LG-ERICSSON

IP Phone
Installer Guide

LIP-8002/8002A
SIP (Session Initiation Protocol)
iPECS

Please read this manual carefully before installation.
Retain it for future reference.

Regulatory and Safety Notices

LIP-8002/8002A

1.   Radio Frequency Emissions

FCC Compliance statement:

This device complies with Part 15 rules. Operation is subject to the following two conditions;

1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interference that may cause undesired operation.

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 in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and the receiver.
- Connect the equipment into an outlet on a different circuit from that to which the receiver is connected

If problems persist, consult the dealer or an experienced radio/TV technician for help.

Canadian Compliance statement:

This Class B digital apparatus complies with Canadian ICES-003.

Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada.

This device complies with Class B Limits of Industry Canada. Operation is subject to the following two conditions;

1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interference that may cause undesired operation.

European Union Declarations of Conformity:

LG-Ericsson Co., Ltd. declares that the equipment specified in this document, which bears the "CE" mark, conforms to the European Union Radio and Telecommunications Terminal Equipment Directive (R&TTE 1999/5/EC) including,

Electromagnetic Compatibility Directive (89/336/EEC) and

Low Voltage Directive (73/23/EEC)

The product fulfills the essential requirements of the harmonized standards shown above.

2.   Product Safety Instructions

This product complies with and conforms to the following international Product Safety standards as applicable:

Safety of Information Technology Equipment, IEC 6095-1, including

Relevant national deviations as listed in Compliance with IEC for Electrical Equipment (IECEE)

Safety of Information Technology Equipment, CAN/CSA-C22.2 No. 60950-1/UL 60950-1

3.   E-911 and use of the LIP-8002/8002A with Multi-Line Telephone System

Please note the use and operation of this phone as part of a multi-line telephone system (MLTS) may be subject to state and/or federal E-911 MLTS laws that require the MLTS to provide a caller's telephone number, extension, and physical location to applicable state and/or local emergency services when a caller initiates a 911 call. The MLTS may not provide a caller's telephone number, extension, or physical location information to emergency services when a caller dials 911, and compliance with state and/or federal E-911 MLTS laws is the sole responsibility of the purchaser of this phone.

4.   Privacy:

LIP-8002/8002A implements security and encryption technologies however, privacy of communications may not be ensured when using this telephone.

Care & Precautions:

Before connecting or using your new phone, take a moment to consider safety and reliability. Use common sense when locating, connecting and using your IP Phone LIP-8002/8002A .

- Locate on a dry level surface away. Keep the phone away from the edges of the surface to avoid the potential of a fall.

- Locate cables to avoid potential for damage. For example, do not locate under rugs or carpet as damage may result from foot traffic or heavy objects. Also, do not locate cables between the desktop and walls where they may be crushed, damaging the insulation.

- Check the cables regularly and, if damage is noted, disconnect your phone. Contact your local representative for a replacement.

- Should liquid spill on the phone, disconnect the unit from power and the network.

- Do not use during lightning storms. Lightning presents a potentially lethal shock hazard.

- Clean the LIP-8002/8002A with a soft dry cloth; do not use liquid cleaners.

- Always use caution when connecting to AC power. Use only with a properly grounded standard AC power outlet.

What is new in this release?

Issue 1.00   2010, 08   first release

Table of Contents

1. Introducing the LIP-8002/8002A

1.1 The IP Phone LIP-8002/8002A Overview

The LG-Ericsson IP phone (LIP-8002/8002A) is Internet Protocol (IP) phones designed to support hosted telephony services over a managed IP network. Based on the open standard Session Initiation Protocol (SIP), LIP-8002/8002A interoperates and exchanges signaling messages with SIP call servers, proxies and gateways to establish, maintain and terminate calls. Real-time Transport Protocol (RTP) packets transport audio between end-points through the service provider's network.

Features available to the IP Phone when used with SIP call servers are similar to those of a conventional business telephone. Features available for a given phone depend on the configuration of the call server as well as configuration of the IP phone and include:

> Anonymous call blocking
> Auto answer
> Call forward
>> Busy
>> No answer
>> Immediate
> Call hold
> Call hold ringback
> Call logs
>> Missed
>> Incoming
>> Outgoing
> Call Park
> Call Pick-up
>> Directed
>> Group
>> Park pick-up
> Call refusal
> Call transfer
> Call waiting
> Caller Id blocking
> Caller Id display
> Click-to-call
> Distinctive ring tones
> Do Not Disturb (DND)
> Hands-free dialing
> Hotline/Warm-Line (Direct Call)
> Last call return
> Last number redial
> LCD contrast control
> Message wait with indicator lamp and tone
> Mute
> One-button Call Transfer

Phone Book
Private line
Private Hold
Speakerphone, full duplex
Speed dial buttons
Speed number dialing
Station-to-Station dialing
Volume control (handset, speakerphone, and ring)
3-Way conference

For ease of use, the LIP-8002/8002A is equipped with fixed buttons, flexible (programmable) buttons, an LCD display, menu navigation buttons, and a dial pad. Figure 1.1-1illustrates these elements of the LIP-8002/8002A.
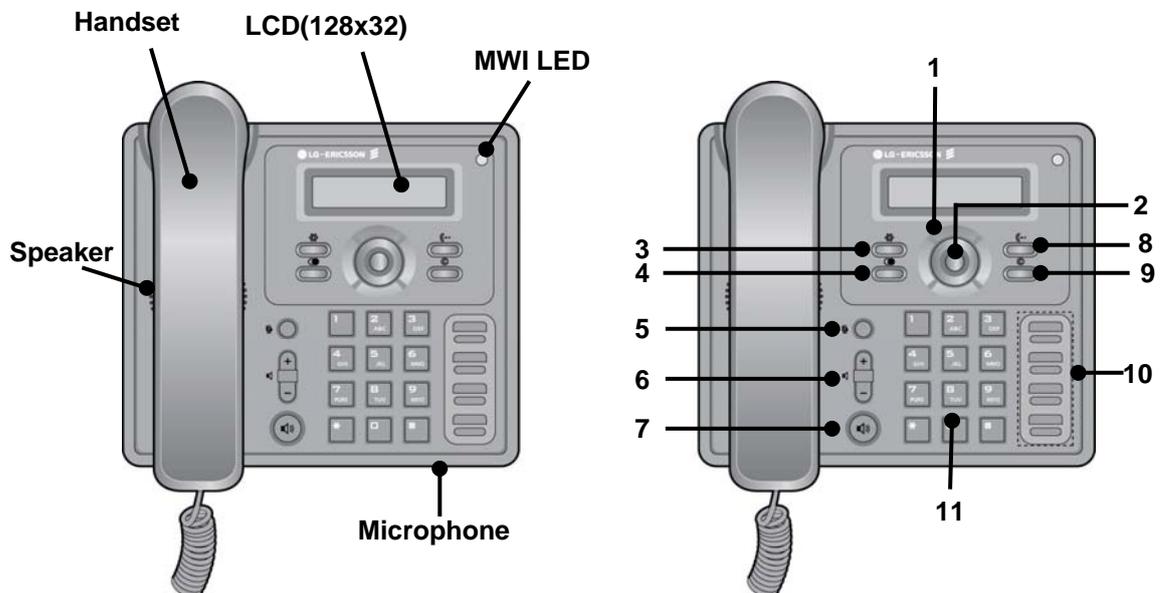
Figure 1.1-1 LIP-8002/8002A

Number	Function
1	**Left** button - Move to the previous menu in the settings menu.
	Right button
	Up button - When the phone is in the idle state, accesses the call log.
	Down button - When the phone is in the idle state, accesses the phonebook.
2	**OK** button - Select a current item or save a value in the settings menu.
	Retrieve a hold Call button (In Hold State) - Use to retrieve a hold call.
3	**Settings** button - 'Settings' accesses and exits the menu for display and changes to the IP Phone configuration.
4	**Redial** button - Use to dial the last number dialed.
5	**Mute** button - Toggle audio from the microphone to the connected party On and Off.
6	**Volume control** button - Use to adjust Ring, Handset, and Speaker volume.
7	**Speakerphone** button - Toggle the IP phone speakerphone On and Off.
8	**Mode** button (In Editing State) - Use to switch the input mode can accept an alphanumeric entry.
	Forward button (In Idle State) - Use to set the call forwarding configuration.
	Transfer button (In Call State) - Use to transfer the current active call.
9	**Delete** button (In Editing State) - Use to delete a letter in front of the cursor or clear an entire string.
	Conference button (In Call State) – Use to initiate, cancel or join the conference call.
10	**Flexible** buttons - Assign as feature in Phone configuration. Refer to section 3.6.5 under Flexible Button Setting for instructions on defining flexible buttons.
11	**Dial pad** buttons (In Editing State)- Use to dial a number, select a menu item, or input a value.

2. Installing the LIP-8002/8002A

2.1 Connecting the IP Phone

The figure below shows the cable connections for your IP Phone. The LAN and desktop PC connections employ standard Category 5 cables terminated with RJ-45 connectors. The LIP-8002 supports PoE (Power over Ethernet) in accordance with the IEEE-802.1af standard. (The LIP-8002A doesn't support PoE and can use only the supplied adaptor.) When connected to a PoE complaint LAN port, the IP Phone derives power from the port. If the LAN port does not support PoE, use the AC/DC adaptor, available separately, for connection to power. The handset connects to the base with the supplied standard handset coiled cord. Use the chart below to make connections to the IP Phone.
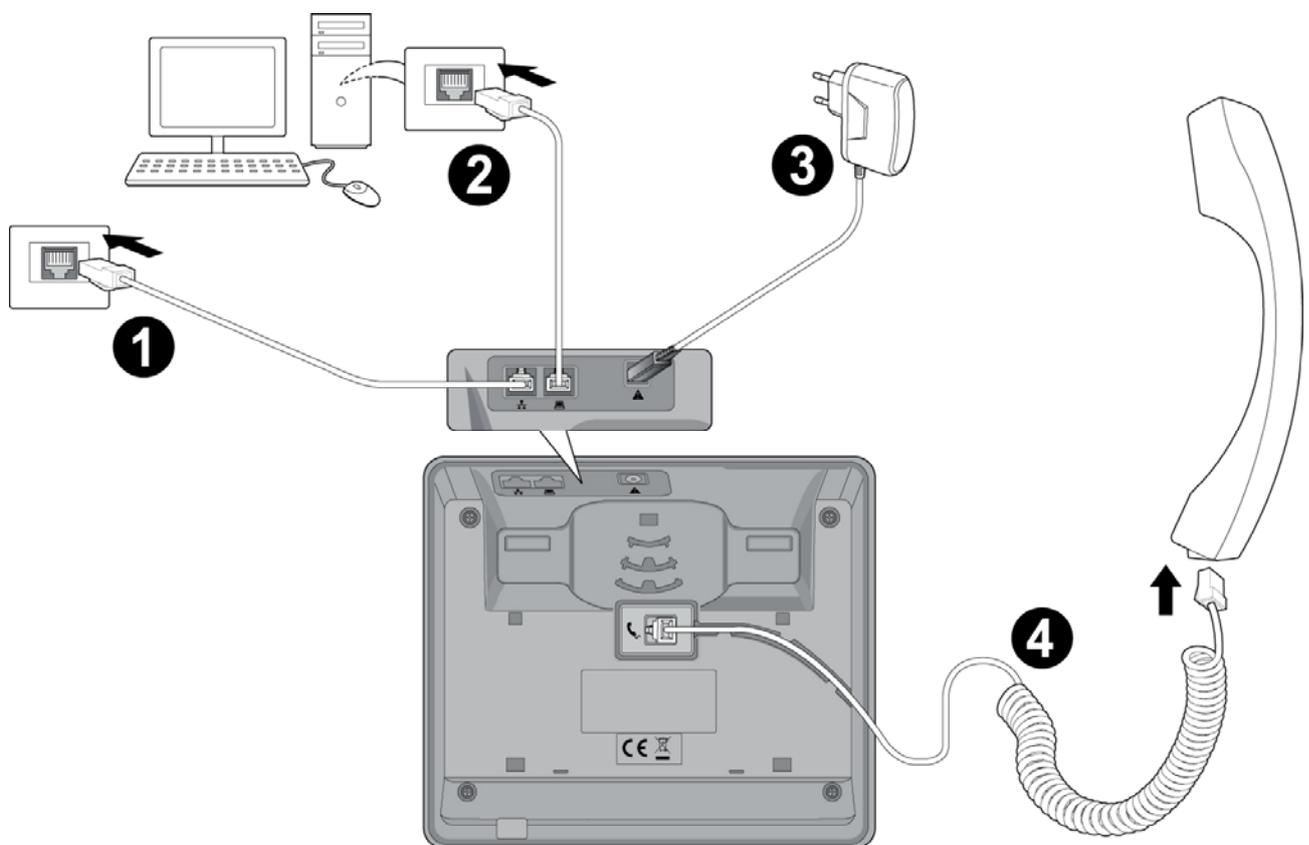
Figure 2.1-1 IP Phone Connections

Wiring Chart

1	LAN	Connect the IP Phone LAN port to the LAN wall jack with the provided LAN cable.
2	PC	Connect the IP Phone PC port to your desktop PC with an RJ-45 terminated UTP-5 cable.

3 Power The LIP-8002 supports PoE. So, if the LAN port supports PoE, IEEE 802.3af compliant, Class 2, the AC/DC adapter is not required. If not supported, connect the IP Phone power port to the DC out of Power Adapter. Connect the Power Adapter AC plug to an AC wall jack.

The LIP-8002A doesn't support PoE. So, must connect the IP Phone power port to the DC out of Power Adapter. Connect the Power Adapter AC plug to an AC wall jack.

4 Handset Connect the Handset coiled cord to the IP Phone base and handset.

RJ-45 Pin Assignments

LAN port	**PC** port
Pin 1 = TX+	Pin 1 = RX+
Pin 2 = TX-	Pin 2 = RX-
Pin 3 = RX+	Pin 3 = TX+
Pin 4 = optional: 48V (or GND)	Pin 4 = No connection
Pin 5 = optional: 48V (or GND)	Pin 5 = No connection
Pin 6 = RX-	Pin 6 = TX-
Pin 7 = optional: GND (or 48V)	Pin 7 = No connection
Pin 8 = optional: GND (or 48V)	Pin 8 = No connection
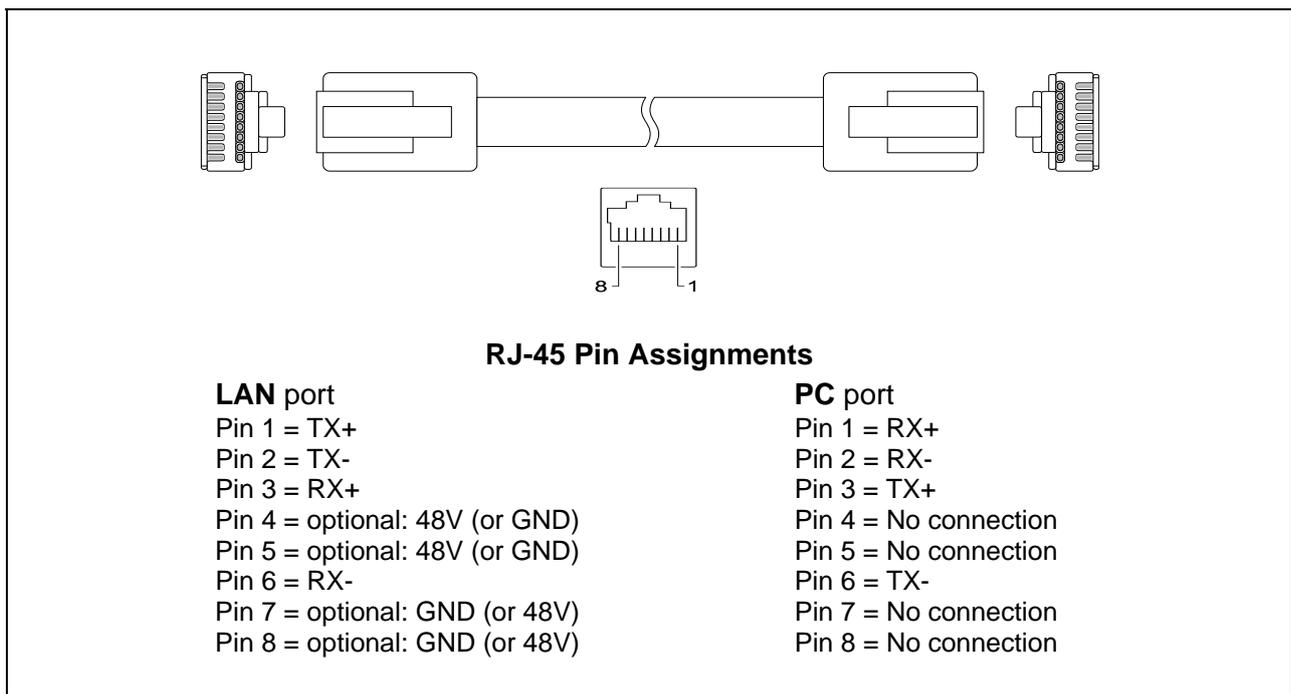
Figure 2.1-2 RJ-45 Terminations

2.2 Wall Mounting

The LIP-8002/8002A incorporates wall mounting in the base of the phone housing. Using the instructions below and Figure 2.2-1, wall mount the LIP-8002/8002A phone.

- Mark and drill two 7 mm holes for plastic wall anchors with a vertical separation of 10 cm.
- Insert two (2) anchors into the holes, then insert and tighten the two (2) screws leaving about 2.5 mm exposed. Refer to Figure 2.2-1.
- Attach all wiring to the IP Phone as described in section 2.1.
- Slide the IP Phone over the screws and assure the phone is secure. Note it may be necessary to remove the phone and tighten or loosen the screws for a secure mounting. Also note, the Foot Stand must not be attached to the IP Phone when wall mounting.

- Remove the Handset hook from the phone as shown in the figure below. Reverse the hook and re-install in the IP Phone so that the hook catches the groove in the handset receiver.
- Complete all wiring as described in section 2.1.
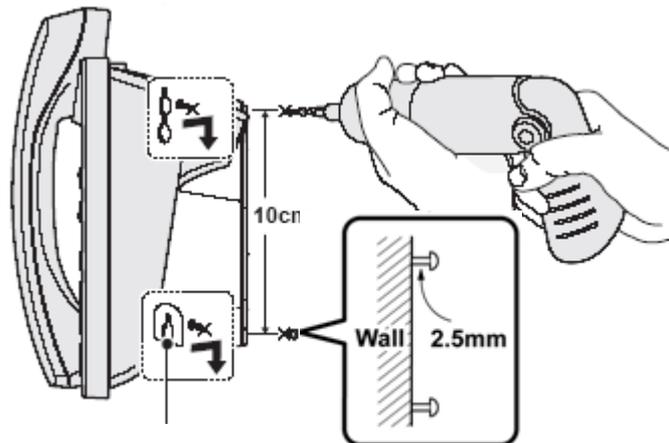
Figure 2.2-1 LIP-8002/8002A Wall Mount Installation

2.3 Provisioning the LIP-8002/8002A

LIP-8002/8002A supports 'plug and play' installation employing the automated provisioning process described in Appendix F. After connecting to the LAN, desktop PC and power, refer to section 2.1, the IP phone retrieves LAN settings including IP address, subnet mask and default gateway address from the local DHCP (Dynamic Host Control Protocol) server. In addition, the IP phone requests address information for the provisioning server where configuration and software files are maintained. The IP phone downloads the configuration and, if required, software files from the provisioning server. When complete, the IP phone automatically registers and subscribes for SIP call services using the information from the configuration files. Refer to Appendix D for more details.

LIP-8002/8002A incorporates means for manual configuration when the automated process is not fully available. As a minimum, the LIP-8002/8002A requires the information below for proper operation. When not assigned through the automated provisioning process, manual entry of the below parameters is required for proper operation.
- SIP parameters including:
  o SIP call server IP address
  o Name (SIP user id)
  o Authentication user name
  o Authentication password

In addition using static IP addressing requires manual entry of the following IP network configuration data.
  o IP address of the IP phone

> o Default gateway address
> o Subnet mask

Two methods are available for manual configuration. The LCD Configuration Menu permits local configuration using the dial pad, see section 3 The second method employs the LIP-8002/8002A Web Manager, which allows access to the IP phone configuration using a Web browser, see section 4. Manual configuration permits assignment of all parameters required to gain full operation of the IP phone. However, manual configuration is time consuming and is intended only to augment the automated process and to allow customizing user preferences.

3. Using the LCD Configuration Menu

This chapter provides detailed information to configure the LIP-8002/8002A using the LCD Configuration Menu. The LCD displays the configuration menu. The dial pad select menu items and input values. A configuration session begins by pressing the **Settings** (⚙) button, which accesses the LCD Configuration Menu shown in Figure 3-1. Select the desired menu item using one of the following methods:

- Use the dial pad to input the menu item number. This is the digit displayed in front of the item or
- Navigate with navigation buttons to highlight the item and select with the **OK** button.

```
Configuration Menu
1.Network Configuration
2.SIP Configuration
3.Phone Settings
4.Call Preferences
5.Directory
6.Set to Default
7.Lock/Unlock Config
8.Information
9.Reboot
```
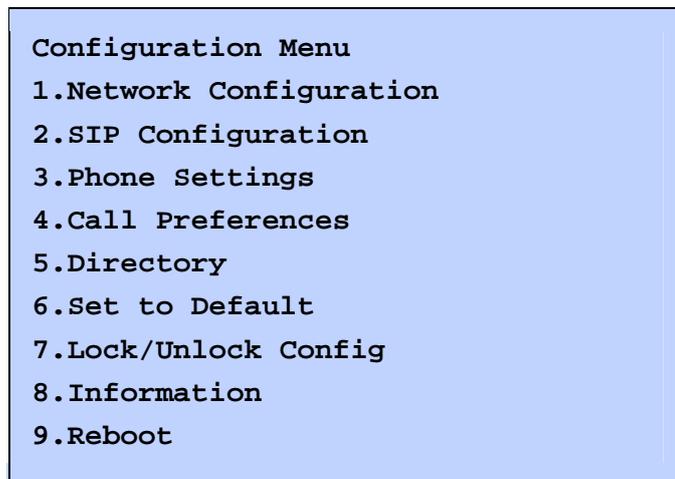
Figure 3-1 LCD Configuration Main Menu

When the desired parameter is reached, enter data with the dial pad or select values. Refer to Figure 3-2 for the complete menu.

Sections 3.1 to 3.3 give general information on use of the LCD Menu. The remaining paragraphs of this section give descriptive and procedural information for each parameter of the menu. Information is organized to follow the structure of the menu as shown in Figure 3-2.

Configuration Menu

- **1. Network Configuration**
  - 1. Network Mode
  - 2. IP Address
  - 3. Subnet Mask
  - 4. Default Gateway
  - 5. Primary DNS Address
  - 6. Secondary DNS Address
  - 7. MAC Address
  - 8. VLAN Settings
    - 1. VLAN Priority (LAN)
    - 2. VLAN ID (LAN)
    - 3. VLAN Priority (PC)
    - 4. VLAN ID (PC)

- **2. SIP Configuration**
  - 1. Line Configuration
    - 1. Call Server Address
    - 2. Proxy Port
    - 3. Display Name
    - 4. Name
    - 5. Auth. Username
    - 6. Auth. Password
    - 7. Reg. Status
  - 2. Outbound Proxy Addr.
  - 3. Outbound Proxy Port
  - 4. Backup Proxy Address
  - 5. Backup Proxy Port
  - 6. Local UDP Port
  - 7. RTP Start Port
  - 8. Proxy Registration
  - 9. Register Expires
  - 10. VMS Address
  - 11. Domain

- **3. Phone Settings**
  - 1. LCD Contrast
  - 2. Ring Setting
    - 1. Default Ring Type
    - 2. Trunk Ring Type
  - 3. Ringer Off
  - 4. Flexible Key Setting
  - 5. Time Configuration
    - 1. SNTP Server Address
    - 2. Time Zone
    - 3. DST Auto Adjustment
    - 4. Time Format
    - 5. Date Format
  - 6. S/W Update
    - 1. Provisioning Type
    - 2. TFTP Server Address
    - 3. HTTPS Provisioning
      - 1. Provisioning URL
      - 2. Username
      - 3. Password
      - 4. DHCP Option Number
    - 4. S/W Download Flag
    - 5. Config Download Flag
    - 6. DHCP Option
  - 7. Lock Outgoing Call
  - 8. Speed Number
  - 9. Key Tone
  - 10. Language
    Appears when language xmls exist

- **4. Call Preferences**
  - 1. Do Not Disturb
  - 2. Call Waiting
  - 3. Call Waiting Tone
  - 4. Call Hold Ringback
  - 5. Msg Waiting Tone
  - 6. Auto Answer
  - 7. Caller ID Blocking
  - 8. Anonymous Call Block

- **5. Directory**
  - 1. Phone Book
  - 2. Call Log
    - 1. Missed Calls
    - 2. Received Calls
    - 3. Placed Calls

- **6. Set to Default**

- **7. Lock/Unlock Config**

- **8. Information**

- **9. Reboot**

Figure 3-2 LCD Configuration Menu
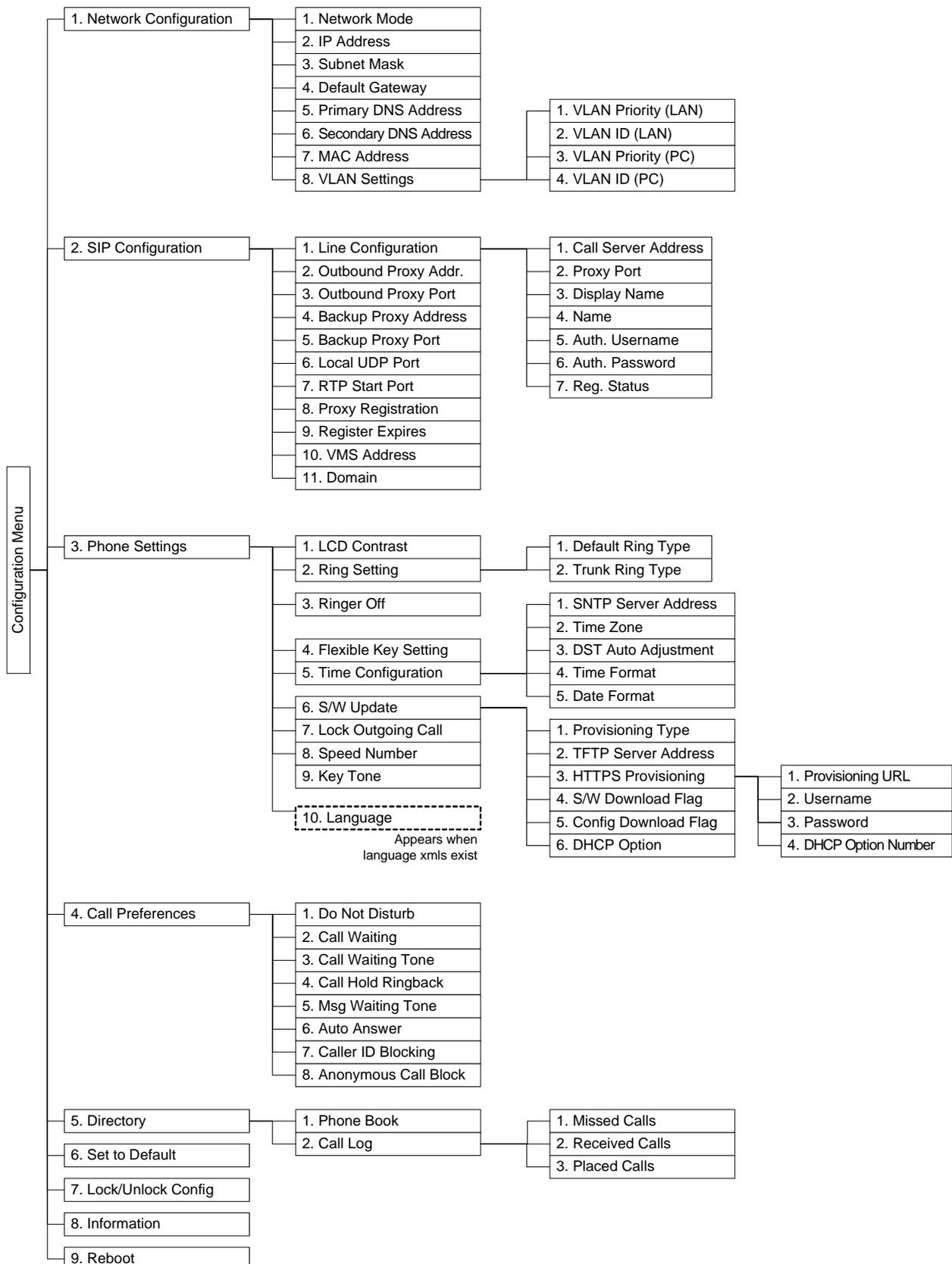
3.1 Navigating the LCD Configuration Menu

The IP phone must be idle to access the configuration menu and may be password protected (Locked). If locked, advanced configuration parameters can be viewed, however, data entry is not possible. Refer to section 3.3.

To access the configuration menu:
- Press the Settings (⚙) button.

To select an item:
- Dial the digit associated with the menu item or
- Use the **Up**/**Down** navigation buttons to move the cursor to the item, then press the **OK** button.

To return to a previous branch in the menu:
- Press the **Left** navigation button.

To exit the configuration menu:
- Press the Settings (⚙) button.

3.2 Entering Values with the Dial Pad and Softkeys

Use the dial pad to input numbers, letters, special characters, and a period. Selecting the **Mode** button switches the dial pad mode. The dial pad mode displays in the upper-right corner as 123 for numeric, ABC for upper case characters and abc for lower case characters. Appendix E. defines the relationship between the dial pad digits and characters.

To enter characters with the dial pad, use the digit with the desired character. Press the same digit repeatedly until the desired character appears in the LCD. After entering a character, pause to allow the cursor to move to next character position.

Other special entries include:
- To enter a period or dot (for example, to input an IP address), press asterisk, ✱, on the dial pad.

To delete a letter in front of the cursor, press the **Delete** button.
- To clear an entire string, press the **Delete** button for a while.

Prior to saving an entry, you may discard changes by pressing the **Left** navigation button to move to the previous menu, or press the Settings (⚙) button to exit the Settings menu.

3.3 Lock/Unlock Configuration

The LCD Configuration Menu can be protected for security using a four to eight digit password. With a password assigned, exiting the LCD Menu automatically locks the menu. Configuration

data displays while the menu is locked however, the menu must be unlocked using the password in order to change advanced parameters. Parameters under the 'Phone Settings' item listed below are not subject to the lock feature, allowing the user access without the need to enter a password.

- LCD Contrast
- Ring Type
- Ringer Off
- Flexible Button settings
- Time Configuration
- Speed Number Dialing
- Key Tone
- Language

With the menu locked, 🔒 (padlock icon) appears in the settings menu.

To assign a password:

- Press the Settings (⚙) button.
- Press the 7 digit, or move the cursor to **7.Lock/Unlock Config** using the navigation buttons and press the **OK** button. If a password has been previously established, the previous password must be entered and press the **OK** button.
- Press the 1 digit, or move the cursor to **1.Change Password** using the navigation buttons and press the **OK** button. **2. Remove Password** menu removes the assigned password.
- Enter your new password using the dial pad and press the **OK** button. A password is composed of four (4) to eight (8) digits.
- To confirm your password, input the new password again and press the **OK** button.to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session, which will lock the LCD Menu automatically if a password is assigned.

To unlock the LCD Menu:

- Press the Settings (⚙) button.
- Press the 7 digit, or move the cursor to **7.Lock/Unlock Config** using the navigation buttons and press the **OK** button.
- Enter the password (4 to 8 digits) and press the **OK** button.
- Press the 3 digit, or move the cursor to **3.Unlock Menu** using the navigation buttons and press the **OK** button, the Unlocked message appears for approximately 2 seconds and the LCD Menu is unlocked.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

ℹ️ With a password assigned, exiting the LCD Configuration Menu automatically locks the menu.
Entering an invalid password returns an error message followed by the "Enter Password" display.

3.4   Network Configuration

Under Network Configuration are the Network mode (DHCP or static), various IP addresses, DNS (Domain Name Service), and other LAN parameters. In addition, the IP phone MAC (Media Access Control) address can be viewed. When using the WiFi Module refer to section 5 for the Network Configuration sub-menus.

> Changing network parameters requires a reboot of the IP phone; when you exit the LCD Configuration Menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.1   Network Mode

The IP phone is capable of obtaining all of its IP configuration data from a DHCP server. When a DHCP server is not available, static addressing is used for the Network Mode. Selecting static addressing requires manual entry of other IP network parameters including:

- IP address of the phone
- Default gateway IP address
- Subnet mask
- DNS address

To set the Network Mode:

- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 1 digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the 1 digit, or move the cursor to **1.Network Mode** using the navigation buttons and press the **OK** button.
- Select the Network Mode using the **Up**/**Down** navigation buttons and press the **OK** button to save your selection.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

> An asterisk, ✱, is displayed in front of the current selection.
> Changing the Network Mode requires a reboot of the IP phone; when you exit the LCD Configuration Menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.2   IP Address

Selecting static addressing for the Network mode requires manual entry of an IP address for the IP phone. The IP phone requires a valid available IP address for proper operation.

To set the IP Address:

- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.

- Press the ☐ digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ☐ digit, or move the cursor to **2.IP Address** using the navigation buttons and press the **OK** button.
- Input the IP Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (☐) button to exit the configuration session.

> The IP Address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP phone retrieves an IP address from the DHCP server.
> Changing the IP Address requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.3 Subnet Mask

Selecting static addressing for the Network mode requires manual entry of a subnet mask for the IP phone. The IP phone requires a valid subnet for proper operation.

To enter the Subnet Mask:
- Press the Settings (☐) button. If the LCD Menu is locked, use the unlock procedure. Refer to <u>section 3.3</u>.
- Press the ☐ digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ☐ digit, or move the cursor to **3.Subnet Mask** using the navigation buttons and press the **OK** button.
- Input the Subnet Mask using the dial pad and press the **OK** button to save your configuration.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (☐) button to exit the configuration session.

> The Subnet Mask can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP phone retrieves a subnet mask from the DHCP server.
> Changing the Subnet Mask requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.4 Default Gateway

Selecting static addressing for the Network mode requires manual entry of a default gateway address for the IP phone. The IP phone requires a valid default gateway address for proper operation.

To assign the Default Gateway address:

- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 1 digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the 4 GHI digit, or move the cursor to **4.Default Gateway** using the navigation buttons and press the **OK** button.
- Input the Default Gateway IP address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> The Default Gateway address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP phone retrieves a default gateway address from the DHCP server.
> Changing the Default Gateway address requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.5 Primary DNS Address

Using an FQDN (Fully Qualified Domain Name) for any of the servers in the IP phone configuration requires entry of a primary DNS IP address. The IP phone contacts the primary DNS server to resolve the FQDN to an IP address.

To assign the Primary DNS Address:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 1 digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the 5 JKL digit, or move the cursor to **5.Primary DNS Address** using the navigation buttons and press the **OK** button.
- Input the Primary DNS Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> The Primary DNS address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP phone retrieves a Primary DNS address from the DHCP server.
> Changing the Primary DNS address requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.6   Secondary DNS Address

With an FQDN assigned as any of the servers in the IP phone configuration, the IP phone contacts the primary DNS server to resolve the FQDN to an IP address. Should the primary not respond, the IP phone contacts a secondary DNS for FQDN resolution. This is an optional but recommended setting.

To assign the Secondary DNS Address:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ☐¹ digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ☐₆ digit, or move the cursor to **6.Secondary DNS Address** using the navigation buttons and press the **OK** button.
- Input the Secondary DNS Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

> The Secondary DNS address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP phone retrieves a Secondary DNS address from the DHCP server.
> Changing the Secondary DNS Address requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.7   MAC Address

IEEE specifications require manufacturers of data equipment to store a unique 6-byte MAC or Ethernet address in the permanent memory of each network device.

To view the MAC Address:
- Press the Settings (⬚) button.
- Press the ☐¹ digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ☐₇ digit, or move the cursor to **7.MAC Address** using the navigation buttons and press the **OK** button.
- View the MAC Address of the IP phone.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

3.4.8   VLAN Settings

VLAN settings define the Ethernet frame priority and VLAN identification in accordance with IEEE 802.1p/Q standards. Separate VLAN tags are assigned for the IP Phone PC port and the LAN (voice) port. With a VLAN Id assigned, only frames with the assigned Id are accepted, and all SIP and voice frames sent by the LIP-8002/8002A include the VLAN Id assigned to the LAN port.

> For proper operation, other network elements (LAN switches and default gateway) must support and be configured with proper VLAN parameters.
> Changing the VLAN parameters requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

3.4.8.1   VLAN Priority (LAN)

The VLAN Priority (LAN) establishes the priority for Ethernet frames from the LIP-8002/8002A voice port including signaling and voice packets. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign VLAN Priority for the LAN (voice) port:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ① digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑧ digit, or move the cursor to **8.VLAN Settings** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.VLAN Priority (LAN)** using the navigation buttons and press the **OK** button.
- Input the VLAN Priority for the LAN port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

> Changing the VLAN Priority requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> The recommended value for VLAN Priority (LAN) port is 5.
> To disable VLANs, assign the VLAN ID as zero (0).

3.4.8.2   VLAN ID (LAN)

The VLAN ID (LAN) assigns the identification for Ethernet frames for the IP Phone voice port including signaling and voice payloads. Setting the VLAN ID to zero (0) disables the VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN ID for the LAN (voice) port:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ⌨ 1 digit, or move the cursor to **1.Network Configuration** using the navigation buttons and press the **OK** button.
- Press the ⌨ 8 digit, or move the cursor to **8.VLAN Settings** using the navigation buttons and press the **OK** button.
- Press the ⌨ 2 digit, or move the cursor to **2.VLAN ID (LAN)** using the navigation buttons and press the **OK** button.
- Input the VLAN ID for the LAN port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> Changing the VLAN ID requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To disable VLANs, assign the VLAN ID as zero (0).

3.4.8.3 VLAN Priority (PC)

The VLAN Priority (PC) establishes the priority for Ethernet frames for the IP Phone PC port. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN Priority for the PC port:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ⌨ 1 digit, or move the cursor to **1.Network Configuration** using the **Up**/**Down** navigation buttons and press the **OK** button.
- Press the ⌨ 8 digit, or move the cursor to **8.VLAN Settings** using the navigation buttons and press the **OK** button.
- Press the ⌨ 3 digit, or move the cursor to **3.VLAN Priority (PC)** using the navigation buttons and press the **OK** button.
- Input the VLAN Priority for the PC port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> Changing the VLAN priority for the PC port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To disable VLANs, assign the VLAN ID as zero (0).
> The recommended value for the VLAN Priority for the PC port is 3.

3.4.8.4   VLAN ID (PC)

The VLAN ID (PC) assigns the identification for Ethernet frames for the IP Phone PC port. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN ID (PC) port:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [1] digit, or move the cursor to **1.Network Configuration** using the   navigation buttons and press the **OK** button.
- Press the [8 TUV] digit, or move the cursor to **8.VLAN Settings** using the navigation buttons and press the **OK** button.
- Press the [4 GHI] button, or move the cursor to **4.VLAN ID (PC)** using the navigation buttons and press the **OK** button.
- Input the VLAN ID (PC) port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session, or press the Settings (⌨) button to exit the configuration session.

> 🛈   Changing the VLAN ID (PC) port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To disable VLANs, assign the VLAN ID as zero (0).

3.5   SIP Configuration

The **SIP Configuration** selection establishes parameters for single line appearance, SIP User Id, allowing the IP Phone to register and subscribe for the line with the appropriate call server. Additionally, general Voice over IP (VoIP) characteristics are assigned..

3.5.1   SIP Configuration, Line Parameters

Assign parameters in this section for SIP line appearance.

3.5.1.1   Call Server Address

The Call Server Address is the IP address or FQDN of the IP Phone host call server.

To assign the Call Server address:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [2 ABC] digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the [1] digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.

- Press the ☐¹ digit, or move the cursor to **1.Call Server Address** using the navigation buttons and press the **OK** button.
- Input the Call Server Address or FQDN using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

> ⓘ   Changing the Call Server Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.1.2   Proxy Port

The Proxy Port defines the TCP/UDP port number employed for SIP signaling transport. In common practice, port number '5060' is used.

To change the Proxy Port number:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ☐² digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the ☐¹ digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the ☐² digit, or move the cursor to **2.Proxy Port** using the navigation buttons and press the **OK** button.
- Input the Proxy Port number using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

> ⓘ   Changing the Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.1.3   Display Name

When assigned, the Display Name is used in SIP headers as the caller id name.

To enter the Display Name:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ☐² digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.

- Press the ⊡ digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the ⊡ digit, or move the cursor to **3.Display Name** using the navigation buttons and press the **OK** button.
- Input the Display Name using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⊡) button to exit the configuration session.

> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.1.4 Name

The Name is used in SIP headers as the SIP user id or account assigned in the call server. A name is required for proper operation and must match the user id or account assigned in the call server.

To enter the Name:
- Press the Settings (⊡) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ⊡ digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the ⊡ digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the ⊡ digit, or move the cursor to **4.Name** using the navigation buttons and press the **OK** button.
- Input the Name using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⊡) button to exit the configuration session.

> Changing the Name requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.1.5 Authentication User Name

Authentication of the line appearance with the SIP call server uses the Authentication User Name.

To assign the Authentication User Name:
- Press the Settings (⊡) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ⊡ digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.

- Press the ① digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Auth. Username** using the navigation buttons and press the **OK** button.
- Input the Authentication User Name using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

> **ⓘ** Changing the Authentication User Name requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
>
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.1.6 Authentication Password

Authentication of the line appearance with the SIP call server uses the Authentication Password in conjunction with the user name.

To assign the Authentication Password:

- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ② digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑥ button, or the move cursor to **6.Auth. Password** using the navigation buttons and press the **OK** button.
- Input the Authentication Password using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

> **ⓘ** Changing the Authentication Password requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
>
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.1.7   Registration Status

The status of the line registration with the SIP call server can be viewed. The status is displayed as OK, NOK (Not OK) or undefined. The latter displays when no call server address is assigned.

To view the Registration status for a line:
- Press the Settings (⚙) button.
- Press the 2ABC digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the 1 digit, or move the cursor to **1.Line Configuration** using the navigation buttons and press the **OK** button.
- Press the 7PQRS button, or move the cursor to **7.Reg. Status** using the navigation buttons, view the registration status.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

3.5.2   SIP Configuration, General Parameters

Parameters in this section assign the overall SIP operation.

3.5.2.1   Outbound Proxy Address

With the Outbound Proxy Address defined, the IP Phone sends all requests to the proxy instead of the SIP call server configured in section 3.5.1.1. The address may be in the form of an IP address or a FQDN.

To assign an Outbound Proxy Address
- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 2ABC digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the 2ABC digit, or move the cursor to **2.Outbound Proxy Addr.** using the navigation buttons and press the **OK** button.
- Input the Outbound Proxy Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

ℹ️   Changing the Outbound Proxy Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
    To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.2.2 Outbound Proxy Port

With the Outbound Proxy Server defined, the associated port must be defined as the Outbound Proxy Port. All SIP requests are sent to the assigned port instead of the proxy port configured in section 3.5.1.2.

To assign the Outbound Proxy Port:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 2 digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the 3 digit, or move the cursor to **3.Outbound Proxy Port** using the navigation buttons and press the **OK** button.
- Input the Outbound Proxy Port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> Changing the Outbound Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.3 Backup Proxy Address

When employing an outbound proxy, a backup proxy can be defined for use should the primary proxy fail to respond. The Backup Proxy Address is an IP address or a FQDN.

To assign a Backup Proxy Address:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 2 digit, or move the cursor to **2.SIP Configuration** using the   navigation buttons and press the **OK** button.
- Press the 4 digit, or move the cursor to **4.Backup Proxy Address** using the navigation buttons and press the **OK** button.
- Input the Backup Proxy Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> Changing the Backup Proxy Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.2.4 Backup Proxy Port

The Backup Proxy Port defines the port used when sending SIP messages to the backup proxy defined under section 3.5.2.3.

To assign the Backup Proxy Port:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ② digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Backup Proxy Port** using the navigation buttons and press the **OK** button.
- Input the Backup Proxy Port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ⓘ Changing the Backup Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.5 Local UDP Port

The Local UDP Port defines the port, normally 5060, the IP Phone uses to send and receive SIP signaling packets. In some instances, particularly when behind a firewall, the normal port may not be available. If port 5060 is not available, a different port can be defined.

To set the Local UDP Port:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ② digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.Local UDP Port** using the navigation buttons and press the **OK** button.
- Input the Local UDP Port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ⓘ Changing the Local UDP Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.6 RTP Start Port

When a SIP call is established, Real-Time Transport Protocol (RTP) packets transport media and digitized voice. The port used for RTP packets is usually 23000, but a different port may be defined.

To assign an RTP Start Port:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to <u>section 3.3</u>.
- Press the [2 ABC] digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the [7 PQRS] digit, or move the cursor to **7.RTP Start Port** using the navigation buttons and press the **OK** button.
- Input the RTP Start Port using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> **ⓘ** Changing the RTP Start Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.7   Proxy Registration

If Proxy Registration is set to Yes, the IP Phone will register with the appropriate call or proxy server upon initial power-up, during a reboot and at periodic intervals based on the timer defined in <u>section 3.5.2.8</u>.

To enable Proxy Registration:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to <u>section 3.3</u>.
- Press the [2 ABC] digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the [8 TUV] button, or move the cursor to **8.Proxy Registration** using the navigation buttons and press the **OK** button.
- To change, select Yes or No using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> **ⓘ** Changing the Proxy Registration requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.8   Register Expires

To assure a connection with the proxy or SIP call server, the IP Phone periodically re-registers. Unless defined by the host server during registration, the Register Expires timer determines the interval between registration attempts and is set from 0 to 6400 seconds, default 3600 seconds.

To modify the Register Expires timer:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to <u>section 3.3</u>.

- Press the ② digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑨ digit, or move the cursor to **9.Register Expires** using the navigation buttons and press the **OK** button.
- Input the registration expiration time using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (🔧) button to exit the configuration session.

> Changing the Register Expirers requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.5.2.9 VMS Address

The Voice Mail Server (VMS) address is the IP address or phone number of the user's voice mail system.

To assign the VMS Address:
- Press the Settings (🔧) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ② digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the digits ①⓪, or move the cursor to **10.VMS Address** using the navigation buttons and press the **OK** button to save your entry.
- Input the VMS Address using the dial pad and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (🔧) button to exit the configuration session.

> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.5.2.10 Domain

The Domain specifies the domain for SIP service when the provider desires to use a specific SIP domain for the IP Phone.

To assign the Domain
- Press the Settings (🔧) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ② digit, or move the cursor to **2.SIP Configuration** using the navigation buttons and press the **OK** button.
- Press the digits ①①, or move the cursor to **11.Domain** using the navigation buttons and press the **OK** button.
- Input the Domain name using the dial pad and press the **OK** button to save your entry.

- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> Changing the Domain requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
>
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6 Phone Settings

Under the **Phone Settings** menu item the user parameters listed below can be changed.

- LCD contrast
- Ring setting
- Ringer Off
- Flexible button settings
- Time Configuration
- S/W Update
- Outgoing Call Lock
- Speed Number Dialing
- Key Tone
- Language

With the exception of S/W Update and Outgoing Call Lock, these settings are not password protected. The user may access these parameters even if the LCD Configuration Menu is locked.

3.6.1 LCD Contrast

The contrast of the LCD is adjustable for best viewing angle.

To adjust the LCD contrast:
- Press the Settings (⌨) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.LCD Contrast** using the navigation buttons and press the **OK** button.
- Press the **Down** navigation button to decrease the contrast, or press the **Up** navigation button to increase the contrast.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

3.6.2 Ring Setting

The IP Phone uses one of 4 ring signals (Ring Types) or no ring ("Ringer Off") for incoming call notification. The Phone Book record may have a different ring type assigned including "Ringer Off" and "Default Ring". The "Default Ring Type" is used when a ring type is "Default Ring". By assigning different ring signals to closely located phones, Phone Book records, the user can identify the ringing phone or caller.

3.6.2.1 Default Ring Type

To select the Default Ring Type for the IP Phone:
- Press the Settings (⚙) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ② digit, or move the cursor to **2.Ring Setting** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.Default Ring Type** using the navigation buttons and press the **OK** button.
- Select the Ring Type using the **Up**/**Down** navigation buttons.
- Press the **OK** button to change to the selected Ring Type.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

ℹ The LIP-8002/8002A uses one of 4 ring types. The Phone Book entry may use a different type.

3.6.2.2 Trunk Ring Type

To select the Trunk ring type for the IP Phone:
- Press the Settings (⚙) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ② digit, or move the cursor to **2.Ring Setting** using the navigation buttons and press the **OK** button.
- Press the ② digit, or move the cursor to **2.Trunk Ring Type** using the navigation buttons and press the **OK** button.
- Select the Ring Type using the **Up**/**Down** navigation button. The Ringer Off selection turns the ringer off for incoming calls. The "Default Ring" uses the Default Ring Type for incoming calls, see section 3.6.6.1
- Press the **OK** button to change to the selected Ring Type.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

ℹ If "Default Ring" is assigned for a trunk ring, the "Default Ring Type" is used.

The Ring Type can be set to any of the 4 Ring types, "Default Ring" or "Ringer Off". When set as Ringer Off, the phone will not ring for incoming calls.
For Ring Type, an asterisk, ✱, displays in front of the current selection.

3.6.3 Ringer Off

The IP Phones can be programmed not to ring when an incoming call is received.

To set the Ringer Off parameter:
- Press the Settings ( ) button.
- Press the ³ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ³ digit, or move the cursor to **3.Ringer Off** using the navigation buttons and press the **OK** button.
- Select Yes or No using the **Up**/**Down** navigation button and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

3.6.4 Flexible Button Setting

The flexible buttons not assigned as line buttons can access various functions. indicates the available functions with a description of each.

Table 3.6.4-1 Flexible Button Functions

	Feature	Description
0	No Function	Unassigned
1	Speed Dial Function	Dials the assigned number
2	Phone Book Function	Views the personal phone book
3	Call Log Function	Views the call log list
4	Mute Function	Toggles the Mute feature on and off
6	Redial Function	Dials the last number dialed by the user.
7	Return Call Function	Dials the number of the last answered call
8	Set Call Forwarding	Sets the call forwarding configuration
9	Conference Function	1. Initiates a conference call 2. Cancels the conference call 3. Joins calls in a conference
10	Transfer Function	Initiates a call transfer
11	Hold Function	Places an active call on hold
12	Do not disturb	Toggles the DND on and off
13	Set Ringer Off	Toggles the ringer on and off
14	Message Function	Views the number of new and old messages stored in the Voice Mail server, and dials the Voice Mail server by pressing the **OK** button.
46	Recording Function	Records voice during call
48	Blind Transfer Function	Initiates a blind call transfer

To assign a function to a flexible button:
- Press the Settings ( ) button.

- Press the [3 DEF] digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the [4 GHI] digit, or move the cursor to **4.Flexible Button** using the navigation buttons and press the **OK** button.
- Select the desired flexible button.
- Select a function using **Up/Down** navigation button and press the **OK** button. Refer to the Table 3.6.5-1 Flexible Button Functions.
- If the selected function is Speed dial, enter the speed number and press the **OK** button.
- Each flexible button can have its own label, enter the label and press the **OK** button to save the entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6.5   Time Configuration

Within the **Time Configuration** menu, the IP Address or FQDN of the desired Simple Network Time Protocol (SNTP) server and the local time zone are defined. In addition, Daylight Savings Time (DST) is disabled.

The following SNTP server addresses or other private SNTP server addresses may be used for Time Configuration:
- clock.via.net
- ntp.nasa.gov
- tick.ucla.edu

3.6.5.1   SNTP Server Address

SNTP is a standard protocol that permits a network device to obtain the network's current time. For operation with SNTP, the IP Phones require an SNTP server IP address or FQDN.

To assign the SNTP Server Address:
- Press the Settings ( ) button.
- Press the [3 DEF] digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the [5 JKL] digit, or move the cursor to **5.Time Configuration** using the navigation buttons and press the **OK** button.
- Press the [1] digit, or move the cursor to **1.SNTP Server Address** using the navigation buttons and press the **OK** button.
- Input the SNTP Server Address using the dial pad and press the **OK** button to save the entry.

- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

🛈   To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6.5.2 Time Zone

For proper operation of the clock, the IP Phone requires the local time zone.

To enter the Time Zone:
- Press the Settings ( ) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Time Configuration** using the navigation buttons and press the **OK** button.
- Press the ② digit, or move the cursor to **2.Time Zone** using the navigation buttons and press the **OK** button.
- Select the Time Zone desired using the **Up**/**Down** navigation buttons and press the **OK** button to save the entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

🛈   An asterisk, ✶, displays in front of the current selection.

3.6.5.3 DST Auto Adjustment

The IP Phone automatically adjusts for DST when enabled.

To enable/disable automatic day light savings time:
- Press the Settings ( ) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Time Configuration** using the navigation buttons and press the **OK** button.
- Press the ③ digit, or move the cursor to **3.DST Auto Adjustment** using the navigation buttons and press the **OK** button.
- Select On or Off using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

3.6.5.4 Time Format

The format of the LCD time display can be set to display time in the 12-hour format or 24-hour format. In addition, the LCD time display can be disabled.

To set the display format for time:
- Press the Settings (⌨) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Time Configuration** using the navigation buttons and press the **OK** button.
- Press the ④ digit, or move the cursor to **4.Time Format** using the navigation buttons and press the **OK** button.
- Press the **Up**/**Down** navigation buttons to select the Time Format and press the **OK** button to set the format.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ℹ An asterisk, ✱, displays in front of the current selection.

3.6.5.5 Date Format

The format of the LCD date displays in the one of the formats below:
>   MM/DD (01/22)
>   DD/MM (22/01)
>   DD Month (22 Jan)
>   Month DD (Jan 22)
>   None

To set the date display format:
- Press the Settings (⌨) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Time Configuration** using the navigation buttons and press the **OK** button.
- Press the ⑤ digit, or move the cursor to **5.Date Format** using the navigation buttons and press the **OK** button.
- Press the **Up**/**Down** navigation buttons to select the display format and press the **OK** button to set the format.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ℹ An asterisk, ✱, displays in front of the current selection.

3.6.6   S/W Update

Within the **S/W Update** selection, the type of provisioning (TFTP/HTTPS), the Server Address, HTTPS Verification settings and DHCP Options can be viewed and changed.

3.6.6.1   Provisioning Type

The LIP-8002/8002A supports two provisioning protocols:
- Trivial File Transfer Protocol (TFTP) – a simple transfer mechanism using UDP but lacks security and TFTP ports are often blocked at firewalls.
- Hyper-Text Transfer Protocol/Secure (HTTP/S) – employs TCP and using SSL (Secure Sockets Layer) encryption is a more secure method. In addition, ports employed by HTTPS are commonly available.

To assign the provisioning protocol:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.Provisioning Type** using the navigation buttons and press the **OK** button.
- Select the Provisioning Type (TFTP or HTTPS) using the the **Up**/**Down** navigation buttons and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

🛈   Changing the Provisioning Type requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
An asterisk, ✱, displays in front of the current selection.

3.6.6.2   TFTP Server Address

With TFTP provisioning selected, the IP address or FQDN of the TFTP server must be defined. The TFTP Server Address may be retrieved automatically from the DHCP server or set manually.

To enter a TFTP Server Address:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.

- Press the [2 ABC] digit, or move the cursor to **2.TFTP Server Address** using the navigation buttons and press the **OK** button.
- Input the TFTP Server Address using the dial pad, and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

ⓘ   Changing the TFTP address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.6.6.3   HTTPS Provisioning

The HTTP/HTTPS server URL is a required parameter for HTTP/HTTPS provisioning. Normally, HTTP employs TCP port 80, and HTTPS employ port 443. In addition, if required by the HTTP/HTTPS server for authentication, a Username and Password are assigned.

ⓘ   When using HTTPS, the server must support SSL (Secure Sockets Layer) for proper operation.

3.6.6.3.1   Provisioning URL

The Provisioning URL is the URL of the HTTP/HTTPS server.

To assign the provisioning server URL:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [3 DEF] digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the [6 MNO] digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the [3 DEF] digit, or move the cursor to **3.HTTPS Provisioning** using the navigation buttons and press the **OK** button.
- Press the [1] digit, or move the cursor to **1.Provisioning URL** using the navigation buttons and press the **OK** button.
- Input the provisioning server URL using the dial pad and press the **OK** button to save the entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

ⓘ   Changing the provisioning server URL requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6.6.3.2 Username

In some cases, the HTTP/HTTPS server requires a Username (Id) and Password for authentication prior to allowing file access.

To set the Username:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the ③ digit, or move the cursor to **3.HTTPS Provisioning** using the navigation buttons and press the **OK** button.
- Press the ② digit, or move the cursor to **2.Username** using the navigation buttons and press the **OK** button.
- Input the Username using the dial pad, and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ⓘ Changing the Username requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6.6.3.3 Password

In some cases, the HTTP/HTTPS server requires a Username (Id) and Password for authentication prior to allowing file access.

To set the Password:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the ③ digit, or move the cursor to **3.HTTPS Provisioning** using the navigation buttons and press the **OK** button.
- Press the ③ digit, or move the cursor to **3.Password** using the navigation buttons and press the **OK** button.
- Input the Password using the dial pad, and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

Changing the Password requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

To enter alphanumeric characters select the **Mode** button. Refer to Appendix E for the character mapping of dial pad digits.

3.6.6.3.4 HTTPS URL DHCP Option Number

An unused DHCP option can be used to define the HTTPS provisioning server URL. As a default, the IP Phone uses DHCP option 160.

To modify the DHCP option used for the HTTPS URL:
- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 3 digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the 6 digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the 3 digit, or move the cursor to **3.HTTPS Provisioning** using the navigation buttons and press the **OK** button.
- Press the 4 digit, or move the cursor to **4.DHCP Option Number** using the navigation buttons and press the **OK** button.
- Input the DHCP option, 1 to 255, using the dial pad, and press the **OK** button to save your entry.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⚙) button to exit the configuration session.

Changing the DHCP option requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.6.6.4 S/W Download Flag

When Power is applied or a reboot is processed, the IP-Phone automatically checks and downloads IP Phone Software from the defined provisioning server. The IP Phone will download software if the software versions are different and the S/W Download flag is enabled. Note this menu item displays only when the Download Flag function is enabled in the Web Manager, see section 4.7, or configuration file, see Appendix D.

To set the S/W Download Flag:
- Press the Settings (⚙) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 3 digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.

- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press ④ button, or move cursor to **4.S/W Download Flag** using the navigation buttons and press the **OK** button.
- Select Yes to enable the S/W Download, or select No to disable the S/W Download using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> Changing the S/W Download Flag requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.6.6.5 Config Download Flag

When Power is applied or a reboot is processed, the IP-Phone automatically downloads a Global (system) and a Phone specific (MAC) Configuration file from the defined provisioning server. To download the configuration files during provisioning, the Config Download flag must be enabled. Note this menu item displays only if the Download Flag function is enabled using the Web Manager, see section 4.7, or the configuration file, see Appendix D.

To set the Config Download Flag:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press ⑤ button, or move cursor to **5.Config Download Flag** using the navigation buttons and press the **OK** button.
- Select Yes to enable the Configuration Download, or select No to disable the Configuration Download using the **Up**/**Down** navigation buttons and press the **OK** button..
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> Changing the Config Download Flag requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

3.6.6.6 DHCP Option

The IP Phone supports DHCP Options 66 & 150 (TFTP server) and a custom Option, default 160, for the HTTP/HTTPS URL to define the provisioning server. In some situations, it is useful to disable use of the DHCP Options.

To set the DHCP Options flag:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.S/W Update** using the navigation buttons and press the **OK** button.
- Press the ⑥ digit, or move the cursor to **6.DHCP Option** using the navigation buttons and press the **OK** button.
- Select the DHCP Option desired using the **Up**/**Down** navigation buttons and press the **OK** button to save the entry. Options available include "Don't use any Option", "Option 66/150 only", "Option 160 only" and "Option 66/150, 160 both".
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

> ℹ️ Changing the DCHP Options requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
> An asterisk, ✱, displays in front of the current selection.

3.6.7 Lock Outgoing Call

The IP Phone can be configured to allow outgoing calls only if a password is entered. When enabled, the LCD screen will display the password request as shown and the password must be entered prior to connecting for outgoing call.

To activate this feature, the LCD password must be set, refer to section 3.3 and Lock Outgoing Call must be enabled.

To enable/disable the Lock Outgoing Call feature:

- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑦ digit, or move cursor to **7.Lock Outgoing Call** using the navigation buttons and press the **OK** button.
- Select Yes to enable the Lock Outgoing Call, or select No to disable the Lock Outgoing Call using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

3.6.8   Speed Number

LIP-8002/8002A includes a Phone Book that supports up to 100 entries. Each Phone Book record includes a two (2) digit Speed number, see section 3.8. When enabled, dialing the Speed Number (00~99) sends the contact information from the associated Phone Book record.

To enable/disable Speed Number operation:
- Press the Settings (⬚) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑧ digit, or move the cursor to **8.Speed Number** using the navigation buttons and press the **OK** button.
- Select Yes to enable the Speed Number, or select No to disable the Speed Number using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

When enabled, dialing an empty Speed Number terminates the call and the empty status of the Speed Number displays for approximately 2 seconds.

3.6.9   Key Tone

Normally, when a button on the phone is pressed a tone is generated to confirm to the user that the phone has recognized the key depression. If desired, this tone can be disabled.

To enable/disable the Key Tone:
- Press the Settings (⬚) button.
- Press the ③ digit, or move the cursor to **3.Phone Settings** using the navigation buttons and press the **OK** button.
- Press the ⑨ digits, or move the cursor to **9.Key Tone** using the navigation buttons and press the **OK** button.
- Select Yes to enable the Key Tone, or select No to disable the Key Tone using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

3.6.10 Language

The IP Phone firmware includes an xml file, which, among other things, determines the languages available to the phone. When the xml file contains more than one (1) language, the Language parameter will be available.

To select an available language:
- Press the Settings (⬚) button.

- Press the ③ digit, or move cursor to **3.Phone Settings** using the navigation buttons, and press the **OK** button.
- Press the ① ⓪ digits, or move cursor to **10.Language** using the navigation buttons, and press the **OK** button.
- Select a Language from the list using the **Up**/**Down** navigation buttons, and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

> An asterisk, ✱, displays in front of the current language selection.

3.7 Call Preferences

Call Preferences define various call handling features of the IP Phones.

3.7.1 Do Not Disturb

When the DND (Do Not Disturb) feature is active, the IP Phone does not notify the user of an incoming call. The IP Phone does not ring and the associated line LED does not flash. This allows you to work without interruption from an incoming call. The call server configuration determines routing of the incoming call while your phone is in DND. To activate DND, DND must be enabled.

To enable the DND operation:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ④ digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.Do Not Disturb** using the navigation buttons and press the **OK** button.
- Select On to enable DND, or select Off to disable DND using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

3.7.2 Call Waiting

With Call Waiting configured, while on a call the IP Phone alerts the user of an incoming call. The line button flashes to indicate the new call, and an optional beep tone is delivered over the existing conversation, see section 3.7.3. In addition, the LCD displays the incoming caller id during the ring signal so you know who is calling before you interrupt the current call.

If Call Waiting is disabled, the IP Phone will reject the new call. The call server configuration determines routing of the incoming call.

To enable/disable Call Waiting:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 4 digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the 2 digit, or move the cursor to **2.Call Waiting** using the navigation buttons and press the **OK** button.
- Select On to enable Call Waiting, or select Off to disable Call Waiting using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

3.7.3   Call Waiting Tone

While off-hook, the IP Phone can provide a tone over the existing conversation to indicate a call waiting. To receive the tone, Call Waiting must also be enabled. Refer to section 3.7.2.

To enable/disable Call Waiting Tone:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 4 digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the 3 digit, or move the cursor to **3.Call Waiting Tone** using the navigation buttons and press the **OK** button.
- Select On to enable the Call Waiting Tone, or select Off to disable the Call Waiting Tone using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session.

3.7.4   Call Hold Ringback

With Call Hold Ringback enabled, the IP Phone rings as a reminder of calls on hold when the user returns the IP Phone to idle.

To enable/disable Call Hold Ringback:
- Press the Settings (⌨) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the 4 digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.

- Press the [4] digit, or move the cursor to **4.Call Hold Ringback** using the navigation buttons and press the **OK** button.
- Select On to enable the Hold Ringback, or select Off to disable the Hold Ringback using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ([⚙]) button to exit the configuration session.

3.7.5   Message Waiting Tone

When notification of a new message is received, the Message LED flashes. As an option, a 'stutter' dial tone provides an audible indication for a message waiting. When the user goes off-hook to place a call, the user receives stutter dial tone.

To enable/disable stutter dial tone for a message wait indication:
- Press the Settings ([⚙]) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [4] digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the [5] digit, or move the cursor to **5.Msg Waiting Tone** using the navigation buttons and press the **OK** button.
- Select On to enable the stutter dial tone, or select Off to disable the stutter dial tone using the **Up**/**Down** navigation buttons and press the **OK** button..
- Move to another parameter with the navigation buttons and continue the session or press the Settings ([⚙]) button to exit the configuration session.

3.7.6   Auto Answer

When idle, the IP Phone can connect an incoming caller to the speakerphone automatically.

To enable/disable Auto Answer:
- Press the Settings ([⚙]) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [4] digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the [6] digit, or move the cursor to **6.Auto Answer** using the navigation buttons and press the **OK** button.
- Select On to enable Auto Answer, or select Off to disable Auto Answer using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ([⚙]) button to exit the configuration session.

3.7.7   Caller ID Blocking

The IP Phone normally sends the assigned display name as the SIP caller id for outgoing calls. Optionally, the IP Phone can send "anonymous" as the caller id in SIP message headers.

To enable/disable blocking of caller id:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [4 GHI] digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the [7 PQRS] digit, or move the cursor to **7.Caller ID Blocking** using the navigation buttons and press the **OK** button.
- Select On to enable blocking of outgoing caller id, or select Off to disable blocking of outgoing caller id using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

3.7.8   Anonymous Call Block

The LIP-8002/8002A can block incoming calls with an anonymous caller id.

To enable/disable Anonymous Call Blocking:
- Press the Settings ( ) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the [4 GHI] digit, or move the cursor to **4.Call Preferences** using the navigation buttons and press the **OK** button.
- Press the [8 TUV] digit, or move the cursor to **8.Anonymous Call Block** using the navigation buttons and press the **OK** button.
- Select On to enable Anonymous Call Blocking, or select Off to disable Anonymous Call Blocking using the **Up**/**Down** navigation buttons and press the **OK** button.
- Move to another parameter with the navigation buttons and continue the session or press the Settings ( ) button to exit the configuration session.

3.8   Phone Book

IP Phones include an internal Phone Book of names and numbers or SIP user ids for up to 100 contacts. The Phone Book menu is accessed through the LCD Menu, with the **Down** navigation button in idle stat or, when assigned, a Phone Book flexible button. The Phone Book menu displays the records.   Adding, editing or deleting the records is possible through the Web Manager, see section 4.8. The phone number or SIP user id in the Phone Book is the contact information used with Speed Number dialing, see section 3.6.10. Each contact can have a Ring type assigned. When a call is received from the contact, the specified ring signal is employed.

3.8.1   Dialing from the Phone Book

To dial a number from the Phone Book:
- Press the Settings (⬚) button.
- Press the ⑤ digit, or move the cursor to **5.Directory** using the navigation buttons and press the **OK** button.
- Press the ① digit, or move the cursor to **1.Phone Book** using the navigation buttons and press the **OK** button.
- Select a contact to dial using the **Up**/**Down** navigation buttons.
- Press the **OK** button to place the call.

3.9   Default

The IP Phone configuration can be returned to the original factory defaults, see Appendix A.

To return to the default configuration:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.
- Press the ⑥ button, or move the cursor to **6.Default** using the navigation buttons, and press the **OK** button.
- Press the **OK** button to set default.
- Move to another parameter with the navigation buttons and continue the session or press the Settings (⬚) button to exit the configuration session.

ℹ️   Returning the configuration to default requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
Returning the IP Phone to the default configuration deletes all previous customized settings.

3.10 Reboot

After changing certain parameters, the IP Phone will require a reboot to utilize the settings. These settings include the following:
- Network Configuration settings
- Call Server Address settings
- Software Update settings
- Default settings

To reboot the IP Phone:
- Press the Settings (⬚) button. If the LCD Menu is locked, use the unlock procedure. Refer to section 3.3.

- Press the ⑨ digit, or move the cursor to **9.Reboot** using the navigation buttons, and press the **OK** button.
- Press the **OK** button to reboot.
- If you press **Left** navigation button in the above step, move to another parameter with the navigation buttons and continue the session or press the Settings (⌨) button to exit the configuration session without activating the changes. Any parameter changes will update during the next reboot.

4. Using Web Manager for Configuration

The Web Manager is a web-based tool for configuring the IP Phone parameters. The Web Manager has access to parameters not available through the LCD Configuration Menu. Appendix A includes information on parameter availability. In addition to remote access to the IP Phone configuration, Web Manager is a more convenient tool when configuring a number of IP Phones or parameters of a single IP Phone.

Pointing your Web browser at the IP Phone IP address and port number 8000 will access Web Manager. Thus, an IP address in the IP Phone is necessary prior to attempting access to Web Manager. In addition, the IP address must be available to the Web browser host. The IP address of the LIP-8002/8002A is available through the LCD, see section 5.1.

Section 4.1 below describes the login procedure and site map structure. Sections 4.2 to 4.15 describe the parameters on each Web page in the order shown in the page link area of the site map.

4.1 Login & Site Map

Point your browser at the LIP-8002/8002A IP address and port 8000, <http://150.150.255.251:8000>, which returns the IP Phone Login page as below.

• Click LOGIN; the password page displays as below.

• Enter a valid U_ser name and Password to display the Web Manager Site map shown in Figure 4-1. The default User name is **private** and the Password is **lip**.

Figure 4.1-1 Site Map

On the left is the Web Manager page-link window and in the main screen is a more detailed site map.

• Select the desired page from either the page-link or the site map window.

4.2   VoIP Configuration

Line Settings

Proxy Address	
Proxy Port	5060
Display Name	
Name	
Authentication Name	
Authentication Password	
Delayed Ring	Not Use
Registration Status	OK

Common Settings

Outbound Proxy Address	
Outbound Proxy Port	5060
Backup Proxy Address	
Backup Proxy Port	5060
Domain	
VMS Address	
Proxy Registration	enable
Registration Timer (sec)	120
Local UDP Port	5060
Local TCP Port	5060
Local TLS Port	5061
TCP Port	5060
TLS Port	5061
Transport Mode	UDP 1. AES_CM_128_HMAC_SHA1_80 2. None
Start Rtp Port	23000
SRTP	disable 1. AES_CM_128_HMAC_SHA1_80 2. ARIA_CM_128_HMAC_SHA1_80 3. ARIA_CM_192_HMAC_SHA1_80
Codec Priority	1. PCMU 2. PCMA 3. G729 4. None 5. None
STUN Server Address	0.0.0.0

Change

Return to the Site Map

Figure 4.2-1 VoIP Configuration

Within the VoIP screen, adjustable parameters are:

- Line based parameters
  o <u>Proxy Server Address</u> is the IP address or FQDN of the SIP call server or SIP proxy.
  o <u>Proxy Port</u> is the server port for incoming requests and outgoing responses
  o <u>Display Name</u> is the name used as part of the IP Phone caller ID. The display name assigned for Line1 displays on the LCD display.
  o <u>Name</u> is the name used during registration. The name is generally the telephone number or the SIP user Id.
  o <u>Authentication Name</u> is the user id for authentication of the line appearance with the call server.
  o <u>Authentication Password</u> is the password for authentication of the line appearance with the call server.
  o <u>Delayed Ring</u> delays audible notification (ring) for a line. The delay is set in 1-second increments from 0 (no delay) to 10 seconds.
  o <u>Registration Status</u> shows the status of the line registration with the call server.

- General Parameters
  o <u>Outbound Proxy Address</u> is the IP address or FQDN of the outbound proxy server to receive SIP requests in place of the line-based Proxy Server.
  o <u>Outbound Proxy Port</u> is the port of the outbound proxy server
  o <u>Backup Proxy Address</u> is the IP address or FQDN of the backup proxy server, used when the connection with the primary proxy times-out.
  o <u>Backup Proxy Port</u> is the port of the backup proxy server
  o <u>Domain</u> is the domain for SIP service when the service provider desires to use a specific SIP domain.
  o <u>VMS Address</u> is the address (IP or phone number) of the Voice Mail Server.
  o <u>Proxy Registration</u> enables SIP registration to the proxy; this parameter is ignored if an Outbound Proxy Address is not defined.
  o <u>Registration Timer</u> is the interval, in seconds, between SIP registration renewal attempts.
  o <u>Local UDP Port</u> is the UDP port used on the LAN for sending and receiving SIP packets when the transport mode is UDP.
  o <u>Local TCP Port</u> is the local TCP port used on the LAN for receiving SIP packets when the transport mode is TCP.
  o <u>Local TLS Port</u> is the local TLS port used on the LAN for receiving SIP packets when the transport mode is TLS.
  o <u>TCP Port</u> is the TCP port used on the LAN for sending SIP packets when the transport mode is TCP.
  o <u>TLS Port</u> is the TLS port used on the LAN for sending SIP packets when the transport mode is TLS.
  o <u>Transport Mode</u> is the SIP signaling transport mode. If the TLS is selected for transport mode, TLS crypto suite is used.

o   Start RTP Port is the port for Real-time Transport Protocol (voice) packets.
o   SRTP can use the Secure RTP by using the SRTP crypto suite.
o   Codec priority selects the priority for audio codecs.

To modify a parameter:

- Select the parameter and modify the value.
- Select the Change button.

Changing the configurations except the display name, delayed ring and VMS address requires a reboot of the IP Phone. When you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.3 LAN Configuration

Figure 4.3-1 LAN Configuration

When the WiFi Module is installed, refer to <u>section 5</u> for LAN Configuration. When the WiFi module is not installed, in the LAN screen the adjustable parameters are:

- LAN Configuration
  - o <u>Network Selection</u> is the IP address mode for the IP Phone. The radial button selects static or DHCP address mode.
  - o <u>Desired IP Lease Time</u> is the IP Address Lease Time value included in DHCP packet made from IP Phone in DHCP mode.
  - o <u>IP address</u> is the IP Phone IP address.
  - o <u>Subnet Mask</u> is the IP Phone subnet mask.
  - o <u>Default Gateway</u> is the IP Phone default gateway IP address.
  - o <u>Mac Address</u> is the MAC address of the IP Phone, view only.
  - o <u>Primary DNS</u> is the IP address of DNS server.
  - o <u>Secondary DNS</u> is the IP address of an auxiliary DNS server.
  - o <u>Domain Name</u> is the domain name for the IP Phone.

To modify the Network Selection (IP address mode):
  - Select the radial button for Static or DHCP address assignment.
  - When complete, select the [Change] button to load new values to the IP Phone memory.

To modify any of the addressing information

- Select the appropriate box and enter the new value. Changes can only be made if static network mode is selected.

- When complete, select the ⟨ Change ⟩ button to load new values to the IP Phone memory.

The MAC address is fixed and cannot be modified.

The IP address, subnet mask, etc. can only be changed if the static Network Mode is assigned.

Changing any value in the LAN screen requires a reboot of the IP Phone. When you leave the menu, you will receive a Reboot notice. The IP Phone must reboot to utilize the new values entered.

4.4   Call Preferences

Figure 4.4-1 Call Preferences

The IP Phone supports various user preferences as listed below.

- Call Preferences
  - o <u>Caller ID Blocking</u> enables and disables blocking the outgoing caller id. When enabled, the IP Phone sends the anonymous id.
  - o <u>Anonymous Call Block</u>, when enabled, blocks incoming calls with an anonymous caller id.

- o <u>Call Waiting Function,</u> when enabled, visually notifies the user of an incoming call while in a conversation. If disable, when in a conversation, a new call is refused and the call server routes the call.
- o <u>Call Waiting Tone,</u> when the IP Phone is in use with call waiting enabled, a beep tone delivered over the existing call indicates an incoming call.
- o <u>Call Hold Ringback</u>, when enabled, rings the phone if the user returns to idle with a call on hold.
- o <u>Message Waiting Tone</u>, when enabled, stutter dial tone notifies the user a new message is waiting. Stutter dial tone notification is in addition to the visual flashing Message LED notification.
- o <u>Auto Answer</u>, when enabled, will activate the speakerphone automatically if an incoming call is received.

- Call Forwarding
  - o <u>Conditions</u> define the condition for Call Forward (Busy, No-Answer, unconditional, or not used).
  - o <u>Address to Forward</u> defines the destination to receive forwarded calls.

- Direct Call (Hotline/Warm-Line)
  - o <u>Direct Call</u>, when enabled, places a call to a predefined destination automatically when the phone goes off hook.
  - o <u>Direct Call Number</u> defines the destination for a direct call.
  - o <u>Direct Call Timer</u> sets a delay timer for Warm-Line operation. The timer specifies the period before placing a direct call. The timer is set in 1 second increments from zero (0) to ten (10) seconds. For Hotline operation, set the timer to zero (0) seconds.

To modify parameters on the Call Preferences screen:
- Select the desired parameter and enter new values, or select values from the drop-down menus.
- When complete, select the Change button to load new values to the IP Phone memory.

4.5   Dial Plan

The IP Phone incorporates a Dial Plan, which monitors the user-dialed digits for patterns matching the dial plan number. If a match occurs, the phone modifies the digits and processes the call using instructions in the dial plan.

4.5.1   Digit Map Table

Figure 4.5.1-1 Digit Map Tables Configuration

Using the Digit Map Table, a string of 512 characters may be entered to define the dial plan. Within the string individual numbers are entered, separated by a '|'. In addition to dial pad digits, special characters may be entered as follows:

- An 'x' matches any single digit, (0 to 9).
- A 'T' inserts a timed pause for additional digits.
- A '.' matches any number of occurrences of a digit, including zero occurrences and includes special character matches.
- A '+' matches any number of occurrences of the previous match, including special characters.
- The '[ ]' matches a digit from a group of digits. This character combination is used with "." to define a range, [2-6] matches any digit from 2 to 6 inclusive.

The dial plan can include modification of a matching dialed number before call set-up. The Digit Map Process defines the modification procedures for a number. The process may include:

- <u>TRNC</u> defines the number of digits to delete from the dialed number.

- Pre defines digits to insert in the front of the truncated number.
- Post defines the digits to insert at the end of the truncated number.

The following are examples of Dial Plans using the digit map tables.

Simple Dial Plan

Customer requirement:

Process the call after any 7 digit number is dialed or if the Operator is dialed at digit '0'.

Digit map routing entry

The digit map of this Dial Plan is (0T|xxxxxxx).

From this map, if the user dials '0' as the first digit, the IP Phone will wait for the pause time for additional digits. After the pause, if no additional digits are dialed, the phone places a call to the Operator. In addition, after the user dials any seven digits, the call is processed.

Complex Dial Plan

Customer requirement:

Process the call after the local operator at 0, long distance operator at 00, a four-digit local extension number starting with 3,4 or 5, a seven-digit local number prefixed by an 8, two-digit number preceded by star ('*'), a ten digit long distance number prefixed by 91, or an international number starting with 9011 is dialed.

Digit map routing entry

The dial plan for this is (0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T).

2nd Dial Tone

A second 512-character string map is employed to provide 2nd Dial Tone.

Customer requirements:

To provide second dial tone after a '9' is dialed, establish a Dial Plan as below.

In the Digit Map (2nd Dial Tone) text box, the dial plan for this is (9|).

To change Digit Map Table parameters:

- Assure the Digit Map radial button is selected under the Dial Plan Type
- Enter the desired values into the Digit Map (Routing) box.
- Select the [Change] button.

To set-up or change a Digit Map to provide second dial-tone

- Enter the digits in the Digit Map (2nd Dial Tone) box.
- Click the [Change] button.

To modify a Digit Map process:

- Enter the digits in the Digit Map (Process) boxes.
- Click the [Change] button.

To delete a Digit Map process:

- Select the DEL check box.
- Click the [Change] button.

4.6   Phone Information

Figure 4.6-1 Phone Information

The Phone Information screen displays the version information of the LIP-8002/8002A and the phone name. In addition, it is possible to change the Web Manager password through this screen.

To change the phone name:
 • Enter the desired name.
 • When complete, select the Change button to load new values to the IP Phone memory.

To change the password:
 • Enter the existing password.
 • Enter the new password and confirm.
 • When complete, select the Change button to load new values to the IP Phone memory.

4.7 Phone Settings

Figure 4.7-1 Phone Settings

In the Phone Settings screen, the following parameters may be modified.

- Phone Settings
  o <u>Frame Duration</u> sets the interval to send/receive RTP packets in milli-seconds.
  o <u>DTMF Relay</u> selects the method for DTMF signaling (in-band, out-of-band or negotiate).

- o <u>DTMF Payload</u> sets the RFC 2833 payload type for DTMF signals.
- o <u>DTMF Duration</u> sets the duration of DTMF signals sent from the phone in milliseconds.
- o <u>Lock Outgoing Call</u> locks the IP Phone, requiring password entry to place a call.
- o <u>Use Speed Number</u> enables or disables use 2-digit dialing to dial Phone Book contacts.
- o <u>Key Tone</u> enables or disables the tone generated when a button on the IP Phone is pressed.
- o <u>Line Lock Out</u> gives a beef notification after disconnect tone when the call is terminated.

- Functions
  - o <u>Download Flag</u> enables the display of the S/W and configuration download flags functions under the LCD Configuration Menu, see section 3.6.8.4 or section 3.6.8.5, respectively
  - o <u>One-touch Pickup</u> enables the call connection by selecting call on LCD using the navigation button when there are multi calls. Disabling One-touch Pickup causes the IP Phone to display the call status when a call is selected; the call does not connect to the speakerphone or handset unless the **OK** button is pressed a second time. This may be useful when the user cannot distinguish LED colors

- Ring Settings
  - o <u>Ringer Type</u> sets the default ring type, select one of 4 ring signals for audible notification of incoming calls,
  - o <u>Trunk Ringer Type</u> sets the ring type for incoming call from trunk line, select one of 4 ring signals, ringer off or default ring.
  - o <u>Ringer</u> turns the ring signal On or Off.

To modify Phone Setting parameters:
- Select the parameter.
- Enter the new value or select a value from the drop-down menu.
- When complete select Change button.

Changing DTMF Duration or Frame Duration requires a reboot of the IP Phone. When you leave the menu, you will receive a Reboot notice. The IP Phone must reboot to utilize the new values entered.

4.8   Phone Book

Figure 4.8-1 Phone Book

The IP Phone includes a Phone Book with 100 contact records. Calls to a contact are made using the Speed number (00-99). The contact number or SIP user id is maintained in the Phone Book. In the Phone Book menu, records may be added, edited, deleted or searched.

To add a Phone Book entry:
- Click the Add button and enter the new values in the Phone Book Entry screen. The Address field is the contact's phone number or SIP user id.
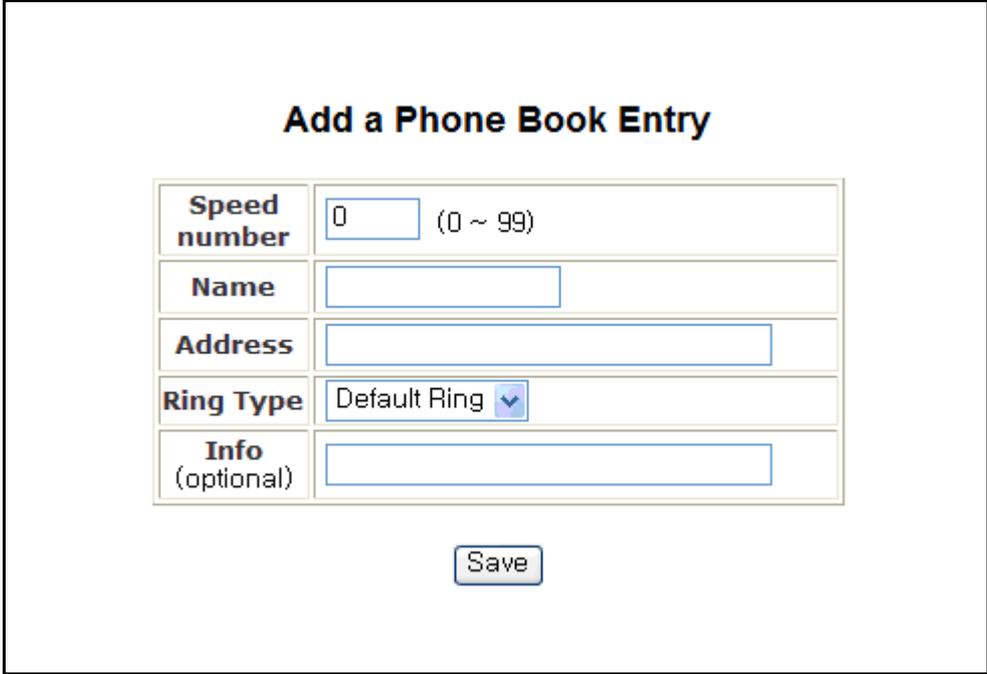
Add a Phone Book Entry

Speed number	0	(0 ~ 99)
Name		
Address		
Ring Type	Default Ring	
Info (optional)		

Save

Figure 4.8-2 Add a Phone Book Entry

- When complete, select the Save button to store the record.

To modify an entry:
- Select Modify for the desired record and enter the new values in the Phone Book Entry screen.
- When complete, select Save to store the modified record.

To place a call using the Phone Book:
- Click the Dial button for the desired contact.

To delete specific entries from the Phone Book:
- Click the 'Clear' box for the desired records and select the Clear button to delete checked records.

To delete all entries from Phone Book:
- Select the ClearAll button.

The Phone Book can be downloaded or uploaded to and from a *.csv formatted file. To Upload/download:
- Click Browse... and locate the desired csv file in your PC.
- Select upload or download, as desired.

An example Phone Book and resulting file are shown below.
Phone Book

Speed Number	Name	Phone Number	Information
1	Lee Myunghee	1000	lmh@abc.com
2	baragirl	2000	
3	lmh	3000	korea

sample_*.csv file
SpeedNumber,Name,PhoneNumber,Information
1,Lee Myunghee,1000,lmh@abc.com
2,baragirl,2000,
3,lmh,3000,korea

4.9   Flexible Button

Figure 4.9-1 Flexible Button

The LIP-8002/8002A has 4 flexible buttons. The buttons can be assigned to various functions. Refer to Table 3.6.5-1 for available functions.

To change button functions:
- Select the function from the drop-down menu next to button.
  - For a Speed dial button, enter the contact number or user id.
- Click the Change button to store changes.

To delete the function assigned to all flexible buttons:
- Select the ClearAll button.   Caution, selecting Clear All will delete all flexible button assignments.

4.10   Call Log

Figure 4.10-1 Call Log

The Call Log is a list of all calls (incoming, outgoing or missed) at the extension. The log indicates the type of call (I/C, O/G and MIS), the contacts name and contact information, time and duration of the call, and state of the call. From the Call log, the user can call a contact or store a call log record directly to the Phone Book.

To add a number from the Call Log to the Phone Book list:
  • Select the ⏍Add⏌ button.

To delete all Call logs:
  • Select the ⏍ ClearAll ⏌ button.
Caution, selecting the ⏍ ClearAll ⏌ button will delete all records in the Call Log.

To call a party from the Call Log:
  • Click the ⏍Dial⏌ button associated with the desired Call log record.

4.11 Network Time Configuration

Figure 4.11-1 Network Time Configuration

The Network Time screen allows you to set up the configuration for time and daylight savings time. Within Network Time Configuration, the following parameters can be set.

- Network Time Configuration
  o <u>SNTP Server Address</u> is the IP address or FQDN of the SNTP (Simple Network Time Protocol) server.
  o <u>Time Zone</u> defines the local time zone of the IP Phone.
  o <u>DST Usage</u> enables or disables Daylight Savings Time.

- **Time Format** sets the format of the LCD time display, off, 12- or 24-hour.

- **Date Format** sets the format of the LCD date display using radial button selection. In addition, the format of the date display can be customized using the "Format" text box by entering the desired format with the characters shown below the text box.

- **DST Configuration**
  The DST configuration sets the format for daylight savings time. The start and end for DST can be configured in the North American standard, Month/Week/Day/Time format.

Known SNTP servers that can be used:
- ntp.nasa.gov
- clock.via.net
- tick.ucla.edu

To modify parameters on the Network Time screen, select the desired parameter and enter new values or select values from the drop-down menus. When complete, select <u>Change</u> to load new values in the IP Phone memory.

4.12 Upgrade Configuration

Figure 4.12-1 Upgrade Configuration

In the Upgrade screen, the results of the most recent download attempt are displayed at the top of the screen. Upgrade result messages are defined in Appendix C. A log displays any error conditions for the configuration file download. In addition, the following parameters can be modified:

- Upgrade Configuration
  - o TFTP/HTTPS sets the protocol of the server where configuration and software files are stored.

o <u>TFTP Server Address</u> assigns the IP address or FQDN of the TFTP server where configuration and software files are located. With the "Received by DHCP on boot" box checked, the IP Phone only uses the TFTP server retrieved with DHCP option 66 or 150.

o <u>HTTPS URL</u> is the URL of the HTTP/HTTPS server where configuration and software files are located. When the "Received by DHCP on boot" box is checked, the IP Phone uses the DHCP Option defined to retrieve the URL. If the option is present, the provisioning type is changed automatically to HTTP/HTTPS using the URL from the DHCP option, default 160.

o <u>ID</u> assigns the user name for HTTPS authentication.

o <u>Password</u> sets the password for HTTPS authentication

o <u>MAC Cfg. Format</u> defines the format of the MAC (Phone specific) configuration filename; "$MAC" is replaced with the MAC address of the IP Phone and must be included as part of the file name or path to the configuration file on the HTTP/HTTPS server.

o <u>File to Upgrade</u> defines the files (software and configuration) to download during a reboot.

o <u>Smart Upgrade</u> enables periodic checks for new software on the provisioning server. Two methods of smart upgrade are available; one checks the server at the interval in hours, the second checks daily at the time specified in a 24-hour clock format. Check the "Use Smart Upgrade" box, select the method with the radial button and enter the interval or time as appropriate.

If the IP Phone finds a new version of software during the smart upgrade, the IP Phone either downloads the file or displays the availability of the new version and awaits user confirmation to download. To await user confirmation, check the "when new found, confirm downloading" box.

To modify the upgrade configuration:
- Select the parameter and enter the new value or select the check box.
- When finished, select the [Change] button.

ⓘ If the "Received by DHCP on boot" box is checked, then an address entered into the TFTP server address or HTTPS URL is ignored and only the server information retrieved from the DHCP server is used.

The radial button next to the provisioning server type (TFTP or HTTPS), must be checked. If no type is selected, file download is disabled.

Changes made to the Upgrade screen require a reboot of the IP Phone. When you leave the menu, you will receive a Reboot notice. The IP Phone must reboot to utilize the new values entered.

4.13 QoS Configuration

Figure 4.13-1 QoS Configuration

Diffserv Configuration

On the QoS screen, the DiffServ Code Point (DSCP) and VLAN parameters are defined. DSCP or ToS establishes the priority of voice packets. Compatible routers apply the DSCP for packet

priority, handling higher priority packets first. However, at congested routers, higher priority packets are discarded first. Thus, a median value for DSCP is recommended.

IEEE 802.1Q
The VLAN tag defines the priority for Ethernet frames and the VLAN ID based on IEEE 802.1p/Q. The IP Phone PC and LAN (voice) ports use separate VLAN tags. With a VLAN ID assigned, only frames with the Id are accepted and frames sent by the IP Phone include the LAN port Id and priority. In addition, the PC, LAN switch ports and the default gateway must be assigned to employ the VLAN tags for proper operation.

Jitter Buffer Configuration
A jitter buffer stores incoming packets to allow for re-sequencing out-of order packets. The Jitter Buffer configuration controls the size of the buffer used. The minimum, normal and maximum delays expected can be changed. Using larger values produce less packet loss as the IP Phone waits longer to receive packets before processing. Note that large values may delay voice packet processing and smaller values may produce more packet loss but quicker voice packet processing.

To modify the QoS configuration:
- Select the parameter and either enter a new value or select a value from the drop-down menu.
- When complete, select Change to load the new values in the IP Phone memory.

Modifications made to the IEEE 802.1p/Q configuration require a reboot of the IP Phone. When you leave the menu, you will receive a Reboot notice. The IP Phone must reboot to utilize the new values entered.

4.14 Load Default

> ## Click to load default settings
>
> After loading, system will restart to apply default values to system
>
> [Load]
>
> ---
>
> Return to the Site Map
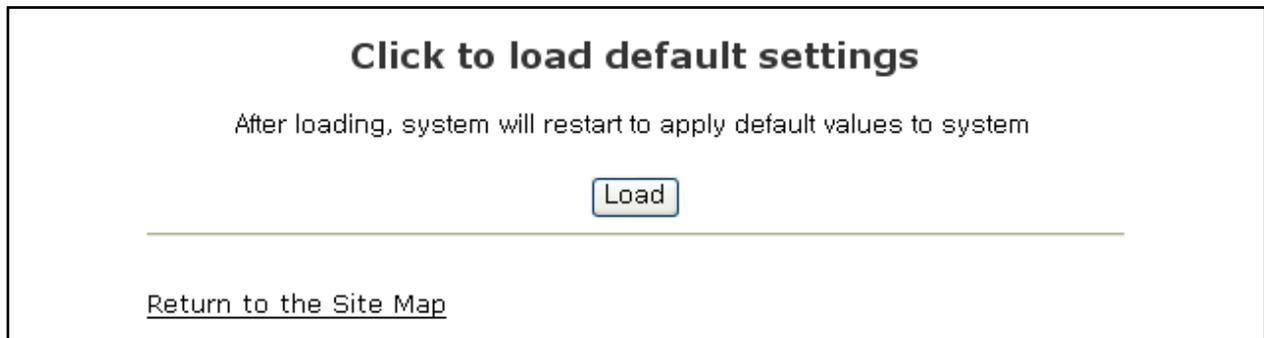
Figure 4.14-1 Load Default

The Load Default page permits the IP Phone configuration to be returned to factory default values. Refer to <u>Appendix A</u> for details.

To set the IP Phone memory to the default configuration:
- Select [Load] and confirm.
- The IP Phone loads the default configuration and reboots.

4.15 Reboot

Figure 4.15-1 Reboot

The Reboot screen is used to reboot the IP Phone manually.

To reboot the IP Phone:
- Select the [Reboot] button.
- The IP Phone saves and loads any changes in the configuration to active memory and reboots using the updated configuration.

5. TroubleShooting

5.1 Information

The LIP-8002/8002A provides the means to quickly view the Network parameters (IP address, default gateway address, and subnet-mask), server IP addresses (call server, TFTP server, and SNTP server), and results of the last file download attempt.

- Press the Settings ⚙ button.
- Press the 8ᴛᴜᴠ digit, or move cursor to **8.Information** using the navigation buttons, and press the **OK** button.

- Press the **Down** navigation button to view the next information.
  The download information displays results of the most recent download attempt, see Appendix C for a list of the messages.
- Press the ⚙ button to exit the menu.

5.2   Troubleshooting Guide

Table 5.2-1 Troubleshooting

Symptom	Corrective Action
LCD Off	1. Switch power source ON 2. Check/replace AC/DC adapter 3. Check/correct input power from the LAN
Handset off-hook, no dial tone	1. Assure the RJ-45 LAN cable is connected correctly. 2. Make sure that the call server IP Address is correct.
Phone displays "DHCP Searching" constantly, more than 2 minutes.	DHCP server is not responding. 1. Check DHCP server. 2. Check VLAN settings. To exit the DHCP Searching mode, press Settings button 3 times, the phone will complete the boot and return to idle.

Network setting error	1. Network Mode: DHCP
	  Make sure Network Mode is DHCP.
	  Make sure the RJ-45 LAN cable is connected securely to your IP Phone and LAN, switch, or router.
	  Make sure that all network ports, routers, or switches are working.
	  Make sure your network supports DHCP.
	2. Network Mode: Static
	  Make sure Network Mode is static.
	  If you entered the IP Address through a phone, make sure you reset your IP Phone.

When I attempt to make a call, I can't hear	1. Check all cables and connections. 2. Make sure the call server IP Address is correct.
When I attempted to reboot or change software, the IP Phone locks up.	Phone lock-up maybe caused by an error in transport during the software upgrade. To correct this problem: 1. Disconnect power from the IP Phone. 2. Press and hold the **Up** navigation button and the ⬚ digit while reapplying power. 3. Release the buttons when the Message LED extinguishes. 4. When the BootC screen appears, enter the digits 1, 2, 3 and 4. 5. Follow the prompts to configure IP parameters for the IP Phone. 6. At the "Server?" prompt, enter the TFTP server address where the IP Phone can find the BootC firmware. If already correct, press the **OK** button. The IP Phone will reboot and return to normal operation.

Appendix A. Factory Default Configuration

Factory default configurations are settings for the configuration, which are set at the factory. Table A-1 below gives the default values for configurable parameters.

Table A-1 Default Configuration

Config File Section Name	Parameter/Field	Default Value
[LAN]	Network mode	DHCP
	IP address	192.168.1.1
	Subnet mask	255.255.255.0
	Default gateway	192.168.1.254
	Primary DNS address	0.0.0.0
	Secondary DNS address	0.0.0.0
	TFTP server address	Not assigned
	ToS (Type of Service)*	0
	PC port VLAN priority	3
	PC port VLAN id	0
	LAN port VLAN priority	5
	LAN port VLAN id	0
[VOIP]	Call server address line 1	0.0.0.0
	Proxy port line 1	5060
	Display name line 1	Not assigned
	Name line 1	Not assigned
	Auth. user name line 1	Not assigned
	Auth. password line 1	Not assigned
	Delayed ring for line 1*	0 seconds
	Outbound proxy address	Not assigned
	Outbound proxy port	5060
	Backup proxy address	Not assigned
	Backup proxy port	5060
	Domain	Not assigned
	Start RTP port	23000
	VMS address	Not assigned
	prack mode**	disabled
	User name	Not assigned
	Replaces header**	enabled
	Local UDP port*	5060
	Local TCP port *	5060
	Local TLS port*	5061
	TCP port*	5060
	TLS port*	5061
	Transport Mode*	UDP
	TLS Crypto Suite*	AES_CM_128_HMAC_SHA1_80/ ARIA_CM_128_HMAC_SHA1_80

Config File Section Name	Parameter/Field	Default Value
	SRTP Crypto Suite*	AES_CM_128_HMAC_SHA1_80/ ARIA_CM_128_HMAC_SHA1_80/ ARIA_CM_192_HMAC_SHA1_80
	SRTP*	Disable
	Register expires*	3600 seconds
	SIP session timer**	0
	SIP T1 timer**	500 msec.
	SIP T2 timer**	4000 msec.
	Preferred codec*	G729/PCMU/PCMA
	De-register*	enabled
[DSP]	Handsfree gain	6
	Handset gain	6
	Ring volume	6
	LCD Contrast	4
	Ringer off	Ringer on
	VAD**	Off
	Frame duration*	20
	DTMF relay*	negotiate
	DTMF payload*	101
	DTMF duration**	120 milli-seconds
	default ring	Ring type 1
	Trunk ring	default ring
	Key tone	enabled
	Undefined cadence (0)**	2000, 4000, 0, 0, 0, 0
	External call cadence (1)**	800, 400, 800, 4000, 0, 0
	Group call cadence (2)**	400, 200, 400, 200, 800, 4000
	Internal call cadence (3)**	300, 200, 1000, 200, 300, 4000
	ACD call cadence (4)**	500, 0, 0, 0, 0, 0
	Priority call cadence (5)**	1500 4500 0 0 0 0 0 0
	Emergency call cadence (6)**	1500 500 1500 2500 0 0 0 0
	Community-1 call cadence (7)**	1500 500 500 3500 0 0
	Community-2 call cadence (8)**	1500 500 500 500 500 2500 0 0
	Community-3 call cadence (9)**	1500 500 500 500 1000 2500 0 0
	Community-4 call cadence (10)**	500 500 500 500 500 500 500 2500
	Auto answer call cadence (11)**	500 500 500 500 2500 3500 0 0
	Jitter buffer, (min, max, nom.)*	30, 60, 120 msec.
[FWRD]	Call forward	Not assigned
[DIAL]	Dial mode	Number
	Dial plan pause timer**	3 seconds
	Inter-digit timer**	6 seconds
[SYSTEM]	Language	English
	Lock outgoing call	unlock
	Speed number	enabled
	Web server**	enabled

Config File Section Name	Parameter/Field	Default Value
	telnet_server**	enabled
	Phone password	Not assigned
	Function download_flag*	enabled
	Function onetouch_pickup*	enabled
	Function dnd*	enabled
	Web Manger login name*	private
	Web Manager login password*	lip
	Line Lock Out	enabled
	Line Lock Out Time	30 sec
[NETTIME]	NTP server address	Not assigned
	Synchronization Interval**	24 hours
	Time zone	GMT+00:00
	DST enable	Off
	DST start configuration*	March, 2nd week, Sunday at 2 am
	DST end configuration*	November, 1st week, Sunday at 2am
	Time format	24-hour
	Date format	DD Mon (21 Feb)
[PROVISION]	Provisioning type	TFTP
	DHCP option 66/150	enabled
	HTTPS URL	Not assigned
	HTTPS URL DHCP option	160
	Use TFTP DHCP option*	enabled
	Use HTTP/S DHCP option*	enabled
	ID	Not assigned
	Password	Not assigned
	Phone specific file name*	$MAC
	Smart upgrade*	Upgrade at midnight
	S/W download flag	enabled
	Config download flag	enabled
	Dhcp lease duration	3600 sec
[PROG]	Flex buttons	Not assigned
[CALL]	Caller id block	disabled
	Anonymous caller id block	disabled
	Call waiting	enabled
	Call waiting tone	enabled
	Call hold ringback tone	enabled
	Message waiting tone	enabled
	Auto answer	disabled
	Forwarding condition*	off
	Forwarding address*	Not assigned
	Direct call*	disabled
	Direct call number*	Not assigned
	Direct call timer*	5 seconds

NOTE: Parameters marked with an asterisk, ✱, cannot be modified via the LCD Configuration Menu. Parameters mark with double asterisk, ✱✱, can only be modified using the configuration file download.

Appendix B. LIP-8002/8002A Specification

Table B-1 provides the basic specifications of the IP Phone.

Table B-1 LIP-8002/8002A Specifications

Item	LIP-8002/8002A
Network Interface	RJ-45 IEEE 802.3 Ethernet/Fast Ethernet switch port
PC/Workstation Interface	RJ-45 IEEE 802.3 Ethernet/Fast Ethernet switch port
Power	802.3af or AC/DC Power Adapter
LED Indicators	Message Indicator
LCD	128 x 32 dot graphic
Physical Size (H x W x D)	19.5cm x 19.5cm x 11.2cm
Weight	0.64 kg
AC/DC Adaptor	Output: 48VDC, 0.3A (max.) Input: 100-240 VAC, 50/60 Hz.
Operation Temperature	0 - 50° C
Storage Temperature	-20 - 60° C
Operation Humidity	0 - 85 % RH, non-condensing
Audio	Acoustic Echo Canceller & Voice Active Detection
Codecs	G.711, G.729a/b,

Appendix C. Download LCD Messages

Table C-1 provides the messages displayed for the most recent software download attempt.

Table C-1 Software Download Message

Message	Description
LOAD FLAG OFF	S/W load is disabled
NULL SERVER ADDR	TFTP/WEB server address is 0.0.0.0
VER FILE XFER ERR	Downloading a new version of S/W has failed
IMAGE XFER ERR	Check the TFTP/WEB server
VER FILE NOT FOUND	The version file doesn't exist in TFTP/WEB server
VER STRING ERR	The version file contains an erroneous format
IMAGE NOT FOUND	The S/W image doesn't exist in TFTP/WEB server
IMAGE SIZE ERR	The S/W image size is 0 or larger than maximum size
INVALID ID OR PW	The current Id or password for authentication is invalid
NEW VERSION LOADED	S/W was successfully loaded
UPDATE NOT NEEDED	The current version is the latest

Table C-2 provides messages displayed for the most recent configuration file download attempt.

Table C-2 Configuration Download Message

Message	Description
LOAD FLAG OFF	Config file load is disabled (section 4.12)
NO SERVER ADDR	TFTP/WEB server address is 0.0.0.0
CFG XFER ERR	Downloading the configuration file has failed; check the TFTP/WEB server status
FILE SIZE ERR	The configuration file size is 0 or there is an error during transfer from the TFTP server
FILE NOT FOUND	The configuration file doesn't exist in TFTP/WEB server
INVALID FILE OR DECRYPT KEY	The received configuration file is an invalid format or encrypted with an unknown key
PARAMETER ERR	The configuration file contains an erroneous value; the value has been automatically converted to previous value
INVALID ID OR PW	The current Id or password for authentication is invalid
DOWNLOAD OK	Configuration was successfully downloaded

Appendix D. Using the Configuration File

During automated provisioning with the Config. Download flag, section 4.12 enabled, IP Phones attempt to download a global and a phone specific configuration file from a TFTP or Web server supporting HTTP/HTTPS. The global file, "sysconf_8002_sip.cfg", is typically for parameters with the same value for all IP Phones connecting to the server. Parameters such as Network mode, Provisioning, Daylight savings time mode are typical globally settings.

The phone-specific configuration file as a default uses the MAC address of the phone for the file name, for example 00405a123477. When using HTTP/S, the IP Phone can be assigned with a name for the phone specific file in place of the MAC address but the MAC address must be included in the file name or path. Parameters specified in the phone specific configuration file override those defined in the global configuration file.

Both the global and MAC configuration files can configure any of the parameters of the IP Phone; the only difference is the MAC file values have priority over the global file. Each is an ASCII file with parameters (fields) grouped under the various sections of the configuration file, see Table D-1. By assigning values after the field name, the IP Phone configuration parameters are established.

A field listed in the configuration file with no value will return the value of the associated parameter to default. If a value is changed for a field marked with an asterisk in Table D-1, the reboot flag will be enabled and the IP Phone will reboot after initialization is complete. The first line of the configuration file (global or phone specific) must be "configuration information" as shown in Figure D-1.

The IP Phone supports SSL and allows assignment of a user name and password to authorize access to the HTTPS files.

In DHCP mode, the values from the DHCP server have a higher priority than the values in the sysconf_8002_sip.cfg and phone-specific configuration files.

Table D-1 below shows the configuration parameters by Section and Field name along with descriptive information. Table D-2 shows selections available for the time zones. Figure D-1 provides a blank configuration file. List a field in the configuration file only if the field requires an assignment other than default. If a field is erased or commented-out, it will have no affect on the IP Phone memory; the phone will use the existing value in memory. Deleting an unused field line from the configuration file also reduces traffic between the provisioning server and the IP Phone during a boot cycle.

Table D-1 Configuration Parameters

Section	Field Name	Description
LAN	network_mode	Sets use of DHCP or static addressing. In static mode, IP parameters are set manually. In DHCP, the IP Phone requests addresses from the local DHCP server including Options below: Option #1 :Subnet Mask Option #2: Time offset in seconds of the local time zone from GMT. Option #3 :Default gateway (router) IP address Option #6 :DNS server IP address Option #42 :SNTP server IP address Option #50 :IP address of the phone Option #66 :TFTP server name Option #150 :TFTP server IP address, has priority over Option #66 when set Option #160 :HTTPS URL for provisioning, has priority over Option #66 and #150 Valid values are static and dhcp. Default is dhcp.
	ipAddress	The phone's IP address. Valid value is an IP address. Default is "192.168.1.1".
	defaultGateway	The phone's default gateway. Valid value is an IP address. Default is "192.168.1.254".
	subnetMask	The phone's subnet-mask. Valid value is a subnet-mask. Default is "255.255.255.0".
	dns1_address	IP address of the primary DNS server used to resolve names to IP addresses. Valid value is an IP address. Default is blank.
	dns2_address	Optional IP address for a secondary DNS server; used if the primary DNS server is unavailable. Valid value is an IP address. Default is blank.
	domain_name	Not used
	tftp_server_address	Defines the IP address or URL of a TFTP provisioning server. The valid value is an IP address or FQDN. The default is blank.
	tos	Sets Type of Service (ToS) value for SIP packets. Valid values are 00 to FF (Hex). Default is 0.
	pc_802_priority	Establishes the IEEE 802.1 priority for Ethernet frames at the PC port of the IP Phone. Valid values are 0 to 7. Default is 3.
	phone_802_priority	Establishes the IEEE 802.1 priority for Ethernet frames at the voice port, including signaling and voice packets, of the IP Phone. Valid values are 0 to 7. Default is 5.

Section	Field Name	Description
	pc_vlan_id	Assigns unique identifier of the VLAN for the PC port. A '0' disables VLANS. Valid values are 0 to 4094 Default is 0, disable.
	phone_vlan_id	Assigns unique identifier of the VLAN for the LAN (voice) port. A '0' disables VLANS. Valid values are 0 to 4094 Default is 0, disable.
NETTIME	sntp_server_address	Sets the IP address or FQDN of the SNTP server from which the phone obtains time data. Valid value is an IP address or FQDN. Default is blank,
	interval	Sets interval, in hours, between time sync with SNTP server. Valid values are 0 to 120 hours. Default is 1.
	timezone	Sets the time zone for the phone. Valid values are time zones shown in Table D-2. Default is 23, London, England time.
	time_format	Sets the LCD display format for time. Valid values are 0 (off), 1 (12-hour) or 2 (24-hour). Default is 2.
	date_format	Sets the LCD display format for date. Valid values are (MM: 2-digit month), (M: month), (DD: 2-digit day), (D: day), (YYYY: 4-digit year), (YY: 2-digit year), (m: month abbreviation), (d: day of week), '/', ',', '-'. Default is DD m (21 Feb)".
	dst_auto_adjust	Enables automatic daylight savings time (DST) adjustment by the phone. Valid values are 0 (disabled) or 1 (enabled). Default is 0.
	dst_start_month	Sets month DST starts. Valid values are 1 to 12, with 1 being January and 12 being December. Default is 3 in the United States.
	dst_start_day_of_week	Sets day of the week DST begins. Valid values are 1 to 7 (1 is Sunday; 7 is Saturday). Default is 1 in the United States
	dst_start_week_of_month	Sets week of month DST begins. Valid values are 1 to 6 and 7. One (1) is the first week; each subsequent number is the subsequent week. Seven (7) is the last week of the month regardless of which week the last week is. Default is 2 for the United States.
	dst_start_time	Sets the hour of the day DST begins. Valid values are 0 to 23. The default is 2 for the United States.
	dst_stop_month	Sets month DST ends. Valid values are 1 to 12 with 1 being January and 12 being December. The default is 11 for the United States.
	dst_stop_day_of_week	Sets day of the week DST ends. Valid values are 1 to 7, with 1 being Sunday and 7 being Saturday. The default is 1 for the United States.

Section	Field Name	Description
	dst_stop_week_of_month	Sets week of month DST ends. Valid values are 1 to 6 and 7, with 1 being the first week, each number thereafter being subsequent weeks. Seven (7) represents the last week in the month The default is 1 for the United States.
	dst_stop_time	Sets the hour of the day DST ends. Valid values are 0 to 23. The default is 2 for the United States.
VOIP	outbound_proxy_server	When defined, sends all SIP requests to the outbound proxy server instead of to the line*n*_proxy_address. The media stream is not routed through the outbound proxy. The valid value is an IP address or FQDN. Default is blank
	outbound_proxy_port	Defines the port number of the outbound proxy server. Valid values are 1024 to 32000. Default is 5060.
	backup_proxy_server	Defines a backup proxy server in case the outbound proxy server fails. The valid value is an IP address or FQDN Default is blank
	back_proxy_port	Defines the port number of the backup proxy server. Valid values are 1024 to 32000. Default is 5060.
	message_url	Defines the voice mail address, IP address or phone number format. The valid value is an IP address or telephone number. Default is blank.
	SIP_service_domain	Specifies the SIP service domain. The valid value is an IP address or FQDN. The default is blank.
	preferred_codec	Defines the order of priority of codecs for negotiation. Valid values are g729, pcmu and pcma. The default order of priority is g729 pcmu pcma.
	rtp_port	Defines the RTP port for media packets. Valid values are from 16300 to 32700. The default is 23000.
	outbound_proxy_register	Enables use of proxy server for SIP requests. Valid values are enable or disable. The default is enable.
	prack_mode	Disables the use of ACK response to provide reliable response to SIP requests. Valid values are enable or disable. Default is disable.
	user_param	Defines "user=phone" parameter in the To, From, and Contact headers of REGISTER messages. Valid values are disable and enable. Default is disable.
	replaces_mode	Enables use of the SIP "Replace" header. Valid values are enable or disable. Default is enable.
	local_udp_port	Defines UDP port used for SIP messages. Valid values are from 1024 to 32000. Default is 5060.

Section	Field Name	Description
	local_tcp_port	Defines local TCP port used for SIP messages. Valid values are from 1024 to 32000. Default is 5060.
	local_tls_port	Defines TLS port used for SIP messages.. Valid values are from 1024 to 32000. Default is 5061.
	tcp_port	Defines TCP port used for SIP messages. Valid values are from 1024 to 32000. Default is 5060.
	tls_port	Defines TLS port used for SIP messages. Valid values are from 1024 to 32000. Default is 5061.
	transport_mode	Defines the transport mode. Valid values are UDP, TCP and TLS. Default is UDP.
	tls_crypto	Defines the order of tls cyprto suite. Valid values are AES_CM_128_HMAC_SHA1_80, AES_CM_128_HMAC_SHA1_32, AES_F8_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80. Default order is AES_CM_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80.
	srtp_crypto	Defines the order of srtp crypto suite. Valid values are AES_CM_128_HMAC_SHA1_80, AES_CM_128_HMAC_SHA1_32, AES_F8_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80, ARIA_CM_192_HMAC_SHA1_80. Default order is AES_CM_128_HMAC_SHA1_80, ARIA_CM_128_HMAC_SHA1_80, ARIA_CM_192_HMAC_SHA1_80.
	srtp_use	Enables use of the SRTP. Valid values are disable and enable. Default is disable.
	timer_register_expires	Defines time, in seconds, that a registration expires and re-registration is required. The Expire header in the registration message includes this value. A value of '0' is a de-registration. Valid values are 0 to 6400. Default is 3600.
	timer_session	Specifies the SIP session timer, in seconds. Valid values are 0 to 640. Default is 0.
	timer_t1	Sets the lowest value, in milliseconds, for the retransmission of SIP messages. Valid values are 100 to 640. Default is 250.
	timer_t2	Sets the highest value, in milliseconds, for the retransmission of SIP messages. Valid values are 641 to 6400. Default is 2000.

Section	Field Name	Description
	de_register	Enables de-registration of each line during a reboot prior to re-registering. Valid values are enable or disable Default is enable
	line1_proxy_address	Assgin the proxy address for the SIP proxy server. Valid value is an IP address or FQDN. Default is blank.
	line1_proxy_port	Assigns the port number for the SIP proxy server. Valid values are 1024 to 32000. The default is 5060.
	line1_displayname	Specifies the caller identification for the line, the phone sends the caller id for the line. Valid value is up to 50 characters. The default is blank.
	line1_name	Defines the SIP user id that is often a number or SIP address for use when registering. Enter a number without dashes, enter 555-0100 as 5550100, or a SIP address without the host name. Valid value is up to 50 characters. The default is blank.
	line1_authname	Specifies the name for authentication if the proxy server challenges the line registration. It is required only if a proxy server requires authentication. Valid value is up to 50 characters. The default is blank.
	line1_password	Specifies the password for authentication if the proxy server challenges the line registration. Valid value is up to 50 characters. The default is blank.
	line1_delayed_ring	Specifies a delay in the audible ring for the line. Delayed ring is adjustable from 0 to 10 seconds in 1-second increments Valid values are 0 to 10 seconds. The default is 0.
DSP	handsfree	Specifies the gain level for the phone's handsfree speaker. Valid values are 1 through 11. The default is 6.
	handset	Specifies the gain level for the phone's handset. Valid values are 1 through 11. The default is 6.
	ring	Specifies the gain level for the phone's ring. Valid values are 1 through 11. The default is 6.
	lcd	Specifies the contrast level for the phone's LCD. Valid values are 1 through 7. The default is 4.
	ringer_off	Toggles the ringer on and off. Valid values are 1 (on) and 0 (off). The default is 1 (on).
	VAD	Sets voice activation detection (VAD). Valid values are 1 (on) and 0 (off). The default is 0.

Section	Field Name	Description
	ring_type	Specifies the default ring type for the IP Phone. Ring type can be a value from 1 to 4. The default is 1.
	alert_ring_type external	Specifies the external ring type for the IP Phone. Ring type can be a value from -1 to 4: • -1, Default ring • 0, ring off. • 1 ~ 4, ring type 1 to 4 (tone ring), The default is -1.
	dtmf_payload	Sets the DTMF payload type for RFC2833. Valid values are 96 through 255. The default is 101.
	dtmf_duration	Sets the DTMF duration. Valid values are 50 through 300 ms. The default is 120.
	cadence0	Sets the ring cadence for calls with undefined in Alert Info header. The valid value is a six value time array (on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 2000-4000-0-0-0-0.
	cadence1	Sets the ring cadence for calls with External call in Alert Info header. The valid value is a six value time array (on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 800-400-800-4000-0-0.
	cadence2	Sets the ring cadence for calls with Group call in Alert Info header. The valid value is a six value time array (on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 400-200-400-200-800-4000.
	cadence3	Sets the ring cadence for calls with Internal call in Alert Info header. The valid value is a six value time array (on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 300-200-1000-200-300-4000
	cadence4	Sets the ring cadence for calls with ACD call in Alert Info header. The valid value is a six value time array (on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 500-0-0-0-0-0
	cadence5	Sets the ring cadence for calls with priority call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 1500-4500-0-0-0-0-0-0
	cadence6	Sets the ring cadence for calls with emergency call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 1500-500-1500-2500-0-0-0-0

Section	Field Name	Description
	cadence7	Sets the ring cadence for calls with community-1 call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 1500-500-500-3500-0-0-0-0
	cadence8	Sets the ring cadence for calls with community-2 call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 1500-500-500-500-500-2500-0-0
	cadence9	Sets the ring cadence for calls with community-3 call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 1500-500-500-500-1000-2000-0-0
	cadence10	Sets the ring cadence for calls with community-4 call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 500 500 500 500 500 500 500 2500
	cadence11	Sets the ring cadence for calls with auto answer call in Alert Info header. The valid value is an eight value time array (on off on off on off on off). The valid range is 0 to 30000 milliseconds total time. The default is 500 500 500 500 2500 3500 0 0
	frame_duration	Sets the RTP frame duration in milliseconds. Valid values are 10 to 100. The default is 20.
	jitterbuffer_min_delay	Sets the minimum jitter buffer size, in milliseconds. The Jitter buffer parameters control the buffer size for out-of-sequence packets. Using larger values produce less packet loss as the IP Phone will wait longer for receipt of packets. However, large values may delay voice packet processing. Smaller values may produce more packet loss but quicker voice packet processing. Valid values are 0 to 150. The default is 30.
	jitterbuffer_nom_delay	Sets the nominal jitter buffer size, in milliseconds. Valid values are 0 to 150. The default is 60.
	jitterbuffer_max_delay	Sets the maximum jitter buffer size, in milliseconds. Valid values are 0 to 150. The default is 120.

Section	Field Name	Description
	dtmf	Sets the method to determine RFC 2833 payload type. Valid values are: • inband — Always generate in-band DTMF. • nego — Negotiate RFC2833 with remote side. • rfc2833 — Always generate DTMF digits out-of-band RFC2833. • info — Always generate DTMF digits out-of-band in INFO message. Default is nego.
	key_tone	Controls the Key tone operation. If enabled, when a button on the IP Phone is pressed the key tone is heard. Valid values are enable and disable. The default is enable
FWRD	condition	Sets conditions for call forward. Valid values are: • -1 — Disables forwarding feature. • 0 — Condition not defined by configuration file, user can define condition. • 1 — Forward calls when the call is not answered in the no-answer. • 2 —Forward calls when the phone is busy. • 4 —Forward all calls. The default is 0 (not defined by configuration file).
	forwarding_address	Sets the address to receive forwarded calls. The valid value is a maximum of 60 characters. The default is blank.
DIAL	dial_mode	Sets the default dial mode. Valid values are number (digits), IP address or URL. The default is number.
	pause_timer	Specifies the pause timer, in seconds, for the dial plan. Valid values are 1 to 30. The default is 3.
	inter_digit_timer	Sets the time delay before dialed digits are sent for normal dialing, in seconds. Valid values are 1 to 30. The default is 6.
	digitmap	defines a digit map string for the dial plan, The valid value is a maximum 512-character string. Default is "x.T|x+#".
	digitmap_tone	Defines the digit map string to deliver second dial tone for a dial plan. The valid value is a maximum 512-character string. Default is blank.
	digitmap_emergency	Defines the emergency call number, see section 4.7. Valid value is a string up to 750 characters. The default is blank.

Section	Field Name		Description
	digitmap_process		Defines the digit map process for a dial plan. Valid values are: • num: the digitmap string, maximum 512-characters. • trunc: number of digits to strip, 0 to 30. • prefix: digits to insert in front of the number, maximum 32 digits. • post: digits to insert at the end of the number, maximum 32 digits. The default is blank.
PROG	add	prog	Specifies the flexible button number being assigned. Valid values are 1 to 4. The default is blank.
		func	Specifies the function for flexible buttons not assigned as a line. Valid values are: • 1 — Speed Dial • 2 — Phonebook • 3 — Call Log • 4 — Mute • 6 — Last number redial • 7 — Last call return • 8 — Forward • 9 — Conference • 10 — Transfer • 11 — Hold • 12 — DND • 13 — Ringer Off • 14 — Message • 46 — Voice Record • 48 — Blind Transfer The default is blank.
		num	Specifies the number for a Speed dial button. The default is blank.
	label	prog	Specifies the flexible button number being assigned. Valid values are 1 to 4. The default is blank.
		string	Specifies the label for flexible buttons. The maximum length is 15-characters. The default is blank.
CALL	callnum_in_line		Defines the number of active calls on a line. Valid value is 1~ 3. The default value is 3.
	caller_id_blk		Controls caller ID blocking. When enabled, the phone sends the anonymous caller id. Valid values are enable or disable. The default is disable.
	anonymous_call_blk		Controls blocking incoming calls with an anonymous caller id. Valid values are enable or disable. The default is disable.
	call_waiting		Controls visual notification of incoming call while busy. Valid values are enable or disable. Default is enable.

Section	Field Name	Description
	call_waiting_tone	Controls audible notification when Call Waiting is enabled Valid values are enable or disable. Default is enable.
	call_hold_ringback	Sets Held call ringback. If a user returns to idle with a call on hold, the held line will ring. Valid values are enable and disable. Default is enable.
	msg_waiting_tone	Controls stutter dial tone when there is a message waiting. Valid values are enable or disable. Default is enable.
	auto_answer	Controls answering an incoming call automatically activating the speakerphone. Valid values are enable or disable. Default is enable.
	call_direct	Controls Hotline/Warm-Line operation. Going off-hook automatically places a call to the designated party. Valid values are enable or disable. Default is disable.
	call_direct_number	Defines the destination for a direct call. Valid value is extension number. The default is blank.
	call_direct_timer	Sets the delay for Warm Line operation. After the timer expires, the direct call is made. Set to zero (0) for Hotline. Valid values are 0 to 10 seconds delay. Default is 5.
SYSTEM	phone_name	Assigns a name to the phone for idle mode display. Valid value is a string of up to 15 characters. The default is blank.
	lang	Specifies the language displayed in the LCD. Support for a given language is limited by the LCD type and the xml file. Valid values are as follows: • 0 — English • 1 — French • 2 — Korean • 3 — Danish Default is 0 (English).
	lock_outgoing_call	Unlocks the phone so a password is not required to place an outgoing call. Valid values are 0 to unlock or 1 to lock. Default is 0 (unlock).
	use_speed_number	Assigns use of 2-digit Speed number dialing. Valid values are 0 to disable or 1 to enable. Default is 1 (enable).
	web_server	Enables access to the Web Manager. Valid values are 0 to disable or 1 to enable. Default is 1 enable.
	telnet_server	Enables access to the telnet server. Valid values are 0 to disable or 1 to enable. Default is 1 enable.
	function_dnd	Disables dnd function for the phone. Valid values are disable or enable. Default is enable.

Section	Field Name	Description
	function onetouch_pickup	Enables call answer or resume from hold by pressing line button. With the parameter disabled, pressing the line button displays the line status. Valid values are enable or disable. The default is enable.
	function download_flag	Enables display of Config and S/W download flag under LCD Menu Phone Settings Valid values are disable or enable. Default is enable.
	llo_use	Enables the beep notification after disconnect tone when the call is terminated. Valid values are disable or enable. Default is enable.
	llo_time	Defines the duration to generate the beep for line lock out. Valid values are 0 to 999. 0 means infinite generation.. Default is 30 sec.
	phone_password	Assigns password for LCD Configuration Menu and Phone Unlock. Valid value is 4 to 8 digits string. Default is blank.
PROVISION	type	Defines the protocol of the provisioning server Valid values are 0 for TFTP or 1 for HTTPS Default is 0 (TFTP).
	cfg_downlaod	Enables download of configuration file from the provisioning server. Valid values are enable or disable. Default is enable.
	sw_downlaod	Enables download of firmware from the provisioning server. Valid values are enable or disable. Default is enable.
	dhcp_get_tftp	Enables use of DHCP options 66 or 150 for definition of the TFTP sever. Valid values are enable or disable. Default is enable.
	dhcp_get_http	Enables use of HTTP/S DHCP option, default 160, for definition of HTTP/S URL Valid values are enable or disable. Default is enable.
	https_url	Defines the HTTP/S provisioning server URL Valid value is a URL of less than 255 characters. Default is blank.
	https_opt	Defines the DHCP Option for the HTTP/HTTPS server URL. Valid values are 1 to 255. Default is 160.
	https_id	Specifies the ID for authenticating to an HTTPS session. The valid value is under 50 characters. The default is blank.
	https_password	Specifies the password for authenticating to an HTTPS session. The valid value is under 50 characters. The default is blank.

Section	Field Name	Description
	https_macform	Specifies the filename of phone specific configuration file. The name or path must include "$MAC", which is replaced by the MAC address during provisioning. The valid value is a maximum of 50 characters and includes "$MAC". The default is "$MAC".
	smartup_use	Enables Smart Update, to check for and download firmware periodically. Valid values are enable or disable. Default is enable.
	smartup_type	Defines the timing of Smart Upgrade as an interval or time of day. Valid values are 1 for interval or 2 for daily Default is 2.
	smartup_interval	Specifies the interval, in seconds, between firmware checks. Valid values are 0 to 86400 seconds. Default is 3600.
	smartup_time	Specifies the time of the day for daily firmware checks. Valid values are 0 to 23 o'clock. Default is 00:00.
	smartup_confirm	Enables user confirmation of firmware download from the provisioning server. Valid values are enable or disable. Default is enable.
	dhcp_lease_duration	Defines the IP address lease time for dhcp packet made from IP Phone. The valid value are 600 to 4294967295 seconds. The default is 3600.

```
;=========================================================================
;LIP-8002 configuration information(v1.0)
;copyright LG-Ericsson since 2010
;=========================================================================
;Description for configuration file
;configuration file has following format
; SECTION-NAME is placed in brackets as shown below and includes LAN, VOIP, DSP, etc...
;name is the parameter or field name as shown below
;value is the value assigned for the parameter.  If left blank, the value will be set to default.
;A blank space must be inserted as a separator between the assigned value and the start of any
comments, ';'.
;comments start with ';' and end with carriage return.
;
;=========================================================================

[LAN]
;Lan configuration
network_mode
ipAddress
defaultGateway
subnetMask
dns1_address
dns2_address
domain_name
tftp_server_address
tos
pc_802_priority
phone_802_priority
pc_vlan_id
phone_vlan_id

[NETTIME]
;SNTP, Timezone and DST Configurations
sntp_server_address
interval
timezone
time_format
date_format
dst_auto_adjust
dst_start_month
dst_start_day_of_week
dst_start_week_of_month
dst_start_time
dst_stop_month
dst_stop_day_of_week
dst_stop_week_of_month
dst_stop_time

[VOIP]
;VoIP configurations
outbound_proxy_server
outbound_proxy_port
backup_proxy_server
backup_proxy_port
message_url
SIP_service_domain
```

```
preferred_codec
rtp_port
outbound_proxy_register
outbound_proxy_mode
prack_mode
user_param
replaces_mode
local_udp_port
tcp_port
local_tcp_port
tls_port
local_tls_port
transport_mode
srtp_use
srtp_crypto
tls_crypto
timer_register_expires
timer_session
timer_t1
timer_t2
de_register
line1_proxy_address
line1_proxy_port
line1_displayname
line1_name
line1_authname
line1_password
line1_delayed_ring

[DSP]
;DSP configurations
handsfree
handset
ring
lcd
ringer_off
VAD
ring_type
alert_ring_type external
dtmf_payload
dtmf_duration
cadence0
cadence1
cadence2
cadence3
cadence4
cadence5
cadence6
cadence7
cadence8
cadence9
cadence10
cadence11
frame_duration
jitterbuffer_min_delay
jitterbuffer_nom_delay
```

```
jitterbuffer_max_delay
dtmf
key_tone


[FWRD]
;Forwarding Configurations
condition
forwarding_address


[DIAL]
;Dial Configurations
dial_mode
pause_timer
inter_digit_timer
digitmap
digitmap_tone
digitmap_emergency
digitmap_process


[PROG]
;add prog func [num]
;       prog = programmable button number, 1 ~ 4
;       func = 0 for no function
;               1 for Speed Dial
;               2 for Phone Book
;               3 for Call Log
;               4 for Mute
;               6 for Redial
;               7 for Return
;               8 for FWRD
;               9 for CONF
;               10 for TRNS
;               11 for Hold
;               12 for DND
;               13 Ringer Off
;               14 Message
;               46 Voice Record
;               48 Blind Transfer
;       num = phone number when a func is PhoneNum, max.50 length string.
;label prog name
;       prog = 1~ 4
;       name = phone label.


[CALL]
;Call Preference Configurations
call_num_in_line
caller_id_blk
anonymous_call_blk
call_waiting
call_waiting_tone
call_hold_ringback
msg_waiting_tone
auto_answer
call_direct
call_direct_number
call_direct_timer
```

```
[SYSTEM]
;System Configurations
phone_name
lang
lock_outgoing_call
use_speed_number
web_server_port
web_server
telnet_server
function download_flag
function onetouch_pickup
function dnd
llo_use
llo_time
phone_password

[PROVISION]
;Provisioning Configurations
type
cfg_download
sw_download
dhcp_get_tftp
dhcp_get_http
https_url
https_opt
https_id
https_password
https_macform
smartup_use
smartup_type
smartup_interval
smartup_time
smartup_confirm
dhcp_lease_duration
```

Appendix E. Dial Pad Mode

The LIP-8002/8002A dial pad inputs digits or alphanumeric characters for specified entries such as hostname. Table E-1 provides the input information for dial pad mode.

Table E-1 Alphanumeric Character Set

Digit	Dial Pad Mode		
	Upper Case '[A]'	**Lower Case '[a]'**	**Symbols '[S]'**
1	@ : /	@ : /	@ : /
2	A B C	a b c	| $ %
3	D E F	d e f	^ & _
4	G H I	g h i	( ) ?
5	J K L	j k l	- + , /
6	M N O	m n o	< > =
7	P Q R S	p q r s	; :
8	T U V	t u v	' " `
9	W X Y Z	w x y z	{ }
0	. , ? !	. , ? !	. , ? !
*	. *	. *	. *
#	Space (' ')	Space (' ')	# [ ]

Appendix F. Auto Provisioning Routine

General

The LIP-8002/8002A incorporates software routines to automatically provision the configuration and software of the phone. To initialize LAN settings, DHCP to gather basic network and provisioning server information, downloads configuration and software files from the provisioning server, and registers and subscribes with the SIP call server using the configuration data.

The phone attempts contact with the provisioning server to download configuration files then checks for software updates. For more details on the configuration files, see Appendix D. Note downloads can be disabled separately for the configuration and software files. As a default, the phone will employ TFTP as the download protocol. Optionally, the phone can employ HTTP/HTTPS as the provisioning server protocol. If the DHCP server returns a URL for the HTTP/S Option (default Opt. #160), the IP Phone shifts provisioning protocol to HTTP/HTTPS using the URL provided to access the server.

DHCP

During the initial and subsequent boot-up cycles, the phone requests various data options from the DHCP server. The following lists the requested options.

- DHCP Option #1 Subnet Mask
- DHCP Option #2 Time offset in seconds of the local time zone from GMT.
- DHCP Option #3 Default Gateway (router) IP Address
- DHCP Option #6 DNS Server IP Address
- DHCP Option #42 SNTP Server IP Address
- DHCP Option #50 IP address of the phone
- DHCP Option #66 TFTP Server Name
- DHCP Option #150 TFTP Server IP Address has priority over Option 66 when set.
- DHCP Option #160 default HTTP/HTTPS URL for provisioning

TFTP Downloads

When TFTP is employed, the IP Phone will contact the server provided from the DHCP server using the name or IP address from DHCP Option #66 or #150 or the TFTP server in memory when the DHCP option is not available. The IP Phone will request the TFTP server send the global configuration file ("sysconf_8002_sip.cfg"). Data from an unencrypted file is transferred to the phone memory.

Changes in the provisioning data from the global configuration file are transferred to the phone memory and the download process is repeated for the phone-specific configuration file. Configuration data in the phone specific file overwrites data from the global file.

If a value is changed for a field marked with an asterisk in Table D-1, the reboot flag will be enabled and the IP Phone will reboot after the download cycle is completed. By default, the phone specific file uses the MAC address of the phone as the file name with no file extension.

After completing download of the configuration files, the IP Phone requests the software version file from the server. This file, which is named "IP_8002_VER_SIP2", is an ASCII text file that identifies the software version available on the server as well as the names of software files to be downloaded. The phone compares the software version identified in the file with the version in the phone. If different, the files identified in the version file are downloaded. Typically, the files required for download include the main firmware file and an xml file but may include other files as needed for the upgrade. These files along with the version file are available from LG-Ericsson. If desired, comments may be added to the version file using "#" as the leading character of the comment line.

If software download is required, the IP Phone will enable the reboot flag and after completing the download cycle, the IP Phone reboots.

HTTP/HTTPS downloads

When HTTP/HTTPS is employed, the IP Phone will contact the URL defined in the memory or in the HTTPS DHCP Option, default #160. When the DHCP option is populated, the phone employs HTTP/HTTPS instead of the default TFTP mode. If the server uses HTTPS, the phone will establish an open SSL link. During the SSL handshake, the phone may optionally verify that the validity of the HTTPS Certificate of Authority (CA). The IP Phone verifies that the CA is from a valid authority such as Verisign®, the server providing the CA is correct and the certificate has not expired. The latter is verifiable only if an SNTP server provides time to the IP Phone. Each of these three verifications is enabled by default but can be disabled using the configuration file or manual configuration. The IP Phone supports several standard CAs including Verisign®. In addition, a CA may be copied to the phone using the Web Manager.

If the HTTP/HTTPS server requires authentication, the IP Phone uses the HTTPS ID and Password assigned; the IP Phone supports the basic and digest method. At this point, the phone will follow the same routine as for TFTP. The IP Phone requests the global configuration file ("sysconf_8002_sip.cfg"). If the file is provided as an unencrypted file, data is transferred to memory.

Changes in the provisioning data from the global configuration file are transferred to the phone memory and the download process is repeated for the phone-specific file. Configuration data in the phone specific file overwrites data from the global file.

Note if a value is changed for a field marked with an asterisk in Table D-1, the reboot flag will be enabled and the IP Phone will reboot after the download cycle is completed. By default, the phone specific file uses the MAC address of the phone as the file name with no file extension. For HTTP/S, a different file name can be assigned in the phone to use as file name but the MAC address must be included as part of the file name or path.

After completing download of the configuration files, the IP Phone requests the software version file from the server. This file, which is named "IP_8002_VER_SIP2", is an ASCII text file that identifies the software version available on the server as well as the names of the files for

download. The phone first requests the file *"mac address"*/IP_8002_VER_SIP2, if not found, the file IP_8002_VER_SIP2 is requested. The phone compares the software version identified in the file with the version in the phone. If different, the files identified in the version file are downloaded. Typically, the files required for download include the main firmware file and an xml file but may include other files as needed for the upgrade. These files along with the version file are available from LG-Ericsson. If desired, comments may be added to the version file using "#" as the leading character of the comment line.

If the software download is required, after completion of the download cycle the IP Phone reboots.

SIP Server Check-in Request

Upon receipt of a SIP check-in or resync request from the SIP server (Notify check sync event message), the IP Phone initiates the reboot process. The phone must be idle to initiate provisioning, if busy, upon return to idle, the process starts. During the process, the IP Phone downloads any configuration file changes as well as new software regardless of the settings for the software and configuration download flags.