MXA920

Ceiling Array Microphone

User guide for Shure MXA920 ceiling array microphones. Learn how to install square and round mics, set up coverage, and quickly get great sound in any room.

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MXA920
Ceiling Array Microphone

Getting Started

Designer Setup Example

This example uses Designer 6.0 to connect an MXA920 and a P300. The process is similar when using other combinations of devices in Designer, so use these steps as a starting point.

After completing this basic setup process, you should be able to:

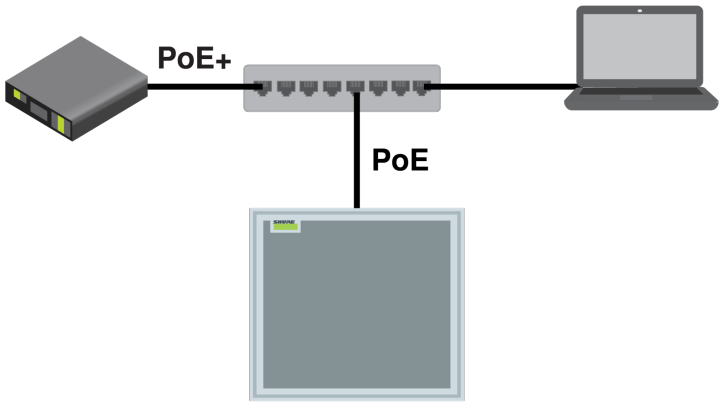
- Discover the MXA920 in Designer
- Add coverage areas
- Adjust DSP settings and route audio
- Understand the basic workflow for working with devices in Designer

This example uses:

- Cat5e (or better) Ethernet cable (shielded cable recommended)
- Network switch that provides Power over Ethernet (PoE) and PoE+ for the P300
- Shure Designer software installed on a computer. Download at shure.com/designer.
- P300

Step 1: Install and Connect

1. Install the MXA920 and other devices. Connect the MXA920 to a PoE port and the P300 to a PoE+ port on the switch using Ethernet cable.
2. Connect your computer running Designer to the same network.
3. Open Designer. Check that you're connected to the correct network in `File` > `Designer preferences`.
4. Open a new design and drag all devices into the design.

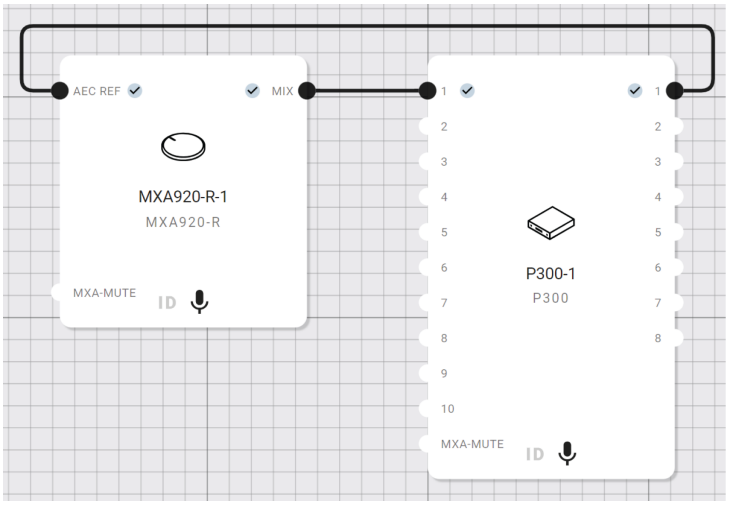


Step 2: Route Audio

The easiest way to route audio and apply DSP is with Designer's auto route feature. This feature automatically routes audio signals, applies DSP settings, turns on mute synchronization, and enables LED control for connected devices.

1. Select Auto route. Designer makes the following audio routes:
   - MXA920 output to P300 input
   - P300 output to MXA920 AEC reference input

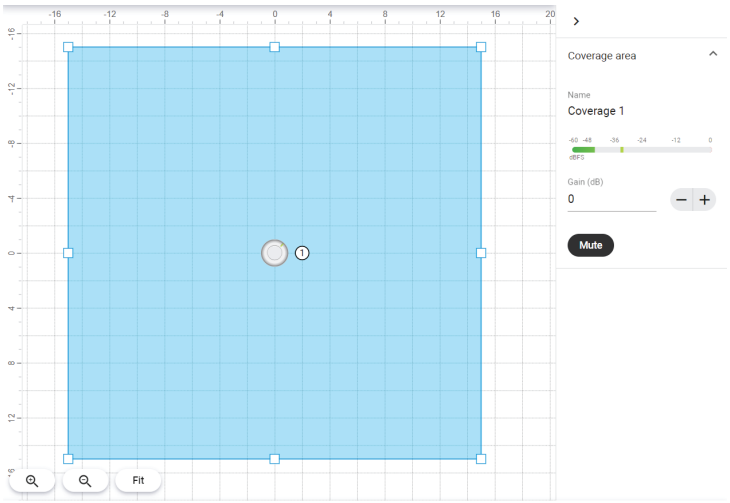
   You can also route audio manually in Designer or use Dante Controller.



2. Check the audio routes, matrix mixer routes, and other settings to make sure they fit your needs. You might need to:
   - Delete unnecessary routes.
   - Verify that AEC reference signals are correctly routed.
   - Fine-tune DSP blocks as needed.
   - Check matrix mixer routes.
3. Send audio from the P300 to other sources using the matrix mixer. A common destination is a computer connected by USB with videoconferencing software.
4. Select Deploy and connect to send device settings to your installed online devices. Designer walks you through associating the design devices with the online devices.

Step 3: Add Coverage

The default setting is a 30 by 30 foot (9 by 9 meter) dynamic coverage area. Any talker inside has coverage, and anything outside that area won't be picked up.



To add more coverage areas:

1. Go to [Your room] > Coverage.
2. Select the MXA920. Go to Properties > Coverage > Add coverage. Choose a dynamic or a dedicated coverage area. You can add any combination of up to 8 coverage areas per microphone. Move and resize as needed.
3. Set up a way to listen to the microphone directly (with a Dante headphone amp, for example). Place a test call with the whole conferencing system. Adjust gain and DSP as needed to get a good room sound.

You can also turn off automatic coverage to manually position up to 8 lobes:

- Designer: [Your room] > Coverage > MXA920 > Properties > Coverage > Auto coverage
- Web application: Settings

Web Application Setup Example

After completing this basic setup process, you should be able to:

- Access the MXA920's web application
- Add coverage areas
- Route audio to other Dante devices using Dante Controller

This example uses:

- Cat5e (or better) Ethernet cable
- Network switch that provides Power over Ethernet (PoE)
- Shure Update Utility and Dante Controller software

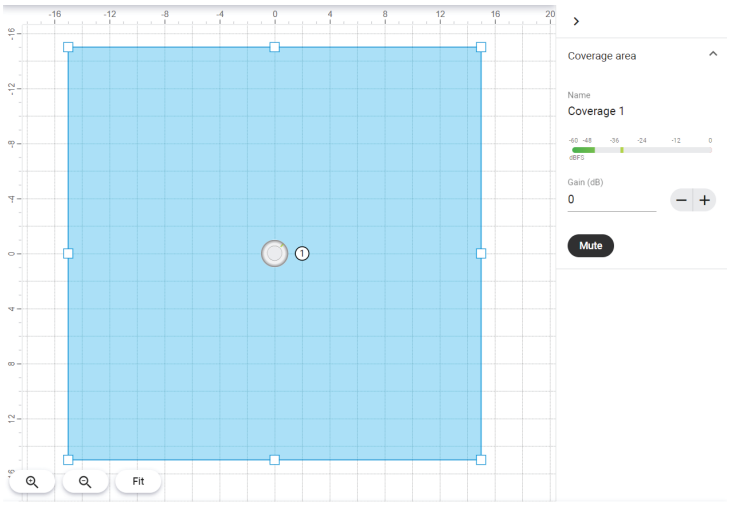
Step 1: Install and Connect

1. Install the microphone and connect it to a PoE port on the network switch using Ethernet cable.
2. Connect the computer running Shure Utility and Dante Controller to the same network.
3. Open Shure Update Utility. Find the MXA920 in the list of devices, and double-click to open the web application.

   Note: *The web application uses an HTTPS connection with self-signed certificates as of firmware 6.1 and newer. You may see a warning message in your browser the first time you open the web application. To proceed, click* Advanced *and then select the option to trust the connection.*

Step 2: Add Coverage

The default setting is a 30 by 30 foot (9 by 9 meter) dynamic coverage area. Any talker inside has coverage, and anything outside that area won't be picked up.



To add more coverage areas:

1. Go to `Coverage` > `Add coverage`.
2. Choose a dynamic or a dedicated coverage area. You can add any combination of up to 8 coverage areas per microphone. Move and resize as needed.
3. Set up a way to listen to the microphone directly (with a Dante headphone amp, for example). Adjust the gain and DSP as needed to get a good room sound. There are gain faders for each coverage area and for the automix output.

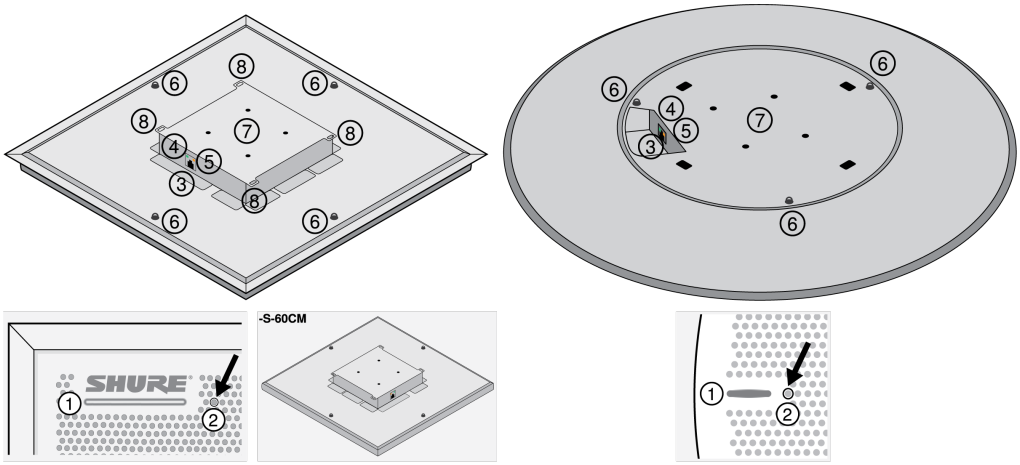
You can also turn off automatic coverage in Settings to manually position up to 8 lobes.

Step 3: Route Audio

To route audio to other Dante devices, use Dante Controller.

1. Open Dante Controller and find the MXA920 in the list of transmitters. With automatic coverage on, the MXA920 only sends audio from the automix output. Transmit channels 1-8 only work when automatic coverage is off.
2. Find the Dante device you're sending audio to in the list of receivers. To make an audio route, check the box where the MXA920's automix output intersects with the receiver device's input channel.
3. Route a far-end reference signal to the MXA920's AEC reference input. The AEC reference is usually the same one that feeds the room's loudspeakers.
4. Place a test call with the whole conferencing system. Adjust coverage, gain, and DSP as needed.

MXA920 Parts



1. Mute status LED

   Customize LED color and behavior in Designer: `Device configuration` > `Settings` > `Lights`.

Default Settings

Microphone Status	LED Color/Behavior
Active	Green (solid)
Muted	Red (solid)
Hardware identification	Green (flashing)
Firmware update in progress	Green (progresses along bar)
Reset	**Network reset:** Red (progresses along bar) **Factory reset:** Triggers device power-up
Error	Red (split, alternate flashing)
Device power-up	Multi-color flash, then blue (moves quickly back and forth across bar)

Note: *If LEDs are disabled, they will still turn on when the device powers up or when an error state occurs.*

2. Reset button
3. RJ-45 network port
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. Eyelet screws for suspension mounting (12 mm diameter)
7. VESA MIS-D mounting holes

8. Safety tether attachment points

Power Over Ethernet (PoE)

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

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

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

Model Variations

SKU	Description
MXA920W-S	White square microphone
MXA920W-S-60CM	White square microphone (60 cm)
MXA920AL-R	Aluminum round microphone
MXA920B-R	Black round microphone
MXA920W-R	White round microphone

Optional Accessories and Replacement Parts

- A900-S-GM Gripple Mount Kit, Square
- A900W-R-GM Gripple Mount Kit, Round, White Cover
- A900B-R-GM Gripple Mount Kit, Round, Black Cover
- A900-S-PM Pole Mount Kit, Square
- A900W-R-PM Pole Mount Kit, Round, White Cover
- A900B-R-PM Pole Mount Kit, Round, Black Cover
- A900-PM-3/8IN Threaded Rod Adapter Mounting Kit
- A910-JB Junction Box Accessory
- A900-CM Ceiling Mount
- A910-HCM Hard Ceiling Mount
- RPM904 frame and grille assembly for MXA920W-S-60CM or MXA910W-60CM
- RPM901W-US frame and grille assembly for MXA920W-S or MXA910W-US

MXA920 Codec Certifications

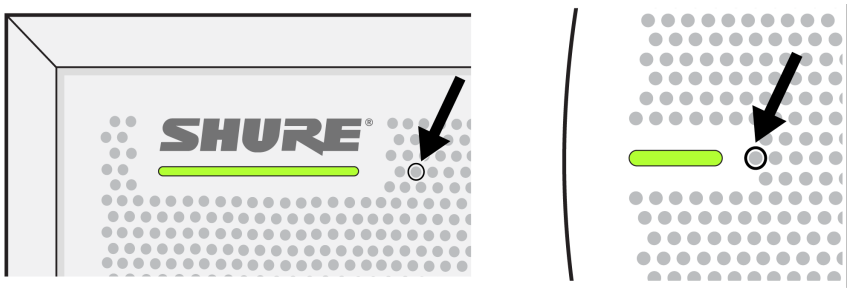
Find MXA920 audio codec certifications at shure.com/mxa920.

What's in the Box

Square or round array microphone	MXA920-S or MXA920-R
Square or round hardware kit **Square:** **Cable ties (8)**	Square: 90A49117 Round: 90A49116

Strain relief tabs (3) **Rubber pad set** **Round:** **Cable ties (8)** **Strain relief tabs (3)**	

Reset Button



The reset button is behind the grille. To push it, use a paper clip or other tool.

Button locations:

- **Square array microphones:** Behind the grille hole with a silkscreened circle around it.
- **Round array microphones:** Behind the first grille hole to the right of the mute status LED.

Reset Modes

- **Network reset** (press for 4-8 seconds): Resets all Shure control and audio network IP settings to factory defaults. Red LED along bar.
- **Full factory reset** (press for more than 8 seconds): Resets all network and configuration settings to the factory defaults. LED turns off, then blue along bar.

MXA920 Control Software

There are 2 ways to control the MXA920:

- **Option 1:** Use Shure Designer software
  - Control all Shure devices in one place
  - Route audio to and from Shure devices
- **Option 2:** Access the MXA920's web application with Shure Update Utility
  - Control 1 microphone at a time
  - Route audio with Dante Controller software

Control Devices with Shure Designer Software

Use Shure Designer software to control this device's settings. Designer enables integrators and system planners to design audio coverage for installations using MXA microphones and other Shure networked devices.

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 `File` > `Designer preferences`.
3. Click Online devices. A list of online devices appears.
4. To identify devices, select a device and click ID in the Properties menu. Double-click the device to open the settings.

From here, you can add devices to designs or online rooms and route audio to other Shure devices. Learn more at shure.com/designer.

Open the Web Application with HTTPS

This device's browser-based web application uses an HTTPS connection for confidentiality and authenticity. HTTPS encrypts information sent between the device and the web application.

To open the web application:

1. Open Shure Update Utility and check that you're connected to the same network as the device.
2. Double-click the device's IP address to open the web application in your browser.
3. The web application uses self-signed certificates. You may see a warning message in your browser the first time you open the web application. Click Advanced and select the option to trust the connection.

To control HTTPS settings, go to `Settings` > `Services` in Designer or the web application.

For more help, refer to our step-by-step video about HTTPS web applications.

Available in firmware 6.2 and newer

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. You can update firmware using Shure Update Utility or ShureCloud.

Download Shure Update Utility at shure.com/suu.

For release notes and more details about releases, visit the Software and Firmware Archive.

Use Designer's Auto Route

Designer's auto route speeds up the process of connecting systems with 1 audio processor and at least 1 microphone. Auto route also creates mute control routes in rooms with MXA network mute buttons. When you select Auto route, you can direct Designer to:

- Create audio routes and mute control routes
- Adjust audio settings
- Turn on mute synchronization
- Enable LED logic control for applicable devices

The settings are optimized for your particular combination of devices. You can adjust settings further, but auto route gives you a good starting point. Auto route works with any device in Designer.

To use auto route:

1. Place all relevant devices in a design.
2. Select Auto route. Designer optimizes microphone and DSP settings for your equipment combination.

   If you remove or add devices, select Auto route again.

   Note: The auto route process clears any manual routes you may have made in your design.

After auto routing a room, check and adjust settings to fit your needs. You may need to:

- Delete unnecessary routes.
- Check levels and adjust gain.
- Check that AEC reference signals are correctly routed and received in a test call.
- Fine-tune DSP blocks.
- Adjust your processor's matrix mixer routes.

If you want to auto route an online room, turn on online room editing in `File` > `Designer preferences`.

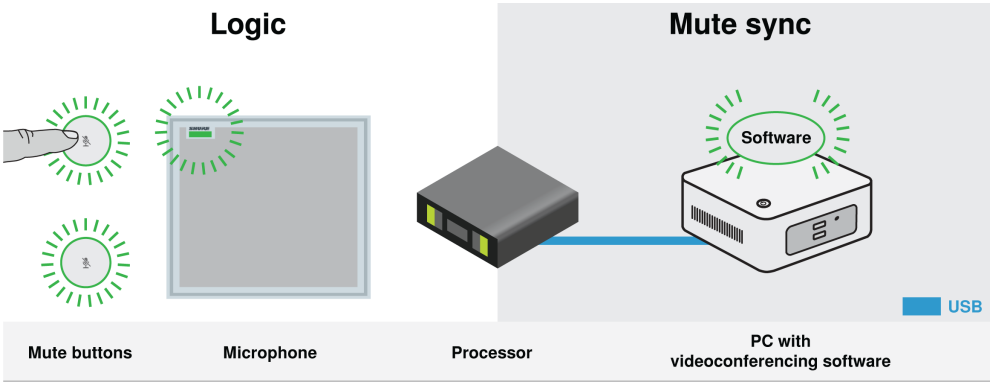
Note: Changes to an online room may cause audio to briefly drop out.

Refer to Designer's Troubleshooting section for help with routing.

Mute Sync

It's important to be able to see if a room is muted or unmuted during a call. You want devices to show the same mute status as the videoconferencing software. Shure devices use logic and mute sync to make this happen.

- **Logic:** Aligns mute status across Shure devices in the room. The processor or access point is the controller.
- **Mute sync:** Aligns mute between the processor and the videoconferencing software, which is on a computer connected to the processor by USB. The processor's automix output is muted or unmuted to change the mute status of the system.



When mute sync is working correctly, you can mute a device (networked mute button or microphone mute button) or the video-conferencing software to mute the room.

To use mute sync:

1. In Designer, create audio and mute control routes between devices in the room.
2. Connect a computer with videoconferencing software to the processor's USB port.
3. Turn on mute sync and logic:
   - Processors: `[Your device]` > `Settings` > `Mute control`
   - MXA320, MXA310, and MXA mute button: `Settings` > `Logic control` > `Mute control function` > `Logic out`
   - MXWAPX devices: `Settings` > `Wireless mics` > `Mute preference`

    - Microphones without physical mute button: Logic is always on

Designer's auto route process configures all necessary mute sync and logic settings for you.

Compatible Shure Logic Devices

- P300 (Also mutes supported soft codecs connected by USB)
- ANIUSB-MATRIX (Also mutes supported soft codecs connected by USB)
- IntelliMix Room software (Also mutes supported soft codecs connected by USB)
- ANIUSB-C
- MXA901
- MXA902
- MXA910
- MXA920
- MXA925
- MXA710
- MXA320
- 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 used with MX400DP base
    - MX412D
    - MX418D

For help with specific mute sync implementations, see our FAQs.

MXA920 Coverage

To access coverage settings:

- **Designer:** Add the microphone to a design or online room and go to Coverage.
- **Web application:** Go to Coverage.

To turn automatic coverage on or off:

- **Designer:**
    - `[Your room]` > `MXA920` > `Settings` > `Automatic coverage`
    - `[Your room]` > `Coverage` > `MXA920` > `Properties` > `Coverage` > `Auto coverage`
- **Web application:** Settings

How Much Space Does the MXA920 Cover?

For most rooms, Shure recommends:

- Distance from talker to microphone: Up to 16 feet (4.9 meters)
- Mounting height: Up to 12 feet (3.7 meters)

These numbers also depend on your room's acoustics, construction, and materials. With automatic coverage on, the default coverage area is a 30 by 30 foot (9 by 9 meter) dynamic coverage area.

In rooms with well-controlled acoustics, you can mount the MXA920 higher than 12 feet and achieve great results. Refer to Microphone Placement Best Practices for details.

After installation, enter the device's mounting height in the control software to help improve beamforming coverage: `Coverage` > `Properties` > `Position`.

How Does Coverage Work?

When you use automatic coverage, the microphone captures any talkers inside the boundaries of the coverage areas that you create. You can add a mix of up to 8 dynamic and dedicated coverage areas per microphone.

For more control, use the virtual acoustic boundary feature to adjust how much sound from outside the coverage area is removed from the automix output signal.

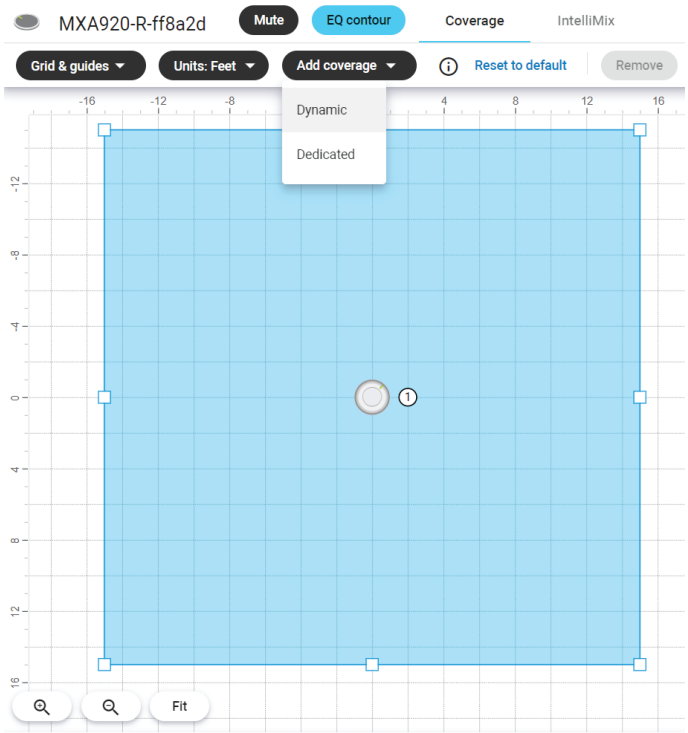
If you turn off automatic coverage, you can manually steer up to 8 lobes.

With automatic coverage on or off, the MXA920 uses Shure's Autofocus™ technology to fine-tune coverage in real time as talkers shift positions or stand. Autofocus is always active, and you don't need to adjust anything for it to work.

Add Coverage Areas

Automatic coverage = On

When you open Coverage, there's a 30 by 30 foot (9 by 9 meter) dynamic coverage area ready to use. Any talker inside has coverage, even if they stand up or walk around.

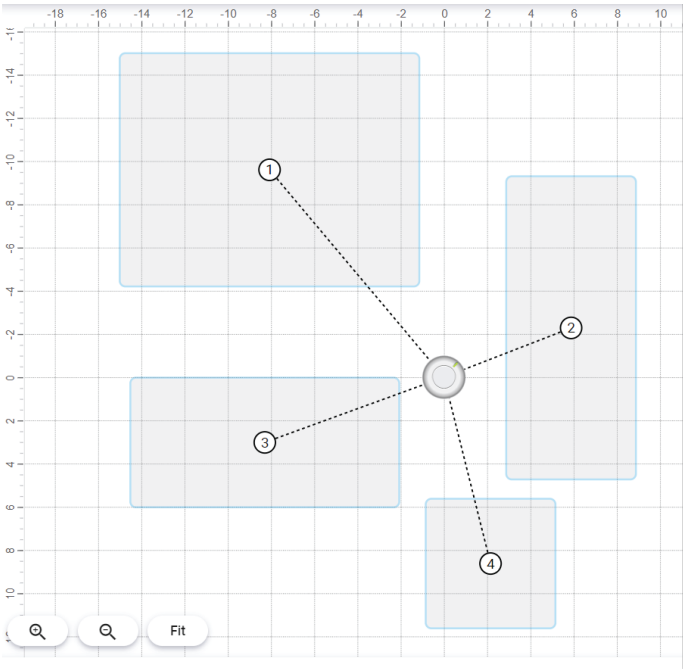


To add coverage areas:

- **Designer:** In the coverage view, select the MXA920, then `Properties` > `Coverage` > Add `coverage`
- **Web application:** Select Add coverage

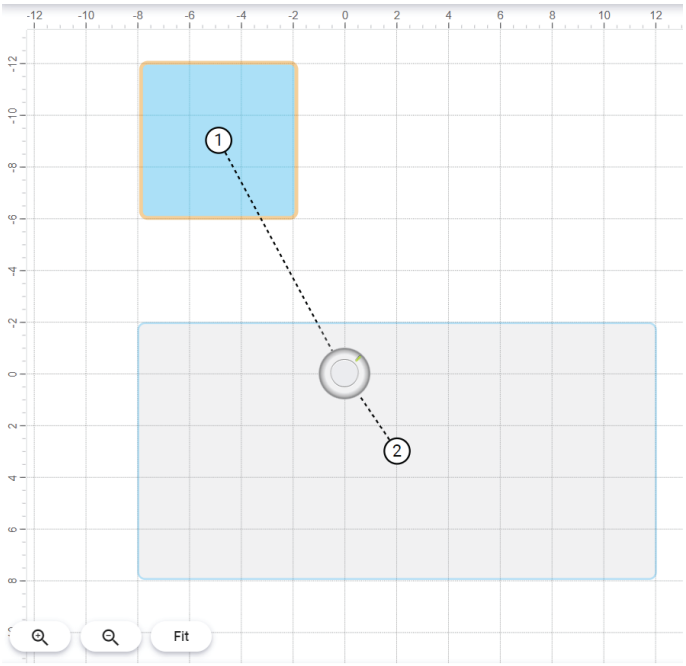
You can use up to 8 coverage areas per microphone, and you can mix both types as needed. Drag and drop to move coverage areas.

Dynamic Coverage Areas



Dynamic coverage areas have flexible coverage. The microphone intelligently adapts to cover all talkers in the coverage area. Change the size to fit your space, and any talker within the boundaries of the coverage area will have microphone coverage (even as they move).

Dedicated Coverage Areas



Dedicated coverage areas have microphone coverage at all times. They have a set size of 6 by 6 feet (1.8 by 1.8 meters). Dedicated coverage works best for talkers that are in one position most of the time, like at a podium or a whiteboard.

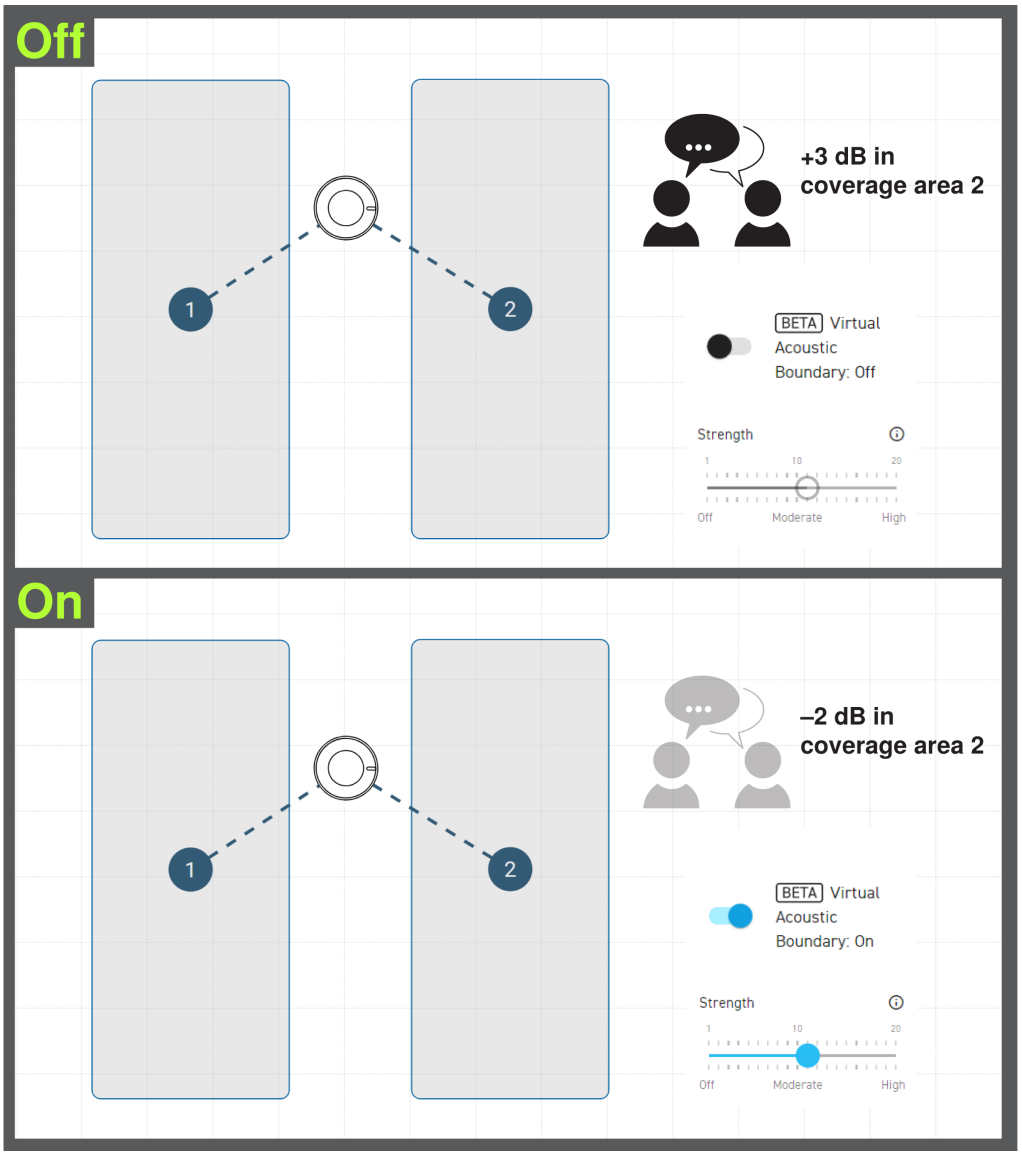
Remove Unwanted Sounds with the Virtual Acoustic Boundary

The virtual acoustic boundary adds more power to the edge of the microphone's coverage area.

When the virtual acoustic boundary is off, the microphone picks up sound that acoustically carries from outside an active coverage area at low levels. This background sound can still be distracting for listeners.

Turn on the virtual acoustic boundary to remove sound from outside coverage areas. The fader adjusts how much sound from outside the coverage area is removed from the microphone's automix output.

Note: The virtual acoustic boundary only works when automatic coverage is on.



To use the virtual acoustic boundary:

1. In the microphone's control software, go to `IntelliMix` > `Properties`. Scroll down to turn on the virtual acoustic boundary.
2. To test, make a call. While you talk, have someone outside of the defined coverage area talk or make noise at the same time.
3. Start with the medium default setting and use the fader to adjust as needed for your room. Have someone on the other end tell you how your signal sounds.

Tips

- A higher setting removes more sound from outside the coverage area. A lower setting removes less sound.
- Results may vary in small, highly reverberant rooms. Install acoustic treatment to improve speech intelligibility.

Use Steerable Lobes

Automatic coverage = Off

To use steerable lobes, turn off automatic coverage:

- **Designer:** [Your room] > Coverage > MXA920 > Properties > Coverage > Auto coverage
- **Web application:** Settings > General > Automatic coverage

With automatic coverage off, you can manually position up to 8 microphone lobes. This mode is best for when you need direct outputs, like for a multi-zone voice lift system. The microphone doesn't use coverage areas when automatic coverage is off.

Refer to the MXA910 guide to learn about using lobes.

Adjust Levels

Before adjusting levels:

1. Set up a way to listen to the microphone directly using a Dante® headphone amp or with Dante Virtual Soundcard.
2. Open Designer and find the MXA920 in the list of online devices. Alternatively, open the device's web application.

Automatic Coverage On

1. Talk in each coverage area at a normal speech volume. You can adjust:
   - **Coverage area gain (post-gate):** From the room's Coverage view, open the MXA920 properties panel on the right side. Select the coverage area to see post-gate gain and mute controls.
   - **IntelliMix gain (post-gate):** Go to the IntelliMix tab to adjust the automix output level and control DSP settings.
2. Adjust EQ settings as needed. You can use EQ to improve speech intelligibility and control problematic frequencies in a room (such as a low-frequency rumble caused by HVAC systems). If your EQ changes cause a large increase or decrease in the level, adjust the levels as in step 1.

Automatic Coverage Off

In this mode, there are 2 sets of gain faders:

- **Channel gain (pre-gate)**: To adjust, open the MXA920 controls in Designer and 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.
- **IntelliMix gain (post-gate)**: To adjust, open the MXA920 controls in Designer and go to IntelliMix. Alternatively, select a lobe in Coverage to see post-gate gain and mute controls in the properties panel. 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.

How to Install the MXA920

There are many ways to install MXA920 microphones. See below for details about the mounting and accessory options for square and round array microphones.

Microphone Placement Best Practices

Room Variables

Decide where to install the microphone by looking at the room's seating arrangements and infrastructure. Follow these guidelines for best results:

- Don't install the microphone behind obstructions.
- Install the microphone at least 4 feet (1.2 m) from noise sources. These include: air vents, projectors, or loudspeakers that have high dB levels.
  - Loudspeakers tend to have higher dB levels when installed above 10 feet (3 m).
- In rooms where talkers turn to face a screen, install the microphone toward the screen so talkers naturally talk toward the microphone.
- In rooms with flexible seating arrangements or multiple microphones, use Designer to ensure that the coverage is adequate for all seating scenarios.
  - If installing multiple microphones, set up coverage areas so they don't overlap.
- Install acoustic treatment to improve speech intelligibility in rooms that are too reverberant.
- If you're planning to use X/Y/Z coordinates for camera tracking, install the MXA920 so that it's level and not angled.

Mounting Height

For most rooms, Shure recommends a mounting height of up to 12 feet (3.7 meters). In rooms with controlled acoustics, you can install the MXA920 higher than 12 feet and get great results.

After installing, enter the device's height in the control software to help improve beamforming coverage: `Coverage` > `Properties` > `Position`.

Consider the following when choosing a mounting height:

- The pickup pattern of the ceiling array is narrower than a shotgun microphone, so it can be placed farther from sound sources. There is no specific barrier at which the audio degrades or gates off.
- Like all microphones, tonality changes as the distance from the source increases.
- The pickup area of the microphone increases slightly as it is mounted higher.

Square Mounting Options:

- In a ceiling grid
- With a VESA mounting device
- On NPT pole
- Suspend from ceiling with A900-GM
- Suspend from ceiling with your own hardware
- On 3/8-inch threaded rod
- Attach to ceiling with A900-CM
- In a hard ceiling

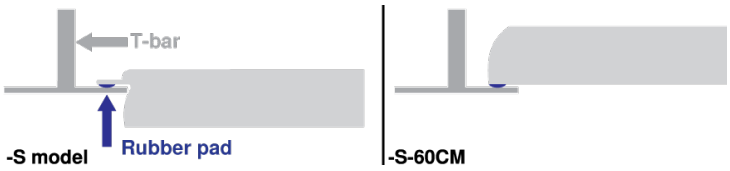
Round Mounting Options:

- With a VESA mounting device
- On NPT pole
- Suspend from ceiling with A900-GM
- Suspend from ceiling with your own hardware
- On 3/8-inch threaded rod
- Attach to ceiling with A900-CM

Installing in a Ceiling Grid

Before you begin:

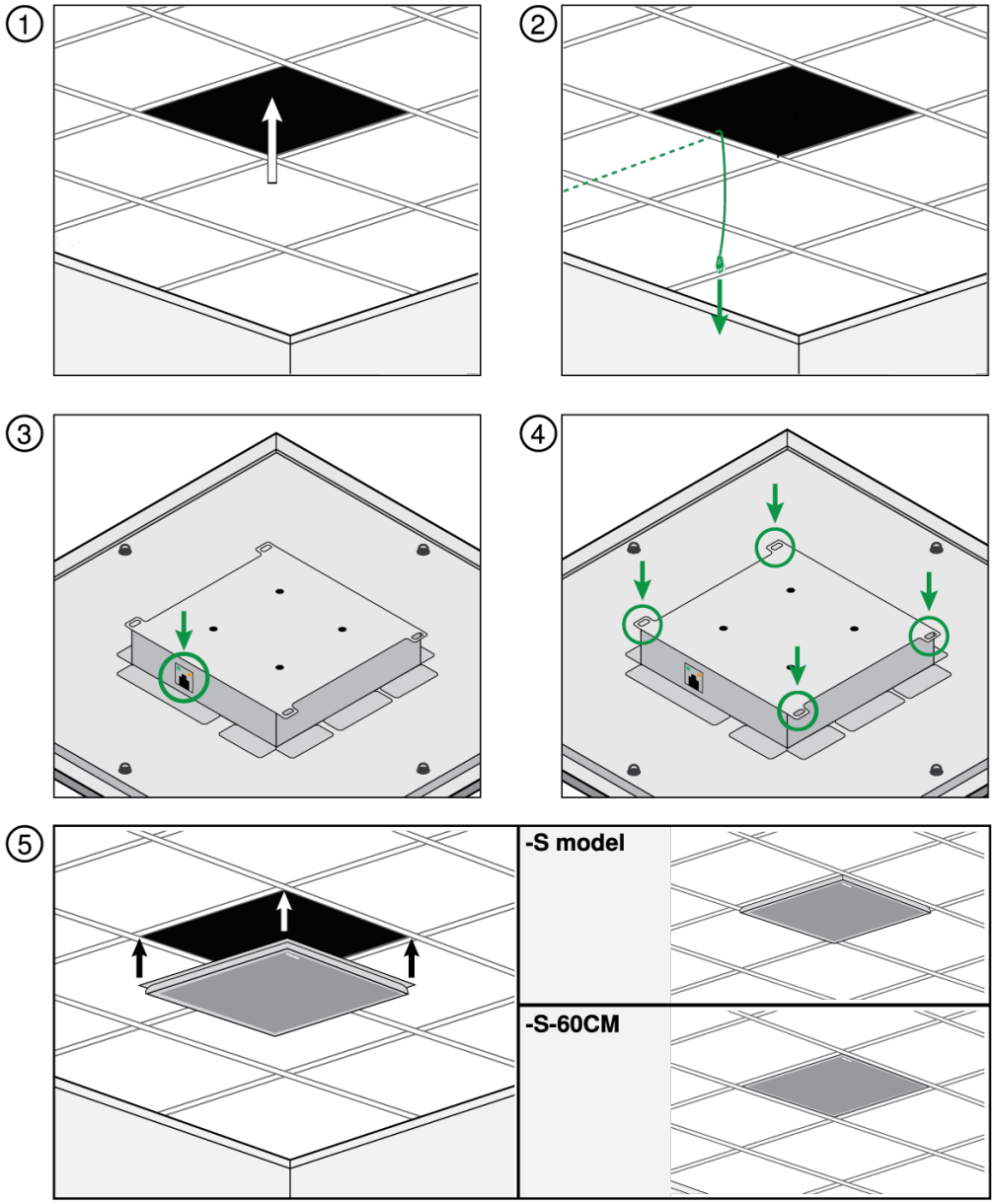
- Remove the plastic cover from the microphone.
- If using, install the rubber pads on the corners of the microphone to prevent scratches.



- Verify that your ceiling grid size matches your model variation.
- If using the A910-JB junction box, install it before ceiling installation.

IMPORTANT: Do not install the 60 cm model in a 2 foot (609.6 mm) ceiling grid.

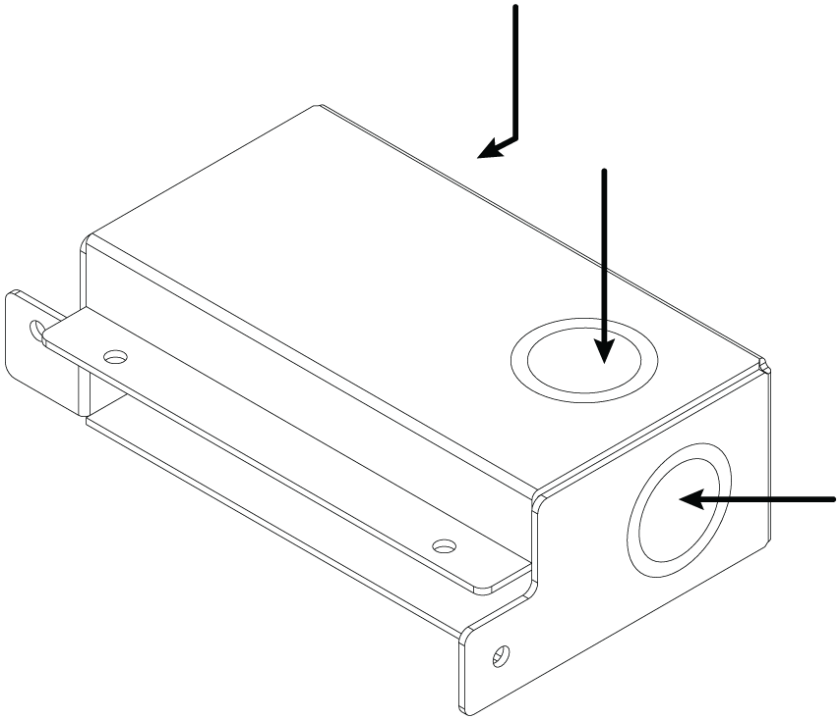
1. Make space in the ceiling grid for the array microphone to be installed.
2. Route the Ethernet cable above the ceiling grid and through the opening in the ceiling.
3. Plug the Ethernet cable in to the microphone.
4. Attach the safety tether between the building structure and one of the tie-off points on the back of the microphone using braided metal cable or other high-strength wire (not included). This safety measure prevents the microphone from falling in an emergency situation. Make sure there is no tension on the safety tether. Follow any local regulations.
5. Install the microphone in the ceiling grid.



Installing the Junction Box Accessory

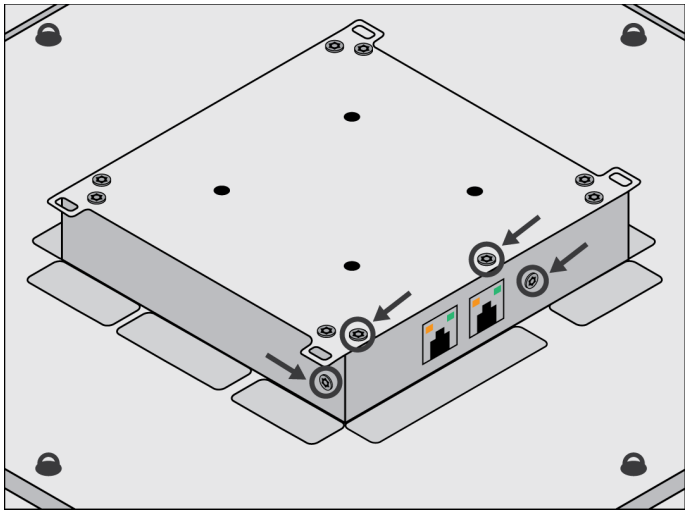
The A910-JB junction box mounts on square ceiling array microphones to connect conduit. There are 3 knockouts on the junction box for attaching conduit. See local regulations to determine if the junction box is necessary.

Note: Install the junction box on the microphone before installing the microphone in the ceiling.



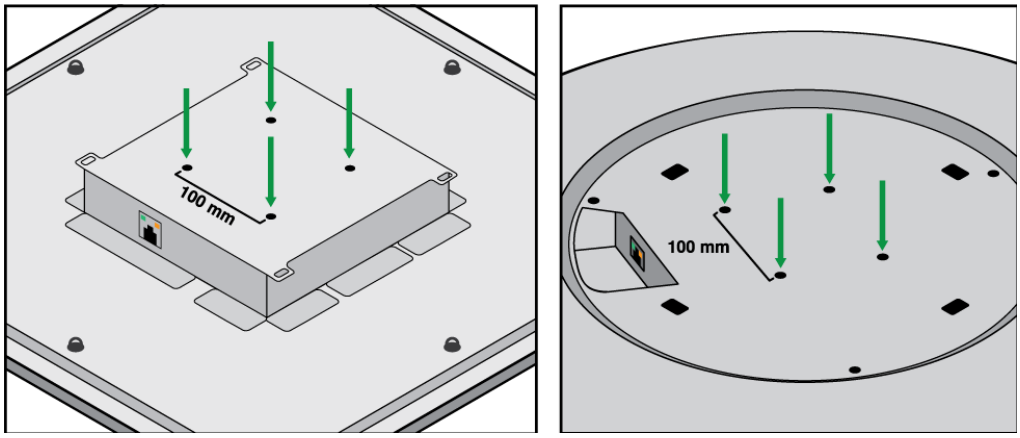
To install:

1. Remove the knockout you plan to use on the junction box.
2. Remove the 4 screws from the microphone as shown.



3. Align the junction box with the screw holes. If possible, plug the network cable into the microphone before securing the junction box.
4. Reinstall the 4 screws to secure the junction box to the microphone.

VESA Standardized Mounting

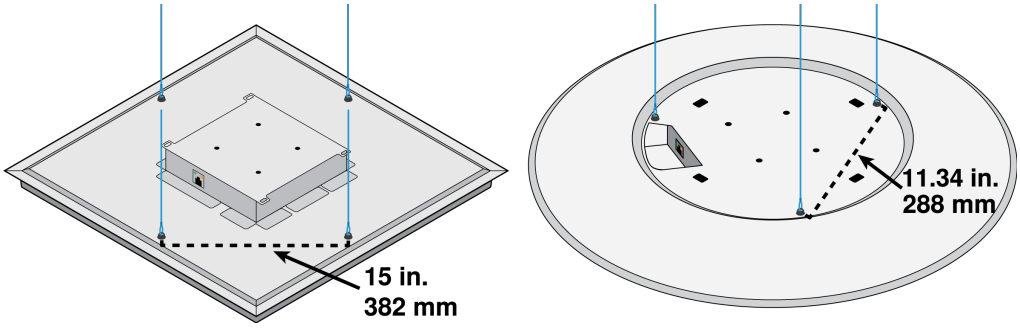


The rear plate has 4 threaded holes for attaching the microphone to a VESA mounting device. The mounting holes follow the VESA MIS-D standard:

- Screw specification: M4 thread (hole depth = 9.15 mm)
- Hole spacing: 100 mm square

The VESA mounting holes work with Shure's A900-PM and A900-PM-3/8IN accessories to mount the microphone on a pole.

Suspending from the Ceiling



Suspend the microphone using your own equipment, or with Shure's A900-GM kit (includes mounting cables and hooks).

To mount using your own equipment, you will need:

- Braided metal cable or high-strength wire
- Hardware to attach cable to ceiling

1. Attach the mounting cables to the 12 mm diameter eyelet screws on the microphone.
2. Attach the cables to the ceiling using the appropriate hardware.

Install on a Ceiling with the A900-CM

Attach array microphones to the ceiling with the A900-CM mounting kit.

Refer to the A900-CM user guide to learn how to install on other ceiling materials.

Hard Ceiling Mounting

You can mount square ceiling array microphones in hard ceilings without a tile grid using the A910-HCM accessory.

Learn more at www.shure.com.

Dante Channels

The automatic coverage setting changes the number of Dante outputs on the MXA920.

Automatic Coverage On

- 1 automix output with IntelliMix DSP for all coverage areas
- 1 AEC reference input

Note: When automatic coverage is on, Dante Controller shows 8 transmit channels and the automix output. The automix output is the only channel that sends audio with automatic coverage on.

Automatic Coverage Off

- Up to 8 separate Dante outputs (1 for each lobe)
- 1 automix output with IntelliMix DSP
- 1 AEC reference input

IntelliMix DSP

This device contains IntelliMix digital signal processing blocks that can be applied to the microphone's output. The DSP blocks include:

- Acoustic echo cancellation (AEC)
- Automatic gain control (AGC)
- Noise reduction
- Compressor
- Delay

To access, go to the IntelliMix view. Double-click a DSP block to open the settings.

DSP Best Practices

- Apply DSP blocks as needed. Run a test of your system without DSP and then add processing as needed to fix any issues that you hear in the audio signal.
- Use Designer's auto route as a DSP starting point. The auto route process applies Shure's recommended DSP settings for the room's mix of devices. You can then customize settings to fit your needs.
- Avoid duplicating DSP blocks.
  - Example: If you plan to use a mic's noise reduction processing, turn off the noise reduction on any connected processors (Shure or third-party). Conversely, if you use noise reduction on a connected processor, turn off the mic's noise reduction.
- Unless you encounter video that lags behind audio, set delay to off.

Acoustic Echo Cancellation (AEC)

Use acoustic echo cancellation (AEC) to keep your microphone signal clear and free of distracting echo on calls. Echo happens when your microphone picks up sound from your loudspeakers and sends it back to others on the call as echo.

AEC is a benefit for everyone else on the call: they hear you clearly with no annoying echo.

AEC Tips

When possible, optimize the acoustic environment using these tips:

- Reduce loudspeaker volume.
- Position loudspeakers farther from microphones.
- Avoid pointing loudspeakers directly at microphone coverage areas.
- Avoid duplicating AEC processing.
  - Example: If you plan to use a mic's AEC processing, turn off the AEC on any connected processors (Shure or third-party). Conversely, if you use AEC on a connected processor, turn off the mic's AEC.
  - Designer's auto route process selects Shure's recommended device for AEC processing based on the mix of Shure devices in the room.

Provide a Reference Signal for AEC

To apply AEC, provide a far end reference signal. For best results, use the same signal that feeds the room's local reinforcement system.

Designer's auto route process automatically routes an AEC reference signal. However, you should check that Designer chooses the reference signal you want to use.

To provide a reference signal, use Designer or Dante Controller to route the far end reference signal to the device's AEC Reference In channel.

Note: Always send an AEC reference signal to microphones with AEC processing, even if you are using a separate DSP for AEC. Some microphones use this reference signal to help improve mic coverage. Designer's auto route feature automatically creates these routes.

AEC Settings

Reference Meter	Use the reference meter to visually verify the reference signal is present. The reference signal should not be clipping.
ERLE	Echo return loss enhancement (ERLE) displays the amount of echo being removed in dB. If the reference source is connected properly, ERLE meter activity corresponds to the reference meter.
Reference	Indicates which channel is serving as the far end reference signal.
Non-Linear Processing	The primary component of the acoustic echo canceller is an adaptive filter. Non-linear processing supplements the adaptive filter to remove any residual echo caused by acoustic irregularities or changes in the environment. **Auto (default)*:** For best results, use this setting. The non-linear processing level automatically adapts to changes in the acoustic environment. **Low:** Use in rooms with controlled acoustics and minimal echoes. This setting provides the most natural sound for full duplex. **Medium:** Use in typical rooms as a starting point. If you hear echo artifacts, try using the high setting.

	High: Use to provide the strongest echo reduction in rooms with bad acoustics, or in situations where the echo path frequently changes.

* Auto setting available in firmware 6.2 and newer

Noise Reduction

Noise reduction keeps the focus on your voice by reducing the level of background noise. Noise reduction is best for constant noise sources such as:

- Loud HVAC systems
- Network switch or server fans
- Display projectors

Noise reduction is a dynamic processor which calculates the noise floor in a room and removes noise throughout the entire spectrum with maximum transparency.

Settings

Options: Low, medium, high, or auto

The noise reduction setting reflects the amount of reduction in dB.

For best results, use the Auto setting so that the noise reduction level adapts to changes in the acoustic environment.

Automatic Gain Control (AGC)

Automatic gain control automatically adjusts channel levels to ensure consistent volume for all talkers in all scenarios. For quieter voices, it increases gain. For louder voices, it attenuates the signal.

Enable AGC on channels where the distance between the talker and the microphone may vary, or in rooms where many different people will use the conferencing system.

Automatic gain control happens post-gate (after the automixer) and does not affect when the automixer gates on or off.

Target Level (dBFS)

Use −37 dBFS as a starting point to ensure adequate headroom and adjust if necessary. This represents the RMS (average) level, which is different from setting the input fader according to peak levels to avoid clipping.

Maximum Boost (dB)

Sets the maximum amount of gain that can be applied

Maximum Cut (dB)

Sets the maximum attenuation that can be applied

Tip: Use the boost/cut meter (not available on all microphones) to monitor the amount of gain added or subtracted from the signal. If the meter is always reaching the maximum boost or cut level, adjust the input fader so the signal is closer to the target level.

Delay

Use delay to synchronize audio and video. When a video system introduces latency (where you hear someone speak, and their mouth moves later), add delay to align audio and video.

Delay is measured in milliseconds. If there is a significant difference between audio and video, start by using larger intervals of delay time (500-1000 ms). When the audio and video are slightly out of sync, use smaller intervals to fine-tune.

Compressor

Use the compressor to control the dynamic range of the selected signal.

Threshold

When the audio signal exceeds the threshold value, the level is attenuated to prevent unwanted spikes in the output signal. The amount of attenuation is determined by the ratio value. Perform a soundcheck and set the threshold 3-6 dB above average talker levels, so the compressor only attenuates unexpected loud sounds.

Ratio

The ratio controls how much the signal is attenuated when it exceeds the threshold value. Higher ratios provide stronger attenuation. A lower ratio of 2:1 means that for every 2 dB the signal exceeds the threshold, the output signal will only exceed the threshold by 1 dB. A higher ratio of 10:1 means a loud sound that exceeds the threshold by 10 dB will only exceed the threshold by 1 dB, effectively reducing the signal by 9 dB.

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

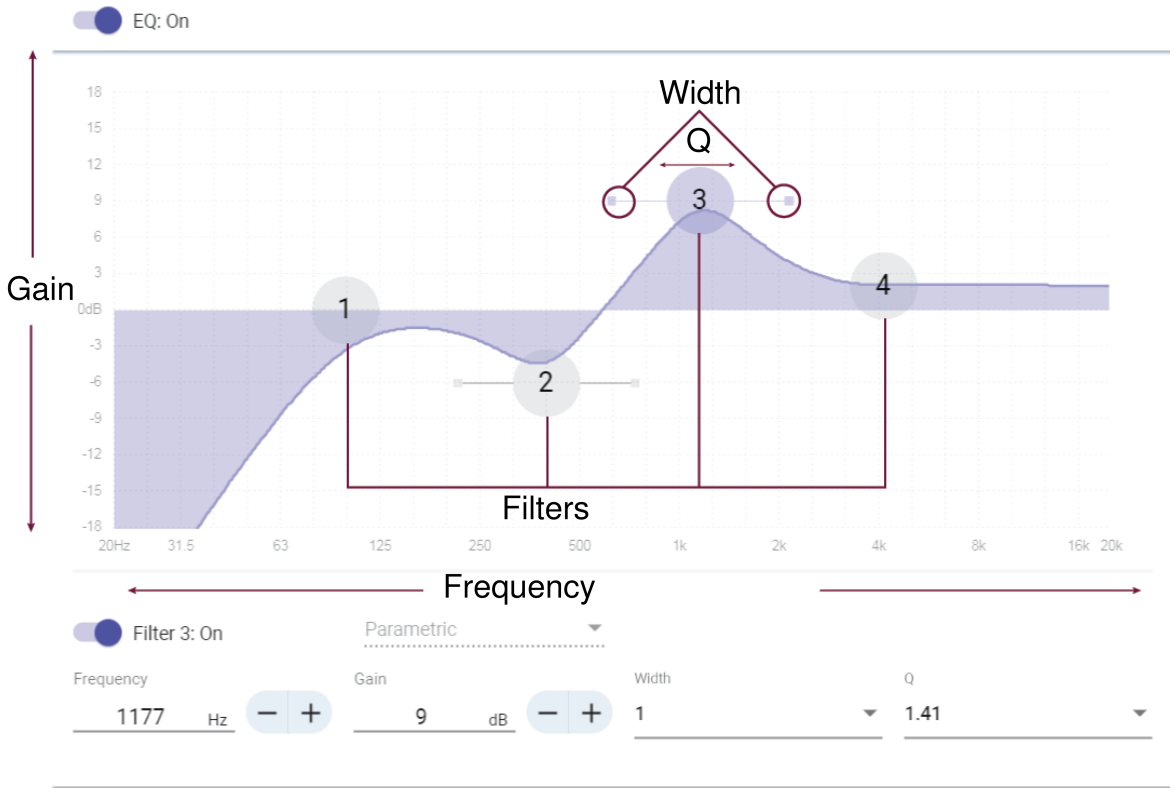
Setting Filter Parameters

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

PEQ Filter Settings

Setting	Function
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 or boost
Gain	Adjusts the level for a specific filter (+/− 18 dB)

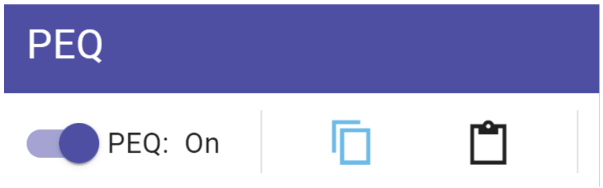
Setting	Function
Q	Adjusts the range of frequencies affected by the filter. As this value increases, the bandwidth becomes thinner.
Width	Adjusts the range of frequencies affected by the filter. The value is represented in octaves. **Note:** The Q and width parameters affect the equalization curve in the same way. The only difference is the way the values are represented.



Copy and Paste Equalizer Channel Settings

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

1. Select the PEQ of the desired channel.
2. Click copy.
3. Select the channel to apply the PEQ setting to and click paste.



Equalizer Applications

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

Uses for EQ

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	Identify the specific frequency range that "excites" the room: 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	Identify the specific frequency range that "excites" the room: 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.

EQ Contour

Use the EQ contour to quickly apply a high-pass filter at 150 Hz to the microphone's signal.

Select EQ contour to turn it on or off.

Automix

Automix Channel

This channel automatically mixes the audio from all selected channels to a single output. To adjust the automix channel settings, go to:

- **MXA710, MXA910, MXA920:** `[Your device]` > `IntelliMix` > `Properties`
- **P300 and IntelliMix Room:** `[Your device]` > `Automixer` > `Properties`

All IntelliMix DSP blocks can be applied to the automix channel.

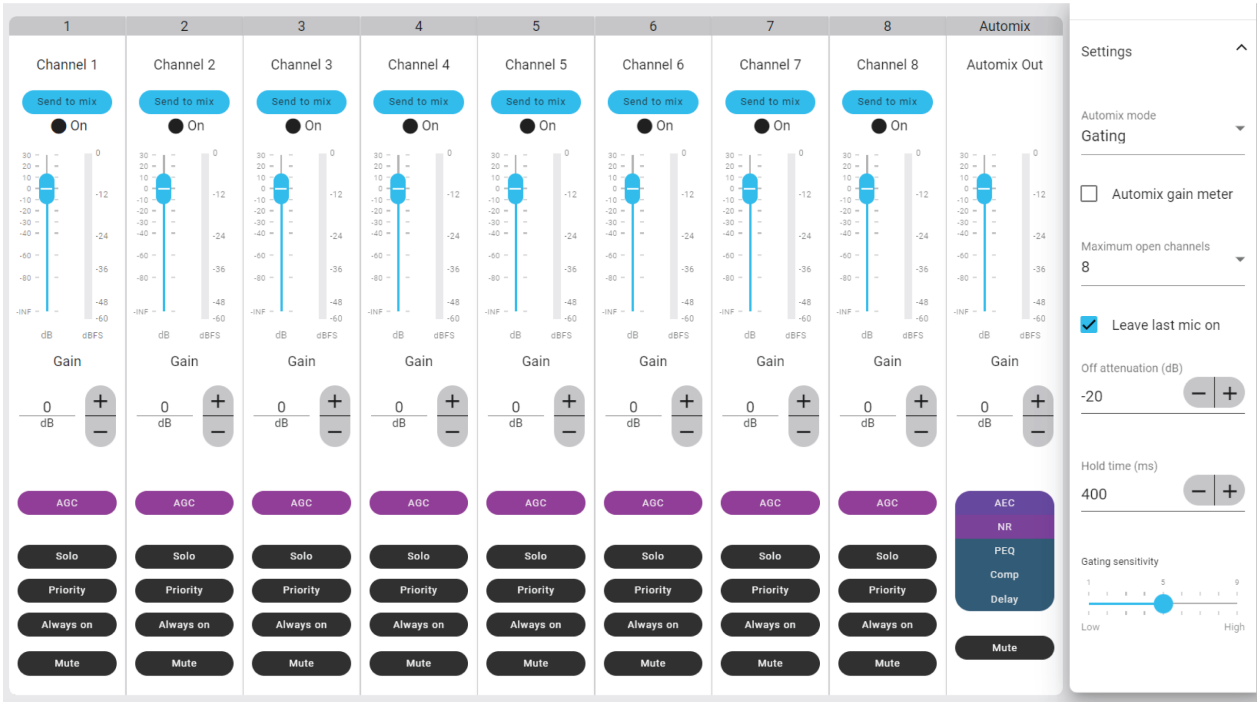
To use the automix channel, do the following:

1. Check that Send to mix is automatically selected (blue) for all channels. To exclude channels from the automix channel and treat them as individual direct outputs, deselect Send to mix (gray).
2. Route the automix channel in Designer or Dante Controller.

Automix Settings

Use these settings to adjust the automixer. Designer adjusts these settings automatically if you use the auto route feature, but you can customize to fit your installation. To adjust, go to:

- **MXA710, MXA910, MXA920:** `[Your device]` > `IntelliMix` > `Properties`
- **P300 and IntelliMix Room:** `[Your device]` > `Automixer` > `Properties`



Automix Settings

Setting	Function
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

Setting	Function
Maximum open channels	Sets the maximum number of simultaneously active 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.
Priority	When selected, this channel gate activates regardless of the number of maximum open channels.
Always on	When selected, this channel will always be active.
Send to mix	When selected, sends the channel to the automix channel.
Solo	Mutes all of the other channels

Note: Not all settings are available on all automixers.

Speech Gating Threshold and Enhanced Noise Filtering

Speech gating threshold and enhanced noise filtering work together to enhance the microphone's sound.

- **Speech gating threshold:**
  - Provides increased control over the automixer gating decision, focusing on speech sources over noise sources
  - Low setting preserves speech better
  - High setting is more aggressive at gating off for noise sources
- **Enhanced noise filtering:**
  - Removes noise from the automix output signal
  - Is designed for noises that are inside a microphone’s coverage area but that are not located next to a person talking

Examples of noise include:

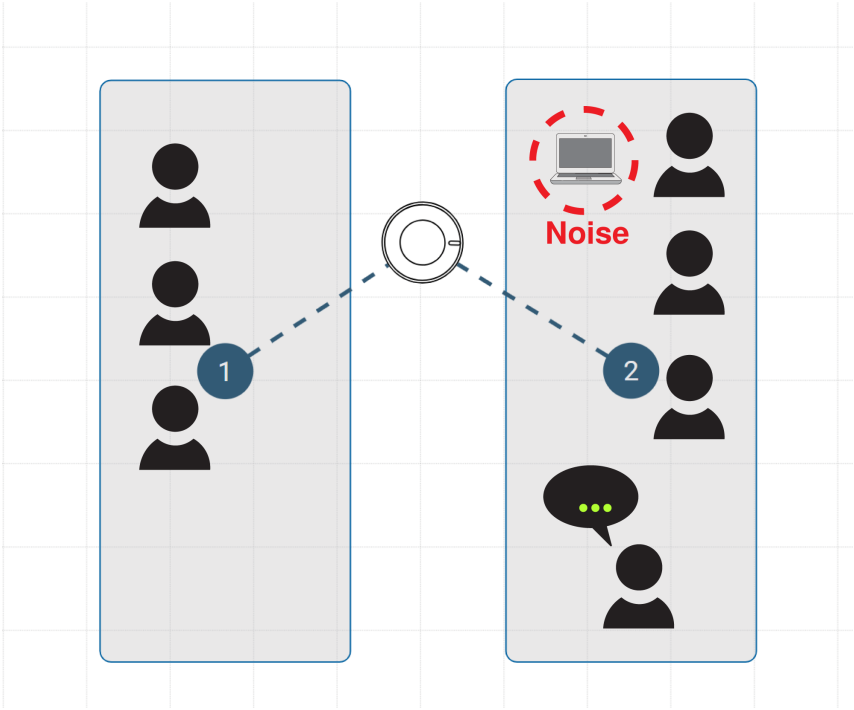
- Shuffling papers across the table from a person talking
- Keyboard across the table from a person talking
- Loud food container across the table from a person talking

Shure DSP Features for Noise

DSP Feature	Purpose	Available in:
Noise reduction	Removes constant background noise (such as HVAC)	All devices with IntelliMix DSP
Enhanced noise filtering	Removes noise sources that: Are in the microphone's coverage area Are not immediately next to a person talking. Example: If a person talking opens a bag of chips, you will still hear the bag.	Firmware 6.2 and newer for: MXA710, MXA901, MXA902: On or off settings MXA920: Low, medium, or high settings

DSP Feature	Purpose	Available in:
AI denoiser	Removes any noise, even noise that happens immediately next to a person talking	IntelliMix Room software

In this example, enhanced noise filtering removes loud typing sounds happening on one end of coverage area 2 and focuses on the person speaking.



Speech gating threshold and enhanced noise filtering work when automatic coverage is on or off.

To use speech gating threshold and enhanced noise filtering:

1. In the microphone's control software, go to `IntelliMix` > `Properties`. Scroll down to adjust the 2 settings.
2. Make a test call in a room with 2 people to test the speech gating threshold and enhanced noise filtering settings. Start with both settings at the highest level.
3. Have both people in the room talk at the same time. The far-end listener should listen for unwanted speech attenuation. If there is speech attenuation, lower the speech gating threshold. Repeat until there is no unwanted speech attenuation.
4. Next, have one person talk and the other person create noise at the same time (use the examples for ideas). The noise source should be away from the person talking. The far-end listener should tell you how the signal sounds. Test and adjust the enhanced noise filtering level as needed.

Tips:

- A high speech gating threshold may detect unvoiced speech sound as noise. If you notice this, lower the speech gating threshold.
- If the microphone is routed to IntelliMix Room and you're using the AI denoiser feature, turn off enhanced noise filtering.
- If using with automatic coverage off and leave last mic on set to off, set the speech gating threshold to low. With this setup, the microphone won't gate off if speech is quiet in comparison to noise.

Automix Modes

To select an automix mode, go to:

- **MXA710, MXA910, MXA920:** [Your device] > IntelliMix > Properties
- **P300 and IntelliMix Room:** [Your device] > Automixer > Properties

Gating

Gating mode delivers fast-acting, seamless channel gating and consistent perceived ambient sound levels. The off attenuation setting is applied to all inactive channels, regardless of the number of active channels.

Gain Sharing

Gain sharing mode dynamically balances system gain between open and closed channels. The system gain remains consistent by distributing gain across channels to equal one open channel. The scaled gain structure helps to reduce noise when there is a high channel count. When fewer channels are used, the off attenuation setting is lower and provides transparent gating.

Manual

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

Security

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 password to make a connection. Encryption is not supported with third-party devices.

In Designer, you can turn on encryption for all devices in a room: [Your room] > Settings > Audio encryption.

To activate encryption in the web application, go to Settings > Audio encryption > Enable encryption.

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.

Set Up the 802.1X Protocol for a Device

Select Shure devices support the IEEE 802.1X port access protocol for network authentication.

Important: To use the 802.1X security protocol with Shure devices, set the network switch to multiple host authentication. You must also make accommodations to allow the audio NIC to connect to the network. The audio NIC doesn't support the 802.1X protocol.

To set up 802.1X, you will need:

- Details about your authentication server's EAP method

- Any required credentials or certificates for that method, for example:
  - MD5 and PWD
    1. User ID and passphrase
  - TLS and PEAP
    1. User ID and passphrase
    2. Certificate (with certificate types) in the .PEM format
- Any passwords to access the devices if they are password locked

Setting up 802.1X is a two-part process.

Step 1: Configure Settings on Test Network

1. Connect the device to a test network that *does not* have 802.1X enabled, and discover it using Designer.
2. Set a device password if desired.
3. Double-click the device and go to `Settings` > `Network` > `802.1X`.
4. Choose your EAP method from the menu.
5. Enter any required credentials and load any necessary certificates.
6. Press Save to save the 802.1X settings to the device.
7. Enable 802.1X and select Reboot now.

Step 2: Connect to a Credentialed Network

1. Connect your device to the credentialed network.
2. Ensure that Designer is connected to the credentialed network.
3. Verify that the device appears in Designer. If it doesn't, reconnect to the test network and check all 802.1X settings for the selected EAP method.

Turn Off or Clear 802.1X Settings

You can turn off 802.1X settings temporarily, or clear them from the device. Open the device and go to `Settings` > `Network` > `802.1X`

- **Disable**: Click the 802.1X switch to turn off 802.1X settings. Click the switch again to enable 802.1X.
- **Clear**: Click Clear 802.1X settings to remove 802.1X settings from the device.

Note: Resetting to factory default clears all 802.1X settings.

Change 802.1X Settings

You may need to change a device's 802.1X settings if the enterprise's 802.1X settings are changing. The best way to do this is to change the 802.1X settings on the devices, and then make changes to the authentication server.

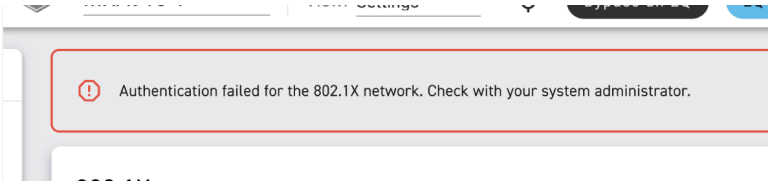
To change device settings:

1. While still connected to the credentialed network, find the device in Designer and go to `Settings` > `Network` > `802.1X`.
2. Make changes and click Save.
3. Make any changes to the authentication server.
4. Reboot your devices. The devices should connect to the credentialed network with the updated 802.1X settings.

Troubleshooting 802.1X Setup Issues

If the device doesn't appear in Designer on the credentialed network, there's a problem with the device's 802.1X settings. To troubleshoot, take the device off the credentialed network and connect it to the test network. You can make any necessary changes to the 802.1X settings, and then reconnect to the credentialed network.

If you attempt to enable 802.1X on a device, but the authentication fails, you will see this notification:



If this occurs, check with your system administrator.

Networking

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.
- For most uses, connect Shure networked devices to the same network and set to the same subnet.
  - If needed, use Designer to discover devices across subnets.
- Disable unused network ports on Shure devices when possible.
- 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.
- Keep device firmware updated to avoid compatibility issues.

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

For more information, see our FAQ about switches to avoid.

Device IP Configuration

This Shure device uses 2 IP addresses: one for Shure control, and one for Dante audio and control. For most installations, the Shure control and Dante audio IP addresses should be in the same subnet range.

- **Shure control**
  - Carries data for Shure control software, firmware updates, and third-party control systems (such as AMX or Crestron)
- **Dante audio and control**
  - Carries Dante digital audio and control data for Dante Controller
  - Requires a wired, gigabit Ethernet connection to operate

To access these settings in Designer, go to `[Your device]` > `Settings` > `IP configuration`.

Devices with 1 RJ-45 port do not natively support Dante split or redundant modes.

Note: Refer to our FAQ if you're using any Shure profiles on NETGEAR M4250-series switches.

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+

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

Ports, Protocols, and Firewall Rules

For information about IP ports and protocols or firewall rules, go to:

- IP Ports and Protocols for Shure Devices
- Firewall Rules for Shure Software Applications

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.

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 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.

Refer to Dante Domain Manager's documentation for more information.

Dante Flows for Shure Devices

Dante flows get created any time you route audio from one Dante device to another. One Dante flow can contain up to 4 audio channels. For example: sending all 5 available channels from an MXA310 to another device uses 2 Dante flows, because 1 flow can contain up to 4 channels.

Every Dante device has a specific number of transmit flows and receive flows. The number of flows is determined by Dante platform capabilities.

Dante Flows for Shure Devices

Dante Platform	Shure Devices Using Platform	Transmit Flow Limit	Receive Flow Limit
Brooklyn II	ULX-D, SCM820, MXWAPT, MXWANI, P300, MXCWAPT	32	32
Brooklyn II (without SRAM)	MXA920, MXA910, MXA902, MXA710, AD4, AD600, APXD2	16	16
IP Core	MXA920-V3, MXA902-V3, MXA901, DCA901	32	32
Ultimo/UltimoX	MXA310, ANI4IN, ANI4OUT, ANIUSB-MATRIX, ANI22, MXN5-C	2	2
DEP	ANIUSB-MATRIX-V3, MXN-AMP, MXN5-C-V3, MXN-6	2	2
DAL	IntelliMix Room	16	16

Learn more about Dante flows in our FAQs or from Audinate.

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.

Paint the MXA920

Paint Square Array Microphones

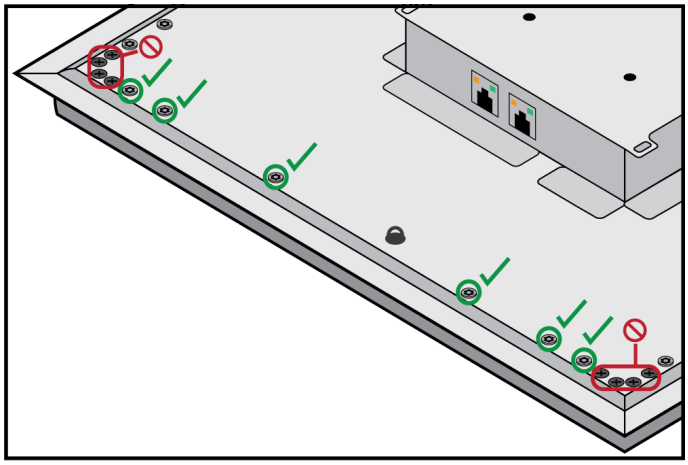
You can paint the grille and frame of square ceiling array microphones to blend in with a room's design.

Note: Don't remove any of the screws on the MXA902's loudspeaker enclosure while painting.

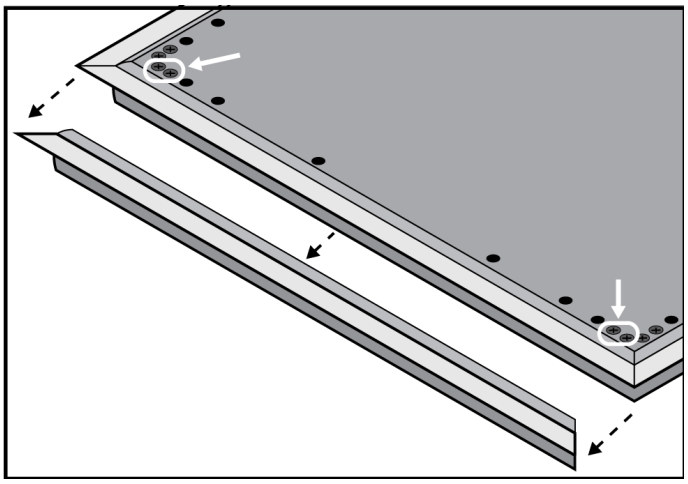
Step 1: Remove the Frame and Grille

1. On each side of the frame, remove the 6 screws and washers that attach the main assembly to the frame.

   Important: Do not remove the 4 recessed screws in each corner.



2. Carefully lift the assembly out of the frame.
3. Remove the gray plastic LED lightpipe. Leave the black plastic guide in place.
4. Remove all 4 recessed screws from one side of the frame. Remove that side of the frame.



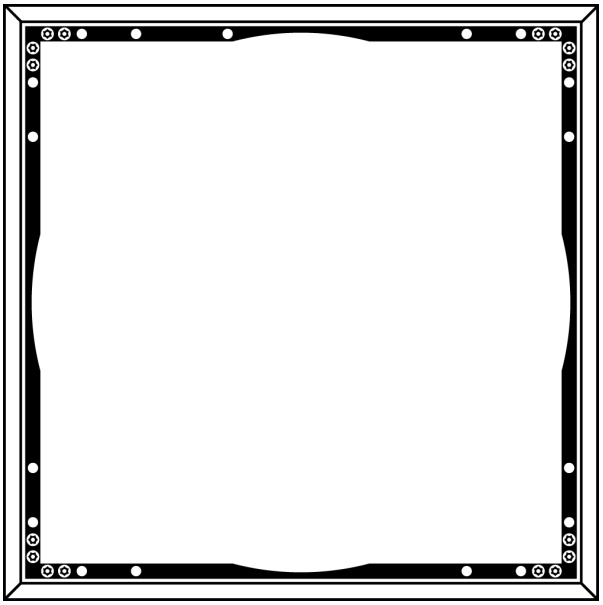
5. Slide the flat grille out of the frame.
6. Carefully remove the foam piece from the grille. Pull from the edges, where it is attached with hook-and-loop fastener strips.

   Important: *Do not paint the foam.*

7. Before painting, reinstall the side of the frame you removed in step 1.4.

Step 2: Mask and Paint

1. Use masking tape to cover the entire extrusion (highlighted in black) that runs along the inside of the frame. This ensures that the necessary metal pieces make contact when reassembled.



2. Use masking tape to cover the hook-and-loop fastener strips on the grille.
3. Paint the frame and grille. Let them dry completely before reassembling. Do not paint any part of the main assembly.

Step 3: Reassembly

1. Attach the foam piece to the grille with the hook-and-loop fastener strips.

2. Remove one side of the frame as in step 1.4. Slide the grille back into the frame.
3. Attach the remaining side of the frame and secure it with the 4 screws.
4. Attach the LED lightpipe to the black plastic guide piece.
5. Align the LED with the lightpipe and put the main assembly back in place on the frame.

   Note: *The label on the assembly is in the corner that corresponds to the LED.*

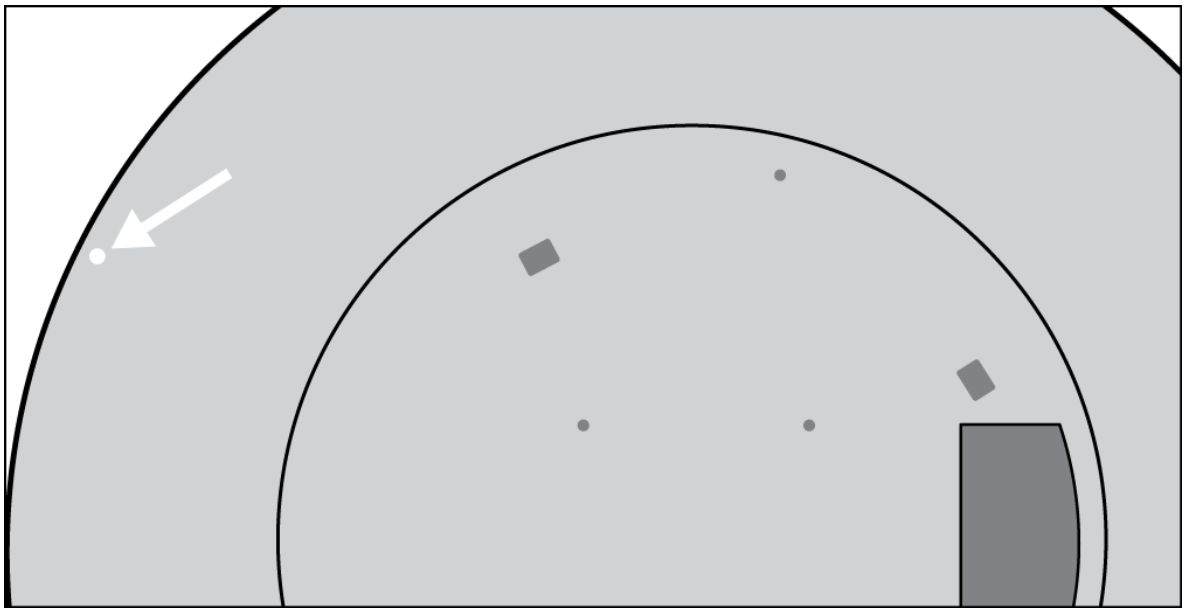
6. Install 6 screws per side to secure the main assembly to the frame. Do not over-tighten.

Paint MXA920-R Microphones

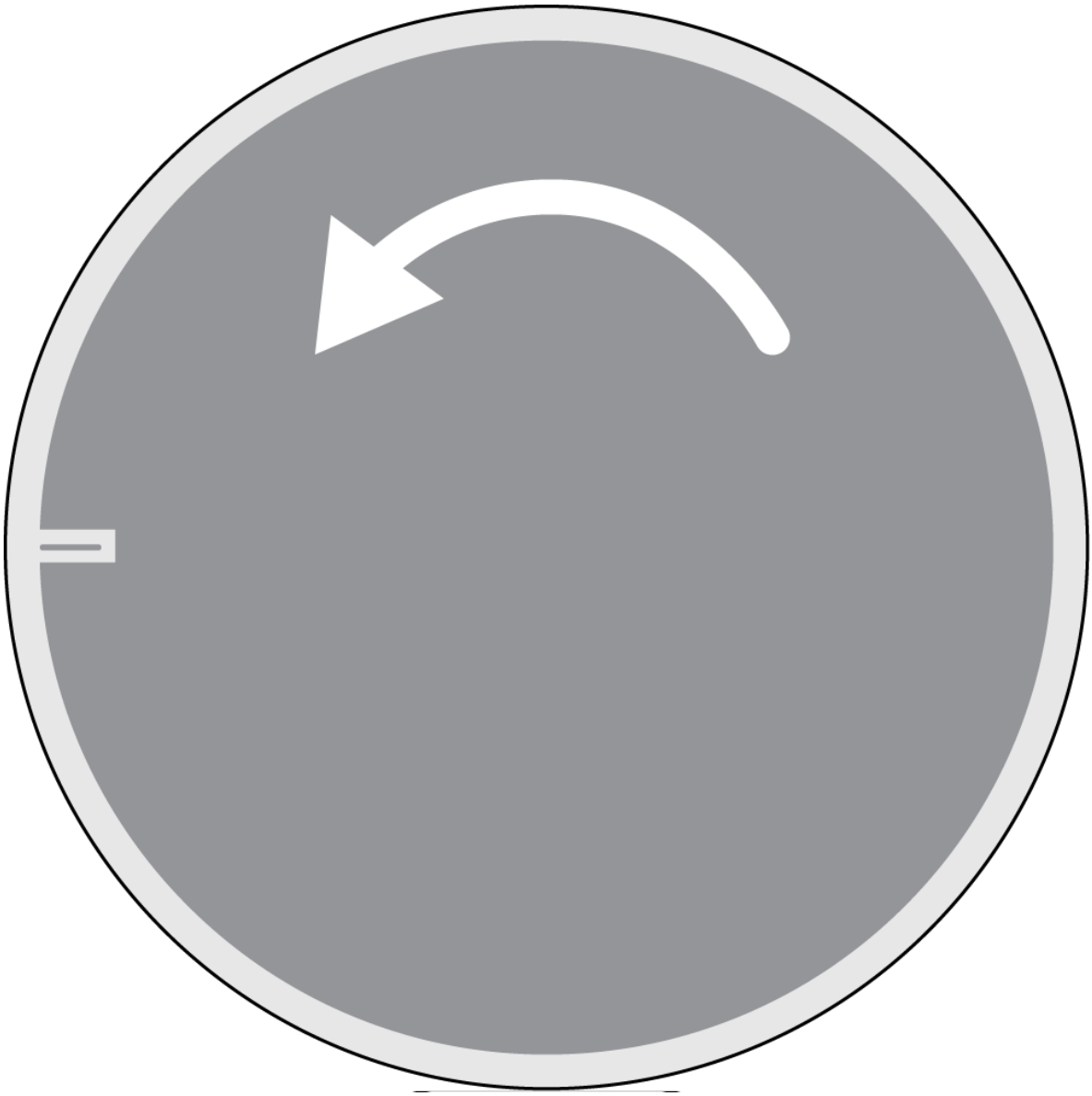
The grille and back cover of round array microphones can be painted to blend in with a room's design.

Step 1: Remove and Paint the Grille

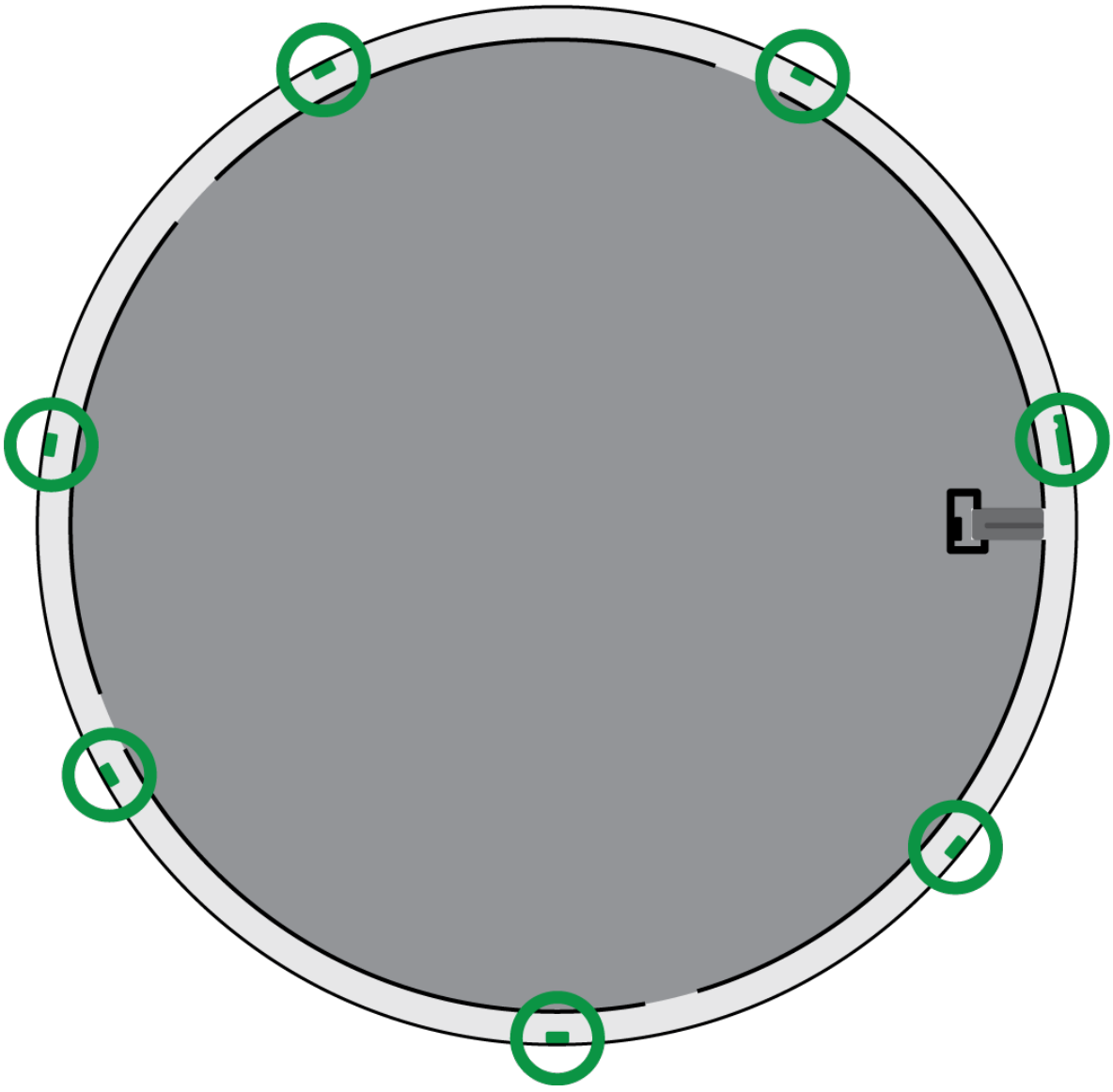
1. Loosen the set screw that connects the grille to the back cover. Turn the microphone over.



2. Rotate the grille as shown to release it from the back cover. Lift it up and out of the tabs holding it in place.



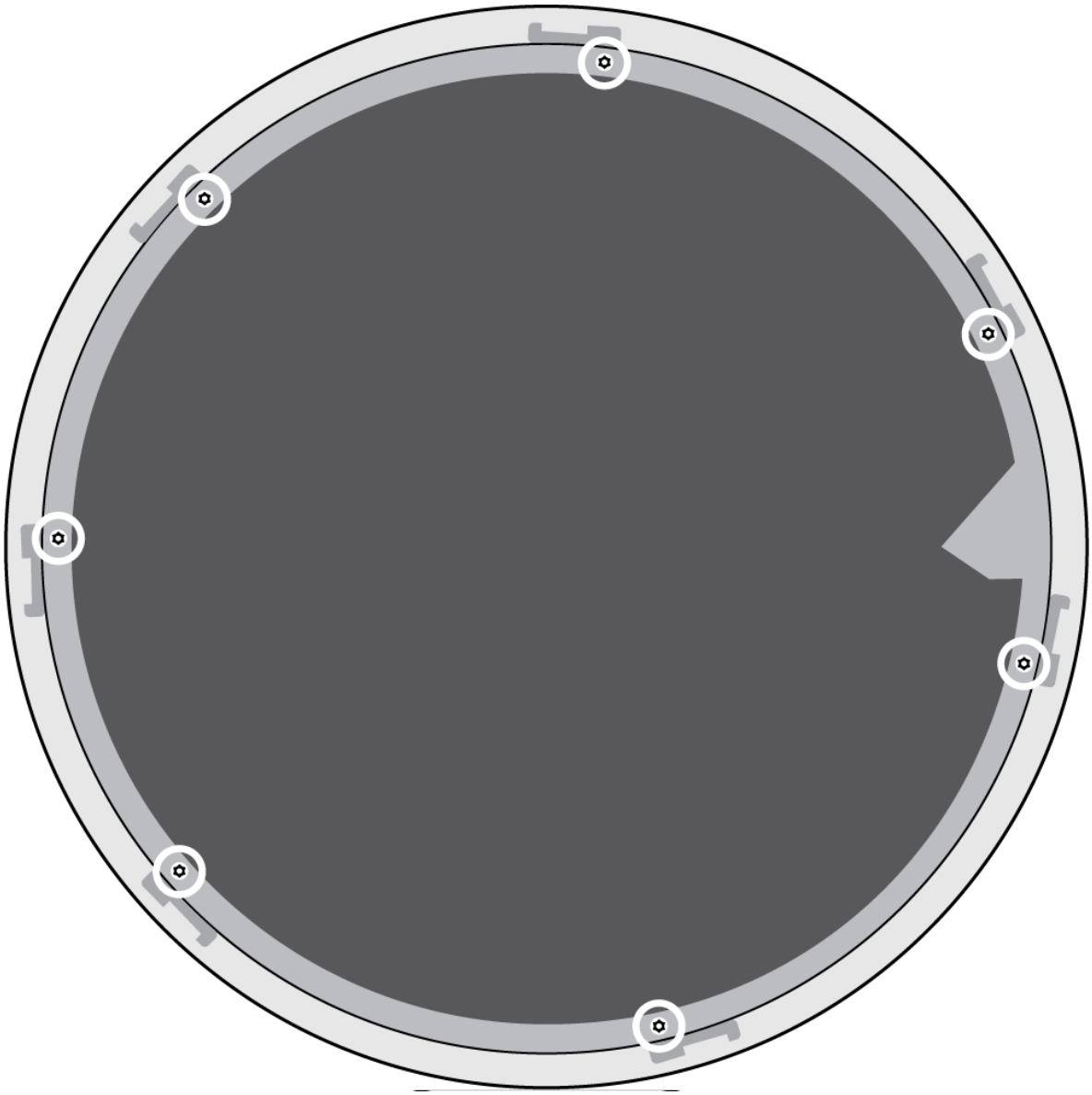
3. Carefully remove the fabric piece from the grille. Pull from the edges where it is attached with Velcro strips. Do not paint the fabric.
4. Hold the edges of the black plastic guide in place and pull up on the clear lightpipe to unsnap it. Leave the guide in place.
5. Mask the 7 bare metal tabs on the grille.



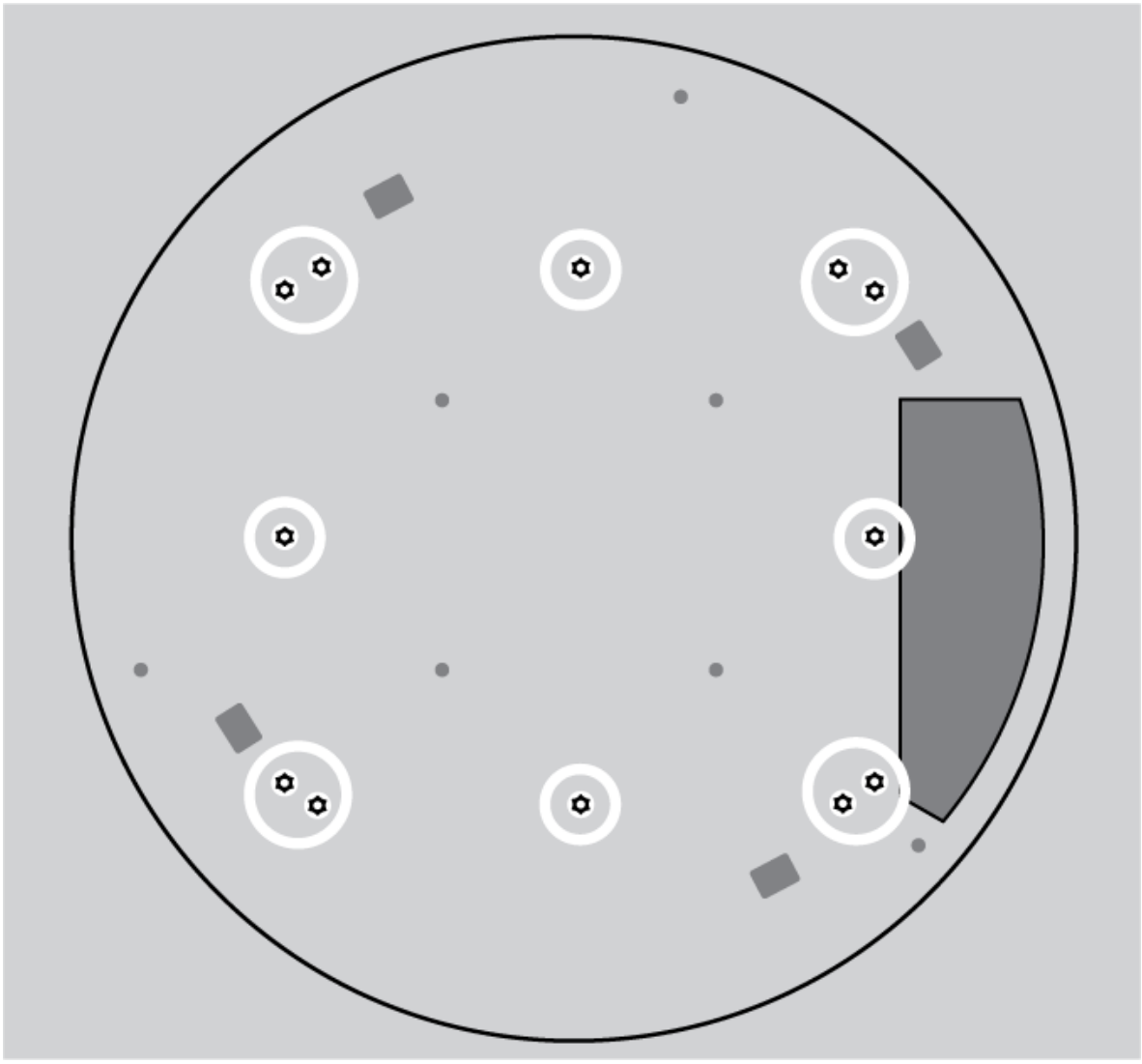
6. Paint the grille.

Step 2: Remove and Paint the Back Cover

1. Remove the 7 screws on the aluminum support panel. Turn the back cover over.



2. Remove the 12 screws that attach the back cover to the processor enclosure. Set the processor enclosure aside with the black board facing up.

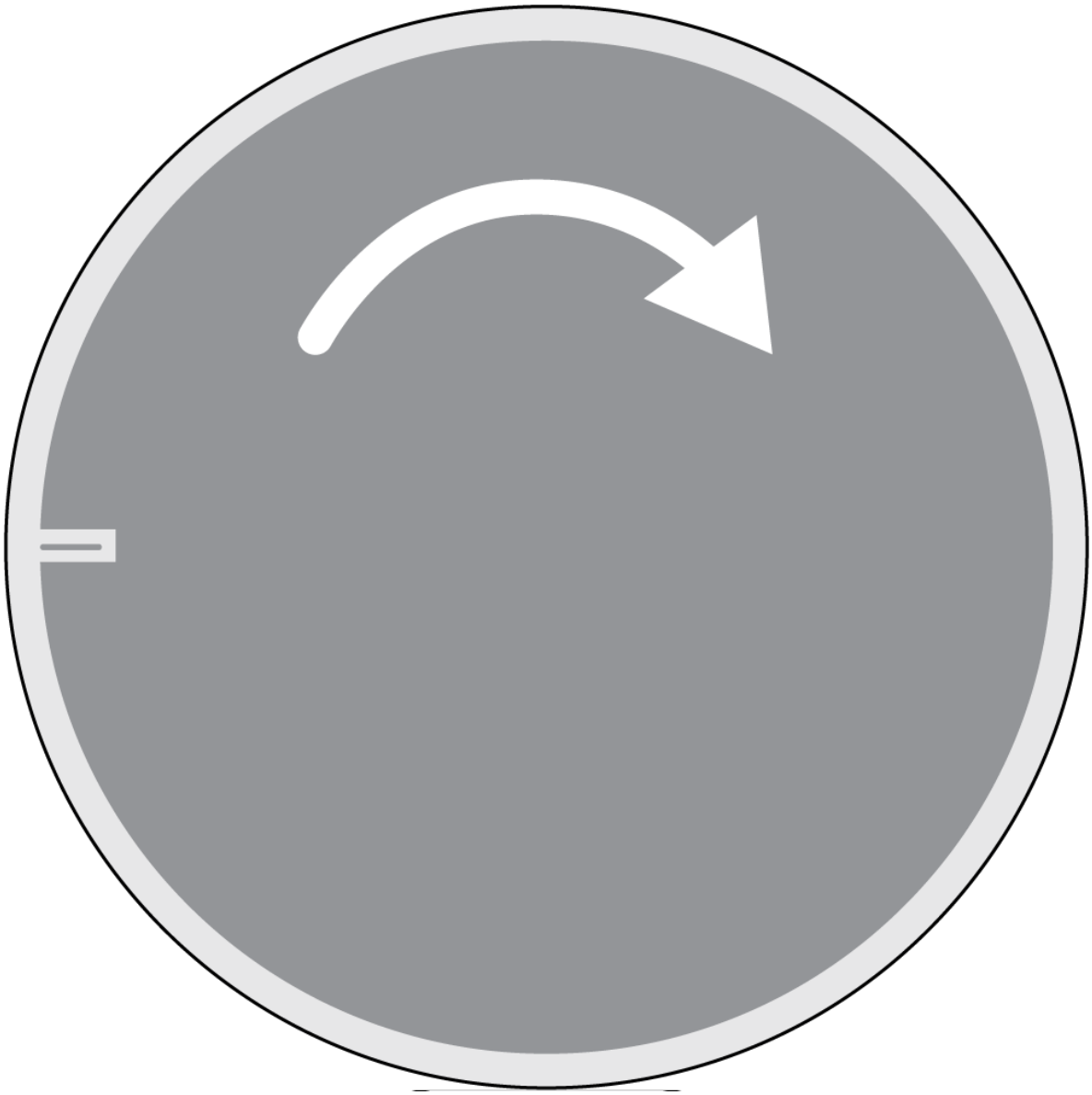


3. Mask the entire flat area in the center of the back cover. Mask the 7 tabs on the inside of the back cover to keep paint out of the screw threads.
4. Paint the outside of the back cover.

Step 3: Reassemble the Microphone

Let the paint dry before reassembling.

1. Use the 12 screws to attach the back cover to the processor.
2. Use the 7 screws to reattach the aluminum support panel.
3. Reinstall the lightpipe on the grille by snapping it into place.
4. Attach the fabric piece to the grille.
5. Align the grille with the 7 tabs on the back cover. Set it down and rotate the grille as shown to engage the tabs.



6. Tighten the set screw.

Monitor and Control with External Systems

Use the REST API

This device has a REST API for easy integration with other enterprise monitoring and control solutions. Use the API to control this device with third-party monitoring and control systems.

Common applications:

- Mute
- LED color and behavior
- Receive coverage information

Shure continues to add API capabilities. Find our complete API documentation at shure.stoplight.io.

The device's API is on by default. To restrict access to the API, go to `Settings` > `Services` in Designer or the device's web application.

Available in firmware 6.2 and newer

Use Command Strings

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:

shure.com/docs/commandstrings/mxa920

Integrate the MXA920 with Camera Control Systems

MXA920 microphones provide data about talker position, lobe position, and other settings. You can use this data to integrate the microphone with camera control systems.

There are 2 ways to access the data:

- **Option 1:** Use the REST API
- **Option 2:** Use command strings

Tips for Integrating with Camera Control Systems

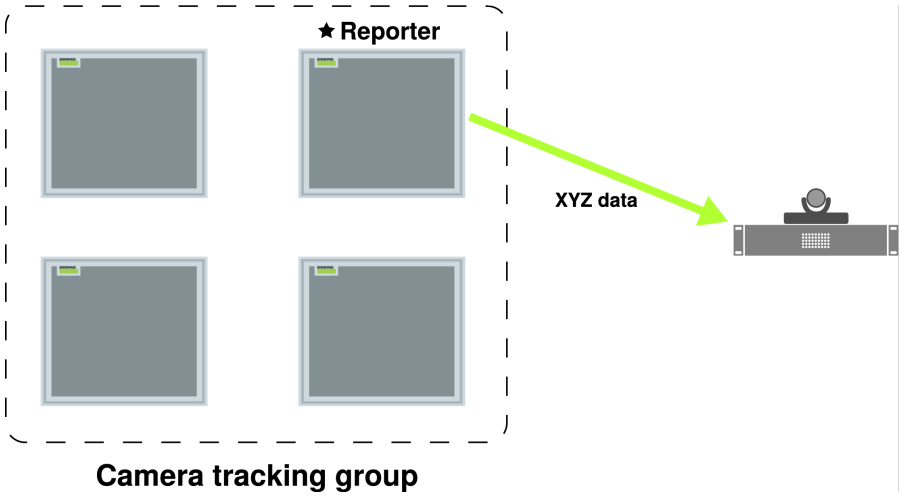
Use these tips to get the best results when integrating a Shure array microphone with a room's camera tracking system.

- **Talk toward the microphone**
  - When possible, ask talkers to speak in the direction of the microphone. The microphone may take longer to send talker positions if they are facing 90 to 180 degrees off-axis from the microphone.
- **Speak up**
  - Ask quieter talkers to speak up. The microphone can take longer to send position information for quieter talkers.
- **Prep the room**
  - Acoustically treated rooms that are less reverberant usually sound better and help improve your microphone signal. In more reverberant rooms, the microphone can take longer to send talker positions or report inaccurate position data.
- **Move closer for single-talker framing**
  - If you need to frame 1 talker at a time with the room's camera, set things up so that the microphone is about 10 feet (3 meters) from the different talker positions. The MXA920 and MXA901 perform well with talkers that are up to 16 feet (4.9 meters) from the microphone, but moving closer helps room cameras with single-talker framing.
  - Position talkers so that they are no more than 90 degrees off-axis from the microphone.

Note: Due to mechanical differences between the round and square MXA920s/MXA925s, round mics report slightly less accurate talker positions than square mics. If talkers are further from the microphone, round mics may report less often than square mics. To mitigate this difference, you can:

- Follow talker position guidelines (maximum of 16 ft. from mic to talker is recommended).
- Make sure that the ceiling height matches what is entered in Designer.
- Use the TALKER_POSITION_SENSITIVITY command to increase the number of reports from the mic. This change may have other unintended effects, so test thoroughly.

Send Talker Position Data from Multiple Microphones



Use the Optimize tracking feature in Designer to send aggregated talker position data. Use the feature to have 1 reporter microphone send talker position data from up to 3 additional follower microphones to camera systems.

Camera tracking groups can have up to 4 microphones in each group. All mics in the group must be the same model (you can't mix MXA925s and MXA901s, for example).

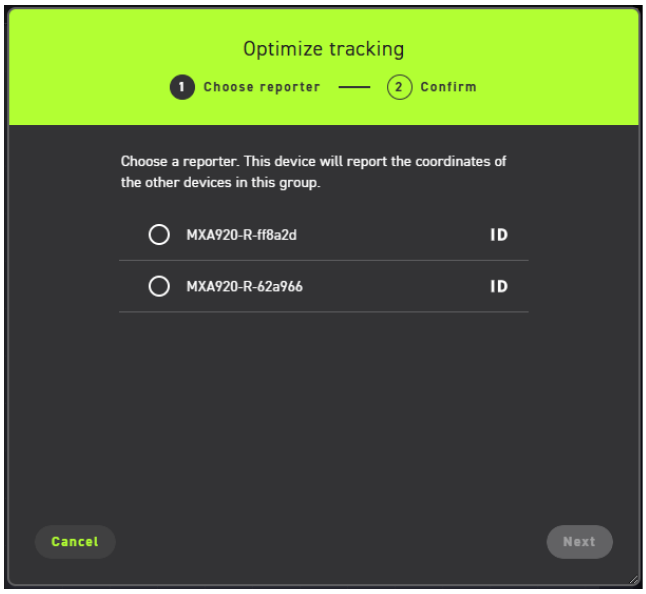
Optimize tracking supports faster deployments since you only need to send command string data from the reporter microphone to the camera system, not all 4 mics individually. The feature also improves X/Y/Z accuracy because the mics can share position data to better find a talker's position.

Before you begin:

- All mics in a camera tracking group must be the same model.
- Follow these placement guidelines:
  - MXA925 or MXA920: Don't install mics more than 12 ft. (3.6 m) from each other.
  - MXA901: Don't install mics more than 8 ft. (2.4 m) from each other.
- Install all mics in your room and arrange them so that the status LED is in the same position on each mic.
- Arrange mics in the Designer room to match how they're installed.
  - Example: If the mics are installed 8.5 ft. apart, they must be spaced 8.5 ft. apart in the Designer room.
- Optimize tracking only works in online rooms.
- Optimize tracking is not compatible with Dante device lock.

To set up optimize tracking:

1. Enter the device height for each mic in `Coverage` > `Properties` > `Position`.
2. Add the mics that you want to group together to the same online room in Designer. Up to 4 mics can be in 1 group.
3. Open the online room from Designer's Online rooms list. Go to the room's Coverage view.
4. Click Optimize tracking. The software guides you through setting up a camera tracking group with up to 4 microphones.



5. After the group is set up, send the appropriate command string data from the reporter microphone to your camera system. The reporter mic sends talker position data from all mics in the camera tracking group.
6. If you delete a mic in a camera tracking group from an online room, you must run Optimize tracking tracking again. This step ensures that the mics provide accurate talker position information.

Troubleshooting

MXA920 Troubleshooting

Problem	Solution
Audio is not present or is quiet/distorted	Check cables. Verify that output channel isn't muted. Check that output levels are not set too low.
Sound quality is muffled or hollow	Check that coverage area is positioned correctly. Use EQ to adjust frequency response.
Microphone does not power on	Check that microphone is plugged in to a PoE source. Check network cables and connections.
Microphone doesn't show up in Designer	Ensure that microphone has power. Make sure microphone is on the same network and subnet as PC. Turn off network interfaces not used to connect to the device (such as Wi-Fi). Check that DHCP server is functioning (if applicable).

Problem	Solution
	Reset the device if necessary. Refer to no online devices FAQ for more help.
Microphone doesn't show up in Shure Web Device Discovery	Ensure that microphone has power. Check that Shure Web Device Discovery is updated. Make sure microphone is on the same network as PC. Check that DHCP server is functioning (if applicable). Reset the device if necessary. Refer to the devices don't appear FAQ for more help.
Flashing red error LED	In the web application, go to `Settings` > `General` > `Export log` to export the device event log. Use the event log to get more information and contact Shure if necessary.
No lights	Go to `[Your device]` > `Settings` > `Lights`. Check if brightness is disabled or if any other settings are turned off. If the device is in a room using Designer's call status feature, mute status LEDs are off when not in a call.
Web application lags in Google Chrome browser	Turn off hardware acceleration option in Chrome.

Additional Resources

- Shure Knowledge Base FAQs
- Command strings for Shure devices
- IP Ports and Protocols for Shure Devices
- Firewall Rules for Shure Software Applications
- Shure API documentation
- Shure Enterprise Networking Troubleshooting Checklist
- Training from Shure Academy
- Shure Collaboration & Conferencing YouTube channel

Download Shure Software

- Shure Designer
- ShureCloud
- Shure Update Utility
- Shure Discovery
- Software and firmware archive

Specifications

General

Coverage Type

Multi-Zone Automatic or Steerable

Power Requirements

Power over Ethernet (PoE), Class 0

Power Consumption

10.1 W maximum

Control Software

Designer or web application

Cable Requirements

Cat5e or higher (shielded cable recommended)

Connector Type

RJ45

Plenum Rating

MXA920-S	UL2043 (Suitable for Air Handling Spaces)
MXA920-R	Not rated

Dust Protection

IEC 60529 IP5X Dust Protected

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

Microphone Elements

113 MEMS

Frequency Response

125 Hz to 20,000 Hz

AES67 or Dante Digital Output

Channel Count	Automatic coverage on	2 total channels (1 output, 1 AEC reference in channel)

	Automatic coverage off	10 total channels (8 independent transmit channels, 1 automix output, 1 AEC reference in channel)
Sampling Rate		48 kHz
Bit Depth		24

Sensitivity
at 1 kHz

−1.74 dBFS/Pa

Maximum SPL
Relative to 0 dBFS overload

95.74 dBSPL

Signal-to-Noise Ratio
Ref. 94 dBSPL at 1 kHz

75.76 dB A-weighted

Latency
Does not include Dante latency

Direct outputs (Automatic coverage off)	13.6 ms
Automix output (Includes IntelliMix processing)	22.9 ms

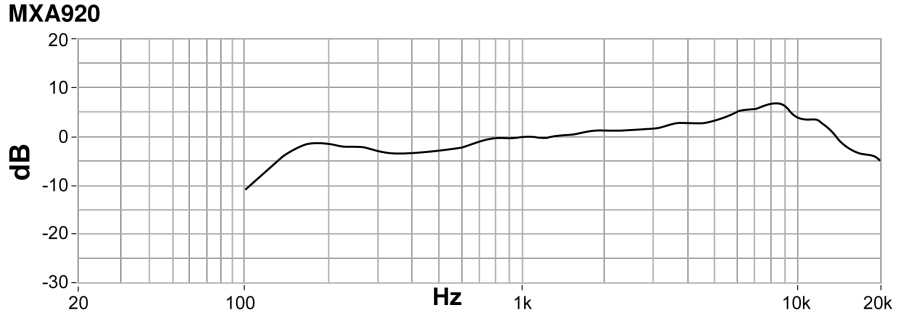
Self Noise
18.24 dB SPL-A

Dynamic Range
77.5 dB

Digital Signal Processing
Automatic mixing, acoustic echo cancellation (AEC), noise reduction, automatic gain control, compressor, delay, equalizer (4-band parametric), mute, gain (140 dB range)

Acoustic Echo Cancellation Tail Length
Up to 250 ms

MXA920 Frequency Response
Frequency response measured directly on-axis from a distance of 6 feet (1.83 m).



Dimensions

Weight

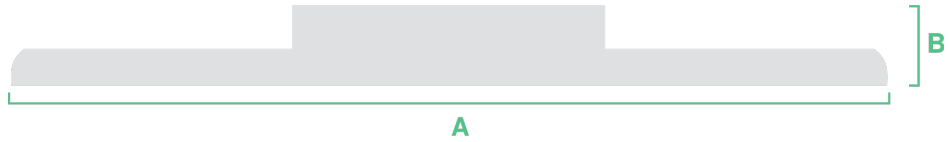
- MXA920-S: 11.8 lbs (5.4 kg)
- MXA920-R: 12.7 lbs (5.8 kg)

MXA920-S



- A (Microphone flange): 0.41 in. (10.5 mm)
- B (Edge to edge): 23.77 in. (603.8 mm)
- C (Height): 2.15 in. (54.69 mm)

MXA920-S-60CM



- A (Edge to edge): 23.38 in. (593.8 mm)
- B (Height): 2.15 in. (54.69 mm)

MXA920-R



- A (Height to top of eyelets): 2.4 in. (61.3 mm)
- B (Outer diameter): 25 in. (635.4 mm)

IMPORTANT SAFETY INSTRUCTIONS

1. READ these instructions.
2. KEEP these instructions.
3. HEED all warnings.

4. FOLLOW all instructions.
5. DO NOT use this apparatus near water.
6. CLEAN ONLY with dry cloth.
7. DO NOT block any ventilation openings. Allow sufficient distances for adequate ventilation and install in accordance with the manufacturer's instructions.
8. DO NOT install near any heat sources such as open flames, radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat. Do not place any open flame sources on the product.
9. DO NOT defeat the safety purpose of the polarized or grounding type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wider blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. PROTECT the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. ONLY USE attachments/accessories specified by the manufacturer.
12. USE only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.



13. UNPLUG this apparatus during lightning storms or when unused for long periods of time.
14. REFER all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
15. DO NOT expose the apparatus to dripping and splashing. DO NOT put objects filled with liquids, such as vases, on the apparatus.
16. The MAINS plug or an appliance coupler shall remain readily operable.
17. The airborne noise of the Apparatus does not exceed 70dB (A).
18. Apparatus with CLASS I construction shall be connected to a MAINS socket outlet with a protective earthing connection.
19. To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture.
20. Do not attempt to modify this product. Doing so could result in personal injury and/or product failure.
21. Operate this product within its specified operating temperature range.

	This symbol indicates that dangerous voltage constituting a risk of electric shock is present within this unit.
	This symbol indicates that there are important operating and maintenance instructions in the literature accompanying this unit.

Important Product Information

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

This device is to be connected only to PoE networks without routing to the outside plant.

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

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

Note: 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.

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. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the manufacturer's instruction manual, may cause interference with radio and television reception.

Notice: The FCC regulations provide that changes or modifications not expressly approved by Shure Incorporated could void your authority to operate this equipment.

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.

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.

This Class B digital apparatus complies with Canadian ICES-003. Cet appareil numérique de la classe B est conforme à la norme NMB-003 du Canada.

CAN ICES-003 (B)/NMB-003(B)

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

Authorized European representative:
Shure Europe GmbH
Global Compliance
Jakob-Dieffenbacher-Str. 12
75031 Eppingen, Germany
Phone: +49-7262-92 49 0
Email: info@shure.de
www.shure.com

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

部件名称	有害物质					
	铅	汞	镉	六价铬	多溴联苯	多溴二苯醚
电路模块	X	○	○	○	○	○
金属模块	X	○	○	○	○	○

线缆及其组件	X	○	○	○	○	○
外壳	○	○	○	○	○	○
电源适配器*	X	○	○	○	○	○
电池组*	X	○	○	○	○	○

本表格依据SJ/T11364的规定编制。

O: 表示该有害物质在该部件所有均质材料中的含量均在GB/T26572规定的限量要求以下。

X: 表示该有害物质至少在该部件某一均质材料中的含量超出GB/T26572规定的限量要求。

注：本产品大部分的部件采用无害的环保材料制造，含有有害物质的部件皆因全球技术发展水平 的限制而无法实现有害物质的替代。

***:表示如果包含部分**