

SL4050/B12/E

SmartLink Series 12-Line VoIP SIP Phone

SL4050/B2/E

SmartLink Series 2-Line VoIP SIP Phone

Quick Start Guide



Approval

The Model SL4050 phones are not approved for, and are not intended for, connection to the Public Switched Telephone Network (PSTN).



WARNING

- **The SmartLink SIP Phone contains no user serviceable parts. The equipment shall be returned to Patton Electronics for repairs, or repaired by qualified service personnel.**
- **Mains Voltage: Do not open the case when the power cord is attached. The mains outlet that is utilized to power the SmartLink SIP Phone 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.**
- **Ultimate disposal of this equipment must be handled according to all applicable national laws and regulations.**

1.0 Before you begin

The VoIP SIP phone can be set up using the keypad and a web browser, such as Internet Explorer. If you purchased this product to make a VoIP call, you must have either an Ethernet-based Cable or a DSL modem with an active connection to the Internet.

1.1 Check your package contents

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

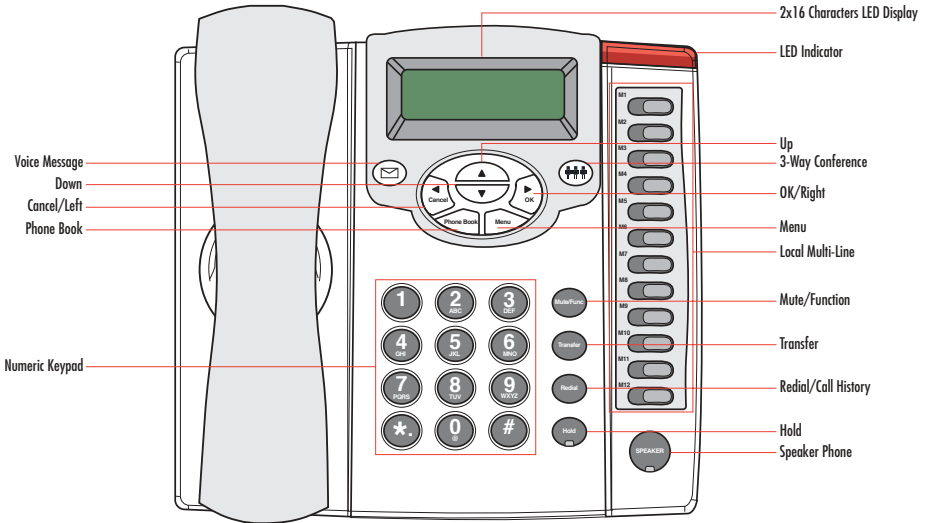


Power Adaptor



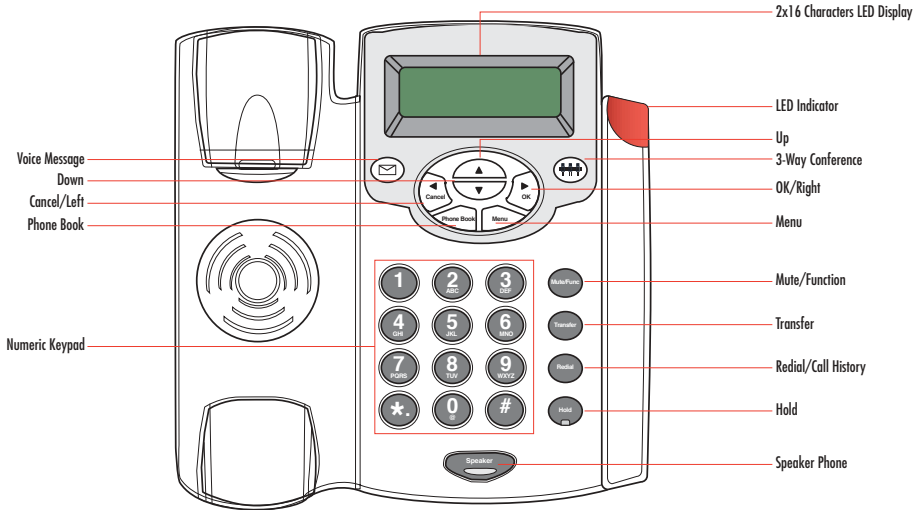
Wall/Desk mounting plate

1.2 Overview of the 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 |
| 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 |













1.3 Overview of the 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 |
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| 3-Way Conference | Enable 3-way conference |
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| 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 |

1.4 Numeric Keypad Definitions

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.

| Key | Text Mode | | Key | Text Mode | |
|---|---------------|---------------|---|--|---------------|
| | Normal (ABC) | Numeric (0-9) | | Normal (ABC) | Numeric (0-9) |
|  | | 1 |  | pqrsPQRS | 7 |
|  | abcABC | 2 |  | tuvTUV | 8 |
|  | defDEF | 3 |  | wxyzWXYZ | 9 |
|  | ghiGHI | 4 |  | @ . _ * # () % & + / \$, | 0 |
|  | jkIJKL | 5 |  | . | * |
|  | mnoMNO | 6 |  | | # |

2.0 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 WAN port on the SIP phone (see **figure 1** for SL4050/B12/E or **figure 2** on page 8 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.

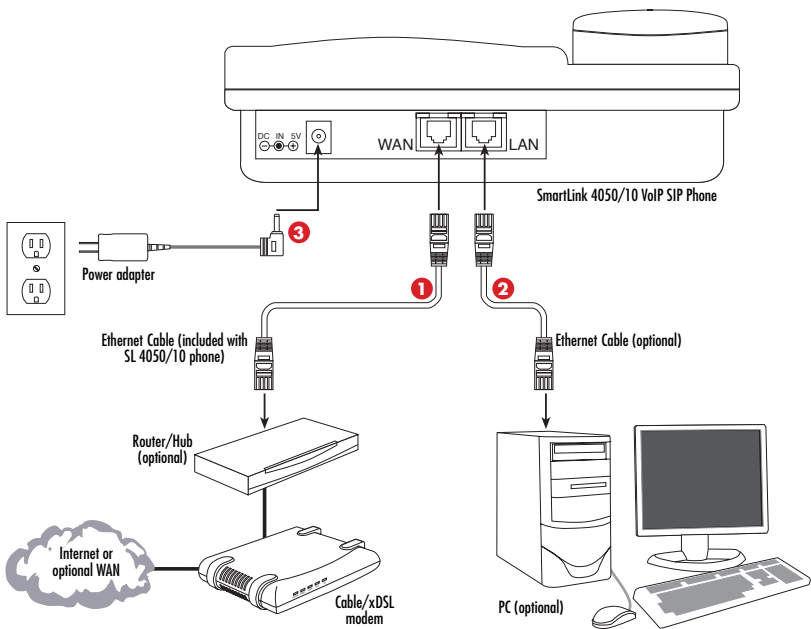


Figure 1. Connecting the SL4050/B12/E SIP Phone

- 2** If you will not be connecting a PC to the phone, go to step 3. Otherwise, connect an Ethernet cable into the LAN port of the SIP phone (see **figure 1** on page 7 for SL4050/B12/E or **figure 2** for SL4050/B2/E). Plug the other end of the cable into the Ethernet port on the PC.

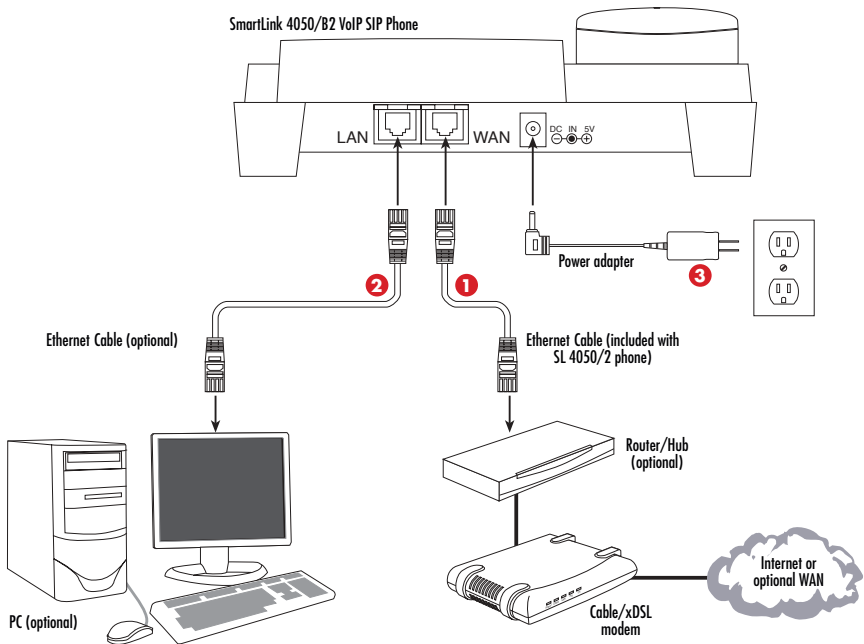


Figure 2. Connecting the SL4050/B2/E SIP Phone

- 3** Plug the power adapter barrel connector into the power connector on the SIP phone (see **figure 1** on page 7 for SL4050/B12/E or **figure 2** for SL4050/B2/E). Plug the other end of the power adapter into an AC electrical outlet.

3.0 Setting up the VoIP SIP phone

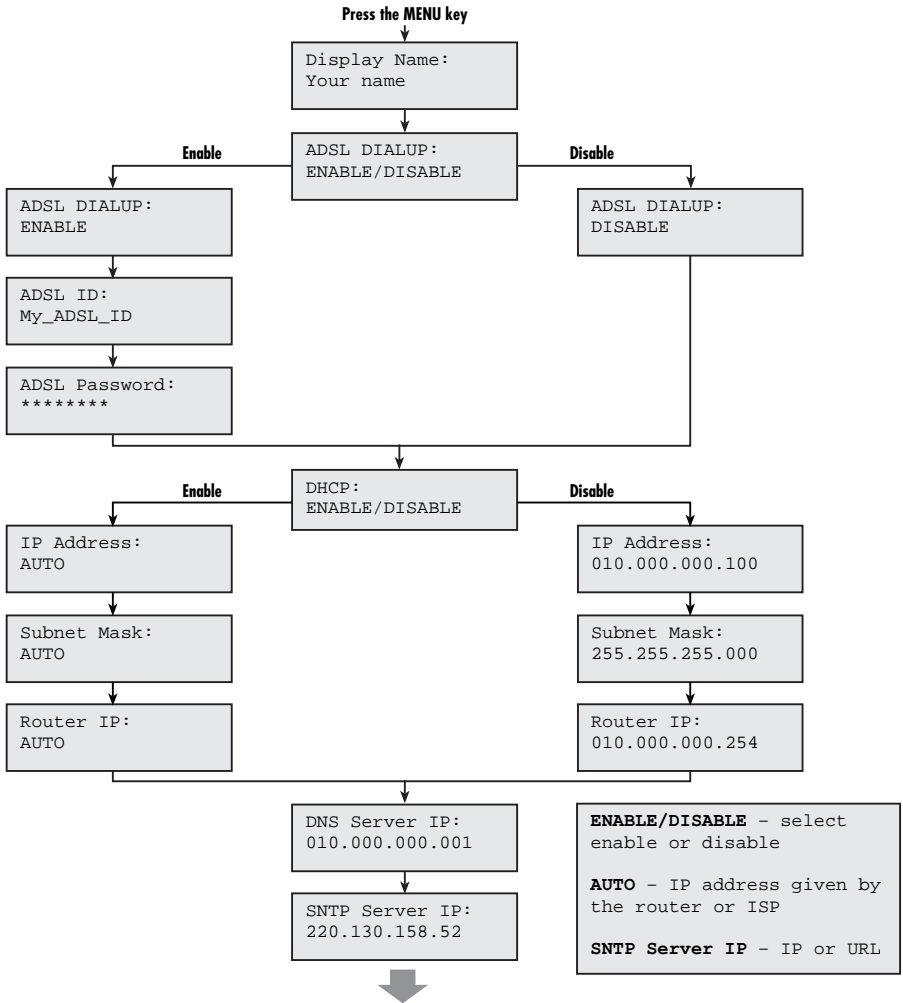


Figure 3. Menu summary, page 1 of 2

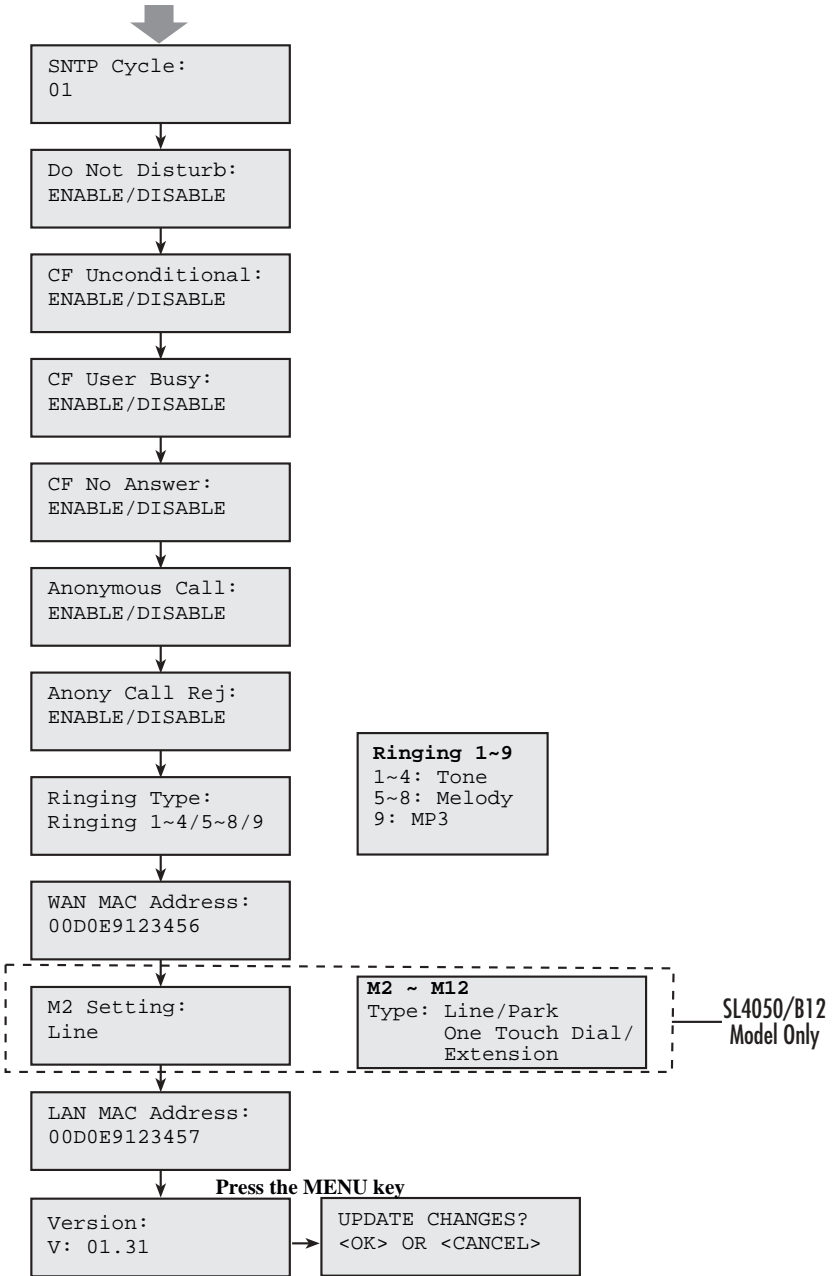




Figure 4. Menu summary, page 2 of 2

4.0 Logging in to the web interface

The configuration menu can be accessed using a web browser.

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 5).



Figure 5. 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**. The following screen displays after logging in:

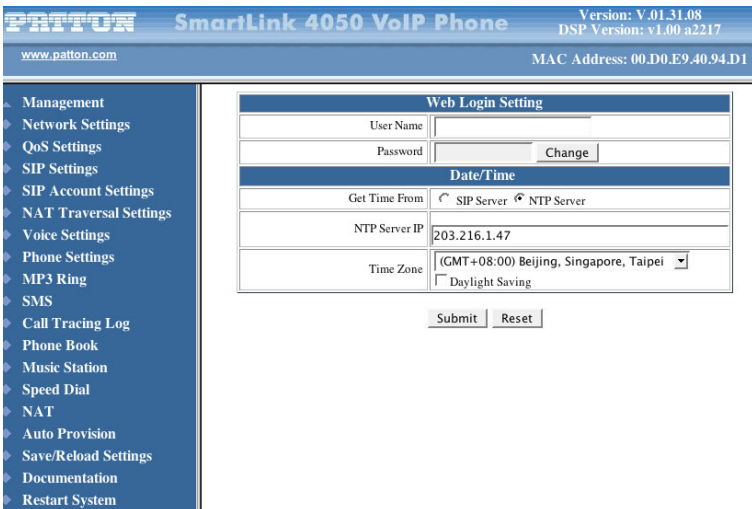


Figure 6. Main window

4.1 Network Settings - 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.

| DHCP / PPPoE / Static IP | |
|--|-------------------|
| <input type="radio"/> DHCP <input type="radio"/> PPPoE <input type="radio"/> Static IP | |
| DNS Setting | |
| DNS Server 1 | 0.0.0.0 |
| DNS Server 2 | 0.0.0.0 |
| MAC Address | |
| WAN MAC | 00.A0.BA.03.CB.D6 |
| LAN MAC | 00.A0.BA.03.CB.D7 |

Figure 7. DHCP configuration window

4.2 Network Settings - PPPoE

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

| DHCP / PPPoE / Static IP | |
|---|-------------------|
| <input type="radio"/> DHCP <input checked="" type="radio"/> PPPoE <input type="radio"/> Static IP | |
| PPPoE ID | |
| PPPoE Password | |
| DNS Setting | |
| DNS Server 1 | 0.0.0.0 |
| DNS Server 2 | 0.0.0.0 |
| MAC Address | |
| WAN MAC | 00.A0.BA.03.CB.D6 |
| LAN MAC | 00.A0.BA.03.CB.D7 |

Figure 8. 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

4.3 Network Settings - Static IP

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

| DHCP / PPPoE / Static IP | |
|---|-------------------|
| <input type="radio"/> DHCP <input type="radio"/> PPPoE <input checked="" type="radio"/> Static IP | |
| IP Address | 10.10.200.49 |
| Router IP | 10.10.1.51 |
| Subnet Mask | 255.255.0.0 |
| DNS Setting | |
| DNS Server 1 | 0.0.0.0 |
| DNS Server 2 | 0.0.0.0 |
| MAC Address | |
| WAN MAC | 00.A0.BA.03.CB.D6 |
| LAN MAC | 00.A0.BA.03.CB.D7 |
| <input type="button" value="Submit"/> <input type="button" value="Reset"/> | |

Figure 9. 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.

4.4 SIP Settings

Click on **SIP Settings** to display the configuration window (see **figure 10**). 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.

| SIP Phone Setting | |
|---|---|
| SIP Phone Port Number | 5060 [1024 - 65535] |
| Registrar Server | |
| Registrar Server Domain Name/IP Address | 10.10.200.6 |
| Registrar Server Port Number | 5060 [1024 - 65535] |
| Authentication Expire Time | 3600 sec. (Default: 3600 sec.) [60 - 9999] |
| Outbound Proxy Server | |
| Outbound Proxy Domain Name/IP Address | |
| Outbound Proxy Port Number | 5060 [1024 - 65535] |
| Send messages via Outbound Proxy | <input checked="" type="radio"/> Disable <input type="radio"/> Enable |

Figure 10. SIP Settings window

- **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

4.5 SIP Account Settings

Click on **SIP Account Settings** to display the configuration window (see **figure 11**). You can have up to four accounts—that is, the SIP phone can receive calls from up to four different phone numbers.

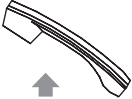



| SIP Account Setting | |
|--------------------------|---|
| Default Account | Account 1 ▾ |
| Account 1 Setting | |
| Account Active | <input type="radio"/> Disable <input checked="" type="radio"/> Enable |
| Display Name | 555 |
| SIP User Name | 555 |
| Authentication User Name | 555 |
| Authentication Password | *** |
| Ring Type | Default ▾ |
| Register Status | Register |

Figure 11. 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.
- **Ring 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.

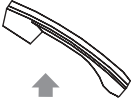



5.0 Making a phone call

5.1 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:

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

5.2 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

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

6.0 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.

6.1 Key Definitions for Internet Radio

| 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 |
| Numerical Keys | The ten numeral keys 0, 1~9 are the quick access keys to the first ten preferred stations on web configuration "Music Station". | | |

6.2 About Internet Radio

- All the keys related to the Internet Radio are described in the table above. 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.

7.0 Additional Information

The complete *SmartLink 4050 Getting Started Guide* is located on the CD-ROM that came with your SIP phone. It can also be downloaded for viewing from www.patton.com.

A.0 Compliance Information

A.1 Compliance

EMC:

- FCC Part 15, Class B
- EN55022, Class B
- EN55024

Safety:

- EN60950-1

A.2 Radio and TV Interference (FCC Part 15)

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).

A.3 EC 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.

A.4 Authorized European Representative

D R M Green, European Compliance Services Limited.
Oakdene House, Oak Road, Watchfield, Swindon, Wilts SN6 8TD, UK

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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)

- 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

Note For additional service and support information, refer to the “Contacting Patton for assistance” chapter of the *SmartLink 4050 Series Getting Started Guide* located on the CD-ROM that came with your SIP phone or available online at www.patton.com.

For additional warranty, trademark, compliance, and technical support information, refer to the *SmartLink 4050 Series Getting Started Guide* located on the CD-ROM that came with your SIP phone or available online at www.patton.com.

