AUDIOCENTER

Dynamic Audio Solutions

User's Manual

Artist Talent Series

Audio Matrix



Revision A0-2026-9

Important Precautions



This symbol is used to alert the operator to follow important operating and precautions detailed in documentation.



This symbol is used to warn operators that uninsulated“dangerous voltages” are present within the equipment that may pose a risk of electric shock.

1.Save the carton and packing materials even if the equipment is arrived in good condition. Should you ever need to ship the device (back to the factory), you can only use the original manufacturer’s packaging.

2.Read all documentation before operating your equipment. Retain all documentation for future reference.

3.Follow all instructions printed on unit chassis for proper operation.

4.Do not spill water or other liquids into or on the unit, or operate the unit while standing in liquid.

5.Make sure power outlets conform to the power requirements listed on the back of the unit.

6.Do not use the unit if the electrical power cord is frayed or broken. The power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles, and the point where they exit from the appliance.

7.Always operate the unit with the AC ground wire connected to the electrical system ground. Precautions should be taken so that the means of grounding of a piece of equipment is not defeated.

8.Mains voltage must be correct and same as that printed on the rear of the unit. Damage caused by connection to improper AC voltage is not covered by any warranty.

9.Have gain controls on processors turned down during power-up to prevent amplifier or speaker damage if there are high signal levels at the inputs.

10.Power down and disconnect units from mains voltage beforemaking connections.

11.Never hold a power switch in the “ON” position if it won’t staythere itself.

12.Do not use the unit near stoves,heat registers,radiators,or other heat producing devices.

13.Do not block fan intake or exhaust ports. Do not operate equipment on a surface or in an environment which may impede the normal flow of air around the unit, such as a bed, rug, weather sheet, carpet, or completely enclosed rack. If the unit is used in an extremely dusty or smoky environment, the unit should be periodically “blown free” of foreign matter.

14.Do not remove the cover. Removing the cover will expose you to potentially dangerous voltages. There are no user serviceable parts inside.

15.Do not connect the inputs / outputs of processor or consoles to any other voltage source, such as a battery, mainssource, or power supply, regardless of whether the processor or console is turned on or off.

16.Non-use periods: The power cord of equipment should beunplugged from the outlet when left unused for a long period of time.

17.Service information: Equipment should be serviced by qualified service personnel when:

A.The power supply cord or the plug has been damaged;

B.Objects have fallen,or liquid has been spilled into the equipment;

C.The equipment has been exposed to rain;

D.The equipment does not appear to operate normally,or exhibits amarked change in performance;

E.The equipment has been dropped,or the enclosure damaged.

18.To obtain service, contact your nearest AUDIOCENTER service centre, distributor or dealer.

AUDIOCENTER

Artist

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◆ Introduction

The Artist Talent Series Network Digital Audio Matrix is offered in four distinct hardware variants. The T88+ provides 8×8 channels of analog I/O, while the T16+ doubles that capacity with 16×16 analog channels. For networked audio applications, the T16A+ expands the platform to 32×32 AES67 channels, and the T16D+ delivers 32×32 Dante I/O with support for 16-channel redundant backup. This broad I/O portfolio supports flexible, reliable system integration for complex audio processing and centralized control applications.

Web-based network control is fully terminal-agnostic and compatible with all operating systems, enabling remote device management and fast parameter configuration over LAN. Linked debugging supports batch setup across multiple units of the same model, while wall panel controllers and RS 232/RS 485 central control systems provide localized scene recall and volume adjustment.

◆ Highlights

- **1 GHz High-Performance DSP Core with Full Processing Power**

  Built around the ADI SHARC® 5th generation 21569 processor, the platform delivers a 1 GHz computing core capable of handling FIR filtering, Dynamic EQ, and fourteen filter types. With an integrated 5A algorithm, including AEC for every channel, and up to 2000ms of adjustable I/O delay, it provides precise acoustic optimization and meets the most demanding installs needs.

- **Five Intelligent Algorithms Integrated – One Click "Plug and Play" Operation**

  • **AFC** (Adaptative Feedback Cancellation) delivers dual loop AI-adaptive howling suppression with a 30% efficiency gain.
  • **ANC** (Active Noise Cancellation) actively suppresses ambient noise for cleaner pickup.
  • **AEC** (Acoustic Echo Cancellation) prevents your own voice from echoing during a call or video-conference, and features a unique adaptive algorithm constantly monitoring audio coming in from far-end.
  • **AMX** (Auto mixer) intelligently manage the volume of multiple microphones open simultaneously, with up to 10 priority levels.
  • **AGC** (Automatic Gain Control) automatically adjusts the volume of every channel signal to keep it at a constant and optimal level.

- **Multi Dimensional Management & Full Interface Compatibility**

  Flexible management meets scalable expansion through zone control, Audiocenter system presets, and user-defined presets for refined scenario based operation. Standard TCP/IP, RS 232, RS 485, and GPIO interfaces ensure seamless integration with central control systems and smart panel, accommodating diverse control and connectivity requirements.

- **Full Scenario Audio Source Compatibility – No Additional Hardware Required**

  A single unit accepts virtually any audio source, with analog, stereo Bluetooth, and USB sound-card I/O provided as standard. Optional AES67 or Dante digital interfaces support high fidelity professional applications. Integrated 48V phantom power supplies condenser microphones directly, eliminating the need for external adapters.

◆ Unpacking

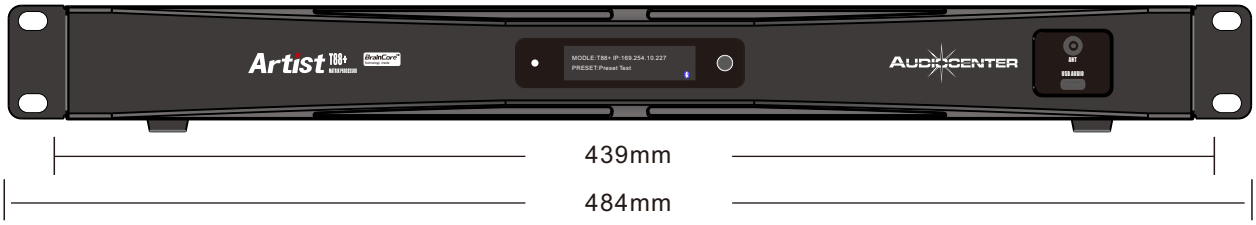
Please inspect the audio matrix carefully immediately after unpacking. If you find any damage, notify your supplier /dealer immediately. Only the shipper may file a damage claim with the carrier for damage incurred during shipping. Be sure to save the carton and all packing materials for the carrier's inspection. If you ever need to ship the unit back to AUDIOCENTER or an authorized service center, you should use only the original manufacturer's packaging.

◆ Installation

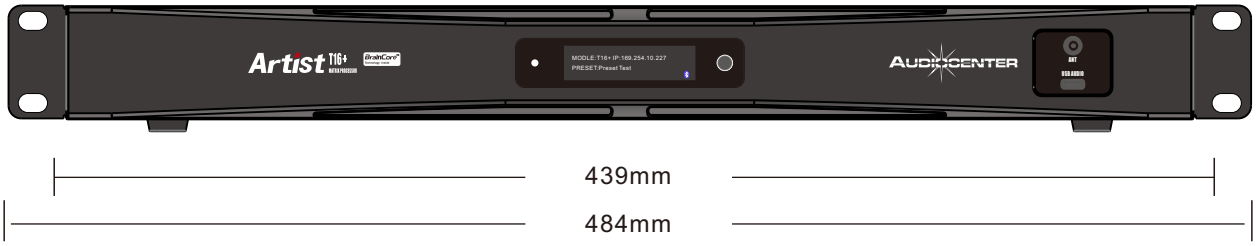
Artist Talent Series is one-rack-space high. Four front-panel mounting holes are provided on each audio matrix. All mount in standard 19-inch racks.

◆ Front Panel

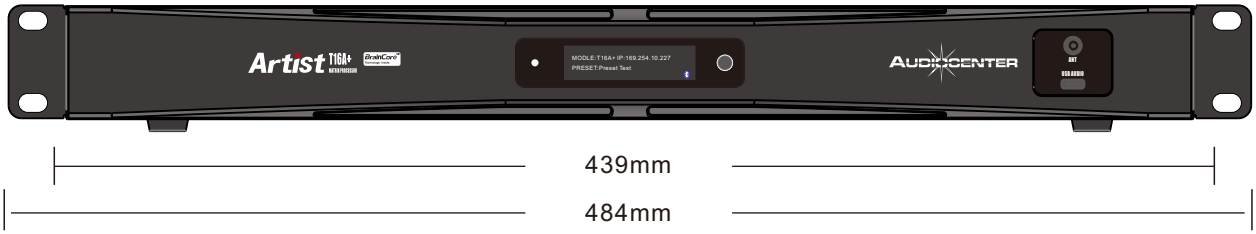
Artist T88+



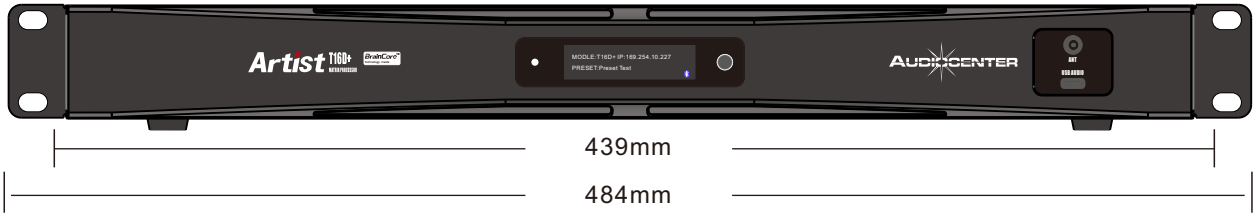
Artist T16+



Artist T16A+

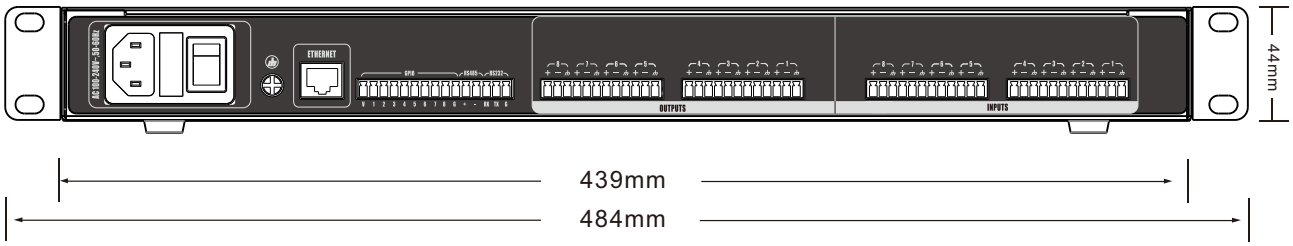


Artist T16D+

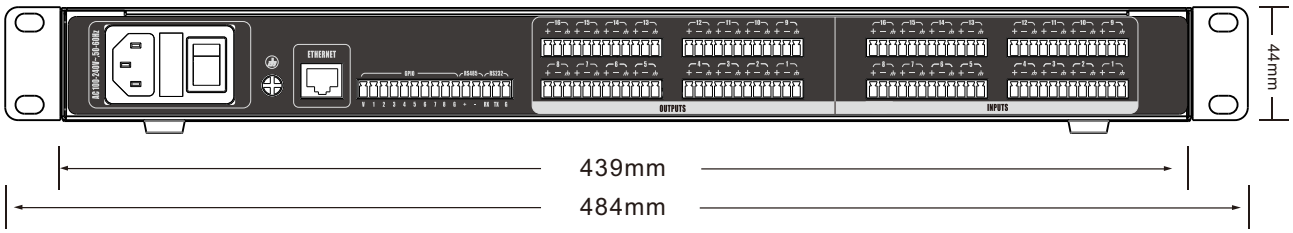


◆ Rear & Side Panel

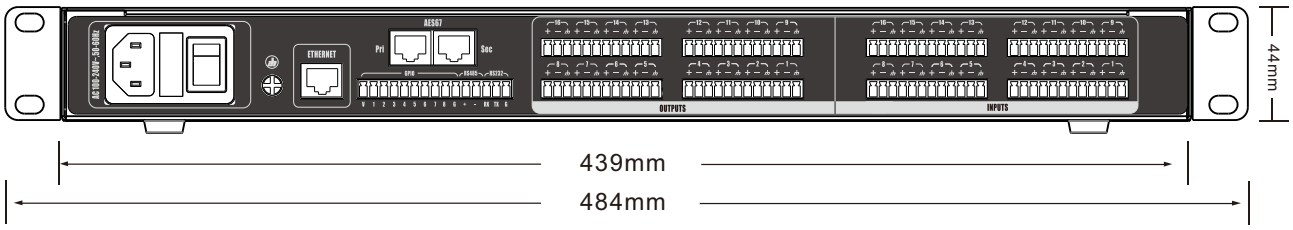
Artist T88+



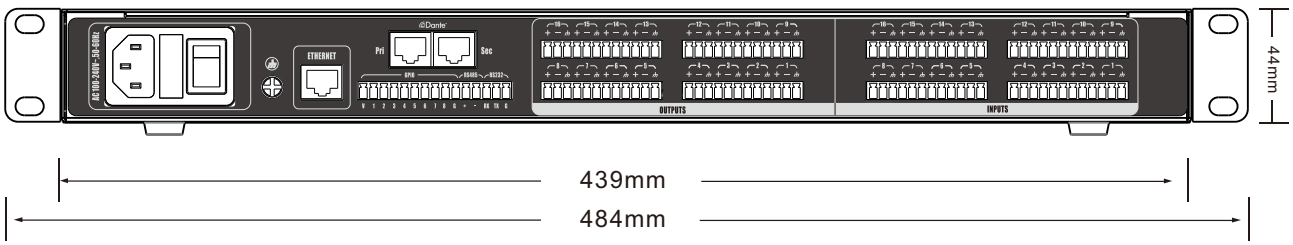
Artist T16+

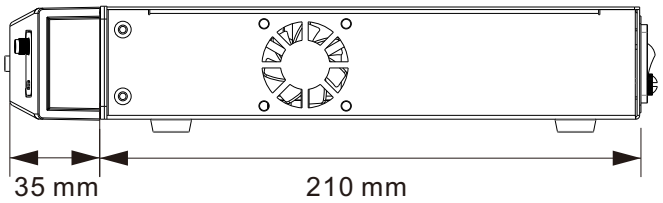


Artist T16A+



Artist T16D+





◆ Front panel function introduction

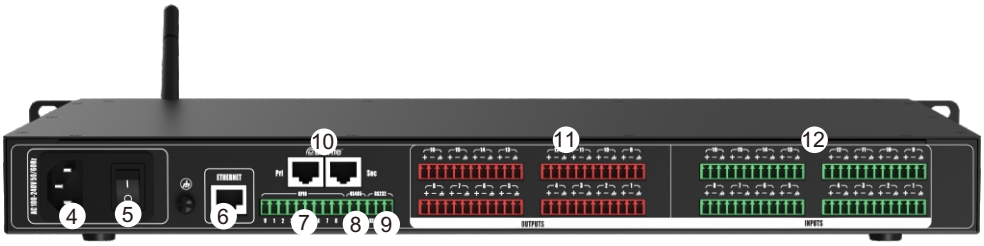


❶ **2.08-inch TFT display**
It can display model number, IP address, current preset, show channel input/output status

❷ **Bluetooth Antenna**

❸ **USB AUDIO**
USB sound card, supports stereo input and output

◆ Rear panel rear panel function introduction



❹ **Power supply socket**
Connected via standard IEC plug

❺ **Power switch**
When the processor is not in use, please keep its power switch off.

❻ **ETHERNET computer control interface**
Rj45 interface, based on TCP / IP protocol, standard CAT5 or CAT6 network cable connection.

❼ **GPIO interface**
GPIO control extension

❽ **RS 485 interface**
It can be used as central control, panel control, camera tracking and expansion interface.

❾ **RS 232 interface**
It can be used as central control, panel control, camera tracking and expansion interface.

❿ **Network Audio Interface**
T16A+: 32×32 channels, AES67
T16D+: 32×32 channels, Dante
T88+/T16+: Not equipped with this interface

⓫ **16-channel analog output（T88+：8-channel analog output）**
Euro-block connector, balanced output

⓬ **16-channel analog input（T88+：8-channel analog input）**
Euro-block connector, balanced input

◆ Quick application guide

The control of the Artist Talent series matrix is operated and accessed via a web browser, with no need to download or install any software.

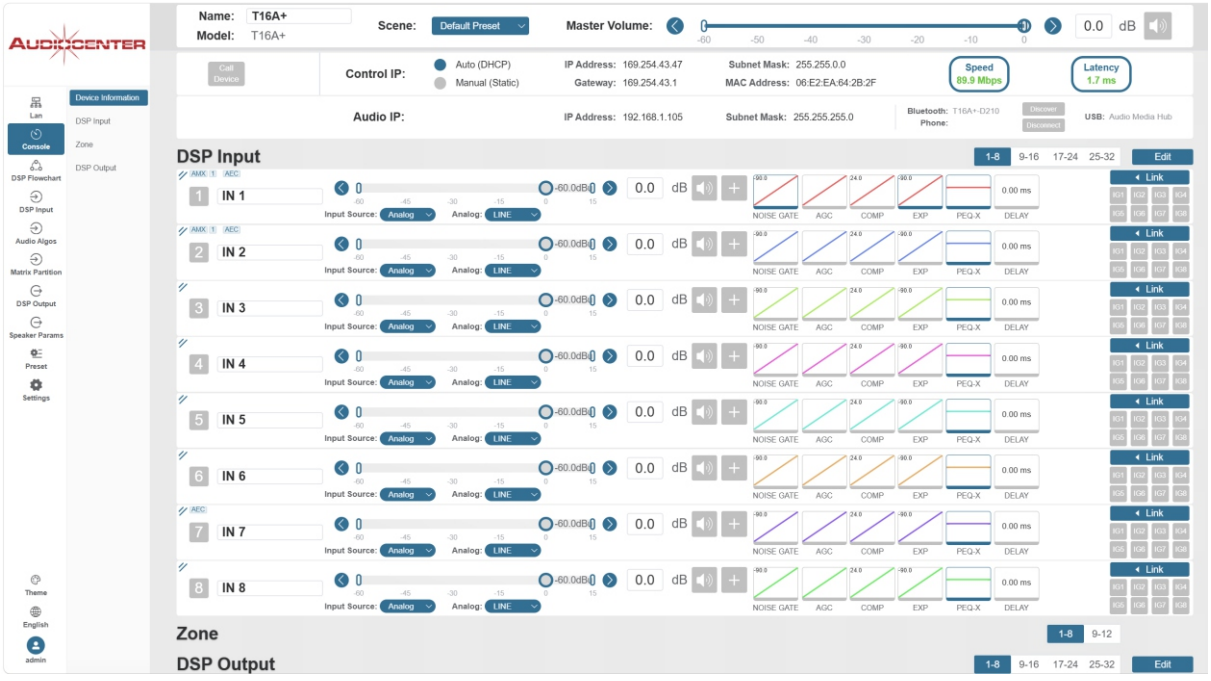
Note: It is recommended to log in and use mainstream browsers such as Google Chrome or Microsoft Edge, as the IE browser is currently unavailable.

◆ Usage steps

1. Turn on the power and start the device, wait for the display to enter the main interface (about 20 seconds);
2. Network connection



- Artist Talent series rear panel is equipped with a network port for internet connection.
- Connect this port on each matrix to your network or computer using Cat 5e cables.
- Enter the network address of the desired matrix in the browser's address bar (view the IP through the display), then press Enter. You can then log in and proceed with operational settings.
- Note: The default IP assignment method of this device is DHCP. If the device is not connected to a computer via a switch, the initial IP is 169.254.X.X with a subnet mask of 255.255.0.0. You need to set the computer's IP address to the same subnet as the device (i.e., IP: 169.254.X.X, subnet mask: 255.255.0.0).



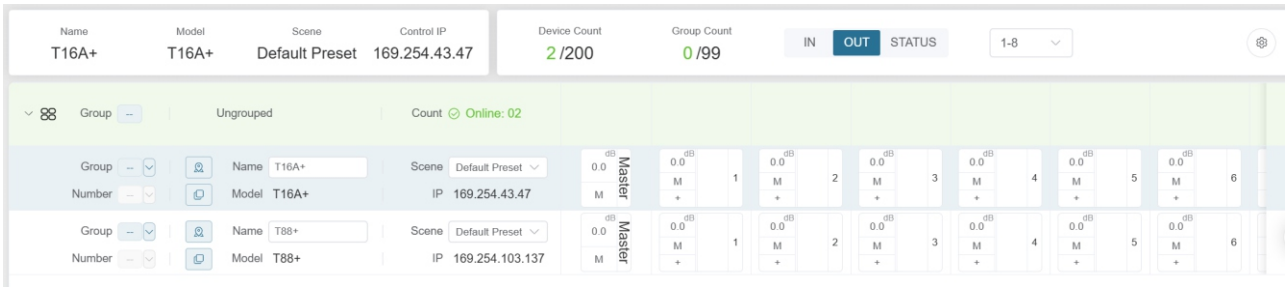
◆ WEB interface description



The modules corresponding to the red characters in the main interface of the software shown in the above picture are as follows:

1. Menu bar
2. Device name, current archive and total volume control
3. Device status and local/network audio IP address
4. Input channel interface
5. Partition channel interface
6. Output channel interface

Local Area Network



Users can search for all online devices on this interface and then manage them in groups.

1.Click on the call recognition device

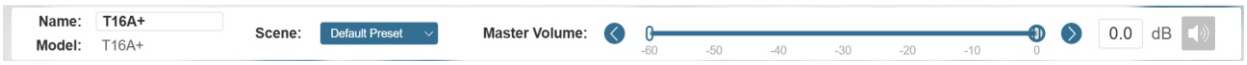
2. Click directly on the control web interface of this matrix

3. Select the input, output and status joint debugging interface;

4. Click to open the batch upgrade interface

◆ Control Panel

Device name, current archive and total volume



1. Click the device name can be used to name the device.



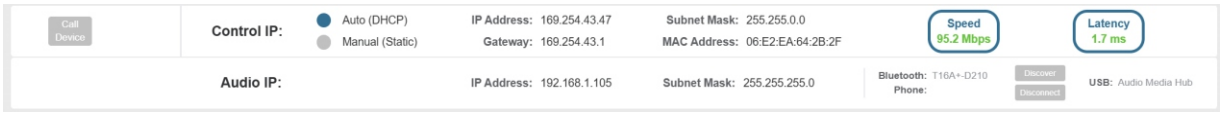
2. Click on the scene to quickly call up the preset you need to use.



3.Here, you can adjust the total volume by swiping, or directly mute or unmute the device. You can also enter the volume in the text box.



Device status and local/network audio IP address



1.Click the blue-gray toggle button; this will make the display and status LED of the connected unit blink so you can quickly locate it among multiple devices.



2. It can be seen whether the IP is allocated manually (static) or automatically (DHCP), as well as the IP address of the network audio. The transmission speed and network delay on the right allow you to see the signal stability of the network connection in real time.

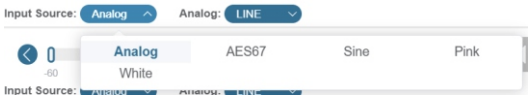
Input channel interface



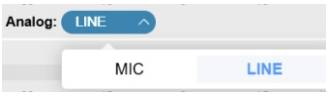
The above picture shows the list of input channels, which can be adjusted for phase, mute and volume. It also has the functions of sensitivity, phantom power and channel joint adjustment, and can display the name of the input signal.

1. Clicking allows you to modify the channel name, which supports characters such as Chinese.

2. Click on the input source on the right. You can choose one of the three input modes: analog input, digital input, and test signal. Each channel can only have one input mode.

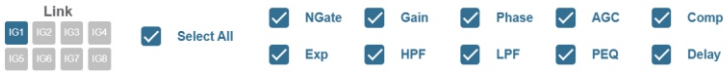


Click on the right to switch between LINE mode and MIC mode. After selecting the MIC mode for the analog input





The sensitivity of the Mic input in the above figure (0 to 40dB) can also be set to whether to turn ON the 48V phantom power supply (OFF/ON).



3. The above figure can be used for DSP input group joint debugging. By clicking on the joint debugging group Settings, you can also select the parameters that need to be debugged for different groups.

4. The size of the input signal can be seen in the right figure.



The circles on the right represent the channel gain size. Slide left or right to adjust the volume.



The volume, mute and phase of the channel can also be set in the right picture.



In the right picture, gray represents unmute and positive phase, red represents mute and blue represents reverse phase.



The right figure can display the parameters input by the DSP. Clicking on a certain module can directly jump to the interface for setting parameters.

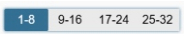


The color in the right picture indicates that this function has been enabled.



Note: The AGC, COMP (Compressor), and EXP (extender) functions can only be enabled one at a time and cannot be enabled simultaneously. If AGC is enabled and then COMP (Compressor) is enabled, the AGC function will be disabled, and at this point, only the COMP (Compressor) function can be used.

5. The right picture can switch between different channel interfaces and display the corresponding parameters.

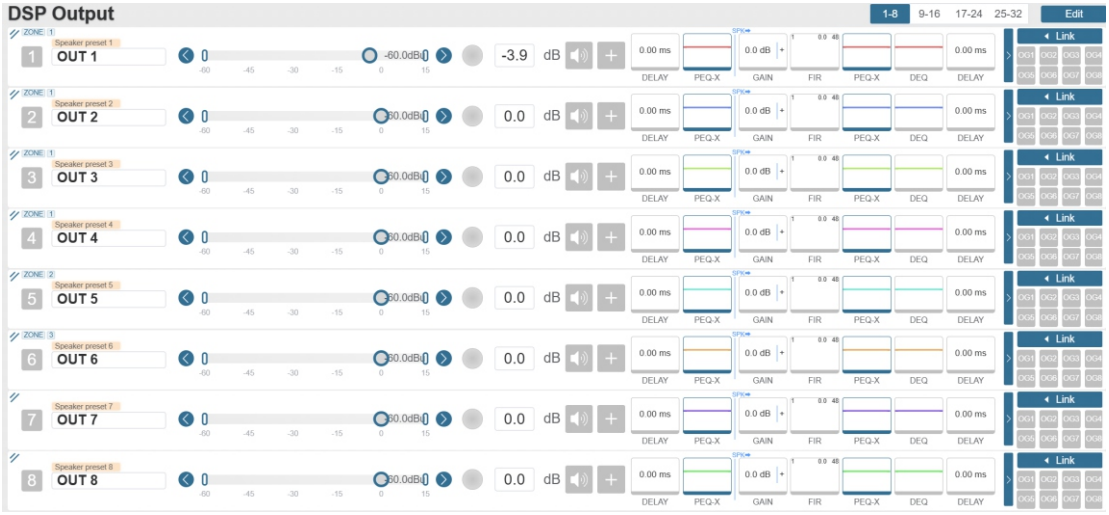


Partition channel interface



Functions and operation methods are related to matrix partitioning. Note: The partitioning function must be set in the matrix partition to be displayed in the control interface.

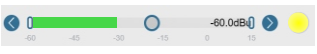
Output channel interface



The above figure shows the list of output channels, which can be adjusted for phase, mute and volume. Moreover, it has the functions of gain setting and DSP output group joint adjustment, and can also display the size of the output signal.

1.Click on the right picture to modify the channel name, which supports characters such as Chinese.

2.The right figure shows the limit level set for the current channel, the corresponding voltage limit level and the actual output level. When the limit is in effect, it will light up yellow. The red color on the level bar represents the compression amount. The specific size can be seen in the red font below. The yellow color on the level bar indicates the position of the currently set limit value on the level bar. The specific parameters are displayed above the level bar.



3.The right picture can be used for group joint debugging (joint debugging of DSP output). By clicking on the joint debugging group Settings, you can also select the parameters that need to be jointly debugged for different groups.



The size of the output signal can be seen in the right picture.

The circles on the right represent the channel gain size. Slide left or right to adjust the volume.



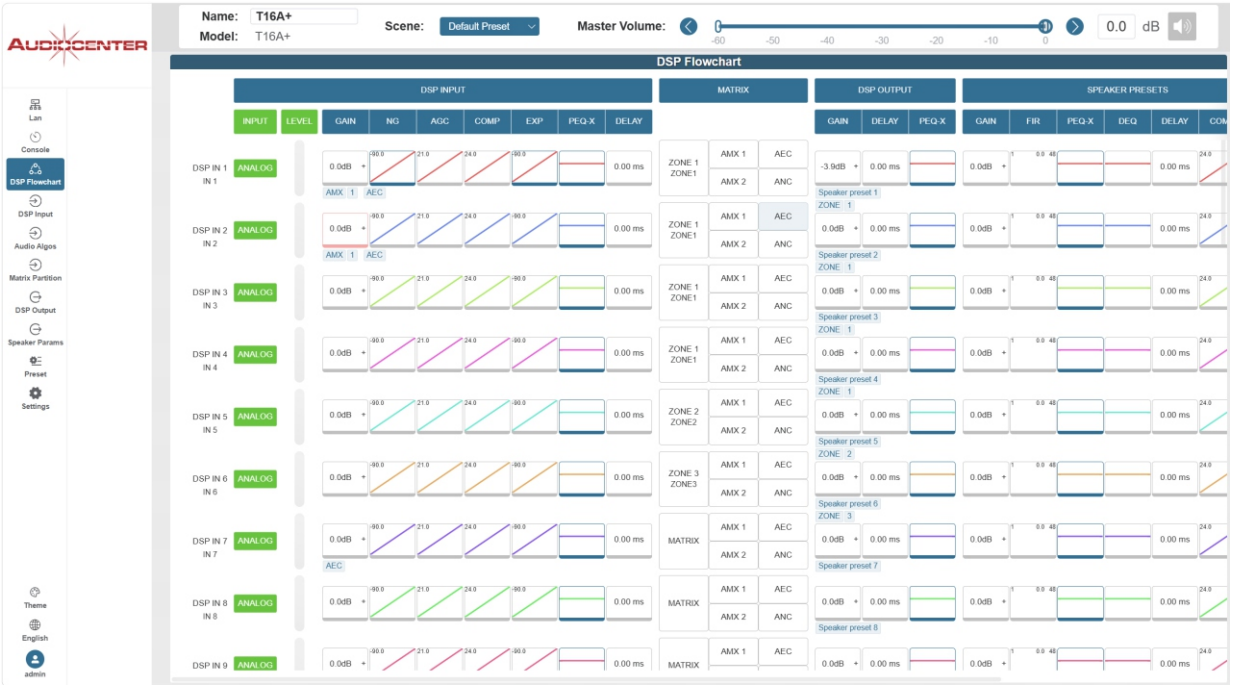
The right figure can also set the volume, mute and phase of the DSP output channel. Gray represents unmute and positive phase, red represents mute and blue represents reverse phase.

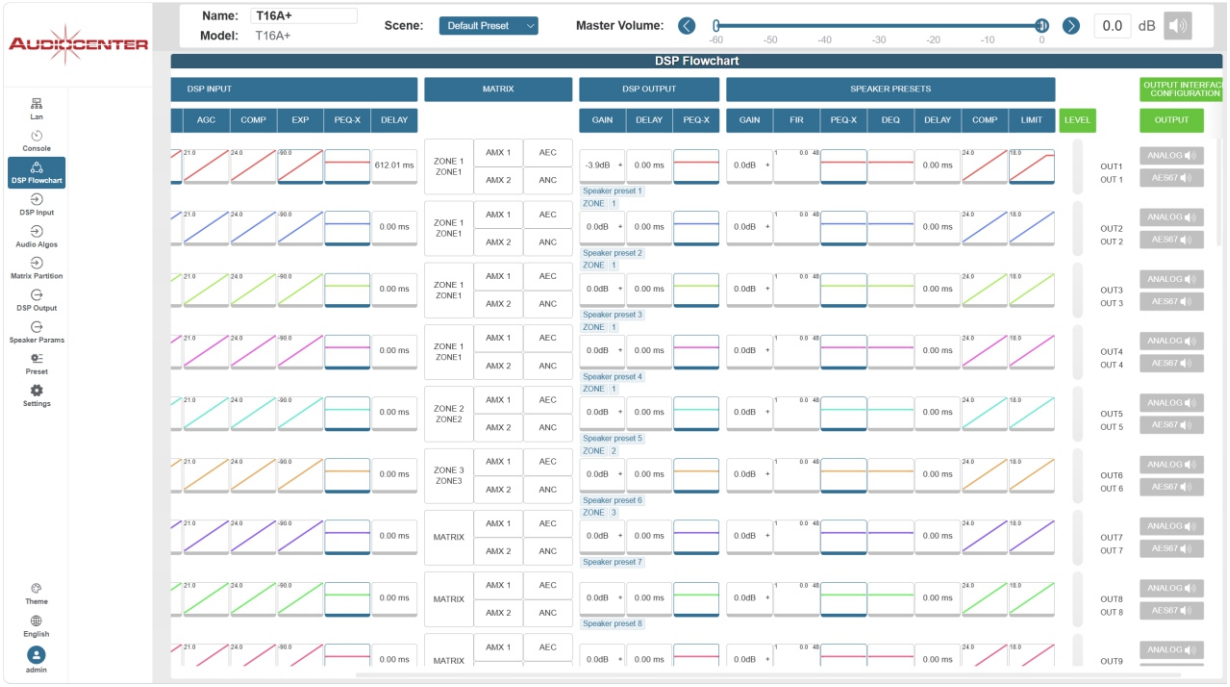
The right figure can display the output parameters of the DSP and the parameters of the speaker. Clicking on a certain module can directly jump to the interface for setting parameters.



The blue color in the right picture indicates that this function has been enabled.

◆ DSP flowchart



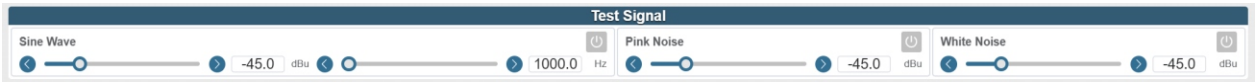


The above picture is the flowchart of the entire Audio Matrix. The flowchart also shows the Settings of each parameter of the equipment. Clicking on a certain DSP function module in the right picture will jump to the specific location of the clicked DSP function.

DEQ

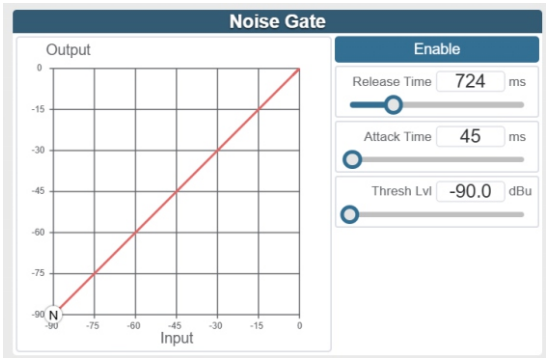
◆ DSP input

Test signal



Turn on the internal test signal source through here

Noise gate



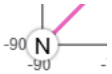
As shown in the above figure, clicking the noise gate switch in the upper right corner can turn on or off the noise gate function of the channel input signal. Blue indicates it is on, and gray indicates it is off.



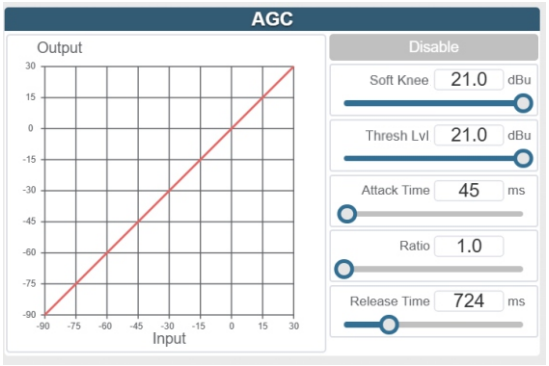
The Attack time, release time, and Threshold are the parameters corresponding to the noise gate function. All of them can be dragged by the slider on the right or input the corresponding values in the value box.



When the noise gate function is enabled, the points on the right chart can be dragged diagonally to adjust the threshold curve.



AGC



In the above picture, clicking the AGC switch in the upper right corner can turn on or off the AGC function of the channel input signal. Blue indicates it is on, and gray indicates it is off.



Start-up threshold: When the input level has not reached this point value, AGC does not intervene in the adjustment, and the input and output levels change linearly and synchronously at a 1:1 ratio. When the input level exceeds this point, AGC officially initiates gain compression.

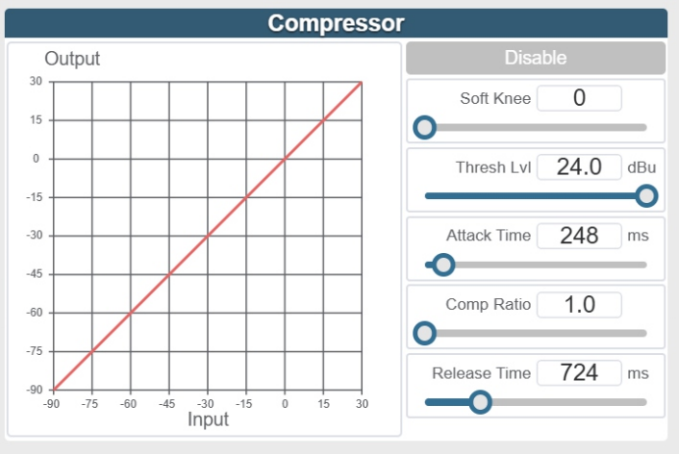
Target threshold: During the compression transition section, as the input continuously increases, the amplitude of the output level rise is gradually restricted, and the upper limit of the output level is locked. Even if the input level continues to increase significantly, the output level basically does not increase again, eliminating overload distortion.

Start-up time: The response time of the AGC when it switches from the direct state to the compressed state after the input signal suddenly becomes strong and exceeds the start-up threshold.

Ratio: The ratio of the input level increment to the output level increment within the compression range, representing the compression intensity. The larger the ratio value, the stronger the compression force and the more stable the output volume.

Release time: The time it takes for the AGC to rebound from the compressed state to the normal direct amplification state after the high-peak loud signal disappears.

Compressor



Soft inflection point: The soft inflection point does not immediately compress the signal at the set compression ratio when it reaches the threshold. Instead, there is a gradually transitioning area near the threshold where the compression ratio gradually changes from 1:1 (i.e., no compression) to the set compression ratio.

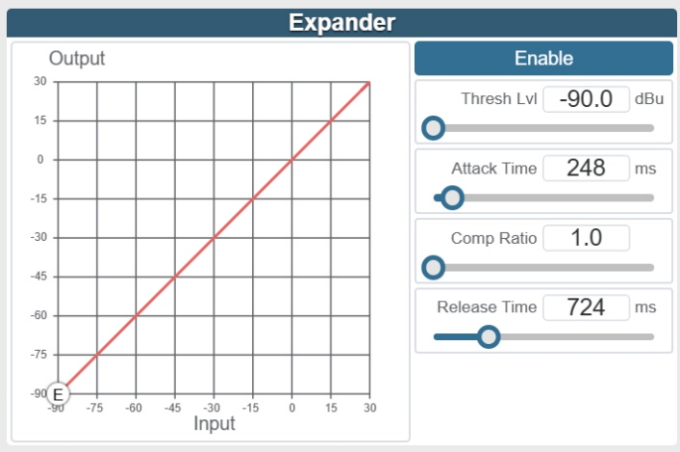
Start control level: When the signal level is higher than this value, the compressor/limiter begins to reduce the gain. Any signal that exceeds the threshold will be regarded as an overshoot signal, and under normal circumstances, its level will be reduced. The greater the range by which the signal exceeds the threshold, the more the level is attenuated.

Compression ratio: The ratio determines the extent to which the overshoot signal attenuates towards the threshold level. The smaller the compression ratio is, the easier it is for the signal to exceed the threshold. Once the signal exceeds the threshold, the compression ratio parameter determines the ratio of the change in the input signal to that in the output signal. For instance, when the compression ratio is 2:1 and the input signal exceeds the threshold by 2dB, the part of the output signal that exceeds the threshold only changes by 1dB. A compression ratio of 1:1 indicates that the compressor does not proportionally attenuate the signal. The adjustable range of the compression ratio is 1 to 20.

Start-up time and release time: To preserve the natural start sensation, it is usually desired that a portion of the initial level can pass through the compressor without being affected (or only slightly affected). To achieve this goal, it is necessary to slow down the response time of the compressor. Similarly, if the signal gain experiences a significant and rapid attenuation and a rapid recovery, the suction effect will occur. The setup time and release time of the compressor are designed to prevent this from happening. The establishment time can determine the rate at which gain decay occurs, while the release time can determine the rate at which gain recovery occurs.

Output gain: Also known as gain compensation fader. If the compressor significantly reduces the signal level, it may be necessary to increase the output gain to maintain the volume. This lifting operation provides a consistent boost to all parts of the signal and is independent of the Settings of other parameters of the compressor.

Extender



Start control level: The signal must exceed this level to open the extender (allowing the signal to pass through). In fact, it is generally set to the magnitude of environmental noise.

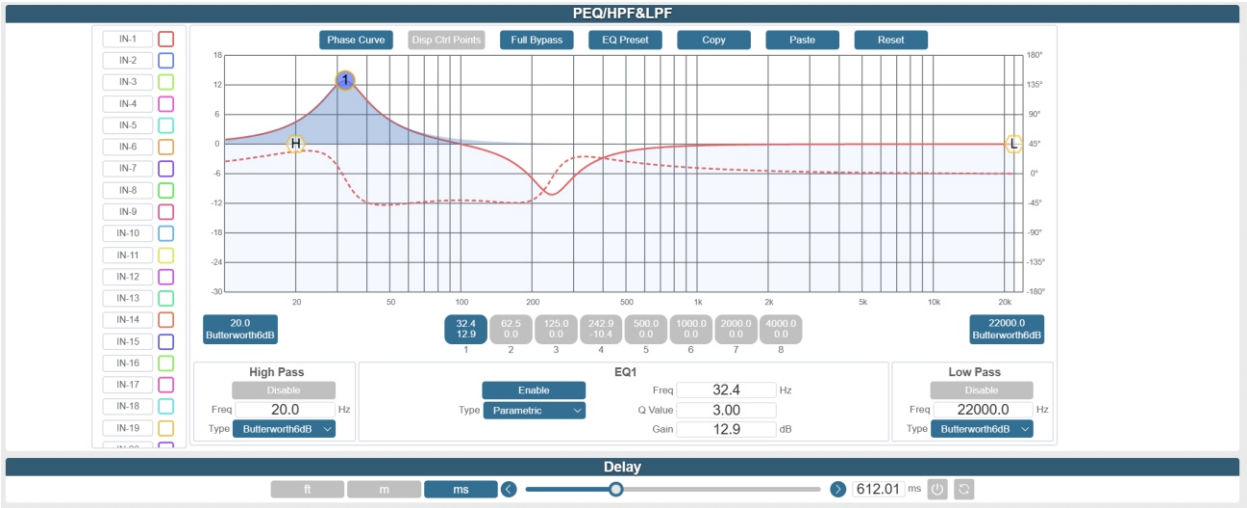
Start-up time: The duration of the input signal, the time required to open the expander when it exceeds the threshold. A faster opening time allows for a quicker transient opening of the extender.

Compression ratio: The slope below the threshold point on the gain curve. When the ratio is set high, start the action of approaching the door.

Release time: The time required for the gain to recover to a value below the threshold after the input signal drops below the threshold.

Note: The AGC, COMP (Compressor), and EXP (extender) functions can only be enabled one at a time and cannot be enabled simultaneously. If AGC is enabled and then COMP (Compressor) is enabled, the AGC function will be disabled, and at this point, only the COMP (Compressor) function can be used.

PEQ/ High and low pass



①The functions of the top buttons are as follows:

Phase curve: Displays the phase curve of the current channel.

Display control points: Show or hide all equalization control points.

Full bypass: Simultaneously turn on or off all equalizers (EQ) of the current channel

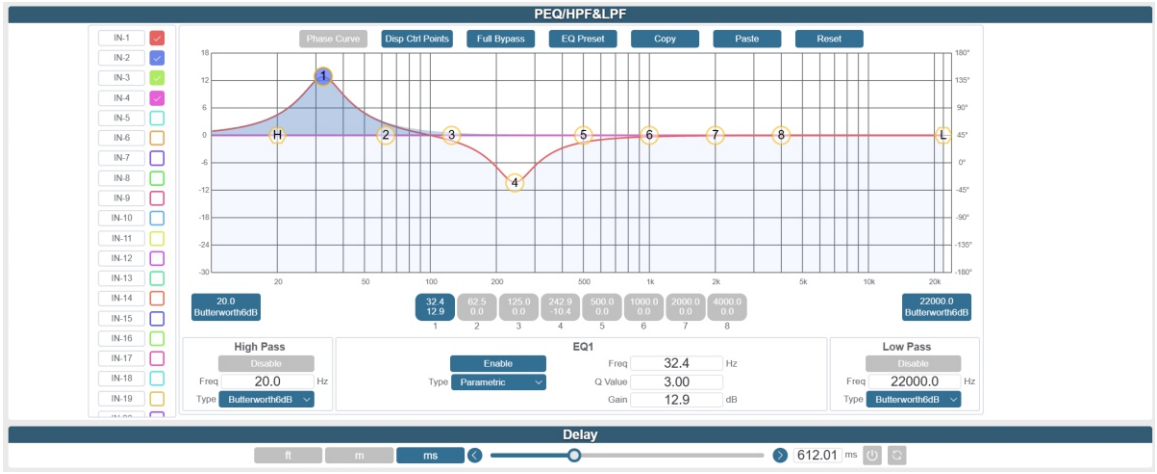
EQ Archive: Save the current equalizer setting parameters to your computer and retrieve them to overwrite the existing equalizer parameters.

Copy: Copy the current equalizer parameter value and paste it into other input channels.

Paste: When used in combination with the copy button, the equalizer parameter values copied by the copy function can be pasted into the current channel.

Reset: Reset the equalizer parameters to the default parameter values.

②Multi-channel EQ curve display



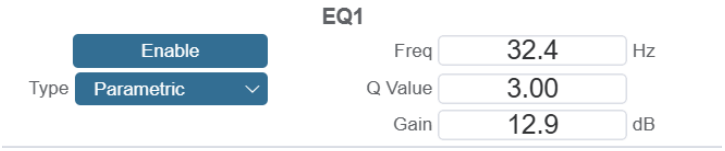
As shown in the above figure, the EQ curve display switch for each channel is at the top. Once turned on, the corresponding channel curve will be drawn in the EQ curve chart.

③EQ control

As shown in the above figure, below the curve chart, all the EQ controls and their detailed values are displayed, which can be used to locate the positions of the controls and adjust the parameter values of each EQ in combination with the EQ parameters below.



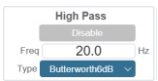
④EQ parameter adjustment items



As shown in the above figure, below the EQ control are the EQ parameter adjustment items, which can precisely control parameters such as the switch, type, frequency, Q value, and gain of the EQ.

⑤High and low pass filter

The high-pass or low-pass section shown in the right figure can control the switch, the frequency and type of high and low-pass.





⑥EQ Archive button

When you click the EQ preset button in the right picture, import and export will pop up.



Input delay device



As shown in the above figure, the delay control contents of all input channels are listed. Dragging the slider on the right can visually adjust the delay values of the corresponding channels.

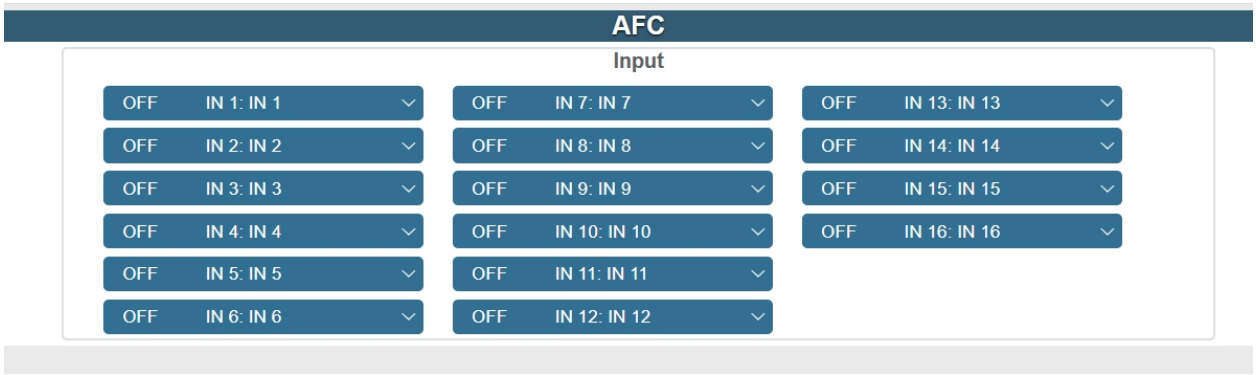


Enter the corresponding value in the numerical box. The switch for the delay function is on the right side. Gray indicates off and blue indicates on.

The reset button on the right can directly reset the default value of the channel delay. Meters and feet are both conversion unit values for milliseconds.

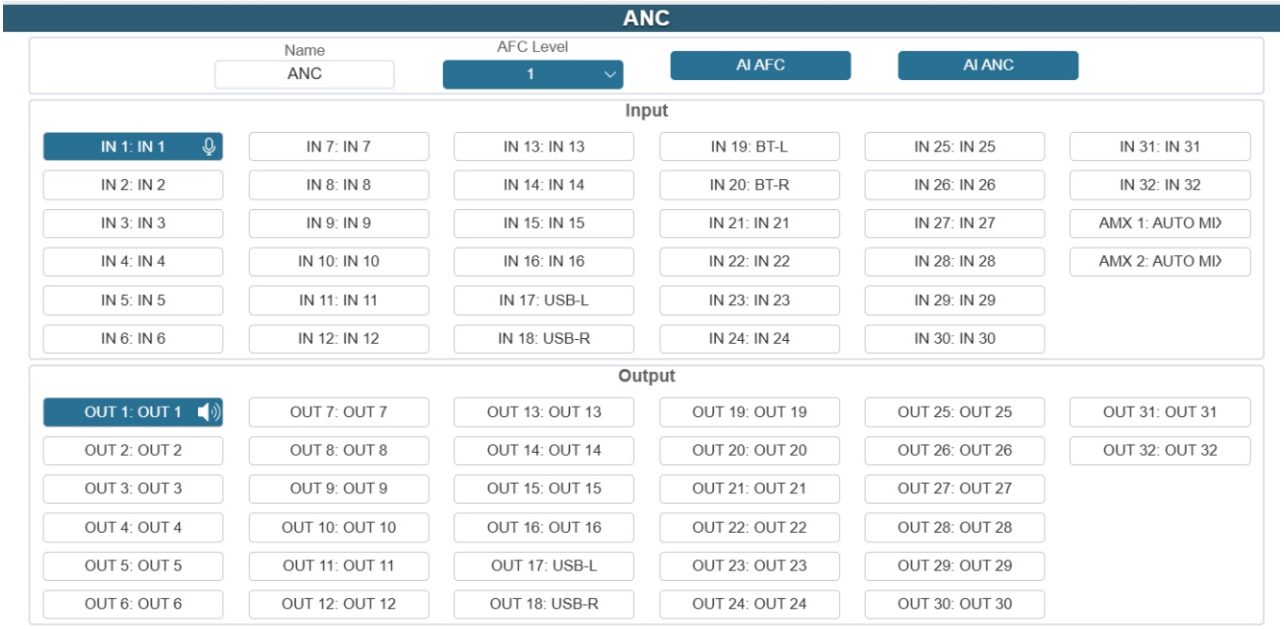
◆ Algorithm

AFC



AFC algorithm: There are two levels of traditional feedback suppression, with frequency shifts of 3Hz and 6Hz.

ANC (AI Noise Cancellation)



As shown in the above figure, after selecting the corresponding input and output channels, AI AFC and AI ANC need to be turned on to achieve the corresponding AI noise reduction effect. Blue indicates the on effect, and gray indicates the off function. Meanwhile, the ANC frequency shift effect has three Settings. In the right figure, 0 is to turn off the AFC effect, 1 is a 3Hz frequency shift, and 2 is a 6Hz frequency shift.

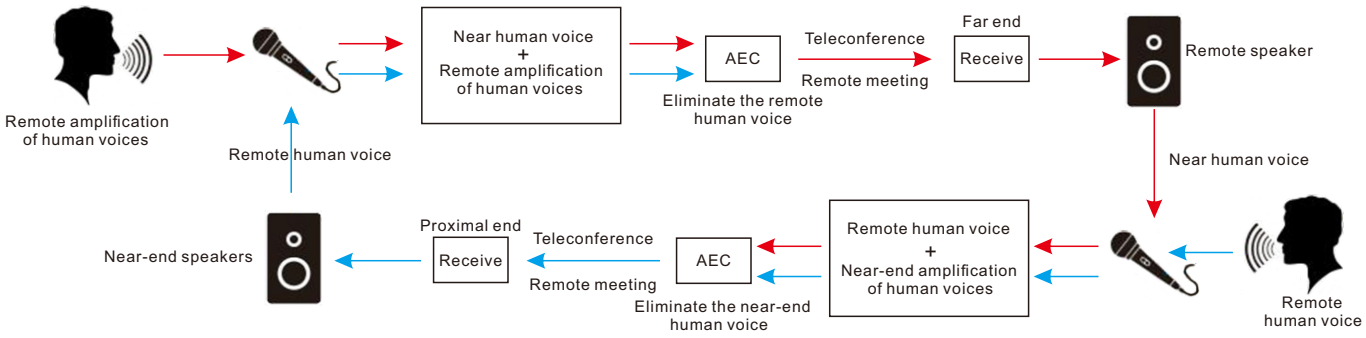


The input channel can be selected as 1-32 microphone or AES67 transmission signal input. Both can be selected as AUTO AMX (automatic mixing) signal input to the local ANC. The output enters the corresponding channel for ANC processing output.

AI AFC: The adaptive acoustic feedback Suppression (AFC) algorithm constructed based on the Kalman optimal estimation theory can track and counteract acoustic loops in real time, providing higher available gain while completely eliminating feedback. Its convergence speed and stability are significantly better than those of traditional NLMS schemes.

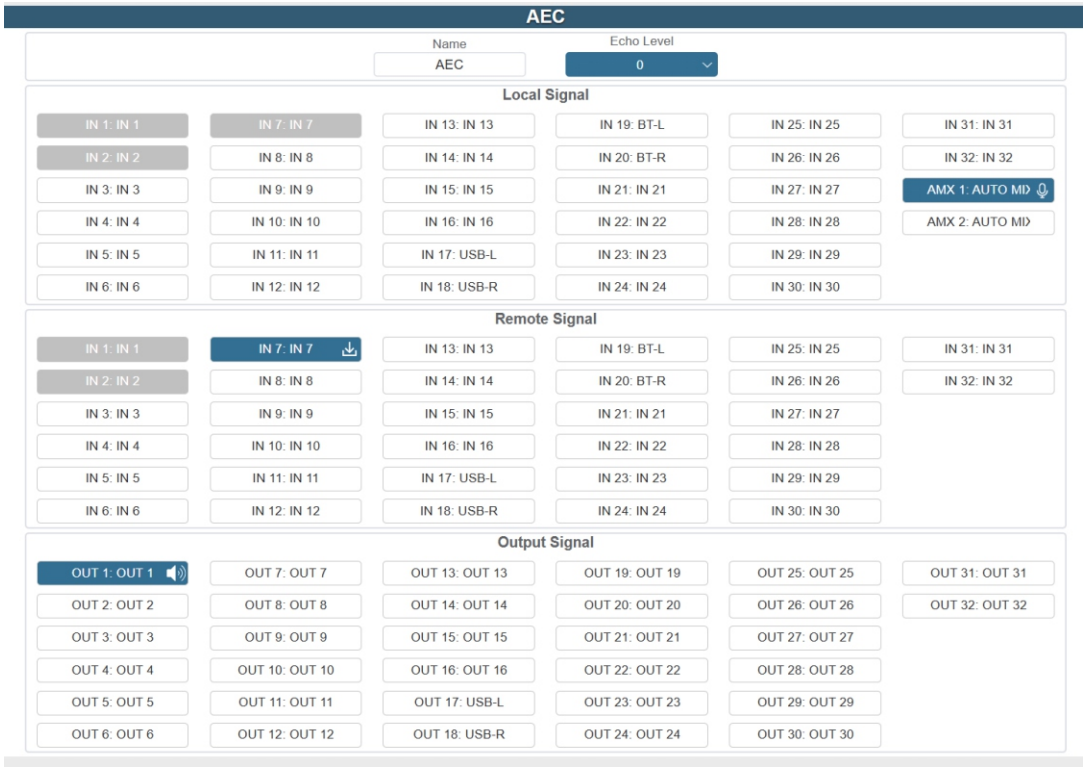
AI ANC: Equipped with an intelligent noise reduction engine based on deep neural networks (RNN), it is different from the traditional fixed-threshold spectrum subtraction method. It can distinguish human voices from environmental noise in real time. Even in complex non-steady-state noise scenarios such as blowing, tapping, knocking, and scratching sounds during meetings, it still maintains clear and natural speech.

AEC (Echo Cancellation)



AEC schematic diagram

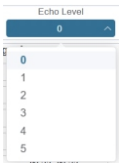
In the above picture, The near-end human voice is transmitted to the far end. Due to spatial reflection, the echo is picked up again by the local microphone and mixed together with the far-end audio signal.At this point, through AEC (Echo Cancellation) processing, the amplified human voice at the near end can be cancelled out, allowing only the sound from the far end to be heard at the near end.



AEC is a core functional module specifically designed for conference audio processing scenarios. It eliminates in real time the acoustic echoes formed when the remote sound signal played by the speaker is picked up by the local microphone for the second time through an adaptive filtering algorithm, while suppressing constant environmental noise to ensure the clarity of two-way voice in full-duplex calls, making remote meeting communication experience closer to face-to-face communication.

As shown in the above figure, AEC: gray indicates the off state, and blue indicates the on state.

AUTO AMX(Auto Mix) : When enabled, the auto mix signal is input to the AEC local signal.



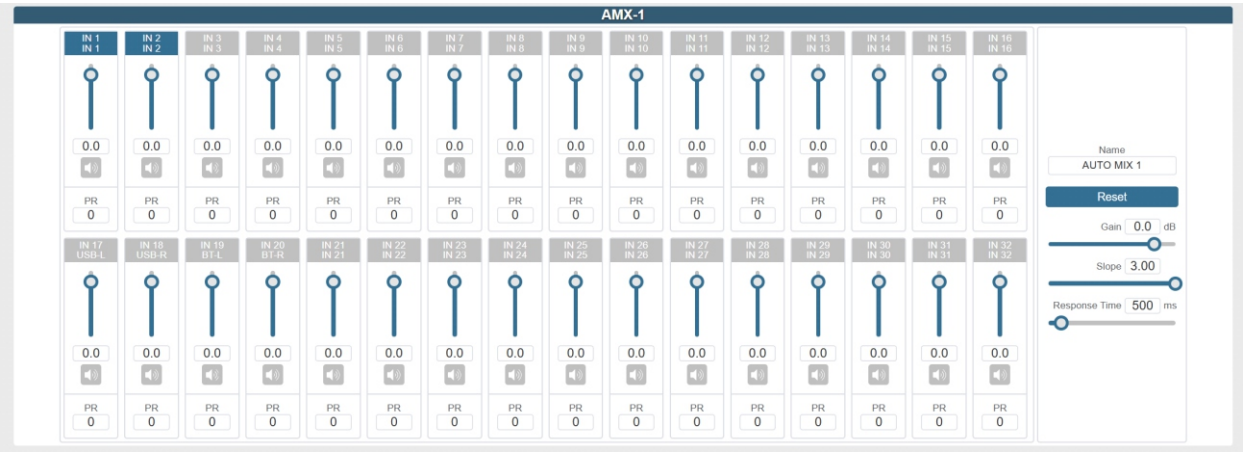
Echo level: 0 to 5,0 is off and 5 is the highest. It is used to configure the intensity of environmental noise suppression to reduce the interference of local constant noise (such as air conditioning sound, equipment background noise) on remote calls. It is recommended to select the lowest effective setting according to the on-site environmental noise level to avoid local voice distortion due to excessive processing.

Local signal: Multiple local input channels (IN1-IN32, USB-L/R, BT-L/R, AMX 1-2, etc.), and the local microphone/pickup device channel that needs to enable AEC processing can be selected.

Remote signal: The remote input channel corresponding to the local signal area (IN1-IN32, USB-L/R, BT-L/R, etc.) is used to select the remote signal channel as the echo reference.

Output: It includes multiple output channels (OUT1-OUT32, USB-L/R, etc.), and the signal output channel processed by AEC can be selected to send pure local voice signals to remote devices. The selected channel (such as the highlighted OUT1 in the interface) will be marked in blue and accompanied by a speaker icon, visually showing the currently active output channel.

AMX(Automatic Mixing)



As shown in the above figure, the core function of automatic mixing is to automatically control the gain of each channel in multi-microphone scenarios (such as meetings and speeches), automatically increasing the volume of the microphone that is speaking and decreasing the volume of the microphone that is not speaking, thereby reducing background noise, lowering the risk of feedback, and ensuring the stability of the overall output volume.

1.Control the manual gain of the corresponding channel. In the figure, IN1 is set to -9.4 dB, and the rest are set to 0dB. In the automatic mixing mode, the fader determines the base volume of this channel, and the automatic mixing will dynamically adjust it based on this.

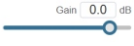
2. The channel mute switch in the right picture can be clicked to quickly turn off the output of this channel.



3. In the right figure, the channel priority setting is shown. The higher the value, the more priority will be given to seizing the gain when the speaker is speaking. This is often used for the speaker's microphone to avoid being interrupted by other speakers.



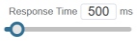
4. The total output gain of the automatic mixing shown in the right figure adjusts the overall volume after mixing all channels. A value of 0dB indicates no additional amplification/attenuation.



5. The right figure controls the "distribution intensity" of automatic mixing, similar to the expansion ratio: the larger the value, the more gain the speaking microphone gets, and the more severe the attenuation of the non-speaking microphones. It is suitable for scenarios with high noise and a large number of microphones. Value 1: Gain is evenly distributed across all channels, with no automatic attenuation effect. Common Settings: 2 to 3. In the figure, 3.00 is a relatively strong attenuation effect, which can effectively reduce background noise.

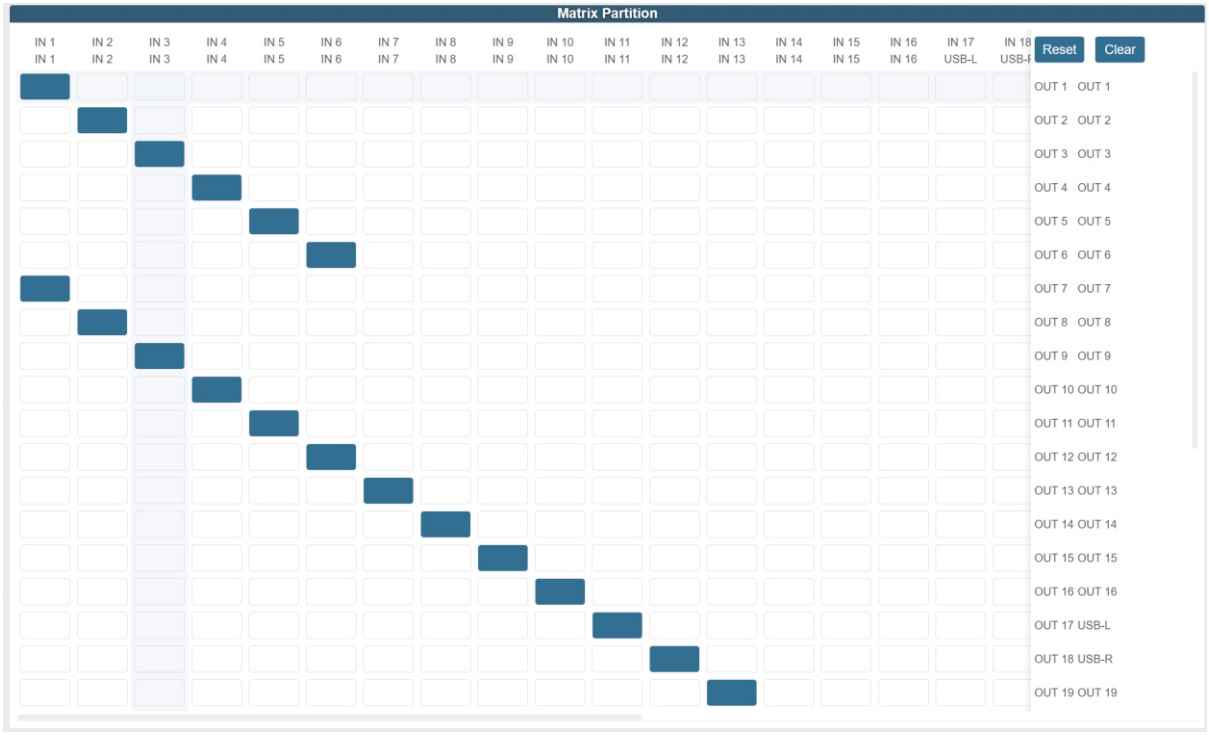


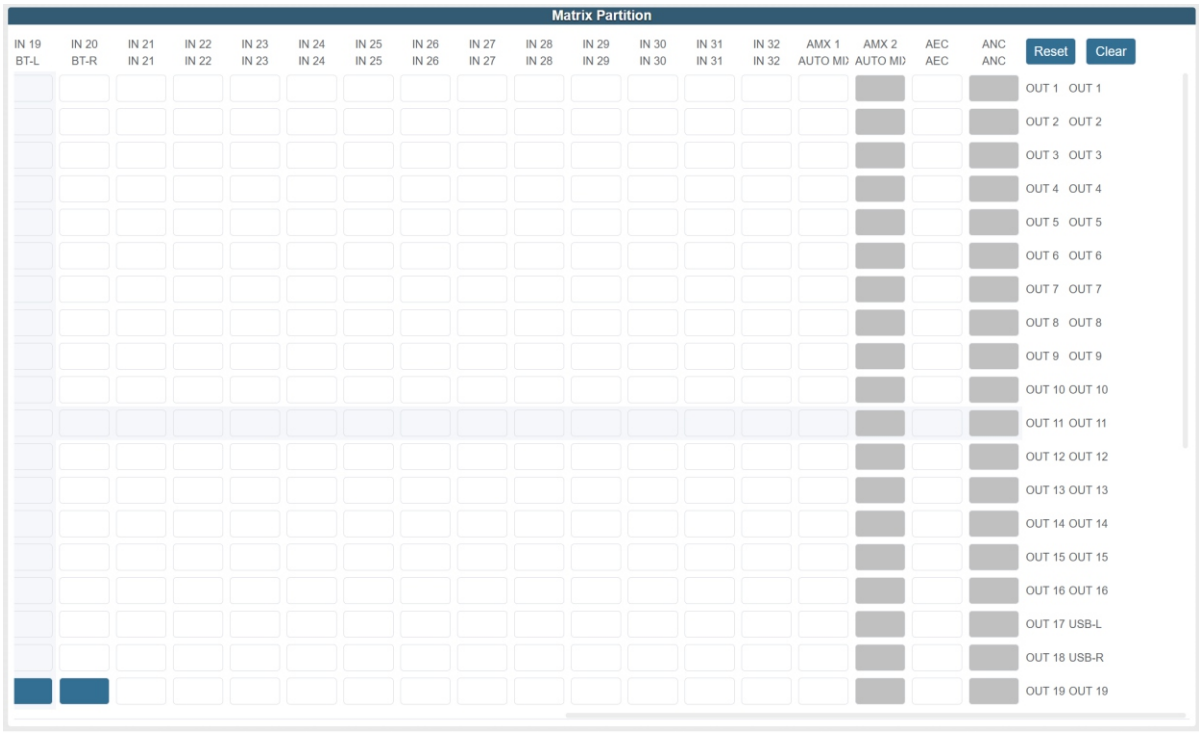
6. The time taken by the system from "detecting speech" to "adjusting gain" determines the response speed of the automatic mixing: the smaller the value, the faster the response, and the volume rises rapidly when someone speaks, which is suitable for scenarios with fast-paced speech.The larger the value, the gentler the response, avoiding volume fluctuations caused by frequent switching. It is suitable for speeches and multi-person meetings. The 500ms setting in the picture is moderately slow, balancing naturalness and stability.



7. Reset: It is a one-click function to restore the faders, mute, and priority of all channels to their default states, quickly resetting the automatic mixing configuration.
8. Name (AUTO MIX 1) : The name of the current auto mix scene can be modified to facilitate saving and invoking presets of different scenes (such as "Conference Room Mode", "Lecture Hall Mode").

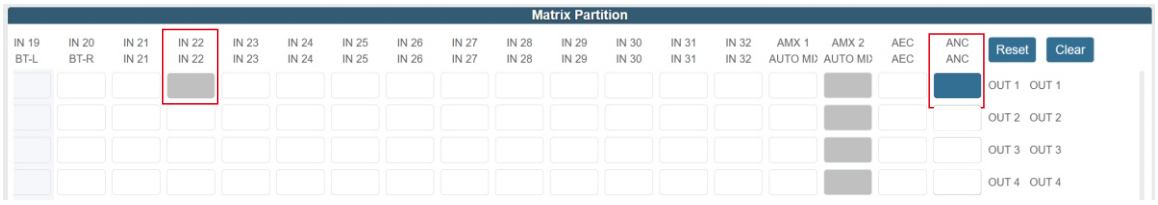
◆ Matrix mixing





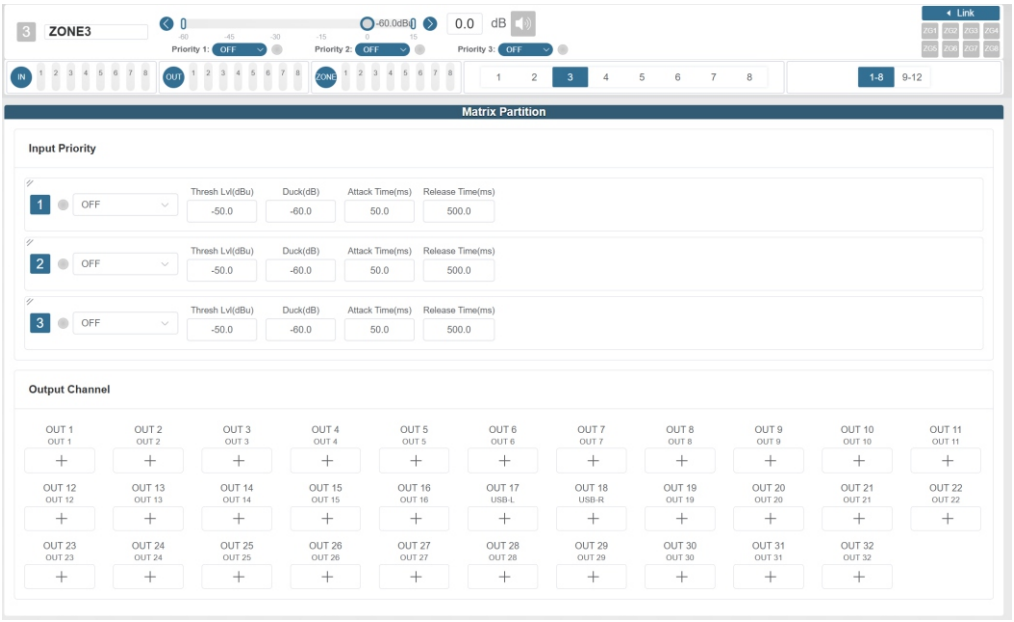
The right side of the above picture corresponds to the output channel, and the upper side corresponds to the input channel. The numerical boxes with values are the input and output channel mixing keys. When the mixing key is blue (clicking the numerical box can switch the state), the input channel and output channel signals perform the mixing function.

The right part of the above picture includes the Reset button and the Clear button. Clicking the reset button will reset the matrix mixing function to the initial one-to-one state. Clicking the Clear button will completely clear the matrix mixing function, and there will be no corresponding relationship between the input and output of the device.



Note: As shown in the above figure, when the algorithm is enabled, the corresponding channel cannot be manually defined and will display a gray tone prohibiting subscription.

Zoning



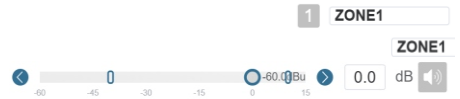
Matrix partitioning (Zone partitioning) is a core module in digital audio processors for multi-input priority management and multi-output channel partition routing. It is mainly used in scenarios such as broadcasting, public broadcasting, meeting rooms, and background music in shopping malls.

Input priority section: Implement audio Ducking. When high-priority signals (such as broadcast microphones) are activated, automatically lower low-priority signals like background music to ensure clear voice.

Output channel section: Implement the zonal routing of signals, allocate the processed audio to different output zones (Zones), and each Zone can independently control the audio source and parameters.

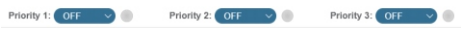
The first partition in the right picture

Click on the right picture to modify the name.

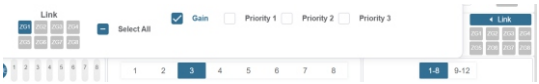


In the right picture, you can adjust the total volume by swiping. You can also directly mute or unmute the device. The text box can also be used to input the volume.

Priority level: The smaller the number, the higher the priority (such as Channel 1; Channel 2; Channel 3). When multiple priority signals are triggered simultaneously, only the channel with the highest priority takes effect, and the ducking action for lower-priority signals will be overridden.

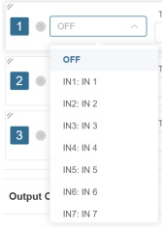


By clicking on the group Settings for joint debugging, you can also select the parameters that need to be debugged for different groups.



There are a total of 12 channels in the zone. Clicking on them allows you to switch to other channels.

Detailed explanation of the input priority module



1. Priority switch: Controls whether the ducking function of this priority channel is enabled. When it is OFF by default, the priority of this channel is not used. When other input channels (IN1-IN32, USB-L/R, BT-L/R, MIX1:AUTO MIX1, MIX2: AUTO MIX2, AEC:AECANC:ANC) are selected, enable the priority control of this channel.

2. Priority: The smaller the number, the higher the priority (such as Channel 1 > Channel 2 > Channel 3). When multiple priority signals are triggered simultaneously, only the channel with the highest priority takes effect, and the evasion of the lower-priority ones will be overridden.

3. Start Control Level (dBu) : The trigger threshold for the evasion function. When the input signal level of this channel is higher than this value, the system determines it as a "valid signal" and triggers the subsequent evasion action. The lower the value, the easier it is to trigger (for example, the background noise of the microphone may also trigger it). If the value is too high, the situation of "not dodging when the signal comes" may occur.

4.Duck(dB) : ducking depth, that is, the extent to which low-priority signals are depressed. The unit is dB. The smaller the value, the greater the reduction. For instance, -60dB almost muts the background sound and is suitable for emergency broadcasts. -15dB is suitable for daily paging, retaining some background sounds.
5. Startup Time (ms) : The response time to trigger the ducking. The unit is milliseconds. Set the speed at which the background sound begins to be depressed after the signal reaches the threshold. The smaller the value, the faster the response, avoiding the problem of "the background sound at the beginning of the voice has not dropped yet". However, a value that is too small may cause the background sound to fluctuate frequently.
6. Release time (ms) : The recovery time when the ducking is lifted. The unit is milliseconds, which is the speed at which the background sound returns to its normal volume from a low state after the high-priority signal disappears. The larger the value, the smoother the recovery, avoiding the jarring feeling of "the background sound suddenly bursts out right after the voice ends".

Detailed Explanation of the Output Channel Module

The interface on the right shows a total of 32 output channels from OUT 1 to OUT 32. The button for each channel is used to configure the audio source routing

1.Core function: Allocate the audio signal that has undergone input priority processing to different output zones (Zones).
For example:
OUT1: Assigned to the speakers in the first-floor lobby
OUT2: Assigned to the speakers in the second-floor corridor
OUT3: Assigned to the meeting room speakers

2. Extended Control: After clicking the + button, you can usually further set the output
Volume, mute, and equalizer (EQ) select whether the area responds to a certain level of priority signal (for example, all areas need to respond for emergency broadcasting, and only the hall responds for daily paging). Zone switch (turn off the audio output of a certain area separately).

◆ DSP output

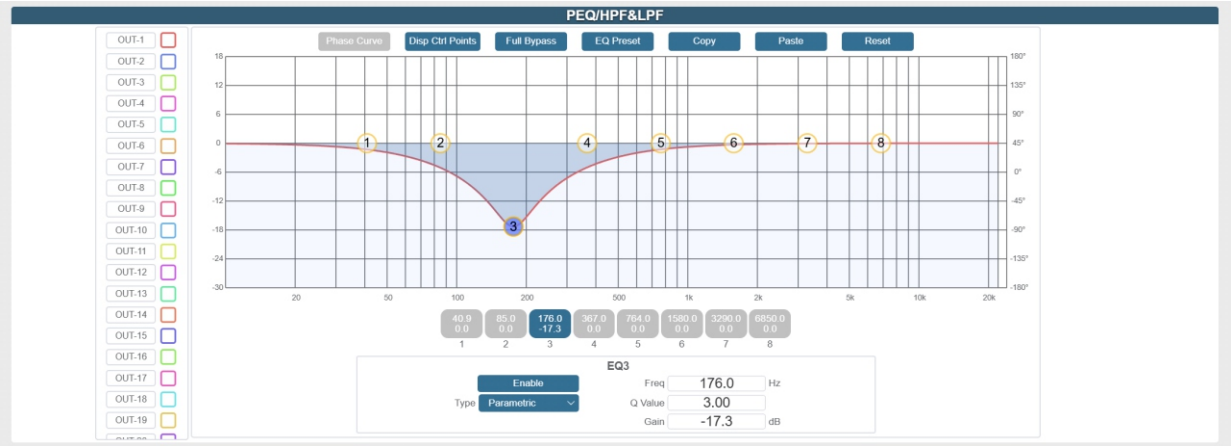
Delay device



As shown in the above figure, the delay control contents of all output channels are listed.
Dragging the left slider can visually adjust the delay values of the corresponding channels.
Enter the corresponding value in the numerical box. The switch for the delay function is on the right side.

Gray indicates off and blue indicates on.
The reset button on the far right can directly reset the default value of the channel delay. Meters and feet are both conversion unit values for milliseconds. 

Output PEQ



①The functions of the top buttons are as follows:

Phase curve: Displays the phase curve of the current channel.

Display control points: Show or hide all equalization control points.

Full bypass: Simultaneously turn on or off all equalizers (EQ) of the current channel

EQ Archive: Save the current equalizer setting parameters to your computer and retrieve them to overwrite the existing equalizer parameters.

Copy: Copy the current equalizer parameter value and paste it into other input channels.

Paste: When used in combination with the copy button, the equalizer parameter values copied by the copy function can be pasted into the current channel.

Reset: Reset the equalizer parameters to the default parameter values.

② Multi-channel EQ curve display



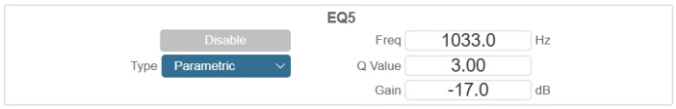
As shown in the above figure, the EQ curve display switch for each channel is at the top. Once turned on, the corresponding channel curve will be drawn in the EQ curve chart.

③EQ Control

As shown in the above figure, below the curve chart, all EQ controls and their detailed values are displayed, which can be used to locate the positions of the controls. Combined with the EQ parameter adjustment items below, the parameter values of each EQ can be precisely adjusted.



④EQ parameter adjustment items



As shown in the above figure, below the EQ control are the EQ parameter adjustment items, which can precisely control parameters such as the switch, type, frequency, Q value, and gain of the EQ.

⑤EQ Save button

As shown in the right figure, clicking the EQ preset button will pop up the import and export options.



◆ Speaker parameters

Speaker parameter gain



Click on the right picture to modify the channel name.

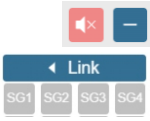


When you need to modify the gain, drag the scroll bar or enter the text box to make the changes.

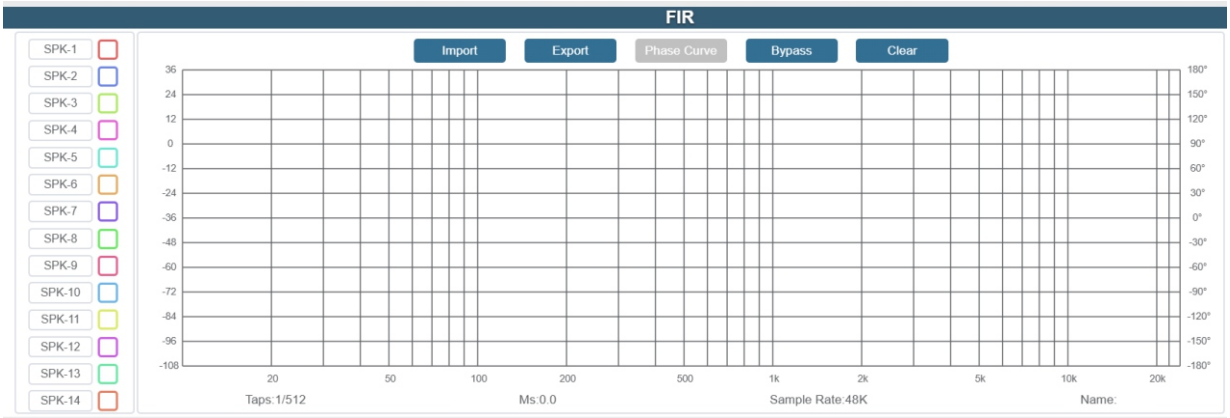


At the same time, phase and mute can also be set.

The right picture shows the joint adjustment of speaker parameters, and the usage method is consistent with the joint adjustment of DSP input and DSP output.



Speaker parameter FIR



Import: Import the curve with the set 48K sampling rate into the device

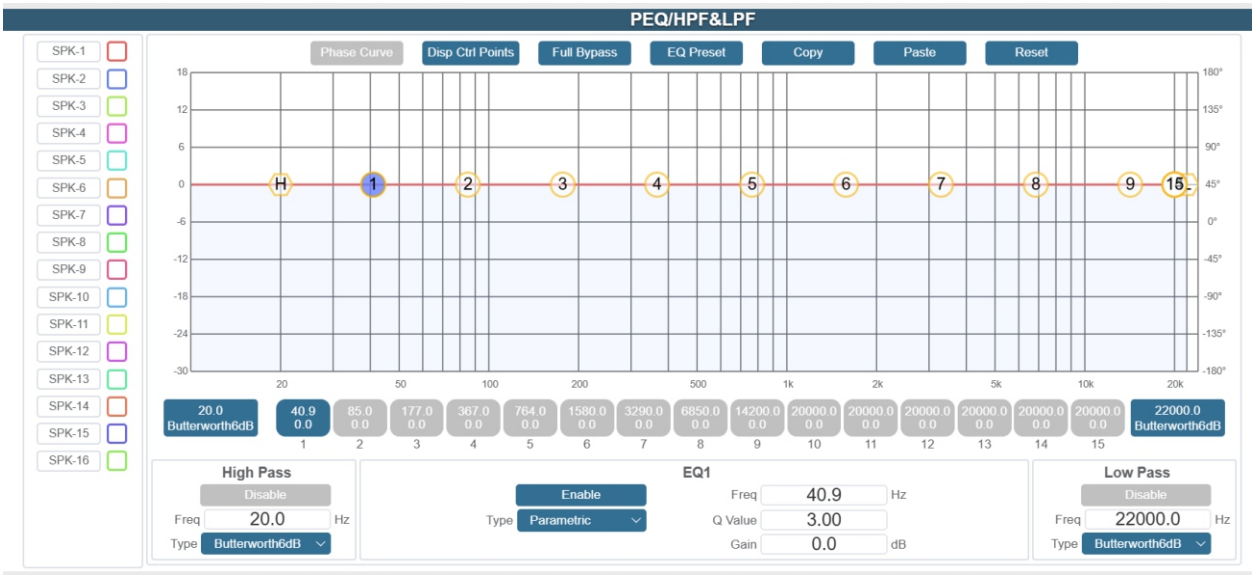
Export: Export and save the FIR curve

Phase curve: It can display the phase curve of FIR

Bypass: By default, it is in the bypass state. Only by turning off the bypass can the curve effect be achieved

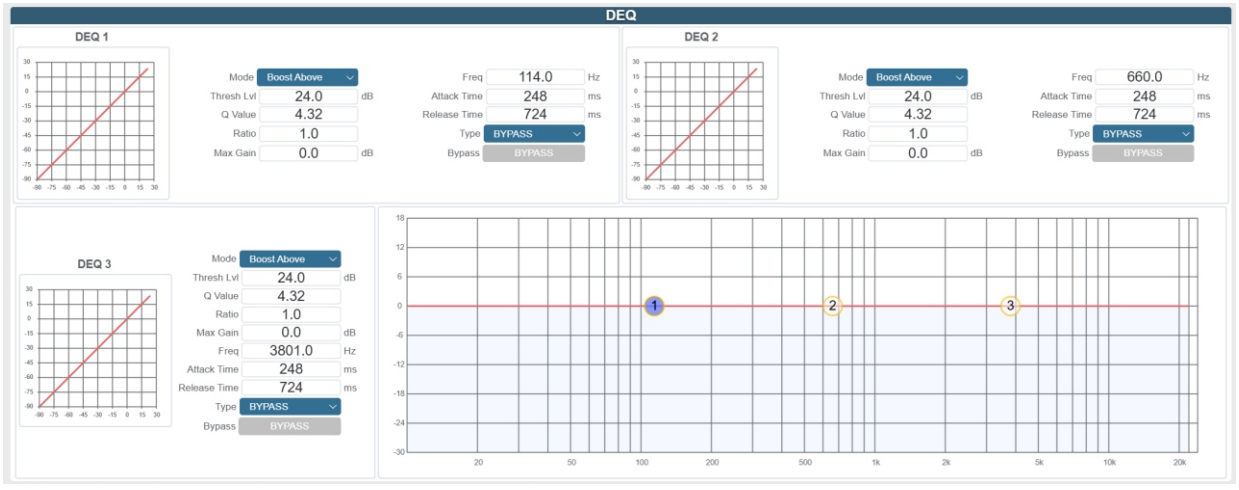
Clear: Clear the FIR curve

Speaker parameters: PEQ/ high and low pass



The functions and operation methods are the same as those for inputting PEQ.

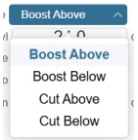
DEQ (Dynamic Equalizer



1.Click on the right picture. Channels 1 to 16 have a speaker parameter library where you can freely switch channels.



2. It can support four modes: greater than threshold boost, less than threshold boost, greater than threshold attenuation, and less than threshold attenuation.



3. Click on the right picture to turn off the DEQ function, remove the bypass, and enable the DEQ function.



4. Starting control level (threshold) : -90 to 24dBu. This function takes effect when the signal of this frequency reaches the starting control level value.

Q value: 0.27 to 15, the range of Q values at varying frequency points;

Ratio: 1 to 100, the adjustment ratio for controlling dynamic changes. The larger the ratio, the more obvious the adjustment.

Maximum effect: 0 to 12, the limit value that can be increased or decreased according to the selected mode;

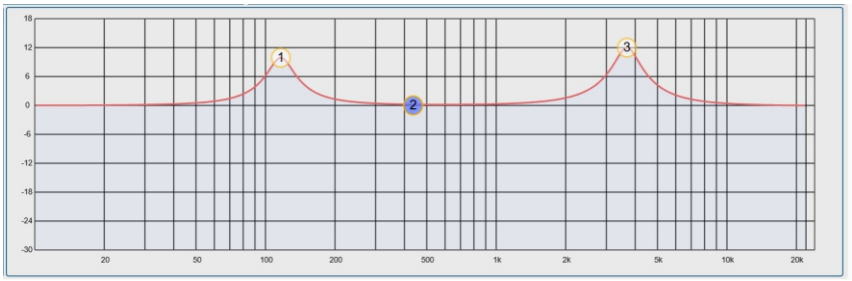
Start-up time: 1 to 2895, with an increased or decreased delay before taking effect

Frequency: Set the center frequency point that takes effect

Release time: The duration of the effect

Type: You can choose PEQ, LSLV, HSLV as the effect type, and BYPASS is to turn off the effect

The above are all the parameters of DEQ. They can be set by modifying the number box in the right figure. The set parameters can be displayed in the curve.

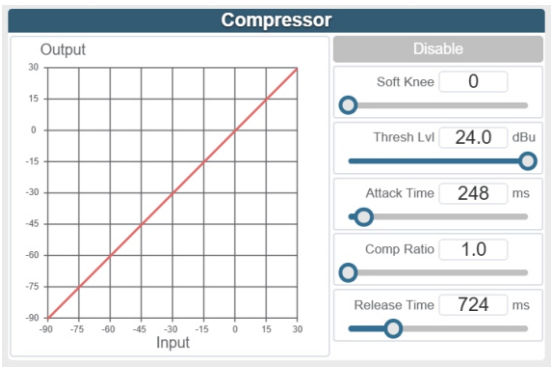


Speaker parameter delay



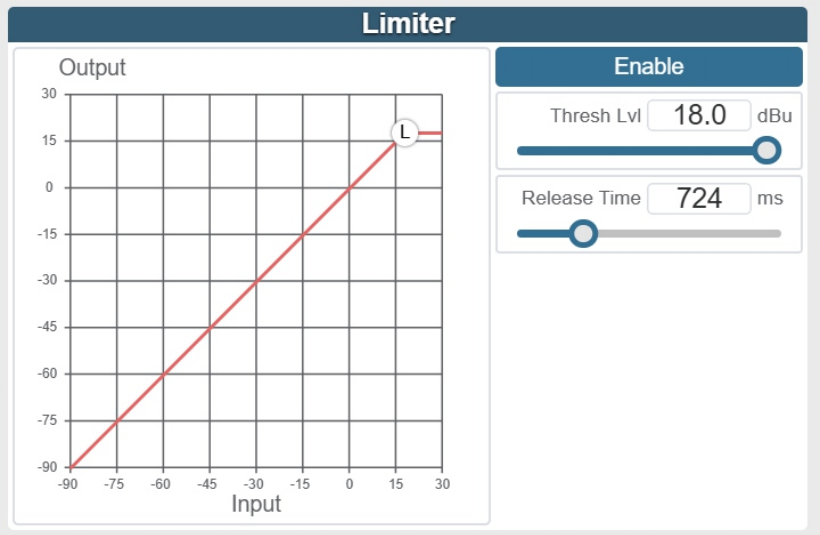
The function and operation mode are the same as the input delay.

Speaker parameter compressor



The functions and operation methods are the same as those of the input compressor.

Speaker parameter limiter

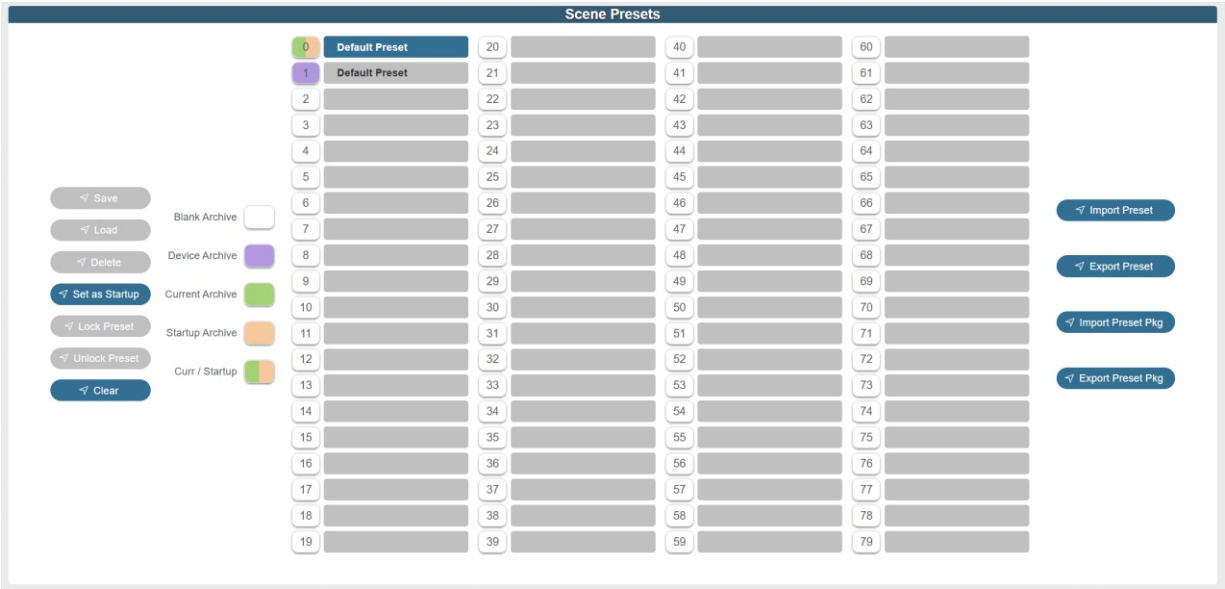


Starting control level: When the signal level is lower than the starting control level, the limiter does not process the signal, and the signal can pass normally. When the signal level reaches or exceeds the starting control level, the limiter will change the amplitude of the signal according to its internal algorithm.

Release time: The time required for the gain to recover to a value below the threshold after the input signal drops below the threshold.

◆ Preset

Scene preset



As shown in the above figure, the left side of the preset interface is the preset position, among which "0 Default preset" for the current preset slot"1 (Default)" is the default preset of the device. It can only be called and cannot be deleted or overwritten. Other preset bits can be freely saved, called, deleted and other operations.

The function buttons on the right side of the preset interface are as follows:

Save: Save the existing device parameters to the corresponding selected preset.

Call: Retrieve the selected preset to the parameters of the current device.

Delete: Delete the selected preset parameters.

Clear: Clear all preset parameter records that are not system presets.

Set as startup preset: Set the selected preset position as the preset that the device will automatically call to work the next time it is powered on.

Preset lock: Set a password to lock the saved preset.

Preset unlock: Enter the set password to unlock the preset.

Import presets: Import the parameter files of individual devices in the computer system to directly overwrite the existing parameter data.

Export presets: Save the parameters of the current device to the computer system to generate a single device parameter preset file.

Import preset packages: Import multiple preset parameter packages from the computer system.

Export preset packages: Export all the parameters of the preset bits in the device preset to the computer system to generate multiple preset parameter package files.

Speaker presets



Restore to default: You need to select a channel to restore the default Settings.

Restore all to default: Restore all channels to default Settings.

Create presets: Select channels to create, save data and protect data.

Import presets: Import the preset channels that have been set.

Import: Import multiple preset files into the preset list simultaneously.

Export: Export the saved speaker presets. Multiple exports can be selected.

Editor: Click on the preset you need to edit. Edit the required functions and modify the protection status. Note: After locking, editing requires entering the password used at the time of locking.

Delete: Delete existing presets.

Setting suggestions

For regular performances, it is recommended to select the 26 to 36dB range to avoid system noise, whistling or distortion caused by excessive gain.
Low-sensitivity speakers (such as line arrays and subwoofers) can be appropriately set to a higher level (38 to 44dB), but it is necessary to cooperate with the processor to limit the output level. After the debugging is completed, it is recommended to retain only 1-2 commonly used gears to avoid on-site misoperation.

◆ Settings

Equipment information



The above picture allows you to modify the device name, view the program version and other information.

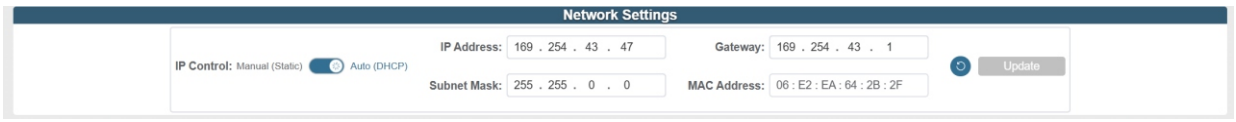
In the backup section, you can select the information you need to back up. After checking the box, click Export.



The right picture shows that importing a backup is to import the previously exported backup file into the device. An update is a program upgrade. After importing the program and loading the upload progress to 100%, wait for another 5 seconds, and the device will be successfully upgraded and automatically restart.

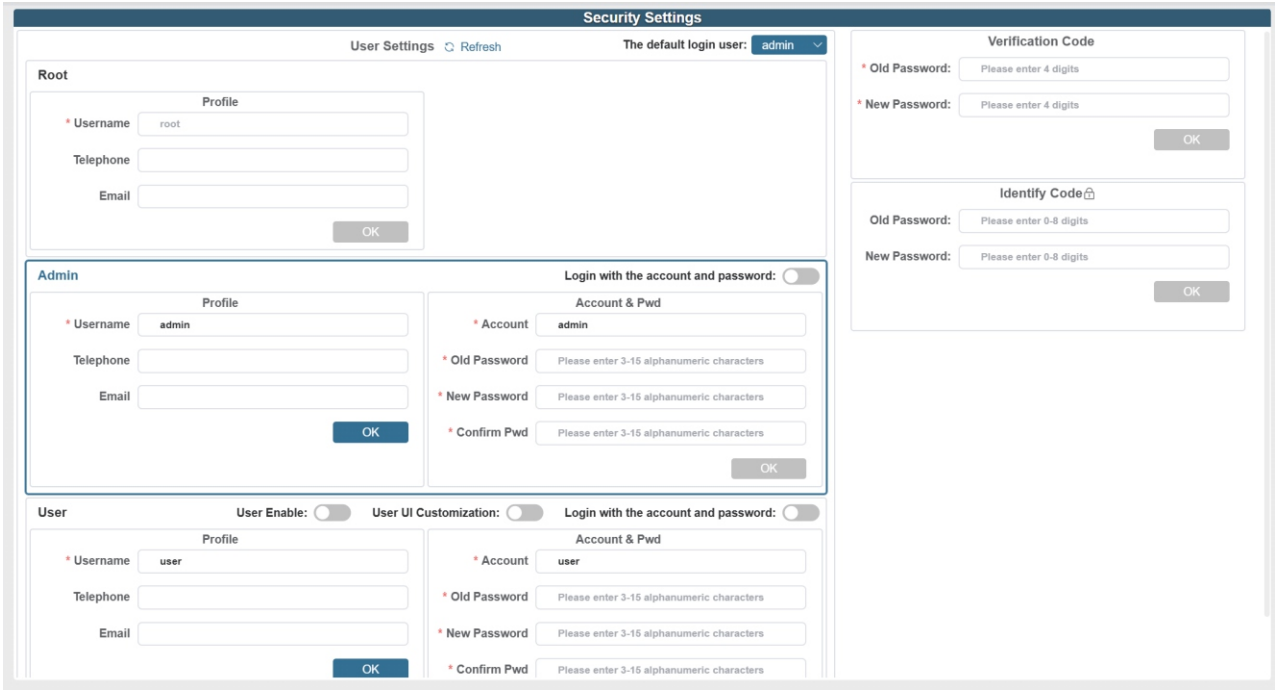


Network information

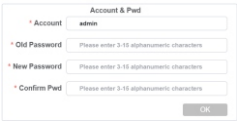


The above picture can set the device to DHCP and static IP. In DHCP mode, it needs to be connected to a router that can allocate IP addresses. If it is not connected, the system will automatically allocate a local IP address of 169.254.X.X. After modification, you need to click "Update" for it to take effect.

Security Settings



The above picture shows the boundary for setting the verification code at level three. At level three, root is the factory management account (not open to users), and admin can manage users (restrict the use of functions). The default password for the admin/user account is 8888.



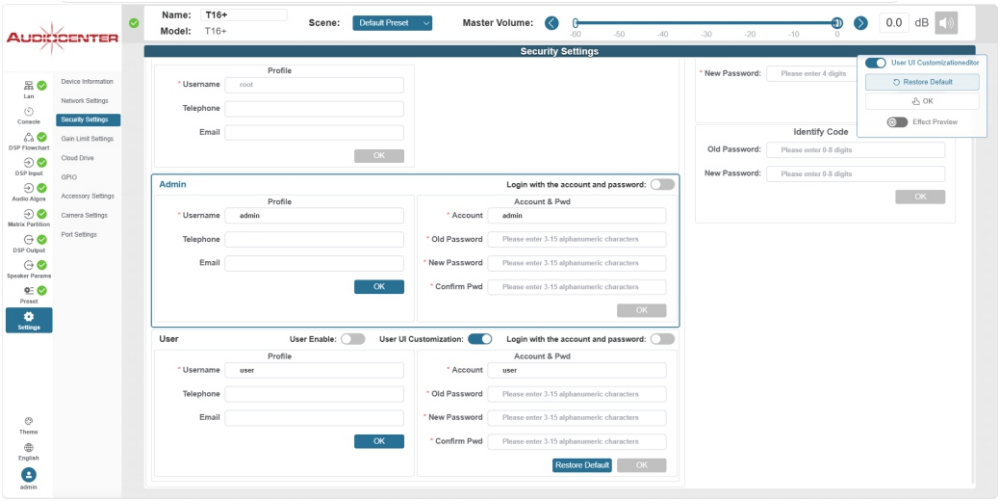
If you need to change your account password, you can do so here and save it.



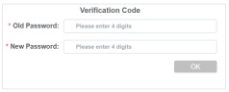
You can fill in your personal information here for easy contact later.



The interface customization on the right is the interface setting for low-level users by advanced users.



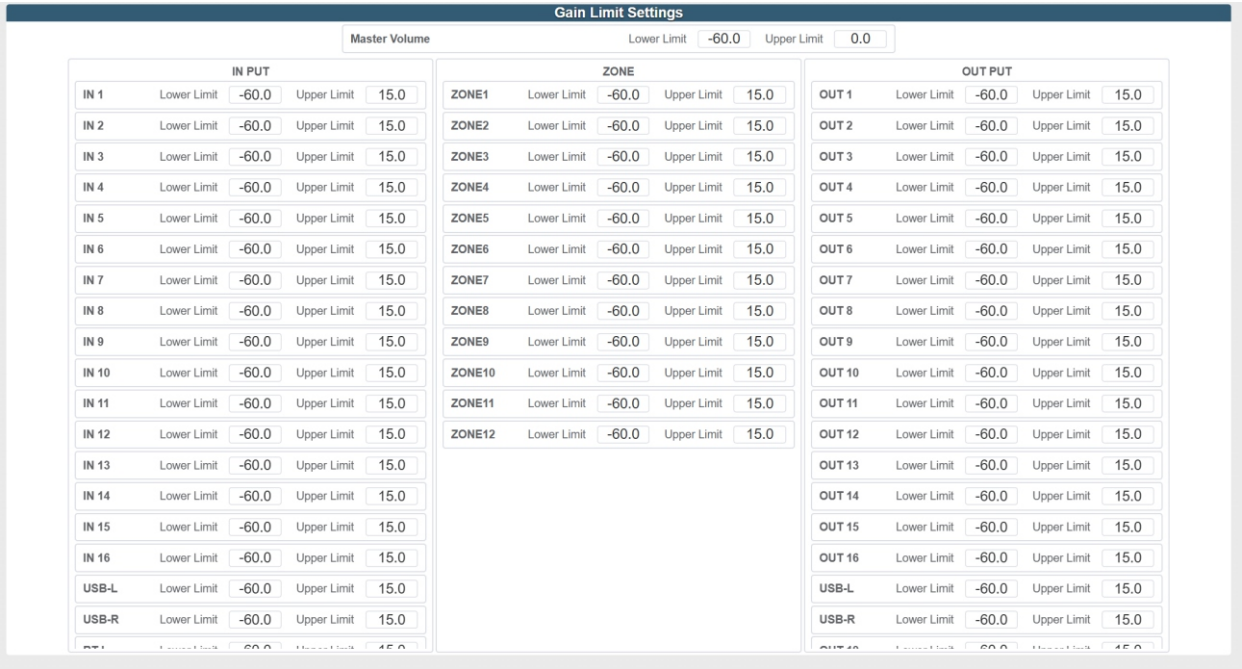
There is a green icon in the upper right corner of the editable interface. Unchecking it means hiding. After editing, click "Confirm" in the upper right corner of the web page to save.



The default password for the verification code in the right picture is 8888. When operating on the WEB end, many operations require this password.

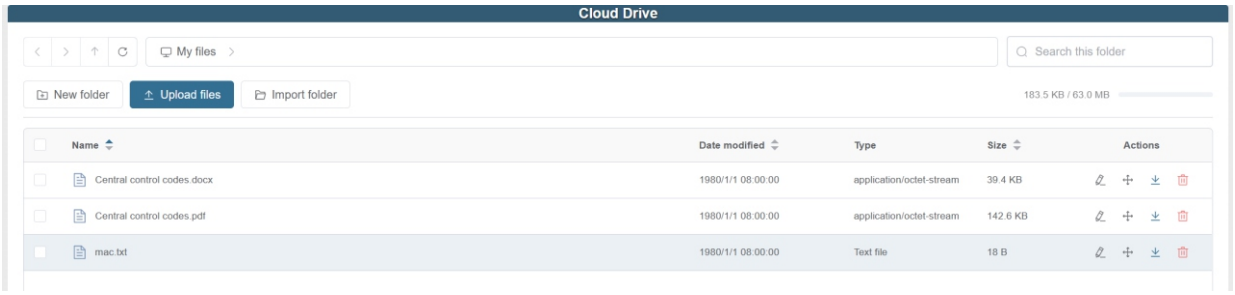
Situation 2: The target machine has already set an identification code, and the number of the identification code does not match that in the backup file: Direct import will fail. You must first modify the identification codes on both sides to unify the numbers before the import can be completed.

Security Settings



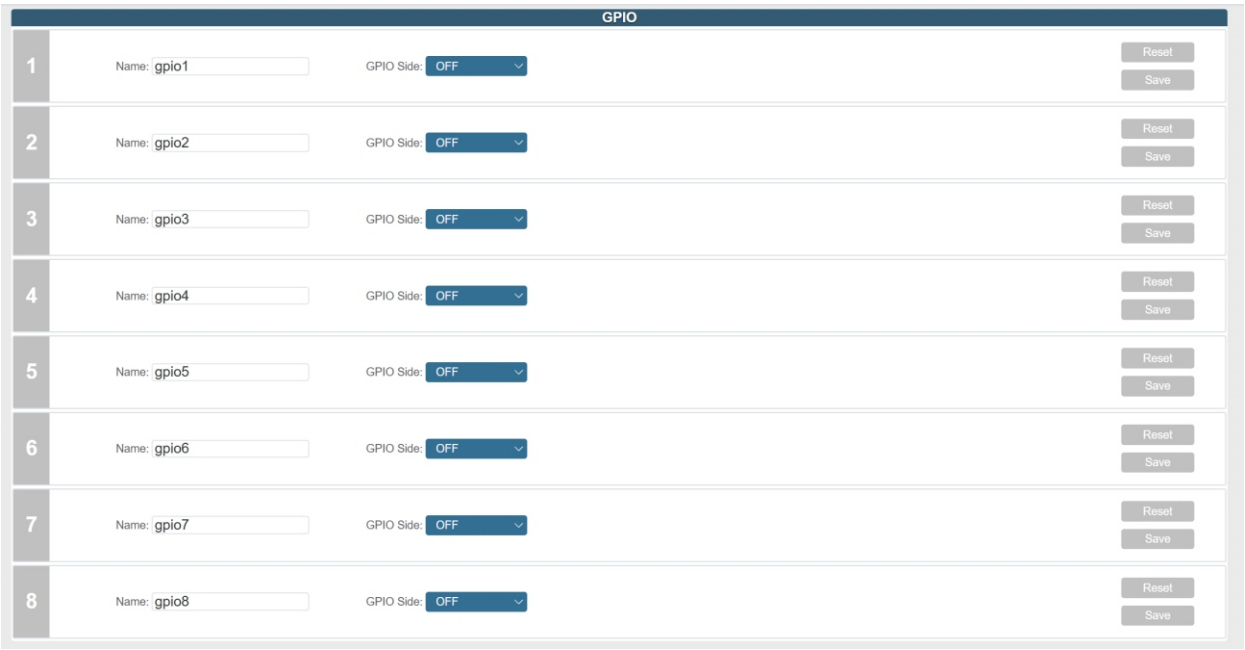
The above figure shows the upper and lower limits of the gain, including the upper and lower limits of the input and output channels and the total volume. The required gain range can be changed. The minimum gain range of the total volume is -60dB and the maximum is 0dB. The minimum gain range of the input and output channels is -60dB and the maximum is 15dB.

Cloud disk



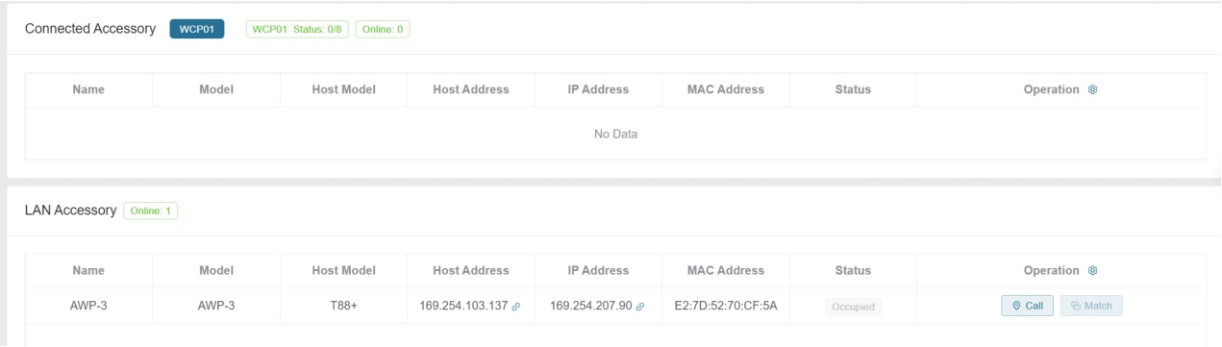
The memory size of the cloud disk is 32M. The virtual USB flash drive with a USB interface is connected to the USB flash drive on the WEB end.

GPIO



The device has eight GPIO interfaces. Here, you can set the names of the GPIO and the functions of the interfaces. After setting, you need to click "Save".

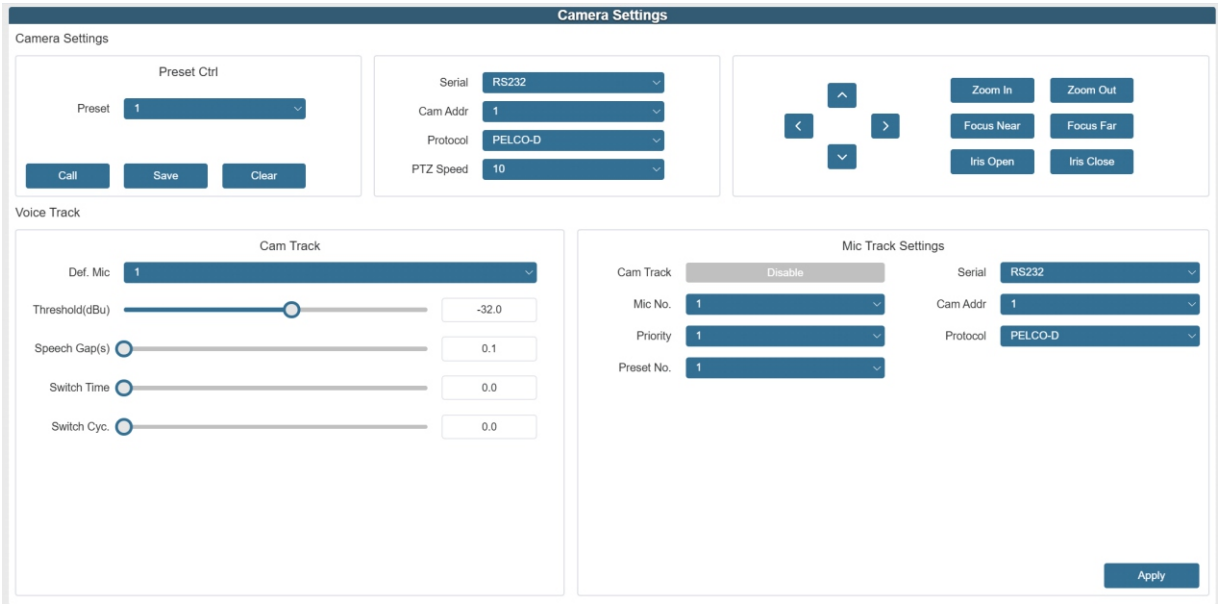
Accessory Settings



As shown in the above figure, the connection status of the AWP-3 wall panel is displayed when the device is connected.

As shown in the above figure, the connection status of the AWP-3 wall panel is displayed when the device is connected.

Camera Settings



The camera Settings is a camera debugging interface. Generally, the camera position is debugged before the tracking starts. Finally, the parameters of this part will be saved to the camera.

First, set up the serial port. There are two serial ports (232 and 485), corresponding to the rear backplane port connected to the pan-tilt. Secondly, there are the camera address and protocol type. Please refer to the actual address of the camera for the camera address. The protocol is related to the camera model. Finally, the preset point number is the identifier defined by the user for the camera. Then, by adjusting parameters such as up, down, left, right, focal length, and aperture, the position and Settings of the camera will be defined. Finally, click "Save" to save the parameters to the camera. "Clear" deletes the information of the current preset point, and "Call" is used to view the current situation.

The camera position saved by the pre-set point.

Note: One camera address may store multiple preset positions, while each preset position belongs to only one camera. Both the camera Settings and the microphone Settings have several parameters such as preset points, serial port numbers, camera addresses, and protocols. When applying them, the actual scenarios need to be taken into consideration.

1.Preset Point Control Module (Upper left)

Function: Save multiple fixed monitoring points for the PTZ camera and quickly call them up with one click.

Preset point dropdown box: Different numbered points from 1 to 32 can be selected, representing different monitoring positions stored by the camera.

Call: After selecting the preset point number and clicking, the pan-tilt will automatically rotate to the monitoring Angle corresponding to that point.

Save: First, manually move the camera to the desired monitoring position, select the preset point number, click Save, and the current picture Angle will be saved to this preset point.

Clear: After selecting the preset point number, click to delete the pan-tilt position information stored in that number.

2. PTZ Communication Serial Port Parameters (upper middle)

This is used to configure the 485/232 control communication parameters of the pan-tilt. When the camera pan-tilt cannot rotate, please check this part first:

Serial port number: The hardware serial port of the control pan-tilt, with RS232 as the default. Switch to RS485/RS232 based on the device wiring.

Camera address number: PTZ device address code. When there are multiple PTZ devices on the same bus, each address must be unique. The default is 1.

Protocol: The PTZ control protocols include PELCO-D, PELCO-P, VISCA, and custom protocols. If these protocols do not match, the PTZ will not act.

Pan-tilt rotational speed: The speed at which the pan-tilt rotates up, down, left and right. The larger the value, the faster the rotation. The range is generally 1 to 100, and the current default is 10.

3.Manual Control Area of Pan-tilt (Upper right)

For real-time manual control of the camera pan-tilt and lens, and for adjusting the position of the picture:

Direction control: Up, down, left and right arrow keys. Long press to continuously control the direction of the pan-tilt, release to stop.

Lens adjustment button:

Zoom in/Zoom out: Control the lens to zoom in and out of the picture.

Focus - Near/Focus - Far: Manual focus. Adjust to make the image clear when the picture is blurry.

Aperture - Large/Aperture - Small: Adjust the amount of light entering the lens. Reduce the aperture when the light in strong light appears white, and increase the aperture when the light in low light appears black.

4. Voice Tracking - Camera tracking Parameters (lower left)

Basic parameters of the microphone pickup linkage pan-tilt tracking function: Sound triggers pan-tilt rotation to follow the sound source:

Default microphone: Select the microphone number connected to this machine. Currently, the default is 1 microphone.

Tracking threshold (dBu) : The sound source volume trigger threshold (-32.0), the higher the value, the more sensitive it is. In a noisy environment, it is recommended to lower the sensitivity (increase the value, such as -20), and in a quiet environment, increase the sensitivity (decrease the value, such as -40).

Voice interval (s) : The duration of the delay for continuous tracking after a sound is detected. 0.1 seconds indicates that the tracking stops 0.1 seconds after the sound disappears.

Waiting time/waiting cycle: The tracking waiting time for multi-microphone switching takes effect when multiple pickup devices patrol the sound source. 0.0 indicates that the patrol switching is turned off.

5. Mic Track Settings

The communication and rules for the complete configuration of microphone-linked pan-tilt tracking are currently disabled.

Once enabled, the sound will automatically switch to the pan-tilt:

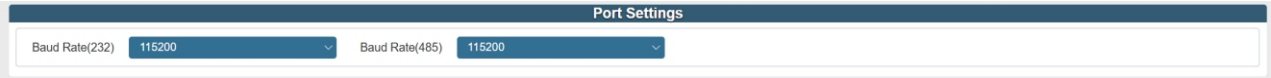
Camera tracking: Select "Enable/Disable" on the switch. Currently, it is in a gray state of disabled status, and blue indicates that the voice tracking function is enabled and effective.

Microphone number: Bind the serial number of the microphone to be linked. Default: 1.

Priority: When multiple microphones are used, the larger the number, the higher the priority, and it prioritizes following the sound source of the high-priority microphone.

The communication parameters on the right (consistent with the serial port parameters of the pan-tilt above for linkage) :
Serial port number: RS232/RS485, compatible with pan-tilt wiring;
Camera address number: Pan-tilt address code, which must be consistent with the pan-tilt address above.
Protocol: PELCO-D, PELCO-P, VISCA and custom, unified pan-tilt control protocol;
Preset point number: The preset point for the pan-tilt to automatically return to position when there is no sound source, with a default of 1.
6. Configure the save operation
After all the parameters have been modified, click the Apply button at the lower right corner of the page. Only then will all the Settings be written to the device and take effect. If you switch pages directly without clicking "Apply", the modifications will be lost.
An example of a regular operation process
Scene 1: Manually set the preset monitoring points
Use the upper right direction and the zoom button to adjust the camera to the scene you need to monitor.
Select the number by pulling down the preset point on the upper left (e.g., 1);
Click "Save" and the point storage is completed.
When you need to review this screen later, just select the corresponding number and click "Call".
Scene 2: Activate the automatic sound tracking gimbal
Change the "Camera Tracking" option at the bottom right to enable.
Check that the serial port number, address number and protocol on the right side are exactly the same as the pan-tilt parameters above.
Adjust the tracking threshold at the lower left to adapt to the on-site noise.
Click "Apply" at the lower right corner to save.
When the on-site microphone detects a human voice, the gimbal will automatically switch to the source of the sound.

Port Settings



It is used to configure the communication baud rates of two independent serial ports: RS232 and RS485. The parameters of the two serial ports can be set separately and independently.
The baud rate can be set as follows: 1200, 2400, 4800, 9600, 19200, 38400, 57600, 115200, 230400, 460800. Pull down and select the corresponding baud rate to successfully set the corresponding baud rate. The default is 115200

◆ Technical parameters

SPECIFICATIONS		Artist T88+	Artist T16+	Artist T16A+	Artist T16D+
Signal Processing		32bit float point dsp 1GHz			
AD/DA Converter		24bit			
AD/DA Converter		48kHz			
Input Channels		8-channel electronically balanced line input USB sound card stereo input Bluetooth stereo input	16-channel electronically balanced line input USB sound card stereo input Bluetooth stereo input	16-channel electronically balanced line input USB sound card stereo input Bluetooth stereo input 32-channel AES67 digital signal input	16-channel electronically balanced line input USB sound card stereo input Bluetooth stereo input 32-channel Dante digital signal input
Input Interface		3.81mm Euroblock(analog input) Type-C		3.81mm Euroblock(analog input) Type-C RJ-45x2	
Output Channels		8-channel electronically balanced line output USB sound card stereo output	16-channel electronically balanced line output USB sound card stereo output	16-channel electronically balanced line output USB sound card stereo output 32-channel AES67 digital signal output	16-channel electronically balanced line output USB sound card stereo output 32-channel Dante digital signal output
Output Interface		3.81mm Euroblock (analog output)		3.81mm Euroblock (analog output) RJ-45x2	
DSP	Input	1. Volume, Mute, and Phase control 2. AFC / AEC/AI-AEC / ANC/AI-ANC / AGC / AUTO MIX / MATRIX MIX 3. Noise Gate / Compressor / Expander 4. 8-band EQ per input,14 EQ types optional,XOVER Butterworth, Bessel, Linkwitz-Riley 6/12/18/24/36/48dB/Oct) 5. 2000ms per channel			
	Output	1. Volume, Mute, and Phase control 2. 8-band EQ per output,14 EQ types optional 3. 2000ms per channel			
	Speaker Presets	1. Volume, Mute, and Phase control 2. FIR: 512Taps per Output channel 3. 15-band EQ per Output channel,14 EQ types optional,XOVER Butterworth, Bessel, Linkwitz-Riley 6/12/18/24/36/48dB/Oct) 4. 3-band DEQ per Output channel 5. 2000ms per Output channel 6. Compressor + Limiter			
Input Impedance		>7.5kΩ			
Output Impedance		<100Ω			
Maximum Input/Output Level		⩾21dBu/line 0dBu/microphone			
Phantom power		+48VDC, 7.5mA (optional per channel)			
Frequency Response		20Hz-20kHz(±0.5dB)/line			
S/N Ratio		⩾115dB(1kHz,A weighted)/line			
THD+N		≤0.003%(1kHz,A weighted)/line			
Crosstalk		⩽-110dB(1kHz,A weighted)/line			
Connection		RS232/RS485/TCP/IP/GPIO			
AC Power Operating Range		100-240V/(±10%,50-60Hz)			
Power consumption		<20W			
Rack space		1U			
Dimension(WxHxD)		484x44x245mm			
Net weight		3.0 kg	3.3 kg		

AUDIOCENTER

Dynamic Audio Solutions

http://www.audiocenter.com E-mail:sales@ac-pro.net