

Product Introduction:

It adopts TCP/IP network protocol, and supports C/S and B/S architectures. The original digital ring network technology, audio clock synchronous transmission technology, built-in high-performance DSP processor, supports 16-channel role separation output mode, 16-channel simultaneous output mode, 16-channel phased output mode and many other functional modes, and is suitable for various medium and large wired and wireless conference scenes.

Functions and Features:

1. Support wired conference and wireless conference at the same time.
2. Adaptive interference avoidance technology is adopted to achieve stronger anti-interference ability, and 29 signals are available.
3. Original digital ring network technology audio clock synchronous transmission technology, audio delay less than 5ms, uncompressed audio transmission with a sampling rate of 48K.
4. Built-in high-performance DSP processor, with audio matrix, howling suppression, EQ, volume, delayer and other adjustment functions.
5. Support a single conference host to realize 16 conference room audio independent zone control and merge audio.
6. The audio input interface includes 1-way RCA and 2-way Phoenix terminals. The audio output interface includes 1-way RCA, 1-way XLR, and 16-way Phoenix terminals. Each output channel can adjust EQ, volume and other parameters.
7. 16-channel role separation output mode, wired or wireless units can output sound independently, and support independent recording of each unit through recording software, and support voice transcription equipment docking to achieve role separation.
8. 16 channels are the same as the transmission output mode, which can independently output the voice of the interpreter machine to provide external equipment, which can be used for recording or monitoring equipment.
9. 16-channel phase-controlled output mode, built-in nx16 audio matrix processor, realizes 16-channel group output function, and outputs to any channel according to any volume ratio.
10. The conference host adopts TCP/IP network protocol, and supports C/S and B/S architecture at the same time, which can be controlled by PC software or browser.
11. It can control audio matrix parameters (including EQ, volume, delayer, microphone sensitivity, etc.), 16-channel output mode switching, switch microphone synchronization, and control role separation host.
12. The system supports a maximum of 5300 wired conference units and 300 wireless conference units. The maximum speaking quantity of the system is 8 wired microphones and 6 wireless microphones.
13. Optional ring-shaped hand-in-hand function is available to ensure that the meeting can continue normally when one of the network cables is disconnected or there is a problem with the unit.
14. Support RS232 central control, realize microphone switch, priority, volume, shutdown and other controls.
15. The computer software can check the information status of the scanning wireless unit's battery power, WiFi signal, etc.; it supports one-key shutdown of all wireless units and a single wireless unit.
16. Support simultaneous interpretation function, the system can transmit 63+1 wired

simultaneous interpretation at the same time.

17. It has a fire alarm linkage trigger interface, provides fire alarm information, and reminds the venue personnel to evacuate at the first time to ensure the safety of the participants.

18. Support PELCO-D, VISCA camera control protocol, can cooperate with high-definition camera tracking host to realize automatic camera tracking.

19. Four microphone management modes: FIFO (first in first out), NORMAL (normal mode), VOICE (voice control mode), APPLY (application mode).

20. The system has the functions of initiating meeting sign-in, voting, election, rating, satisfaction, and customization.

21. It has a 4.3-inch full-color touch screen, which can realize any touch operation for parameter setting or viewing.

22. With USB recording function.

23. Support 10-segment EQ adjustment function, 16 multi-function output channels and 2 LINEOUT output channels have 10-segment EQ adjustment function.

24. The system unit has a hot-swappable function, and the original function will be automatically restored when it is plugged in again.

25. When using without a computer, the conference system can also realize the function of limiting the number of speakers.

26. Support docking voice transcription server to realize voice transcription function

27. The dual-machine true backup function is optional. When a main machine is powered off and the main machine fails, it can automatically switch to the slave machine to run, realizing the dual backup function.

28. DANTE protocol audio output module(Optional function)

29. With 2*50W power amplifier output function, it can directly bring sound reinforcement

30. With independent headphone monitoring function

Technical Specifications:

1. Microphone capacity: wired microphone \leq 5300; wireless microphone \leq 300

2. Simultaneous interpretation channels: 63+1 channels

3. Frequency response: 80~16KHz

4. Signal-to-noise ratio: >80dB(A)

5. Dynamic range: >80dB

6. Total harmonic distortion: <0.05%

7. Main power supply: 100-240VAC

8. EXTENSION port: connect conference system expansion equipment

9. DANTE port: connect to the external device of DANTE protocol(Optional finction)

10. WIFI network port: connect to wireless AP

11. PC network port: connect to the computer

12. Static power consumption: 30W

13. Output power consumption: 350W

14. Wired microphone connection method: special cable

15. Touch screen control: 4.3 inches full-color touch screen