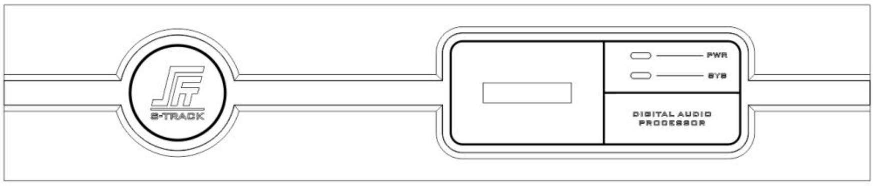

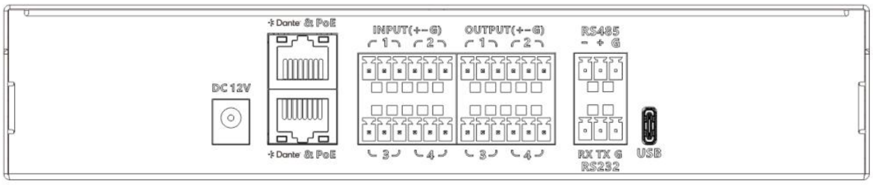

DBOX X44-U
Digital Signal Processor

User Manual

Preface

The purpose of this section is to ensure that you can safely and correctly use the product by following this manual, thus avoiding operational risks or damage to property.Before using this product, please read the product manual carefully and keep it for future reference.

Outlined

This manual applies to Digital Signal Processor.

It describes the functions and usage of various functional modules of the Digital Signal Processor and guides you through its installation and commissioning process.

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 input power of the equipment power adapter is 100V-240V, 50/60Hz AC power.
- 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.

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Chapter 1 Product Introduction

1.1 Introduction

The device supports analog channel 4 into 4 out, USB playback and recording, comprehensive matrix mixing function, equipped with high-performance A/D D/A converter and 32-bit floating-point DSP processor, support 24bit/48KHz sampling frequency, restore high-quality sound, user-friendly operation software interface, easy to operate and powerful performance. Mainly used in a variety of large places, can meet the theater, concert halls, remote video conferencing, stadiums, churches, conference centers, theme parks, public sound reinforcement systems and other aspects of the application needs.

1.2 Product Features

- Highly integrated, integrating a variety of traditional Analog audio processing equipment in a Digital Signal Processor;
- High-performance 32-bit floating-point DSP processor, all-digital processing, fast response to AFC (Acoustic Feedback Canceler), AEC (Acoustic Echo Canceler), ANS (Adaptive Noise Suppressor), Ducker and other audio processing;
- High-performance A/D, D/A converter, 24bit/48KHz sampling frequency, high-quality Analog → Digital, Digital → Analog conversion;
- 4 Analog input channels and 4 Analog output channels, very small distortion and ultra-low noise floor;
- 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 functions;
- Scene storage is different from the Analog equipment is one of the most practical and significant features, can store 8 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

- Comprehensive matrix mixing function, 24bit/48KHz sampling frequency, high performance A/D, D/A converter and 32-bit floating point DSP processor;
- Inputs per channel: Preamplifier, 5-band Parametric Equalizer,Compressor, Ducker, AFC, AEC, ANS;

- ✧ Outputs per channel:31-band Graphic Equalizer, Limiter;
- ✧ Display shows the IP address;
- ✧ Input mute button, phantom power button;
- ✧ Output mute button, invert button per channel;
- ✧ Ethernet multi-purpose data transmission and control port, can support real-time management of single and multiple devices;
- ✧ Intuitive image, simple and easy to understand the graphical software control interface, for customers to bring fast, real-time operating experience;
- ✧ Support PC client software control, device built-in client, support for online downloads;
- ✧ Extendable USB-C interface, support USB Audio Card;
- ✧ Configuration of bi-directional RS232 interface, RS485 interface, standard Ethernet control interface;
- ✧ POE mode power supply, can be used with 802.3af/at standard POE switch;
- ✧ Support 8 groups of scene presets, scene create, save, delete and other functions;
- ✧ Intuitive, graphical software control interface, works on Windows XP, 7, 8, 10, 11, etc.;
- ✧ Provides network audio expansion via Dante protocol up to 8 (4*4) channels.

Chapter 2 Specification

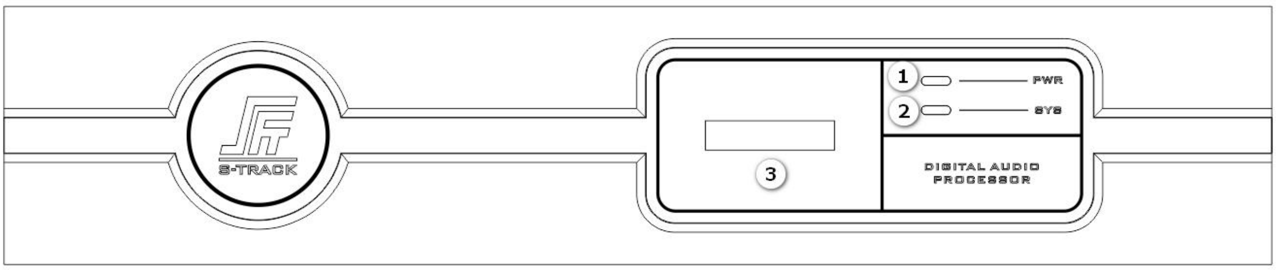
Category	Parameter Item	Parameter Description
Peripherals	Input Interfaces	4 Analog and 4 Dante
	Output Interfaces	4 Analog and 4 Dante
	Display	Displays the device IP address
	Control Interfaces	2 RJ45 interface, 1 RS232 interface, 1 RS485 interface
Audio processing	Processor	TI 456MHz FLOPS dual-core 32-bit DSP processor; 24-bit A/D and D/A converter, 48kHz sampling rate

	Input Channel	Functional module: Preamplifier, 5-band Parametric Equalizer, Compressor, Ducker, Acoustic Feedback Canceler (AFC), Acoustic Echo Canceler (AEC), Adaptive Noise Suppressor (ANS), Ducker. Physical interface: Balanced Phoenix terminals.
	Output Channel	31-band Graphic Equalizer, Limiter. Physical interface: Balanced Phoenix terminals
	Phantom Power	DC 48V
	Input Impedance	Balanced: 20KΩ
	Output Impedance	Balanced: 100Ω
	Common Mode Rejection Ratio	52dB@50Hz
	Frequency Response	20Hz～20KHz, ±0.15dB
	Noise Floor	-89dBu
	Signal-to-Noise Ratio	105dB
	THD+N	≤0.003% @1kHz, +4dBu
	Channel Isolation	100dB@1kHz, 4dBu
	Input Range	≤+18dBu (A-Weighting)
	Equalizer	Parametric Equalizer: Frequency: 20 to 20kHz, Gain: -15 to +15dB, Bandwidth: 0.02 to 4 Graphic Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB
	Maximum Output Level	18dBu

	Maximum Input Level	18dBu
	Analog/Digital Dynamic Range	120dB
	Digital/Analog Dynamic Range	120dB
General specification	Operating Voltage	Power adapter input: AC 100V～240V, 50Hz/60Hz; Output: DC 12V/2A; OR POE power supply
	Maximum Power	15W
	Operating Temperature and Humidity	-10℃～40℃, 10%～90%RH, No condensation
	Chassis	1/2U
	Product Dimensions (L×W×H)	216mm×190.5mm×44mm
	Net weight	1.2kg
	Package Dimensions (L×W×H)	330mm×230mm×70mm
	Gross Weight	1.6kg

Chapter 3 Interface Description

3.1 Front Panel

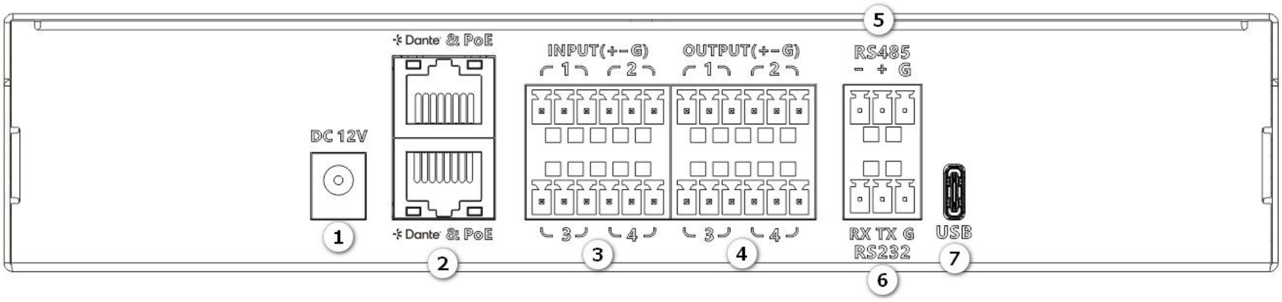


① PWR: Power indicator, the indicator light is always on to indicate that the device is powered normally;

② SYS: System operation indicator, the indicator will flash once per second to indicate that the system is operating normally;

③ Display: Displays the IP address.

3.2 Rear Panel



① DC 12V: This power connector is used to attach a DC 12V/2A power adapter;

② Dante POE network power supply interface: This network interface facilitates Dante digital signal transmission, connection to PC interactive control software, and provides direct power to the device;

③ INPUT signal input interface: This can be connected to microphones, PCs, and other devices;

④ OUTPUT signal output interface: This can be connected to amplifiers, active speakers, and other devices;

⑤ RS485: The RS485 serial interface allows for the connection of control terminals or centralized control devices;

⑥ RS232: The RS232 serial interface enables the attachment of control terminals or centralized control devices;

⑦ USB-C interface: This supports a USB-C sound card.

Chapter 4 Instructions for Use

4.1 Software Download

To download the installation software, source files are embedded within the Digital Signal Processor device itself. Simply enter the device’s factory default IP address (default IP: 192.168.1.200) into the URL bar of your browser. This action will direct you to the download interface. Follow the instructions on the web interface to initiate the software download.

Before installing the software, please ensure that your PC meets the required specifications. Your PC must have the .NET Framework 3.5 or above installed for Windows systems.

Note: Ensure that your PC is on the same network segment as the device's IP address (default IP: 192.168.1.200, subnet mask: 255.255.255.0) when downloading the software; otherwise, access will be restricted.

Web downloads:

Software DOWNLOAD:

4.2 Main Interface

① Search the device list area;

② Control and status display area;

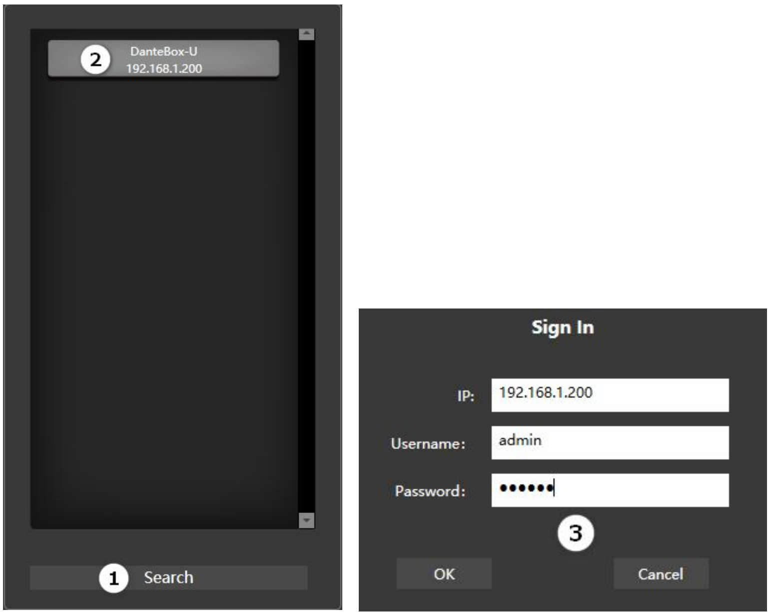
③ A1...D4: Channel selection area, A1...A4 are Analog channels, D1...D4 are Dante channels;

④ INPUT...OUTPUT: Audio Processing Modules;

⑤ Scene and Setting area;

4.3 Search the device list area

4.3.1 Sign In



① Search button: Click this button to search for devices that are online on the LAN

② Displays the device name and IP address of the searched list of online devices;

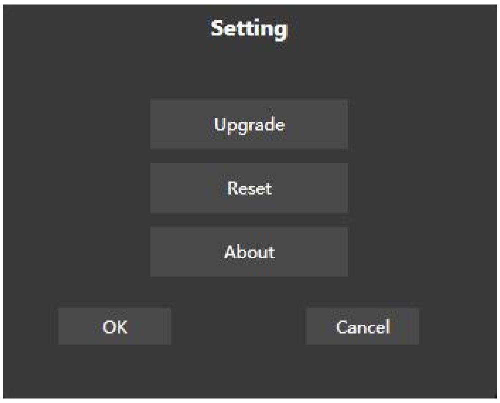
③ Double-click the device name in the device list, Sign In pop-up window, enter the user name and password: (default user name: **admin**, password: **123456**) Click [OK] button to enter;

4.3.2 Modify the IP address

① If you need to modify the device IP address information, **right-click** the device name, will pop-up window Setting;

② Modify the device IP address as needed.

4.3.3 Setting

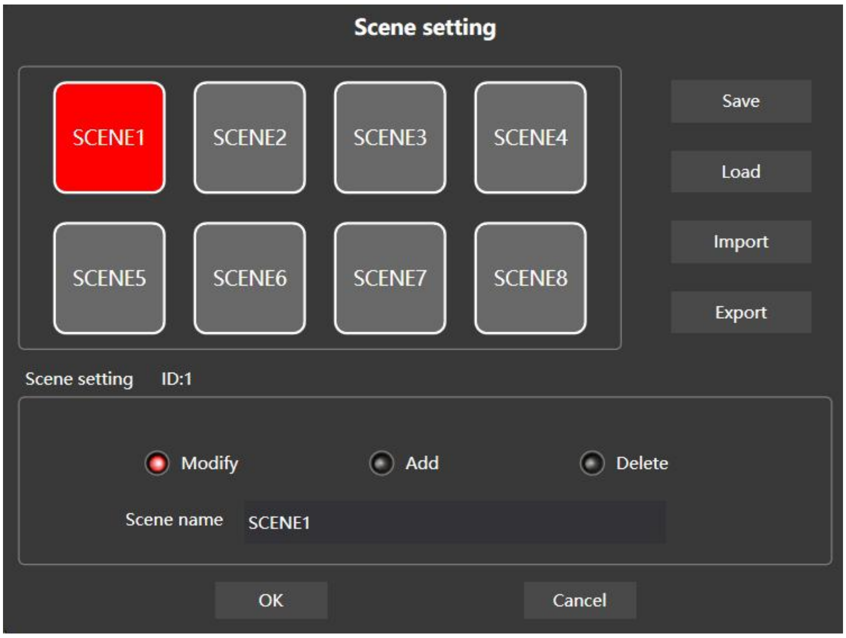


① Upgrade: Device firmware upgrade;

② Reset: Restore factory default status;

③ About: View device serial number information.

4.3.4 Scene setting



① Save: Saves the current parameters to the selected scene;

② Load: Loads the selected scene;

③ Import: Import the saved scene file;

④ Export: Export the scene locally;

⑤ Modify: Modify the scene name;

⑥ Add: Add new scene, maximum 8 scenes can be added;

⑦ Delete: Delete the scene.

4.4 Pre-processing

4.4.1 Input

The Analog Input provides 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 one three-terminal 3.5mm Euro style connectors.

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

① Indicator light: The indicator light changes color to reflect the signal level in the input channel: it is gray when no signal is present, turns green with a small signal, and red with a large signal;

② 48V: Use the 48V Phantom power supply switch judiciously. Turning it on for non-condensor microphones may cause damage;

③ Mute: Mutes the input signal;

④ Channel fader: Channel gain value range (0~-72dBFS), can be controlled by the fader.

4.4.2 Parametric Equalizer

The Parametric Equalizer is a variable equalizer allowing you to individually adjust the Gain, Bandwidth and center Frequency of up to 5 frequency bands. You can also Bypass individual bands.

The Parametric Equalizer is mainly used to modify over-emphasised or missing frequency ranges. Whether the frequency range is narrowed or widened, the Parametric Equalizer can help to repair the narrowed frequencies or widen the frequency range to achieve the ideal signal tone.

① Bypass All/Active: Bypass All or Active the Parametric Equalizer for the current channel; When the Parametric Equalizer is bypassed, audio is passed through without any change;

② Reset: All band filter parameters are restored to the default;

③ Bypass (1~5)/Active (1~5): Bypass or Active the Parametric Equalizer for an individual frequency band; When an individual frequency band is bypassed, audio is passed through without any change;

④ Frequency (20~20000): Sets the center Frequency of an individual band;

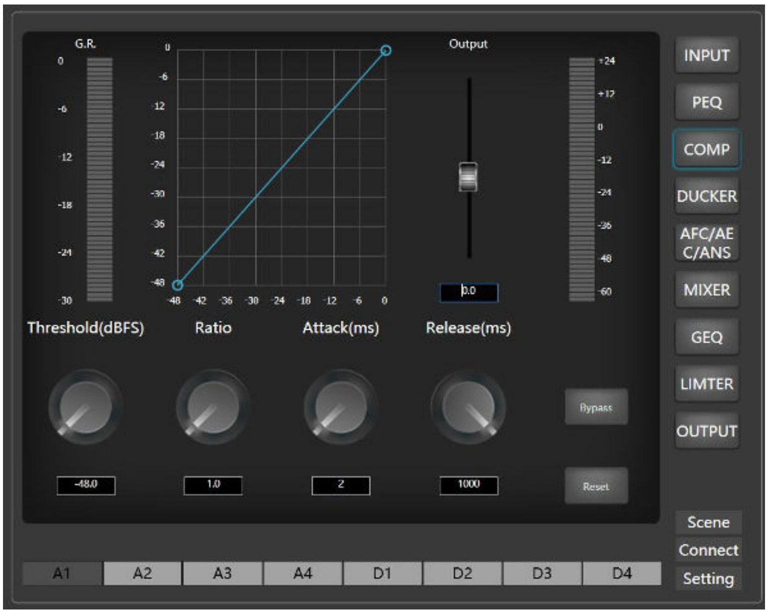
⑤ Gain (-15~15): Controls the Gain for an individual frequency band;

⑥ Bandwidth (Octave): Sets the bandwidth of an individual band of the equalizer, from 0 octaves to 4.00 octaves (default is 1.00). This is not active when either Low-shelf or High-shelf Type is selected. Adjusting bandwidth generally adjusts Q-Factor in an inverse manner.

4.4.3 Compressor

The purpose of the Compressor 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 (20:1 - very little amplitude variation) Output.



① Threshold (-48~0): Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. 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 -15 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 - (-15 dB)] / 2.5} + (-15 dB) = -5 dB.

② Ratio (1~20): The ratio between the Input and the Output as measured from the Threshold Level. The closer the Ratio is to 20, the smaller dynamic changes in the Output level. As the Ratio is adjusted closer to 1, the dynamic range of the Output increases;

③ Attack Time (1~1000): Time between when the input level reaches the threshold and when the compressor is activated;

④ Release Time (1~1000): Time between when the input level is less than the threshold and when the compressor stops working completely;

⑤ Output Gain (-24~30): Controls the Gain of the output;

⑥ Bypass/Active: Bypass or Active the Compressor for the current channel, When the Compressor is bypassed, audio is passed through without any change;

⑦ Reset: Resets the parameters to the default.

4.4.4 Ducker

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

- How long the Paging Input channels are held at the gain level;
- How fast the reduced gain is applied to the Paging Input 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.

Typical application:

- 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 Paging Input: Priority Input channel (example microphone);

② Background Input: Input signal channel (example background music);

③ Threshold (-60~0dB): The Threshold Level is the RMS level of the Priority channel at which the Ducker activates;

④ Depth (0~96dB): The Depth sets the amount of attenuation applied to the Background input channels when the Ducker is activated. The Paging Input channel is mixed with the main outputs, so the Depth setting controls how much of the main channels are heard in the output;

⑤ Attack Time (10~500ms): The Attack Time is the time it takes the Background Input channel output to fall to 63% of the Depth level when the Ducker is activated.Use this control to provide a smooth transition from the Background Input channel audio to the Paging Input channel audio;

⑥ Hold Time (1~30000ms): The Hold Time determines how long the Background Input channel stays at Depth once the level drops below the Threshold.This is to prevent the Background Input channel from opening and closing due to momentary pauses in the Paging Input channel input;

⑦ Release Time (10~10000): The Release Time is the time it takes the Background Input channel output to return to 63% of its normal level when the Ducker is deactivated and the Hold Time expires.Use this control to provide a smooth transition from the Paging Input channel audio back to the Background Input channel audio;

⑧ Reduction: Graphically displays the amount of attenuation applied to the Background Input channels.

⑨ Local Output: Select the channel for local output of the Background Input; the call input needs to be mixed to the local output channel in the matrix mix;

⑩ Bypass/Active: Bypass or Active the Ducker. When the Ducker is bypassed, audio is passed through without any change;

⑪ Reset: Resets the parameter to the default.

4.5 AFC, AEC, ANS

4.5.1 AFC

The Acoustic Feedback Canceler (AFC) is used to cancel the whistling generated between the microphone and speakers in the sound reinforcement system, thus capturing the frequency that causes the whistling for attenuation to ensure the quality of the sound as well as to prevent burning out of the amplifier or speakers.

Feedback: Selects the signal that needs to be processed by the Acoustic Feedback Canceler, and the processed signal selects the output channel in the mixer;

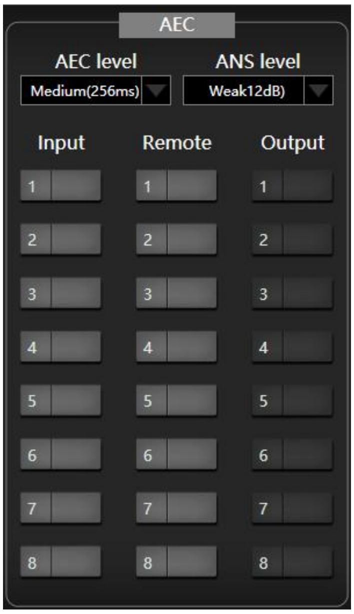
① Local Input: The channel for local Mic output, i.e. the signal that needs to be processed by the Acoustic Feedback Canceler;

② Local Output: The signal processed by the Acoustic Feedback Canceler, output to the local output channel.

4.5.2 AEC

The Acoustic Echo Canceler (AEC) 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. 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.



Acoustic Echo Canceler: Set the signal that needs to be processed by the Acoustic Echo Canceler, and the processed signal selects the output channel in the Mixer;

① Local Input: local Mic output channel, i.e. the signal that needs to be processed by the Acoustic Echo Canceler;

② Remote Input: The echo remote input, i.e. the reference signal;

③ Local Output: The signal after Acoustic Echo Canceler processing is output to the local output or output to the remote end;

④ AEC level: Echo level range [small room (128ms), medium room (256ms), large room (512ms)];

⑤ ANS level: Noise reduction level range (6~30dB).

4.5.3 ANS

The Adaptive Noise Suppression (ANS) effectively removes non-vocal sounds. Distinguish between human voice and non-human voice, and treat non-human voice as noise. A piece of audio containing both vocals and noise is processed by this module, and theoretically, only the vocals remain.

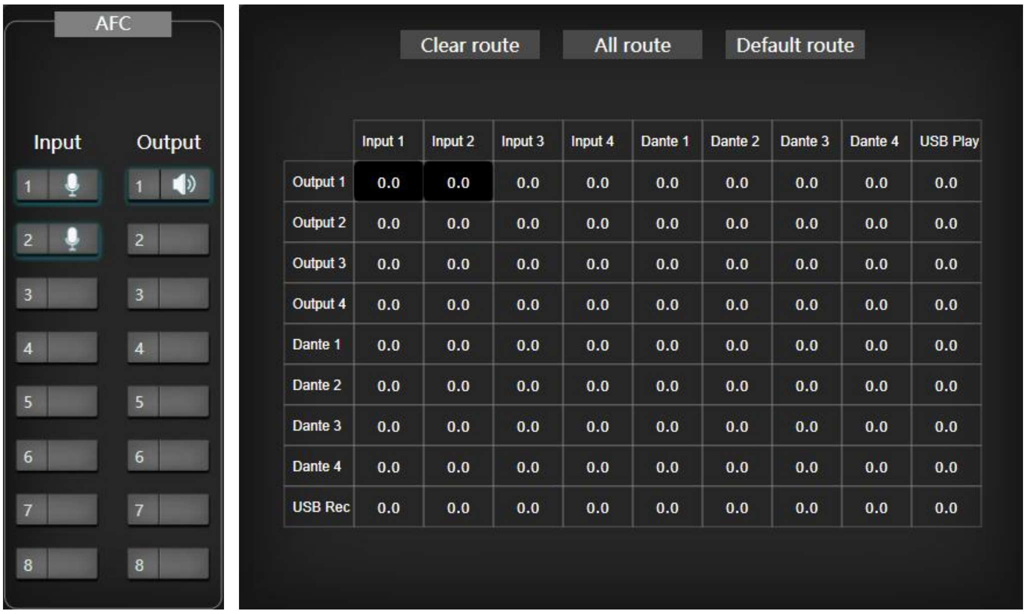
Adaptive Noise Suppression: Select the signal that needs Adaptive Noise Suppression processing, and the processed signal is output in the mixer by selecting the corresponding channel;

① Local Input: Local Mic output channel, i.e. the signal that needs Adaptive Noise Suppression processing;

② Local Output: The signal after Adaptive Noise Suppression processing is output to the local output channel;

③ ANS level: Range of noise reduction level (6~30dB).

Example 1 AFC with Matrix Mixer association operation



The signals from input channels IN1 and IN2 are processed in the AFC and output in output channel OUT1, configured as shown above:

① AFC window [Local Input] list selects the input channels IN1 and IN2, indicating that the signals of input channels IN1 and IN2 will be sent to the AFC for processing;

② The [Local Input] list selects the point corresponding to the output channel OUT1, indicating that the input signal is routed to the OUT1 channel for output after being processed by the AFC. After the AFC is enabled, the corresponding channel of the Matrix Mixer Window list is displayed in black.

Example 2 AEC with Matrix Mixer association operation

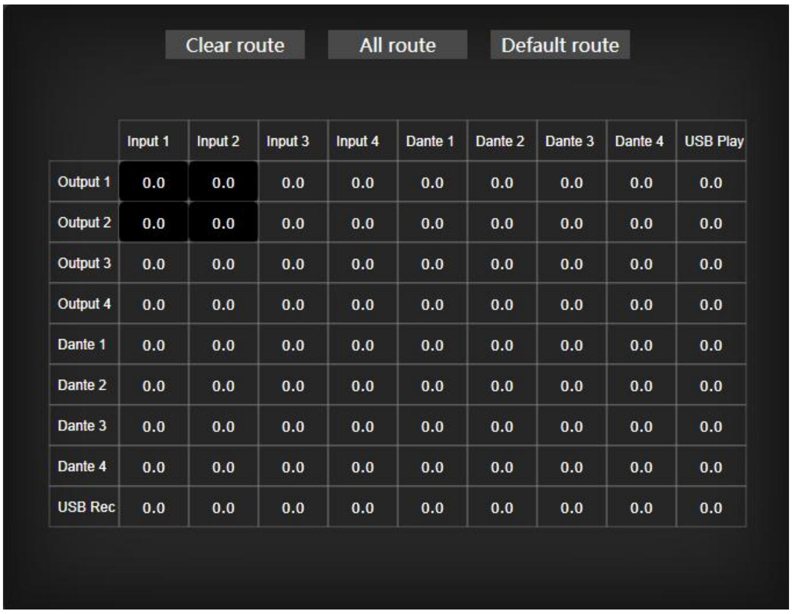
	Input 1	Input 2	Input 3	Input 4	Dante 1	Dante 2	Dante 3	Dante 4	USB Play
Output 1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
USB Rec	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

The local input signal is IN1 channel and the remote input signal is IN2 channel, at this time, the local input and the corresponding channel of the remote input will be grayed out to prevent the AEC from being activated abnormally due to the checking, and the local input signal will be AEC from OUT1 channel to the remote place, and the configuration will be as described in the above figure:

① AEC window to select the local input IN1 channel and then select the remote input IN2 channel, that is, the remote input IN1 channel and the remote input IN2 reference signal will be sent to the AEC for processing;

② The local input IN1 signal is then output to the remote end through the OUT1 channel, and the remote input IN2 reference signal is then routed through the matrix to the OUT2 channel for output to the local loudspeaker.

Example 3 ANS with Matrix Mixer association operation



The local input IN1 and IN2 channel signals are processed for ANS and output on OUT1 and OUT2 channels, configured as above:

① Input IN1 and IN2 channels are selected in the ANS window, indicating that the input signals of input IN1 and IN2 channels are sent to the ANS for processing;

② Check OUT1 and OUT2 in the local output list, which means that the result of the ANS processing will be sent to the output OUT1 and OUT2 channels for output. After enabling the ANS, the corresponding channels in the Matrix Mixer window list are shown in black.

4.6 Matrix Mixer

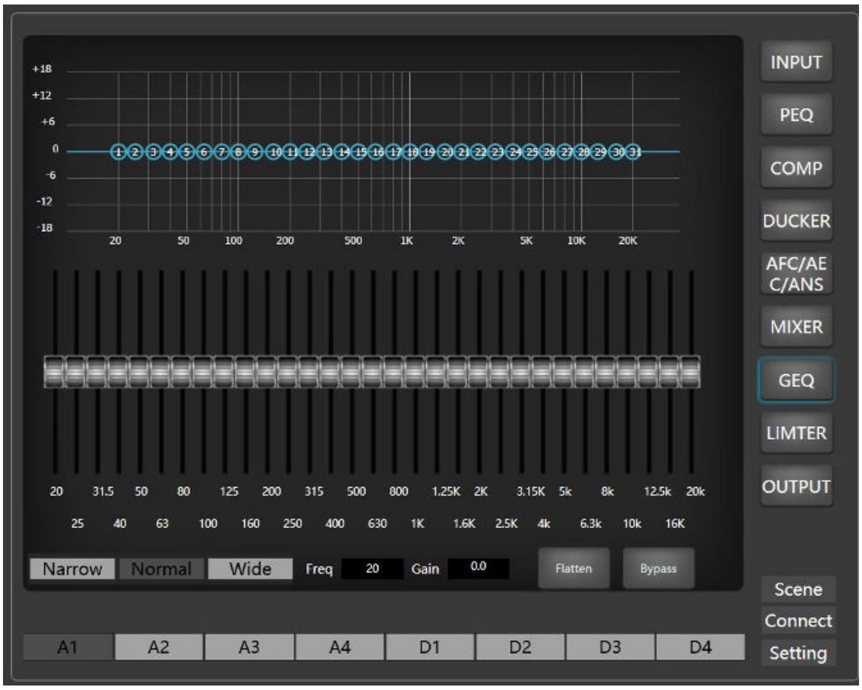
The Matrix Mixer receives an audio input and distributes the audio signal to each Output channels.

Matrix both signal routing and mixing double multiplexing function, X-axis is the input channel, Y-axis is the output channel, the matrix initialization state is (as shown in the figure above: one-to-one) input and output.

4.7 Post-processing

4.7.1 Graphic Equalizer

The Graphic Equalizer Component provides from 31 bands with 1/3th octave per band. Each band can be adjusted from -18dB to +18 dB.



① Narrow: Narrowband equalization filter; Normal: Normal equalization filter; Wide: Wideband equalization filter;

② Freq.: Frequency indication of the current equalization filter;

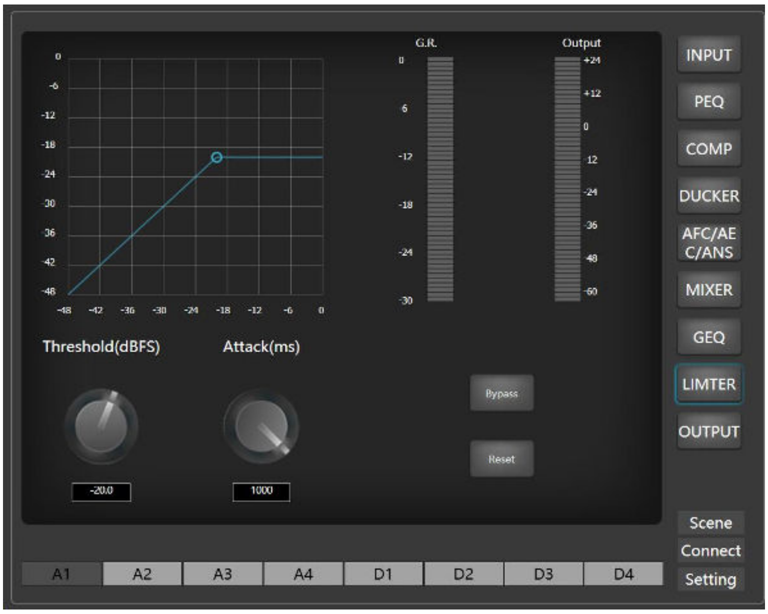
③ Gain: Gain indication or control of the current equalization filter;

④ Flatten: Rests all the band gains to the default;

⑤ Bypass/Active: Bypass or Active the Graphic Equalizer for the current channel. When the Graphic Equalizer is bypassed, audio is passed through without any change.

4.7.2 Limiter

The Limiter limits the output level to the Threshold Level, prevent signal overload and transient interference.When the input signal is above the threshold, the output signal is equal to the threshold; when the input signal is below the threshold, the output signal is equal to the input signal.



① Bypass/Active: Bypass or Active the Limiter for the current channel. When the Limiter is bypassed, audio is passed through without any change;

② Threshold (-48~0): Sets the level at which the Limiter has an effect, and the level at which the output is held;

③ Release Time (1~1000): When the input signal falls below the Threshold Level, the sound channel is not turned off immediately, but is delayed for the Release Time. During this time, the sound channel stays on as long as there is a signal above the Threshold Level.

④ G.R.: Graphically displays the amount of attenuation applied to the Channel;

⑤ Reset: Resets the parameters to the default.

4.7.3 Output

The Analog Output provides one channel 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 one three-terminal 3.5mm Euro style connectors.

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

① Indicator light: When there is no signal into the output channel, the indicator light is gray; when there is a small signal into the output channel, the indicator light is green; when there is a large signal into the output channel, the indicator light is red;

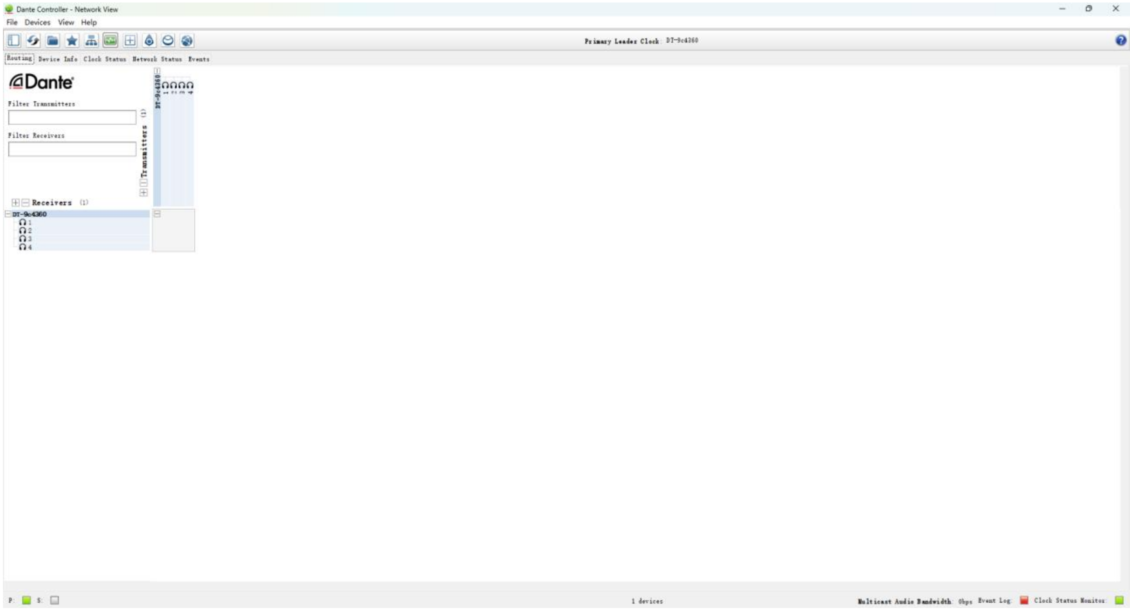
② Mute: Mutes the output signal;

③ Invert: Inverts the polarity of the output signal;

④ Channel fader: Channel gain value range (0~-72dBFS), can be controlled by fader.

Chapter 5 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 6FAQ

6.1 Abnormal power indicator (PWR)

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: Reboot the device.

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

6.2 Abnormal status indicator (SYS)

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.

6.3 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.

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

6.5 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:** If the LAN is automatically obtaining IP, please use 0.0.0.0 for the processor IP).

6.6 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.

6.7 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.

6.8 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.

6.9 RS485 center control command does not work

First, check whether the connection is normal, the "+" of the center control host connects to the "+" of the device, the "-" of the center control host connects to the "-" of the device. ", the central control host and device ground interconnection; second, check the software configuration of the device interface items: baud rate, start bit, stop bit and other settings with the central control host interface configuration is consistent. If the problem is not solved, please contact the manufacturer.

Chapter 7 Packing List

Device	DC12V/2A Power Adapter	Quick Guide	6pin Phoenix Connector	3pin Phoenix Connector	Small Screwdriver
1PCS	1PCS	1PCS	4PCS	1PCS	1PCS

Warranty Regulations

The warranty period for this product is 1 year.

During the warranty period,non-man-made damage or performance failures can enjoy a Three Guarantees Service(Repair, Replacement, and Refund Service) .

The warranty card is effective only after being stamped by the sales unit. Alterations will invalidate it!

The following conditions (including, but not limited to) are not covered by the Three Guarantees Service :

1. No warranty card or valid invoice missing, or if the date has exceeded the validity period of the three guarantees services.
2. Damage caused by not following the product instructions for use, maintenance, storage, and management.
3. Mismatch between the product model or code on the warranty voucher and the physical goods.
4. Damage from dismantling or repair by unauthorized service providers.
5. Normal discoloration, wear and tear, and consumption during use of the product are not covered.
6. If the product cannot be used due to the user's network issues, please consult customer service staff.

SHENZHEN S TRACK SCIENCE TECHNOLOGY CO., LTD

Web: www.s-track.com.cn Tel:+86 755 29983191 Mail:service@s-track.cn

Add: 9F, 1B, Shangzhi Technology Park, Guangming District, Shenzhen City, Guangdong Province, China 518107