MP-B88Pro
Digital Sound Processor

User Manual

www.milestonepro.co.in

Contents

1. Technical Overview

1. Introduction To Technology

The audio DSP series uses several core technical features that allow audio engineers to do their work better. DSP based remote audio hardware is routing and processing controlled through a computer. The manual mainly introduces the technologies needed to achieve this goal.

DSP Controller is a Windows-based application that configures and controls DSP hardware. DSP Controller Built-in 16 presets, each preset can be flexibly designed processing modules and order according to the requirements of designers. After the design is completed, you only need to save, and can be used all the time. DSP Controller The built-in processing module order and parameters are in line with most application scenarios, and can be easily used without any changes.

DSP Controller is a full function application, including the processing parameter adjustment of all modules and the setting of peripheral accessories, such as RS232, RS485, drag panel configuration and Dante network audio control. Of great interest is the user interface, which allows engineers to create custom interfaces that can be edited by an integrator and operated by field technicians or end users who do not understand the technology. Advanced security features allow end users to access only the controls allowed by engineers or system designers.

2. Audio Input And Output Section

2.1. Audio Input

The device can have up to 16 fixed analog audio inputs, connected through a detachable balanced Phoenix connector. The analog input part can support the microphone, or the line level signal with nominal levels of 0 dBu, 6 dBu, 12 dBu, 18 dBu, 24 dBu, 30 dBu, 36 dBu, 42 dBu, 48 dBu. Each input can use a phantom power supply of + 48VDC, and the front stage amplification gain and phantom power supply can be very conveniently controlled through DSP Controller.

2.2. Audio Output

The device can have up to 16 fixed analog audio outputs, connected through a detachable balanced Phoenix connector. The first stage of the analog output part is the converter (DAC) of D / A. The DSP uses an advanced 24-bit 256X sampler converter. Like the A / D converter, the multi-bit architecture enables a wider dynamic range, but has the same superior distortion features as the conventional unit digital analog converter. Set the unit gain (0dB) by volume control, the analog output portion was corrected to + 4 dBu with a dynamic margin of 14dB. This means that the 0 dBFS digital signal is equivalent to the + 18 dBu output signal. If other signal levels are required, they is easily achieved by changing the volume.

3. Float Point DSP

The DSP device uses the analog device SHARC DSP, and these DSPs have 32-and 40-bit floating-point processing. Comparable to the 40-bit floating-point processing of other devices. Floating-point processing can provide users with significant advantages in terms of sound quality and ease of use.

Fixed-point handling limit:

The problem with fixed points is that clipping distortion can occur if there are significant gain changes, data loss, or worse. For example, consider the 24-bit audio signal processing via the 24-bit fixed-point processing. If, under some processing, you decay the signal by 42dB, then the new signal will only contain 17 bits of information.7 bits are lost forever due to the decay of the gain. Even worse is the problem of sharpening distortion. If a significant signal is applied to a signal that is already close to 0dBFS, the signal will be sharpened at 0dBFS, causing audio distortion. Even if the signal level is reduced below 0dBFS by posterior adjustment, the cutting has occurred and distortion remains. Fixed points may try to create some dynamic residues above 0dBFS, but in doing so, they inevitably give up some digit numbers. For example, if a dynamic margin of 12dB (2 bits) is created, then a 24-bit system is actually only 22 bits.

Floating point:

Instead, by floating-point processing, all digits of available were consistently assigned to the signal regardless of its signal level. Basically, the floating point uses some number of digits as an index to set the approximate signal level, and then allocates the remaining number of digits to store signal values that are actually independent of the level. Therefore, regardless of the signal level, from -200dB and 200dB to dB value above 0dBFS, the saved signal has optimized accuracy and no cutting distortion. SHARC Provide 32-bit and 40-bit precision processing, and through 32-bit processing, 25 bits are assigned to the storage signal, regardless of its signal level. This means that the low-level signal through at least 1 bit is always significantly better than the 24-bit fixed-point treatment. With an extended 40-bit precision processing, signals can be stored through 33 bits.

Practical significance:

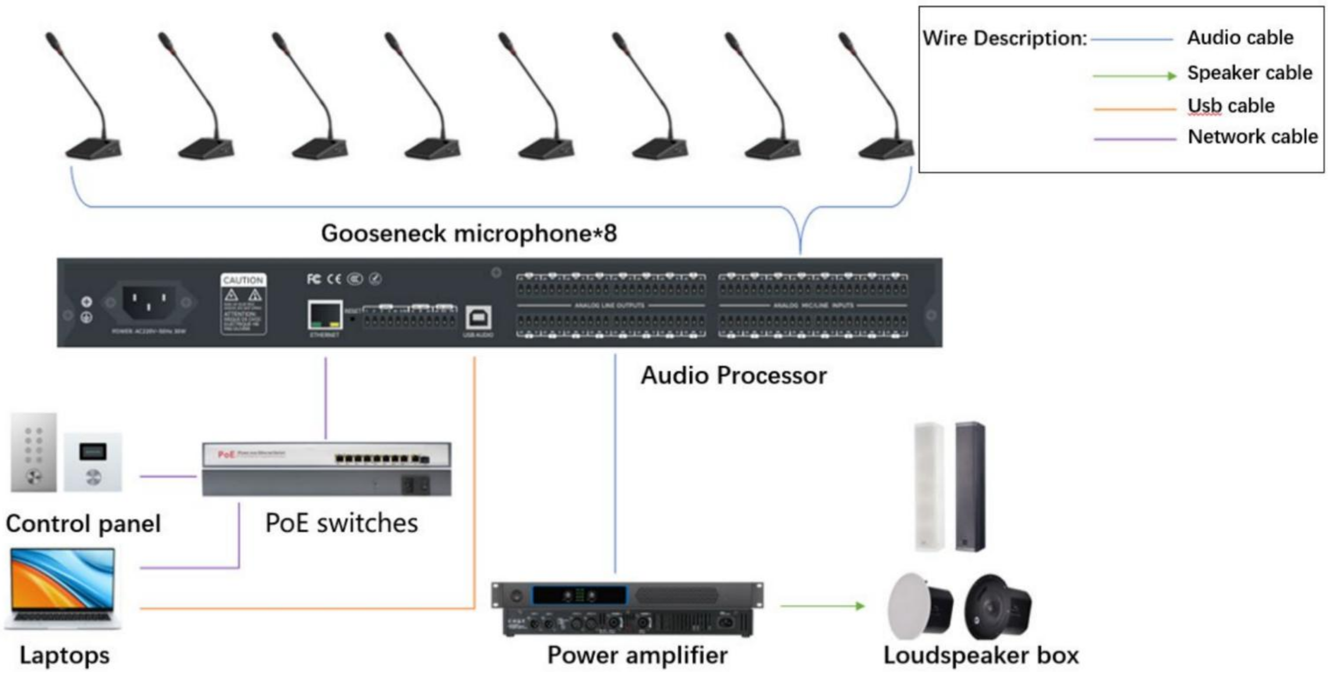
What is the practical significance of floating point processing to users? The gain-level problem between multiple modules can be ignored. If one module reduces the signal level by 50dB, and another subsequent processing increases it to the original value, there is no data loss. In a fixed-point system, the user must view its signal level before it is sent to the A / D converter, because all the digital-to-analog converters are fixed-point. In the DSP system, if you notice that your signal is cut before the output is sent to the digital-to-analog converter, you can immediately turn it off at the output to correct the problem. When using the fixed system, you need to search for each processing module to find the cutting source.

4. Typical System Application

Conference sound expansion system

Microphone access to the local input of the DSP processor, the computer can use the USB interface for audio recording or play background music, Debugging the entire audio system through PoE switches or directly connected to processors, after the processor speaker processing module of the signal modification and processing, by the local output connected to the power amplifier, the power amplifier output to the speaker.

The control panel controls the processor through the PoE switches, which can realize the volume adjustment and scene call functions.

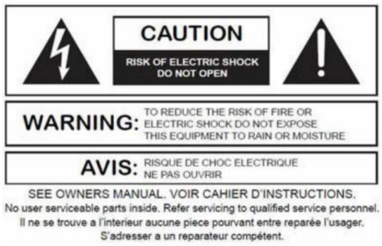


3. Hardware

1.Safety Instructions

Important safety instructions

1. Read the instructions.
2. Keep the instructions well.
3. Pay attention to all the warnings.
4. Follow all of the instructions.
5. Do not use the equipment near the water. Equipment should not be exposed to water droplets or splashes, ensuring that there is no liquid item, such as vases.
6. Use only the dry cloth cleaning equipment.
7. Do not block the air vents. Install only as per vendor instructions.
8. Do not install any heat sources such as radiators, heat registers, stoves or other heat generating equipment (including power amplifier).
9. Use the protective ground connection to connect the device to a power socket. Do not cause the polarizing plug or the ground plug. One polarized plug comes with two blades, one wider than the other. One ground-type plug has two blades and a third ground end. Wide blades or third ground ends provide safety for the user. If the plug provided does not match the power outlet, please contact the electrician to replace the old outlet.
10. Protect the power cord from being trampled or squeezed, especially the plugs, sockets and wire connections with the equipment.
11. Use only accessories / accessories specified by the manufacturer.
12. Use only carts, tripods, brackets or tables designated by the manufacturer or sold with the equipment. When using a trolley, be careful when moving the trolley / device combination to avoid injury from tipping.
13. During thunderstorms or when not in use for a long time, please unplug the device.
14. Find qualified warranty personnel to handle all repair issues. When the equipment is damaged in any way, maintenance is necessary, such as when the power cord or plug cord is damaged, liquid spills or objects fall into the equipment, the equipment is exposed to rain or moisture, the operation is incorrect, or the equipment falls off.



The lightning sign with an arrow symbol inside the equilateral triangle is designed to make the user aware that the uninsulated "dangerous voltage" inside the product shell is enough to cause an electric shock. The exclamation mark within the equilateral triangle is intended to make the user of the importance of operation and maintenance (repair) instructions in the product literature.

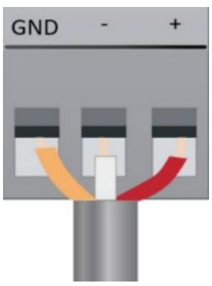
Warning: To prevent electric shock, do not use the polar plug, socket or other socket outlet provided on the device with an extension cord, unless the tip cannot be fully inserted.

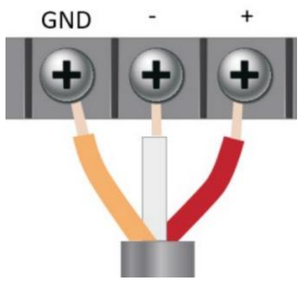
2.Audio Wiring Reference

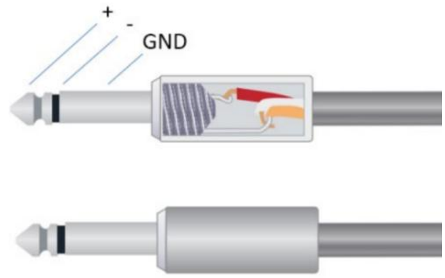
Balanced connection

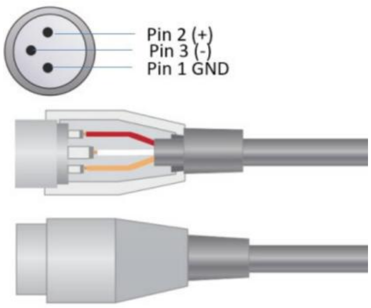
Any one of these interfaces may occur on both sides of the equilibrium connection.

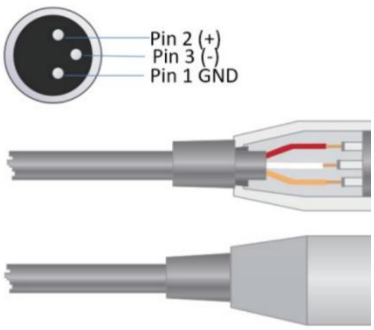
Note: For one XLR interface, the header connects the output and the header connects the input.





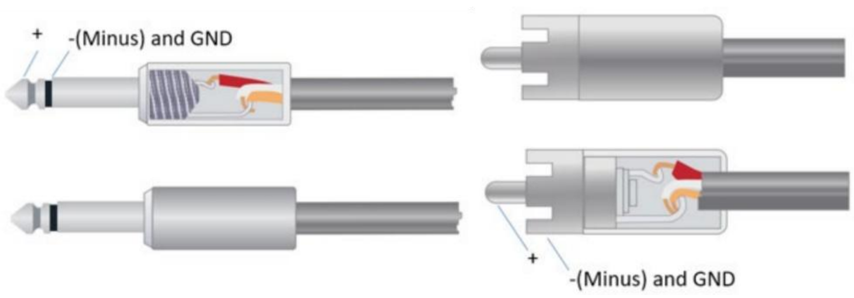






Unbalanced connection

The RCA, interface and 1 / 4 inch TS interface are unbalanced interfaces, installed with a multiple stranded shielding wire and can be placed at both ends of the unbalanced connection.



3.Specification Parameters

Number of GPIO ports and serial ports	RS232*1	RS232*1+RS485*1+GPIO*4
Processor	ADI SHARC 21489	ADI SHARC 21489
Sampling rate/Quantization Bits	48K/24bit	48K/24bit
Input gain	0/6/12/18/24/30/36/42/48dB	0/6/12/18/24/30/36/42/48dB
Vision power supply	+48V/10mA max	+48V/10mA max
Frequency response (20 ~ 20 kHz)	±0.5dB	±0.5dB
Maximum level	+18dBu	+18dBu
THD+N	0.002 %@4dBu	<-92dB @17dBu
Input dynamic range	110dB	110dB
Output dynamic range	110dB	112dB
Channel isolation degree @ 1 kHz:	108dB	108dB
Input impedance (balanced connection method)	5.4KΩ	5.4KΩ
Output impedance (balanced connection method)	600Ω	600Ω
Latency	<3ms	<3ms
Working power supply	AC 220V 50Hz	AC 220V 50Hz
Dimensions (width x depth x height)	482 x 200 x 45mm	482 x 200 x 45mm
Hauled weight	3KG/4KG	3KG/4KG

4.Mechanical Parameters

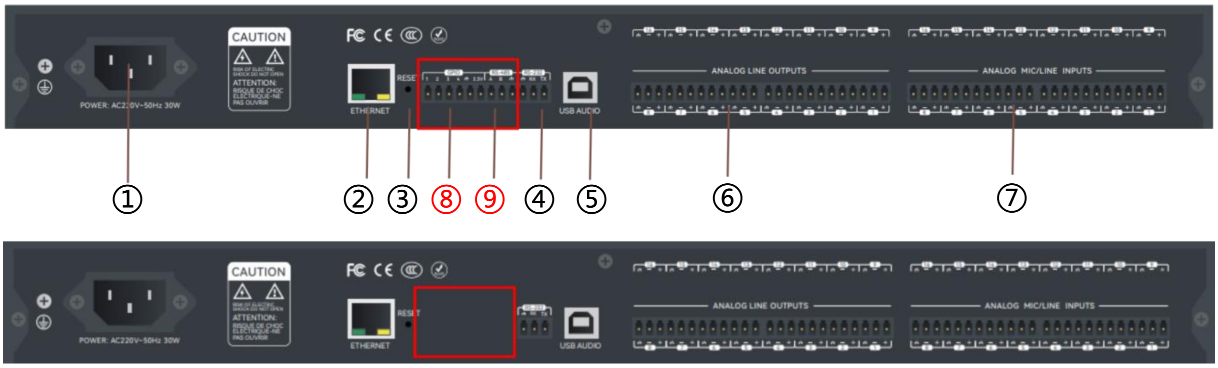
- **Required space:** 1U (width and depth and height: 18.91 ″ x 9.5 ″ x 1.72 ″ / 48.02 cm x 24.13 cm x 4.37 cm). Depth does not include reserved space for power plugs, terminals A minimum of 3 inches of additional space is required for connections on the rear panel. The reserved depth depends on the wire used and the connection mode.
- **Electrical:** maximum universal input power for AC 220V 50Hz.
- **Operating environment:** The recommended maximum operating environment temperature is 30℃ / 86 ℉, ensuring that there is no barrier on the left and right sides of the equipment (at least 5.08 cm, 2 inch gap should be reserved). Do not cover newspapers, tablecloth, curtains and other items with the equipment sink.
- **Shipping weight: 3kg/4kg.**

5.Front Panel

① PWR: LED power indicator light, the green light is always on when the machine is running normally;

② SYS: The device operation status indicator light, normally flashing slowly (with a frequency of approximately 1 second flashing).

6.Rear Panel



① POWER Power interface: connect the AC 220V 50Hz power adapter;

② ETHERNET Network control interface: By connecting to this network port, the client computer can debug and monitor the device;

③ Reset button: used to restore the factory settings;

④ RS232 interface: connect to the control terminal or the central control equipment;

⑤ USB sound card: playing and recording;

⑥ OUTPUT Analog signal output interface: can be connected to power amplifier, active speakers and other devices;

⑦ INPUT Analog signal input interface: can connect the microphone, DVD and other equipment;

⑧ GPIO interface: 4 definable input/output logic level control ports, which can be connected to levels below 12V, such as dry contact signals, to trigger device actions, or output 5V after triggering actions to drive relays and other devices;

⑨ RS485 interface: connect the control terminal or central control equipment.

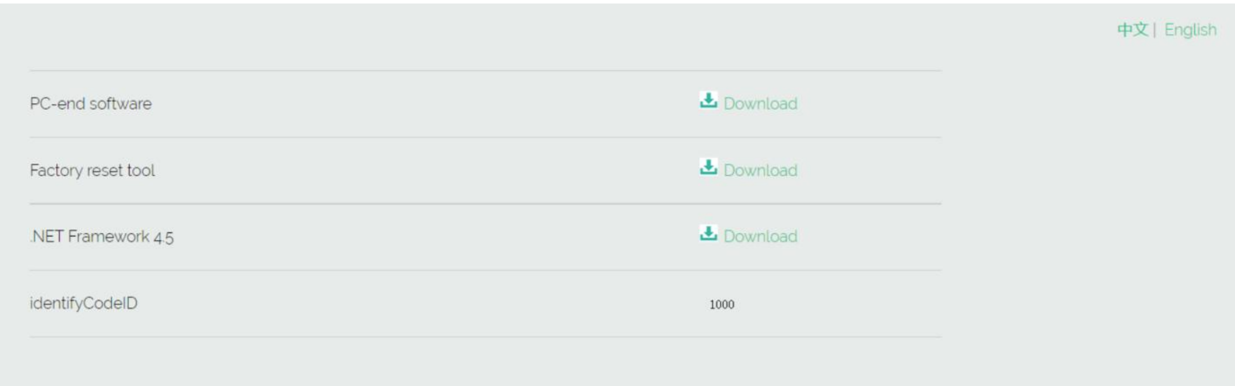
3. Software

1.Software Installation

A Windows PC with a 1 GHz or higher processor and:

- Windows 7 Or a higher version;
- 1 GB of free storage space;
- 1024 x 768, resolution;
- 24 bits or higher colors;
- 2 GB or higher memory;
- Network (Ethernet) interface;
- The CAT 5 line or the existing Ethernet network.

Audio processor built-in control software, without CD installation, acces s to the audio processor IP address can be quickly downloaded, through the browser address bar input device IP address access to the audio processor, find the download link will install the installed software download to the local installation. The default IP address of the device is 169.254.10.227 Subnet mask: 255.255.0.0. Please add the address of the network segment to the PC to provide for the normal access of the device. After the device is started, enter "http://169.254.10.227/" in the browser address bar.



Before installing the PC software, make sure the PC has “Microsoft installed. Net Framework” Newer version.

Note: If you fail to download and install the software, please try with IE or Google browser, or enter the browser Settings, cancel the browser download popup prompt to try again.

2.Using the Software

After opening the software, the main interface is displayed:

Click the button in the upper right corner of the main interface, you will automatically find all the processors on the network. The user will connect to the specified processor according to the needs. When the device is successfully connected, this image will light up in the device list , and a processor supports up to 8 users to connect and control online at the same time.

3.Audio Module Parameters

3. 1. Input Source

The input channel module parameters are as follows:

Sensitivity: Microphone gain, 0 / 6 / 12 / 18 / 18 / 24 / 30 / 36 / 42 / 48dB 9 gear optional.

Phantom power supply: feed the external capacitor microphone, click the button when needed. Line input or no power supply to prevent damage to external equipment.

Sine wave: the drag frequency can produce a sine wave at the specified frequency (20 ~ 20 kHz). The output level can be adjusted as needed, in dBFS. Use the pusher adjustment or click the text input box to specify a value.

White noise: White noise has equal energy at each frequency component. Observe it on a spectrometer with a constant bandwidth, which has a flat frequency spectrum. At this time, the frequency adjustment is ineffective, and the level is available.

Pink noise: The frequency component power of pink noise is mainly distributed in the middle and low frequency bands, and it drops at 3dB / Oct speed in the spectrum. At this time, the frequency adjustment is ineffective, and the level is available.

In addition, the right-click on each pusher shows the following menu settings on the main interface.

Copy: copies all parameters of the channel;

Paste: Paste the copied parameters to the channel;

Group Settings: Quickly open the group Settings interface.

Minimum and maximum gain: limit the maximum and minimum gain of this channel. When debugging, do not want to be external change and affect the stability of the system, the maximum gain can be set.

3. 2. Expander

The extender, in principle contrary to the compressor, is able to extend the dynamic range of the signal. The most fundamental difference between the two devices is that the compressor acts on the signal above the threshold and the extender acts for the signal below the threshold. Extenders will lower weak signals, as can be seen in FIG, an input signal below the threshold 20dB produces an output signal 40dB below the threshold. the signal part below the threshold will extend downward, resulting in a smaller level. When using a 1:20 extension ratio, the extender transmission feature looks like a noise gate. In fact, the noise gate is an extender that uses a large extension ratio.

Extender module parameters are as follows:

Threshold: The signal must exceed this level to open the expander (allow signal to pass). Actually generally set to the size of the ambient noise;

Ratio: the slope below the threshold point on the gain curve, The action of approaching the door begins when the ratio is set high;

Attack: the duration of the input signal, above the threshold, the time required to open the expander. A faster opening time allows for a faster transient opening of the extender;

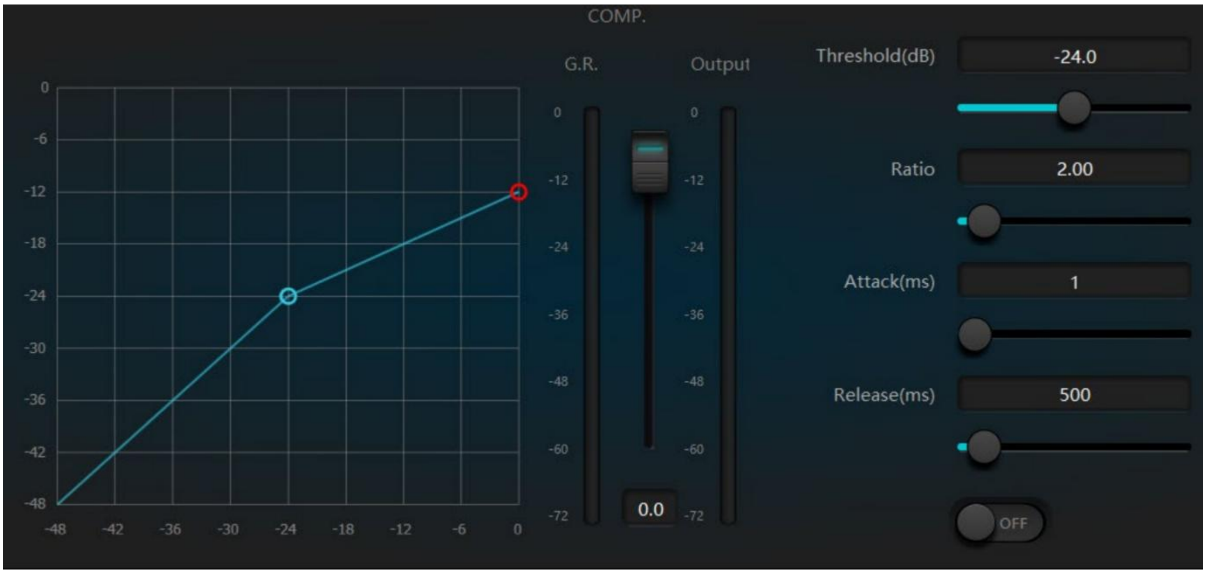
Release: the time required for the input signal to have the gain go back to the value below the threshold;

Both attack and release function only to reduce the rate of change in the gain decay. That is, the speed of the gain increase from-40dB to 0dB is controlled by the attack, while the gain is reduced from 0dB, and the speed of decay to-40dB is controlled by the release . The attack or the release are not related to the threshold setting. If the signal changes high and low below the threshold, the attack and the release will also affect the gain attenuation respectively, and once the signal level rises above the threshold, The gain attenuation generated by the expander will disappear at the speed controlled by the establishment time.When the gain attenuation decreases to 0dB, the extender stops expanding. Subsequently, when the signal drops below the threshold again, the extender starts again and the release starts to work.

3.3. Compressor And Limiter

Compressor

The compressor reduces the dynamic range of signals greater than the threshold, and the signal level below the threshold remains constant. The compressor has the following control parameters:



The compressor module parameters are as follows:

Threshold: The compressor / limiter starts to reduce the gain when the signal level is above this value. Any signal beyond the threshold will be considered an overshot signal, and its level is reduced under normal conditions. The larger the signal exceeded the threshold, the more the level decayed;

Ratio: namely, the compression ratio. The ratio determines the degree to which the overshoot signal has decayed to the threshold level. The smaller the compression ratio, the easier the signal is to be higher than the threshold. Once the signal exceeds the threshold, the compression ratio determines the ratio of the input signal change to the output signal. For example, when the compression ratio is 2:1, and the input signal exceeds the threshold of 2dB, then the part of the output signal exceeding the threshold changes only by 1dB. A compression ratio of 1:1 indicates that the compressor has not scaled the signal. The adjustable range of the compression ratio is 1 to 20;

Attack, release: In order to retain the natural onset, the first part of the level will be unaffected through the compressor (or only slightly affected). To do this, Need to slow down the reaction time of the compressor. Similarly, the suction effect occurs if the signal gain shows substantial rapid decay, and rapid recovery. The establishment time and release of the compressor are designed to avoid this situation. The establishment time can determine the rate at which the gain decay occurs, while the release can determine the rate of the gain recovery;

Output gain: also called the gain compensation pusher. If the compressor significantly reduces the signal level, an increased output gain may need to maintain the volume. This lifting operation has the same amount for all parts of the signal, independent of the setting of other parameters of the compressor;

G.R. And output level meter: G.R. Indicates the compression amount of the compressor; output represents the output level of the signal through the compressor module. The compression is shown in an inverted level table, if the input signal is-6dB, the threshold is set to-30dB and the ratio is 2:1, the compression is 12dB, G.R. The level meter indicates at-12dB and output indicates at-18dB.

Limiter

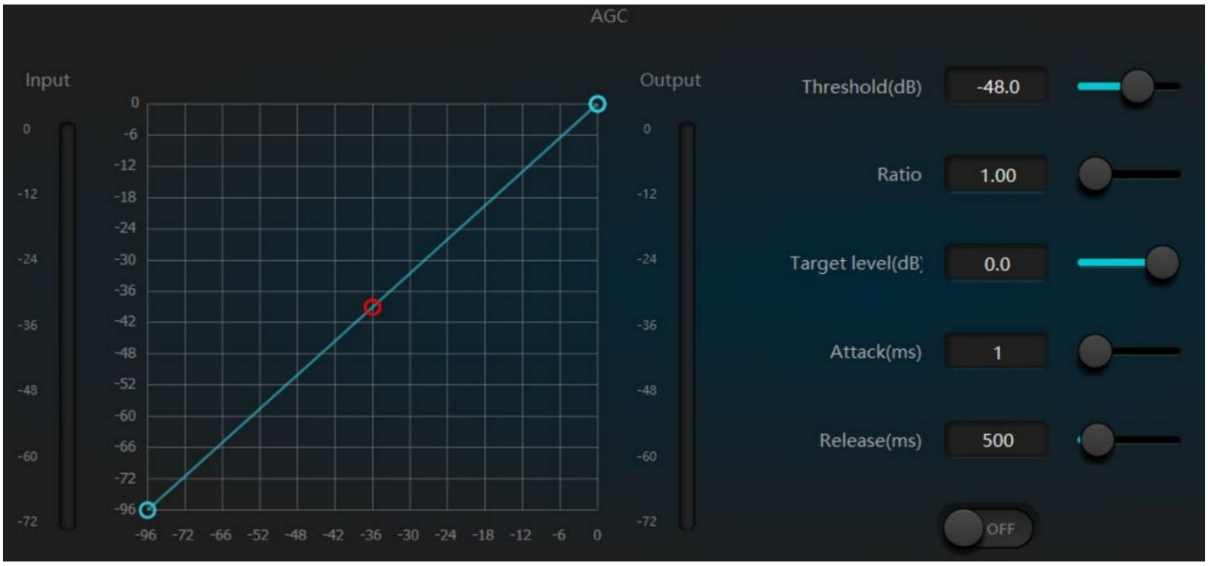
Also called a limiter, it has only one key task: make sure the signal does not exceed the threshold level under any circumstances. By adjusting the control parameters of the compressor, it can work very similar to the limiter. The core of the limiter working principle is that it really concerns the signal content below the threshold level and how the gain attenuation starts to produce before the signal overshoots. The limit period is completed by two stages of processing. In the first stage, there is only a slight limit, but it does not process the overshot signal, and then in the second stage, if the signal is overshot, they attenuate in a very violent way.

The limiter can only the two parameters, threshold and release. For signal processing, the occasional cut should be solved by the limiter, while the frequent cut usually requires the attenuation of the level of the signal

3.4. Auto Gain Control

Automatic gain control (AGC) is a special case of the compressor, whose threshold is set at a very low level, medium to slow establishment time, long release, and low ratio. The aim is to raise the signal with uncertain levels to a target level while staying dynamic. Most of the automatic gain controls contain some sort of silent detection to prevent loss of gain attenuation during silent periods. This is the only feature that distinguishes the automatic gain control from a normal compressor / limiter.

Automatic gain control is used to normalize the level of the CD player playing background, foreground, or waiting music to eliminate some changes in the paging microphone level.



Automatic gain, the module parameters are as follows:

Threshold: When the signal level is below this threshold, the input / output ratio is 1:1. When the signal level is above this threshold, the input / output ratio changes with the ratio control setting. Set this threshold to a background noise just above the input signal level;

Ratio: The ratio between the input signal level portion that exceeds the threshold and the output signal level portion;

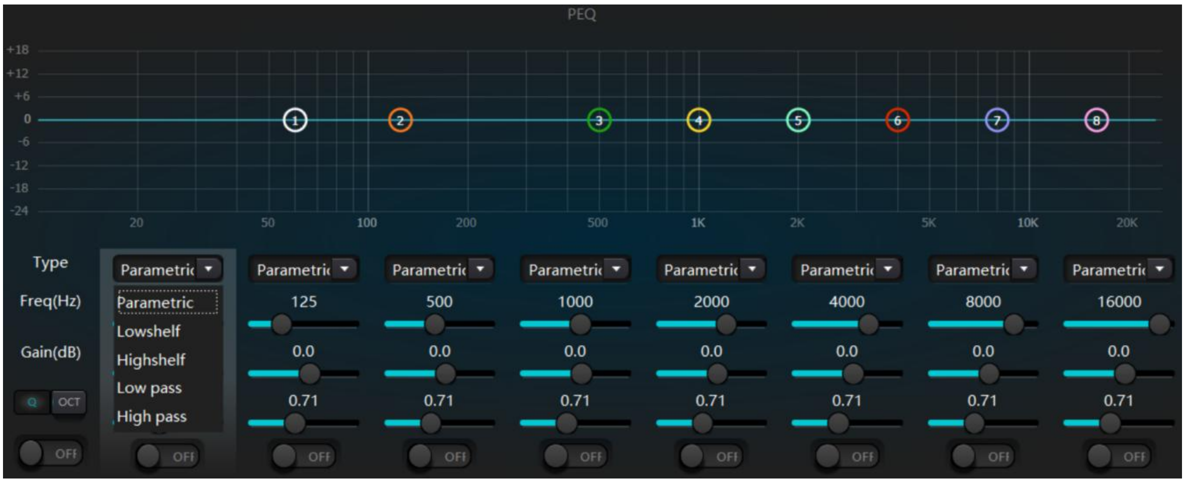
Target level: the required output signal level. If the signal is above this threshold, the controller will compress the signal according to a ratio;

Attack: control the reaction time above the threshold level;

Release: controls the level reaction time below the threshold signal.

3.5. Parametric Equalizer

The main role of the equalizer is to correct the frequency ranges that are overemphasor missing, whether they are wide or narrow. Equalizers can also help us narrow or widen the frequency range, or change the size of some components of their spectrum. In simple terms, the equalizer can change the timbre of the signal.



The equalizer module parameters are as follows:

Type: Default parameter balance, optional high & low shelf filter and high & low pass filter. Each filter has a different form and can perform different functions;

High & Low pass: the reference frequency of the pass filter is called the cut-off frequency. The pass filter allows the frequency components on the cut-off frequency side to pass completely through the filter, and the frequency component on the other side of the cut-off frequency is continuously attenuated. High pass can pass through the frequency component above the cutoff frequency and filter out the frequency component below the cutoff frequency. The Low pass, on the contrary, allows the frequency component passing through below the cutoff frequency to filter out the frequency component above the cutoff frequency;

High & Low shelf: High shelf filter means a partial gain increase or decay of the frequency above the set frequency. A low shelf filter is a partial gain increase or decay of the frequency below the set frequency. The set frequency is not the cut-off frequency of 3dB, but is located at the center point of the falling edge or rising edge of the filter. The Q value affects the peaking and has a corresponding mathematical relationship with the peak;

Frequency : the center frequency of the filter;

Gain : a gain boost or decay dB value at the central frequency location;

Q: Quality factor of the filter. The adjustable range of Q-value is 0.02~50;

When the type is parameter equilibrium, the Q value indicates the width of the bell-shaped frequency sound curve between the cut-off frequencies on both sides.

If the type is high or high pass, Q> 0.707, there will be some peak phenomenon in the filter response. If Q <0.707, the slope will be flatter and the roll will occur earlier;

There is a switch under each segment equalizer, The parameter settings for each segment do not work when closed, and the parameter setting of each segment does not work. The equalizer has a master switch indicating that the module is enabled or not enabled.

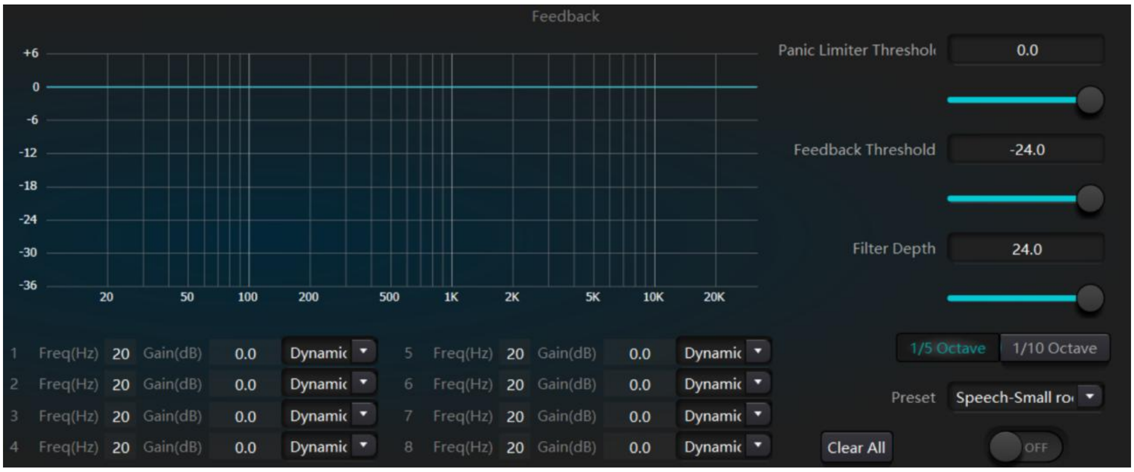
3.6. Feedback

When using feedback suppression modules, always combine good system design with engineering practice, rather than replacing good system design. Traditional methods such as limiting the number of microphones opened, minimizing the distance between the audio source and the microphone, locating the microphone and speaker for minimum feedback, and balancing the room for a flat response should still be used. After that, the feedback suppressor can be used to gain additional gain. The feedback suppressor does not magically solve the poor system design or improve the sound transmission gain beyond the physical limits of the system.

The feedback suppression module automatically detects and suppresses the acoustic feedback in the audio system. The module distinguishes between the feedback and the expected audio based on the characteristics of the signal. When a feedback presence is detected at a certain frequency, a notch filter is automatically added to the frequency point at which the feedback occurs to attenuate it. On the first addition, the notch filter only decayed slightly. If the feedback still exists, the trap device continues to attenuate according to the set parameters until the feedback disappears or reaches the maximum value of the parameter setting. A variety of user parameters can be used to accurately fine-tune the effect of the module.

The ringing output can lock the filters to prevent them from changing during the show. The filter settings can be copied to a dedicated notch filter module (such as the equalizer). Eight filters automatically cycle using filters set as automatic, in this way, to remove the filters that are only temporary.

Each channel has a feedback inhibition, drag the mouse input module to find the feedback inhibition module or quickly enter the feedback inhibition module by clicking the shortcut key on the right. To enable the feedback suppression module, click the open button to automatically detect the feedback point and eliminate it by using a narrow band filter. Each feedback suppression module has 8 narrow band filters.



The feedback inhibition module parameters are as follows:

Panic limiter threshold: This parameter tells the module, " Any level above this threshold is definitely feedback."When the signal level is above the feedback threshold, several situations occur: a) output gain is temporarily attenuated to control the speed at which feedback is established, b) output level is limited to prevent runaway, c) filter sensitivity increases to detect feedback faster. Once the output level drops below the threshold, the gain returns and the sensitivity returns to normal. This value refers to the peak level of the full-range digital signal. When this value is set to 0, it turns off this function;

Feedback Threshold: This reference tells the module, " Any level below this threshold is definitely not feedback." This prevents the module from detecting low-level buzz in Light music clips;

Filter depth: Set the maximum attenuation that a single filter will achieve.shallower settings may prevent the filter from doing too much damage to the signal, and may lead to worse control of feedback, especially in large narrow resonance systems;

Bandwidth: optional 1 / 10 and 1 / 5 Oct, using a constant Q value, the filter will not widen due to the depth increase. It is recommended that the filter group is used up in the voice environment, and when feedback often occurs, the bandwidth is set to 1 / 5 Oct, because it has a wider bandwidth and a larger impact range;

Preset: built-in four presets, "music big room", "music small room", "speech big room", " speech small room". These four presets fit the default settings for most applications;

Notch mode: each trap including three modes, "automatic, manual and fixed", set as automatic., This mode will automatically detect audio signals based on the set parameters. When the 8 filters are used up, a new feedback is detected. The module will find the "automatic" filter setting and use it to suppress the new feedback. When set to manual, the trap gain can be set manually; the filter parameters are always valid, not occupied by the new feedback point, and the next restart is valid. If you need to save these feedback parameters, click the Save Preset button;

Clear: Click this button to transiently clear all the filters. It removes the suppressed feedback point of the previous check, which is usually performed when re-debugging the feedback module;

Feedback suppressor can be used as a tool for system debugging to find feedback frequency points or as a preventive measure during normal operation. If you want to obtain a high system sound transmission gain and feedback suppression effect, it is recommended to follow the following steps:

(A) Reduce the system gain and reset all the filter parameters with the clear button

(B) Set the parameter value of the feedback inhibition module. Also reduce the panic threshold to reduce the feedback occurrence level

(C) Turn on all the microphones and slowly increase the system gain until the feedback occurs. Stop increases when feedback occurs

(D) Wait for the feedback suppression module to move, after the feedback disappears, continue to increase the gain

(E) Repeat until the system reaches the desired gain or all the filters have been assigned

(F) Change the panic threshold to a maximum level just above the desired non-feedback signal

At this point, each filter can be set to a fixed mode if necessary, if necessary, or the dynamic state can be saved to handle feedback that may occur during the performance. Another possible line is to copy the filter to the trap module (such as the equalizer). This can add more filter capabilities.

If speakers are included in the device used, it is recommended to use a compressor / limiter module for additional protection. Setting the appropriate limiter will ensure that the speaker is not damaged, even if all the trap filters are used out or the feedback suppressor cannot control the feedback, if the system gain is too high.

3.7. AutoMixer

Meeting room scenario,if multiple microphones are opened to the same gain level, and only one person in the speech, the result may not be very clear, other microphone will pick up the room noise, reverberation, etc., when these signals mixed with normal microphone signal, will greatly reduce the audio output quality after mixing, and the amplification system is very easy to scream, unable to get enough acoustic gain. To solve this problem, other temporarily unused microphones will need to be turned off. The automixer can complete the closing process and react much faster than manual operation.

A gain-sharing automixer is built in the processor. On each channel in the automixing matrix has a direct output that is not affected by the automatic gain and channel pusher, and only affected by the channel mute. A channel appropriate to a fixed volume, such as background music, needs to be maintained at a fixed level without the control of the automatic mixing; for example, keeping the chairman microphone normally open and its gain independent from the automatic mixing, the output of the channel can be adjusted directly in the output matrix routing. At this point, the automatic mixing button of the channel can also be closed, its gain will not be adjusted, and the signal level on that channel will not affect the gain on other channels.

The automatic mixing module has two sets of control parameters: the main control parameters and the channel control parameters.

(1) Main control parameters

Switch button: The bottom button turns the automatic mix on or off

Gain: Control the automatic mixing master output volume

Slope: Slope control affects the attenuation of lower levels. The els with low levels also decay more at a higher slope. The slope control works similarly to the ratio control on the extender. It is recommended to set this value around 2.0 or 2.0. If it is set to 1.0, the effect is equivalent to closing the automatic mixing of all channels; when set to 3.0, greater gain adjustment is occur with unnatural effects. The larger the set value, the more open channels, and the more the overall decay. When the slope is set to 2.0, there is an ideal gain sharing, which is the preferred value in use.

Response: Fast time ensures that the first word is not removed. When the time is slower, the operation is smoother. Practice shows that the effect is best when the response time is between 100ms and 1000ms. The automatic gain is designed to open the microphone much faster than turning the microphone off, so even with a 100ms response, the talking prefix is usually not subtracted. If set to a slower time of several seconds, the automixer response time has a longer hold time, and the last active channel remains open in several seconds.

(2) Channel control parameters

Automatic mixing: Each channel has an automatic mixing on/off button, which must be enabled for channels that need to participate in automatic mixing. Channels that do not have this button enabled will not participate in automatic mixing;

Prior: dodge is preferred, the corresponding channel will have a higher level of dodge priority.

Threshold: triggers the threshold of the dodge switch

After channel mute and pusher after automatic gain, any adjustment of these two parameters will not affect the automatic mixing operation. For example, even if one channel has been muted, the level of that channel can still reduce the level gain of the other channels if it is large. Need to mute a signal and prevent it from affecting the automatic mixing, and turn off the automatic mixing. The mute button on each channel mutes the channel in the mix, and will also output the mute directly. The channel pusher also controls the mixing level and the direct output level of the channel. Click on the text box and enter a dB value to accurately control the channel level.

3.8. Acoustic Echo Cancellation

Acoustic echo cancellation or AEC is a digital Audio signal processing technology. When the conversation occurs in two or more places, remote video conference is conducted through communication equipment and network. The AEC program increases the speech clarity of remote speakers by eliminating acoustic echoes generated in local rooms.

The echo cancellation module applied in the remote call can easily expand the sound of the remote voice signal locally and reduce the interference of the acoustic echo. Its basic working principle is to simulate the echo channel, estimate the echo possibly formed by the remote signal, and then subtract the estimate signal from the input signal of the microphone, so that the input voice signal no longer contains the echo, so as to achieve the purpose of echo elimination.

There is only one echo cancellation module in DSP Controller, and local input and remote input mixers are preset to realize multiplex signals to participate in echo cancellation, as shown in Fig. There is one parameter available for adjustment:

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

Note: The echo cancellation module setting needs to be used together with the matrix module setting signal routing.

3.9. Noise Suppression

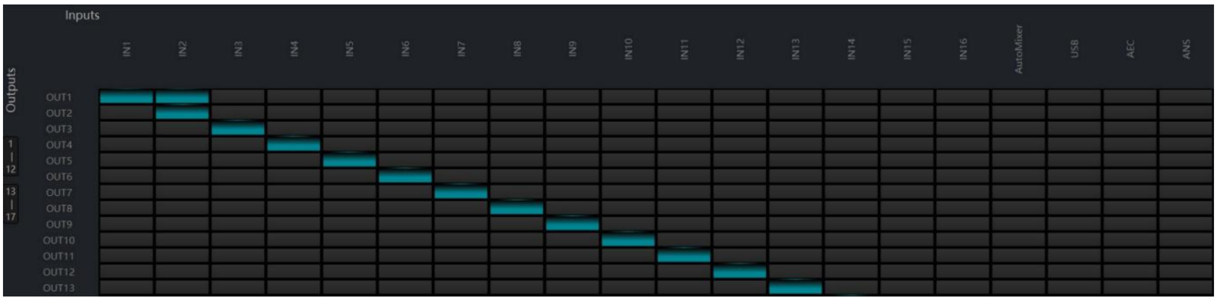
The noise suppression module can effectively remove non-human sound. Distinguish a human voice from a nonhuman voice, and treat a nonhuman voice as noise. An audio section containing human voice and noise is processed by the module, and theoretically, only human voice is left.

There is only one noise cancellation module in DSP Controller, and a multi-channel mixer is preset to achieve the multichannel signal involved in noise cancellation, as shown in Fig.

Inpression level: there are five types of 6, 12, 18,24 and 30dB. The meaning of dB is how much noise reduction dB, the greater the value, the greater the damage to the speech, which is unavoidable.

3.10. Matrix

The matrix has the dual operation function of routing and mixing. Lateral indicates input channel, longitudinal indicates output channel, default one-to-one input output, as shown in Fig. If it is necessary to mix the sound of input channel 1 and input channel 2 to output channel 1, point both lateral 1 and 2 on output channel 1. If inputs 1 and 2 are involved in automatic mixing, the output is unaffected by automatic mixing. Similarly, after setting the automatic mixing, echo elimination, and noise suppression module, it is also necessary to set the matrix to obtain the correct signal routing relationship.



3.11. High Pass & Low Pass Filter

Each output channel provides a high and low-pass module, consisting of a high-pass filter and a low-pass filter. The filter parameters are as follows:

Freq: Cut-off frequency of the filter, Bessel and Butterworth cut-off frequency are defined at-3dB, while Linkwitz-riley cut-off frequency is defined at-6dB.

Gain: Gain setting affects the full frequency band increase or attenuation of the signal.

Type: Select the filter type, with three types: Bessel, Butterworth, and Linkwitz-riley. Butterworth Has the flattest pass-band.

Slope: The excessive band attenuation size of the filter is 6,12,18,24,30,36,42,48dB / Oct. For example, 24dB / Oct indicates the amplitude attenuation of 24dB in each frequency difference in the transition band.

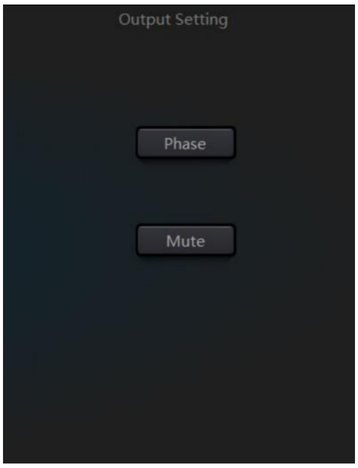
If you want to activate a High-pass or low-pass module, click the activation button at the bottom.

3.12. Delay

Ms: sets the delay time of the delay er. This value ranged from 1 to 1,200 ms.meters and feet are converted unit values of milliseconds.

Activation button: activate the specified delay module in the module and insert it into the audio signal path to increase a fixed delay time to the signal.

3.13. Output



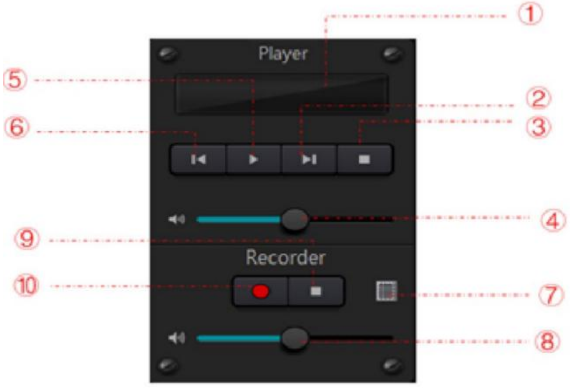
Phase: audio signal phase reversal 180°.

Mute: set the mute / non-mute.·

The output channel is the same as the input channel, and the same menu bar will appear when you right-click on the channel. It can be set according to needs.

3.14. USB Soundcard

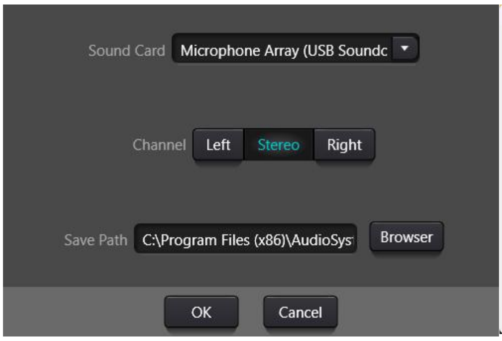
Using USB soundcard has two functions: one is to achieve recording; the other is the PC teleconference. The USB sound goes through the echo cancellation and noise cancellation modules and is easily connected to the teleconference. The USB play of the software interface is only used when the recording function is used.

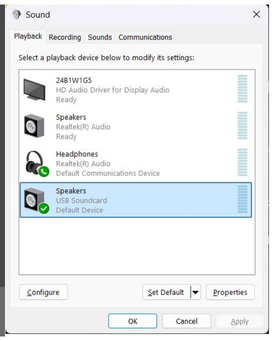


① Song play information, double-click to enter the playlist
② Next song
③ Suspend
④ Song playback volume size adjustment
⑤ play
⑥ Previous song
⑦ recording list
⑧ Sound recording volume adjustment
⑨ stops recording
⑩ started recording

Sound card setting:

Connect the DSP processor and the computer host through the USB cable. For the first connection, the computer will pop up the new hardware and automatically install the driver. After the installation, the USB Soundcard new device appears in the computer sound card list, as shown in the figure. Then select the USB sound card in the software playlist and sound card setting.





The playlist can operate on the song file, or save the song as a list, and open it directly in the next use. Click the open folder at the bottom of the playlist to select play songs, clear the song list, and enter the sound card setting interface, as shown in the figure.

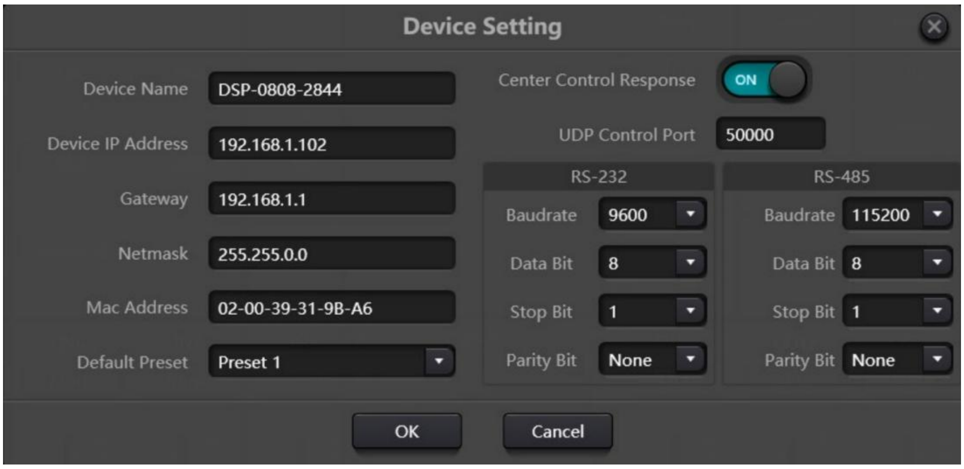
4.Setting Menu

4.1. File Menu

Open: In offline mode, click the Open pop-up File dialog box selection to open an existing preset file (suffix name: * dsppro). You can also right-button a preset file, using the DSP. The exe application opens;

Save As: saves the presets on the application to the local hard disk for easy copy and storage.

4.2. Device Setting



Set device name, network address, and serial port port rate information. The longest device name is 16 characters and 5 Chinese.

Default Preset: you can select two start preset modes, one is to specify any one of the 16 presets as the start preset, which will be started with each boot. The second is to select the last load preset, the last use before the power off as the preset for the next boot.

4.3. Group Setting

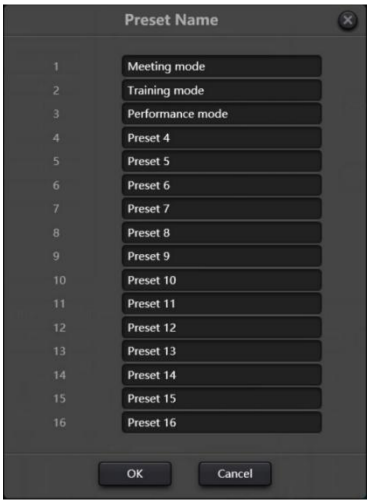
The grouping interface is divided into two labels: input and output,A channel can only participate in only one group. Under the same group, their channel volume adjustment and mute adjustment are synchronized. Other module parameters are not synchronized, which is the biggest difference from the Link function.

Each group can choose from 1 to the maximum number of devices.The maximum number of channels for the device depends on the model you purchased. The channel is set as a group, which will be distinguished by a color on the Home screen.

The relationship between grouping and LINK: When a channel has set up grouping, it will not participate in LINK. Means that grouping have higher priority than LINK . The difference between grouping and LINK is that grouping can only control the channel gain and silence, while LINK links all the parameters on that channel.

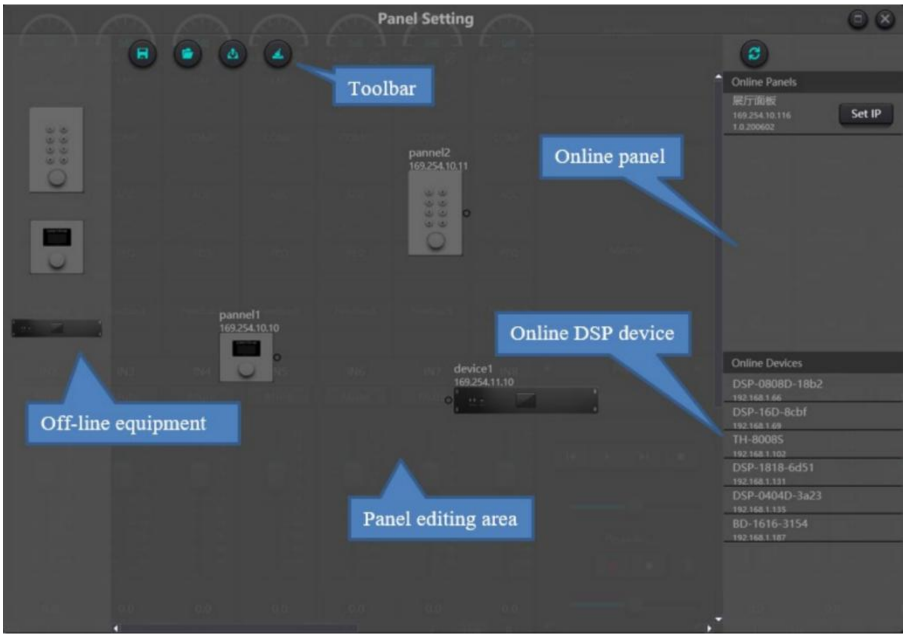
4.4. Preset Name

Supports name editing for 16 sets of presets, facilitating scene retrieval.



4.5. Panel setting

The panel settings include 2 panel types: key version and OLED screen version. Through the panel setting interface, multiple solid panels and DSP devices are connected through the connection, and then the purpose of panel control of DSP devices can be achieved through a simple setting panel.



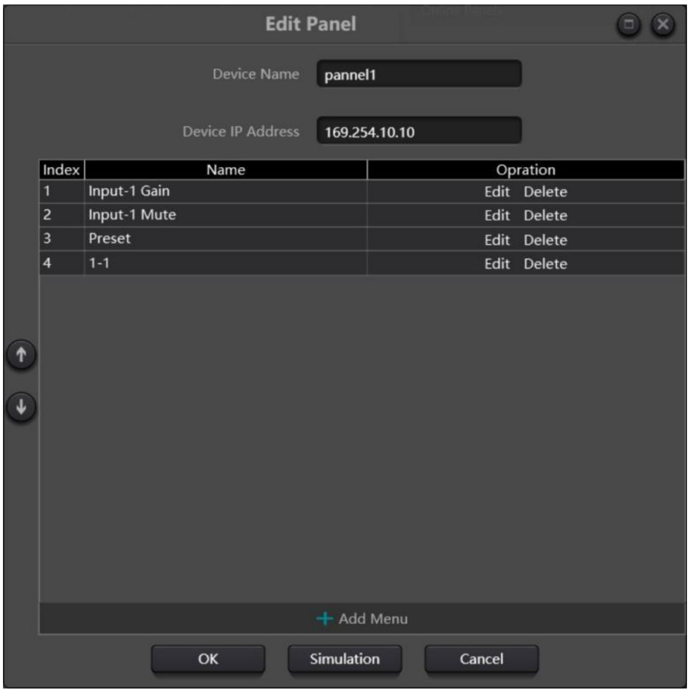
Offline equipment: suitable for offline editing status, debugging engineers first configure the panel parameters locally, and then download them to the online panel. Of course, you can also directly edit the online panel, in the online panel that column drag out to the panel design area double-click edit.

Notice that there is a small circle on both the panel and the device. Click on the small circle to pull out a line and select the destination device, which establishes the connection between the two devices.

Double-click on the panel in the design area to enter the panel configuration interface. The configuration of the two panels will be described separately below. After the configuration is complete, click on the download icon of the toolbar to download the panel configuration to the hardware

OLED screen panel:

The OLED panel contains a 1.3-inch OLED screen and a knob. O LED The screen display strategy is graded according to the menu. There are three types of menus: volume, button, and preset. In the panel design area, double-click a O LED panel to enter the panel detail settings. As shown in the figure below.



Click Add menu, pop-up menu selection box, select the corresponding menu items, OK. After the software menu configuration is complete, click the download icon of the toolbar to download the configuration to the panel hardware.

Panel operation steps:

1. In the main interface, the panel name and IP address are displayed through the left or right knob.

2. Press the button on the knob and the second row of the menu interface starts to flash to indicate the edit mode.

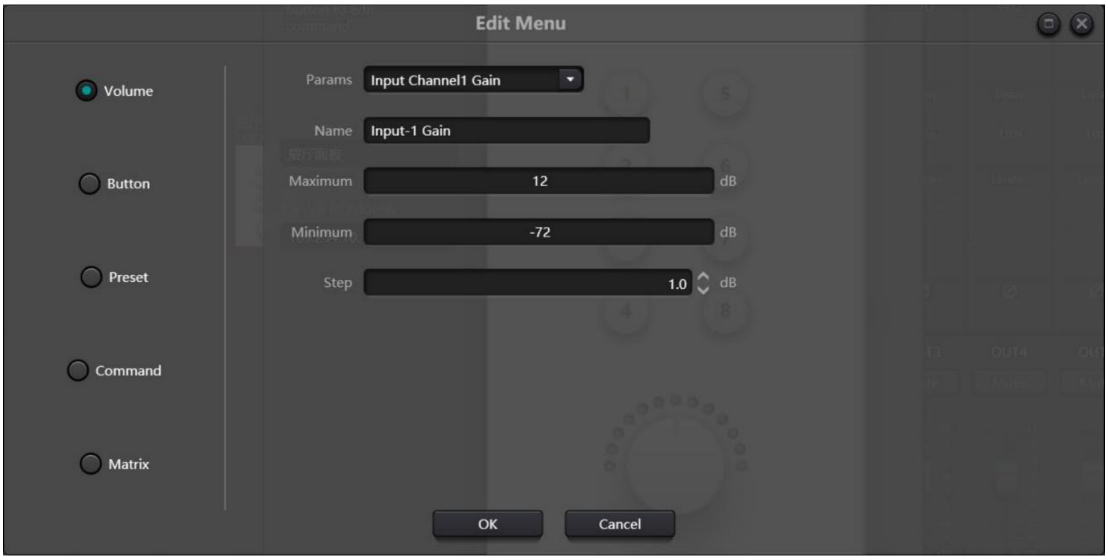
3. Use the knob left or right rotation to change the value.

4. Press the button on the knob again to exit edit mode and return to menu mode.

Key panel:

The button panel consists of eight buttons and one knob. The knob makes gain adjustment for use, and 8 keys can be programmed to achieve different functions. The key functions can be divided into four types: volume adjustment, mute, preset, and command. Drag an item to the specified button in the ribbon to program the button.

Similarly, after completing all the programming work, you can use the simulation button to see the correct configuration.



Panel operating light indicator:

1. The key light often indicates that the button is configured with the mute function.

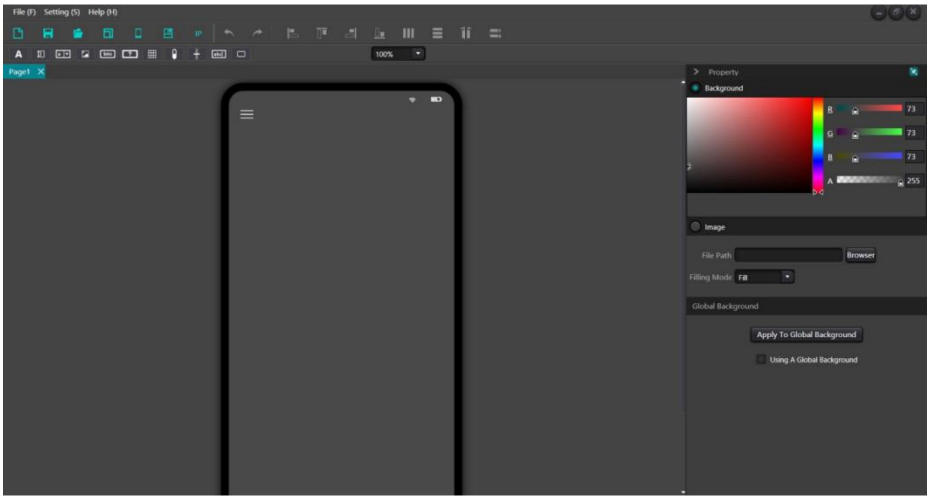
2. The button light flashing indicates that the button is configured with the gain function, and the configuration knob is used together to adjust the gain of the channel. The 13 lights around the knob represent the gain, changing with the size of the gain. All lights off represent a gain of -72 dB, while all lights on represent a gain of 12dB.

3. Press the button light and light it on instantly, indicating that the button is configured with a preset or command function.

Command-type function: Command data comes from the central control command. Refer to section

4.6. User Interface

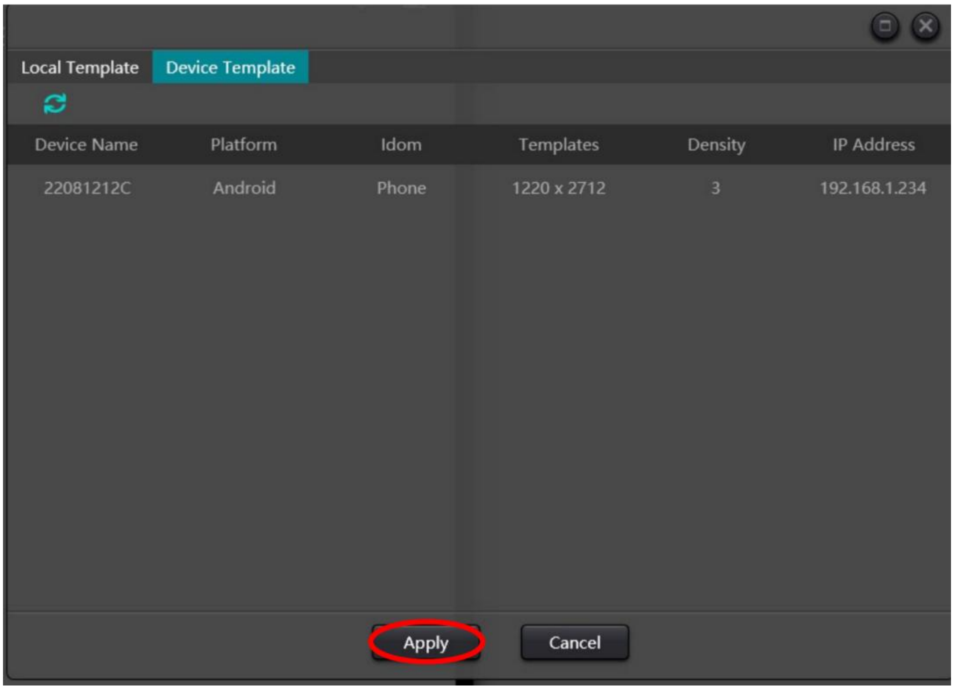
User Interface, which allows engineers to create custom interfaces that can be edited by an integrator and operated by field technicians or end users who understand the technology. Advanced security features can only be accessed by end users within the scope allowed by engineering or system designers.



4.7. Mobile Device Use

Online template selection

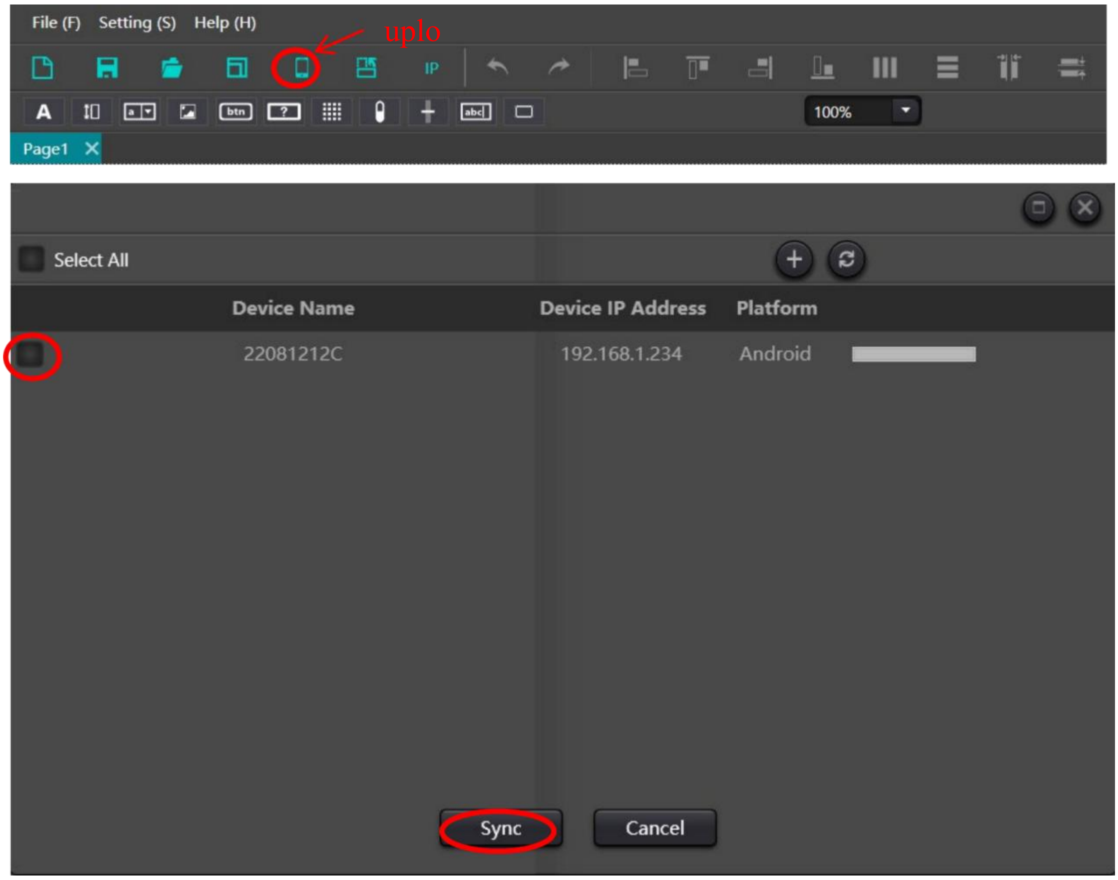
When the software is installed on the mobile terminal, be careful not to let the screen go out. In addition, the mobile terminal and PC must be the same LAN.



Edit the interface

Drag the upper control to change the corresponding properties

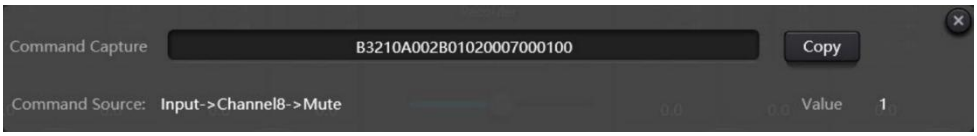
upload



Click the synchronous data to upload to the mobile terminal. When uploading, open the application on the mobile phone or tablet. If the upload timeout or the mobile device cannot be searched, you can turn off the software and open it again or check whether the devices are in the same network segment.

4.8. Help Menu

(1) Get the central control command



Open the central command window, click the parameters to be controlled on the interface, and the central command window displays the current command immediately. Copy the command to the control program to send the code to the device through the UDP/RS232/RS485 protocol.

(2) About

Display the version number, technical support contact information, copyright information, etc.

4.9. GPIO Setting

Open the GPIO setting software interface, the device has 4 GPIO, which can independently configure input or output.

Enter GPIO has preset, routing, gain, mute, command, analog to digital gain available.

Output GPIO has a preset, level, mute, command selection.

Enter the GPIO settings

Preset

Trigger type: high level trigger / low level trigger / high level trigger, low level cancel / low level trigger, high level trigger, namely up edge / down trigger / rise trigger, fall edge cancellation / down trigger, rising edge cancellation.

Preset: Switch to the preset when the jump type matches the trigger type of the software settings.

<table>
<tr>
<td>route</td>
<td>
</td>
<td>Trigger type: Ibid.
Input, output: Select the input channel mix that the output is for.
Mix / cancel the mixing action when the trigger condition is met.</td>
</tr>
<tr>
<td>gain</td>
<td>
</td>
<td>Trigger type: Ibid.
Channel: Select an input or output channel.
Step length: On the basis of the original gain of the channel.</td>
</tr>
<tr>
<td>Mute / Unmute</td>
<td>
</td>
<td>Trigger type: Ibid.
Channel: Select an input or output channel.</td>
</tr>
<tr>
<td>Command</td>
<td>
</td>
<td>Trigger type: Ibid.
Command: When the trigger type setting is satisfied, send the command code via R S232.</td>
</tr>
<tr>
<td>Analog to digital gain</td>
<td>
</td>
<td>Analog to digital gain ： very useful when external Potentiometer is connected. The appearance is similar to the rotary encoder, which is different from the encoder in that the potentiometer is analog, adjusting the size of the voltage and current, the encoder is digital, and the transmitted "0", "1" binary code.</td>
</tr>
</table>

Output the GPIO settings

Preset

Output type: high level / low level

Preset: When switched to the preset, the corresponding GPIO port output a high or low level.

level

Output type: high level / low level

Channel: Specifies the input or output channel

Level: When the specified channel level reaches the set level threshold, GPIO output high / low level. Conversely, to output the opposite level.

Mute

Output type: high level / low level

Channel: Specifies the input or output channel. When the channel is silent, the output is set at a high / low level. Unmute, and output the opposite level.

4. Control

1.External Control Programer

Series external control programming supports UDP, RS232 and RS485, The control protocol covers all the control parameters of the processor, including three parts: parameter control, parameter acquisition and preset call.

When using UDP control, the default port is 50000, which can be set in Device Settings through the host software.

When using RS232 and RS485 control, the default port rate is 115200, data bit 8, stop bit 1, no check bit. It can also be set up in the Device Settings set up. **When sending RS232 and RS485, the interval between messages needs to be maintained at least 200 milliseconds.**

If the central control needs to reply, please open the central control reply switch in the "Device Setting".

2.Control Protocol

For historical reasons, the latest control protocol is longer and fully compatible with the old fixed length control protocol. In the protocol, the fourth byte is used as version differentiation, 0-V 1 (historical version) and 1-V2 (current protocol version).

The differences between V1 and V2 is that V1 can control all the processing module parameters, but only one command can control one parameter. If a command is required to control multiple channels, use the V2 version. Or, there is a need to press a key to trigger the GPIO of the device to output high and low levels, or send a command through RS232 / RS485, then the V2 version will be suitable.

Software coding rules (12 bytes):

byte 1	byte 2	byte 3	byte 4	byte 5~132
0xb3	Message type	length	version number	data

V1 version:

Message type (byte2): three types, 0x21 (parameter control), 0x22 (parameter acquisition), 0x13 (switching scenarios)

Length (byte 3): Invalid.

0x21 (Parameter control):

At this time, Data byte5~12 are respectively:

byte 5~6	byte 7~8	byte 9~10	byte 11~12
module ID	parameter type	Parameter value 1	Parameter value 2

The module ID (byte 5 to 6) is assigned in Appendix A.

Parameter type (byte7~8) is shown in Appendix B.

Parameter Value 1 (byte9~10) has only one parameter, only the parameter value 1 is valid, such as controlling the compressor switch.

Parameter Value 2(byte11~12) is valid when there are 2 parameters, such as control input channel 1 silence. Parameter Value 1 Fill in input channel number (from 0) , Parameter Value 2 fill in 1 (mute).

Special case: matrix routing has three parameters, the first is the input channel number, the second is the output channel number, the third is the on and off of routing. At this point, byte9 of parameter 1 fills in the input channel number, byte10 fills in the output channel number, and parameter 2 fills in the on and off of routing.

0x22 (parameter acquisition):

The parameter acquisition rule is the same as the parameter control. The difference is that the acquired value is filled in the parameter value 1 and the parameter value 2.

0x13 (Switch to the scene):

Simply fill in the scene number (0-15) at byte5, and fill in 0 for byte6-12.

Note: v1 version can get code through the PC software menu bar: Help-. If custom development, use this protocol rule.

V2 version:

Message type (byte 2): three types, 0x21 (parameter control), 0x22 (parameter acquisition), 0x13 (switching scenario), 0x74 (other controls), 0x6e (Dante routing).

Length (byte 3): fill in the corresponding data area length according to the message type. The actual transmission time is variable length, according to the length of the data plus 4 bytes of header information, that is, the total amount of data.

1. Parameter control (0x21)

The format of the data area is now as follows:

byte5	byte6	byte7	byte8	byte9~72
Input / Output	Start channel	End Of The Channel	Parameter Type	Parameter Values

Byte 5: represents the control input or output channel, 0x2-input channel, 0x1-output channel.

Byte 6-7: Start channel number and end channel number, channel number starts from 0.

Byte 8: Parameter type as in V1 version, see Appendix B.

Byte 9-40: Fill in the parameter value from the start channel to the end channel, write from the 9th byte, each parameter value occupies 2 bytes.

2. Parameter acquisition (0x22)

The data area format is the same as the parameter control, Parameter values do not need to be filled in. The obtained parameters will be filled in this position.

3. Switch the scenarios (0x13)

Byte 5: complete the scene number (0-15).

Byte 6-8: fill in 0.

4. Other controls (0x74)

Other controls include but are not limited to: GPIO, RS232, RS485, central control response. The protocol format is as follows:

GPIO:

byte5	byte6	byte7	byte8	byte9	byte10	byte11	byte12
Control Type	Length	Reserve	Reserve	GPIO Direction	Origin GPIO	Finish GPIO	Price

The byte 5 control type is 1

The byte6 data length is fixed to 4 bytes.

byte9 GPIO Direction, set the input or output, value 0 represents the input, value 1 represents the output.

For byte10-11 starting GPIO and ending GPIO, the DSP device has 4 GPIO, indicated by serial numbers 0-7.

byte12 According to the byte9 GPIO direction, when set to the output, the field is filled in a high (1) / low (0) level. When set to input, the field is a return field, reading the GPIO level value on the device.

RS232/RS485:

byte5	byte6	byte7	byte8	byte9-132
Control type	Length	Reserve	Reserve	Data

Byte5 control type RS232 is 2, RS485 is 3.

The byte 6 data length is the current length of data to be sent through RS232 / 485.

By te 9-132 Fill in the data sent by RS232 / 485.

Central control reply:

byte5	byte6	byte7	byte8	byte9
Control type	Length	Reserve	Reserve	Reply switch

The byte5 Control type is 4.

The byte 6 data length is 1.

Byte9 turns on the central control reply switch for 1 and turns off the reply for 0.

5. Dante routing (0x6e)

Data area format is:

byte5	byte6	byte7	byte8	byte9-24	byte25-40
Dante Number Of Channel	Routing Switch	Reserve	Reserve	Subscribe To The Channel Name	Subscribe To The Device Name

byte5 Dante Channel number, the difference is that the Dante channel number starts from 1.

byte6 Dante Channel, subscribe / unsubscribe to the specified channels for the Dante devices represented by byte25-40. This specified channel is indicated by the byte 9-24 channel name.

3.Serial Port to UDP (RS232 To UDP)

The device supports RS232 to UDP, and the protocol format is as follows:

4 Byte prefix	4 Bytes	2 Bytes	1 Bytes	1 Bytes	128 Bytes
UDP:	IP address	Port	Length	Reserve	Data

RS 232 After receiving this protocol format packet, forward the data in the protocol to the device with the specified IP address and port.

For example, send data "HELLO DSP" to port 50000 of the device "192.168.10.22". The agreement order is:

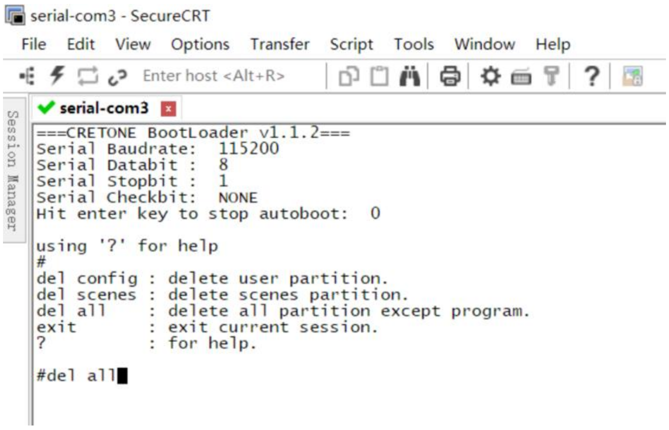
4 Byte prefix	4 Bytes	2 Bytes	1 Bytes	1 Bytes	128 Bytes
0x3a504455（′ :PDU′）	0x 1610A8C0	0x C350	0x09	0x00	“HELLO DSP”

Application situation: This function can be applied in many cases where the central control host has no network interface. As shown in the figure, the central control host is connected to the DSP device through the RS 232, and the DSP device is connected to the Ethernet through the network cable. The central control host controls any network device through a serial port to network command.

5. Frequently Asked Question

1. How to restore the factory Settings?

Connect to the computer via RS232 and run the serial port software (SecureCRT is recommended). The default serial port port rate is 115200 with 8 data bits, no parity and 1 stop bits. SecureCRT After connecting the serial port, press the enter key in the terminal interface to restart the machine and enter the bootloader guide dialog box, as shown in the figure.



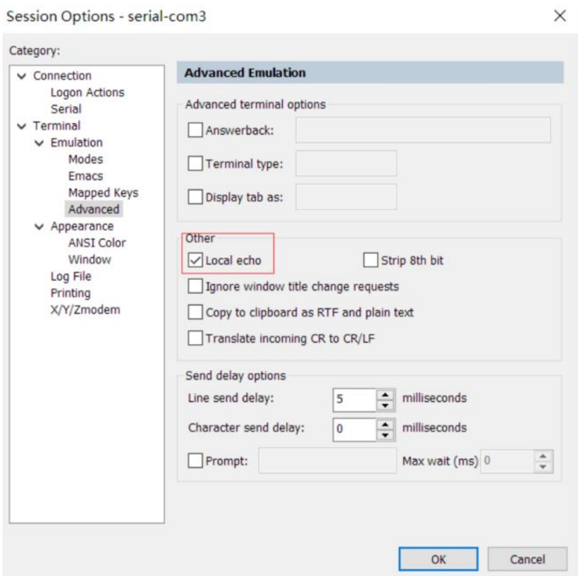
Command details:

del config: Delete configuration information, such as IP addresses and other network configuration. After being deleted, the device is restored to the default IP: 169.254.20.227.

del secens: Delete the preset. DSP, all 16 presets restored to default.

del all: Delete all partitions except the program.

Note: Some SecureCRT may not appear after installation, please tick "Local echo" in Options-> Session Options, as shown in the figure.



Appendix A: Module ID assignment

module name	ID	module name	ID
input source	299	Output channel 1-32 high and 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 time-delay device	231~262
Input channel 1-32 automatic gain	65~95	Output channel 1-32 limiter	263~294
Input channel 1-32 equalizer	97~128		
Input channel 1 – 32 for feedback inhibition	129~160		
Automatic mixing	161	Echo cancellation selector	162
echo cancellation	163	Noise suppression selector	164
muting	165		
mixer	166		
output	295		
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Appendix B: Module parameter type (1)

module name	parameter type	explain	module name	parameter type	explain
Input source	0x1	gain	Output	0x10	Gain step length
	0x2	mute		0x11	Link
	0x3	sensitivity		0x12	Channel level
	0x4	Phantom power switch		0x1	gain
	0x5	Signal generator type		0x2	mute
	0x6	Signal generator frequency		0x3	The channel name
	0x7	Sine-wave gain magnitude		0x4	Inverse polarity
	0x8	The channel name		0x5	sensitivity
	0x9	Inverse polarity		0x6	Gain step length
	0x10	Gain step length		0x7	Link
	0x11	Link		0x8	Channel level
	0x12	Channel level	Expander	0x1	switch
Delay	0x1	Bypass switch		0x2	threshold
	0x2	millisecond		0x3	ratio
	0x3	microsecond		0x4	built-up time
Parametric equalization	0x1	master switch		0x5	release
	0x2	Subsection switch	Compresser	0x1	Compressor switch
	0x3	frequency		0x2	Compressor threshold
	0x4	gain		0x3	Compressor ratio
	0x5	Q-value		0x4	built-up time
	0x6	type		0x5	recovery time
				0x6	gain compensation

Appendix B: Module parameter type (2)

module name	parameter type	explain	module name	parameter type	explain
Mixer	0x1	Mix switch	Feedback inhibition	0x1	switch
	0x2	Mix 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	clean up
	0x4	High-pass frequency		0x8	Panic threshold
	0x5	High-pass gain		0x9	Feedback depth
	0x11	Low pass switch	Automatic 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	built-up time
Automatic Mixing	0x1	Overall silent		0x6	release
	0x2	overall gain	Echo cancellation	0x1	Echo elimination switch
	0x3	slope		0x2	Echo elimination mode
	0x4	response time	Noise Suppression	0x1	Noise suppression switch
	0x5	Channel automatic switch		0x2	Noise suppression mode
	0x6	The channel mute	System control	0x1	The system mute
	0x7	Channel gain		0x2	System gain
	0x8	priority			
	0x9	Automatic mixing switch			