TOPPING

Professional

KB Control Center

V1.0

User manual

Contents

1. Install and open KB Control Center

System requirements

- macOS 12 or later / Windows 10 or later
- CPU: Intel-based Mac or PC(Pentium or equivalent), ≥ 1 GHz
- RAM: 2 GB minimum,4 GB or more recommended
- USB Port: One available USB 2.0 / 3.0 port
- Storage: At least 512 GB of free space recommended

Download and install

1. Visit the website to download the KB Control Center application.

Win: https://topping.pro/download/Roy05zQ

Mac: https://topping.pro/download/Em1vree

1. Unzip and double-click to run the installation package, and follow the prompts to install KB Control Center.
2. Use the included USB cable to connect the USB-C port of your computer and device.
3. Double-click the KB Control Center shortcut on your computer desktop to launch KB Control Center.
4. When the computer is connected to the Internet, connected to the device and turned on, KB Control Center will notify you if new firmware is available for update.

Note: The Windows version of KB Control Center has more buffering settings than the Mac version, but the rest are the same.

Error message description

If "Please install driver" or "Device not connected" is displayed when opening KB Control Center, please check:

1. Check whether the operating system meets the requirements: macOS version 12 or higher; Windows 10 or higher.
2. Check that the USB cable is fully inserted and that it is connected to the USB-C port on the device, not the OTG port.
3. Check if the device is powered on.
4. Close all anti-virus software, uninstall and delete KB Control Center, and reinstall KB Control Center.
5. Check whether the USB cable is damaged and try using another USB cable no longer than 2 meters.
6. Try connecting the POWER interface on the back of KB22 to a DC 5V independent power supply.

7. Try using another USB port on your computer. It is best to use the USB port on the back of the computer host.
8. Check whether it is a computer problem, and if conditions permit, try to connect to another computer.

2. Mode introduction

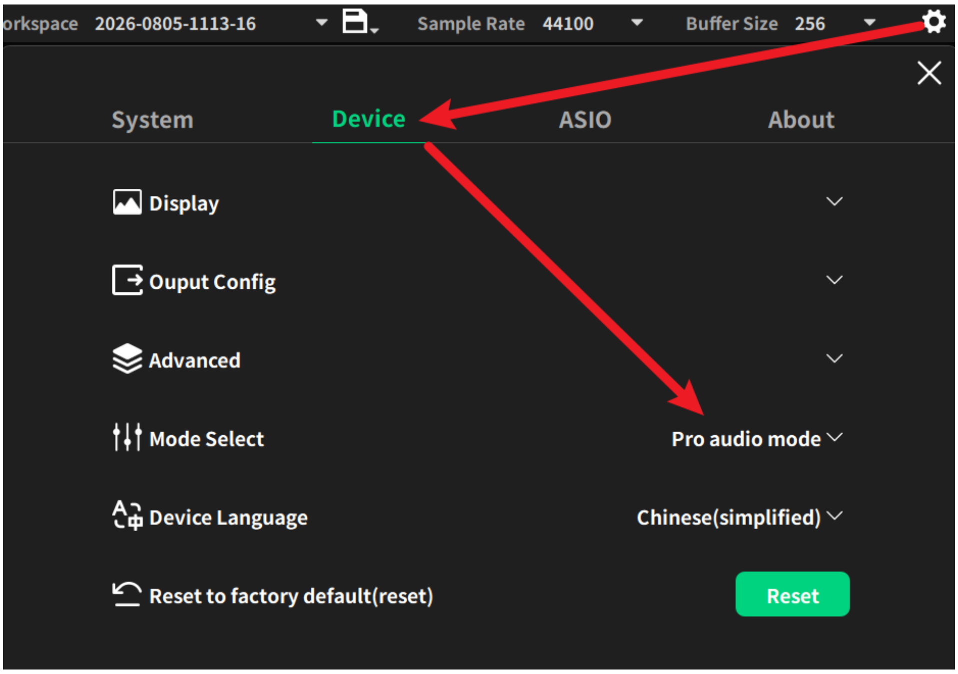
2.1 Model comparison

In order to meet the needs of different users, the device is equipped with three working modes, which can be switched according to actual usage scenarios. The default is professional mode.

model	HiFi mode	Professional mode	eSports mode
Applicable scenarios	Suitable for daily listening to music and other lightweight needs	For professional users, suitable for recording studios, multi-channel routing and music production	For game scenarios, suitable for various game live broadcast needs
Sampling rate	PCM 44.1-384kHz DSD Native 64-256/ Dop 64-128	44.1-192kHz	44.1kHz/48kHz
Input channel	USB-C、OTG	IN1、IN2、3.5-Mic、OTG、Playback 1/2~7/8	IN1、IN2、3.5-Mic、OTG、Playback 1/2~5/6
mix channel	none	Mix A–C (3 sets)	3 groups
internal recording	none	Loopback 1/2、3/4、5/6	Recording
Effect	EQ	EQ, ducking, noise reduction, compression, expansion, delay, reverb	ducking, noise reduction, vocal enhancement，output audio effect enhancement, output spatial enhancement

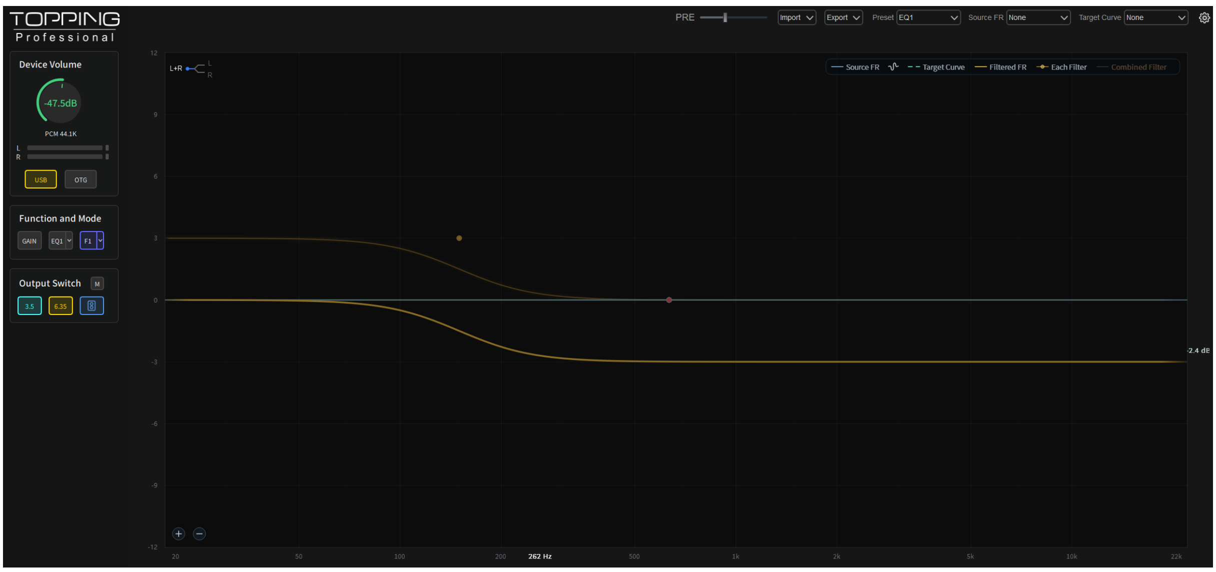
model	HiFi mode	Professional mode	eSports mode
PEQ adjustment	Headphones/Line Out: 10-band PEQ	Microphone input: 4-band PEQ Headphone output: 10-band PEQ	none

2.2 Mode switch

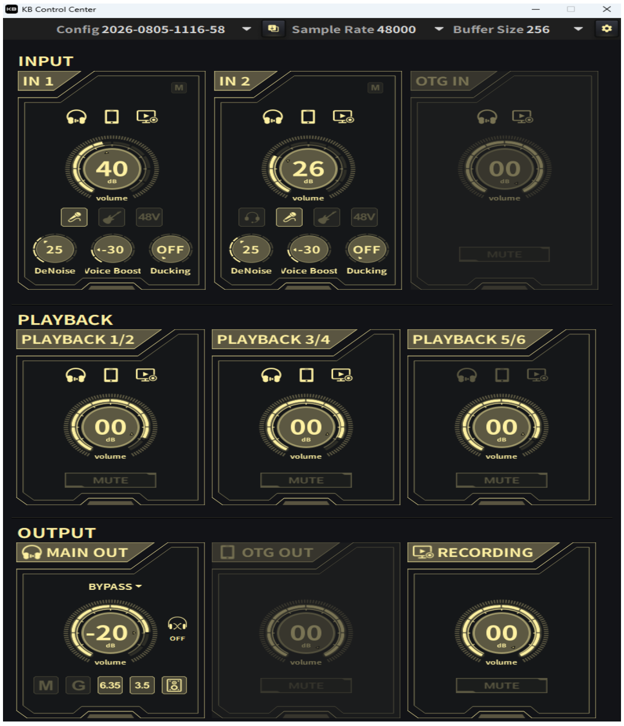


2.3 Page overview

HIFI mode



Game mode



Pro audio mode

3. Basic operations

·Click the button to turn on/off the corresponding function.

·Click the knob, drag the mouse up and down or roll the mouse wheel to turn the knob.

·Double-click a knob/fader to return to its default position.

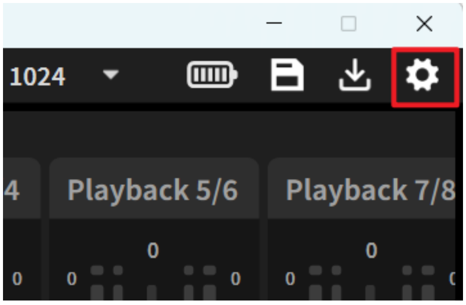
·Click the value corresponding to the push rod to modify the value directly.

4. System settings

4.1 Go to settings

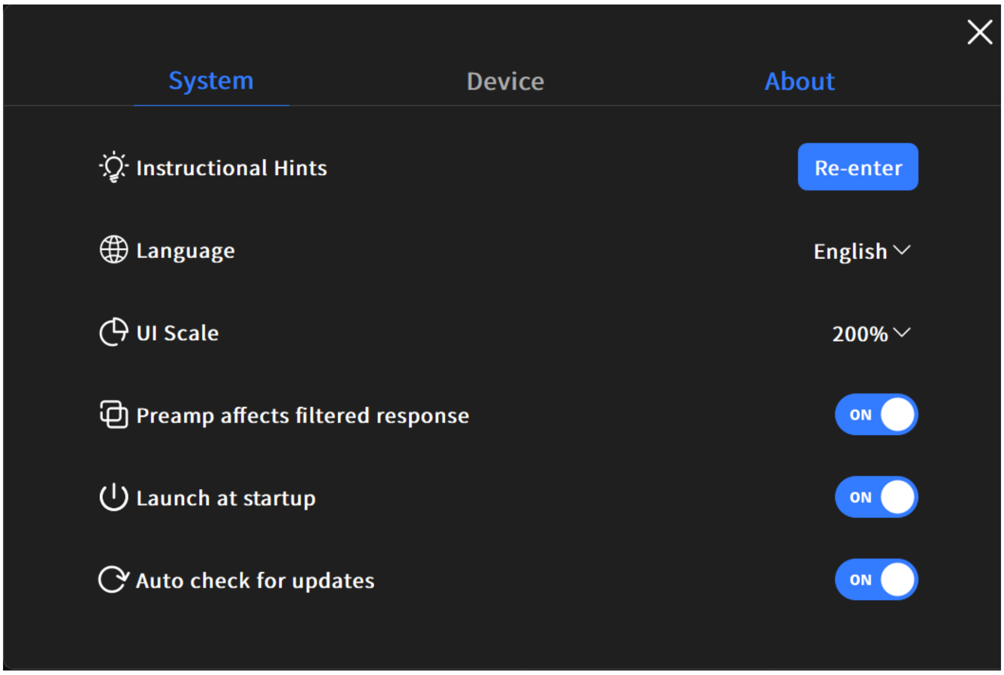
Click the settings icon in the upper right corner of the page to enter settings.

The system setting items in different modes may be different. Please refer to the actual software interface for details. The following pictures take HIFI mode as an example.



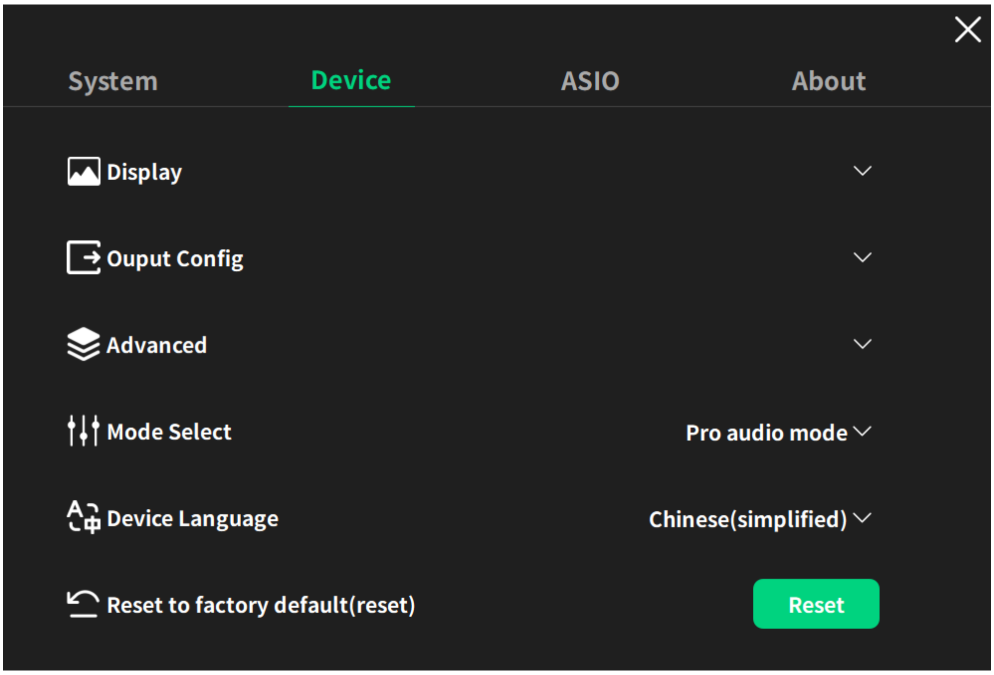
4.2 System settings

In this settings page, you can configure teaching prompts, language selection, auto-start at boot, automatic check for updates, etc. For related settings, please refer to the picture example below.

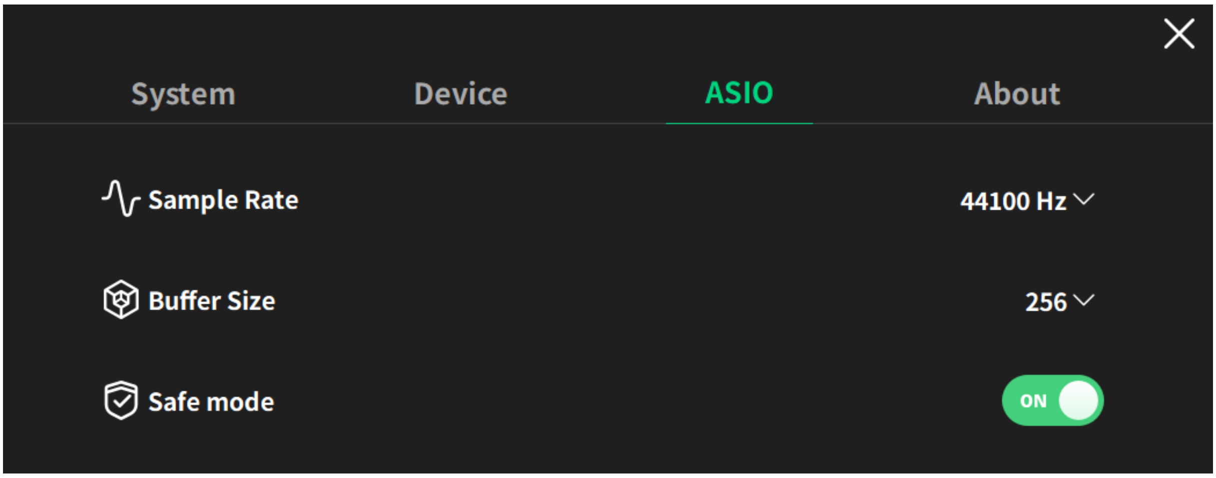


4.3 Device settings

Please refer to the picture below for relevant device settings. There may be differences in different modes. The specific settings in the actual software shall prevail. For specific setting instructions, please refer to "6. Menu Settings" in the KB22 manual.



4.4 ASIO settings



Sampling rate

The gaming mode can be set to 44.1 or 48kHz, and the professional recording mode can be set to 44.1, 48, 88.2, 96, 176.4 or 192kHz. The higher the sampling rate, the higher the fidelity of the recorded audio, but at the same time, audio recorded at a high sampling rate requires more storage space.

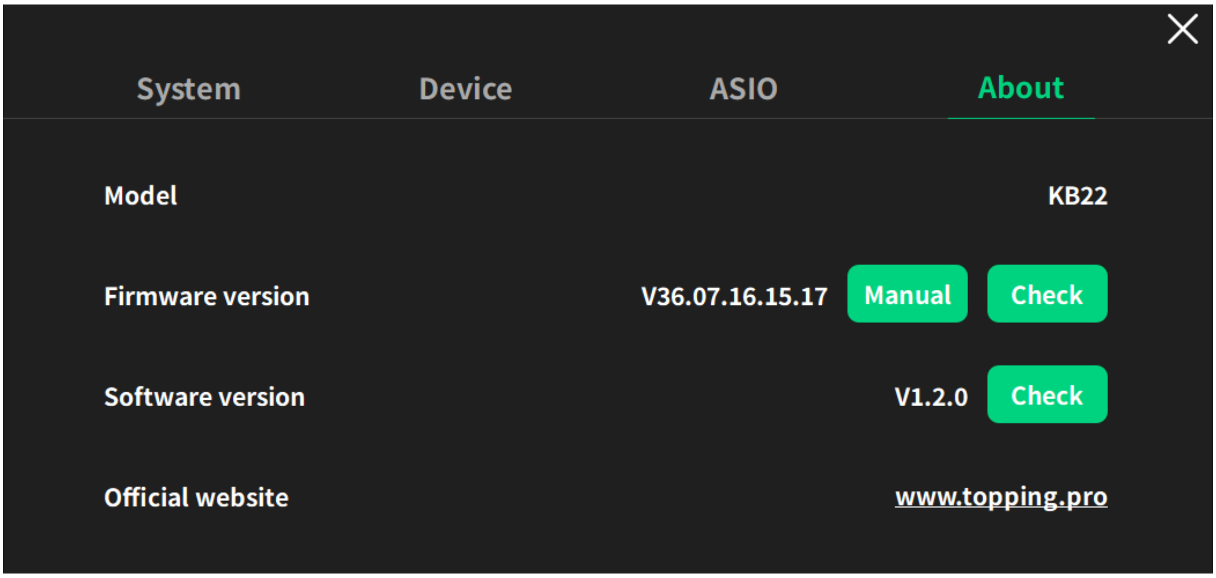
Buffer size (Windows OS only)

The smaller the buffer, the smaller the delay, but the higher the requirements on computer performance. When you experience playback stuttering/crackling, try increasing the buffer size.

Safe mode

Turning on safe mode can improve stability and reduce the probability of cracking sound.

4.5 About

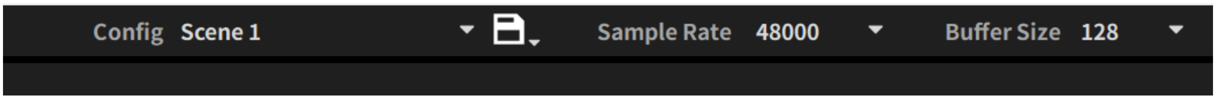


You can view the device model, device software version and KB Control Center version.

You can check for updates or visit the Topping Professional official website for more information.

5. Global Status Bar

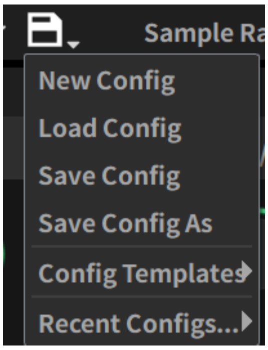
Saving and Loading Configurations (Gaming Mode and Professional Mode)



This feature is designed for users with multiple usage scenarios; you can save settings for input, the mixer, internal recording, and output within the KB Control Center for different scenarios.

When switching scenarios, you can quickly load the corresponding settings without having to adjust each parameter individually.

New Config / Load Config / Save Config / Save Config As / Config Templates / Recent Configs



New Config: Create a new configuration.

Enter a name and press Enter to finalize creation; this configuration will then become the active one.

Load Config: Load a selected configuration file and apply it to the current interface.

Save Config: Save the current interface state to the configuration file currently in use.

Save Config As: Save the current settings as a new configuration.

Enter a new name and press Enter to save; the original configuration remains unchanged.

Config Templates: Save current settings as a template, or select an existing template to quickly apply preset settings.

Recent Configs: View and quickly open recently loaded configurations.

Sampling rate

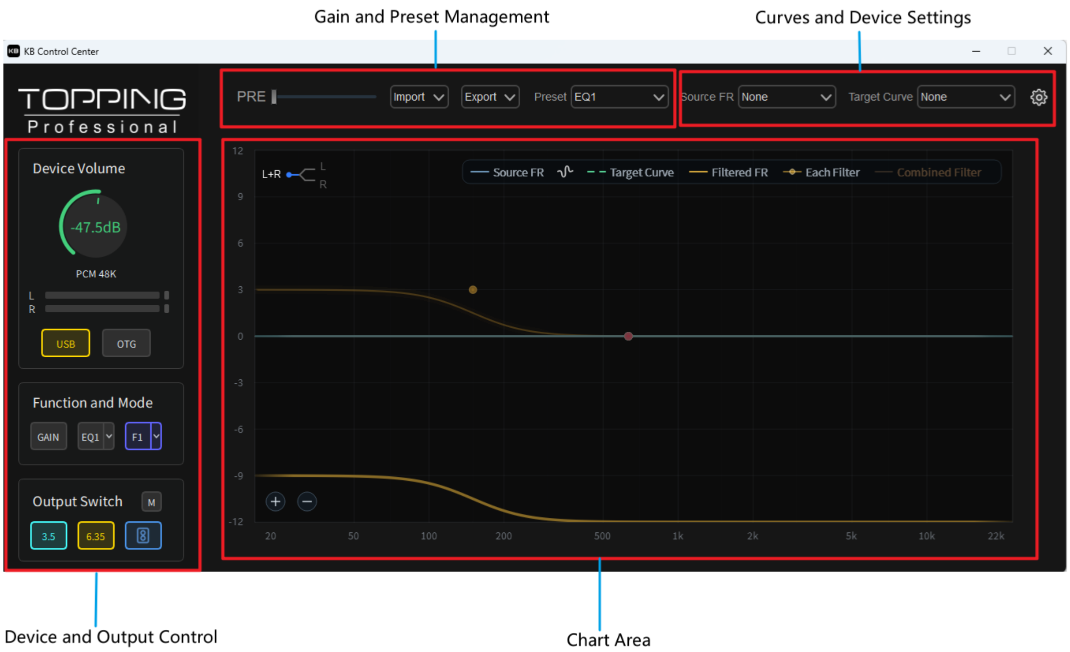
The gaming mode can be set to 44.1 or 48kHz, and the professional recording mode can be set to 44.1, 48, 88.2, 96, 176.4 or 192kHz. The higher the sampling rate, the higher the fidelity of the recorded audio, but at the same time, audio recorded at a high sampling rate requires more storage space.

Buffer size (Windows OS only)

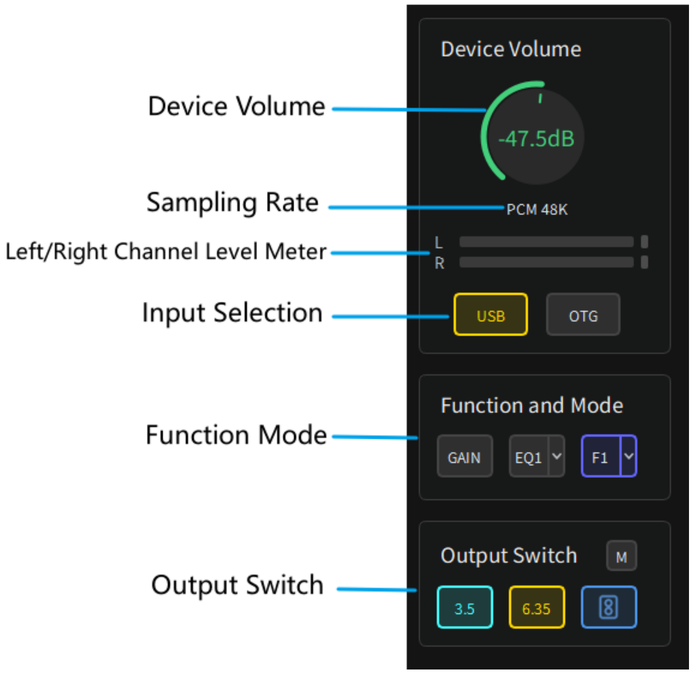
The smaller the buffer, the smaller the delay, but the higher the requirements on computer performance. When you experience playback stuttering/crackling, try increasing the buffer size.

6. HIFI mode

Overview



6.1 Device and Output Control



Device Volume: Adjusts the current device output volume.

Sampling Rate: Displays the current audio sampling rate.

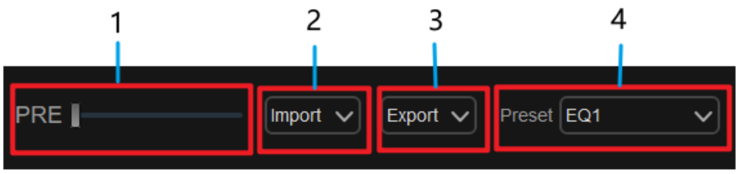
Left and Right Channel Level Meter: Displays the left and right channel levels.

Input Selection: Switches between USB and OTG inputs.

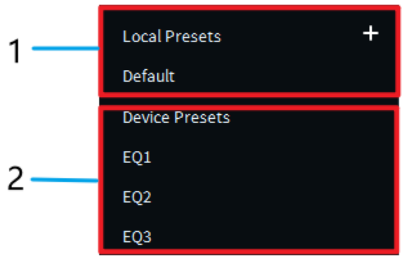
Function Mode: Selects the headphone amplifier gain, PEQ, and filter.

Output Switch: Selects 3.5 mm, 6.35 mm, line output, or mute.

6.2 Gain and Preset Management

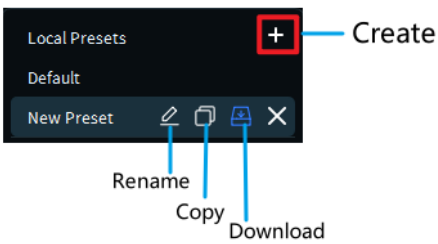


1.Preamplifier Gain
Refers to the overall gain or attenuation. Adjustment applies to all frequencies, and the adjustment range is ±12dB.
2.Import
Click "Import" to import the source frequency response curve, target curve, and filter parameters to both the local system and the device.
3.Export
Click "Export" to export the frequency response curve and filter parameters to your local storage and the device.
4. Manage local and preset PEQ

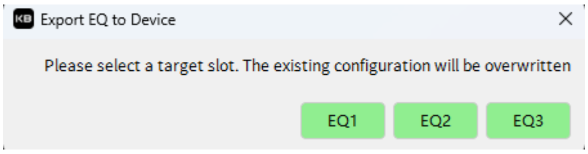


1. Local configuration
This function is suitable for users with different listening habits, music types and usage scenarios. You can configure different frequency response curves according to your own needs. When switching scenes, you can quickly select different configurations without having to reconfigure them one by one.

Create local configuration

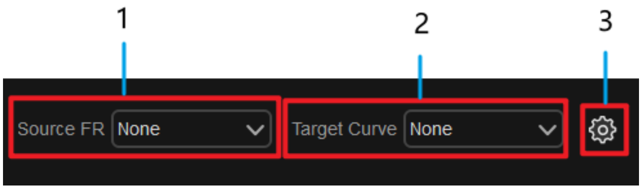


Click the + sign to the right of the local configuration to create it successfully.
Click the X to delete the local configuration, click the copy icon to copy the local configuration, and click the download icon to choose the local configuration to overwrite the original configuration.



2.Device configuration
This module is used to display device configurations that are available to the device when offline. Three configurations are provided by default and cannot be deleted.

6.3 Curves and Equipment Settings



1. Source frequency response curve file

You can select your desired source frequency response curve file.

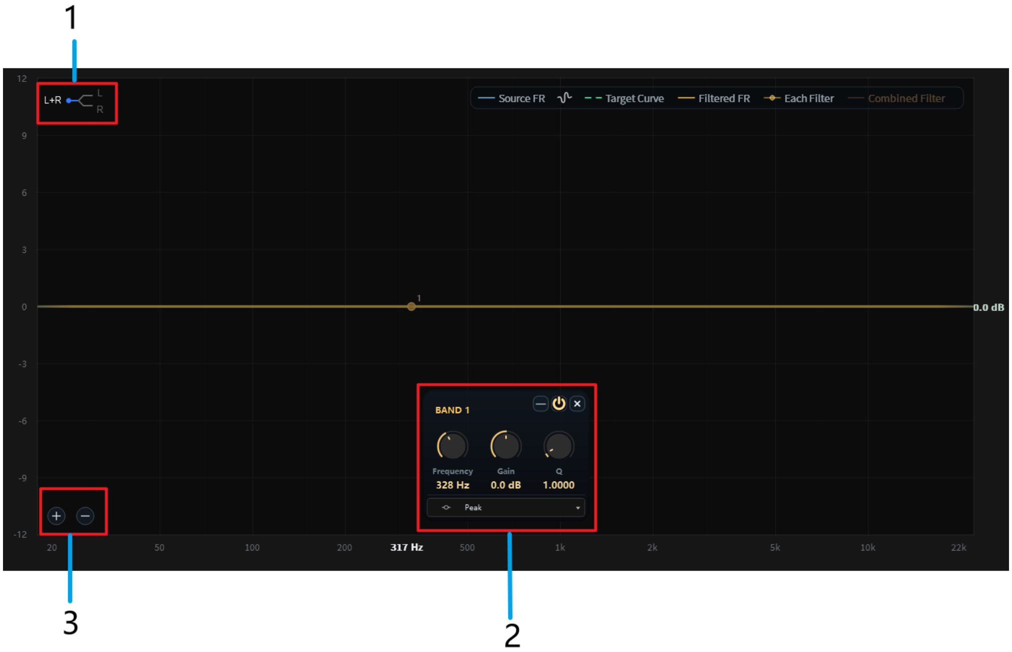
2.Target curve file

You can select the target curve file you want. For example, Harman OE is the Harman over-ear target curve.

3.Set up

System settings: For detailed instructions, please refer to the "4.2 System Settings" chapter.
Device settings: For specific settings, please refer to the "6. Menu Settings" chapter in the KB22 manual.
about: Please refer to the "4.5 About" chapter for detailed instructions.

6.4 Chart Area



1. **PEQ mode**
   L+R: Set the PEQ parameters of the left and right channels uniformly.
   L/R: Click L/R to set the PEQ parameters of the left/right channels respectively.
2. **PEQ setting area**
   Drag the operating point to adjust the frequency/gain/Q value; right-click to switch filter types or delete frequency bands, double-click in the blank space to create a new frequency band, and click to view detailed frequency band parameters.
3. **Zoom button**
   Adjust the viewing area of the chart.

7. Game mode

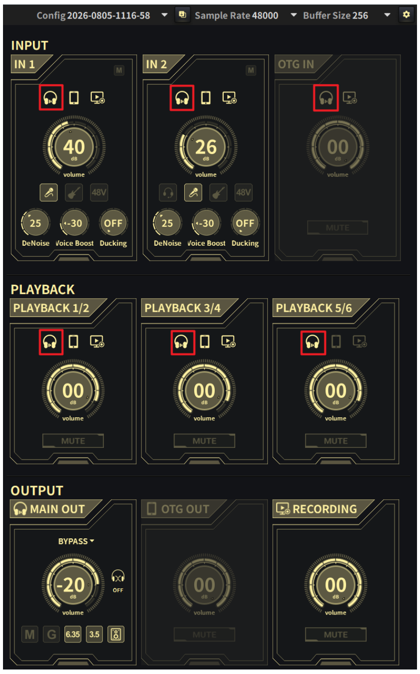
7.1 Signal routing

Users can choose different signal sources to output to headphones, OTG output or RECORDING according to their needs.

Users can select the target output channel for each signal in the input area or playback area. Click in the corresponding signal row **Headphone icon, OTG output icon or Recording icon**, the signal can be sent to the corresponding channel.

- Input signal: IN1/IN2, OTG IN
- Playback signal: Playback 1/2, Playback 3/4, Playback 5/6
- Target channels: Headphone amplifier output, OTG output, Recording channel
  For example: the processed IN1/IN2 input signal, OTG IN, and playback signal: Playback

1/2, Playback 3/4, Playback 5/6
Send to the headphone amp output. As shown below:

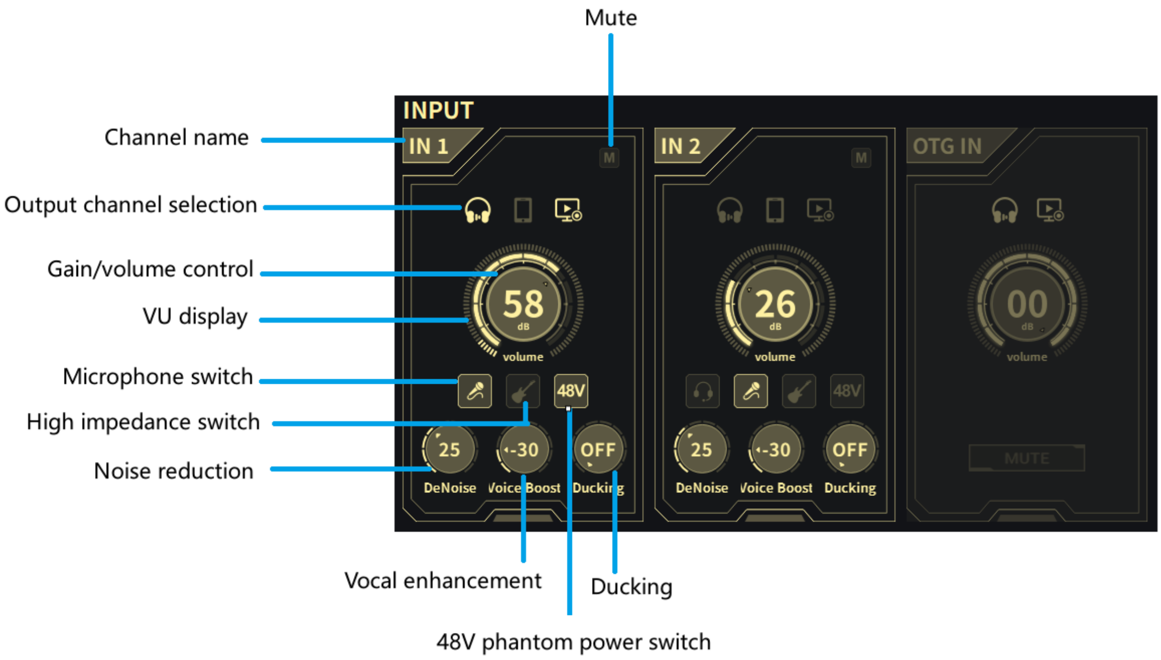


The supported audio routing distribution relationships are as shown in the table below. Users can route different signals to HP, OTG OUT or Recording according to actual needs. To avoid audio loops, OTG IN does not support routing to OTG OUT.

	HP	**OTG OUT**	**Recording**
IN1 after effects processing	✓	✓	✓
IN2 after effects processing	✓	✓	✓
OTG IN	✓		✓
Playback 1/2	✓	✓	✓
Playback 3/4	✓	✓	✓
Playback 5/6	✓	✓	✓

7.2 Input

The hardware input of the device can be configured.



Channel name
Displays the name of the input channel.

M

Mute this channel when the M button is lit. Light the M button again or adjust the volume to cancel mute.

Output channel selection

You can select the signal channel you want to output to the headphone jack (HP), OTG jack (OTG OUT) or RECORDING.

Gain/volume control

Adjust the gain/volume of the corresponding channel.

VU display
Used to display the audio signal level of the current channel.

Microphone switch
For microphones or line-level inputs, turn this button on.

High impedance switch
For high impedance inputs, such as musical instruments, turn this button on.

48V phantom power switch

Click this button to turn on or off the phantom power supply of the corresponding input interface.

IN1 / IN2: When 48V is turned on, 48V phantom power will be provided to the connected microphone.

Notes:

- If the connected device does not require phantom power (48V), turning on phantom power may cause damage to the device. Please only enable this feature if you are sure that your microphone requires corresponding power.
- Before unplugging or unplugging the microphone, please turn the device volume to minimum and turn off phantom power to avoid device damage or noise.

Noise reduction

Attenuate irrelevant noise in the environment (such as background noise, environmental interference sounds), retain the human voice captured by the microphone, and improve the clarity of the sound。

The higher the Level, the stronger the noise reduction effect and the greater the sound quality loss to the original signal.

Vocal enhancement

Enhance the input vocal signal to make the vocal clearer and more prominent.

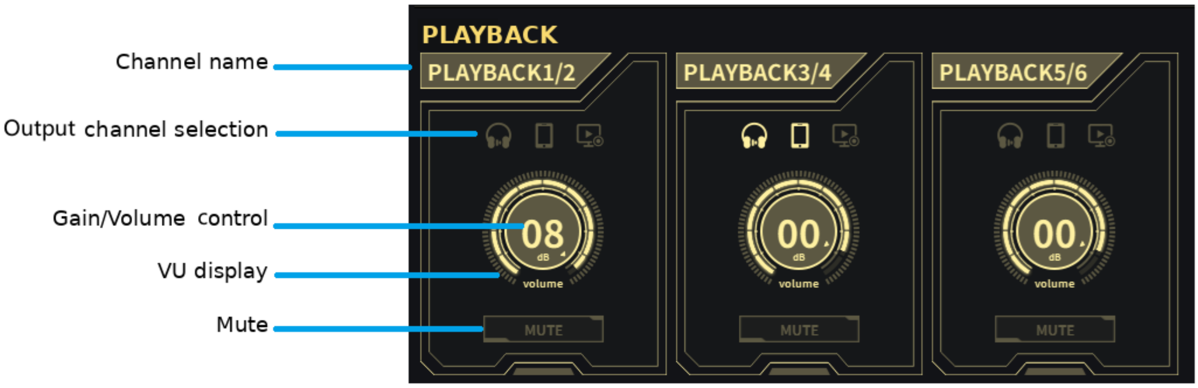
- The higher the value： the stronger the vocal enhancement effect.: the stronger the vocal enhancement effect.
- The lower the value: the weaker the vocal enhancement effect.

Ducking

This function can automatically reduce the volume of other sound sources (such as background music, system audio, etc.) when it detects an input signal from the microphone (IN1/2), thereby ensuring that the signal is continuously audible and not masked by other sound sources. It is often used in live chat and other scenarios. Click to turn on or off the DUCKING function.

- Turn the knob up: background sounds are muffled more and vocals stand out more.
- Turn the knob down: the background sound is suppressed less, and the overall listening experience is more natural.

7.3 Playback



Channel name

Display the name of the playback channel.

Output channel selection

You can select the signal channel you want to output to the headphone jack (HP), OTG jack (OTG OUT) or RECORDING.

Gain/volume control

Adjust the gain/volume of the corresponding channel.

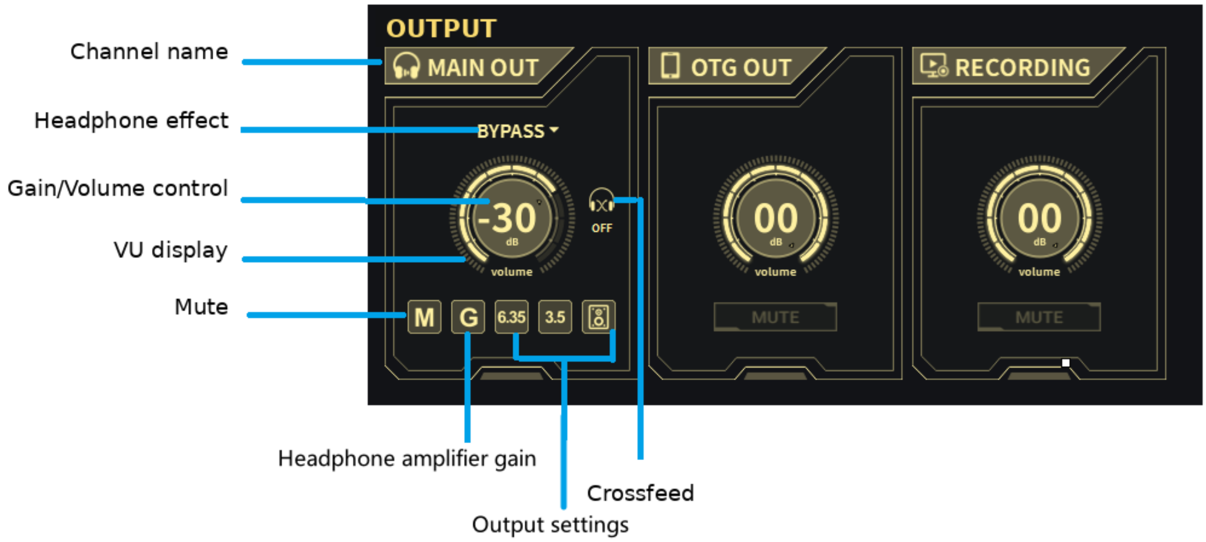
VU display

Used to display the audio signal level of the current channel.

Mute

Mute this channel when the light is on. Click the MUTE button again or adjust the volume to unmute.

7.4 Output



Channel name

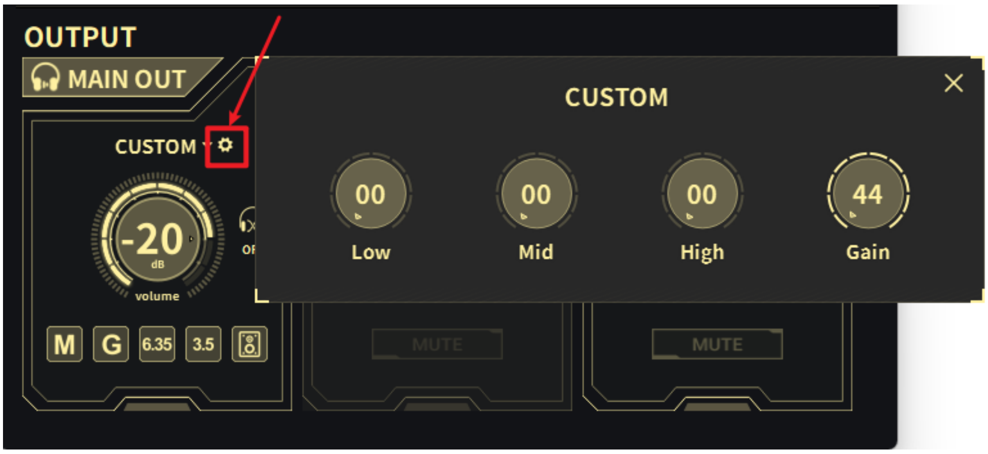
Displays the name of the output channel.

Crossfeed

Headphone amp effects can be selected according to different needs

Mode weak (LOW), mode (MID), mode strong (HIGH), mode customization (CUSTOM), off (BYPASS)

Click the settings icon shown in the image on the right to open the CUSTOM settings. Adjust Low Frequencies, Mid Frequencies, High Frequencies, and Enhancement to suit the game's audio characteristics, your headphones' performance, and your listening preferences. These adjustments can make key sounds, such as footsteps and gunfire, easier to distinguish, helping you determine enemy positions and movements and make timely tactical decisions.



- Low Frequencies: Adjusts the low-frequency level of sounds such as footsteps. Increase this setting slightly when complex ambient sounds mask footstep details.
- Mid Frequencies: Adjusts the mid-frequency range of sounds such as footsteps and reloading, helping you identify nearby actions and track enemy movement.
- High Frequencies: Adjusts the high-frequency detail of gunfire.
- Enhancement: Adjusts the amount of audio compression. Higher settings reduce the difference between loud and quiet sounds, making subtle sounds such as footsteps easier to notice. Lower settings preserve more of the dynamic range, helping you judge distance based on differences in volume.

Gain/volume control

Adjust the gain/volume of the corresponding channel.

VU display

Used to display the audio signal level of the current channel.

Mute

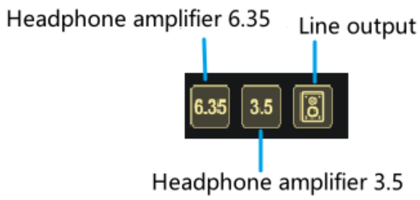
Mute this channel when the light is on. Click the MUTE button again or adjust the volume to unmute.

Headphone gain

When the light is off, it is low gain; when the light is on, it is high gain.

Output settings

There is output when it is on, and it is silent when it is off.



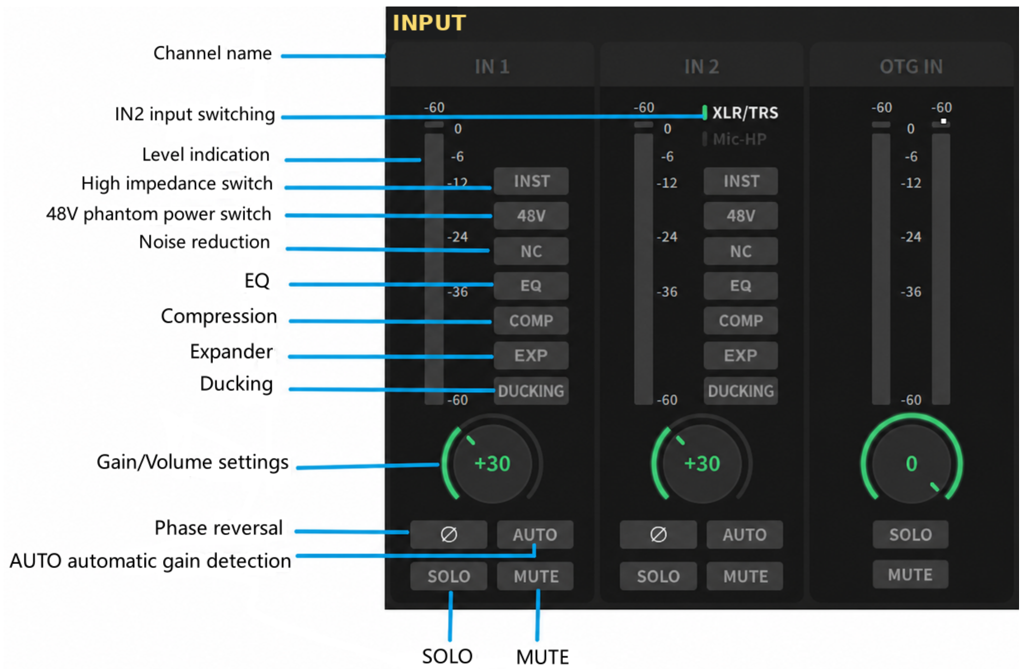
mutual feedback effect

Only effective for headphone amps, you can choose STUDIO, ROOM, or OFF.

8. Professional mode

8.1 Input

Device hardware inputs can be configured.



Channel name

Displays the name of the input channel.

IN2 input switching

IN2 selectable XLR/TRS or Mic-HP.

Mic-HP: Provides a 2.5V bias voltage when headset microphone input is enabled.

Level indication

Displays the current signal level (in dBFS). When it exceeds 0dBFS, the clipping indicator light at the top lights up. The mono input displays 1 level bar, and the stereo input displays 2 level bars, corresponding to the left and right channels respectively.

High impedance switch

For high impedance inputs, such as musical instruments, turn this button on.

48V phantom power switch

Click this button to turn on or off the phantom power supply of the corresponding input interface.

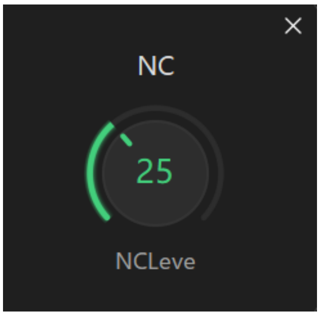
- **IN1 / IN2**: When 48V is turned on, 48V phantom power will be provided to the connected microphone.

Notes:

- If the connected device does not require phantom power (48V), turning on phantom power may cause damage to the device. Please only enable this feature if you are sure that your microphone requires corresponding power.
- Before unplugging or unplugging the microphone, please turn the device volume to minimum and turn off phantom power to avoid device damage or noise.

Noise reduction (44.1-48kHz)

Adjust the intensity of noise reduction as needed for the best experience.



Noise reduction effect processor processing capacity. The higher the Level, the stronger the noise reduction effect and the greater the sound quality loss to the original signal.

EQ（44.1-48kHz）

The microphone input offers 4-band PEQ adjustment, allowing the gain, frequency and Q of each band to be adjusted for a clean, clear, professional base dry sound. The adjusted effects can be recorded to the computer.

Click the EQ button to turn on or off the EQ function; double-click or right-click to enter the EQ setting page, as follows:

Switch button: Enable or disable this EQ function.

enter: Real-time display of microphone input level.

Curve display area: Displays the frequency response curve of the audio signal, allowing you to visually view and adjust the gain of the frequency. The orange curve in the figure represents the current frequency adjustment status. The vertical axis represents gain (dB) and the horizontal axis represents frequency (Hz), reflecting the change in signal strength at each frequency point.

EQ settings

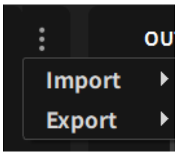
The 4 columns represent the PEQ settings of the four frequency bands, and each frequency band can be set

- **type:**Peak, low pass, high pass, low shelf and high shelf.
- **Gain**: Adjust the gain or attenuation of a specific frequency band and control the loudness of the frequency.
- **frequency**: The target frequency point you want to adjust, or the center point of the range, determines where in the spectrum the EQ processing will occur.
- **Q value**: Adjust the bandwidth of the frequency band to determine the frequency range of the frequency band. The lower the Q value, the wider the frequency range of the influence and the gentler the change; the higher the Q value, the narrower the frequency range of the influence and the sharper the change.

Output: The microphone output level after EQ adjustment is displayed in real time, and the volume of the output signal can be adjusted through the fader on the right to ensure that the final output audio volume is appropriate.

Import and export functions

The import and export function is used to quickly reuse equalizer parameter configurations, supports the migration of adjusted filter settings between different audio processing scenarios and devices, improves audio processing efficiency, and facilitates parameter backup and sharing.

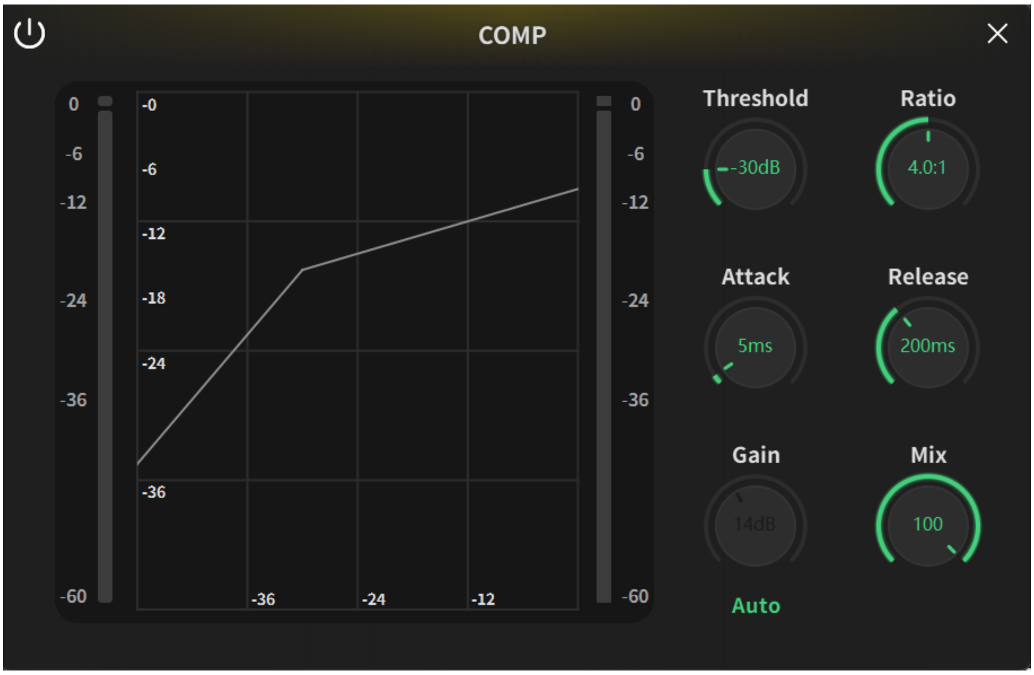


Click the three-dot icon on the interface to expand the submenu:

- Import filter: Import the saved equalizer filter parameter configuration file (including gain, frequency, Q value, etc. settings), and reuse the previously debugged frequency band adjustment scheme with one click, without having to manually set parameters again.
- Export filter: Save 1 - 4 groups of filter parameters (type, gain, frequency, Q value and enabled status) in the current equalizer as files, so that they can be easily imported and used in the equalizers of other audio projects and devices, and the parameters can be reused across scenarios.

Compression (44.1-48kHz)

The core function of a compressor is to control the dynamic range of the sound, making the volume fluctuations smoother and the loudness more uniform, while retaining the strength and texture of the sound. The following is the function description of each parameter:



Threshold

The "volume threshold" that triggers compression. When the level of the input sound exceeds the threshold, the compressor is activated to "reduce" the excessive volume; sounds below the threshold remain in their original state and are not compressed.

Ratio

Represents how "compressed" sounds are that exceed the threshold. For example: if the input sound exceeds the threshold by 4dB, after 4:1 compression, the output sound will only increase by 1dB. The larger the ratio, the stronger the compression.

Attack

How long it takes for the compressor to achieve full compression once the signal exceeds the threshold. A fast attack time (such as 1ms) will cause the compressor to start compressing the audio signal almost immediately, while a slower attack time (such as 50ms or 100ms) will cause the compressor to delay compressing for a while. This setting is critical and affects the transient feel of the audio (such as the punch of a drum beat).

Release

Release time determines how soon the compressor stops compressing after the signal drops below the threshold. A fast release time (such as 20ms) will cause the compressor to stop compressing quickly, which is suitable for fast dynamic changes; while a slower release time (such as 500ms or longer) will cause the compressor to stop compressing slowly, which is suitable for smoother signals. The release time setting affects the natural feel of the audio.

Gain

After compression, the overall volume of the audio signal is restored by increasing the compensation gain to maintain the overall balance.

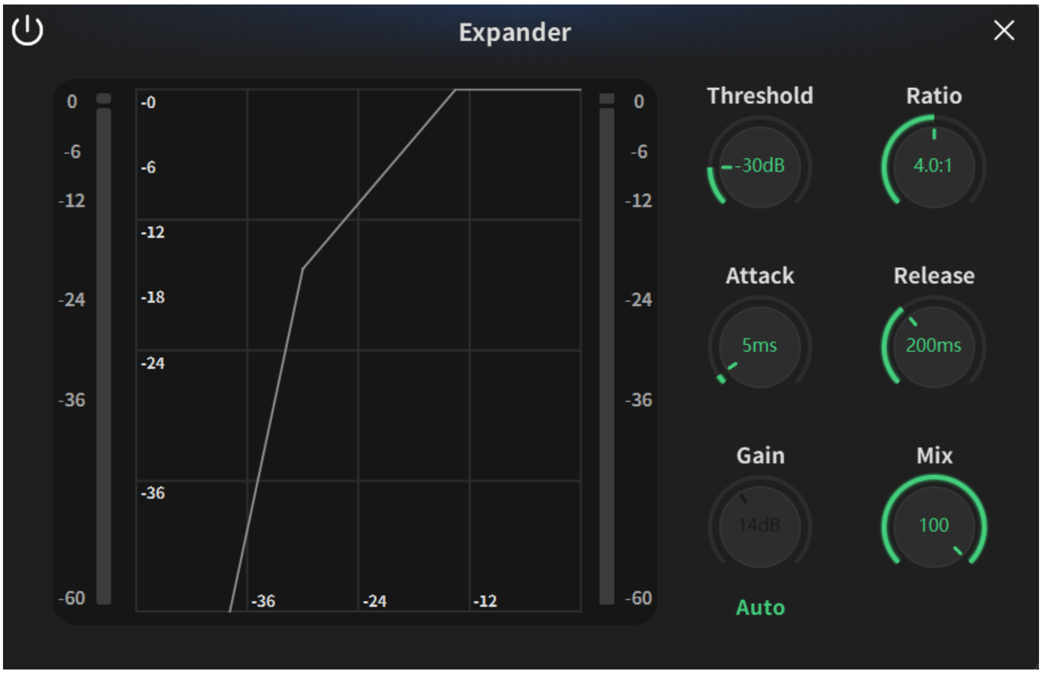
Click "Auto" below to turn on automatic gain compensation. The compressor automatically calculates and applies appropriate gain based on the current threshold, compression ratio and other parameters to compensate for the "overall volume loss caused by compressing large volumes."

Mix

Adjust the mixing ratio of "original dry sound" and "compressed wet sound".

Expander (44.1-48kHz)

The expander plug-in is used to automatically reduce the volume of the signal when it falls below a set threshold, making background noise, ambient sounds and weak interference quieter, thus improving the clarity and dynamic contrast of the sound. The following is the function description of each parameter:



Threshold

The threshold is used to set the level limit at which the expander begins to operate. When the input signal is below this threshold, the expander begins to reduce the signal volume; when the input signal is above this threshold, the signal usually remains the same or is less affected.

Expansion ratio

The expansion ratio determines how much signals below the threshold are reduced. The larger the expansion ratio, the more obviously low-level signals are suppressed.

Attack

Attack time determines how quickly the expander starts reducing the volume once the signal falls below the threshold.

Release

The release time determines how soon the expander returns to a state of no or less processing when the signal returns above the threshold.

Gain

Adjust the overall output volume of the processed signal. Increasing gain increases overall loudness, while decreasing gain avoids overloading or clipping the output. Pay attention to the output level when using it to avoid distortion caused by excessive gain.

Click "Auto" below to turn on automatic gain compensation. The expander will automatically calculate and apply appropriate output gain based on the current threshold, expansion ratio and other parameters to compensate for the overall loudness change caused by the attenuation of low-level signals, so that the volume sense before and after processing remains relatively consistent.

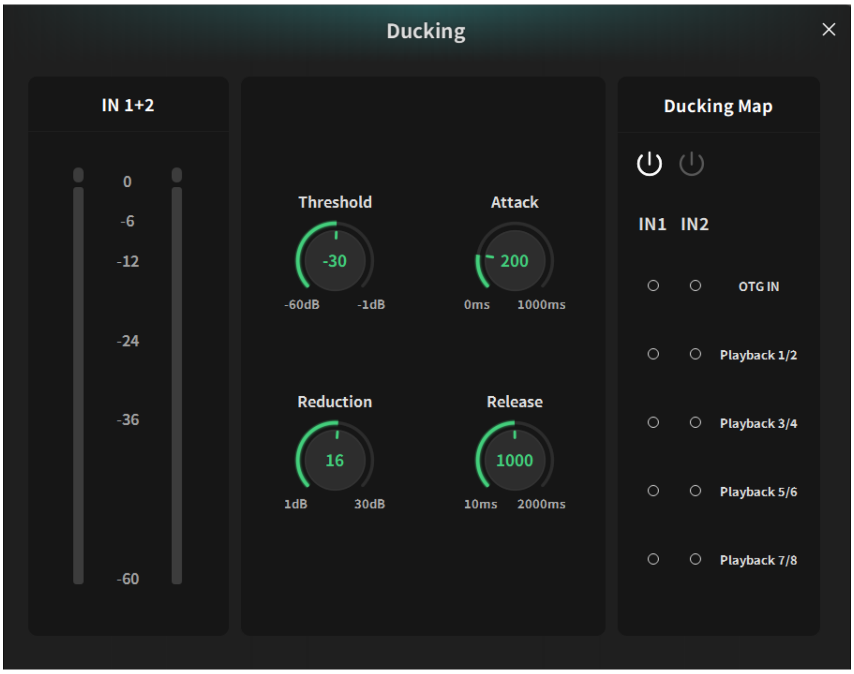
Mix

Adjust the mixing ratio of the original signal and the expanded signal.

Ducking (44.1-48kHz)

This function can automatically reduce the volume of other sound sources (such as background music, system audio, etc.) when it detects an input signal from the microphone (IN1/2), thereby ensuring that the signal is continuously audible and not masked by other sound sources. It is often used in live chat and other scenarios.

Click to turn on or off the DUCKING function; double-click or right-click to enter the DUCKING setting page, as follows:



Switch button: Enable or disable ducking functionality.

IN1+2： Real-time display of microphone input level to determine whether trigger conditions are met.

Threshold: When the microphone input level exceeds the ducking threshold, the ducking function is triggered and automatically starts to suppress other sound sources.

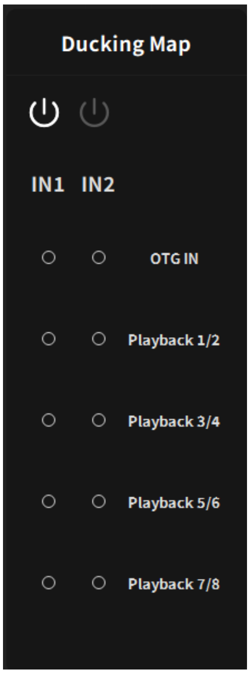
Attack: When the input level exceeds the ducking threshold, the time required for the process from initiating ducking to fully taking effect. The smaller the value, the faster the response.

Reduction: The level attenuation of the suppressed bass source. The larger the value, the more obvious the suppression.

Release: When the input level is lower than the ducking threshold, the time required for the process from turning off ducking to completely ineffective ducking. The larger the value, the smoother the recovery.

Ducking Map: You can set which channels automatically reduce the volume after IN1/IN2 reaches the ducking trigger condition. As shown in the figure, when it is detected that the

IN1 channel reaches the ducking trigger condition, the system will automatically reduce the volume of the OTG input.



Gain/Volume settings

Adjust the gain/volume of the corresponding channel.

Phase reversal

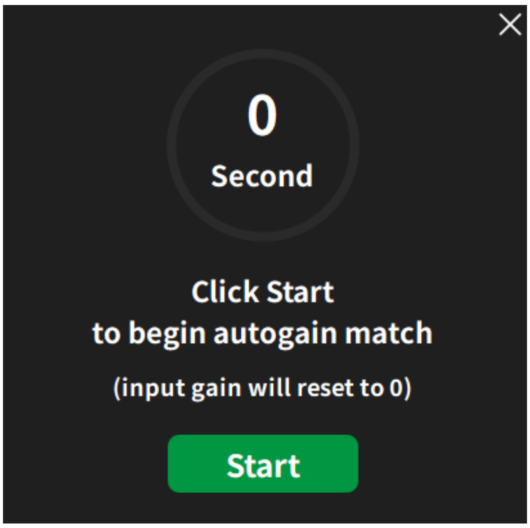
Inverts the polarity of the signal 180° when the light is on.

AUTO automatic gain detection

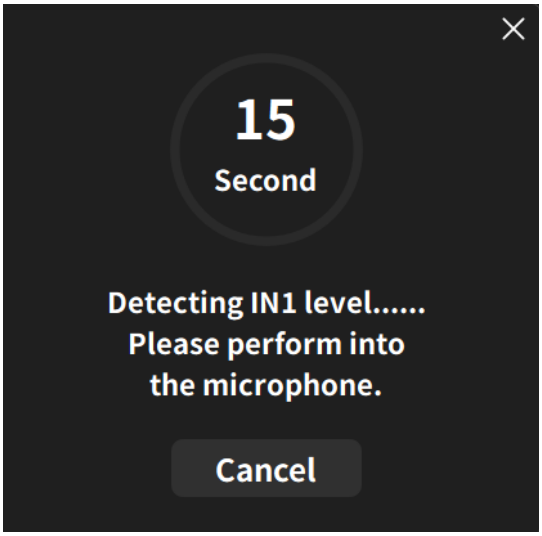
This function can intelligently adjust the gain level of the microphone according to the input signal strength.

Make sure the microphone has been connected to KB22 and the IN1 selection and 48V switch settings are correct. If you don't know, you can first check the connecting microphone in 4.2 of the KB22 manual.

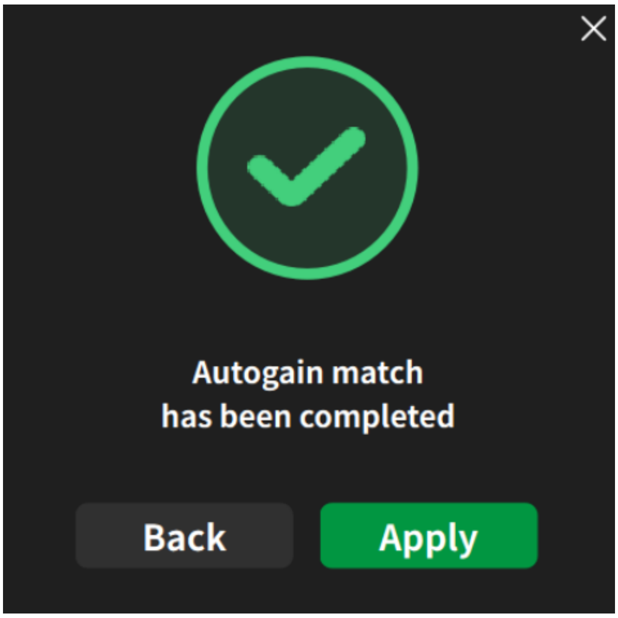
Click AUTO to enter the automatic gain detection process

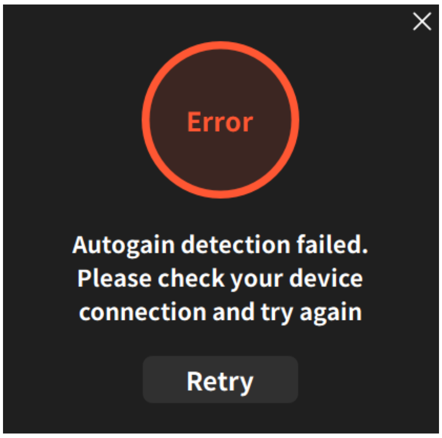


It is recommended to simulate actual usage situations (such as recording and live broadcasting) during testing, and speak/sing at a normal volume and a normal distance from the device to ensure more accurate gain matching.



If the detection process is normal, the program will provide the appropriate gain for your microphone to achieve the ideal recording level for the current state. If the detection process is abnormal, you will be prompted to try again.





SOLO

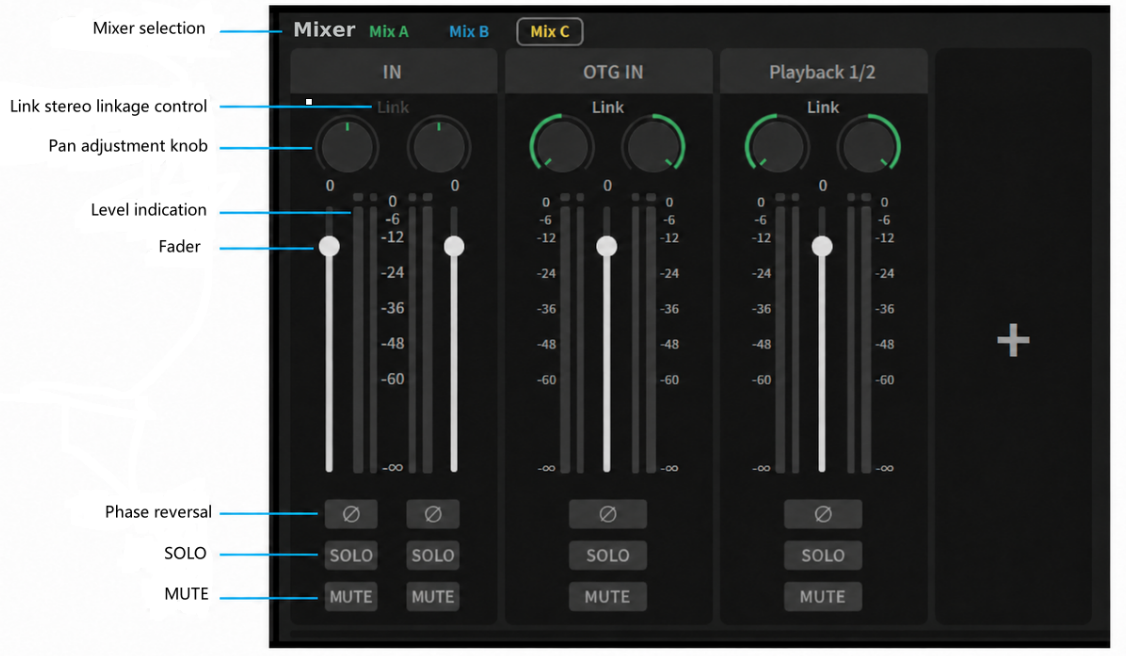
Will mute all channels except the channel currently SOLOing. Multiple channels can be SOLOed at the same time.

MUTE

Mute this channel when the light is on. Click the MUTE button again or adjust the volume to unmute.

8.2 Mixer

The mixer supports mixing multiple audio input signals (including hardware input signals and computer audio signals) into one output.



Mixer selection

Three different mixers are available. The three mixers share the same input, but different mixers can have different settings.

Link stereo linkage control

Lighting this button will set the two adjacent input channels to one stereo channel to achieve linkage control. For example, one fader can control the signal levels of two adjacent inputs at the same time without having to push two faders. In addition, after turning on Link, the pan knobs of the two channels will automatically be set to extreme left and extreme right respectively.

Pan pan adjustment knob

Used to adjust the left and right distribution of sound sources. For example, when it is turned to the extreme left, it means that all the signals of this channel are transmitted to the left channel output. Similarly, when it is turned to the extreme right, it means that all the signals of this channel are transmitted to the right channel output. When it is in the middle position, it means that the signals of this channel are outputted by both the left and right channels.

Level indication

Displays the current signal level (in dBFS). When it exceeds 0dBFS, the clipping indicator light at the top lights up.

The left and right are the level indicators of the two channels respectively. One channel has two level indicators, the outside is the input level indicator and the inside is the output level indicator.

Putter

The fader controls the level that the corresponding channel sends to the selected mixer.

Phase reversal

Inverts the polarity of the signal 180° when the light is on.

SOLO

Will mute all channels except the channel currently SOLOing. Multiple channels can be SOLOed at the same time.

MUTE

Mute this channel when the light is on.

Delete channel

There are many channels available in the mixing area. In order to facilitate operation and simplify the interface, users can freely add or delete mixing channels according to actual needs.

When the mouse is hovering over the channel name, a delete icon will appear in the upper right corner. Click this icon to delete the channel.

Add channel

There are many channels available in the mixing area. In order to facilitate operation and simplify the interface, users can freely add or delete mixing channels according to actual needs.

Click the "+" on the right to choose to add the desired channel.

8.3 Effect area

Delay and reverb effects can be added to the KB22's hardware input signals.

The 2 faders on the left are used to adjust the level of the input track sent to the effects processing section (FX). After all input signals enter the effects area, they will be mixed and output as dual mono signals, and the processed signals are named "FX".

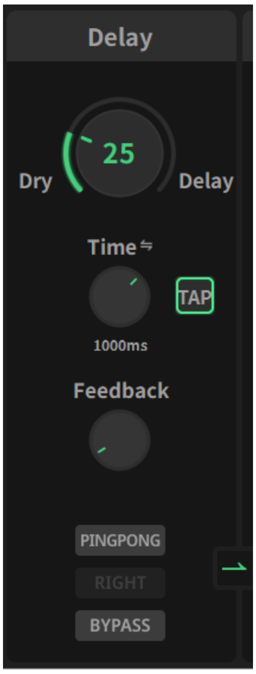
- **effects input**: Displays the total level of the 2 input tracks before entering effect processing.
- **Effect output**: Displays the total level of the FX area output after processing.
- **MUTE FX**: Mute the FX channel.

Effect processing:

1. Delay (44.1-48kHz)

Delay plug-ins are used to repeatedly play the input signal at set time intervals to produce echo, spatial or rhythmic repetition effects. By adjusting the dry to delay ratio, delay time, and feedback amount, you can control how much the original sound is mixed with the delayed sound, how long the echo appears, and how many times it is repeated.

The following are the functions and operation instructions of this delay effect:



Parameter adjustment area

Dry/Delay Balance Knob

Adjust the mix between the original and delayed signals. Dry represents the original, unprocessed sound, and Delay represents the repetitive sound produced after processing with effects.

- Small value: more original sound, weaker delay/echo effect, more natural and forward sound.
- Large values: more delayed sounds, more obvious echo effects, and a stronger sense of space and repetition.

TAP

Used to quickly set the delay time by clicking. When clicking the TAP button continuously, the system will calculate the delay time based on the click interval.
In BPM mode, TAP can be used to set the sync speed so that the delay effect matches the click rhythm; in time mode, TAP will convert the click interval directly into a millisecond value and apply it to the delay time parameter.

feedback

Sets the proportion of the delayed signal returned to the delay processor, which is used to control the number of echo repetitions.

- The higher the feedback value, the more often and longer the echo will repeat.
- The lower the feedback value, the fewer echo repetitions and faster decay.

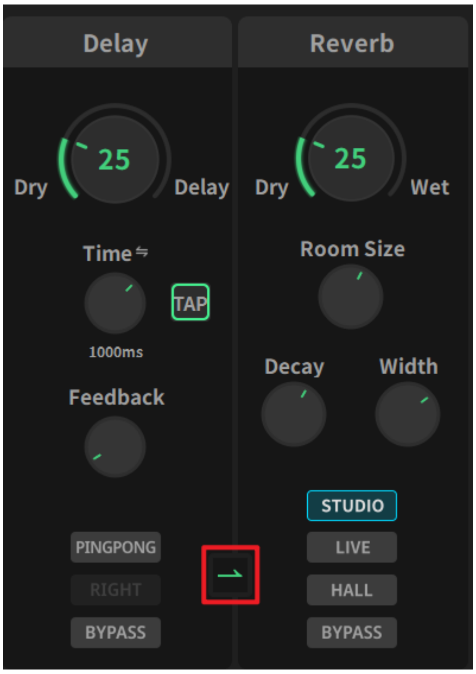
Excessive feedback may cause echoes to continue to stack up, or even cause self-excitation or overload. Please adjust appropriately according to actual needs.

Delay mode

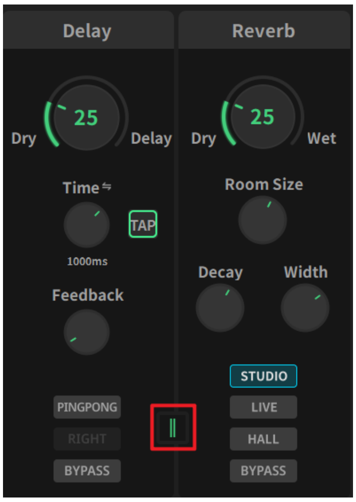
Used to select how delayed sounds are played in the left and right channels. Different modes will change the sound image position and spatial feeling of the delayed signal.

- **PINGPONG**: Ping pong delay mode, the delayed sound will alternately repeat between the left and right channels, producing a stereo echo effect that moves back and forth from left to right.
- **RIGHT**: Right start mode. exist**PINGPONG** When on, the first delay starts on the right channel and then repeats alternately between the left and right channels.
- **LEFT** : Left start mode. exist**PINGPONG** When on, the first delay starts on the left channel and then repeats alternately between the left and right channels.

Sequential processing: Delay is processed first and then reverb is processed in the order of the signal chain; therefore, the signal generated by the delay will also enter the reverb.



andRow processing: Delay and reverberation act on the original signal at the same time, and they are independent of each other and do not affect each other.



Adjustment suggestions

1. First general **Dry to delay ratio** Turn it down to ensure the original sound is clear.
2. Set a suitable time, or use **TAP** Click along with the beat to set the delay time.
3. slowly increase **feedback**, until the number of echoes meets the needs.
4. Choose according to the desired spatial effect **PINGPONG** mode and set it after turning it on **RIGHT/LEFT** 。
5. Final fine-tuning**Dry to wet ratio**, to prevent the delayed sound from overpowering the original sound.

2. Reverb (44.1-48kHz)

Simulate the reflection and attenuation process of sound in different acoustic spaces, add natural reflection to "dry" sound, and create a sense of space and depth. The following are the functions and operation instructions of this reverb effector:

Parameter adjustment area

Dry/Reverb balance knob

Dry sound: refers to the original sound without added reverberation effect; reverberation: refers to the sound with added spatial reflection effect. Adjust this knob to control the ratio of "dry" to "reverbed" sounds:

- The smaller the value, the higher the proportion of dry sound, and the sound will sound “dry” and “closer”.
- The larger the value, the higher the proportion of reverberation sound, and the sound will sound "wet" and "distant" (suitable for creating a sense of atmosphere, but too much can easily lead to muddy sound, so it needs to be balanced according to needs).

Space size

The physical size of the simulated space (such as a small room, a hall) has 6 levels from small to large.

- Turn the knob up: The sense of space is stronger, and the sound seems to be reverberating in a "large space" (such as a hall).
- Turn the knob smaller: The space feels more compact, and the sound seems to be reflected in a "small space" (such as a bedroom).

Attenuation

Controls the duration of reverberation (the time a sound goes from loud to dead).

- Turn the knob up: the reverberation time is longer and the sound "died" (like the long echo of a cathedral).
- Turn the knob down: the reverberation time is shorter and the sound "dies out quickly" (like a brief reflection in a small room).

Width

Adjusts the stereo width of the reverb (the diffuseness of the left and right channels).

- Turn the knob up: the reverb is more "open", filling the left and right channels, creating a wider sense of space.
- Turn the knob down: the reverb is more "focused", leaning towards the middle channel, and the sound is tighter.

Preset mode button

Just click the button to quickly apply the preset classic reverb scene, which is suitable for novices to get started quickly:

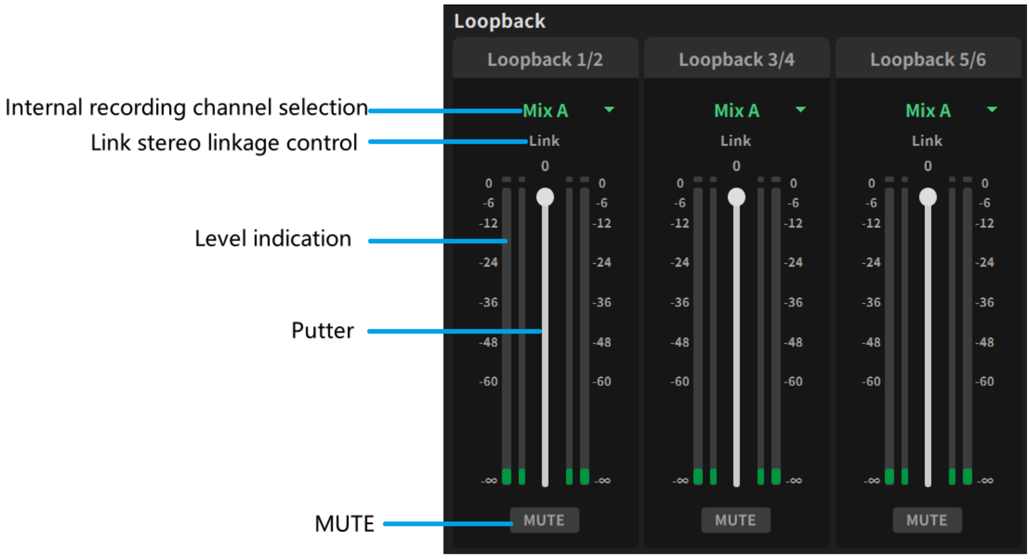
- **STUDIO**： Close room reverberation creates a specific small space.
- **LIVE**： Mid-range stage reverberation gives the sound a sense of presence and creates a medium-sized space.
- **HALL**： The long-distance hall reverberation gives the sound overwhelming momentum and creates a large space.
- **BYPASS**： After clicking, the input signal will bypass the effector (ie not be processed by the effector).

Adjustment suggestions

1. Start with a preset: click first**STUDIO/LIVE/HALL** Choose a rough scene, then fine-tune the knobs to optimize the details.
2. Adjust while listening: Keep the sound playing (such as playing music, talking) while adjusting the knob, and gradually adjust according to the listening experience to avoid excessive reverberation.

8.4 Internal recording area

Sends hardware input signals and computer audio signals (such as playback software or browser audio signals) back to the computer for transmission to a DAW for recording.



Internal recording channel selection

Select the signal that needs to be transmitted back to the computer.

Options:

·All mixes

·All hardware input signals (if you select mono input, the mono signal will be assigned to the left and right channels.)

·All computer playback signals

Link stereo linkage control

Lighting this button will set the two adjacent input channels to one stereo channel to achieve linkage control. For example, one fader can control the signal levels of two adjacent inputs at the same time without having to push two faders. In addition, after turning on Link, the pan knobs of the two channels will automatically be set to extreme left and extreme right respectively.

Level indication

Displays the current signal level (in dBFS). When it exceeds 0dBFS, the clipping indicator light at the top lights up.

The left and right are the level indicators of the two channels respectively. One channel has two level indicators, the outside is the input level indicator and the inside is the output level indicator.

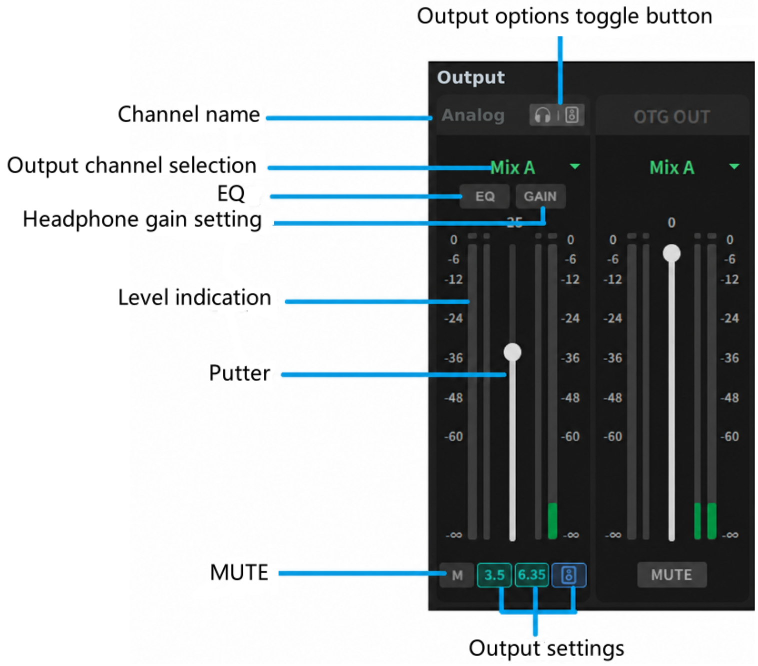
Putter

The fader controls the level that the corresponding channel sends to the selected mixer.

MUTE

Mute this channel when the light is on. Click the MUTE button again to unmute.

8.5 Output



Channel name

Displays the channel name of the output area.

Output channel selection

Select the signal channel you want to output to the headphone jack (HP) or OTG jack (OTG OUT).

Options:

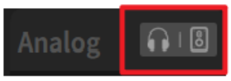
·All mixes

·All hardware input signals (if you select mono input, the mono signal will be assigned to the left and right channels.)

·All computer playback signals

Output options toggle button

Click this button to quickly switch between different output options. The available output options are: all, line, amp 6.35, amp 3.5, amp all, line + amp 6.35, line + amp 3.5.



EQ

The headphone output provides 10-band PEQ adjustment to adjust the sound you hear in your headphones without changing the original signal recorded to your computer. Used to create a comfortable, accurate and personalized monitoring environment.

Click the EQ button to turn on or off the EQ function; double-click or right-click to enter the EQ setting page.

Headphone gain setting

When the light is off, it is low gain; when the light is on, it is high gain.

Level indication

Displays the current signal level (in dBFS). When it exceeds 0dBFS, the clipping indicator light at the top lights up.

The left and right are the level indicators of the two channels respectively. One channel has two level indicators, the outside is the input level indicator and the inside is the output level indicator.

Putter

The fader controls the output level of the corresponding channel.

MUTE

Mute this channel when the light is on. Click the MUTE button again or adjust the volume to unmute.

Output settings

There is output when it is on, and it is silent when it is off.