Safety Information

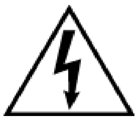
IMPORTANT SAFETY INSTRUCTIONS

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.

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 / 2 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 Shure 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 IntelliMix 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

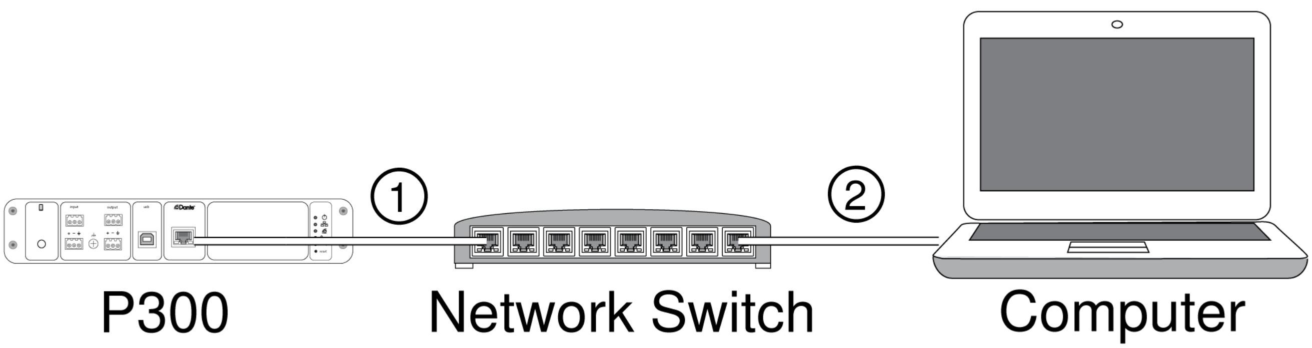
Getting Started

This device features a browser-based web application, which controls audio and network properties. Upon completing this basic setup process, you will be able to:

- Access the web application to customize audio settings, signal routing, and network properties
- Use Dante Controller software to connect with other Dante devices and pass audio
- Access additional configuration information

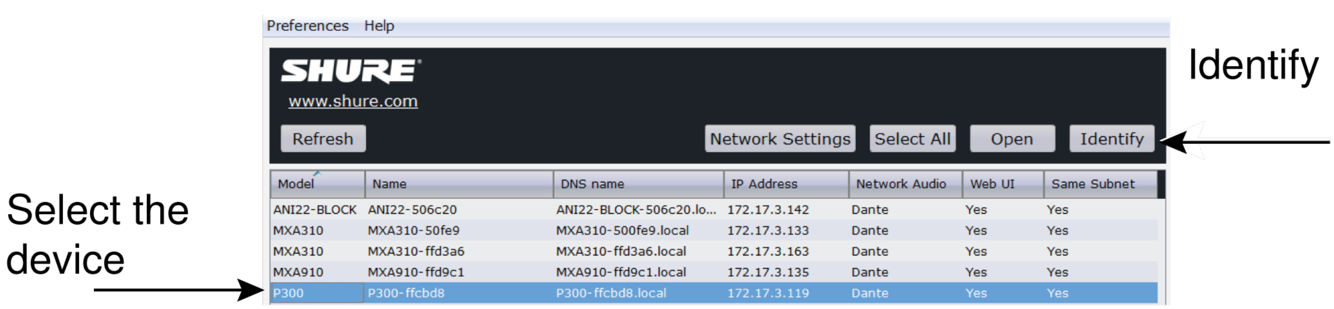
Step 1: Connect to a Network

1. Use an Ethernet cable (CAT5e or higher) to connect the P300 to a network switch.
   Note: The network switch must provide Power over Ethernet Plus (PoE+). Make sure to connect to a PoE+ port, since many switches do not supply power on all ports.
2. Connect a computer to the network switch with an Ethernet cable



Step 2: Access the Web Application

1. Download and install the **Shure Device Discovery application** (http://www.shure.com)
2. Open the Shure Device Discovery application
3. Double-click the device to open the web application.
   Tip: If setting up multiple Shure devices, use the Identify button in the application to flash the lights on the device.

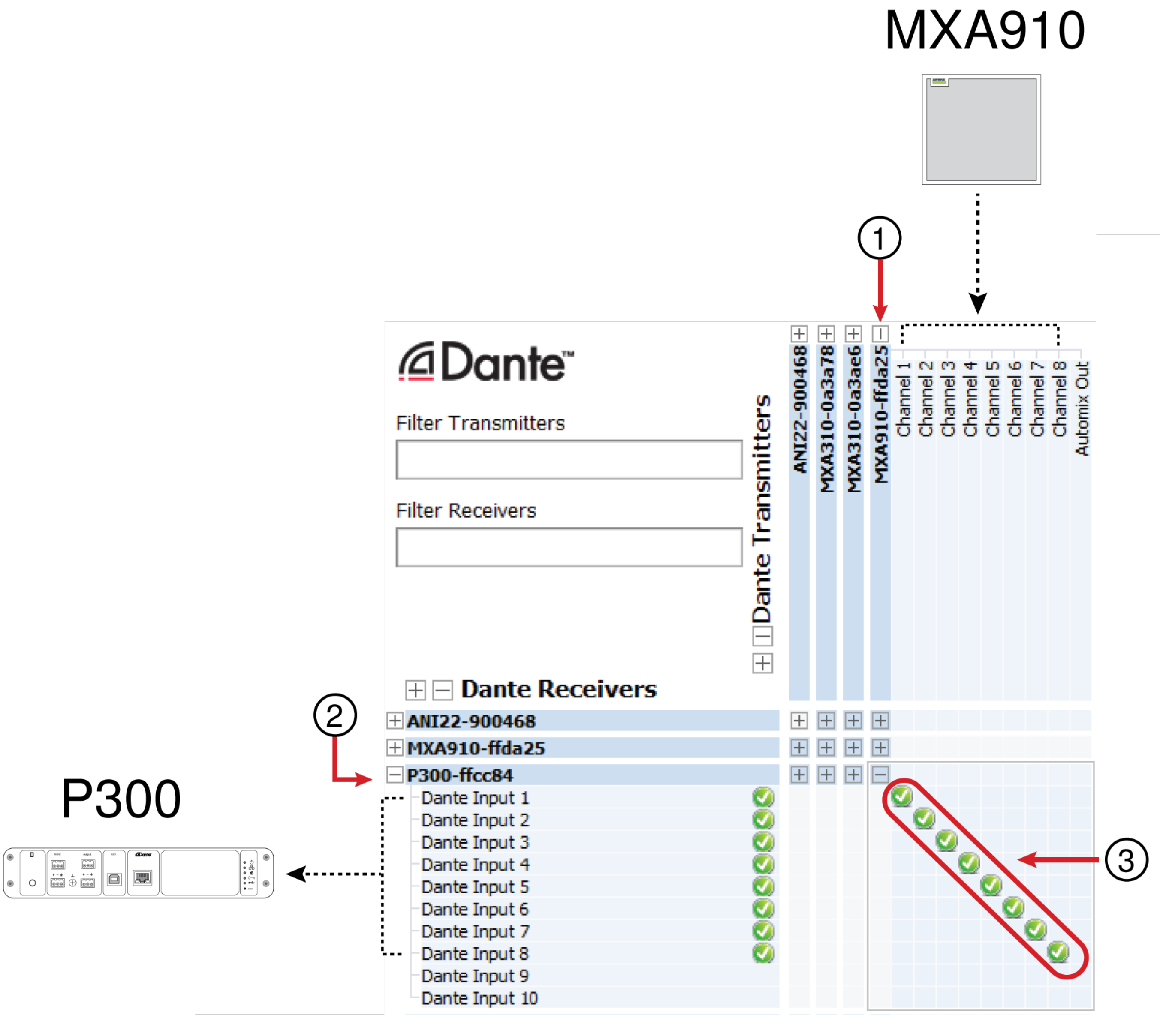


Step 3: Connect Devices in Dante Controller Software

1. Download and install Dante Controller Software from http://www.audinate.com
2. Use Dante Controller to create connections with other Dante devices.
3. **Important:** For Shure devices with integrated automatic mixing (such as the MXA910), connect independent channels to the P300 Dante input channels to ensure effective acoustic echo cancellation.

Note: Refer to the Dante Controller user guide for more information on channel routing (available at http://www.audinate.com/resources/technical-documentation)

Example: Connecting the P300 and Shure MXA910



1. Find the MXA910 in the list of Dante transmitters, and select the plus sign (+) to show all channels.
2. Find the P300 in the list of Dante receivers, and select the plus sign (+) to show all channels.
3. Connect channels 1-8 from the MXA910 to Dante input channels 1-8 on the P300. Do not use the automix output from the MXA910 into the P300.

Step 4: Configure Audio

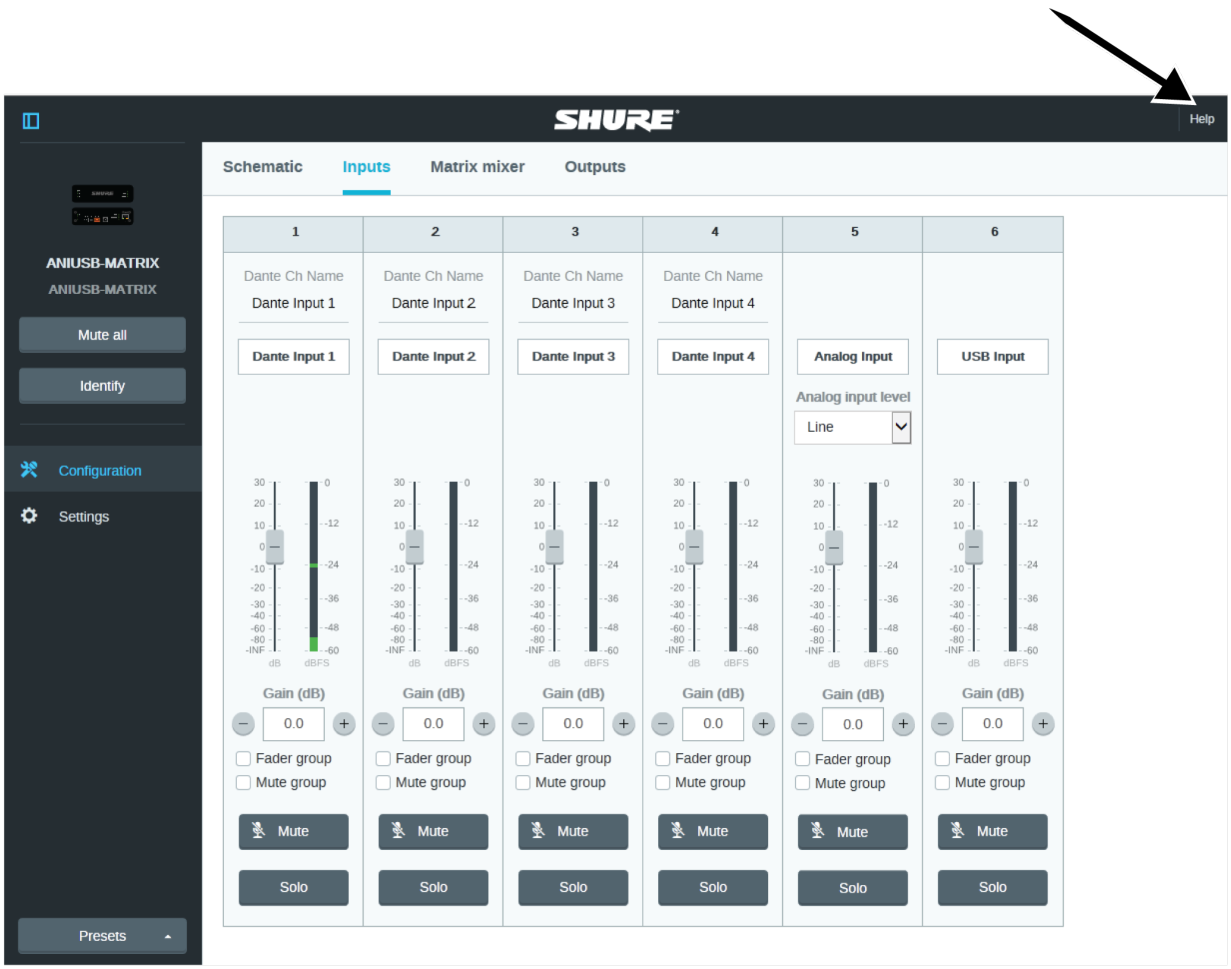
The final configuration steps will vary, depending on the signal processing required and hardware connected to the P300. These steps provide a general guideline. Specific steps are included in the system examples.

1. Connect analog, USB, and mobile audio devices

2. Route signals in the matrix mixer
3. Adjust input and output levels in the input and output tabs
4. Turn on digital signal processing blocks as needed
5. Set the AEC reference channel by opening the AEC menu in the schematic view or inputs tab and selecting a channel from the pull-down menu. Use the channel that carries audio to loudspeakers as the AEC reference. Analog -- To Speaker is the most common channel for this application, in configurations using an analog loudspeaker system or the built-in speaker on a display.

More comprehensive information is available in the help section of the web application.

Access the help section



Get More Information

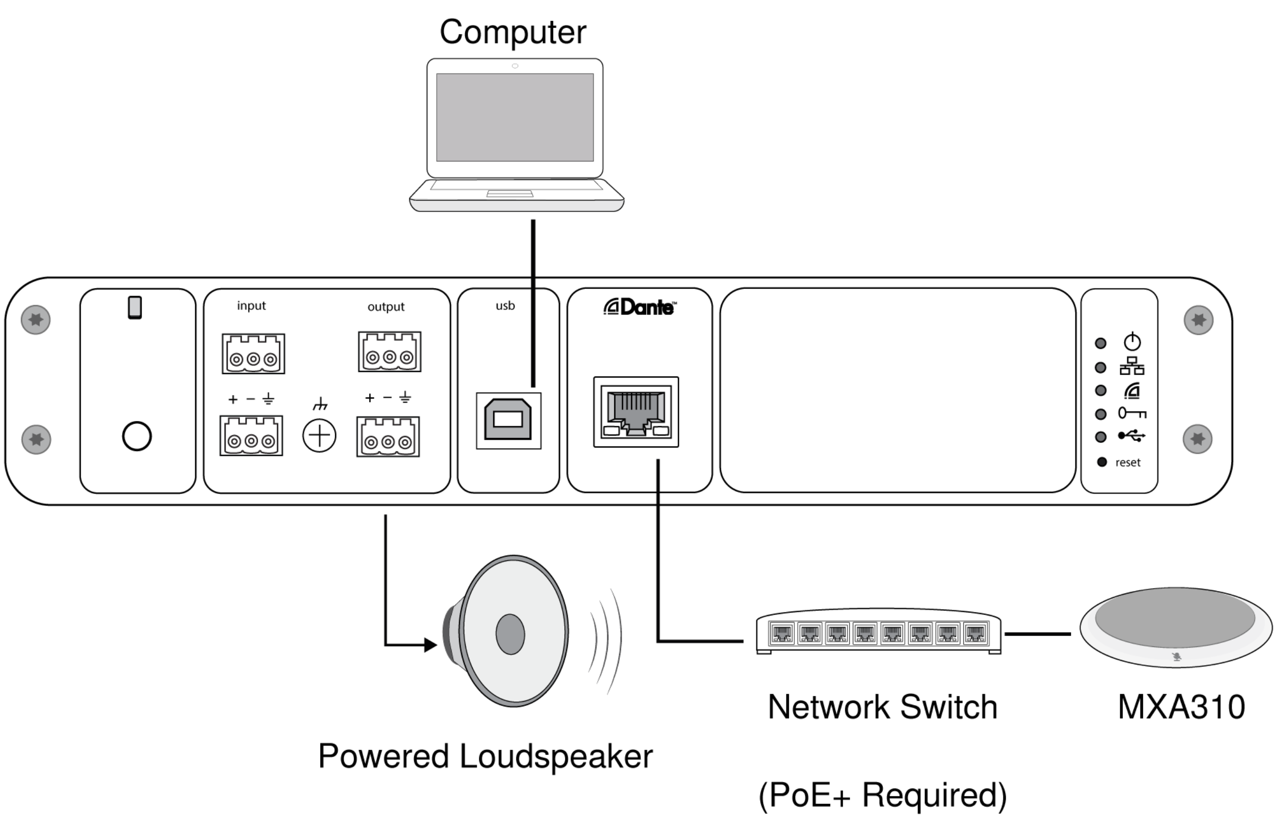
Now that the basic setup is complete, you should have access to the web application and be able to pass audio between devices. More comprehensive information is available online and in the help section, including:

- Maximizing audio quality with the built-in parametric equalizer
- External control system command strings
- Signal routing
- System scenario diagrams
- Software configuration
- Networking information
- Troubleshooting
- Replacement parts and accessories

The complete user guide is available at http://pubs.shure.com/guide/P300 (http://pubs.shure.com/guide/ANIUSB-MATRIX)

System Planning and Gear Requirements

USB Computer System



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.
3. In the P300 web application, open the matrix mixer to make connections between devices.
   Note: Some connections are established in the matrix mixer by default. Refer to the matrix mixer help topic in the web application for additional information.

   Required Matrix Mixer Connections:

Input	Output
Automix	USB output
USB input	Analog - To Speaker

4. In the schematic view, right-click any AEC block and set the AEC reference channel to Analog - To Speaker.
5. In the web application, adjust input and output levels and perform a sound check. Refer to the help topics in the web application for additional information.

Connecting a USB Device

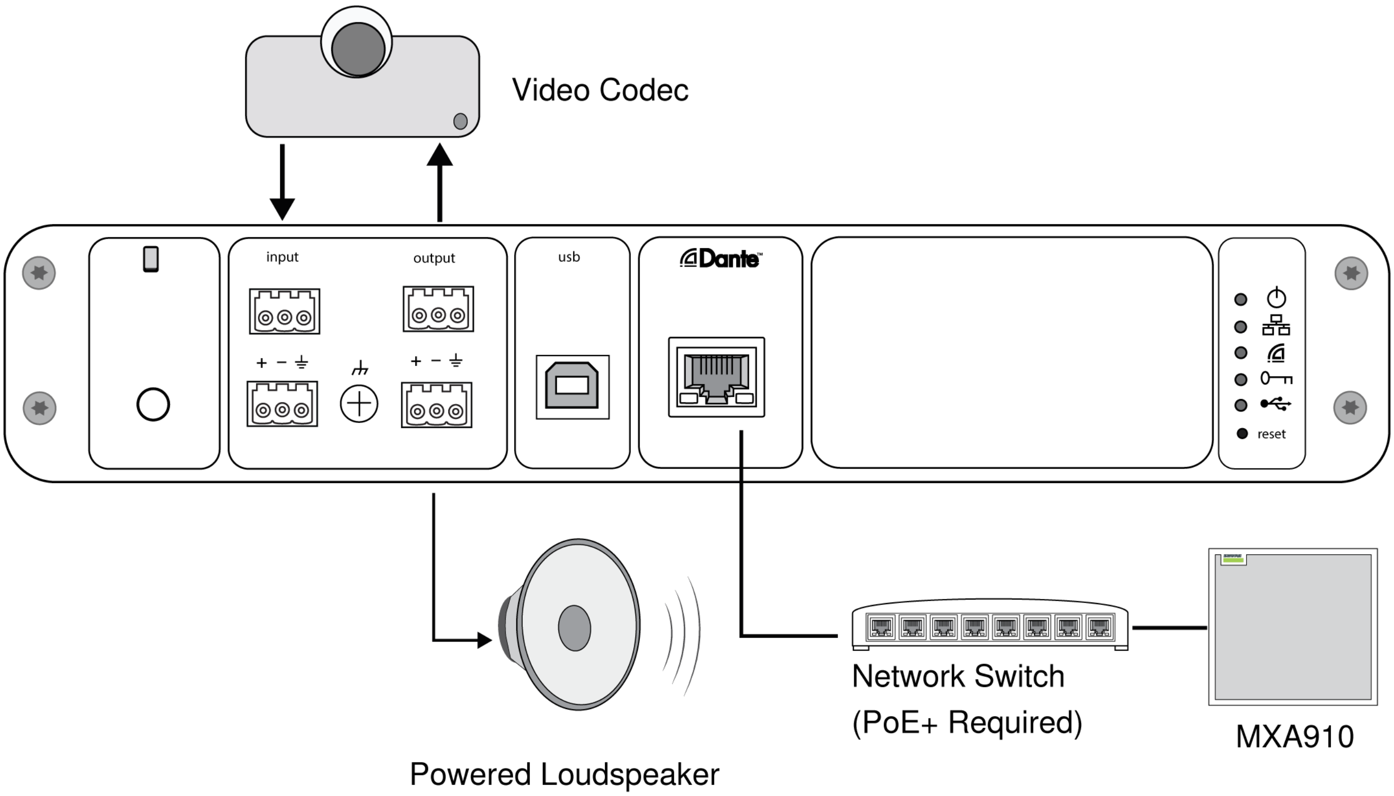
The USB port connects the host computer to the entire room audio system, including microphones and loudspeakers.

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



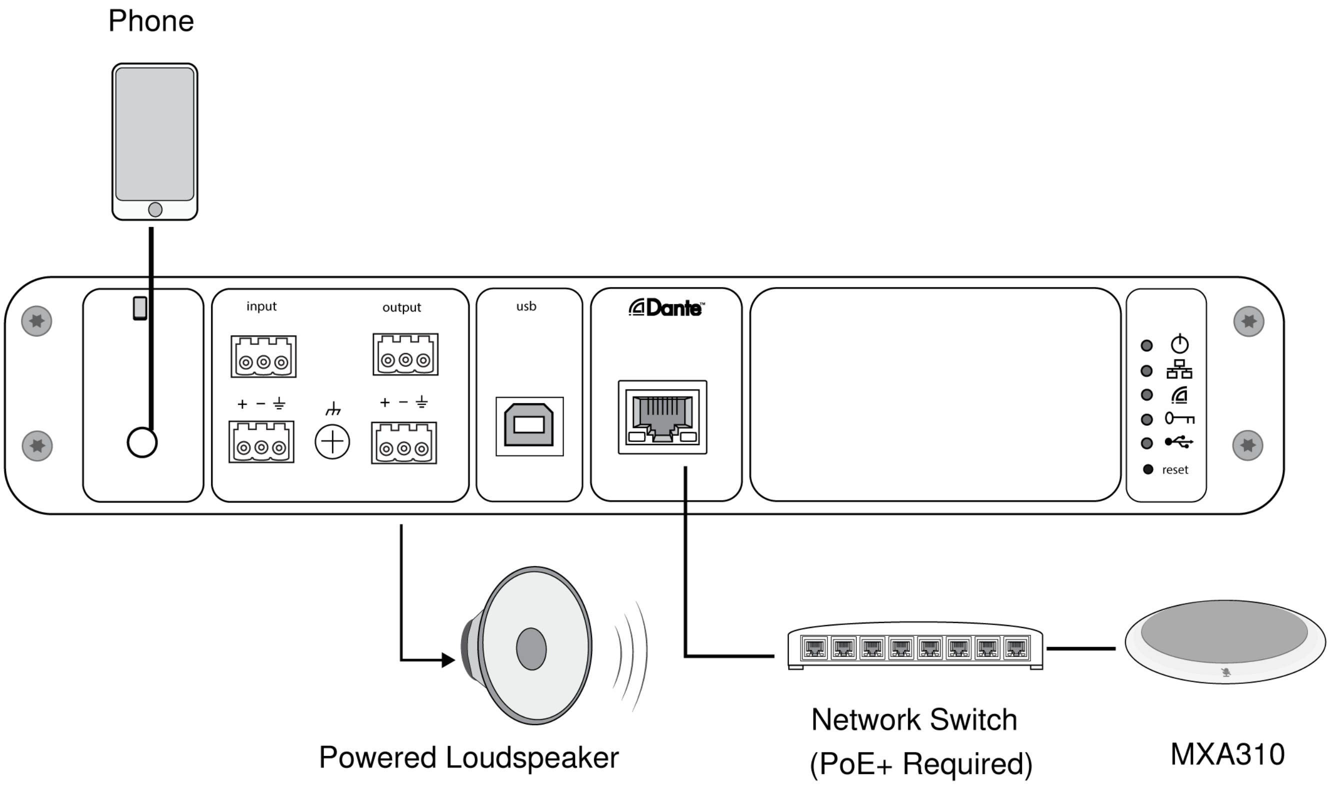
1. Connect the hardware codec audio output to the P300 analog input 1. In the matrix mixer, this is labeled Analog - From Codec.
2. Connect the hardware codec audio input to the P300 analog output 1. In the matrix mixer, this is labeled Analog - To Codec.
3. Connect a powered loudspeaker or amplifier to the P300 analog output 2. In the matrix mixer, this is labeled Analog - To Speaker.
4. In the P300 web application, open the matrix mixer to make connections between devices.
   Note: Some connections are established in the matrix mixer by default. Refer to the matrix mixer help topic in the web application for additional information.

   Required Matrix Mixer Connections:

Input	Output
Automix	Analog - To codec
Analog - From Codec	Analog - To Speaker

5. In the schematic view, right-click any AEC block and set the AEC reference channel to Analog - To Speaker.
6. In the web application, adjust input and output levels and perform a sound check. Refer to the help topics in the web application 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.
3. In the P300 web application, open the matrix mixer to make connections between devices.
   Note: Some connections are established in the matrix mixer by default. Refer to the matrix mixer help topic in the web application for additional information.

   Required Matrix Mixer Connections:

Input	Output
Automix	Mobile output
Mobile input	Analog - To Speaker

4. In the schematic view, right-click any AEC block and set the AEC reference channel to Analog - To Speaker.

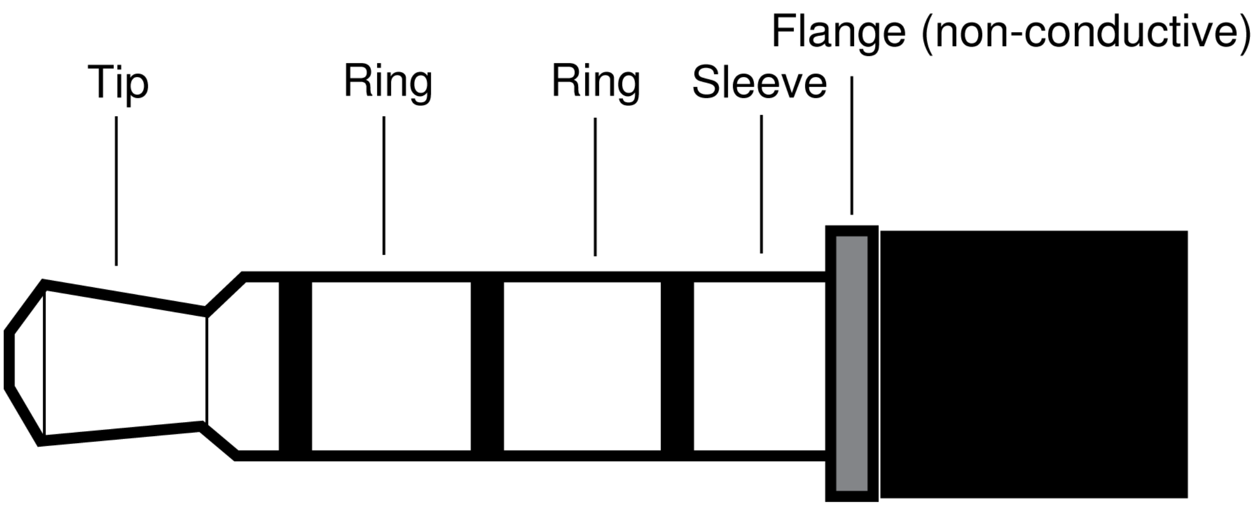
5. In the web application, adjust input and output levels and perform a sound check. Refer to the help topics in the web application for additional information.

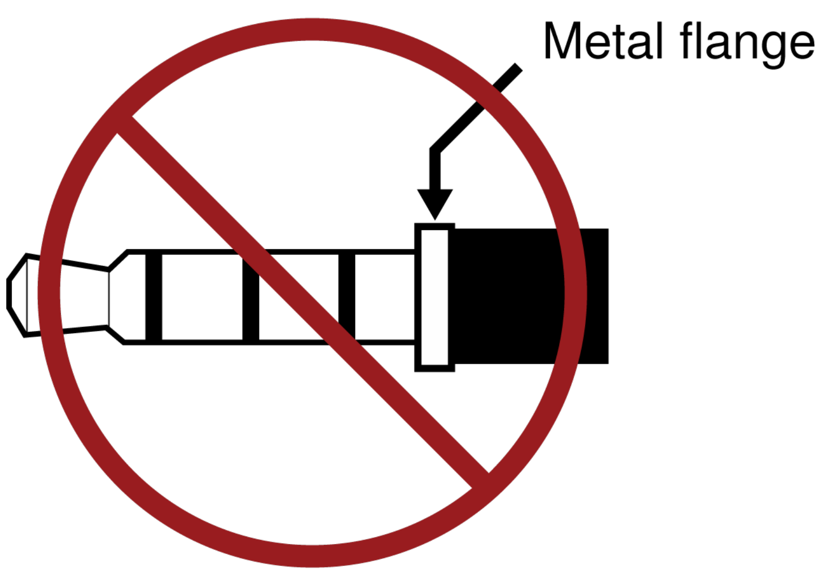
Mobile Connection Cable Requirements

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 (located in the mobile channel of the input section in the web application) must be turned off in this case.

Optimizing P300 Audio Performance with MXA910 and MXA310 Microphones

The P300 features templates for pairing with Shure microphone systems, and microphone optimization modes in the automixer to ensure best performance with all equipment combinations. Disabling the signal processing on the MXA910 and MXA310 is critical to achieve the best possible audio quality. See the process for each microphone later in this topic.

P300 Configuration

Templates

Select a template to automatically adjust all DSP settings to match the microphone. Because room acoustics are highly variable, equalizer settings are not included in templates and must be set manually.

Automix Mic Optimization Setting

Note: If using a template, the mic optimization setting is automatically entered.

Select the microphone that is used with the automixer for best performance.

Use **MXA910** or **MXA310** when using a Shure Microflex®Advance™ Ceiling Array or Table Array microphone.

Important: Disable the low-shelf filter (MXA910), low-cut filter (MXA310), and all equalization on the microphones for the best performance with mic optimization.

Use the **Off** setting when using a Shure Microflex Wireless system, or traditional wired microphones. If using wired microphones, use the Shure ANI4IN Network Interface to bring the microphones onto the Dante network.

MXA910 and MXA310 Configuration

To Optimize P300 + MXA910 Audio Performance:

1. Open the MXA910 web application
2. Disable the Low Shelf Filter
3. On every channel, disable the equalizer and set the input gain to 0
4. If equalization is needed, use the P300 equalizers

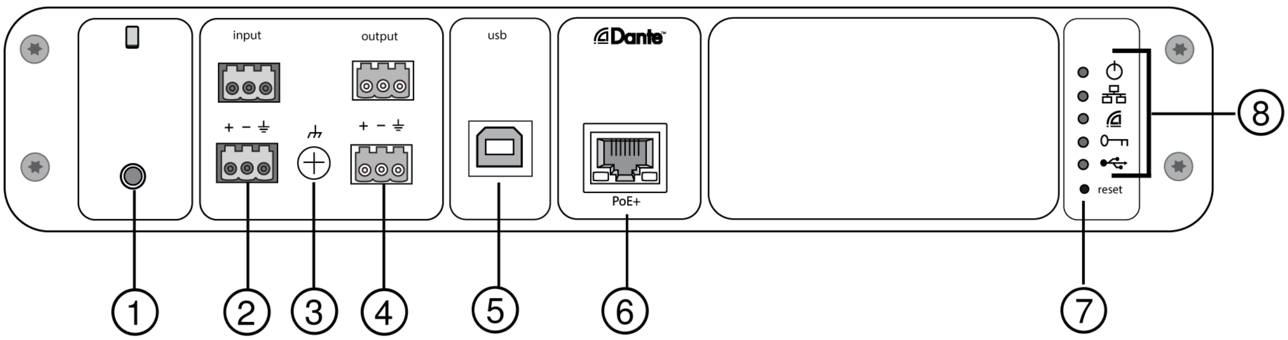
To Optimize P300 + MXA310 Audio Performance:

1. Open the MXA310 web application

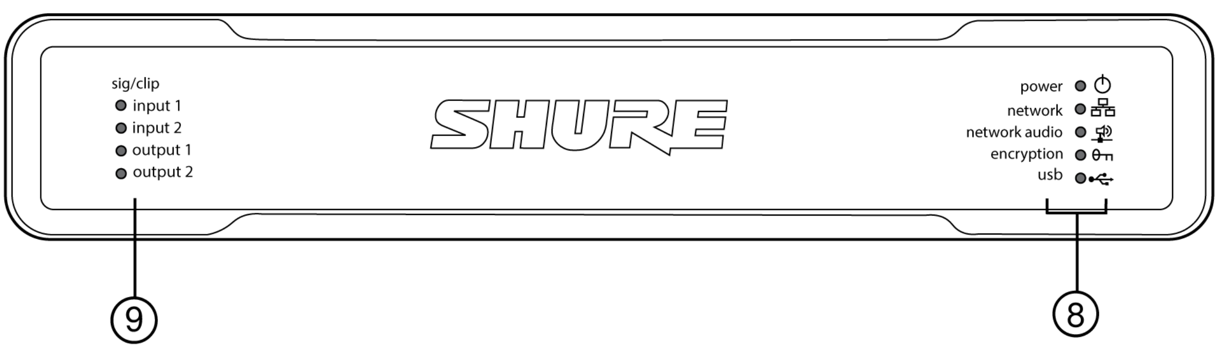
2. Disable the Low Cut Filter
3. On every channel, disable the equalizer and set the input gain to 0
4. If equalization is needed, use the P300 equalizers

Hardware and Installation

P300 Hardware



Rear Panel



Front panel

<table>
<tr><td>① Mobile Input</td><td>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 topic for additional information

Pin Assignments:
<table>
<tr><td>Tip</td><td>Audio Input (Left)</td></tr>
<tr><td>Ring 1</td><td>Audio Input (Right)</td></tr>
<tr><td>Ring 2</td><td>Ground</td></tr>
<tr><td>Sleeve</td><td>Audio Output (To Phone)</td></tr>
</table>
Note: Left and Right audio signals are summed to a mono signal.</td></tr>
<tr><td>② Audio Inputs (Block Connector)</td><td>Balanced audio input connects to an analog audio device. Set the analog input level in the web application to match the output level of the analog device. Input sensitivity:
Line (+4 dBu)

Aux (-10 dBV)

Block Pin Assignments:
<table>
<tr><td>+</td><td>Audio +</td></tr>
<tr><td>−</td><td>Audio -</td></tr>
<tr><td>⏚</td><td>Audio ground</td></tr>
</table></td></tr>
<tr><td>③ Chassis Ground Screw</td><td>Provides an optional connection for microphone shield wire to chassis ground.</td></tr>
<tr><td>④ Audio Outputs (Block Connector)</td><td>Balanced audio output connects to an analog device. Set the output level in the web application to match the input sensitivity of the analog device (Line, Aux, or Mic level).
<table>
<tr><td>+</td><td>Audio +</td></tr>
<tr><td>−</td><td>Audio -</td></tr>
<tr><td>⏚</td><td>Audio ground</td></tr>
</table></td></tr>
</table>

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 in the web application.

<table>
<tr>
<td>⑨ Level Indicators (Signal/Clip)</td>
<td>Tri-color LEDs indicate the audio signal level for the analog channels. Adjust output levels in the web application to avoid clipping.

Analog Input/Output
<table>
<tr><th>LED State</th><th>Audio Signal Level</th></tr>
<tr><td>Off</td><td>less than -60 dBFS</td></tr>
<tr><td>Green</td><td>-60 dBFS to -18 dBFS</td></tr>
<tr><td>Yellow</td><td>-18 dBFS to -6 dBFS</td></tr>
<tr><td>Red</td><td>-6 dBFS or more</td></tr>
</table>

Note: The input and output LEDs stay off when metering is set to Post Fader and the channel is muted in the web application.</td>
</tr>
</table>

Installation and Rack Mounting

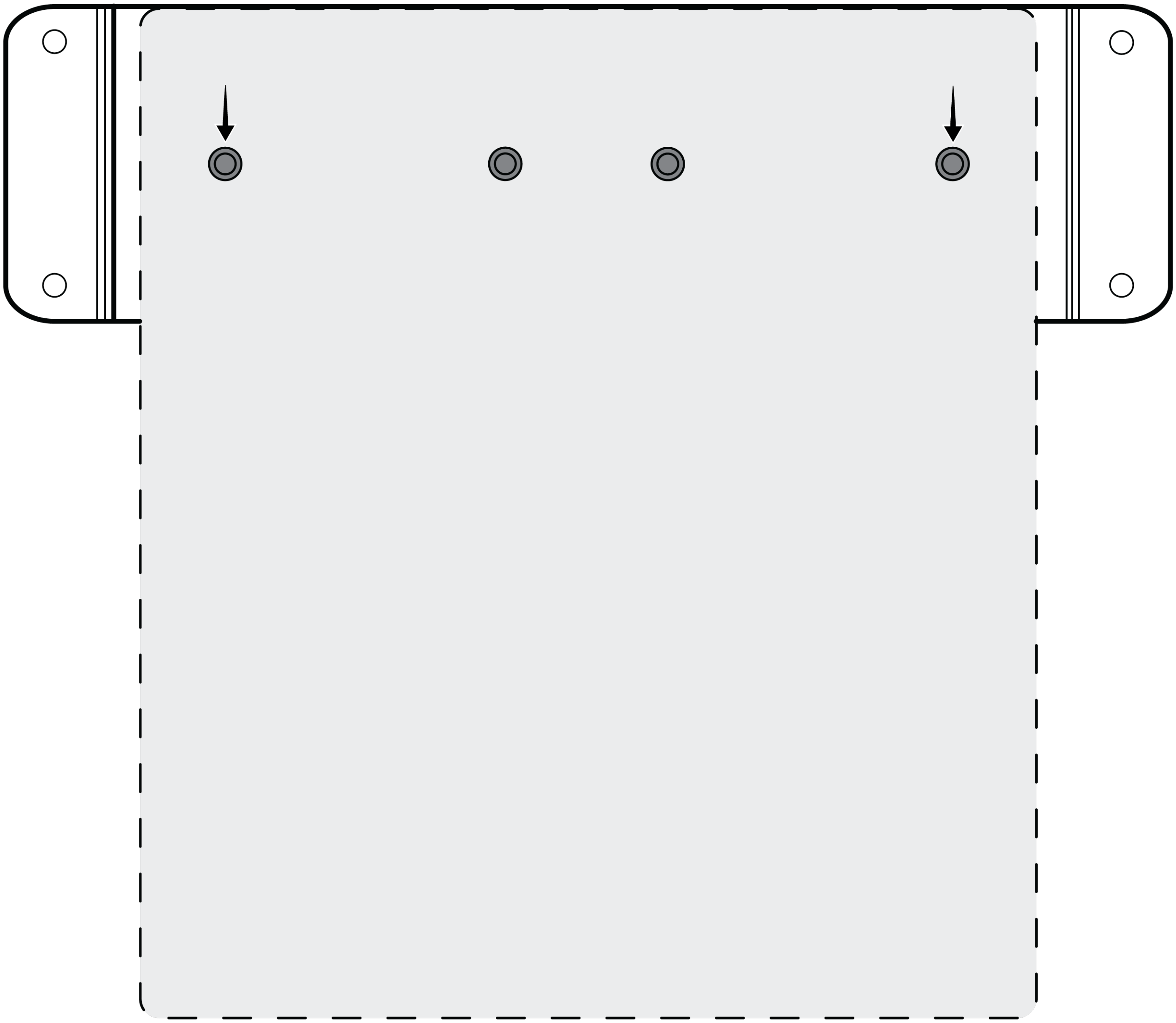
Two mounting solutions are available for installing the P300:

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

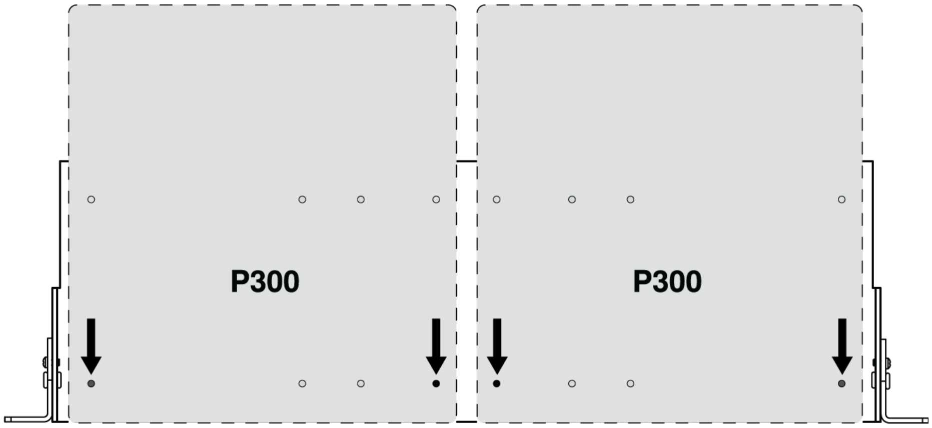
Single-unit Mounting Tray (included accessory): Supports a single device for mounting under a table

Securing the Devices

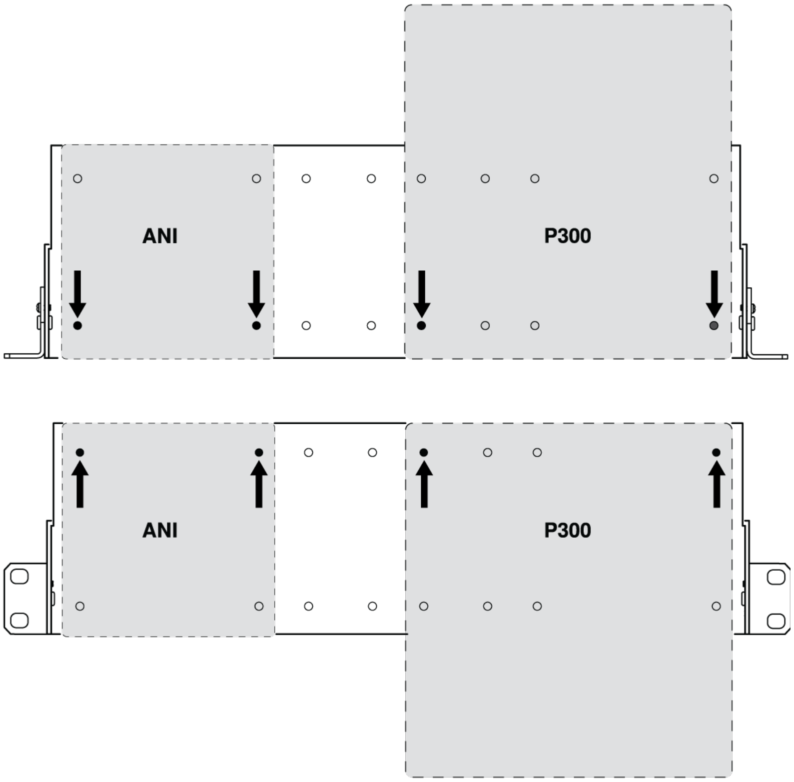
Use the included screws from the mounting hardware kit to secure each P300 or Audio Network Interface (ANI). Devices can be mounted to face either direction. Insert the screws from the bottom in the appropriate holes, according to the following diagrams:



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.

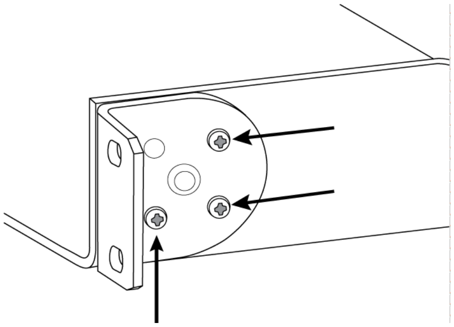


Rack Ear Configuration (CRT1)

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

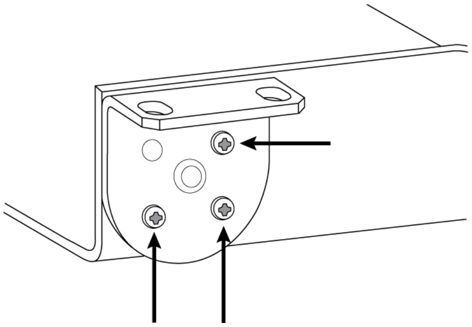
Standard 19" Rack Mount

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

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 two 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)	Restores all network and web application settings to the factory defaults.

Software Reset Options

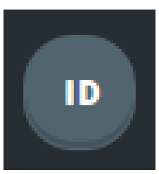
Reboot Device: In the web application (settings > factory reset), there is a Reboot Device button, which simply power-cycles the device as if it were unplugged from the network. All settings are retained when the device is re-booted.

Restore Factory Defaults: In the web application (settings > factory reset), this restores all network and web application settings to the factory defaults. This is the same as performing a full factory reset using the reset button on the device.

Default Settings Preset: To revert audio settings back to the factory configuration (excluding Device Name, IP Settings, and Passwords), select Load Preset and choose the default settings preset.

Device Identification

To identify the hardware by flashing the lights, select the Identify button in the navigation menu located on the left side of the web application.



The Identify button appears as an icon when the menu is collapsed.

Schematic View

The schematic view in the web application provides an overview of the entire 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.

Per Channel	
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 Signal Flow

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.

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

Routing Channels

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

Important: Dante devices must be routed in Dante Controller software to pass audio to or from a Dante device.

Default Setting

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 Channel
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

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.

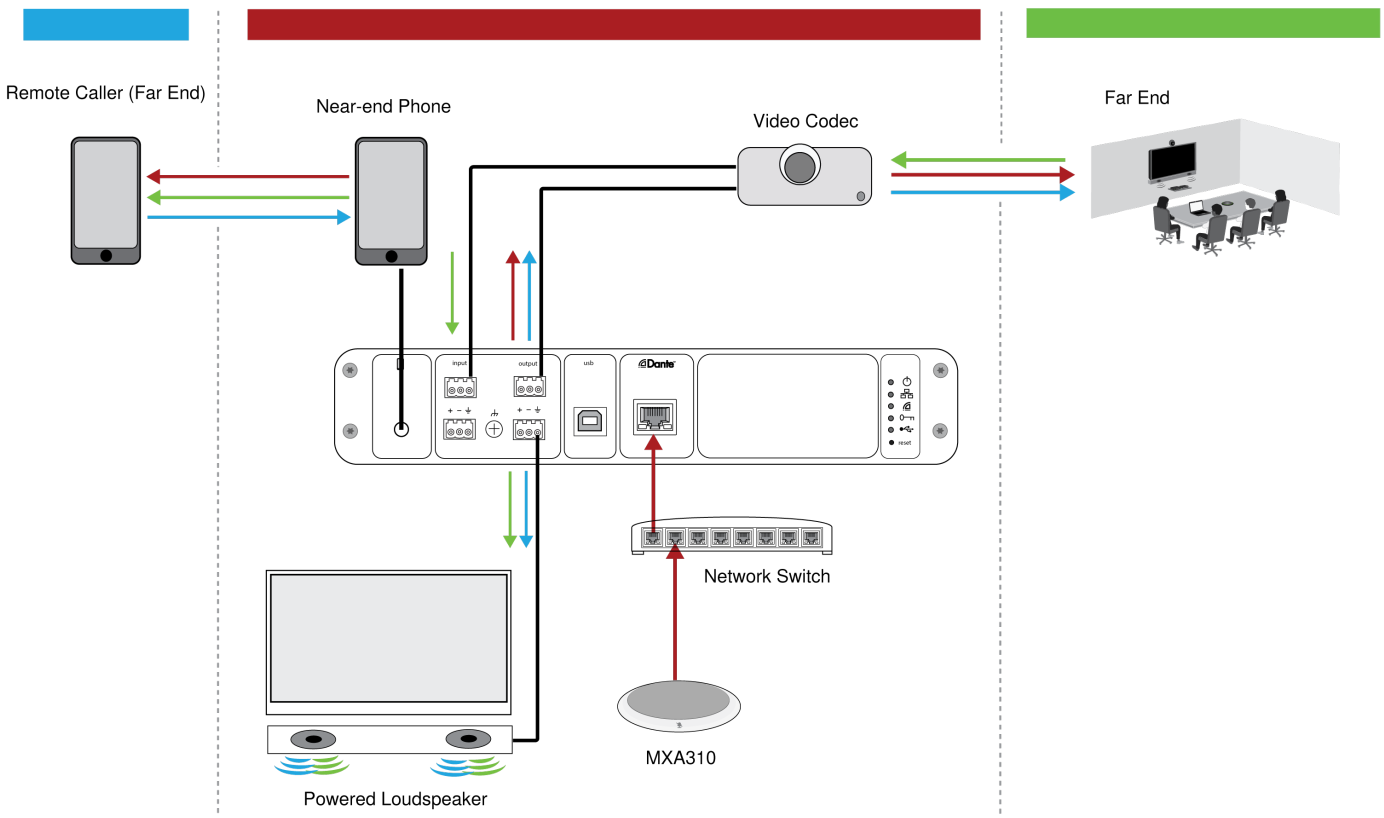
Gain staging: input fader > crosspoint gain > output fader

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

Input / Source Channel	**Output / Destination Channel**
Mobile input	Analog - To Codec (analog output 1) Analog - To Speaker (analog output 2)

Mute and Fader Groups

Mute Groups	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.
Fader Groups	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:

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

Logic

The enable logic feature on all Dante input channels makes it possible to use a microphone mute button to send a mute command to the P300. The button on the microphone triggers the P300 to mute the audio after the acoustic echo canceller and the automatic mixer, so the processors continue operating when the system needs to be muted. In many configurations, this eliminates the need for an external control system, which is often needed in a system with an acoustic echo canceller. Enabling logic on any channel activates the setting on all Dante channels (1-8).

This feature works with Shure MXA310 table array microphones and MXA910 ceiling array microphones, or with the Shure ANI4IN network interface (when paired with analog microphones that support logic functionality).

Note: the MXA910 requires an MXA310 or control system to send the logic signal to the P300.

The automixer mix out channel in the P300 web application has a mute button, which responds to the logic control device (MXA310 or control system) when logic is enabled. When the automix mute button is pressed in the P300 web application, the LED on the microphone (MXA910 or MXA310) changes to show that the system is muted.

Important MXA310 microphone configuration steps:

- The microphone logic setting must be set to external control in the MXA310 web application. This must be done for each microphone.
- If using the automix out channel from the MXA310 (instead of sending 4 individual MXA310 channels to the P300 automixer), the MXA310 LED must be set to Ring instead of Segments.

Adjusting Input levels

Levels for Dante, analog, USB, and mobile channels are adjustable in the Input tab and in the schematic view.

To monitor input levels before they reach the P300, set the metering to pre-fader in the settings menu. When adjusting the faders, set metering to post-fader.

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.
   - Levels for Microflex Advance™ and Microflex Wireless microphones are adjustable through their web application.
2. **Adjust the digital gain in the P300 web application:**
   - Use the faders or manually enter a gain value.
   - The digital gain 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 provides crosspoint gain, to adjust separate submixes for different outputs.

Analog Sources (Line Inputs)

Before you begin, verify that levels from the analog devices with adjustable output levels are operating at nominal levels. The fader adjusts the digital gain before the signal reaches the matrix mixer.

1. Match the analog input level setting according to the incoming signal level:
   Line (+4 dBu)

Aux (-10 dBV)

2. Use the fader (digital gain) to adjust the mix going to the USB or Dante output channels.

Mobile Devices

The mobile device input gain is optimized for most devices when the fader is set at zero, to provide 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 a phone at approximately 50% volume
2. Make a test call to verify the following:
    - Can the far-end talker be clearly heard?
    - Can the far-end talker clearly hear audio from the near end?

If the signal being sent to the far-end is quiet, verify the gain levels for the near-end microphones and the automixer have been properly set.

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-Fader Metering

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

Input Metering

When set to **pre-gain**, the meter displays the signal level at the inputs of the P300. If signals are too low or clipping, adjust them at the source.

When set to **post-gain**, the meter is affected by gain adjustments from the input channel faders. This does not include automatic gain control or any other processing.

Output Metering

When set to **pre-gain**, the meter displays signal levels after gain has been applied at the input stage, but before the output faders. This includes input faders, digital signal processing blocks, automixing, and crosspoint gain.

When set to **post-gain**, the meter indicated the signal sent to each output. This includes gain adjustments made on the output faders .

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.

The 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)	Use -30 dBFS as a starting point to ensure adequate headroom, and adjust if necessary. This represents the RMS (average) level, which is different from setting the input fader according to peak levels to avoid clipping.
Maximum Boost (dB)	Sets the maximum amount of gain that can be applied
Maximum Cut (dB)	Sets the maximum attenuation that can be applied

Tip: Use the boost/cut meter to monitor the amount of gain added or subtracted from the signal. If this meter is always reaching the maximum boost or cut level, consider adjusting the input fader so the signal is closer to the target level.

Parametric Equalizer

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

2-Band Equalizers

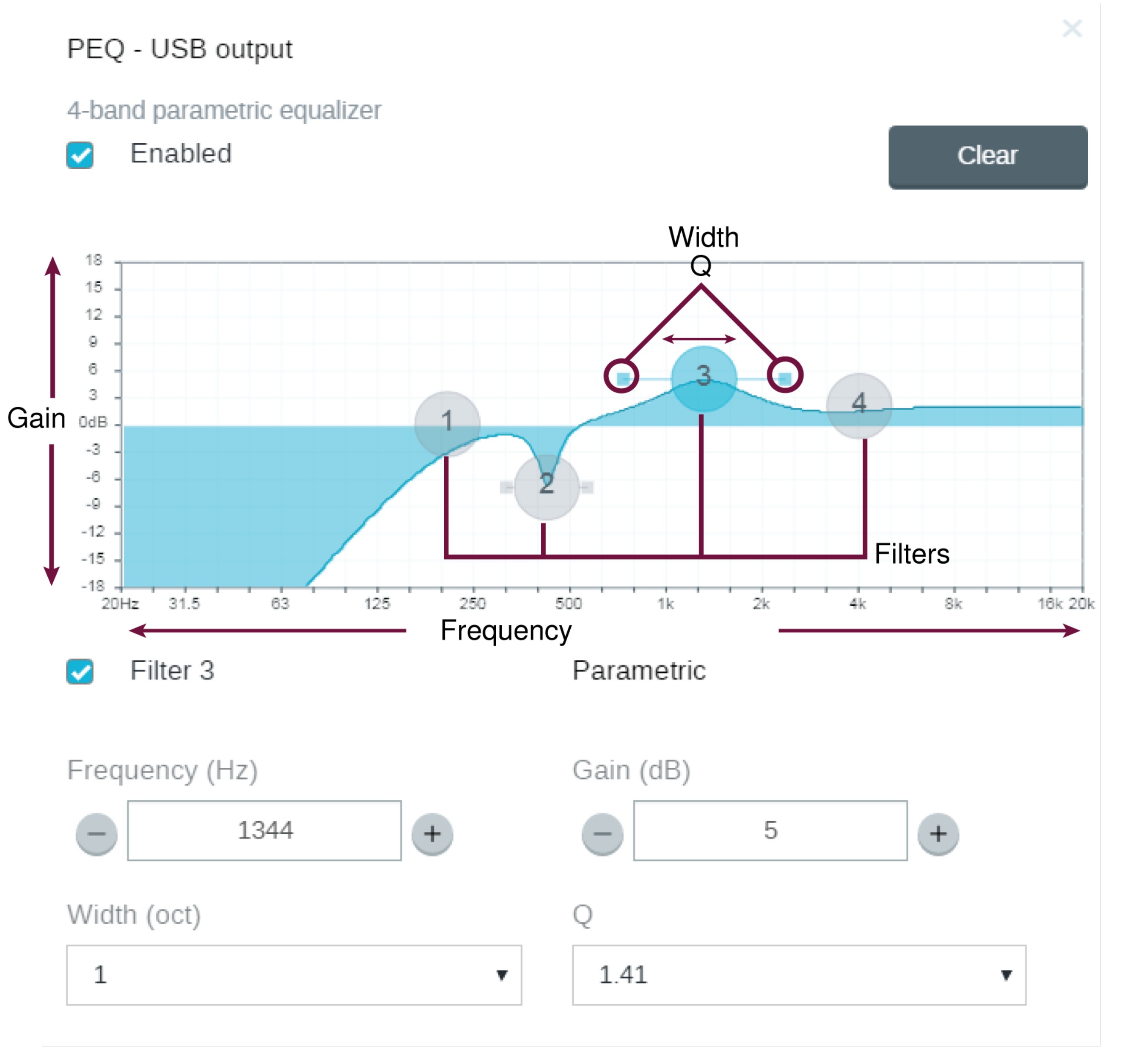
Adjust filter settings by manipulating the icons in the frequency response graph, or by entering numeric values. Disable a filter using the check-box next to the filter.

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	Only the first and last band have selectable filter types. **Parametric:** Attenuates or boosts the signal within a customizable frequency range **Low Cut:** Rolls off the audio signal below the selected frequency **Low Shelf:** Attenuates or boosts the audio signal below the selected frequency **High Cut:** Rolls off the audio signal above the selected frequency **High Shelf:** Attenuates or boosts the audio signal above the selected frequency
Frequency	Select the center frequency of the filter to cut/boost
Gain	Adjusts the level for a specific filter (+/- 30 dB)
Q	Adjusts the range of frequencies affected by the filter. As this value increases, the bandwidth becomes thinner.
Width	Adjusts the range of frequencies affected by the filter. The value is represented in octaves. **Note:** the Q and width parameters affect the equalization curve in the same way. The only difference is the way the values are represented.



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

EQ Application	Suggested Settings
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

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.

Reference Meter	Use the reference meter to visually verify the reference signal is present.
ERLE	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.
Reference	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. **Note:** Selecting a reference on any channel applies that same reference to all channels with AEC.
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. **Medium:** Use in typical rooms as a starting point. If echo artifacts appear, 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.

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 automixer output 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.

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

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.
Gate Inhibit	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 inhibitor fader. Increase the fader level so that the far end indicator turns on, and far-end audio does not gate the near-end automixer channels on. 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.

Mic Optimization Mode

Select the microphone that is used with the automixer for best performance.

Use **MXA910** or **MXA310** when using a Shure MicroflexAdvance Ceiling Array or Table Array microphone.

Important: Disable the low-shelf filter (MXA910), low-cut filter (MXA310), and all equalization on the microphones for the best performance with mic optimization.

Use the **Off** setting when using a Shure Microflex Wireless system, or traditional wired microphones. If using wired microphones, use the Shure ANI4IN Network Interface to bring the microphones onto the Dante network.

Using A Password

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

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

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

To activate encryption:

1. Open the Settings menu and select the General tab.
2. Select the Enable Encryption checkbox.
3. Enter a passphrase. All devices must use the same passphrase to establish an encrypted connection.

Important: For encryption to work:

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

Networking and Dante

Networking Best Practices

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

- Always use a "star" network topology by connecting each component directly to the switch or router.
- Connect all Shure networked devices to the **same network** and set to the **same subnet**. This applies to all devices that audio signals must be routed between (managed through Dante Controller). It is also required in order to open the web application for a device.

- Devices on separate networks require an audio processor or conferencing software to carry audio between them. See the system planning and gear requirements section for network setup information and configuration examples.
- Use only 1 DHCP server per network. Disable DHCP addressing on additional servers.
- Power on the switch and DHCP server prior to MXA equipment.
- To expand the network, use multiple Ethernet switches in a star topology.
- All devices must be at the same firmware revision level.

Network Audio and Shure Control Data

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

To configure the IP properties, follow these steps:

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

Manually Assigning Static IP Address

To manually assign IP addresses, follow these steps:

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

Operating the Control Software over Wi-Fi

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

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

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

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

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

Shure recommends the following Wi-Fi router brands:

- Cisco
- Linksys
- Apple

Digital Audio Networking

Dante[tm] 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.

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

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

Latency Recommendations

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

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 in the Shure device's web application, 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 (http://dev.audinate.com/GA/dante-controller/userguide/pdf/latest/AUD-MAN-DanteController-3.10.x-v1.0.pdf).

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.

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.

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
22	tcp	SSH	Secure Shell Interface	Closed
23	tcp	Telnet	Not supported	Closed

Port	TCP/ UDP	Protocol	Description	Factory De- fault
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†	Required for device discovery	Open
5568	udp	SDT†	Required for inter-device communication	Open
8023	tcp	Telnet	Debug console interface	Password
8180*	tcp	HTML	Required for web application	Open
8427	udp	Multcast SLP†	Required for inter-device communication	Open
64000	tcp	Telnet	Required for Shure firmware update	Open

Dante Audio & Controller

Port	TCP/UDP	Protocol	Description
162	udp	SNMP	Used by Dante
[319-320]*	udp	PTP†	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†	Used by Dante

Port	TCP/UDP	Protocol	Description
[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.

P300 Command Strings

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

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

Port: 2202

If using static IP addresses, the “ Shure Control” and the “Audio Network” settings must be set to manual in the P300 web application (`Settings > Network`). Use the Control IP address for TCP/IP communication with Shure devices.

Conventions

The device has 4 types of strings:

GET

Finds the status of a parameter. After the AMX/Crestron sends a GET command, the P300 responds with a RE-PORT string

SET

Changes the status of a parameter. After the AMX/Crestron sends a SET command, the P300 will respond with a REPORT string to indicate the new value of the parameter.

REP

When the P300 receives a GET or SET command, it will reply with a REPORT command to indicate the status of the parameter. REPORT is also sent by the P300 when a parameter is changed on the P300 or through the web application.

SAMPLE

Used for metering audio levels.

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

Most parameters will send a REPORT command when they change. Thus, it is not necessary to constantly query parameters. The P300 will send a REPORT command when any of these parameters change.

The character

“x”
in all of the following strings represents the channel of the P300 and can be ASCII numbers 0 through 4 as in the following table

00	All Channels
01-08	Dante Inputs with Mic Processing
09-10	Dante Inputs
11-12	Analog Inputs
13	USB Input
14	Mobile Input
15-16	Dante Outputs
17-18	Analog Outputs
19	USB Output
20	Mobile Output
21	Automixer Output
22	AEC Reference/Gate Inhibit Reference

Example Scenario: Muting a System

The Acoustic Echo Canceler (AEC) and P300 automixer require constant audio signal from the microphone to operate. Do NOT send commands to the microphone to mute locally. Instead, use logic communication between the P300 and Microflex Advance devices. This allows the AEC to continue processing audio even while the system is muted, and deliver the best results when the system is unmuted.

After logic functionality is set up between Shure devices, send the command from the control system to mute the P300 automixer output. If set up correctly, the P300 automixer output will mute, and the microphone LED color will change to indicate the system is muted.

Note: Although the MXA310 LED status shows the system is muted, the audio signal is still passed to the P300 to allow continuous processing.

Crestron/AMX Control System

Crestron/AMX sends the mute command to the P300.

P300

The LED command to indicate mute state is sent from the P300 to the MXA310.

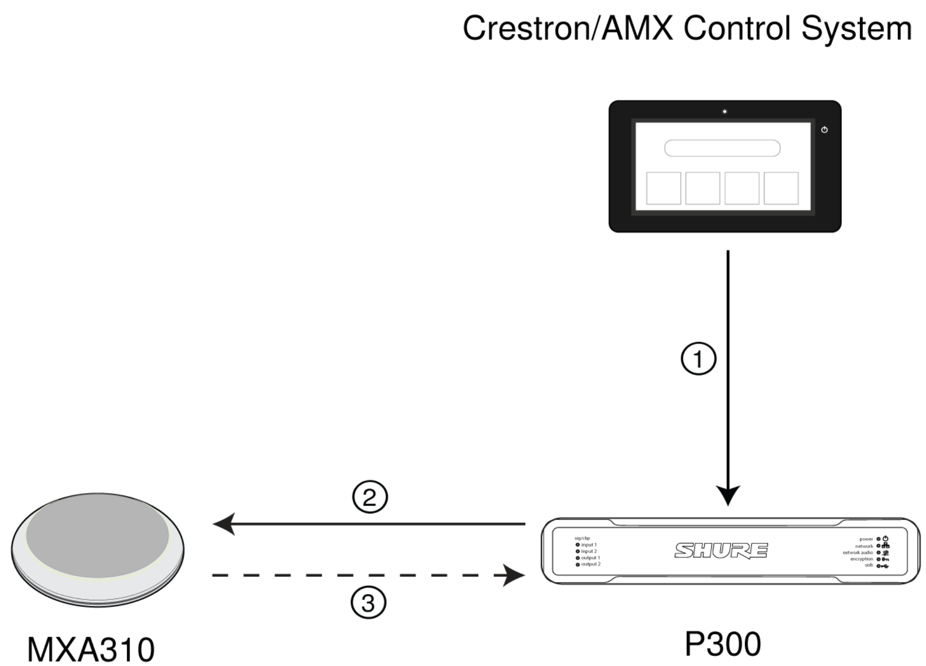
MXA310

The MXA310 sends audio to the P300 for continuous processing.

Required Steps for Logic Functionality

1. In the MXA310 web application, go to `Configuration > Button Control`, then set mode to Logic Out.
2. In the P300 web application, go to the Input tab and enable Logic for every channel routed from the MXA310 microphone. The device type appears at the bottom of the input channel strip.

Note: The MXA910 does not require set up for logic functionality.



① Mute Command

Crestron/AMX sends the mute command to the P300.

② LED Command

The P300 sends the LED command to the MXA310 so that the microphone LED color matches the system mute state.

③ Continuous Audio Signal

The MXA310 sends audio to the P300 for continuous processing. The system is muted from the P300 at the end of the audio chain.

Best Practices for Muting:

① Mute Button:

Press the mute button on the Crestron/AMX panel.

② Crestron/AMX sends following command to P300:

< SET 21 AUTOMXR_MUTE TOGGLE >

Note: The TOGGLE command simplifies logic within the Crestron/AMX. ON/OFF commands can be used instead, but supplemental processes must be implemented within the Crestron/AMX.

③ P300 Automixer channels mute, and P300 sends following REPORT back to Crestron/ AMX:

< REP 21 AUTOMXR_MUTE ON >

This REPORT command can be used in various ways for button feedback on the control surface.

Command Strings (Common)

Get All	
Command String: < GET xx ALL >	*Where xx is ASCII channel number: 00 through 21. Use this command on first power on to update the status of all parameters.*
P300 Response: < REP ... >	*The P300 responds with individual Report strings for all parameters.*
Get Model Number	
Command String: < GET MODEL >	
P300 Response: < REP MODEL {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy is 32 characters of the model number. The P300 always responds with a 32 character model number.*
Get Serial Number	

Command String: < GET SERIAL_NUM >	
P300 Response: < REP SERIAL_NUM {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where* yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy *is 32 characters of the serial number. The P300 always responds with a 32 character serial number.*
Get Channel Name	
Command String: < GET xx CHAN_NAME >	*Where xx is ASCII channel number: 00 through 20.*
P300 Response: < REP xx CHAN_NAME {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where* yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy *is 31 characters of the user name. The P300 always responds with a 31 character name.*
Get Device ID	
Command String: < GET DEVICE_ID >	*The Device ID command does not contain the x channel character, as it is for the entire P300.*
P300 Response: < REP DEVICE_ID {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where* yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy *is 31 characters of the device ID. The P300 always responds with a 31 character device ID.*
Get Firmware Version	
Command String: < GET FW_VER >	
P300 Response: < REP FW_VER {yyyyyyyyyyyyyyyyyy} >	*Where* yyyyyyyyyyyyyyyyyy *is 18 characters. The P300 always responds with 18 characters.*
Get Preset	
Command String: < GET PRESET >	

P300 Response: < REP PRESET nn >	*Where nn is the preset number 01-10.*
Set Preset	
Command String: < SET PRESET nn >	*Where nn is the preset number 1-10. (Leading zero is optional when using the SET command).*
P300 Response: < REP PRESET nn >	*Where nn is the preset number 01-10.*
Get Preset Name	
Command String: < GET PRESET1 > < GET PRESET2 > < GET PRESET3 > etc	*Send one of these commands to the P300.*
P300 Response: < REP PRESET1 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET2 {yyyyyyyyyyyyyyyyyyyyyyyyy} > < REP PRESET3 {yyyyyyyyyyyyyyyyyyyyyyyyy} > etc	*Where* yyyyyyyyyyyyyyyyyyyyyyyyy *is 25 characters of the preset name. The P300 always responds with a 25 character device ID*
Get Audio Gain	
Command String: < GET xx AUDIO_GAIN_HI_RES >	*Where xx is ASCII channel number: 00 through 22.*
P300 Response: < REP xx AUDIO_GAIN_HI_RES yyyy >	*Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.*
Set Audio Gain	
Command String: < SET xx AUDIO_GAIN_HI_RES yyyy >	*Where yyyy takes on the ASCII values of 0000 to 1400. yyyy is in steps of one-tenth of a dB.*

P300 Response: < REP xx AUDIO_GAIN_HI_RES yyyy >	*Where yyyy takes on the ASCII values of 0000 to 1400.*
Increase Audio Gain by n dB	
Command String: < SET xx AUDIO_GAIN_HI_RES INC nn >	*Where nn is the amount in one-tenth of a dB to increase the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).*
P300 Response: < REP xx AUDIO_GAIN_HI_RES yyyy >	*Where yyyy takes on the ASCII values of 0000 to 1400.*
Decrease Audio Gain by n dB	
Command String: < SET xx AUDIO_GAIN_HI_RES DEC nn >	*Where nn is the amount in one-tenth of a dB to decrease the gain. nn can be single digit (n), double digit (nn), triple digit (nnn).*
P300 Response: < REP xx AUDIO_GAIN_HI_RES yyyy >	*Where yyyy takes on the ASCII values of 0000 to 1400.*
Get Analog Input Gain Switch	
Command String: < GET xx AUDIO_IN_LVL_SWITCH >	*Where xx is ASCII channel number: 00 or 11-12.*
P300 Response: < REP xx AUDIO_IN_LVL_SWITCH LINE_LVL > < REP xx AUDIO_IN_LVL_SWITCH AUX_LVL >	*The P300 will respond with one of these strings.*
Set Analog Input Gain Switch	
Command String: < SET xx AUDIO_IN_LVL_SWITCH LINE_LVL > < SET xx AUDIO_IN_LVL_SWITCH AUX_LVL >	*Where xx is ASCII channel number: 00, 11, or 12. Send one of these commands to the P300.*
P300 Response: < REP xx AUDIO_IN_LVL_SWITCH LINE_LVL > < REP xx AUDIO_IN_LVL_SWITCH AUX_LVL >	*The P300 will respond with one of these strings.*

Get Channel Audio Mute	
Command String: < GET xx AUDIO_MUTE >	*Where xx is ASCII channel number: 00 through 20.*
P300 Response: < REP xx AUDIO_MUTE ON > < REP xx AUDIO_MUTE OFF >	*The P300 will respond with one of these strings.*
Mute Channel Audio	
Command String: < SET xx AUDIO_MUTE ON >	
P300 Response: < REP xx AUDIO_MUTE ON >	
Unmute Channel Audio	
Command String: < SET xx AUDIO_MUTE OFF >	
P300 Response: < REP xx AUDIO_MUTE OFF >	
Toggle Channel Audio Mute	
Command String: < SET xx AUDIO_MUTE TOGGLE >	
P300 Response: < REP xx AUDIO_MUTE ON > < REP xx AUDIO_MUTE OFF >	*The P300 will respond with one of these strings.*
Get Device Audio Mute	
Command String: < GET DEVICE_AUDIO_MUTE >	

P300 Response: < REP DEVICE_AUDIO_MUTE ON > < REP DEVICE_AUDIO_MUTE OFF >	*The P300 will respond with one of these strings.*
Set Device Audio Mute	
Command String: < SET DEVICE_AUDIO_MUTE ON > < SET DEVICE_AUDIO_MUTE OFF > < SET DEVICE_AUDIO_MUTE TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP DEVICE_AUDIO_MUTE ON > < REP DEVICE_AUDIO_MUTE OFF >	*The P300 will respond with one of these strings.*
Get Analog Output Gain Switch	
Command String: < GET xx AUDIO_OUT_LVL_SWITCH >	*Where xx is ASCII channel number: 00, 17, or 18.*
P300 Response: < REP xx AUDIO_OUT_LVL_SWITCH LINE_LVL > < REP xx AUDIO_OUT_LVL_SWITCH AUX_LVL > < REP xx AUDIO_OUT_LVL_SWITCH MIC_LVL >	*The P300 will respond with one of these strings.*
Set Analog Output Gain Switch	
Command String: < SET xx AUDIO_OUT_LVL_SWITCH LINE_LVL > < SET xx AUDIO_OUT_LVL_SWITCH AUX_LVL > < SET xx AUDIO_OUT_LVL_SWITCH MIC_LVL >	*Where xx is ASCII channel number: 00, 17, or 18. Send one of these commands to the P300.*
P300 Response: < REP xx AUDIO_OUT_LVL_SWITCH LINE_LVL > < REP xx AUDIO_OUT_LVL_SWITCH AUX_LVL > < REP xx AUDIO_OUT_LVL_SWITCH MIC_LVL >	*The P300 will respond with one of these strings.*
Set Flash Lights on P300	

Command String: < SET FLASH ON > < SET FLASH OFF >	*Send one of these commands to the P300. The flash automatically turns off after 30 seconds.*
P300 Response: < REP FLASH ON > < REP FLASH OFF >	*The P300 will respond with one of these strings.*
Get Flash Lights on P300	
Command String: < GET FLASH >	
P300 Response: < REP FLASH ON > < REP FLASH OFF >	*The P300 will respond with one of these strings.*
Set Metering Rate Inputs	
Command String: < SET METER_RATE_IN yyyyy >	*Where yyyyy is a value from 00000 to 99999 representing milliseconds. 00000=off; 00100=minimum value; 99999=maximum value. Note: values 00001 to 00099 are not valid and result in <REP ERR> response.*

P300 Response: < REP METER_RATE_IN yyyyy > < SAMPLE_IN aaa bbb ccc ddd eee fff ggg hhh iii jjj kkk lll mmm nnn >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 14 input channels:* *1-8: Dante Inputs with Mic Processing* *9-10: Dante Inputs* *11-12: Analog Inputs* *13: USB Input* *14: Mobile Input*
Get Metering Rate Inputs	
Command String: < GET METER_RATE_IN >	

P300 Response: < REP METER_RATE_IN yyyyy > < SAMPLE_IN aaa bbb ccc ddd eee fff ggg hhh iii jjj kkk lll mmm nnn >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 14 input channels:* *1-8: Dante Inputs with Mic Processing* *9-10: Dante Inputs* *11-12: Analog Inputs* *13: USB Input* *14: Mobile Input*
Set Metering Rate Outputs	
Command String: < SET METER_RATE_OUT yyyyy >	*Where yyyyy is a value from 00000 to 99999 representing milliseconds. 00000=off; 00100=minimum value; 99999=maximum value. Note: values 00001 to 00099 are not valid and result in <REP ERR> response.*

P300 Response: < REP METER_RATE_OUT yyyyy > < SAMPLE_OUT aaa bbb ccc ddd eee fff >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 6 output channels:* *1-2: Dante Outputs* *3-4: Analog Outputs* *5: USB Output* *6: Mobile Output*
Get Metering Rate Outputs	
Command String: < GET METER_RATE_OUT >	

P300 Response: < REP METER_RATE_OUT yyyyy > < SAMPLE_OUT aaa bbb ccc ddd eee fff >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 6 output channels:* *1-2: Dante Outputs* *3-4: Analog Outputs* *5: USB Output* *6: Mobile Output*
Set Metering Rate Processing Blocks	
Command String: < SET METER_RATE_PROC yyyyy >	*Where yyyyy is a value from 00000 to 99999 representing milliseconds. 00000=off; 00100=minimum value; 99999=maximum value. Note: values 00001 to 00099 are not valid and result in <REP ERR> response.*

P300 Response: < REP METER_RATE_PROC yyyyy > < SAMPLE_PROC aaa bbb ccc ddd eee fff ggg hhh iii jjj kkk lll >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 12 channels:* *1-8: pre-AGC on Eight Dante Input Channels* *9: Automixer Output* *10: pre-Compressor* *11: AEC reference* *12: Gate Inhibit reference*
Get Metering Rate Processing Blocks	
Command String: < GET METER_RATE_PROC >	

P300 Response: < REP METER_RATE_PROC yyyyy > < SAMPLE_PROC aaa bbb ccc ddd eee fff ggg hhh iii jjj kkk lll >	*Where yyyyy is rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the value of the audio level received and is 000-060, which represent actual audio level of -60 to 0 dBFS.* *aaais channel 1 data* *bbbis channel 2 data* *cccis channel 3 data* *dddis channel 4 data* *The sample data (aaa, bbb, ccc, ddd, etc.) appears in the following order, representing the 12 channels:* *1-8: pre-AGC on Eight Dante Input Channels* *9: Automixer Output* *10: pre-Compressor* *11: AEC reference* *12: Gate Inhibit reference*
Get LED Brightness	
Command String: < GET LED_BRIGHTNESS >	
P300 Response: < REP LED_BRIGHTNESS n >	*Where n can take on the following values:* 0 = LED disabled 1 = LED dim 2 = LED default
Set LED Brightness	

Command String: < SET LED_BRIGHTNESS n >	*Where n can take on the following values:* 0 = LED disabled 1 = LED dim 2 = LED default
P300 Response: < REP LED_BRIGHTNESS n >	
Get Audio IP Address	
Command String: < GET IP_ADDR_NET_AUDIO_PRIMARY >	
P300 Response: < REP IP_ADDR_NET_AUDIO_PRIMARY {yyyyyyyyyyyyyyy} >	*Where yyyyyyyyyyyyyyy is a 15 digit IP address.*
Get Audio Subnet Address	
Command String: < GET IP_SUBNET_NET_AUDIO_PRIMARY >	
P300 Response: < REP IP_SUBNET_NET_AUDIO_PRIMARY {yyyyyyyyyyyyyyy} >	*Where yyyyyyyyyyyyyyy is a 15 digit subnet address.*
Get Audio Gateway Address	
Command String: < GET IP_GATEWAY_NET_AUDIO_PRIMARY >	
P300 Response: < REP IP_GATEWAY_NET_AUDIO_PRIMARY {yyyyyyyyyyyyyyy} >	*Where yyyyyyyyyyyyyyy is a 15 digit gateway address.*
Get Encryption Status	
Command String: < GET ENCRYPTION >	

P300 Response: < REP ENCRYPTION ON > < REP ENCRYPTION OFF >	*The P300 will respond with one of these strings.*
Reboot P300	
Command String: < SET REBOOT >	
P300 Response: < REP REBOOT >	
Get Error Events	
Command String: < GET LAST_ERROR_EVENT >	*Gets the last error that is logged on the P300.*
P300 Response: < REP LAST_ERROR_EVENT {yyyyyyyyyyyyyyy} >	*Where yyyyyyyyyyyyyyy is up to 128 characters.*
Get PEQ Filter Enable	
Command String: < GET xx PEQ yy >	*Where xx is the PEQ block; 00 means all PEQ blocks on P300; 01-08 are PEQ blocks on Dante Mic Inputs; 09-10 are EQ blocks on Dante Inputs; 11-12 are EQ blocks on Analog Inputs; 13 is EQ block on USB Input; 14 is EQ block on Mobile Input; 15-16 are PEQ blocks on Dante Outputs; 17-18 are PEQ blocks on Analog Outputs; 19 is PEQ block on USB Output; 21 is PEQ block after autmoxer. Where yy is the filter number in the selected PEQ block xx. Valid values are as follows; 01-04: individual filter when xx is in range of 01-08, or 15-21; 01-02: individual filter when xx is in range of 09-14.*

P300 Response: < REP xx PEQ yy ON > < REP xx PEQ yy OFF >	
Set PEQ Filter Enable	
Command String: < SET xx PEQ yy ON > < SET xx PEQ yy OFF >	*Send one of these commands to the P300.*
P300 Response: < REP xx PEQ yy ON > < REP xx PEQ yy OFF >	*Where xx is the PEQ block 15-16, 17-18, 19, or 21. Where yy is the PEQ filter 01-04 within the block. 00 can be used for all blocks or all filters.*
Get Input Meter Display Mode	
Command String: < GET INPUT_METER_MODE >	
P300 Response: < REP INPUT_METER_MODE PRE_FADER > < REP INPUT_METER_MODE POST_FADER >	*The P300 will respond with one of these strings.*
Set Input Meter Display Mode	
Command String: < SET INPUT_METER_MODE PRE_FADER > < SET INPUT_METER_MODE POST_FADER >	*Send one of these commands to the P300.*
P300 Response: < REP INPUT_METER_MODE PRE_FADER > < REP INPUT_METER_MODE POST_FADER >	*The P300 will respond with one of these strings.*
Get Output Meter Display Mode	
Command String: < GET OUTPUT_METER_MODE >	

P300 Response: < REP OUTPUT_METER_MODE PRE_FADER > < REP OUTPUT_METER_MODE POST_FADER >	*The P300 will respond with one of these strings.*
Set Output Meter Display Mode	
Command String: < SET OUTPUT_METER_MODE PRE_FADER > < SET OUTPUT_METER_MODE POST_FADER >	*Send one of these commands to the P300.*
P300 Response: < REP OUTPUT_METER_MODE PRE_FADER > < REP OUTPUT_METER_MODE POST_FADER >	*The P300 will respond with one of these strings.*
Get USB Connection Status	
Command String: < GET USB_CONNECT >	
P300 Response: < REP USB_CONNECT ON > < REP USB_CONNECT OFF > < REP USB_CONNECT ERROR >	*The P300 will respond with one of these strings.*
Get Matrix Mixer Routing	
Command String: < GET xx MATRIX_MXR_ROUTE yy >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20.*
P300 Response: < REP xx MATRIX_MXR_ROUTE yy ON > < REP xx MATRIX_MXR_ROUTE yy OFF >	*The P300 will respond with one of these strings.*
Set Matrix Mixer Routing	
Command String: < SET xx MATRIX_MXR_ROUTE yy ON > < SET xx MATRIX_MXR_ROUTE yy OFF >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20. Send one of these commands to the P300.*

P300 Response: < REP xx MATRIX_MXR_ROUTE yy ON > < REP xx MATRIX_MXR_ROUTE yy OFF >	*The P300 will respond with one of these strings.*
Get Matrix Mixer Gain	
Command String: < GET xx MATRIX_MXR_GAIN yy >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20.*
P300 Response: < REP xx MATRIX_MXR_GAIN yy zzzz >	*Where zzzz takes on the ASCII values of 0000 to 1400. zzzz is in steps of one-tenth of a dB.*
Set Matrix Mixer Gain	
Command String: < SET xx MATRIX_MXR_GAIN yy zzzz >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20. Where zzzz takes on the ASCII values of 0000 to 1400. zzzz is in steps of one-tenth of a dB.*
P300 Response: < REP xx MATRIX_MXR_GAIN yyzzzz >	
Increment Matrix Mixer Gain	
Command String: < SET xx MATRIX_MXR_GAIN yy INC nn >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20. Where nn is in steps of one-tenth of a dB.*
P300 Response: < REP xx MATRIX_MXR_GAIN yy zzzz >	*Where zzzz takes on the ASCII values of 0000 to 1400. zzzz is in steps of one-tenth of a dB.*
Decrement Matrix Mixer Gain	
Command String: < SET xx MATRIX_MXR_GAIN yy DEC nn >	*Where xx is input channel numbers 21 or 9-14. Where yy is output channel numbers 15-20. Where nn is in steps of one-tenth of a dB.*

P300 Response: < REP xx MATRIX_MXR_GAIN yy zzzz >	*Where zzzz takes on the ASCII values of 0000 to 1400. zzzz is in steps of one-tenth of a dB.*
Get Control Network MAC Address	
Command String: < GET CONTROL_MAC_ADDR >	
P300 Response: < REP CONTROL_ MAC_ADDR yy:yy:yy:yy:yy:yy >	*Where yy:yy:yy:yy:yy:yy is a 17 char literal string formatted as 6 octets, each separated by a colon. Example: 00:0E:DD:FF:F1:63*
Get Network Audio Channel Name	
Command String: < GET xx NA_CHAN_NAME >	*Where xx is channel number All channels: 0 P300: 1-10, 15-16*
P300 Response: < REP xx NA_CHAN_NAME {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where xx is channel number. Where {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} is 31 char channel name. Value is padded with spaces as needed to ensure that 31 char are always reported.*
Get Network Audio Device Name	
Command String: < GET NA_DEVICE_NAME >	
P300 Response: < REP NA_DEVICE_NAME {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} >	*Where {yyyyyyyyyyyyyyyyyyyyyyyyyyyyyyy} is a text string. Most devices allow device id to be up to 31characters. Value is padded with spaces as needed to ensure that 31 char are always reported.*
Restore Default Settings	
Command String: < SET DEFAULT_SETTINGS >	*Request the device to set itself to default settings.*

P300 Response: < REP PRESET xx >	*where xx = 00 if restore is successful*
Get AEC State	
Command String: < GET xx AEC >	*Where xx is the channel number: All Dante Mic Channels: 00; P300 Dante Channel with Mic Processing: 01-08.*
P300 Response: < REP xx AEC ON > < REP xx AEC OFF >	*The P300 will respond with one of these strings.*
Set AEC State	
Command String: < SET xx AEC ON > < SET xx AEC OFF > < SET xx AEC TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx AEC ON > < REP xx AEC OFF >	*The P300 will respond with one of these strings.*
Get AEC Reference Signal	
Command String: < GET xx AEC_REF >	*Where xx is channel number that can be 00 or 22.*

P300 Response: < REP xx AEC_REF n >	*Where xx is channel number. Where n can take on the following values:* DANTEOUT1 DANTEOUT2 ANALOGOUT1 ANALOGOUT2 ANALOGIN1 ANALOGIN2 USBIN MOBILEIN
Set AEC Reference Signal	
Command String: < SET xx AEC_REF n >	*Where xx is channel number. Where n can take on the following values:* DANTEOUT1 DANTEOUT2 ANALOGOUT1 ANALOGOUT2 ANALOGIN1 ANALOGIN2 USBIN MOBILEIN

P300 Response: < REP xx AEC_REF n >	*Where xx is channel number. Where n can take on the following values:* DANTEOUT1 DANTEOUT2 ANALOGOUT1 ANALOGOUT2 ANALOGIN1 ANALOGIN2 USBIN MOBILEIN
Set ERLE Meter Rate	
Command String: < SET METER_RATE_ERLE yyyyy >	*Where yyyyy is a value from 00000 to 99999 representing milliseconds. 00000 = off; 00100 = minimum value; 99999=maximum value. Note: values 00001 to 00099 are not valid and result in <REP ERR>response.*

P300 Response: < REP METER_RATE_ERLE yyyyy > < SAMPLE_ERLE aaa bbb ccc ddd eee fff ggg hhh >	*Where yyyyy = rate in milliseconds. Value 00000 means metering is off.* *Where aaa, bbb, etc is the sample for each channel. ERLE data is in 1 dB increment and is in the range of 00 to 40 dB* aaa = channel 1 data bbb = channel 2 data ccc = channel 3 data ddd = channel 4 data eee = channel 5 data fff = channel 6 data ggg = channel 7 data hhh = channel 8 data
Get ERLE Meter Rate	
Command String: < GET METER_RATE_ERLE >	
P300 Response: < REP METER_RATE_ERLE yyyyy >	*Where yyyyy = rate in milliseconds. Value 00000 means metering is off.*
Get NLP State	
Command String: < GET xx AEC_NLP >	*Where xx is the channel number: All Dante Mic Channels: 00; P300 Dante Channel with Mic Processing: 01-08.*

P300 Response: < REP xx AEC_NLP LOW > < REP xx AEC_NLP MEDIUM > < REP xx AEC_NLP HIGH >	*The P300 will respond with one of these commands.*
Set NLP State	
Command String: < SET xx AEC_NLP LOW > < SET xx AEC_NLP MEDIUM > < SET xx AEC_NLP HIGH >	*The P300 will respond with one of these commands.*
P300 Response: < REP xx AEC_NLP LOW > < REP xx AEC_NLP MEDIUM > < REP xx AEC_NLP HIGH >	*The P300 will respond with one of these commands.*
Get Noise Reduction State	
Command String: < GET xx NOISE_RED >	*Where xx is the channel number: All Dante Mic Channels: 00; P300 Dante Channel with Mic Processing: 01-08.*
P300 Response: < REP xx NOISE_RED ON > < REP xx NOISE_RED OFF >	*The P300 will respond with one of these commands.*
Set Noise Reduction State	
Command String: < SET xx NOISE_RED ON > < SET xx NOISE_RED OFF >	*The P300 will respond with one of these commands.*
P300 Response: < REP xx NOISE_RED ON > < REP xx NOISE_RED OFF >	*The P300 will respond with one of these commands.*
Get Noise Reduction Level	

Command String: < GET xx NOISE_RED_LVL >	*Where xx is the channel number: All Dante Mic Channels: 00; P300 Dante Channel with Mic Processing: 01-08.*
P300 Response: < REP xx NOISE_LVL LOW > < REP xx NOISE_RED_LVL MEDIUM > < REP xx NOISE_RED HIGH >	*The P300 will respond with one of these commands.*
Set Noise Reduction Level	
Command String: < SET xx NOISE_RED_LVL LOW > < SET xx NOISE_RED_LVL MEDIUM > < SET xx NOISE_RED_LVL HIGH >	*The P300 will respond with one of these commands.*
P300 Response: < REP xx NOISE_RED_LVL LOW > < REP xx NOISE_RED_LVL MEDIUM > < SET xx NOISE_RED_LVL HIGH >	*The P300 will respond with one of these commands.*
Get AGC State	
Command String: < GET xx AGC >	*Where xx is the channel number: All Dante Mic Channels: 00; P300 Dante Channel with Mic Processing: 01-08.*
P300 Response: < REP xx AGC ON > < REP xx AGC OFF >	*The P300 will respond with one of these strings.*
Set AGC State	
Command String: < SET xx AGC ON > < SET xx AGC OFF > < SET xx AGC TOGGLE >	*Send one of these commands to the P300.*

P300 Response: < REP xx AGC ON > < REP xx AGC OFF >	*The P300 will respond with one of these strings.*
Set AGC Metering Rate	
Command String: < SET METER_RATE_AGC yyyyy >	*Where yyyyy is a value from 00000 to 99999 representing milliseconds. 00000 = off; 00100 = minimum value; 99999=maximum value. Note: values 00001 to 00099 are not valid and result in <REP ERR> response.*

P300 Response: < REP METER_RATE_AGC yyyyy > < SAMPLE_AGC aaa bbb ccc ddd eee fff ggg hhh >	*Where yyyyy = rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the sample for each channel. ERLE data is in 1 dB increment and is in the range of 00 to 40 dB* aaa = channel 1 data bbb = channel 2 data ccc = channel 3 data ddd = channel 4 data eee = channel 5 data fff = channel 6 data ggg = channel 7 data hhh = channel 8 data *AGC Gain data is in 1 dB increment. The reported data is scaled by 20 so the range is 00 to 40, representing an actual range of -20 to + 20 dB. -20 dB is represented as 00; 0 dB is represented as 20; +20 dB is represented as 40.*
Get AGC Metering Rate	
Command String: < GET METER_RATE_AGC >	

P300 Response: < REP METER_RATE_AGC yyyyy > < SAMPLE_AGC aaa bbb ccc ddd eee fff ggg hhh >	*Where yyyyy = rate in milliseconds. Value 00000 means metering is off. Where aaa, bbb, etc is the sample for each channel. ERLE data is in 1 dB increment and is in the range of 00 to 40 dB* aaa = channel 1 data bbb = channel 2 data ccc = channel 3 data ddd = channel 4 data eee = channel 5 data fff = channel 6 data ggg = channel 7 data hhh = channel 8 data *AGC Gain data is in 1 dB increment. The reported data is scaled by 20 so the range is 00 to 40, representing an actual range of -20 to + 20 dB. -20 dB is represented as 00; 0 dB is represented as 20; +20 dB is repre-sented as 40.*
Get AGC Max Cut Value	
Command String: < GET xx AGC_MAX_CUT >	*Where xx is channel number: All channels: 0; P300: 01-08.*
P300 Response: < REP xx AGC_MAX_CUT yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Cut data defined in SET com-mand.*

Set AGC Max Cut Value	
Command String: < SET xx AGC_MAX_CUT yyy >	*Where xx is channel number: All channels: 0; P300: 01-08. Where yyy is AGC Max Cut data in 0.1 dB increment. The actual range of -20.0 to 0.0 dB is shifted by 20.0 and then multiplied by 10 so user data has a range of 000 to 200.* -20.0 dB is represented as 000 -12.3 dB is represented as 077 -1.2 dB is represented as 188.
P300 Response: < REP xx AGC_MAX_CUT yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Cut data in 0.1 dB increment. The actual range of -20.0 to 0.0 dB is shifted by 20.0 and then multiplied by 10 so user data has a range of 000 to 200.* -20.0 dB is represented as 000 -12.3 dB is represented as 077 -1.2 dB is represented as 188.
Set AGC Max Cut Value Increment	
Command String: < SET xx AGC_MAX_CUT inc nnn >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dB. The requested is multiplied by 10 and is three digits long.* 1.2 is represented as 012 12.3 is represented as 123 *The resulting Cut when the nnn is applied must be in the range of 000-200.*
P300 Response: < REP xx AGC_MAX_CUT yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Cut data defined in SET command.*

Set AGC Max Cut Value Decrement	
Command String: < SET xx AGC_MAX_CUT dec nnn >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dB. The requested is multiplied by 10 and is three digits long.* 1.2 is represented as 012 12.3 is represented as 123 *The resulting Cut when the nnn is applied must be in the range of 000-200.*
P300 Response: < REP xx AGC_MAX_CUT yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Cut data defined in SET command.*
Get AGC Max Boost Value	
Command String: < GET xx AGC_MAX_BOOST >	*Where xx is channel number: All channels: 0; P300: 01-08.*
P300 Response: < REP xx AGC_MAX_BOOST yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Boost data defined in SET command.*
Set AGC Max Boost Value	
Command String: < SET xx AGC_MAX_BOOST yyy >	*Where xx is channel number: All channels: 0; P300: 01-08. Where yyy is AGC Max Boost data. The range is 000 to 200, representing an actual range of 0.0 to +20.0 dB in 0.1 dB increment.* +12.3 dB is represented as 123 +1.2 dB is represented as 012

P300 Response: < REP xx AGC_MAX_BOOST yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Boost data. The range is 000 to 200, representing an actual range of 0.0 to +20.0 dB in 0.1 dB increment.* +12.3 dB is represented as 123 +1.2 dB is represented as 012
Set AGC Max Boost Value Increment	
Command String: < SET xx AGC_MAX_BOOST inc nnn >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dB. The resulting Boost when the nnn is applied must be in the range of 000-200.*
P300 Response: < REP xx AGC_MAX_BOOST yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Boost data defined in SET command.*
Set AGC Max Boost Value Decrement	
Command String: < SET xx AGC_MAX_BOOST dec nnn >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dB. The resulting Boost when the nnn is applied must be in the range of 000-200.*
P300 Response: < REP xx AGC_MAX_BOOST yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Max Boost data defined in SET command.*
Get AGC Target Level	
Command String: < GET xx AGC_TARGET >	*Where xx is channel number: All channels: 0; P300: 01-08.*
P300 Response: < REP xx AGC_TARGET yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Target data defined in SET command.*

Set AGC Target Level	
Command String: < SET xx AGC_TARGET yyy >	*Where xx is channel number: All channels: 0; P300: 01-08. Where yyy is AGC Target Level data in 0.1 dBFS increment. Actual range of -50.0 to 0.0 dBFS is shifted by 50 then multiplied by 10, resulting in user data in the range of 000 to 500..* -50.0 is represented as 000 -12.3 is represented as 377 -1.2 is represented as 488
P300 Response: < REP xx AGC_TARGET yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Target Level data in 0.1 dBFS increment. Actual range of -50.0 to 0.0 dBFS is shifted by 50 then multiplied by 10, resulting in user data in the range of 000 to 500..* -50.0 is represented as 000 -12.3 is represented as 377 -1.2 is represented as 488
Set AGC Target Level Increment	
Command String: < SET xx AGC_TARGET inc nnn >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dBFS. The requested nnn is multiplied by 10 and is three digits long:* 1.2 is represented as 012 12.3 is represented as 123 *The resulting Target when the nnn is applied must be in the range 000-200.*

P300 Response: < REP xx AGC_TARGET yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Target data defined in SET command.*
Set AGC Target Level Decrement	
Command String: < SET xx AGC_TARGET dec n >	*Where xx is channel number: All channels: 0; P300: 01-08. Where nnn is in units of one-tenth of a dBFS. The requested nnn is multiplied by 10 and is three digits long:* 1.2 is represented as 012 12.3 is represented as 123 *The resulting Target when the nnn is applied must be in the range of 000-200.*
P300 Response: < REP xx AGC_TARGET yyy >	*Where xx is channel number in the range of 01-08. Where yyy is AGC Target data defined in SET command.*
Get Gate Inhibit State	
Command String: < GET GATE_INHIBIT >	
P300 Response: < REP GATE_INHIBIT ON > < REP GATE_INHIBIT OFF >	*The P300 will respond with one of these strings.*
Set Gate Inhibit State	
Command String: < SET GATE_INHIBIT ON > < SET GATE_INHIBIT OFF > < SET GATE_INHIBIT TOGGLE >	*Send one of these commands to the P300.*

P300 Response: < REP GATE_INHIBIT ON > < REP GATE_INHIBIT OFF >	*The P300 will respond with one of these strings.*
Get Automixer Mode	
Command String: < GET xx AUTOMXR_MODE >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_MODE MANUAL > < REP xx AUTOMXR_MODE GAINSHARE > < REP xx AUTOMXR_MODE GATING >	*The P300 will respond with one of these strings.*
Set Automixer Mode	
Command String: < SET xx AUTOMXR_MODE MANUAL > < SET xx AUTOMXR_MODE GAINSHARE > < SET xx AUTOMXR_MODE GATING >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_MODE MANUAL > < REP xx AUTOMXR_MODE GAINSHARE > < REP xx AUTOMXR_MODE GATING >	*The P300 will respond with one of these strings.*
Get Automixer Off Attenuation	
Command String: < GET xx AUTOMXR_OFF_ATT >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_OFF_ATT yyy >	*Where xx is automixer channel number. Where yyy is automixer off attenuation.*
Set Automixer Off Attenuation	

Command String: < SET xx AUTOMXR_OFF_ATT yyy >	*Where xx is automixer channel number. Where yyy is automixer off attenuation in 1 dB increment. Actual range of -110 to -3 dB is shifted by 110, so user data is in the range of 0 to 107.* -110 is represented as 000 -10 is represented as 100 -3 is represented as 107
P300 Response: < REP xx AUTOMXR_OFF_ATT yyy >	*Where xx is automixer channel number. Where yyy is automixer off attenuation.*
Get Automixer Gating Sensitivity	
Command String: < GET xx AUTOMXR_GATE_SEN >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_GATE_SEN y >	*Where xx is automixer channel number. Where y is automixer gating sensitivity.*
Set Automixer Gating Sensitivity	
Command String: < SET xx AUTOMXR_GATE_SEN y >	*Where xx is automixer channel number. Where y is automixer gating sensitivity in the range of 1 to 9, in increment of 1.*
P300 Response: < REP xx AUTOMXR_GATE_SEN y >	*Where xx is automixer channel number. Where y is automixer gating sensitivity.*
Set Automixer Gating Sensitivity Increment	
Command String: < SET xx AUTOMXR_GATE_SEN inc n >	*Where xx is automixer channel number. Where nis increment step. The value after n is applied cannot exceed the range of 1 to 9.*
P300 Response: < REP xx AUTOMXR_GATE_SEN y >	*Where xx is automixer channel number. Where y is automixer gating sensitivity.*

Set Automixer Gating Sensitivity Decrement	
Command String: < SET xx AUTOMXR_GATE_SEN dec n >	*Where xx is automixer channel number. Where nis decrement step. The value after n is applied cannot exceed the range of 1 to 9.*
P300 Response: < REP xx AUTOMXR_GATE_SEN y >	*Where xx is automixer channel number. Where y is automixer gating sensitivity.*
Get Automixer Maximum Number of Mics	
Command String: < GET xx AUTOMXR_MAX_NOM >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_MAX_NOM y >	*Where xx is automixer channel number. Where y is automixer Max NOM.*
Set Automixer Maximum Number of Mics	
Command String: < SET xx AUTOMXR_MAX_NOM y >	*Where xx is automixer channel number. Where y is automixer Max NOM in the range of 1 to 8, in increment of 1.*
P300 Response: < REP xx AUTOMXR_MAX_NOM y >	*Where xx is automixer channel number. Where y is automixer Max NOM.*
Get Automixer Last Mic Lock On	
Command String: < GET xx AUTOMXR_LMLO >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_LMLO ON > < REP xx AUTOMXR_LMLO OFF >	*The P300 will respond with one of these strings.*
Set Automixer Last Mic Lock On	

Command String: < SET xx AUTOMXR_LMLO ON > < SET xx AUTOMXR_LMLO OFF > < SET xx AUTOMXR_LMLO TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_LMLO ON > < REP xx AUTOMXR_LMLO OFF >	*The P300 will respond with one of these strings.*
Get Automixer Hold Time	
Command String: < GET xx AUTOMXR_HOLDTIME >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_HOLDTIME yyyy >	*Where xxis automixer channel number. Where yyyy is automixer Hold Time.*
Set Automixer Hold Time	
Command String: < SET xx AUTOMXR_HOLDTIME yyyy >	*Where xx is automixer channel number. Where yyyy is automixer Hold Time in the range of 0100 to 1500 ms, in increment of 1 ms.*
P300 Response: < REP xx AUTOMXR_HOLDTIME yyyy >	*Where xx is automixer channel number. Where yyyy is automixer Hold Time.*
Get Automixer Gating Optimization	
Command String: < GET xx AUTOMXR_GATE_OPT >	*Where xx is the automixer channel number, 00 or 21.*
P300 Response: < REP xx AUTOMXR_GATE_OPT NORMAL > < REP xx AUTOMXR_GATE_OPT MXA310 > < REP xx AUTOMXR_GATE_OPT MXA910 >	*Where xx is automixer channel number. Where yyyy is automixer Hold Time.*
Set Automixer Gating Optimization	

Command String: < SET xx AUTOMXR_GATE_OPT NORMAL > < SET xx AUTOMXR_GATE_OPT MXA310 > < SET xx AUTOMXR_GATE_OPT MXA910 >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_GATE_OPT NORMAL > < REP xx AUTOMXR_GATE_OPT MXA310 > < REP xx AUTOMXR_GATE_OPT MXA910 >	*The P300 will respond with one of these strings.*
Get Automixer Channel Always On	
Command String: < GET xx AUTOMXR_ALWAYS_ON >	*Where xx is the channel number: All: 00; P300: 01-08.*
P300 Response: < REP xx AUTOMXR_ALWAYS_ON ON > < REP xx AUTOMXR_ALWAYS_ON OFF >	*The P300 will respond with one of these strings.*
Set Automixer Channel Always On	
Command String: < SET xx AUTOMXR_ALWAYS_ON ON > < SET xx AUTOMXR_ALWAYS_ON OFF > < SET xx AUTOMXR_ALWAYS_ON TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_ALWAYS_ON ON > < REP xx AUTOMXR_ALWAYS_ON OFF >	*The P300 will respond with one of these strings.*
Get Automixer Channel Priority	
Command String: < GET xx AUTOMXR_PRIORITY >	*Where xx is the channel number: All: 00; P300: 01-08.*
P300 Response: < REP xx AUTOMXR_PRIORITY ON > < REP xx AUTOMXR_PRIORITY OFF >	*The P300 will respond with one of these strings.*

Set Automixer Channel Priority	
Command String: < SET xx AUTOMXR_PRIORITY ON > < SET xx AUTOMXR_PRIORITY OFF > < SET xx AUTOMXR_PRIORITY TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_PRIORITY ON > < REP xx AUTOMXR_PRIORITY OFF >	*The P300 will respond with one of these strings.*
Get Automixer Post Gate Mute	
Command String: < GET xx AUTOMXR_MUTE >	*Where xx is the channel number and can only be channel 00 or 21.*
P300 Response: < REP xx AUTOMXR_MUTE ON > < REP xx AUTOMXR_MUTE OFF >	*The P300 will respond with one of these strings.*
Set Automixer Post Gate Mute	
Command String: < SET xx AUTOMXR_MUTE ON > < SET xx AUTOMXR_MUTE OFF > < SET xx AUTOMXR_MUTE TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx AUTOMXR_MUTE ON > < REP xx AUTOMXR_MUTE OFF >	*The P300 will respond with one of these strings.*
Get Automixer Gate Status	
Command String: < GET xx AUTOMXR_GATE >	*Where xx is the channel number: All: 00; Individual Channel: 01-08; Gate Inhibit Channel: 22.*

P300 Response: < REP xx AUTOMXR_GATE ON > < REP xx AUTOMXR_GATE OFF >	*The P300 will respond with one of these strings.*
Get Compressor State	
Command String: < GET xx COMPRESSOR >	*Where xx is the channel number 00 or 21.*
P300 Response: < REP xx COMPRESSOR ON > < REP xx COMPRESSOR OFF >	*The P300 will respond with one of these strings.*
Set Compressor State	
Command String: < SET xx COMPRESSOR ON > < SET xx COMPRESSOR OFF > < SET xx COMPRESSOR TOGGLE >	*Send one of these commands to the P300.*
P300 Response: < REP xx COMPRESSOR ON > < REP xx COMPRESSOR OFF >	*The P300 will respond with one of these strings.*
Get Compressor Threshold	
Command String: < GET xx COMP_THRESHOLD >	*Where xx is the channel number 00 or 21.*
P300 Response: < REP xx COMP_THRESHOLD yyy >	*Where xx is the channel number 00 or 21. Where yyy is Compressor Threshold data, as defined in SET command.*
Set Compressor Threshold	

Command String: < SET xx COMP_THRESHOLD yyy >	*Where xx is the channel number 00 or 21.Where yyy is Compressor Threshold data in 0.1 dB increment. Actual range of -60.0 to 0.0 dB is shifted by 60 then multiplied by 10, resulting in user data in the range of 000 to 600.* -60.0 is represented as 000 -12.3 is represented as 477 -1.2 is represented as 588
P300 Response: < REP xx COMP_THRESHOLD yyy >	*Where xx is the channel number 00 or 21. Where yyy is Compressor Threshold data, as defined in SET command.*
Get Compressor Ratio	
Command String: < GET xx COMP_RATIO >	*Where xx is the channel number 00 or 21.*
P300 Response: < REP xx COMP_RATIO yyyy >	*Where xx is the channel number 00 or 21. Where yyyy is Compressor Ratio data, as defined in SET command.*
Set Compressor Ratio	
Command String: < SET xx COMP_RATIO yyyy >	*Where xx is the channel number 00 or 21.Where yyyy is Compressor Ratio data in 0.1 increment. Ratio data is in the range of 1000 to 0010, representing actual range of 100.0:1 to 1.0:1 ratio.* 100.0:1 is represented as 1000 12.3:1 is represented as 0123 1.2:1 is represented as 0012
P300 Response: < REP xx COMP_RATIO yyyy >	*Where xx is the channel number 00 or 21. Where yyyy is Compressor Ratio data, as defined in SET command.*

Get Delay	
Command String: < GET xx DELAY >	*Where xxis the channel number :* *All Channels: 0* *Analog Out 1: 17* *Analog Out 2: 18* *USB Out: 19*
P300 Response: < REP xx DELAY yyyy >	*Where xx is channel number defined in GET command. Where yyyy is Delay, as defined in SET command.*
Set Delay	
Command String: < SET xx DELAY yyyy >	*Where xx is channel number defined in the GET command. Where yyyyis Delay data in 1 ms increment. Delay is in the range of 0 to 1000 ms, 0 means Delay unit is disabled.*
P300 Response: < REP xx DELAY yyyy >	*Where xx is channel number defined in GET command. Where yyyy is Delay, as defined in SET command.*

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
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.
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

Problem	Solution
No audio	• Verify the ANIUSB-MATRIX 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 • 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

Important Product Information

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

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

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

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

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

Authorized under the verification provision of FCC Part 15B.

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

Information to the user

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

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

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

Authorized European representative:

Shure Europe GmbH

Headquarters Europe, Middle East & Africa

Department: EMEA Approval

Jakob-Dieffenbacher-Str. 12

75031 Eppingen, Germany

Phone: +49-7262-92 49 0

Fax: +49-7262-92 49 11 4

Email: info@shure.de

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

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

Specifications

Analog Connections

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 input channels, 2 output channels

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 application

HTML5 Browser-based

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	44.1, 48 kHz
Bit Depth	16, 24

Latency

Does not include Dante latency	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

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.

Accessories

Furnished Accessories

KIT, HARDWARE, P300-IMX	90D33522
BRACKET, HALF RACK UNIT	53A27741

Optional Accessories and Replacement Parts

19" rack tray	CRT1