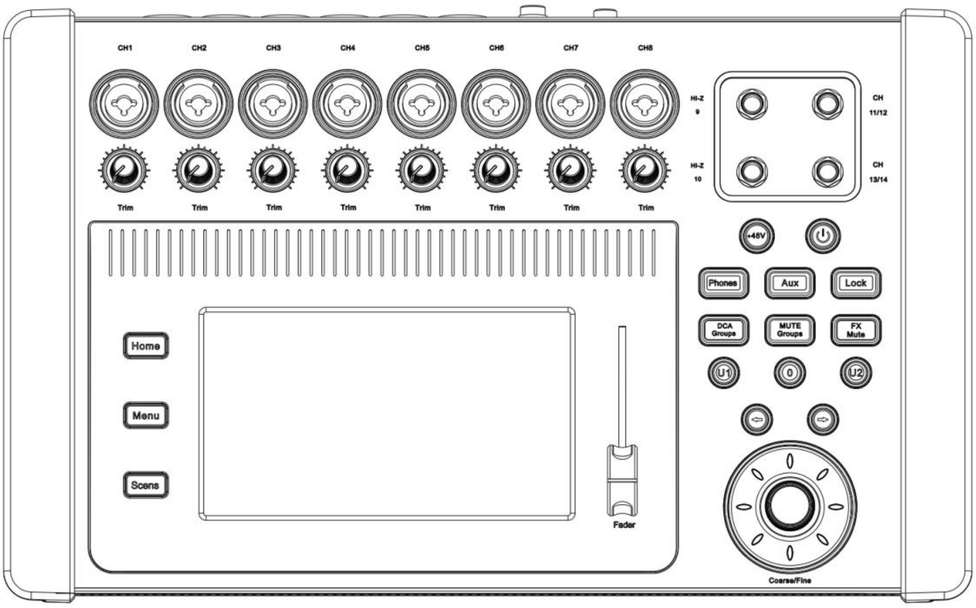

HIPPO D1608
Digital Mixer

User Manual

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 Mixer.

This manual describes the function and use of the various function modules of the Digital Mixer, and guides you through the installation and commissioning of the Digital Mixer.

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, the electrical safety regulations of the country and the region of use must be strictly observed.
- When installing the equipment, make sure that the ground wire in the power cord is well grounded and the chassis grounding point is well grounded, do not use a two-pronged plug. Make sure that the input power supply of the equipment is 100V-240V 50/60Hz AC.
- Please use the power adapter provided by the regular manufacturer, please refer to the product parameter table for the specific requirements of the power adapter.
- Do not connect multiple devices to the same power adapter (exceeding the adapter load may generate excessive heat or cause a fire).

- Keep the working environment well ventilated so that the heat generated by the equipment at work can be discharged in time to avoid damage to the equipment due to high temperature.
- Always unplug the unit's AC power cord from the AC power outlet before: A. Removing or reinstalling any part of the unit; B. Disconnecting or reconnecting any electrical plug or connection to the unit. Do not operate with electricity.
- There are AC high-voltage parts in the equipment, non-professionals should not disassemble it without permission to avoid the danger of electric shock. Do not repair the equipment privately to avoid aggravating the degree of damage.
- Do not spill any corrosive chemicals or liquids on or near the unit.
- If the unit emits smoke, odors, or noises, immediately turn off the power and unplug the power cord, and contact your dealer or service center.
- If the unit is not working properly, contact the dealer or service center from which you purchased the unit and do not disassemble or modify the unit in any way. (We are not responsible for problems caused by unauthorized 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).
- Please do not use the device in high temperature, low temperature or high humidity environment, the specific temperature and humidity requirements refer to the device's parameter table.
- The unit needs to be used indoors and should not be installed in an exposed location where it may get wet or very humid.
- Turn off the main power supply of the equipment in a humid dewy environment or when it is not used for a long time.
- 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, the equipment can be packed using the packaging materials and sent to the agent or returned to the manufacturer for processing. We will not be responsible for any accidental damage during transportation that is 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

This Digital Mixer effectively integrates digital mixing systems with innovative design and powerful DSP functions. It adopts a new concept that combines modern digital technology with traditional operation, offering users a highly professional functional experience. Its compact size, simple operation interface, and professional mixing effects deliver outstanding performance in professional shows, while also providing powerful features that fully satisfy less experienced individual users.

The mixer boasts strong processing power and advanced features. The software operation process is designed for convenience and speed, allowing you to quickly adjust the mixing interface. This convenient and fast operation ensures that everyone can enjoy the ease of use and powerful capabilities of the Digital Mixer.

1.2 Product Features

- 16 inputs (8 balanced XLR/TRS digital gain channels, 2 sets (4 channels) of stereo inputs, 2 high-resistance mono inputs, 2 USB playback channels);
- Extremely low distortion and ultra-low noise floor, versatile parameter adjustability, and good consistency due to digital gain to effectively prevent misuse;
- 7-inch high-definition touch screen, friendly software interface, clarity navigation design;
- The operating panel consisting of a digital encoder and dedicated keys enables all settings to be made quickly and easily;
- 5 built-in effects for singing and performing, the built-in effects can simplify the system wiring; the device comes with classic reverb, chorus, modulation and other effect modules; FX effects can be used to return to the mix using a dedicated return channel and does not occupy the mono and stereo input channels;
- Scene storage is different from the analog mixer is one of the most practical and significant features, can store 100 complete scenes, all the scenes can be exported to an external storage device for storage backup, in order to later call at any time.

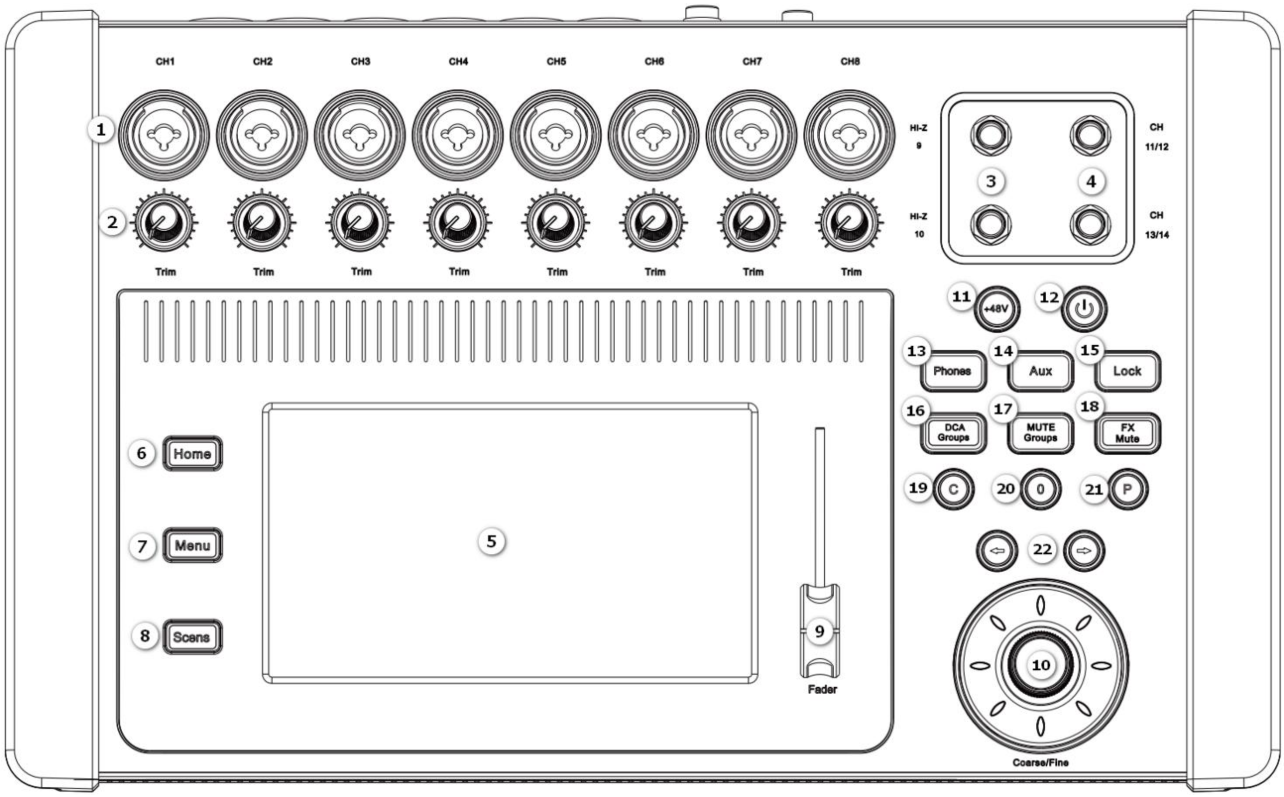
1.3 Functions

- 16 input channels, low noise floor, versatile parameter adjustability;
- The use of high-end audio AD/DA chip makes it have excellent audio indicators, the maximum input and output levels reach 20dBu;

- ✧ Built-in USB recording and playback function, support APE, FLAC, MP3, WAV and other audio formats; built-in 4G storage space, you can upload audio files into the machine to play;
- ✧ 8-channel DCA grouping, 8-channel mute grouping, inputs and outputs, effects channels can be programmed;
- ✧ Each input channel has a 4-band parametric equalizer, compressor, noise gate, phase, and delay;
- ✧ Each output channel has a 6-band parametric equalizer, 31-band graphic equalizer, high and low pass filters, voltage limiter, and time delay;
- ✧ Built-in adaptive trap feedback suppression algorithm;
- ✧ 5 effect types: mono delay, stereo delay, chorus, reverb, pitch shift; 4 effect channels;
- ✧ Dual-machine hot backup synchronizes data in real time over the network;
- ✧ Unique LOCK key to hold live scene data or prevent misuse;
- ✧ 7-inch 1024x600 resolution high-definition capacitive touch screen, full-function operation control mixer;
- ✧ Supports 30-255 groups of scene presets, which can be imported into USB storage for easy backup recall;
- ✧ Built-in: Sine wave, pink noise, white noise signal generator;
- ✧ Unique Link connection function for adjacent channel binding settings;
- ✧ Channel name customization;
- ✧ Cross-platform development platform, support for Windows, Linux, macOS, Android, iOS mainstream operating system full-featured control software.

Chapter 2 Interface/Keypad Description

2.1 Front Panel



① CH1-CH8: Balanced XLR/TRS combination mono input connectors;

② Trim knob: Channels 1-8 to adjust analog input signal level before analog to digital conversion;

③ HI-Z 9-10: High resistance mono channels for instrument input connectors;

④ CH11-CH14: TRS stereo input connectors;

⑤ 7-inch high-definition LCD touch screen;

⑥ Home Button: Return to the main interface;

⑦ Menu Button: Enter the display control and system setting interface;

⑧ Scene Button: Enter the scene preset interface;

⑨ Fader: Volume attenuation fader;

⑩ Encoder Data Wheel: Press to change the selected value or position, Rotate left and right can be fine-tuned parameters, and can scroll to view the list;

⑪ +48V Button: Open the phantom power supply interface, view the status of all channels with the phantom power supply on/off;

⑫ Close screen button;

⑬ Phones Button: Monitor level volume control;

⑭ Aux Button: Navigates to the Aux overview screen;

⑮ Lock Button: Operating system interface lock to prevent misuse (default password: 123456);

⑯ DCA Groups Button: Navigates to the interface where DCA groups can be controlled and edited;

⑰ Mute Groups Button: Navigates to the interface where mute groupings can be controlled and edited;

⑱ FX Mute Button: Mute or unmute all effect channels;

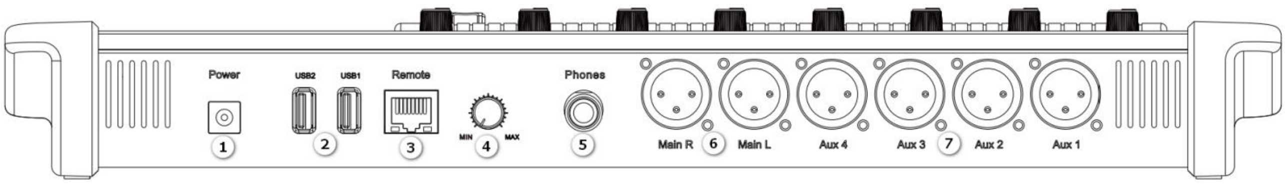
⑲ C Button: Copy button to copy the current channel configuration parameters to other channels, limited to the same type of channel;

⑳ 0 Button: Resets the current output channel gain value to the default (0 dB);

㉑ P Button: Paste button, can paste the parameters of the copied channel to the current channel, limited to the same type of channel;

㉒ ⇐ ⇒ Button: Navigate to the left or right.

2.2 Rear Panel



① Power: Please use the power supply that came with the mixer, do not use other power supplies instead;

② USB2.0 (Class A): for connecting USB storage devices and Wi-Fi adapters;

③ Remote: Ethernet interface for connecting to PC-based interactive software;

④ Trim knob: for adjusting the signal level of the monitoring interface;

⑤ Monitor: Stereo TRS interface when the monitor channel is activated the line or headphone output is transferred to this output channel;

⑥ Main R/L: Balanced XLR male connectors;

⑦ Aux: Auxiliary outputs 1 to 4 channels, balanced XLR male connector.

Chapter 3 Instructions for Use

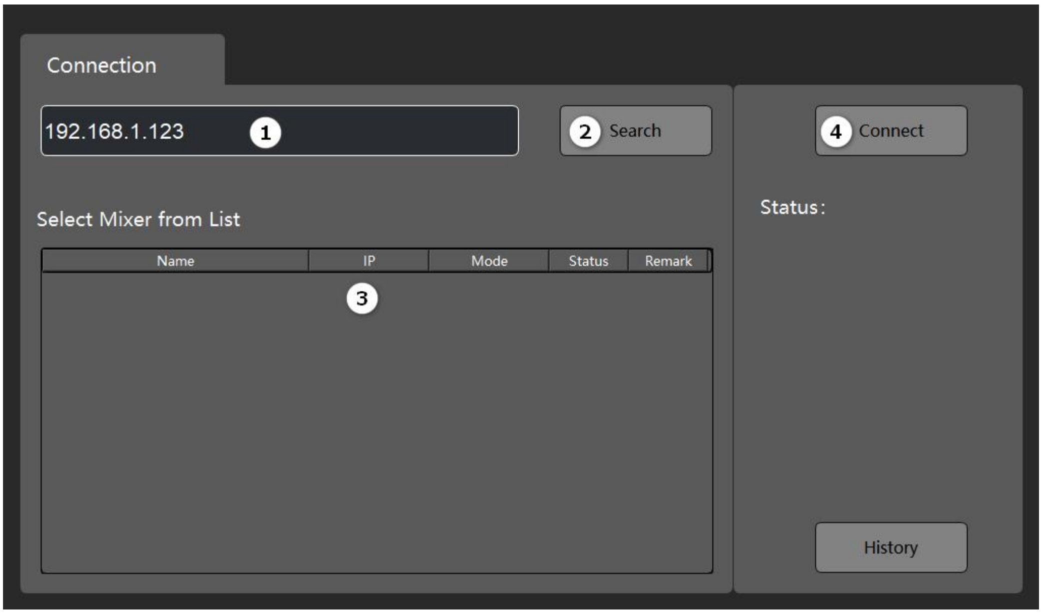
3.1 Software and Documents Download

① View device IP address: Enter [Menu] ⟹ [Network] to view the device IP address information, the default is automatically assigned by DHCP, you can choose to manually set the IP address;

② Download: Open the browser, enter the IP address of the device, click "Enter" to navigate to the download interface, which provides three download options: audio files, quick guide / user manual, interactive software downloads (support systems Windows, Android, macOS, Linux);

③ Default password: (LOCK, scene reset password) 123456.

3.2 PC Software Login Connection



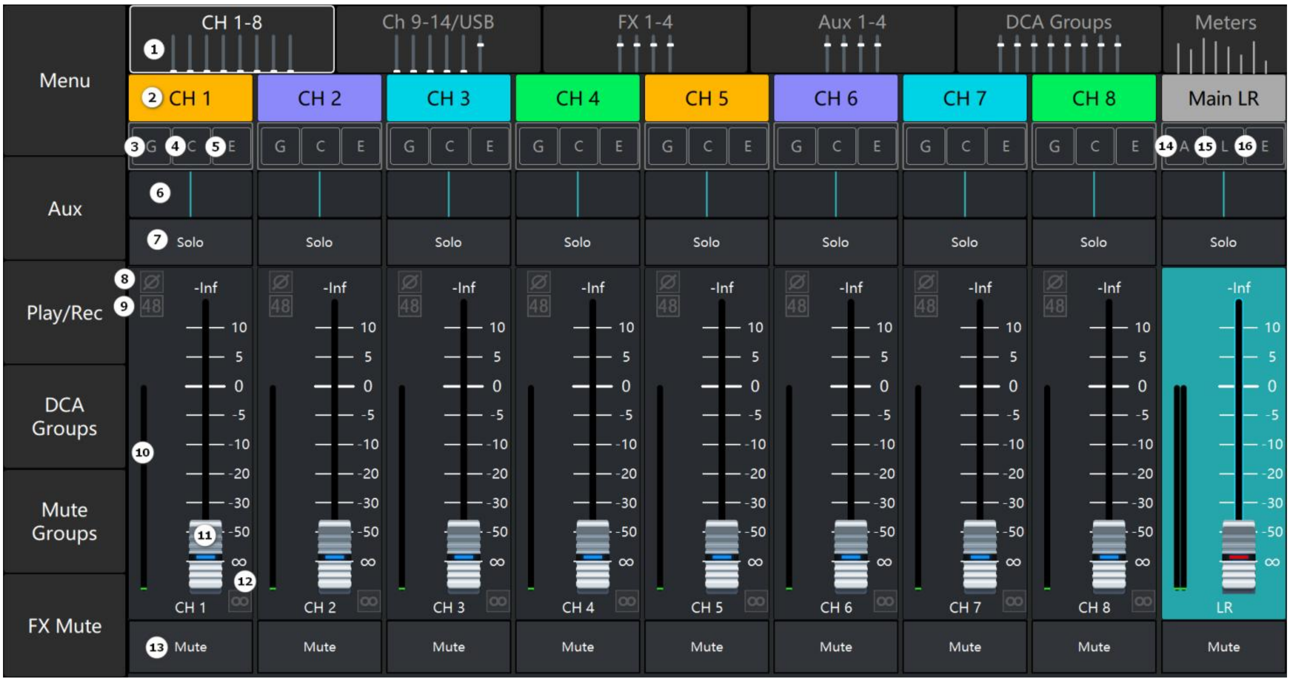
① IP input box: IP address can be entered and displayed;

② Search: Search for devices, devices in the same LAN can be found;

③ Device List: Displays online device name, IP address and other information;

④ Connect: Select the device in the list, click "Connect" to connect, and automatically jump to the main interface.

3.3 Main Interface



① Navigation Bar: Displays the type of channel and the channel range;

② Channel Name: Displays the channel name (customizable name and color), touch to navigate to the channel configuration interface;

③ G: Displays whether the channel Noise Gate is enabled or disabled;

④ C: Displays whether the channel Compressor is enabled or disabled;

⑤ E: Displays whether the channel Parametric Equalizer is enabled or disabled;

⑥ Panning: Adjust the Panning of the sound source distribution in space, touch and slide or use the data wheel to adjust;

⑦ Solo: Route the signal of this channel to the monitor interface;

⑧ Ø: Displays the input signal polarity has been changed for this channel;

⑨ 48: Displays whether the channel Phantom power is enabled or disabled (red indicates enabled);

⑩ Level Meter: Displays the real-time signal level of the current channel;

⑪ Channel Fader: Touch the fader to adjust the current channel gain;

⑫ Link: Links the channel to an adjacent channel and the channel settings will all be copied to the adjacent channel;

⑬ Mute: Mutes the channel(red indicator), and the orange color indicates that the channel is turned on for system mute or Mute Group or DCA Group mute;

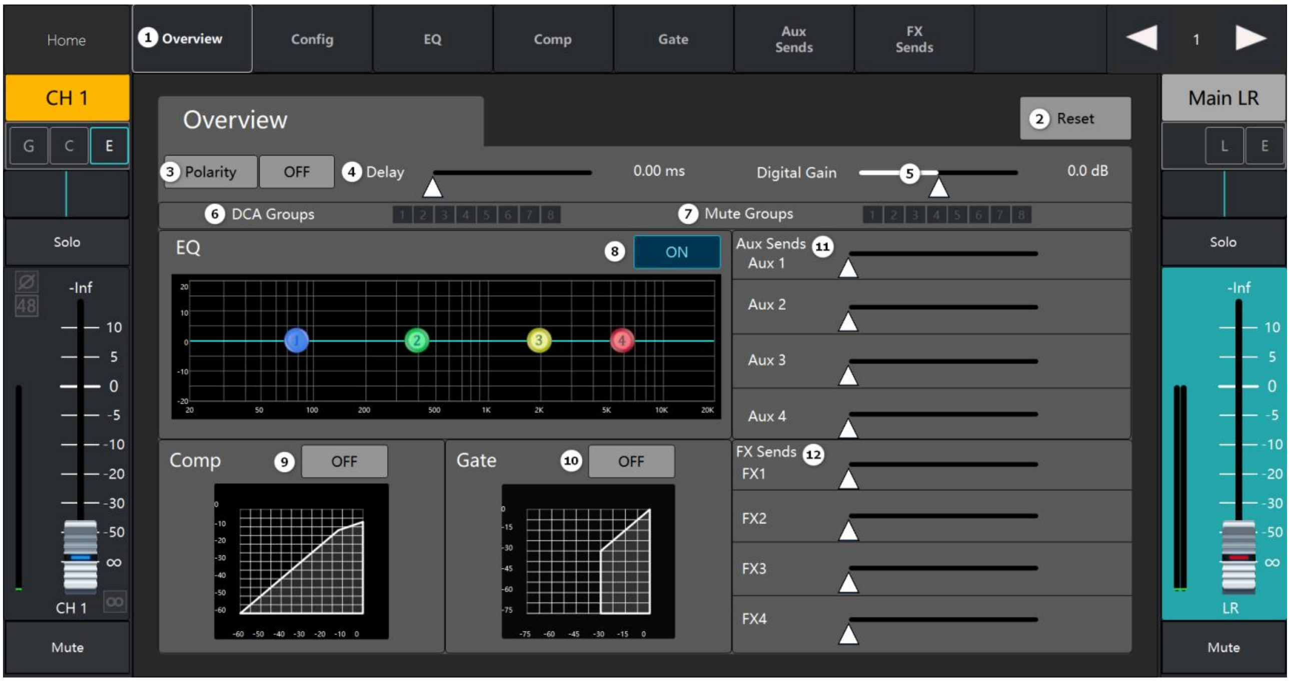
⑭ A: Displays whether the channel Anti-Feedback is enabled or disabled;

⑮ L/C: Displays whether the channel Limiter/Compressor is enabled or disabled;

⑯ E: Displays whether the channel Parametric Equalizer is enabled or disabled.

3.4 Input Channel

3.4.1 Overview



① Overview: Overview of the current channel's controls;

② Reset: Rests current interface parameter configuration to default;

③ Polarity: Change the polarity of the current input signal;

④ Delay On/Off Button: Displays Delay setting and Delay information;

⑤ Digital Gain: Control the channel digital gain (-3～+3dB) by slider;

⑥ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑦ Mute Groups: Displays the channel has been assigned to the Mute Groups;

⑧ Parametric Equalizer: Enable or disable the Parametric Equalizer and display the Equalizer curve graph. Click on the curve graph to jump to the Equalizer configuration interface;

⑨ Compressor: Enable or disable the Compressor and display the Compressor curve graph. Click on the curve graph to jump to the Compressor configuration interface;

⑩ Noise Gate: Enable or disable the Noise Gate and display the Noise Gate curve graph. Click on the curve graph to jump to the Noise Gate configuration interface;

⑪ Aux Sends: Send the current channel signal to the Aux Auxiliary output channel;

⑫ FX Sends: Send the current channel signal to the FX effects channel.

3.4.2 Configuration

① Configuration: Parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Channel Name: Displays the channel name, touch and then display the keyboard to customize the channel name;

④ Link: Links the channel to an adjacent channel and the channel settings will all be copied to the adjacent channel;

⑤ Ch Marker: Channel color marking, customizable input channel display colors;

⑥ Polarity: Change the current channel input signal polarity;

⑦ 48V: Enable or disable 48V phantom power for this channel;

⑧ Delay: Enable or disable the current channel Delay by lightly touching the slider or using the data wheel to adjust the delay range (0～1000 ms). This provides an alternative method for setting delay times in units of distance, ranging from 0 meters to 340 meters. Entering delay times in distance units is often more convenient in practical scenarios. If needed, distances can also be entered in feet;

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.

⑨ Digital Gain: Control the channel digital gain (-3～+3dB) by slider;

⑩ Main L/R: Route the channel signal to the Main channel output;

⑪ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑫ Mute Groups: Displays the channel has been assigned to the Mute Groups.

3.4.3 Parametric Equalizer

The Parametric Equalizer component is a tool used to precisely adjust audio frequency response. It is a variable equalizer that fine-tunes specific frequency ranges by adjusting the gain, bandwidth, and center frequency of each frequency band, balancing frequency components, and resolving audio issues. By flexibly configuring the parameters of each frequency band, users can achieve frequency adjustments ranging from simple to complex, meeting the needs of different scenarios such as music production, live sound reinforcement, and voice processing, and achieving ideal audio effects.

① Equalizer parameter configuration interface;

② On/Off Button: Enables or disables the Equalizer;

③ Real Time Analyzer: Enables or disables the real time analyzer to display the channel signal amplitude and peak;

④ Reset: Rests current interface parameter configuration to default;

⑤ Parametric Equalizer Display: Displays the equalizer curve graphically based on the equalizer parameter settings;

⑥ Low Cut Filter On/Off Button: The filter uses the frequency set by the frequency controller as the cutoff frequency, attenuating frequencies below the cutoff frequency. This is typically used to remove low-frequency interference or extract high-frequency characteristics;

⑦ Low Cut Frequency: Sets the cutoff frequency for the low-cut filter;

⑧ Band 2～4 On/Off Button: Enables or disables the relevant parametric equalizer bands. The bands are fully parametric, with a frequency range of 20 Hz～20 kHz;

⑨ Gain: Adjusts the gain within the frequency settings of the relevant equalizer bands, with a range of -15～+15 dB;

⑩ Frequency: Sets the center frequency of the parametric equalizer band. If the shelving filter is enabled, control the cutoff frequency for the shelving filter;

⑪ Bandwidth (Q-Factor): The Q-factor refers to the range of frequencies affected around the center frequency. Set the Q-factor for a single band in the equalizer, ranging from 0.4 octaves to 4 octaves (default value is 1.00). This setting is disabled when selecting Low Shelf and High Shelf. The Q-factor determines the precision and scope of equalization adjustments. Higher Q-factor yield narrower bandwidth and more precise frequency range control; lower Q-factor provide wider bandwidth and broader frequency influence;;

⑫ Low Shelf: Enables or disables the low shelf filter, converting equalizer band "2" from a parametric filter to a low shelf filter, increasing or attenuating the gain of the low-frequency portion below the set center frequency, commonly used to increase the richness of low frequencies or reduce low-frequency rumble;

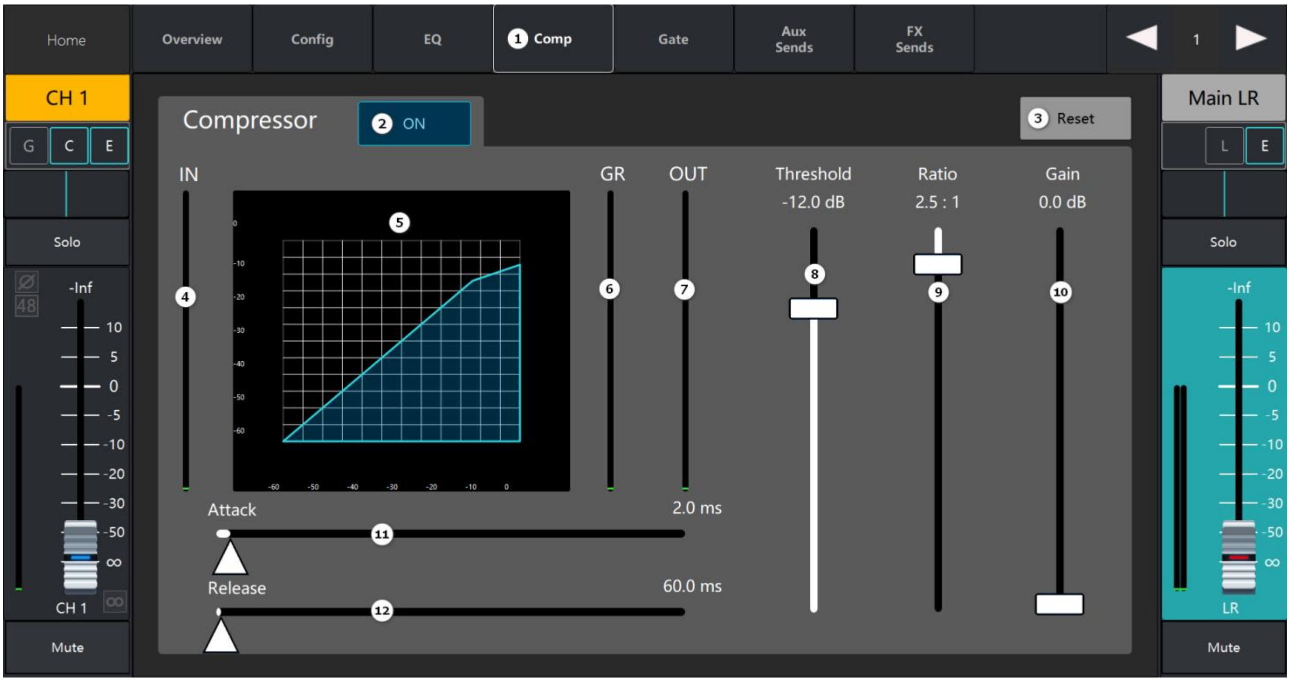
⑬ High Shelf: Enables or disables the high-shelf filter, and change equalizer band "4" from a parametric filter to a high-shelf filter, which increases or attenuates the gain of the high-frequency portion above the set center frequency. This is typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies;

⑭ High Cut Filter On/Off Button: The filter uses the frequency set by the frequency controller as the cutoff frequency, attenuating frequencies above the cutoff frequency. This is typically used to remove high-frequency noise or increase low-frequency components;

⑮ High Cut Frequency: Sets the cutoff frequency for the high cut filter.

3.4.4 Compressor

The purpose of the Compressor component is to control the dynamic range of the Output above a set Threshold Level, thereby optimizing audio balance and consistency. Compressor are widely used in music production, live sound reinforcement, broadcasting, and voice processing, helping users control audio signal peaks, prevent distortion, and enhance overall audio clarity and audibility by raising the average signal level



① Compressor parameter configuration interface;

② On/Off Button: Enables or disables the Compressor;

③ Reset: Rests current interface parameter configuration to default;

④ Input: Displays the input level;

⑤ Compressor Graph: Compressor graph with level scale from -60dB～0dB;

⑥ Gain Attenuation: Displays the signal level attenuated by the Compressor;

⑦ Output: Displays the output level after Compressor processing;

⑧ Threshold: Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. 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.

⑨ Ratio: 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;

⑩ Gain: Controls the Gain of the output, used to compensate for the reduction in signal level caused by compression processing. When an audio signal is compressed, its overall volume decreases. The function of Output Gain is to restore the compressed signal to a volume level close to that before compression by raising the output signal level;

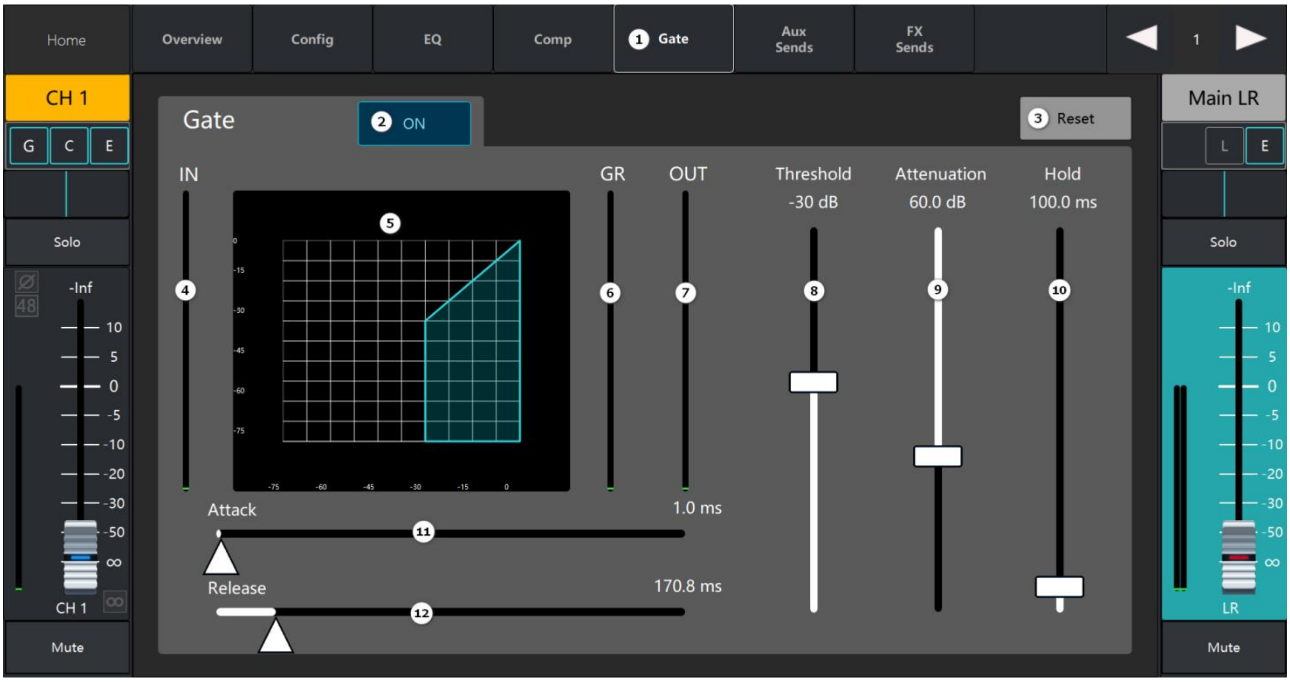
⑪ Attack Time: Attack time is how fast the compressor reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

⑫ Release Time: Release time is how fast the compressor reacts when the signal drops below the threshold and gain is restored to its non-limited level. Fast release time can increase signal loudness, but is prone to suction effects; Slow release time provides a smooth transition and reduces suction effects, but may sound sluggish. Settings should be balanced according to audio characteristics.

3.4.5 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 Attenuation 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.



① Noise Gate parameter configuration interface;

② On/Off Button: Enables or disables the Noise Gate;

③ Reset: Rests current interface parameter configuration to default;

④ Input: Displays the input level;

⑤ Noise Gate Graph: Noise Gate graph with level scale from -75dB～0dB;

⑥ Gain Attenuation: Displays the signal level attenuated by the Noise Gate;

⑦ Output: Displays the output level after Noise Gate processing;

⑧ Threshold: Sets the point from which the attenuation is calculated. This is where the Noise Gate starts working. When the input signal falls below the Threshold Level, the Noise Gate attenuates the signal according to the set attenuation amount;

⑨ Attenuation: Sets the amount of output signal attenuation when the input signal drops below the Threshold Level;

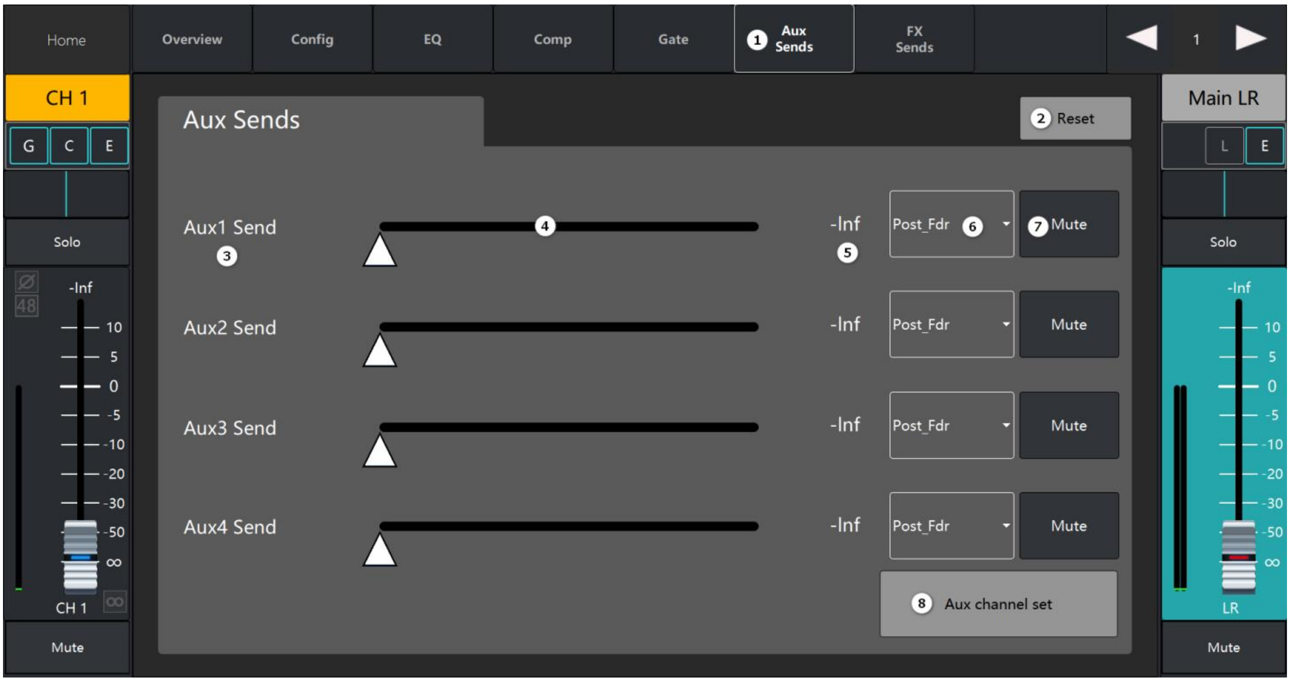
⑩ Hold Time: The Hold Time determines the minimum time the Noise Gate stays open once it is opened, or the length of time the Noise Gate stays open after the RMS input level drops below the Threshold Level. This is to prevent the gate from opening and closing due to momentary pauses in the input;

⑪ Attack Time: Attack time is how fast the Noise Gate to activate when the input signal falls below the Noise Gate Threshold Level. Shorter attack time enables the Noise Gate to respond quickly to signal changes, making it suitable for processing rapidly changing audio signals; Longer attack time provides a smoother transition, preventing abrupt processing effects and making it ideal for processing gradual signals such as vocals or music;

⑫ Attack time is how fast the Compressor/Limiter reacts to a signal crossing the set threshold going up

⑬ Release Time: Release time is how fast the Noise Gate ceases attenuating the audio signal when the input signal exceeds the Noise Gate Threshold Level. Shorter release time enables rapid signal dynamics recovery, suitable for fast-changing audio but prone to causing suck effects (i.e., rapid fluctuations in signal level); Longer release time provides smoother transitions and reduces suck effects, but may make the signal recovery process appear sluggish.

3.4.6 Aux Sends



① Aux Sends: Auxiliary sends parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Aux output channel name;

④ Aux Send Slider: Sets the audio signal level sent from this channel to the Aux output channel;

⑤ Displays the gain of the current sending channel;

⑥ Touch the drop-down box to select to send Pre-Fader/Post-Fader/Pre-Dynamics/Pre-All signals to the Aux output channel;

⑦ Mute: Mute the Aux Sends channel without affecting any other Aux outputs or sends;

⑧ Aux channel setting: Touch the button to jump to the Aux sends configuration interface.

I. Pre-All

Pre-All refers to the signal being extracted at the very beginning of the channel processing chain—before any processing modules (such as equalizers or dynamics processors), and even before the concept of channel faders. It typically follows the input gain stage (preamp).

Signal Path: Input Signal → Input Gain → [Pre-All Tap Point] → Equalizer → Dynamic Processor → Channel Fader → Output.

Pre-All is controlled solely by input gain and remains unaffected by channel EQ, dynamics processing, mutes, or fader movements. It is commonly used in multitrack recording to capture the raw, unadulterated signal.

II. Pre-Dynamics

Pre-Dynamics refers to the signal being captured before it passes through dynamics processors (compressors, noise gates) but after the equalizer.

Signal Path: Input Signal → Input Gain → Equalizer → [Pre-Dynamics Tap Point] →Dynamic Processor→Channel Fader→Output.

Pre-Dynamics is affected by input gain and EQ settings, but not by dynamic processing, mute, or fader actions. Commonly used in multitrack recording to capture signals that have undergone tonal shaping (EQ) but not dynamic control.

III. Pre-Fader

Pre-Fader is the signal extracted before the channel fader, but after the equalizer and dynamic processors.

Signal path: Input Signal → Input Gain → Equalizer → Dynamic Processor → 【Pre-Fader Tap Point】 → Channel Fader → Output.

Pre-Fader is affected by input gain, EQ, and dynamics processor settings, but is unaffected by the channel fader and mute. It is commonly used for stage monitors, where the signal has undergone all tonal and dynamic processing but its level is not controlled by the master fader.

IV. Post-Fader

Post-Fader is where the signal is tapped after passing through the channel fader. This is the final tap point in the processing chain.

Signal path: Input Signal → Input Gain → Equalizer → Dynamic Processor → Channel Fader → [Post-Fader Tap Point] → Output.

Post-Fader is fully controlled by all processing modules on the channel (input gain, EQ, dynamics) as well as the channel fader and mute. Commonly used for effects mixing, where both the main signal and effect signal levels change with the fader movement.

3.4.7 FX Sends

① FX Sends: Effects sends parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ FX channel name;

④ FX Send Slider: Sets the audio signal level sent from this channel to the Effects;

⑤ Displays the gain of the current send channel;

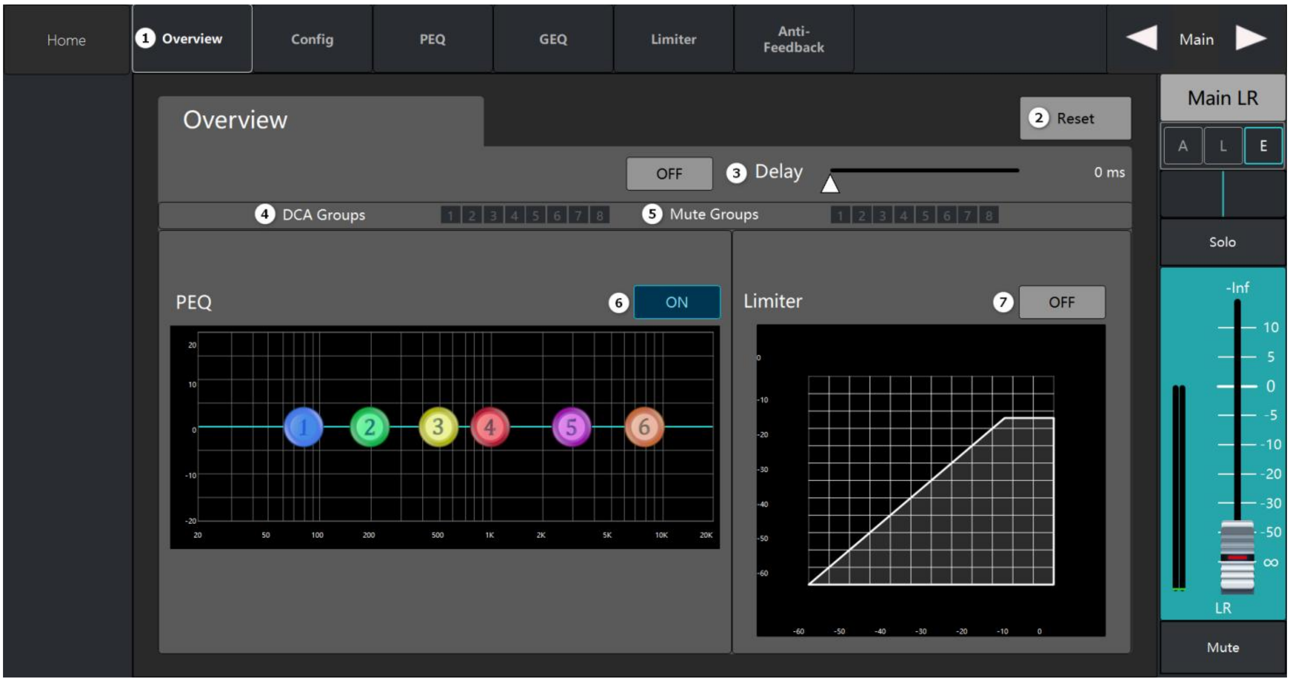
⑥ Touch the drop-down box to select the effect type: Mono Delay, Stereo Delay, Chorus, Reverb, Pitch Shift;

⑦ Touch to change the send Pre-Fader/Post-Fader signal to the FX channel;

⑧ FX Channel Setting: Touch the button to jump to the FX sends configuration interface.

3.5 Main L R Output channel

3.5.1 Overview



① Overview parameter interface;

② Reset: Rests current interface parameter configuration to default;

③ Delay On/Off Button: Displays Delay settings and Delay information;

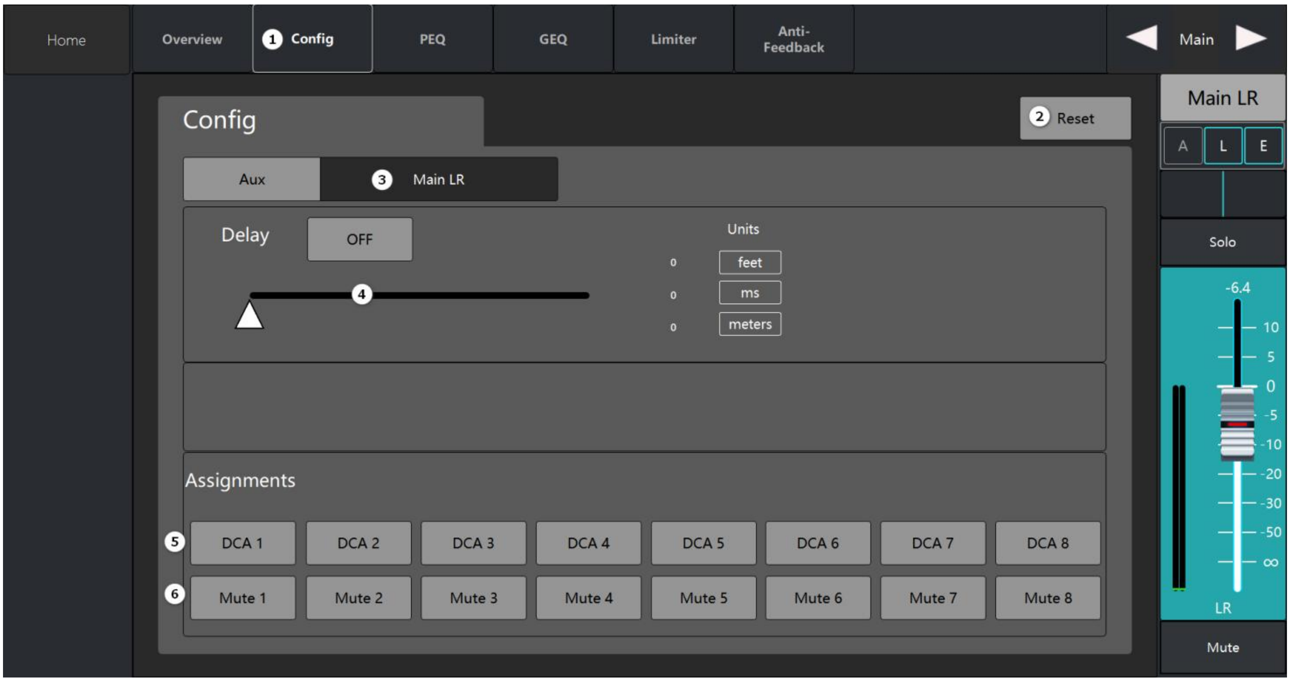
④ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑤ Mute Groups: Displays the channel has been assigned to the Mute Groups;

⑥ Parametric Equalizer: Enable or disable the Parametric Equalizer and display the Equalizer curve graph. Click on the curve graph to jump to the Equalizer configuration interface;

⑦ Compressor/Limiter: Enable or disable the Compressor/Limiter and display the Compressor/Limiter curve graph. Click on the curve graph to jump to the Compressor/Limiter configuration interface.

3.5.2 Configuration



① Configuration: Parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Channel Name: Displays the channel name, touch the display keyboard to customize the channel name;

④ Delay: Enable or disable the current channel Delay by lightly touching the slider or using the data wheel to adjust the delay range (0～1000 ms);

⑤ DCA Groups: Displays the channel has been assigned to the DCA Groups;

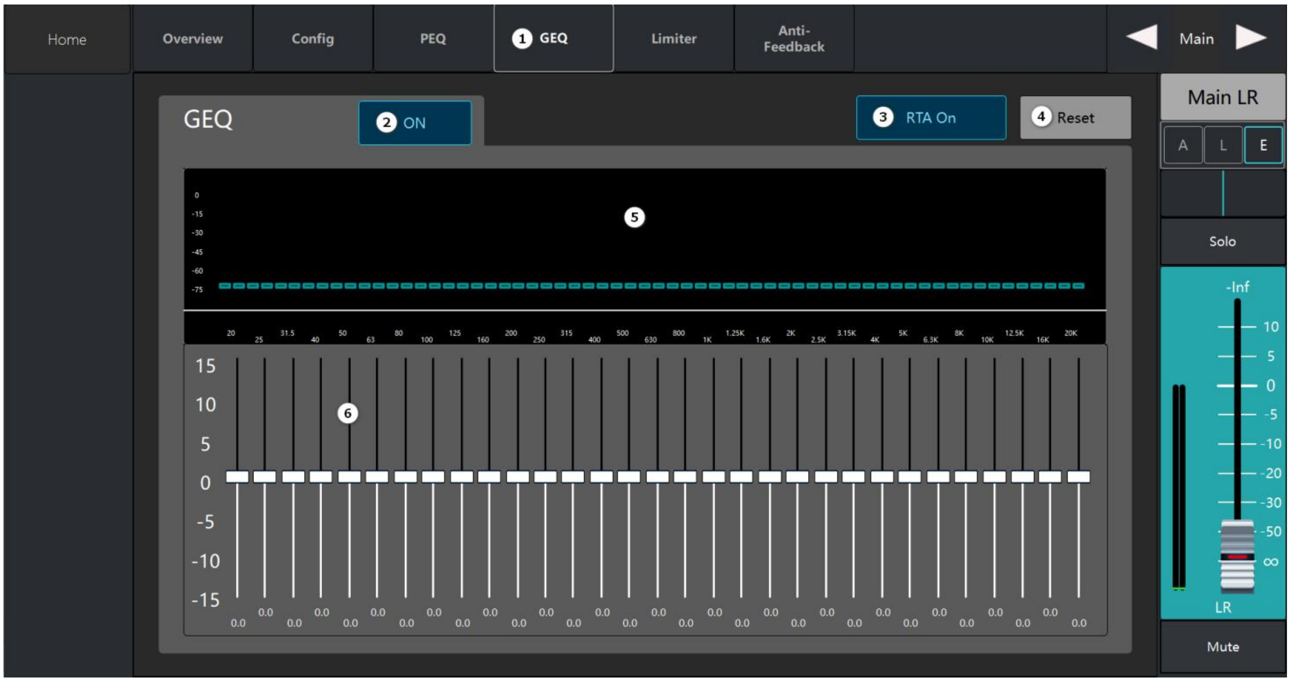
⑥ Mute Groups: Displays the channel has been assigned to the Mute Groups.

3.5.3 Parameter Equalizer

Reference Input Channel — Parametric Equalizer.

3.5.4 Graphic Equalizer

The Graphic Equalizer is a common audio processing tool widely used in music production, live sound mixing, home theaters, and professional audio systems. It fine-tunes audio signals using 31 fixed-frequency filters, each band 1/3 octave, with the gain or attenuation of each frequency band controllable via intuitive sliders. The design of this equalizer draws inspiration from analog-era mixing consoles, with its graphical interface enabling users to quickly and intuitively adjust the audio spectrum.



① Graphic Equalizer parameter configuration interface;

② On/Off Button: Enables or disables the Equalizer;

③ Real Time Analyzer On/Off Button: Enables or disables the real time analyzer;

④ Reset: Rests current interface parameter configuration to default;

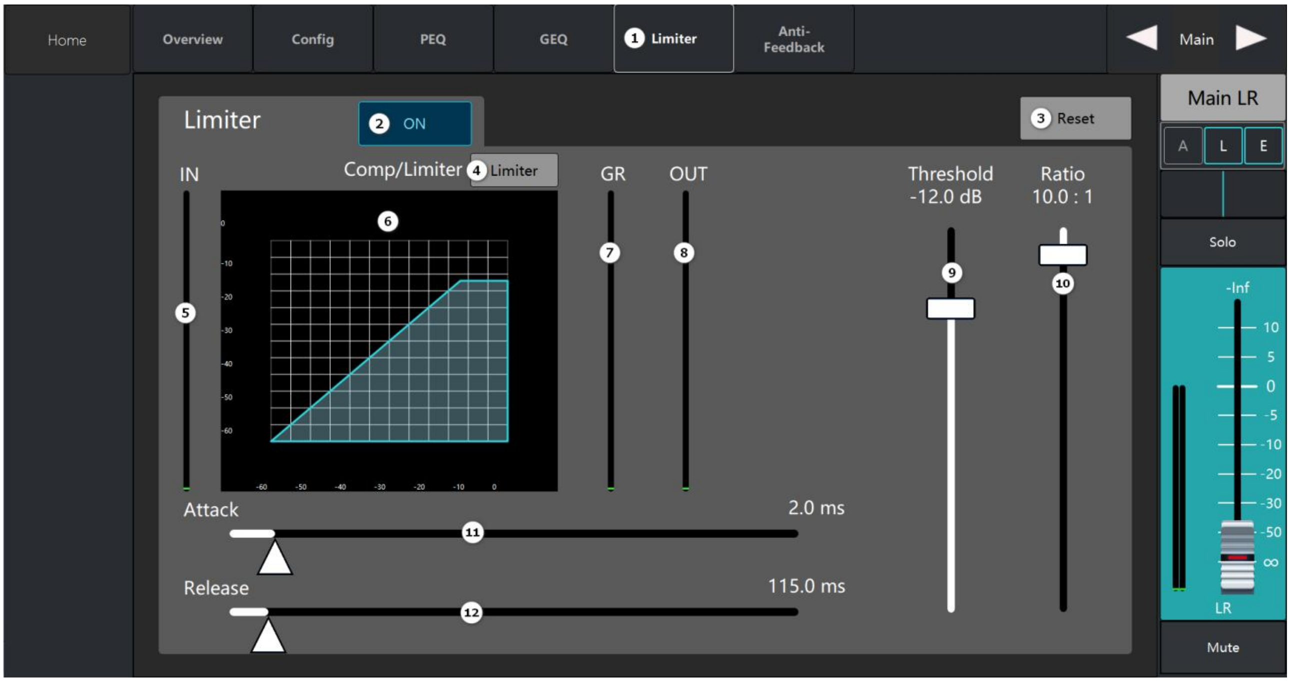
⑤ Real Time Analyzer: Enables or disables the real time analyzer to display the channel signal amplitude and peak;

⑥ Graphic Equalizer: 31-band Graphic Equalizer control. Select the corresponding frequency band and adjust the gain by moving the slider. Positive values indicate boosting the gain at that frequency band, while negative values indicate attenuating the gain at that frequency band.

3.5.5 Compressor/Limiter

Optional use of Compressor component or Limiter component.

The Limiter component 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 Compressor/Limiter parameter configuration interface;

② On/Off Button: Enables or disables the Compressor/Limiter;

③ Reset: Rests current interface parameter configuration to default;

④ Compressor/Limiter: Change between using the Compressor component or Limiter component;

⑤ Input: Displays the input level;

⑥ Compressor/Limiter Graph: Compressor/Limiter graph with level scale from -60dB～0dB;

⑦ Gain Attenuation: Graphically displays the amount of attenuation applied to the Channel, Gain Reduction reflects the degree to which the Compressor/Limiter attenuates the signal. For example, if the input signal exceeds the threshold by 3dB, the limiter may attenuate the signal by 3dB, resulting in a compression of 3dB;

⑧ Output: Displays the output level after Compressor/Limiter processing;

⑨ Threshold: Sets the level at which the Limiter has an effect, and the level at which the output is held. When the audio signal level exceeds the Threshold Level, the Compressor/Limiter reduces the signal level;

⑩ Ratio: 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;

⑪ Attack Time: Attack time is how fast the Compressor/Limiter reacts to a signal crossing the set threshold going up. Shorter attack time Compressor/Limiter can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Longer attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

⑫ Release Time: Release time is how fast the Compressor/Limiter reacts when the signal drops below the threshold and gain is restored to its non-limited level. Shorter release time can increase signal loudness, but is prone to suction effects; Longer release time provides a smooth transition and reduces suction effects, but may sound sluggish. Settings should be balanced according to audio characteristics.

3.5.6 Anti-Feedback

The Anti-Feedback component 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.

The Anti-Feedback is commonly used in scenarios such as conference rooms and concerts. Its core functions and features include real-time monitoring of audio input signals, analyzing frequency components that may cause feedback using digital signal processing (DSP) technology, and attenuating these components. The Anti-Feedback can detect feedback frequencies and process them within an extremely short time (0.01 seconds), thereby preventing feedback from occurring. It dynamically adjusts gain based on changes in the audio signal to ensure that feedback is suppressed without compromising audio quality.

① Anti-Feedback: Anti-Feedback parameter configuration interface;

② On/Off Button: Enables or disables the Anti-Feedback;

③ Auto: Automatically find the feedback point and suppress it;

④ Manual: When a suspicious feedback frequency is recognized, touch this button to apply a filter at that frequency;

⑤ Unlocked/Locked: Unlocked feedback frequency point or Locked feedback frequency point;

⑥ Reset: Rests current interface parameter configuration to default;

⑦ Frequency Grid: Displays the feedback points captured by the filter at different frequencies and the amount of attenuation;

⑧ Notch Feedback: Enables or disables the filter for different frequency bands;

⑨ Frequency: Sets the center frequency of the filter;

⑩ Attenuation : Sets the filter attenuation amount;

⑪ Filter Depth: Increases or decreases the depth of all filters;

⑫ Filter Bandwidth (Q): Sets the bandwidth of all filters;

⑬ On/Off Button: Enables or disables the Noise gain;

⑭ Noise: Sets the Noise gain.

3.6 Aux Output channel

3.6.1 Overview

① Overview parameter interface;

② Aux Pickup: Selects audio signal Pre-Fader/Post-Fader/Pre-Dynamics/Pre-All;

③ Delay On/Off Button: Displays Delay settings and Delay information;

④ Reset: Rests current interface parameter configuration to default;

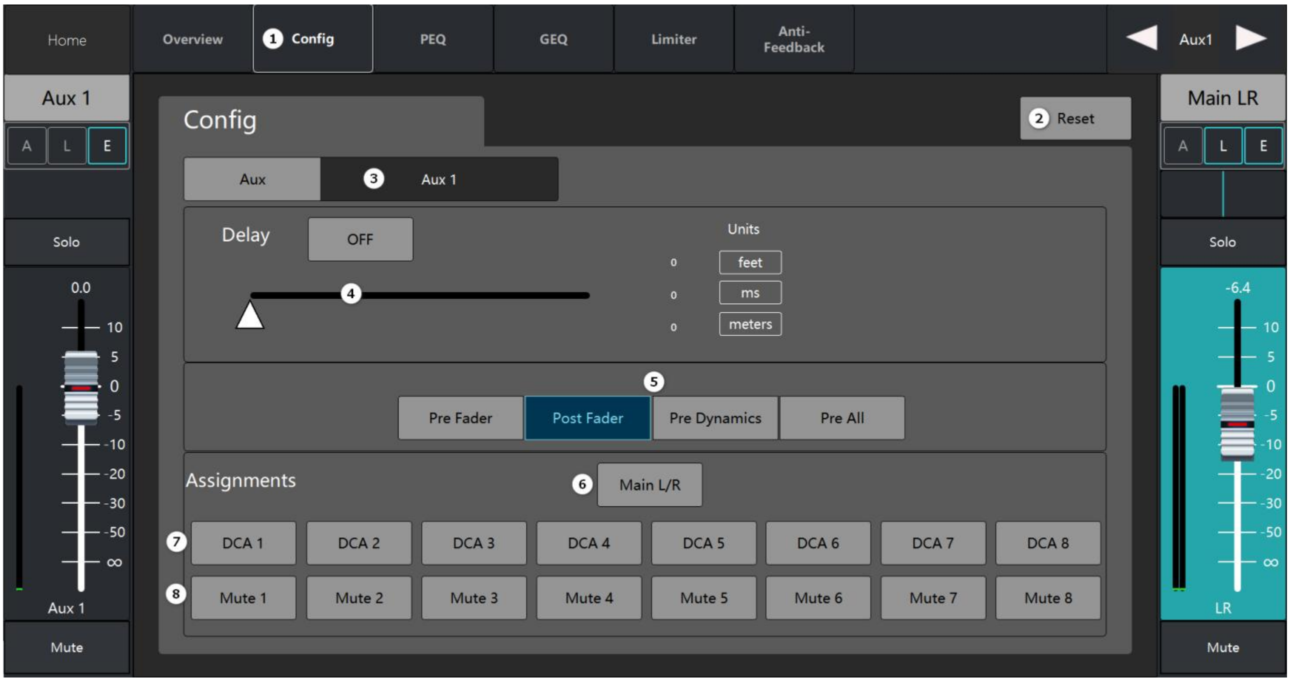
⑤ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑥ Mute Groups: Displays the channel has been assigned to the Mute Groups;

⑦ Parametric Equalizer: Enable or disable the Parametric Equalizer and display the Equalizer curve graph. Click on the curve graph to jump to the Equalizer configuration interface;

⑧ Compressor/Limiter: Enable or disable the Compressor/Limiter and display the Compressor/Limiter curve graph. Click on the curve graph to jump to the Compressor/Limiter configuration interface.

3.6.2 Configuration



① Configuration: Parameter configuration interface;

② Channel Name: Displays the channel name, touch the display keyboard to customize the channel name;

③ Reset: Rests current interface parameter configuration to default;

④ Delay: Enable or disable the current channel Delay by lightly touching the slider or using the data wheel to adjust the delay range (0～1000 ms);

⑤ Aux Pickup: Selects audio signal Pre-Fader/Post-Fader/Pre-Dynamics/Pre-All;

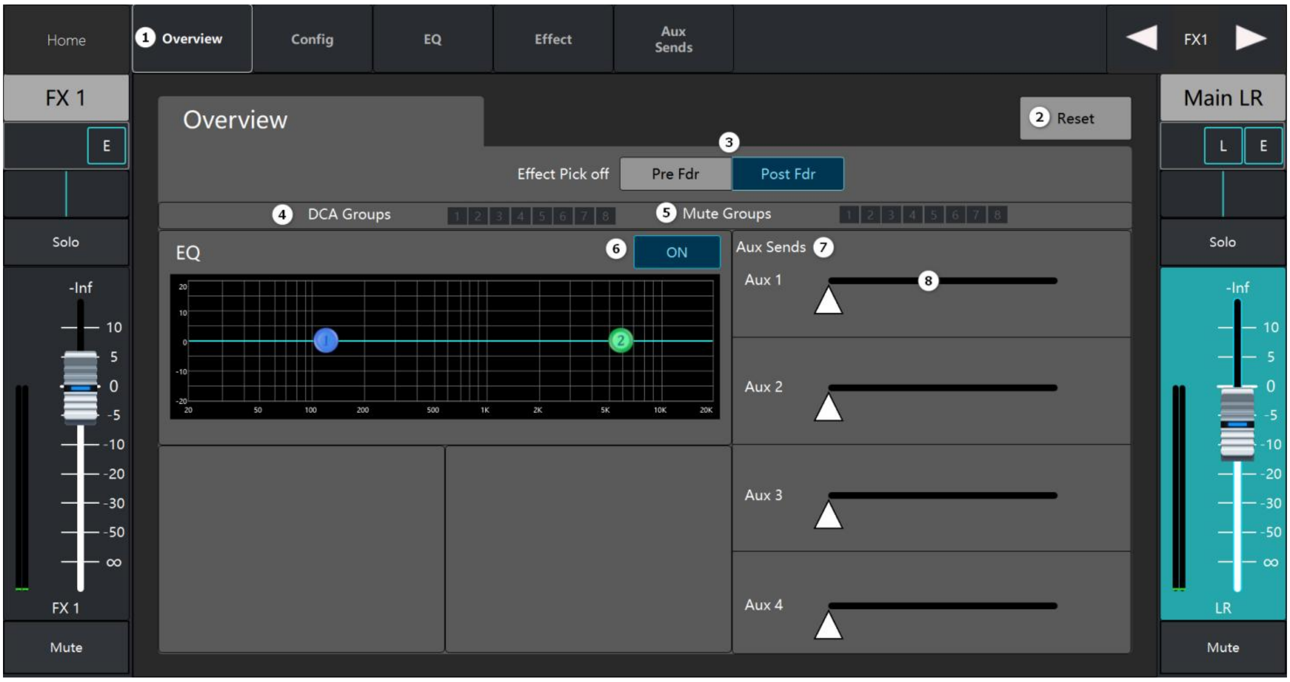
⑥ Main L/R: Sends the current channel signal to the Main output channel;

⑦ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑧ Mute Groups: Displays the channel has been assigned to the Mute Groups.

3.7 FX Channel

3.7.1 Overview



① Overview parameter interface;

② Reset: Rests current interface parameter configuration to default;

③ Effects Pickup: Selects audio signal Pre-Fader/Post-Fader;

④ DCA Groups: Displays the channel has been assigned to the DCA Groups;

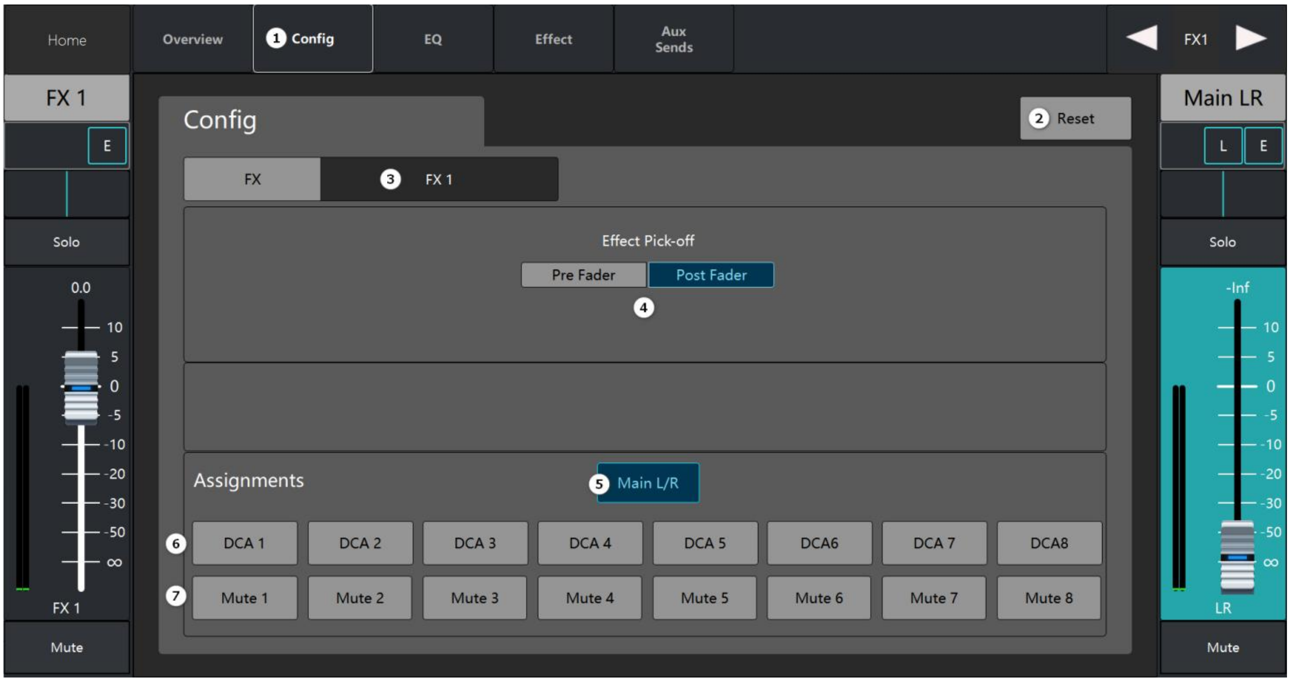
⑤ Mute Groups: Displays the channel has been assigned to the Mute Groups;

⑥ Parametric Equalizer: Enable or disable the Parametric Equalizer and display the Equalizer curve graph. Click on the curve graph to jump to the Equalizer configuration interface;

⑦ Aux Sends: Send the current channel signal to the Aux Auxiliary output channel;

⑧ Aux Send Slider: Sets the audio signal level sent from this channel to the Aux Auxiliary output channel.

3.7.2 Configuration



① Configuration: Parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Channel Name: Displays the channel name, touch the display keyboard to customize the channel name;

④ Effects Pickup: Selects audio signal Pre-Fader/Post-Fader;

⑤ Main L/R: Sends the current channel signal to the Main output channel;

⑥ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑦ Mute Groups: Displays the channel has been assigned to the Mute Groups.

3.7.3 Effects

I. Mono Delay

Mono Delay is an effect that plays the original audio again one or more times after a set period of time. All delayed sounds are output from a single channel.

① Effect parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Select the type of Effect including Mono Delay, Stereo Delay, Chorus, Reverb and Pitch Shift;

④ Effect Presets: Click the drop-down list to select the Effect presets ;

⑤ Input Level: The level of the signal not processed by the Effect;

⑥ Delay: Sets the delay time in milliseconds;

⑦ Tap Tempo: Sets the tempo of the regeneration using a tap. Press the "Tempo" button multiple times to set the delay time based on the song's current tempo, ensuring the delay stays in sync with the music's rhythm;

⑧ Regeneration: Sets the echo attenuation ratio, the echo will attenuation slowly and gradually according to the ratio;

⑨ Low Cut: Attenuates or cuts out the sound below this set frequency, the range is 20Hz～2KHz;

⑩ High Cut: Attenuates or cuts out the sound above the set frequency, the range is 200Hz～20KHz;

⑪ Output Level: The level after Effect processing;

⑫ FX Returns to Monitors: The current Effect is sent to the Aux output channel.

II. Stereo Delay

Stereo Delay is an enhanced version of mono delay. It distributes the delayed sound across the left and right channels, creating a wider and more complex spatial effect.

① Effect parameter configuration interface;

② Select the type of Effect including Mono Delay, Stereo Delay, Chorus, Reverb and Pitch Shift;

③ Effect Presets: Click the drop-down list to select the Effect presets ;

④ Reset: Rests current interface parameter configuration to default;

⑤ Input Level: The level of the signal not processed by the Effect;

⑥ Delay: Sets the delay time in milliseconds;

⑦ Tap Tempo: Sets the tempo of the regeneration using a tap. Press the "Tempo" button multiple times to set the delay time based on the song's current tempo, ensuring the delay stays in sync with the music's rhythm;

⑧ Regeneration: Sets the echo attenuation ratio, the echo will attenuation slowly and gradually according to the ratio;

⑨ Low Cut: Attenuates or cuts out the sound below this set frequency, the range is 20Hz～2KHz;

⑩ High Cut: Attenuates or cuts out the sound above the set frequency, the range is 200Hz～20KHz;

⑪ Output Level: The level after Effect processing;

⑫ FX Returns to Monitors: The current Effect is sent to the Aux output channel.

III. Chorus

Chorus creates the illusion of multiple sound sources (or instruments) sounding simultaneously by subtly modulating the delay time, pitch waveform, and modulation frequency of the original signal, resulting in a richer, warmer sound.

① Effect parameter configuration interface;

② Select the type of Effect including Mono Delay, Stereo Delay, Chorus, Reverb and Pitch Shift;

③ Effect Presets: Click the drop-down list to select the Effect presets;

④ Reset: Rests current interface parameter configuration to default;

⑤ Input Level: The level of the signal not processed by the Effect;

⑥ Ratio: Sets the speed of the pitch shift;

⑦ Depth: Sets the time range of audio signal adjustment;

⑧ Sine/Saw Option: Selects the mode of tone change;

⑨ Low Cut: Attenuates or cuts out the sound below this set frequency, the range is 20Hz～2KHz;

⑩ High Cut: Attenuates or cuts out the sound above the set frequency, the range is 200Hz～20KHz;

⑪ Output Level: The level after Effect processing;

⑫ FX Returns to Monitors: The current Effect is sent to the Aux output channel.

IV. Reverb

Reverb simulates the effect of countless echoes blending together as sound reflects off surfaces within a physical space, such as a room, hall, or cave. It is the most important tool for creating a sense of space.

① Effect parameter configuration interface;

② Select the type of Effect including Mono Delay, Stereo Delay, Chorus, Reverb and Pitch Shift;

③ Effect Presets: Click the drop-down list to select the Effect presets;

④ Reset: Rests current interface parameter configuration to default;

⑤ Input Level: The level of the signal not processed by the Effect;

⑥ Feedback: Adjusts the total amount of delayed signal returned to the input channel, thereby changing the reverb effect;

⑦ Damping: High-frequency attenuation amount control;

⑧ Dry/Wet: The ratio of the original signal to the signal processed with reverb;

⑨ Output Level: The level after Effect processing;

⑩ FX Returns to Monitors: The current Effect is sent to the Aux output channel.

V. Pitch Shift

Pitch Shift alters the pitch of the original audio signal, raising or lowering it.

① Effect parameter configuration interface;

② Select the type of Effect including Mono Delay, Stereo Delay, Chorus, Reverb and Pitch Shift;

③ Effect Presets: Click the drop-down list to select the Effect presets;

④ Reset: Rests current interface parameter configuration to default;

⑤ Input Level: The level of the signal not processed by the Effect;

⑥ Left: Control the pitch-shift gain for the left channel using the slider;

⑦ Right: Control the pitch-shift gain for the right channel using the slider;

⑧ Low Cut: Attenuates or cuts out the sound below this set frequency, the range is 20Hz～2KHz;

⑨ High Cut: Attenuates or cuts out the sound above the set frequency, the range is 200Hz～20KHz;

⑩ Left Delay: Sets the left channel pitch shift delay time;

⑪ Right Delay: Sets the right channel pitch shift delay time;

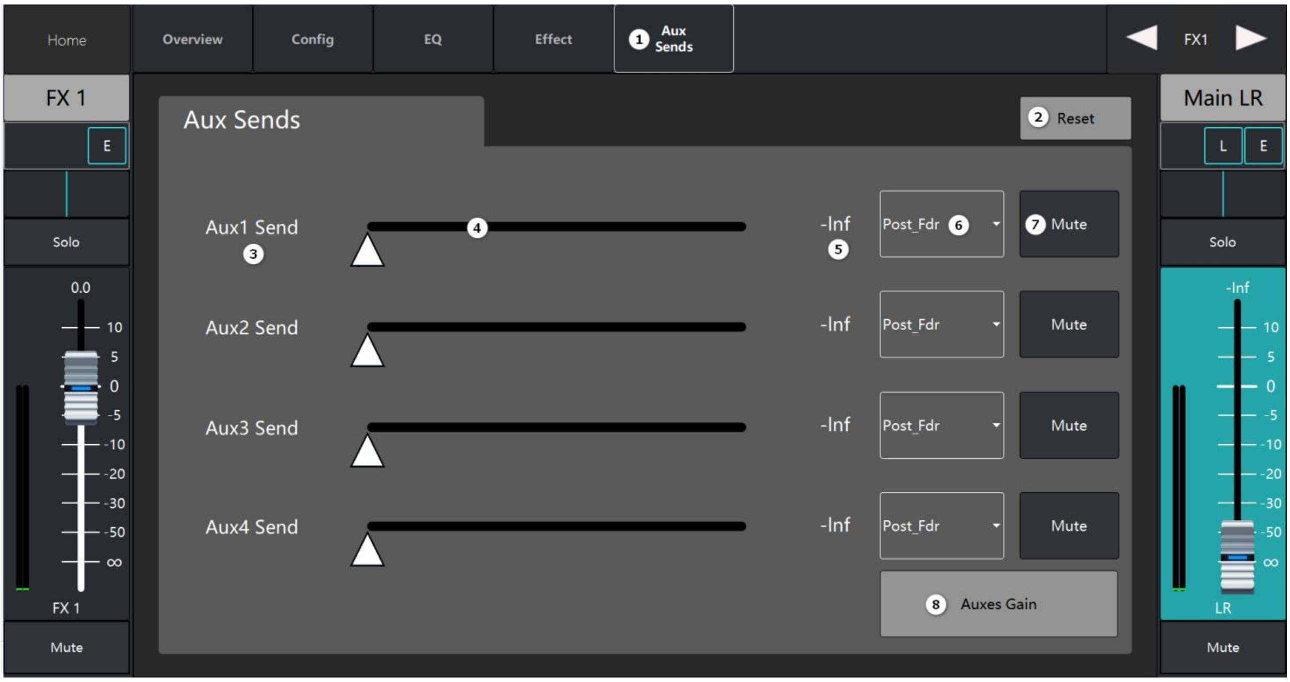
⑫ Left Sharp/Lower: Left channel selects Sharp or Lower Shift;

⑬ Right Sharp/Lower: Right channel selects Sharp or Lower Shift;

⑭ Output Level: The level after Effect processing;

⑮ FX Returns to Monitors: The current Effect is sent to the Aux output channel.

3.7.4 FX Aux Sends



① Aux Sends: Auxiliary sends parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Aux output channel name;

④ Aux Send Slider: Sets the audio signal level sent from this channel to the Aux output channel;

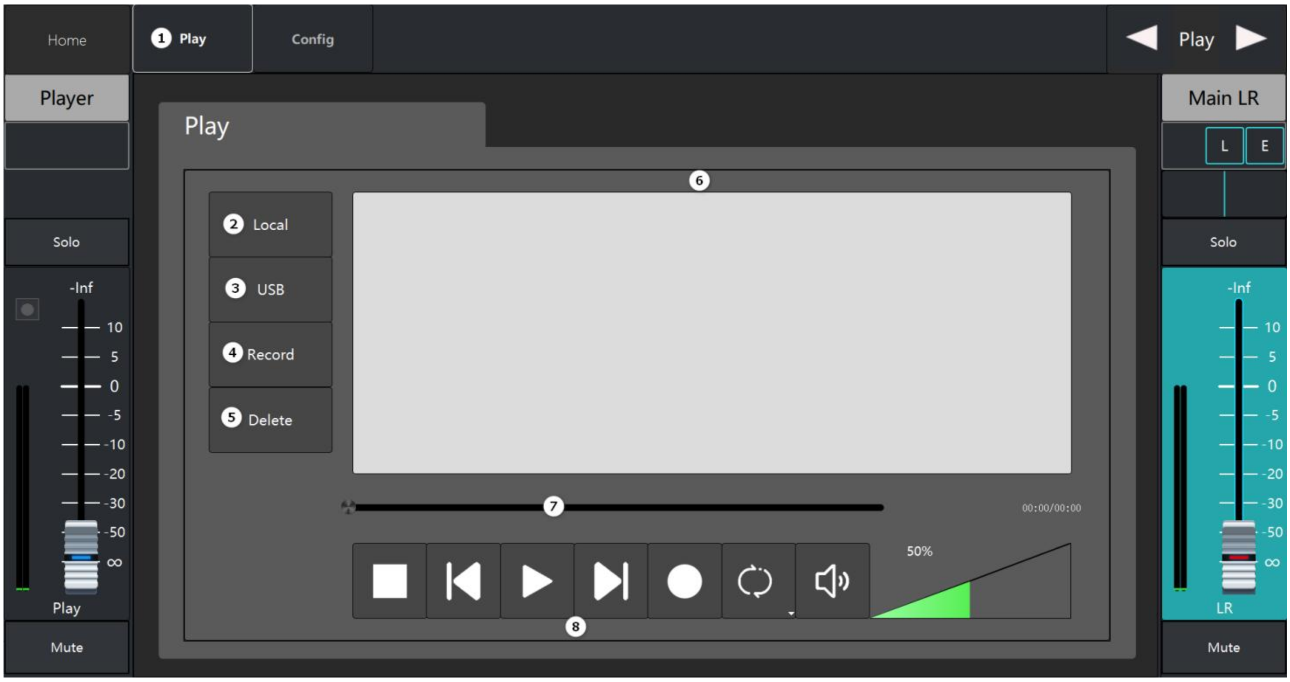
⑤ Displays the gain of the current sending channel;

⑥ Touch the drop-down box to select to send Pre-Fader/Post-Fader/Pre-Dynamics/Pre-All signals to the Aux output channel;

⑦ Mute: Mute the Aux Sends channel without affecting any other Aux outputs or sends;

⑧ Auxes Gain: Touch the button to jump to the Aux sends configuration interface.

3.8 Player/Recorder



① Player or Recorder interface;

② Local: List of local music files;

③ USB: List of audio files in external USB (Class A) removable hard disk devices;

④ Record: List of recorded files;

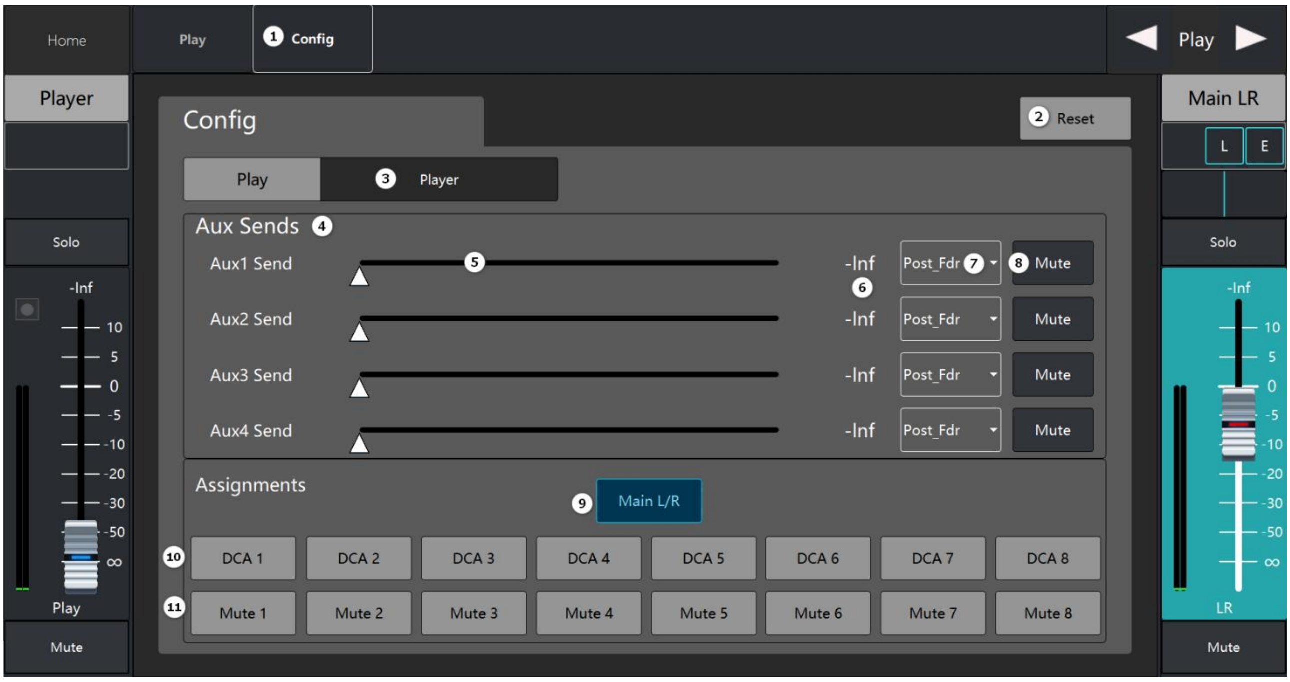
⑤ Delete: Only local files as well as recorded files are allowed to be deleted;

⑥ List: Displays the current list of tracks, touch to select the track to play;

⑦ Audio file playback progress bar, slide to determine the playback position;

⑧ Playback controls: Stop, Previous, Pause, Next, Record, Play Mode, Mute, Volume Adjustment.

3.8.1 Player Configuration



① Configuration: Parameter configuration interface;

② Reset: Rests current interface parameter configuration to default;

③ Channel Name: Displays the channel name, touch the display keyboard to customize the channel name;

④ Aux output channel name;

⑤ Aux Send Slider: Sets the audio signal level sent from this channel to the Aux output channel;

⑥ Displays the gain of the current sending channel;

⑦ Touch the drop-down box to select to send Pre-Fader/Post-Fader/Pre-Dynamics/Pre-All signals to the Aux output channel;

⑧ Mute: Mute the Aux Sends channel without affecting any other Aux outputs or sends;

⑨ Main L/R: Sends the current channel signal to the Main output channel;

⑩ DCA Groups: Displays the channel has been assigned to the DCA Groups;

⑪ Mute Groups: Displays the channel has been assigned to the Mute Groups.

3.9 Other Functions

3.9.1 Menu Settings

① DCA Groups: Navigates to the DCA Groups interface;

② Real Time Analyzer: Navigates to the Real Time Analyzer interface;

③ Noise Generator: Including Sine wave, Pink Noise and White Noise signal;

④ FX Overview: Navigates to the Effects interface;

⑤ Scenes: Scene setting interface, provides 30 scenes, can be increased to 100 scenes;

⑥ Clear: One-click clear function (Algorithm, Mute, Monitor, Gain);

⑦ Copy/Paste: Copy the current channel configuration parameters to other channels, but only between channels of the same type;

⑧ Initialize Configuration: Clear the current scene file configuration and reset to the default configuration;

⑨ Upload File/Network: PC software is to upload audio files to the device locally; Digital Mixer interface is to display the device IP address and turn on the Wi-Fi adapter;

⑩ Search Mixer/Fader Speed: PC software is to jump to the search Mixer interface; Digital Mixer interface is Fader calibration;

⑪ Language: Selects the languages (English, Simplified Chinese, Traditional Chinese);

⑫ Update/Firmware: PC software is to upload firmware to Mixer for upgrades; Digital Mixer interface is to upload firmware to the device via USB to upgrade the firmware.

⑬ Version: Displays the version information;

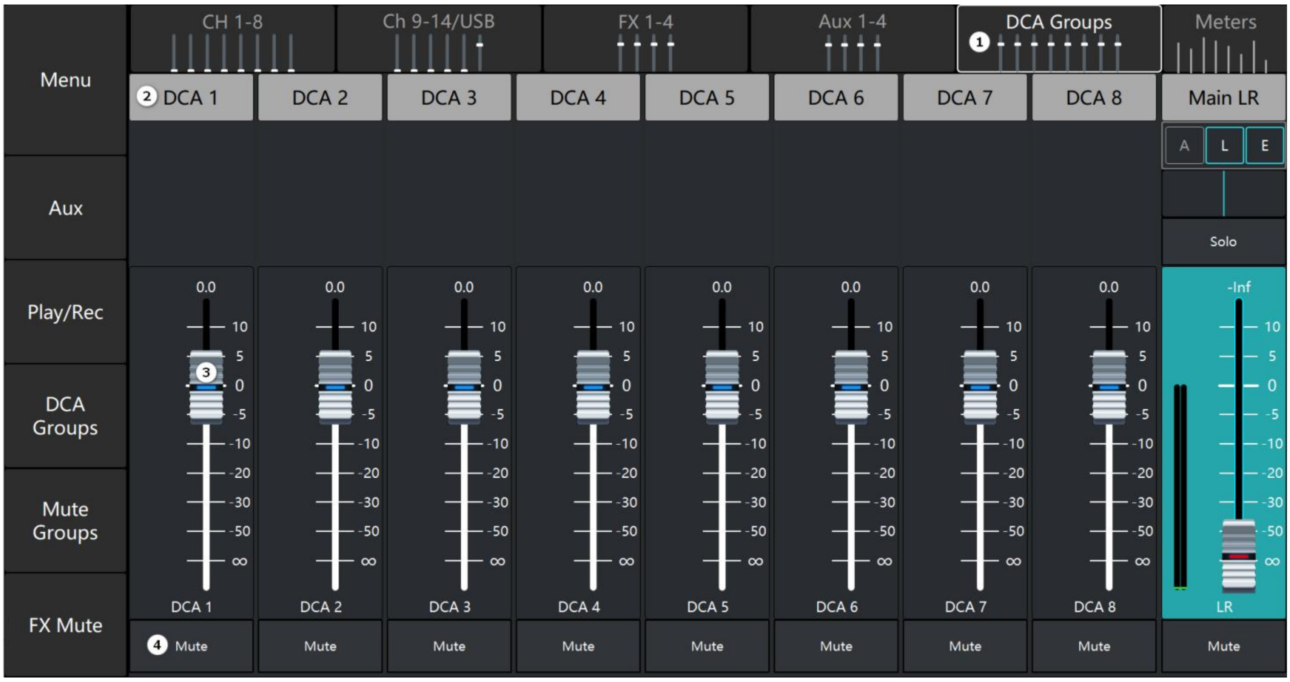
⑭ Service: UDP Service: Check/cancel the central command control check box; Master: Set the current Mixer as the Master; Slave: Set the current Mixer as the Slave, sets the IP address of the Master to enable real-time synchronization of Master data over the network, thereby achieving dual-Mixer hot backup;

⑮ Login Management: Password Login: Password login is required to start up the Mixer. The default login password is 123456, which can be changed on the login interface. Lock (Enabled): Enables the LOCK button, requiring a login password to unlock. Lock (Disabled): Disables the LOCK button.

3.9.2 DCA Groups

DCA Groups are control groups that virtually bundle multiple channel faders together, allowing simultaneous level control of these channels via a single master fader.

I. DCA Groups Main Interface



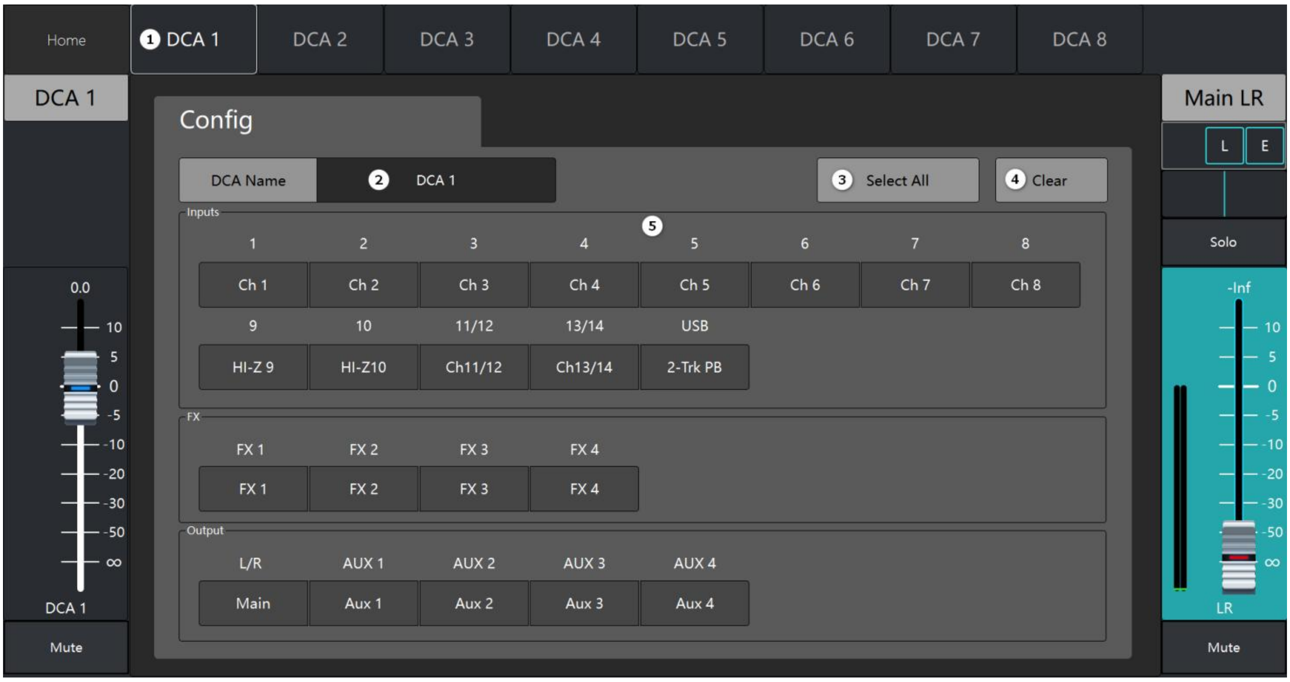
① DCA Groups 1-8: DCA Groups main interface;

② DCA Groups channel name;

③ DCA Groups channel Fader that control the gain of all channels that are grouped into that Groups;

④ Mute: Mutes the DCA Groups.

II. DCA Groups Configuration Interface



① DCA Groups: Selects a DCA Group ;

② Select All: Select all channels to be assigned to the DCA Group;

③ Clear: Clear all assignments in the selected DCA Group;

④ Group Name: The name of the Group channel is displayed, and the name can be customized and modified;

⑤ Touch the assign button of a channel to add the channel to the selected DCA Group, and both Input and Output and Effect return can be assigned to the DCA Group.

Note: When a channel is assigned to one or more DCA Groups, the output of that channel is equal to the sum of all Faders in the DCA Group plus the value of the channel's own fader. When a channel is assigned to one or more DCA Groups, if you need the audio signal from that channel to pass through, you must unmute both the channel and all DCA Groups. The same principle applies when a channel belongs to both a DCA Group and a mute group. If the audio signal from the channel needs to pass through, the mute must be canceled for all groups.

3.9.3 Mute Groups

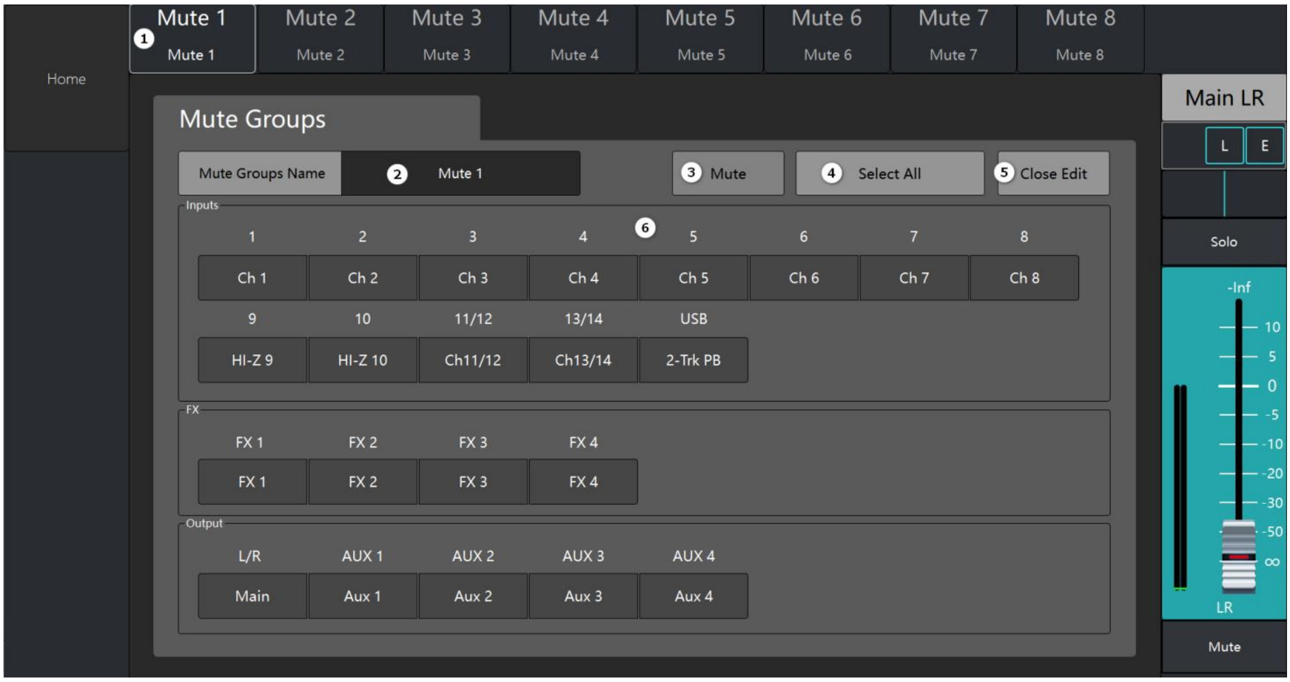
Mute Groups link the mute buttons of multiple channels together. When a mute group is activated, all channels within the group are simultaneously muted or unmuted.

I. Mute Group Main Interface

① Mute Groups Edit: Click to enter the Mute Group detail configuration interface;

② Mute: Mute selected Group.

II. Mute Group Details Interface



① Mute Groups: Selects a Mute Group ;

② Group Name: The name of the group channel is displayed, and the name can be customized and modified;

③ Mute: Mute or Unmute the channels that have been assigned to the Mute Group;

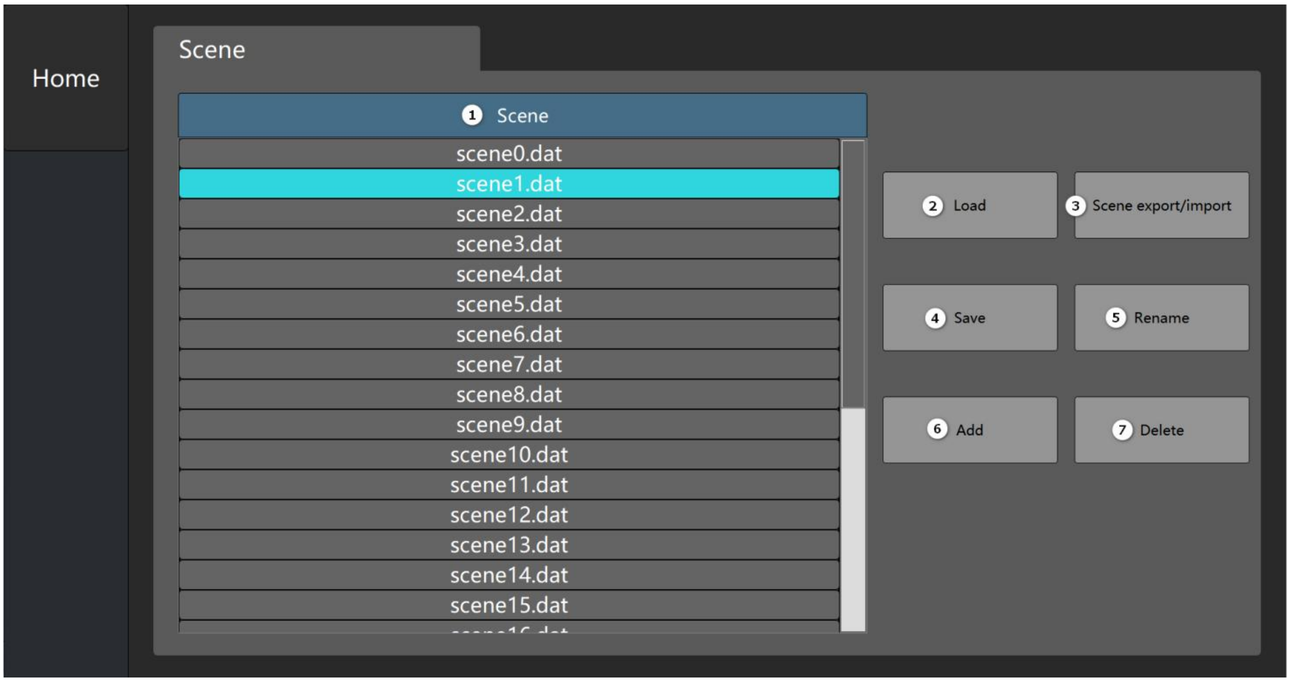
④ Select All/Clear Assignments: Select or clear all Mute Groups assignments;

⑤ Close Edit: Touch to navigate to the main Mute Group interface;

⑥ Touch the assign button of a channel to add the channel to the selected DCA Group, and both Input and Output and Effect return can be assigned to the DCA Group.

Note: When a channel is muted via the Mute Groups, the mute button for that channel will turn orange. When a channel is muted via the channel mute button and the Mute Groups, the mute button for that channel turns red. For an audio signal to pass through, all mutes associated with the channel must be canceled.

3.9.4 Scene Configuration



① Scene: Scene list, select the scene for editing by drop-down slider;

② Load Button: Touch the scene number in the list to load;

③ Import/Export Scene Button:Import or Export scenes;

④ Save Button: Save the current scene content to another scene;

⑤ Rename Button: Modify the name of the selected scene;

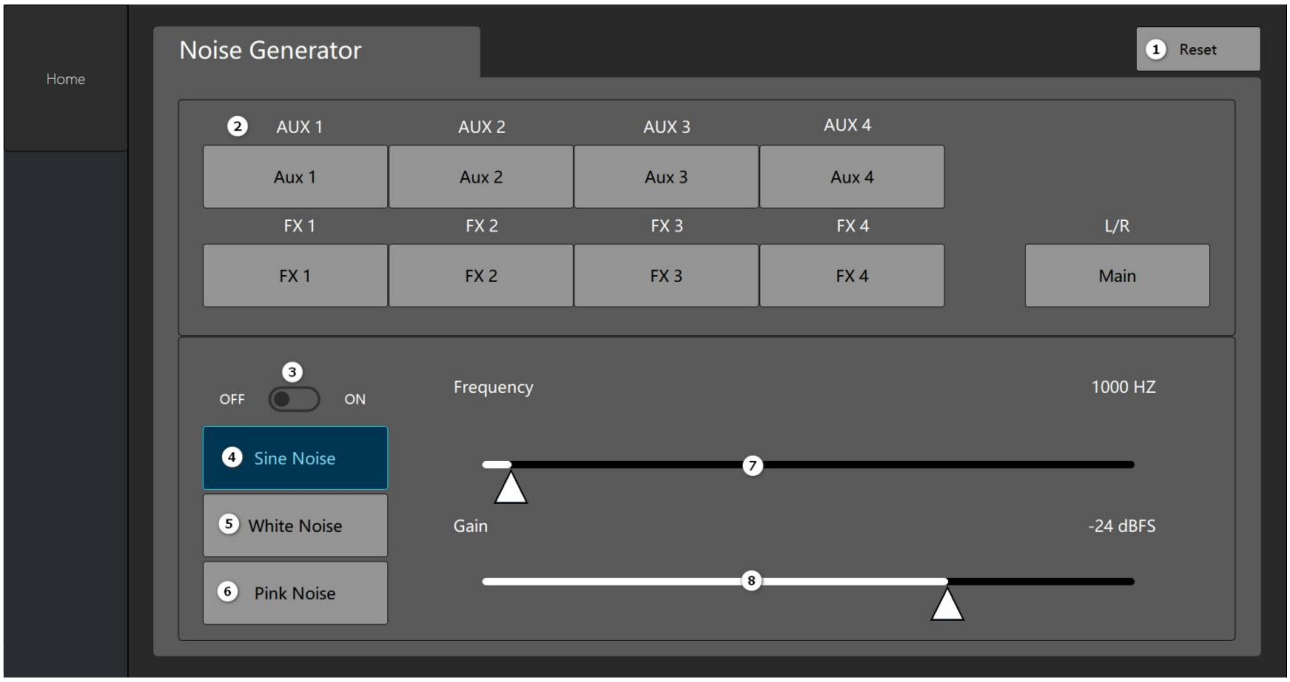
⑥ Add Button: Add a new scene;

⑦ Delete Button: Deletes the selected scene.

3.9.5 Noise Generator

Noise Generator signals include Sine Wave, Pink Noise and White Noise, which play an important role in audio system commissioning and measurement.

- 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.



① Reset: Resets the Noise Generator settings;

② Send Button: Selects the channel name to send the noise to that channel output;

③ On/Off Button: Enables or disables the Noise Generator;

④ Sine Noise: Enables the Sine Wave signal output;

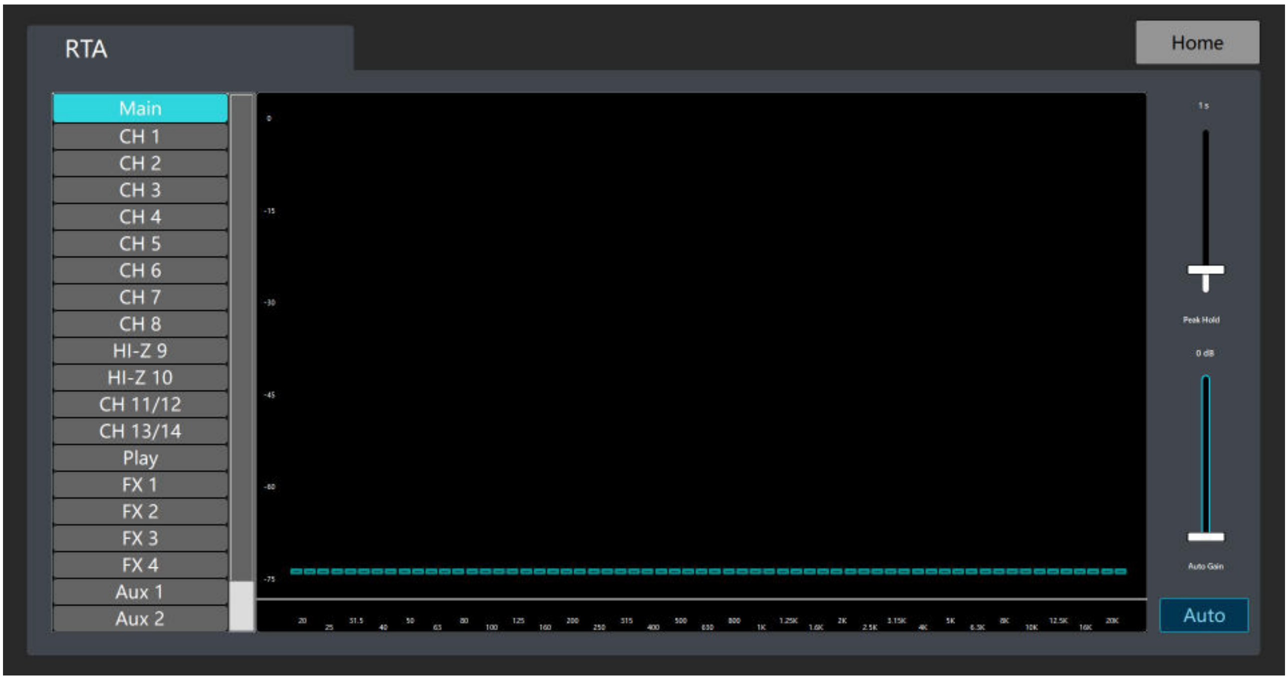
⑤ White Noise: Enables White Noise signal output;

⑥ Pink Noise: Enables Pink Noise signal output;

⑦ Frequency: Sets the noise output frequency;

⑧ Gain: Sets the noise output signal level.

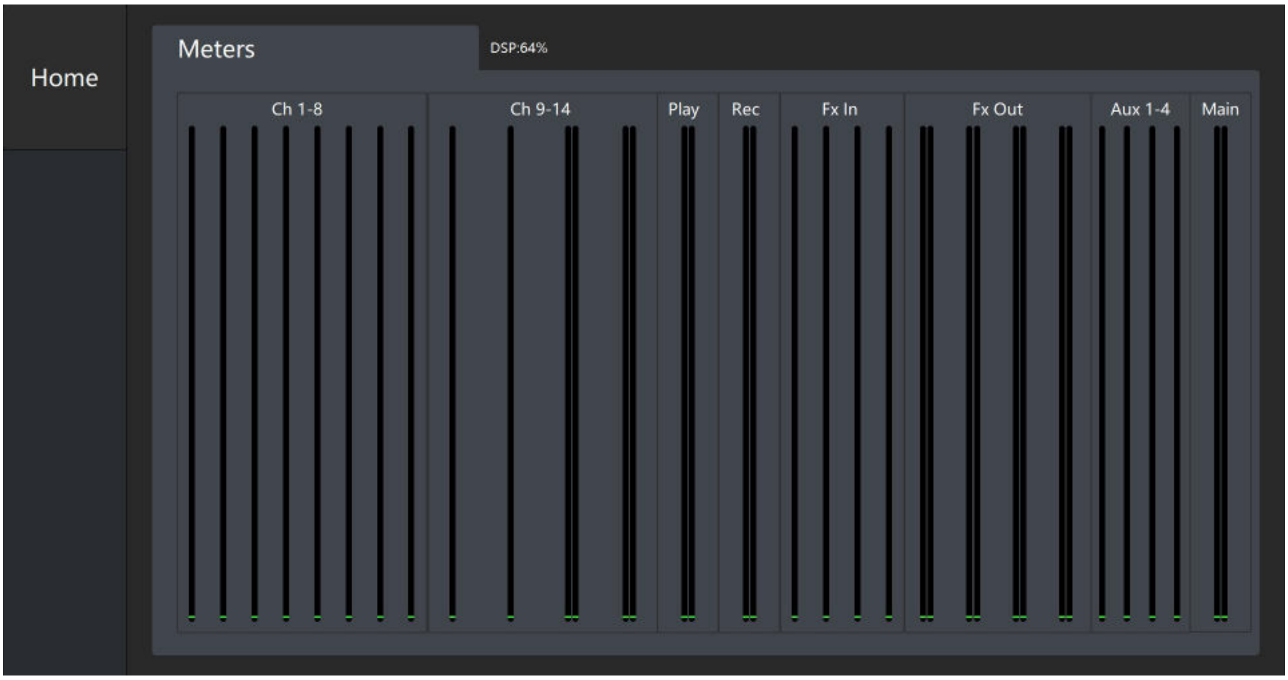
3.9.6 Real Time Analyzer



The real time analyzer sources mainly include input channels, playback channels, Main L/R, Aux Auxiliary outputs, and FX effects.

3.9.7 Level Meters

The level overview is used to display real-time level information for all input/output channels.



3.9.8 Auxiliary Output Matrix Overview

① Channels 1-8, high resistance channels 9-10, stereo channels 11-14, USB, and FX effects channels 1-4 can be selected in the navigation bar;

② Aux output channel name;

③ Input Channel Send Control: Sets the audio signal level sent from the input channel to the Aux output channel;

④ Aux Masters: Controls the Aux channel gain via a slider;

⑤ Pre-Fader/Post-Fader: Displays the position of the signal send (Pre-Fader or Post-Fader).

Chapter 4 Packing List

Device	Power Adapter	User Manual
1PCS	1PCS	1PCS

Chapter 5 Specification

Category	Parameter Item	Parameter Description
Peripherals	Input Interfaces	16 input channels: 8 balanced XLR/TRS combination digital gain microphone channels; 2 groups (4 channels) TRS6.35mm stereo input channels; 2 TRS6.35mm high-resistance mono input channels; 2 USB playback channels
	Output Interfaces	10 output channels: 2 L/R XLR Main output channels; 4 XLR Aux output channels; 1 group (2 channels) TRS6.35mm stereo monitoring channels; 2 USB recording channels
	Display	7-inch high-definition touch screen, 1024 x 600 resolution
	Control Interfaces	1 RJ45 interface (controlled via UDP protocol), 2 USB A interfaces
Audio processing	Processor	ADI SHARC ADSP-21489 450 MHz high performance 32-bit/40-bit floating-point DSP processor; 24-bit A/D and D/A converter, 48kHz sampling rate
	Input Channel	Functional component: Delay, Polarity, Phantom power, 4-band Parametric Equalizer, Compressor, Noise Gate
	Output Channel	Functional component: Delay, 6-band Parametric Equalizer, 31-band Graphic Equalizer, Compressor/Limiter, Acoustic Notch Feedback Canceler
	Phantom Power	DC 48V

	Signal-to-Noise Ratio	108dB
	Frequency Response	20Hz～20KHz, ±0.2dB
	THD+N	≤0.003% @1kHz, +4dBu
	Maximum Output Level	20dBu
	Maximum Input Level	20dBu
	Analog/Digital Dynamic Range	110dB
	Digital/Analog Dynamic Range	110dB
	Input to Output Dynamic Range	108dB
	Input Impedance	Balanced: 2KΩ
	Output Impedance	Balanced: 100Ω
	Noise Floor	-90dBu
	Channel Isolation	70dB@1kHz
	Common Mode Rejection Ratio	>60dB@50Hz
	System Latency	≤6ms
	Filter	Low Cut, High Cut, Low Shelf, High Shelf
	Equalizer	Parametric Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB, Q Factor: 0.4～4 Graphic Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB
	Effects	5 effect types: Mono Delay, Stereo Delay, Chorus, Reverb, Pitch Shift
General	Operating Voltage	AC 100V～240V, 50Hz/60Hz

specification	Maximum Power	30W
	Operating Temperature and Humidity	0℃～55℃, 10%～90%RH, No condensation
	Installation	Table placement, adding hanging ears, rack installation
	Power Supply	External power supply, output DC 19V, 3.42A
	External Power Supply Dimensions (L×W×H)	107.5mm×45.5mm×30mm
	Product Dimensions (L×W×H)	410mm×250.4mm×69mm
	Net Weight	2.7kg
	Package Dimensions (L×W×H)	590mm×430mm×110mm
	Package Weight	3.1kg

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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