

Blueface Configuration Guide: SIP Speaker

Document Part # 931912B

CyberData Corporation

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Blueface Configuration Guide: SIP Speaker Document #931912B

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1.0 Setup Diagram

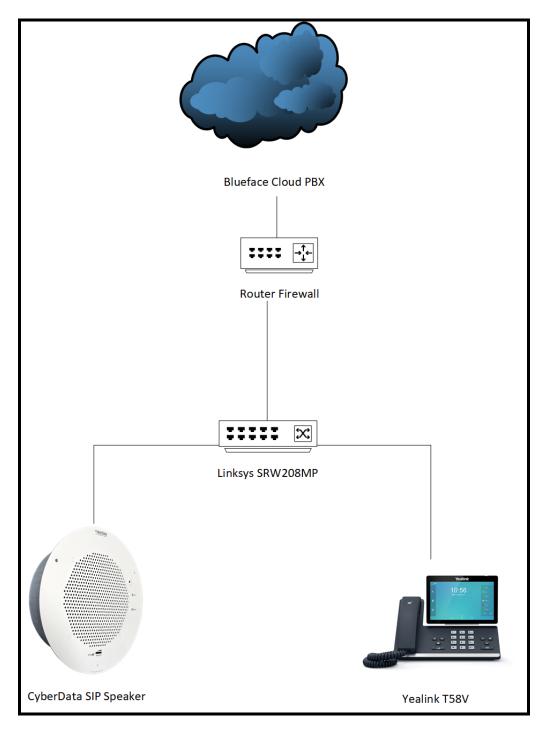


Figure 1-1: Interoperability Test Infrastructure



2.0 Test Setup Equipment

This section describes the products used for interoperability testing with Blueface.

Table 2-1: <u>Setup Equipment</u>

EQUIPMENT	MODEL or PART NUMBER	FIRMWARE VERSION
CYBERDATA SIP SPEAKER	011394	12.1.1
CYBERDATA SIP TALKBACK SPEAKER	011398	12.1.1



3.0 Before You Start

This configuration guide documents the integration process of the CyberData SIP Speaker.

Network Advisories

Blueface uses a Fully Qualified Domain Name (FQDN) for the SIP server address. The CyberData SIP Speaker's need to perform a DNS A query to resolve the IP address of Blueface's SIP Server FQDN. It is necessary to ensure the configured DNS server(s) have an A record for the SIP Server address.

In addition, be sure to verify the following ports are available for the speaker to use:

- UDP 5062 (SIP)
- UDP 10500 (RTP)

The speaker will need to traverse the public internet in order to operate with Blueface in the cloud.

The speaker's paging extension uses SIP port 5060 to receive SIP messages. The device will send SIP messages to port 5062, the port used by Blueface's SIP Server.

SIP ports 5060 and RTP port 10500 are the default values on all noted firmware levels.

Alternatively, SIP ports for the device are configurable on the SIP page of the web interface.

The CyberData Discovery Utility can be used to locate CyberData devices on your network. You may download it from the following web address: https://www.cyberdata.net/pages/discovery

Note: DHCP addressing mode is enabled on default on all noted firmware levels.



Product Documentation and Utilities

Before you start, download the Operation and Quick Start guides from the speaker's product webpage:

SIP Speaker (011394):

https://files.cyberdata.net/assets/011393,011394/011394_931181L_SIP_Speaker_Ops_Guide.pdf

SIP Talkback Speaker (011398): https://files.cyberdata.net/assets/011397,011398/011398_931191M_SIP_Talk-Back_Speaker_Operations_Guide.pdf



4.0 Configuration Procedure: Callflow Setup

Blueface does not allow users to add their own devices to the platform. The MAC addresses of the devices must be provided to the account manager, who can then add the devices to the platform for you. An email will then be generated and sent to you that will contain the registration information for the CyberData device.

Blueface requires a callflow to be created to call or make a call from any device. This section will outline how to create the dial plan.

1. Log into Blueface.

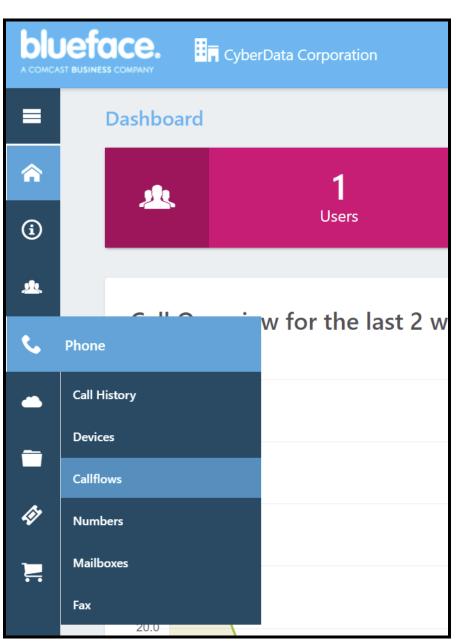
https://portal.nsvconnect.com/login

Username		
Password		٩
L	.OG IN	~
Forgot Password?	English - USA	~

Figure 4-1: Login



2. From the landing page **Phone** and then **callflows**.





3. On the Callflows page press Create New Flow.

Figure 4-3:	Callflow Page

allflows					
Search					٩
	Parks	A Flow Switches	s 🖒 Ci	reate New Flow	+
	Parks	A Flow Switches		reate New Flow	

4. Name the new callflow and set a description.

Figure 4-4: <u>Callflow designer</u>

Designer						
Name: *			Description:			Linked Numbers:
SIP Speaker			SIP Speaker			Linked Numbers
Callflow						× × ₹
			+			*
			💊 🙆 📼			
			+			_
						_
						_
						*
Undo	6	Erase Callflow		Cancel	0	Save >

5. From Elements drag **Ring** into the Callflow.



- 6. Click the yellow exclamation point to open the **Ring Element Settings** popup.
- **7.** Select the Phone tab in the popup.
- 8. Select the Device that will be used in the group.

Ring Ele	ement Settings				×
Timeout: 20 This config	*) seconds			
1	د ₹	만	Selecte	ed destinations	
Search		٩	c	Generic SIP Device - SIP Speaker - Line 1	â
Devices					
	SIP Paging Adapter G Generic SIP Device	÷			
٩	SIP Paging Amplifier	•			
	SIP Paging Server B Generic SIP Device	+			
	SIP Strobe	+			
	Video Intercom with Keypad	٠			
< 1	2 3 >				
Extra opt	tions				*
				Cancel	Ok 🗸

Figure 4-5: <u>Ring Element Settings</u>

- 9. Press Ok to save the device to the callflow.
- **10.** Press Save to save the callflow.



Figure 4-6: Callflow Designer

Designer			
Name: *	Description:		
SIP Speaker	SIP Speaker		
Califlow			× × ×
	ے ایج بی میں ب	c SIP Device.	
			-
Unde 🔍	Erase Califlow	Cancel 🛇	save >

- 11. Click the Save button to create the Phone.
- **12.** Next link a number to the new callflow.
- **13.** Save the number to the callflow.

Figure 4-7: Link a Number

Numbers linked to SIP Speaker			×
These are the numbers linked to call flow SIP Speaker. Calls to any of these numbers will flow through the elements defined in this call flow.	want to reuse a linke	ble numbers you can lini d number, unlink first.	
Number ↓	Search	e numbers only	٩
	Number 🖡	Callflow 1 Linked	
203 Unlink 🎉	200	All Page Zone	Unlink 🔗
	201	SIP Paging Adapter	Unlink 🔗
	202	SIP Paging Server	Unlink 🔗
	204	SIP Paging Amplifier	Unlink 🔗
	205		Link <i>S</i>
	× 1 2	3 4 5 >	
		Cancel	Save 🗸



5.0 Configuration Procedure: Setting up the Paging Extension

For configuring through the web interface, use the following steps to login to the web interface of your CyberData device.

CyberData Setting	Blueface Email
Primary SIP Server	SIP Server
Primary SIP User ID	Username
Primary SIP Auth ID	Authentication ID
Primary SIP Auth Password	Password

Table 5-1: <u>Setting Name correlation</u>

1. Click Launch Browser from the CyberData Discovery Utility or point your browser to the CyberData device's IP address to access the Home Page of the web interface.

CD Discovery U	tility			- 🗆 ×
Cyber[VoIP Discovery Utility
IP Address	DHCP	MAC Address	Serial Number	Device Name
192.168.1.15	Enabled	00:20:f7:04:5d:ce	398001862	CyberData SIP Speaker
Discover	Open Browser			Quit

Figure 5-1: CyberData Discovery Utility

2. Enter the default credentials when prompted and click the Log In button.

Username: admin Password: admin



Figure 5-2: Home Tab

Home De	evice Audio Network	SIP Multicast SSL Sensor	Audiofiles Events Autoprov Firmware
	Cybe	erData SIP S	peaker
Current Sta	atus	Admin Settings	Import Settings
Serial Number:	398001862	Username: admin	Choose File No file chosen
Mac Address:	00:20:f7:04:5d:ce	Password:	
Firmware Version:	v12.1.1	Confirm Password:	Import Config
IP Addressing:	DHCP		E
IP Address:	192.168.1.15		Export Settings
Subnet Mask:	255.255.255.0	Save Reboot Toggle Help	
Default Gateway:	192.168.1.1		Export Config
DNS Server 1: DNS Server 2:	192.168.1.1		
SIP Mode:	Enabled		
Multicast Mode:	Enabled		
Event Reporting: Nightringer:	Disabled Disabled		
nightinger.	Disaulu		
Primary SIP Serve	: Not registered		
Backup Server 1:	Not registered		
Backup Server 2:			
Nightringer Server			
Monitor SIP Server	: Not registered		

3. Navigate to the SIP tab.

Note: All SIP credentials are listed in an email sent by Blueface after the device was added to the platform.

- 4. Set the **Primary SIP Server** to the value listed for SIP Server.
- 5. Set the **Primary SIP User ID** to the value listed for the Username.
- 6. Set the Primary SIP Auth ID to the value listed for the Authentication ID.
- 7. Set the **Primary SIP Auth Password** to the value listed for the Password.
- 8. Set the **Remote SIP Port** to 5062.



Figure 5-2: <u>SIP Tab</u>

SIP Settings		Nightringer Sett	ings
Enable SIP operation:		Enable Nightringer:	
SIP Transport Protocol:	UDP 🗸	SIP Server:	10.0.0.253
TLS Version:		Remote SIP Port:	5060
Verify Server Certificate:		Local SIP Port:	5061
Register with a SIP Server: Use Cisco SRST:		Outbound Proxy:	
Primary SIP Server:	cust-uc-us.nsvconnect.com	Outbound Proxy Port:	0
Primary SIP User ID:		User ID:	241
-	HwD8a5ZHpfAGKDcGEUyB	Authenticate ID:	241
Primary SIP Auth ID:	HwD8a5ZHpfAGKDcGEUyB	Authenticate Password:	
Primary SIP Auth Password:		Re-registration Interval (in s	
Backup SIP Server 1:			
Backup SIP User ID 1:		DTD Cattings	
Backup SIP Auth ID 1:		RTP Settings	
Backup SIP Auth Password 1:		RTP Port (even): 10500	
		Jitter Buffer: 50	
Backup SIP Server 2:		SRTP: Disabled	
Backup SIP User ID 2:			
Backup SIP Auth ID 2:		Call Disconnect	ion
Backup SIP Auth Password 2:		Terminate Call after delay:	
		leminate can alter delay.	<u> </u>
Remote SIP Port:	5062		
Local SIP Port:	5060	Codec Selection	
Outbound Proxy:		Force Selected Codec:	
Outbound Proxy Port:	0	Codec: PCI	/IU (G.711, u-law) 🗸
Marilla a Iba			
Monitor User ID: Monitor Authenticate ID:		Button Settings	
Monitor Authenticate Password:			
Monitor Authenticate Password.		Dial Out Extension: 204	100
Disable rport Discovery:		Extension ID: Classro	om 102
Buffer SIP Calls:			
Re-registration Interval (in seconds)	: 360		
Unregister on Boot:			
Keep Alive Period:	10000		
Save Reboot Toggle Help			

9. Save and Reboot.

Once the speaker finishes rebooting the unit should show Registered on the home tab.



Figure 5-3: <u>Home Tab – Registered</u>

Home Device Au	lio Network	SIP Multicast	SSL	Sensor	Audiofiles	Events	Autoprov	Firmware
Current Status Serial Number: 398001862 Mac Address: 00:20:77:04:5c Firmware Version: v12.1.1	-	Admin Settin Username: a Password: Confirm Password:		P S		ker ort Settir se File No file (ngs	
IP Addressing: DHCP IP Address: 192.168.1.15 Subnet Mask: 255.255.250 Default Gateway: 192.168.1.1 DNS Server 1: 192.168.1.1 DNS Server 1: 192.168.1.1 DNS Server 2: SIP Mode: Enabled Multicast Mode: Enabled Event Reporting: Disabled Nightringer: Disabled Primary SIP Server: Registered Backup Server 1: Not registered Backup Server 2: Not registered Nightringer Server: Not registered Monitor SIP Server: Not registered			Foggle Help		Exp	ort Settir	ıgs	



5.1 Configuration Procedure: Setting up the Nightringer Extension

The Nightringer Extension is a secondary extension that will ring when called. This makes the Nightringer extension ideal for use in ring groups.

1. Navigate to the web interface of the speaker.

Figure 5-4: Home Tab

Home De	vice Audio N	etwork SIP Multicast SSL	Sensor Audiofiles Events Autoprov Firmware
	Су	berData SI	P Speaker
Current Sta	atus	Admin Settings	Import Settings
Serial Number: Mac Address: Firmware Version: IP Addressing:	398001862 00:20:f7:04:5d:ce	Username: admin Password: Confirm Password:	Choose File No file chosen
IP Address:	192.168.1.15		Export Settings
Subnet Mask: Default Gateway:	255.255.255.0 192.168.1.1	Save Reboot Toggle Help	
DNS Server 1: DNS Server 2:	192.168.1.1		Export Config
SIP Mode: Multicast Mode: Event Reporting: Nightringer:	Enabled Enabled Disabled Disabled		
Primary SIP Server Backup Server 1:			
Backup Server 2: Nightringer Server Monitor SIP Server	Not registered		
monitor an aerver	. Not registered		

2. Navigate to the SIP tab.

Note: All SIP credentials are listed in an email sent by Blueface after the device was added to the platform.

- 3. Check the box Enable Nightringer.
- 4. Set the **SIP Server** to the value listed for SIP Server.
- 5. Set the User ID to the value listed for the Username.
- 6. Set the Authenticate ID to the value listed for the Authentication ID.
- 7. Set the Authenticate Password to the value listed for the Password.
- 8. Set the **Remote SIP Port** to 5062.



Figure 5-5: SIP Tab - Nightringer

SIP Settings			Nightringer	Settings	
Enable SIP operation:			Enable Nightringer:		
SIP Transport Protocol:			SIP Server:		cust-uc-us.nsvconnect.com
TLS Version:		~	Remote SIP Port:		5062
Verify Server Certificate: Register with a SIP Server:			Local SIP Port:		5061
Use Cisco SRST:			Outbound Proxy:		
Primary SIP Server:	cust-uc-us.nsvconnect.com		Outbound Proxy Po	rt:	0
Primary SIP User ID:	HwD8a5ZHpfAGKDcGEUyB		User ID:		PbJrP4FW7z45nuK6Jc9Z
Primary SIP Auth ID:	HwD8a5ZHpfAGKDcGEUyB		Authenticate ID:		PbJrP4FW7z45nuK6Jc9Z
Primary SIP Auth Password:			Authenticate Passw	ord:	
· · · · · · · · · · · · · · · · · · ·			Re-registration Inter	rval (in seconds):	360
Backup SIP Server 1:					
Backup SIP User ID 1:			DTD Sotting	10	
Backup SIP Auth ID 1:			RTP Setting	IS	
Backup SIP Auth Password 1:			RTP Port (even): 10	500	
			Jitter Buffer: 50		
Backup SIP Server 2:			SRTP: Di	sabled 🗸	
Backup SIP User ID 2:					
Backup SIP Auth ID 2:			Call Discon	nection	
Backup SIP Auth Password 2:			Terminate Call after	dalaw 0	
			Terminate Gall after	delay: 0	
Remote SIP Port:	5062				
Local SIP Port:	5060		Codec Sele	ction	
Outbound Proxy:			Force Selected Cod	ec:	
Outbound Proxy Port:	0		Codec:	PCMU (G.711	, u-law) 🗸
Monitor User ID:			Button Sett	inge	
Monitor Authenticate ID:				-	
Monitor Authenticate Password:			Dial Out Extension:	204	
Disable met Discovery			Extension ID:	Classroom 102	
Disable rport Discovery: Buffer SIP Calls:					
Re-registration Interval (in seconds)	: 360				
Unregister on Boot:					
Keep Alive Period:	10000				
Save Reboot Toggle Help					

9. Save and reboot the speaker.

If the credentials were added correctly, when the unit finishes rebooting Registered in Green should appear next to Nightringer Status on the Home Tab.



Figure 5-6: Nightringer Registered

Home De	vice Audio Ne	etwork SIP N	Aulticast SSL	Sensor	Audiofiles	Events	Autoprov	Firmware
Current Sta	-	berDa	ata SI	PS	•	ker		
Serial Number:	398001862	Username:	admin			e File No file o	100	
Mac Address: Firmware Version:	00:20:f7:04:5d:ce v12.1.1	Password: Confirm Pa	ssword:		Impor	t Config		
IP Addressing: IP Address: Subnet Mask: Default Gateway: DNS Server 1: DNS Server 2:	DHCP 192.168.1.15 255.255.255.0 192.168.1.1 192.168.1.1	Save F	Reboot Toggle Help			o rt Settir t Config	ngs	
SIP Mode: Multicast Mode:	Enabled Enabled							
Event Reporting: Nightringer:	Disabled Enabled							
Primary SIP Server Backup Server 1: Backup Server 2: Nightringer Server Monitor SIP Server	Not registered Not registered Registered							



6.0 Using CyberData SIP Speakers.

CyberData Speakers are designed for one-way communication. When a call is made to the device an announcement can be made. The units can be used by directly calling the SIP extension, in a page group, or with multicast. This makes the speaker's extremely versatile paging endpoints

CyberData SIP Speakers also come in a talkback format which allows the speaker to be used for two-way communication. This will typically require the use of the accessory Remote Call Button (011508). The addition of the accessory call button allows a call to be made from the speaker, although the speaker can still be called and used in a two-way conversation without the call button.

6.1 Setting up a page group

After registering the device to Blueface, a page group can be created which allows a call to be made which can reach multiple endpoints simultaneously. This allows for zoned paging directly through the service and does not require additional hardware.

- 1. Select Callflows in Phones on Blueface.
- 2. Name the new callflow and set a description.
- 3. In the callflow designer select the Paging Element.

Figure 6-1: Page Group Designer

Designer		
Name: *	Description:	
All Page Zone	All Page	
Califlow	2 2 P	Elements
	4	Voicemail
		🗰 Time of Day
	at 🙆 🔤	III Menu
		O Flow Switch
		🖻 Switch Menu
		A Hot Desk
		4 Conference
		Video Conference
Undo	Cancel Save >	ரி Paging

- 4. After adding the paging element, click on it to assign users.
- 5. Add all necessary users for the paging group.



Paging Element Set	tings		×		
③ PIN:	Oisplay Name: All Page				
T r	∀	Selecte	d destinations		
Search	٩	c	Generic SIP Device - SIP Paging Amplifier - Line 1		
Devices		c	Generic SIP Device - SIP Paging Adapter - Line 1		
Yealink VP-T49G IP Video Phone		Generic SIP Device - SIP Speaker - Line 1			
Generic SIP Dev	ice	c	Generic SIP Device - SIP Strobe - Line 1		
< 1 2 >		c	Generic SIP Device - Office Ringer - Line 1		
		¢.	Generic SIP Device - SIP Paging Server - Line 1		
Extra options					
	No media available		Add new media + Record 🎈		
	Media Uploader				
Maximum fil			pp Files or Browse um file size 5mb ac, wma, flac, mp1, mp2, mp3, mp4, opus, ra		
			Cancel 🛇 Add Media 🕇		
Accessible From Off Net: OFF	Bypass on Call Pr OFF	otection:			
			Cancel 🛇 Ok 🗸		

Figure 6-2: Paging Element Creation

- 6. Once all the desired users are added, press Ok.
- 7. Next save the new call flow.



Figure 6-3: Callflow Created

Designer							
Name: *			Description:				
All Page Zone			All Page				
Callflow						×	,* ₽
			+				*
			Ring: line Generic SIP Device.				
							*
Undo	0	Erase Callflow	t i	Cancel	0	Save	>

8. After saving the callflow click **Linked Numbers** to set an extension number for the paging group callflow.

Figure 6-4: Linked Numbers



9. Set a number for the paging group.



Figure 6-5: Linking Number

These are the numbers linked to call flow All Page Zone. Calls to any of these numbers will flow through the elements defined in	These are the available numbers you can link to the callflow. want to reuse a linked number, unlink first.					
his call flow.	Search		٩			
Number 🕹	Show Available	numbers only				
	Number 🕹	Callflow 1 Linked 1				
200 Unlink 🔗	201	SIP Paging Adapter	Unlink 🔗			
	202	SIP Paging Server	Unlink 🔗			
	203	SIP Speaker	Unlink 🔗			
	204	SIP Paging Amplifier	Unlink 🔗			
	205		Link S			
	< 1 2	3 4 5 >				

10. Press Save to save the number to the callflow.

The callflow is now ready to be used. When called it will send a SIP call to all group elements and allow a page to be made.



6.2 Multicast Setup

Most CyberData devices support Multicast which is a protocol that allows for easy paging on a local area network (LAN). This section will illustrate how to setup the device to listen for multicast and the different settings that work with multicast.

Priority	Address	Port	Name	Buffer	Deen	Deley
	239.168.3.10	11000		Buffer	Beep	
9			Emergency Warning			
8	239.168.3.9	10000	All Page			
7	239.168.3.8	9000	Warehouse Only			
6	239.168.3.7	8000	Unused			
5	239.168.3.6	7000	Unused			
4	239.168.3.5	6000	Unused			
3	239.168.3.4	5000	Unused			
2	239.168.3.3	4000	Unused			
1	239.168.3.2	3000	Unused			
0	239.168.3.1	2000	Background Music			
Polycom Default Channel 1 Polycom Priority Channel 24 Polycom Emergency Channel 25 SIP calls are considered priority 4.5						
Port range can be from 2000-65535 Priority 9 is the highest and 0 is the lowest						
Priority 9 is the highest and 0 is the lowest A higher priority audio stream will always supersede a lower one						

Figure 6-6: Multicast Tab

The multicast engine workings on priority, higher priority supersedes a lower priority. CyberData recommends setting all pages or emergency pages to a higher priority, this will prevent a non-emergency message playing over any emergency notifications. There are also options to Buffer the message, play a beep tone before the message or enable the onboard relay for the duration of the message.



6.3 Remote Call Button Setup

This section walks through the process of wiring and enabling the Remote Call Button (011508). While this accessory is not required to use the speaker in a two-way call, the button is typically installed to allow a call to be made from the speaker.

To use the button it must first be wired to the speaker. The button uses a four-wire connection which notifies the speaker of the call button being pressed but also applies power to the button to illuminate the LED.

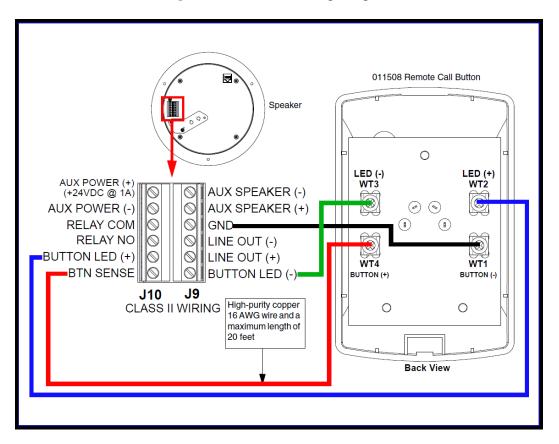


Figure 6-7: Button Wiring Diagram

1. With the speaker disconnected from power use a small screwdriver to wire the button to the speaker.

Note: CyberData recommends a 16 AWG wire with a maximum length of 20 feet.

- 2. After wiring the button to the speaker, connect the speaker back to power and mount both the speaker and button in place.
- 3. As the speaker boots up the button's LED ring should illuminate.



- 4. Navigate to the SIP Talkback Speaker's web interface.
- 5. From the Home tab, click on the Device Tab.
- 6. On the Device Tab, check the box for **Button Installed** located in the Button Settings section.
- 7. Adjust any settings as necessary, CyberData recommends using the setting **Play Ringback Tone** to alert the user of the speaker that the call is being setup.

Figure 6-8: Button Settings



Note: The setting *Prevent Call Termination* is useful in some scenarios where a call should not be ended from the speaker. CyberData finds this setting to be used frequently with the talkback speakers when located in schools. This will prevent a second press of the button on the speaker from ending the call.

- 8. Save.
- 9. Navigate to the SIP Tab.
- **10.** Set the **Dialout Extension** to the number that will be called when the accessory button is pressed.
- 11. Set the Extension ID to the location of the speaker

Figure 6-9: <u>Button Settings – Dialout</u>

Button Settings			
Dial Out Extension:	204		
Extension ID:	Classroom 102		

12. Save.

13. Reboot.

Once the speaker has booted back up, pressing the call button accessory will trigger a call from the speaker.



7.0 Contact CyberData Corporation

Sales

For sales-related questions, please visit our <u>Contact CyberData Sales</u> web page for more information.

Technical Support

For CyberData Technical Support, please submit a <u>Contact CyberData VoIP Technical Support</u> form on our website.

The CyberData VoIP Technical Support Contact form initiates a troubleshooting ticket which CyberData uses for quality assurance purposes.

Additionally, the Contact VoIP Tech Support form tells us which phone system you are using, the make and model of the network switch, and other essential troubleshooting information we need to efficiently assist with a resolution. Please also include as much detail as possible in the Describe Problem section of the form. Your installation is extremely important to us.

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