



LS2000

Handset Speakerphone

Models: IPVH01, IPVH02, IPVH03, IPVH04

Admin Guide

Installation | Configuration | Support | Maintenance | Use



Code Blue

800.205.7186 ● www.codeblue.com

WARNING

ONLY QUALIFIED PERSONNEL SHOULD INSTALL THESE UNITS. THE INSTALLATION SHOULD CONFORM TO ALL LOCAL CODES. IN SOME COUNTRIES, A CERTIFIED ELECTRICIAN MAY BE REQUIRED.

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Table of Contents

Section	Section Title	Page #
2	Introduction	5
	• LS2000 Model Photos	5
3	Safety Information	6
4	Getting Started	7
5	Power Requirements	8
6	Installation Instructions	12
	• Tools Needed	12
	• Basic Flush Mount Instructions	12
	• Basic Surface Mount Instructions	12
7	How to Update Connectors	13
8	Wiring Diagrams	15
9	Schematics	16
10	Setting Up Your LS2000	18
	• Determine IP Address	18
	• Default Login Info	19
	• Network - General Settings	19
	• Accounts	21
	• SIP Settings	22
	• Manage Accounts	23
	• System Status	24
	• Audio	25
	• Multicast Audio	26
	• General Events	26
	• Action Scripts	27
	• Place A Call Action	28
	• Control AUX Output	30
	• Play Message	30



Table of Contents (continued)

Section	Section Title	Page #
	• Recordings	33
	• Administration General	34
	• Date & Time	35
	• Diagnostics	36
	• Firmware	38
	• Hardware Settings	41
11	In Call Commands	43
12	Compatibility	44
13	Configuring for Cisco Unified Call Manager	45
14	Troubleshooting	49
15	Remote Mount Beacon/Strobe Installation	50
	• S-1000 & S-1050 Strobe Operation	51
	• Strobe Programming	52
16	Maintenance Schedule	53
17	Warranty	55
18	Download Information	56
19	Legal & Regulatory Information	57

2 Introduction

Thank you for choosing the Code Blue **LS2000** full duplex VoIP handset device for indoor and outdoor applications. This telephone is part of our Emergency Signaling group of products built to meet the latest regulations, withstand the harshest elements and be proactive solutions for when you need them most. This guide provides basic and advanced configuration information for obtaining the best performance with the LS2000 phone



Handset Phone



Handset w/ Volume Button



Handset w/ Keypad




Handset w/ Keypad and Volume Button

Call Privacy Laws

Some states require all parties to be aware that they are being recorded. Code Blue phones offer the ability to play a message stating that the caller is being recorded and giving the caller the option to continue or end the recorded call.

3 Safety Information

HAZARD LEVELS LEGEND

DANGER	Indicates a hazardous situation which, if not avoided, <i>will</i> result in death or serious injury.
WARNING	Indicates a hazardous situation which, if not avoided, <i>could</i> result in death or serious injury.
CAUTION 	Indicates a hazardous situation which, if not avoided, could result in minor or moderate injury.
NOTICE	Indicates a situation which, if not avoided, could result in damage to property.
IMPORTANT	Indicates significant information that is essential for proper product functionality.
<i>NOTE</i>	Indicates useful information that helps get the most out of a product.

Safety Instructions

WARNING • Code Blue products shall be installed by trained professionals. The installation should conform to all local codes. In some countries, a certified electrician may be required.

- NOTICE** • When transporting a Code Blue product, use the original packaging or equivalent to prevent damage to the product.
- Code Blue products shall be used in compliance with local laws and regulations.
 - Store the Code Blue product in a dry and ventilated environment.
 - Do not install the product on unstable brackets, surfaces or walls.
 - Use only applicable tools when installing Code Blue products.
 - Do not use chemicals, caustic agents, steel wool or aerosol cleaners other than those tested and recommended by Code Blue.
 - Use only accessories that comply with technical specifications of the product. These can be provided by Code Blue or a third party.
 - Use only spare/replacement parts provided by or recommended by Code Blue.

Transportation

- NOTICE** • When transporting a Code Blue product, use the original packaging or equivalent to prevent damage to the product.



4 Getting Started

This guide contains information and a general overview for **LS2000**, including applications, installation, and wiring.

LS2000 Features

- Handset
- SIP 2.0 Protocol Support
- 3 Ethernet Ports
- 3 auxiliary N.O. input contact closures
- 3 auxiliary N.O. output contact closures
- NEMA 3S Compliant
- Peer-to-peer audio communication
- Multicast audio
- Multiple ADA compliant lettering choices with Braille

5 Power Requirements

The following table includes **LS2000** and ALL OTHER Code Blue Devices and Enclosures for reference

Faceplates	Voltage	Max Current	Max Watts	Norm Current	Norm Watts	KWHrs
IA4100	24V AC	0.40	9.60	0.22	5.28	0.13
	12V DC	0.90	10.80	0.39	4.68	0.11
	24V DC	0.90	21.60	0.39	9.36	0.22
IP5000	24V AC	0.10	2.40	0.07	1.68	0.04
	12V DC	0.19	2.28	0.15	1.80	0.04
	24V DC	0.19	4.56	0.15	3.60	0.09
Centry	12VDC	0.50	6.00	0.38	4.56	0.11
LS1000/LS2000	12V DC	0.50	3.60	0.40	4.80	0.12
Lights	Voltage	Max Current	Max Watts	Norm Current	Norm Watts	KWHrs
s-1000/S-2000 LED Strobe	24V AC	0.28	6.72	0.22	5.28	0.13
	12V DC	0.26	3.12	0.24	2.88	0.07
	24V DC	0.26	6.24	0.24	5.76	0.14
A-700 Area Light	24V AC	1.80	43.20	0.83	19.92	0.48
	12V DC	2.68	32.16	0.38	4.56	0.11
	24V DC	2.68	64.32	0.38	9.12	0.22
S-1050 LED Strobe W/ Photocell	24V AC	0.28	6.72	0.22	5.28	0.13
	12V DC	0.27	3.22	0.24	2.88	0.07
	24V DC	0.27	6.43	0.24	5.76	0.14
LED Light Bar	24V AC	0.04	0.96	0.04	0.96	0.02
	12VDC	0.04	0.48	0.04	0.48	0.01
	24V DC	0.04	0.96	0.04	0.96	0.02
WM180 PAS With LED Strobe	12-24V DC	7.30	175.20	2.10	50.40	1.21



Models With IA4100 Faceplate	Voltage	Current	Watts	KWHrs
CB 1-e	24V AC	0.48	11.52	0.28
	12VDC	0.67	8.04	0.19
	24V DC	0.67	16.08	0.39
CB 1-s	24V AC	1.31	31.44	0.75
	12V DC	1.05	12.60	0.30
	24V DC	1.05	25.20	0.60
CB 5-s	24V AC	0.48	11.52	0.28
	12V DC	0.67	8.04	0.19
	24V DC	0.67	16.08	0.39
CB 9-s	24V AC	0.26	6.24	0.15
	12V DC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 2-e	24V AC	0.44	10.56	0.25
	12VDC	0.63	7.56	0.18
	24V DC	0.63	15.12	0.36
CB 2-a	24V AC	0.48	11.52	0.28
	12V DC	0.67	8.04	0.19
	24V DC	0.67	16.08	0.39
CB 2-s	24V AC	1.31	31.44	0.75
	12V DC	1.05	12.60	0.30
	24V DC	1.05	25.20	0.60
CB 2 w/ Audio Paging	12-24V DC	6.44	154.56	3.71
CB 4-s	24V AC	0.22	5.28	0.13
	12V DC	0.39	4.68	0.11
	24V DC	0.39	9.36	0.22
CB 4-r	24V AC	0.26	6.24	0.15
	12V DC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 4-u	24V AC	0.26	6.24	0.15
	12V DC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 6-F & CB 6-S	24V AC	0.22	5.28	0.13
	12V DC	0.39	4.68	0.11
	24V DC	0.39	9.36	0.22
CB RT	24V AC	0.48	11.52	0.28
	12V DC	0.67	8.04	0.19
	24V DC	0.67	16.08	0.39



Models With IP5000 Faceplate	Voltage	Current	Watts	KWHrs
CB 1-e	24V AC	0.33	7.92	0.19
	12VDC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 1-s	24V AC	1.16	27.84	0.67
	12V DC	0.81	9.72	0.23
	24V DC	0.81	19.44	0.47
CB 5-s	24V AC	0.33	792.00	0.19
	12V DC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 5-s	24V AC	0.11	2.64	0.06
	12VDC	0.19	2.28	0.05
	24V DC	0.19	4.56	0.11
CB 2-e	24V AC	0.29	6.96	0.17
	12V DC	0.39	4.68	0.11
	24V DC	0.39	9.36	0.22
CB 2-a	24V AC	0.33	7.92	0.19
	12VDC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25
CB 2-s	24V AC	1.16	27.84	0.67
	12V DC	0.81	9.72	0.23
	24VDC	0.81	19.44	0.47
CB 2 w/ Audio Paging	12-24V DC	6.44	154.56	3.71
CB 4-S	24V AC	0.07	1.68	0.04
	12V DC	0.15	1.80	0.04
	24V DC	0.15	3.60	0.09
CB 4-r	24V AC	0.11	2.64	0.06
	12VDC	0.19	2.28	0.05
	24V DC	0.19	4.56	0.11
CB 4-u	2av AC	0.11	2.64	0.06
	12VDC	0.19	2.28	0.05
	24V DC	0.19	4.56	0.11
CB 6-F & CB 6-S	24VAC	0.07	1.68	0.04
	12V DC	0.15	1.80	0.04
	24V DC	0.15	3.60	0.09
CB RT	24V AC	0.33	7.92	0.19
	12V DC	0.43	5.16	0.12
	24V DC	0.43	10.32	0.25



Models with LS1000/LS2000	Voltage	Current	Watts	KWHrs
CB 1-e	12V DC	0.68	8.16	0.20
CB 1-s	12V DC	1.06	12.72	0.31
CB 5-s	12V DC	0.68	8.16	0.20
CB 9-S	12V DC	0.44	5.28	0.13
CB 2-e	12V DC	0.64	7.68	0.18
CB 2-a	12V DC	0.68	8.16	0.20
CB 2-s	12V DC	1.06	12.72	0.31
CB 4-s	12V DC	0.40	4.80	0.12
CB 4-r	12V DC	0.44	5.28	0.13
CB 4-U	12V DC	0.44	5.28	0.13
CB 6-F & CB 6-S	12V DC	0.40	4.80	0.12
CB RT	12V DC	0.68	8.16	0.20

High Voltage Models	Voltage	Current	Watts	KWHrs
CB 1 w/ Audio Paging	12-24V DC	3.83	460	11
CB 5 w/ Audio Paging	12-24V DC	3.33	400	9.6
CB 1 w/ NightCharge	120V AC	2.5	300	2.4
CB 4-U w/ NightCharge	120V AC	2.5	300	2.4

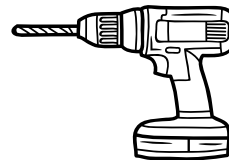
High Voltage AC Components	Voltage	Current	Watts	KWHrs
Multi-Tap Power Supply	120V AC	1.75A/210VAC	210	5.04
DIN Rail Power Supply	120V AC	1.2A/115VAC	115	2.76
Audio Paging Amp	12-24V DC	3.83	459.6	11.03

6 Installation Instructions

Tools Needed

LS2000 faceplate tools needed for installation:

1. Drill and screws to mount unit to wall.
2. Code Blue Security bit and any 1/4" bit drive handle.



Basic Flush Mount Installation Instructions for use by Installers:

1. Remove six security screws with Code Blue security bit and any 1/4" bit driver tool.
2. Separate LS2000 from enclosure and set aside carefully. The flush mount enclosure will come with no mounting holes. It can be mounted from the sides, top or bottom. Once your mounting needs have been determined, drill the holes needed for your specific installation requirements into the flush mount back box.
3. Mark the mounting holes onto the mounting surface. In order to comply with the Americans with Disabilities Act (ADA) of 1990, the speakerphone button(s) should be positioned between 34 and 48 inches from grade level. (Consult an ADA specialist in your area to verify local and federal guidelines.)
4. Connect earth ground to the provided copper grounding lug using a minimum 14-gauge conductor.
5. Connect electrical and communications wiring using the wiring diagram below.
6. Follow all federal and local codes that apply.
7. Attach faceplate to enclosure using six security screws.

Basic Surface Mount Installation Instructions for user by Installers:

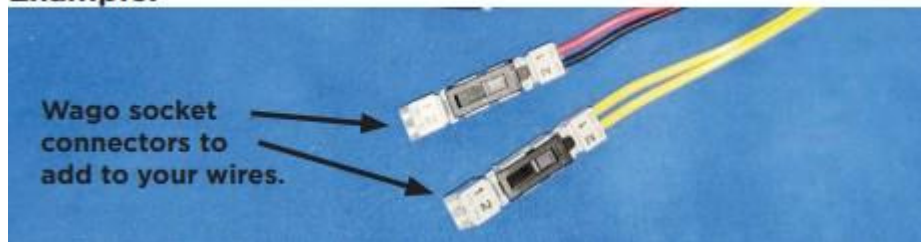
1. Remove six security screws with Code Blue security bit and any 1/4" bit driver tool.
2. Separate LS2000 from enclosure and set aside carefully.
3. Mark the mounting holes onto the mounting surface. In order to comply with the Americans with Disabilities Act (ADA) of 1990, the speakerphone button(s) should be positioned between 34 and 48 inches from grade level. (Consult an ADA specialist in your area to verify local and federal guidelines.)
4. Drill all marked holes.
5. Secure the housing to the wall – If you are securing to drywall, utilize drywall screws and anchors. If you are securing to metal, utilize sheet metal screws. If you are securing to brick, use Tapcon screws. Fasten all screws securely.
IMPORTANT: Ensure that conduit is aligned with wiring.
6. Connect earth ground to the provided copper grounding lug using a minimum 14-gauge conductor.
7. Connect electrical and communications wiring using the wiring diagram below.
8. Follow all federal and local codes that apply.
9. Attach faceplate to enclosure using six security screws.



7 How to Update Connectors

As of 2020, many Code Blue products come with Wago connectors. These connectors provide ease of use and a much stronger connection. Below are the steps needed to change to the new connectors.

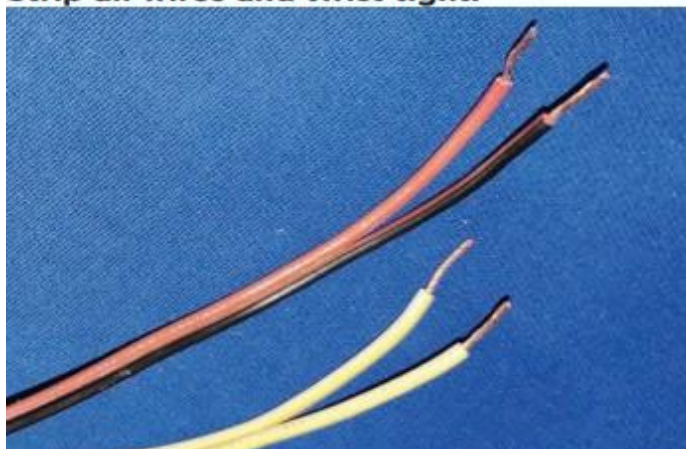
Example:



Cut off both wires.



Strip all wires and twist tight.





Place small screwdriver into square hole and push down. Insert cut wire into round hole and remove screwdriver. Repeat on the rest of the connectors.

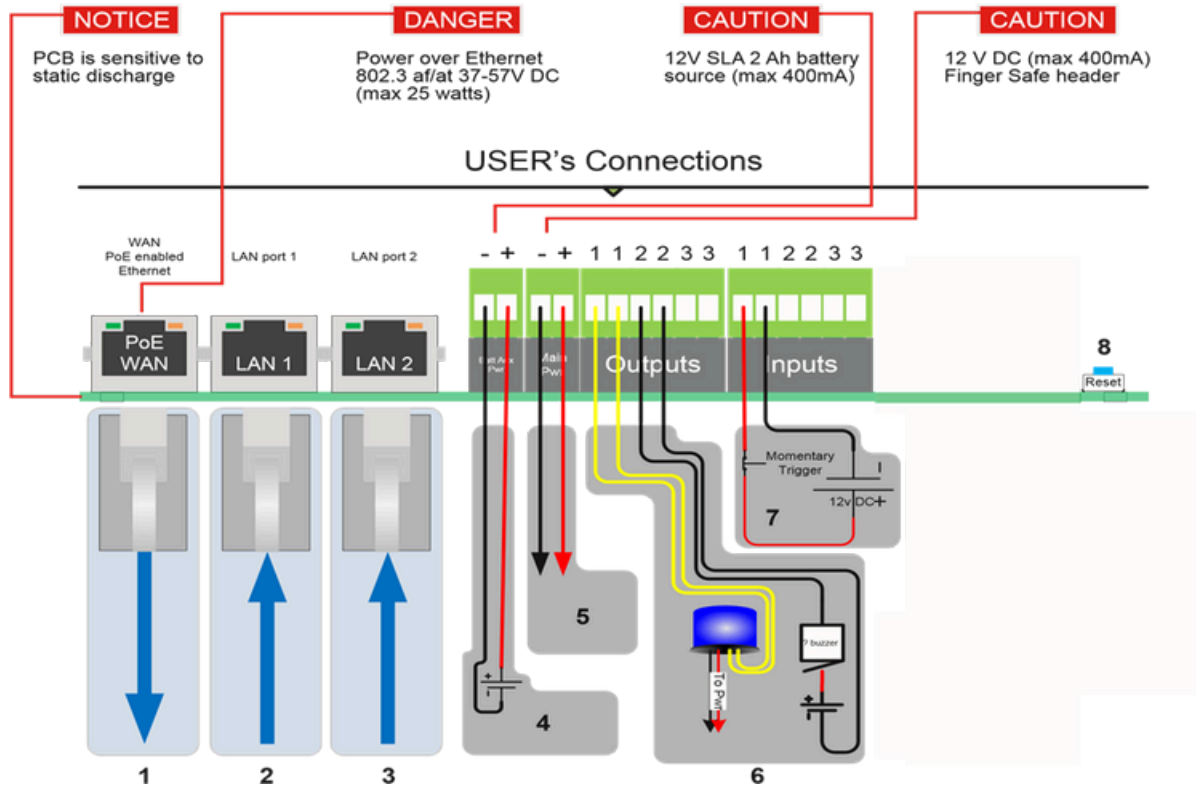


Once all connectors have been switched, you are ready to apply power.





8 Wiring Diagrams



1 - Cat 5e - Connect to Switch (100 Mb)

2 - Cat 5e - Connect to Peripheral (100 Mb)

3 - Cat 5e - Connect to Peripheral (100 Mb)

4 - Optional 12V battery

5 - 12V DC power

6 - Aux outputs (3)

• The auxiliary outputs are Normally Open and can be triggered in your action script to close. Duration is defined in your action script.

7 - Aux inputs (3)

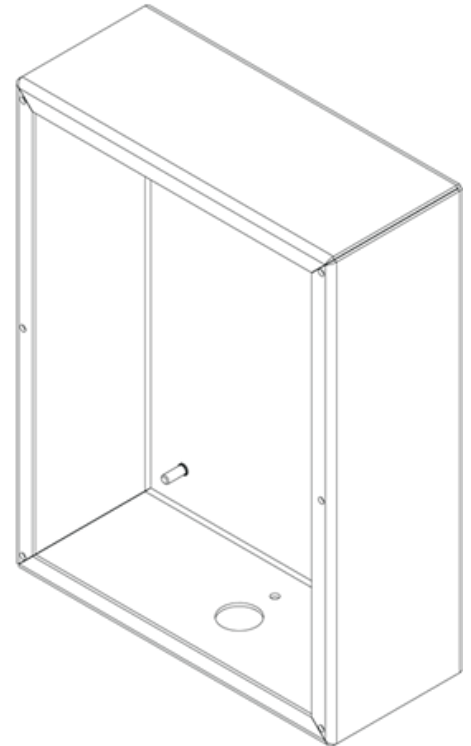
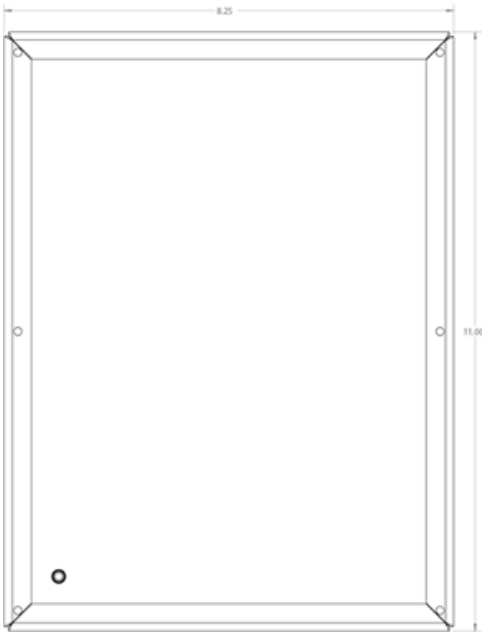
• The auxiliary inputs are Normally Open and are triggered by voltage. The voltage range is 9 to 32V AC or DC with current protection of 1.25 Amps.

8 - Factory Reset Switch

• With power connected to LS2000, hold down this reset switch for 12 seconds and then release. The unit will reboot. All settings will be lost.

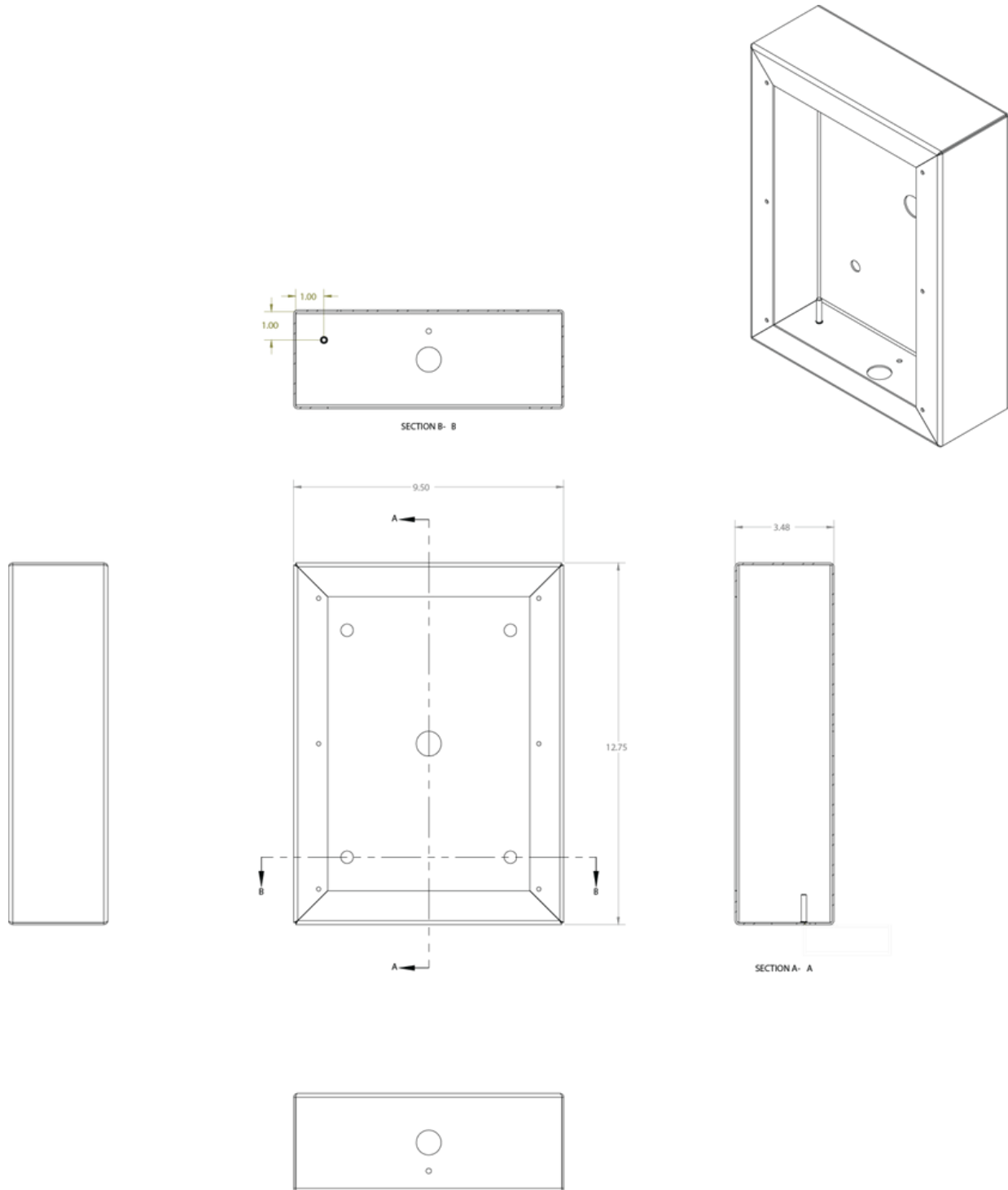
9 Schematics

9.1 Flush Mount Mounting Schematics





9.2 Surface Mount Mounting Schema Mou





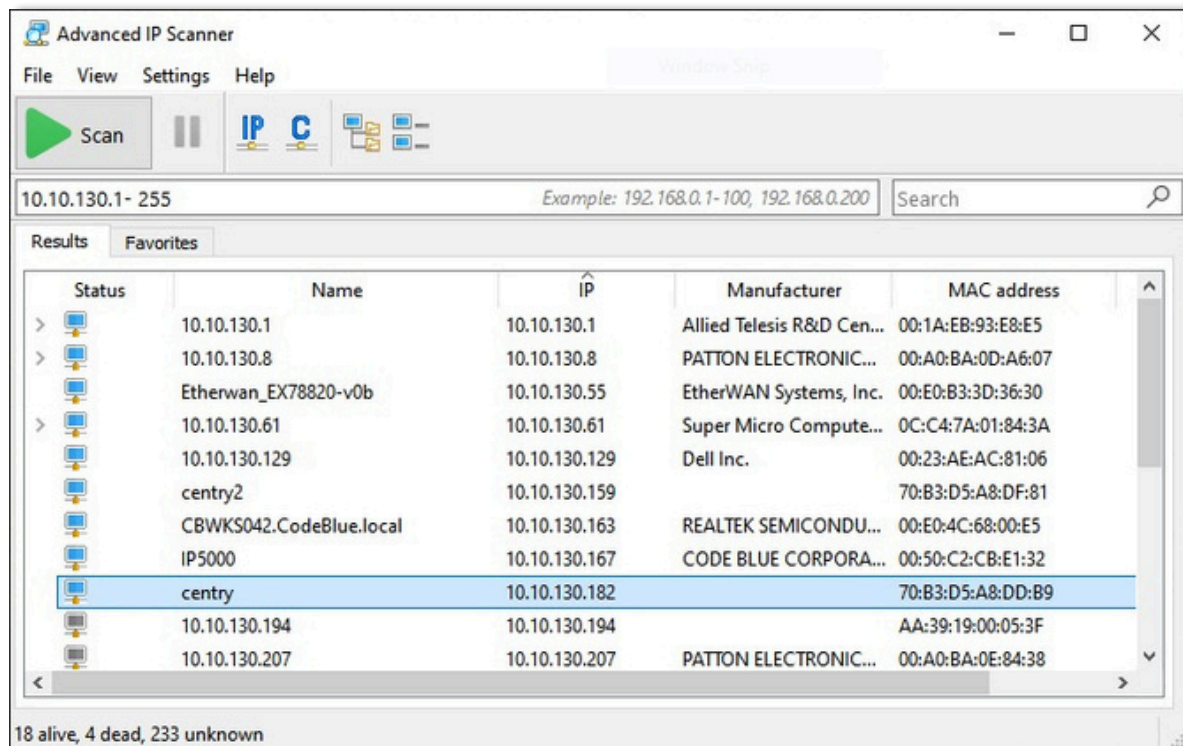
10 Setting Up Your LS2000

Determine the IP Address

Once you can browse into the device, you can switch to a static IP if desired.

Connect the LS2000 to your DHCP Network. The “Call Placed” red LED will flash while the device is booting up. The LS2000 will acquire its network settings from the DHCP server on the network. If the network is not running DHCP the device will default to the IP address of **192.168.0.249** after one minute.

If you do not have network scanner software, you can download many free ones. “Advanced IP Scanner” is one example. Run a scan of the subnet the LS2000 device is plugged into, and your PC if plugged into the same network.



You can find the IP address for your LS2000 speakerphone by matching its MAC address. You can then browse to the IP address of the device. The recommended web browsers are Google Chrome and Mozilla Firefox. You may experience erratic behavior with other browsers.



Default Login Information

Username: admin

Password: admin

The screenshot shows a login form with the following fields and elements:

- Username:** A text input field containing the value "admin".
- Password:** A password input field with masked characters "****".
- Submit:** A blue button labeled "Submit".

Network - General Settings

By default, the LS2000 is set to DHCP. If no DHCP service is present the LS2000 will default to 192.168.0.249. You can set a static IP address or to enable and set a VLAN ID under Network>General.

The screenshot shows the "Network: General" configuration page. The left sidebar contains a navigation menu with categories: Accounts, Network, Audio, Video, Event, and Administration. The main content area is titled "Network: General" and includes the following settings:

- General**
 - Connection Type: DHCP Static IP
 - Enable SSH:
 - SSH Port:
 - Hostname:
- Static IP**
 - Address:
 - Netmask:
 - Default Router:
 - DNS Primary:
- VLAN**
 - Enable VLAN:
 - VLAN ID:

At the bottom of the page, there are two buttons: "Save Changes" and "Apply Changes".



Generally, you can leave the Host and Domain fields empty.

SSH Port 22 is enabled by default. You can change the Port and disable it, if desired.

If limiting Network Ports, then the ports to have open are:

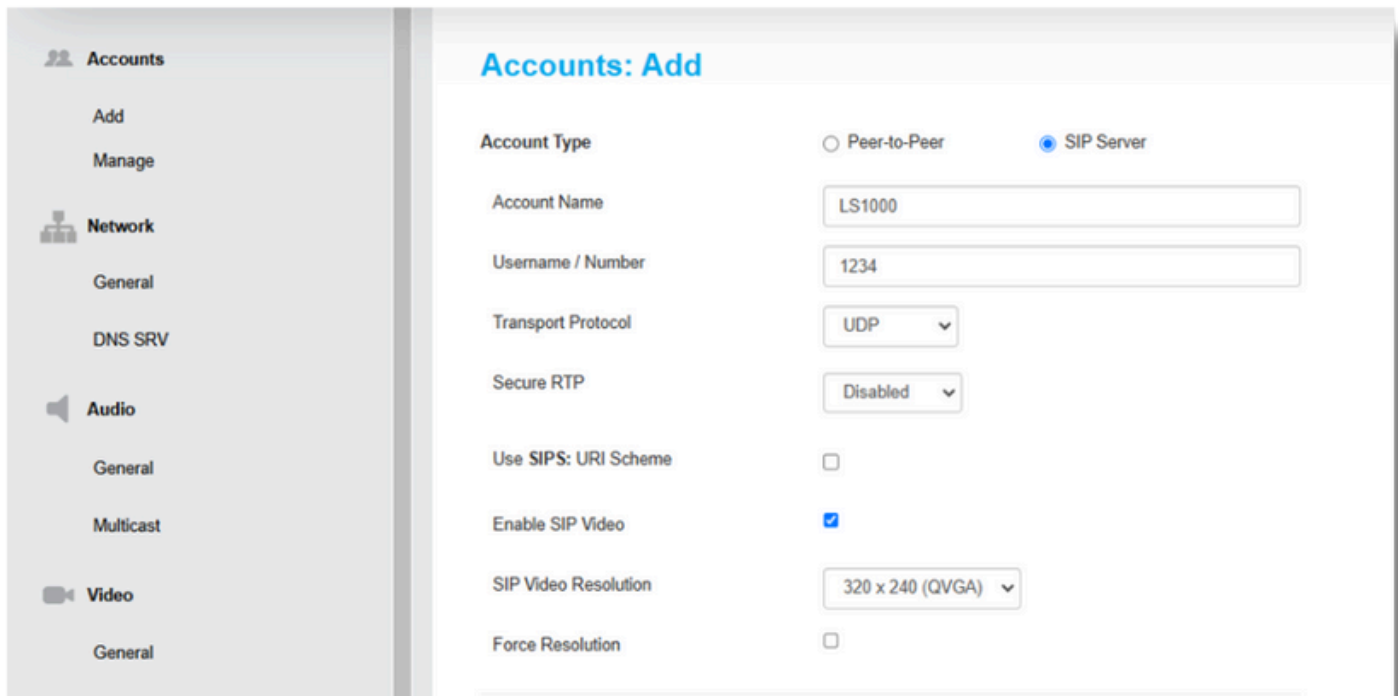
- tcp 22 SSH Secure Shell Protocol (Configurable)
- tcp 53 DNS Domain Name Service over TCP
- udp 53 DNS Domain Name Service over UDP
- tcp 80 HTTP Hyper Text Transfer Protocol
- tcp 161 SNMP Simple Network Management Protocol
- tcp 162 SNMP Trap Simple Network Management Protocol Trap
- tcp 443 SSL Secure Socket Layer (Deprecated)
- udp 554 RTSP Real Time Streaming Protocol
- udp 554 RTSP Real Time Streaming Protocol
- udp 5060 SIP Session Initiation Protocol
- tcp 5061 SIP Session Initiated Protocol
- rtp 16384-32767 Real Time Transport (Configurable)

Accounts

Add Account

These settings are usually provided by IP PBX, SIP server or SIP gateway personnel. When setting up a SIP or peer-to-peer account, make sure the correct Account Type is selected

- **Account Name:** Free form field to identify the account.
- **Username/Number:** Phone number or extension for the LS2000. This is still required for peer-to-peer; Any number value can be entered.
- **Speaker Volume:** You can set the speaker volume gain here in the GUI, as well as from a touch-tone phone using in-call commands. This becomes the default volume level.
- **Mic Volume:** You can set the mic volume gain here in the GUI, as well as from a touchtone phone. using in-call commands. This becomes the default volume level.
- **Enable SIP Video:** Select this option to allow SIP video packets to be transmitted. RTSP video will still be available but note that only one format will be available at a time.
- **SIP Video Resolution:** Use drop down menu to select resolution for SIP video. This does not apply to RTSP video quality.
- **Force Resolution:** Some desk phones will overwrite the resolution set in the LS2000 to their native resolution. Select this option to force the selected resolution.



Accounts: Add

Account Type Peer-to-Peer SIP Server

Account Name

Username / Number

Transport Protocol

Secure RTP

Use SIPS: URI Scheme

Enable SIP Video

SIP Video Resolution

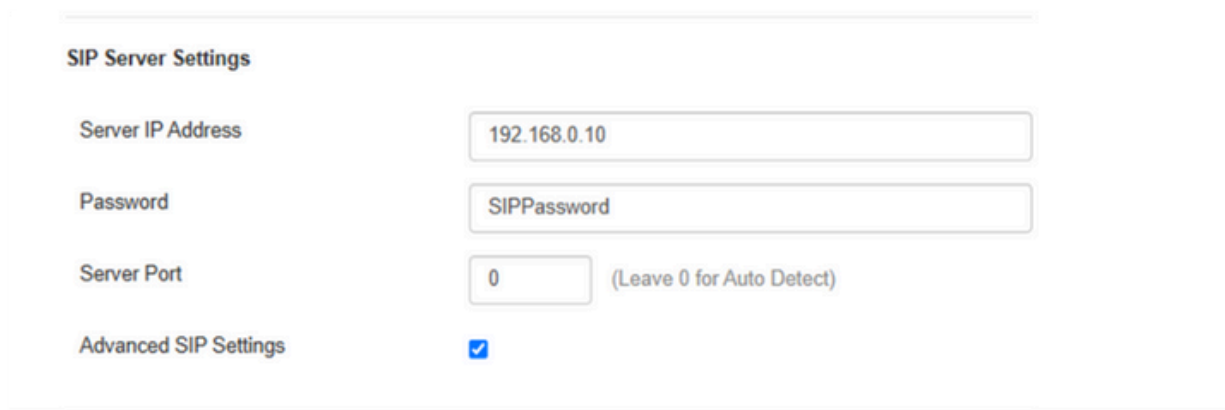
Force Resolution

SIP Server Settings

Server IP Address: You can use the domain name or IP address of your IP PBX, SIP Gateway, SBC or whatever you register SIP devices to.

Password: This is the shared SIP secret password that the phone and your SIP registration server share.

Server Port: 0 for auto detect. Commonly may be 5060. If necessary, you may have to specify a port here.



The screenshot shows a web interface for configuring SIP server settings. It includes the following fields and options:

- SIP Server Settings** (Section Header)
- Server IP Address:** Text input field containing "192.168.0.10".
- Password:** Text input field containing "SIPPassword".
- Server Port:** Text input field containing "0", with a note "(Leave 0 for Auto Detect)".
- Advanced SIP Settings:** A checkbox that is checked.

Advanced SIP Settings

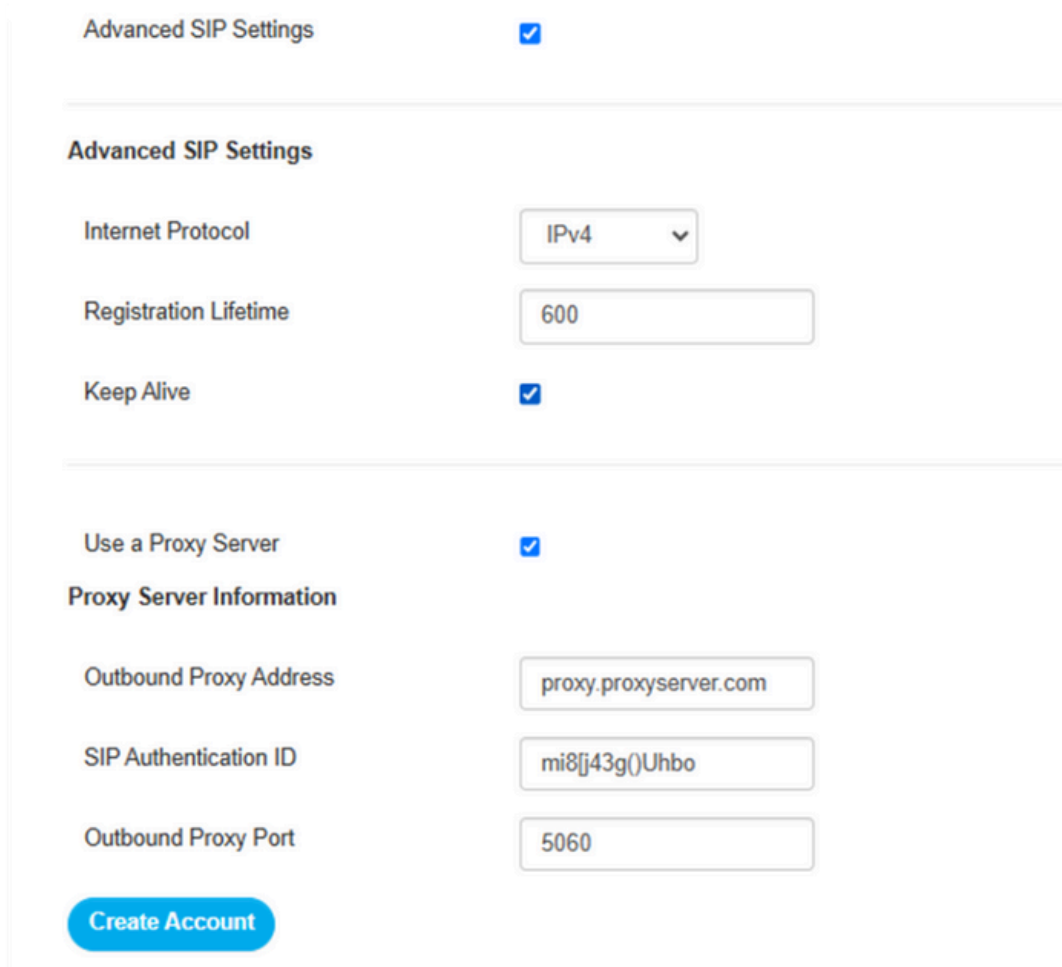
Transport Protocol: IPv4 or IPv6.

Communication Protocol: UDP or TCP.

Registration Lifetime: Default is 600 seconds. This is the time the phone will attempt re-registration if not contacted by the SIP server.

Keep Alive: Check if you want keep-alive messages sent between the SIP server and the phone.

Use a Proxy Server: If using a proxy server in addition to a SIP server, then specify the IP Address or Domain, SIP Authentication ID and Outbound Proxy Port used.



The screenshot shows a web interface for configuring SIP settings. At the top, there is a section for "Advanced SIP Settings" with a checked checkbox. Below this, the "Advanced SIP Settings" section includes: "Internet Protocol" set to "IPv4" (dropdown), "Registration Lifetime" set to "600" (text input), and "Keep Alive" checked. A second section, "Use a Proxy Server", is also checked. Underneath, the "Proxy Server Information" section includes: "Outbound Proxy Address" set to "proxy.proxyserver.com" (text input), "SIP Authentication ID" set to "mi8[j43g()Uhbo" (text input), and "Outbound Proxy Port" set to "5060" (text input). A blue "Create Account" button is located at the bottom left of the form area.

Manage Accounts

- After adding an account, you can manage the created accounts on this page.

Note: You can create a near infinite to different SIP servers. Generally, only one is used.

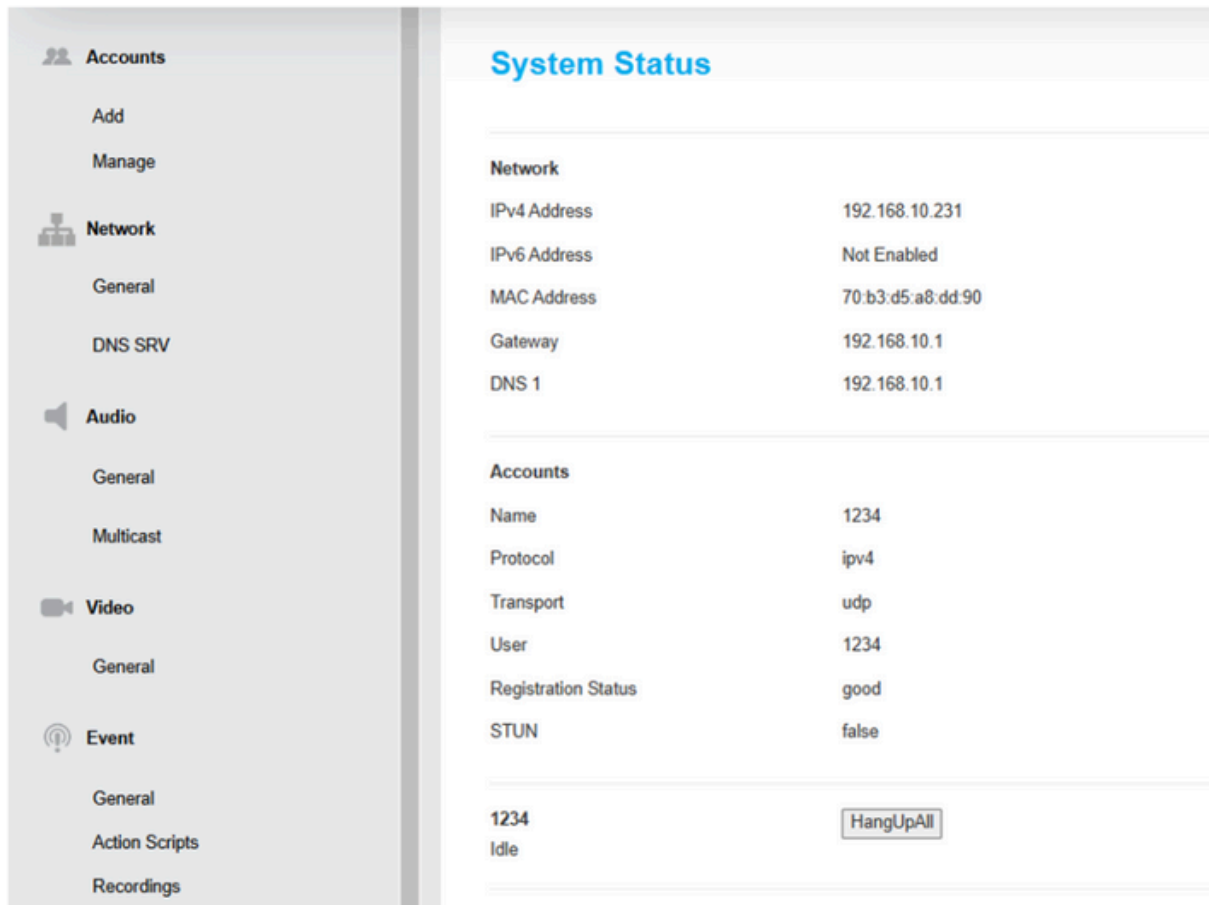
- Click onto an account to manage it for viewing or to make changes.

System Status

After creating your account and providing the SIP server, check the registration status on the System Status page. This is located in the top right corner of the screen.

In this screen, you can see the current IP address of the phone, which is the same for the camera.

It also will display the current registration status of the configured accounts. Registration status of “good” indicates the account is properly registered to the SIP server. If not, investigate the matching account settings configured in the SIP server and if the network has any restrictions.



The screenshot displays the 'System Status' page. On the left is a navigation menu with categories: Accounts, Network, Audio, Video, and Event. The main content area is titled 'System Status' and is divided into two sections: 'Network' and 'Accounts'.

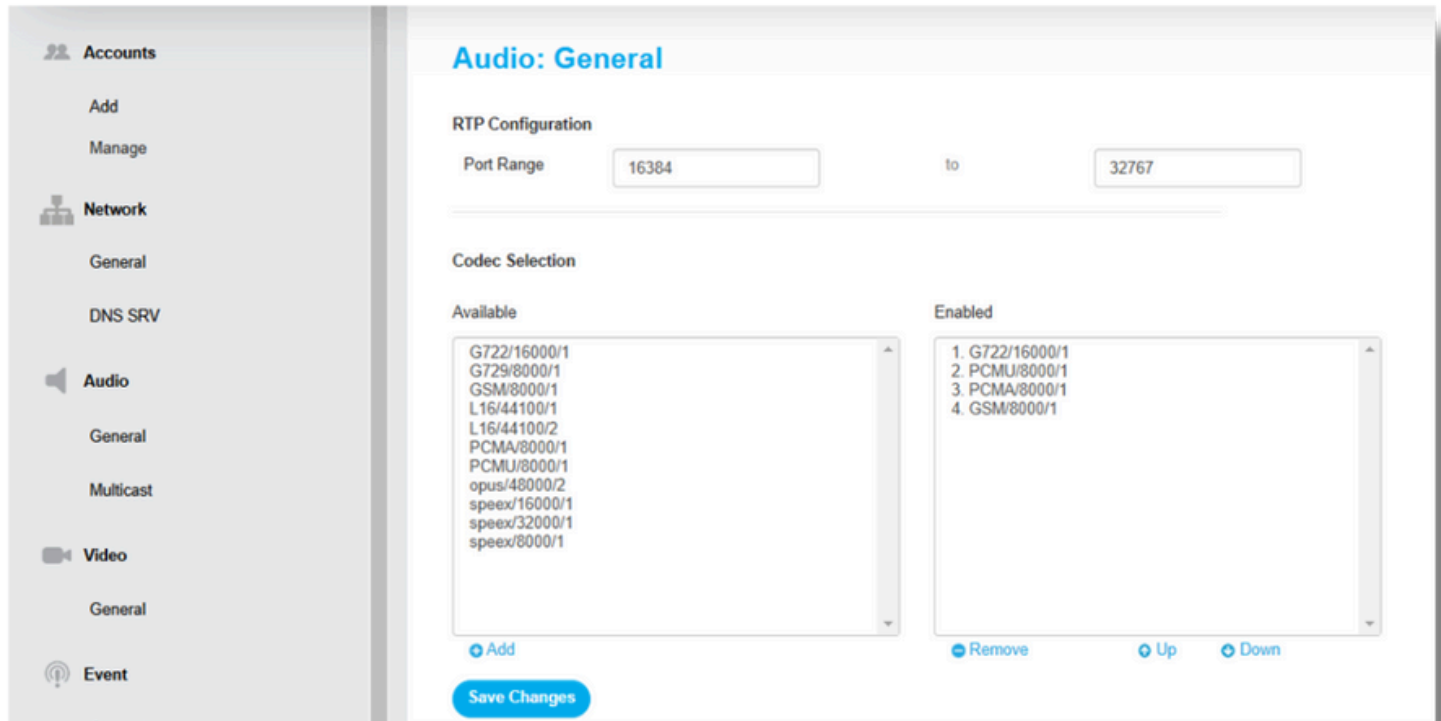
Network	
IPv4 Address	192.168.10.231
IPv6 Address	Not Enabled
MAC Address	70:b3:d5:a8:dd:90
Gateway	192.168.10.1
DNS 1	192.168.10.1

Accounts	
Name	1234
Protocol	ipv4
Transport	udp
User	1234
Registration Status	good
STUN	false

At the bottom of the Accounts section, there is a row for account '1234' with the status 'Idle' and a 'HangUpAll' button.

Audio

Default RTP Ports and Codecs are listed here. These can be adjusted based on the needs of the SIP or PBX server. The default port range is 16384-32767.



The screenshot displays the 'Audio: General' configuration page in the Code Blue administrator interface. The left sidebar contains navigation options: Accounts (Add, Manage), Network (General, DNS SRV), Audio (General, Multicast), Video (General), and Event. The main content area is titled 'Audio: General' and includes the following sections:

- RTP Configuration:** Port Range is set to 16384 to 32767.
- Codec Selection:** Two lists are provided: 'Available' and 'Enabled'.
 - Available:** G722/16000/1, G729/8000/1, GSM/8000/1, L16/44100/1, L16/44100/2, PCMA/8000/1, PCMU/8000/1, opus/48000/2, speex/16000/1, speex/32000/1, speex/8000/1.
 - Enabled:** 1. G722/16000/1, 2. PCMU/8000/1, 3. PCMA/8000/1, 4. GSM/8000/1.

Buttons for 'Add', 'Remove', 'Up', and 'Down' are located below the codec lists. A 'Save Changes' button is at the bottom of the configuration area.

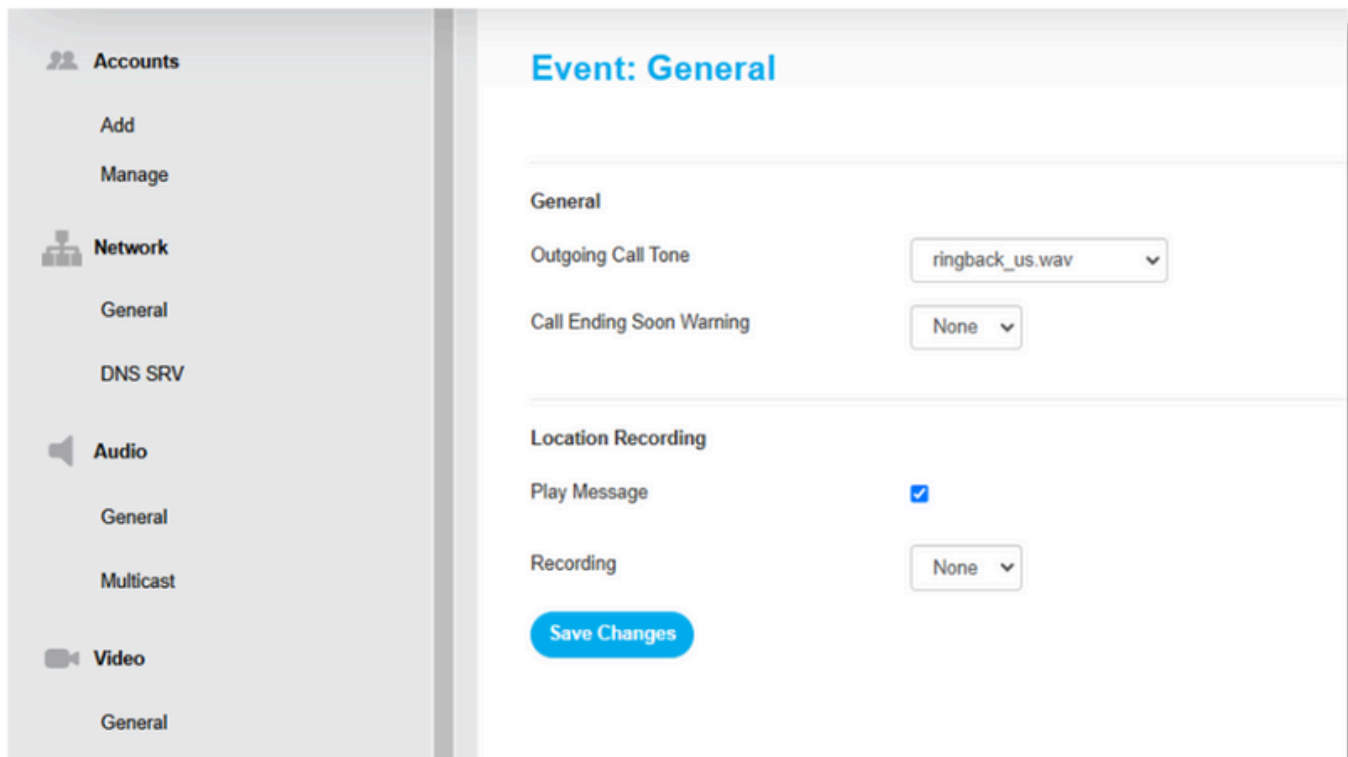
Multicast Audio

To add a new multicast configuration, enter a Label, the Listening IP address and Port. To change the priority of the multicast, simply drag and drop. A phone call priority level also can be set with the Order Priority box checked. By default, the Phone Call Priority is set to 0.



General Events

In the General Events setting you can alter the outgoing call tone, apply a call ending warning, allow or disallow incoming calls and apply location recordings.





Action Scripts

Action Scripts determine what happens when a button is pushed, an aux input is activated or an incoming call is received. If you have a multi-button phone by default the Red Button is labeled as Button 1 and the Black Button is labeled as Button 2.

Event: Action Scripts

Script For

Buttons Button 1 Button 2

Aux In Aux 1 Aux 2 Aux 3

Incoming Calls Incoming Call

Other Inputs:

- Select the radial button you would like to associate an action with.
- Choose “Add Action”
- You can choose to have an aux output triggered, place a call, or simply play a message. You can also string multiple actions together.
- For Incoming Call, the ability to Auto Dial will be replaced with Receive Call. Options may vary depending on peripherals selected as available in the Hardware Settings such as handset and PAS options.

Event: Action Scripts

Script For

Buttons

Aux In

Incoming Calls Incoming Call

Other Inputs:

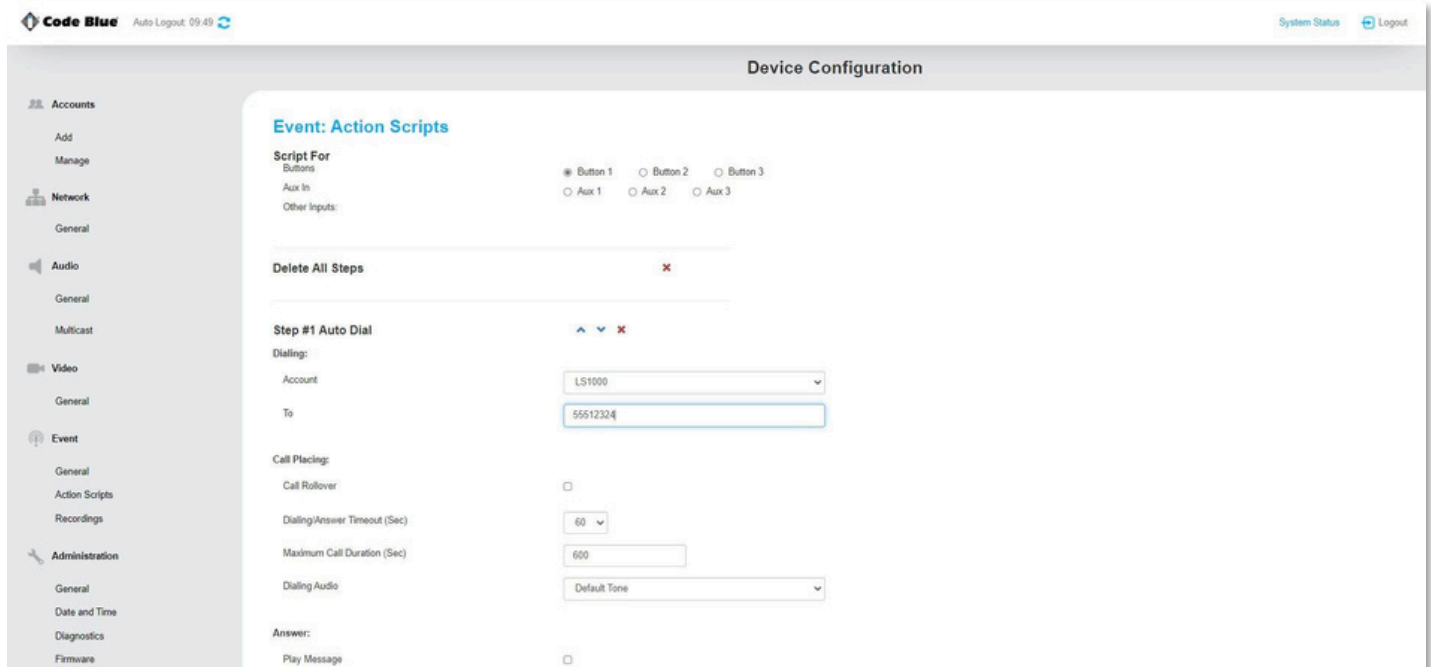
Delete All Steps

Add Action Test Script

Save Script

Place Call Action

- Select the Account that was created earlier that the calls will be routed through.



- If a Keypad is present, then the option for Manual Dial is available. The number of digits that can be dialed can be limited. For example, only allowing 4-digit extensions to be dialed.
- For Auto Dial calls enter in the “To” field the number you would like LS2000 to dial for Auto Calls. For peer-to-peer, enter the IP Address:Port the call is intended to go to.
- If that number does not answer, you can “Add Call” and select another account and phone number to dial. This is useful if you have the phone registering to multiple SIP servers/SIP PBXs. For example, you can have it call 911 from Account 1 and call 911 from Account 2 if Account 1 is down. The phone will roll over to Account 2 and call out that account.
- Repeat for the number of times you want the call to rollover before answering.
- Each iteration of the rolled over call uses the “Dialing/Answer Timeout.”

Note: If voicemail answers, then that is considered an answered call.



“Maximum Call Duration” is 600 seconds or 10 minutes by default. This is used as a timer in case the receiving party never hangs up the call. This timer is editable.

“While Dialing”

You can select the audio of ring-back or use a custom recording that is uploaded into the phone in the Recordings page. A common recording script would be something like, “Security has been contacted; please stand by.”

This message will repeat just like ring-back would until the call is answered. This message will stop when the call is answered.

“When Answered”

The default mode is two-way audio upon the call being answered.

Additional options include playing a custom recording locally out of the LS2000 speaker and/or playing a custom recording remotely to the receiving party that identifies where this call is coming from.

Note: This is a useful feature if the receiving party is not going to get a unique caller ID name/number for this location.

Once the recording has played, two-way audio will commence.

Note: If playing a remote message but not a local message, then the person at the LS2000 will hear dead air until the remote message is done playing. It is advisable to play the same message to both parties in this case.

Note: It is advisable to keep your recordings short to establish connection to a two-way call as quickly as possible.

“In-Call Commands”

Default is to have the in-call commands available during the phone call. These can be disabled for this action. In-call commands are also available on an inbound call into LS2000. See section on in-call commands for the full list.



Control AUX Output

Choose the AUX output you want to control.

Event: Action Scripts

Script For

Buttons Button 1 Button 2

Aux In Aux 1 Aux 2 Aux 3

Incoming Calls Incoming Call

Other Inputs:

Delete All Steps ✕

Step #1 Control AUX Output ^ v ✕

Output Number

Set To

For Duration (Sec) (0 = toggle only)

Add Action Test Script

Save Script

Choose enabled or disabled for this action. Enabled closes the normally open contact; disabled opens the normally open contact.

Choose the duration you want the action to last. (0 seconds = indefinite.)

Note: This is the most common setting for controlling an auxiliary, such as a blue light strobe, to stay activated for the duration of the call if linked to a place call action

Play Message

Play a message when this step is reached as part of your greater action script or as the only action in your action script.

Upload your recordings by going to the Recordings page. The message only plays out of the LS2000 speaker.

Note: There are blue arrows that can move your action up and down in the action script.

Note: You can string together many actions in the action script.

A typical action script for a LS2000 may just have a Place Call Action, or if your LS2000 has a remote beacon/strobe light, then it would look like the example below to have the strobe linked to the duration of the call.

At the bottom of the action script page, you can test your action script and it will launch through the script and place a call, play a message, or trip auxiliaries.

Code Blue Auto Logout: 09:30
System Status [Logout](#)

Device Configuration

- Accounts**
- Add
- Manage
- Network**
- General
- Audio**
- General
- Multicast
- Event**
- General
- Action Scripts
- Recordings
- Administration**
- General
- Date and Time
- Diagnostics
- Firmware
- Hardware

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Event: Action Scripts

Script For
 Aux In: Aux 1 Aux 2 Aux 3
 Other Inputs: Hook Switch Incoming Call

Delete All Steps ✕

Step #1 Control AUX Output ⬆ ⬇ ✕

Output Number:

Set To:

For Duration (Sec): (0 = toggle only)

Step #2 Auto Dial ⬆ ⬇ ✕

Dialing:

Account:

To:

Call Placing:

Call Rollover:

Dialing/Answer Timeout (Sec):

Maximum Call Duration (Sec):

Dialing Audio:

Answer:

Play Message:

In Call:

Incoming DTMF Commands:

DTMF Aux Out Momentary Enabled (Sec): (0 = toggle only)

Audio Output: Handset

Outgoing DTMF:

Blink Station Light:

Hang Up:

Allow Hang Up on Repress:

Step #3 Control AUX Output ⬆ ⬇ ✕

Output Number:

Set To:

For Duration (Sec): (0 = toggle only)

Recordings

- Upload recordings into the phone on this page.
- Recordings uploaded are then available in action scripts.



- The phone only can play uncompressed 16-bit PCM or compressed g.711 A-law/U-law format .wav files.
- If you upload unsupported audio file formats, they will not play in the action script.



Administration General

- Upload a secure PEM certificate and PEM key.
- Change Web GUI login password
- Generate OS and SIP logs
- Create and restore configuration backup
- Reboot device
- Factory Reset device

Administration: General

Secure HTTP Server

PEM Certificate

PEM Key

Password Settings

Current Password

New Password

Confirm New Password

[Update Password](#)

Logging

[Generate OS Log](#)

[Generate SIP Log](#)

Backup

[Create Backup](#)

Restore Backup

Reboot

[Reboot Device](#)

Factory Reset

[Reset](#)



Date and Time

- Adjust the date and time on this page or point the phone to a NTP server to have a permanent time source.

Administration: Date and Time

Current Date & Time Thursday, September 5, 2024 at 11:23:08 AM EDT

Set Date & Time

Time Zone

Daylight Savings Time

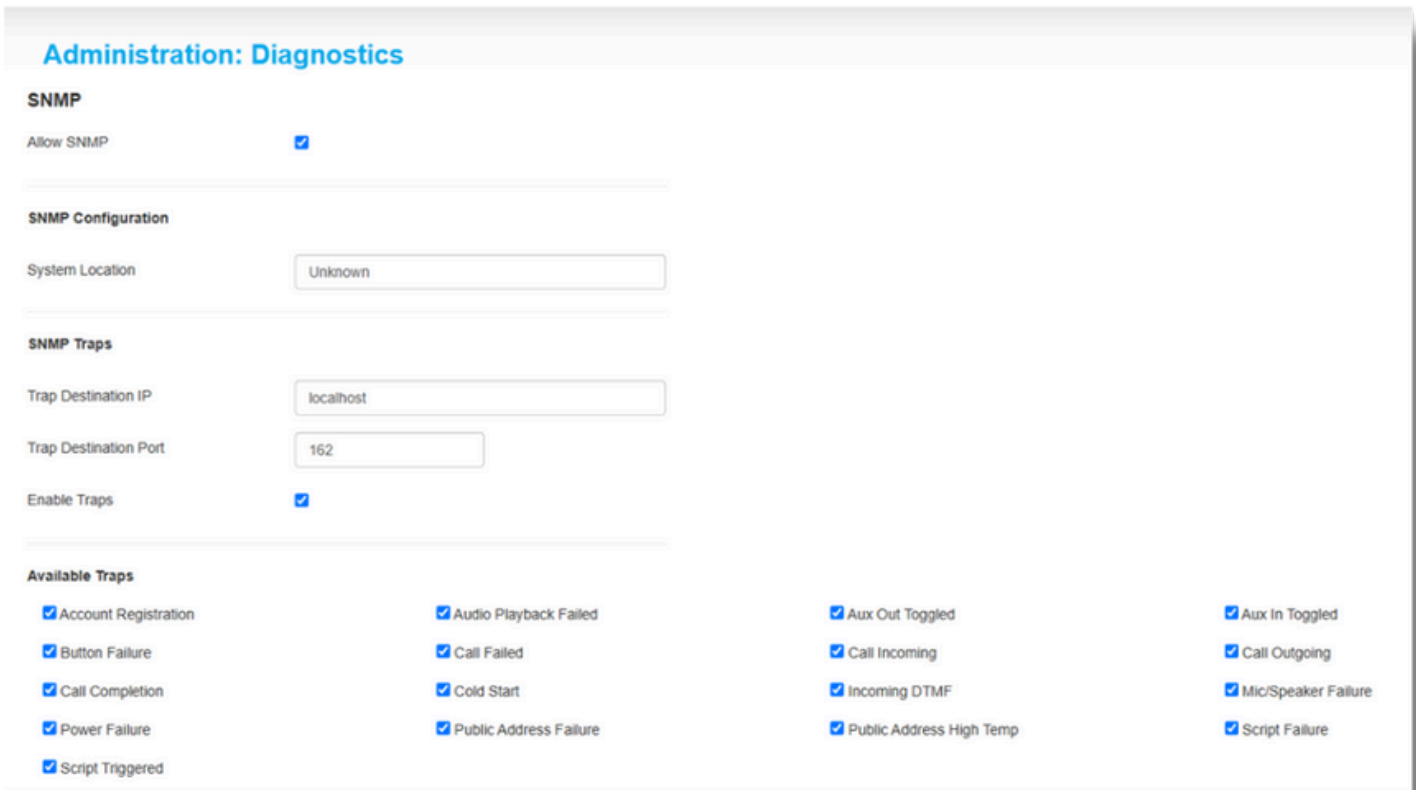
Use NTP Server

Server Address

[Save Changes](#)

Diagnostics

- With SNMP Traps enabled, enter the SNMP server name, port number (162 by default) and Read/Write Community Names.
- Each trap can be enabled or disabled to customize the output.



The screenshot shows the 'Administration: Diagnostics' page. It is divided into several sections:

- SNMP**: 'Allow SNMP' is checked.
- SNMP Configuration**: 'System Location' is set to 'Unknown'.
- SNMP Traps**: 'Trap Destination IP' is 'localhost', 'Trap Destination Port' is '162', and 'Enable Traps' is checked.
- Available Traps**: A grid of 16 traps, all of which are checked:
 - Account Registration
 - Audio Playback Failed
 - Aux Out Toggled
 - Aux In Toggled
 - Button Failure
 - Call Failed
 - Call Incoming
 - Call Outgoing
 - Call Completion
 - Cold Start
 - Incoming DTMF
 - Mic/Speaker Failure
 - Power Failure
 - Public Address Failure
 - Public Address High Temp
 - Script Failure
 - Script Triggered

- The time interval for Power, Microphone and Speaker can be adjusted. By default, these are set to 900 seconds for Power Failure tests and 0 Seconds (Disabled) for Mic/Speaker tests.
- The volume of the Mic/Speaker test can be adjusted via a drop-down menu.
- Alerts POST; Set a destination to receive an HTTP post of Call outgoing, Call Answered and Call Hung-up.

SNMP Version Configuration

SNMP Version

V1/V2 ▾

Read Community Name

public

Write Community Name

private

Diagnostic Test Intervals**Power Supply Failure Timeouts**

12V DC (Sec)

900

(0 = disabled)

12V DC Battery (Sec)

900

(0 = disabled)

PoE (Sec)

900

(0 = disabled)

Microphone & Speaker Test

Test Interval

Disabled ▾

Alerts POST

Enable

URL

<https://ls1000.free.beeceptor.com>**Save Changes**

Firmware

Upload new firmware into the phone from www.codeblue.com/resources. Once you download onto your PC, unzip the file. From the LS2000 firmware page, browse to the .bin file on your PC and upgrade your LS2000.



Firmware Release Notes:

v1.2.7

- Adds 2s delay after enabling PA to allow amplifiers to warmup
- Allows Button 4 & Aux 3 scripts, fixes save error
- Checks if camera is installed when making phone calls before requesting video resources (eliminates 12 second call delay)
- Hides SIP Video settings in UI if camera is not installed
- Fixes Green In-Call LED
- Fixes default 1Gpbs link auto-negotiation
- Updates OpenSSH to 9.7p1 from 8.9p1

v1.2.6

- Fixes 'Delete All' action script steps
- Fixes GS Audio issues if Video Call but no SIP Video enabled on phone
- Fixes SegFault if Adding/Removing scripts when script is active
- Fixes crash during Multicast
- Fixes next-event handling in multi-step action script
- Aborts script if active when a script step is removed from Web UI
- Rejects second Incoming Call if Incoming Call script is currently active
- Ignores keypad events if keypad is not installed
- Adds ability to backup & restore phone configuration through WebUI
- Plays audible beep through speaker after successful boot
- Plays busy signal if calling phone that rejects call or if line is busy
- Updates NGINX Server to v1.26 (Released April 2024)

v1.2.5

- Fix to handle internal conference bridge port changes
- Improves echo cancellation
- Verified incoming/outgoing calls with Zoom
- Allows user to completely disable SNMP functionality
- Adds 'Alert POST' feature
- Fixes an issue with call rollover never ending.
- In-Call Volume changes are saved and applied after reboot.
- Web UI Volume Changes on Hardware page reflect current volume levels and applied immediately when saved.
- Fixes race condition between network interface configuration and net-snmpd
- Renames Web UI page `Network: SIP` to `Network: DNS SRV`
- Fixes incoming calls if only a play message action is present
- Fixes libpjsip lockup issues
- Adds Avahi Daemon to provide mDNS lookup (Use `avahi-browse -ra` on Linux to identify Centry devices)
- Allows Disabled/15 Minute/Hourly/Daily Mic & Speaker test options in Web UI
- Fix for Incoming Call script not showing up
- Removes Speaker and Microphone levels from SIP Accounts (Default values can now be set from Web UI Hardware page)
- Adds ability to enable/disable RTSP stream from Web UI
- Adds ability to enable/disable ONVIF from Web UI
- Disables ONVIF by default
- Removes limit of '80% of maximum' speaker output and microphone input
- Improves Web UI Date & Time page by verbosely stating current time vs set time
- Additional improvements to control video resource between videod & phoned
- Fixes Web UI Reboot button
- Improves Firmware Update & Reboot experience by auto-reloading webpage and more verbose messaging
- Adds Station Light control to call scripts
- Fixes Vlan by changing order in which net devices are initialized
- Updates AEC Tail Length from 56ms to PJSIP default of 200ms to potentially address echo issue
- Removes `Events: General -> Allow Incoming Calls` checkbox
- Mutes both parties when playing local and/or remote messages
- Improves handshaking of video resources between videod and phoned
- Fixes SNMP CallOutgoing trap
- Updates jQuery to v3.1.7 from v1.12.3 to address vulnerability in ajax requests (CVE-2020-11023)
- Removes placeholders in UI (example: static IP address)
- Defaults 'Incoming Call' to True
- Defaults 'Blink Station Light' during script to True
- Fix for setting Timezone
- Disables Station Light output if Station Light is not installed
- Fix for multiple keypad presses
- Fix for SIP Video
- Fix for call rollovers
- Fix for limit of steps in script
- Removes ec_log and ec_logd/postgres logging
- General code clean-up for maintainability
- Adds `df` and `du` Linux commands

v1.2.4

- Support for ONVIF 22.12
- Attempts to improve audio quality during phone calls
- Adds DNS SRV support
- Adds VLAN Tagging (802.1Q) support
- SNGREP is now allowed via admin user account via command `sudo sngrep`
- Adds UI support to persistently change hostname
- Removes ILBC Codec
- Attempts to acquire DHCP IP address 5 minutes after default static IP address is set
- Updates CA-Certificates to use latest available (3/11/2023)
- Systemd logs are now cleared once an hour. This may reduce the occurrence of the ECoS firmware update failures.

v1.2.3

- Monitors on-board ethernet switch status for external Link Good and resets DHCP upon Link Down
- Fixes issue with Remote/Local playback restarting during script
- Adds RTSP Constant Bit Rate (CBR) options 1-4 Mbps in UI
- Fixes issue with RTSP freeze
- Allows up to 10 active RTSP streams
- Syncs UI Admin password to Admin Linux Profile
- Allows Micrel KSZ8794 Chip Id 7

v1.2.2

- Ignores IP address check on start-up if the device has been configured for a static IP address.

v1.2.1

- Updates PJLIB to v2.12.1 from v2.7.2
- Supports DTMF over SIP-INFO
- Defaults telephone-event to 101
- Adds SRTP Support
- Adds TLS Support
- Adds Support for special characters in password
- Allows trigger of Aux port in parallel with making phone calls
- Fix for early media
- Fix for CHAN_PJSIP video

v1.1.4

- Adds four button support



Hardware Settings

- Hardware settings allow you to enable or disable the Station Light, adjust the brightness, and blink rate.
- Each button can be configured as a Non-Emergency Button, Emergency or Volume Control.
- Any additional peripherals attached to LS2000 must be marked as “Installed” to ensure functionality.
- The default volumes for speaker, microphone and Public Address speaker can be set here.



Administration: Hardware

Buttons

Button 1



Mode

Emergency

Button 2



Mode

Non-Emergency

Button 3



Button 4



Installed Peripherals

Speaker



Default Speaker Volume



Hook Switch



Handset



Keypad



Public Address



Default Microphone Volume



Station Light



Save Changes

11 In-Call Commands

You can make adjustments during a live phone call with the **LS2000** device. This can be done by calling into the **LS2000** or by a call initiated by the **LS2000** to the receiving party. The party with the touch-tone phone can use the keys on their keypad via DTMF tones to make adjustments.

In-Call Command	Function	Description
1	Play Location Message	Play pre-recorded message set in Event>General
2	Switch Audio Output	Toggle between onboard speaker & PAS
3	Deactivate Call Timer	Deactivates the Maximum Call Duration set in the Action Scripts.
4	Aux 1 Output	Toggle contact closed/open
5	Aux 2 Output	Toggle contact closed/open
6	Increase Mic Gain	Small adjustments to LS2000 microphone
7	Decrease Mic Gain	Small adjustments to LS2000 microphone
8	Increase Speaker Gain	Small adjustments to LS2000 speaker
9	Decrease Speaker Gain	Small adjustments to LS2000 speaker

Note: These adjustments do not persist after the call has ended. The LS2000 phone will return to the default levels set in the account screen via the phone's web GUI. See the [Accounts chapter](#) for further information.

12 Compatibility

LS2000 is a SIP version 2.0 (RFC3261) device and is compatible with IP Gateways and PBXs that can register third party SIP devices.

You must verify that the IP PBX you are registering LS2000 to can handle third party SIP devices, whether through licensing and/or hardware add-ons. Please note that LS2000 currently does not work with a separate Proxy Server. SIP registration and calls must point to the same domain/IP Address SIP Server.

Examples of mainstream IP PBXs LS2000 has registered to as a third-party SIP device include:

- Avaya
- Asterisk
- Cisco Call Manager
- and many others...

13 Configuring for Cisco Unified Call Manager

PREPARATION

1. Record the MAC address and determine the current IP address for each LS2000 device you wish to use with CUCM.
2. Determine which partition you will put the LS2000 directory numbers into.
3. Obtain one directory number for each Ce LS2000 device.
 - a. If you are going to use the LS2000's dual account configuration to register to redundant CUCM servers, obtain a second directory number for each LS2000 device.
4. Determine which calling search space you will assign to the LS2000.

LS2000 CONFIGURATION

Configure Account(s)

1. Log in to LS2000 via its web interface. The default username and password are admin and admin.
2. Select Account 1.
3. For VoIP Protocol, select SIP and RTP.
4. Under SIP Configuration, for Username/Number enter the directory number assigned in the CUCM.
5. For Display Name, enter caller ID text.
6. For Domain, enter the hostname or IP address of the CUCM subscriber you wish to register this account.
7. Insure Keep-Alive is enabled.
8. For Password, enter the password you entered into Digest Credentials under the CUCM end user.
9. Click Save.
10. Repeat steps 3-9 with Account 2 if you are using the second account.

Other Settings

Refer to Section 10 to complete the setup of the LS2000, including Numbers, General Settings, Hardware Settings and Action Scripts. When finished, it should now register to CUCM and be able to place calls in the assigned calling search space as well as receive calls at the directory number it is configured with.

Note: If you are setting up the LS2000 with secondary account support, make sure that you create each failover number twice.

UCM CONFIGURATION

All UCM-side configuration is done within the Cisco Unified CM Administration web interface.

Create Phone Security Profile

1. Navigate to System > Security > Phone Security Profile.
2. Do a Find on “Third-party” to locate the Third-party SIP Device Basic - Standard SIP Non- Secure Profile. Click the Copy icon.
3. Check Enable Digest Authentication.
4. Change the Name and Description to Code Blue LS2000 Profile.
5. Click Save.

Configure End Users

For each LS2000 device, configure a new end user for SIP authentication.

1. Navigate to User Management > End User.
2. Click Add New.
3. For the User ID, enter the Directory Number of the LS2000 phone.
4. Fill in the Last Name field with the Directory Number.
 - a. The Directory Number, User ID, and Last Name fields must all be the same.

Create and record a secure SIP password and fill in the Digest Credentials and Confirm Digest Credentials fields with this password. You will be entering this password later into the LS2000.

5. Click Save.

Configuring End Users for Secondary Accounts

If you are going to use the Cen LS2000's try's secondary account functionality to register to a separate directory number to a separate CUCM node for failover support, repeat the above process using a local-use-only MAC address. A local-use-only MAC address has the U/L bit set to 1 to indicate the address is locally administered.

Configure Phones and Directory Numbers

For each LS2000 device, configure a new Phone and associated directory number.

1. Navigate to Device > Phone.
2. Click Add New.
3. For Phone Type, select Third-party SIP Device (Basic).
4. Enter the MAC Address of the phone in hexadecimal format; e.g. 00:50:C2:17:7B:E8 would become 0050c2177be8.
5. For Device Pool, select Default (or some other locally-configured device pool).
6. For Phone Button Template, select Third-party SIP Device (Basic).
7. For Calling Search Space, select the calling search space the LS2000 is to use.
8. For Device Security Profile, select Code Blue LS2000 Profile.
9. For SIP Profile, select Standard SIP Profile.
10. For Digest User, select the end user matching the MAC address of the phone.
11. Click Save.
12. On the left side of the screen, click Line [1] - Add a new DN.
13. Fill in the Directory Number.
14. For Route Partition, select the partition the directory number resides in.
15. Under Line 1, for Display (Internal Caller ID), enter a descriptive name for Caller ID purposes.
16. If you wish to return a busy signal for silent monitoring if the LS2000 is in use, disable Call Waiting: under Multiple Call/Call Waiting Settings, For both Maximum Number of Calls and Busy Trigger, enter 1.
17. Click Save.



Configuring Phones and Directory Numbers for Secondary Accounts

If you are going to use the LS2000's secondary account functionality, repeat the above process with a local-use-only MAC address as outlined in Configuring End Users for Secondary Accounts, and specify a distinct directory number.

Integrating InformaCast Utilizing Cisco Call Manager

Access to the InformaCast emergency notification system produced by Singlewire Software frequently is included with Cisco Unified Communication Manager (CUCM). Code Blue's VoIP speakerphones (LS2000) can be registered and configured with CUCM as SIP devices that are compatible with IP Gateways and PBXs that can register third-party SIP devices.

To send audio pages from InformaCast to Code Blue speakerphones, select the Code Blue devices as end point phones for the messages. The Code Blue phones will have the ability to answer and play the audio by default.

14 Troubleshooting

The LS2000 is a network device. The following are tips for troubleshooting:

Power - Ensure the power to your device is working and rated for 802.11af/at PoE specifications if using PoE. LS2000 should only have one source of power other than a backup battery. If not using PoE, then main power needs to fall into 12-24 Volts DC.

Ping Test - This determines connectivity and the packet loss and latency time to and from your destination and the quality of your network connection to LS2000. If you receive no response and power is confirmed, contact your network administrator. You can also ping from within the phone towards your IP PBX to test that it can reach its IP PBX/SIP Gateway.

Network - If you are putting LS2000 on a network that restricts ports, then the following ports must be open for LS2000 to communicate to its appropriate IP PBX/SIP Gateway:

1. IAX2/UDP outgoing to port 4569 on IP PBX/SIP Gateway (only needed if using IAX2 instead of SIP on the accounts page).
2. SIP/UDP outgoing to port 5060 IP PBX/SIP Gateway.
3. RTP/UDP incoming from IP PBX/SIP Gateway to UDP ports 23456-23556 (configurable).

DHCP - LS2000 is set up for DHCP by default. If you cannot determine the IP address of your LS2000, contact your network administrator.

Account - Ensure your SIP account is set up correctly. Account username and password must match the account credentials on your VoIP system. This is the most common mistake with setting up SIP accounts.

Codec - Ensure the codec settings on your VoIP system match the LS2000 codec settings.

Firewall - Firewalls commonly block or partially block VoIP calls. Check with your network administrator if you cannot communicate with your LS2000 from behind a firewall.

Note: If you do not have a DHCP server running, use a standard home/wireless router and plug your speakerphone and laptop into the same router. Once you know the IP Address, you can browse to it via a web browser.

15 Remote Mount Beacon/Strobe Installation

ATTACH J-BOX TO THE POLE

- Thread the banding (B) through the pole bracket (A) located on the backside of the J-box (C).
- Wrap the banding around the pole. Cut the banding to desired length.
- Using a screwdriver or nut driver, tighten the banding and make sure that the unit is in the desired location.

NOTE: J-box must be positioned so weep hole faces down.

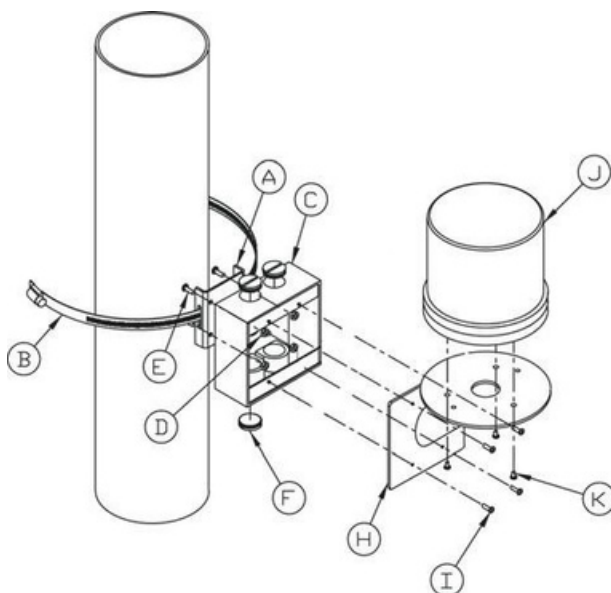
ATTACH LIGHT TO BRACKET

- Using the three M4 X 8 screws enclosed (K), fasten the strobe (J) to the round portion of the strobe bracket.

NOTE: If the beacon/strobe is mounted upside-down, a drain hole must be drilled into the lens to prevent it from filling with water.

ATTACH LIGHT AND BRACKET TO THE J-BOX

- Connect all wiring from the strobe to the wiring from the unit inside of the J-box using wire nuts.
- Attach strobe bracket to the J-box using four 6-32 X ½ screws as shown

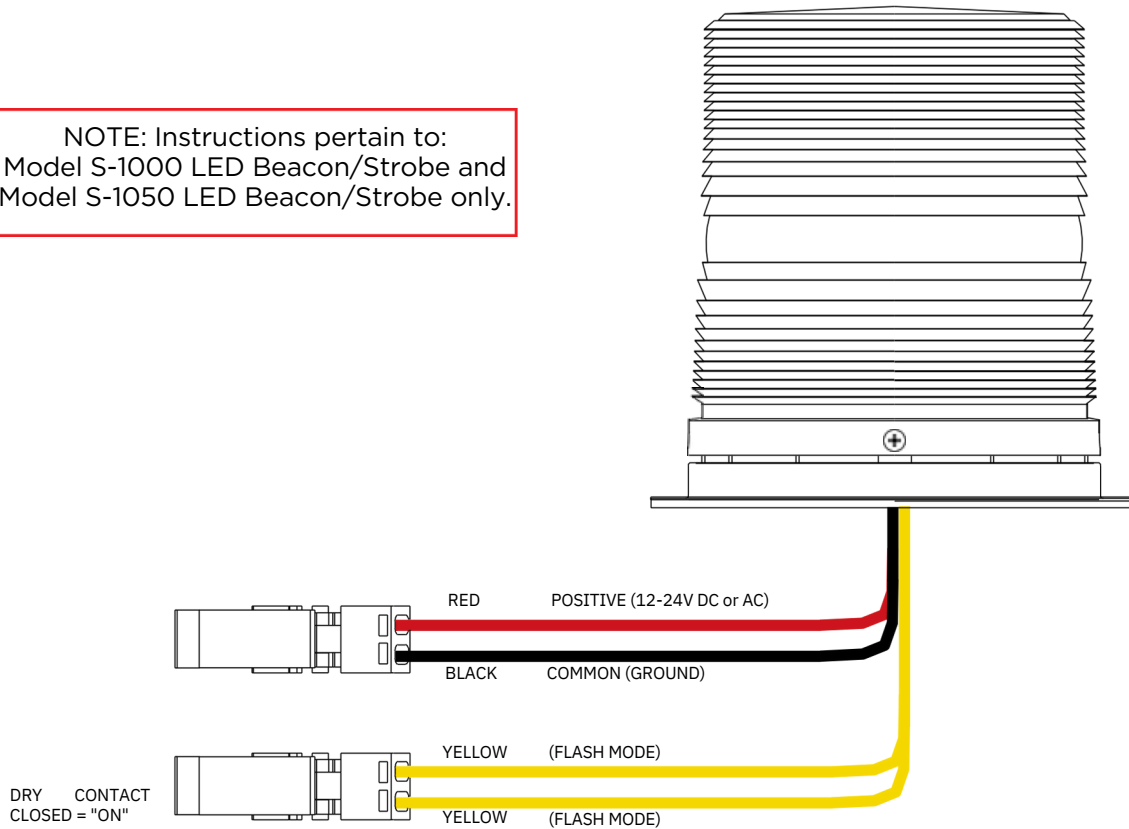


- A - pole-bracket
- B - banding
- C - J-box
- D - pole-bracket mount nut (4 each)
- E - pole-bracket mount screw (4 each)
- F - conduit plug
- H - strobe-bracket I - 6-32 X ½ screws (4 each)
- J - strobe light
- K - M4 X 8 screws (3 each) (Low voltage)
- K - 10-24 X ¾ screws (2 each) (High voltage)

All wiring must be installed and connected by experienced and certified personnel to meet local and national electrical codes, and will include a service disconnect.

S-1000 & S-1050 Strobe Operation

NOTE: Instructions pertain to:
Model S-1000 LED Beacon/Strobe and
Model S-1050 LED Beacon/Strobe only.



CAUTION ⚠️ **REMOVE ALL POWER FROM UNIT BEFORE SERVICING.**

OPERATION

To activate the LEDs in the PRIMARY-STEADYBURN MODE, connect the BLACK and RED wires to 12-24 volts AC or DC.

When in PRIMARY-STEADYBURN MODE, to change the LEDs to SECONDARY-FLASH MODE, connect both YELLOW control wires together (i.e., CLOSED = ON).

PHOTOCELL FEATURE (S-1050 MODEL)

The Steadyburn Mode will be ON in dark or night ambient environments and OFF in bright or daylight ambient environments. The S-1050 LED Beacon/Strobe has two built-in photo response features: (a) dawn/dusk transition delay of 15-30 minutes and (b) transient light acknowledgement delay of at least 3 minutes.

PROGRAMMING PRIMARY & SECONDARY MODES

1. Remove power from unit.
2. Short the Yellow wires together.
3. Restore power to the unit and wait until the unit begins to flash. Once the unit begins to flash, remove the short. The unit will alternately demonstrate the Secondary-Flash Mode and Primary-Steadyburn Mode that will be displayed during operation. For approximately 4 seconds the Secondary-Flash Mode will be demonstrated, followed by the Primary-Steadyburn Mode.
4. To select the next mode of operation, momentarily short the yellow wires. The unit will cycle to the next mode in the list above.

Mode Number	Primary-Steadyburn Mode	Secondary-Flash Mode
1	High	Single - 60 FPM
2	OFF	Single - 60 FPM
3	Low	Single - 60 FPM
4	High	Single - 150 FPM
5	OFF	Single - 150 FPM
6	Low	Single - 150 FPM
7	High	Single - 375 FPM
8	OFF	Single - 375 FPM
9	Low	Single - 375 FPM
10	High	Neobe - 75
11	OFF	Neobe - 75
12	Low	Neobe - 75
13	High	Neobe - 150
14	OFF	Neobe - 150
15	Low	Neobe - 150
16	High	Double - 125
17	OFF	Double - 125
18	Low	Double - 125
19	High	Double - 250
20	OFF	Double - 250
21	Low	Double - 250

5. There are seven Flash Modes and three Steadyburn Modes combinations to choose from.
6. When you reach the desired mode of operation, remove power from the unit. You **MUST** leave power disconnected for 20 seconds **BEFORE** reapplying. When power is reapplied, the unit will operate as programmed above.

NOTE: If you do not leave power disconnected for 20 seconds before reapplying power, the light will default to Program Mode.

INPUT VOLTAGE RANGE: 12-24V AC or DC		
TEMPERATURE RATING: -40°C to +65°C (-40°F to 149°F)		
TYPICAL POWER CONSUMPTION AT 25°C		
Voltage Flash Mode		Steady Mode - High
12V DC	0.24 A Max	0.24 A
24V DC	0.12 A Max	0.12 A
12V AC	1.1 A rms Max	0.53 A rms
24V AC	0.22 A rms Max	0.22 A rms
NOTE: Average current draw in Flash Mode will vary by selected Flash mode. The above maximum amperage draw is stated at Single 60 FPM.		

16 Maintenance Schedule

LEGEND

G Guard Tasks**T** Technician Tasks

DAILY OR WEEKLY

G Perform functional communications check.

- Action: Press Red Button
 - Strobe activates
 - Red LED "Call Placed" light turns on
 - Message plays
 - Call connects, green LED "Call Received" light turns on
 - Confirm conversation clarity with dispatch

MONTHLY OR QUARTERLY

G Visually check lighting functions:

- Faceplate light
- Beacon/Strobe

G Visually inspect unit for damage to:

- Faceplate
- Piezo Button
- Microphone
- Speaker

T Check Batteries:

- Functioning with full charge.
- Recharging fully, including NightCharge®/Solar Units (Note: Mid-to-late afternoon inspection is recommended)

IMPORTANT NOTE: Depending on the environment in which the batteries are installed, it is recommended that batteries for Solar & NightCharge® products are fully replaced every 2-3 years.

BIANNUALLY

T Remove access door and faceplate assembly to inspect the following:

- Ensure all electrical connections are secure
- Check all phone connections for corrosion (*If corroded, clean and coat with dielectric gel or replace*)
- Ensure all battery connections are tight and clean
- Verify no stains exist around gasket areas (stains indicate leaking & gasket should be replaced)
- Verify moisture weep hole on cabinet bottom is open and unobstructed
- Verify bottom of bollards are at least 1/2 inch above footing and free of obstructions (only applies to CB1, CB5, CB9, & CBRT units)

G Apply automotive paint sealant to unit exterior for protecting finish against environmental pollutants (Suggested products include Black Magic Wet Shine Liquid Wax, Nu Finish NFP-80, and 5 Star Shine)**G** Clean & coat exterior stainless steel cabinets with cleaner/polish (Suggested products include Chase Products' Champion Spray-on Stainless Steel Cleaner to help protect finish against environmental pollutants)**T** Visually confirm line-of-sight is still clear to base station (i.e., confirm that new tree growth, new building construction or other obstructions are not blocking view of base station)





UNIT SURFACE MAINTENANCE

The painted and stainless steel Code Blue models require periodic care to sustain their aesthetic appearance. Units located outdoors are vulnerable to harsh environmental conditions, including UV rays, acid rain, diesel fumes and airborne iron particles (i.e., dust) which over time may cause unit discoloring. To prevent pollutants developing harmful chemical reactions on Code Blue units, an appropriate surface maintenance schedule should be adhered to. The Surface Care Frequency table below provides general guidelines to assist in configuring a schedule. Please note that the frequency of care required to guard the Code Blue unit's surface from damage will also be dictated by local environmental characteristics.

LEGEND: POLLUTANTS LEVEL

Low	
Low/Moderate	
Moderate	
Moderate/High	
High	

SURFACE CARE FREQUENCY

	MONTHLY	BIMONTHLY	QUARTERLY	BIANNUAL	ANNUAL
Painted					
Stainless Steel					

See scheduled tasks under Biannually for suggested paint sealants or stainless steel cleaners.

AVERAGE COMPONENT LIFE

Component life is based on various mechanical, operational and environmental factors. Your local Code Blue reseller can assist you with a regularly scheduled maintenance program customized to your individual site requirements.

Code Blue strongly recommends contacting a local CB dealer to establish a proactive maintenance schedule.

17 Warranty

Code Blue Corporation provides a limited warranty on this product. Refer to your sales agreement to establish the terms. In addition, Code Blue's standard warranty language, as well as information regarding support for this product while under warranty, is available at www.codeblue.com/support

In Case of Breakdown

In case of system breakdown, discontinue use and contact Tech Support at:

technicalsupport@codeblue.com or call **800-205-7186, option 3.**

In Case of Abnormal Operation

If the unit emits smoke or an unusual smell, if water or other foreign material enters the enclosure, or if you drop the unit or damage the enclosure, power off the unit immediately and contact Code Blue Customer Service at:

customerservice@codeblue.com or call **800-205-7186, option 2.**

18 Download Information

Code Blue now has a centralized location where you can find installation, setup, information, configuration and operation instructions.

Admin Guides: www.codeblue.com/resources/guides

Firmware: www.codeblue.com/resources/firmware

Maintenance Tips: www.codeblue.com/support

Product Sheets: www.codeblue.com/resources/sheets

Specifications: www.codeblue.com/resources/specifications

For Legacy Product Information:

www.codeblue.com/legacy-products

These guides should contain all the information needed for your application. If further information is required, please contact customerservice@codeblue.com.

19 Legal & Regulatory Information

Legal Considerations

Video and audio surveillance can be regulated by laws that vary from country to country. Check the laws in your local region before using this product for surveillance purposes.

Liability

Every care has been taken in the preparation of this document. Please inform Code Blue Corporation of any inaccuracies or omissions. Code Blue cannot be held responsible for any technical or typographical errors and reserves the right to make changes to the product and manuals without prior notice. Code Blue makes no warranty of any kind with regard to the material contained within this document, including, but not limited to, the implied warranties of merchantability and fitness for a particular purpose. Code Blue shall not be liable or responsible for incidental or consequential damages in connection with the furnishing, performance or use of this material. This product is only to be used for its intended purpose.

Intellectual Property Rights

Code Blue Corporation has intellectual property rights relating to technology embodied in the product described in this document. This product contains open source code that also contains additional open source libraries.

Equipment Modifications

This equipment must be installed and used in strict accordance with the instructions given in the user documentation. This equipment contains no user-serviceable components. Unauthorized equipment changes or modifications will invalidate all applicable regulatory certifications and approvals.

Trademark Acknowledgments

Code Blue and Centry products are registered trademarks or trademark applications of Code Blue Corporation in various jurisdictions. All other company names and products are trademarks or registered trademarks of their respective companies.

Regulatory Information

Electromagnetic Compatibility (EMC)

This equipment has been designed and tested to fulfill applicable standards for:

- Radio Frequency emission when installed according to the instruction and used in the intended environment.
- Immunity to electrical and electromagnetic phenomenon when installed according to the instructions and used in its intended environments.

USA

This equipment has been tested using a shielded network cable (STP) and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at their own expense. The product shall be connected using a shielded network cable (STP) that is properly grounded.

Canada

This digital apparatus complies with CAN ICES-3 (Class A). The product shall be connected using a shielded network cable (STP) that is properly grounded.

Cet appareil numérique est conforme à la norme NMB ICES-3 (classe A). Le produit doit être connecté à l'aide d'un câble réseau blindé (STP) qui est correctement mis à la terre.

Disposal and Recycling

When this product has reached the end of its useful life, dispose of it according to local laws and regulations. For information about your nearest designated collection point, contact your local authority responsible for waste disposal. In accordance with local legislation, penalties may be applicable for incorrect disposal of this waste.

This guide should contain all the information needed for your application. If any further information is needed, please contact customerservice@codeblue.com.

Support

Should you require any technical assistance, please contact Code Blue.

Visit codeblue.com to:

- Download user documentation and software.
- Find answers to resolved problems in the FAQ database.

Report problems to Code Blue Technical Support via email at:

technicalsupport@codeblue.com or **800-205-7186**