manually, Daylight Saving Time function will be disabled. <b style="color: red;">Note: If ATA electric power was removed, this time will be lost.
Date & Time	Year; Setup the Year; Only use number, Data Settings secondstion is 2011~2022, maximum length is 4 bytes. Month; Setup the Month; Only use number, Data Settings secondstion is 1~12, maximum length is 2 bytes. Date; Setup Date; Only use number, Data Settings secondstion

Item	Decription
	<p>is 1~31, maximum length is 2 bytes. Hour; Setup Hour; Only use number, Data Settings secondstion is 0~23, maximum length is 2 bytes. Minute; Setup Minute; Only use number, Data Settings secondstion is 0~59, maximum length is 2 bytes. Secondsond; Setup Secondsond; Only use number, Data Settings secondstion is 0~59, maximum length is 2 bytes.</p>
Get PC Time [Button]	Get local personal computer date and time information.
Daylight Saving Time	<p>Default: Disable. When Enable this function, the Daylight Saving is on. Provide options: Disable, Enable.</p>
OffSet up	<p>Default: +2 Hour. Set up the Daylight Saving Time difference. Provide options: -2 hour, -1 hour, +1 hour, +2 hour.</p>
Start Time	<p>Setup Daylight Saving Time. You can select the start date by day or week. Setup beginning month: Default Setting is January. Here offers options from January to December. Day of Month : Default Setting is 01. Here Provide options from 1th to 31th. Week of Month: Selects the effective week. Here Provide options for Last Week, Last Secondsond Week, Week1, Week2 and Week3 ◦ Day : Provide options: Sun, Mon, Tue, Wed, Thu, Fri, Sat. Start Time : 00~23.</p>
End Time	<p>Stop Daylight Saving Time Setting. You can select the end date by day or week. Setup ending month: Default Setting is January. Here offer options froms Jan to Dec. Day of Month : Default Setting is 01. Here Provide options from 1th to 31th. Week of Month: Selects the effective week. Here Provide options for Last Week, Last Secondsond Week, Week1, Week2 and Week3 ◦ Day : Provide options: Sun, Mon, Tue, Wed, Thu, Fri, Sat. Start Time : 00~23.</p>
Submit [Button]	Save the Settings.

7.5.3 Operate Instruction

Example 1: NTP Setting

Step 1: In [NTP Setting] web page, Setup [NTP Active: Auto, Primary NTP: north-america.pool.ntp.org, Secondsondary NTP: asia.pool.ntp.org, Time Zone: GMT+ 08:00, Update Interval: 6 Hour] (See Figure 1).

NTP Active:	Auto
Primary NTP:	north-america.pool.ntp.org
Secondary NTP:	asia.pool.ntp.org
Time Zone:	GMT + 08 : 00 (HH:MM)
Update Interval:	6 Hour

(Figure 1)

Step 2: In [System Status] web page, View the [Current Time] field, The field show time information now (See Figure 2).

System Information			
Model Name:	LP399 Plus	Version:	LP399_Plus_V3.5
Firmware Version:	2.0.18-1-1312160	DSP Version:	WM-1208270
Current Time:	2014-02-10 11:09	Update Date:	
System Up Time:	0 day(s) 0 hour(s) 0 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 0 minute(s)		

(Figure 2)

Example 2: DST Setting (Daylight Saving Time)

Step 1: In [NTP Setting] web page, Setup [NTP Active: Auto, Primary NTP: north-america.pool.ntp.org, Secondsondary NTP: asia.pool.ntp.org, Time Zone: GMT+ 08:00, Update Interval: 6 Hour, Daylight Saving Time: Enable, OffSet up: +1 Hour, Start Time: Aug, By Week, Last Week, Mon, 01, End Time: Oct, By Week, Last Week, Fri, 18] (See Figure 3).

NTP Setting

NTP Active:	Auto
Primary NTP:	north-america.pool.ntp.org
Secondary NTP:	asia.pool.ntp.org
Time Zone:	GMT + 08 : 00 (HH:MM)
Update Interval:	6 Hour

Manually Time	(Not use Daylight Saving Time)
Date & Time	2014 Year 2 Month 10 Date 11 Hour 11 Minute 25 second
	Get PC Time

Daylight Saving Time :	Enable
Offset:	+ 1 Hour
Start Time:	Aug By Week 01 Last Week Mon 01
End Time:	Oct By Week 01 Last Week Fri 18

Submit

(Figure 3)

Step 2: In [System Status] web page, View the [Current Time] field, The field show [Daylight Saving Time] time information.

Example 3: Manually Setup NTP

Step 1: In [NTP Setting] web page, Setup [NTP Active: Manual] , Press [Get PC Time] button to get personal computer's date and time (For example : Date & Time: 2005 Year, 1 Month, 1 Date, 0 Hour, 0 Min, 23 second) (See Figure4).

Manually Time	(Not use Daylight Saving Time)											
Date & Time	2005	Year	1	Month	1	Date	0	Hour	0	Minute	23	second
	<input type="button" value="Get PC Time"/>											

(Figure 4)

Step 2: In [System Status] web page, View the [Current Time] field, The field show [Manually Time] time information (See Figure 5).

Manually Time	(Not use Daylight Saving Time)											
Date & Time	2014	Year	2	Month	10	Date	11	Hour	22	Minute	10	second
	<input type="button" value="Get PC Time"/>											

(Figure 5)

8. NAT

Provide [LAN, DMZ & MAC Clone, Virutal Server] function.

8.1 LAN

8.1.1 Function

Provide LAN port configuration setting including DHCP server function.

8.1.2 Instruction

LAN Setting

Device Active: Router ▼

LAN IP Address:

LAN MAC Address:

Enable DHCP Server: Enable ▼

IP Address: ~ (Start ~ End, 1~254)

Lease Time: (10~10080 Minute)

Index	Assign IP Address	MAC Address	Lease Time (sec)
1	192.168.123.150	54:42:49:87:C5:1F	85972

Item	Decription
Device Active	Default: NAT. Setup the routing function of LAN Port. Provide options: Bridge, NAT. Bridge: When you Setup Bridge, [WAN & LAN] Port are all in the same IP domain, just like in the same network switch. NAT: When you Setup NAT, [WAN & LAN] Port are in the different IP domain. LAN Port work as DHCP server when you Enable DHCP Server Active function.

LAN IP Address	Default: 192.168.123.1. The type is xxx.xxx.xxx.xxx of 15 bytes.
LAN MAC Address	Show LAN MAD ID address.
Enable DHCP Server	Default: Enable; When you Disable this function, LAN Port will not be a DHCP Server. Provide options: Enable, Disable.
IP Address	Default: 150~200; Setup the IP address range of DHCP Server. This column can only be numbers; maximum length is 3 bytes with the range of 1~254.
Lease Time	Default is 1440 (min), IP address rented deadline. Only use number, Data setting range is 10~17820, maximum length is 5 bytes.
Submit [Button]	Save the Settings.
Index	Shows the serial number.
Assign IP Address	LP399 DHCP server appoint IP address.
MAC Address	Connect ATA LAN port device's MAC address.
Lease Time(sec)	LAN port device connection time.
Refresh [Button]	Reload DHCP device status.

8.1.3 Operate Instruction

Example 1: LAN Mode: Bridge

Step 1: In [LAN Setting] web page, Setup [Device Active: Bridge] (See Figure 3).

LAN Setting

Device Active:

(Figure 3)

Step 2: In [System Status] web page, [LAN Inofrmation] was no show. (See Figure 4).

System Status

WAN Information

Link Status:	Connected	Active:	Fixed IP Client
IP Address:	192.168.22.37	Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254	Primary DNS:	164.124.101.2
Second DNS:	203.248.252.2	MAC Address:	00:01:a8:71:02:32

System Information

Model Name:	ATA172 Plus	Version:	ATA172_Plus_V3.3
Firmware Version:	2.0.14-1-1210172	DSP Version:	NV-1106080
Current Time:	2015-01-07 11:24		
System Up Time:	0 day(s) 0 hour(s) 0 minute(s)		
Network Link Up Time:	0 day(s) 0 hour(s) 0 minute(s)		

(Figure 4)

8.2 DMZ & MAC Clone

8.2.1 Function

Provide DMZ and MAC Clone configuration.

8.2.2 Instruction

DMZ and MAC Clone Setting

DMZ Active:	Disable ▾
DMZ IP Address:	1.1.1.1
MAC Clone Active:	Disable ▾

Submit

Item	Decription
DMZ Type	Default is Disable. When set to Enable, all network packages will be sent to the IP address which was defined from [Assigned IP Address].
DMZ IP Address	Default: 0.0.0.0. The assigned IP address type is xxx.xxx.xxx.xxx of maximum 15 bytes.
MAC Clone Type	Default is Disable. When it was set to Enable, ATA will get your PC computer's MAC address. Option: Disable, Enable. NOTE: 1. When use MAC Clone function, ATA must set "LAN Mode" to NAT mode and enable DHCP server. 2. You have to login ATA to configure via LAN port to set the MAC Clone function.
Submit [button]	Save the configuration.

NOTE :

If you want to resume your original MAC address, please use "Restore Default Setting" command at Webpage configuration.

8.3 Virtual Server

8.3.1 Function

Provide 12 sets of Virtual Server.

8.3.2 Instruction

Virtual Server Setting

Index	Active	Protocol	Internet Port		Extranet Port		Server IP Address	Action
			Start ~ End	Start ~ End	Start ~ End	Start ~ End		
1	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
2	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
3	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
4	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
5	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
6	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
7	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
8	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
9	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
10	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
11	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	
12	<input type="checkbox"/>	TCP ▾	<input type="text"/> ~ <input type="text"/>	<input type="text"/> ~ <input type="text"/>	<input type="text"/>	<input type="text"/>	Delete	

Item	Description
Index	Index number to support 12 sets configuration.
Enable	Default is Disable to all sets. When setting Enable, this function will be started.
Protocol	Default is TCP. Protocol option is : TCP or UDP.
Internet Port	Define the intranet port. Range is 1~65533. Here can define a range of ports or fixed port.
Extranet Port	Define the extranet port. Range is 1~65533. Here can define a range of ports or fixed port.
Server IP	Define internet server IP address which can only be IP address and the format is xxx.xxx.xxx.xxx. The address range is 15 digits.
Action	Click the delete button to clear the setting.
Submit [button]	Save the configuration.
Delete All [button]	Clear all configuration in virtual server setting web page.

NOTE :

When you define the Internet & Extranet Ports, please avoid from using ATA default ports. For example: 5060, 9999, 20000.

8.3.3 Operate Instruction

Step 1: In [Virtual Server Setting] web page, Setup [Index: 1, Active :select, Protocol: TCP, Internet Port: 80~80, Extranet Port: 8080~8080, Server IP Address: 192.168.123.150; Index: 2, Active:select, Protocol: TCP, Internet Port: 600~600, Extranet Port: 600~600, Server IP Address: 192.168.123.15] (See Figure 1).

Index	Active	Protocol	Internet Port		Extranet Port		Server IP Address	Action
			Start ~ End	Start ~ End				
1	<input checked="" type="checkbox"/>	TCP ▾	80 ~ 80	8080 ~ 8080	192.168.123.150	Delete		
2	<input checked="" type="checkbox"/>	TCP ▾	600 ~ 600	600 ~ 600	192.168.123.15	Delete		

(Figure 1)

Step 2: Use other personal computer to Setup IP 192.168.123.150, Then connect LP399's LAN port, Start personal computer Web Server function.

Step 3: Link to [<http://192.168.123.150:8080>], Show [Web Function Testing] web page.

9. SIP Setting

Provide Service, Code, Advanced and STUN function.

9.1 Service (SIP register setting)

9.1.1 Function

Provide 5 SIP register accounts to register 5 different SIP Server or IP-PBX.

9.1.2 Instruction

Figure 1: 1FXS (ATA171plus, ATA-171), 1FXS+1PSTN (ATA-171P) and 1FXS+1FXO (ATA-171M).

Service Domain Setting

Realm:

Realm Active:	<input type="text" value="Disable"/>
Display Name:	<input type="text"/>
Phone Number:	<input type="text"/>
Authentication ID:	<input type="text"/>
Authentication Password:	<input type="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Subscribe for MWI :	<input type="text" value="Disable"/>

(Figure 1)

Item	Decription
Realm No.	Default is 1 (The first register account). If you want to switch to 1 st account, please dial [1*] then hang up phone. Please refer to [Phone – Dial Plan Setting] -> [Realm 1~5 prefix].
Active	Default is Disable. This account will be active when you set to enable.
Display Name	Account's display name which can be numbers or strings. Maximum length is: 31 digits.
Phone Number	Account's phone number which only can be numbers. Maximum length is: 31 digits.
Authentication ID	Account's register ID which can be numbers or strings. Maximum length: 47 digits.
Authentication Password	Account's register password which can be numbers or strings. Maximum length is: 31 digits.
Domain Server	Input Domain Server. It can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For Example: nat.voiptalk.org:5065.
Proxy Server	Enter Proxy Server information. It can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If

Item	Decription
	special Port Address is needed, please add it behind. For instance: nat.voiptalk.org:5065.
Subscribe of MWI	Subscribe for MWI function (message waiting indicator). Your Register SIP Proxy server must support this function.
Submit [button]	Save the configuration.

Figure 2: 2FXS (ATA172plus or ATA-172)

Service Domain Setting

Phone: ▼

Realm: ▼

Realm Active: ▼

Display Name:

Phone Number:

Authentication ID:

Authentication Password:

Domain Server:

Proxy Server:

Subscribe for MWI : ▼

(Figure 2)

Item	Decription
Phone No	Default is Phone 1. Define phone 1~2 configuration.
Realm No.	Default is 1 (The first register account). If you want to switch to 1 st account, please dial [1*] then hang up phone. Please refer to [Phone – Dial Plan Setting] -> [Realm 1~5 prefix].
Active	Default is Disable. This account will be active when you set to enable.
Display Name	Account's display name which can be numbers or strings. Maximum length: 31 digits.
Phone Number	Account's phone number which can only be numbers. Maximum length: 31 digits.
Authentication ID	Account's register ID which can be numbers or strings. Maximum length: 47 digits.
Authentication Password	Account's register password which can be numbers or strings. Maximum length: 31 digits.
Domain Server	Enter Domain Server which can be IP Address or Domain Name. Format is : xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For Example: nat.voiptalk.org:5065.
Proxy Server	Enter Proxy Server information which can be IP Address or Domain Name. Format is : xxx.xxx.xxx.xxx. Maximum length is

Item	Description
	63 digits. If special Port Address is needed, please add it behind. For instance, nat.voiptalk.org:5065.
Subscribe of MWI	Subscribe for MWI function (message waiting indicator). Your Register SIP Proxy server must support this function.
Submit [button]	Save the configuration.

9.1.3 Operate Instruction

Example 1: Register at port other than 5060

Step 1: In [Service Domain Setting] web page, Set up [Realm Active: Enable, Display Name: 22061, Phoner Number: 22061, Authentication ID: 22061, Authentication Password: test, Domain Server: 61.62.236.71:6000, Proxy Server: 61.62.236.71:6000, Subscribe for MWI: Disable] (See Figure 1).

Since port 5060 is widely used in VoIP SIP port, hacker is easily to detect your device is an VoIP device (via port 5060) and invade your device to make expensive call. We strongly suggest not to use port 5060. Use another port to avoid this issue happening. In this example, We use SIP port 6000 instead. However, your registration SIP Server or IP-PBX Server should be able to accept SIP port other than 5060.

Realm:

Realm Active:	<input type="text" value="Enable"/>
Display Name:	<input type="text" value="22061"/>
Phone Number:	<input type="text" value="22061"/>
Authentication ID:	<input type="text" value="22061"/>
Authentication Password:	<input type="text" value="*****"/>
Domain Server:	<input type="text" value="61.62.236.71:6000"/>
Proxy Server:	<input type="text" value="61.62.236.71:6000"/>
Subscribe for MWI :	<input type="text" value="Disable"/>

(Figure 1)

Step 2: In [Service Domain Setting] web page (See Figure 2), the register status of that account is [System Status].

Realm:

Realm Active:	<input type="text" value="Enable"/>
Display Name:	<input type="text" value="22061"/>
Phone Number:	<input type="text" value="22061"/>
Authentication ID:	<input type="text" value="22061"/>
Authentication Password:	<input type="text" value="*****"/>
Domain Server:	<input type="text" value="61.62.236.71:6000"/>
Proxy Server:	<input type="text" value="61.62.236.71:6000"/>
Subscribe for MWI :	<input type="text" value="Disable"/>

(Figure 2)

Example 2: Enable Subscribe For MWI

Step 1: In [Service Domain Setting] web page, Setup [Realm Active: Enable, Display Name: 22061, Phoner Number: 22061, Authentication ID: 22061, Authentication Password: test, Domain Server: 61.62.236.71:6000, Proxy Server: 61.62.236.71:6000, Subscribe for MWI: Enable] (See Figure 3).

Realm:

Realm Active:	<input type="text" value="Enable"/>
Display Name:	<input type="text" value="22061"/>
Phone Number:	<input type="text" value="22061"/>
Authentication ID:	<input type="text" value="22061"/>
Authentication Password:	<input type="text" value="••••"/>
Domain Server:	<input type="text" value="61.62.236.71:6000"/>
Proxy Server:	<input type="text" value="61.62.236.71:6000"/>
Subscribe for MWI :	<input type="text" value="Enable"/>

(Figure 3)

9.2 Codec (Voice Format)

9.2.1 Function

Provide Voice Codec priority, RTP payload type and Codec ID setting.

9.2.2 Instruction

Figure 1: G.723 voice codec.

Codecs Setting

Disable Codecs		Enable Codecs
G.726 - 16 G.726 - 24 G.726 - 32 G.726 - 40	<input type="button" value=" >>"/> <input type="button" value=" <<"/>	G.711 u-law G.711 a-law G.723 G.729
<input type="button" value="Move"/>		<input type="button" value="Up"/> <input type="button" value="Down"/>

G.711 and G.729:	<input type="text" value="20"/> ms
G.723:	<input type="text" value="30"/> ms
G.723 5.3K:	<input type="text" value="Disable"/>
Silence Suppression (VAD):	<input type="text" value="Disable"/>
Echo Canceller :	<input type="text" value="Enable"/>

Codec Type	ID Value	
G726-16:	<input type="text" value="Default"/> <input type="text" value="23"/>	(95~127)
G726-24:	<input type="text" value="Default"/> <input type="text" value="22"/>	(95~127)
G726-32:	<input type="text" value="Default"/> <input type="text" value="2"/>	(95~127)
G726-40:	<input type="text" value="Default"/> <input type="text" value="21"/>	(95~127)
RFC 2833:	<input type="text" value="Default"/> <input type="text" value="101"/>	(95~127)

Item	Decription
Disabled Codecs	Default: G.726.16, G.726.24, G.726.32, G.726.40. Disable these voice Codecs.
>> <<	>>: Move to Enable Codec area. <<: Move to Disable Codec area.
Enabled Codec	Default: G.711 u-law, G.711 a-law, G.723, G.729. Provide using Codec items. The priority is according to the order of the screen. The default Setting of the first priority is G.729.
Move	>>: Select [Disable Codecs] item, press [>>] button, can move to Enable Codec area.

Item	Description
	<<: Select [Enable Codecs] item, press [<<] button, can move to Disable Codec area.
Up [Button]	Select [Enable Codec] item, press [Up] button, can move up the codec priority.
Down [Button]	Select [Enable Codec] item, press [Down] button, can move down the codec priority.
G.711 and G.729	Default: 20 ms; Set up the RTP Package Length of G.711 & G.729. Provide options: 10, 20, 30, 40, 50, 60, 70, 80, 90(ms).
G.723	Default: 30 ms; Set up the RTP Package Length of G.723. Provide options: 30, 60, 90(ms).
G.723 5.3K	Default: Disable; Set up the G.723 5.3K function open or close.
Silence Suppression (VAD)	Default: Disable; Set up Silence Suppression (VAD). Provide options: Disable, Enable. When VAD detects that the users are in talking, Codec will send out messages to network. Theoretically, there is only one user talking and another one is listening in the same time, the listening one don't send out any voice, so VAD will send the messages of the talking one to network, therefore, VAD can lower amount of message about 30%.
Echo Canceller	Default: Disable; Setup Echo Cancel. Provide options: Disable, Enable.
Codec Type	Setup the information of Codec ID.
G726-16	Default: 23; When you Setup Custom, you can modify the Codec ID Value. This column can only be numbers; maximum length is 3 bytes with the range of 95~127. Provide options: Default, Custom. *: Please select Custom, before you modify Codec ID Value.
G726-24	Default: 22; This column can only be numbers; maximum length is 3 bytes with the range of 95~127. Provide options: Default, Custom.
G726-32	Default: 2; This column can only be numbers; maximum length is 3 bytes with the range of 95~127. Provide options: Default, Custom.
G726-40	Default is 21; This column can only be numbers; maximum length is 3 bytes with the range of 95~127. Provide options: Default, Custom.
RFC 2833	Default is 101; This column can only be numbers; maximum length is 3 bytes with the range of 95~127. Provide options: Default, Custom.
Submit [Button]	Save the Settings.

9.2.3 Operate Instruction

Example 1: Adjust Codec Order

Step 1: In [Code Setting] web page, Setup G.726 - 16 enable, move mouse on [Disabled Codecs: G.726 -16], then press [>>] button, can move to [Enabled Codecs] area, move mouse on this Codec, can use [Up] or [Down] button, adjust codec negotiation priority during call was established (See Figure 1).

Codecs Setting

Disable Codecs		Enable Codecs
G.726 - 24 G.726 - 32 G.726 - 40	<input type="button" value=" >>"/> <input type="button" value=" <<"/>	G.726 - 16 G.711 u-law G.711 a-law G.723 G.729
<input type="button" value=" Move"/>		<input type="button" value=" Up"/> <input type="button" value=" Down"/>

(Figure 1)

Step 2: The other site makes a call to you, when you pick up the call, LP399 will use this codec as the first priority to talk.

Example 2: Change Codec ID

Step 1: In [Code Setting] web page, Setup [RFC2833, Custom: 100] (See Figure 2).

Codec Type	Default	ID Value	
G726-16:	Default	23	(95~127)
G726-24:	Default	22	(95~127)
G726-32:	Default	2	(95~127)
G726-40:	Default	21	(95~127)
RFC 2833:	Default	101	(95~127)
iLBC:	Custom	100	(95~127)

(Figure 2)

Note: If the other side sends you RFC-2833 not 100, LP399 will modify it to adjust the other side to communicate.

9.3 Advanced

9.3.1 Function

Provide [SIP Expire Time, SIP/RTP Port, QoS, Register SIP Expire Time, Use DNS SRV, DTMF, User=Phone, PRACK] Function.

9.3.2 Instruction

Figure 1: 1FXS(ATA171plus, ATA-171), 1FXS+1PSTN(ATA-171P), 1FXS+1FXO(ATA-171M).

SIP - Advanced Setting

SIP Expire Time:	<input type="text" value="60"/>	(60~86400 Seconds, 0=define by Server)
SIP Expire Time Type:	General <input type="button" value="v"/>	(General: Expire Time - [Expire Time/6])
SIP Registration Retry Timer:	<input type="text" value="64"/>	(5~250 Second)
SIP Session Timer T1:	<input type="text" value="1000"/>	(ms)
SIP Session Timer T2:	<input type="text" value="8000"/>	(ms)
SIP Session Timer B, F, H:	<input type="text" value="32000"/>	(ms)
SIP INVITE Timeout:	<input type="text" value="8000"/>	(ms)
Local SIP Port:	<input type="text" value="5060"/> ~ <input type="text" value="5060"/>	(1024~40000, Start ~ End)
Local RTP Port:	<input type="text" value="20000"/> ~ <input type="text" value="21999"/>	(1024~40000, Start ~ End)
Hold Type:	RFC 2543 (0.0.0.0) <input type="button" value="v"/>	
DTMF Type:	RFC 2833 <input type="button" value="v"/>	
RPort:	Enable <input type="button" value="v"/>	
Voice QoS (Diff-Serv):	<input type="text" value="40"/>	(0~63)
SIP QoS (Diff-Serv):	<input type="text" value="40"/>	(0~63)
Use DNS SRV:	Disable <input type="button" value="v"/>	
Keep-alive Message:	Disable <input type="button" value="v"/>	
Keep-alive Interval:	<input type="text" value="60"/>	(15~250 Second)
Jitter Buffer:	<input type="text" value="1"/> ~ <input type="text" value="64"/>	(1~96 Packet)
SIP Server Type:	General <input type="button" value="v"/>	
Use user=phone (Register):	Disable <input type="button" value="v"/>	
Use user=phone (Invite):	Disable <input type="button" value="v"/>	
Send SIP PRACK to Proxy:	Disable <input type="button" value="v"/>	
Only Accept Trusted Certificates:	Enable <input type="button" value="v"/>	
Set User Agent Content:	<input type="text"/>	

(Figure 1)

Item	Description
SIP Expire Time	<p>Default is 60; When this function is Set up to 0, the SIP Expire Time is according to the default value of Server.</p> <p>This column can only be numbers; maximum length is 5 bytes with the range of 60~86400 (seconds).</p>
SIP Expire Time Type	<p>Default: General.</p> <p>Provide options: General, 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10.</p> <p>*This function must be supported by SIP Server or IP-PBX Server.</p> <p>The count formula of SIP Expire Time: General: $SIP\ Expire\ Time - [(SIP\ Expire\ Time / 30) * 6]$ as $SIP\ Expire\ Time > 60$ seconds, if $SIP\ Expire\ Time < 60$ seconds, the SIP Expire Time subtract 5 seconds uniformly. 1/2: $SIP\ Expire\ Time * 1/2$. 2/3: $SIP\ Expire\ Time * 2/3$. 3/4: $SIP\ Expire\ Time * 3/4$. 4/5: $SIP\ Expire\ Time * 4/5$. 5/6: $SIP\ Expire\ Time * 5/6$. 6/7: $SIP\ Expire\ Time * 6/7$. 7/8: $SIP\ Expire\ Time * 7/8$. 8/9: $SIP\ Expire\ Time * 8/9$. 9/10: $SIP\ Expire\ Time * 9/10$.</p>
SIP Registration Retry Timer	<p>Default: 64 seconds.</p> <p>Set up the period of registering SIP Server again if LP399 fails to register SIP Server or IP-PBX server.</p> <p>This column can only enter numbers; maximum length is 4 bytes with the range of 5~3600 (seconds).</p>
SIP Session Timer T1	<p>Default: 1000 ms; Set up round-trip time (RTP) estimate.</p> <p>This column can only enter numbers; maximum length is 4 bytes with the range of 500~2000.</p> <p>*This function must be supported by Server.</p>
SIP Session Timer T2	<p>Default: 8000 ms.</p> <p>Set up the maximum retransmit interval for non-INVITE requests and INVITE responses.</p> <p>This column can only enter numbers; maximum length is 5 bytes with the range of 4000~16000.</p> <p>*This function must be supported by Server.</p>
SIP Session Timer B, F, H	<p>Default: 32000 ms.</p> <p>Set up the maximum retransmit interval for non-INVITE requests and INVITE responses.</p> <p>This column can only enter numbers; maximum length is 6 bytes with the range of 8000~128000.</p> <p>B: $64 * SIP\ T1$; INVITE transaction timeout timer ◦ F: $64 * SIP\ T1$; non-INVITE transaction timeout timer ◦ H: $64 * SIP\ T1$, Wait time for ACK receipt.</p>

Item	Description
	<p>*This function must be supported by Server.</p> <p>For example, if T1 is 500 ms, T2 is 4 seconds and B,F,H is 32 seconds, then non-INVITE retransmissions occur at intervals of 500 ms, 1s, 2s, 4s, 4s, 4s, 4s, 4s, 4s. This means that retransmissions occur with an exponentially increasing interval that caps at T2. In this particular scenario, there are 10 retransmissions which is a total of 11 requests from UAC.</p>
SIP INVITE Timeout	<p>Default: 30000 ms; Set up SIP Invite, If how long do not respond, enter the failed state. This column can only enter numbers; maximum length is 5 bytes with the range of 8000~64000.</p>
Local SIP Port	<p>Default: 5060~5060.</p> <p>Set up the Start and End SIP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000. °</p> <p>If you want to Set up a fixed port, please Set up the same value of Start and End Port.</p> <p>If you want to Set up a range, the left column is Start Port, the right Port is End Port.</p>
Local RTP Port	<p>Default: 20000~21999.</p> <p>Set up the Start and End RTP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000.</p> <p>If you want to Set up a fixed port, please Set up the same value of Start and End Port.</p> <p>If you want to Set up a range, the left column is Start Port, the right Port is End Port.</p>
Hold Type	<p>Default: RFC 2543 (0.0.0.0).</p> <p>Set up Hold (define by RFC).</p> <p>When this function is on, the information of [Connection Information (c): IN IP4 xxx.xxx.xxx.xxx] will change IP to the device of executing the function.</p> <p>Provide options: RFC2543 (0.0.0.0), Type1 (Send only), Type2 (inactvie).</p>
DTMF Type	<p>Default: RFC 2833.</p> <p>InBand: When you enter key information, the [Ethereal] will not show it.</p> <p>RFC2833: When you enter key information, the [Ethereal] will show [RTP Event].</p> <p>SIP Info: When you enter key information, the [Ethereal] will show [Request: Info].</p> <p>Provide options: InBand, RFC2833, SIP Info.</p> <p>RFC2833 + Inband: When you enter key information, LP399 sends Inband message and [RTP Event] message.</p> <p>SIP Info + Inband: When you enter key information, LP399</p>

Item	Description
	sends Inband message and [Request: Info] message.
RPort	Default: Disable; Set up RPort function. When this function is on, the [Rport] message will add in [Message Header]. Provide options: Disable, Enable. *This function must be supported by Server.
Voice QoS (Diff-Serv)	Default: 40; This column can only enter numbers; maximum length is 2 bytes with the range of 0~63.
SIP QoS (Diff-Serv)	Default: 40; This column can only enter numbers; maximum length is 2 bytes with the range of 0~63.
Use DNS SRV	Default: Disable. When this function is on, the package will show [DNS, Standard query SRV_sip_upd.xxx.xxx.xxx]. Provide options: Disable, Enable. *This function must be supported by Server.
Keep-alive Message	Default: Disable; When this function is on and system is in NAT, LP399 will send a package to Server periodically according to [Send Keep Alives Packet]. Provide options: Disable, Send UDP, Send SIP Option. Send UDP: Use UDP format to send; For example: UDP, Source Port: sip Destination Port: xxxx. Send SIP Option: Use SIP Option format to send; For example: SIP, Request-Line: OPTIONS sip:xxx.xxx.xxx.xxx;user=phone SIP/2.0.
Keep-alive Interval	Default: 60; This column can only enter numbers; maximum length is 3 bytes with the range of 15~250.
Jitter Buffer	Default: 1~64; Set up Jitter Buffer. In VoIP system, the time of every voice package arrives destination will affect by Network Delay. Therefore, Jitter Buffer is used in destination to modify the order of packages and adjust the time of Voice Playout Delay, this function will raise the voice quality. This column can only enter numbers; maximum length is 3 bytes with the range 0~32.
SIP Server Type	Default: General. Set up the type of SIP Server. In accordance with market available different SIP Servers or IP-PBX server, WellGate M1 will adjust its configuration to be compatible with these SIP Server. Provide options: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink. * Please make sure which model of SIP Server or IP-PBX server for Wellgate M1 to work with in order to select suitable SIP Server Type.
Use user=phone (Register)	Default: Disable; When this function is on, the Register Header will add "user=phone" message in Register packages.

Item	Description
	Provide options: Disable, Enable. *This function must be supported from SIP Server.
Use user=phone (Invite)	Default: Disable; When this function is on, the Invite Header will add "user=phone" message in Invite packages. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Send SIP PRACK of Proxy	Default: Disable; When this function is on, there will add "PRACK Header" messages. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Only Accept Trusted Certificates	Default: Enable. Set up IP incoming call if it comes from trusted SIP Server or IP-PBX or not. When Enable , this device only accept IP incoming call from trusted SIP Server or IP-PBX. It will reject P2P call. When Disable , this device accept P2P call. And accept any IP incoming call even from not trusted SIP Server or IP-PBX. Provide drop-down options: Disable, Enable.
Set up User Agent Content	Default: (null); When send SIP packets, the packets header "User-Agent" message will join this word. This column can enter numbers and strings (support: 0~9, a~z, @, _, -, *, #, ., +, :, () [,] and blank); maximum length is 46 bytes.
Submit [Button]	Save the Settings.

Figure 2: 2FXS (ATA172plus and ATA-172).

SIP - Advanced Setting

SIP Expire Time:	<input type="text" value="60"/>	(60~86400 Seconds, 0=define by Server)
SIP Expire Time Type:	<input type="text" value="General"/>	(General: Expire Time - [Expire Time/6])
SIP Registration Retry Timer:	<input type="text" value="64"/>	(5~250 Second)
SIP Session Timer T1:	<input type="text" value="1000"/>	(ms)
SIP Session Timer T2:	<input type="text" value="8000"/>	(ms)
SIP Session Timer B, F, H:	<input type="text" value="32000"/>	(ms)
SIP INVITE Timeout:	<input type="text" value="8000"/>	(ms)
Local SIP Port of Phone 1:	<input type="text" value="5060"/> ~ <input type="text" value="5060"/>	(1024~40000, Start ~ End)
Local RTP Port of Phone 1:	<input type="text" value="20000"/> ~ <input type="text" value="21999"/>	(1024~40000, Start ~ End)
Local SIP Port of Phone 2:	<input type="text" value="5062"/> ~ <input type="text" value="5062"/>	(1024~40000, Start ~ End)
Local RTP Port of Phone 2:	<input type="text" value="22000"/> ~ <input type="text" value="23999"/>	(1024~40000, Start ~ End)
Hold Type:	<input type="text" value="RFC 2543 (0.0.0.0)"/>	
DTMF Type:	<input type="text" value="RFC 2833"/>	
RPort:	<input type="text" value="Enable"/>	
Voice QoS (Diff-Serv):	<input type="text" value="40"/>	(0~63)
SIP QoS (Diff-Serv):	<input type="text" value="40"/>	(0~63)
Use DNS SRV:	<input type="text" value="Disable"/>	
Keep-alive Message:	<input type="text" value="Disable"/>	
Keep-alive Interval:	<input type="text" value="60"/>	(15~250 Second)
Jitter Buffer:	<input type="text" value="1"/> ~ <input type="text" value="64"/>	(1~96 Packet)
SIP Server Type:	<input type="text" value="General"/>	
Use user=phone (Register):	<input type="text" value="Disable"/>	
Use user=phone (Invite):	<input type="text" value="Disable"/>	
Send SIP PRACK to Proxy:	<input type="text" value="Disable"/>	
Only Accept Trusted Certificates:	<input type="text" value="Disable"/>	
Set User Agent Content:	<input type="text"/>	

(Figure 2)

Item	Description
SIP Expire Time	Default is 60; When this function is Set up to 0, the SIP Expire Time is according to the default value of Server. This column can only be numbers; maximum length is 5 bytes with the range of 60~86400 (seconds).
SIP Expire Time Type	Default: General. Provide options: General, 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10. *This function must be supported by SIP Server or IP-PBX Server.

Item	Description
	<p>The count formula of SIP Expire Time: General: SIP Expire Time-[(SIP Expire Time/30)*6] as SIP Expire Time > 60 seconds, if SIP Expire Time < 60 seconds, the SIP Expire Time subtract 5 seconds uniformly. 1/2: SIP Expire Time * 1/2. 2/3: SIP Expire Time * 2/3. 3/4: SIP Expire Time * 3/4. 4/5: SIP Expire Time * 4/5. 5/6: SIP Expire Time * 5/6. 6/7: SIP Expire Time * 6/7. 7/8: SIP Expire Time * 7/8. 8/9: SIP Expire Time * 8/9. 9/10: SIP Expire Time * 9/10.</p>
SIP Registration Retry Timer	<p>Default: 64 seconds. Set up the period of registering SIP Server again if LP399 fails to register SIP Server or IP-PBX server. This column can only enter numbers; maximum length is 4 bytes with the range of 5~3600 (seconds).</p>
SIP Session Timer T1	<p>Default: 1000 ms; Set up round-trip time (RTP) estimate. This column can only enter numbers; maximum length is 4 bytes with the range of 500~2000. *This function must be supported by Server.</p>
SIP Session Timer T2	<p>Default: 8000 ms. Set up the maximum retransmit interval for non-INVITE requests and INVITE responses. This column can only enter numbers; maximum length is 5 bytes with the range of 4000~16000. *This function must be supported by Server.</p>
SIP Session Timer B, F, H	<p>Default: 32000 ms. Set up the maximum retransmit interval for non-INVITE requests and INVITE responses. This column can only enter numbers; maximum length is 6 bytes with the range of 8000~128000. B: 64 * SIP T1; INVITE transaction timeout timer ◦ F: 64 * SIP T1; non-INVITE transaction timeout timer ◦ H: 64 * SIP T1, Wait time for ACK receipt. *This function must be supported by Server. For example, if T1 is 500 ms, T2 is 4 seconds and B,F,H is 32 seconds, then non-INVITE retransmissions occur at intervals of 500 ms, 1s, 2s, 4s, 4s, 4s, 4s, 4s, 4s. This means that retransmissions occur with an exponentially increasing interval that caps at T2. In this particular scenario, there are 10 retransmissions which is a total of 11 requests from UAC.</p>
SIP INVITE Timeout	<p>Default: 30000 ms; Set up SIP Invite, If how long do not respond, enter the failed state. This column can only enter</p>

Item	Description
	numbers; maximum length is 5 bytes with the range of 8000~64000.
Local SIP Port of phone 1	Default: 5060~5060. Set up the Start and End SIP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000. ◦ If you want to Set up a fixed port, please Set up the same value of Start and End Port. If you want to Set up a range, the left column is Start Port, the right Port is End Port.
Local RTP Port of phone 1	Default: 20000~21999. Set up the Start and End RTP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000. If you want to Set up a fixed port, please Set up the same value of Start and End Port. If you want to Set up a range, the left column is Start Port, the right Port is End Port.
Local SIP Port of phone 2	Default: 5062~5062. Set up the Start and End SIP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000. ◦ If you want to Set up a fixed port, please Set up the same value of Start and End Port. If you want to Set up a range, the left column is Start Port, the right Port is End Port.
Local RTP Port of phone 2	Default: 22000~23999. Set up the Start and End RTP Port Range of phone 1. This column can only enter numbers; maximum length is 5 bytes with the range of 1024~40000. If you want to Set up a fixed port, please Set up the same value of Start and End Port. If you want to Set up a range, the left column is Start Port, the right Port is End Port.
Hold Type	Default: RFC 2543 (0.0.0.0). Set up Hold (define by RFC). When this function is on, the information of [Connection Information (c): IN IP4 xxx.xxx.xxx.xxx] will change IP to the device of executing the function. Provide options: RFC2543 (0.0.0.0), Type1 (Send only), Type2 (inactvie).
DTMF Type	Default: RFC 2833. InBand: When you enter key information, the [Ethereal] will not show it. RFC2833: When you enter key information, the [Ethereal] will

Item	Description
	<p>show [RTP Event].</p> <p>SIP Info: When you enter key information, the [Ethereal] will show [Request: Info].</p> <p>Provide options: InBand, RFC2833, SIP Info.</p> <p>RFC2833 + Inband: When you enter key information, LP399 sends Inband message and [RTP Event] message.</p> <p>SIP Info + Inband: When you enter key information, LP399 sends Inband message and [Request: Info] message.</p>
RPort	<p>Default: Disable; Set up RPort function. When this function is on, the [Rport] message will add in [Message Header].</p> <p>Provide options: Disable, Enable.</p> <p>*This function must be supported by Server.</p>
Voice QoS (Diff-Serv)	<p>Default: 40; This column can only enter numbers; maximum length is 2 bytes with the range of 0~63.</p>
SIP QoS (Diff-Serv)	<p>Default: 40; This column can only enter numbers; maximum length is 2 bytes with the range of 0~63.</p>
Use DNS SRV	<p>Default: Disable.</p> <p>When this function is on, the package will show [DNS, Standard query SRV_sip_upd.xxx.xxx.xxx].</p> <p>Provide options: Disable, Enable.</p> <p>*This function must be supported by Server.</p>
Keep-alive Message	<p>Default: Disable; When this function is on and system is in NAT, LP399 will send a package to Server periodically according to [Send Keep Alives Packet].</p> <p>Provide options: Disable, Send UDP, Send SIP Option.</p> <p>Send UDP: Use UDP format to send; For example: UDP, Source Port: sip Destination Port: xxxx.</p> <p>Send SIP Option: Use SIP Option format to send; For example: SIP, Request-Line: OPTIONS sip:xxx.xxx.xxx.xxx;user=phone SIP/2.0.</p>
Keep-alive Interval	<p>Default: 60; This column can only enter numbers; maximum length is 3 bytes with the range of 15~250.</p>
Jitter Buffer	<p>Default: 1~64; Set up Jitter Buffer.</p> <p>In VoIP system, the time of every voice package arrives destination will affect by Network Delay. Therefore, Jitter Buffer is used in destination to modify the order of packages and adjust the time of Voice Playout Delay, this function will raise the voice quality.</p> <p>This column can only enter numbers; maximum length is 3 bytes with the range 0~32.</p>
SIP Server Type	<p>Default: General.</p> <p>Set up the type of SIP Server.</p> <p>In accordance with market available different SIP Servers or IP-PBX server, WellGate M1 will adjust its configuration to be compatible with these SIP Server.</p>

Item	Description
	Provide options: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink. * Please make sure which model of SIP Server or IP-PBX server for Wellgate M1 to work with in order to select suitable SIP Server Type.
Use user=phone (Register)	Default: Disable; When this function is on, the Register Header will add "user=phone" message in Register packages. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Use user=phone (Invite)	Default: Disable; When this function is on, the Invite Header will add "user=phone" message in Invite packages. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Send SIP PRACK of Proxy	Default: Disable; When this function is on, there will add "PRACK Header" messages. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Only Accept Trusted Certificates	Default: Enable. Set up IP incoming call if it comes from trusted SIP Server or IP-PBX or not. When Enable, this device only accept IP incoming call from trusted SIP Server or IP-PBX. It will reject P2P call. When Disable, this device accept P2P call. And accept any IP incoming call even from not trusted SIP Server or IP-PBX. Provide drop-down options: Disable, Enable.
Set up User Agent Content	Default: (null); When send SIP packets, the packets header "User-Agent" message will join this word. This column can enter numbers and strings (support: 0~9, a~z, @, _, -, *, #, ., +, :, () [,] and blank); maximum length is 46 bytes.
Submit [Button]	Save the Settings.

9.3.3 Operate Instuction

Example 1: SIP Expire Time

◆ SIP Expire Time: 60

Step 1: In [Service Domain Setting] web page, Set up [Realm Active: Enable, Display Name: 22061, Phoner Number: 22061, Authentication ID: 22061, Authentication Password: test, Domain Server: 61.62.236.71:6000, Proxy Server: 61.62.236.71:6000, Subscribe for MWI: Disable] (See Figure 1).

Realm:

Realm Active:	<input type="text" value="Enable"/>
Display Name:	<input type="text" value="22061"/>
Phone Number:	<input type="text" value="22061"/>
Authentication ID:	<input type="text" value="22061"/>
Authentication Password:	<input type="text" value="••••"/>
Domain Server:	<input type="text" value="61.62.236.71:6000"/>
Proxy Server:	<input type="text" value="61.62.236.71:6000"/>
Subscribe for MWI :	<input type="text" value="Disable"/>

(Figure 1)

Step 2: In [SIP – Advanced Setting] web page, Set up [SIP Expire Time: 60] (See Figure 2).

SIP Expire Time: (60~86400 Seconds, 0=define by Server)

(Figure 2)

Step 3: When registering to Server successfully, LP399 will send a register package every 55 seconds.

◆ **SIP Expire Time: 0 (by server)**

Step 1: In [Service Domain Setting] web page, Set up [Realm Active: Enable, Display Name: 22061, Phoner Number: 22061, Authentication ID: 22061, Authentication Password: test, Domain Server: 61.62.236.71:6000, Proxy Server: 61.62.236.71:6000, Subscribe for MWI: Disable] (See Figure 1).

Step 2: In [SIP – Advanced Setting] web page, Set up [SIP Expire Time: 0 (the register time is defined by Server)] (See Figure 3).

SIP Expire Time: (60~86400 Seconds, 0=define by Server)

(Figure 3)

Step 3: System will register to Server according to the period that defined by Server.

Example 2: Use DNS SRV

◆ **Use DNS SRV: Enable**

Step 1: Set up register SIP account Settings first, then use [Domain] to register SIP Server. (See Figure 4).

Realm:

Realm Active:	<input type="text" value="Enable"/>
Display Name:	<input type="text" value="8061"/>
Phone Number:	<input type="text" value="8061"/>
Authentication ID:	<input type="text" value="8061"/>
Authentication Password:	<input type="text" value="••••"/>
Domain Server:	<input type="text" value="voiptalk.org"/>
Proxy Server:	<input type="text" value="nat.voiptalk.org"/>
Subscribe for MWI :	<input type="text" value="Disable"/>

(Figure 4)

Step 2: In [SIP – Advanced Setting] web page, Set up [Use DNS SRV: Enable] (See Figure 5)

Use DNS SRV:	Enable
Keep-alive Message:	Disable

(Figure 5)

Example 3: Keep Alives Message

◆ Keep Alives Message: Send UDP

Step 1: Set up register SIP account settings (See Figure 6).

Realm:	1
Realm Active:	Enable
Display Name:	22061
Phone Number:	22061
Authentication ID:	22061
Authentication Password:	•••••
Domain Server:	61.62.236.71:6000
Proxy Server:	61.62.236.71:6000
Subscribe for MWI :	Disable
<input type="button" value="Submit"/>	

(Figure 6)

Step 2: In [SIP – Advanced Setting] web page, Set up [Keep Alives Message: Send UDP, Keep Alives Interval: 150] (See Figure 7).

Keep-alive Message:	Send UDP
Keep-alive Interval:	150 (15~250 Second)

(Figure 7)

◆ Keep Alives Message: Send SIP Info

Step 1: Set up register SIP account Settings (See Figure 6).

Step 2: In [SIP – Advanced Setting] web page, Set up [Keep Alives Message: Send SIP Option, Keep Alives Interval: 150] (See Figure 8).

Keep-alive Message:	Send SIP OPTION
Keep-alive Interval:	150 (15~250 Second)

(Figure 8)

9.4 STUN (STUN & Froce configuration)

9.4.1 Function

STUN Provides function to set up STUN and Force feature.

9.4.2 Instruction

STUN Setting

STUN Active:	<input type="text" value="Disable"/>
STUN Server Name:	<input type="text" value="stun.xten.com"/>
STUN Port:	<input type="text" value="3478"/> (80~65535)
Force Active:	<input type="text" value="Disable"/>
Public IP Address:	<input type="text"/>
Public Port:	<input type="text" value="5060"/> (80~65535)

Item	Description
STUN Active	Default: Disable; When this function is on, STUN functions Enable. Provide options: Disable, Enable.
STUN Server Name	Default: stun.xten.com; This column can enter IP or Domain Name with the format of xxx.xxx.xxx.xxx; maximum length is 63 bytes.
STUN Port	Default: 3478; This column can only enter numbers; maximum length is 5 bytes with the range 80~65535.
Force Active	Default: Disable; When this function is on, the IP of [SIP infor] in [Ethereal] will replace by the assigned IP address. Provide options: Disable, Enable.
Public IP Address	Set up Router's public IP address; This column can only enter IP with the format of xxx.xxx.xxx.xxx; maximum length is 63 bytes.
Public Port	Default: 5060; Set up Router's public Port. This column can only enter numbers; maximum length is 5 bytes with the range of 80~65535.
Submit [Button]	Save the settings.

9.4.3 Operate Instruction

Example 1: STUN

Step 1: Please Set up the SIP account first.

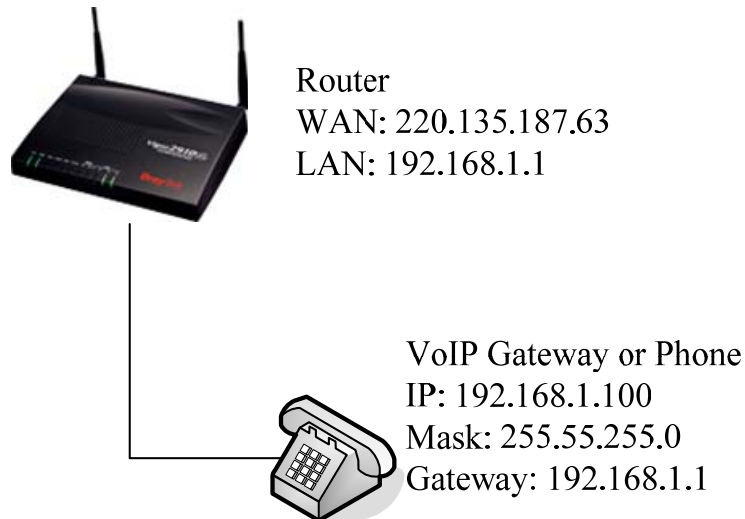
Step 2: In [STUN Setting] web page, Set up [STUN Active: Enable, STUN Server Name: stun.xten.com, SUTN Port: 3478] (See Figure 1).

STUN Active:	<input type="text" value="Enable"/>
STUN Server Name:	<input type="text" value="stun.xten.com"/>
STUN Port:	<input type="text" value="3478"/> (80~65535)

(Figure 1)

Example 2: Force

The structure figure (See Figure 2):



(Figure 2)

Step 1: Please set up the SIP account first.

Step 2: In [STUN Setting] web page, set up [Force Active: Enable, Public IP Address: 118.169.209.251, Public Port: 7777] (See Figure 3).

STUN Active:	<input type="text" value="Enable"/>
STUN Server Name:	<input type="text" value="118.169.209.251"/>
STUN Port:	<input type="text" value="7777"/> (80~65535)

(Figure 3)

10. Management (Advanced configuration)

Provide [[Status Log](#), [Auto Config](#), [Auto Update](#), [New Firmware](#), [Advanced](#), [Passowrd](#), [Tones](#), [Default](#), [Language](#)] functions.

10.1 Status Log

10.1.1 Function

Status Log Provide the running status of the system.

10.1.2 Instruction

Staus Log

Phone Status IDLE

System Log

Page:

Index	Message
0	<2014-02-10 11:49>Get Time from SNTP server, Succeed!
1	<2005-01-01 08:00>Get SNTP server IP=216.66.0.142
2	<2005-01-01 00:00>DHCP Got Ip=192.168.23.21
3	<2005-01-01 00:00>DHCP state 1=2
4	<2005-01-01 00:00>DHCP_SendRequest()
5	<2005-01-01 00:00>Rx OFFER from 192.168.1.17
6	<2005-01-01 00:00>DHCP_SendDiscover()
7	<2005-01-01 00:00>Enable DHCP_SERVER
8	<2005-01-01 00:00>Init Lan Interface!
9	<2005-01-01 00:00>Iface type : DHCP_CLIENT
10	<2005-01-01 00:00>Init Wan Interface!
11	<2005-01-01 00:00>Application starting ...
12	
13	
14	
15	
16	
17	
18	
19	
20	
21	
22	
23	
24	

Item	Description
Phone Status	Show the Phone user status now; The status has: IDLE, Off-Hook. IDLE: Ready Off Hook: in use.
Refresh [Button]	Refresh the Phone status.
Status Log	System work status message.
Page	Default: 1 (Page 1), Select page. Provide drop-down options:1~xx; The page increases to next one in accordance with the amount of data to increase automatically.
Index	Show index number.
Message	Show the information of the system. Example: <2014-02-10 11:49> Get Time from SNTP server, Succeed! <2014-02-10 11:49>: Show the time of message. Get Time from SNTP server, Succeed! : Shows the content of message.
Export System Log [Button]	Save [Stauts Log] data to log file; System default file name is Syslog.log.

10.1.3 Operate Instruction

Example 1: Check Phone Status

Step 1: In [Status Log] web page, When the phone don't in use, [Phone Status] will show [IDLE] (See Figure 1).

Phone Status IDLE

(Figure 1)

Step 2: When pick up the handset, In [Status Log] web page, Press [Refresh] button, The [Phone Status] show [Off Hook] (See Figure 2).

Phone Status Off Hook

(Figure 2)

Example 2: Check System Log

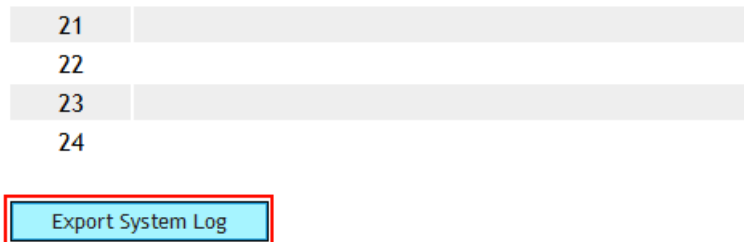
Step 1: In [Status Log] web page, Select web page, the page will show other pages data. (See Figure 3).

System Log	
Page:	1
Index	Message
0	<2014-02-10 11:49>Get Time from SNTP server, Succeed!
1	<2005-01-01 08:00>Get SNTP server IP=216.66.0.142
2	<2005-01-01 00:00>DHCP Got Ip=192.168.23.21
3	<2005-01-01 00:00>DHCP state 1=2
4	<2005-01-01 00:00>DHCP_SendRequest()
5	<2005-01-01 00:00>Rx OFFER from 192.168.1.17
6	<2005-01-01 00:00>DHCP_SendDiscover()
7	<2005-01-01 00:00>Enable DHCP_SERVER
8	<2005-01-01 00:00>Init Lan Interface!
9	<2005-01-01 00:00>Iface type : DHCP_CLIENT
10	<2005-01-01 00:00>Init Wan Interface!
11	<2005-01-01 00:00>Application starting ...
12	

(Figure 3)

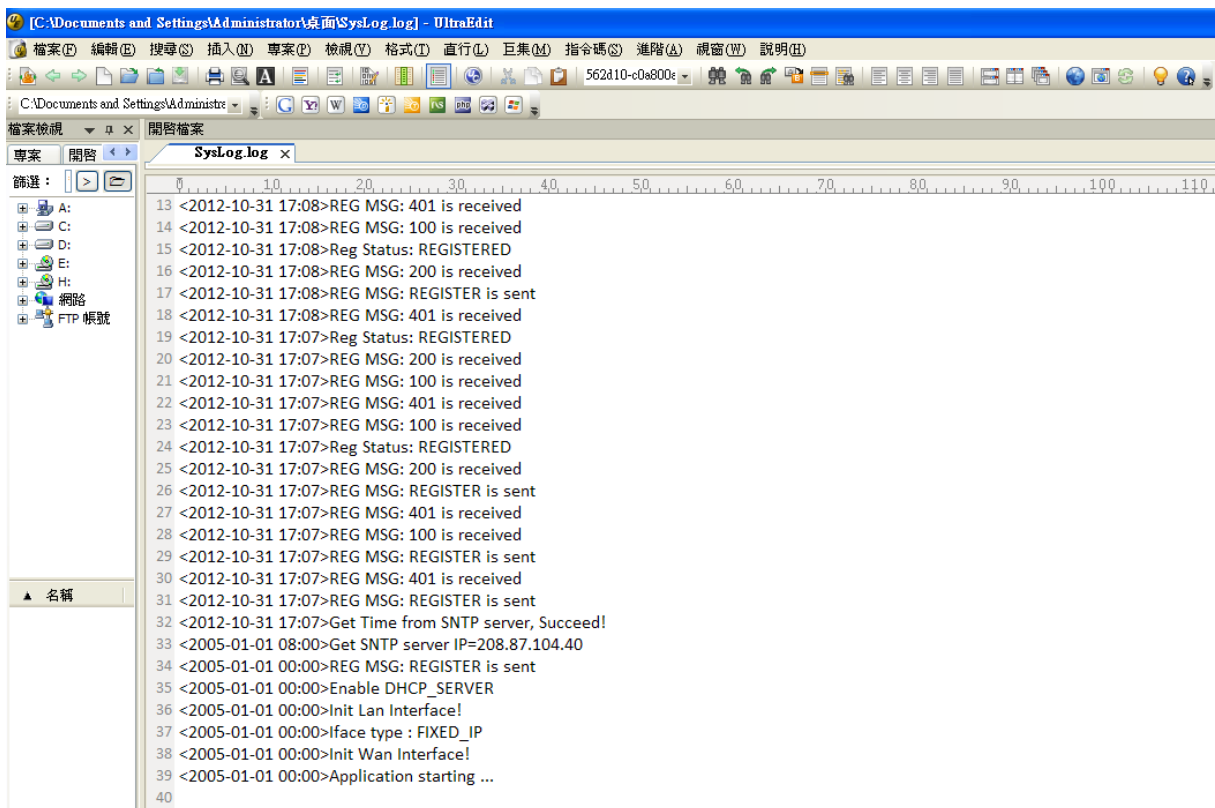
Example 2: Save System Log File

Step 1: In [Status Log] web page, press [Export System Log] button, Enter [Save File] menu, Then press [Save] button (See Figure 4).



(Figure 4)

Step 2: Save file completely, Any text editor can open [SysLog.log] file (See Figure 5).



(Figure 5)

10.2 Auto Configuration

10.2.1 Function

Provide 3 kind of provision methods. (TFTP, FTP and HTTP)

10.2.2 Instruction

Auto Provision Setting

Provision Active:	<input type="text" value="Disable"/>	
2 Steps Configuration:	<input type="text" value="Disable"/>	
Server Auto Discovery:	<input type="text" value="Disable"/>	
Scheduling:	<input type="text" value="Disable"/>	
TFTP Server:	<input type="text"/>	
TFTP File Path:	<input type="text"/>	Exp. file/load/
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/
FTP Server:	<input type="text"/>	Exp. 60.35.187.30
FTP User Name:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. file/load/
Next Configuration time:	<input type="text"/>	
<input type="button" value="Submit"/>		

Item	Decription
Provision Active	Default: Disable; When this function is on, LP399 will download the MACID.dat from the designated Server. Provide options: Disable, TFTP, FTP and HTTP.
2 Steps Configuration	Default: Disable; Set up 2 Steps configuration, get the common settings first, then get the SIP accounts and passwords secondly. Provide options: Disable, Enable. *This function must be supported from SIP Server.
Server Auto Discovery	Default: Disable; DHCP TFTP Option 66 (TFTP): DHCP Server will offer the Option 66 Server IP address into the column of system [TFTP Server] when it assigns IP. Broadcasting: Discovering the Server by broadcasting, the Server will send the information of Type and Server to LP399, it will fill in these information to the corresponding columns. Provide options: Disable, DHCP TFTP Option 66 (TFTP),

Item	Description
	Broadcasting. *This function must be supported by Server.
Scheduling	Default: Disable; Execute the Configuration regularly. When this function is on, system will check the Configuration Server by [Next Config Time]. Provide options: Disable, Enable. Note: The time parameter use [Auto Update Setting] web page's [Scheduling Time & Date].
TFTP Server	This column can only enter IP with the format of xxx.xxx.xxx.xxx; maximum length is 15 bytes.
TFTP File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, ex: 123/.
HTTP Server	This column can enter IP or Domain Name; maximum length is 63 bytes.
HTTP File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, ex: 123/.
FTP Server	This column can enter IP or Domain Name; maximum length is 63 bytes.
FTP User Name	This column can enter IP or Domain Name; maximum length is 63 bytes.
FTP Password	This column can enter IP or Domain Name; maximum length is 63 bytes.
File File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, ex: 123/.
Next Configuration Time	System will check the Configuration Server when the Next config time is up. The start counting date is the next day, so the Next configuration time will add one day. Count rule : the next day + days + time period + MACaddress + random number = Next config time.
Submit [Button]	Save the settings.

10.2.3 Operate Instruction

Example 1: Configuration by HTTP

Step 1: Please complete MACID.dat first, and place it at the path of the designated Server.

Step 2: In [Auto Provision Setting] web page, Set up [Provision Active: HTTP, HTTP Server: 192.168.50.2, HTTP Path: /download/] (See Figure 1).

Provision Active:	<input type="text" value="HTTP"/>	Exp.	
2 Steps Configuration:	<input type="text" value="Disable"/>	Exp.	
Server Auto Discovery:	<input type="text" value="Disable"/>	Exp.	
Scheduling:	<input type="text" value="Disable"/>	Exp.	
HTTP Server:	<input type="text" value="192.168.50.2"/>	Exp.	60.35.187.30
HTTP File Path:	<input type="text" value="/download/"/>	Exp.	/download/

(Figure 1)

Step 3: In [Service Domain Setting] web page, Check [Realm: 1] Settings, Use [MACID.dat] file to load configuration (See Figure 2).

Realm: 1 ▾

Realm Active:	Enable ▾
Display Name:	22061
Phone Number:	22061
Authentication ID:	22061
Authentication Password:	••••
Domain Server:	61.62.236.71:6000
Proxy Server:	61.62.236.71:6000
Subscribe for MWI :	Disable ▾

Submit

(Figure 2)

Example 2: Configuration by FTP

Step 1: Please complete MACID.dat first, and place it at the path of the designated Server.

Step 2: In [Auto Provision Setting] web page, Set up [Provision Active: FTP, FTP Server: 192.168.50.2, FTP User Name: test, FTP Password: test, FTP File Path: download/] (See Figure 3).

Provision Active:	FTP ▾
2 Steps Configuration:	Disable ▾
Server Auto Discovery:	Disable ▾
Scheduling:	Disable ▾

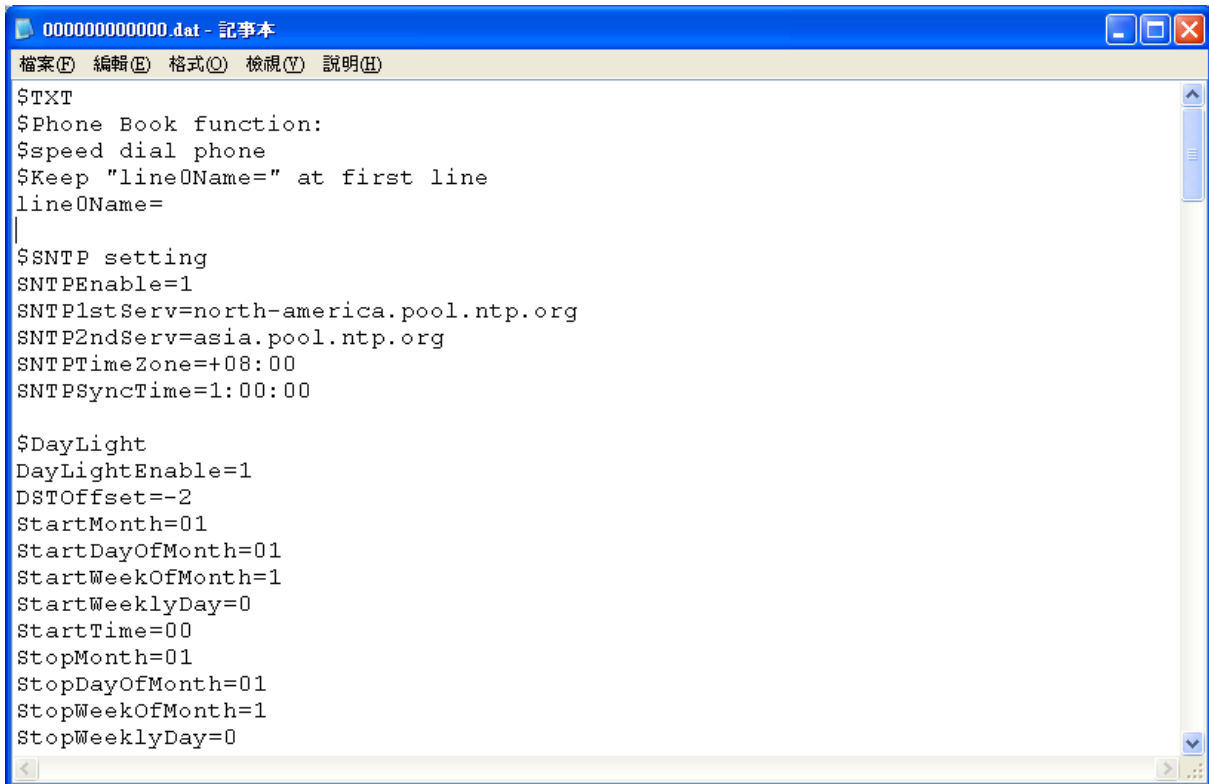
FTP Server:	192.168.50.2	Exp. 60.35.187.30
FTP User Name:	test	
FTP Password:	••••	
FTP File Path:	download/	Exp. file/load/

(Figure 3)

Step 3: In [Service Domain Setting] web page, Check [Realm: 1] Settings, Use [MACID.dat] file to load configuration.

Example 3 : Set up 2 Steps configuration (not encryption)

Step 1: Build a common MAC file with name [000000000000.dat], the file does not contain [\$Service Domain Setting] data, (See Figure 4).



```

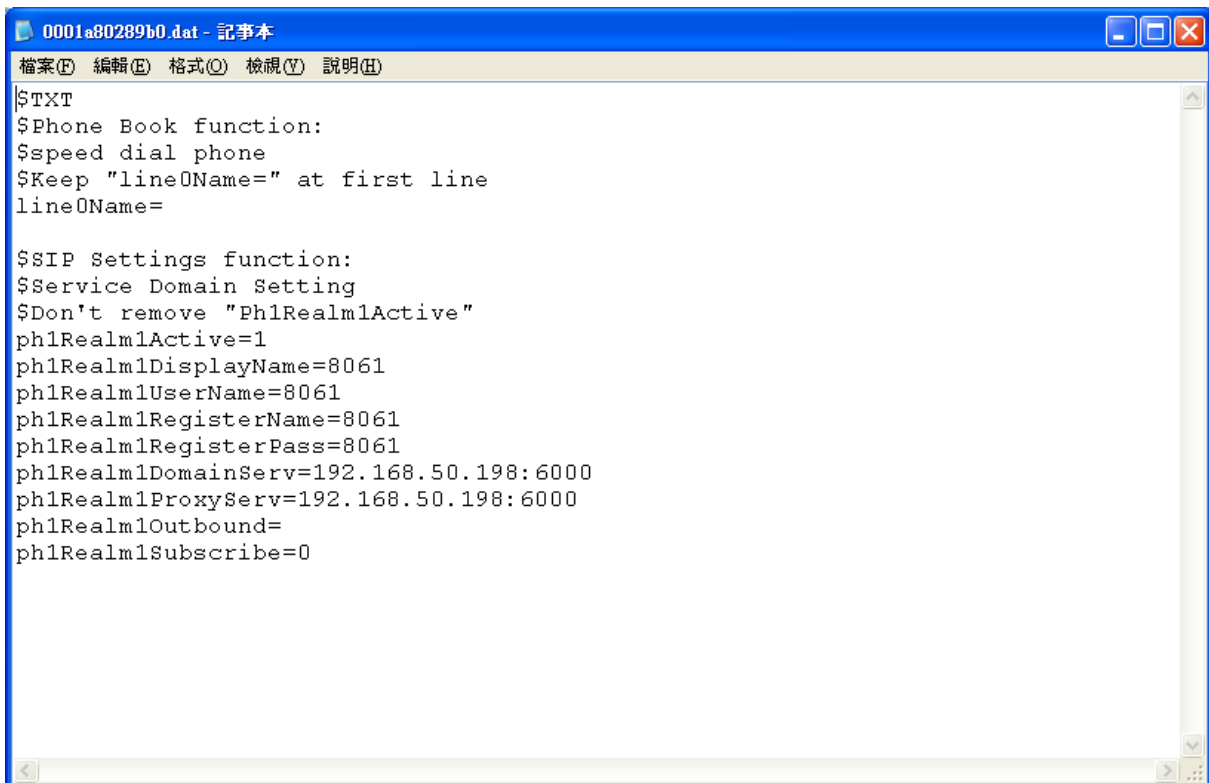
000000000000.dat - 記事本
檔案(F) 編輯(E) 格式(O) 檢視(V) 說明(H)

$TXT
$Phone Book function:
$speed dial phone
$Keep "line0Name=" at first line
line0Name=
|
$SNTP setting
SNTPEnable=1
SNTP1stServ=north-america.pool.ntp.org
SNTP2ndServ=asia.pool.ntp.org
SNTPTimeZone=+08:00
SNTPSyncTime=1:00:00

$DayLight
DayLightEnable=1
DSTOffset=-2
StartMonth=01
StartDayOfMonth=01
StartWeekOfMonth=1
StartWeeklyDay=0
StartTime=00
StopMonth=01
StopDayOfMonth=01
StopWeekOfMonth=1
StopWeeklyDay=0
    
```

(Figure 4)

Step 2. Set up the [\$Service Domain Setting] data, and [line0Name=, ph1Realm1Active=1] cannot be deleted (See Figure 5).



```

0001a80289b0.dat - 記事本
檔案(F) 編輯(E) 格式(O) 檢視(V) 說明(H)

$TXT
$Phone Book function:
$speed dial phone
$Keep "line0Name=" at first line
line0Name=

$SIP Settings function:
$Service Domain Setting
$Don't remove "Ph1Realm1Active"
ph1Realm1Active=1
ph1Realm1DisplayName=8061
ph1Realm1UserName=8061
ph1Realm1RegisterName=8061
ph1Realm1RegisterPass=8061
ph1Realm1DomainServ=192.168.50.198:6000
ph1Realm1ProxyServ=192.168.50.198:6000
ph1Realm1Outbound=
ph1Realm1Subscribe=0
    
```

(Figure 5)

Step 3: On [Auto Configuration Setting], Set up [Type: TFTP, 2 Steps Configuration: Enable, TFTP Server: 192.168.50.99] (See Figure 6).

Provision Active:	TFTP
2 Steps Configuration:	Enable
Server Auto Discovery:	Disable
Scheduling:	Disable
TFTP Server:	192.168.50.99
TFTP File Path:	<input type="text"/> Exp. file/load/

(Figure 6)

Step 4: In [Service Domain Setting] web page, Check [Realm: 1] Settings, Use [MACID.dat] file to load configuration (See Figure 7).

Realm:

Realm Active:	Enable
Display Name:	8061
Phone Number:	8061
Authentication ID:	8061
Authentication Password:	••••
Domain Server:	voiptalk.org
Proxy Server:	nat.voiptalk.org
Subscribe for MWI :	Disable

(Figure 7)

Example 4 : Server Auto Discover

◆ Broadcasting

Step 1: In [Auto Provision Setting] web page, Set up [Provision Active: TFTP, Server Auto Discover: Broadcasting] (See Figure 8).

Provision Active:	TFTP
2 Steps Configuration:	Disable
Server Auto Discovery:	Broadcasting
Scheduling:	Disable

(Figure 8)

Step 2: In [Auto Provision Setting] web page, Check [Provision Active: TFTP, TFTP Server: 192.168.55.91, TFTP File Path: config/] (See Figure 9).

Note: If starts TFTP Server at the same time, this field parameter can't see data. Because of this data upload provision function by TFTP Server.

Provision Active:	TFTP
2 Steps Configuration:	Disable
Server Auto Discovery:	Broadcasting
Scheduling:	Disable
TFTP Server:	192.168.55.91
TFTP File Path:	config/ Exp. file/load/

(Figure 9)

Step 3: Enter [Service Domain Setting] web page, check [Realm No.: 1] Settings, Use [MACID.dat] file to load configuration.

◆ **DHCP Option 66 (TFTP) (Please refer to DHCP Turbo and TFTP Turbo documents)**

Step 1: Install [DHCP Turbo + TFTPd32] software, then placed [MACID.dat] file into the specified directory.

Step 2: In [Auto Provision Setting] web page, Set up [Provision Active: TFTP, Server Auto Discover: DHCP Option 66 (TFTP)] (See Figure 10).

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	DHCP Option 66 (TFTP) ▼	
Scheduling:	Disable	▼
TFTP Server:	<input type="text"/>	
TFTP File Path:	<input type="text"/>	Exp. file/load/

(Figure 10)

Step 3: In [Auto Provision Setting] web page, Check [TFTP Server] field, upload this data (See Figure 11).

Note : If start TFTP Server at the same time, This field parameter can't see data, Because this data upload provision function by TFTP Server.

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	DHCP Option 66 (TFTP) ▼	
Scheduling:	Disable	▼
TFTP Server:	192.168.50.91	
TFTP File Path:	<input type="text"/>	Exp. file/load/

(Figure 11)

Step 6: Enter [Service Domain Setting] web page, Check [Realm: 1] Setting, Use [MACID.dat] file to load configuration.

◆ **DHCP Option 66 (TFTP)-2 (Please refer to DHCP Turbo document)**

Step 1: Install [DHCP Turbo + TFTP Turbo] software, Then placed [MACID.dat] file into the specified directory.

Step 2: In [Auto Provision Setting] web page, Set up [Provision Active: TFTP, Server Auto Discover: DHCP Option 66 (TFTP)] (See Figure 12).

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	DHCP Option 66 (TFTP) ▼	
Scheduling:	Disable	▼
TFTP Server:	192.168.50.91	
TFTP File Path:	<input type="text"/>	Exp. file/load/

(Figure 12)

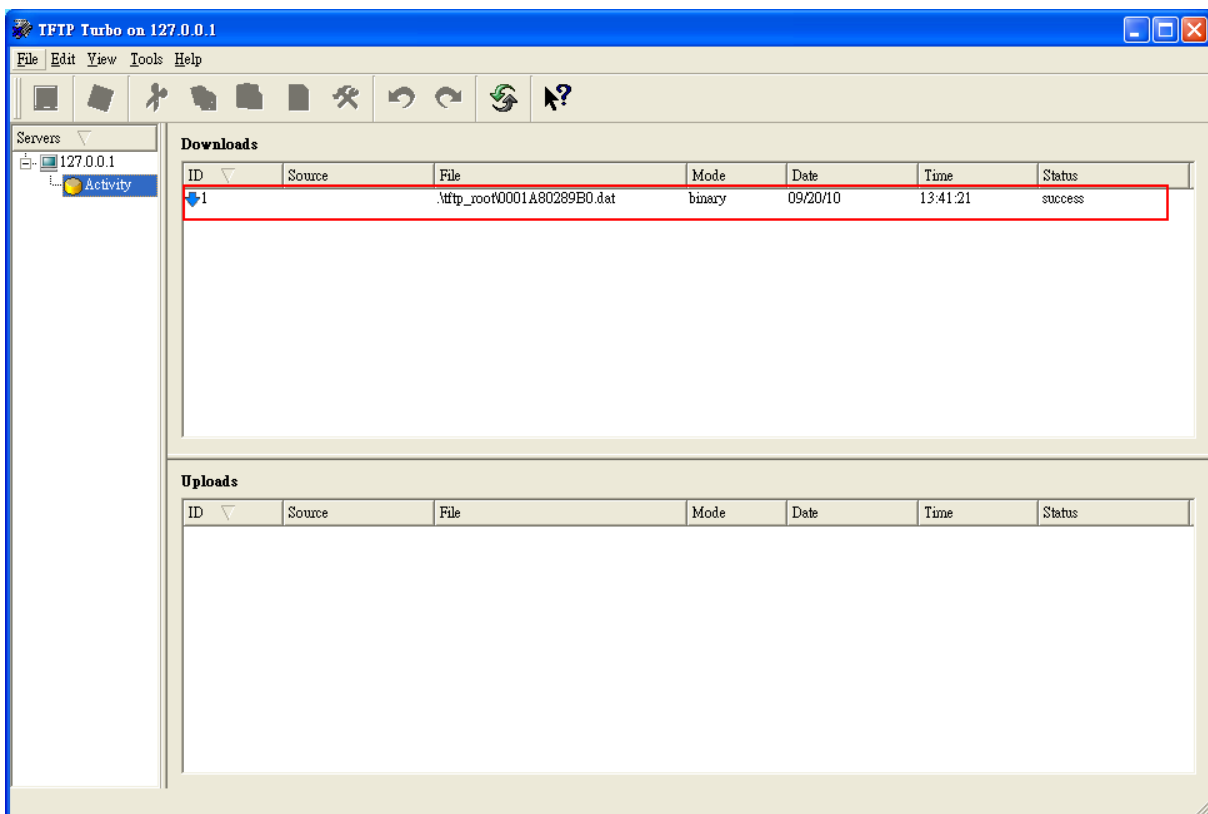
Step 3: In [Auto Provision Setting] web page, Check [TFTP Server] field, the field will upload this data (See Figure 13).

Note : If start TFTP Server at the same time, This field parameter can't see data, Because this data upload provision function by TFTP Server.

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	DHCP Option 66 (TFTP) ▼	
Scheduling:	Disable	▼
TFTP Server:	192.168.50.91	
TFTP File Path:		Exp. file/load/

(Figure 13)

Step 4: In [TFTP Turbo Server] menu, can check [Download] status, The page can view and the device will come to download [MACID.dat] file (See Figure 14).



(Figure 14)

Example 5 : Auto-Provision By Scheduling

Step 1: In [Auto Update Setting] web page, Set up [Scheduling (Date): 14, Scheduling (Time): AM 00:00 -5:59] (See Figure 15).

Check New Firmware Type:	Scheduling only ▼	
Scheduling (Date):	14	(1-30 Days)
Scheduling (Time):	AM 00:00-05:59 ▼	
Automatic Update Type:	Automatic ▼	

(Figure 15)

Step 2: In [Auto Provision Setting] web page, Set up [Provision Active: TFTP, Scheduling:

Enable, TFTP Server: 192.168.50.91] (See Figure 16).

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	Disable	▼
Scheduling:	Enable	▼
TFTP Server:	192.168.50.91	
TFTP File Path:		Exp. file/load/

(Figure 16)

Step 3: Return [Auto Update Setting] web page; In [Next Configuration Time] field, the field will show upgrade date and time at next time (See Figure 17).

Provision Active:	TFTP	▼
2 Steps Configuration:	Disable	▼
Server Auto Discovery:	Disable	▼
Scheduling:	Enable	▼
TFTP Server:	192.168.50.91	
TFTP File Path:		Exp. file/load/
Next Configuration time:	2014-02-24 01:34	

(Figure 17)

Note: How to establish MAC File data, Please refer 『Auto_provision_1.ppt』 file.

10.3 Auto Update (Firmware Auto Upgrade)

10.3.1 Function

Provide the types of TFTP, FTP, HTTP to update the firmware in **ssh** type.

10.3.2 Instruction

Auto Update Setting

Update Active:	<input type="text" value="Disable"/>	
TFTP Server:	<input type="text"/>	
TFTP File Path:	<input type="text"/>	Exp. file/load/
HTTP Server:	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path:	<input type="text"/>	Exp. /download/
FTP Server:	<input type="text"/>	Exp. 60.35.187.30
FTP User Name:	<input type="text"/>	
FTP Password:	<input type="text"/>	
FTP File Path:	<input type="text"/>	Exp. file/load/
Check New Firmware Type:	<input type="text" value="Scheduling only"/>	
Scheduling (Date):	<input type="text" value="14"/> (1~30 Days)	
Scheduling (Time):	<input type="text" value="AM 00:00-05:59"/>	
Automatic Update Type:	<input type="text" value="Automatic"/>	
Firmware File Prefix:	<input type="text" value="PHONE"/>	

Next Update time:

Item	Decription
Update Active	Default: Disable; When this function is on, LP399 will update the firmware from the designate Auto provision Server. Provide options: Disable, TFTP, FTP and HTTP.
TFTP Server	This column can only enter IP with the format of xxx.xxx.xxx.xxx; maximum length is 15 bytes.
TFTP File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, for instance: 123/.
HTTP Server	This column can enter IP or Domain Name; maximum length is 63 bytes.
HTTP File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, ex: 123/.
FTP Server	This column can enter IP or Domain Name; maximum length is 63 bytes.
FTP User Name	This column can enter IP or Domain Name; maximum length is 63 bytes.
FTP Password	This column can enter IP or Domain Name; maximum length is

Item	Description
	63 bytes.
File File Path	This column can enter numbers or strings; maximum length is 63 bytes with the "/" in the end, ex: 123/.
Check new Firmware Type	Default: Scheduling Only. Set up the type for checking new firmware. - Power on and Scheduling: Check the new firmware when powers on and base on Scheduling - Scheduling: According to [Next Update Time] to check the new firmware. Provide options: Power on and Schedule, Scheduling Only. * Power on and Scheduling: When LP399 discovers a new firmware which is different with current one, it will not update immediately. But you will hear a hint tone or see a [Found new s/w] message on LCD. You should update firmware by your decision.
Scheduling (Date)	Default: 14 (day). This column can only enter numbers; maximum length is 2 bytes with a range of 1~30.
Scheduling (Time)	Default: AM 00:00 – 05:59. Provide options: AM 00:00 – 05:59, AM 06:00 – 11:59, AM 12:00 – 17:59, AM 18:00 – 23:59.
Automatic Update Type	Default: Notify only. Set up the type to update firmware. Provide option: Notify only, Automatic. - Notify only: When LP399 discover a new firmware, it will not update firmware immediately. But you will hear a hint tone from LP399 or see a [Found new s/w] message on LCD display. - Automatic: Update firmware automatically.
Firmware File Prefix	Default is production model. This is used to judge which model ask for update, such as Phone or ATA. This column can enter numbers or strings; maximum length is 8 bytes.
Next Update Time	LP399 will check the Update Server when the Next Update time is up. The start counting date is the next day, so the Next Update time will add one day. Count rule : the next day + days + time period + MACaddress + random number = Next Update time.
Submit [Button]	Save the Settings.

NOTE : Firmware updated manually at Auto Provision mode.

1. You will hear "DuDuDu" alert tone from handset when you pickup handset after updated firmware was available. If you give up update firmware procedure here, you don't hear any alert tone at next time.
2. If you want to proceed firmware upgrade procedures, dial #190# and hang on handset.
3. Pick up Handset again, dial #160# to enter firmware upgrade procedures.

Once ATA enter firmware upgrade procedures, it takes about 2 to 3 minutes to complete. ATA don't implement any job or function in this period. Please don't unplug power adaptor during firmware upgrade procedures in order to prevent from failure.

10.3.3 Operate Instruction

Example : Build an Auto Update File [***_ver.dat]

Step 1: The filename for checking firmware version, the filename is according to [Firmware File Prefix] on [Auto Update Setting] and adds [_ver.dat] to be [filename_ver.dat], (See Figure 1).

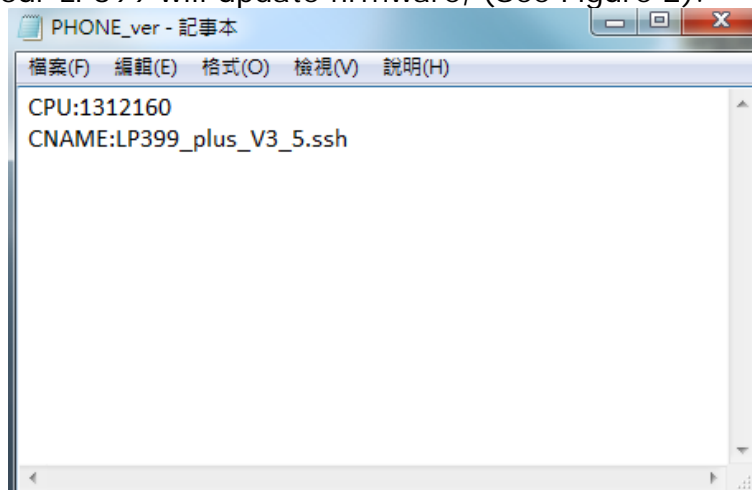
Ex : Firmware File Prefix: PHONE, the filename for checking firmware version is TA1S_ver.dat.

Pay attention to [Firmware File Prefix], the name must be the same with ***_ver.dat.

Check New Firmware Type:	Power ON and Scheduling ▾
Scheduling (Date):	5 (1-30 Days)
Scheduling (Time):	PM 12:00-17:59 ▾
Automatic Update Type:	Notify only ▾
Firmware File Prefix:	PHONE

(Figure 1)

Step 2: After build a [PHONE_ver.dat] file, there must have [CPU, CNAME] two data to compare with updating file in Server. If the Server one is newer than your existing one, then your LP399 will update firmware, (See Figure 2).



(Figure 2)

Note:

CPU: The date of updating file, it can only enter numbers.

CNAME: The updating filename, please enter a complete name without space or signs.

Step 3: Please place ***_ver.dat and *.ssh files in the designated folder of the Server.

Note: This example device is IP-Phone(LP399), ATA series is the same method.

10.4 New Firmware

10.4.1 Function

The Firmware upgrade only support `.ssh` format.

10.4.2 Instruction

Firmware Upgrade

Update Active:	CPU+DSP xxxx.ssh ▾
Load From File:	<input type="text"/> 瀏覽...
<input type="button" value="Upgrade"/>	

Item	Decription
Update Active	Default: CPU+DSP xxxx.ssh; Upgrade file format.
Load From File	The location of the updating file.
Upgrade [Button]	Execute updating firmware.

PS : It takes 2 or 3 minutes when you start update firmware. User can't make call during the upgrade. Please don't turn off the power.

Caution : ATA171plus, ATA172plus firmware are different with ATA-171, ATA-172, please only use correct firmware for these models.

10.5 Advanced (Advanced Settings)

10.5.1 Function

Provide anonymous call, billing signal, encryption, syslog and FXS/FXO parameter function setting.

10.5.2 Instruction

Figure 1: 1FXS(ATA171plus, ATA-171), 2FXS(ATA172plus, ATA-172) and 1FXS+1PSTN(ATA-171P).

Management - Advanced Setting

ICMP Not Echo:	Disable
Anonymous Call:	Disable
Management from WAN:	Enable
WEB Login Port:	9999 (1-65535)
Telnet Login:	Enable
Stop Feature Tone:	Disable (MWI, forward, Do Not Disturb....)
Billing Signal:	Disable
CPC Delay:	2 (seconds)
CPC Duration:	0 (0-120; x 10ms)
IP Dialing Format:	Type 1 (x@x.x.x.x)
Send Flash Event:	Disable
Encryption Type:	Disable
Encryption Key:	
PPPoE Retry Period:	5 (0-250 seconds)
DHCP Gateway ARP Check Period:	0 (0 or 30-300 seconds)
Syslog Server IP Address:	
System Log:	Disable
FXS Port Country:	USA
Flash Hook Time (MAX):	60 (4-255; x 10ms)
Flash Hook Time (MIN):	7 (3-12; x 10ms)
NET Bandwidth Limit:	Disable Kbps

(Figure 1)

Item	Description
ICMP Not Echo	Default is Disable. When ICMP was set to Enable, ATA doesn't response PING command. Option: Disable, Enable
Send Anonymous CID	Default is Disable. When ATA was set to Type 1 or Type 2 , ATA will send out anonymous to SIP Server (or to remote party) instead of CID. Type 1 (anonymous@anonymous.invalid) Type 2 (anonymous@x.x.x.x) * Your Register Proxy server must support this function.
Management from WAN	Default is Enabl which allows web management access from WAN port. When it was set to Disable, ATA only allow web access vial LAN port. Option: Disable, Enable
Stop Feature Tone	Default is Disable. This feature is to provide alert tone at the following messages.

Item	Description
	<p>Enable: If you enable [Subscribe for MWI, forward, DND] function, you will hear the alert tone(DuDuDu.....) when you pick up the phone.</p> <p>Option: Disable, Enable.</p>
Billing Signal	<p>Default is Disable. This feature is to provide start billing signal when call was established.</p> <p>Option: Disable, Polarity Reversal, Tone_12K, Tone_16K.</p>
CPC Delay	<p>Default is 2 seconds. Delay how many seconds to send CPC signal (Loop Current Drop signal) to Analog telephone set when ATA received drop call signal from IP SIP command. Only numbers are accepted, data range is (2~5 seconds), maximum data length is 1 digit.</p>
CPC Duration	<p>Default is 120ms. Setting CPC feature was activated duration (Loop Current Drop duration), data range is (0~120ms), maximum length: 3 digits.</p>
IP Dialing Format	<p>Default is Type 1 (x@x.x.x.x); Define the IP dialing format.</p> <p>Option: Disabled, Type 1 (x@x.x.x.x), Type 2 (x.x.x.x).</p>
Send Flash event	<p>Default is Disable; When you press Flash Key at analog phone set to do Transfer feature, ATA will send different event messages to IP side.</p> <p>Option:</p> <p>Disable: Send [SIP/DSP, Content-Type=applicatio-sdp].</p> <p>DTMF Event: Send [RTP event, Payload type=RTP event Flash].</p> <p>SIP Info: send [SIP, Request: INFO sip:xxx@xxxx].</p>
Encrypt Type	<p>Default is Disable.</p> <p>Option: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP and Welltech. VGCP is popular in the market. Once this feature was selected, both voice codec and SIP command were encrypted during transmit on IP network.</p> <p>* Note: Your Registered Proxy server must support the same encryption type with ATA.</p>
Encrypt Key	<p>Set encryption password. Only VGX encryption format need password. Maximum data is 63 digits which can be numbers or strings.</p>
PPPoE Retry Period	<p>Default is 5 (Seconds). Set the time for PPPoE to retry when PPPoE failed. Only numbers are accepted, data range: (5~255) seconds, maximum length is 3 digits.</p>
System Log Server	<p>Sending ATA debug messages to System Log Server which can be IP Address or Domain Name Address. Format: xxx.xxx.xxx.xxx; Maximum length is 63 digits.</p>
System Log Type	<p>Default is Disable; Define Syslog type or Log message type.</p> <p>Option: Disable, Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug, All.</p>
FXS Port Country	<p>Default is USA. To select FXS Port impedance of the analog telephone by different country's specification.</p>
Flash Signal Detect (Max)	<p>Default is 60 (equal 600ms). To detect Hook Flash Time at maximum time. When Flash time is less than 600ms, it was regarded as HOOK FLASH command.</p>

Item	Decription
	When Flash time is longer than 600ms, it was regarded as On-Hook (drop call) command. Configuration range is from (4~255), Unit: 10ms. Maximum length is 3 digits.
Flash Signal Detect (Min)	Default is 7(equal to 70ms). When Flash Time is longer than 70ms, it will be regarded as FLASH command. When Flash Time is less than 70 ms, it will be regarded as On-Hook. Configuration range is from (3~12), Unit: 10ms. Minimum length is 3 digits.
NET Bandwidth Limit	Default is Disable. LAN port bandwidth limitation. Option: Disable, 128 , 256 , 512 , 1024 , 2048 , 4096 , 8192 kbps .
Submit [button]	Save the configuration.

Figure 2: 1FXS + 1FXO, ATA-171M.

Management - Advanced Setting

ICMP Not Echo:	<input type="text" value="Disable"/>
Anonymous Call:	<input type="text" value="Disable"/>
Management from WAN:	<input type="text" value="Enable"/>
WEB Login Port:	<input type="text" value="9999"/> (1-65535)
Telnet Login:	<input type="text" value="Enable"/>
Stop Feature Tone:	<input type="text" value="Disable"/> (MWI, forward, Do Not Disturb....)
Billing Signal:	<input type="text" value="Disable"/>
CPC Delay:	<input type="text" value="2"/> (seconds)
CPC Duration:	<input type="text" value="0"/> (0~120; x 10ms)
IP Dialing Format:	<input type="text" value="Type 1 (x@x.x.x.x)"/>
Send Flash Event:	<input type="text" value="Disable"/>
Encryption Type:	<input type="text" value="Disable"/>
Encryption Key:	<input type="text"/>
PPPoE Retry Period:	<input type="text" value="5"/> (0-250 seconds)
DHCP Gateway ARP Check Period:	<input type="text" value="0"/> (0 or 30-300 seconds)
Syslog Server IP Address:	<input type="text"/>
System Log:	<input type="text" value="Disable"/>
PSTN Port Country:	<input type="text" value="USA"/>
PSTN Silence Timeout:	<input type="text" value="30"/> (1-250 minutes)
PSTN CID forward:	<input type="text" value="Disable"/>
Generate Flash Signal for PSTN:	<input type="text" value="10"/> (9~120; x 10ms)
FXS Port Country:	<input type="text" value="USA"/>
Flash Hook Time (MAX):	<input type="text" value="60"/> (4-255; x 10ms)
Flash Hook Time (MIN):	<input type="text" value="7"/> (3-12; x 10ms)
NET Bandwidth Limit:	<input type="text" value="Disable"/> Kbps

(Figure 2)

Item	Decription
ICMP Not Echo	Default is Disable; Enable: ping will not reply. Option: Disable, Enable
Send Anonymous CID	Default is Disable; When you set Type 1 or Type 2, ATA will send CID as anonymous to your server.

Item	Description
	Type 1 (anonymous@anonymous.invalid) Type 2 (anonymous@x.x.x.x) * Your Register Proxy server must support this function.
Management form WAN	Default is Enable; ATA allow web management via WAN port. Disable: ATA only allow web access vial LAN port. Option: Disable, Enable
Stop Feature Tone	Default is Disable Enable: If you enable [Subscribe for MWI, forward, DND] function, you will hear the alert tone when you pick up the phone. Option: Disable, Enable.
Billing Signal	Default is Disable. Option: Disable, Polarity Reversal, Tone_12K, Tone_16K.
CPC Delay	Default: 2(sec); Setting how long it takes for the voltage reaches 0V when receiving hang up signal. Only numbers are accepted, data range (2~5 sec.), maximum length: 1 byte.
CPC Duration	Default: 120ms. Setting how long it takes for the voltage reaches 0V, data range (0~120), maximum length: 3 bytes.
IP Dialing Format	Default is Type 1 (x@x.x.x.x); Define the IP dialing format. Option: Disabled, Type 1 (x@x.x.x.x), Type 2 (x.x.x.x)
Send Flash event	Default is Disable; When you press [Hook/Flash (Transfer)] ATA will send different event. Option: Disable: [SIP/DSP, Content-Type=applicatio-sdp] DTMF Event: [RTP event, Payload type=RTP event Flash] SIP Info: [SIP, Request: INFO sip:xxx@xxxx]
Encrypt Type	Default is Disable. Option: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP and Welltech . * Your Register Proxy server must support this function.
Encrypt Key	Set encryption password ◦ Only support GVX encryption format, maximum data: 63 bytes.
PPPoE Retry Period	Default: 5 (Seconds) ; Set up how long it takes for PPPoE to retry when PPPoE failed. Only numbers are accepted, data range: (5~255), maximum length: 3 digits.
System Log Server	Sending ATA debug message to System Log Server which can be an IP Address or Domain Name Address. Format: xxx.xxx.xxx.xxx; Maximum length: 63 digits.
System Log Type	Default is Disable; Define Syslog type and Log messages. Option: Disable, Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug, All.
FXO Port Coutry	Default is USA country telephony specification. To select FXO Port impedance of the analog telephone by different countries.
FXO Silence Time	Default to 30 minutes. Configure FXO silence time setting to release FXO port automatically. Time length is 1 to 250 minutes.
FXO CID forward	Default is Disable. When FXO port received an incoming call, ATA will forward this call as well as caller ID to IP side. Go to webpage configuration to enable the setting at Phone –

Item	Description
	General] -> [Auto Answer] or [Phone – Caller Service] -> [Forward] . If ATA was set Forward & Auto-answer , the CID of incoming call will be forwarded to SIP Server.
Generate Flash Signal for FXO	Default is 10 (equal to 100ms). Generate Flash Singal for FXO: 100ms When the Flash signal : < (less than) 100 ms, it will be regarded as Hook Flash. > (longer than) 100 ms, it will be regarded as On-Hook. Unit: 10ms. Maximum length is 3 digits.
FXS Port Coutry	Default is USA type impedance. To select FXS Port impedance of the analog telephone by different country's specification.
Flash Signal Detect (Max)	Default is 60 (equal 600ms). To detect Hook Flash Time at maximum time. When Flash time is less than 600ms , it was regarded as HOOK FLASH command. When Flash time is longer than 600ms, it was regarded as On-Hook (drop call) command. Configuration range is from (4~255), Unit: 10ms. Maximum length is 3 digits.
Flash Signal Detect (Min)	Default is 7(equal to 70ms) . When Flash Time is longer than 70ms, it will be regarded as FLASH command. When Flash Time is less than 70 ms, it will be regarded as On-Hook. Configuration range is from (3~12), Unit: 10ms. Minimum length is 3 digits.
NET Bandwidth Limit	Default is Disable. LAN port bandwidth limitation. Option: Disable, 128 , 256 , 512 , 1024 , 2048 , 4096 , 8192 kbps .
Submit [button]	Save the configuration.

10.5.3 Operate Instruction

Example 1: System Log (Please Start TFTP or System Log Server)

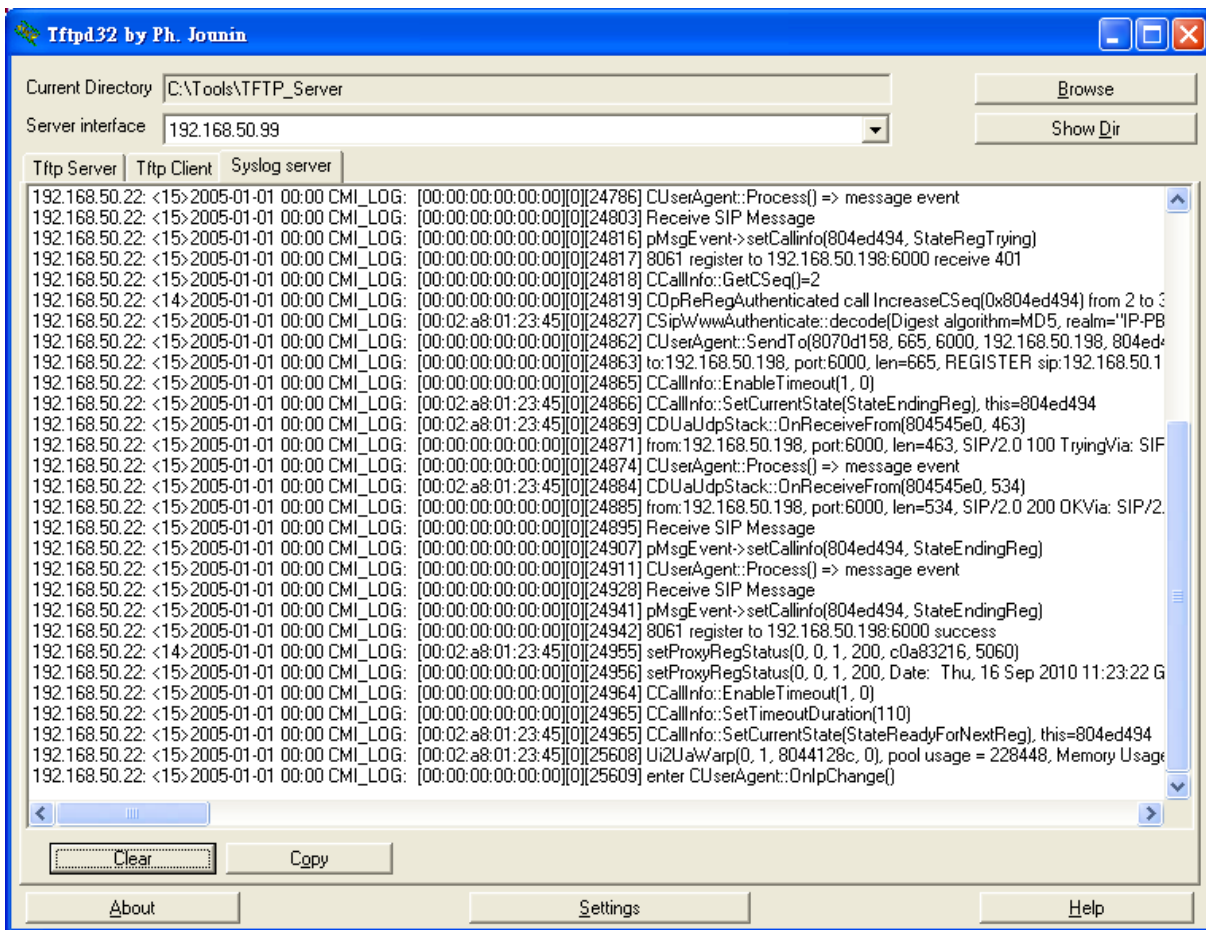
◆ System Log Type: All

Step 1: In [Management - Advanced Setting] web page, Set up [System Log Server: 192.168.50,99, System Log Type: All] (See Figure 1).

Syslog Server IP Address:	192.168.50.99
System Log:	<ul style="list-style-type: none"> Disable Call Statistics General Debug Call Statistics + General Debug SIP Debug Call Statistics + SIP Debug General Debug + SIP Debug All
NET Bandwidth Limit:	
<input type="button" value="Submit"/>	

(Figure 1)

Step 2: In [TFTP]'s [Syslog server] web page, this page show log message data (See Figure 2).



(Figure 2)

Note: User can use ethereal capture to check syslog data.

Example 2: DHCP Gateway ARP Check Period

Step 1: In [WAN Setting] web page, Set up [WAN Active: DHCP] (See Figure3).

WAN Setting

WAN Active:	DHCP
IP Address:	192.168.23.21
Subnet Mask:	255.255.248.0
Default Gateway:	192.168.16.254
DNS Active:	Static
Primary DNS:	168.95.192.1
Second DNS:	168.95.1.1
MAC Address:	00:01:a9:39:90:00
System Name:	VOIP_PHONE

(Figure 3)

Step 2: In [Management - Advanced Setting] web page, Set up [DHCP Gateway ARP Check Period: 30] (See Figure 4).

PPPoE Retry Period:	<input type="text" value="5"/>	(0~250 seconds)
DHCP Gateway ARP Check Period:	<input type="text" value="30"/>	(0 or 30~300 seconds)

(Figure 4)

10.6 Password (Change Login Account)

10.6.1 Function

Password Provides 3 Authority functions to change their User name and Password, respectively.

10.6.2 Instruction

Figure 1: Admin

Account & Password Setting

Admin	
New User Name:	<input type="text"/>
New Password:	<input type="text"/>
Confirm Password:	<input type="text"/>
System	
New User Name:	<input type="text"/>
New Password:	<input type="text"/>
Confirm Password:	<input type="text"/>
User	
New User Name:	<input type="text"/>
New Password:	<input type="text"/>
Confirm Password:	<input type="text"/>
<input type="button" value="Submit"/>	

(Figure 1)

Item	Decription
Admin	Administrator (the highest authority): it can only Set up an Administrator account. Default Username: root , default password: test
New User Name	Enter new user name, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, *, #, .); maximum length is 32 bytes.
New Password	Enter new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Confirmed Password	Confirm new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
System	System (the middle authority): it can only Set up a System account. Default Username: system , default password: test
New User Name	Enter new user name, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, *, #, .); maximum length is 32

Item	Decription
	bytes.
New Password	Enter new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Confirmed Password	Confirm new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
User	Normal User (the lowest authority): it can only Set up a Normal User account. Default Username: user, default password: test
New User Name	Enter new user name, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, *, #, .); maximum length is 32 bytes.
New Password	Enter new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Confirmed Password	Confirm new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Submit [Button]	Save the Settings.

Figure 2: System Authority Account & Password Setting

System

New User Name:

New Password:

Confirm Password:

User

New User Name:

New Password:

Confirm Password:

(Figure 2)

Item	Decription
System	System (the middle authority): it can only Set up a System account. Default Username: system , default password: test
New User Name	Enter new user name, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, *, #, .); maximum length is 32 bytes.
New Password	Enter new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %);

Item	Decription
	maximum length is 32 bytes.
Confirmed Password	Confirm new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
User	Normal User (the lowest authority): it can only Set up a Normal User account. Default Username: user, default password: test
New User Name	Enter new user name, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, *, #, .); maximum length is 32 bytes.
New Password	Enter new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Confirmed Password	Confirm new user password, This column can enter numbers or strings (support: 0~9, a~z, @, _, -, ., +, :, [,], *, #, !, %); maximum length is 32 bytes.
Submit [Button]	Save the Settings.

10.7 Tones (Audio Frequency Set up)

10.7.1 Function

Tone setting provides Dial, Ring Back, Busy, Congestion, Ring, Call Waiting Tone and Multi-Frequency configuration.

10.7.2 Instruction

Tones Setting

	Dial	Ring Back	Busy	Congestion	Ring	Call Waiting
Cadence On:	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Hi-Tone Freq.:	<input type="text" value="440"/>	<input type="text" value="480"/>	<input type="text" value="620"/>	<input type="text" value="620"/>	<input type="text" value="480"/>	<input type="text" value="440"/>
Lo-Tone Freq.:	<input type="text" value="350"/>	<input type="text" value="440"/>	<input type="text" value="480"/>	<input type="text" value="480"/>	<input type="text" value="440"/>	<input type="text" value="350"/>
Hi-Tone Gain:	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="15360"/>	<input type="text" value="2261"/>
Lo-Tone Gain:	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="2261"/>	<input type="text" value="15360"/>	<input type="text" value="1130"/>
On Time 1:	<input type="text" value="0"/> x 10ms	<input type="text" value="200"/>	<input type="text" value="50"/>	<input type="text" value="30"/>	<input type="text" value="200"/>	<input type="text" value="30"/>
Off Time 1:	<input type="text" value="0"/> x 10ms	<input type="text" value="400"/>	<input type="text" value="50"/>	<input type="text" value="20"/>	<input type="text" value="400"/>	<input type="text" value="20"/>
On Time 2:	<input type="text" value="0"/> x 10ms	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="30"/>
Off Time 2:	<input type="text" value="0"/> x 10ms	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="400"/>
On Time 3:	<input type="text" value="0"/> x 10ms	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>
Off Time 3:	<input type="text" value="0"/> x 10ms	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>

Tone Gain Value: 372767-> 0bB, 16384-> -6dB, 8192-> -12dB

Item	Decription
Dial Tone	Set up the Dial Tone Settings.
Cadence On	Default: Disable; When check the box, Cadence On will Enable.
Hi-Tone Freq	Default: 440; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096. This is high frequency tone.
Lo-Tone Freq	Default: 350; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096. This is low frequency tone.
Hi-Tone Gain	Default: 4522; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535. °
On Time 1	Default: 0; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms). °
Off Time 1	Default: 0; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).

Item	Description
On Time 2	Default: 0; Set up the second category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Off Time 2	Default: 0; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Ring Back	Set up the Ring Back Tone Settings
Cadence On	Default: Enable; When check the box, Cadence On will Enable.
Hi-Tone Freq	Default: 480; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Lo-Tone Freq	Default: 440; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Hi-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
On Time 1	Default: 200; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Off Time 1	Default: 400; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
On Time 2	Default: 0; Set up the second category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Off Time 2	Default: 0; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~999 (10ms).
Busy	Set up the Busy Tone Settings
Cadence On	Default: Enable; When check the box, Cadence On will Enable.
Hi-Tone Freq	Default: 620; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Lo-Tone Freq	Default: 480; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Hi-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 2261; This column can only enter numbers; maximum

Item	Description
	length is 5 bytes with a range of 0~65535. °
On Time 1	Default: 50; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 1	Default: 50; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 2	Default: 0; Set up the second category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 2	Default: 0; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms). °
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Congestion	Set up the Congestion Tone Settings.
Cadence On	Default: Enable; When check the box, Cadence On will Enable.
Hi-Tone Freq	Default: 620; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Lo-Tone Freq	Default: 480; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Hi-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
On Time 1	Default: 30; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 1	Default: 20; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 2	Default: 0; Set up the second category of time start. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 2	Default: 0; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Ring	Set up the Ring Tone Settings.
Cadence On	Default: Enable; When check the box, Cadence On will Enable.

Item	Description
Hi-Tone Freq	Default: 480; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Lo-Tone Freq	Default: 440; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Hi-Tone Gain	Default: 15360; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 15360; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535. °
On Time 1	Default: 200; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 1	Default: 400; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 2	Default: 0; Set up the second category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 2	Default: 0; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Call Waiting	Set up the Call Waiting Tone Settings.
Cadence On	Default: Enable; When check the box, Cadence On will Enable.
Hi-Tone Freq	Default: 440; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Lo-Tone Freq	Default: 350; This column can only enter numbers; maximum length is 4 bytes with a range of 0~4096.
Hi-Tone Gain	Default: 2261; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
Lo-Tone Gain	Default: 1130; This column can only enter numbers; maximum length is 5 bytes with a range of 0~65535.
On Time 1	Default: 30; Set up the first category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 1	Default: 20; Set up the first category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 2	Default: 30; Set up the second category of ON time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Off Time 2	Default: 400; Set up the second category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
On Time 3	Default: 0; Set up the third category of ON time. This column can only enter numbers; maximum length is 3 bytes

Item	Decription
	with a range of 0~ 999 (10ms).
Off Time 3	Default: 0; Set up the third category of OFF time. This column can only enter numbers; maximum length is 3 bytes with a range of 0~ 999 (10ms).
Submit [Button]	Save the Settings.

10.8 Default

10.8.1 Function

Default Provides the way of eliminating all Settings and reloading the default Settings except the data of Phone Book.

10.8.2 Instruction

Reset to Default

Reset to Factory Setting:

Backup or Restore Setting

Export Setting

Restore Setting

Item	Decription
Reset [Button]	Clear all Setting and reset to default, Then reboot system automatically.
Export [Button]	[Export] button, Provide device configuration back up. Default file name is config.db.
Restore [Button]	Provide restore feature, Import file format is .db.

Note: If user use [Update.htm] web page's [Restore Configuration] to upload [config.db] file, it can't clear configuration from this feature.

10.8.3 Operate Instruction

Example 1: Export System Configuration

Step 1: Please finish the Web parameter Set up, Then reboot LP399.

Step 2: In [Reset to Default] web page, Press [Export System Configuration]'s [Export] button (See Figure 1).

Backup or Restore Setting

Export Setting

(Figure 1)

Step 3: Enter [Save File] menu, Default file name is config.db, press [Save] button, Save the system parameter to .db file (See Figure 2).



(Figure 2)

Example 2: Restore Setting

Step 1: In [Reset to Default] web page, Press [Browse] button. After choosing the

configuration file, Press [Open] button. Return [Reset to Default] web page, make sure to perform the update, Press [Restore] button (See Figure 3).

Reset to Default

Reset to Factory Setting:

Backup or Restore Setting

Export Setting

Restore Setting C:\Users\KevinLiu\Desktop\config.db

(Figure 3)

Step 2: Restore complete, Must save setting and reboot system (See Figure 4), Reboot system can take effect the [config.db] parameter.

Note Information

Please wait for a moment while rebooting ...

(Figure 4)

10.9 Language

10.9.1 Function

Provide language option for web configuration, ATA will auto reboot after press [Submit]. [Don't remove Power adaptor at this moment.](#)

10.9.2 Instruction

Language Setting

WEB Language:

Item	Decription
Choice Language	Default is English. ATA has to restart after you have changed Web language. Option: English, Chinese, Simplified Chinese
Submit [button]	Save the configuration.

11. Save & Reboot

Save configuration and Reboot ATA.

11.1 Function

Save Change: Save configuration and auto reboot to take effect.

Reboot System: Reboot ATA

11.2 Instruction

Save and Reboot System

Save Change:

Reboot System Now:

Item	Decription
Save [Button]	Save all Settings and restart device.
Reboot [Button]	Restart device.

12. Logout (Login System)

12.1 Function

Logout system and return to login page.

12.2 Instruction

Logout

Logout System Now ?

Logout

Item	Decription
Logout [Button]	Logout system Settings menu; Return web page login.

