



# P300

## IntelliMix<sup>®</sup> Audio Conferencing Processor

Complete user guide for the P300. Includes system diagrams, specifications, installation instructions, command strings, and more.  
Version: 7.2 (2021-B)

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# P300 IntelliMix<sup>®</sup> Audio Conferencing Processor

## Getting Started

To control the P300, use Shure Designer software to adjust settings and route audio between Shure devices. After completing this basic setup process, you should be able to:

- Control the P300 in Designer
- Apply digital signal processing (DSP)
- Route audio to and from devices

Before you get started, you'll need:

- Cat5e (or better) Ethernet cable
- Network switch that provides Power over Ethernet Plus (PoE+)
- Shure Designer software installed on a computer. Download at [www.shure.com](http://www.shure.com).

## Step 1: Connect to a Network and Discover in Designer

1. Connect the P300 to a PoE+ port on the network switch using Cat5e (or better) cable.
2. Connect your computer running Designer to the network switch.
3. Open Designer, and check that you're connected to the correct network in Settings.
4. Click Online devices. A list of online devices appears.
5. To identify devices, click the product icon to flash the lights on a device.



## Step 2: Route Audio and Apply DSP

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

For this example, we'll connect an MXA910 ceiling array microphone. The process is the same for any networked Shure microphone in Designer.

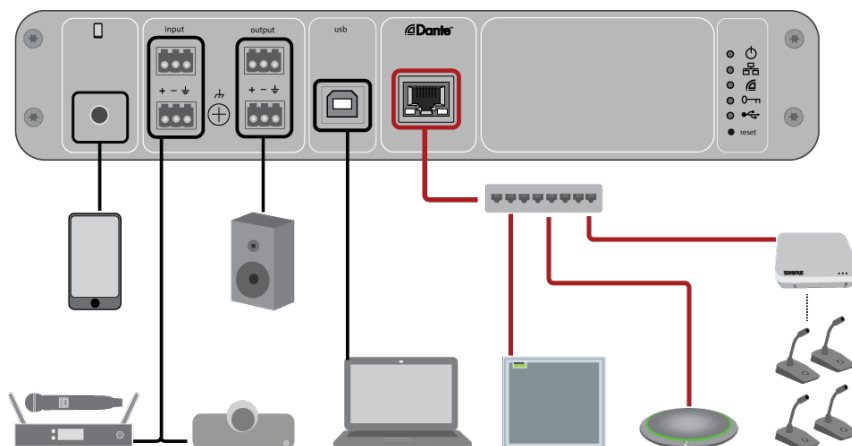
1. From My projects, select New to create a new project. Click New to add a location to your project.
2. Select Live mode. Any online devices appear in the list. Drag and drop the P300 and MXA910 to add them to your location.
3. Select Optimize.
4. Check the audio routes and settings to make sure they fit your needs. You might need to:
  - Delete unnecessary routes.
  - Fine-tune DSP blocks as needed.
  - Test and adjust microphone coverage.

You can also route audio manually in Designer outside of the Optimize workflow, or use Dante Controller.

## Step 3: Adjust DSP and Route Signals

The final steps vary depending on what other devices you connect to the P300. Regardless of what devices you connect, the final steps are to check DSP settings and route signals in the matrix mixer.

1. Connect any other analog, USB, or mobile audio devices to the P300.



2. Check the signal routes in the Matrix mixer tab. Designer establishes many common routes automatically.
3. Adjust levels as needed in the Inputs and Outputs tabs.
4. Listen to your signal and adjust DSP settings as needed. Optimize applies DSP automatically, so you may not need to make any adjustments.
5. Check and adjust the acoustic echo cancellation (AEC) reference source in Schematic > AEC. Optimize sets one automatically, but you may need to adjust it depending on your setup.

Use the channel that carries audio to loudspeakers as the AEC reference. If your room has an analog loudspeaker system or built-in display speakers, Analog -- To Speaker is the most common source. If you're using Dante loudspeakers, one of Dante outputs is the reference.

## Get More Information

Learn more about the P300 in the Designer Help section or in the complete user guide at [pubs.shure.com/guide/P300](https://pubs.shure.com/guide/P300).

## Overview

### General Description

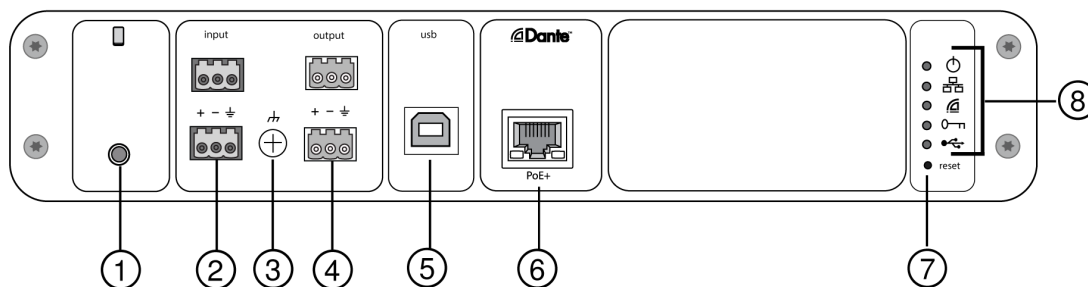
The P300 IntelliMix Audio Conferencing Processor offers IntelliMix DSP algorithms optimized for audio/video conferencing applications, featuring 8 channels of acoustic echo cancellation, noise reduction and automatic gain control to ensure a high-quality audio experience.

The P300 provides Dante (10 in/8 out), analog (2 block in/2 block out), USB (1 in/out) and mobile (3.5 mm) connectivity options that makes connecting to room systems and collaborating with laptops and mobile devices easier than ever.

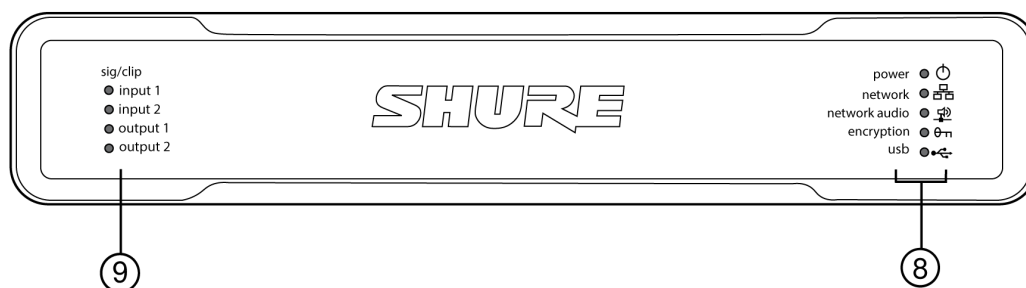
## Features

- Connects 10 Dante™ audio inputs, 2 analog inputs, USB, and a mobile device to an A/V conferencing system or a PC-based videoconferencing application
- Includes IntelliMix DSP algorithms to enhance audio quality in A/V conferences: 8 channels of AEC (acoustic echo cancellation), noise reduction, and automatic gain control, combined with automatic mixing, matrix mixing, delay, compressor and parametric equalization
- Flexible signal routing and connectivity: analog audio (2 block in/2 block out) to connect to room A/V conferencing system; USB (1 in/out) to connect to laptop or room PC; one 3.5 mm TRRS jack to connect to a mobile device to enable an additional participant to join
- Power over Ethernet Plus (PoE+) eliminates the need for an outboard power supply
- Compact form factor is easy to mount without equipment rack

## P300 Hardware



Rear panel



Front panel

### ① Mobile Input

TRRS mobile input connects to a mobile device. Supports bidirectional audio with a TRRS cable, or sends audio into the P300 with a TRS cable.

*Note: See the cable requirements for additional information.*

#### Pin Assignments:

Tip	Audio Input (Left)
Ring 1	Audio Input (Right )
Ring 2	Ground
Sleeve	Audio Output (To Phone)

*Note: Left and Right audio signals are summed to a mono signal.*

### ② Audio Inputs (Block Connector)

Balanced audio input connects to an analog audio device. Set the analog input level to match the output level of the analog device.

#### Input sensitivity:

Line (+4 dBu)

Aux (-10 dBV)

#### Block Pin Assignments:

+	Audio +
–	Audio -
⊥	Audio ground

### ③ Chassis Ground Screw

Provides an optional connection for microphone shield wire to chassis ground.

### ④ Audio Outputs (Block Connector)

Balanced audio output connects to an analog device. Set the output level to match the input sensitivity of the analog device (Line, Aux, or Mic level).

+	Audio +
–	Audio -
⊥	Audio ground

### ⑤ USB Port

Connects to a computer to send and receive audio signals. Use the matrix mixer to sum any combination of signals from the P300 into a single mono channel and send through the USB output.

### ⑥ Dante Network Port

Connects to a network switch to connect Dante audio, Power over Ethernet (PoE), and data from the control software.

### ⑦ Reset Button

Resets the device settings back to the factory default.

## ⑧ LED Indicators

- ⓘ Power
- 📶 Network
- 🔊 Network Audio
- 🔒 Encryption
- 🔌 USB

**Power:** Power over Ethernet Plus (PoE+) present

*Note: Use a PoE+ injector if your network switch does not supply PoE+.*

**Network:** Ethernet connection active

**Network Audio:** Dante audio present on the network

### Network Audio LED Behavior

LED Status	Activity
Off	No active signal
Green	Device is operating successfully
Red	Error has occurred. See event log for details.

**Encryption:**

LED Status	Activity
Off	Audio not encrypted
Green	Encryption enabled
Red	Encryption error. Possible causes: Encryption is enabled on one device and not on another Passphrase mismatch

**USB Audio**

LED State	Status
Off	No USB device connected
Green	USB device operating successfully
Red (flashing)	Problem detected with connected USB audio device

**Note:** Error details are available in the Event Log.

## ⑨ Level Indicators (Signal/Clip)

Tri-color LEDs indicate the audio signal level for the analog channels. Adjust output levels to avoid clipping.



## Analog Input/Output

LED State	Audio Signal Level
Off	less than -60 dBFS
Green	-59 dBFS to -24 dBFS
Yellow	-23 dBFS to -1 dBFS
Red	0 dBFS or more

**Note:** The input and output LEDs stay off when metering is set to Post-gain and the channel is muted.

## Power Over Ethernet Plus (PoE+)

This device requires PoE Plus to operate. It is compatible with both **Class 4** PoE+ sources.

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

- A network switch that provides PoE+
- A PoE+ injector device (must be a Gigabit device)

## Cable Requirements

Always use Cat5E cable or higher.

## Reset

The reset button is located inside a small hole in the rear panel. Use a paperclip or other small tool to press the button.

There are 2 hardware reset functions:

### Network reset (press button for 4-8 seconds)

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

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

Resets all network and Designer settings to the factory defaults.

## Software Reset Options

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

**Restore Factory Defaults (Settings > Factory Reset):** Restores all network and Designer settings to the factory defaults. This is the same as performing a full factory reset using the reset button on the device.

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



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

# Installation

## Installation and Rack Mounting

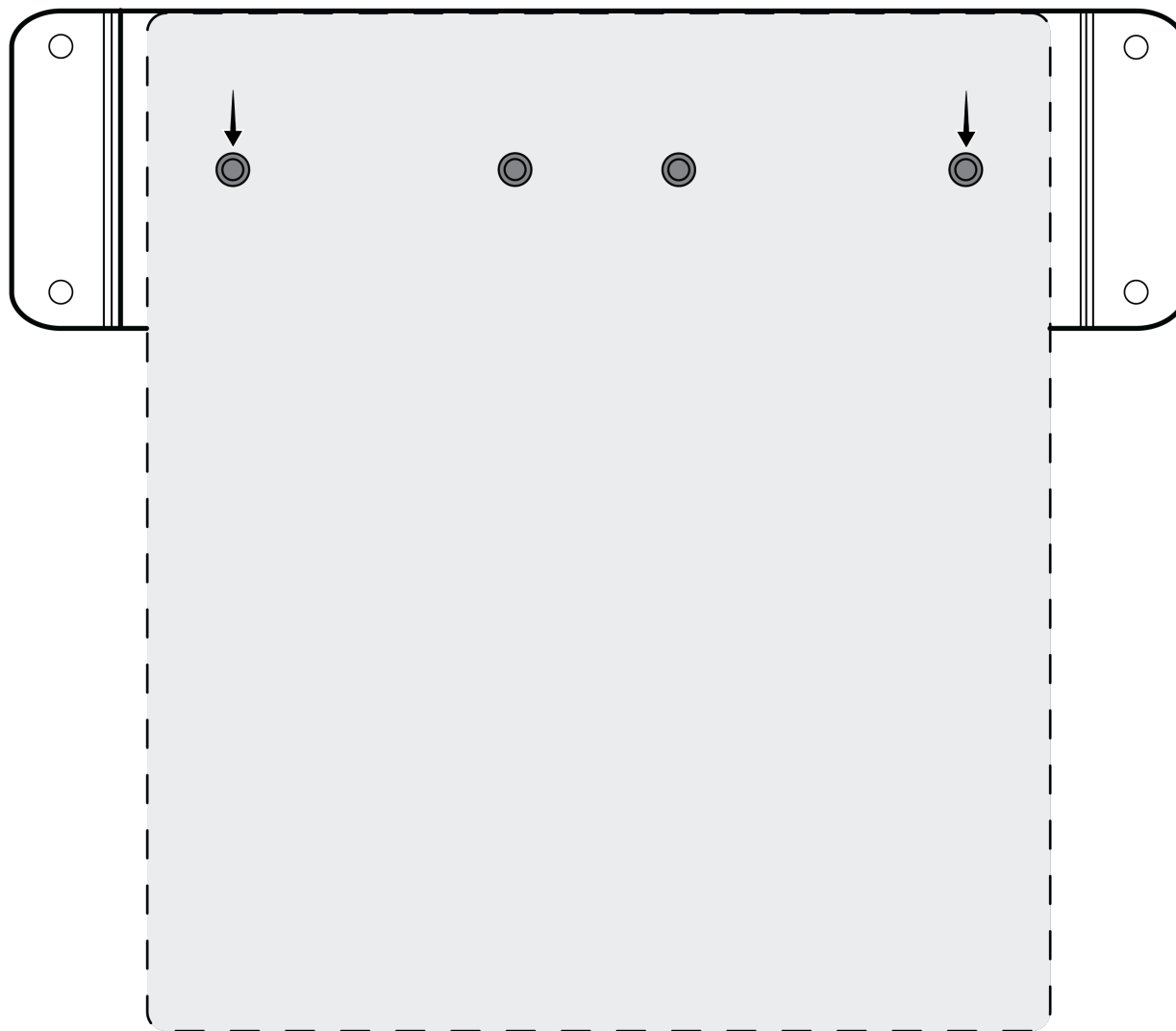
Two mounting solutions are available for installing the P300:

**CRT1 19" Rack Tray (optional accessory):** Supports up to 2 devices (two P300s or one P300 and one ANI4IN, ANI4OUT, ANI22, or ANIUSB); mountable in a rack or under a table

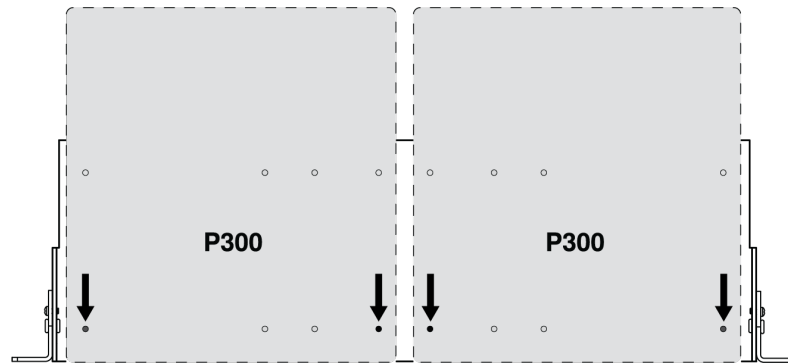
**Single-unit Mounting Tray (included accessory):** Supports a single device for mounting under a table

## Securing the Devices

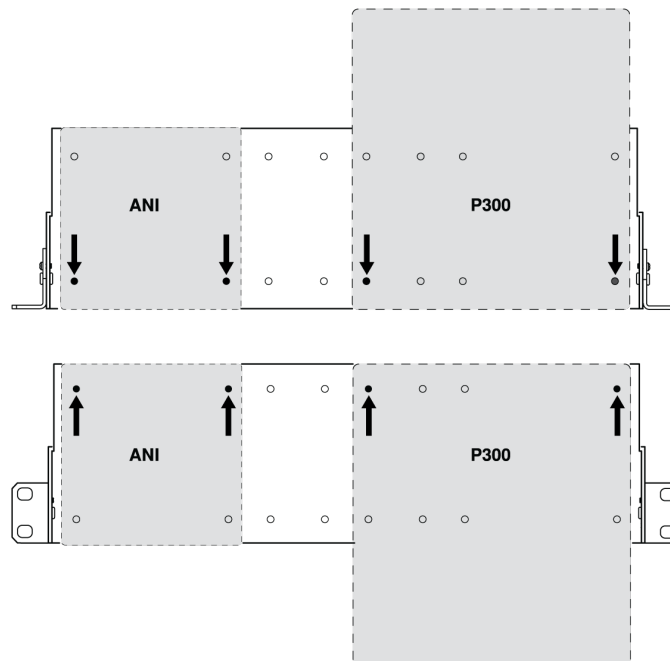
Use the included screws from the mounting hardware kit to secure each P300 or Audio Network Interface (ANI). Devices can be mounted to face either direction. Insert the screws from the bottom in the appropriate holes, according to the following diagrams:



*Align the holes as shown for securing a single device in the single-unit mounting tray*



Align the holes as shown for securing up to two devices in the 19" rack tray.

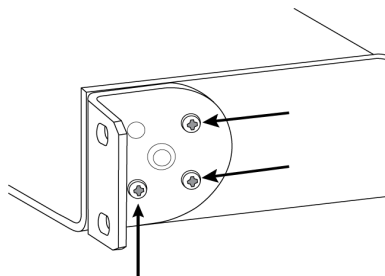


## Rack Ear Configuration (CRT1)

The adjustable rack ears support mounting in a standard equipment rack or underneath a table.

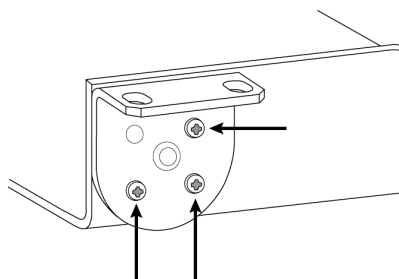
### Standard 19" Rack Mount

1. Align the ears with the mounting holes pointed forward.
2. Install the two screws that hold the ear to the tray as shown.



## Under-table Mounting

1. Align the ears with the mounting holes pointed upward.
2. Install the two screws that hold the ear to the tray as shown.




## Installing Underneath a Table



1. Hold the tray in the desired location under a table
2. Use a pencil to mark the location of the mounting holes on the table.
3. Drill 4 holes for the screws. The diameter of the holes in the tray are 7.1 mm.
4. Install the components into the tray
5. Install with 4 screws to secure the tray underneath the table




## Use Cases

### Overview of Shure Conferencing Devices

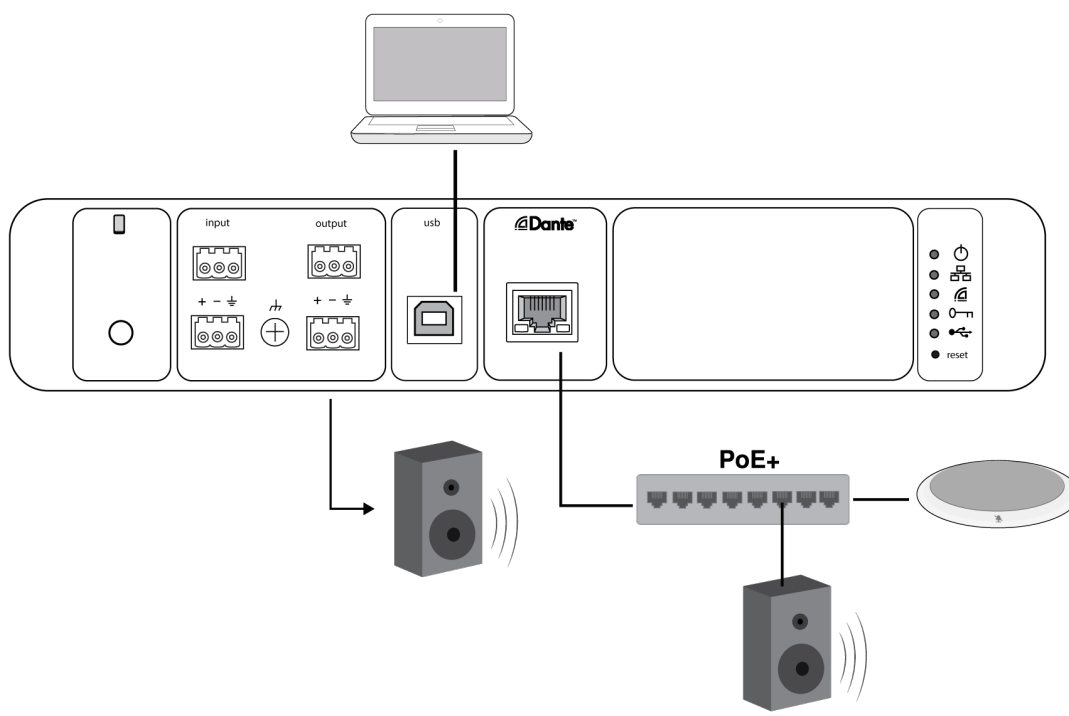
Shure offers a range of connectivity options for conferencing. MXA microphones, audio processors, and network interfaces all use Dante to send audio over standard IT networks. You can use Shure's free [Designer software](#) to control most Shure devices and route audio between them.

Device	Purpose	Physical Connections	Dante I/Os
 <b>MXA910</b>	Ceiling array microphone with IntelliMix DSP	1 PoE port	8 individual channel outputs or 1 automix channel output with IntelliMix DSP 1 AEC reference input

Device	Purpose	Physical Connections	Dante I/Os
 <b>MXA710</b>	Linear array microphone with IntelliMix DSP	1 PoE port	<b>2 Foot:</b> 4 individual channel outputs or 1 automix channel output with IntelliMix DSP 1 AEC reference input  <b>4 Foot:</b> 8 individual channel outputs or 1 automix channel output with IntelliMix DSP 1 AEC reference input
 <b>MXA310</b>	Table array microphone	1 PoE port	4 individual channel outputs or 1 automix channel output
 <b>P300</b>	Audio processor with IntelliMix DSP and matrix mixer	1 USB in/out 2 analog block in 2 analog block out 1 mobile TRRS port (3.5 mm) 1 PoE+ port	8 Dante inputs with IntelliMix DSP 2 auxiliary Dante inputs 8 Dante outputs
 <b>IntelliMix Room</b>	Audio processing software with IntelliMix DSP and matrix mixer	Varies depending on device	8 or 16 Dante inputs with IntelliMix DSP 8 auxiliary Dante inputs 8 Dante outputs 1 virtual audio input and output 1 PC input and output
 <b>ANIUSB-MATRIX</b>	Matrix mixer with USB and analog input/output	1 USB in/out 1 analog block in 1 analog block out 1 PoE port	4 Dante inputs 2 Dante outputs
 <b>ANI4IN (block or XLR connectors)</b>	Converts analog signals to Dante signals	4 analog in 1 PoE port	4 Dante inputs
 <b>ANI4OUT (block or XLR connectors)</b>	Converts Dante signals to analog signals	4 analog out 1 PoE port	4 Dante outputs

Device	Purpose	Physical Connections	Dante I/Os
 <b>ANI22 (block or XLR connectors)</b>	Converts 2 analog signals to Dante signals Converts 2 Dante signals to analog signals	2 analog in 2 analog out 1 PoE port	2 Dante inputs 2 Dante outputs
 <b>MXN5-C</b>	Networked ceiling loudspeaker powered by PoE	1 PoE port	2 Dante inputs 1 Dante output
 <b>MXA Network Mute Button</b>	PoE-powered network mute button for Shure devices	1 PoE port 1 power cable connector for base	n/a

## P300 and Computer with Soft Codec



1. Connect the computer to the USB port on the P300.
2. Connect a powered loudspeaker or amplifier to the P300 analog output 2. In the matrix mixer, this is labeled Analog - To Speaker. You can also use Dante PoE loudspeakers connected to the network switch.
3. In Designer, go to Matrix mixer to make connections between devices. Some connections are established automatically.

Required Matrix Mixer Connections:

Input	Output
Automix	USB output

Input	Output
USB input	Analog - To Speaker <b>or</b> Dante output

- In Schematic, right-click any AEC block and choose an AEC reference source. The reference source is Analog - To Speaker for analog loudspeakers, or one of the Dante outputs for Dante loudspeakers.
- Adjust input and output levels using Designer and perform a sound check. Refer to the Help section for additional information.

## Connecting a USB Device

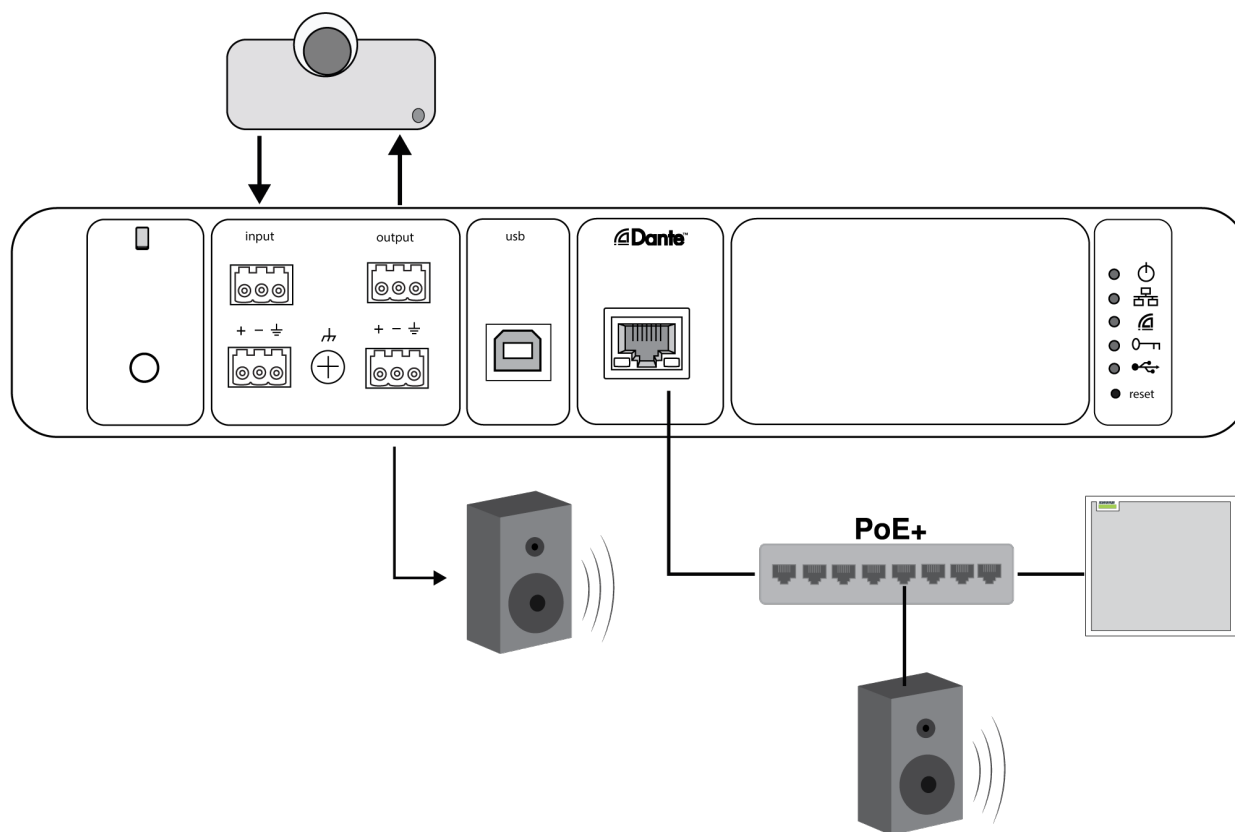
The USB port connects the host computer to the entire room audio system, including microphones and loudspeakers.

When the P300 is connected for the first time, the computer recognizes it as a USB audio device. You may need to select it as the input/output (recording/playback) device to pass audio. Assign the P300 as the default device to ensure it passes audio every time it is plugged in. Refer to the manual for your computer to configure the audio settings.

## Adapter Compatibility

This device is compatible with USB-B to USB-C adapters. Using an adapter is only recommended for desktop and laptop computers, as many mobile devices do not support bi-directional audio through USB or lightning ports.

## Hardware Codec System



- Connect the hardware codec audio output to the P300 analog input 1. In the matrix mixer, this is labeled Analog - From Codec.
- Connect the hardware codec audio input to the P300 analog output 1. In the matrix mixer, this is labeled Analog - To Codec.

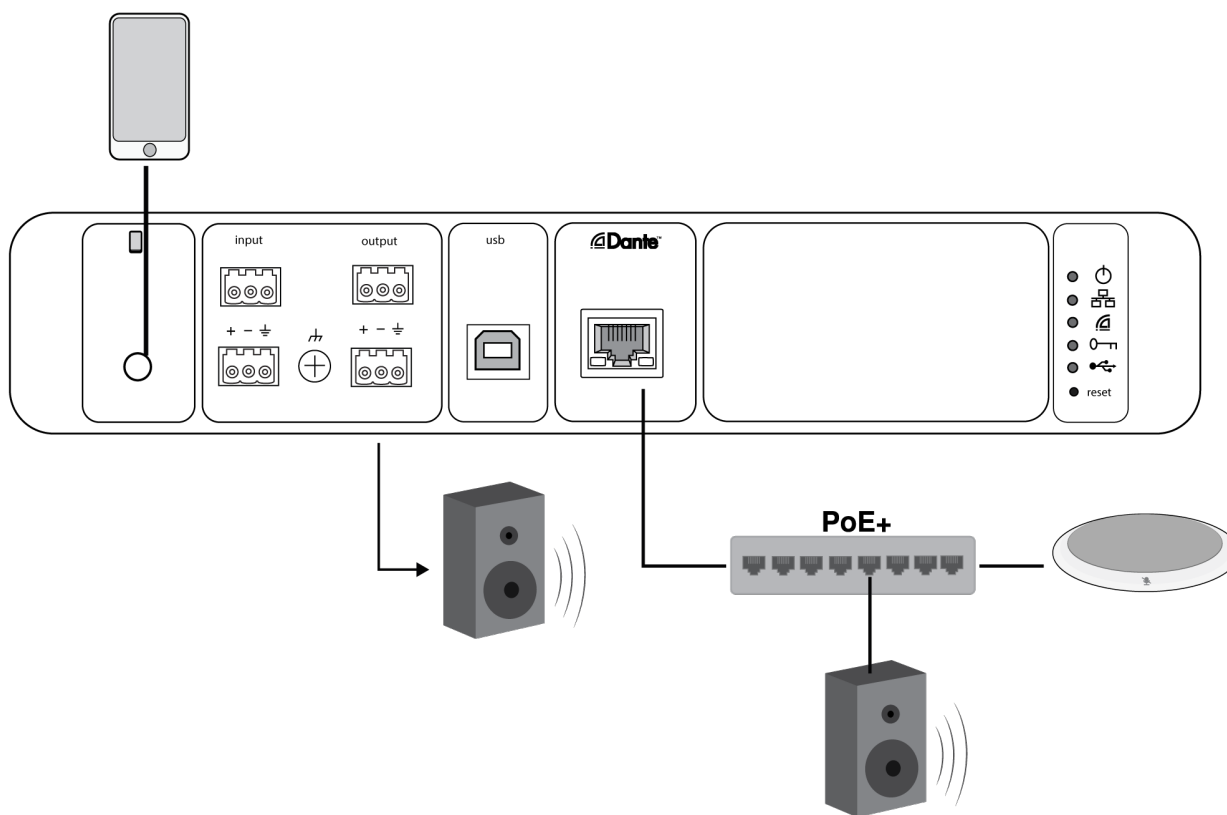
3. Connect a powered loudspeaker or amplifier to the P300 analog output 2. In the matrix mixer, this is labeled Analog - To Speaker. You can also use Dante PoE loudspeakers connected to the network switch.
4. In Designer, go to Matrix mixer to make connections between devices. Some connections are established automatically.

Required Matrix Mixer Connections:

Input	Output
<b>Automix</b>	Analog - To codec
<b>Analog - From Codec</b>	Analog - To Speaker <b>or</b> Dante output

5. In Schematic, right-click any AEC block and choose an AEC reference source. The reference source is Analog - To Speaker for analog loudspeakers, or one of the Dante outputs for Dante loudspeakers.
6. Adjust input and output levels using Designer and perform a sound check. Refer to the Help section for additional information.

## Mobile Phone System



In this example, when the phone is plugged in, the built-in microphone and speaker are disabled -- the phone simply carries the call. The MXA310 microphone captures near-end audio, and the loudspeaker delivers audio from the far end of the call.

1. Connect the phone to the P300 with a 1/8-inch **TRRS** cable.
2. Connect a powered loudspeaker or amplifier to the P300 analog output 2. In the matrix mixer, this is labeled Analog - To Speaker. You can also use Dante PoE loudspeakers connected to the network switch.
3. In Designer, go to Matrix mixer to make connections between devices. Some connections are established automatically.



## Required Matrix Mixer Connections:

Input	Output
Automix	Mobile output
Mobile input	Analog - To Speaker <b>or</b> Dante output

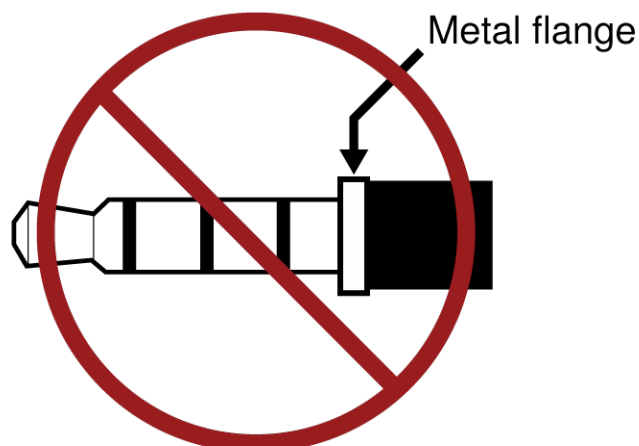
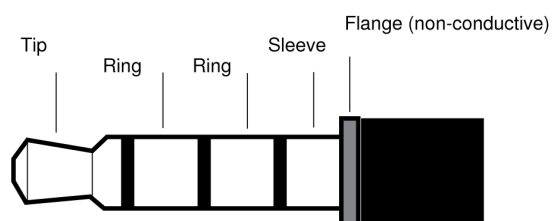
- In Schematic, right-click any AEC block and choose an AEC reference source. The reference source is Analog - To Speaker for analog loudspeakers, or one of the Dante outputs for Dante loudspeakers.
- Adjust input and output levels using Designer and perform a sound check. Refer to the Help section for additional information.

## Mobile Connection Cable Requirements

A 1/8-inch TRRS cable is required to connect a phone to the P300. Avoid using cables with a metal flange, as it may create an electrical connection to the exterior of the phone and interrupt the signal.

To ensure proper operation, only use:

- Apple-approved cables
- Cables with a plastic or non-conductive flange



**Note:** If necessary, a TRS (tip/ring/sleeve) cable may be used to plug a stereo device into the P300, but the device will only be able to send audio to the P300. The Enable auto-mute feature on the mobile input channel must be turned off in this case.

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## Controlling 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 Settings.
3. Click Online devices. A list of online devices appears.
4. To identify devices, click the product icon to flash the lights on a device. Select your device in the list and click Configure to control device settings.

Learn more at [shure.com/designer](https://shure.com/designer).

You can also access basic device settings using Shure Web Device Discovery. Full control is available in Designer.

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## How to Update Firmware Using Designer

Applies to version 4.2 and newer.

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

To update:

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

## Firmware Versioning

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

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

## Differences Between 3.x and 4.x Firmware

### 6 Dante Outputs Added

- With 4.x firmware, the P300 has 10 Dante inputs and 8 Dante outputs for increased flexibility

### Automixer Direct Out Tap Points Added

- Choose where the microphone signals going to the matrix mixer come from: pre- or post-processing, and pre- or post-automixer gating
- These options add flexibility for reinforcement, voice lift, speaker zoning, and other applications.

### Support for Dante Domain Manager

- You can now add the P300 to domains in Dante Domain Manager software.

### Audio Encryption Improvements

- Firmware 4.x and newer has improved audio encryption, so make sure to update all Shure devices using encryption to the same major firmware version.

### Web Applications Removed

- With firmware 4.x and newer, Shure Designer software is the only way to control the P300.
- You can access basic settings for the device by opening it from the Shure Web Device Discovery tool, but you must use Designer for full control.
- Download Designer at [shure.com/designer](https://shure.com/designer).

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## Using Designer's Optimize Workflow

Designer's Optimize workflow speeds up the process of connecting systems with at least 1 microphone and 1 audio processor. Optimize also creates mute control routes in locations with MXA network mute buttons. When you select Optimize in a location, Designer does the following:

- Creates audio routes and mute control routes
- Adjusts audio settings
- Turns on mute synchronization
- Enables LED logic control for applicable devices

The settings are optimized for your particular combination of devices. You can customize settings further, but the Optimize workflow gives you a good starting point.

After optimizing a location, you should check and adjust settings to fit your needs. These steps may include:

- Deleting unnecessary routes.
- Checking levels and adjusting gain.
- Verifying that AEC reference signals are correctly routed.
- Fine-tuning DSP blocks as needed.

### Compatible devices:

- MXA910
- MXA710
- MXA310
- P300
- IntelliMix Room
- ANIUSB-MATRIX
- MXN5-C
- MXA Network Mute Button

### To use the Optimize workflow:

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

If you remove or add devices, select Optimize again.

---

# Schematic View

The schematic view in Designer provides an overview of the entire audio signal chain, with the ability to adjust settings and monitor signals.

## Adjusting Settings

Right-click an input, output, or processing block to access the following options:

Per Channel

### Copy / paste

Copy and paste settings between items. For example, set the equalizer curve on the USB output, and then use the same setting for the analog output. Or, copy the gain and mute status from one input channel to several others.

### Mute / unmute

Mutes or activates the channel

### Enable / disable

Turns processing on or off (does not apply to matrix mixer or automixer)

### Edit

Opens the dialog to adjust parameters

Global (right-click in blank area)

### Mute all inputs

Mutes all input channels

### Mute all outputs

Mutes all output channels

### Unmute all inputs

Unmutes all input channels

### Unmute all outputs

Unmutes all output channels

### Close all dialogs

Clears all open dialogs from the workspace

## Customizing the Workspace

Create a custom environment to monitor and control a set of inputs, outputs, and processing blocks from a single screen.

There are two ways to break out dialogs:

- Right click > edit
- Double-click the input, output, or processing block.

Open as many dialogs as you need to keep important controls available.

## Metering and Audio Routing

A meter appears underneath each input and output to indicate signal levels (dBFS).

The lines connecting inputs and outputs to the matrix mixer appear colored when connections are established. When a signal is not routed, the line appears gray. Use these tools to troubleshoot audio signals and verify connections and levels.

## Mute and Fader Groups

<b>Mute Groups</b>	Check the Mute group box to add the channel to a group. Muting any channel within the Mute group mutes all channels in the group.
<b>Fader Groups</b>	Check the Fader group box to add the channel to a group. All faders within the group are linked, and move together when a single fader is adjusted.

## Custom Presets

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

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

Open the presets menu to reveal preset options:

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

## Parametric Equalizer (PEQ)

Maximize audio quality by adjusting the frequency response with the parametric equalizer (PEQ). Use the input equalizers to make adjustments to specific channels, while using the output equalizers to adjust frequency response of all signals that are summed through a given output.

Common equalizer applications:

- Improve speech intelligibility

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

**Note:** If you're connecting a microphone that has a built-in equalizer (such as an MXA310), disable any EQ on the mic and use the P300's EQ instead.

## 2-Band Equalizers

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

Each band has a selectable filter:

**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)

## 4-Band Equalizers

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

**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).

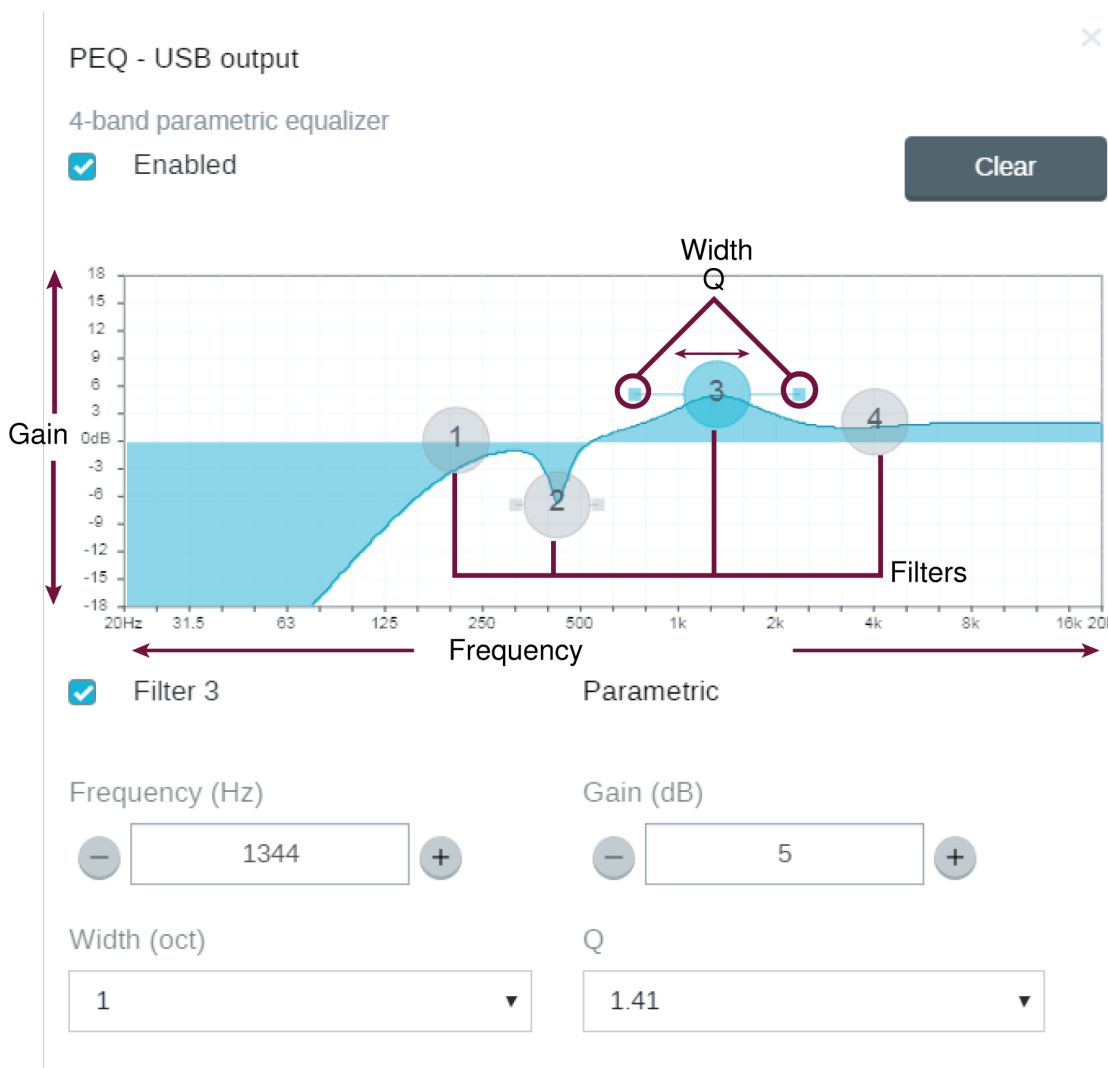
### Q

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.



## 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: 1. Set a narrow Q value.

EQ Application	Suggested Settings
	<ol style="list-style-type: none"> <li>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.</li> <li>3. Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound.</li> </ol>
<p><b>Reduce hollow, resonant room sound</b></p>	<p>Identify the specific frequency range that "excites" the room:</p> <ol style="list-style-type: none"> <li>1. Set a narrow Q value.</li> <li>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.</li> <li>3. Reduce the gain at the identified frequency (start between -3 and -6 dB) to minimize the unwanted room sound.</li> </ol>



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

## Automatic Gain Control (AGC)

Automatic gain control 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.

Automatic gain control is post-fader, and adjusts the channel level after the input level has been adjusted. Enable it 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.

### Target Level (dBFS)

Represents the level that you want the gain to reach. This level is different from adjusting the input fader according to peak levels to avoid clipping. Suggested starting points:

- P300: -32 dBFS
- MXA910: -37 dBFS
- IntelliMix Room: -32 dBFS

### 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 to monitor the amount of gain added or subtracted from the signal. If this meter is always reaching the maximum boost or cut level, adjust the input fader so the signal is closer to the target level.



## 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 is a DSP algorithm which identifies and eliminates echoes to deliver clear, uninterrupted speech. The P300 features 8 channels of acoustic echo cancellation, with independent processing on each channel for maximum effectiveness. Use the following tips when setting up a system:

- Optimize the acoustic environment, when possible: avoid pointing speakers directly at microphones, reduce speaker volume, and position speakers farther from microphones.
- If connecting to a Shure MXA910, disable the Echo Reduction on the microphone.

## Training the Acoustic Echo Cancellation

Training is the process where the AEC optimizes processing based on the acoustic environment. It only trains when far-end audio is present and near-end talkers are quiet. The AEC is constantly adapting, so if the acoustic environment changes, the AEC automatically adjusts.

## Adjusting Settings

To adjust acoustic echo cancellation settings, open the AEC menu in the schematic view or inputs tab.

<b>Reference Meter</b>	Use the reference meter to visually verify the reference signal is present.
<b>ERLE</b>	Echo reduction loss enhancement displays the dB level of signal reduction (the amount of echo being removed). If connected properly, the ERLE meter activity generally corresponds to the reference meter.
<b>Reference</b>	Select the channel that carries audio to the loudspeakers as the reference. Analog - To Speaker is the most commonly used channel, for configurations with an analog loudspeaker system or a display with a built-in speaker.  <b>Note:</b> Selecting a reference on any channel applies that same reference to all channels with AEC.
<b>Non-linear Processing</b>	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. <b>Low:</b> Use in rooms with controlled acoustics and minimal echoes. This setting provides the most natural sound. <b>Medium:</b> Use in typical rooms as a starting point. If echo artifacts appear, try using the high setting. <b>High:</b> Use to provide the strongest echo reduction in rooms with bad acoustics, or in situations where the echo path frequently changes.

## Noise Reduction

Noise reduction significantly reduces the amount of noise in the signal caused by projectors, HVAC systems, or other environmental noise. It is a dynamic processor, which calculates the noise floor in the room and removes noise throughout the entire spectrum with maximum transparency.

## Settings

The noise reduction setting (low, medium, or high) represents the amount of reduction in dB. Use the lowest possible setting that effectively lowers noise in the room.

## 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.

## Delay

Use the delay feature on the analog and USB outputs to synchronize audio and video. When a video system introduces latency (where you hear someone speak, and their mouth moves later), simply add delay to the analog outputs to align with the video. Delay can also be used in larger rooms to align the arrival time or phase between multiple speakers.

The 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 it is closer to full synchronization, use smaller intervals to fine-tune.

The USB output channel features delay to ensure the near-end camera and near-end audio are synchronized.

## Matrix Mixer

The matrix mixer routes audio signals between inputs and outputs for simple and flexible routing:

- Send a single input channel to multiple outputs
- Send multiple input channels to a single output

## Crosspoint Gain

Crosspoint gain adjusts the gain between a specific input and output, to create separate submixes without changing input or output fader settings. Select the dB value at any crosspoint to open the gain adjustment panel.

**Gain staging:** Input fader > crosspoint gain > output fader

## Routing Channels

Connect inputs and outputs by selecting the box where they intersect.

## P300 Default Settings

The default configuration enables calling to multiple far ends with near-end Shure microphones. Connections are established for operating hardware codecs, software codecs, and mobile phones simultaneously.

Input/Source Channel	Output/ Destination Channels
<b>Automix (summed Dante input channels)</b>	Analog - To Codec (Analog output 1) USB output

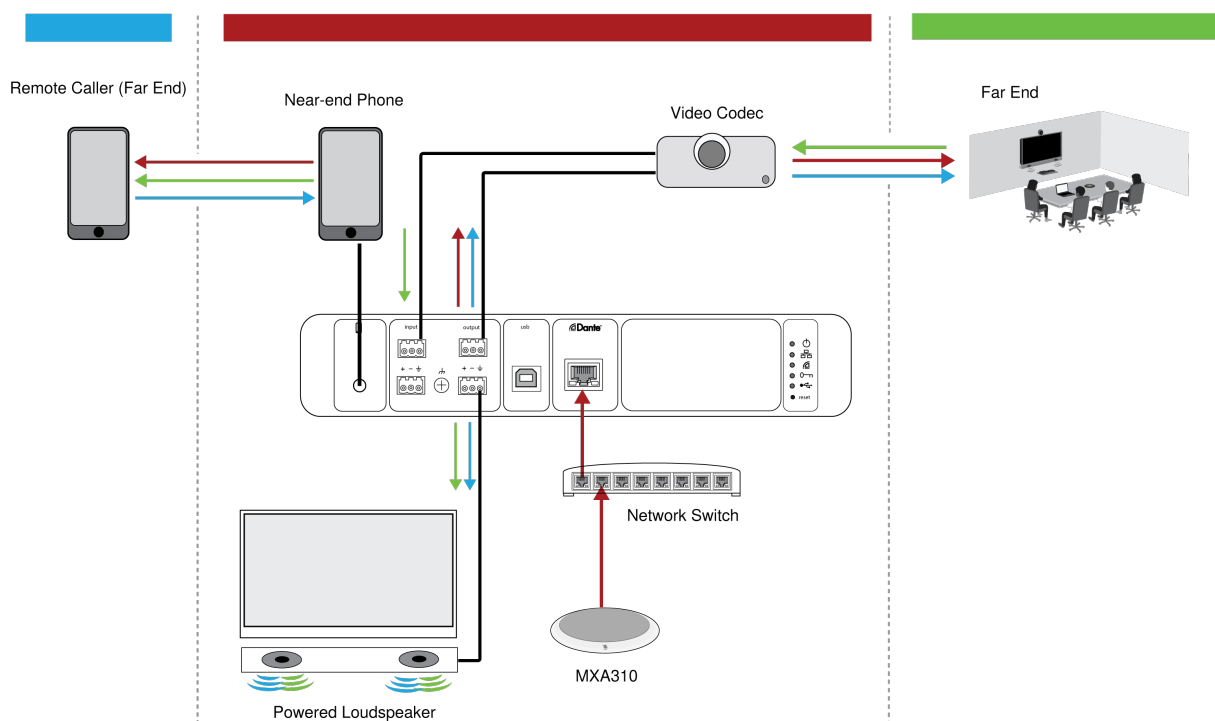
Input/Source Channel	Output/ Destination Channels
	Mobile output
<b>Analog - From Codec (Analog input 1)</b>	Analog - To Speaker (Analog output 2) USB output Mobile output
<b>USB input</b>	Analog - To Codec (Analog output 1) Analog - To Speaker (Analog output 2) Mobile output
<b>Mobile input</b>	Analog - To Codec (analog output 1) Analog - To Speaker (analog output 2) USB output

### Example Scenario: Connecting a Three-Way Call

**Near-end audio** from Dante microphones (Shure MXA 310) and the mobile phone are both routed to the video codec and sent to the far end. The mobile phone is simply carrying the audio from the remote caller -- its built-in microphone and speaker are disabled.

**Far-end audio** from the video codec is routed to a powered loudspeaker or amplifier (analog or Dante-enabled). It is also routed to the mobile phone (connected to the P300) to relay the signal to the remote caller.

The **remote caller (far end)** receives audio from both the near-end and far-end locations. The P300 connects all locations by routing both near and far-end audio sources through the mobile output. The audio from the remote caller is routed to the mobile input, and then sent to the loudspeakers in the near-end room and through the video codec to the far-end room.



Input / Source Channel	Output / Destination Channel
<b>Automix (four summed Dante input channels from MXA310)</b>	Analog - To Codec (analog output 1) Mobile output
<b>Analog - From Codec (analog input 1)</b>	Analog - To Speaker (analog output 2) Mobile output
<b>Mobile input</b>	Analog - To Codec (analog output 1) Analog - To Speaker (analog output 2)

## Mute Synchronization

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

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

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

### Shure logic devices:

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

## Setting Up Mute Sync on the MXA310 and P300

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

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

5. Go to Lights. Set the style to Ring.

---

## Using Call Status

Designer's Call status feature uses microphone LEDs to show if you're in a videoconferencing call or not. This is a location-level feature, so it applies to all microphones in a Designer location.

To use:

1. Put all microphones and processors (ANIUSB-MATRIX, IntelliMix Room, or P300) in the same Designer location.
2. Route microphone signals to the processor manually or with Designer's Optimize workflow.
3. Go to [Your location] > Settings > Call status to enable or disable Call status.

When Call status is enabled:

- Microphone LEDs off = Not in a call
- Microphone LEDs on = In a call

Call status is compatible with the following codecs:

- Microsoft Teams
- Microsoft Teams Rooms
- Zoom Client for Meetings
- Zoom Rooms

**Note:** If your codec is running on a computer with a Chrome operating system, call status will not work.

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## Adjusting Input levels

The Inputs tab controls a channel's gain before it reaches the matrix mixer. However, you should also adjust the source's gain before it reaches the P300.

**To monitor a source's input level before it reaches the P300:** Set metering to **Pre-gain** in the Settings menu.

**To adjust channel gain in the P300's Inputs tab:** Set metering to **Post-gain** in the Settings menu.

## Digital Sources (Dante and USB)

1. Check the source level before it reaches the P300:
  - Verify that the networked microphones or other Dante sources are operating at nominal output levels.
  - Set USB volume on the computer to the maximum setting if the volume control is accessible.
  - Adjust levels for Microflex® Advance™ and Microflex Wireless microphones in their web applications.
2. Adjust the P300's input gain in Designer:
  - Use the faders or manually enter a gain value. This gain stage adjusts the level of the signal before it reaches the matrix mixer.
  - Mix the levels as high as possible without the loudest channel reaching the peak level (0 dB) on the meter.

**Note:** The Matrix mixer tab lets you adjust crosspoint gain, which controls the levels for the separate submixes being sent to the different outputs.

## Analog Sources

1. Check the source level before it reaches the P300, and set any analog device with adjustable output levels to a nominal level.

2. In Designer, choose Line (+4dBu) or Aux (-10dBV) to match the incoming signal level for each analog input channel.
3. Use each channel's fader to adjust the gain before it goes to the matrix mixer.

## Mobile Devices

The mobile device input gain is optimized for most devices when the fader is set at 0 dB. It provides adequate volume with sufficient headroom. As a general target, the audio signal received by the P300 from the phone should reach an average level of approximately -24 dBFS.

1. Set the phone to approximately 50% volume.
2. Set the input gain to 0 dB.
3. Make a test call to verify the following:
  - Can you clearly hear the far-end talker?
  - Can the far-end talker clearly hear audio from the near end?

If the signal being sent to the far end is too quiet, check and adjust the gain levels for the near-end microphones and the automixer.

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## Adjusting Output Levels

**Tip:** Set the output metering in the settings menu to ensure accurate metering.

Adjust faders in the Outputs section as high as necessary, but make sure to avoid clipping (when the signal reaches 0 dBFS). Always adjust the input gain and crosspoint gain in the matrix mixer before the output gain.

**Analog output level:** Select Line, Aux, or Mic level output signal to match the sensitivity of the receiving device.

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## Pre- and Post-Gain Metering

The 2 metering modes allow you to monitor signal levels before and after the gain stages.

### Input Metering

- **Pre-gain:** Displays the pre-fader signal level. If signals are too low or clipping, adjust them at the source.
- **Post-gain:** Displays the post-fader signal level, so the meters reflect gain adjustments on input faders. This meter does not include automatic gain control or any other processing.

### Output Metering

- **Pre-gain:** Displays signal levels after input gain has been applied, but before the output faders. This meter includes input faders, digital signal processing blocks, automixing, and crosspoint gain.
  - **Post-gain:** Displays the post-fader signal level sent to each output. This meter includes gain adjustments made on the output faders.
-

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# Automix Modes

## Gating

Gating mode delivers fast-acting, seamless channel gating and consistent perceived ambient sound levels. Off-attenuation in this mode is fixed at -20 dB per channel, regardless of the number of open 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 lower off-attenuation 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.

## Automix Settings

**Note:** Not all settings are available on all automixers.

### 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.

### Always On

When selected, this channel will always be active.

### Send to Mix

When selected, sends the channel to the automix channel.

### Solo

Mutes all of the other channels

### 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.

### Mic Optimization Mode (P300 only)

Select the microphone that is used with the automixer for best performance. For best results, use Designer's Optimize workflow (this automatically selects the correct mic optimization mode).

Use the Off setting when using a Shure Microflex Wireless system, or traditional wired microphones.

### Gate Inhibit (P300 only)

Enable gate inhibit to prevent far-end audio from gating on near-end microphone channels.

1. Make sure all input gain levels are properly adjusted and all other automixer settings are configured.
2. Disable Leave last mic on.
3. Perform a test call with the far end to adjust the gate inhibit fader. Increase the fader level so that the far end indicator turns on, and far-end audio does not gate on the near-end automixer channels.
4. Verify that near-end talkers still activate the automixer channels. If the channels do not turn on, lower the gate inhibit fader.
5. Re-enable Leave last mic on if necessary.

## Automixer Direct Out Tap Points

In the Automixer tab, use the menus below each channel to choose where the signal to the matrix mixer should come from.

All options include input channel gain, mute, solo, and PEQ.

### Pre-Processing/Pre-Gate

Sends a signal without AEC, noise reduction, or AGC to the matrix mixer.

### Post-Processing/Pre-Gate

Sends a signal with AEC and noise reduction but without automixer gating or AGC to the matrix mixer.

### Post-Processing/Post-Gate

Sends a signal with automixer gating, AEC, and noise reduction but without AGC to the matrix mixer.

### Pre-Processing/Post-Gate

Sends a signal with automixer gating but without AEC, noise reduction, or AGC to the matrix mixer.

**Note:** Direct out tap points are not available on all Shure automixers.

## Encryption

Encryption operates at the location level in live mode only, meaning that all devices included in this location must have these settings. 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. Encryption is not supported with third-party devices.

To activate encryption:

1. In a location, click  (Settings), located in the upper right.



2. Select Audio encryption.
3. Select Enable Encryption.

The other options allow you to re-key the encryption or disable it if encryption had previously been enabled and you no longer want it.

**Important:** For encryption to work:

- Encryption must be universally enabled or disabled on all connected Shure devices in the same location.
- AES67 must be disabled in Dante Controller to turn encryption on or off. AES67 encryption is currently not supported.

**Note:** Encryption will not work between devices on 3.x and 4.x firmware. Update all devices to same the major firmware version to use encryption.

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## Networking and Dante

### Networking Best Practices

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

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

### Network Audio and Shure Control Data

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

#### Shure Control

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

#### Network Audio

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

## Device IP Settings

### Configure IP

Sets IP mode of the selected network interface:

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

### IP Settings

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

### MAC Address

The network interface's unique identification.

## Configuring IP Settings

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

To configure the IP properties, follow these steps:

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

## Manually Assigning Static IP Address

To manually assign IP addresses, follow these steps:

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

## Digital Audio Networking

Dante<sup>™</sup> 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)

## Setting Latency

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

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

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

## Latency Recommendations

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

## Pushing Device Names to the Dante Network

To send a device name to appear in Dante Controller, go to Settings>General and enter a Device Name. Select Push to Dante to send the name to appear on the network.

Note: names appear in Dante Controller with "-d" attached.

## 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.

## Compatibility with Dante Domain Manager

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

Considerations for Shure devices controlled by DDM:

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

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

**Note:** Applies to firmware 4.1.x and newer.

## 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.

Unicast and multicast transmission settings also affect the number of Dante flows a device can send or receive. Using multicast transmission can help overcome unicast flow limitations.

Shure devices use different Dante platforms:

Dante Platform	Shure Devices Using Platform	Unicast Transmit Flow Limit	Unicast Receive Flow Limit
Brooklyn II	ULX-D, SCM820, MXWAPT, MXWANI, P300, MXCWAPT	32	32
Brooklyn II (without SRAM)	MXA910, MXA710, AD4	16	16
Ultimo/UltimoX	MXA310, ANI4IN, ANI4OUT, ANIUSB-MATRIX, ANI22, MXN5-C	2	2
DAL	IntelliMix Room	16	16

[Learn more about Dante flows in our FAQs](#) or from [Audinate](#).

## Packet Bridge

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

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

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

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

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

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

\*UDP: User Datagram Protocol

## 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](http://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

## IP Ports and Protocols

### Shure Control

Port	TCP/UDP	Protocol	Description	Factory Default
21	tcp	FTP	Required for firmware updates (otherwise closed)	Closed

Port	TCP/UDP	Protocol	Description	Factory Default
22	tcp	SSH	Secure Shell Interface	Closed
23	tcp	Telnet	Not supported	Closed
68	udp	DHCP	Dynamic Host Configuration Protocol	Open
80*	tcp	HTTP	Required to launch embedded web server	Open
443	tcp	HTTPS	Not supported	Closed
161	tcp	SNMP	Not supported	Closed
162	tcp	SNMP	Not supported	Closed
2202	tcp	ASCII	Required for 3rd party control strings	Open
5353	udp	mDNS <sup>†</sup>	Required for device discovery	Open
5568	udp	SDT <sup>†</sup>	Required for inter-device communication	Open
8023	tcp	Telnet	Debug console interface	Closed
8180	tcp	HTML	Required for web application	Open
8427	udp	Multicast SLP <sup>†</sup>	Required for inter-device communication	Open
64000	tcp	Telnet	Required for Shure firmware update	Open

## Dante Audio & Controller

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

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

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

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## Using a Third-Party Control System

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

### Common applications:

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

A complete list of command strings is available at:

[pubs.shure.com/command-strings/P300](https://pubs.shure.com/command-strings/P300).

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## Troubleshooting

### Event Log

The event log provides a detailed account of activity from the moment the device is powered on. The log collects up to 1,000 activity entries and time-stamps them relative to the last power cycle. The entries are stored in the internal memory, and are not cleared when the device is power-cycled. The Export feature creates a CSV (comma separated values) document to save and sort the log data.

Refer to the log file for details when troubleshooting or consulting with Shure Systems Support.

### To view the event log:

1. Open the Help menu
2. Select View Event Log

Severity Level

### Information

An action or event has been successfully completed

### Warning

An action cannot be complete, but overall functionality is stable

### Error

A problem has occurred that could inhibit functionality.

Log Details

### Description

Provides details on events and errors, including IP address and subnet mask.

### Time Stamp

Power cycles:days:hours:minutes:seconds since most recent boot-up.

## Event ID

Indicates event type for internal reference.

**Tip:** Use the filter to narrow down results. Select a category heading to sort the log.

## Troubleshooting

Problem	Solution
<b>Software lags in Google Chrome browser</b>	Problem is browser-related. Turn off hardware acceleration option in Chrome.
<b>Sound quality is muffled</b>	Use equalizer to adjust frequency response. See the equalizer applications for the appropriate use.
<b>Audio sounds too high or too low in pitch</b>	Make sure that the sample rate settings for Playback and Recording are the same in your computer's sound settings. If these sample rates do not match, the audio may sound too high or too low in pitch.
<b>Hardware does not show up in device discovery</b>	<ul style="list-style-type: none"> <li>Ensure the devices are powered</li> <li>Ensure PC and equipment are on the same network and set to the same subnet</li> <li>Turn off other network interfaces not used to connect to the device (including WiFi)</li> <li>Check that DHCP server is functioning (if applicable)</li> <li>Reset the device</li> </ul>
<b>No audio</b>	<ul style="list-style-type: none"> <li>Verify the P300 is selected as the audio device in the audio devices or properties panel on the computer</li> <li>Audio channels must be routed to an output through the matrix mixer</li> <li>Connections between devices must be established in Dante Controller™ software</li> <li>Check cables</li> <li>Verify that input/output channels are not muted</li> <li>Check that fader levels are not set too low</li> <li>Make sure there is not an encryption error -- a passphrase mismatch or encryption only enabled on one device disrupts audio.</li> </ul>
<b>Cannot route Dante audio channels</b>	Install latest version of Dante Controller from Audinate, available at <a href="http://www.audinate.com">www.audinate.com</a> .
<b>Hardware does not power on</b>	<ul style="list-style-type: none"> <li>The network switch must supply Power over Ethernet. Otherwise, a PoE injector must be used</li> <li>Check network cables and connections</li> </ul>



# Contact Customer Support

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

## Specifications

### General

<b>Input</b>	(2) 3-pin block connector (Active Balanced)
<b>Output</b>	(2) 3-pin block connector (Impedance Balanced)
<b>Mobile</b>	(1) TRRS 3.5 mm (1/8")

### USB Connections

(1) USB 2.0, Type B

Single port carries 2 input and 2 output channels (Summed mono)

### Network Connections (Dante Digital Audio)

(1) RJ45

10 Dante Inputs , 8 Dante Outputs

### Polarity

Non-inverting, any input to any output

### Power Requirements

802.3 at Type 2 (PoE Plus), Class 4

### Power Consumption

17.5 W, maximum

### Weight

1710 g (3.8 lbs)

### Dimensions

H x W x D

4 x 21 x 22.6 cm (1.6 x 8.3 x 8.9 in.)

### Control Software

Shure Designer

### Operating Temperature Range

-6.7°C (20°F) to 50°C (122°F)

### Storage Temperature Range

-29°C (-20°F) to 74°C (165°F)

## Thermal Power Dissipation

<b>Maximum</b>	17.5 W (60 BTU/hr)
<b>typical</b>	14.6 W (50 BTU/hr)

## Audio

## Frequency Response

+1, -1.5 dB

20 to 20,000 Hz

## Dante Digital Audio

<b>Sampling Rate</b>	48 kHz
<b>Bit Depth</b>	24

## USB Audio

<b>Sampling Rate</b>	48 kHz
<b>Bit Depth</b>	16, 24

## Latency

Does not include Dante latency

<b>Firmware 4.1 and newer</b>	Dante 1-8 in to Dante out (AEC enabled)	15.4 ms
	<b>Dante 1-8 in to Dante out (AEC disabled)</b>	8.7 ms
	<b>Dante 9-10 in to Dante out</b>	3.4 ms
	<b>Analog in to Analog out</b>	3.8 ms
<b>Firmware 3.1 and older</b>	Dante 1-8 in to Dante out (AEC enabled)	12.5 ms
	<b>Dante 1-8 in to Dante out (AEC disabled)</b>	5.8 ms
	<b>Dante 9-10 in to Dante out</b>	1.8 ms
	<b>Analog in to Analog out</b>	2.2 ms

## Acoustic Echo Cancellation Tail Length

Up to 300 ms

## Analog Connections (Block Connectors)

## Dynamic Range

20 Hz to 20 kHz, A-weighted, typical

<b>Analog-to-Dante</b>	113 dB
<b>Dante-to-Analog</b>	117 dB

## Equivalent Input Noise

20 Hz to 20 kHz, A-weighted, input terminated with 150Ω

<b>Line</b>	-86 dBV
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<b>Aux</b>	-98 dBV
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### Total Harmonic Distortion

@ 1 kHz, 0 dBV Input, 0 dB analog gain

<0.05%

### Common Mode Rejection Ratio

150Ω balanced source @ 1 kHz

>50 dB

### Input Impedance

9.6 kΩ

### Input Clipping Level

<b>Line</b>	+27 dBV
<b>Aux</b>	+15 dBV

### Output Impedance

80 Ω

### Output Clipping Level

<b>Line</b>	+20 dBV
<b>Aux</b>	+0 dBV
<b>Mic</b>	-26 dBV

### Mobile Connection (3.5 mm Connector)

#### Pin Assignments

<b>Tip</b>	Audio Input (Left)
<b>Ring 1</b>	Audio Input (Right)
<b>Ring 2</b>	Ground
<b>Sleeve</b>	Audio Output (To Phone)

### Dynamic Range

20 Hz to 20 kHz, A-weighted, typical

<b>Analog-to-Dante</b>	99 dB
<b>Dante-to-Analog</b>	90 dB

### Equivalent Input Noise

20 Hz to 20 kHz, A-weighted, input terminated with 20Ω

-95 dBV

### Total Harmonic Distortion

@ 1 kHz, 0 dBV Input, 0 dB analog gain

<0.05%

### Input Impedance

3.7 k $\Omega$ 

### Input Clipping Level

+4 dBV

### Output Impedance

1.4 k $\Omega$ 

### Output Clipping Level

*Output terminated with 2.2 k $\Omega$* 

-20 dBV

## Networking

### Cable Requirements

Cat 5e or higher (shielded cable recommended)

## Mobile Pin Assignments (TRRS)

<b>Tip</b>	Audio Input (Left)
<b>Ring 1</b>	Audio Input (Right )
<b>Ring 2</b>	Ground
<b>Sleeve</b>	Audio Output (To Phone)

**Note:** The audio input (tip and ring 1) are summed to a mono signal in the P300, to send the signal to any destination on a single channel.

## Accessories

### Furnished Accessories

<b>KIT, HARDWARE, P300-IMX</b>	90D33522
<b>BRACKET, HALF RACK UNIT</b>	53A27741
<b>USB cable</b>	95A39698

### Optional Accessories and Replacement Parts

<b>19" rack tray</b>	CRT1
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# Safety Information

## 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.

**Note:** Model information and power ratings are labeled on the bottom of the unit.

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

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## Important Product Information

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

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

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

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

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

Authorized under the verification provision of FCC Part 15B.

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

## Information to the user

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

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

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

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

The CE Declaration of Conformity can be obtained from: [www.shure.com/europe/compliance](http://www.shure.com/europe/compliance)

Authorized European representative:

Shure Europe GmbH

Headquarters Europe, Middle East & Africa

Department: EMEA Approval

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Phone: +49-7262-92 49 0

Fax: +49-7262-92 49 11 4

Email: [info@shure.de](mailto: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](http://www.shure.com)