

Conference Controller

Full digital conference system audio transmission embedded software

TS-0300M embedded V4.41

TS-0300M



Feature:

- Uncompressed audio transmission with a sampling rate of 48K. Category 5e shielded cables are used to ensure reliable transmission of conference information over long distances while providing perfect sound quality. The device uses segmented compression mixing processing technology and clock synchronization transmission technology, and the delay from conference unit pickup to host output is 5ms. Built-in DSP processor, with 16-channel audio matrix, howling suppression, 10-band EQ adjustment, volume dB value adjustment, and delay adjustment functions.

- Supports two recording methods: host USB flash drive and client software; with conference microphones and recording boxes, it can record the speech audio of a single conference unit and the mixed speech audio of all conference units
- 4. Supports two recording memors nost USB hash drive and cellent software; with conference microphones and recording boxes, it can record the speech audio of a single conference unit and the mixed speech audio of all conference units.

 5. The device has 1 USB interface; the rear panel has 2 RS-232 interfaces, 1 RS-485 interface, and 4 RJ45 communication interfaces; it has 1 RCA input, 1 XLR input, and 2 Phoenix terminal input interfaces; 1 RCA output, 1 XLR output, and 16

 Phoenix terminal output interfaces; 1 DIP switch, 1 grounding column.

 6. It has 16 audio output channels, which can be expanded to 272 audio output channels. The audio output channels can be configured as wired role separation output mode, wireless role separation output mode, and simultaneous transmission output mode; each audio output channel can independently adjust audio parameters, including 30-level volume adjustment, 10-band equalizer adjustment, and 100-level delay adjustment functions.

 7. It has 3 backup mechanisms; it supports the host dual-machine hot standby function, which can set one device as the host and the other as the slave. When the host falls, it can automatically switch to the slave to realize the dual backup function; it supports the ring dual-link function to ensure that the meeting can continue normally when one of the network cables is disconnected or the unit has problems; it supports the T-link backup function. Even if multiple conference units fail in the link, other
- conference units are not affected, ensuring the normal progress of the meeting.

 8. Supports the use with conference microphone processor. The host and microphone processor transmit audio through the network cable connection. It can simultaneously transmit the audio signals of 16 wired conference units and 8 wireless

- conference units, and provide feedback suppression, intelligent mixing and automatic gain audio adjustment processing functions.

 9. 16-channel role separation output mode allows wired or wireless units to output independently according to ID numbers, supporting up to 128 channels

 10. Independent audio output of wired units or wireless units and supports independent recording port independent evording software, or voice transcription equipment docking to achieve role separation.

 11. 16-channel simultaneous transmission output mode allows simultaneous audio to be output independently according to channel numbers, which can be used for recording or monitoring equipment. And the number of output channels can be expanded through external devices
- 12. The host has a 16-channel audio group output interface; using conference partition phase control technology, it can be split into 16 independent conference systems for use, or it can be used as a large conference system, realizing multiple ways of
- 12. The host has a 16-channel audio group output interface; using conference partition phase control technology, it can be split into 16 independent conference systems for use, or it can be used as a large conference system, realizing multiple ways or conference room merging/splitting.

 13. It has C/S and B/S control architecture, including client, WEB, local full-color touch screen, and Android phone/tablet control mode; through the client and WEB, you can adjust the audio matrix parameters (including EQ, volume, delay, conference unit sensitivity, etc.), 16-channel output mode switching, switch oofference unit, switch between Chinese, English, Russian and French, and control role separation host function; use the local full-color touch screen to adjust the conference with switch between Chinese, English, Russian and French, display brightness/output volume adjustment, display remaining days, enter registration code for host registration function; use Android phone/tablet to control the conference unit switch, start sign-in, vote, receive conference service information, and one-click close the wireless conference unit function, no PC operation required.

 14. It has C/S and B/S architecture management software, and 8 operating systems that can run on both client and WEB software, including Windows7/10/11, Kylin Desktop Operating System (Peitian Edition), page 2005 Systems 2005 United Transit and Systems (Peitian Edition), page 2005 Systems 2005 United Transit and Systems (Peitian Edition), page 2005 Systems 2005 United Transit and Systems (Peitian Edition), page 2005 Systems 20
- Edition), macOS System, Tongxin UOS, and Ubuntu Desktop Operating System.
- 15. 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, vote, receive conference service information, and turn off the wireless
- 15. The device has the function of being continued by Android income private Art 1 software and table Art 1 software and
- 17. Ultra-large data processing capacity: The system supports 24 conference units speaking at the same time, including 16 wired conference units and 8 wireless conference units speaking at the same time; it has the function of customizing the nu of conference unit speakers, and the number of wired conference unit speakers can be set to any number between 1 and 16; the number of wireless conference unit speakers can be set to any number between 1 and 8.

 18. The conference system multi-loop detection and network replenishment technology are adopted to achieve rapid recovery when the hand-in-hand link of the conference unit fails, and the loop recovery time is 5ms.

 19. It supports arbitrary switching display in multiple languages ??including Chinese, English, Russian and French.

 20. It supports the function of customizing the identity of conference microphones, which can be defined as the chairman unit, representative unit or "VIP" unit according to the needs of the site.

 21. The PC software can view the battery power, WIFi signal and other information status of the online wireless unit; it supports one-click shutdown of all wireless units and shutdown of a wireless unit separately.

 22. It supports simultaneous interpretation function, and the system can transmit up to 6341 wired simultaneous interpretation at the same time.

 23. It has a fire alarm linkage trigger interface, supports real-time detection of smoke alarms, and after triggering, the fire alarm information can be synchronized to the microphone interface and the host interface, reminding the venue personnel to evacuate urgently and ensuring the safety of the participants.

 24. It has 1 RS-485 interface, supports one camera to realize camera tracking, and supports PELCO-D and VISCA control protocols. Cooperate with the camera tracking host to achieve multi-channel video automatic tracking function.

 25. Multiple microphone management modes: FIFO (first in, first out), NORIMAL (normal mode), VOICE (voice control mode), APPLY (application mode), etc. large data processing capacity: The system supports 24 conference units speaking at the same time, including 16 wired conference units and 8 wireless conference units speaking at the same time; it has the function of customizing the number

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- 31. Support AP channel scanning, monitor the use of wireless channels on site, support automatic or manual configuration of the best channel, and support online display list of AP names for easy reference.

 32. 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; support entering the registration code on the touch device screen for host registration.

 33. The system is deeply adapted to the voice transcription system. The systems interact with each ther through network cables to realize the role-separated voice transcription function without the need for cumbersome wiring process.

 34. The host is compatible with wired and wireless conference units at the same time, and the two can be used in parallel; cross-domain audio synchronization technology is adopted, and the audio of wired and wireless conference units as the same time.
- 34. The nost is companied with which and whereas conference which at the same that the
- 37. It has the function of hot standby of host dual machines, which can set the host or slave function. When the host fails, it can automatically switch to the slave operation to realize the dual backup function.
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 38. It has the function of operation and maintenance management platform, which can remotely upgrade the firmware through the web end; it has log management function, which can automatically collect and store system logs; for example, real-time monitoring of equipment operation status and equipment failure information, including insufficient memory, fire alarm prompts, ID duplication, etc.

 39. The conference host software is integrated into the audio integrated management platform to achieve unified management of audio equipment. The platform can scan the online status of digital conference hosts, audio processors, mixers, suppressors, and amplifiers. Multiple online devices of the same product can also be scanned, and the device hardware name, hardware IP address, online and offline status information will be displayed; it has the function of uploading configuration information to the cloud or saving it locally for backup with one click, and restoring configuration information with one click.

Specification:

Model	TS-0300M
Microphone capacity	Wired microphone ≤4096; wireless microphone ≤300
Simultaneous interpretation channel	63+1 channel
Frequency response	80~16KHz
SNR	>78dB(A)
Dynamic range	>80dB
THD	<0.05%
Main power	100-120VAC/200-240VACbyswitch
Audio input	LINEIN1: 775mVrms balanced; 2 input Phoenix terminal: 775mVrms balanced; LINEIN2: 775mVrms unbalanced
Audio output	LINEOUT1:1Vrms balanced; 16 multi-function output Phoenix terminal: 1Vrms balanced; LINEOUT2:1Vrms unbalanced
Output load	>1ΚΩ
EXTENSION port	1 for connect conference system extension equipment
DANTE/NC port	1 for connect to external devices with DANTE protocol
WIFI network port	1 for connect to wireless AP
PC network port	1 for connect to the computer
DELEGATES output interface	4 for connect conference speaking units
RS-232 interface	2 channels, 1 channel for camera tracking, 1 channel for docking external equipment
RS-485 interface	1 for camera tracking
Static power	30W
Output power consumption	320W
Wired microphone connection method	Special cable (6 cores)
Touch screen control	4.3 inch full color touch screen
Colour	black
Net weight	5.6Kg
Dimension (LxWxH)	484x03x88mm
Installation method	19 inch standard cabinet