Main 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 IntelliMix® DSP suite includes Steerable Coverage™ , with selectable polar patterns on four independent channels to capture participant audio. The innovative new toroid polar pattern delivers 360° coverage, while rejecting sound from directly above the microphone. Browser-based control software provides an intuitive user interface for microphone attributes, including channel configuration, automatic mix settings, and preset templates. 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
- IntelliMix® 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

- Intuitive software interface provides comprehensive microphone and pattern control
- Includes templates to speed initial set-up and 10 customizable presets to import or export configurations between multiple microphones

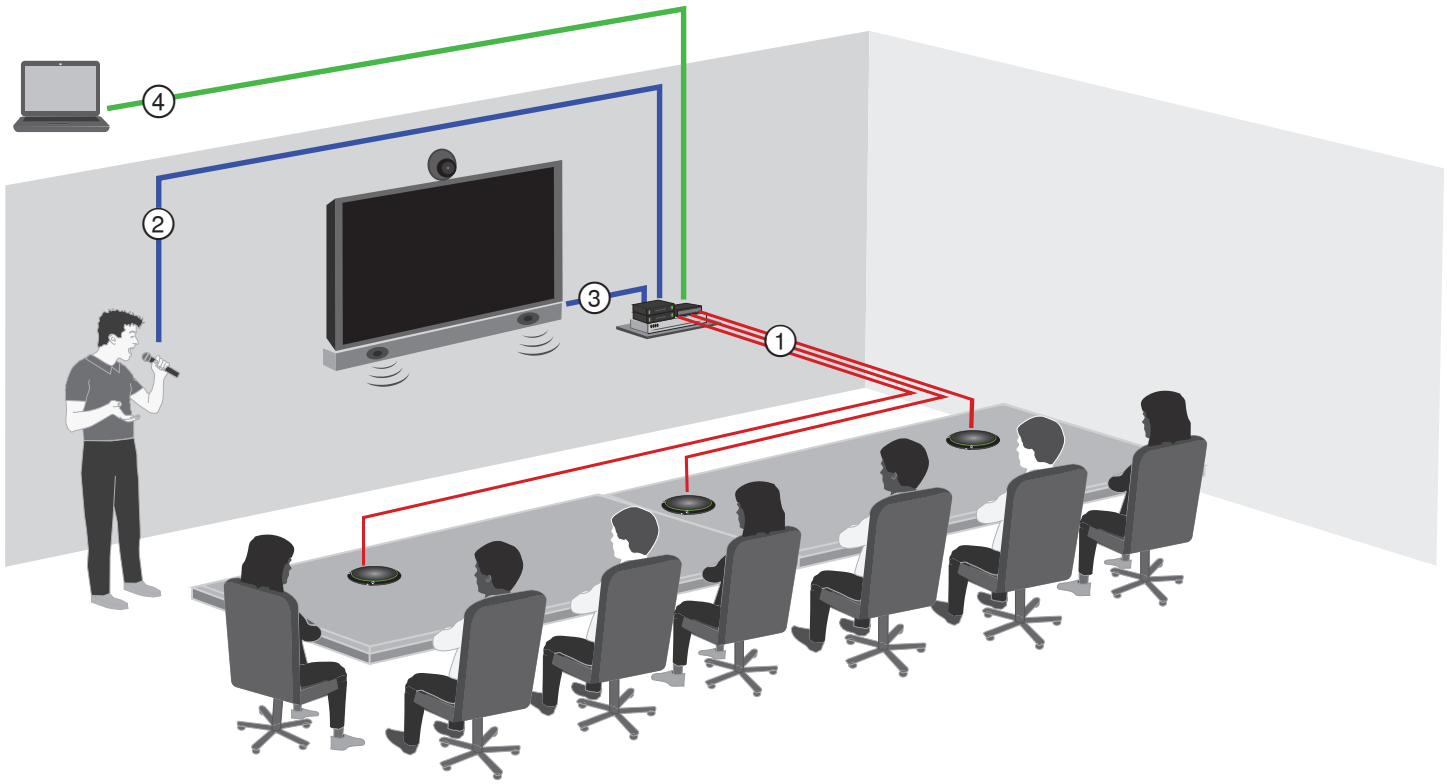
Network Connectivity

- 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 the Shure browser-based interface provides control over the following:

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.

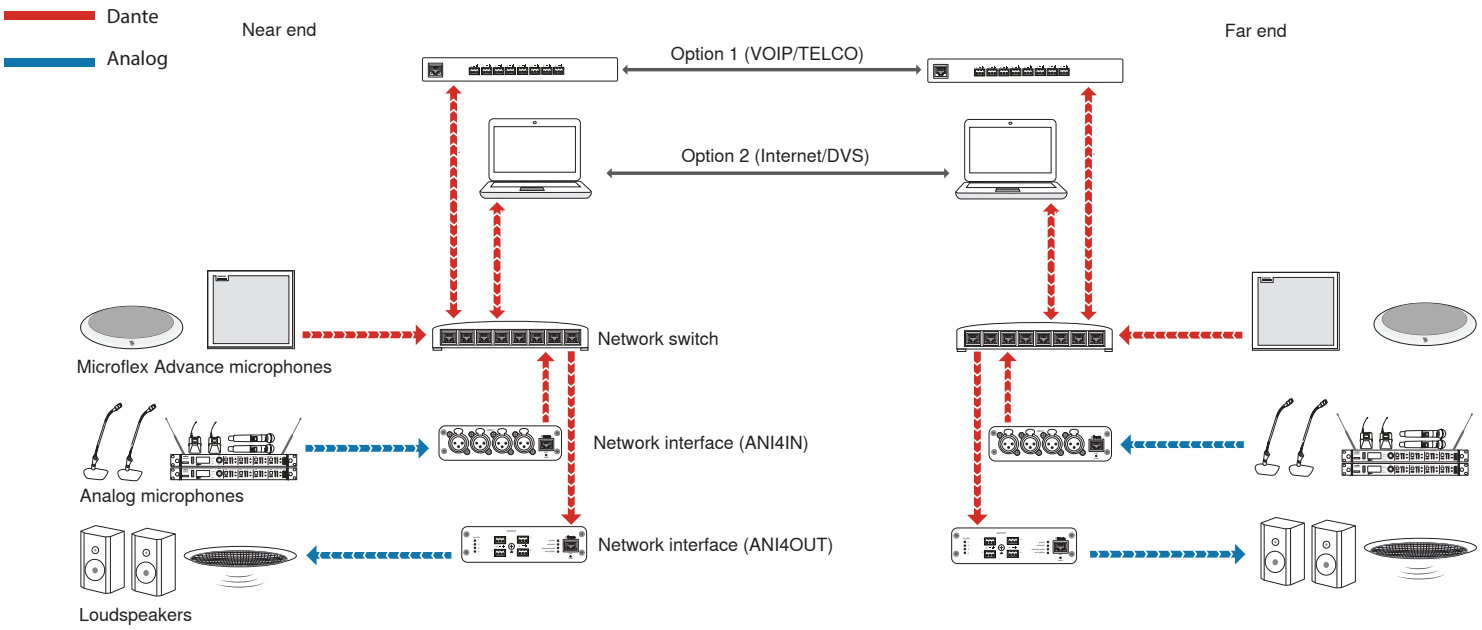
Shure components shown in this diagram:

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

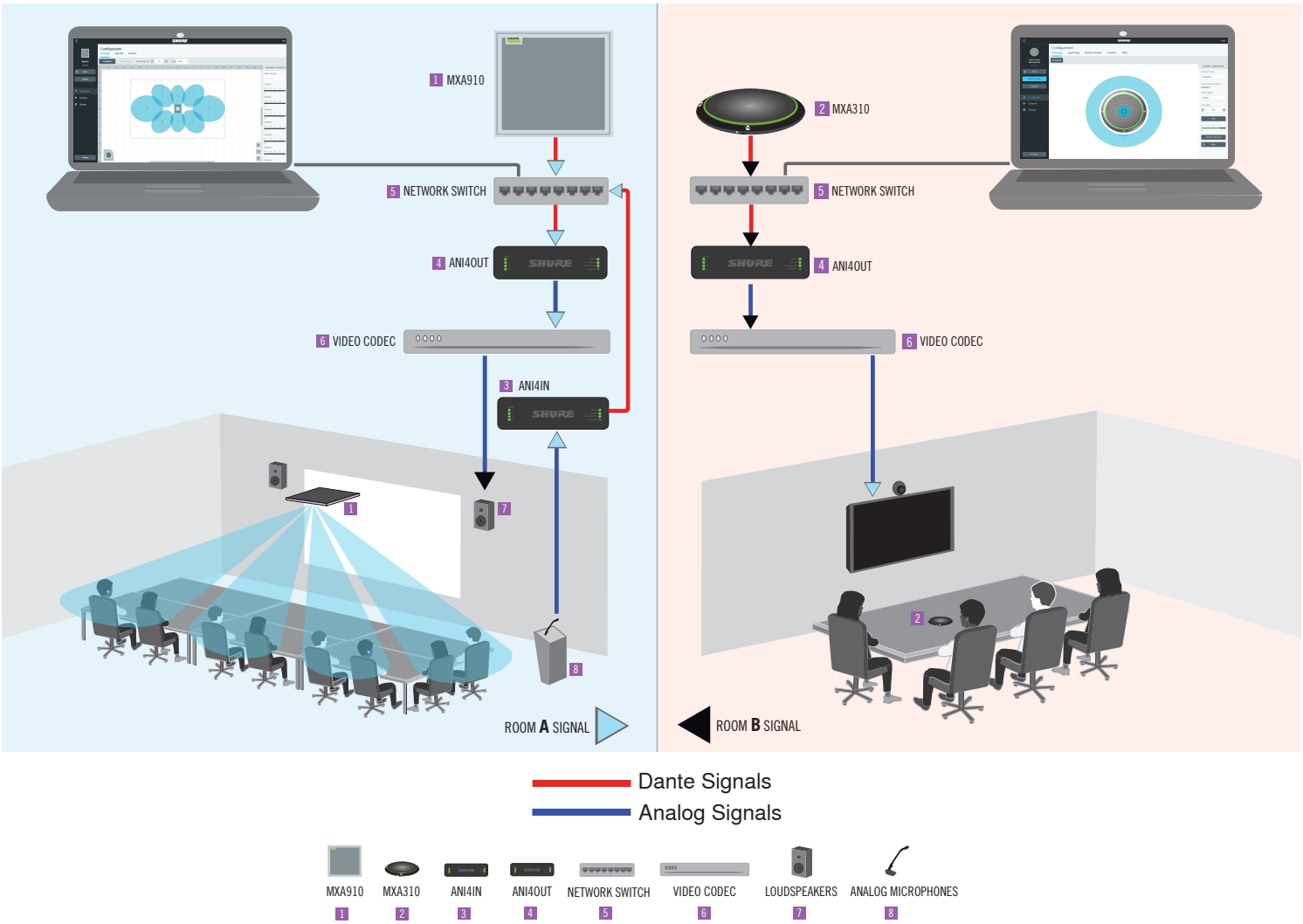
Audio Network Interfaces
The interfaces are used to connect analog devices such as loudspeakers and analog microphones to the network.

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.

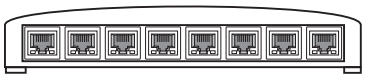
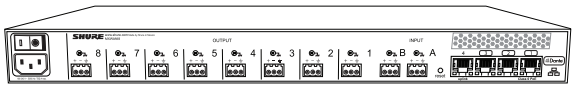
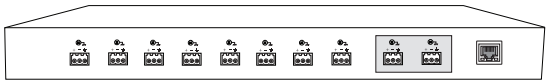
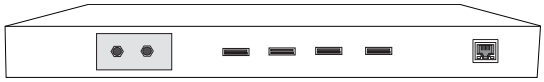
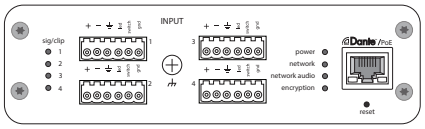
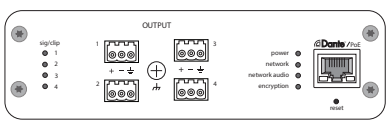
Expanded Control for Analog Devices

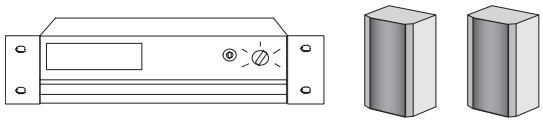
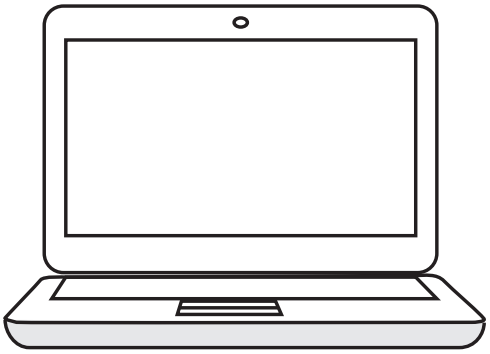
Analog devices that are connected to the network through a Shure network interface (ANI4IN/ANI4OUT) benefit from additional remote control: Volume levels, equalization, and signal routing are managed through the web application. For example, adjusting loudspeaker volume or muting a wired microphone, which would normally be done from the hardware, can now be controlled remotely over the network.

Dante™ Signal Routing

Signal routing between devices is managed through Dante Controller software, provided by Audinate™ .

Hardware Component Index

Network Switch	
	The network switch provides central connectivity for all networked components. Audio from any networked Shure microphones that are connected to the switch can be routed to any Dante™ -enabled device. The switch sends and receives audio and control data, while simultaneously powering the microphones and audio network interfaces through PoE (Power over Ethernet). See the network switch requirements for additional details. **Power over Ethernet (PoE) Requirements:** All Shure components included in these scenarios **require Power over Ethernet** (class 0). If not provided through the network switch, a PoE injector is required to power the devices.
Shure Microflex Wireless Audio Network Interface (MXWANI)	
	The Microflex Wireless Audio Network Interface (MXWANI) is a digital-to-analog breakout box with a built-in gigabit network switch. It converts digital audio from the network into analog signals for signal processing or amplification, and provides PoE over one network port to power a device. For details, refer to the Microflex Wireless user guide, available at www.shure.com.
Audio Processor	
	The audio processor sends and receives audio through a VOIP server or a standard phone line. They also provide digital signal processing, such as acoustic echo cancellation. **Dante™ -enabled** Processors that support Dante™ connect directly to the network switch to receive audio from Microflex Advance microphones. **Analog** When using an analog processor, a converter (such as the Shure ANI4OUT or MXWANI) is required to deliver the analog audio from Microflex Advance microphones to the processor.
Video Codec	
	Like the audio processor, the codec sends and receives audio signals alongside video signals between the near end and the far end. Audio from the near end must connect to the audio input on the video codec, which is typically a stereo analog connection. The Shure ANI4OUT Audio Network Interface converts the audio to an analog signal for connecting to a codec.
Shure ANI4IN Audio Network Interface (Analog-to-Dante Converter)	
	The Shure ANI4IN Audio Network Interface converts 4 channels of analog audio into independent digital audio channels on a Dante™ network. Adjustable gain and +48V phantom power deliver the flexibility to support line, auxiliary, and microphone-level devices. For networked conferencing systems, the Audio Network Interface provides a simple way to connect previously installed analog equipment onto the audio network, such as wireless microphones for presenters. The web application gives technicians and administrators control over channel levels and settings from any computer connected to the same network.
Shure ANI4OUT Audio Network Interface (Dante-to-Analog Converter)	
	The Shure ANI4OUT Audio Network Interface converts 4 channels of Dante™ digital audio into discrete analog signals. Available in both XLR and block connector versions, each box uses a single network cable to receive audio and power using Power over Ethernet (PoE). The web application gives technicians and administrators control over channel levels and settings from any computer connected to the same network.
Amplifiers and Loudspeakers	

	Audio from the far end is routed to local loudspeakers. Dante™ -enabled speakers or amplifiers connect directly to the network switch, while analog systems require an audio network interface to receive networked audio.
Computers and Control Systems	
	**Browser-Based Web Application** A computer connected to the network provides control of Shure networked components through the web application for each device. **Dante™ Software** A computer running Dante™ Virtual Soundcard, Dante™ Controller, and web conferencing software is used to send and receive audio between the near end and far end. **Control Systems (AMX, Crestron, etc.)** If using a third-party control system, Microflex Advance microphones send and receive commands over Ethernet. If an analog logic signal must be sent over the network, the Shure ANI4IN Audio Network Interface receives analog logic signals and converts them into Ethernet control strings.

System Scenarios

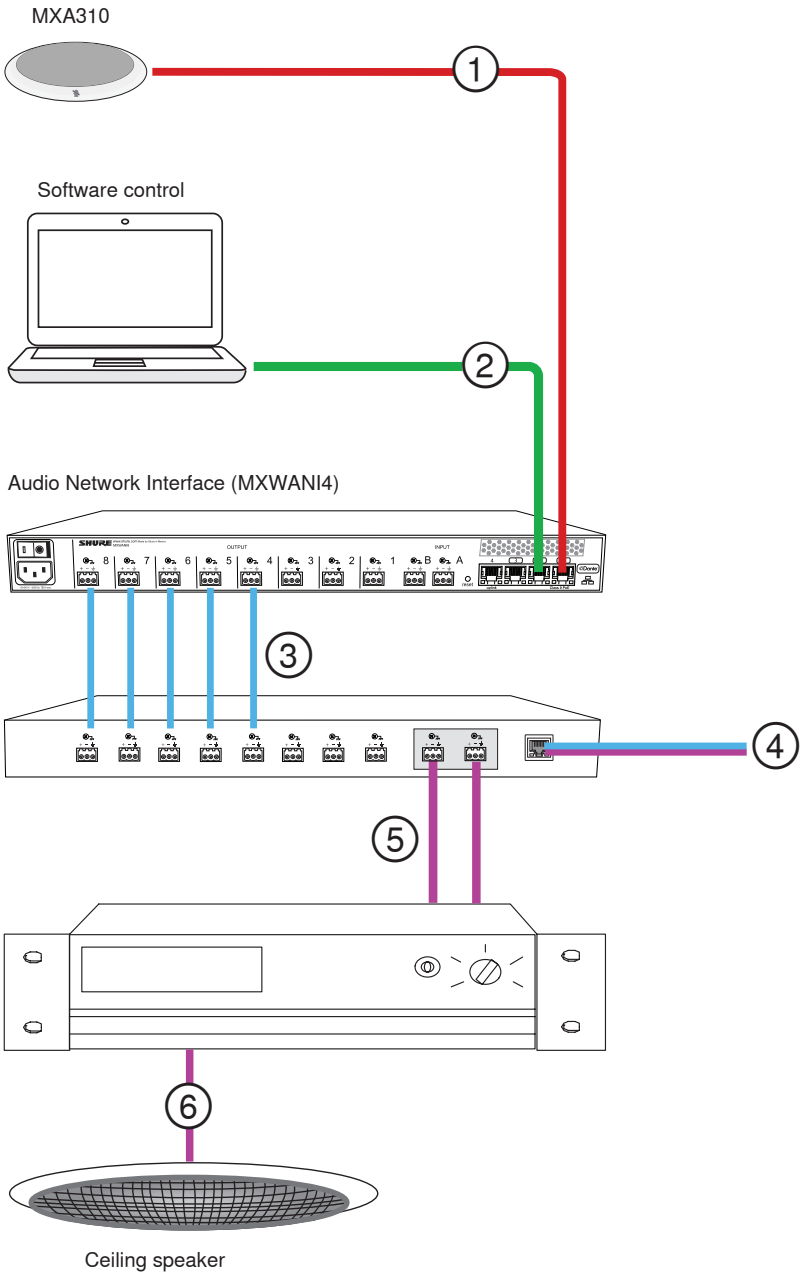
The following diagrams show a selection of common conferencing room systems. Use them as a reference when planning hardware and cable requirements for an installation. Each diagram includes:

- Signal flow and connections
- Required hardware
- Component roles

Power Over Ethernet and Hardware Requirements

All Shure components included in these scenarios **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).

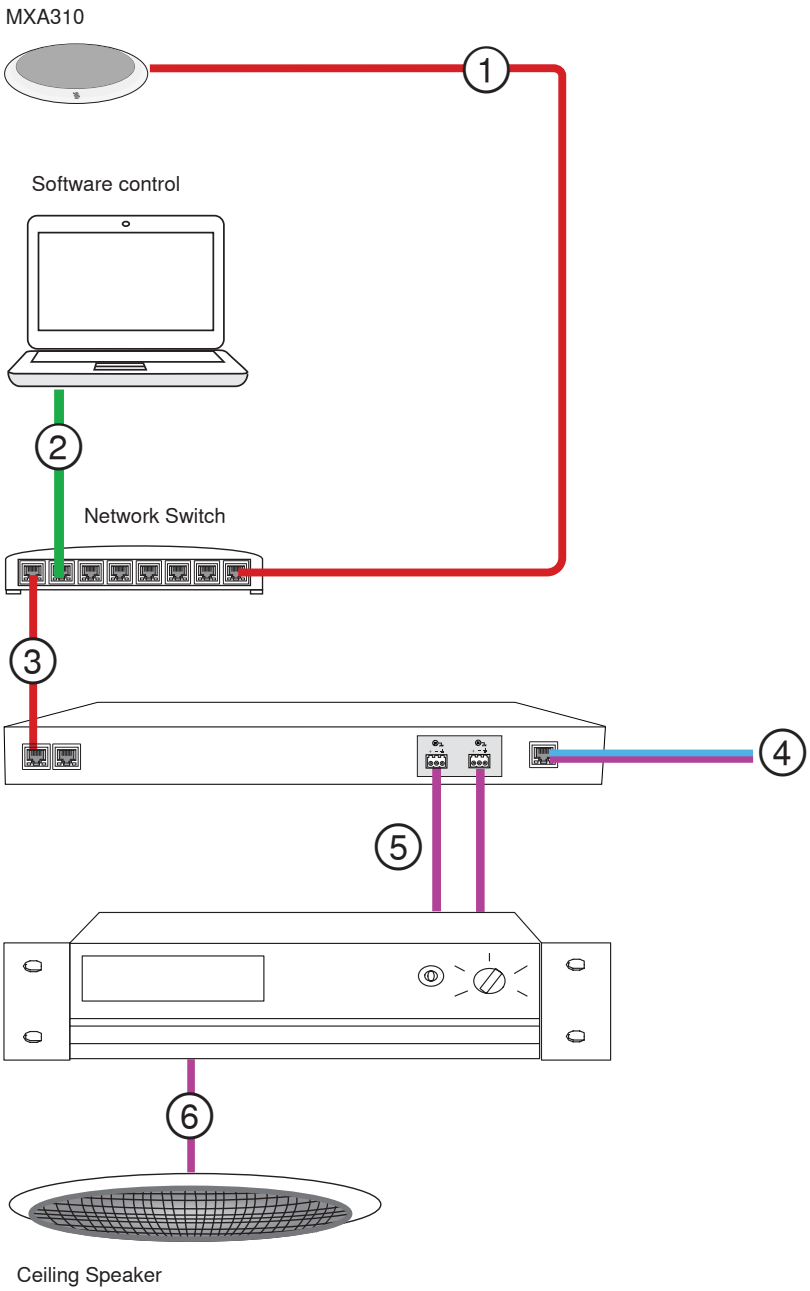
④ Connection to far end

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)

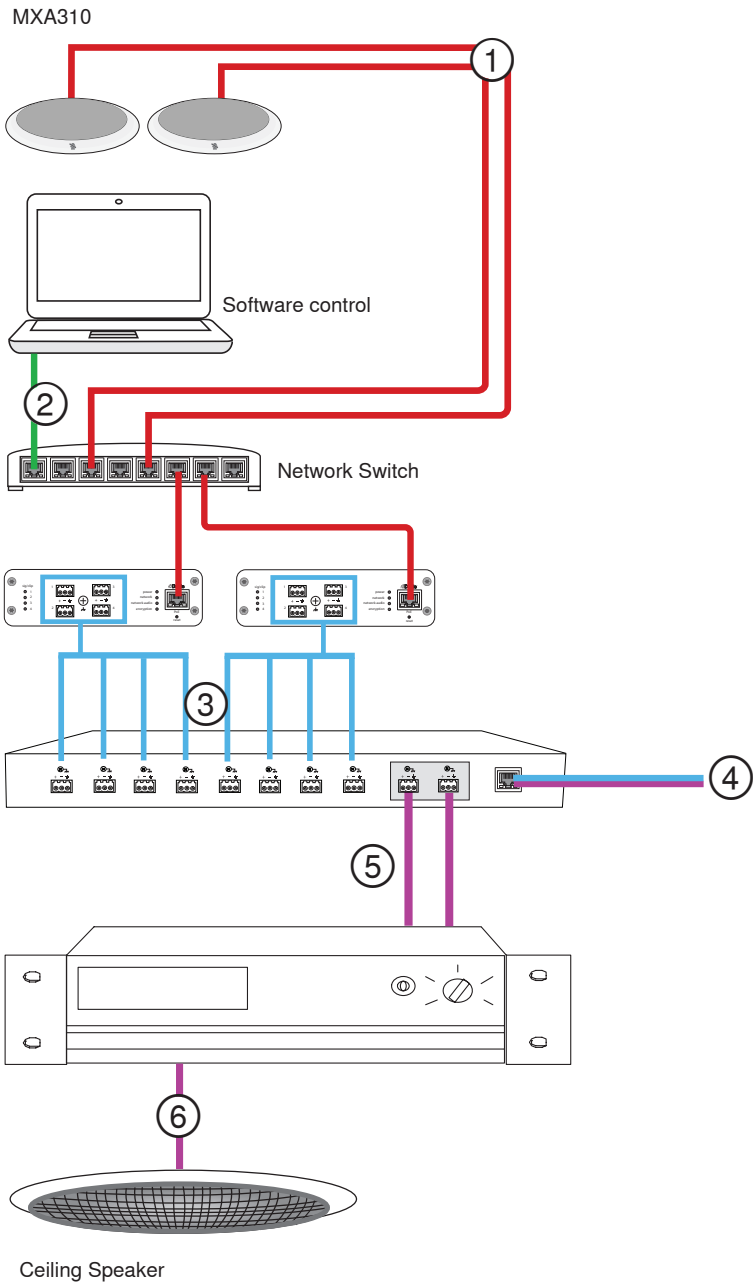
- Digital-to-analog conversion to deliver Dante™ audio over an analog (VOIP or telephone line) output.
- Analog-to-digital conversion to deliver analog audio from the far end onto the Dante™ network.

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.

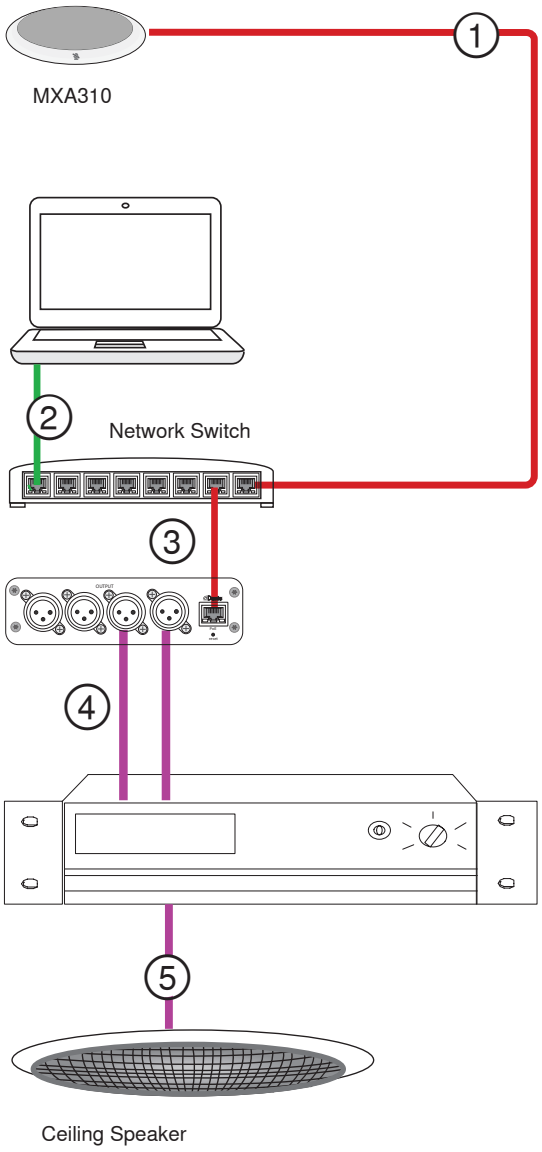
To a processing device: Route the ANI4OUT outputs to the processing device inputs to provide digital signal processing (acoustic echo cancellation).

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 the controller software 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.

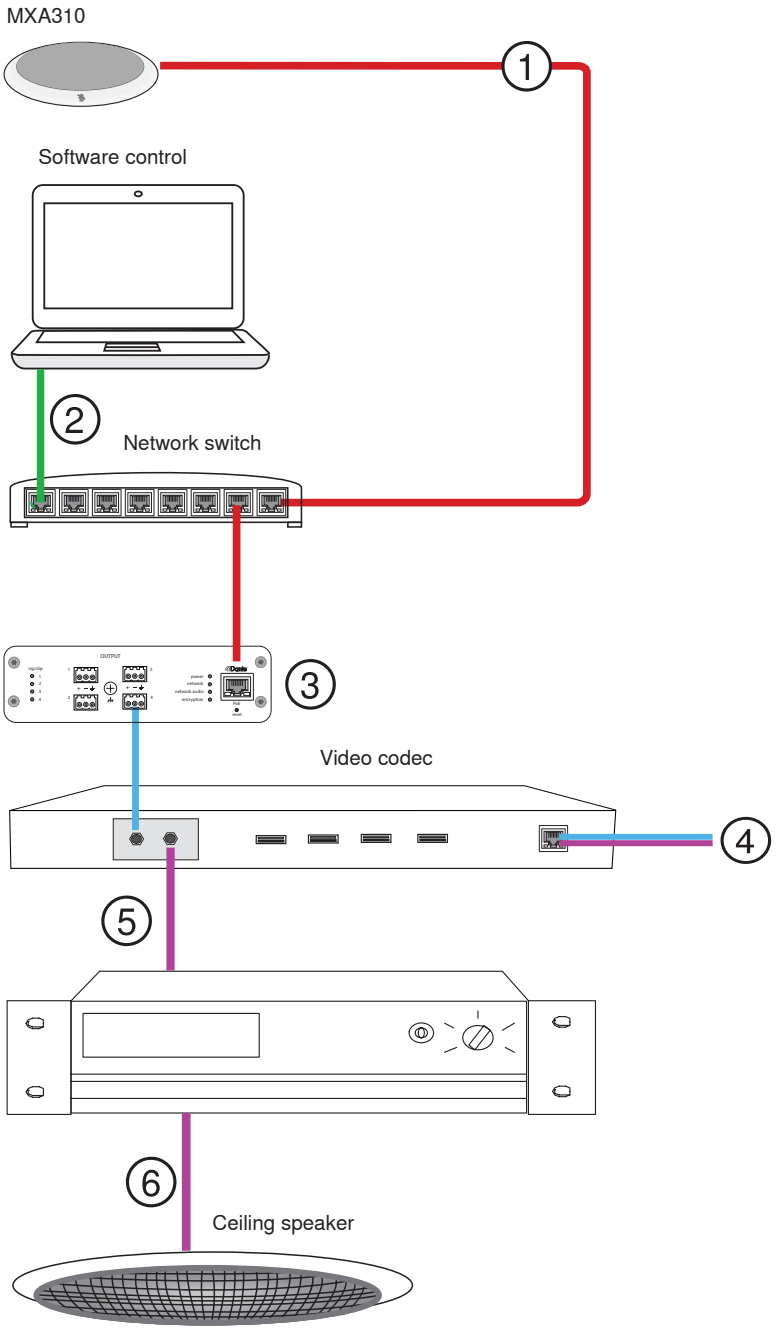
③ Network switch to ANI4OUT

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

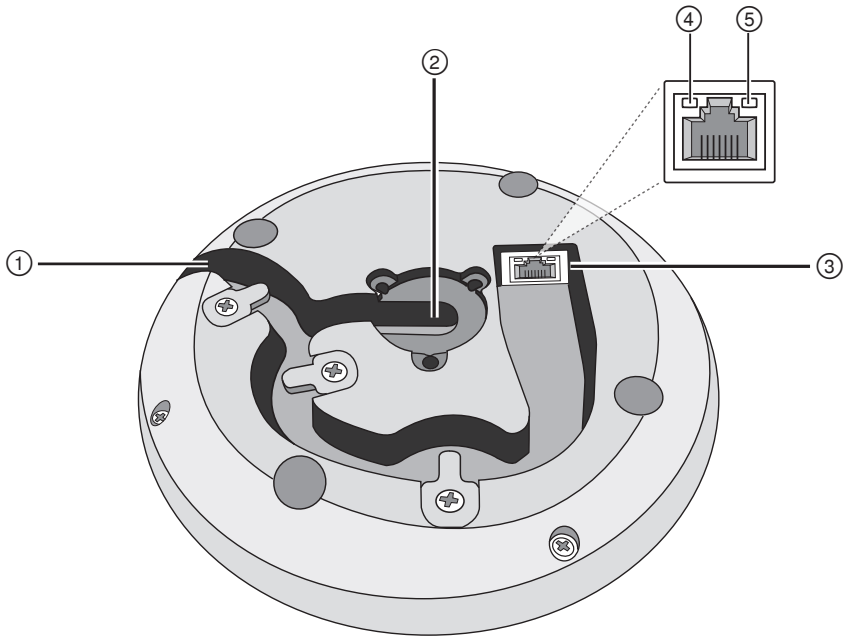
④ Video codec connection to far end
Connect the codec to the appropriate network to connect with the far end.

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.

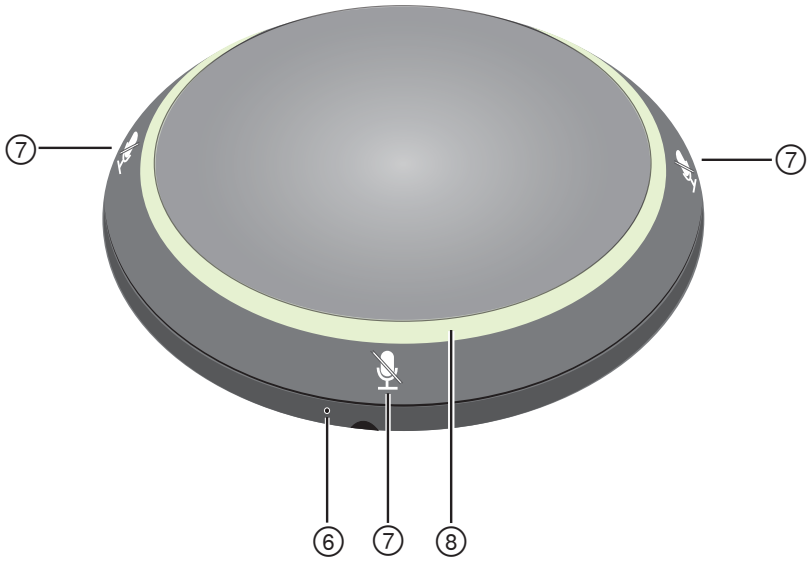
② Bottom Cable Exit
Guide the cable under the tabs and through the bottom exit for permanent table installations.

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



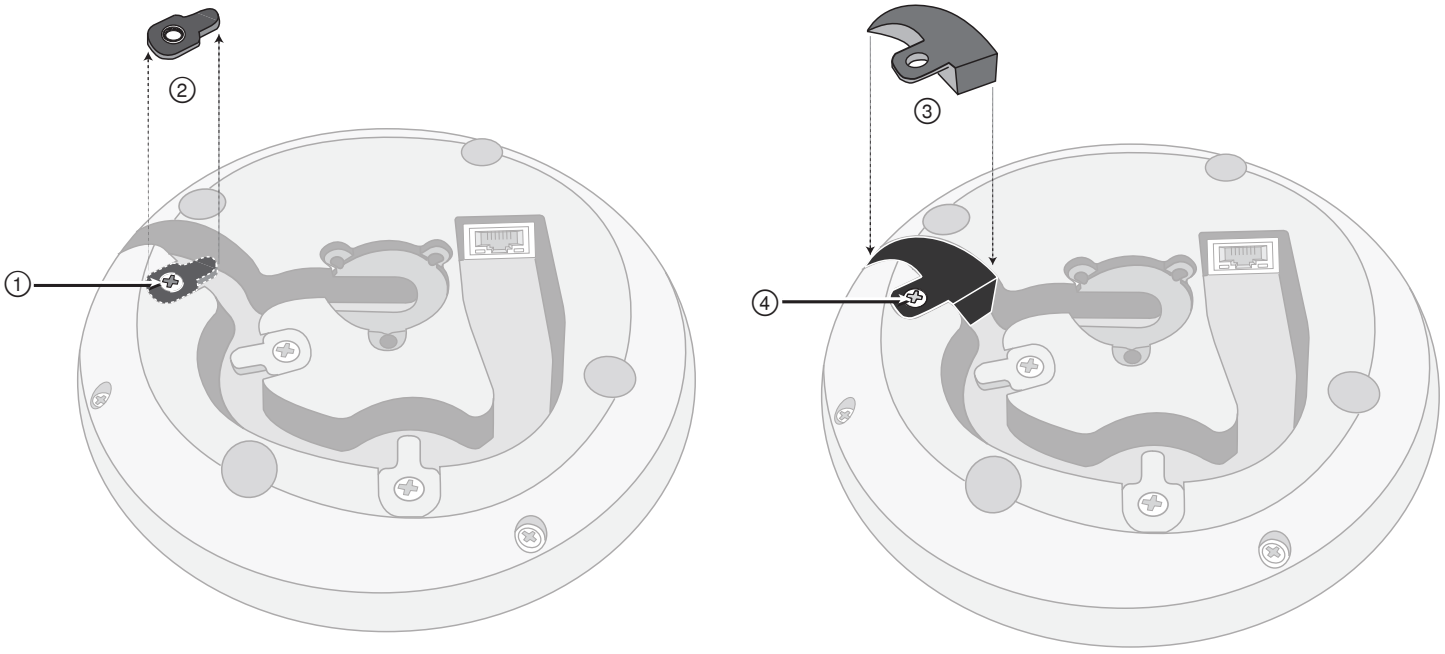
⑥ Reset Button
Use a paperclip or similar tool to push the reset button.

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.

Installing the Cable-Exit Plug

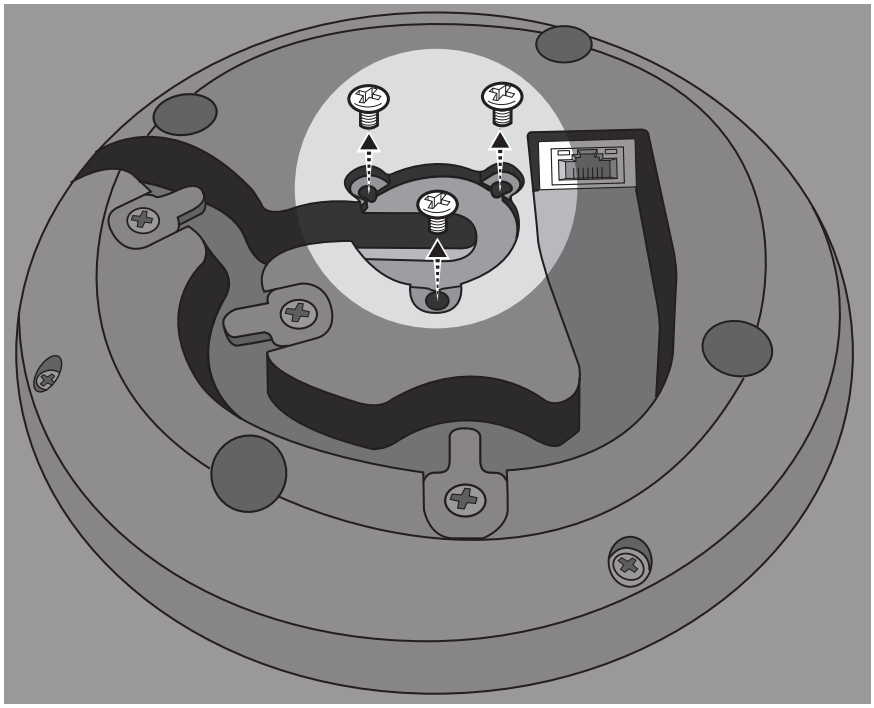
The plug covers the cable exit for permanent installations in which the cable is routed down through a table.



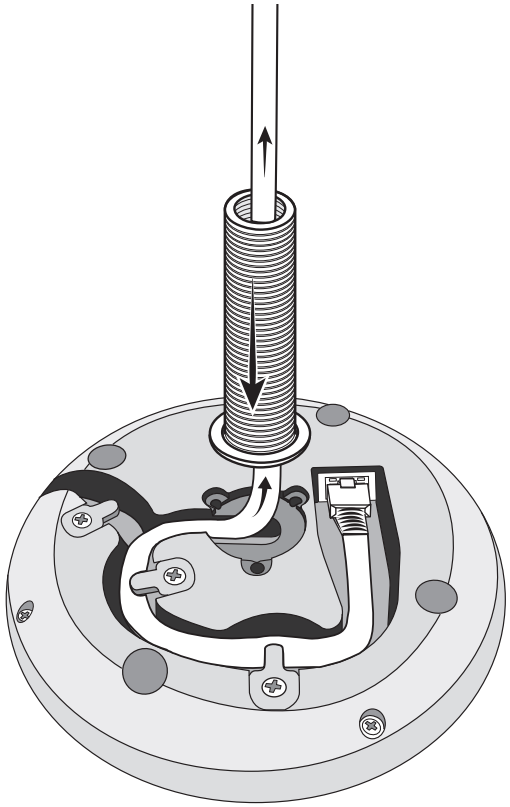
1. Remove the screw that holds in the cable-retaining tab closest to the cable exit
2. Remove the cable-retaining tab
3. Insert the plug
4. Replace the screw to secure the plug

Permanent Table Installation

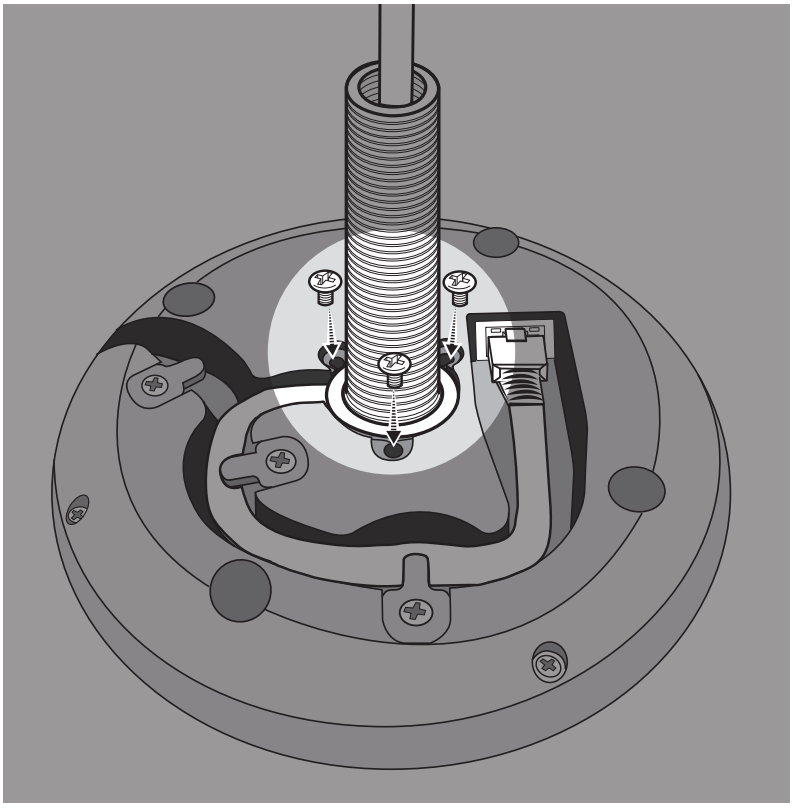
1. Remove the 3 screws located in the center on the bottom of the microphone



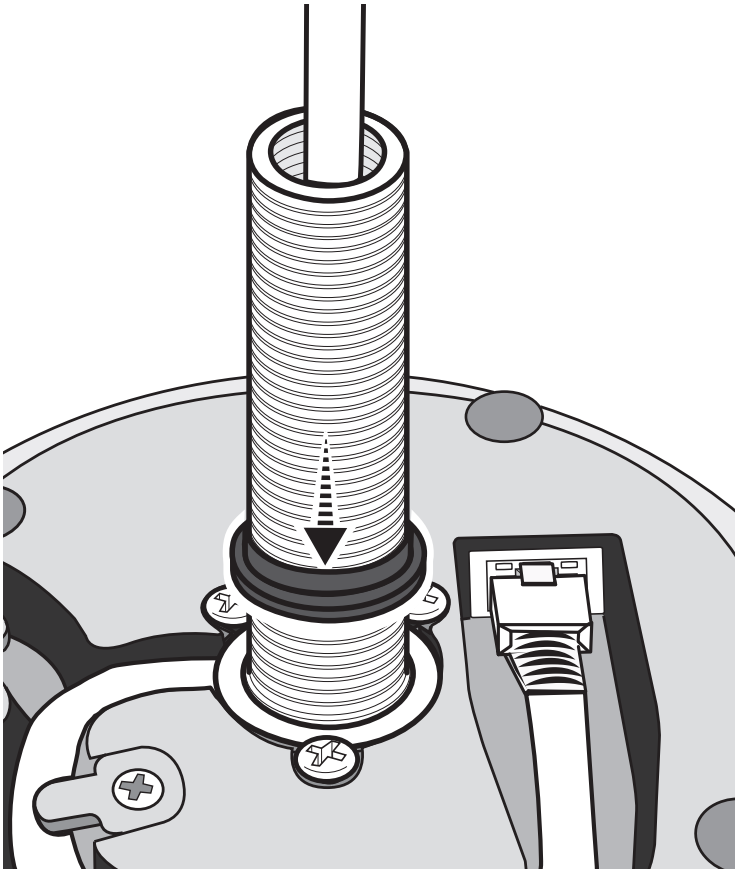
2. Plug a network cable into the microphone and guide it through the center exit path. When the cable is secured, guide it through the tube.
   Note: If necessary, remove the retaining tabs to install thicker cable. Replace them after the cable is installed.



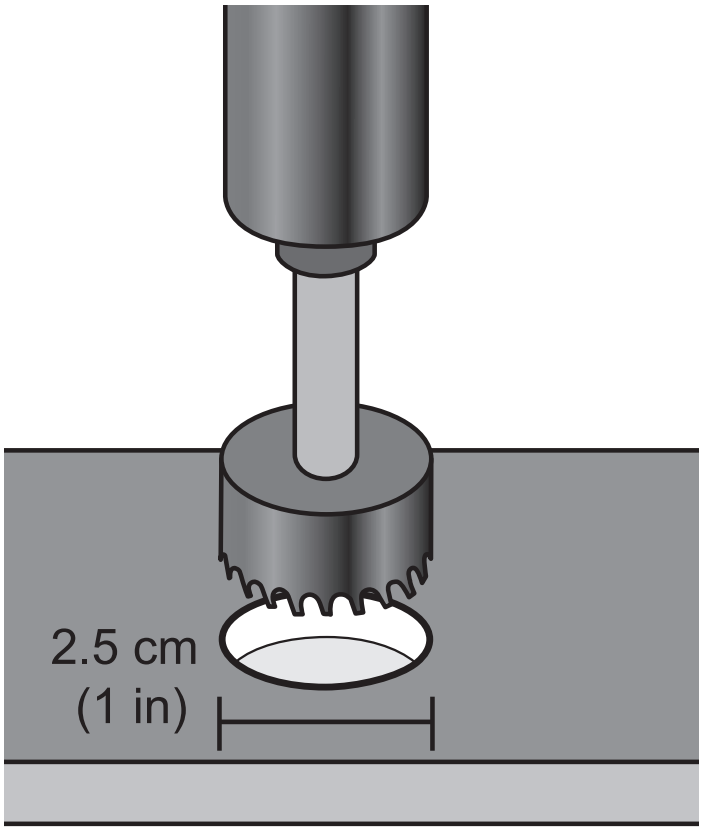
3. Align the tube into the recessed area in the center of the microphone. Install the 3 screws (removed in step 1) to secure the tube.



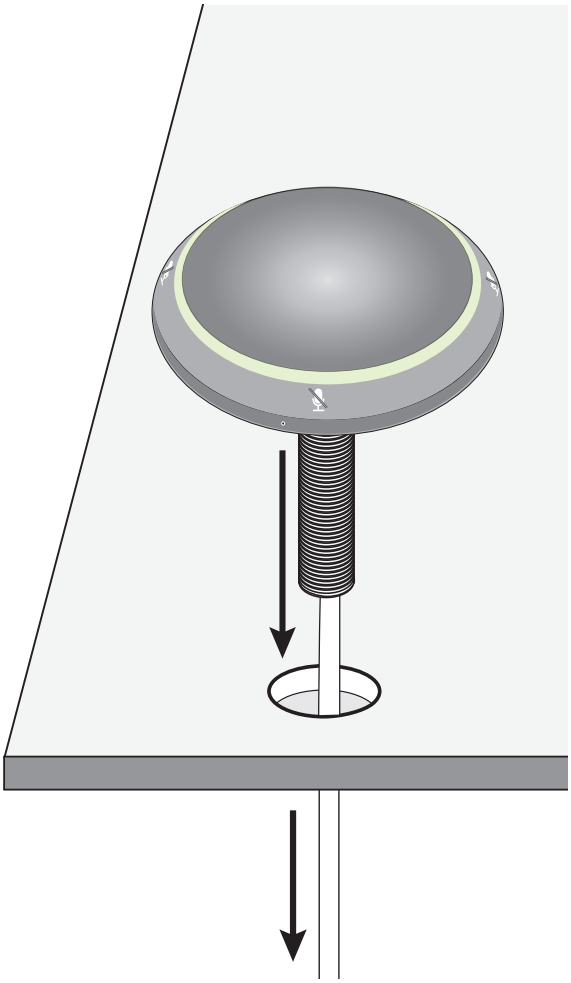
4. Slide one of the rubber washers to the base of the tube.



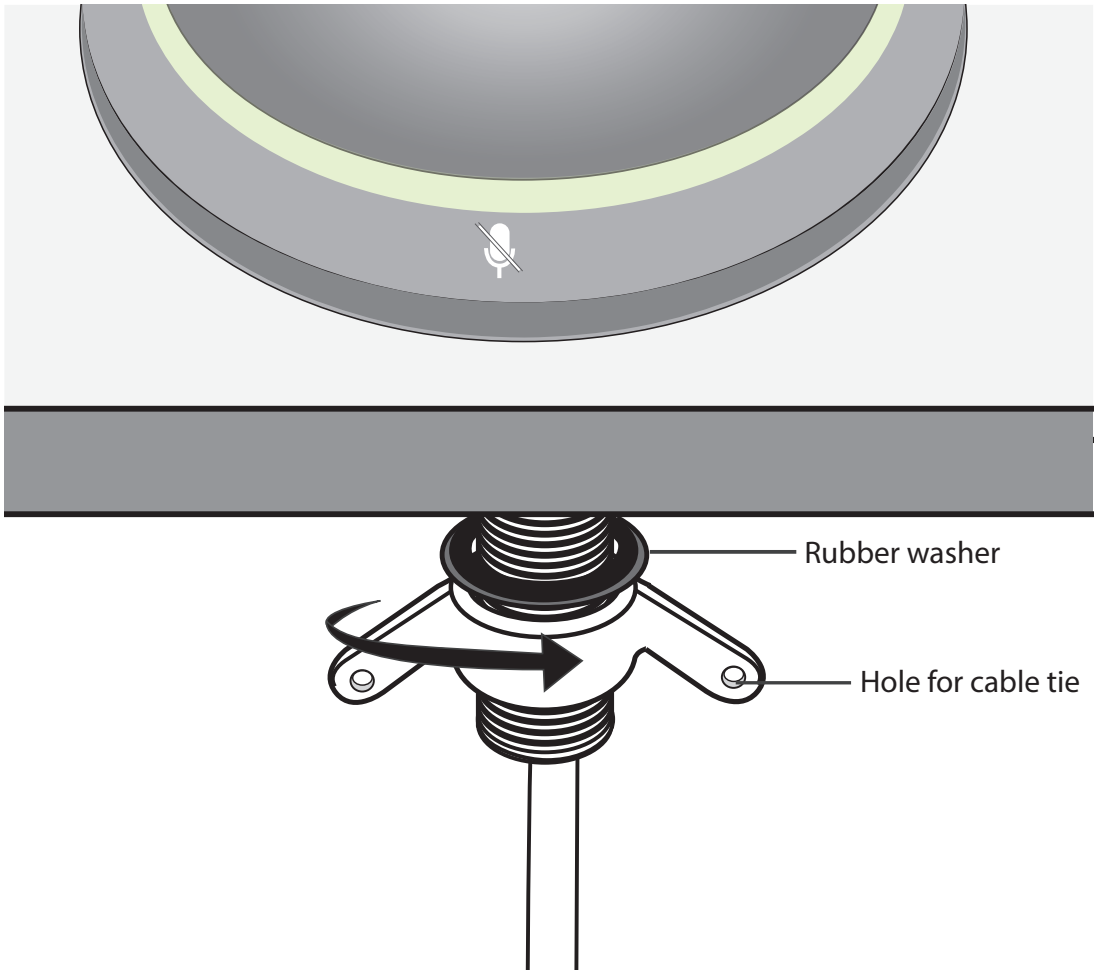
5. Drill a 1-inch (2.5 cm) hole through the table.



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

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, accessible through the web application.

The web application provides increased positioning flexibility over traditional conferencing microphones:

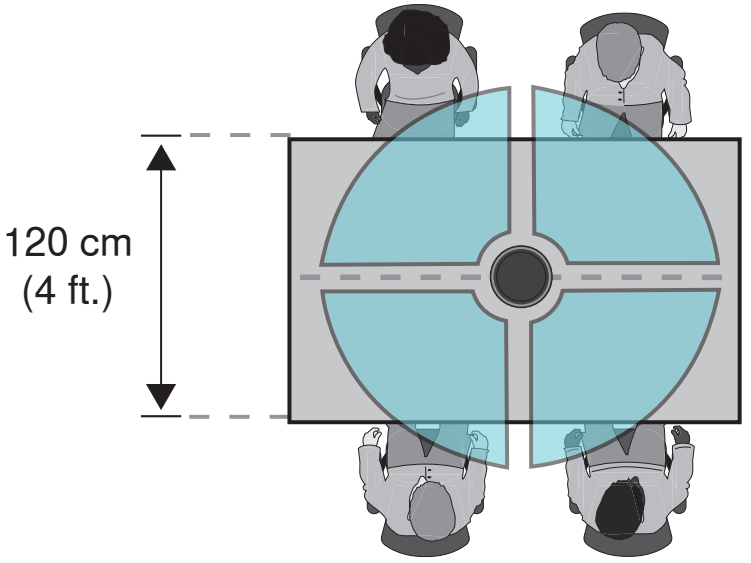
- Configurable pickup areas can be rotated and modified for the number of talkers.
- Network connectivity, device identification, and presets allow moving, adding and removing microphones with ease.
- Independent channels and automixing make Dante™ signal routing simple and flexible.
- Customized presets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

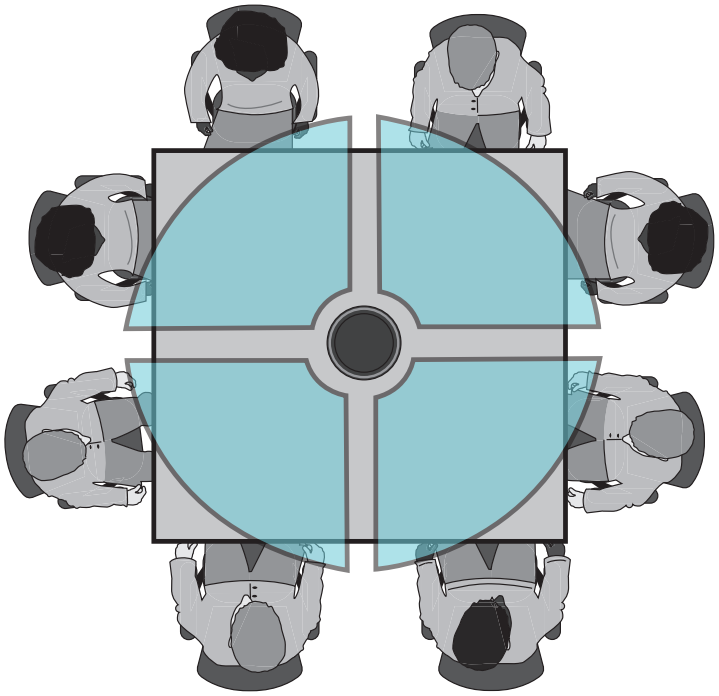
Seating Scenarios

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 through the web application, or through an external control system.

Single-Microphone Applications (Multiple Channels)

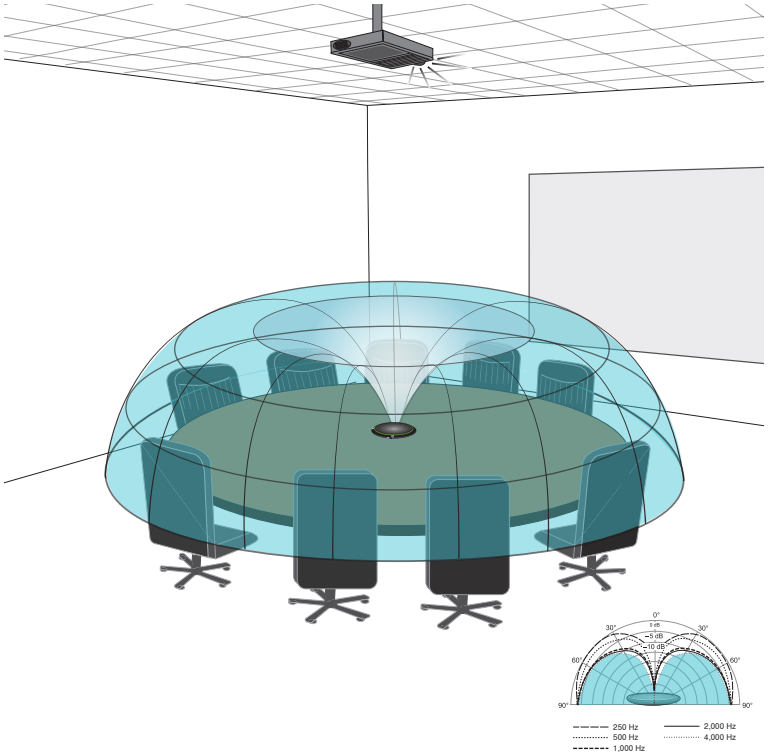
With four independent channels and polar patterns, coverage can be customized to match the table shape, size, and seating arrangement. The automatic mixing feature helps reduce extraneous noise (such as typing or paper shuffling) from interfering with speech intelligibility on the far end.



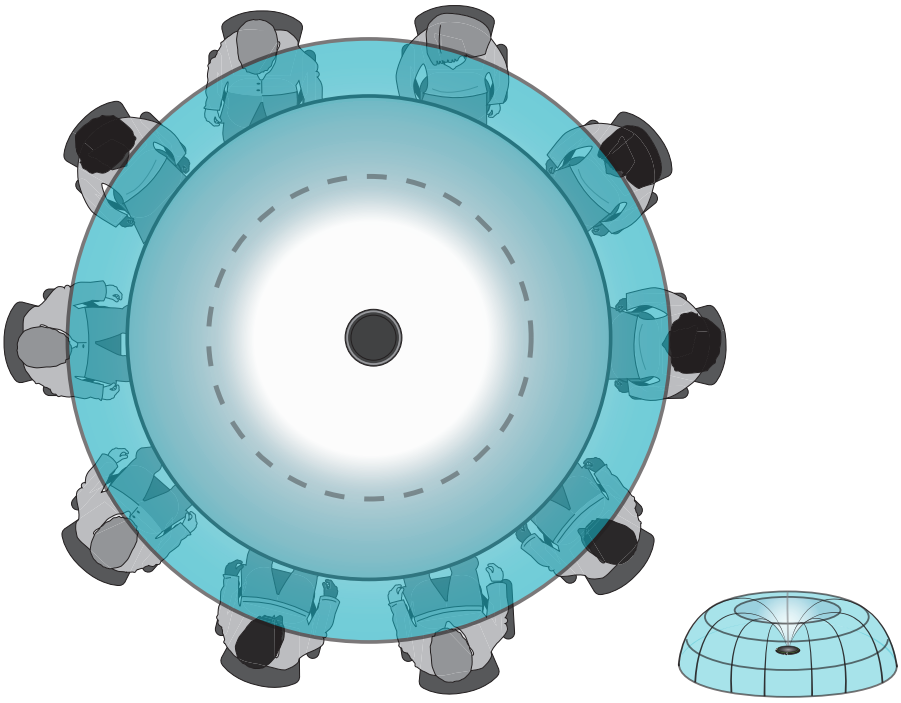


Toroid Pattern Applications

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.



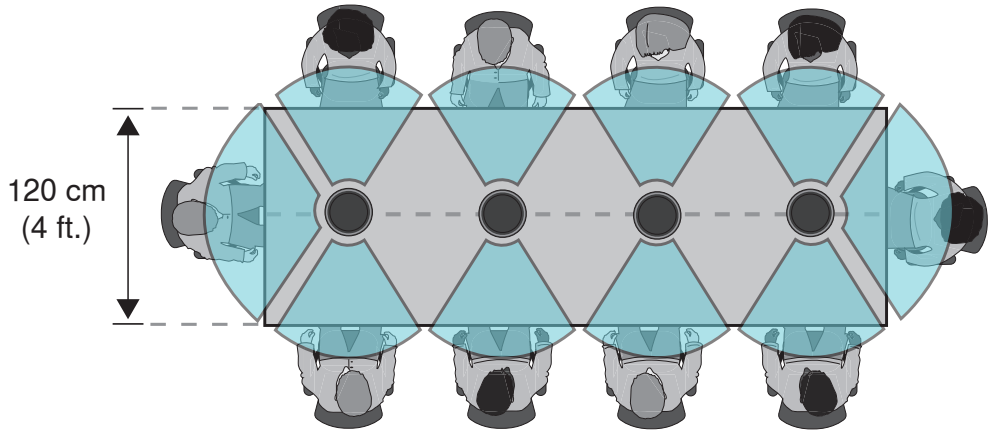
Noise from a ceiling-mounted projector is rejected, while all talkers are covered.



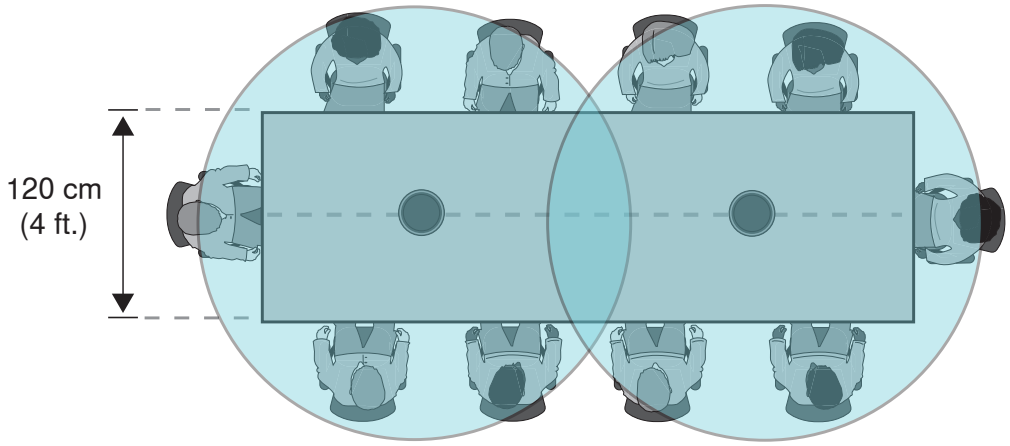
For a table with a single microphone and more than 4 talkers, the toroid pattern to ensures that all voices are heard equally.

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

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

Using A Password

All settings are configurable by default. To protect settings with a password, open the Settings menu and select the General tab. In this screen, passwords can be created or changed.

Once a password has been set, a Read-Only option appears on the log-in screen. In Read-Only mode, device parameters can be viewed, but not edited. Device identification remains active.

Firmware Updates

Firmware is embedded software in each component that controls functionality. Periodically, new versions of firmware are developed to incorporate additional features and enhancements. To take advantage of design improvements, new versions of the firmware can be uploaded and installed using the Shure Update Utility. Software is available for download from http://www.shure.com.

Important: When components are connected through the Shure MXW Audio Network Interface, their firmware must be updated on one device at a time prior to updating the MXW Audio Network Interface firmware. Attempting to update all devices at once will cause the interface to reboot after its firmware is updated, and the connection to other networked components will be lost.

Perform the following steps to update the firmware:

CAUTION! Ensure the device has a stable network connection during the update. Do not turn off the device until the update is complete.

1. Connect the device and computer to the same network (set to the same subnet).
2. Download Shure Update Utility app and install it.
3. Open the application.
4. Click Check For Updates... button to view new firmware versions available for download.
5. Select the desired firmware and press Downloadto download it to the Firmware Library.
6. From the Update Devices tab, select the new firmware and press Send Updates... to begin the firmware update, which overwrites the existing firmware on the device.

Firmware Release Requirements

All devices comprise a network with multiple communications protocols that work together to ensure proper operation. The recommended best practice is that all devices are on an identical release. To view the firmware version of each device on the network, open the component user interface, and look under Settings>About.

The format for Shure device's firmware is MAJOR.MINOR.PATCH. (Ex. 1.6.2 where 1 is the Major firmware level, 6 is the Minor firmware level, and 2 is the Patch firmware level.) At minimum, devices that operate on the same subnet should have identical MAJOR and MINOR release numbers.

- Devices of different MAJOR releases are not compatible.
- Differences in the PATCH firmware release level may introduce undesired inconsistencies.

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
 **Omnidirectional**	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.
 **Toroid**	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).
 **Bi-directional**	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.
 **One, Two, Three, or Four Talkers**	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: 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 Channels

The templates feature controls the number of active channels. Changing the template to match the desired channel count does not affect the channel level settings.

1. In the Microphone Configuration screen, select Templates.
2. Select the template with the desired number of channels.

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.

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

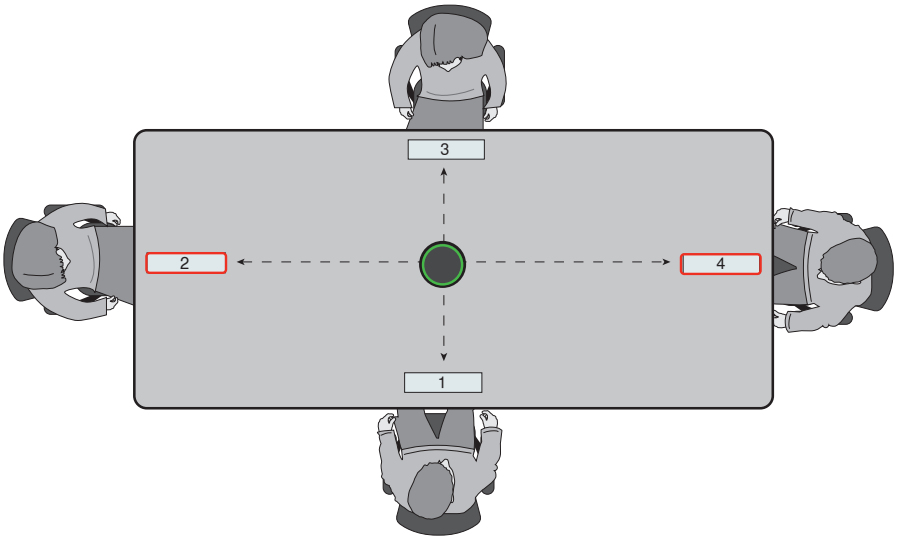
Use a template as a starting point when configuring coverage. Templates only adjust coverage, and do not affect gain levels or other settings.

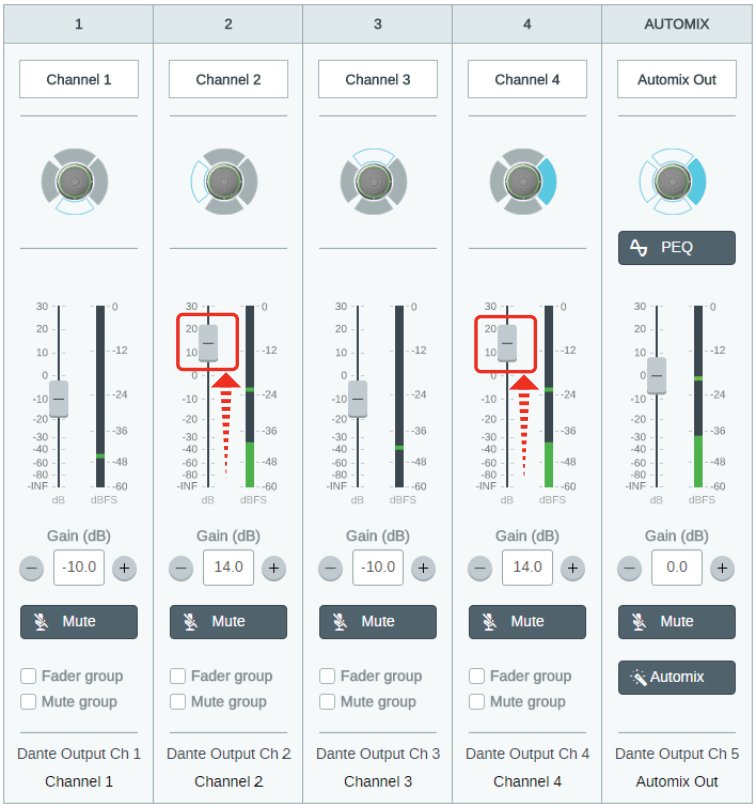
1. Select the template that is the closest match to the seating scenario
2. Select OK
3. If channels must be added, select the template with the desired channel count. This is the only way to add channels to a configuration.

Adjusting Channel Levels

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.

To adjust the level, move the fader up or down or manually enter a gain setting.





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

Adjusting Levels

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.

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.

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.

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

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

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

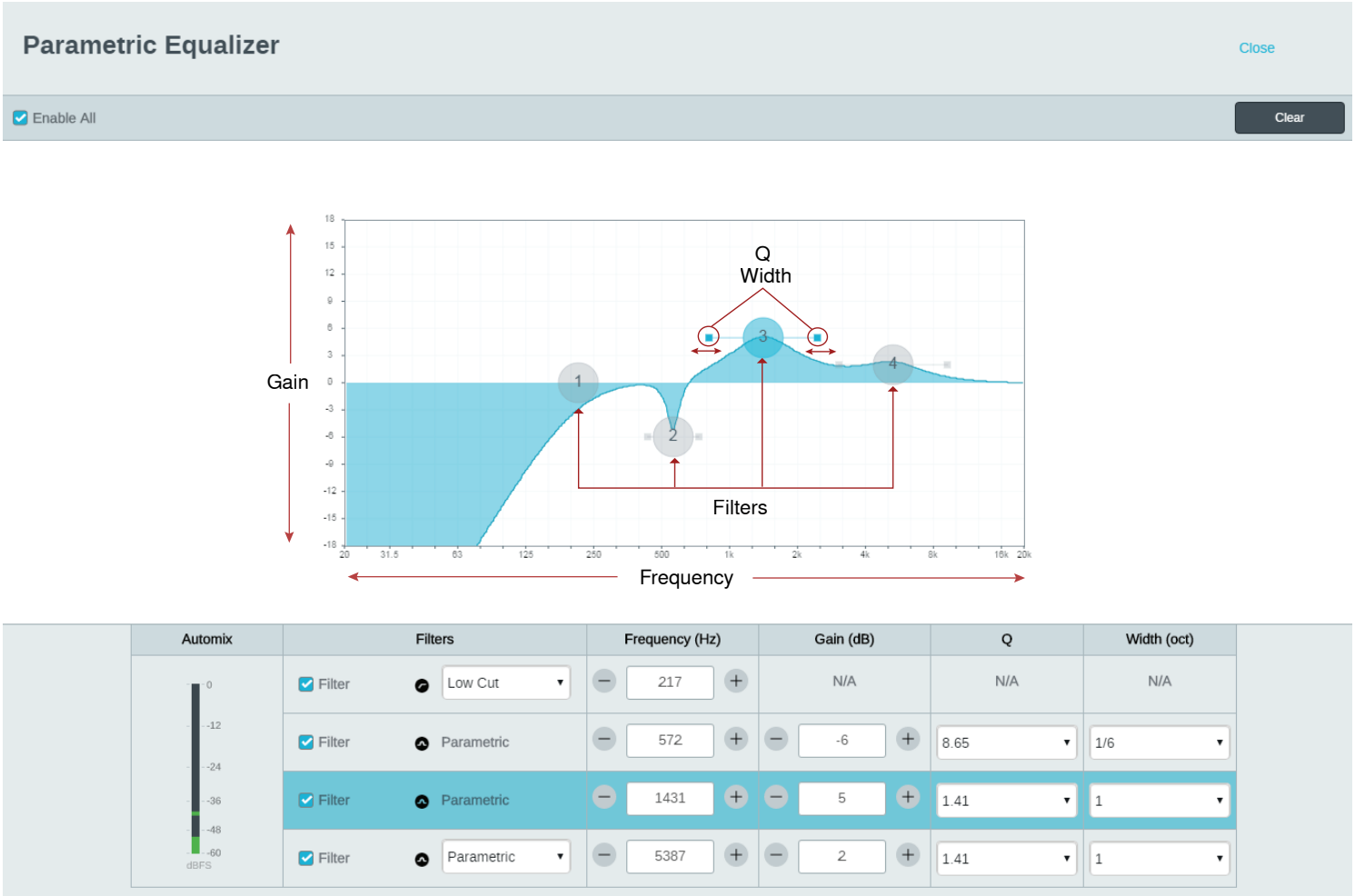
Frequency
Select the center frequency of the filter to cut/boost

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.



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.

Note: The equalizer can only be assigned to a single channel on the MXA310.

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

EQ Application	Suggested Settings
Reduce flutter echoes and sibilance	Identify the specific frequency range that "excites" the room: Set a narrow Q value 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 Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound
Reduce hollow, resonant room sound	Identify the specific frequency range that "excites" the room: Set a narrow Q value 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 Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound

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 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 -15 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	-6.0
5	-7.0
6	-7.8
7	-8.4
8	-9.0

Custom

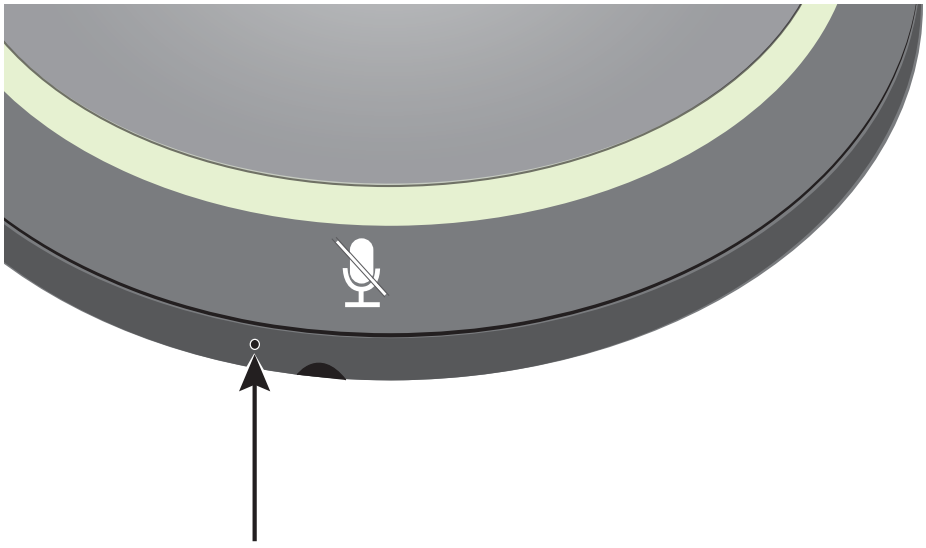
Custom mode provides control over all automixing parameters. This mode is useful when adjustments must be made to one of the preset modes to fit a particular application. If parameters are changed in smooth or classic mode, custom mode automatically activates.

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.

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 two 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.

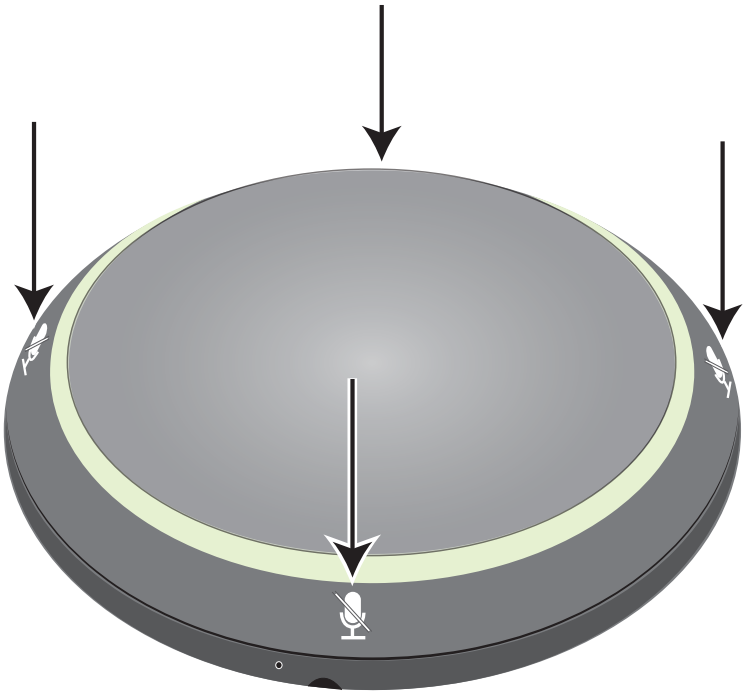
Low-cut Filter

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

Four touch-sensitive mute buttons are indicated by the mute icons around the edge of the microphone. Pressing any of these buttons mutes the entire device. In the web application, channels can be muted individually. 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 configure button properties through the web application, go to mic configuration>Button Control>Button Properties.

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

Using a Third-Party Control System

The microphone can send an external logic control signal to any networked devices that receive logic signals through an Ethernet connection. This allows the microphone mute switch to mute a DSP audio signal, instead of (or in addition to) muting the microphone at the source. The microphone also receives logic commands over the network. Many parameters controlled through the web application can be controlled through 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 in the device help or from www.shure.com.

To send a logic signal out when the mute button is pressed:

1. In the web application, select Configuration > Button Control.
2. Under the Button Properties menu, change the Mute Control Function setting to Logic out.

MXA310 Microflex® Advance™ Command Strings

This document can also be found at: http://shure.custhelp.com/app/answers/detail/a_id/6059

The device is connected via Ethernet to a control system, such as AMX, Crestron or Extron.

Connection: Ethernet (TCP/IP; select “Client” in the AMX/Crestron program)
Port: 2202

Conventions

The device has 4 types of strings:

GET
Finds the status of a parameter. After the AMX/Crestron sends a GET command, the MXA310 responds with a REPORT string

SET
Changes the status of a parameter. After the AMX/Crestron sends a SET command, the MXA310 will respond with a REPORT string to indicate the new value of the parameter.

REP
When the MXA310 receives a GET or SET command, it will reply with a REPORT command to indicate the status of the parameter. REPORT is also sent by the MXA310 when a parameter is changed on the MXA310 or through the GUI.

SAMPLE
Used for metering audio levels.

All messages sent and received are ASCII. Note that the level indicators and gain indicators are also in ASCII

Most parameters will send a REPORT command when they change. Thus, it is not necessary to constantly query parameters. The MXA310 will send a REPORT command when any of these parameters change.

The character “x” in all of the following strings represents the channel of the MXA310 and can be ASCII numbers 0 through 5 as in the following table.

0	All channels
1 through 4	Individual channels
5	Automix output

Command Strings (Common)

Get All		
	Command String: `< GET x ALL >`	Where x is ASCII channel number: 0 through 5. Use this command on first power on to update the status of all parameters.
	MXA310 Response: `< REP ... >`	The MXA310 responds with individual Report strings for all parameters.
Get Channel Name		
	Command String: `< GET x CHAN_NAME >`	Where x is ASCII channel number: 0 through 5.
	MXA310 Response: `< REP x CHAN_NAME {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >`	Where yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy is 31 characters of the user name. The MXA310 always responds with a 31 character name.
Get Device ID		
	Command String: `< GET DEVICE_ID >`	The Device ID command does not contain the x channel character, as it is for the entire device.
	MXA310 Response: `< REP DEVICE_ID {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >`	Where yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy is 31 characters of the device ID. The microphone always responds with a 31 character device ID.
Get Audio Gain		

	Command String: < GET x AUDIO_GAIN_HI_RES >	Where x is ASCII channel number: 1 through 5. Channel number 0 (all channels) is not valid for this command.
	MXA310 Response: < REP x AUDIO_GAIN_HI_RES yyyy >	Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.
Set Audio Gain		
	Command String: < SET x AUDIO_GAIN_HI_RES yyyy >	Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.
	MXA310 Response: < REP x AUDIO_GAIN_HI_RES yyyy >	Where yyyy takes on the ASCII values of 0000 to 1400.
Increase Audio Gain by n dB		
	Command String: < SET x AUDIO_GAIN_HI_RES INC nn >	Where nn is the amount in one-tenth of a dB to increase the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).
	MXA310 Response: < REP x AUDIO_GAIN_HI_RES yyyy >	Where yyyy takes on the ASCII values of 0000 to 1400.
Decrease Audio Gain by n dB		
	Command String: < SET x AUDIO_GAIN_HI_RES DEC nn >	Where nn is the amount in one-tenth of a dB to decrease the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).
	MXA310 Response: < REP x AUDIO_GAIN_HI_RES yyyy >	Where yyyy takes on the ASCII values of 0000 to 1400.
Get Channel Audio Mute		
	Command String: < GET x AUDIO_MUTE >	Where x is ASCII channel number: 0 through 5. See table on page 1. Channel Audio Mute is pre-meter.
	MXA310 Response: < REP x AUDIO_MUTE ON > < REP x AUDIO_MUTE OFF >	The MXA310 will respond with one of these strings.
Mute Channel Audio		
	Command String: < SET x AUDIO_MUTE ON >	
	MXA310 Response: < REP x AUDIO_MUTE ON >	
Unmute Channel Audio		
	Command String: < SET x AUDIO_MUTE OFF >	
	MXA310 Response: < REP x AUDIO_MUTE OFF >	
Toggle Channel Audio Mute		
	Command String: < SET x AUDIO_MUTE TOGGLE >	
	MXA310 Response: < REP x AUDIO_MUTE ON > < REP x AUDIO_MUTE OFF >	The MXA310 will respond with one of these strings.
Get Device Audio Mute		
	Command String: < GET DEVICE_AUDIO_MUTE >	Device Audio Mute is equivalent to the physical mute button on the mic. Device Audio Mute is post-meter.

	MXA310 Response: < REP DEVICE_AUDIO_MUTE ON > < REP DEVICE_AUDIO_MUTE OFF >	The MXA310 will respond with one of these strings.
Mute Device Audio		
	Command String: < SET DEVICE_AUDIO_MUTE ON >	
	MXA310 Response: < REP DEVICE_AUDIO_MUTE ON >	
Unmute Device Audio		
	Command String: < SET DEVICE_AUDIO_MUTE OFF >	
	MXA310 Response: < REP DEVICE_AUDIO_MUTE OFF >	
Toggle Device Audio Mute		
	Command String: < SET DEVICE_AUDIO_MUTE TOGGLE >	
	MXA310 Response: < REP DEVICE_AUDIO_MUTE ON > < REP DEVICE_AUDIO_MUTE OFF >	The MXA310 will respond with one of these strings.
Get Output Clip Status		
	Command String: < GET x AUDIO_OUT_CLIP_INDICATOR >	Where x is ASCII channel number: 1 through 5. See table on page 1. It is not necessary to continually send this command. The microphone will send a REPORT message whenever the status changes.
	MXA310 Response: < REP x AUDIO_OUT_CLIP_INDICATOR ON > < REP x AUDIO_OUT_CLIP_INDICATOR OFF >	The MXA310 will respond with one of these strings.
Flash Lights on Microphone		
	Command String: < SET FLASH ON > < SET FLASH OFF >	Send one of these commands to the MXA310. The flash automatically turns off after 30 seconds.
	MXA310 Response: < REP FLASH ON > < REP FLASH OFF >	The MXA310 will respond with one of these strings.
Turn Metering On		
	Command String: < SET METER_RATE sssss >	Where sssss is the metering speed in milliseconds. Setting sssss=0 turns metering off. Minimum setting is 100 milliseconds. Metering is off by default.
	MXA310 Response: < REP METER_RATE sssss > < SAMPLE aaa bbb ccc ddd eee >	Where aaa, bbb, etc is the value of the audio level received and is 000-060. aaa = output 1 bbb = output 2 ccc = output 3 ddd = output 4 eee = output 5
Stop Metering		
	Command String: < SET METER_RATE 0 >	A value of 00000 is also acceptable.

	MXA310 Response: < REP METER_RATE 00000 >	
Get Audio Peak Level		
	Command String: < GET x AUDIO_IN_PEAK_LVL >	
	MXA310 Response: < REP x AUDIO_IN_PEAK_LVL nn >	Where nn is the audio level and is 00-60.
Get Audio RMS Level		
	Command String: < GET x AUDIO_IN_RMS_LVL >	
	MXA310 Response: < REP x AUDIO_IN_RMS_LVL nn >	Where nn is the audio level and is 00-60.
Get Preset		
	Command String: < GET PRESET >	
	MXA310 Response: < REP PRESET nn >	Where nn is the preset number 01-10.
Set Preset		
	Command String: < SET PRESET nn >	Where nn is the preset number 1-10. (Leading zero is optional when using the SET command).
	MXA310 Response: < REP PRESET nn >	Where nn is the preset number 01-10.
Get Preset Name		
	Command String: < GET PRESET1 > < GET PRESET2 > < GET PRESET3> etc	Send one of these commands to the MXA310.
	MXA310 Response: < REP PRESET1 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET2 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET3 {yyyyyyyyyyyyyyyyyyyyyyyyy} > etc	Whereyyyyyyyyyyyyyyyyyyyyyyyyy is 25 characters of the device ID. The MXA310 always responds with a 25 character device ID
Get Gate Out Status		
	Command String: < GET x AUTOMIX_GATE_OUT_EXT_SIG >	Where x is ASCII channel number: 0 through 4. It is not necessary to continually send this command. The MXA310 will send a REPORT message whenever the status changes.
	MXA310 Response: < REP x AUTOMIX_GATE_OUT_EXT_SIG ON >< REP x AUTOMIX_GATE_OUT_EXT_SIG OFF >	The MXA310 will respond with one of these strings.
External Switch Out		
	Command String: < GET EXT_SWITCH_OUT_STATE >	It is not necessary to continually send this command. The MXA310 will send a REPORT message whenever the status changes.
	MXA310 Response: < REP EXT_SWITCH_OUT_STATE ON > < REP EXT_SWITCH_OUT_STATE OFF >	The MXA310 will respond with one of these strings.
Mute Button Status		

	Command String: < GET MUTE_BUTTON_STATUS >	It is not necessary to continually send this command. The MXA310 will send a REPORT message whenever the status changes.
	MXA310 Response: < REP MUTE_BUTTON_STATUS ON > < REP MUTE_BUTTON_STATUS OFF >	The MXA310 will respond with one of these strings.
	Mute Button LED State	
	Command String: < GET MUTE_BUTTON_LED_STATE >	
	MXA310 Response: < REP MUTE_BUTTON_LED_STATE ON > < REP MUTE_BUTTON_LED_STATE OFF >	The MXA310 will respond with one of these strings.
	Get Ring LED State(Use when GUI Lighting Style is set to RING)	
	Command String: < GET DEV_LED_IN_STATE >	This command is only available when both “Mute Control Function” is set to “Logic Out” or "Disabled" AND Light Ring “Lighting Style” is set to “Ring” in the GUI.
	MXA310 Response: < REP DEV_LED_IN_STATE ON > < REP DEV_LED_IN_STATE OFF >	The MXA310 will respond with one of these strings.
	Set Ring LED State(Use when GUI Lighting Style is set to RING)	
	Command String: < SET DEV_LED_IN_STATE ON > < SET DEV_LED_IN_STATE OFF >	Send one of these commands to the MXA310. This command is only available when both “Mute Control Function” is set to “Logic Out” or "Disabled" AND Light Ring “Lighting Style” is set to “Ring” in the GUI.
	MXA310 Response: < REP DEV_LED_IN_STATE ON > < REP DEV_LED_IN_STATE OFF >	The MXA310 will respond with one of these strings.
	Get Segments LED State(Use when GUI Lighting Style is set to SEGMENTS)	
	Command String: < GET x CHAN_LED_IN_STATE >	This command is only available when both “Mute Control Function” is set to “Logic Out” or "Disabled" AND Light Ring “Lighting Style” is set to “Segments” in the GUI.
	MXA310 Response: < REP x CHAN_LED_IN_STATE ON > < REP x CHAN_LED_IN_STATE OFF >	The MXA310 will respond with one of these strings.
	Set Segments LED State(Use when GUI Lighting Style is set to SEGMENTS)	
	Command String: < SET x CHAN_LED_IN_STATE ON > < SET x CHAN_LED_IN_STATE OFF >	Where x is ASCII channel number: 1 through 4. Send one of these commands to the MXA310. This command is only available when both “Mute Control Function” is set to “Logic Out” or "Disabled" AND Light Ring “Lighting Style” is set to “Segments” in the GUI.
	MXA310 Response: < REP x CHAN_LED_IN_STATE ON > < REP x CHAN_LED_IN_STATE OFF >	The MXA310 will respond with one of these strings.
	Get LED Brightness	
	Command String: < GET LED_BRIGHTNESS >	

	MXA310 Response: < REP LED_BRIGHTNESS n >	Where n can take on the following values: 0 = LED disabled 1 = LED dim 2 = LED default
Set LED Brightness		
	Command String: < SET LED_BRIGHTNESS n >	Where n can take on the following values: 0 = LED disabled 1 = LED dim 2 = LED default
	MXA310 Response: < REP LED_BRIGHTNESS n >	
Get LED Mute Color		
	Command String: < GET LED_COLOR_MUTED >	
	MXA310 Response: < REP LED_COLOR_MUTED nnnn >	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
Set LED Mute Color		
	Command String: < SET LED_COLOR_MUTED nnnn >	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	MXA310 Response: < REP LED_COLOR_MUTED nnnn >	
Get LED Unmute Color		
	Command String: < GET LED_COLOR_UNMUTED >	
	MXA310 Response: < REP LED_COLOR_UNMUTED nnnn >	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
Set LED Unmute Color		
	Command String: < SET LED_COLOR_UNMUTED nnnn >	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	MXA310 Response: < REP LED_COLOR_UNMUTED nnnn >	
Get LED Mute Flashing		
	Command String: < GET LED_STATE_MUTED >	
	MXA310 Response: < REP LED_STATE_MUTED nnn >	Where nnn can be ON, OFF, or FLASHING
Set LED Mute Flashing		
	Command String: < SET LED_STATE_MUTED nnn >	Where nnn can be ON, OFF, or FLASHING
	MXA310 Response: < REP LED_STATE_MUTED nnn >	
Get LED Unmute Flashing		
	Command String: < GET LED_STATE_UNMUTED >	
	MXA310 Response: < REP LED_STATE_UNMUTED nnn >	Where nnn can be ON, OFF, or FLASHING

Set LED Unmute Flashing		
	Command String: < SET LED_STATE_UNMUTED nnn >	Where nnn can be ON, OFF, or FLASHING
	MXA310 Response: < REP LED_STATE_UNMUTED nnn >	

Digital Audio Networking

Dantetm 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 Recommendations for Dante Networking

In addition to the basic networking requirements, Dante audio networks should use a Gigabit network switch or router with the following features:

- Gigabit ports
- Quality of Service (QoS) with 4 queues
- Diffserv (DSCP) QoS, with strict priority
- Recommended: A managed switch to provide detailed information about the operation of each network link (port speed, error counters, bandwidth used)

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.

Table provided courtesy of Audinate®

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 or 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

Networking Best Practices

Use the following best practices when setting up a network to ensure reliable communication:

- Always use a "star" network topology by connecting each component directly to the switch or router.
- Connect all Shure networked devices to the **same network** and set to the **same subnet**. This applies to all devices that audio signals must be routed between (managed through Dante Controller). It is also required in order to open the web application for a device.
- Devices on separate networks require an audio processor or conferencing software to carry audio between them. See the system planning and gear requirements section for network setup information and configuration examples.
- Use only 1 DHCP server per network. Disable DHCP addressing on additional servers.
- Power on the switch and DHCP server prior to MXA equipment.
- To expand the network, use multiple Ethernet switches in a star topology.
- All devices must be at the same firmware revision level.

Network Audio and Shure Control Data

Microflex® Advance™ 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 through the web application. 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 the web application.
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 web application.
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.

The latency setting for Dante devices should be set according to the number of switches in the network.

Use Audinate's Dante Controller software to change the latency setting.

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

IP Ports and Protocols

Port	TCP/UDP	Protocol	Description	Factory Default
21	tcp	FTP	Required for firmware updates (otherwise closed)	Closed
22	tcp	SSH	Not supported	Closed
23	tcp	Telnet	Standard console interface	Closed
68	udp	DHCP	Dynamic Host Configuration Protocol	Open
80*	tcp	HTTP	Required to launch embedded web server	Open
427	tcp/udp	SLP†	Required for inter-device communication	Open
443	tcp	HTTPS	Not supported	Closed
161	tcp	SNMP	Not supported	Closed
162	tcp	SNMP	Not supported	Closed
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	Password
8180*	tcp	HTML	Required for web application	Open
8427	udp	Multcast SLP†	Required for inter-device communication	Open
64000	tcp	Telnet	Required for Shure firmware update	Open

Port	TCP/UDP	Protocol	Description
162	udp	SNMP	Used by Dante
[319-320]*	udp	PTP†	Dante clocking
4321, 14336-14600	udp	Dante	Dante audio

Port	TCP/UDP	Protocol	Description
[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

0 1

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 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

Accessories

Replacement Parts

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

Specifications

All specifications measured from cardioid polar pattern. Values for all patterns are within ± 3 dB of these specifications unless otherwise noted.

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

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

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

Dante Digital Output

Channel Count	5 total channels (4 independent transmit channels, 1 IntelliMix® 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) [2], Mute, Gain (140 dB range)
System	IntelliMix Automatic mixing, Low-Cut Filter (-12 dB/octave @150 Hz)

2 3

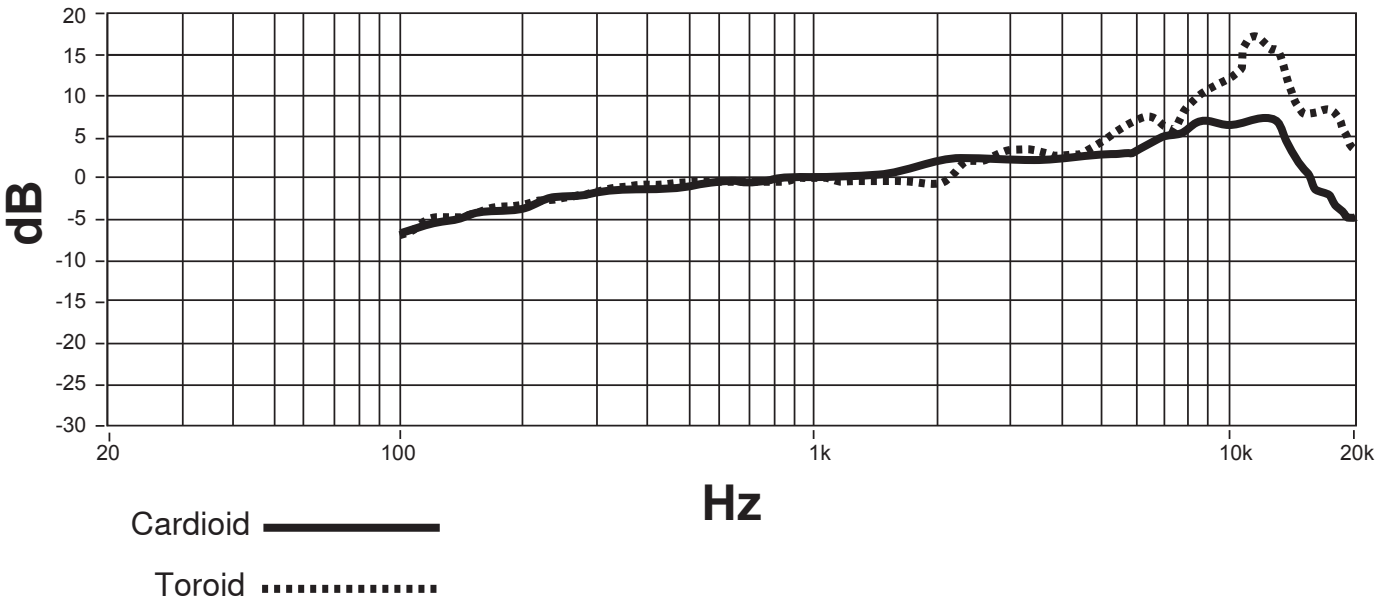
Networking

Cable Requirements

Cat 5e or higher (shielded cable recommended)

Frequency Response

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



IP Ports and Protocols

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22	tcp	SSH	Not supported	Closed
23	tcp	Telnet	Standard console interface	Closed

2 [1] 1 Pa=94 dB SPL

3 [2] Assignable to one channel at a time

Port	TCP/UDP	Protocol	Description	Factory Default
68	udp	DHCP	Dynamic Host Configuration Protocol	Open
80*	tcp	HTTP	Required to launch embedded web server	Open
427	tcp/udp	SLP†	Required for inter-device communication	Open
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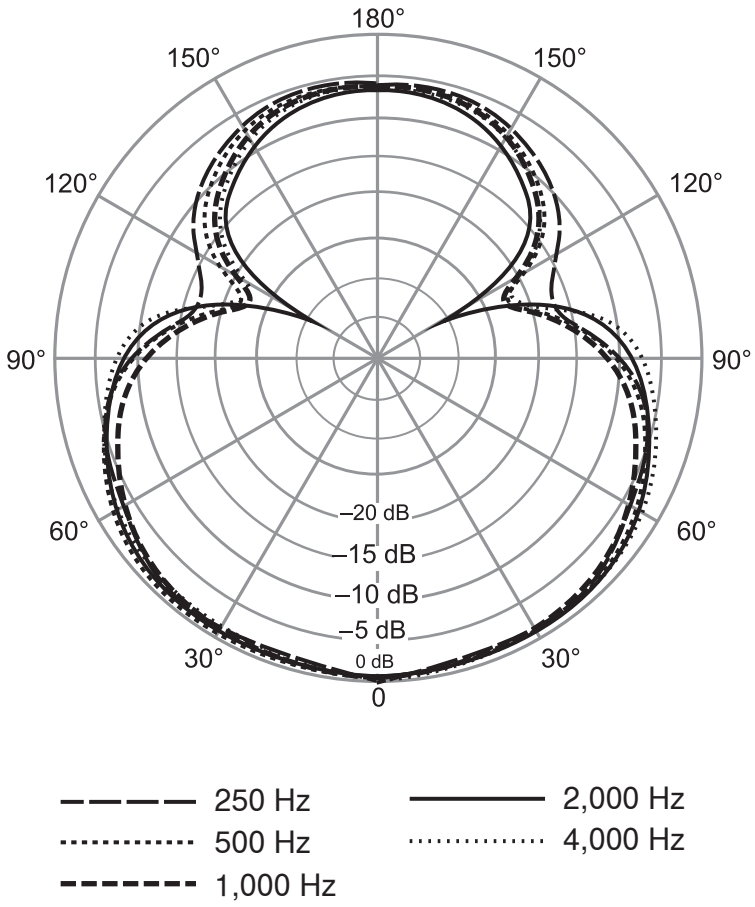
[8700-8706, 8800]*	udp	Dante	Dante Control and Monitoring
8751	udp	Dante	Dante Controller
16000-65536	udp	Dante	Used by Dante

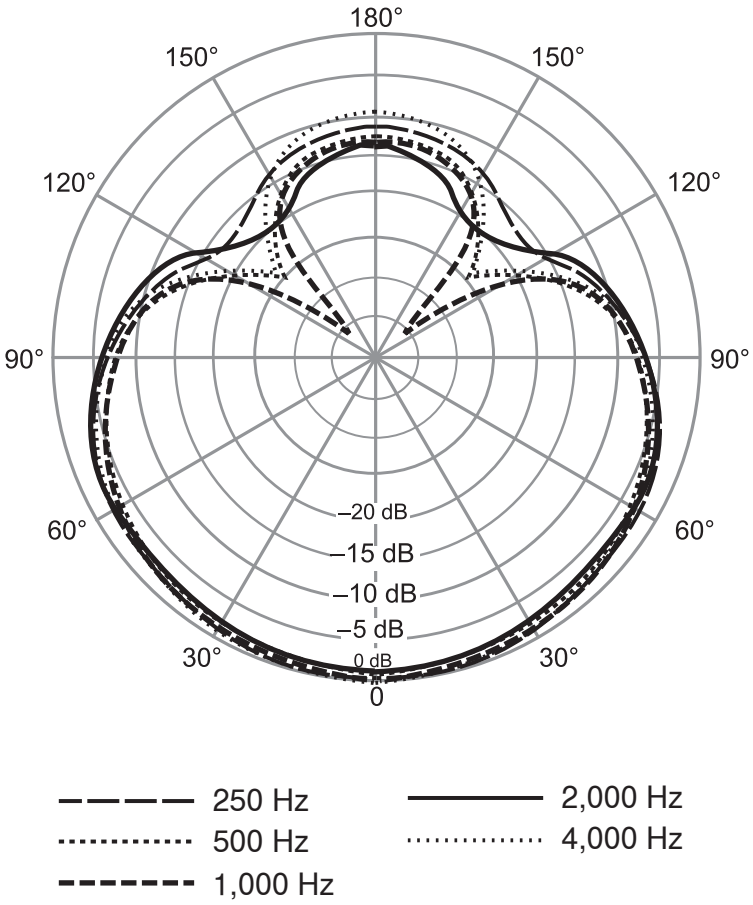
[4] [5]

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

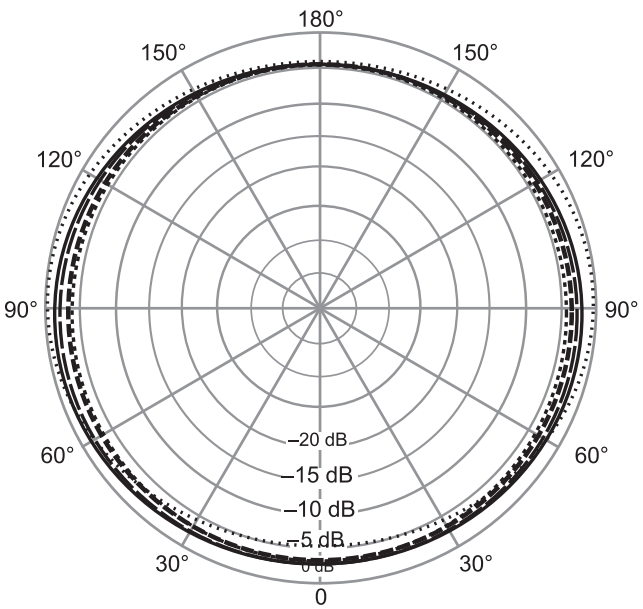
[5] †These protocols require multicast. Ensure multicast has been correctly configured for your network.

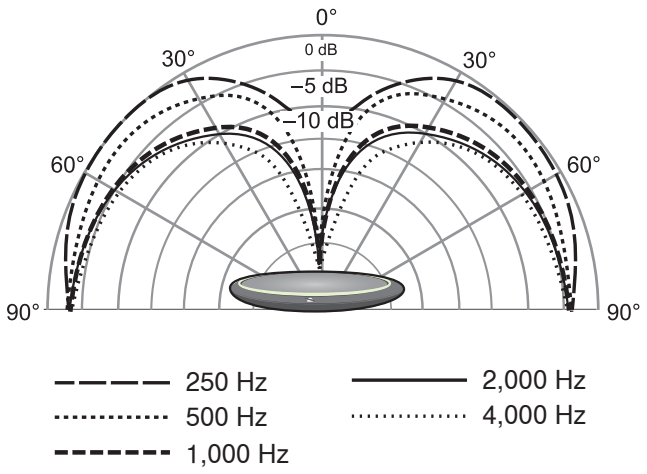


Hypercardioid

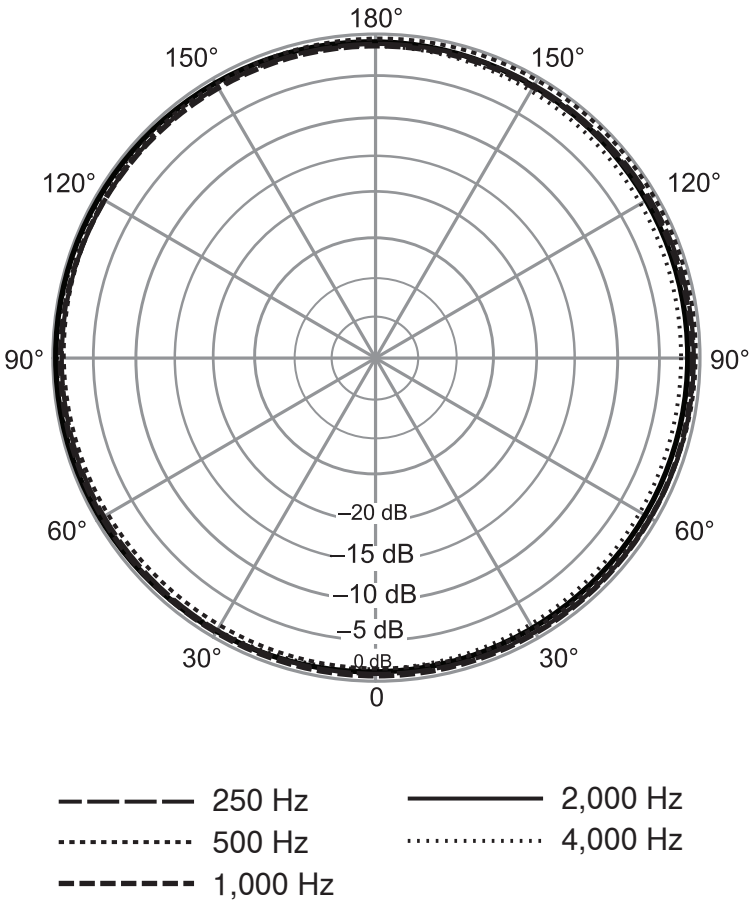


Supercardioid

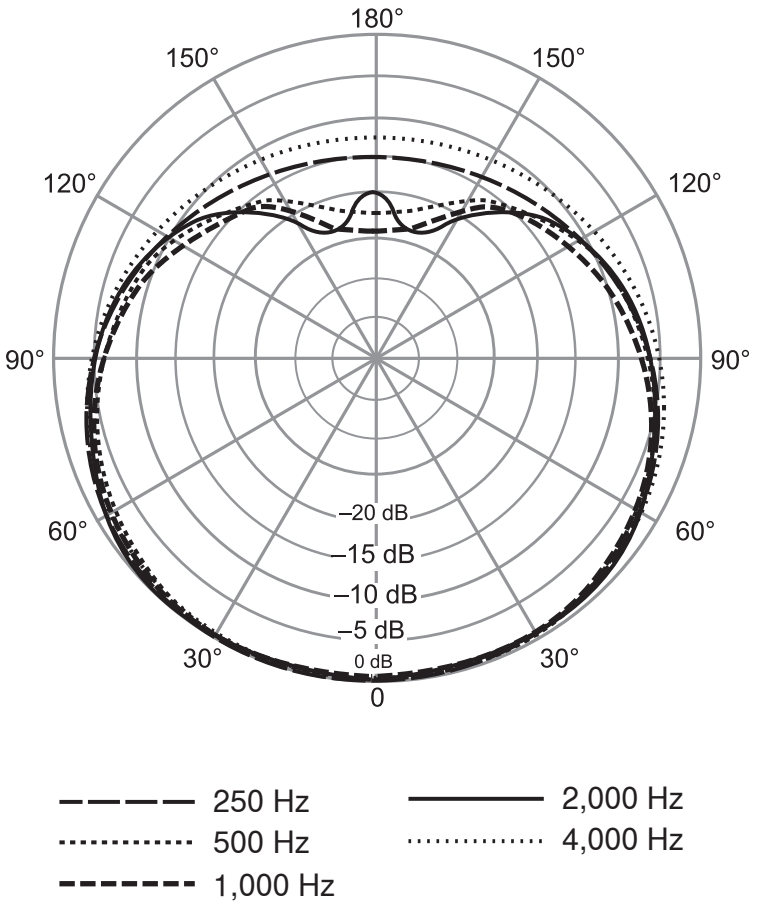




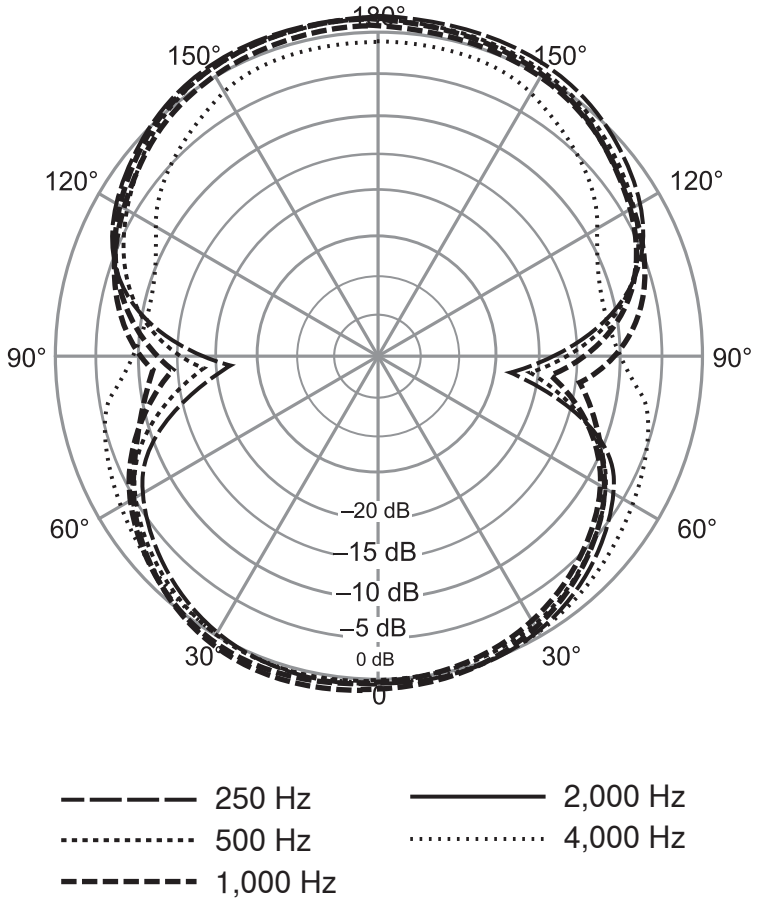
Toroid



Omnidirectional


180°
150°
150°
120°
120°
90°
90°
60°
60°
30°
30°
0
–20 dB
–15 dB
–10 dB
–5 dB
0 dB
250 Hz
500 Hz
1,000 Hz
2,000 Hz
4,000 Hz

Cardioid



Bidirectional