Full Digital Smart Conference Controller

TS-0400M

Description

This full-digital conference system is built on a gigabit network audio transmission architecture, supporting multi-channel concurrent processing and enabling 128 microphones to be used simultaneously and speak stably. The system highly integrates audio processing and conference discussion control functions, possessing professional-grade audio processing capabilities, with end-to-end latency as low as 2–3 milliseconds from conference unit sound pickup to audio processing output. Simultaneously, the system innovatively integrates practical functions such as multi-room management and identity recognition, comprehensively empowering a high-quality, high-efficiency conference sound reinforcement experience.

Features

*Built on a gigabit network audio transmission architecture, all core components use domestically produced chips, and can connect to digital wired microphones, digital wireless microphones, and analog microphones; it supports multi-channel concurrent processing, enabling 128 microphones to be turned on and speak stably at the same time.

*With an ultra-large system capacity, it supports simultaneous access and management of 65,535 wired conference units and 300 wireless conference units; it also supports simultaneous participation of 65,535 conference units in the meeting agenda (check-in, voting, services, etc.) and speech control.

*It features highly integrated DSP audio processing capabilities, including ducking, automatic gain control, equalizer, compressor, expander, automatic mixing, feedback suppression, echo cancellation, ambient noise reduction, matrix routing, EQ adjustment (supporting 8-band, 10-band, 15-band, and 31-band EQ adjustment), volume dB adjustment, and delay adjustment.

*Equipped with professional-grade audio processing capabilities, it achieves a low latency of 2-3 milliseconds across the entire audio processing chain, from conference unit pickup through ducking, automatic gain control, equalizer, compressor, expander, automatic mixing, feedback suppression, matrix routing, EQ adjustment (supporting 8-band, 10-band, 15-band, and 31-band EQ adjustment), volume dB adjustment, to audio processing output.

*This industry-innovative system adopts a "one-machine-multiple-zone" and "multi-machine interconnection" approach. A single host can be divided into eight independent conference zones, each with its own dedicated microphone and audio channel. Simultaneously, conference functions operate independently without interference, including check-in, voting, and priority settings, ensuring spatial isolation and independent operation. Furthermore, the system supports flexible audio interaction modes, enabling cross-zone two-way communication to meet the needs of group discussions and centralized training sessions. Multiple hosts can be interconnected via a network for unified management, significantly improving resource utilization and making system deployment and expansion more flexible and efficient.

*Multi-mode meeting management and high-concurrency processing: The system supports the creation of 8 independent meeting rooms and can flexibly switch between 7 speaking modes for each meeting room, including first-in-first-out, normal mode, request mode, voice control mode, last-in-first-out, free discussion and queuing mode, to adapt to different meeting formats.

*Ultra-large data processing capability: The system supports simultaneous speaking from 128 conference units and has a function to customize the number of speakers per conference unit. The number of speakers per wired conference unit can be set to any number between 1 and 128; the number of speakers per wireless conference unit can be set to any number between 1 and 8.

*It supports recording audio from individual microphones when used with recording software. It can record the audio of 128 conference sessions separately, and after recording, the audio of each microphone will be generated into a separate WAV file and saved locally.

*It features USB flash drive recording and playback capabilities, supporting individual or combined audio recording of any of the eight channels (IN1~IN8), and allowing users to select individual recording of wired/wireless microphones or perform audio mixing. It also supports audio playback via USB, with selectable individual or combined playback of the ten output channels (LINE OUT 1, LINE OUT 2, OUT 1~OUT 8), meeting diverse needs for meeting recording and multimedia applications.
*All 8 input channels support individual configuration to enable or disable 48V phantom power supply, and each input channel supports sensitivity adjustment.
*It features a 9-input, 10-output audio matrix routing function, and the audio output channels can be configured as wired role separation output mode, wireless role separation output mode, simultaneous transmission output mode, and phase control mode; each audio output channel can independently adjust audio parameters, including volume adjustment, equalizer adjustment, delay adjustment, limiter adjustment, and phase inversion adjustment functions.
*The role separation output mode allows wired or wireless units to output independently based on their ID numbers. By expanding the wired units, it supports 128 independent audio outputs, and the wireless units support 8 independent audio outputs. It also supports independent recording of each unit through recording software or docking with a speech-to-text device to achieve role separation.
*It supports 8-channel simultaneous output mode, with simultaneous audio output independently based on channel number, which can be used by recording or monitoring devices. Furthermore, the number of output channels can be cascaded with external audio processors.
*When the output mode is phase control mode, the matrix router can customize any route for 12 input channels and output channels (LINE OUT1, LINE OUT2, USB recording, wired download, wireless download, OUT1~OUT8). Each channel can be customized as mixing group 1 and mixing group 2, and the functions of feedback suppression, echo cancellation and environmental noise reduction can be customized and enabled.
*The main unit is compatible with simultaneous connection of digital wired microphones, digital wireless microphones, and analog microphones, and all three can be used in parallel. It adopts cross-domain audio synchronization technology, provides input configuration functions, supports selection of any input channel, can switch between microphone input and line input, and adjust phase inversion, volume, ducking, automatic gain, equalizer, compressor, and expander. In microphone input mode, it can also control the phantom power switch, configure camera tracking, voice tracking thresholds, and switch functions.
*It features automatic mixing capabilities, including gain-shared automatic mixing and gated automatic mixing; it has two mixing group settings, with mixing modes including gain-shared automatic mixing and gated automatic mixing. Gain-shared automatic mixing allows setting output gain, gain step size, slope, response time, and mute on/off audio parameters, including input source gain, mute, priority, and automatic mixing settings; gated automatic mixing allows setting output gain and mute on/off audio parameters, including input source gain, mute, sensitivity, and automatic mixing settings.
*It supports adjustable output channel equalizers, with each channel featuring an 8-band parametric equalizer or graphic equalizer, offering 10, 15, and 31-band graphic equalizers. The parametric equalizer supports selection of overhead, undermount, and parametric filter types, as well as setting parameters such as frequency, gain, and Q/OCT; the graphic equalizer supports three modes: narrowband, normal, and wideband.
*The system has functions for meeting check-in, voting, election, rating, satisfaction, and customization.
*It has a device positioning function and supports initiating device positioning via software. After initiating positioning, the device will display a prompt message.
*It supports custom microphone identity and permission configuration, and microphones can be defined as chairman units or representative units according to on-site needs; chairman units can be set to default priority speaking permissions, intermediate priority speaking permissions, and advanced priority speaking permissions; it supports custom setting of unit name and function configuration (including custom selection of initiating hardware check-in/voting, participating in hardware check-in/voting, and initiating meeting services).
*It features two backup mechanisms; it supports dual-machine hot standby, allowing one device to be set as the master and the other as the slave. When the master fails, it can automatically switch to the slave to run, achieving dual backup; it has a built-in two-way ring dual-link backup function, eliminating the need for additional expansion equipment. When one link in the conference unit fails, it can achieve millisecond-level automatic switching of the backup link (switching time ≤ 5ms), ensuring the continuity and stability of system operation.
*It has a fire alarm linkage trigger interface, which supports real-time detection by connecting to smoke detectors. After triggering, the fire alarm information can be synchronized to the microphone interface and the host interface, reminding the participants to evacuate immediately and ensuring their safety.
*It has one RS-485 interface, which can support the connection of up to five cameras for video tracking, and supports PELCO-D and VISCA control protocols.
*It has two RS-232 interfaces, one of which supports connection to a camera to achieve video tracking; the other supports connection to a speech-to-text system to achieve speech-to-text function with separate roles.
*It features a client application, a 4.3-inch touchscreen, and control via Android phones/tablets. The client application allows adjustment of audio matrix parameters (including EQ, volume, delay, conference unit sensitivity, etc.), audio channel output mode switching, conference unit on/off control, language switching (Chinese, English, Russian, French, and Traditional Chinese), and host role separation. The touchscreen allows adjustment of conference modes, the number of wired/wireless conference units that can be used, ID assignment, host/slave settings, language settings (Chinese, English, Russian, French, and Traditional Chinese), display brightness/output volume adjustment, display of remaining usage days, and host registration via registration code. Android phones/tablets allow control of conference unit on/off, enabling check-in, voting, receiving conference service information, and one-click shutdown of wireless conference units, all without the need for a PC.

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*It supports the use of extended cascaded conference network audio processors, and the host and the conference network audio processor transmit audio over the network. The wired unit supports simultaneous transmission of 128 audio channels, and the wireless unit supports simultaneous transmission of 8 audio channels. It also provides functions such as feedback suppression, intelligent mixing, and automatic gain audio adjustment processing.
*It supports simultaneous interpretation, and the system can transmit 63+1 wired simultaneous interpretations at the same time.
*The system is deeply integrated with the speech-to-text system, and the systems exchange data through the network to achieve role-separated speech-to-text functionality without the need for cumbersome wiring.
*It has node management function, which supports viewing the power usage of each channel (aviation port 1 to aviation port 4) of the device through the display screen; it also has an automatic overload alarm function.
*The brightness can be adjusted via the display screen, allowing users to flexibly set any brightness value within the range of 1 to 100 according to the environment; it supports microphone control functions, which can manage microphones from 8 different conference rooms, including one-click shutdown of all wireless microphones, and unified control of the raising or lowering of the microphone.
*It supports displaying device statistics via touchscreen, including the total number of online microphones, chairman's microphone, delegate's microphone, interpreter's microphone, network audio processor, and online status of extended hosts.
*The touchscreen offers gain adjustment functionality, allowing individual adjustment of the gain values for 10 output channels, with an adjustment range of -72dB to 12dB.
*It features an AP list display function, which can show the AP's serial number, SSID, and channel information; it supports AP configuration and allows multiple channels to be added to the blacklist.
*It supports wireless microphone audio download function. When enabled, the wireless microphone with built-in speaker can play the audio of the speech locally to achieve local sound amplification (requires the use of TS-W301S).
*It supports interface language switching and offers five font options: Simplified Chinese, Traditional Chinese, English, Russian, and French.
*The conference host has a registration day display function, which makes it convenient for users to check the remaining usage days after registration at any time; it also supports directly entering the registration code on the touch screen to complete the host registration.
*The front panel has 6 status indicator lights to show the usage status of the main unit and wired and wireless conference units. One of them is the main unit status light, which flashes when the main unit is powered on normally. Four of them are wired conference unit communication indicator lights, which flash when the wired conference unit is in normal communication use. One of them is the wireless conference unit communication indicator light, which flashes when the wireless transceiver is connected and in normal use. It is off when no device is connected, which can quickly detect the link usage status.

Specification

Microphone capacity	Wired microphones ≤65535; Wireless microphones ≤300
Simultaneous Interpretation Channel	63+1 channels
Frequency response	20~20kHz
Signal-to-noise ratio (maximum)	18 dBu, ≥110 dB(A)
THD (Analog Input/Output)	+4dBu, ≤0.002%
Main power supply	100-120V~ 50/60Hz/200-240V~ 50/60Hz
Audio input	8-channel Phoenix terminal block (775mVrms balanced)
Audio output	1-line OUT (1Vrms balanced)
Audio output	1-channel LINE OUT (1Vrms unbalanced)
Audio output	8-channel Phoenix terminal block (775mVrms balanced)
EXTENSION port	Two RJ45 network ports (10/100/1000Mbps) for connecting conference system expansion equipment.
WiFi port	One RJ45 Ethernet port (10/100/1000Mbps) for connecting to a wireless access point.
PC network port	One RJ45 network port (10/100/1000Mbps) for connecting to a computer.
RS-232 interface	Two channels: one for camera tracking and the other for connecting to external devices.
RS-485 interface	1 channel, for the camera
DELEGATES output interface	4 channels, used to connect conference speaking units.
Wired microphone connection method	Specialized cable (6 cores)
Output power consumption	350W
Touchscreen control	4.3-inch display
External dimensions (L×W×H)	482.6×307.6×95.7mm
Installation method	19-inch standard rack
Net weight	Approximately 5.3 kg