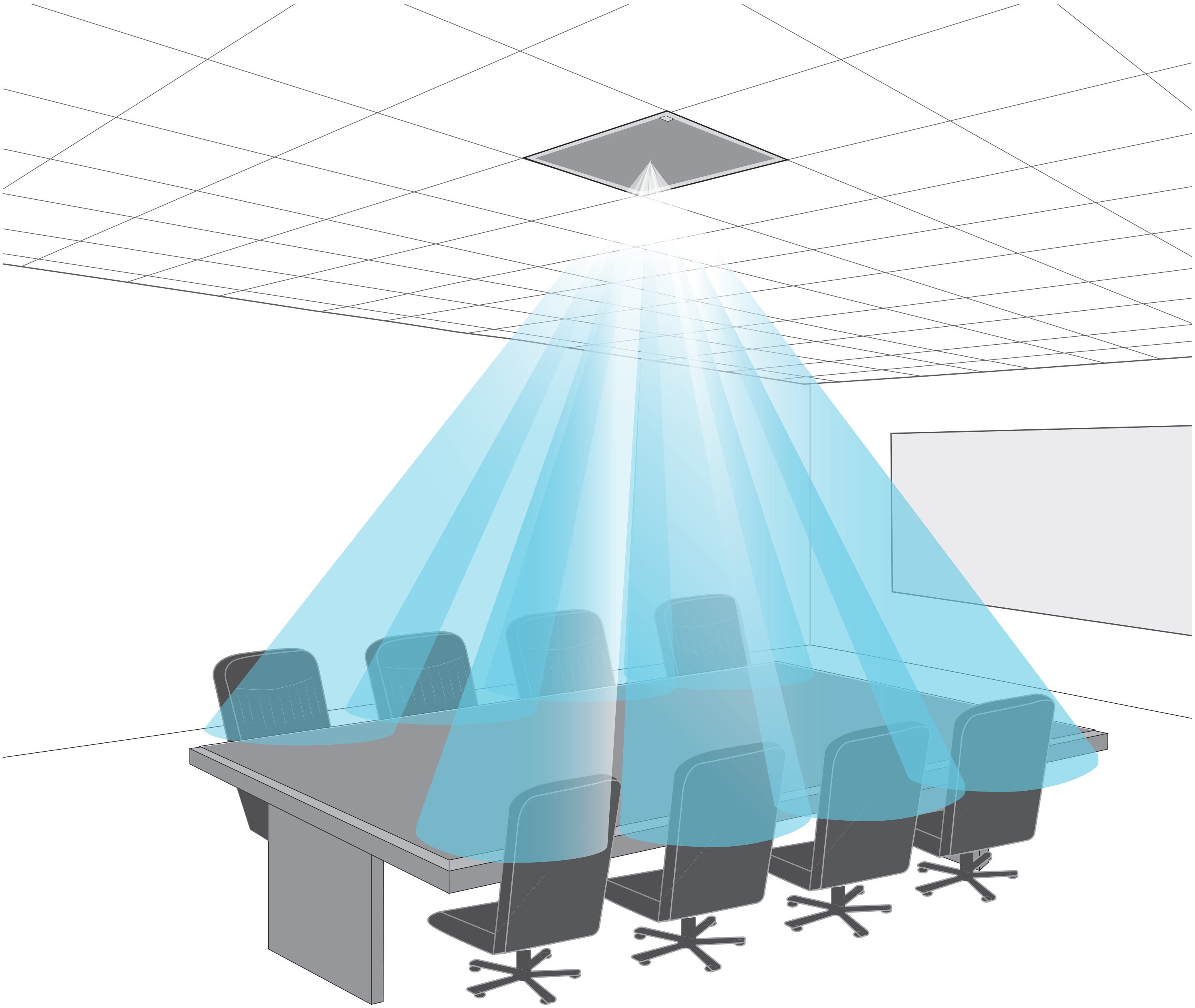
Overview

General Description

The Microflex® Advance™ Ceiling Array is a premium networked array microphone for AV conferencing environments, including boardrooms, huddle rooms, and multi-purpose spaces. Revolutionary technology from the IntelliMix® DSP suite includes Steerable Coverage™ , with eight highly directional pickup lobes that capture participant audio from overhead. Browser-based control software provides an intuitive user interface for microphone attributes, including lobe configuration, automatic mix settings, and preset templates. The microphone integrates seamlessly with Dante™ digital networked audio and third-party preset controllers, including Crestron and AMX, to deliver a high-quality AV conferencing experience that appeals equally to integrators, consultants, and meeting participants.



Features

Configurable Coverage

- Steerable Coverage™ delivers precise pickup for up to 8 independent lobes
- IntelliMix® DSP Suite provides fast-acting automatic mixing, echo reduction, and channel equalization

Software Control

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

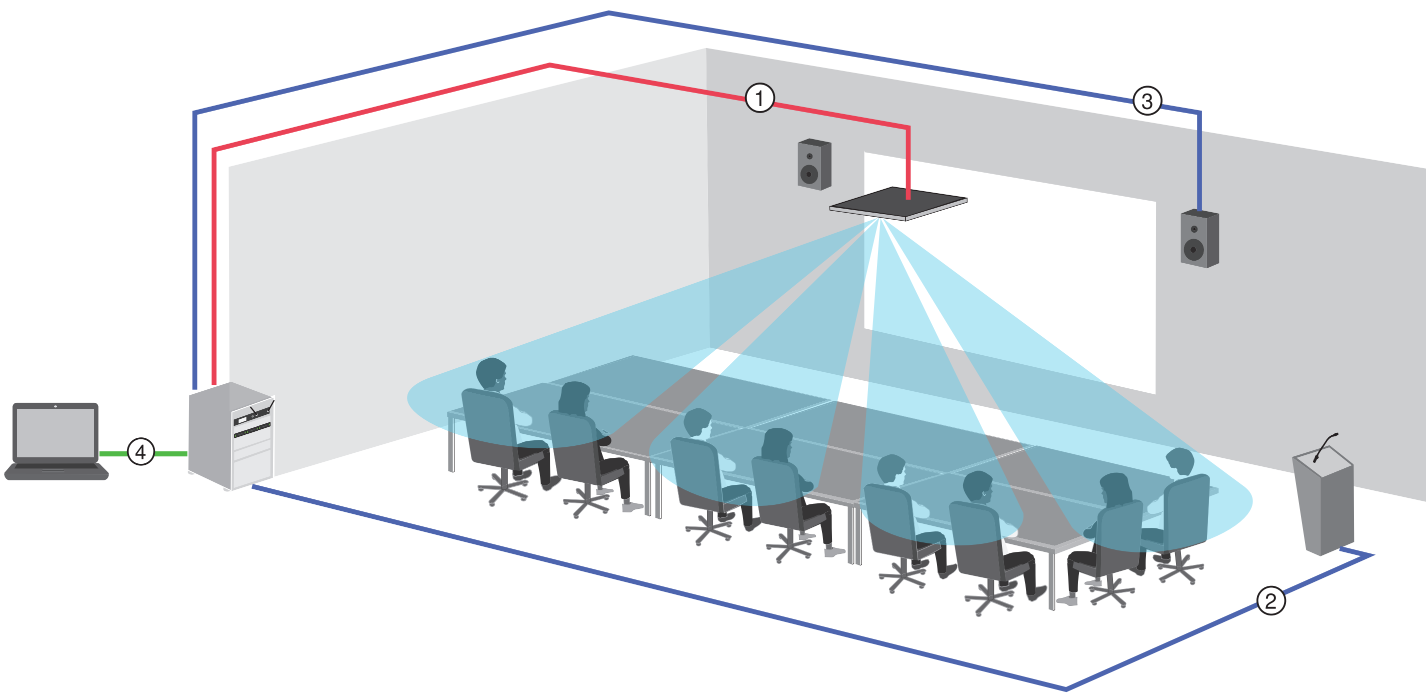
Network Connectivity

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

Professional Design

- Sleek industrial design blends with contemporary board rooms and meeting spaces
- Seamless flush-mount with standard ceiling tiles
- Available in white, black, and aluminum finishes (detachable grille can be custom painted)

System Overview



① Dante audio, power, and control

Each array microphone connects to the network over a single network cable, which carries Dante™ audio, Power over Ethernet (PoE), and control information to adjust coverage, audio levels, and processing.

② Analog audio (microphone to network)

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

③ Far-end audio (network to loudspeakers)

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

④ Device control and Dante™ audio

Control: A computer connected to the network controls the microphone through a web-based control application. Coverage, muting, LED behavior, lobe settings, gain, and network settings are controlled remotely.

Audio: Dante™ audio is routed through Dante™ controller software. Dante™ Virtual Soundcard enables audio monitoring and recording directly on the computer.

System Planning and Gear Requirements

Setting up the Audio Network

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

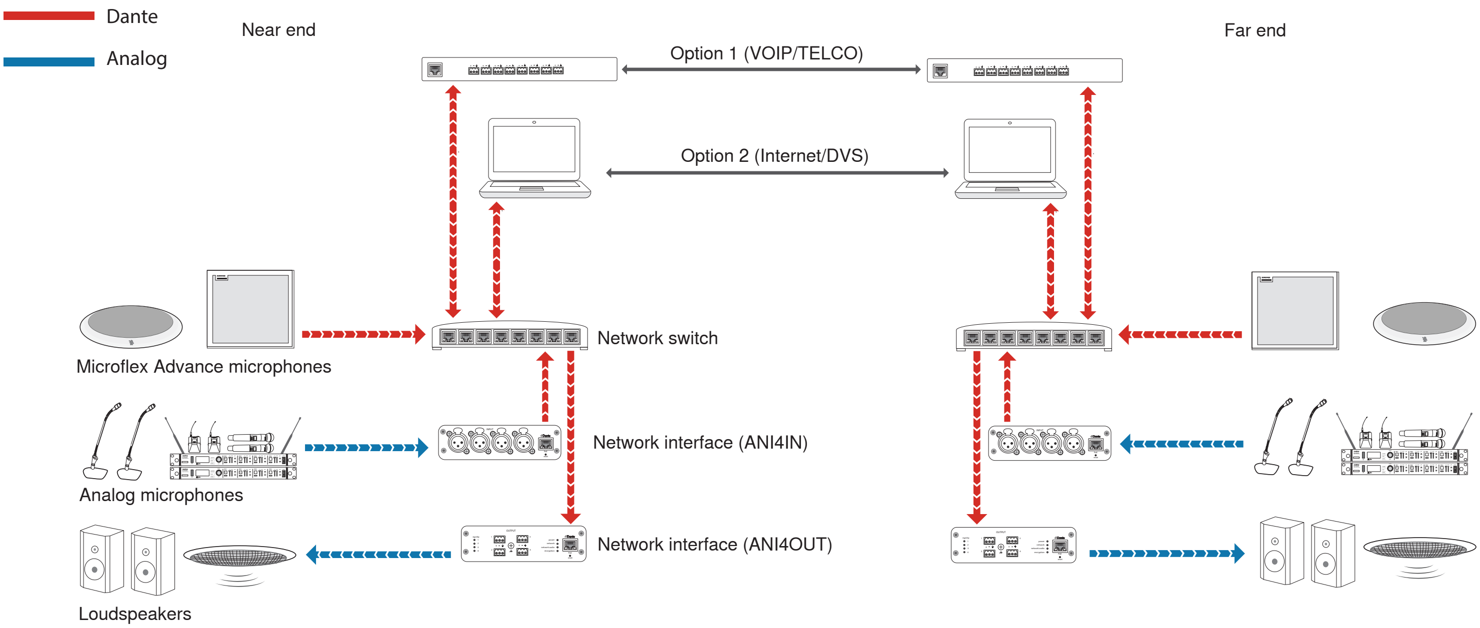
Shure components shown in this diagram:

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

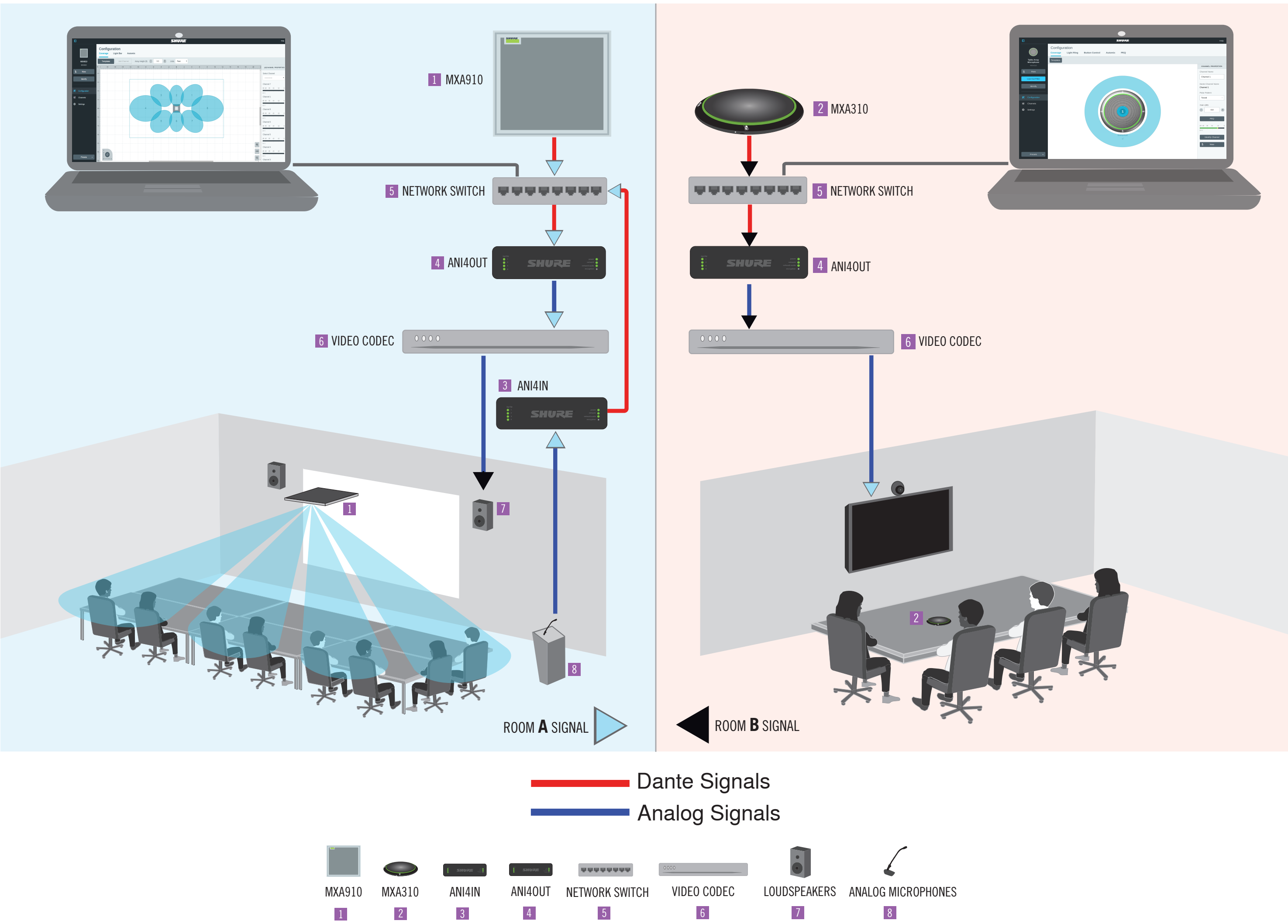
Audio Network Interfaces
The interfaces are used to connect analog devices such as loudspeakers and analog microphones to the network.

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

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



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



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

Controlling Hardware and Audio Over the Network

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

Shure Hardware and Audio

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

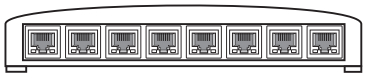
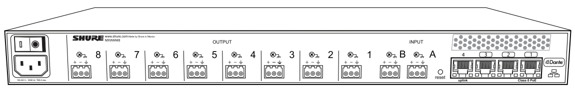
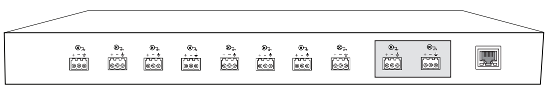
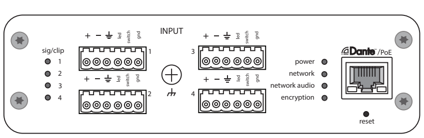
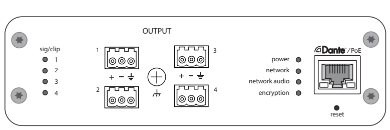
Expanded Control for Analog Devices

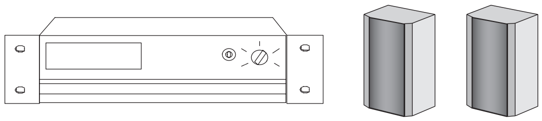
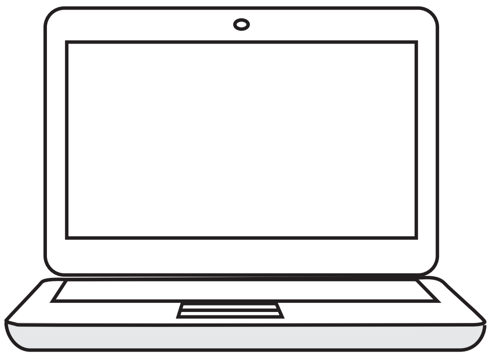
Analog devices that are connected to the network through a Shure network interface (ANI4IN/ANI4OUT) benefit from additional remote control: Volume levels, equalization, and signal routing are managed through the web application. For example, adjusting loudspeaker volume or muting a wired microphone,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

Dante™ Signal Routing

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

Hardware Component Index

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

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

System Scenarios

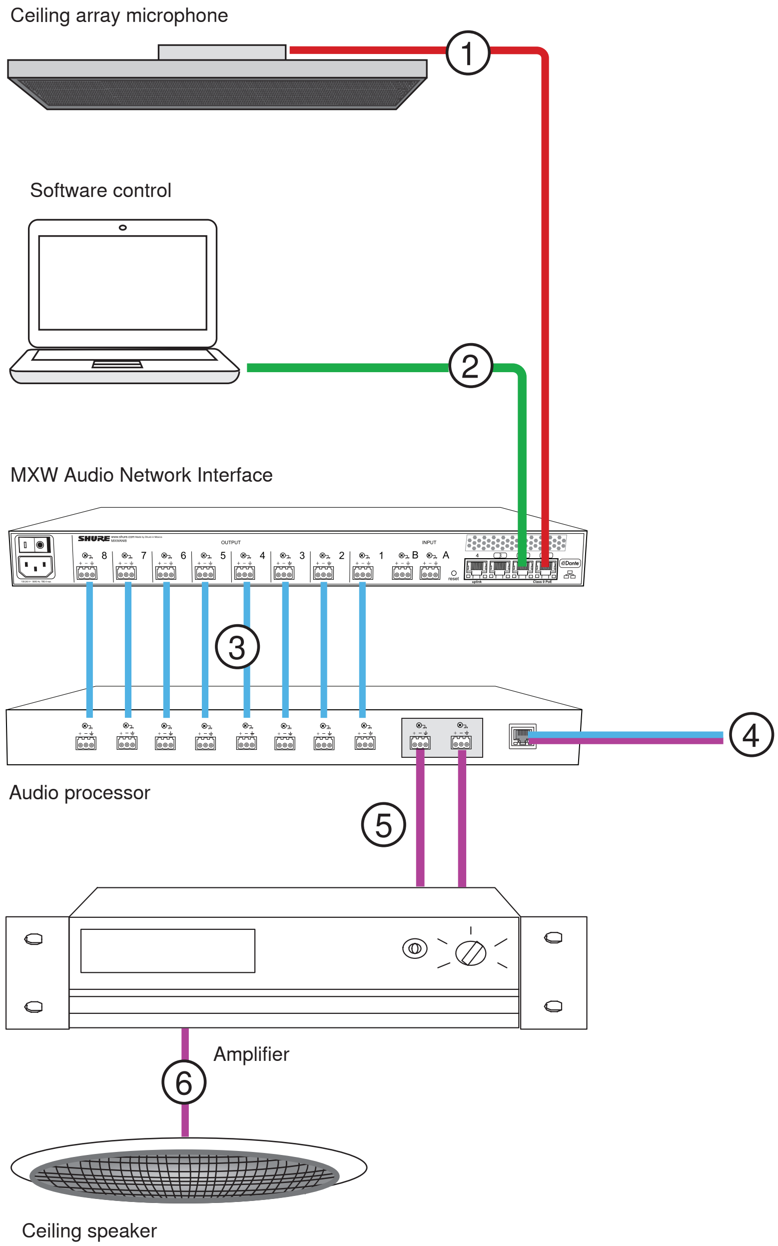
The following diagrams show a selection of common conferencing room systems. Use them as a reference when planning hardware and cable requirements for an installation. Each diagram includes:

- Signal flow and connections
- Required hardware
- Component roles

Power Over Ethernet and Hardware Requirements

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

Telephone Conference with Shure MXW Network Interface



① Array microphone to Shure MXWANI

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

② Computer to Shure MWXANI

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

③ Shure ANI analog outputs to audio processor

Step 1: Route signals with Dante™ Controller software

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

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

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

④ Connection to far end

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

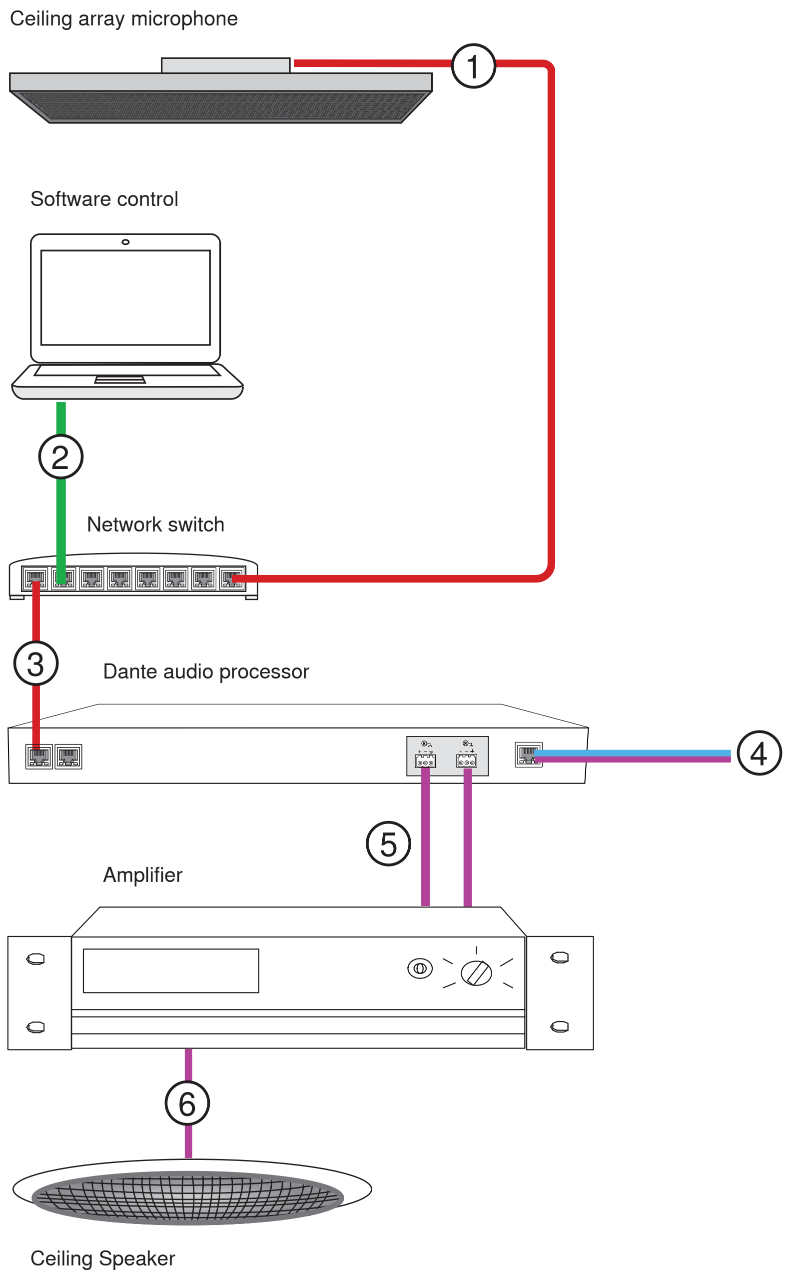
⑤ Audio from far end to amplifier

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

⑥ Amplified audio signal to loudspeakers

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

Telephone Conference with Dante™ -enabled Audio Processor



① Array microphone to network switch

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

② Computer to network switch

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

③ Network switch to Dante™ audio processor

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

- Digital signal processing (acoustic echo cancellation)
- Digital-to-analog conversion to deliver Dante™ audio over an analog (VOIP or telephone line) output.

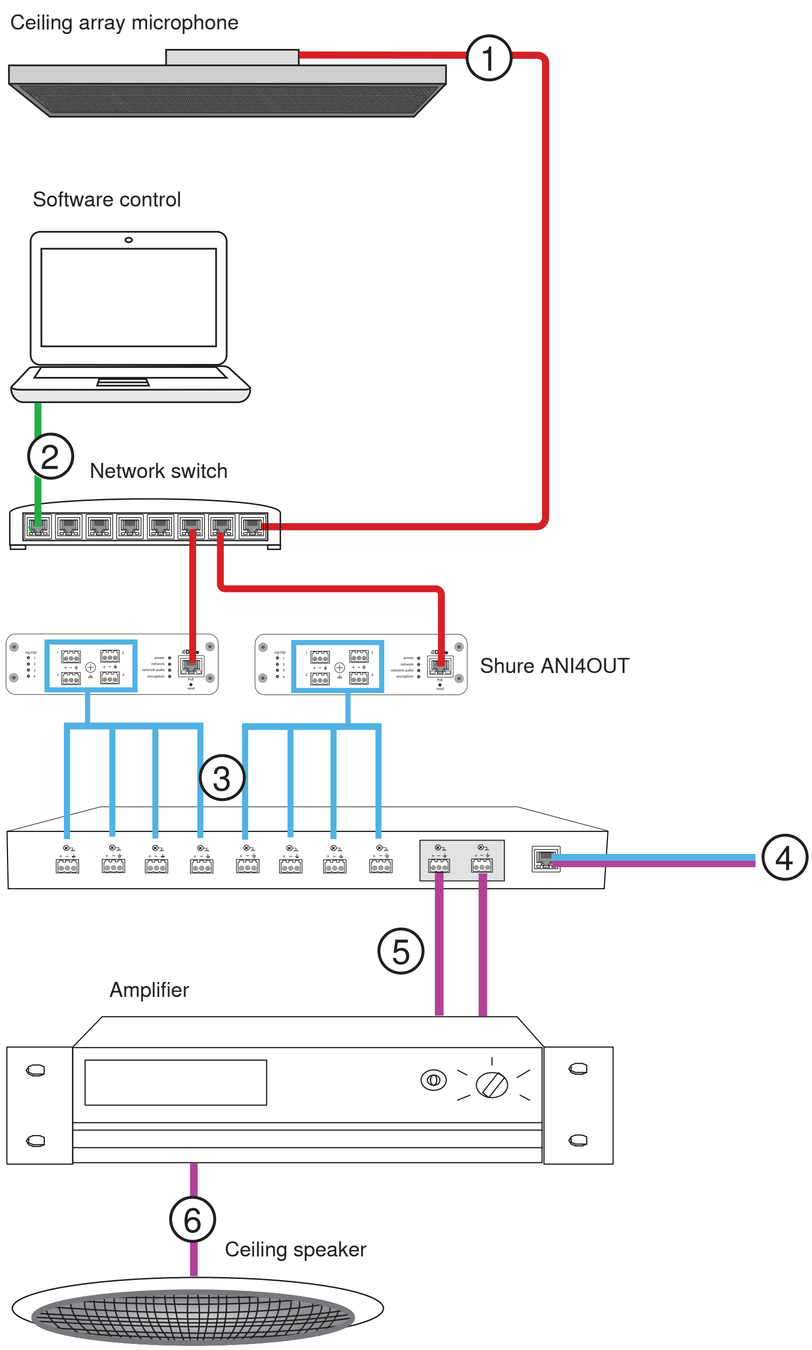
- Analog-to-digital conversion to deliver analog audio from the far end onto the Dante™ network.

④ Connection to far end
Connect the output from the audio processor to a VOIP server or telephone line to deliver audio between the near end and far end.

⑤ Audio from far end to amplifier
Route the far end audio through the audio processor output to an amplifier.

⑥ Amplified audio signal to loudspeakers
Connect the loudspeakers to the amplifier to deliver the audio from the far end.

Telephone Conference with Network Interfaces and Audio Processor



① Array microphone to network switch
Connect the array microphone output with a network cable to any port on the switch that supplies power over ethernet (PoE).

② Computer to network switch

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

③ ANI4OUT (digital-to-analog conversion)

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

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

④ Connection to far end

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

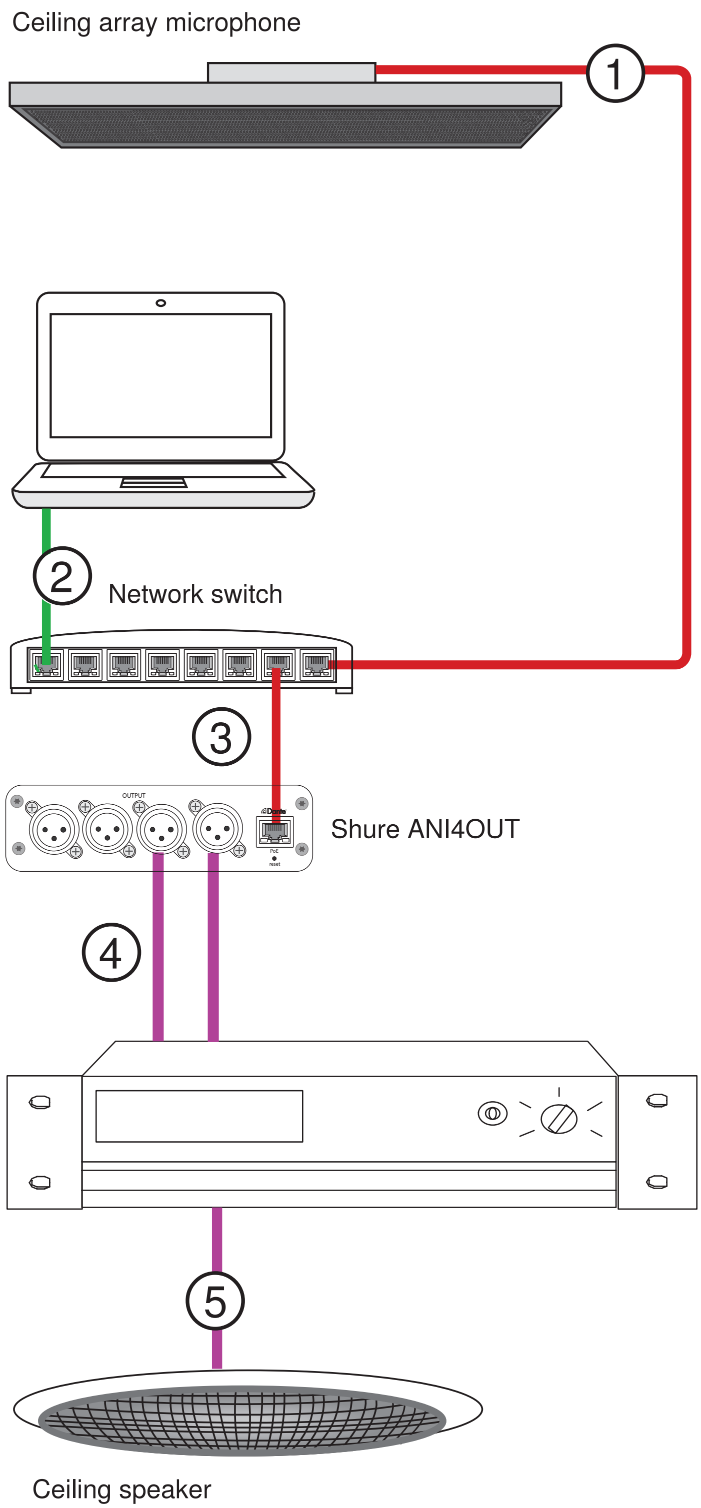
⑤ Audio from far end to amplifier

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

⑥ Amplified audio signal to loudspeakers

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

Web Conferencing Software With Dante™ Virtual Soundcard



① Array microphone to network switch

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

② Computer to network switch

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

- **Dante™ Virtual Soundcard / Controller:** Turn on the Dante™ Virtual Soundcard and use the controller software to route the array microphone signal to the computer.
- **Web Conferencing Software:**Assign the audio input and output device settings to the appropriate Dante transmitter and receiver channels.

③ Network switch to ANI4OUT

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

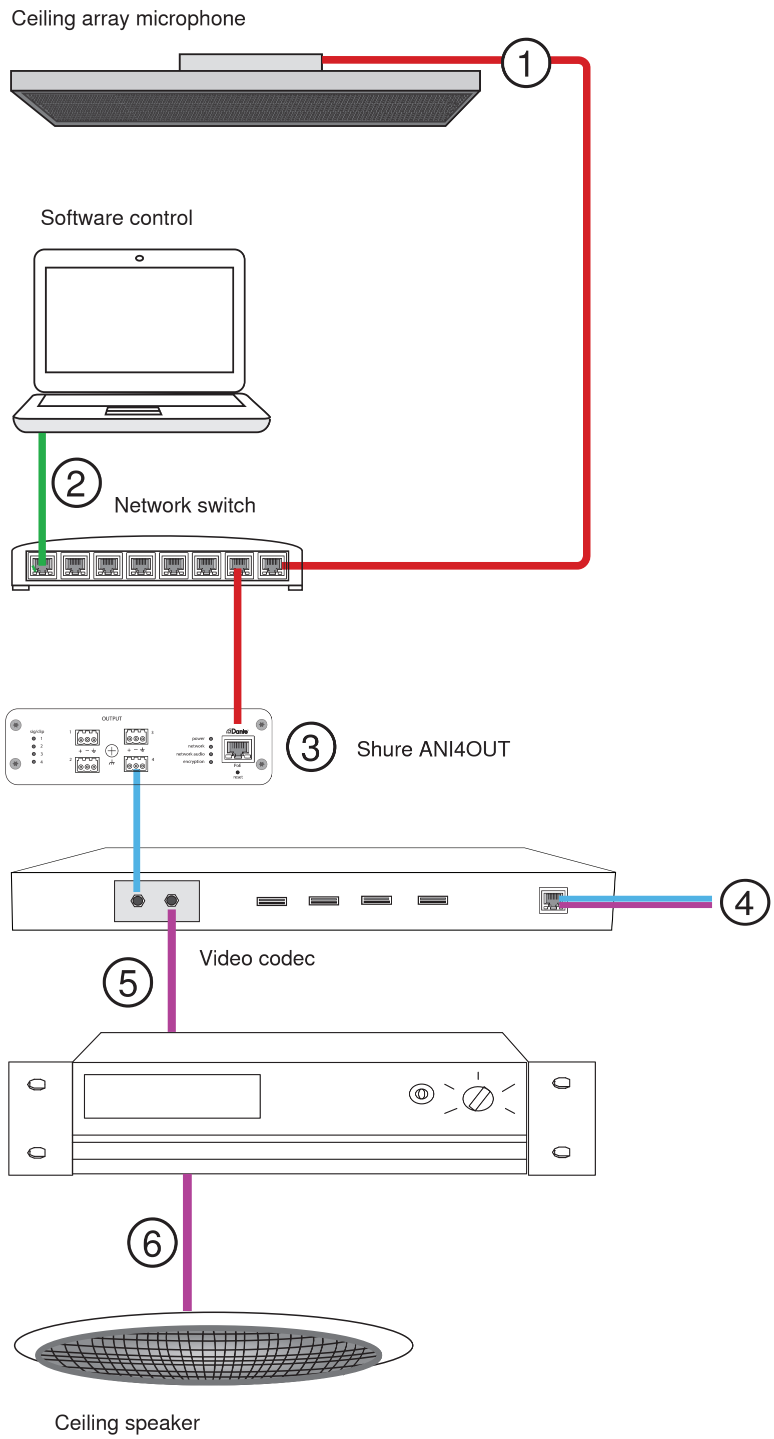
④ Audio from far end to amplifier

Route the far end audio to an amplifier.

⑤ Amplified audio signal to loudspeakers

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

Video Conference



① Array microphone to network switch

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

② Computer to network switch

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

③ ANI4OUT (digital-to-analog conversion)

Each ANI4OUT receives 4 channels of Dante audio, and converts them to 4 analog signals, delivered through XLR outputs or block connectors.
Input: Connect the ANI4OUT to the network switch with a network cable
Ouput: Connect the analog output to the audio input on the video codec

④ Video codec connection to far end

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

⑤ Audio from far end to amplifier

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

⑥ Amplified audio signal to loudspeakers

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

Installation

Microphone Placement

Room Variables

Optimal microphone placement is determined by the seating arrangements and infrastructure. Follow these guidelines for the best possible results:

- In rooms with flexible furniture arrangements or multiple array microphones, use the microphone configuration tool in the web application to ensure that the coverage is adequate for all seating scenarios.
- The lobes should be pointed towards the front of each talker. Carefully consider placement in rooms where talkers may face a screen during a video conference.
- Avoid installing the microphone directly next to unwanted sound sources, such as air vents or noisy video projectors.
- Consider installing acoustic treatment to improve speech intelligibility in rooms that are too reverberant.

Mounting Height

The maximum mounting height that can be set in the ceiling array microphone web application is 30 feet (9.14 meters). In a typical acoustic environment1, the microphone maintains an "A" rating based on the STIPA2 (Speech Transmission Index for Public Address systems) international standard at distances up to 16 feet between the microphone and talker. In better acoustic environments, the STIPA "A" rating may extend beyond 16 feet.

Consider the following when determining a mounting height:

- The pickup pattern of the ceiling array is narrower than a shotgun microphone, and therefore it can be placed farther from the source than any other microphone. While the web application shows an ideal coverage zone for each channel, keep in mind that there is no specific barrier at which the audio degrades or gates off. Lobe sensitivity data is available for each width setting in the product specifications.
- Like all microphones, tonality changes as the distance from the source increases.
- The intelligibility scale helps to predict how the microphone will sound at a given height.
- The coverage area of the lobes increases at farther distances.

[1] Room conditions: RT60 (reverb time) = 500 ms @ 1kHz, A weighted room noise = 40dBSPL(A)

[2] IEC-602682-16 standard

Intelligibility Scale

The intelligibility scale objectively compares the acoustic performance of the array microphone with a cardioid gooseneck microphone at various distances. This information is useful for predicting how the array microphone will perform at a given distance and to determine an ideal mounting height. The data in the intelligibility scale table is derived from measuring the microphones to meet an equivalent value from the **Speech Transmission Index IEC-602682-16 standard**.

Distances With Equivalent Speech Transmission Index Values

Ceiling Array Microphone (Distance to Talker)	Cardioid Gooseneck Microphone (Distance to Talker)
6 ft (1.83 m)	3.75 feet (1.14 m)
8 ft (2.44 m)	5 feet (1.52 m)
10 ft (3.05 m)	6.25 feet (1.91 m)
12 ft (3.66 m)	7.5 feet (2.29 m)

Data was collected in a typical huddle room with the following measurements:

- **Reverberation decay time:** 500 ms @ 1kHz
- **Noise floor:** 40 dB SPL (A-weighted)

Note: These values are specific to the described room. In a well-controlled acoustic environment, the array microphone may perform with equivalent Speech Transmission Index values at even greater distances. In highly reverberant rooms, the performance is less predictable.

A = Distance between array microphone and talker

B = Distance between cardioid microphone and talker

In this example, the acoustic performance of the array microphone mounted (A) feet from the talker matches the cardioid gooseneck microphone placed at a distance of (B) feet from the talker.

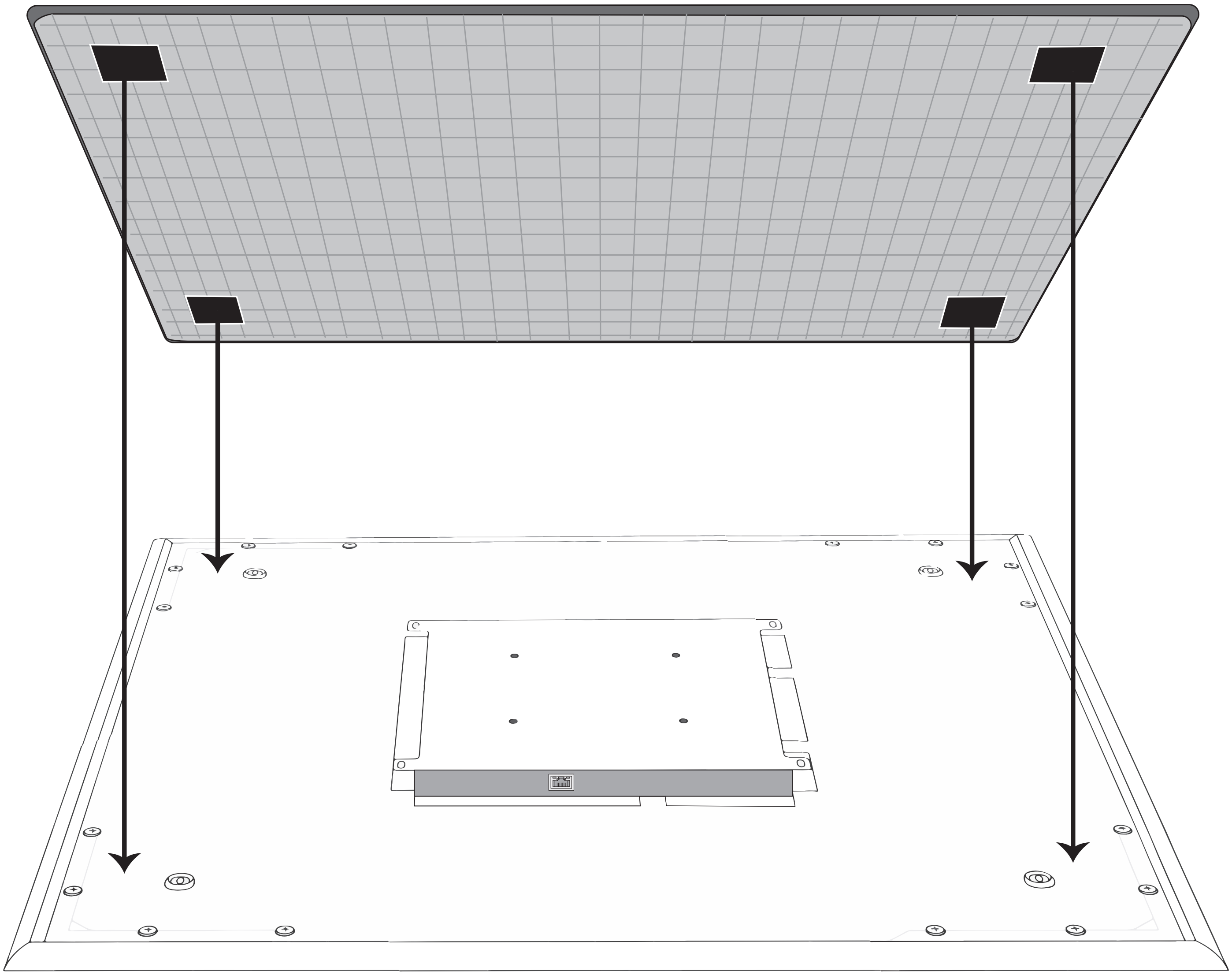
FyreWrap® Fire Protective Wrap System Installation

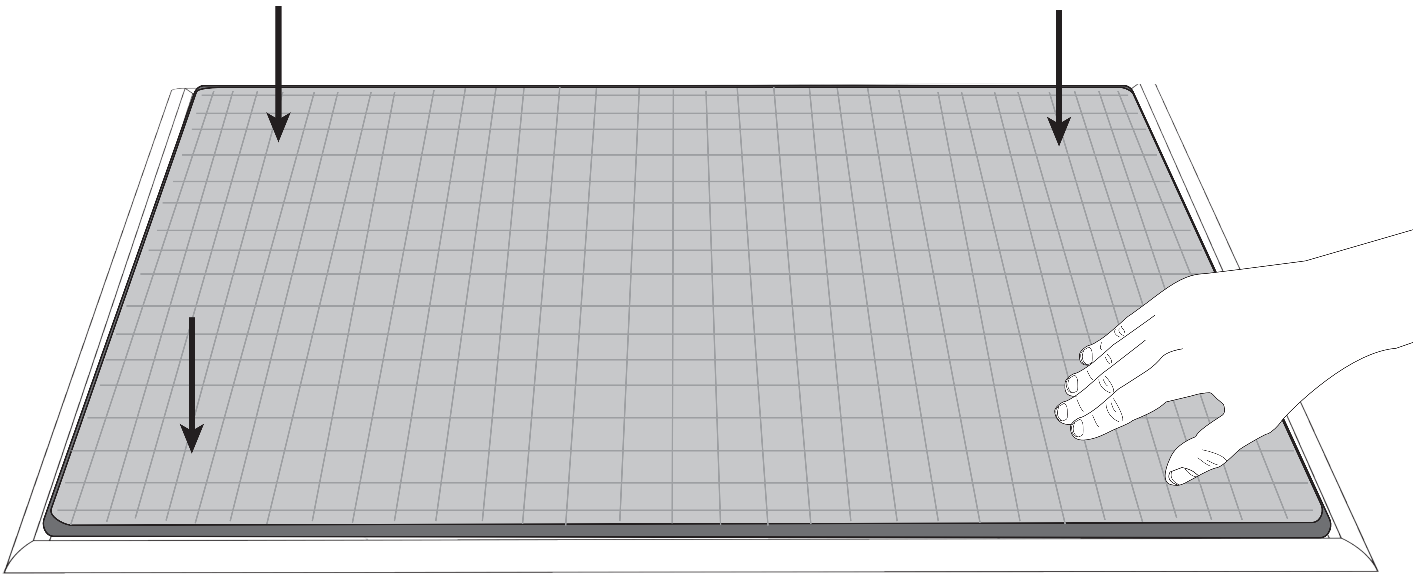
The FyreWrap® fire protective wrap system included with the Microflex® Advance™ MXA910 ceiling array microphone must be installed to meet the UL 2043 plenum rating (suitable for air handling spaces).

Installation

1. Make sure the microphone surface is clean to ensure proper adhesion
2. Remove the paper backing from the 4 adhesive pads on the fire wrap system
3. Align the fire wrap system over the microphone and secure it by applying gentle pressure over the adhesive pads

   Note: make sure to leave enough space to install the Ethernet cable and safety tether (if required).





Installing the Array Microphone

Before you begin:

- Remove the protective plastic cover from the microphone
- Verify the ceiling tile size matches the appropriate model variation
- If using the optional junction box or adapter accessories, install them on the microphone prior to ceiling installation

Model Variations

Model	Ceiling Grid Size	Color
MXA910B	2 x 2 ft (60.9 x 60.9 cm)	Black
MXA910W	2 x 2 ft (60.9 x 60.9 cm)	White
MXA910AL	2 x 2 ft (60.9 x 60.9 cm)	Aluminum
MXA910B-60CM	60 x 60 cm (23.6 x 23.6 in)	Black
MXA910W-60CM	60 x 60 cm (23.6 x 23.6 in)	White
MXA910AL-60CM	60 x 60 cm (23.6 x 23.6 in)	Aluminum
A910-25MM (Adapter)	25 mm expander fits onto 60 cm model, for 62.5 x 62.5 cm installation	---
A910-JB (Junction box accessory)	---	---

Note: see product specifications for array microphone dimensions.

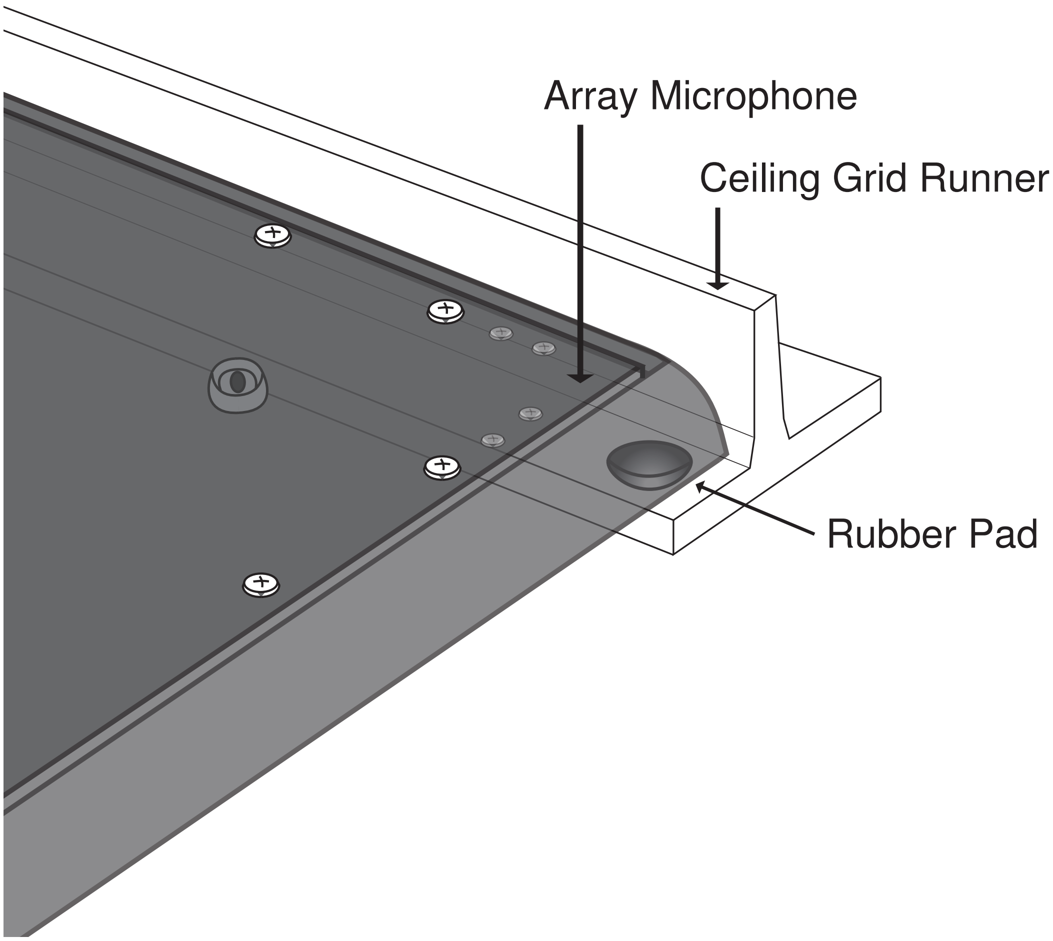
Ceiling Tile Mounting

The array microphone mounts directly in a ceiling tile grid. The microphone is available in two sizes, with an optional adapter kit also available to provide solutions for the most common ceiling tile sizes.

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

Rubber Scratch Protectors

Optional: Before installing the microphone into the ceiling, attach the included rubber pads on the corners of the microphone to prevent scratching.

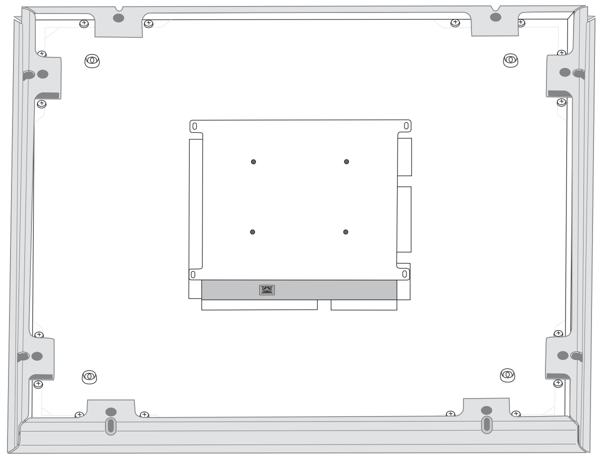


Installation

1. Remove the tile from the ceiling grid where the array microphone will be installed.
2. Run the Ethernet cable above the ceiling grid and through the opening in the ceiling.
   Note: An optional junction box accessory (model A910-JB) mounts on the microphone to directly connect conduit.
3. Plug the Ethernet cable into the array microphone output.
4. Install the microphone into the ceiling grid.

Using the Adapter (625 mm Tile Size)

For ceiling grids which measure 625 x 625 mm, attach the adapter to the array microphone and then follow the ceiling tile mounting instructions.
Note: Only for use with the 60 x 60 cm model

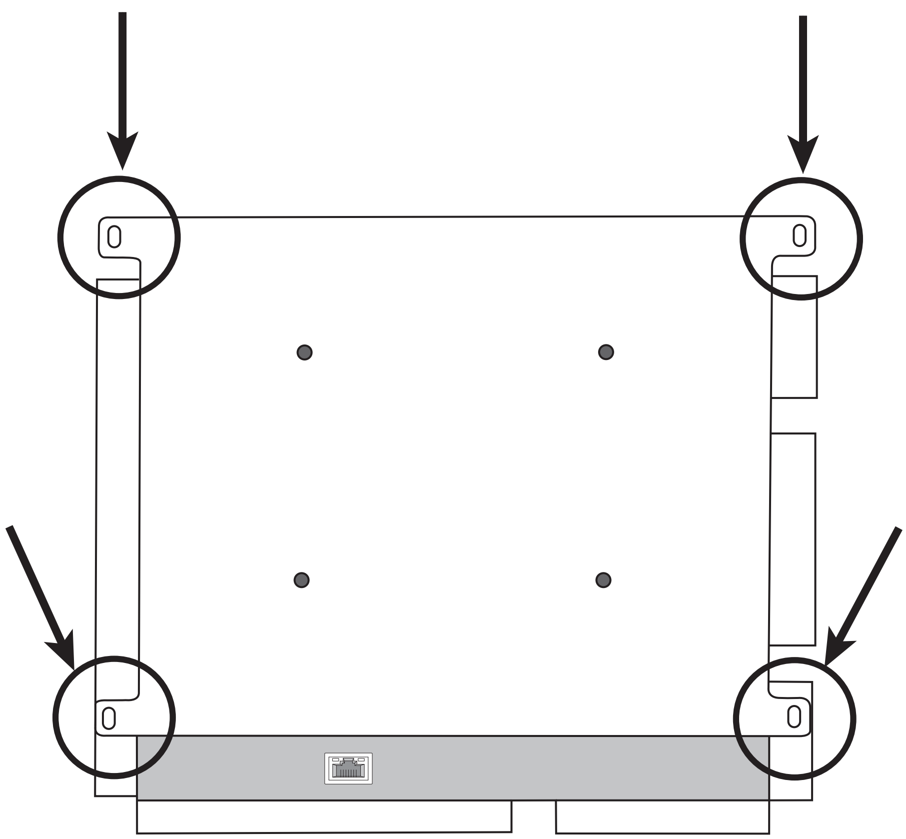


Screw the adapter pieces on as shown, using 2 screws per side.

Other Ceiling Tile Sizes

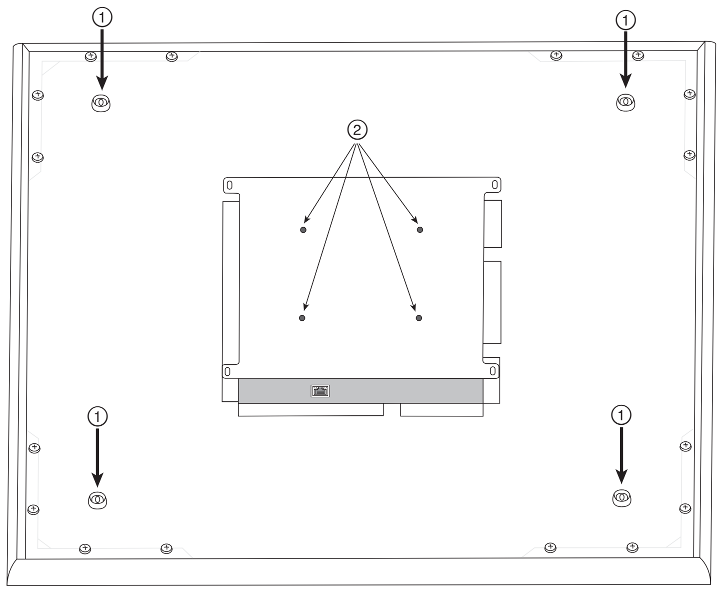
Mounting the microphone in a grid of a different size requires using a suspension mount option or modifying the ceiling grid.

Safety Tether



Attach a safety tether between the building structure and one of the tie-off points on the back panel of the microphone. This safety measure prevents the microphone from falling in an emergency situation. Make sure there is no tension on the safety tether to ensure the microphone is resting properly in the ceiling grid. **Important:** Follow local regulations when attaching the safety tether.

Suspension Mounting



① Wire Suspension Hanging Points (4 mm hole size)
② VESA Mounting Holes

4-point Wire Suspension

Secure the microphone to the ceiling using braided metal cable or other high-strength wire. Use the 4 integrated hanging points on the back of the microphone to securely attach the cable. The hole size in the hanging points is 4 mm (0.15 in).

VESA Standardized Mounting

The rear plate on the microphone 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 (Microphone threaded hole depth = 9.15 mm)
- Hole spacing: 100 mm (square)

Cable Management

To keep the Ethernet cable out of view, use the appropriate method based on the installation type.

Installation	Cable Solution
Ceiling tile	Run the cable above the ceiling grid
VESA (pole mounting)	Guide the cable through the pole to run it above the ceiling grid
4-point wire suspension	Use cable ties to attach the CAT5 cable along one of the hanging wires

Note: If using conduit to contain the cable, the optional junction box accessory (model A910-JB) mounts directly to the rear panel of the microphone.

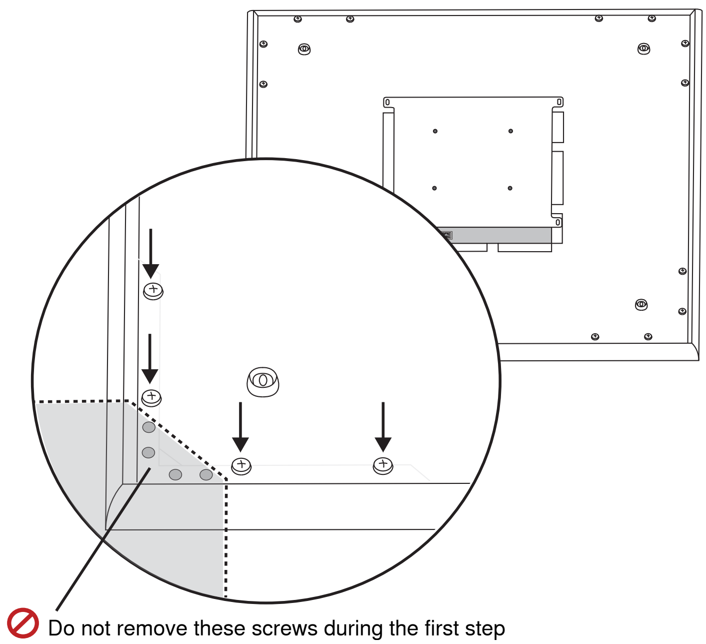
Painting Components

The grille and frame of the array microphone can be painted to blend in with room design. Some basic disassembly is required for painting.

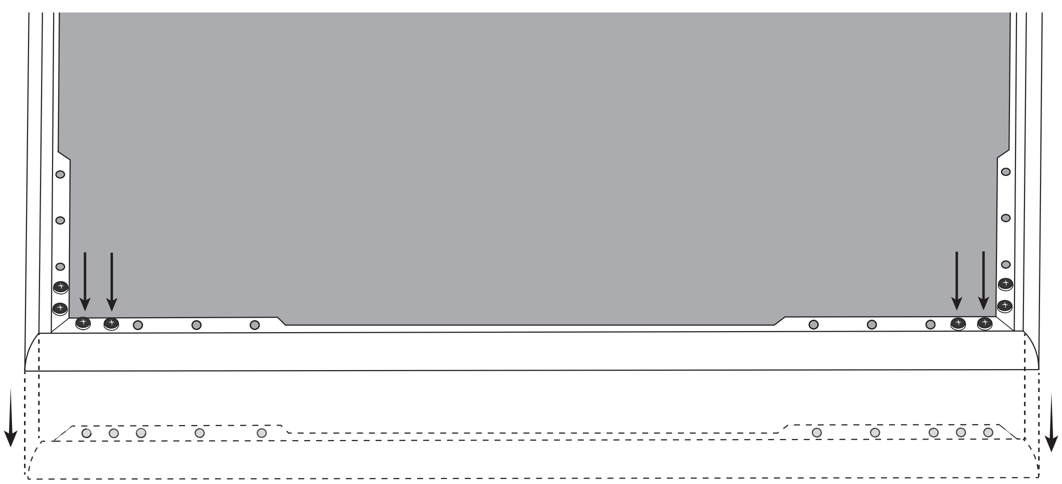
Step 1: Removing the Frame and Grille

1. Remove the screws that hold the main assembly on the frame (6 screws per side). There are washers between these screws and the rear panel.
   Important: Do not remove the screws that are farthest in the corner and recessed into the panel (see graphic).
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 screws from one side of the frame (see image).
5. Remove the side of the frame.
6. Slide the flat grille out of the frame for easy removal of the foam piece.
7. 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.



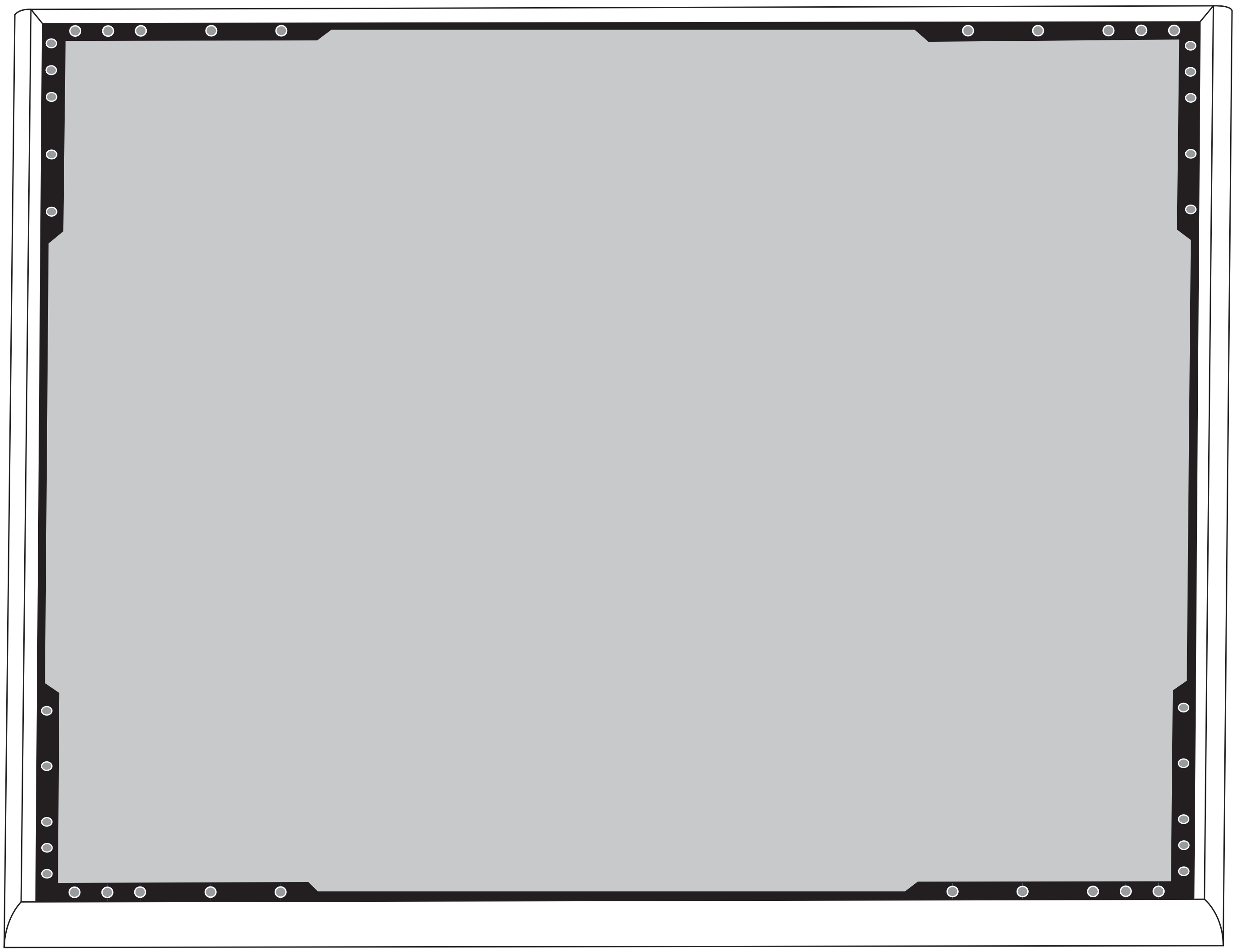
(1.1) Removing the screws to detach the main assembly



(1.4 - 1.5) Removing screws and detaching one side of the frame

Step 2: Masking and Painting

1. Use masking tape to cover the entire extrusion 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 section and allow it to completely dry before reassembling. Do not paint any part of the main assembly.



(2.1) Masking off the extrusion (highlighted in black)

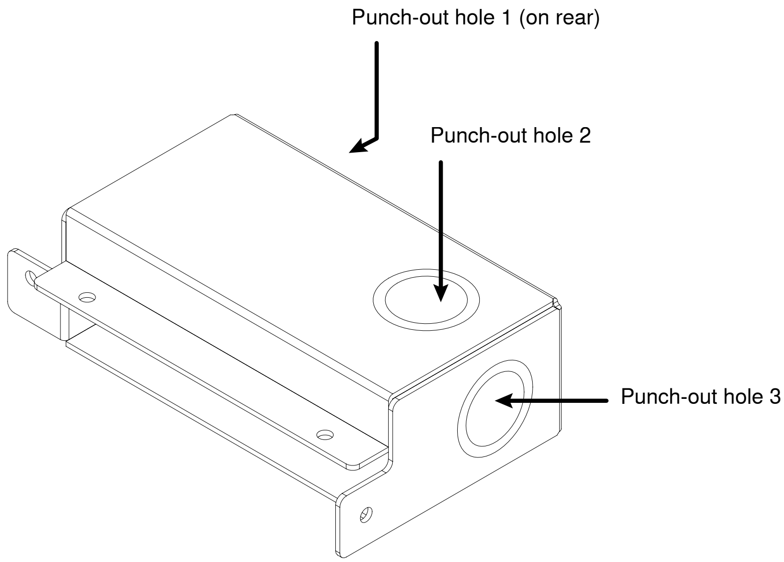
Step 3: Reassembly

1. Secure the foam piece with the hook-and-loop fastener strips around the edges.
2. 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 into the black plastic guide piece.
5. Align the LED with the lightpipe and put the assembly back in place on the frame.

   Note: The label on the assembly is placed on the corner that corresponds to the LED. Use it for reference to ensure proper orientation during reassembly.
6. Install the screws (6 per side) to secure the main assembly to the frame. Do not over-tighten.

Installing the Junction Box Accessory

The A910-JB junction box mounts on the microphone, enabling conduit connections for cable runs. Refer to local building codes and regulations to determine if the junction box is necessary. There are three punch-out sections on the junction box for attaching conduit.

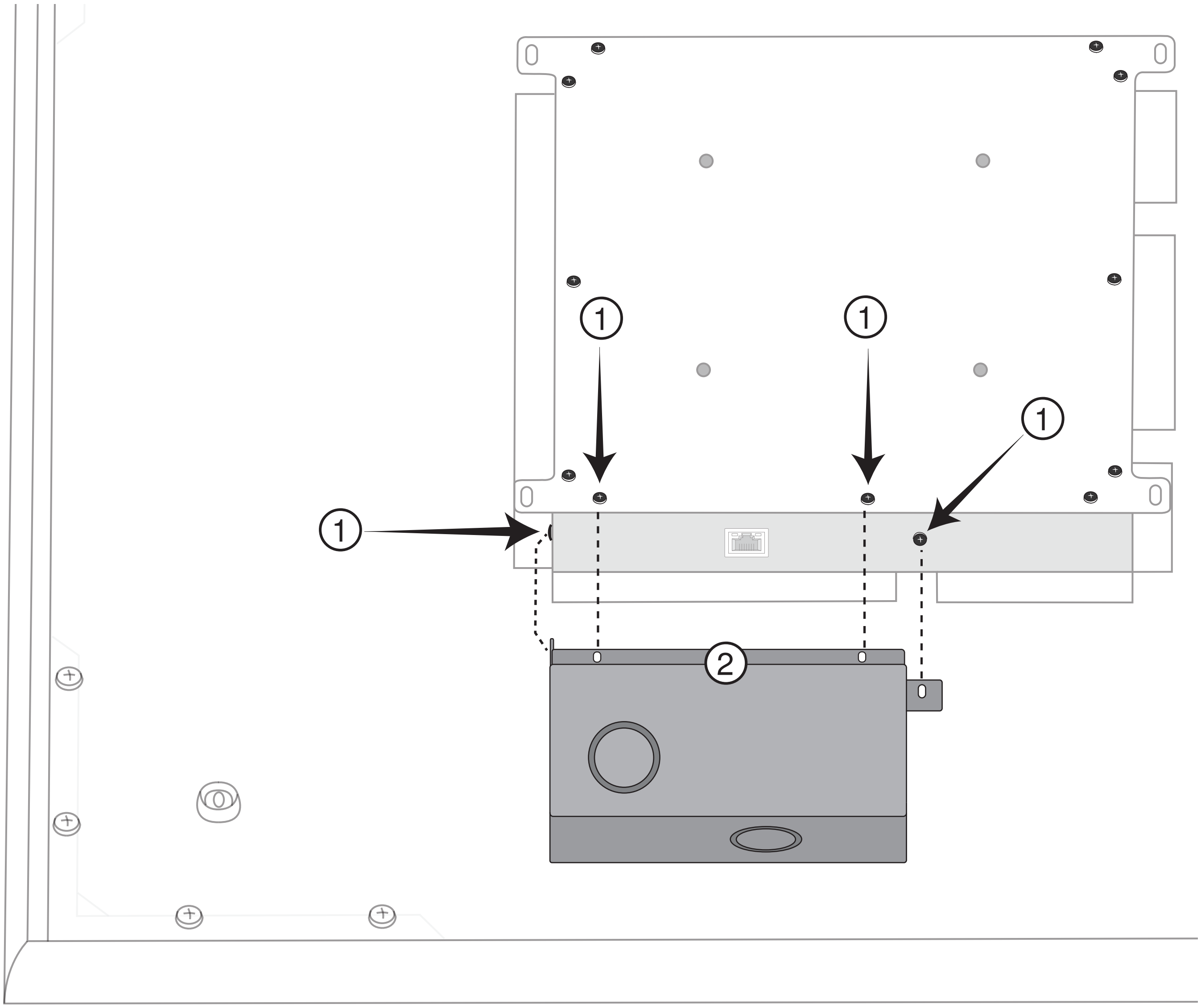


Important: Punch out the necessary holes in the junction box prior to installing it onto the microphone.

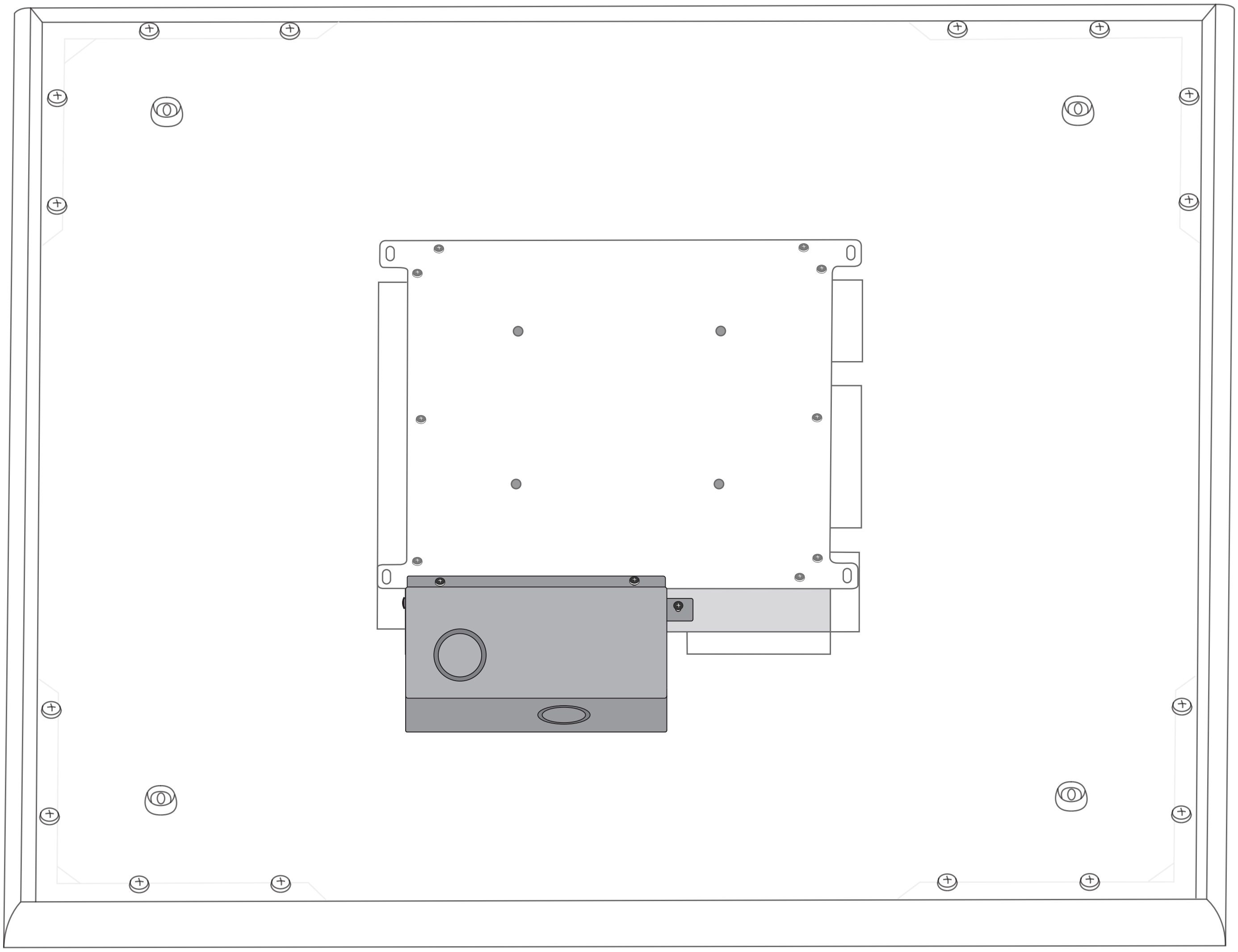
To install:

1. Remove the 4 screws from the microphone as shown.

2. Align the junction box with the screw holes. If possible, plug the network cable into the microphone before securing the junction box.



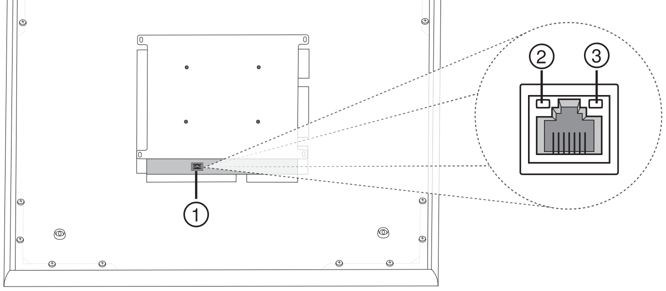
3. Install the 4 screws to secure the junction box on the microphone.



Hardware

Network Ethernet Port

The network port carries all audio, power, and control data. It is located on the back panel as shown.



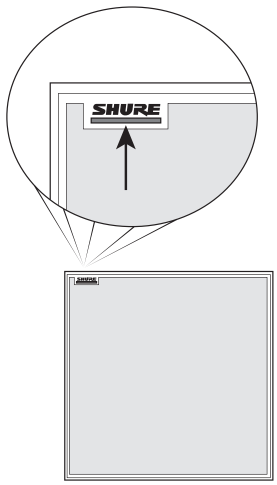
① **Network Port**
RJ-45 jack for network connection.

② **Network Status LED (Green)**
Off = no network link
On = network link established
Flashing = network link active

③ **Network Speed LED (Amber)**
Off = 10/100 Mbps
On = 1 Gbps

LED Light Bar

The LED on the microphone indicates whether the microphone is active or muted, identifies the hardware, and provides confirmation of firmware updates.



Mute LED Behavior

The lighting for mute and active microphone states is configurable to match the behavior of other devices in conference rooms. In the LIGHT BAR PROPERTIES menu, use the drop-down menus to select LED settings.

Microphone Status	LED Behavior / Color
Active	Green (solid)
Mute	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, Blue (moves quickly back and forth across bar)

Note: When the LED is disabled, the LED still illuminates while the device is powering up and when an error state occurs.

Customizing Lighting Settings

Custom LED brightness, colors, and behaviors are assignable in the control software. They can also be controlled through an external control system:

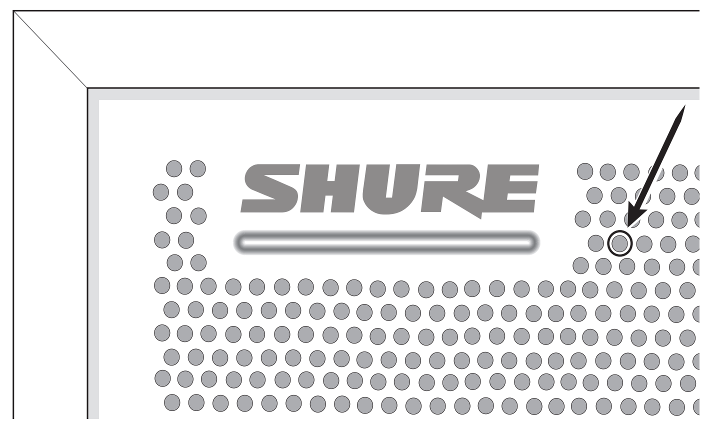
1. Open the Configuration tab
2. Select Light Bar

Dimming and Disabling

To dim or turn off the LED, use the drop-down menu in the Light Bar configuration screen.

Reset Button

The hardware reset button is located inside a grille hole and can be pushed with a paperclip or other small tool. The hole is identified with a gray circle. When looking at the Shure logo, it is the second hole in the fourth row from the top.



Reset Modes

Network reset (press button for 4-8 seconds)
Resets all Shure control and audio network IP settings to factory defaults

Full factory reset (press button for longer than 8 seconds)
Restores all network and web application settings to the factory defaults.

Power Over Ethernet (PoE)

This device requires PoE to operate. It is compatible with both **Class 0** and **Class 3** PoE sources.

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

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

Software Installation, Management, and Security

Accessing the Web Application

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

① Install the Shure Discovery application

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

② Connect the network

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

③ Launch the Discovery application

The app displays all Shure devices that feature a GUI.

④ Identify the hardware

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

⑤ Bookmark the device's web application (recommended)

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

Web Application Browser Compatibility

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

Accessing the Web Application without the Discovery App

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

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

MXA910: http://MXA910-aabbcc.local

Using A Password

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

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

Firmware Updates

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

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

Perform the following steps to update the firmware:

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

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

Firmware Release Requirements

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

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

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

Microphone Configuration

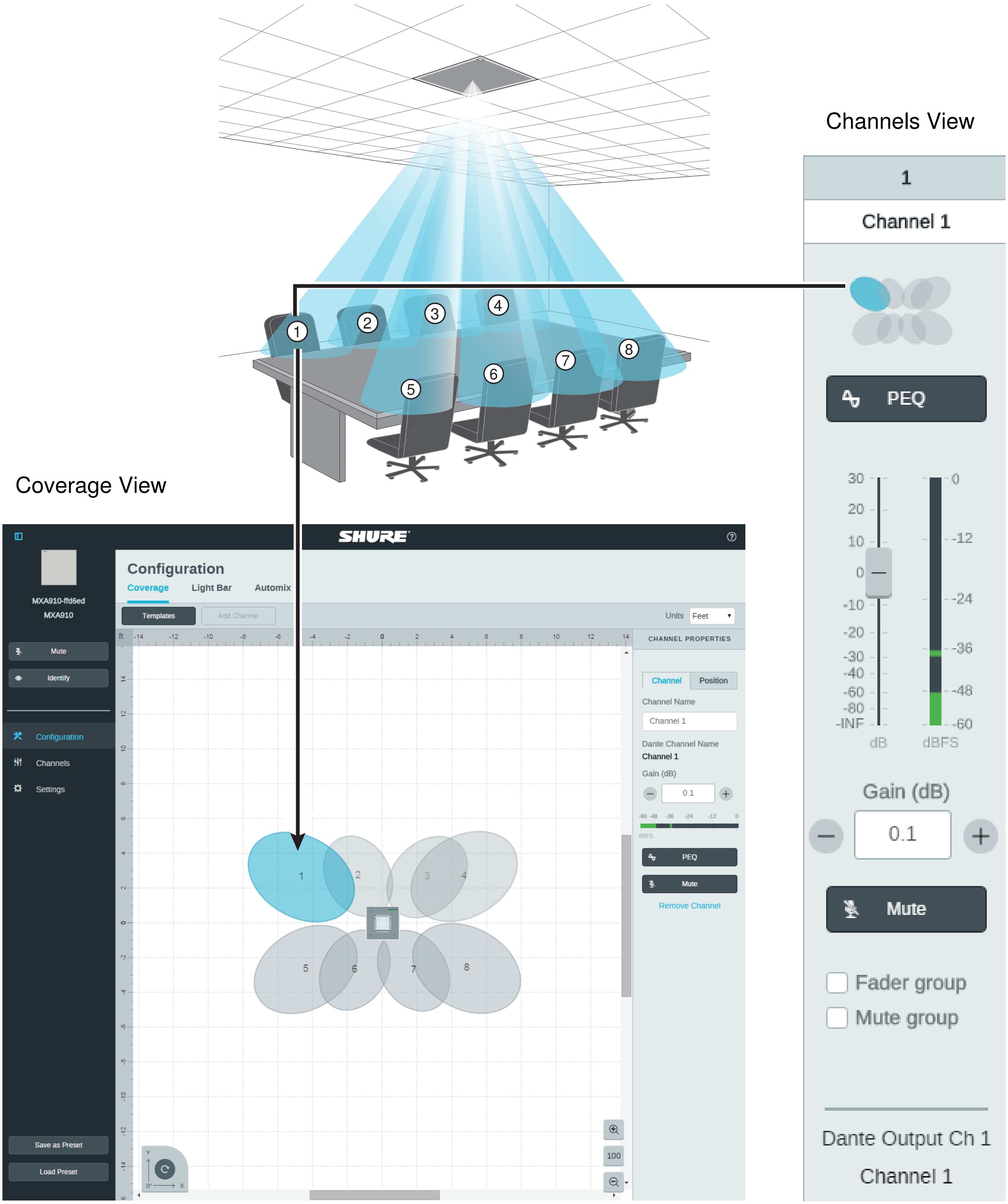
Software Overview

The web application is accessed through a web browser, from a computer on the same network. It allows administrators and technicians to control:

Coverage: Adjust lobe width and location, select templates, save or load presets, customize light bar settings, and run automatic setup.
Channels: Adjust and monitor channel levels, mute channels or channel groups, configure automix settings, and adjust equalizer settings.
Settings: Control network IP settings, device name, passwords, languages, firmware identification, and device reset.

Software Workflow Basics

Think of each lobe as an individual microphone. If there were 8 microphones on the table, each one could be physically moved according to seating arrangements, and would be plugged into a mixer with independent gain and channel controls. With the Microflex Advance Ceiling Array Microphone, the web application delivers control over the physical coverage and audio channel settings, with user presets to quickly switch between configurations. Each lobe is moved according to seating arrangements, with three width settings to change the size of the coverage area. Independent mixer channels control the level and audio properties for each lobe.



Each lobe is represented graphically and can be dragged into place. A corresponding mixer channel provides control over audio settings for each lobe.

Configuring Microphone Coverage

Select the Coverage section inside the Configuration tab to access the coverage configuration tools.

Step 1: Set Microphone Properties

1. Set the Units to feet or meters, according to your preference.
2. Enter a value for the microphone height. This distance is between the floor and the microphone height.
3. If necessary, use the orientation tool at the bottom of the workspace to rotate the configuration view to match the installer's perspective. Use the degree value or the microphone LED in the workspace to confirm the position.

Step 2: Select a Template

Select a template that is the closest match to the seating scenario.

Step 3: Set Coverage

Lobes can be automatically or manually positioned. The Auto Position feature detects sound from a specific area when the microphone has already been installed. Manual positioning is useful for designers, when the room has been specified but the microphone has not yet been installed.

Automatic Positioning

Automatic positioning determines the talker position (X/Y values) and the talker height through a simple soundcheck.

1. Select a channel to position the lobe.
2. Choose Auto Position in the Channel Properties section to start the process.
3. Follow the directions provided for each step presented in the software.
4. Repeat for each channel as needed.

Best Practices

- Verify that the microphone height has been accurately entered in the Array Properties.
- Use speech as the sound source. Do not use test tones.
- Eliminate noise sources in the room if possible, such as projectors or fans.
- For the best acoustics, talkers should not be directly against a wall or in a corner, as acoustic reflections can reduce aiming accuracy.
- Speak from the exact location where the lobe should be positioned. It is important to match the height (sitting or standing), as talker height is determined through automatic positioning.
- If a lobe needs to cover an area with multiple talkers, talk from the center of the region. Manual adjustments may be necessary after the lobe is positioned to ensure the entire area is covered.

Manual Positioning

1. Move lobes to cover the appropriate areas.
    - Lobes are independently selectable and can be moved anywhere within the maximum allowed coverage area. If dragged outside of this region, lobes turn red and revert back to the last acceptable position.
    - Lobes can be positioned in two ways: drag them into place, or manually enter a distance from the microphone. Distance values are X and Y along the grid, with the microphone centered at 0.
    - Use the grid to measure the precise placement of the lobe.
2. Enter the talker height by selecting a lobe and providing the value in the Channel Properties. This ensures accurate aiming.

Using the Reference Grid

The grid provides a frame of reference to show distances relative to the center of the microphone. The selected unit (feet/meters) applies to the grid, so make sure it has been set appropriately. If the distances between the talkers and the microphone are known, the grid can visually verify that the lobe placement is correct.

Note: Values to the left of the microphone on the horizontal X-axis, and below the microphone on the vertical Y-axis are measured as negative values. When the rotate feature is used, the grid values change to match the new perspective.

Adding and Removing Channels

To add channels into the workspace, use the Add Channel button located above the workspace.

To remove a channel, select a lobe and select Remove Channel, located under the Channel Properties. The delete key may also be used to remove the selected channel.

Step 4: Adjust Lobe Width

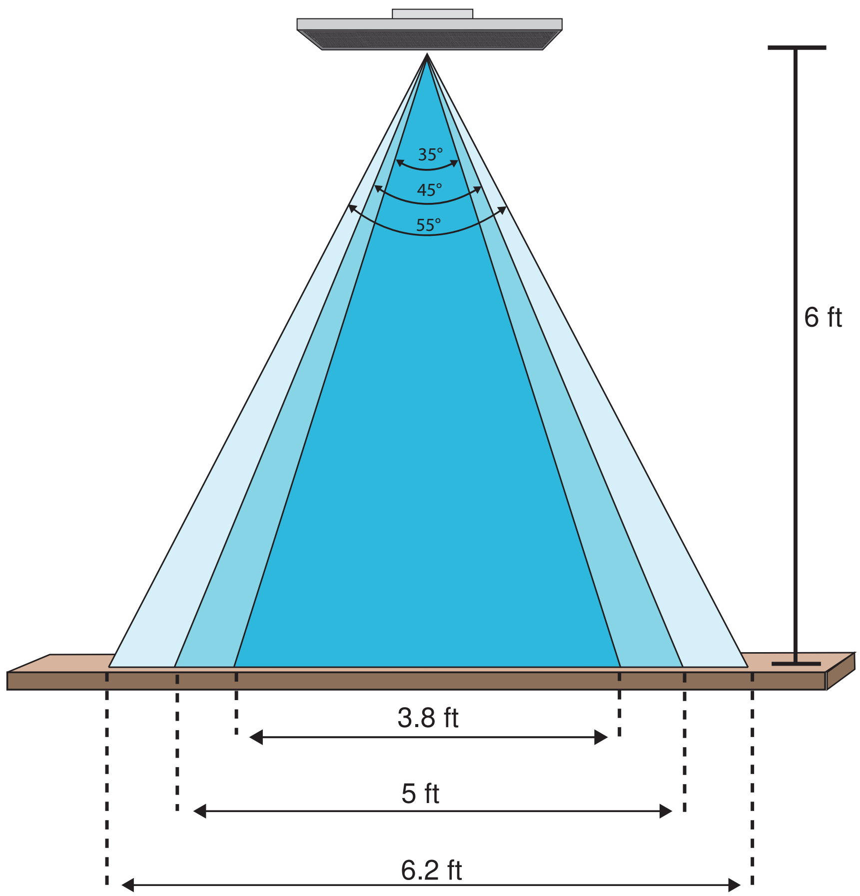
Independent width control makes it possible for some lobes to capture individual talkers (narrow), while other lobes cover multiple talkers (wide).

To change a lobe width:

1. Select Configuration in the web application
2. Select a lobe in the room view to reveal the channel properties menu
3. Choose a channel width setting from the pull-down menu. The width is calculated and displayed, based on the lobe location and ceiling height.

Lobe Width Settings:

- Narrow (35°)
- Medium (45°)
- Wide (55°)



Lobe widths for the three settings with the microphone 6 feet above a table

Adjusting Levels

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

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

Parametric Equalizer

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

Common equalizer applications:

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

Setting Filter Parameters

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

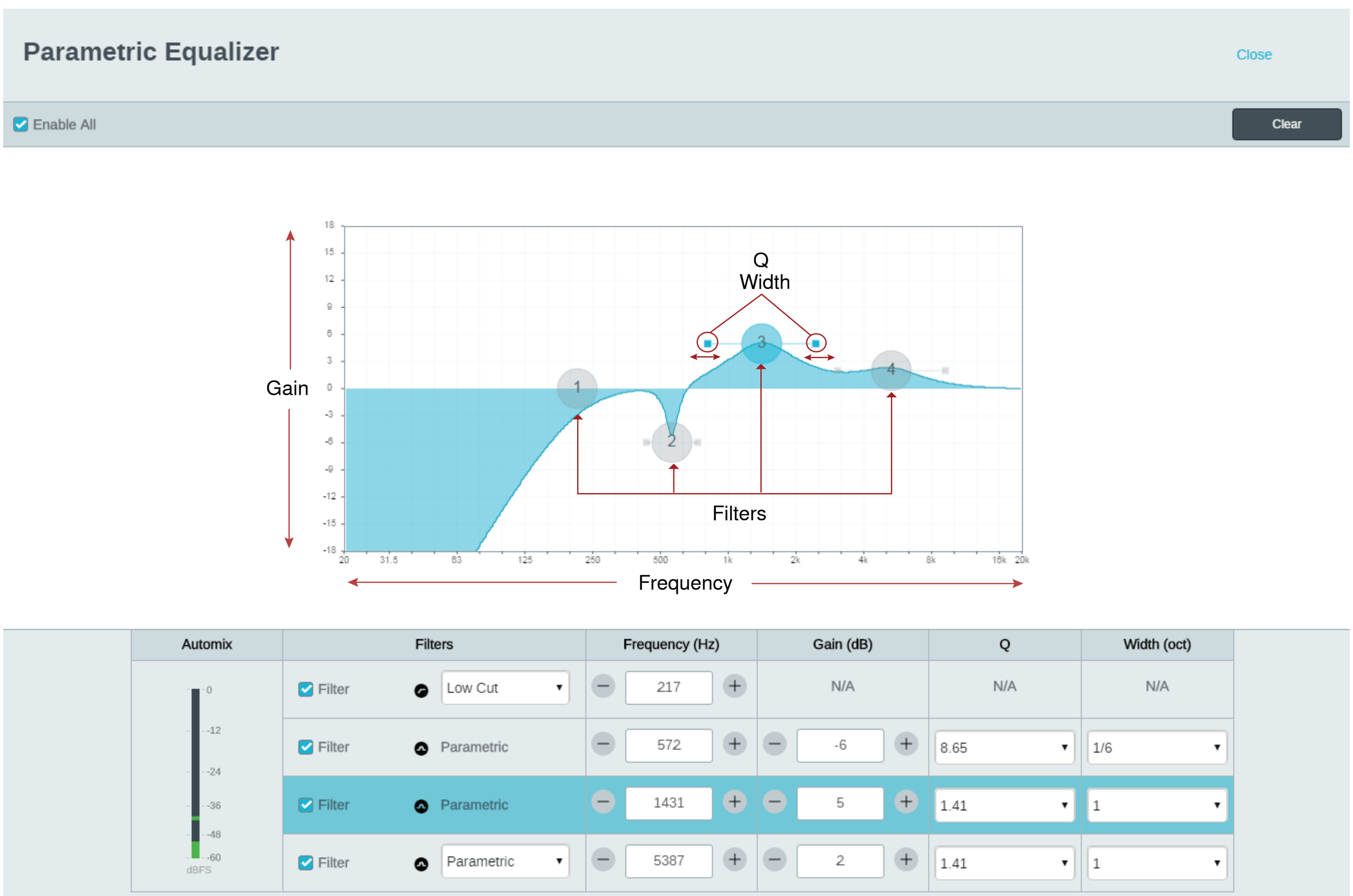
Filter Type
Only the first and last band have selectable filter types.
Parametric: Attenuates or boosts the signal within a customizable frequency range
Low Cut: Rolls off the audio signal below the selected frequency
Low Shelf: Attenuates or boosts the audio signal below the selected frequency
High Cut: Rolls off the audio signal above the selected frequency
High Shelf: Attenuates or boosts the audio signal above the selected frequency

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

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

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

Width
Adjusts the range of frequencies affected by the filter. The value is represented in octaves.
Note: the Q and width parameters affect the equalization curve in the same way. The only difference is the way the values are represented.



When to Use the Channel and Automix Equalizers

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

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

Equalizer Applications

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

EQ Application	Suggested Settings
Treble boost for improved speech intelligibility	Add a high shelf filter to boost frequencies greater than 1 kHz by 3-6 dB
HVAC noise reduction	Add a low cut filter to attenuate frequencies below 200 Hz
Reduce flutter echoes and sibilance	Identify the specific frequency range that "excites" the room: Set a narrow Q value Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 1 kHz and 6 kHz to pinpoint the range of flutter echoes or sibilance Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound
Reduce hollow, resonant room sound	Identify the specific frequency range that "excites" the room: Set a narrow Q value Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 300 Hz and 900 Hz to pinpoint the resonant frequency Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound

Custom Presets

Use presets to quickly save and recall settings. Up to 10 presets can be stored on each device to match various seating arrangements. A preset saves all device settings except for the Device Name, IP Settings, and Passwords. Importing and exporting presets into new installations saves time and improves workflow.

Open the presets menu to reveal preset options:

save as preset:
Saves settings to the device

load preset:
Opens a configuration from the device

import from file:
Downloads a preset file from a computer onto the device. Files may be selected through the browser or dragged into the import window.

export to file:
Saves a preset file from the device onto a computer

Mute and Fader Groups

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

Automix Channel

This channel automatically mixes the audio from all channels to deliver a convenient, single output. The automix channel is always active. To use it, simply route the channel in Dante™ Controller to the desired output.

There are two ways to access AUTOMIX settings:

From the configuration screen:

1. Select Configuration
2. Open the AUTOMIX tab

From the channels screen:

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

Automix Modes

Classic

Classic mode emulates the Shure SCM820 automixer (in its default settings). It is renowned for fast-acting, seamless channel gating and consistent perceived ambient sound levels. Off-attenuation in this mode is fixed at -15 dB per channel, regardless of the number of open channels.

Smooth

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

Number of channels enabled	Off-attenuation (dB)
2	-3.0
3	-4.8
4	-6.0
5	-7.0
6	-7.8
7	-8.4
8	-9.0

Custom

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

Manual

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

Automix Settings

Leave Last Mic On
Keeps the most recently used microphone channel active. The purpose of this feature is to keep natural room sound in the signal so that meeting participants on the far end know the audio signal has not been interrupted.

Gating Sensitivity
Changes the threshold of the level at which the gate is opened

Off Attenuation
Sets the level of signal reduction when a channel is not active

Hold Time
Sets the duration for which the channel remains open after the level drops below the gate threshold

Maximum Open Channels
Sets the maximum number of simultaneously active channels

Priority
When selected, this channel gate activates regardless of the number of maximum open channels.

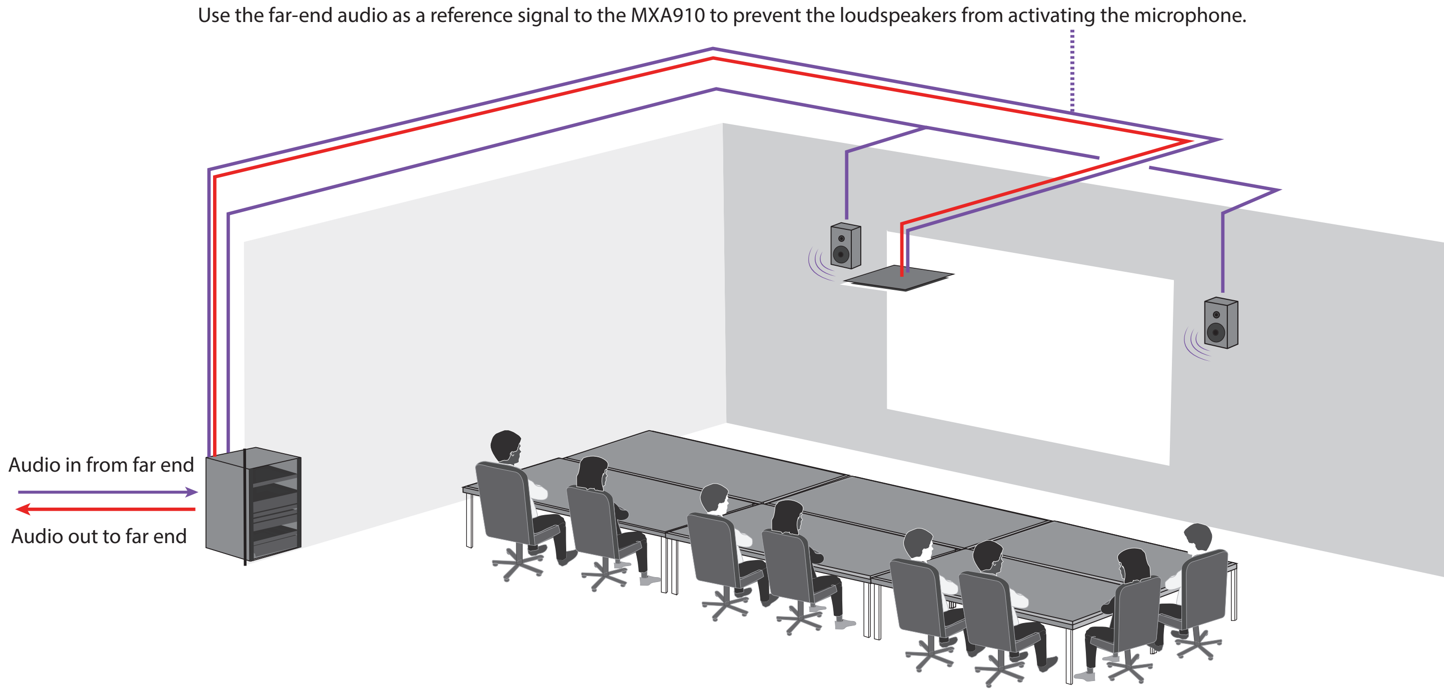
Echo Reduction

In audio conferencing, a talker may hear their voice echo as a result of the microphone capturing far-end audio from loudspeakers.

The echo reduction feature prevents the far-end signal from activating the microphone. Ideal for installations in which per-channel DSP echo cancellation is not within a project budget, echo reduction is highly effective for connecting directly to a computer or video codec which hosts a single-channel echo canceller.

How It Works

An echo reference signal from the far end is routed through Dante Controller software to the microphone's Intellimix processing algorithm. The processor uses this signal to prevent the microphone from gating on and capturing audio from the loudspeakers.



Enabling Echo Reduction

1. Use Dante Controller software to route the incoming far end audio signal to the Echo Suppression Reference In channel on the MXA910.
2. In the MXA910 web application, enter Configuration> AUTOMIX
3. Enable echo reduction by selecting a strength setting in the pull-down menu. Soft, medium, and hard settings provide far-end attenuation and apply gain to the reference channel to ensure proper gating.

   Note: the off-attenuation setting changes to -56 dB, and the Leave last microphone on setting is set to OFF when echo reduction is enabled.

Dante and Networking

Digital Audio Networking

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

Switch Recommendations for Dante Networking

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

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

Cable Requirements

Always use Cat5E cable or higher.

QoS (Quality of Service) Settings

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

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

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

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

Table provided courtesy of Audinate®

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

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

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

Networking Terminology

PTP (Precision Time Protocol): Used to synchronize clocks on the network
DSCP (Differentiated Services Code Point): Standardized identification method for data used in layer 3 QoS prioritization

Networking

Networking Best Practices

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

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

Network Audio and Shure Control Data

Microflex® Advance™ devices transport two types of data over the network: Shure Control and Network Audio.

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

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

Device IP Settings

Configure IP

Sets IP mode of the selected network interface:

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

IP Settings

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

MAC Address

The network interface's unique identification.

Configuring IP Settings

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

To configure the IP properties, follow these steps:

1. Open the web application.

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

Manually Assigning Static IP Address

To manually assign IP addresses, follow these steps:

1. Open the web application.
2. Go to the Settings tab and select Network.
3. Select Manual as the Configure IP setting.
4. Enter the IP settings.

Setting Latency

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

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

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

Latency Recommendations

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

Operating the Control Software over Wi-Fi

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

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

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

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

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

- Cisco
- Linksys
- Apple

IP Ports and Protocols

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

Port	TCP/UDP	Protocol	Description	Factory Default
162	tcp	SNMP	Not supported	Closed
2202	tcp	ASCII	Required for 3rd party control strings	Open
5353	udp	mDNS†	Required for device discovery	Open
5568	udp	SDT†	Required for inter-device communication	Open
8023	tcp	Telnet	Debug console interface	Password
8180*	tcp	HTML	Required for web application	Open
8427	udp	Multcast SLP†	Required for inter-device communication	Open
64000	tcp	Telnet	Required for Shure firmware update	Open

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

0 1

Using a Third-Party Control System

The microphone receives logic commands over the network. Many parameters controlled through the web application can be controlled through a third party control system, using the appropriate command string.

Common applications:

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

A complete list of command strings is available in the device help or from www.shure.com.

MXA910 Microflex® Advance™ Command Strings

This document can also be found at: http://shure.custhelp.com/app/answers/detail/a_id/6058

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

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

Conventions

The device has 4 types of strings:

GET
Finds the status of a parameter. After the AMX/Crestron sends a GET command, the MXA910 responds with a REPORT string

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

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

SET
Changes the status of a parameter. After the AMX/Crestron sends a SET command, the MXA910 will respond with a REPORT string to indicate the new value of the parameter.

REP
When the MXA910 receives a GET or SET command, it will reply with a REPORT command to indicate the status of the parameter. REPORT is also sent by the device when a parameter is changed on the MXA910 or through the GUI.

SAMPLE
Used for metering audio levels.

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

Most parameters will send a REPORT command when they change. Thus, it is not necessary to constantly query parameters. The MXA910 will send a REPORT command when any of these parameters change.

The character “x” in all of the following strings represents the channel of the MXA910 and can be ASCII numbers 0 through 9 as in the following table.

0	All channels
1 through 8	Individual channels
9	Automix output

Command Strings (Common)

Get All		
	Command String: `< GET x ALL >`	Where x is ASCII channel number: 0 through 9. Use this command on first power on to update the status of all parameters.
	MXA910 Response: `< REP ... >`	The MXA910 responds with individual Report strings for all parameters.
Get Channel Name		
	Command String: `< GET x CHAN_NAME >`	Where x is ASCII channel number: 0 through 9.
	MXA910 Response: `< REP x CHAN_NAME {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >`	Where yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy is 31 characters of the user name. The MXA910 always responds with a 31 character name.
Get Device ID		
	Command String: `< GET DEVICE_ID >`	The Device ID command does not contain the x channel character, as it is for the entire device.
	MXA910 Response: `< REP DEVICE_ID {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >`	Where yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy is 31 characters of the device ID. The MXA910 always responds with a 31 character device ID.
Get Audio Gain		
	Command String: `< GET x AUDIO_GAIN_HI_RES >`	Where x is ASCII channel number: 1 through 9. Channel number 0 (all channels) is not valid for this command.
	MXA910 Response: `< REP x AUDIO_GAIN_HI_RES yyyy >`	Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.
Set Audio Gain		
	Command String: `< SET x AUDIO_GAIN_HI_RES yyyy >`	Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.
	MXA910 Response: `< REP x AUDIO_GAIN_HI_RES yyyy >`	Where yyyy takes on the ASCII values of 0000 to 1400.
Increase Audio Gain by n dB		
	Command String: `< SET x AUDIO_GAIN_HI_RES INC nn >`	Where nn is the amount in one-tenth of a dB to increase the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).

	MXA910 Response: `< REP x AUDIO_GAIN_HI_RES yyyy >`	Where yyyy takes on the ASCII values of 0000 to 1400.
Decrease Audio Gain by n dB		
	Command String: `< SET x AUDIO_GAIN_HI_RES DEC nn >`	Where nn is the amount in one-tenth of a dB to decrease the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).
	MXA910 Response: `< REP x AUDIO_GAIN_HI_RES yyyy >`	Where yyyy takes on the ASCII values of 0000 to 1400.
Get Channel Audio Mute		
	Command String: `< GET x AUDIO_MUTE >`	Where x is ASCII channel number: 0 through 9. Channel Audio Mute is pre-meter

	MXA910 Response:
	`< REP x AUDIO_MUTE ON >`
	`< REP x AUDIO_MUTE OFF >`
		The MXA910 will respond with one of these strings.

Mute Channel Audio		
	Command String: `< SET x AUDIO_MUTE ON >`	
	MXA910 Response: `< REP x AUDIO_MUTE ON >`	
Unmute Channel Audio		
	Command String: `< SET x AUDIO_MUTE OFF >`	
	MXA910 Response: `< REP x AUDIO_MUTE OFF >`	
Toggle Channel Audio Mute		
	Command String: `< SET x AUDIO_MUTE TOGGLE >`	

	MXA910 Response:
	`< REP x AUDIO_MUTE ON >`
	`< REP x AUDIO_MUTE OFF >`
		The MXA910 will respond with one of these strings.

Get Device Audio Mute		
	Command String: `< GET DEVICE_AUDIO_MUTE >`	Device Audio Mute is post-meter.

	MXA910 Response:
	`< REP DEVICE_AUDIO_MUTE ON >`
	`< REP DEVICE_AUDIO_MUTE OFF >`
		The MXA910 will respond with one of these strings.

Mute Device Audio		
	Command String: `< SET DEVICE_AUDIO_MUTE ON >`	
	MXA910 Response: `< REP DEVICE_AUDIO_MUTE ON >`	
Unmute Device Audio		
	Command String: `< SET DEVICE_AUDIO_MUTE OFF >`	
	MXA910 Response: `< REP DEVICE_AUDIO_MUTE OFF >`	
Toggle Device Audio Mute		

	Command String: `< SET DEVICE_AUDIO_MUTE TOGGLE >`	

	MXA910 Response:
	`< REP DEVICE_AUDIO_MUTE ON >`
	`< REP DEVICE_AUDIO_MUTE OFF >`
		The MXA910 will respond with one of these strings.

Get Output Clip Status		
	Command String: `< GET x AUDIO_OUT_CLIP_INDICATOR >`	Where x is ASCII channel number: 0 through 9. It is not necessary to continually send this command. The MXA910 will send a REPORT message whenever the status changes.

	MXA910 Response:
	`< REP x AUDIO_OUT_CLIP_INDICATOR ON >`
	`< REP x AUDIO_OUT_CLIP_INDICATOR OFF >`
		The MXA910 will respond with one of these strings.

Flash Lights on Microphone		

	Command String:
	`< SET FLASH ON >`
	`< SET FLASH OFF >`
		Send one of these commands to the MXA910. The flash automatically turns off after 30 seconds.

	MXA910 Response:
	`< REP FLASH ON >`
	`< REP FLASH OFF >`
		The MXA910 will respond with one of these strings.

Turn Metering On		
	Command String: `< SET METER_RATE sssss >`	Where sssss is the metering speed in milliseconds. Setting sssss=0 turns metering off. Minimum setting is 100 milliseconds. Metering is off by default.

	MXA910 Response:
	`< REP METER_RATE sssss >`
	`< SAMPLE aaa bbb ccc ddd eee fff ggg hhh iii >`
		Where aaa, bbb, etc is the value of the audio level received and is 000-060.
		aaa = output 1
		bbb = output 2
		ccc = output 3
		ddd = output 4
		eee = output 5
		fff = output 6
		ggg = output 7
		hhh = output 8
		iii = output 9

Stop Metering		
	Command String: `< SET METER_RATE 0 >`	A value of 00000 is also acceptable.
	MXA910 Response: `< REP METER_RATE 00000 >`	
Get Audio Peak Level		
	Command String: `< GET x AUDIO_IN_PEAK_LVL >`	
	MXA910 Response: `< REP x AUDIO_IN_PEAK_LVL nn >`	Where nn is the audio level and is 00-60.
Get Audio RMS Level		
	Command String: `< GET x AUDIO_IN_RMS_LVL >`	

	MXA910 Response: < REP x AUDIO_IN_RMS_LVL nn >	Where nn is the audio level and is 00-60.
Get Preset		
	Command String: < GET PRESET >	
	MXA910 Response: < REP PRESET nn >	Where nn is the preset number 01-10.
Set Preset		
	Command String: < SET PRESET nn >	Where nn is the preset number 1-10. (Leading zero is optional when using the SET command).
	MXA910 Response: < REP PRESET nn >	Where nn is the preset number 01-10.
Get Preset Name		
	Command String: < GET PRESET1 > < GET PRESET2 > < GET PRESET3 > etc	Send one of these strings to the MXA910.
	MXA910 Response: < REP PRESET1 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET2 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET3 {yyyyyyyyyyyyyyyyyyyyyyyyy} > etc	Whereyyyyyyyyyyyyyyyyyyyyyyyyy is 25 characters of the device ID. The MXA910 always responds with a 25 character device ID
Get Gate Out Status		
	Command String: < GET x AUTOMIX_GATE_OUT_EXT_SIG >	Where x is ASCII channel number: 0 through 8. It is not necessary to continually send this command. The MXA910 will send a REPORT message whenever the status changes.
	MXA910 Response: < REP x AUTOMIX_GATE_OUT_EXT_SIG ON > < REP x AUTOMIX_GATE_OUT_EXT_SIG OFF >	The MXA910 will respond with one of these strings.
Get LED State		
	Command String: < GET DEV_LED_IN_STATE >	
	MXA910 Response: < REP DEV_LED_IN_STATE OFF > < REP DEV_LED_IN_STATE ON >	The MXA910 will respond with one of these strings.
Set LED State		
	Command String: < SET DEV_LED_IN_STATE OFF > < SET DEV_LED_IN_STATE ON >	Send one of these commands to the MXA910.
	MXA910 Response: < REP DEV_LED_IN_STATE OFF > < REP DEV_LED_IN_STATE ON >	The MXA910 will respond with one of these strings.
Get LED Brightness		
	Command String: < GET LED_BRIGHTNESS >	

	MXA910 Response: `< REP LED_BRIGHTNESS n >`	Where n can take on the following values: 0 = LED disabled 1 = LED dim 2 = LED default
	Set LED Brightness	
	Command String: `< SET LED_BRIGHTNESS n >`	Where n can take on the following values: 0 = LED disabled 1 = LED dim 2 = LED default
	MXA910 Response: `< REP LED_BRIGHTNESS n >`	
	Get LED Mute Color	
	Command String: `< GET LED_COLOR_MUTED >`	
	MXA910 Response: `< REP LED_COLOR_MUTED nnnn >`	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	Set LED Mute Color	
	Command String: `< SET LED_COLOR_MUTED nnnn >`	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	MXA910 Response: `< REP LED_COLOR_MUTED nnnn >`	
	Get LED Unmute Color	
	Command String: `< GET LED_COLOR_UNMUTED >`	
	MXA910 Response: `< REP LED_COLOR_UNMUTED nnnn >`	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	Set LED Unmute Color	
	Command String: `< SET LED_COLOR_UNMUTED nnnn >`	Where nnnn can be RED, GREEN, BLUE, PINK, PURPLE, YELLOW, ORANGE, or WHITE
	MXA910 Response: `< REP LED_COLOR_UNMUTED nnnn >`	
	Get LED Mute Flashing	
	Command String: `< GET LED_STATE_MUTED >`	
	MXA910 Response: `< REP LED_STATE_MUTED nnn >`	Where nnn can be ON, OFF, or FLASHING
	Set LED Mute Flashing	
	Command String: `< SET LED_STATE_MUTED nnn >`	Where nnn can be ON, OFF, or FLASHING
	MXA910 Response: `< REP LED_STATE_MUTED nnn >`	
	Get LED Unmute Flashing	
	Command String: `< GET LED_STATE_UNMUTED >`	
	MXA910 Response: `< REP LED_STATE_UNMUTED nnn >`	Where nnn can be ON, OFF, or FLASHING

Set LED Unmute Flashing		
	Command String: `< SET LED_STATE_UNMUTED nnn >`	Where nnn can be ON, OFF, or FLASHING
	MXA910 Response: `< REP LED_STATE_UNMUTED nnn >`	
Get X-Axis Beam (Lobe) Steering		
	Command String: `< GET x BEAM_X >`	Where the X-Axis is parallel with the Shure logo.
	MXA910 Response: `< REP x BEAM_X nnnn >`	Where nnnn is 0000-3048 in centimeters. The value 1524 is the centerline of the MXA910.
Set X-Axis Beam (Lobe) Steering		
	Command String: `< SET x BEAM_X nnnn >`	Where nnnn is 0000-3048 in centimeters. The value 1524 is the centerline of the MXA910.
	MXA910 Response: `< REP x BEAM_X nnnn >`	
Get Y-Axis Beam (Lobe) Steering		
	Command String: `< GET x BEAM_Y >`	Where the Y-Axis is perpendicular to the X-Axis.
	MXA910 Response: `< REP x BEAM_Y nnnn >`	Where nnnn is 0000-3048 in centimeters. The value 1524 is the centerline of the MXA910.
Set Y-Axis Beam (Lobe) Steering		
	Command String: `< SET x BEAM_Y nnnn >`	Where nnnn is 0000-3048 in centimeters. The value 1524 is the centerline of the MXA910.
	MXA910 Response: `< REP x BEAM_Y nnnn >`	
Get Beam (Lobe) Height		
	Command String: `< GET x BEAM_Z >`	Where height is the distance down from the MXA910.
	MXA910 Response: `< REP x BEAM_Z nnn >`	Where nnn is 000-914 in centimeters.
Set Beam (Lobe) Height		
	Command String: `< SET x BEAM_Z nnn >`	Where nnn is 000-914 in centimeters.
	MXA910 Response: `< REP x BEAM_Z nnn >`	
Get Beam (Lobe) Width		
	Command String: `< GET x BEAM_W >`	
	MXA910 Response: `< REP x BEAM_W nnnn >`	Where nnnn can be WIDE, MEDIUM, or NARROW
Set Beam (Lobe) Width		
	Command String: `< SET x BEAM_W nnnn >`	Where nnnn can be WIDE, MEDIUM, or NARROW
	MXA910 Response: `< REP x BEAM_W nnnn >`	

Troubleshooting

Troubleshooting

Problem	Solution
Software lags in Google Chrome browser	Problem is browser-related. Turn off hardware acceleration option in Chrome.
Sound quality is muffled or hollow	Check that lobes have been aimed to the desired area. Use equalizer to adjust frequency response on a single channel or on the automix channel. See the equalizer applications for the appropriate use.
Microphone does not show up in device discovery	Ensure the devices are powered Ensure PC and equipment are on the same network and set to the same subnet Turn off other network interfaces not used to connect to the device (including WiFi) Check that DHCP server is functioning (if applicable) Reset the device if necessary
Audio is not present or is quiet/distorted	Check cables Verify that channels are not muted Make sure channels are aimed in the right direction Check that fader levels are not set too low If using automixing, check the settings to ensure channels are gating on/off properly
No lights	Check if brightness is disabled or if any Light Bar settings are turned off.
Auto-positioning identifies incorrect location	If talker is in a corner or very close to a wall, acoustic reflections may interfere with localization accuracy. Try automatic positioning again, and if the issue persists, manual positioning may be necessary.
Microphone does not power on	The network switch must supply Power over Ethernet. Otherwise, a PoE injector must be used Check network cables and connections

Important Product Information

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

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

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

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

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

Authorized under the verification provision of FCC Part 15B.

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

Information to the user

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

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

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

Authorized European representative:
Shure Europe GmbH
Headquarters Europe, Middle East & Africa
Department: EMEA Approval
Jakob-Dieffenbacher-Str. 12
75031 Eppingen, Germany
Phone: 49-7262-92 49 0
Fax: 49-7262-92 49 11 4
Email: info@shure.de

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

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

Specifications

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

Beam Width

Adjustable	Narrow	35 degrees
	Medium	45 degrees
	Wide	55 degrees

Connector Type

RJ45

Power Requirements

Power over Ethernet (PoE), Class 0

Power Consumption

9W, maximum

Weight

4.26 kg (9.4 lbs)

Product Dimensions

MXA910xx	603.8 x 603.8 mm (23.77 x 23.77 in.)
MXA910xx-60CM	593.8 x 593.8 mm (23.38 x 23.38 in.)
A910-25MM	619.7 x 619.7 mm (24.4 x 24.4 in.)

[2]

control application

HTML5 Browser-based

Plenum Rating

Requires Fyrewrap® fire protective wrap system (Included)

UL 2043 (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

Frequency Response

180 to 17,000 Hz

Dante Digital Output

Channel Count	9 total channels (8 independent transmit channels, 1 IntelliMix® Automatic mixing transmit channel)
Sampling Rate	48 kHz
Bit Depth	24

Sensitivity

at 1 kHz

0.75 dBFS/Pa

[2] Note: the adapter accessory converts the 600 mm model to fit into a 625 x 625 mm ceiling grid.

Maximum SPL

Relative to 0 dBFS overload

93.25 dB SPL

Signal-To-Noise Ratio

Ref. 94 dB SPL at 1 kHz

83 dB A-weighted

Latency

Not including Dante latency

6 ms

Self Noise

11 dB SPL-A

Dynamic Range

82.25 dB

Built-in Digital Signal Processing

Per Channel	Equalizer (4-band Parametric), Mute, Gain (140 dB range)
System	IntelliMix® Automatic mixing, Echo Reduction

Intelligibility Scale

Equivalent acoustic performance, compared to a cardioid gooseneck microphone (environment dependent)

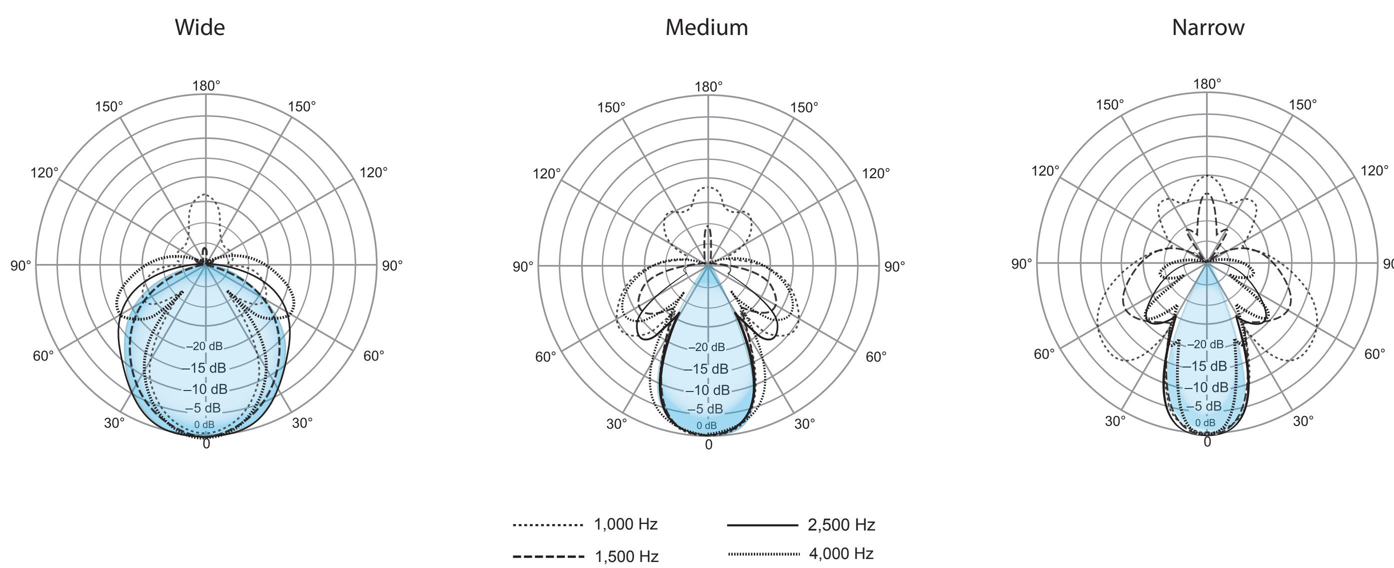
Cardioid distance multiplied by 1.6

[3]

Networking

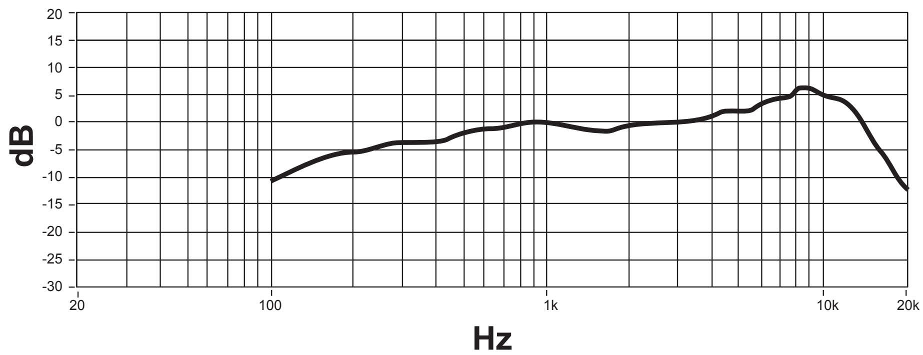
Cable Requirements

Cat 5e or higher (shielded cable recommended)



Polar Pattern

[3] [1] 1 Pa=94 dB SPL



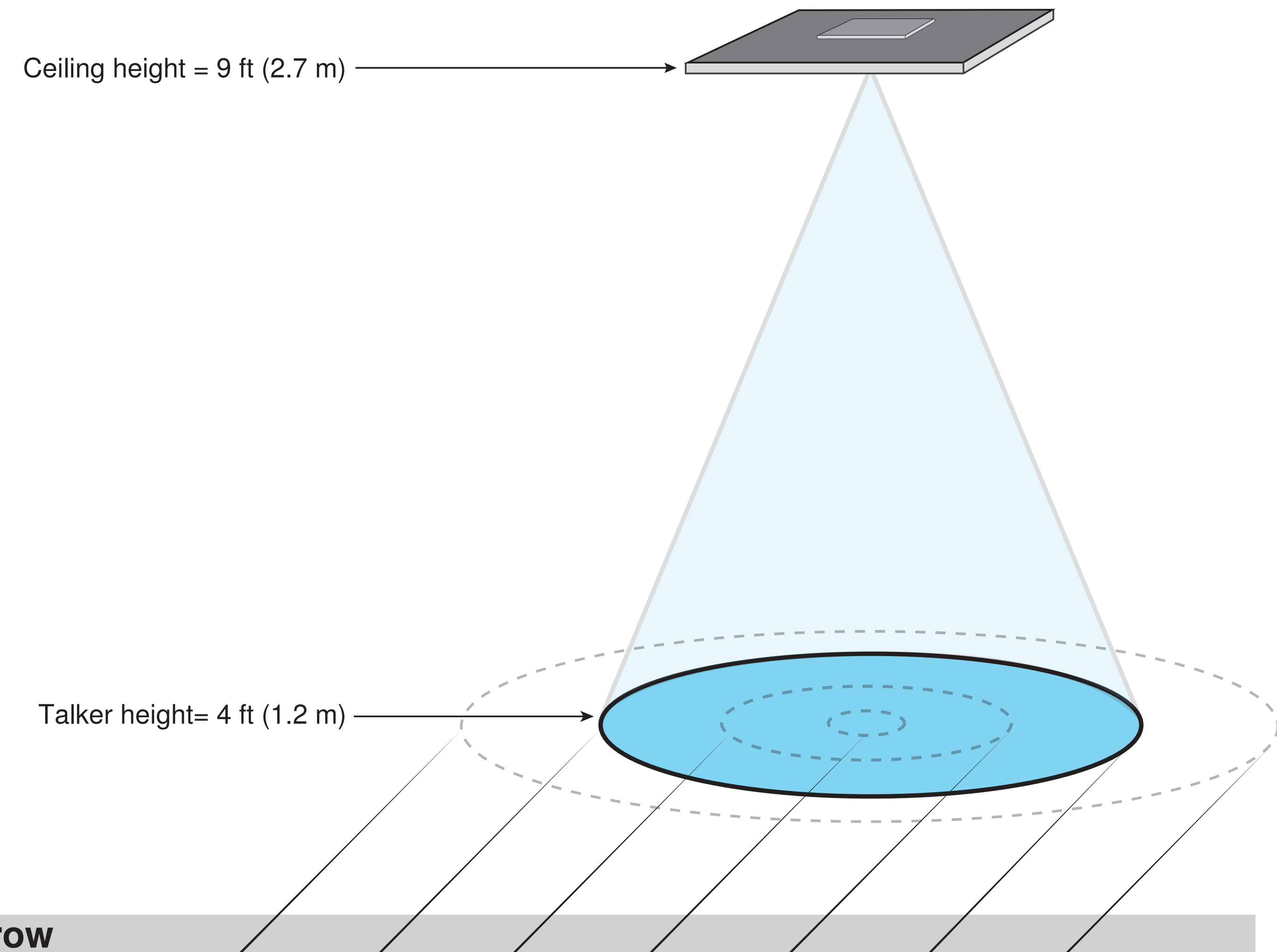
Frequency Response

Lobe Sensitivity

The edge of the blue coverage area for each channel in the web application 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.

Measured at 1 kHz, on-axis



Narrow							
Centimeters	55	44	30	0	30	44	55
Inches	22	17	12	0	12	17	22
dB	-12	**-6**	-3	0	-3	**-6**	-12
Medium							
Centimeters	69	47	38	0	38	47	69
Inches	27	18	15	0	15	18	27
dB	-12	**-6**	-3	0	-3	**-6**	-12
Wide							
Centimeters	92	62	44	0	44	62	92
Inches	36	24	17	0	17	24	36
dB	-12	**-6**	-3	0	-3	**-6**	-12

IP Ports and Protocols

Port	TCP/UDP	Protocol	Description	Factory Default
21	tcp	FTP	Required for firmware updates (otherwise closed)	Closed
22	tcp	SSH	Not supported	Closed
23	tcp	Telnet	Standard console interface	Closed
68	udp	DHCP	Dynamic Host Configuration Protocol	Open
80*	tcp	HTTP	Required to launch embedded web server	Open
427	tcp/udp	SLP†	Required for inter-device communication	Open
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162	tcp	SNMP	Not supported	Closed
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5568	udp	SDT†	Required for inter-device communication	Open
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8180*	tcp	HTML	Required for web application	Open
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64000	tcp	Telnet	Required for Shure firmware update	Open

Port	TCP/UDP	Protocol	Description
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4321, 14336-14600	udp	Dante	Dante audio
[4440, 4444, 4455]*	udp	Dante	Dante audio routing
5353	udp	mDNS†	Used by Dante
[8700-8706, 8800]*	udp	Dante	Dante Control and Monitoring
8751	udp	Dante	Dante Controller
16000-65536	udp	Dante	Used by Dante

[4] [5]

Accessories

Replacement Parts

Rubber pad set	95A28365
Cable management clip	95A29877
Junction Box Accessory	A910-JB
25 mm expander (fits onto 60 cm model, for 62.5 x 62.5 cm installation)	A910-25MM Adapter

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

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