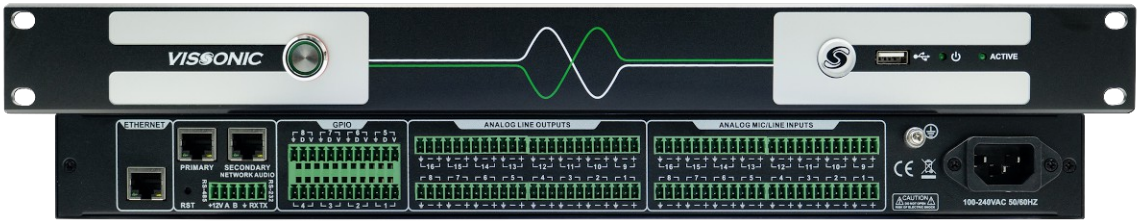
VISSONIC

Professional Audio Video Manufacturer

VIS-DAP Audio Processor
User Manual V1.3



VISSONIC ELECTRONICS LIMITED

The meaning of symbols

■ Safety instructions

For your safe and correct use of equipment, we use a lot of symbols on the equipment and in the manuals, demonstrating the risk of body hurt or possible damage to property for the user or others. Indications and their meanings are as follow. Please make sure to correctly understand these instructions before reading the manual.

	This is A level product, which may cause radio interference in the living environment. In this case, users may need to take the feasible measures to get around the interference.
	Remind users that the dangerous voltage without insulation occurring within the equipment may cause people suffer from shock.
	CE certification means that the product has reached the directive safety requirements defined by the European Union. Users can be assured about the use of it.
	SGS certification means that the product has reached the quality inspection standards proposed by the world's largest SGS.
	This product passed the ISO9001 international quality certification (certification body: TUV Rheinland, Germany).
	Warning: in order to avoid electrical shock, do not open the machine cover, nor is the useless part allowed to be placed in the box. Please contact the qualified service personnel.

■ General information instructions

	It lists the factors leading to the unsuccessful operation or set and the relevant information to pay attention.

Important Note

 **Warning**

To ensure safe and reliable use, follow the guidelines during installation, operation, and maintenance.

Note: This device may cause radio interference in residential environments.

Precautions During Installation

- Ensure the ground wire in the power cord is properly grounded and the chassis grounding point is secure. Do not use a two-prong plug. Make sure the device is powered by AC 100V~240V/(50/60Hz).
- Maintain good ventilation in the working environment to allow heat dissipation and prevent device damage due to overheating.
- In humid or dewy environments, or when the device will not be used for an extended period, turn off the main power.
- Always unplug the AC power cord before:
- A. Removing or reinstalling any device components.
- B. Disconnecting or reconnecting any electrical plugs or connections.
- The device contains high-voltage components. Do not disassemble without authorization. Non-professionals should not attempt repairs to avoid electric shock or further damage.
- Do not spill corrosive chemicals or liquids on or near the device.
- The disconnect device for this product is the appliance coupler, located at the rear of the product. When not in use, unplug the power cord via the appliance coupler. Keep the area around the coupler clear to ensure proper disconnection.

Preface

This manual provides an overview of VISSONIC's digital audio processor, including its features, panels, performance specifications, PC software operation, and troubleshooting.

If the technical parameters and system usage in this manual are changed, the manufacturer will update the version of the manual. Please use the latest user manual.

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1.2	Remove iOS compatibility	2025.10.15
1.3	Updated control panel to processor diagram	2026.04.21

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1. System Overview

1.1. Production Introduction

This series supports up to 16×16 analog audio I/O, with optional AUDIOLINK or Dante modules for 16×16 network audio. Compatible with CLEACON system, Dante Domain Manager, and AES67.

- A/D and D/A conversion: 120dB dynamic range, up to 48kHz sampling rate;
- Processor: High-speed ADI DAP, 450MHz;
- Input source selection: Switchable between balanced mic and line input via Phoenix connector;
- 64-bit floating-point DAP: Enables simultaneous multi-channel processing with ultra-low, consistent latency, ensuring perfect audio-video sync;
- Module copy/paste: Speeds up setup by allowing quick duplication of settings between modules;
- Input channel features: Preamp, signal generator, expander, ducker, compressor, 12-band EQ, AGC, auto-mix, AFC, ANC, AEC, and audio matrix;
- Output channel features: Speaker management with 31-band EQ, delay, crossover, filters, limiter;
- Compatibility: Supports control via Windows and Android systems;
- Network: Ethernet port for real-time single/multi-device control;
- Interface: Intuitive graphical UI for easy operation;
- USB: Supports upgrades, recording/playback, and 1x1 USB audio interface;
- Mixing: Full matrix mixing with auto-mix and mix level control;
- Camera tracking: Built-in auto-tracking function;
- Control ports: Bi-directional RS232/RS485, Ethernet, 8-channel programmable GPIO;
- Tablet support: Compatible with tablet interfaces;
- Scene memory: Stores up to 16 scene presets;
- OS compatibility: Works with XP, Windows 7/8/10.

1.2. Floating-point DAP

The DAP device uses Analog Devices' SHARC processors, offering 32- and 40-bit

floating-point processing—comparable to other high-end 40-bit systems.

Floating-point processing delivers significant advantages in audio quality and ease of use.

Limitations of Fixed-Point Processing

Fixed-point processing can lead to data loss or, worse, clipping distortion when significant gain changes occur. For example, processing a 24-bit audio signal through a 24-bit fixed-point system and applying a 42 dB attenuation reduces the signal's resolution to only 17 bits—permanently losing 7 bits of detail.

Clipping becomes an even greater concern. If gain is increased on a signal already close to 0 dBFS, the signal may clip at the 0 dBFS ceiling, causing distortion. Even if the level is later reduced, the distortion remains, as the clipping has already altered the signal.

To avoid clipping, some fixed-point systems introduce headroom above 0 dBFS (e.g., 12 dB), but this comes at the cost of resolution. A 24-bit system reserving 12 dB (2 bits) of headroom effectively provides only 22 bits of usable resolution.

Floating-Point Processing

Unlike fixed-point processing, floating-point processing allocates the same number of bits to represent the signal regardless of its level. It uses part of the bits as an exponent to define the approximate signal level, while the remaining bits store the actual value—independent of amplitude.

This allows floating-point DAPs to maintain optimal precision across a wide dynamic range, from below -200 dB up to and beyond 0 dBFS, without clipping distortion.

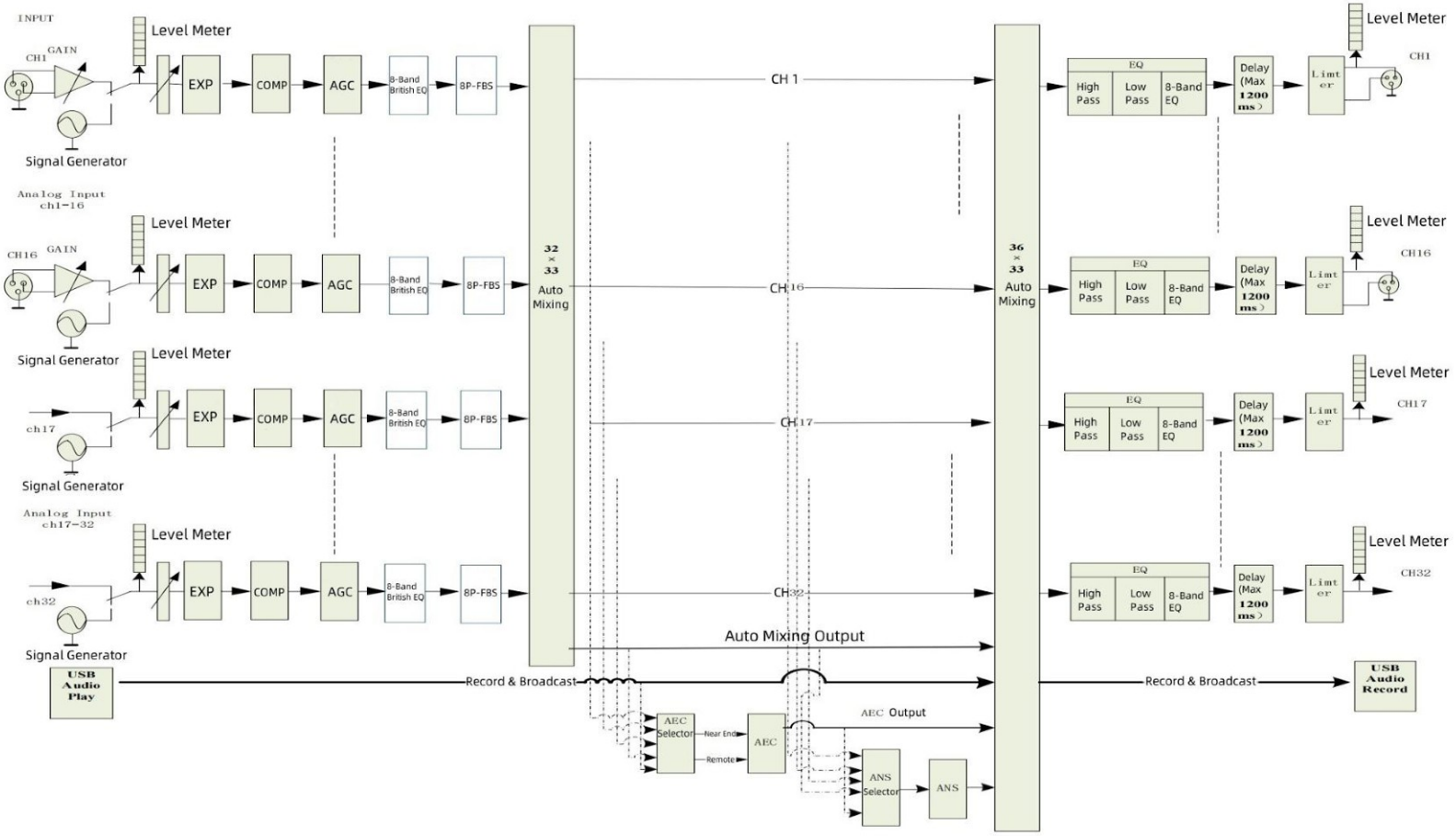
SHARC DAPs offer 32-bit and 40-bit precision. In 32-bit mode, 25 bits are used to store the signal value, ensuring much higher accuracy than 24-bit fixed-point systems—even for very low-level signals. The 40-bit mode extends this further, allowing up to 33 bits for signal representation.

Practical Significance

What does floating-point processing mean for users in real-world applications? It eliminates concerns about gain staging between processing modules. For example, if one module attenuates a signal by 50 dB and a later module restores it, there is no loss of data.

In fixed-point systems, users must carefully manage signal levels before the A/D converter, as all converters operate in fixed-point. If clipping occurs near the output of a DAP system, users can simply reduce the output level to fix the issue. In contrast, fixed-point systems require tracking through each processing module to locate and resolve the clipping source.

1.3. Signal Processing Flow

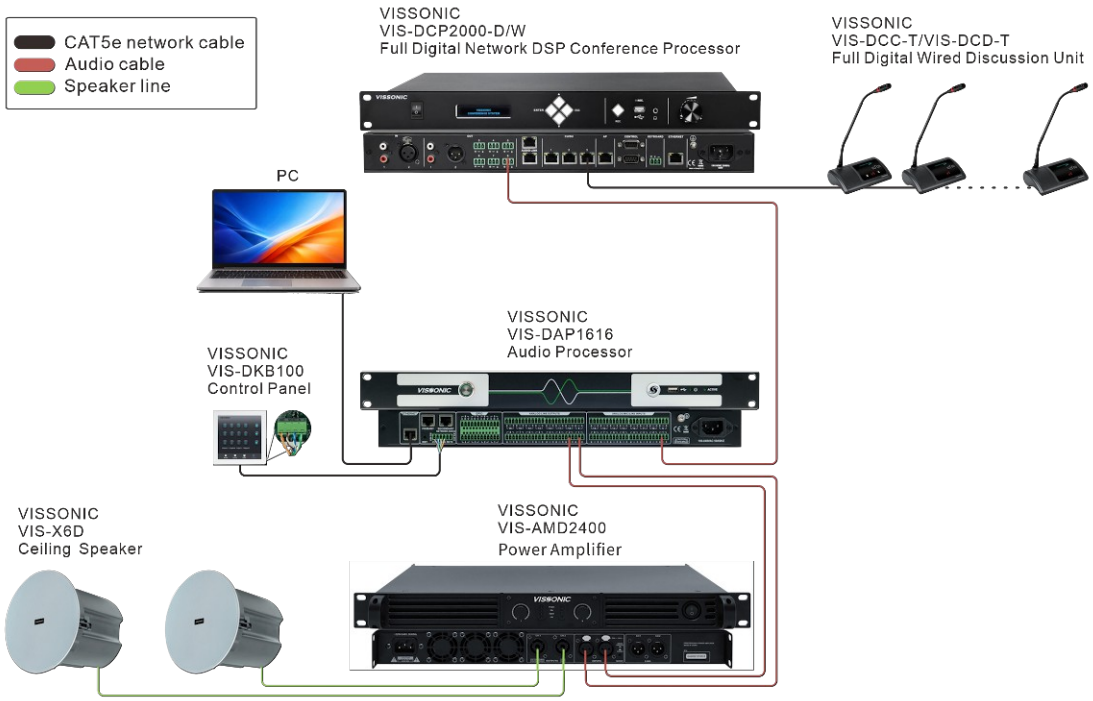


1.4. Typical System Application

Conference Sound Reinforcement System

Microphones are connected to the local inputs of the DAP processor. A computer can record audio or play background music via the USB interface. System-wide audio debugging is performed through the network port of the processor, either via a switch or direct connection. After signal processing through the processor's speaker management module, the output is routed to the power amplifier and finally to the speakers.

An optional control panel can be equipped to manage the processor via RS485, enabling functions such as volume adjustment and scene recall.



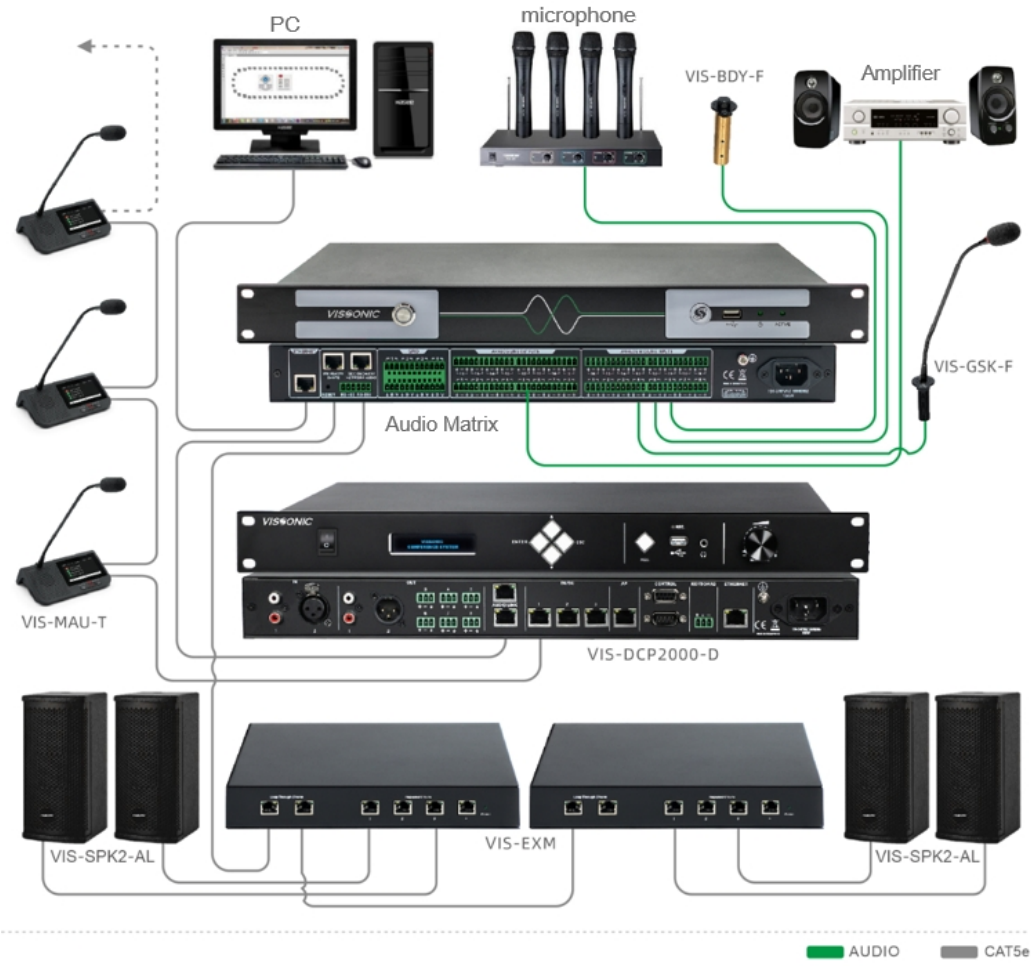
Control Panel Pinout:

Control panel	Audio Processor
VCC	+12V
GND	⏚
A	A
B	B

1.5. Audiolink Network Audio System Application

Audiolink input/output provides 8 channels of uncompressed, low-latency digital audio for conference hosts, expansion units, amplifiers, and PoE speakers that are equipped with Audiolink.

Audiolink provides 8 channels of uncompressed, low-latency digital audio between Audiolink-enabled conference hosts, expansion units, amplifiers, and PoE speakers. It supports converting analog inputs from an audio processor to Audiolink output, enabling direct audio transmission to expansion units and PoE speakers. Alternatively, Audiolink audio can be output to analog amplifiers and speakers via the processor.



2. Hardware Description

2.1. Specifications

	VIS-DAP84	VIS-DAP88	VIS-DAP1212	VIS-DAP1616
Analog Inputs	8	8	12	16
Analog Outputs	4	8	12	16
RS-232 Ports	1			
RS-485 Ports	1			
GPIO	8 shared I/O			
RJ45	1			
USB	1，recording support			
Optional	DANTE or AUDIOLINK Network Interface (Optional) 2			
Max Input Gain	48dBu			
Phantom Power	48V			
I/O Quantization	48KHz/24bit			
CMRR	60Hz： 80dB			
Max Input Level	+24dBu			
A/D Dynamic Range(A-weighted)	113dB			
D/A Dynamic Range (A-weighted)	113dB			
End-to-End Dynamic Range	113dB			
Noise Floor(A-weighted)	-90dBu			
Channel Isolation	1kHz： 108dB			
Frequency Response	20～20kHz (±0.3dB)			
THD+N	<-95dB @17dBu			
Max Balanced Output	+18dBu			

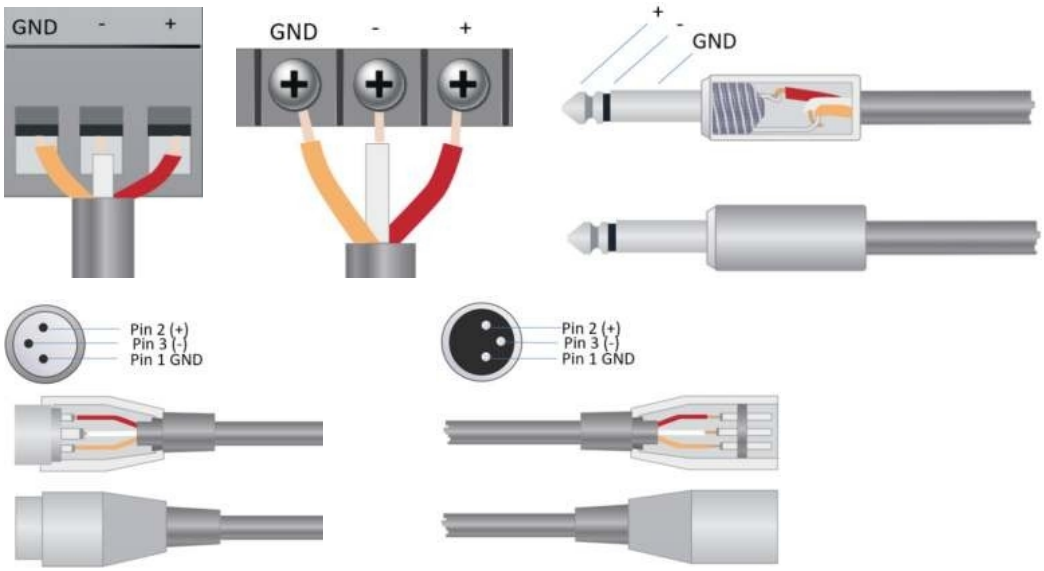
Latency	1.2s
Analog in to Analog out	3ms
Input Impeance(balanced)	5.4KΩ
Output Impeance(balanced)	600Ω
Dimensions: WxDxH (mm)	483*260*44.5(mm)
Shipping Weight	3.5Kg
Power Supply	AC 110V/220V 50Hz/60Hz
Operating Temp.	0~40°C

2.2. Audio Cable Reference

Balanced Connection

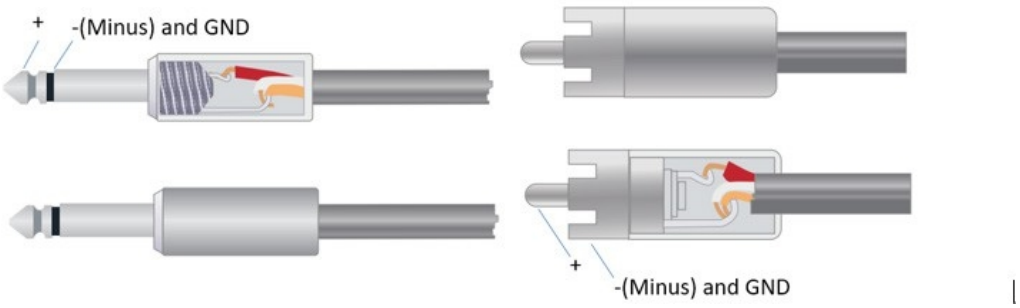
Any of the following connectors may appear on either end of a balanced audio connection.

Note: For XLR connectors, female connects output, male connects input.

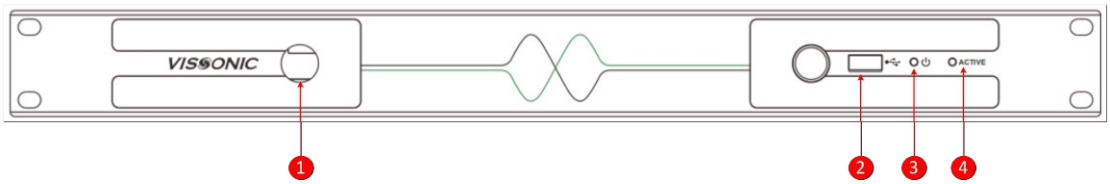


Unbalanced Connection

RCA and 1/4" TS connectors are unbalanced. They use a shielded cable with two conductors and can be used on both ends of an unbalanced connection.



2.3. Front Panel Description



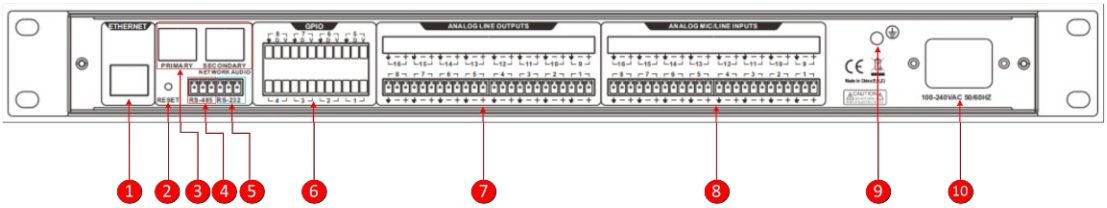
(1) Power Button

(2) USB Port – Supports recording functions

(3) PWR：Lit when power is normal; off indicates power issue

(4) ACTIVE：Blinking indicates normal operation; off indicates device failure

2.4. Back Panel Interface Description



(1) ETHERNET: 10M/100M Ethernet port for connecting to control devices (PC, router, etc.)

(2) RESET: Press and hold for 5 seconds to restore factory settings

(3) AUDIOLINK or DANTEModule (Optional): For DANTE: left port is primary, right port is secondary (do not connect both at the same time)

(4) RS485: Supports camera tracking

(5) RS232: Supports central control commands and camera tracking; RX: Receive, TX: Transmit, G: Ground

(6) GPIO: 8 configurable input/output ports for logic level control; Accepts input signals under 24V (e.g. dry contact); Outputs 5V to drive relays or other devices

(7) INPUT: Analog input, connects to microphone, DVD, etc.

(8) OUTPUT: Analog output, connects to amplifier or active speakers

(9) Chassis ground point

(10) POWER: AC input 100–240V, 50/60Hz

3. Software

3.1. Software Installation

- A Windows PC with a 1 GHz or faster processor, and:
- Windows 7 or later
- 1 GB of free storage space
- 1024 x 768 screen resolution
- 24-bit color or higher
- 2 GB RAM or more
- Network (Ethernet) interface

- CAT5 cable or existing Ethernet network

The audio processor comes with built-in control software and does not require installation via CD. Simply access the audio processor's IP address to quickly download the installation package.

Enter the device IP address in your browser's address bar to access the audio processor, then download the software via the provided link.

Default factory IP address: 169.254.10.227

Subnet mask: 255.255.0.0

Please add a compatible IP address in the same subnet on your PC to ensure proper access. Once the device has booted up, open a browser and enter:

http://169.254.10.227/



Before installing the PC software, please ensure that a recent version of Microsoft .NET Framework is already installed on your computer.

Note: If the software download fails, please try using Internet Explorer or Google Chrome. Alternatively, go to your browser settings and disable the download prompt pop-up, then try downloading again.

3.2. Using The Software

After the software installation is complete, locate the DAP shortcut icon on the destop and double-click it to launch the software. Once the software is launched, first connect to the device via the Device List in the top right corner.





The main interface appears as shown in the image below:



Click the "Device List" button in the upper right corner of the main interface. The software will automatically search for all processors on the network. Users can connect to the desired processor as needed. Once connected, the icon to the left of the device name will turn green. Each processor supports up to 8 users connected and controlling it simultaneously.

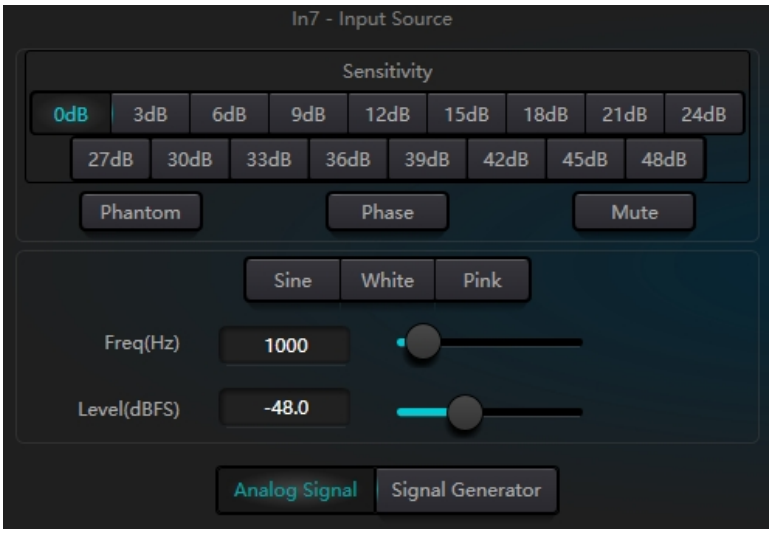
3.3. Configurable Software

Click Edit, then right-click on a module in the input/output channel to modify, replace, delete, or copy it. After editing, click Upload to Host. Note: If CPU usage exceeds 100% and turns red, the configuration cannot be uploaded. Please adjust the modules to reduce CPU load.



3.4. Audio Module Parameters

3.4.1. Input Source



Input Channel Module Parameters

Sensitivity: 17 gain levels from 0 to 48 dB (e.g., 0/3/6/.../48 dB).

Phantom Power: Provides power to condenser mics. Enable only when needed to avoid equipment damage.

Sine Wave: Drag the frequency slider to generate a sine wave at a specific frequency (20–20kHz). The output level, in dBFS, can be adjusted using the fader or by entering a value in the text box.

White Noise: White noise contains equal energy at each frequency component. It appears as a flat spectrum on a constant bandwidth analyzer. Frequency adjustment is disabled; only level control is available.

Pink Noise: Pink noise has more power in lower frequencies, decreasing at a rate of 3dB per octave on the spectrum. Frequency adjustment is disabled; only level control is available.

Additionally, right-clicking any fader on the main interface will bring up a settings menu.



Group Settings: Quickly open the group settings interface.

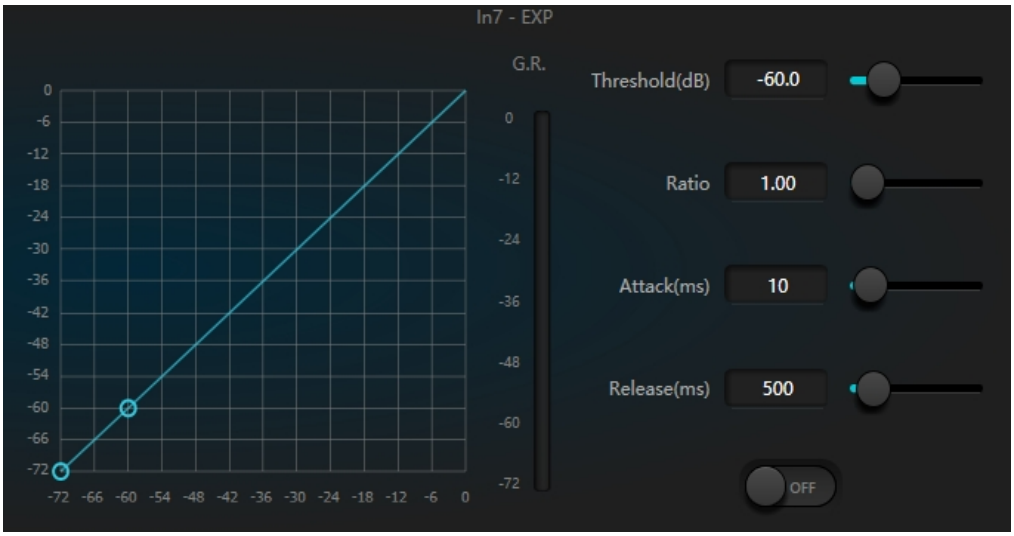
Minimum Gain and Maximum Gain: Set limits on the minimum and maximum gain for the channel. After system tuning is complete, you can set a maximum gain to prevent

external changes that could affect system stability.

3.4.2. Expander

An expander operates on the opposite principle of a compressor, extending a signal's dynamic range. The fundamental difference between the two lies in their threshold operation: compressors affect signals above the threshold, while expanders process signals below it.

At a 1:2 expansion ratio, an input signal 20dB below threshold will produce an output 40dB below threshold. As shown in the diagram, signals below threshold are extended downward, resulting in lower levels. When using a 1:20 expansion ratio, the expander's transfer characteristic resembles a noise gate. In fact, a noise gate is essentially an expander employing an extremely high expansion ratio.



Expander Control Parameters

Threshold: The signal must exceed this level to activate the expander (allowing signal passage). Typically set to match ambient noise levels.

Ratio: Determines the slope of the gain reduction below the threshold. Higher ratios make the expander behave more like a gate.

Attack Time: The time required for the expander to open when the input signal exceeds the threshold. Faster attack times allow quicker response to transients.

Release Time: The time taken for the gain to return to its reduced state after the signal falls below the threshold.

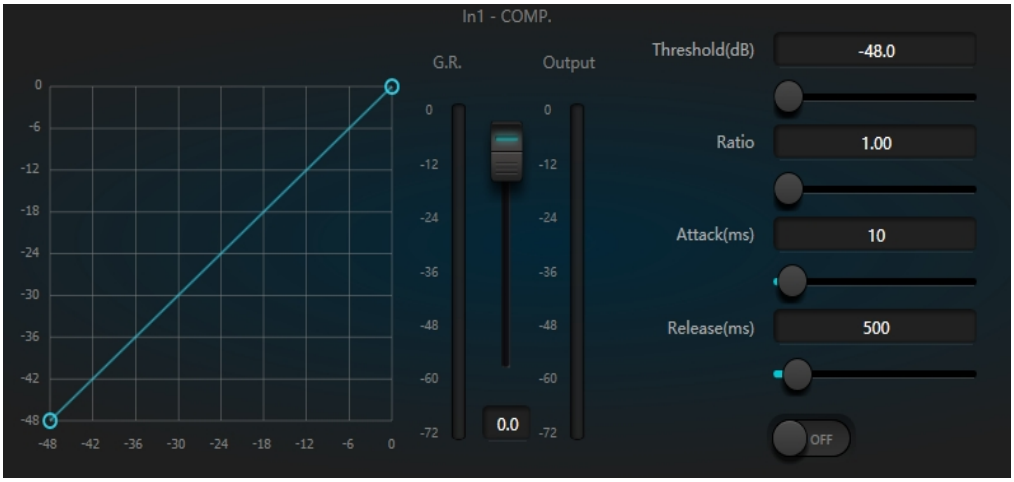
The expander's attack and release times exclusively regulate the rate of gain variation: the attack time governs how quickly the gain recovers from-40dB to 0dB when the

signal exceeds the threshold, while the release time controls the speed at which the gain attenuates from 0dB to -40dB when the signal falls below the threshold. These timing parameters operate independently of the threshold setting. During signal fluctuations below the threshold, the time parameters continuously influence gain reduction. When the signal surpasses the threshold, the gain reduction gradually diminishes at the attack-determined rate (terminating expansion). Upon the signal receding below the threshold again, the expansion reactivates with the release time governing the attenuation process.

3.4.3. Compressor and Limiters

Compressor

Reduces the dynamic range of signals that exceed the set threshold, while signals below the threshold remain unaffected.



Compressor module parameters

Threshold: When the signal level exceeds this value, the compressor/limiter begins to reduce the gain. Any signal exceeding the threshold is considered an overshoot and is normally attenuated. The more the signal exceeds the threshold, the greater the attenuation applied.

Ratio: This is the compression ratio, determining how much the overshoot signal is reduced toward the threshold level. A lower ratio allows the signal to exceed the threshold more easily. Once the signal exceeds the threshold, the ratio defines the relationship between the input and output level changes. For example, with a 2:1 ratio, if the input signal exceeds the threshold by 2dB, the output signal increases by only 1dB. A 1:1 ratio means no compression is applied. The adjustable range for the ratio is 1 to

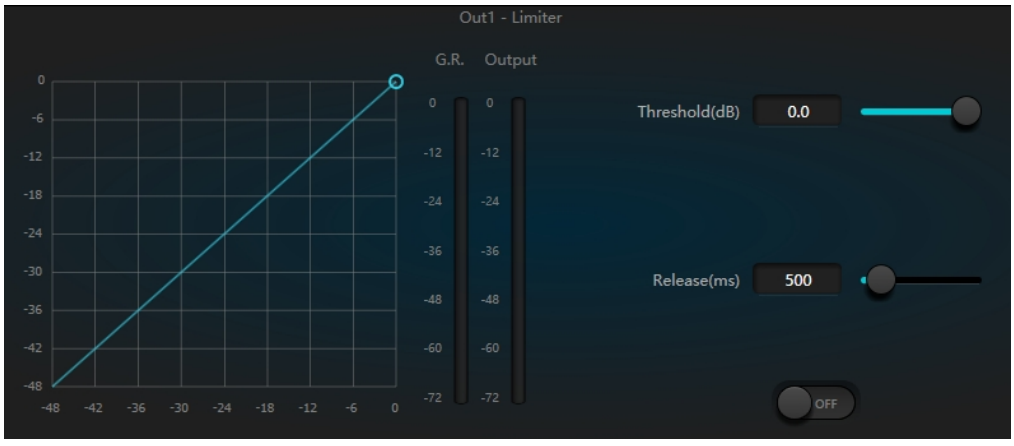
20.

Attack Time & Release Time: To preserve the natural sense of attack in a sound, it is often desirable for the initial portion of a signal to pass through the compressor unaffected or only slightly affected. This requires slowing down the compressor's response time. Similarly, if the gain drops or recovers too quickly, it can cause a "pumping" or "breathing" effect. The attack and release times of the compressor are designed to avoid this issue. Attack time determines how quickly gain reduction is applied, while release time controls how quickly the gain returns to normal.

Output Gain: Also known as makeup gain. If the compressor significantly reduces the signal level, it may be necessary to boost the output gain to maintain the desired volume. This gain boost is applied equally to the entire signal, regardless of other compressor settings.

G.R. and Output Level Meter: G.R. (Gain Reduction) indicates how much compression is being applied, while Output shows the signal level after compression. The gain reduction is typically displayed as an inverted level meter. For example, if the input signal is at -6dB, the threshold is set at -30dB, and the ratio is 2:1, the gain reduction is 12dB. The G.R. meter will show -12dB, and the Output meter will indicate -18dB.

Clipper

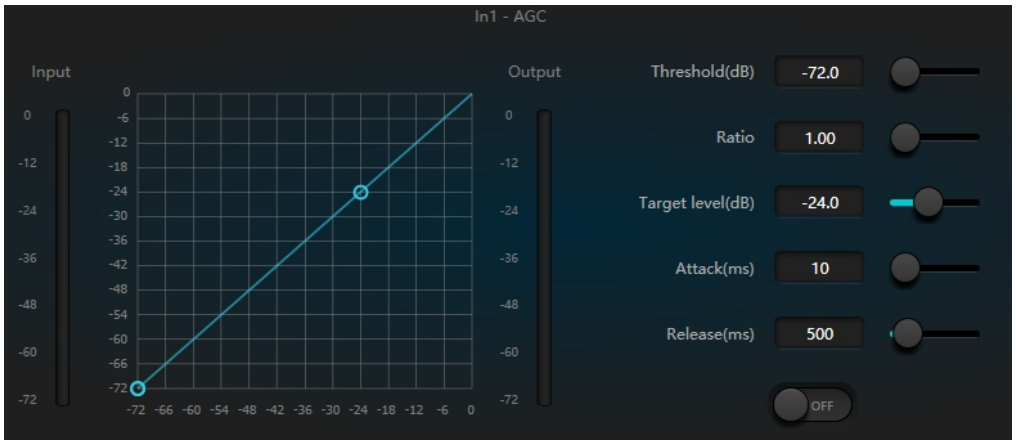


Clipper is as known as a limiter, its sole purpose is to prevent signals from exceeding a set threshold, achieved through two-stage processing: initial gentle attenuation to control dynamics, followed by aggressive gain reduction if peaks persist. It operates with just two controls—threshold and release time—and differs from a compressor only in degree; extreme compressor settings can emulate limiting behavior. For optimal results, limiters should correct occasional clipping, while frequent clipping requires

lowering the input signal level rather than excessive limiting.

3.4.4. Auto Gain Control

Automatic Gain Control (AGC) is a special type of compressor with a very low threshold, medium to slow attack time, long release time, and a low ratio. Its purpose is to raise signals with uncertain levels to a target level while maintaining their dynamics. Most AGC systems include some form of silence detection to prevent gain increase during silent periods—this is the key feature that differentiates AGC from standard compressors or limiters.



AGC is often used to normalize the output level of background music, foreground music, or on-hold music from CD players, helping to eliminate level fluctuations from paging microphones.

Parameters of the Automatic Gain Control

Threshold: When the input signal level is below this threshold, the input/output ratio remains 1:1. Once the signal level exceeds this threshold, the input/output ratio changes according to the set ratio. This threshold should be set just above the background noise level of the input signal.

Ratio: The proportion between the input signal level change and the resulting output level change when the input is above the threshold.

Target Level: The desired output signal level. If the signal exceeds this level, the controller compresses it based on the set ratio.

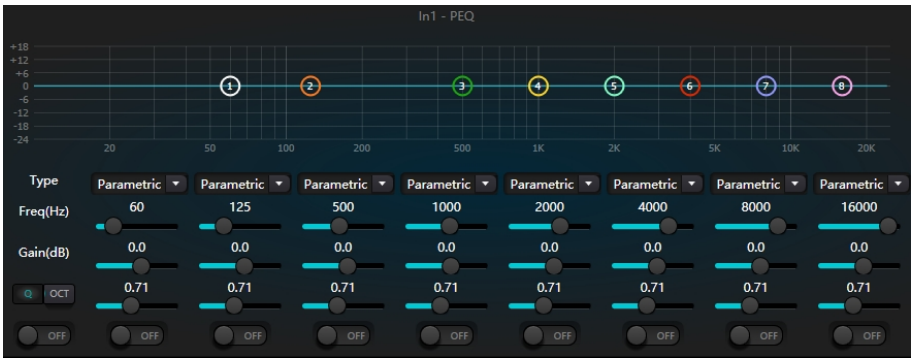
Attack Time: Determines how quickly the AGC reacts to signal levels above the threshold.

Release Time: Determines how quickly the AGC reacts to signal levels falling below

the threshold.

3.4.5. Parametric Equalizer

The primary function of an equalizer is to correct frequency ranges that are either overly emphasized or lacking, whether these ranges are broad or narrow. Additionally, equalizers help narrow or widen frequency bands or adjust the level of certain components within the frequency spectrum. Simply put, an equalizer alters the tonal character of a signal.



Parametric equalizers are available in 5-band, 8-band, and 12-band options. The module parameters are as follows:

Type: Default is parametric EQ, with optional high/low shelf filters and high/low pass filters. Each filter type has different shapes and serves different functions.

High & Low Pass Filters: These filters allow frequencies on one side of a defined cutoff frequency to pass while gradually attenuating frequencies on the other side.

- High-pass filters allow frequencies above the cutoff to pass and attenuate lower frequencies.
- Low-pass filters do the opposite, allowing lower frequencies to pass and attenuating higher ones.

High & Low Shelf Filters: Also known as shelving filters, these adjust the gain of frequencies above (high shelf) or below (low shelf) a specified point. Unlike the cutoff point in pass filters, the set frequency here refers to the midpoint of the filter's slope. The Q value affects the sharpness or peaking of the filter, with a mathematical relationship to the peak level.

Gain: Refers to the amount of boost or attenuation (in decibels) at the center frequency.

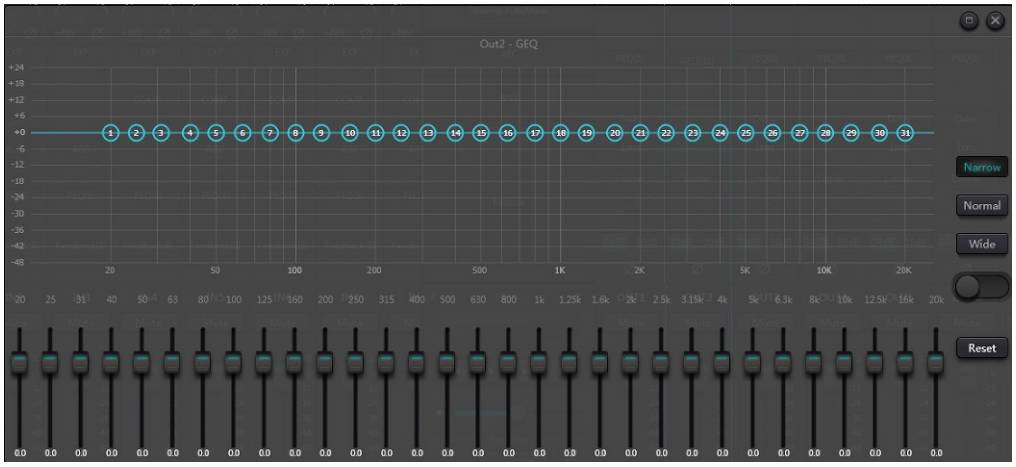
Q (Quality Factor): Defines the bandwidth of the filter. The adjustable range is from 0.02 to 50.

- In parametric EQ mode, the Q value determines the width of the bell-shaped frequency response between the cutoff frequencies on either side.
- In shelf or pass filter mode, if Q > 0.707, the filter response may exhibit peaking. If Q < 0.707, the slope becomes flatter and the roll-off begins earlier.

Each EQ band has an individual on/off switch; when off, the band's parameters are inactive. Additionally, there is a global EQ switch to enable or disable the entire module.

3.4.6. Graphic Equalizer

Utilizing constant Q technology, each frequency point is equipped with a slide potentiometer. Regardless of whether a frequency is boosted or attenuated, the filter's bandwidth remains unchanged. The graphic equalizer is currently available in 10-band, 15-band, and 31-band configurations.



3.4.7. FeedBack Suppression

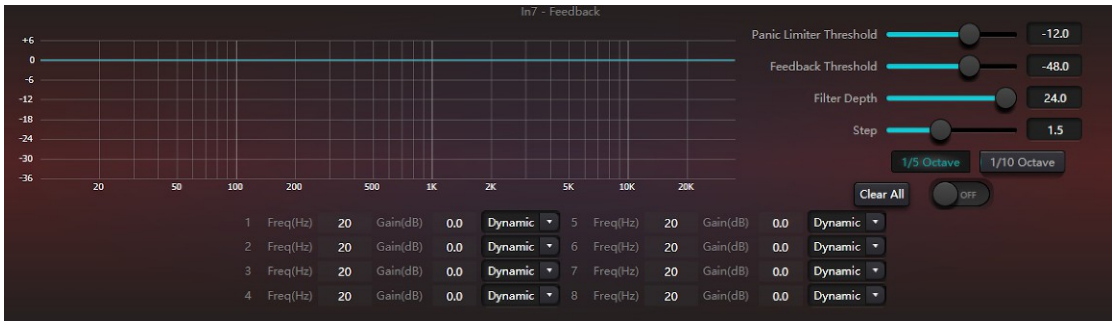
When using the Feedback Suppression module, it should always be combined with good system design and engineering practices, rather than replacing them. Traditional methods—such as limiting the number of open microphones, minimizing the distance between the sound source and the microphone, positioning microphones and loudspeakers to minimize feedback, and using equalization to achieve a flat room response—should still be employed. Only after these measures are in place should the feedback suppressor be used to gain additional system headroom. A feedback suppressor is not a magical solution for poor system design or for exceeding the physical limitations of a sound system.

The Feedback Suppression module automatically detects and suppresses acoustic

feedback in the audio system. Based on the characteristics of the signal, the module distinguishes between feedback and desired audio content. When feedback is detected at a certain frequency, a notch filter is automatically applied at that frequency to attenuate it. The initial attenuation is mild, and if the feedback persists, the filter continues to attenuate according to the configured parameters until the feedback is eliminated or the maximum attenuation value is reached. Various user parameters are available for fine-tuning the module's performance.

Once ringing has occurred, filters can be locked to prevent them from being altered during live performances. The filter settings can also be copied to a dedicated notch filter module (such as an equalizer). Eight automatic filters are used in a rotating manner, allowing temporary filters to be replaced when necessary.

Each channel has its own Feedback Suppression module. You can locate the module by dragging from the input section or quickly access it via the shortcut button on the right. To enable the module, click the "Enable" button. The system will then automatically detect feedback frequencies and suppress them using narrow-band filters. Each Feedback Suppression module allows you to choose from 8, 12, or 16 narrow-band filters.



Feedback Suppression Module Parameters:

Panic Threshold: This parameter tells the module, "Any level above this threshold is definitely feedback." When the signal level exceeds the panic threshold, several actions occur:

a) The output gain is temporarily reduced to control the rate at which feedback builds up;

b) The output level is limited to prevent runaway volume;

c) Filter sensitivity is increased for faster feedback detection.

Once the output level drops below the threshold, the gain and sensitivity return to

normal. This value refers to the peak level of a full-scale digital signal. Setting this value to 0 disables this function.

Feedback Threshold: This threshold tells the module, "Any level below this value is definitely not feedback." This prevents the module from mistakenly identifying low-level hums or noise in quiet music passages as feedback.

Filter Depth: Sets the maximum attenuation a single filter can apply. A shallower setting may reduce signal degradation caused by the filter but can also result in poorer feedback control—especially in large, narrow-band resonant systems.

Bandwidth: Selectable between 1/10 and 1/5 octave. Using a constant Q value ensures the filter bandwidth does not increase with depth. In speech-focused environments where filters may quickly run out and feedback is common, it's recommended to set the bandwidth to 1/5 octave, as the wider range can cover more potential feedback frequencies.

Presets: Four built-in presets are available: "Music – Large Room," "Music – Small Room," "Speech – Large Room," and "Speech – Small Room." These presets provide default settings suitable for most common applications.

Notch Filter Modes:

Each notch filter offers three modes: Auto, Manual, and Fixed.

- In Auto mode, the filter automatically detects and suppresses feedback based on the configured parameters. When all 8 filter slots are used and new feedback is detected, the module will clear the earliest-used Auto filter and reassign it to suppress the new feedback.
- In Manual mode, the gain of the filter can be manually adjusted by the user.
- In Fixed mode, the filter parameters remain constantly active, will not be overridden by new feedback detection, and will persist after a system reboot.

To save these feedback filter settings, click the Save Preset button.

Clear: Clicking this button instantly clears all filters. It removes all previously detected and suppressed feedback points. This function is typically used when retuning or resetting the feedback suppression module.

Application: The feedback suppressor can be used as a system tuning tool to locate feedback frequencies or as a preventive measure during normal operation. For optimal

acoustic gain and effective feedback suppression, it is recommended to follow these tuning steps:

a. Lower the system gain and reset all filter parameters using the Clear button.

b. Configure the feedback suppression module parameters. Lower the panic threshold to reduce the level at which feedback occurs.

c. Turn on all microphones and slowly increase system gain until feedback begins. Stop increasing gain once feedback starts.

d. Wait for the feedback suppression module to act. Once the feedback disappears, continue increasing gain.

e. Repeat the process until the desired gain is achieved or all filters are assigned.

f. Set the panic threshold to just above the maximum expected level of non-feedback signals.

At this point, if needed, each filter can be set to Fixed mode, or the current dynamic state can be saved to handle potential feedback during a live performance.

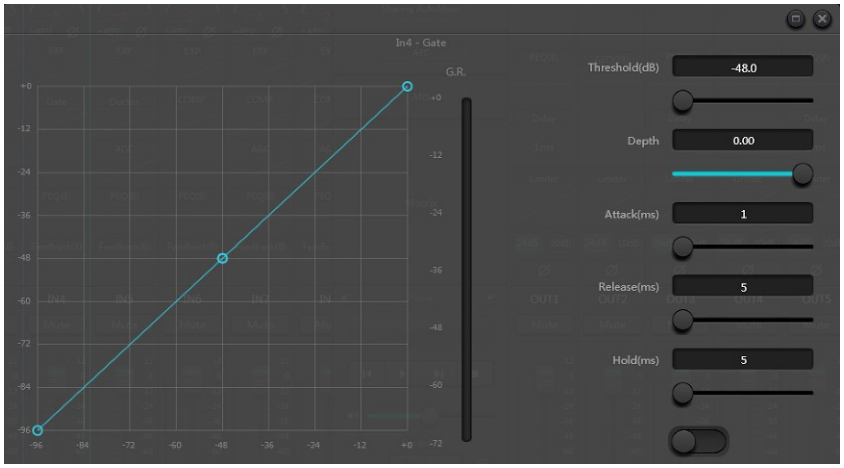
Another option is to copy the filters to a dedicated notch filter module (such as an equalizer), which allows for additional filter capacity.

If the system includes built-in loudspeakers, it is recommended to use a compressor/limiter module for added protection.

Properly setting a limiter will ensure that the loudspeakers are not damaged—even if all notch filters have been used or the feedback suppressor cannot fully control feedback, such as in cases of excessive system gain.

3.4.8. Noise Gate

The primary purpose of a noise gate is to attenuate signals that fall below a set threshold—these signals are usually unwanted noise.



Threshold: Signals above the threshold pass through; signals below the threshold trigger the gate and get attenuated.

Depth: The amount of attenuation applied to signals below the threshold.

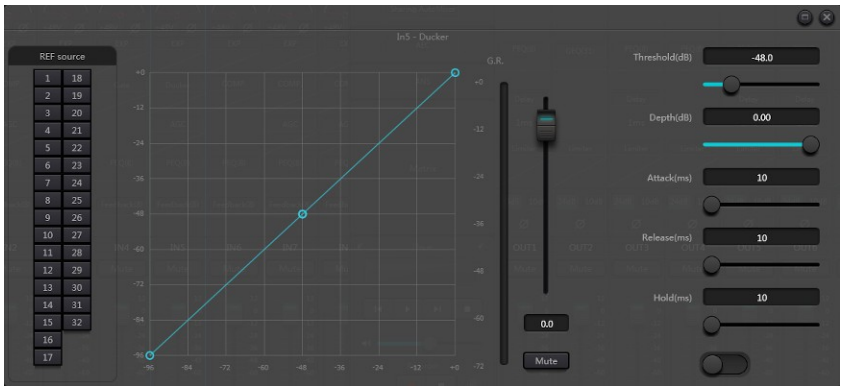
Attack Time: The speed at which the gate opens when the signal exceeds the threshold.

Release Time: The opposite of attack time; it determines how quickly the gate closes once the signal drops below the threshold.

Hold Time: When the signal drops below the threshold, attenuation normally begins immediately. However, if a hold time (e.g., 10 ms) is set, the gate will maintain the original output for that duration before attenuation begins.

3.4.9. Ducker

When the level of a certain channel exceeds a specified threshold, the levels of other selected channels will be attenuated—this is called the ducking effect.



Threshold: When the reference signal exceeds the threshold, attenuation begins; when it falls below the threshold, the signal recovers.

Depth: The amount by which the ducked signal is reduced.

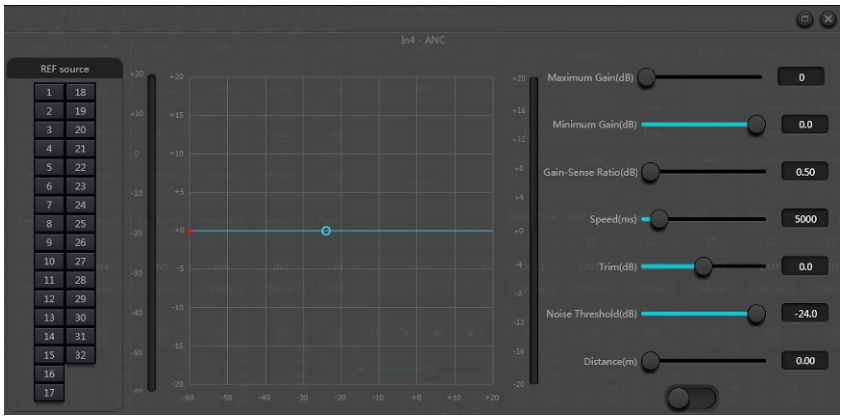
Attack Time: The time it takes to start attenuating the ducked channel after the reference signal exceeds the threshold.

Release Time: The time it takes for the ducked signal to return to its original level after the reference signal falls below the threshold.

Hold Time: The duration for which the ducking effect on the ducked channel continues to be maintained after the reference signal drops below the threshold.

3.4.10. Active Noise Compensation(ANC)

Automatically adjusts output volume based on real-time environmental noise detection and processing.



Maximum Gain: The maximum adjustable amount;

Minimum Gain: The minimum adjustable amount;

Gain Sensitivity Ratio: The ratio of gain increase or decrease;

Speed: The rate at which gain is increased or decreased;

Trim: Gain adjustment;

Noise Threshold: Gain starts to increase when noise exceeds this threshold, and decreases when below it;

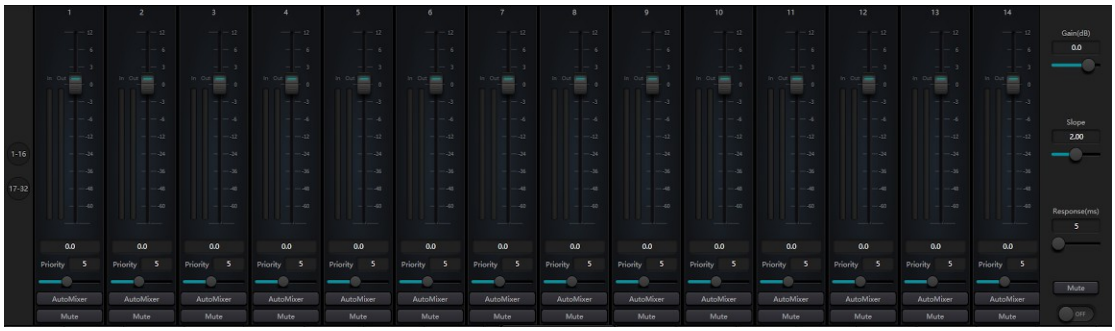
Distance: The distance between the reference signal and the local signal.

3.4.11. AutoMixer

In conference room audio systems, when multiple microphones operate at identical gain levels with only one active speaker, inactive microphones pick up ambient noise and reverberation, degrading audio quality and increasing feedback risks. The built-in 32-channel gain-sharing auto-mixer instantly mutes idle microphones with response times far exceeding manual operation. The system provides independent direct outputs per channel to accommodate fixed-level requirements like background music or special

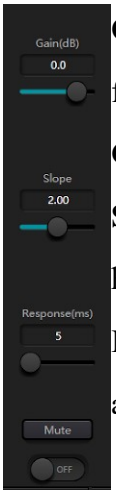
applications such as permanently active chairman microphones with independent gain control.

The auto-mixer module features a two-tier control system: master parameters and channel parameters.



Automatic Mixing (Gain Sharing Interface)

① Main Control Parameters



On/Off Button: The bottom button enables or disables the automatic mixing function.

Gain: Controls the main output volume of the automatic mixer.

Slope: This controls the amount of attenuation applied to lower-level signals. A higher slope results in more attenuation for lower-level channels.

Its behavior is similar to the ratio control in an expander. Recommended setting: around 2.0 for optimal gain sharing.

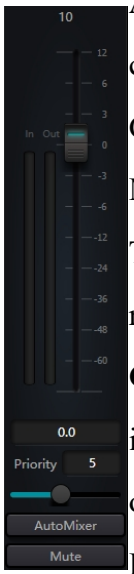
- Setting it to 1.0 is equivalent to turning off automatic mixing on all channels.
- Setting it to 3.0 or higher may cause exaggerated gain adjustments, resulting in unnatural audio.
- In general, the higher the slope, the more open channels there are, and the greater the overall attenuation applied.

Response Time: Determines how quickly the automatic mixer reacts to signal changes.

- Faster response ensures that the beginning of speech (attack transients) is not cut off.
- Slower response provides smoother operation.
- A range between 100 ms and 1000 ms is typically most effective.
- The system is designed so that microphone activation is faster than deactivation, ensuring speech clarity even with a 100 ms response time.

- Setting this parameter to a few seconds introduces a longer hold time, keeping the last active channel open for a few seconds.
-

② Channel Control Parameters



AutoMixer: Each channel has an on/off toggle for automatic mixing. Only channels with this toggle enabled will participate in the auto mixing algorithm. Channels with the toggle turned off will be excluded from automatic mixing.

Mute: Mute and fader adjustments are applied after the automatic gain control. This means even if a channel is muted, its signal level can still affect the gain reduction on other channels if it's high enough.

Gain: Adjusts the fader level to increase or decrease the channel's contribution in the auto mix. This controls how prominently the channel appears in the overall mix.

Priority: Setting the priority allows higher-priority channels to override lower-priority ones, thereby affecting the automatic mixing algorithm. The parameter range is 0–10, with higher values indicating higher priority.

Channel mute and fader controls are post-auto gain, meaning any adjustments to these two parameters will not affect the operation of the automatic mixing. For example, even if a channel is muted, if its signal level is high, it can still reduce the gain of other channels. To mute a signal and prevent it from influencing the automatic mixing, you must mute the channel and disable its participation in auto mixing.

Each channel’s mute button silences the channel in the mix and also mutes its direct output. The channel fader controls both the mix level and the direct output level. You can precisely adjust the channel level by clicking the text box and entering a dB value.

Priority control allows higher-priority channels to override lower-priority ones, influencing the automatic mixing algorithm. The control can be set in a range from 0 (lowest priority) to 10 (highest priority), with a default value of 5 (standard priority). You can adjust the priority using the slider or by clicking the input box and entering a specific value between 0 and 10. Increasing the value raises the channel's priority. When two channels have the same input signal level, the one with the higher priority will receive a higher auto gain. If the priority difference is 1 unit, the higher-priority

channel will receive an additional 2 dB of auto mixing gain (assuming both channels have a slope setting of 2.0). For example, if Channel 1 is set to priority 6 and Channel 2 is set to priority 3, and both have the same input level, Channel 1 will gain an extra 6 dB in auto mixing gain compared to Channel 2. Note that the slope setting in the main control parameters also affects the gain difference resulting from priority weighting.

If the slope is set to 3.0, then each unit of priority difference results in a 4 dB gain difference. If all channels have the same priority, it's recommended to leave the default setting at level 5.

Note: Be cautious when using extreme priority differences between channels (e.g., 0 vs. 10). If a high-priority channel picks up ambient noise (such as from a loudspeaker), it may suppress lower-priority channels—even if it is not actively in use. The higher the slope, the more severe the problem. If this issue arises during system setup or tuning, consider inserting a noise gate or expander before the auto mixer on the highest-priority channel, and set the threshold high enough so background noise or speaker pickup does not trigger the gate or expander.

An additional threshold-based automatic mixer option is also available:



Automatic Mixing (Threshold-Based)

The threshold-based automatic mixer is developed based on a noise gate principle. Each channel has its own noise gate and is either open or closed — there is no intermediate state.

Gain: Controls the overall output gain of the automatic mixer.

Hold: The Hold function determines how long a channel remains open after the speaker stops talking. This ensures that short pauses between words or sentences during speech do not cause the channel to close prematurely.

Off gain: Sets the level of a channel when it is not active. For example, if Microphone 1 is active and speaking, and then Microphone 2 becomes active, Microphone 2 will

receive gain based on its speaking level plus the value set by Off Gain. If this value is set too high, inactive microphones may pick up room noise and reverberation. If it's too low, the newly activated microphone may not output sufficient level.

Sensitivity: Controls how far above the channel's ambient noise threshold the input level must be to activate the channel. Higher sensitivity values make the channel easier to activate, even under quiet conditions.

NOM Atten: The Number of Open Microphones (NOM) attenuation fader determines how much all channels are attenuated as more microphones become active. For example, if set to 3dB, opening two microphones results in 3dB attenuation; opening four results in 6dB, and eight results in 9dB. Each doubling of open microphones adds an additional 3dB attenuation.

NOM Limit: Sets the maximum number of microphones that can be active at once.

Noise Threshold: If the input level exceeds this threshold, the channel opens; if it drops below the threshold, the channel closes.

3.4.12. Acoustic Echo Cancellation

Acoustic Echo Cancellation (AEC) is a digital audio signal processing technique used in audio and video conferencing when conversations take place between participants in a local meeting room and one or more speakers at a distance. AEC enhances the clarity of remote speakers' voices by eliminating acoustic echo generated within the local room.

The echo cancellation module, applied to remote calls, enables local amplification of remote voice signals while attenuating the interference caused by acoustic echo. Its basic working principle involves modeling the echo path to estimate the potential echo produced by the remote signal. This estimated echo is then subtracted from the microphone input signal, effectively removing the echo from the transmitted audio and thus achieving echo cancellation.



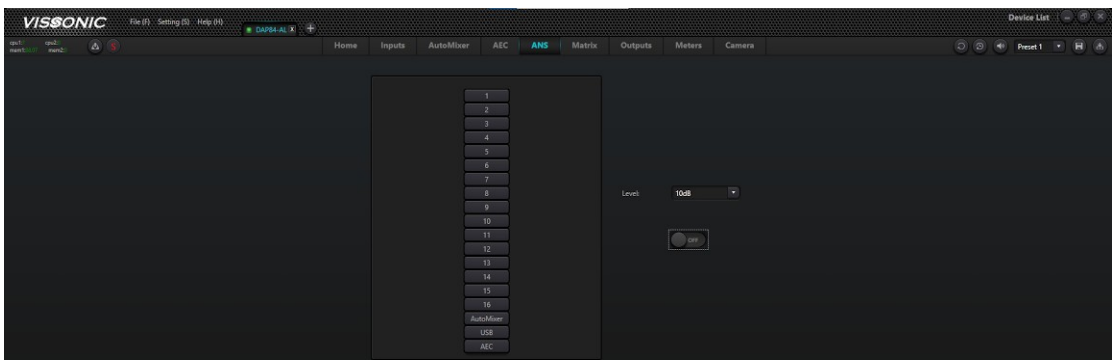
In the DAP Controller, there is only one Echo Cancellation (AEC) module, which is pre-configured with local input and far-end input mixers to allow multiple signal paths to participate in echo cancellation, as shown in the diagram. Only one parameter is available for adjustment.

Nonlinear Filtering: This parameter determines the aggressiveness of echo suppression and offers three selectable levels: Conservative, Moderate, Aggressive.

3.4.13. Noise Suppression

The Noise Suppression module effectively removes non-speech sounds by distinguishing between human voice and non-voice signals, treating the latter as noise. After processing an audio signal that contains both speech and noise through this module, theoretically, only the speech remains.

In the DAP Controller, there is only one noise suppression module, which comes with a pre-configured multi-channel mixer that allows multiple signal sources to participate in the noise suppression process, as illustrated.



Suppression Level: There are four selectable options—6, 10, 15, and 18 dB. The dB

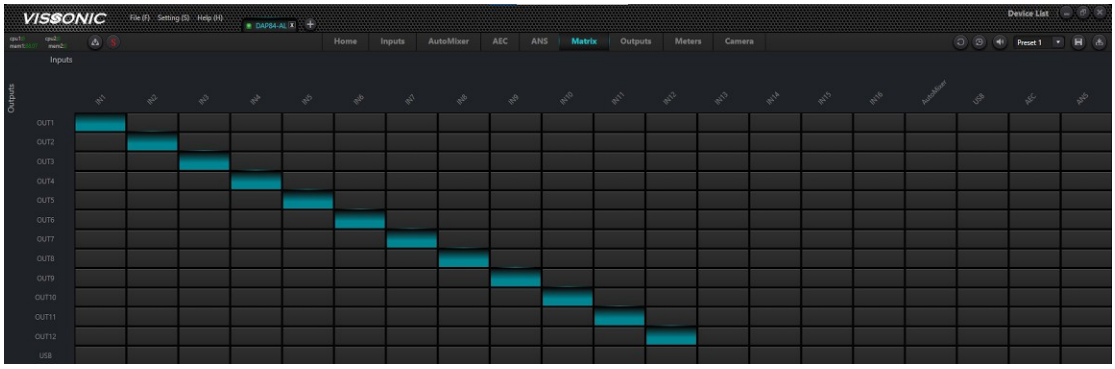
value indicates how much the noise is reduced. The higher the value, the greater the noise suppression, but it also results in more degradation of the speech signal, which is an unavoidable trade-off.

3.4.14. Matrix

The matrix provides dual functionality for routing and mixing. The horizontal axis represents input channels, while the vertical axis represents output channels. By default, each input is routed to the corresponding output one-to-one, as shown in the diagram. For example, if you want to mix Input Channel 1 and Input Channel 2 into Output Channel 1, simply enable both Input 1 and Input 2 on Output 1.

Note that if Input 1 and Input 2 are part of an automatic mixing process, the output is not affected by the automatic mixing settings.

Similarly, after configuring the automatic mixing, echo cancellation, and noise suppression modules, you must also configure the matrix settings to establish the correct signal routing.



3.4.15. High Pass & Low Pass Filter

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

Frequency: The cutoff frequency of the filter. For Bessel and Butterworth filters, the cutoff is defined at -3dB; for Linkwitz-Riley filters, it's defined at -6dB.

Gain: Adjusts the overall gain across the full frequency range, either boosting or attenuating the signal.

Type: Choose from three filter types — Bessel, Butterworth, and Linkwitz-Riley. Among them, Butterworth has the flattest passband.

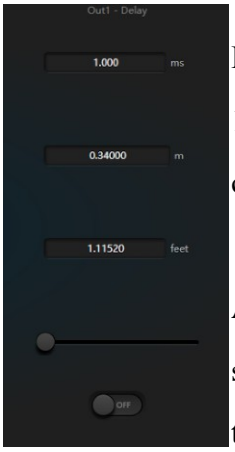
Slope: Determines how sharply the signal attenuates beyond the cutoff frequency.

Available options include 6, 12, 18, 24, 30, 36, 42, and 48 dB/Oct. For example, a slope of 24 dB/Oct means that in the transition band, the amplitude drops by 24 dB for each octave.

To activate the high-pass or low-pass filter module, click the activation button at the bottom.



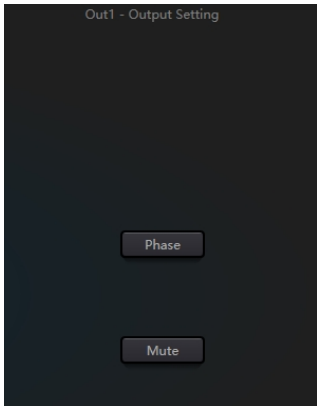
3.4.16. Delay



Ms: Sets the delay time of the delay module. The available range is 1 to 1200 milliseconds. The values in meters and feet are simply converted units of the millisecond setting.

Activate Button: Activates the specified delay module within the system, inserting it into the audio signal path to apply a fixed delay to the signal.

3.4.17. Output



Phase: Reverses the phase of the audio signal by 180°.

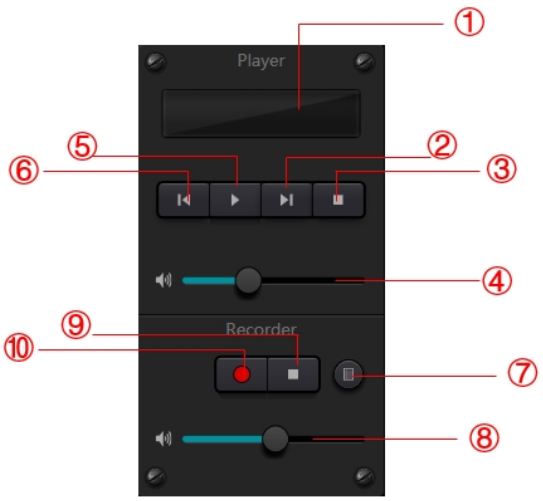
Mute: Toggles mute/unmute for the channel.

On the output channels, similar to the input channels, some settings can also be accessed by right-clicking. You can configure these options as needed.



3.4.18. USB Soundcard

Using the USB sound card serves two main purposes: first, for recording and playback; second, for remote conferencing on the PC side. The USB audio signal passes through both the echo cancellation and noise suppression modules, making it easy to integrate into remote meetings. The USB Playback function in the software interface is only used when the recording/playback feature is active.



① Playback Information – double-click to enter the playlist

② Next Track

③ Pause

④ Adjust Song Playback Volume

⑤ Play

⑥ Previous Track

⑦ Recording List

⑧ Adjust Recording Volume

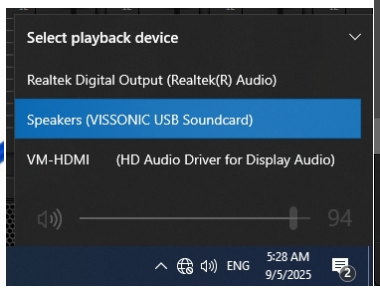
⑨ Stop Recording

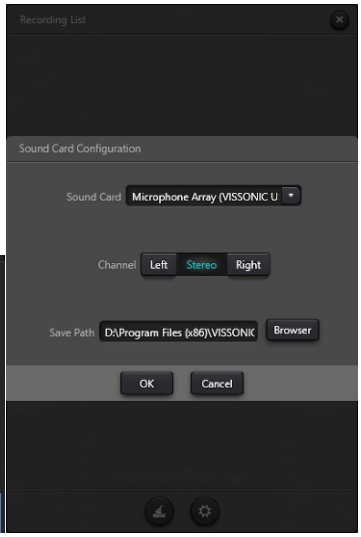
⑩ Start Recording

Sound Card Settings:

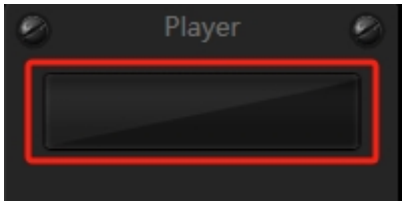
Connect the DAP processor to the computer using a dual Type-A USB cable. Upon the first connection, the computer will detect new hardware and automatically install the necessary drivers. Once the installation is complete, a new device named USB Soundcard will appear in the computer's sound card list, as shown in the figure. Then, go to the software's playlist and select USB Soundcard in the sound card settings.

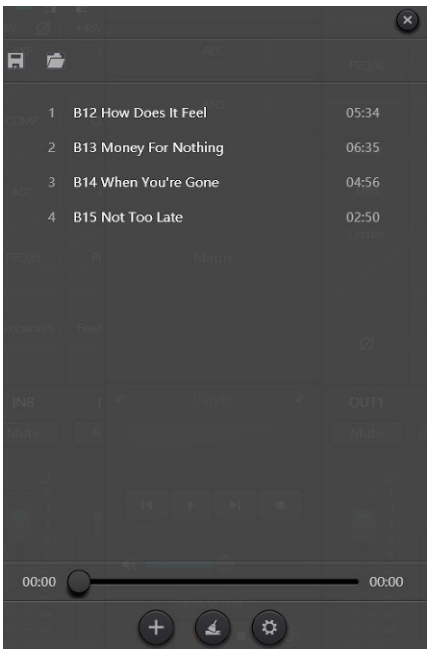




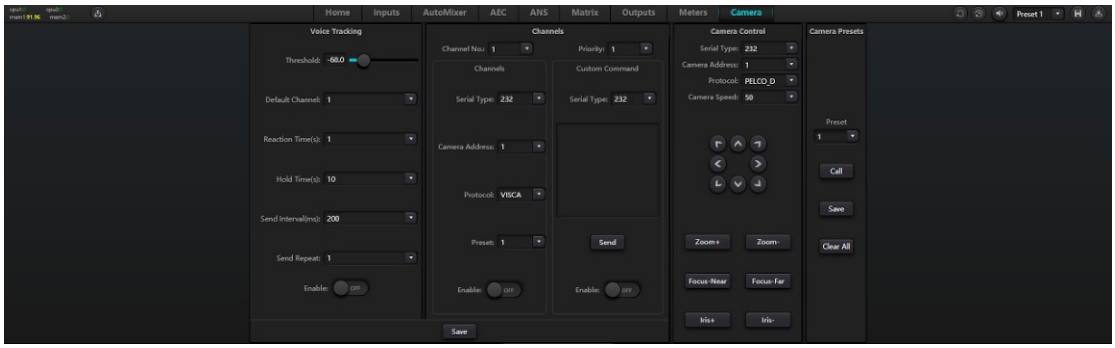


The playlist allows you to manage song files and save them as a list for future use. Simply open the saved list next time to continue playback. Click  at the bottom of the playlist to open a folder and select songs for playback, click  to clear the song list, and click  to enter the sound card settings interface, as shown in the figure.





3.4.19. Camera Tracking



Voice Tracking Settings

Threshold: When a microphone input signal is greater than or equal to the threshold, the system automatically activates the tracking parameters.

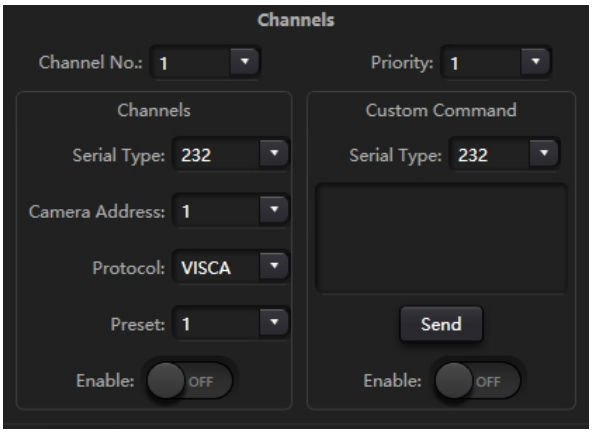
Default Channel: When no microphones are active, the camera will return to the preset position associated with the default microphone.

Reaction Time: The maximum allowed interruption time for a valid signal. For example, if a user speaks into the microphone and pauses for up to 3 seconds (when set to 3s), the signal is still considered active. A pause longer than this time renders the signal inactive.

Hold Time: The minimum speaking duration required for the camera to switch to a specific position. For instance, if a user speaks and the speech duration exceeds the switch time, the system deems it a valid signal, and the camera will automatically move to the preset position. This time is usually set longer than the response time.

Send Interval: The interval between each camera switching command.

Send Repeat: The number of times the camera switch command is sent. If set to 0, the command is triggered only once (special handling).



Microphone Settings

Channel No.: Typically corresponds to the device's input channel number, which indicates the channel the microphone is connected to.

Priority: The smaller the number, the higher the priority. When multiple microphones are triggered simultaneously, the system gives precedence to the one with the higher priority (i.e., smaller number). If two microphones have the same priority, the system handles the one that was triggered first.

The preset position, serial port number, camera address, and protocol must match the actual camera connection.

Enable: When enabled, the microphone settings take effect.

Custom Command: When the device detects an input signal (typically when someone is speaking), it will automatically send the defined command to the specified serial port. If the command is pre-set but "Enable Custom Command" is not checked, the device won't send it automatically, but you can still click the "Send" button to manually transmit the command.

Click "Save" to store the parameters in the device. At this point, the microphone is linked with the corresponding camera address.

You can then use the "Enable Microphone Settings" option to specify whether the microphone's settings are active when voice tracking is enabled.

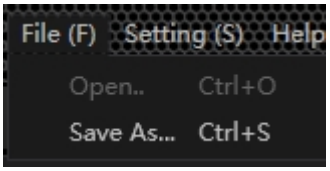
The camera setup section is used to test and adjust camera positioning, typically done before starting voice tracking. The final parameters will be stored in the camera.



To configure the PTZ camera control settings, first set up the serial port. There are two available ports (RS-232 and RS-485), which should correspond to the ports on the back panel connected to the PTZ camera. Then, configure the camera address and protocol type—the address should match the actual address set on the camera, and the protocol depends on the specific camera model. Next, assign a preset point number to serve as an identifier for the defined camera position. Use the directional controls (up, down, left, right) along with zoom, focus, and aperture adjustments to position the camera precisely. Once the desired position is set, click "Save" to store the preset; "Clear" to delete the current preset point information; and "Recall" to retrieve and view the saved camera position for the current preset point.

3.5. Setting Menu

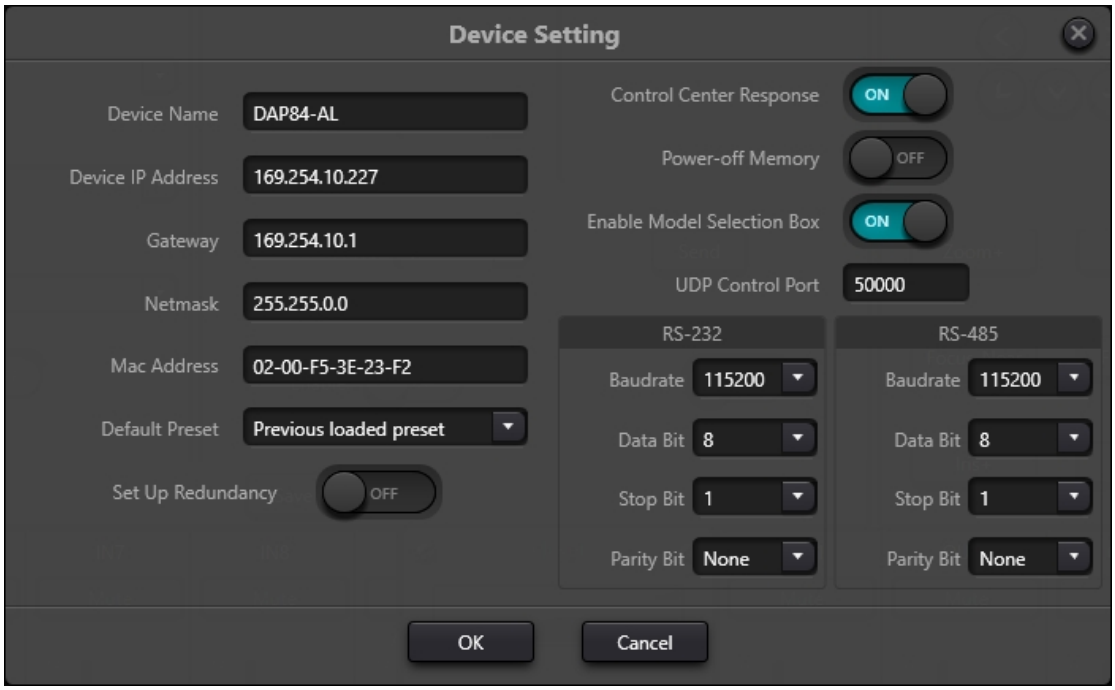
3.5.1. File Menu



Open: In offline mode, click "Open" to bring up a file dialog and select an existing preset file. You can also right-click the preset file and choose to open it with the DAP.exe application.

Save As: Saves the current preset from the application to the local hard drive, making it convenient for backup or copying to other devices.

3.5.2. Device Setting

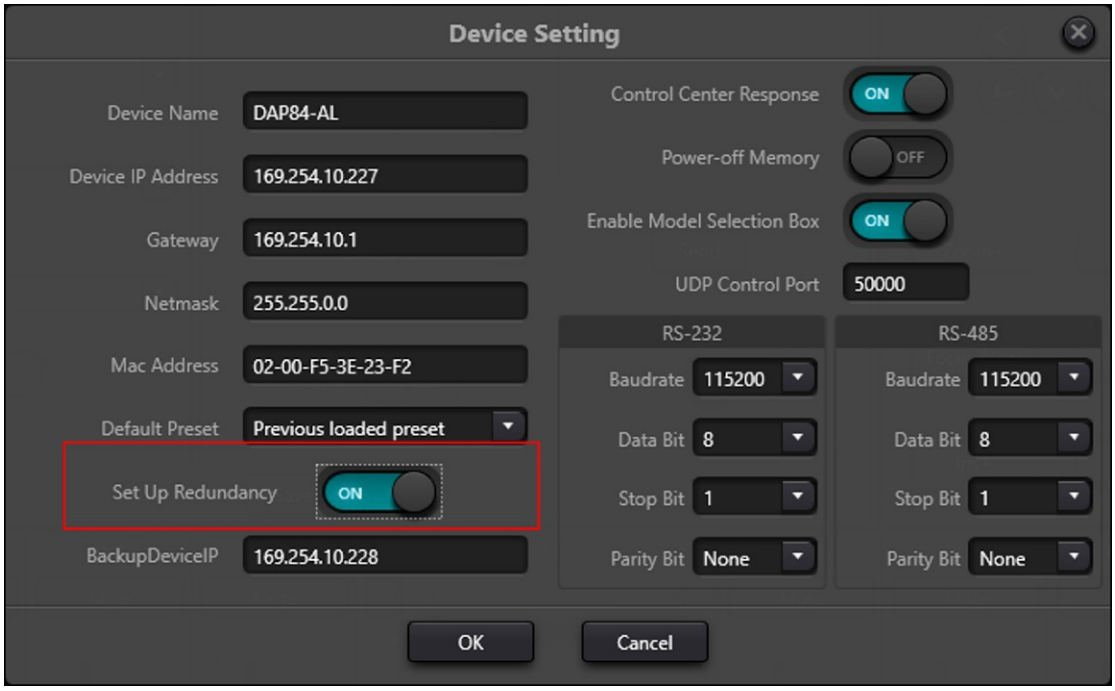


Set device name, network address, dual-machine hot backup, serial port baud rate, and other information. The device name supports up to 16 characters or 5 Chinese characters.

Default Startup Preset: Two startup preset modes are available.

1. Designated Preset: Select any one of the 16 presets to be used as the default. The device will load this preset every time it powers on.
2. Last Used Preset: The device will load the preset that was last used before power-off as the startup preset for the next boot.

Dual-Machine Hot Backup:



Click "Set as Redundancy", then enter the IP address of the standby device to complete the setup (as shown in the figure above).

3.5.3. GPIO Setting

Open the GPIO settings interface. The device includes a total of 8 GPIOs, each of which can be independently configured as input or output.

Input GPIO options include: Preset, Routing, Gain, Mute, Command, and Analog-to-Digital Gain.

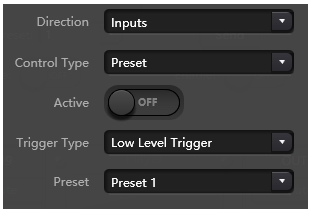
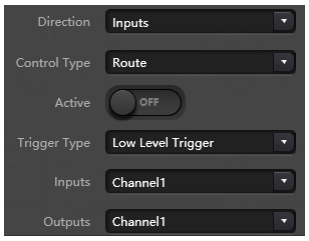
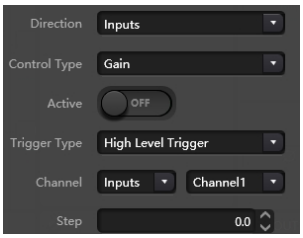
Output GPIO options include: Preset, Level, Mute, and Command.

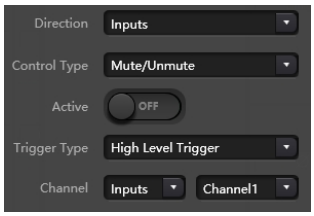
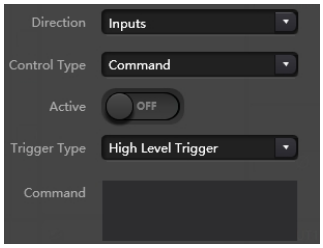
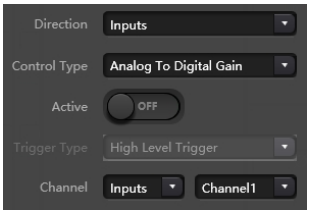
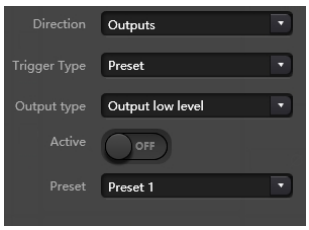
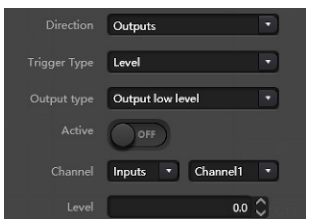


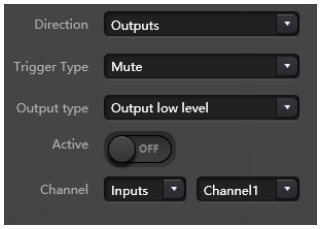






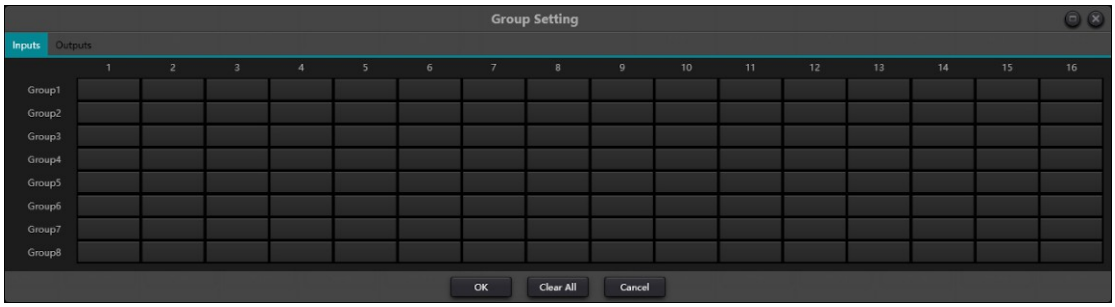
Input GPIO Settings	
Preset 	Trigger Type: High-level trigger / Low-level trigger / High-level trigger & low-level reset / Low-level trigger & high-level reset (These correspond to rising edge, falling edge, rising edge trigger with falling edge cancel, and falling edge trigger with rising edge cancel.) Preset: When the input signal at the hardware GPIO port matches the configured trigger type, the system will switch to the specified preset.
Route 	Trigger Type: Same as described above. Input / Output: Select the input channel to be mixed into the corresponding output channel. When the trigger condition is met, a mix or unmix action will be executed accordingly.
Gain 	Trigger Type: Same as described above. Channel: Select the input or output channel. Step Size: Increase the gain of the selected channel based on its original gain by the specified step size, measured in dB.

Mute 	Trigger Type: Same as described above. Channel: Select the input or output channel.
Command 	Trigger Type: Same as described above. Command: When the configured trigger condition is met, the specified command code will be sent out via RS232.
Analog-to-Digital Gain 	Analog-to-Digital Gain is especially useful when an external potentiometer is connected, as it allows for adjusting the gain of input or output channels. Although a potentiometer resembles a rotary encoder in appearance, their working principles differ significantly: a potentiometer is an analog device that adjusts voltage and current levels, while a rotary encoder is a digital device that transmits binary codes such as “0” and “1”.
Output GPIO Settings	
Preset 	Output Type: High Level / Low Level Preset: When switching to the specified preset, the corresponding GPIO port outputs a high or low level signal.
Level 	Output Type: High Level / Low Level Channel: Specify an input or output channel Level: When the level of the specified channel reaches the set threshold, the GPIO outputs a high or low level. Otherwise, it outputs the opposite level.

Mute 	Output Type: High Level / Low Level Channel: Specify an input or output channel. When the channel is muted, the GPIO outputs the configured high or low level. When unmuted, it outputs the opposite level.

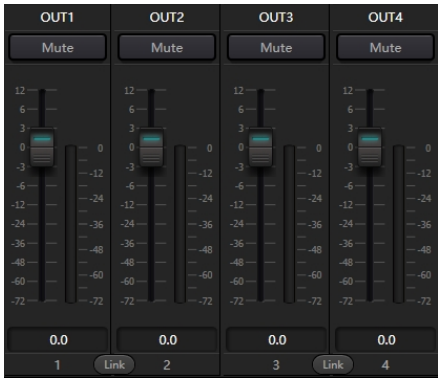
3.5.4. roup Setting

The grouping interface is divided into two tabs: Input and Output. Each channel can only belong to one group. Within the same group, volume and mute adjustments for the grouped channels are synchronized. Other module parameters are not synchronized, which is the key difference from the LINK function.



Each group can include from 1 up to the maximum number of channels supported by the device, depending on the model purchased. Once channels are grouped, they are color-coded in the main interface for identification.



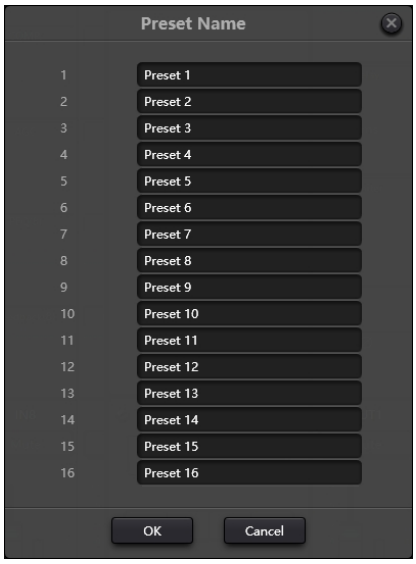


Relationship between Group and LINK:

When a channel is assigned to a group, it will not participate in LINK. This means grouping takes priority over LINK. The main difference is that grouping only synchronizes gain and mute controls, whereas LINK synchronizes all parameters on the linked channels.

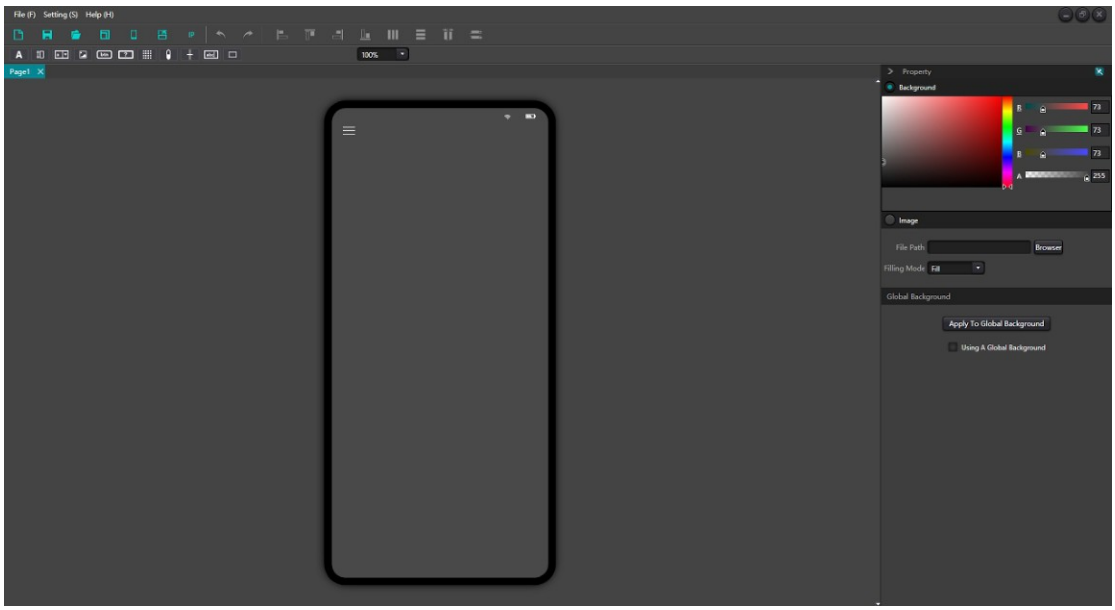
3.5.5. Preset Name

Supports name editing for 16 preset groups, making it easier to recall different scenarios.



3.5.6. User Interface

This feature allows engineers to create customized interfaces that can be edited by integrators and operated by on-site technicians or end users with no technical background. Advanced security functions can only be accessed by end users within the limits permitted by the system designer or integrator.

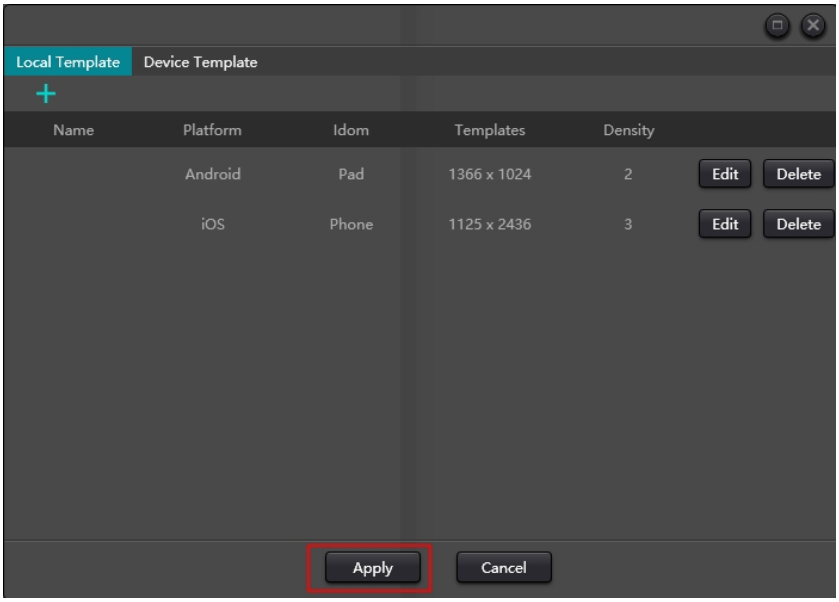


Online Template Selection

After installing the software on your mobile device, open it and make sure the screen stays active. Additionally, ensure that both the mobile device and the PC are

connected to the same local area network (LAN).





Editing Interface

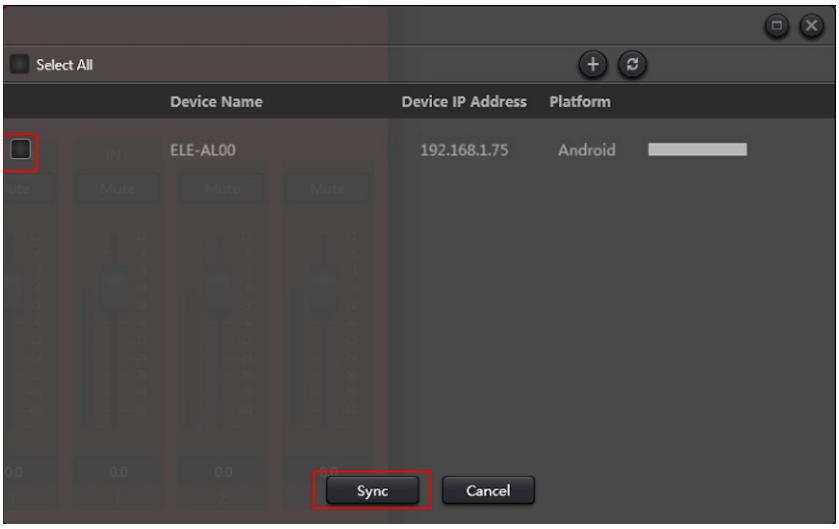
Drag the controls from the top section to modify the corresponding properties.





Upload





Click "Sync" to upload to the mobile device. When uploading, make sure the app is open on your phone or tablet. If you encounter a timeout or the mobile device cannot be found, try restarting the software or check whether the devices are on the same network segment.

3.5.7. Help Menu

Command Capture



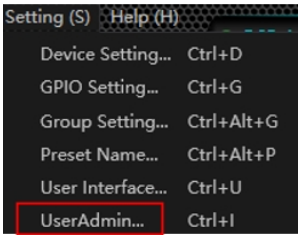
Open the command capture window. Click on the parameter you want to control in the interface, and the command window will instantly display the current command. Copy the command into your control program to send it to the device via UDP/RS232/RS485 protocol.

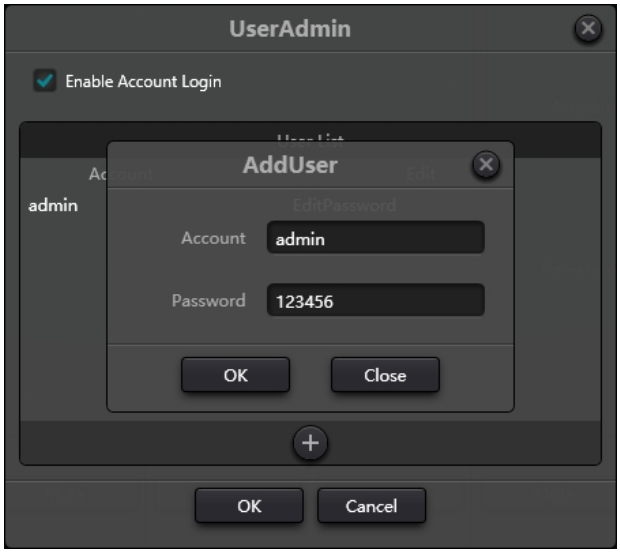
About

Displays version number, technical support contact information, copyright details, and more.



3.5.8. User Management





Enable Account Login – Click the "+" icon to add new users (sub-accounts). There can only be one main account, which has full privileges to add or remove sub-accounts as needed.



Once account login is enabled, users must enter the corresponding username and password to access the device (as shown above).

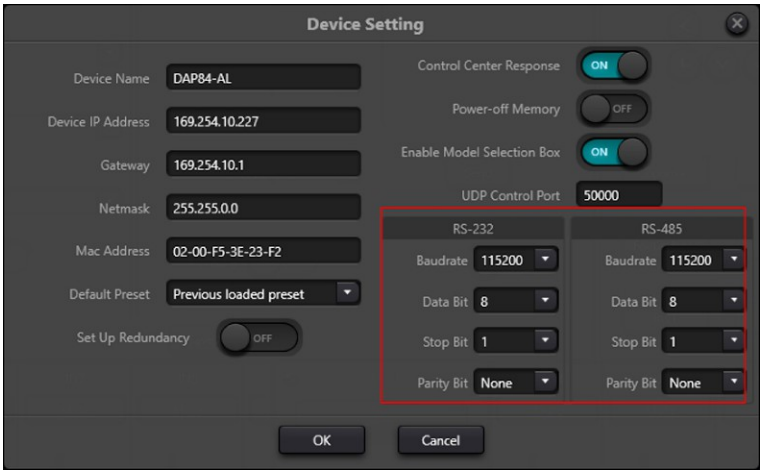
4. Control

4.1. External Control Programmer

External control programming supports UDP, RS232, and RS485 protocols. The control protocol covers all parameters of the processor and includes three parts: parameter control, parameter retrieval, and preset recall.

When using UDP control, the default port is 50000, which can be viewed and modified in the “Device Settings” of the upper computer software. For RS232 or RS485 control, the default communication settings are: baud rate 115200, data bits 8, stop bit 1, and no parity. These settings can also be configured in the “Device Settings” as needed. When sending commands via RS232 or RS485, ensure there is a minimum 200 ms interval between messages.

If a response from the device is required, enable the Control Center Response switch in the “Device Setting”.



5. Audiolink Audio Network Module (Optional)

AudioLink is similar to Dante. The AudioLink Input source must be connected and configured through the conference host. You can select sources such as Line In1, Line In2 of the conference host, audio from conference unit IDs, individual microphone audio, interpreter unit audio, mixed output audio from the conference host, zone output audio, or audio input from other DAP devices with AudioLink interfaces.

Note: By default, AudioLink network PoE speakers output only the first Dante channel (Dante1). Without a conference host, all connected AudioLink PoE speakers will output from this single channel only. When used with a conference host, more output options can be configured.

For product series integrated with the AudioLink audio network module, some input/output channels in the quick control area are labeled as Dante channels. The interface is shown below:



About VISSONIC Electronics Limited

Our mission is to develop and manufacture the most comprehensive and innovative audio visual products for our clients. We provide the best performance/price ratio products because it could give you satisfaction just from the time you use them, we believe the good design with cutting edge technology on products will provide value to all our partners and end users. Listen to your demands, we fulfill it.

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