



MXA310

Table Array Microphone

Complete user guide for the MXA310. Includes installation instructions, specifications, best practices, command strings, and more.
Version: 5.1 (2020-H)

Table of Contents

MXA310 Table Array Microphone	4	How to Update Firmware Using Designer (Version 4.2 and Newer)	26
Overview	4	Firmware Versioning	27
General Description	4	Microphone Configuration	27
Features	4	Control Software Overview	27
System Overview	5	Microphone Configuration	27
System Planning and Gear Requirements	6	Adding or Removing a Channel	28
Setting up the Audio Network	6	Custom Presets	29
System Use Cases	9	Templates	29
Telephone Conference with Shure MXW Audio Network Interface	9	Adjusting Levels	29
Telephone Conference with Dante-enabled Audio Processor	11	Mute and Fader Groups	30
Telephone Conference with Breakout Boxes and Audio Processor	12	Identifying Channels	31
Web Conferencing Software with Dante Virtual Soundcard 3	13	Device Identification	31
Video Conference	14	LED Light Ring	31
Hardware and Installation	15	Parametric Equalizer (PEQ)	31
Hardware	15	Reset	34
LED Light Ring	16	Low-cut Filter	36
Installing the Cable-Exit Plug	17	Mute Buttons	36
Permanent Table Installation	17	AES67	37
Power Over Ethernet (PoE)	20	Encryption	37
Microphone Placement	20	Mute Synchronization	38
Seating Scenarios	20	Setting Up Mute Sync on the MXA310 and P300	38
Software Installation, Management, and Security	25	Automix Channel	38
Controlling Devices with Shure Designer Software	25	Automix Settings	39
Accessing the Web Application	26	Automix Modes	39
Web Application Browser Compatibility	26	Using a Third-Party Control System	40
Accessing the Web Application without the Discovery App 6	26	Networking and Dante	40
		Digital Audio Networking	40
		Switch and Cable Recommendations for Dante Networking	40
		Compatibility with Dante Domain Manager	41

QoS (Quality of Service) Settings	41	Troubleshooting	45
Dante Transmit Flows	42		
Networking Best Practices	42	Important Product Information	46
Network Audio and Shure Control Data	42	Information to the user	47
Device IP Settings	42	What's in the Box	47
Configuring IP Settings	43		
Manually Assigning Static IP Address	43	Optional Accessories	48
Setting Latency	43	Specifications	48
Operating the Control Software over Wi-Fi	43	Frequency Response	49
Packet Bridge	44		
IP Ports and Protocols	44	Contact Customer Support	55

MXA310

Table Array Microphone

Overview

General Description

The Microflex® Advance™ table array is a premium networked tabletop microphone for AV conferencing environments, including boardrooms, huddle rooms, and multi-purpose spaces. Revolutionary technology from the Shure DSP suite includes Steerable Coverage™, with selectable polar patterns on 4 independent channels to capture participant audio. The innovative new toroid polar pattern delivers 360° coverage, while rejecting sound from directly above the microphone. Control the microphone with Shure Designer software, or a browser-based web application. The microphone integrates seamlessly with Dante™ digital networked audio and third-party preset controllers, including Crestron and AMX, to deliver a high-quality AV conferencing experience that appeals equally to integrators, consultants, and meeting participants.

Features

Configurable Coverage

- Steerable Coverage delivers precise pick-up for up to 4 independent lobes
- Shure DSP Suite provides fast-acting automatic mixing and channel equalization
- Innovative toroid polar pattern delivers 360° coverage, while rejecting sound from directly above the microphone to reduce noise caused by HVAC systems or video projectors.

Software Control

- Shure Designer software provides comprehensive microphone and pattern control
- With Designer, you can also design coverage with online and offline devices, and route audio between Shure devices
- If Designer isn't available, use the browser-based web application to control the microphone

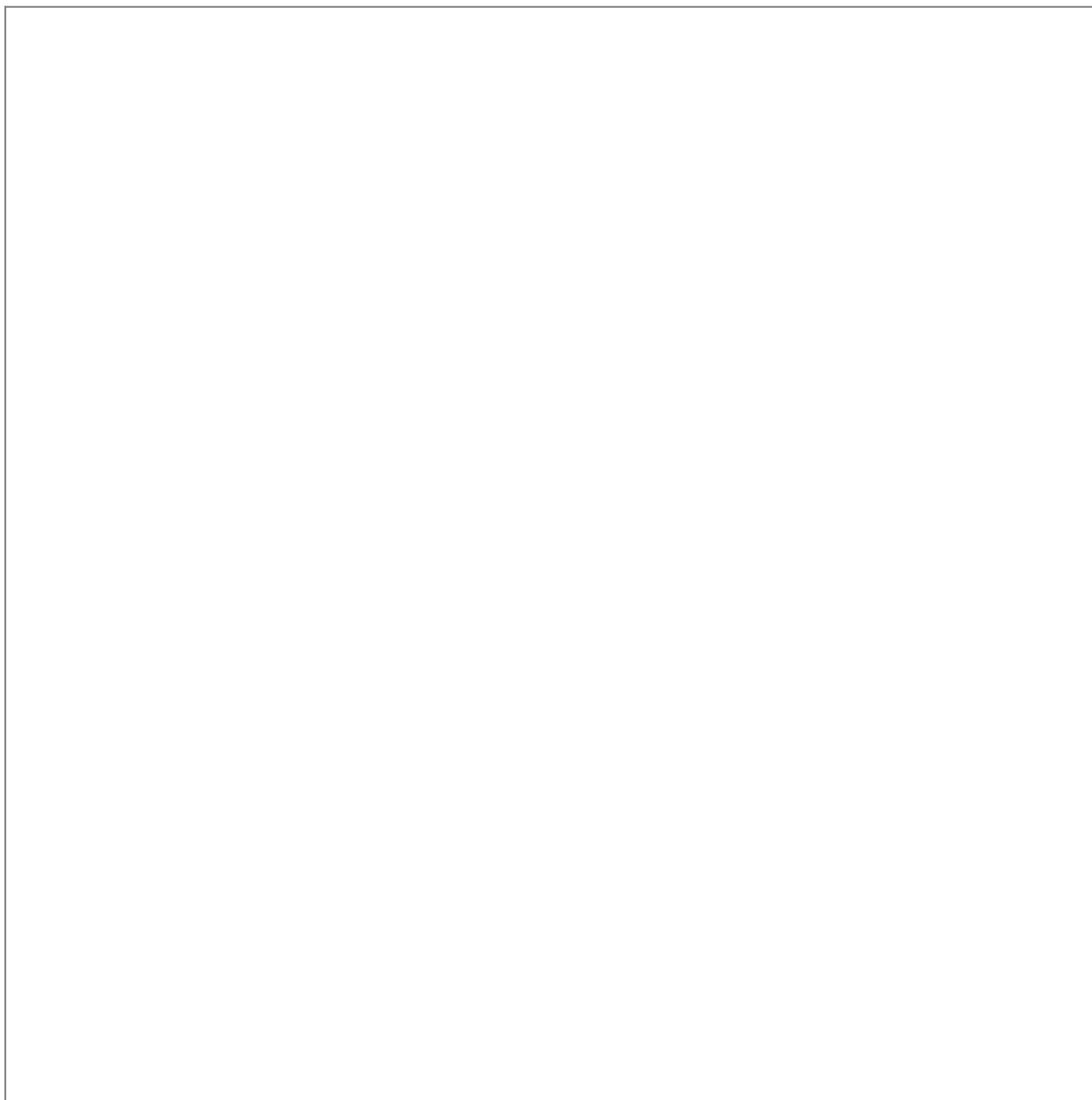
Network Connectivity

- Four discrete audio channels and an additional automix channel are delivered over a single network cable
- Dante™ digital audio coexists safely on the same network as IT and control data, or can be configured to use a dedicated network
- Control strings available for third-party preset controllers including Crestron and AMX

Professional Design

- Sleek, low-profile industrial design blends with contemporary board rooms and meeting spaces
- Configurable multi-colored LED light ring matches the environment, displays mute settings, and confirms coverage settings
- Available in white, black, and aluminum finishes

System Overview



① Dante audio, power, and control

A single network cable delivers 4 discrete audio channels from each microphone onto the Dante network, where they can be routed to any Dante-compatible devices.

② Analog audio (microphone to network)

Analog equipment, such as a wireless microphone system or a gooseneck microphone on a podium, connects to the Dante™ audio network through a Shure Network Interface (model ANI4IN) for a completely networked conferencing system.

③ Far-end audio (network to loudspeakers)

Dante™-enabled loudspeakers and amplifiers connect directly to a network switch. Analog loudspeakers and amplifiers connect through a Shure Network Interface (model ANI4OUT), which converts Dante™ audio channels into analog signals, delivered through 4 discrete XLR or block connector outputs.

④ Device control and Dante audio

A computer running Dante Controller and Shure Designer provides control over the following:

Control:

A computer connected to the network controls the microphone with Shure Designer software. You can remotely control coverage, muting, LED behavior, gain, and network settings.

Audio:

Route audio with Dante™ Controller or Shure Designer software. Dante™ Virtual Soundcard enables audio monitoring and recording directly on the computer.

System Planning and Gear Requirements

Setting up the Audio Network

Shure networked conferencing systems are comprised of Microflex Advance microphones and network interfaces, which operate entirely on a Dante network. Additional hardware, including network switches, computers, loudspeakers, and audio processors are described in the hardware component index.

Shure components shown in this diagram:

Microflex Advance Microphones

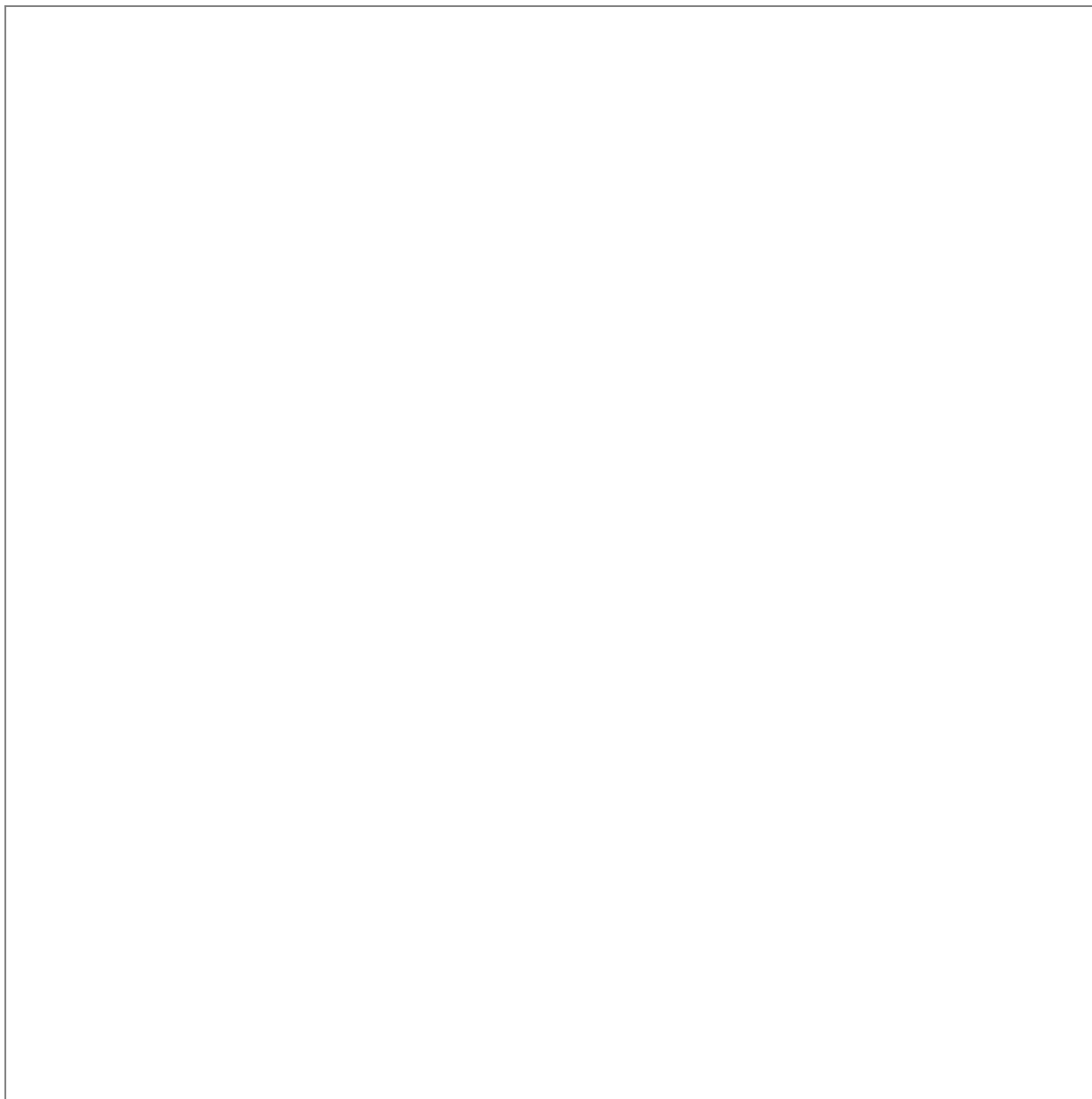
The MXA910 and MXA310 are equipped with Dante outputs, and connect directly to a network switch.

Audio Network Interfaces

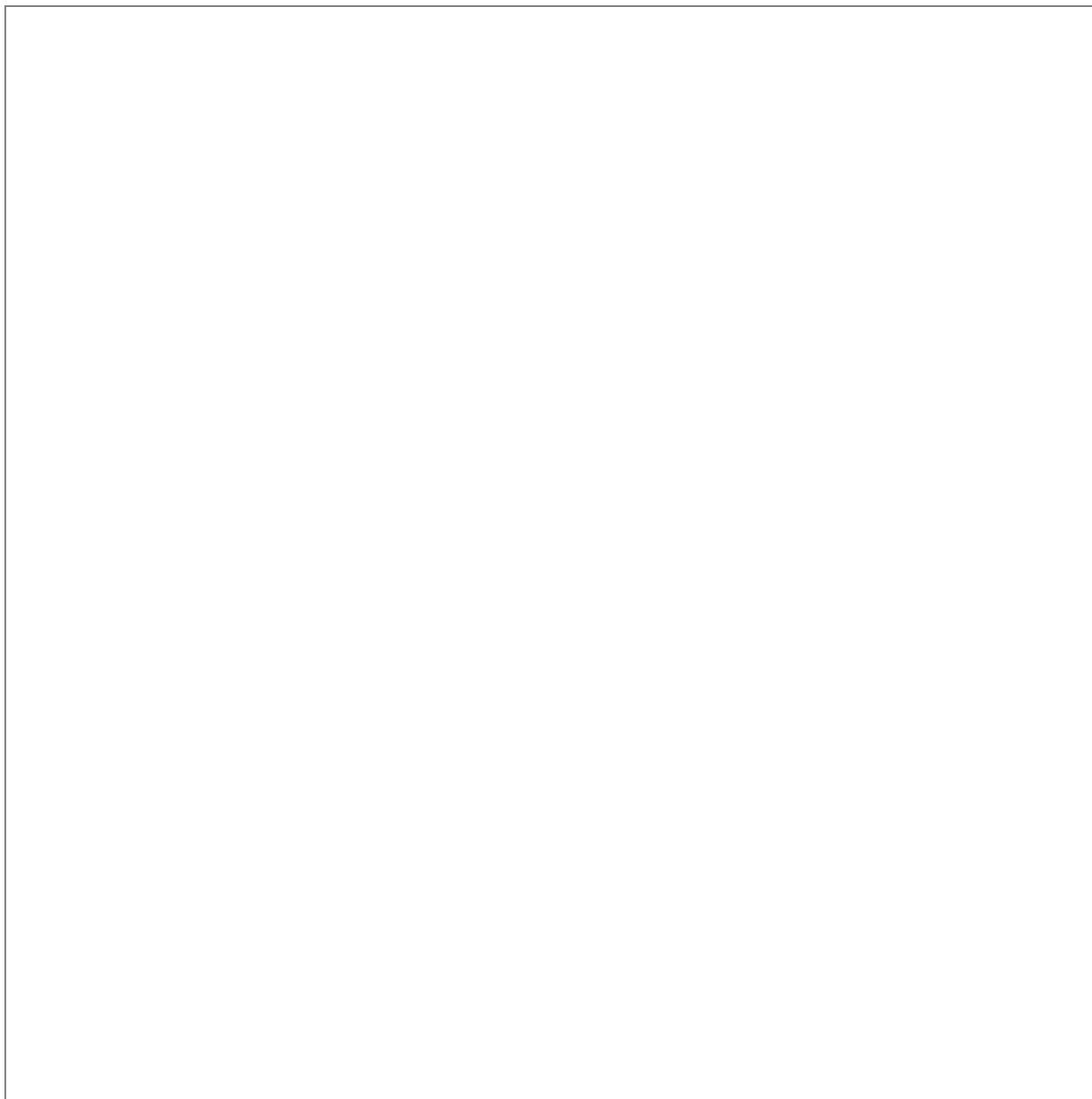
The interfaces are used to connect analog devices such as loudspeakers and analog microphones to the network.

ANI4IN: Converts 4 analog signals (separate XLR and block connector models available) into Dante digital audio signals.

ANI4OUT: Converts 4 channels of Dante audio from the network into analog signals.



This diagram shows the entire signal path through a networked conference system. Signals from the near end and far end are exchanged through an audio processor connected to a phone system, or through a computer connected to the internet. Analog microphones connect to the network through the Shure ANI4IN, while loudspeakers connect through the Shure ANI4OUT.



This diagram shows Microflex Advance components in context, with two rooms communicating through video codecs.

Controlling Hardware and Audio Over the Network

Audio and hardware settings are managed through a computer connected to the same network.

Shure Hardware and Audio

Each Microflex Advance component has a web application which provides mixing and configuration tools to optimize sound quality.

Expanded Control for Analog Devices

Analog devices that are connected to the network through a Shure network interface (ANI4IN/ANI4OUT) benefit from additional remote control: Volume levels, equalization, and signal routing are managed through the web application. For example, adjust-

ing loudspeaker volume or muting a wired microphone, which would normally be done from the hardware, can now be controlled remotely over the network.

Dante Signal Routing

You can manage signal routing with Dante Controller or Shure Designer software.

System Use Cases

These use cases will help you understand how Shure devices fit in conference rooms, huddle rooms, and multipurpose spaces.

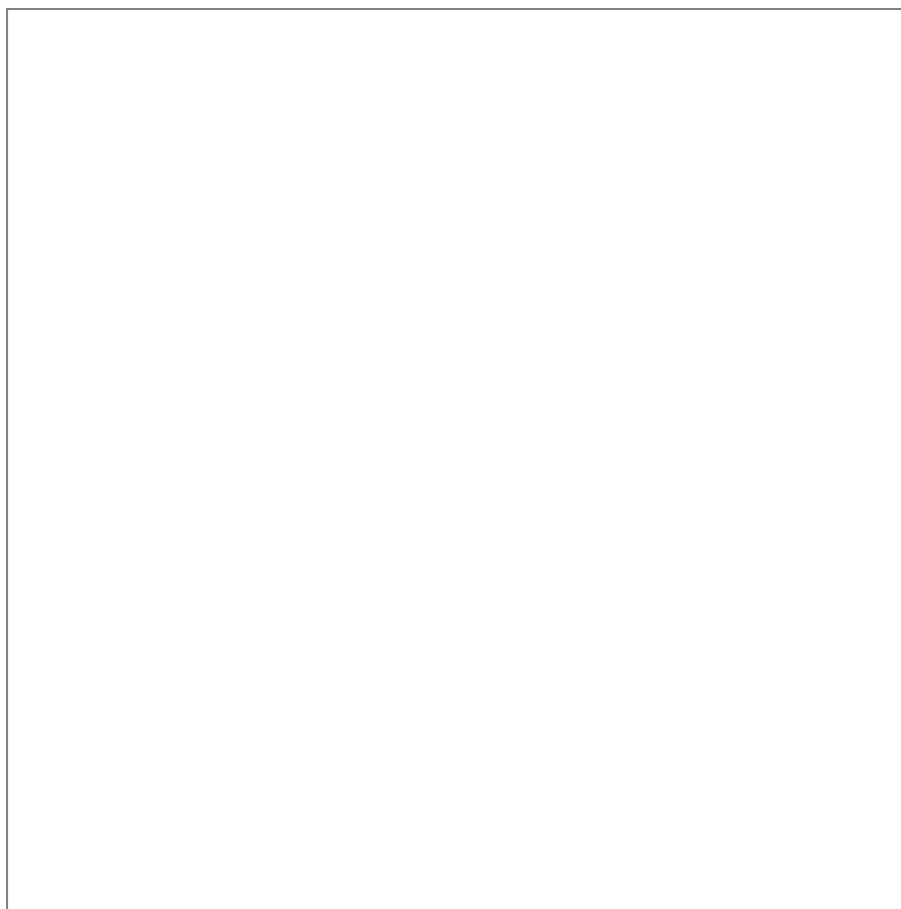
Each diagram includes:

- Signal flow and connections
- Required devices

Power Over Ethernet and Hardware Requirements

All Shure devices included in these use cases **require Power over Ethernet (PoE, class 0)**. Refer to the Dante and Networking section for additional information on cable and network switch requirements.

Telephone Conference with Shure MXW Audio Network Interface



① Array microphone to Shure MXWANI

Connect the microphone output to **port 1** on the MXWANI with a network cable. Port 1 provides the necessary Power over Ethernet (PoE).

② Computer to Shure MWXANI

Connect a computer to the ANI on port 2 or 3 with a network cable to provide control of the array microphone and other networked components.

③ Shure ANI analog outputs to audio processor

Step 1: Route signals with Dante Controller software

Route the channels from the microphone (Dante transmitter) to the MXWANI channels (Dante receiver). This establishes the discrete channels to deliver through the analog outputs.

Step 2: Connect the MXWANI outputs to the processing device inputs

Block connector outputs on the MXWANI send balanced audio signals to the inputs on the processing device, which provides digital signal processing (such as acoustic echo cancellation).

④ Connection to far end

Connect the audio processor to a VOIP server or telephone line to send and receive audio between the near end and far end.

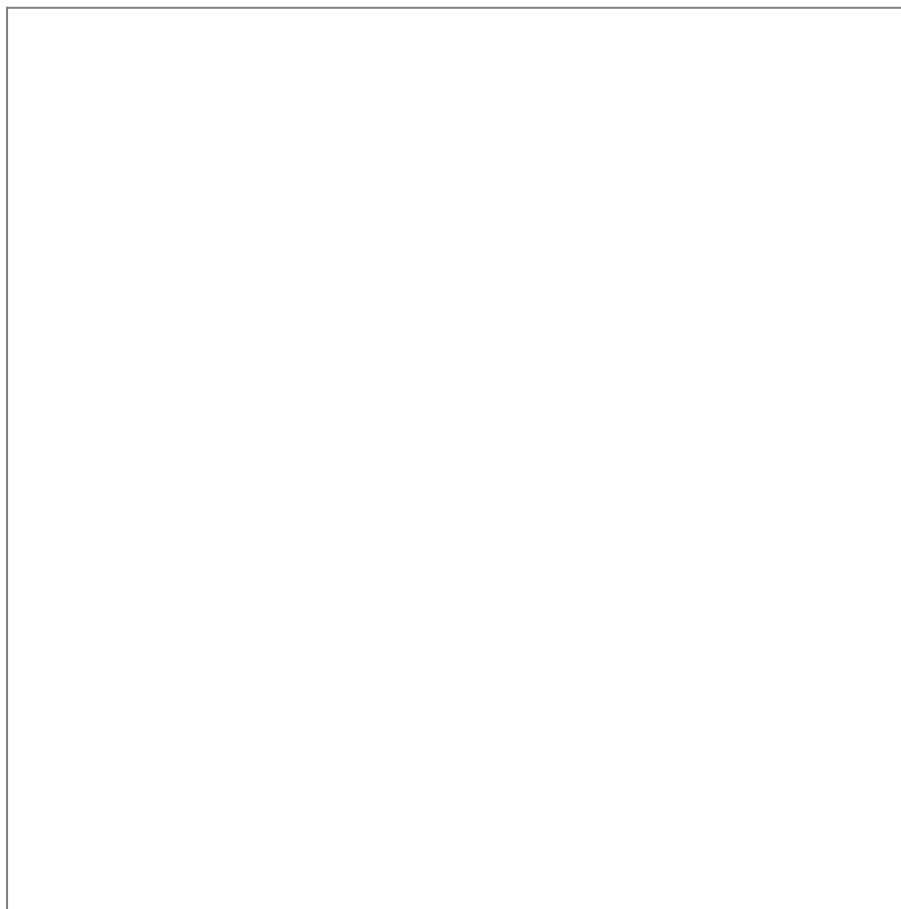
⑤ Audio from far end to amplifier

Route the far-end audio through the audio processor output to an amplifier.

⑥ Amplified audio signal to loudspeakers

Connect the loudspeakers to the amplifier to hear the audio from the far end.

Telephone Conference with Dante-enabled Audio Processor



① Array microphone to network switch

Connect the microphone output with a network cable to any port on the switch that supplies Power over Ethernet (PoE).

② Computer to network switch

Connect a computer to the network switch to provide control of the microphone and other networked components.

③ Network switch to Dante audio processor

Connect the Dante audio processor to the network switch to provide:

- Digital signal processing (acoustic echo cancellation)
- Digital-to-analog conversion to deliver Dante audio over an analog (VOIP or telephone line) output.
- Analog-to-digital conversion to deliver analog audio from the far end onto the Dante network.

④ Connection to far end

Connect the output from the audio processor to a VOIP server or telephone line to deliver audio between the near end and far end.

⑤ Audio from far end to amplifier

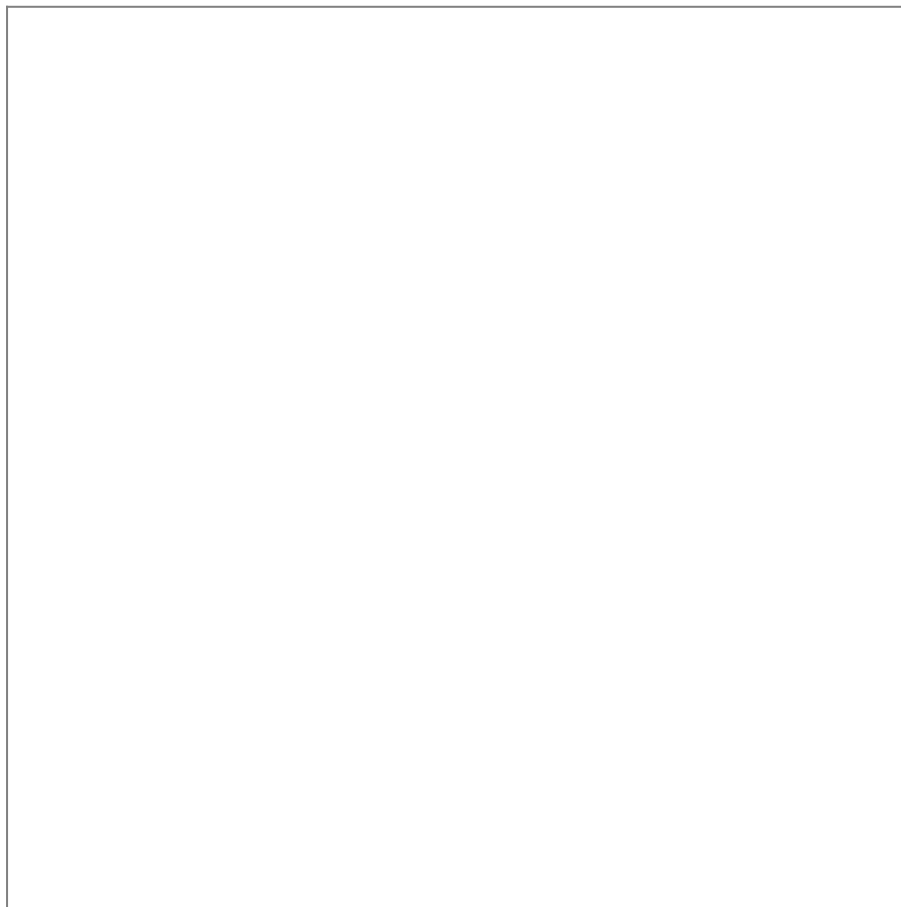
Route the far end audio through the audio processor output to an amplifier.

⑥ Amplified audio signal to loudspeakers

Connect the loudspeakers to the amplifier to deliver the audio from the far end.

Telephone Conference with Breakout Boxes and Audio Processor

In this scenario, two MXA310 microphones are used for a total of 8 Dante audio channels. Using two network interfaces, the Dante channels are converted to analog signals for acoustic echo cancellation.



① Microphone to network switch

Connect the array microphone output with a network cable to any port on the switch that supplies power over ethernet (PoE).

② Computer to network switch

Connect a computer to the network switch to provide control of the microphone and other networked components through the software control panel.

③ ANI4OUT (digital-to-analog conversion)

From the network switch: Use network cables to connect each ANI4OUT to the network switch. A single ANI4OUT receives 4 channels of Dante audio, and converts them to 4 analog signals, delivered through XLR outputs or block connectors. Using two of them, all 8 channels from the microphones can be connected to analog inputs on an audio processing device.

To a processing device: Route the ANI4OUT outputs to the processing device inputs to provide digital signal processing (acoustic echo cancellation).

④ Connection to far end

Connect the output from the audio processor to a VOIP server or telephone line to deliver audio between the near end and far end.

⑤ Audio from far end to amplifier

Route the far end audio through the audio processor output to an amplifier.

⑥ Amplified audio signal to loudspeakers

Connect the loudspeakers to the amplifier to deliver the audio from the far end

Web Conferencing Software with Dante Virtual Soundcard



① Microphone to network switch

Connect the microphone output with a network cable to any port on the switch that supplies Power over Ethernet (PoE).

② Computer to network switch

Connect a computer to the network switch to provide control of the microphone and other networked components through the software control panel. The computer also runs Dante Virtual Soundcard, Dante Controller, and the web conferencing software.

- **Dante Virtual Soundcard / Controller:** Turn on the Dante Virtual Soundcard and use Dante Controller to route the microphone signal to the computer.

- **Web Conferencing Software:** Assign the audio input and output device settings to the appropriate Dante transmitter and receiver channels.

③ Network switch to ANI4OUT

Use network cables to connect each ANI4OUT to the network switch. Each interface receives 4 channels of Dante audio, and converts them to 4 analog signals, delivered through XLR outputs or block connectors.

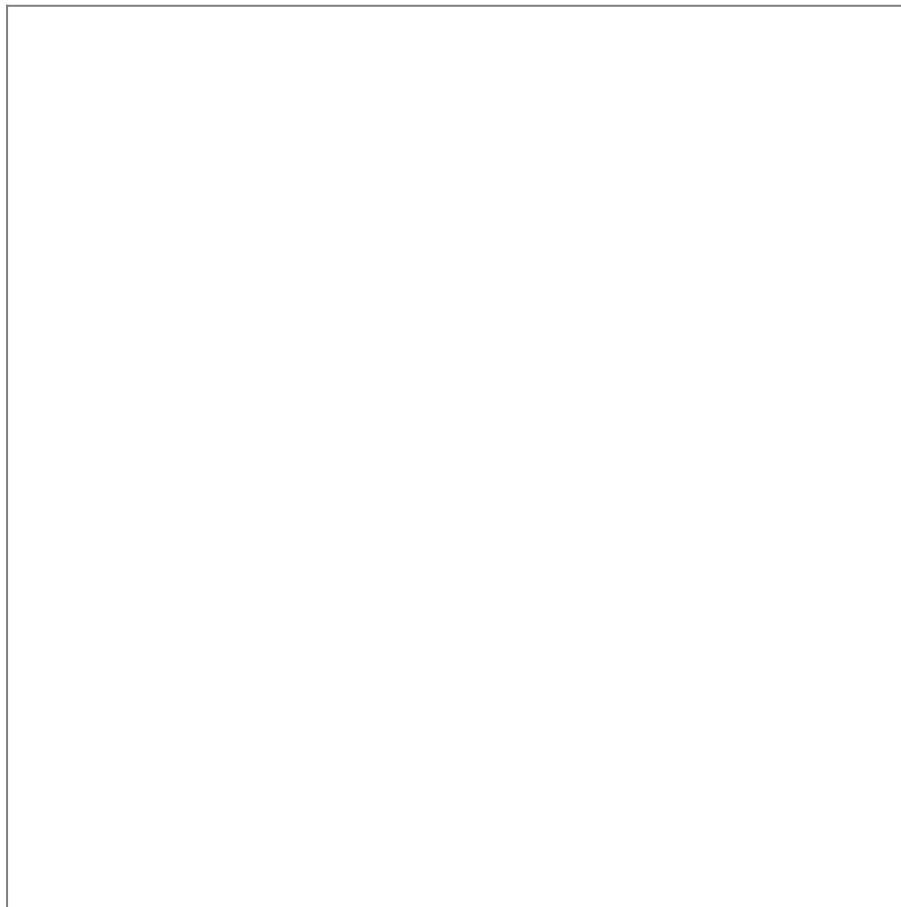
④ Audio from far end to amplifier

Route the far-end audio to an amplifier.

⑤ Amplified audio signal to loudspeakers

Connect the loudspeakers to the amplifier to deliver the audio from the far end.

Video Conference



① Microphone to network switch

Connect the microphone output with a network cable to any port on the switch that supplies power over Ethernet (PoE).

② Computer to network switch

Connect a computer to the network switch to provide control of the microphone and other networked components through the software control panel.

③ ANI4OUT (digital-to-analog conversion)

Each ANI4OUT receives 4 channels of Dante audio, and converts them to 4 analog signals, delivered through XLR outputs or block connectors.

Input: Connect the ANI4OUT to the network switch with a network cable

Output: Connect the analog output to the audio input on the video codec

④ Video codec connection to far end

Connect the codec to the appropriate network to connect with the far end.

⑤ Audio from far end to amplifier

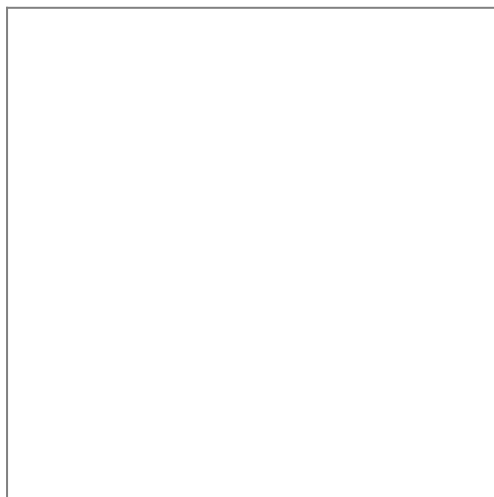
Route the far end audio through the video codec audio output to an amplifier.

⑥ Amplified audio signal to loudspeakers

Connect the loudspeakers to the amplifier to deliver the audio from the far end.

Hardware and Installation

Hardware



① Cable Exit

Guide the cable under the tabs and through the routing to exit from the side.

② Bottom Cable Exit

Guide the cable under the tabs and through the bottom exit for permanent table installations.

Note: Use the cable plug accessory when the cable is routed through the bottom.

③ Network Port

RJ-45 jack for network connection.

④ Network Status LED (Green)

Off = no network link

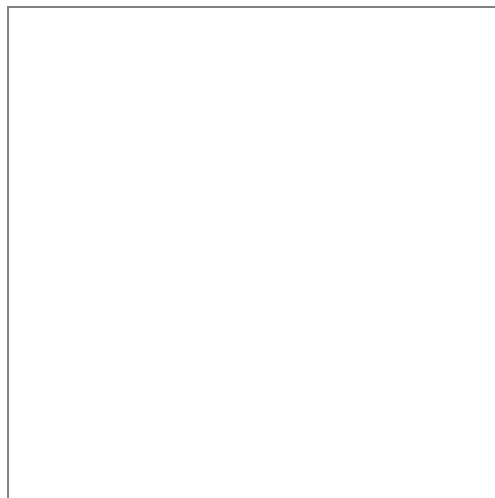
On = network link established

Flashing = network link active

⑤ Network Speed LED (Amber)

Off = 10/100 Mbps

On = 1 Gbps



⑥ Reset Button

Use a paperclip or similar tool to push the reset button.

⑦ Mute Buttons

Four touch-sensitive buttons control the mute status for each channel.

⑧ LED Light Ring

Indicates mute status, with configurable color and behavior states.

LED Light Ring

You can adjust LED light ring settings using Shure Designer software or the device's web application.

In Designer, go to **Settings > Lights**.

In the web application, go to **Configuration > Light Ring**.

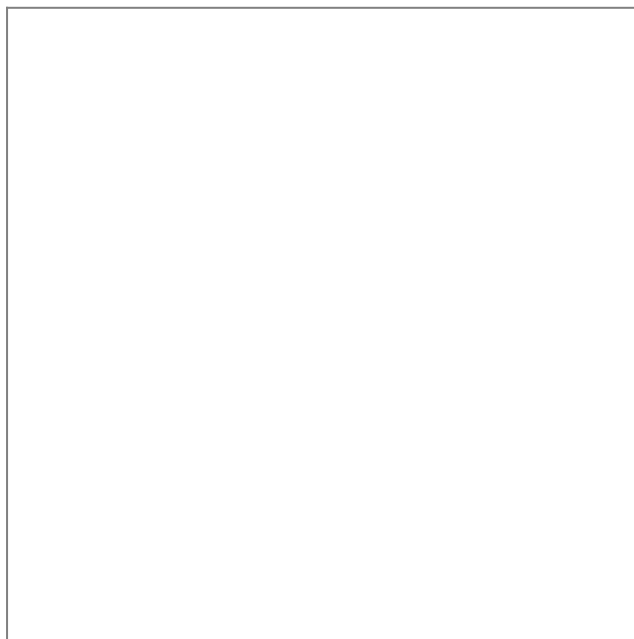
Default Settings

Microphone Status	LED Behavior / Color
Active	Green (solid)

Microphone Status	LED Behavior / Color
Mute	Red (solid)
Hardware identification	Green (flashing) Device identify: Entire light ring Channel identify: Light ring segment
Firmware update in progress	Green (ring fills up, turns off, repeats)
Reset	Network reset: Red (rotates around ring) Factory reset: Blue (rotates around ring)
Error	Red (split, alternate flashing)
Device power-up	Blue (rotates around ring)

Installing the Cable-Exit Plug

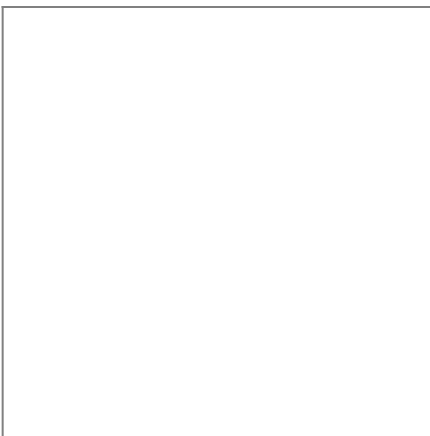
The plug covers the cable exit for permanent installations in which the cable is routed down through a table.



1. Remove the screw that holds in the cable-retaining tab closest to the cable exit
2. Remove the cable-retaining tab
3. Insert the plug
4. Replace the screw to secure the plug

Permanent Table Installation

1. Remove the 3 screws located in the center on the bottom of the microphone

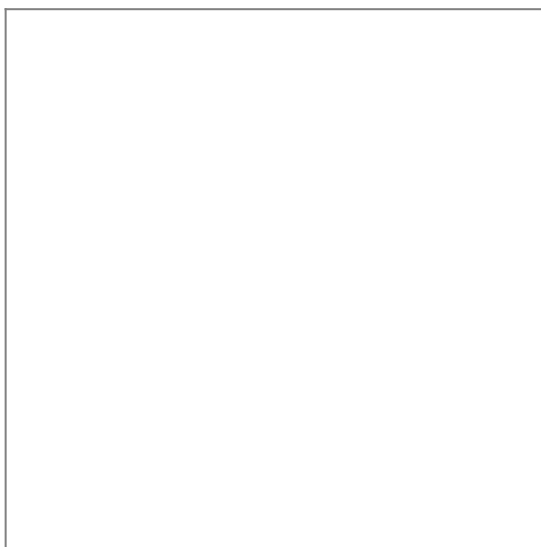


2. Plug a network cable into the microphone and guide it through the center exit path. When the cable is secured, guide it through the tube.

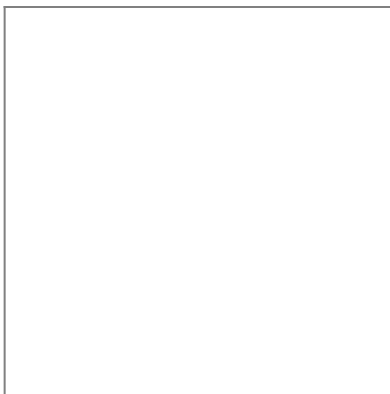
Note: If necessary, remove the retaining tabs to install thicker cable. Replace them after the cable is installed.



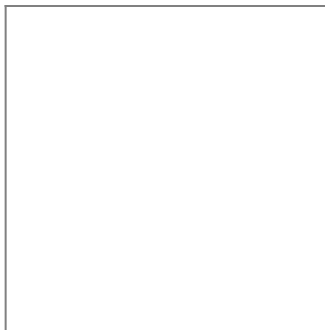
3. Align the tube into the recessed area in the center of the microphone. Install the 3 screws (removed in step 1) to secure the tube.



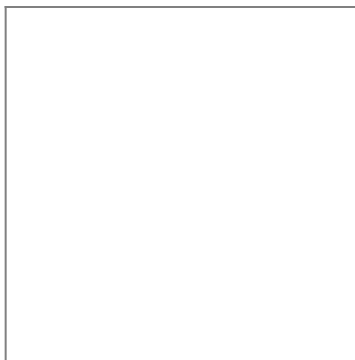
4. Slide one of the rubber washers to the base of the tube.



5. Drill a 1-inch (2.5 cm) hole through the table.

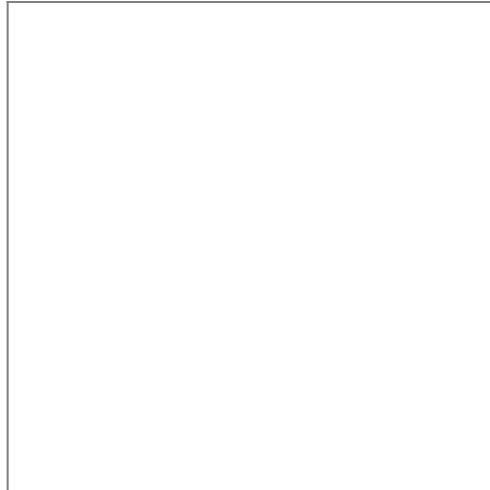


6. Guide the cable through the hole in the table. Then, place the tube through the hole in the table and gently press the microphone down.



7. Attach the remaining rubber washer and wing nut from underneath the table. Then, tighten the wing nut to secure the microphone on the table.

Optional: use the hole in the wing nut to insert a cable tie for cable management.



Power Over Ethernet (PoE)

This device requires PoE to operate. It is compatible with both **Class 0** and **Class 2** PoE sources.

Power over Ethernet is delivered in one of the following ways:

- A network switch that provides PoE
- A PoE injector device

Microphone Placement

Each microphone has 4 channels that can be aimed independently, based on the seating arrangement. Each channel features independent polar patterns and additional channel settings. You can control these settings in Shure Designer software or in the device's web application.

Designer software provides increased positioning flexibility over traditional conferencing microphones:

- Configurable pickup areas can be rotated and modified for the number of talkers.
- Network connectivity, device identification, and presets allow moving, adding and removing microphones with ease.
- Independent channels and automixing make Dante signal routing simple and flexible.
- Customized presets can be saved to immediately recall different room configurations.

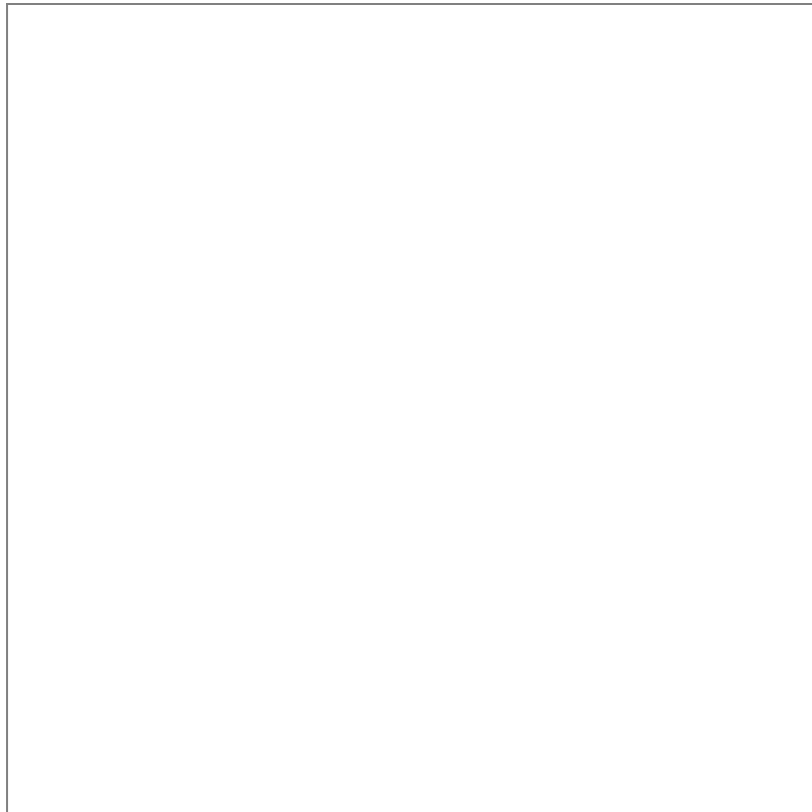
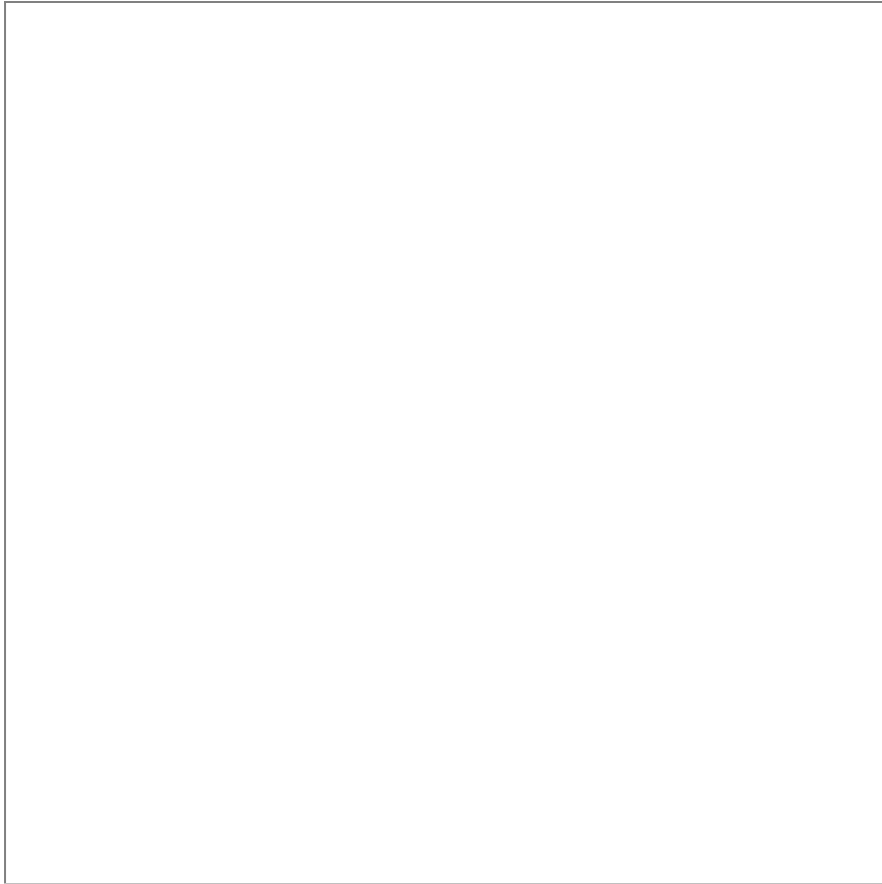
Seating Scenarios

Each channel can capture one or several talkers. In rooms with flexible furniture arrangements, microphones can be moved to cover various seating arrangements as long as they are plugged into the same network.

Note: Settings are saved on each microphone, and are retained when plugged into a different network port. Presets can be recalled and deployed using Designer software, the web application, or an external control system.

Single-Microphone Applications (Multiple Channels)

With four independent channels and polar patterns, coverage can be customized to match the table shape, size, and seating arrangement. The automatic mixing feature helps reduce extraneous noise (such as typing or paper shuffling) from interfering with speech intelligibility on the far end.

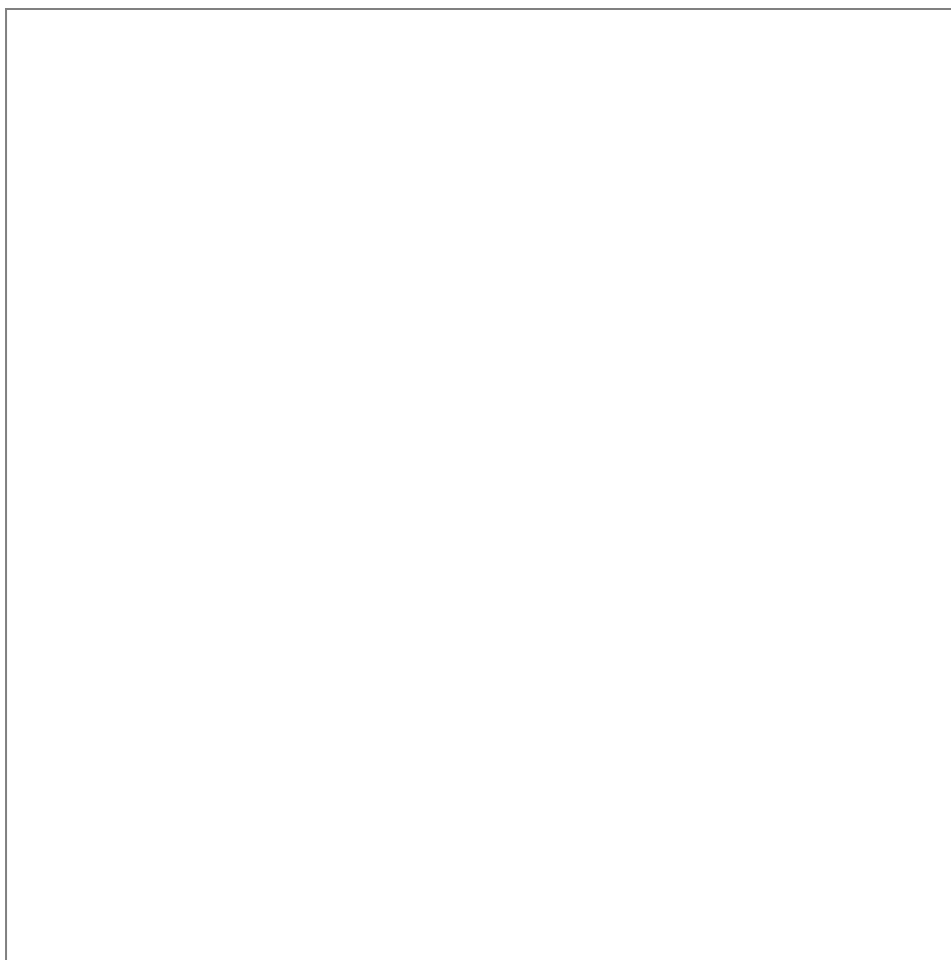


Toroid Pattern Applications

The toroid pattern rejects sound from directly above the microphone to reduce noise from video projectors or other sources of unwanted sound. It is the simplest way to ensure equal coverage among all talkers, while retaining the benefits of the rejection provided by a directional polar pattern. When this pattern is used, the audio is sent over a single channel. Therefore, when automatic mixing is desired, configure the microphone to use multiple directional patterns instead of the toroid pattern.



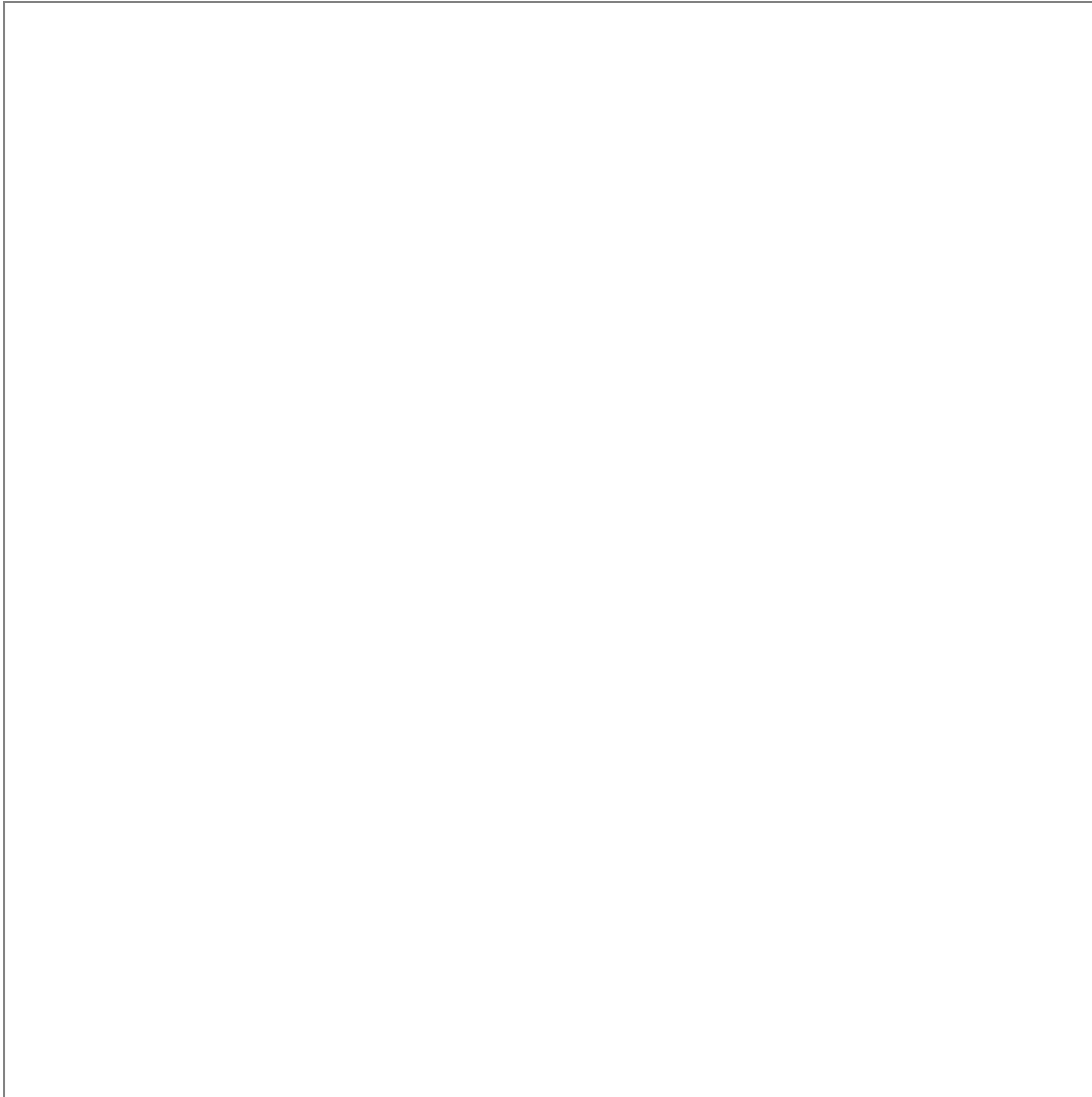
Noise from a ceiling-mounted projector is rejected, while all talkers are covered.



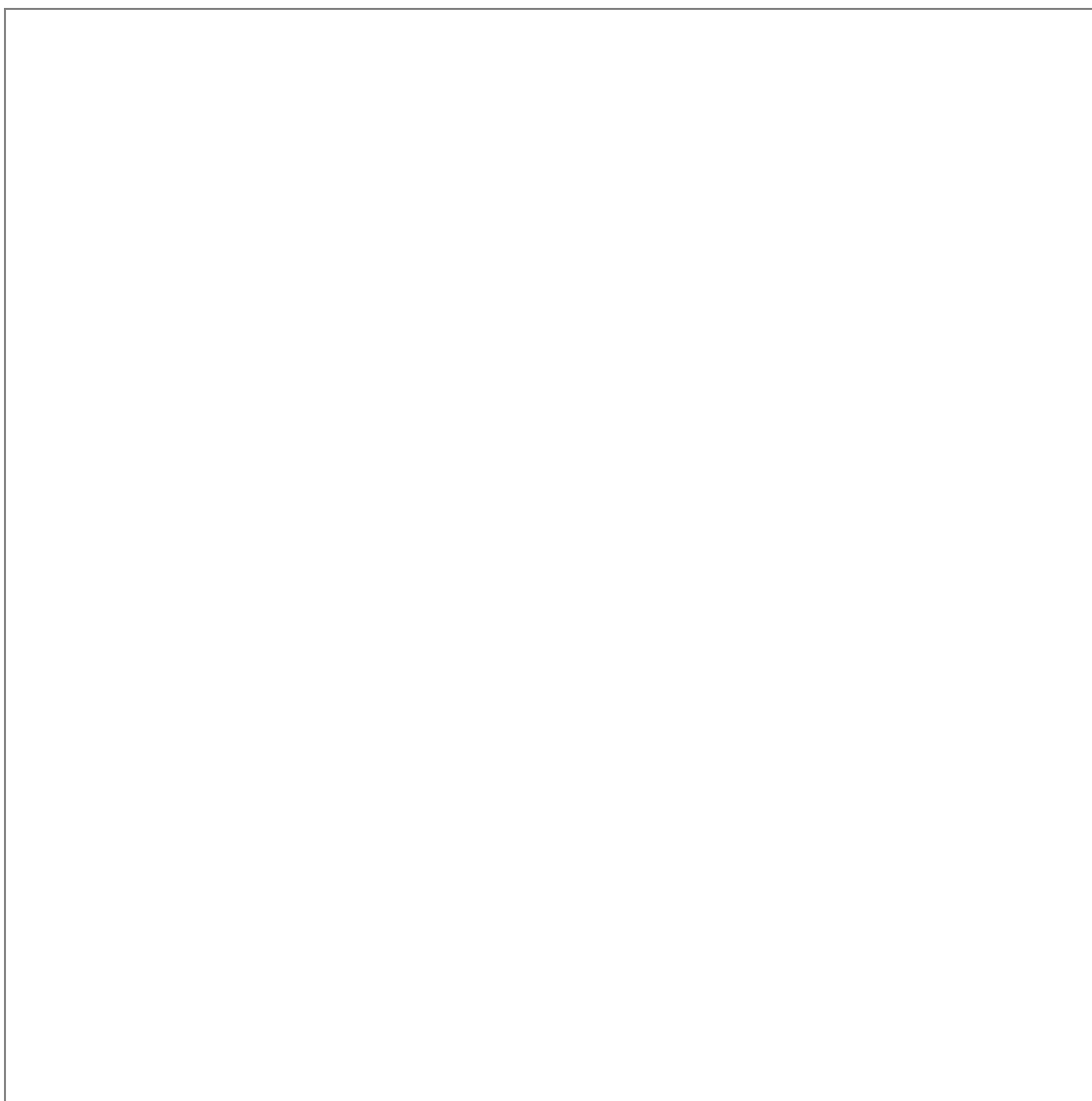
For a table with a single microphone and more than 4 talkers, the toroid pattern to ensures that all voices are heard equally.

Coverage With Multiple Microphones

For large tables, a series of microphones captures all talkers. Place the microphones in the center of the table for balanced pickup and accurate aiming. For the best audio quality and clarity, use enough microphones so that each talker has their own channel.



A table with 10 people is covered by 4 microphones, with an independent channel for each person.



For a large table with 2 microphones, place the microphones to cover equally sized areas. Use the Toroid or Omnidirectional setting to cover the entire table.

Software Installation, Management, and Security

Controlling Devices with Shure Designer Software

You can control this device using Shure Designer software. Designer enables integrators and system planners to design audio coverage for installations using MXA microphones and other Shure networked components.

With Designer, you can:

- Design audio coverage, whether online or offline
- Control Shure device settings and coverage
- Route audio between Shure devices
- Push settings to many devices at once
- Create and reuse templates across multiple locations and projects
- Import floor plans

To access your device in Designer:

1. Download and install Designer on a computer connected to the same network as your device.
2. Open Designer, and check that you're connected to the correct network in Settings.
3. Click Online devices. A list of online devices appears.
4. To identify devices, click the product icon to flash the lights on a device. Select your device in the list and click Configure to open the device's configuration window.

Learn more and download at www.shure.com/designer.

Accessing the Web Application

The Shure Web Server Discovery application finds all Shure devices on the network that feature a web-based GUI. Follow these steps to install the software and access the web application:

① Install the Shure Discovery application

Download and install the Shure Discovery application from www.shure.com. This automatically installs the required Bonjour device discovery tool on the computer.

② Connect the network

Ensure the computer and the hardware are on the same network.

③ Launch the Discovery application

The app displays all Shure devices that feature a GUI.

④ Identify the hardware

Double-click on a device to open its GUI in a web browser.

⑤ Bookmark the device's web application (recommended)

Bookmark the device's DNS name to access the GUI without the Shure Discovery app.

Web Application Browser Compatibility

The web application is compatible with all HTML5-supported browsers. To ensure the best performance, disabling hardware acceleration and unused plug-ins is recommended.

Accessing the Web Application without the Discovery App

If the Discovery application is not installed, the web application can be accessed by typing the DNS name into an internet browser. The DNS name is derived from model of the unit, in combination with the last three bytes (six digits) of the MAC address, and ending in .local.

Format Example: If the MAC address of a unit is 00:0E:DD:AA:BB:CC, then the link is written as follows:

MXA310: <http://MXA310-aabbcc.local>

How to Update Firmware Using Designer (Version 4.2 and Newer)

Before setting up devices, check for firmware updates using Designer to take advantage of new features and improvements. You can also install firmware using [Shure Update Utility](#) for most products.

To update:

1. Open Designer. If there's new firmware that you haven't downloaded yet, Designer shows a banner with the number of updates available. Click to download firmware.
2. Go to Online devices and find your devices.
3. Choose a firmware version for each device from the Available firmware column. Make sure that no one is editing device settings during an update.
4. Select the checkbox next to each device you plan to update and click Update firmware. Devices may disappear from Online devices during an update. Don't close Designer while updating firmware.

Firmware Versioning

When updating firmware, update all hardware to the same firmware version to ensure consistent operation.

The firmware of all devices has the form of MAJOR.MINOR.PATCH (e.g., 1.2.14). At a minimum, all devices on the network, must have the same MAJOR and MINOR firmware version numbers (e.g., 1.2.x).

Microphone Configuration

Control Software Overview

The boundary microphone user interface provides flexible, in-depth control of the microphone to deliver exceptional results for nearly any room. The following control options are quickly accessible through a web browser on a desktop or mobile device:

- Channel levels, monitoring, and mute status
- Polar pattern selection
- Pickup area aiming
- Security and network settings
- Automix settings
- Light settings
- External control switch configuration





Microphone Configuration

The microphone features multiple configurations to adapt to any meeting space, based on these variables:

- Table size and shape
- Number of meeting participants
- Participant seating arrangement

Selecting Pickup Patterns

1. Select Configuration > Coverage
2. Select a channel to reveal the Channel Properties
3. Use the Polar Pattern pull-down menu to make a selection

Pickup Pattern	Directional Characteristic	Use When
 <p>Omnidirectional</p>	Picks up sound with equal sensitivity from all directions	Participants are likely to move around, or when additional sound sources are located away from the microphone. The omnidirectional pattern performs best in a quiet, controlled environment. Note: Omnidirectional channels are not sent to the automix channel.
 <p>Toroid</p>	Picks up sound from the edges of the microphone, while rejecting sounds from directly above it.	Rooms have a higher level of ambient noise, or when noise from above is a concern (a video projector, for example).
 <p>Bi-directional</p>	Captures sound on two opposite sides of the microphone in a figure-8 pattern	Two talkers are facing each other, sitting on opposite sides of a table. The bi-directional pattern provides better off-axis rejection than the two talkers setting, but does not allow independent gain adjustment for each talker.
 <p>One, Two, Three, or Four Talkers</p>	<p>Each pickup area features independent polar pattern control. Select each polar pattern setting based on the number of talkers in each pickup area and the table size or shape. Available patterns include:</p> <ul style="list-style-type: none"> Cardioid Supercardioid Toroid Omnidirectional Bi-directional Hypercardioid 	Maximum noise rejection and channel separation are desired, and when the seating configuration is unlikely to change. This configuration is optimal for use with automixing.

Aiming Pickup Areas

All pickup patterns (except omnidirectional and toroid) can be aimed directly at individual talkers to provide the clearest possible signal with minimal room ambience. In the Configuration menu, aim the pickup lobes by selecting and dragging the channel. The angle can also be adjusted in 15° increments from the channel properties menu on the right side of the workspace.

Adding or Removing a Channel

To add or remove a channel, go to Configuration.

- Select Add Channel to add another channel.
- Select Remove Channel or press the Delete key on your keyboard to remove a channel.

Removing a channel also deletes any EQ or gain settings applied to that channel.

Custom Presets

Use presets to quickly save and recall settings. Up to 10 presets can be stored on each device to match various seating arrangements. A preset saves all device settings except for the Device Name, IP Settings, and Passwords. Importing and exporting presets into new installations saves time and improves workflow. When a preset is selected, the name displays above the preset menu. If changes are made, an asterisk appears next to the name.

Note: Use the default settings preset to revert to the factory configuration (excludes Device Name, IP Settings, and Passwords).

Open the presets menu to reveal preset options:

save as preset:	Saves settings to the device
load preset:	Opens a configuration from the device
import from file:	Downloads a preset file from a computer onto the device. Files may be selected through the browser or dragged into the import window.
export to file:	Saves a preset file from the device onto a computer

Templates

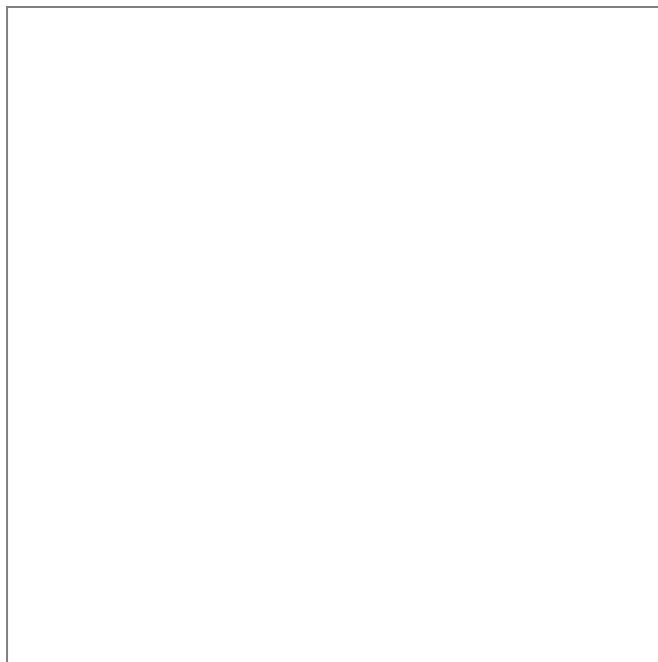
Use a template as a starting point when configuring coverage. Templates only adjust coverage, and do not affect gain levels or other settings.

1. Select the template that is the closest match to the seating scenario.
2. Select OK.
3. Select Add Channel or Remove Channel to adjust coverage.

Adjusting Levels

Input gain levels on Microflex[®] Advance microphones must be set for each saved coverage preset to ensure optimized gain structure for all seating scenarios. Always adjust the levels before making any changes to automix settings to ensure the best performance.

Each of the 4 microphone channels feature independent gain control. This feature is useful when meeting participants are seated at unequal distances from the microphone.



When the microphone is centered on a rectangular table, use the channel gain to balance the levels and compensate for the unequal distances.

1. Perform a level check for each coverage area, using a typical speech volume. Adjust the faders so the meters are peaking at approximately -20 dBFS.
2. Adjust the equalizer settings to optimize speech intelligibility and minimize noise (such as low-frequency rumble caused by HVAC systems).
3. If equalizer settings cause a significant increase or decrease in levels, make any necessary level adjustments according to step 1.

When to Use the Channel and Automix Gain Faders

There are 2 different gain faders that serve different purposes:

Channel Gain (Pre-Gate)

To adjust, go to Channels. These faders affect a channel's gain before it reaches the automixer and therefore affect the automixer's gating decision. Boosting the gain here will make the lobe more sensitive to sound sources and more likely to gate on. Lowering gain here makes the lobe less sensitive and less likely to gate on. If you're only using direct outputs for each channel without the automixer, you only need to use these faders.

Automix Gain (Post-Gate)

To adjust, go to Configuration > Automix. These faders adjust a channel's gain after the lobe has gated on. Adjusting the gain here *will not* affect the automixer's gating decision. Only use these faders to adjust the gain of a talker after you are satisfied with the automixer's gating behavior.

Note: The level meters in Configuration > Automix only display pre-gate channel gain, but the faders will adjust post-gate channel gain.

Mute and Fader Groups

Add channels to a Mute group or Fader group to link the corresponding controls together. For example, if channels 1, 2, and 3 are added to a Mute group, muting any of those individual channels will mute all of the grouped channels.



If you're using Shure Designer software to configure your system, please check the Designer help section for more about this topic.

Identifying Channels

Identify a channel on the microphone by flashing the corresponding LED. This quickly verifies that level or equalizer adjustments are being made to the intended channel.

1. Select Configuration > Coverage
2. Select a channel
3. Use the Identify Channel button to flash the LEDs on the microphone

Device Identification

To identify the microphone by flashing the light ring, select the Identify button in the device options section.

LED Light Ring

Light ring properties are configurable to match room or enterprise-wide behavior conventions and aesthetics.

Brightness	Adjusts the intensity level of the LED light ring
Lighting Style	Segments are divided to show individual channels. Ring is a continuous LED
Display Automix Gating	Indicates a channel is off (audio signal has dropped below the gate threshold). When enabled, the lighting style automatically switches to segment mode. Off: LED light ring turns off when a channel gates off Follow mute color: LED light ring switches to assigned mute color when a channel gates off
Unmute Behavior	LED activity when the microphone is active
Unmute Color	LED color when the microphone is active
Mute Behavior	LED activity when the microphone is muted
Mute Color	LED color when the microphone is muted

Parametric Equalizer (PEQ)

Maximize audio quality by adjusting the frequency response with the parametric equalizer.

Common equalizer applications:

- Improve speech intelligibility
- Reduce noise from HVAC systems or video projectors
- Reduce room irregularities
- Adjust frequency response for reinforcement systems

To turn off all EQ filters, select Bypass all EQ.



If you're using Shure Designer software to configure your system, please check the Designer help section for more about this topic.

Setting Filter Parameters

Adjust filter settings by manipulating the icons in the frequency response graph, or by entering numeric values. Disable a filter using the check-box next to the filter.

Filter Type

Only the first and last band have selectable filter types.

Parametric: Attenuates or boosts the signal within a customizable frequency range

Low Cut: Rolls off the audio signal below the selected frequency

Low Shelf: Attenuates or boosts the audio signal below the selected frequency

High Cut: Rolls off the audio signal above the selected frequency

High Shelf: Attenuates or boosts the audio signal above the selected frequency

Frequency

Select the center frequency of the filter to cut/boost

Gain

Adjusts the level for a specific filter (+/- 30 dB)

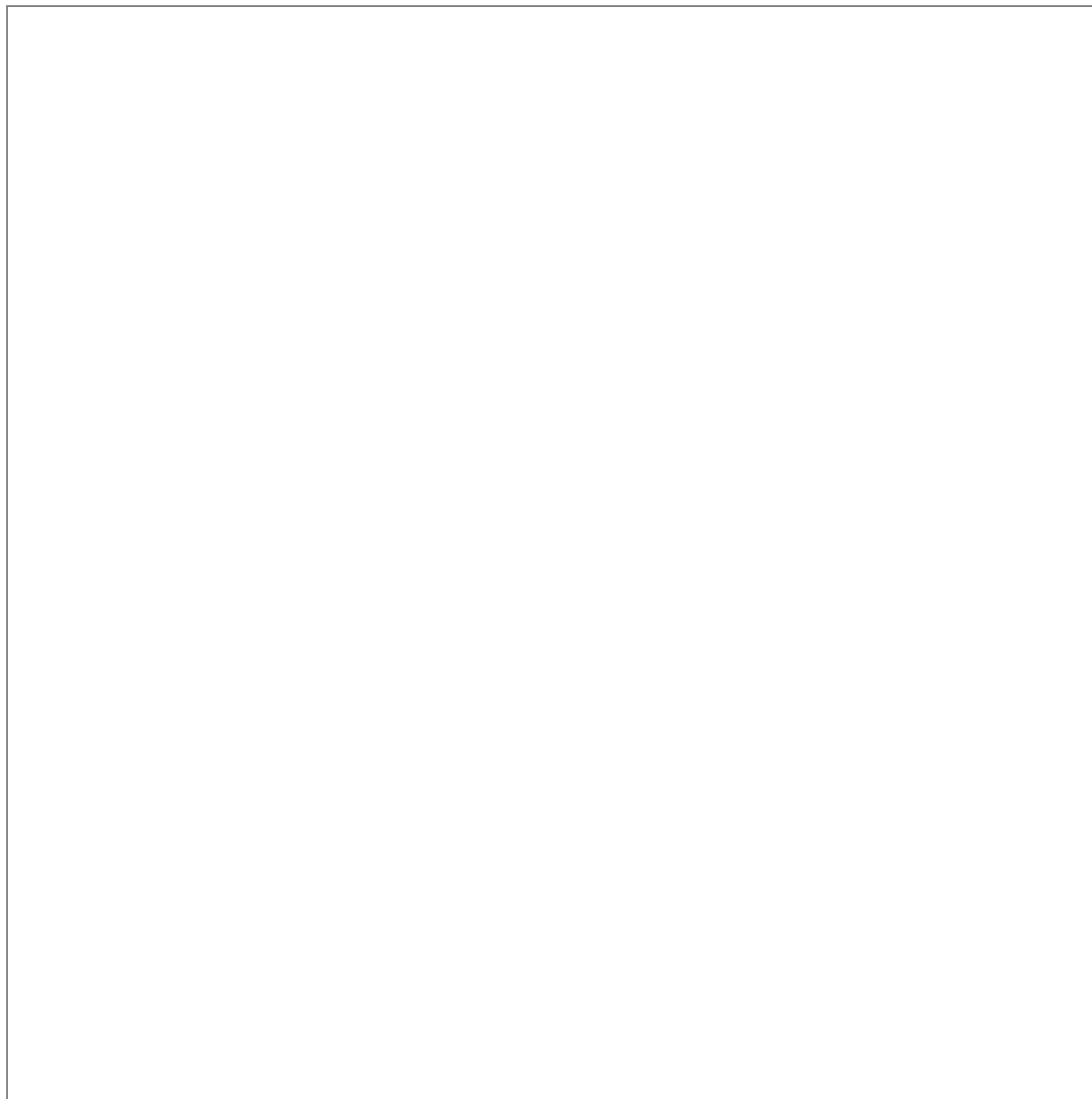
Q

Adjusts the range of frequencies affected by the filter. As this value increases, the bandwidth becomes thinner.

Width

Adjusts the range of frequencies affected by the filter. The value is represented in octaves.

Note: The *Q* and *width* parameters affect the equalization curve in the same way. The only difference is the way the values are represented.



Copy, Paste, Import, and Export Equalizer Channel Settings

These features make it simple to use effective equalizer settings from a previous installation, or simply accelerate configuration time.

Copy and Paste

Use to quickly apply the same PEQ setting across multiple channels.

1. Select the channel from the pull-down menu in the PEQ screen.
2. Select Copy
3. In the pull-down menu, select the channel to apply the PEQ setting and select Paste.

Import and Export

Use to save and load PEQ settings from a file on a computer. This is useful for creating a library of reusable configuration files on computers used for system installation.

Export

Choose a channel to save the PEQ setting, and select Export to file.

Import

Choose a channel to load the PEQ setting, and select Import from file.

When to Use the Channel and Automix Equalizers

Apply **Automix EQ** to make system-wide changes, such as a treble boost to improve speech clarity. Use **Channel EQ** to make adjustments to a specific channel. For example, to reduce unwanted noise picked up by only one channel.

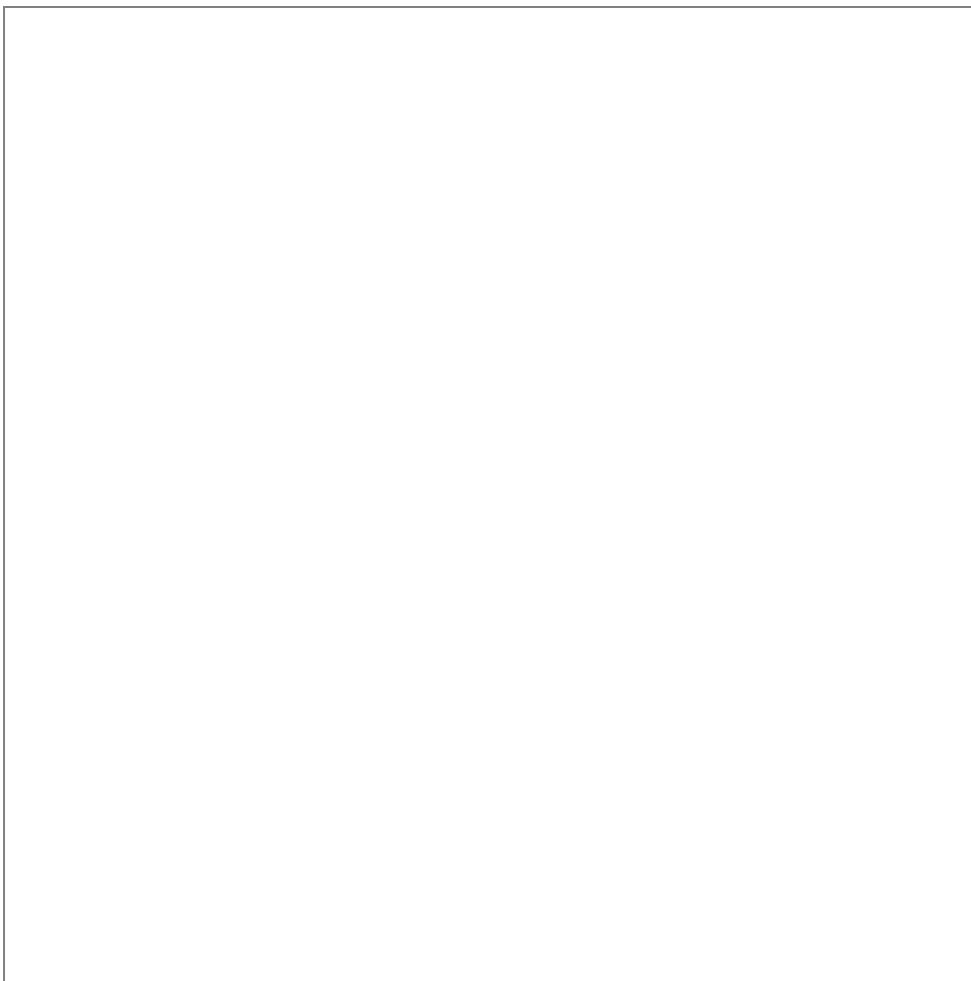
Equalizer Applications

Conferencing room acoustics vary based on room size, shape, and construction materials. Use the guidelines in following table.

EQ Application	Suggested Settings
Treble boost for improved speech intelligibility	Add a high shelf filter to boost frequencies greater than 1 kHz by 3-6 dB
HVAC noise reduction	Add a low cut filter to attenuate frequencies below 200 Hz
Reduce flutter echoes and sibilance	<p>Identify the specific frequency range that "excites" the room:</p> <ol style="list-style-type: none"> 1. Set a narrow Q value. 2. Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 1 kHz and 6 kHz to pinpoint the range of flutter echoes or sibilance. 3. Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound.
Reduce hollow, resonant room sound	<p>Identify the specific frequency range that "excites" the room:</p> <ol style="list-style-type: none"> 1. Set a narrow Q value. 2. Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 300 Hz and 900 Hz to pinpoint the resonant frequency. 3. Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound.

Reset

The reset button is located inside a small hole in the lower half of the microphone. Use a paperclip or other small tool to press the button.



There are 2 reset functions:

Network reset (press button for 4-8 seconds)

Resets all Shure control and audio network IP settings to factory defaults.

Full factory reset (press button for longer than 8 seconds)

Restores all network and web application settings to the factory defaults.

Software Reset Options

To simply revert settings without a complete hardware reset, use one of the following options:

Reboot Device (Settings > Factory Reset): Power-cycles the device as if it were unplugged from the network. All settings are retained when the device is rebooted.

Default Settings (Presets > Load Preset > Default Settings): Reverts audio settings back to the factory configuration (excluding Device Name, IP Settings, and Passwords).



If you're using Shure Designer software to configure your system, please check the Designer help section for more about this topic.

Low-cut Filter

The low-cut filter rolls off low frequencies to reduce unwanted noise from sources such as table vibrations, HVAC systems, and other environmental noise.

To enable, open the web application and select the low-cut filter button in the device options section.

Mute Buttons

The microphone has 4 touch-sensitive mute buttons around the edge of the microphone. Pressing any button mutes the entire device. You can mute channels individually using Shure Designer software or the web application. If the Light Ring is set to display Segments, the individual channel mutes are visible on the device. If it is set to Ring, the Light Ring only displays the device mute status.



To adjust button properties in Designer, go to **Settings > Logic control**.

To adjust button properties in the web application, go to **Configuration > Button Control**.

Mute Control Function

- **Local:** mutes/unmutes audio from the microphone
- **Logic out:** sends a command string to a control system to mute the audio farther down the signal path
- **Disabled:** button is inactive

Mute Control Mode

- **Toggle on/off:** Press the button to switch between mute and active states

- **Push to talk:** Hold the button to activate the microphone when speaking
- **Push to mute:** Hold the button to mute the microphone

Default Toggle State

Determines whether the microphone is muted or active when powered on

AES67

AES67 is a networked audio standard that enables communication between hardware components which use different IP audio technologies. This Shure device supports AES67 for increased compatibility within networked systems for live sound, integrated installations, and broadcast applications.

The following information is critical when transmitting or receiving AES67 signals:

- Update Dante Controller software to the newest available version to ensure the AES67 configuration tab appears.
- Before turning encryption on or off, you must disable AES67 in Dante Controller.
- AES67 cannot operate when the transmit and receive devices both support Dante.

Shure Device Supports:	Device 2 Supports:	AES67 Compatibility
Dante and AES67	Dante and AES67	No. Must use Dante.
Dante and AES67	AES67 without Dante. Any other audio networking protocol is acceptable.	Yes

Separate Dante and AES67 flows can operate simultaneously. The total number of flows is determined by the maximum flow limit of the device.

Sending Audio from a Shure Device

All AES67 configuration is managed in Dante Controller software. For more information, refer to the Dante Controller user guide.

1. Open the Shure transmitting device in Dante Controller.
2. Enable AES67.
3. Reboot the Shure device.
4. Create AES67 flows according to the instructions in the [Dante Controller user guide](#).

Receiving Audio from a Device Using a Different Audio Network Protocol

Third-party devices: When the hardware supports SAP, flows are identified in the routing software that the device uses. Otherwise, to receive an AES67 flow, the AES67 session ID and IP address are required.

Shure devices: The transmitting device must support SAP. In Dante Controller, a transmit device (appears as an IP address) can be routed like any other Dante device.

Encryption

Audio is encrypted with the Advanced Encryption Standard (AES-256), as specified by the US Government National Institute of Standards and Technology (NIST) publication FIPS-197. Shure devices that support encryption require a passphrase to make a connection. Encryption is not supported with third-party devices.

To activate encryption:

1. Open the Settings menu and select the General tab.
2. Select Enable Encryption.
3. Enter a passphrase. All devices must use the same passphrase to establish an encrypted connection.

Important: For encryption to work:

- All Shure devices on your network must use encryption.
- Disable AES67 in Dante Controller. AES67 and AES-256 cannot be used at the same time.



If you're using Shure Designer software to configure your system, please check the Designer help section for more about this topic.

Mute Synchronization

Mute synchronization ensures that all connected devices in a conferencing system mute or unmute at the same time and at the correct point in the signal path. Mute status is synchronized in the devices using logic signals or USB connections.

To use mute synchronization, enable logic on connected devices using the web application or Shure Designer software. Many Shure devices have logic enabled automatically.

If you use Designer's Optimize workflow, Designer configures all of the necessary mute synchronization settings for you.

Shure logic devices:

- P300 (Also mutes [supported soft codecs](#) connected by USB)
- ANIUSB-MATRIX (Also mutes [supported soft codecs](#) connected by USB)
- MXA910
- MXA710
- MXA310
- Network Mute Button
- ANI22-BLOCK
- ANI4IN-BLOCK
- Logic-enabled MX microphones connected to ANI22-BLOCK or ANI4IN-BLOCK
 - MX392
 - MX395-LED
 - MX396
 - MX405/410/415

Setting Up Mute Sync on the MXA310 and P300

Use this setup to mute the P300 by pressing the mute button on the MXA310. In the P300's signal chain, muting happens after the DSP so that the AEC stays converged.

1. Open the P300 in Designer and go to Inputs.
2. On any channel strip, select Logic enable. This enables logic on all channels.
3. Open the MXA310 in Designer and go to Settings > Logic control.
4. Set the Mute control function to Logic out.
5. Go to Lights. Set the style to Ring.

Automix Channel

This channel automatically mixes the audio from all channels to deliver a convenient, single output. The automix channel must be routed in Dante Controller to the desired output.

Note: Automix is disabled when using the toroid polar pattern. Inversely, the toroid pattern cannot be selected when automix is enabled.

To enable automixing and modify settings:

1. Select Configuration
2. Open the AUTOMIX tab
3. Check the Enable box

To modify settings from the channels screen:

1. Select Channels
2. In the AUTOMIX channel, select the AUTOMIX button

Automix Settings

Leave Last Mic On	Keeps the most recently used microphone channel active. The purpose of this feature is to keep natural room sound in the signal so that meeting participants on the far end know the audio signal has not been interrupted.
Gating Sensitivity	Changes the threshold of the level at which the gate is opened
Off Attenuation	Sets the level of signal reduction when a channel is not active
Hold Time	Sets the duration for which the channel remains open after the level drops below the gate threshold
Maximum Open Channels	Sets the maximum number of simultaneously active channels
Priority	When selected, this channel gate activates regardless of the number of maximum open channels.
Automix Gain Meter	When enabled, changes gain meters to display automix gating in real time. Channels that gate open will display more gain than channels that are closed (attenuated) in the mix.

Automix Modes

Classic

Classic mode emulates the Shure SCM820 automixer (in its default settings). It is renowned for fast-acting, seamless channel gating and consistent perceived ambient sound levels. Off-attenuation in this mode is fixed at -12 dB per channel, regardless of the number of open channels.

Smooth

In Smooth mode, Off-attenuation settings for each channel are scaled, depending on the number of open channels. The scaled gain structure helps to reduce noise when there is a high channel count. When fewer channels are used, the lower off-attenuation provides transparent gating.

Number of channels enabled	Off-attenuation (dB)
2	-3.0
3	-4.8
4	-6

Custom

Custom mode provides control over all automixing parameters. This mode is useful when adjustments must be made to one of the preset modes to fit a particular application. If parameters are changed in smooth or classic mode, custom mode automatically activates.

Manual

Manual mode sums all active tracks and sends the summed signal over a single Dante output. This provides the option to route the signal for reinforcement or recording, without enabling automixing. The settings from the faders in the standard monitoring view apply to the summed output.

Using a Third-Party Control System

This device receives logic commands over the network. Many parameters controlled through Designer can be controlled using a third-party control system, using the appropriate command string.

Common applications:

- Mute
- LED color and behavior
- Loading presets
- Adjusting levels

A complete list of command strings is available at:

pubs.shure.com/command-strings/MXA310.

Networking and Dante

Digital Audio Networking

Dante™ digital audio is carried over standard Ethernet and operates using standard Internet Protocols. Dante provides low latency, tight clock synchronization, and high Quality-of-Service (QoS) to provide reliable audio transport to a variety of Dante devices. Dante audio can coexist safely on the same network as IT and control data, or can be configured to use a dedicated network.

Switch and Cable Recommendations for Dante Networking

Switches and cables determine how well your audio network performs. Use high-quality switches and cables to make your audio network more reliable.

Network switches should have:

- Gigabit ports. 10/100 switches may work on small networks, but gigabit switches perform better.
- Power over Ethernet (PoE) or PoE+ ports for any devices that require power
- Management features to provide information about port speed, error counters, and bandwidth used
- Ability to switch off Energy Efficient Ethernet (EEE). EEE (also known as "Green Ethernet") may cause audio dropouts and problems with clock synchronization.
- Diffserv (DSCP) Quality of Service (QoS) with strict priority and 4 queues

Ethernet cables should be:

- Cat5e or better
- Shielded

Compatibility with Dante Domain Manager

This device is compatible with Dante Domain Manager software (DDM). DDM is network management software with user authentication, role-based security, and auditing features for Dante networks and Dante-enabled products.

Considerations for Shure devices controlled by DDM:

- When you add Shure devices to a Dante domain, set the local controller access to Read Write. Otherwise, you won't be able to access to Dante settings, perform a factory reset, or update device firmware.
- If the device and DDM can't communicate over the network for any reason, you won't be able to control Dante settings, perform a factory reset, or update device firmware. When the connection is reestablished, the device follows the policy set for it in the Dante domain.
- If Dante device lock is on, DDM is offline, or the configuration of the device is set to Prevent, some device settings are disabled. These include: Dante encryption, MXW association, AD4 Dante browse and Dante cue, and SCM820 linking.

See [Dante Domain Manager's documentation](#) for more information.

QoS (Quality of Service) Settings

QoS settings assign priorities to specific data packets on the network, ensuring reliable audio delivery on larger networks with heavy traffic. This feature is available on most managed network switches. Although not required, assigning QoS settings is recommended.

Note: Coordinate changes with the network administrator to avoid disrupting service.

To assign QoS values, open the switch interface and use the following table to assign Dante-associated queue values.

- Assign the highest possible value (shown as 4 in this example) for time-critical PTP events
- Use descending priority values for each remaining packet.

Dante QoS Priority Values

Priority	Usage	DSCP Label	Hex	Decimal	Binary
High (4)	Time-critical PTP events	CS7	0x38	56	111000
Medium (3)	Audio, PTP	EF	0x2E	46	101110
Low (2)	(reserved)	CS1	0x08	8	001000
None (1)	Other traffic	BestEffort	0x00	0	000000

Note: Switch management may vary by manufacturer and switch type. Consult the manufacturer's product guide for specific configuration details.

For more information on Dante requirements and networking, visit www.audinate.com.

Networking Terminology

PTP (Precision Time Protocol): Used to synchronize clocks on the network

DSCP (Differentiated Services Code Point): Standardized identification method for data used in layer 3 QoS prioritization

Dante Transmit Flows

For each device, there is a limit of **2 transmit flows** and **2 receive flows**. A single flow consists of either a single unicast or multicast transmission, and supports up to 4 audio channels.

- A **unicast flow** is a point-to-point connection between 2 devices, supporting up to 4 channels per flow. To send 8 channels of audio between two devices, 2 unicast flows are required.
- A **multicast flow** is a one-to-many transmission, which supports sending up to 4 channels to multiple receiving devices across the network. To send 8 channels from one device to all others on the network, 2 multicast flows are required.

Networking Best Practices

When connecting Shure devices to a network, use the following best practices:

- Always use a "star" network topology by connecting each device directly to the switch or router.
- Connect all Shure networked devices to the **same network** and set to the **same subnet**.
- Allow all Shure software through the firewall on your computer.
- Use only 1 DHCP server per network. Disable DHCP addressing on additional servers.
- Power on the switch and DHCP server before powering on the Shure devices.
- To expand the network, use multiple switches in a star topology.
- All devices must be at the same firmware revision level.

Network Audio and Shure Control Data

MicroflexAdvance devices transport two types of data over the network: Shure Control and Network Audio.

Shure Control

The Shure Control carries data for the control software operation, firmware updates and 3rd party control systems (AMX, Crestron).

Network Audio

This network carries both the Dante digital audio and the control data for Dante Controller. The network audio requires a wired, gigabit Ethernet connection to operate.

Device IP Settings

Configure IP

Sets IP mode of the selected network interface:

- **Auto (DHCP):** For automatic assignment of IP addresses.
- **Manual (Static):** For Static IP addresses.

IP Settings

View and edit the IP Address, Subnet Mask, and Gateway for each network interface.

MAC Address

The network interface's unique identification.

Configuring IP Settings

IP configurations are managed in Shure Designer software. By default, they are set to Automatic (DHCP) mode. DHCP mode enables the devices to accept IP settings from a DHCP server, or automatically fall back to Link-Local settings when no DHCP is available. IP addresses may also be manually set.

To configure the IP properties, follow these steps:

1. Open device's configuration window.
2. Go to the Settings tab and select Network.
3. Select Auto or Manual. If Auto is used, addresses will be automatically assigned. For Manual setup, follow the instructions on manual configuration.

Manually Assigning Static IP Address

To manually assign IP addresses, follow these steps:

1. Open the device's configuration window in Designer.
2. Go to the Settings tab and select Network.
3. Select Manual as the Configure IP setting.
4. Enter the IP settings.

Setting Latency

Latency is the amount of time for a signal to travel across the system to the outputs of a device. To account for variances in latency time between devices and channels, Dante has a predetermined selection of latency settings. When the same setting is selected, it ensures that all Dante devices on the network are in sync.

These latency values should be used as a starting point. To determine the exact latency to use for your setup, deploy the setup, send Dante audio between your devices, and measure the actual latency in your system using Audinate's Dante Controller software. Then round up to the nearest latency setting available, and use that one.

Use Audinate's Dante Controller software to change latency settings.

Latency Recommendations

Latency Setting	Maximum Number of Switches
0.25 ms	3
0.5 ms (default)	5
1 ms	10
2 ms	10+

Operating the Control Software over Wi-Fi

When operating the web application over Wi-Fi, it's important to set up the wireless router properly for best performance. The system employs several standard-based protocols that rely on multicast. Wi-Fi treats broadcast and multicast packets differently than general packets for backward compatibility reasons. In some cases, the Wi-Fi router will limit the multicast packet transmission rate to a value that is too slow for web application to properly operate.

Wi-Fi routers typically support 802.11b, 802.11a/g, and/or 802.11n standards. By default, many Wi-Fi routers are configured to allow older 802.11b devices to operate over the network. In this configuration, these routers will automatically limit the multicast data rates (or sometimes referred to as 'basic rate', or 'management rate') to 1-2Mbps.

Note: A Wi-Fi connection can only be used for the control software. Network audio cannot be transmitted over Wi-Fi.

Tip: For larger wireless microphone configurations, it's recommended to increase the multicast transmission rate to provide adequate bandwidth.

Important: For best performance, use a Wi-Fi router that does not limit the multicast rate to 1-2 Mbps.

Shure recommends the following Wi-Fi router brands:

- Cisco
- Linksys
- Apple

Packet Bridge

Packet bridge enables an external controller to obtain IP information from the control interface of a Shure device. To access the packet bridge, an external controller must send a query packet over **unicast UDP*** to **port 2203** on the Dante interface of the Shure device.

1. Send a UDP packet with a minimum 1-byte payload .

Note: The maximum accepted payload 140 bytes. Any content is allowed.

2. The Shure device will send a response packet over unicast UDP to the controller, using a destination UDP port identical to the source port of the query packet. The payload of the response packet follows this format:

Bytes	Content
0-3	IP address, as 32-bit unsigned integer in network order
4-7	Subnet mask, as 32-bit unsigned integer in network order
8-13	MAC address, as array of 6 bytes

Note: The Shure device should respond in less than one second on a typical network. If there is no response, try sending the query again after verifying the destination IP address and port number.

***UDP:** User Datagram Protocol

IP Ports and Protocols

Shure Control

Port	TCP/UDP	Protocol	Description	Factory Default
21	tcp	FTP	Required for firmware updates (otherwise closed)	Closed
22	tcp	SSH	Secure Shell Interface	Closed
23	tcp	Telnet	Not supported	Closed
68	udp	DHCP	Dynamic Host Configuration Protocol	Open
80*	tcp	HTTP	Required to launch embedded web server	Open
443	tcp	HTTPS	Not supported	Closed
161	tcp	SNMP	Not supported	Closed
162	tcp	SNMP	Not supported	Closed

Port	TCP/UDP	Protocol	Description	Factory Default
2202	tcp	ASCII	Required for 3rd party control strings	Open
5353	udp	mDNS [†]	Required for device discovery	Open
5568	udp	SDT [†]	Required for inter-device communication	Open
8023	tcp	Telnet	Debug console interface	Closed
8180	tcp	HTML	Required for web application	Open
8427	udp	Multicast SLP [†]	Required for inter-device communication	Open
64000	tcp	Telnet	Required for Shure firmware update	Open

Dante Audio & Controller

Port	TCP/UDP	Protocol	Description
162	udp	SNMP	Used by Dante
[319-320]*	udp	PTP [†]	Dante clocking
2203	udp	Custom	Required for packet bridge
4321, 14336-14600	udp	Dante	Dante audio
[4440, 4444, 4455]*	udp	Dante	Dante audio routing
5353	udp	mDNS [†]	Used by Dante
[8700-8706, 8800]*	udp	Dante	Dante Control and Monitoring
8751	udp	Dante	Dante Controller
16000-65536	udp	Dante	Used by Dante

*These ports must be open on the PC or control system to access the device through a firewall.

[†]These protocols require multicast. Ensure multicast has been correctly configured for your network.

Troubleshooting

Problem	Solution
Software lags in Google Chrome browser	Problem is browser-related. Turn off hardware acceleration option in Chrome.

Problem	Solution
Sound quality is muffled or hollow	<p>Check that channels have been aimed to the desired area. Make sure channels are not accidentally muted.</p> <p>Use equalizer to adjust frequency response on a single channel or on the automix channel. See the equalizer applications for the appropriate use.</p>
Microphone does not show up in device discovery	<p>Ensure the devices are powered</p> <p>Ensure PC and equipment are on the same network and set to the same subnet</p> <p>Turn off other network interfaces not used to connect to the device (including WiFi)</p> <p>Check that DHCP server is functioning (if applicable)</p> <p>Reset the device if necessary</p>
Audio is not present or is quiet/distorted	<p>Check cables</p> <p>Verify that channels are not muted</p> <p>Make sure channels are aimed in the right direction, with the intended polar pattern.</p> <p>Check that fader levels are not set too low</p> <p>If using automixing, check the settings to ensure channels are gating on/off properly</p>
No lights	<p>Check if No Lights Mode is enabled, or if any Light Ring settings are turned off.</p>
Automixing is disabled or is missing a channel	<p>Automix is automatically disabled when you turn on toroid</p> <p>Omnidirectional channels are not sent to the automix channel</p>
Microphone does not power on	<p>The network switch must supply Power over Ethernet. Otherwise, a PoE injector must be used</p> <p>Check network cables and connections</p>

Important Product Information

The equipment is intended to be used in professional audio applications.

Note: This device is not intended to be connected directly to a public internet network.

EMC conformance to Environment E2: Commercial and Light Industrial. Testing is based on the use of supplied and recommended cable types. The use of other than shielded (screened) cable types may degrade EMC performance.

Changes or modifications not expressly approved by Shure Incorporated could void your authority to operate this equipment.

Industry Canada ICES-003 Compliance Label: CAN ICES-3 (B)/NMB-3(B)

Authorized under the verification provision of FCC Part 15B.

Please follow your regional recycling scheme for batteries, packaging, and electronic waste.

Information to the user

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference.
2. This device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and the receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

The CE Declaration of Conformity can be obtained from: www.shure.com/europe/compliance

Authorized European representative:

Shure Europe GmbH

Headquarters Europe, Middle East & Africa

Department: EMEA Approval

Jakob-Dieffenbacher-Str. 12

75031 Eppingen, Germany

Phone: +49-7262-92 49 0

Fax: +49-7262-92 49 11 4

Email: info@shure.de

This product meets the Essential Requirements of all relevant European directives and is eligible for CE marking.

The CE Declaration of Conformity can be obtained from Shure Incorporated or any of its European representatives. For contact information please visit www.shure.com

What's in the Box

Cable-exit plug (black)	65A29429
Cable-exit plug (white)	65B29429
Cable-exit plug (silver)	65C29429
Mounting tube wing nut	65A27351
Mounting tube	31A2165
Rubber Isolation Ring	66A405
Nylon cable ties (4)	80A583

Optional Accessories

Flush mounting tray kit (aluminum)	A310AL-FM
Flush mounting tray kit (black)	A310B-FM

Specifications

General

Polar Pattern

All channels independently adjustable

Cardioid, Hypercardioid, Supercardioid, Toroid, Omnidirectional, Bidirectional

Connector Type

RJ45

Power Requirements

Power over Ethernet (PoE), Class 0

Power Consumption

4W, maximum

Weight

362 g (0.8 lbs)

Dimensions

H x W x D

3.6 x 13.4 x 13.4 cm (1.4 x 5.3 x 5.3 in.)

control application

HTML5 Browser-based

Operating Temperature Range

−6.7°C (20°F) to 40°C (104°F)

Storage Temperature Range

−29°C (−20°F) to 74°C (165°F)

Audio

Frequency Response

100 to 20,000 Hz

AES67 or Dante Digital Output

Channel Count	5 total channels (4 independent transmit channels, 1 Automatic mixing transmit channel)
Sampling Rate	48 kHz
Bit Depth	24

Sensitivity

at 1 kHz, , -15 dB Gain Setting

-21 dBFS/Pa

Maximum SPL

1 kHz at 1% THD, -15 dB Gain Setting

115.2 dB SPL

Signal-To-Noise Ratio

Ref. 94 dB SPL at 1 kHz, -15 dB Gain Setting

Cardioid	75 dB
Toroid	67 dB

Latency

Not including Dante latency

<1 ms

Self Noise

-15 dB Gain Setting

Cardioid	19.2 dB SPL-A
Toroid	26.8 dB SPL-A

Dynamic Range

-15 dB Gain Setting

Cardioid	96 dB
Toroid	90 dB SPL

Built-in Digital Signal Processing

Per Channel	Equalizer (4-band Parametric) , Mute, Gain (140 dB range)
System	Automatic mixing, Low-Cut Filter (-12 dB/octave @150 Hz)

Networking

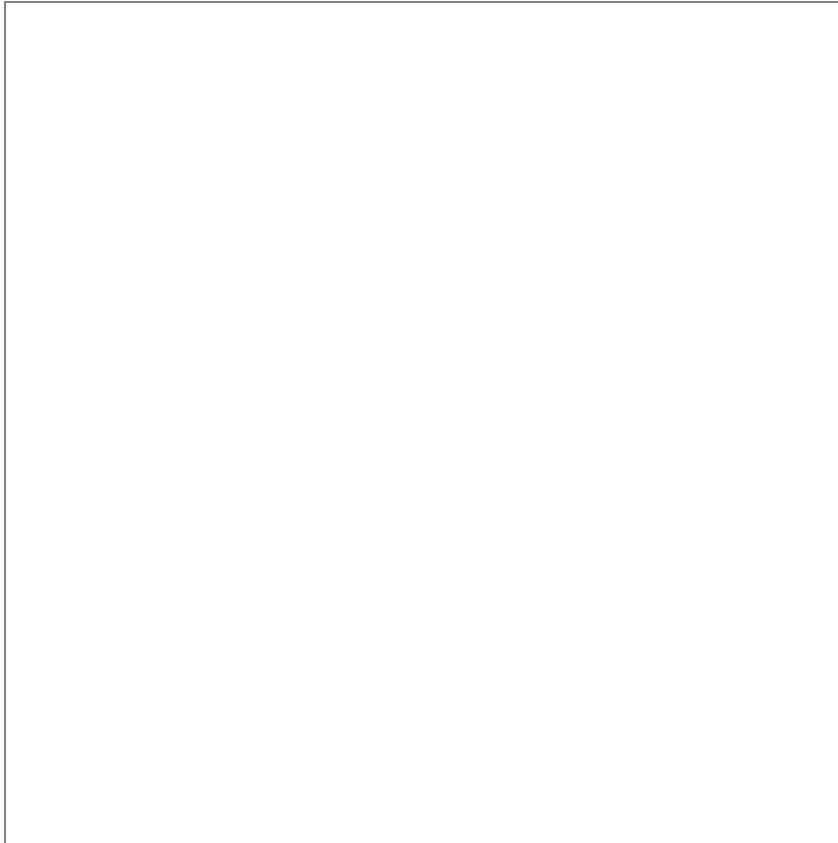
Cable Requirements

Cat 5e or higher (shielded cable recommended)

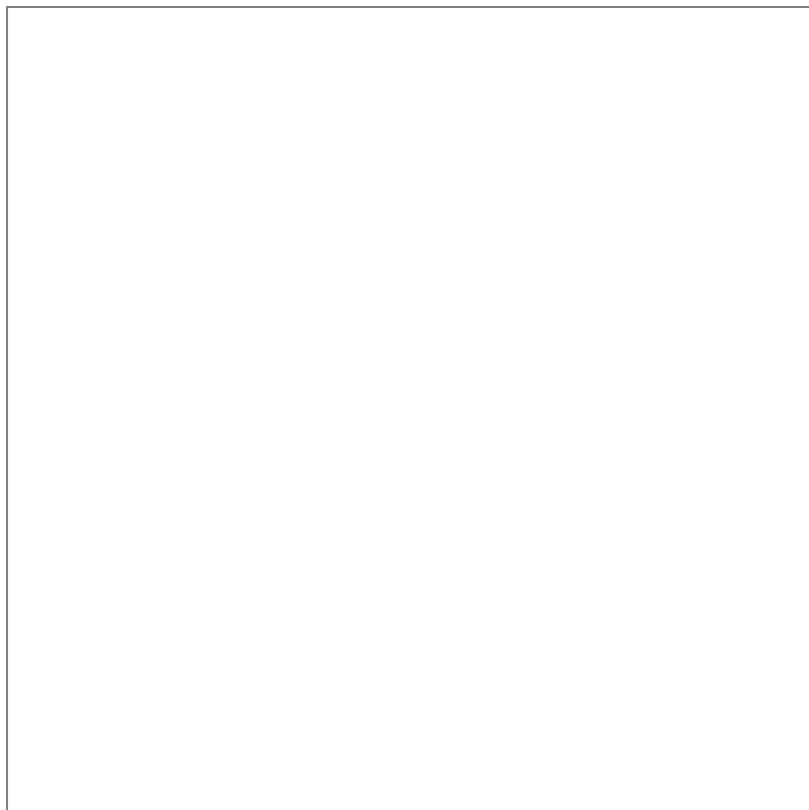
Frequency Response

Frequency response measured from a distance of 2 feet (61 cm).

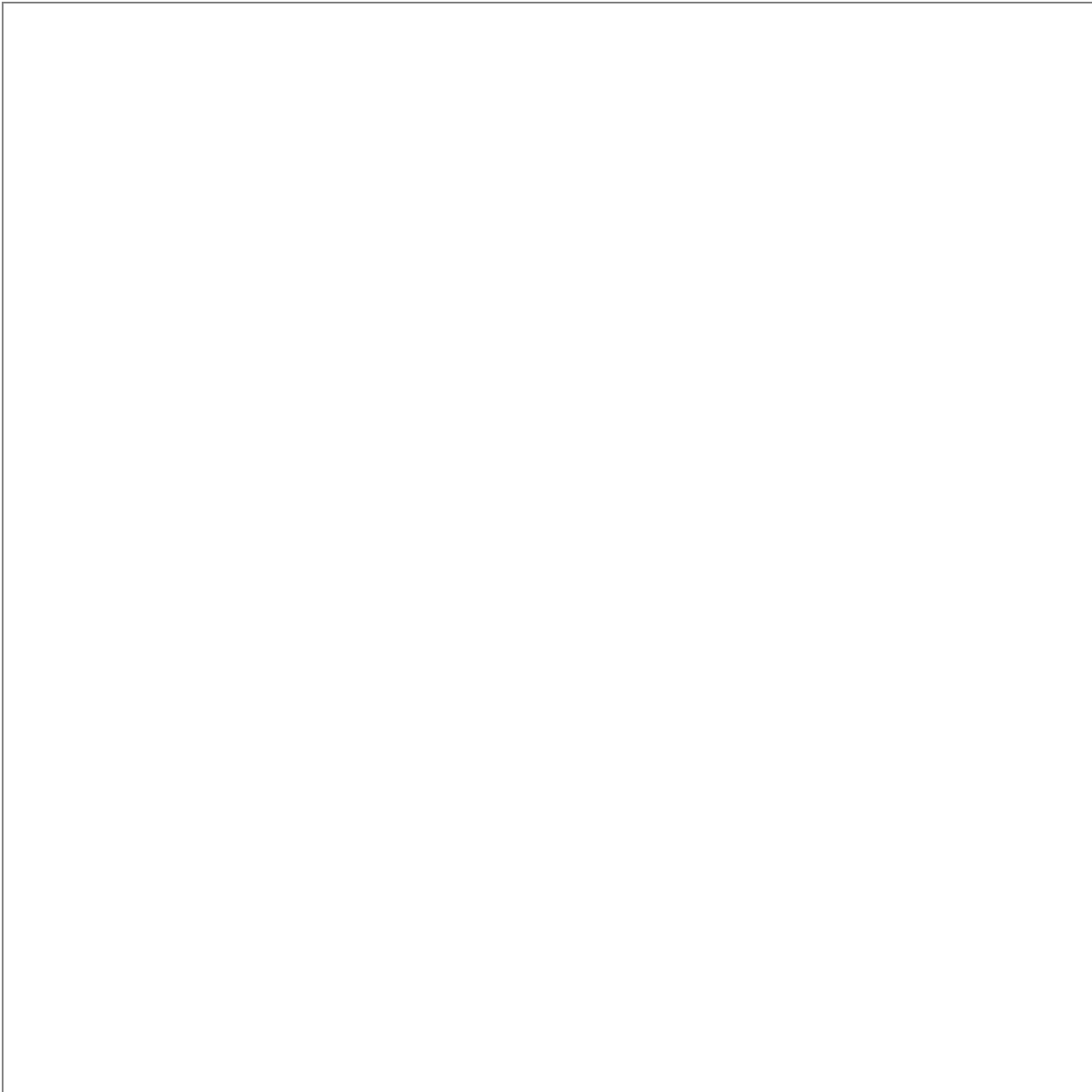




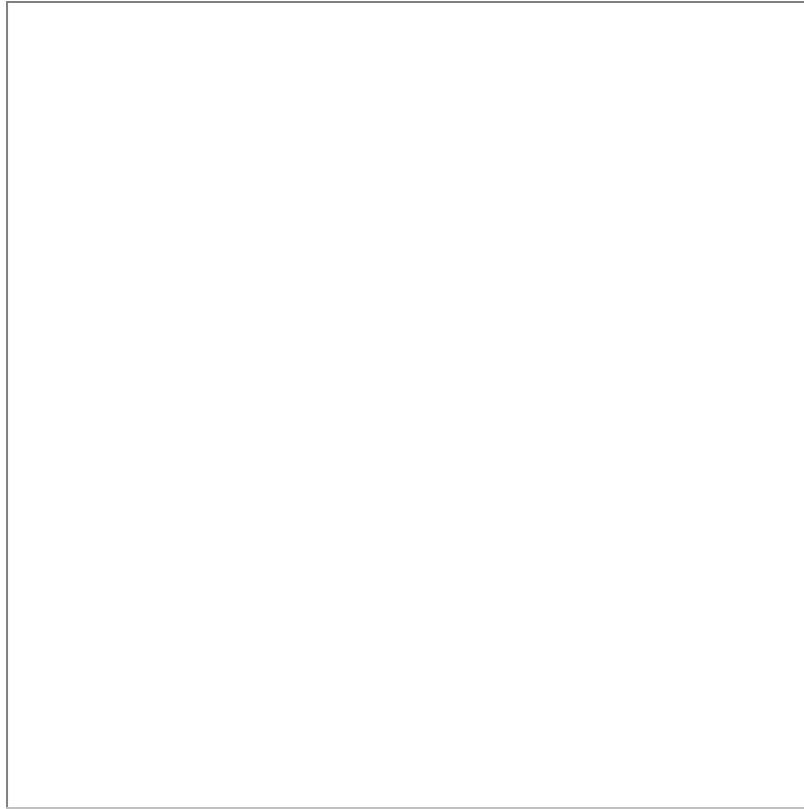
Hypercardioid



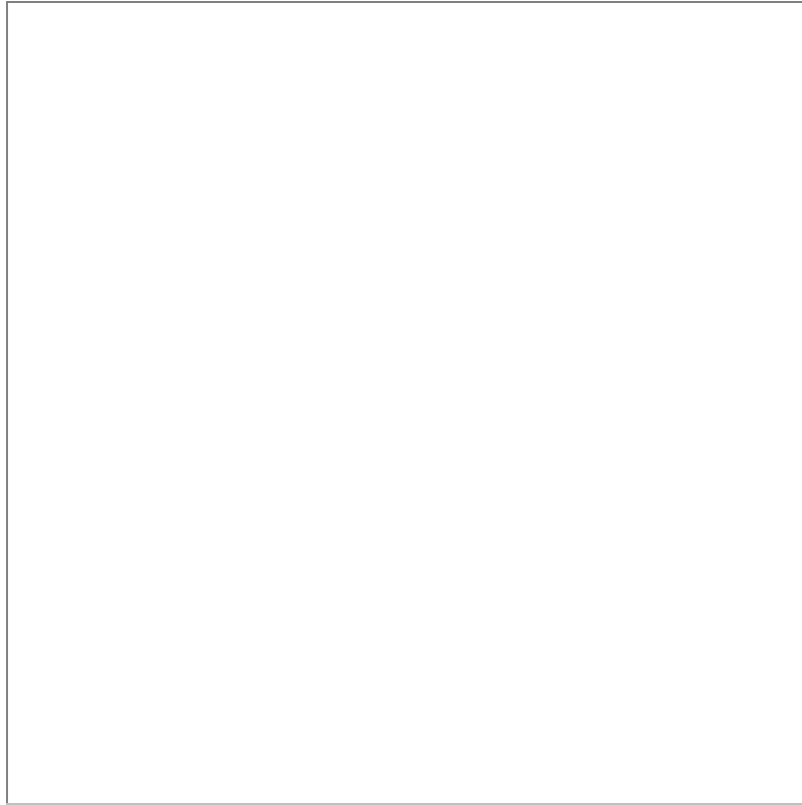
Supercardioid



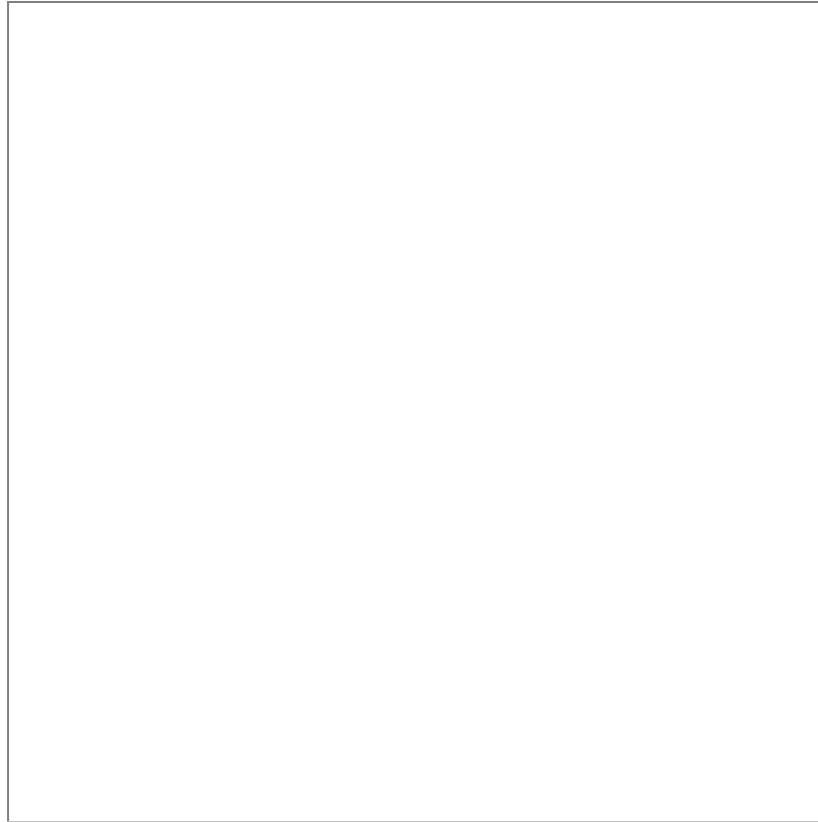
Toroid



Omnidirectional



Cardioid



Bidirectional

Contact Customer Support

Didn't find what you need? [Contact our customer support](#) to get help.