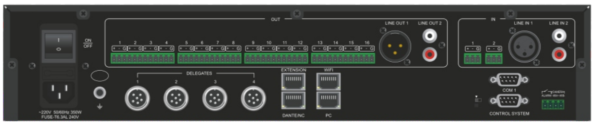
Conference Host

Full digital conference system audio transmission embedded software

TS-0300M embedded V4.41

TS-0300M



Feature:

* Using clock synchronization and transmission technology, the audio delay is less than 5ms; adopt the uncompressed audio transmission with 48K sampling rate; use cat5e shielded cable to ensure long-distance reliable transmission of conference information, while providing perfect sound quality.
* Built-in high-performance DSP processor, with 16-channel audio matrix, howling suppression, EQ, volume, delay and other adjustment functions.
* Support conference speech recording function; with conference microphone, it supports recording the speech audio of a single microphone or recording the mixed output audio of all microphones; support recording through the host U disk or PC software.
* The communication interface includes 2 RS232 interfaces, 1 RS-485 interface, and 4 RJ45 interfaces 1 RCA, 1 XLR, and 2 Phoenix terminals. The audio output interface includes 1 RCA, 1 XLR, and 16 phoenix terminals.
* Support 16-channel output function, support flexible configuration as 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.
* The 16-channel role separation output mode enables wired or wireless units to output independently according to ID numbers; it can be used by recording or speech transcription equipment; the number of output channels can be extended by external devices.
* The 16-channel simultaneous transmission output mode enables simultaneous interpretation audio to be output independently according to the channel number; it can be used by recording or monitoring equipment; the number of output channels can be extended by external devices.
* The 16-channel phase control output mode, based on the original conference matrix technology, built-in nx16 audio matrix processor, can achieve 16-channel group output function. Any input source (including all input sources and online microphones) can be output to any channel according to any volume ratio.
* Adopt TCP/IP network protocol, support C/S and B/S architecture; it can be controlled by PC software or browser.
* The device has Android mobile phone and tablet APP software, which supports controlling the microphone switch, enabling sign-in, vote, receive conference service information, turn off the wireless microphone with one click and other functions without PC operation.
* Support client and WEB side control methods. Audio matrix parameters (including EQ, volume, delay, microphone sensitivity, etc.) can be adjusted through the client or WEB side,16-channel output mode switching, switch microphone synchronization, four language switching of Chinese, English, Russian and French, control role separation controller.
* Large system capacity, the system supports a maximum of 4096 wired conference units and 300 wireless conference units. The maximum number of speakers in the system is 16 wired microphones and 8 wireless microphones.Support custom microphone speaker number function, the wired microphone speaker number range can be set to any number between 1 and 16; the wireless microphone speaker number range can be set to any number between 1 and 8.
* Support circular hand-in-hand function to ensure that the conference can continue normally when one of the network cables is disconnected or the unit fails.
* Support Chinese, English, Russian, French and other languages.
* Support custom conference microphone identity function, which can be defined as chairman unit, representative unit or "VIP" unit according to on-site needs.
* Through the PC software, you can view the information such as battery level, WiFi signal of the online wireless units; it supports one-key shutdown of all wireless units and a single wireless unit.
* Support simultaneous interpretation function, the system can simultaneously transmit 63+1 wired simultaneous interpretation.
* With a fire alarm linkage trigger interface, it supports real-time detection of connected smoke alarms. After triggering, the fire alarm information can be synchronized to the microphone interface and the host interface to remind the venue personnel to evacuate immediately to ensure the safety of the participants.
* With 1 RS-485 interface, support a camera to realize camera tracking, support PELCO-D, VISCA camera control protocol; work with HD camera tracking server to realize automatic camera tracking.
* Multiple microphone management modes: FIFO (first in, first out), NORMAL (normal mode), VOICE (voice control mode), APPLY (application mode).
* Support functions such as meeting sign-in, voting, evaluation, and other custom functions.
* With a 4.3-inch full-color touch screen, it can realize parameter setting or viewing, and other touch operation.
* Powerful ID editing function, support ID coding for wired units, wireless units, interpreter machines, and role separation servers.
* With USB recording function, it can record and play meeting records.
* Support 10-band EQ adjustment function; 16 multi-function output channels and 2 LINEOUT output channels support 10-band EQ adjustment function.
* Support AP channel scanning, learn about the use of wireless channels on site, support automatic or manual configuration as the best channel, support online display of AP name list, convenient to check.
* The conference host supports the registration days display function to know the remaining days after registration at any time; Support entering the registration code on the touch screen for host registration.
* Support connecting with speech transcription system. The systems exchange data through network cables to realize the role-separated speech transcription function without the need for cumbersome wiring processes.
* Support the master-slave dual hot backup function. The host or slave machine function can be set. when the master fails, it can automatically switch to the slave operation to achieve dual backup functions.
* With the operation and maintenance management platform, support remotely upgrading firmware through the web; support log management functions and can automatically collect and store system logs; support real-time monitoring of equipment operating status and equipment failure information, including insufficient memory, fire alarm prompts, ID duplication, etc.

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	>1KΩ
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