

**5517**

**1 FXS / 1 FXO**

**VoIP Telephone Adaptor**

**User Manual**

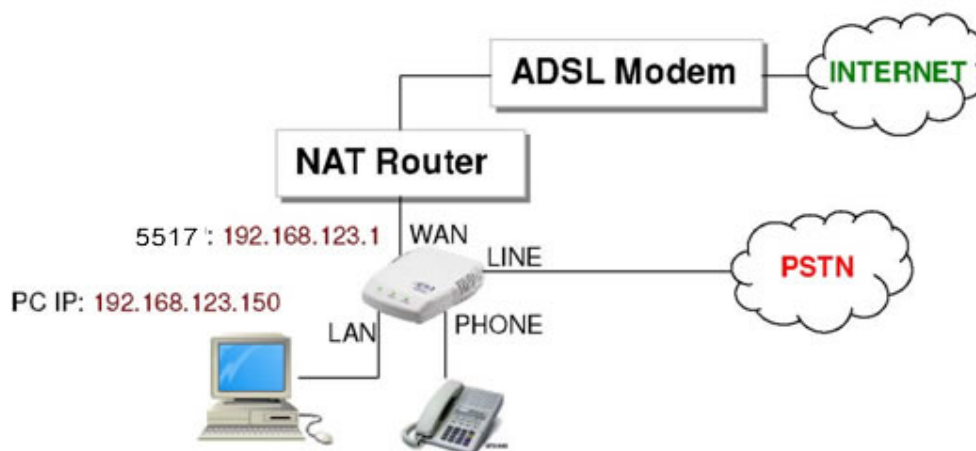
**V1.1u**

## Quick Guide

### Step 1: Broadband (ADSL/Cable Modem) Connections for ANSEL 5517

- A. Connect ANSEL 5517 WAN port to ADSL NAT Router as the following connection.
- B. Connect ANSEL 5517 LAN port to Notebook PC LAN port using a Category 5 LAN cable.
- C. Connect ANSEL 5517 RJ11 PHONE port to a Telephone Set.
- D. Connect ANSEL 5517 RJ11 LINE port to a PSTN Telephone Line.
- E. Connect 12VDC Power Adaptor. After power on, the POWER LED will be **Green** ON.
- F. The PHONE LED will be **Green** flashing for a successful SIP registration. Pick up the phone, and it will be **Green** ON indicating VoIP mode.
- G. Press **0\*** key, and the PHONE LED will be **Yellow** ON indicating PSTN mode.
- H. Press **#121#** and **#120#** from the phone to listen to IVR and to check the DHCP status and the IP address (e.g. 192.168.123.1) for ANSEL 5517. After the IP announcement, please hang up.

Figure A. ADSL Connections with NAT Router for 5517



### Step 2: Settings for ANSEL 5517 from PC Web Browser

- A. This is an example for FWD ITSP provider using ADSL connection with NAT router as in Figure
- A. Both parties sign up to FWD SIP server <http://www.freeworlddialup.com> with individual registered FWD phone numbers and passwords.

B. Pick up the phone and press **#120#** from the phone to listen to IVR and get the IP address (e.g. 192.168.123.1) for ANSEL 5517.

C. Enter the IP address from PC Web browser for configuration settings.

**Example:** Enter <http://192.168.123.1> from IE Web browser to display login page.

D. Enter the user name and password into the blank field. The default settings are Username: **root** Password: **test** . Click the “**Login**” button to enter for configurations.

The image shows a web browser window displaying the 'Login VoIP Gateway' page. The page has a blue header with the text 'Login VoIP Gateway'. Below the header, there are two input fields: 'Username' and 'Password'. Below these fields are two buttons: 'Login' and 'Clear'. At the bottom of the form, there is a checkbox labeled 'Remember last login'.

E. You need to set up the following web configurations: **Network Settings, SIP Settings, NAT Settings/STUN Settings**. Remember to submit, save and reboot for new configurations.

F. The PHONE LED will be **Green** flashing showing a successful registration in the SIP server. For further detail configurations, please refer to the VoIP applications chapter.

### Step 3: Making Point-To-Point SIP Calls

A. Pick up the phone in VoIP mode, and you should hear a dial tone.

B. Press **123456#** to call the party with the number **123456** registered in the SIP server. Note **#** is used to send out the call immediately. In a moment, you should hear the ring back tone, and wait for the called party to answer. For more applications, please refer to the user manual.

### Step 4: Making Public Switching Telephone Calls

- A. Pick up the phone.
- B. Press **0\*** key for PSTN mode, and the PHONE LED will be **Yellow** ON, and you should hear a dial tone. Now you may dial any public phone number.

**Note: Difficulties in configuring ANSEL 5517? Please refer to the last chapter for trouble shootings.**

## **TABLE OF CONTENTS**

<b>1. Introductions</b>	<b>1</b>
<b>2. Features</b>	<b>1</b>
<b>3. Standard Compliances</b>	<b>2</b>
<b>4. Packing Contents</b>	<b>2</b>
<b>5. LED Indicators</b>	<b>2</b>
<b>6. Installations &amp; SIP Configurations</b>	<b>3</b>
<b>7. Default Reset by Telephone</b>	<b>3</b>
<b>8. Configurations by Web Browser</b>	<b>4</b>
<b>9. Configurations by Telephone &amp; IVR</b>	<b>38</b>
<b>10. VoIP Applications Examples</b>	<b>40</b>
<b>PSTN Calling/Answering</b>	<b>◆ 41</b>
<b>SIP-to-SIP Calling/Answering</b>	<b>◆ 41</b>
<b>SIP to PSTN Calling</b>	<b>◆ 42</b>
<b>SIP to Direct IP Calling</b>	<b>◆ 43</b>
<b>PSTN to SIP Calling</b>	<b>◆ 43</b>
<b>Direct IP to Direct IP Calling/Answering</b>	<b>◆ 44</b>
<b>Direct IP to Direct IP Calling within NAT</b>	<b>◆ 44</b>
<b>Direct IP to PSTN Calling</b>	<b>◆ 45</b>
<b>PSTN to Direct IP Calling</b>	<b>◆ 45</b>
<b>3-Way Conference Call, Transfer, Waiting, Hold</b>	<b>46</b>
<b>SIP-to-SIP Calling for FWD</b>	<b>◆ 47</b>
<b>11. Trouble Shooting for Web Configurations</b>	<b>52</b>

# 1. Introductions

The ANSEL 5517 is a 1-port FXS / 1-port FXO Telephone Adaptor (TA) with SIP Protocols for Voice over IP (VoIP) applications. Connecting to the Internet and the PSTN line with an analog telephone set, the ANSEL 5517 can connect a VoIP call over the Internet with extension to the public switched telephone line. ANSEL 5517 provides one WAN port for Internet connections, one LAN port for Notebook PC, and two RJ11 connectors for Phone (FXS) and PSTN (FXO). With an embedded NAT/DHCP server, ANSEL 5517 can be easily configured for different network diagrams by PC Web browser and telephone set. This is very suitable for ITSP (Internet Telephony Service Providers) and SOHO users to make 2-stage VoIP with PSTN extension calls.

Note that ANSEL 5517 requires an IP address, a subnet mask, and its gateway Router IP address for its own use to connect to Internet. These three are available from your Internet service provider. ANSEL 5517 may enable PPPoE or DHCP features to automatically get an assigned dynamic IP from the ITSP. Please refer to Section 8 Configurations by Web browser for detailed information.

# 2. Features

The ANSEL 5517 VoIP TA is equipped with two RJ11 connectors and two RJ45 connectors and is featuring as the following

- ❖ Three LED Indicators for ANSEL 5517: POWER, PHONE, LAN
- ❖ RJ45 x 2 for WAN and LAN ports + RJ11 x 2 for FXS and FXO ports
- ❖ Configurations by Web Browser and Telephone
- ❖ Embedded NAT/DHCP Server
- ❖ PPPoE/DHCP Client for Dynamic IP plus NAT, DNS, and DDNS Clients
- ❖ Support STUN server for NAT Traversal
- ❖ Support registrations for up to 3 SIP Servers.
- ❖ Hot Line Mode
- ❖ Dial Plan Settings
- ❖ T.38 FAX over IP
- ❖ Interactive Voice Recording (IVR) for telephone IP status
- ❖ Phone Book, Call Forward/Waiting, Call Transfer/Hold, and 3-Way Conference Calls
- ❖ Auto Configurations by TFTP, HTTP, or FTP server
- ❖ Remote Firmware Upgraded with HTTP or TFTP server by Web PC
- ❖ Direct IP/URL Dial without SIP Proxy or Dial number via SIP server
- ❖ Telephone features: Volume Adjustment, Phone book, and Flash
- ❖ Out-Band DTMF (RFC 2833) / In-Band DTMF / Send DTMF SIP Info

### 3. Standard Compliances

The ANSEL 5517 VoIP TA supports for the following standards

**VoIP Protocol:** IETF RFC3261 and RFC 2543 for SIP

**SIP Authentication:** IETF RFC2069 and RFC 2617 for MD5

**Speech Codec:** ITU-T G.711, G.723, G.729A/B, VAD and CNG

**Echo Cancellation:** ITU-T G.165/168

### 4. Packing Contents

Inside the package you should find:

- (1) One ANSEL 5517 SIP TA
- (2) One AC to 12VDC/1A Power Adaptor
- (3) One User Manual CD

Please check if the packing is damaged or any component is missing. If so, please contact your distributor.

### 5. LED Indicators

On the front panel of ANSEL 5517, there are three LED indicators as the following:

**POWER:** “On” indicates the power is normal

**PHONE:** “**Green On**” indicates the Phone is in use (off-hook) for VoIP call. “**Red On**” indicates the PSTN Line is in use or ringing from incoming call. “**Yellow On**” indicates the Phone and the PSTN Line are in use for PSTN call. “**Green Flashing**” indicates successful registration at SIP server. “**Yellow Flashing**” indicates PSTN incoming call with successful SIP registration.

**LAN:** “On” indicates the Ethernet port is in Connection.  
“Flashing” indicates the data activity of Ethernet port.

### 6. Installations & SIP Configurations

- 1 Connect ANSEL 5517 RJ45 WAN port to NAT Router using a Category 5 LAN cable.
- 2 Connect ANSEL 5517 RJ45 LAN port to Notebook PC using a Category 5 LAN cable.
- 3 Connect ANSEL 5517 RJ11 PHONE port to a Plain Old Telephone Set (POTS).
- 4 Connect ANSEL 5517 RJ11 LINE port to a Public Switched Telephone Network (PSTN) line.
- 5 Connect the power adaptor (12VDC) to power on ANSEL 5517, and the POWER LED will be ON.
- 6 The PHONE LED indicators will be OFF for about 5 seconds and start flashing for 5 times, and remain OFF for VoIP configurations. The LAN LED will be ON when RJ45 Ethernet ports are connected. Pick up the phone, and the PHONE LED will be **Green** ON indicating VoIP mode. If you hear a busy tone, please check if the

- WAN port is connected.
- 7 Press **0\*** key to PSTN line. The PHONE LED will become **Yellow** ON and you should hear a PSTN dial tone. If not, please check if the PSTN Line is connected.
  - 8 Press **#120#** to check the assigned IP address for the ANSEL 5517. The default IP address is **192.168.123.1**. Please refer to Chapter 8, and you may enter this IP address in IE Web browser for web configurations.
  - 9 Register ANSEL 5517 into your SIP server, and refer to Chapter 10 for VoIP applications examples of SIP registrations.
  - 10 If ANSEL 5517 has successfully registered in the SIP server, the PHONE LED will start **Green** flashing.
  - 11 Pick up the phone, and you should hear a dial tone. Press **123456#** to call the party with the number **123456** registered in the SIP server. Note that **#** 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 answer.

## 7. Default Reset by Telephone

ANSEL 5517 provides an easy way to reset to factory defaults by using Telephone.

Pick up the phone and press **#198#** to reset back to factory defaults, and the ANSEL 5517 will enter into POWER ON cycle. The PHONE LED indicators will be OFF for about 4 seconds and start flashing for 5 times. The POWER LED then will be lit constantly, and the PHONE LED will be OFF.

## 8. Configurations by Web Browser

You may enter the IP address from PC Web browser to configure ANSEL 5517.

For example, enter <http://192.168.123.1> from IE web browser to display login page as follows.

- 8.1. Please enter the default IP address <http://192.168.123.1> from PC Web browser. The following Web page shall be displayed on PC. If you have difficulties accessing the Web page from the PC Web browser, the subnet IP of PC might be different from **192.168.123.xxx**. In this case, please refer to Chapter 11 for trouble shooting.



Login VoIP Gateway

Username

Password

☐ Remember last login



- 8.2. Please enter the username and password into the blank field. The default settings are: Username: **root** Password: **test**
- 8.3. Click the “**Login**” button will enter the management information page for system setup. Note that whenever you change the setting in each Web page, please remember to click the “Submit” button in the page, and click the “Save” button to save into the non-volatile memory and click the “Reboot” button to activate the new settings.

## System Information

8.4. You will see the system information such as firmware version, Codec, etc in this page.

8.5. You may click the button list at the left hand side to configure the ANSEL 5517.

**Phone Book**

**Call Settings**

**Network**

**SIP Settings**

**Auto Config**

**User Password**

**Save Change**

**Update**

**Reboot**

## System Information

This page illustrate the system related information.

Model Name:	VoIP FXS/FXO TA
Firmware Version:	Mon Mar 26 20:30:16 2007 (01uc)
Codec Version:	Tue Oct 31 16:53:19 2006.

## Phone Book

### Phone Book Settings

- 8.6. You may add/delete **Name (in numeric only)** up to maximum 140 entries in Phone book list.
- 8.7. To add a phone name, you need to enter the position, the name, and the phone URL. When you finished a new phone list, just click the “Add Phone” button.
- 8.8. To delete a phone name, please select the phone name then click “Delete

Selected” button.

8.9. To delete all phone names, please click “Delete All” button.

8.10. After dialing a phone number, the TA will first match with the **Name** in the phone book. If matched, the TA will send out the corresponding URL. If not, the dialed phone number will be sent out.

8.11. **Example 1:** Name: **101**, URL: **192.168.1.100**

Press **101#** on telephone, and the phone at **192.168.1.100** will start ringing.

8.12. **Example 2:** Name: **201**, URL: **14081234567**

Press **201#** on telephone, and the TA will call the registered number **14081234567**.

8.13. **Example 3:** Name: **301**, URL: **192.168.1.100:5062**

Press **301#** on telephone, and the TA will call at **192.168.1.100** with port **5062**.

8.14. **Example 4:** No Name: **401** in phone book.

Press **401#** on telephone, and the TA will call the registered number **401**.

### Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1

Phone	Name	URL	Select
0			<input type="checkbox"/>
1	101	192.168.1.100	<input type="checkbox"/>
2	102	james@sip.server.com	<input type="checkbox"/>
3	103	612345@fwd.pulver.com	<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"/>

- Phone Book
- Call Settings
- Network
- SIP Settings
- Auto Config
- User Password
- Save Change
- Update
- Reboot

## Call Settings

8.15. The subpages are as follows; Call Forward, SNTP, Volume, DND Settings, Auto Answer, Caller ID, Dial Plan, Flash Time (or hook switch), Call Waiting,

T.38 FAX over IP, and Hot Line settings.

### Call Forward function:

8.16. You can select the forward mode and enter the forward URL.

**All Forward:** All incoming call will forward to the URL you choose.

**Busy Forward:** The incoming call will forward to the URL when the callee is busy.

**No Answer Forward:** The incoming call will forward to the URL when no answer.

8.17. You need to set the Time Out ring which will initiate No-Answer forwarding to the number you choose. When you finished the setting, please click the “Submit” button.

## Forward Setting

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

Phone Book	
Call	Call Forward
Netw	SNTP Settings
SIP s	Volume Settings
Auto	DND Settings
User	Auto Answer
Save	Caller ID
Upda	Dial Plan Settings
Rebo	Flash Time Settings
	Call Waiting
	T.38(FAX) Settings
	Hot line Settings

All Forward:	<input type="radio"/> Off	<input checked="" type="radio"/> IP	<input type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text" value="61234"/>	<input type="text" value="61234"/>
Busy Fwd No.:	<input type="text" value="Jane"/>	<input type="text" value="192.168.62.100"/>
No Answer Fwd No.:	<input type="text" value="James"/>	<input type="text" value="james@sipserver.com"/>

No Answer Fwd Time Out:	<input type="text" value="3"/>	(2~8 Ring)
-------------------------	--------------------------------	------------

### SNTP Setting:

8.18. You can setup the primary and second SNTP Server IP Address, to get the date/time information. You may also set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the “Submit” button.

# SNTP Settings

You could set the SNTP servers in this page.

Phone Book	
Call	Call Forward
Netw	SNTP Settings
SIP S	Volume Settings
Auto	DND Settings
User	Auto Answer
Save	Caller ID
Upda	Dial Plan Settings
Rebo	Call Waiting

SNTP: ☒ On ☐ Off

Primary Server:

Secondary Server:

Time Zone: GMT    (hh:mm)

Sync. Time:    (dd:hh:mm)

## Volume Setting:

8.19. You can setup the Handset Volume/Gain, PSTN-Out Volume, and PSTN-In Gain in this page. Handset Volume is to set the volume hearing from the handset. PSTN-Out Volume is to set the PSTN volume for you to 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.

Phone Book	
Call	Call Forward
Netw	SNTP Settings
SIP S	Volume Settings
Auto	DND Settings
User	Auto Answer
Save	Caller ID
Upda	Dial Plan Settings

Handset Volume:  (0~12)

PSTN-Out Volume:  (0~12)

Handset Gain:  (0~15)

PSTN-In Gain:  (0~15)

## DND Setting:

8.20. You can configure the DND (Do Not Disturb) setting to keep the phone silence. You can choose either DND Always or a DND period.

8.21. DND Always: All incoming call will be blocked until this feature is disabled.

8.22. DND Period: Set a time period and the phone will be blocked during the time

period. If the time in “From” is greater than that in “To” time, the DND time will be from Day 1 to Day 2.

8.23. After you finished the setting, please click the “Submit” button.

Phone Book

Call Forward

SNTP Settings

Volume Settings

DND Settings

Auto Answer

Caller ID

Dial Plan Settings

Flash Time Settings

Call Waiting

T.38(FAX) Settings

Hot line Settings

## 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)

Submit

Reset

### Auto Answer Setting:

8.24. You may enable the Auto Answer function to answer the incoming call by ANSEL 5517 FXO port. When the ring count exceeds the number set in Auto Answer Counter, the FXO port will auto answer and allow for extension calls from PSTN to VoIP and vice versa. If the incoming call is from PSTN, then ANSEL 5517 FXO port will answer with a short beep tone and allow caller to redial to VoIP number. If the incoming call is from the Internet, then FXO port will answer with a PSTN dial tone and allow caller to redial to PSTN phone number. PIN Code is used to prevent from call piracy. The caller needs to enter the right PIN code and the postfix “#” to get the PSTN dial tone. Incorrect PIN Code will result in call disconnect. The Auto Answer is disabled at default.

## Auto Answer

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

Phone Book	
Call f	Call Forward
Netw	SNTP Settings
SIP s	Volume Settings
Auto	DND Settings
User	Auto Answer
Save	Caller ID
Upda	Dial Plan Settings
Rebo	Flash Time Settings
	Call Waiting
	T.38(FAX) Settings
	Hot line Settings

Auto Answer:	<input type="radio"/> On <input checked="" type="radio"/> Off
Auto Answer Counter:	<input type="text" value="3"/> (0~8)
PIN Code Enabled:	<input type="radio"/> On <input checked="" type="radio"/> Off
PIN Code:	<input type="text"/>
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

### Caller ID Setting:

8.25. You may show caller ID in your PSTN Phone or IP Phone by selecting “Yes” in Single Caller ID, and the desired Caller ID option for either FSK or DTMF. When you finished the setting, please click the “Submit” button.

## Caller ID Setting

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

Phone Book	
Call f	Call Forward
Netw	SNTP Settings
SIP s	Volume Settings
Auto	DND Settings
User	Auto Answer
Save	Caller ID
Upda	Dial Plan Settings
Rebo	Flash Time Settings
	Call Waiting
	T.38(FAX) Settings
	Hot line Settings

Caller ID:	<input type="text" value="Caller ID after 1st Ring (FSK)"/>
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Type 2:	<input type="radio"/> Yes <input checked="" type="radio"/> No
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

### Dial Plan Settings:

8.26. Dial plan and auto dial timer settings can be set in this page. The dial plan allows

you to map the dialing into an easy-to-remember phone number system. The auto dial timer specifies the elapse time between the dialing digits. When Drop prefix is ON and the dialing prefix is matched, the prefix will be dropped and replaced by the rule digits and followed by the rest of dialing digits. When Drop prefix is OFF and the dialing prefix is matched, the rule digits will be added before the dialing digits in accord with the settings.

#### 8.27. Symbol Representations:

Symbol	Representations
x or X	0,1,2,3,4,5,6,7,8,9
+	or

**Example 1:** Drop Prefix: **No**, Replace rule 1: **002, 8613+8862**

- a) Pressing **8613xxx** will result in dialing out **002+8613+xxx**.
- b) Pressing **8862xxx** will result in dialing out **002+8862+xxx**.

**Example 2:** Drop Prefix: **Yes**, Replace rule 2: **006, 002+003+004+005+009**

- a) Pressing **002xxx** will result in dialing out **006+xxx**.
- b) Pressing **003xxxx** will result in dialing out **006+xxxx**.

**Example 3:** Drop Prefix: **No**, Replace rule 3:  
**009, 12**

- a) Pressing **12xxx** will result in dialing out  
**009+12xxx**.

**Example 4:** Drop Prefix: **No**, Replace rule 4: **007, 5xxx+35xx+21xx**

- a) Pressing **5xxx** will result in dialing out **007+5xxx**.
- b) Pressing **534** will result in dialing out **534** (not matched for the rest 3 digits).
- c) Pressing **35xx** will result in dialing out **007+35xx**.
- d) Pressing **356** will result in dialing out **356** (not matched for the rest 2 digits).
- e) Pressing **35668** will result in dialing out **35668** (not matched for the rest 2 digits).

**Example 5:** Dial Now: **\*xx+#xx+11x+xxxxxxxxx**



- a) Pressing **\*00**, **\*01**, **\*02 .. \*99** will result in dialing out the same **\*xx** **immediately**.
- b) Pressing **#00**, **#01**, **#02 .. #99** will result in dialing out the same **#xx** **immediately**.
- c) Pressing **110**, **111**, .. **119** will result in dialing out the same **11x** **immediately**.
- d) Pressing **12345678** (8 digits) will result in dialing out **12345678** **immediately**. This implies that the phone numbers with 9 or more digits are prohibited.

8.28. Auto Dial Timer: The inter-digit timer. Default is 5 seconds.

8.29. When you finish the setting, please click the Submit button.

8.30. Click the Save button. The changes you have made will be saved and the ANSEL 5517 will reboot automatically.

**Phone Book**

**Call** : Call Forward

**Netw** : SNTP Settings

**SIP S** : Volume Settings

**Auto** : DND Settings

**User** : Auto Answer

**Save** : Caller ID

**Upda** : Dial Plan Settings

**Rebo** : Flash Time Settings

**Call Waiting**

**T.38(FAX) Settings**

**Hot line Settings**

## Dial Plan

You could the set the dial plan in this page.

---

Drop prefix : ☐ Yes ☒ No

Replace rule 1:  +

Drop prefix : ☒ Yes ☐ No

Replace rule 2:  +

Drop prefix : ☐ Yes ☒ No

Replace rule 3:  +

Drop prefix : ☐ Yes ☒ No

Replace rule 4:  +

Dial now:

Auto Dial Time:  (3~9 sec)

Use # as send key: ☒ Yes ☐ No

Use \* for IP dialing: ☒ Yes ☐ No

### Flash Time Setting:

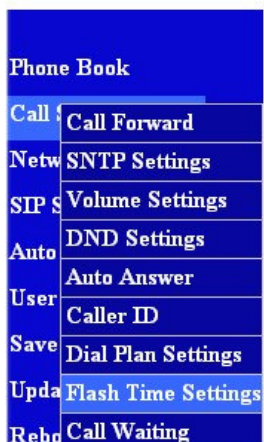
8.31. You can set the flash time duration for the telephone flash key or hook switch in this page. The telephone flash key is used to switch to the other phone line or HOLD, and is quite useful for the 3-way conference call and the call waiting function. When



you finished the setting, please click the “Submit” button.

## Flash Time Setting

You could set the flash time in this page.



### FXO Flash Time

Flash Time:  x 10MS (9~120)

### FXS Flash Time

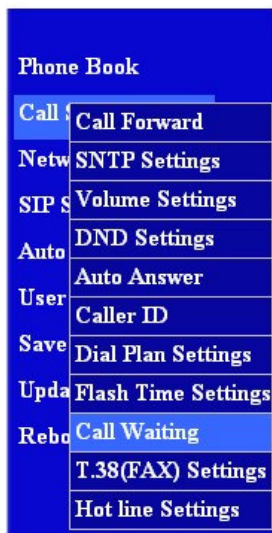
Max Flash Time:  x 10MS (4~255)

### Call Waiting Setting:

8.32. You can enable the call waiting function in this page. It allows answering another coming call by pressing flash key while holding the current call. You may switch back to previous call by pressing flash key again. When you finished the setting, please click the “Submit” button.

## Call Waiting Setting

You could enable/disable the call waiting setting in this page.



Call Waiting: ☒ On ☐ Off

**Call Transfer function:**

The call transfer function allows users to answer an incoming call and to hold the current call by pressing flash key, and then transfer the current call to the desired party by dialing the desired party number ended with # key. The call transfer function is exclusive with call waiting function. You may enable call transfer function by disabling the call waiting function (#139#), or disable call transfer function by enabling the call waiting function (#138#).

**T.38 (FAX) Setting:**

8.33. T.38 function can be used for FAX transmission over IP. Note that T.38 function must be enabled for both side of FAX over IP. You may enable or disable the T.38 function. When you finished the setting, please click the Submit button.

### T.38 (FAX) Setting

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

Phone Book

Call Forward

SNTP Settings

Volume Settings

DND Settings

Auto Answer

Caller ID

Dial Plan Settings

Flash Time Settings

Call Waiting

T.38(FAX) Settings

Hot line Settings

T.38 (FAX):

☒ On ☐ Off

T.38 Port:

(1024~65533)

Submit

Reset

**Hot Line Setting:**

- 8.34. The Hot Line mode allows to making a direct call at the Phone Number or IP stored in this page without dialing. Hot-Line Mode is very convenient for IP calling to Public Switching Telephone Network (PSTN) number through FXO Gateway
- 8.35. When the Hot Line mode is enabled, you just pick up the phone and the ANSEL 5517 will call the party directly to the preset IP (or URL) address. The default for Hot Line mode is disabled.

8.36. You need to “Enable” and click the “Submit” button and reboot to activate the function.

## Hot line Setting

You could set the hot line in this page.

**Use Hot Line :** ☒ Enable ☐ Disable

Hot line number:

Phone Book

Call **Call Forward**

Netw **SNTP Settings**

SIP S **Volume Settings**

Auto **DND Settings**

User **Auto Answer**

Save **Caller ID**

Upda **Dial Plan Settings**

Rebo **Flash Time Settings**

**Call Waiting**

**T.38(FAX) Settings**

**Hot line Settings**

Pick up the phone. In 1-2 seconds (default wait time), the ANSEL 5517 will automatically call the preset IP or phone number.

## Network

8.37. ANSEL 5517 is equipped with an embedded NAT router between WAN and LAN ports to meet the IP Network requirements. If you have an external NAT router, then you may select Bridge mode in WAN setting. Thus the two WAN and LAN Ethernet ports will be bridged and transparent. Otherwise, you may select NAT mode to enable embedded NAT and go on DDNS settings. The default is for NAT mode.

### Network Status:

8.38. You can check and show the current Network settings in this page.

## Network Status

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



Interface WAN	
Type:	DHCP Client
IP:	192.168.0.200
Mask:	255.255.255.0
Gateway:	192.168.0.254
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface LAN	
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

### WAN Settings:

- 8.39. The WAN setting is used to configure the Ethernet port connects to the ADSL Modem/Router.
- 8.40. The default setting is for NAT mode to enable the embedded NAT router between the WAN port and LAN port. You may select Bridge Mode if you need NOT use the embedded NAT router. When setting to Bridge Mode, only the WAN settings will get effective and the LAN settings will be ignored.
- 8.41. There are three selections for WAN IP Type: Fixed IP, DHCP Client, and PPPoE modes. This WAN setting is for the WAN port when set in NAT mode. The default is at DHCP Client.
- 8.42. For Fix IP Mode, please make sure the IP address, Net Mask, Gateway, and DNS settings are suitable in your current network environment.
- 8.43. For PPPoE Mode, you have to enter correct username and password to get the IP address from your Internet Service Provider.
- 8.44. When you finished the settings, please click the Submit button.

## WAN Settings

You could configure the WAN settings in this page.

Phone Book
Call Settings
Network
SIP Settings
Auto
User
Save
Update
Reboot

LAN Mode: ☐ Bridge ☒ NAT

### WAN Setting

IP Type: ☐ Fixed IP ☒ DHCP Client ☐ PPPoE

IP: 192.168.62.170

Mask: 255.255.255.0

Gateway: 192.168.62.1

DNS Server1: 0.0.0.0

DNS Server2: 0.0.0.0

MAC: 0009261201cb

Host Name: VOIP\_TA1S10

### PPPoE Setting

User Name:

Password:

Submit

Reset

### LAN Settings:

- 8.45. The default IP address is 192.168.123.1, with Net Mask 255.255.255.0., and DHCP Server enabled. The IP addresses for DHCP are from 150 to 200.
- 8.46. Connect your PC to the LAN port, set your PC as DHCP mode, then the PC will automatically get an IP address from the TA.
- 8.47. When you finished the settings, please click the Submit button.

Phone Book
Call Settings
Netw
SIP S
Auto
User
Save
Update
Reboot

## LAN Settings

You could configure the LAN settings in this page.

<b>LAN Setting</b>	
IP:	192.168.123.1
Mask:	255.255.255.0
MAC:	000926ccdde
<b>DHCP Server</b>	
DHCP Server:	<input checked="" type="radio"/> On <input type="radio"/> Off
Start IP:	150
End IP:	200
Lease Time:	1 : 0 (dd:hh)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

### DDNS Setting:

8.48. You need to have a DDNS account before configuring the DDNS setting. Usually, most of the VoIP applications are working with a SIP Proxy Server. Nonetheless, you may have a DDNS account with a public IP address, and others can call you via the DDNS account. When you finished the setting, please click the Submit button.

## DDNS Settings

You could set the configuration of DDNS in this page.

Phone Book
Call Settings
Netw
SIP S
Auto
User
Save
Update
Reboot

<b>DDNS:</b> <input checked="" type="radio"/> On <input type="radio"/> Off	
Host Name:	
User Name:	
Password:	
E-mail Address:	
DDNS Server:	
DDNS Server List:	User Input
Type:	dyndns
Wild Card:	on
BACKMX:	<input type="radio"/> On <input checked="" type="radio"/> Off
Off Line:	<input type="radio"/> On <input checked="" type="radio"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

## **VLAN Setting:**

- 8.49. There are two parts for VLAN settings. One is to set for VoIP packets related to ANSEL 5517, and the other is for the VLAN setting in the NAT Mode.
- 8.50. There are two kinds of destination packets coming to the TA's WAN port, one is VoIP packets for ANSEL 5517, and the other will go through the WAN port to the LAN port.
- 8.51. VLAN Packets: If you enable VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.
- 8.52. VID: Please set your VID in accordance with your service provider.
- 8.53. User Priority: Defines user priority with eight ( $2^3$ ) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually, this will be defined by your service provider.
- 8.54. CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.
- 8.55. When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the TA's IP address and the same VID will be accept by the TA. If the incoming packets with TA's IP address but different VID then the packets will be discard by the TA. The Other incoming packets with different IP address will go through the WAN port to the LAN port.
- 8.56. NAT VLAN Setting: When ANSEL 5517 is set in embedded NAT mode, the TA will filter out the wrong incoming packets. You can separate the other devices connected behind the embedded NAT into 4 VLAN groups by assigning different VID for these 4 groups. The ANSEL 5517 will check the VID of the incoming packets from ADSL modem. If the packet is not with the ANSEL 5517 IP address and the right VID, or the VID is not of of these 4 VIDs you set, then the packets will be discarded by the ANSEL 5517.

## VLAN Settings

You could set the VLAN settings in this page.



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

NAT VLAN Setting		
VLAN Packets:	<input checked="" type="radio"/> Off	<input type="radio"/> On
VID1:	<input type="text" value="4"/>	(2 ~ 4094), 0->Off
VID2:	<input type="text" value="5"/>	(2 ~ 4094), 0->Off
VID3:	<input type="text" value="6"/>	(2 ~ 4094), 0->Off
VID4:	<input type="text" value="7"/>	(2 ~ 4094), 0->Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>		

## SIP Settings

8.57. 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.

### Service Domain Settings:

8.58. You may register up to three SIP Servers for three Realms in the ANSEL 5517. You can receive the incoming calls from all the three SIP Servers. For outgoing calls, you may select the registration SIP server first, and then call the associated registration phone number.

8.59. To select SIP Server **1** (default), please pickup the phone, press **1\***, then hangup.

8.60. To select SIP Server **2** (or **3**), please pickup the phone, press **2\*** (or **3\***), then hangup.

8.61. Click "**Active**" ON to enable the Service Domain, then enter the following items:

8.62. **Display Name:** enter the name you want to display.



- 8.63. **User Name:** enter the User Name given by your ITSP.
- 8.64. **Register Name:** enter the Register Name given by your ITSP.
- 8.65. **Register Password:** enter the Register Password given by your ITSP.
- 8.66. **Domain Server:** enter the Domain Server given by your ITSP.
- 8.67. **Proxy Server:** enter the Proxy Server given by your ITSP.
- 8.68. **Outbound Proxy:** enter the Outbound Proxy of ITSP. If not provided, you may skip this.
- 8.69. **Register Period:** enter the Register Period in minute given by your ITSP.
- 8.70. When it shows “**Registered**” in the Register Status, it indicates a successful registration to the ITSP, and the “**PHONE**” LED will start flashing. The ANSEL 5517 is then ready for VoIP call.
- 8.71. After you finished the setting, please click the “Submit” button.

#### **DTMF Settings:**

- 8.72. You can setup the options for DTMF function in this page. The options include RFC2833 (Outband DTMF), Inband DTMF, and Send DTMF SIP info. The default is set at Inband DTMF. If you are making two-stage callings for extension to PSTN, you might need to select Outband DTMF option.

#### **Port Settings:**

- 8.73. You can setup the SIP and RTP port number in this page. Each ITSP provider might have different SIP/RTP port setting, please refer to the ITSP to setup the port number correctly. When you finished the setting, please click the “Submit” button. The defaults for SIP port and RTP port are 5060 and 60000, respectively.

#### **STUN Settings:**

8.74. The STUN function must be enabled to work properly behind NAT when registered in SIP server. You may enter the STUN server IP address and the STUN port number as shown in the following example.

Phone Book

Call Settings

Network

SIP Set

Auto C

User P

Save C

Update

Reboot

Service Domain

Codec Settings

Codec ID

Other Settings

Service Domain Settings

You could set information of service domains in this page.

Realm 1

Active: ☒ On ☐ Off

Display Name: 636346

User Name: 636346

Register Name: 636346

Register Password: ●●●●

Domain Server: fwd.pulver.com

Proxy Server: fwd.pulver.com

Outbound Proxy: fwdnat.pulver.com:5082

Status: Registered

Realm 2

Active: ☐ On ☒ Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Status: Not Registered

Phone Book

Call Settings

Network

SIP Set

Service Domain

Auto C

Codec Settings

User P

Codec ID

Save C

Other Settings

Update

Reboot

**Realm 3**

Active: ☐ On ☒ Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Status: Not Registered

**DTMF Setting**

☒ 2833

☐ Inband DTMF

☐ Send DTMF SIP Info

**Port Setting**

SIP Port:  (1024~65535)

RTP Port:  (1024~65535)

**STUN Setting**

STUN: ☐ On ☒ Off

STUN Server:

STUN Port:

## Codec Settings:

8.75. You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ITSP recommendations to setup these items.

Phone Book

Call Settings

Network

SIP S

Auto

User

Save

Update

Reboot

Service Domain

Codec Settings

Codec ID

Other Settings

# Codec Settings

You could set the codec settings in this page.

Codec Priority

Codec Priority 1:

G.729

Codec Priority 2:

G.723

Codec Priority 3:

G.711 u-law

Codec Priority 4:

G.711 a-law

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:

☐ On ☒ Off

Voice VAD

Voice VAD:

☐ On ☒ Off

Submit

Reset

## Codec ID Settings:

8.76. You can setup the Codec ID in this page. You need to follow the ITSP suggestion to setup these items.

# Codec ID Setting

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

Phone Book

Call Settings

Network

SIP S

Auto

User

Save

Service Domain

Codec Settings

Codec ID Setting

Other Settings

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

Submit

Reset

## Other Settings:

8.77. You can setup the Hold by RFC and QoS in this page. To change these settings please follows your ITSP information. When you finished the setting, please click the Submit button. The QoS is used to set the voice packet priority. Higher value other than zero will get higher priority for the voice packets in Internet. However, the QoS function still needs to cooperate with the other Internet devices.

### Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS (Diff-Serv):	<input type="text" value="40"/> (0~63)
SIP QoS (Diff-Serv):	<input type="text" value="40"/> (0~63)
SIP Expire Time:	<input type="text" value="60"/> (30~86400 sec)
Use DNS SRV:	<input type="radio"/> On <input checked="" type="radio"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- Phone Book
- Call Settings
- Network
- SIP Settings**
  - Service Domain
  - Auto Codec Settings
  - User Codec ID
  - Other Settings**
- Save
- Update

## Auto Configuration Setting

8.78. Auto Configuration function can be used to download the original configurations stored in the TFTP, HTTP or FTP server. This is useful for the new user to automatically download a predefined configuration setting. Remember to click the "Submit" button and "Save" in the **Save Change** section. The ANSEL 5517 will then reboot and automatically download the original configurations from the TFTP or FTP server.

Phone Book

Call Settings

Network

SIP Settings

Auto Config

User Password

Save Change

Update

Reboot

## 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:

FTP Username:

FTP Password:  Exp. 60.35.17.1

FTP File Path:  Exp. /file/load

Submit

Reset

## 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. 60.35.137.30

HTTP File Path:  Exp. /download/

FTP Server:

FTP Username:

FTP Password:  Exp. 60.35.17.1

FTP File Path:  Exp. /file/load

Submit

Reset

Phone Book

Call Settings

Network

SIP Settings

Auto Config

User Password

Save Change

Update

Reboot

## Auto Configuration Setting

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

<b>Phone Book</b>	Auto Configuration:	<input type="radio"/> Off	<input type="radio"/> TFTP	<input checked="" type="radio"/> FTP	<input type="radio"/> HTTP
<b>Call Settings</b>	TFTP Server:	<input type="text"/>			
<b>Network</b>	HTTP Server:	<input type="text"/>	Exp. 60.35.187.30		
<b>SIP Settings</b>	HTTP File Path:	<input type="text"/>	Exp. /download/		
<b>Auto Config</b>	FTP Server:	<input type="text" value="voip.sipserver.com"/>			
<b>User Password</b>	FTP Username:	<input type="text" value="test"/>			
<b>Save Change</b>	FTP Password:	<input type="password" value="••••"/>	Exp. 60.35.17.1		
<b>Update</b>	FTP File Path:	<input type="text" value="/ta1s1o/"/>	Exp. /file/load		
<b>Reboot</b>	<input type="button" value="Submit"/> <input type="button" value="Reset"/>				

### User Password

8.79. You may change the login name and password in this page.

## User Password

You could change the login username/password in this page.

<b>Phone Book</b>	New username:	<input type="text"/>
<b>Call Settings</b>	New password:	<input type="password"/>
<b>Network</b>	Confirmed password:	<input type="password"/>
<b>SIP Settings</b>	<input type="button" value="Submit"/> <input type="button" value="Reset"/>	
<b>Auto Config</b>		
<b>User Password</b>		

### Save Changes

8.80. You can save the changes you have made, and click the Save button. After clicking the “Save” button, the ANSEL 5517 will automatically save the new settings.

## Save Changes

You have to save changes to effect them.

---

Phone Book

Call Settings

Network

SIP Settings

Auto Config

User Password

Save Change

Update

Reboot

Save Changes:

### Update New Firmware:

8.81. ANSEL 5517 provides two methods, HTTP or TFTP, to update new firmware as the following steps:

8.82. Select the firmware code type, Risc or DSP code. (mostly for Risc code)

8.83. Click the "Browse" button to choose the updated file location for HTTP download, or

8.84. Select TFTP and enter the IP address of TFTP server for firmware download, then click the "Update" button.



Phone Book  
 Call Settings  
 Network  
 SIP Settings  
 Auto Config  
 User Password  
 Save Change  
 Update New Firmware  
 Reboot Auto Update

## Update Firmware

You could update the newest firmware.

Method: ☒ Local PC ☐ TFTP

### Local PC

Code Type: Risc

File Location:  Browse...

### TFTP

TFTP Server: 192.168.1.250

Update

Reset

## Update Firmware

You could update the newest firmware.

Method: ☐ HTTP ☒ TFTP

### HTTP

Code Type: Risc

File Location:  Browse...

### TFTP

TFTP Server: 61.66.159.237

Update

Reset

### Auto Update:

8.85. ANSEL 5517 also provides TFTP, HTTP or FTP for firmware auto-update as the following steps:

8.86. Select the update method and enter the server IP address,

8.87. Click the "Submit" button to get auto-update effective.

Phone Book

Call Settings

Network

SIP Settings

Auto Config

User Password

Save Change

Update

New Firmware

Reboot

Auto Update

Default Settings

## Auto Update Settings

You could set auto update settings in this page.

Update via: ☐ Off ☒ TFTP ☐ FTP ☐ HTTP

TFTP Server:

HTTP Server:  Exp. 60.35.187.30

HTTP File Path:  Exp. /download/

FTP Server:

FTP Username:

FTP Password:  Exp. 60.35.17.1

FTP File Path:  Exp. /file/load

Check new firmware: ☐ Power ON ☒ Scheduling

Scheduling (Date):  (1~30 days)

Scheduling (Time):  ▼

Automatic Update: ☒ Notify only ☐ Automatic

Firmware File Prefix:

Next update time:

Phone Book

Call Settings

Network

SIP Settings

Auto Config

User Password

Save Change

Update New Firmware

Reboot Auto Update

Default Settings

## Auto Update Settings

You could set auto update settings in this page.

Update via: ☐ Off ☐ TFTP ☒ FTP ☐ HTTP

TFTP Server:

HTTP Server:  Exp. 60.35.187.30

HTTP File Path:  Exp. /download/

FTP Server:  61.66.159.237

FTP Username:  test

FTP Password:  Exp. 60.35.17.1

FTP File Path:  /ta1s1p/ Exp. /file/load

Check new firmware: ☐ Power ON ☒ Scheduling

Scheduling (Date):  14 (1~30 days)

Scheduling (Time):  AM 00:00- 05:59

Automatic Update: ☒ Notify only ☐ Automatic

Firmware File Prefix:  TA1S1P

Next update time:

Submit

Reset



## Auto Update Settings

You could set auto update settings in this page.

Update via: ☐ Off ☐ TFTP ☐ FTP ☒ HTTP

TFTP Server:

HTTP Server:  Exp. 60.35.187.30

HTTP File Path:  Exp. /download/

FTP Server:

FTP Username:

FTP Password:  Exp. 60.35.17.1

FTP File Path:  Exp. /file/load

Check new firmware: ☐ Power ON ☒ Scheduling

Scheduling (Date):  (1~30 days)

Scheduling (Time):

Automatic Update: ☒ Notify only ☐ Automatic

Firmware File Prefix:

Next update time:

- 8.88. After clicking the “Update” button, the firmware list will be displayed from server to indicate the available firmware for download.
- 8.89. Select the new file you want to download to the ANSEL 5517 then click the “Select” button.
- 8.90. In 3 to 4 minutes, the PHONE LED indicators will start flashing 5 times to indicate successful firmware update. Then, you need to login again new IP address which is available from IVR by pressing **#120#** from phone.

## Firmware List

You could choose one of the firmware to update.

No	Risc Version List	Select
0	vs211_01td.gz	<input type="radio"/>
1		
2		<input type="radio"/>
3		
4		<input type="radio"/>
5		
6		<input type="radio"/>
7		
8		<input type="radio"/>
9		

No	DSP Version List	Select
0		<input type="radio"/>
1		
2		<input type="radio"/>
3		
4		<input type="radio"/>

Submit

Reset

8.91. NOTE: Do NOT power OFF the ANSEL 5517 after clicking the “Select” button, or you may damage the ANSEL 5517.

8.92. NOTE: The remote TFTP download works only for public IP address.

## Note Information

It takes 3-4 min to update and reboot automatically.



Configure OK.

When update finished, the POWER, PHONE LEDs will be OFF for a while and start flashing 5 times. Then, please use phone IVR to get new IP and relogin again.

Now, please wait ...

### Default Setting:

8.93. You can restore the ANSEL 5517 to factory default in this page. By clicking the “Restore” button, the ANSEL 5517 will restore to default and automatically restart again.

## Restore Default Settings

You could click the restore button to restore the factory settings.



Restore default settings:

### Reboot

8.94. You may click the Reboot button to restart, then ANSEL 5517 will automatically reboot with the stored configurations.

## Reboot System

You could press the reboot button to restart the system.



Reboot system:

## 9. Configurations by Telephone & IVR

You can use telephone to configure and to check the status of ANSEL 5517. Make sure that the LAN port is connected to Ethernet, or you may hear a busy tone from the telephone.

Group	IVR Action	Phone Command	Remarks
Status	Check TA LAN IP Address	#120#	IVR will report the current TA local IP address. Hang up while hearing end tone.
Status	Check IP Type	#121#	IVR will report if WAN DHCP is enabled or disabled. Hang up while hearing end tone.
Status	Check Phone Number	#122#	IVR will report registered phone number.
Status	Check Network Mask	#123#	IVR will report WAN network mask.
Status	Check Gateway IP Address	#124#	IVR will report WAN Gateway IP address.
Status	Check Primary DNS Server IP Address	#125#	IVR will report WAN Primary DNS Server IP address.
Status	Check TA WAN IP Address	#126#	IVR will report the TA WAN IP address.
Status	Check Firmware Version	#128#	IVR will announce the firmware version.
Setting	Set WAN DHCP client	#111#	This set WAN to DHCP Client mode.

Setting	Set WAN Static IP Address	#112xxx*xxx*xxx*xxx#	Ex: #112061*066*159*009# Note xxx must be 3 decimal digits. This setting will disable DHCP Client.
Setting	Set WAN Network Mask	#113xxx*xxx*xxx*xxx#	Ex: #113255*255*255*000# Note xxx must be 3 decimal digits and Must set WAN Static IP first (#112).
Setting	Set Router IP Address	#114xxx*xxx*xxx*xxx#	Ex: #114061*066*159*254# Note xxx must be 3 decimal digits and Must set WAN Static IP first (#112).
Setting	Set Primary DNS Server	#115xxx*xxx*xxx*xxx#	Ex: #115159*168*001*001# Note xxx must be 3 decimal digits and Must set WAN Static IP first (#112).
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.
Setting	Set Handset Gain	#131+[00~15]#	Ex: #13107# and default is 06
Setting	Set Handset Volume	#132+[00~12]#	Ex: #13209# and Default is 10
Setting	Set TFTP Server IP Address	#135xxx*xxx*xxx*xxx#	Ex: #135061*066*159*254#
Setting	Set FTP Server IP Address	#136xxx*xxx*xxx*xxx#	Ex: #136159*168*001*001#
Setting	Auto Configure Mode	#137+[0~2]#	0: Disable, 1: TFTP mode 2: FTP mode

Group	IVR Action	Phone Command	Remarks
Setting	Enable Call Waiting	#138#	This will disable Call Transfer.
Setting	Disable Call Waiting	#139#	This will enable Call Transfer.
Setting	Unlock Keypad	#190#	You must unlock keypad first in order to change settings by keypad.
Setting	Lock Keypad	#191#	Keypad can NOT be used for setting.
Setting	Reboot	#195#	After you hear "Option Successful," hang-up and TA will reboot automatically.
Setting	Factory Reset	#198#	TA will reset back to factory defaults. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST!
Command	Blind Call Transfer	#510#xxxxxx#	Ex: #510#54321# transfer to 54321
Command	Attendant Call Transfer	#511#xxxxxx#	Ex: #511#54321# transfer to 54321
Command	3-Way Conference Call	#512#xxxxxx#	Ex: #512#54321# conference with 54321
Setting		0*	Switch to PSTN Line.
Setting		1*	Select the First SIP server.



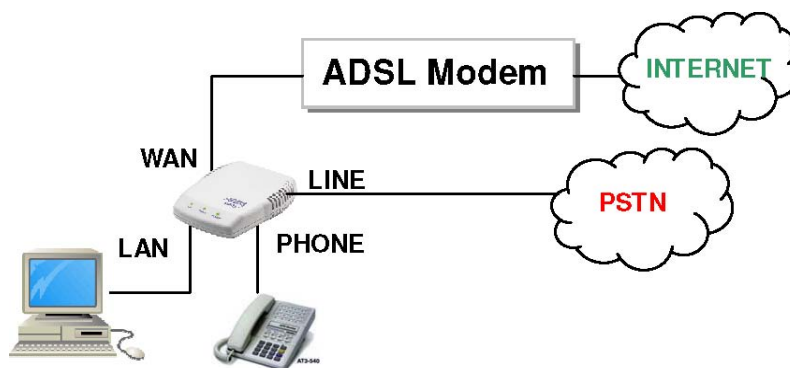
Setting		2*	Select the Second SIP server.
Setting		3*	Select the Third SIP server.

## 10.VoIP Application Examples

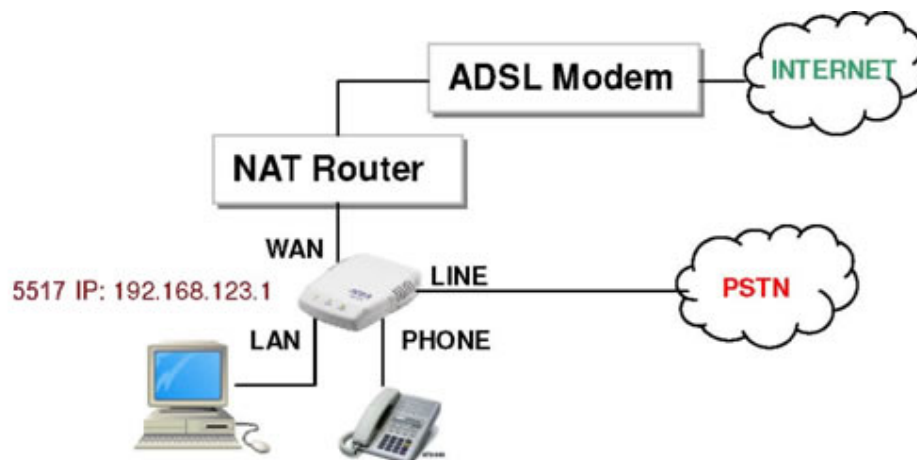
You can use PC Web browser to configure ANSEL 5517.

For example, enter **http://192.168.123.1** from PC web browser.

### A. ADSL Connections without NAT Router for ANSEL 5517



### B. ADSL Connections with NAT Router for ANSEL 5517



## Example 1: Public Switched Telephone Network (PSTN) Calling/Answering

### Applications:

ANSEL 5517 is default at the VoIP mode. For PSTN calls, you may just pick up the phone, press **0\*** key, and dial directly to the PSTN number like a normal telephone.

### Configurations:

- 1 Select **"ON"** for the **"Auto Answer"** in Call settings.
- 2 Select a value for Auto Answer Ring Counter, and the default value is set at **3**.
- 3 Click the **"Submit"** button.

### Calling/Answering

- 4 Pick up the phone and press **0\*** key for PSTN mode, and you should hear a dial tone.
- 5 Press, e.g. **1234567**, to call the PSTN party with **1234567**. In a moment, you should hear a ring back tone, and wait for the called PSTN party to answer.
- 6 When receiving PSTN incoming call, you must pick up the phone within **4** rings to answer, otherwise the VoIP mode will answer automatically for IP extension call. Please refer to **Example 5** for more details of PSTN to SIP extension calls.
- 7 If the **"Auto Answer"** is OFF, the FXO port will become PSTN only, and the function of extension call from PSTN to SIP call will be disabled.

## Example 2: SIP-to-SIP Calling/Answering

### Applications:

The applications can be for ADSL connections as in either Diagrams A or B. Both parties are registered to SIP server with 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. For Diagram A without NAT router, you may select NAT mode to enable the embedded NAT router. For Diagram B with external NAT router, you may select Bridge mode to disable the embedded NAT. Please refer to **Example 11** for more detailed SIP server registrations.

### Configurations:

- 1 Select either **"NAT"** or **"Bridge"** in accord with your network in **"WAN settings"** page,
- 2 Select **"DHCP Client"** to automatically get an IP address from NAT router.
- 3 Remember to click the **"Submit"** button,
- 4 Select Active **"ON"** in the **"SIP settings / Service Domain"** page,
- 5 Enter the items of **Register Name, Register Password, Proxy Server, and Outbound Proxy**,
- 6 Select **"ON"** in the **"STUN setting"**, if Outbound Proxy is NOT available.
- 7 Upon successful SIP registration, the PHONE LED will start **Green** flashing.

### Callings:

- 1 Pick up the phone, and you should hear a dial tone for VoIP mode.
- 2 Press **1688#** or **1688** to call the party with the registered SIP phone number **1688**. Note that **#** key will dial out the number immediately. Dialing without **#** will not dial out until the auto dial timer (default=5 seconds) elapsed.

### Example 3: SIP to PSTN Calling

#### Applications:

The applications can be for ADSL connections as in both Diagrams A and B. Both parties are registered to SIP server with either fixed real IP or private IP under NAT router. The SIP-to-PSTN calling works when both calling and answering parties are registered to SIP server with given registered phone numbers.

### Configurations:

- 1 Same as in **Example 2**.
- 2 Select "**ON**" in the "**SIP settings / STUN setting**" page, if Outbound Proxy is NOT available.
- 3 Select "**ON**" for the "**Auto Answer**" and "**PIN Code**" in Call settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**.
- 4 After registration, the PHONE LED will be flashing to indicate a successful SIP registration.

### Callings:

- 1 Pick up the phone for VoIP mode, and press **1688#** or **1688** to call the party with the registered SIP phone number **1688**.
- 2 After **4** rings of Auto Answer, the FXO port will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for PSTN. Note you must add the postfix "**#**". PIN Code is used to prevent from call piracy. Incorrect PIN Code will result in call disconnect. If PIN code is OFF, the caller may press PSTN number directly.
- 3 Press **1234567** to call the PSTN party number of **1234567**.

### Example 4: 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:

- 1 Same as in **Example 2**.

- 2 Select **"ON"** in the **"SIP settings / STUN setting"** page, if Outbound Proxy is NOT available.
- 3 Make sure the PHONE LED is flashing continuously with a successful SIP registration.

#### Callings:

- 1 Pick up the phone for VoIP mode.
- 2 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.

### Example 5: PSTN to SIP Calling

#### Applications:

The applications can be for ADSL connections as in both Diagrams A and B. Both parties are registered to SIP server with either fixed real IP or private IP under NAT router.

#### Configurations:

- 1 Same as in **Example 2**.
- 2 Select **"ON"** in the **"SIP settings / STUN setting"** page, if Outbound Proxy is NOT available.
- 3 Select **"ON"** for the **"Auto Answer"** and **"PIN Code"** in Call settings. Set the Auto Answer Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**.
- 4 Make sure the PHONE LED is flashing continuously with a successful SIP registration.

#### Callings:

- 5 Make a Call from PSTN line to the ANSEL 5517 FXO number, e.g. **7654321**. In a moment, you should hear a ring back tone, and wait for the ANSEL 5517 to answer. After **4** rings, the VoIP mode will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for VoIP mode. Incorrect PIN Code will result in call disconnect. Note that if PIN code is OFF, there will be not short beep tone and the caller may press SIP number directly.
- 6 Press **1688#** or **1688** to call the party with the registered SIP phone number **1688**. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

### Example 6: 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:

- 1 Select "**Fixed IP**", and bridge "**ON**" in the "**Network / WAN settings**" page,
- 2 Enter the items of **IP, Subnet Mask, Gateway IP**,
- 3 Click the "**Submit**" button.
- 4 Make sure the SIP server is OFF (default is OFF) and PHONE LED is NOT flashing.

Callings:

- 1 Pick up the phone for VoIP mode.
- 2 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.

## Example 7: 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:

- 1 Select "**Fixed IP**", and bridge "**ON**" in the "**Network / WAN settings**" page,
- 2 Enter the items of **IP, Subnet Mask, Gateway IP**,
- 3 Click the "**Submit**" button.
- 4 Make sure the SIP server is OFF (default is OFF) and PHONE LED is NOT flashing. Callings:
- 5 Pick up the phone for VoIP mode.
- 6 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.

## Example 8: Direct IP to PSTN Calling

Applications:

The Direct IP to PSTN calling is for the application when the answering parties are with known fixed real IP addresses. The SIP server registrations may not be necessary.

Configurations:

- 1 Same as in **Example 6**.
  - ❖ Select “**ON**” for the “**Auto Answer**” and “**PIN Code**” in Call settings. Set the Auto Answer
  - ❖ Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**.

Callings:

- 2 Pick up the phone for VoIP mode.
  - ❖ Press 211\*21\*191\*4# or 211\*21\*191\*4 to call the party with the real IP address of 211.21.191.4. After 4 rings, the FXO port will auto answer with a short beep tone (not dial tone). Press 1234# for PIN code and then you will hear a dial tone from PSTN. PIN Code is used to prevent from call piracy. Incorrect PIN Code will result in call disconnect. Note that if PIN code is OFF, the caller may press PSTN number directly.
- 3 Dial directly the number, e.g. **1234567** to call the PSTN party number of **1234567**. In a moment, you should hear a ring back tone, and wait for the called PSTN party to answer.

## Example 9: PSTN to Direct IP Calling

Applications:

The PSTN to Direct IP calling is for the application when the answering parties are with known fixed private or real IP addresses. The SIP server registrations may not be necessary.

Configurations:

- 1 Same as in **Example 6**.
  - ❖ 2. Select “**ON**” for the “**Auto Answer**” and “**PIN Code**” in Call settings. Set the Auto Answer
  - ❖ Ring Counter, e.g. **3**, and the PIN code, e.g. **1234**.

Callings:

- 2 Make a call from PSTN line to the ANSEL 5517 FXO number, e.g. **7654321**. In a moment, you should hear a ring back tone, and wait for the ANSEL 5517 to answer. After **4** rings of Auto Answer, the FXO port will auto answer with a short beep tone (not dial tone). Press **1234#** for PIN code and then you will hear a dial tone for VoIP mode.
- 3 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 another ring back tone, and wait for the VoIP called party to answer.

## Example 10: 3-Way Conference Call, Call Transfer, Call Waiting, Hold

### 3-Way Conference Call, Call Transfer Applications:

These are for call transfer and conferencing among Parties A, B, and C. Three parties are registered to SIP server with either fixed real IP or private IP. There are two kinds of call transfer; Blind Transfer and Attendant Transfer.

#### **Blind Transfer:**

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#510# [Party C number] #** to transfer to Party C.

#### **Attendant Transfer:**

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#511# [Party C number] #** to call and talk to Party C. Hang up from Party B, then Party A will transfer and connect to Party C.

#### **3-Way Conference Call:**

Party A calls Party B. While in conversation, Party B may press Flash key to hold the call and then press **#512# [Party C number] #** to call and talk to Party C. Press Flash key from Party B, then Party C will join for the conference call.

#### **Call Waiting Application:**

When a new call is coming while you are talking, you will hear an interrupt tone and you can push the Flash key to switch to answer the new call. You may push the Flash key to switch between the two calls.

#### **Call Hold Application:**

You may push the Hold (Flash) key to hold the current call for a while, then push Hold key again to resume talking.

## **Example 11: SIP-to-SIP Calling for FreeWorld Dialup (FWD)**

### Applications:

This shows how to use FWD as an example for free ITSP provider. The applications are for both parties registered to FWD SIP server.

- 1 Visit <http://www.freeworlddialup.com> and sign up for a new registered account number. Follow the instructions for registration.
- 2 After finished, you will receive a mail sent by the FWD mail system, and you will get one FWD phone number and password in the mail. For example, the register name/phone number is **636346** with password **xxxx**.
- 3 Login to the Web configuration page.

- 4 WAN and LAN Settings
- 5 SIP Settings You have to enter the Display Name, User Name, Registered Name, Registered Password, Domain Server (fwd.pulver.com), Proxy Server (fwd.pulver.com), Outbound Proxy (fwdnat.pulver.com:5082). After finished the setting, click the Submit button and the Save Change button. The system will reboot automatically. After system boot up, the SIP setting page will show "Registered", and the PHONE LED will start flashing.

Configurations:

## WAN Settings

You could configure the WAN settings in this page.

<b>Phone Book</b>	
<b>Call Settings</b>	
<b>Network</b>	<b>Status</b>
<b>SIP Settings</b>	<b>WAN Settings</b>
<b>Auto</b>	<b>LAN Settings</b>
<b>User</b>	<b>DDNS Settings</b>
<b>Save</b>	<b>VLAN Settings</b>
<b>Change</b>	
<b>Update</b>	
<b>Reboot</b>	

LAN Mode: ☐ Bridge ☒ NAT

**WAN Setting**  
IP Type: ☐ Fixed IP ☒ DHCP Client ☐ PPPoE  
IP:   
Mask:   
Gateway:   
DNS Server1:   
DNS Server2:   
MAC:   
Host Name:

**PPPoE Setting**  
User Name:   
Password:



Phone Book

Call Settings

Network

SIP Settings

Auto

User

Save

Update

Reboot

Status

WAN Settings

LAN Settings

DDNS Settings

VLAN Settings

Change

# LAN Settings

You could configure the LAN settings in this page.

LAN Setting

IP:

192.163.123.1

Mask:

255.255.255.0

MAC:

000926ccddee

DHCP Server

DHCP Server:

☒ On ☐ Off

Start IP:

150

End IP:

200

Lease Time:

1

:

0

(dd:hh)

Submrit

Reset

# Service Domain Settings

You could set information of service domains in this page.

Phone Book

Call Settings

Network

SIP Settings

Auto

User

Save

Update

Reboot

Service Domain

Codec Settings

Codec ID

Other Settings

Realm 1

Active:

☒ On ☐ Off

Display Name:

636346

User Name:

636346

Register Name:

636346

Register Password:

••••

Domain Server:

fwd.pulver.com

Proxy Server:

fwd.pulver.com

Outbound Proxy:

fwdnat.pulver.com:5082

Status:

Registered

Realm 2

Active:

☒ On ☐ Off

Display Name:

User Name:

Register Name:

Register Password:

Phone Book  
Call Settings  
Network  
SIP S  
Auto  
User  
Save  
Update  
Reboot

Service Domain
Codec Settings
Codec ID
Other Settings

Domain Server:   
Proxy Server:   
Outbound Proxy:   
Status: Not Registered

DTMF Setting

☒ 2833  
☐ Inband DTMF  
☐ Send DTMF SIP Info

Port Setting

SIP Port:  (1024~65535)  
RTP Port:  (1024~65535)

STUN Setting

STUN: ☐ On ☒ Off

STUN Server:   
STUN Port:

FWD SIP Server Register Name: 636346 Password: xxxx Domain Server: fwd.pulver.com, Proxy Server: fwd.pulver.com, Stun Server: stun.fwdnet.net

# Codec Settings

You could set the codec settings in this page.

---

## Codec Priority

Codec Priority 1:	G.729	▼
Codec Priority 2:	G.711 u-law	▼
Codec Priority 3:	G.711 a-law	▼
Codec Priority 4:	G.723	▼
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: ☐ On ☒ Off

## Voice VAD

Voice VAD: ☐ On ☒ Off

Submit

Reset

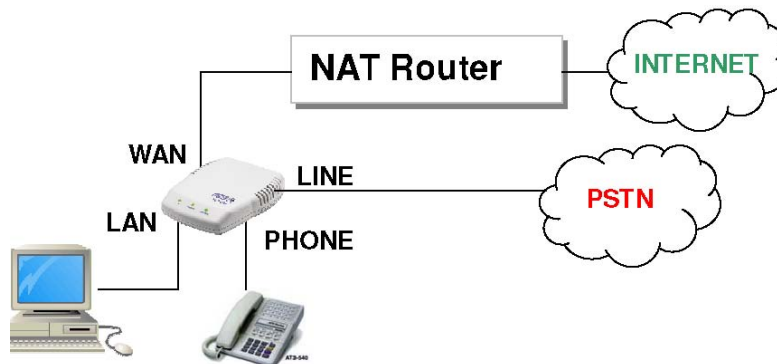
## Callings:

- 1 Pick up the phone for VoIP mode. (Your FWD phone number 636346).
- 2 Press **654321** to call the party with the registered FWD phone number **654321**. In a moment, you should hear the ring back tone, and wait for the called party to answer.

## 11. Trouble Shooting for Web Configurations

11.1. DO NOT HEAR DIAL TONE? The phone port of ANSEL 5517 is set to VoIP mode at default. When you pick up the phone and hear a busy tone, it indicates the WAN port is NOT connected. Make sure the ADSL Ethernet cable is connected to the WAN port of ANSEL 5517 and Power Reset again. You may press **0\*** key to switch to PSTN line, and you should hear another dial tone from PSTN. If not, please make sure the PSTN line is connected to the LINE port.

11.2. CAN NOT ACCESS WEB PAGE? IE Web Browser is a useful tool to configure ANSEL 5517. When you have difficulties in accessing the default IP address <http://192.168.123.1> of ANSEL 5517 as in the following figure, the most possibility is that the PC might have different subnet IP settings from [192.168.123.xxx](http://192.168.123.xxx). In this case, you must change ANSEL 5517 IP address to the same subnet as PC and NAT router.

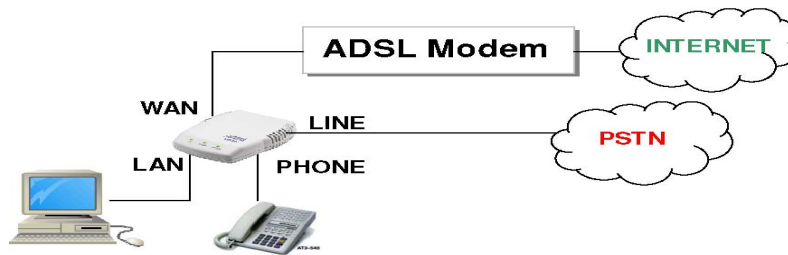


**Example:** To change ANSEL 5517 IP address to the same subnet as PC and NAT router

- 1 Pick up the phone and press **#111#** from the phone to enable DHCP Client mode. ANSEL 5517 will reboot, and LED will start flashing to get an IP address from NAT DHCP server.
- 2 Pick up the phone and press **#120#** to obtain the ANSEL 5517 IP address from telephone IVR, for example, [192.168.62.51](http://192.168.62.51).
- 3 Enter from IE web browser <http://192.168.62.51> to login ANSEL 5517 web page for configurations.

### 11.3. ONLY ONE IP AVAILABLE FROM ADSL/CABLE SERVICE PROVIDER?

Sometimes only one IP address is available from Internet Service Provider (ISP) without NAT router as the following figure. Usually, a DHCP or PPPoE server at the central side of ISP is used to assign one IP address to each user. In this case, you may need to enable the embedded NAT router of ANSEL 5517 to provide more than one IP addresses for PC and ANSEL 5517.



**Example:** To change PC IP address to the same subnet as **192.168.123.1** for ANSEL 5517

1. As in Window 2000 (my computer), -At "Network and Dialup Connections", right click on "Local Area Connection", then click on property. -The "Local Area Connection Properties" window will pop up. -Double click on "Internet Protocol (TCP/IP)". -The "Internet Protocol (TCP/IP) Properties" window will pop up. -Click on "Use the following IP Address". -Enter IP: **192.168.123.50** (50 can be any number other than 1, which is used by ANSEL 5517). -Enter Subnet mask: **255.255.255.0** -Enter Default gateway: **192.168.123.1** -Click on OK button.

- ❖ You will lose internet connection at this time.
- ❖ At IE browser, type <http://192.168.123.1>
- ❖ Follow the example in "Advanced Settings for Embedded NAT" for web login.
- ❖ At LAN setting, turn on DHCP server.
- ❖ At WAN setting, choose "DHCP client" to work with your ADSL/Cable modem.
- ❖ Save change, wait for ANSEL 5517 to reboot.
- ❖ Change your PC's "Internet Protocol (TCP/IP) Properties" back to "obtain an IP address automatically".
- ❖ You may press **#120#** and **#126#** to listen to the IVR for LAN and WAN IP addresses.

**11.4. VOIP EXTENSION CALLS TO PSTN ARE NOT WORKING?** You must enable the Auto Answer function in Call Settings in order to answer automatically by VoIP mode. The Auto Answer is disabled at default. Make sure the PSTN is connected to LINE port. When the ring count exceeds the number set in Auto Answer Counter, the FXO port will auto answer and allow for VoIP extension call.

#### **Extension Calls from Internet VoIP to PSTN**

If the incoming call is from VoIP, then FXO port will answer with a PSTN dial tone and allow caller to redial to PSTN phone number.

#### **Extension Calls from PSTN to Internet VoIP**

If the incoming call is from PSTN, then ANSEL 5517 FXO port will answer with a short beep tone and allow caller to redial to VoIP Phone number.