SpectraLink 8020/8030 Wireless Telephone

Administration Guide

Session Initiation Protocol (SIP)

January 2008 Edition
1725-36038-001
Version B

POLYCOM®

Trademark Information

Polycom® and the logo designs
SpectraLink®
LinkPlus
Link
NetLink
SVP
Are trademarks and registered trademarks of Polycom, Inc. in the United States of America and various countries. All other trademarks used herein are the property of their respective owners.

Patent Information

The accompanying product is protected by one or more US and foreign patents and/or pending patent applications held by Polycom, Inc.

Copyright Notice

Every effort has been made to ensure that the information in this document is accurate. Polycom, Inc. is not responsible for printing or clerical errors. Information in this document is subject to change without notice and does not represent a commitment on the part of Polycom, Inc.

Notice

Polycom, Inc. has prepared this document for use by Polycom personnel and customers. The drawings and specifications contained herein are the property of Polycom and shall be neither reproduced in whole or in part without the prior written approval of Polycom, nor be implied to grant any license to make, use, or sell equipment manufactured in accordance herewith.

Polycom reserves the right to make changes in specifications and other information contained in this document without prior notice, and the reader should in all cases consult Polycom to determine whether any such changes have been made.

No representation or other affirmation of fact contained in this document including but not limited to statements regarding capacity, response-time performance, suitability for use, or performance of products described herein shall be deemed to be a warranty by Polycom for any purpose, or give rise to any liability of Polycom whatsoever.

Contact Information

Please contact your Polycom Authorized Reseller for assistance.

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

About this Guide

This document explains how to configure and maintain the SpectraLink 8020/8030 Wireless Telephones with Session Initiation Protocol (SIP); a protocol for the control of voice over IP (VoIP) calls via a proxy server(s).

Polycom Model Numbers

This document covers the following registered model numbers:

802X, 803X

Related Documents

SpectraLink 8000 SVP Server: Administration Guide for SIP (1725-36033-001)

SpectraLink 8020/8030 Wireless Telephone: SIP User Agent: Features and Standards (1725-36037-001)
Available at
http://www.spectralink.com/resources/manual_netlink.jsp.

Polycom WLAN Compatibility Table (1725-36040-001)

Access Point Configuration Guide 1725-36xxx-001 where xxx indicates a number corresponding to the type of access point.)
Available at
http://www.spectralink.com/resources/wifi_compatibility.jsp.

Best Practices for Deploying Enterprise-Grade Wi-Fi Telephony
Available at
http://www.spectralink.com/resources/white_papers.jsp.

Open Application Interface (OAI) Specification (Version 2.0) (72-0052-00)
Technical specifications available by request. Please contact Customer Support.

Asterisk cmd VoiceMailMain (Shows how to enter the Asterisk voicemail system, its menu structure, and related information.) Available at http://www.voip-info.org/wiki/index.php?page=Asterisk+cmd+VoiceMailMain.

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 SpectraLink hardware.

Contents

1

SpectraLink 8020/8030 Wireless Telephone Overview

The SpectraLink 8020/8030 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. The handset radio transmits and receives packets at up to 54 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. Each handset may have up to 10 sets of credentials to identify itself by current user.

SpectraLink Voice Priority (SVP) and Quality of Service

SVP is the SpectraLink quality of service (QoS) mechanism that is implemented in the handset and an access point (AP) to enhance voice quality over the wireless network. SVP gives preference to voice packets over data packets on the wireless medium, increasing the probability that all voice packets are transmitted efficiently and with minimum or no delay. SVP is fully compatible with the IEEE 802.11 standards.

The SpectraLink 8000 SVP Server is an Ethernet LAN device that works with the AP to provide quality of service (QoS) on the wireless LAN. Voice packets to and from the SpectraLink 8020/8030 Wireless Telephones are intercepted by the SpectraLink 8000 SVP Server and encapsulated for prioritization as they are routed to and from a SIP proxy server. See the *SpectraLink 8000 SVP Server Administration Guide for SIP* document for detailed information about this device.

SpectraLink 8020/8030 Wireless Telephone Security

SpectraLink 8020/8030 Wireless Telephones support the 802.11i standard including Wi-Fi Protected Access (WPA and WPA2) in the pre-shared key (PSK) mode. As vendors introduce access points (APs) that are eligible to become Wi-Fi CERTIFIED for WPA-PSK and/or WPA2-PSK, Polycom will determine compatibility with the SpectraLink 8020/8030 Wireless Telephones and include these APs on the *Polycom WLAN Compatibility Table.*

SpectraLink 8020/8030 Wireless Telephones support basic WMM as part of the 802.11e standard. If the AP supports WMM, the handset automatically discovers and uses it. WMM does not replace the SpectraLink 8000 SVP Server as described in the first paragraph. WMM settings must be configured on the SpectraLink 8000 SVP Server.

The SpectraLink 8020/8030 Wireless Telephone supports Wired Equivalent Privacy (WEP) as defined by the 802.11 standard. Polycom offers the product with both 40-bit and 128-bit encryption.

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

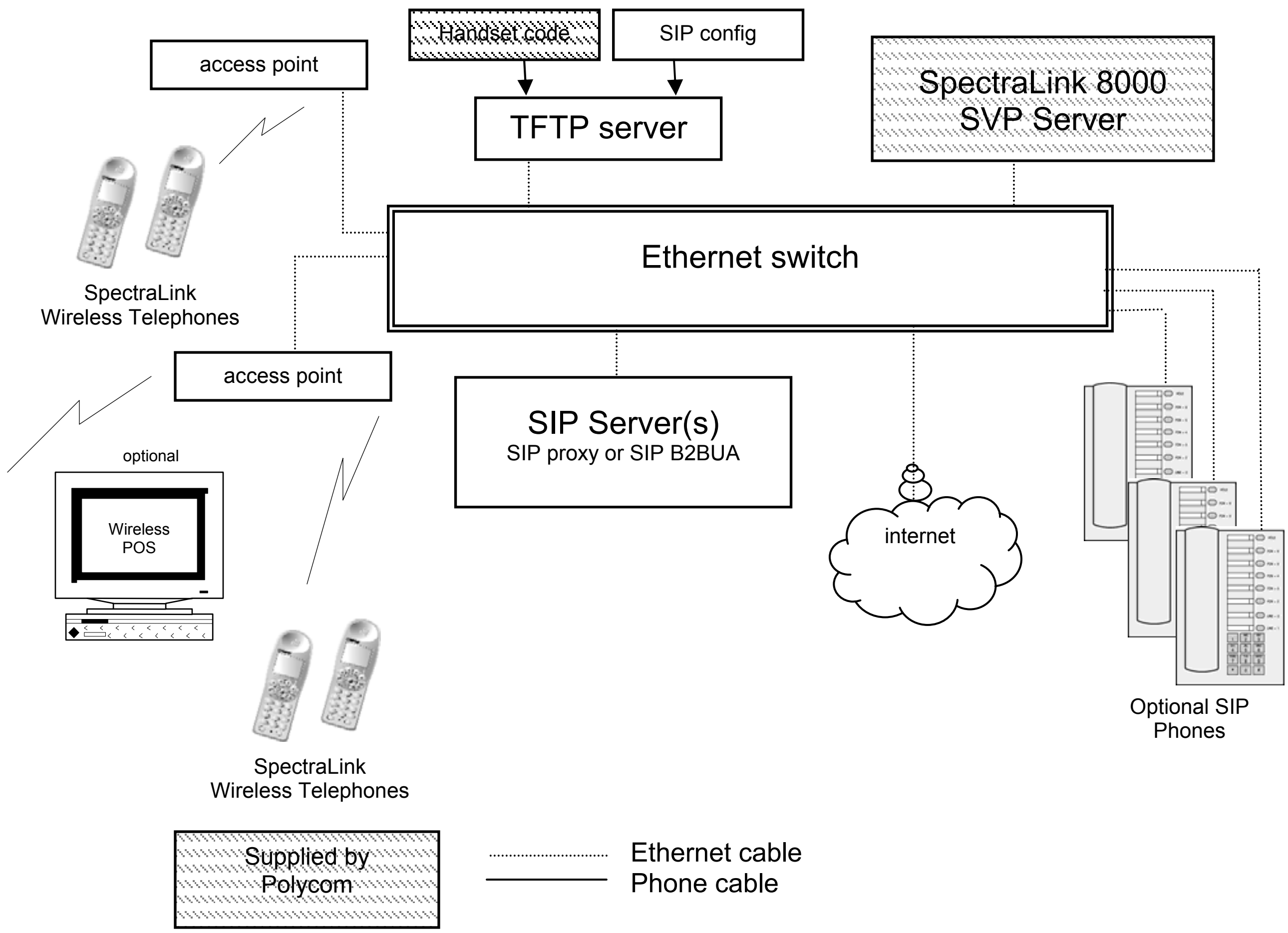
Quick Start Guide

1. A wireless LAN must be properly configured and operational through the use of 802.11 wireless APs.
2. A TFTP server must be available on the network in order to load the appropriate software onto the handsets. See Chapter 3 *Software License and Protocol Management* for detailed instructions for loading software on handsets.
3. The supported SIP system components must be connected to your network and completely operational.

4. The SpectraLink 8000 SVP Server, which facilitates the QoS on the wireless LAN for the handsets, must be on the same subnet as the handsets and have the proper versions of software.
   Ensure you have the following versions for the SpectraLink 8000 SVP Server:
   173 svp100.toc
   174 zvmlinux
   175 flashfs
5. Visit http://www.spectralink.com/softwareUpdates to download the latest handset software and any updates to the SpectraLink 8000 SVP Server software.
6. Install the correct handset software per Chapter 3 *Software License and Protocol Management*, section *Configuration Process*.
7. Install any updates to the SpectraLink 8000 SVP Server software per *SpectraLink 8000 SVP Server Administration Guide for SIP*, Chapter 5 *Software Maintenance*. Ensure the software is properly loaded on the TFTP server.
8. Configure your handset to ensure that it is associated with the wireless LAN, has the appropriate software, and has the correct IP address for the supported proxy server. See Chapter 3 *Software License and Protocol Management* and Chapter 2 *SpectraLink 8020/8030 Wireless Telephone Configuration* for detailed instructions for loading software onto and configuring handsets.
9. Create configuration files on the SIP TFTP server to define parameters for the SIP application. See Chapter 5 *Programming the SIP Handset Features*, section *SIP TFTP Server Configuration Files*.

System Diagram

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



System Components

SpectraLink 8020/8030 Wireless Telephone

The SpectraLink 8020/8030 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 8020/8030 Wireless Telephones use direct-sequence spread spectrum radio technology (DSSS) to transmit audio packets over wireless LAN APs that support SpectraLink Voice Priority (SVP).

SpectraLink 8000 SVP Server

SVP is the SpectraLink quality of service (QoS) mechanism that is implemented in the AP to enhance voice quality over the wireless network. SVP gives preference to voice packets over data packets on the wireless medium, increasing the probability that all voice packets are transmitted efficiently and with minimum or no delay. SVP is fully compatible with the IEEE 802.11b standard.

The SpectraLink 8000 SVP Server is an Ethernet LAN appliance that works with the AP to provide QoS on the wireless LAN. All SIP packets to and from the SpectraLink 8020/8030 Wireless Telephones pass through the SpectraLink 8000 SVP Server and are encapsulated for prioritization as they are routed to and from other SIP devices.

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. Typically, the requirements for use of SpectraLink 8020/8030 Wireless Telephones are similar to that of wireless data devices.

Access points must utilize SpectraLink Voice Priority (SVP). See the *Polycom WLAN Compatibility Table* for information about APs that support SVP.

Ethernet switch

Interconnects multiple network devices, including the SpectraLink 8000 SVP Server, 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 8020/8030 Wireless Telephones.

Although a single Ethernet switch network is recommended, the handsets and the SpectraLink 8000 SVP Server 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 for the quality of service (QoS) features of the SpectraLink 8000 SVP Server to be compromised and 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 8020/8030 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 handsets can operate with SIP proxy servers such as SER (SIP Extensible Router) or with SIP Back-to-Back User Agents (B2BUA) – the most common form of SIP server for PBX-based systems. The handsets can also operate with no SIP server at all. In this case the IP address location services provided by the SIP server are not available, so direct IP address dialing must be used.

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

SIP phone

The optional wired-LAN desksets such as those provided by Cisco, Inter-Tel, and Polycom.

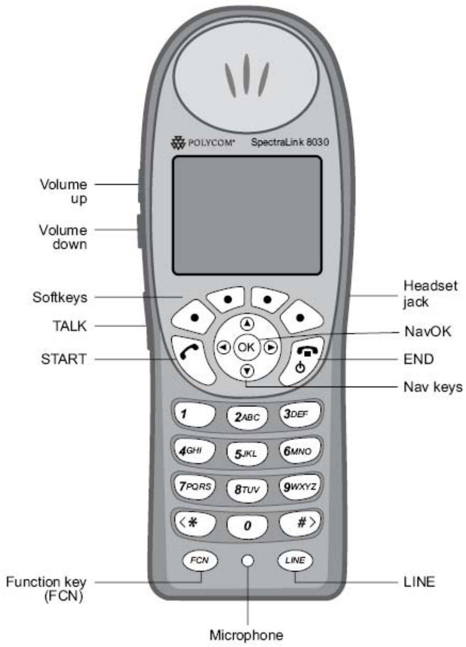
TFTP server

Required in the system to distribute software to the handsets. May be on a different subnet than the gateway, APs and/or handsets.

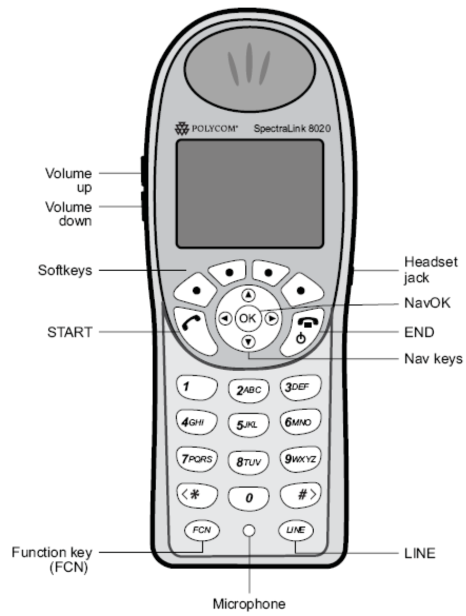
SIP TFTP server

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 8020/8030 Wireless Telephone Configuration*, section *The Admin Menu*.) Normally, the SIP TFTP server is the same as the network TFTP server, see above.

SpectraLink 8020/8030 Wireless Telephone Specifications



SpectraLink 8030 Wireless Telephone



SpectraLink 8020 Wireless Telephone

Table of Specifications

Radio mode (selectable)	(802.11b, 802.11g)	2.4–2.4835 GHz
	(802.11a)	5.150–5.250 GHz 5.250–5.350 GHz 5.470–5.725 GHz 5.725–5.825 GHz
Transmission type	Direct-sequence spread spectrum (DSSS)	
Transmit data rate	up to 54 Mb/s	
Radio QoS	SpectraLink Voice Priority (SVP), WMM	
Wireless security	Wired Equivalent Privacy (WEP), 40-bit and 128-bit; Cisco FSR; WPA-PSK, WPA2-PSK	
FCC certification	Part 15.247	
Voice encoding	ADPCM (Proprietary)	
Transmit power	See Admin menu	
Display	Up to five lines of text plus two icon status rows and one row for softkey labels.	
8020 Dimensions	5.7" x 2.0" x 0.9" (14.5 x 5.1 x 2.3 cm)	
8030 Dimensions	5.4" x 2.0" x 0.9" (13.7 x 5.1 x 2.3 cm)	
8020 Weight*	3.9 oz. (110.6 g) with Standard Battery Pack	
8030 Weight*	4.2 oz. (119.1 g) with Standard Battery Pack	
Standard Battery Pack capacity	4 hours talk, 80 hours standby	
Extended Battery Pack capacity	6 hours talk, 120 hours standby	
Ultra-Extended Battery Pack capacity	8 hours talk, 160 hours standby	

Startup Sequence

The SpectraLink 8020/8030 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**	Note: Only valid on non-SRP protocol. Indicates one of the following: Static IP configuration, or SpectraLink 8000 SVP Server address found in DHCP response, or SpectraLink 8000 SVP Server address found via DNS lookup.
1234**5**	All networking functions are complete (notably, DHCP), and the handset is proceeding with establishing the SRP link to the SpectraLink 8000 SVP Server.
123**4**	The SRP link is established; all network stack initialization is complete, proceeding with application-specific initialization.
12**3**	SIP application startup. Icon 3 is extinguished if a generic SIP config file is found.
1**2**	Icon 2 is extinguished if a handset specific SIP config file is found.
(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 the configuration file that contains user-specific information. If the file is not found, an error message will appear and the handset will restart. See Chapter 5 *Programming the SIP Handset Features*, section *SIP TFTP Server Configuration Files*.

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.

When an incoming call arrives, the handset rings; the handset enters the active mode and remains so until the call is ended. The call is answered by pressing the **START** key or the **Answ** softkey. The handset will ring according to user preference as specified in the standby menus. The ringing can be silenced by pressing the **END** 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.)

Active mode (off-hook)

The handset is in the active mode when an incoming call is answered 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 the **END** key is pressed, the first call is terminated and the handset reverts to a full ring.

The active mode utilizes 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** key.

Push-to-talk (PTT) mode

The SpectraLink 8030 Wireless Telephones utilize channels for incoming and outgoing radio communication. While PTT is active, the handset is in PTT mode. It can receive regular phone calls in this mode. When a regular phone call is answered, the handset enters active mode.

Configuration menu mode

When user preferences are being configured in the Config menu, the handset is on but is not active. It cannot receive calls while in the Config menu.

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.

2

SpectraLink 8020/8030 Wireless Telephone Configuration

Each 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 Handset Administration Tool is a software utility that enables rapid configuration of handsets by utilizing the USB port on the Dual Charger. See the *Handset Administration Tool* document for specific instructions. Please see your service representative or contact Polycom customer service for more information about this time-saving tool.

The Admin 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 options must be set in each handset requiring different settings.

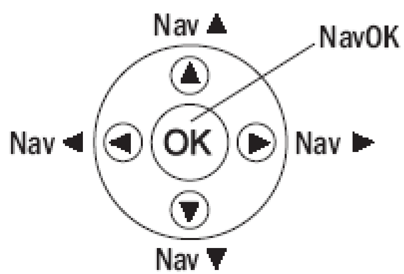
Opening the Admin menu

1. With the handset powered off, press and hold the **START** key. While holding the **START** key, press and release the **END** key.
2. When the Admin menu appears, release the **START** key.

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.

Navigation

The navigation keys just below the softkeys are used to navigate through and select menu options. These are referred to as **Nav▲**, **Nav▼**, **Nav◄**, **Nav►**, and **NavOK**.



Toggle options

Some menu items have only two options, which operate on a toggle basis. The current setting is shown below the menu heading on the info line. The other available setting is highlighted in the menu list. Press **NavOK** to activate the highlighted setting.

For example, when predial is disabled, the info line displays **Predial Disabled** and the highlighted menu item is the **Enable Predial** option. Press **NavOK** to enable predial. The info line will change to display **Predial Enabled**.

In another example, when the info line displays **Ring in Speaker**, the highlighted menu option is **Ring in Headset**. Press **NavOK** to select **Ring in Headset**, The ring will now sound in the headset and the info line will change to **Ring in Headset**.

Data entry and editing

An asterisk (*) next to an option on the display indicates that it is selected. Use the **Nav** keys and the softkeys to navigate and select desired options.

Enter numbers by pressing the buttons on the keypad. The blinking underscore identifies the current cursor position. When entering alphanumeric strings, the **CAPS/caps** softkey will appear and may be pressed to toggle the case. Enter letters by repeatedly pressing the corresponding key until the desired letter displays on the screen. Use the **CAPS** softkey to change the case as needed.

To edit during entry, delete the character to the left of the cursor by pressing the **Del** softkey. To replace an entry, delete it by pressing the **Clr** softkey and then enter the new data. To edit an existing entry, use **Nav◀** and **Nav▶** to move the cursor position, and then press the **Del** softkey to delete the character to the left. Insert new data by pressing the buttons on the keypad.

Alphanumeric entries:

Key	caps	CAPS
1	1	1
2	2 a b c	2 A B C
3	3 d e f	3 D E F
4	4 g h I	4 G H I
5	5 j k l	5 J K L
6	6 m n o	6 M N O
7	7 p q r s	7 P Q R S
8	8 t u v	8 T U V
9	9 w x y z	9 W X Y Z
0	0	0
*****	* . ! $ % & ' ( ) + , : ; / \ = @ ~	
#	<space>	

Admin menu

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

Admin Menu

1st level	2nd level	3rd level	4th level	5th level
Phone Config	Telephony Protocol	*Type 036		
	Push-to-talk	PTT Enable/*Disable		
		Allowed Channels	*Channel 1 *Channel 2 *…. *Channel 24	
		Name Channels	[list]	Enter Name
		Priority Channel	Priority Channel On/*Off	
			Name Channel	[Enter Name]
	Time Zone	[list] *GMT		
	Daylight Savings	*DST No Adjust DST Auto (USA) DST Auto (AUS) DST Auto (EURO)		
	Password *Enable/Disable			
	[If Password is enabled] Change Password			
	SIP Registration	Login Reg 2 Reg 3 Reg 4	[for each option] Username Password	
	Clear SIP Regist.			
	OAI	*Enable OAI Disable OAI		
	Location Service	Enable RTLS *Disable RTLS		
		Transmit Interval	1 minute 5 minutes *10 minutes	
		Location Server IP	Enter IP	
		ELP Port	Enter Port *8552	

1st level	2nd level	3rd level	4th level	5th level
Network Config	IP Addresses	*Use DHCP		
		Static IP	Phone IP Default Gateway Subnet Mask TFTP Server IP Syslog Server IP Time Server IP SVP Server IP SIP TFTP Svr IP OAI Server IP	
	SS ID	[enter]		
	Security	*None		
		WEP	Authentication	*Open System Shared Key
			Enable WEP *Disable WEP	
			Key Information	Default Key Key Length Key 1-4
			Rotation Secret	
		WPA2-PSK	Passphrase *Pre-Shared Key	
		WPA-PSK	Passphrase *Pre-Shared Key	
		Cisco FSR	Username Password	

1st level	2nd level	3rd level	4th level	5th level
Network Config	Reg. Domain	01 02 03 04 05 06 07		
		→	[802.11 Config]	
			a →	[802.11a] 5.150–5.250 5.250–5.350 DFS 5.470–5.725 DFS 5.470-5.650 DFS 5.725–5.825 5.725-5.850
			*b & b/g mixed g only	
			→	[Transmit Power] 5mW (7dBm) 10mW (10dBm) 20mW (13dBm) *30mW (15dBm) 40mW (16dBm) 50mW (17dBm) 100mW (20dBm)

* Subbands have not been established for the b and b/g mixed or the g-only mode at this writing. Provision is made in the software to accommodate these ranges once established. Until added, selecting either of these two modes will immediately bring up Transmit Power options.

1st level	2nd level	3rd level	4th level	5th level
Diagnostics	Run Site Survey			
	Diagnostics Mode	*Disable Enable		
	Syslog Mode	*Disabled Errors Events Full		
	Error Handling Mode Halt on Error/ *Restart on Error			
Restore Defaults				

Phone Config

Telephony Protocol

Telephony Protocol lets you select the VoIP protocol that your site is licensed to download and run. The SIP protocol used for the SpectraLink 8020/8030 Wireless Telephones requires license option selection **36**. Any other protocol will cause the handset to malfunction.

Push-to-talk (PTT)

PTT is disabled by default. When enabled, all 24 PTT channels are allowed by default. To toggle the allowed status of any channel, select **Allowed Channels**, scroll to the channel to be disallowed and press **NavOK**. Allowed channels are displayed with an asterisk (*) in the left column. Only those channels allowed in the Admin menu will appear on the Config menu where they can be subscribed to by the end user. The priority channel, labeled by default as channel 25, may be set and will be available to all PTT handsets. When a PTT broadcast is made on the priority channel, it will override any active PTT transmission on all other channels.

Time Zone

Worldwide time zone options are available. Greenwich Mean Time (GMT) is the default.

Daylight Savings

The handset may be adjusted for daylight savings time.

Password Enable/Disable/Change

The password option controls access to the Admin menu. It is enabled by default with the password 123456. The **Password** option operates as a toggle between **Enabled** and **Disabled**. The info line will display the current state. Press **NavOK** to change the password protection state. To modify the password requirement, the default or previously set password must be entered to verify the change. **Change Password** will appear only if the password is enabled. The password is disabled by default. The password must be set in each handset for which controlled access is desired.

SIP Registration

Individual handsets may be configured to correspond with the SIP configuration information in the TFTP server. The handset is then automatically identified at startup. If username and password information is not configured in the Admin menu, then this information will be requested at startup

In either case, the username must agree with a corresponding configuration file. See Chapter 5 *Programming the SIP Handset Features*, section *SIP TFTP Server Configuration Files*.

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 the **Bksp** softkey and then the **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 **Login** username is not provided.

OAI Enable/Disable

Polycom'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 (Version 2.0)* documentation for information about administering the OAI Gateway and the services it can provide.

If you have an OAI Gateway installed in your system, OAI may be optionally enabled in each handset. You may select whether the handset should attempt to connect to the SpectraLink 8000 OAI Gateway by choosing either the **Enable** or **Disable** options in this menu.

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.

Location Service

Location service may be used to enable or disable the Ekahau Real-Time Location System (RTLS), select a transmit interval, or enter a static IP address for the Ekahau Positioning Engine (EPE). Location services capability is provided by the EPE 4.0 using Ekahau Location Protocol (ELP). See Ekahau's user documentation for more information.

RTLS [Enable/Disable] The RTLS is disabled by default. Press **NavOK** to toggle to the alternate setting. When RTLS is enabled, the handset will display the RTLS icon in the top center of the screen.

The ring indicator icon will take precedence over the RTLS icon, i.e. the new icon will not be visible while the handset is ringing. When ringing has ceased and the ring indicator becomes inactive, the RTLS icon will again appear (regardless of hook state).

Transmit interval Allows selection of **1 minute**, **5 minutes**, or **10 minutes** for maximum time between transmit intervals. Default transmit interval is 10 minutes. Press **NavOK** to select the desired transmit interval.

> To optimize battery life, the interval between sending out ELP updates will vary based on handset state. It is expected that ELP updates will occur at most every two to six seconds and at least every few minutes. If improved tracking capability is desired, set the transmit interval for a shorter time between ELP updates. Increasing the frequency of transmissions will decrease battery life.

Location Server IP Allows the user to statically enter the IP address of the EPE. Enter the IP address and press **NavOK** to save.

> Ekahau clients are not expected to find the EPE automatically. Regardless of the handset's selection of DHCP or static IP, the EPE IP address must be statically entered in the Ekahau admin menus or HAT.

ELP Port Allows the user to select the port number which ELP updates get sent to at the Location Server IP address. It must match the value configured in the Ekahau Positioning Engine for proper functionality. The ELP port number must be greater than zero and less than 65536. Default is 8552. Enter the port number and press **NavOK** to save.

Network Config

IP Addresses

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.

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. 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.

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.

SpectraLink 8020/8030 Wireless Telephones cannot roam with uninterrupted service between subnets unless specific LAN components are present. Certain AP/Ethernet switch combinations establish a Layer-2 tunnel across subnets that enable the handsets to roam. Without this capability, any call in progress will be dropped when the user moves out of range and the handset must be power cycled in order to resume functionality in the new subnet area.

Ensure that all your 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 APs on the new subnet. Note that the wireless telephones cannot “roam” across subnets, since they cannot change IP addresses while operational.

Please see *Best Practices for Deploying Enterprise-Grade Wi-Fi Telephony* for detailed configuration information.

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.

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

Time Server IP The IP address of the time server.

SVP Server IP The IP address of the SpectraLink 8000 SVP Server. If using Static IP configuration, this is simply the IP address of the SpectraLink 8000 SVP Server. Note that the SpectraLink 8000 SVP Server must be statically configured to have a permanent IP address. If DHCP is being used, the handset will try the following, in order: the DHCP option 151, then a DNS lookup of "SLNKSVP2" if the DHCP options 6 (DNS server) and 15 (Domain Name) are configured.

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. See Chapter 4, *SIP Integration Factors*.

OAI Server IP The IP address of the SpectraLink 8000 OAI Gateway. If using Static IP configuration, this is simply the IP address of the SpectraLink 8000 OAI Gateway. If DHCP is being used, the handset will try the DHCP option 152.

SSID

Enter the SSID.

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.

> Encryption codes display as they are entered. For security reasons codes will not display when a user returns to the Admin menu, Encryption options.

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 Enable/Disable
Select either **Enable WEP** or **Disable WEP**.

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 1-4 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 through F.

Rotation Secret

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

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.

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.

Cisco FSR (Fast Secure Roaming) In order to provide the highest level of security without compromising voice quality on Cisco Aironet wireless LAN APs, Polycom and Cisco Systems have cooperated to implement the Fast Secure Roaming mechanism. FSR is designed to minimize call interruptions for SpectraLink 8020/8030 Wireless Telephone users as they roam throughout a facility. Existing Aironet 350, 1100, and 1200 APs may require a firmware upgrade to support FSR. Cisco FSR requires advanced configuration of the Cisco APs in your site. See your Cisco representative for detailed documentation on configuring the APs and other required security services on the wired network. To configure Cisco FSR on a handset, you must enter a Radius Server username and password into each handset.

Username

Enter a username that matches an entry on the Radius server. Usernames are alphanumeric strings, and can be entered using the alphanumeric string entry technique.

Password

Enter the password that corresponds to this Username.

> Consult the *Configuration Guide* for the APs installed in your facility for information on which of the WPA versions are recommended by Polycom engineering. Configure the recommended version on the AP and select the corresponding option on the Admin menu.

Regulatory Domain/802.11 Config/Transmit Power

Regulatory domain, 802.11 configuration and transmit power are interdependent. See *Appendix A: Regulatory Domains* for regulatory domain setting specifications. Polycom recommends that you check with local authorities for the latest status of national regulations for both 2.4 and 5 GHz wireless LANs.

FCC requirements dictate that the menu for changing the regulatory domain be available by password, which in our case is the **LINE** key. Press **LINE** and then navigate to the desired domain. Press **NavOK** to set the domain.

01 - North America

02 – Europe

03 – Japan

04 – Singapore

05 – Korea

06 – Taiwan

07 – Hong Kong

802.11 config

Once the regulatory domain is set, the **802.11 Config** modes are displayed. Only one may be chosen. **802.11(b & b/g mixed)** is the default. Press **NavOK** to set the mode. If the mode has subbands, the **Subband** list will open. If the mode does not have subbands, the **Transmit Powe**r list will open.

Use **g only** if all of your infrastructure devices use only 802.11g. The handsets will operate up to 54 mb/s in this mode.

Use **b & b/g mixed** if some of your infrastructure components only understand 802.11b. The handsets will operate up to 11 mb/s.

Subbands have not been established for the **b and b/g mixed** or the **g only** mode at this writing. Provision is made in the software to accommodate these ranges once established. Newly added subbands may not appear in the above table.

Subband

Once a mode is set the subband list will display, if applicable. Only those ranges which are allowed in the set regulatory domain and that pertain to the set mode are displayed. Note that for 802.11a the bands labeled **DFS** will vary depending on the set regulatory domain. Multiple subbands may be set. Navigate to the desired subband and set with **NavOK.** The **Transmit Power** menu will open. Once the **Transmit Power** setting is done, you will be returned to the subband list.

To deselect a subband, navigate to it and press **NavOK**.

Once the subband settings are as desired, press the **Done** softkey to exit to the **Network Setup** menu.

Transmit power

For subbands: The **Transmit Power** list opens when **NavOK** is pressed from the **Subband** menu. A transmit power setting is required for each subband. Only one level may be set per subband. Only those power levels which apply to the regulatory domain and 802.11 mode are listed. Navigate to the desired level and press **NavOK** to set and return to the subband list. Another subband may be selected which repeats the process.

If the highlighted power transmit level is legal on all of the subbands for the set mode, an **All** softkey will appear. Press the **All** softkey to apply that level to all subbands and return to the subband menu where all subbands will now be selected. **All** overrides any previously set power transmit levels.

Without subbands: When the 802.11 mode has no subbands, the **Transmit Power** list opens when **NavOK** is pressed to set the mode. Only those power levels which apply to the domain and 802.11 mode are listed. Navigate to the desired level and press **NavOK**. This sets the transmit power level and exits the **Regulatory Domain** menus. The **Network Setup** menu will again display.

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 the *Diagnostic Tools* section for more information about site survey.

Diagnostics Mode

Diagnostics can be enabled or disabled. See Chapter 8 *Diagnostic Tools*, section *Diagnostics Enabled* for a detailed explanation of the **Diagnostics** mode options.

Syslog Mode

See Chapter 8 *Diagnostic Tools*, section *Syslog Mode* for a detailed explanation of the **Syslog** mode options.

Error Handling Mode

The **Error Handling** mode determines how the handset will behave when an error occurs. The **Halt on Error** option will cause the handset to stop operating if an error message is received. Unless the error is a fatal one, normal operation may be resumed by power-cycling the handset. The **Restart on Error** option will cause the handset to make every effort to reboot quietly and quickly to standby mode. In either scenario, a call in progress will be lost.

Error detail may be shown on the display, captured by the syslog server and may also be available for downloading with the Handset Administration Tool.

Restore Defaults

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

Admin Menu Default Table

When the **Restore Defaults** option is selected, administrative parameters will be reset to their factory defaults as shown in the table below. The **Telephony Protocol** setting will not change. User parameters will be reset per the table on page 41.

Menu option	Setting	Sub-option	Sub-sub-option	Default
Phone Config				
	Push-to-Talk			Disabled
		Allowed Channels		[all]
		Name Channels		[None set]
		Priority Channel		Disabled
	Time Zone			GMT
	Daylight Saving			DST No Adjust
	Password			Enabled
	Change Password			[n/a]
	SIP Registration			[None set]
	Clear Regist.			[n/a]
	OAI			Enabled
	Location Service			
		RTLS		Disabled
		Transmit Interval		10 minutes
		Location Server IP		[None set]
		ELP Port		8552

Menu option	Setting	Sub-option	Sub-sub-option	Default
Network Config	IP Addresses			Use DHCP
	SSID*			[None set]
	Security			None
		WEP	Authentication	Open System
			WEP	Disabled
			Key Information	[None set]
			Rotation Secret	[None set]
		WPA2-PSK		Pre-Shared Key
		WPA-PSK		Pre-Shared Key
		Cisco FSR	Username Password	[none set]
	Reg. Domain*			[none set]
		802.11 mode		b & b/g mixed
		Transmit Power		30 mW (15 dBm)
Diagnostics	Run Site Survey			[n/a]
	Diagnostics			Disabled
	Syslog Mode			Disabled
	[Error Handling Mode]			Restart on Error

User-Defined Preferences

The SpectraLink 8020/8030 Wireless Telephone features a configuration menu ("Config menu") that is available to the user to configure user preferences and display handset information. The Config menu is opened by pressing the **Cfg** softkey from standby mode. See the *SpectraLink 8020/8030 Wireless Telephone and Accessories User Guide*.

Config Menu

Config menu	2nd level	3rd level	4th level	5th level	6th level
Lock Keys					
User Profiles	Silent Vibrate Loud Soft Custom				
		Set as Active			
		Ring Settings	Telephone Ring Message Alert 1 Message Alert 2		
				Ring Cadence	Off PBX Continuous Short Pulse Long Pulse
				Ring Tone	Tones 1-10
				Ring Volume	Volume ■■■■■■■■
				Vibrate Cadence	Off PBX Continuous Short Pulse Long Pulse

Config menu	2nd level	3rd level	4th level	5th level	6th level
		Noise Mode[1]	Normal High Severe		
		Ring in Headset Ring in Speaker			
		Warning Tones Disable/Enable			
		Key Tones Disable/Enable			
		PTT Disable/Enable			
Phone Settings	Keypad Autolock	Disable 5 Seconds 10 Seconds 20 Seconds			
	Display Contrast	Set Contrast			
	Use Hearing Aid Use No Hearing Aid				
	Play Startup Song Inhibit Song				
	Predial Disable/Enable				

[1] High and Severe noise modes increase microphone, speaker, and ring volume settings above Normal mode baseline. All measures are approximate.

	Microphone	In-ear speaker	Ring volume
High	+12dB	+6dB	+3dB
Severe	+18dB	+12dB	+6dB

Config menu	2nd level	3rd level	4th level	5th level	6th level
Push-to-talk	Default Channel	Channel 1 …. Channel 24			
	Subscribed Channels	Channel 1 Channel 2 Channel 3 …. Channel 24			
	PTT Audio Volume	Audio Volume ■■■■■■■■			
	PTT Tone Volume	Tone Volume ■■■■■■■■			
System Info	Phone IP Address				
	Alias IP Address				
	SVP IP Address				
	OAI IP Address				
	Firmware Version				

Default settings

The profile options on the standby menu may be reset to their default values by the **Restore Defaults** option in the Admin menu. These are the default settings:

Setting/profile	Silent	Vibrate	Soft	Loud	Custom
Ring Cadence	Off	Off	PBX	PBX	PBX
Ring Tone	Tone 1	Tone 1	Tone 1	Tone 1	Tone 1
Ring Volume	1	1	3	7	5
Vibrate Cadence	Off	PBX	Off	Off	PBX
Ring Delay	0	0	0	0	5
Noise Mode	Normal	Normal	Normal	Normal	Normal
Headset/Speaker	Speaker	Speaker	Speaker	Speaker	Speaker
Key Tones	Off	Off	On	On	On
Warning Tones	Off	Off	On	On	On
Push-to-talk	Off	Off	On	On	On

3

Software License and Protocol Management

SpectraLink 8020/8030 Wireless Telephones support a number of different IP protocol integrations. All SpectraLink 8020/8030 Wireless Telephones are shipped from Polycom with a generic software load that allows them to associate to a wireless LAN and download functional software from a TFTP server. **The handsets will not function properly without downloading appropriate software.**

The following details the process to properly configure SpectraLink 8020/8030 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 802.11a/b/g wireless APs.
- A TFTP must be available on the network in order to load the appropriate software into the handsets.
- Software versions required:

Component	Version
SpectraLink 8000 SVP Server	17x.028 or higher
OAI Server MOG 600	54.032 or higher
OAI Server MOG 700	82.017 or higher

- Finally, ensure that the Battery Pack on the handset is fully charged.

Configuration Process

1. Download the latest SpectraLink 8020/8030 Wireless Telephone IP software from http://www.spectralink.com/softwareUpdates.
2. Load the latest version of the SIP code and place it on the TFTP server and ensure the TFTP server is started. The four files that are needed must be named:

usb downloader	pd14udsp.bin
functional filename	pd14csp.bin
phintl filename	pi1400sp.bin
ota downloader	pd14odsp.bin
config file	slnk_cfg.cfg

3. Use the Handset Administration Tool to set up the configuration of each handset to meet all essential requirements. If not using the Handset Administration Tool, ensure the following parameters are correctly set in the Admin menu for each handset: See Chapter 2 *SpectraLink 8020/8030 Wireless Telephone Configuration* for detailed configuration instructions.
   – If statically assigning IP addresses, ensure that the **Phone IP**, **Subnet Mask**, and **Default Gateway** information are accurate. If using a DHCP Server, ensure that the DHCP option is set.
   – Ensure the handset has properly configured **SSID** and **Reg Domain** information.
   – Ensure the **Telephony Protocol** menu option is set to **36**. This ensures the handset will check for the proper SRP files each time it powers on.
   – Ensure security settings are properly programmed.
4. Power cycle the handset.
5. 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.

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 on, it will check with the TFTP server to ensure it has the proper software version.

4

SIP Integration Factors

CODECs

The SpectraLink 8020/8030 Wireless Telephones are compatible with the G.711µ-law and G.711a-law codecs. There is no setting required on the handset.

DHCP

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
15	Domain name
66	TFTP server
151	SpectraLink 8000 SVP Server
152	SpectraLink 8000 OAI Gateway
siaddr	Boot server or next server

DNS

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 user-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 and SpectraLink 8000 SVP 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.

5

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. Both of these files must be written locally.

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.

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. See Chapter 2 *SpectraLink 8020/8030 Wireless Telephone Configuration*, section *The Admin Menu*, subsection *SIP Registration*.
- Username parameters are: alphanumeric, no spaces, no punctuation, case is ignored, 1-16 characters.

- Generic file information should contain proxy server information and other SIP system data.
- Specific file information 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" or by enclosing the string in quotes (").
- If necessary, other special characters may be included by using a hexadecimal representation: (%hh) where hh is the representation of the character.
- Lines may appear in any order although maintenance may be simplified by preserving the order in the supplied example file.

Program 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_3001.cfg or 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.	n = 1, 2, 3 xxx.xxx.xxx.xxx = IP4 address pppp = port (optional, 5060 is the default) proxyname = computer 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 the proxy does not act as a domain server.
PROXYn_TYPE	Inter-Tel SER Asterisk[2]	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. Only the three shown values may be used. Omit this parameter if the value is not one of these recognized settings.
PROXYn_ KEYPRESS_2833	enable disable	Controls generation of in-stream RFC2833 formatted key press events.	n = 1, 2, 3
PROXYn_ KEYPRESS_INFO	enable disable	Controls generation of SIP INFO requests to the SIP server for keypress events.	n = 1, 2, 3
PROXYn_HOLD_IP0	enable disable	Controls setting of the media stream IP destination address to 0 (zero) when a call is put on hold.	n = 1, 2, 3 Use for compatibility with older SIP servers that may not recognize newer stream attribute parameters for HOLD status.

[2] For up-to-date Asterisk voicemail commands, go to: http://www.voip-info.org/wiki/index.php?page=Asterisk+cmd+VoiceMailMain

Parameter	Value	Description	Notes
PROXYn_PRACK	enable disable	Enables reliable provisional responses to INVITE requests	n = 1, 2, 3
PROXYn _ MAIL_SUBSCR	name@xxx.xxx.xxx.x xx or sip:name@domain	Contact to whom the handset should subscribe for mail notification.	n = 1, 2, 3 name = mail server contact name. xxx.xxx.xxx.xxx = IP4 address. Domain where the mail resides[3].
PROXYn_ MAIL_ACCESS	name@xxx.xxx.xxx.x xx or sip:name@domain	Contact to whom the handset should invite to access the mail center.	See above.
PROXYn _ MAIL_NOTIFY	name@xxx.xxx.xxx.x xx or sip:name@domain	Contact from whom mail notification originates.	[4] [5]
AUTH [see warning note below]	username;password	Credentials. In general credentials are needed for each registered line.[6].	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 e.g. g711u, g711a, g729	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.

---

3 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.

4 Values as above.

5 For Inter-Tel SIP servers, both PROXYn_MAIL_SUBSCR and PROXYn_MAIL_NOTIFY parameters must be set in the configuration file.

6 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
LINEn	username	The dial # or name. All LINEn user names should be unique for a given LINEn_ PROXY. This may be enforced in future software revisions.	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_PROXY can be omitted if the line is not to be registered and you wish to do direct phone to phone calls.
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.
FAVORITE	Dialstring;identifier;LINEn	Phonebook list of numbers accessible from the Favorites menu.	Up to 15 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. LINEn = can be omitted if dialing can be done on any registered line. Usually omitted but if present, the number will be dialed on the programmed line.

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

# 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 = 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
```

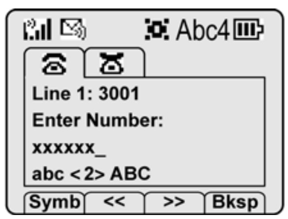
```
# Two lines may map to the same extension to allow second incomming calls.
LINE2         = 3001
LINE2_PROXY   = 1
LINE2_CALLID  = Brady, Marsha

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
```

6

Using the SIP Handset

The Handset Display

When active, the handset screen will display either a call status screen or one of several menu screens. The call status screen has the following format:



This example shows two call tabs indicating that two calls are in progress. The un-selected call tab indicates that we have put another call on hold. The call-status icon for the selected call indicates that this call is being dialed. The text indicates the selected call is on line 1, extension 3001. Enter Number indicates that the handset is ready to be dialed. Once this call is connected, the connected party's information will appear on the third line, and the fourth line contains help or error messages, as appropriate. The softkeys during this action offer text editing functions.

System icons

Indicator	Function
	The signal-strength icon indicates the strength of the signal and can assist the user in determining if the handset is moving out-of-range.
	The voicemail icon is activated when a new voicemail message is received if the feature is supported by the phone emulation.
	The battery icon indicates the amount of charge remaining in the Battery Pack. When only one level remains, the Battery Pack needs to be charged.
	The speakerphone icon displays when the speakerphone is active.

Indicator	Function
	Up and down arrows are displayed when the menu has additional options above or below. Left or right arrows are displayed during editing when the cursor may be moved left or right.
	The Push-to-talk (PTT) ring icon. A PTT call is coming in.
	The priority PTT ring icon. A call is coming in on the priority PTT channel. This call will override any other.
	Location Service icon: indicates the Ekahau Real-Time Location System (RTLS) is enabled.
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, an 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 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.
XXXX	During character entry, Indicates current data entry symbol mode.

Call status icons

Indicator	Function
	On-hook icon, Solid when idle. Flashes while in standby mode to indicate that at least one call is still active or on hold. Flashing when incoming call is ringing.
	Off-hook icon. Solid when a call is being dialed.
	Hold icon. Call is on hold
	Transfer icon. Call is in the process of being transferred
	Audio flowing icon. Audio is flowing both ways on a call.
	Audio receive-only icon. Locally muted (flash) or far end hold with no music on hold.
	No audio icon. No audio is flowing. Call is terminating or far end hold with audio disable.

NavOK functions

The **NavOK** key acts as a fifth softkey with implicit functionality as follows:

State	NavOK key function
Dialing	Place phone call.
Answering	Answer a phone call
Holding	Resume audio.
Displaying menu	Select the highlighted menu option.
Displaying call status	Resume audio on the currently selected call and place previous call on hold. If the selected call is ringing, the call will be answered.
Entering login name or login password	Save name or password and proceed with startup.

Softkeys

Softkey	Name	Displayed during…	Press to…
<<	Cursor backward	Entering a dial number.	Move the cursor back one position.
>>	Cursor forward	Entering a dial number.	Move the cursor forward in alphanumeric mode, if the cursor is at the end of the line, adds a space character.
Answ	Answer	Incoming call 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.
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.
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.

Softkey	Name	Displayed during…	Press to…
Hold	Hold	In an active call.	Place the call on hold. The line status shows 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 dialed[7].	Initiate a call to the specified message center contact address for retrieval or administration of voicemail.
Mute	Toggle muting	In an active call.	Toggle audio transmission to the far end. The line status shows when not muted or 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.
Resm	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.
Symb	Symbols	Entering a username or password. Entering the digits of a number.	Select the set of symbols available on the keypad while entering data.

[7] 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.

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** key to exit without starting a new call. Press the **More** softkey to page through additional items on the Line menu.

Symbol menu

The symbol menu allows you to change the set of characters available for data entry through multiple key presses of the dial pad keys.

While dialing a number or entering login information, press the **Symb** softkey to view a menu of possible sets of characters that can be entered using multiple key presses of the dial pad keys. Normally, a simple numeric mode is selected; selecting other symbol modes allows convenient access to the complete printable US ASCII character set. The following table shows what characters are available through repeated key presses in various symbol modes.

<table>
<tr><th>Key</th><th>Number</th><th>English</th><th>Number + English</th><th>Punctuation</th></tr>
<tr><td>1</td><td>1</td><td>1 ; : / \ ! ‘</td><td>1</td><td>@ : 1</td></tr>
<tr><td>2</td><td>2</td><td>a b c 2 A B C</td><td>2 A B C a b c</td><td>; , 2</td></tr>
<tr><td>3</td><td>3</td><td>d e f 3 D E F</td><td>3 D E F d e f</td><td>& | ` ~ 3</td></tr>
<tr><td>4</td><td>4</td><td>g h I 4 G H I</td><td>4 G H I g h i</td><td>( ) 4</td></tr>
<tr><td>5</td><td>5</td><td>j k l 5 J K L</td><td>5 J K L j k l</td><td>< > 5</td></tr>
<tr><td>6</td><td>6</td><td>m n o 6 M N O</td><td>6 M N O m n o</td><td>{ } 6</td></tr>
</table>

Key	Number	English	Number + English	Punctuation
7	7	p q r s 7 P Q R S	7 P Q R S p q r s	[ ] 7
8	8	t u v 8 T U V	8 T U V t u v	‘ “ \ 8
9	9	w x y z 9 W X Y Z	9 W X Y Z w x y z	^ _ 9
0	0	@ - _ 0 = , < >	0 - _	[space] 0
*	. *	. $ * & % + ( )	* .	* . = + / -
#	@ *	[space] , ( ) |	# [space]	# ! ? $ %

Favorites menu

The Favorites menu assists you in dialing by providing access to a predefined list of dial numbers. The predefined list can include either complete dial numbers for named parties or partial numbers that need additional data entry. This might be the case, for 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.

While in a dialing state, press 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 the **NavOK** key or **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

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

Transfer

Do Not Disturb

Set/Clear Forward

<OAI>

<OAI>

<OAI>

Items on this menu are accessible through navigation and selection keys or through short-cut keys as displayed with the menu items. OAI functions are automatically added as items at the end of this menu when defined on an OAI server.

Handset Operation

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 press the **Cfg** softkey, then press **NavOK**.
Place a call	1 To dial a number, follow any one of these sequences: » Press the **START** key, wait for a dial tone, then dial the number. Then press either **START** again or **Nav OK**. » Dial the number and then press the **START** key or **Nav OK**. » Press the **Spkr** softkey, then dial the number. Then press **START** or **Nav OK**. » Press the **START** key; press the **Favr** softkey; use the **Nav** ▲▼ keys to select the number or user from the list; press **Nav OK** to dial the number. 2 Listen for the ring to indicate the alerting of the called party. Note: **Line 1** is the default line.
Place a call from Favorites menu	1 Press **START** or **Spkr**. 2 Listen for dial tone. 3 Press the **FAVR** softkey. 4 Use the **Nav** keys to navigate to the desired entry. 5 Select the entry by pressing **NavOK**. 6 Press **START** or **NavOK** to place the call.

If you want to…	Then…
Place a second call	1 To get a dial tone for the second call, press **LINE** and **NavOK**. a. If the handset has multiple lines, press **LINE** + [the line number]. b. Press **NavOK**. • The first call is automatically placed on hold. • The second call appears in a new call tab. 2 Dial the number to place the second call. 3 Use the **Nav** ◄► keys to toggle between calls.
Place a call on a different line	1 Press the **LINE** key. 2 Navigate to the desired line and press **NavOK** or press the number of the line. 3 Dial the number. 4 Press **START** or **NavOK** to initiate the call.
Answer a call	• Press **START**, and hold the handset to your ear. • Press **NAV OK**, and hold the handset to your ear. • Press the **Answ** softkey and hold the handset to your ear. • Press the **Spkr** softkey and speak towards the handset.
Reject a call	Press the **Rej** softkey to reject the call and allow the SIP server to redirect the call elsewhere, usually to voicemail.
Answer a call on a second line	If another call comes in on a different line, a new call icon flashes and a tone sounds in the audio stream until the call is answered, the first call is terminated, the caller hangs up, or the call transfers to voicemail. 1 To view the caller ID of the incoming call without interrupting the active call, press **Nav** ►. The original call's audio remains active. The display now shows information about the incoming call. 2 Press **NavOK**, **START** or **Answ** to place the current call on **Hold** and answer the second call, or Press **Rej** to redirect the incoming call to voicemail or as otherwise programmed.
Navigate among call tabs	Use **Nav**► and **Nav**◄.

<table>
<tr><th>If you want to…</th><th>Then…</th></tr>
<tr><td>Forward calls</td><td>Calls may be forwarded on a per-line basis. The Line menu display will indicate if a line is forwarded by a > character followed by the destination address.
When a line is forwarded, the standby screen displays the forwarding status, unless the do-not-disturb feature is enabled.
If your handset has multiple lines, set or clear forwarding for each line individually.
Press START or press Line and a line number to select the line that you want to be forwarded.
Press FCN and select the Set/Clear Forward item from the FCN menu using a digit key or NavOK key.
Enter the forwarding number for the indicated line.
Press NavOK to save.
To clear call forwarding for a given line #, again press START or select the line #. Again select the Set/Clear Forward item from the menu.</td></tr>
<tr><td>Clear forwarding on a line</td><td>Press START or select the line number. Select the Set/Clear Forward item from the menu.</td></tr>
<tr><td>Listen to voicemail</td><td>Press START.
Press Msg softkey, or
Dial your voice message system number
Press START or Nav OK.</td></tr>
<tr><td>Do Not Disturb
(reject all incoming calls)</td><td>Press START.
Press FCN.
Select the Do-Not-Disturb item from the menu and press NavOK.
When the do-not-disturb feature is enabled, the standby screen will display Do Not Disturb. The handset will not ring for new incoming calls.</td></tr>
<tr><td>Clear Do Not Disturb</td><td>Press START.
Press FCN.
Select the Do-Not-Disturb item from the menu and press NavOK .</td></tr>
<tr><td>Redial the last number you dialed</td><td>Press START.
Press the Redl softkey.
Verify that the displayed number is the one you want and the line number is valid.
Press NavOK or START.</td></tr>
</table>

If you want to…	Then…
Activate installed custom applications (registered OAI application on an OAI server)	1 Press **START** or **Spkr**. 2 Press **FCN**. 3 Navigate to the desired custom application using **Nav** ▲▼ keys or the **More** softkey. 4 Select the application using **NavOK** or shortcut key.
Transfer a call (blind)	1 While in a call, press **FCN** and then select the transfer item by shortcut key or **NavOK**. (The current call will be placed on hold and it will be marked with a transfer icon ✕. A new call will be started and you will hear a dial tone.) 2 Dial 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** and select the **Transfer** option again. The marked call will be transferred to the number that you have entered. 4 Press the **END** key to return to standby. If the transfer fails, you will see an error message and you can then pick up the original call by navigating to the marked call and pressing **NavOK**.
Transfer a call (consulted)	1 While in a call, press **FCN** and then select the transfer item by shortcut key or **NavOK**. (The current call will be placed on hold and it will be marked with a transfer icon ✕. A new call will be started and you will hear a dial tone.) 2 Dial 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 **NavOK** or **START** to establish a second call. 4 Inform the person on the other end that you would like to transfer the call. 5 Press **FCN** and select the **Transfer** option to transfer the call. 6 Press the **END** key to return to standby.
Transfer an active call to a call on Hold	1 Press **FCN** and then select the **Transfer** option by pressing **NavOK**. The current call will be placed on hold and it will be marked with a transfer icon ✕. 2 Navigate to the second call (already on hold). 3 Press the **Resm** softkey and tell the other party that the call will be transferred. 4 Press **FCN** and select the **Transfer** option by pressing **NavOK**. 5 Press the **END** key to return to standby.
Silence the ringing	Press the **END** key to silence the external speaker ring and convert to in-ear speaker ringing. External speaker ringing will resume when the next incoming call is received while the handset is in standby mode.

If you want to…	Then…
Change the ring volume	Press the up/down volume buttons on the side of the handset during ringing, or while the handset is in standby mode.
Adjust the speaker volume	While speakerphone is active, press the up/down volume buttons on the side of the handset during the call.
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.
Mute/Unmute a call	Press the **Mute** softkey. When the handset is muted, the audio flowing icon ⇆ changes to the audio receive-only icon ⇇. Press the **Mute** softkey again to restore audio pickup.
End a call	Press the **End** softkey to maintain the active mode and view the active calls. Press the **END** key on the keypad to return to the standby mode.
Change the profile	Press the **Prof** softkey and use the **Nav** keys to select a new profile while in standby mode. The selected profile is marked with an asterisk (*****).
Open Config menu	Press the **Cfg** softkey from 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 within that period.
Resume a call on hold from standby.	Press the **Resm** softkey. If more than one call is on hold, use the **Nav** ◄ ► keys to select the call you wish to resume and press the **Resm** softkey or **NavOK**.

7

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 11 *Troubleshooting* for corrective action.

8

Diagnostic Tools

Run Site Survey, **Diagnostics Enabled** and **Syslog Mode** are three diagnostic tools provided to assist the wireless LAN administrator in evaluating the functioning of the SpectraLink 8020/8030 Wireless Telephone and the system surrounding it. Diagnostic Tools are enabled in the Admin menu.

Run Site Survey

Site survey is 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 SSID. The information available through the site survey includes:

- SSID
- 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 test is started, it is by default in "single SSID" mode. When the **Any** soft key is pressed (softkey A) all APs, regardless of SSID, are displayed and the softkey changes to say **MyID**. Pressing the **MyID** soft key will revert to the "single SSID" mode and change the softkey back to **Any**.

The display would look like the following for the multiple AP 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 SSID 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.

```
ab7bc8  -28  02  100
2ae578  -48  06  200
2ae596  -56  11  100

Any               Detl
```

When the **Any** SSID mode is selected, the summary display contains the first six characters of the APs SSID instead of the beacon interval as in the example below.

```
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 will move between AP indices.

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

Where:

- i – index of selected AP (value will be from 0 to 3 inclusive)
- 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 – 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 racing 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 them. See Chapter 11 *Troubleshooting*, section *Access Point Problems* for more information.

Diagnostics Enabled

Diagnostics is used to evaluate the overall quality of the link between the handset, AP, and infrastructure side equipment, such as IP PBX, SpectraLink 8000 SVP Server, and gateways. Unlike **Site Survey**, **Diagnostics** is used while the functional code is running, and during a call.

When **Diagnostics** is enabled in the Admin menu, the handset can display diagnostic screens any time it is in active mode.

The display of information is instigated by pressing the **Nav◄** or **Nav►** key. Only four of the diagnostic counters listed below can be shown at a time. Pressing the **Nav** keys multiple times will cycle through the various counters and the normal off-hook (IP-PBX) display. The numeric icon at the top of the display indicates what screen number is being displayed. For example: The first time the **Nav** key is pressed, the **1** icon is shown, and the first four counters are displayed. The next time it is pressed, the **2** icon is shown, and the next four counters are displayed. The counters will be cycled through in this fashion until there are no more counters to be displayed. After all the counters have been displayed, the screen returns to the normal off-hook IP-PBX screen.

The information provided by **Diagnostics** includes:

Screen 1

- 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)

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

Screen 2

- Jitter – average error or "wobble" in received packet timing, in microseconds
- Last successful transmit data rate (LastRate)
- Gateway type (GatewyType)

```
J i t t e r              n n n n n
L a s t R a t e          n n n n n
G a t e w y T y p e      m n e m o
```

Where:

mnemo – a mnemonic that indicates what type of gateway is being used

- 11Mb – this system can run at full speed

Screen 3

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 from the current AP or a mnemonic for the reason code indicating why the handset didn't hand off to this other AP

```
C : m m m m   c h   - s s   a i d
1 : m m m m   c h   - s s   m n e m
2 : m m m m   c h   - s s   m n e m
3 : m m m m   c h   - s s   m n e m
```

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

- 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)

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

Screen 5

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

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

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.
- **Errors** – causes the handset to log only events that we consider to be an error (see below).
- **Events** – logs all errors plus 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
Error Handling Mode	Yes	Yes	Yes

All syslog messages will include:

- Date and time (to 1/100th of second) since handset power on (The handset time is set when it is powered on to Jan-1 00:00.00 GMT adjusted. If it has obtained a time from the network time server, that time will display instead.)
- Handset's MAC address
- Handset's IP address
- 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%)

9

Certifying the Handsets

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

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 11 *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 Chapter 8 *Diagnostic Tools*, section *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 -70 dBm in all areas) 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.

10

Software Maintenance

The SpectraLink 8020/8030 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** button. **Firmware Version** is also an option on the Config menu.

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 8020/8030 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 configuration. 81 = File put into memory did not 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.

11

Troubleshooting

On occasion, you may run into transmission problems due to any number of factors originating from the wireless LAN. SpectraLink 8020/8030 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.

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 will recover the call if the user moves back into range within a few seconds.

Capacity

In areas of heavy use, the call capacity of a particular AP may be filled. If this happens, the user will hear three chirps from the handset. The user can wait until another user terminates a call or move within range of another AP and try the call again. If a user is on a call and moves into an area where capacity is full, the system attempts to find another AP. Due to range limitations, this may be the same as moving out of range.

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 *Configuration Guide* for the AP in use at the site.

Handset Status Messages

SpectraLink 8020/8030 Wireless Telephone status messages provide information about the SpectraLink 8020/8030 Wireless Telephone's communication with the AP and host telephone system. The following table summarizes, in alphabetical order, the status messages.

Message	Description	Action
3 chirps (audio)	Handset is not able to communicate with the best AP, probably because that AP has no bandwidth available.	None. This is only a warning, the call will hand off to the best AP once it becomes available.
Address Mismatch	Handset software download files are incorrect or corrupted.	Download new software from the Polycom 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 Guide*. 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 Guide*. 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 Guide*. 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 Guide*. 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.	Download new software from the Polycom website per *Software Maintenance*.

Message	Description	Action
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.
Bad Phintl File	Handset software download files are incorrect or corrupted.	Download new software from the Polycom website per *Software Maintenance*.
Bad Program File	Handset software download files are incorrect or corrupted.	Download new software from the Polycom website per *Software Maintenance*.
Bad SIP TFTP IP	A bad unicast address has been entered for the SIP TFTP server in static entry mode.	Re-enter the correct IP address in the administrative menus for static IP addresses.
(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. 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 Management 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 Engineers recommend 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 (at least one renewal succeeded).	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.

Message	Description	Action
DL Not On Sector	Handset software download files are incorrect or corrupted.	Download new software from the Polycom website per *Software Maintenance*.
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 require 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 turn the handset off during this operation.
Files Too Big	Handset software download files are incorrect or corrupted.	Download new software from the Polycom website 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.
Initializing SIP	The handset is performing a power-on initialization of the SIP application. The phone is initializing its data structures and attempting to access the SIP TFTP server and download the SIP configuration files.	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.
Multiple GW Res	More than one SpectraLink 8000 SVP Server has responded.	Caused by two or more handsets sharing the same IP address. Assign unique IP addresses to each handset.

Message	Description	Action
Multiple SVP Reg yyy.yyy.yyy.yyy	y…y = SVP IP address Handset received responses from multiple SVP Servers; displays IP address of one responding SVP Server.	This can happen if the handset has been reconfigured to use a different SVP server and then powered on before the previous server has had time to determine that the handset is no longer connected to it. The problem should go away after about 30 seconds.
Must Upgrade SW!	Handset software is incompatible with hardware.	Download new software from the Polycom website per *Software Maintenance*.
Net Busy xxxxxxxxxxxx	x…x = AP MAC address. Handset cannot obtain sufficient bandwidth to support a call; displays MAC of failing AP.	Try the call again later.
No DHCP Server	Handset is unable to contact the DHCP server.	Check that DNCP 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.	Let handset come completely up. Statically configure an ESSID in the Admin menu.
No Func Code	Handset software download files are incorrect or corrupted.	Reconfigure the handset to gain access to the WLAN and download new code.
No Host IP	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 IP.	Check the IP address of the handset and reconfigure if required.
No SIP DHCP	DHCP is configured but no valid SIP option 43 was found.	Check DHCP configuration for option 43 and reconfigure if required.
No Net Access	Cannot authenticate / associate with AP.	Verify the AP configuration. 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.
	No ESSID: Auto-learn not supported (or) incorrect ESSID.	Verify the ESSID of the wireless LAN and enter or Autolearn it again if required.
	AP does not support appropriate data rates.	Check the AP configuration against Configuration Guide for AP.
	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.

Message	Description	Action
	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 per *Configuration Guide*. Try Site Survey mode to determine a more specific cause.
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 of the handset.
No SIP TFTP IP	No IP address has been entered for the SIP TFTP server.	In static IP mode the SIP TFTP server address must be entered in the administrative menus.
No SIP user file	The phone is attempting to download a SIP configuration file from the SIP TFTP server. A file must be available for the username that was entered either in the admin menus or as requested at power-on.	Ensure a SIP configuration file is available on the SIP TFTP server and is named as specified (sip_username.cfg).
No SVP IP	The handset is configured for "Static IP" (as opposed to "use DHCP"), and no valid SpectraLink 8000 SVP Server address has been entered.	Enter a valid SpectraLink 8000 SVP Server IP address in the configuration setting or change to "use DHCP."
No SVP Response yyy.yyy.yyy.yyy	y…y = SVP Server IP address. Handset has lost contact with the SVP Server.	This may be caused by bad radio reception or a problem with the SpectraLink 8000 SVP Server. The handset will keep trying to fix the problem for 20 seconds, and the message may clear by itself. If it does not, the handset will restart. Report this problem to the system administrator if it keeps happening.
No SVP Server	Handset can't locate SpectraLink 8000 SVP Server. SpectraLink 8000 SVP Server is not working. No LAN connection at the SpectraLink 8000 SVP Server.	IP address configuration of SpectraLink 8000 SVP Server is wrong or missing. Check error status screen on SpectraLink 8000 SVP Server. Verify SpectraLink 8000 SVP Server connection to LAN.
No SVP Server No DNS Entry	Handset unable to perform DNS lookup for SVP Server, server had no entry for SVP Server.	The network administrator must verify that a proper IP address has been entered for the SVP Server DHCP option.

Message	Description	Action
No SVP Server No DNS IP	Handset unable to perform DNS lookup for SVP Server, no IP address for DNS server.	The network administrator must verify proper DHCP server operation.
No SW Found	A required software component has not been identified.	Check that the handset license type has a corresponding entry in the slnk_cfg.cfg file. Check that the pd11sid.bin and pi110000.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	The far end of a call has hung up.	Hang up the near end.
Press END to quit	The handset is waiting to acquire bandwidth required for voice communication.	Press **END** or wait until bandwidth is available.
Registering	The handset has completed initialization of the SIP application and is attempting to register lines to the SIP proxy servers.	If registrations are failing, the phone can stay in this state for a considerable length of time. After the phone leaves this state, press the **LINE** key to view what lines have failed to register. Ensure usernames and passwords have been entered in administrative menus for registrations that have failed and that proxy information is correct in the SIP configuration files.
RTP Open Failed	The handset attempted to open an RTP port for audio but was unsuccessful.	Verify that SpectraLink 8000 SVP Server capacity has not been exceeded.
Select License	The correct protocol has not been selected from the license set.	Using the Admin menu, select one license from the set 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.
SIP Login	Prompt for login information – username and password.	At power-on initialization, no username was detected in the admin menu items for SIP registrations. Enter a valid username and password for an existing SIP configuration file.
Skt Open Fail	Socket open fail. Occurs when the handset attempts to open a connection to the proxy server but fails.	Verify that SpectraLink 8000 SVP Server capacity has not been exceeded.

Message	Description	Action
Service Rej.	The SpectraLink 8000 SVP Server has rejected a request from the handset.	The handset will restart and attempt to re-register with the SpectraLink 8000 SVP Server, which should fix the problem. Report to your administrator if it keeps happening.
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.
SVP Service Rej.	The SpectraLink 8000 SVP Server has rejected a request from the handset.	The handset will restart and attempt to re-register with the SVP Server, which should fix the problem. Report to your administrator if it keeps happening.
System Busy yyy.yyy.yyy.yyy	y…y = SVP Server IP Address. SVP Server has reached call capacity.	All call paths are in use, try the call again in a few minutes.
System Locked (with Busy Tone)	SpectraLink 8000 SVP Server is locked.	Try call again later, system has been locked for maintenance.
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 phintl file.	Download new software from the Polycom website per *Software Maintenance*.
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 internally updating its software images.	None. The handset may do this briefly after a download. This is informational only.

Message	Description	Action
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 turn the handset off during this operation.
Wait for bandwidth	The phone is waiting for bandwidth sufficient for voice communication.	No action required. You will have the option of pressing **END** to abort the phone call.
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.

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 both 2.4 and 5 GHz wireless LANs.

Domain Identifier	802.11 Mode	Band	Channels	DFS Required?	Max. Power Limit (peak power)	Countries
01	g only b & b/g mixed	2.4000 – 2.4745 GHz	1 – 11	n/a	100mW (+20dBm)	US Canada Mexico Brazil
	a	5.1500 – 5.2500 GHz	36 – 48	No	50mW (+17dBm)	
		5.2500 – 5.3500 GHz	52 – 64	Yes	100mW (+20dBm)	
		5.4700 – 5.7250 GHz	100 – 140	Yes		
		5.7250 – 5.8250 GHz	149 – 161	No		
02	g only b & b/g mixed	2.4000 – 2.4845 GHz	1 – 13	n/a	100mW (+20dBm)	Europe Australia New Zealand
	a	5.1500 – 5.2500 GHz	36 – 48	No		
		5.2500 – 5.3500 GHz	52 – 64	Yes		
		5.4700 – 5.7250 GHz	100 – 140	Yes		
03	g only b & b/g mixed	2412.0 – 2472.0 GHz	1 – 13	n/a	100mW (+20dBm)	Japan
	a	5.1500 – 5.2500 GHz	36 – 48	No		
		5.2500 – 5.3500 GHz	52 – 64	Yes		
04	g only b & b/g mixed	2.4000 – 2.4835 GHz	1 – 13	n/a	100mW (+20dBm)	Singapore
	a	5.1500 – 5.2500 GHz	36 – 48	No		
		5.2500 – 5.3500 GHz	52 – 64	Yes		
05	g only b & b/g mixed	2.4000 – 2.4845 GHz	1 – 11	n/a	100mW (+20dBm)	Korea
	a	5.1500 – 5.2500 GHz	36 – 48	No		
		5.2500 – 5.3500 GHz	52 – 64	Yes		
		5.4700 – 5.6500 GHz	100 – 124	Yes		
		5.7250 – 5.8250 GHz	149 – 161	No		

Domain Identifier	802.11 Mode	Band	Channels	DFS Required?	Max. Power Limit (peak power)	Countries
06	g only b & b/g mixed	2.4000 – 2.4745 GHz	1 – 11	n/a	100mW (+20dBm)	Taiwan
	a	5.2500 – 5.3500 GHz	52 – 64	Yes		
		5.4700 – 5.7250 GHz	100 – 140	Yes		
		5.7250 – 5.8500 GHz	149 – 165	No		
07	g only b & b/g mixed	2.4000 – 2.4845 GHz	1 – 13	n/a	100mW (+20dBm)	Hong Kong
	a	5.1500 – 5.2500 GHz	36 – 48	No	50mW (+17dBm)	
		5.2500 – 5.3500 GHz	52 – 64	Yes	100mW (+20dBm)	
		5.4700 – 5.7250 GHz	100 – 140	Yes		
		5.7250 – 5.8250 GHz	149 – 161	No		

Index