KEYER SERIES

Digital Processing & Control

Installation and Operations Manual

Catalog

1 Disclaimer

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1.1 Copyright Statement

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1.2 Safety Precautions

Please read all the instructions below before attempting to unpack, install, operate the device, or connecting the device to the power supply:

- Always take safety precautions to reduce the risks of fire, electric shock, and personal injury.
- Do not expose the device to rain or moisture, nor install it near water to reduce the risk of fire or electric shock.
- Do not expose the device to dripping or splashing of liquids.
- Do not insert any object into opening or empty slot of the device in any manner to avoid damage to the parts inside.
- Do not attach the power cable to the building surface.
- Only use the power adapter supplied with the device. Do not use a damaged power adapter.
- Do not place any item on the power cable. Protect the power cable from being walked on.
- Do not stack devices on top of each other to prevent overheating. Alternatively, provide adequate ventilation and leave around the device sufficient space to allow proper air circulation.

2. Packaging List

Item	Quantity	Image
DSP	1 Unit	
12-PIN PHOENIX TERMINAL	Individual	
3 POSITION PHOENIX TERMINAL	1 PC	
POWER CORD	1.2m IEC Power Cord	
SMALL SCREWDRIVER	1 Hand	
WARRANTY CARD	1 Sheet	

4. Product parameters

Parameters	Keyer 44	Keyer 88	Keyer 88D	Keyer 1616
Analog Input Channel	4	8	8	16
Analog Output Channel	4	8	8	16
Dante Input Channel	N/A	N/A	8	N/A
Dante Output Channel	N/A	N/A	8	N/A
Number of GPIOs	8	8	8	16
232 Number of serial ports	1			
485 Number of serial ports	1			
ETHERNET	1			
USB	1 x USB Bidirectional interface			
Maximum analog gain	51dB			
Phantom power	48V			
Input and output quantification	48KHz/24bit			
A/D dynamic range	120dB			
Input Common Mode Rejection	70 dB @ +18dBu@80Hz			
Input impedance	20kΩ balance, 10kΩ unbalance			
Maximum input	18dBu			
D/A dynamic range	120dB			
Channel isolation	>100dB			
Frequency response	20 to 20kHz (±0.2dB)			
Total Harmonic Distortion (THD+N)	≤0.004% @1kHz, +4dBu			
Output impedance	100 Ω balance, 50 unbalance			
Maximum output	18dBu			
Working power	AC 110V/220V 50Hz/60Hz			
Operating temperature	0~40°C			

5. Product appearance & description

5.1 Description of the front panel

① PWR: power supply indicator (red), the light is on to indicate that the device is powered properly, otherwise the power supply is abnormal.
② SYS: a status indicator (green), the light flashes to indicate that the equipment is operating normally, otherwise the equipment is faulty.
③ USB Bidirectional interface used for audio recording, audio broadcasting & video conferencing.

5.2 Description of the backplane interface

① Isolated ground.
② POWER: 220V AC power input connector and power switch.
③ ETHERNET: 10M/100M Ethernet interface for connection to a console (PC, router, etc.).
④ RESET: restores the factory settings, press, and hold for 6 seconds, and the device will automatically reboot.
⑤ RS232: support for central control commands and camera tracking, RX: receive data, TX: send data, G: ground.
⑥ RS485: support for central control commands and camera tracking.
⑦ GPIO: GPIO control. (GREEN color coded)
⑧ OUTPUT: Analogue output. (BLACK color coded)
⑨ INPUT: analog input. (ORANGE color coded)

6. Software operating instructions

6.1 Web control and software downloads

The device's factory default IP address is 192.168.1.200. The subnet mask is 255.255.255.0. Please enter the address of this network Range into the PC first so that the gadget can connect properly.
Once the device has done booting, use your web browser to navigate to the address "http://192.168.1.200/", as shown below.

Net Framework PC software, which supports XP, Win7, and Win8 operating systems.

Before installing the software on the PC, please ensure that Microsoft. When installing the software, some systems (e.g., WIN8) will prompt you with "User account control information", please click on the "OK" button to upgrade the software permissions.

6.2 Main interface

① Menu bar: file, settings, mode, view, help.
② Flow control area: flow chart of the audio data, with detailed settings for the parameters of each process by clicking on the icons.
③ System lock: to prevent misuse.
④ Scene drop-down box: switching scenes.
⑤ Saving scenes: the user can save the current scene on demand.
⑥ Connecting the device: entering the IP address and account/password to log in.
⑦ Search for a device: click to refresh the current device list
⑧ Login able devices: under the list of devices, double-click to select the device name and enter the account/password to log in.
⑨ Device list: the display of current online devices.
⑩ Input channel quick control area: the display of the level and gain of each channel, as well as quick enable/disable of the individual processors by supporting copy/paste functions.
⑪ Output channel quick control area: the display of the level and gain of each channel, as well as quick enable/disable of the individual processors by supporting copy/paste functions.

Login screen.

Default Username:admin Password:123456

Account passwords can be changed by going under the settings module. Before logging in, please confirm whether the IP address of the device and the client IP are in the same network segment, if not, double-click on the device name in the device list to modify or change the client IP to the same as the device side.

Click OK and the status bar prompts the following.

6.3 System flow

Signal processing flow chart

Standard configuration	Inputs: test signal/mute/expander/10-band EQ high/low/compressor/auto gain. Outputs: Delay / Crossover / 31-band Graphic EQ / Limiter / Output Opposite / Mute
Advanced configuration	(AFC) Feedback Canceller, (AEC) Echo Canceller, (ANS) Noise Canceller, Automixers, Ducker

6.4 Input section

6.4.1 Input settings

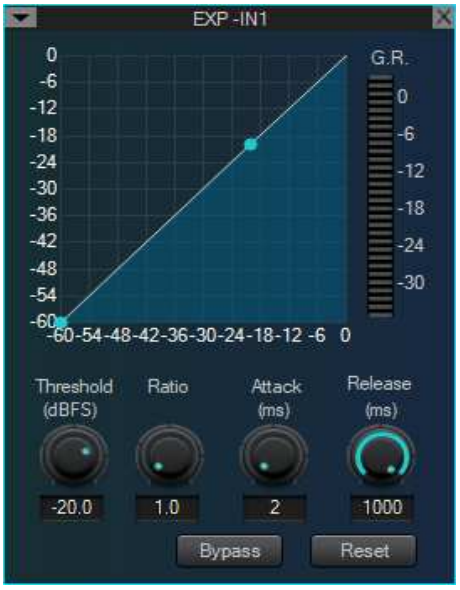
The input signal can be either an analog signal, a test signal generated internally by the device or in the case of Dante version, a digital network signal.
The amplification of the input analog signal can be adjusted by adjusting the sensitivity, selectable; from 24 to -27, in 3dB steps.

Mute: muting of the channel when selected.

Inverted phase: processing of the signal by flipping it 180 degrees out of phase.

Phantom power: For powering condenser microphones
Note: Do not switch on Phantom power for non-condenser microphones.

Test signals: including sine, pink and white noise, with the test signal, enabled the system automatically blocks the signal from the analog input.



6.4.2 Expenders

The expander is used to increase the dynamic range of the input according to the user's needs. When the input signal is less than the "threshold", the expander compresses the input signal by a set "ratio", and the output level = threshold - (threshold - input level)/ratio.
When the input signal is greater than the "threshold", the output is 1:1, output level = input level.

Threshold: The expander can be started only when the signal exceeds this threshold (allowing the transmission of the signal). As a matter of fact, the signal is often set as the ambient noise.

Ratio: It refers to the slope below the threshold point on the gain curve. When the slope is set at a high level, the gate movement will start.

Starting time: It refers to the time required to start the expander when the duration of the input signal exceeds the threshold. Shorter starting time allows us to start the expander more quickly.

Release time: It refers to the time required for the gain to be restored to a value lower than the threshold when the input signal is lower than the threshold.

6.4.3 Parametric equalizer

The equalizer is mainly used to correct the frequency range that is overemphasized or gets lost, no matter 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 get it simplified, the equalizer can be used to change the tone of the signal.

The equalizer has the following control parameters:

Reset:all filter parameters are reset to their default's.

Active: whether the equalizer is active or not.

Bypass/Active: whether the segmented equalizer is active or not.

Centre frequency: the center frequency to be equalized.

Gain: the gain/attenuation at the center of the frequency.

Octaves: i.e., the range of influence of the segment around the central frequency, the higher the greater the bandwidth and the greater the range of influence.

Low shelf: Applies a quantitative boost or attenuation to all frequencies below the cut-off frequency.

High shelf: Applies a quantitative boost or attenuation to all frequencies above the cut-off frequency.

6.4.4 Compressors

The compressor is used to reduce the dynamic range of the signal higher than the threshold set by the users, and to maintain the dynamic range of the signal lower than the threshold.

The compressor has the following control parameters:

Ratio: The compression ratio of the input to the output.

Attack: The speed of gain reduction processing associated with the start of this compressor. The shorter the start-up time, the more significant the instantaneous change in signal and the short gain decay makes listening uncomfortable.

Release: The release time determines the moment-to-moment gain variation of the compressor. Fast release times increase the level of subjectivity, while slow-release times are more helpful in keeping the level under control.

Output fader: The fader controls the output gain of the module. If the compressor reduces the signal level significantly, the perceived volume may need to be maintained in the output gain boost.

6.4.5 Automatic gain

The purpose of an automatic gain control is to bring a signal of an uncertain level to the target level while maintaining the dynamic range of the volume. Typical applications: For example, when a user is speaking into a microphone, the distance between the mouth and the microphone may vary from time to time, causing the output volume to vary from time to time or even a feeling of intermittent speech.

Automatic gain means that the input signal below the threshold is output in a ratio of 1:1 by setting the threshold. The level above the threshold is boosted directly according to the ratio, setting the target level so that the sound signal can be output in a stable manner.

Threshold: when the signal level is below the threshold, the input/output ratio is 1:1. When the signal level is above the threshold, input/output = ratio. Set this threshold to a level slightly above the noise ratio of your input signal.

Target level: the desired output signal level.

Automatic gain control is to automatically control the amplitude of the gain by changing the input and output compression ratio. Amplification of the signal when a weak signal is an input ensures the strength of the output sound signal; compression of the signal reduces the sound output amplitude when the input signal strength reaches a certain level.

6.4.6 Automatic mixing

The range now offers automatic mixers using "adaptive gain sharing" technology.

Automatic mixers are primarily used to automatically operate and control how a conventional mixer with many voice inputs will output the desired result. Consider a typical conference room scenario with ten attendees and one microphone each. If all ten microphones are turned on at the same time, and only one person is speaking. As a result, the output will not be ideal because the other nine microphones pick up room acoustics, reverberation, etc., reducing the production of the whole system.

6.4.6.1 Gain-shared Auto Mix

LVL: Auto Mix Channel Level Bar

G: The fader can be adjusted, or specific numbers can be entered in the text box to boost and attenuate the volume, and the channel fader controls the mix level and direct output level for that channel.

PR Priority: When multiple microphones speak simultaneously, the higher priority channel outputs the sound first. This control defines the priority with a between 0 (lowest priority) and 10 (highest priority); the default is 5 (standard priority). If all channels have equal importance, set the priority of all channels to 5.

Auto: controls whether the channel is added to the Auto Mix algorithm

Enable/Pass-through: whether the auto-mix algorithm is active.

Gain: control the gain or attenuation of the output channel by turning the knob or entering a specific in the text box

Response time: the time it takes for the auto-mix algorithm to take effect.

Slope: gain ratio distribution

Reset: resets all parameters to their default s and switches to gain-shared auto-mix

6.4.6.2 Threshold-type automatic mixing

G: Input channel gain level bar
Threshold: Passing is only allowed if the channel level reaches the threshold. Push the fader or enter a specific parameter in the input box to set this channel threshold.
Depth: control the attenuation of the Auto Mix channel
Background percentage: boost increases the threshold from the original threshold for each channel.
Gain: control the output channel signal gain or attenuation by turning the knob or entering specific parameters in the input box
Hold time hold: automatic mixing of channel gain level bar hold times.
Last Mic On: The channel with the last input signal is always open.
ID Gating: simultaneous input of signals from multiple channels, passing only the channel with the highest signal level.
Drop-down box options: Auto mode: channel level reaches the threshold to allow pass;
FIL blocking mode: multiple channels input signals at the same time, only one channel is allowed to pass; PR priority: multiple channels input signals at the same time, the channel with the higher priority has priority to output signals.

6.5 Ducker

Paging input: controlled signal input channel
Background input: Controlled signal input channel
Threshold: control signal level above the threshold, blocking out the controlled signal
Depth: Controlled signal attenuation
Start time: time for the ducker algorithm to take effect
Hold time: algorithm hold time when there is no control signal input to the input channel
Release time: algorithm release time
Reduction: Reduction level.

6.6 Feedback, Echo, Noise cancellation

Feedback: The feedback eliminator selects the signal to be processed, with the processed signal selecting the output channel in the mixer.

Echo: sets the signal to be processed by the echo canceller, and the processed signal selects the output channel in the mixer.

Remote input: local MIC output, i.e., the signal to be echo- processed. Echo far input: the reference signal.

Noise: select the signal to be processed for noise cancellation; the processed signal is output in the mixer by selecting the corresponding channel.

Mixing: mixing the signal of the selected input channel to the corresponding output channel.

Feedback suppression: used to suppress the whine generated between the microphone and the loudspeaker in the PA system, thus capturing the frequency causing the whine for attenuation, thus ensuring sound quality, and preventing burnout of the amplifier or loudspeaker.

Example 1

Feedback eliminator and mixer association operation.

The signal from channel two is processed for feedback and output in channel 3, configured as shown on the left.

1. the selection of input channel 2 in the feedback eliminator indicates that the signal from input channel two is fed to the feedback eliminator for processing.

2. Check the point corresponding to OUT3 in the "Feedback Cancellation" column of the mixer to indicate that the result of the feedback cancellation process is sent to the output of output channel 3.

Example 2

Echo canceller and mixer association operation.

The local signal is input channel one, and the remote signal is input channel 5, i.e., the signal about channel 5 in channel 1 is removed and output in channel 7, configured as shown on the left.

1. the selection of input channels 1 and 2 in the echo canceller indicates that the signals from input channels 1 and 2 are fed to the echo canceller for processing.

2. Check the point corresponding to OUT7 in the "AEC/Echo Cancellation" column of the mixer to indicate that the result of the echo canceller processing is sent to the output of channel 7.

Example 3

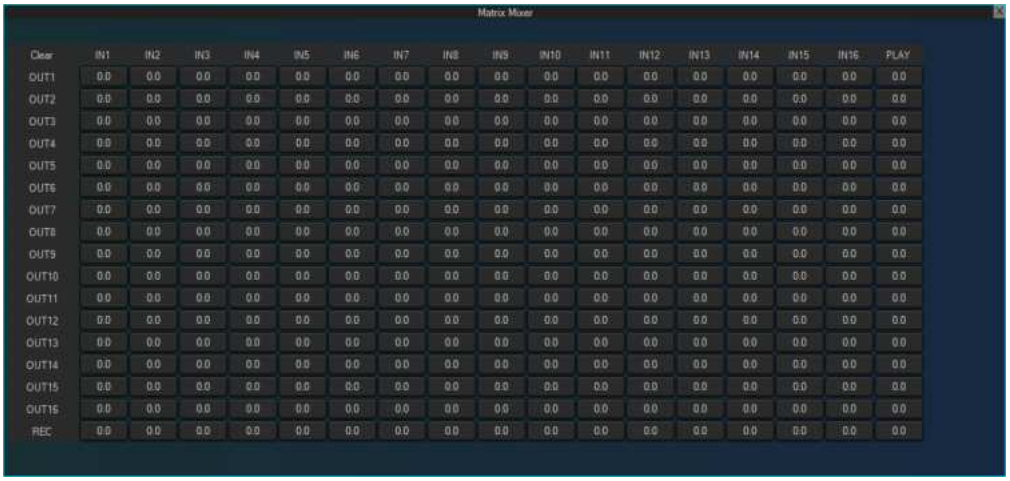
Noise canceller and mixer-associated operation.

The signal from channels 3 and 5 is noise cancelled and output in channels 1 and 2 in the configuration shown on the left.

1. the selection of input channels 3 and 5 in the noise canceller indicates that the signals from input channels 3 and 5 are fed to the noise canceller for processing.

2. Check the points corresponding to OUT1 and OUT2 in the "ANS/noise cancellation" column of the mixer to indicate that the result of the noise cancellation process is sent to the output of output channel 1 and output channel 2.

6.7 Matrix mixing



Control of mixing logic
Column: Input channels
Rows: Output channels

Mixing output gain: adjustable gain via faders (12 to -72dB).

6.8 Output section

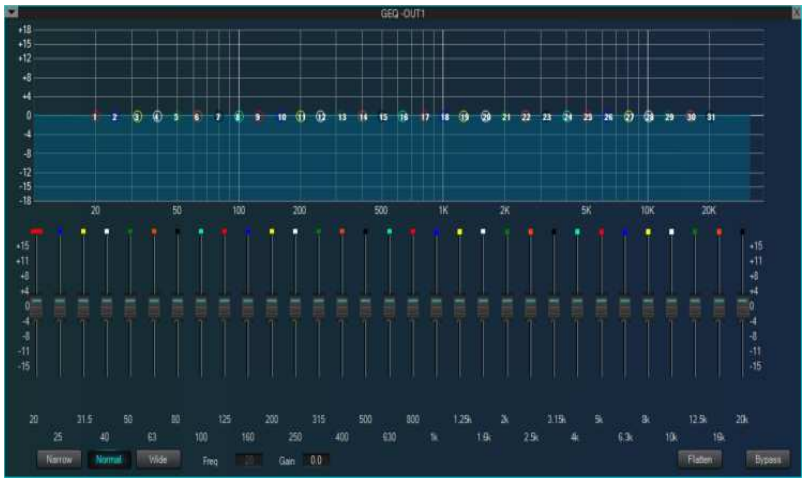
6.8.1 Delayers

The time interval between the input of the signal to the processor and the output of the processor is generally used to produce effects such as reverberation or echo but also for processing auxiliary speakers in larger applications.

6.8.2 Crossovers

High-frequency pass-through/enable enables and disables the high pass filter.
Low-frequency pass-through/enable enables and disables the low-pass filter.
High-pass frequency: the cut-off frequency of the high-pass filter.
Low-pass frequency: the cut-off frequency of the low-pass filter.

6.8.3 Equalizer



Equalizer: 31-band graphic equalizer.
Narrowband: narrowband equalization filter.
Normal: conventional equalization filter.
Broadband: broadband equalization filters.
Center frequency: indication of the center frequency of the current equalization filter.
Gain: indication or control of the gain of the current equalization filter.
Leveling: indicates or controls whether the equalizer is enabled.
Reset: restores all band gains to their default state.

6.8.4 Limiter

Straight through/enabled: enabling or disabling the limiter.
Threshold: the starting level for limiting; the limiting processing module is activated when the signal is above this limit.
Recovery time: When the input signal falls below this setting, the sound channel will not be switched off immediately but will be delayed by this setting. During this time, the sound channel will remain on as long as the signal is above the "threshold" limit.
Compression: the difference between the signal after processing by the limiter and the input signal.

6.8.5 Output settings

The possibility of setting the outputs to be muted and inverted.

7. Other functions

7.1 Channel control

Input.
(1) The name of the channel that can be modified by double-clicking on the channel name.
2) M, P, E, Q, C, G indicate the corresponding input channel shortcuts.

M Checked for Mute	Q Checked to enable and disable the equalizer
P Checked to turn phantom power on and off	C Checked to enable and disable the compressor
E Checked to enable and disable the expander	G Checked to enable and disable Auto Gain

(3) Level meter showing the current input level of the output channel.

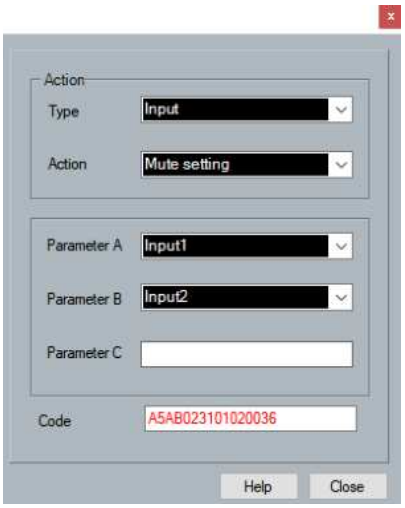
(4) The fader adjusts the digital gain of the current output channel.
(5) The level meter shows the current input channel's input level. The hidden channels can be displayed by dragging left and right or scrolling the middle mouse button, and the channel parameters can be copied and pasted by clicking on the channel.

Output.
(1) Double-click on the channel name to modify the channel name.
2) M, D, X, Q, L, and V indicate the corresponding input channel shortcuts.

M Checked for Mute	Q Checked to enable and disable the equalizer
D Checked to enable and disable the time delay	L Checked to enable and disable the limiter

(3) A level meter showing the output level of the current output channel.
(4) The fader adjusts the digital gain of the current output channel.
(5) The level meter shows the current output channel's output level. The hidden channel can be displayed by dragging the left and right or rolling the middle mouse button, and the channel parameters can be copied and pasted by clicking on the channel.

7.2 Central control commands Generator

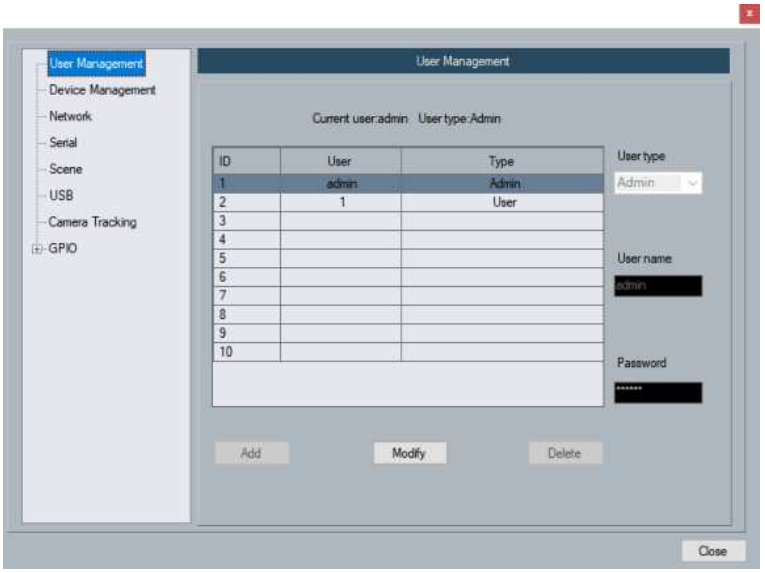


The central command generator converts frequently used operations into a 16-character command code that external devices can quickly call. Each of these commands contains up to three different sets of parameters.
Types of control commands: scene, input, output, mix, parametric EQ, graphic EQ, expander, compressor, auto gain, delay, crossover, and limiter.

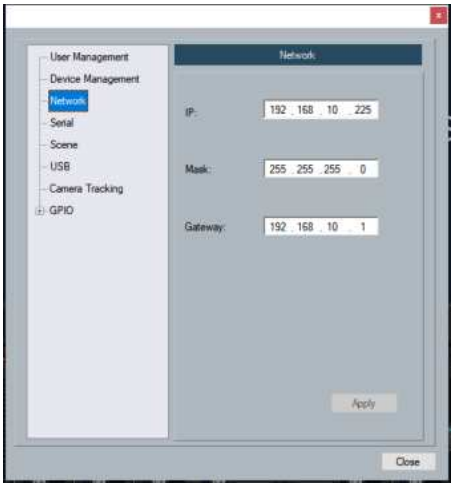
7.3 Device setup

Device settings include user settings, network settings, serial port settings, scene settings, camera tracking, and GPIO.

(1) User settings

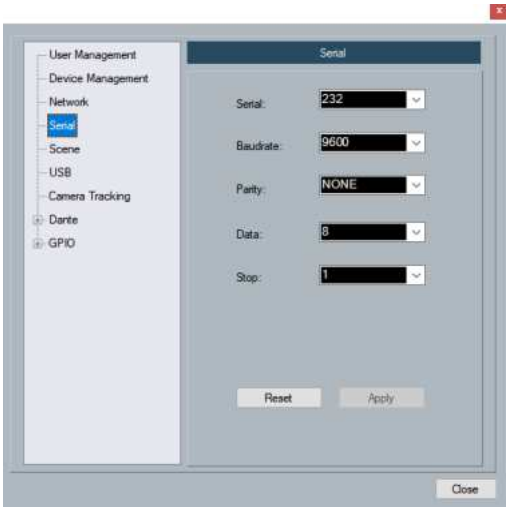


1. The initial username of the device is admin/password 123456. admin can add, delete, and modify all user information; ordinary users can only modify personal information.
2. Modify user: First, select the user you need to modify in the user list; the username and password edit box display the information of the currently selected user, enter the new information, and click the "Modify" button.
3. Delete user: select the line in the user list to be deleted and click the "Delete" button to delete the user.
4. Add a user: select an empty line in the list on the left, enter the new user's information in the username and password edit box on the right (should be empty), and click the "Add" button to add a new user.



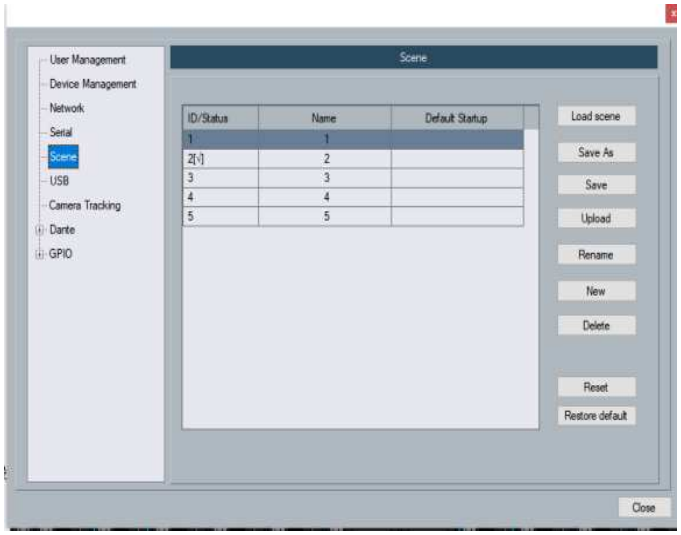
(2) Network settings

To view and modify the network address information of the device, enter the IP address, subnet mask, and gateway in the corresponding fields and click the **Apply** button to complete the repair.



(3) Serial port settings

To view and modify the serial port information of the current device, click the "Apply" button after setting to modify the serial port information of the current device; to restore to the default, click the "Reset All" button directly, and the items cannot be empty when setting.



(4) Scene setting

1. Modify the name: modify the name of the selected scene.
2. Uploading scenes from the PC and overwriting the selected scenes.
3. Save scenario: Save the current run parameters to the selected scenario.
4. Save as: save the parameters of the current run as a scene to the PC.
5. loading scenes: enabling the currently selected scene, usually for scene replacement.
6. Restore factory settings: restore all scene configurations to their default configuration.

(5) Voice tracking

Save speech tracking parameters: Each scene can save different speech tracking parameters; first, click "Apply" after setting in the speech tracking interface; then click "Save" in the "Scene Control" interface. "Save" in the "Scene Control" screen, the speech tracking parameters are automatically saved to the corresponding scene

1. camera tracking type: there are camera tracking and custom commands. Camera tracking is used for channel input signal control of camera rotation; custom command sending is used for channel input signal control to send the corresponding custom command to the corresponding port.
2. Tracking threshold means that the detected input signal must be greater than or equal to the tracking threshold for the system to enable the tracking parameter automatically.

3. Default MIC: When all the microphones have not input, the camera is rotated to the position set by the default MIC, or the associated command defined by the default MIC is sent. A # symbol is a virtual number that can only be used to set the default microphone.

4. Reaction time: the maximum interruption time of a valid signal. If a microphone is used for speech, the response time is set to 3 seconds, and the signal is continuously valid within 3S of a pause in the middle of the speech but invalid if it exceeds 3S.

Switching time: The shortest speaking time required for the camera to switch to a valid position. If you use a microphone to speak, the duration of the speech must be greater than the "switching time", then the channel signal is considered valid, and the camera will automatically switch to the set position. Usually, the "switching time" usually exceeds the "reaction time".

6. Rotation time: the interval of sending a camera switch command or a custom command, such as 0, means special processing is only triggered once.

7. The microphone number generally corresponds to the input channel of the device, i.e., the channel number to which the microphone is connected. The microphone number with a # is a virtual number and can only be used to set the default microphone.

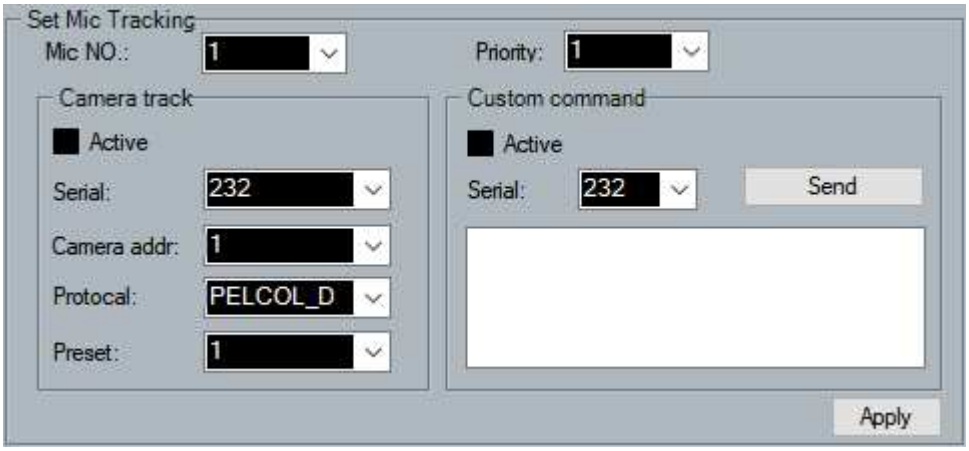
8. the smaller the priority number, the higher the priority level; when the priority level is the same by the trigger priority order, such as two microphones speaking at the same time, the camera automatically rotates to the preset bit corresponding to the microphone with the smaller priority number (i.e., the higher priority level) or sends the command corresponding to the microphone with the smaller priority number (i.e., the higher priority level); but if the priority levels of two microphones are the same, the signal checked first shall prevail.

9. Enable this MIC setting: you can set all the MIC parameters in advance, but when you use it, enable only some of them, depending on the actual situation.

10. preset points, serial number, camera address, protocol, and camera-related, must correspond to the actual connection with the camera

Custom command means that when the matrix's microphone checks that there is an input signal (usually when someone speaks), it will automatically send the corresponding command to the defined serial port. Secondly, you can also set the command in advance. However, the device must check the "Enable custom command" to send automatically. However, you can still click the "Send" button to send the commands in the input box to the specified serial port at any time.

12. Click on "Save" to save the parameters to the device, at which point the channel's microphone is associated with the corresponding camera address. Then use the "Enable microphone settings" option to determine if the microphone settings are valid when tracking is enabled.



13. Camera Setup is a camera commissioning interface, which generally debugs the camera position before tracking begins, and the final parameters for this section will be saved on the camera.

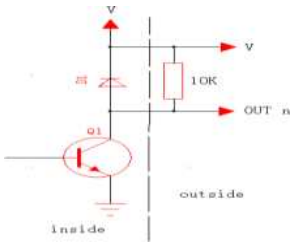
14. Firstly, serial port settings, there are 2 serial ports (232, 485), corresponding to the back panel port to which the head is connected.

15. followed by the camera address and protocol type, camera address please refer to the actual address of the camera, protocol, and camera model related.

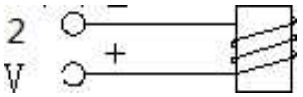
16. The final preset point number is a user-defined identifier for the camera, and then adjusting the up, down, left, right, and focal length, aperture, and other parameters will define the camera position and settings.

17. finally, click "Save" to save the parameters to the camera, "Clear" to delete the information of the current preset point, and "Call" is used to view the current preset fact saved by the camera position.
Note: A camera address can contain more than one preset point, but a preset point corresponds to only one camera address. Camera settings and microphone settings have several parameters: preset point, serial number, camera address, and protocol, which need to be considered for the actual application.
232 and 485 port switching methods.
If a camera is first connected to an audio processor on the 232 port and then disconnected from the other audio processor in the same position, the camera is then connected to the other audio processor.
The camera is connected to the 485 ports. In this case, the parameters in the camera remain unchanged, and only the microphone settings need to be adjusted, but the 485 port should be selected.

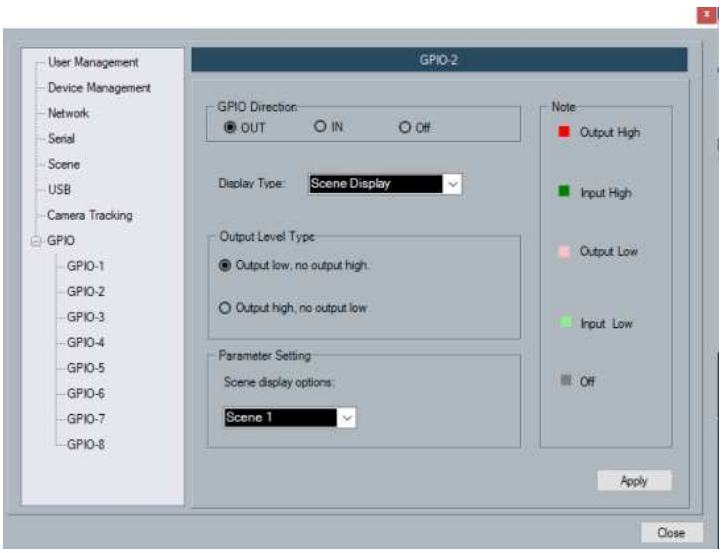
(6) GPIO settings

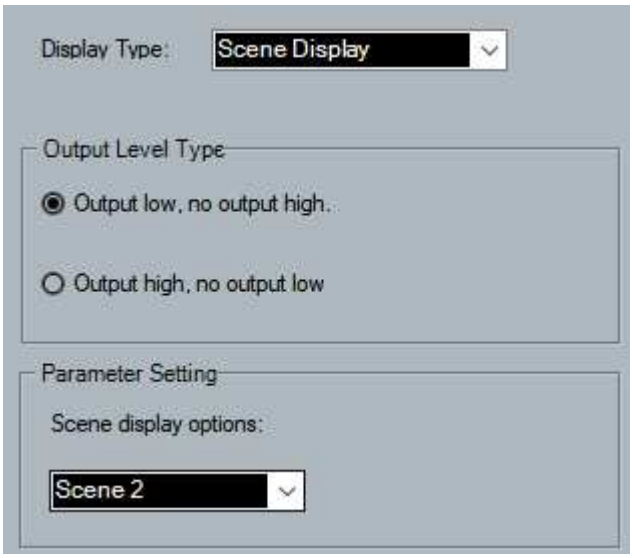


Output connection I.
First, connect a 10K/0.25W resistor between one of the GPIO pins (e.g., port 2) and "V" on the device (as shown); this pin will output a low 0 or a high 1 depending on the state of the matrix, which can be used to trigger another GPIO or another device.



Output connection II (for control).
Driving relays: for controlling alarm devices, etc., with built-in current-continuity diodes.
Output settings and use.
PC connected to devise -> setup menu -> device setup -> corresponding GPIO (in this case GPIO-2) -> direction set to "output".





IN KEYER 44,88 & 88D THERE ARE 8 GPIO SETTINGS

IN KEYER 1616 THERE ARE 16 GPIO SETTINGS

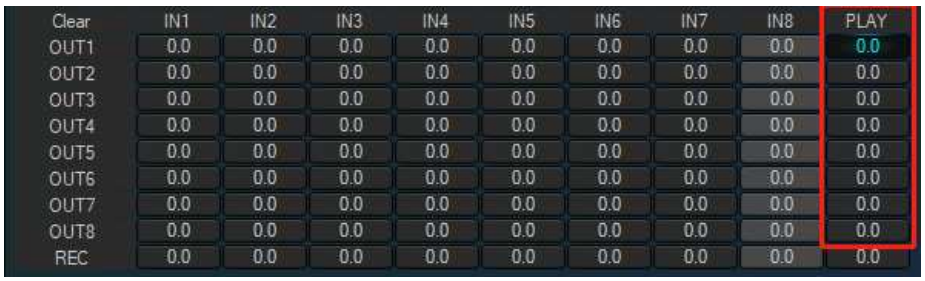
Display: select the scene to display, set it as shown, and click "Apply" below to make the setting effective.
When the matrix is loaded with scene 4, the GPIO(n) output low relay has current flowing through it; when the matrix is switched to other scenes (e.g., scene 3), the GPIO(n) has no current flowing through it and does not close. The action is reversed if "output high, no output low" is selected.
Suppose GPIO(n) is bound to other parameters, such as level display, mute display, system mute display, etc. In that case, the relay can also switch its operating state according to the parameters set in the matrix.

7.4 USB sound card recording function.

Sound card playback function

Audio signals from other devices, such as PCs, are transferred to the processor via a USB cable.
Example: USB sound card playback

1. USB dual male cable to connect the computer to the processor
2. The PC side selects this driver as the speaker
3. PC software matrix configuration, select a channel output in the Play column



Clear	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8	PLAY
OUT1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
REC	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

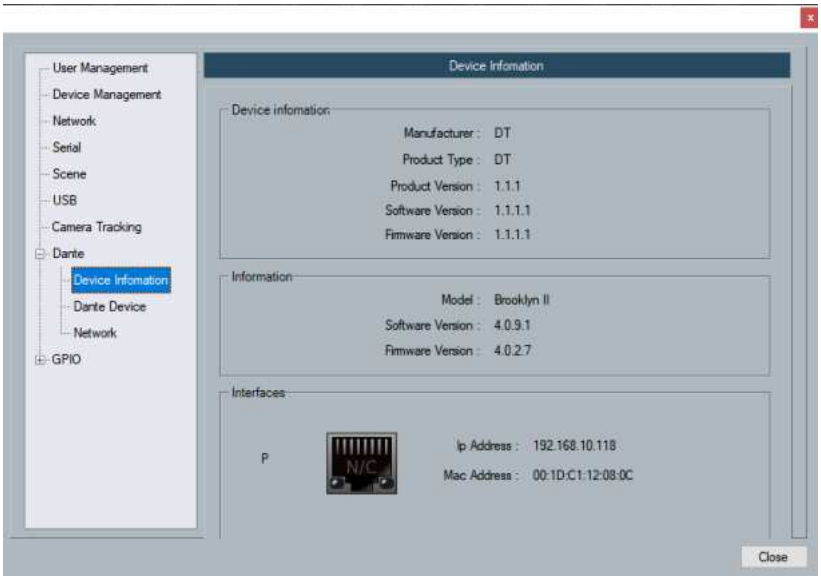
Sound card recording function.

The audio signal from the processor is transferred to other devices, such as the PC, via a USB cable.
Example: USB sound card recording

Clear	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8	PLAY
OUT1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT5	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT6	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT7	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
OUT8	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
REC	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

1. USB dual male cable to connect the PC side to the processor
2. Processor access to analog audio signal or Dante audio signal
3. PC software matrix configuration, input channel matrix of audio signals to REC outputs
4. The PC is switched to the processor sound card when recording.
5. Use audio recording software to record.

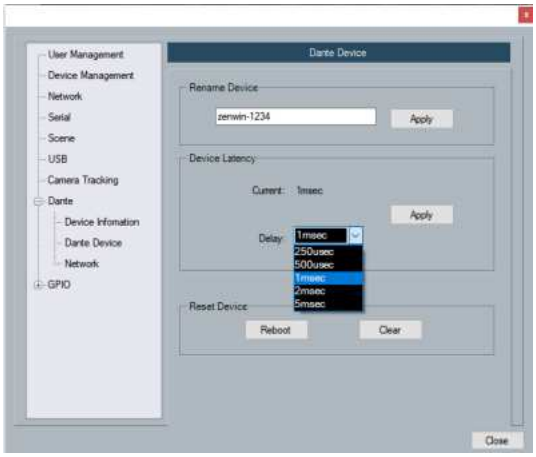
7.5 Dante configuration



The Dante channels in the output section will also be searched for by other Dante devices in the network and will be set to source their input signals according to their needs.
The Setup menu allows you to view the Dante module information for the current device and to set up the Dante module.

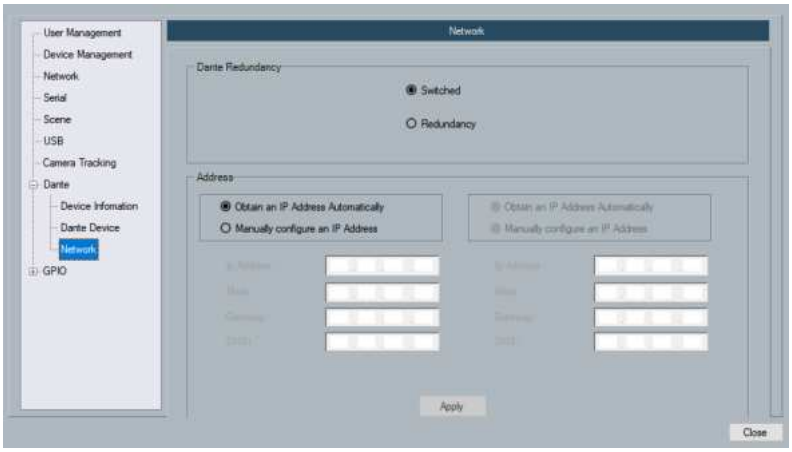
As shown in the picture below
Dante equipment information

Dante device configuration



The lower the device's latency, the higher the network bandwidth requirements; using the 1 MS default is recommended. Restarting the Dante module takes about 10s, the whole unit will not restart, but all channels will be temporarily disabled.

Dante network configuration



Dante Backup When 'Switch' is selected, both Dante ports of the device will use the same configuration; when 'Backup' is selected, the devices.

The two Dante ports will be primary and secondary, allowing simultaneous connection to both networks and uninterrupted signal transmission in case of a communication failure on the primary port.

7.6 PC1 Parameter Controller

Please get in touch with our after-sales staff for instructions on how to use this control panel.

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