Dante 4-Channel Audio DSP

User Manual

500556-V2

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1. Safety Precautions

To ensure the best performance from the product, please read all instructions carefully before using the device. Save this manual for future reference.

- Follow basic safety precautions to reduce the risk of fire, electrical shock, and injury.
- Do not dismantle the housing or modify the module. It may result in electrical shock or burns.
- Do not open or remove the housing of the device as you may be exposed to dangerous voltage or other hazards.
- To prevent fire or shock hazard, do not expose the unit to rain, moisture and do not install this product near water. Keep the product away from liquids.
- Spillage into the housing may result in fire, electrical shock, or equipment damage. If an object or liquid falls or spills on the housing, unplug the module immediately.
- Do not use liquid or aerosol cleaners to clean this unit. Always unplug the power to the device before cleaning.
- Using supplies or parts not meeting the product specifications may cause damage, deterioration or malfunction.
- Refer all servicing to qualified service personnel.
- Install the device in a place with adequate ventilation to avoid damage caused by overheat.
- Unplug the power when left unused for a long period of time.
- Information on disposal of devices: do not burn or mix with general household waste, please treat them as normal electrical waste.

2. Introduction

The Dante 4-Channel Audio DSP (Model 500556-V2) is a product developed for the transmission, routing, and processing of audio signals in Dante networks. It can perform high-quality, low-delay audio transmission, supports 4-way input and output balanced analog channels, and can be powered via PoE from any PoE network switch or via a 12V DC power supply. This processor supports 4-way analog audio signal balance input and output, 4-way Dante digital audio signal input, and the 4-way Dante digital audio signal output. It also features a USB audio card that supports one channel of audio input and one channel of audio output. The device has one of the best DSP algorithms in the industry that includes options such as AEC, AFC, ANC, Ducker, Compressor, Limiter, Mixer and Graphic Equalization among others; In addition, the product provides PC version control software to monitor and operate all the functions on the DSP in an easy and intuitive way.

3. Features

- Built-in Dante module
- Supports PoE power from PoE network switches or via a 12VDC power supply
- Features 4 balanced inputs and 4 balanced outputs
- Features a USB sound card
- The input supports 48VDC phantom power supply
- DSP processing functions include gain adjustment, stage parameter equalization, compressor, ducker, mixer, graphic equalization, limiter, setting, etc
- Provides software for the Windows platform to manage all audio operations
- API provided for RS232, RS485 and TCP/IP control

4. Package Contents

- One (1) Dante 4-Channel Audio DSP
- Two (2) Cabinet mounting bracket
- Four (4) 6 pin connection plug
- Two (2) 3 pin connection plug
- One (1) Screwdriver
- One (1) User manual (available via download)

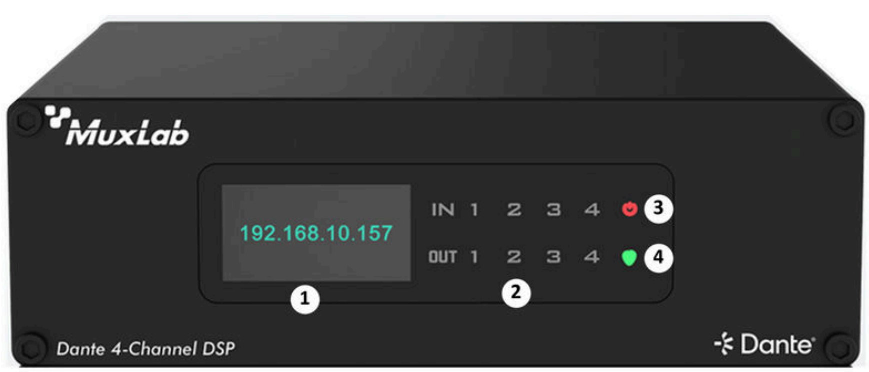
Notes: Confirm that the product and accessories are all included. If not, please contact the supplier from which you purchased the unit.

5. Specifications

Specification	
Analog Audio	4 balanced inputs + 4 balanced outputs
Dante Audio	4 inputs + 4 outputs
USB Sound Card	1 input + 1 output
Analog Maximum Gain	36dB
Quantization Bits	24bit
Sample Rate	48k
Phantom Power	48VDC
Frequency Response	20Hz~20KHz±0.15dB
THD+N	≤0.003% @ 4dBu
Digital/Analog Dynamic Range	114dB
Analog/Digital Dynamic Range	120dB
Input Impedance	20kΩ Balanced，10kΩ Unbalanced
Output Impedance	100Ω Balanced，50Ω Unbalanced
Maximum Input Level	+18dBu，Balanced
Maximum Output Level	+18dBu，Balanced
Input Common Mode Rejection	52dB @ 60Hz
Channel Isolation	>100dB
Background Noise	-89dBu
Number of RS485 serial ports	1
Number of RS232 Serial Ports	1
Number of USB sound cards	1
Working Temperature	0℃ - 40℃
Working Power Supply	2VDC/PoE power supply
Warranty	2 years
Order Information	500556-V2 Dante 4-Channel Audio DSP (UPC: 627699015568)

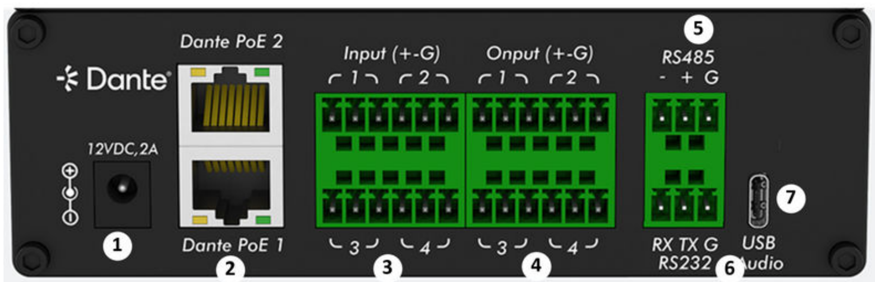
6. Hardware Interface

6.1 Front Panel



1. LED display: after normal operation of the system, the current IP address information of the device will be displayed.
2. Channel indicator: when the analog IN/OUT channel flows into a very small signal, the indicator is not lit; when the analog IN/OUT channel flows into a very large signal, the indicator is red; in other cases, the indicator is green.
3. Power indicator light: when the device is powered on, the power indicator is always on.
4. System indicator light: When the device is in normal operation, this indicator shows a flashing state

6.2 Rear Panel



1. 12VDC: DC power supply interface.
2. Dante POE network power supply interface: this network interface can be used for Dante digital signal transmission, for connecting to PC interactive control software and as a direct power supply to the device.
3. INPUT signal input interface: can be connected to microphone, DVD and other audio devices.
4. OUTPUT signal output interface: can be connected to amplifiers, active speakers and other audio devices
5. RS485 interface: RS485 balanced interface.
6. RS232 interface: RS232 balanced interface.
7. USB interface: support USB sound card function.

7 Software Interface

7.1 Software Download

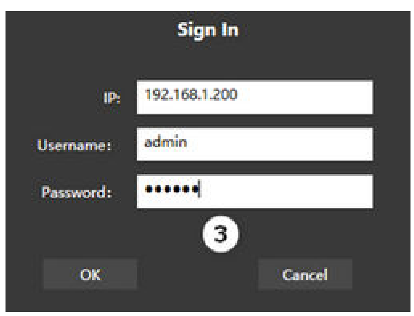
The source file of the installation software is embedded in the device. To download the software, you only need to enter the factory default IP address (default IP: 192.168.1.200 Subnet Mask: 255.255.255.0) in the browser url address bar, and press Enter to navigate to the download interface. According to the content information of the web interface. Just click to download the software, and please note that before installing the PC software, please ensure that the PC client has installed Microsoft .Net Framework 3.5 or above.

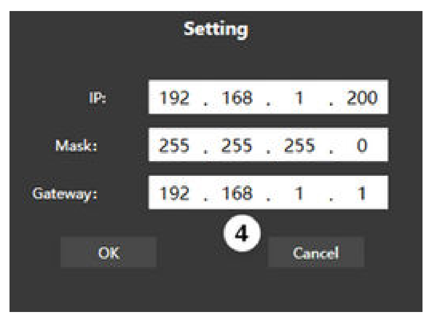
Note: Make sure the IP address of the client PC and the device are on the same network segment; otherwise it will not be accessible.

Install the "Dante 4-Channel DSP" software you just downloaded so you can control all the DSP options.

7.2 Connecting to the unit

Main Interface



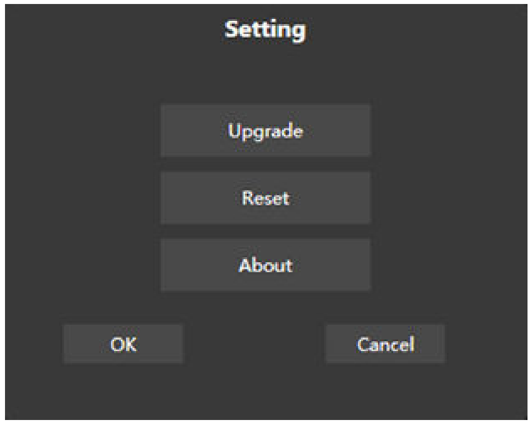


Steps:

1. Log in and from the main interface, click search (1) to discover DSP units on the network.
2. Double-click the device name in the device list, the Sign In login window will pop up.
3. Enter the user name and password: (default user name: admin, password: 123456) click the [OK] button to enter.
4. If you need to modify the device IP address information, right-click Device name, and the Setting window will pop up, and you can change it according to the requirements of the use environment.

7.3 Settings

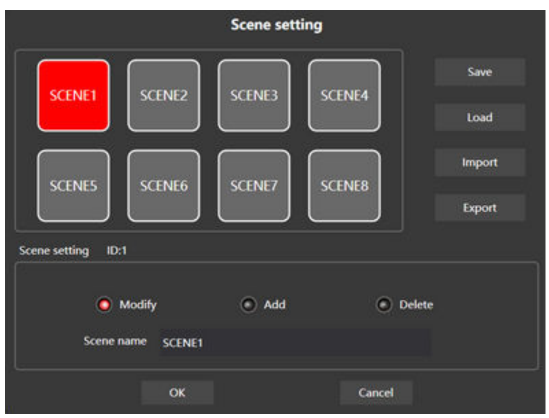
From the main interface, click Setting. The following pop-up window will appear.



Upgrade: for the device firmware upgrade if necessary.
Reset: to restore the factory default parameter state.
About: to view device serial number information

7.4 Scene

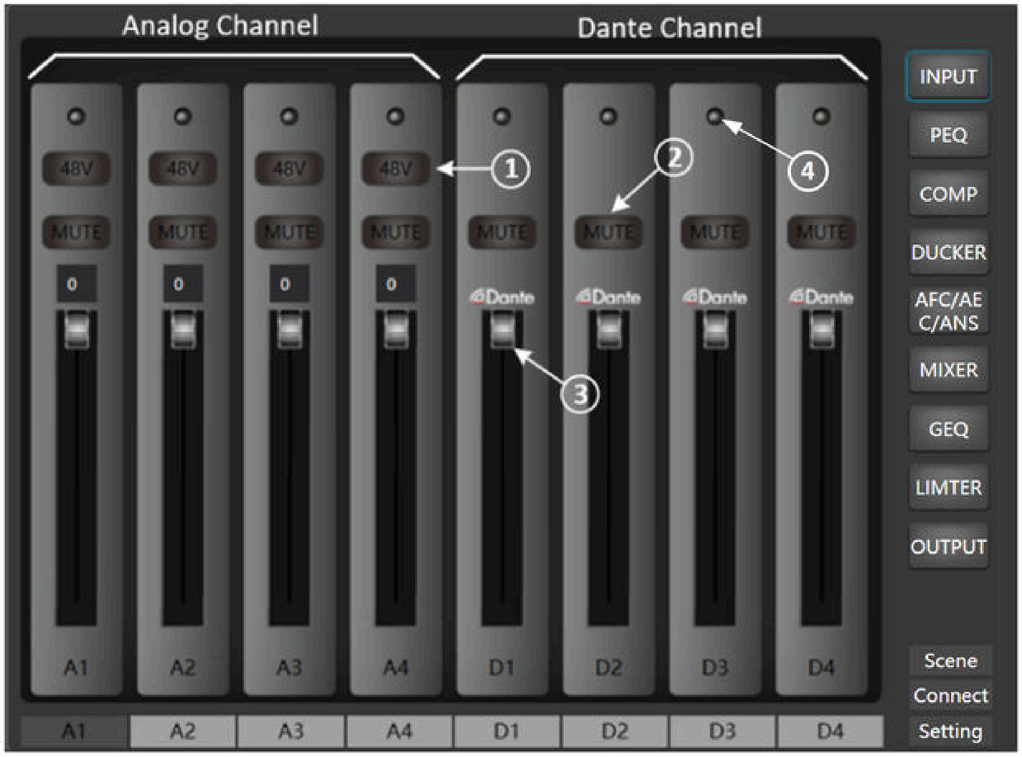
From the main interface, click Scene. The following pop-up window will appear.



Save: saves the current parameters to the selected scene.
Load: loads the selected scene.
Import: import the saved scene file.
Export: export the scene locally.
Modify: modify the scene name.
Add: add new scene, maximum 8 scenes can be added.
Delete: delete the scene.

7.5 Pre-Processing

7.5.1 Input Settings



1. 48V: Turn on or off the 48V phantom power supply of the analog channel,
2. Mute: Mute button.
3. Channel fader: The channel gain value range (0~-72dBFS), can be controlled by the fader.
4. Indicator light: When there is no signal into the input channel, the indicator light is gray; when there is a small signal into the input channel, the indicator light is green; when there is a large signal into the input channel, the indicator light is red.

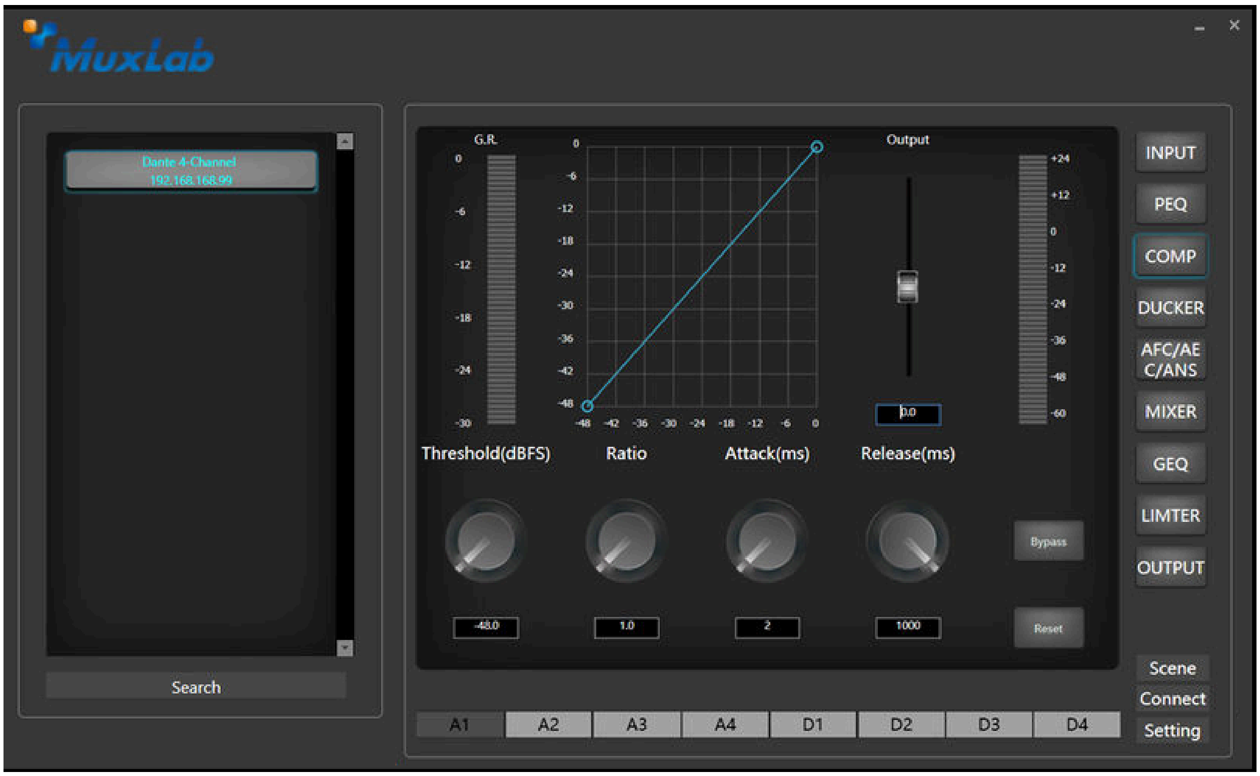
7.5.2 Parametric Equalizer

Parametric equalizers are digital filters used in audio for adjusting the frequency content of a sound signal. Parametric equalizers provide capabilities beyond those of graphic equalizers by allowing the adjustment of gain, center frequency, and bandwidth of each filter.

Equalizer type: 5-terminal parametric equalizer
Passthrough/Enable: enable/disable the equalizer;
Band Pass/Enable: Enable/disable the equalizer for the current band
Reset: restore the current parameters to the default state
Center frequency: the center frequency that needs to be equalized
Gain: the gain/attenuation value of the frequency center point
Bandwidth: That is, the range of influence of the segment around the center frequency, the larger the value, the greater the bandwidth and the greater the range of influence.

7.5.3 Compressor

Compressors are used to reduce the dynamic range of signals above a user-determined threshold, the signal levels below the threshold remain unchanged.

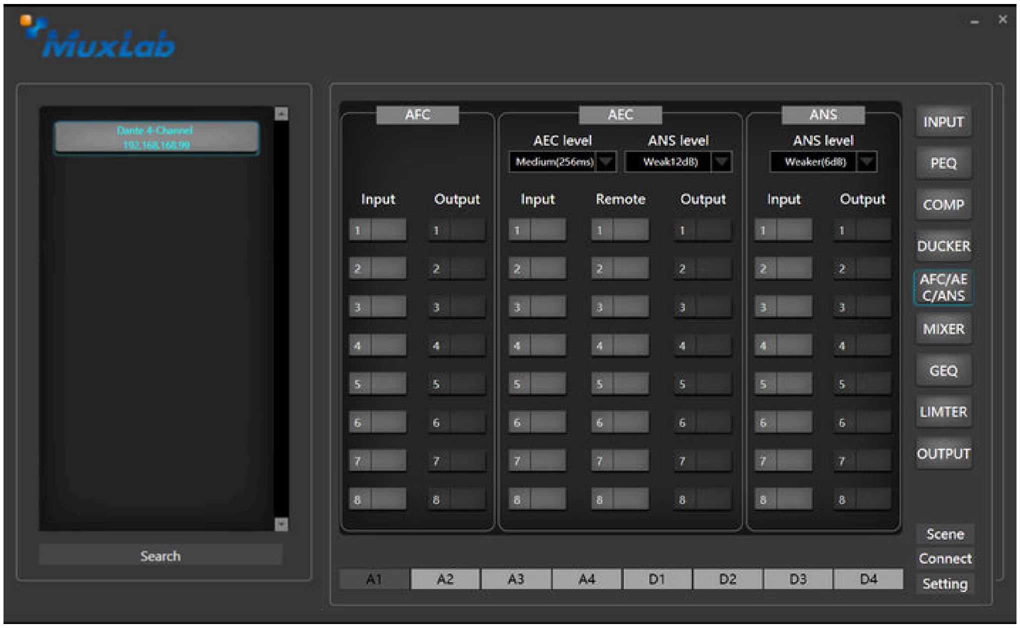


Threshold: Threshold value of the compressor.
Ratio: The input and output compression ratio of the compressor.
Attack Time: The start time of the compressor.
Release: The recovery time of the compressor.
Reset: reset to default parameters
Bypass/Active: Compressor enable indicator or control

7.5.4 Ducker

Paging Input: control signal input channel.
Background Input: controlled signal input channel.
Threshold: when the control signal level is higher than the threshold value, the controlled signal will be blocked.
Depth: attenuation value of controlled signal.
Attack Time: the effective time of dodger algorithm.
Hold Time: when there is no signal input from the control signal input channel, the algorithm will hold time.
Release Time: release time of dodger algorithm.
Output Selection: output channel.
Reset: reset as default parameter.
Bypass/Active: enable/disable the compressor.

7.5.5 AFC/AEC/ANS



Acoustic Feedback Cancellation (AFC)

Selects the signal that needs to be processed through the feedback eliminator, and the processed signal selects the output channel in the mixer

- Multi-point filtering and multi sub-band frequency shifting keep the harmonic property of the original pitch period without causing sound distortion.
- Through acoustic modeling of room feedback path, the acoustic feedback can be eliminated adaptively.
- It can quickly track the feedback path changes and greatly enhance the ability to suppress the noise. The microphone transmission gain can be increased by 6-18db, greatly enhancing the microphone gain, suitable for various large, medium and small meeting rooms.

Audio Echo Cancellation(AEC)

Sets the signal that needs to be processed through the echo canceller, and the processed signal selects the output channel in the mixer.

Input: Local MIC input, i.e. the signal that needs to be processed by echo canceller
Remote: Reference signal.

- Using sub-band algorithm, it has less MIPS consumption.
- The length of echo path can be set, the maximum echo off tail can be supported up to 512ms, suitable for all kinds of large, medium, and small meeting rooms.

- Using the stable Double Talk detection method, it is effective even in the environment of strong background noise and nonlinear distortion, and the residual echo will not increase during the simultaneous speech of both sides.
- Strong robustness, can work in all possible applications and environments.
- The embedded noise suppression algorithm can eliminate the additional noise in the noise environment.
- The variable step size and post-processing algorithm greatly improve the rate of convergence and the echo rejection ratio (ERLE) of the nonlinear distortion of the terminal speaker.

Acoustic Noise Suppression (ANS)

Select the signal that needs noise cancellation processing, and the processed signal selects the corresponding channel output in the mixer.

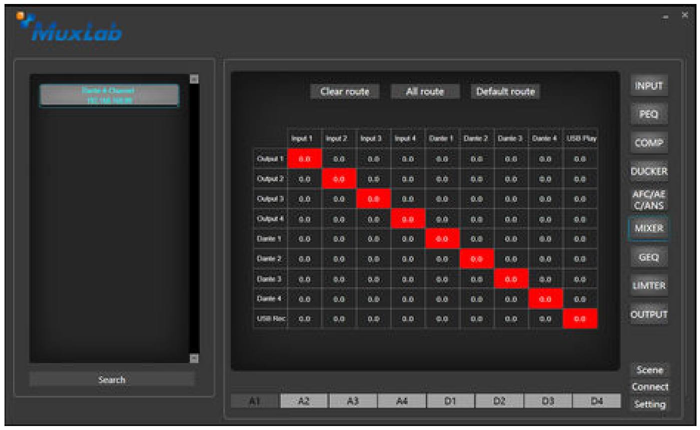
- It is a noise suppression technique to deal with noisy speech signals.

- It decompositions the input signal into a series of frequency sub-bands, estimates the environmental noise and signal level in each sub-band, and then attenuates the sub-band signal according to the real-time SNR. The output signal is synthesized by smoothing and post-processing of these processed sub-band signals.

- Because of the unique post-processing algorithm, the noise suppression algorithm can track the environmental noise changes quickly and accurately while maintaining good output sound quality. Noise suppression reaches -30db, speech is almost completely distortion free.

7.6 Matrix Mixing

Controls the mixing logic.



Columns: Input Channels
Row: Output Channels
Note: you can right click the Input and change the attenuation number display in the button.

7.7 Post Processing

7.7.1 Graphic Equalizer

The gain of 31 frequency points can be adjusted individually, to achieve the purpose of strengthening or weakening certain frequency points and achieve different effects.

Bypass/Enable: Enables and disables the equalizer.
Gain: Gain/attenuation at the frequency center point.
Flatten: restore the gain of all frequency bands to 0dB.
Narrow: A type of bandwidth that is lower than normal bandwidth.
Normal: Commonly used ordinary bandwidth.
Broadband: the widest

7.7.2 Limiter

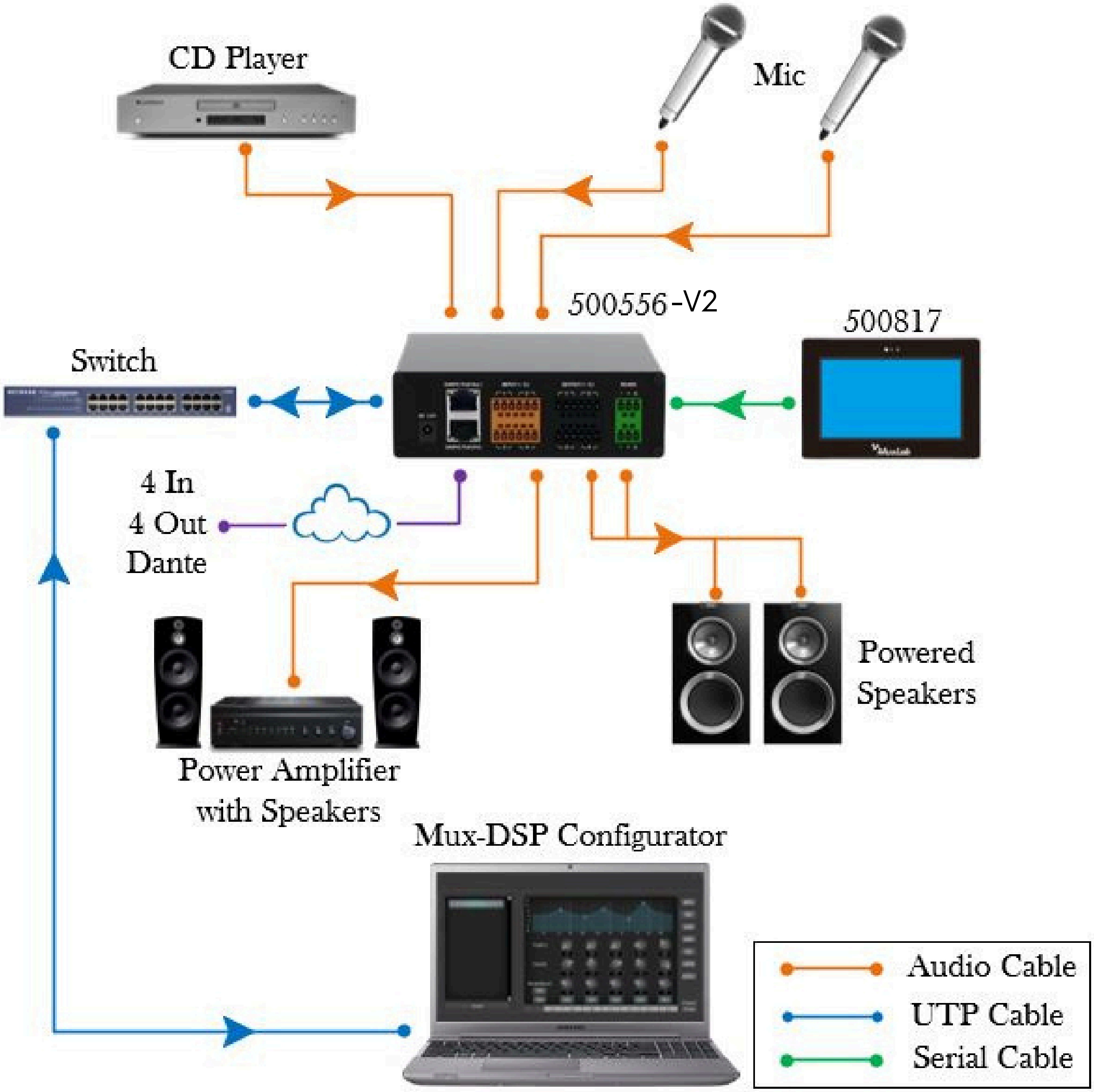
Bypass/Active: Enables or disables the limiter.
Reset: Reset to default parameter.
Threshold: The initial level of the limiter. When the signal is higher than this limit value, the limiter processing module is started.
Attack: When the input signal is lower than this setting value, the sound channel will not be turned off immediately, but the closing time will be delayed according to this setting value. During this time, as long as there is a signal above the "threshold" limit value, the sound channel can be continuously opened.
G.R.: The difference between the signal processed by the limiter and the input signal.

7.8 Output Settings

Muting and inverting the outputs as well as the output audio gain can be set.

1. **Indicator light:** When there is no signal into the output channel, the indicator light is gray; when there is a small signal into the output channel, the indicator light is green; when there is a large signal into the output channel, the indicator light is red,
2. **Invert:** Invert button.
3. **Mute:** Mute button.
4. **Channel fader**: channel gain value range (0~-72dBFS), can be controlled by fader.

8. Application Diagram



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