SL4050/10
SmartLink Series 10-Line VoIP SIP Phone
SL4050/2
SmartLink Series 2-Line VoIP SIP Phone

Getting Started Guide

SmartLink 4050/10

SmartLink 4050/2

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

Sales Office: **+1 (301) 975-1000**
Technical Support: **+1 (301) 975-1007**
E-mail: **support@patton.com**
WWW: **www.patton.com**

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Patton Electronics Company, Inc.
7622 Rickenbacker Drive
Gaithersburg, MD 20879 USA
Tel: +1 (301) 975-1000
Fax: +1 (301) 869-9293
Support: +1 (301) 975-1007
Web: www.patton.com
E-mail: support@patton.com

Trademark Statement

Warranty Information

Patton Electronics warrants all SmartLink SIP phone components to be free from defects, and will—at our option—repair or replace the product should it fail within one year from the first date of the shipment.

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 4050/10 10-Line VoIP SIP Telephone and SmartLink 4050/2 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 15 provides information about the SIP phone
- Chapter 2 on page 21 provides installation procedures
- Chapter 3 on page 32 provides procedures for configuring the SIP Phone
- Chapter 4 on page 57 describes how to operate the SIP Phone
- Chapter 5 on page 62 describes how to use the Phone Book feature
- Chapter 6 on page 65 contains information on troubleshooting problems with the SIP Phone
- Chapter 7 on page 67 contains information on contacting Patton technical support for assistance
- Appendix A on page 70 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

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.

WARNING

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.

WARNING

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

CAUTION

Ultimate disposal of this equipment must be handled according to all applicable national laws and regulations.

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

Chapter contents

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 4050/10 and SmartLink 4050/2 packaging. Contact your supplier immediately if an item is missing.

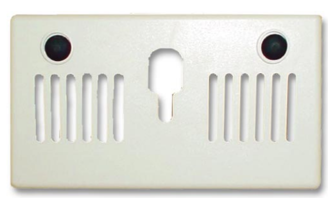
SmartLink 4050/10 VoIP SIP Phone

SmartLink 4050/2 VoIP SIP Phone

Ethernet cable, 10-foot (3-meter), Qty: 2

SmartLink documentation CD-ROM

Power Adaptor (5V DC)



Wall mounting plate (SL4050/10 only)

Overview of SL4050/10 key functions

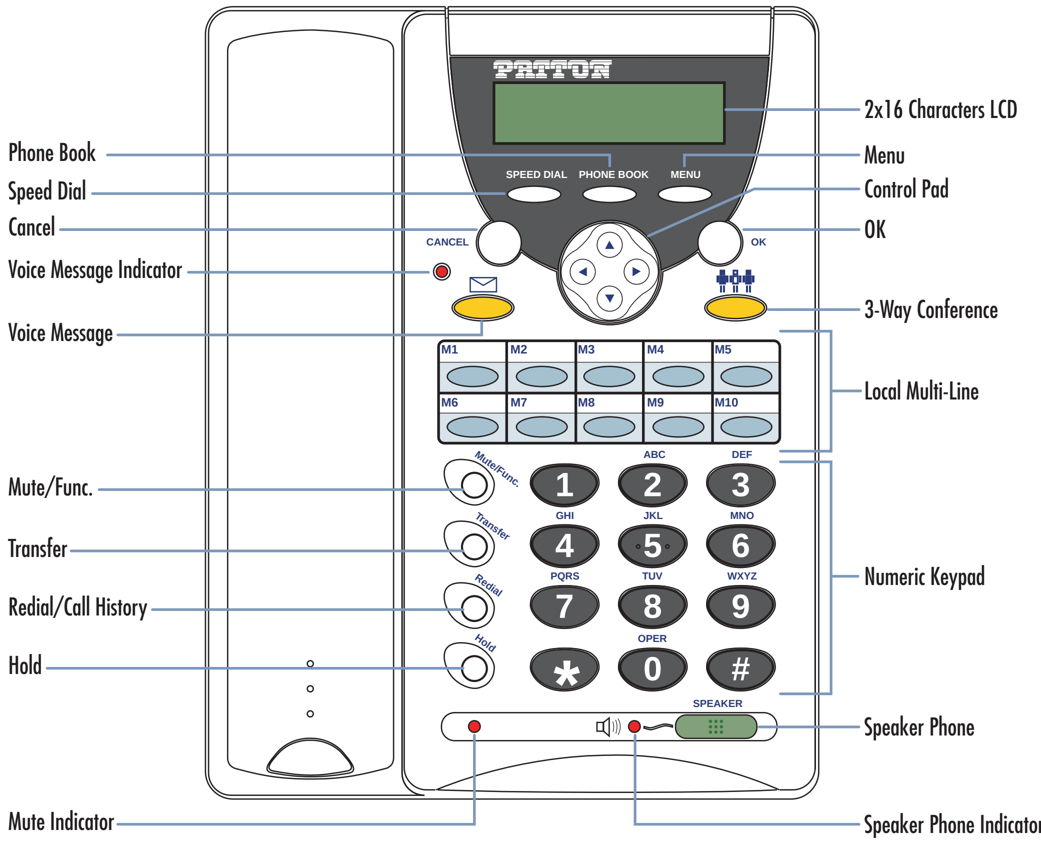


Figure 1. SmartLink 4050/10 SIP Phone controls and indicators

Table 2. Summary of SL4050/10 key functions

Item	Description
LCD Display	Displays menu, time, clock, name, phone number, call status
Menu	Access the phone menu
OK	Confirm setting change, exit menu, dial, save changes
Control Pad	Backspace, scroll up or down, select enable or disable
3-Way Conference	Enable 3-way conference
Local Multiline	Switch to different lines
Numeric Keypad	Input IP/phone number/alphabet characters
Speaker Phone	Enable user to use the phone without using the handset
Speaker Phone Indicator	Indicates that phone is currently in speaker phone mode
Phonebook	Access the phonebook

Table 2. Summary of SL4050/10 key functions (Continued)

Item	Description
Speed Dial	Access the speed dial menu
Cancel	Deny changes, cancel phone calls, ignore phone calls
Voice Message Indicator	Indicates that there is a voice message
Voice Message	Check voice message
Mute/Func.	Disable user's handset microphone so that the person on the other line can not hear anything
Transfer	Transfer the person on the other line to another number
Redial/Call History	Redial last dialed number, access redial menu
Hold	Place the person on the other line on hold
On Hold Indicator	Indicates that the person on the other line is currently placed on hold

Overview of SL4050/2 key functions

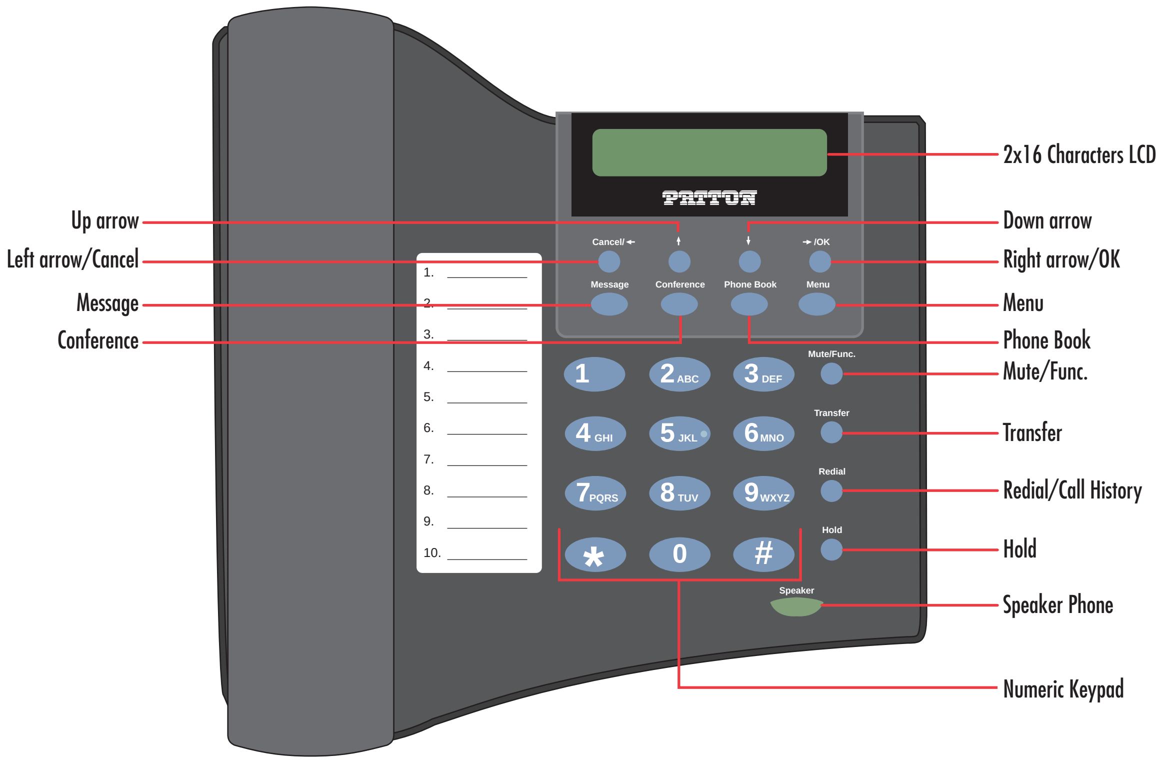


Figure 2. SmartLink 4050/2 SIP Phone controls and indicators

Table 3. Summary of SL4050/2 key functions

Item	Description
LCD Display	Displays menu, time, clock, name, phone number, call status
Left arrow/Cancel	In left-arrow mode, moves cursor on display one character to the left each time the button is pressed/In Cancel mode, pressing this button cancels changes, cancels phone calls, or ignores phone calls
Up arrow	Moves cursor up one line of text each time the button is pressed
Down arrow	Moves cursor down one line of text each time the button is pressed
Right arrow/OK	In right-arrow mode, moves cursor on display one character to the right each time the button is pressed/In OK mode, pressing this button confirms setting changes, confirms exiting from a menu, dials, or saves changes
Menu	Access the phone menu
Phone Book	Access the phone book

Table 3. Summary of SL4050/2 key functions (Continued)

Item	Description
Mute/Func.	Disable user's handset microphone so that the person on the other line can not hear anything
Transfer	Transfer the person on the other line to another number
Redial/Call History	Redial last dialed number, access redial menu
Hold	Place the person on the other line on hold
Speaker Phone	Enable user to use the phone without using the handset
Numeric Keypad	Input IP/phone number/alphabet characters
Conference	Enable 3-way conference
Message	Check voice messages

Chapter 2 Installing the SmartLink SIP Phone

Chapter contents

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 SL4050/10 or figure 4 on page 23 for SL4050/2). 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 PC port of the SIP phone (see figure 3 for SL4050/10 or figure 4 on page 23 for SL4050/2). Plug the other end of the cable into the Ethernet port on the PC.

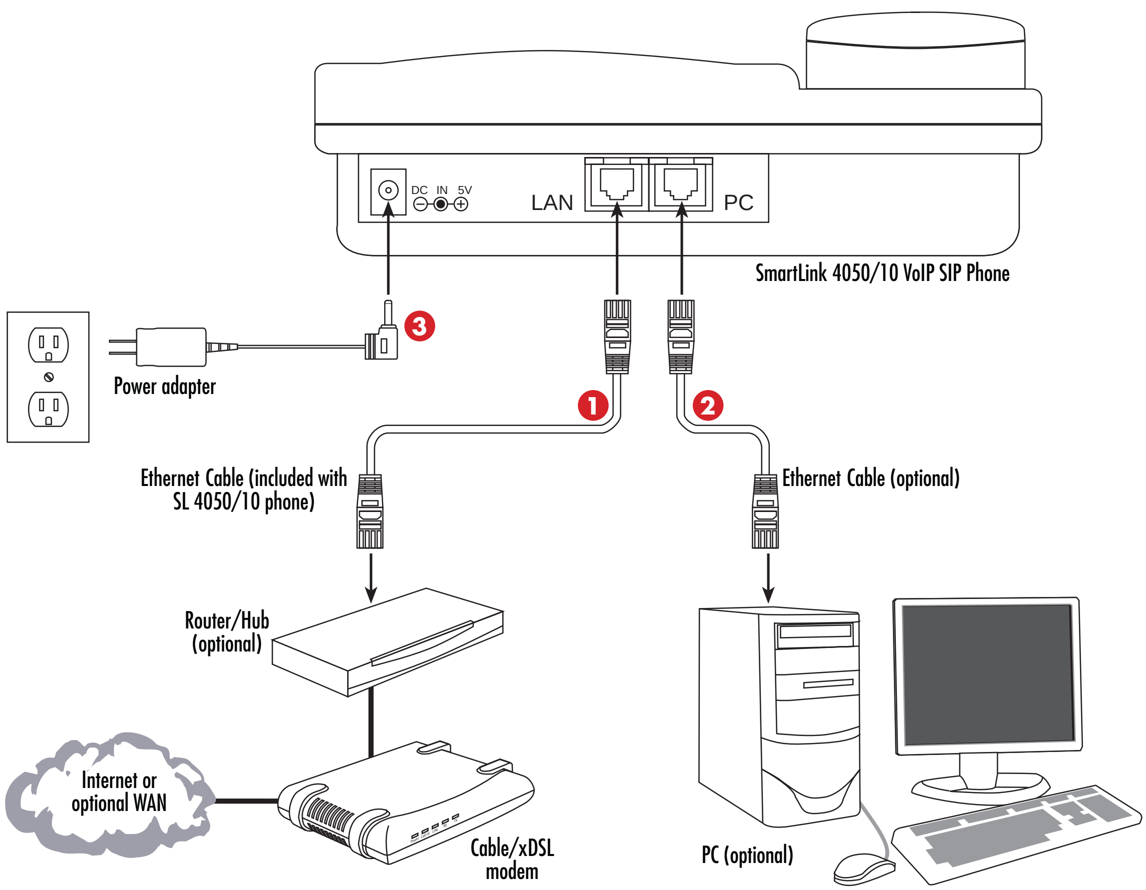


Figure 3. Connecting the SL4050/10 SIP Phone

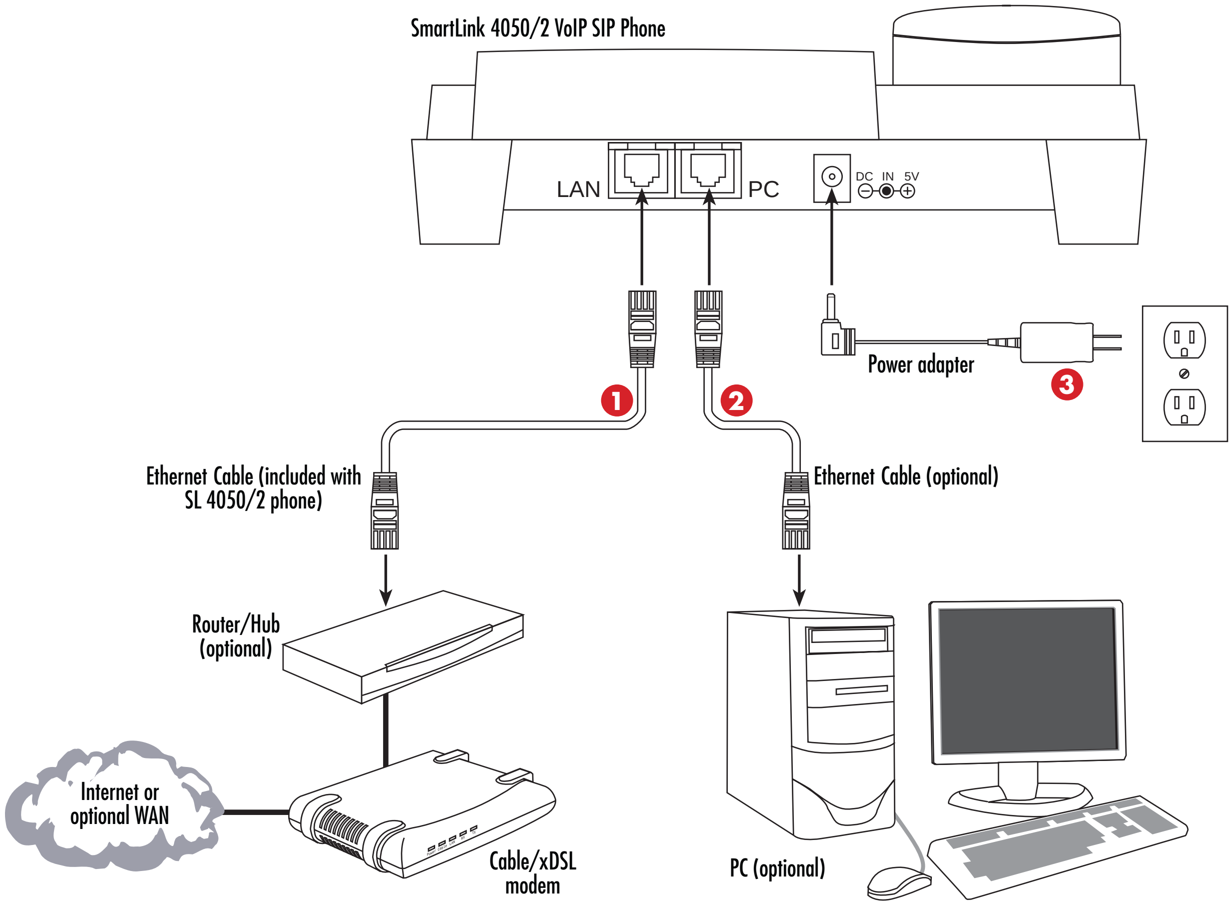


Figure 4. Connecting the SL4050/2 SIP Phone

3 Plug the power adapter barrel connector into the power connector on the SIP phone (see figure 3 on page 22 for SL4050/10 or figure 4 for SL4050/2). Plug the other end of the power adapter into an AC electrical outlet.

Setting up the VoIP SIP phone

Menu summary

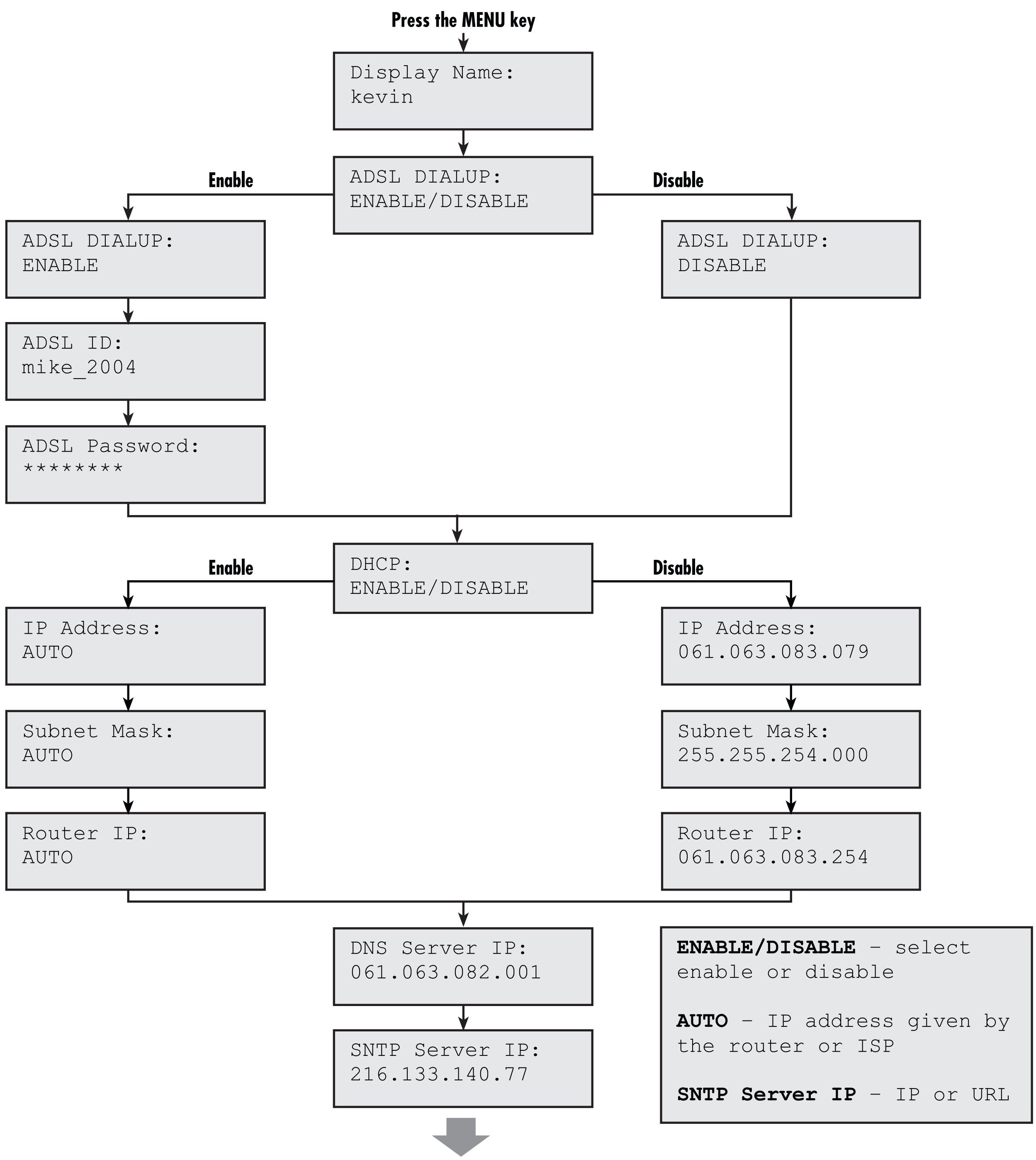


Figure 5. Menu summary, page 1 of 2

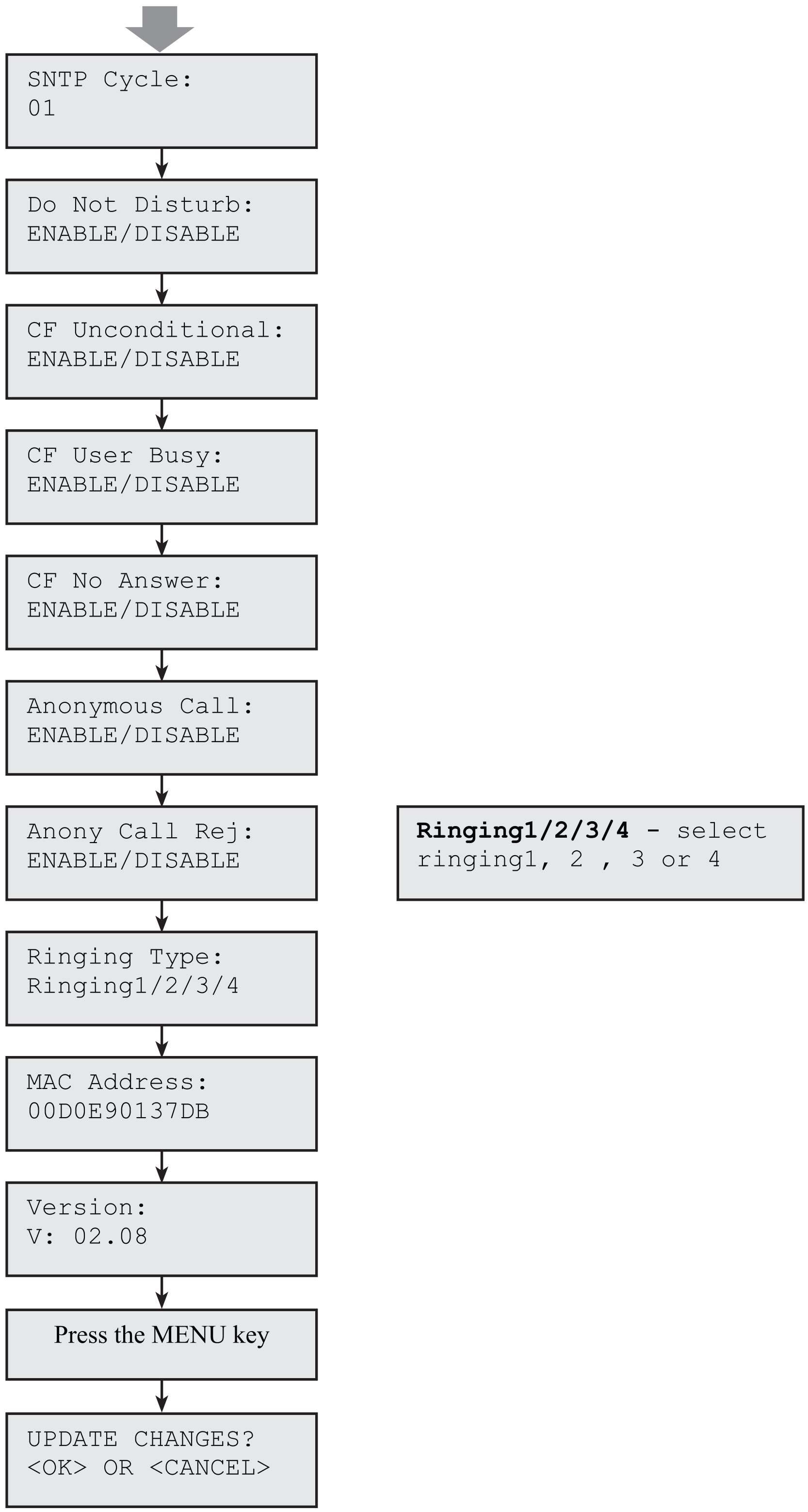


Figure 6. Menu summary, page 2 of 2

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 left and right arrows on the control pad to select *ENABLE* or *DISABLE*. The left arrow 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:
kevin
```

Note To type text characters, press the appropriate key on the numeric keypad (see figure 1 on page 17). 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.

Display Name

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

2. Use to select *ENABLE*

```
ADSL DIALUP:
ENABLE
```

3. Press

4. Enter the ADSL ID

```
ADSL ID:
My_ID
```

5. Press

6. Enter the ADSL password

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

DISABLE ADSL dialup

1. Press ▼

2. Use ► 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 ◄ or ► to set DHCP to *ENABLE*

```
ADSL DIALUP:
ENABLE
```

3. Press ▼. The IP address is automatically acquired

```
IP Address:
061.063.083.079
```

4. Press ▼. The subnet mask is automatically acquired

```
Subnet Mask:
255.255.254.000
```

5. Press ▼. The router IP address is automatically acquired

```
Router IP:
061.063.083.254
```

DISABLE DHCP

1. Press ▼

2. Use ◄ or ► to set DHCP to *DISABLE*

```
DHCP:
DISABLE
```

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

```
IP Address:
061.063.083.079
```

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

```
Subnet Mask:
255.255.254.000
```

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

```
Router IP:
061.063.083.254
```

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:
061.063.082.001
```

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. 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:
216.133.140.77
```

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:
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 3, "Using the configuration menu" on page 32 for more information on call forwarding.

1. Press ▾.
2. Use ◂ or ▸ to select *ENABLE* or *DISABLE*

```
CF Unconditional:
DISABLE
```

CF (call forward) User Busy

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

1. Press ▼.
2. Use ◄ or ► to select *ENABLE* or *DISABLE*

```
CF User Busy:
DISABLE
```

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:
DISABLE
```

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:
DISABLE
```

Ringing Type

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

1. Press ▼.
2. Use ◄ or ► to select *ENABLE* or *DISABLE*

```
Ringing Type:
Ringing1/2/3/4
```

Note Pressing MENU to exit menu. When asked to save or cancel, press OK to *SAVE*.

MAC Address

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

1. Press ▾.
2. The MAC address is displayed

```
MAC Address:
00D0E90137DB
```

Version

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

1. Press ▾.
2. The firmware version is displayed

```
Version:
V: 02.08
```

Language Selection

The VoIP SIP phone supports the English and Japanese languages.

1. Press Mute/Func. followed by 1

```
Language:
English
```

2. Use ◂ or ▸ to select the preferred language.
3. Press OK when finished.

Time Format

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

1. Press Mute/Func. followed by 2

```
Time Format:
24Hours
```

2. Use ◂ or ▸ to select the time format.
3. Press OK when finished.

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
2. Use 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.

Congratulations, your SmartLink VoIP SIP Phone is ready to use!

Chapter 3 **Using the configuration menu**

Chapter contents

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 the *MENU* button and scroll down to the IP address.

   The login window displays (see Figure 7).

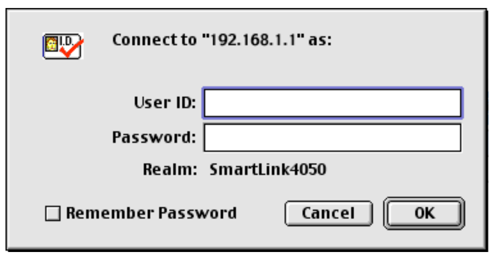


Figure 7. Login window

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

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*). The main window displays (see Figure 8).

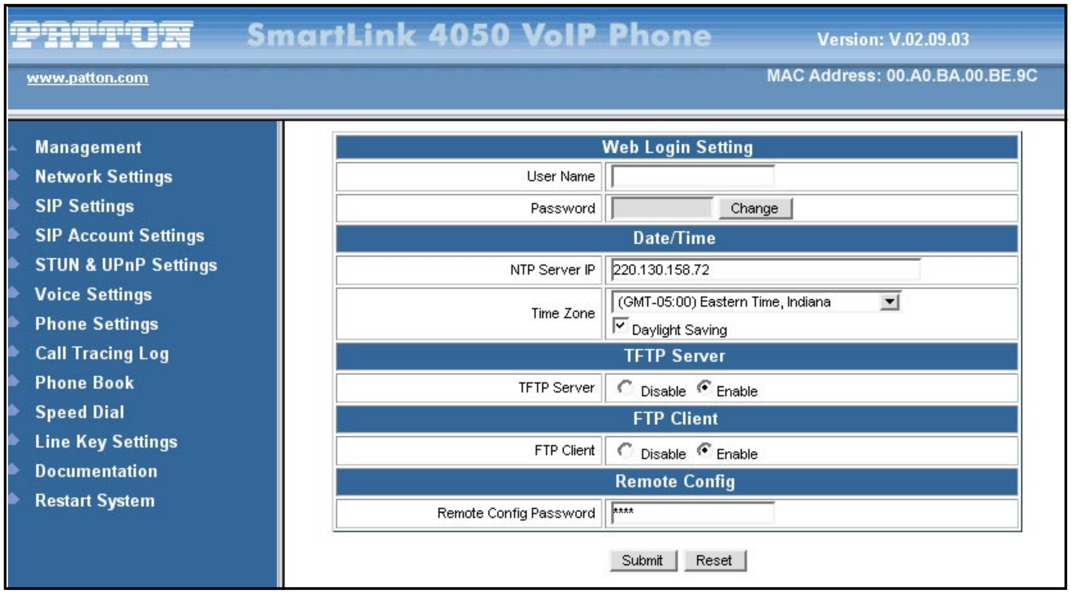


Figure 8. Main window

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 (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.

Remote Config Password

Remote password to access the configuration menu from VoIP software. (You can download this software from your supplier's website). Default password is *1234*.

Management Settings—Restore Factory Setting

Click on *Management > Select Restore Factory Setting*. The message *Press [Restore] button to restore the default setting*! displays (see figure 9).

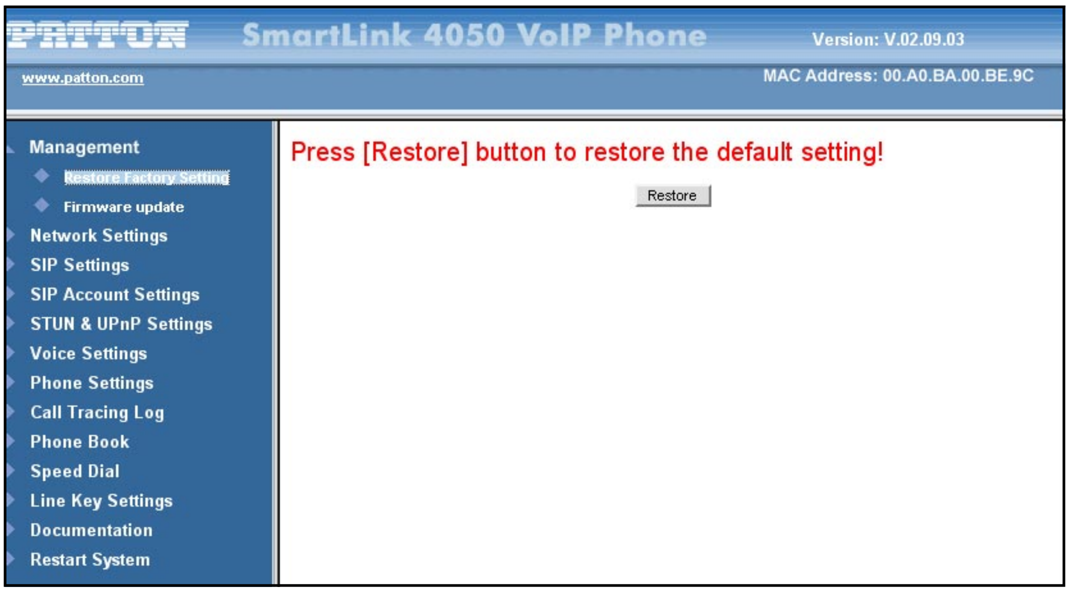


Figure 9. Restore Factory Setting window

Restore Factory Setting

Click the **Restore** button to return all settings back to factory default settings.

Management Setting—Firmware update

Click on *Management* > *Firmware update* to display the *Firmware update* window (see figure 10).

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

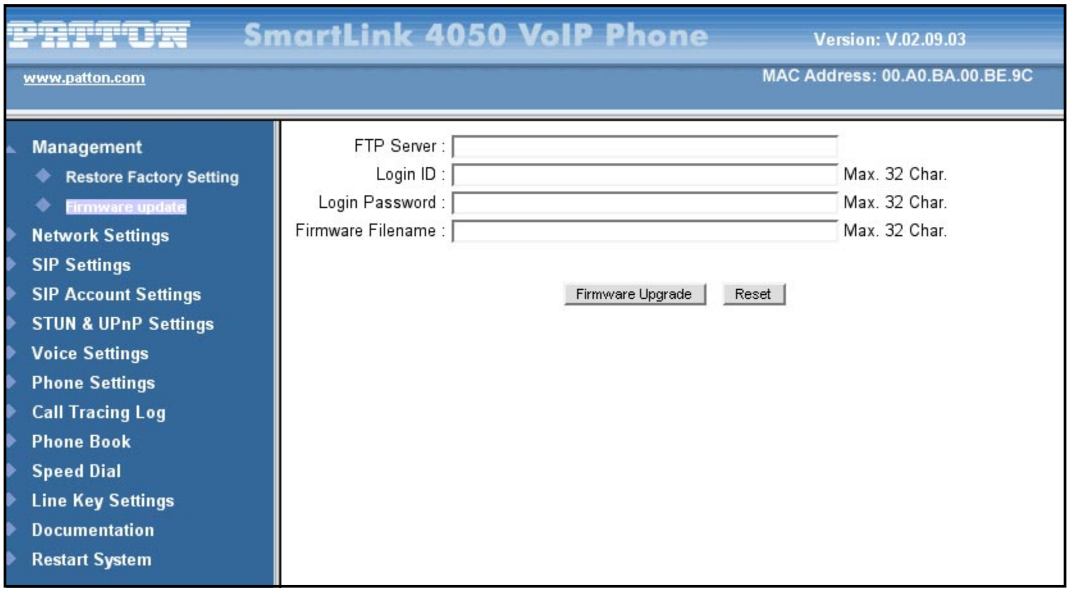


Figure 10. Firmware update window

FTP Server

FTP server address.

Login ID

Login ID provided by your supplier.

Login Password

Login password provided by your supplier.

Firmware Filename

Updated firmware filename. Do not change the file name unless told to do so by your supplier.

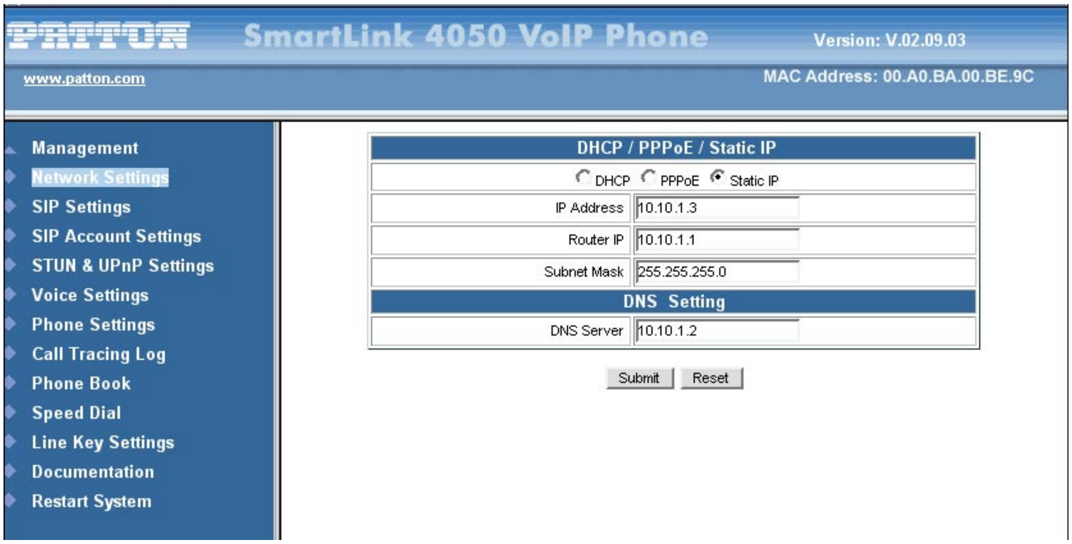


Figure 11. Network Settings window

Network Setting—DHCP

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 Server" for details)
- **PPPoE**—Select *PPPoE* if your ISP uses PPPoE (most DSL users choose *PPPoE*) (see section "DHCP Server" 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 "DHCP Server" for details)

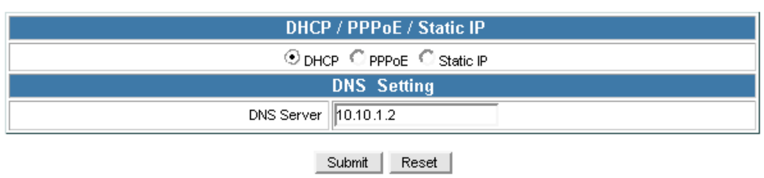


Figure 12. DHCP configuration window

DHCP Server

Dynamic host configuration protocol (DHCP) server address. This IP address information is obtained automatically from your ISP.

DNS Setting

The DNS address is provided by your ISP.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

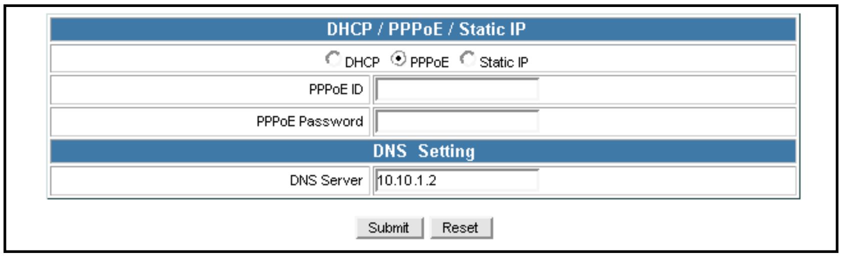


Figure 13. PPPoE configuration window

PPPoE

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.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

> **Note** After modifying the IP address, click *Restart System*, then click the **Restart** button so the new settings can take effect.

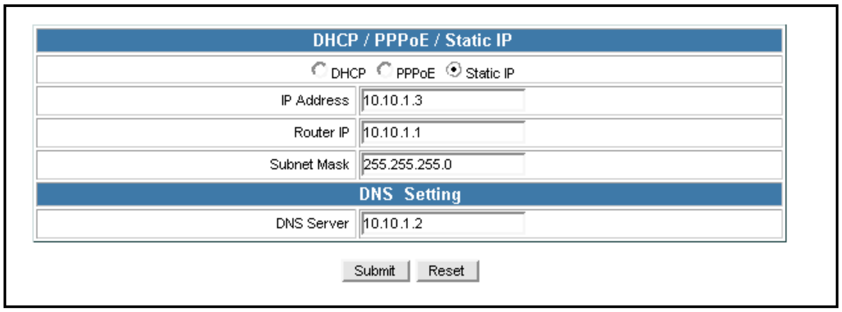


Figure 14. Static IP configuration window

Static IP

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.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

> **Note** After modifying the IP address, click *Restart System*, then click the **Restart** button so the new settings can take effect.

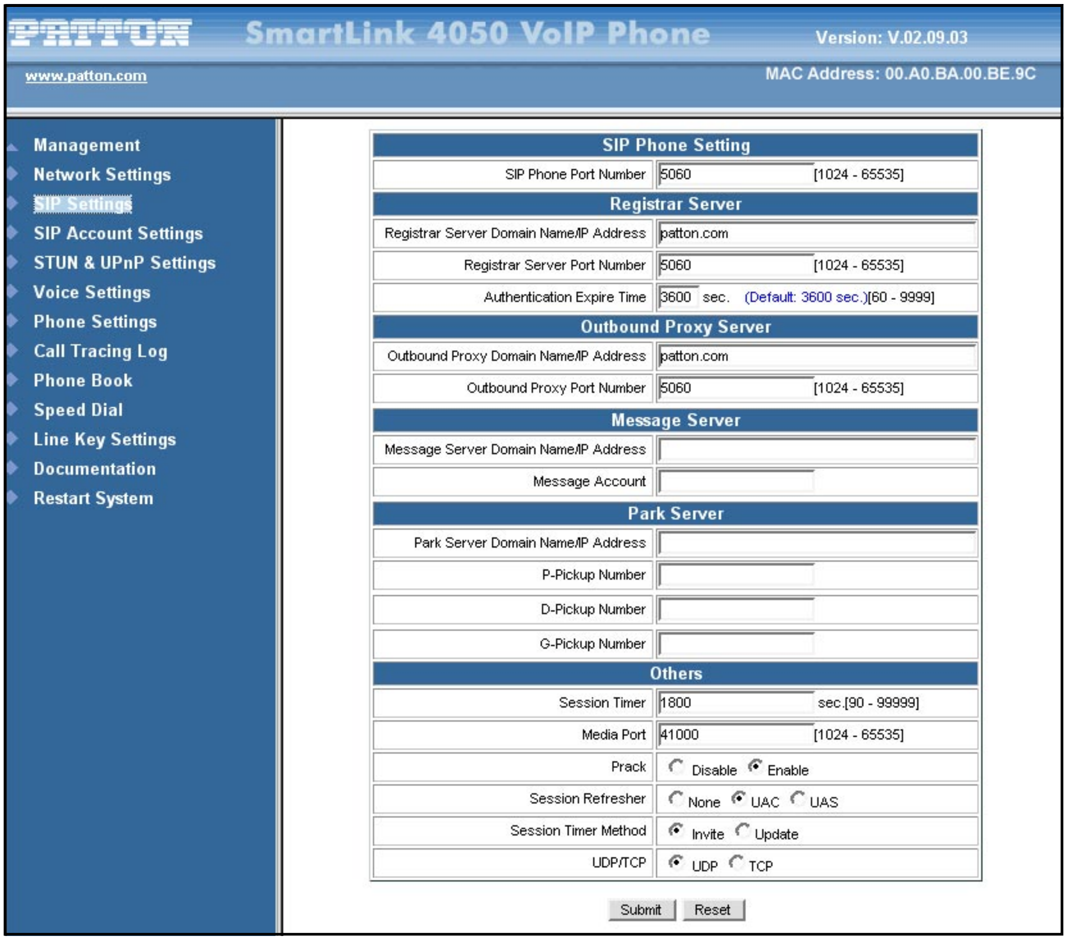


Figure 15. SIP Settings window

SIP Settings

Click on *SIP Settings* to display the configuration window (see figure 15). 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. 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

SIP phone port number.

Registrar Server

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 Server

Outbound Proxy Domain Name/IP Address

Outbound proxy domain name or IP address.

Outbound Proxy Port Number

Outbound proxy port number.

Message Server

Domain name or IP address.

Park Server

Domain name or IP address.

Others

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

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*.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

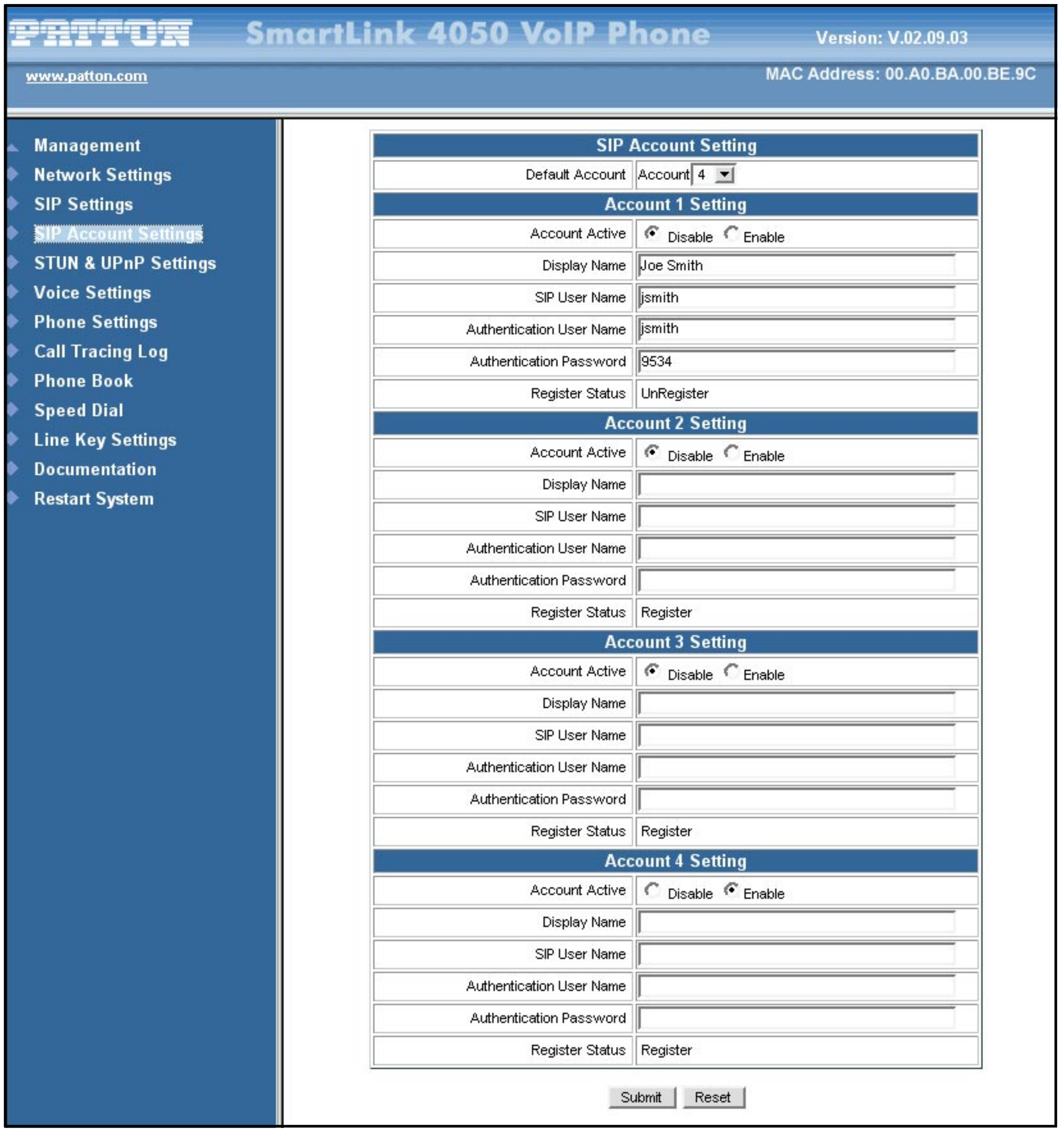


Figure 16. SIP Account Settings window

SIP Account Settings

Click on *SIP Account Settings* to display the configuration window (see figure 16). You can have up to four accounts—that is, the SIP phone can receive calls from 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.

Saving your work
When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

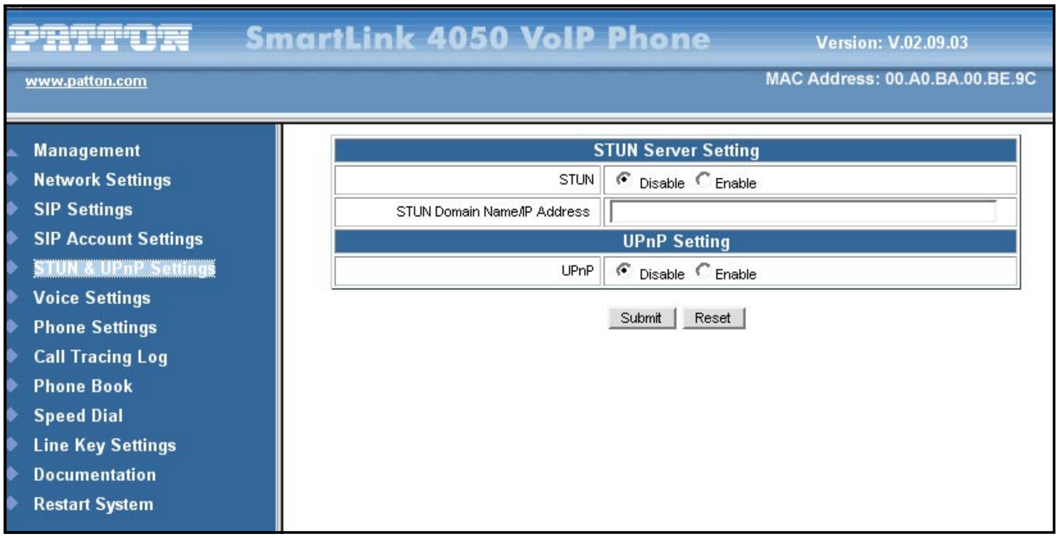


Figure 17. STUN & UPnP Settings

STUN & UPnP Settings

Click on *STUN & UPnP Settings* to display the configuration window (see figure 17).

STUN Server Setting

STUN

Simple traversal of user datagram protocol through network address translators (STUN) 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. Click to **Enable** or **Disable** STUN.

STUN Domain Name/IP Address

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

UPnP Setting

UPnP

Click to **Enable** or **Disable** universal plug and play (UPnP).

> **Note** Some NAT supports UPnP so STUN is not required and must be disabled.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

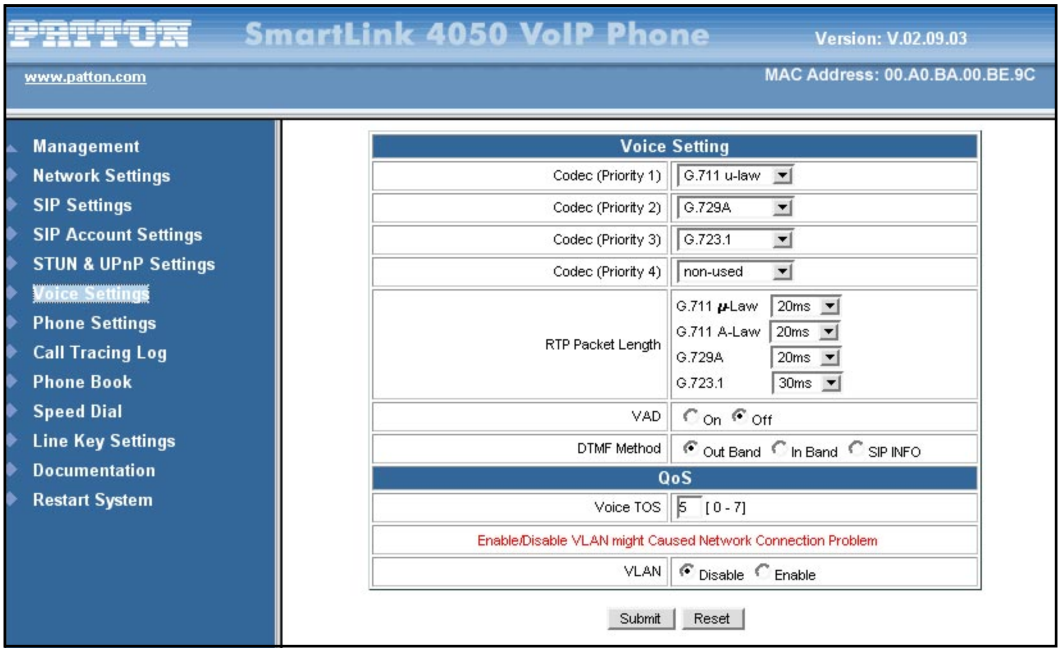


Figure 18. Voice Setting and QoS

Voice Settings

Click on *Voice Setting and QoS* to display the configuration window (see figure 18).

Voice Setting

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:

- Out Band
- In Band
- SIP INFO

QoS

Voice TOS

Sets the type of service for this Internet datagram.

VLAN

Enable or **Disable** virtual LAN.

IMPORTANT

Enabling or disabling VLAN may cause network connection problems.

VLAN Priority

Set the virtual LAN priority.

VLAN ID

Virtual LAN ID.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

Figure 19. Phone Settings window

Phone Settings

Click on *Phone Settings* to display the configuration window (see figure 19). You can only enable or disable call forwarding from the SIP phone **MENU** key. With the web browser, you can enter the forwarded phone numbers in the *Phone Setting* window.

Phone Setting

Tone Setting

Select the tone for particular country.

Ringer Type

Selects the type of ring (1 to 4).

Hold Tone

Select whether a **Melody** or **Tone** will play when the **HOLD** key on the SIP phone is pressed.

Do Not Disturb

Click **Enable** to reject all incoming calls. Click **Disable** to accept incoming calls.

Call Waiting

Click to **Enable** or **Disable** call waiting.

Anonymous Call

Select how much information about the SIP phone user will be sent to the called party's phone:

- **Disable**—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.
- **Full URI**—If **Full URI** is selected, only user name is displayed on the receiver's phone when the user makes a phone call.
- **Display Name**—If **Display Name** is selected, only name is displayed on the receiver's phone when the user makes a phone call.

Anonymous Call Reject

Click **Enable** to reject anonymous calls. Click **Disable** to accept anonymous calls.

Call Forward

Select how call forwarding is handled:

- **No Answer**—Click **No Answer** to enable call forwarding to another number when no one answers the phone after 180 seconds (default). The timer can be changed from 0–600 seconds (see section "Timer" to change the timer setting). Enter the call forwarding number in the text box.
- **Busy**—Click **Busy** to enable call forward to another number when user is busy on the phone. Enter the call forwarding number in the text box.
- **Unconditional**—Click **Unconditional** to transfer all incoming calls to another number. Enter the call forwarding number in the text box.

Timer

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).

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

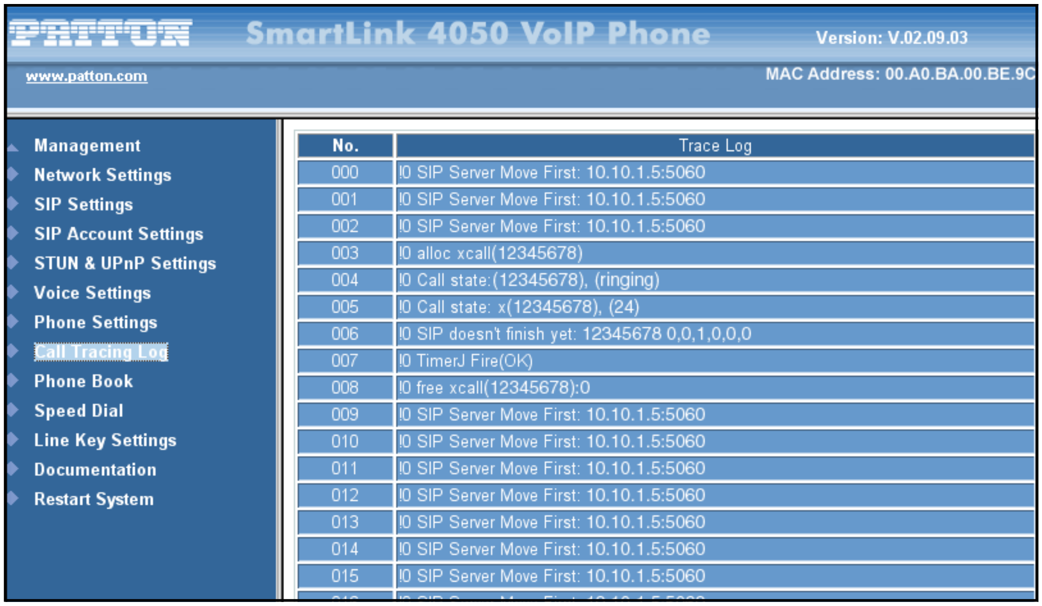


Figure 20. Call Tracing Log window

Call Tracing Log

Click on *Call Tracing Log* to display the configuration window (see figure 20). The call tracing log keeps a record of all the phone activities. This log is used by our Patton technicians to troubleshoot hardware problems.

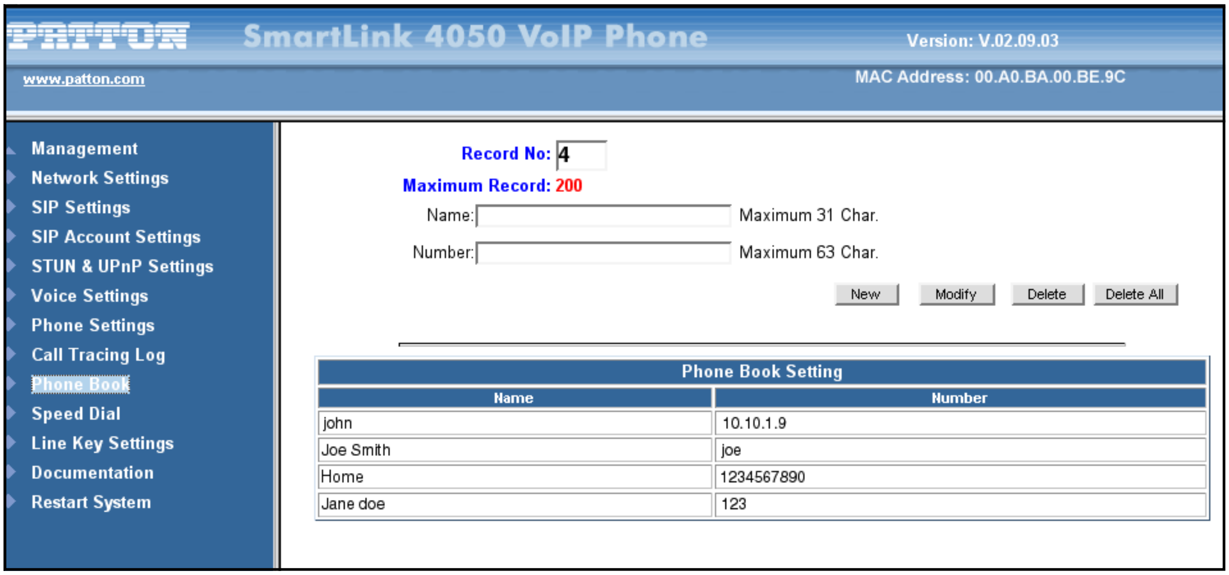


Figure 21. Phone Book window

Phone Book

Click on *Phone Book* to display the configuration window (see figure 21). The Phone Book window enables users to add, modify, or delete phone numbers:

- To add a name, type the name (up to a maximum of 31 characters) and number (up to a maximum of 63 characters), then click **New**
- To modify or delete a name, select the name from the list and click **Modify** to edit the listing or **Delete** to delete the listing
- To delete all names from the listing, click **Delete All**.

Phone Book Setting

Name

The name you would like to add.

Number

The phone number that corresponds to the name.

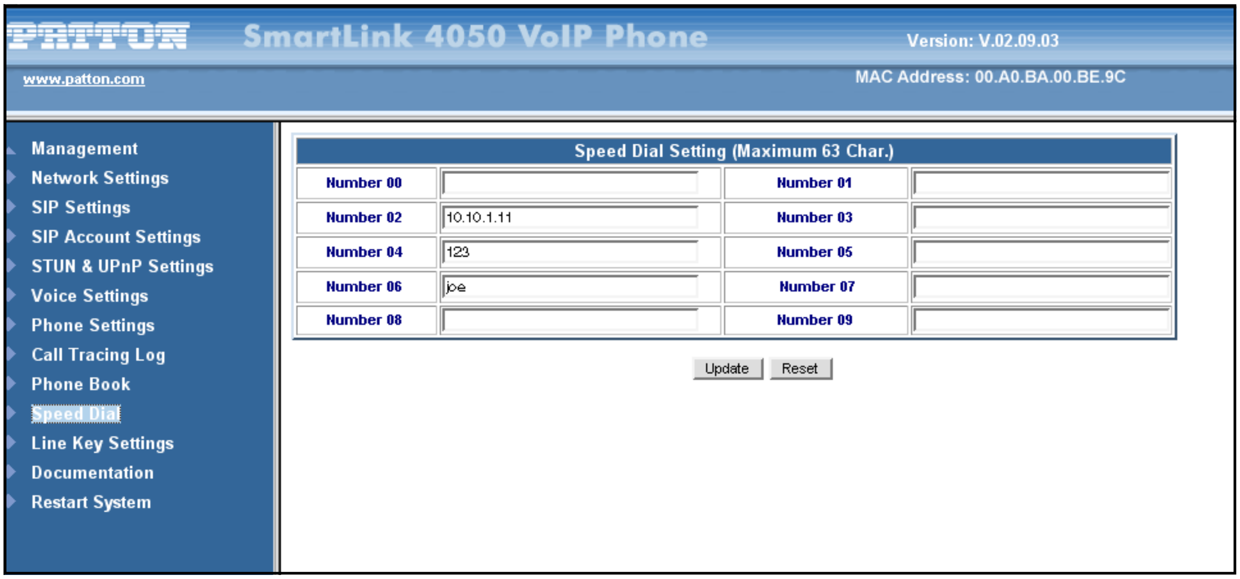


Figure 22. Speed Dial window

Speed Dial

Click on *Speed Dial* to display the configuration window (see figure 22). Speed dial numbers can be accessed from the IP phone. Refer to section "Speed Dialing" on page 58 for speed dial info.

Speed Dial Setting (Maximum 63 Char.)

Number 0x

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

Saving your work

When you finish configuring the settings, click the **Update** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

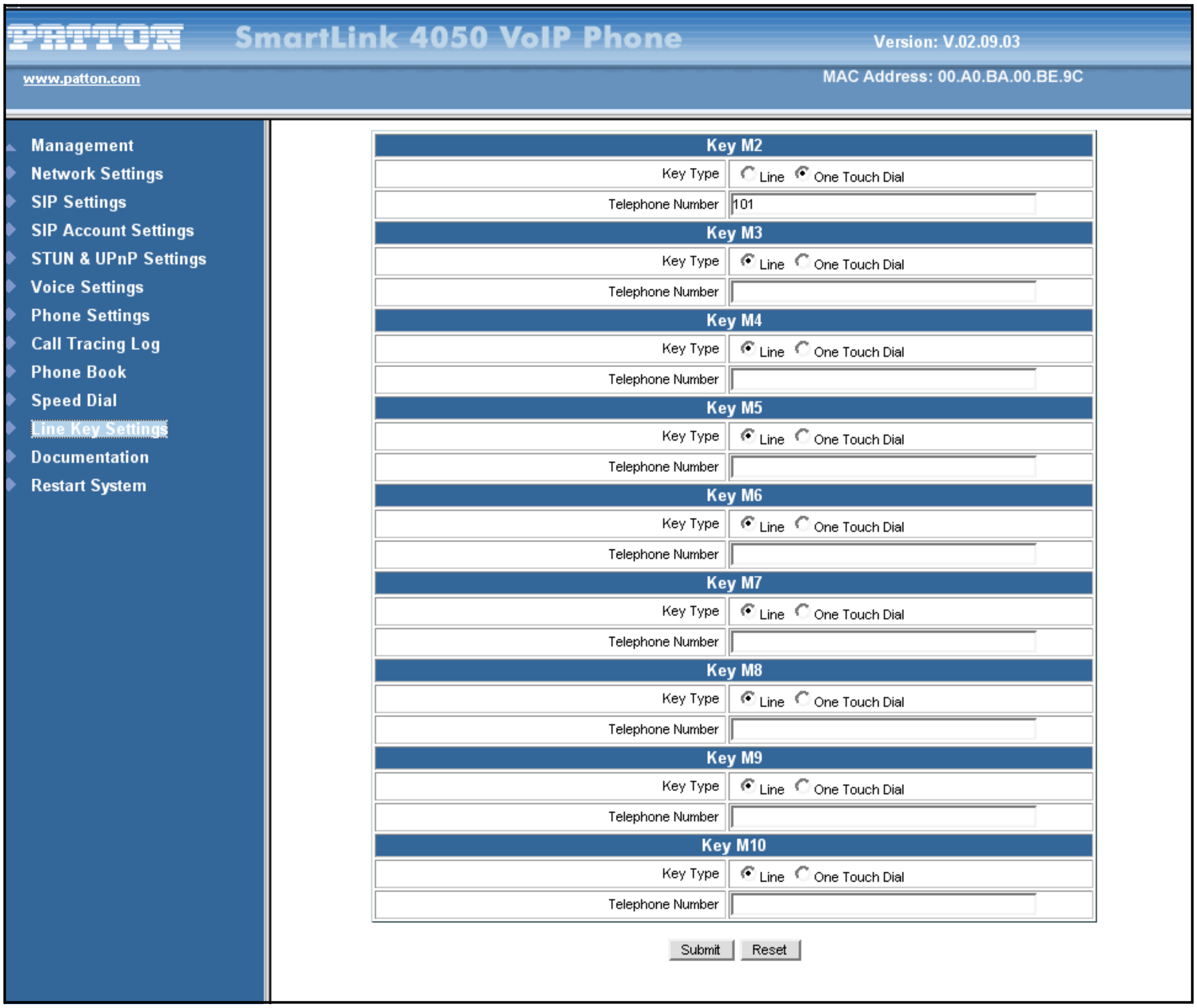


Figure 23. Line Key Settings window

Line Key Settings

Click on *Line Key Settings* to display the configuration window (see figure 23). Line Key Settings enable the user to customize line keys as a line or one-touch dial. Refer to section "One-Touch Dialing" on page 60 for one-touch dial info.

Key Type

Select local multiline key (M2–M10) as **Line** or **One-Touch Dial** function.

Telephone Number

Enter the phone number to be dialed if the *On-Touch Dial* function is selected.

Saving your work

When you finish configuring the settings, click the **Submit** button to save the changes. Otherwise, click the **Reset** button to cancel the changes.

Documentation

Figure 24. Documentation link

Click the **Documentation** link (see figure 24) to download and display the *SmartLink 4050 Series Getting Started Guide* in portable document format (PDF).

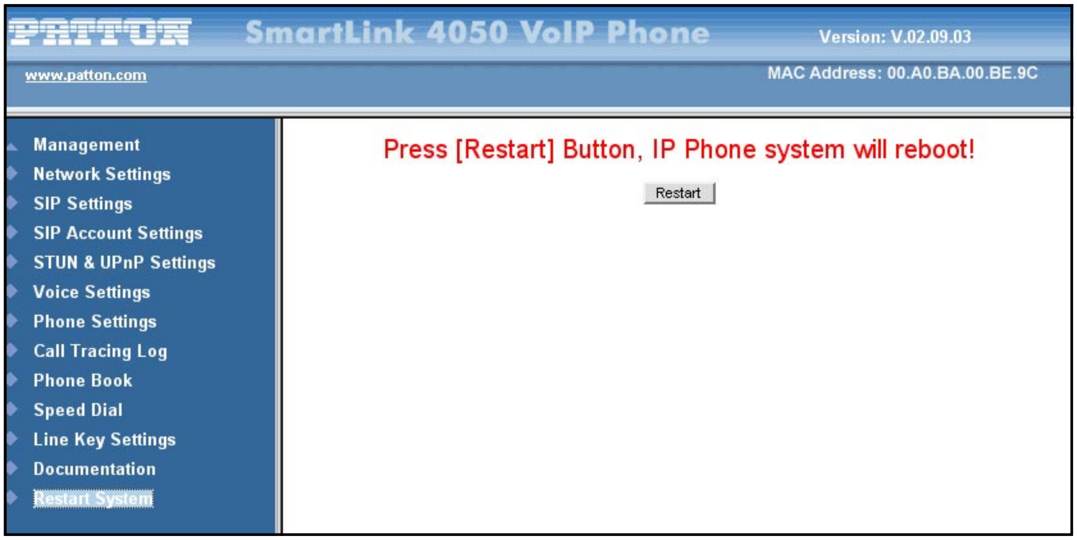


Figure 25. Restart System window

Restart System

Click on *Restart System*. The message *Press [Restore] Button, IP Phone system will reboot!* displays (see figure 25). Click the **Restart** button so all modifications will take effect.

Chapter 4 Operating the VoIP SIP phone

Chapter contents

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 a speed dial number. For example, to dial *08* press
   * 0 8

Answering a phone call

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

Switching to another line

While having a conversation, press the flashing local multiline key **M1** to **M10** to switch to another line.

Mute

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

While having a conversation, press the **Mute** button. To resume the conversation, press **Mute** again.

Call Transfer

While having a conversation:

1. Press the **Transfer** button 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 the **Transfer** button again to transfer the call.

Redial

Note To return to idle mode, press the **CANCEL** button.

Last Dialed Number

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

Through Call History

1. Press the **Redial** button. Do not lift the handset when you press **Redial**.
2. Press the **Redial** button again to cycle through the dialed, missed, and received calls.
3. Press the down button to scroll down the dialed, missed, and received numbers until the desired number is displayed on the screen.
4. Press the left or right buttons to show detail information on every call.
5. Lift the handset or press the **OK** button.

On Hold

While having a conversation, press the **Hold** button. To resume the conversation, press **Hold** again.

Call Forwarding

Refer to sections "Call forwarding" on page 28 and "Call Forward" on page 50 to set up call forwarding.

Call Waiting (internal/external)

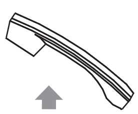
While having a conversation:

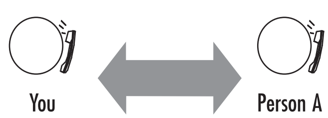
1. Press the flashing local multiline key **M1** to **M10** button to pick up another incoming call. The first caller is automatically placed on hold.
2. Press the flashing local multiline key **M1** to **M10** button of the first caller to retrieve the call again.

One-Touch Dialing

Using a local multiline key (M2–M10) set for one-touch dialing, press the pre-programmed local multiline key **M1** to **M10** to make a call.

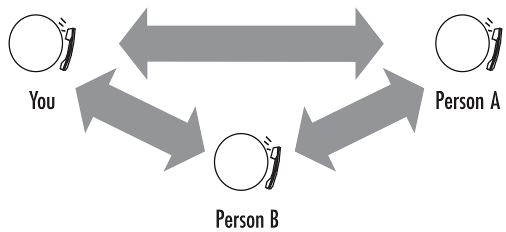
Three-Way Conferencing

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



2. After **Person A** picks up the phone, press the 3-way conference key 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 3-way conference key to begin the 3-way conference.



Chapter 5 Using the Phone Book

Chapter contents

Dialing from the Phone Book

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

Storing a number

1. Press and hold the **PHONE BOOK** PHONE BOOK button until *Name* displays on the screen.
2. Use the numeric keypad to type a name, then press **OK** OK

Note To type text characters, press the appropriate key on the numeric keypad (see figure 1 on page 17). 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. Use the numeric keypad to type the number that corresponds to the name, then press **OK** OK
4. Press **OK** OK again to save the changes to the Phone Book.
5. Repeat steps 1 through 4 to store additional phone numbers.

Editing a Phone Book listing

1. Press the **PHONE BOOK** PHONE BOOK button to access the phone book.
2. Press the down button to scroll down the list until the desired name is displayed on the screen.
3. Press the **PHONE BOOK** PHONE BOOK button again.
4. Select *Edit* and press **OK** OK to begin editing.
5. Use the numeric keypad to type a new name, then press **OK** OK

Note To type text characters, press the appropriate key on the numeric keypad (see figure 1 on page 17). 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**.

6. Use the numeric keypad to type the new number that corresponds to the name, then press **OK** OK
7. Press **OK** OK to save changes, overwriting the previous name and phone number.

Deleting a Phone Book listing

1. Press the **PHONE BOOK** PHONE BOOK button to access the phone book.
2. Press the down button to scroll down the list until the name you want to delete is selected.
3. Press the **PHONE BOOK** PHONE BOOK button again.
4. Select *Delete* and press **OK** OK to delete the listing.
5. Press **OK** OK to save the change to the Phone Book.

Chapter 6 Troubleshooting

Chapter contents

Introduction

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

Symptom	Recommended action
No dial tone	Do the following: • Check to see if there are any loose connections. • Verify that the power cord is connected properly. • Verify that 120 VAC is available at the power outlet. • Contact your service provider to see if there is a problem with your WAN or Internet connection. If the problem still exists, replace the SIP phone.
Nothing displayed on the LCD screen	Do the following: • Verify that the power cord is connected properly. • Verify that 120 VAC is available at the power outlet. If the problem still exists, replace the SIP phone
How do I update the SIP Phone firmware?	The SIP Phone automatically updates firmware when it powers up (while connected to the Internet).
Why can't I dial my friend's SIP number?	Do the following: • 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 SIP Phone automatically detects for new firmware when you unplug the power. If new version is available the phone will automatically update the firmware. If the firmware is not updating, do the following: • Verify that the FTP address is correct. • Check with your supplier to verify that the 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.

Chapter 7 Contacting Patton for assistance

Chapter contents

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**

Chapter contents

Compliance

EMC Compliance:

FCC Part 15, Class B

EN55022, Class B

EN55024

Safety Compliance

EN60950-1

FCC Warning

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the Instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause radio interference in which case the user will be required to correct the interference at his or her own expense.

Radio and TV Interference

The SmartLink SIP Phone 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. The SmartLink SIP Phone have been tested and found to comply with the limits for a Class B computing device in accordance with 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 SmartLink SIP Phone does cause interference to radio or television reception, which can be determined by disconnecting the unit, the user is encouraged to 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-Mark Warning

This is a Class B product. In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

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.

Appendix B Specifications

Chapter contents

Protocol

IETF SIP RFC3261

H.323

Network Interface

RJ45 x 2

10/100BaseT

Call Features

Call transfer (unattended/blind & announced)

Call forward (busy/no answer/unconditional)

Anonymous call blocking

Out-of-band DTMF (RFC 2833)

Message waiting indicator

Call park/pickup (support SIP required)

Group pickup (Support SIP server required)

Voice Codec

G.711µ-law

G711a-law

G.723.1 (5.3k)

G.723.1 (6.3k)

G.729a/b

SIP Server Support

Registrar Server (setting from web)

Outbound Proxy (setting from web)

IP Assignment

Static IP

DHCP

PPPoE

Security

HTTP 1.1 basic/digest

Authentication for Web setup

MD5 for SIP authentication (RFC 2069/ RFC 2617)

QoS

ToS field

IEEE 802.1q VLAN

Tone

DTMF –(inband, out of band, SIP info)

4 selectable ring tones

Ring back tone (local & remote)

Dial tone

Busy tone

Dial Methods

Direct IP call without SIP registration

Dial registered 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

Firmware Upgrade

TFTP

Auto/manual provisioning system

NAT Traversal

UPnP

STUN

TCP/IP

IP/TCP/UDP/DHCP/RTP/RTCP

ICMP/HTTP/SNTP/TFTP/DNS

Configuration

Key & LCD configuration

Web browser configuration

Auto/manual provisioning system