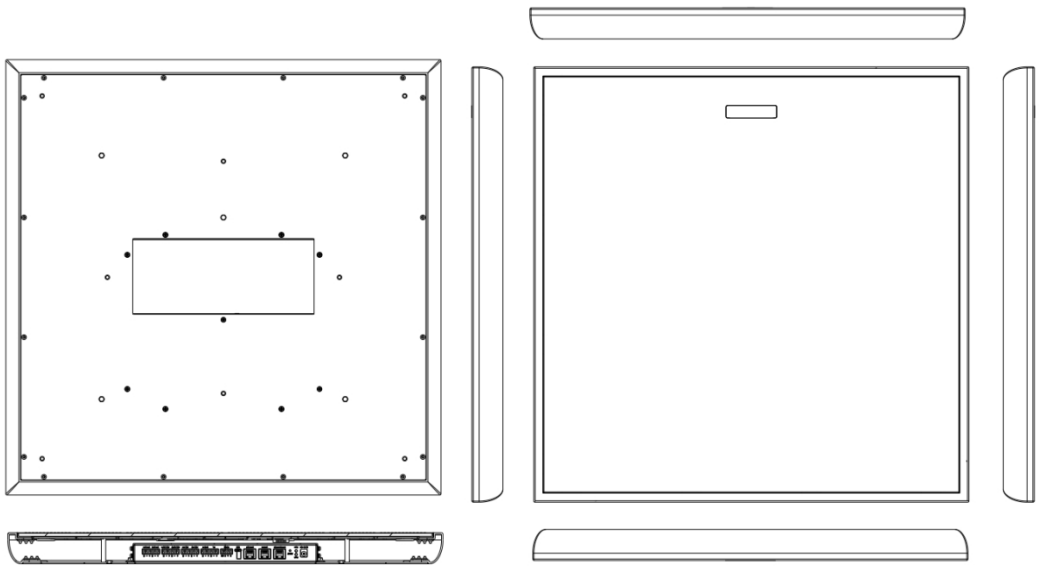
Intelligent Ceiling Linear Array Microphone

User Manual



TS-CA129

Please read this manual carefully before using the product.

Thank you for using the ceiling Array microphone. Please read this manual carefully to ensure safe and correct operation of the equipment. Please keep this manual properly for future reference.

The ceiling array microphone is composed of 129 high-fidelity microphones and features a composite circular array layout that provides the necessary pickup and sampling conditions for the algorithm; The CPU features a professional audio processor (8 cores, 6T high computing power), along with industry-leading audio signal processing algorithms. Such as the low-latency ClearVoice AI algorithm, real-time feedback suppression engine, adaptive sound field control technology, multimodal AI noise reduction system, spatial sound field modeling technology, traditional audio processing +AI deep algorithm dual-engine architecture, dynamic beamforming, etc., can effectively enhance the quality of voice amplification interaction; With the design of professional audio composite circular array layout microphones, it can achieve 33 sound amplification zones and 33 sound pickup zones, with a sound amplification distance of up to 4 meters and a sound pickup distance of up to 8 meters, which can effectively pick up sound at any position in the sound pickup zone; Built-in cascading function module enables the cascading of multiple ceiling array microphones; Built-in Dante module for network audio transmission function. Built-in advanced algorithms automatically match various scenarios, whether it's local amplification or output to a remote terminal, to achieve clear vocals and high fidelity.

Note: The company may upgrade the product and firmware later, which may be slightly different from this user guide. All rights of interpretation belong to the company.

Notes:

1. Please comply with relevant national laws and do not use this product for any illegal purposes;
2. Operating temperature: 0 to 40 degrees Celsius, use at the specified temperature;
3. Operating humidity: <90%. This product is not fully waterproof.
4 Avoid heavy pressure during transportation and storage to prevent damage to the product;
5. This product must be fastened with ropes in all other installation methods except for the hanging rod installation;
6. Built-in cascading function enables multiple TS-CA129 ceiling array microphones to be cascaded;
7. Built-in Dante module for digital audio network transmission function;
8 Do not disassemble the product without permission. Contact after-sales service if you encounter any problems during use.
9. When installing and deploying, make sure the hanging microphone is securely installed to prevent it from falling;
10 When installing and deploying, wires and equipment should be kept as far away as possible from strong electrical signal sources.

Catalog

1.Unpacking Inspection

When opening the outer packaging box, please check the appearance for any obvious damage and confirm that the items are consistent with the list. See the table below for the specific list.

Accessory names	Specifications	Quantity
Ceiling array microphone	Unit	1
Power adapter (output: DC24V/2.7A)	Pcs	1
3-pin power cord (1.5M)	Pcs	1
Infrared remote control	Pcs	1
Pitch 3.81, 6P Phoenix plug	Pcs	4
Spacing 3.81, 4P Phoenix plug	Pcs	1
Lifting accessory package (4 M5 lifting rings, 4 3-meter wire ropes, 4 spring hooks, 3 disc codes, 12 self-tapping taps, 12 plastic expansion tubes)	Package	4
Instruction for Use	Pcs	1

2.Features

1.Wide sound pickup: The ceiling array microphone is composed of 129 units of omnidirectional array microphone, with an amplification distance of 4 meters and a pickup distance of 8 meters. It can reach wherever you go, achieving truly seamless sound amplification.

2.Loud volume: Acoustic feedback gain ≥15dB, greater volume, less whistling, making speaking effortless.

3.Clean sound: Embedded with ClearVoice AI large model deep algorithm for precise sound recognition, removing all kinds of noise and enhancing voice signals.

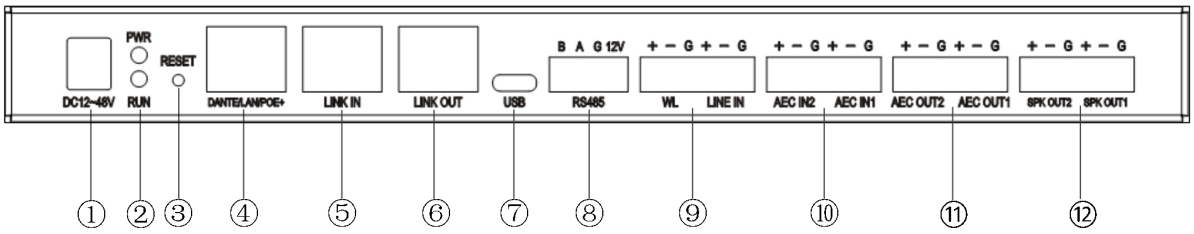
4.High flexibility: Supports custom positioning pickup, supports 33 pickup zones, 33 amplification zone mode Settings, and supports multiple units cascaded to adapt to different scenarios.

5.Easy to expand: Rich analog and digital (Dante, USB) audio interfaces can be used independently and connected to wireless microphones, etc., without the need for additional audio processors;

6.Easy to manage: The system is simple and offers a variety of external control methods such as infrared, 485, and network to meet different needs and make the system more flexible.

3.Product Introduction

Ceiling array microphone makes meeting sound administrators simple and efficient, with an ultra-wide pickup range. Whether it's a small room for 30 people or a large auditorium for 800 people, every corner can be clearly heard. Equipped with the low-latency ClearVoice AI algorithm, it automatically blocks out distractions such as keyboard sounds and air conditioning noise, significantly enhancing your meeting concentration. With dynamic beams, intelligent zone pickup during discussions and automatic voice tracking of the speaker during speeches. In remote meetings, the sound is as natural and real as a face-to-face conversation. Simplify the system, no more microphones, devices hidden in the ceiling, almost imperceptible during meetings, clear and smooth communication experience.



Ceiling array microphone interface diagram

1)DC12-48V interface: Power interface, input voltage range DC12V-48V.

2)PWR/RUN: Normal power-up, red light stays on; normal system operation, green light stays on.

3)RESET: Restore factory Settings.

4)DANTE/LAN/POE+: Dante signal transfer interface/Network interface /POE power supply network port

5)LINKE IN: Cascaded link input port.

6)LINK OUT: Cascaded link output port.

7)USB: Type C firmware upgrade interface, USB2.0 audio transfer interface.

8)RS485:485 device control interface; Output voltage DC12V/0.4A.

9)WL/LINE IN: Wireless microphone analog input; External audio source (PC) analog input.

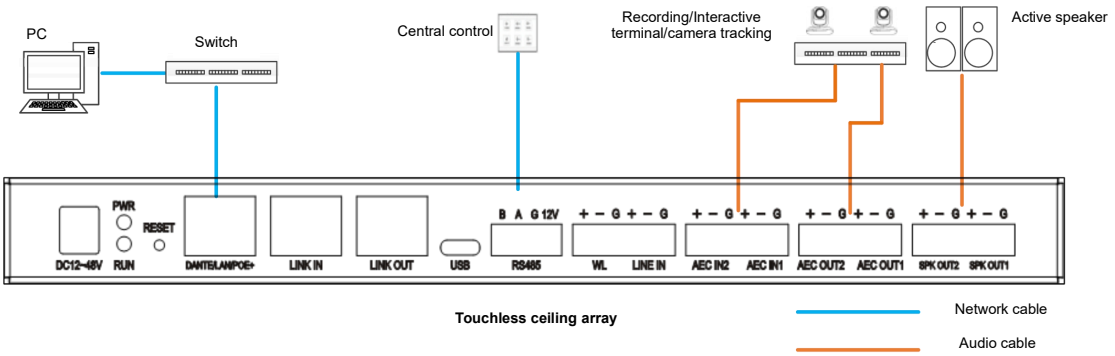
10)AEC in 1/2: Remote audio (reference signal) analog input.

11)AEC OUT1/2: Remote analog audio output.

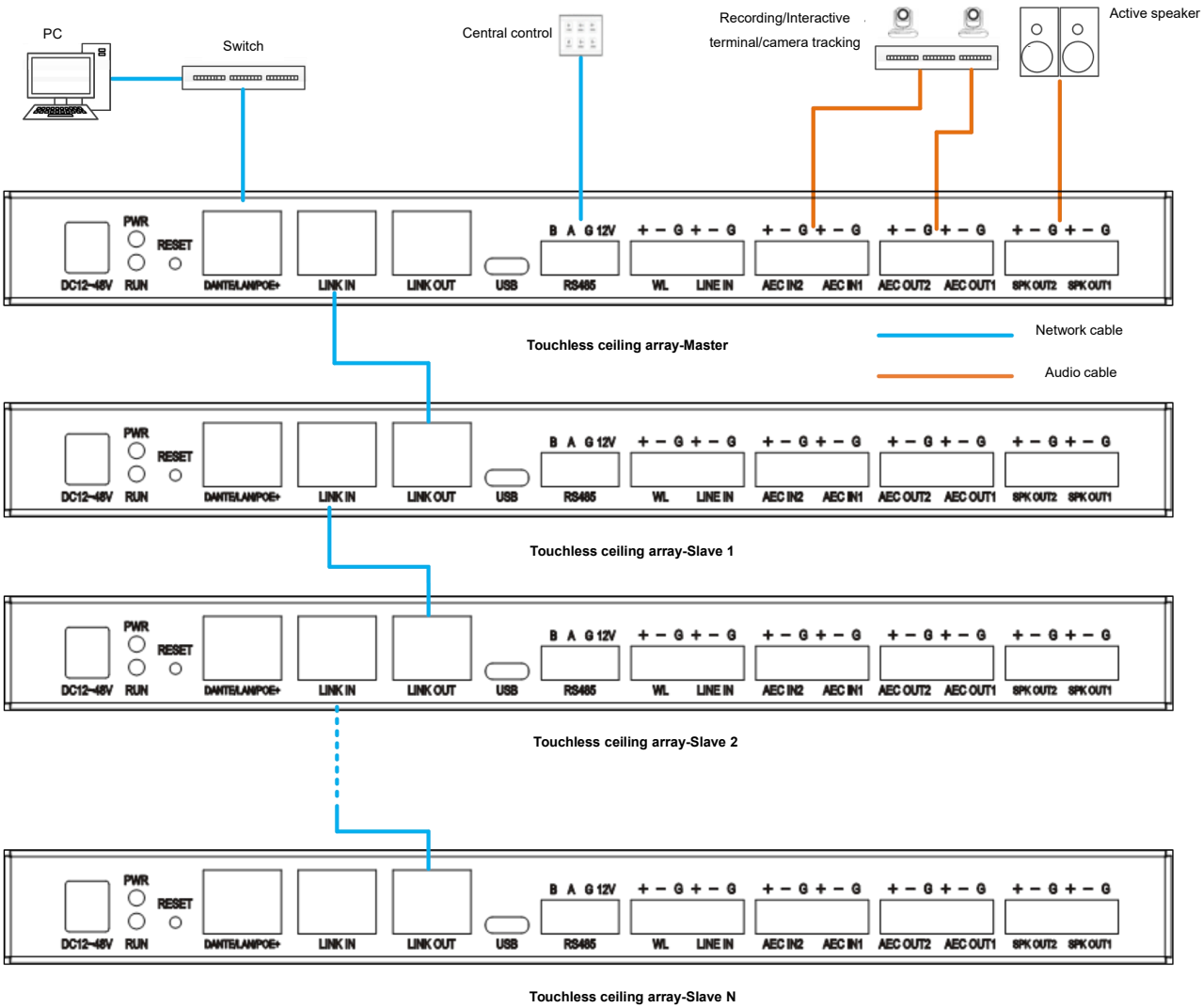
12)SPK OUT1/2: Speaker analog audio output.

4.Product Usage and Installation

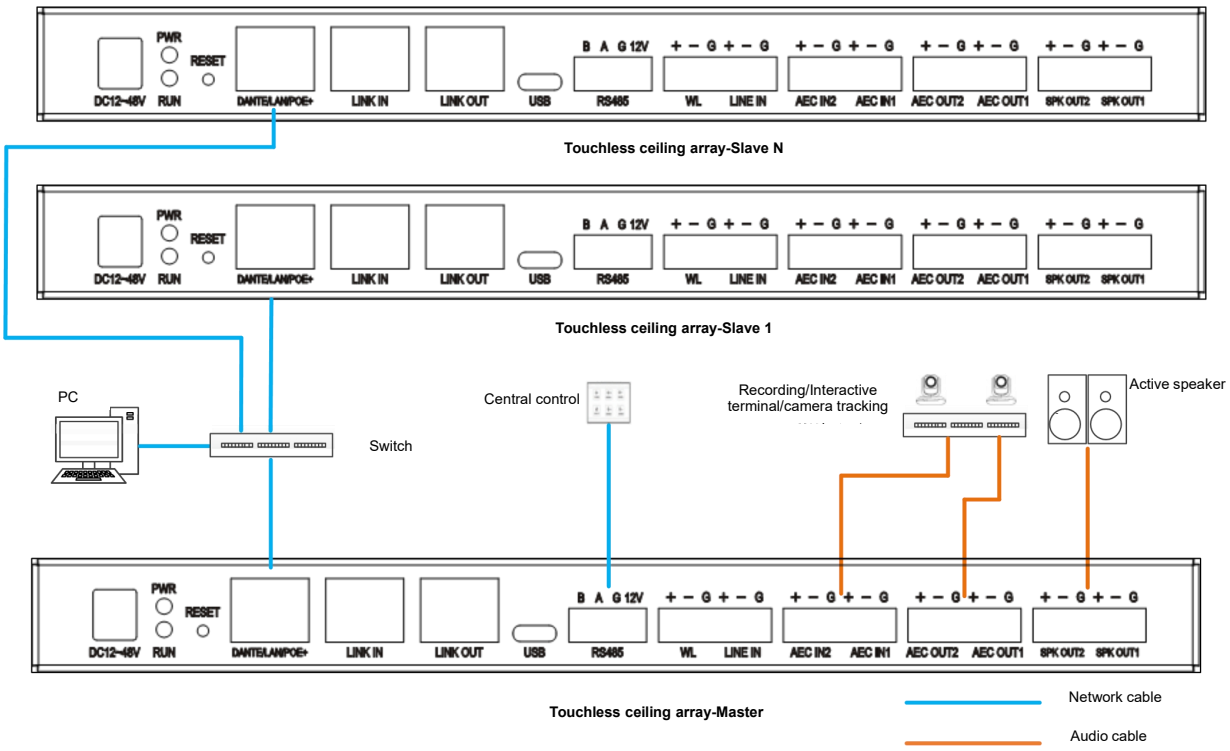
4.1.Standalone use



4.2.Use in series cascades



4.3.Use in parallel cascades



4.4.Installation method

1.Embedded to ceiling **(ropes must be added for hoisting and fixing)**

2.Rope hoisting

3.Boom installation

5.Product Parameters

Module	Functions	Specific Parameters
Algorithm specifications	Supported algorithms	AFC, ANS, AEC, AGC, ARR
	Acoustic feedback gain	≥18dB
	Noise reduction amplitude	≥30dB
	Echo cancellation amplitude	≥90dB
	Echo cancellation length	≥1s
	Reverberation suppression	≥18dB
	Maximum gain	≥30dB
	AI noise reduction	Supported
	Auto-mix	Supported
	Beamforming	Supported
	Sound source localization	Support
Microphone specifications	Microphone quantity	129
	Sensitivity	-32±2dB
	Signal-to-noise ratio	70dB
	Frequency response	75-20KHz
Hardware specifications	Sampling rate	48K
	Frequency response	20Hz to 20KHz ±0.5dB
	Signal-to-noise ratio	100dB
	Distortion	≦0.1%
	Input impedance (balanced)	20kΩ
	Output impedance (balanced)	200Ω
	Maximum input level (balanced)	4dBu
	Maximum output level (balanced)	10dBu
	Dante audio	Supported
	Cascading function	Supported
	485 central control interface	Supported
	Power input	DC 12V-48V/POE power supply
	Power	25W
Other specifications	Size (length × width × height)	600 * 600 * 42.8 mm
	Net weight of equipment	6.03 kg
	Operating temperature	0℃ to 40℃
	Storage temperature	- 20℃ ~ 60℃
	Color	Pearl white

6.Product Use

1. Client-side configuration software

1.1 Compatible operating systems: Windows 7 (64-bit), Windows 10 (64-bit).

1.2 Compatible hardware devices: Mainstream office and home-use PCs can run normally.

1.3 Installation method: No installation required. The exe file format can be opened and used directly.

1.4 Remote network requirements: The same local area network.

2 Configure debugging

2.1 After the device is installed and deployed as required, run the client configuration software micDemo.exe.

2.2 The factory default IP address of the device is: 192.168.1.100, subnet mask: 255.255.255.0, gateway: 192.168.1.1; Please add the IP address of this network segment in your PC first. The PC and the smart audio host need to be in the same network segment to connect normally.

2.3 Default parameters: When the device leaves the factory, a set of recommended parameter configurations is built in based on typical application scenarios as the default configuration, and the parameters can be adjusted and saved on-site according to the actual sound field environment.

2.4 Restore Default Parameters: If you need to restore the factory default parameters, you can restore the factory Settings in the client software.

3. Firmware upgrade: Connect the device and perform a firmware upgrade on the system after the system finds the device.

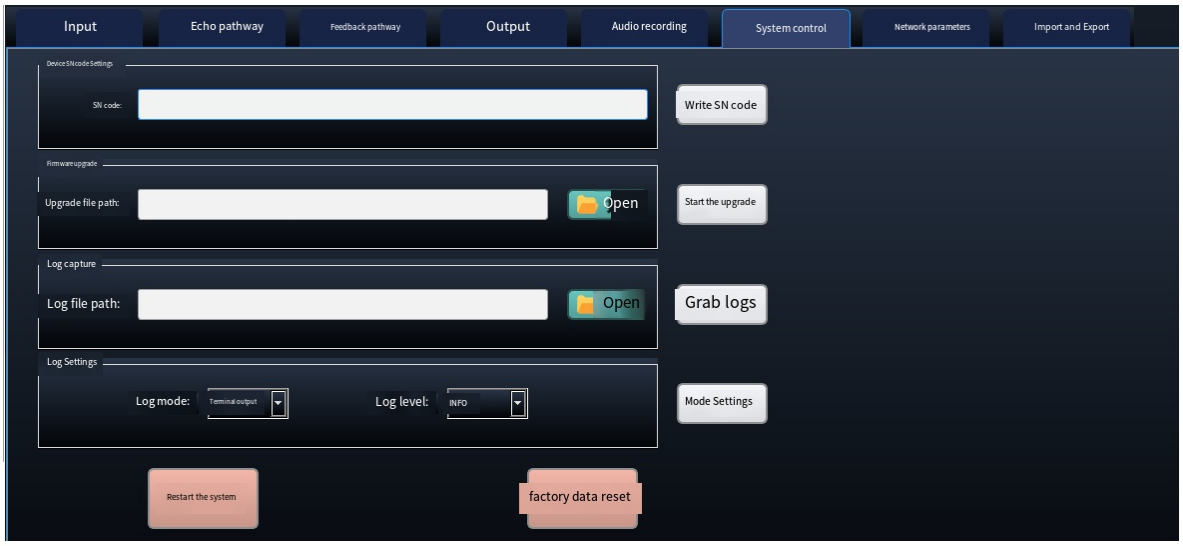


Figure 6.1 Upgrade interface diagram

Firmware upgrade method: Open Client - System Control - Firmware Upgrade, find the Upgrade File path box, System import the specified upgrade file, click Start Upgrade to automatically complete the firmware upgrade; The parameter configuration remains unchanged during the upgrade, and the system will automatically restart upon successful upgrade; Factory Settings need to be restored after reconnecting the device.

7.Introduction and Notes

To help you better understand the specific functions and main operating procedures of the smart ceiling linear array microphone system control software, this manual is specially compiled for your reference!

This document is written based solely on the product specification. If you find that some of the descriptions do not match the actual system during use and cannot meet your needs, please consult the relevant engineers in a timely manner. We apologize for any inconvenience caused during your reading and use, and appreciate your understanding!

8.Software Introduction

The ceiling linear array microphone makes meeting sound management simple and efficient. With an ultra-wide pickup range, whether it's a small room with 30 people or a large auditorium with 800, every corner can clearly hear the speaker. Equipped with the low-latency ClearVoice AI algorithm, it can automatically filter out distractions such as keyboard noise and air conditioning sounds, significantly improving focus during meetings. Using dynamic beamforming, it intelligently partitions audio during discussions and automatically tracks the speaker’s voice during presentations. In remote meetings, the sound feels as natural and real as face-to-face communication. The system is simplified, no longer requiring multiple microphones, with the equipment hidden in the ceiling, making it almost invisible during meetings while providing a clear and smooth communication experience.

The paired intelligent ceiling linear array microphone system control software supports controlling input channels, beam levels, beam control, channel selection, echo channel, feedback channel, mixing matrix, output channels, audio recording, system control, network parameter settings, as well as import and export functions.

The main compatible device model is the TS-CA129 ceiling linear array microphone device.

9.Software Usage

9.1.Software Deployment Requirements

Software platform requirements: The intelligent ceiling linear array microphone system control software is compatible with the following operating systems: Windows 7 (64-bit) and Window 10 (64-bit).

Hardware platform requirements : Mainstream office and home PCs can run normally.

Installation method : No installation required, the exe file format can be opened and used directly.

Remote network requirements: Same local area network.

9.2.Configuration Debugging Requirements

1. After the device is installed and deployed as required, run the client configuration software micDemo.exe.

2. The factory default IP address of the device is: 192.168.1.100, subnet mask: 255.255.255.0, network gateway: 192.168.1.1. Please first add the IP address of this network segment on the PC (it should be different from the device IP address to avoid conflicts. It is recommended to disable firewalls, anti-virus software and other blocking systems). The PC and controller must be in the same network segment to connect normally.

3. Default parameters: When the device leaves the factory, a set of recommended parameter configurations is built in as the default configuration based on typical application scenarios. Parameters can be adjusted and saved on-site based on the actual sound field environment.

4. Restore default parameters: If you need to restore the factory default parameters, you can restore the default factory settings in the client software.

5. Device indicator light function (normal connection after power on):

(1)When the power is turned on, the red, blue and green lights will light up in a cycle.

(2)When working normally, the green light goes on.

(3)When the microphone local sound reinforcement is muted, the red light goes on.

9.3.System Flow

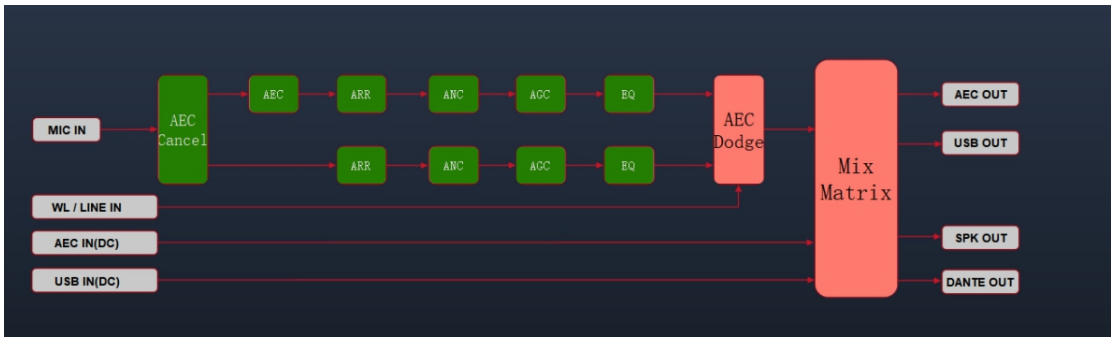


Figure 3-1 System Flow

1.Microphone input channel;

2.Whether the microphone channel algorithm is enabled or not, green means enabled, red means disabled;

3.WL\LINE IN, AEC IN, USB IN input channels;

4.Echo channel algorithm function configuration: Click to switch, green means enabled, red means disabled;

5.Feedback channel algorithm function configuration: Click to switch, green means enabled, red means disabled;

6.Mixing selection: Click to switch to the mixing matrix interface in the processor parameter control area;

7.Output channel.

10.Home Interface Functions and Contents

Double-click to run the client configuration software micDemo.exe and open the software to enter the home page interface .

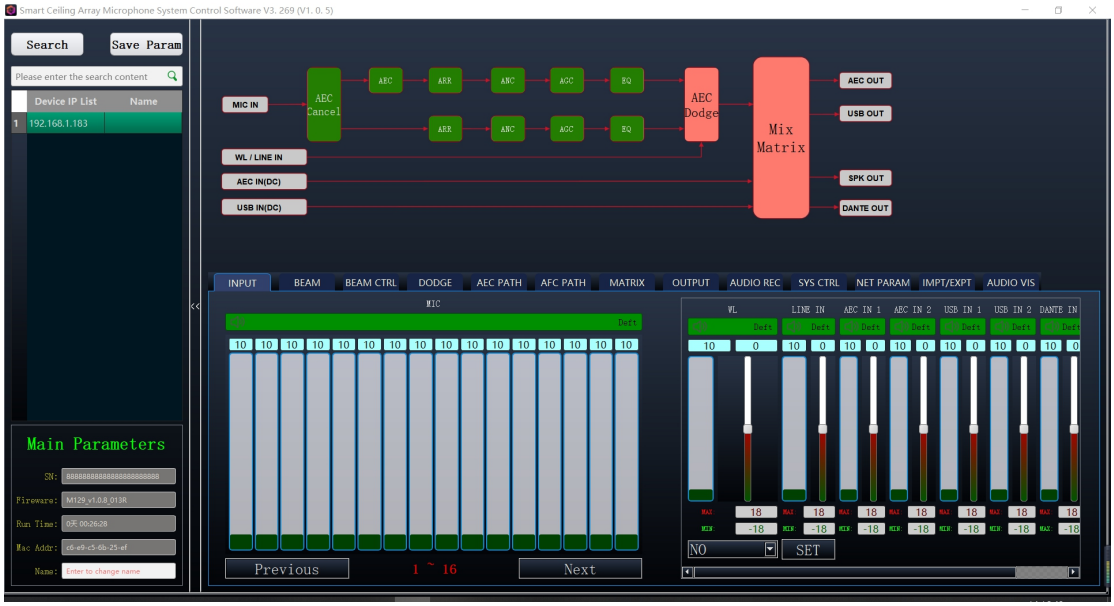


Figure 4-1 Home page interface

1.Area ① is for device search and parameter saving.

2.Area ② is the area where the device's IP address is searched. Select the IP address and double-click it to log in to the device.

3.Area ③ is the system flow area and equipment signal flow.

4.Area ④ is the main parameter display area of the device, which displays the device software version information.

5.Area ⑤ is the flow control area: audio data flow chart, you can click the icon to set the parameters of each processing module in detail.

6.Area ⑥ is the processor parameter control area: you can drag or scroll the mouse to display the hidden part.

10.1.Device List Function

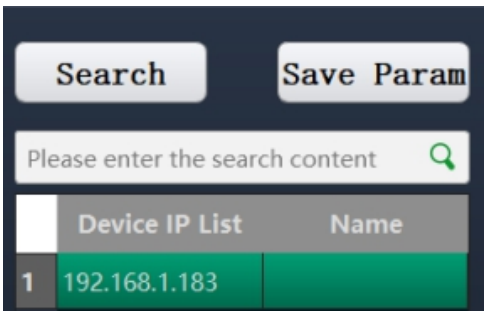


Figure 4-2 Device list page

1.Device Search:

(1)When there is no device on the page , click [Device Search] on the page to enter the search page to search for all devices that can be connected in the same network segment and display the corresponding IP addresses;

(2)When there are devices on the page, click on the online device in the device list to load the device and view the function parameter settings related to the device.

2.Parameter Save: Click [Parameter save] to save the parameters of the current device settings.

10.2.Device Parameters

Figure 4-3 Device parameters page

After successfully connecting to the device , you can view the SN, FiteWare, Run Time, Mac Addr, and Name parameters of the current device.

(1)SN: Displays the serial number of the device;

(2)Fire Ware: Displays the software firmware version number;

(3)Run Time: Indicates the normal operating time;

(4)Mac Addr: Displays MAC address information;

(5)Name: Modify the device name. Press Enter to confirm the modification.

10.3.Functions of Each Channel

10.3.1.Input Channel

Figure 4-4 Input channel interface

1.Linear Array microphone input channel:

(1)Mute function: Turn on/off the linear array microphone signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Level display bar: Displays the current volume changes of the linear array microphone.

2.LINE IN1 input channel:

(1)Mute function: Turn on/off the LINE IN1 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the LINE IN1 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

3.LINE IN2 input channel:

(1)Mute function: Turn on/off the LINE IN2 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: can change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the LINE IN2 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

4.AEC IN1 input channel:

(1)Mute function: Turn on/off the AEC IN1 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the AEC IN1 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

5.AEC IN2 input channel:

(1)Mute function: Turn on/off the AEC IN2 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the AEC IN2 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

6.USB IN1 input channel:

(1)Mute function: Turn on/off the USB IN1 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of USB IN1 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

7.USB IN2 input channel:

(1)Mute function: Turn on/off the USB IN2 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of USB IN2 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

8.DANTE IN1: These two inputs are only available in Dante mode and are the input channels for Dante module signals.

(1)Mute function: Turn on/off the DANTE IN1 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the DANTE IN1 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be

less than -18.

9.DANTE IN2: These two inputs are only available in Dante mode and are the input channels for Dante module signals.

(1)Mute function: Turn on/off the DANTE IN2 channel input signal.

(2)Level value: Displays the volume level in real time and cannot be adjusted.

(3)Volume setting value: Enter a number to change the volume gain, the maximum adjustment range is -18 to 18.

(4)Gain fader: Change the volume gain, the maximum adjustment range is -18 to 18.

(5)Level display bar: Displays the current volume change of the DANTE IN2 channel.

(6)Maximum value: Set the maximum value of volume gain adjustment, which shall not exceed 18.

(7)Minimum value: Set the minimum value of volume gain adjustment, which shall not be less than -18.

10.3.2.Beam Level

Figure 4-5 Beam level interface

1.ANGEL: HORI (horizontal angle from the sound source to the device), VERT (vertical angle from the sound source to the device).

2.33 beams with real-time level hopping.

3.Beam level page turning

4.Beam level page turning

10.3.3.Beam level control

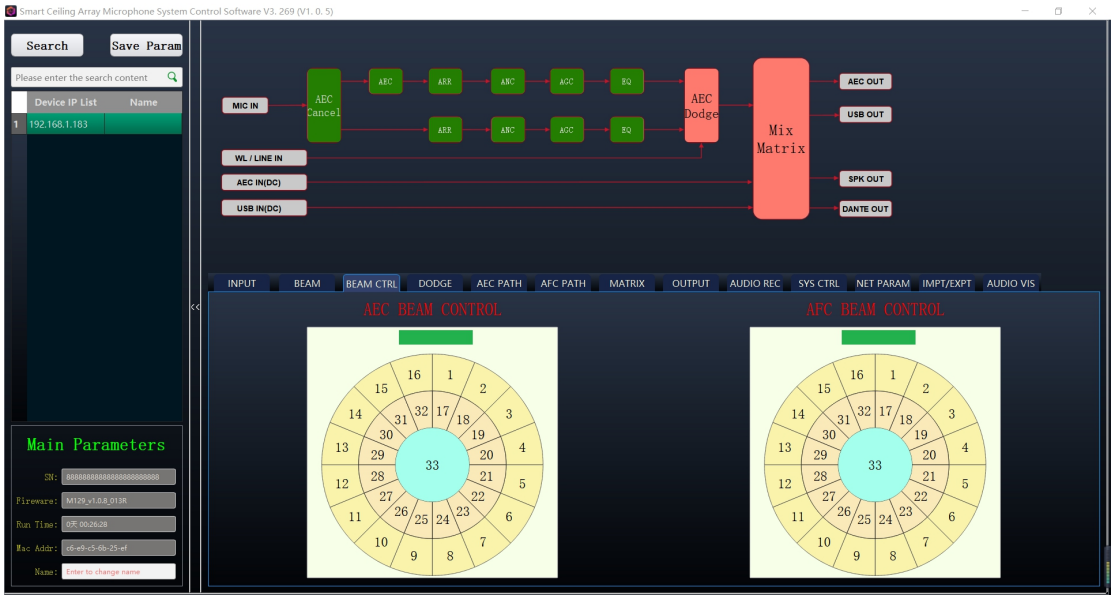


Figure 4-6 Beam control interface

1.Echo channel beam control: There are 33 beam switches in the echo channel, gray indicates off, bright indicates on.

2.Feedback channel beam control: There are 33 beam switches in the feedback channel, gray indicates off, bright indicates on.

10.3.4.Channel Selection

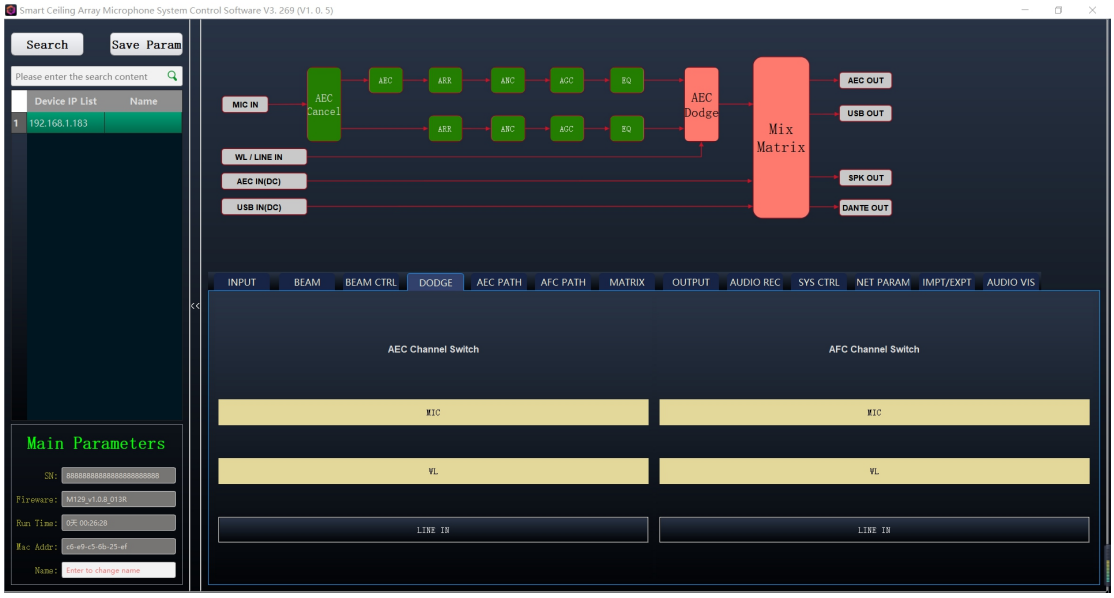


Figure 4-7 Channel selection interface

1.Echo channel switch: Click the selected channel to output the signal of the channel after echo cancellation processing.

2.Feedback channel switch: Click the selected channel output the signal of the channel after feedback processing.

10.3.5.Echo Channel

Figure 4-8 Echo channel interface

1.Echo suppression level: The value shows the current level, which can be changed by the fader, with an adjustment range of 0 to 4. The maximum value of the echo suppression level should not exceed 4, and the minimum value should not be lower than 0.

2.Reverberation suppression amount: The value shows the current level, which can be changed by the fader, with an adjustment range of 3 to 18. The maximum value of the reverberation suppression level should not exceed 18, and the minimum value should not be lower than 3.

3.Steady noise suppression amount: The value shows the current level, which can be changed by the fader, with an adjustment range of 3 to 30; the maximum value of the smooth noise suppression level is set not to exceed 30, and the minimum value is not less than 3.

4.AI noise reduction: The amount of transient noise suppression. The value shows the current level, which can be changed by the fader, with an adjustment range of 0 to 30. The maximum value of the noise reduction level cannot exceed 30, and the minimum value cannot be lower than 0.

5.Automatic maximum gain: The value shows the current level, which can be changed by the fader, with an adjustment range of 6 to 15; the maximum value is set not to exceed 15, and the minimum value is not less than 6.

6.13-band equalizer adjustment: Click the label with the mouse and drag it up or down to change the amplitude of the frequency band; the gain parameter can be directly input, and the amplitude of the frequency band will be changed after entering the parameter; push the fader to change the amplitude of the frequency band.

10.3.6.Feedback Channel

Figure 4-9 Feedback channel interface

1.Feedback suppression level: The value shows the current level, which can be changed by the fader, with an adjustment range of 0 to 4. The maximum value of the feedback suppression level should not exceed 4, and the minimum value should not be lower than 0.

2.Reverberation suppression amount: The value shows the current level, which can be changed by the fader, with an adjustment range of 3 to 18. The maximum value of the reverberation suppression level should not exceed 18, and the minimum value should not be lower than 3.

3.Steady noise suppression amount: The value shows the current level, which can be changed by the fader, with an adjustment range of 3 to 30; the maximum value of the smooth noise suppression level is set not to exceed 30, and the minimum value is not less than 3.

4.AI noise reduction: The amount of transient noise suppression. The value shows the current level, which can be changed by the fader, with an adjustment range of 0 to 30. The maximum

value of the noise reduction level cannot exceed 30, and the minimum value cannot be lower than 0.

5.Automatic maximum gain: The value shows the current level, which can be changed by the fader, with an adjustment range of 6 to 15; the maximum value is set not to exceed 15, and the minimum value is not less than 6.

6.Feedback gain: The value shows the current level; the level can be changed by the fader, with an adjustment range of -12 to 6; the maximum value of the feedback gain level is set to no more than 6; the minimum value is not less than -12.

7.Sound reinforcement angle: The value shows the current angle; the angle can be changed by the fader, and the adjustment range is 180° to 360°; the maximum value of the set angle does not exceed 360°; the minimum value is not less than 180°.

8.13-band equalizer adjustment: Click the label with the mouse and drag it up or down to change the amplitude of the frequency band; the gain parameter can be directly input, and the amplitude of the frequency band will be changed after entering the parameter; push the fader to change the amplitude of the frequency band.

10.3.7.Mixing Matrix

Figure 4-10 Mixing matrix interface

1.Input Channels:

(1)USB IN: USB input channel.

(2)DANTE IN1: Dante mode remote interactive input channel.

(3)DANTE IN2: Dante mode local sound reinforcement input channel

(4)WL\LINE IN: Line input channel, mainly for computers or other line-out devices.

(5)AEC IN: Echo reference input channel.

(6)AEC CHANNEL: Signal after echo cancellation, an internal channel of the software, with no structured interface definition.

(7)AFC CHANNEL: Signal after feedback suppression, a channel within the software, with no structured interface definition.

2.Output channels:

(1)AEC OUT: The output signal sent to the remote interaction, with echo cancellation applied.

(2)SPK OUT: Local sound reinforcement output when there is no cascade connection, connected to the power amplifier.

(3)USB OUT: USB interface output channel.

(4)DANTE OUT1: The remote interactive output in Dante mode, with echo cancellation applied to the signal.

(5)DANTE OUT2: The local amplification output in Dante mode, with feedback suppression applied to the signal.

10.3.8.Output Channel

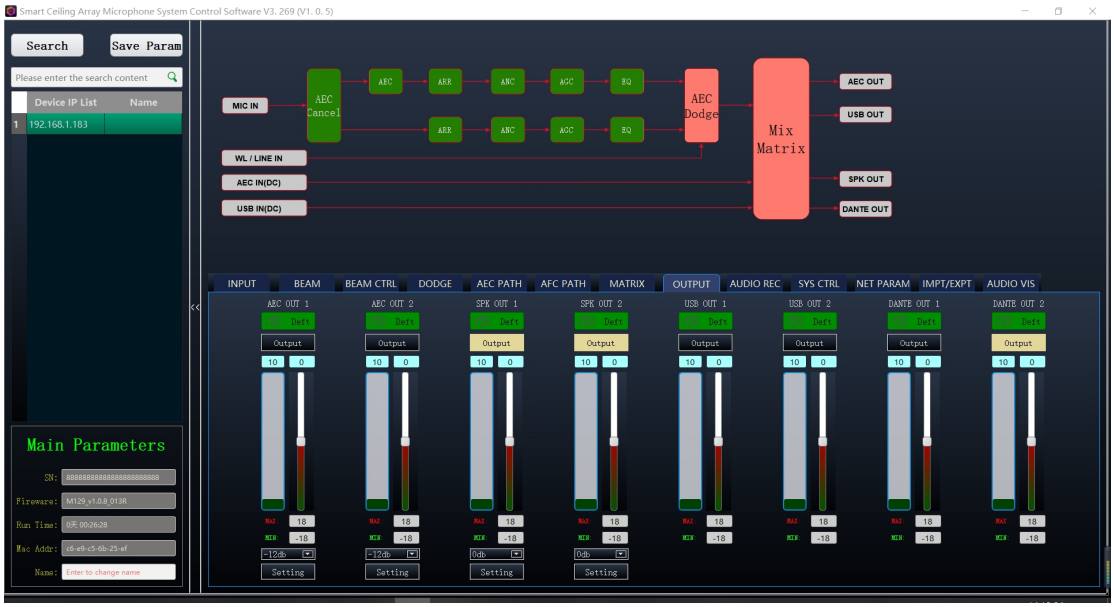


Figure 4-11 Output channel interface

1.AEC OUT output channel:

① Mute function: Turn on/off the AEC OUT channel input signal.

② Audio output: Turning on the audio output function (yellow) means that the output has a feedback function; turning off the audio output function (gray) means that the AEC OUT function defined by the channel is outputting normally.

③ Level value: Displays the volume level in real time and cannot be adjusted.

④ Volume display: Displays the current volume gain.

⑤ Level display bar: Displays the current volume level changes of the output channel.

⑥ Gain fader: Can change the volume gain, the maximum adjustment range is -18 to 18.

⑦ Maximum value: Set the maximum value of the volume gain adjustment to no more than 18.

⑧ Minimum value: Set the minimum value of volume gain adjustment to no less than -18.

⑨ Output decibel value: 0db, -12db, -18db, -24db four values are optional, after selecting, click the set button to apply

2.The settings of SPK OUT, USB OUT, and DANTE OUT are the same as above.

10.4.Other Features

10.4.1.Audio Recording Function

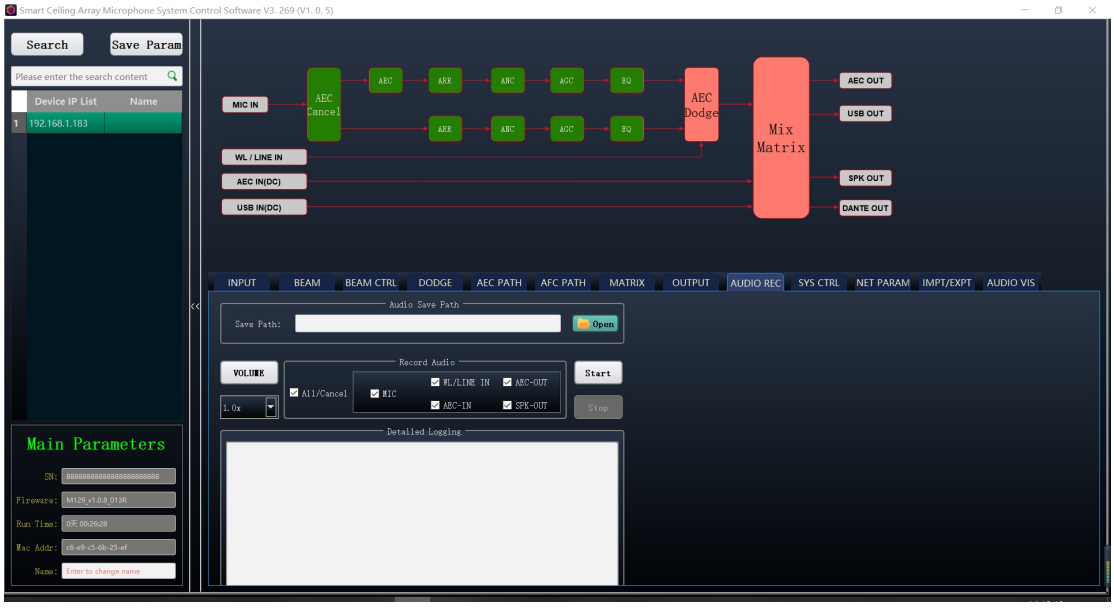


Figure 4-12 Audio recording interface

1.Audio source saving path: Click to open the optional path and set the path where the recorded audio file is saved on the computer.

2.Recording volume: Supports 0.5, 1.0, 1.5, 2.0, and 4.0.

3.Recording audio sources: Five audio sources for recording are optional (MIC1/2/WL-LN, PC-IN, AEC-OUT, etc.). By default, all are selected. Tick the box to select the audio source you want to record. After selecting the audio source, click Start to begin recording. Click Stop to stop recording. The recorded audio source file will be saved in the specified folder.

4.Log details: Record some information of the device studio.

10.4.2.System Control Function

Figure 4-13 System control interface

1.Device SN code setting: Write the actual SN code according to actual needs; you can click [Write SN Code] to write it in. A corresponding pop-up window will be displayed if the writing is successful (SN code modification is successful).

2.Firmware upgrade: After selecting the corresponding matching file to open, click [Start Upgrade] to perform the firmware upgrade operation. After the upgrade is successful, a pop-up window will be displayed (Upgrade successful) and the device will automatically restart. At this time, you need to search for the device again.

3.Log capture: Click [Open] to set the save path for the captured log, and then click [Capture Log] to capture the log.

4.Speaker selection: Supports selection of AC31S passive speakers, SS36 passive column speakers, model three, model four, model five, etc.

5.Log settings: Supports log setting modes, including: no output, terminal output, and writing to

files.

6.Restart system: Click to restart the system automatically, but no configuration will be modified.

7.Restore factory settings: Click to automatically restart the system and restore the parameter configuration to the factory default configuration.

10.4.3.Network Parameter Setting Function

Figure 4-14 Network parameter interface

1.Obtain IP address automatically: After selecting obtain IP address automatically, the IP address cannot be modified manually (the default display IP: 192.168.1.100) .

2.Manually modify the IP address: Manually enter the IP address, subnet mask, and default gateway.

3.IP address: Manually enter the IP address to be set (for example, 192.168.xx).

4.Subnet Mask: Enter the subnet mask (for example, 255.255.255.0).

5.Default Gateway: Enter the gateway (for example, 192.168.1.1).

6.After entering the IP address, subnet mask, and gateway, click OK. After the upgrade is successful, a pop-up window will pop up to prompt the upgrade is successful! The device will automatically restart, and you will need to search for the device again.

7.Cascade settings: Set the corresponding device as the master microphone or slave microphone and click OK.

10.4.4.Import and Export Functions

Figure 4-15 Import and export interface

1.Export configuration: After selecting the path, click the inverted triangle of the current configuration, select the scene data to be exported, and then click Export Configuration to export the data. After the export is successful, it will prompt that the configuration is successful.

2.Import configuration: After selecting the path, click the inverted triangle of the current configuration, select the scene data to be imported, and then click Import Configuration to import the data and overwrite the current scene. After the import is successful, a pop-up window will prompt that the import is successful.

3.Import file: Click Open, find the path where the file is saved on the computer, click the selected file, and then click Import file to import the local file in the selected path. After the import is successful, the data in the storage device will be overwritten.

4.Scenario:

(1)Scene switching: Supports switching scenes 1/2/3/4/5/6/7/8/9/10, with clear mode scene as the default;

(2)Scene save: After modifying the data of the current scene, click Save to save the debugged data

(3)Scenario application: After selecting the corresponding scene, you can apply this scene.

11.Troubleshooting

Phenomena	Reason	Solution
Fails to open PC tool	Antivirus software limit	Turn off antivirus software
	Tool damage	Update PC tools
The network is not functioning properly	Network issue	Check the device status light
		Check the PC network to ensure that the network segment is consistent with the device
		Hardware RESET restores default and searches using the default network
		When cascaded IN series, the LINK IN of the previous machine is connected to the LINK OUT of the next machine, and the last machine cannot be connected to the network
		When cascaded in parallel, each device can only have one network port connected to the network.
		Three ports cannot be connected to the same network at the same time, only one port can be connected to the network
	Network port failure	Check the status of the PC port light
	Device failure	Return to factory for repair
Low output volume	Configuration issue	Check the input and output configuration
	Output gain gear	For each output of AEC OUT and SPK OUT, there are four hardware analog gain Settings: 0dB, -12dB, -18dB, -24dB; The default is 0dB.
	Wiring failure	Check if the input and output wiring is correct

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Warning

1. Do not block the vent.
2. If the power cord is damaged, stop using it immediately.
3. It should not be placed in the following places: poor ventilation; dusty; direct sunlight; high ambient temperature or close to heat sources; subject to vibration.
4. It is required to be protected from rain and moisture.
5. It cannot be placed in a container that can be filled with liquid.
6. The housing is required to be safely grounded.
7. The power plug and socket are required to be compatible.
8. Please do not open the housing cover by yourself to avoid electric shock. When it fails, please contact the supplier or the manufacturer for maintenance services, or you can entrust professionals in professional departments with professional certificates. Non-professionals are forbidden to disassemble the machine by themselves, otherwise electric shock may endanger the their life.

Specifications and parameters are subject to change without notice

Ver 0.1

USER MANUAL