



Configuring CyberData Devices for Intermedia Hosted PBX

This procedure was written by:



INTERMEDIA

The Business Cloud™

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Revision Information

Revision 931097B was released on July 28, 2015, and has the following changes:

- Adds [Section 4.0, "Configuring the SIP Paging Adapter"](#)
- Adds [Section 7.0, "Configuring the SIP-enabled IP Outdoor Keypad Intercom"](#)
- Adds [Section 8.0, "Configuring the Outdoor Intercom to Auto Dial Numbers and Extensions"](#)

Revision 931097C was released on January 7, 2020 and has the following changes:

- Updated CyberData device screenshots
- Adds [Section 4.5 Disable DTMF Menu](#)
- Adds [Section 5.5 Disable DTMF Menu](#)

Contents

Contents

Revision Information	3
1.0 About Intermedia Bring Your Own Phone (BYOP)	1
2.1 Obtaining SIP Credentials	2
3.1 Creating Hosted PBX Paging Groups	4
4.0 Configuring the SIP Paging Adapter	7
4.1 About the SIP Paging Adapter	7
4.2 Accessing the SIP Paging Adapter	7
4.3 Verify Firmware Version	8
4.4 Configure the SIP Parameters	9
4.5 Disable DTMF Menu	11
5.0 Configuring the SIP Paging Server	12
5.1 About the SIP Paging Server	12
5.2 Accessing the Paging Server	13
5.3 Verify Firmware Version	14
5.4 Configure the SIP Parameters	14
5.5 Disable the DTMF Menu	16
6.0 Configuring the SIP Speaker	17
6.1 About the SIP Speaker	17
6.2 Accessing the SIP Speaker	17
6.3 Verify Firmware Version	18
6.4 Configure the SIP Parameters	19
7.0 Configuring the SIP-enabled IP Outdoor Keypad Intercom	22
7.1 About the SIP-enabled IP Outdoor Keypad Intercom	22
7.2 Accessing the SIP-enabled IP Outdoor Keypad Intercom	23
7.3 Verify Firmware Version	24
7.4 Configure the SIP Parameters	25
8.1 Configuring the Outdoor Keypad Intercom to Auto Dial Numbers and Extensions	28
8.1 To Program a Phone Number or Extension	28
8.2 To Program an Auto Attendant Number	29

1.0 About Intermedia Bring Your Own Phone (BYOP)

The Intermedia Bring Your Own Phone (BYOP) feature provides the ability to supply your own SIP devices for use with Intermedia's Hosted PBX service. This feature may also be utilized to successfully configure SIP phones, SIP softphones and other SIP devices, such as paging or intercom units, for use with Intermedia Hosted PBX service.

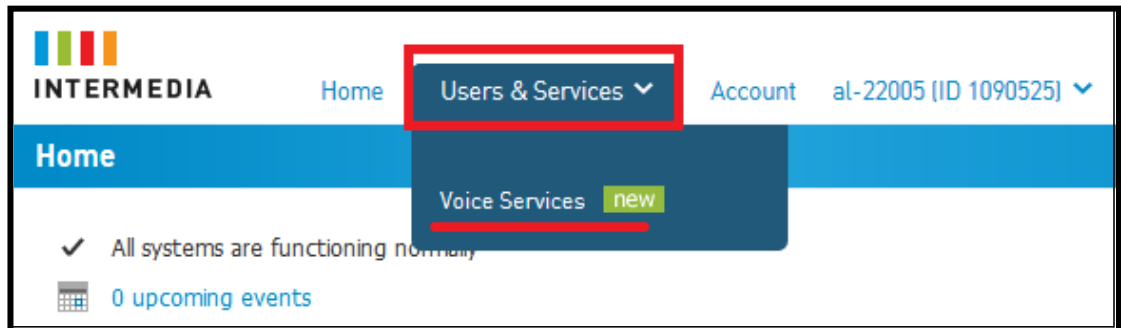
The greatest amount of freedom and access to device configurations are attained with Intermedia's AnyPhone BYOP, which is available to any SIP enabled telecommunications device that is not listed on Intermedia's Approved Phone and Equipment list below:

Cisco SPA112	Polycom IP331	Polycom IP7000
Cisco SPA232D	Polycom IP335	Polycom VVX300
Cisco SPA302D	Polycom IP550	Polycom VVX310
Cisco SPA303	Polycom IP560	Polycom VVX400
Cisco SPA504G	Polycom IP650	Polycom VVX410
Cisco SPA525G2	Polycom IP5000	Polycom VVX500
	Polycom IP6000	Polycom VVX600

2.1 Obtaining SIP Credentials

Once your account has been created with an AnyPhone BYOP device, or your device has been added to your existing account, you may log into the HostPilot End User Control Panel and retrieve your device's SIP credentials. These credentials are required in order to make your CyberData device register with your Hosted PBX service. To find your credentials:

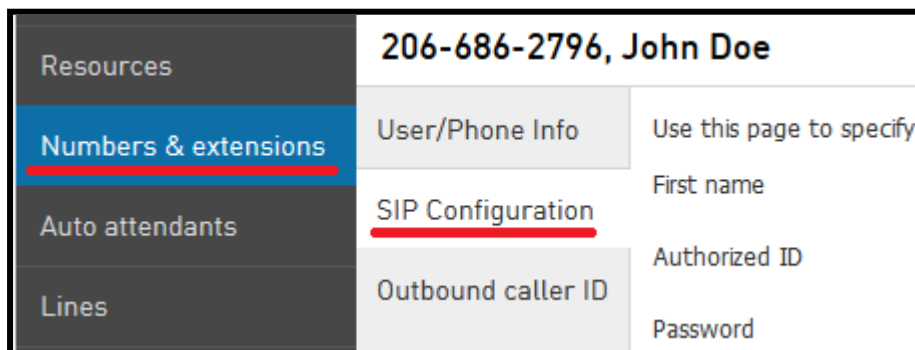
1. Log into HostPilot at <https://exchange.intermedia.net/asp/Login.aspx> using the credentials you received via email
2. Under the **Users & Services** section, click on **Voice Services**.



3. Select the **Numbers & Extensions** tab and locate your SIP device by looking for a device type, Extension number or specific DID. A paging device will be provisioned with a device Type of 'Generic SIP Paging (AnyPhone)' making it easier to spot. Once located, click on the phone number for the device.

206-686-2397	112	Joe Blow	Generic SIP Paging (AnyPhone)
206-577-8736	103	Jane Doe	Polycom VVX410 4-Line Phone
206-686-2796	111	John Doe	Generic SIP Phone (AnyPhone)
206-577-8738	106	Mark Etting	Cisco SPA504G 4-Line Phone

4. Click on the **SIP Configuration** tab to locate your SIP credentials.



Once you have located your device’s credentials, you will need to manually transcribe them into the administrative web interface for your CyberData device.

User/Phone Info

SIP Configuration

Outbound caller ID

Failover routing

911 address

Call history

Groups

This page provides the SIP credentials to allow you to connect your SIP device to your Hosted PBX system. For more information, please click the appropriate link: [Bria 4 SoftPhone](#), [CyberData Paging Devices](#), [Other SIP Devices](#).

SIP User Name ?

161

SIP Authorization ID

900193965

SIP Password

XXXXXXXXXX

SIP Domain

uc70.telecomsvc.com:6060

Outbound Proxy

uc70.telecomsvc.com:6060

Type ?

Generic

Preferred Codec ?

☐ G.729
 ☒ G.711

Secondary Codec ?

☒ G.729
 ☐ G.711
 ☐ None

Paging ?

☒ Enabled

Save changes

The CyberData preferred codec is G.711. Intermedia's network supports G.711 uLaw, and our recommendation is to set the **Preferred Codec** on your SIP Configuration tab to G.711 to match the CyberData paging device preference. This setting will help ensure higher quality and more legible pages through your paging devices.

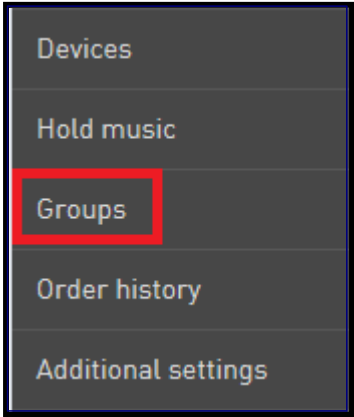
Note After saving a codec change on the SIP Configuration tab, it is very important to power cycle your CyberData device.

The **Paging Enabled** checkbox controls whether this paging device may be added to a Hosted PBX Paging Group. Hosted PBX has a built-in paging feature which sends pages to devices via direct SIP INVITE messages and utilizes auto-answer paging options. CyberData paging speakers may work in this capacity and if you would like to include them within Hosted PBX Paging Groups, check this option.

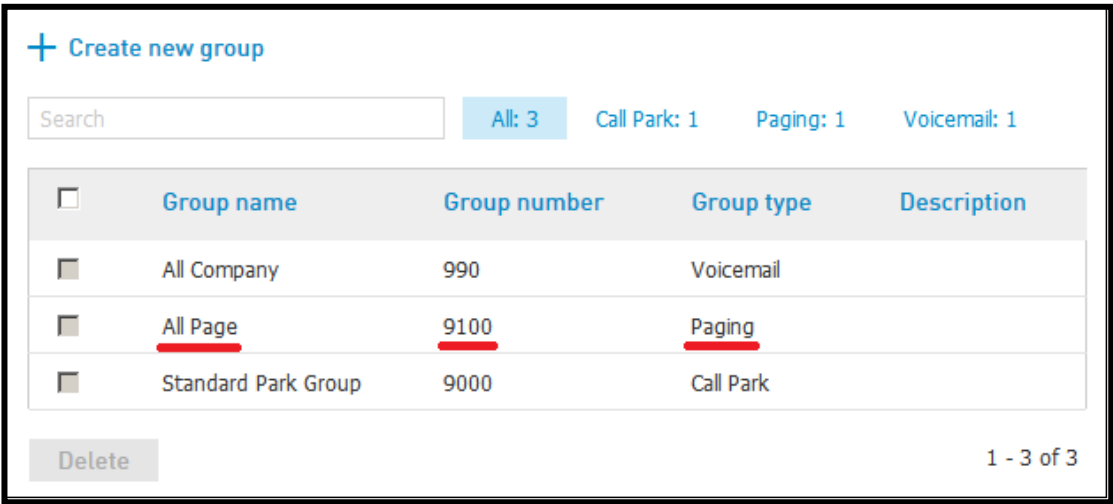
3.1 Creating Hosted PBX Paging Groups

If you wish to add a CyberData paging device into a Hosted PBX paging group, please perform the following within your HostPilot account:

1. Click on the **Groups** section.



2. The **Groups** section will open and display all current Groups of all types. Your Hosted PBX service automatically comes with one paging group pre-configured, named **All Page**, which includes all of your devices. The group number is **9100**.



3. Dialing the **All Page** group number of **9100** from any of your phones will send a page to all of your provisioned devices that are currently powered on and registering with your Hosted PBX service.

- Your devices may be included within multiple Hosted PBX paging groups. To create a group, click on the **Create new group** button.

Note: the following default groups may not be deleted: All Company, All Page, and

+ Create new group

Search All: 3 Call Park: 1 Paging: 1

<input type="checkbox"/>	Group name	Group number	Group type
<input type="checkbox"/>	All Company	990	Voicemail
<input type="checkbox"/>	All Page	9100	Paging

- The new group dialog will open. Select the **Paging group** radio button, and provide a **Group Name**. When finished, click the **Create Group** button.

About group types **▼**

Group Type: ☒ Paging group
☐ Voicemail group
☐ Call Park group

Group Name:

Description:

Create Group

- Once the paging group has been created, you may now choose which phones and devices you wish to include in the group. Use the drop-down menu to choose those devices, and click the **Add** button for each device.

Add numbers to the group

Start typing user name or number ^ Add

Jane Doe, 103 (206-577-8736)

John Doe, 111 (206-686-2796)

Mark Etting, 106 (206-577-8738)

Select from numbers list

7. Each device automatically saves within the paging group when added. Once finished adding devices to your paging group, the group is ready to use.

<input type="checkbox"/>	Name	Phone Number	Extension
<input type="checkbox"/>	Jane Doe	206-577-8736	103
<input type="checkbox"/>	John Doe	206-686-2796	111

Exclude...1 - 2 of 2

8. Take note of your new group's Group Number. This may be found on the group's **General** tab. This Group Number is the 4-digit number you must dial from your phones to send a page to the devices in this page group.

General

Use this page if you want to assign a new Group Name or optional description.

Members

Group Type:

Paging

Group Name:

Shop Pages

Description:

Optional

Group Number:

9101

Save changes

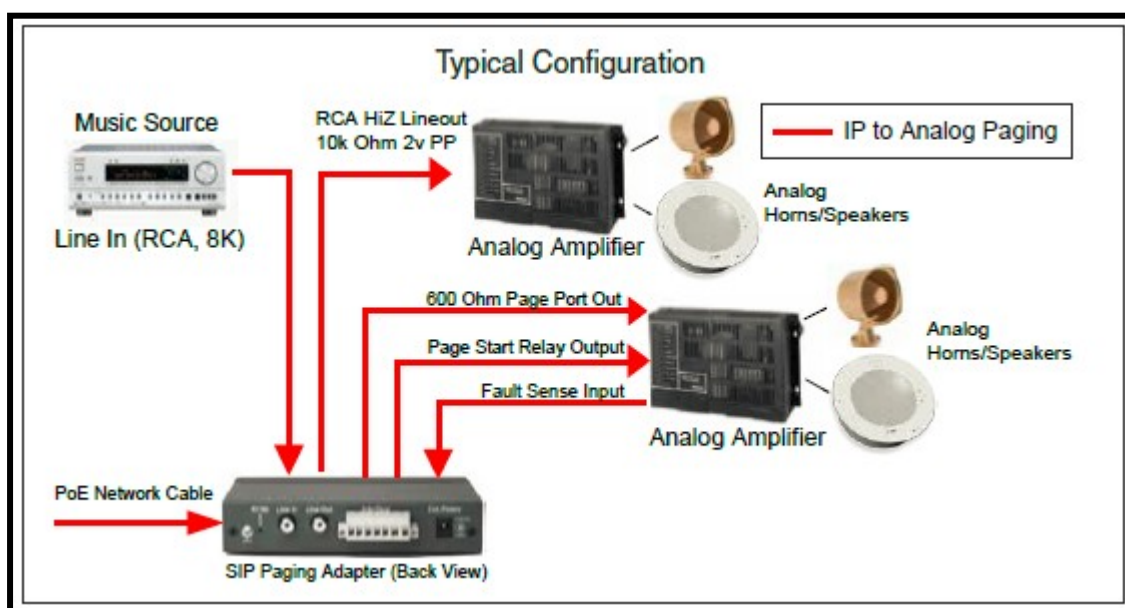
Delete Group

4.0 Configuring the SIP Paging Adapter

4.1 About the SIP Paging Adapter

The CyberData SIP Paging Adapter is a VoIP endpoint that interfaces analog paging systems with SIP and Multicast-based audio sources.

Please note: The SIP Paging Adapter is a Power over Ethernet (PoE) device and requires a PoE router or switch within your local network. If you do not have nor do not wish to obtain a PoE network device, you must purchase a PoE Power Injector along with your SIP Paging Adapter to provide it with electricity.

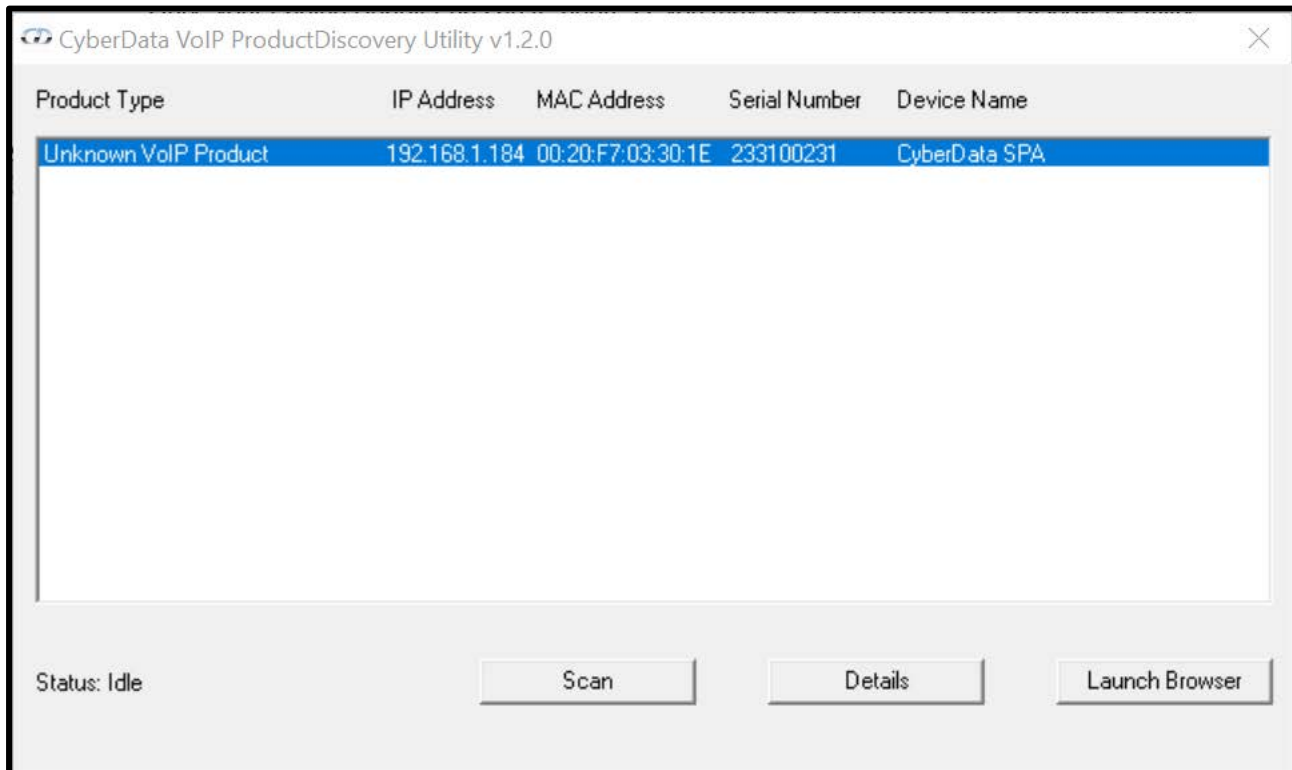


4.2 Accessing the SIP Paging Adapter

To begin setting up your SIP Paging Adapter, please ensure you have:

- Connected the Paging Adapter to a PoE port on your network
- Provided power to the device, via PoE or a PoE Power Injector from CyberData
- Provided an IP address to the Paging Adapter via DHCP

Once your Paging Adapter has an IP address, you may use CyberData's VoIP Discovery Utility which can be downloaded from CyberData's website. Run the utility and Scan for your Paging Adapter. Once located, highlight the device and click the Launch Browser button to log into the web user interface. Please refer to your CyberData documentation for the default login and password.



4.3 Verify Firmware Version

Before proceeding, it is recommended to verify your device is utilizing the latest firmware version. The firmware version may be read on the Home page of the user interface.

Current Status

Serial Number: 233100231

Mac Address: 00:20:f7:03:30:1e

Firmware Version: v11.8.0

Please check the CyberData website for the latest firmware version. Download and update the firmware via the Update Firmware section in the user interface if necessary.

4.4 Configure the SIP Parameters

Once your firmware version is verified, proceed to the SIP Config section to begin entering your Intermedia Hosted PBX SIP credentials. Please utilize this reference table below to assist with terminology differences.

Intermedia Credential Name	CyberData Credential Name
SIP Authorization ID	SIP User ID
SIP Authorization ID	Authenticate ID
SIP Password	Authenticate Password
SIP Domain	SIP Server

In addition to the SIP credentials provided by Intermedia, we recommend the following additional settings.

SIP Configuration Tab	Setting
Remote SIP Port	6060
Local SIP Port	6000 + SIP User Name **
Re-registration Interval	30

** Some routers may have difficulty handling requests that return to the same IP address and Port for multiple SIP devices. We recommend setting each SIP device with a unique Local SIP Port, and our preferred method is to match the value of your SIP User Name supplied by Intermedia's Hosted PBX. I.E. your SIP User Name is 112, your Local SIP Port becomes $6000 + 112 = 6112$.

When complete, click the Save button. Your SIP Config should be similar to the following:

CyberData Paging Adapter

SIP Settings

Enable SIP operation: ☒

SIP Transport Protocol: UDP ▾

TLS Version: 1.2 only (recommended) ▾

Verify Server Certificate: ☐

Register with a SIP Server: ☒

Use Cisco SRST: ☐

Primary SIP Server: uc70.telecomsvc.com

Primary SIP User ID: 900193964

Primary SIP Auth ID: 900193964

Primary SIP Auth Password: *****

Backup SIP Server 1:

Backup SIP User ID 1:

Backup SIP Auth ID 1:

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: 6060

Local SIP Port: 6159

Outbound Proxy:

Outbound Proxy Port: 0

Disable rport Discovery: ☐

Buffer SIP Calls: ☐

Re-registration Interval (in seconds): 30

Unregister on Boot: ☐

Keep Alive Period: 0

Nightringer Settings

Enable Nightringer: ☐

SIP Server: 10.0.0.253

Remote SIP Port: 5060

Local SIP Port: 5061

Outbound Proxy:

Outbound Proxy Port: 0

User ID: 241

Authenticate ID: 241

Authenticate Password: *****

Re-registration Interval (in seconds): 360

Call Disconnection

Terminate Call after delay: 0

Codec Selection

Force Selected Codec: ☐

Codec: PCMU (G.711, u-law) ▾

RTP Settings

RTP Port: 10500

(even):

Jitter Buffer: 50

Once verified, click the Reboot button. Your SIP Paging Adapter should reach out to Intermedia's network and achieve successful registration when it powers up. Registration status may be confirmed on the Home tab of the user interface.

Primary SIP Server: (Registered with SIP Server)

Backup Server 1: (NOT Registered with SIP Server)

Backup Server 2: (NOT Registered with SIP Server)

Once registered, your SIP Paging Adapter will receive inbound calls placed to the 10-digit number, or 3-digit extension assigned by your Hosted PBX service.

4.5 Disable DTMF Menu

When using the paging adapter in a paging group the DTMF menu to play stored messages will need to be disabled. If not disabled, the unit will not play audio when a page is received. Check the box “Bypass DTMF Menus (Go straight to page)”. Then save and reboot for the setting to take effect.

CyberData

Paging Adapter

Line-in Settings

Enable Line-in to Line-out Loopback

Clock Settings

Set Time with NTP server on boot

NTP Server

north-america.pool.ntp.org

Posix Timezone String (see manual)

PST8PDT,M3.2.0/2:00:00,M11.1.0

Periodically sync time with server

Time update period (in hours)

24

Current Time

18:21:48

Misc Settings

Device Name

CyberData SPA

Beep on Init

Beep on Page

Disable HTTPS (NOT recommended)

Relay Settings

Activate Relay on Local Audio

DTMF Settings

DTMF Duration

500

Bypass DTMF Menus (Go straight to page)

Send pre-configured DTMF for Analog Zone

Zone

Manual DTMF Entry for Analog Zone

Require Security Code

Security Code

Test Audio

Test Relay

Save

Reboot

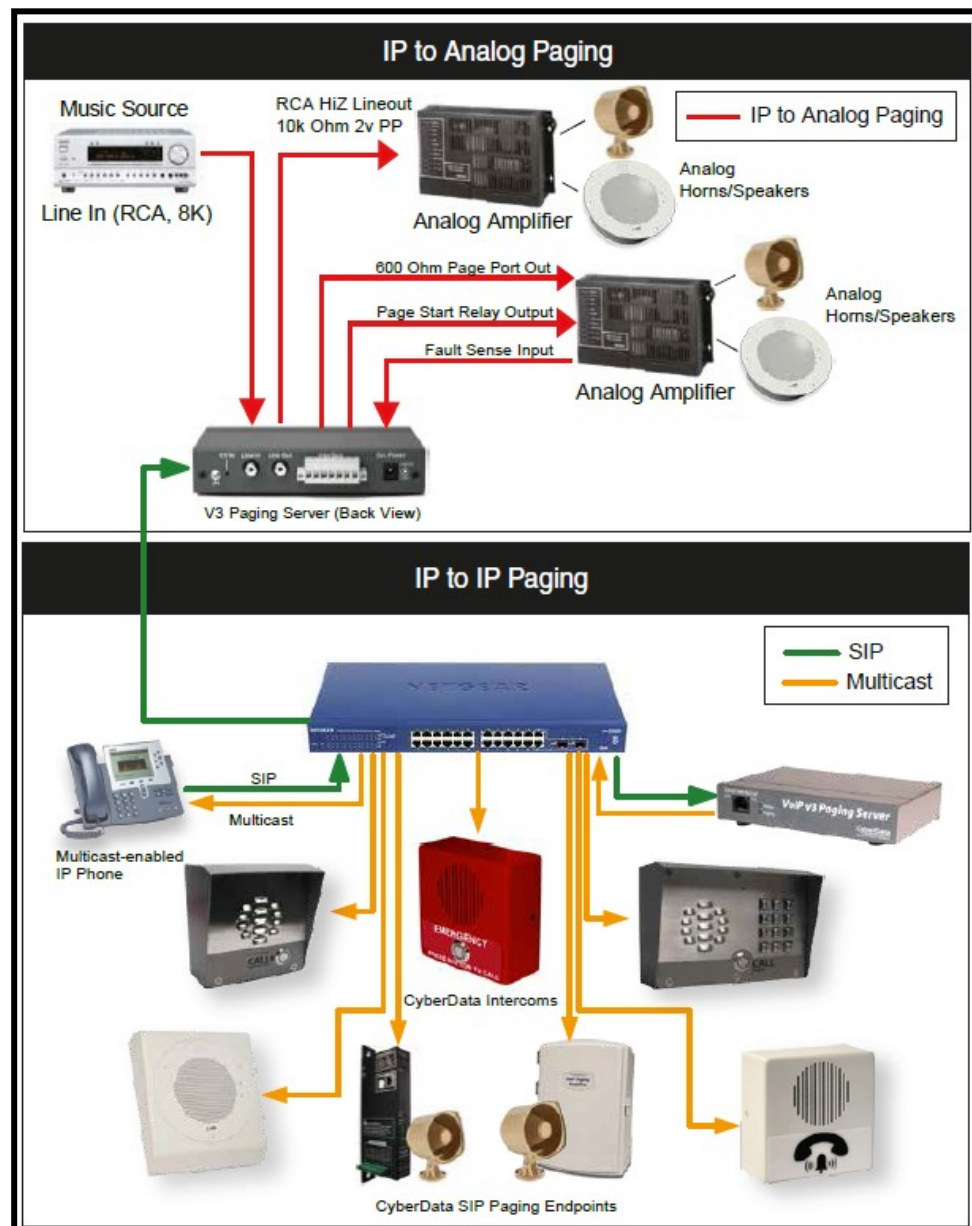
Toggle Help

5.0 Configuring the SIP Paging Server

5.1 About the SIP Paging Server

The CyberData SIP Paging Server enables users through a single SIP phone extension, to access multiple zones for paging in a VoIP network and to connect to legacy analog overhead paging systems. Configuration is achieved via an intuitive web-based graphical user interface. The Paging Server may be connected via analog or IP, per the diagram below.

Note: The Paging Server is a Power over Ethernet (PoE) device and requires a PoE router or switch within your local network. If you do not have or do not wish to obtain a PoE network device, you must purchase a PoE Power Injector along with your Paging Server to provide it with electricity.

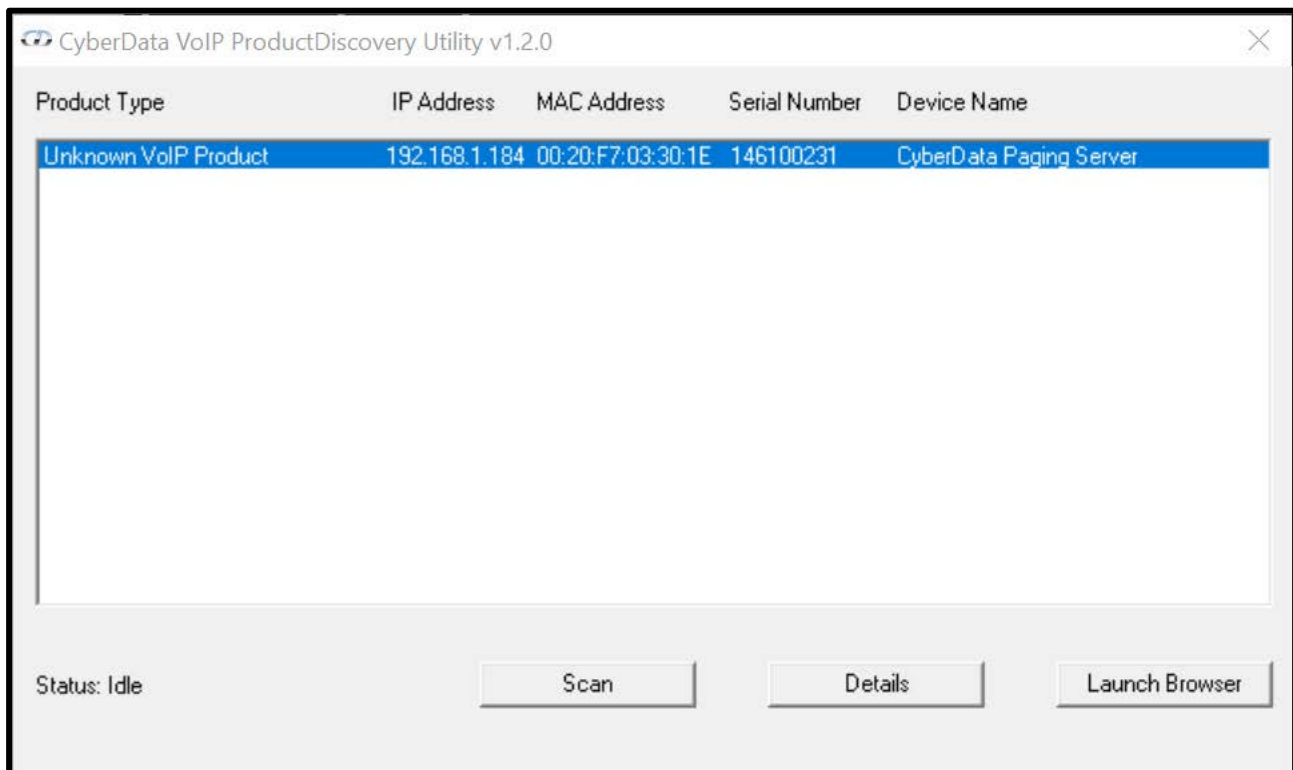


5.2 Accessing the Paging Server

To begin setting up your Paging Server, please ensure you have:

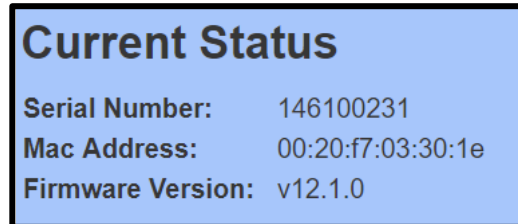
- Connected the paging server to a PoE port on your network
- Provided power to the device, via PoE or a PoE Power Injector from CyberData
- Provided an IP address to the Paging Server via DHCP

Once your Paging Server has an IP address, you may use CyberData's VoIP Discovery Utility which can be downloaded from CyberData's website. Run the utility and Scan for your Paging Server. Once located, highlight the device and click the Launch Browser button to log into the web user interface. Please refer to your CyberData documentation for the default login and password.



5.3 Verify Firmware Version

Before proceeding, it is recommended to verify your device is utilizing the latest firmware version. The firmware version may be read on the Home page of the user interface.



Please check the CyberData website for the latest firmware version. Download and update the firmware via the **Update Firmware** section in the user interface if necessary.

5.4 Configure the SIP Parameters

Once your firmware version is verified, proceed to the **SIP Config** section to begin entering your Intermedia Hosted PBX SIP credentials. Please utilize this reference table below to assist with terminology differences.

Intermedia Credential Name	CyberData Credential Name
SIP Authorization ID	SIP User ID
SIP Authorization ID	Authenticate ID
SIP Password	Authenticate Password
SIP Domain	SIP Server

In addition to the SIP credentials provided by Intermedia, we recommend the following additional settings.

SIP Configuration Tab	Setting
Remote SIP Port	6060
Local SIP Port	6000 + SIP User Name ^a
Re-registration Interval	30

- a. Some routers may have difficulty handling requests that return to the same IP address and Port for multiple SIP devices. We recommend setting each SIP device with a unique Local SIP Port, and our preferred method is to match the value of your SIP User Name supplied by Intermedia’s Hosted PBX. I.E. your SIP User Name is 112, your Local SIP Port becomes 6000 + 112 = 6112.

When complete, click the **Save** button. Your SIP configuration should be similar to the following:

CyberData v3.1 Paging Server

SIP Settings

Enable SIP operation: ☒

SIP Transport Protocol: UDP ▾

Register with a SIP Server: ☒

Use Cisco SRST: ☐

Primary SIP Server:

Primary SIP User ID:

Primary SIP Auth ID:

Primary SIP Auth Password:

Backup SIP Server 1:

Backup SIP User ID 1:

Backup SIP Auth ID 1:

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:

Local SIP Port:

Outbound Proxy:

Outbound Proxy Port:

Disable rport Discovery: ☐

Buffer SIP Calls: ☐

Re-registration Interval (in seconds):

Unregister on Boot: ☐

Keep Alive Period:

Nightringer Settings

Enable Nightringer: ☐

SIP Server:

Remote SIP Port:

Local SIP Port:

Outbound Proxy:

Outbound Proxy Port:

User ID:

Authenticate ID:

Authenticate Password:

Re-registration Interval (in seconds):

Relay rings to multicast: ☐

Multicast Address:

Multicast Port:

Call Disconnection

Terminate Call after delay:

Codec Selection

Force Selected Codec: ☐

Codec: PCMU (G.711, u-law) ▾

RTP Settings

RTP Port (even):

Jitter Buffer:

Save
Reboot
Toggle Help

Once verified, click the **Reboot** button. Your Paging Server should reach out to Intermedia's network and achieve successful registration when it powers up. Registration status may be confirmed on either the **Home** tab of the user interface.

Primary SIP Server:	(Registered with SIP Server)
Backup Server 1:	(NOT Registered with SIP Server)
Backup Server 2:	(NOT Registered with SIP Server)

Once registered, your Paging Server will receive inbound calls placed to the 10-digit number, or 3-digit extension assigned by your Hosted PBX service.

5.5 Disable the DTMF Menu

When using the paging server in a paging group the DTMF menu to play stored messages will need to be disabled. If not disabled, the unit will not play audio when a page is received. Check the box “Bypass DTMF Menus (Go straight to page)”. Then save and reboot for the setting to take effect.

CyberData Paging Adapter

Line-in Settings

Enable Line-in to Line-out Loopback

Clock Settings

Set Time with NTP server on boot

NTP Server: north-america.pool.ntp.org

Posix Timezone String (see manual): PST8PDT,M3.2.0/2:00:00,M11.1.0

Periodically sync time with server

Time update period (in hours): 24

Current Time: 18:21:48

Misc Settings

Device Name: CyberData SPA

Beep on Init:

Beep on Page:

Disable HTTPS (NOT recommended):

Relay Settings

Activate Relay on Local Audio:

DTMF Settings

DTMF Duration: 500

Bypass DTMF Menus (Go straight to page):

Send pre-configured DTMF for Analog Zone:

Zone:

Manual DTMF Entry for Analog Zone:

Require Security Code:

Security Code:

Test Audio

Test Relay

Save

Reboot

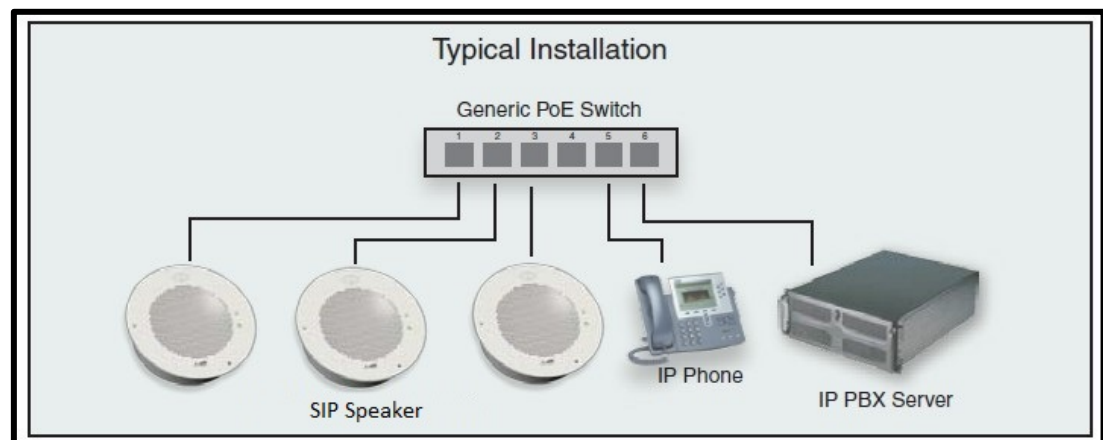
Toggle Help

6.0 Configuring the SIP Speaker

6.1 About the SIP Speaker

The CyberData SIP-enabled IP Speaker is a Power over Ethernet (PoE 802.3af) VoIP mass notification device with network-controlled speaker volume, simultaneous SIP and priority- based multicast streaming, and integrates with both the CyberData Wall Mount and the CyberData Wall Mount Clock Kit. In a non-SIP environment, the speaker is capable of broadcasting audio through multicast and can work in conjunction with the CyberData Paging Server.

Note: The SIP Speaker is a Power over Ethernet (PoE) device and requires a PoE switch within your local network. If you do not have or do not wish to obtain a PoE network device, you must purchase a PoE Power Injector along with your SIP Speaker to provide it with electricity.

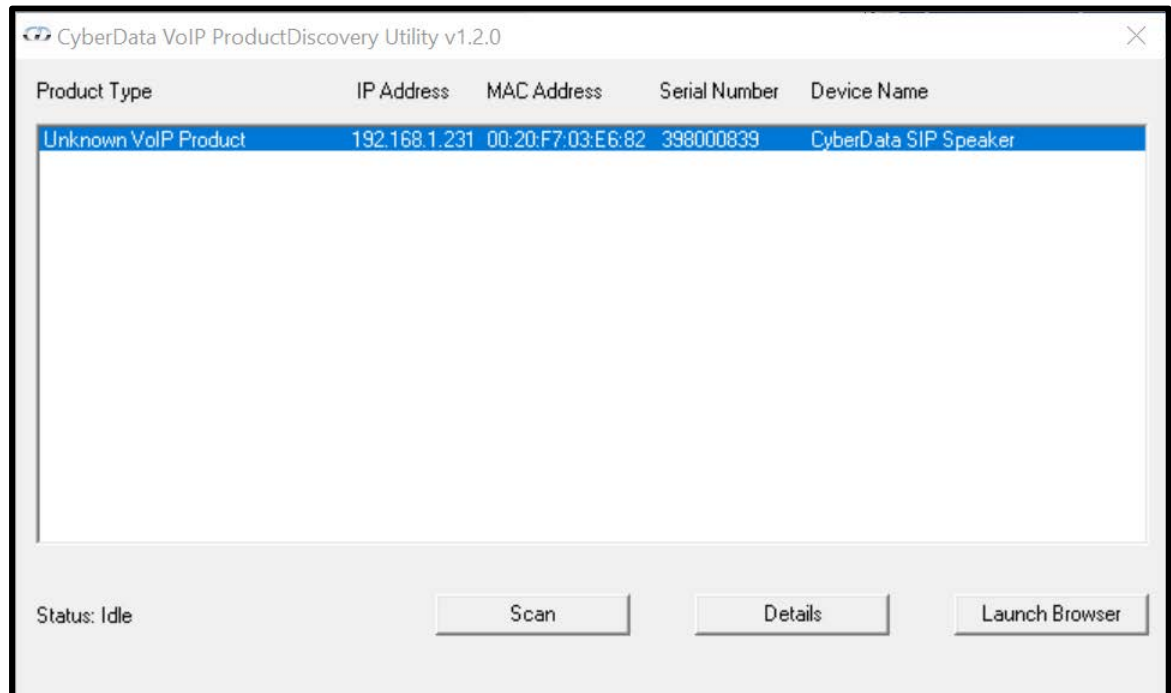


6.2 Accessing the SIP Speaker

To begin setting up your Ceiling Speaker, please ensure you have:

- Connected the speaker to a PoE port on your network
- Provided power to the speaker via PoE or an optional PoE Power Injector available from your CyberData supplier
- Provided an IP address to the speaker via DHCP

Once your speaker has an IP address, you may use CyberData's VoIP Discovery Utility which can be downloaded from CyberData's website. Run the utility and Scan for your speaker. Once located, highlight the device and click the **Launch Browser** button to log into the web user interface. Please refer to your CyberData documentation for the default login and password.



6.3 Verify Firmware Version

Before proceeding, it is recommended to verify your device is utilizing the latest firmware version. The firmware version may be read on the Home page of the user interface.

Current Status

Serial Number: 398000839

Mac Address: 00:20:f7:03:e6:82

Firmware Version: v12.0.2

Please check the CyberData website for the latest firmware version. Download and update the firmware via the **Update Firmware** section in the user interface if necessary.

6.4 Configure the SIP Parameters

Once your firmware version is verified, proceed to the **SIP Config** section to begin entering your Intermedia Hosted PBX SIP credentials. Please utilize this reference table below to assist with terminology differences.

Intermedia Credential Name	CyberData Credential Name
SIP Authorization ID	SIP User ID
SIP Authorization ID	Authenticate ID
SIP Password	Authenticate Password
SIP Domain	SIP Server

In addition to the SIP credentials provided by Intermedia, we recommend the following additional settings.

SIP Configuration Tab	Setting
Remote SIP Port	6060
Local SIP Port	6000 + SIP User Name ^a
Re-registration Interval	30

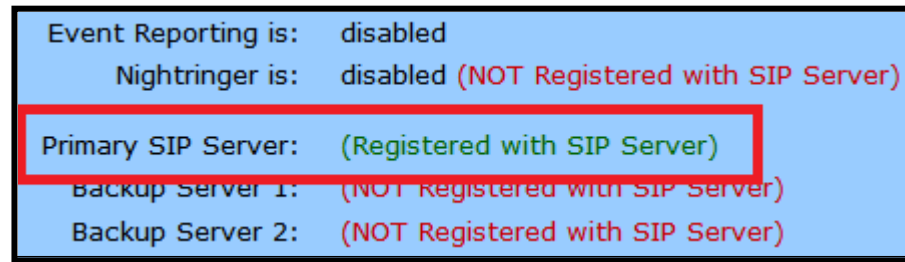
- a. Some routers may have difficulty handling requests that return to the same IP address and Port for multiple SIP devices. We recommend setting each SIP device with a unique Local SIP Port, and our preferred method is to match the value of your SIP User Name supplied by Intermedia's Hosted PBX. I.E. your SIP User Name is 112, your Local SIP Port becomes $6000 + 112 = 6112$.

When complete, click the **Save** button. Your SIP configuration should be similar to the following:

SIP Settings

Enable SIP operation:	<input checked="" type="checkbox"/>
SIP Transport Protocol:	UDP ▾
TLS Version:	1.2 only (recommended) ▾
Verify Server Certificate:	<input checked="" type="checkbox"/>
Register with a SIP Server:	<input checked="" type="checkbox"/>
Use Cisco SRST:	<input type="checkbox"/>
Primary SIP Server:	uc70.telecomsvc.com
Primary SIP User ID:	900012222
Primary SIP Auth ID:	900012222
Primary SIP Auth Password:
Backup SIP Server 1:	
Backup SIP User ID 1:	
Backup SIP Auth ID 1:	
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:	6060
Local SIP Port:	6112
Outbound Proxy:	
Outbound Proxy Port:	0
Monitor User ID:	
Monitor Authenticate ID:	
Monitor Authenticate Password:	
Disable rport Discovery:	<input type="checkbox"/>
Buffer SIP Calls:	<input type="checkbox"/>
Re-registration Interval (in seconds):	30
Unregister on Boot:	<input type="checkbox"/>
Keep Alive Period:	0

Once verified, click the **Reboot** button. Your SIP Speaker should reach out to Intermedia's network and achieve successful registration when it powers up. Registration status may be confirmed on the **Home** tab of the user interface.



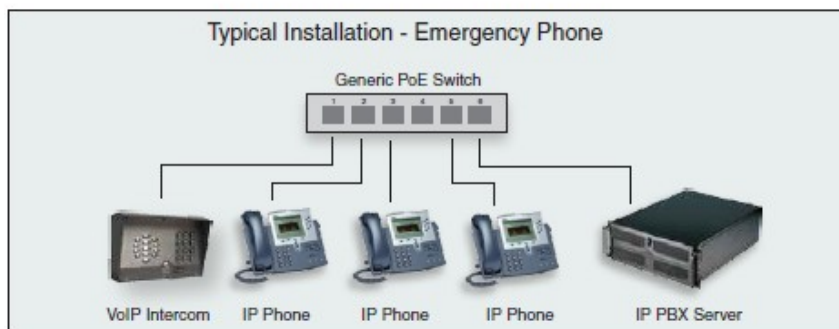
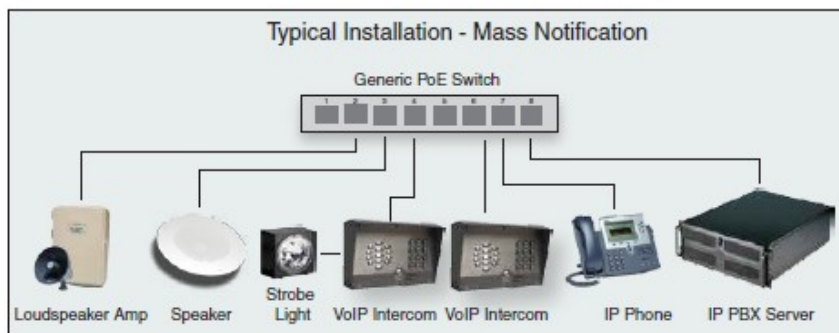
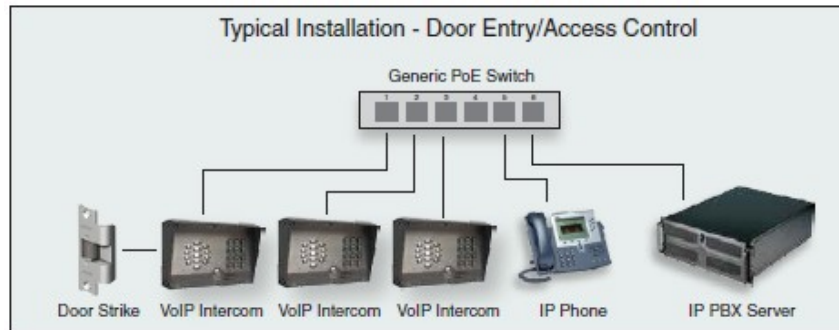
Once registered, your Ceiling Speaker will receive inbound calls placed to the 10-digit number, or 3-digit extension assigned by your Hosted PBX service.

7.0 Configuring the SIP-enabled IP Outdoor Keypad Intercom Intercom

7.1 About the SIP-enabled IP Outdoor Keypad Intercom

The SIP-enabled IP Outdoor Intercom with Keypad is a two-way communication and secure access device. Combining the versatility of a SIP based keypad intercom with the increased weather protection rating of IP 65, this device is perfect for settings such as commercial/ residential facilities, schools and universities, retail establishments, warehouse and manufacturing plants, parking garages and shipyards, and so much more.

Please note: The SIP-enabled IP Outdoor Intercom is a Power over Ethernet (PoE) device and requires a PoE switch within your local network. If you do not have or do not wish to obtain a PoE network device, you must purchase a PoE Power Injector along with your SIP-enabled IP Outdoor Intercom to provide it with electricity.

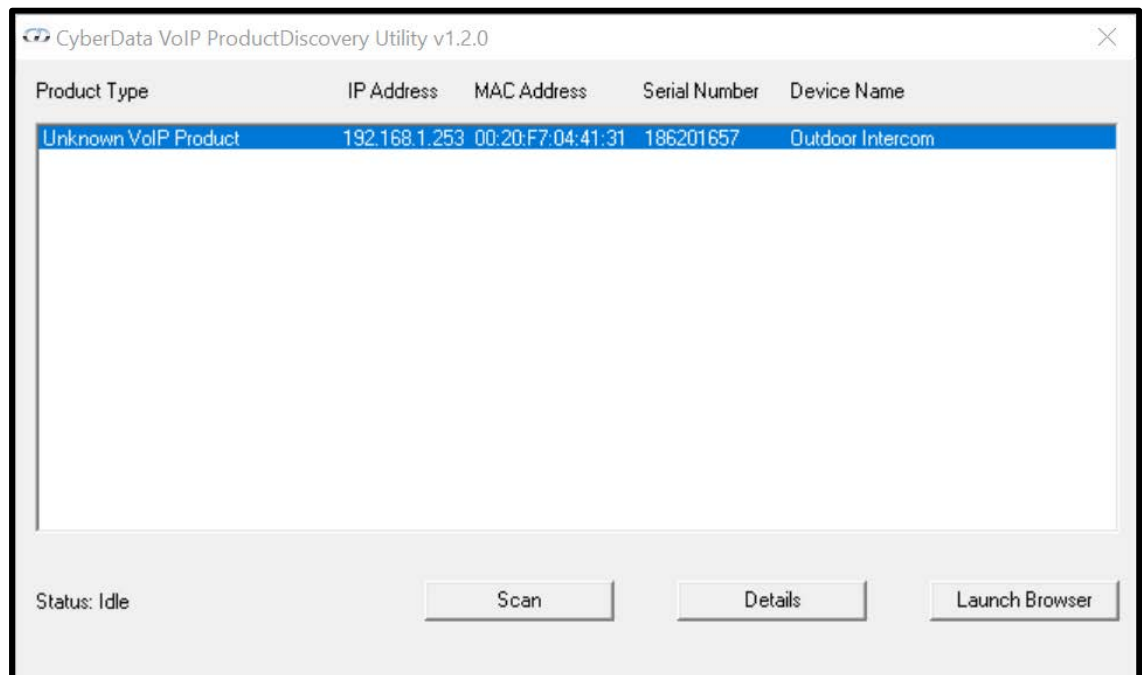


7.2 Accessing the SIP-enabled IP Outdoor Keypad Intercom

To begin setting up your SIP-enabled IP Outdoor Intercom, please ensure you have:

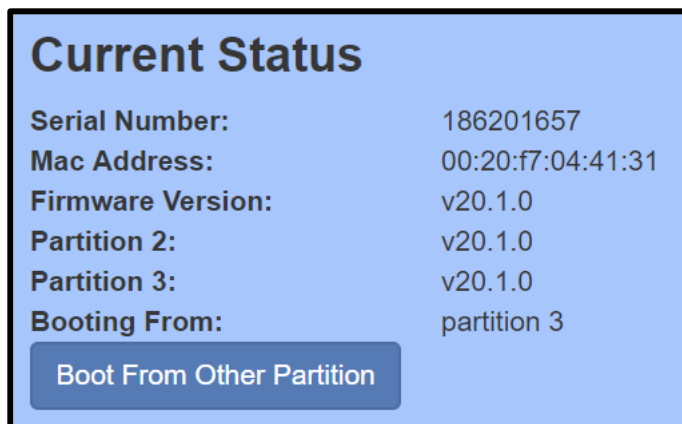
- Connected the Outdoor Intercom to a PoE port on your network
- Provided power to the device, via PoE or a PoE Power Injector from CyberData
- Provided an IP address to the Outdoor Intercom via DHCP

Once your Outdoor Intercom has an IP address, you may use CyberData's VoIP Discovery Utility which can be downloaded from CyberData's website. Run the utility and Scan for your Outdoor Intercom. Once located, highlight the device and click the Launch Browser button to log into the web user interface. Please refer to your CyberData documentation for the default login and password.



7.3 Verify Firmware Version

Before proceeding, it is recommended to verify your device is utilizing the latest firmware version. The firmware version may be read on the Home page of the user interface.



Please check the CyberData website for the latest firmware version. Download and update the firmware via the Update Firmware section in the user interface if necessary.

7.4 Configure the SIP Parameters

Once your firmware version is verified, proceed to the SIP Config section to begin entering your Intermedia Hosted PBX SIP credentials. Please utilize this reference table below to assist with terminology differences.

Intermedia Credential Name	CyberData Credential Name
SIP Authorization ID	Primary SIP User ID
SIP Authorization ID	Primary SIP Auth ID
SIP Password	Primary SIP Auth Password
SIP Domain	Primary SIP Server

In addition to the SIP credentials provided by Intermedia, we recommend the following additional settings.

SIP Configuration Tab	Setting
Remote SIP Port	6060
Local SIP Port	6000 + SIP User Name ^a
Re-registration Interval	30

- a. Some routers may have difficulty handling requests that return to the same IP address and Port for multiple SIP devices. We recommend setting each SIP device with a unique Local SIP Port, and our preferred method is to match the value of your SIP User Name supplied by Intermedia's Hosted PBX. I.E. your SIP User Name is 112, your Local SIP Port becomes 6000 + 112 = 6112.

When complete, click the Save button. Your SIP Config should be similar to the following:

CyberData Outdoor Intercom

SIP Settings

Enable SIP operation:

☒

Register with a SIP Server:

☒

Primary SIP Server:

uc70.telecomsvc.com

Primary SIP User ID:

900193964

Primary SIP Auth ID:

900193964

Primary SIP Auth Password:

.....

Re-registration Interval (in seconds):

30

Backup SIP Server 1:

Backup SIP User ID:

Backup SIP Auth ID:

Backup SIP Auth Password:

Re-registration Interval (in seconds):

360

Backup SIP Server 2:

Backup SIP User ID:

Backup SIP Auth ID:

Backup SIP Auth Password:

Re-registration Interval (in seconds):

360

Remote SIP Port:

6060

Local SIP Port:

6159

SIP Transport Protocol:

UDP

TLS Version:

1.2 only (recommended)

Verify Server Certificate:

☐

Outbound Proxy:

Outbound Proxy Port:

0

Use Cisco SRST:

☐

Disable rport Discovery:

☐

Unregister on Boot:

☐

Keep Alive Period:

0

Nightringer Settings

SIP Server:

SIP User ID:

SIP Auth ID:

SIP Auth Password:

Re-registration Interval (in seconds):

360

Dial Out Settings

Dial out Extension:

204

Extension ID:

id204

Send Multicast Audio:

☐

Multicast Address:

224.5.5.5

Multicast Port:

5050

Repeat Message:

1

Call Disconnection

Terminate Call after delay:

0

Audio Codec Selection

Codec:

Auto Select

RTP Settings

RTP Port (even):

10500

Jitter Buffer:

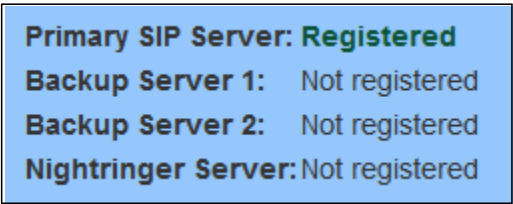
50

Save

Reboot

Toggle Help

Once verified, click the Reboot button. Your Outdoor Intercom should reach out to Intermedia's network and achieve successful registration when it powers up. Registration status may be confirmed on the Home tab of the user interface.

A screenshot of a registration status window with a light blue background and a thin black border. It contains four lines of text: 'Primary SIP Server: Registered' (where 'Registered' is in green), 'Backup Server 1: Not registered', 'Backup Server 2: Not registered', and 'Nightringer Server: Not registered'.

Primary SIP Server:	Registered
Backup Server 1:	Not registered
Backup Server 2:	Not registered
Nightringer Server:	Not registered

Once registered, your Outdoor Intercom will receive inbound calls placed to the 10-digit number, or 3-digit extension assigned by your Hosted PBX service.

8.1 Configuring the Outdoor Keypad Intercom to Auto Dial Numbers and Extensions

By default, the Outdoor Keypad Intercom will behave like a typical SIP phone and will emit a dial tone upon pressing the Call Button, allowing for the dialing of Hosted PBX extensions or full 10-digit phone numbers. You may wish to more closely constrain the destinations which are reachable from the Outdoor Intercom to prevent misuse. Some logical dial destinations may include:

- A Hosted PBX extension number
- A Hosted PBX phone number such as a desk phone or cell phone
- An Auto Attendant phone number

The easiest way to accomplish this is to utilize the Outdoor Intercom's Speed Dial feature. To locate this feature, within the Outdoor Intercom's web user interface:

1. Click on the Buttons section at the top.
2. Under Dial Mode, select 'Enable Speed Dial Operation'. The 'Speed Dial Settings' section will now become enabled.

Speed Dial Settings

Speed Dial Timeout:

Keypad 1:	<input type="text" value="100"/>	ID: <input type="text" value="2066861234"/>
Keypad 2:	<input type="text" value="101"/>	ID: <input type="text" value="2066861234"/>
Keypad 3:	<input type="text" value="102"/>	ID: <input type="text" value="2066861234"/>

The Speed Dial Settings section allows you to program the actions of a specific button on the Outdoor Intercom's keypad. For instance, you may program the '1' button on the keypad to automatically dial your Hosted PBX extension 100 when pressed. Alternately, you may program a full 10-digit phone number, such as your main Auto Attendant. You may even create complex dial strings, including pauses and menu choices.

8.1 To Program a Phone Number or Extension

1. Set the Speed Dial Timeout value. This value will determine how long, in seconds, a button must be pressed in order to place a call. Use a value of 0 for no delay.
2. Next choose the keypad digit you wish to program. You may also program the Call Button itself to automatically dial a number or extension.
3. Locate the 'Keypad X:' field, where 'X' is the button you wish to program.
4. Enter the full 10-digit phone number or extension number in the first column.
5. Enter the Caller ID you wish to be displayed when someone uses this Speed Dial in the second column named 'ID'. This field will accept numbers and letters.

6. Click the Save button.
7. Click the Reboot button.

Speed Dial Settings		
Speed Dial Timeout:	0	
Keypad 1:	100	ID: 2066861234
Keypad 2:	101	ID: 2066861234
Keypad 3:	102	ID: 2066861234
Keypad 4:	2066861234	ID: Front Door
Keypad 5:	4251234567	ID: Front Door
Keypad 6:	3609876543	ID: Front Door

The Outdoor Intercom will now place a phone call to the phone numbers or extensions you have programmed when each corresponding button is pressed on the keypad.

8.2 To Program an Auto Attendant Number

More complex speed dials may be attained with the use of an Intermedia Auto Attendant. Auto Attendants allow for the use of calls to pre-programmed hunt groups or ring groups, which are a more efficient way of reaching multiple people within your organization should building access be required. To program the Outdoor Intercom to use an Auto Attendant hunt groups or ring groups:

1. First, create your hunt group or ring group within your Auto Attendant, and link the hunt group or ring group to a menu option within your active menu.

Menu Option List		
Menu Action	Destination	
0 Replay Menu		
1 Extension	Extension 555, Sales	
2 Extension	Extension 123, Service Custom	
3 Voice Mail		
5 Hunt Group	Hunt Group, Front Door	
		DELETE

2. Next, decide which keypad option on the Outdoor Intercom will correspond to the action to call this Auto Attendant. You may also program the Call Button itself to automatically dial the Auto Attendant.
3. Locate the 'Keypad X:' field, where 'X' is the button you wish to program.
4. Enter the full 10-digit phone number of the Auto Attendant in the first column.

5. Next, enter a comma, followed by the Auto Attendant menu option number for your hunt group or ring group. The comma creates a pause of 3 seconds before sending the next digits. E.x. 2066861234,5

Keypad #:		ID:	
Call Button:	2066861234,5	ID:	Front Door

6. Click the Save button.
7. Click the Reboot button.

Test your setup thoroughly. If necessary, you may need to add additional commas to compensate for post dial delay to create a longer delay or access nested menu choices. Each comma will increase the delay by an additional 3 seconds. For example, the following dial string will produce the following results:

- 2066861234,1,,123
 - Dial 2066861234
 - Pause 3 seconds
 - Dial 1
 - Pause 6 seconds
 - Dial 123

This mechanism can be used to automatically access nested menus, but remember to adjust your programming in the Outdoor Intercom any time menu routing or voiced greetings are updated.