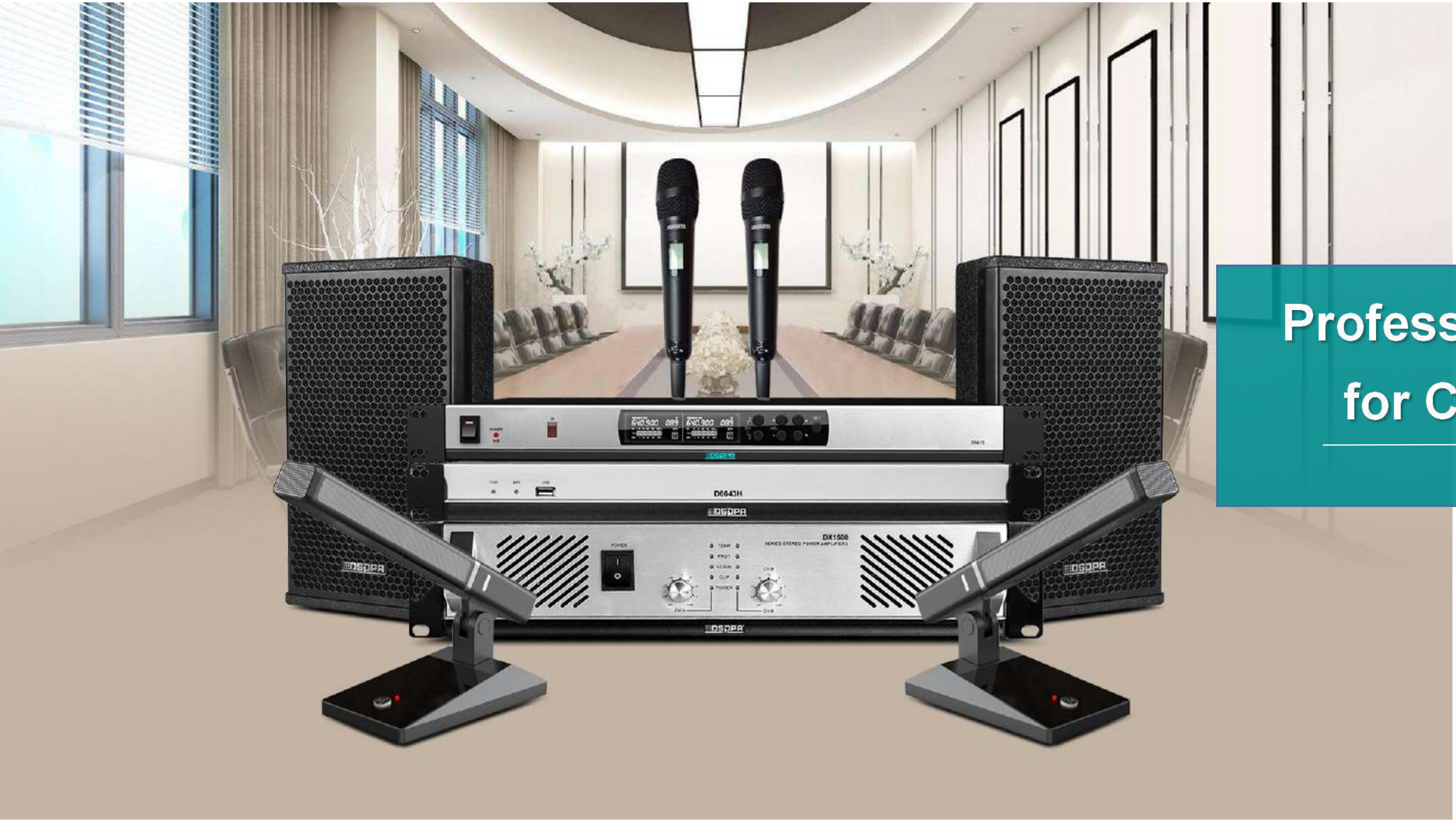


Professional Sound System for Conference Rooms

I. Solution Overview & Pain Points

System Solution Overview

- With the expansion of the company scale and the need for business development, the frequency of conference room usage has been increasing. However, in large meetings or training events, the sound transmission effect in the conference room is not ideal, leading to participants being unable to clearly hear the speaker's remarks. Therefore, in order to improve the sound transmission effect in the conference room, we need to formulate a sound reinforcement solution.

Three Major Key Issues in Conference Room

1 Howling

2 Noise

3 Short Pickup Distance

II. Solution Requirements

Conference Room Area

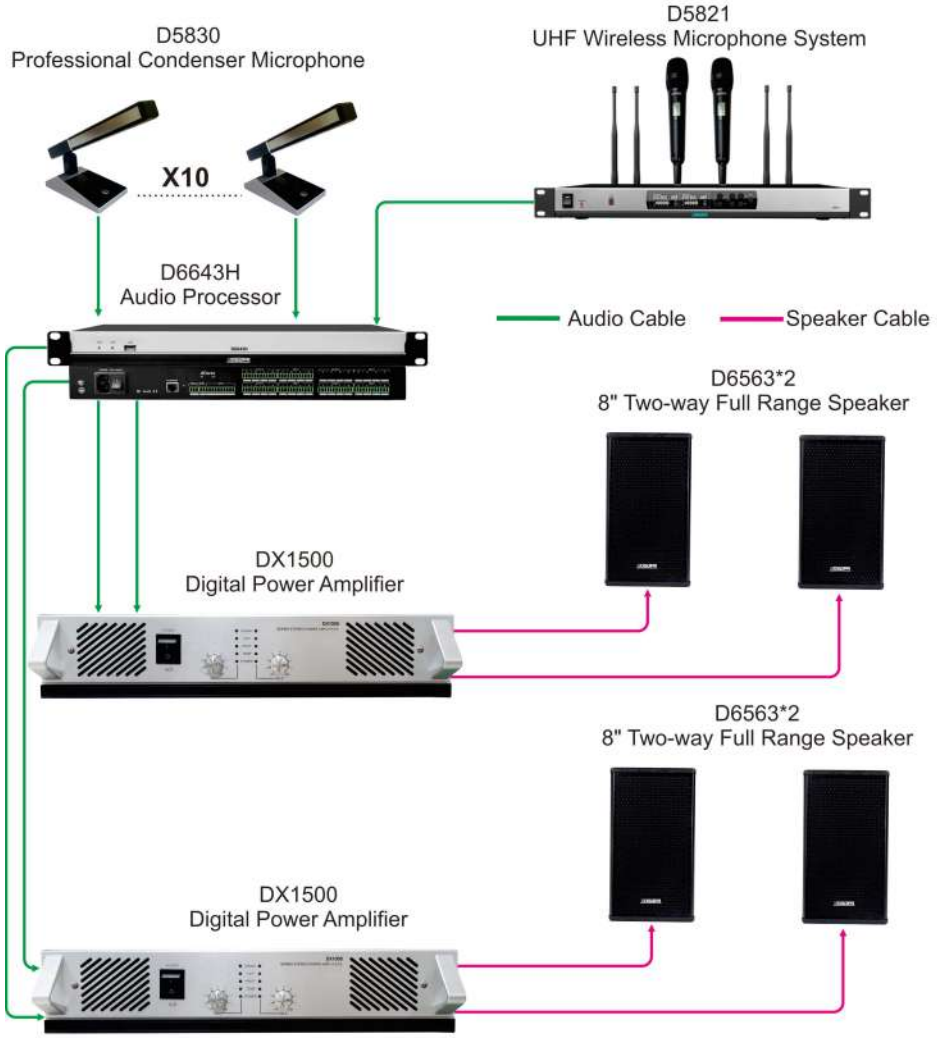
- Length 12m, Width 8m, Height 3.5m; Useable Area: 100㎡
- Independent sound system suitable for conferences.

Project Requirements

- 1. Provide clear and stable sound transmission effect to ensure that participants can hear the speaker's speech.
- 2. Convenient remote control of each speaking microphone, including volume adjustment, sensitivity adjustment, mute, etc.
- 3. Easy installation and operation, reducing operation and maintenance costs.
- 4. Beautiful and elegant product combination, in line with the decoration style of the conference room.

III. Project Configuration

No.	Model	Product Name	Quantity
Meeting Room			
1	D5821	UHF Wireless Microphone System	1PC
2	D5830	Professional Condenser Microphone	10PCS
3	D6643H	Audio Processor	1PC
4	DX1500	Digital Power Amplifier	2PC
5	D6563	8"Two-way Full Range Speaker	4PCS



IV. Product Highlight 1

Howling Suppression

- The sensitivity of each input microphone can be changed according to the different positions used.
- When the microphone is close to the speaker, reducing the microphone sensitivity will reduce the microphone's sensitivity to sound, making it less likely to produce howling.
- Adjustable range: 24dBu to -27dBu.

Volume Adjustment

- Each input microphone can adjust the volume according to different usage habits.
- When the microphone is far away from people, the microphone volume can be appropriately increased to increase the pickup distance.
- Adjustable range: -72dBu to 12dBu.

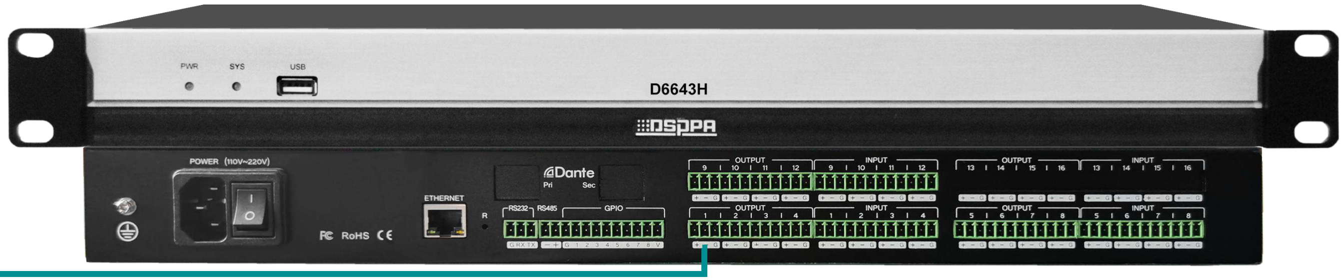
IV. Product Highlight 2

Noise Suppression

- Each input microphone can change the noise suppression ratio according to the environment in which it is used.
- When the microphone environment is noisy, adjust the threshold and ratio of the input source, and sounds below the threshold will be filtered out, thereby reducing noise.
- Threshold adjustable range: -63dBFS to 0dBFS.

Operation Instructions

- Set the input threshold to -48dBFS, and the noise is lower than -48dBFS, which will be directly filtered.
- When the ratio is set to 2, the sound will be sharply reduced according to the ratio, thus suppressing the noise.
- Start time: set to 2ms; Recovery time: set to 2 ms.



IV. Product Highlight 3

Input Balance

- According to the different sound fields of each microphone, the input equalization can be adjusted to reduce the howling caused by excessive frequency points.
- Adjusting the input microphone channel can prevent the music channel from being affected, thereby ensuring that the timbre of the music will not change.

Operation Instructions

- When the low-frequency resonance of the environmental sound field is severe, adjust 120-180 to reduce 2-3dB to reduce the low-frequency sound input.
- Increase 1-2dB at 600-800 to supplement the mid-to-high frequencies of the sound.
- It reduces the turbidity caused by low frequencies and improves the clarity of human voices.

IV. Product Highlight 4

Auto Gain

- When the user speaks into the microphone, the distance between the mouth and the microphone may fluctuate, causing the output volume to fluctuate between high and low, or even causing the user to feel that the speech is intermittent.
- Automatic gain is to set the threshold, and the input signal below the threshold is output according to the ratio of 1:1. For the level above the threshold, the level is directly increased according to the ratio. After setting the target level, the sound signal can be output stably.

Operation Instructions

- Set the input threshold to -45dBFS, which is greater than the noise threshold of -48dBFS.
- The target level is set to -15dBFS, that is, the sound is in the range of -45dBFS to -15dBFS, and the ratio is set to 2. The sound will be directly promoted according to the ratio, and the output will be stable.
- Startup time: set to 200ms; Recovery time: set to 300ms.

IV. Product Highlight 5

Auto Mixing

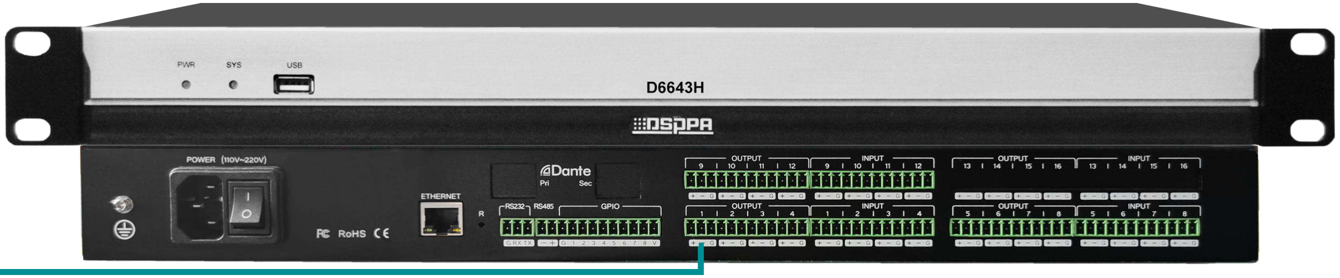
- There are ten participants, each with one microphone. If ten microphones are turned on at the same time, and only one person is speaking, then the output will definitely not be ideal, as the other nine microphones pick up room insulation, reverberation, etc., these will reduce the output effect of the entire system.

- When the microphone is turned on, the automatic mixer can suppress the microphone pickup when no one is talking, thereby obtaining the ideal output effect.

Operation Instructions

- The automatic button "Auto" of input channels 1 - 5 is selected, indicating that the signal of this channel will be sent to the automatic mixer for processing.

- Priority: 0-10 level. When multiple microphones are turned on to speak at the same time, the microphone used by the leader can be set to have a higher priority, thus giving priority to the sound output.



V. System Advantages

Long Sound Pickup Distance

- With the support of automatic gain and automatic mixing technology, the microphone's pickup distance is 20-60CM, which can restore human voices and make them clear and easy to hear.

- It also ensures no howling, no noise, and no interference at a comfortable pickup distance.

Sleek & Stylish Product Design

- Metal texture, zinc alloy shell, tempered glass inlay, beautiful and durable.

- Specially designed shielded circuits and one-piece zinc alloy shell can effectively suppress various types of electromagnetic interference.

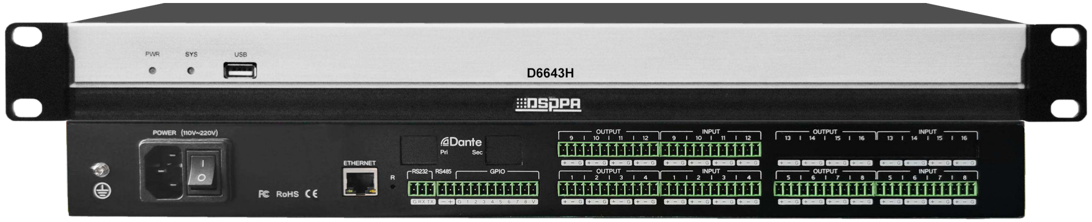
VI. Main Products

Product_Professional Condenser Microphone-D5830

- Can suppress lateral ambient noise greater than 80dB.
- Metallic texture with tempered glass inlay, beautiful and durable.
- Zinc alloy die-casting, oxidized processing on the surface, thick and beautiful; with non-slip pads for four
- feet, more stable.
- Distinctive directional characteristics in the vocal speech frequency band (100-12000) Hz, with excellent
- pickup within the specified range (20-60cm).
- Powered by 48V phantom power supply.
- Output impedance (20%): 680 Ω (balanced).
- With low AC internal resistance, and strong anti-interference capability.

Product_Digital Audio Processor -D6643H

D6643H



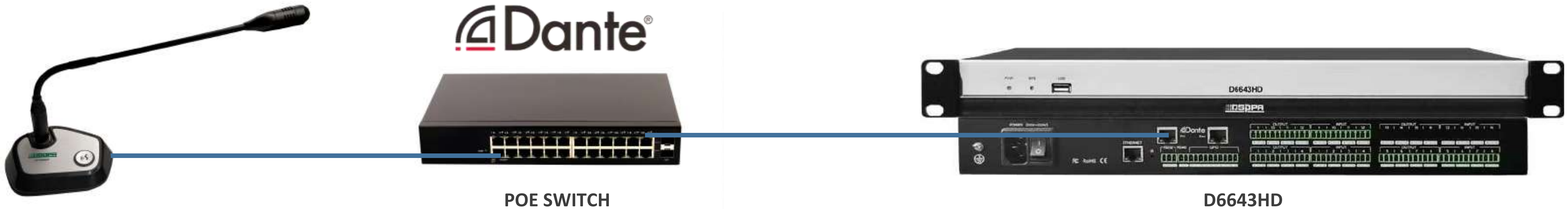
- 24 bit/ 48KHz sampling frequency, high-performance A / D, D / A converter and 32-bit floating-point DSP processor, full-function matrix mixing function to provide users with excellent and clear sound.
- High-precision input sensitivity adjustment with 21 levels, 3dB steps; the maximum input gain is 60dB.
- Efficient algorithm processing: AFC, AEC, ANS, AUTOMIXER, EQ, GATE, AGC and so on.
- Abundant interface extension: Input and output GPIO of 8 channels can be customized. Support external input level of 3.3 ~ 24V; USB interface supports recording and broadcasting, scene preservation and other functions. RS-485 supports automatic camera tracking, easy to achieve video conferencing; RS-232 two-way serial control interface, it can control or be controlled, such as video matrix, camera and other equipment.
- Support scene preset for multiple group, user-friendly operating software interface.
- Fast operation: web control mode, support Android, IOS system.
- Support 1 USB recording channel.

Optional Configuration

Dante Advantage

- Dante signal transmission overcomes the disadvantage of audio cables not being able to transmit over long distances
- Dante signal system enables unlimited long-distance transmission in a network environment.
- Dante signal transmission guarantees minimal delay, ensuring that audio is practically transmitted in real-time.
- Dante signal transmission is uncompressed digital audio, ensuring audio quality and clarity.

Digital Audio Processor with Dante -D6643HD



POE SWITCH

D6643HD

Digital Audio Processor with Dante

DT4200P

Dante Microphone

Creating the Best, Sounding the World!

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