IP1500 and IP2500 Series Speakerphones

Installation | Configuration | Support | Maintenance | Use

Administrator Guide

Code Blue®
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2 Introduction

Thank you for choosing the Code Blue IP1500 and IP2500 Series full duplex VoIP speakerphone(s), intercom and paging device(s) for indoor and outdoor applications. These speakerphones are 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 IP1500 and IP2500 Series speakerphone(s).

These devices are listed as non-emergency signaling and are non-monitored indoor/outdoor communication units.

IP2501-s Flush Mount

IP2501-d Flush Mount

IP1501 Flush Mount

IP2500-s Surface Mount

IP2500-d Surface Mount

IP1500 Surface Mount

3 Getting Started

This chapter provides information for obtaining the best performance with the IP1500 and IP2500 Series speakerphone. It is strongly recommended that the entire guide is read before configuring your IP1500 and IP2500 Series speakerphone to ensure you get maximum performance.

Throughout this guide you will see the following two references:
Calling party: This is the person activating the speakerphone by pressing a button.
Called party: This is the person receiving the call from the speakerphone; typically a guard, 911 operator, dispatch officer, etc.

These speakerphones provides powerful, yet flexible IP communication, and deliver excellent voice quality for your speakerphone, intercom and paging solution.

3.1 What's included with the IP1500 and IP1501 Series

IP1500 Surface Mount

Quantity	Part No.	Description	
1		Surface Mount Box	Included
1		Mounting Bracket	Included
4		No. 4 Security Screws	Included
4		No. 4 Screws	Not Included
1		Four Square Box	Not Included
1		Installation and Set Up Guide	Included

A Standard Drilled Spanner Insert Bit is required to the access unit – **Not Included**

IP1501 Flush Mount

Quantity	Part No.	Description	
1		Faceplate	Included
1		Weatherproof Back Box	Included
4		No. 4 Security Screws	Included
1		No. 4 Screws	Included

A Standard Tamper-Resistant Insert Bit is required to access unit – **Not Included**

3.2 What's included with the IP2500 and IP2501 Series

IP2500-s and IP2500-d Surface Mount

Quantity	Part No.	Description	
1		Surface Mount Box	Included
1		Mounting Plate	Included
2		Retaining Screws	Included
4		No. 8 Screws	Included
1		Standard Security Bit	Included
1		Installation and Set Up Guide	Included

IP2501-s and IP2500-d Flush Mount

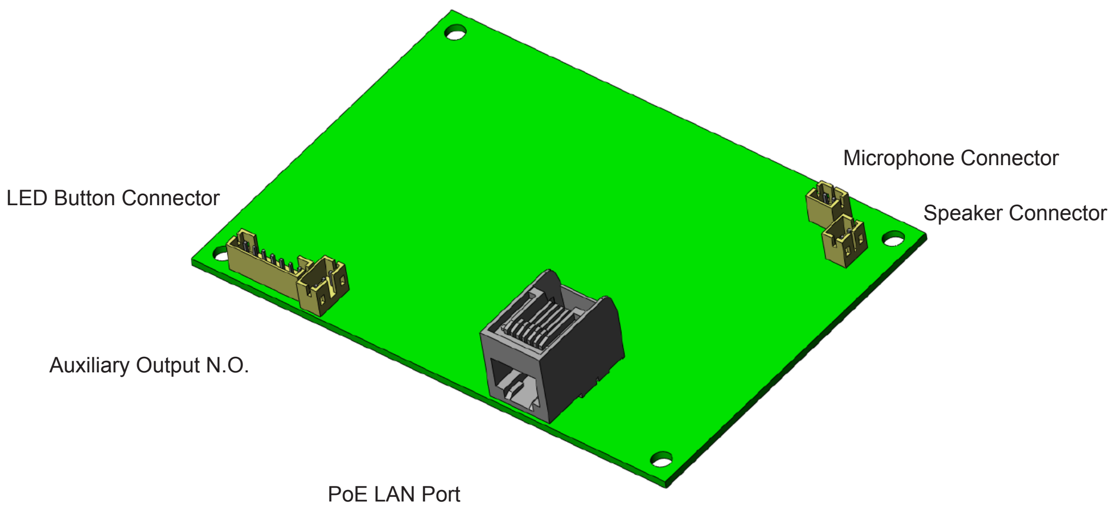
Quantity	Part No.	Description	
1		Faceplate Assembly	Included
1		Weatherproof Back Box	Included
4		Retaining Screws	Included
1		Standard Security Bit	Included
1		Installation and Set Up Guide	Included
		Mounting hardware for Back Box	Not Included

4 Connectors, Ports and Switch List

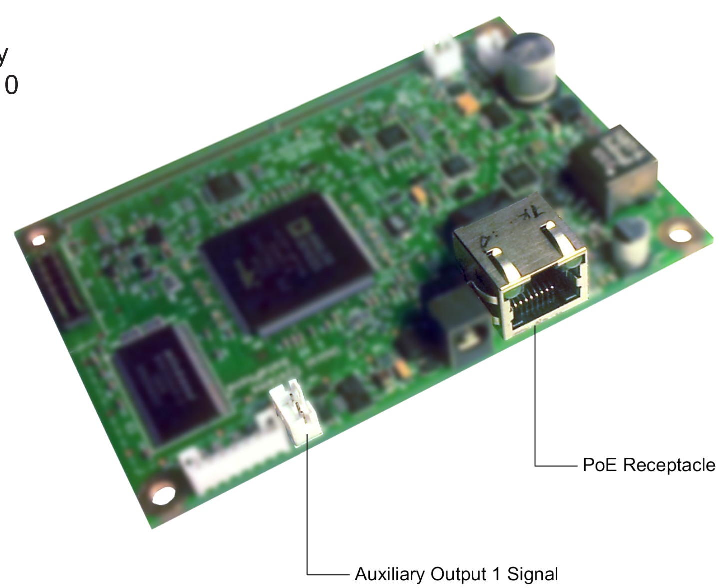
The IP1500 and IP2500 Series has a compact design small enough and is light enough to ensure an easy installation at any new or existing location. The IP1500/2500 has one PoE LAN port (IEEE 802.3 10/100 Ethernet Port) for connecting to a network. A single, normally open auxiliary output is available to trigger door locks, lights, gates or any other security device.

The internal components consist of a microphone connector, a speaker connector, one PoE LAN port, a button connector(s), one normally open auxiliary output and PCB mounting hardware.

4.1 IP1500 and IP2500 Series Printed Circuit Board Layout

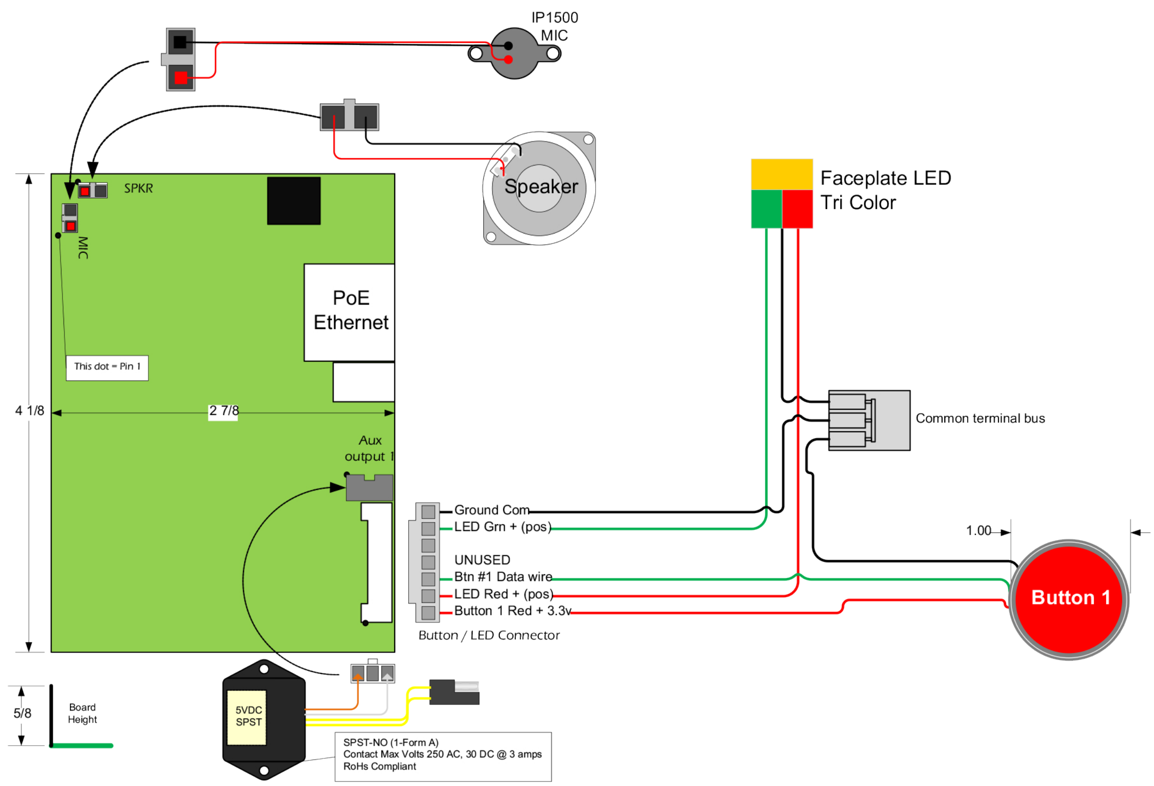


The IP1500 and IP1501 Series is solely powered by PoE, 802.3at and af Class 0 upon initialization.



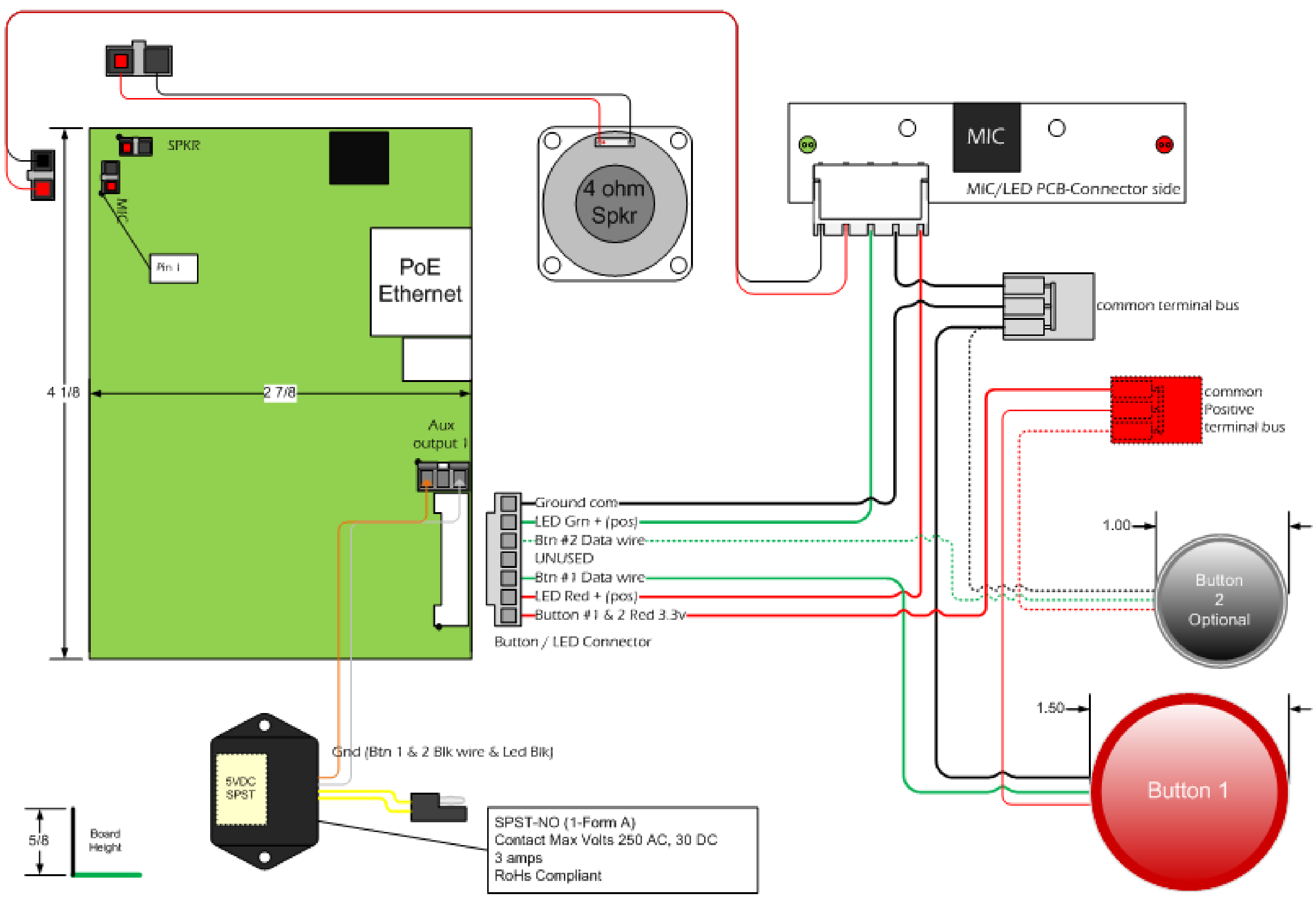
4.2 Wiring Diagram IP1500 and IP1501 Series

The IP1500 and IP1501 Series has one PoE LAN port for network connectivity and one normally open auxiliary output relay for triggering devices such as a LED beacon/strobe, camera preset activation inputs, gate control or third party controllers. The relay can be programmed to hold the relay closed for the duration of the call, for a specific time period or toggled for a specific time period upon the operator's request.



4.3 Wiring Diagram IP2500 and IP2501 Series

The IP2500 and IP2501 Series has one PoE LAN port for network connectivity and one normally open auxiliary output relay for triggering devices such as a LED beacon/strobe, camera preset activation inputs, gate control or third party controllers. The relay can be programmed to hold the relay closed for the duration of the call, for a specific time period or toggled for a specific time period upon the operator's request. The IP2500 and IP2501 Series also has the option of an additional button for non-emergency calls.

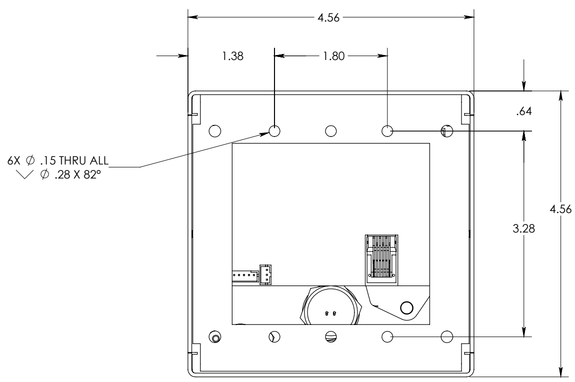


5 Installation

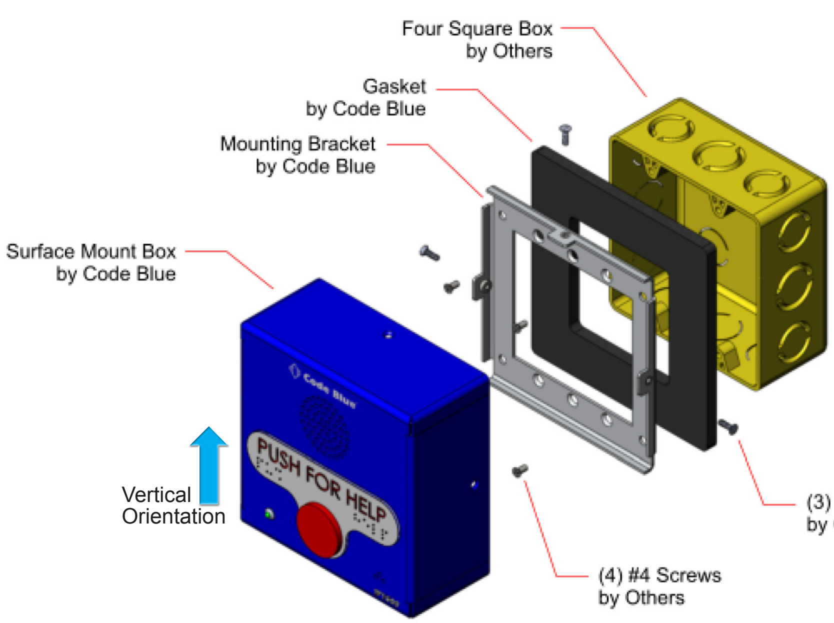
The IP1500 and IP2500 Series comes in surface and flush mount options. The surface mount allows the mounting bracket to be installed during rough in and the faceplate with electronics during completion. The faceplate on the flush mount is five inches square and provides an overlap to the mounting box to eliminate additional trim work.

5.1 Surface Mount IP1500 Series

IP1500 Surface Mount: 4.50"w x 4.50"h x 2.00"d



Back View

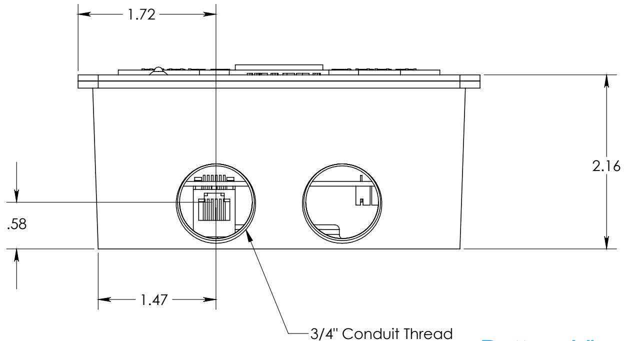


- Mount standard four square box to stud
- Place gasket between Mounting Bracket and Four Square Box.
- Attach mounting bracket to four square box with (4) #4 screws
- Connect CAT 5e cable to the Ethernet port on the PCB
- Attach surface mount box to mounting bracket with (3) #4 security screws provided

NOTE: Thread seal must be applied to external case screws.

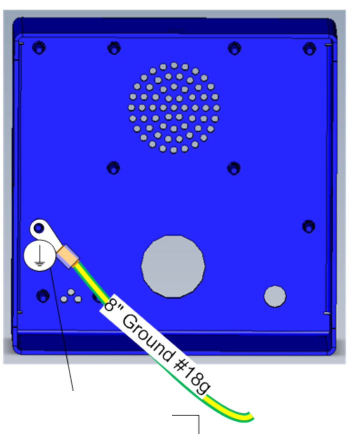
5.2 Flush Mount IP1501 Series

IP1500 Flush Mount: 4.00"w x 5.00"h

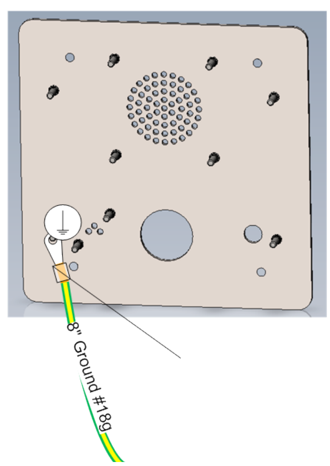


Bottom View

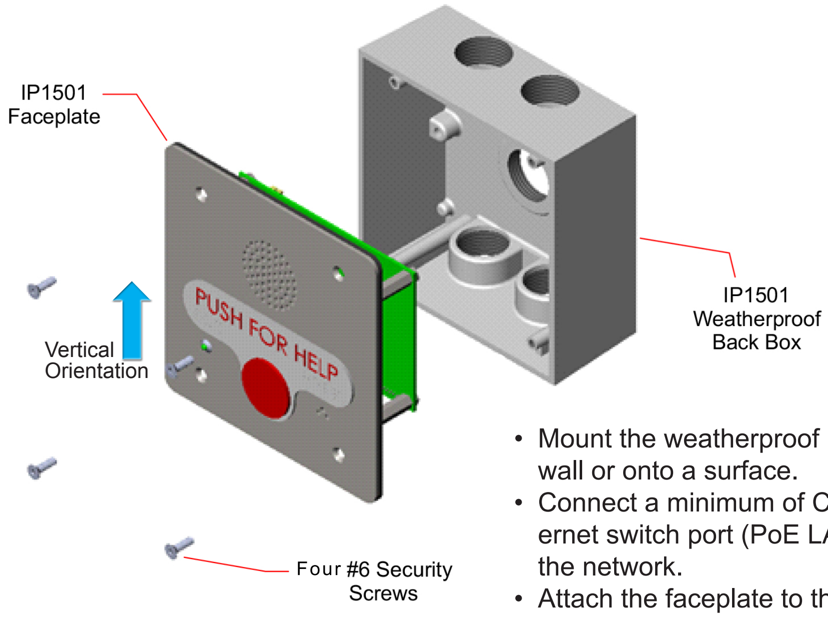
IP1500 Faceplate - Backside Grounding



IP1501 Base Faceplate - Backside Grounding



Ground lug will be placed on a stud with bare metal on one side and a kepnut holding it in place. The lug used for the 1500 requires a #6 eyelet type lug.

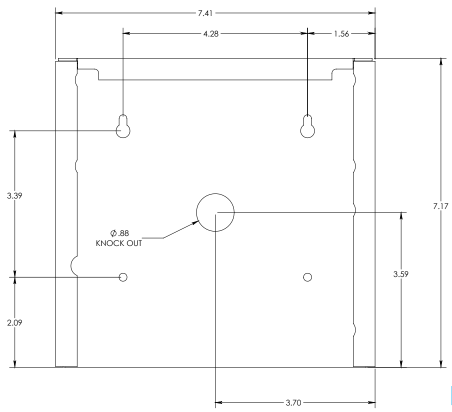


- Mount the weatherproof back box to an inside wall or onto a surface.
- Connect a minimum of CAT 5e cable to the Ethernet switch port (PoE LAN) for connectivity to the network.
- Attach the faceplate to the back box with the four - #6 security screws provided.
- Connect any auxiliary device to the N.O. aux output 1

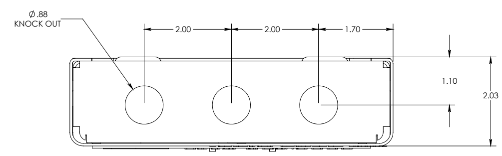
NOTE: Thread seal must be applied to the four faceplate security screws.

5.3 Surface Mount IP2500 Series

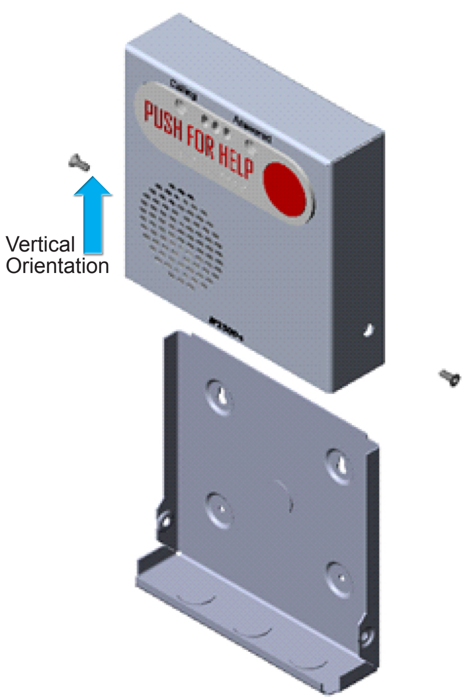
IP2500 Series Surface Mount: 7.25”w x 7.25”h x 2.00”d



Back View



Bottom View

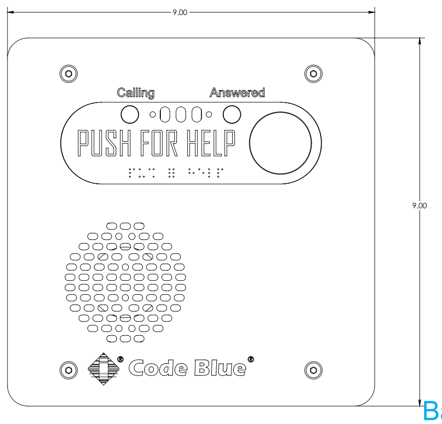


- First, remove the two retaining screws, one from each side of the case
- Remove the rear mounting plate by sliding it downward
- Use the mounting plate as your guide on the wall, mark the mounting and conduit holes
- Remove the rear mounting plate and create the required holes, attach the mounting plate to the wall
- Pull CAT 5e cable through the conduit hole in the rear mounting plate
- Connect CAT 5e cable to the Ethernet port on the PCB
- Slide the IP2500 Series case down, starting from the top of the rear mounting plate
- Replace the two retaining screws

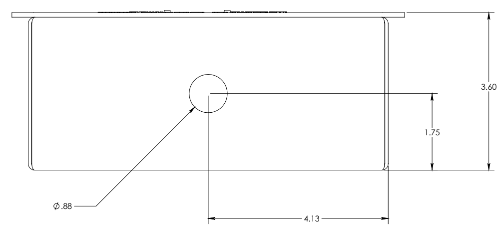
NOTE: By design, no main gasket is needed to seal the enclosures. The installer is responsible for sealing all screws with thread seal, as well as applying the proper conduit for a weather-tight seal.

5.4 Flush Mount IP2501 Series

IP2501 Series Flush Mount: 9.00”w x 9.00”h

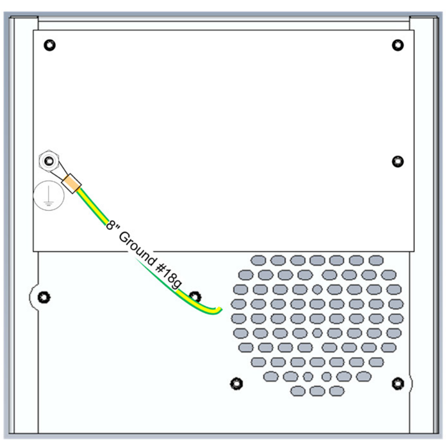


Back View

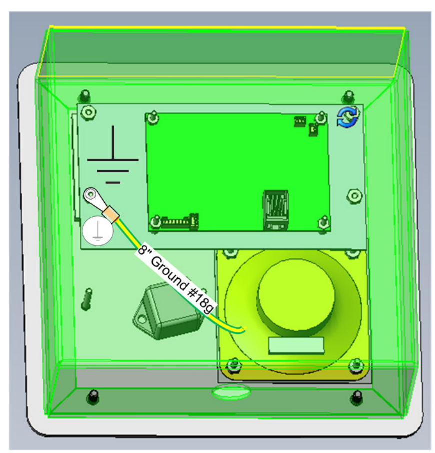


Bottom View

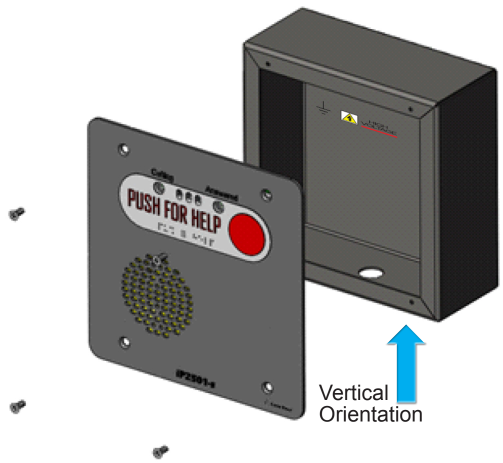
IP2500 Series Faceplate - Backside Grounding



IP2501 Series Base Faceplate - Backside Grounding



Ground lug will be placed on a stud with bare metal on one side and a kepnut holding it in place. The lug used for the 2500 requires a #6 eyelet type lug.



- First, remove the four faceplate screws from the back box
- Use the back box as your guide on the wall, mark the rough opening size
- Create the required opening and conduit runs
- Pull CAT 5e cable through the conduit hole in the back box
- Connect CAT 5e cable to the Ethernet port on the PCB
- Attach faceplate to back box
- Replace the four faceplate screws

Special Warning: High Voltage – Range 37 - 57V DC <12.96 watts (27mA)

Grounding
The IP2500 series is UL 2017 safety rated. Due to the standard PoE 802.3af power supplied by the remote (IEEE 802.3af or "at" rated) PoE Switch, the IP2500 has a protective earthing terminal cable available. There must be an uninterruptible safety earth ground attached to the grounding lanyard provided. Whenever it is likely that the protection has been impaired, disconnect the Ethernet cable until the ground has been restored. This is a non-emergency, non-monitored product line.

Ground terminations points have been marked in accordance with UL 2017 standards. Approved icons have been applied to indication the grounding location.

Ground Wire Connection
Locate the 18g green ground wire, which is terminated with a ring lug that has been secured to a faceplate stud. The loose end of the green ground wire must be secured to an earth ground nearby.

If the enclosure has marked grounding location, and a green grounding screw is available, please use it whenever local codes dictate earth ground is required. Ground termination locations can be labelled using the following graphic symbol:

Ground Images sample: ⏚

There are no user-serviceable parts inside these products. Any servicing, adjustment, maintenance or repair must be performed only by service-trained personnel. These products do not have a power switch; they are powered on when the Ethernet PoE cable is plugged in.

ELECTRICAL SAFETY WARNINGS
This device is suitable for use in non-hazardous locations only.

WARNING: Explosion Hazard – Do not replace the device unless power has been switched off or the area is known to be non-hazardous.

WARNING: Do not operate the equipment in the presence of flammable gasses or fumes. Operating electrical equipment in such an environment constitutes a definite safety hazard.

WARNING: If the equipment is used in a manner not specified by Code Blue Corp., the protection provided by the equipment may be impaired.

WARNING: Do not perform any services on the unit unless qualified to do so. Do not substitute unauthorized parts or make unauthorized modifications to the unit.

WARNING: Properly ground the unit before connecting anything else. Units not properly grounded may result in a safety risk and could be hazardous and may void the warranty. See the grounding technique section for proper ways to ground the unit.

WARNING: Do not operate the equipment in a manner not specified by this manual.

WARNING: Do not work on equipment or cables during periods of lightning activity.

WARNING: Install only in accordance with Local & National Codes of Authorities Having Jurisdiction. (Revised 2010-11-15) 5

Shield Patch Cable

Note: Before applying power to the grounded switch, you must use a volt meter to verify there is no voltage difference between the power supply's negative output terminal and the switch chassis grounding point. If the use of shielded cables is required, it is generally recommended to only connect the shield at one end to prevent ground loops and interfere with low level signals (i.e. thermocouples, RTD, etc.). Cat5e cables manufactured to EIA-568A or 568B specifications are required for use with switches.

In the event all Cat5e patch cable distances are short(i.e. all Ethernet devices are located the same local cabinet and/or referenced to the same earth ground), it is permissible to use fully shielded cables terminated to chassis ground at both ends in systems void of low level analog signals.

6 Provisioning the Phone

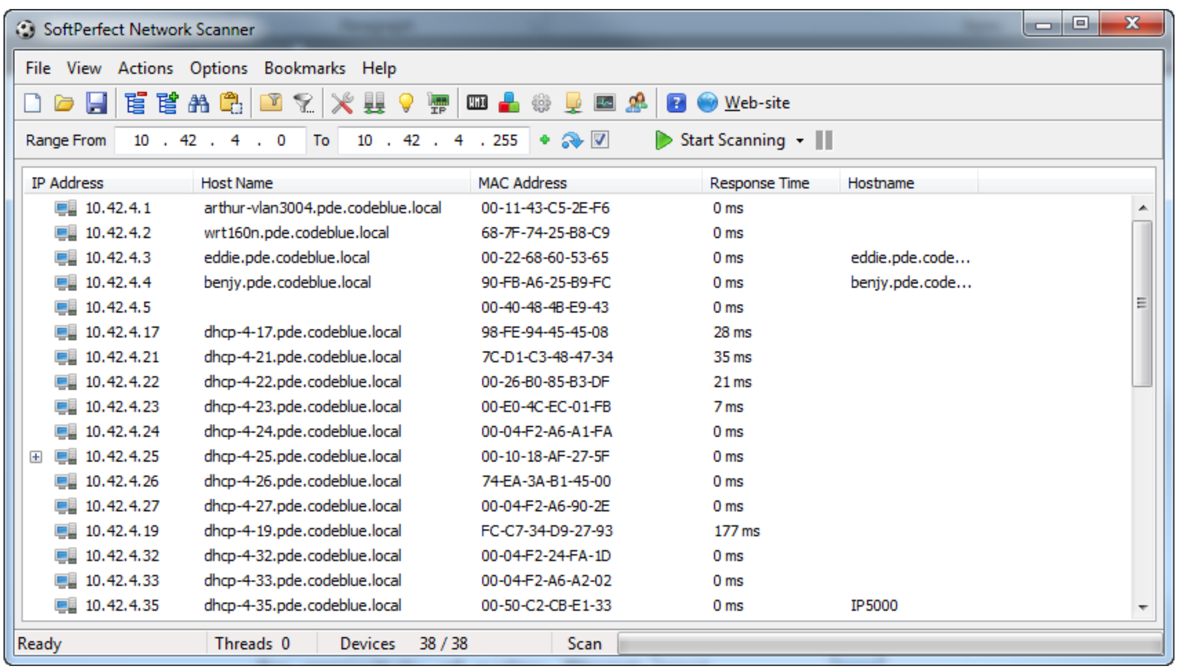
6.1 Determine the IP Address

All IP1500 and IP2500 Series speakerphones are DHCP by default. A programming video is available at www.codeblue.com/support/how-to-videos.

1. Connect the speakerphone to your network. The LED will flash momentarily and an audible beep will be heard out of the speaker to indicate the OS is loading. The IP1500/2500 speakerphone will acquire IP Network settings from your DHCP server.

2. Check your DHCP lease records or utilize a network scanner such as SoftPerfect's Network Scanner to match the MAC address of the speakerphone to the correct IP address in your lease table or output of the network scanner.

IP Address	Ethernet	Hostname	Start Date	End Date
172.1.100.234	00:0f:1f:17:55:63	IP5000	2010/09/29 04:52:45	2010/09/29 16:52:45
172.1.100.228	00:1c:c0:b0:41:e6	IP5000	2010/09/29 05:26:40	2010/09/29 17:26:40
172.1.100.238	00:1c:c0:b0:3a:20	IP5000	2010/09/29 09:17:08	2010/09/29 21:17:08
172.1.100.234	00:0f:1f:17:55:63	IP5000	2010/09/29 09:53:35	2010/09/29 21:53:35



Lease Table and Network Scanner Example

Logging Into The System

1. Log in using a web browser.

   A. Place the IP Address of your speakerphone into the URL address bar and press ENTER.

   B. Depending on the browser being used, a certificate warning may pop up. Go ahead and approve in order to load up the login dialog box.

   C. Enter user name “admin” and password “admin” and press ENTER.

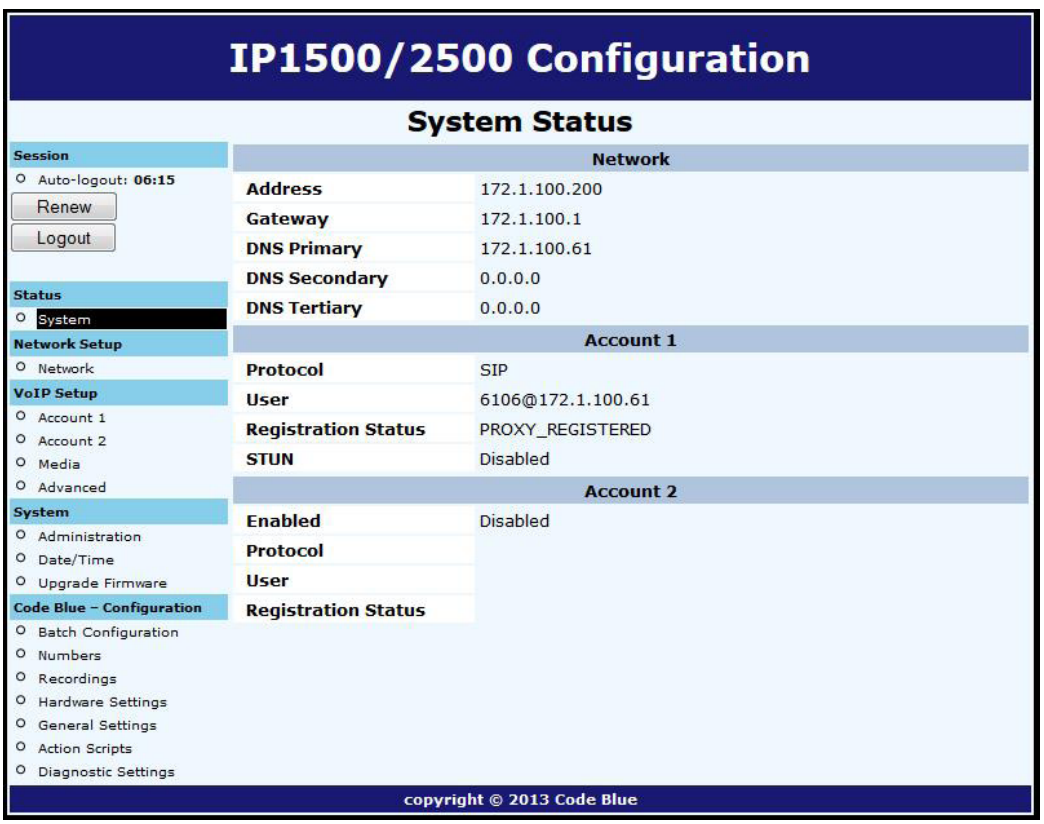
2. System Status Screen.

   A. Current session time before Auto-Logout is executed.

   B. Clicking **Renew** will restart the timer to 10 minutes, effectively keeping you logged in. This state helps prevent others from logging in and taking over the session, therefore erasing any unsaved changes made.

   C. Clicking **Logout** will log you out of the GUI.

   D. Network: Displays current IP address, DNS address, DNS Tertiary address, Account 1’s current status and Account 2’s current status.



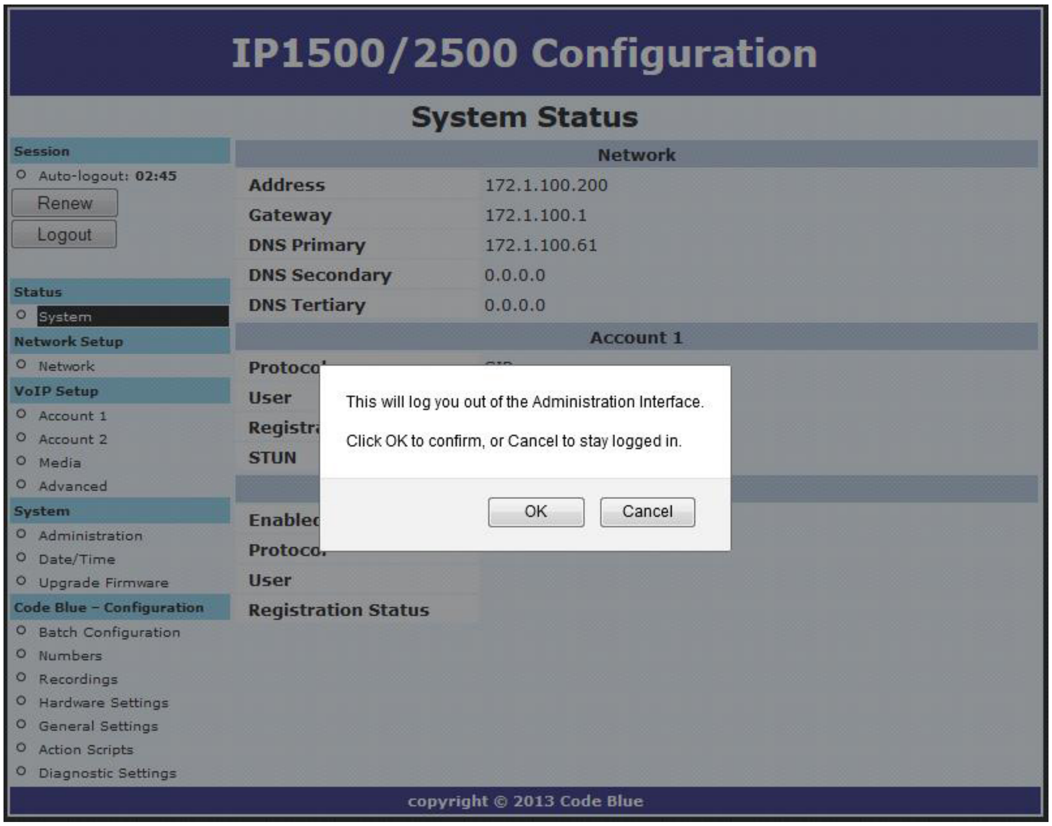
Logging Out Of The System

1. To log out of the speakerphone, simply click on **Logout** under **Session** (see far left-hand column).

   The speakerphone will also log you out automatically after 10 minutes.

   You will be prompted for confirmation.

2. Click **OK** to complete the logout process or **Cancel** to continue configuring your speakerphone.

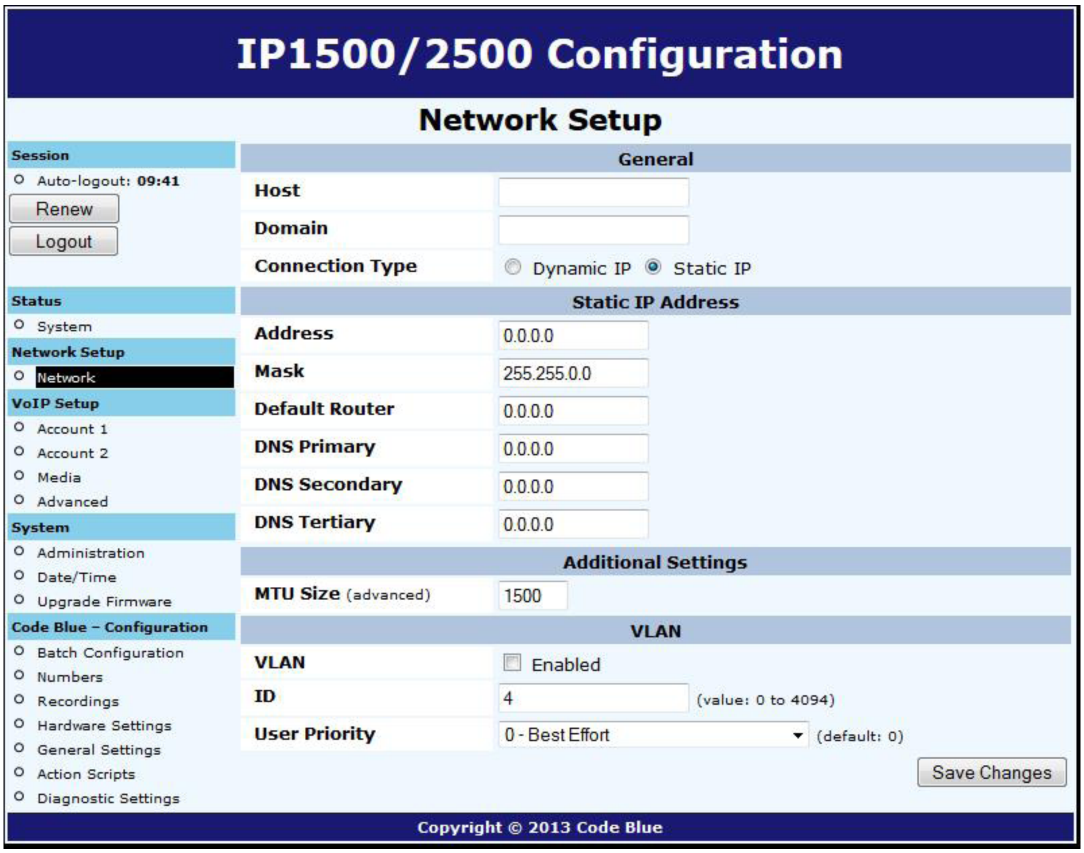


6.2 Network Configuration

Once you have obtained the DHCP address of the speakerphone you can log in and set a static IP address.

1. Click on the **Network** menu item under Network Setup (see far left-hand column).
2. Under General, click on **Static IP** for **Connection Type.**
3. Enter your desired IP settings under Static IP Address.
4. Once you have entered your settings, click on **Save Changes.**

Note that if you have moved your speakerphone to a network your PC cannot access, you will have to configure your PC to access that network before configuration can continue.



VLAN Configuration

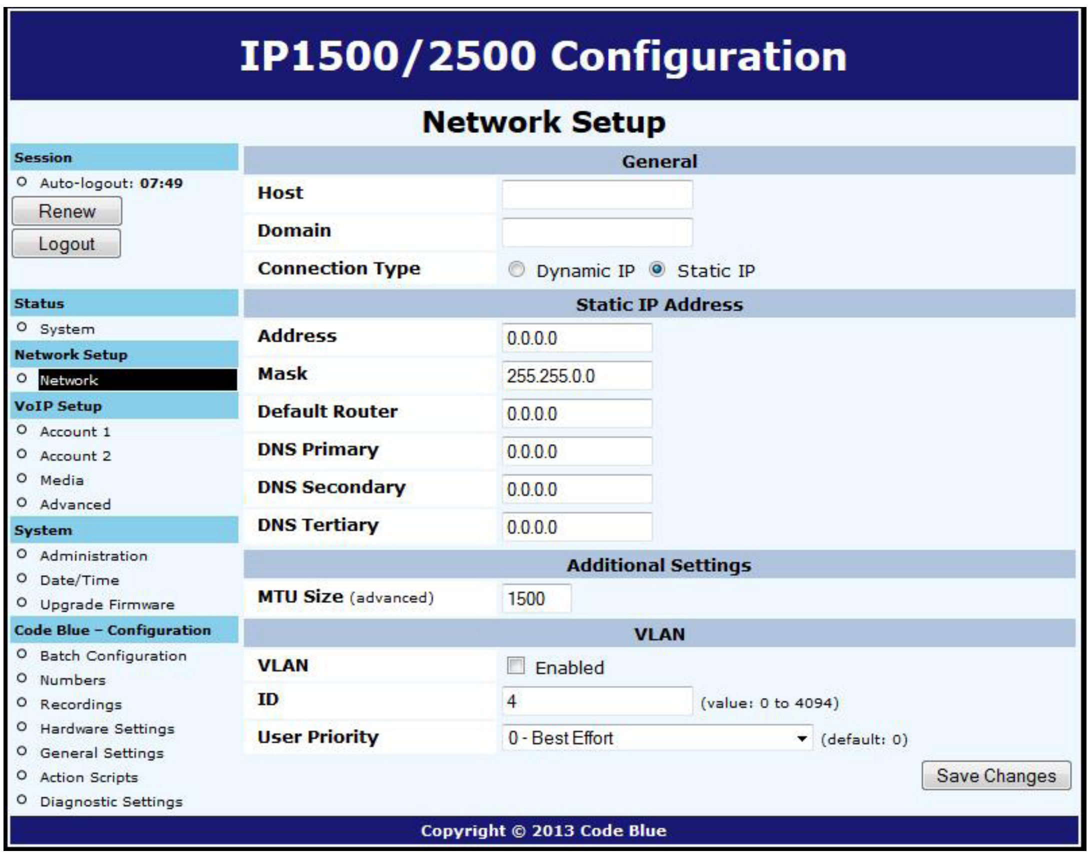
The speakerphone is capable of performing IEEE 802.1Q frame tagging and user priority settings.

1. Click on the **Network** menu item under **Network Setup** (see far left-hand column).

2. Then click on the **VLAN Enabled** check box in the VLAN section and select your desired VLAN ID and User Priority.

3. Once you have entered your settings, click on **Save Changes.**

Note that if your PC cannot access the new VLAN, you will have to correct this problem before continuing configuration, as you will lose access to the speakerphone. If you wish to disable VLAN support and cannot reach the speakerphone on its configured VLAN, factory-reset the unit to clear network configuration.



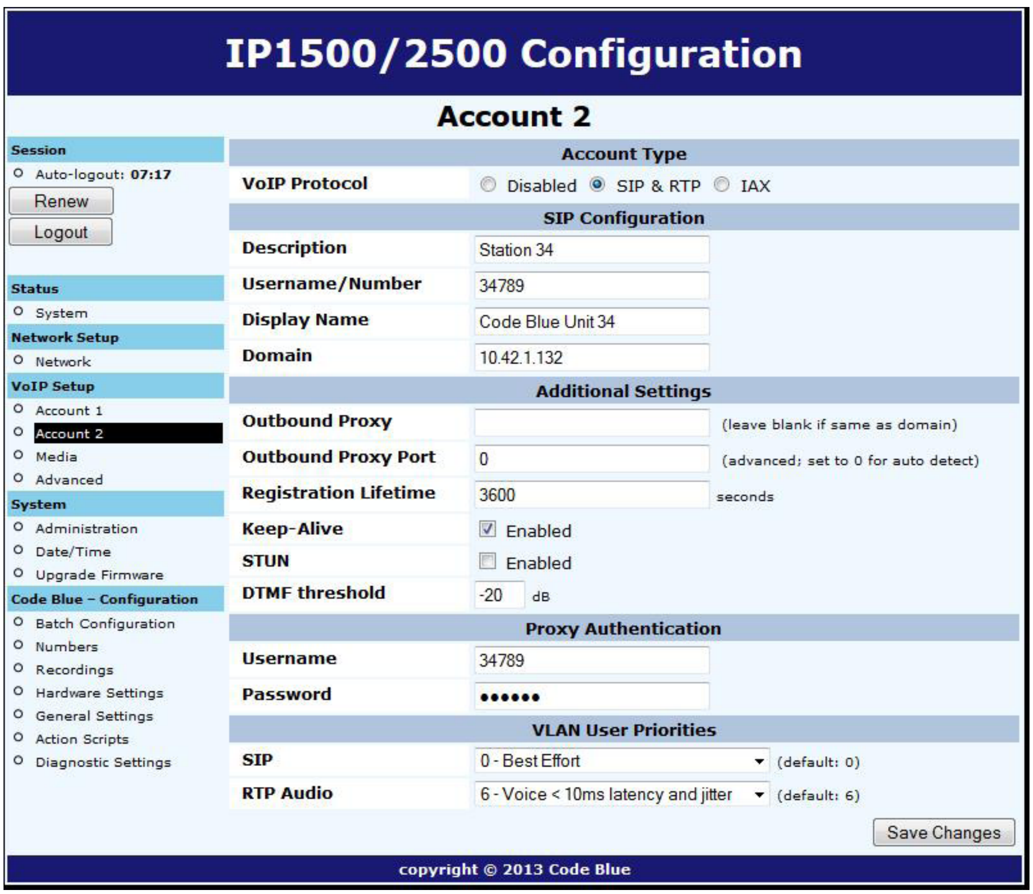
6.3 Configuring VoIP Settings

The IP1500 and IP2500 Series speakerphones are an advanced VoIP devices capable of connectivity to VoIP systems via SIP and IAX2 protocols. Built-in codecs provide multiple options for communicating with your VoIP system or Code Blue's ToolVox Media Gateway. STUN server capabilities are also built in for helping traverse firewalls when connecting the unit outside of the hosting network.

Configuring VoIP Accounts

The speakerphone can register to VoIP systems using either the SIP or IAX protocols, and has the ability to register to two separate VoIP systems simultaneously to provide redundancy.

Each of the speakerphone's two accounts, available under VoIP Setup as Account 1 and Account 2, can be configured as either SIP or IAX, subject to the limitation that you can only have one of the two accounts configured as IAX. If you wish to use only one account, set Account 2 to Disabled.

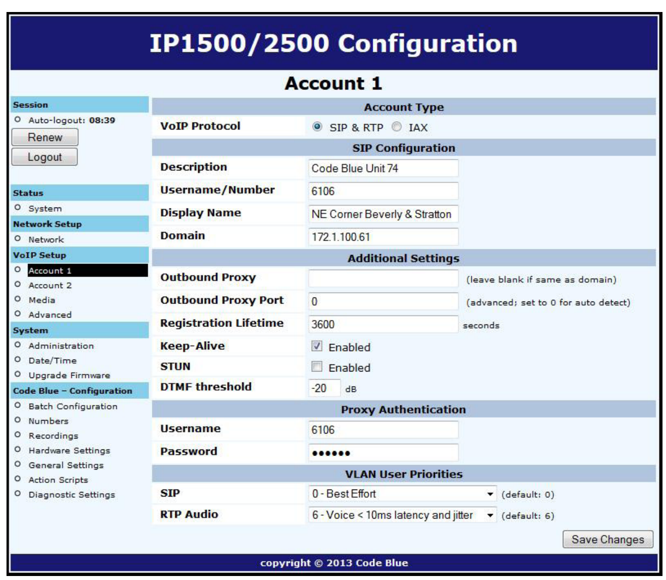


Configuring a SIP Account

Either of the speakerphone's two accounts can be configured to register to a VoIP system via SIP.

Configuration is as follows:

- Set the VoIP Protocol to SIP & RTP.
- For Description, enter a name the speakerphone will use internally to refer to this account.
- For Username/Number, enter the number that the speakerphone will use for SIP addressing. This will often be the extension number in a VoIP-based PBX.
- For Display Name, enter the display name the speakerphone will send in SIP transactions. This will often be the calling name of the extension.
- For Domain, enter the domain the speakerphone will register to.
- For Outbound Proxy, enter a SIP proxy the speakerphone should send outbound calls to. If this is the same as the domain, you can leave this field blank.
- For Outbound Proxy Port, enter an IP port number the speakerphone will send outbound calls to. Typically, this should be left at 0.
- For Registration Lifetime, enter the time in seconds the speakerphone will request that its registration be valid for. The speakerphone will automatically re-register before this time period expires.
- Check Keep-Alive if you want the speakerphone to periodically send OPTIONS requests to the SIP server, e.g. to keep a NAT connection alive.
- Check STUN if you want to enable STUN support for this account.
- You can adjust the DTMF Threshhold value if you have difficulties with the speakerphone activating in-call commands when no DTMF is present.
- For Username and Password, set the username and password the speakerphone will use to authenticate to the domain and outbound proxy. Note that the username is used for authentication only and need not match the Username/Number field if the VoIP system does not expect it to.
- VLAN user priorities can be adjusted for SIP and RTP audio.

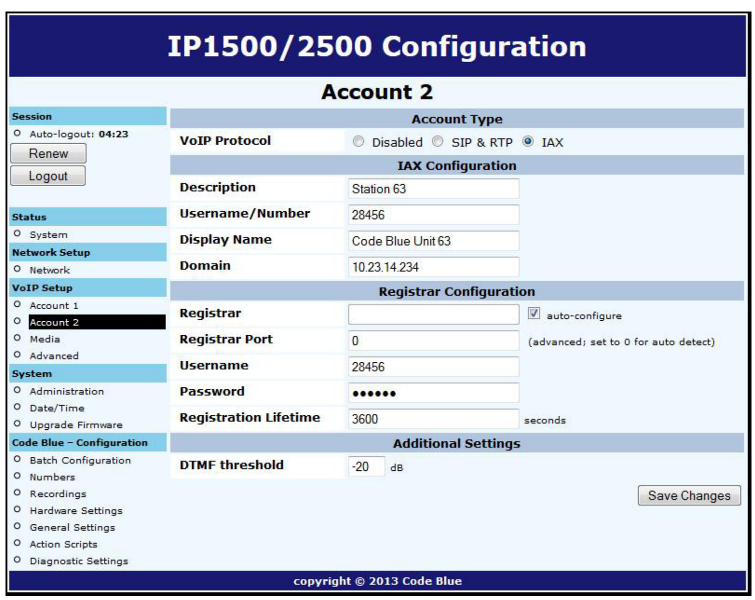


Configuring an IAX Account

Either of the speakerphone's two accounts can be configured to register to a VoIP system via IAX. (Note, however, that only one of the two accounts may be configured as IAX - the speakerphone does not support two simultaneous IAX accounts.)

Configuration is as follows:

- Set the VoIP Protocol to IAX.
- For Description, enter a name the speakerphone will use internally to refer to this account.
- For Username/Number, enter the number that the speakerphone will use for IAX addressing. This will often be the extension number in a VoIP-based PBX.
- For Display Name, enter the display name the speakerphone will send in IAX transactions. This will often be the calling name of the extension.
- For Domain, enter the domain the speakerphone will use in its IAX address.
- For Registrar, enter the address of the IAX server the speakerphone should register and send outbound calls to. If this is the same as the domain, you can leave this field blank.
- For Registrar Port, enter an IP port number the speakerphone will register and send outbound calls to. Typically, this should be left at 0.
- For Username and Password, set the username and password the speakerphone will use to authenticate to the domain and outbound proxy. Note that the username is used for authentication only and need not match the Username/Number field if the VoIP system does not expect it to.
- For Registration Lifetime, enter the time in seconds the speakerphone will request that its registration be valid for. The speakerphone will automatically re-register before this time period expires.
- You can adjust the DTMF Threshhold value if you have difficulties with the speakerphoneactivating in-call commands when no DTMF is present.



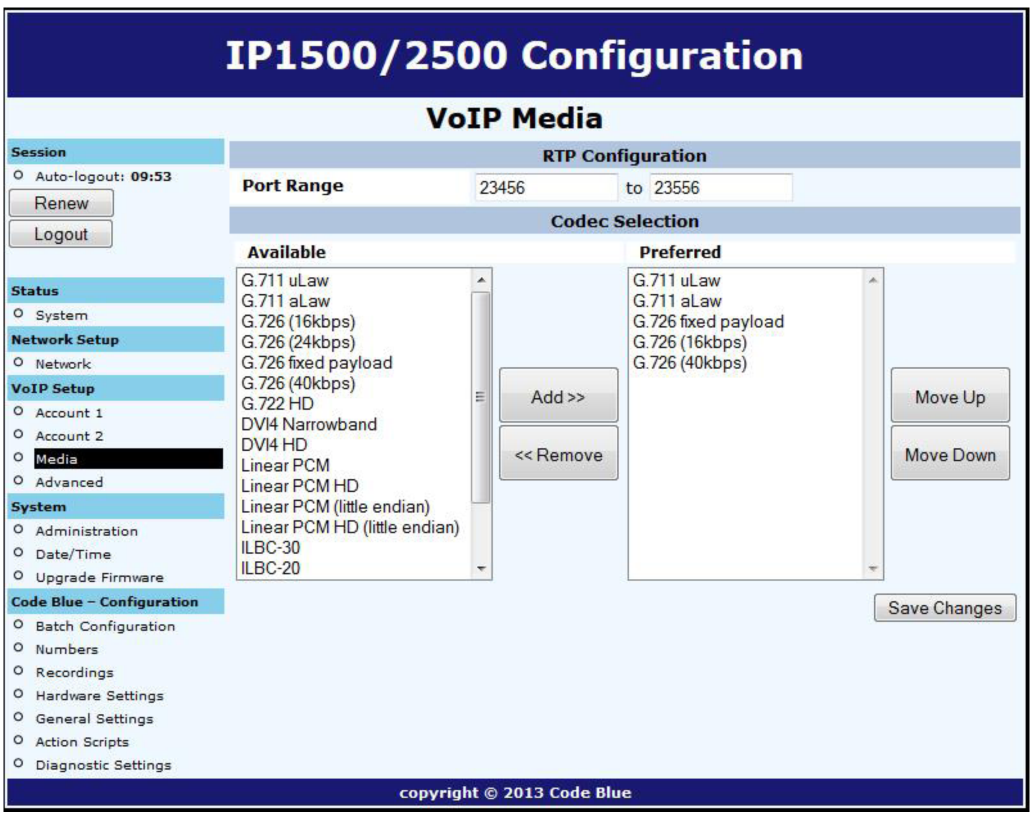
Configuring Media Settings

For the SIP protocol, you can specify a port range from which the speakerphone will select IP ports to offer to the other system for use with RTP communication.

The speakerphone can use any one of a suite of codecs for voice communication. Which codec is used is dependent on negotiation with the remote system, but you can use Codec Selection to specify a list of preferred codecs that will be offered in negotiation.

- To add codecs to the **Preferred** list, highlight them in the **Available** list and click **Add.**
- To remove codecs from the **Preferred** list, highlight them and click **Remove.**
- To change the order preferred codecs are offered, highlight them and click either **Move Up** or **Move Down** to reorganize them.

Note that some codecs corrupt DTMF tones, e.g. G.729. If RFC2833 out-of-band DTMF signaling is not in use, be sure to configure your codecs appropriately or you may not be able to use in-call commands. Be sure to test your configuration to make sure all features are available.



Configuring Advanced Settings

The speakerphone can be configured to utilize a STUN server for transversal of firewall devices for the setup of a VoIP call.

1. Click on **Advanced** under **VoIP Setup** (see far left-hand column) to configure the STUN server IP address and Port.

2. Upon completion, click **Save Changes.**

6.4 Configuring the System Settings

The speakerphone system administration is provided under the System Settings dialog, which allows you to change the following:

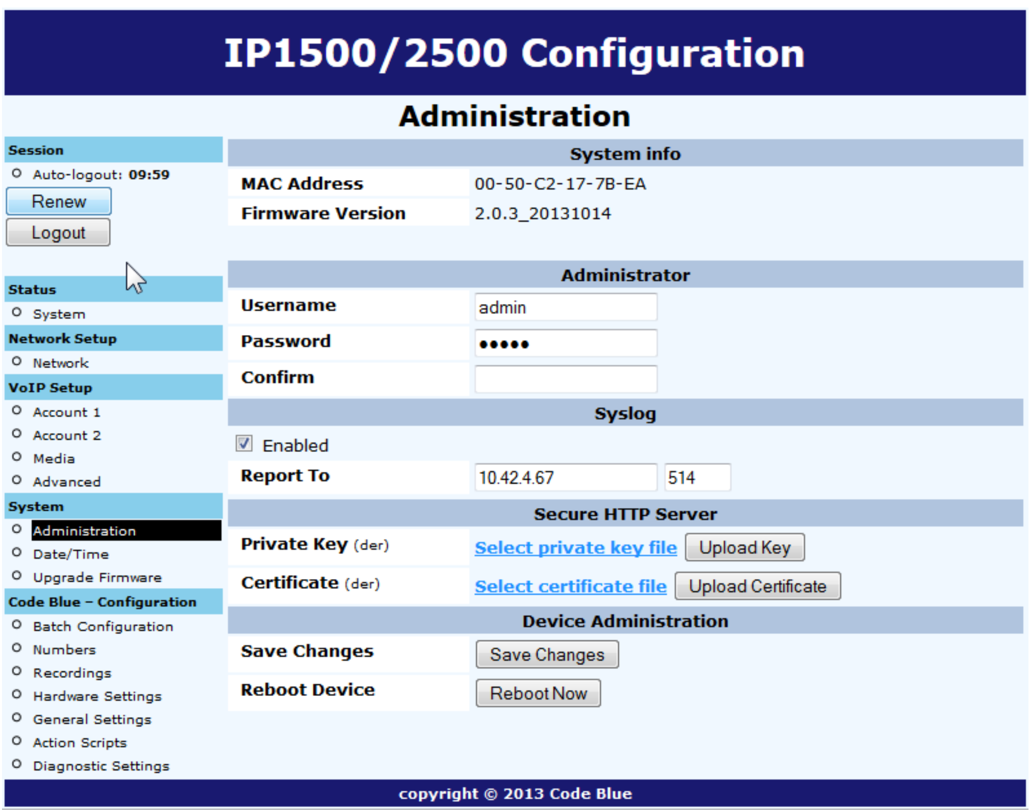
- **Administrative Logon Credentials**
- **Syslog Service Reporting**
- **Secure HTTP Server**
- **Date and Time**
- **Upgrade Firmware**

System Administration Settings

The Administration page under System contains several system settings:

- The **System Info** section displays the MAC address and firmware version running on the speakerphone.
- The **Administrator** section allows changing of the administrator username and password. Enter a new **Username,** if desired, and enter the new **Password** and again in the **Confirm** box to change these parameters.
- The speakerphone can send RFC 5424 Syslog messages to a **Syslog** server by specifying it in the Syslog section.
  Note that Syslog messages are only useful for advanced troubleshooting and are not intended for general monitoring.
- A new private key and certificate can be uploaded to the speakerphone's **Secure HTTP Server** if you do not wish to use the system's built-in key and certificate. The key should be PKCS#8, DER-formatted and the certificate X.509, DER-formatted.

When you are finished making changes, click **Save Changes.** You can also reboot the device directly from this page by clicking **Reboot Now.**



Date and Time Configuration

The speakerphone date and time are managed by:

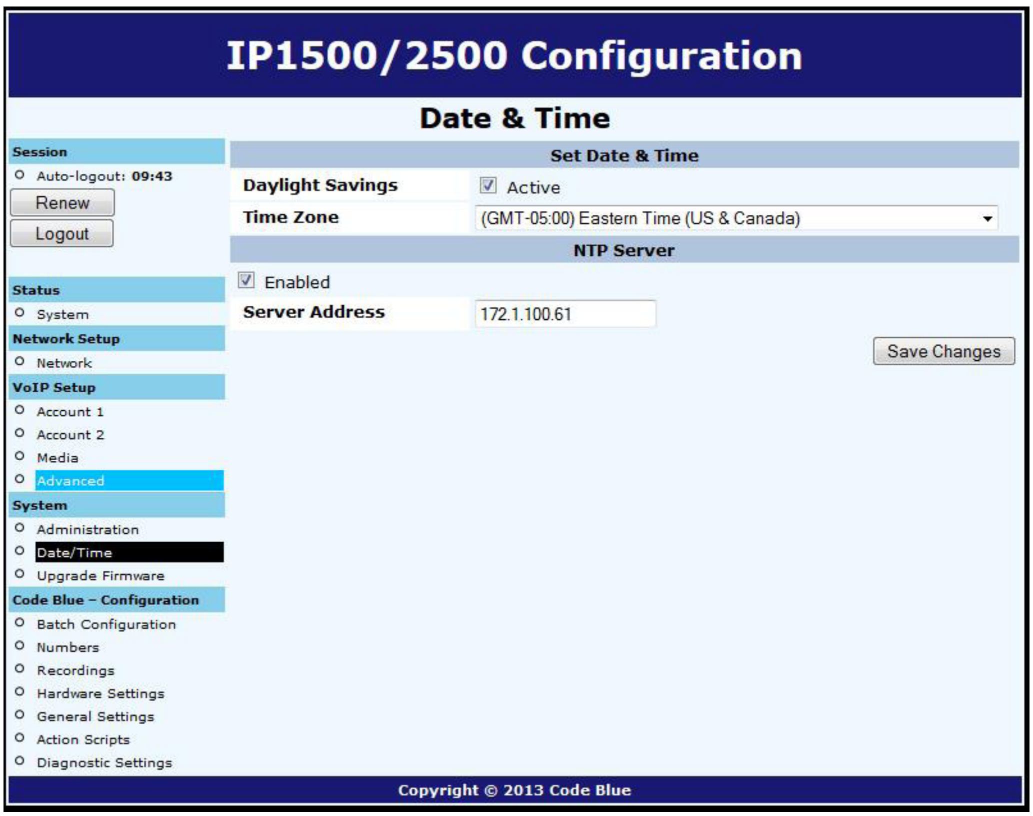
1. Clicking **Date/Time** under **System** (see far left-hand column).

   Under the **Set Date & Time** section, you can manually set the Date, Time, Daylight Savings (if applicable) and Time Zone.

2. To automatically synchronize with an NTP (Network Time Protocol) server, check **Enabled** and enter the IP or URL of the NTP server (i.e. **Server Address**).

3. Click **Save Changes.**

   If the “Apply Now” box doesn’t appear once saved, move to the administration section and use the “Reboot” button.



Upgrading the IP1500/2500 Firmware

The speakerphone firmware file can be changed by:

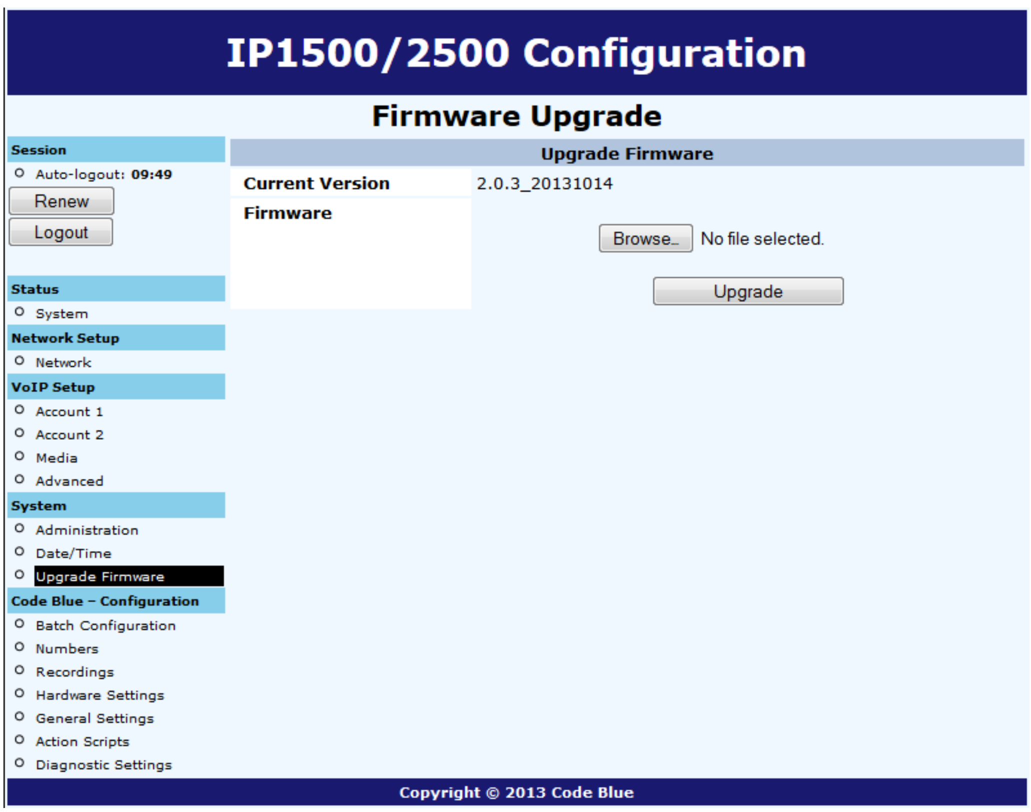
1. Select **Upgrade Firmware** under **System** (see far left hand column).

2. Click **Browse** (or **Select File**) and select the appropriate firmware file.

3. Click the **Upgrade** button.

4. The speakerphone will update, automatically back up the new firmware and reboot. Once this is complete, your new firmware will be in use and should be displayed next to **Current Version.**

Note: Firmware version is also reported in the **Administration** section.



6.5 Configuring System Options and Scripts

The speakerphone has advanced configuration settings, which allow for complete control of the hardware and how the system performs. A memory capacity of 1 MB provides for multiple phone number and recorded message capabilities. Incoming call routing, SNMP and advanced diagnostics enhanced with advanced scripting capabilities provide for flexible configurations.

Batch Configuration

The speakerphone can be configured from a TFTP server, e.g. UPD.

1. Click on **Batch Configuration** under **Code Blue** (see far left-hand column).

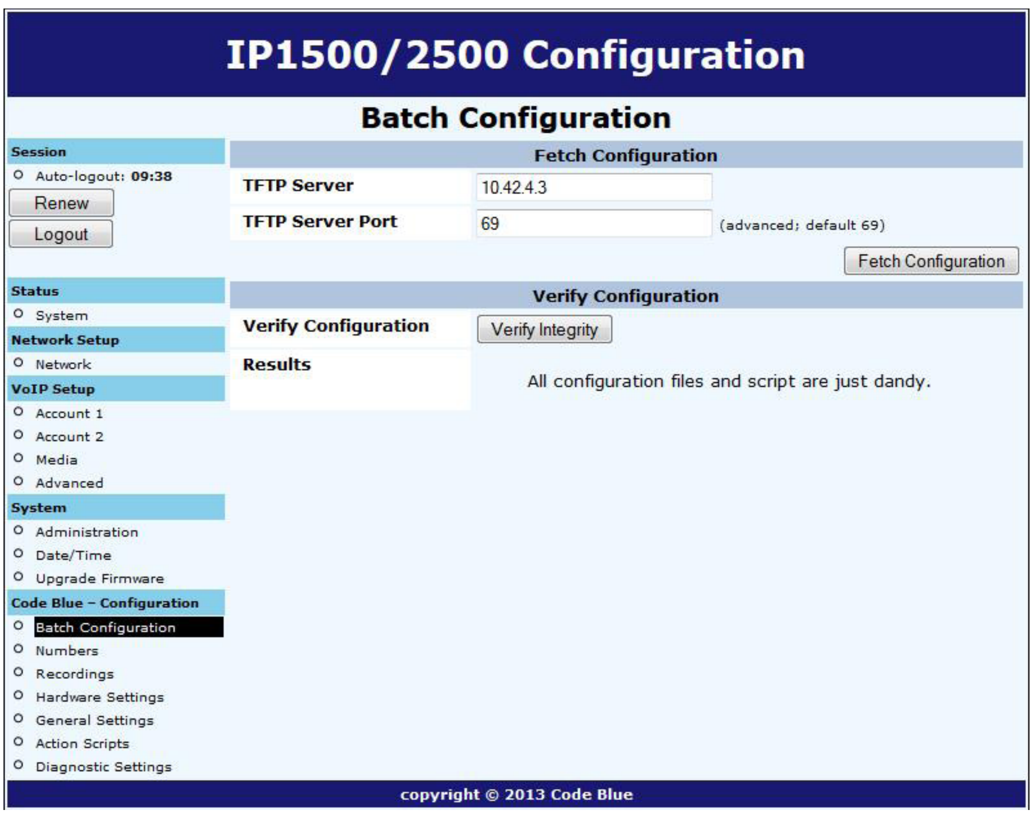
2. Enter the **TFTP Server** IP address and **TFTP Server Port.**

3. Click on **Fetch Configuration** to pull the configuration files from your TFTP server.

4. Click on **Verify Integrity** to validate the configuration files are suitable for use.

   If you are not offered the change to "Apply Now", move to the Administration dialog and manually click on the "Reboot" button.

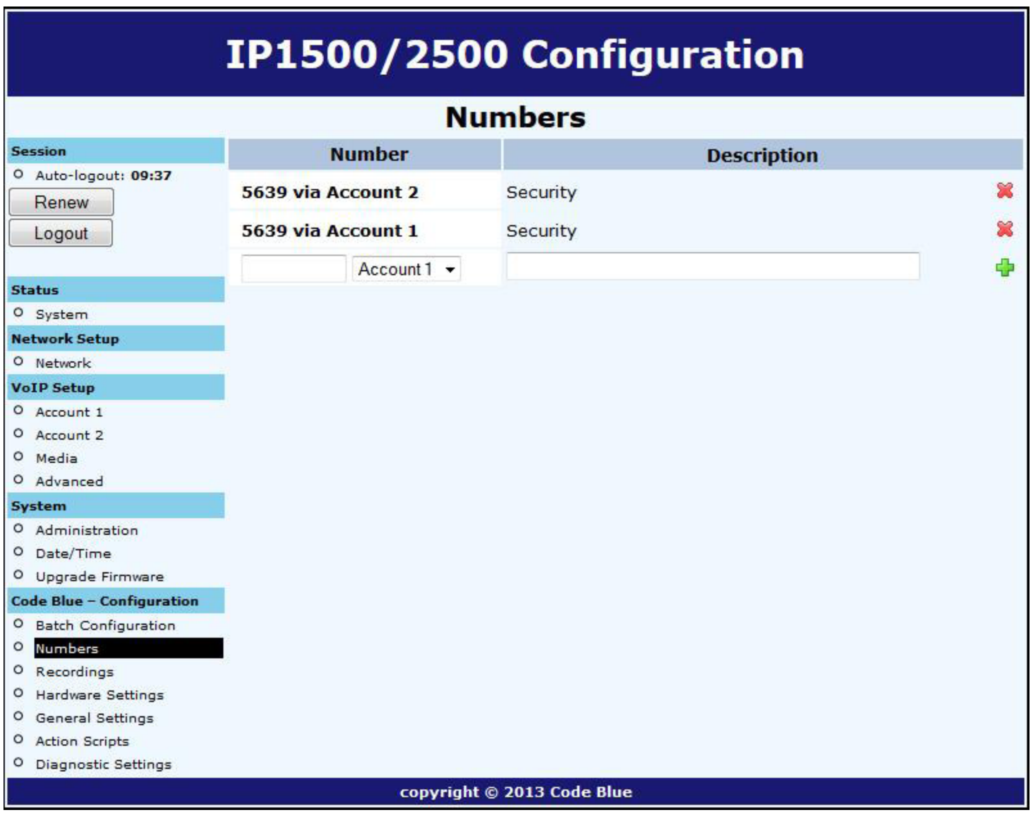
This functionality can be used in lieu of UPD's program functionality to have the speakerphone pull its configuration instead of having it pushed from UPD.



Entering Phone Numbers

The speakerphone number configuration is made by:

1. Clicking **Numbers** under **Code Blue** (see far left-hand column).
2. Enter the extension (i.e. SIP account, user extension) number. Choose which account this extension number will be related to. Enter a description for this extension. See account reference on page 15.
3. Select the **green plus sign** to add the number.
4. To delete a number, simply click the **red X.**
5. Select the **green check mark** when prompted **Are you sure?**



Recordings Administration

The speakerphone recording configuration is made by:

1. Selecting **Recordings** under **Code Blue** (see far left-hand column).

2. Click on **Select recording file** and choose the file you wish to upload to the speakerphone. Click **Open.**

3. Enter the Description within the Description Field.

4. Click on the **green plus sign** to add the recording and wait for it to finish.

   During the upload process the screen will display **Uploading file…**

   At this point do not refresh the page or click away from the page or the file will not be uploaded. Once the file upload is complete you will see **Download Recording** and a new line for uploading additional recordings.

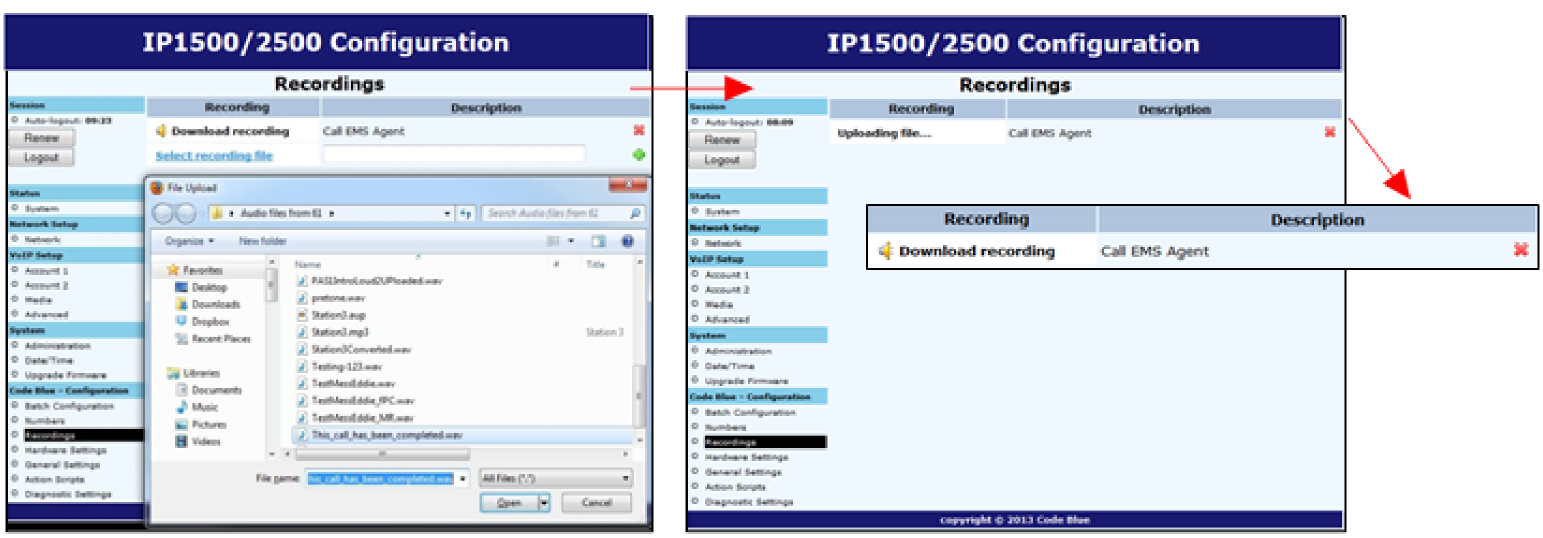
5. To delete a number, simply click the **red X.**

6. Select the **green check mark** when prompted **Are you sure.**

The speakerphone supports the following formats and all files must contain mono (single channel) data.

- File containing raw PCM uLaw data (extension .ulaw)
- Wave file containing mono 8 KHz or 16 KHz Linear PCM data (extension .wav)

Note: Audio files will consume memory space within the 1 MB shared memory allocation.



Hardware Settings

The speakerphone hardware settings are configured by:

1. Selecting **Hardware Settings** under **Code Blue** (see far left-hand column).

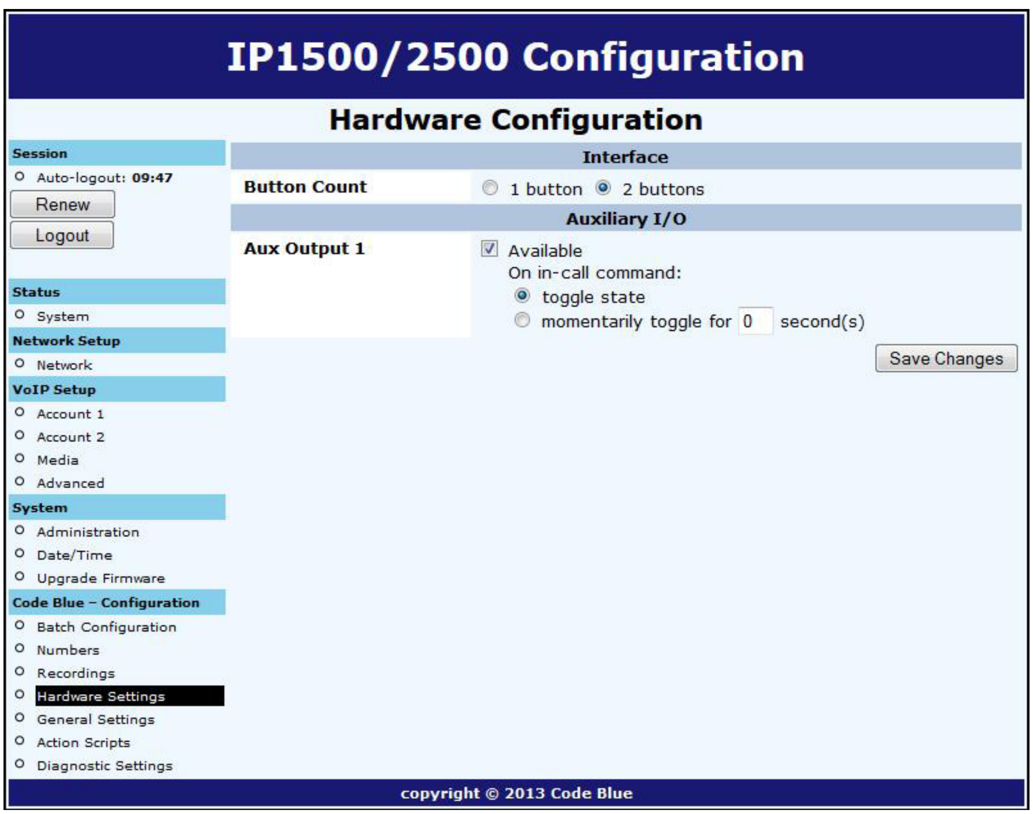
2. Select the appropriate **Button Count, Keypad Available** settings under the **Interface** section.

3. Checking Aux Output 1 will enable the aux output relay. By default, the port is set to enable (Toggle State) when used in an Action Script.

   When **Momentary toggle** choice has been selected, the called party now has the ability to activate the aux output remotely for the time period chosen via DTMF tones from their phones keypad.

   Note: **Momentary toggle** is intended for remote control use by the called party. It's important to understand that scripted use of the aux output not be used on any aux output port that has been selected to act in the momentary (remote control aspect) toggle function. Also it is not recommended to use the **General Settings > Incoming Calls > Aux Output 1 Enable on Incoming Call** check box.

4. With selections made, click **Save Changes.**



General Settings

The IP1500/2500 speakerphone general configuration can be accessed by:

1. Clicking on **General Settings** under **Code Blue** (see far left-hand column).
2. In this section you can select how many rings the speakerphone will wait before answering an incoming call.
3. Click the down arrow next to Answer In to change settings.
4. The **Aux Output 1** check box, when checked, will enable the **Aux Output 1** on incoming call and is disabled when incoming call is terminated.

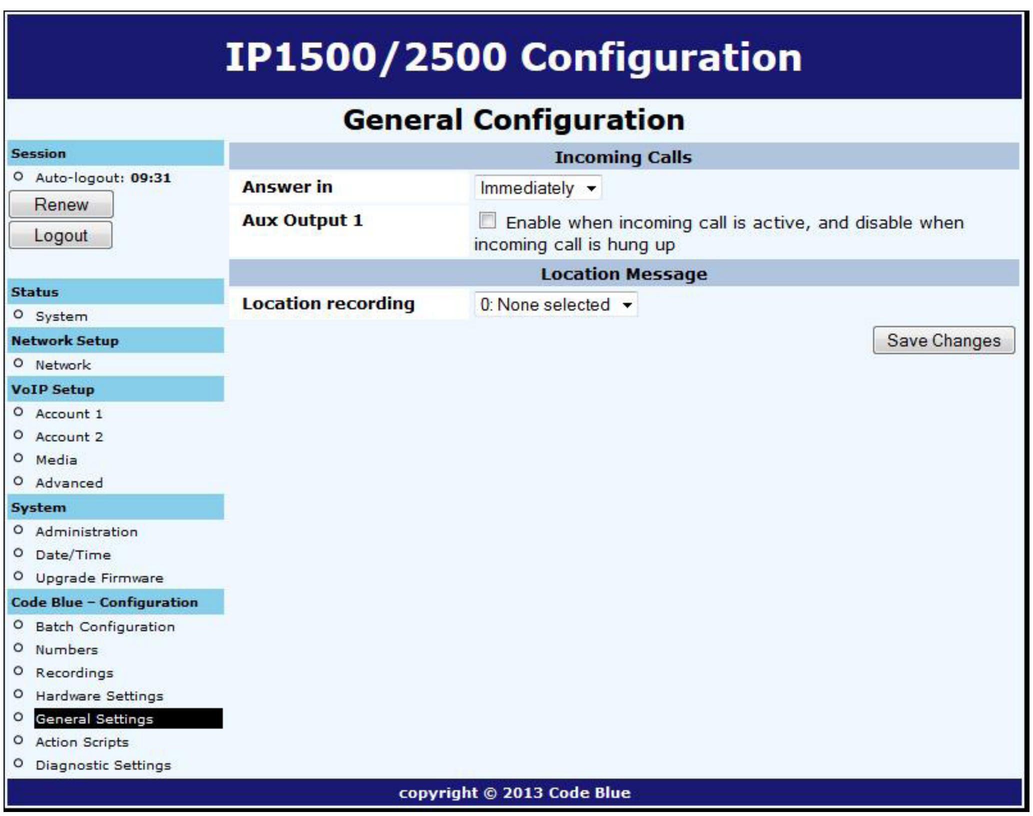
   This feature was not intended to be used with **Aux Outputs** configured with the **momentarily toggle (Hardware Settings Dialog)** choice.

The speakerphone can also be configured with a standard location message.

1. Click on the **down arrow** next to **Location Recording** to select this recording as the default Location Message.

   » The location message must be uploaded before this choice can be made. See **Recording's dialog.**

2. Once you have configured the options on this page, click **Save Changes.**



Action Script Configuration

Action Scripts are based on Hardware Settings made earlier in the setup process. For example, if your speakerphone has two physical buttons and only one button was selected in **Hardware Settings "Interface" "Button Count"** some scripts choices will be missing.

Scripting Requirements
The Action Script in the speakerphone can be very extensive, yet only if all the correct features are enabled. Understanding all the abilities of the phone is required, only then can the user configure the speakerphone for maximum functionality.

Numbers
Load phone numbers for all of your planned calls from this speakerphone.

Recordings
Record all message and upload them to this speakerphone.

Hardware Settings
Ensure the speakerphone features are represented in the Hardware Settings portion of the GUI.

Diagnostic Settings
When using remote monitoring services, for example SNMP Server service or Code Blue's ToolVox Server w/UPD application, the speakerphone will send SNMP traps or use the "Action Scripts" to generate calls to a monitoring service and play pre-recorded messages as a notification an issue has been detected.

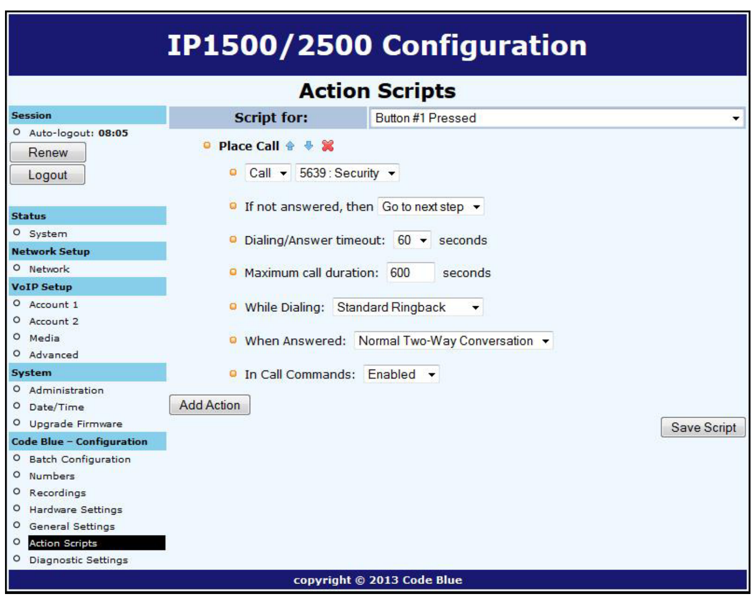
SCRIPTING BASIC CALL

The speakerphone has GUI interface for building scripts. Scripting can consist of a single action or combination of actions related to a button press or Auxiliary Output Trigger alone.

- Click on **Action Scripts** under **Code Blue** (see far left-hand column) to program the action scripts you wish the unit to perform during button activation or diagnostic condition.

- To program, select a **Button** or **Diagnostic** condition from the option list by clicking on the down arrow across from Script for: For this example select **Button #1 Pressed.**

- Click on **Add Action.**



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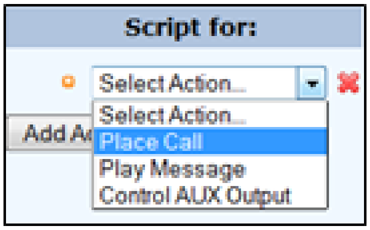
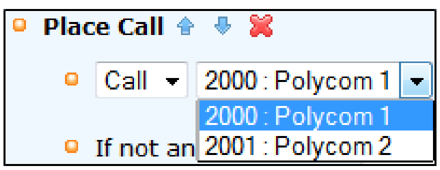
SCRIPTING BASIC CALL *(continued)*

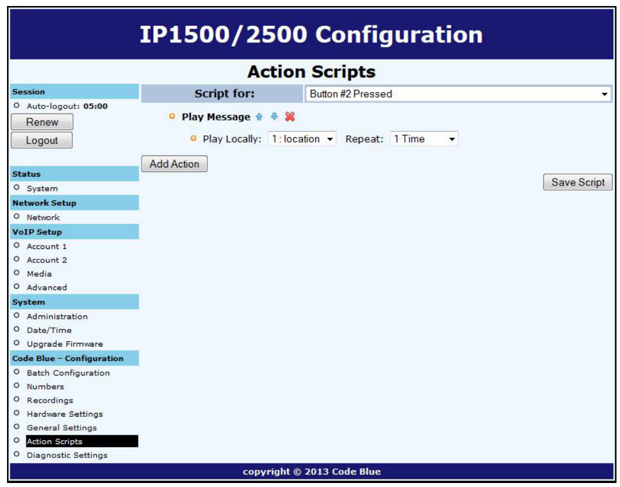
- From the **Select Action** drop down, choose **Place Call.**



- By default, the first number placed in memory will be present here. If another number is desired, use the drop-down arrow to locate and select another phone number.

- Click on the **Save Script** button. This completes the basic programming needed to place a call.

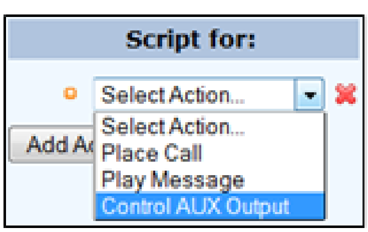




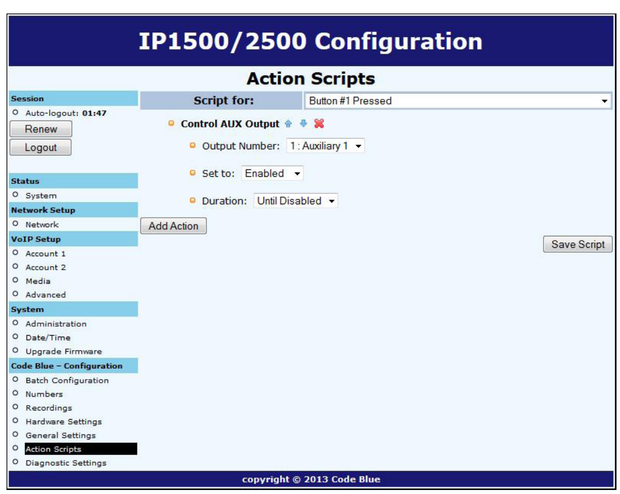
Other Basic Script Choices
Scripting in the speakerphone allows for non-phone call scripting to be programmed to meet unique needs of the customer.

1. For example, use **"Button #1 Pressed"** as seen in the example **"Basic Call".**

2. Instead of choosing **"Place Call,"** select **"Control Aux Output".**



3. By default, the Auxiliary 1 is presented (but note only those Aux Outputs selected in Hardware Settings will be available in this list).



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SCRIPTING BASIC CALL *(continued)*

4. Next choice is to **Enable** this Aux Output and/or set the **Duration** for this **Aux Output Action.** In this example, request a 10-second duration upon the touch of button 1.

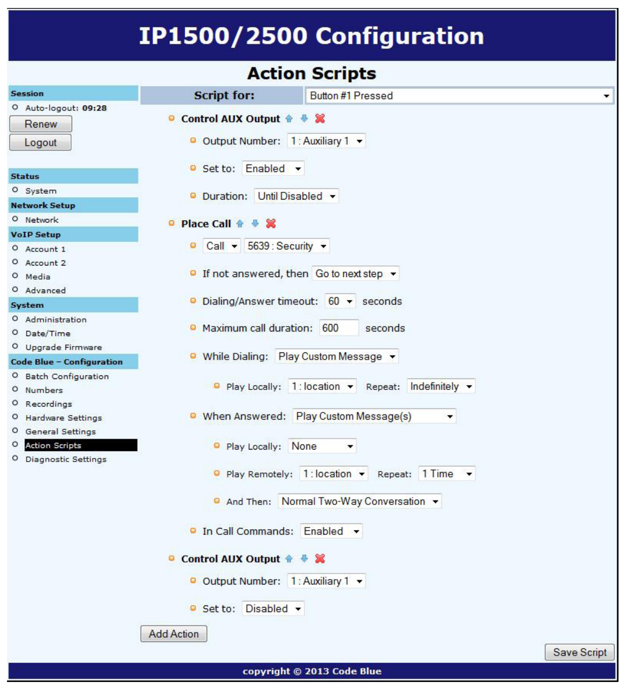
5. Next click on **Save Script.** This script is now ready to be tested. Touch Button 1 to test.

Combining Multiple Actions in One "Script -- Advance Programming"
The following example would be the most common configuration deployed.

1. Using **Action Scripts > Script for:** "Button #1 Pressed".
   Add the following as seen in the example:

   A. Control Aux Output – Enable
   B. Place Call – with messages for Calling party and Called Party
   C. Control Aux Output – Disable

2. The Script should look like this:
   Click **Save Script** when finished.



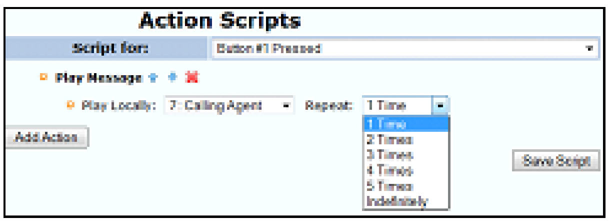
ACTION SCRIPT PARAMETERS

Within the Scripts are many settings controlling the next step in the process of the Action Script: Duration of the process, Enable/Disable features, or even a reactivation of an Aux Output with a timed limitation. The following will provide detailed explanations into these Script controls.

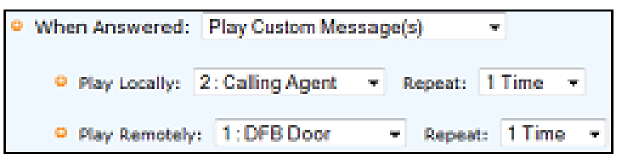
Note: Scripts, Phone Numbers and Recordings all share a 1Mb memory cap.

Playing a Message
Messages can be set to play any time upon the activation of a Script or during a call.

Plus, they can be set to repeat as shown here:



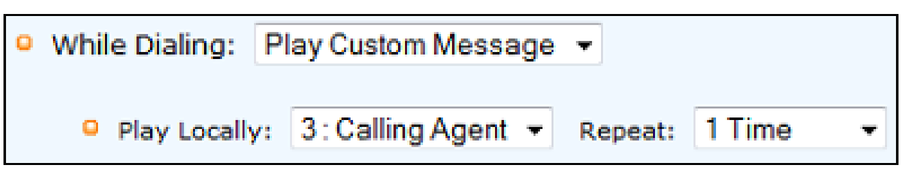
Place Call
Placing a Call is where the administrator sets up which numbers will be attempted and the order. The administrator could choose multiple numbers stored in "Numbers" or the same number can be repeated many times. "If not answered, then" Call. Select additional numbers to be dialed.

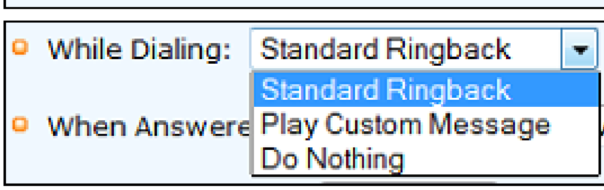


Dialing/Answer Timeout: The default time is 60 seconds and can be stepped down to as little as five seconds, before the call attempt times out.

Maximum Call Duration: The default time is 600 seconds (10 minutes). Duration range 0001 to 9999 seconds (1 second up to 166.65 minutes). Thirty seconds before the timer exhausts an audible tone will play to notify both parties the call is about to terminate, unless the timer is disabled through a During call Command (DTMF tone 3).

While Dialing: Standard Ringback is the default setting. Other choices: A message can be set to play to the person at the IP1500/2500 and/or Do Nothing, until the call is connected.





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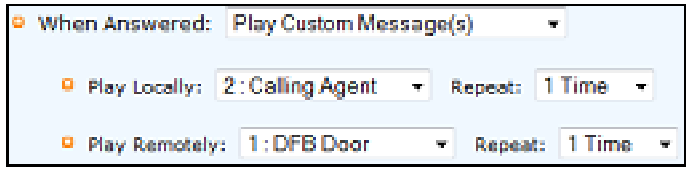
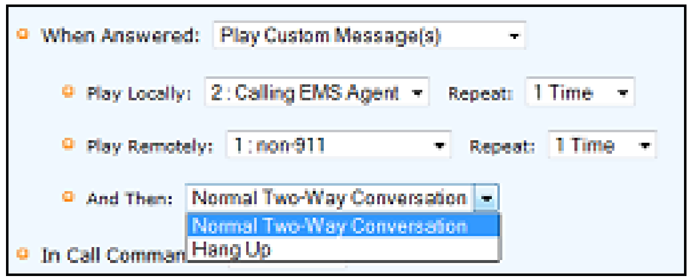
ACTION SCRIPT PARAMETERS *(continued)*





Place Call (continued)
When Answered: The default setting is **Normal Two-Way Conversation,** the option is to **Play Custom Messages**. A message can be set to play **Locally** (at the speakerphone) and/or **Remotely** (to the **Called Party**).

Choosing this option will add another option to the Place call sequence, **And Then.**
The **And Then** choice allows the call to continue through to normal two-way conversation mode or **Hang Up** and reset the speakerphone.

Note: In this feature, it is prohibited to use the same exact message in both local and remotely selection.

In Call Commands: The default is Enabled. All Remote Control DTMF tone commands are available for use by the called party. The alternate choice is Disabled, effectively locking out all DTMF tone commands from the Called Parties control.

Control AUX Output
- Aux Outputs can be activated and deactivated throughout a Script.
- Aux Outputs can also be set to activate on incoming answered calls.
- It is strongly advised that when this feature is used no other configurations are enabled for an Aux Output with Momentary Toggle selected in **Hardware Settings.**

Sample Application using Dual Accounts on the IP1500/2500 Phone

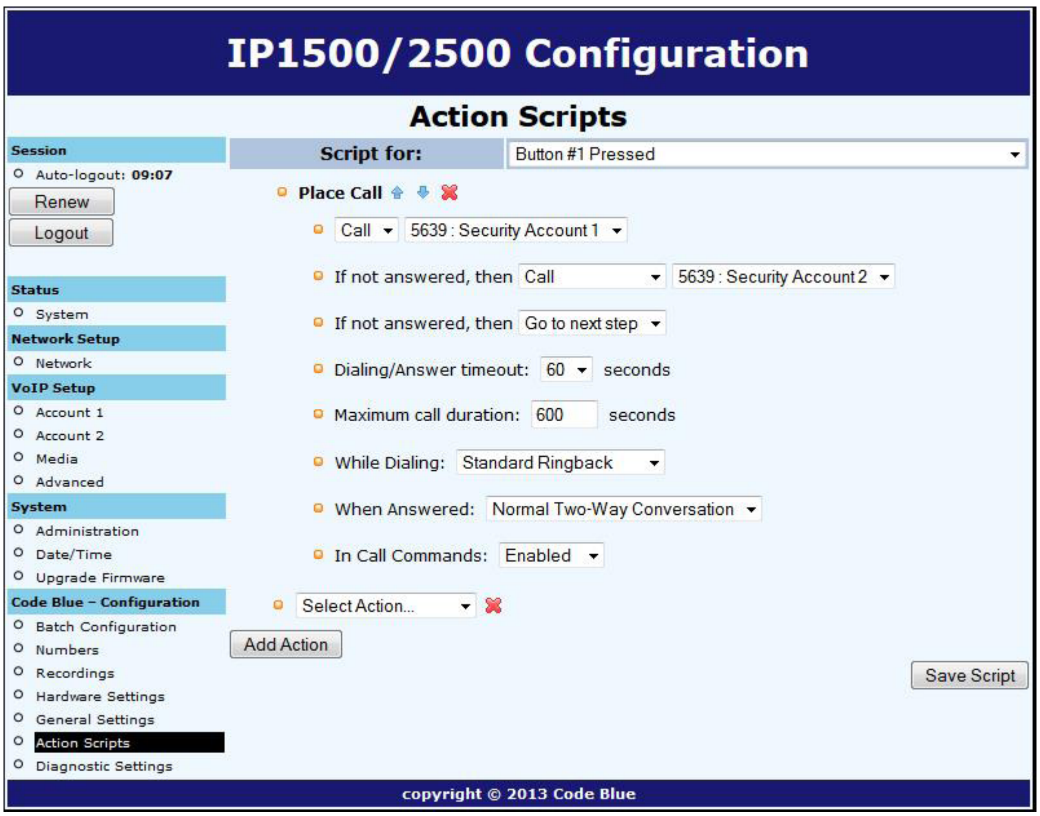
If using both accounts on a speakerphone, you must then set up 2 numbers (one **"via Account 1"** and the other **"via Account 2"**), and an action script with a single dial step with **"call first number"** and **"if not answered then call second number".**

Use outcomes dependent on the network:

1. If server 1 is considered registered and responds, the call goes through to server 1 immediately.

2. If server 1 is considered registered and unresponsive, it will be tried for the time listed in **Dialing/Answer timeout,** but no more than 30 seconds; then server 2 will be tried.

3. If server 1 is not considered registered, server 1 will be skipped and server 2 will be tried immediately.



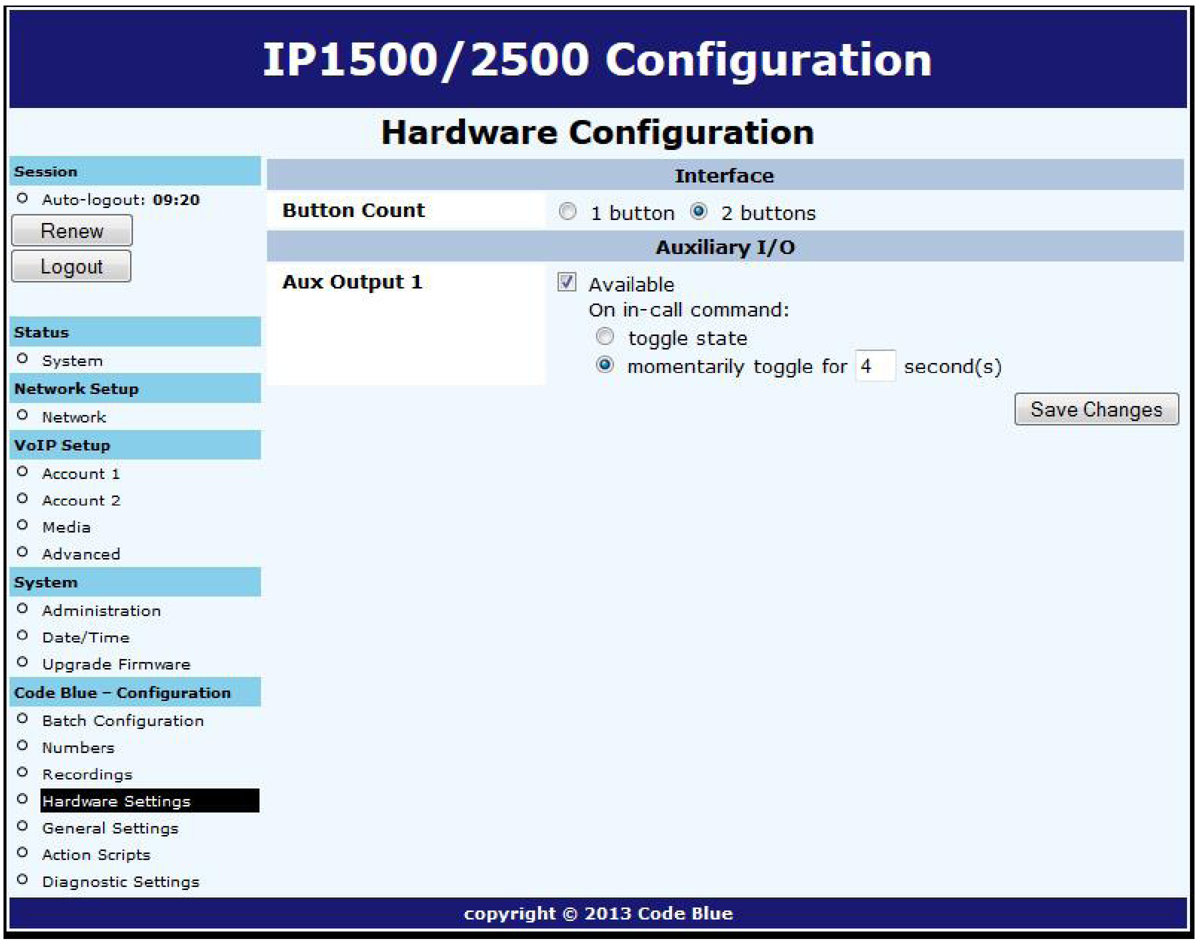
Auxiliary Output Expanded Functionality & Use Case

The speakerphone v2 Aux Output abilities has been expanded for unique use cases: Security Personal Access Control.

Example:

Gate or Door Control
Either output can be configured to activate upon the called parties use of the DTMF keys 4 or 5 on His or Her phone, for a predetermined time period needed by the Gate Mechanism (example - 4 seconds).

Setting up Auxiliary Output 1 to Momentarily Toggle for 4 seconds.



Aux Output Momentary Toggle** is best used for remote control operations and should not be combined with **Scripted Timed Aux Output** timers or **Incoming Calls > Aux Output > Enable** when an **Incoming Call is active.

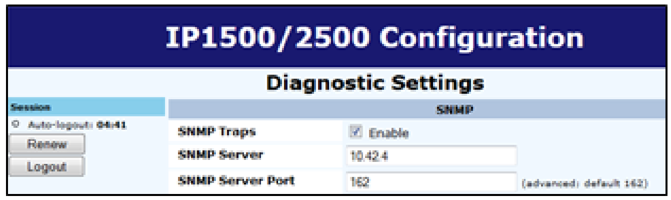
Configuring Diagnostics

Diagnostic Settings
The speakerphone diagnostic settings are configured by:

- Selecting **Diagnostic Settings** in the **Code Blue Configuration.**

- Click the **Enable** check box.

- Input the **SNMP Server** IP address and **SNMP Server Port** number to monitor the speakerphone with an **SNMP** management software or with Code Blue's ToolVox Gateway, with Unit Programming & Diagnostic (UPD) Software.



- PoE Power Failure: PoE power is the sole power source and if an interruption in service is experienced, no Trap will be sent due to loss of PoE energy. The PoE switch should alert you to PoE switch state.

Others – (Tests)
Microphone testing is disabled by default, and enabling will show a number of reoccurring test routines. The microphone is supported by the speaker's ability to generate tones at the schedule intervals.

- The test consists of beeps from the speaker, which will be received by the microphone.

  The maximum number of beeps: 10 beeps
  Once the microphone detects the beeps, the test is complete until the next scheduled test is present.

  The beep tone volume choices are soft, loud, or soft to loud.
  Beep tone volume setting should be set to anticipate ambient noise level at the time of the test.

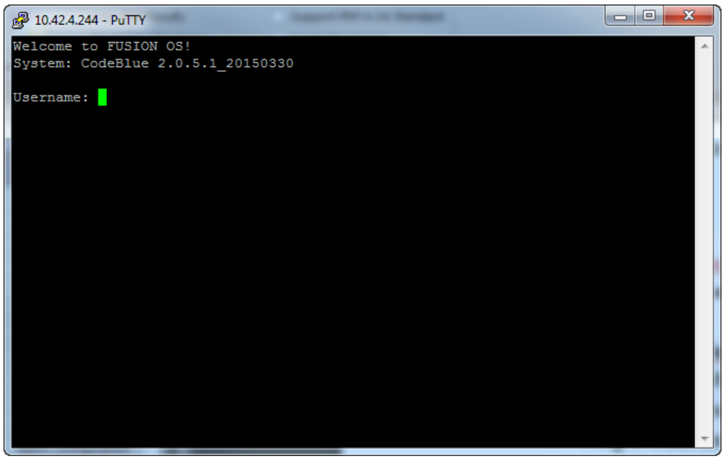
- The test schedule choices are:

  Every 15 minutes
  Hourly
  Daily
  Weekly

Testing on demand: When microphone speaker testing is enabled, the administrator may select to **Run Test** while logged into the speakerphone. The results of the test will only be present in a failed SNMP trap, which would appear in the SNMP server logs or UPD Diagnostic Reports logs. The MIB value is **CODEBLUE-MIB::micSpeakerFailure.**

7 CLI (Command Line Interface)

The speakerphone has extensive commands that can be used by telnetting into the device.
You can use windows telnet or download a common free telnet client, “putty”.
Telnet to the IP Address of the speakerphone: use port 23 if unsure.



Login is the same as through the Web GUI.

admin
admin

You can type “help” to see a list of available commands.
The most commonly used:

Format c: codeblue – Using this command, you format the phone and return it to factory default. This command must then be followed up with a reboot.

Reboot – Make the phone reboot.

Ping IP Address or Domain Name – Ping the IP PBX to see if the phone can reach its registrar.

Button 1 – Select button 1-4 and initiate a button push remotely. This is very handy for remote testing. Button 1 is the red button. Button 2 is the black button if equipped.

8 In-Call Commands

The speakerphone provides enhanced functionality through the utilization of In Call Commands. These commands are DTMF or phone keypad entries made by the Called party. Below is a list and explanation of each command.

In-Call Command	Function	Description
1	Play Location Message	Plays the Location Recordings selected in General Settings
3	Deactivate Call Timer	Deactivates the Maximum call duration timer setting in the operational script currently running
4	Activate/Deactivate Auxiliary 1	Toggle Auxiliary 1 state; activate or deactivate
6	Mic Volume Up	Increase the microphone gain; used to increase the Called party volume
7	Mic Volume Down	Decrease the microphone gain; used to decrease the Called party volume
8	Speaker Volume Up	Increase the speaker volume; used to increase the Calling party volume
9	Speaker Volume Down	Decrease the speaker volume; used to decrease the Calling party volume

Note: Some VoIP codecs do not fully support DTMF Tone signally and may not function as intended.

9 Factory Reset

The system can be reset via two different methods.

1st Method:

The speakerphone can be reset by following the steps below:
Use the 7 pin reset plug (sent with your order) in order to perform a full reset.

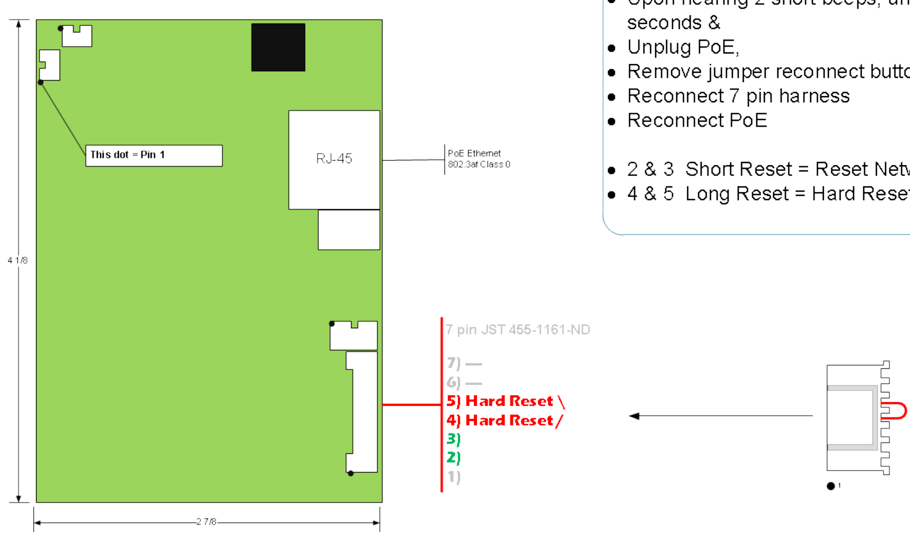
4 & 5 Long Reset = Hard Reset, sets everything back to default
2 & 3 Short Reset = Resets network configuration

- Unplug the RJ45 from the PoE switch port
- Unplug button and disconnect the LED harness
- Short the appropriate 2 pins together for short or long reset (see pic below)
- Plug the RJ 45 back into the PoE switch port
- Upon hearing two short beeps, unit has been reset
- Wait 10 seconds for phone to reboot
- Unplug power source
- Remove the jumper and reconnect the buttons
- Reconnect 7 pin harness
- Reconnect PoE

The Reset is now complete.

- Unplug PoE
- Unplug button / Led harness and replace with shorting pin connector
- Reconnect PoE
- Upon hearing 2 short beeps, unit has been reset, wait 10 seconds &
- Unplug PoE,
- Remove jumper reconnect buttons
- Reconnect 7 pin harness
- Reconnect PoE

- 2 & 3 Short Reset = Reset Network Config
- 4 & 5 Long Reset = Hard Reset, clears everything to default



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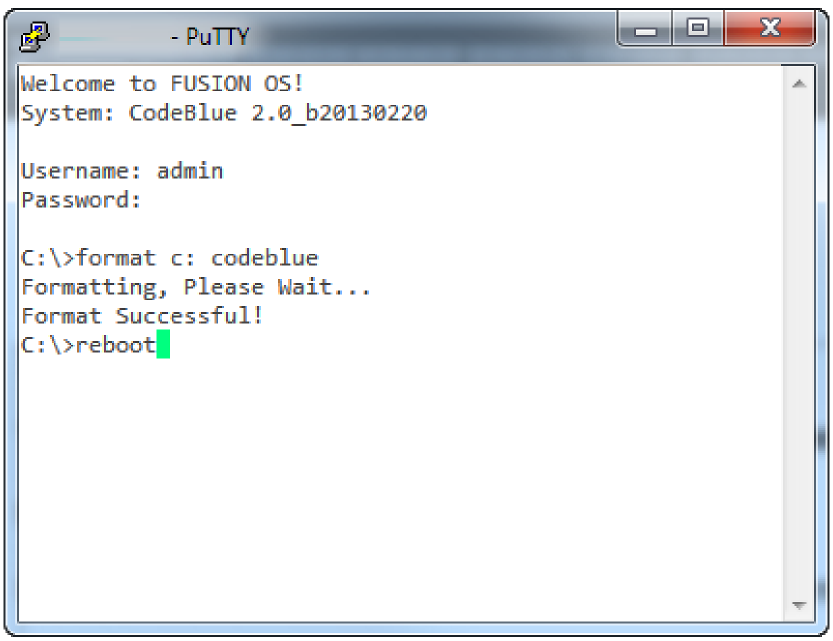
Factory Reset *(continued)*

2nd Method:

If you have telnet access to the unit, you can default the unit through the command line.

- Using Windows Telnet
  Open <IP Address> <port>
- Enter Username: **admin** and Password: **admin**
- At the prompt, type **format c: codeblue**
- After successfully formatting the phone, type **reboot**



10 Compatibility

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

You must verify that the IP PBX you are registering the speakerphone to can handle third party SIP devices whether through licensing and/or Hardware add-ons.

Some examples of mainstream IP PBXs the speakerphone has registered to as a third party SIP device are:

Asterisk
Cisco Call Manager

and many others…

11 Configuring for Cisco Unified Communications Manager 9

PREPARATION

1. Record the MAC address and determine the current IP address for each IP1500/2500/5000 device you wish to use with CUCM.

2. Determine which partition you will put the IP1500/2500/5000 directory numbers into.

3. Obtain one directory number for each IP1500/2500/5000 device.

    a. If you are going to use the IP1500/2500/5000's dual account configuration to regis ter to redundant CUCM servers, obtain a second directory number for each IP1500/2500/5000 device.

4. Determine which calling search space you will assign to the IP1500/2500/5000.

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 IP1500-2500-5000 Profile.

5. Click Save.

Configure End Users

For each IP5000 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 hexadecimal version of the MAC address; e.g. 00:50:C2:17:7B:E8 would become 0050c2177be8.

    a. Use of the MAC address as user ID is only a recommendation. If local configuration permits, you can use any other form of user ID; just be sure to record which user ID goes with which phone and which of the phone's accounts.

4. Fill in the Last name field with a description of the station.

1. 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 IP1500/2500/5000.

2. Click Save.

Configuring End Users for Secondary Accounts

If you are going to use the IP1500/2500/5000'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.

Since all IP1500/2500/5000 units' MAC addresses start with 0, you can create a locally-administered address that is unlikely to conflict with other locally-administered addresses simply by setting the U/L bit simply means changing the second 0 to a 2, e.g. 0250c2177be8.

Configure Phones and Directory Numbers

For each IP5000 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 IP1500/2500 is to use.

8. For Device Security Profile, select Code Blue IP1500-2500-5000 Profile.

9. For SIP Profile, select Standard SIP Profile.

10. For Digest User, select the end user matching the MAC address of the phone, or the alter nate user ID you created when you were configuring the end user.

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.

1. Under Line 1, for Display (Internal Caller ID), enter a descriptive name for Caller ID pur poses.

2. If you wish to return a busy signal for silent monitoring if the IP1500/2500/5000 is in use, disable Call
Waiting: under Multiple Call/Call Waiting Settings, For both Maximum Number of Calls and Busy Trigger, enter 1.

3. Click Save.

Configuring Phones and Directory Numbers for Secondary Accounts

If you are going to use the IP1500/2500/5000'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.

IP5000 CONFIGURATION

Refer to the IP1500/2500/5000 Administration AND User Guide located on our website

Clear Existing Configuration

If necessary, clear the IP1500/2500/5000's existing configuration. This will reset it to DHCP, so make sure you have the capability to find the device's IP address again if you do this. For each unit:

1. Open a Telnet client and connect to the IP1500/2500/5000.

2. Log in using the username admin and the default password admin.

3. Type format c: codeblue and press Enter.

4. Type reboot and press Enter.

Configure Account(s)

1. Log in to the IP1500/2500/5000 via its web interface. The default username and password are admin and admin.

2. Select Account 1.

3. For VoIP Protocol, select SIP & RTP.

4. Under SIP Configuration, for Username/Number, enter the directory number you assigned earlier.

5. For Display Name, enter caller ID text.

6. For Domain, enter the hostname or IP address of the CUCM node you wish to register this account to.

1. Insure Keep-Alive is enabled.
2. Under Proxy Authentication, for Username, enter the username you assigned the CUCM end user, e.g. the hexadecimal representation of the MAC address or the local-use variant for a secondary account.
3. For Password, enter the password you entered into Digest Credentials under the CUCM end user.
4. Click Save.
5. Repeat steps 3-10 with Account 2 if you are using the second account.

Other Settings

Refer to the IP1500/2500/5000 Administration AND User Guide to complete the setup of the IP1500/2500/5000, including Numbers, General Settings, Hardware Settings, and Action Scripts. When finished, click Apply Now to restart the phone; 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 IP1500/2500/5000 with secondary account support, make sure that you create each failover number twice.

12 Avaya IP Office Integration Guide

Introduction

This Avaya IP Office Integration Guide provides general instructions for integration of the **IP1500/2500/5000 Series Phones** with an IP Office installation. Read this instruction set completely before starting any installation. For detailed **IP1500/2500/5000** setup instructions, please consult the **IP1500/2500/5000 Guides**.

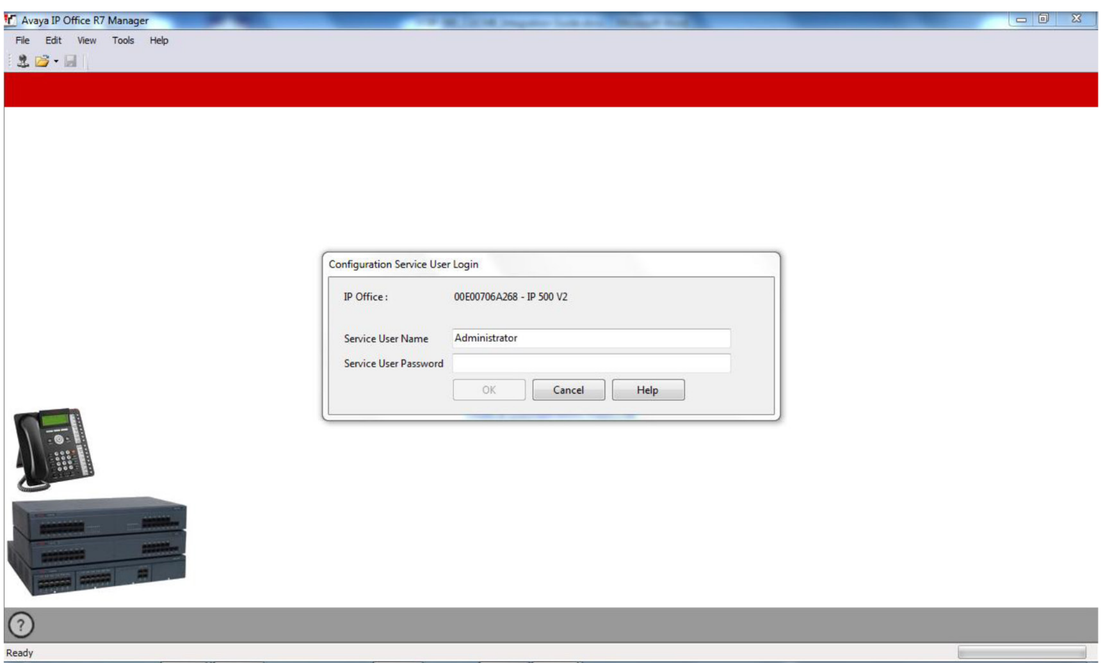
Prerequisites

- Avaya IP Office Manager Version 9 pre-installed
- SIP Device Licensing for 3rd Party IP Endpoints
- Network access to the IP Office Manager, **IP1500/2500/5000 Series Phones** and all network services (SIP, HTTP, FTP, RTP/SRTP)

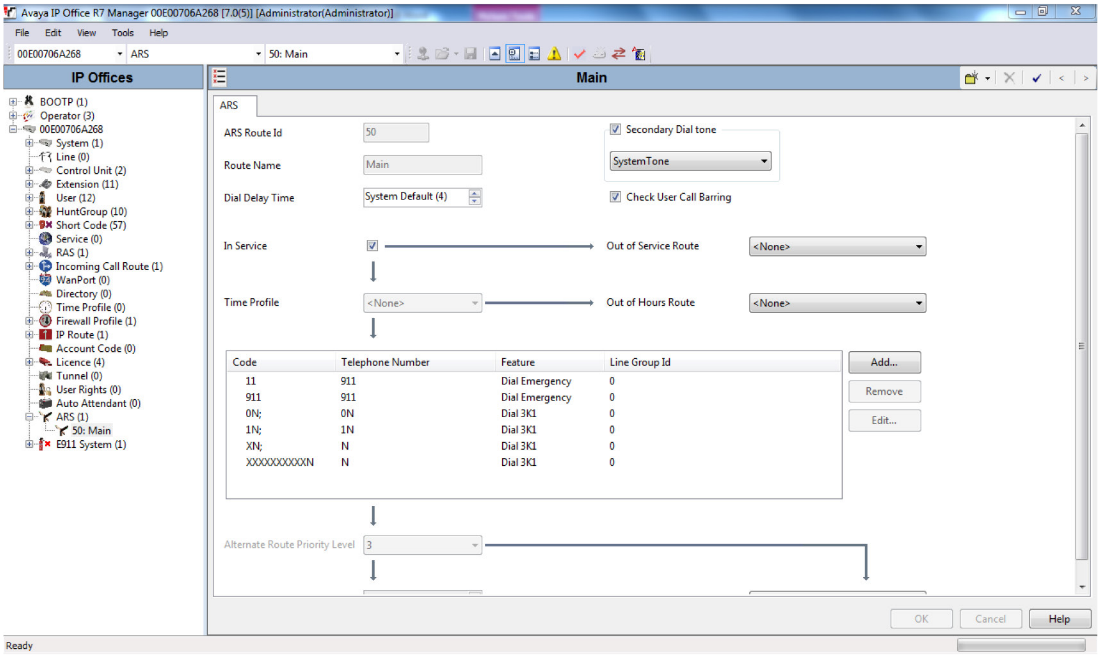
IP Office Manager Basic Configuration

Basic instructions for integrating **IP1500/2500/5000 Series Phones** with an Avaya IP Office R7 Manager are included. Advanced setup of IP Office Manager features is outside the scope of this document.

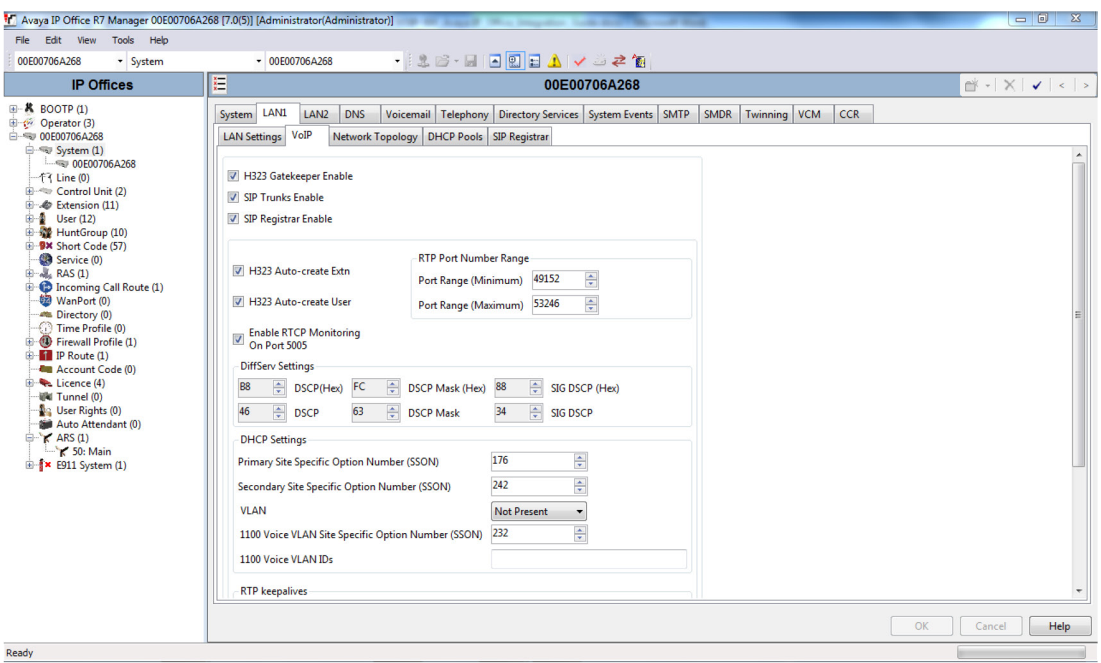
1. Using IP Office R7 Manager, connect to the IP Office Control Unit.



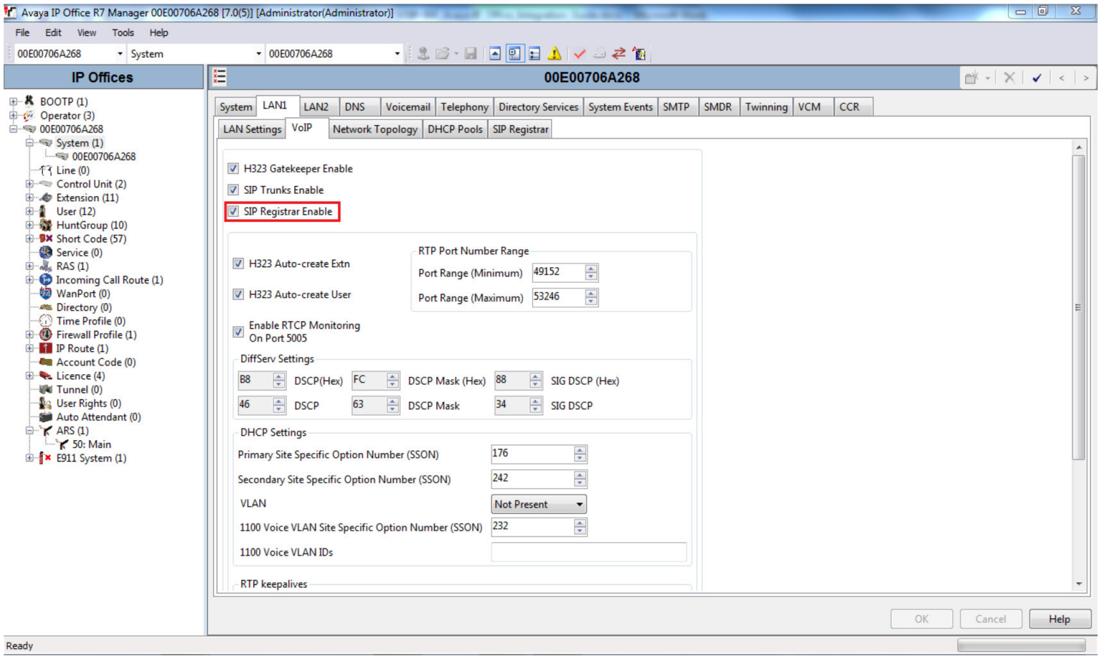
2. Log in to Avaya IP Office Manager:



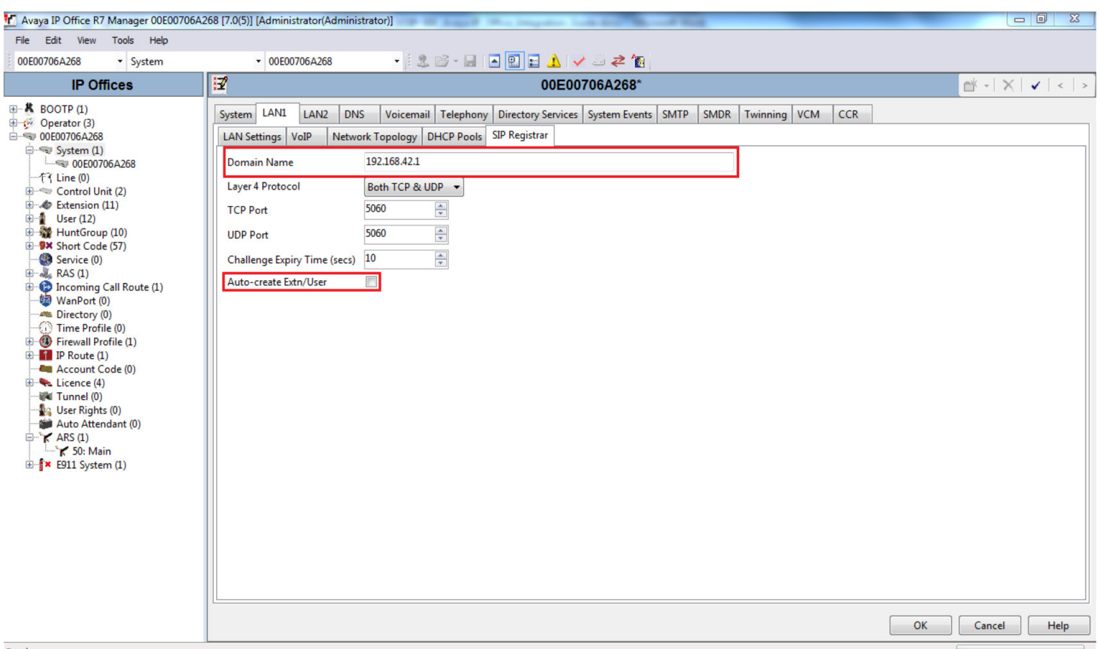
3. SIP Extension Support is required for **IP1500/2500/5000** integration. Select **System > LAN1 (or LAN2) > VoIP** in IP Office Manager:



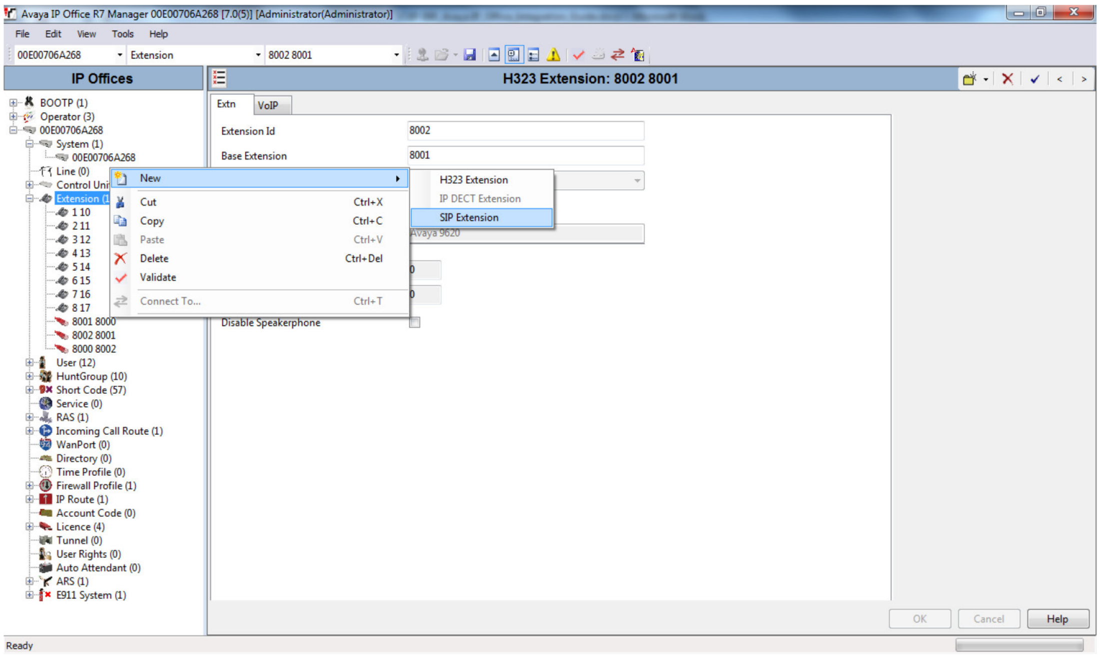
4. Check that **SIP Registrar Enable** is enabled.



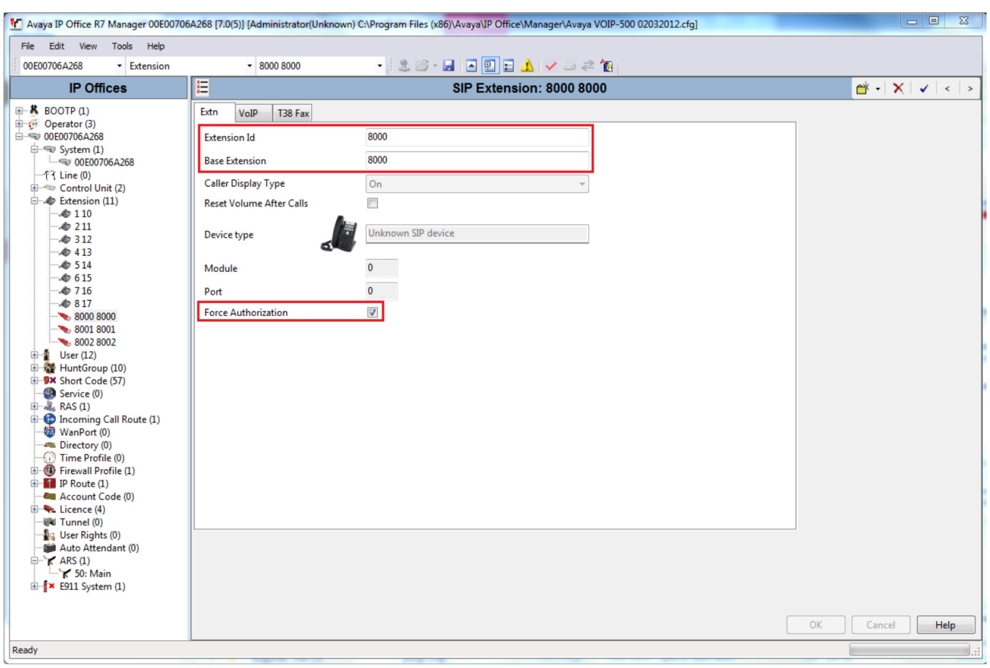
5. Select the **SIP Registrar** sub-tab.
6. In **Domain Name,** enter the Fully Qualified Domain Name (FQDN) or the IP ad dress associated with the correct LAN port on the IP Office Control Unit. Deselect **Auto-create Extn/User.** Click **OK.**



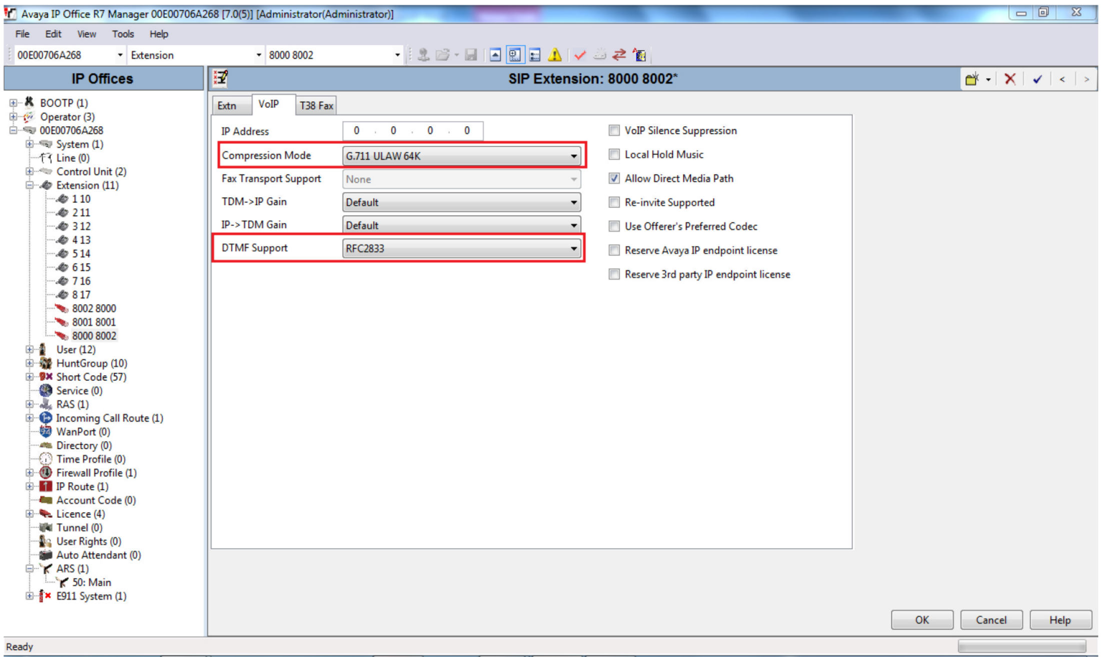
7. A SIP extension will need to be created for each **IP1500/2500/5000 Series Phone**. Right click on **Extension**, select **New** and then click on **SIP Extension**.



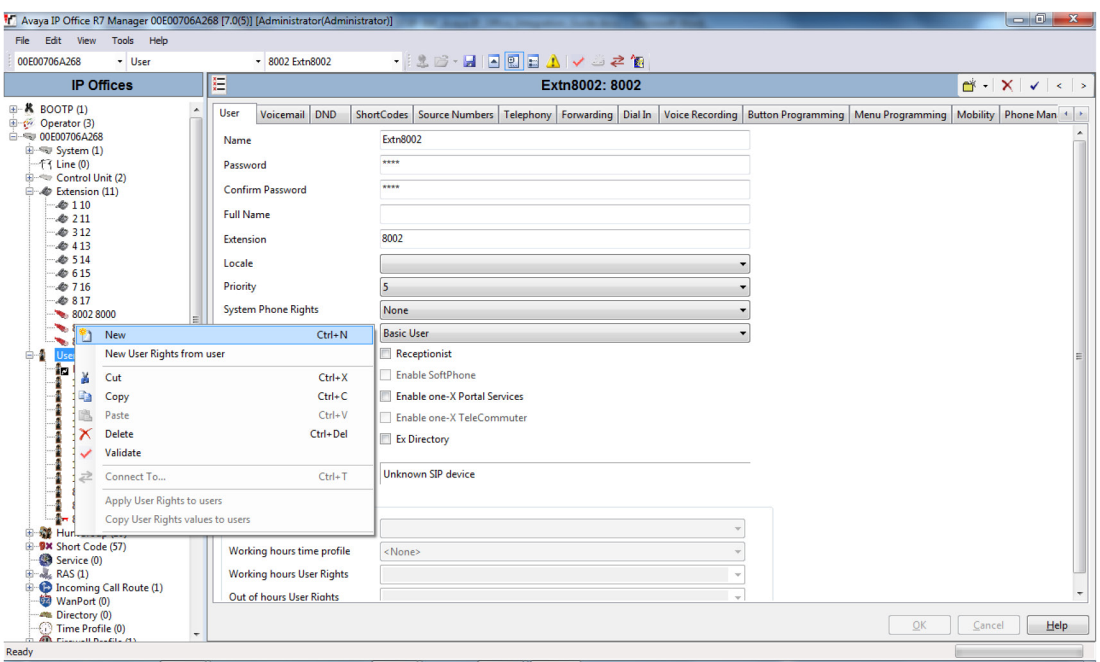
8. Enter the following fields to create a new extension:
   - **Extension ID:** A unique extension to identify the logical extension in IP Office. By default, IP extensions start at 8000.
   - **Base Extension:** This is the extension used to call the **IP1500/2500/5000 Series Phone**.
   - **Force Authorization:** Select to force authentication of the **IP1500/2500/5000 Series Phone**.



9. Select the **VoIP** tab and select the **Compression Mode**. The default of the **IP1500/2500/5000 Series Phone** is **G.711 U-LAW** and will work in most cases. More information on audio codecs can be found in the **IP1500/2500/5000 Series Phone Guides**. Set **DTMF Support to RFC2833.**

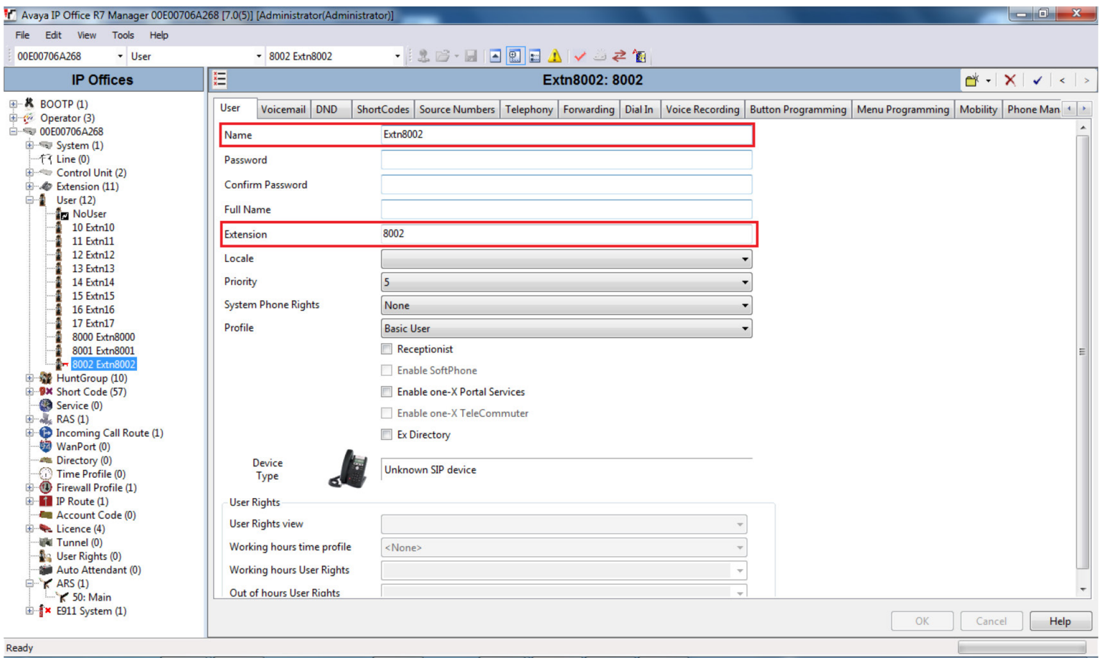


10. Each **IP1500/2500/5000 Series Phone** should have a unique User. Right click on **User** and select **New.**

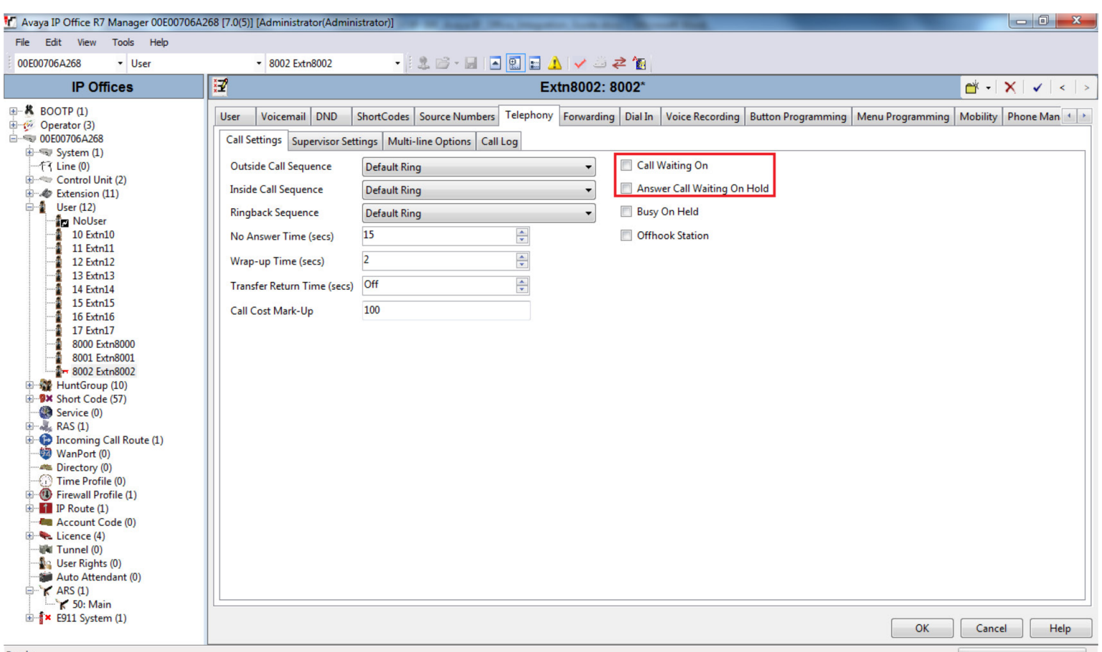


11. Enter the following fields to create a new user;

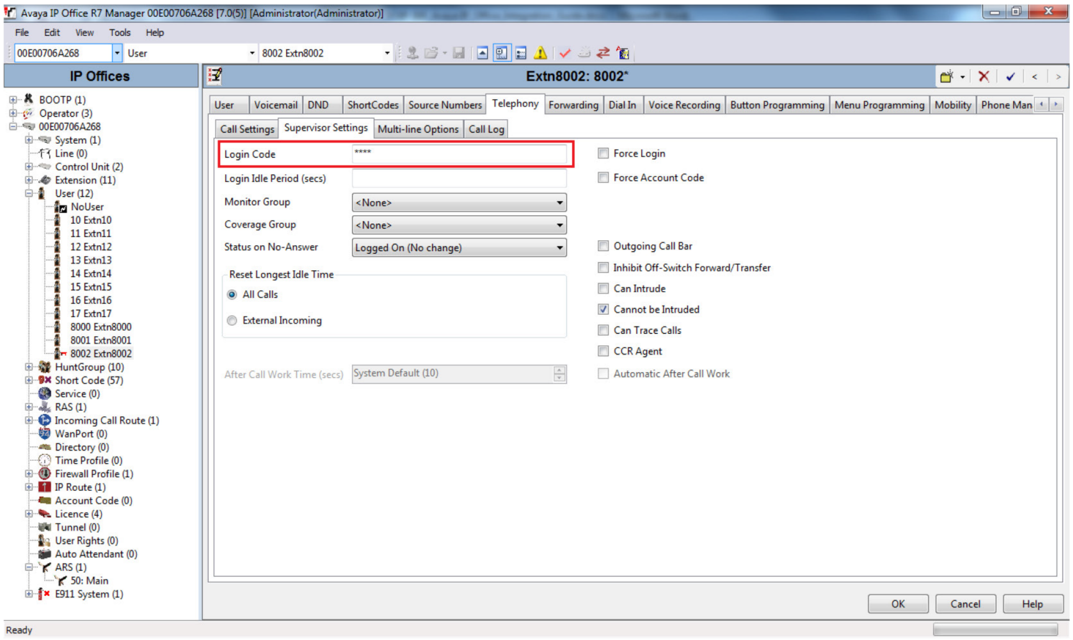
- **Name:** This will be displayed as the user's name in IP Office Manager, and is used as the username for SIP registration when configuring the **IP1500/2500/5000 Series Phone**.
- **Extension:** This should match the **Base Extension** configured for the SIP extension in Step 8. This is also used as the phone number when configuring the **IP1500/2500/5000 Series Phone**.



12. Select the **Telephony** tab and then the **Call Settings** sub-tab. Disable **Call Waiting On** and **Answer Call Waiting on Hold**. Call waiting is not supported on the **IP1500/2500/5000 Series Phone**.



13. Select the **Supervisor** sub-tab. In the **Login Code** field enter a password to be used by the **IP1500/2500/5000 Series Phone** for authentication. Avaya IP Office will only accept numbers in this field.



14. If adding multiple **IP1500/2500/5000 Series Phones**, repeat Steps 7-13 for each device.

13 Using the IP1500 and IP2500 Series Speakerphones

The speakerphone can be configured for multiple uses. The main function is to provide 2-way voice communications. Pressing the red button will activate the configured script programmed for button #1.

Button #1 activation overrides any other action the speakerphone is performing at the time of the button press. For example if the speakerphone:

1. Is being programmed at the time
2. Was in a monitoring call
3. Was in the middle of a diagnostic test
4. Is currently in an information (button #2) call.

Button #2, **INFO** or **CALL** buttons are typically utilized for placing informational calls. Any action other than Button #1 activation is consider **Non-Priority** calling and commonly utilized for director service, student/employee escort requests, gate entry, guest services and similar requests.

The speakerphone's Auxiliary Output is typically utilized for activating Code Blue's LED Beacon/Strobe, and can be used as a normally open (N.O.) dry contact closure (see spec for relay ratings) used, for example, to activate centralized building/security management equipment.

Incoming calls: The speakerphone auto-answers an incoming call. (Based on the settings configured under **General Settings** in **General Configuration > Incoming Calls > Answer in** Immediately or after a number of rings.)

14 Troubleshooting

TROUBLESHOOTING THE IP1500 AND IP2500 SERIES SPEAKERPHONE

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

Power - Ensure the power to your device is working and rated for 802.11af PoE specifications.

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 your speakerphone. If you receive no response and PoE 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 registrar. See CLI Commands.

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

Account - Ensure your SIP or IAX2 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 your codec settings on your VoIP system match the IPspeakerphone codec settings.

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

Contact information for Code Blue's Technical Services and Support staff can be located at the end of this Guide if you need further assistance troubleshooting your speakerphone. Depending on your issue, a firmware upgrade may be needed.

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 your web browser.

15 Technical Specifications

Power Features	
Power over Ethernet IEEE 802.3af / at	
Communications	
IP Communications	
Environmental	
-40°C to 70°C (-40°F to 158°F)	
0% - 95% RH Non-condensing	
Standard Features	

- Full duplex speakerphone, intercom and paging device
- 1MB memory storage for phone numbers and audio messages
- Phone numbers up to 255 digits long
- SIP/IAX2 Protocol support
- STUN client for NAT transversal
- UDP, TCP and TLS
- 1 x IEEE 802.3 10/100 Ethernet port
- Embedded web server
- Security includes:
  - » HTTPS
  - » Transport Layer Security (TLS)
  - » SRTP (RFC3711), SIPS
  - » RTCP
  - » VLAN
  - » Password protection
- DTMF inband/out of band/INFO
- 1 x Hybrid pressure sensitive data button inputs
- 1 x auxiliary N.O. output contact closures with programmable timing capability
- Self-monitoring and fault reporting:
  - » Communication service
  - » Button failure
  - » Speaker failure
  - » Microphone failure
- Message Playback options:
  - » Multiple and repeating during call placed
  - » Multiple and repeating during call received
  - » Message playback during a call via DTMF commands
- In-Call commands via DTMF:
  - » Auxiliary output control
  - » Incremental increase/decrease speaker, microphone
  - » Message playback
- Built-in scripting language provides advanced button and diagnostic report programming
- Corrosion resistant connectors
- Enhanced speakerphone and microphone sensitivity
- Non-volatile memory ensures programming is retained during power loss
- Conformal coated PCBs for environmental protection and operation
- ADA compliant with Braille signage and LED indicators
- ToolVox Media Gateway emails fault status report about phone
- NEMA 4/IP 55 rated
- Dual account registration for redundancy
- Built with powerful DSP technology

For IP1500 Only

- » Standard Bezel Options
- » Standard Surface Mount Housing Color: Safety Blue
- » Standard Flush Mount Faceplate Color: Stainless

For IP2500 Only

- » Two highly visible LED indicators for ADA compliance for hearing impaired
- » Optional dual button faceplate

16 Regulatory

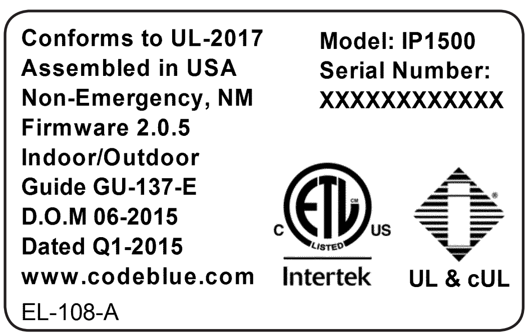
The IP1500 and IP2500 Series speakerphones conform to the following list of directives and product safety standards as applicable:

EU: EN 55022:2006+A1:2007
EN 55024:1998+A1:2001+A2:2003
EN 61000-4-2:1995
EN 61000-4-3:2006+A1:2008
EN 61000-4-4:2004
EN 61000-4-5:2006
EN 61000-4-6:2007
EN 61000-4-8:1993+A1:2001
EN 61000-4-11:2004
EN 61000-3-2:2006+A1:2007
EN 61000-3-3:2008

USA: CFR 47, Part 15
CANADA: ICES-003e

14.1 ETL Required Labeling

The IP1500 and IP2500 Series of full duplex VoIP speakerphones are labeled in accordance with the UL 2017 standard.



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/downloads**.

Notice: Every effort was made to ensure that the information in this document was complete and accurate at the time of printing. Information is subject to change.

18 Technical Services and Support

For additional support, please feel free to contact Code Blue's Technical Services and Support Staff at **ts@codeblue.com** or **(800) 205-7186, Opt 3.**

8 a.m. to 5 p.m. Monday through Friday Eastern Time