



Fully digital conference system host TZ-0400M

Digital Conference System V1.0



Features

1. The equipment has audio clock synchronization transmission technology, end-to-end audio transmission is less than 5ms, and sampling rate is 48K uncompressed audio transmission. It uses Category 5e shielded cables to ensure long-distance and reliable transmission of conference information while providing perfect sound quality.
2. Built-in DSP processor with 16-channel audio matrix, howling suppression, 10-band EQ adjustment, volume dB value adjustment, and delay adjustment functions.
3. The communication interface includes 2 RS232 interfaces, 1 RS-485 interface, and 4 RJ45 interfaces; the audio input interface includes 1 RCA, 1 XLR, and 2 Phoenix terminals; the audio output interface includes 1 RCA, 1 XLR, and 16 Phoenix terminals.
4. Supports 16-channel audio output function, which can be flexibly configured as wired role separation output mode, wireless role separation output mode, simultaneous transmission output mode, and phase control output mode. Each output channel can adjust 10-band EQ, volume dB value adjustment, and delay parameter adjustment.
5. The 16-channel role-separated output mode allows the wired or wireless unit to output independently according to the ID number, supports up to 128 channels of independent audio output from wired or wireless units, and supports independent recording of each unit through recording software, or docking with voice transcription equipment to achieve role separation.
6. The 16-channel simultaneous transmission output mode allows the simultaneous transmission audio to be output independently according to the channel number, which can be used by recording or monitoring equipment. The number of output channels can be expanded through external devices.
7. 16-channel phase-controlled output mode, built-in $n \times 16$ audio matrix processor, 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.
8. The conference host uses TCP/IP network protocol and supports both C/S and B/S architectures, and can be controlled by PC software or browser.
9. The device has the function of being controlled by Android mobile phone APP software and tablet APP software. The software can control the microphone switch, sign-in, voting, receiving conference service information, one-click wireless microphone shutdown and other functions, without PC operation.
10. The device has client and WEB control modes. Through the client or WEB, you can adjust the audio matrix parameters (including EQ, volume, delay, microphone sensitivity, etc.), switch the 16-channel output mode, switch the microphone synchronously, switch between four languages (Chinese, English, Russian and French), and control the role separation host.
11. The system has a large system capacity and can be expanded to 4096 wired conference units and 300 wireless conference units. The system supports 24 microphones speaking at the same time, including 16 wired microphones and 8 wireless microphones speaking at the same time; it has a custom microphone speaker number function, the number of wired microphone speakers can be set to any number between 1 and 16; the number of wireless microphone speakers can be set to any number between 1 and 8.
12. A ring-shaped hand-in-hand function can be realized to ensure that the meeting can continue normally when one of the network cables is disconnected or a unit fails.
13. It supports multiple languages including Chinese, English, Russian and French, and can be switched at will.
14. The PC software can check the battery power, WiFi signal and other information status of online wireless units; it supports one-click shutdown of all wireless units and shutdown of a wireless unit individually.
15. It has a fire alarm linkage trigger interface and supports real-time detection of smoke alarms. After triggering, the fire alarm information can be synchronized to the microphone interface and the host interface, reminding the venue personnel to evacuate urgently in the first time to ensure the safety of the participants.



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16. It has 1 RS-485 interface, supports one camera to realize camera tracking, supports PELCO-D and VISCA control protocols, and cooperates with the camera tracking host to achieve multi-channel video automatic tracking function.
17. Multiple microphone management modes: FIFO (first in, first out), NORMAL (normal mode), VOICE (voice control mode), APPLY (application mode), etc.
18. The system has functions such as initiating meeting check-in, voting, election, rating, satisfaction, customization, etc.
19. It has a 4.3-inch full-color touch screen, which can realize parameter setting or viewing and any touch operation.
20. Powerful ID function, can ID wired unit, wireless unit, interpreter machine, role separation host.
21. Equipped with USB recording function, it can record and play meeting minutes.
1. Audio input: LINEIN1: 775mVrms balanced; 2-way input Phoenix terminal: 775mVrms balanced; LINEIN2: 775mVrms unbalanced
2. Audio output: LINEOUT1: 1Vrms balanced; 16-way multi-function output Phoenix terminal: 1Vrms balanced; LINEOUT2: 1Vrms unbalanced
22. Supports AP channel scanning, monitors the use of wireless channels on site, supports automatic or manual configuration of the best channel, and supports online display of AP name list for easy reference.
23. The conference host has the function of displaying the number of days registered, so you can know the remaining days after registration at any time; it supports entering the registration code on the touch device screen to register the host.
24. The system is deeply adapted to the speech transcription system. The systems exchange data via network cables to achieve role-separated speech transcription functions without the need for tedious wiring processes.

Specification

Main power supply	100-120VAC/200-240VACbyswitch
Frequency Response	80~16KHz
Signal-to-Noise Ratio	>78dB(A)
Dynamic Range	>80dB
Total Harmonic Distortion	<0.05%
Output Load	>1KΩ
Output power consumption	320W
Static power consumption	30W
Installation	19-inch standard cabinet
Wired Microphone Connection Method	Special cable (6 cores)
RS-232 interface	2 channels, 1 channel for camera tracking, 1 channel for connecting to external devices
RS-485 interface	1 channel, for camera tracking
PC network port	1 channel, connected to computer
DELEGATES Output Interface	4 channels, used to connect conference speaking units
E×TENSION port	1 channel, connect conference system expansion device
DANTE/NC port	1 channel, for connecting to external devices using the DANTE protocol
WIFI port	1 channel, connected to wireless AP
Conference speech recording function	When used with a conference microphone, it can record the speech audio of a single microphone or the mixed output audio of all microphones; it supports recording via the host USB flash drive or PC software.
Simultaneous interpretation function	The system can transmit up to 63+1 wired simultaneous interpretations at the same time.
EQ adjustment function	The 16 multi-function output channels and 2 LINEOUT output channels all have 10-band EQ adjustment function.
Host dual-machine hot standby function	The host or slave function can be set. When the host fails, it can automatically switch to the slave operation to achieve dual backup function.
Custom conference microphone identity function	It can be defined as a chairman unit, a representative unit or a "VIP" unit according to on-site needs.
Functions of the operation and maintenance management platform	The firmware can be upgraded remotely via the web. It has a log management function that can automatically collect and store system logs. For example, it can monitor the device's operating status and device failure information in real time, including insufficient memory, fire alarms, and duplicate IDs.
color	black
Product size (L × W × H)	484 × 303 × 88mm
net weight	5.6 kg