

Prestel VCS-MA8D

Ceiling Microphone Array Operation Instructions



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1. Log in

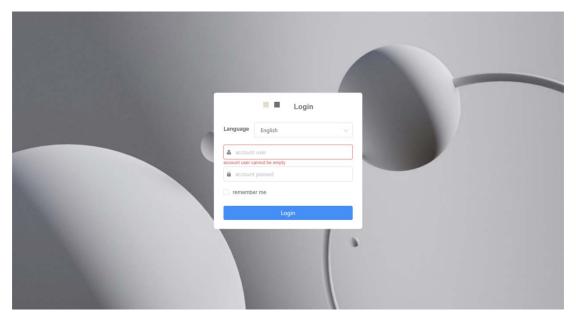


Open a browser and enter the IP address in the address bar to access.



The default IP address is:

192.168.1.80



Username: admin Password: admin123

Notice

- Before visiting, please ensure that the IP address of your computer is in the same subnet and that the network is functioning properly.
- If you do not have an address in the 192.168.1.X subnet, please set or add at least one such address in the network adapter first.

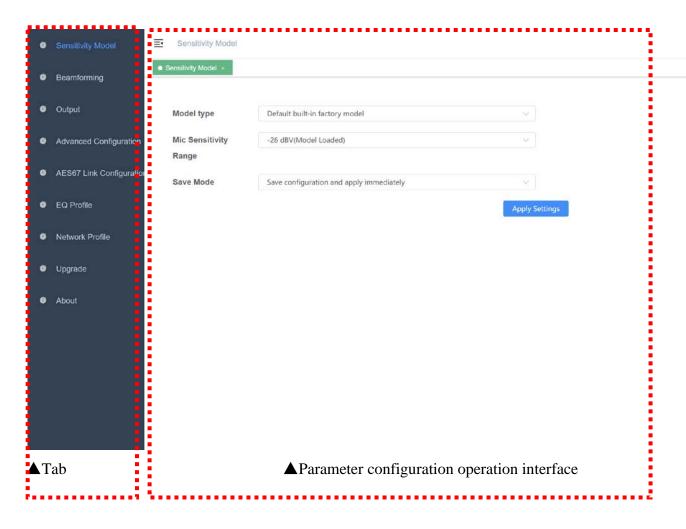
Taking the setup of Windows 10 as an example.

- Network and Internet Settings -> Change Adapter Options -> Right-click on the computer's network card (network adapter) -> Properties -> Select "Internet Protocol Version 4 (TCP/IPv4)" -> Properties -> Advanced -> Click Add in the IP address field -> Enter the IP address such as "192.168.1.123" and subnet mask "255.255.255.0" -> Add -> OK, close all open windows to complete the setup.
- Alternatively, in the "Internet Protocol Version 4 (TCP/IPv4)" properties, select "Use the following IP address" and add an address in the 192.168.1.X subnet, such as "192.168.1.123" with the default subnet mask of 225.225.225.0. Click OK, then close all open windows to complete the setup.

Note: You can select the interface language on the login screen.

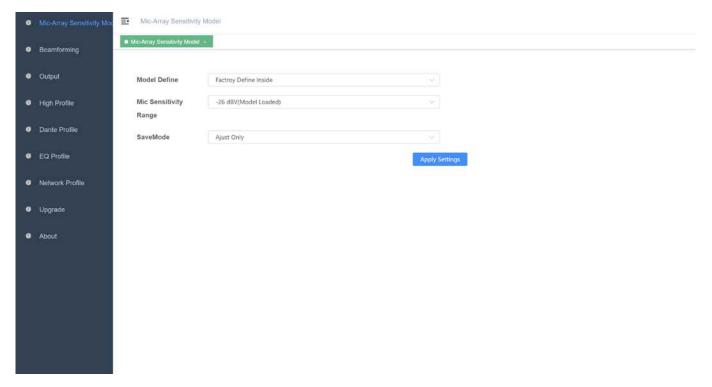
2. Web Settings

The left side is the tab area, and right side is the operation interface.



3. Setup and Detailed Explanation

(1) Array Sensitivity Model



Array Sensitivity Model

Model Type

Default Tone: This model adapts to most scenarios, offering a clear and natural listening experience with high audio fidelity and a comfortable auditory sensation.

Transparent Tone: This model is more suitable for classrooms, training rooms, etc. It is not suitable for environments with a lot of reverberation as it may make the audio sound mixed.

Sharp Tone: This model is suitable for environments with a lot of reverberation and long reverberation tails, focusing more on high frequencies in speech, better adapting to spaces with strong reverberation.

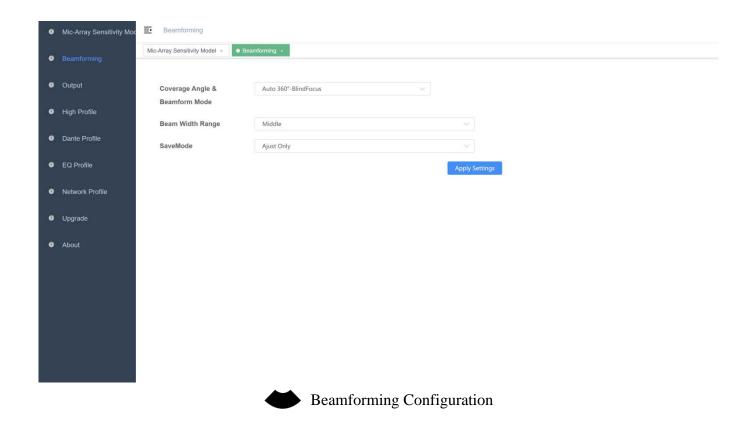
MIC Optimal Spatial Model Range

Microphone sensitivity setting, default at -26dBv, the smaller the value, the lower the sensitivity. Choose according to the on-site environment.

Save Method: Effective for this session only/Save Configuration/Save Configuration & Apply Immediately

Apply Settings: Confirm the current settings and apply them according to the save method.

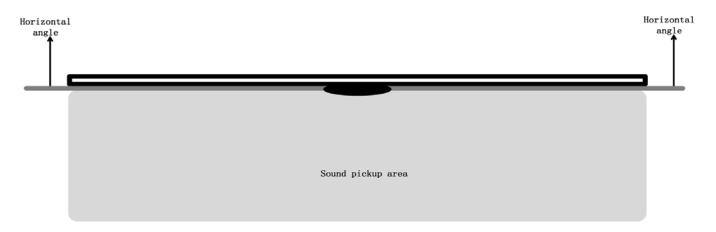
(2) Beamforming Setup



Pickup Angle & Beamform Mode

Omnidirectional Beam Tracking Array

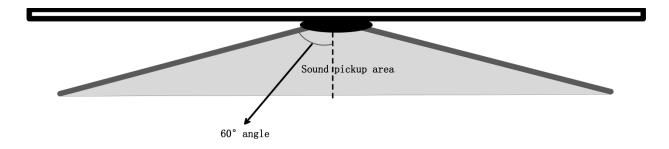
Application: Default option, emphasizes horizontal directional pickup.



(Pickup Area: 180° - area indicated by the angle)

Omnidirectional Beam Tracking Array – 60° Elevation

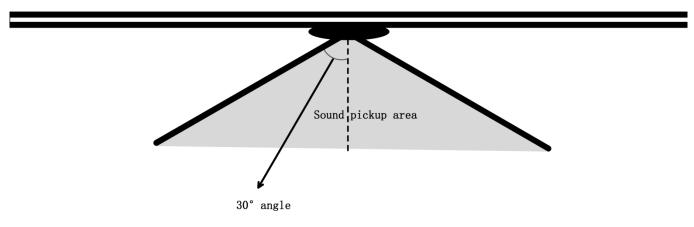
Application: The gray area represents the pickup zone, with a total angle of 120°, comprised of 60° on each side of the microphone's vertical center point. The system prioritizes sound tracking within a 60m² area while actively blocking noise interference from outside the pickup angle.



(Pickup Area: 60° - area indicated by the angle)

Omnidirectional Beam Tracking Array - 30° Elevation

Application: The gray area represents the pickup zone, with a total angle of 60°, comprised of 30° on each side of the microphone's vertical center point. The system prioritizes sound tracking within a 12m² area while actively blocking noise interference from outside the pickup angle.



(Pickup Area: 30° - area indicated by the angle)

Omnidirectional Beam Tracking Array Aggregation

Application: Focuses on low-frequency clarity with less high-frequency feedback.

Omnidirectional Beam Tracking Array Aggregation Beta

Application: Focuses on low-frequency clarity with less high-frequency feedback.

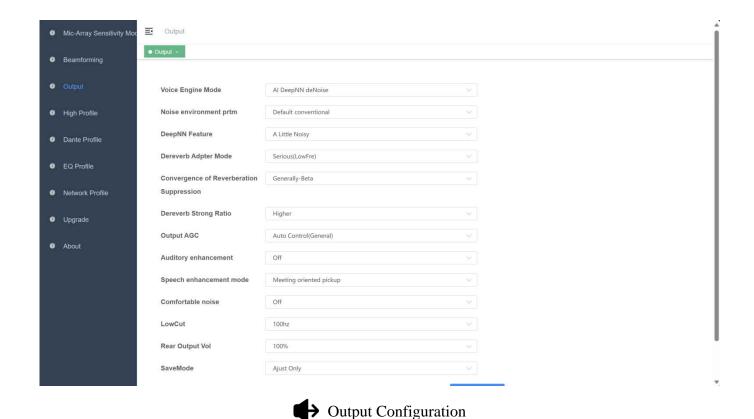
The Beta beam theoretically achieves a 60° downward sound pickup curtain, which can partially shield against reflected sound in the space.

Save Mode: Only Effective Now / Save Configuration / Save Configuration & Apply Immediately

Settings: Confirm current settings according to the save mode.

Notes: The pickup area angle coefficient is a reference theoretical value, not an absolute angle.

(3) Output Configuration



Environmental Noise & Voice Enhancement Mode

AI Neural Network Model for Noise Reduction (AI Smart Noise Reduction Algorithm)

Noise Environment Preprocessing

Select noise reduction level based on overall environmental noise (None/Quiet/Default Normal/Poor/Severe/Extremely Severe)

Process steady-state noise levels in the environment and adjust settings accordingly. (Generally, the Default Normal can cover most scenarios; choose a higher level if the onsite noise is severe.)

Noise Reduction Feature Context Processing

Select AI noise reduction level based on environmental noise (None/Level 1/Level 2/Level 3/Level 4/Level 5)

Choose the AI noise reduction level based on the onsite noise environment (default is Level 2 noise reduction; increase level if there is significant noise onsite).

Reverberation Suppression Switch & Adaptive

Configure onsite reverberation: (Off/Low/Moderate/High (Low Frequency Focused)/High (Mid Frequency Focused)/High & Small Dynamic Range)

Choose a suitable or high reverberation suppression level based on the onsite environment.

Reverberation Suppression Convergence

Fast: fastest converging reverberation suppression, suitable for relatively quiet sites with significant reverberation reflection.

Balanced: basic configuration, suitable for most scenarios.

Balanced Alpha: based on the default reverberation suppression algorithm, adjusts the division of

reverberation frequency domain, allowing increased sharpness of reverberation & sound)

Balanced Beta: reverberation reflection calculation uses an updated model.

De-reverberation Intensity

Configure onsite reverberation environment: (Slight/Low/Medium/High/Maximum), default set to Medium, increase level if reverberation is significant onsite. Reduce level if the acoustic environment is excellent.

Output Gain Control

Adjust microphone volume levels to (General/Moderate/Large) based on onsite recording conditions.

Auditory Enhancement

Enhances mid to high-frequency harmonics when activated on dynamic listening devices to improve audio quality.

Voice Enhancement Mode

For Environmental Monitoring: Used in monitoring scenarios to pick up detailed environmental sounds, making the sound more distinct.

For General Environments: Used in regular scenarios.

Comfort Noise

Adds comfort noise to the audio.

Low Cut Configuration

Adjust microphone to cut off the low-frequency portion of the audio (effective in conference mode).

Post Output Volume

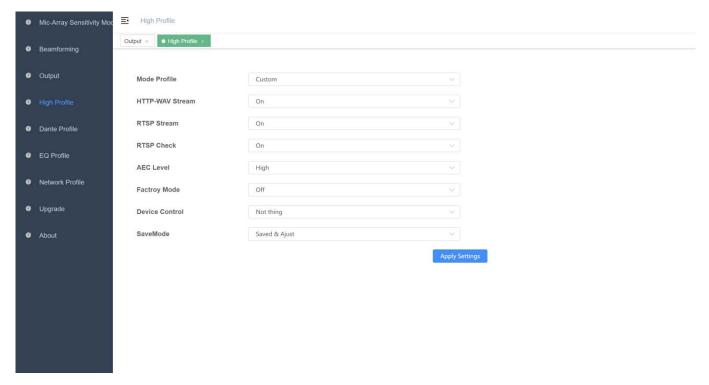
Adjust the overall microphone output volume (including USB and analog signals).

Note: increase output gain by 1dB, 2dB, or 3dB on top of the output gain control.

Save Method: Effective for this session only/Save Only/Save and Apply.

Settings: Confirm the current settings and apply them according to the save method.

(4) Advanced Configuration



Advanced Configuration

Application Mode Settings

Custom configuration not yet available

HTTP-WAV Stream

Enable or disable audio streaming via HTTP protocol, with the HTTP audio stream output at a 32K sampling rate. Microphone audio stream URL: (http://IP/mix.wav).

RTSP Stream

Enable or disable device RTSP stream configuration, with RTSP audio stream output at a 16K sampling rate. Microphone stream URL: (rtsp://IP/audio).

RTSP Authentication

Enable or disable RTSP stream authentication with username and password Username: admin, Password: admin123

Factory Mode

Off

Test MIC Mode (test microphone sound)

Output 0dB 1Khz (microphone outputs 1Khz audio)

Clear AES67 Configuration (one-click to clear AES67 settings)

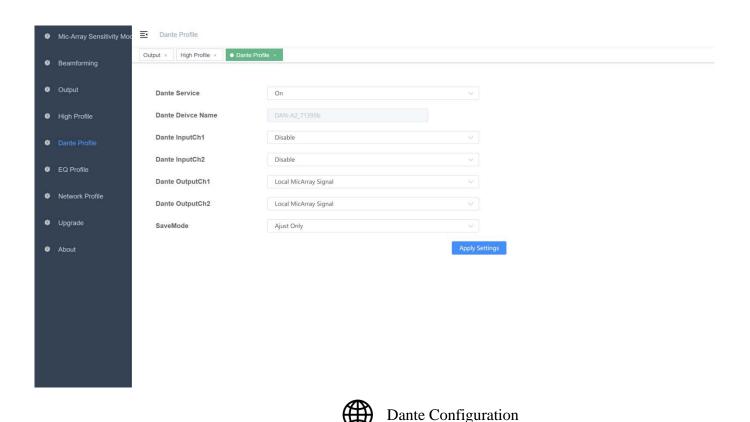
Device Operations

Control device (disable/restore factory settings & restart/restart)

Control restart and restore to factory settings

Save Method: Effective for this session only/Save Only/Save and Apply **Settings:** Confirm the current settings and apply them according to the save method.

(5) Dante Configuration



Dante Service

Dante is an audio connectivity technology that uses Ethernet cables instead of traditional analog cables to achieve high-speed transmission of digital signals and simultaneous transmission of multiple audio channels. In addition, the Dante system can be managed by software, offering scalability and low cost, significantly simplifying the deployment and maintenance of audio and video systems.

Dante Device Name

Display the Dante name of this device.

Dante Input Ch1

Modify input channel type (AEC: Acoustic Echo Cancellation reference signal, Smart Mixing: Microphone Cascade).

Dante Input Ch2

Modify input channel type (AEC: Acoustic Echo Cancellation reference signal, Smart Mixing: Microphone Cascade).

Dante Output Ch1

Output current array microphone signal.

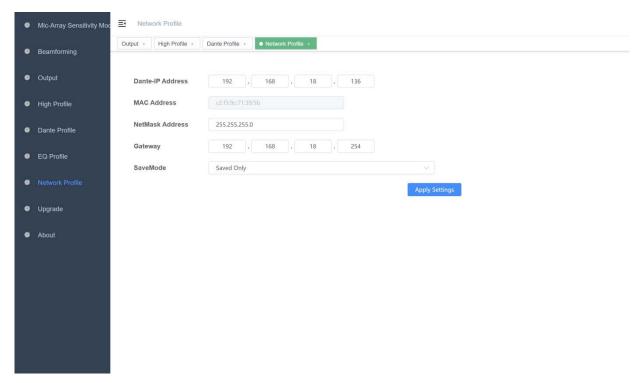
Dante Output Ch2

Output current array microphone signal.

Save Method: Effective for this session only/Save Only/Save and Apply

Settings: Confirm the current settings and apply them according to the save method.

(6) Network Configuration



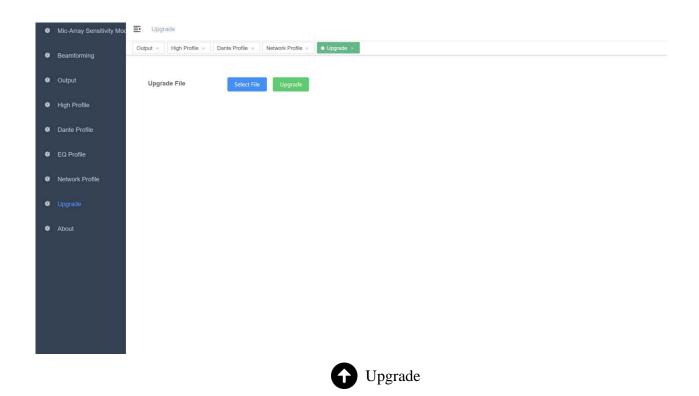
Network Configuration

Configure the product network IP. (Currently, Dante-IP and array mic IP are the same; Dante does not support IP modification.)

Save Mode: Only Effective Now / Save Only / Save and Apply

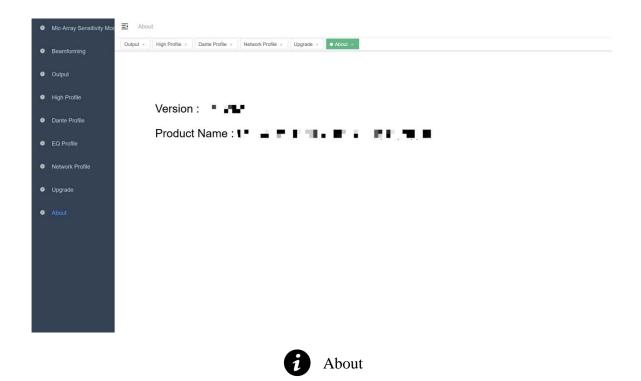
Settings: Confirm current settings according to the save mode.

(7) Upgrade



Select "Upgrade File" and choose to upload, then click "Upgrade Update" to complete the update. The system will automatically restart.

(8) About



Version and product model can be viewed here

4. Interface Introduction

