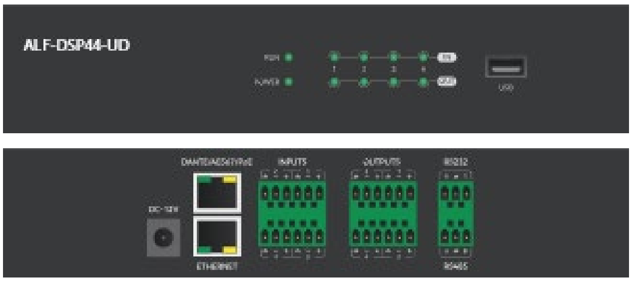
ALF-DSP USER GUIDE

ALF-DSP44-U

4x4 Digital Sound Processor

ALF-DSP44-UD

4x4 Digital Sound Processor with Dante

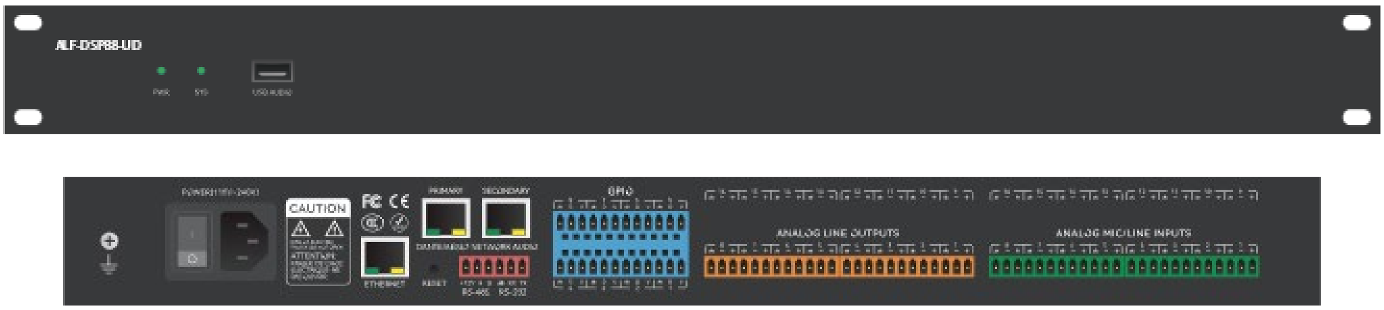


ALF-DSP88-U

8x8 Digital Sound Processor

ALF-DSP88-UD

8x8 Digital Sound Processor with Dante



Version: ALF-DSPxxxxx - V2.0 19072022

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Preface

Read this user manual carefully before using the product. Pictures shown in this manual are for reference only. Different models and specifications are subject to the real product.

This manual is only for operational instruction, please contact the local distributor for maintenance assistance. The functions described in this version are updated as of September 2021. In the constant effort to improve the product, we reserve the right to make functions or parameters changes without notice or obligation. Please refer to the dealers for the latest details.

FCC Statement

This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. It has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a commercial installation.

Operation of this equipment in a residential area is likely to cause interference, in which case the user at their own expense will be required to take whatever measures may be necessary to correct the interference.

Any changes or modifications not expressly approved by the manufacturer would void the user's authority to operate the equipment.

Do not dispose of this product with the normal household waste at the end of its life cycle. Return it to a collection point for the recycling of electrical and electronic devices. This is indicated by the symbol on the product, user manual or packaging. The materials are reusable according to their markings. By reusing, recycling or other forms of utilisation of old devices you make an important contribution to the protection of our environment. Please contact your local authorities for details about collection points.

1. Hardware

1.1 Safety Instructions

Safety Instructions

Important safety instructions:

1. Read these instructions.

2. Keep these instructions secure.

3. Pay attention to all warnings.

4. Follow all instructions.

5. Please keep the device away from water. The device cannot be exposed to water drips or water splashes; make sure that there is no object with liquid near the device, such as a vase.

6. Please use dry cloth to clean the device.

7. Please do not block the vent. Please install the device based on the manufacturer's instructions.

8. Please do not install near any heat source, such as radiator, furnace, or other devices (including amplifiers) that generate heat.

9. Please make use of a protective grounding connection to connect the device to the power socket. Please do not use polarized plug or grounding plug. A polarized plug has two leaves, and one is wider than another. A grounding plug has two leaves and a third ground terminal. The wide leaf or third ground terminal can provide safety for the users. If the plug provided does not correspond with the power socket, please contact an electrician to replace the old socket with a new one.

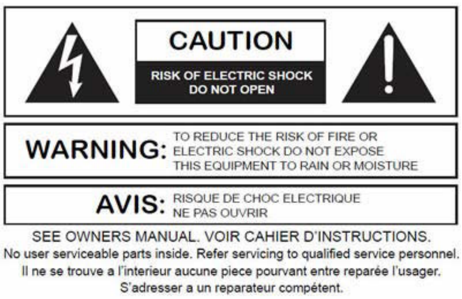
10. Protect the power cord so that it will not be trampled or jutted, particularly the plug, the socket and the connections of cord and device.

11. Please use the accessories designated by the manufacturer.

12. Please only use a cart, a tripod, or a desk designated by the manufacturer, or sold together with the device. When using a cart, please take care with the mobile cart / device to avoid injury from rollover.

13. Please unplug the device during a thunderstorm or during the idle period.

14. Please contact our qualified maintenance personnel to deal with all maintenance related issues. For example, the power cord gets damaged, liquid has spilt over, or an object falls onto the device; the device is exposed to rainwater or moisture; the operations are not correct, or the device completely fails.



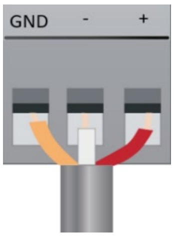
The lightning logo (an equilateral triangle with an arrow) is used to make the users aware of the uninsulated "dangerous voltage" within the product shell, which may cause electric shock. An equilateral triangle with an exclamation mark is adopted to make the users understand the importance of the operations and maintenance instructions given in the appendixes attached to the product.

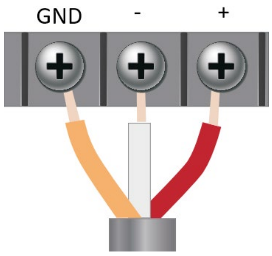
Warning: In order to prevent electric shock, please do not use a polarized plug provided by a device with an extension cord. The socket outlet cannot be inserted except for the sharp end.

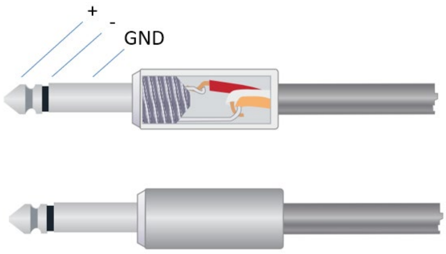
1.2 Audio Wiring Reference

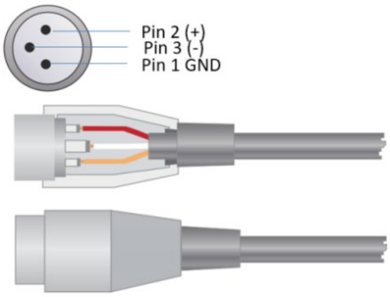
Balanced Audio Connection

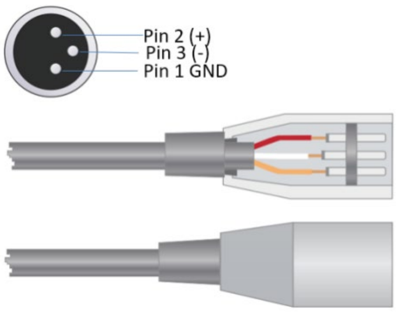
Any of these audio connections may occur on either input or output terminals.





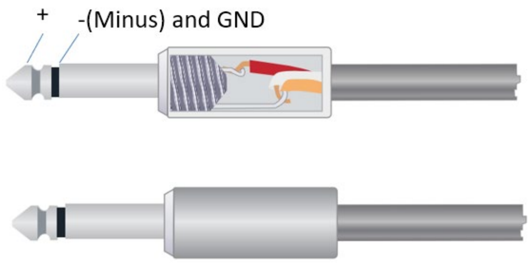


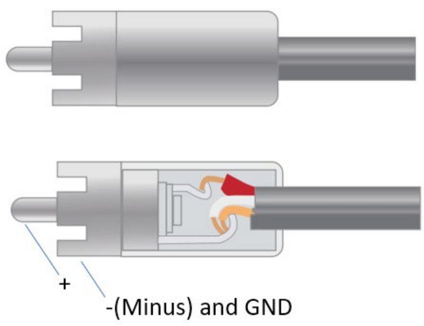




Unbalanced Audio Connection

RCA audio connections and 1/4-inch TS audio connections are unbalanced connections. A multi-strand shielded conductor may be installed on both ends of the unbalanced connection. In such a case, please note to join the negative and shield conductors as indicated.





1.3 Specifications

	ALF-DSP44-U	ALF-DSP44-UD	ALF-DSP88-U	ALF-DSP88-UD
ANALOGUE	4-in / 4-out	4-in / 4-out	8-in / 8-out	8-in / 8-out
DIGITAL	-	4-in / 4-out	-	8-in / 8-out
USB	1-in / 1-out	1-in / 1-out	1-in / 1-out	1-in / 1-out
PROCESSOR	ADI SHARC 21489(x1)	ADI SHARC 21489(x2)	ADI SHARC 21489(x2)	ADI SHARC 21489(x2)
SAMPLING RATE/ DIGITIZING BIT	48KHz / 24bit	48KHz / 24bit	48KHz / 24bit	48KHz / 24bit
SYSTEM LATENCY	<3ms	<3ms	<3ms	<3ms
INPUT GAIN	0 – 48dB in 6dB Steps	0 – 48dB in 3dB Steps		
PHANTOM POWER	48V			
FREQUENCY RESPONSE (20-20KHz)	±0.2dB			
MAXIMUM LEVEL	+24dBu			
THD + NOISE	0.003%@4dBu			
DYNAMIC RANGE	110 dB			
BACKGROUND NOISE (A-WEIGHTED)	-91 dBA			
CHANNEL ISOLATION @1KHz	108 dB			
INPUT IMPEDANCE	9.4KΩ			
OUTPUT IMPEDANCE	102Ω			
ACOUSTIC ECHO CANCELLATION (AEC)	Y (2-channels)	Y (2-channels)	Y (1-channel)	Y (1-channel)
AMBIENT NOISE COMPENSATION (ANC)	-	-	Y	Y
AUTO MIXER	Y	Y	Y	Y
AUTOMATIC GAIN CONTROL (AGC)	Y	Y	Y	Y
AUTOMATIC NOISE SUPPRESSION (ANS)	Y	Y	Y	Y
COMPRESSOR	Y	Y	Y	Y
DELAY	Y	Y	Y	Y
DUCKER	-	-	Y	Y
EXPANDER	Y	Y	Y	Y
FEEDBACK SUPPRESSOR	Y	Y	Y	Y
GRAPHIC EQUALIZER	-	-	Y	Y
HIGH/ LOW PASS FILTER	Y	Y	Y	Y
LIMITER	Y	Y	Y	Y
MATRIX	Y	Y	Y	Y
NOISE GATE	-	-	Y	Y
PARAMETRIC EQUALIZER	Y	Y	Y	Y
CAMERA CONTROL	-	-	Y	Y
ETHERNET	1 LAN	1 LAN / 1 DANTE	1 LAN	1 LAN / 2 DANTE
SERIAL PORT	1 RS-232 / 1 RS-485	1 RS-232 / 1 RS-485	1 RS-232 / 1 RS-485	1 RS-232 / 1 RS-485
GPIO	-	8GPI / 8GPO	8GPI / 8GPO	8GPI / 8GPO
POWER REQUIREMENTS	110-240V AC 50Hz-60Hz	110-240V AC 50Hz-60Hz	110-240V AC 50Hz-60Hz	110-240V AC 50Hz-60Hz
MAXIMUM POWER CONSUMPTION	<40W	<40W	<40W	<40W
RACK SPACE	Half Rack	Half Rack	1U	1U
DIMENTIONS (W x D x H)	215 x 184 x 45mm	215 x 184 x 45mm	482 x 260 x 45mm	482 x 260 x 45mm
SHIPPING WEIGHT	1,8KG	1,8KG	3KG	3KG

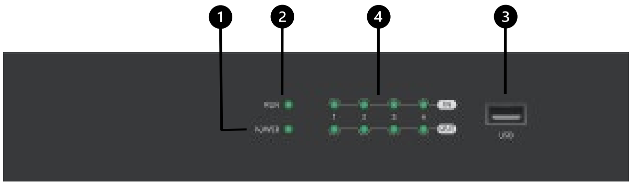
1.4 Mechanical

Ventilation:

The recommended highest operating ambient temperature is 30℃ / 86℉.

Make sure that there is no obstruction on both sides (a gap of at least 5.08cm / 2 inches shall be reserved). Please do not cover the thermal vents of the device with newspapers, a tablecloth, or any other objects.

1.5 Front Panel



ALF-DSP44-U / ALF-DSP44-UD

ALF-DSP88-U / ALF-DSP88-UD

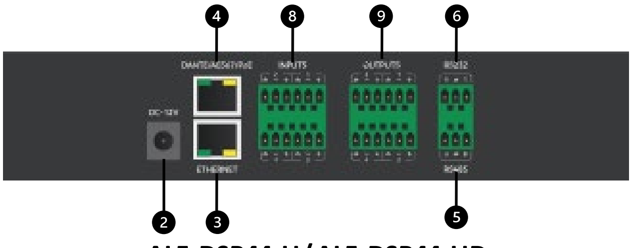
① **Power:** LED power indicator.

② **STATUS:** The operational status indicator of the device.

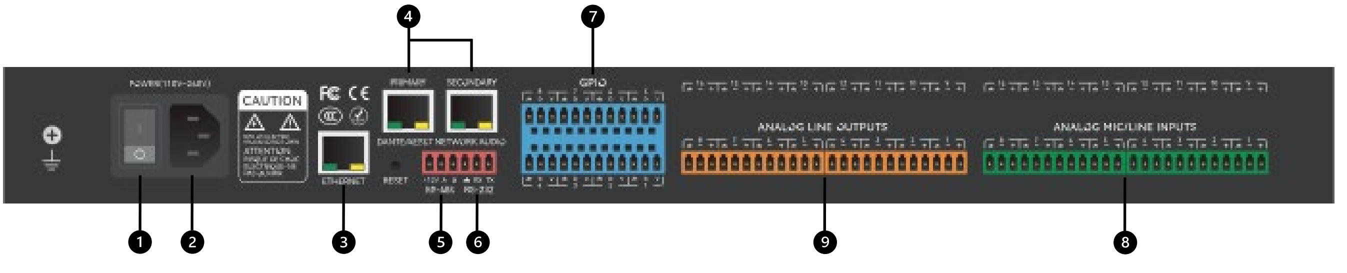
③ **USB AUDIO:** USB audio for connection to host PC. (1-in / 1-out)

④ **I/O:** Shows signal status of Input / Outputs.

1.6 Rear Panel



ALF-DSP44-U/ ALF-DSP44-UD



ALF-DSP88-U/ ALF-DSP88-UD

① **Power switch:** Turn unit on / off. *(ALF-DSP88-U and ALF-DSP88UD ONLY)*
② **Power connector:** (Supports 110 - 240V AC 50/60Hz, and supports a maximum power of 40W)
③ **Ethernet Connector:** 10/100 Base-T Ethernet connector is used for IP-based PC software and host control, and third-party accessory controller.
④ **Dante Connections:** Dante connections for connecting to Dante Digital Media Network.
(ALF-DSP44-UD and ALF-DSP88-UD ONLY)
⑤ **RS-485:** Used for the serial communication port Tx = sending or data output or Rx = receiving or data input that connects to a third-party control device. Port setting: 115200 baud (default), 8 data bits, 1 stop bit, no parity, no flow control.
⑥ **RS-232:** Used for serial communication; port Tx = sending or data output or Rx = receiving or data input that connects to a third-party control device. Port settings: 115200 baud (default), 8 data bits, 1 stop bit, no parity, no flow control.
RS485 & RS-232 can be used for voice tracking control (or other output commands), or for bus input control. A central command can be used to conveniently integrate into your software.
⑦ **GPIO:** 8-channel logic connections, with 4 pairs of universal grounding pins. After being activated, the logic output will be low (0V), and the internal voltage will be high (5V) when not activated. You may directly power and light up external LEDs. The logic output can be driven by the logic output control module in the device design. Polarity and threshold can be set in the software.
⑧ **INPUTS:** Balanced mic/line level audio inputs with +48V Phantom power.
⑨ **OUTPUTS:** Balanced line level audio outputs.

2. Technology Overview

2.1 Introduction to DSP Technology

Audio Digital Signal Processors (DSP) are equipped with several core technical features to simulate the work of an audio engineer. DSP-based audio management, routing, processing, and control is facilitated via a computer running the GUI Software for the Audio DSP Hardware. This Manual mainly introduces the techniques used to achieve this goal.

The DSP Controller software is a Windows-based application, which is used to configure and control the DSP hardware. The DSP Controller has 16 built-in presets, and the modules and sequences for each preset can be flexibly designed in accordance with the designer's requirements. Once the design is completed, it can be saved for future use. The sequences and parameters of the DSP Controller's built-in processing modules work with most application scenarios without needing any changes.

The DSP Controller is a full-feature application, including parameter adjustments and peripheral accessory settings of all modules, such as RS232, RS485, and click-and-drag panel configuration etc. The most interesting part is the user interface, which allows the engineer to customize the user interface so that the integrator can edit it, or the onsite technicians or end users who have no idea of the relevant techniques can operate it. Advanced security features make it possible for the end user to only access the controls allowed by the engineer, or designer.

2.2 Audio Input Section

The DSP's supports up to 4 or 8 fixed analogue audio inputs (model dependent), which can be connected via removable balanced phoenix connectors. The analogue input section supports microphone or line-level signals whose nominal levels range from 0dBu to +48dBu in 3dBu steps.

+48VDC phantom power can be independently switched on/ off for each input.

Preamp gain and phantom power can be conveniently controlled via the DSP Controller.

The Analogue to Digital (A/D) converter, or ADC, adopts an advanced 24-bit 256X sampling rate converter.

Analogue to Digital (A/D) Converter Technical Specs:

Sampling rate: 48kHz

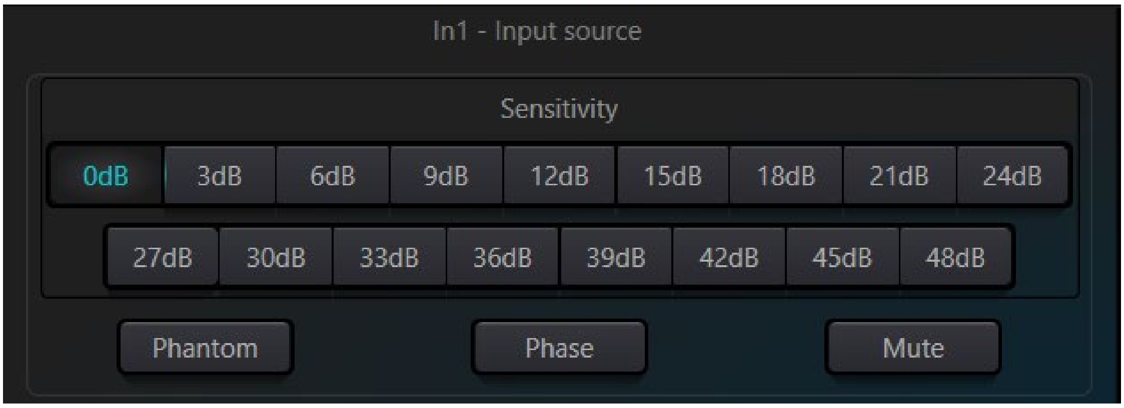
THD+N: 105dB

Dynamic range: 120dB

Audio format: 24Bit MSB TDM

2.3 Audio Output Section

The Analogue output section refers to the digital to analogue (D/A) converter, or DAC, also adopts an advanced 24-bit 256X sampling converter. Just like the A/D converter, it also uses multi-bit architecture for broader dynamic range. Unit gain (0dB) is set via volume control, and the analogue output section is corrected as +4dBu with 20dB headroom. That is to say that 0dBFS digital signal is equivalent to +24dBu output signal. If other signal levels are required, you may change the output volume to achieve it.



Digital to Analogue (D/A) Technical Specs:

Sampling rate: 48kHz

THD+N: -100dB

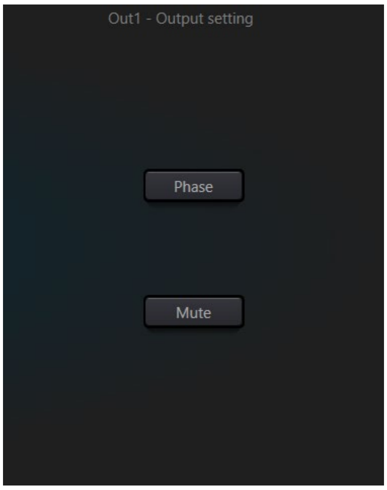
Dynamic range (A-weighted): 118dB

Audio format: 24Bit MSB TDM

On the software output interface there are two controls: Phase and Mute.

Phase: The phase button inverts the phase of the output signal by swopping the polarity.

Mute: Mutes the analogue output of the respective channel



2.4 Floating Point DSP

The ALF-DSP88-U adopts analogue devices' ADI SHARC audio processors, enabling 32-bit and 40-bit floating-point processing, which can be compared to 40-bit floating-point processing of other devices. Floating-point processing provides prominent advantages for the users in terms of sound quality and usability.

Fixed-Point Processing Limitations

Fixed-point processing has its own disadvantages. If there is a significant change in gain, data loss or a more severe error may occur, including clipping or distortion. For example, the processing of 24-bit fixed point-based audio signal, in some cases, if you attenuate the signal to 42dB, the new signal only includes 17-bit information. Due to gain attenuation, 7 bits of information will be lost forever. Worse than that is the clipping distortion. For a signal nearly close to 0dBFS, the signal will be clipped at 0dBFS, and audio distortion will occur. Even if the signal level is adjusted to below 0dBFS through post-regulation, the clipping has occurred, and the distortion still exists. Fixed point processing can create some headroom above 0dBFS, but by doing so, some bits have to be abandoned. For example, if a 12dB (2 bits) headroom is created, a 24-bit system will only have 22 bits.

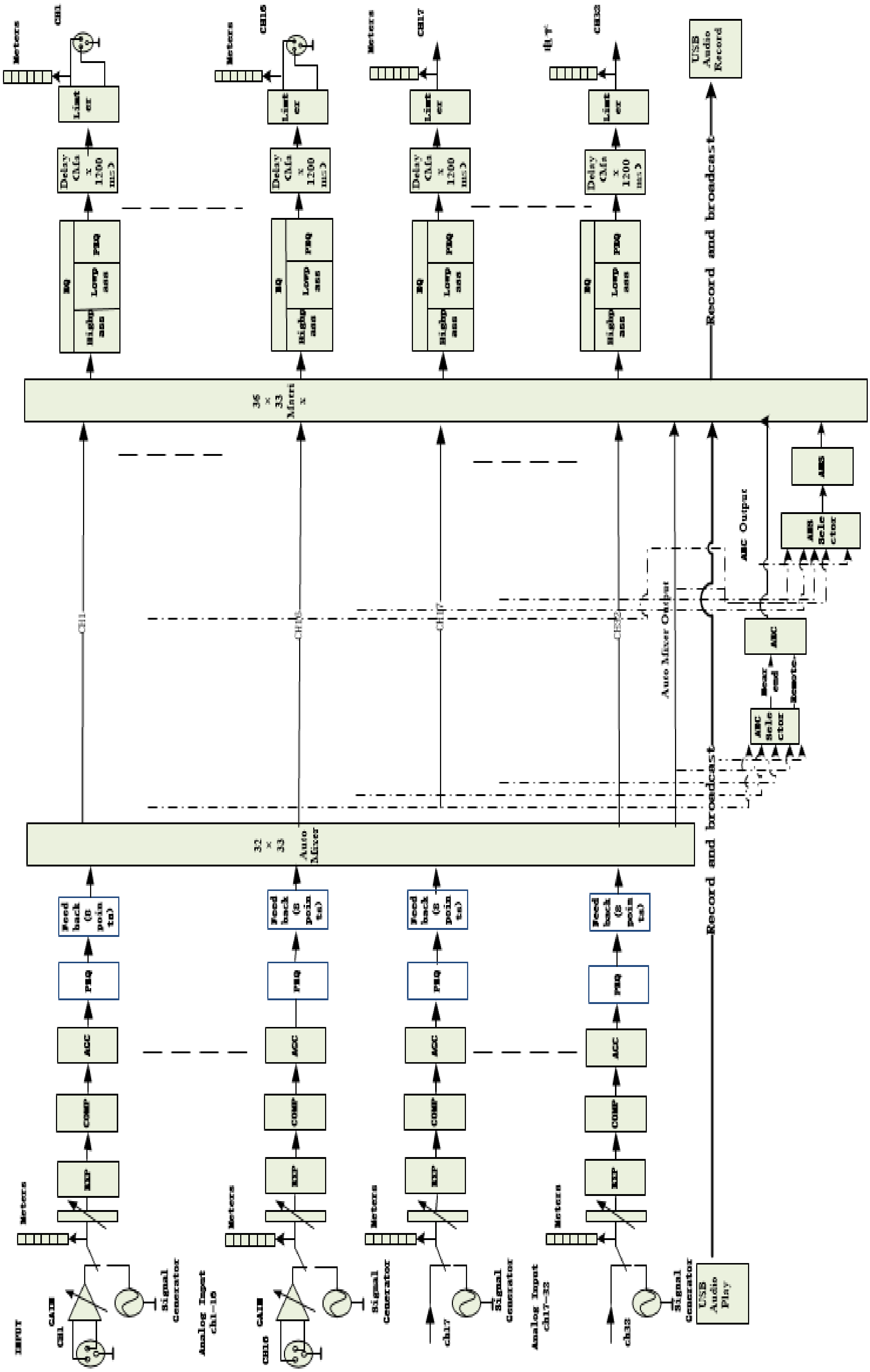
Floating-Point Processing

On the contrary, by taking advantage of floating-point processing, no matter what the signal level is, all available bits are uniformly distributed to the signals. Basically, the floating points use some bits as indexes to set up general signal level and distribute the remaining bits to signals with independently stored level. As a result, no matter what kind of level (from -200dB and 200dB below to 0dBFS above, the stored signal's accuracy is optimized without clipping distortion. SHARC provides 32-bit and 40-bit accurate processing; through 32-bit processing, 25 bits are distributed to storage signals no matter what its signal level is. This means that, based on at least 1-bit low level signal, its accuracy is always significantly superior to 24-bit fixed point processing. Through expanded 40-bit accurate processing, 33-bit storage signals can be achieved.

Practical Significance

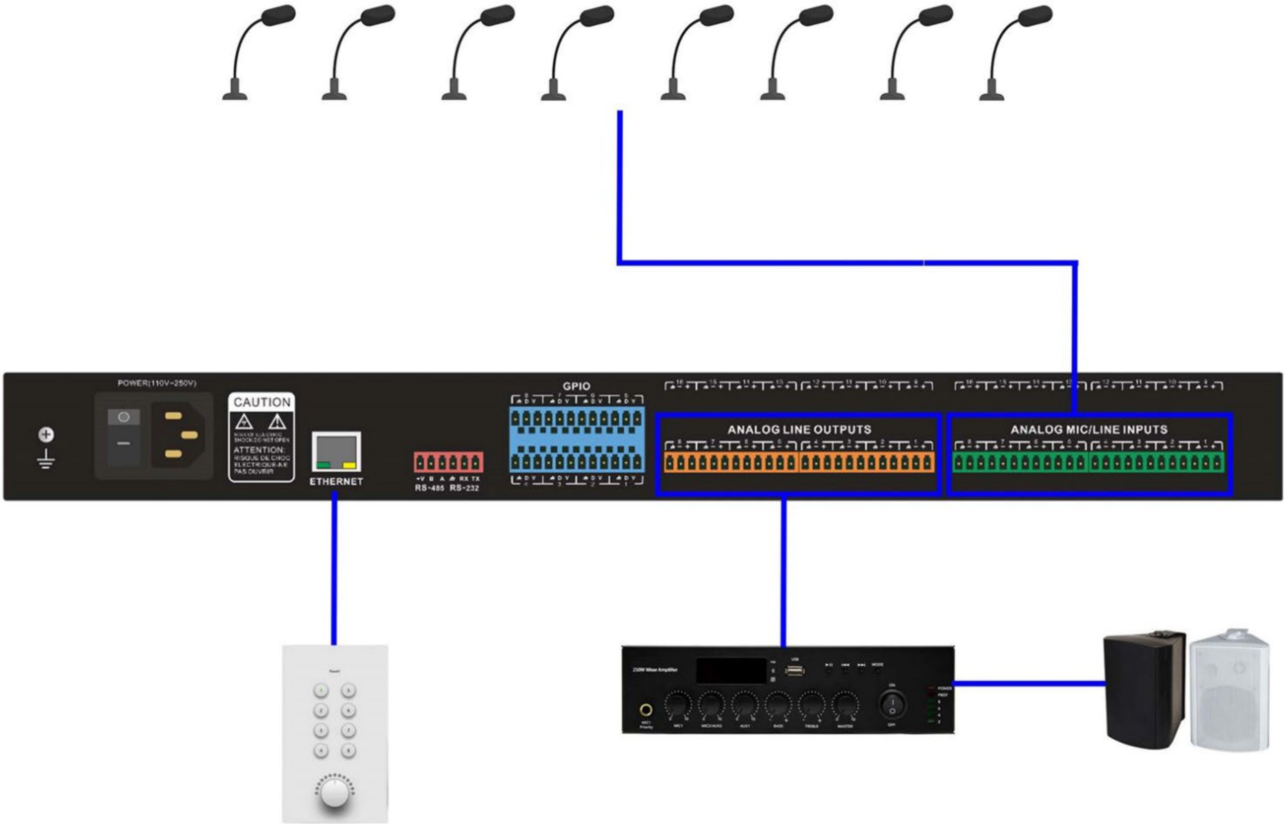
What's the practical significance of floating-point processing for the users? The gain stages between multiple modules can be ignored. If the signal level of a module is reduced by 50dB and is then restored to its original value through another processing, data loss will not occur. In the fixed-point system, the users must check other signal levels before sending it to the A/D converter because all digital-to-analogue converters adopt fixed points. In the DSP system, if you notice that your signal has been clipped before it is outputted and transmitted to the digital-to-analogue converter, you may close it off immediately at the output section to correct the situation. If using the fixed-point system, you would have to search each processing module to find the clipping source.

2.5 Audio Flow



2.6 Typical System Application

Conferencing System: The processor can be connected to microphones, amplifiers, speakers, and other audio peripherals (like USB audio from a PC) so as to process all audio signals and advanced algorithms like AEC in a typical conferencing system. Additional audio matrixing, auto mixing and signal routing is also achieved in this application to facilitate excellent audio quality in a conference.



3. Software

3.1 Software Installation

A Windows PC with:

- Intel i5 processor or higher
- 8 GB or higher memory
- 1 GB free storage space
- Windows 7 or higher version
- Minimum 1920 x 1080 resolution
- 24 bit or higher color
- Network (Ethernet) port

Software Installation:

Visit the device web page address: 169.254.10.227

1. Download software from the product website and install files.
2. Double click the downloaded file and install by following the instructions on the screen.

After the software is installed, read other parts of the help file, or execute the software.

After the software is installed, use one of the following methods to open the software:

1. Double Click Desktop Icon
2. Open from Start menu

3.2 Using the Software

After enabling the software, the main menu is shown as below:

Click the button  in the right corner of the main menu to find all processors on the network automatically. The user may connect to the designated processor based on their own needs; after the connection, the indicator will flash; each processor supports simultaneous connection and control for up to four users.

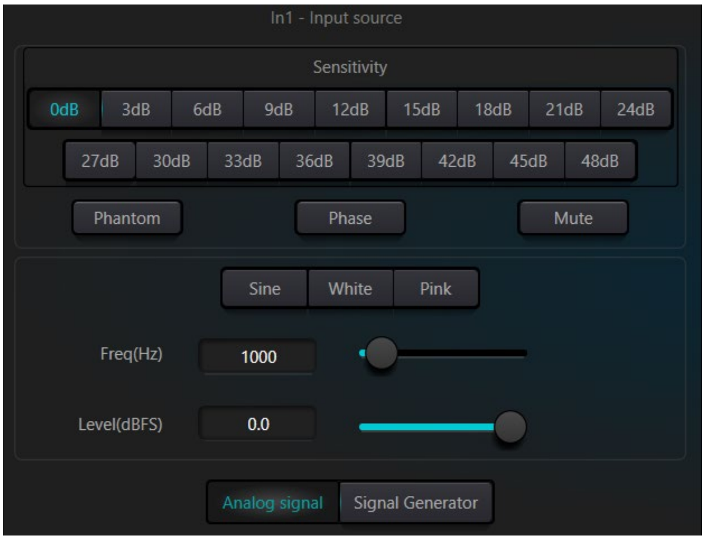
3.3 Custom edit processing module

Click on the edit button, input or output channel processor module right-click selection, edit dialog box, can replace the current processing module, can delete, copy and other operations, edit good click upload to the host. Note: when the CPU display more than 100 will turn red, this time the resource can not be uploaded to the host, need to be re-edited.

3.4 Audio Module Parameters

There are two ways to access the audio module parameters: first, click the input or output channel module you wish to access, and enter the parameter interface of the module; secondly, right click the channel module and the configuration interface will pop out. This is used for the following module parameters.

3.4.1 Input Source



Sensitivity: Microphone gain, 0dB - 48dB in 3dB Steps.

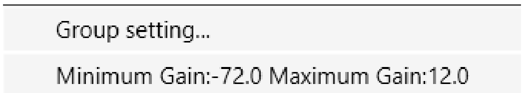
Phantom Power: Provides +48V phantom power for external condenser microphones. (Do not enable phantom power when the power is not required, this is to prevent damaging any non-Phantom powered devices when connected.)

Sine Wave: Select the Sine Wave button and drag the frequency to generate sine wave with selected frequency (selectable 20Hz - 20 kHz). You may regulate the output level (unit: dBFS) based on your requirements. Use the level fader to adjust or click the text field to designate a value.

White Noise: When observed on the frequency spectrograph with constant bandwidth, and set to a flat frequency spectrum, white noise signal has equal energy across all frequencies.

Pink Noise: The frequency intensity level of pink noise is mainly distributed in the middle and low frequency bands. It decreases with a speed of 3dB/Oct in the middle and low frequency bands.

In addition, you may also find the following menu by right clicking each fader on the main menu.



Group Setting: Open the group setting interface window.

Minimum and Maximum Gains: Limit the maximum and minimum values of a channel. After it is commissioned, if you do not wish that the system's stability is affected due to external factors, you may set up a maximum gain.

3.4.2 Expander

Although similar in concept, the expander has a different operating principle from the compressor. It can expand the dynamic range of a signal. The most fundamental difference in these two devices lies in that, the compressor works on the signal higher than the threshold, while the expander works on the signal lower than the threshold. Louder and softer signals become relatively louder and softer respectively. It can be seen from Fig.3.2 that, when the expansion ratio reaches 1:2, the input signal 20dB lower than the threshold will generate an output signal 40dB lower than the threshold. Thus, as shown below, the signal lower than the threshold will extend downwards and cause softer level. When an expansion ratio 1:20 is adopted. The expander seems to be a noise gate in terms of the transmission features. In fact, a noise gate is an expander with a great expansion ratio.

Fig.3.2 Expander

The expander has the following control parameters:

Threshold: The expander activates only when the signal exceeds this threshold (allowing the transmission of the signal). As a standard practice, the signal is often set at the ambient noise level

Ratio: Ratio refers to the slope below the threshold point on the gain curve. When the slope is set at a high level, gating will activate.

Attack: Attack refers to the time that the expander will wait before activating when the input signal exceeds the threshold. Shorter attack time allows to start the expander more quickly.

Release time: Release time refers to the delay in time required for the gain to be restored to normal when the input signal drops lower than the threshold.

No matter the starting time or the release time, it just helps to reduce the speed of gain attenuation. That is to say, the speed of the gain from -40dB to 0dB is slowed down due to the influence of Attack. The Attack time and Release time is unrelated to the threshold. If the signal level falls below the threshold, the Attack time and Release time will have their own respective influence on gain attenuation; when the signal level rises above the threshold, the gain attenuation produced by the expander will disappear in accordance with the speed controlled by the Release time. When the gain attenuation is reduced to 0dB, the expander will stop expansion. Later, when the signal reduces to below the threshold, the expander will start again, and the release time will begin to work.

3.4.3 Compressor & Limiter

Compressor

The compressor is used to reduce the dynamic range of the signal higher than the threshold set by the user, and to maintain the dynamic range of the signal lower than the threshold. The compressor has the following control parameters:

Threshold: When the signal level is higher than the threshold, the compressor / limiter begins to reduce the gain. Any signal that exceeds the threshold is regarded as overshoot signal, and its level will be reduced based on the ratio set. The more the signal level exceeds the threshold, the more level is attenuated.

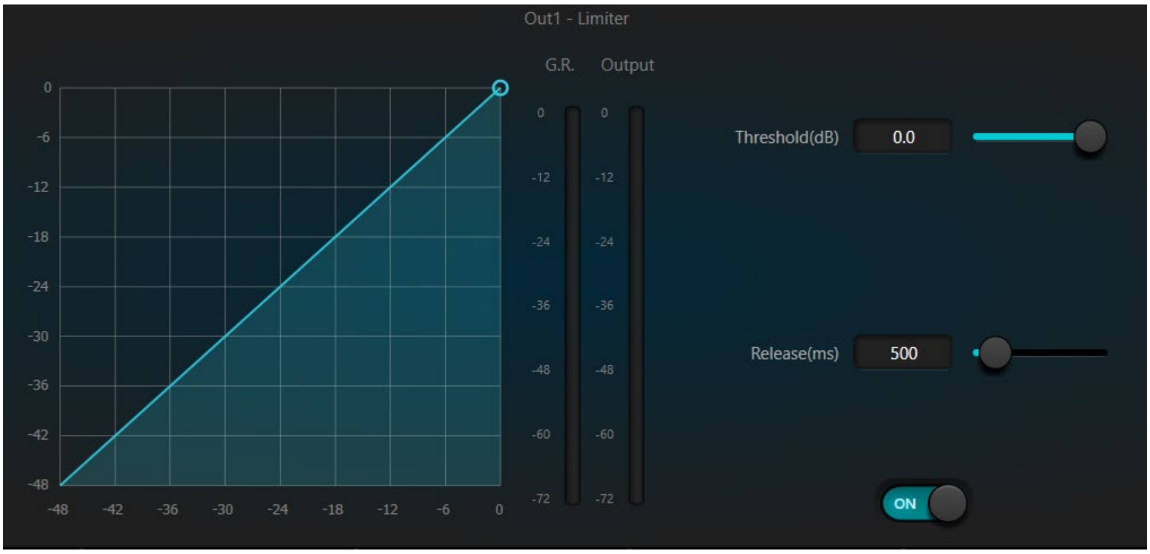
Ratio: It refers to the compression ratio. The ratio sets the attenuation degree of the overshoot signal to the threshold level. The smaller the compression ratio is, the less compression will take place and signal will remain higher than the threshold. Once the signal exceeds the threshold, the compression ratio decides the ratio of input signal variation to output signal variation. For example, when the compression ratio is 1:2, if the input signal is 2dB higher than the threshold, the exceeding part only changes by 1dB. A compression ratio of 1:1 suggests that the compressor does not attenuate the signal in proportion. The adjustable range of compression ratio is 1-20.

Attack Time & Release Time: In order to maintain natural oscillation, it is generally accepted that part of the original level will pass through the compression without any influence, or just minor influence. Likewise, if there is rapid sharp attenuation and rapid recovery in the signal gain, the suction effect will occur. The attack and release time of the compressor is to avoid this effect. The attack time sets the speed of gain attenuation, while the release time sets the speed of gain recovery.

Output Gain: Also called gain compensation. If the compressor significantly reduces the level of the signal, it may need to enhance the output gain to maintain the volume. Output gain applies to all parts of the signal and is unrelated to other parameter settings of the compressor.

Gain Reduction (G.R.) and output Level Meter: G.R. indicates the compressor's compression amount; output refers to the output level of the signal that has passed through the compressor module (post compression). The compression amount is displayed in an inverse level meter. If the input signal and threshold are set as -6dB and -30dB, respectively, and the ratio is 2:1, then the compression amount is 12dB; the G.R. level meter indicates around -12dB and output indicates around -18dB.

Limiter



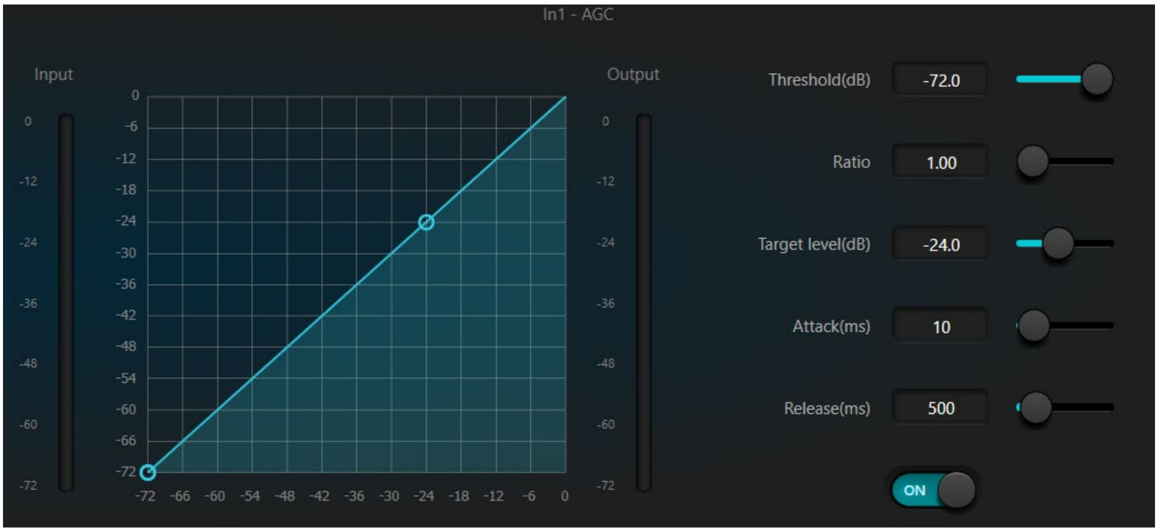
The limiter only has one key task: make sure that the signal will not exceed the threshold level in any way. By adjusting the compressor's control parameters, its working modes will be very similar to those of the limiter. The core working principle of a limiter is that it focuses on the signal below the threshold level as well as how the gain attenuation is produced before the occurrence of overshoot signal. The limit period consists of two processing stages: during the first stage, there is a minor limit, but the overshoot signal will not be processed; during the second stage, if there is overshoot signal, it will attenuate with a very high ratio.

The limiter only provides two parameters: Threshold and Release Time. In terms of signal processing, occasional clipping will be solved with a limiter, while the signal level will be attenuated in terms of frequent clipping.

3.4.4 Automatic Gain Control

Automatic gain control (AGC) is a type of compressor. Its threshold is set at a very low level with middle-to-slow attack time, long release time, and low ratio. The purpose is to improve the signal with an uncertain level to a target level, while maintaining the dynamic range at the same time. Most of the auto gain control includes silent detection to prevent the gain attenuation loss during the silent period. This is the only function that distinguishes auto gain control from an ordinary compressor / limiter.

Auto gain control may be adopted to normalize the level of CD players that play background music, foreground music and music on hold, as to eliminate the changes in the level of some paging microphones.



Auto gain control includes the following control parameters and switches:

Threshold: When the signal level is lower than the threshold, the input-to-output ratio is 1:1. When the signal level is higher than the threshold, the input-to-output ratio changes with the ratio control settings. The threshold is set as the background noise just higher than the level of input signal.

Ratio: It refers to the ratio of the changes in the level of the input signal higher than the threshold to the changes in the level of the output signal.

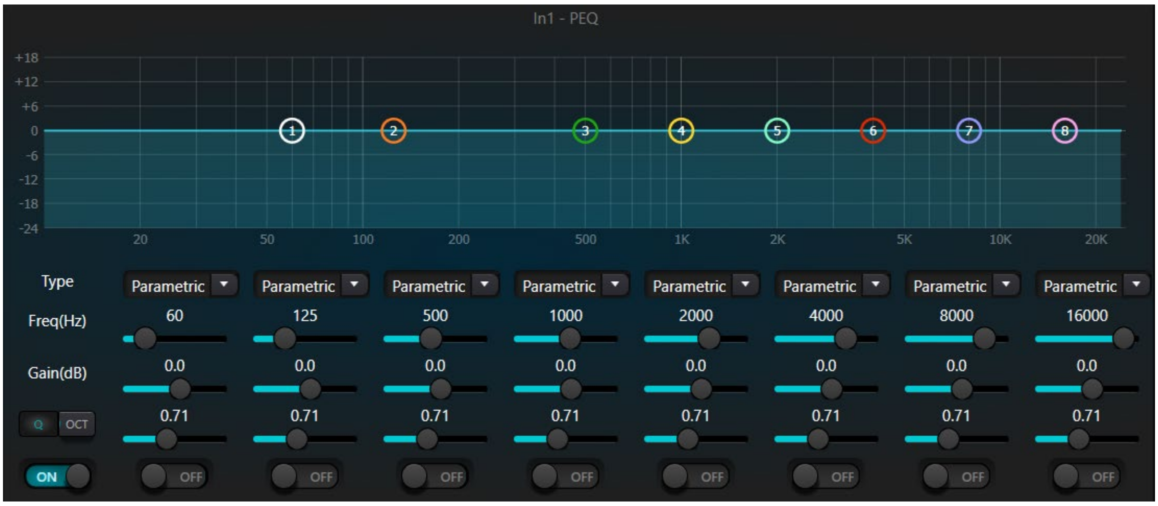
Target Threshold: It refers to the level of output signal required. If the signal is higher than the threshold, the controller will compress the signal in proportion.

Attack Time: It refers to the response time required to control the level higher than the threshold.

Release Time: It refers to the response time required to control the level lower than the threshold.

3.4.5 Equalizers

The equalizer is mainly used to correct the frequency range that is overemphasized or gets lost, no matter if it is wide or narrow. In addition, the equalizer can also help us to narrow or widen the frequency range or change the amount of a component in the frequency spectrum. To simplify, the equalizer can be used to change the tone of the signal.



The equalizer has the following control parameters:

Type: Parametric EQ is the default setting. High and low shelf filters and high and low pass filters can also be selected. Each kind of filter has different forms to achieve different functions.

High and Low Pass Filter: The reference frequency of a pass-type filter is called the cut-off frequency. Pass-type filters allow the frequencies on one side of the cut-off frequency to fully pass the filter; in the meantime, the frequencies on the other side of the cut-off frequency are attenuated at a constant dB ratio per frequency octave. High pass filters allow the frequencies above the cut-off frequency to pass and filter the frequencies below the cut-off frequency. To the contrary, low pass filters allow the frequencies below the cut-off frequency to pass and filter the frequencies above the cut-off frequency.

High and Low Shelf Filter: High shelf filter means that the gain increases or attenuates for the frequencies above the set frequency. Low shelf filter means that the gain increases or attenuates for the frequencies below the set frequency. The set frequency is not 3dB cut-off frequency but refers to the center of the failing edge or rising edge of the filter. Q value affects the peak and has a mathematical relation with the peak.

Frequency (Hz): Refers to the center frequency of the filter.

Gain (dB): Refers to the increased or attenuated decibel value of the gain at the center frequency.

Q: It refers to the quality factor of a filter. The adjustable range of the Q value is 0.02-50.

When the filter is set as a parametric EQ filter, Q value refers to the width of the bell-shaped frequency response curve on both sides of the cut-off frequency.

When the filter is a high and low shelf filter or a high and low pass filter, if Q>0.707, there will be peaks in the filter responses. If Q<0.707, the slope will become flatter, and the roll-off will occur in advance.

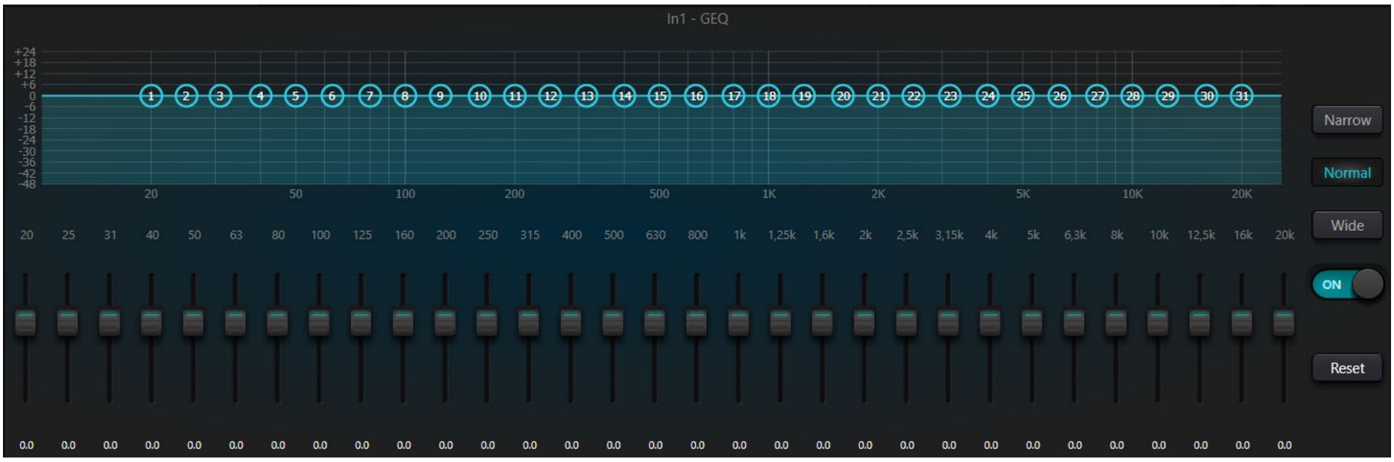
Each segment of the equalizer has a switch, which is used to turn on or turn off the corresponding segment. When turned off, that frequency parameter settings are disabled.

The equalizer has a master switch, which is used to enable or disable the whole module.

3.4.6 Graphic Equalizer

By using a constant Q-value, each frequency point is equipped with a push-pull potentiometer. The bandwidth of the filter remains unchanged regardless of if the frequency is raised or attenuated. The common professional graphic equalizer is to divide 20 Hz~20 kHz signals into 10, 15, 27 or 31 bands / frequencies to adjust.

The graphical equalizer has 10, 15 and 31 band options



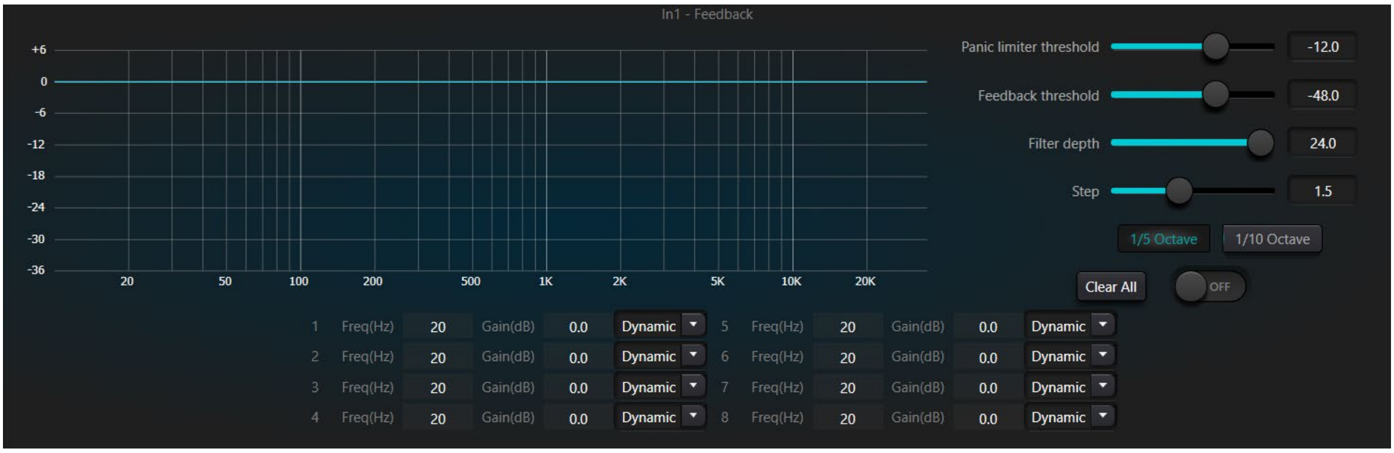
3.4.7 Feedback Suppressor

While using the feedback suppressor module, it is advised to remember that feedback suppression is not a replacement for a good audio system design and commissioning. Traditional audio practices, such as limiting the number of open microphones, minimizing the distance between sound source and microphone, positioning the microphone and loudspeaker to get minimum feedback, and balancing the room to get a flat response, is the first step in setting up a good audio system. Later, we can adopt feedback suppression to get additional gain. Feedback suppression cannot be used to magically solve a system's design defects or improve the sound transmission gain in a way exceeding the system's physical limitations.

The feedback suppressor module automatically detects and inhibits audio feedback in the sound system. The module distinguishes feedback from expected sounds based on the characteristics of the audio signals. When feedback at a certain frequency is detected, a notching filter will be automatically added at the feedback point (frequency) to attenuate the signal level at that frequency. During the first addition, the notching filter only attenuates the feedback a bit. If the feedback persists, the notching filter will continue to attenuate the feedback in accordance with the preset parameters until the feedback disappears or reaches the maximum preset parameter. Multiple user parameters can be used for accurate fine tuning of the effects of the feedback suppression module.

The Feedback suppression filters may be locked up to prevent any change during operation. Alternatively, the filter settings can be copied to a dedicated notching filter module (such as the parametric equalizer). The Eight filters can be set as auto filters in an automatic cycle. In this way, those filters for temporary use can be removed.

Each input channel has a feedback suppression module. Use a mouse to navigate the input module and find the feedback suppressor module or quickly enter the feedback suppressor module by clicking the shortcut key on the right. If the feedback suppressor module needs to be enabled, activate the module to automatically detect the feedback point, and use a narrow-band filter for elimination. Each feedback suppressor module has 8 narrow-band filters.



The feedback suppressor module has the following adjustable parameters:

Panic Limiter Threshold: According to this parameter, any level higher than the threshold is absolutely “feedback”. When a signal level is higher than the feedback threshold, any of the following circumstances will occur; a) the output gain is temporarily attenuated to control the speed of feedback; b) the output level is restricted to prevent out of control feedback; c) the filter's sensitivity is increased for faster detection and feedback. Once the output level is lower than the threshold, the gain will be recovered, and the sensitivity is restored to normal state. This value refers to the peak value of the digital range signal. If the value is set as 0, this function is disabled.

Feedback Threshold: According to this parameter, "any level lower than the threshold is absolutely not feedback". This may prevent the module from detecting feedback in a soft music or due low noise level.

Filter Depth: It refers to the maximum attenuation of a single filter. A shallow setting may prevent excessive frequency or signal degradation caused by the notching filter to the original signal. A deep notching filter may cause worse feedback control, especially in a large narrow resonance system.

Bandwidth: 1/10 and 1/5 Oct options can be chosen. A constant Q value is adopted. The filter will not become wider due to the increase of depth. It is recommended to use the filter in the phonetic environment. In the case of frequent feedback, the bandwidth is set at 1/5 Oct because it has wider bandwidth and greater influence.

Notching Filter Mode: Each notching filter has two modes: Dynamic, or manual mode. When 'Dynamic' mode is set for the notching filter and with all eight filters in use, the feedback suppression module will "check" the dynamic filters and re-use an available filter to inhibit the new feedback detected. When Manual mode is set for the filter, the gain and frequency is manually set by the user.

Clear All: Click the button to instantly clear all filters. It will clear up all feedback points detected previously. This operation is generally done when recommissioning the feedback module.

Feedback suppression can be used as a tool during system commissioning to identify feedback points or as a preventive measure during normal operations. If you want to get higher system transmission gain and feedback suppression, it is recommended that you debug the system by following the steps below:

(a) Reduce the system gain, and use the "Clear" button to reset all filter parameters

(b) Set up fine-tuned parameters for the feedback suppressor module. Also, decrease the panic threshold to reduce the feedback level.

(c) Open all microphones, and slowly increase system gain until feedback occurs. Stop increasing system gain when the feedback occurs.

(d) Wait for the feedback suppressor module to take effect; after the feedback disappears, continue to increase gain.

(e) Repeat the operation until the system reaches the required gain or until all filters are fully distributed

(f) Change the panic threshold to a maximum level just higher than the expected non-feedback signal.

At this time, if needed, you may set Fixed mode for each filter or save the dynamic status to deal with possible feedback during the normal operation. Additionally, you may copy the filter to a notching filter module (such as the equalizer). In this way, you may add more feedback filter capacity.

If a speaker is included among the devices used, it is recommended to use a compressor / limiter module for additional protection. You may set an appropriate limiter to make sure that the speaker will not get damaged even if all notching filters are used up, or the feedback inhibitor cannot effectively control the feedback, such as in the case of excessive system gain.

3.4.8 Noise Gate

The main purpose of a noise gate is to attenuate signals below the set threshold, and to allow signal to pass normally when the signal level is above the set threshold.

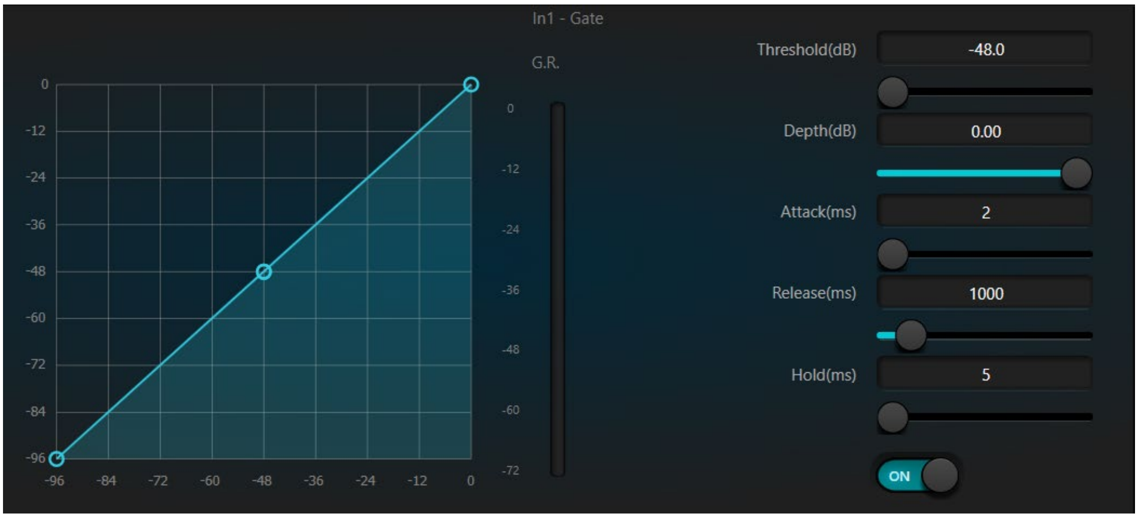
Threshold: Audio signal greater than the threshold setting is passed while audio signal less than the threshold is attenuation

Depth: The Depth determines how much the audio signal below the threshold is attenuated.

Attack time: Attack time refers to the time for the noise gate to open after the threshold has been passed.

Release time: The release time is opposite to the attack time and refers to the time taken for the noise gate to close.

Hold time: The hold time determines how long the noise gate remains open after the signal has dropped below the threshold.



3.4.9 Ducker

A ducker is used to attenuate or cut the level of the channel based on signal level from a secondary signal input or channel. When the level of the reference channel exceeds the specified threshold, the level of the specified channel will be attenuated, in essence 'ducking' the signal level.

Threshold: The reference signal begins to decay above the threshold and recovers below the threshold.

Depth: The amount the signal is reduced by when the ducker is activated.

Attack time: Attack time refers to the time taken for the attenuation to activate after the threshold has been passed.

Release time: The release time is opposite to the attack time and refers to the time taken for the attenuation to be released after the threshold is no longer passed by the reference signal.

Hold time: The hold time determines how long attenuation remains after the signal has dropped below the threshold.

3.4.10 Ambient Noise Compensation (ANC

An ambient noise compensator is used to automatically adjust output volume according to a reference signal input. When the reference signal increases or decreases with changing ambient noise picked up by a reference microphone, the output volume of the specified channel can be increased or decreased based on the set parameters.

Maximum gain: Sets the maximum amount to which the signal level can be adjusted.

Minimum gain: Sets the minimum amount to which the signal level can be adjusted.

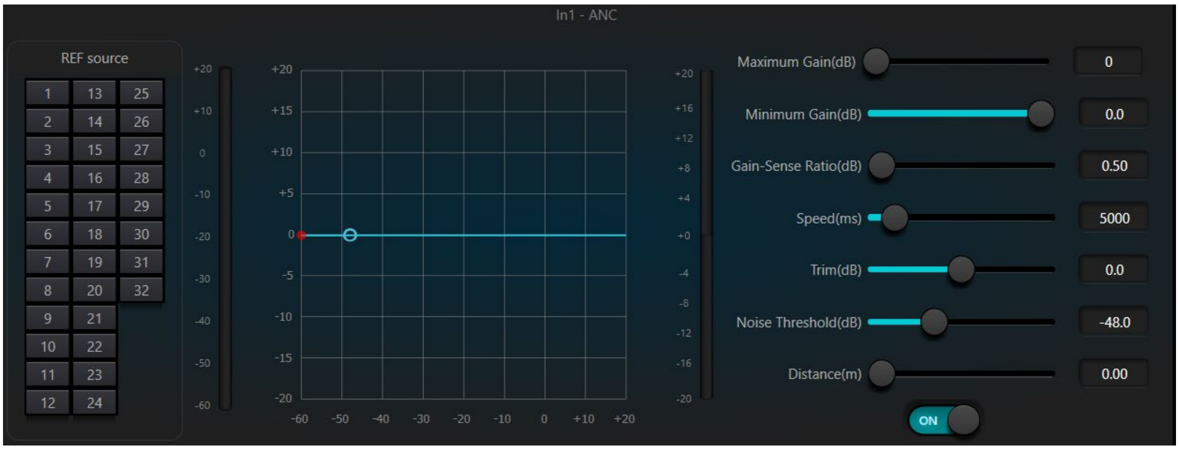
Gain sensing ratio: Sets the ratio by which signal level is attenuated or gained.

Speed: The speed at which the signal level is attenuated or gained.

Trim: The amount of gain or attenuation the output signal is adjusted by.

Noise threshold: The threshold / mean dB at which the ANC will activate and adjust signal. Signals greater than the threshold will be increased, while signals less than the threshold will be reduced.

Distance: The physical distance between reference signal and local signals.



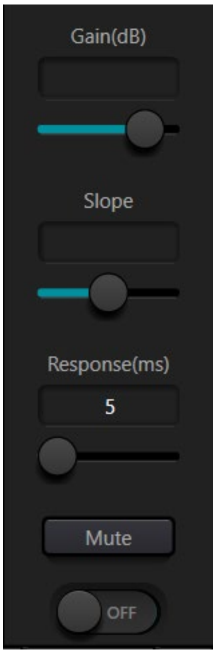
3.4.11 Auto Mixer

In a conference room, if several microphones are opened to the same gain level and there is only one person speaking, the audio pickup of the microphone may be not clear. Other microphones will pick up noise and reverberation in the room. When these signals are mixed with the desired ‘speech’ signals, audio output quality may be greatly reduced and become un-intelligible. In addition, the whole audio signal chain may be over energized by excessive gain and could start feeding back, result in a feedback loop, or ‘ringing’ sound. To solve the issue, other unused microphones should be closed or ‘muted” when not in use. The auto mixer is used to automatically open and close microphones based on signal input level.

There is a built-in gain share auto mixer inside the processor. It supports up to 8-channels. There is also a direct output at each channel of the auto mixing matrix, which is only influenced by channel mute and bypasses the auto mixer functions like auto gain and channel fader. Channels suitable for fixed volume like background music need to be kept at a fixed level without being controlled by the auto mixer. For example, it will keep a microphone normally open. Its gain will then not be influenced by the auto mixer. At this point, users may directly adjust the output of the channel in the matrix router as well as turn off the auto mixer button of the channel. Its gain will not be adjusted and gains at other channels will not be influenced by the signal level at the channel.

There are two groups of control parameters in the auto mixer module: main control parameters and channel control parameters.

(1) Main (Global) control parameters



Click the ON / OFF button to turn the auto mixer on or off.

Gain: Controls the main output volume of the auto mixer

Slope: The slope control influences the attenuation of lower levels. If the slope is high, the attenuation of lower-level channels will rise. The slope control and the ratio control at the expander has the same working mode. It is suggested that the value be set at or around 2.0. If it is set at 1.0, the effect is equal to closing the auto mixer at all channels. If it is set at 3.0, the action will result in larger gain adjustment, which may bring unnatural volume levels. The bigger the value, the more the channel is opened and the more the total attenuation. When the slope is set at 2.0, ideal gain sharing may be realized, so it is the preferred value to use.

Response Time: Faster response time may ensure that the first letters of spoken words are not cut off. Slower response time allows smoother operation. Practice shows that the best effect will be produced when response time is between 100ms and 1000ms. The design of auto gain aims for faster microphones switching. Therefore, first letters of spoken word will not be cut off even if the response time is set to 100ms. If it is set at several seconds, then there will be a longer hold time of the response time of the auto mixer, previous active channels will be saved at open status for several seconds.

(2) Channel control parameters

Auto Mixer: Each channel has an auto mixer on / off button which must be turned on for that channel to participate in the auto mixer. When in Off mode, that channel will not participate in the auto mixer.

Mute: Both the channel mute and volume fader are post the auto gain processor. If the channel level is higher, the level gain of other channels may also be reduced even if the channel mute is on.

Gain: The Gain fader adjustment may increase / decrease the volume amount of that channel in the auto mixer.

Priority: Priority settings may give priority treatment to high priority channels over low priority channels, and therefore the auto mixer algorithm will be affected accordingly. Priority ranges from 0 to 10. The higher the value, the higher the priority.

Both the channel mute and fader are post auto gain. Any adjustment made towards these two parameters won't influence the operation of the auto mixer. For example, If the channel level is higher, the level gain of other channels may also be reduced even if the channel mute is on. Channel mute must be turned on and the auto mixer shall be turned off to set signal mute and prevent its influence on the auto mixer. The mute button at each channel and directly controls output mute when mixing sound. Channel faders also control sound mixing level and direct output level of channels. Click the textbox and input a dB value to control channel level precisely.

Priority control allows high priority channels to override low priority channels, and thus the auto mixer algorithm will be affected. Priority value can be set from 0 (the lowest priority) to 10 (the highest priority), and the default value is 5 (standard priority). Users may use the slider or click the textbook to input a specified priority between 0 and 10. Increasing the value means increasing priority.

If two channels have the same signal level, then the channel with higher priority will receive more auto gain. If there is one-unit difference in priority level between them, then the channel with the higher priority level will get an extra gain of 2dB (supposing the slope of the two channels are set at 2.0). For example, if channel 1 and 2's priorities are respectively set at 6 and 3, and the input level of those two channels are the same, then Channel 1 will receive 66dB gain more than Channel 2. During operation, please note that the slope setting of the main control parameters influence gain difference brought by the priority weight of channels. If the slope is set at 3.0, then one priority level difference will result in a gain difference of 4dB.

Note: In some cases, be very careful when using wide priority differences between channels, such as a priority of 0 and 10. If channels with ultra-high priority recognize signals like background music from a loudspeaker, then it is possible for them to mask channels with lower priority. The effect will get worse if the slope is higher too. If the issue occurs during installation and commissioning, users may consider installing a noise gate or expander between the auto mixers of the highest priority channels. Also, consider setting a threshold at a value that it won't be opened by the noise gate or expander.

3.4.12 Acoustic Echo Cancelation

Acoustic Echo Cancelation (AEC for short) is used in audio / video conferencing to facilitate full duplex communication. The AEC module increases the remote speaker's phonetic intelligibility by cancelling out or removing unwanted acoustic echo generated in the local room's audio system (between the microphones and speakers).

Echo cancellation module for remote calls can be used to carry out local amplification of remote voice signals and attenuate the interference caused by acoustic echo. Its basic operation principle is simulating echo channel, estimating possible echo generated by remote signals and then minus the estimated signal from input signal of microphones, and thus there will be no echo generated in input voice signal to achieve the goal of cancelling echo.

There is only one echo cancellation module in the DSP Controller. Two local inputs and remote output mixers are preset to realize multichannel signal participating in echo cancellation as shown in figure.

One parameter can be adjusted, namely the Non-linear filter (NLP). There are three preset filters to choose from: Conservative, Moderate and Aggressive can be selected to determine the acoustic echo suppression level.

Note: The settings of the Acoustic Echo Cancellation module can be used cooperatively with the matrix module to determine signal routes.

3.4.13 Noise Suppression

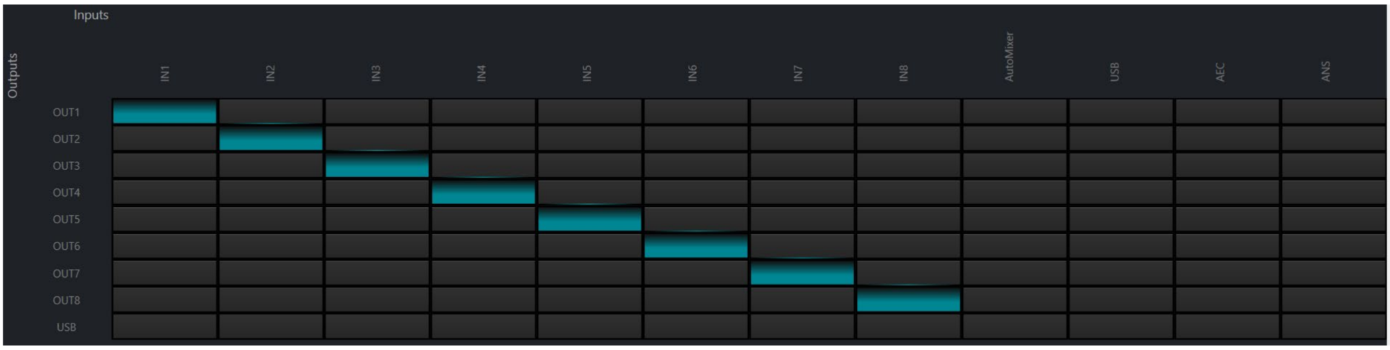
The Noise suppression module is used to remove non-human voice interference noise. It may distinguish human voice from non-human voice and treat the latter as noise. After processing, only human voice audio signal is passed to the output routing.

There is only one Noise Suppression module in the DSP Controller. Multichannel mixers are preset to realize multiple channels to participate in the noise cancellation processor as shown in the below figure.

Suppression level: There are three levels of noise suppression including: Mild (-6dB), Medium (-10dB) and Aggressive (-15dB). dB here refers to the reduction in decibel level of the suppressed noise. The bigger the value, the more degradation of voice signal is caused, therefore use noise suppression very cautiously.

3.4.14 Matrix

The audio Matrix has dual functions including routing and mixing. As shown in the figure below, the horizontal axis indicates the input channels, and the vertical axis indicates output channels. One-to-one routing of input to output is set as the default settings. As an example, to mix and route audio signals of channel 1 and channel 2 and then output both select OUT1 on both IN1 and IN2. If input 1 and 2 participate in auto mixing, then output 1 will not be influenced by the auto mixer. In other words, after setting auto mixing, echo cancellation and noise suppression modules, routing needs to be set in the matrix from the correct input sources, like auto mixer, AEC, or ANS to get the correct signal routing.



3.4.15 High and Low Pass Filter

Each output channel provides a pass-through filter module which consists of high-pass and low-pass filters. Each filter has four parameters to adjust as follows:

Frequency: This sets the cutoff frequency of each filter. The cutoff frequency of Bessel and Butterworth are selectable in -3 dB steps, and the cutoff frequency of Linkwitz-Riley are selectable in -6dB steps.

Gain: The Gain setting influences the increase and attenuation of each filter band.

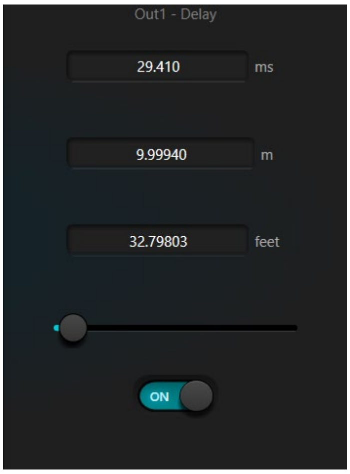
Type: There are three types of filters to choose from; Bessel, Butterworth, and Linkwitz-Riley. Butterworth has the flattest knee on the passband.

Slope: Slope refers to the attenuation values of the transition zone of the filter. There are a total of 8 attenuation values including 6, 12, 18, 24, 30, 36, 42 and 48dB/Oct. For example, 24dB/Oct indicates that the attenuation range is -24dB for each octave of frequency in the transition zone.

Select the On / Off button to activate or deactivate the high-pass or low-pass filters individually.

3.4.16 Delay

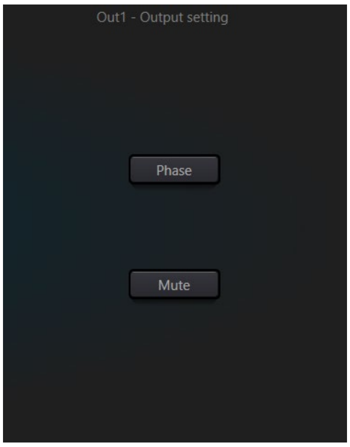
The Delay module is used to add delay to the output audio signal. This can be used to align audio signals in physical time.



Activate button: Turns On / Off the delay module and inserts it into audio signal path to increase fixed delay time for signals.

Millisecond (ms): Sets the delay time in milliseconds. The value ranges from 1 to 1200 milliseconds. Both meter and feet are alternate units provided to milliseconds to set a delay time based on physical distance.

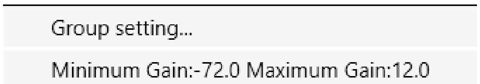
3.3.17 Output



Phase: 180-degree audio signal phase inversion.

Mute: Set output mute/unmute.

Users can use right button to set part menu at output channels, which can be carried out based on requirements.



3.4.18 USB interface

The USB interface is used to carry out two functions, recording and as audio interface for unified Communications using personal computers. After going through echo and noise cancellation modules, the USB interface may be accessed by unified communications software when connected to a computer.

① Song playing information, double click to enter playlist

② Next song

③ Pause

④ Song volume adjustment

⑤ Play

⑥ Prev. song

⑦ Sound recording list

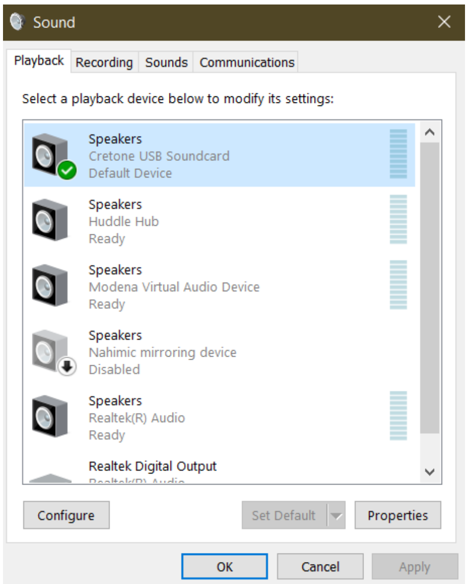
⑧ Sound recording volume adjustment

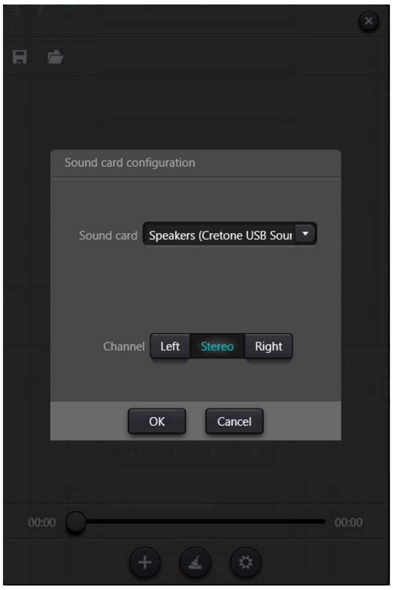
⑨ Stop recording

⑩ Start recording

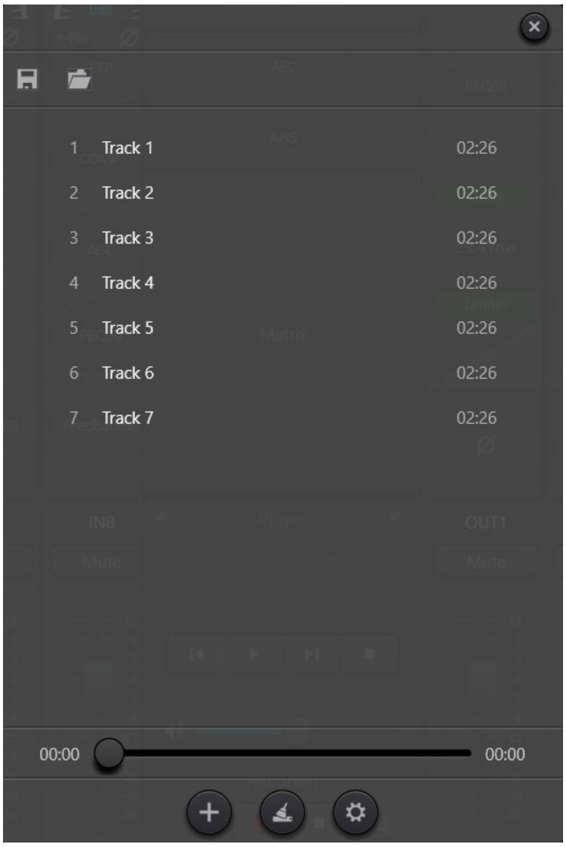
Soundcard Setting

USB cable with double ends of Type-A can be used to connect DSP processor and computer host. For initial connection, the "Found New Hardware" windows will pop up on the computer screen, and the driver will be installed automatically. After installation, the USB soundcard will be accessible in the computer soundcard list as shown in the figure below. Users may select 'Cretone USB Soundcard' in the Sound Settings panel of the computer for both Playback and Recording.





Users may also playback audio files loaded manually in the on-board playlist from the connected computer. Users may also directly open them when using the device next time. As shown in the figure, click at the bottom of the playlist to open file folder and select the audio files to be played, press to clear the playlist and to enter the settings of the USB interface.



3.4.19 Camera Tracking

Voice tracking threshold: Detects a microphone input signal greater than or equal to the tracking threshold, the system will automatically enable tracking parameters when threshold is exceeded.

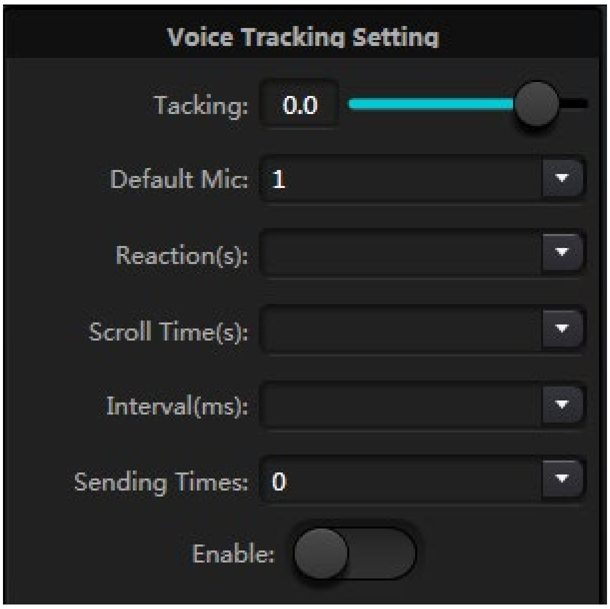
Default Mic: When no Mic input signal is received or tracking threshold is not exceeded, turn the camera to the default MIC position

Reaction time: The maximum intermittent time of the valid signal. If the microphone is used to speak, the reaction time is set to 3 seconds, and the signal is still considered to be valid for 3 seconds in the middle of the speech, and more than 3 seconds, is regarded as invalid.

Scroll time: The shortest speaking time required for the camera to switch to a valid position. If the microphone is used to speak, the speaking time must be longer than the "scroll time ". When the channel signal is considered to be valid, the camera automatically turns to the set position. Usually "Scroll time" is greater than "Reaction time ".

Interval: Sets the interval between each camera command sent**.**

Sending Times: Sets the number of times a camera command is sent.



Mic No.: Sets the microphone number; generally corresponding to the input channel of the device.

Priority: Sets the priority for the selected Mic No. (channel). If two microphones / channels activate at the same time, the higher priority level will execute the preset command.

*Preset number, serial port number, camera address and protocol, must correspond to the actual connection of the camera.

Enable: Turn enable on to enable the selected channel to form part of the Voice Tracking Camera control.

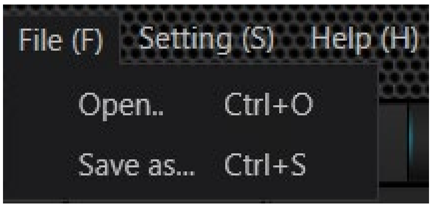
Save: Save the parameters to the device

Camera Settings

The camera settings section is used to set up the control parameters for each camera connected. Multiple cameras can be added, controlled and presets assigned accordingly.

3.5 Settings Menu

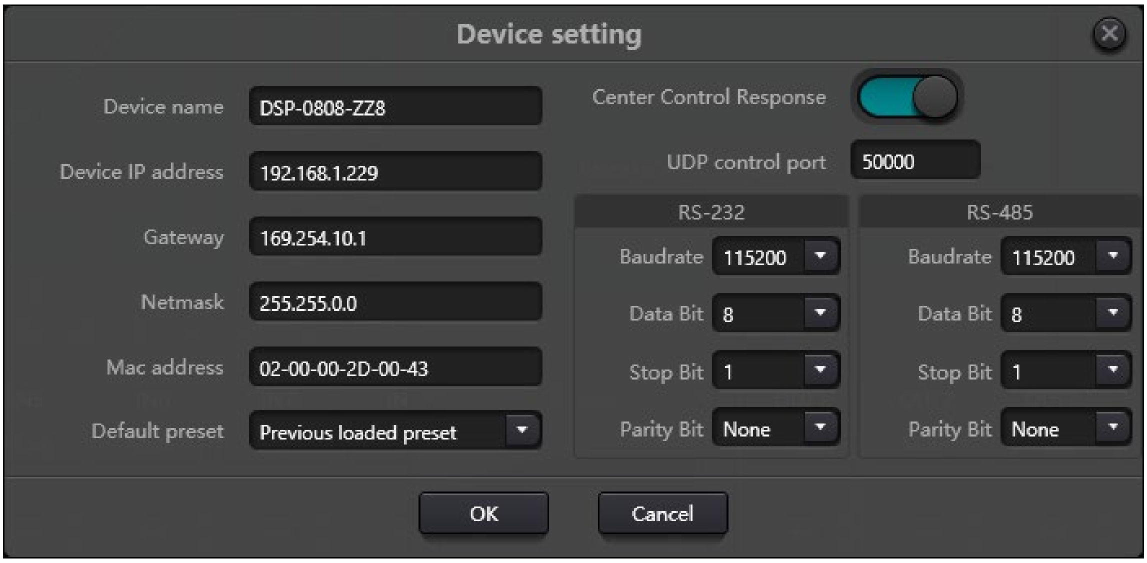
3.5.1 File Menu



In offline mode, click the pop-up file dialog and open an existed default document with suffix *.uma, or right click on the default document to open DSP.exe.

"Save as" refers to saving presets on the application to local hard risk to realize easy copy and store.

3.5.2 Device Setting



Information like device name, network address and serial baud rate can be set under device setting. The maximum length of the device name is 16 characters.

Default startup: Two startup preset modes are available for selection. One is any preset from 16 presets acting as startup preset. Each boot will start with it. Another is selecting previous upload preset and taking the last preset before power outage as the preset for next startup.

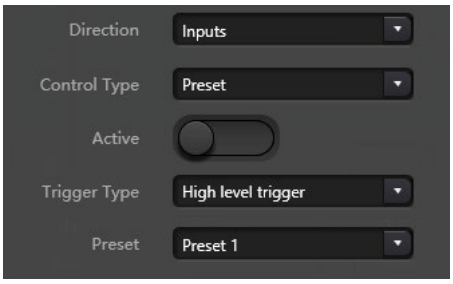
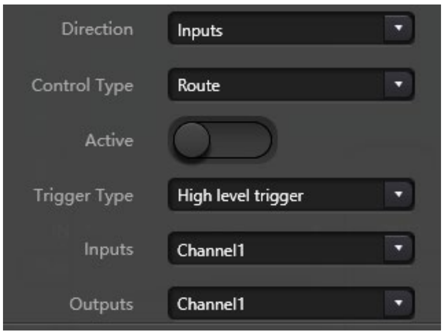
3.5.3 GPIO Setting

Open the software interface of GPIO settings. The device has total 8 GPIOs that allow independent input or output configuration.

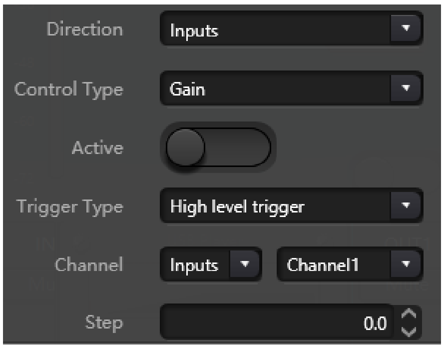
Input GPIOs have preset, router, gain, mute, command, and analogue-to-digital gain as control options.

Output GPIOs have preset, level, mute, and command for selection as control options.

Input GPIO Setting

Preset	 	**Trigger types:** High level trigger, Low level trigger, High level trigger / Low level trigger cancellation, Low level trigger, High level cancellation, i.e., rising edge / falling edge trigger / rising edge trigger, falling edge cancellation / falling edge trigger, rising edge cancellation **Preset:** It will change to preset when jump type of hardware GPIO port and trigger type of software setting are consistent.
Routing	 	**Trigger types:** Same as above **Input and output:** Select input channel routing corresponding to output channel. Executes mixing / cancel mixing action when the trigger condition is satisfied.

Gain

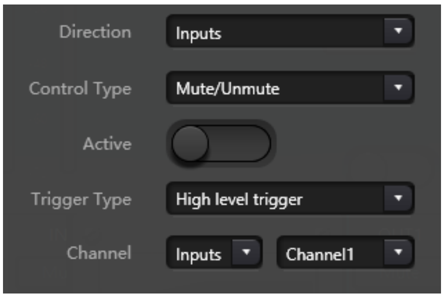


Trigger types: Same as above

Channels: Select input or output channels

Step: Level increase / decrease in 0.1dB steps based on original channel gain.

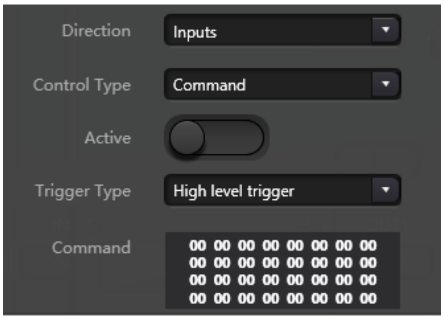
Mute/ Unmute



Trigger types: Same as above

Channels: Select input or output channel

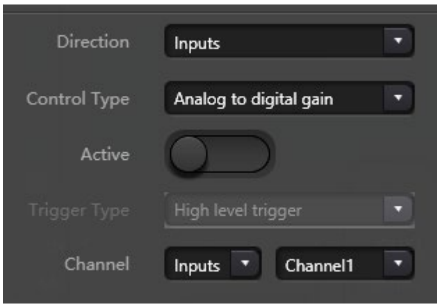
Command



Trigger types: Same as above

Command: The command code will be sent via RS232 when the trigger condition is satisfactory

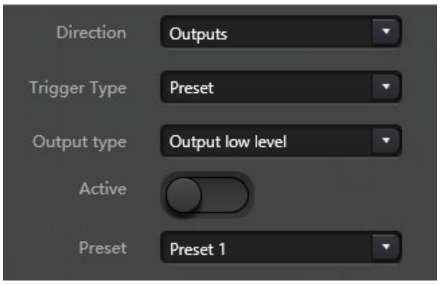
Analogue-to-digital Gain



Analogue-to-digital gain is very useful when connecting a potentiometer externally. It may adjust input or output channel gain. It looks like a rotary encoder. The difference between them is that a potentiometer is analogue and adjusts voltage and current while an encoder is digital and transmits binary codes of 0 and 1.

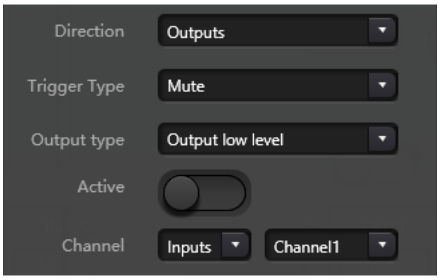
Output GPIO Setting

Preset



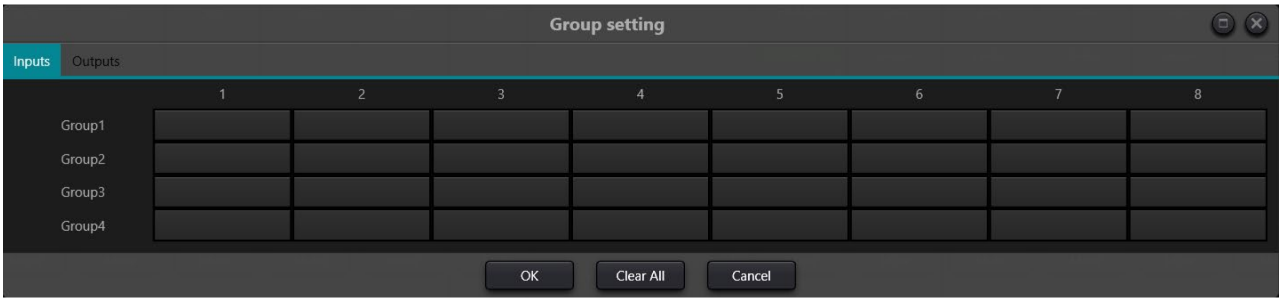
Output types: High level / Low level

Preset: Corresponding GPIO port outputs high level or low level when changing to it.

Level		**Output types:** High level / Low level **Channels:** Appointed input or output channel **Level:** GPIO outputs high / low level when appointed channel level reaches preset level threshold.
Mute		**Output types:** High level / Low level **Channel:** Appointed input / output channel. Preset high / low level is output when the channel is muted. On the contrary, the opposite level will be output when cancelling mute.

3.5.4 Group Setting

The interface for group assignments contains two sections including input and output labels. a Maximum of 4 groups are available. A channel can only participate in one group. In each same group, the channel volume adjustment and mute are synchronous. Other module parameters are not synchronized, which is the biggest difference between the Group assignments and the stereo 'Link' option.



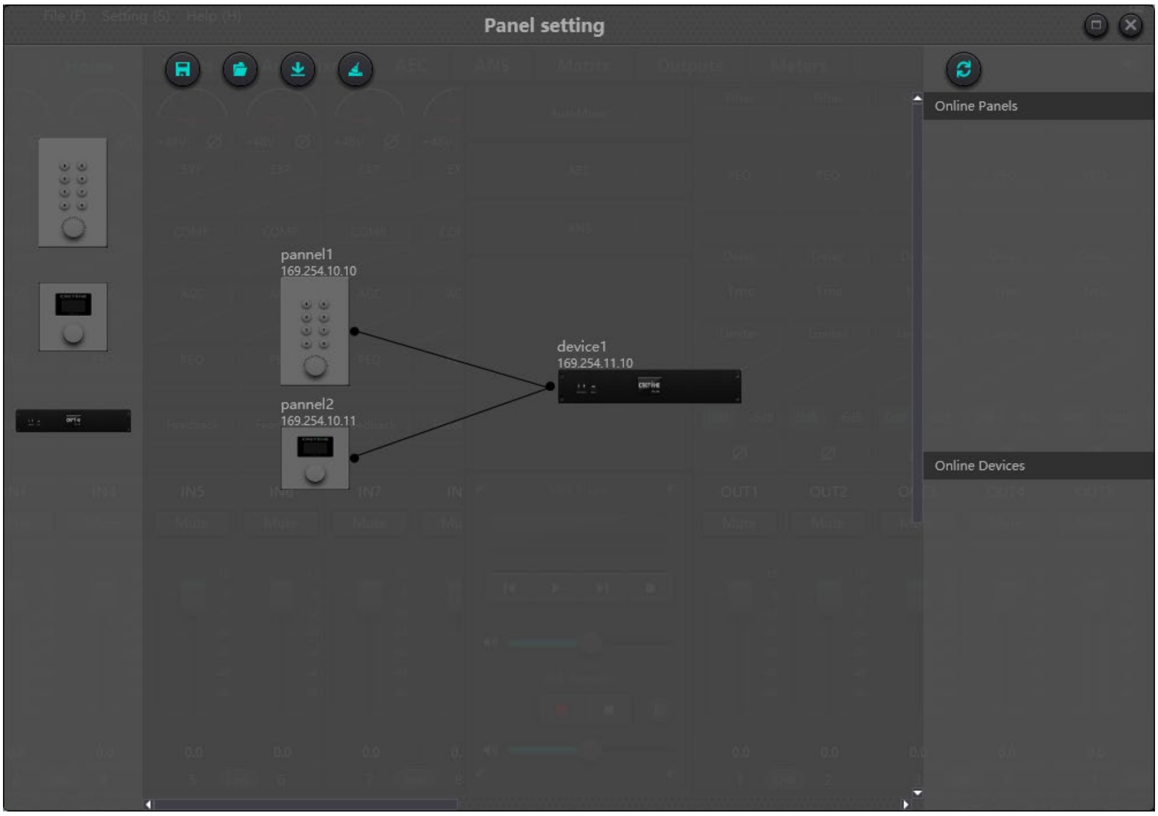
There are 4 groups in total. 1-device maximum number of channels can be selected for each group. The maximum number of channels is determined by the device type you purchased. Channels are set into one group which will be differentiated by a color difference of the volume slider in the main interface.

Group vs. LINK

The relationship between groups and link: The channel participating in a group will not participate in a 'LINK', which means group priories are higher than LINK. The difference between Groups and LINK is that Groups can only control channel gain and mute, while LINK links all parameters at the channel.

3.5.5 Panel Setting

Panel setting includes two panel types which are an 8-button panel with Volume control and an OLED panel. Use virtual cables to connect multiple physical online panels with the online DSP device via panel settings, to configure the panels for controlling selected DSP features.



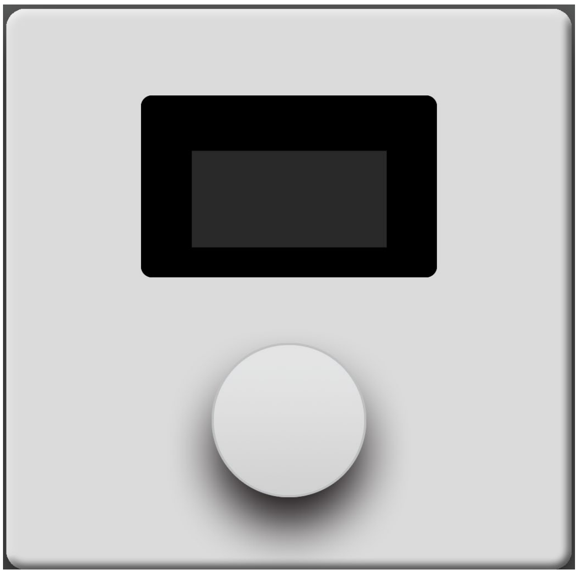
Offline device: In offline edit status, the commissioning engineer configures the panel parameters locally, and then downloads it to the online panel. The panel can then be edited directly online. Drag an offline device from the column of online panels to the panel design area and then double click to edit it.

Please note that there is a small circle on both panel and device. Click the circle and then draw a line, select target device, the connection between the two devices is established this way.

Double click the panel in design area to enter the panel's configuration interface. The configuration of the two panels are described below. After completing configuration, click the toolbar download icon to download the panel configuration to the hardware panels.

OLED Screen:

The OLED screen consists of a 1.3" OLED screen and Rotary encoder. The OLED screen display is organized according to the function assigned. There are three types of functions including menu, buttons, and presets. Double click the OLED screen in the design area to enter its setting as shown below.



Click "add menu" to access the pop-up menu selection box, choose the corresponding function and confirm it. After finishing the settings of the software menu configuration, click on the toolbar download icon to download the configuration to the hardware panel.

Operating the panel:

1. The panel displays name and IP address on main interface. Turn the Rotary Encoder left or right to change menus.

2. Press the Rotary Encoder to enter and display the second row of the menu. The interface starts to flash, which indicates that it entered edit mode of that menu.

3. Turn the Rotary encoder left or right to change value.

4. Press the Rotary Encoder again to exit the edit mode and go back to the main menu.

Key Panel:

There are eight keys and one Rotary Encoder on the key panel. The Rotary Encoder is used to adjust gains, and the eight keys can be used to achieve different functions through programming. There are four types of key functions, including volume adjustment, mute, preset, and command. Drag an item into the function area to assign the key and complete the programming steps of the key.

After completing all programming, users may use the emulation button to check the configuration of the panel is correct.

Panel operation and Indicators:

1. Button remains illuminated, indicating that it is configured with a mute function.

2. Button keeps flashing, which indicates that the button is configured with gain function. The configured Rotary Encoder adjusts the gain of the assigned channel. 13 indicators around the Rotary Encoder indicates gain level. They are on or off with gain level set. When all 13 indicators are off, it indicates a gain of -72dB while all on indicates a gain of 12dB.

3. A momentary flash when pressing the button indicates the button is configured for preset or command recall.

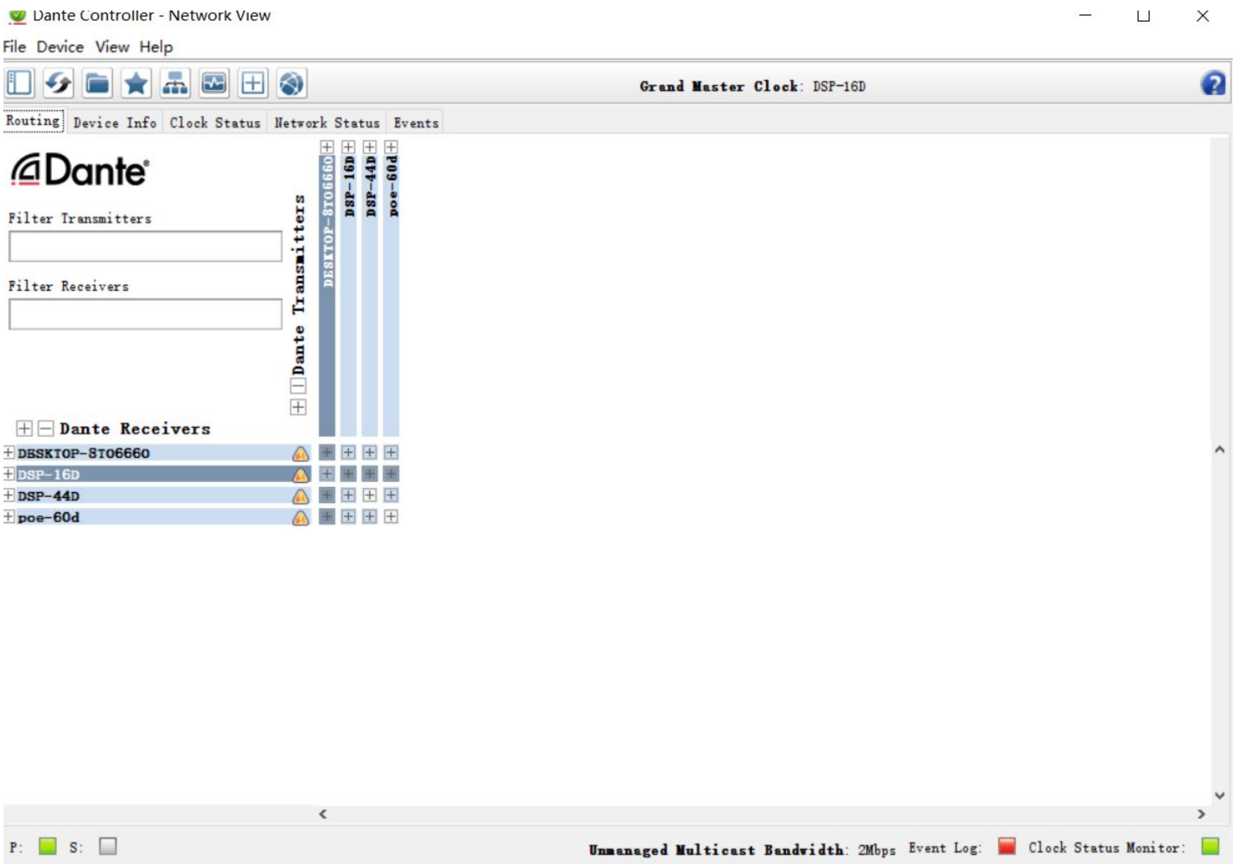
Command functions data strings comes from the central control command. Please refer to section 5 - Control.

3.5.6 Dante Setting

Note: Before accessing Dante settings, please check that the computer network port and Dante network port of the DSP has been connected to the Dante network. On the DSP software interface, you can only check the signal indicator of the Brooklyn card.

In order to set up a device lock or to check the signal indicators, please use Dante Controller.

Dante routing is also set up in Dante Controller. Dante Controller provides routing, channel information, network settings and other Dante network related information. The Dante devices receiving channels are displayed on the left part of the interface, while the Dante devices transmitter channels are displayed on the top part of the interface.



On the routing interface, the small boxes at the intersection of the transmitter and receiver channels indicate that a routing relationship can be created. A green tick icon will appear at the intersection of the matrix after a single click and once routing has been negotiated and a successful subscription is established. Users may see a grey icon for a very short time period while negotiating, which indicates that routing is in process.

A warning icon, or error icon will appear if there is a routing or subscription problem. If several devices are subscribed at the same time, a yellow icon may appear temporarily.

Note: Routed and locked devices cannot be moved, but existing routing can be deleted, or replaced.

Cancelling Audio Subscriptions

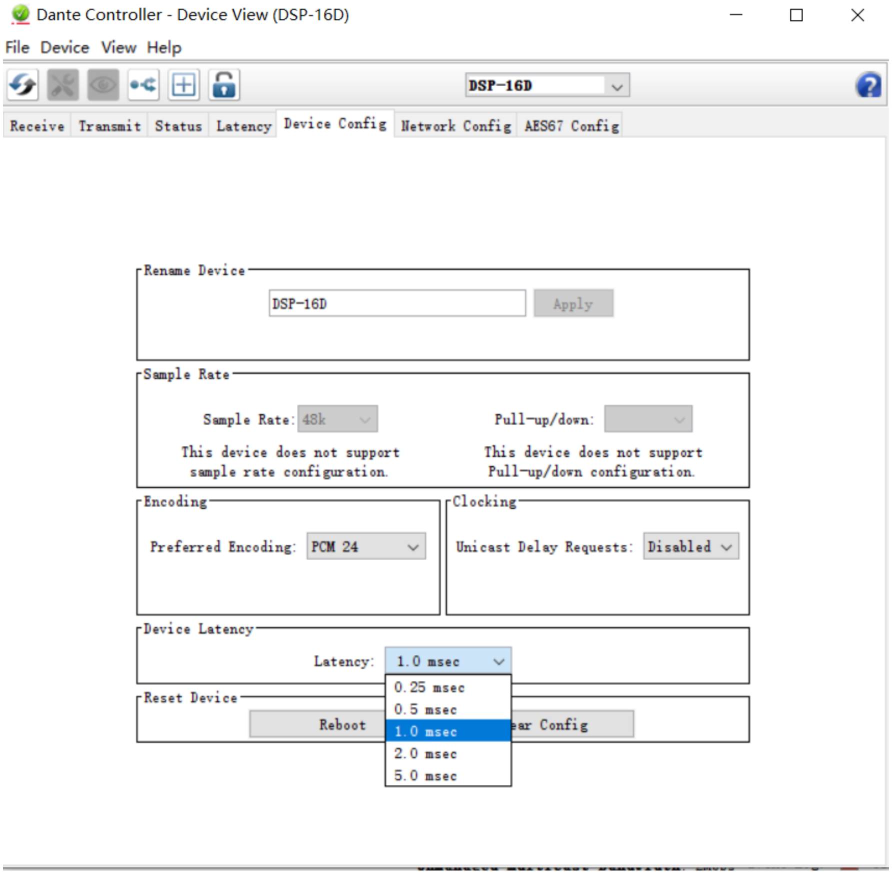
Users may click on a subscribed intersections to cancel audio subscriptions. The subscription icon will be removed, and a blank intersection block will be displayed.

Subscription Status

	Processing	Subscription is in processing
	Subscribed	Connection established
	Warning	Subscription is not processed normally because the transmitter device is not visible on the network
	Error	Transmission error, for example, not enough bandwidth on the network
	Coming soon	The device is processing subscription of other channels. In most cases, many channels are subscribed at once.

Users may view information like a device's IP address and Dante firmware version on the 'Device config' tab.

Double click device name on routing interface to enter detailed settings of the device as shown in figure below:



Channel names can be modified on the transmitter and receiver labels. Channel naming rules are as follows:

The maximum total length for all DSP device names is 16 characters. The name length of devices supported by Dante is as much as 31 characters. Therefore, ensure that the name length of Dante device names and channel names are no more than 16 characters when routing by using the interface, or the DSP Controller will cut the process off, which will result in incorrect subscription.

Names are not case-sensitive. "Guitar" and "guitar" are the same name.

Valid characters include A-Z, a-z, 0-9, and '-'.

Device names can't start or end with '-'.

Device names shall also be unique on network.

Any characters can be used for naming sending channel labels, except '=', ' 'and '@'.

Sending channel labels must be unique in the device.

Receiver channel naming has the same rules as transmitter channels.

Device Configuration:

Device configuration refers to device name modification, audio sample rate, and delay. Device names must be modified following the device name modification rules. Delay needs to be emphasized. If Dante network delay compensation is needed for various delays at receiving end, there is device setting delay (the interface delay) at each receiving end. The delay refers to the time difference between samples received at the receiving end and broadcasted from the transmitter end. The default delay for Dante devices are 1ms, which is adequate for large-scale networks.

However, automatic negotiation will be carried out at sending and receiving ends when establishing connections, which ensures delay time is negotiated to prevent packet loss.

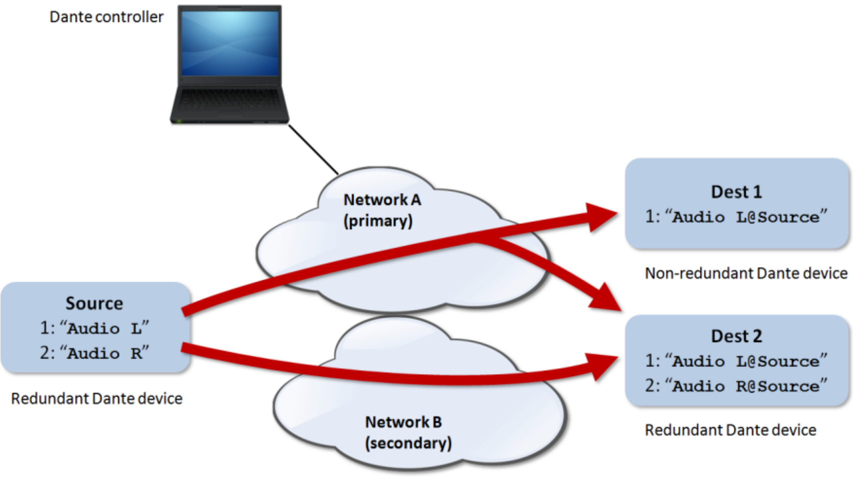
For example, Ultimo devices support minimum 1ms delay. If the delay for a faster device like a PCIe card is set at 0.25s and the device is establishing connection with an Ultimo device, then the delay of subscription will be 1ms which is the minimum delay supported by subscriptions. If minimum delay possibly reaches 1s in a megabyte network, then a subscription error may occur when transmitting based on the condition that delay time is no more than 1s.

Network Configuration:

Network configuration refers to network IP address, mask, and gateway settings. Brooklyn supports redundancy mode and exchange mode settings.

Redundancy Mode

Many Dante devices have two network ports named “Primary” and “Secondary”. “Primary” port connects the physical network. If “Secondary” port has been used, then “Secondary” port shall connect to another physical network. The “Secondary” port cannot communicate with the “Primary” port.



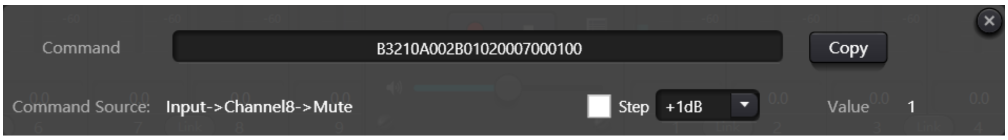
Multicast Stream:

What is stream? Dante audio routing automatically creates streams. A stream moves audio data of several channels from transmitter ends to one or more receiving ends. Unicast streams are given to single receiving devices while multicast streams are given to multiple receiving devices. Multicast streams may be created or configured manually through the interface, but it uses network bandwidth whether there is receiving device or not. Meanwhile, it doesn’t need extra bandwidth when more receiving ends are added.

As shown in the figure, select the multicast stream label page, check device channel, click create, and then created multicast stream will display in the right list of the interface. It can also be deleted when users don’t need it. A stream mostly includes four channels by default. If more than four channels are checked, they will be divided to several streams automatically.

3.5.7 Help Menu

(1) Central command



Open the central control command window and select the parameter to be controlled on the DSP GUI interface. The window will display the current command immediately. Copy the command and then use UDP or RS232 to send the command to the device form external control interface.

(2) Device upgrade

Device upgrades can be carried out through UDP. Connect the device, click setting-help-device upgrade. A file choice pop-up box will appear, navigate to, and choose the processor upgrade file (*.bin). Execute the update.

(3) About

Display version number, tech support contact information and copyright info., etc.

4. Control

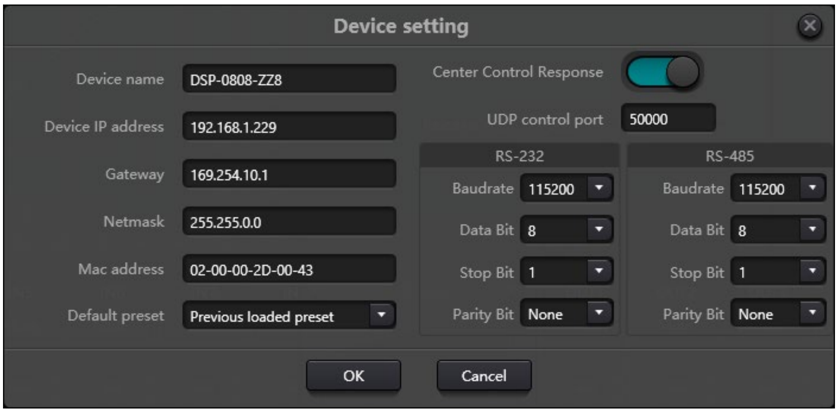
4.1 External Control Programmer

The External control programmer supports UDP and RS232, and controls protocols for all control parameters of the processor, including module parameter controls, parameter acquisition, and preset calling.

When UDP controls are used, the default port is 50000. Ports can be set in “Device Setting” in the DSP software.

When using RS232 controls, the default baud rate is 115200, data bits is 8, stop bit is 1, and parity set to ‘None’. Similarly, they can be set in “Device Setting”. **The interval between messages must be more than 100ms for RS232 command sending.**

If central control needs feedback and acknowledgement, please turn on ‘central control response’ in “Device Settings”.



4.2 Control Protocol

Because of historical reasons, the latest control protocol adopts variable-length and is fully compatible with old fixed-length control protocols. In protocols, the fourth byte is used to distinguish versions. 0- indicates V1 version (previous versions) and 1- indicates V2 version (current protocol version).

The difference between V1 and V2 is that V1 may control all processing module parameters, but one command can only control one parameter. If a parameter is needed to control continuous multiple channels, then V2 version must be used. In other words, in a condition that users need to press one key on the key panel to trigger GPIO output high- / low-level of devices, or send a command via RS232 / RS485, then V2 version will be the best choice.

Software coding rules (total 12 bytes)

byte1	byte2	byte3	byte4	byte5~132

0xb3	Message Type	Length	Version No.	Data

V1:

Information types (byte2): There are three information types including 0x21 (parameter controls), 0x22 (parameter acquisition) and 0x13 (scenario switch).

Length (byte3): invalid.

0x21 (parameter control): Data byte 5 thought 12 are respectively:

byte 5~6	byte 7~8	byte 9~10	byte 11~12
Module ID	Parameter Type	Parameter 1	Parameter 2

Please refer to Appendix A to get the distribution of **Module ID** (byte 5 ~ 6).

Please refer to Appendix B for **Parameter types** (byte 7 ~ 8).

When **Parameter 1** (byte 9 ~ 10) has only one parameter, then only parameter 1 is valid, such as control compressor switch.

Parameter 2 (byte 11 ~ 12) only valid when there are two parameters, such as control output channel 1 mute. Parameter value 1 shall be filled in input channel number from 0. Parameter value 2 shall be filled in 1 (mute).

Exception: Matrix routing has three parameters. The first one is input channel number, the second one is output channel number, and the third one is routing switch. At this point, byte 9 of parameter value 1 shall be filled in input channel numbers, byte 10 shall be filled in output channel number, and parameter 2 shall be filled in routing switch.

0x22 (Parameter Acquisition):

Parameter acquisition rules are the same with parameter controls. The difference between them is values acquired shall be filled in parameter 1 and parameter 2.

0x13 (Scenario Switch):

Users only need to fill scenario numbers (0 ~ 15) in byte 5 and 0 in byte 6 ~ 12.

Note: Central control command of V1 version can acquire code through software menu bar of PC. For customized development, please use this protocol rule.

V2:

Message types (byte 2): There are three message types (byte 2) including 0x21 (parameter controls), 0x22 (parameter acquisition), 0x13 (scenario switch), 0x74 (other controls), and 0x6e (Dante routing).

Length (byte 3): Fill in corresponding data section length based on information type. The length can be longer when actual sending is carried out. Total data volume can be found through adding 4 byte header information to data length.

1. Parameter Control (0x21)

At this point, the formats of data section are as follows.

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9 ~ 72
Input / Output	Start Channel	End Channel	Parameter Type	Parameter Value

Byte 5: It indicates control input or output channel, 0x2- input channels and 0x1-output channels

Byte 6 - 7: They indicate start and end channel numbers. Channel numbers start from 0.

Byte 8: This kind of parameter is the same with V1 version. Please refer to Appendix B.

Byte 9 - 40: Fill in parameter values of start to end channels. It shall be filled in from the ninth byte. Each parameter value shall take two bytes.

2. Parameter Acquisition (0x22)

Data section format is the same with parameter controls. Parameter values may not be filled in. Acquired parameters will be filled in this position.

3. Scenario Switch (0x13)

Byte 5: Fill in scenario numbers (0 - 15).

Byte 6 - 8: Fill in 0.

4. Other Controls (0x74)

Other controls include but not limited to GPIO, RS232, RS485, and central control replies. The protocol formats are as follows:

GPIO:

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9	Byte 10	Byte 11	Byte 12
Control Type	Data Length	Reserved	Reserved	GPIO Direction	Start GPIO	End GPIO	Value

The controlling type for Byte 5 is 1.

The data length of Byte 6 is fixed as four bytes.

Byte 9 GPIO direction, set input or output. Value 0 indicates input, and value 1 indicates output.

Byte 10 - 11 start GPIO and end GPIO. DSP devices have eight GPIOs in total, which are indicated with number 0 - 7.

Byte 12 is determined according to Byte 9 GPIO direction. The field shall be filled in high (1) / low (0) level for output settings. The field is a return field to read GPIO level value on devices for input settings.

RS232 / RS485:

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9 - 132
Control Type	Data Length	Reserved	Reserved	Data

Byte 5 is 2 for RS232 control, and 3 for RS485 control.

The data length of Byte 6 refers to data length that shall be sent via RS232 / RS485 currently.

Byte 9 - 132 shall be filled in data sent by RS232 / RS485.

Central control replies:

Byte 5	Byte 6	Byte 7	Byte 8	Byte 9
Control Type	Data Length	Reserved	Reserved	Reply Switch

Byte 5 control type is 4.

The data length of Byte 6 is 1.

When Byte 9 is 1, it means turning on central control replies switch; and 0 means turning off replies.

4.3 Serial Port-to-UDP (RS232 To UDP)

DSP devices support RS232 translating into UDP. The protocol formats are as follows.

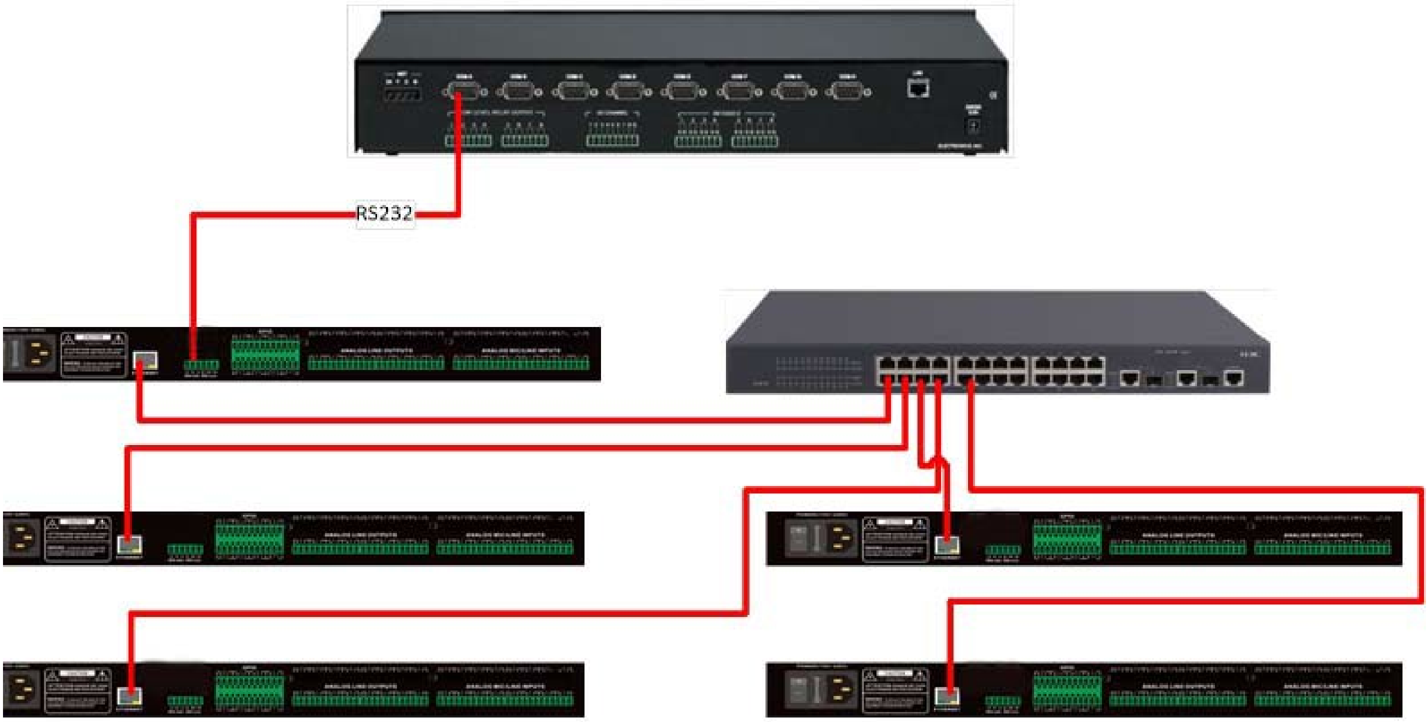
4bytes prefix	4bytes	2bytes	1byte	1byte	128bytes
UDP:	IP Address	Port	Data Length	Reserved	Data

After receiving the protocol format data packet, RS232 sends data in the protocol to appointed IP addresses and devices at ports.

For example, when sending data “HELLO DSP” to device port 50000 of device ”192.168.10.22”, the protocol commands are as follows:

4 bytes prefix	4 bytes	2 bytes	1 byte	1 byte	128 bytes
0x3a504455 （’:PDU’）	0x1610A8C0	0xC350	0x09	0x00	”HELLO DSP”

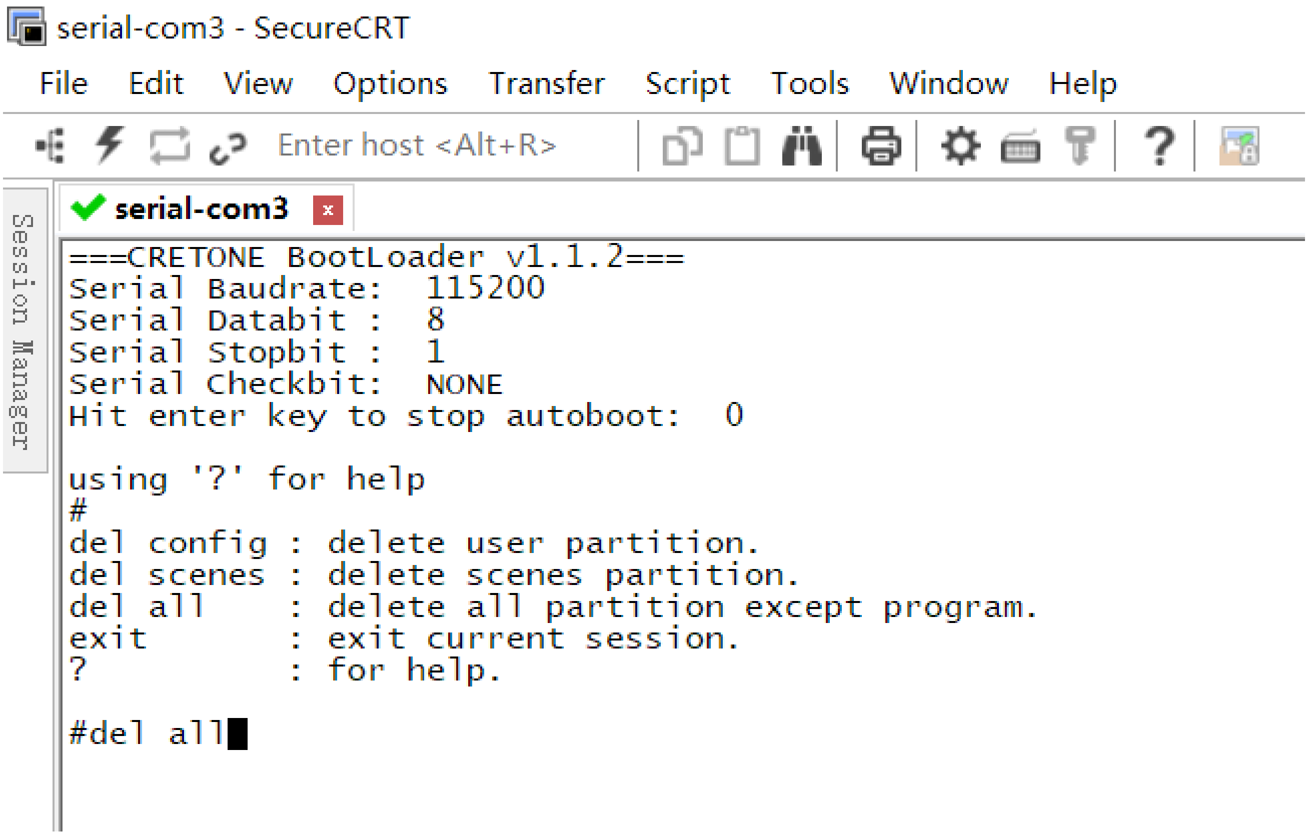
Application scenario: The function can be applied in scenarios when many central control hosts have no network port. As shown in the below figure, central control hosts translate network commands through serial ports to control any network device.



5. FAQs

1. How to restore factory setting?

Connect the processor to a computer through RS232 and run the serial port software (SecureCRT is recommended for use). The default baud rate of the serial port is 115200, 8 data bits, no parity check, and 1 stop bit. After connecting SecureCRT to the serial port, long press to enter a terminal interface to reboot the computer and enter the bootloader boot dialog box as shown in the below figure:



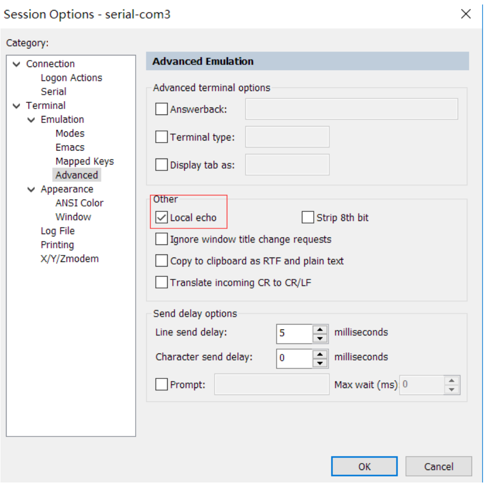
Command explanation:

del config: delete configuration information, such as network configurations like IP address. The device restores to default IP 169.254.20.227 after deleting.

del scenes: delete preset. All 16 presets of DSP devices restore to default values.

del all: delete all sections except the program.

Note: There may be no echo after the installation of SecureCRT. Please check settings under “Local echo” by going to Options->Session Options->Enable ‘Local Echo’, as shown in the picture below.



Appendix A: Module ID Distribution

Module Name	ID	Module Name	ID
Input source	299	Output Channel 1-32 High & Low Pass	167~198
Input Channel 1-32 Expander	1~32	Output Channel 1-32 Equalizer	199~230
Input Channel 1-32 Compressor	33~64	Output Channel 1-32 Delayer	231~262
Input Channel 1-32 Auto Gain	65~95	Output Channel 1-32 Limiter	263~294
Input Channel 1-32 Equalizer	97~128		
Input Channel 1-32 Feedback Inhibition	129~160		
AutoMixer	161	Echo Canceller	162
Echo Cancellation	163	Noise Suppressor	164
Noise Suppression	165		
Mixer	166		
Output	295		
System Control	296		

Appendix B: Module Parameter Types (1)

Module Name	Parameter Type	Description
Input Source	0x1	Gain
	0x2	Mute
	0x3	Sensitivity
	0x4	Phantom Power Switch
	0x5	Signal Generator Type
	0x6	Signal Generator Frequency
	0x7	Sine Wave Gain Size
	0x8	Channel Name
	0x9	Invert
	0x10	Gain Compensation
	0x11	Link
	0x12	Channel Level
Delayer	0x1	Bypass Switch
	0x2	Millisecond
	0x3	Microsecond
Equalizer	0x1	Total Equalizer Switch
	0x2	Child Segment Switch
	0x3	Frequency
	0x4	Gain
	0x5	Q Value
	0x6	Type

Module Name	Parameter Type	Description
Output	0x10	Gain Compensation
	0x11	Link
	0x12	Channel Level
	0x1	Gain
	0x2	Mute
	0x3	Channel Name
	0x4	Invert
	0x5	Sensitivity
	0x6	Gain Compensation
	0x7	Link
	0x8	Channel Level
Expander	0x1	Switch
	0x2	Threshold
	0x3	Ratio
	0x4	Setup Time
	0x5	Release Time
Compressor	0x1	Compressor Switch
	0x2	Compressor Threshold
	0x3	Compressor Ratio
	0x4	Setup Time
	0x5	Recovery Time
	0x6	Gain Compensation

Appendix B: Module Parameter Types (2)

Module Name	Parameter Type	Description	Module Name	Parameter Type	Description
Mixer	0x1	Mixer Switch	Feedback Inhibition	0x1	Switch
	0x2	Mixer Gain		0x2	Feedback Point Frequency
High & Low Pass	0x1	High Pass Switch		0x3	Feedback Point Gain
	0x2	High Pass Type		0x6	Preset
	0x3	High Pass Slope		0x7	Clear
	0x4	High Pass Frequency		0x8	Panic Threshold
	0x5	High Pass Gain		0x9	Feedback
	0x11	Low Pass Switch	Auto Gain	0x1	Switch
	0x12	Low Pass Type		0x2	Threshold
	0x13	Low Pass Slope		0x3	Target Threshold
	0x14	Low Pass Frequency		0x4	Ratio
	0x15	Low Pass Gain		0x5	Setup Time
Auto Mix	0x1	Total Mute		0x6	Release Time
	0x2	Total Gain	Echo Cancellation	0x1	Echo Cancellation Switch
	0x3	Slope		0x2	Echo Cancellation Mode
	0x4	Response Time	Noise Suppression	0x1	Noise Suppression Switch
	0x5	Channel Auto Switch		0x2	Noise Suppression Mode
	0x6	Channel Mute	System Control	0x1	System Mute
	0x7	Channel Gain		0x2	System Gain
	0x8	Priority			
	0x9	Auto Mix Switch			

7. After-sales Service

Should you experience problems using the ALF-DSPXX-U/D, please note that any transport costs of the equipment to the distributor are borne by the user during the warranty.

1) **Product Limited Warranty:** The manufacturer warrants that its products will be free from defects in materials and workmanship for **seven years,** which starts from the first day of purchase.

   Proof of purchase in the form of a bill of sale or receipted invoice which is evidence that the unit is within the warranty period must be presented to obtain warranty service.

2) **What the warranty does not cover (servicing available for a fee):**
   - Warranty expiration.
   - Factory applied serial number has been altered or removed from the product.
   - Damage, deterioration, or malfunction caused by:
     - Normal wear and tear.
     - Use of supplies or parts not meeting product specifications.
     - No certificate or invoice as the proof of warranty.
     - The product model showed on the warranty card does not match with the product or if the product had been altered.
     - Damage caused by force majeure.
     - Servicing not authorized by the manufacturer.
     - Any other causes which do not relate to a product defect.
     - Delivery, installation or labour charges for installation or setup of the product.

3) **Technical Support:** Contact our after-sales department.

8. Warranty

1.1 This limited warranty covers defects in materials and workmanship in this product.

1.2 Should warranty service be required, proof of purchase must be presented to the Company. The serial number on the product must be clearly visible and not have been tampered with in any way whatsoever.

1.3 This limited warranty does not cover any damage, deterioration or malfunction resulting from any alteration, modification, improper or unreasonable use or maintenance, misuse, abuse, accident, neglect, exposure to excess moisture, fire, improper packing, and shipping (such claims must be presented to the carrier), lightning, power surges, or other acts of nature. This limited warranty does not cover any damage, deterioration or malfunction resulting from the installation or removal of this product from any installation, any unauthorized tampering with this product, any repairs attempted by anyone unauthorized by the Company to make such repairs, or any other cause which does not relate directly to a defect in materials and/or workmanship of this product. This limited warranty does not coverequipment enclosures, cables or accessories used in conjunction with this product.

This limited warranty does not cover the cost of normal maintenance. Failure of the product due to insufficient or improper maintenance is not covered.

1.4 The Company does not warrant that the product covered hereby, including, without limitation, the technology and/or integrated circuit(s) included in the product, will not become obsolete or that such items are or will remain compatible with any other product or technology with which the product may be used.

1.5 Only the original purchaser of this product is covered under this limited warranty. This limited warranty is not transferable to subsequent purchasers or owners of this product.

1.6 Unless otherwise specified, the goods are warranted in accordance with the manufacturer's product specific warranties against any defect attributable to faulty workmanship or materials, fair wear and tear being excluded.

1.7 This limited warranty only covers the cost of faulty goods and does not include the cost of labor and travel to return the goods to the Company's premises.

1.8 In the event of any improper maintenance, repair or service being carried out by any third persons during the warranty period without the Company's written authorization, the limited warranty shall be void.

1.9 A 7 (seven) year limited warranty is given on the aforesaid product where used correctly according to the Company's instructions, and only with the use of the Company's components.

1.10 The Company will, at its sole option, provide one of the following three remedies to whatever extent it shall deem necessary to satisfy a proper claim under this limited warranty:

1.10.1 Elect to repair or facilitate the repair of any defective parts within a reasonable period of time, free of any charge for the necessary parts and labor to complete the repair and restore this product to its proper operating condition.; or

1.10.2 Replace this product with a direct replacement or with a similar product deemed by the Company to perform substantially the same function as the original product; or

1.10.3 Issue a refund of the original purchase price less depreciation to be determined based on the age of the product at the time remedy is sought under this limited warranty.

1.11 The Company is not obligated to provide the Customer with a substitute unit during the limited warranty period or at any time thereafter.

1.12 If this product is returned to the Company this product must be insured during shipment, with the insurance and shipping charges prepaid by the Customer. If this product is returned uninsured, the Customer assumes all risks of loss or damage during shipment. The Company will not be responsible for any costs related to the removal or reinstallation of this product from or into any installation. The Company will not be responsible for any costs related to any setting up this product, any adjustment of user controls or any programming required for a specific installation of this product.

1.13 Please be aware that the Company's products and components have not been tested with competitor's products and therefore the Company cannot warrant products and/or components used in conjunction with competitor's products.

1.14 The appropriateness of the goods for the purpose intended is only warranted to the extent that the goods are used in accordance with the Company's installation, classification, and usage instructions.

1.15 Any claim by the Customer which is based on any defect in the quality or condition of the goods or their failure to correspond with specification shall be notified in writing to the Company within 7 days of delivery or (where the defect or failure was not apparent on reasonable inspection by the Customer) within a reasonable time after discovery of the defect or failure, but, in any event, within 6 months of delivery.

1.16 If delivery is not refused, and the Customer does not notify the Company accordingly, the Customer may not reject the goods and the Company shall have no liability and the Customer shall pay the price as if the goods had been delivered in accordance with the Agreement.

1.17 THE MAXIMUM LIABILITY OF THE COMPANY UNDER THIS LIMITED WARRANTY SHALL NOT EXCEED THE ACTUAL PURCHASE PRICE PAID FOR THE PRODUCT.