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 audio matrix, howling suppression, EQ, volume, delay and other adjustment functions.
* 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.
* Support 16-channel output function, support flexible configuration as role separation output mode, simultaneous transmission output mode and phase control output mode, and support adjusting parameters such as EQ, volume and delay for each output channel.
* 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.
* Support the WEB control of audio matrix parameters (including EQ, volume, delay, microphone sensitivity, etc.), 16-channel output mode switching, switch microphone synchronization, Chinese, English, Russian and French switching, and the 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 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.
* 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 provides fire alarm information to evacuate the venue personnel urgently.
* 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.
* Four 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. g.
* 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.
* Support entering the registration code on the touch screen for host registration.
* Support connecting with speech transcription server to realize speech transcription function
* Support setting the master-slave dual backup function, when the master fails, it can automatically switch to the slave operation.

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