

VoIP Telephone Adapter User's Manual



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1 Introduction

This user's manual is for VoIP Telephone Adapter (TA). This user's manual will explain the IVR instruction, web configuration and command line configuration for the TA. Before using the TA, some setup processes are required to make the TA work properly. Please refer to the Setup Menu for further information.

1.1 Product Overview (Single Phone Port Model)

The TA has the following interfaces for Networking, telephone interface, LED indication, and power connector.

Networking

These two interfaces support 10/100Mps Fast Ethernet.

WAN: Connect to the ADSL or Router.

LAN: Connect to your PC computer. You can then connect to Internet through the Internet Sharing function of the Adapter.

Phone

PHONE: Connect to the standard (analog) phone or Fax machine

LINE: (Optional) Connect to the landline (PSTN), usually presented as a phone jack on the wall of your home or office



LED Light Indicators

There are four LED indicators in the TA to show the Power, WAN, LINE and VOIP indication.

Power: On when Power is on

WAN: On when the WAN network link is up

LINE: On when the phone is switched to regular landline (PSTN).

VOIP: On when VoIP service is registered, blink when the VoIP call is active.



1.2 Product Overview (Dual Phone Port Model)

The TA has the following interfaces for Networking, telephone interface, LED indication, and power connector.

Networking

These two interfaces support 10/100Mbps Fast Ethernet.

WAN: Connect to the ADSL or Router.

LAN: Connect to your PC computer. You can then connect to Internet through the Internet Sharing function of the Adapter.

Phone

PHONE1: Connect to a standard (analog) phone or Fax Machine

PHONE2: Connect to second standard (analog) phone or Fax Machine



LED Light Indicators

There are four LED indicators in the TA to show the Power, WAN, PHONE1 and PHONE2 indication.

Power: On when Power is on

WAN: On when the WAN network link is up

PHONE1: On when VoIP service of Phone 1 is registered, blink when Phone 1 is active.

PHONE2: On when VoIP service of Phone 2 is registered, blink when Phone 2 is active.



1.1 Software Overview

<p>Network Protocol</p> <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 	<p>Tone</p> <ul style="list-style-type: none"> • Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • Programming Tone
<p>Codec</p> <ul style="list-style-type: none"> • G.711: 64k bit/s (PCM) • G.723.1: 6.3k / 5.3k bit/s • G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: adds VAD & CNG to G.729 	<p>Phone Function</p> <ul style="list-style-type: none"> • Volume Adjustment • Speed dial key • Phone book • Flash
<p>Voice Quality</p> <ul style="list-style-type: none"> • VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer 	<p>IP Assignment</p> <ul style="list-style-type: none"> • Static IP • DHCP • PPPoE
<p>Call Function</p> <ul style="list-style-type: none"> • Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference 	<p>Security</p> <ul style="list-style-type: none"> • HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/ RFC 2617)
<p>DTMF Function</p> <ul style="list-style-type: none"> • In-Band DTMF • Out-of Band DTMF • SIP Info 	<p>QoS</p> <ul style="list-style-type: none"> • ToS field
<p>SIP Server</p> <ul style="list-style-type: none"> • Registrar Server (three SIP account) • Outbound Proxy 	<p>NAT Traversal</p> <ul style="list-style-type: none"> • STUN
	<p>Configuration</p> <ul style="list-style-type: none"> • Web Browser • Console/Telnet • IVR/Keypad
	<p>Firmware Upgrade</p> <ul style="list-style-type: none"> • TFTP • Console • HTTP

2 IVR Interface for TA

You can use the connected phone to configure the TA. Please follow the instruction to configure your Telephone Adapter.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Function	unlock keypad	#190#	None	If you want to let user change the setting by keypad. You should hear two beep sounds when unlock is succeed
Function	lock keypad	#191#	None	If you want to lock user change the setting by keypad.
Function	Reboot	#195#	None	After you hear "Option Successful," hang-up. The system will reboot automatically. (Need Unlock First)
Function	Factory Reset	#198#	None	System will automatically Reboot. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data. (Need Unlock First)
Function	enable call waiting	#138#	None	Enable Call waiting
Function	disable call waiting	#139#	None	Disable call waiting
Info	Check LAN port IP Address	#120#	None	IVR will announce the current IP address of the TA
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 TA.
Info	Check Gateway IP Address	#124#	None	IVR will announce the current gateway IP address of the TA.
Info	Check Primary DNS Server Setting	#125#	None	IVR will announce the current setting in the Primary DNS field.
Info	Check WAN port IP Address	#126#	None	IVR will announce the current IP address (WAN Port) of the TA
Info	Check Firmware Version	#128#	None	IVR will announce the version of the firmware running on the TA.
Setting	Set DHCP client	#111#	None	The system will change to DHCP Client type (Need Unlock First)
Setting	Set Static IP Address	#112xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type. (Need Unlock First)
Setting	Set Network Mask	#113xxx*xxx*xxx*xxx#	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. (Need Unlock First)
Setting	Set Gateway IP Address	#114xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. (Need Unlock First)

Setting	Set Primary DNS Server	#115xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. (Need Unlock 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. (Need Unlock First)
Setting	Set Handset Gain	#131+[00~15]#	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6 (Need Unlock First)
Setting	Set Handset Volume	#132+[00~12]#	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10 (Need Unlock First)

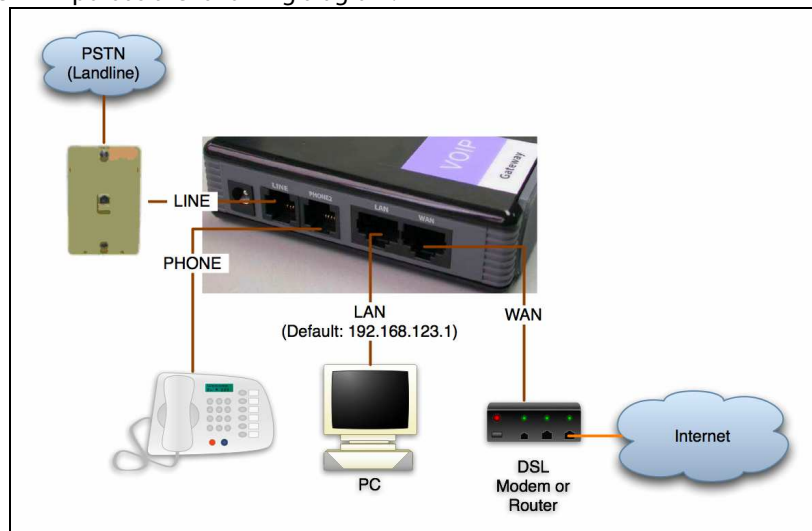
3 Setup the TA by Web Browser

The TA provides a built-in web server. You can use Web browser to configure the TA. First please input the IP address in the Web page. In the end of IP address, please add the port number ":9999". Ex: http://192.168.123.1:9999

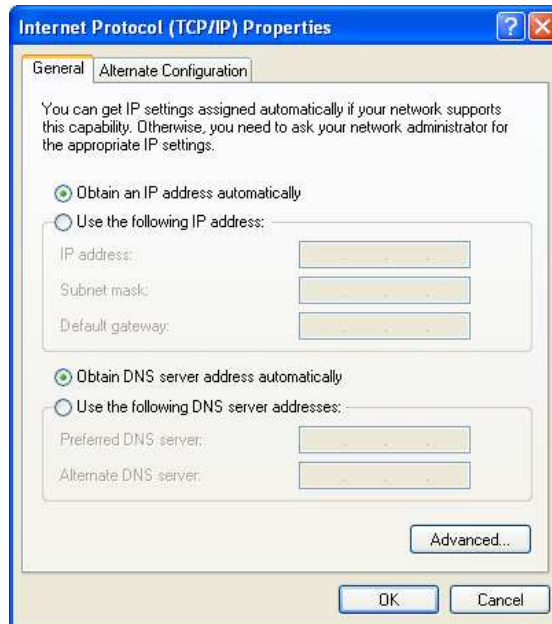
The default LAN port IP is **192.168.123.1**, with DHCP Server enabled. The default WAN port has DHCP client enabled; the IP would be automatically assigned if it is connected to a network with DHCP server.

3.1 First Time Login

1. Connected your PC to the LAN port of the Adapter, and your DSL modem or router to the WAN port as the following diagram.

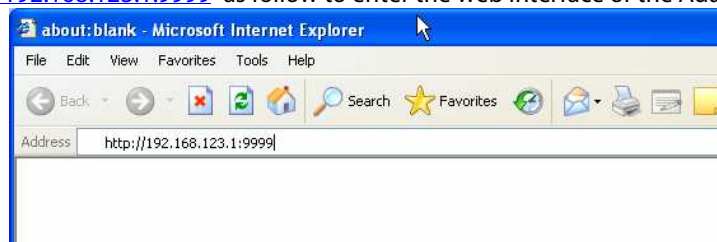


2. Enable the "Obtain an IP address automatically" in your "Local Area Connection" inside Windows, under "Control Panel" -> "Network Connections" -> "Local Area Connection" -> Property -> "Internet Protocol" -> Properties



Your PC would then obtain an IP from the Adapter.

3. Inside your PC, open the browser (Internet Explorer), and enter "<http://192.168.123.1:9999>" as follow to enter the web interface of the Adapter



4. Please input the username and password into the blank field. The default setting is:
 - For normal user, **the username is: user; and the password is: test.** If you use the account login, but you can not configure the SIP setting.
 - For Administrator, **the username is: root; and the password is: test.** If you use the account login, you can configure all the setting. *Please inquiry your service provider if the password is not correct.*



Troubleshooting Tips:

If there is no login page (shown on the left) appears. Please check:

- i. The link typed in the web browser (Internet Explorer). Make sure "<http://>" and the port ":9999" is entered, such that the link is in the form of <http://192.168.123.1:9999>
- ii. The IP address may be changed due to whatever reason. You may plug in a standard phone to the phone, and check the LAN ip address by enter "#120#"

5. Click the "Login" button will move into the TA web based management interface. **If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button.** When you finished the setting, please click the Reboot

function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.

3.2 Default Reset from Keypads

- Pick up the handset of the Phone connected to the Phone Adapter
- Press keys “#190#” to unlock the phone. You should hear two beep sounds then. Hook on the handset.
- Pick up the handset again, and press keys “#198#” to reset the Adapter to factory default. The Adapter would then reset and reboot.

3.3 Default Setting

LAN IP Address: 192.168.123.1 (LAN)

WAN IP Address: Dynamic Address/DHCP (WAN)

Web Management Page

Login Name: root

Password: test

4 Application Example

4.1 SIP-to-SIP Calling/Answering

Applications:

Both parties are registered to SIP server with either fixed real IP or private IP under NAT router. The SIP-to-SIP calling works when both calling and answering parties are registered to SIP server with given registered phone numbers.

Configurations:

- Make sure “DHCP Client” is “ON” in the “Network / Network settings” pages,
- Select Active “ON” in the “SIP settings / Service Domain” pages,
- Enter the Register Name, Register Password, Proxy Server, and Outbound Proxy,
- Select “ON” in “NAT settings / STUN setting” page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LED indicator of “PHONE1” would be lighted.

Callings:

- Pick up the phone, and you should hear a dial tone. If you hear a busy tone, most likely the server registration failed. Please go to “SIP settings / Service Domain” page to review the settings
- If you're going to call user 1688, please press '1'+ '6'+ '8'+ '8'+ '#'. Note that '#' key will let the TA dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed.

4.2 SIP to Direct IP Calling

Applications:

The application is for the calling party with ADSL connection as in either Diagrams A or B. The calling party is registered to SIP server with either fixed real IP or private IP under NAT router. The answering party is with fixed real IP.

Configurations:

- Same as in Example 4.1
- Select “ON” in “NAT settings / STUN setting” page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LCD will show registered <phone number>.

Callings:

- Press Hand-Free key for speakerphone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.3 Direct IP to Direct IP Calling/Answering

Applications:

The applications are for ADSL connection without NAT router as in Diagram A. Both parties are with fixed real IP. The Direct IP calling works when both calling and answering parties are with known fixed IP. SIP server registrations are not required in this application.

Configurations:

- Select "Fixed IP", and bridge "ON" in the "Network / Network settings" page
- Enter the items of IP, Subnet Mask, Gateway IP
- Click the "Submit" button.

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.4 Direct IP to Direct IP Calling within NAT Router

Applications:

For the calling party in ADSL connection with NAT router as in Diagram B, this Direct IP calling can work when the answering parties are with fixed private IP addresses within the same VPN network, or with fixed real IP addresses.

Configurations:

- Select "Fixed IP", and bridge "ON" in the "Network / Network settings" page
- Enter the items of IP, Subnet Mask, Gateway IP
- Click the "Submit" button

Callings:

- Pick up the phone, and you should hear a dial tone
- Press 192*168*1*51# or 192*168*1*51 to call the party with the private IP address of 192.168.1.51. Press 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the called party to answer.

4.5 Calling through the FXO Port (LINE Port)

Applications:

User can make call through the FXO if the TA has the FXO Port (Line Port on the back),.

Configurations:

- Connect the FXO Port (LINE Port) to the PSTN Landline

Callings:

- Pick up the phone, and press '0' + '*', you should then hear a dial tone
- Press the landline phone number, and the call would be made.

4.6 3-Way Conference Call, Call Waiting, Call Hold

4.6.1 3-Way Conference Calling Application

This is for 3-way conference call among Parties A, B, and C. Three parties are registered to SIP server with either fixed real IP or private IP.

- Make a phone call from Party A to the first phone number Party B
- After the first call is established, press Flash key from Party A to hold the call, and Party A should hear a dial tone
- Press #512# + Party C's Phone Number + '#'. Party C would be connected.
- After the second call is established, press FLASH key from Party A to join in Party B and Party C together for three-way conference call

4.6.2 Call Waiting Application:

When a new call is coming while you are talking, you can push the FLASH key on your phone to switch to the new call. You can push the FLASH key to switch between the two calls.

4.6.3 Call Hold Application:

You may push the FLASH key to hold the current call for a while, then push FLASH key again to resume talking.

4.6.4 Call Transfer – Blind Transfer

You can transfer the current call to another user using the Call Transfer function

- Party B Call Party A. Party A and Party B are in conversation
- Party A press FLASH to hold the call. A should hear a dial tone
- Party A can then press #510# + Party C's Phone Number + '#', to make blind transfer to Party C.

4.6.5 Call Transfer – Attendant Transfer

- Party B Call Party A. Party A and Party B are in conversation
- Party A press FLASH to hold the call. A should hear a dial tone
- Party A can then press #511# + Party C's Phone Number + '#', Party C's phone would ring, and Party A can talk to Party C.
- When Party A hang up the call, Party B and Party C would in conversation.

4.6.6 Call Forward

You can forward the incoming call to a third-party

- Set up the Call Forward Number using the Web Configuration (Refer to 3.2.1) or Using the LCD Menu ("Phone Setting" -> "Call Forward")
- Press the Key FORWARD to enable the Call Forward

5 Phone Configuration

5.1 System Information

When you login the web page, you can see the TA current system information like firmware version, company... etc in this page.

Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

VOIP

- Home
- Phone Book »
- Phone Settings »
- Network »
- SIP Settings »
- Advanced »
- Administration »
- Save Change
- Reboot

System Information

This page illustrate the system related information.

Model Name: VoIP
 Firmware Version: 2.2.499 (712280)
 Codec Version: Wed Dec 05 10:39:05 2007.

Call Status

Protocol: SIP
 Status: Registered
 Phone In Use: 0 Phone In Use

WAN Status

MAC adress: 00:11:22:33:44:7a
 IP Address / Netmask: 192.168.1.181/255.255.255.0[DHCP]
 Gateway: 192.168.1.1
 DNS: 168.95.192.1/168.95.1.1

LAN Status

IP Address / Netmask: 192.168.123.1/255.255.255.0[Static]

5.2 Phone Book

In Phone Book contains Speed Dial Settings. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press "#".

Add New Phone

Position: (0~139)

Name:

Number or URL:

- **Phone:** Phone Book Entry Index. The total number of Phone Book Entry is 140
- **Name:** Phone Book Entry Name. Check Example for details
- **URL:** Phone Book Entry URL: the lin number or IP. Check example for details

Example

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1 ▼

Phone	Name	Number or URL	Select
0	301	192.168.1.2	<input type="checkbox"/>
1	206	1747643364	<input type="checkbox"/>
2	202	192.168.1.202:5062	<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected Delete All Reset

Example 1: Name: 301, URL: 301@192.168.1.2
If user dialed [301#], the phone would dial out [192.168.1.2] actually.

Example 2: Name: 206, URL: 17476433364
If user dialed [206#], the phone would dial out [17476433364] actually.

Example 3: Name: 202, URL: 192.168.1.202:5062
If user dialed [202#], the phone would dial out [192.168.1.2:5064] actually.

5.3 Phone Setting

In Phone setting contains Call Forward, SNTP Settings, Volume Settings, Block Setting, Auto Answer, Caller ID, Auto Dial Setting, Flash Time Setting and Call Waiting Setting functions.

5.3.1 Call Forward function

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

Forward Setting

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

All Forward: Off IP PSTN
Busy Forward: Off IP
No Answer Forward: Off IP PSTN

	Name	Number or URL
All Fwd No.:	<input type="text" value="Operator"/>	<input type="text" value="2001"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

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

All Forward:

All incoming call will forward to the number you choose. You can input the name and the phone number in URL/Number field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

Busy Forward:

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

No Answer Forward:

If you can not answer the phone, the incoming call will forward to the number you choose. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choose.

Forward Setting

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

All Forward: Off IP PSTN

Busy Forward: Off IP

No Answer Forward: Off IP PSTN

	Name	Number or URL
All Fwd No.:	<input type="text" value="Mobile"/>	<input type="text" value="998354409"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

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

5.3.1.1 Call Forward from VoIP to PSTN (Only supported by model with FXO port)

1. Select the case to forward (e.g. All Forward), Click option of "PSTN"
2. Enter the PSTN phone number in the "All Fwd No."->"URL/Number" field, and the "Name Field"
3. Click Submit to take effect.

5.3.1.2 Call Forward from PSTN to VoIP (Only supported by model with FXO port)

1. Select the case to forward (e.g. All Forward), Click option of "IP"
2. Enter the PSTN phone number in the "All Fwd No."->"URL/Number" field, and the "Name Field"
3. Click Submit to take effect.

5.3.2 SNTP setting function

User can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

5.3.3 Volume setting function

User can setup the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

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

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

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

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

Volume Setting

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

Handset Volume: (0~12)

PSTN-Out Volume: (0~12)

Handset Gain: (0~15)

PSTN-In Gain: (0~15)

5.3.4 DND function

You can setup the Block Setting to keep the phone silence. You can choose Always Block or Block a period.

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

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

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

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always: On Off

DND Period: On Off

From: : (hh:mm)

To: : (hh:mm)

5.3.5 Auto Answer function (Supported by FXO Model Only)

You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the TA can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the TA can let user to redial to IP Phone number. Auto Answer Counter is to set after the ring counts meet the number you set then the auto answer will enable.

Auto Answer:

- Off – Disable Auto Answer
- IP IN – Enable Auto Answer on incoming VoIP Call

- FXO IN – Enable Auto Answer on incoming FXO (landline) Call
- Both – Enable Auto Answer for both incoming VoIP and FXO (Landline) Calls
- Trunk Gateway – Use ATA as a Trunk Gateway, refer to 3.3.6 for details

Auto Answer Counter:

The number of ring count before the ATA automatically answer the call

PIN Code Enabled:

If enabled, auto-answered call will prompt user (a beep-beep sound) to enter a PIN. PIN has to be ended by a # key. (e.g., if the PIN is “123”, user has to press 1+2+3+#) If PIN is correct, a second dial tone will be heard to prompt entering the phone number.

PIN Code:

The code requires the user to enabled if PIN Code is enabled

Auto Answer

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

Auto Answer: Off IP IN FXO IN Both Trunk
Gateway

Auto Answer Counter: (0~8)

PIN Code Enabled: Off On

PIN Code:

5.3.6 Trunk Gateway (Supported by FXO Model Only)

You can use the ATA with FXO as a Trunk Gateway – Let an IP PBX or SIP Proxy to dial to PSTN through the ATA's FXO port.

1. Under Auto Answer page, set the Auto Answer option as “Trunk Gateway”
2. On an IP PBX, add a Outgoing SIP Trunk. On the SIP Outgoing Trunk, set the Gateway IP to the IP or domain name of the ATA. So, the ATA will dial out incoming SIP call number through the FXO port.

Here is a example of setting a Outgoing SIP Trunk with an IP PBX

Modify Outgoing rules via VOIP	
Description	FXO-Trunk
Authority	>= 0
IP/Domain Address	192.168.1.207:5060
UserName	
Password	
Dial Prefix	7 <small>TIPS:Dial Prefix can not be in FXO, or system will not work normally.</small>
Dial Strip Bit	1
Append Prefix	
User Prefix	
User Strip Bit	
Feature	Standard
Outside User Limit	<input type="radio"/> Enable <input checked="" type="radio"/> Disable

When User of the IP PBX dial 75698088, the IP PBX will route the call to the ATA (192.168.1.207), the ATA will make an outgoing call to number 5698088

5.3.7 Caller ID function

User can set the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.

Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:

Single Caller ID: Yes No

CID Without Time: Yes No

CID Type 2: Yes No

5.3.8 Flash Time Setting function

When you use the PSTN Phone and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function

Flash Time Setting

You could set the flash time in this page.

FXO Flash Time
Generate Flash Time: x 10 ms (9~120)

FXS Flash Time
Flash Signal Detect (MAX): x 10 ms (4~255)
Flash Signal Detect (MIN): x 10 ms (7~12)

5.3.9 Call Waiting setting function

You can Enable/Disable the Call Waiting function, when you are talking with someone, there is a new incoming call, you will hear the call waiting tone.

5.3.10 T.38 Fax setting function

Our Phone Adapter support T.38 fax function. To use Fax in T.38 mode,

1. Under "T.38 Setting" Page, select "T.38" as "On"
2. Submit and let the Adapter Reboot

T.38 Port Port: The network port that used for T.38 communication.

T.38 (FAX) Setting

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

T.38 (FAX): On Off

T.38 Port: (1024~65533)

5.3.11 Dial Plan

5.3.11.1 Dial Now

TA would dial the number out immediately if it matches the **Dial Now Rule**. Dial Now Rule has the same syntax as the Dial Plan above.

Dial Now Rule

Dialed Numbers Matching following Pattern would be dialed out automatically

Rule:

Rule Syntax

0-9	Number Digit	e.g. 911 Dialed numbers 911
'+' or ' '	"Or" Separator	e.g. 911+18503 Dialed number 911 or 18503
[]	Range	e.g. 00[2-4] Dialed number 002, 003 or 004
x	Any Number	e.g. 86xx Any 4 digits dialed number start with 86

5.3.11.2 Add or Replace Prefix

You can use the Dial Plan to add prefix to phone number automatically before dialing out.

Append/Replace Rule

Dialed Numbers Matching following Prefix or Pattern would be adjusted

No	Rule Option	Append Prefix	Matching Rule
1	<input type="radio"/> Replace <input checked="" type="radio"/> Append	<input type="text" value="002"/>	<input type="text" value="8613 8662"/>
2	<input checked="" type="radio"/> Replace <input type="radio"/> Append	<input type="text" value="006"/>	<input type="text" value="00[2-9]"/>
3	<input type="radio"/> Replace <input checked="" type="radio"/> Append	<input type="text" value="009"/>	<input type="text" value="12"/>
4	<input type="radio"/> Replace <input checked="" type="radio"/> Append	<input type="text" value="007"/>	<input type="text" value="5xxx 35xx 21xx"/>

There are 4 replace rules, each the dialed number matches the rule, a prefix would be added (or replace) to the dialed number.

Rule Syntax

0-9	Number Digit	e.g. 86 Dialed numbers start with 86
'+' or ' '	"Or" Separator	e.g. 002+005 Dialed number start with 002 or 005
[]	Range	e.g. 00[2-4] Dialed number start with 002, 003 or 004
x	Any Number	e.g. 86xx Any 4 digits dialed number start with 86

Example 1: Append, Append Prefix: 002, Rule: 8613|8662

- 1: If what we dialed begins with 8613, number "002" when is added to the beginning; so the actual number dialed would be [002]8613|xxx].
- 2: If what we dialed begins with 8662, number "002" when is added to the beginning; so the actual number dialed would be [002]8663|xxx].

Example 2: Replace, Append Prefix: 006, Rule: 00[2-9]

- 1: If what we dialed begins with 002 (e.g. 0021386), the number "002" would be replaced with "006"; so the actual number dialed would be like 0061386.
- 2: If what we dialed begins with 003, the number "003" would be replaced with "006"; so the actual number dialed would be [006]xxx].

Example 3: Append, Add Prefix: 009, Rule: 12

1: If what we dialed begins with 12, the number "12" would be prefixed by number "009"; so the actual number dialed would be [009]12|xxx].

Example 4: Append, Add Prefix 007, Rule: 5xxx|35xx|21xx

1: If what we dialed begins with 5, and followed any 3 numbers, the number would be prefixed by number "007"; so the actual number dialed would be [007]5xx].

2: If what we dialed 534, the number 5 is only followed by 2 numbers, it does not match with the dialing rule; so the actual number dialed is still [534].

3: If what we dialed begins with 35, and followed any 2 numbers, the number would be prefixed by number "007"; so the actual number dialed would be [007]35xx].

4: If what we dialed 358822, the number 5 is only followed by 2 numbers, so it does not match with the dialing rule; so the actual number dialed is still [358822].

5.3.11.3 Auto Dial Time

TA would auto dialed out the numbers if no more key is pressed for a few seconds – Auto Dial Timeout. The Auto Dial Time can be set in this Dial Plan page, the default timeout is 5 seconds

Other Dial Options:

Auto Dial Time: (3~9 sec)

Use # as send key: Yes No

Use * for IP dialing: Yes No

5.3.12 Dial Plan – Call Routing (Supported by FXO Model Only)

Routing Rule

Dialed Numbers Matching following Prefix or Pattern would be specially routed

Routing to : IP FXO Disable

Routing rule :

If the dialed number matched the Routing rule, the call would be dial to the route specific in the "Routing to" field.

For example.

Routing to: FXO
Routing rule: [2-4]xxxxxx|911

"[2-4]xxxxxx" mean if any dialed number begins with 2 to 4, and followed by 6 digits would be dialed to FXO(PSTN). E..g 3234567 would be dialed through the FXO(PSTN)

"911" mean if that if user dialed 911, the number would be routed through FXO port.

5.4 Network

In Network you can check the Network status, configure the Network Settings and DDNS settings.

5.4.1 Network Status

You can check the current Network setting in this page.

Network Status

This page shows current status of network interfaces of the system.

System Up Time: 0 day(s) 0 hour(s) 39 minute(s)
 Network Link Up Time: 0 day(s) 0 hour(s) 39 minute(s)
 NAT Type: Port restricted cone

WAN Interface	
Type:	DHCP Client
IP:	192.168.1.181
Mask:	255.255.255.0
Gateway:	192.168.1.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

LAN Interface	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

5.4.2 WAN Settings

You can configure the TA Network setting in this page.

LAN Mode: Bridge NAT

WAN Setting	
IP Type:	<input type="radio"/> Fixed IP <input checked="" type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	<input type="text" value="192.168.1.181"/>
Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="192.168.1.1"/>
DNS Server1:	<input type="text" value="168.95.192.1"/>
DNS Server2:	<input type="text" value="168.95.1.1"/>
MAC:	<input type="text" value="00112233447a"/>
Host Name:	<input type="text" value="VOIP_TA1S10"/>

5.4.2.1 Mode

There are two modes for the WAN port: Bridge or NAT.

Bridge mode: The two WAN and LAN Ethernet ports will be bridged and transparent.

NAT mode: The embedded NAT will be enabled.

The IP type for IP Phone is default at Fixed IP (192.168.1.100). You may select a proper IP type for your network requirements.

5.4.2.2 NAT Settings

To enable embedded NAT, you must set mode of WAN port to "NAT". This embedded NAT is useful for ADSL users without NAT router, and it separates the WAN port from the LAN port to perform router IP address translation.

For the WAN port, please select a proper IP type:

DHCP Client(default): Automatically get an assigned IP from the your router. Please select this if there is a router in your network.

PPPoE: Most commonly way to connect to your Internet Service Provider through ADSL. Enter the given username and password from the Internet Service Provider.

Fixed IP: Assign a Fixed IP to the Adapter.

MAC settings for LAN and WAN have been preprogrammed and must be different from each other. Please leave it default unless necessary.

After you finished the setting, please click the "Submit" button.

5.4.3 LAN Settings

The screenshot shows a web-based configuration interface. The top section is titled "LAN Setting" and contains three input fields: "IP:" with the value "192.168.123.1", "Mask:" with "255.255.255.0", and "MAC:" with "001122324455". The bottom section is titled "DHCP Server" and contains a "DHCP Server:" label with "On" selected (radio button) and "Off" unselected. Below this are three input fields: "Start IP:" with "150", "End IP:" with "200", and "Lease Time:" with "1" and "0" in separate boxes, followed by "(dd:hh)".

5.4.3.1 LAN Setting

IP: The IP Address of the LAN Port. Default is 192.168.123.1

5.4.3.2 DHCP Server

There is built-in DHCP Server in the Phone Adapter. The default is enabled.

Start IP/End IP: The range of IP to be assigned. E.g. Assign IP from 192.168.1.150 to 192.168.1.200

Lease Time: The lease time of each DHCP assignment

5.4.4 DDNS Setting

You can configure the DDNS setting in this page. You need to have the DDNS account and input the information properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.

DDNS:	<input checked="" type="radio"/> On <input type="radio"/> Off
Host Name:	<input type="text" value="voip.dyndns.org"/>
User Name:	<input type="text" value="voip"/>
Password:	<input type="password" value="••••"/>
E-mail Address:	<input type="text"/>
DDNS Server:	<input type="text"/>
DDNS Server List:	<input type="text" value="ddns.com.cn"/>
Type:	<input type="text" value="dyndns"/>
Wild Card:	<input type="text" value="on"/>
BACKMX:	<input type="radio"/> On <input checked="" type="radio"/> Off
Off Line:	<input type="radio"/> On <input checked="" type="radio"/> Off

5.4.5 VLAN Setting

Our Phone Adapter support VLAN. Note that building a VLAN network requires a switch support VLAN. VLAN can be used to separate VOIP traffic from other DATA traffic. Then, user can configure the VLAN switch to handle the VOIP VLAN tag in higher priority. This is one of the ways to guarantee Quality-of-Service within a private network.

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets:	<input checked="" type="radio"/> On <input type="radio"/> Off
VID (802.1Q/TAG):	<input type="text" value="136"/> (2 ~ 4094)
User Priority (802.1P):	<input type="text" value="3"/> (0 ~ 7)
CFI:	<input type="text" value="1"/> (0 ~ 1)

5.4.6 PPTP Setting

You can connect the Phone Adapter to a VPN (PPTP) Server. Using VPN, user can connect to a SIP Server (SIP Proxy or IP PBX) inside a private network, or making a secure phone call through

the Adapter.

PPTP Server: VPN Server to connect.

PPTP Username/Password: Username and Password for the VPN account of the VPN Server

PPTP Settings

You could set the PPTP server in this page.

PPTP: On Off

PPTP Server:

PPTP Username:

PPTP Password:

5.4.7 STUN Setting

You can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your TA working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

5.5 SIP Settings

You can setup the Service Domain, Port Settings, Codec Settings, RTP Setting, RPort Setting and Other Settings for SIP Proxy Server registrations in this page.

5.5.1 Service Domain

You may register up to three SIP accounts in the IP Phone. You can call your friends via firstly enabled SIP account and receive the phone calls from all the three SIP accounts. It supports 3 services; allow user register on different service providers. Click "Active" ON to enable the Service Domain, and then input the following fields:

Realm (1 ~ 3)

- Display Name:** Name you want to display.
- User Name:** User Name given by your ITSP.
- Register Name:** Register Name given by your ITSP.
- Register Password:** Register Password given by your ITSP.
- Domain Server:** Domain Server given by your ITSP.
- Proxy Server:** Proxy Server given by your ITSP.
- Outbound Proxy:** Outbound Proxy of ITSP. If not provided or it is the same as Proxy Server, you may skip this.
- MWI:** Whether you would like to subscribe for Voicemail notification.

When it shows "Registered" in the Register Status, it indicates a successful registration to the

ITSP, and the "PHONE" LED will start flashing. The IP Phone is then ready for VoIP call.

If you have more than one SIP account, please follow the steps to register to other ITSPs. After you finished the setting, please click the "Submit" button.

SIP Accounts Settings

You could set information of service domains in this page.

Realm 1 (Default)

Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	<input type="text" value="Operator"/>
User Name:	<input type="text" value="2007"/>
Register Name:	<input type="text" value="2007"/>
Register Password:	<input type="password" value="....."/>
Domain Server:	<input type="text" value="sip.myoffice.com"/>
Proxy Server:	<input type="text" value="sip.myoffice.com"/>
Outbound Proxy:	<input type="text"/>
Subscribe for MWI:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	Registered

Switching Default SIP Account (Domain)

If you have registered more than one SIP Accounts, you may want to switch to default SIP account for outgoing call.

1. Please pick up the handset, and
 - Press "1*#" - Make first SIP account as default.
 - Press "2*#" - Make second SIP account as default
 - Press "3*#" - Make third SIP account as default
2. Put down the handset, and the new default account would be used for outgoing call.

5.5.2 Port Settings

You can setup the SIP and RTP port number in this page.

SIP Port: Network Port for SIP communication. Default is 5060

RTP Port: Network Port for RTP data communication. Default is 60000

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

5.5.3 Codec Settings

You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

Codec Priority	
Codec Priority 1:	G.711 u-law
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.723
Codec Priority 4:	G.729
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40
Codec Priority 9:	GSM

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

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

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

5.5.4 Codec ID Setting

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

Codec ID Setting

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

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

5.5.5 DTMF Setting

User can setup the InBand DTMF, 2833 Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please follows your ISP information. When you finished the setting, please click the Submit button.

DTMF Setting: Configure how the Phone Adapter sends out the DTMF signal.

- 2833: Send in format compliant to RFC2833
- Inband DTMF: Send the DTMF as Inband DTMF
- Send DTMF SIP Info: Send the DTMF as SIP Info

5.5.6 RPort Function:

You can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

5.5.7 Other Settings

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

Other Settings

You could set other settings in this page.

Hold by RFC: On Off

Voice QoS (Diff-Serv): (0~63)

SIP QoS (Diff-Serv): (0~63)

SIP Expire Time: (15~86400 sec)

Use DNS SRV: On Off

Send Keep Alives Packet: On Off

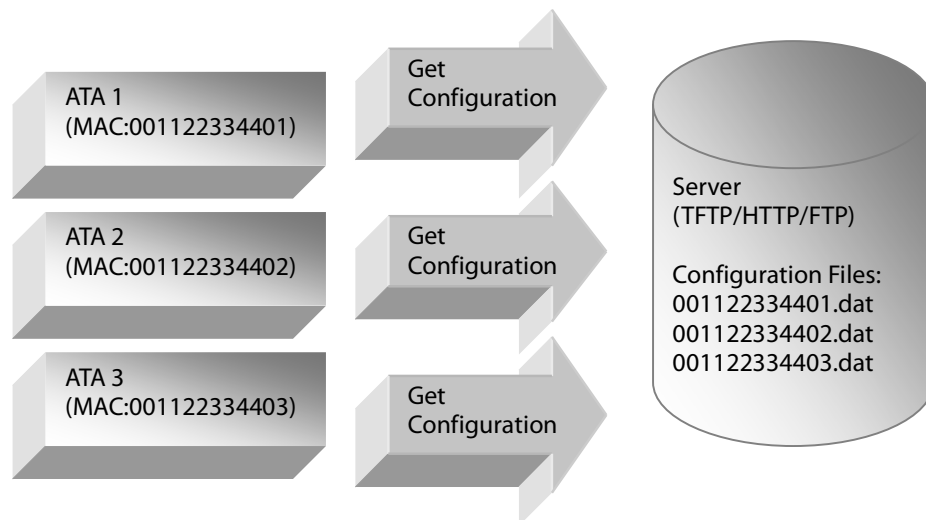
Keep Alives Period: (15~250 sec)

Jitter Buffer: (0~250 packets)

5.6 Advanced

5.6.1 Auto Configuration Setting

You can setup the Auto Configuration so that the ATA would automatically download the configure file from server. This function helps server provider for their deployment.



For example, you would like to setup the auto configuration using an FTP Server
FTP Server Domain name: <ftp.myserver.com>
FTP Username: user
FTP File Path: /file/firmware

1. Setup the configuration files, and put them on your FTP Server, under the "File Path".
(Please refer to Auto Configuration and StandardCfg.dat document for details)
2. Under the Auto Configuration Setting of ATA:
Set the following:

Auto Configuration Setting

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

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:

HTTP Server: Exp. 60.35.187.30

HTTP File Path: Exp. /download/

FTP Server: Exp. 60.35.17.1

FTP Username:

FTP Password:

FTP File Path: Exp. /file/load

3. The ATA would automatically reboot, and get the updated configuration file from server.

5.6.2 FXS/FXO Setting:

You can setup the FXS/FXO setting in this page. When you are using different country's PSTN Phone or connect to different country's PSTN Line, you have to set the country's setting to meet the requirement. When you finished the setting, please click the Submit button.

FXO & FXS Setting

You could select the FXO & FXS impedance of the analog telephone by different country in this page.

FXO Port:

FXS Port:

FXO Silence Timeout : (1~250 minutes)

5.6.3 Advanced Setting

Advanced Setting

You could change advanced setting in this page.

ICMP Not Echo: Yes No

Management from WAN: Yes No

Web Server port:

PPPoE retry period: Seconds

Send Anonymous CID: Yes No

Anonymous Call Rejection: Yes No

Switch to PSTN when No SIP registration: Yes No

Billing Signal:

CPC Delay: (2~5 Seconds)

CPC Duration: x 10 ms (0~120)

Send Flash event:

SIP Encrypt:

Encryption Key:

You

- **ICMP Not Echo:** Enable/Disable the ICMP Echo in this page. This function can disable echo when someone ping this device, it can avoid hacker try to attack the device. When you finished the setting, please click the Submit button.
- **Management from WAN:** Enable/Disable web management page access from WAN

port

- **Web Server Port:** Configure the Web management server port. Default is 9999
- **PPPoE Retry period:** Configure the PPPoE Retry period
- **Send Anonymous CID:** Enable/Disable sending anonymous Caller ID.
- **Anonymous Call Rejection:** Enable/Disable Anonymous Call Rejection. Enable this would automatically reject anonymous call.
- **Billing Signal:** Configure how the adapter generates the Billing Signal.
- **Switch to PSTN when No SIP registration:** Adapter would automatically switch to PSTN/FXO mode if there is no SIP registration
- **Send Flash event:** Configure how the adapter send the Flash Event

5.7 System Auth

5.7.1 In System Authority

You can change your login name and password. . When you finished the setting, please click the Submit button.

5.8 Save Change

5.8.1 In Save Change

You can save the changes you have done. If you want to use new setting in the TA, You have to click the Save button. After you click the Save button, the TA will automatically restart and the

5.9 Update

5.9.1 In Update

You can update the TA's firmware to the new one or do the factory reset to let the TA back to default setting.

5.9.2 In New Firmware

You can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:

- Select the firmware code type, Risc or DSP code.
- Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
- Select the correct file you want to download to the TA then click the Update button.

Update Firmware

You could update the newest firmware. [\[Advanced\]](#)

Method: Local PC TFTP

Local PC

Code Type: ▾

File Location:

TFTP

TFTP Server:

5.9.3 In Default Setting

You can restore the TA to factory default in this page. You can just click the Restore button, then the TA will restore to default and automatically restart again.

5.10 Reboot

5.10.1 Reboot

You can restart the TA. If you want to restart the TA, you can just click the reboot button, then the TA will automatically.

6 Troubleshooting

6.1 The WAN LED does not light

Make sure the WAN port of Phone Adapter is connected properly to your ADSL Modem or Router. Power Reset and check again.

6.2 Do not hear DIAL TONE?

No dial tone happens when there is Internet connection problem or when the Service registration failed.

- Check if the Account information is set up correctly. Goto "SIP Settings"->"Service Domain" of the Web Management Interface. Make sure all the information is entered properly, especially is the password. Press Submit and Reboot the IP Phone
- Make sure the Phone Adapter can connect to the Internet. Check if your ADSL Modem is working properly.
- Call your VoIP Service provider, to check if there is any problem with their server.

6.3 How to Reset the Phone Adapter to Factory Default?

- Refer to **Ch. 3.2**

6.4 How to switch to another SIP account for outgoing call?

If you have registered more than one SIP Accounts, you may want to switch to default SIP account for outgoing call.

1. Please pick up the handset, and

- Press "1*#" - Make first SIP account as default.
- Press "2*#" - Make second SIP account as default
- Press "3*#" - Make third SIP account as default

2. Put down the handset, and the new default account would be used for outgoing call.