PATTON Electronics Co.

SL4050/B12/E

SmartLink Series 12-Line VoIP SIP Phone

SL4050/B2/E

SmartLink Series 2-Line VoIP SIP Phone

Getting Started Guide

TImportant

This is a Class B device and is intended for use in a light industrial or residential environment. It is not intended nor approved for use in an industrial environment. The Model SL4050 phones are not approved for, and are not intended for, direct connection to the Public Switched Telephone Network (PSTN).

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This warranty is limited to defects in workmanship or materials, and does not cover customer damage, abuse or unauthorized modification. If the product fails to perform as warranted, your sole recourse shall be repair or replacement as described above. Under no condition shall Patton Electronics be liable for any damages incurred by the use of this product. These damages include, but are not limited to, the following: lost profits, lost savings and incidental or consequential damages arising from the use of or inability to use this product. Patton Electronics specifically disclaims all other warranties, expressed or implied, and the installation or use of this product shall be deemed an acceptance of these terms by the user.

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About this guide

This guide describes using the SmartLink 4050/B12 12-Line VoIP SIP Telephone and SmartLink 4050/B2 2-Line VoIP SIP Telephone.

Audience

This guide is intended for the following users:

- Operators
- Installers
- Maintenance technicians

Structure

This guide contains the following chapters and appendices:

- Chapter 1 on page 13 provides information about the SIP phone
- Chapter 2 on page 19 provides installation procedures
- Chapter 5 on page 38 provides procedures for configuring the SIP Phone
- Chapter 3 on page 31 describes how to operate the SIP Phone
- Chapter 4 on page 35 describes how to use the Phone Book feature
- Chapter 6 on page 63 contains information on troubleshooting problems with the SIP Phone
- Chapter 7 on page 66 contains information on contacting Patton technical support for assistance
- Appendix A on page 69 contains compliance information for the SIP Phone
- Appendix B on page 72 contains specifications for the SIP Phone

For best results, read the contents of this guide *before* you install the SIP Phone.

Precautions

Notes, cautions, and warnings, which have the following meanings, are used throughout this guide to help you become aware of potential problems. ***Warnings*** are intended to prevent safety hazards that could result in personal injury. ***Cautions*** are intended to prevent situations that could result in property damage or impaired functioning.

Note A note presents additional information or interesting sidelights.

CAUTION

The alert symbol and CAUTION heading indicate a potential hazard. Strictly follow the instructions to avoid property damage.

WARNING

The shock hazard symbol and WARNING heading indicate a potential electric shock hazard. Strictly follow the warning instructions to avoid injury caused by electric shock.

Safety when working with electricity

WARNING

- **This device contains no user serviceable parts. The equipment shall be returned to Patton Electronics for repairs, or repaired by qualified service personnel.**
- **The external power adapter shall be a listed Limited Power Source. Ensure that the power cable used meets all applicable standards for the country in which it is to be installed, and that it is connected to a wall outlet which has earth ground. The mains outlet that is utilized to power the devise shall be within 10 feet (3 meters) of the device, shall be easily accessible, and protected by a circuit breaker.**
- **Do not work on the system or connect or disconnect cables during periods of lightning activity.**

In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

General observations

- Clean the case with a soft slightly moist anti-static cloth
- Place the unit on a flat surface and ensure free air circulation
- Avoid exposing the unit to direct sunlight and other heat sources
- Protect the unit from moisture, vapors, and corrosive liquids

Typographical conventions used in this document

This section describes the typographical conventions and terms used in this guide.

General conventions

The procedures described in this manual use the following text conventions:

Table 1. General conventions

Convention	Meaning
Garamond blue type	Indicates a cross-reference hyperlink that points to a figure, graphic, table, or section heading. Clicking on the hyperlink jumps you to the reference. When you have finished reviewing the reference, click on the **Go to Previous View** button in the Adobe® Acrobat® Reader toolbar to return to your starting point.
Garamond bold type	Indicates the names of command buttons that execute an action.
< >	Angle brackets indicate function and keyboard keys, such as <SHIFT>, <CTRL>, <C>, and so on.

Chapter 1 **General information**

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SmartLink 4050 Series SIP Phones overview

Voice over IP (also known as *Internet telephony*) is a technology that enables anyone to make a telephone call over the Internet. This is a quick user guide for the SmartLink 4050 Series SIP Phones. It will help you configure the telephone and have it ready to run within a few minutes.

The following items are included in the SmartLink 4050/B12 and SmartLink 4050/B2 packaging. Contact your supplier immediately if an item is missing.

SmartLink 4050/B12 VoIP SIP Phone

SmartLink 4050/B2 VoIP SIP Phone

Ethernet cable, 5-foot (1.5-meter), Qty: 1

SmartLink documentation CD-ROM

Power Adaptor

Wall/Desk mounting plate

Overview of SL4050/B12/E key functions

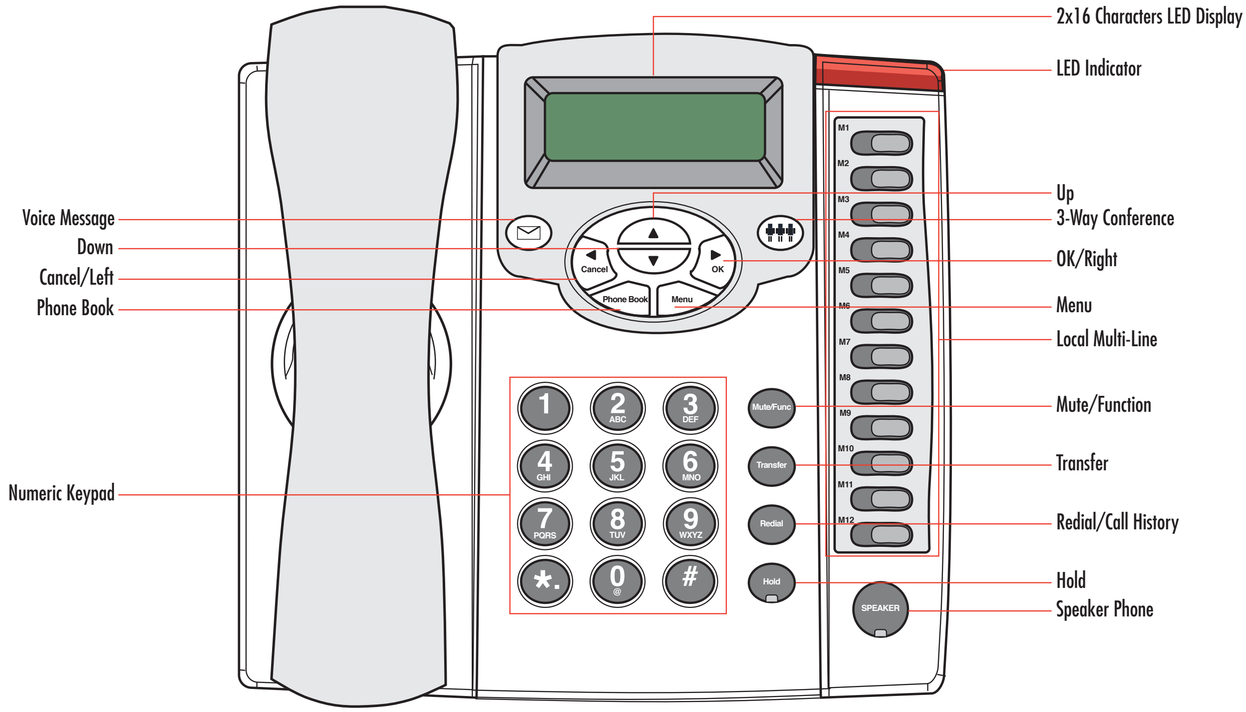

Figure 1. SmartLink 4050/B12 SIP Phone controls and indicators

Table 2. Summary of SL4050/B12/E key functions

Item	Description
2 x 16 LCD Display	Displays menu, time, clock, name, phone number, call status
LED Indicator	Indicates that phone is currently in use or ringing
Up	Cycle through the phone menu, adjust volume
3-Way Conference	Enable 3-way conference
OK/Right	Confirm setting change, exit menu, dial, save changes
Menu	Access the phone menu
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
Transfer	Transfer the person you are currently having a conversation to another line

Table 2. Summary of SL4050/B12/E key functions (Continued)

Item	Description
Redial/Call History	Redial last dialed number, access redial menu
Hold	Place the person on the other line on hold, answer call waiting
Speaker Phone	Enable user to use the phone without using the handset
Voice Message	Check voice message
Down	Cycle through the phone menu, adjust volume
Cancel/Left	Deny changes, cancel phone calls, ignore phone calls, backspace
Phone Book	Access the phonebook
Numeric Keypad	Input IP/phone number/alphabet characters
Local Multiline	Switch to different lines

Overview of SL4050/B2/E key functions

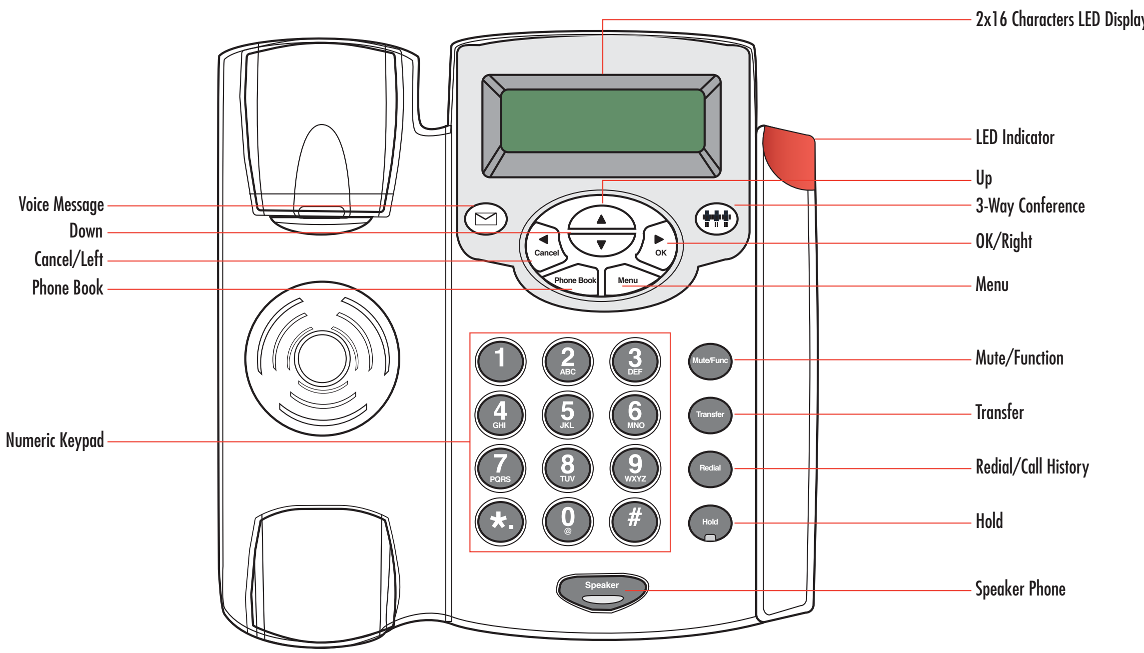

Figure 2. SmartLink 4050/B2 SIP Phone controls and indicators

Table 3. Summary of SL4050/B2/E key functions

Item	Description
2 x 16 LCD Display	Displays menu, time, clock, name, phone number, call status
LED Indicator	Indicates that phone is currently in use or ringing
Up	Cycle through the phone menu, adjust volume
3-Way Conference	Enable 3-way conference
OK/Right	Confirm setting change, exit menu, dial, save changes
Menu	Access the phone menu
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

Table 3. Summary of SL4050/B2/E key functions (Continued)

Item	Description
Transfer	Transfer the person you are currently having a conversation to another line
Redial/Call History	Redial last dialed number, access redial menu
Hold	Place the person on the other line on hold, answer call waiting
Speaker Phone	Enable user to use the phone without using the handset
Voice Message	Check voice message
Down	Cycle through the phone menu, adjust volume
Cancel/Left	Deny changes, cancel phone calls, ignore phone calls, backspace
Phone Book	Access the phonebook

Numeric Keypad Defintions

You can use alphanumeric characters to enter details into the Phone Book, to create text and e-mail messages. The table below shows the characters that you can enter in the different text modes.

Table 4. Keypad Definitions

	Text Mode			Text Mode	
Key	**Normal (ABC)**	**Numeric (0-9)**	**Key**	**Normal (ABC)**	**Numeric (0-9)**
1		**1**	7 PQRS	**pqrsPQRS**	**7**
2 ABC	**abcABC**	**2**	8 TUV	**tuvTUV**	**8**
3 DEF	**defDEF**	**3**	9 WXYZ	**wxyzWXYZ**	**9**
4 GHI	**ghiGHI**	**4**	0 @	**@ . _ - * # ( ) % & + / $,**	**0**
5 JKL	**jklJKL**	**5**	*.	**.**	*****
6 MNO	**mnoMNO**	**6**	#		**#**

In Normal and Numeric modes, each time you press in quick succession, the next character available is displayed. When you do not press a key for more than one second, the current character will be selected and the cursor will move to the right for the next selection.

For example, to enter "c", press 2 ABC four times in quick succession. To enter the displayed character, release the key or press another key.

Chapter 2 **Installing the SmartLink SIP Phone**

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Installing the VoIP SIP phone

CAUTION

The interconnecting cables shall be acceptable for external use and shall be rated for the proper application with respect to voltage, current, anticipated temperature, flammability, and mechanical serviceability.

1. Plug one end of the Ethernet cable included with the VoIP SIP phone into the LAN port on the SIP phone (see figure 3 for SL4050/B12/E or figure 4 on page 21 for SL4050/B2/E). Plug the other end of the cable into the xDSL modem or cable modem (or into an optional router or hub).

WARNING

Do not work on the system or connect or disconnect cables during periods of lightning activity.

2. If you will not be connecting a PC to the phone, go to step 3. Otherwise, connect an Ethernet cable into the WAN port of the SIP phone (see figure 3 for SL4050/B12/E or figure 4 on page 21 for SL4050/B2/E). Plug the other end of the cable into the Ethernet port on the PC.

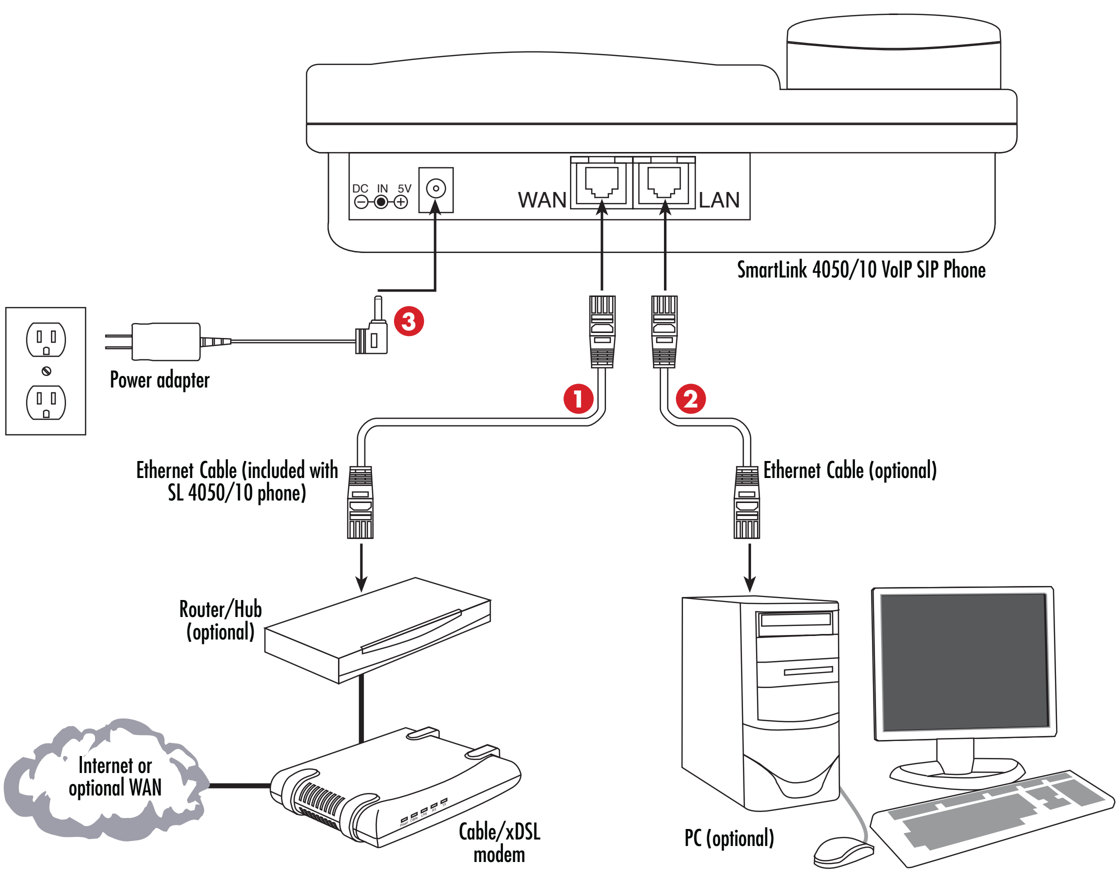

Figure 3. Connecting the SL4050/B12/E SIP Phone

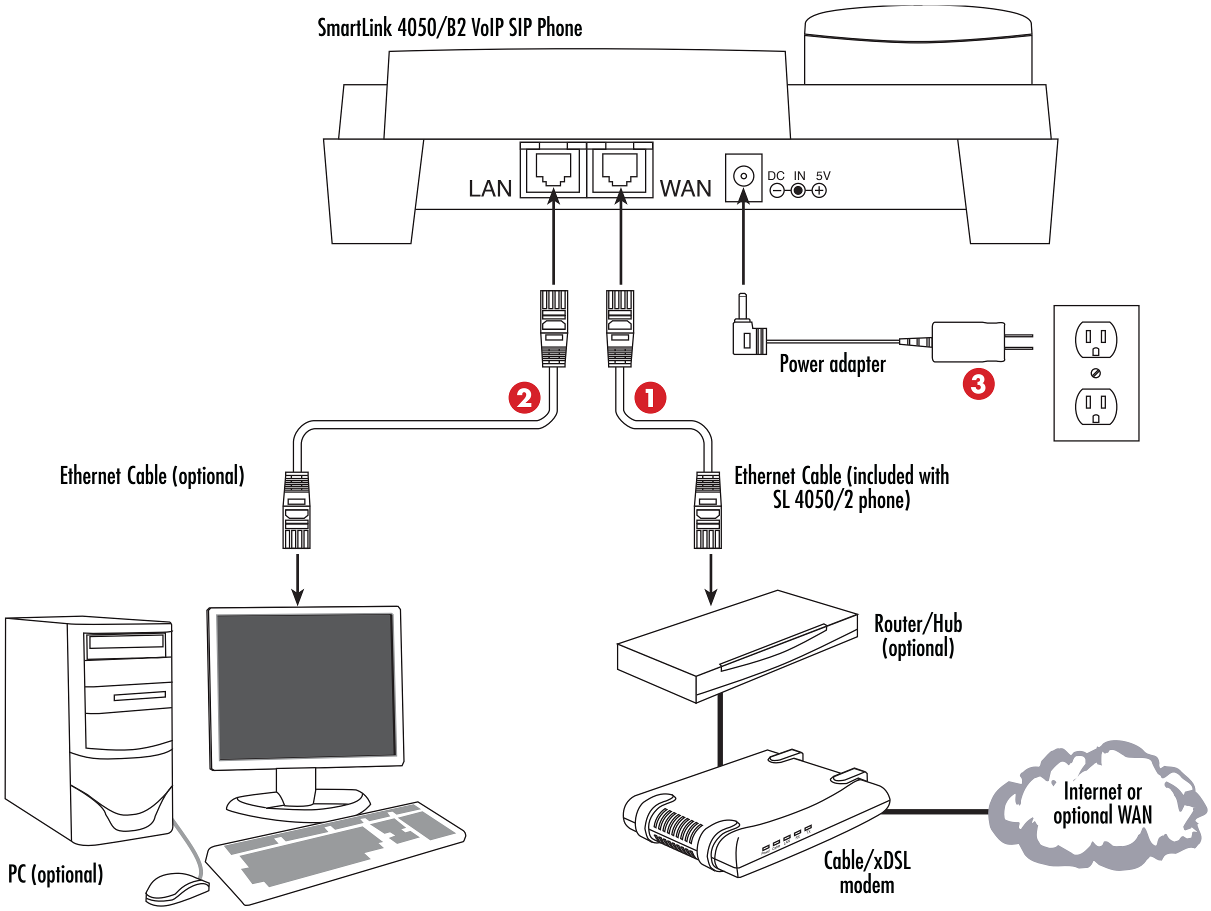

Figure 4. Connecting the SL4050/B2/E SIP Phone

3. Plug the power adapter barrel connector into the power connector on the SIP phone (see figure 3 on page 20 for SL4050/B12/E or figure 4 for SL4050/B2/E). Plug the other end of the power adapter into an AC electrical outlet.

Setting up the VoIP SIP phone

Menu summary

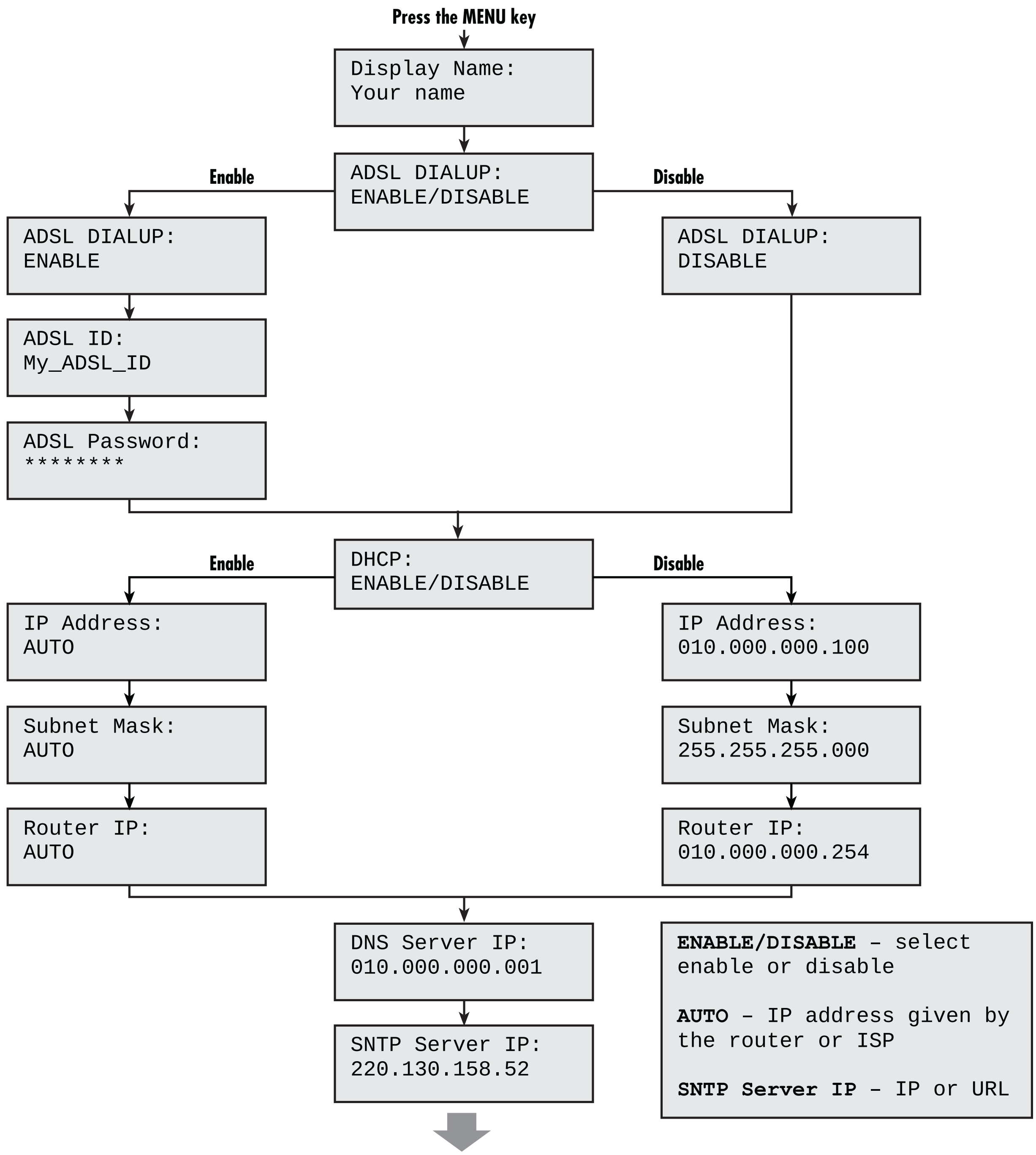

Figure 5. Menu summary, page 1 of 2

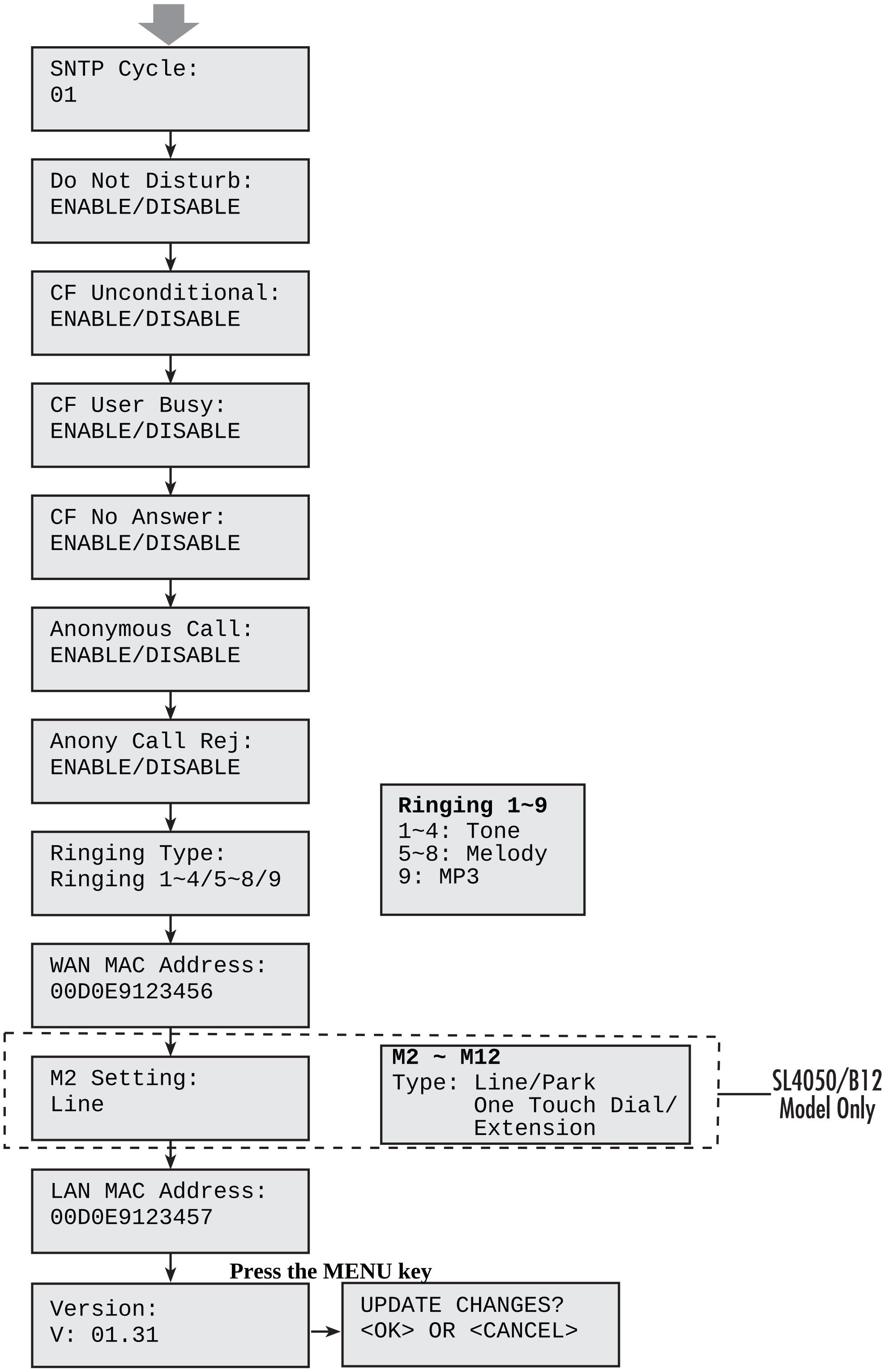

Figure 6. Menu summary, page 2 of 2

Note You can stop the setup process at any time by pressing **MENU** + **OK** to save any changes and exit, or by pressing **MENU** + **CANCEL** to quit without saving. The phone will automatically time-out and exit the menu screen if there are no inputs from the user.

Use Cancel and OK keys on the control pad to select *ENABLE* or *DISABLE*.

The Cancel key can also be used as a backspace key to delete characters.

Display Name

1. Press Menu
2. Use the numeric keypad to enter the display name

```
Display Name:
Your name
```

Note To type text characters, press the appropriate key on the numeric keypad (see figure 1 on page 15). For example, to type a "z" press the *9* key until the lowercase *z* appears—the displayed sequence would be:
9 W X Y Z w x y z.

ADSL DIalup

Some Internet Service Providers (mostly ADSL) use 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 ▼
2. Use OK to select *ENABLE*

```
ADSL DIALUP:
ENABLE
```

Set up ADSL ID

1. Press ▼
2. Enter the ADSL ID

```
ADSL ID:
My_ID
```

Set up ADSL password

1. Press
2. Enter the ADSL password

```
ADSL Password:
********
```

DISABLE ADSL dialup

1. Press
2. Use OK to select *DISABLE*

```
ADSL DIALUP:
DISABLE
```

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
2. Use Cancel or OK to set DHCP to *ENABLE*

```
ADSL DIALUP:
ENABLE
```

3. Press . The IP address is automatically acquired

```
IP Address:
010.000.000.100
```

4. Press . The subnet mask is automatically acquired

```
Subnet Mask:
255.255.255.000
```

5. Press . The router IP address is automatically acquired

```
Router IP:
010.000.000.254
```

DISABLE DHCP

1. Press
2. Use Cancel or OK to set DHCP to *DISABLE*

```
DHCP:
DISABLE
```

3. Press . Use the numeric keypad to enter the IP address

```
IP Address:
010.000.000.100
```

4. Press  . Use the numeric keypad to enter the subnet mask

```
Subnet Mask:
255.255.255.000
```

5. Press  . Use the numeric keypad to enter the router IP address

```
Router IP:
010.000.000.254
```

DNS Server IP

The domain name system (DNS) is the way that Internet domain names are located and translated into Internet Protocol addresses. There is probably a DNS server within close geographic proximity to your ISP that maps the domain names in your Internet requests or forwards them to other servers in the Internet.

1. Press  .
2. Use the numeric keypad to enter the DNS server IP address

```
DNS Server IP:
010.000.000.001
```

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. The SNTP Server IP address can be URL or IP address format.

1. Press  .
2. Use the numeric keypad to enter the SNTP server IP or URL address

```
SNTP Server IP:
220.130.158.52
```

Do Not Disturb

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

1. Press  .
2. Use 

or 

to select *ENABLE* or *DISABLE*

```
Do Not Disturb:
ENABLE/DISABLE
```

Call forwarding

CF (call forward) Unconditional

Enable CF Unconditional to forward all the incoming calls to another number. Otherwise set to disable.

Note You will need to use a web browser to input the forwarded phone number. Refer to chapter 5, "Using the configuration menu" on page 38 for more information on call forwarding.

1. Press  .
2. Use 

 or 

 to select *ENABLE* or *DISABLE*

```
CF Unconditional:
ENABLE/DISABLE
```

3. Press  to enter the number where the call will be forwarded.

```
Ring Type:
Ringing 1/2/3/4/5/6/
```

CF (call forward) User Busy

Forward all the incoming calls to another number when the phone is in use with another call.

1. Press  .
2. Use 

 or 

 to select *ENABLE* or *DISABLE*

```
CF User Busy:
ENABLE/DISABLE
```

3. Press  to enter the number where the call will be forwarded.

```
M2 Setting: (M2~M12)
Line/Park/One Touch
```

CF (call forward) No Answer

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

1. Press  .
2. Use 

 or 

 to select *ENABLE* or *DISABLE*

```
CF No Answer:
ENABLE/DISABLE
```

3. Press  to enter the number where the call will be forwarded.

```
Time Format:
24Hours
```

Anonymous Call

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

1. Press .
2. Use or to select *ENABLE* or *DISABLE*

```
Anonymous Call:
ENABLE/DISABLE
```

Anony Call Rej (anonymous call rejection)

Reject any anonymous incoming calls.

1. Press .
2. Use or to select *ENABLE* or *DISABLE*

```
Anony Call Rej:
ENABLE/DISABLE
```

Ringing Type

Select the ring tone. There are nine possible ring tones.

1. Press .
2. Use or to select *ENABLE* or *DISABLE*

```
Ring Type:
Ringing 1/2/3/4/5/6/7/8/9
```

Note At this point, you may press to exit the menu and press to *SAVE*.. The next two sections explain how to obtain the MAC address and firmware version.

M2~M12 Setting (Model SL4050/B12/E only)

Select the functionality of the line keys. There are four types of settings: Line / Park / One-Touch Dial / Extension.

1. Press .
2. Use or to select the functionality.

```
M2 Setting: (M2~M12)
Line/Park/One Touch Dial/Extension
```

3. Press to enter the number (i.e. number of the parking area, destination of One-Touch Dial, or the monitored extension.

MAC Address

The *MAC Address* menu displays the MAC address, which cannot be modified.

1. Press .
2. The MAC address is displayed on the screen.

```
WAN MAC Address:
000FC9017D4A
```

```
LAN MAC Address:
000FC9017D4B
```

Version

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

1. Press .
2. The firmware version is displayed on the screen.

```
Version:
V: 01.31
```

Language Selection

The VoIP SIP phone supports two languages: English and Japanese. (The language change will effect the web interface, as well).

1. Press Mute/Func followed by 1

```
Language:
English
```

2. Use or to select the preferred language.

Time Format

You may select a 12-hour or 24-hour time format.

1. Press Mute/Func followed by 2 ABC
2. Use or to select the time format.

```
Time Format:
24Hours
```

3. Press OK when done.

Volume Adjustment

Ringer Volume

While the handset is in place, press ▲ to increase the ringer volume or ▼ to decrease the ringer volume.

Speaker Volume

1. While the handset is in place, press SPEAKER
2. Press ▲ to increase the speaker volume or ▼ to decrease the speaker volume.

Handset Volume

Pick up the handset and press ▲ to increase the volume or ▼ to decrease the volume.

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Dialing an IP address

1. Lift the handset or press the **SPEAKER** button.
2. Dial an IP address. For example, to dial *192.168.0.1* press:

   1 9 2 *. 1 6 8 *. 0 *. 1

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

Dialing a SIP number

Note You must register with a SIP server before using a SIP number.

1. Lift the handset or press the **SPEAKER** button.
2. Dial a SIP number. For example, to dial *1866* press

   1 8 6 6

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

Speed Dialing

1. Lift the handset or press the **SPEAKER** button.
2. Dial "*" then the speed dial number. For example, to dial *08,* press:

   *. 0 8

Answering a phone call

Note The Cancel key may be used to reject a call.

Lift the handset or press the **SPEAKER** button to begin a conversation.

Switching to another line

While having a conversation, press and the line key to switch to another line.

Mute

Note While mute is activated, sounds the caller makes can be heard through your speaker but your sound cannot be heard by the caller.

While having a conversation, press . To resume the conversation, press again.

Call Transfer

While having a conversation:

1. *For Model SL4050/B12/E:* Press to put the person on the other line on hold.

   For Model SL4050/B2/E: Press to put the person on the other line on hold.

2. Dial the IP address or the number of the extension where you would like to transfer the call to.

3. Press to transfer the call.

Redial

Note To return to idle mode, press the key.

Last Dialed Number

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

2. Press to dial the last dialed number.

Through Call History

1. Press . Do not lift the handset when you press **Redial**.

2. Press again to cycle through the dialed, missed, and received calls.

3. Press to scroll down the dialed, missed, and received numbers until the desired number is displayed on the screen.

4. Lift the handset or press .

On Hold

Note To transfer a call while on hold, press the **Transfer** key. Dial the extension/phone number and press again to transfer the call.

While having a conversation, press the **Hold** button.

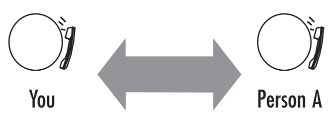
To resume the conversation, press again.

Call Forwarding

Refer to sections "Call forwarding" on page 27 to set up call forwarding.

Three-Way Conferencing

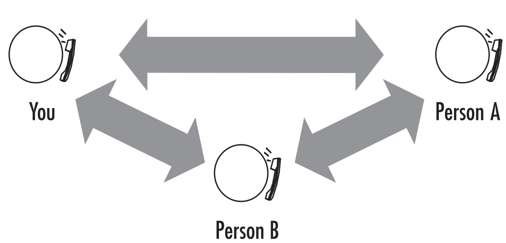
1. Lift the handset and call **Person A**.



2. *For Model SL4050/B12/E:* After **Person A** picks up the phone, press the Conference key to place **Person A** on hold.

   For Model SL4050/B2/E: After **Person A** picks up the phone, press the Hold button to place **Person A** on hold.

3. Dial the extension or phone number of **Person B**.
4. When **Person B** picks up the phone, press the Conference key to begin the 3-way conference.



Chapter 4 Using the Phone Book

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Storing a number in the Phone Book....36
Editing a number in the Phone Book....36
Deleting a Phone Book listing....37

Dialing from the Phone Book

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

Storing a number in the Phone Book

1. Press and hold the **PHONE BOOK** key until “**Name:**” displays on the screen.
2. Enter a name, then press .

Note To type text characters, press the appropriate key on the numeric keypad (see figure 1 on page 15). For example, to type a “z” press the *9* key until the lowercase *z* appears—the displayed sequence would be:
9 W X Y Z w x y z.

3. Enter the number that corresponds to the name, then press .
4. Select a ringer type from the nine options *(Tone: 1~4, Melody: 5~8, MP3: 9)*.
5. Press the **PHONE BOOK** key again to save the number into the phone book.
6. Repeat steps 1 through 5 to store additional phone numbers.

Editing a number in the Phone Book

1. Press the **PHONE BOOK** key to access the phone book.
2. Press to scroll down the list until the desired name is displayed on the screen.
3. Press the **PHONE BOOK** key again.
4. Select”**Edit**” to begin editing.
5. Enter a new name, then press .
6. Enter the new number, then press .
7. Select a new ringer type from the nine options *(Tone: 1~4, Melody: 5~8, MP3: 9)*.
8. Press the **PHONE BOOK** key to save and override the previous name and phone number.

Deleting a Phone Book listing

1. Press the **PHONE BOOK** key 

to access the phone book.
2. Press  to scroll down the list until the name you want to delete is selected.
3. Press the **PHONE BOOK** key 

again.
4. Select “**Delete**”.
5. Press 

to save the change to the Phone Book.

Chapter 5 **Using the configuration menu**

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Introduction

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

Accessing the configuration menu

1. Open a web browser (Internet Explorer, Netscape Navigator, or equivalent).
2. Type in the IP address of the phone followed by: **9999** (for example *http://192.168.1.1:9999*).

> **Note** The IP address is provided by your Internet service provider (ISP). If your ISP supports DHCP, you can obtain the IP address from your phone.
>
> Press 
>
> + 
>
> to get the IP address. You can also login from the LAN port with *http://192.168.15.1:9999*.

The login window displays (see Figure 7).

Figure 7. Login window

3. Enter a **User Name** and **Password**. Leave the user name and password blank if you are installing the phone for the first time.
4. Click **OK**.

Web login setting

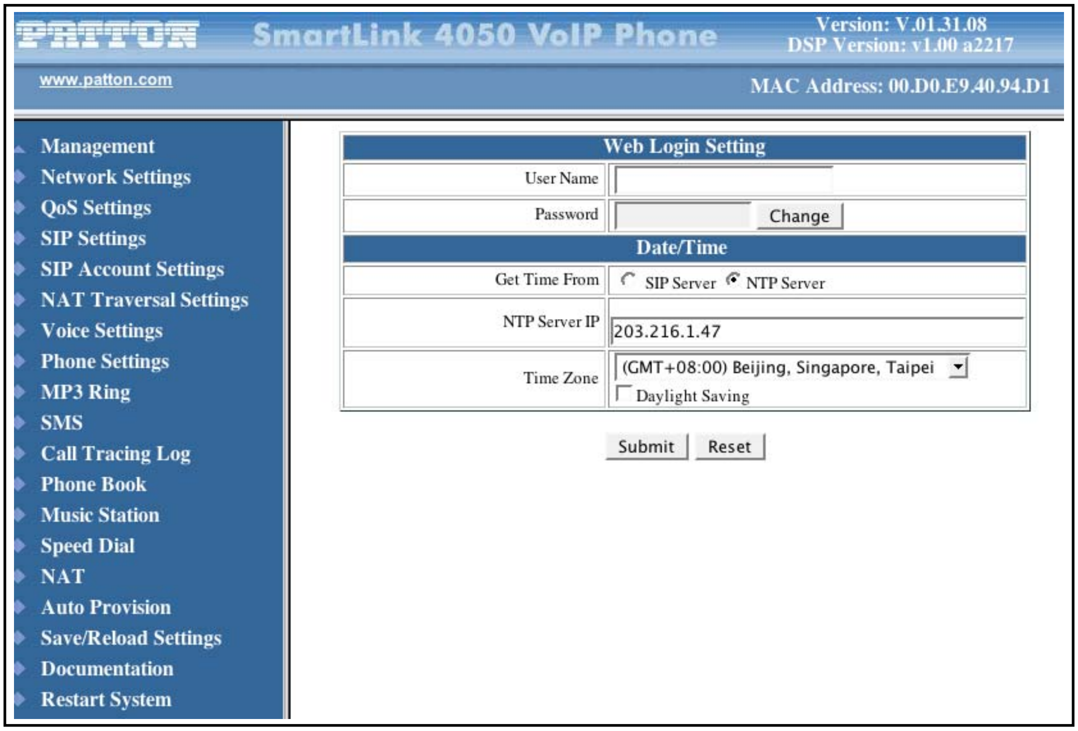


Figure 8. Main window

- **User Name:** Configuration menu login name.
- **Password:** Configuration menu login password.
- **Get Time From**: Get time setting from SIP or NTP server.
- **NTP Server IP:** Network Time Protocol (NTP) is a protocol used to help match your system clock with an accurate time source (e.g. atomic clock, 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 files 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.

Management Settings—Restore Factory Setting

1. Click on **Management** > **Restore Factory Setting**. The message *Press [Restore] button to restore the default setting*! displays (see figure 9).
2. Click the **Restore** button to return all settings back to factory default settings.

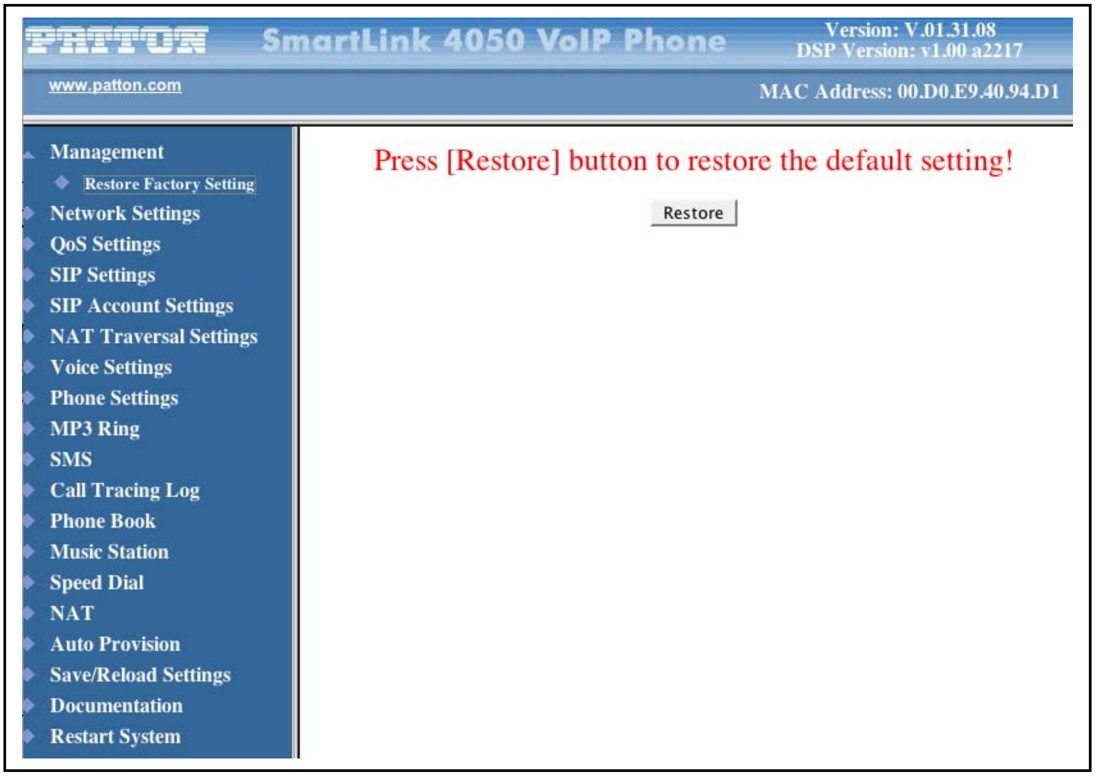


Figure 9. Restore Factory Setting window

Figure 10.

Network Settings

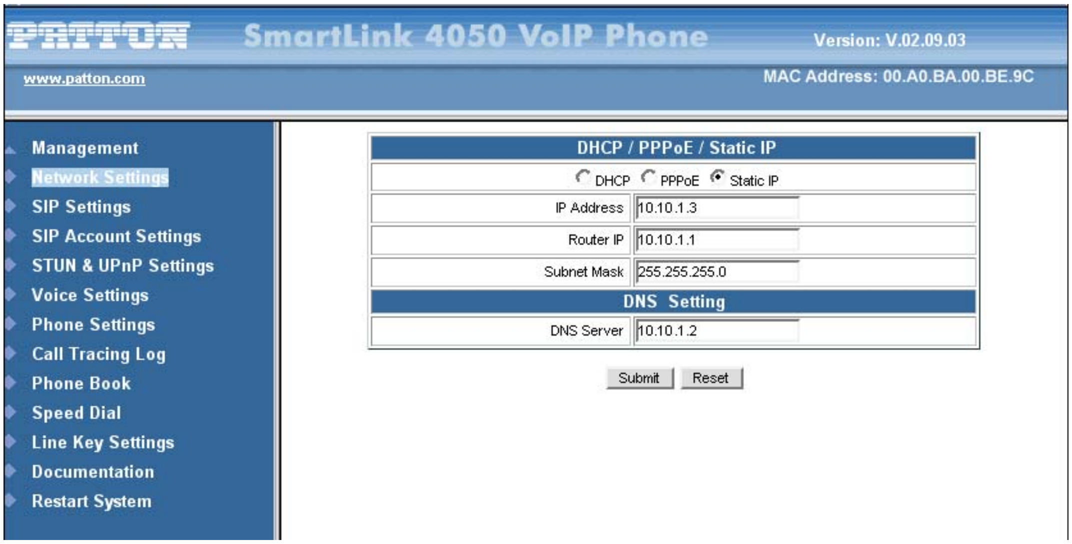


Figure 11. Network Settings window

Click on **Network Settings** to display the configuration window (see figure 11). Select the method used to connect to the Internet:

- **DHCP**—Select *DHCP* if you have cable Internet (see section "DHCP" for details)
- **PPPoE**—Select *PPPoE* if your ISP uses PPPoE (most DSL users choose *PPPoE*) (see section "PPPoE" for details)
- **Static IP**—Choose the *Static IP* network setting if the wide area network IP address is provided to you by your ISP (see section "Static IP" for details)

DHCP

Click on Dynamic host configuration protocol (DHCP) for IP address information that is obtained automatically from your ISP. The **DNS Server** information is obtained from your ISP.

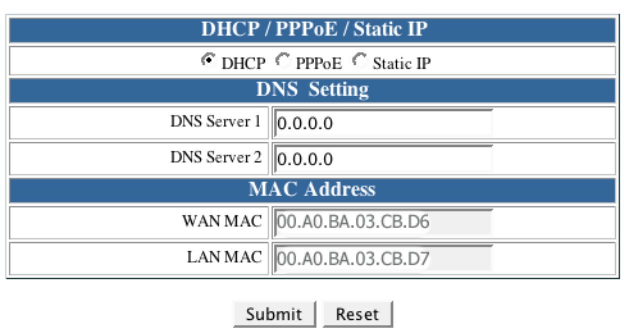


Figure 12. DHCP configuration window

PPPoE

Select **PPPoE** if your ISP uses PPPoE. Most DSL users use PPPoE.

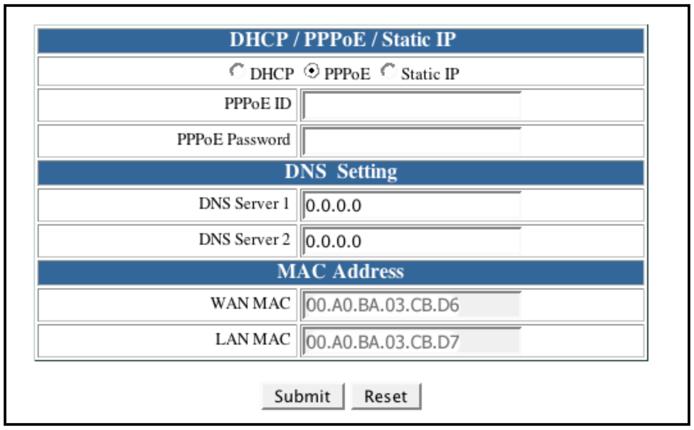


Figure 13. PPPoE configuration window

- **PPPoE ID:** PPPoE ID/username provided by your ISP
- **PPPoE Password:** Password for the PPPoE ID
- **DNS Server 1-2:** DNS address provided by your ISP

Static IP

Select **Static IP** if all Wide Area Network IP information is provided to you by your ISP.

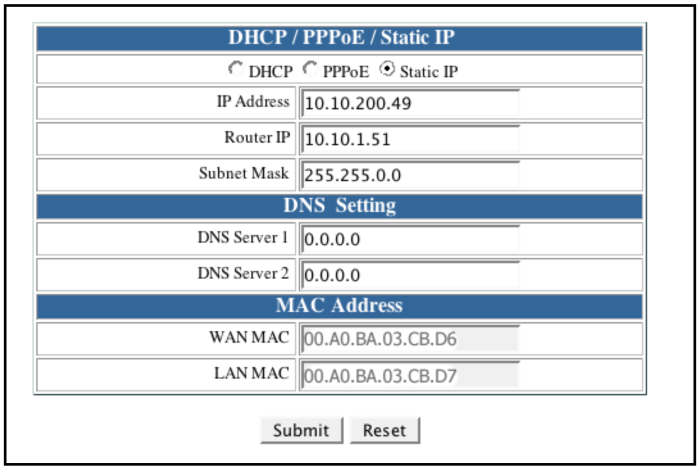


Figure 14. Static IP configuration window

- **IP Address:** IP address assigned to you by your ISP
- **Router IP:** Router IP address
- **Subnet Mask:** Subnet mask
- **DNS Server 1-2:** DNS server address provided by your ISP

Note **RESTART** the system for new settings to take effect after you modify the IP address.

QoS Settings

Figure 15. QoS Settings window

- **Voice TOS:** Set the type of service for the Internet data
- **SIP TOS:** Set the type of service for the higher signaling priority packet
- **VLAN:** Enable or disable VLAN

SIP Settings

Click on **SIP Settings** to display the configuration window (see figure 16). Session initiation protocol (SIP) is the most popular VoIP standard. It enables two or more people to make phone calls, share multimedia, and make multimedia conference over the Internet.

Note You should have an administrator set up these settings for you or obtain the information directly from your SIP service provider.

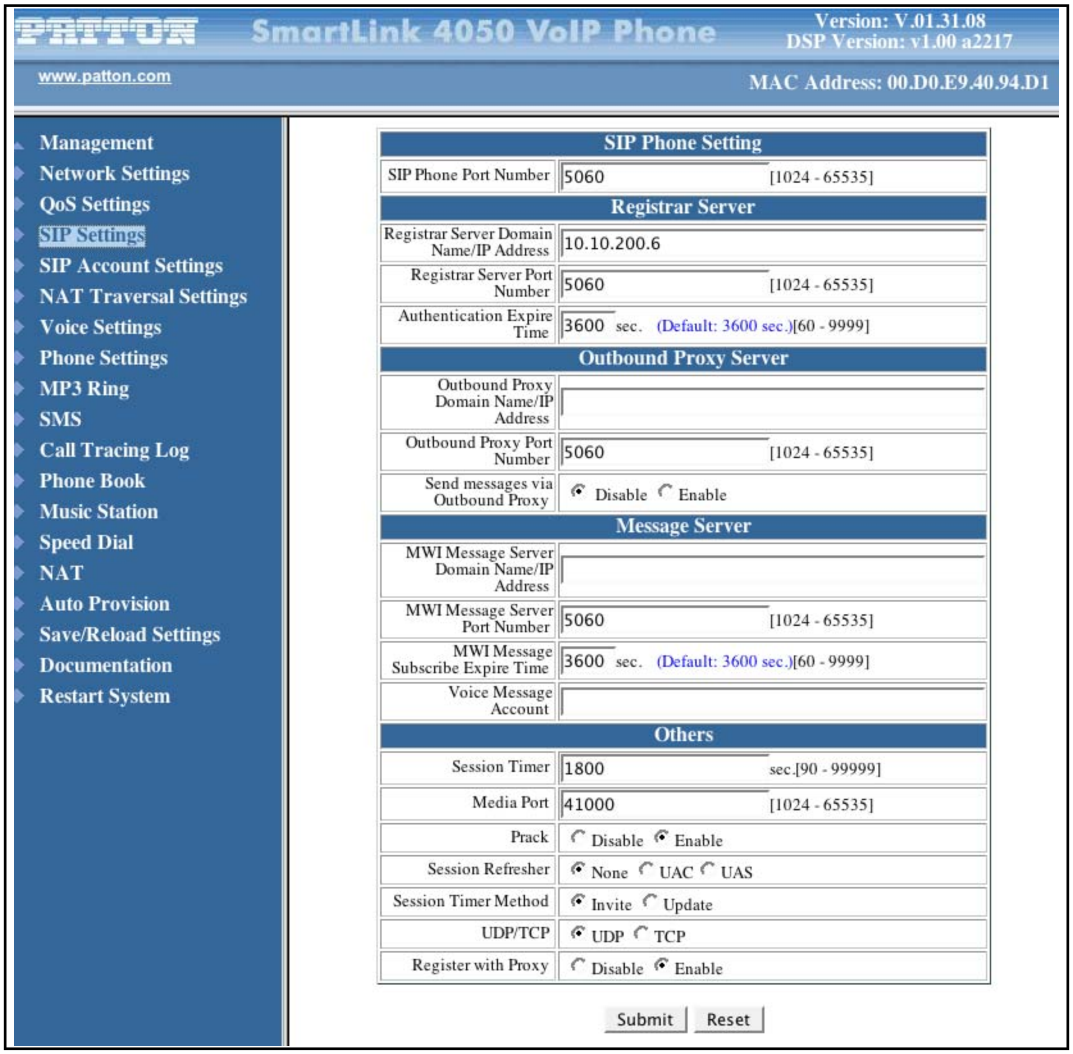
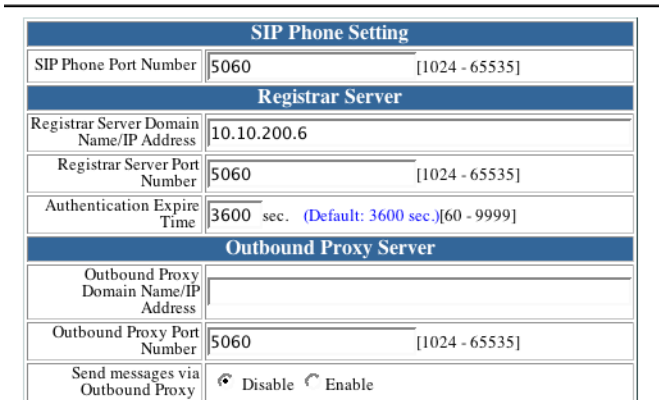


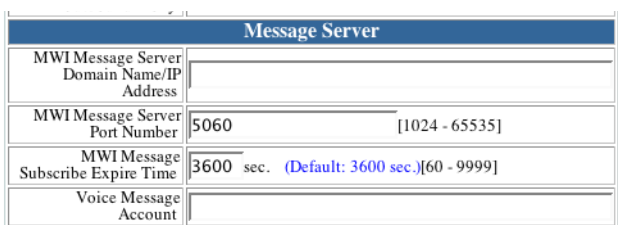
Figure 16. SIP Settings window

SIP Phone Setting, Registrar Server, and Outbound Proxy Server



- **SIP Phone Port Number:** SIP phone listening port
- **Registrar Server Domain Name/IP Address:** Registrar server domain name or IP address.
- **Registrar Server Port Number:** Registrar server listening port
- **Authentication Expire Time:** The time that the SIP registration expires. The phone must send SIP REGISTER to keep the registration at half of the setting time.
- **Outbound Proxy Domain Name/IP Address:** Outbound proxy domain name or IP address.
- **Outbound Proxy Port Number:** Outbound proxy listening port
- **Send messages via Outbound Proxy:** Select *Enable* to send all SIP requests through Outbound Proxy

SIP Message Server



- **MWI Message Server Domain Name/IP Address:** Message server domain name or IP address
- **MWI Message Server Port Number:** Message server listening port
- **MWI Message Subscribe Expire Time:** The time that the subscription expires.
- **Voice Message Account:** Voice message account

Park Server & Presence Server (Model SL4050/B12/E only)

Park Server	
Park Server Domain Name/IP Address	
Park Account	
Presence Server	
Presence Server Domain Name/IP Address	

- **Park Server Domain Name / IP Address:** Park server host name or IP address.
- **Park Account:**The number of the parking area on Park server
- **Presence Server Domain Name / IP Address:** Presence server host name or IP address.

Other Settings

Others	
Session Timer	1800 sec.[90 - 99999]
Media Port	41000 [1024 - 65535]
Prack	○ Disable ◉ Enable
Session Refresher	◉ None ○ UAC ○ UAS
Session Timer Method	◉ Invite ○ Update
UDP/TCP	◉ UDP ○ TCP
Register with Proxy	○ Disable ◉ Enable

Submit Reset

This section should be configured by 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 audio.
- **Prack:** A SIP method which is applied to the condition of acknowledging to the provisional responses like 180 Ringing. 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.
- **Register with Proxy:** When “Set messages via Outbound Proxy” is enabled, all the SIP requests including Register will be sent through Outbound Proxy. Enable the option will against the rule and send SIP Register directly to the Registrar.

SIP Account Settings

Click on **SIP Account Settings** to display the configuration window (see figure 17). You can have up to four accounts—that is, the SIP phone can receive calls from up to four different phone numbers.

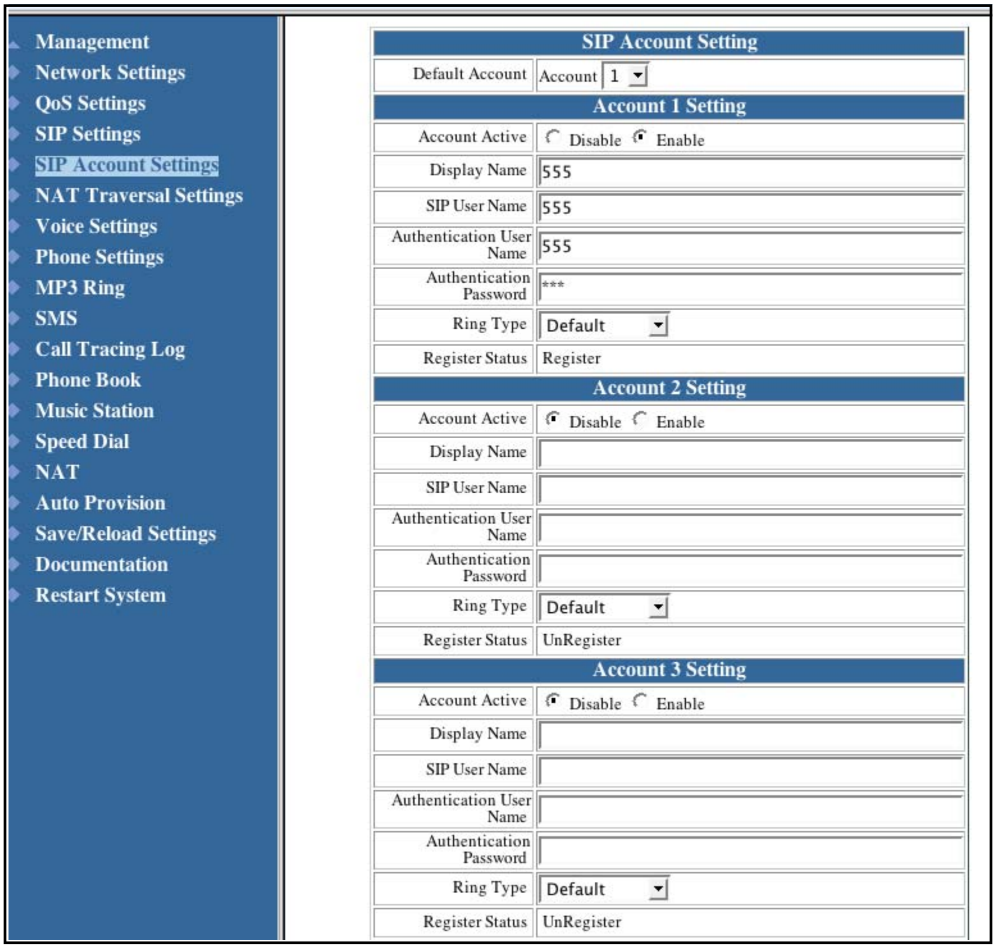


Figure 17. SIP Account Settings window

- **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:** Name displayed on the LCD of called party.
- **SIP User Name:** The number in the URI displayed on the LCD for the caller.
- **Authentication User Name:** User name to log into the SIP server.
- **Authentication Password:** Password to log into the SIP server.
- **Ringer Type:** Eight types of tone and melody can be selected for the specified account
- **Register Status:** Displays if the current phone is registered or unregistered with SIP server.

NAT Traversal Settings

Click on **NAT Traversal Settings** to display the configuration window (see figure 18).

NAT traversal is a challenge that all Service Providers looking to deliver public IP-based voice service must solve. The challenge is to provide secure connection to subscribers behind NAT (Network Address Translation) devices and Firewalls. Overcoming this traversal problem will lead to widespread deployment of profitable voice over IP service to any subscriber with a broadband connection. Therefore, this IP Phone implements NAT traversal function for solving the Firewall and NAT traversal problems

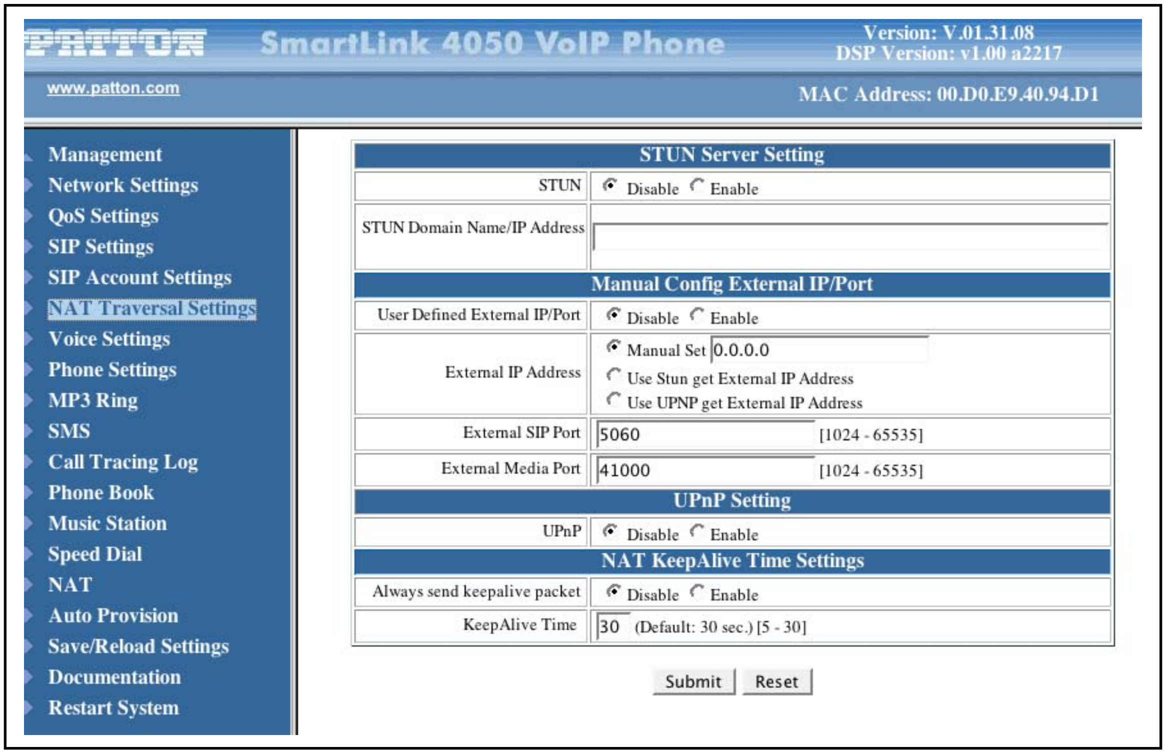


Figure 18. STUN & UPnP Settings

- **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.
- **User Defined External IP/Port:** Enable or disable the settings for configuring the user defined external IP address and port number.
- **External IP Address:**
  - Setup the external IP address manually.
  - Use Stun server to get external IP address.
  - Use UPnP to get external IP address.

- **External SIP Port:** External SIP port
- **External Media Port:** External media port

> **Note** It has to be complied with the settings of virtual server of the NAT devices if IP Phone enables the configuration manually.

- **UPnP:** Enable or disable universal plug and play. Some NAT supports UPnP so STUN is not required and must be disabled.
- **Always send keepalive packet:** Enable or disable to keep the channel which is created for SIP signaling alive.
- **KeepAlive Time:** The time interval that the IP phone always sends the keepalive packet in order to ensure NAT works properly.

Voice Settings

Click on **Voice Settings** to display the configuration window (see figure 19).

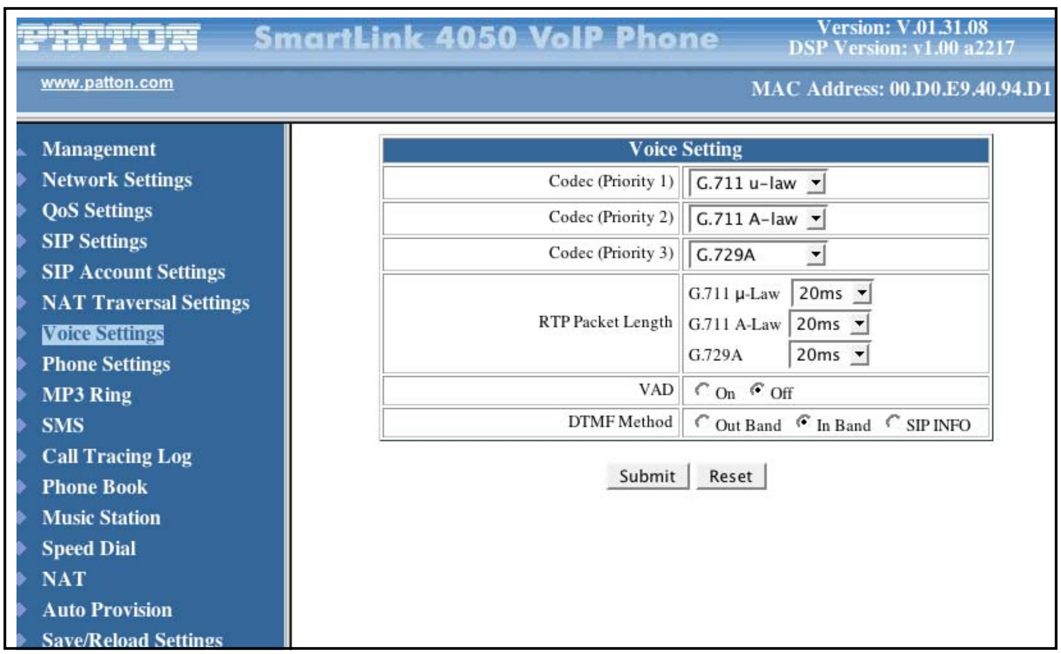


Figure 19. Voice Settings

- **Codec (Priority 1 ~ 3):** Voice Compression Algorithm priority settings. Select from the most used codec to the least used codec.
- **RTP Packet Length:** The payload size for each RTP packet.
- **VAD:** Support VAD for silence suppression. When Enable is selected, it also supports SID frame for CNG.
- **DTMF Method:** Select the method to generate DTMF. Out Band DTMF is based on RFC2833.
- **Payload Type:** Setting the payload type for the Out Band DTMF (Default is 101).

Phone Settings

Click on **Phone Settings** to display the configuration window (see figure 20).

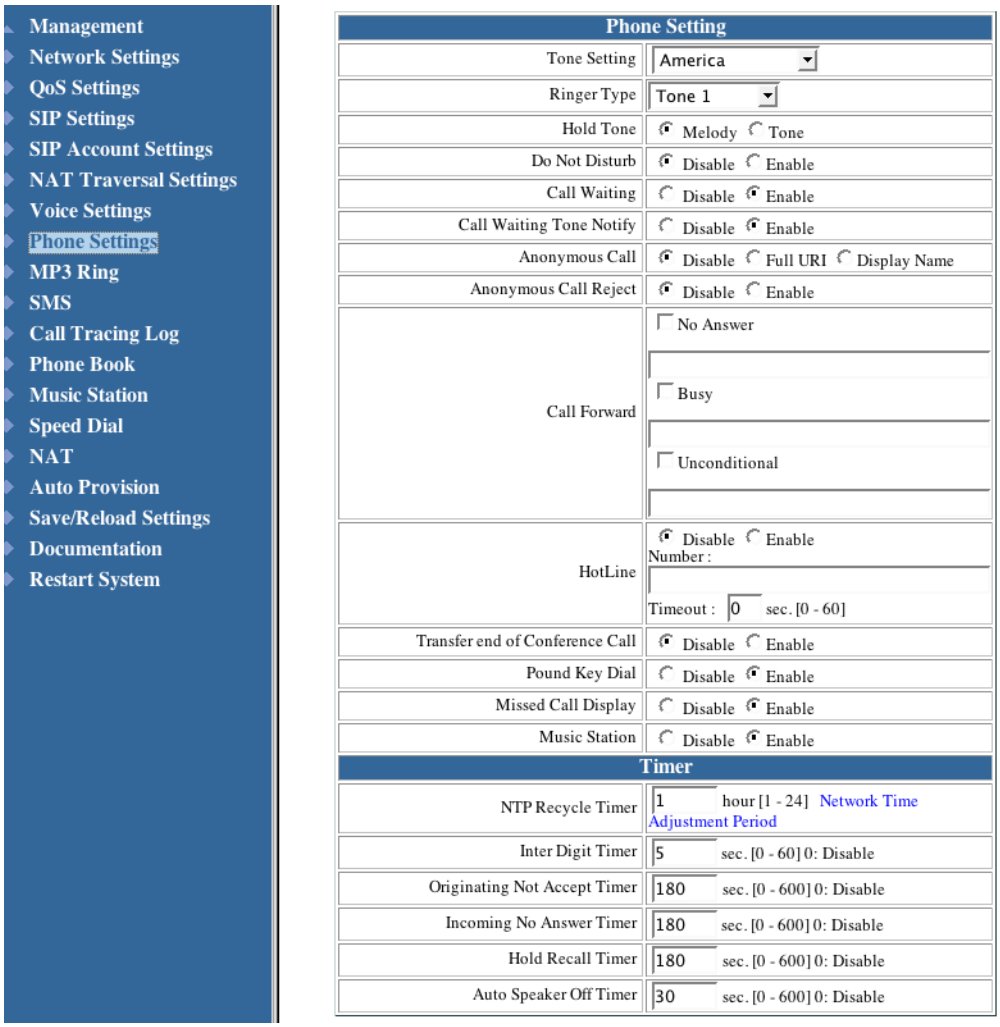


Figure 20. Phone Settings window

Phone Setting

- **Tone Setting:** Select the tone for particular country
- **Ringer Type:** Select the type of ring (Tone: 1 ~ 4, Melody: 5 ~ 8 & MP3: 9).
- **Hold Tone:** Select melody or tone when the phone is on hold.
- **Do Not Disturb:** Reject all incoming calls.
- **Call Waiting:** Enable or disable call waiting.
- **Call Waiting Tone Notify:** Enable or disable the reminding tone for 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, it uses "Anonymous" as its display name and URI when the user makes a phone call. It may display "Anonymous" or nothing on the receiver's phone.
  - When *Display Name* is selected, only display name is replaced by "Anonymous" when the user makes a phone call. It may display "Anonymous" or nothing on the receiver's phone.
- **Anonymous Call Reject:** Select Enable to reject anonymous calls.
- **Call Forward:** Enter the call forward number on the text box.
  - 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 the section,"Timer", 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.
- **Hot Line:**
  - Enable or disable Hot Line
  - *Number:* a phone number which is the destination of Hot Line
  - *Timeout:* If user doesn't dial during the time, the phone will dial the Number automatically.
- **Transfer end of Conference Call:** Enable or disable the feature of transferring call after the three-way conference call is ended.
- **Pound Key Dial:** Enable or disable Pound key Dial. Pound Key (#) can be defined as a <send> key.
- **Miss Call Display:** Enable or disable to display miss calls on the LCD.

Timer

- **NTP Recycle Timer:** The time interval that the IP phone synchronize with NTP server.
- **Inter Digit Timer:** 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 Timer:** 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 Timer:** 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 Timer:** The time interval that the call party which is put on held by the phone recalls.
- **Auto Speaker Off Timer:** The time interval that the speaker phone is on before turning off automatically (due to inactivity).

MP3 Ring

Click on **MP3 Ring** to display the configuration window (see figure 21).

Figure 21. MP3 Ring window

- **Ring File:** Click “Browse” to choose one MP3 file and click “Upload File”. The maximum size of the MP3 file is 30KB.

The MP3 file is used for the Ringer type “MP3 Ring 9”.

SMS

Click on **SMS** to display the configuration window (see figure 22).

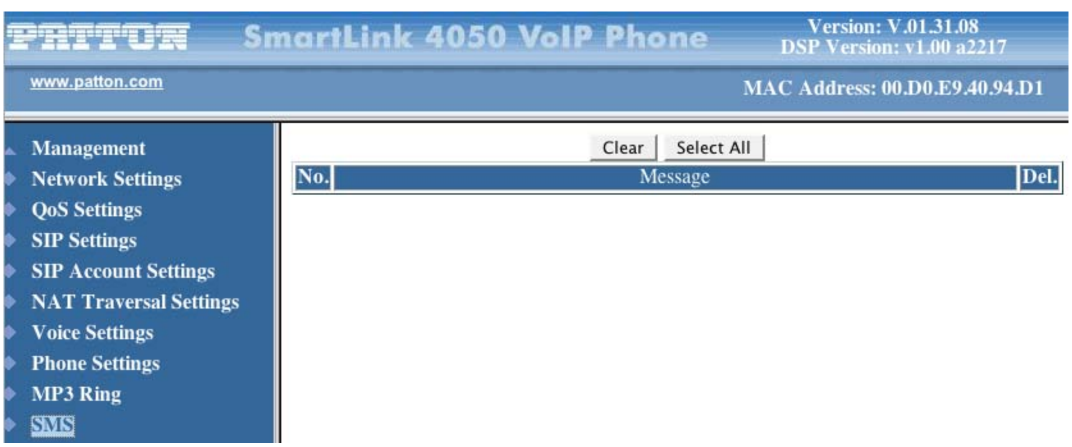


Figure 22. SMS window

- Show the received messages including sender information, received time and message body. (Max. 20 messages).

> **Note** The SL4050 is unable to send messages to other VoIP devices. It can only receive messages.

Call Tracing Log

Click on **Call Tracing Log** to display the configuration window (see figure 23). The call tracing log keeps a record of all the phone activities. This log is used by our Patton technicians to troubleshoot hardware problems.

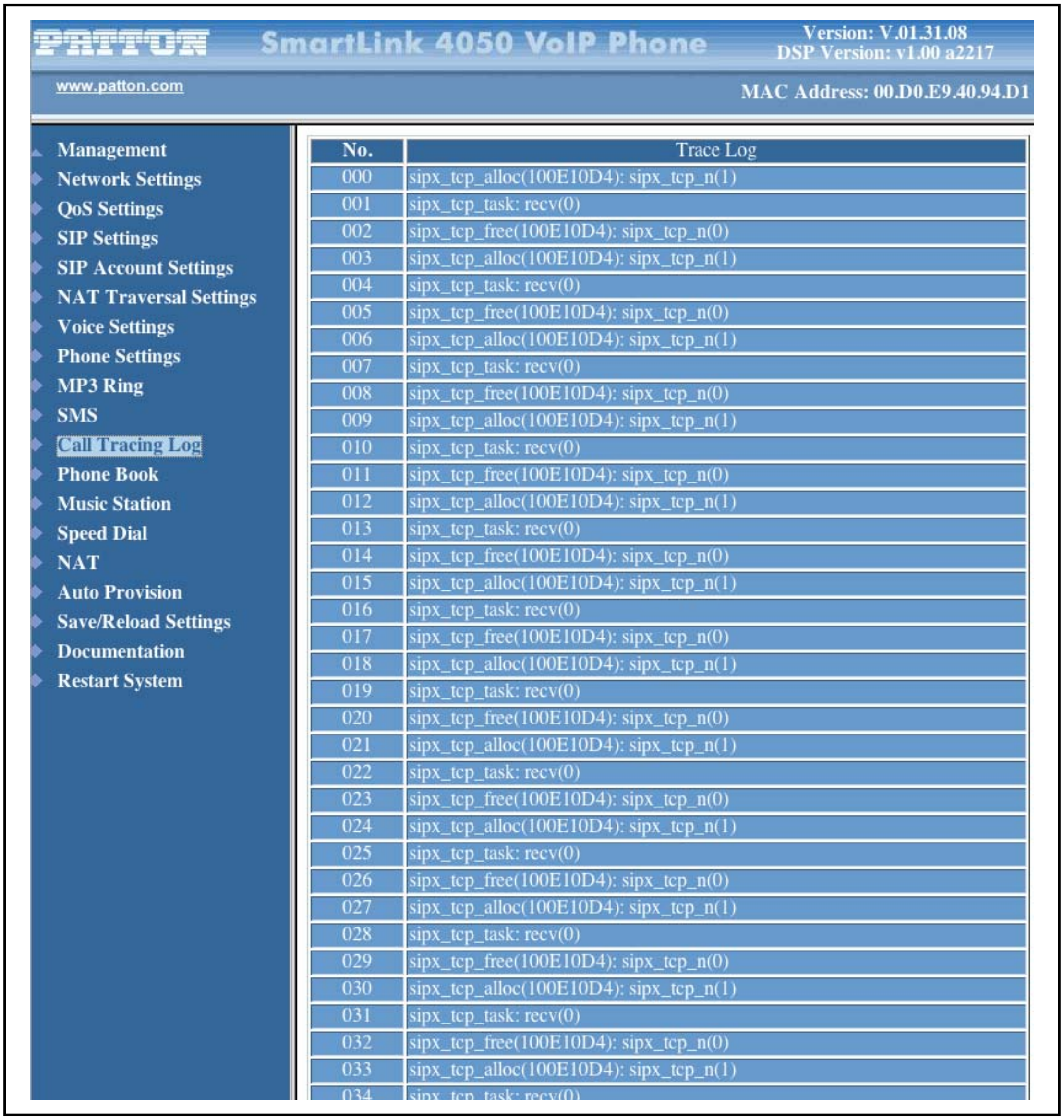


Figure 23. Call Tracing Log window

Phone Book

Click on **Phone Book** to display the configuration window (see figure 24). The Phone Book window enables users to add, modify, or delete phone numbers:

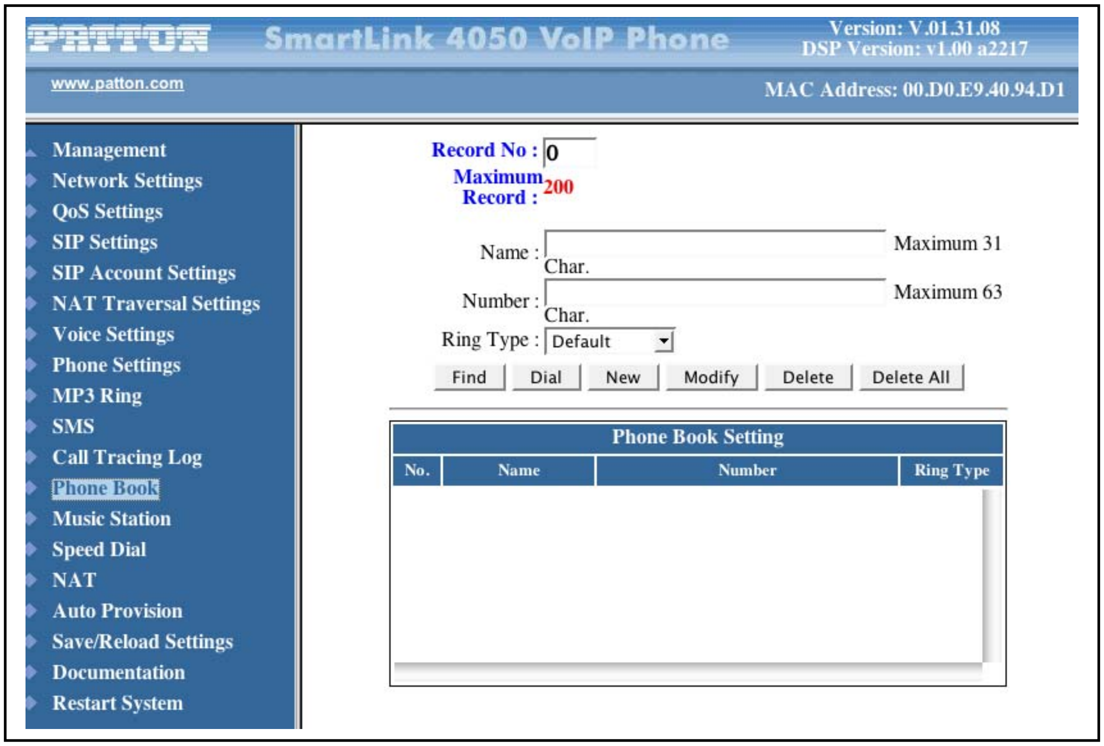


Figure 24. Phone Book window

To add an entry to the phone book, type in the name and number then click **New** to add.

To modify/delete an entry, select the name from the list and click **Modify** or **Delete**.

- **Name:** Name that you would like to add.
- **Number:** Phone number that corresponds to the name.
- **Ring Type:** Ring type of the phone number

Music Station

Click on **Music Station** to display the configuration window (see figure 25). The SL4050 supports 20 stations in maximum. (10 default stations are provided).

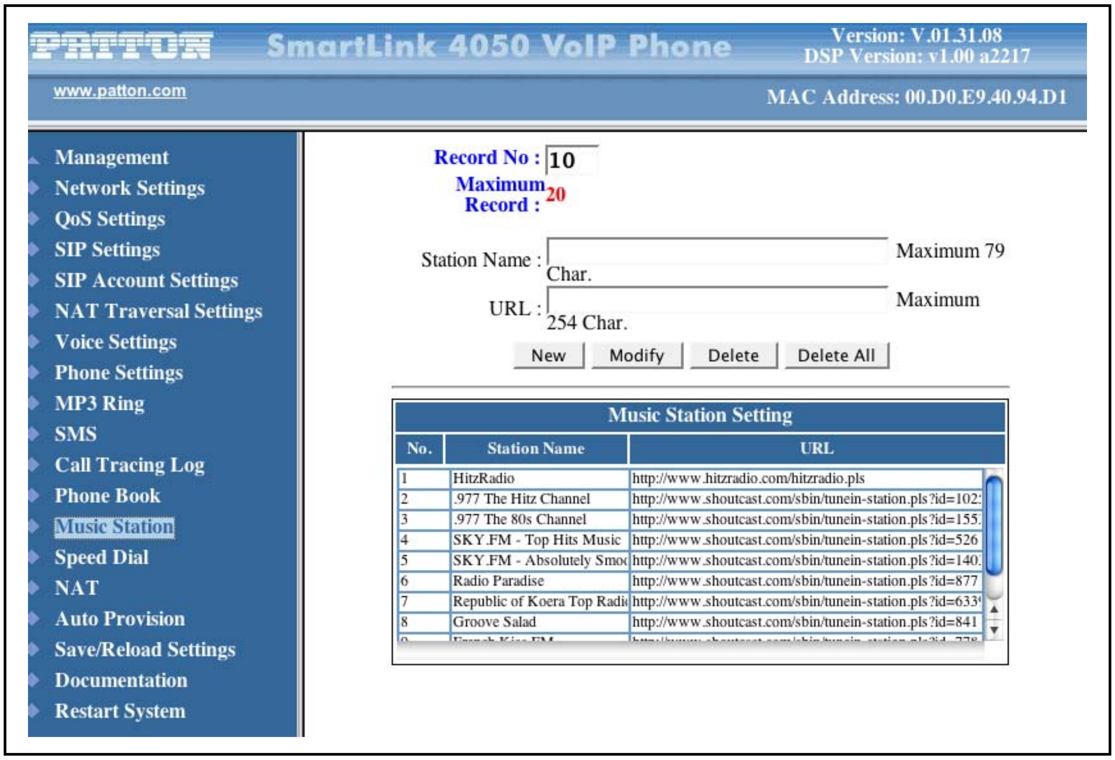
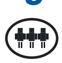


Figure 25. Music Station window

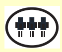
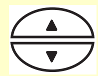
- **Station Name:** An easy-to-memorized name to the station, ex: Station1.
- **URL:** A complete URL to access the station.

Operating the Internet Radio

1. Press  to turn on the Internet radio.
2. Press  and   to choose a preferred station.
3. Press  to turn off the Internet radio.

Key Definitions for Internet Radio

Table 5. Internet Radio Key Definitions

Key	Definition	Key	Definition
	Turn on the Internet Radio		Increase / decrease the volume
	Pause / Play		Display the name of the current station
	Turn off the Internet Radio		Tune the Internet Radio to the preferred station
Numeral Keys	The ten numeral keys 0, 1~9 are the quick access keys to the first ten preferred stations on web configuration "Music Station".		

About Internet Radio

- All the keys related to the Internet Radio are described in table 5. Those key functions will only be available when the phone is hung up. If the phone is hung on, those key functions will back to the original designed which has stated in Page.7.
- When the phone is receiving the incoming call, the Internet Radio function will be turned off automatically.
- When the user picks up the handset or presses "SPEAKER" to make a phone call, the Internet Radio will be also turned off automatically.
- Please turn off the Internet Radio before you do the next steps as below:
  - Use pre-dialing to make a phone call
  - Enter MENU to configure
  - Access Phone Book
  - Adjust the Ringer Volume
- When the user is listening to the Internet Radio, the phone will have the current song and singer's name showing on the screen.

Speed Dial

Click on **Speed Dial** to display the configuration window (see figure 26). Speed dial numbers can be accessed from the IP phone.

Figure 26. Speed Dial window

- **Number 0x:** Speed dials phone number. 0x is the speed dial number.

Line Key Settings (Model SL4050/B12/E only)

Click on **Line Key Settings** to display the configuration window (see figure 27).

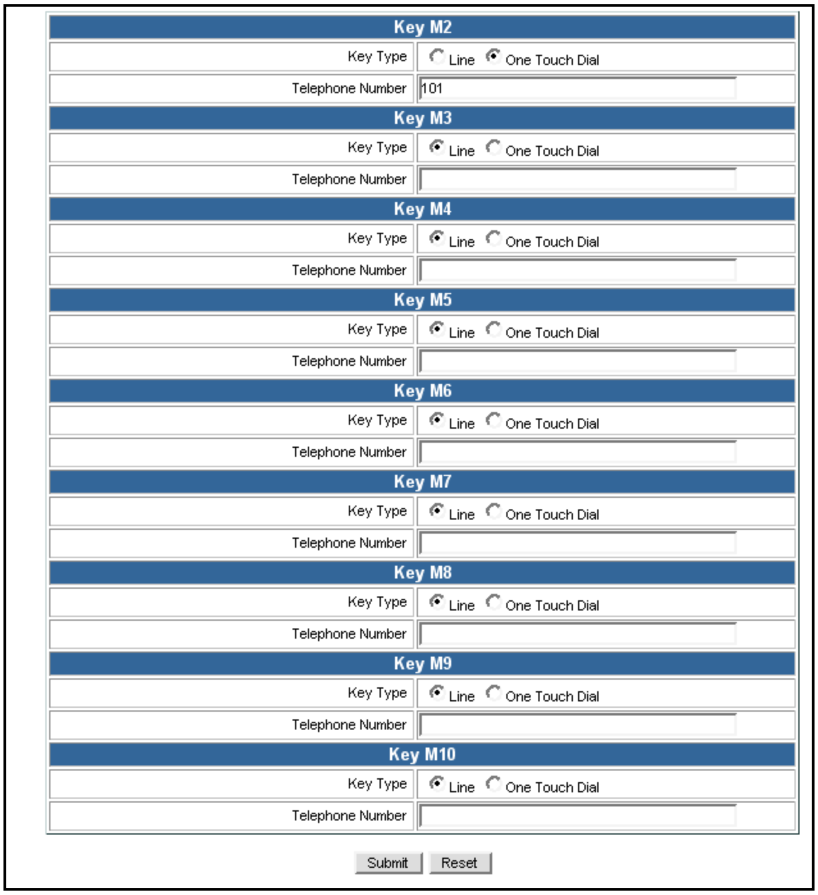


Figure 27. Line Key Settings window

- **Type:** It supports four types for those programmable keys that are “Line”, “Park”, “One Touch Dial” and “Extension”. Default is “Line”
  - *Park:* It is an advanced feature to park the active call in the parking area which is a special extension on Park server. The phones which have been assigned to monitor the parking area can retrieve calls if there are calls on parked. The Park server is generally co-located with SIP proxy.
  - *Extension:* It is an advanced feature called “DSS/BLF”. It watches the specified extension by receiving the notification of status from Presence server, which is generally co-located with SIP proxy and shows the status by LED indicator. The pre-configured key can be treated as the representative of the watched extension. It can be used to call the extension directly and pick up calls of the extension by pressing the key.
- **Park Number:** The phone number of the parking area that is corresponding to “Park”.
- **Phone Number:** The phone number of the destination which can be called by one-touch-dial that is corresponding to “One Touch Dial”.
- **Monitor Number:** The phone number of the monitored extension that is corresponding to “Extension”.

NAT Settings

Click on **NAT Settings** to display the configuration window (see figure 28). Select NAT mode for ROUTE Mode or Bridge Mode.

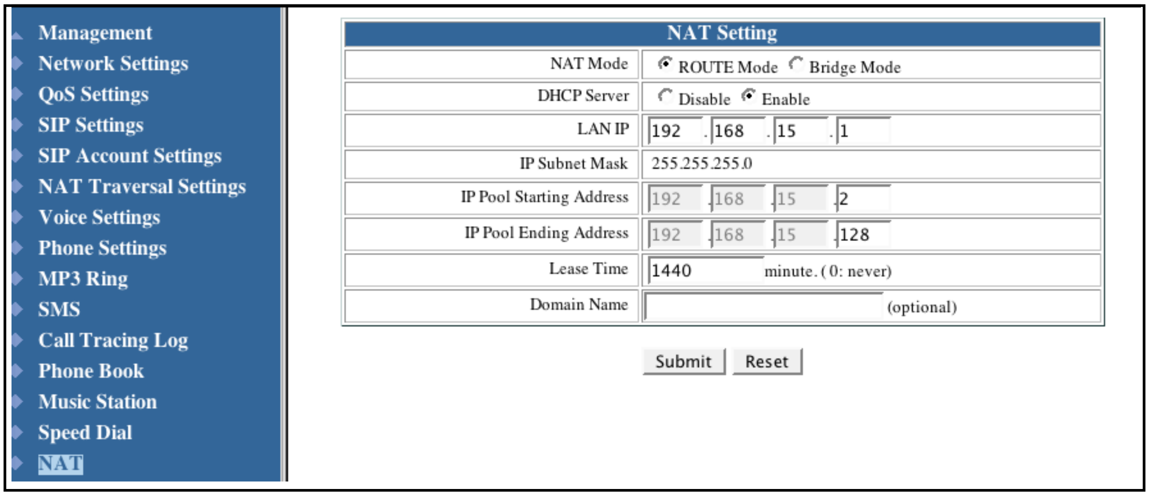


Figure 28. NAT Settings window

Auto-Provision

Click on **Auto Provision** to display the configuration window (see figure 28).

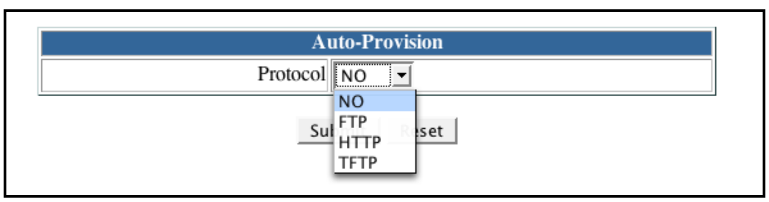


Figure 29. Auto-Provision window

- **Protocol:** Support FTP, HTTP and TFTP for downloading firmware and configuration automatically. Default is NO to disable the function
- **FTP / HTTP / TFTP:** IP address of the Auto Provision server
- **FTP / HTTP / TFTP Port:** Listening port of the Auto Provision server
- **Username (Protocol = FTP or HTTP):** The username required by Auto Provision server for authorization.
- **Password (Protocol = FTP or HTTP):** The password required by Auto Provision server for authorization.
- **Encryption:** Choose YES to receive and decrypt the encrypted configuration files
- **Encryption Key:** The key which is provided by administrator for decrypting the encrypted configuration files
- **Refresh Interval (hr):** The time at which the IP phone connects to the Auto Provision server for checking update.

Save/Reload Settings

Click the **Save/Reload Settings** link to save and reload system settings (see figure 30).

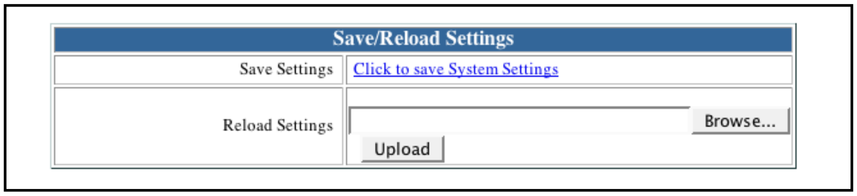


Figure 30. Save/Reload window

Documentation

Click the **Documentation** link (see figure 31) to download and display the *SmartLink 4050 Series Getting Started Guide* in portable document format (PDF).

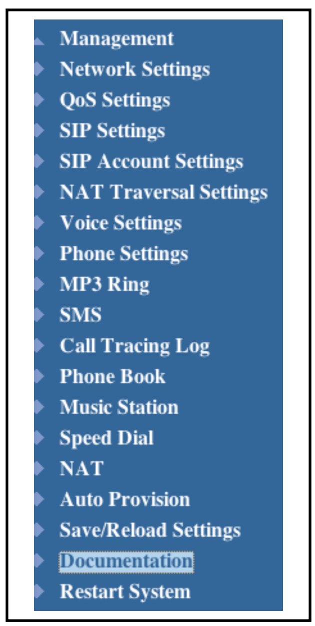


Figure 31. Documentation link

Restart System

Click on *Restart System.* (see figure 32). Click the **Restart** button so all modifications will take effect.

Figure 32. Restart System button

Chapter 6 **Troubleshooting**

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Introduction

CAUTION

The interconnecting cables shall be acceptable for external use and shall be rated for the proper application with respect to voltage, current, anticipated temperature, flammability, and mechanical serviceability.

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

Symptom	Recommended action
No dial tone	• Check to see if there are any loose connections. • Verify that the power cord is connected properly. • Contact your service provider to see if there is a problem with your WAN or Internet connection.
Nothing displayed on the LCD screen	• Verify that the power cord is connected properly. • Verify that proper AC power is available at the power outlet.
How do I update the SIP Phone firmware?	The SIP Phone automatically updates firmware when it powers up (while connected to the internet) if auto-provisioning is available.
Why can't I dial my friend's SIP number?	• 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. • 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. • 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?	• 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. • Check if FTP address is correct. • Check with your supplier if firmware filename is correct.
I accidentally set DSL to enable and now the phone does not boot up	Unplug the power cord from the IP phone. Wait 2 seconds and plug the power cord back in the IP phone. Press and hold the **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 does the "Can't Upgrade Now" message display when I click **Submit** in the configuration menu?	Make sure you exit setting mode (phonebook, menu, speed dial, etc.) before clicking **Submit** in the configuration menu.

Symptom	Recommended action
The WAN port of my SL4050 (PhoneB) is connected with the LAN port of another SL4050 (PhoneA). And, my SL4050 (PhoneB) become malfunction on the networking so that I can not get the VoIP services. How can I do to fix it?	To solve this problem, please change the IP segment of the PhoneA LAN port other than the “192.168.15.xxx” first (for example, “192.168.10.xxx”). Then, the PhoneB will automatically start to get the VoIP connection and the associated VoIP services. It is because that in the factory default settings the SL4050 has an integrated DHCP server to assign the IP address of the LAN port with the IP segment of “192.168.15.xxx”. That is, for this kind of connection of PhoneA & PhoneB (the WAN port of PhoneB is connected with the LAN port of PhoneA), the WAN port & the LAN port of PhoneB will be in the same IP segment (192.168.15.xxx), which will get the system of PhoneB confused so as to be malfunction on the networking. From above, that is the reason why we should change the IP segment of the PhoneA LAN port.

Chapter 7 **Contacting Patton for assistance**

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Introduction

This chapter contains the following information:

- "Contact information"—describes how to contact Patton technical support for assistance.
- "Warranty Service and Returned Merchandise Authorizations (RMAs)"—contains information about the RAS warranty and obtaining a return merchandise authorization (RMA).

Contact information

Patton Electronics offers a wide array of free technical services. If you have questions about any of our other products we recommend you begin your search for answers by using our technical knowledge base. Here, we have gathered together many of the more commonly asked questions and compiled them into a searchable database to help you quickly solve your problems.

Patton support headquarters in the USA

- Online support: Available at **www.patton.com**
- E-mail support: E-mail sent to **support@patton.com** will be answered within 1 business day
- Telephone support: Standard telephone support is available five days a week—from **8:00 am** to **5:00 pm EST** (**1300** to **2200 UTC/GMT**)—by calling **+1 (301) 975-1007**
- Support via VoIP: Contact Patton free of charge by using a VoIP ISP phone to call **sip:support@patton.com**
- Fax: **+1 (253) 663-5693**

Alternate Patton support for Europe, Middle East, and Africa (EMEA)

- Online support: Available at **www.patton-inalp.com**
- E-mail support: E-mail sent to **support@patton-inalp.com** will be answered within 1 business day
- Telephone support: Standard telephone support is available five days a week—from **8:00 am** to **5:00 pm CET** (**0900** to **1800 UTC/GMT**)—by calling **+41 (0)31 985 25 55**
- Fax: **+41 (0)31 985 25 26**

Warranty Service and Returned Merchandise Authorizations (RMAs)

Patton Electronics is an ISO-9001 certified manufacturer and our products are carefully tested before shipment. All of our products are backed by a comprehensive warranty program.

> **Note** If you purchased your equipment from a Patton Electronics reseller, ask your reseller how you should proceed with warranty service. It is often more convenient for you to work with your local reseller to obtain a replacement. Patton services our products no matter how you acquired them.

Warranty coverage

Our products are under warranty to be free from defects, and we will, at our option, repair or replace the product should it fail within one year from the first date of shipment. Our warranty is limited to defects in workmanship or materials, and does not cover customer damage, lightning or power surge damage, abuse, or unauthorized modification.

Returns for credit

Customer satisfaction is important to us, therefore any product may be returned with authorization within 30 days from the shipment date for a full credit of the purchase price. If you have ordered the wrong equipment or you are dissatisfied in any way, please contact us to request an RMA number to accept your return. Patton is not responsible for equipment returned without a Return Authorization.

Return for credit policy

- Less than 30 days: No Charge. Your credit will be issued upon receipt and inspection of the equipment.
- 30 to 60 days: We will add a 20% restocking charge (crediting your account with 80% of the purchase price).
- Over 60 days: Products will be accepted for repairs only.

RMA numbers

RMA numbers are required for all product returns. You can obtain an RMA by doing one of the following:

- Completing a request on the RMA Request page in the *Support* section at **www.patton.com**
- By calling **+1 (301) 975-1007** and speaking to a Technical Support Engineer
- By sending an e-mail to **returns@patton.com**

All returned units must have the RMA number clearly visible on the outside of the shipping container. Please use the original packing material that the device came in or pack the unit securely to avoid damage during shipping.

Shipping instructions

The RMA number should be clearly visible on the address label. Our shipping address is as follows:

Patton Electronics Company
RMA#: xxxx
7622 Rickenbacker Dr.
Gaithersburg, MD 20879-4773 USA

Patton will ship the equipment back to you in the same manner you ship it to us. Patton will pay the return shipping costs.

Appendix A **Compliance information**

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- Safety Compliance 70
- PSTN Regulatory Compliance 70

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Industry Canada Notice 71

Compliance

EMC Compliance:

FCC Part 15, Class B

EN55022, Class B

EN55024

Safety Compliance

UL60950-1/CSA C22.2 No. 60950-1

IEC/EN 60950-1

AS/NZS 60950-1

PSTN Regulatory Compliance

FCC Part 68

CS-03

AS/ACIF S004

AS/ACIF S040

Radio and TV Interference

This equipment generates and uses radio frequency energy, and if not installed and used properly-that is, in strict accordance with the manufacturer's instructions-may cause interference to radio and television reception. This equipment has been tested and found to comply with the limits for a Class A computing device in accordance with the specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the equipment causes interference to radio or television reception, which can be determined by disconnecting the cables, try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna, and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

CE Notice (Declaration of Conformity)

We certify that the apparatus identified in this document conforms to the requirements of Council Directive 1999/5/EC on the approximation of the laws of the member states relating to Radio and Telecommunication Terminal Equipment and the mutual recognition of their conformity.

The safety advice in the documentation accompanying this product shall be obeyed. The conformity to the above directive is indicated by the CE sign on the device.

Authorized European Representative

D R M Green, European Compliance Services Limited.

Oakdene House, Oak Road, Watchfield, Swindon, Wilts SN6 8TD, UK

FCC Part 68 (ACTA) Statement

This equipment complies with Part 68 of FCC rules and the requirements adopted by ACTA. On the bottom side of this equipment is a label that contains-among other information-a product identifier in the format US: AAAEQ##TXXXX. If requested, this number must be provided to the telephone company.

The method used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA.

If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this equipment, for repair or warranty information, please contact our company. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.

Industry Canada Notice

This equipment meets the applicable Industry Canada Terminal Equipment Technical Specifications. This is confirmed by the registration number. The abbreviation, IC, before the registration number signifies that registration was performed based on a Declaration of Conformity indicating that Industry Canada technical specifications were met. It does not imply that Industry Canada approved the equipment.

This Declaration of Conformity means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction. Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above condition may not prevent degradation of service in some situations. Repairs to some certified equipment should be made by an authorized maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment. Users should ensure for their own protection that the ground connections of the power utility, telephone lines and internal metallic water pipe system, are connected together. This protection may be particularly important in rural areas.

Appendix B Specifications

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Protocol

IETF SIP RFC3261

Network Interface

RJ45 x 2

10/100BaseT

LCD Display

2 x 16 characters

Call Features

Call Hold / Resume

Call Mute

Call Transfer (Unattended / Blind & Attended)

Call Waiting

Call Forward (Busy / No Answer / Unconditional)

Caller ID Display

Anonymous Call

Anonymous Call Blocking

In band DTMF / Out-of-band DTMF (RFC 2833) / SIP INFO

3-way Conference

Redial

Message Waiting Indicator (RFC3842)

SMS (RFC 3482)

Call Park / Retrieve (RFC3515)

Direct Station Select (DSS)

Busy Lamp Field (BLF-RFC4235)

Call Pickup (Support SIP server required)

Auto Answer (Support SIP server required)

Codec

G.711µ-law

G711a-law

G.729a/b

Phone Functions

Multi-user (up to 4 SIP accounts)

One touch dial (up to 11 records)

Speakerphone communication

Pre-dial before sending

Hot Line

Handset / Speakerphone Volume adjustment

Speed dial (10 records)

Phone book (200 records)

Multi-line (up to 12 lines)

Call history (Incoming calls / Outgoing calls / Missed calls)

MP3 Ringer

Internet Radio

Security

HTTP 1.1 basic/digest authentication for Web setup

MD5 for SIP authentication (RFC 2069/ RFC 2617)

Dial Methods

Direct IP call without SIP registration

Dial number via SIP server

Dial URI from phone book / speed dial

Voice Quality

VAD (Voice Activity Detection)

CNG (Comfort Noise Generation)

AEC (Acoustic Echo Cancellation)

G.168

Jitter buffer

QoS

ToS field

IEEE 802.1Q VLAN

Tone

DTMF

Ring Tone, 9 selectable tones

Ring Back Tone (local and remote)

Dial Tone

Busy Tone

IP Assignment

Static IP

DHCP

PPPoE

NAT Traversal

UPnP

STUN

Static port mapping

TCP/IP

IP/TCP/UDP/DHCP/RTP/ FTP/ICMP/HTTP/SNTP/TFTP/ DNS

Configuration

Key & LCD configuration

Web browser configuration (Multi-language)

Auto/Manual provisioning system (Support FTP/HTTP/TFTP)

Firmware Upgrade

TFTP

Auto/Manual provisioning system (Support FTP/HTTP/TFTP)

Power

Adapter

Input AC 100-120V / 220-240V

Output DC 9V

LAN port

PoE, Power over Ethernet (IEEE 802.3af)

Environmental

Operating temperature: 0_40¢J

Storage temperature: -20_60¢J

Operating humidity: 20%_80%

Physical Dimensions

Size: 200(L) x 220(W) x 100(H) mm

Wall Mount

Weight: 860g

Appendix C **Wall Mount Installation**

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Mounting the SL4050 78

Mounting the SL4050

This appendix illustrates the installation step by step if you would like to mount the SL4050 on the wall. Please print this page (figure 33) before the installation.

1. Put the template (figure 33), which you have printed before the installation, on the wall. The template shows the two keyholes with plus sign indicating the center where the screw must be located.

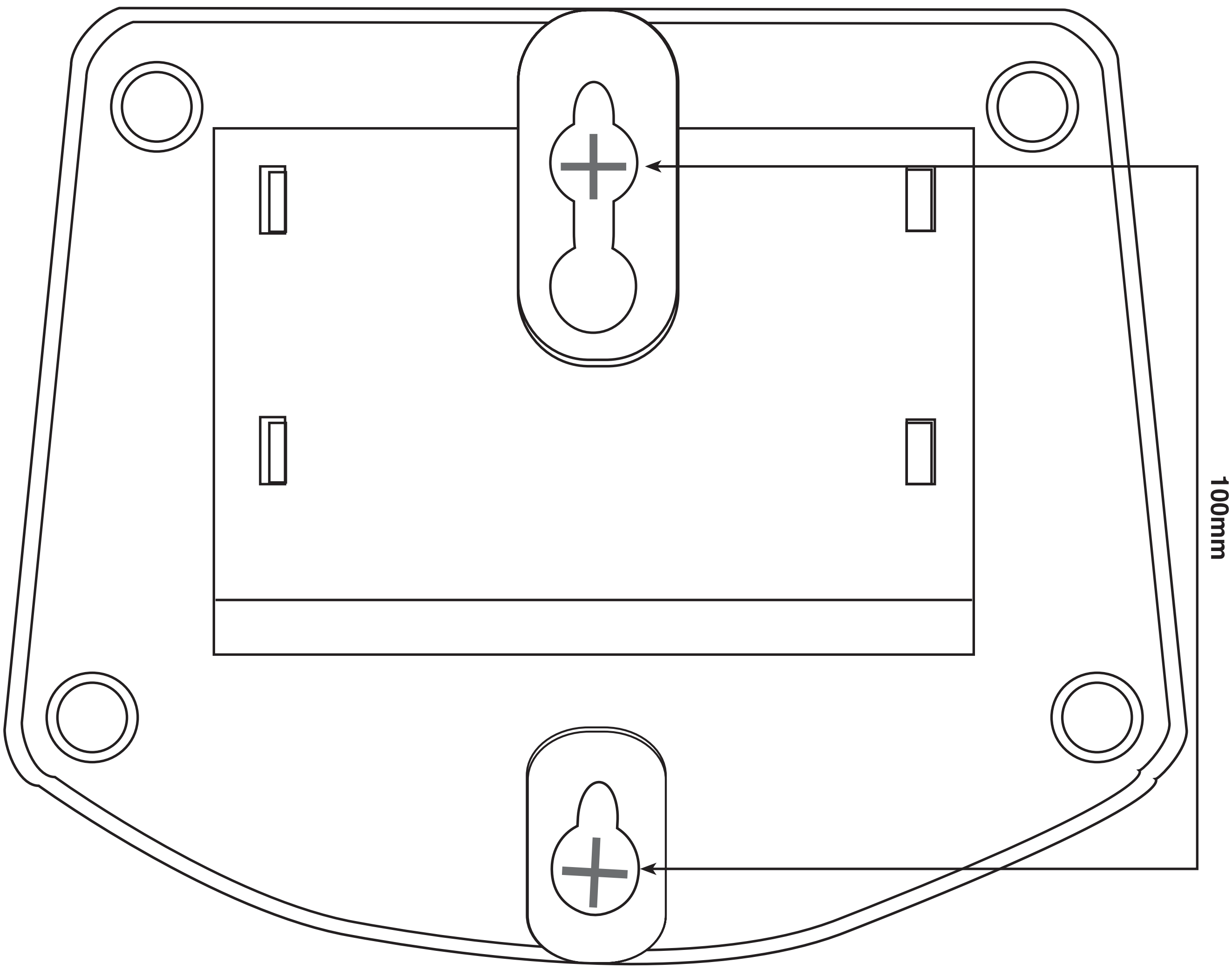

Figure 33. Wall Mount Installation

2. Use a screwdriver to fasten the screw on the wall. Please use the screw with the suitable size and reserve the sufficient distance between the wall and the underside of the screw head as described in figure 34.

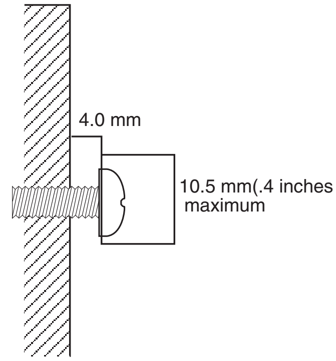

Figure 34. Fasten screw for wall mount installation

3. Place the mount on the wall so that the keyholes of the mount are above the mounting screws.
4. Slide down the mount until it stops against the top of the keyhole
5. Place the SL4050 on the wall mount as figure 35.

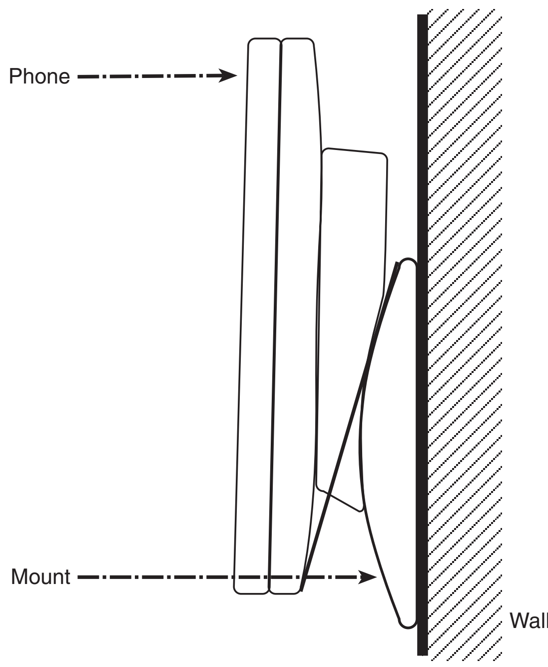

Figure 35. Place SL4050 on the wall mount