

P160S SIP Phone

Quick User Guide

Version 2.2

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1.0 INTRODUCTION

Voice over IP (also known as Internet Phone) is a technology that allows anyone to make a telephone call over the Internet. This is a quick user guide for the P160S SIP Phone. It is intended to help you configure the telephone and have it ready to run within a few minutes. Please follow the user guide carefully as troubleshooting the telephone can be very difficult and time consuming.

2.0 PACKAGE CONTENT

The following materials are included in the package. Please check the package to ensure that all the materials are listed below. Contact your supplier immediately if an item is missing.



IP Phone (Model: P160S)



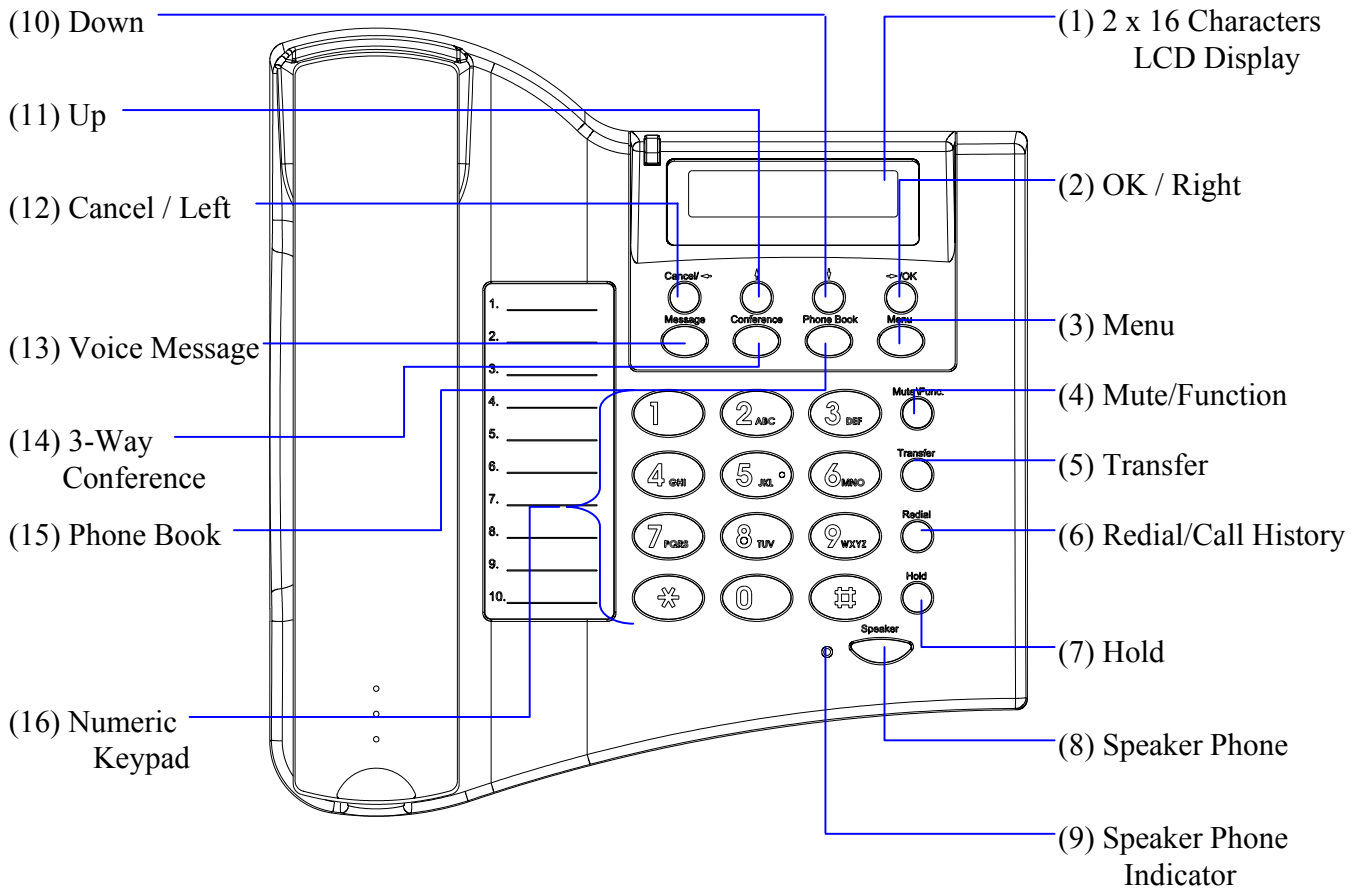
Ethernet Cable (1.8 metre)



Power Adaptor (5V DC)

3.0 LIST OF FIGURES

Diagram for ACT IP Phone (**Model: P160S**)

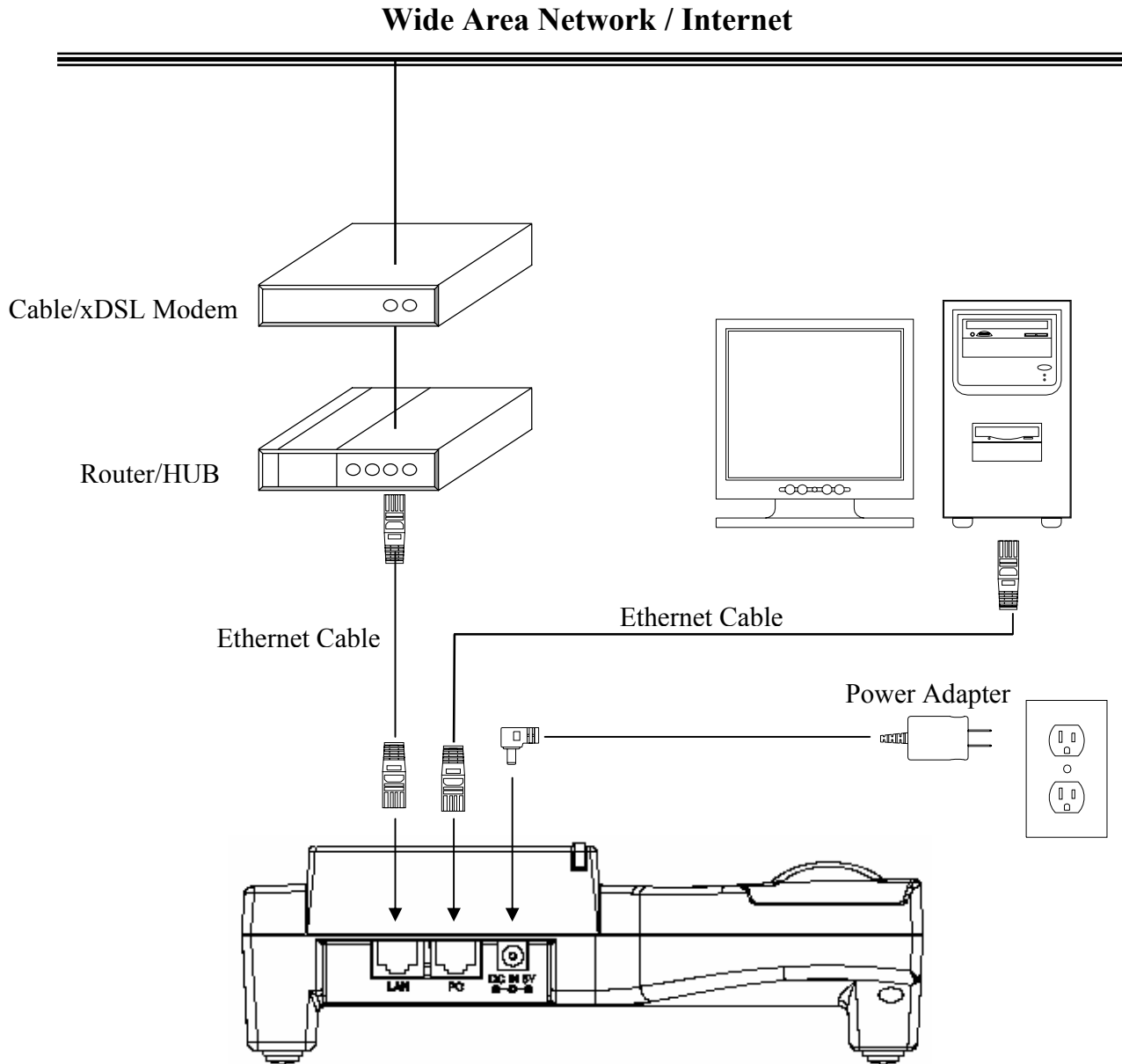


4.0 SUMMARY OF KEY FUNCTIONS

Keys	Functions
(1) LCD Display	Displays menu, time, clock, name, phone number, call status
(2) OK/Right	Confirm setting change, exit menu, dial, save changes
(3) Menu	Access the phone menu
(4) Mute/Function	Disable user's microphone so that the person on the other line can not hear anything, access the language selection, access the time format
(5) Transfer	Transfer the person you are currently having a conversation to another line
(6) Redial/Call History	Redial last dialed number, access redial menu
(7) Hold	Place the person on the other line on hold, answer call waiting
(8) Speaker Phone	Enable user to use the phone without using the handset
(9) Speaker Phone Indicator	Indicates that phone is currently in speaker phone mode
(10) Down	Cycle through the phone menu, adjust volume
(11) Up	Cycle through the phone menu, adjust volume
(12) Cancel/Left	Deny changes, cancel phone calls, ignore phone calls, backspace
(13) Voice Message	Check voice message
(14) 3-Way Conference	Enable 3-way conference
(15) Phonebook	Access the phonebook
(16) Numeric Keypad	Input IP/phone number/alphabet characters

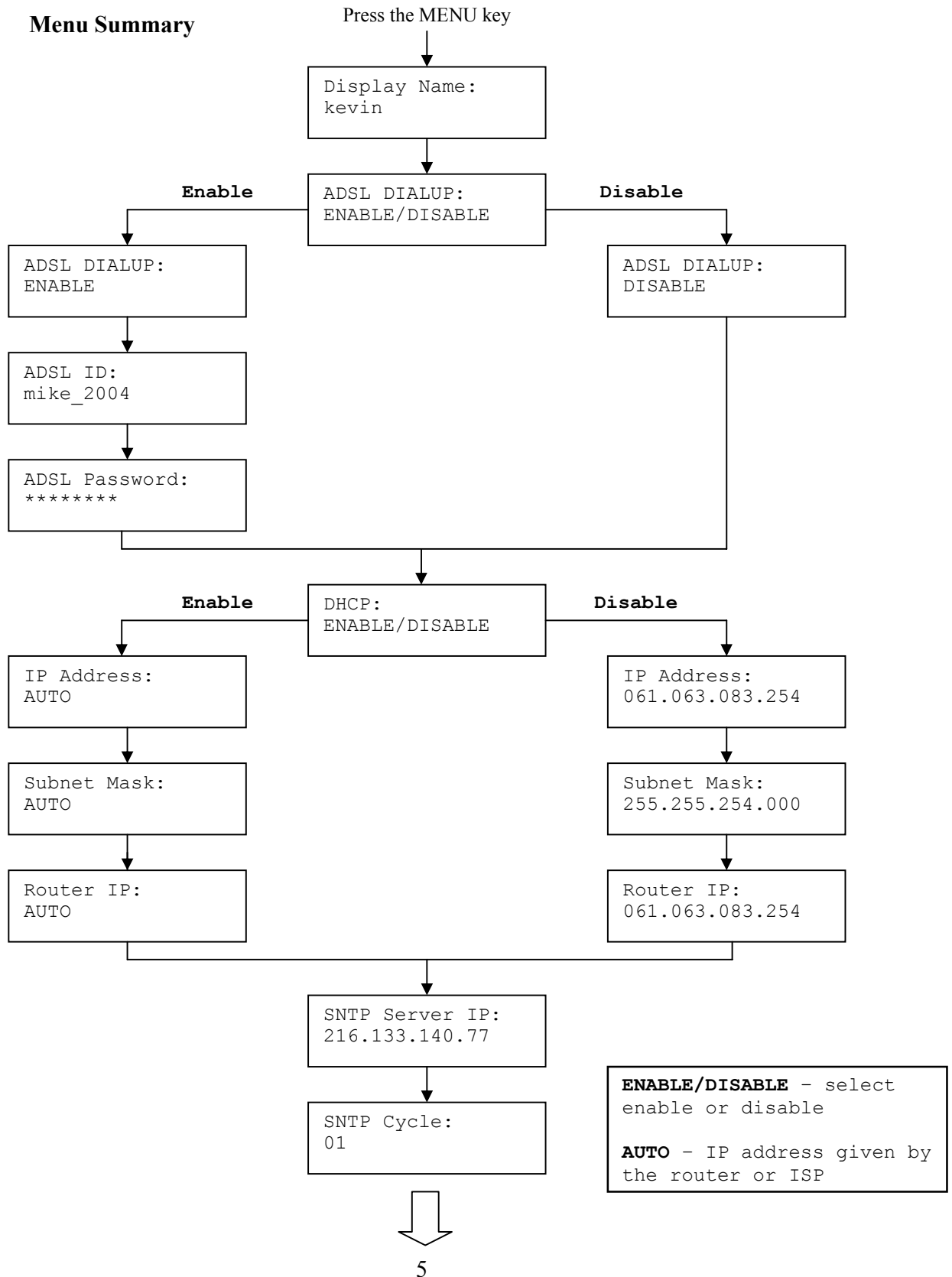
5.0 CONNECTING THE IP PHONE

Connect the IP Phone as the following diagram:



6.0 IP PHONE SETUP

6.1 Menu Summary





Do Not Disturb:
ENABLE/DISABLE



CF Unconditional:
ENABLE/DISABLE



CF User Busy:
ENABLE/DISABLE



CF No Answer:
ENABLE/DISABLE



Anonymous Call:
ENABLE/DISABLE



Anony Call Rej:
ENABLE/DISABLE



Ring Type:
Ringin1/2/3/4

Ringin1/2/3/4 - select
ringin1, 2, 3 or 4



MAC Address:
00D0E90137DB



Version:
V: 02.08



Press the MENU key




UPDATE CHANGES?
<OK> OR <CANCEL>

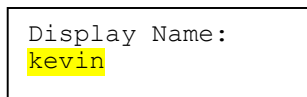
NOTE 1: If you made any modifications, you may quit setup at any time by pressing **MENU + OK** to save and exit or **MENU + CANCEL** to quit without saving. The phone will automatically exit from the menu screen if there are no inputs from the user.

NOTE 2: Use  or  to select **ENABLE** or **DISABLE**.

NOTE 3: Left arrow key  can be used as **Backspace** key.

6.2 Display Name

- Press 
- Enter the display name





Display Name:
kevin

6.3 ADSL Dialup

Some Internet Service Provider (mostly ADSL) uses PPPoE which requires that the user enter an ID and a password to access the Internet. In this case, enable ADSL DIALUP and enter the PPPoE ID and PPPoE password.

ENABLE ADSL Dialup


1

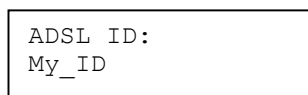
- Press 
- Use  to select **ENABLE**



ADSL DIALUP:
ENABLE


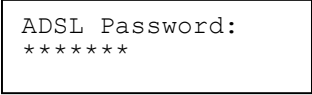
2

- Press 
- Enter the ADSL ID





ADSL ID:
My_ID

3

- Press 
- Enter ADSL password 

DISABLE ADSL Dialup




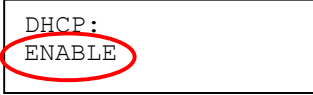
- Press 
- Press  to select **DISABLE** 

6.4 DHCP (Dynamic Host Configuration Protocol)


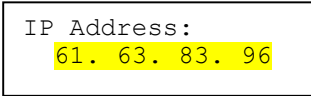
DHCP allows the network administrator to distribute IP addresses when a computer is plugged into a different place in the network. If your ISP provides static IP address, you must disable DHCP and enter the IP address provided.

ENABLE DHCP


1

- Press 
- Use  or  to set DHCP to **ENABLE** 

2


- Press 
- IP address automatically acquired 

3

- Press 
- Subnet mask automatically acquired

Subnet Mask:
255.255.254. 0




4

- Press 
- Router IP automatically acquired

Router IP:
61. 63. 83.254


DISABLE DHCP

1

- Press 
- Use  or  to set DHCP to **DISABLE**


DHCP:
DISABLE

2

- Press 
- Enter the IP address


IP Address:
061.063.083.019

3

- Press 
- Enter the subnet mask

Subnet Mask:
255.255.254.000


4

- Press 
- Enter the router IP address

Router IP:
061.063.088.019

6.5 SNTP Server IP




Simple Network Time Protocol (SNTP) is a protocol used to help match your system clock with an accurate time source. If you do not know your SNTP Server IP, please ignore this section. SNTP Server IP address can be either URL or IP.

- Press 
- Enter SNTP server IP or URL

SNTP Server IP:
216.133.140.78

6.6 Do Not Disturb




This setting allows the user to reject all incoming phone calls.

- Press 
- Use  or  to select **ENABLE** or **DISABLE**

Do Not Disturb:
DISABLE




6.7 CF (call forward) Unconditional

Enable CF Unconditional to forward all the incoming calls to another number. Otherwise set to disable. *You will need to use a web-browser to input the forwarded phone number. Refer to section 7.0 for more information on call forwarding.*

- Press 
 - Use  or  to select [ENABLE](#) or [DISABLE](#)
- CF Unconditional:
DISABLE
-




6.8 CF (call forward) User Busy

Forward all the incoming calls to another number when user is busy on the phone.

- Press 
 - Use  or  to select [ENABLE](#) or [DISABLE](#)
- CF User Busy:
DISABLE
-

6.9 CF (call forward) No Answer

Forward all incoming calls to another phone number after a certain number of rings.

- Press 
 - Use  or  to select [ENABLE](#) or [DISABLE](#)
- CF No Answer:
ENABLE
-

6.10 Anonymous Call

Enables the caller (user) to hide the name and phone number from the receiver.

- Press 



- Use  or  to select ENABLE or DISABLE

Anonymous Call:
ENABLE

6.11 Anony Call Rej (Anonymous Call Rejection)

Reject any anonymous incoming calls.

- Press 

- Use  or  to select ENABLE or DISABLE

Anony Call Rej:
DISABLE

6.12 Ringing Type



Select the ring tone. There are four ring tones in total.

- Press 

- Use  or  to select the ring type


Ringing Type:
Ringing4

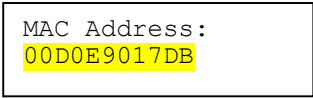
NOTE: At this point, you may save the settings and exit. The next two sections explain how to obtain the MAC address and firmware version.

- Press  to exit menu
- When asked to save or cancel, press  to **SAVE**

6.13 MAC Address

This menu displays the MAC address. User cannot modify MAC address.


- Press 
- **MAC address** is displayed on the screen

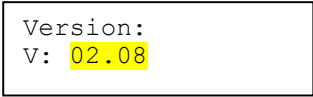


```
MAC Address:  
00D0E9017DB
```

6.14 Version

Version menu displays the firmware version. You cannot modify the version number.



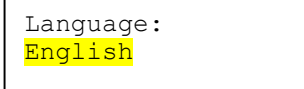



- Press 
- Firmware **version** is displayed on screen



```
Version:  
V: 02.08
```



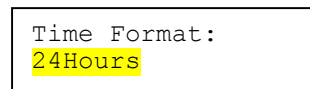



6.15 Language Selection

The VoIP Phone (model no. P160S) supports two languages: Japanese and English.

- Press  followed by  
- Use  or  to select the preferred language
- Press  when done

6.16 Time Format



You may select the 12hr or 24hr time format.

- Press  followed by  
- Use  or  to select the time format
- Press  when done

6.17 Volume Adjustment




6.17.1 Ringer Volume

While the handset is in place,



- Press  to increase the ringer volume and  to decrease the ringer volume

6.17.2 Speaker Volume

While the handset is in place,

- Press  **Speaker**
- Press  to increase the speaker volume and  to decrease the speaker volume

6.17.3 Handset Volume

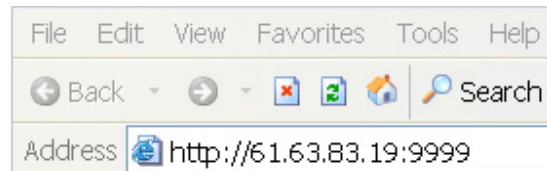
- Pick up the handset and press  to increase the volume or press  to decrease the volume

7.0 USING THE CONFIGURATION MENU

The configuration menu can be accessed using a web browser. Some advanced features such as CF Unconditional, CF User Busy and CF No Answer must be setup from the web browser.

7.1 Accessing Configuration Menu

- Open the web browser (ie. Internet Explorer, Netscape...)
- Type in the **IP Address** of the phone followed by :9999

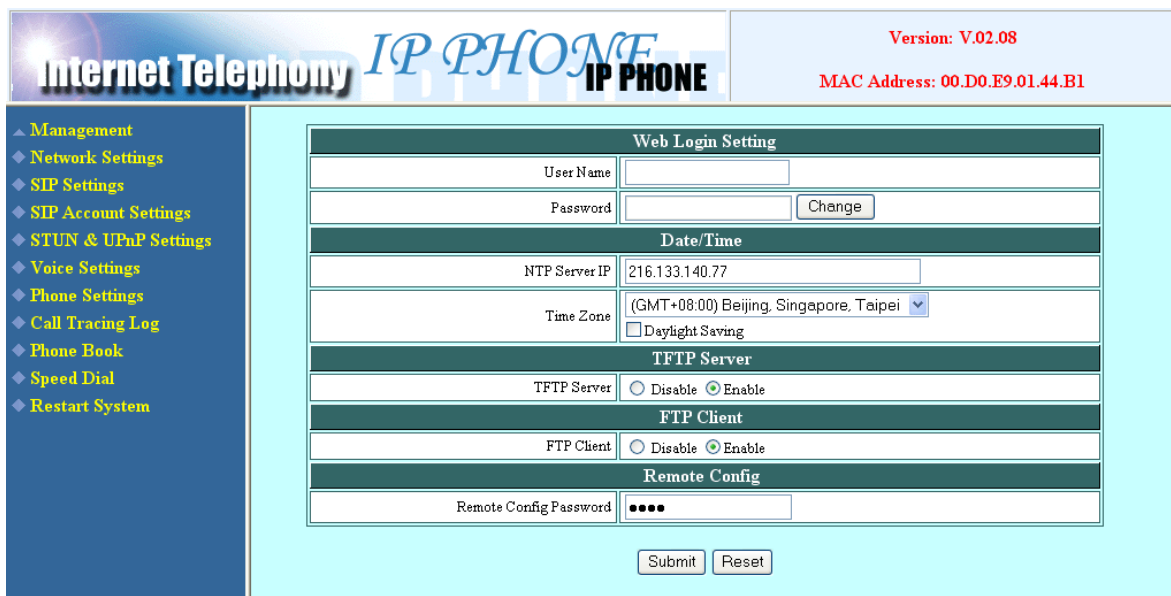


IP address is provided by your Internet Service Provider (ISP). If your ISP supports DHCP, you may obtain the IP address from you phone. Press MENU and scroll down to IP address.

- Enter **User Name** and **Password** (leave User Name and Password blank if you are installing the phone for the first time)
- Click **OK**



7.2 Web Login Setting

A screenshot of the IP PHONE web interface. The page has a header with "Internet Telephony IP PHONE" and "Version: V.02.08" and "MAC Address: 00.D0.E9.01.44.B1". On the left is a navigation menu with items like "Management", "Network Settings", "SIP Settings", etc. The main content area is titled "Web Login Setting" and contains several sections: "Web Login Setting" with "User Name" and "Password" fields and a "Change" button; "Date/Time" with "NTP Server IP" (216.133.140.77), "Time Zone" (GMT+08:00 Beijing, Singapore, Taipei), and a "Daylight Saving" checkbox; "TFTP Server" with "TFTP Server" radio buttons (Disable, Enable); "FTP Client" with "FTP Client" radio buttons (Disable, Enable); and "Remote Config" with a "Remote Config Password" field. At the bottom are "Submit" and "Reset" buttons.

User Name	Configuration menu login name.
Password	Configuration menu login password.
NTP Server IP	Network Time Protocol (NTP) is a protocol used to help match your system clock with an accurate time source (eg atomic clock, time server). It is good practice to have all your networked computers synchronized with one server.
Time Zone	Select your time zone. If there is daylight saving in your area, click the check box.
TFTP Server	Enable or disable TFTP server to allow transfer of firmware from a computer to the IP phone.
FTP Client	Enable or disable IP phone to download files from FTP server and update the firmware automatically.
Remote Config Password	Remote password to access the configuration menu from VoIP software (You may download this software from your supplier's website). Default password is 1234 .

7.3 Management Setting – Restore Factory Setting



Click on Management. Select Restore Factory Setting and the above screen will display on the screen.

Restore Factory Setting Restores all the settings back to factory default settings.

7.4 Management Setting – Firmware update

Management

- Restore Factory Setting
- Firmware update**
- Network Settings
- SIP Settings
- SIP Account Settings

FTP Server :

Login ID : Max. 32 Char.

Login Password : Max. 32 Char.

Firmware Filename : Max. 32 Char.

Firmware Upgrade Cancel

FTP server, login ID, login password and firmware filename are preset when you purchase the phone. These are required to download and update the firmware.

FTP Server

FTP Server address.

Login ID

Login ID provided by your supplier.

Login Password

Login password provided by you supplier.

Firmware Filename

Updated firmware filename. Do not change the file name unless specified by your supplier.

7.5 Network Setting – DHCP

Management

- Restore Factory Setting
- Firmware update
- Network Settings**
- SIP Settings
- SIP Account Settings

DHCP / PPPoE / Static IP

DHCP PPPoE Static IP

DNS Setting

DNS Server

Submit Reset

Select DHCP if you have cable internet.

DHCP Server

Dynamic Host Configuration Protocol (DHCP) Server address. This IP address information is obtained automatically from your ISP.

DNS Server

DNS address provided by your ISP.

7.6 Network Setting – PPPoE

DHCP / PPPoE / Static IP	
<input type="radio"/> DHCP <input checked="" type="radio"/> PPPoE <input type="radio"/> Static IP	
PPPoE ID	<input type="text"/>
PPPoE Password	<input type="text"/>
DNS Setting	
DNS Server	<input type="text" value="61.63.82.1"/>

Choose PPPoE if your ISP uses PPPoE. Most DSL users use PPPoE.

PPPoE ID PPPoE ID/username provided by your ISP.

PPPoE Password PPPoE password.

DNS Server DNS address provided by your ISP.

7.7 Network Setting – Static IP

DHCP / PPPoE / Static IP	
<input type="radio"/> DHCP <input type="radio"/> PPPoE <input checked="" type="radio"/> Static IP	
IP Address	<input type="text" value="61.63.83.19"/>
Router IP	<input type="text" value="61.63.83.254"/>
Subnet Mask	<input type="text" value="255.255.254.0"/>
DNS Setting	
DNS Server	<input type="text" value="61.63.82.1"/>

Choose Static IP network setting if all Wide Area Network IP is provided to you by your ISP.

IP Address IP address assigned to you by your ISP.

Router IP Router IP address.

Subnet Mask Subnet mask address.

DNS Server DNS server address provided by your ISP.

NOTE: RESTART the system for new settings to take effect after you modify the IP address.

7.8 SIP Setting – SIP Phone Setting, Registrar and Outbound Proxy Server

SIP Phone Setting	
SIP Phone Port Number	5060
Registrar Server	
Registrar Server Domain Name/IP Address	
Registrar Server Port Number	5060
Authentication Expire Time	3600 sec. (Default: 3600 sec.)
Outbound Proxy Server	
Outbound Proxy Domain Name/IP Address	
Outbound Proxy Port Number	5060

Session Initiation Protocol (SIP) is the most popular Voice over IP standard. It enables two or more people to make phone calls, share multimedia and make multimedia conference over the internet. Please have an administrator setup these settings for you or obtain this information from your SIP service provider.

SIP Phone Port Number

SIP phone port number.

Registrar Server Domain Name/IP Address

Registrar server domain name or IP address.

Registrar Server Port Number

Registrar server port number.

Authentication Expire Time

The time that the phone waits to connect to the SIP server after the user dialed a number. If still not connected, the phone will disconnect and redial.

Outbound Proxy Domain Name/IP Address

Outbound proxy domain name or IP address.

Outbound Proxy Port Number

Outbound proxy port number.

7.9 SIP Setting – Others

Others	
Session Timer	1800 <small>sec.[30 - 99999]</small>
Media Port	41000 <small>[1024 - 65535]</small>
Prack	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
Session Refresher	<input checked="" type="radio"/> None <input type="radio"/> UAC <input type="radio"/> UAS
Session Timer Method	<input checked="" type="radio"/> Invite <input type="radio"/> Update
UDP/TCP	<input checked="" type="radio"/> UDP <input type="radio"/> TCP

This section is for network administrators.

Session Timer

The time interval in which the phone periodically refresh SIP sessions by sending repeated INVITE requests. These INVITE requests allow the user agent or proxies to determine the status of the SIP session.

Media Port

Real-time Transport Protocol port number. Provides end-to-end transfer of data with real-time characteristics, such as streaming audio and video.

Prack

Prack ensures that media information is exchanged and that network checks before connecting the call. Select Enable for a more reliable connection.

Session Refresher

Select None to disable SIP session timer support.

Select UAC to initiate SIP request.

Select UAS to receive SIP request and then return a response.

Session Timer Method

Select SIP request method. Default method is Invite.

UDP/TCP

Select SIP signal transmission method. Default method is UDP.

7.10 SIP Account Settings

SIP Account Setting	
Default Account	Account 1 ▼
Account 1 Setting	
Account Active	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
Display Name	Michael
SIP User Name	608
Authentication User Name	608
Authentication Password	608
Register Status	Register
Account 2 Setting	
Account Active	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
Display Name	
SIP User Name	
Authentication User Name	
Authentication Password	
Register Status	UnRegister

You may have up to 4 accounts. i.e., the IP phone can receive up to four different phone numbers.

Default Account

When you dial a number, the default account is used to dial. User Name of default account is displayed on the receiver's IP phone.

Account Active

Enable or disable this account.

Display Name

Display name on the IP phone.

SIP User Name

User name.

Authentication User Name

Name used to access SIP server.

Authentication Password

User password to access SIP server.

Register Status

Displays if the current phone is registered or unregistered with SIP server.

7.11 STUN Setting – STUN Server Setting, UPnP Setting

STUN Server Setting	
STUN	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
STUN Domain Name/IP Address	<input type="text"/>
UPnP Setting	
UPnP	<input checked="" type="radio"/> Disable <input type="radio"/> Enable

STUN

Simple Traversal of User Datagram Protocol through Network Address Translators is a protocol that allows applications to determine the types of NATs and firewalls are in between them and the internet. STUN also provides the ability for applications to determine the public IP addresses allocated to them by the NAT.

STUN Domain Name/IP Address

Enter STUN domain name or IP address if STUN is enabled.

UPnP

Enable or disable universal plug and play. Some NAT supports UPnP so STUN is not required and must be disabled.

7.12 Voice Setting and QoS

Voice Setting	
Codec (Priority 1)	G.729A ▼
Codec (Priority 2)	G.723.1 ▼
Codec (Priority 3)	G.711 u-law ▼
Codec (Priority 4)	non-used ▼
RTP Packet Length	G.711 μ -Law 20ms ▼
	G.711 A-Law 20ms ▼
	G.729A 20ms ▼
	G.723.1 30ms ▼
VAD	<input type="radio"/> On <input checked="" type="radio"/> Off
DTMF Method	<input type="radio"/> Out Band <input checked="" type="radio"/> In Band <input type="radio"/> SIP INFO
QoS	
Voice TOS	5 [0 - 7]
Enable/Disable VLAN might Caused Network Connection Problem	
VLAN	<input checked="" type="radio"/> Disable <input type="radio"/> Enable

- Codec** Voice Compression Algorithm priority settings. Select from the most used codec to the least used codec.
- RTP Packet Length** Real-Time Transfer Protocol (RTP) packet length.
- VAD** VAD detects voice activity and adjusts the signal to a target power level. It ensures that background noise or echo does not get amplified to the target power level.
- DTMF Method** Select the tone method for IP phone.
- Voice TOS** Sets the type of service for this Internet datagram.
- VLAN** Enable or disable virtual LAN.
- VLAN Priority** Set the virtual LAN Priority.
- VLAN ID** Virtual LAN ID.

7.13 Phone Settings – Phone Setting

Phone Setting	
Tone Setting	America <input type="button" value="v"/>
Ringer Type	RingType 2 <input type="button" value="v"/>
Hold Tone	<input checked="" type="radio"/> Melody <input type="radio"/> Tone
Do Not Disturb	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
Call Waiting	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
Anonymous Call	<input checked="" type="radio"/> Disable <input type="radio"/> Full URI <input type="radio"/> Display Name
Anonymous Call Reject	<input checked="" type="radio"/> Disable <input type="radio"/> Enable
Call Forward	<input type="checkbox"/> No Answer <input type="text"/>
	<input type="checkbox"/> Busy <input type="text"/>
	<input type="checkbox"/> Unconditional <input type="text"/>
	<input type="text"/>

Recall you can only enable or disable call forwarding from the IP phone MENU key. With the web-browser, you can enter the forwarded phone numbers in the Phone Setting menu.

Tone Setting

Select the tone for particular country

Ringer Type

Select the type of ring (1 to 4).

Hold Tone

Select melody or tone when HOLD key is pressed.

Do Not Disturb

Reject all incoming calls.

Call Waiting

Enable or disable call waiting.

Anonymous Call

If DISABLE is selected, full URI and name are sent to the receiver's phone when the user makes a phone call. The URI and name of the caller are displayed on the receiver's phone.

When Full URI is selected, only user name is displayed on the receiver's phone when the user makes a phone call.

When Display Name is selected, only name is displayed on the receiver's phone when the user makes a phone call.

Anonymous Call Reject

Select Enable to reject anonymous calls.

Call Forward

Click No Answer to enable call forward to another number when no one answers the phone after 180s (default). The timer can be changed from 0-600s. Refer to section 7.14 to change the timer.

Click Busy to enable call forward to another number when user is busy on the phone.

Click Unconditional to transfer all incoming calls to another number.

Enter the call forward number on the text box.

7.14 Phone Setting – Timer

Timer	
NTP Recycle Timer	<input type="text" value="1"/> hour [1 - 24] Network Time Adjustment Period
Inter Digit Timer	<input type="text" value="4"/> sec. [0 - 60] 0: Disable
Originating Not Accept Timer	<input type="text" value="180"/> sec. [0 - 600] 0: Disable
Incoming No Answer Timer	<input type="text" value="180"/> sec. [0 - 600] 0: Disable
Hold Recall Timer	<input type="text" value="180"/> sec. [0 - 600] 0: Disable
Auto Speaker Off Timer	<input type="text" value="30"/> sec. [0 - 600] 0: Disable

NTP Recycle

NTP recycle time.

Inter Digit

The time interval that the IP phone waits to detect the end of DTMF digits. No more digits are accepted after this period and the phone begins to dial.

Originating Not Accept

The time interval that the caller's phone waits to establish a call. If the receiver fails to answer the phone during this

time interval, the caller's phone will automatically disconnect.

Incoming No Answer

The time interval that the receiver's phone will ring. If the receiver fails to answer the phone during this time interval, the phone will automatically disconnect.

Hold Recall

The time interval that the caller is put on hold before the phone automatically disconnect.

Auto Speaker Off

The time interval that the speaker phone is on before turning off automatically (due to inactivity).

7.15 Call Tracing Log

No.	Trace Log
000	!0 FW Version: 02.08.00
001	!2 ReadSetupInfo: 0.
002	!6 Basic number for random: (829)
003	!0 Language:(0)
004	!0 Remote Config Task Ruming.
005	!6 WriteSetupInfo: 0. len(00000A2C)
006	\$
007	!1 Err: invalid IP.

Call Tracing Log keeps a record of all the phone activities. This log is used by our engineers to troubleshoot hardware problems.

7.16 Phone Book

Record No: 2
Maximum Record: 200

Name: Maximum 31 Char.
Number: Maximum 63 Char.

Phone Book Setting	
Name	Number
KEVIN LAI	709
MIKE LAI	708

Phonebook menu allows the user to add, modify and delete phone numbers. To add, type in the name and number then click NEW to add. To modify/delete, select the name from the list and click modify/delete.

Name Name that you would like to add.

Number Phone number that corresponds to the name.

7.17 Speed Dial

Speed Dial Setting (Maximum 63 Char.)			
Number 00	<input type="text"/>	Number 01	<input type="text"/>
Number 02	<input type="text"/>	Number 03	<input type="text"/>
Number 04	<input type="text"/>	Number 05	<input type="text"/>
Number 06	<input type="text"/>	Number 07	<input type="text"/>
Number 08	<input type="text"/>	Number 09	<input type="text"/>

Speed dial numbers can be accessed from the IP phone. Refer to section 8.2 for speed dial info.

Number 0x Speed dial phone number. 0x is the speed dial number.

7.18 Restart System

Press [Restart] Button, IP Phone system will reboot!

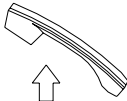

Restart

Click **Restart** to update all the modifications and reboot the system.

8.0 OPERATING THE PHONE

8.1 Dialing


8.1.1 IP Address

1 . Lift handset  or press **SPEAKER** button 

2 . Dial IP address.

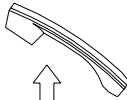

For example: dialing **192.168.0.1**



3 . Press **OK**  or wait until the timer expires to dial.

8.1.2 SIP Number


Note: You have to register with SIP server to use SIP number.

1 . Lift handset  or press **SPEAKER** button 

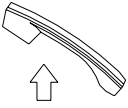

2 . Dial SIP number.

For example: dialing **1866**



3 . Press **OK**  or wait until the timer expires.

8.2 Speed Dialing

- 1 . Lift handset  or press **SPEAKER** button 
- 2 . Dial Speed Dial number.

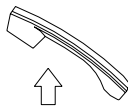

For example: dialing speed dial number **08**



8.3 Answering a Phone Call

Note: The CANCEL key may be used to reject a call.

When phone rings:

- 1 . Lift handset  or press **SPEAKER** button 
- to begin conversation.

8.4 Switching to Another Line

While having a conversation:

- 1 . Press **Hold** to switch to another line.

8.5 Mute



Note: While mute is activated, sound from the caller can be heard from your speaker but your sound can't be heard by the caller.

While having a conversation:

- 1 . Press **Mute**  . You may press **Mute** key again to resume conversation.

8.6 Call Transfer

While having a conversation:

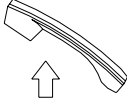


- 1 . Press **Hold**  to put the person on the other line on hold.
- 2 . Dial the IP address or the extension number where you like the call to be transferred.
- 3 . Press **Transfer**  to transfer the call.

8.7 Redial




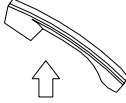

Note: To return to idle mode, press **CANCEL** key



8.7.1 Last Dialed Number

- 1 . Lift handset  or press **SPEAKER** button 
- 2 . Press **Redial**  to dial the last dialed number.

8.7.2 Through Call History

- 1 . Press **Redial**  . Do not lift the handset when you press Redial.
- 2 . Press **Redial**  again to cycle through the dialed, missed and received calls.
- 3 . Press **DOWN** key  to scroll down the dialed, missed or received lists until the number is displayed on the screen.
- 4 . Pickup the handset  or press **OK** 

8.8 On Hold

Note: To [transfer a call while on hold](#), press the **TRANS** key. Dial the extension/phone number and press the **TRANS** key again to transfer the call.

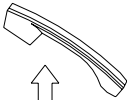
While having a conversation:

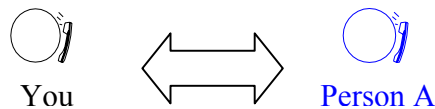
1. Press **HOLD**  (Press **HOLD** again to resume conversation)


8.9 Call Forward

Please refer to [IP Phone Setup and Web Browser Configuration](#) section to setup call forwarding.

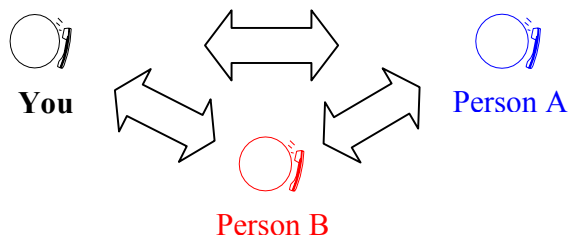
8.10 Three Way Conference

1. Pick up the handset  and call **Person A**.






2. After **Person A** pick up the phone, press **Hold** key  to place **Person A** on hold.
3. Dial the extension or phone number of **Person B** and wait until **Person B** picks up the phone.

4. Press **Conference** key  to begin 3-way conference.







9.0 USING THE PHONEBOOK








9.1 Dialing from the Phonebook

- 1 . Press the **PHONE BOOK** key  to access the phone book.
- 2 . Press  to scroll down the list until the name is displayed on the screen.
- 3 . Press **OK**  to dial.






9.2 Storing a Number

- 1 . Press and hold the **PHONE BOOK** key  until “**Name :**” is displayed on the screen.
- 2 . Enter a name then press **OK** .
- 3 . Enter the number that corresponds to the name and press **OK** .
- 4 . Press **OK**  again to save the phonebook.
- 5 . Repeat Step 1 to 4 to store another phone number.

9.3 Editing a Number

- 1 . Press the **PHONE BOOK** key  to access the phonebook.
- 2 . Press  until the name is displayed on the screen.
- 3 . Press the **PHONE BOOK** key  again.
- 4 . Select “**Edit**” and press **OK**  to edit.
- 5 . Enter a new name and press **OK** .
- 6 . Enter the new phone number and press **OK** .
- 7 . Press **OK**  to save and override the previous name and phone number.

9.4 Deleting a Number

- 1 . Press the **PHONE BOOK** key  to access the phonebook.
- 2 . Press  until the name you want to delete is selected.
- 3 . Press the **PHONE BOOK** key  again.
- 4 . Select “**Delete**” and press **OK**  to delete.
- 5 . Press **OK**  again to save the new list on the phonebook.

10.0 Troubleshooting

The following troubleshooting information can be used to help solve most common problems.

QUESTION	RECOMMENDED ACTION
There are no DIAL tone	1. Check if there are any loose connections.
Nothing is displayed on the LCD screen	1. Check if power cord is connected properly. 2. Check if there is 120V AC coming from the power outlet.
How to update Firmware?	1. ATC IP Phone automatically updates firmware when it powers up (while connected to the internet).
Why can't I dial my friend's SIP number?	1. Check Registrar Server Domain Name/IP address and Outbound Proxy Domain Name/IP Address (under SIP Settings in Configuration Menu). Make sure you have the right Name or IP Address. 2. Check the LCD display on your phone to see if there is a name or number displayed on the screen. If the name or number is not displayed, use a web browser and access the configuration menu. Make sure that the Registrar Server Domain Name/IP Address is correct. 3. Check the register status under SIP Account Settings in the configuration menu (from web browser). If your status is unregistered, it means you do not have a SIP account. Contact your SIP service provider to get an account.
Why isn't my firmware updating?	1. Your IP phone automatically detects for new firmware when you unplug the power. If new version is available the phone will automatically update the firmware. 2. Check if FTP address is correct.

	3. Check with your supplier if firmware filename is correct.
I accidentally set DSL to enable and now the phone does not boot up	1. Unplug the power cord from the IP phone. Wait 2 seconds and plug the power cord back in the IP phone. Press and hold MENU key. The system should bypass boot up and go straight into phone setup menu. Modify the phone setting and make sure you save it before you exit.
Why do I get “Can’t Upgrade Now” screen when I click [Submit] in the configuration menu?	1. Make sure you exit setting mode (phonebook, menu, speed dial...) before you click [Submit] in the configuration menu.

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