

CyberData SIP Paging Ceiling Speaker V3 Integration with 8x8



This document covers the integration of CyberData's SIP Paging Speaker V3 with 8x8. This document was written for 8x8 and the following CyberData Products.

- 011397 (RAL 9002, Gray White) Talk Back Speaker
- 011398 (RAL 9003, Signal White) Talk Back Speaker
- 011393 (RAL 9002, Gray White) Ceiling Speaker (no talk back)
- 011394 (RAL 9003, Signal White) Ceiling Speaker (no talk back)

All support and supporting documentation for CyberData should be obtained from CyberData itself. This document also assumes the reader is familiar with setting up CyberData Paging equipment and/or has access to the Manuals for the CyberData equipment, as several sections are left out of this manual such as setting up the network configuration of the CyberData Equipment and pin outs for relay, and audio out usage.

CyberData devices do integrate with both Yealink and Polycom devices, 8x8 suggests using Yealink devices over Polycom if more than one zone is needed. For more information on the integration process see integration section.

1 Contents

2	INTEGRATION.....	2
2.1	SIP PAGE AND CONVERTED TO MULTICAST	3
2.2	PURE MULTICAST	3
2.3	TRADITIONAL PAGING EQUIPMENT	3
3	MULTICAST PAGING	4
3.1	HOW MULTICAST PAGING WORKS.....	4
3.2	CAVEATS OF MULTICAST PAGING.....	4
3.3	ADVANTAGES OF MULTICAST PAGING.....	4
4	SIP PAGING	4
4.1	HOW SIP PAGING WORKS	4
4.2	CAVEATS OF SIP PAGING	5
4.3	ADVANTAGES OF SIP PAGING	5
5	CREATING A USER PROFILE ON 8X8 FOR SIP CALLS AND NIGHT RING CAPABILITIES	5
5.1	CREATE USER PROFILE	5
6	ASSIGN USER TO THE DEVICE	6
7	CYBERDATA SIP CELLING SPEAKER SETUP	8
7.1	CONNECTING TO THE CYBERDATA SIP CELLING SPEAKER	9
7.2	HOME SCREEN	10
7.3	DEVICE CONFIGURATION	12
7.4	NETWORK CONFIGURATION	13
7.5	SIP CONFIGURATION	14
7.6	NIGHTRINGER CONFIGURATION.....	16
7.7	MULTICAST (PAGING GROUPS)	18

2 Integration

CyberData SIP Ceiling Speaker can be integrated in multiple ways with 8x8, each integration option has its unique benefits and draw backs. For the best integration between all types of phones (Polycom, Yealink, and other 3rd party devices) as well as routing and remote devices 8x8 recommends SIP Page and Converted to Multicast.

The CyberData SIP Ceiling Speaker can listen in to Multicast Streams that are pre-defined and relay this pages to its Audio Out ports to traditional Paging equipment (this does not require DTMF input), or it be called directly by using an extension assigned to it, to start the page.

Technical Publications

If equipped with the talk back function when the SIP extension is dialed, you can configure talk back (2-way audio) with the speaker. Additional options like Clock, Relays, Call button, and a strobe can be added.

Starting with CyberData firmware 7.2.0 you can integrate Polycom Group Paging with traditional Multicast paging services. This is accomplished via CyberData firmware enhancements to provide Multicast and Group Paging features at the same time, for more information see CyberData's website.

2.1 SIP Page and Converted to Multicast

This will afford the possibility to integrate the Yealink, Polycom and other 3rd party equipment along with the CyberData Paging equipment.

Traditional paging equipment can be integrated into this solution using CyberData SIP Ceiling Speaker's Audio Out port and Relays.

Users will simply dial the page extension, and make their page. A SIP call will be placed to the CyberData Ceiling Speaker (and any other device in the page group).

2.2 Pure Multicast

When using CyberData paging equipment you can integrate as a pure multicast solution, in that you will no longer use the paging services of 8x8, and rely purely on Multicast capabilities of the Polycom, Yealink and CyberData equipment. When using CyberData's SIP Ceiling Speaker and Yealink phones you may either dedicate a unique paging button on the phone per page zone. The Yealink phones only support listening to 5 multicast paging zones. If using Polycom phones, you can only use one of the Polycom paging groups.

Users will press a predefined paging button on the Polycom and Yealink devices to initiate the page, this will start a multicast from the device to all other devices listening to the same multicast IP address and Port pair.

2.3 Traditional Paging Equipment

Traditional Paging equipment can be integrated into either integration option (SIP or Multicast) by the CyberData SIP Paging Server. It is recommended to use the CyberData SIP Paging Server to integrate with traditional paging equipment input and offers the ability to include relays. The CyberData Paging Server supports a 600 Ohms at 5 VPP output referred to as Page Port and a 10K Ohms at 2 VPP output referred to as Line Out.

If integrating multiple locations with traditional paging equipment it is recommended to use the CyberData Paging Adapter to integrate additional locations using SIP paging from the 8x8, and the CyberData Paging Adapter to integrate to the traditional paging equipment.

3 Multicast Paging

3.1 How Multicast Paging Works

After a user presses a configured “Paging” key on the phone, the phone sends a page message (which is an RTP stream, hereinafter referred to as a “page”) to a preconfigured multicast address. Any device in the local network listens for the page on the preconfigured multicast address. The device will display the multicast page sent/received address to the user. You can define multiple multicast zones by using a different multicast IP or port number, a single device can listen to multiple IP:Port combinations.

The device uses G711 uLaw CODEC for multicast paging.

The recipient can drop the incoming page if required. The recipient can also press Do Not Disturb (DND) or other “ignore” options on the device to ignore/reject any incoming pages.

3.2 Caveats of Multicast Paging

Multicast paging is designed for Yealink and Polycom devices. There is no guaranteed interoperability with any other 8x8 supported phones. CyberData Paging Equipment is an exception, as it has been tested and certified to work properly with the Yealink and Polycom phones. The Virtual Office Desktop Softphone does not support multicast paging.

This service is typically non-routable, and cannot be used to page across the WAN, complex VLANs, or to remote devices.



Note: Multicast page is one-way only - from sender to the receiver.



Note: For outgoing pages, all other existing calls on the phone are put on hold.

If a page session already exists on the phone, and the phone receives another incoming page, the priority is given to the first multicast session and the second multicast session is ignored. The behavior for the incoming calls in this case is also based on the setting for the “Allow Barge In” parameter. The incoming call is handled as if there were an existing call already on the phone.

3.3 Advantages of Multicast Paging

Multicast paging allows for virtually unlimited paging capability in a local network, does not require a session license to operate, and is almost instantaneous, as it does not require the phones to acknowledge the page request.

4 SIP Paging

4.1 How SIP Paging Works

SIP paging works as follows: the 8x8 places a SIP call to the device with an auto answer flag, the Cyber Data Ceiling Speaker will auto answer when properly configured for auto.

4.2 Caveats of SIP Paging

- Limited to 1 device currently, unless using the Configuration Manager.

4.3 Advantages of SIP Paging

- Works with remote devices.
- Works with the Yealink and Polycom product line.

5 Creating a User Profile on 8x8 for SIP Calls and Night Ring Capabilities

If using the SIP Call and or Night Ring capabilities of the SIP Ceiling Speaker, a softphone device should be ordered and a user is required to be created on 8x8. Create a user profile and assign the new user profile to the softphone only device ordered. This will be needed to be done for Each Registration required on the CyberData Device. If not using the SIP Call or Night Ring capabilities of the CyberData equipment this section can be skipped.

5.1 Create User Profile

In account manager, click on Accounts and then User Profiles. Click Create New User Profile. Provide the following information:

- First Name (Required)
- Last Name (Required)
- Nickname (Optional)
- Email Address (Required, and must be unique)
- Job Title (Optional)
- Department (Optional)
- Location (Optional)
- User Name (Required)
- Salesforce ID (Ignore)
- Zendesk ID (Ignore)
- NetSuite ID (Ignore)
- Mobile (Ignore)
- Language (Optional, Leave as Default)
- Time Zone (Optional, Leave as default)

Technical Publications

Create a New User Profile

First Name *

Last Name *

Nickname

Email Address *

Job Title

Department

Location

User Name *

SalesForce ID

Zendesk ID

NetSuite ID

Mobile

Language

Time Zone

English (U.S.)

US/Eastern

*=Indicates Required Fields

Save

Save / Add Another

Cancel

Click on Save (or Save / Add Another if going to add a Page user as well).

6 Assign User to the Device

After creating the user profile that will interface with CyberData Equipment, assign the user to the device. In Account Manager select Phone System, and then click on View All Extensions.

HOME | PHONE SYSTEM | BILLING | REPORTING | ORDERS | ACCOUNTS | SUPPORT | VIRTUAL OFFICE ONLINE

Home > Phone System > Extensions

Enter keyword Search

Help

PHONE SYSTEM

Extensions

Auto Attendant

Virtual and Toll-Free Numbers

Ring Groups

Music on Hold

Call Queues

Branches

Switchboard

Paging

Company Settings

Number Transfer Request

Call Recording

Edit Voicemail / Fax Notifications

Group Call Pickup

Call Park Extensions

Cordless Devices

Extensions

Quick Find / Edit Extension

Search Reset

Enter extension number, phone number or caller ID.

View All Extensions

Edit Multiple Extensions

Change Extension Numbers

Download Call Recordings

Line Key Configuration

Outbound Calling Options

From the list of extensions find the extension ordered for the Cyber Data Device, and click Edit.

Technical Publications

Edit	Active	Unlimited Extension			Unassigned	Unassigned	Unassigned
------	--------	---------------------	--	--	------------	------------	------------

Set the following item, the rest can be left as “default”.

- Enable Virtual Office: No/Unchecked
- Enable Virtual Office Mobile: No/Unchecked
- Verify Preferred Codec is set to G.711U (90 kbps)

The screenshot shows the 'Edit Extension' page in a web application. On the left is a sidebar with a 'PHONE SYSTEM' menu containing various options like 'Extensions', 'Auto Attendant', 'Ring Groups', etc. The main content area is titled 'Edit Extension' and includes a 'Help' link. Below the title is a note about expand/collapse icons and a 'Save Changes' button. The page is divided into two main sections: 'Extension Information' and 'Extension Settings'. In the 'Extension Information' section, there is a 'User Profile' dropdown menu with a 'Select User Profile' link. A blue callout bubble points to this link with the text 'Click to add the User Profile Created'. Below this, there are fields for 'External Caller ID' (Phone Number, Caller ID Full Name), 'Internal Caller ID' (First Name, Last Name, Caller ID Full Name), and 'Caller ID Option Locked to User?'. In the 'Extension Settings' section, the 'Preferred Codec' is set to 'G.711U (90 kbps)'. A blue callout bubble points to this dropdown with the text 'Must be G.711'. To the right of the settings, there are checkboxes for 'Enable Virtual Office' and 'Enable Virtual Office Mobile', both of which are unchecked. A blue callout bubble points to these checkboxes with the text 'Uncheck'. At the bottom right of the settings, there are several other checkboxes for features like 'Travelling Outside the Country', 'View Billing Statements', 'Hide in Auto Attendant Directory', 'Enable Inbound Caller ID', 'Allow Music on Hold Selection', 'Do Not Disturb', and 'Permanent Caller ID Blocking'.

Then click on “Select User Profile to add the appropriate User Profile, by clicking the “select” next to the profile you want to use.

Select User Profile

To search for a user profile, type the user profile information and click on the search button. You can also click on the view all button to see all the user profiles.

Search

Reset

View All

Actions	First Name	Last Name	Email Address	User Name
Select	Agam
Select	Alan
Select	Ali
Select
Select
Select
Select
Select	CyberData	PageServer
Select
Select
Select
Select
Select
Select
Select

Cancel

It will return you to the previous screen and click on “Save Changes”.

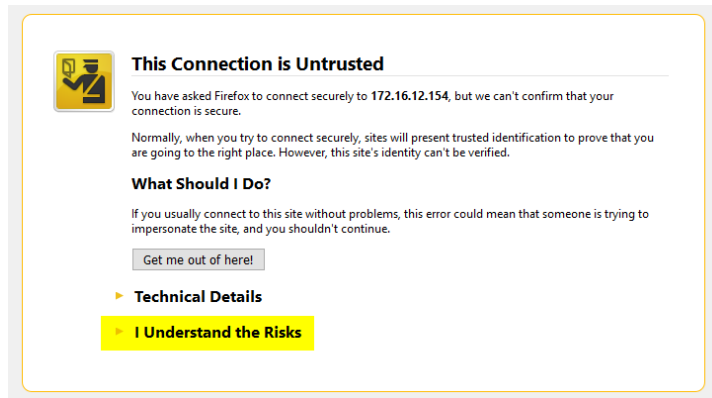
7 CyberData SIP Ceiling Speaker Setup

When deploying the CyberData SIP Ceiling Speaker it is recommended to use DHCP. CyberData provides a “Discovery Utility” that can be downloaded from their website (http://www.cyberdata.net/support/voip/discovery_utility.html) to initially discover the IP address of the SIP Ceiling Speaker. Using the CyberData Discovery Utility to obtain the current IP address of the CyberData SIP Ceiling Speaker login using a web browser using the default username of “**admin**” and the default password of “**admin**”. For more information on using the discovery utility and basic setup of the CyberData equipment, please refer to the operating manuals from CyberData. If using the pure multicast integration option, the CyberData equipment will not be registering with 8x8.

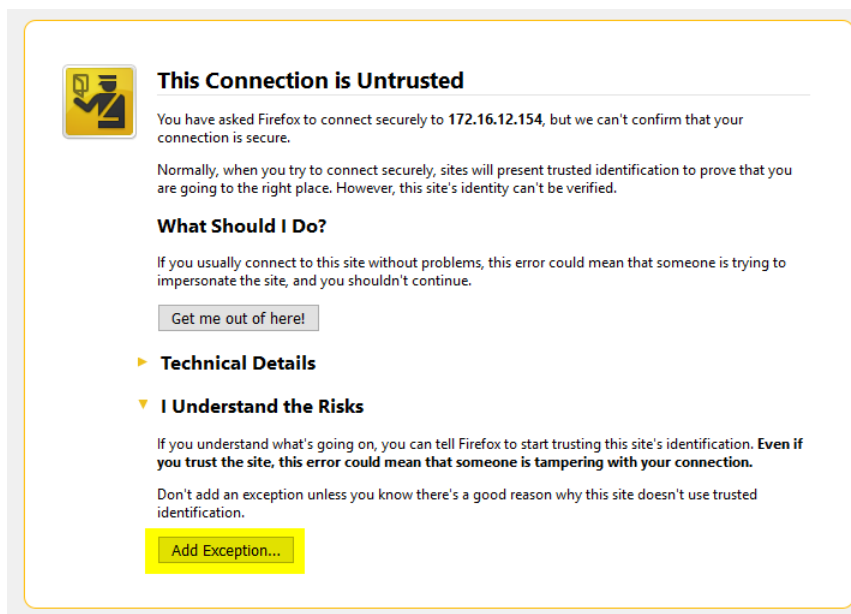
Technical Publications

7.1 Connecting to the CyberData SIP Ceiling Speaker

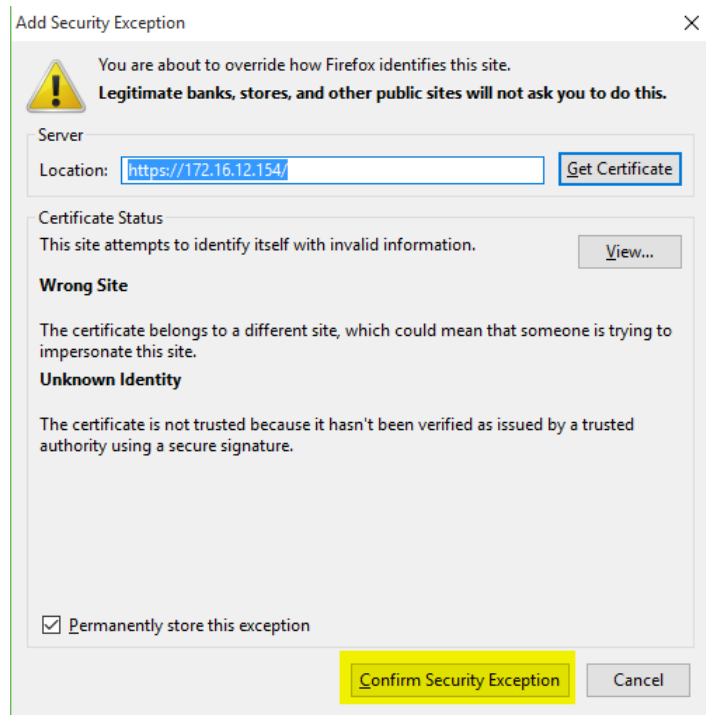
The CyberData SIP Ceiling Speaker now uses HTTPS to provision the device. When connecting to the CyberData Ceiling Speaker you will be required to accept the Self Signed certificate by clicking on “I understand the risks” link.



Then click “Add Exception”.



And then click Confirm Security Exception.



7.2 Home Screen

After logging into the CyberData SIP Ceiling Speaker using your favorite browser you are immediately taken to the Home Screen which will display the following information

On the Top, you will find your navigation options,

Change Username: Type in this field to change the username (25-character limit).

- Default: **admin**

Change Password: Type in this field to change the password (19-character limit).

- Default: **admin**

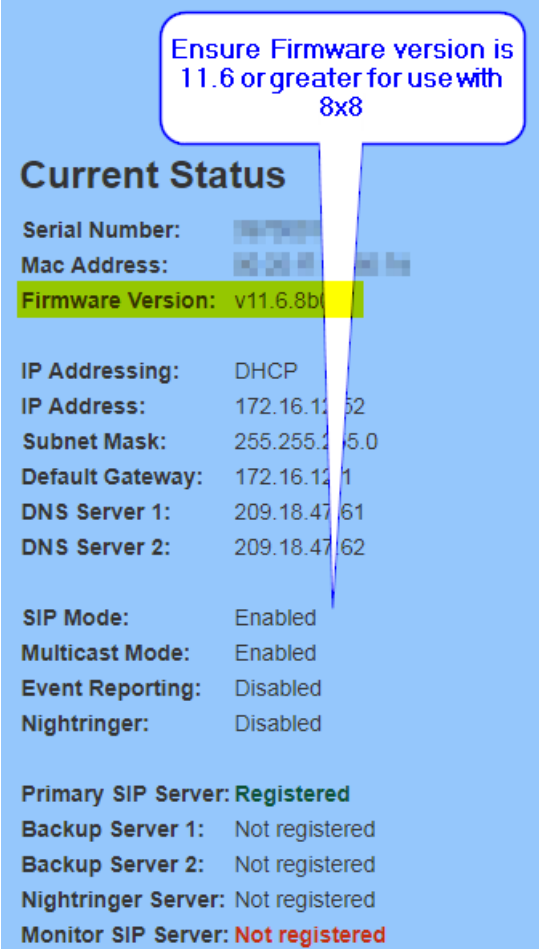
Re-enter Password: Type the password again in this field to confirm the new



password (19-character limit).

Current Settings:

Provides you with the current IP addressing of the device, Mac address and serial number.



Ensure Firmware version is 11.6 or greater for use with 8x8

Current Status

Serial Number: [REDACTED]
Mac Address: [REDACTED]
Firmware Version: v11.6.8b

IP Addressing: DHCP
IP Address: 172.16.12.52
Subnet Mask: 255.255.255.0
Default Gateway: 172.16.12.1
DNS Server 1: 209.18.47.61
DNS Server 2: 209.18.47.62

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**

The home screen will also show the current registration status, and features enabled on the CyberData SIP Paging Speaker.



Click on the Save button to save your configuration settings.



Note: You need to reboot for changes to take effect.

Reboot

Click on the Reboot button to reboot the system.

7.3 Device Configuration

On the device configuration screen, you can configure several default options for the paging speaker.

It is 8x8's recommendation to leave all these settings as default.

If using a Cyber Data add on such as the Clock, Talkback, Strobe you may see additional options here.

Please refer to the Cyber Data Manual for additional information on this section.

Device Name: Shows the device name (25-character limit). If using multiple paging speakers, please provide a unique name for each speaker. This name should reference the physical location of the device for future administration purposes.

CyberData V3.1 Speaker

Volume Settings (0-9)

Disable Volume Control Dial ☐

SIP Volume:

Multicast Volume:

Ring Volume:

Sensor Volume:

Push to Talk Volume:

Volume Boost:

Microphone Settings (0-9)

Microphone:

Microphone Gain:

Push to Talk Microphone Gain:

Microphone Boost 1 (+20dB):

Microphone Boost 2 (+20dB):

DTMF Settings

Require Security Code: ☐

Security Code:

Enable DTMF Push to Talk: ☐

Monitor DTMF Toggle Key:

Enable Stored Message Playback: ☐

Power Settings

802.3AT Mode:

Force 802.3AT Mode (NOT recommended): ☐

Auxiliary Power Supply: ☐

Time Settings

Set Time with NTP server on boot: ☐

NTP Server:

Posix Timezone String (see manual):

Periodically sync time with server: ☐

Time update period (in hours):

Current Time:

Set Time Manually:

Relay Settings

Activate Relay with DTMF code: ☒

Relay Pulse Code:

Relay Pulse Duration (in seconds):

Relay Activation Code:

Relay Deactivation Code:

Activate Relay During Ring: ☐

Activate Relay During Night Ring: ☐

Activate Relay While Call Active: ☐

Clock Settings

Clock Kit:

Misc Settings

Device Name:

Auto-Answer Incoming Calls: ☒

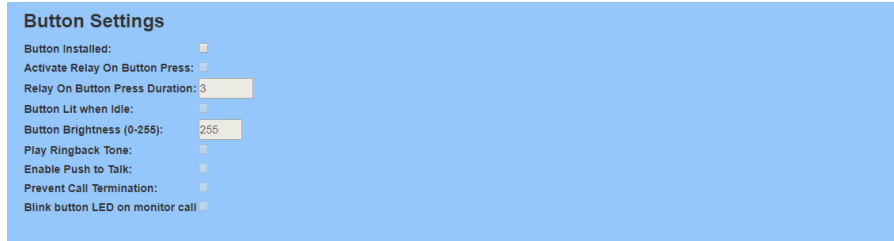
Beep on Init: ☒

Beep on Page: ☒

Disable HTTPS (NOT recommended): ☐

Dual Speakers: ☐

RGB Strobe:



Button Settings

Button Installed: ☐

Activate Relay On Button Press: ☐

Relay On Button Press Duration: 3

Button Lit when Idle: ☐

Button Brightness (0-255): 255

Play Ringback Tone: ☐

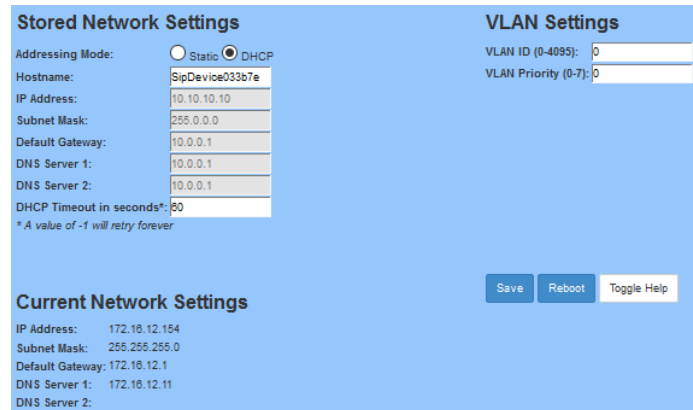
Enable Push to Talk: ☐

Prevent Call Termination: ☐

Blink button LED on monitor call: ☐

7.4 Network Configuration

Addressing Node Select either DHCP IP Addressing or Static Addressing by marking the appropriate radio button. DHCP Addressing mode is enabled on default and the device will attempt to resolve network addressing with the local DHCP server upon boot. If DHCP Addressing fails, the device will revert to the last known IP address or the factory default address if no prior DHCP lease was established.



Stored Network Settings

Addressing Mode: ☐ Static ☒ DHCP

Hostname: SipDevice033b7e

IP Address: 10.10.10.10

Subnet Mask: 255.0.0.0

Default Gateway: 10.0.0.1

DNS Server 1: 10.0.0.1

DNS Server 2: 10.0.0.1

DHCP Timeout in seconds: 60

* A value of -1 will retry forever

VLAN Settings

VLAN ID (0-4095): 0

VLAN Priority (0-7): 0

Current Network Settings

IP Address: 172.16.12.154

Subnet Mask: 255.255.255.0

Default Gateway: 172.16.12.1

DNS Server 1: 172.16.12.11

DNS Server 2:

Buttons: Save, Reboot, Toggle Help

Hostname This is the hostname provided by the DHCP server. See the DHCP/ DNS server documentation for more information. Enter up to 64 characters.

IP Address Enter the Static IPv4 network address in dotted decimal notation.

Subnet Mask Enter the Subnet Mask in dotted decimal notation.

Default Gateway Enter the Default Gateway IPv4 address in dotted decimal notation.

DNS Server 1 Enter the primary DNS Server IPv4 address in dotted decimal notation.

DNS Server 2 Enter the secondary DNS Server IPv4 address in dotted decimal notation.

DHCP Timeout in seconds Specify the desired time-out duration (in seconds) that the device will wait for a response from the DHCP server before reverting to the stored static IP address. The stored static IP address may be the last

Technical Publications

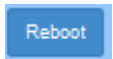
known IP address or the factory default address if no prior DHCP lease was established. Enter up to 8 characters. A value of -1 will retry forever.



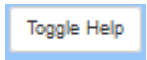
Click on the Save button to save your configuration settings.



Note: You need to reboot for changes to take effect.



Click on the Reboot button to reboot the system.



Click on the Toggle Help button to see a short description of some of the web page items. First click on the Toggle Help button, and you will see a question mark (?) appear next to some of the web page items. Move the mouse pointer to hover over a question mark to see a short description of a specific web page item.

The screenshot displays a network configuration interface with three main sections:

- Stored Network Settings:** Includes fields for Addressing Mode (Static/DHCP), Hostname (SipDevice033b7e), IP Address (10.10.10.10), Subnet Mask (255.0.0.0), Default Gateway (10.0.0.1), DNS Server 1 (10.0.0.1), DNS Server 2 (10.0.0.1), and DHCP Timeout in seconds (00). A note states: "* A value of -1 will retry forever".
- VLAN Settings:** Includes fields for VLAN ID (0-4095) and VLAN Priority (0-7).
- Current Network Settings:** Displays the current configuration: IP Address (172.16.12.154), Subnet Mask (255.255.255.0), Default Gateway (172.16.12.1), DNS Server 1 (172.16.12.11), and DNS Server 2.

At the bottom right, there are buttons for Save, Reboot, and Toggle Help.

7.5 SIP Configuration

SIP configuration screen is used to configure the SIP registration parameters used by the CyberData SIP Ceiling Speaker to register with 8x8 for paging purposes. The SIP User ID and Authentication ID are the same values which is the GUN ID provided by your 8x8 for the device and assigned to the user created previously. Authentication Password is provided by your 8x8 Engineer.

Enable SIP Operation:

Checked

Register with a SIP Server:

Checked

Use Cisco SRST: Unchecked

SIP Server: unsbc.8x8.com

Backup SIP Server 1: Not

Used

Technical Publications

Backup SIP Server 2: Not Used

Remote SIP Port: 5299

Local SIP Port: 5060

Outbound Proxy: must be left blank.

Outbound Proxy Port: 0

SIP User ID: the GUN ID provided by your 8x8 Engineer.

Authentication ID: Same as User ID.

Authentication Password: the SIP Proxy Password provided by your 8x8 engineer.

Monitor User ID: Leave as default.

Monitor Authenticate ID: Leave as default.

Monitor Authenticate Password: Leave as default.

Button Dial Out Extension: If equipped, the extension the device will dial.

Button Extension ID: If equipped, the extension the device will show the call coming from.

Re-registration Interval: 360

Unregister on Reboot:

Unchecked



SIP Settings		Nightringer Settings	
Enable SIP operation:	<input checked="" type="checkbox"/>	Enable Nightringer:	<input checked="" type="checkbox"/>
Register with a SIP Server:	<input checked="" type="checkbox"/>	SIP Server:	unsrc.8x8.com
Use Cisco SRST:	<input type="checkbox"/>	Remote SIP Port:	5299
Primary SIP Server:	unsrc.8x8.com	Local SIP Port:	5061
Primary SIP User ID:		Outbound Proxy:	
Primary SIP Auth ID:		Outbound Proxy Port:	0
Primary SIP Auth Password:		User ID:	
Backup SIP Server 1:		Authenticate ID:	
Backup SIP User ID 1:		Authenticate Password:	
Backup SIP Auth ID 1:		Re-registration Interval (in seconds):	360
Backup SIP Auth Password 1:			
Backup SIP Server 2:			
Backup SIP User ID 2:			
Backup SIP Auth ID 2:			
Backup SIP Auth Password 2:			
Remote SIP Port:	5299		
Local SIP Port:	5060		
Outbound Proxy:			
Outbound Proxy Port:	0		
Monitor User ID:	200		
Monitor Authenticate ID:	200		
Monitor Authenticate Password:	*****		
Disable rport Discovery:	<input type="checkbox"/>		
Buffer SIP Calls:	<input type="checkbox"/>		
Re-registration Interval (in seconds):	360		
Unregister on Boot:	<input type="checkbox"/>		
Keep Alive Period:	10000		

RTP Settings	
RTP Port (even):	10500
Jitter Buffer:	50

Call Disconnection	
Terminate Call after delay:	0

Codec Selection	
Force Selected Codec:	<input type="checkbox"/>
Codec:	PCMU (G.711, u-law) ▼

Button Settings	
Dial Out Extension:	204
Extension ID:	id204

Note: if checked will create an issue on registration, and the device will fail to register.

Buffer SIP Calls: Optional, if checked the CyberData SIP Server will buffer the page, and once the call is disconnected, it will make the page.

RTP Port (even): 10500

Jitter Buffer: 50

Call Disconnection

Terminate call after delay: 0

Codec Selection Force

Selected Codec: Unchecked

Codec: PCMU (G.711, u-law)

7.6 Nightringer Configuration

Nightringer configuration screen is used to configure the SIP registration parameters used by the CyberData SIP Ceiling Speaker to register with 8x8 for Night Bell or Nightringer purposes. The SIP User ID and Authentication ID are the same values which is the GUN ID provided by your 8x8 for the device and assigned to the user. Authentication Password is provided by your 8x8 Engineer.

Technical Publications

Enable Nightringer: Checked

SIP Server: unsbc.8x8.com

Remote SIP Port: 5299

Local SIP Port: 5061, must be Port 5061.

User ID: the GUN ID provided by your 8x8 engineer.

Authentication ID: Same as User ID.

Authentication Password:

The SIP Proxy Password for the Device as provided by your 8x8 engineer.

Re-registration Interval: 360

Relay rings to multicast: If you wish all multicast devices to receive the ringer page, CHECK this check box.

Multicast Address: the IP Address to send the nightringer page to.

Multicast Port: The Port Address to send the nightringer page to.

Save

Click on the Save button to save your configuration settings.



Note: You need to reboot for changes to take effect.

SIP Settings	Nightringer Settings
Enable SIP operation: <input checked="" type="checkbox"/>	Enable Nightringer: <input checked="" type="checkbox"/>
Register with a SIP Server: <input checked="" type="checkbox"/>	SIP Server: unsbc.8x8.com
Use Cisco SRST: <input type="checkbox"/>	Remote SIP Port: 5299
Primary SIP Server: unsbc.8x8.com	Local SIP Port: 5061
Primary SIP User ID: [REDACTED]	Outbound Proxy:
Primary SIP Auth ID: [REDACTED]	Outbound Proxy Port: 0
Primary SIP Auth Password: [REDACTED]	User ID: [REDACTED]
Backup SIP Server 1:	Authenticate ID: [REDACTED]
Backup SIP User ID 1:	Authenticate Password: [REDACTED]
Backup SIP Auth ID 1:	Re-registration Interval (in seconds): 360
Backup SIP Auth Password 1:	
Backup SIP Server 2:	
Backup SIP User ID 2:	
Backup SIP Auth ID 2:	
Backup SIP Auth Password 2:	
Remote SIP Port: 5299	
Local SIP Port: 5060	
Outbound Proxy:	
Outbound Proxy Port: 0	
Monitor User ID: 200	
Monitor Authenticate ID: 200	
Monitor Authenticate Password: [REDACTED]	
Disable rport Discovery: <input type="checkbox"/>	
Buffer SIP Calls: <input type="checkbox"/>	
Re-registration Interval (in seconds): 360	
Unregister on Boot: <input type="checkbox"/>	
Keep Alive Period: 10000	

RTP Settings
RTP Port (even): 10500
Jitter Buffer: 50

Call Disconnection
Terminate Call after delay: 0

Codec Selection
Force Selected Codec: <input type="checkbox"/>
Codec: PCMU (G.711, u-law) ▼

Button Settings
Dial Out Extension: 204
Extension ID: id204

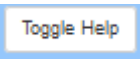
Save

Reboot

Toggle Help



Click on the Reboot button to reboot the system.



Click on the Toggle Help button to see a short description of some of the web page items. First click on the Toggle Help button, and you will see a question mark (?) appear next to some of the web page items. Move the mouse pointer to hover over a question mark to see a short description of a specific web page item.

7.7 Multicast (Paging Groups)

A multicast group is a way of assigning multicast IP addresses and port numbers when configuring CyberData multicast paging. To assign a multicast address, you must first configure the Yealink, Polycom and CyberData VoIP speakers that you want to put into a paging zone by entering a particular multicast address and port number combination in the Yealink, Polycom web interface, and web configuration for CyberData VoIP speakers. Each zone must have a unique IP address and Port number. The Port number must be even. The Multicast Configuration page consists of four pages. Each page must be saved independently.



To edit a paging group, click on the Edit button for the group you wish to edit. In the popup windows enter your configuration options for that paging group.

Polycom will use a Default IP of 224.0.1.116 and a port of 5001 for its paging functions. 8x8 recommends that when using Polycom phones to set Priority 0 to be your Polycom Paging group by entering the IP of 224.0.1.116 and Port 5001 into Priority 0.

Technical Publications

Address: Enter the IP address of the PGROUP.

- **Note:** To disable a relay on a group, use an IP address of 0.0.0.0.

Port: Enter the port number of the PGROUP.

- **Note:** The port range can be from 2000 to 65534 and must be even.

Name: Enter a name for the PGROUP.

Buffer: Should this be buffered before played.

Beep: should a beep be played before page.

Relay: should the relay be engaged with this page.

Polycom Default Channel: 1

Polycom Priority Channel: 24

Polycom Emergency Channel: 25



Note: You need to reboot for changes to take effect.

Reboot

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Multicast Settings

Enable Multicast Operation: ☒

Priority	Address	Port	Name	Buffer	Beep	Relay
9	234.2.1.10	2018	Emergency	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
8	234.2.1.9	2016	MG8	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
7	234.2.1.8	2014	MG7	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
6	234.2.1.7	2012	MG6	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
5	234.2.1.6	2010	MG5	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
4	234.2.1.5	2008	MG4	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
3	234.2.1.4	2006	MG3	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
2	234.2.1.3	2004	MG2	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
1	234.2.1.2	2002	MG1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
0	224.0.1.116	5001	Polycom Default	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Polycom Default Channel

Polycom Priority Channel

Polycom Emergency Channel

SIP calls are considered priority 4.5

Port range can be from 2000-65535

Priority 9 is the highest and 0 is the lowest

A higher priority audio stream will always supersede a lower one

** You need to reboot for changes to take effect*

Save

Reboot

Toggle Help