Main Controller

Model: VCS-DG5680M





SUMMARY

• This is a new intelligent digital wired and wireless dual transmission video tracking conference system, which perfectly integrates all digital technology, high-frequency transmission, and comprehensive network technology into the conference system, allowing the system to support both wireless and wired conference speakers. Users can greatly increase the flexibility of system installation according to their own needs, and seamlessly connect with intelligent central control systems, video conference systems, and monitoring fire prevention systems, providing a more complete solution for efficient modern conference system engineering.

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FEATURES

- Adopting all-digital conference technology, developed based on digital network architecture, it can realize the transmission of 64-channel audio and various information in one network cable, and support wired and wireless conferences at the same time.
- Adopting high-fidelity lossless digital audio transmission technology, and with digital network audio clock synchronization transmission technology, the end-to-end audio delay is less than 5ms, and supports non-compressed audio transmission with a sampling rate of 48K, so that the conference sound quality can be restored with high fidelity
- Adaptive interference avoidance technology to achieve stronger anti-interference ability, based on WiFi6 that complies with IEEE 802.11n standards, and supports 2.4GHZ, 5GHZ, and multiple signals are available
- Adopting intelligent anti-interference technology, combined with digital circuit design, to eliminate electromagnetic interference from mobile phones and other electronic products
- With 24 digital transmission signals, multiple wireless systems can be used in the same place, and mutual interference with other wireless products in the venue can be avoided.
- Supporting the ring hand-in-hand function to ensure that the meeting can continue normally when one of the network cables is disconnected or the microphone fails.
- With a 4.3-inch 800*480 full-view IPS capacitive touch screen to intuitively display and conveniently adjust the system's parameters, and supports arbitrary switching of Chinese, English, Russian, and French languages.
- The PC software can view the battery power, WiFi signal and other information status of the online wireless microphone; it supports one-click turning off all wireless microphones and turning off a wireless microphone separately.
- Support USB recording, high-fidelity WAV format output
- The system supports a maximum of 5,300 wired conference units and 300 wireless conference units, and supports a maximum of 8 wired conference units and 6 wireless conference units speaking at the same time.
- The wired unit adopts a "hand-in-hand" connection method, supports hot-swap hot-swap function, and automatically restores the original function when plugged in again, which is convenient for installation and maintenance.
- With multiple conference modes: FIFO (first-in-first-out mode), NORMAL (normal mode), VOICE (voice control mode), APPLY (application mode).
- With conference service functions, and the speaking unit can apply for tea, pens, help and other services.
- The system has functions such as initiating conference sign-in, voting, election, rating, satisfaction, customization, etc., and with PC software, more voting methods can be realized, for example:
- a)Voting method: Agree/Abstain/Oppose
- b)Election method: 1/2/3/4/5
- c)Rating/Satisfaction method: --/-/0/+/++

- Adopting high-performance DSP digital audio processing technology, it supports 16-channel audio matrix, howling suppression, 10-band EQ adjustment, volume dB value adjustment, and delay adjustment.
- Built-in howling suppression function (Optional)
- Support a single conference controller to achieve 16 independent partition output and merged audio of grouped audio.
- With 16-channel wired and wireless role separation output mode, which can make wired or wireless microphones output independently according to ID numbers, and supports 128 independent audio outputs of wired microphones or wireless microphones, and supports independent recording of each microphone through recording software, or voice transcription equipment docking to achieve role separation.
- With 16-channel simultaneous transmission output mode, which can make the translator audio output independently according to the channel number, which can be used for recording or monitoring equipment. And the number of output channels can be expanded through external devices.
- With 16-channel phased output mode, built-in nx16 audio matrix processor, and realizes 16-channel group output function. Any input source (including all input sources and online microphones) can be output to any channel at any volume ratio
- Supports sound partition output function, and the volume of each partition is automatically adjusted according to the microphone opening position, so as to achieve a farther pickup distance without howling.
- The system uses TCP/IP network protocol and supports C/S and B/S architectures. It can be set and controlled by PC software or web pages. The computer supports importing live pictures for simulated ranking.
- The device has client and WEB control methods. Through the client or WEB, the audio matrix parameters (including EQ, volume, delay, microphone sensitivity, etc.), 16 channel output mode switching, switch microphone synchronization, four language switching of Chinese, English, Russian and French, and control of role separation host can be adjusted.
- Support to control audio matrix parameters (including EQ, volume, microphone sensitivity, etc.), 16 channel output mode switching, switch microphone synchronization, and control role separation host.
- With 10-band EQ adjustment function. Both 16 multifunction output channels and 2 LINEOUT output channels have 10-band EQ adjustment function.
- Support simultaneous interpretation system, and can transmit 63+1 wired simultaneous interpretation at the same time.



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- The system is deeply adapted to the voice transcription system. The systems exchange data through the network cable to realize the role-separated voice transcription function.
- With host dual-machine hot standby backup function, and can set the host or slave function. When the host fails or loses power, it can automatically switch to the slave operation to realize the dual backup function.
- With 4 x RJ45 microphone unit interfaces and 4 8-core microphone unit interfaces.
- With 1 x RJ45 data expansion interface, which can be connected to the conference system expansion device.
- With Dante protocol audio output and can communicate with other Dante audio devices. (Optional)
- With 1 x RJ45 WIFI interface, which can be connected to a wireless router.
- With 1 x RJ45 PC interface, which can be connected to a PC for control and management.
- With camera 232 and 485 communication interfaces (6P Phoenix plug), which can connect standard-definition or highdefinition cameras and support SONY VISCA and PELCO P/D communication protocols.
- With central control code 232 interface (3P Phoenix plug), which can be connected to the central control system and supports secondary development of the SDK development package.
- With fire alarm linkage trigger interface, which supports realtime detection of smoke alarms. After triggering, the alarm information will be synchronized to the microphone interface and the host interface.
- With 2 x balanced audio output interfaces (3P Phoenix plug and XLR Canon), 2 x unbalanced audio output interfaces (3P Phoenix plug and 6.35mm audio interface), and 16 x audio matrix output interfaces (3P Phoenix plug). Each output channel can adjust 10-band EQ, delay, volume dB value and other parameters.
- With 2 x balanced audio input interfaces (3P Phoenix plug) and 2 x unbalanced audio input interfaces (2P Phoenix plug), which can input external audio signals (such as background music or remote voice signals).
- With optical fiber extension interface, the sound quality is not attenuated during long-distance transmission, and two remote conference rooms can be merged into one.
- With 1 USB sound card interface and 2 sound card unbalanced output interfaces (2P Phoenix plug), it can be connected to a computer for remote conferencing.
- Supports 4 camera access, 4K image quality, 4-input and 4output HDMI seamless video tracking switching, camera audio output, and optical fiber video transmission. (Optional)



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SUPPLIED ACCESSORIES

1 Power cable

1 USB cable

1 Installation and operating manual

SYSTEM ENVIRONMENTAL CONDITIONS

Working conditions fixed/stationary/transportable

Temperature range

Max.relative hurmidity < 95% (not condensing)

TECHNICAL SPECIFICATIONS

General

Power supply. AC110V-220V/50Hz

Transmission method.....5G

Frequency response 50Hz-18kHzs S/N ratio >80dBA Dynamic >80dB

Wireless Units ≤300

Interpretation channel. 63+1 CH

Others

Weight 5.5 kg [12.125 lbs] Colour Black

Interface

Recording connector. USB

Wired Unit Input connector DIN-8 x4, RJ45 x4 (group)

Central controller connector RS-232 (3P connector) x1 Camera switch connector RS-232 (3P connector) x1

Camera control connector 6P connector x1 Computer connecting USB x1

Audio input 2P connector (unbal) x2

3P connector (bal) x2

Audio output 3P connector (bal) x1

2P connector (unbal) x2

XLR (bal) x1 6.35mm (unbal) x1

Coaxial x1

Optical Fiber x1

Sound card

2P connector (unbal) x2

3P connector (bal) x16

Audio matrix

Digital audio interface Dante x1(If necessary)

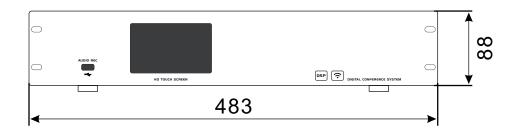
AP interface RJ45 x1 Sound card USB x1

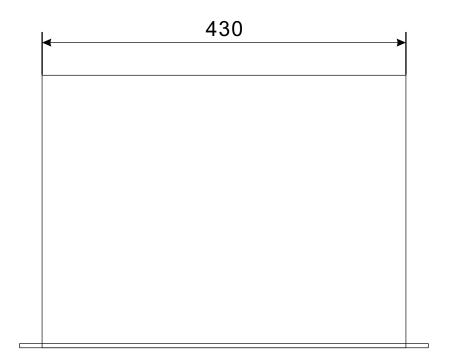


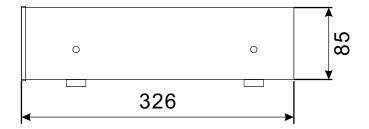
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DIMENSIONS (mm)







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