P300

IntelliMix® Audio Conferencing Processor

Manual for Shure IntelliMix P300 Audio Conferencing Processor. Includes system diagrams, specifications, how to use IntelliMix DSP, and installation details

Version: 8.4 (2025-E)

Table of Contents

P300 IntelliMix® Audio Conferencing Processor **4**

Overview **4**

General Description 4

Features 4

Getting Started **4**

Videoconferencing Room Equipment 4

P300 Designer Setup Example 5

P300 Parts **8**

Power Over Ethernet Plus (PoE+) 11

Cable Requirements 11

Control Devices with Shure Designer Software 11

What's in the Box **11**

Reset **12**

Software Reset Options 12

Use Designer's Auto Route **12**

DSP Settings **13**

Acoustic Echo Cancellation 13

Automatic Gain Control (AGC) 14

Noise Reduction 14

Compressor 14

Delay 15

Schematic View **15**

Adjusting Settings 16

Customizing the Workspace 16

Metering and DSP Blocks 16

Matrix Mixer **16**

Crosspoint Gain 16

Routing Channels 17

P300 Default Settings 17

Installation and Rack Mounting **18**

Securing the Devices 19

Rack Ear Configuration (CRT1) 20

Installing Underneath a Table 21

Mute Sync **21**

Use Call Status **22**

Adjusting Input levels **22**

Digital Sources (Dante and USB) 23

Analog Sources 23

Mobile Devices 23

Parametric Equalizer (PEQ) 23

Adjusting Output Levels **26**

Pre- and Post-Gain Metering **26**

Input Metering 27

Output Metering 27

Automix **27**

Automix Channel 27

Automix Settings 27

Automix Modes 29

Automixer Direct Out Tap Points 29

Use Cases **29**

P300 and Computer with Soft Codec 30

Connecting a USB Device 31

Hardware Codec System 31

Mobile Phone System 32

Security **34**

Encryption 34
Set Up the 802.1X Protocol for a Device 34

Networking 35
Networking Best Practices 35
Switch and Cable Recommendations for Dante Networking 36
Device IP Configuration 36
Setting Latency 36
QoS (Quality of Service) Settings 37
Ports, Protocols, and Firewall Rules 37
Packet Bridge 37
Digital Audio Networking 38
AES67 39

Use Command Strings 40

Troubleshooting 40
Event Log 41

Additional Resources 42

P300 Specifications 42
Mobile Pin Assignments (TRRS) 45

Optional Accessories and Replacement Parts 45

Safety Information 46
IMPORTANT SAFETY INSTRUCTIONS 46

Important Product Information 47
Information to the user 47

P300
IntelliMix® Audio Conferencing Processor

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.

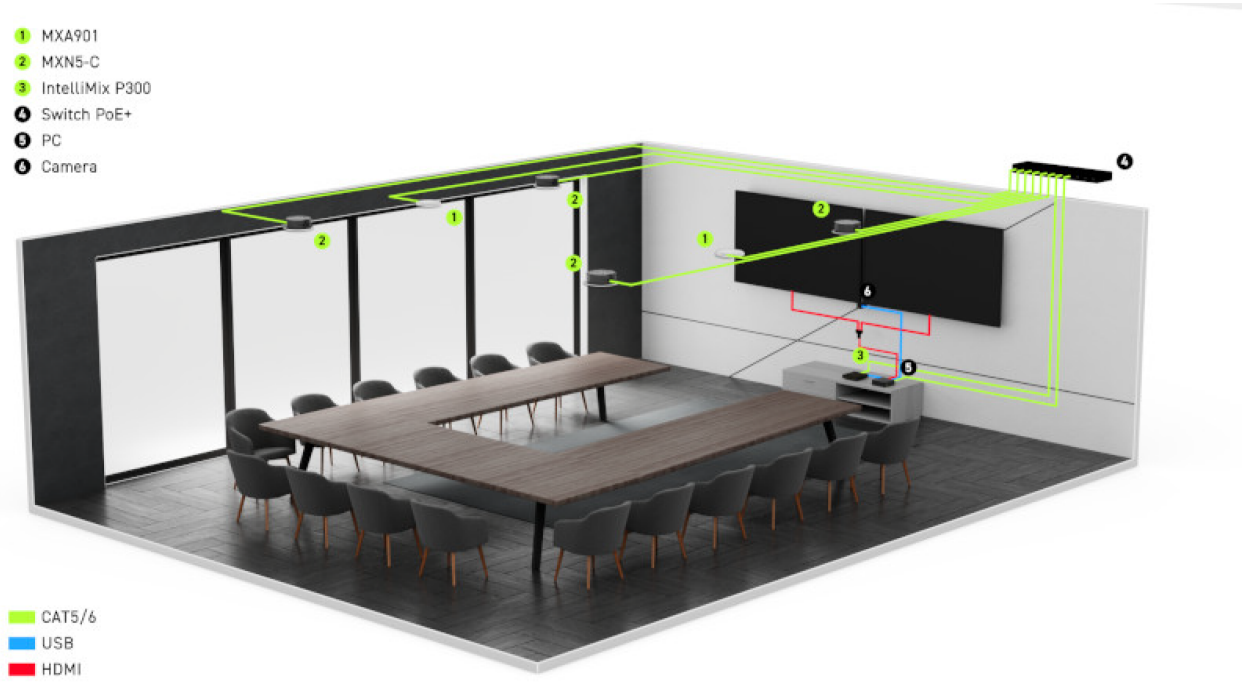
Features

- 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

Getting Started

Videoconferencing Room Equipment

Your P300 is one part of a complete room system for videoconferencing. It's helpful to think about all the parts of your room system as you get started setting up the P300.



Large room equipment example

A standard room includes some or all of this equipment:

- Microphone(s)
- DSP or other processor
- Loudspeakers (networked or analog)
- Computer with videoconferencing software
- Network switch
- Camera
- Other accessories such as table mute buttons, tablets to control meetings, and other peripherals

The Microflex® Ecosystem offers a complete portfolio of networked audio solutions that can be precisely tailored to your needs.

P300 Designer Setup Example

This example uses Shure Designer 6.0 software to set up a P300, 2 MXA901 microphones, and 4 MXN5-C loudspeakers. You might use Designer if:

- You're managing many Shure devices in different spaces
- You're routing audio between many Shure devices

You can use other combinations of devices with the P300, but the setup process in Designer is similar.

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

- Control the P300 in Designer
- Route audio and adjust DSP settings
- Understand the basic workflow for working with devices in Designer

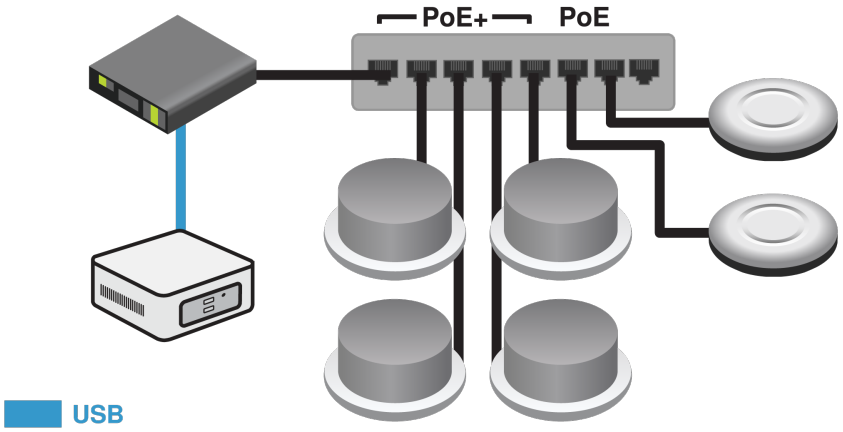
This example uses:

- Cat5e (or better) Ethernet cable

- Network switch that provides Power over Ethernet Plus (PoE+)
- Computer with Designer 6.0 or newer (download at shure.com/designer)
- Computer with videoconferencing software
- 2 MXA901 microphones
- 4 MXN5-C loudspeakers

Step 1: Install and Connect

1. Install the P300 and other devices. Connect the P300 and MXN5-C loudspeakers PoE+ ports on the network switch. The MXA901s only require PoE.
2. Connect the computer with videoconferencing software (such as Zoom or Microsoft Teams) to the P300's USB port.

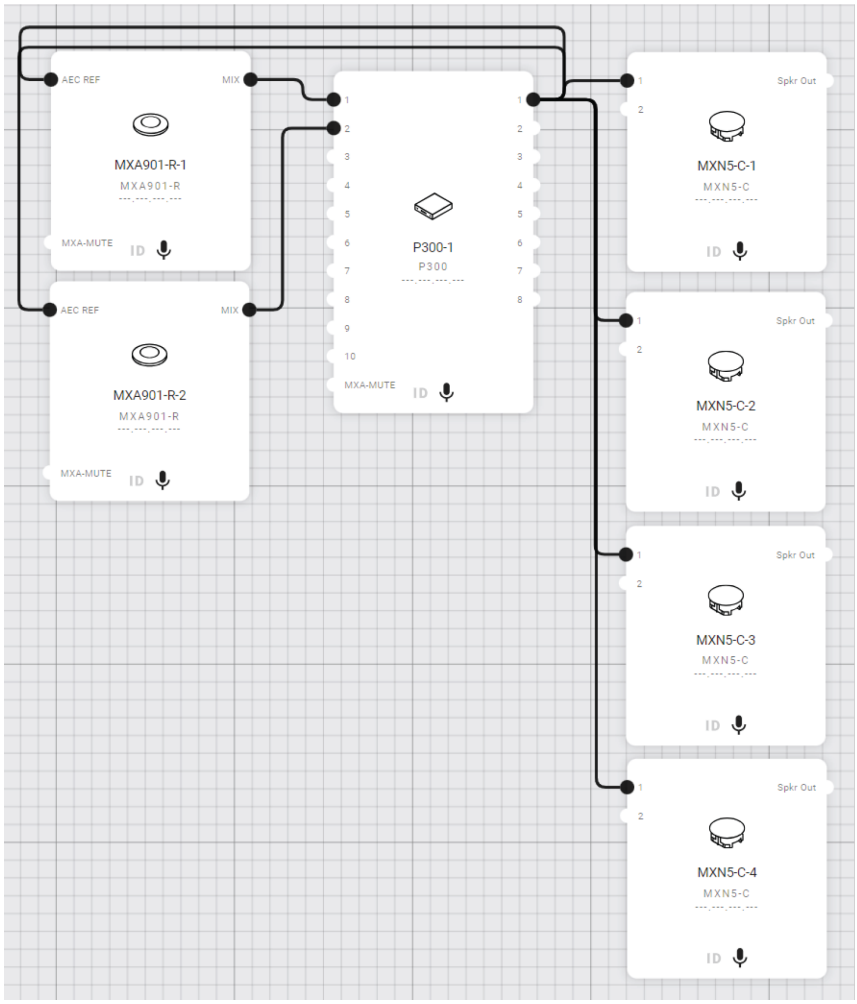


3. Connect the computer running Designer to the same network.
4. Open Designer. Check that you're connected to the correct network in `File` > `Designer preferences`.
5. Open a new design and drag the devices into the design.

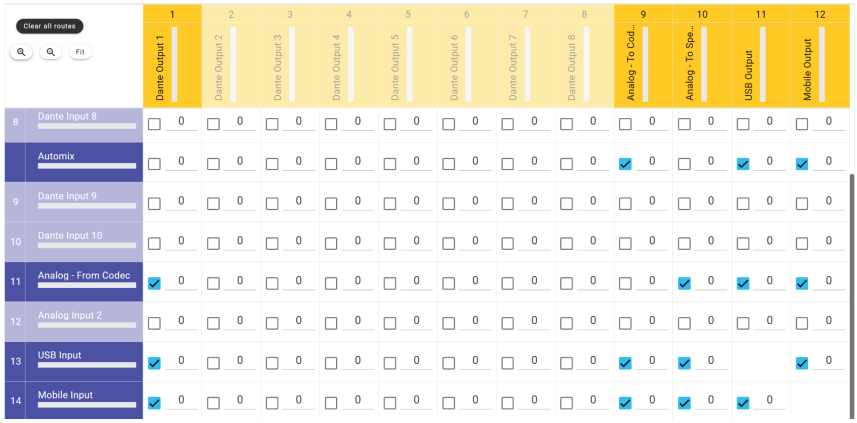
Step 2: Route Audio and Apply DSP

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

1. Select Auto route. Designer makes the following audio routes:
   - MXA901s to P300 inputs
   - P300 output to MXA901 AEC reference inputs
   - P300 output to MXN5-C inputs



The P300's matrix mixer creates these routes:



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

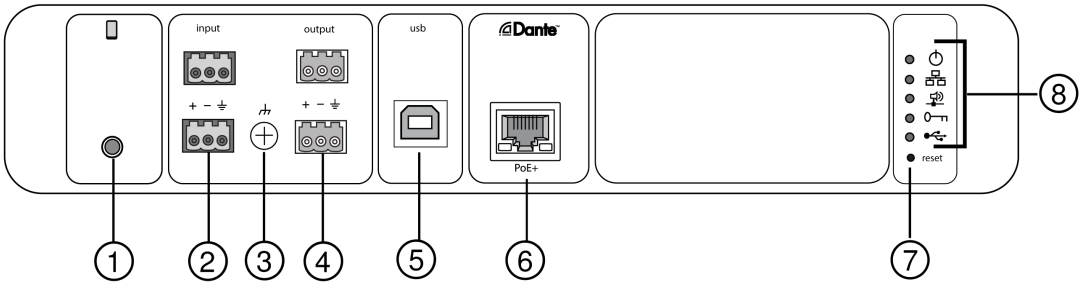
3. Select Deploy and connect to send the settings to the installed devices. Designer walks you through associating the design devices with the online devices.

Step 3: Listen and Adjust

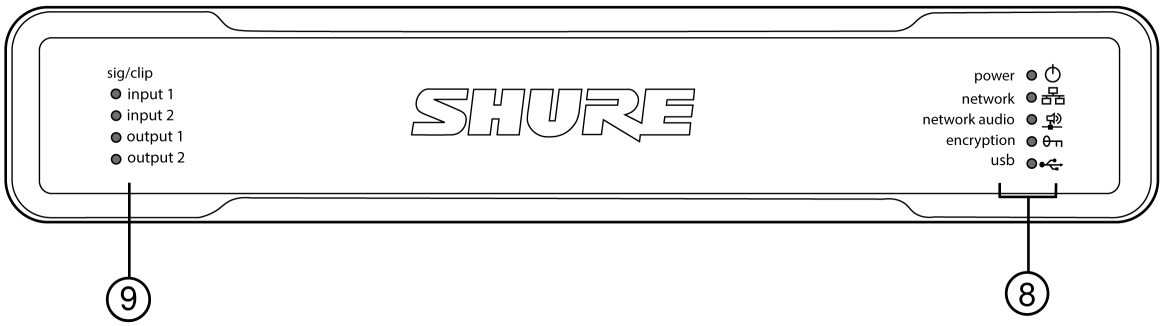
1. Open the videoconferencing software on the computer connected to the P300. In the settings, choose the P300 as the speaker and microphone.
2. Place a test call with the whole system. Have the far-end caller tell you how the microphone signals sound and make adjustments as needed:
   - P300: `[Your room]` > `P300`. The schematic tab shows all DSP blocks, and the inputs and outputs tabs have channel gain faders.
   - Microphone coverage: `[Your room]` > `Coverage`
   - MXA901s: `[Your room]` > `MXA901-R`
   - MXN5-Cs: `[Your room]` > `MXN5-C`
3. This combination of devices uses the MXA901s for AEC. The P300 output signal that feeds the 4 loudspeakers is the AEC reference source.

   For other setups, you should check that the AEC reference source is correct. Use the channel that carries audio to loudspeakers as the AEC reference.

P300 Parts



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:

- Plus sign: Audio +
- Minus sign: Audio −
- Ground symbol: 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).

- Plus sign: Audio +
- Minus sign: Audio −
- Ground symbol: 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.

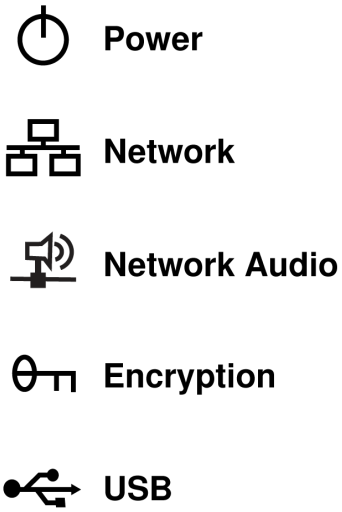
⑥ Dante Network Port

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: Power over Ethernet Plus (PoE+) present

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 in device web application for details.

Encryption LED Behavior

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 LED Behavior

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 in the device web application.*

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 (PoE+) to operate. It is compatible with **Class 4** PoE+ sources.

PoE+ 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.

Control Devices with Shure Designer Software

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

To access your device in Designer:

1. Download and install Designer on a computer connected to the same network as your device.
2. Open Designer, and check that you're connected to the correct network in `File` > `Designer preferences`.
3. Click Online devices. A list of online devices appears.
4. To identify devices, select a device and click ID in the Properties menu. Double-click the device to open the settings.

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

You can also access device settings using Shure Web Device Discovery.

What's in the Box

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

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` > `Restore default settings`): Resets audio settings back to the factory configuration (excluding device name, IP settings, and passwords).

Use Designer's Auto Route

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

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

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

To use auto route:

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

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

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

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

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

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

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

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

DSP Settings

Acoustic Echo Cancellation

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

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

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

Select a Reference Signal for AEC

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

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

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

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

AEC Settings

Reference Meter	Use the reference meter to visually verify the reference signal is present. The reference signal should not be clipping.
ERLE	Echo return loss enhancement (ERLE) displays the dB level of signal reduction (the amount of echo being removed). If the reference source is connected properly, the ERLE meter activity generally corresponds to the reference meter.
Reference	Indicates which channel is serving as the far end reference signal.

Non-Linear Processing	The primary component of the acoustic echo canceller is an adaptive filter. Non-linear processing supplements the adaptive filter to remove any residual echo caused by acoustic irregularities or changes in the environment. Use the lowest possible setting that is effective in your room. **Low:** Use in rooms with controlled acoustics and minimal echoes. This setting provides the most natural sound for full duplex. **Medium:** Use in typical rooms as a starting point. If you hear echo artifacts, try using the high setting. **High:** Use to provide the strongest echo reduction in rooms with bad acoustics, or in situations where the echo path frequently changes.

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.

Noise Reduction

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

Settings

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.

Schematic View

The schematic view in Designer shows a device's entire audio signal chain, with the ability to adjust settings and monitor signals. The schematic views for the P300, ANIUSB-MATRIX, and IntelliMix Room look slightly different because each device has a different mix of inputs and outputs.



Example of the P300's schematic view

Adjusting Settings

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

Channel Settings

Setting	Function
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. You can also double-click a block to edit it.

Right-click in a blank area to access global settings:

- Mute/unmute all input channels
- Mute/unmute all output channels
- Close all open dialogs

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 2 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 DSP Blocks

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

DSP blocks have a color when they are turned on. When off, they are gray. Use these tools to troubleshoot audio signals and verify connections and levels.

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.

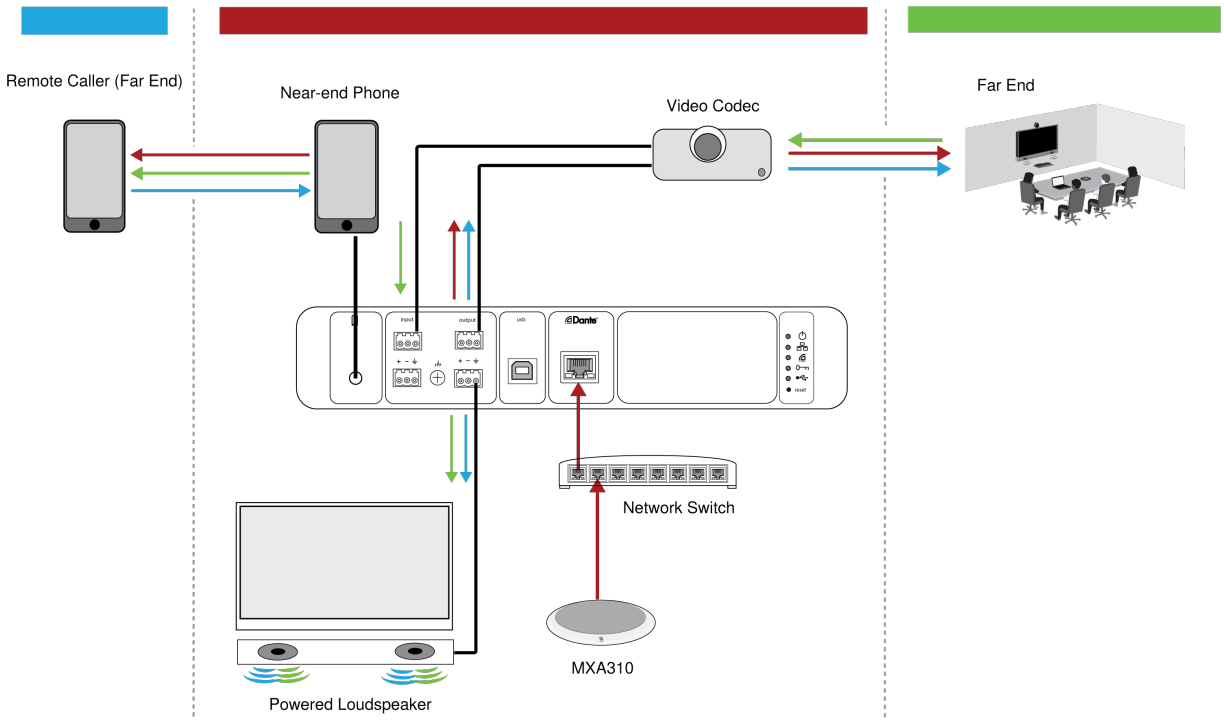
Input/Source Channel	Output/ Destination Channels
Automix (summed Dante input channels)	Analog - To Codec (Analog output 1) USB output Mobile output
Analog - From Codec (Analog input 1)	Analog - To Speaker (Analog output 2) USB output Mobile output
USB input	Analog - To Codec (Analog output 1) Analog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

Installation and Rack Mounting

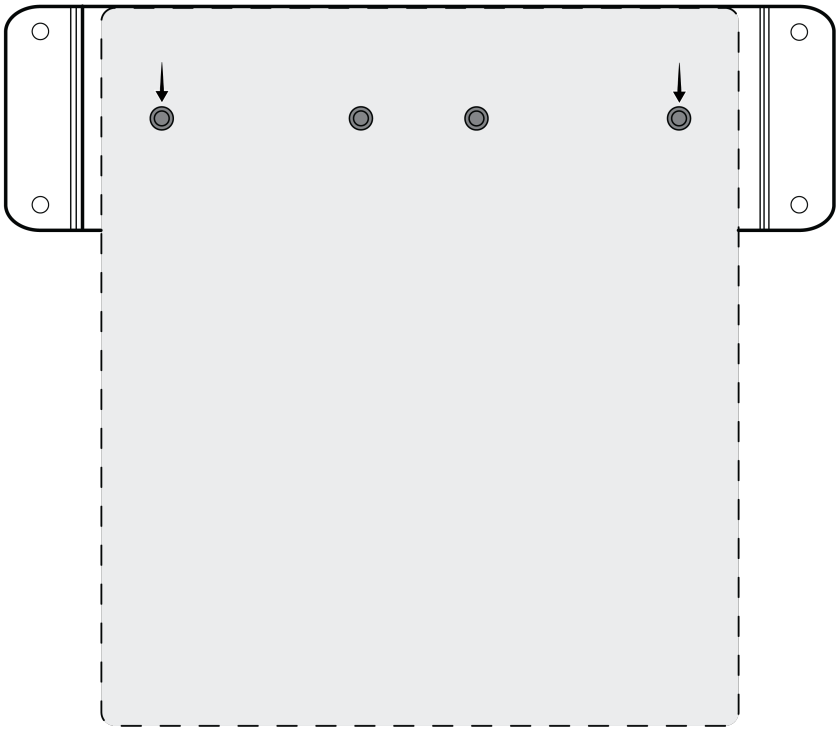
Two mounting solutions are available for installing the P300:

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

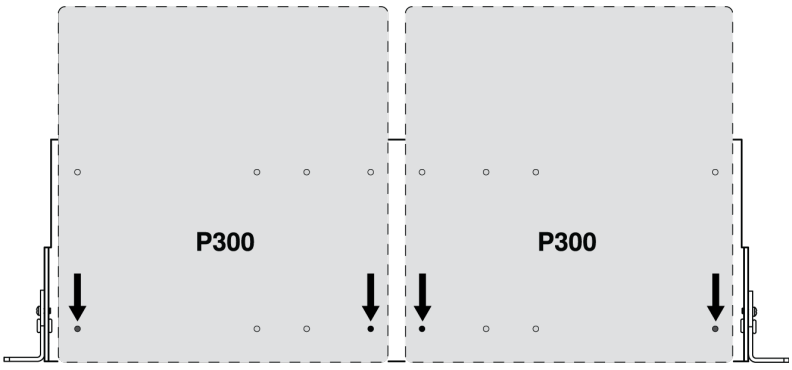
Single-unit Mounting Tray (included accessory): Supports a single device for mounting under a table

Securing the Devices

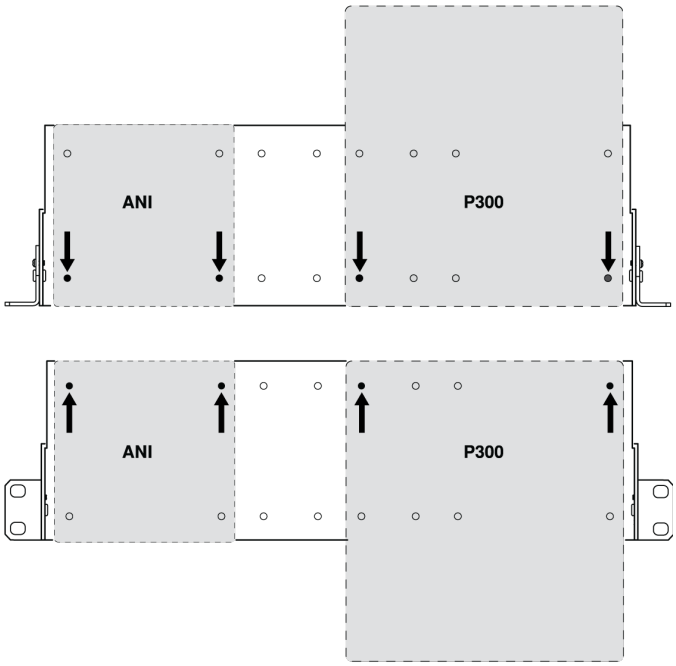
Use the included screws from the mounting hardware kit to secure each P300 or Audio Network Interface (ANI). Devices can be mounted to face either direction. Insert the screws from the bottom in the appropriate holes, according to the following diagrams:



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.

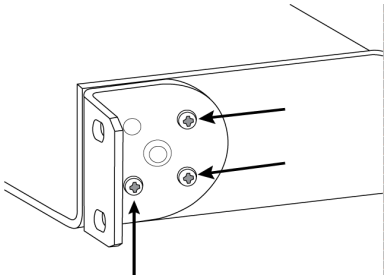


Rack Ear Configuration (CRT1)

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

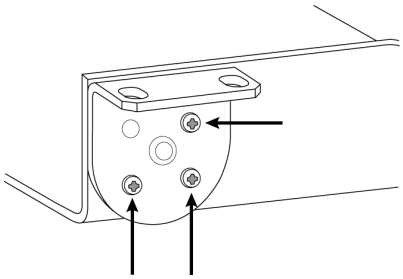
Standard 19" Rack Mount

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.



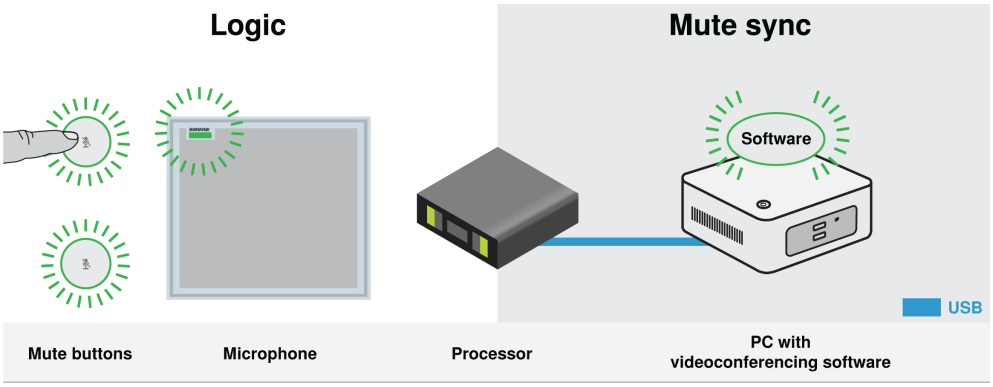
Installing Underneath a Table

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

Mute Sync

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

- **Logic:** Aligns mute status across Shure devices in the room. The processor (such as a P300, IntelliMix Room software, or an ANIUSB-MATRIX) is the controller.
- **Mute sync:** Aligns mute between the processor and the videoconferencing software, which is on a computer connected to the processor by USB. The processor's automix output is muted or unmuted to change the mute status of the system.



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

To use mute sync:

1. In Designer, create audio and mute control routes between devices in the room.
2. Connect a computer with videoconferencing software to the processor's USB port.
3. Turn on mute sync and logic:
   - Processors: `[Your device]` > `Settings` > `Mute control`
   - Microphones without physical mute button: Logic is always on

    - MXA310 and MXA mute button: `Settings` > `Logic control` > `Mute control function` > `Logic out`

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

Compatible Shure Logic Devices

- P300 (Also mutes supported soft codecs connected by USB)
- ANIUSB-MATRIX (Also mutes supported soft codecs connected by USB)
- IntelliMix Room software (Also mutes supported soft codecs connected by USB)
- MXA901
- MXA902
- MXA910
- MXA920
- MXA710
- MXA310
- Network Mute Button
- ANI22-BLOCK
- ANI4IN-BLOCK
- Logic-enabled MX microphones connected to ANI22-BLOCK or ANI4IN-BLOCK
    - MX392
    - MX395-LED
    - MX396
    - MX405/410/415

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

Use Call Status

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

To use call status:

1. Put all microphones and processors (ANIUSB-MATRIX, IntelliMix Room, or P300) in the same Designer room.
2. Select Auto route to create audio and mute control routes. You can also route audio manually.
3. Go to `[Your room]` > `Settings` > `Call status` to turn on call status.

When call status is on:

- 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.

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 MicroflexAdvance™ 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.

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

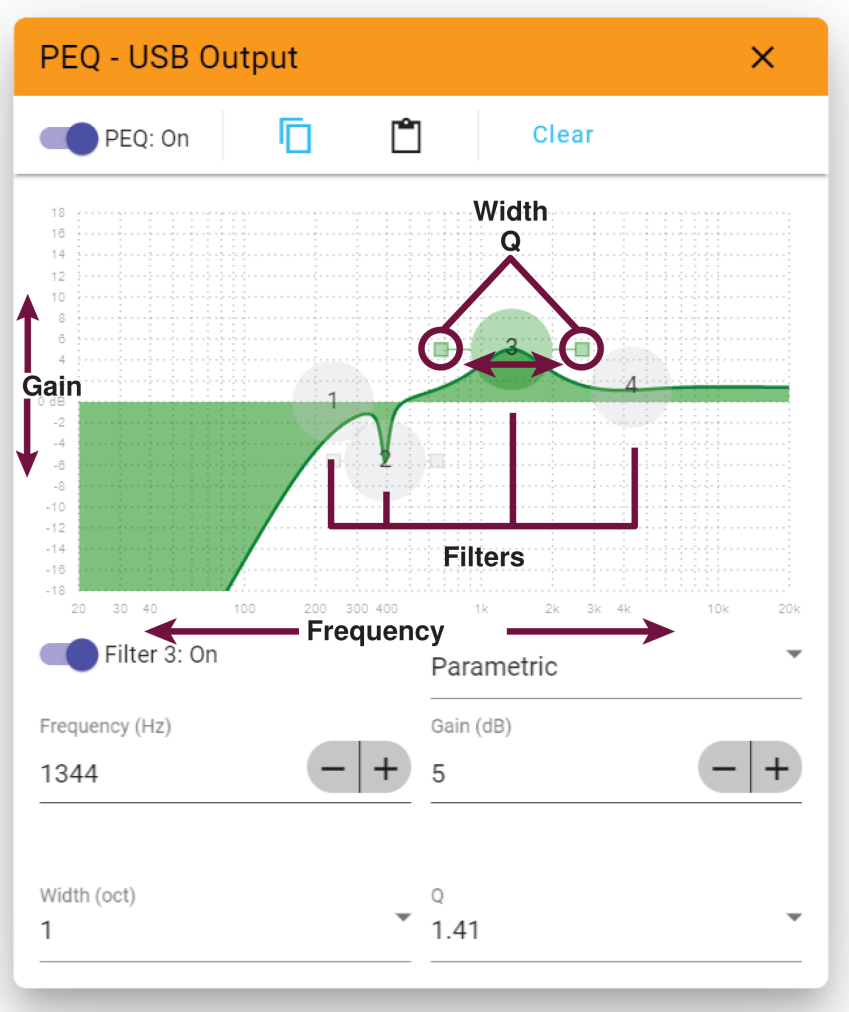
Select the center frequency of the filter to cut/boost

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.



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 Applications

EQ Application	Suggested Settings
Treble boost for improved speech intelligibility	Add a high shelf filter to boost frequencies greater than 1 kHz by 3 to 6 dB.
HVAC noise reduction	Add a low cut filter to attenuate frequencies below 200 Hz.
Reduce flutter echoes and sibilance	Identify the specific frequency range that "excites" the room: 1. Set a narrow Q value. 2. Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 1 kHz and 6 kHz to pinpoint the range of flutter echoes or sibilance. 3. Reduce the gain at the identified frequency (start between −3 and −6 dB) to minimize the unwanted room sound.
Reduce hollow, resonant room sound	Identify the specific frequency range that "excites" the room: 1. Set a narrow Q value. 2. Increase the gain to between +10 and +15 dB, and then experiment with frequencies between 300 Hz and 900 Hz to pinpoint the resonant frequency. 3. Reduce the gain at the identified frequency (start between −3 and −6 dB) to minimize the unwanted room sound.

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.

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.

Automix

Automix Channel

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

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

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

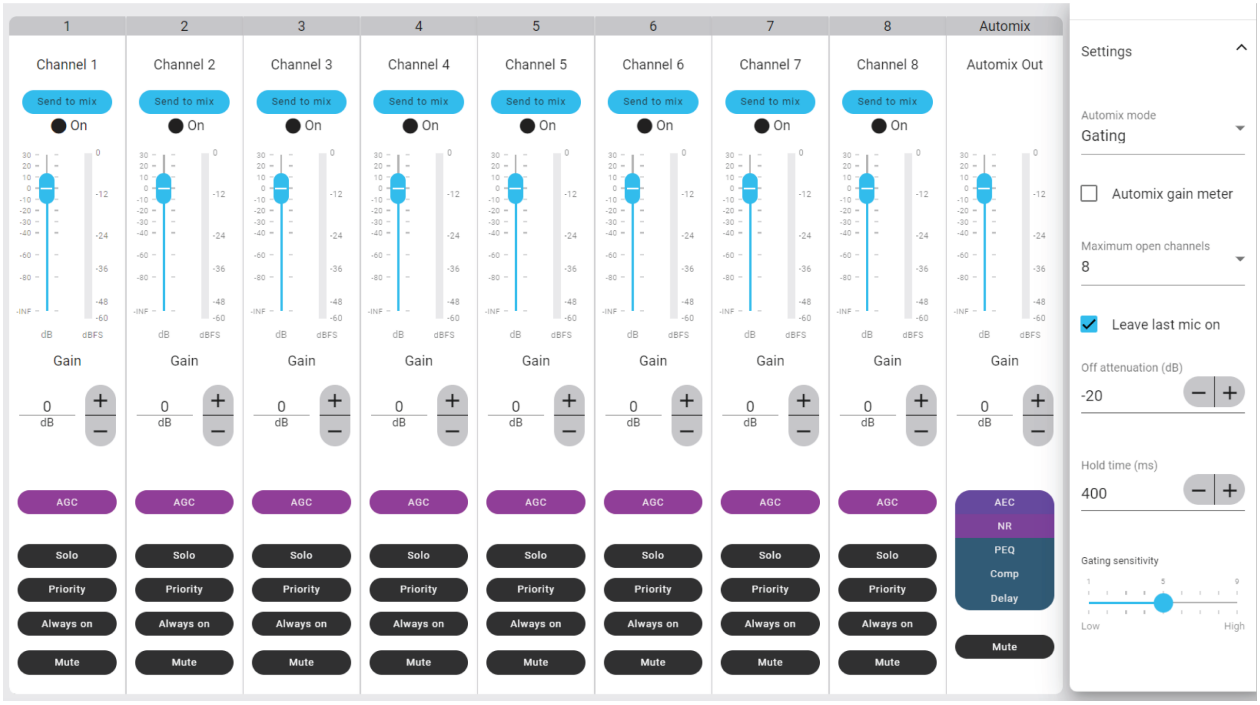
To use the automix channel, do the following:

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

Automix Settings

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

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



Automix Settings

Setting	Function
Leave last mic on	Keeps the most recently used microphone channel active. The purpose of this feature is to keep natural room sound in the signal so that meeting participants on the far end know the audio signal has not been interrupted.
Gating sensitivity	Changes the threshold of the level at which the gate is opened
Off attenuation	Sets the level of signal reduction when a channel is not active
Hold time	Sets the duration for which the channel remains open after the level drops below the gate threshold
Maximum open channels	Sets the maximum number of simultaneously active channels
Automix gain meter	When enabled, changes gain meters to display automix gating in real time. Channels that gate open will display more gain than channels that are closed (attenuated) in the mix.
Priority	When selected, this channel gate activates regardless of the number of maximum open channels.
Always on	When selected, this channel will always be active.
Send to mix	When selected, sends the channel to the automix channel.

Setting	Function
Solo	Mutes all of the other channels

Note: Not all settings are available on all automixers.

Automix Modes

To select an automix mode, go to:

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

Gating

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

Gain Sharing

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

Manual

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

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, automixer gating, 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.

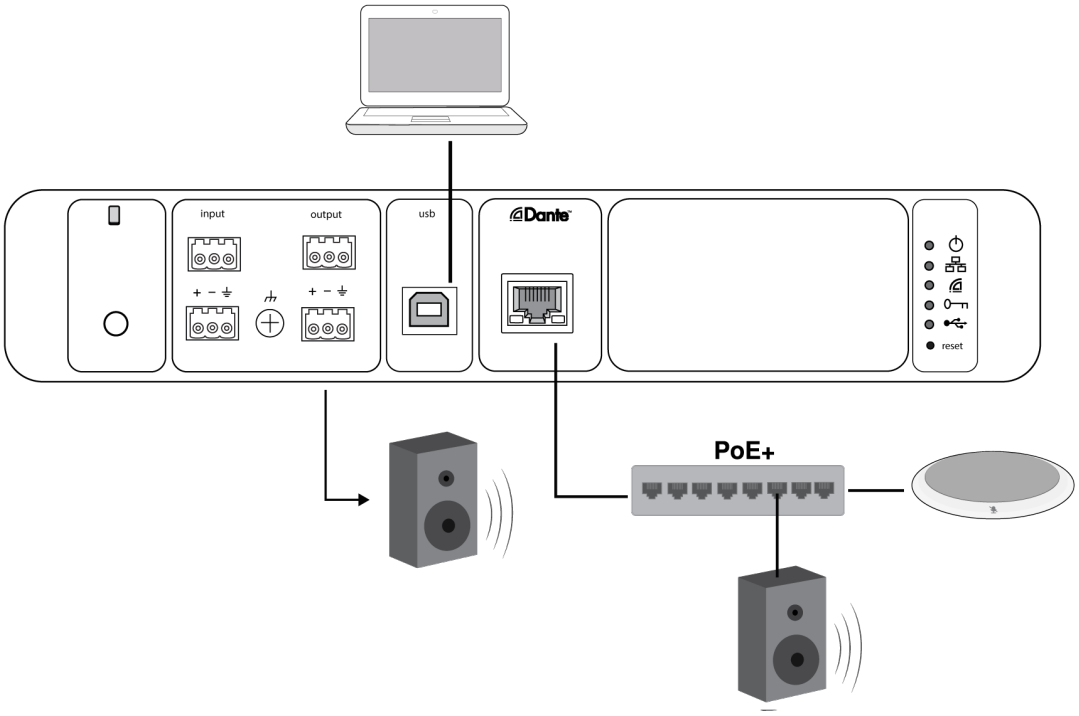
Pre-Processing/Post-Gate

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.

Use Cases

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
USB input	Analog - To Speaker **or** Dante output

4. 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.
5. Adjust input and output levels using Designer and perform a sound check. Refer to the Help section for additional information.

USB Device Type Settings

When you use a USB-connected codec with a P300 or ANIUSB-MATRIX, you may need to set the USB device type on the Shure device.

The USB device type setting tells the codec if acoustic echo cancellation (AEC) is needed or not. The codec can then turn on or turn off its own AEC (if supported).

For example: You have an MXA710 routed to an ANIUSB-MATRIX and want to use the MXA710's AEC. Set the USB device type to Echo-canceling speakerphone to tell the codec to turn off its AEC.

To change USB device type:

1. In Designer, find the P300 or ANIUSB-MATRIX and go to `Settings` > `General`.
2. Choose a USB device type:
   - Speakerphone: For using the codec's AEC
   - Echo-canceling speakerphone: For using the Shure device's AEC
3. After changing the USB device type, select your device as the default audio device in the codec's settings.
4. Place a test call to make sure AEC is working properly.

Connecting a USB Device

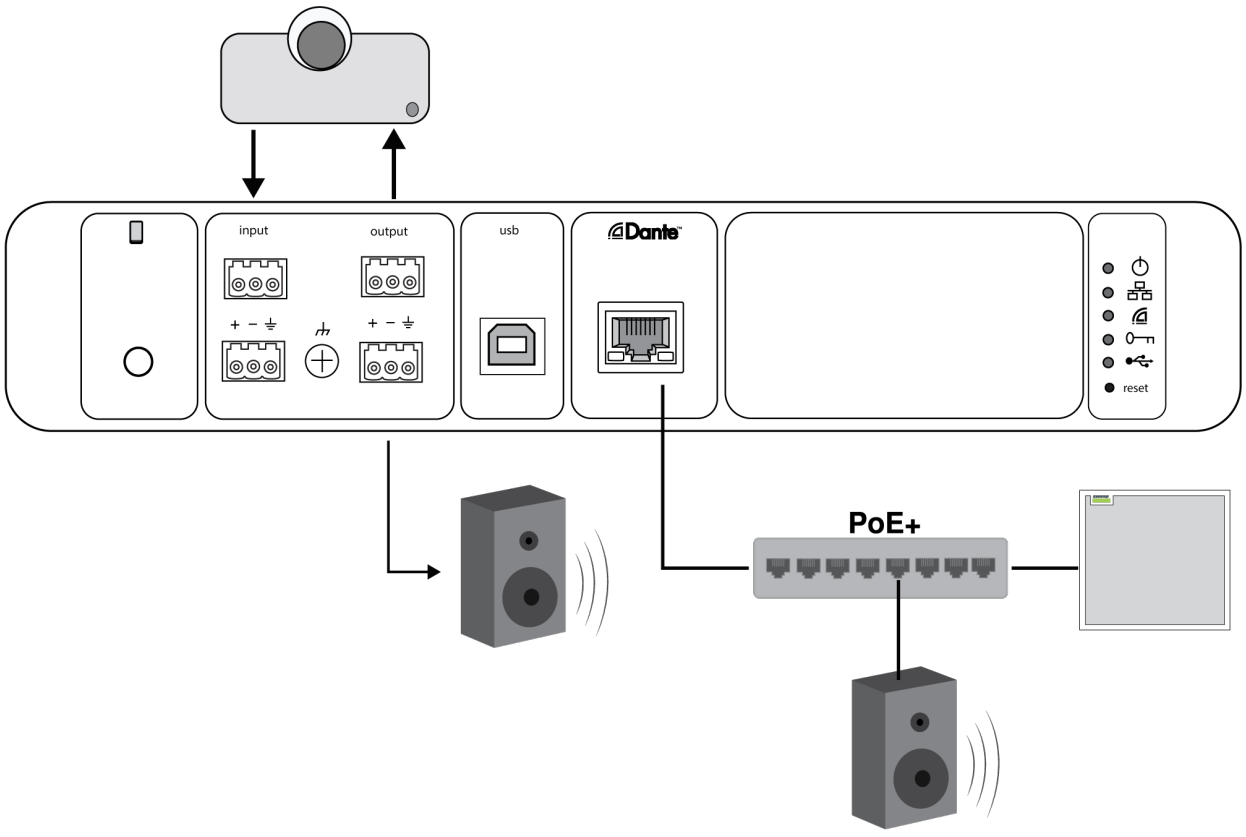
The USB port connects the host computer to the entire room audio system, including microphones and loudspeakers.

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



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

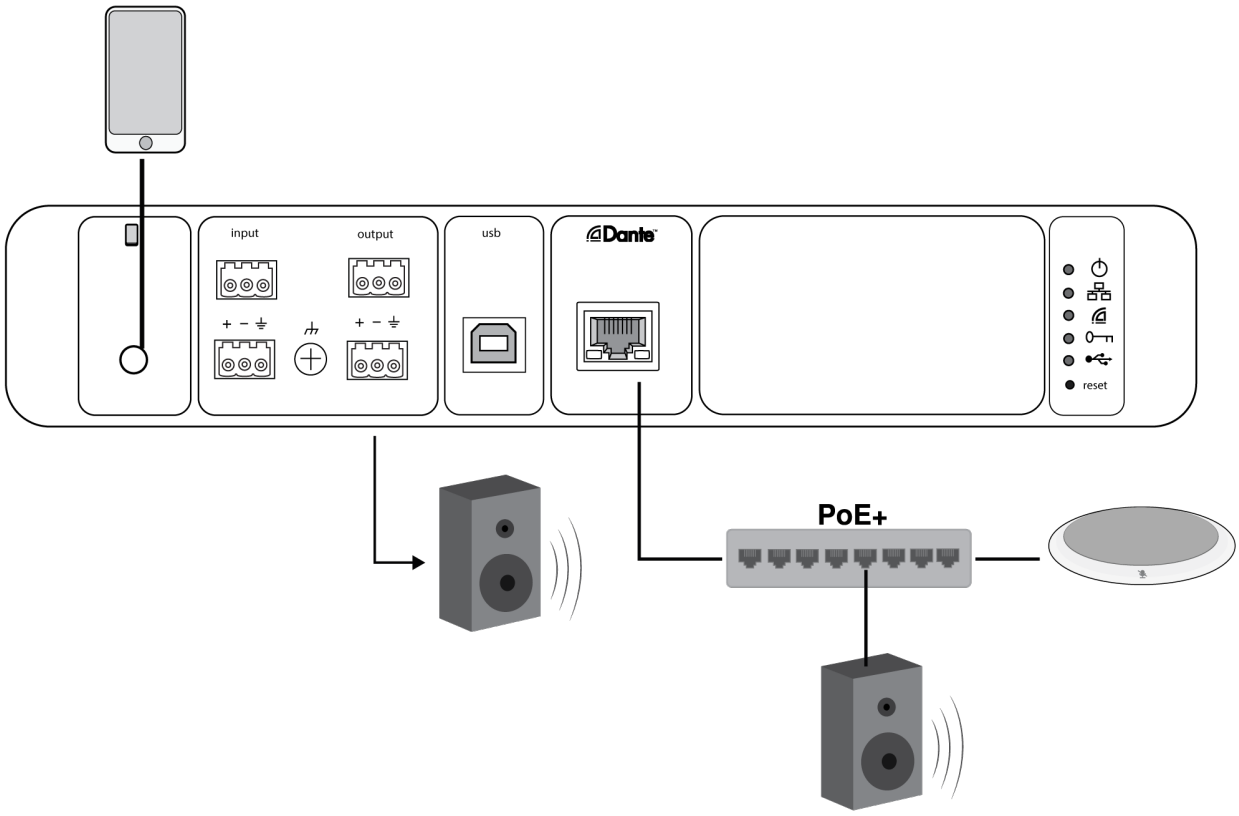
3. Connect a powered loudspeaker or amplifier to the P300 analog output 2. In the matrix mixer, this is labeled Analog - To Speaker. You can also use Dante PoE loudspeakers connected to the network switch.
4. In Designer, go to Matrix mixer to make connections between devices. Some connections are established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

Required Matrix Mixer Connections:

Input	Output
Automix	Analog - To codec
Analog - From Codec	Analog - To Speaker **or** 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 **or** Dante output

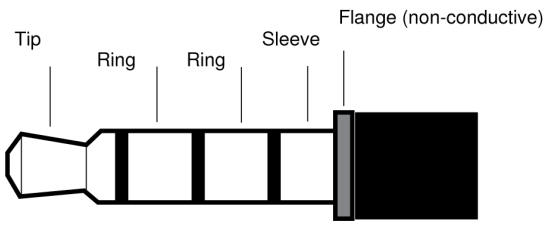
4. In Schematic, right-click any AEC block and choose an AEC reference source. The reference source is Analog - To Speaker for analog loudspeakers, or one of the Dante outputs for Dante loudspeakers.
5. Adjust input and output levels using Designer and perform a sound check. Refer to the Help section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

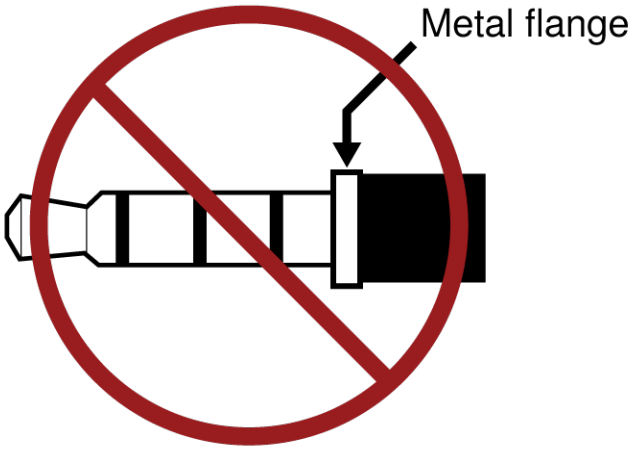
Mobile Connection Cable Requirements

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.

Security

Encryption

Audio is encrypted with the Advanced Encryption Standard (AES-256), as specified by the US Government National Institute of Standards and Technology (NIST) publication FIPS-197. Shure devices that support encryption require a password to make a connection. Encryption is not supported with third-party devices.

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

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

Important: For encryption to work:

- All Shure devices on your network must use encryption.
- Disable AES67 in Dante Controller. AES67 and AES-256 cannot be used at the same time.

Set Up the 802.1X Protocol for a Device

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

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

Setting up 802.1X is a two-part process.

To set up 802.1X, you will need:

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

Step 1: Configure Settings on Test Network

1. Connect the device to your test network and discover it using Designer.
2. Set a device password if desired.
3. Double-click the device and go to `Settings` > `Network` > `802.1X`.
4. Choose your EAP method from the menu.
5. Enter any required credentials and load any necessary certificates.
6. Press Save to save the 802.1X settings to the device.
7. Enable 802.1X and select Reboot later.

Step 2: Connect to a Credentialed Network

1. Connect your device to the credentialed network.

2. Ensure that Designer is connected to the credentialed network.
3. Go to `Settings` > `Network` > `802.1X` and enable 802.1X. Reboot the device for the 802.1X settings to take effect.
4. If the device doesn't appear in Designer after the reboot, reconnect to the test network and check all 802.1X settings for the selected EAP method.

Turn Off or Clear 802.1X Settings

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

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

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

Change 802.1X Settings

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

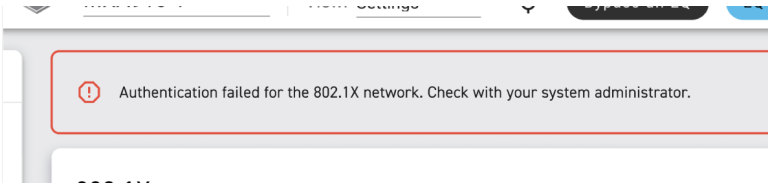
To change device settings:

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

Troubleshooting 802.1X Setup Issues

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

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



If this occurs, check with your system administrator.

Networking

Networking Best Practices

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

- Always use a "star" network topology by connecting each device directly to the switch or router.
- Connect all Shure networked devices to the **same network** and set to the **same subnet**.
- Allow all Shure software through the firewall on your computer.

- Use only 1 DHCP server per network. Disable DHCP addressing on additional servers.
- Power on the switch and DHCP server before powering on the Shure devices.
- To expand the network, use multiple switches in a star topology.
- All devices must be at the same firmware revision level.

Switch and Cable Recommendations for Dante Networking

Switches and cables determine how well your audio network performs. Use high-quality switches and cables to make your audio network more reliable.

Network switches should have:

- Gigabit ports. 10/100 switches may work on small networks, but gigabit switches perform better.
- Power over Ethernet (PoE) or PoE+ ports for any devices that require power
- Management features to provide information about port speed, error counters, and bandwidth used
- Ability to switch off Energy Efficient Ethernet (EEE). EEE (also known as "Green Ethernet") may cause audio dropouts and problems with clock synchronization.
- Diffserv (DSCP) Quality of Service (QoS) with strict priority and 4 queues

Ethernet cables should be:

- Cat5e or better
- Shielded

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

Device IP Configuration

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

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

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

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

Setting Latency

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

These latency values should be used as a starting point. To determine the exact latency to use for your setup, deploy the 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

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

QoS (Quality of Service) Settings

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

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

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

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

Dante QoS Priority Values

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

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

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

Networking Terminology

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

Ports, Protocols, and Firewall Rules

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

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

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

The packet bridge does not allow cross-subnet command strings.

Digital Audio Networking

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

Compatibility with Dante Domain Manager

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

Considerations for Shure devices controlled by DDM:

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

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

Dante Flows for Shure Devices

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

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

Dante Flows for Shure Devices

Dante Platform	Shure Devices Using Platform	Transmit Flow Limit	Receive Flow Limit
Brooklyn II	ULX-D, SCM820, MXWAPT, MXWANI, P300, MXCWAPT	32	32

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

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

AES67

AES67 is a networked audio standard that enables communication between hardware components which use different IP audio technologies. This Shure device supports AES67 for increased compatibility within networked systems for live sound, integrated installations, and broadcast applications.

The following information is critical when transmitting or receiving AES67 signals:

- Update Dante Controller software to the newest available version to ensure the AES67 configuration tab appears.
- Before turning encryption on or off, you must disable AES67 in Dante Controller.
- AES67 cannot operate when the transmit and receive devices both support Dante.

Shure Device Supports:	Device 2 Supports:	AES67 Compatibility
Dante and AES67	Dante and AES67	No. Must use Dante.
Dante and AES67	AES67 without Dante. Any other audio networking protocol is acceptable.	Yes

Separate Dante and AES67 flows can operate simultaneously. The total number of flows is determined by the maximum flow limit of the device.

Sending Audio from a Shure Device

All AES67 configuration is managed in Dante Controller software. For more information, refer to the Dante Controller user guide.

1. Open the Shure transmitting device in Dante Controller.
2. Enable AES67.
3. Reboot the Shure device.
4. Create AES67 flows according to the instructions in the Dante Controller user guide.

Receiving Audio from a Device Using a Different Audio Network Protocol

Third-party devices: When the hardware supports SAP, flows are identified in the routing software that the device uses. Otherwise, to receive an AES67 flow, the AES67 session ID and IP address are required.

Shure devices: The transmitting device must support SAP. In Dante Controller, a transmit device (appears as an IP address) can be routed like any other Dante device.

Use Command Strings

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

Common applications:

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

A complete list of command strings is available at:

pubs.shure.com/command-strings/P300.

Troubleshooting

Problem	Solution
Software lags in Google Chrome browser	Problem is browser-related. Turn off hardware acceleration option in Chrome.
Sound quality is muffled	Use equalizer to adjust frequency response. See the equalizer applications for the appropriate use.
Audio sounds too high or too low in pitch	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.
Hardware 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
No audio	Verify the P300 is selected as the audio device in the audio devices or properties panel on the computer Audio channels must be routed to an output through the matrix mixer Connections between devices must be established in Dante Controller® software Check cables

Problem	Solution
	Verify that input/output channels are not muted Check that fader levels are not set too low Make sure there is not an encryption error -- a passphrase mismatch or encryption only enabled on one device disrupts audio.
Cannot route Dante audio channels	Install latest version of Dante Controller from Audinate, available at www.audinate.com.
Hardware does not power on	The network switch must supply Power over Ethernet. Otherwise, a PoE injector must be used Check network cables and connections

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) file to save and sort the log data.

Refer to the log file for details when troubleshooting or when working with Shure support.

To view the event log:

1. Open the device's web application. As of Designer 6 and newer, there is no longer an event log in Designer.
2. Select `Help` > `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.

Additional Resources

- Shure Knowledge Base FAQs
- Command strings for Shure devices
- IP Ports and Protocols for Shure Devices
- Firewall Rules for Shure Software Applications
- Shure API documentation
- Shure Enterprise Networking Troubleshooting Checklist
- Training from the Shure Audio Institute
- Shure Systems YouTube channel

Download Shure Software

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

P300 Specifications

General

Input	(2) 3-pin block connector (Active Balanced)
Output	(2) 3-pin block connector (Impedance Balanced)
Mobile	(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

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

Audio

Frequency Response
+1, -1.5 dB

20 to 20,000 Hz

Dante Digital Audio

Sampling Rate	48 kHz
Bit Depth	24

USB Audio

Sampling Rate	48 kHz
Bit Depth	16, 24

Latency
Does not include Dante latency

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

	Analog in to Analog out	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

Analog-to-Dante	113 dB
Dante-to-Analog	117 dB

Equivalent Input Noise

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

Line	-86 dBV
Aux	-98 dBV

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

Line	+27 dBV
Aux	+15 dBV

Output Impedance

80 Ω

Output Clipping Level

Line	+20 dBV
Aux	+0 dBV
Mic	-26 dBV

Mobile Connection (3.5 mm Connector)

Pin Assignments

Tip	Audio Input (Left)
Ring 1	Audio Input (Right)

Ring 2	Ground
Sleeve	Audio Output (To Phone)

Dynamic Range

20 Hz to 20 kHz, A-weighted, typical

Analog-to-Dante	99 dB
Dante-to-Analog	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 kΩ

Input Clipping Level

+4 dBV

Output Impedance

1.4 kΩ

Output Clipping Level

Output terminated with 2.2 kΩ

-20 dBV

Networking

Cable Requirements

Cat 5e or higher (shielded cable recommended)

Mobile Pin Assignments (TRRS)

Tip	Audio Input (Left)
Ring 1	Audio Input (Right)
Ring 2	Ground
Sleeve	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.

Optional Accessories and Replacement Parts

19" rack tray	CRT1

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.

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

Important Product Information

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

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

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

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

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

Authorized under the verification provision of FCC Part 15B.

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

Dante is a registered trademark of Audinate Pty Ltd.

Information to the user

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the manufacturer's instruction manual, may cause interference with radio and television reception.

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

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

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

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

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

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

Authorized European representative:
Shure Europe GmbH
Global Compliance
Jakob-Dieffenbacher-Str. 12

75031 Eppingen, Germany
Phone: +49-7262-92 49 0
Email: info@shure.de
www.shure.com

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

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