DVDO

DVDO-DSP44-1

4x4 Digital Audio Processor

User Manual

Version v1.0

DVDO | +1.408.213.6680 | support@dvdo.com | www.dvdo.com

Contents

1. Product Overview

DVDO-DSP44-1 is a compact, high-performance digital signal processor with four balanced analog inputs and four balanced analog outputs.

It is designed for small to medium-size commercial audio installations requiring flexible routing and mixing as well as advanced signal optimization.

It features advanced audio technologies including Acoustic Feedback Canceler (AFC), Adaptive Echo Cancellation (AEC), Adaptive Noise Suppression (ANS), Automatic Gain Control (AGC), Active Noise Cancellation (ANC) & Auto Mixer (AM).

Its configurable DSP engine supports EQ, dynamics, filtering, delay, summing and full matrix routing, all controlled through an intuitive software interface or via RS232/RS485.

Additional features include voice tracking for PTZ cameras, a USB port for audio recording/playback & video conferencing, and front-panel status indicators.

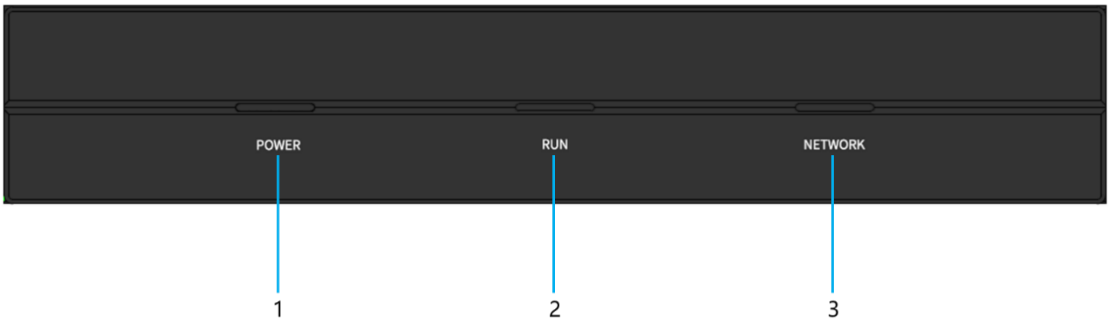
2. Product Features

- 4 balanced mic/line inputs
- 4 balanced line outputs
- 48V phantom power supply for MIC inputs
- AFC, adaptive suppression system noise
- AEC, adaptive eliminate the echo generated in the meeting
- AGC, adaptive automatic gain control, ensure sound system output volume is balanced
- ANS, dynamic adaptive noise reduction with signal level up to 18dB
- ANC, adaptive noise gain compensation, automatically adjust the output gain according to the change of ambient noise
- AM, multi-microphone intelligent mixing function, easy to manage multi-microphone combination output, with microphone selection function
- Support matrix input and output signal routing function, audio signal switching and allocation
- Excellent and rich signal processing module, which mainly includes compressor, evader, equalizer, filter, expander, noise gate, limiter, delay and other functions
- Built-in USB sound card, supporting music playback, recording and soft video conferencing (such as: ZOOM, Tencent conference, Nail conference, etc.)
- RS485/RS232 serial commands control
- The UI provides intuitive graphical operation interface, clear signal flow, diversified control panel, and flexible configuration of the audio processor

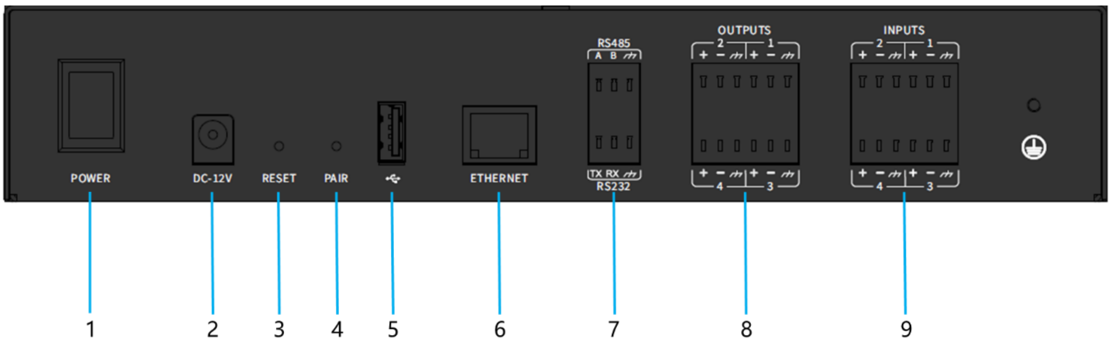
3. Packing List

1x DVDO-DSP44-1 4x4 digital audio processor
1x 12v power adapter
2x 3pin terminal blocks
4x 6pin terminal blocks
1x screwdriver

4. Panel Description



Serial number	Name	Instructions
1	POWER indicator	Power indicator, keep on after power on.
2	RUN indicator	Run indicator light, flashing slowly indicates normal operation.
3	NETWORK indicator light	The network Linkage indicator is steady on when the PC control terminal is Linkage through the network.



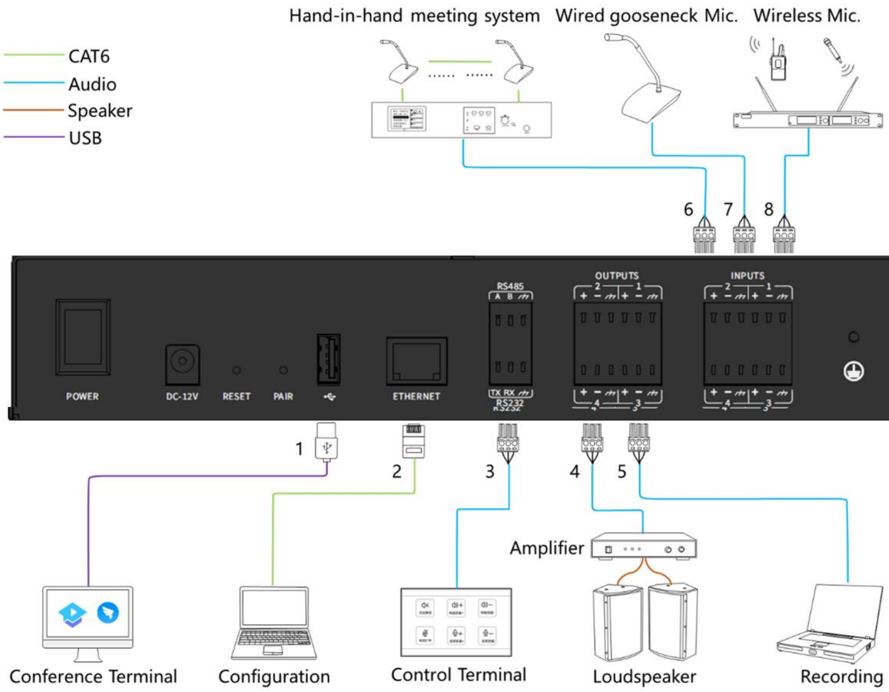
No	Name	Description
1	Power port	100-240V AC, 50/60Hz
2	Power switch	
3	RESET	Restore factory Settings Press and hold down the 3s device to restart and restore factory Settings
4	PAIR	Remote debugging pairing button
5	USB 2.0 Type-A	Supports two-way audio data transmission
6	ETHERNET	External configuration computer
7	RS232 + RS485	Serial control interface, external control terminal or central control device

8	ANALOG LINE OUTPUTS	Line output interface (1-4), can connect power amplifier, active speaker, recording and broadcasting server and other equipment
9	ANALOG MIC/LINE INPUTS	Microphone/line input interface (1-4) for connecting devices such as microphones, DVDs, and conference hosts

5. Technical Specifications

Parameter	Value
Number of analog input and output channels	4x4
Sampling rate	48K/24bit
Input gain	0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dB
Phantom power	+48V/10mA max
Frequency response(20~20kHz)	±0.5dB
Maximum level	+18dBu
THD+N	<-100dB @4dBu
Input dynamic range	110dB
Output dynamic range	112dB
Channel isolation@1kHz	108dB
Input impedance (balanced connection method)	5.4KΩ
Output impedance (balanced connection method)	600Ω
Working power supply	DC12V, 2A
Net weight	1.2kg
Dimensions (W x D x H)	250 x 160 x 44mm

6. System Diagram



Devices are connected through interfaces s on the rear panel:

1. USB 2.0 Type-A interface, supports bidirectional audio data transmission
2. External configuration computer with RJ45 interface
3. RS232 & RS485 serial commands control interface
4. Configured as the line output loudspeaker interface, external power amplifier or active speaker, used for local sound expansion output or playback of remote audio signals
5. Configured as the line output Mix interface to connect to external recording devices
6. Configured as a microphone interface to connect to the hand-in-hand conference host
7. Configured as microphone interface, can access the wired gooseneck microphone
8. Configured as a wireless microphone interface to connect to a wireless microphone host

7. Software Control

7.1. Software Download

Device Linkage and configuration

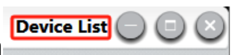
The default IP address of the processor is 168.182.102.36. The subnet Mask is 255.255.255.0. Make sure both the PC and the processor are on the same network segment of the same local area network. If the PC and processor are on different network segments, go to the PC Network Settings to modify the Network Parameter.

Software Download

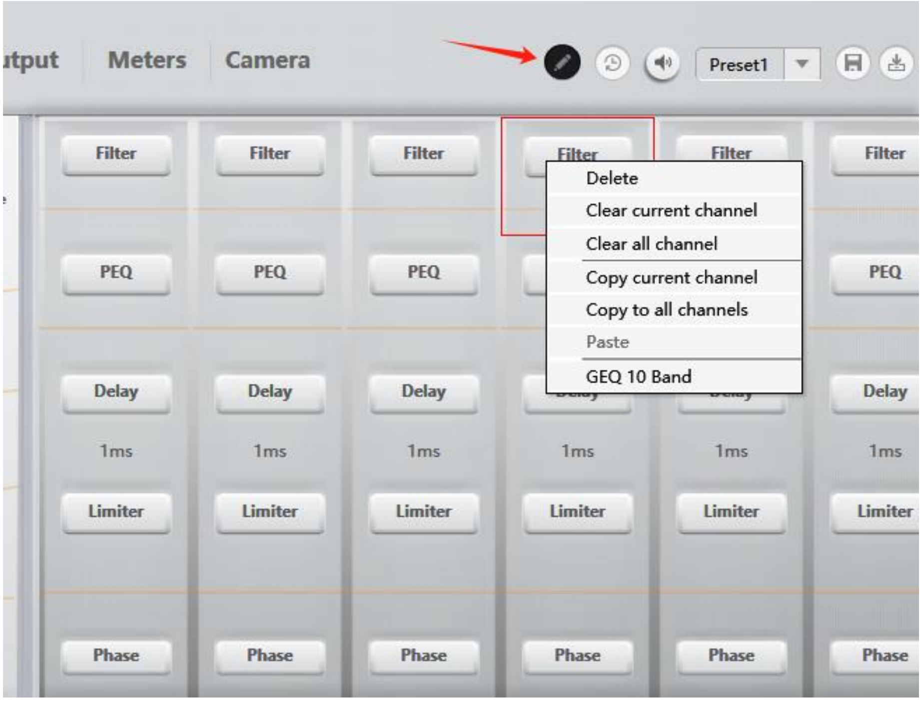
After the previous step, open the PC browser and Input “168.182.102.36” to open the software download page. To log in, Input username “admin”, password “123456” and click “Login”. Download the installer according to the prompts (some models need to choose to download PC software) and install it after the download is completed.

7.2. Software Operation

After opening the software, the following main screen is displayed:

Click the button  in the upper right corner of the main interface, all processors on the Network will be automatically searched. The user will Linkage to the specified processor as required, and the network indicator will light up after the connection.

Custom edit processing module



Click the button  in the upper right corner, right-click on the Input or Output channel processor module, the editing dialog box appears, you can replace the current processing module, delete, copy and other Output, click the editing button again you can exit the editing mode of the prime and you can choose whether to upload the proper parameter to the host.

When editing the module, note the change in the lower right corner 

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When it turns red 

, that is, when CPU1 is greater than or equal to 100, editing or uploading the device cannot be completed, an error message will be displayed.

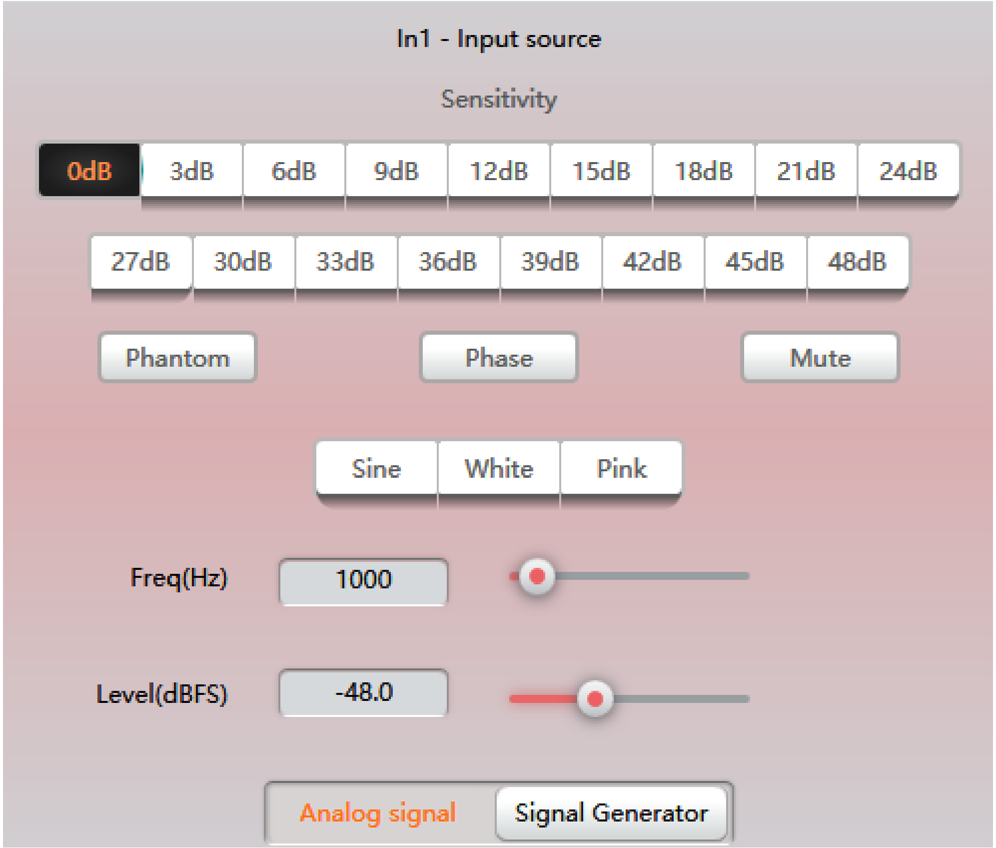
Other

Click  to restore the device to factory Settings; Click  to mute all Output Channel; Click  to save the current configuration; Click  to import the local configuration. Click 

to synchronize the two Channel gain controls.

7.3. Audio Module Parameters

7.3.1. Input Source



Sensitivity: the microphone gain, 0/3/6/9 12/15/18/21/24/27/30/33/36/39/42/45/48 is a, a total of 17 files are optional.

Phantom: External capacitive microphone feed, when needed to activate. Do not open the line with Input or no power supply to prevent damage to external devices.

Phase: The phase of the audio signal is reversed 180°.

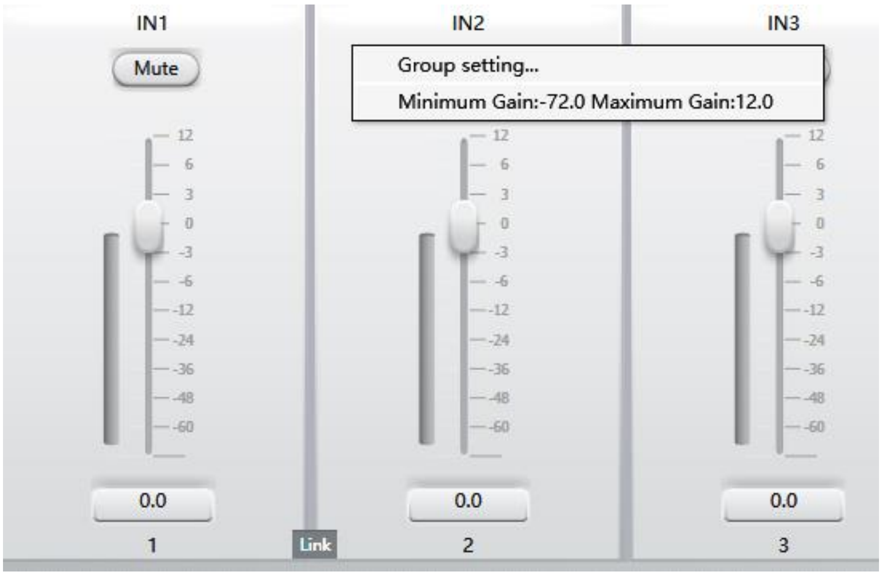
Mute: Mute the Channel.

Sine Wave: Drag the frequency to produce a sine wave at the specified frequency (20 to 20 KHZ). The Output level can be adjusted as required in dBFS. Use the fader to adjust or click the text Input box to specify a value.

White Noise: White noise has equal energy in each frequency component. Observe it on a spectrum meter with constant bandwidth, it has a flat spectrum. At this point the frequency adjustment is ineffective and the level is available.

Pink Noise: The frequency component power of pink noise is mainly distributed in the low and medium frequency bands, where it decreases at a rate of 3dB/Oct across the spectrum. At this time, the frequency adjustment is not effective and the level is available.

In addition, right click on each fader in the main interface to see the following menu Settings.

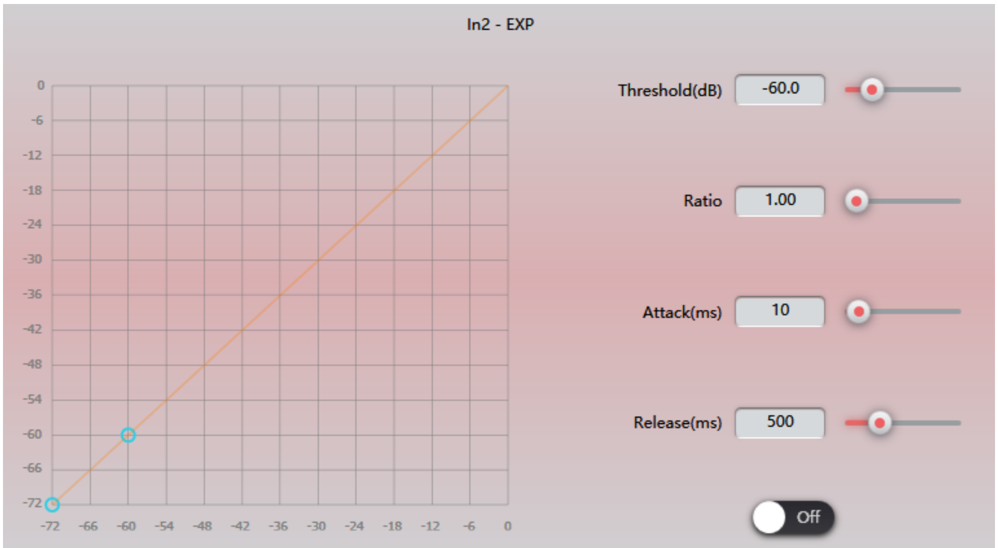


Group settings: Quickly opens the group Settings screen and groups Input or Output channels.

Minimum Gain and Maximum Gain: Limits the maximum and maximum gain of the Channel.

7.3.2. Expander

Expanders are the opposite of compressors in principle, being able to extend the dynamic range of a signal. The most basic difference between these two devices is that the compressor works on signals above a threshold, while the expander works on signals below a threshold. The expander is able to make small signals even smaller.



The extender has the following control Parameter:

Threshold: This level must be exceeded by the signal to open the expander (to allow the signal to pass through). In practice it is generally set to the Environment Noise of the magnitude.

Ratio: The slope below the Threshold point on the gain curve. The Action approaches the Gate when the ratio is set high.

Attack: Duration of the Input signal, above the Threshold, the time **it** takes to open the extender. A faster opening time allows for faster transient opening of the expander.

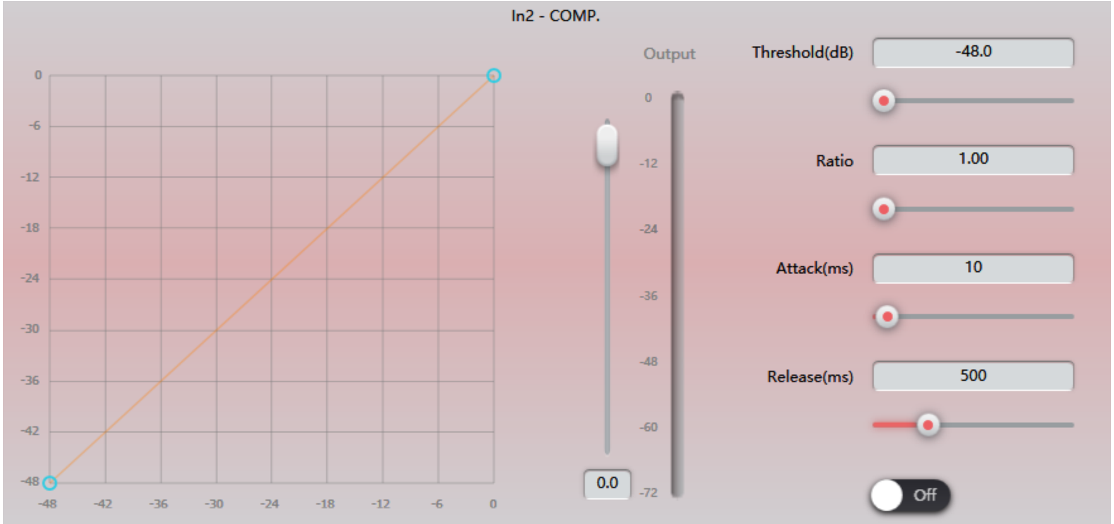
Release: The time it takes for the gain to return to a value below the Threshold after the Input signal drops below the Threshold.

Whether it is startup time or release time, its effect is only to reduce the rate of change of the amount of gain Damping.

7.3.3. Compressor & Limiter

Compressor

The compressor reduces the dynamic range of signals above a user-set Threshold, and signal levels below that Threshold remain.



The compressor has the following control Parameter:

Threshold: The signal level is above this value the compressor starts to reduce the gain. Any signal that exceeds the Threshold is considered an overshoot signal and its level is reduced under normal circumstances. The greater the range of the signal beyond the Threshold, the more the level will be Damping.

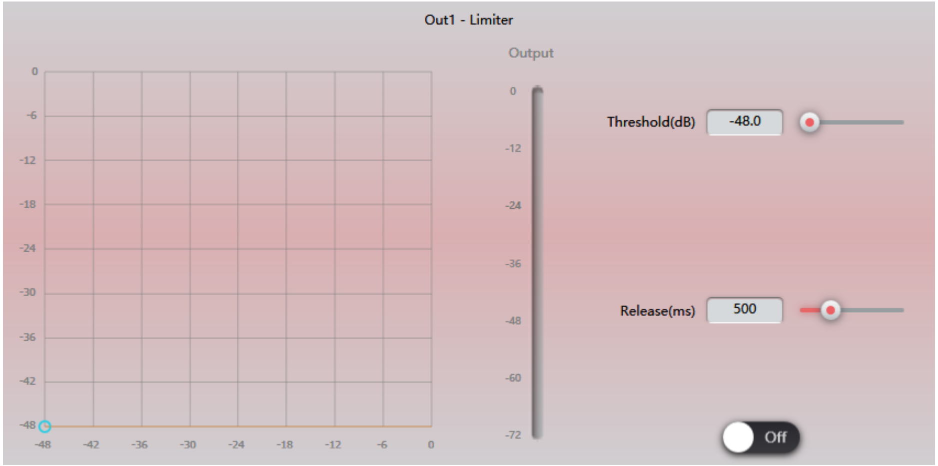
Ratio: This is compression ratio. The ratio determines the extent to which the overshoot signal Damping towards the threshold level. The smaller the compression ratio, the easier it is for the signal to be higher than the threshold. Once the signal exceeds the threshold, the Parameter of the compression ratio determines the ratio of the change in the Input signal to the change in the output signal.

Attack and Release: In order to retain the natural onset of vibration, it is usually desirable that an initial portion of the level will pass through the compressor unaffected (or only slightly affected). To achieve this, you need to slow down the compressor's reaction time. Similarly, if there is a large and rapid Damping of the signal gain, as well as a rapid recovery, the suction effect will occur. The set-up time and release time of the compressor are designed to avoid this happening. The buildup time can determine the speed at which the gain Damping occurs, while the release time can determine the speed at which the gain recovers.

Output gain slip Block: If the compressor significantly reduces the signal level, it may be necessary to increase the Output gain to maintain the volume. This lifting Operation is consistent for all parts of the signal and is independent of the setting of other Parameter of the compressor.

Limiter

It has only one key task: to ensure that the signal does not exceed the threshold level, no matter what the circumstances. By adjusting the control Parameter of the compressor, it can be made to work in a very similar way to the limiter. The core of the working principle of the limiter is the content of the signal below its true Relation threshold level, and how the gain Damping begins to be generated before the signal overshoots. The restriction period is completed through two processing stages, in the first stage is only slightly limited, but does not deal with overshoot signals, and in the second stage, if the signals produce overshoot, they will be in a very drastic manner Damping.

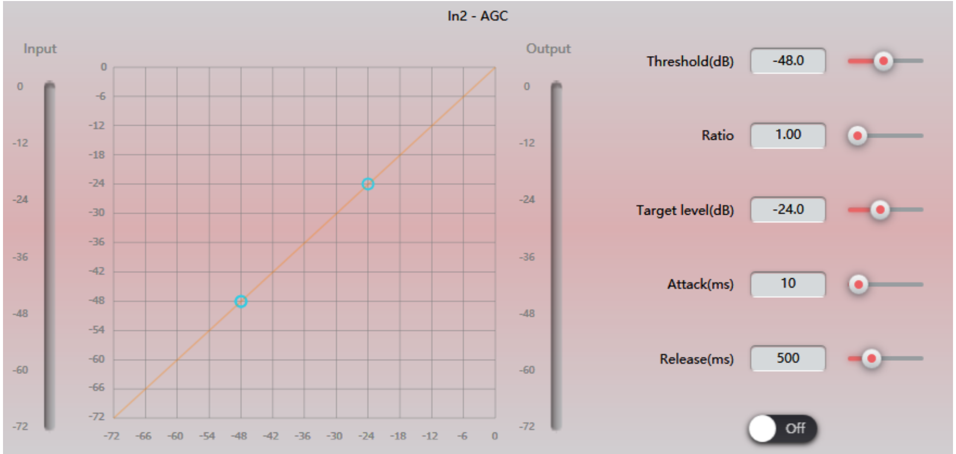


The limiter only provides two Parameter: Threshold and release time. For signal processing, the occasional clipping should be solved by the limiter, while the frequent clipping usually requires the attenuating of the signal level.

7.3.4. Auto Gain Control

Automatic gain control (AGC) is a special case of a compressor whose threshold is set at a very low level, medium to slow build time, long release time, and low ratio. The purpose is to raise a signal with an uncertain level to a target level while maintaining dynamics. Most automatic gain controls include some kind of silence detection to prevent loss of gain attenuation during silence. This is the only feature that sets an automatic gain control apart from a normal compressor/limiter.

Use automatic gain control to normalize the level of a CD player playing background, foreground, or waiting music to eliminate some paging microphone level variations.



Automatic gain control includes the following Parameter:

Threshold: When the signal level is lower than the threshold, the Input/Output ratio is 1:1. When the signal level is above this threshold, the Input/Output ratio varies with the ratio control setting. Set this threshold to background noise just above the Input signal level.

Ratio: The ratio between the change in the level of the Input signal above the threshold and the change in the level of the Output signal.

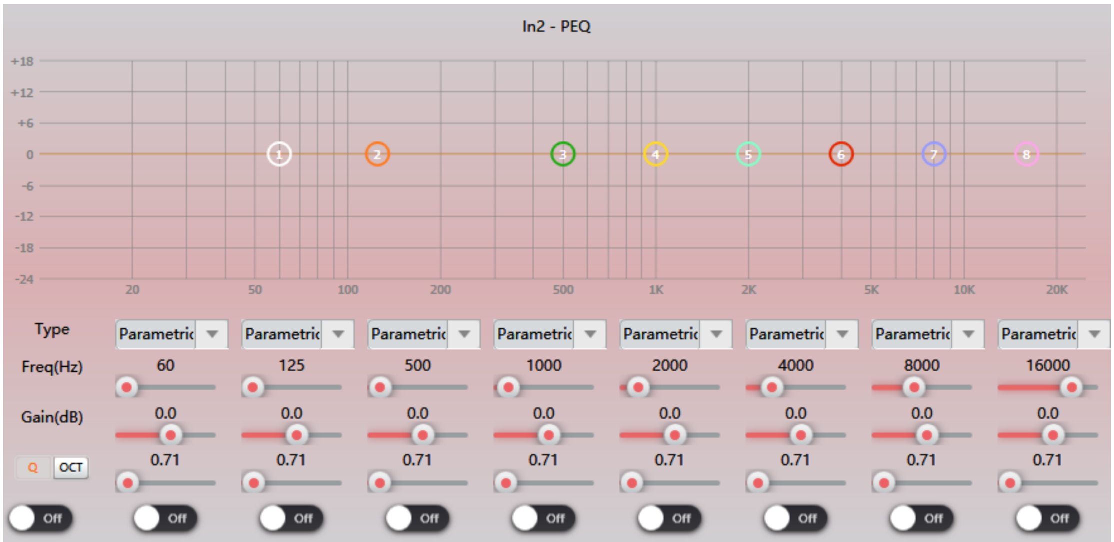
Target level: Desired Output signal level. If the signal is above this Threshold, the controller compresses the signal according to the ratio.

Attack: Controls the level reaction time above the Threshold.

Release: Controls the level response time of signals below the Threshold.

7.3.5. PEQ

The main purpose of the equalizer is to correct for overemphasized or missing frequency ranges, whether they are wide or narrow. In addition, equalizers can help us narrow or widen the frequency range, or change the Size of certain components of their spectrum. In simple terms, an equalizer changes the timbre of a signal.



The equalizer has the following control Parameter:

Type: Default parameter equalization, optional high and low frame filter and high and low pass filter. Each type of filter has different forms and can accomplish different functions.

High & Low pass: The reference frequency of the filter is called the cut-off frequency, and the frequency component on one side of the cut-off frequency can be completely passed through the filter, while the frequency component on the other side of the cut-off frequency is continuously Damping. Among them, High pass can let the frequency component above the cutoff frequency pass, and filter out the frequency component below the cutoff frequency. On the contrary, Low pass allows the frequency components below the cutoff frequency to pass, while filtering out the frequency components above the cutoff frequency.

High & Low shelf: also known as shelf filter. The overhead filter is defined as a partial gain increase or Damping of the frequency above the set frequency. The low frame filter is a partial gain increase or Damping of the frequency below the set frequency. The set frequency is not the cut-off frequency of 3dB, but the center point of the falling or rising edge of the filter. The Q value affects the peak value and has a mathematical relationship with the peak value.

Freq (Hz): The center frequency of the filter.

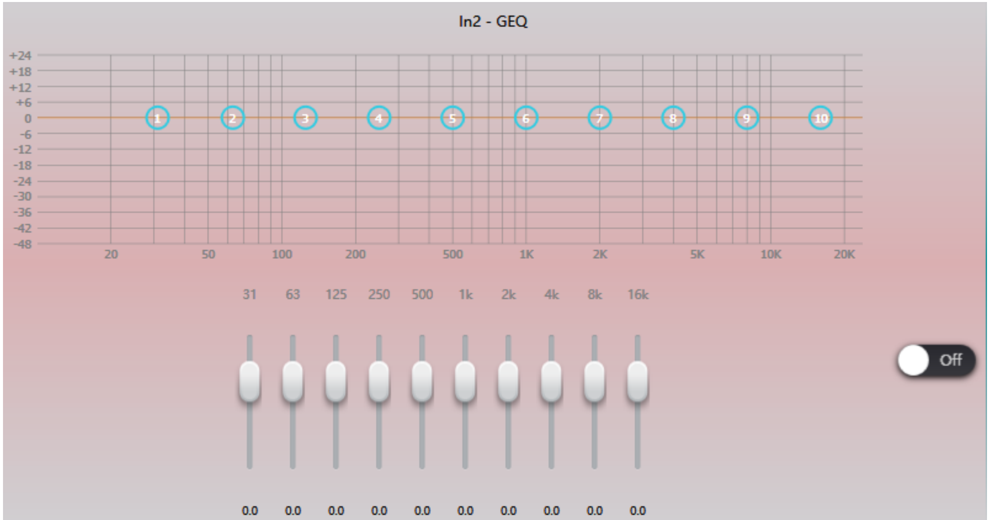
Gain (dB): The decibel value at which the gain increases or Damping at the center frequency.

Q/OCT: Quality factor of the filter. The Q value can be adjusted from 0.02 to 50; The adjustable range of OCT value was 0.029 to 11.289

There is a switch under each equalizer that enables or disables the equalizer. The Parameter setting of each equalizer takes no effect when the equalizer is disabled. The equalizer has a master switch that enables or disables the module.

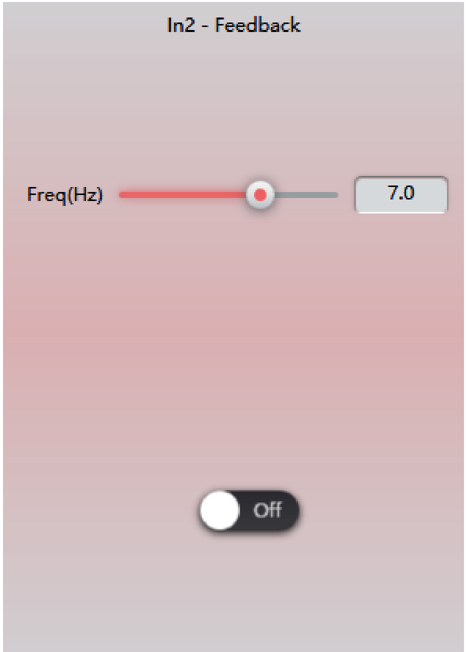
7.3.6. GEQ

Using constant Q-value technology, each frequency point is provided with a push-pull potentiometer, no matter raising or Damping a certain frequency, the bandwidth of the filter is always unchanged.



7.3.7. Feedback

Feedback suppression modules should always be used in conjunction with good system design and engineering practices, not as a substitute for good system design. Traditional methods such as limiting the number of microphones on, minimizing the Distance from the source to the microphones, positioning microphones and speakers for minimal feedback, and equalizing the room for a flat response should still be used. After that, feedback suppressors can be used for additional gain. Feedback suppressors do not magically solve a poorly designed system or increase the sound gain beyond the physical limits of the system.



The feedback suppression module automatically detects and suppresses acoustic feedback in the audio system. The module distinguishes between feedback and expected audio based on the characteristics of the signal. When feedback is detected at a certain frequency, a notch filter is automatically added to the feedback frequency point to Damping it.

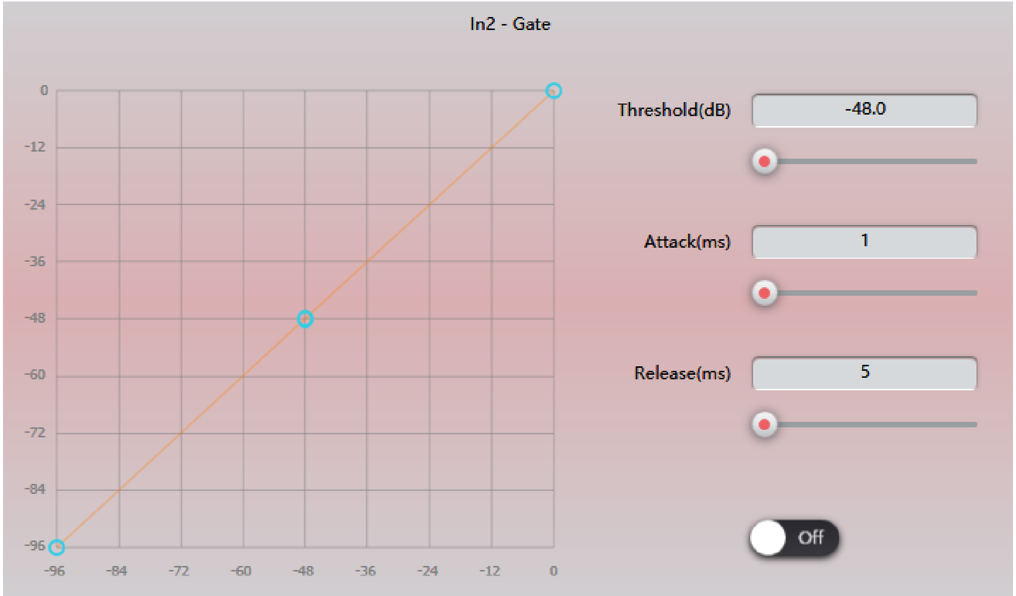
Each Channel has a feedback suppression. To enable the feedback suppression module, click the Open button to enable the feedback suppression of the corresponding Channel.

The following are the adjustable Parameter for feedback suppression:

Freq: The adjustment can modify the feedback suppression frequency

7.3.8. Gate

The main purpose of the Noise gate is to Damping signals below the threshold, and such Damping signals are usually Noise.



The Noise gate can be adjusted Parameter as follows:

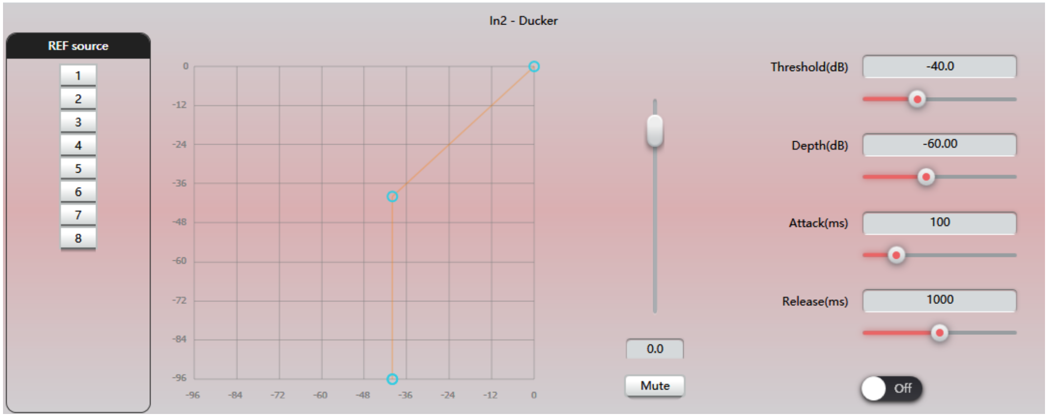
Threshold: The signal exceeds the Threshold and starts and Damping when the signal is smaller than the Threshold.

Attack: Start time refers to the speed at which the Noise door opens.

Release: The release time is the opposite of the start time, which refers to the speed at which the Noise door closes.

7.3.9. Ducker

When the level of a Channel exceeds the specified Threshold, the level of another Channel will be attenuated, which is the evasive effect.



REF source: Reference signal selected by the Ducker device. Channel from 1-8 can be selected as the reference signal

Threshold: When the reference signal is higher than the Threshold, it Damping. When the reference signal is lower than the Threshold, it recovers.

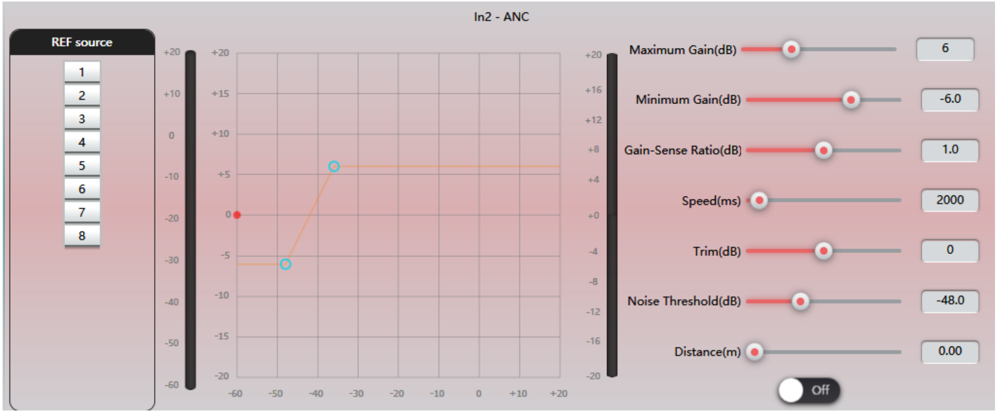
Depth: The amount of maximum reduction by the evaded signal. The amount that is lowered can be adjusted via the left slider.

Attack time: The time when the reference signal is higher than the Threshold and begins to Damping the avoided channel signal.

Release time: After the reference signal is lower than the Threshold, the time when the evaded signal recovers to the Size of the original signal.

7.3.10. ANC

Automatically adjust Output volume according to ambient Noise sensing and handling.

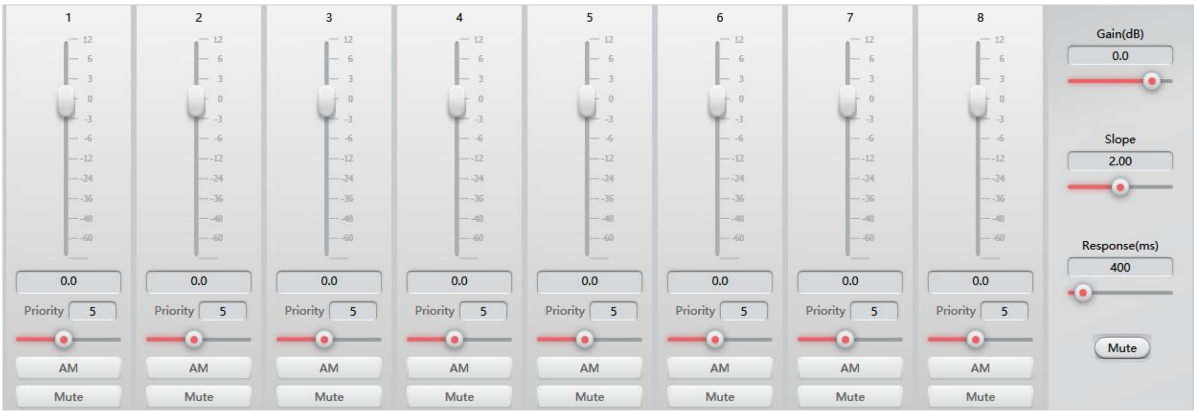


Maximum Gain: The maximum volume that can be adjusted.
Minimum Gain: The minimum amount that can be adjusted.
Gain-Sense Ratio: The ratio of increase or Damping.
Speed: Speed of increase or Damping.
Trim: Gain.
Noise Threshold: More than the start boost gain, less than Damping.
Distance: Distance between the reference signal and the local signal.

7.3.11. Auto Mixer

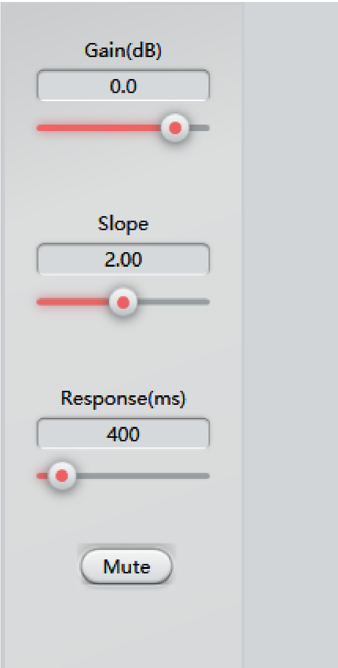
In the conference room, if multiple microphones are turned on to the Same gain level, and only one person is speaking, the result may not be very clear, other microphones will pick up room noise, reverberation, etc., when these signals are mixed with normal microphone signals, it will greatly reduce the quality of the mixed audio Output, and the whole sound amplification system is very easy to scream. And the whole sound reinforcement system is very easy to whistle, cannot get enough sound gain. In order to solve this problem, it is necessary to turn off other microphones that are not in use for a while. Automatic mixers can complete this shutdown process and react much faster than manual Operation.

A gain-sharing automatic mixer built into the processor supports 8 Channel of audio Input. Each Channel in the auto mix matrix has a direct Output, unaffected by auto gain and Channel faders, only by Channel muting. A Channel suitable for a fixed volume, such as a Channel for background music, needs to be kept at a fixed level without being controlled by automatic mixing; For example, if the chair microphone needs to be kept in the normally on State and its gain is not affected by automatic mixing, the Output of the channel can be adjusted directly in the Output matrix routing. At this point, the automatic mix button of the Channel can also be turned off, its gain will not be adjusted, and the signal level on this Channel will not affect the gain on other Channels.



The automatic mixing module has two sets of control Parameter:

1. Master control Parameter



Gain: Controls the main Output volume **of automatic mixing**

Slope: Slope control affects the Damping of lower levels. A low Channel will also be Damping more at a higher slope. The slope control works in a similar way to the ratio control on the expander.

When the slope is set to 2.0, it achieves a relatively ideal gain sharing and is the preferred value in use.

Response: A faster time ensures that the head of the spoken word is not cut off. The operation is more Smoothing when the time is slow. Practice shows that the response time is around 400ms for the best results. Autogain is designed to turn the microphone on much faster than it is turned off, so even with a 400ms response, the head of the spoken word is usually not subtracted. If you set a slower time of a few seconds, the auto mixer response time will have a longer hold time, and the last active Channel will save the open State for a few seconds.

Mute: Mute the automatic mixing Channel

2. Channel control Parameter

AM: Each Channel has an auto mix on/Off button that needs to be turned on for Channel participating in auto mix. It can also be turned off, and the Channel does not participate in automatic mixing.

Mute: Mute the Channel, but does not affect the Channel through the auto mix Output sound.

gain: Adjust the gain fader to increase/decrease the proportion of volume in automatic mixing.

Priority: The automatic mixing algorithm is affected when a high-priority Channel overtakes a low-priority channel. The Parameter ranges from 0 to 10. The larger the value, the higher the priority.

Priority control allows high priority Channel to Coverage low priority Channel, thus affecting the automatic mixing Channel. The control can take on a range of values from 0 (lowest priority) to 10 (highest priority), with a default value of 5 (standard priority). You can adjust the priority by using the slider, or you can Input a specified priority between 0 and 10 by clicking the edit box. Increasing this Numeric Value increases the priority.

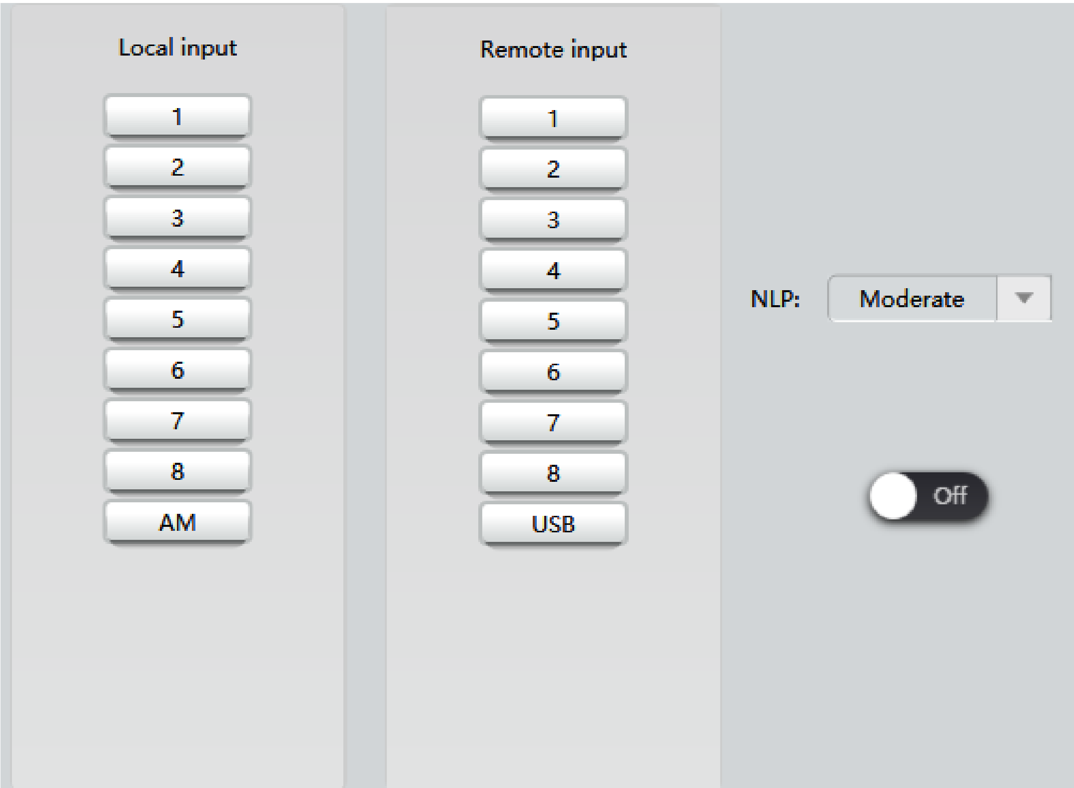
If two Size of the same signal level size, the channel with a higher priority will have a higher automatic gain. If two Channel differ by one unit of priority, the Channel with one higher priority gains an additional 2dB (Hypothesis the slope of both Channel is set to 2.0) of automatic mixing gain. For example, if the priority of Channel 1 is set to 6 and the priority of Channel 2 is set to 3, the Channel Input levels of both Channel have the same Channel, and the Channel will get an additional mixing gain of 6dB more than the Channel 2. Note that the slope setting of the main control Parameter will also affect the difference in mixing gain caused by the Channel's preferred weight. If the slope is set to 3.0, then a priority unit difference between Channel results in a gain difference of 4dB. If all Channel have the same priority, leave all Settings at the default level 5.

Note: Extreme priority differences between Channel, such as 0 and 10, need to be used with extreme care in some Settings. If a very high priority Channel is picking up signals from the speaker, such as background Noise, it is possible to mask a lower priority Channel, even if the very high priority Channel is not in use, the higher the slope of the problem is more serious. If this problem is encountered during installation and commissioning, consider adding a Noise gate or expander between the automatic mixers on the highest priority Channel, while setting the threshold cell to a threshold or level where the expander will not be opened by background Noise or speaker recognition.

7.3.12. Echo Cancellation

Acoustic Echo Cancellation or AEC is a digital audio signal processing technique used for audio and video teleconference when the conversation takes place between participants in a local conference room and one or more speakers at a certain Distance. AEC programs increase the Phonetic of remote speakers by eliminating Acoustic echoes generated in the local room.

The echo cancellation module applied in remote call can facilitate local amplification of remote Phonetic signal and Damping off the interference of acoustic echo. Its basic working principle is to simulate the echo channel, estimate the echo that may be formed by the remote signal, and then subtract the estimated signal from the Input signal of the microphone, so that the Input Phonetic signal no longer contains the echo, in order to achieve the purpose of echo elimination.



There is only one echo cancellation module in DSP Controller. The local Input and remote Input mixer are preset to realize multiple signals participating in echo cancellation, as shown in the figure. There is a Parameter to adjust:

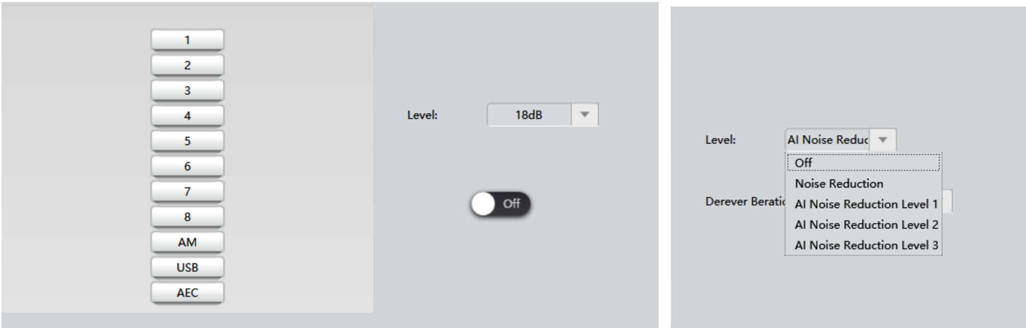
NLP: Conservative, Moderate, Aggressive. These three optional types select the level of suppression of the echo.

Note: Echo cancellation module Settings need to be used in conjunction with matrix module Settings for signal routing.

7.3.13. Noise Suppression

Noise suppression module can effectively remove the sound except human voice. Distinguish between human voices and non-human voices, and treat non-human voices as Noise. A piece of audio containing a human voice and Noise is processed by the module and, in theory, only the human voice is left.

There is only one Noise cancellation module in DSP Controller, and a multi-channel mixer is preset to realize the participation of multiple signals in Noise elimination. Some models have AI noise reduction modules, as shown in the figure.



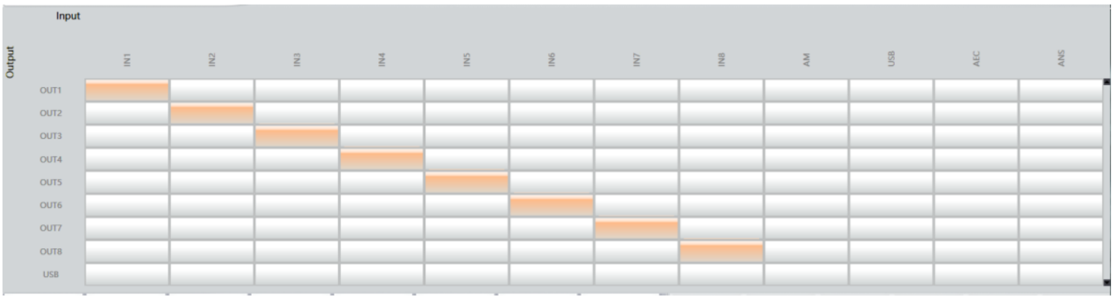
Level: A total of 6dB, 10dB, 15dB and 18dB are available. The meaning of dB is to suppress the Noise to reduce how much dB, the greater the value, the greater the damage to Phonetic, which is unavoidable.

Level (AI): There are traditional noise reduction, AI noise reduction level 1, AI noise reduction level 2, and AI noise reduction level 3 to choose from. The effect of AI denoising is better than that of traditional denoising, and the higher the level, the more obvious the effect is. The level is adjusted according to the actual situation.

De-reverberation: When speaking in an open environment, there may be reverberation. Turning on can alleviate this reverberation.

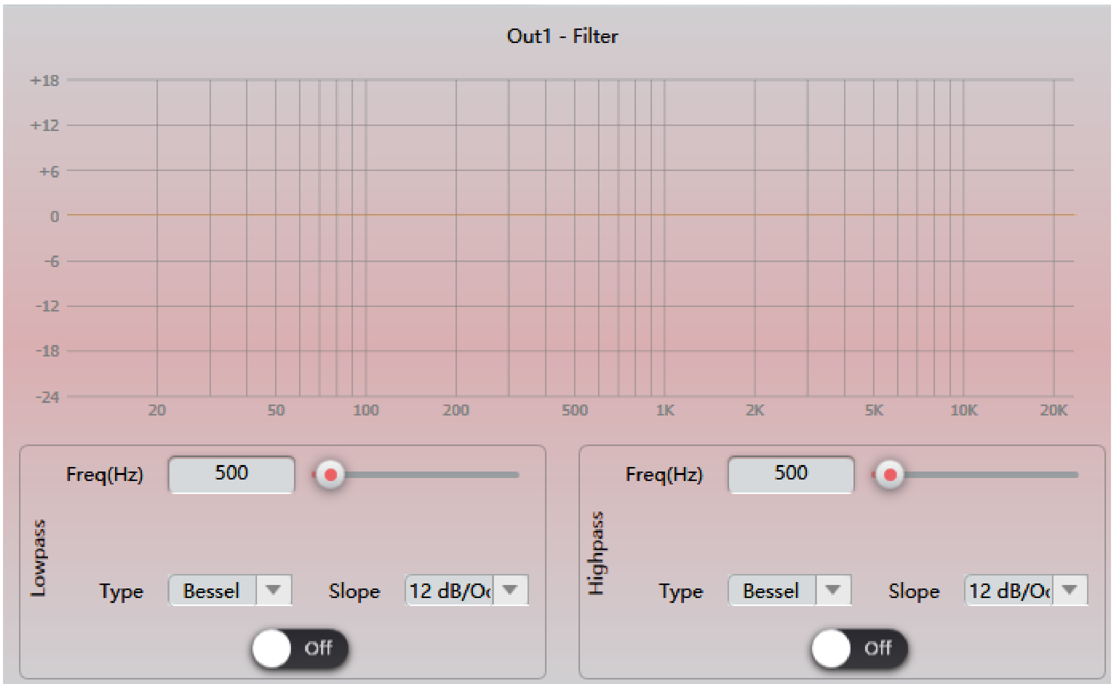
7.3.14. Matrix

The matrix has the dual Operation function of routing and mixing. Horizontal represents Input Channel, vertical represents output Channel, as shown in the figure. If you want the sound source to OUT1 through IN1 Output, then dot the corresponding matrix.



7.3.15. High Pass & Low Pass Filter

The high-low pass filter module can filter out sounds below or above a set frequency.



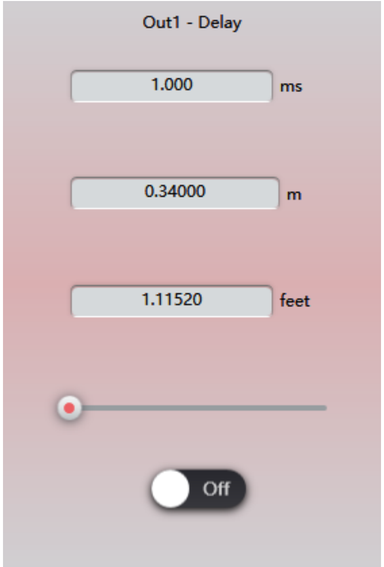
Each Output Channel provides a high-low pass module, consisting of a high-pass filter and a low-pass filter. Each filter has the following four Parameter:

Freq: The cutoff frequency of the filter.

Type: Select the type of filter, there are Bessel, Butterworth, and Linkwitz-riley types.

Slope: The Damping Size of the overbend of the filter. There are 8 choices: 6, 12, 18, 24, 30, 36, 42, 48dB/Oct. For example, 24dB/Oct indicates that in the transition zone, the frequency is different by one octave, and the amplitude Damping is 24dB.

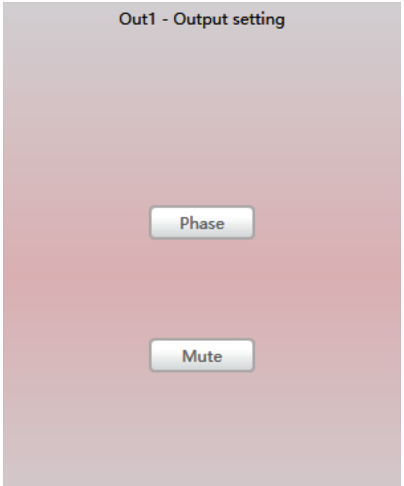
7.3.16. Delay



With the delay module enabled on the Channel type, the Input sound will delay the output cell based on the set parameter youdaoplaceholder3.

Millisecond: Set the delay time of the delay device. This value ranges from 0 to 500 milliseconds. Both meters and feet are unit values for milliseconds.

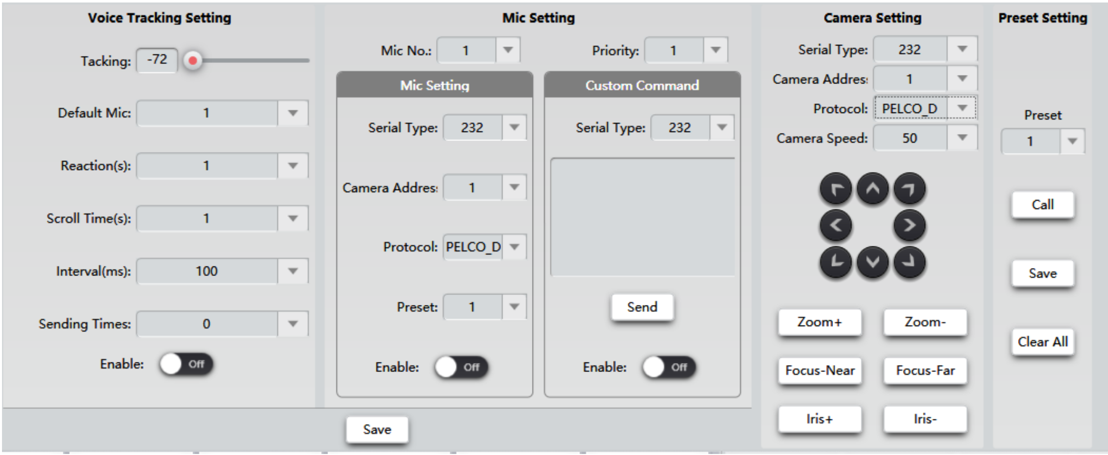
7.3.17. Output



Phase: The phase of the audio signal is reversed by 180°.

Mute: Set to mute/unmute.

7.3.18. Camera Tracking



Voice Tracking Setting

Tacking: Detects that the signal with microphone Input is greater than or equal to the tracking Threshold, and enables the tracking parameter.

Default Mic: When all microphones have no signal Input, turn the currently configured camera to the default microphone setting position;

Reaction: If you use the microphone to speak, set the reaction time to 1 second, the pause within 1 second is still considered as the signal continues to be valid, more than 1 second, it is considered as the signal is invalid, it will not start.

Scroll Time: The minimum speaking time required for the camera to switch to a valid position. If the microphone is used to speak, the speaking time must be greater than the "switching time", it is considered that the Channel signal is effective, and the camera will automatically turn to the set position. Usually, the "switching time" is greater than the "reaction time".

Interval: The interval time between each camera switch command is sent.

Sending Times: The number of times the camera switch command is sent, if 0 indicates special processing, it is only triggered once.

Mic Setting

Mic No: Generally corresponds to the input channel of the device, that is, the channel number of the microphone Linkage.

Priority: the smaller the priority level is, the higher the priority level is. When the priority is the same, it will be processed in accordance with the trigger priority order; If two microphones talk at the same time, the camera will automatically turn to the preset position corresponding to the microphone with the smaller priority level (that is, the higher priority level) or send the command corresponding to the mic with the smaller priority level (that is, the higher priority level); However, if the priority of the two microphones is the same, the signal detected first will prevail. The preset point, serial port number, camera address, and protocol are related to the camera and must correspond to the actual Linkage of the camera.

Enable: The settings take effect after being opened.

Custom Command: it means that when the device detects an input signal, it will automatically send the corresponding command to the defined serial port. Secondly, it can also set the command in advance. If you do not check the "Enable custom command", the device will not automatically send, but you can still click the "Send" button to send the command in the input box to the specified serial port at any time. Click "Save" to save the parameters to the device. At this time, the mic of the channel has been associated with the corresponding camera address. Then go through the "Enable Microphone Settings" option to determine if the microphone Settings are valid when tracking is enabled.

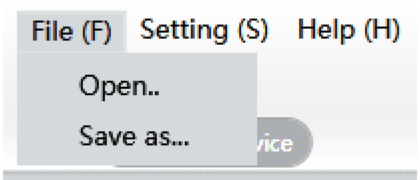
Camera Setting

Generally, the camera position is debugged before the tracking starts, and the parameters of this part will be saved on the camera at the end. First, set the serial port. The serial port protocol type can be 232 or 485, which corresponds to the rear backplane port Linkage of the PGND. Followed by the camera address and protocol, camera address please refer to the actual address of the camera, protocol and camera model related.

Preset Setting

Preset points are user-defined identifiers for the camera, then adjusting the up, down, left, right, focal length, zoom, and aperture parameters in the camera Settings will define the position and Settings of the camera; Finally, click "Save" to save the parameters, "Clear" to delete the information of the current preset point, and "call" to view the camera position saved by the current preset point.

7.3.19. File Menu

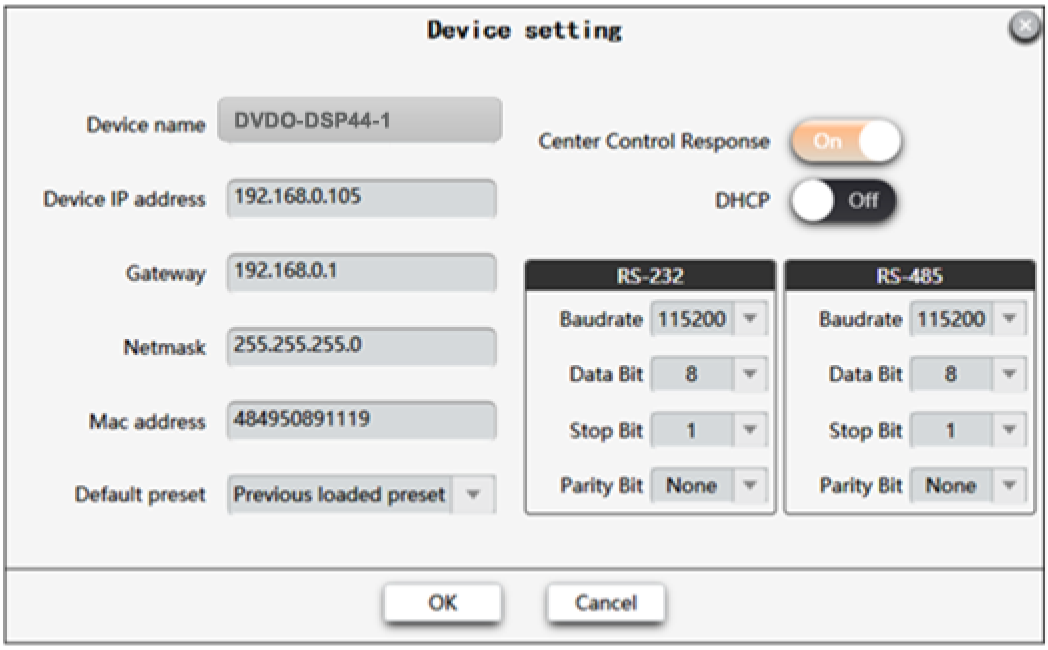


If you are not Linkage to the device, click Open to open an existing default file (extension: *.dps list).

Click "Save as" to save the preset on the application to the local hard drive for easy copy and storage.

7.3.20. Setting Menu

Device setting



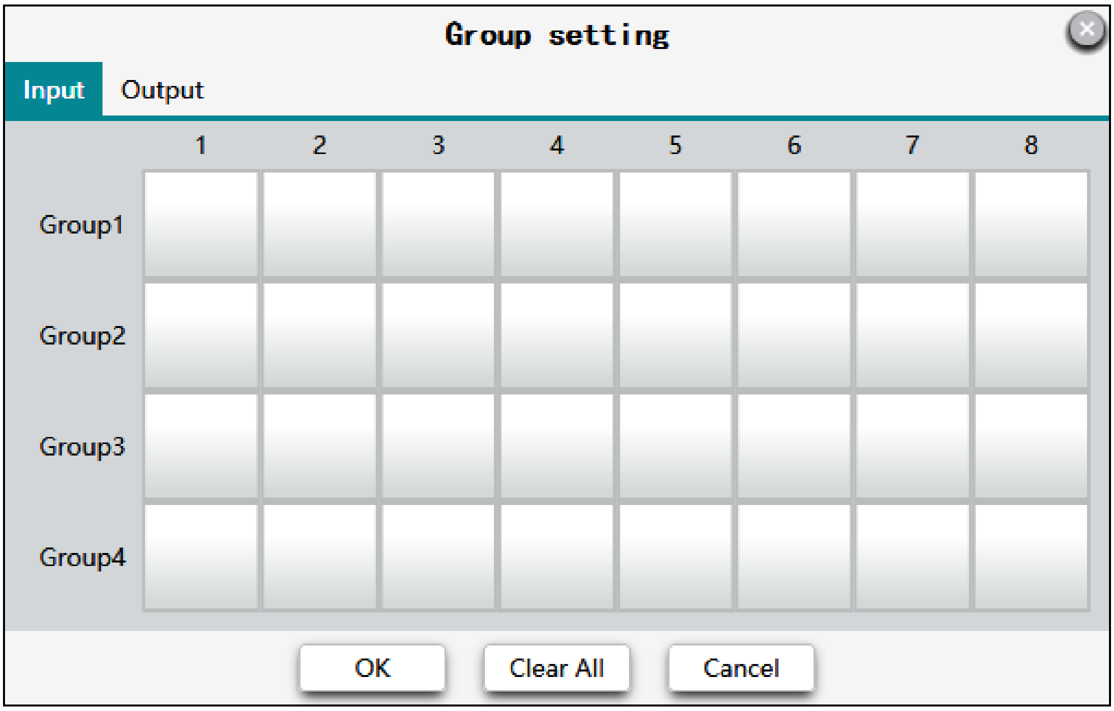
The device name, device IP, RS-232 configuration parameters and RS485 configuration parameters can be set.

Default preset: Two startup preset modes can be selected. One is to specify any one of the 16 presets as the startup preset, and each startup will be started with the preset. The second is to select the preset of last loading, and the preset used last time before power off is used as the preset for the next boot.

DHCP: After enabling the device will automatically obtain an IP address, without the need to manually set the IP address, effective after restart.

Group setting

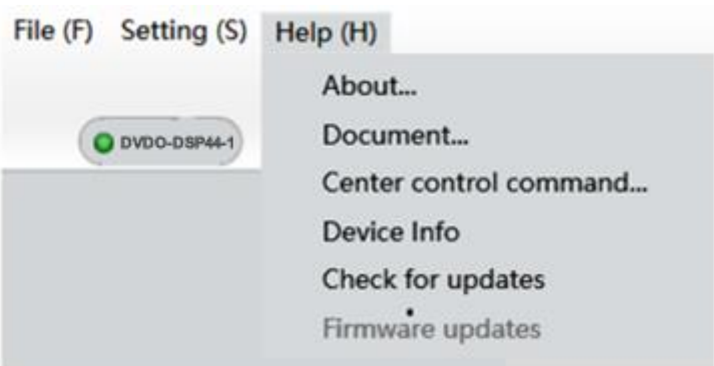
The group interface is divided into two labels: input and output, and the maximum of four groups can be set under each label. One channel can only participate in one group. Under the same group, their channel volume adjustment and mute adjustment are synchronized.



Preset Name

Change the preset name for preset 1-16.

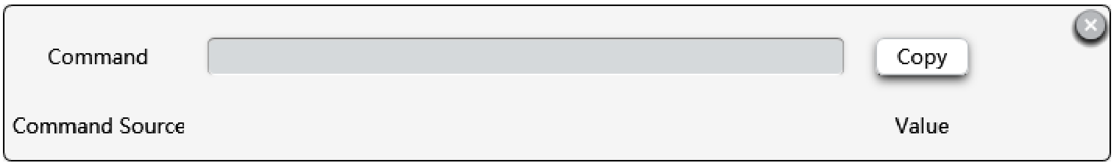
7.3.21. Help Menu



About: Displays version number, technical support contact information, etc.

Document: Get the current device help documentation.

Center control command: View the corresponding control command for the current function.

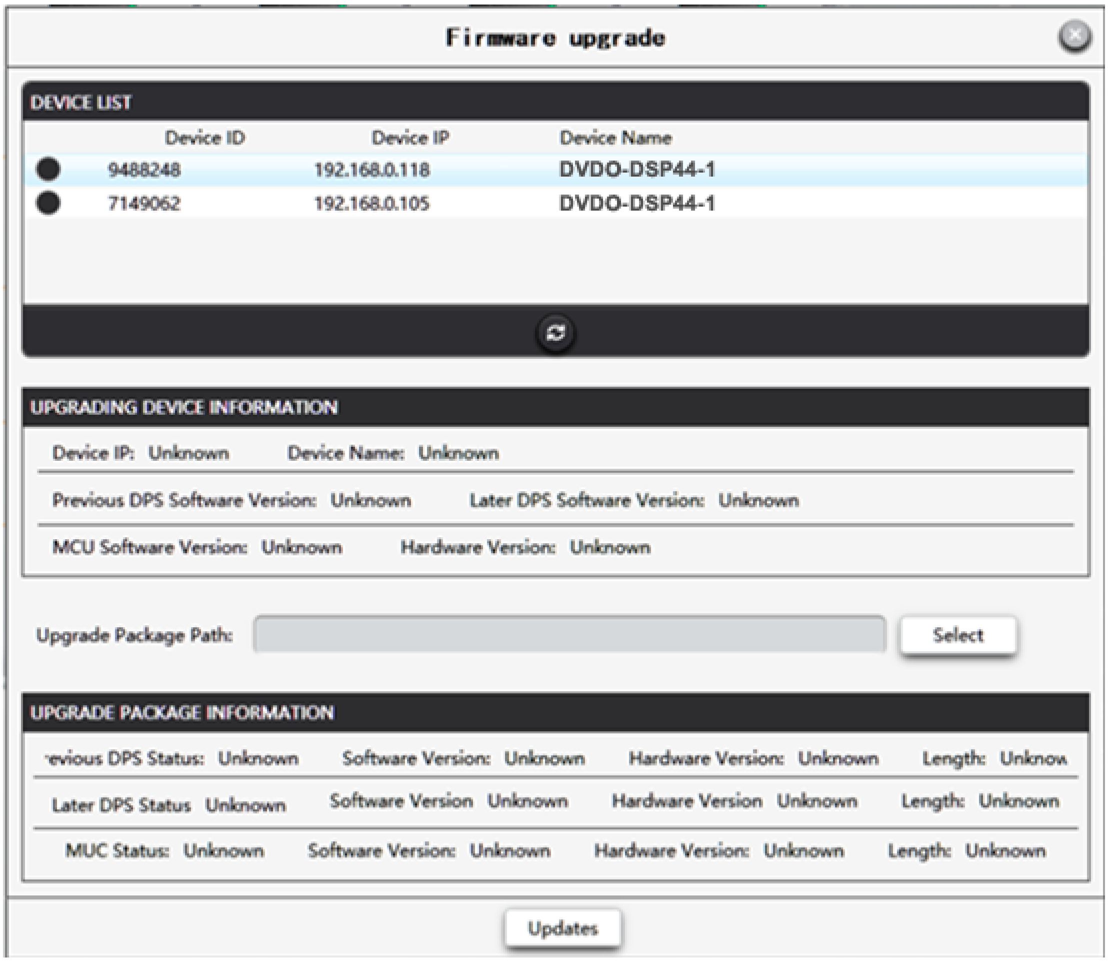


Click on the corresponding function to display the corresponding control command. Note: There is no central control command for camera tracking.

Device Info: View current device version information, etc.

Check for updates: Check if a new version of the DSP control software exists.

Firmware updates: Update the device firmware.



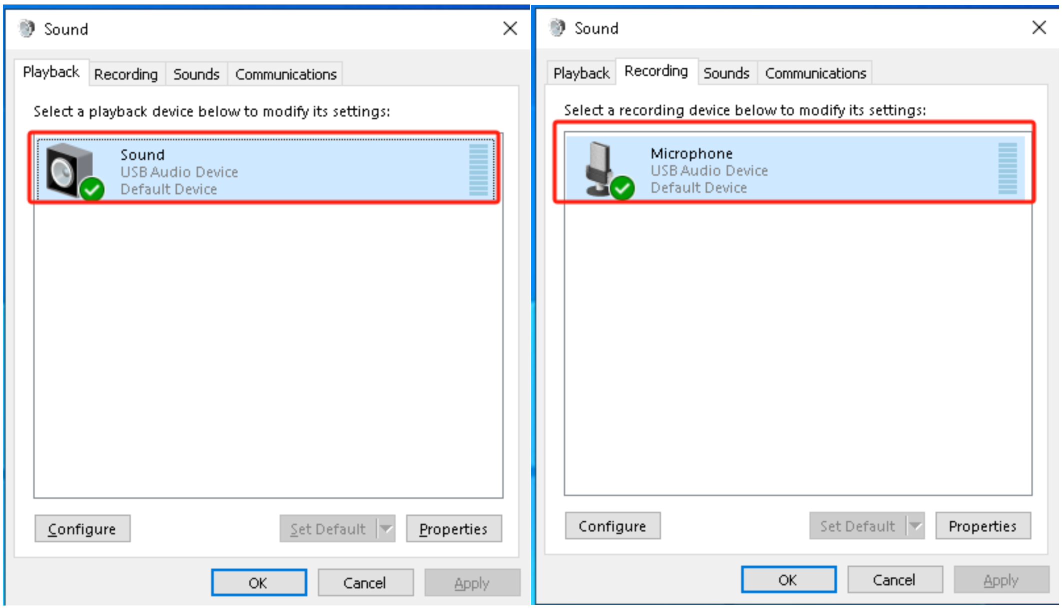
Select the device to display the device information of the current device. After selecting the firmware update package, click Upgrade to upgrade the device to the corresponding version.

7.4. USB Soundcard

The use of USB sound card has two functional purposes: one is to achieve recording and playing; The second is remote conference on PC, USB sound after echo cancellation and noise cancellation.

Sound Card Settings

Linkage to the host computer through the USB data cable of the double-headed Type-A interface. The first Linkage, the computer will automatically install the driver. After the installation is complete, the USB sound card of the current device is displayed in the Player sound card list and recording sound card list of the PC. Select the USB sound card of the current device (the name of the current device sound card is selected based on the actual situation) as the default device to complete the Settings.



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