SpectraLink 8002 Wireless Telephone

Administration Guide

Session Initiation Protocol (SIP)

June 2008 Edition
1725-36034-001
Version C

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Contact Information

Please contact your Polycom Authorized Reseller for assistance.

Polycom, Inc.
4750 Willow Road,
Pleasanton, CA 94588
http://www.polycom.com

Essential Installation Pointers

You must update the software with the most recent code.

- Visit http://www.polycom.com/usa.en/support/voice/wi-fi/spectralink_8002_wireless.html. Additional information about updating handset code is in Chapter 3 *Software License and Protocol Management*.
- Ensure the License Option on the Admin Menu is set to **22**. See Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration*; Section *The Admin (Administration) Men*u.

WMM QoS must be supported and enabled on the Wi-Fi access point(s).

- When using the SpectraLink 8002 Wireless Telephones, WMM QoS is mandatory to ensure voice packets are delivered in a timely fashion. The AP must have WMM enabled. If WMM is not enabled or if the AP does not support it, the handset will stop and display an error (e.g. **No Net Found**) after firmware update.
- Be aware that not all APs enable WMM by default. In fact, many APs require updated firmware to support WMM. Follow the AP's manufacturer instructions to ensure WMM is supported and enabled.

Data APs may not support VoWiFi.

- The 8002 handset has been tested with many SMB access points. We have found that VoWiFi performance varies between AP models and even firmware versions. An AP being used for data does not guarantee satisfactory Voice over Wi-Fi performance.
- We recommend that you update to the latest AP firmware version if possible. If you experience unexpected behaviors, try a different AP type.

A TFTP server must be on the local network with the SIP configuration files present.

- A TFTP server is a required component of the LAN. The SIP configuration files (and the handset code) reside on the TFTP server and the handset retrieves the SIP files at every power up. It is not recommended or supported to have the TFTP server or call-server across a WAN.
- The handset must be configured to point to a TFTP server for the SIP configuration files. This is accomplished by setting the handset for DHCP or by using static IP addressing. See Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration*, section *Network Config*.
- If the handset cannot find the files, it will stop and report an error.

A knowledge base to assist troubleshooting is available at:

http://knowledgebase.polycom.com/

About this Guide

This document explains how to configure and maintain the SpectraLink 8002 Wireless Telephones with Session Initiation Protocol (SIP); a protocol for the control of voice over IP (VoIP) calls.

Polycom Model Numbers

This document covers the following registered model number:

SNP2400

Related Documents

SpectraLink 8002 Wireless Telephone and Accessories User Guide.
(1725-36027-001)

SpectraLink 8002 Wireless Telephone Quick Reference Guide
(1725-36026-001)

SpectraLink 8002 Wireless Telephone Deployment Guide
(1725-36166-001)

Available at http://www.polycom.com/usa/en/support/voice/wi-fi/spectralink_8002_wireless.html.

Open Application Interface (OAI) Specification
(72-0052-00)
Technical specifications available by request.
Please contact Customer Support.

Customer Support Hotline

Polycom wants you to have a successful installation. If you have questions please contact our Customer Support Hotline at (800) 775-5330. The hotline is open Monday through Friday, 6 a.m. to 6 p.m. Mountain time.

Icons and Conventions

This manual uses the following icons and conventions.

⚠	Caution! Follow these instructions carefully to avoid danger.
📌	Note these instructions carefully.
Label	This typeface indicates a key, label, or button on Polycom hardware.

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1

SpectraLink Wireless Telephone Overview

SpectraLink 8002 Wireless Telephones are mobile handsets for workplace IP telephone systems. The handsets operate over an 802.11b wireless Ethernet LAN providing users a wireless voice over IP (VoIP) extension. By seamlessly integrating into a SIP environment, handset users are provided with high-quality mobile voice communications throughout the workplace. The handset gives users the freedom to roam throughout the workplace while providing all the features and functionality of a SIP desk phone.

The handsets reside on the wireless LAN with other wireless devices using direct sequence spread spectrum (DSSS) radio technology. Access points that transmit these signals must use WMM for QoS. The handset radio transmits and receives packets at up to 11 Mb/s.

In the SIP environment, the handsets support up to three proxy servers. Each handset may have five line appearances and two calls per line. If required, each handset may have multiple sets of credentials to identify itself by current user.

> The latest software versions are required to support the features described in this document.

WMM-Compatible Access Points

SpectraLink 8002 Wireless Telephones support basic WMM™ as part of the 802.11e protocol. The handset requires an AP that supports WMM.

The WMM specification for the SpectraLink 8002 Wireless Telephone does not accommodate admission control. The AP may not use the mechanism of admission control in the wireless environment where SpectraLink 8002 Wireless Telephones are operating.

WMM must be enabled when configuring access points for the SpectraLink 8002 Wireless Telephones. The handset will not work properly if WMM is not enabled.

Refer to the access point manufacturer's directions for instructions on enabling WMM.

Visit the Wi-Fi Alliance website www.wi-fi.org to search for APs that support WMM.

Security Options

The SpectraLink 8002 Wireless Telephone supports Wired Equivalent Privacy (WEP) as defined by the 802.11 specification. Polycom offers the product with both 40-bit and 128-bit encryption.

SpectraLink 8002 Wireless Telephones also support the 802.11i protocol including Wi-Fi Protected Access (WPA™ and WPA2™)—PSK.

Quick Start Guide

1. A wireless LAN must be properly configured and operational through the use of 802.11 wireless APs. Ensure WMM is enabled on the APs.
2. Configure a TFTP server on the LAN. The TFTP server is used to load the handset software and SIP configuration files into the handsets. See Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration* for DHCP information.
3. The supported SIP system components must be connected to your network and completely operational.
4. Visit http://www.polycom.com/usa.en/support/voice/wi-fi/spectralink_8002_wireless.html to download the latest handset software and example SIP configuration files.
5. Ensure the handset software is properly loaded on the TFTP server that has been designated for the handset code. See Chapter 3 *Software License and Protocol Management,* section *Configuration Process*.
6. Create/modify configuration files on the SIP TFTP server to define parameters for the SIP application. See Chapter 4 *Programming the SIP Handset Features*, section *SIP TFTP Server Configuration Files*. The TFTP server may be the same one referenced in step 2 above.
7. Configure your handset per Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration*. Ensure these configuration options are set:

   Set **License Option** to **22**.

   a With the handset powered off, press and hold the **Power Off** key. While holding the **Power Off** key, press and release the **Power On** key.

   b Select **Phone Config** option. Press **OK** softkey.

   c Select **License Option**. Press **OK** softkey.

   d Select **Set Current**. Press **OK** softkey.

   e Select **Type 022**. Press **OK** softkey.

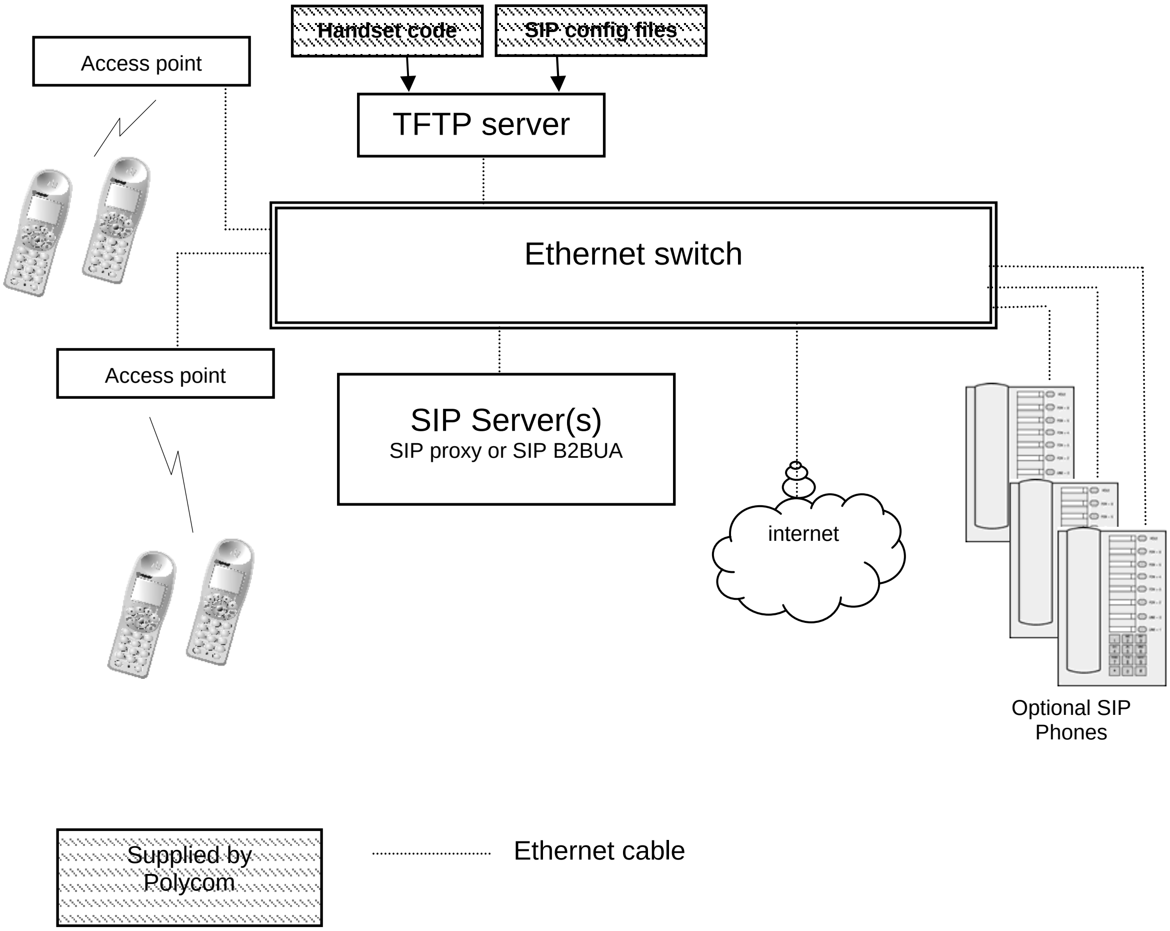
f Verify **Set Current** (**022**). Press **Up** softkey. Press **Up** repeatedly to exit the Admin Menu.

Consult the Admin Menu section in Chapter 2 and follow a similar sequence to ensure the following options are set:

- SIP Registration, Reg 1 and Login
- IP Addresses: DHCP or static for each listed component, including SIP TFTP server and TFTP server.
- Security
- Regulatory Domain
- Preferred Channels

System Diagram

The following diagram shows the SpectraLink components residing on a network with APs and a wireless LAN Ethernet Switch.


Handset code
SIP config files
Access point
TFTP server
Ethernet switch
Access point
SIP Server(s)
SIP proxy or SIP B2BUA
internet
Optional SIP Phones
Supplied by Polycom
Ethernet cable

System Components

SpectraLink 8002 Wireless Telephone

The SpectraLink 8002 Wireless Telephone is a lightweight, durable handset specifically designed for mobile workplace use within a facility using SIP and an 802.11 wireless LAN. The handsets are to be used on-premises; they are not cellular or satellite phones.

SpectraLink 8002 Wireless Telephones use direct-sequence spread spectrum radio technology (DSSS) to transmit audio packets over wireless LAN APs that support WMM.

Access points

Supplied by third-party vendors, APs provide the connection between the wired Ethernet LAN and the wireless (802.11) LAN. Access points must be positioned in all areas where handsets will be used. The number and placement of APs will affect the coverage area and capacity of the wireless system.

Access points must support WMM, WMM must be enabled and admission control must be disabled.

Ethernet switch

Interconnects multiple network devices, including the proxy server(s), wired IP phones and the APs. Ethernet switches provide the highest performance networks, which can handle combined voice and data traffic, and are required when using the SpectraLink 8002 Wireless Telephones.

Although a single Ethernet switch network is recommended, the handsets can operate in larger, more complex networks, including networks with multiple Ethernet switches, routers, VLANs and/or multiple subnets. However, in such networks, it is possible that voice quality may suffer. Any network that consists of more than a single Ethernet switch should be thoroughly tested to ensure any quality issues are detected.

Note that the SpectraLink 8002 Wireless Telephones cannot “roam” from one subnet to another. If routers and multiple subnets are in use, the handsets must only use APs attached to a single subnet, or be powered off and back on to switch to a different subnet.

SIP server

The SIP server is a component from a third-party vendor that provides access to telephony services. The handsets can recognize up to three distinct SIP servers in a single system.

The SIP proxy server connects to another device such as a PBX or gateway and from there, other wired phones and the PSTN.

> The SpectraLink 8002 Wireless Telephone has been tested and certified with Digium® Asterisk® Business Edition server.

SIP phone

The optional wired-LAN desksets such as those provided by Polycom.

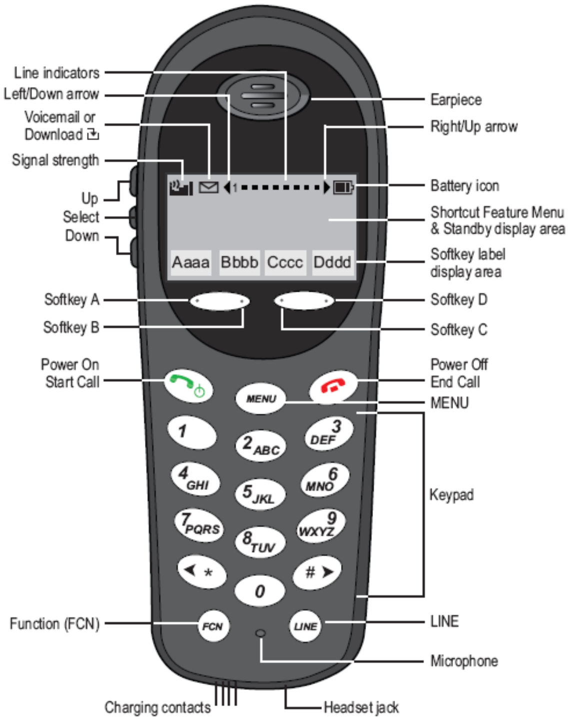
TFTP server

A TFTP server is required in the system to distribute software to the handsets. May be on a different subnet than the gateway and/or APs.

SIP TFTP server

A SIP TFTP server is required in the system to deliver SIP configuration parameters to the SIP handset whenever a SIP handset is powered-up. The location of the SIP TFTP server is separately specified in SIP handset administration parameters (See Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration*, section *The Admin Menu*.) Normally, the SIP TFTP server is the same as the network TFTP server, see above.

Handset Specifications


Line indicators
Left/Down arrow
Voicemail or Download
Signal strength
Up
Select
Down
Softkey A
Softkey B
Power On
Start Call
Function (FCN)
Charging contacts
Headset jack
Earpiece
Right/Up arrow
Battery icon
Shortcut Feature Menu & Standby display area
Softkey label display area
Softkey D
Softkey C
Power Off
End Call
MENU
Keypad
LINE
Microphone
Aaaa
Bbbb
Cccc
Dddd

SpectraLink 8002 Wireless Telephone

Radio frequency:	2.4000 – 2.4835 GHz
Transmission type:	Direct-sequence spread spectrum (DSSS)
Transmit data rate:	Up to 11 Mb/s
Radio QoS:	WMM
Wireless security:	Wired Equivalent Privacy (WEP), 40-bit and 128-bit, WPA-PSK, WPA2-PSK
FCC certification:	Part 15.247
Management:	DHCP, TFTP
Voice encoding:	G.711 (A- and mu-law),
VoIP Protocols:	SIP
Transmit power:	100 mW peak, < 10 mW average
Display:	Pixel-based (up to 4-line x 18-character) alphanumeric, plus line and status indicators.
Dimensions:	5.5” x 2.0” x 0.9” (14.0 x 5.1 x 2.3 cm)
Weight:	4.2 ounces (119.0 g)
Battery capacity:	Up to 3 hours talk time, up to 50 hours standby

Startup Sequence

The SpectraLink 8002 Wireless Telephone goes through an initialization sequence at startup. The line icons 1-9 display and count down as the handset steps through this sequence. This is usually very rapid. If there is difficulty at any step that prevents initialization from continuing, an error message will display and the related icon(s) will stay on. Please see the error table at the back of this document for instructions on how to handle error messages that occur during initialization.

Icon	**The icon(s) shown in bold turns off when:**
12345678**9**	The handset has located and authenticated and associated with at least one AP, and is proceeding to bring up higher-layer networking functions.
1234567**8**	The handset is either configured for Static IP, or if configured for DHCP, the DHCP discovery process has started.
123456**7**	If DHCP is configured, a DHCP response was received which contains a good DNS server configuration.
12345**6**	Indicates Static IP configuration.
1234**5**	All networking functions are complete (notably, DHCP) and the handset is establishing its connection to the LAN.
123**4**	The LAN connection is complete.
12**3**	The handset has located a generic SIP config file.
1**2**	The handset has located a handset-specific SIP config file.
(no icons) Registering	Handset is attempting to register each of the specified line contacts.
(no icons) EXT. XXXXX	Handset has registered with at least one contact on one proxy server. Initialization is complete. The handset is in standby mode ready to receive and place calls. The line one contact is displayed.

During the last three steps of this process, the handset contacts the SIP TFTP server and downloads general information about the proxy server(s), downloads specific information pertaining to the handset, registers with the proxy server(s), and verifies handset credentials. Once this process is complete, the handset is ready to use.

If the username and password have not been defined in the Admin menu, you will be prompted to enter both of these items before the extension number can display. The user name must correspond to a handset-specific configuration file. If the file is not found, an error message will appear and the handset will restart. See Chapter 4 *Programming the SIP Handset Features*.

2

SpectraLink 8002 Wireless Telephone Configuration

Each SpectraLink 8002 handset may be configured for site-specific requirements by opening the Admin menu and selecting options or entering specific information. Any settings entered in the Admin menu must conform to system settings. Only the handset being configured is affected by the Admin menu settings.

The Wireless Telephone user may select several usability options from the Standby menu, described below in the *User-defined Preferences* section. This information is also provided in the end-user manual.

The SpectraLink Configuration Cradle is an accessory device designed to automate configuration of SpectraLink 8002 Wireless Telephones. The Configuration Cradle is connected to a PC via a serial cable. A downloadable Configuration Cradle program runs on the PC and enables the system administrator to establish and store configuration options for system, group and user categories. A configuration plan may be set up in the program and downloaded into a handset or a configured handset may be placed in the cradle and its configuration may be uploaded and edited or saved.

Please see your service representative or contact Polycom Customer Service for more information about this time-saving device.

The Admin (Administration) Menu

The Admin menu contains configuration options that are stored locally (on each handset). Each handset is independent, and if the default settings are not desired, the Admin menu options must be set in each handset requiring different settings.

Opening the Admin menu

1. With the handset powered off, press and hold the **Power Off** key. While holding the **Power Off** key, press and release the **Power On** key.
2. The Admin menu displays.

If an Admin password has been set, the display will require its entry before opening the Admin menu. The default password is 123456. If no password is set, the display will proceed directly into the Admin menu.

Entering and editing Admin menu options

An asterisk (*) next to an option on the display indicates that it is selected. Use the **Up**, **Down**, and **Select** side buttons and the softkeys to navigate and select:

Up/Down button**:**	Display previous/next menu item.
Select button:	Selects the menu item or option.
OK softkey:	Selects the menu item or option.
Save softkey:	Saves the entry.
Bksp softkey:	Backspaces to allow editing of entry.
Cncl softkey:	Cancels edit and returns to previous menu level.
Up softkey:	Returns to previous menu level.
Exit softkey:	Exits the menu (at the top level).
End Call key:	Exits to standby mode (from any level)

Alphanumeric string entry

1. Press the first digit/letter. The digit displays. Press the key again to scroll through the letters associated with that key.

Example: if the **2** key is repeatedly pressed, the handset will display **2**, **A**, **B**, and **C**, then **a**, **b**, and **c**.

The following table shows which key will allow you to enter non-numeric characters or other characters not represented on the keypad.

To Enter	Press
. - _ ! # $ % & ‘ ( ) * + , : ; / \ = @ ~	1
Space	0
Q,q	7
Z,z	9

2. When the correct entry displays, press the right arrow key (**#**) to move on to the next character. Repeat for each digit/letter of the entry. To erase, press the left arrow key (*****) or **Bksp** softkey to erase the previous character.

3. Press the **Save** softkey to save the entry.
Press the **Up** softkey to abort and return to the menu without saving any changes.

The following table lists the Admin menu items. The default settings are shown in the table below with an * prior to the option. Detailed descriptions of each item appear below the table.

Admin menu

Admin Menu Items	2nd Level	3rd Level	4th Level	5th Level
Phone Config	License Option	Set Current	22	
	SIP Registration	Reg 1 and Login Reg 2 Reg 3 Reg 4	[for each option] Username Password	
	OAI On/Off	*Enable OAI Disable OAI		
	Admin Password	Enter PW	Re-enter Password	
Network Config	IP Addresses	*Use DHCP		
		Static IP	Phone IP TFTP Server IP Default Gateway Subnet Mask Syslog Server IP SIP TFTP Svr IP OAI Server IP	
	ESS ID	*Learn Once		
		Learn Always		
		Static Entry		
	Security	*None		
		WEP	Authentication	Open System Shared Key
			WEP On/Off	WEP Off WEP On
			Key Information	Default Key Key Length Key 1-4
			Rotation Secret	
		WPA-PSK	Passphrase Pre-Shared Key	
		WPA2-PSK	Passphrase Pre-Shared Key	

Admin Menu Items	2nd Level	3rd Level	4th Level	5th Level
	Reg. Domain	None 01 02 04 05 06		
	Transmit Power	*Maximum 50 mW † 30 mW † 20 mW 15 mW 10 mW 5 mW		
	Preferred Channels	Channel 1‡ … Channel 14‡		
Diagnostics	Run Site Survey			
	Diagnostics Mode	Diagnostics On *Diagnostics Off		
	Syslog Mode	*Disabled Errors Events Full		
Restore Defaults				

**default setting*

† *50 mW and 30 mW only appear if Regulatory Domain is set to None or 01.*

‡ *Available channel selections depend on domain setting.*

Phone Config

License Option

The License Option indicates the VoIP protocol that your site is licensed to download and run. The SIP protocol used for the SpectraLink 8002 Wireless Telephones requires license option selection **22**. Any other protocol will cause the handset to malfunction.

> The handset will give you a **No SVP Response** error if the **License Option** is not set to **22**. Be sure to set this option immediately, before trying to associate the handset with an access point.

SIP Registration

Individual handsets can be configured to correspond with a handset-specific configuration file in the TFTP server. The handset is then automatically identified at startup.

If the username and password information is not configured in the Admin menu, then this information will be requested at startup. The username and password entered must agree with a corresponding handset-specific configuration file. See Chapter 4 *Programming the SIP Handset*, section *SIP TFTP Server Configuration Files*.

Reg 1 and Login allows you to specify a username and password for automatically acquiring SIP configuration information. If no username is specified, the SIP handset will request username and password at startup and any additional registrations specified here are ignored.

The username should correspond to the primary (line 1) dial number assigned to the user. The username and password should also correspond to the authentication credentials as created by your system administrator for your primary line registration. Usernames or passwords can be erased by selecting the item, then pressing **Bksp** softkey and then **Save** softkey.

Reg 2, **Reg 3**, and **Reg 4** allow you to specify additional authentication usernames and passwords that may be required by your handset for any additional line appearances (registrations) that may appear in the specific user’s configuration file. This information will be ignored if a **Reg 1 and Login** username is not provided.

OAI On/Off

SpectraLink's Open Application Interface (OAI) enables third-party computer applications to display alphanumeric messages on the handset display and take input from the handset keypad. Refer to the OAI Specification documentation for information about administering the OAI Gateway and the services it can provide.

If OAI is enabled, and an IP address (called the **OAI Server IP**) is available to the handset (either via DHCP or Static IP configuration), the handset will communicate with the OAI Gateway at power-on, and periodically while it is powered-on. If you don't have a SpectraLink 8000 OAI Gateway installed at your site, you should disable the OAI feature to preserve network bandwidth and battery life.

To conserve handset battery life, do not enable OAI unless an OAI gateway is present and an application is available.

Admin Password

The Admin Password controls access to the administration functions in the Admin menu. The password must be set in each handset for which controlled access is desired. Handsets are shipped without any Admin menu password. Data entry for the password uses the alphanumeric string entry technique. A password may be up to 32 characters. Type the password and press the **Save** softkey. A confirmation prompt will appear. Type the password again and press the **Save** softkey. If the passwords match, the Admin password has been set.

If you **Save** with no entry, the password is erased and the display will not require it before displaying the Admin menu.

Network Config

IP Address

There are two modes in which the handset can operate: DHCP-enabled or Static IP. Select the mode for operation from the IP Address menu:

***Use DHCP** Will use Dynamic Host Configuration Protocol to assign an IP Address each time the handset is turned on. If DHCP is enabled, the handset also receives all other IP Address configurations from the DHCP server.

Dynamic Host Configuration Protocol (DHCP) is a standardized protocol that enables clients to be dynamically assigned with various configuration parameters, such as an IP address, subnet mask, default gateway, and other critical network configuration information. DHCP servers centrally manage such configuration data, and are configured by network administrators with settings that are appropriate for a given network environment. The handset will use the following DHCP options if DHCP use is enabled.

Option	Meaning
1	Subnet mask
3	Default gateway
6	DNS server
7	Syslog server
15	Domain name
66	TFTP server
152	SpectraLink 8000 OAI Gateway
siaddr	Boot server or next server

Static IP Allows you to manually set a fixed IP Address. If selected, the handset will prompt for the IP addresses for each configurable network component. When entering addresses, enter the digits only including leading zeroes, e.g. 192.168.001.002. No periods are required.

Regardless of the mode in which the handset is operating, the following components are required and must be configured as part of the SIP system:

Phone IP The IP address of the handset. This is automatically assigned if DHCP is used. If using Static IP configuration, you must obtain a unique IP Address for each handset from your network administrator.

TFTP Server IP The IP address of a TFTP server on your network, which holds software images for updating the handsets and contains the handset files. If this feature is configured (not set to 0.0.0.0 or 255.255.255.255) with either Static IP configuration or using DHCP option 66 (TFTP server), or the boot server/next server (siaddr) field, the handset will check for newer software each time it is powered on or comes back into range of your network. This check takes only a second and ensures that all handsets in your network are kept up-to-date with the same version of software.

The TFTP server and the SIP TFTP server may be the same device with the same IP address or may be different devices with different IP addresses.

Default Gateway and Subnet Mask Used to identify subnets, when using a complex network, which includes routers. Both of these must be configured either with an IP address under Static IP (not set to 000.000.000.000 or 255.255.255.255) or with DHCP for the handset to contact any network components on a different subnet. If configured on the DHCP server, use option 3 for the Default Gateway and option 1 for the Subnet Mask. Contact the network administrator for the proper settings for the network.

> Note that the wireless telephones cannot “roam” across subnets, since they cannot change IP addresses while operational. Ensure that all APs are attached to the same subnet for proper operation. The handset can change subnets if DHCP is enabled and the handset is powered-off then back on when within range of only APs on the new subnet.

Syslog Server IP The IP address of the syslog server. See Chapter 5 *Diagnostic Tools* for more information.

SIP TFTP Server IP The IP address of a TFTP server on your network, which holds SIP configuration files. In static mode, this parameter must be configured with an IP address. In DHCP mode, the SIP TFTP server may be specified by defining the address on the DNS server for the name “siptftp” If this is not defined, the address specified in option 66 will be used.

The TFTP server and the SIP TFTP server may be the same device with the same IP address or may be different devices with different IP addresses.

OAI Server IP The IP address of the SpectraLink OAI Gateway. If using Static IP configuration, this is simply the IP address of the SpectraLink OAI Gateway. If DHCP is being used, the handset will try the DHCP option 152.

Domain Name System (DNS), an industry-standard protocol, locates computers on an IP-based network. IP networks rely on number-based addresses to move information on the network. However, it is easier to remember friendly names than number-based addresses, so it is necessary to translate user-friendly names into addresses that the network can recognize. The handset will use DNS to automatically translate names into IP addresses for the TFTP server.

In DHCP mode, the SIP handset will use DNS to look up an address for the logical name “siptftp” to locate the SIP TFTP file server. If this logical name is undefined, then the address specified by option 66 is used for the SIP TFTP server.

ESSID

Select the option that will enable the handset to acquire APs with the correct ESSID (Extended Service Set ID, aka SSID) each time it is turned on.

Note about Automatic Learn options: Broadcast ESSID must be enabled in the APs for ESSID learning to function when multiple ESSIDs can be seen by the handset from one location. Overlapping wireless systems complicate the use of ESSID learning as the handset in an overlapping area could receive conflicting signals. If this is the situation at your site, use Static Entry or Learn Once in an area without overlapping ESSIDs.

***Learn Once** Allows the handset to scan all ESSIDs for a DHCP server. Once either is found, the handset retains the ESSID from whichever AP it associates with at that point. When overlapping wireless systems exist, the Learn Once feature allows the handset to use only the ESSID established at first learn at all subsequent power ons. This ESSID is retained by the handset until the ESSID option is reselected in the Admin menu, thus erasing the stored ESSID.

Learn Always Allows the handset to automatically learn the ESSID at each power on or loss of contact with the wireless LAN (out of range). This may be useful if the handset will be used at more than one site.

Static Entry If the APs do not broadcast their ESSID, or if there are overlapping wireless systems in use at the site, enter the ESSID manually.

Security

***NONE** disables any 802.11 encryption or security authentication mechanisms.

For WEP, WPA-PSK and WPA2-PSK set each of the following options to match exactly the settings in the APs.

The encryption key or encryption passphrase displays as it is entered. For security reasons codes will not display again once entered, even if the entry screen is revisited.

WEP (Wired Equivalent Privacy) is a wireless encryption protocol that encrypts data frames on the wireless medium allowing for greater security in the wireless network. If WEP is required at this site, you must configure each handset to correspond with the encryption protocol set up in the APs. Select the entries from the options below to enable the handset to acquire the system.

Authentication Select either **Open System** or **Shared Key**.

WEP On/Off Select either **WEP Off** or **WEP On**.

Key Information

Default Key Enter the key number specified for use by the handsets. This will be **1** through **4.**

Key Length Select either **40-bit** or **128-bit** depending on the key length specified for use at this location.

Key 1 through 4 Scroll to the key option that corresponds to the Default Key that was entered above. Enter the encryption key as a sequence of hexadecimal characters. (Use the 2 and 3 keys to access hexadecimal digits A-F, use the Right Arrow key to advance to the next digit, and the Left Arrow key to backspace.) For 40-bit keys, enter 10 hex characters. For 128-bit keys, enter 26 hex characters. The display will scroll as needed.

Rotation Secret This is used for proprietary WEP key rotation. Refer to your custom document if this feature is supported in your system.

WPA-PSK The security features of WPA (Wi-Fi Protected Access) using PSK (pre-shared key) are available and may be used if supported by the APs in the facility. Select either **Passphrase** and enter a passphrase between eight and 63 characters in length or **Pre-shared Key** and enter the 256-bit key code.

WPA2-PSK The security features of WPA2 (Wi-Fi Protected Access) using PSK are available and may be used if supported by the APs in the facility. Select either **Passphrase** and enter a passphrase between eight and 63 characters in length or **Pre-shared Key** and enter the 256-bit key code.

Regulatory Domain

The Regulatory Domain will default to **None** on the handset display. FCC requirements dictate that the menu for changing the domain be available by password, which in this case is the **LINE** key. To change the domain, press **LINE** and then enter the digits that represent the site's domain. Note that both digits must be entered.

01 North America

02 Europe (except Spain and France); Japan (channels 1-13)

04 Spain

05 France

06 Korea

> As of this writing, Spain and France are adopting the general European regulatory rules. Check with your wireless LAN administrator or supplier regarding which domain to enter in these countries.

Transmit Power

Available Transmit Power is regulated by domain. The Regulatory Domain setting above affects the options available for this setting. The default setting is **Maximum** which in North America is 100 mW. The Maximum in other domains is 30 mW. Transmit Power may be set to a lower number if necessary by selecting one of the other levels. If changed from the default, ensure the Transmit Power setting is the same on all handsets and all APs.

> Power is limited to 30mW in regulatory domains other than North America.

> See *Appendix A* for the full list of certified countries and the corresponding domain for each.

Preferred Channel

The Preferred Channel option lists the channels that are permitted within the regulatory domain previously set. To speed up handoff, the handset will look for APs on the preferred channel list first. If no AP is found, the handset will then will look for APs on the non-preferred channels. Selected channels are marked with an asterisk. You may select more than one channel.

Diagnostics

Run Site Survey

The **Site Survey** mode is activated by selecting this option. The site survey starts running immediately upon selecting this option. See Chapter 5 *Diagnostic Tools* for more information about site survey.

Diagnostics Mode

See Chapter 5 *Diagnostic Tools* for a detailed explanation of the **Diagnostics Mode** options.

Syslog Mode

See Chapter 5 *Diagnostic Tools* for a detailed explanation of the **Syslog Mode** options.

Restore Defaults

The **Restore Defaults** option will set all user and administrative parameters except License Option to their factory defaults.

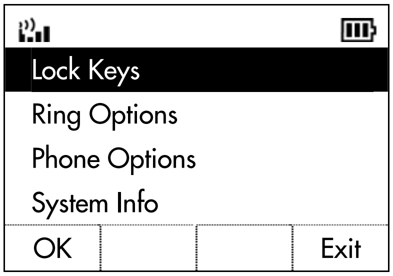
User-defined Preferences (Standby Menu)

The following user-defined preferences are also covered in the *SpectraLink 8002 Wireless Telephone User Guide*. The system administrator can refer to this list for more information about customizing handset settings.

To configure the following options, the handset must acquire the system (no error message may display) and be at the extension display. This is the standby mode. While in the standby mode, press and hold **FCN** briefly to open the user options menu. Use the following keys to display and select options:

Up/Down buttons:	Display previous/next menu item.
Select button:	Selects the menu item or option.
OK softkey:	Selects the menu item or option.
Save softkey:	Saves the entry.
Bksp softkey:	Backspaces to allow editing of entry.
Cncl softkey:	Cancels edit and return to previous menu level.
Up softkey:	Returns to previous menu level.
Exit softkey:	Exits the menu (at the top level).
End Call key:	Exits to standby state (from any level).

Standby mode, options menu



Standby menu

Standby Menu Items	2nd Level	3rd Level	4th Level
Lock Keys			
Ring Options	Telephone Ring	Ring Cadence	Off *PBX Continuous Short Pulse Long Pulse
		Ring Tone	*Tone 1 Tone 2 Tone 3 Tone 4 Tone 5
		Vibrate Cadence	*Off PBX Continuous Short Pulse Long Pulse
		Ring Delay**	*No Delay 5 Second Delay 10 Second Delay
	Auxiliary Ring 1	[same as Telephone Ring]	
	Auxiliary Ring 2		
Phone Options	Noise Mode	*Normal High Severe	
	Key Tones	*Enable Tones Disable Tones	
	Warning Tones	*Enable warnings Disable warnings	
	Display Contrast	Contrast %	
	Keypad Autolock	*Disable 5 seconds 10 seconds 20 seconds	

**Default setting.*

***Option appears when Ring Cadence and Vibrate Cadence are both set to a value other than "Off".*

Standby Menu Items	2nd Level	3rd Level	4th Level
System Info	Phone IP Addr		
	Server IP Addr		
	Firmware Version		

(Additional options may be present. Contact your service representative for information.)

Lock Keys

When enabled, the Keypad Lock option will lock the keypad immediately. If the keypad is locked, it may be unlocked by the end user pressing the **Unlk** softkey and then the **#** key.

Ring Options

The Ring Option allows the user to set the ring for three separate functions. Telephone ring is used for usual telephony functions. The Auxiliary Rings may be used to set different ringing patterns for OAI applications.

Telephone Ring

Telephone Ring allows the user to set a distinctive ring style, volume and sequence. Select from an audible ring or a vibrate-only ring or a vibrate ring along with or followed by an audible ring.

Ring Cadence The Ring Cadence is the rhythm of the ring. It may be set to a pre-programmed ring cadence, or it may be set to obtain its cadence from the proxy server. The PBX option is designed to utilize any distinctive rings sent by the proxy server while allowing the user to set unique rings for auxiliary applications.

Off: Silent

PBX: Proxy server determines ring cadence (e.g. the proxy server may send rings that differentiate between internal and external calls)

Continuous: Rings continually until answered

Short Pulse: Rings in short bursts

Long Pulse: Rings in long bursts

Ring Tone Select from five available tones (scroll to Tone 5 option). The Play softkey allows the user to preview the tone before selecting. If Ring Cadence is turned off, the Ring Tone option will not appear on the menu.

Vibrate Cadence Vibrate Cadence options are the same as for Ring Cadence (see above).

Ring Delay Ring Delay determines how long the vibrate cadence will play before the audible ring starts. If Ring Cadence and Vibrate Cadence are turned off, the Ring Delay option will not appear on the menu.

Auxiliary Ring 1 and Ring 2

Auxiliary rings are designed to be utilized by OAI applications, enabling the user to set a distinctive ring for these applications.

Phone Options

Noise Mode

Provides options that describe the noise level in your environment. Changing this mode to **High** or **Severe** adjusts the handset's microphone and in-ear speaker volume to account for background noise. Select **Normal** for most office environments; **High** for moderate background noise; or **Severe** for extremely noisy conditions. If you use **High** or **Severe** modes in a quiet environment, the person on the other end of the line may find it difficult to hear your voice.

Key Tones

Key Tones may be turned on or off and determine whether tones play when keys are pressed. Key Tones are enabled by default.

Warning Tones

The handset plays various Warning Tones such as system up or down, out of range, etc. These tones may be turned on or off and are enabled by default.

Display Contrast

The display may need to be adjusted for different lighting situations. Contrast may be set by pressing the **Up** and **Down** side buttons until the desired contrast is displayed and then pressing the **OK** softkey.

Keypad Autolock

The handset may be set to lock the keypad automatically when in standby mode. The automatic locking function of the keypad may be disabled (the default) or adjusted for a 5, 10 or 20-second delay before locking.

System Info

Phone IP Addr

Displays the IP address currently assigned to the handset.

Server IP Addr

Displays the IP address of the SpectraLink OAI Server (if installed).

Firmware Version

Displays the software version running the handset. The MAC address is the hardware identification number and is set at the factory. Three code numbers correspond to the three files that each version uses.

Downloader: pd11gl3.bin
Functional: pd11wsd3.bin
Phintl: pi11001.bin

3

Software License and Protocol Management

All SpectraLink 8002 Wireless Telephones are shipped from Polycom with SIP software already installed. You may need to download and install updates per the following instructions. The handsets may not function properly without downloading updated software.

The following details the process to properly configure SpectraLink 8002 Wireless Telephones and download software via over-the-air file transfer.

Requirements

- A wireless LAN must be properly configured and operational through the use of 802.11b wireless APs.
- A TFTP server must be available on the network in order to load the appropriate software into the handsets.
- Finally, ensure that the Battery Pack on the handset is fully charged.

Configuration Process

1. Download the latest SpectraLink 8002 Wireless Telephone software from http://www.polycom.com/usa.en/support/voice/wi-fi/spectralink_8002_wireless.html.
2. Download the latest version of the SIP code and place it on the TFTP server and ensure the TFTP server is started. The five files that are needed must be named:

   slnk_cfg.cfg
   pd11wsd.bin
   pd11wsd3.bin
   pd11gl3.bin
   pi110001.bin
3. Configure the handsets using the Admin menu. See Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration* for detailed configuration instructions.
   - a Ensure the **License Option** menu option is set to **22**.
   - b Configure the **SIP Registration** settings.
   - c If statically assigning IP addresses, ensure that the **Phone IP address**, **TFTP server IP**, **Subnet Mask**, **Default Gateway**, and **SIP TFTP server IP addresses** are accurate. If using a DHCP server, ensure that appropriate options are set for the TFTP server.
   - d Ensure the handset has properly configured **ESSID**. If you are accepting broadcast **ESSIDs** at your APs, the handset will automatically learn the ESSID information when powering on.
   - e Set the **Reg Domain**.
   - f Configure the handset security settings.
   - g Set the **Preferred Channels**.
4. Power cycle the handset.
5. If the handset code needs to be updated, the SIP code will now download to the handset. The status bar will increment fully across the display for each function that is being performed in the download process. Upon completion of the update process, the handset will re-boot with the new firmware.

6. During the second download evolution, the handset receives code from the TFTP server for system configuration and for its own settings. Once this second evolution is complete, the handset is ready to use.

> For future software upgrades, simply update the files that are stored on the TFTP server. Each time the handset is powered up, it will check with the TFTP server to ensure it has the proper software version.

4

Programming the SIP Handset Features

In order for the handset to function in the SIP environment, it downloads two files from the root directory of the SIP TFTP server during startup. The first file contains generic system information and is downloaded by every handset during the power-up sequence. A second file, unique for each handset, is then downloaded. It contains specific information for each handset such as username, password, and line appearances. Example files are available for download.

Both of these files must be created or the example files must be customized according to the site's configuration. This is usually done by the system integrator or administrator.

SIP TFTP Server Configuration Files

The two file types, generic and specific, are identical in format. Any or all of the configuration information can be contained in either file. Any information in the specific file that conflicts with the information in the generic file will take precedence over that in the generic file. Authentication information will be accepted from both files. For ease of administration, it is recommended both file types be utilized and that common settings be placed in the generic file..

Guidelines

- The files are in plain text, US-ASCII. The general form of the configuration file data is "parameter = value."
- The generic filename must be SIP_allusers.cfg.
- Each specific filename must have the form of SIP_[username].cfg where [username] is as assigned to each individual user by the system administrator (e.g. sip_3001.cfg). See **SIP Registration** in Chapter 2 *SpectraLink 8002 Wireless Telephone Configuration*, section *The Admin Menu*.

- Username requirements are: alphanumeric, no spaces, no punctuation, case is ignored, 1-16 characters.
- The generic file should contain proxy server information and other SIP system data.
- The specific file should contain data specific to each user such as authentication credentials and line appearance data.
- Some parameter lines accept more than one value, separated by a colon or semicolon character as defined in the following table.
- Any line that begins with a pound sign (#) is ignored.
- In general space characters are ignored. Space characters may be included in string values by replacing the space with "%20."
- If necessary, other special characters may be included by using a hexadecimal representation: (%hh) where hh is the representation of the character.
- The lines in the .cfg file may appear in any order although maintenance may be simplified by preserving the order in the supplied example file.

Create or modify each of the files according to the following instructions.

The generic file (sip_allusers.cfg)

The generic configuration file provides system information common to all handsets.

The handset-specific files (e.g. sip_3001.cfg)

The handset-specific configuration file provides specific information for the handset to identify itself and communicate to other phones. Each handset must have its own file with a unique filename. You may use the same parameters as the generic file when programming the handset files if you wish to override a common setting.

> You must configure a unique handset file for each handset being deployed. Typically each of these files is named with the extension number or name of the person assigned the handset. For example John Doe's handset could have a handset filename of sip_JohnDoe.cfg.

Proxy server commands

Use the following parameters when programming the configuration files. See the sample configuration files following this table for more detailed information.

Parameter	Value	Description	Notes
PROXYn_ADDR	xxx.xxx.xxx.xxx: pppp or proxyname:pppp	Proxy address. One proxy address is required but additional proxy addresses may be used to register secondary line appearances.	n = 1, 2, 3 xxx.xxx.xxx.xxx = IP address pppp = port (optional, 5060 is the default) proxyname = proxy server name (DHCP only)
PROXYn_DOMAIN	Domain name	Domain served by this proxy server.	n = 1, 2, 3 DOMAIN = [example: spectralink.com] Can be omitted if a specific proxy domain name is not defined at the proxy server. If omitted, defaults to the IP address of the proxy server (see above)
PROXYn_TYPE	Asterisk SER	Specify the manufacturer of each defined proxy server.	Used by the handset to perform proxy-specific actions based on known behavior for specific proxy types. Omit this parameter if the value is not Asterisk or SER.
PROXYn_KEYPRESS_2833	*enable disable *default (in example file)	Controls generation of in-stream RFC2833 formatted key press events. Match what is required by your PBX. See RFC2833. See config file example for more information.	n = 1, 2, 3 Must be disabled for OAI.
PROXYn_KEYPRESS_INFO	enable disable	Controls generation of SIP INFO requests to the SIP server for keypress events. Normally enabled.	n = 1, 2, 3 Must be enabled for OAI.

Parameter	Value	Description	Notes
PROXYn_REREG_SECS	35 seconds to 3600 seconds *default 3600 seconds	The requested expiration interval in REGISTER request messages.	n = 1, 2, 3 If the server has a lower maximum setting or a higher minimum setting than that requested, the server response takes precedence. See example file
PROXYn_KEEPALIVE_SECS	0 or 10 to 3600 seconds *default = 0	Specifies that the handset should send keepalives to the PROXYn server.	n = 1, 2, 3 The keepalive interval defaults to ZERO which disables the feature. When specified, the interval can be set to zero or to 10 to 3600 seconds. If a keepalive fails to get a response within the SIP 32 second timeout, then keepalives are terminated until the next successful registration to the proxy. See example file
PROXYn_HOLD_IP0	*enable disable *default (in example file)	Controls setting of the media stream IP destination address to 0 (zero) when a call is put on hold. See config file example for more information.	n = 1, 2, 3 Use for compatibility with older SIP servers that may not recognize newer stream attribute parameters for HOLD status.
PROXYn _MAIL_SUBSCR	name@xxx.xxx.xxx.xxx or sip:name@domain	Contact to whom the handset should subscribe for mail notification. See config file example for more information.	n = 1, 2, 3 name = mail server contact name. xxx.xxx.xxx.xxx = IP address. Domain where the mail resides[1].

---

[1] For mail notifications, in general you will need to define only one contact parameter for each proxy. If the proxy server automatically creates and renews subscriptions when the handset registers, then only the PROXYn _MAIL_NOTIFY contact need be specified. If the handset must subscribe to a particular contact to get mail notification, then only the PROXYn_MAIL_SUBSCR contact needs to be specified.

Parameter	Value	Description	Notes
PROXYn_MAIL_ACCESS	name@xxx.xxx.xxx.xxx or sip:name@domain	Contact to whom the handset should invite to access the mail center. This is the main voicemail dial number.	n = 1, 2, 3 name = mail server contact name. xxx.xxx.xxx.xxx = IP address Domain where the mail resides.
PROXYn _MAIL_NOTIFY	name@xxx.xxx.xxx.xxx or sip:name@domain	Contact from whom mail notification originates. See config file example for more information.	n = 1, 2, 3 name = mail server contact name. xxx.xxx.xxx.xxx = IP address. Domain where the mail resides.
AUTH [see warning note below]	username; password	Credentials. In general credentials are needed for each registered line.[2].	username = The dial number or string that identifies the line appearance. Generally an extension or phone number. password = a secure password created by the system administer which enables a handset to register and/or function.
CODECS	codec1, codec2	Comma-separated list of supported codecs in order of preference.	Defaults to “g711u, g711a”. if either is omitted it will be added to the end of the list.
LINEn	username	The dial # or name.	n = 1, 2, 3, 4, 5 The registered contact becomes: sip:username@domain
LINEn _PROXY	i	SIP proxy server for this line.	n = 1, 2, 3, 4, 5 i = the number of the proxy server 1, 2, 3.
LINEn _CALLID	callerid	String that displays at the far end.	n = 1, 2, 3, 4, 5 callerid = the text that will display as the caller ID on the called handset.

---

[2] WARNING: providing credentials by using the AUTH parameters in the configuration files is a security risk and should be avoided by entering usernames and passwords in admin menu or by allowing the user to login at startup time. Credentials entered here are in plain text and accessible by anyone who can access the TFTP server files. Credentials stored in the SIP server or in the handsets are protected.

Parameter	Value	Description	Notes
FAVORITE	Dialstring; identifier	Phonebook list of numbers accessible from the Favorites menu. Favorites in the allusers file will be present in the Favorites on all handsets. See config file example for more information.	Up to 8 entries permitted which may be divided between generic and phone specific files. Dialstring = complete SIP URI or local extension # Identifier = name. If omitted, the dial string appears on Favorites menu.

Sample Configuration Files

Configuration files are illustrated below. These files are available as a download from the software updates site and may be customized for your application. Please note that these are merely samples and will not work on your system as written here or as downloaded. Your configuration files must be locally programmed according to your site requirements.

SIP_allusers.cfg

```
# SIP ALL USERS Configuration file example

# Configuration file format example with explanatory text

# Codec preference order only.  This does not enable/disable codecs
# (Optional)
# can be G.711-ulaw, g.711u, G.711U, g711u, G.711U, etc.
# if g711u is omitted it will be added to end of list.
# if g711a is omitted it will be added to end of list after u.
CODECS = g711u, g711a

# One PROXYn (PBX/Call Server) is required, additional ones are optional as
# you can register secondary line appearances with other PROXY servers
PROXY1_ADDR     = 10.0.0.138:5060
#PROXY2_ADDR    = 172.29.0.140:5060

#ProxyDomain can be omitted if a specific proxy domain name is not defined at
the
# proxy server. If omitted, the ProxyDomain defaults to the IP address of the
# proxy server.
# (below are examples of different ways to specify a domain)
#PROXY1_DOMAIN = plcmengr.com
#PROXY1_DOMAIN = 10.0.0.138
#PROXY1_DOMAIN = axlx.engr.local

# PROXY1_MAIL_SUBSCR is who we should subscribe to for mail center
# notifications
# This is needed only if the user is not subscribed automatically at
# registration.
# It is almost never required in current versions of Asterisk to specify
# this.
# If you are using Asterisk (non-business edition) before v1.2, this is
# necessary.
# This example is actually specific for a line number:3001
#PROXY1_MAIL_SUBSCR = sip:3001@vmail.asterisk.com
```

```
# PROXYn_MAIL_NOTIFY is from whom we might get unsolicited mail center
# notifications
# This option is deprecated and no longer needed in versions beyond
# e/h340/i640 phones v108.011, Polycom 8002 phones v130.001, and Polycom
# 8020/8030 phones 131.001.
# Examples:
#PROXY1_MAIL_NOTIFY = asterisk@10.0.0.138
#PROXY1_MAIL_NOTIFY = sip:asterisk@10.0.0.138

# PROXY1_MAIL_ACCESS is the main voicemail dial number
# Examples:
# PROXY1_MAIL_ACCESS = 7999
# PROXY1_MAIL_ACCESS = sip:7999@10.0.0.138
# PROXY1_MAIL_ACCESS = 7999@10.0.0.138
PROXY1_MAIL_ACCESS = 7999

#PROXY1_KEYPRESS_2833 controls generation of in-stream RFC2833 formatted key
# press events. Normally you want this to be disabled for Asterisk but it
# depends on your configuration and what you want to be able to do.
# If you are going to do OAI integration, this must be disabled.
# The default is disable
PROXY1_KEYPRESS_2833 = disable

#PROXY1_KEYPRESS_INFO controls generation of SIP INFO requests to the SIP
# server for keypress events.  Normally you want this to be enabled.
# If you are going to do OAI integration, this must be enabled.
# The default is enable
PROXY1_KEYPRESS_INFO = enable

# PROXYn_HOLD_IP0 controls setting of media stream IP destination to 0.0.0.0
# when a call is put on hold.
# PROXYn_HOLD_IP0 is not required for current versions of Asterisk.
# For older PBXs that require this, set this to enable
#PROXYn_HOLD_IP0 = enable

# PROXY1_PRACK enables ACK'd provisional responses to INVITE requests.  The
# PRACK mechanism will be used if this switch is enabled and the Proxy server
# specifies support for the PRACK mechanism.  PRACK is NOT SUPPORTED in
# current versions of Asterisk, but is to be supported on subsequent
# versions.  PRACK should not be required on local area networks.
PROXY1_PRACK = disable

# PROXYn_KEEPALIVE_SECS specifies that the phone should send Keep alives to
#  the PROXYn server.  It can be set to 0 or anywhere from 10 to 3600
#  seconds. The default value (which is applied if no parameter is supplied
#  in this file) is 0 which means the phone will not send keep_alives to
#  PROXYn.
#  If the phone fails to get a keepalive response within the SIP 32 second
#  timeout, the keepalives are terminated for that PROXY until the next
#  successful registration to the PROXY.
#
#  This parameter is REQUIRED on Asterisk Business Edition with the Polycom
#  8002 phone but not with the Polycom 8020/8030 phones.  This is because
#  Linux, the OS running Asterisk, has a ARP Cache flushing mechanism.  This
#  feature serves to refresh the ARP Cache for the 8002 phones.  For 8020 and
#  8030 phones, the ARP Cache is refreshed by the SVP server.  8020/8030
#  phones do not know how to parse this parameter.
#
#  The value of this parameter for 8002 phones should be just less than the
#  minimum ARP Cache flushing time on the Linux box running Asterisk.  For
#  Redhat this minimum flushing time is 15 seconds, so the parameter needs to
#  be set for 14 seconds or less.  For other Linux Distributions, you may
#  have to determine what this minimum value is, however 14 seconds should
#  work fine.
#
#  The lower this setting, the more the impact on the 8002's standby battery
#  life.(8002 phones running version 130.003 and earlier code ignore this
#  parameter if present)
#  (8020/8030 phones ignore this parameter if present no matter the SW
#  version)
PROXY1_KEEPALIVE_SECS=14

# PROXYn_REREG_SECS specifies the re-registration interval for a PROXY.
#  This is the requested expiration interval sent in the REGISTER request
#  message.  It can be set to anything between 35 and 3600 seconds.
#  The default value (which is applied if no parameter is supplied in this
#  file) is 3600 which means the phone will attempt to register every 3580
#  seconds, if the server's response interval doesn't take precedence (see
#  below).
#
#  If the server has a lower maximum setting or a higher minimum setting that
#  this value then the server's response will take precedence and this
#  parameter will be ignored.
#
#  The phone will always attempt to re-register with the PROXY 20 seconds
#  prior to the expiration interval.  Thus, if this parameter is set to 35
#  seconds, and the server's response interval doesn't take precedence, then
#  the phone will attempt to re-register every 15 seconds.
#
#  The lower this setting, the more the impact on the 8002's standby battery
#  life.
#  (8002 phones running version 130.003 and earlier code ignore this
# parameter if present)
#(8020/8030 phones ignore this parameter if present no matter the SW version)
PROXY1_REREG_SECS=3600

# Favorites in the allusers file will be present in the favorites on all
# handsets
# The username can be blank and can include escaped chars
# Useful features can be included such as call forwarding or dialing
# voicemail
# FAVORITE = dial_string; username
FAVORITE = 1234; Site Security
FAVORITE = *98; Call Forwarding
```

Sample handset-specific file

```
# Configuration file format example

# Codec preference order only.  This does not enable/disable codecs.
# (Optional) can be G.711-ulaw, g.711u, G.711U, g711u, G.711U, etc.
# if g711u is omitted it will be added to end of list.
# if g711a is omitted it will be added to end of list after u.
#CODECS = g711u, g711a

# One PROXYn (PBX/Call Server) is required, additional ones are optional as
# you can register secondary line appearances with other PROXY servers
#
#PROXY1_ADDR     = 10.0.0.138:5060
#PROXY2_ADDR     = 172.29.0.140:5060

# ProxyDomain can be omitted if a specific proxy domain name is not defined
# at the proxy server. If omitted, the ProxyDomain defaults to the IP address
# of the proxy server.
# (below are examples of different ways to specify a domain)
#PROXY1_DOMAIN = plcmengr.com
#PROXY1_DOMAIN = 10.0.0.138
#PROXY1_DOMAIN = axlx.engr.local

# PROXY1_MAIL_SUBSCR is who we should subscribe to for mail center
# notifications
# This is needed only if the user is not subscribed automatically at
# registration.
# It is almost never required in current versions of Asterisk to specify
# this.
# If you are using Asterisk (non-business edition) before v1.2, this is
# necessary.
# This example is actually specific for a line number:3001
#PROXY1_MAIL_SUBSCR = sip:3001@vmail.asterisk.com


# PROXYn_MAIL_NOTIFY is from whom we might get unsolicited mail center
# notifications
# This option is deprecated and no longer needed in versions beyond
# 8002/e340/h340/i640 handsets v108.011, Polycom 8002 phones v130.001, and
# Polycom 8020/8030 phones 131.001.
# Examples:
#PROXY1_MAIL_NOTIFY = asterisk@10.0.0.138
#PROXY1_MAIL_NOTIFY = sip:asterisk@10.0.0.138

# PROXY1_MAIL_ACCESS is the main voicemail dial number
# Examples:
# PROXY1_MAIL_ACCESS = 7999
# PROXY1_MAIL_ACCESS = sip:7999@10.0.0.138
# PROXY1_MAIL_ACCESS = 7999@10.0.0.138

#PROXY1_KEYPRESS_2833 controls generation of in-stream RFC2833 formatted key
```

```
# press events. Normally you want this to be disabled for Asterisk but it
# depends on your configuration and what you want to be able to do.
# The default is disable
#PROXY1_KEYPRESS_2833 = disable

# PROXY1_KEYPRESS_INFO controls generation of SIP INFO requests to the SIP
# server for keypress events. Normally you want this to be enabled.
# The default is enable
#PROXY1_KEYPRESS_INFO = enable

# PROXYn_HOLD_IP0 controls setting of media stream IP destination to 0.0.0.0
# when a call is put on hold.
# PROXYn_HOLD_IP0 is not required for current versions of Asterisk.
# For older PBXs that require this, set this to enable
#PROXYn_HOLD_IP0 = enable

# PROXY1_PRACK enables ACK'd provisional responses to INVITE requests.  The
# PRACK mechanism will be used if this switch is enabled and the Proxy server
# specifies support for the PRACK mechanism.  PRACK is NOT SUPPORTED in
# current versions of Asterisk, but is to be supported on subsequent
# versions.  PRACK should not be required on local area networks.
#PROXY1_PRACK = disable


#  ////////////////////////
#  //////  ABOVE this line should probably be in the sip_allusers.cfg file
#  //////  with items uncommented in this file only for overriding a setting
#  //////  for a particular user.
#  ////////////////////////

# Authentication credentials
# (Normally not stored in this file for security reasons)
# AUTH = username; password
AUTH = 3001; 3001

# Line definitions
# Each definition should have LINEn, LINEn_PROXY and LINEn_CALLID
# LINEn is the dial number
# LINEn_PROXY is the PROXYn server this line should register with, typically
# defined in sip_allusers.cfg.
# LINEn_CALLID is shown on the standby display of 8020/8030 phones but not
# 8002 or e/h340/i640 phones.  The Asterisk Server converts the callID
# information to alternative forms defined in the Asterisk configuration
# files for display at the far end of a phone call.
# Up to 5 line definitions can be made for each user
# Line definitions do not necessarily have to have different extensions
LINE1         = 3001
LINE1_PROXY   = 1
LINE1_CALLID  = Mouse, Mickey

# Two lines may map to the same extension to allow second incomming calls.
LINE2         = 3001
LINE2_PROXY   = 1
```

```
LINE2_CALLID  = Brady, Marsha

LINE3         = 3002
LINE3_PROXY   = 1
LINE3_CALLID  = Drew, Nancy

#LINE4         = 804
#LINE4_PROXY   = 3
#LINE4_CALLID  = Sip User 4

#LINE5         = 1014
#LINE5_PROXY   = 2
#LINE5_CALLID  = Sip User 5

# Favorite Dialed Number list.
# You can define up to 8 total entries including any defined in
# sip_allusers.cfg.
# You can enclose a string in quotes to allow for spaces.
# Each favorite can be complete SIP URI
# Format is:
# FAVORITE = dial_string; username
#
# The username can be blank and can include escaped chars.
FAVORITE = 3001; Bob
FAVORITE = 3032; Jill in Accounting
FAVORITE = 3013; SoundPoint 3013
FAVORITE = 3020; Jane
FAVORITE = 93035551212; Richard's Cell
```

5

Using the SIP Handset

The Handset Display

Overview

When active, the handset screen has six rows that display information in this format:

Row #	Information provided
1	Icons
2	Line number
3	Instructions
4	Call one info
5	Call two info or dial aids
6	Softkeys

Row 1 displays signal, battery, voicemail and line number icons. Row 2 displays the line number of the line for which information is being displayed. Row 3 displays instructions where needed. Two calls are permitted on each line. Row 4 displays information for call one and row 5 displays information for call two. Row 5 is also used to provide dialing aid for call one. Row 6 displays available softkeys.

The SpectraLink 8002 Wireless Telephones support a two to four line display of up to 19 alphanumeric characters. Display information provided by the proxy server when the handset is off-hook, such as calling party identification, will be processed and put on the handset display.

Line indicators

The line indicators on the handset will convert to a solid or flashing number to indicate the line is in use or ringing. The number or text associated with the line is displayed on the handset by pressing the **LINE** key.

Handset icons

Indicator	Function
📶 📶 📶 📶 📶	The signal-strength icon indicates the strength of the signal from the AP and can assist the user in determining if the handset is moving out-of-range of that AP.
✉	The voicemail icon is activated when a new voicemail message is received if the feature is supported by the proxy server.
🔋 🔋 🔋 🔋	The battery icon indicates the amount of charge remaining in the Battery Pack. When only one bar remains, the Battery Pack should be charged.
Locked	Locked indicates that the keypad is locked to prevent accidental activation. Use the Unlk softkey plus the # key to unlock it.
[No Service message]	If warning tones are not disabled, a falling tone alarm will sound and a descriptive message displays when the handset cannot receive or place calls. You may be outside of the covered area. Walk back into the covered area. The in-service rising tone indicates service is reestablished.
⊞	The download icon indicates that the handset is downloading code. This icon only appears while the handset is running the over-the-air downloader. It appears to the right of the signal strength icon in the same location as the voicemail icon.

Signal strength

The signal-strength icon indicates the strength of the signal from the AP and can assist the user in determining if the handset is moving out of range. It is always present on the display in the upper left corner.

Battery charge

The battery icon indicates the amount of charge remaining in the Battery Pack. There are three levels and when only one level remains, the Battery Pack needs to be charged. It is always present on the display in the upper right corner.

Voicemail

The voicemail icon is activated whenever any of the specified mail subscribers/notifiers (see SIP configuration files) sends notification to the handset that new messages are available. By activating the **LINE** menu, and observing which lines are flagged with a plus (**+**) sign, the user can determine which proxy has outstanding mail.

The system administrator defines access to voicemail.

Download

Indicates that the handset is checking for or downloading code. This icon only appears while the handset is running the over-the-air downloader. It appears to the right of the signal-strength icon in the same location as the voicemail icon.

Ringing and tones

The ringing type (normal or vibrate) is selected by the handset user. The audible and vibrating ringer on the handset will follow the protocol command provided by the PBX via the proxy server if the user selects **PBX** cadence described in the *User-defined Preferences* section.

Handset modes

Standby mode (on-hook)

In standby mode, the handset is waiting for an incoming call or for the user to place an outgoing call. The extension number is shown on the display and there is no dial tone. In this mode, the handset is conserving battery power and wireless LAN bandwidth.

By default, the username assigned to line 1 is displayed on the standby screen.

Active mode (off-hook)

When an incoming call occurs, the handset will ring, and the handset enters the active mode. The handset will ring according to user preference as specified in the standby menus. The ringing can be silenced by pressing the **End Call** key. If you do not wish to accept the call, press the **Rej** softkey. The SIP server (if present) will redirect the call as configured by the system administrator, often to voicemail. (Treated like the handset is busy.)

The handset is in the active mode when an incoming call is answered by pressing the **Start** key or when it is in communication with the SIP server without being in an active call.

When an incoming call occurs during an active call, the handset will play the second call ringing sound until the call is answered, the caller hangs up, or the call transfers to voicemail. If **End Call** is pressed, the first call is terminated and the handset reverts to a full ring.

The active modes utilize the most bandwidth and battery power. To conserve these resources, return the handset to the standby mode when a call is completed by pressing the **End Call** key.

Messaging mode

If text messaging functions have been programmed, as in a nurse call system, the handset is able to receive text messages. While these messages are being accessed, the handset is in messaging mode. Incoming calls will ring with the second call ringing sound.

Messaging mode uses more battery power than standby mode, but less than Active mode.

Softkeys

Softkey functions change as the state of the handset changes. Softkeys are displayed only when they are available to be activated. Softkeys make common PBX functions available on the handset. For example, a call may be placed on hold, the handset may be muted, and a second line may be activated for an outgoing call or a second call on the same line may be answered. Calls may also be transferred to another number, either as a consulted transfer or without consult. Other handset features (DND, Transfer) may be accessed by pressing the **FCN** key.

Number dialing may be specified as alpha or numeric code by pressing **Start Call** plus the **MENU** key. Select **Alpha Dialing** if letters and symbols are to be entered in the dialing string.

Softkey	Name	Displayed…	Press to…
Answ	Answer	When a call is incoming on the selected line.	Answer the call (equivalent to START key).
Bksp	Backspace character	Entering a dial number.	Delete the character prior to the cursor position.
Back	Back one screen	Displaying a menu.	Exit the menu.
Del	Delete	Editing dial number.	Delete the character to the left of the cursor.
Dial	Dial Call	A dial number is being entered on the selected line.	Initiate a phone call to the entered dial number.
End	End Call	An active call on the selected line.	Terminate the call without going back to standby mode.
Favr	Favorites	Prior to entering the first character of a dial number.	Activate the Favorites menu.

Softkey	Name	Displayed…	Press to…
Fwd	Forward Definition	Prior to entering the first character of a dial number.	Delete a previously defined forward destination. Or initiate definition of a new forward destination.
Hold	Hold	In an active call.	Place the call on hold. The line status shows x when the call is on hold or * when audio is flowing.
Msg	Message	Initial dial screen when new line is selected and a dial tone is active prior to entering first character of the number to be dialed3.	Initiate a call to the specified message center contact address for retrieval or administration of voicemail.
Mute	Toggle muting	When in an active call.	Toggle audio transmission to the far end. The line status shows * when not muted or x when muted.
OK	OK	Power up registration if username is not configured in admin menu.	Send the username and password to the SIP server for authorization to register the handset.
Redl	Redial	Prior to entering the first character of a dial number.	Redial the last number that was dialed.
Rej	Reject	Incoming call on the selected line.	Reject the incoming call. The SIP server will then redirect the call elsewhere if configured to do so.
Rsme	Resume	In an active call and you have placed the call on hold or in standby mode if any call is on hold.	Resume a call that was previously placed on hold or that went on hold when another line was activated.
Save	Save	Entering a dial number as a forward destination.	Save the dial number as the forwarding destination for the selected line.

---

3 Appears only if the PROXYn_MAIL_NOTIFY is configured. A message center contact address must be defined for the proxy used by the selected line.

Menus

Line menu

The Line menu allows you to activate a call on a selected line or to view the status of lines.

View the five line appearances by pressing the **LINE** key. The line number and extension number of the first four lines are displayed. Press the **LINE** key again to view the fifth line. Active lines are in bold face. Defined but unregistered lines are faded. Lines that have been forwarded are prefixed with the greater-than character (**>**) followed by the forwarding destination.

Pressing the **LINE** key from the active mode displays a menu of line appearances as programmed in the SIP TFTP configuration file. The **LINE** key can be pressed while the handset is in the standby mode to activate the handset and to activate a new call on the selected line.

The currently selected line is indicated by an asterisk (*****). Lines for which the corresponding proxy server has outstanding new mail are flagged with plus (**+**) characters. Lines that should be registered to a proxy but have failed registration for any reason are displayed in faded text and are not selectable from the menu.

Exit the **LINE** display by pressing a line number key to start a new call on the selected line and put any other call on hold, or by pressing the **End Call** key to exit without starting a new call. Press the **More** softkey to page through additional items on the Line menu.

Favorites menu

The Favorites menu provides access to a predefined list of dial numbers. The list is programmed by the system administrator and may include common or handset-specific entries. For handset-specific entries, the user must provide a list of numbers to the system administrator for programming. Numbers on the Favorites list cannot be programmed in or through the handset by the end user.

The Favorites list can include either complete dial numbers for named parties or partial numbers that need additional data entry. Here is a partial entry example: If a PBX feature access code for call forwarding is defined in the Favorites list but you need to add the forwarding destination information before sending the call to the PBX to activate the feature, you can create a Favorite with the access code only.

Access the Favorites list while in a dialing state by pressing the **Favr** softkey to display a menu of pre-defined numbers or names that can be dialed (as programmed in the SIP TFTP configuration file.) When an item is selected from the list, the dial number is displayed. You may edit or add digits to the displayed number if necessary before pressing **START** to place the call. When using the Favorites menu to perform a blind transfer to someone on the list, select an entry from the Favorites menu and then press **FCN** and select **Transfer**.

FCN menu

Pressing the **FCN** key from the active mode displays a list of telephony options available to the user. Included here are transfer and do-not-disturb functions.

The FCN menu is accessible while in the active mode and provides these features:

Transfer

Do Not Disturb

Set/Clear Forward

[defined option 1]

[defined option 2]

[defined option 3]

OAI functions are automatically added as items at the end of this menu if defined on an OAI server.

Items on this menu are accessible through navigation and selection keys or through short-cut keys as displayed with the menu items.

MENU display

Pressing the **MENU** key from the active mode allows you to set the dialing mode to either numeric only or alphanumeric entry.

Handset Operation

Alphanumeric dialing

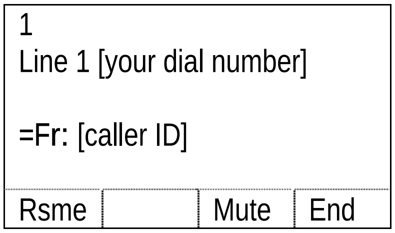
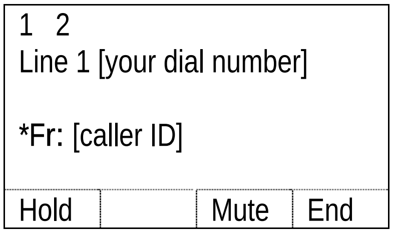
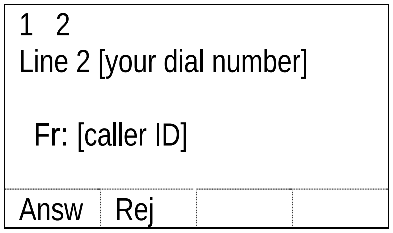
During dialing the carat (**^**) indicates the insertion point for the next character. Up to 80 characters may be entered with automatic scrolling. While in alphanumeric mode, alphabetic characters can be entered by repeatedly pressing the 1 through 9 keys. Punctuation characters are available by repeatedly pressing the asterisk (*****), zero (**0**) or pound (**#**) keys. Row 5 displays the available characters as the key is pressed. Pausing will enter the highlighted character in the dial string. A space character is available on the pound key or by pressing the **>>** softkey when the insertion point is at the end of the dial string. The **<<** and **>>** softkeys may be used to position the insertion point for editing the entry. Press the **Dele** softkey to delete the character prior to the insertion point.

Feature activation

If you want to…	Then…
Turn the handset on	Press and hold the **END** key until two chirps sound.
Turn the handset off	Press and hold the **END** key. One chirp will sound. If you are in a call, hang up first, then turn off the handset.
Unlock the keypad	Press the **Unlk** softkey, then **#**.
Lock the keypad	While in standby mode press **FCN**, then **Select**.

If you want to…	Then…
Place a call	1 Press the **START** key and wait for a dial tone. 2 Dial the numbers and press **START**. Listen for the ring to indicate the alerting of the called party. Note: Line 1 is the default line Example: 1 Line 1 [your dial number] *To: [dialed number] Hold | Rej | | Mute
Place a second call on the same line	Only outgoing calls are permitted as the second call on a given line. If there is an incoming call to the number assigned to the line, the caller will get a busy tone and be routed to voicemail or to the second appearance of that call number, if defined. 1 To access the second call dial tone, press **LINE** + [the line number]. 2 Dial the number for the second call. 3 Press **START** to place the call. 4 When a second call is active on the same line appearance, the second call appears below the first on the handset display. 5 Use **LINE** + [the line number] to toggle between calls. The selected call is in bold type the other call is faded. Example: The following example shows an active incoming call as call one on line 1 (Fr or From) and an outgoing call on hold as call two on line 1 (To). The bold text on row 4 indicates that call one is the currently selected call and that any softkeys or menu functions will apply to call one. The asterisk (*****) on row 4 indicates that audio is currently flowing for call one. 1 Line 1 [your dial number] *Fr: [caller ID of caller] =To: [number you dialed] Hold | | Mute | End
Place a call on a different line	1 Press **START** and listen for dial tone and then press **LINE**. or to predial the number just press the **LINE** key. 2 Navigate to desired line and press **Select**. 3 Dial the number. 4 Press the **Dial** softkey or **START** to initiate the call.

If you want to…	**Then…**
Place a call from Favorites menu	1 Press **START**. 2 Listen for dial tone. 3 Press **Favr** softkey. 4 Use the side buttons to find and select desired entry. 5 Edit entry if needed. 6 Press **START** to place the call.
Answer/Reject a call	When the handset rings, the line's icon will flash and row 2 will display the line number and your dial number. Row 4 will display Fr (From) and the caller ID of the incoming call. 1 Press **START** or the **Answ** softkey to answer the call. or 2 Press **Rej** to reject the call and allow the SIP server to redirect the call elsewhere, usually to voicemail. Example: 1 Line 1 [your dial number] Fr: [caller ID] Answ Rej When the call is answered an asterisk (*) displays at the start of row 4 which indicates that this line is in an active call with established audio connection. The softkeys for **Hold**, **End** and **Mute** appear. Example: 1 Line 1 [your dial number] *Fr: [caller ID] Hold Mute End

If you want to…	**Then…**
Place a call on hold	Press the **Hold** softkey. When a call is placed on hold, an equal character (=) replaces the asterisk (*) and the **Rsme** softkey replaces the **Hold** softkey. Press **Rsme** to return to the call. Example:  1 Line 1 [your dial number] =Fr: [caller ID] Rsme Mute End
Mute/Unmute a call	Press the **Mute** softkey. When the handset is muted, an x character replaces the asterisk (*). Press the **Mute** softkey again to restore audio pickup.
Silence the ring	Press the **END** key to silence the loud ring and convert it to in-ear ringing. Loud ringing will resume when the next incoming call is received while the handset is in standby mode.
Answer a call on a second line	If another call comes in on a different line, the called line icon flashes and a tone sounds in the audio stream. In this example, a call is coming in on line 2 and the line 2 icon is flashing. Example:  1 2 Line 1 [your dial number] *Fr: [caller ID] Hold Mute End 1 To view the caller ID of the incoming call without interrupting the active call, press **LINE** + [the line number]. The display now shows information about the incoming call and you will continue hearing audio on the first call. Example:  1 2 Line 2 [your dial number] Fr: [caller ID] Answ Rej 2 Press **Answ** or **START** to answer the incoming call and place the active call on the other line on hold. Pressing the **Rej** softkey will redirect the incoming call to voicemail or as otherwise programmed.

If you want to…	Then…
Transfer a call (blind)	1 While in a call, press **FCN** and then select **Transfer** (or press the shortcut key **1**). (The current call will be placed on hold]. A new call will be started and you will hear a dial tone.) 2 Press the digits for the number to which you wish to transfer the call or press the **Favr** softkey and select an entry from the Favorites menu.. 3 Press **FCN 1** again. The call will be transferred to the number that you have entered and your call will be disconnected.
Transfer a call (consulted)	1 While in a call, press **FCN** and then select **Transfer** (or press the shortcut key **1**). (The current call will be placed on hold]. A new call will be started and you will hear a dial tone.) 2 Press the digits for the number to which you wish to transfer the call or press the **Favr** softkey and select an entry from the Favorites menu.. 3 Press **Dial** to establish a second call. 4 When the party answers, have your conversation and then press **FCN 1** to transfer the held call and disconnect the first call and return to standby mode.
Transfer a call to another call already on Hold	1 While in a call, press **FCN** and then select **Transfer** (or press the shortcut key **1**). The current call will be placed on hold and you can converse with the party on the second line. 2 Have your conversation and then press **FCN 1** to transfer the held call and disconnect the first call and return to standby mode.
Forward all calls	Calls may be forwarded on a per-line basis. The **LINE** display will indicate if a line is forwarded by a > character followed by the destination address. If any line is forwarded, the standby screen will display forwarding status. If the do-not-disturb feature is enabled, the forwarding status will be pre-empted by the **Do Not Disturb** display. 1 Press **LINE** + [the line number]. 2 Press the **Fwd** softkey and enter the forwarding destination. 3 Press the **Save** softkey to save the forwarding destination. If a given contact number has multiple appearances on the handset, forward the lowest line appearance. All calls to that number on any line appearance will then be forwarded.
Clear forwarding on a line	1 Press **LINE** + [the line number], 2 Press the **Fwd** softkey and enter the forwarding destination. 3 Press the **Fwd** softkey again.

If you want to…	Then…
Listen to voicemail	When any proxy has outstanding messages, the voicemail icon will display. The **LINE** menu will display a plus (+) indication when there is a new voicemail for that line. Indications are cleared when the handset receives notification that all new voicemail messages have been read. 1 Press **START** and dial the number assigned to message retrieval and press **START** again. 2 Follow system instructions to retrieve messages. Alternately, if a PROXYn_MAIL_ACCESS value has been configured for the corresponding proxy, the voicemail system can also be accessed by pressing the **Msg** softkey, which will appear after selecting a line and before beginning to dial.
Do Not Disturb (reject all incoming calls)	Press **FCN 2**. The handset will not ring for new incoming calls. When the do-not-disturb feature is enabled, the standby screen will display **Do Not Disturb**.
Clear Do Not Disturb	Press **FCN 2** again to resume ringing for new incoming calls.
Redial the last number you dialed	1 Press **START**. 2 Press **Redl**. 3 Verify that the displayed number is the one you want and the line # is valid. 4 Press **START**.
Activate a registered OAI function on an OAI server	1 Press **START**. 2 Press **FCN**. 3 Navigate to desired OAI function using **Up** and **Down** buttons. 4 Select the OAI function using the **Select** button or shortcut key.
Silence while ringing	Press the **END** key to silence the ring.
Adjust the headset volume	While the headset is plugged in, press the up/down volume buttons on the side of the handset during the call.
End the call	Press the **END** key to return to the standby mode.
Turn on the backlight	The backlight comes on when any key is pressed or when there is an incoming call, and stays on for 10 seconds. It turns off if another key is not pressed for 10 seconds.

6

Testing a Handset

Verify proper registration and operation of each handset by performing the following tests on each handset in an active wireless area.

1. Power on the handset by pressing the **END** key. A series of messages will be displayed as the handset acquires the system. The handset should display the user extension.
2. Place a call and listen to the audio quality. End the call by pressing the **END** key.
3. Place a call to the handset and verify ring, answer, clear transmit, and clear receive audio.
4. Use the softkeys to verify all softkey programmed features on the handset.
5. Press the **END** key. Any line indicators should turn off and the extension number display will return.

If any of these steps fails to operate as described, refer to Chapter 10 *Troubleshooting* for corrective action.

7

Diagnostic Tools

Run Site Survey, **Diagnostics Mode** and **Syslog Mode** are three diagnostic tools provided to assist the WLAN administrator in evaluating the functioning of the handset and the system surrounding it. Diagnostic tools are enabled in the Admin menu.

Site Survey

Site survey can be used to evaluate the facility coverage before certifying that an installation is complete. It can also be used at any time to evaluate coverage by testing signal strength, to gain information about an AP, and to scan an area to look for all APs regardless of ESSID. The information available through the site survey includes:

- ESSID
- Beacon Interval
- Information regarding support of 802.11d, 802.11g, 802.11h, and other 802.11 amendment standards as required.
- Current security configuration

Start the site survey by selecting **Run Site Survey** from the Admin menu. The mode starts immediately.

When the site survey is started, it is by default in “single ESSID” mode. When the **Any** softkey is pressed (softkey A), all APs regardless of ESSID are displayed, and the softkey changes to say **MyID**. Pressing the **MyID** softkey will revert to the “single ESSID” mode and change the softkey back to **Any**.

The display has the following layout for "single ESSID" mode.

```

1 1 1 1 1 1   - 2 2   3 3   4 4 4
1 1 1 1 1 1   - 2 2   3 3   4 4 4
1 1 1 1 1 1   - 2 2   3 3   4 4 4
1 1 1 1 1 1   - 2 2   3 3   4 4 4
A n y                       D e t l
```

Where:

- 111111 = The last three octets of the on-air MAC address for a discovered AP.
- 22 = The signal strength for the specified AP.
- 33 = The channel number of the specified AP.
- 444 = The beacon interval configured on the specified AP.
- Any/MyID = Softkey to toggle between "single SSID" and "any SSID" mode.
- Detl/Smry = Softkey to toggle between the multiple AP (summary) display, and the single (detail) displays for each AP.

The following screen shows how the display would look when there are three APs configured with an ESSID that matches that of the handset. The first has a signal strength of –28 dBm, is configured on channel 2, with a beacon interval of 100 ms. The second has a signal strength of –48 dBm, is configured on channel 6, with a beacon interval of 200 ms. The third has a signal strength of –56 dBm, is configured on channel 11 with a beacon interval of 100 ms.

```

a b 7 b c 8   - 2 8   0 2   1 0 0
2 a e 5 7 8   - 4 8   0 6   2 0 0
2 a e 5 9 6   - 5 6   1 1   1 0 0

A n y                       D e t l
```

When the "any ESSID" mode is selected, the summary display contains the first six characters of the AP's ESSID instead of the beacon interval as in the example below. Also, the number of octets of the APs BSSID is reduced to 2 octets.

```
ab7b  -28  02  ALPHA
2ae5  -48  06  WSMTES
2ae5  -56  11  voice

MyID             Detl
```

In detail mode the display would appear as follows. The left/right arrow keys (* and #) will move between AP indices.

```
i: bbbbbb  sn  ch  bcn
eeeeeeeeeee  DGHI
rrrrrrrrrrrrr+xxxx
mmm  G:gggg  P:pppp
Any               Smry
```

Where:

- i = index of selected AP (value will be from 0 to 3)
- bbbbbb = the last three octets of the BSSID for a discovered AP
- sn = signal strength in –dBm
- ch = channel
- bcn = beacon interval
- eeeeeeeeeee = SSID (up to first 11 characters)
- DGHI = 802.11 standards supported
- rrrrrrrr = rates supported. Basic rates will have a “b” following the rate
- + = more rates are supported than those displayed
- xxxx = WMM or UPSD if those QoS methods are supported
- mmm = security mode
- G:gggg = group key security
- P:pppp = pairwise key security
- Any/MyID = softkey to toggle between “single SSID” and “any SSID” modes
- Detl/Smry = softkey to toggle between the multiple AP display (summary), and the single AP display (detail)

Numbers scrolling across the handset display indicate AP information is being obtained. A **Waiting** message indicates the system is not configured properly and the handset cannot find any APs.

Solving coverage issues

Coverage issues are best resolved by adding and/or relocating APs.

Overlap issues may be resolved by reassigning channels to the APs or by relocating the APs. See Chapter 10 *Troubleshooting* under *Access Point Problems* for more information.

Diagnostics Mode

The **Diagnostics Mode** is used to evaluate the overall quality of the link between the handset, AP, and infrastructure side equipment, such as proxy server. Unlike site survey, the **Diagnostics Mode** is used while the functional code is running, and during a call.

When **Diagnostics Mode** is turned on in the Admin menu, the handset can display diagnostic screens any time it is active (in a call).

Start the display of diagnostic information by pressing the **MENU** key. Pressing the **MENU** key multiple times will cycle through the various screens and the normal off-hook (proxy server) display. The numeric icon at the top of the display indicates what screen number is being displayed. For example, the first time the **MENU** key is pressed, the **1** icon is shown, and the first set of counters is displayed. The next time it is pressed, the **2** icon is shown, and the next set of counters is displayed. The screens will be cycled through in this fashion until there are no more screens to be displayed, after which the screen returns to the normal off-hook proxy server screen.

Note that the normal use of the **MENU** key is not available if **Diagnostics Mode** is enabled.

The screens provided by the **Diagnostics Mode** include:

Screen 1

```
MissedRcvCnt   nnnnn
MissedXmtCnt   nnnnn
RxRetryCount   nnnnn
TxRetryCount   nnnnn
```

- Missed receive packet count since power up (MissedRcvCnt)
- Missed transmit packet count since power up (MissedXmtCnt)
- Receive retry count since power up (RxRetryCount)
- Transmit retry count since power up (TxRetryCount)

Screen 2

```
Jitter         nnnnn
LastRate       nnnnn
GatewyType     mnemo
```

- Jitter is the average error or “wobble” in received packet timing, in microseconds (jitter)
- Last successful transmit data rate (LastRate)
- Gateway type (not used)

Screen 3

```
C: mmmm ch -ss aid
1: mmmm ch -ss mnem
2: mmmm ch -ss mnem
3: mmmm ch -ss mnem
```

Screen 3 contains a list of the APs that are heard and the following parameters from each AP:

- Indicator as to whether this is the current AP, or an index into the list of other APs heard (C indicates current).
- Last 2 octets of the MAC address of the AP (mmmm)
- Channel number (ch)
- Signal strength (ss)
- Either the 802.11 Association ID (aid) from the current AP or a mnemonic (mnem) for the reason code indicating why the handset did not hand off to this other AP.

Where:

AP mnem: a mnemonic indicating the reason code:

- Unkn: reason unknown
- Weak: signal strength too weak
- Rate: one or more basic rates not supported
- Full: AP can not handle bandwidth requirements
- AthT: authentication timeout
- AscT: association timeout
- AthF: authentication failure
- AscF: association failure
- SecT: security handshake timeout
- SecF: security handshake failure
- Cnfg: AP not configured correctly for security, QoS mode or infrastructure network

Screen 4

```
AssocCount      nnnnn
ReAssocCount    nnnnn
AssocFailure    nnnnn
ReAssocFail     nnnnn
```

- Association count since power up (AssocCount)
- Re-association count since power up (ReAssocCount)

- Association failures since power up (AssocFailure)
- Re-association failures since power up (ReAssocFail)

Screen 5

```
Sec-ErrCount   nnnnn
LstSecErrSeq   nnnnn
```

- Security error count since power up (Sec-ErrCount)
- MAC sequence number of frame with last security error (LstSecErrSeq)

Syslog Mode

A syslog server must be present on the network in order for the handset to send the log messages and have them saved. The syslog server will be found with DHCP option 7 if the handset is using DHCP. If static addresses are configured, the syslog server's IP address can be configured statically in the Admin menu.

> If the syslog server address is blank (000.000.000.000 or 255.255.255.255), or the handset is using DHCP and no option 7 is received from the DHCP server, the handset will not send any syslog messages.

Admin menu options:

- ***Disabled** turns syslog off. (default)
- **Errors** causes the handset to log only events that are considered to be an error (see below).
- **Events** logs all errors plus also some other interesting events (see below).
- **Full** logs all the above plus a running stream of other quality information (see below).

The table below lists the syslog messages and which level of logging will produce them:

Message type	Errors	Events	Full
Failed Handoff	Yes	Yes	Yes
Successful Handoff	No	Yes	Yes
Security Error	Yes	Yes	Yes
Call Start/End	No	Yes	Yes
Audio stats	No	No	Yes (every 5 secs)
Audio error threshold exceeded	Yes	Yes	Yes
Radio stats	No	No	Yes (every 5 secs)
Radio error threshold exceeded	Yes	Yes	Yes

All syslog messages will include:

- Date and time (to 1/100th of second) since handset power on (currently set to Jan-1 00:00.00)
- Handset's MAC address
- Handset's IP address
- Syslog sequence number

The table below lists the additional items in each message type:

Failed Handoff (Sent whenever the handset attempted to handoff, but failed trying.)	Failed AP MAC Failed AP signal strength Current AP MAC Current AP signal strength Failure reason
Successful Handoff	New AP MAC New AP signal strength Old AP MAC Old AP signal strength Reason for handoff Other candidate APs: MAC Signal strength Reason not used
Security Error	AP MAC AP signal strength Security mode Error details (mode dependent)

Call Start	Call type (telephony, OAI, PTT) AP MAC AP signal strength
Call End	AP MAC AP signal strength
Audio stats	AP MAC AP signal strength Payload size (in msec) Payloads sent Payloads received Payloads missed (not received) Payloads missed rate (over last 5 seconds) Payloads late Payloads late rate (over last 5 seconds) Average jitter
Audio error threshold exceeded (Sent if payloads missed rate or payloads late rate exceeds 2%, or if the average jitter is over 2 msec)	Same as audio stats
Radio stats	AP MAC AP signal strength Directed packets sent Directed packets received Multicast packets sent Multicast packets received Broadcast packets sent Broadcast packets received TX dropped count TX drop rate (over last 5 seconds) TX retry count TX retry rate (over last 5 seconds) RX retry count RX retry rate (over last 5 seconds)
Radio error threshold exceeded (Sent if TX drop rate exceeds 2% or TX or RX retry rate exceeds 5%)	Same as radio stats

Messages are formatted like the following example:

```
Jan 1 00:01:26.72 0090.7a02.2a1b (172.16.0.46) [001a]
RStat: AP 00:40:96:48:1D:0C (-56 dBm), Sent 783523,
Recvd 791342, MSnt 245, MRcd 5674, BSnt 43, BRcd 10783,
TX drop 43 (0.0%), TX retry 578 (1.2%), RX retry 1217
(1.6%)
```

8

Certifying the Installation

Prior to determining that an installation is complete, test the handsets following the sequence given in Chapter 6 *Testing a Handset* and conduct a **Site Survey** mode test according to the directions given in Chapter 7 *Diagnostic Tools*.

The installation may need some adjustments. Note any areas where coverage is conflicting or inadequate. Note any system difficulties and work with your wireless LAN and/or LAN system administrator to determine the cause and possible remedy. See Chapter 10 *Troubleshooting* for clues to possible sources of difficulties. If any adjustments are made to the system, re-test the device in the same vicinity to determine if the difficulty is resolved.

The installer should not leave the site before performing installation verification.

These tests must be performed in typical operating conditions, especially if heavy loads occur. Testing sequence and procedure is different for every installation. Generally, you should organize the test according to area and volume, placing numerous calls to others who can listen while you perform coverage tests. Note any areas with excessive static or clarity problems and report it to a Polycom service engineer.

The coverage test will also require you to put the handset in **Site Survey** mode and walk the entire coverage area to verify all APs.

Conducting a site survey

Conduct a site survey of the installation, by walking the site looking for interfering 802.11 systems, adequate coverage and channel assignment, and correct AP configuration.

1. Referring to *Run Site Survey*, put a handset into **Site Survey** in the **Any/Smry** ESSID mode. Walk throughout the site checking for any expected APs or other ESSIDs.

2. Then, walk the site again, in **MyID/Smry** ESSID mode, this time checking that every location has adequate coverage (there should

be at least one AP stronger than -60 dBm in all areas for 11mb/s operation) and has good channel allocation. (At any point, the strongest AP shown should be on a different channel than the next best choice.)

3. Finally, use the single AP (**MyID**/**Detl**) display to check each AP, to ensure it is configured for the proper data rates, beacon interval, 802.11 options enabled, QoS method, and security method.

Make any necessary adjustments to AP locations and configurations and repeat steps 1 through 3 until the site survey shows adequate coverage and correct configuration at every location.

The installation is not complete until these certification steps have been performed. Do not hand out handsets at a site that has not been certified.

9

Software Maintenance

SpectraLink 8002 Wireless Telephones use proprietary software programs written and maintained by Polycom Corporation. The software versions that are running on the handsets can be displayed during power on by holding down the **END** key.

Polycom Customer Service or an authorized dealer will provide information about software updates and how to obtain the software (for example, downloading from a website).

Upgrading handsets

After software updates are obtained from Polycom, they must be transferred to the appropriate location in the LAN to update the code used by the handsets.

SpectraLink 8002 Wireless Telephones allow over-the-air transfer of software updates from the designated TFTP server to the handsets. The downloader function in the handset checks its software version every time the handset is turned on. If there is any discrepancy the handset immediately begins to download the update.

Normal download messages

When the handset is powered on, it displays a series of messages indicating that it is searching for new software, checking the versions, and downloading. The normal message progression is:

Message	Description
Checking Code	Handset is contacting the TFTP server to determine if it has a newer version of software that should be downloaded.
Erasing Memory	Handset has determined that a download should occur and is erasing the current software from memory. This message also displays a progress bar. When the progress bar fills the display line the erase operation is complete.

Message	Description
Updating Code	Handset is downloading new software into memory. The number icons at the bottom of the display indicate which file number is currently being downloaded. This message also displays a progress bar. When the progress bar fills the display line the update operation is complete on that file.

When the update is complete, the handset displays the extension number, and is ready for use.

Download failure or recovery messages

The following display messages indicate a failure or recovery situation during the download process.

Message	Description
Server Busy	Handset is attempting to download from a TFTP server that is busy downloading other phones and refusing additional downloads. The handset will automatically retry the download every few seconds.
TFTP ERROR(x):yy	A failure has occurred during the TFTP download of one of the files. (x) = The file number which was being downloaded; yy is an error code describing the particular failure. Possible error codes are: 01 = TFTP server did not find the requested file. 02 = Access violation (reported from TFTP server). 07 = TFTP server reported "No such user" error. Check the TFTP server configuration. 16 = No TFTP server address. Check the TFTP server IP configuration in the handset or in the DHCP server. 81 = File put into memory did not pass CRC. The handset will attempt to download the file again. FF = Timeout error. TFTP server did not respond within a specified period of time.
Erase Failed	Download process failed to erase the memory in the handset. This operation will retry.
Waiting	Handset has attempted some operation several times and failed, and is now waiting for a period of time before attempting that operation again. This waiting period preserves Battery Pack life.

10

Troubleshooting

On occasion, you may run into transmission problems due to any number of factors originating from the wireless LAN. SpectraLink 8002 Wireless Telephones can exhibit transmission problems in several ways. They can cease functioning properly, display error messages, or display incorrect data. When using and troubleshooting handsets, consider the following problem sources to determine the best method of approaching any specific situation. In additional to the information in this document, you may refer to the knowledge base at http://knowledgebase.polycom.com.

The most up-to-date support information can be found in the knowledge base at http://knowledgebase.polycom.com.

Access point problems

Most, but not all, handset audio problems have to do with AP range, positioning, and capacity. Performing a site survey as described in this document can isolate the AP causing these types of problems. If the handset itself is suspected, conduct a parallel site survey with a handset that is known to be properly functioning.

In range/out-of-range

Service will be disrupted if a user moves outside the area covered by the wireless LAN APs. Service is restored if the user moves back within range. If a call drops because a user moves out-of-range, the handset may be able to recover the call if the user moves back into range within a few seconds.

Capacity

Basic WMM does not provide the capability of regulating capacity on an AP. If capacity issues are a concern, please consult your service representative.

Transmission obstructions

Prior to system installation, the best location for APs for optimum transmission coverage should have been determined. However, small pockets of obstruction may still be present, or obstructions may be introduced into the facility after system installation. This loss of service can be restored by moving out of the obstructed area or by adding/rearranging APs.

Configuration problems

Certain problems are associated with improper configuration of either the SIP system or the handset.

Configuration problems are generally corrected by changing the configuration on the SIP system or on the handset. There may also be incorrect programming of the AP. See the manufacturer's instructions for the AP in use at the site.

Handset Status Messages

SpectraLink Wireless Telephone status messages provide information about the SpectraLink 8002 Wireless Telephone's communication with the AP and host telephone system. The following table summarizes, in alphabetical order, the status messages.

Message	Description	Action
Address Mismatch	Handset software download files are incorrect or corrupted.	Download new software from the SpectraLink website per *Software Maintenance*.
ASSERT xxx c Line yyy	The handset has detected a fault from which it cannot recover.	Record the error code so it can be reported. Turn the handset off then on again. If error persists, try registering a different handset to this telephone port. If error still persists, contact Polycom Technical Support and report the error.
Assoc Failed xxxxxxxxxxxx	x…x = AP MAC address. Handset association was refused by AP; displays MAC of failing AP.	Check handset and AP security settings. Ensure AP is configured per Configuration Note. Try another AP.
Assoc Timeout xxxxxxxxxxxx	x…x = AP MAC address. Handset did not receive association response from AP; displays MAC of failing AP.	Check handset and AP security settings. Ensure AP is configured per Configuration Note. Try another AP.
Auth Failed xxxxxxxxxxxx	x…x = AP MAC address. Handset authentication was refused by AP; displays MAC of failing AP.	Check handset and AP security settings. Ensure AP is configured per Configuration Note. Try another AP.
Auth Timeout xxxxxxxxxxxx	x…x = AP MAC address. Handset did not receive authentication response from AP; displays MAC of failing AP.	Check handset and AP security settings. Ensure AP is configured per Configuration Note. Try another AP.
Bad Code Type xx Expected Code Type yy	xx, yy = software license types. Handset software does not match current handset license selection.	Check current handset license selection, slnk.cfg.cfg file contents. If necessary obtain new software per *Software Maintenance*.
Bad Config	Some needed configuration parameter has not been set.	Check all required handset configuration parameters for valid settings.
Bad ESSID	The handset is configured for "static ESSID" (as opposed to "Learn once" or "Learn always"), and no ESS ID has been entered.	Enter an ESSID in the configuration settings or change to one of the "Learn" modes.

Message	Description	Action
Bad Network IP	The value of the Network IP address entered in the WT through the menus or the configuration is missing or invalid	Enter a valid network IP address
Bad Network Mask	The value of the Network network mask entered in the WT through the menus or the configuration is missing or invalid	Enter a valid network mask
Bad Phintl File	Handset software download files are incorrect or corrupted.	Download new software from the SpectraLink website per *Software Maintenance*.
Bad Program File	Handset software download files are incorrect or corrupted.	Download new software from the SpectraLink website per *Software Maintenance*.
(battery icon), Battery Low, beep (audio)	Low battery.	In call: the battery icon displays and a soft beep will be heard when the user is on the handset and the battery charge is low. User has 15–30 minutes of battery life left. The Battery Pack can be changed while the call is still in progress. Do not press End Call. Place call on Hold or Park. Quickly remove the discharged battery and replace with a charged battery, power on the handset, and press Start Call to resume the call in progress. Not in call: The battery icon displays whenever the battery charge is low The message Battery Low and a beep indicate a critically low battery charge when user is not on the handset. The handset will not work until the Battery Pack is charged.
Battery Failure	The Battery Pack is not functioning.	Replace the Battery Pack with a new or confirmed SpectraLink Battery Pack. Only SpectraLink Battery Packs will work.
Battery Failed	Battery Pack is damaged or incompatible with handset.	Replace the Battery Pack with a new or confirmed SpectraLink Battery Pack. Only SpectraLink Battery Packs will work.
Can't Renew DHCP yyy.yyy.yyy.yyy	y…y = DHCP server IP address. DHCP server is not responding to initial renewal attempt.	Configuration problem. Check the IP address configuration in the DHCP server.
Charging …	The handset is charging in the desktop charger.	No action needed.
Charge Complete	The handset is now fully charged.	No action needed.
Checking Code	Handset is contacting the TFTP server to determine if it has a newer version of software that should be downloaded.	None, this message should only last for approximately one second. If message remains displayed, power off and contact customer support for a replacement phone.

Message	Description	Action
Checking DHCP IP	The handset is retrieving DHCP information from the DHCP server.	None. This is informational only.
CRC Code Error	The software which has been TFTP downloaded has a bad redundancy code check.	Try the download again; it is possible the software was corrupted during download. If the error repeats, check that the download image on the TFTP server is not corrupted.
Code Mismatch!	The software loaded into the handset is incorrect for this model handset.	Verify the License option value is correct. Replace the software image on the TFTP server with software that is correct for the handset model.
DCA Timeout	The handset has detected a fault for which it cannot recover, possibly due to a failure to acquire any network.	Turn the handset off, then on again. If error persists, contact Polycom Technical Support and report the error.
DHCP Error (1-5)	DHCP Error 1.	The handset cannot locate a DHCP server. It will try every four seconds until a server is located.
	DHCP Error 2.	The handset has not received a response from the server for a request to an IP address. It will retry until a server is found.
	DHCP Error 3.	The server refuses to lease the handset an IP address. It will keep trying.
	DHCP Error 4.	The server offered the handset a lease that is too short. The minimum lease time is 10 minutes but Polycom recommends at least one-hour minimum lease time. The handset will stop trying. Reconfigure the server and power cycle the handset.
	DHCP Error 5.	Failure during WEP Key rotation process (proprietary feature).
DHCP Lease Exp yyy.yyy.yyy.yyy	y…y = DHCP server IP address. DHCP is not responding to renewal attempts .	The handset failed to renew its DHCP lease, either because the DHCP server is not running, or because the configuration has been changed by the administrator. The handset will attempt to negotiate a new lease, which will either work, or it will change to one of the above DHCP errors (1 through 4).
DHCP NACK error yyy.yyy.yyy.yyy	y…y = DHCP server IP address. DHCP server explicitly refused renewal.	The DHCP lease currently in use by the handset is no longer valid, which forces the handset to restart. This problem should resolve itself on the restart. If it does not, the problem is in the DHCP server.
DL Not On Sector	Handset software download files are incorrect or corrupted.	Obtain new software per *Software Maintenance*.

Message	Description	Action
DO NOT POWER OFF	The handset is in a critical section of the software update.	None. Do not remove the Battery Pack or attempt to power off the phone while this is displayed. Doing so may render the handset inoperable.
Duplicate IP	The handset has detected another device with its same IP address.	If using DHCP, check that the DHCP server is properly configured to avoid duplicate addresses. If using Static IP, check that the handset was assigned a unique address.
Erase Failed	Download process failed to erase the memory in the handset.	Operation will retry but may eventually report the error "int. error: 0F" Power cycle the handset.
Erasing Memory	Handset has determined that a download should occur and is erasing the current software from memory.	None. When the progress bar fills the display line the erase operation is complete. Do not remove the Battery Pack or attempt to turn the handset off during this operation.
Files Too Big	Handset software download files are incorrect or corrupted.	Download new software per *Software Maintenance*.
Flash Config Error	Handset internal configuration is corrupt.	Perform "Restore Defaults" operation via administrator menus (or re-program with Configuration Cradle).
Initializing …	The handset is performing power-on initialization.	None. This is informational only.
Internal Err. # #	The handset has detected a fault from which it cannot recover.	Record the error code so it can be reported. Turn the handset off then on again. If error persists, try registering a different handset to this telephone port. If error still persists, contact Polycom Technical Support and report the error.
Must Upgrade SW!	Handset software is incompatible with hardware.	Download new software from the SpectraLink website per *Software Maintenance*.
No DHCP Server	Handset is unable to contact the DHCP server.	Check that your DHCP server is operational and connected to WLAN or use Static IP configuration in the handset.
No ESSID	Attempted to run Site Survey application without an ESSID set.	Statically configure an ESSID in the Admin menu.
No Func Code	Handset software is incorrect or corrupted.	Reconfigure the handset to gain access to the WLAN and download new handset software.

Message	Description	Action
No Host IP (Addr)	The handset is configured for “static IP” (as opposed to “use DHCP”) and no valid host IP address (the handset’s IP address) has been entered.	Enter a valid IP address in the configuration settings or change to “use DHCP.”
No IP Address	Invalid handset IP address.	Check the IP address of the handset and reconfigure if required, or change to “use DHCP”.
No Net Access	Cannot communicate with LAN devices through the WLAN (such as PBX).	Verify the AP connectivity to the LAN. Verify that all the WEP settings in the handset match those in the APs.
No Net Found No APs	Handset cannot find any APs This indicates any of the following:	
	No radio link.	Verify that the AP is turned on. Verify WMM is enabled on AP. WMM is mandatory.
	No ESSID: Auto-learn not supported (or) incorrect ESSID.	Verify the ESSID of the wireless LAN and enter or Auto learn it again if required.
	AP does not support appropriate data rates.	Check the AP configuration.
	Out of range.	Try getting closer to an AP. Check to see if other handsets are working within the same range of an AP. If so, check the ESSID of this handset.
	Incorrect Security settings.	Verify that all the Security settings in the handset match those in the APs.
No Net Found xxxxxxxxxxxx yy	x…x = AP MAC address. yy = AP signal strength. Handset cannot find a suitable AP; displays MAC and signal strength of “best” non-suitable AP found.	Check AP and handset network settings such as ESSID, Security, Reg domain and Tx power. Ensure APs are configured correctly. Try Site Survey mode to determine a more specific cause. Enable WMM in the APs.
No PBX Response	The handset has exceeded its retransmission limit with no ACK response from proxy server.	Verify that proxy server IP address and port are properly configured.
No Reg Domain	Regulatory Domain not set.	Configure the Regulatory Domain in the handset.
No SIP DHCP	DHCP is configured but no valid SIP option 43 was found.	Check DHCP server configuration for option 43 and reconfigure if required.
No SVP IP	Not SVP related. Deprecated (old) error message.	Update handset software to 130.005 or later.
No SVP Response	Not SVP related. No handset License Option set.	Set License Option to 22. See Chapter 2 for detailed information.

Message	Description	Action
No SW Found	A required software component has not been identified.	Check that the handset license type (number) has a corresponding entry in the slnk_cfg.cfg file. Check that the pd11wsd.bin and pi110001.bin entries exist in under this license type in the slnk_cfg.cfg file.
Not Installed!	A required software component is missing.	Check that all required software files are on the TFTP server, if over-the-air downloading is being used. If the error repeats, contact Polycom Technical Support.
Press End Call	The far end of a call has hung up.	Hang up the near end by pressing the **END** key.
Select License	A protocol has not been selected from the license option setting.	Using the Admin menu, select one license option to allow the phone to download the appropriate software.
Server Busy	Handset is attempting to download from a TFTP server that is busy downloading other devices and refusing additional downloads.	None, the handset will automatically retry the download every few seconds.
Storing Config	Handset is storing changes to handset configuration.	None. Informational only. The handset may display this briefly following a configuration change or software download.
TFTP ERROR(x):yy	A failure has occurred during a TFTP software download. (x) = The file number which was being downloaded; yy is an error code describing the particular failure. Possible error codes are: 01 = TFTP server did not find the requested file. 02 = Access violation (reported from TFTP server). 07 = TFTP server reported "No such user" error. 16 = No TFTP server address. 81 = File put into memory did not CRC. FF = Timeout error. TFTP server did not respond within a specified period of time.	Error code 01, 02, 07, or 16 - check the TFTP server configuration. Error code 81, the handset will attempt to download the file again. For other messages, power off the handset, then turn it on again to retry the download. If the error repeats, note it and contact Polycom Customer Support.
Too Many Errors	The handset continues to reset and cannot be recovered.	Fatal error. Return handset to Polycom.
Unknown xx:yy:zz	A phrase is missing from the language file.	Obtain new software per *Software Maintenance*.

Message	Description	Action
Unsupported Codec	The proxy server has requested using a codec not supported by the handset.	Check proxy server configuration for supported codecs and reconfigure if necessary.
Updating …	The handset is updating its software .	None. The handset may do this briefly after a download. This is informational only.
Updating Code…	Handset is downloading new software into memory. The number icons at the bottom of the display indicate which file number is currently being downloaded. This message also displays a progress bar. When the progress bar fills the display line the update operation is complete on that file.	None. When the progress bar fills the display line the update operation is complete on that file. Do not remove the Battery Pack or attempt to turn the handset off during this operation.
Waiting…	Handset has attempted some operation several times and failed.	None. The handset is waiting for a specified period of time before attempting that operation again.
Wrong Code Type	The software loaded into the handset is incorrect for this model phone.	Replace the software image on the TFTP server with software that is correct for the handset model and restart the handset.

Appendix A: Regulatory Domains

This table details the specifications for regulatory domain settings. Polycom recommends that you check with local authorities for the latest status of their national regulations for 2.4 GHz wireless LANs.

Domain Identifier	Band	Channels	Max. Power Limit (peak power)	Countries
01	2.4000 – 2.4745 MHz	1-11	100mW (+20dBm)	Argentina Canada Mexico Taiwan US
02	2.4000 – 2.4845 MHz	1-13	100mW (+20dBm)	Australia Europe Hong Kong Japan New Zealand Singapore
03				(reserved)
04	2.4445 – 2.4745 MHz	10-11	30mW (+15dBm)	Spain
05	2.4145 – 2.4845 MHz	10-13	30mW (+15dBm)	France
06	2.4000 – 2.4845 MHz	1-13	30mW (+15dBm)	Korea

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