# iSpeaker CM365

Ceiling Array Microphone

SOFTWARE OPERATION USER MANUAL

Version 1.1 | 2026

www.infobitav.com

# 1. Product Overview

## 1.1 Product Overview

The **INFOBIT iSpeaker CM365** is an intelligent ceiling array microphone engineered for transparent sound reinforcement and professional conferencing environments. The device integrates **129 high-fidelity microphones** in a composite circular array layout, providing the acoustic sampling density required for advanced beamforming and signal processing.

The processor core features a professional audio processing unit (8-core CPU, 6T computing power) running a suite of industry-leading algorithms:

- ClearVoice AI algorithm
- Real-time Anti-Feedback Suppression Engine
- Adaptive Acoustic Field Control technology
- Multi-modal AI noise reduction system
- Spatial acoustic field modeling
- Traditional DSP + AI deep-learning dual-engine architecture
- Dynamic beamforming

**Key performance capabilities:**

- **Multiple zones:** 33 amplification zones and 33 pickup zones
- **Amplification range:** up to 4 meters
- **Pickup range:** up to 8 meters — captures audio from any position within the pickup zone
- **Built-in cascade module** — supports multi-unit cascading for large venues
- **Built-in Dante module** — supports digital audio network transmission

# 2. Product Interface Description

The **iSpeaker CM365** provides the following physical connectors and status indicators:

| Interface / Indicator | Description |
|---|---|
| **DC 12–48V** | Power input connector. Accepts DC 12V to 48V power supply. |

| | |
|---|---|
| **PWR / RUN** | Indicators — **PWR**: solid red when device is powered on. RUN: solid green when operating normally. |
| **RESET** | Factory reset button. Restores all settings to factory defaults. |
| **DANTE / LAN / POE+** | Dante digital audio transmission port / Ethernet network port / PoE+ power input. |
| **LINK IN** | Cascade link input port for multi-unit daisy-chain configurations. |
| **LINK OUT** | Cascade link output port for multi-unit daisy-chain configurations. |
| **USB (Type-C)** | Firmware upgrade interface (Type-C connector) and USB 2.0 audio transmission interface. |
| **RS485** | RS-485 device control interface. Provides DC 12V / 0.4A auxiliary output. |
| **WL / LINE IN** | Wireless microphone analog input and external audio source (PC line-out) analog input. |
| **AEC IN 1/2** | Remote audio analog input for echo cancellation reference signal. |
| **AEC OUT 1/2** | Remote audio analog output — carries the post-echo-cancellation signal. |
| **SPK OUT 1/2** | Speaker analog audio output — carries the post-feedback-suppression local amplification signal. |

# 3. PC Management Software

> **NOTE:** *Default device network settings:* ***IP Address: 192.168.1.100*** *|* ***Subnet Mask: 255.255.255.0*** *|* ***Gateway: 192.168.1.1****. Before connecting, configure the PC with an IP address in the same subnet (must differ from the device IP to avoid conflicts). It is recommended to disable the Windows Firewall and any antivirus software to ensure normal device connectivity.*

## 3.1 Software Overview

The PC management software interface is divided into the following functional areas:


| Area | Function |
| --- | --- |
| **1 — Search & Save Parameters** | Discovers devices on the local network. Saves configured parameters to the device. |
| **2 — Device List** | Displays discovered device IP addresses. Double-click an IP to connect. |
| **3 — Signal Flow** | Graphical display of the audio signal processing chain. Click module icons |

| | |
|---|---|
| **Diagram** | to access parameter settings. |
| **4 — Main Parameters** | Shows device SN, firmware version, runtime, MAC address, and editable device name. |
| **5 — Flow Control Area** | Tab-based parameter control for each processing module (INPUT, AEC PATH, AFC PATH, OUTPUT, etc.). Click the icons to set detailed parameters for each processing module. |
| **6 — Processor Control** | Expandable/scrollable panel for detailed parameter adjustment of the selected module. Drag or scroll the mouse to show hidden sections. |

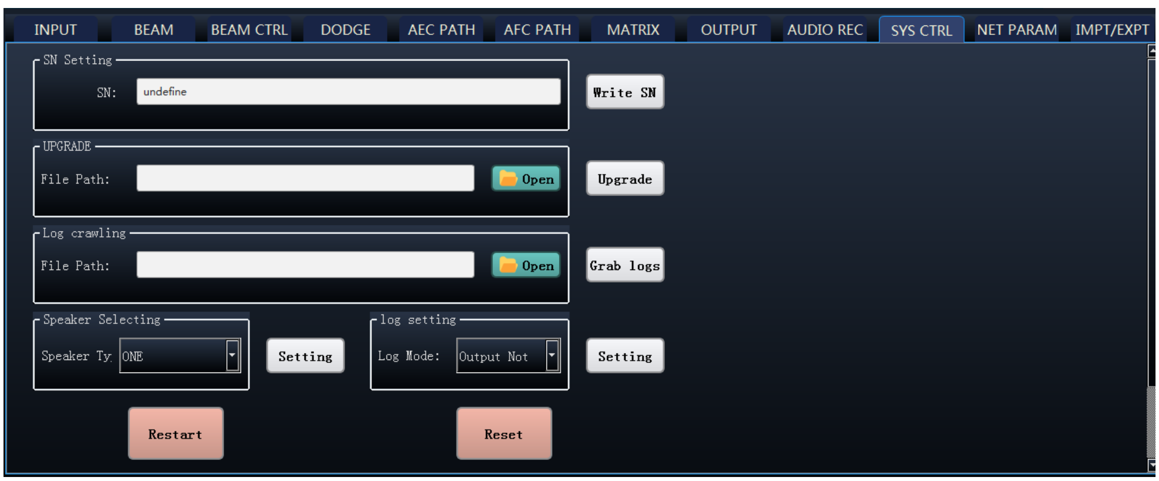
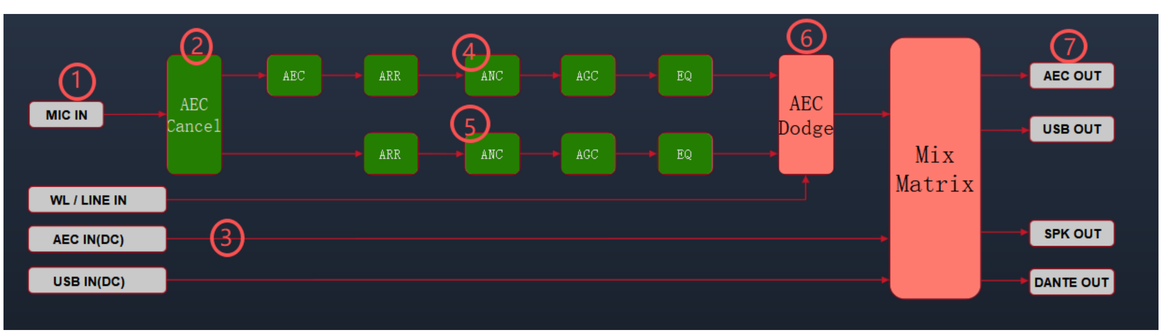



*Figure 3.1 — Software Main Interface (SYS CTRL tab shown)*

## 3.2 System Signal Flow



The following describes the audio processing chain within the **iSpeaker CM365**:

- MIC1 / MIC2 → AEC Cancel → AEC Dodge → AEC → ARR → ANC → AGC → EQ → Mix Matrix → AEC OUT
- AFC Cancel → AFC Dodge → AFC → ARR → ANC → AGC → EQ → Mix Matrix → SPK OUT
- Additional inputs: WL/LINE IN, AEC IN, USB IN feed into the processing chain as configured.

**1. Microphone Input Channel.**

**2. Microphone Channel Algorithm Enable/Disable Switch:** Green indicates enabled, red indicates disabled.

**3. WL/LINE IN, AEC IN, and USB IN Input Channels.**

**4. Echo Channel Algorithm Configuration:** Click to toggle; green indicates enabled, red indicates disabled.

**5. Feedback Channel Algorithm Configuration:** Click to toggle; green indicates enabled, red indicates disabled.

**6. Mix Selection:** Click to toggle to switch to the Mix Matrix interface in the Processor Parameter Control Area.

**7. Output Channels.**

*Note: Signal indicator colors: Green = enabled / active.  Red = disabled / muted.*

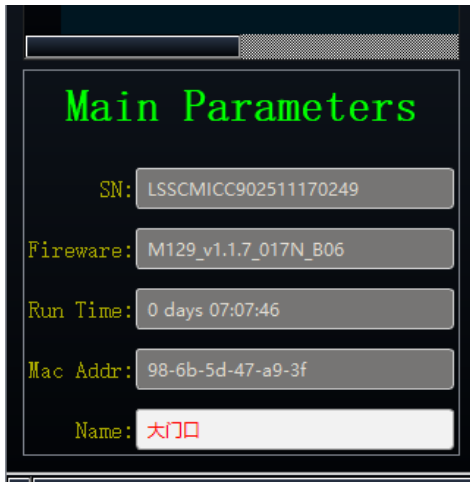
## 3.3 Software Features

### 3.3.1 Device Search & Connection

Click **Search** to discover all devices on the same local area network. Discovered devices appear with their IP addresses. Select the target IP and double-click to connect. Click **Save Param** to save the current parameter configuration to the device.

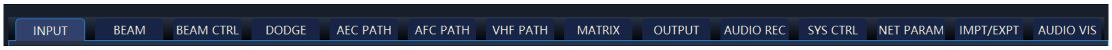
### 3.3.2 Main Parameters Panel




The left-side Main Parameters panel provides the following read-only and editable fields:

| Field | Description |
| --- | --- |
| **SN** | Device serial number (read-only) |
| **Firmware** | Installed firmware version number (read-only) |
| **Run Time** | Elapsed device runtime since last restart (read-only) |
| **Mac Addr** | Device hardware MAC address (read-only) |
| **Name** | Device display name — editable. Type a new name in the field and press Enter to apply. |

### 3.3.3 Flow Control Area




Click any module icon in the signal flow diagram to select it and configure its parameters in the tabbed processor control area.

# 3.4 Processor Modules

## 3.4.1 INPUT — Input Settings

The INPUT tab provides gain, mute, and level metering controls for all audio input channels.


*Figure 3.4.1 — Main Interface with INPUT Tab Selected*

| Parameter | Range / Values | Description |
|---|---|---|
| **MIC1 / MIC2** | Array microphone | Ceiling array microphone channels 1 and 2. Mute toggles the signal on/off. Level meter displays real-time signal level (read-only). These channels feed into the AEC processing path. |
| **WL-MIC** | Gain: −18 to +18 dB | Wireless microphone analog input. Adjustable gain fader and numeric input. Set MAX (≤+18) and MIN (≥−18) gain limits to restrict adjustment range. |
| **PC-IN (LINE IN)** | Gain: −18 to +18 dB | External audio source (PC) analog input. Same gain controls and limit settings as WL-MIC. |
| **AEC-IN** | Gain: −18 to +18 dB | Remote audio reference signal input for acoustic echo cancellation. Same gain controls and limit settings as WL-MIC. |

| | | |
|---|---|---|
| **USB IN 1/2** | Gain: −18 to +18 dB | USB audio input channels. In remote interaction mode, can be selected as the echo reference signal source. |
| **DANTE IN 1/2** | Gain: −18 to +18 dB | Available in Dante mode only. DANTE IN1 is the remote interaction input channel. DANTE IN2 is the local amplification input channel. |
| **Teacher Dual Mic** | Toggle ON/OFF | Enables teacher dual-microphone mode for classroom deployments requiring split-zone microphone tracking. |

For each adjustable channel, the following controls are provided:

- Mute: Enables or disables the input channel signal.
- Level Meter: Displays real-time signal level — read-only display, not adjustable.
- Gain Fader: Drag to adjust the channel gain within the configured MAX/MIN limits.
- Numeric Input: Type a gain value directly to set the level precisely.
- MAX: Sets the maximum allowable gain limit (up to +18 dB).
- MIN: Sets the minimum allowable gain limit (down to −18 dB).

**STEPS:**

**1. Array Microphone Input Channel**

**Mute Function:** Enables array microphone input signal and disables array microphone signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Level Indicator Bar:** Displays the current volume change of the array microphone.

**2. LINE IN1 Input Channel**

**Mute Function:** Enables LINE IN1 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting:** Input a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the LINE IN1 channel.

 | 

**Maximum Value:** Sets the maximum value for volume gain adjustment; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum value for volume gain adjustment; the minimum value cannot be lower than -18.

**3. LINE IN2 Input Channel**

**Mute Function:** Enables LINE IN2 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting:** Input a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the LINE IN2 channel.

**Maximum Value:** Sets the maximum volume gain adjustment value; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum volume gain adjustment value; the minimum value cannot be lower than -18.

**4. AEC IN1 Input Channel**

**Mute Function:** Enables AEC IN1 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting:** Input a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the AEC IN1 channel.

**Maximum Value:** Sets the maximum volume gain adjustment value; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum volume gain adjustment value; the minimum value cannot be lower than -18.

**5. AEC IN2 Input Channel**

**Mute Function:** Enables AEC IN2 channel input signal.

**Level Value:** Real-time display of volume level; not adjustable.

**Volume Setting:** Input a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the AEC IN2 channel.

**Maximum Value:** Sets the maximum value for volume gain adjustment; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum value for volume gain adjustment; the minimum value cannot be lower than -18.

**6. USB IN1 Input Channel**

**Mute Function:** Enables USB IN1 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting Value:** Inputs a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator:** Displays the current volume change of the USB IN1 channel.

**Maximum Value:** Sets the maximum volume gain adjustment value; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum volume gain adjustment value; the minimum value cannot be lower than -18.

**7. USB IN2 Input Channel**

**Mute Function:** Enables USB IN2 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting Value:** Inputs a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the USB IN2 channel.

**Maximum Value:** Sets the maximum volume gain adjustment value, not exceeding 18.

**Minimum Value:** Sets the minimum volume gain adjustment value, not lower than -18.

### 8. DANTE IN1

These two inputs are only available in Dante mode and are the input channels for the Dante module signal.

**Mute Function:** Enables the DANTE IN1 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting Value:** Inputs a number to change the volume gain; the maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; the maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the DANTE IN1 channel.

**Maximum Value:** Sets the maximum volume gain adjustment value, not exceeding 18.

**Minimum Value:** Sets the minimum volume gain adjustment value; the minimum value is not lower than -18.

### 9. DANTE IN2

These two inputs are only available in Dante mode and are the input channels for the Dante module signal.

**Mute Function:** Enables the DANTE IN2 channel input signal.

**Level Value:** Displays the real-time volume level; not adjustable.

**Volume Setting:** Inputs a number to change the volume gain; maximum adjustment range is -18 to 18.

**Gain Fader:** Changes the volume gain; maximum adjustment range is -18 to 18.

**Level Indicator Bar:** Displays the current volume change of the DANTE IN2 channel.

**Maximum Value:** Sets the maximum value for volume gain adjustment; the maximum value cannot exceed 18.

**Minimum Value:** Sets the minimum value for volume gain adjustment; the minimum value cannot be lower than -18.

### 3.4.2 Beam Level

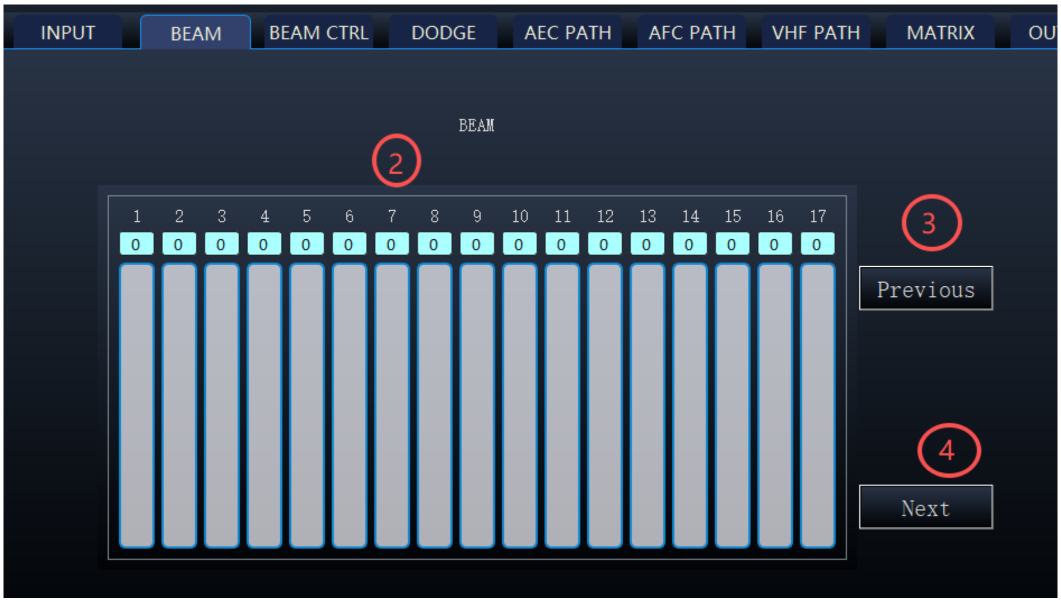



*Figure 3.4.2 — Beam Level*

**1. ANGEL:** HORI (horizontal angle from sound source to device), VERT (vertical angle from sound source to device). Real-time level fluctuations for 33 beams.

2. Beam level display **previous page**.

3. Beam level display **next page**.

### 3.4.3 Beam Control

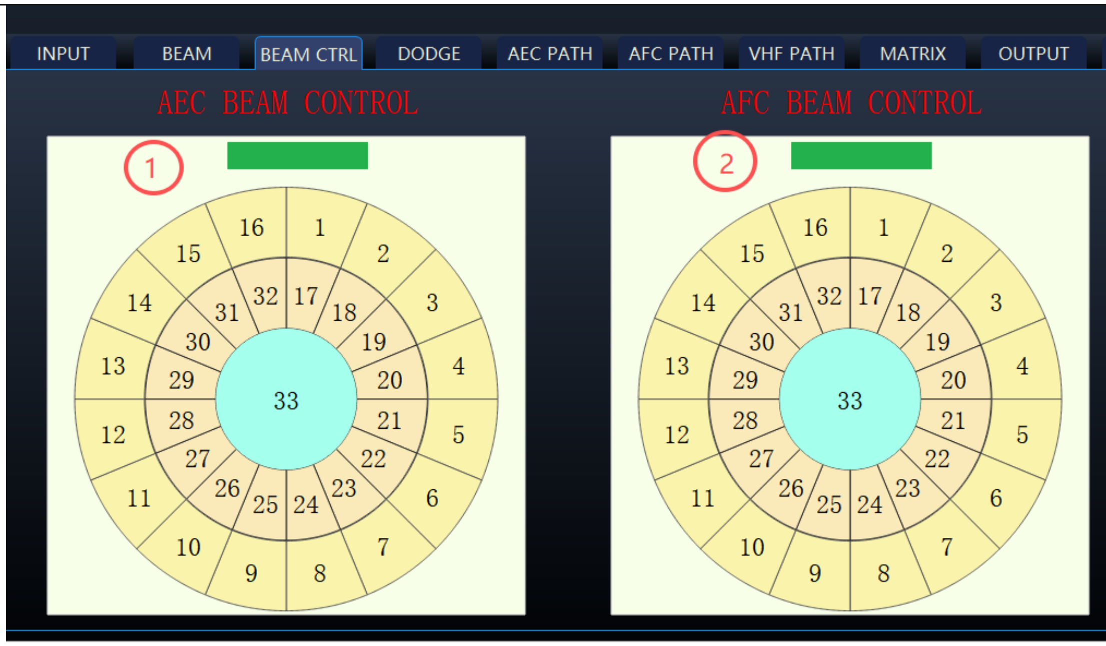



*Figure 3.4.3 — Beam Control*

**1. Echo Path Beam Control:** 33x beam areas in the echo path; gray indicates off, bright indicates on.

**2. Feedback Path Beam Control:** 33x beam areas in the feedback path; gray indicates off, bright indicates on.

## 3.4.4 DODGE — Pathway Selection

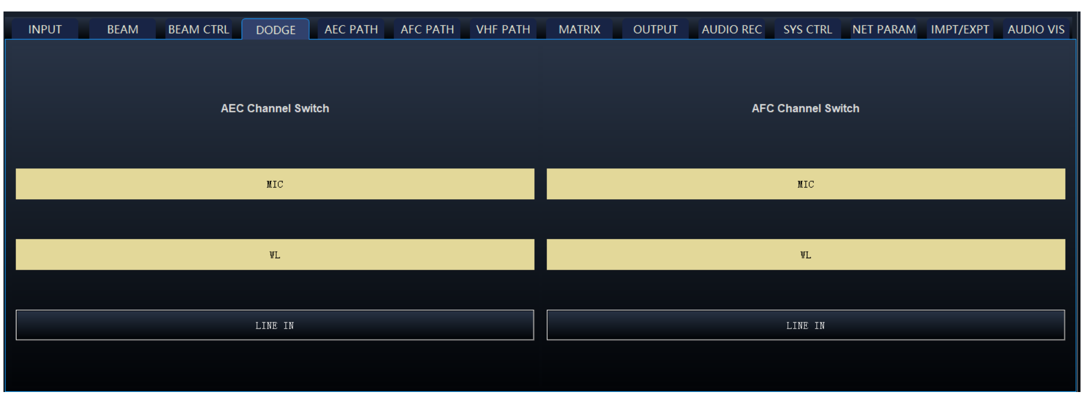



*Figure 3.4.4 — DODGE*

**1. Echo Path Switch:** Clicking the selected channel indicates that the signal from that channel will undergo echo cancellation processing before being output.

**2. Feedback Path Switch:** Clicking the selected channel indicates that the signal from that channel will undergo feedback processing before being output.

## 3.4.5 AEC PATH — Acoustic Echo Cancellation

The **AEC PATH** tab configures the Acoustic Echo Cancellation processing chain. These parameters control the output quality of AEC OUT / USB OUT / DANTE OUT 1 (remote interaction output).


*Figure 3.4.5 — AEC PATH Tab*

**1. Echo Suppression Level:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 0 to 4. The maximum value sets the echo suppression level, with a maximum value not exceeding 4; the minimum value sets the echo suppression level, with a minimum value not lower than 0.

**2. Reverberation Suppression Amount:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 3 to 18; the maximum value sets the level, with a maximum value not exceeding 18; the minimum value sets the level, with a minimum value not lower than 3.

**3. Steady Noise Suppression Amount:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 3 to 30; the maximum value sets the level, with a maximum value not exceeding 30; the minimum value sets the level, with a minimum value not lower than 3.

**4. AI Noise Reduction:** Transient noise suppression amount; the current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 0 to 30; the maximum value sets the level, with a maximum value not exceeding 30; the minimum value sets the level, with a minimum value not lower than 0.

**5. Automatic Maximum Gain:** The current gain level is displayed numerically; the fader can change the gain level, adjustable from 6 to 15; the maximum value setting level is provided, with a maximum value not exceeding 15; the minimum value setting level is provided, with a minimum value not lower than 6.

**6. 13-band Equalizer Adjustment:** Click and drag the label up or down to change the amplitude of that frequency band; gain function: directly input the gain parameter to change the amplitude of that frequency band; pushing the fader can also change the amplitude of that frequency band.

| Parameter | Range / Values | Description |
|---|---|---|
| **AEC Level** | 0 – 4 (Max: 4, Min: 0) | Acoustic echo cancellation depth. Increase when the far-end participant reports hearing their own voice echoed back. Default level suits most rooms. |
| **REVERB Level** | 3 – 18 (Max: 18, Min: 3) | Reverberation suppression strength. Increase in acoustically reflective environments (open spaces, hard surfaces). Decrease in treated rooms to preserve voice naturalness. |
| **SNS Level** | 3 – 30 (Max: 30, Min: 3) | Stationary Noise Suppression level. Removes continuous background noise such as HVAC, fans, and projectors. |
| **Denoise (AI)** | 0 – 30 (Max: 30, Min: 0) | AI-powered transient noise suppression. Reduces impulse and non-stationary noise events. Higher values suppress more noise but may affect voice clarity. |
| **AGC Level** | 6 – 15 (Max: 15, Min: 6) | Automatic Gain Control ceiling. Maintains consistent output level by dynamically adjusting gain. |
| **ANGLE** | 180° or 360° | Pickup beam angle for the AEC processing path. 180° = directional, 360° = omnidirectional. |

| | | |
|---|---|---|
| **13-Band EQ** | Gain per frequency band | Parametric equalizer with 13 frequency bands spanning 50 Hz – 20 kHz. Drag band markers up/down to adjust amplitude; type gain values directly; or use the faders. |
| **Reset** | Button | Resets all AEC PATH parameters to factory default values. |

> **NOTE:** *Commissioning tip: Higher REVERB, SNS, and Denoise values reduce noise but degrade voice naturalness. In acoustically treated rooms, leave suppression values at defaults and only adjust gain.*

## 3.4.6 AFC PATH — Anti-Feedback Control

The AFC PATH tab configures the Anti-Feedback Control processing chain. These parameters control the output quality of SPK OUT / DANTE OUT 2 (local amplification output).

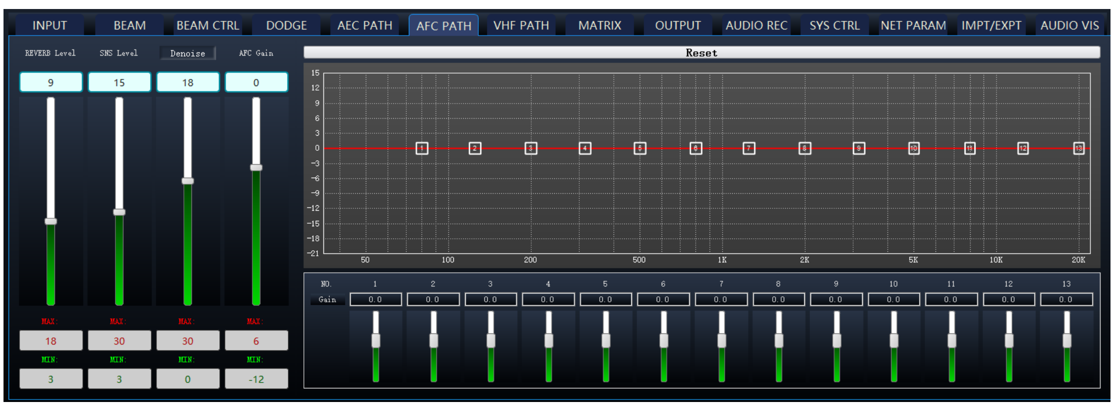



*Figure 3.4.6 — AFC PATH Tab*

**1. Feedback Suppression Level:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 0 to 4; the maximum value sets the echo suppression level, with a maximum value not exceeding 4; the minimum value sets the echo suppression level, with a minimum value not lower than 0.

**2. Reverberation Suppression Amount:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 3 to 18; the maximum value sets the level, with a maximum value not exceeding 18; the minimum value sets the level, with a minimum value not lower than 3.

**3. Steady Noise Suppression Amount:** The current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 3 to 30; the maximum value sets the level, with a maximum value not exceeding 30; the minimum value sets the level, with a minimum value not lower than 3.

**4. AI Noise Reduction:** Transient noise suppression amount; the current level is displayed numerically; the level can be adjusted using the fader, with an adjustment range of 0 to 30; the maximum value sets the level, with a maximum value not exceeding 30; the minimum value sets the level, with a minimum value not lower than 0.

**5. Automatic Maximum Gain:** Displays the current gain level; the fader can adjust the gain level from 6 to 15; a maximum value is set, not exceeding 15; a minimum value is set, not lower than 6.

**6. Feedback Gain:** Displays the current gain level; the fader can adjust the gain level from -12 to 6; a maximum value is set, not exceeding 6; a minimum value is set, not lower than -12.

**7. Reinforcement Angle:** Displays the current angle; the fader can adjust the angle, ranging from 180 degrees to 360 degrees.

8. For the **13-band equalization adjustment**, click on the label and drag it up or down to change the amplitude of that frequency band; for the gain function, directly input the gain parameter to change the amplitude of that frequency band; moving the fader will also change the amplitude of that frequency band.

| Parameter | Range / Values | Description |
|---|---|---|
| **AFC Level** | 0 – 4 (Max: 4, Min: 0) | Anti-feedback suppression depth for the local amplification output. |
| **REVERB Level** | 3 – 18 (Max: 18, Min: 3) | Reverberation suppression strength for the local amplification signal. |
| **SNS Level** | 3 – 30 (Max: 30, Min: 3) | Stationary noise suppression level for the AFC path. |
| **Denoise (AI)** | 0 – 30 (Max: 30, Min: 0) | AI transient noise suppression for local amplification output. |
| **AFC Gain** | −18 to +6 dB | Output gain for local amplification. Controls the speaker |

| | | |
|---|---|---|
| | (Max: +6, Min: −18) | volume. Excessive values cause audio feedback (howling). Commission by gradually increasing from minimum. This value is linked to the remote control volume buttons — set a safe upper limit to prevent over-amplification. |
| **ANGLE** | 180° or 360° | Amplification beam angle: 180° (half-space / directional) or 360° (full space / omnidirectional). |
| **13-Band EQ** | Gain per frequency band | Parametric EQ for the local output path. 13 bands from 50 Hz – 20 kHz. Same controls as AEC PATH EQ. |
| **Reset** | Button | Resets all AFC PATH parameters to factory default values. |

> **NOTE:** *Commissioning tip: Increase AFC Gain gradually during initial setup. Reduce immediately if feedback (howling) occurs. For EQ: if the room has noticeable echo/reverberation, reduce EQ bands above 1 kHz. If audio sounds thin, add a slight low-frequency boost.*

## 3.4.7 MATRIX


*Figure 3.4.7 — MATRIX*

**1. INPUT CHANNELS:**

**USB IN:** USB input channel.

**DANTE IN1:** Dante mode remote interactive input channel.

**DANTE IN2:** Dante mode local amplification input channel.

**WL/LINE IN:** Line-in input channel, mainly for computers or other line-out devices.

**AEC IN:** Echo reference input channel.

**AEC CHANNEL:** Signal after echo cancellation, an internal software channel with no structured interface definition.

**AFC CHANNEL:** Signal after feedback suppression, an internal software channel with no structured interface definition.

### 2. OUTPUT CHANNELS:

AEC OUT1, AEC OUT2, SPK OUT1, SPK OUT2, USB OUT1, USB OUT2, Dante OUT1, Dante OUT2.

## 3.4.8 OUTPUT — Output Settings

The **OUTPUT** tab provides gain, mute, and level controls for the two main output channels.


*Figure 3.4.8 — OUTPUT Tab*

### AEC OUT Output Channel

① **Mute Function:** Enables AEC OUT channel input signal.

② **Audio Output:** Enables audio output (yellow), indicating feedback functionality; disables audio output (gray), indicating normal AEC OUT output according to channel definition.

③ **Level Value:** Real-time display of volume level, not adjustable.

④ **Volume Display:** Displays current volume gain.

⑤ **Level Display Bar:** Displays the current volume level change of the output channel.

⑥ **Gain Fader:** Adjusts volume gain; maximum adjustment range is -18 to 18.

⑦ **Maximum Value:** Sets the maximum volume gain adjustment value. Input a number to change the maximum value; the maximum value cannot exceed 18.

⑧ **Minimum Value:** Sets the minimum volume gain adjustment value. Input a number to change the minimum value; the minimum value cannot be lower than -18.

⑨ **Output Decibels:** Four values are selectable: 0dB, -12dB, -18dB, and -24dB. After selecting, click the "Set" button to apply.

*Note: The settings for SPK OUT, USB OUT, and DANTE OUT are the same as above.*

| Parameter | Range / Values | Description |
|---|---|---|
| **AEC-OUT** | Gain: −18 to +18 dB | Remote interaction output (post-echo-cancellation signal). Feeds the AEC OUT 1/2 physical connectors, USB OUT, and DANTE OUT 1. |
| **SPK-OUT** | Gain: −18 to +18 dB | Local amplification output (post-anti-feedback-processing signal). Feeds SPK OUT 1/2 physical connectors and DANTE OUT 2. |
| **Mute** | Toggle ON/OFF | Mutes the output channel. Green = active/unmuted. |
| **Audio Output Mode** | Yellow / Gray toggle | Yellow: output signal is processed through the AFC anti-feedback path. Gray: standard output per the defined channel function. |
| **Level Meter** | Read-only | Displays the real-time output signal level. |
| **Gain Fader** | −18 to +18 dB | Adjusts the channel output gain within the configured MAX/MIN limits. |

| | | |
|---|---|---|
| **MAX / MIN** | Input fields | Sets the maximum (+18 dB limit) and minimum (−18 dB limit) for the output gain adjustment range. |
| **Output dB Preset** | 0 / −12 / −18 / −24 dB | Selects a preset output attenuation level. Click the Set button to apply the selected preset. |

### 3.4.9 AUDIO REC — Audio Recording

The **AUDIO REC** tab enables simultaneous multi-channel audio recording from the device for diagnostic analysis, system tuning, or archival purposes.

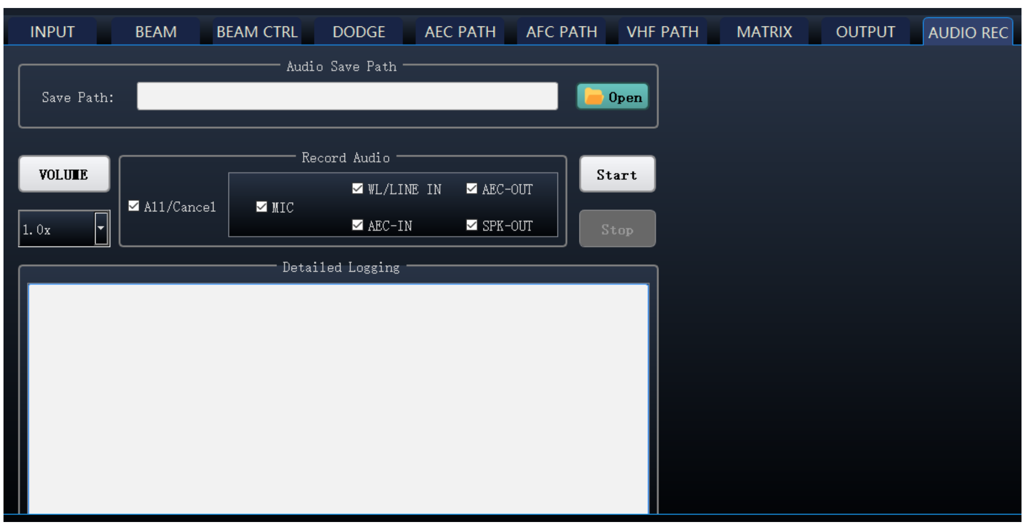



*Figure 3.4.9 — AUDIO REC Tab*

**1. Audio Source Save Path:** Click to open and set the path where recorded audio files will be saved on your computer.

**2. Recording Audio Source:** There are 5 audio sources to choose from for recording. The default is to record all. Check the boxes to select the audio sources you want to record. After selecting the audio sources, click "Start" to begin recording. Click "Stop Recording" to stop recording, and the audio files will be saved in the folder you set on your computer.

**3. Detailed Log Recording:** Records some information about the device's operation.

| Parameter | Range / Values | Description |
|---|---|---|
| **Audio Save Path** | Folder path | Click Open to browse and select the destination folder on the PC where recorded audio files will be saved. |
| **VOLUME** | Multiplier (e.g. 1.0x) | Playback volume multiplier for monitoring recorded audio. |
| **All/Cancel** | Checkbox | Selects or deselects all available recording channels simultaneously. |
| **MIC1/2/WL-I** | Checkbox | Records the MIC1, MIC2 array microphone channels and the WL-MIC input channel. |
| **PC-IN** | Checkbox | Records the PC/LINE IN input channel. |
| **AEC-OL** | Checkbox | Records the AEC output level monitor signal. |
| **AEC-I** | Checkbox | Records the AEC reference input signal. |
| **SPK-OL** | Checkbox | Records the SPK output level monitor signal. |
| **Start / Stop** | Buttons | Start: begins recording all selected channels simultaneously. Stop: ends recording and saves the audio files to the configured path. |
| **Detailed Logging** | Log display area | Displays real-time device operational log information during the recording session. |

### 3.4.10 SYS CTRL — System Control

The **SYS CTRL** tab provides device management functions including firmware upgrade, diagnostic log retrieval, and system reset.


*Figure 3.4.10 — SYS CTRL Tab*

**1. SN Setting:** Enter the actual SN code according to your needs and click "Write SN Code." A pop-up message will indicate successful SN code modification after successful writing.

**2. Upgrade:** Click to open, locate the save path of the firmware to be upgraded, select it, open it, and click "Start Upgrade." A pop-up message will indicate successful upgrade! The device will automatically restart; please search for the device again.

**3. Log Crawling:** Click to open, set the capture destination and save path on your computer, and click "Grab Logs."

**4. Log Settings:** There are two log modes: terminal output and file writing; there are four daily log levels: INFO, DEBUG, WRN, and ERR.

**5. Restart:** Reboot a normally functioning device.

**6. Reset:** Restore the parameters to the factory default configuration.

| Parameter | Range / Values | Description |
|---|---|---|
| **SN Setting / Write SN** | Text input + button | Enter the device serial number in the text field and click Write SN to write it to the device. A confirmation dialog appears on success. |
| **Upgrade (Firmware)** | File path + button | Click Open to locate the firmware upgrade file (.bin) on the PC. Select the file, then click Upgrade to begin the upgrade process. A success dialog appears and the device restarts automatically. Use Search to rediscover the device after the upgrade. |

| | | |
|---|---|---|
| **Log Crawling / Grab Logs** | File path + button | Click Open to select the destination folder for log files. Click Grab Logs to retrieve and save the current device diagnostic log. |
| **Speaker Selecting** | Dropdown + Setting btn | Selects the speaker output configuration mode (e.g., ONE). Click Setting to apply the selection. |
| **Log Mode** | Dropdown + Setting btn | Sets the diagnostic log output mode: Output (terminal display) or Write File. Log levels: INFO, DEBUG, WRN (Warning), ERR (Error). Click Setting to apply. |
| **Restart** | Button | Restarts the device software while preserving all current parameter settings. |
| **Reset** | Button | Performs a factory reset — restores all parameters to their factory default values. All configured settings will be overwritten. |

> **NOTE:** *After performing a firmware upgrade or factory reset, the device will restart automatically. Use the Search function in the software to rediscover the device on the network.*

### 3.4.11 NET PARAM — Network Parameters

The **NET PARAM** tab configures the device IP address, subnet mask, and gateway for network communication.

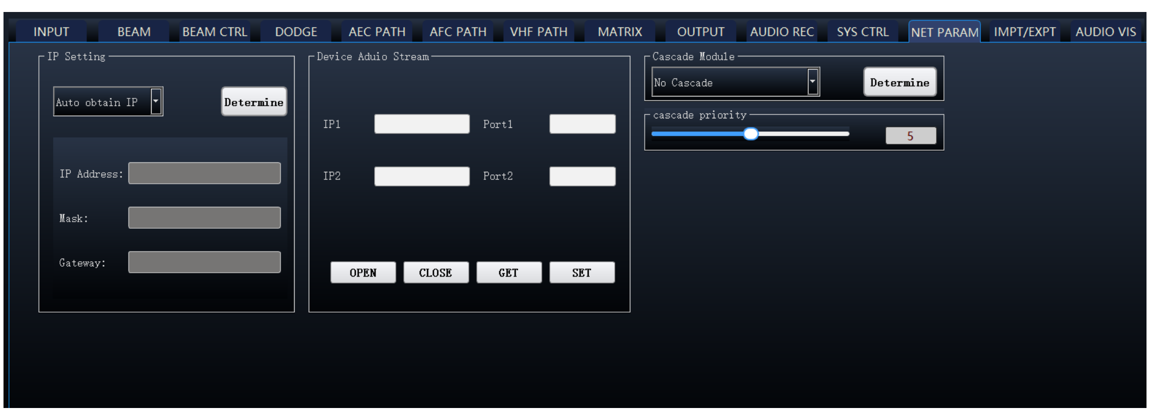



*Figure 3.4.11 — NET PARAM Tab*

**1.** Set to **automatically obtain** an IP address; the IP address cannot be manually changed (default IP: 192.168.1.100).

2. When setting to use the **following IP address**, the IP address can be manually changed.

**3. IP Address:** Enter the desired IP address, such as 192.168.1.3.

**4. Subnet Mask:** Enter the subnet mask, such as 255.255.255.0.

**5. Default Gateway:** Enter the gateway, such as 192.168.1.1.

After entering the IP address, subnet mask, and gateway, click OK. A pop-up message will indicate a successful upgrade! The device will automatically restart; please search for the device again.

### Cascade & Audio Streaming Settings

Additional advanced settings available in the NET PARAM section:

Set the corresponding device as the master or slave microphone and click OK.

- **Audio Streaming:** Enter the target device IP address and select the streaming port to enable audio network streaming output.
- **Cascade Settings:** Assign the device role as Master Mic or Slave Mic, then click Confirm. After setting the master device, all connected slave device IP addresses will be listed.

| Parameter | Range / Values | Description |
|---|---|---|
| **IP Mode** | Dropdown (Auto / Manual) | Auto obtain IP: device receives its IP address via DHCP. The IP address fields are read-only in this mode (default: 192.168.1.100). Switch the dropdown to the static IP option to enable manual configuration. |
| **IP Address** | e.g. 192.168.1.3 | Static IP address to assign to the device. |
| **Mask** | e.g. 255.255.255.0 | Subnet mask for the device network. |
| **Gateway** | e.g. 192.168.1.1 | Default gateway address for the device network. |

| | | |
|---|---|---|
| **Determine** | Button | Applies the configured IP settings. A confirmation dialog appears on success, and the device restarts automatically. Use Search to rediscover the device at its new IP address. |

## 3.4.12 IMPT/EXPT — Import / Export Configuration

The **IMPT/EXPT** tab enables saving and loading of device configuration presets. The device supports up to 10 independent configuration scenes.


*Figure 3.4.12 — IMPT/EXPT Tab*

**1. Export Configuration:** Click "Open" set the path to save the exported file on your computer. Click the down arrow next to the current configuration, select the scene you want to export, and finally click "Export Configuration." A pop-up message will indicate that the configuration export was successful.

**2. Import Configuration:** Click "Open" and locate the file saved on your computer. Click the selected file; click the down arrow next to the current configuration, select the scene to import it into, and finally click "Import Configuration"; a pop-up message will indicate successful configuration import.

**3. Import File:** Click "Open" and locate the file saved on your computer. Click the selected file; click "Import File"; a pop-up message will indicate successful configuration import.

**4. Scene Switching:** There are 10 scenes in total; the default is the Clear Mode scene.

**5. Scene Saving:** Saves the scene after debugging.

**6. Scene Application:** After selecting a scene, click this button to switch scenes.

| Parameter | Range / Values | Description |
|---|---|---|
| **Export Configuration** | Path + scene + Export btn | Click Open to select the destination folder. Use the scene dropdown (CURRENT or a numbered scene) to select which configuration to export. Click Export. A confirmation dialog confirms success. |
| **Import Configuration** | Path + scene + Import btn | Click Open to navigate to the saved configuration file. Use the scene dropdown to select the target scene slot for import. Click Import. A confirmation dialog confirms success. |
| **Import File** | Path + Import btn | Click Open to locate a raw device configuration file. Click Import to load it directly into the device. |
| **Scene Selector** | 1 – 10 scenes (ONE default) | Selects one of 10 available configuration scene slots. The default scene is labeled CURRENT (Clear Mode). |
| **Save** | Button | Saves the current active parameter configuration to the selected scene slot. |
| **Apply** | Button | Activates the selected scene configuration immediately on the device. |

## 3.4.13 AUDIO VISUAL- Camera tracking


*Figure 3.4.13 (1) — AUDIO VISUAL*

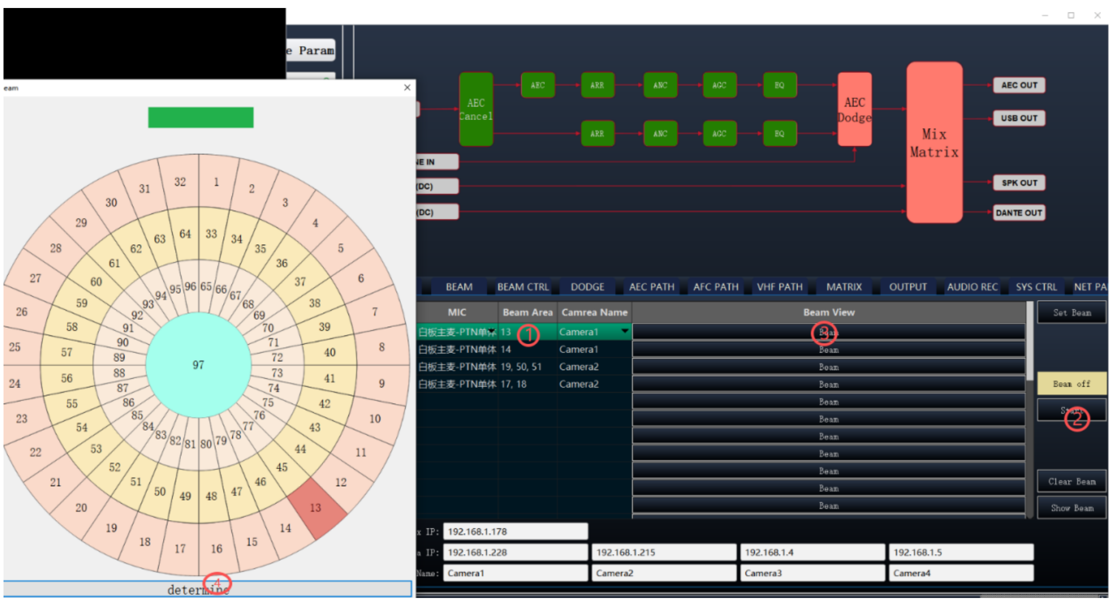



*Figure 3.4.13 (2) — Camera tracking settings*

**1. Predefined positions:** Select from seats 1-30, corresponding to network camera predefined positions 0-29 (for a single camera). When using two or more cameras on-site, the predefined position of the second camera continues for any unused seats. For example, Camera 1 covers seats 1–10, Camera 2 covers seats 11–20, and so forth.

**2. MIC:** Select the microphone for camera tracking and bind other microphones for tracking.

**3. Signal pickup area:** Displays the bound beam number.

1) Select the audio pickup area to display green

2) When you start speaking, the system identifies your seat and automatically displays the corresponding beam; tapping stops speech recognition.

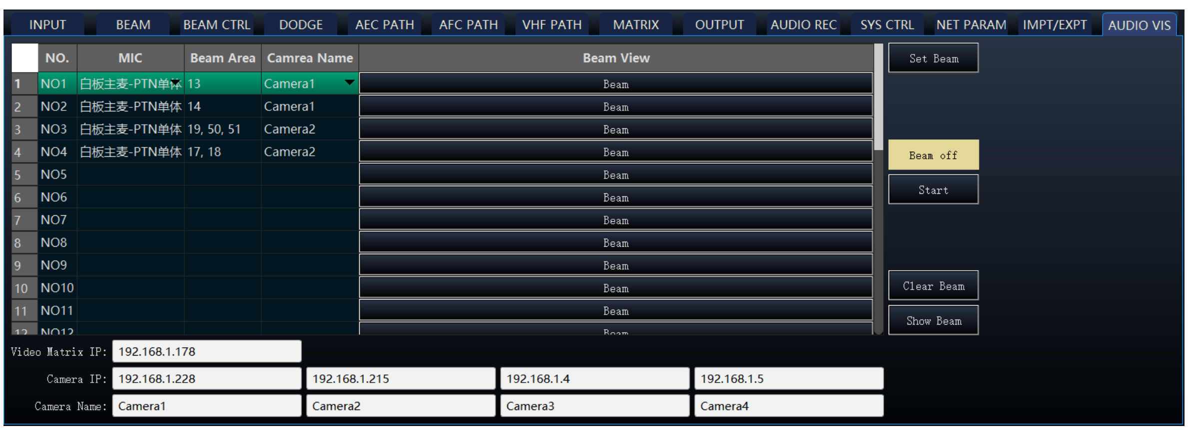
3) Click to view or modify to display the corresponding area and manually adjust the illuminated beam.

4) Confirm save. This binds the sound beam as shown below.

**4. Camera names:** Factory defaults are Camera1, Camera2, Camera3, and Camera4; you can choose accordingly. In the blank field below, customize camera names and IP addresses. After entering them, click Enter to confirm with a success message. Note that cameras and processors must share the same network segment.

**5. For setups requiring two or more cameras**, add a matrix switch to coordinate image switching. Simply enter the matrix IP address; the control code is pre-written at the system level and cannot be modified. (Third-party matrices must confirm support for TCP/IP control and control codes; alternatively, contact the manufacturer for customization.)

**6. After setup, click to enable audio-video synchronization** (this completes the operation and allows you to test the effect).




**7.** After manually adjusting the beam binding area, click **Bind Beam** on the right.

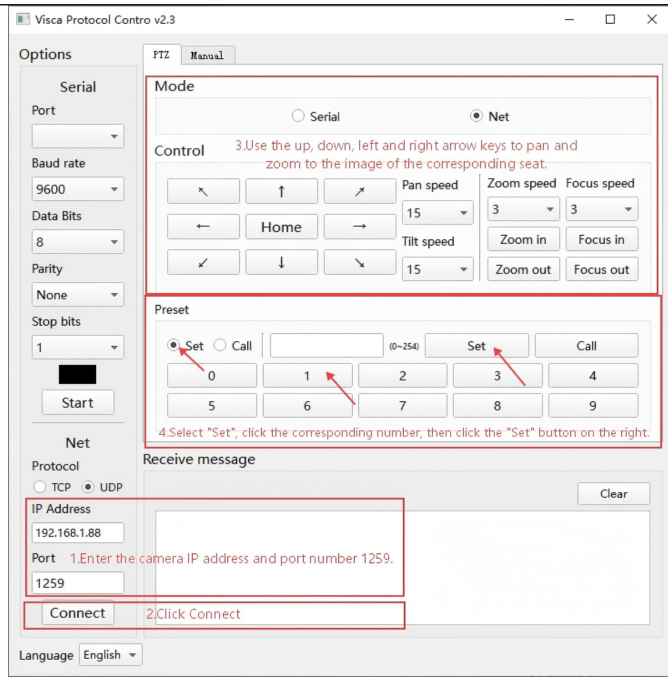
**8. Clear with One Click:** deletes all linked information.

**9. Beam Distribution:** Shows binding information for all seats, beams, and cameras.

**10. Set up preset positions** on the camera (you can also use the built-in remote control):

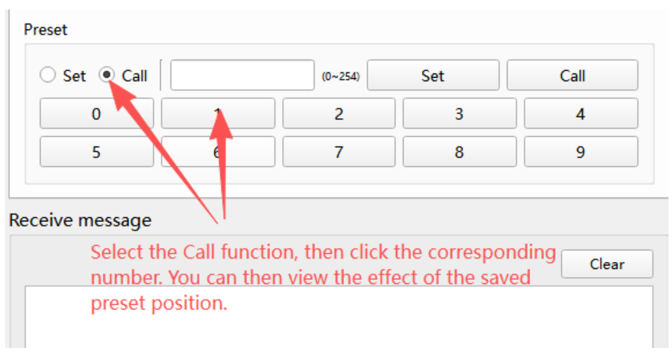
1) Operate using the Visca control software, as shown in the figure.




2) After setting the preset values sequentially, the values 0–9 correspond to seat numbers 1–10 in the software.

3) Call the number to see the result and confirm if the setup was successful, as shown in the figure.




4) If the zero preset cannot be activated, you can reset it (some cameras require final configuration of the zero preset).

# 4. Commissioning Quick Reference

Recommended parameter settings for common deployment scenarios.

## 4.1 Local Sound Reinforcement

For local speaker amplification, configure the **AFC PATH** parameters:

**Pure Amplified Signal Routing Configuration Sequence:** Output channel 2 of the last cascaded device => Input channel 2 of the previous stage, and so on until it reaches the cascade master device. The master device's output channel 2 (SPK OUT) => Other devices receiving Dante signals; then, the output channel 2 of the previous stage device must be used as the input channel 2 of the next stage device.

| Parameter | Commissioning Recommendation |
|---|---|
| **REVERB Level** | Increase for open, acoustically reflective spaces (e.g., large halls, conference rooms with glass walls). Decrease in acoustically treated rooms to preserve voice naturalness. |
| **AFC Gain** | Start at minimum and increase gradually. Reduce immediately if audio feedback (howling) occurs. Set a safe MAX limit to prevent over-amplification via the remote control. |
| **EQ (above 1 kHz)** | If the room has noticeable reverberation tail, reduce the high-frequency EQ bands (above 1 kHz) slightly. |
| **EQ (low frequencies)** | If the audio sounds thin or lacking body, add a moderate boost to the low-frequency EQ bands. |

## 4.2 Remote Conferencing / AEC Path

For remote interaction **(AEC path)**, the same principles apply using AEC PATH parameters:

**Pure Interactive Signal Routing Configuration Sequence:** Output channel 1 of the last cascaded device => Input channel 1 of the previous stage, and so on until it reaches the cascade master device. The master device's output channel 1 (AEC OUT) => Input channels of other devices receiving Dante signals; then, the output channel 1 of the previous stage device must be used as the input channel 1 of the next stage device; Other device output channels for Dante signals => Input channel 1 of the master device.

- **AEC Level:** Increase when the far-end reports hearing their own voice echoed back.
- **REVERB Level, SNS Level, Denoise:** Apply the same logic as section 4.1 above.
- **EQ tuning** principles for AEC OUT are identical to AFC PATH EQ.

> **NOTE:** *General principle: Suppression parameters (REVERB, SNS, Denoise) reduce noise artifacts but degrade voice naturalness at high values. In acoustically treated environments, minimize suppression and focus tuning on gain levels only.*

# 5. Troubleshooting

| Symptom | Cause / Solution |
|---|---|
| **Software cannot connect / Device not found in search** | Verify the PC firewall and antivirus software are disabled. These applications commonly block the device discovery protocol. Disable them and retry the Search function. |
| **Input level meters show no activity, but audio is present in the room** | Multiple software instances may be running simultaneously. Close all instances and reopen the software once. Also verify the device is properly connected and the correct IP is selected. |
| **Local amplification feedback / howling** | The connected amplifier or speaker output level is too high, or the AFC Gain is set too high. Reduce AFC Gain incrementally. If the issue persists, use the IMPT/EXPT tab to restore factory defaults for the mix matrix configuration. |
| **No audio output from the device** | Verify the output cable is connected to the SPK OUT port (not AEC OUT). Check the device LED: green or amber LED = normal operation. Confirm that the amplification mode is enabled in the software. |
| **Connected audio has accompaniment but no vocal audio** | The source device outputs an unbalanced 3.5mm signal, but it is wired using a balanced connector wiring scheme. This causes the left and right channels to cancel each other (phase inversion). Rewire using the correct unbalanced (mono) connection method. |
| **Dante Controller cannot discover the device** | The device Dante interface and the control PC are on different network subnets. Open Dante Controller, view the device information to check the current Dante IP address, and update it to match the PC's subnet. |

# 6. Product Specifications

## 6.1 Algorithm Specifications

| Parameter | Specification |
|---|---|
| **Supported Algorithms** | AFC, ANS, AEC, AGC, ARR |
| **Feedback Gain Suppression** | ≥ 18 dB |
| **Noise Reduction** | ≥ 30 dB |
| **Echo Cancellation Depth** | ≥ 90 dB |
| **Echo Tail Cancellation Length** | ≥ 1 second |
| **Reverberation Suppression** | ≥ 18 dB |
| **Maximum Gain** | ≥ 30 dB |
| **AI Noise Reduction** | Supported |
| **Automatic Mixing (Auto Mix)** | Supported |
| **Beamforming** | Supported |
| **Sound Source Localization** | Supported |

## 6.2 Microphone Specifications

| Parameter | Specification |
|---|---|

| | |
|---|---|
| **Number of Microphone Elements** | 129 |
| **Sensitivity** | −32 ± 2 dB |
| **Signal-to-Noise Ratio** | 70 dB |
| **Frequency Response** | 75 Hz – 20 kHz |

## 6.3 Hardware & System Specifications

| Parameter | Specification |
|---|---|
| **Sample Rate** | 48 kHz |
| **Frequency Response** | 20 Hz – 20 kHz, ±0.5 dB |
| **Signal-to-Noise Ratio** | 100 dB |
| **Total Harmonic Distortion (THD)** | ≤ 0.1% |
| **Input Impedance (Balanced)** | 20 kΩ |
| **Output Impedance (Balanced)** | 200 Ω |
| **Maximum Input Level (Balanced)** | +4 dBu |
| **Maximum Output Level (Balanced)** | +10 dBu |
| **Dante Audio Networking** | Supported |
| **Unit Cascade** | Supported |
| **RS-485 Control Interface** | Supported |

| | |
|---|---|
| **Power Input** | DC 12V – 48V / PoE+ |
| **Power Consumption** | 25 W |
| **Dimensions (L × W × H)** | 600 × 600 × 42.8 mm |
| **Net Weight** | 6 kg |
| **Operating Temperature** | 0°C – 40°C |
| **Storage Temperature** | −20°C – 60°C |
| **Color** | Pearl White |

*For technical support, visit: www.infobitav.com | info@infobitav.com*