MXA710

2-Foot and 4-Foot Linear Array Microphone

Shure MXA710 linear array microphone manual. Learn how to install 2-foot and 4-foot versions and how to use Shure's trusted IntelliMix DSP platform

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MXA710
2-Foot and 4-Foot Linear Array Microphone

General Description

The Shure Microflex® Advance™ MXA710 Linear Array Microphone represents the next evolution in Shure array microphone technology, designed for high-quality audio capture in premium AV conferencing environments. The linear form factor of the MXA710 allows for placement virtually anywhere in a meeting space, including on a wall, around a display, on a ceiling, or in a conference room table. Available in 2- and 4-foot lengths in 3 colors, the MXA710 includes proprietary IntelliMix® DSP and Autofocus™ technology that provides all the processing needed for echo- and noise-free audio.

Features

- Steerable Coverage™ technology to capture audio anywhere in the room (up to 4 lobes with a 2-foot array, 8 lobes with a 4-foot array)
- Autofocus technology fine-tunes each lobe position in real time, even if meeting participants lean back or stand up.
- Default room coverage template enables quick and easy lobe optimization for wall, ceiling, or table installations.
- IntelliMix DSP includes automatic mixing, acoustic echo cancellation, noise reduction, and automatic gain control.
- Shure Designer System Configuration software for easy setup and configuration
- PoE powered
- LED status bars with configurable colors and brightness
- Dante & AES67 audio networking protocols
- Shure network audio encryption compatible
- Multiple mounting accessories available for wall, ceiling, or table installation

MXA710 Designer Setup Example

This example uses Designer 6.0 to set up an MXA710 and an ANIUSB-MATRIX. You can use other combinations of devices with the MXA710, but the setup process in Designer is similar.

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

- Discover the MXA710 in Designer
- Design microphone coverage
- Apply DSP and route signals

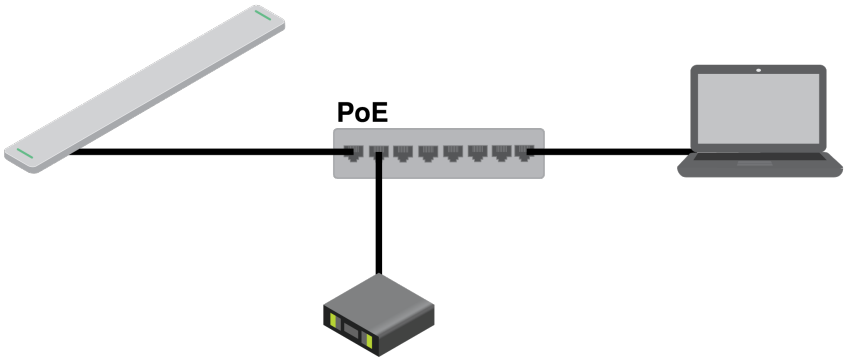
This example uses:

- Cat5e or better Ethernet cable (shielded cable recommended)
- Network switch that provides Power over Ethernet (PoE)
- Shure Designer software (download at shure.com/designer)
- ANIUSB-MATRIX

Step 1: Install and Connect

1. Install the MXA710 and other devices. Connect devices to PoE ports on the network switch using Ethernet cable.

2. Connect the 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 both devices into the design. Designer prompts you to choose an installation method for the MXA710. You can change this setting later in `[Your room]` > `Coverage` > `MXA710` > `Properties` > `Device control` > `Change installation`.

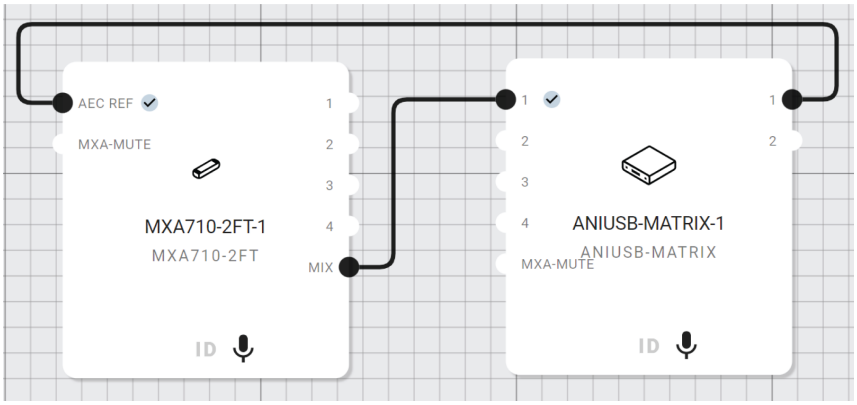


Step 2: Route Audio and Apply DSP

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

The MXA710 includes IntelliMix DSP that can be applied to the automix channel output.

1. Select Auto route. Designer makes the following audio routes:
   - MXA710 output to ANIUSB-MATRIX input
   - ANIUSB-MATRIX output to MXA710 AEC reference input

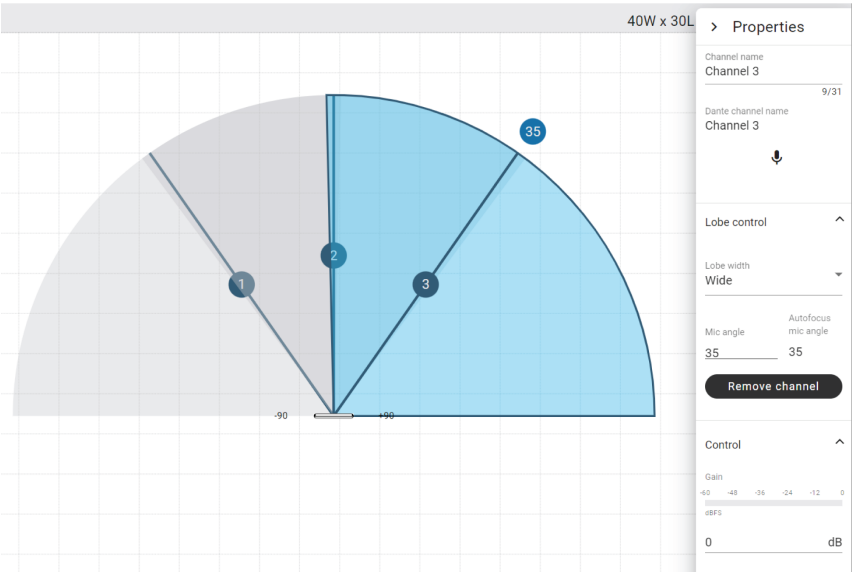


2. Check the audio routes and settings to make sure they fit your needs. You might need to:
   - Delete unnecessary routes.
   - Verify that AEC reference signals are correctly routed.
   - Fine-tune DSP blocks as needed.
   - Check matrix mixer routes.

   You can also route audio manually in Designer or use Dante Controller to route to third-party devices.
3. Send audio from the ANIUSB-MATRIX 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.

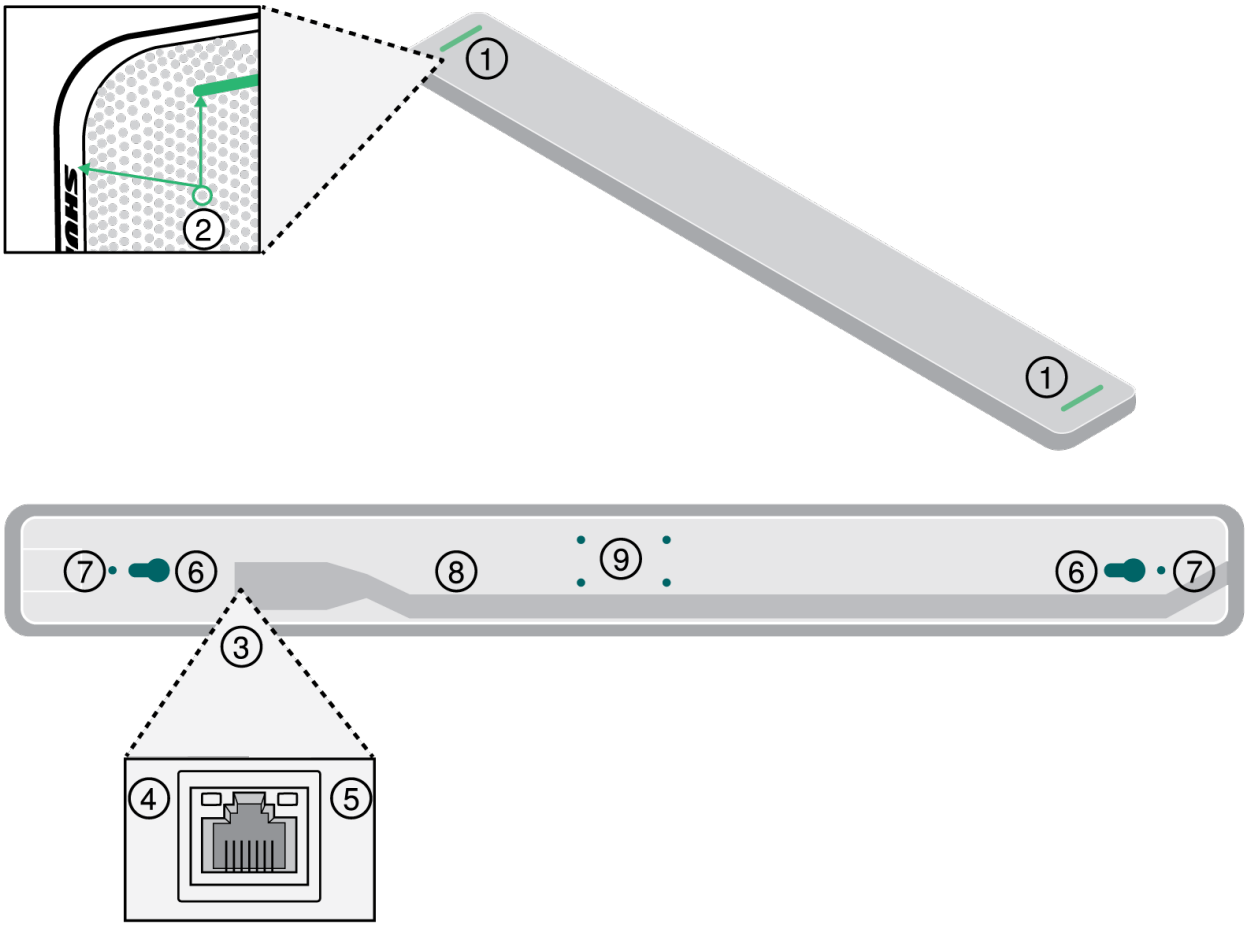
Step 3: Adjust Microphone Coverage

1. Go to `[Your room]` > `Coverage` to adjust the MXA710's coverage.
2. Set up a way to listen to the microphone directly (with a Dante headphone amp, for example). Listen to each of your microphone's channels and adjust the lobe position, width, and gain as needed. The solid line in each lobe shows where pickup is the strongest. The lobe's edge is −6 dB down from the solid line.



3. Place a test call with the whole conferencing system. Adjust gain and DSP as needed to get a good room sound.

MXA710 Parts



1. Mute status LED

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

Mute Status LED 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

   Sits behind the microphone grille. To access, find a grille hole that aligns with the left edge of the mute status LED and the "S" of the Shure logo. Use a small paperclip or other tool to press and hold the button. You may need to try a few different holes to press the reset button.

3. Network port

   RJ-45 jack for network connection. Power over Ethernet is required to power the microphone.

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. Mounting keyholes

   Use to attach the microphone to the wall-mounting bracket.

7. Screw holes for suspension mounting

   Use to attach eyelet screws to hold braided metal cable or other high-strength wire for suspension mounting.

8. Cable exit

   Route the Ethernet cable here to keep it flush with the microphone.

9. Screw holes (VESA MIS-B compatibility)

   Use to attach the desk stand, the microphone stand mount, or other VESA MIS-B-compatible adapters.

Model Variations

SKU	Description
MXA710B-2FT	Black 2-foot microphone (60 cm)
MXA710W-2FT	White 2-foot microphone (60 cm)
MXA710AL-2FT	Aluminum 2-foot microphone (60 cm)
MXA710B-4FT	Black 4-foot microphone (120 cm)
MXA710W-4FT	White 4-foot microphone (120 cm)
MXA710AL-4FT	Aluminum 4-foot microphone (120 cm)

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

Cable Requirements

Always use Cat5E cable or higher.

Control Devices with Shure Designer Software

To control this device's settings, use Shure Designer software. 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.

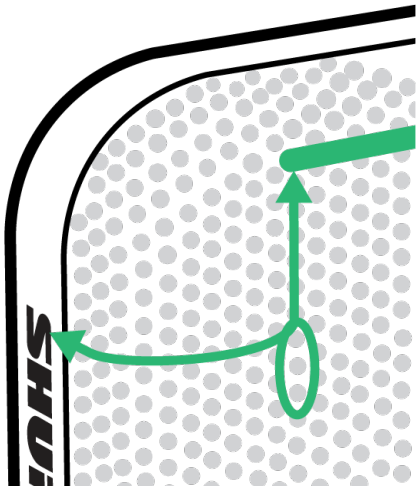
What's in the Box

2-foot or 4-foot linear array microphone	MXA710-2FT or MXA710-4FT
Wall-mounting bracket (2 or 4-foot)	RPM710-2M or RPM710-4M
Hardware kit with: **Wall cover plate and screws (US and UK versions)** **Mounting eyelet screws (2)** **Washers for eyelet screws (2)** **Cable ties (2)**	RPM710-H

Reset Button

The reset button is behind the grille and can be pushed with a small paperclip or other tool. To access the button:

1. Find the end of the microphone that has the Shure logo printed on the side.
2. Find the left edge of the microphone light, which sits behind the grille.
3. Insert the paperclip into the grille hole that aligns with the left edge of the microphone light and the "S" of the Shure logo. Press and hold to reset the microphone. If you don't feel a button, try the grille holes below and around the first one. You may need to try a few different holes to press the reset button.



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. Multicolor flash, then blue LED along bar.

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.

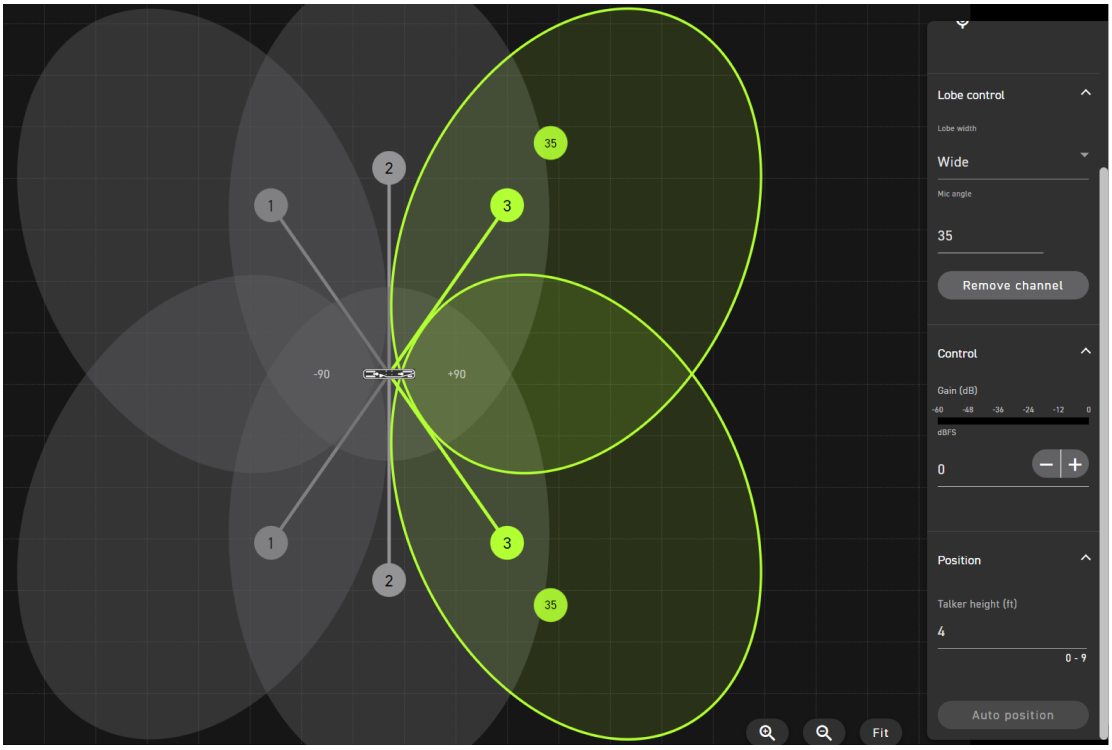
How to Adjust Microphone Coverage

To control microphone coverage, use Designer. Microphone coverage is at the room level, meaning that there is one coverage map for all Shure devices in a room.

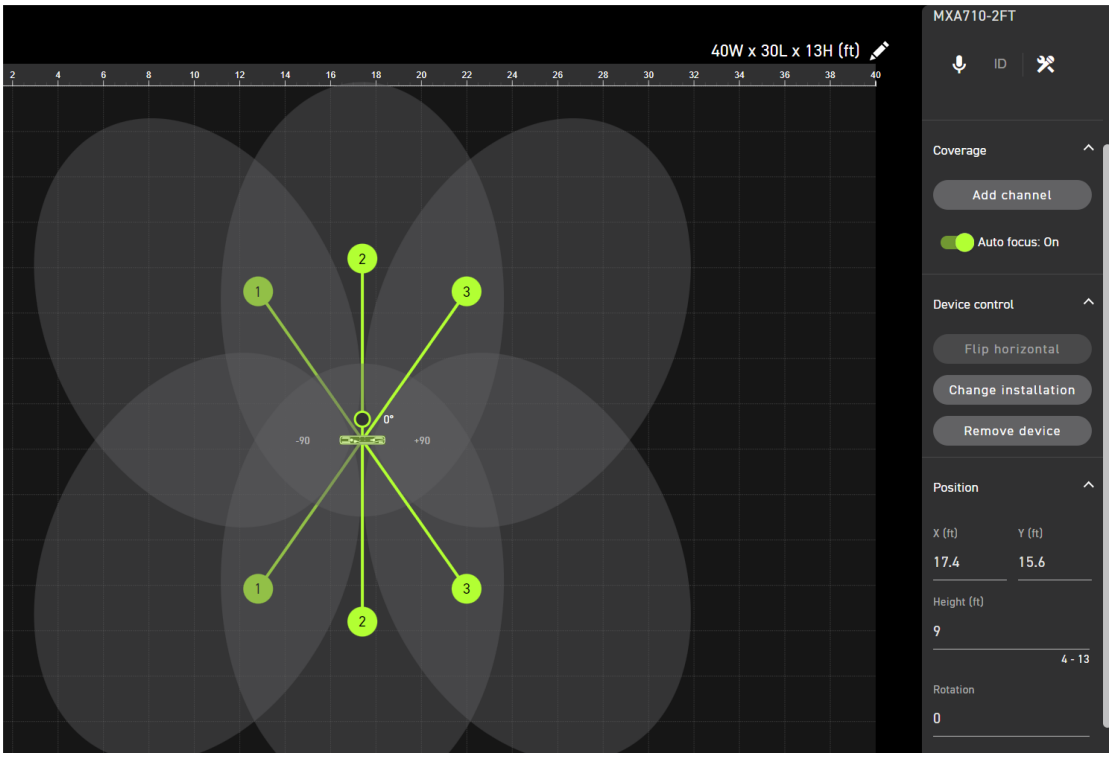
1. Add the MXA710 to a design or an online room. Designer prompts you to choose an installation orientation. There are 4 options:
   - Wall horizontal
   - Wall vertical
   - Ceiling
   - Table

   The installation method determines which coverage template is used. The templates are designed and tested to fit most common installations.

2. Go to `[Your room]` > `Coverage`. Adjust the microphone's position and orientation to match your room's layout.
3. **Ceiling installations:** Set a talker height for each lobe and a device height for the most accurate representation of microphone coverage:
   - **Talker height:** Select each lobe and go to `Properties` > `Position` > `Talker height`.



   - **Device height:** Select the MXA710 and go to `Properties` > `Position` > `Height`. The device height can be different than the room height for suspension mounting installations.



4. Click and drag each lobe to cover the talkers in the room. Adjust each lobe's width and position in the Properties panel.
5. Make a test call with your system. Have someone talk from each position and listen to each of your microphone's channels. Adjust the lobe position, width, and gain as needed in the Properties panel.

   For non-ceiling installations, the solid line represents where the coverage is the strongest. The edge of the coverage area for each lobe represents where the lobe's sensitivity reaches −6 dB.

   Autofocus technology fine-tunes each lobe position in real time, even if meeting participants lean back or stand up.

Tips for Great Coverage

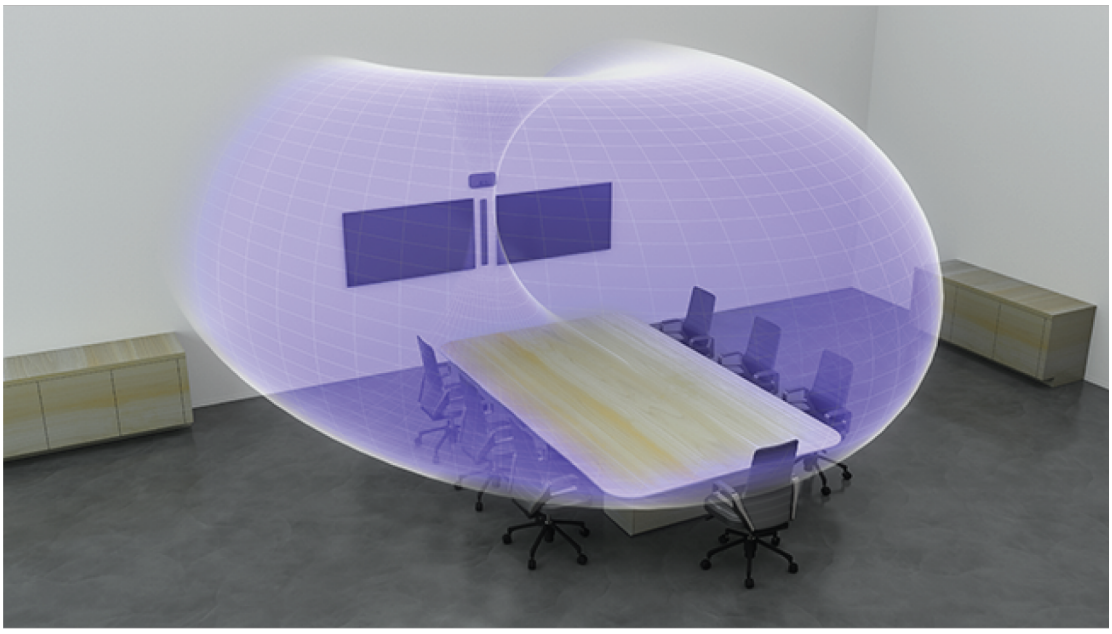
- To change the installation type, go to `[Your room]` > `Coverage` > `MXA710` > `Properties` > `Device control` > `Change installation`.
- Lobes can cover 1 talker or many talkers depending on width. Test and listen to your settings, and adjust as needed.
- **Ceiling installations:** For best results, avoid using narrow lobes.
- Lobes are bidirectional in some positions because of the microphone's pickup pattern.

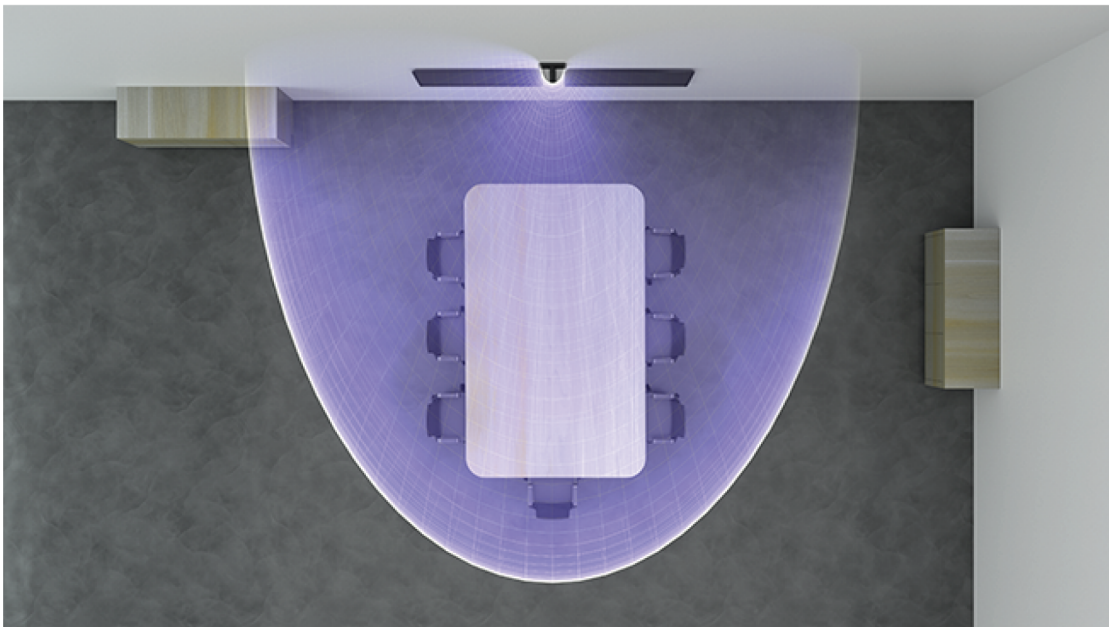
MXA710 Coverage Examples

Use these images to understand how the coverage patterns work in different installations. Always listen to lobes as you move them into position. Have someone talk from each lobe position to make sure that you have good coverage.

Wall Vertical (2-foot microphone)

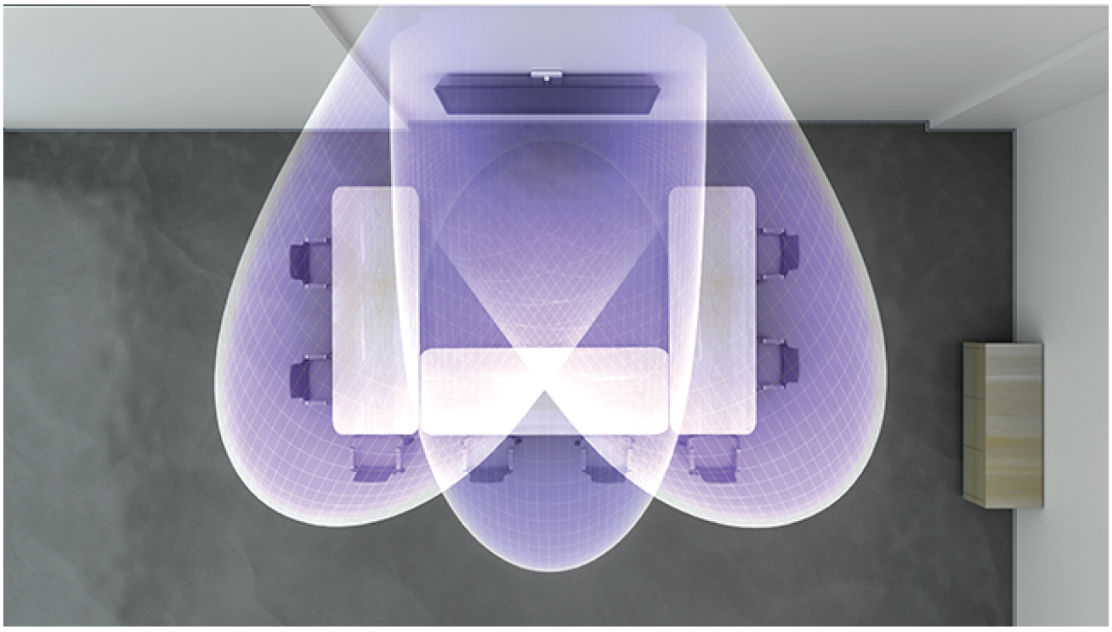
1 lobe. Go to Open side view to adjust the vertical angle.





Wall Horizontal (4-foot microphone)

3 lobes



Ceiling (2-foot microphone installed flush with ceiling)

3 lobes. Some are bidirectional in certain positions.

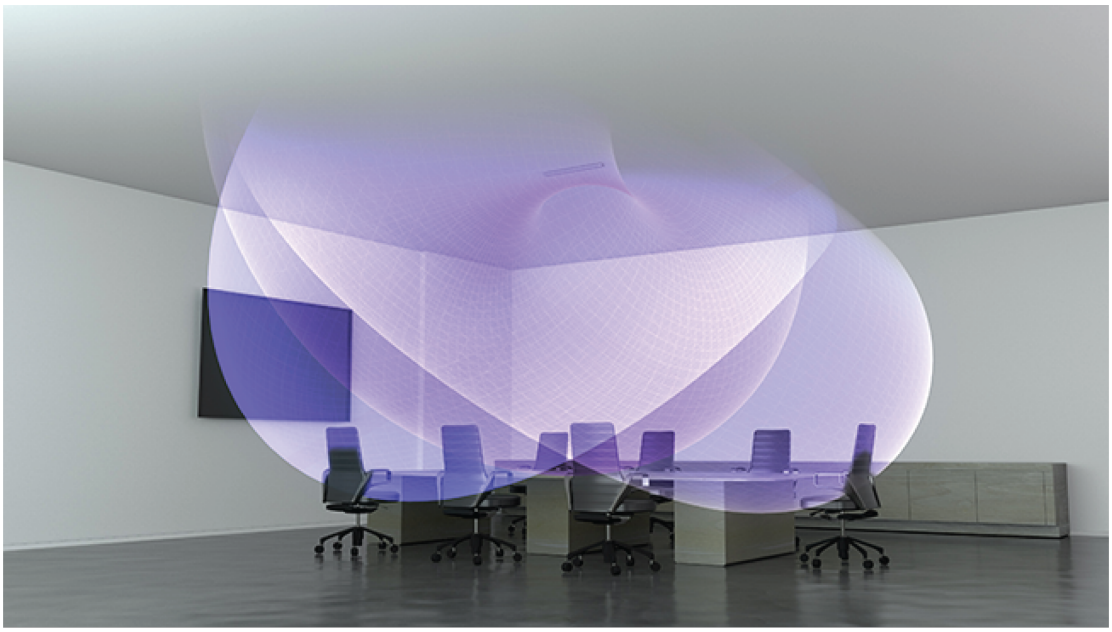
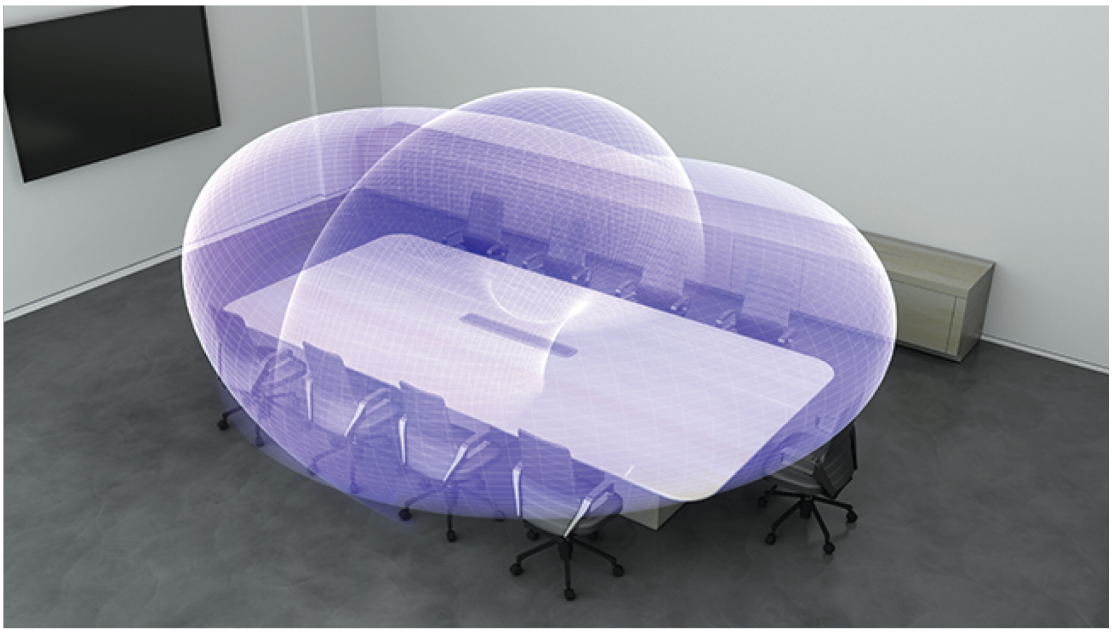


Table (2-foot microphone)

3 lobes. Some are bidirectional in certain positions.



Using Autofocus to Improve Coverage

This microphone uses built-in Autofocus technology to fine-tune microphone coverage in real time, even if meeting participants lean back or stand up. On microphones that show steerable lobes for coverage (such as the MXA710 or MXA920), you'll see the lobes moving in the coverage map as participants change positions. Autofocus only responds to in-room sound sources.

To turn Autofocus on or off, go to: `Coverage` > `Select device` > `Properties` > `Coverage`.

Important: Always route a reference signal to the microphone's AEC reference in channel to take full advantage of Autofocus. You should set up this reference signal even if you're using a separate DSP or direct outputs (on applicable microphones).

Adjust Levels

Before adjusting levels, set up a way to listen to the microphone's signal. You can make a test call with your system or use a Dante headphone amp.

Adjust the level for each microphone channel so that the gain setting works for all seating scenarios. Shure recommends setting the channel levels before making changes to automixer settings.

To adjust:

1. Perform a level check for each channel 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 control problematic frequencies in a room (such as a low-frequency rumble caused by HVAC systems).
3. If equalizer settings cause a significant increase or decrease in levels, make any necessary level adjustments according to step 1.

When to Use the Channel and IntelliMix Gain Faders

There are 2 different gain faders that serve different purposes:

Channel Gain (Pre-Gate)

To adjust, go to: `[Your room]` > `Device configuration` > `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, go to: `[Your room]` > `Device configuration` > `IntelliMix`

These faders adjust a channel's gain after the lobe has gated on. Adjusting the gain here does not affect the automixer's gating decision. Only use these faders to adjust the gain of a lobe after you are satisfied with the automixer's gating behavior.

How to Install the MXA710

The MXA710 is an extremely versatile microphone. You can install it in many places in a conference room and easily get good coverage for all talkers.

MXA710 Microphone Coverage

	MXA710-2FT	MXA710-4FT
Room size	Medium to large	Medium to large
Maximum number of lobes	4	8
Recommended distance from talkers	2 to 16 feet	2 to 16 feet

Best Practices for Installation

- Before installing, open the microphone's coverage map in Designer. Look at the 4 device installation templates to understand how the lobes behave when you move them around and use different widths. Lobes also have Autofocus technology, which fine-tunes each lobe position in real time, even if meeting participants lean back or stand up. Templates are available for:
  - Wall horizontal
  - Wall vertical

  - Ceiling
  - Table
- Measure your space and make sure that all talkers will fit in the microphone's coverage area.
- Coverage also depends on your room's acoustics, construction, and materials. Take these into consideration when planning coverage.
- Don't place the microphone behind obstructions. Keep the microphone's grille at least 36 inches away from any occupancy sensors.
- Plan for any future coverage needs.

Ways to Install the MXA710

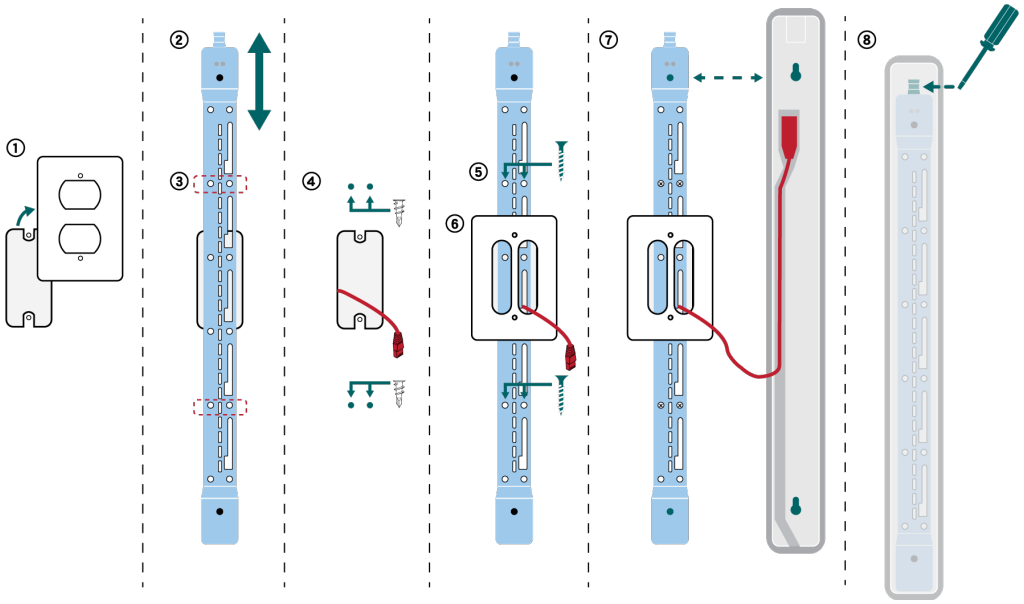
Accessory	Install location	Other hardware required?
Wall-mounting bracket	Wall	Drywall mounting screws and anchors
Display mount kit	Attach to display mount	Peerless Universal Sound Bar Kit, Chief Thinstall Center Channel Speaker Adapter, or other similar adapter with VESA MIS-B compatibility
Suspended Cable	Ceiling	Braided metal cable Hardware to attach cable to ceiling or A710-TB Tile Bridge
A710-TB Tile Bridge	Drop ceiling tile	No
A710-FM Flush Mount	Table, wall, or hard ceiling	No
A710-MSA Mic Stand Accessory	Mic stand	Mic stand
A710-DS Desk Stand	Credenza or other flat surface	No

Installing the Wall-Mounting Bracket

You can mount the bracket directly over a junction box, or at any other cable exit on the wall. The bracket works positioned vertically or horizontally.

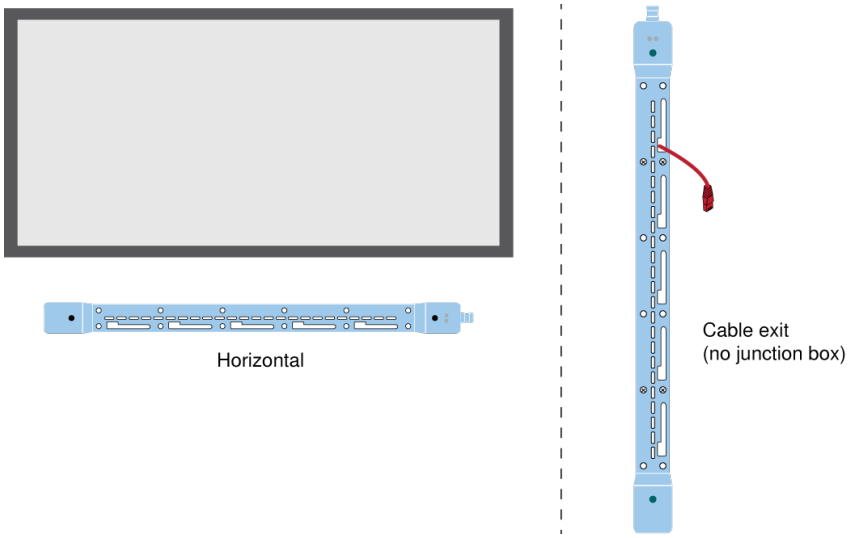
To get started, you will need:

- Wall-mounting bracket
- Cover plate (US or UK) and screws (if using)
- 4 drywall anchors and screws
- Screwdriver
- Drill
- Cat5e or better Ethernet cable



1. If you're mounting over a junction box, remove the existing cover plate.
2. Position the wall-mounting bracket. You can mount it vertically or horizontally on the wall.
3. Using a pencil, mark the wall for the position of the drywall anchors and screws. If you're installing over a junction box, balance the bracket with 2 screws above the box and 2 screws below it.
4. Remove the bracket and drill the holes for the drywall anchors. Install the drywall anchors.
5. Place the bracket on the wall and insert the drywall screws into the anchors to secure the bracket.
6. Thread the Ethernet cable through one of the large openings in the bracket. If you're mounting over a junction box, thread the cable through the provided cover plate and install the cover plate over the bracket.
7. Connect the Ethernet cable to the microphone. Align the holes on the back of the microphone with the raised posts on the bracket and slide the microphone into the bracket until it clicks into place.
8. To remove the microphone, press the tab at the top of the bracket with a screwdriver or other tool and slide the microphone up.

Other mounting options:



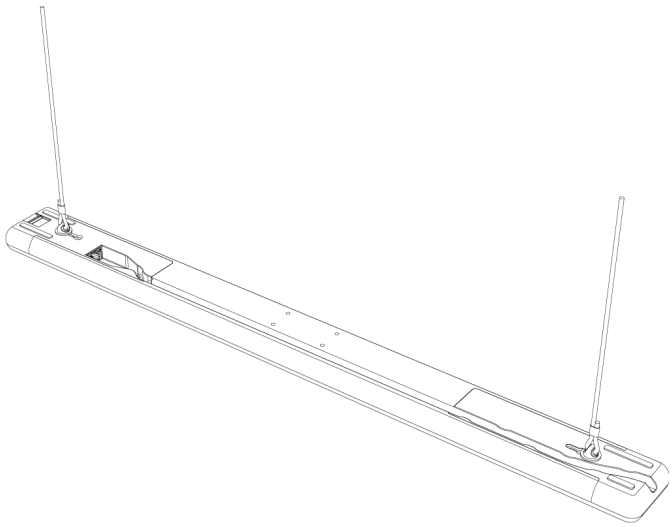
Shure also sells the A710-FM flush mount kit, which attaches to the microphone's mounting keyholes like the wall bracket in step 7 above.

Suspending the Microphone from the Ceiling

To get started, you will need:

- 2 mounting eyelet screws (4 mm diameter)
- 2 washers
- Braided metal cable or high-strength wire*
- Hardware to attach cable to ceiling*

1. Place the washers over the microphone's mounting holes and attach the eyelet screws to the microphone.
2. Attach the mounting cables to the eyelets.
3. Attach the cables to the ceiling using the appropriate hardware.



* Not included

Shure also sells the A710-TB tile bridge, which attaches to the microphone's screw holes like the suspension cable in step 1 above. Use the hardware included with the tile bridge to attach to the microphone.

VESA MIS-B Compatibility

The 4 screw holes (for M4 x 10 mm screws) on the bottom of the microphone are compatible with VESA MIS-B mounting products, such as the Peerless Universal Sound Bar Kit or the Chief Thinstall Center Channel Speaker Adapter.

Covering the Microphone with Fabric

In certain installations, covering the microphone or the mounting hardware with fabric may be desirable. Shure has tested the acoustic performance of this microphone with some fabrics from Guilford of Maine and Kvadrat.

In our tests, there was little effect on the microphone's acoustic performance if the fabric met one of the following specifications. **To cover the microphone, the fabric should meet at least one of these specifications:**

- Specific airflow resistance of ≤254 Pa*s/m (Pascal-second per meter)
- Any Guilford of Maine fabric with ≥0.95 NRC (noise reduction coefficient) rating

These are examples of fabrics that met our specifications at the time when Shure evaluated fabrics: Guilford of Maine's Bee-Have, and Kvadrat's Ginger, Mi Casa, Casita, and Time.

For best results:

- Use only 1 layer of fabric over the microphone or mounting hardware.
- Always confirm the fabric's acoustic specifications and testing process with the fabric manufacturer. Shure doesn't keep track of any changes in fabric specifications.

IntelliMix DSP

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

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

To access the DSP blocks, select the IntelliMix tab. When enabled, each DSP block will be colored.

Selecting Bypass IntelliMix will bypass the following DSP blocks: AEC, AGC, noise reduction, compressor, and delay.

DSP Best Practices

- Apply DSP blocks only 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.
- Unless you encounter video that lags behind audio, set Delay to off.
- DSP blocks do not affect whether the automixer gates a channel on or off.

Acoustic Echo Cancellation

In audio conferencing, a far-end talker may hear their voice echo as a result of a near-end microphone capturing audio from loudspeakers. Acoustic echo cancellation (AEC) is a DSP algorithm which identifies the far-end signal and stops it from being captured by the microphone to deliver clear, uninterrupted speech. During a conference call, the AEC works constantly to optimize processing while far-end audio is present.

When possible, optimize the acoustic environment using the following tips:

- Reduce loudspeaker volume
- Position loudspeakers farther from microphones
- Avoid pointing loudspeakers directly at microphone coverage areas
- If the microphone is providing AEC on the automix output, turn off AEC on the processor. Designer's auto route applies these settings automatically.

Select a Reference Signal for AEC

To apply AEC, provide a far end reference signal. For best results, use the signal that also feeds your local reinforcement system.

- **P300:** Go to Schematic and click any AEC block. Choose the reference source, and the reference source changes for all AEC blocks.
- **MXA710, MXA901, MXA910, MXA920:** Route a far-end signal to the AEC Reference In channel.
- **IntelliMix Room:** Go to Schematic and click an AEC block. Choose the reference source. Each block can use a different reference source, so set the reference for each AEC block. When stereo audio is on, you can set different left and right channel reference sources as needed.

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

Note: Always send an AEC reference signal to microphones with AEC processing, even if you are using a separate DSP for AEC. 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 dB level of signal reduction (the amount of echo being removed). If the reference source is connected properly, the ERLE meter activity generally 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. Use the lowest possible setting that is effective in your room. **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 (default)*:** For best results, use this setting. The non-linear processing level automatically adapts to changes in the acoustic environment.

* Auto setting available in firmware 6.2 and newer for MXA920, MXA902, MXA901, and MXA710

Noise Reduction

Noise reduction significantly reduces the amount of background noise in your signal caused by projectors, HVAC systems, or other environmental sources. It is a dynamic processor, which calculates the noise floor in the 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.

* Auto setting available in firmware 6.2 and newer for MXA920, MXA902, MXA901, and MXA710

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:

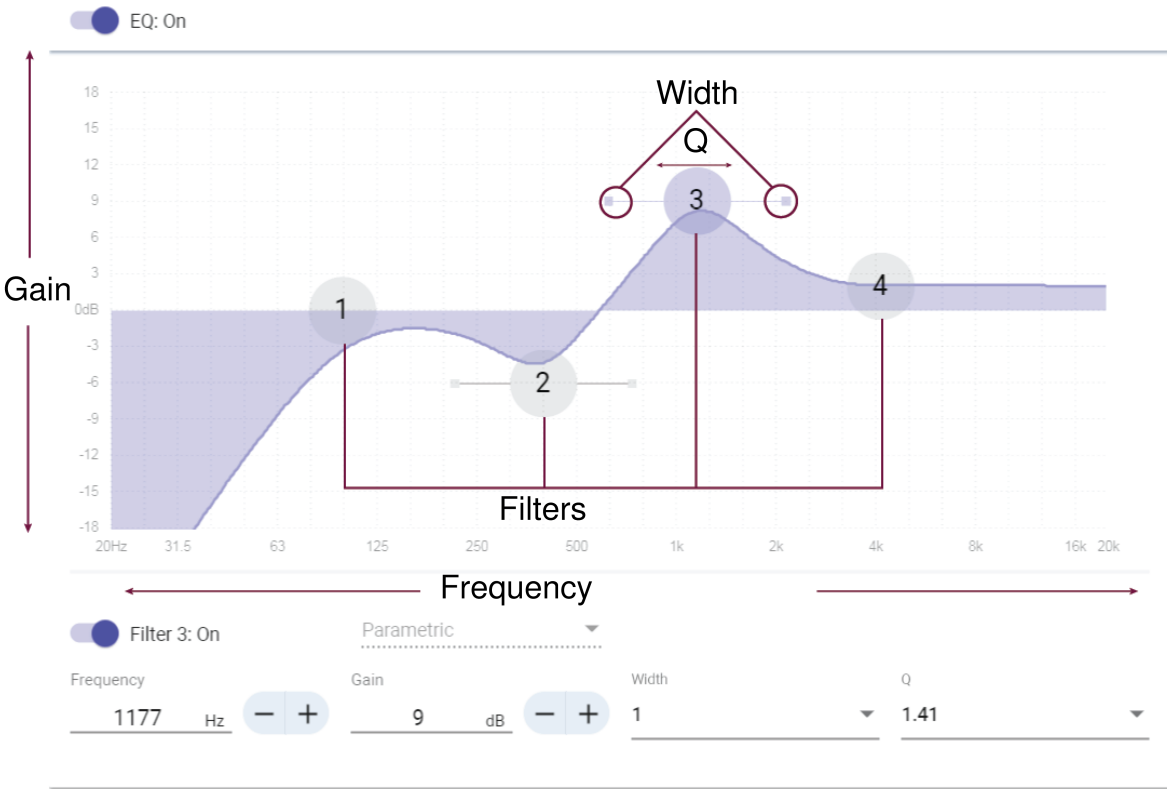
- Improve speech intelligibility
- Reduce noise from HVAC systems or video projectors
- Reduce room irregularities
- Adjust frequency response for reinforcement systems

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

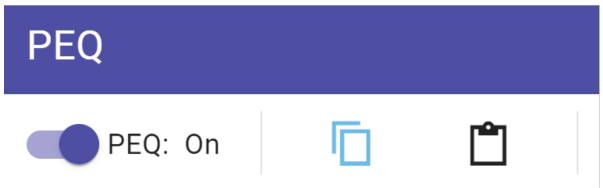
Setting	Function
Filter Type	Only the first and last band have selectable filter types. **Parametric:** Attenuates or boosts the signal within a customizable frequency range **Low Cut:** Rolls off the audio signal below the selected frequency **Low Shelf:** Attenuates or boosts the audio signal below 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)
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 built-in equalizer contour to quickly apply EQ changes to all channels. The EQ contour is separate from the per-channel EQ settings. Applying both the EQ contour and per-channel EQ has a cumulative effect, meaning that the EQ changes stack on top of each other.

To use, open the microphone in Designer and click EQ contour to turn it on or off.

- **MXA710 EQ contour:** Low shelf at 300 Hz, -6 dB

Click Bypass all EQ to quickly bypass any EQ contours or channel EQ settings.

Best Practices

- Listen to and test your system as you make EQ changes, and make sure they work for your specific room.
- When using with a P300 audio conferencing processor, turn off the microphone's channel EQ and EQ contours. Use the P300 to make EQ adjustments.

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.

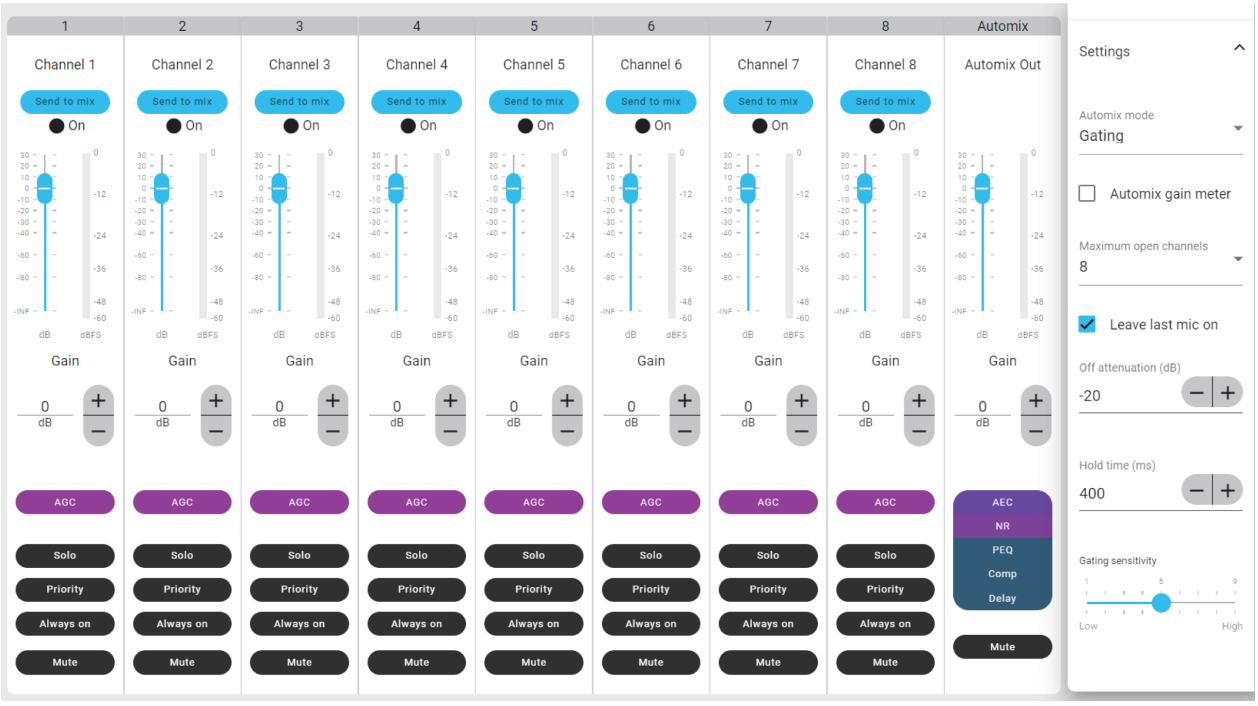
To use the automix channel, do the following:

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
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.

Setting	Function
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:**
  - Improves automixer gating
  - Focuses on speech sources over 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

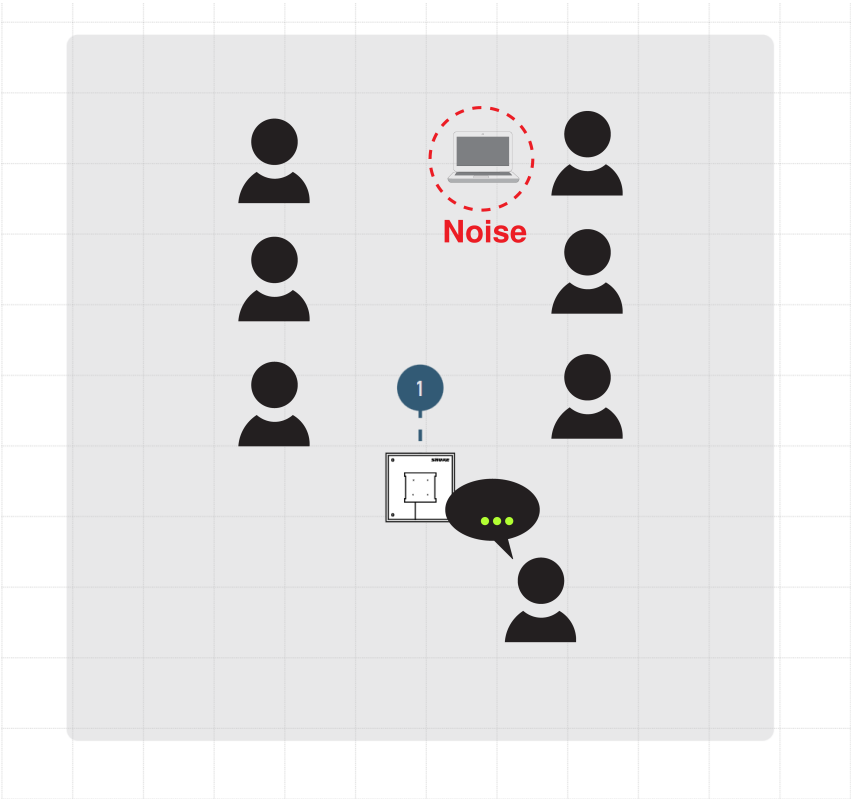
Examples of noise include:

- 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
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 the coverage area and focuses on the person speaking.



To use speech gating threshold and enhanced noise filtering:

1. In the microphone's control software, go to `IntelliMix` > `Properties`. Scroll down to turn on the 2 settings.
2. Make a test call. Have someone on the far end tell you how the signal sounds and adjust as needed.

Note: If the microphone is routed to IntelliMix Room and you're using the AI denoiser feature, turn off enhanced noise filtering.

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.

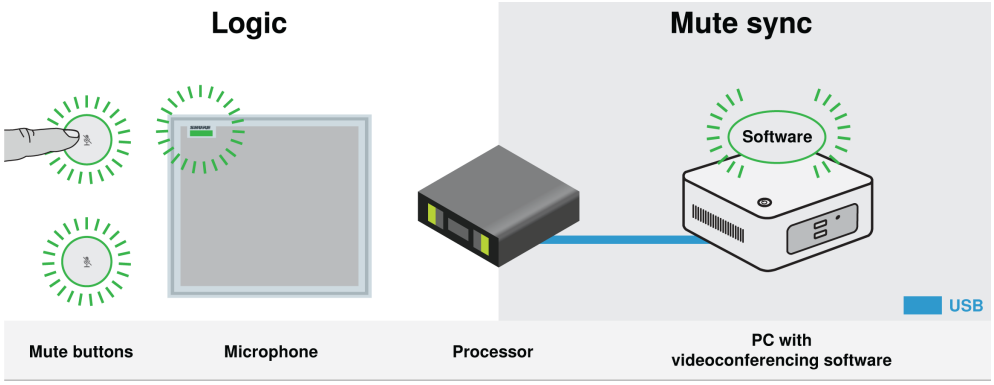
Manual

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.

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 (such as a P300, IntelliMix Room software, or an ANIUSB-MATRIX) 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 (mute button or microphone mute button) or the videoconferencing 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`
   - Microphones without physical mute button: Logic is always on
   - MXA310 and MXA mute button: `Settings` > `Logic control` > `Mute control function` > `Logic out`

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)
- MXA901
- MXA902
- MXA910
- MXA920

- MXA710
- MXA310
- Network Mute Button
- ANI22-BLOCK
- ANI4IN-BLOCK
- Logic-enabled MX microphones connected to ANI22-BLOCK or ANI4IN-BLOCK
  - MX392
  - MX395-LED
  - MX396
  - MX405/410/415

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

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.

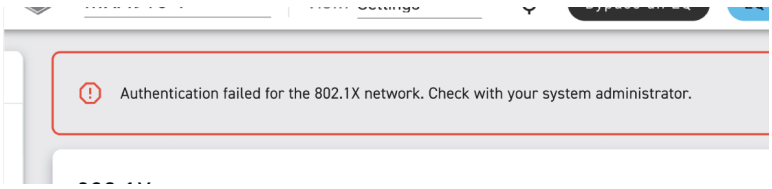
To change device settings:

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

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`.

Note: Refer to our FAQ if you're using 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 set-up, 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.

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/en-US/docs/commandstrings/MXA710.

Optional Accessories

- A710-FM-2FT Flush Mount Kit
- A710-FM-4FT Flush Mount Kit
- A710B-DS Desk Stand (black)
- A710AL-DS Desk Stand (aluminum)
- A710-TB Tile Bridge
- A710-MSA Mic Stand Adapter
- A710B-2FT-HOUSING (black)
- A710W-2FT-HOUSING (white)
- A710AL-2FT-HOUSING (aluminum)
- A710B-4FT-HOUSING (black)
- A710W-4FT-HOUSING (white)
- A710AL-4FT-HOUSING (aluminum)

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 the Shure Audio Institute
- Shure Systems YouTube channel

Download Shure Software

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

MXA710 Specifications

All specifications measured from narrow lobe width. Values for all widths are within ± 3 dB of these specifications unless otherwise noted.

General

Lobe Width

Adjustable	Narrow	30 degrees
	Medium	40 degrees
	Wide	70 degrees

Connector Type
RJ45

Power Requirements
Power over Ethernet (PoE), Class 0

Power Consumption
10 W maximum

Weight

MXA710-2FT	2 lbs (0.91 kg)
MXA710-4FT	3.7 lbs (1.67 kg)

Product Dimensions

MXA710-2FT	0.87 x 2.36 x 25.04 in. (22.09 x 60 x 636 mm) H x W x L
MXA710-4FT	0.87 x 2.36 x 49.12 in. (22.09 x 60 x 1247.76 mm) H x W x L

Control Software
Shure Designer

Plenum Rating
UL2043 (Suitable for Air Handling Spaces)

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

MXA710-2FT	50 MEMS
MXA710-4FT	100 MEMS

Frequency Response
100 Hz to 20 kHz

AES67 or Dante Digital Output

Channel Count	MXA710-2FT	6 total channels (4 independent transmit channels, 1 Automix output, 1 AEC reference in channel)
	MXA710-4FT	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

MXA710-2FT	-7.4 dBFS/Pa
MXA710-4FT	-7.9 dBFS/Pa

Maximum SPL
Relative to 0 dBFS overload

MXA710-2FT	101.4 dB SPL
MXA710-4FT	101.9 dB SPL

Signal-To-Noise Ratio
Ref. 94 dB SPL at 1 kHz

71.2 dB A-weighted

Latency
Does not include Dante latency

Direct Outputs	8.7 ms
Automix output (Includes IntelliMix processing)	19.3 ms

Self Noise

MXA710-2FT	22.9 dB SPL-A
MXA710-4FT	22.8 dB SPL-A

Dynamic Range

MXA710-2FT	78.5 dB
MXA710-4FT	79.1 dB

Built-in 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

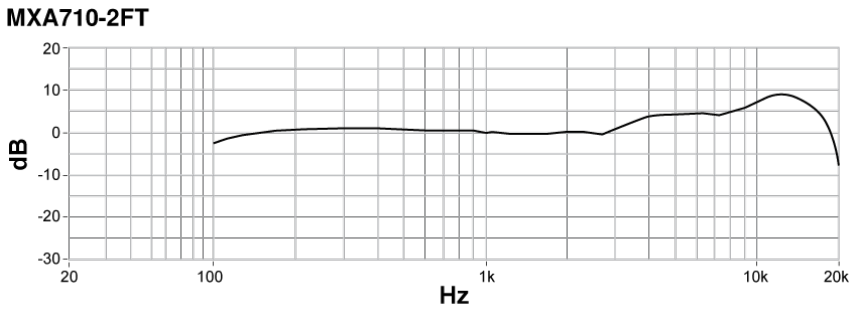
Networking

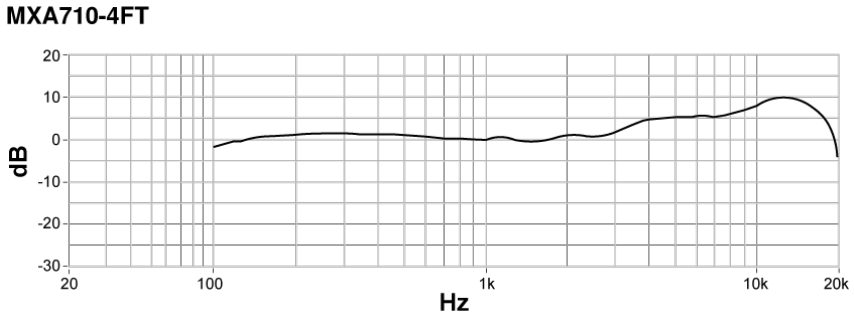
Cable Requirements

Cat 5e or higher (shielded cable recommended)

MXA710 Frequency Response

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



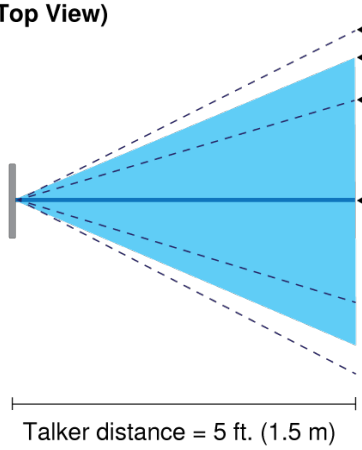


Lobe Sensitivity

The edge of the blue coverage area for each channel represents where the sensitivity reaches -6 dB. Understanding how lobe sensitivity is displayed helps to:

- Provide complete coverage in a space, either by adding lobes or changing the lobe width. This ensures the sensitivity is within 6 dB in all areas. It is acceptable for lobes to slightly overlap.
- Ensure that spacing and isolation are adequate to reduce noise and maximize automatic mixing performance.

Wall Horizontal Installation (Top View)



	Wide		Medium		Narrow	
dB	Inches	Centimeters	Inches	Centimeters	Inches	Centimeters
-12	82	210	32	81	22	55
-6	42	107	22	55	16	41
-3	28	71	16	41	11	27
0	0	0	0	0	0	0

Measured at 1 kHz, on-axis

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.

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.

Dante is a registered trademark of Audinate Pty Ltd.

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.

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

Authorized European representative:
Shure Europe GmbH
Global Compliance
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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