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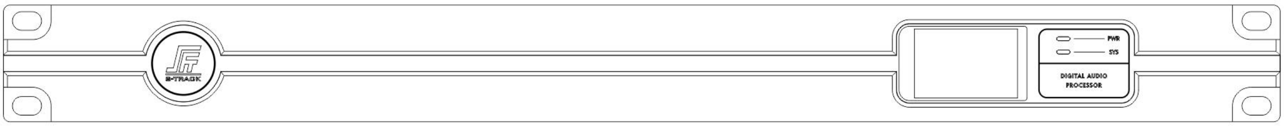
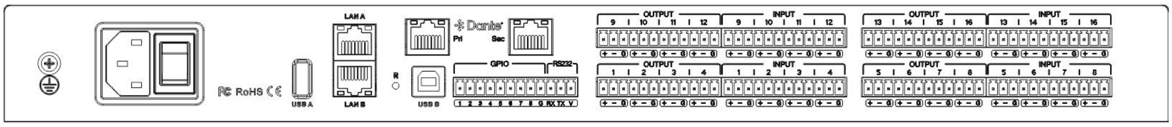




PUMA XD1616
Digital Signal Processor

User Manual

This user manual applies to software version V1.7 and above

Preface

The purpose of this section is to ensure that the user is able to use the product correctly through this manual in order to avoid danger in operation or property damage. Before using this product, please read the product manual carefully and keep it for future reference.

Outlined

This manual applies to Digital Signal Processor.

This manual describes the functions and use of the various functional modules of the Digital Signal Processor, and guides you through the installation and commissioning of the Digital Signal Processor.

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description
Note	Provides additional information to emphasize or supplement important points of the main text.
Caution	Indicates a potentially hazardous situation, which if not avoided, could result in equipment damage, data loss, performance degradation, or unexpected results.
Danger	Indicates a hazard with a high level of risk, which if not avoided, will result in death or serious injury.

Safety Instructions

Danger

To ensure reliable use of the equipment and the safety of personnel, please observe the following during installation, use and maintenance:

- During the installation and use of the equipment, all electrical safety regulations of the country and the region of use must be strictly observed.
- When installing the equipment, make sure that the power supply of the equipment is AC 100V-240V, 50/60Hz.
- Keep the working environment well ventilated so that the heat generated by the equipment during operation can be discharged in time to avoid damage to the equipment due to excessive temperature.

- Always unplug the unit's power adapter from the AC power outlet before: A. Removing or reinstalling any part of the equipment; B. Disconnecting or reconnecting any electrical plug or connection of the equipment. Do not operate with electricity.
- There are AC high-voltage parts in the equipment, non-professionals should not disassemble them without permission to avoid the risk of electric shock. Do not repair the equipment privately to avoid aggravating the damage.
- Do not spill any corrosive chemicals or liquids on or near the equipment.
- If the unit emits smoke, odour or noises, turn off the power immediately and unplug the power cord, and contact your dealer or service centre.
- If the appliance is not working properly, contact the shop where you purchased the appliance or the nearest service centre and do not disassemble or modify the appliance in any way. (We cannot be held responsible for problems caused by unapproved modifications or repairs).

Caution

- Do not drop objects on the equipment or vibrate the equipment vigorously, and keep the equipment away from locations with magnetic field interference. Avoid installing the equipment in a place where the surface vibrates or is susceptible to shock (neglecting this may damage the equipment).
- Do not use the equipment in high temperature, low temperature or high humidity environments. Refer to the equipment's data sheet for specific temperature and humidity requirements.
- Use the unit indoors, not in an exposed installation where it may be exposed to rain or extreme humidity.
- When the equipment is not used for a long period of time or in a humid and dewy environment, the main power supply of the equipment should be switched off.
- When cleaning the equipment, please use a sufficiently soft dry cloth or other alternatives to wipe the internal and external surfaces, do not use alkaline detergent to wash, and avoid hard objects to scratch the equipment.
- Please keep all the original packaging materials of the equipment properly, so that in case of problems, use the packaging materials to pack the equipment and send it to the agent or return it to the manufacturer for processing. We will not be responsible for any accidental damage in transit not caused by the original packaging materials.

Note

- Requirements for the quality of installation and commissioning personnel

  Qualifications or experience in the installation and commissioning of audio and video systems and qualifications to perform related work, in addition to the knowledge and operational skills listed below.

  - Basic knowledge and installation skills of audio and video systems and components.

- Basic knowledge and skills in low voltage cabling and wiring of low voltage electronics.
- Basic audio and networking knowledge and skills and the ability to read and understand the contents of this manual.

TABLE OF CONTENTS

Chapter 1 Product Introduction

1.1 Introduction

This is a powerful and flexible open architecture Digital Signal Processor, the series adopts a new ID design, light vitality and line sense is our design style. The hardware utilizes board-based design, equipped with ADI 40-bit high-speed floating-point DSP chip, high-performance A/D, D/A chip, with the underlying Linux operating system, and advanced DSP processing technology, which gives play to the powerful audio processing capability of ADI chip. The front panel comes with a high-definition color screen, which can display the device name and IP address of this machine to help users quickly connect to the control software.

The control software adopts a new UI design, light vitality and line sense is our design style. The software uses modular, algorithmic modular design, through the open signal flow design, drag-and-drop algorithmic modules, flexible with the functional design, able to freely build algorithmic function modules, free design of the operating logic, mainly used in large-scale conferences, stadiums, multi-purpose halls, command centers, auditoriums, etc. sound reinforcement, to achieve customized services.

128 Dante network audio channels provide high bandwidth, low latency, high compatibility and low cost solution for network audio transmission.

Through our hardware and software R & D capabilities, we provide rich audio algorithms that infinitely increase the degree of freedom and imagination, and these capabilities will be reflected in the continuous upgrading and iteration of the product.

1.2 Product Features

- Highly integrated, integrating a variety of traditional Analog audio processing equipment in a Digital Signal Processor;
- High-performance 40-bit floating-point DSP processor, all-digital processing, fast response to AGC (Automatic Gain Control), ANC (Ambient Noise Compensator), AM (Automatic Mixer), AFC (Acoustic Notch Feedback Controller), AEC (Acoustic Echo Canceler) and other audio processing;
- Analog input channels and 16 Analog output channels, very small distortion and ultra-low noise floor;
- Rich interface expansion;
- Supports Dante network transmission, which makes audio transmission more stable and faster;
- Humanization, graphical, intuitive and easy-to-operate control software interface;

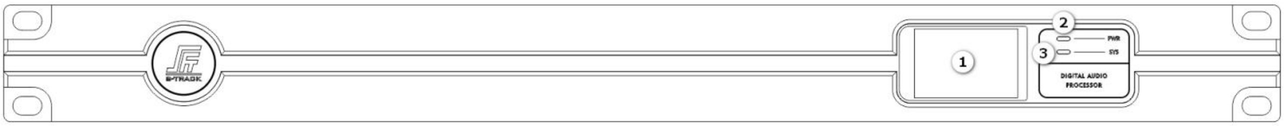
- Comprehensive matrix mixing;
- Scene storage is different from the Analog equipment is one of the most practical and significant features, can store many complete scenes, all the scenes can be exported to an external storage device for storage backup, so that the later call at any time.

1.3 Functions

- Advanced and reliable hardware: ADI high performance DSP SSHARC 21569 + NXP i.MX ARM processor;
- Supports 64×64 Dante network audio channels (AES67 compatible);
- Open signal flow design, do whatever you want to do;
- Visualized user control interface design, easily meet customer needs;
- Built-in USB Audio Class;
- Support any signal real-time monitoring, debugging is no longer blind;
- Display IP address & device name, no more worries about not finding the device;
- Built-in all the required software, no more lost CD trouble;
- Highly optimized DSP, worry-free processing up to 3400 equalization bands;
- 4 independent feedback elimination, 2 independent echo cancellation;
- Open Websocket protocol control, easy cloud platform integration, support for RS232, UDP center control control;
- Independent dual control network port, easy network backup;
- Dual Gigabit Ethernet ports, network audio transmission can be redundant backup;
- Built-in Parametric Equalizer, Graphic Equalizer, FIR High Pass Filter, Band Pass, Expander, Compressor, Noise Gate, Peak Limiter, Priority Ducker, Multifunctional Mixer, Delay, AEC, AFC and so on;
- Support USB recording and playback function, playback and recording path selectable, recording storage space real-time display;
- Support audio dynamic level real-time feedback;
- Support real-time analysis of signal spectrum of each channel.

Chapter 2 Interface Description

2.1 Front Panel

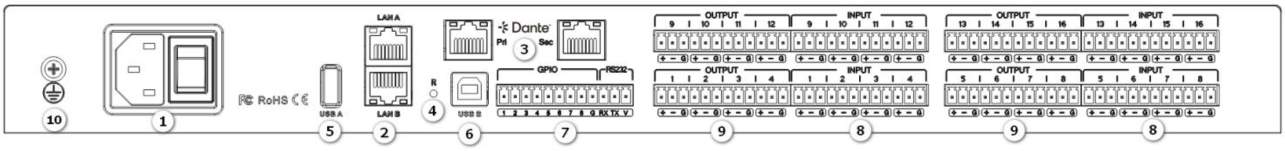


① Display: Displays the device name, IP address;

② PWR: Power indicator, the indicator light is always on to indicate that the device is powered normally;

③ SYS: System operation indicator, the indicator will flash once per second to indicate that the system is operating normally.

2.2 Rear Panel



① POWER: Power supply interface, connect AC 110V-220V power supply, warp switch control processor power;

② LAN A and B: Network control interface, by connecting this network port, the client computer can debug and monitor the device;

③ Dante: Dante network audio transmission interface, equipped with primary and secondary dual network interface, can be used for redundant backup of Dante network signals;

④ RESET: Reset button, long press to restore factory settings and restart the processor;

⑤ USB-Type A interface: USB recording and playback interface;

⑥ USB-Type B interface: USB Audio Class interface;

⑦ RS232+GPIO interface: Connect to the control terminal or central control device;

⑧ INPUT: Analog input interface, can be connected to microphone, PC and other devices;

⑨ OUTPUT: Analog output interface, can be connected to the amplifier, active speakers and other devices;

⑩ Ground screw: Used to ground the chassis, play accidental leakage safety protection, electrostatic balance and other protective measures.

Chapter 3 Software Download

By default, the device automatically obtains the IP address assigned by the DHCP server (you need to assign an IP address to it through the DHCP server of the router or managed switch, or else you can't connect to the device), and the device's front-panel display will show the current IP address.

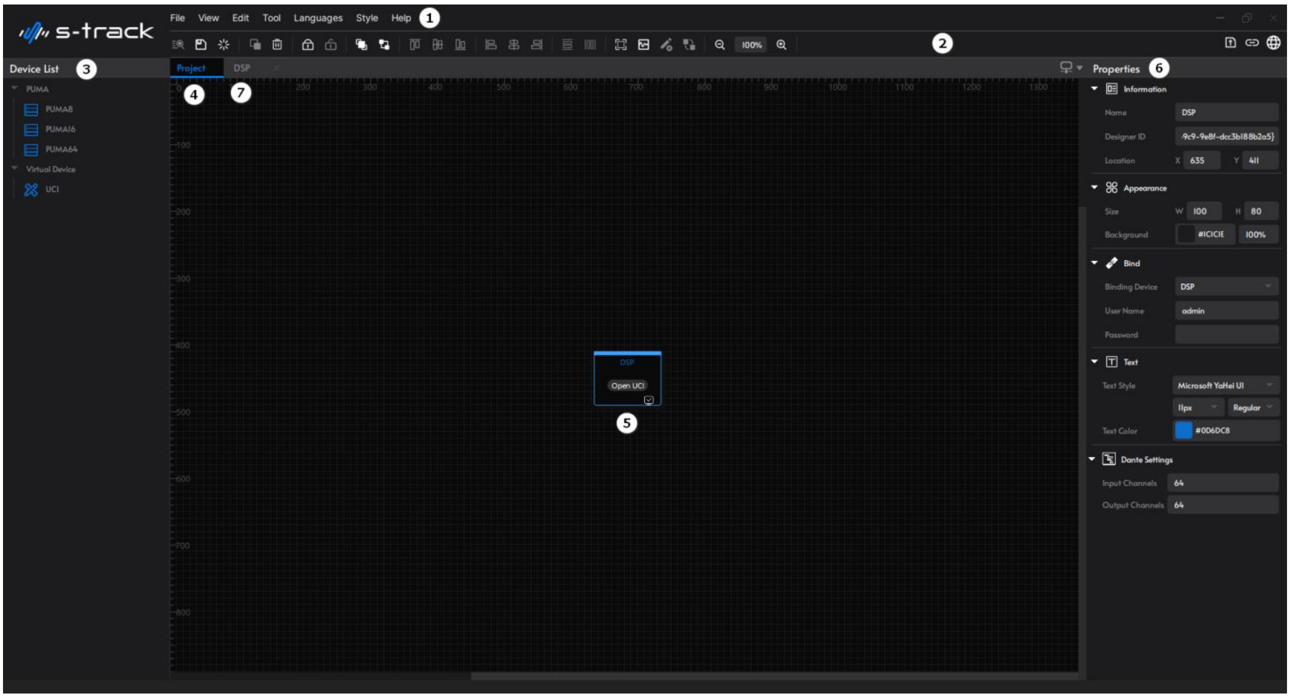
Access the processor's IP address from the client host's browser to the following page:

Click the "DOWNLOAD" download button, choose to download the Windows, macOS, Linux, Android platform client software Designer and UCI on the download page, and follow the prompts to install it.

Note: Before installing the client software, please make sure that your Windows operating system has installed Microsoft.Net Framework 4.7.2 or above, if not, you can click Download and Install.

Chapter 4 Main Interface

Device List Interface:



① Menu;

② Quick Operation;

③ Device List;

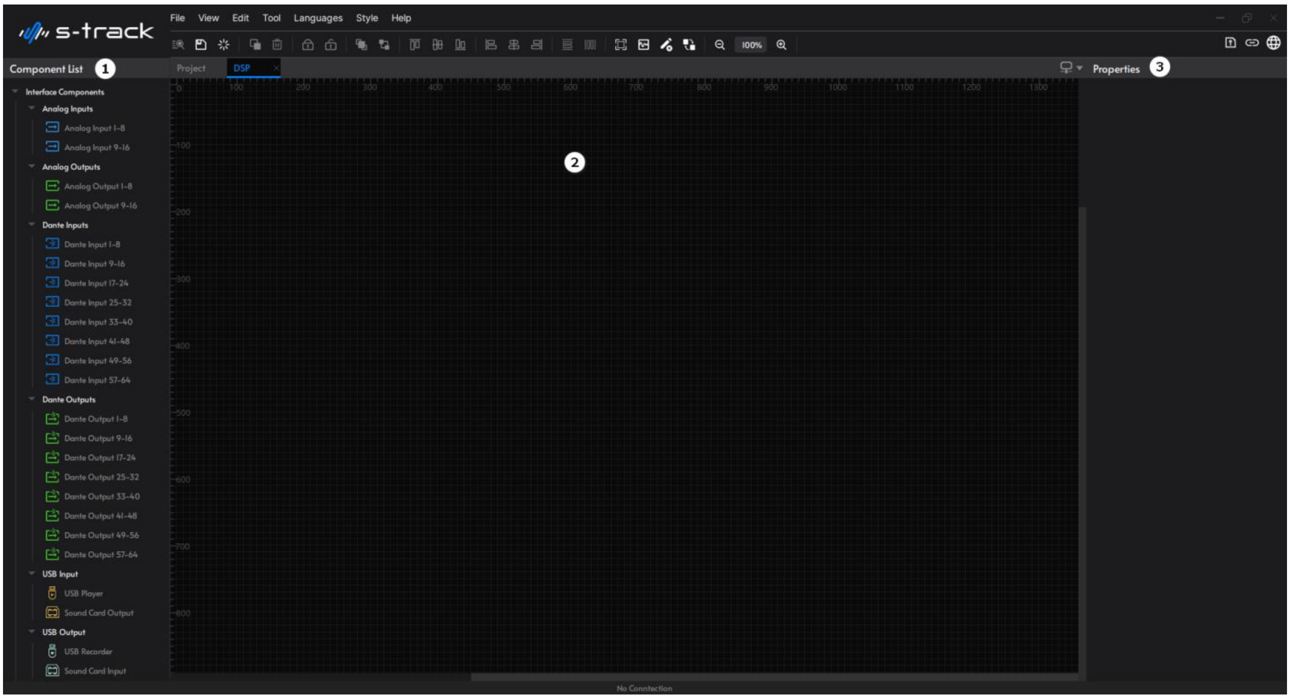
④ Device List Interface;

⑤ Device Online;

⑥ Online Device Properties;

⑦ Designer Interface Tabs.

Designer Interface:



① Component List;

② Designer Interface;

③ Component Properties;

4.1 Menu

4.1.1 File

① New (Ctrl+N): Opening a new design interface in the Designer interface will clear the existing components on the interface;

② End Run (F7): Exit the Emulate mode or Run mode;

③ Emulate (F5): Emulate mode, all adjustable controls are controllable, but there is no audio transmission, so there is no impact on audio. Levels are inactive. (Emulate is displayed in the center at the bottom of the interface);

④ Save (Ctrl+S): Save the scene locally (Load the scene saved locally, and save it again without popping up the save path dialog box);

⑤ Save as...: Save the scene locally;

⑥ Load (Ctrl+O): Load local scene;

⑦ Upload (Ctrl+U): Upload a scene to replace a previously loaded scene in the device and enter Run mode, all controls are active. Adjustable controls can be changed and have an impact on the audio. Levels are active and display device name, device IP address, current scene name, CPU usage;

⑧ Upgrade (Ctrl+Shift+S): For upgrading the firmware.

4.1.2 View

① Switch Network View: Switch to device network information view;

② Switch UCI Designer View: Switch between the Designer interface and the UCI interface.

4.1.3 Edit

① Copy (Ctrl+C): Copy all components and controls selected in the interface (Except hardware components);

② Delete (Delete): Delete all selected components and controls from the interface;

③ Select All (Ctrl+A): Select all components and controls in the interface;

④ Undo (Ctrl+Z): Undo the previous step;

⑤ Redo (Ctrl+Y): Resume the previous operation.

4.1.4 Tool

① Vertical Align-Top: Moves selected components (controls) vertically to align with the top of the last selected component (control);

② Vertical Align-Center: Moves selected components (controls) vertically to align with the middle of the last selected component (control);

③ Vertical Align-Bottom: Moves selected components (controls) vertically to align with the bottom of the last selected component (control);

④ Horizontal Align-Left: Moves selected components (controls) horizontally to align with the left of the last selected component (control);

⑤ Horizontal Align-Center: Moves selected components (controls) horizontally to align with the middle of the last selected component (control);

⑥ Horizontal Align-Right: Moves selected components (controls) horizontally to align with the right of the last selected component (control);

⑦ Lock: Lock the selected component (control);;

⑧ Unlock: Unlock the selected component (control);;

⑨ Bring to Front: Displays the component (control) at the top layer;

⑩ Send to Back: Displays the component (control) at the bottom layer;

⑪ Zoom Out: Zoom out of the current interface, minimum to 20%;

⑫ Zoom In: Zoom in on the current interface, Maximum to 300%;

⑬ Fit in Scene: Displays all components dragged and dropped into the Designer interface on one page;

⑭ Change IP Address: Change the IP address of the binding DSP for controls designed in the UCI Designer Interface (controls via UDP protocol).

Note: Only change the IP address of all controls in the current layer, if there are multiple layers, you need to change each layer.

4.1.5 Style

① Dark: Dark background;

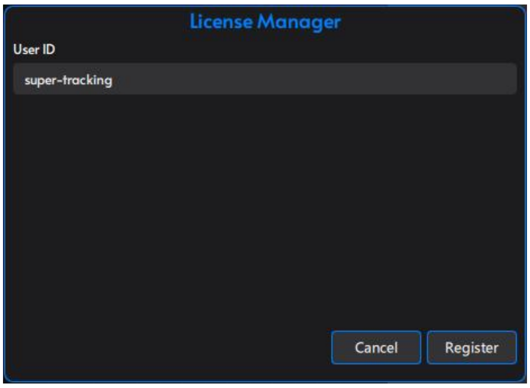
② Bright: Bright background.

4.1.6 Help

① About: View the version number;

② Help: View the user manual;

③ License Manager: Register the UCI software license;



④ Update APK: Upload an Android UCI to an Android device.

4.2 Quick Operation

① Search: You can use this option to find the location of the control that has been placed in the user interface in the component to which the design interface belongs;

② Save: Save the scene locally (Load the scene saved locally, and save it again without popping up the save path dialog box);

③ Load: Load local scene;

④ Copy: Select the component or control you want to copy, select Edit->Copy from the main menu, or click the Copy icon in Quick Operation, or press Ctrl+C, or right-click and select Copy;

⑤ Delete: Select the component or control you want to delete, select Edit->Delete from the main menu, or click the Delete icon in Quick Operation, or press Delete, or right-click to select Delete;

⑥ Lock: Select the component or control you want to lock, select Tool->Lock from the main menu, or click the Lock icon in Quick Operation, or right-click and select Lock;

Note: When the component (control) is locked, you can not select or move it.

⑦ Unlock: Select the component or control you want to unlock, select Tool->Unlock from the main menu, or click the Unlock icon in Quick Operation, or right-click and select Unlock;

⑧ Bring to Front: Displays the component (control) at the top layer. Select Tool->Bring to Front from the main menu, or click on the Bring to Front icon in Quick Operation, or right-click and select Bring to Front;

⑨ Send to Back: Displays the component (control) at the bottom layer. Select Tool->Send to Back from the main menu, or click on the Bring to Back icon in Quick Operation, or right-click and select Send to Back;

⑩ Vertical Align-Top: Moves selected components (controls) vertically to align with the top of the last selected component (control);

⑪ Vertical Align-Center: Moves selected components (controls) vertically to align with the middle of the last selected component (control);

⑫ Vertical Align-Bottom: Moves selected components (controls) vertically to align with the bottom of the last selected component (control);

⑬ Horizontal Align-Left: Moves selected components (controls) horizontally to align with the left of the last selected component (control);

⑭ Horizontal Align-Center: Moves selected components (controls) horizontally to align with the middle of the last selected component (control);

⑮ Horizontal Align-Right: Moves selected components (controls) horizontally to align with the right of the last selected component (control);

⑯ Fit in Scene: Displays all components dragged and dropped into the Designer interface on one page;

⑰ Run or Emulate Mode: The default mode of the software is Design Mode, click on this Quick Operation icon to enter the Run or Emulate Mode; click again to exit the Run or Emulate Mode;

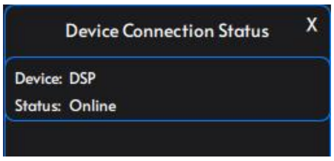
⑱ **UCI Designer Mode: The pop-up window of the default component of the software is non-UCI Designer Mode, which can only be used for parameter adjustment; after clicking this Quick Operation icon, you can modify the style of the controls in UCI interface; and you can select controls and drag them in Designer interface**;

⑲ Zoom Out: Zoom out of the current interface, minimum to 20%;

⑳ Zoom In: Zoom in on the current interface, Maximum to 300%;

㉑ Switch UCI Designer Page: Switch between the Designer interface and the UCI interface;

㉒ Up load to Device: Upload a scene to replace a previously loaded scene in the device and enter Run mode, all controls are active. Adjustable controls can be changed and have an impact on the audio;



㉓ Connect to Device or Disconnect from Device: Connect or Disconnect the device;

㉔ Network View: View device network information.

Note: The above options can be found in the right-click Quick Operation after selecting the component (control).

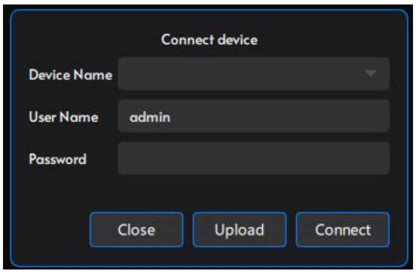
4.3 Device Configuration

4.3.1 Device Connection

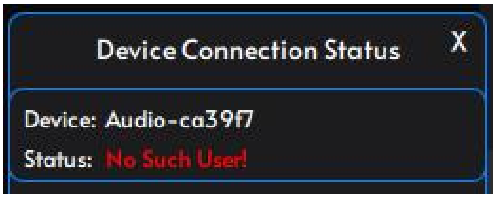
There are three types of connection scenarios:

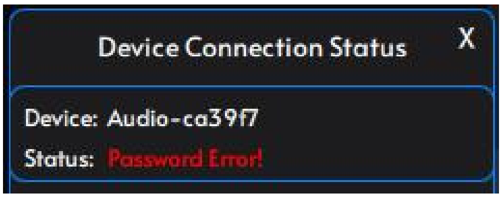
1. Unbound device: Click the connection button on the top right corner, and the list below will display "Can’t find Device";

Click on the red error prompt message, and a pop-up window for connecting the device will appear. You can then re-set the device to be connected and the corresponding username and password (the default username is admin and the password is 123456). Click the "Connect" button to load the current scene of the device; click the "Upload" button to import this scene into the device.



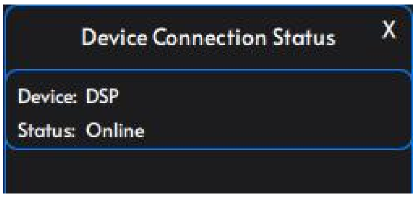
2. Input incorrect username or password: Select the device component, right-click on the device you want to connect to, and then click the connection button on the upper right corner. The list below will then display "No Such User!" or "Password Error!";





Click on the red error message, and a pop-up window for connecting the device will appear. You can then reset the username and password (the default username is admin and the default password is 123456).

3. Normal Connection: Select the device component, right-click on the device to be connected, enter the correct username and password in the right attribute bar, and then click the connection button on the upper right corner. After the connection is successful, the device component will display "Online".



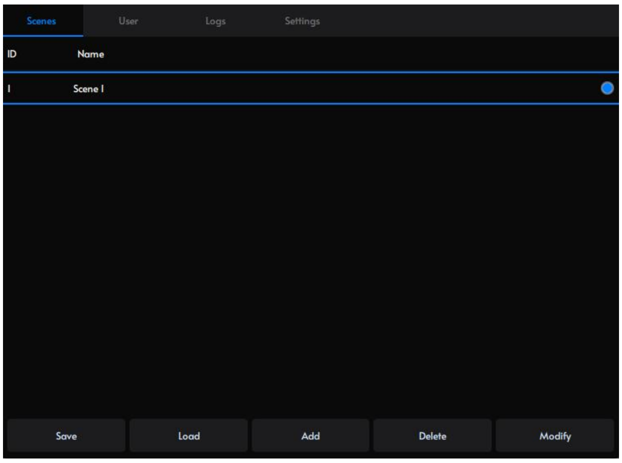
4.3.2 Device Config

Click Network View to enter the device configuration interface and display device network information.

Click "Device Config" to enter the Device Settings interface.

I. Scenes Management

Administrator can save, load, add, delete and modify the scene, other users can only switch the existing scene and cannot change the scene settings.



① Save: Save the currently running parameters to the currently running scene;

② Load: Load the currently selected scene, usually used for scene replacement;

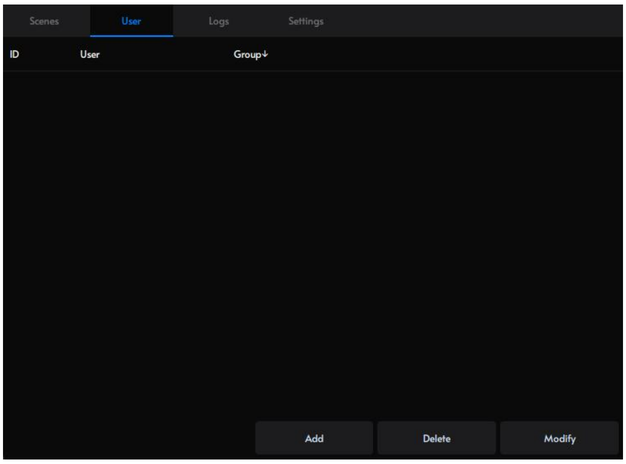
③ Add: Upload the scene from locate PC to the device;

④ Delete: Delete the selected scene;

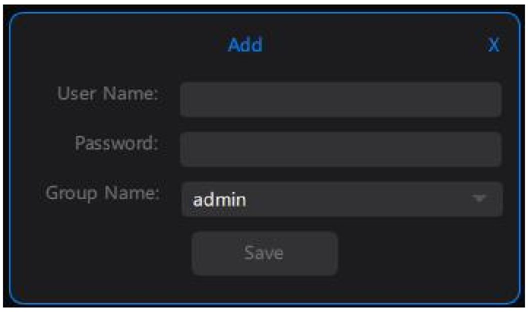
⑤ Modify: Modify the name of the selected scene.

II. User Management

Administrator can add, delete and modify all user information.



① Add: The first user to be added must be an administrator. Click "Add" button, input 1~15 characters user name and 1~14 characters password, select user identity (administrator, user, visitor), click "Save" button to add successfully.



② Delete: Click to select the user you want to delete, click "Delete" button to delete the user.

③ Modify: Click to select the user you want to modify the password, click "Modify", enter the new password and click "Save" button to modify it successfully.

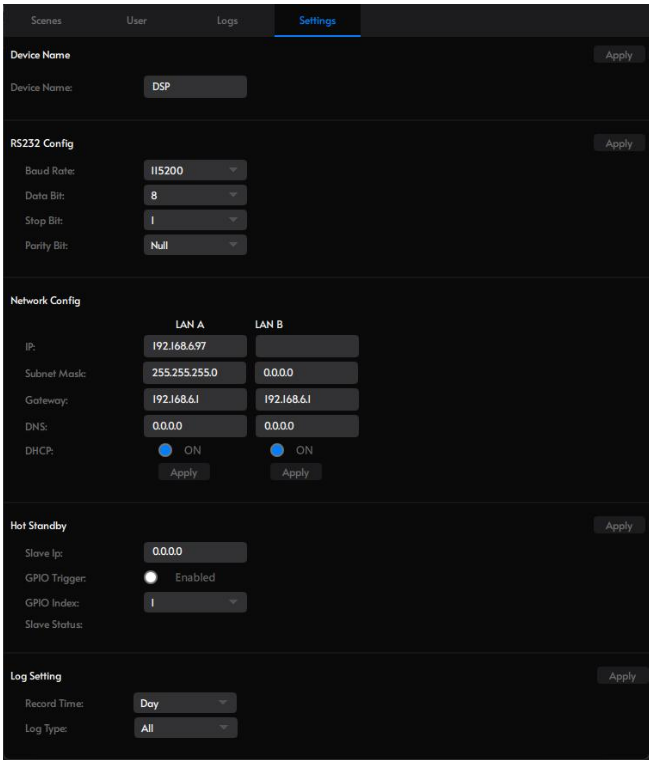
III. Logs Management

The Event Log page provides a listing of events that take place on the device. The Event Log is stored on the device, abstracted from the running design file and previous design files.

① Clear All: Delete all logs;

② Refresh: Refresh the log display.

IV. Device Settings



1. Device Name

You can rename the device name. Input the new device name and click "Apply", then the renamed name will be displayed in the device screen, the software will automatically disconnect, and the device list will be refreshed to display the renamed device name.

2. RS232 Config

View and modify the serial port information of the current device, click "Apply" button to modify the serial port information of the current device after the setting is completed.

3. Network Config

View and modify the network address of the device.

① IP, Subnet Mask, Gateway: You can input the IP address, Subnet Mask and Gateway in the corresponding position and click the "Apply" button to complete the modification;

② DNS: Analyze the domain name as IP address, so that users can access the Internet more conveniently;

③ DHCP: The device automatically obtains the IP address and subnet mask assigned by the DHCP server when it connects.

4. Hot Standby

① Slave IP: Input the IP of the slave device;

② GPIO Trigger: Whether to turn on the GPIO trigger;

③ GPIO Index: Select from GPIO 2-8 (Note: GPIO 1 cannot be selected);

④ Salve Status: The status of the slave device.

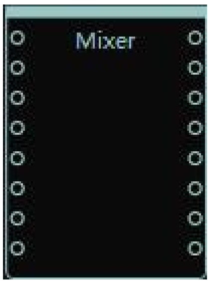
5. Log Setting

① Record time: You can query according to the Day, Week and Month;

② Log Type: You can query according to the All, Normal and Emphasis.

4.4 Components and Usage

In the components list, components are categorized into Interface Component, Audio Component.



Audio signal pins are represented by a circle, representing Channels and wiring is represented by a line. The pins on the left side of the component represent the input signal channel; the pins on the right side represent the output signal channel. However, not all components have two or any inputs and outputs. Additionally, you can set the Properties to add Channels.

4.4.1 Components

Select a component by clicking the left mouse in the Component list and drag it into the Designer interface by holding down the left mouse. When you want to add many components of the same type to the Designer interface, you can use copy and paste (except for hardware components).

After adding a component to the Designer interface, you can copy and paste it. The pasted component will retain the configuration settings of the copied component. If you make any control settings after you have made them in Emulate mode, they will be copied when you copy and paste them again.

I. Moving Components

When you select a component in the interface, you can move it in several different ways:

① Select the component and hold down the left mouse button, then drag it to the location you wish to place it in the interface;

② Select the component by entering the desired coordinates for the component in the Position field in the Properties.

II. Moving through the interface

When you design a scene that is larger than what can be displayed in the interface, there are several ways to navigate:

① Scroll: Use the sliders on the right and bottom of the interface;

② Zoom: Use the interface quick operation options to display the zoom ratio of the current page.

III. Selecting Components

The following rules apply when selecting any component:

③ You can select multiple components as long as they are on the same interface;

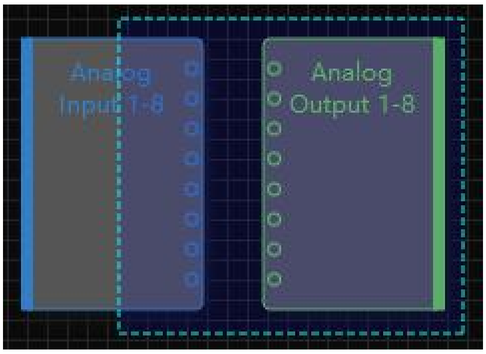
④ To select all components on the interface, press Ctrl+A. You can also use the wrap-around method described below;

⑤ To select a single component, simply click on it. When the mouse cursor is positioned over the component to be selected, it will change to a move cursor;

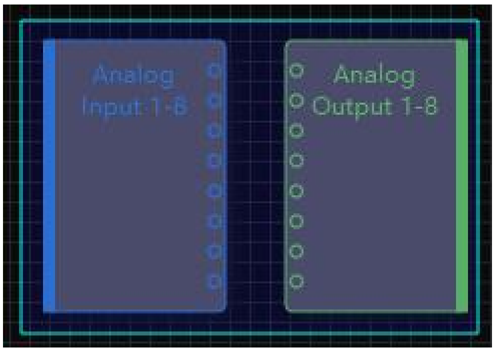
⑥ To select multiple components:

a. Hold down the Ctrl or Shift key and click one component at a time;

b. Click and hold the left mouse button, then drag a box in a **right-to-left** direction to intersect the components you selected;



c. Click and hold the left mouse button, then drag a box in a **left-to-right** direction, all within the box for the component you selected.

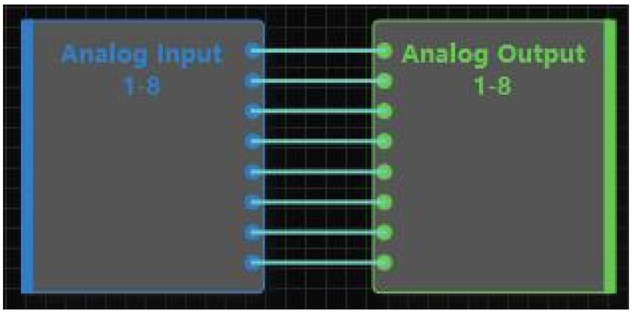


⑦ To deselect a component, simply click on another component or press the ESC button, or click on a blank space.

Note: Selecting component channels follows the same principles as selecting component rules.

4.4.2 Wiring

The following describes the use of wiring, which is represented by a line.



① You must select channels on one or more components and connect to the channels to be connected;

② You can connect in any direction: From inputs to outputs, or from outputs to inputs (only one output can be connected to an input, and multiple inputs can be connected to an output);

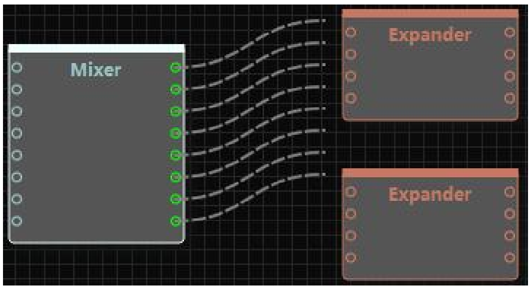
③ When you move a component, the line moves with it;

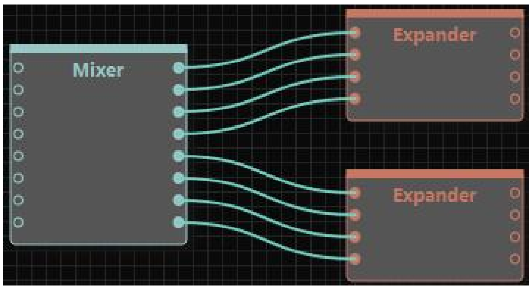
④ If you delete a component, the line of that component to other components is deleted;

⑤ If you choose to line and press the "Delete" key, the line will be deleted.

Note: Limits the types of connections that can be made. For example, you cannot connect an input to an input or an output to an output. As long as you connect the correct type and direction of connections, the components you connect together are up to you.

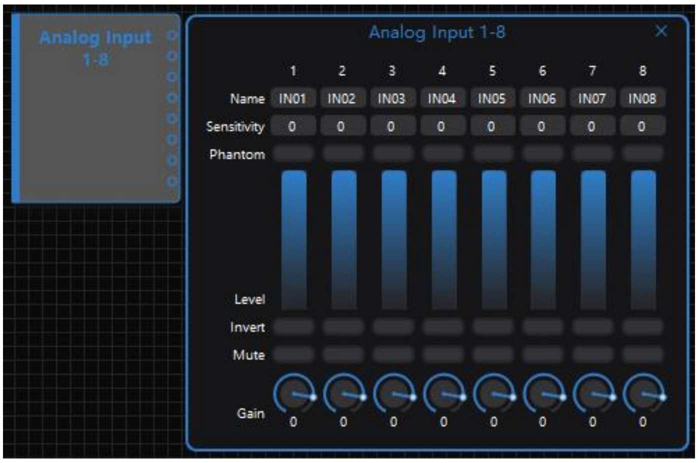
Connect one or more channels of a component by selecting one or more channels on the component.





4.4.3 Using controls

Each component has controls. When you double-click a component, the control panel for that component is displayed.



① Knob/Fader: Click and hold the control, move the mouse up, down, left or right and the control will follow to adjust the value;

② Button: Click on the button. Buttons change state depending on the type of toggle, momentary, etc;

③ Drop-down list: Click on the drop-down list, scroll or move the mouse to the entry you want and click on it;

④ Level Meter: Displays read-only controls for signal level, increase or decrease;

⑤ Label: Displays numeric controls;

⑥ Text Edit Box: Text entry field. Enter the desired text and press the "Enter" key to take effect.

Note: You cannot make adjustments to the control values on the design mode. You can modify the control values and test the control logic in the Emulate mode, but the audio is not processed. It is in Run mode that all functions of the system are running.

4.4.4 Keyboard Input

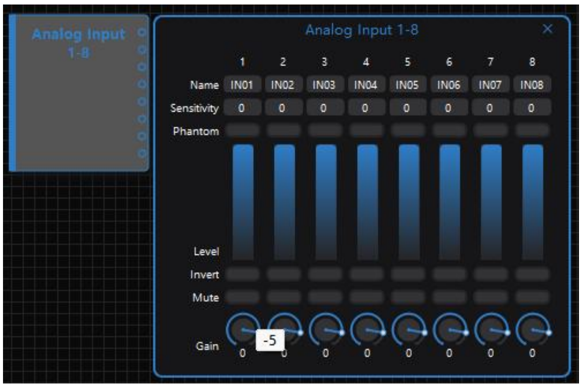
Entering control values provides a more precise method of adjusting the control. With digital controls, the accuracy depends on the unit of measurement or the range of values, and you may or may not be able to use decimals.

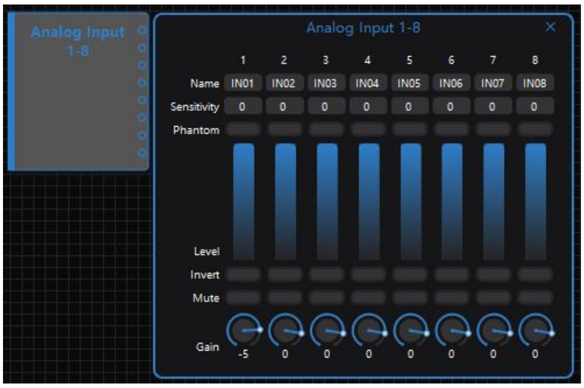
To control using the keyboard:

① Click the knob left or right to control;

② Click and hold the knob to control the knob to the left or right;

③ Enter the value and press enter to effect.



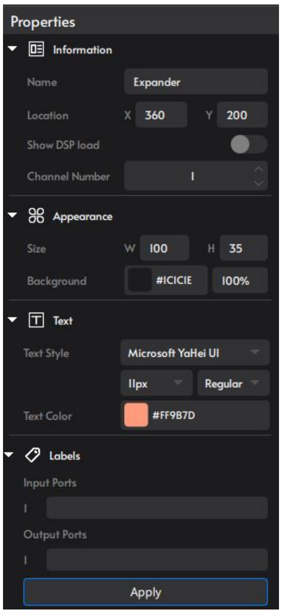


Note: Keyboard input is limited by the control range. If a value greater than the range is entered, the control is set to the maximum value allowed, and if a value less than the minimum range value is entered, the control is set to the minimum value.

The precision of the displayed value (below the control) is usually limited by the number of characters allowed in that display area, usually up to three digits, excluding decimal points. For example, you can enter 3.95 or 100, etc..

4.5 Component Properties

When you place a component in the design interface and select it, the component's properties are displayed in the right pane of the interface. The properties are divided into 3 small lists, Information, Appearance, and Text, each of which can be collapsed.



① Name: Allows you rename the component in the Designer. The Name property defaults to the component name;

② Location: The coordinates reference a specific place in the Designer;

③ Show DSP load: Real-time show of current component DSP load;

④ Channel number: Allows you to set Multi-Channel of the component;

⑤ Size: Allows you to set the width and height of the component;

⑥ Background: Set the fill color and transparency of the component;

⑦ Text style: Set the font of the component;

⑧ Text color: Set the font color of the component.

⑨ Input Ports: Customized Input ports label;

⑩ Output Ports: Customized Output ports label.

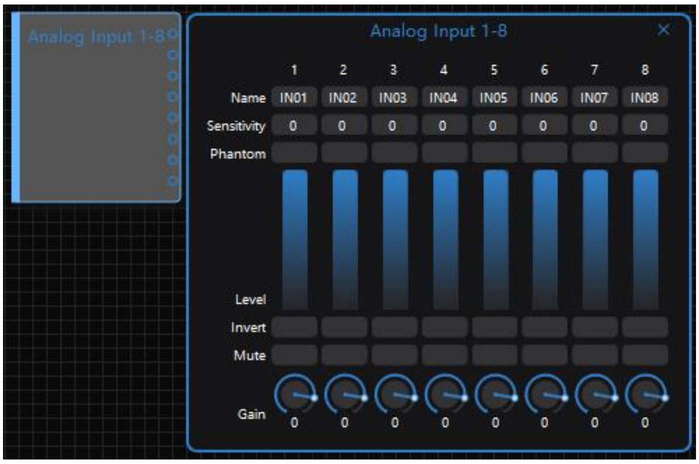
Note: In Design mode, you can change any of these properties. In Run or Emulate mode, component properties are unavailable.

Chapter 5 Component Functions

5.1 Interface Components

5.1.1 **Analog Input**

The Analog Input component provides eight line-level input for devices with line-level outputs, and inputs for microphones. The Analog Input converts the analog input signal to a processed digital and provides software controls before and after the convertor. Connections are made using eight three-terminal 3.5mm Euro style connectors.



① **Name:** Displays the name of the device's default Analog Input channel, which can be renamed;

② **Sensitivity:** Controls the maximum analog level received by the analog input channel, which determines the gain level of the input signal. Users can optimize the input signal level by adjusting the sensitivity according to actual requirements and the characteristics of the audio source. Select this field to enter the desired value;

③ **Phantom:** The input component provides 48V phantom power for condenser microphones. Condenser microphones typically require an external power source to drive their internal circuitry, while phantom power delivers a stable power supply to the microphone through the audio cable. Please do not turn on the Line Input or non-condenser microphone to prevent burning;

④ **Level:** Measures the Digital Signal level after the D/A converter;

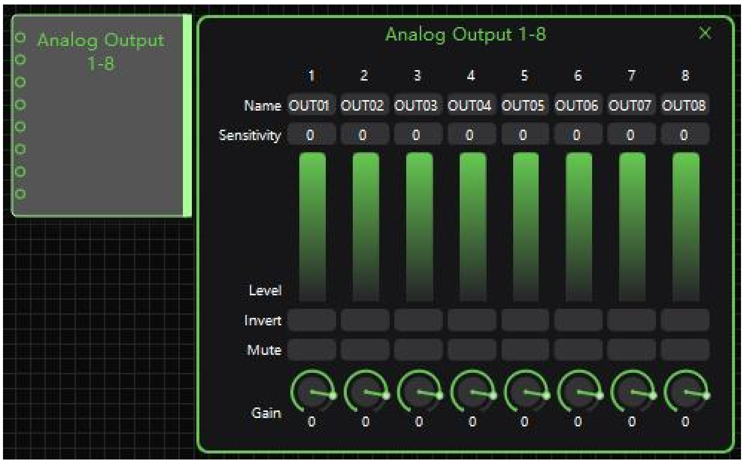
⑤ **Invert:** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

⑥ **Mute:** Mutes the output signal;

⑦ **Gain:** Controls the Gain of the output signal.

5.1.2 Analog Output

The Analog Output component provides eight channels of line-level output for device. The Analog Output converts the processed digital signal to analog and provides software controls before and after the convertor. Connections are made using eight three-terminal 3.5mm Euro style connectors.



① **Name:** Displays the name of the device's default Analog Output channel, which can be renamed;

② **Sensitivity:** Adjusts the output signal volume;

③ **Level:** Measures the Digital Signal level prior to the D/A converter;

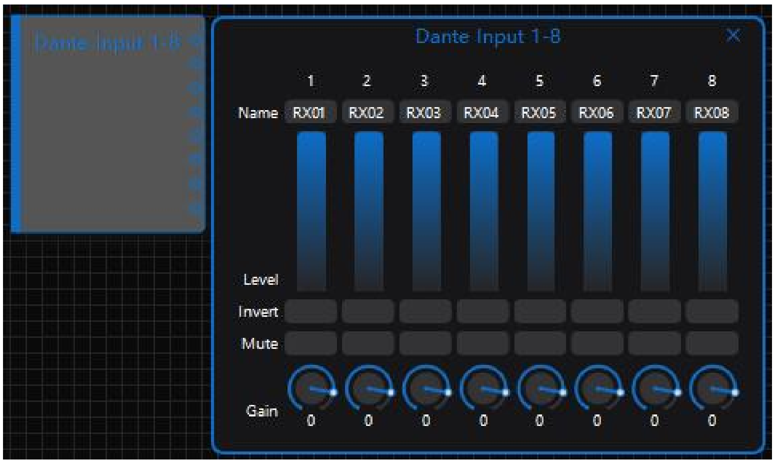
④ **Invert:** Inverts the polarity of the output signal;

⑤ **Mute:** Mutes the output signal;

⑥ **Gain:** Controls the Gain of the output signal.

5.1.3 Dante Input

The Dante Input component provides a means of receiving and transmitting to and from other Dante enabled devices on a Dante network.



① **Name:** Displays the name of the device's default Dante Input channel, which can be renamed;

② **Level:** Meters for each channel indicating the Dante input level, in dBFS. The measurement is taken directly from the Dante input signal before the Digital Controls (Invert, Gain or Mute);

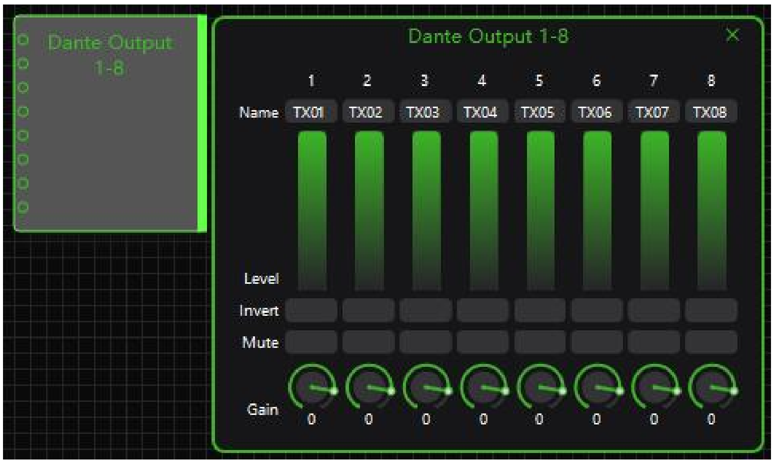
③ **Invert:** Inverts the polarity of the output signal;

④ **Mute:** Mutes the output signal;

⑤ **Gain:** Controls the Gain of the output signal.

5.1.4 Dante Output

The Dante Output component provides a means of receiving and transmitting to and from other Dante enabled devices on a Dante network.



① **Name:** Displays the name of the device's default Dante Output channel, which can be renamed;

② **Level:** Measures Output level to the output device;

③ **Invert:** Inverts the polarity of the output signal;

④ **Mute:** Mutes the output signal;

⑤ **Gain:** Controls the Gain of the output signal.

5.1.5 USB Player

The USB Player component provides playback of audio files from a USB flash drive inserted into the USB A port of the device, supporting USB flash drive format FAT32 and audio file formats MP3 and WAV.

① **Play:** The Play button starts playing the selected audio file from either the beginning, or from where it was paused;

② **Pause:** The Pause button temporarily stops playing the selected file. The must be playing in order to pause the playback. You can resume playing the file by clicking either the Play or Pause buttons;

③ **Stop:** The Stop button stops playing the selected audio file. The file is reset to the beginning;

④ **Previous:** Click this button to play the previous file in Play List order. If the current song is the first in order, this button is inactive;

⑤ **Next:** Click this button to play the next file in Playlist order. If the current song is the last in order, this button is inactive;

⑥ **Delete:** Delete the Play List files;

⑦ **Level:** Graphically displays the level of the output signal;

⑧ **Mute:** Mutes the output of the USB Player, the file continues to play;

⑨ **Invert:** Inverts the polarity of the audio signal;

⑩ **Gain:** Controls the Gain of the output signal;

⑪ **Copy From USB:** Copy the audio files from USB flash disk to the device playlist;

⑫ **Copy To USB:** Copy the audio files in the list to the USB flash disk;

⑬ **Mode:** Three play modes: None, Repeat (track): repeat playback of the selected file, and Repeat (playlist): repeat playback of files in the playlist order;

⑭ **Auto Play:** Enables the selected file to begin playing when the design is saved to the device and run;

⑮ **Current Time:** Displays how long the file has been playing. The format is HH:MM:SS. The Time readout will stop when the Pause button is engaged, and is reset to zero when the Stop button is engaged;

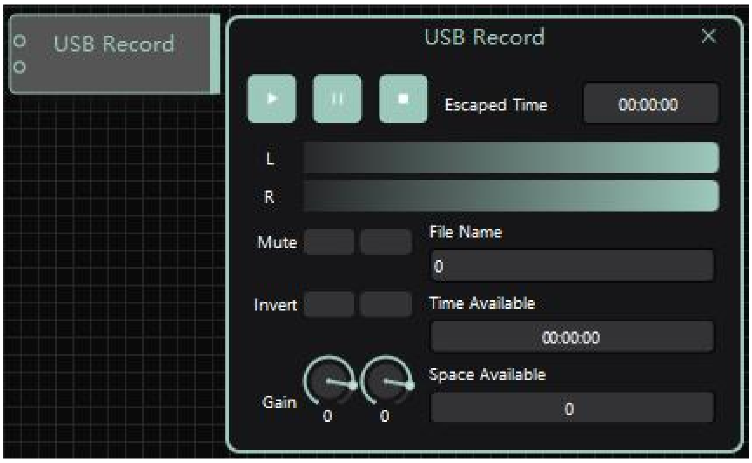
⑯ **Total Time:** Displays the total play-time of the selected file. The format is HH:MM:SS;

⑰ **Current Play:** Displays the name of the audio file being played;

⑱ **Play List:** Displays the available Playlists, and select the one you want to play. You must click the Play button to begin playback.

5.1.6 **USB Recorder**

The USB Recorder component allows you to record two channels of audio.



① **Record:** Starts the recording;

② **Pause:** Pauses the recording;

③ **Stop:** Stops the recording;

④ **Escaped Time:** Displays the length of the recording. HH:MM:SS;

⑤ **Level:** Graphically displays the level of the input signal;

⑥ **Mute:** Mutes the input signal;

⑦ **Invert:** Inverts the polarity of the audio signal;

⑧ **Gain:** Controls the Gain of the input signal;

⑨ **File Name:** Displays the name of the recorded file in date _ time, audio file format is WAV;

⑩ **Escaped Time:** Display the duration of the recording.

⑪ **Time available:** The time available on the device for recording. HH:MM:SS;

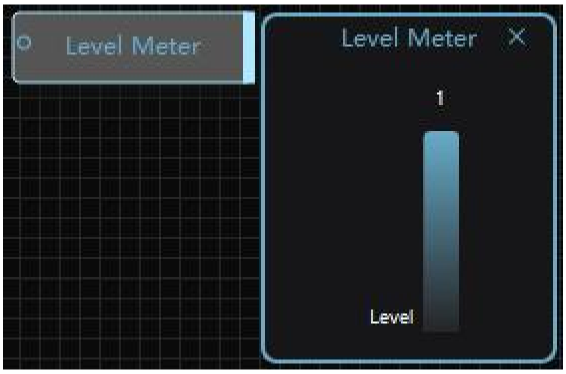
⑫ **Space available:** The space available on the device for recording.

5.2 Signal Analyze Component

5.2.1 **Level Meter**

The Level Meter wiring other Output channel to display the signal level graphically.

The Level Meter is an RMS meter used to measure audio signal level, expressed in dB.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel.

5.3 Dynamic Control Components

5.3.1 **Expander**

The purpose of the Expander component is to control the dynamic range of the Output below a set Threshold Level.

You can specify the maximum attenuation applied by adjusting the Depth setting. You can also set the proportion (Input to Output) of attenuation that is applied to the Input by adjusting the Ratio. The Ratio allows adjustments from 1 (1:1 - unity with the Input) to 100 (100:1 - very close to the Threshold Level).

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Threshold:** Sets the point from which the attenuation is calculated based on the Ratio setting. This is where the Expander starts working. Assuming only one Input, a level below the Threshold Level is attenuated, anything above the Threshold Level is not attenuated;

Example:

- If the: Threshold Level is -30 dB; Ratio is 2.5; Input level is -40 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-40 dB - (-30 dB)] * 2.5} + (-30 dB) = -55 dB.

② **Ratio:** The ratio between the Input and the Output as measured from the Threshold Level;

③ **Depth:** The maximum amount of attenuation that can be applied;

④ **Soft Knee:** Provides a smooth transition for the Output during the transition from above the Threshold Level to the expanded Output, below the Threshold Level. The Soft Knee begins adding attenuation before the Threshold Level as the Input decreases then continues to increase attenuation until the Input level decreases by the amount set by the Soft Knee. At that point, the Soft Knee has no effect;

⑤ **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

⑥ **Attack Time:** Sets the time it takes for the output amplitude to rise to 63% of the specified Ratio of the input amplitude once the Threshold Level is exceeded;

⑦ **Release Time:** Sets the time it takes for the output to fall to 63% of the level set by the Depth control after the Detector level drops below the Threshold Level;

⑧ **In Level:** Graphically displays the level of the input signal;

⑨ **In Gain:** Sets the Gain of the input. This is added to the level of the input source;

⑩ **Out Gain:** Controls the Gain of the output;

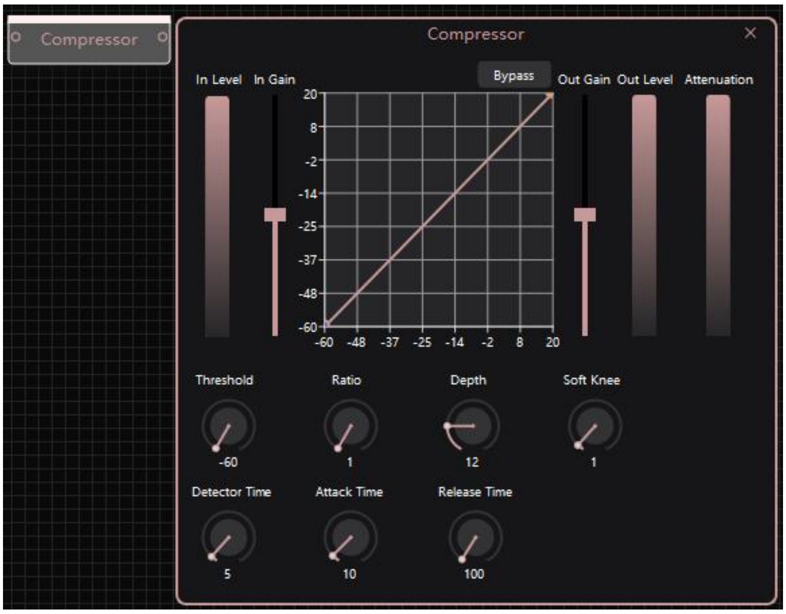
⑪ **Attenuation:** Graphically displays the amount of attenuation applied to the Channel;

⑫ **Bypass:** When engaged, the Expander is bypassed, audio is passed through without any change..

5.3.2 Compressor

The purpose of the Compressor component is to control the dynamic range of the Output above a set Threshold Level.

The Compressor can be adjusted from unity (1:1) with the Input, to an almost flat (100:1 - very little amplitude variation) Output. Using the Depth control, you can control the point at which compression stops increasing, even though the Input level continues to increase.



1. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Threshold:** Sets the level where compression begins. Assuming only one Input, a level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;

Example:

- If the:Threshold Level is -30 dB; Ratio is 2.5; Input level is -10 dB
- Then the Adjusted Output is:
- {[(Input Level - Threshold Level) / Ratio] + Threshold Level} = Output Level
- {[-10 dB - (-30 dB) / 2.5] + (-30 dB)} = -22 dB.

② **Ratio:** The ratio between the Input and the Output as measured from the Threshold Level. The closer the Ratio is to 100, the smaller dynamic changes in the Output level. As the Ratio is adjusted closer to 1, the dynamic range of the Output increases;

③ **Depth:** Sets the maximum amount of attenuation applied to the Input. As the Input increases, the Compressor applies more attenuation based on the Ratio setting. When the amount of attenuation applied reaches the Depth setting, no more attenuation is added;

④ **Soft Knee:** Provides a smooth transition, in the Output, from the uncompressed level to the compressed level. The Soft Knee slowly adds attenuation, as the Input increases, until the attenuation added by the Soft Knee is equal to the attenuation added by the Ratio setting;

⑤ **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

⑥ **Attack Time:** Sets the time it takes for the Input to drop to 63% of the Depth level. A long Attack Time causes the Output to stay closer to unity with the Input as the Input increases, then slowly drop to the desired Output level. A fast Attack Time causes the Output to get to the desired level faster without getting as close to unity with the Input;

⑦ **Release Time:** Sets the time it takes for the Output to reach 63% of the Threshold Level. This prevents the Output from dropping to quickly when the Input drops. Provides a smoother transition from a high Output to a lower Output;

⑧ **In Level:** Graphically displays the level of the input signal;

⑨ **In Gain:** Sets the Gain of the input. This is added to the level of the input source;

⑩ **Out Gain:** Controls the Gain of the output;

⑪ **Attenuation:** Graphically displays the amount of attenuation applied to the Channel;

⑫ **Bypass:** When engaged, the Compressor is bypassed, audio is passed through without any change.

5.3.3 **Noise Gate**

The Noise Gate component is used to either pass or attenuate audio signals based on the RMS level of the input signal. If the signal is above the specified Threshold, the signal is passed un-attenuated. If the signal is below the Threshold, it is attenuated by the amount specified by the Depth control.

The Noise Gate component controls its output based on the input level. If the input is lower than the Threshold level, it is attenuated. If the input is above the Threshold Level, it is passed un-attenuated. You can use the Gate to prevent open microphones and other types of inputs, from introducing unwanted sounds into your system.

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Threshold**: Sets the level at which the Noise Gate opens based on the Input level;

② **Depth:** Sets how much Attenuation is applied to the inputs when they are below the Threshold Level. This is a negative amount added to the Detector Level;

③ **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

④ **Attack Time:** The Attack Time adjusts the time it takes for the output amplitude to rise to equal 63% of the input amplitude once the Threshold Level is exceeded. This does not affect the decay of the signal;

⑤ **Hold Time:** The Hold Time determines the minimum time the Noise Gate stays open once it is opened, or the length of time the Noise Gate stays open after the RMS input level drops below the Threshold Level. This is to prevent the gate from opening and closing due to momentary pauses in the input;

⑥ **Release Time:** The Release Time adjusts the time it takes for the output to fall to 63% of the level set by the Depth control, after the Detector level drops below the Threshold Level and the Hold Time is expired;

⑦ **In Gain:** Sets the Gain of the input. This is added to the level of the input source;

⑧ **In Level:** Graphically displays the level of the input signal;

⑨ **Attenuation:** Graphically displays the amount of attenuation applied to the Channel;

⑩ **Out Gain:** Controls the Gain of the output;

⑪ **Bypass:** When engaged, the Active the Noise Gate is bypassed, audio is passed through without any change.

5.3.4 **Automatic Gain Control**

The purpose of the Automatic Gain Control component is to control the overall dynamic range of the Output when the Input level changes.

The AGC can be adjusted from unity (1:1) with the Input, to an almost flat (100:1 - very little amplitude variation) Output. The AGC automatically adjusts the Gain to the Target Level by compensating for low Inputs, compressing for high Inputs.

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Threshold:** The level at which the AGC component becomes active. This should be set at a level so that the anticipated background noise does not activate the AGC. When an Input exceeds the Threshold Level, the Gain for that input is adjusted to calculated level based on the Ratio and the Target Level;

② **Ratio:** The ratio between the Input and the Output as measured from the Target Level. The closer the Ratio is to 100, the closer the Output will be to the Target Level, which also means smaller dynamic changes in the Output level. When the Input is below the Target Level and the Gain applied by the Ratio setting is greater than the Maximum Gain setting, the Output is clipped per the Maximum Gain setting;

③ **Max Gain:** Sets the maximum amount of gain provided to an Input when the Input is above the Threshold Level, but below the Target Level. The Maximum Gain is not used when the gain applied by the Ratio setting is less than the Maximum Gain setting.

Example:

Ratio calculation used:

- Threshold Level is -50 dB
- Target Level is -15 dB
- Ratio is 2 (2:1)
- Maximum Gain is 10 dB
- Input is -20 dB
- Adjusted Output is -17.5 dB
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-20 dB - (-15 dB)] / 2} + (-15 dB) = -17.5 dB
- The applied gain is 2.5 dB, which is less than the 10 dB Maximum Gain.

Maximum Gain used:

- Maximum Gain is 10 dB
- Input is -40 dB
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-40 dB - (-15 dB)] / 2} + (-15 dB) = -27.5 dB
- would be a gain of 12.5 dB based on the Ratio
- Adjusted Output is actually -30 dB a gain of 10 dB, which is equal to the Maximum Gain setting of 10 dB. The same calculation is used, but the Output is clipped at the Maximum Gain setting.

④ **Target Level:** Sets the point from which the Gain is calculated based on the Ratio setting. Assuming only one Input, a level below the Target Level has a positive Gain applied, a level above the Target Level has a negative Gain (attenuation) applied;

Example 1:

- If the Threshold Level is -40dB, Target Level is -15 dB; Ratio is 2.5; Input level is -30 dB
- Adjusted Output is
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-30 dB - (-15 dB)] / 2.5} + (-15 dB) = -21 dB

Example 2:

- If the Threshold Level is -40dB, Target Level is -15 dB; Ratio is 2.5; Input level is -5 dB
- Adjusted Output is
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-5 dB - (-15 dB)] / 2.5} + (-15 dB) = -11 dB

⑤ **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

⑥ **Response Time:** Adjusts the time constant that determines the time it takes for the Gain to go from its current level to the new level when there is a change in the Input RMS level;

⑦ **Hold Time:** When the Enable button is engaged, Hold Time sets minimum time the applied Gain stays at the calculated level after the RMS Input level drops below the Threshold Level. This is to prevent the Output from fluctuating due to momentary pauses in the Input. If there is another Channel in use with a lower RMS Input level, above the Threshold Level, the Hold Time does not apply. The applied Gain changes to the proper level for the new Channel based on the Response Time;

⑧ **Recovery Time:** When the Enable button is engaged, the Recovery Time sets the time required for the applied gain to return to zero. If there is another channel in use with a lower RMS input level above the threshold level, the Recovery Time does not take effect. The applied gain is changed to the appropriate level for the new channel based on the response time;

⑨ **Enable:** Turns on and off the Recovery Time Enable functionality. In the Off position, the Hold Time and Recovery Time do not apply. If there is only one Channel above the Threshold Level, and the Recovery button is in the Off position, when that Channel drops below the Threshold Level, the Gain stays at the last calculated level before the Channel dropped below the Threshold Level;

⑩ **In Level:** Graphically displays the level of the input signal;

⑪ **In Gain:** Sets the Gain of the input. This is added to the level of the input source;

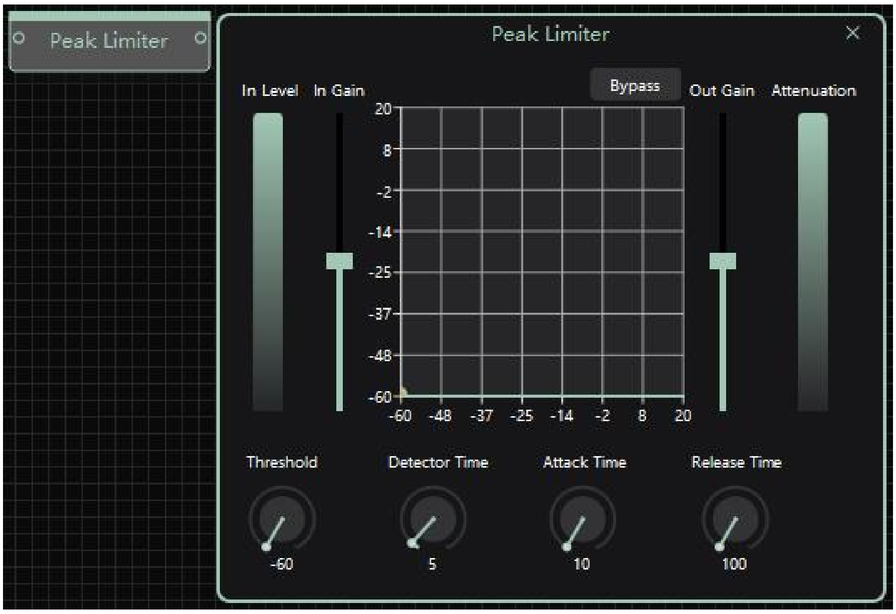
⑫ **Out Gain:** Controls the Gain of the output;

⑬ **Attenuation:** Graphically displays the amount of attenuation applied to the Channel;

⑭ **Bypass:** When engaged, the Automatic Gain Control is bypassed, audio is passed through without any change.

5.3.5 Peak Limiter

The Peak Limiter component limits the output level to the Threshold Level plus the Attenuation, prevent signal overload and transient interference.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Threshold:** Sets the level at which the Peak Limiter has an effect, and the level at which the output is held;

② **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

③ **Attack Time:** The Attack Time adjusts the time it takes for the output amplitude to rise to equal 63% of the input amplitude once the Threshold Level is exceeded. This does not affect the decay of the signal;

④ **Release Time:** The Release Time adjusts the time it takes for the output to fall to 63% of unity with the Input, after the Detector level drops below the Threshold Level;

⑤ **In Level:** Graphically displays the level of the input signal;

⑥ **In Gain:** Sets the Gain of the Input. This is added to the output Gain when the Input Level is below the Threshold Level, and added to the level of the Applied Gain or attenuation when the Input signal is above the Threshold Level;

⑦ **Out Gain:** Controls the Gain of the output;

⑧ **Attenuation:** Graphically displays the amount of attenuation applied to the Channel;

⑨ **Bypass:** When engaged, the Peak Limiter is bypassed, audio is passed through without any change.

5.3.6 **Priority Ducker**

The Priority Ducker component is used to attenuate one or more channels of audio input when the audio input of a "priority" channel reaches a specified Threshold. The audio on the Priority channel is then mixed to the outputs in place of, or louder than, the attenuated audio. You can control:

- The amount of gain applied to the main channels;
- How long the main channels are held at the gain level;
- How fast the reduced gain is applied to the main channels, and how fast the initial gain is re-applied, providing smooth transitions;
- The gain of the Priority channel.

The Priority Ducker can have many audio inputs and 1 Priority channel. You can choose a pre-set Detector Time, or you can choose the manual control. The Priority channel is the last input on the left side of the Priority Ducker component.

A typical application:

- **Conference System:** When a Conference begins, all of the other background music are reduced in gain and only the Speaker is heard;
- **Program Performances:** When a program performance begins, all of the other background music are reduced in gain and only the host is heard
- **Emergency Announcements:** When an emergency announcement begins, all of the other channels are reduced in gain and only the emergency announcement is heard.

1. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and one Output Channel.

① Input Pins: Input signal channel (example background music);

② Last Input Pins: Priority Input signal channel (example microphone);

③ Output Pins: Output signal channel.

2. Controls

① **Threshold:** The Threshold Level is the RMS level of the Priority channel at which the Priority Ducker activates;

② **Depth:** The Depth sets the amount of attenuation applied to the main input channels when the Priority Ducker is activated. The Priority channel is mixed with the main outputs, so the Depth setting controls how much of the main channels are heard in the output. Setting the Depth to 60 dB of attenuation essentially mutes the main channels;

③ **Priority Gain:** The Priority Gain controls the output gain of the priority channel. Adjust this gain to the level you wish the priority channel to be heard when mixed to the outputs with the attenuated audio;

④ **Detector Time:** The Detector Time adjusts the time constant which determines the rate of change of the Detector level in response to changes in the input signal. Sets the time it takes for the Detector level to reach 63% of a change in the input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the input from causing unwanted, momentary output. A fast Detector Time may result in an Output that rapidly fluctuates;

⑤ **Attack Time:** The Attack Time is the time it takes the main channel output to fall to 63% of the Depth level when the Priority Ducker is activated.Use this control to provide a smooth transition from the main channel audio to the Priority channel audio;

⑥ **Hold Time:** The Hold Time determines how long the main channel stays at Depth once the Detector level drops below the Threshold.This is to prevent the main channel from opening and closing due to momentary pauses in the Priority channel input;

⑦ **Release Time:** The Release Time is the time it takes the main channel output to return to 63% of its normal level when the Priority Ducker is deactivated and the Hold Time expires.Use this control to provide a smooth transition from the Priority channel audio back to the main channel audio;

⑧ **In Gain:** Controls the input Gain of the Priority channel. This control has no effect on the main channel inputs to the Priority Ducker;

⑨ **In Level:** Graphically displays the level of the input signal of the priority channel;

⑩ **Applied Gain:** Graphically displays the amount of gain applied to the main channels;

⑪ **Out Gain:** Control the Gain of the overall output of the Priority Ducker, both Priority and main channels;

⑫ **Bypass:** When engaged, the Priority Ducker is bypassed, audio is passed through without any change.

5.3.7 Ambient Noise Compensator

The Ambient Noise Compensator component is used to control the level of a Program signal, for example background music or announcements, in a situation where the ambient signal level varies.

The ambient signal level is measured, and as it rises, the gain applied to the Program signal is increased, so that the Program signal can be heard over the ambient signal. The ambient signal level is measured continuously, even when Program signal is present. The Ambient Noise Compensator removes as much Program signal from the sensing microphone signal as possible by modeling the path from the loudspeaker(s) to the sensing microphone.

6. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and one Output Channel.

① Input Pins: Program signal Input channel;

② Last Input Pins: Sensing microphone Input channel;

③ Output Pins: Program signal Output channel.

7. Controls

① **Threshold**: The recovered ambient signal level at which the compensator starts compensating (The Program signal already has been removed);

② **Depth:** The Depth sets the amount of compensatory applied to the Program signal Input channels when the Ambient Noise Compensator is activated. The Depth setting controls how much of the Program signal channel are heard in the output;

③ **Priority Gain:** The Priority Gain controls the output gain of the Program channel. Adjust this gain to the level you wish the Program channel to be heard;

④ **Detector Time:** The Detector Time determines the rate of change of the Detector level in response to changes in the ambient input signal. This adjustment is to prevent changes (spikes, low frequency signals) in the ambient input from causing unwanted, momentary changes in the Program output;

⑤ **Attack Time:** Determines the time it takes for the gain to reach 63% of a rising ambient signal detector level;

⑥ **Hold Time:** The Hold Time determines how long the Program channel stays at Depth once the Detector level drops below the Threshold.This is to prevent the Program channel from opening and closing due to momentary pauses in the Sensing microphone channel input;

⑦ **Release Time:** Determines the time it takes for the gain to reach 63% of a falling ambient signal detector level;

⑧ **In Gain:**Control the Gain of the sensing microphone input;

⑨ **In Level:** Graphically displays the level of the sensing microphone input signal;

⑩ **Applied Gain:** Graphically displays the amount of gain applied to the Program channel;

⑪ **Out Gain:** Control the Gain of the Program output;

⑫ **Bypass:** When engaged, the Ambient Noise Compensator is bypassed, audio is passed through without any change.

5.3.8 Gated Ambient Compensator

The purpose of the Gated Ambient Compensator component is to control the gain of a Program, for example, announcements, in a situation where the ambient noise varies.

The ambient noise is measured, and as it increases, the applied gain to the announcements increases at a set Ratio, so that when the announcement is made it is loud enough to overcome the ambient noise. When the announcement level exceeds the Threshold Level, the Ambient Detector becomes inactive so that the announcement is not measured along with the ambient noise. When the announcement level falls below the Threshold Level, the Ambient Detector again becomes active.

When the ambient noise is low, and an announcement is made, depending on the input level of the announcement, and the gain settings in the Compensator, attenuation may be applied to the announcement so that it is not too loud.

1. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and one Output Channel.

① Input Pins: Program signal Input channel;

② Last Input Pins: Sensing microphone Input channel;

③ Output Pins: Program signal Output channel.

2. Controls

① **Program Threshold**: Sets the threshold, of the Program input, above which the Ambient Detector is not active. When the channel Program Level falls below this setting, the Ambient Detector becomes active;

② **Ambient Threshold**: The crossing point of the ambient level and the Minimum Gain. At this point, as the ambient level rises, the gain applied to the Program input moves from the Minimum Gain towards the Maximum Gain. The point at which the ambient level is high enough to have an impact on the Program input gain;

③ **Ratio:** The ratio between the Minimum Gain and the Maximum Gain as measured from the Ambient Threshold Level. The closer the Ratio is to 2, the lower the ambient level needs to be for the Maximum Gain to be used when the Program input passes the Program Threshold. For example: If the Ambient Threshold is -30 dB, Minimum Gain is -10 dB, Maximum Gain is 10 dB, and the Ratio is 2.0, when the ambient level reaches -20 dB, Maximum Gain is applied when the Program Threshold is reached.Same example except the Ratio is 0.5. When the ambient level reaches 10 dB the Maximum Gain is applied when the Program Threshold is reached;

④ **Minimum Gain:** The minimum amount of gain you want to apply to the Program output when the ambient noise is at it's lowest point. Usually a negative amount, or attenuation - shown in red on the Gain meter. Gain is added to the Program input level up to this setting. If the Minimum Gain is -15 dB, and the Program input level is 5 dB, the Program Output is -10 dB, 5 + (-15) = -10;

⑤ **Maximum Gain:** The maximum amount of gain you want to apply to the Program output when the ambient noise is at it's highest point. Gain is added to the Program input level up to this setting. If the Maximum Gain is set to 10 dB, and the Program input level is 5 dB, the Program Output is 15 dB;

⑥ **Sense Delay:** Sets a delay in the time it takes the Ambient Detector to go from inactive to Active. This is to compensate for momentary pauses in the Program input;

⑦ **Attack Time:** Sets the time it takes for the output amplitude to rise to 63% of the specified Ratio of the input amplitude once the Ambient Threshold Level is exceeded;

⑧ **Release Time:** Sets the time it takes for the output to fall to 63% of the level set by the Minimum Gain control after the Ambient Detector level drops below the Ambient Threshold Level;

⑨ **In Gain:** Control the Gain of the sensing microphone input;

⑩ **Ambient Level:** Graphically displays the level of Ambient Noise;

⑪ **Applied Gain:** Graphically displays the amount of gain applied to the Program channel;

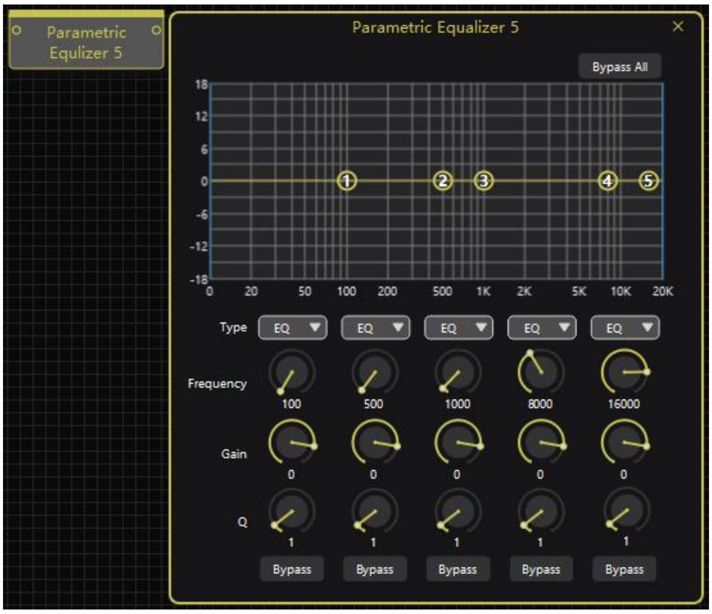
⑫ **Out Gain:** Control the Gain of the Program output;

⑬ **Bypass:** When engaged, the Gated Ambient Compensator is bypassed, audio is passed through without any change.

5.4 Filter Components

5.4.1 Parametric Equalizer

The Parametric Equalizer component is a variable equalizer allowing you to individually adjust the Gain, Bandwidth and center Frequency of up to 8 frequency bands. You can also Bypass individual bands. Additionally, you can change any or all of the bands to either a High-shelf or Low-shelf equalizer.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Type:** Selects the type of equalizer for that band, includes Parametric Equalizer, High-Shelf, Low-Shelf Filter;

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- **High-Shelf:** The High shelf is a gain increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies;
- **Low-Shelf:** The Low shelf is a gain increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling.

② **Frequency:** Sets the center Frequency of an individual band;

③ **Gain:** Controls the Gain for an individual frequency band;

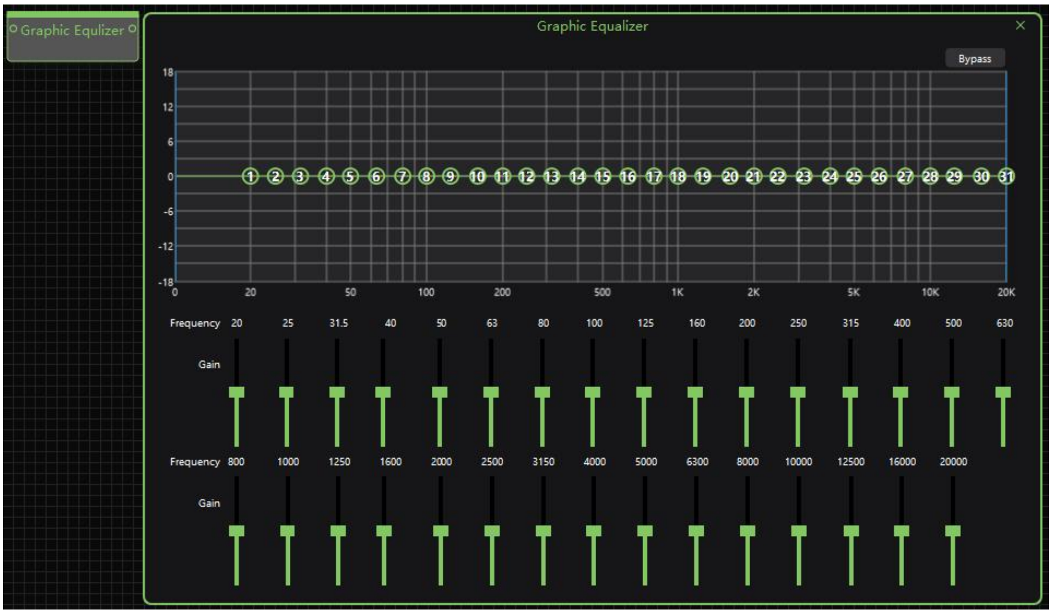
④ **Q-Factor**: Sets the Q-factor of an individual band of the equalizer. This is not active when either Low-shelf or High-shelf Type is selected. Adjusting Q-Factor generally adjusts Bandwidth in an inverse manner;

⑤ **Bypass:** When engaged, an individual frequency band is bypassed, audio is passed through without any change;

⑥ **Bypass All:** When engaged, the Parametric Equalizer is bypassed, audio is passed through without any change.

5.4.2 Graphic Equalizer

The Graphic Equalizer component provides from 31 bands with 1/3th octave per band. Each band can be adjusted from -20dB to +20 dB.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Frequency:** The center frequency is the center point of the current equalization filter and also the reference frequency for gain adjustment. Adjusting the center frequency can change the filter's range of action;

② **Gain:** Controls the output gain of an individual frequency band;

③ **Bypass:** When engaged, the Graphic Equalizer is bypassed, audio is passed through without any change.

5.4.3 FIR High Pass Filter

The FIR High Pass Filter component amplifies frequencies above a set frequency, and attenuates the frequencies below that set frequency.

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Invert:** Inverts the polarity of the output signal;

② **Mute:** Mutes the output signal;

③ **Octave:** Sets the Octave;

④ **Gain:** Sets the output Gain;

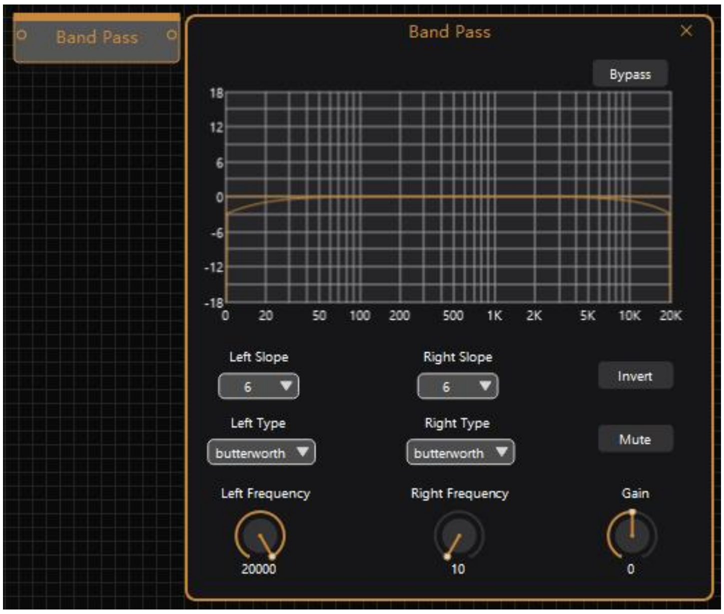
⑤ **Attenuation:** Sets the amount of attenuation of the frequencies below that set frequency;

⑥ **Frequency:** Sets the frequency at which the gain of the output reaches a point 3 dB below the maximum as set by the Gain control. Frequencies below this frequency are attenuated. All frequencies above this frequency are amplified based on the setting of the Gain control;

⑦ **Bypass:** When engaged, the FIR High Pass Filter is bypassed, audio is passed through without any change.

5.4.4 **Band Pass Filter**

The Band Pass filter component divides the audio input signals into 3 frequency bands: low-pass, band-pass and high-pass, you can set the Slope rates and filter types (Butterworth, Linkwitz-Riley, Bessel) for each filter in each band.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Slope:** Determines the rate of change of attenuation at the high-pass and low-pass frequencies of the band. The slope setting establishes the crossover region between two adjacent bands. The Slope includes 0, 6, 12, 18, 24, 32, 36, 42, 48. Note: When the Slopes are both set to 0, the Band Pass Filter passes is bypassed; When the Left Slope is set to 0, it is a high-pass filter; When the Right Slope is set to 0, it is a low-pass filter;

② **Type:** The Type setting includes: Butterworth, Linkwitz-Riley, Bessel filters. You can select any combination of two of these filters for a band's high-pass and low-pass frequencies;

③ **Frequency:** Determines the frequency the band will start (high-pass) and stop (low-pass) passing the audio signal;

④ **Invert:** Inverts the polarity of the output signal;

⑤ **Mute:** Mutes the output signal;

⑥ **Gain:** Controls the output gain;

⑦ **Bypass:** When engaged, the Band Pass Filter is bypassed, audio is passed through without any change.

5.5 Remix Components

5.5.1 **Gain Sharing Automatic Mic Mixer**

The Gain Sharing Automatic Mic Mixer component takes a number of audio inputs, allowing one or more of the inputs to pass while attenuating others based on the level.

The Gain Sharing Automatic Mic Mixer is primarily used for multiple live microphones operating in the same room together as a system – for example, in boardrooms, classrooms, churches, and courtrooms. The Gain Sharing Automatic Mic Mixer controls the live microphones by turning up microphones when someone is talking and turning down microphones that are not used. It is a voice-activated, real-time process without an operator. The Gain Sharing Automatic Mic Mixer controls the additive effect of multiple microphones being on at the same time and adapts to changing background noise conditions. The gain of each microphone input is calculated as the ratio of its RMS level to the combined RMS levels of all inputs. This ensures unity system gain at all times.

1. **Inputs and Outputs**

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and three Output Channels.

① Input Pins: Input signal channel;

② Output Pins: Output signal channel;

③ Last Output pins: Mix Output channel.

2. **Controls**

① **Threshold:** Sets the Input RMS level at which the Applied Gain begins to move away from the Rest Gain. Use this setting to control background noise (such as side conversations) when a mic is not in use;

② **Depth:** Sets the minimum amount of Applied Gain applied to the Input channels;

③ **Rest Gain:** Sets the amount of Applied Gain for all Input channels when none are open. When one or more channels are open, Applied Gain is being distributed towards the open channels, away from the closed channels;

④ **Attack Time:** Determines how quickly Applied Gain moves towards 0 dB (unity gain) once the Input level exceeds the Threshold Level;

⑤ **Hold Time:** Sets the minimum time an Input channel stays open once it is opened, or the length of time an Input channel stays open after the Input RMS level drops below the Threshold Level. Use this control to prevent the gate from opening and closing due to momentary pauses in the input;

⑥ **Release Time:** Determines how quickly Applied Gain moves away from 0 dB once the Input level drops below the Threshold Level and the Hold Time has expired;

⑦ **Detector Time:** This control determines the rate of change of the Input RMS detector output in response to changes in the Input signal. Use this adjustment to prevent abrupt changes in the input signal from causing unwanted, momentary output;

⑧ **Mix Gain:** Adjusts the Mix output channel gain. This applies to all Input channels on the Mix output, but does not affect the individual output channels;

⑨ **Mix Mute:** Mutes the Mix channel output. This mutes all Input channels for the Mix output but does not affect the individual Output Channels;

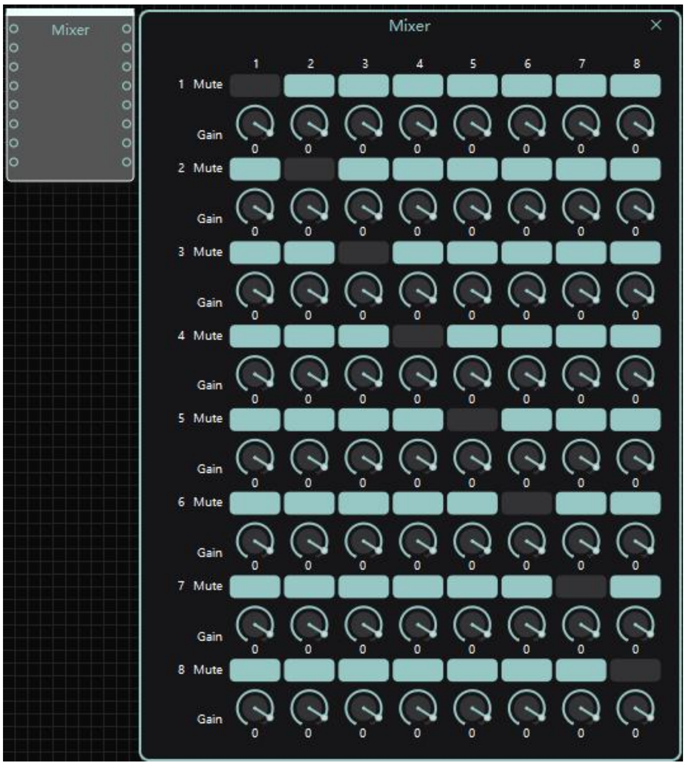
⑩ **Priority Gain:** Adjusts the gain or attenuation applied to a particular Input channel. For example, you can boost the gain of a mic being used by a soft spoken person and attenuate the gain of a louder person so they can be heard equally in the room;

⑪ **Applied Gain:** Measures the attenuation applied to a particular Input channel. The gain is either zero or a negative value. Graphically the meter starts from the top, displaying attenuation in red going down.

5.5.2 **Mixer**

The Matrix Mixer component receives an audio input and distributes the audio signal to each Output channels. For each Output, you can control the Gain and Mute condition of the Input. Likewise, a single Output has Gain and Mute control for each Input.

X-axis is the input channel, Y-axis is the output channel.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides 8 Input Channels and 8 Output Channels.

① **Mute:** Mutes the Input channel;

② **Gain:** Adjusts the Input channel Gain.

5.5.3 Gating Automatic Mic Mixer (Legacy)

The Gated Automatic Mic Mixer (Legacy) component is typically used in applications with multiple microphones, and at times, multiple microphones being used at the same time. The microphones in use can be prioritized using three different modes: Automatic, Priority, and Obstruction or combinations of these. The channels open and close based on Threshold levels set for each channel or manual control. Attenuation controls help control feedback based on the number of microphones open at one time.

1. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and three Output Channels.

① Input Pins: Input signal channel;

② Output Pins: Output signal channel;

③ Last Output Pins: Mix Output channel.

2. Controls

① **Mute:** Mutes the Mix channel output. This mutes all Input channels for the Mix output but does not affect the individual Output channels;

② **Gain:** Adjusts the Mix output channel gain. This applies to all Input channels for the Mix output, but does not affect the individual Output Channels;

③ **ID Gating:** When the ID Gating button is on, will prevent multiple gates from opening if they are fed by the same source such as a talker who is positioned between two microphones;

④ **Last Mic On:** Leaves the last microphone that was used in the open condition, until another microphone exceeds the Threshold;

⑤ **Background Percentage:** This control provides adaptive threshold functionality.This feature allows you to set the microphone Threshold levels fairly low for rapid response, but prevents the gates from opening when the background noise level rises.The background signal is the sum of all the microphone input signals. The Background Percentage knob determines the percentage of background signal used to raise the Threshold levels.Note: Set the Threshold levels when the room is quiet, if the gates open when the room gets noisy, increase the Background Percentage until the gates close;

⑥ **Depth:** The knob sets the amount of attenuation applied to any input channel when the channel's Gate is closed;

⑦ **Hold Time:** The knob the sets minimum time an input stays open once it is opened, or the length of time an input stays open after the RMS input level drops below the Threshold Level. This is to prevent the gate from opening and closing due to momentary pauses in the input;

⑧ **Max NOM:** NOM is counted beginning at input 1 and once the maximum NOM has been reached gates will be closed for the remaining inputs. If you want a limited NOM to always contain the highest priority (open) inputs, assign input 1 the highest priority, and assign the rest of the inputs in descending priority order;

⑨ **Attenuation Type:** When Logarithmic is selected, the output is attenuated by the amount specified by the Attenuation Step control for each open channel every time NOM is doubled; When Linear is selected, the output is attenuated by the amount specified by the Attenuation Step control for each open channel every time NOM increases by 1;

⑩ **Attenuation Step:** The amount of attenuation added every time another mic opens, excluding the first mic opened;

⑪ **Max Attenuation:** Sets the maximum amount of attenuation that can be applied to the output regardless of the NOM;

⑫ **Response Time:** Adjusts the time constant that determines the time it takes for the attenuation to be applied to the output when there is a change in the NOM;

⑬ **NOM:** The total number of open microphones;

⑭ **Gain:** Measures the output gain with respect to the amount of attenuation applied. For example, 40 dB of applied attenuation would result in a -40 dB change in the output gain;

⑮ **Open:** Indicates if the associated channel is open or closed.Note: The Depth knob sets the amount of attenuation applied to any input channel when the channel's Gate is closed;

⑯ **Manual:** Opens the channel until manually closed;

⑰ **Mode:** You can choose one of three modes for the channel;

- **Automatic Mode** opens the channel when the RMS input level exceeds the Threshold;
- **Priority Mode** opens the channel when the RMS input level exceeds the Threshold and the Priority set for the channel is higher than any other Priority channel currently open;
- **Obstruction Mode** opens the channel when the RMS input level exceeds the Threshold no other channels are open.

⑱ **Priority:** Used to set the priority order of the channels. All inputs with the same priority can be open at the same time. This setting has no effect when the channel is not in Priority Mode.Note: The higher the number, the higher the Priority. Tip: Leave gaps in your priority lists so you can easily change the priority of one mic without having to shift other priorities to make room for a single change;

⑲ **Threshold:** Sets the peak level at which the gate opens and allows audio to pass on this channel.

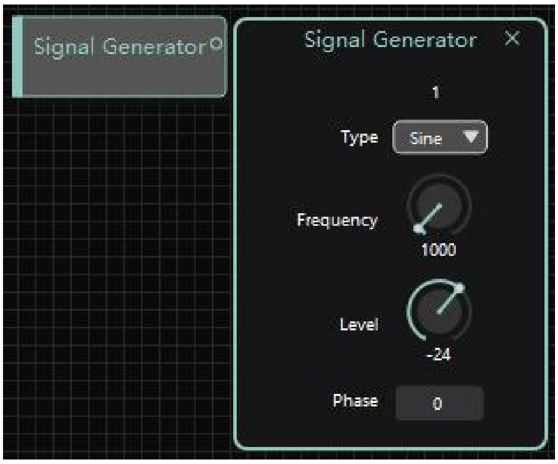
5.6 Other Components

5.6.1 Signal Generator

The **Sine Wave** is a pure audio signal with a single, constant frequency, amplitude, and phase, and a smooth, periodic waveform.

The **Pink Noise** produces random frequencies distributed uniformly by octave throughout the audio spectrum.

The **White Noise** is a random noise whose power spectral density is constant throughout the frequency domain, that is, all frequencies have the same energy density.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Output Channel.

① **Type:** Controls the type of signal generated by the output channel, includes Sine, Pink, White;

② **Frequency:** Controls the signal frequency for the Output channel;

③ **Level:** Adjusts the Gain for the individual Output channel;

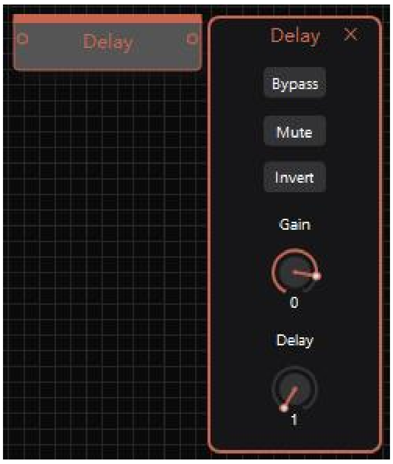
④ **Phase:** Controls the phase for the individual Output channel.

5.6.2 Delay

The Delay component introduces a delay in the audio signal in order to compensate for the placement of speakers.

Delay Effect Implementation:

- **Reverb Effect:** By setting an appropriate delay time, the reverb effect simulates the reflection and diffusion of sound in a space, enhancing the sound's spatial and three-dimensional quality, as if you were immersed in a specific acoustic environment.
- **Echo Effect:** By using a delay unit to generate repeated sound signals, natural echoes are simulated, enhancing the layering and depth of the sound.
- **Sound Optimization:** In larger performance venues, delay units can be used to assist with speaker processing. By applying different delays to different speakers, sound is distributed evenly throughout the space, avoiding sound overlap and interference, and optimizing the overall sound field effect.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Mute:** Mutes the individual audio signal;

② **Invert:** Inverts the polarity of the individual audio signal;

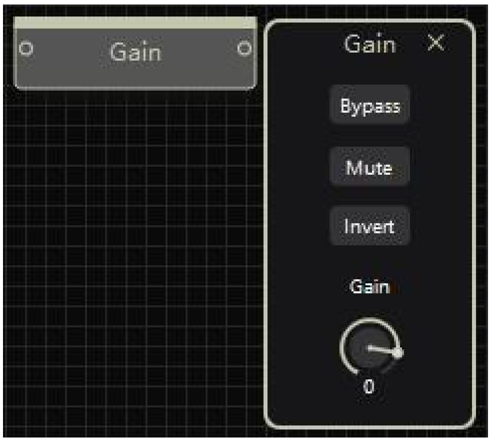
③ **Gain:** Controls the gain of the individual audio signal;

④ **Delay:** Controls the delay time of the individual audio signal;

⑤ **Bypass:** When engaged, the Delay is bypassed, audio is passed through without any change.

5.6.3 **Gain**

The Gain component is used to increase or attenuate the audio signal of all Channels simultaneously. You can simultaneously Bypass, Invert, and Mute all the Channels as well.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Mute:** Mutes the Output for the individual channel;

② **Invert:** Inverts the polarity of the Output signal for the individual channel;

③ **Gain:** Adjusts the Output gain for the individual channel;

④ **Bypass:** When engaged, the Gain is bypassed, audio is passed through without any change.

5.6.4 Gain Ramp

The Gain Ramp component provides the capability to ramp an input signal from one level to another in a specified amount of time, and to ramp back to the original level. You can adjust both Target Gains individually, as well as the Ramp Time in each direction. Once a Target Gain is reached, the output remains at that level until it is manually released to ramp to the other Target Gain.

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Rate A/B (dB/s):** Rate A displays the decibels per second it takes the output level to go from Gain B to Gain A. Rate B displays the same as above except from Gain B to Gain A. Rate A or B = ((| Gain A| - | Gain B|)/ Time A or B);

② **Time A/B:** Sets the time it takes for ramp A (or B) to rise to the Target Gain set for ramp A (or B);

③ **Time Unmute:** Sets the time when the output is unmuted;

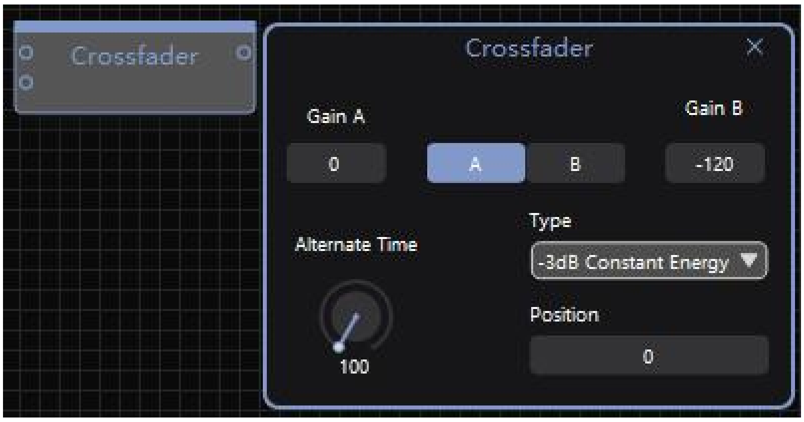
④ **Gain A/B:** Gain A adjusts the output level when the B button is disengaged. It is the Target Gain when the ramp goes from Gain B to Gain A. Gain B adjust the output level when the B button is engaged. It is the Target Gain when the ramp goes from Gain A to Gain B;

⑤ **Mute:** Mutes the output of the Gain Ramp component.

5.6.5 **Crossfader**

The Crossfader component allows you to fade between two input channels going to one output. You can specify a mono Crossfader, having two inputs and one output. The number of inputs is always double the number of outputs.

You manually trigger the fade from Input A to Input B, and back. You can control the time it takes to fade from one channel to another.



1. **Inputs and Outputs**

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and one Output Channel.

① Input Pins: Input A signal channel;

② Last Input Pins: Input B signal channel;

③ Output Pins: Output signal channel.

2. **Controls**

① **Gain A/B (dB):** Displays the current gain value for each channel. During a crossfade operation, one value is increasing, the other decreasing;

② **Position (%):** Displays the positional balance between Input A and Input B. Zero percent (0%) the output is equal to Input A. One hundred (100%) the output is equal to Input B;

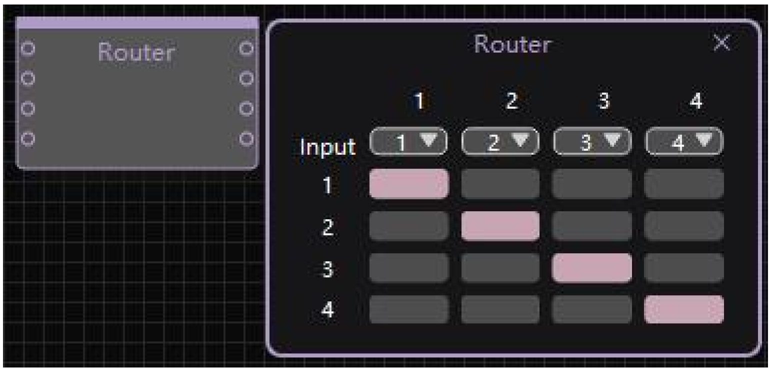
③ **Alternate Time:** The Time controls the length of time it takes the Crossfader to fade from one channel to the other;

④ **Type:** The Type control allows selection of the crossfade **Type:** -6 dB constant gain or -3 dB constant power. With -6 dB constant gain type selected, the Crossfader sums the input signals with a constant combined gain. The midpoint gain is -6dB. This setting is suited for correlated input signals but results in a 3 dB dip at the midpoint for uncorrelated signals. With -3 dB constant power type selected, the Crossfader sums the input signals to a constant combined power. The midpoint gain is -3dB. This setting is suited for uncorrelated input signals and is the preferred one.

5.6.6 **Router**

The Router component allows you to route any of the available audio Inputs to any of the available Outputs. You can choose Mono to route inputs to outputs one at a time, Stereo to simultaneously route the left and right Channels of an Input to the left and right Channels of any available Output, or Multi-channel to simultaneously route all Input Channels to any Output.

Y-axis is the input channel, X-axis is the output channel.



The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides four Input Channels and four Output Channels.

① Input: You can select the Input column for each Output row. You can have one Input feeding multiple Outputs.

5.6.7 **Acoustic Notch Feedback Controller**

The Acoustic Notch Feedback Controller component allows you to set feedback suppression based on the room, and automatically suppress feedback. The Notch Feedback Controller reduces the gain at the notch frequency while affecting the gain at surrounding frequencies as little as possible. Audio engineers sometimes insert parametric notch filters manually at peak frequencies, prior to the live event, to help prevent undesirable feedback from occurring. This is called "ringing out the room". Notch Feedback Control is an automated way of "ringing out the room".The Notch Feedback Controller allows the undesirable feedback to begin, and then detects the frequency at which the undesirable feedback is occurring. The AFC then places a notch filter at this frequency.

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides one Input Channel and one Output Channel.

① **Auto Set Able:** Set auto-detect mode to automatically detect feedback points and reduce the gain of the segment;

② **Q Link:** When the Q-Link button is engaged, adjusting the bandwidth value of one bandwidth segment will change the other segments in synchronization with this segment;

③ **Threshold:** When a frequency's amplitude exceeds this level, a filter is activated. in addition to exceeding the Feedback Threshold the amplitude at a particular frequency must exceed the average amplitude of the audio spectrum by an internally defined amount;

④ **Max Spec:** Numerically displays the peak level of the input signal;

⑤ **Max Spec Freq:** Numerically displays the frequency corresponding to the peak of the input signal;

⑥ **Set Flag:** Set the number of filters to use, one for each double-click;

⑦ **Band Able:** Enable or disable a segment filter;

⑧ **Frequency:** Detects the frequency of the input signal;

⑨ **Gain:** Controls the overall gain through the Notch Feedback Controller component;

⑩ **Bypass:** When engaged, the Notch Feedback Controller is bypassed, audio is passed through without any change.

5.6.8 **Acoustic Echo Canceler**

The multi-channel Acoustic Echo Canceler (AEC) component is used in conference rooms (Near-End) and other installations where people call in from remote locations. The remote caller's (Far-End caller's) voice is broadcast over loudspeakers in the conference room. The sound is picked up by microphones in the conference room and echoed back to the Far-End caller. The purpose of the AEC is to eliminate these echoes while at the same time allowing the Far-End caller to hear clearly, what people in the room are saying.

Each microphone in the conference room is plugged into one channel of the AEC component. Each channel also receives the loudspeaker signal that carries the remote talker's voice. This is called the AEC reference signal. To remove the echoes, the AEC subtracts a filtered version of the reference signal from the microphone signal.

1. Inputs and Outputs

The number of signal pins is variable and set in the component's Channel number Property. By default, the component provides two Input Channels and one Output Channel.

① Input Pins: Local microphone Input signal channel;

② Last Input Pins: Remote reference Input signal channel;

③ Output Pins: Local Output signal channel.

2. Controls

① **AEC Level:** Selectable 128ms, 256ms, 512ms;

② **Duplex Detection:** Selectable 0, 5, 10;

③ **Nonlinear Processing Level:** Selectable 0, 5, 10;

④ **Bypass:** When engaged, the Acoustic Echo Canceler is bypassed, audio is passed through without any change.

Chapter 6 User Control Interface (UCI) Design

6.1 Controls

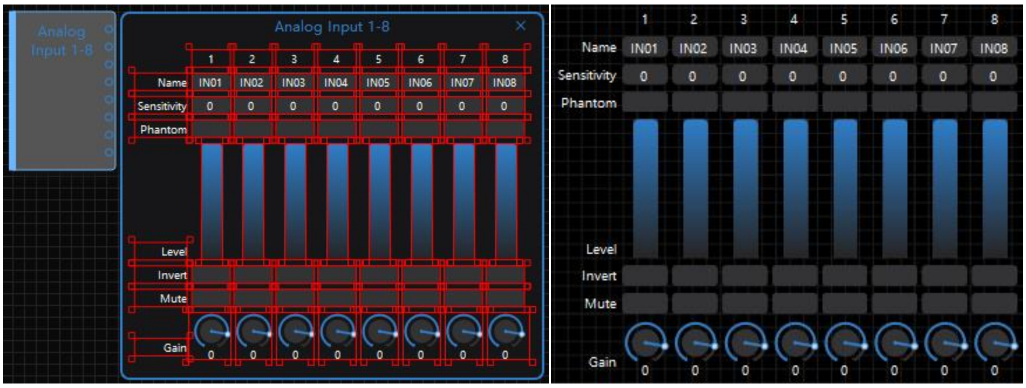
You can drag controls into the UCI interface by selecting them from the component's control panel, and you can arrange the controls you need in the UCI interface by arranging them in other ways. They can be copied and pasted, and the control you paste is associated with the copied control. This means that you adjust one control and the other controls change with it.

1. Open the component control panel of the control you want to copy;
2. Select the control you want to copy, the text;
3. Select the control and hold down the left mouse to drag it out to the UCI interface.

   ① After dragging the control out, you can change the size of the control by selecting the control and dragging its handle;

   ② If you copy a control and then delete the component it came from, the control will then become an invalid control.

Note: You cannot perform copy and paste functions in simulation mode.



6.2 User Control Interface

Layers and pages can be added or deleted in the user interface, and you can also set the page size and background color by selecting the page. When adding a layer, please select the page you want to add the layer to, otherwise it will be added to the first page by default.

6.3 Component types

UCI:

Type	Function
Label	Provide read-only type of text box
Panel	Background panel for User Control Interface design with background color and image settings
PageSwitchButton	You can specify the page to jump to, set the background color and add text and pictures
LayerShowButton	You can specify the layer you want to jump to, set the background color and add text and pictures before and after the jump
SceneLoadButton	You can specify the ID of the scene you want to jump to, set the background color and add text and pictures before and after the jump
ScenceSaveButton	You can save the scene parameters, set the background color and add text and pictures

Controls:

Type	Function
CheckButton	Provides settings to enable and disable a control
EditBox	Control values can be entered with the keyboard
Meter	Provides level meter display status
Knob	Provide adjustable knob control
ComboBox	Provides a drop-down list of available parameters
Button	Buttons that can be set for the press action
RadioButton	Buttons that can be set for the press action cannot be canceled after being selected
List	Displays the list of read files
Slider	Provides adjustable fade-out type control Can be vertical or horizontal

6.4 Component properties

Shared:

Type	Function
Location	The position of the control can be modified
Type	The type of the control can be modified
Size	Modify the size of the control
Test Style	You can modify the text font size and font type of the control
Text Color	You can modify the color of the control text
Vertical Alignment	You can modify the vertical top, vertical center and vertical bottom alignment of the control text

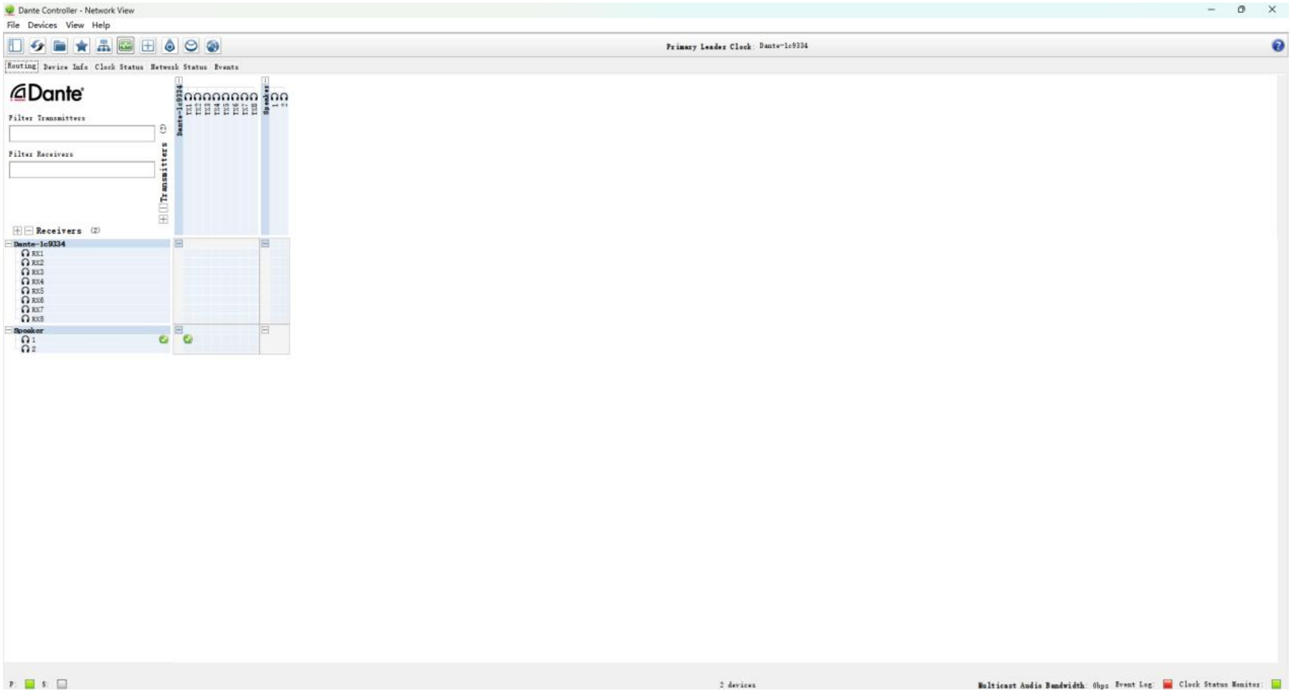
Horizontal Alignment	You can modify the horizontal left, horizontal center, and horizontal right alignment of the control text

Others:

Type	Function
Background	The background color of the control can be modified
Border Width	The border width of the control can be modified
Border Color	You can modify the border color of the control
Check Color	You can modify the color of the control when it is clicked
Uncheck Color	You can modify the color of the control when it is not clicked
Check Text	Modifies the text of the control when it is clicked
Uncheck Text	Modifies the text of the control when it is not clicked
Check Image	Modifies the image of the control when it is clicked
Uncheck Image	Modifies the image of the control when it is not clicked
Max Value	The maximum value that the control can control can be modified
Min Value	Modifies the minimum value of the control
IsTextReadOnly	Modifies the EditBox control to allow only text to be read, not edited
IsNumTextBox	Modifies the EditBox control to allow only numbers to be entered

Chapter 7 Dante Network Audio Routing

In a Dante audio network, the Dante Controller software is required to set up the routing of the various signals accessing the processor. It can realize 1-to-1, 1-to-N mapping operation from input to output within Dante network.



The software "Dante Controller" is free to download from the company of Audinate (the owner of Dante technology).To install the software on the computer,please visit the link: https://www.getdante.com/products/software-essentials/dante-controller.

And the "User Guide" of "Dante Controller" is available on the Audinate websit: https://dev.audinate.com/GA/dante-controller/userguide/webhelp/content/front_page.htm.

Note: 1, Dante can not run in the Wi-Fi connection environment, is dependent on a reliable and secure wired network environment to transmit perfect audio;

2, Dante Controller software corresponds to the platform of Windows 7, Windows 10, Windows 11, macOS, please select the appropriate software version according to your system platform.

Chapter 8 FAQ

8.1 Abnormal power indicator

No light: First, check whether the power connection and device power supply are normal; Second, check whether the power switch on the rear panel of the device is on.

Blinking: Unplug all GPIO connection cables and reboot the device.

If the problem is not solved, please contact the distributor or manufacturer.

8.2 Abnormal status indicator

After 18 seconds of power-on, the system is working normally and the system light of the device should be flashing once per second.

Does not light up, often light up, or blinks rapidly:

First, system error, contact the after-sales service to upgrade the software version; Second, long press the reset button (R hole on the rear panel of the device) for more than 6 seconds, the device will restore the factory settings and restart automatically. If the problem is not solved, please contact the distributor or manufacturer.

8.3 Channel no sound

First, check whether the audio source, audio input and output wiring is normal; Second, check whether the mute function of the corresponding audio channel is enabled, if the mute switch has been turned on, please turn off the mute switch; Third, check whether the settings of the corresponding channel's input processing, matrix mixing and output processing are normal; Fourth, check that the audio signals are routed correctly in the Dante Controller. If the problem is not solved, please contact the distributor or manufacturer.

8.4 The control software cannot search the device

First, check whether the system light of the device is in normal blinking state; Second, check whether the network connection is normal; Third, ensure the network accessibility between the configuration host and the device; Fourth, press and hold down the reset button (R-hole on the rear panel of the device) for more than 6 seconds, and the device will restore the factory settings and restart automatically. If the problem is not solved, please contact the distributor or manufacturer.

8.5 Network connection failure

Network connection failure is usually caused by different network segments of the device. If the LAN and the processor network segments are different, you can connect the processor directly via PC, login to the device configuration interface, change the processor network segment to be the same as the LAN and then access the LAN. (**Note:** The processor's IP address is automatically obtained by default, if the LAN is automatically obtained IP address, you do not need to log in the device to change the IP address).

8.6 Current noise in output channel

Please check whether the processor is well grounded, which usually requires the grounding screw on the left side of the rear panel of the chassis to be connected to the metal enclosure such as the cabinet through a metal wire. If the problem still exists, please check the wiring of the input devices. If the input devices are unbalanced (two wires), please connect the "+" and "G" of the input connector of the processor.

8.7 How to recognize system noise

After the system is set up, there is noise troubleshooting: First, unplug the device output audio cable, there is noise, please check the causes of the back stage equipment; Second, restore the output wiring, mute the corresponding output channel, there is noise, if unbalanced connection, try to shorten the connecting line, to avoid the introduction of interference, if balanced connection, try to disconnect the ground wire; Third, cancel the corresponding channel mute, unplug the device input audio cable, there is noise, long press the reset button (R hole on the rear panel of the device) for more than 6 seconds, the device will restore the factory settings and restart automatically;Fourth, restore the input wiring, turn off the audio source, there is noise, check the input connection, refer to the second point of the processing; Fifth, check the audio source is there is noise. If the problem is not solved, please contact the distributor or manufacturer.

8.8 RS232 center control command does not work

First, check whether the connection is normal, the central control host TX connected to the device's RX, the central control host RX connected to the device's TX, the central control host and the device's ground interconnect; Second, check the software configuration of the device interface items: baud rate, start bit, stop bit, etc. Settings are the same as the interface configuration of the central control host. If the problem is not solved, please contact the distributor or manufacturer.

8.9 U disk audio can not be played

The device only supports FAT32 format U disk, only supports MP3 or WAV format audio files, audio files need to be stored in the root directory; ensure that the U disk is writable, the device needs to create a playlist. U disk has more than one partition only recognizes the first partition.

Chapter 9 Packing List

Device	Power Cable	Quick Guide	12pin Phoenix Connector	3pin Phoenix Connector	Small Screwdriver
1PCS	1PCS	1PCS	9PCS	1PCS	1PCS

Chapter 10 Specification

Category	Parameter Item	Parameter Description
Input	Analog Input Channels	Mic/Line 16
	Dante Input channels	64
	Input Frequency Response	20Hz～20KHz@+18dBu, ±0.2dB 50Hz～20KHz@+18dBu, ±0.1dB
	Input THD+N	@ +18dBu Sensitivity & +8dBu Input < 0.003%
	Input Equivalent Input Noise	(Unweighted 20Hz-20kHz) < -125dB
	Inputs Crosstalk @1kHz	>110dB typical, 100dB max
	Input Dynamic Range	@ +18dBu Sensitivity > 110 dB
	Input Common Mode Noise Rejection	@ +18dBu Sensitivity 60dB
	Input Impedance (Balanced)	2.4kΩ nominal
	Input Sensitivity Range (3dB)	-39dBu～+18dBu
	Phantom Power	+48V DC, input current max 10mA
	Sampling Rate	48KHz
	AD/DA Conversion	24-bit
Output	Analog Output Channels	16
	Dante Output channels	64
	Frequency Response	20Hz-20KHz @ all environments, ±0.2dB

	THD+N	0.003%, +18dBu max output level
	Outputs Crosstalk @ 1kHz	>100dB typical, 90dB maximum
	Output Dynamic Range	>108dB
	Output Impedance (Balanced)	100Ω
	Output Level Range	18dBu/4dBu
USB	USB A	32bit float
	Channels	2×2
	Sample rate	48KHz
General specification	Display	Displays the device IP address and device name
	Noise Floor	-90dBu
	Power	35W (Nominal power), 65W (Maximum power)
	Power Supply	AC 100V～240V, 50Hz/60Hz
	Operating Temperature	0～40℃
	Operating Humidity	10%～90%RH, No condensation
	Product Weight	2.5kg
	Product Dimensions (LWH)	482.4mm×260.5mm×44mm
	Package Weight	3.2kg
	Package Dimensions (LWH)	590mm×340mm×110mm

Warranty Regulations

The warranty period of this product is 1 year.

In the warranty period of non-man-made damage caused by the product performance failure can enjoy three packages of service.

Warranty card by the sales unit stamped after the effective. The alteration is invalid!

The following conditions (**including, but not limited to, this**) **are not covered by the three-package service:**

1. No warranty card or missing valid invoice or the date has exceeded the validity period of the three packages of services;
2. Not in accordance with the requirements of the product instructions for use, maintenance, management and damage caused;
3. The product model or code on the warranty voucher does not match the physical goods;
4. Damage caused by the dismantling and repair of non-authorised service providers;
5. Normal discolouration, wear and tear and consumption during the use of the product are not covered by the warranty;
6. The product cannot be used due to the user's own network reasons, please consult customer service staff.

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