Conference System Server TS-0300MS

Embedded software: Full digital conference system V4.41

Feature

* Original clock synchronization and transmission technology, sampling rate 48K uncompressed audio transmission.
* Built-in high-performance DSP processor, with EQ, volume, delay and other adjustment functions.
* The audio input interface includes 1 RCA unbalanced input port and 1 XLR balanced input port. The audio output interface includes 1 RCA unbalanced output port and 1 XLR balanced output port.
* 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.), microphone status synchronization, Chinese, English, Russian and French switching, and the role separation controller.
* Large system capacity, the system supports a maximum of 300 wireless conference units. The maximum number of speakers in the system is 8 wireless microphones.
* 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.
* With 1 POE network port, support power supply and data transmission for wireless AP.
* With a fire alarm linkage trigger interface, it provides fire alarm information to evacuate the venue personnel urgently.
* With 1 RS-485 interface, support up to 5 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.
* With a 2.2-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.
* Support 10-band EQ adjustment function; 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 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 to realize double backup function.
* Support remote firmware upgrade through the web.

Specification

Microphone capacity	Wireless microphone≤300
Frequency response	20Hz ~ 20KHz
SNR	>85dB(A)
Dynamic range	>80dB(A)
THD	<0.05%
Main power	100-240AC/50-60Hz
Audio output	LINE OUT 1: 1Vrms XLR balanced output; LINE OUT 2: 1Vrms lotus unbalanced output
Audio input	LINE IN 1: 775mVrms XLR balanced input; LINE IN 2: 775mVrms lotus unbalanced input
EXTENSION port	1 channel, connected to conference system expansion equipment
WIFI network port	2 for connect to wireless AP
PC network port	1 for connect to the computer
RS-232 interface	2 channels, 1 channel for camera tracking, 1 channel for docking external equipment
RS-485 interface	1 for camera tracking
Balanced input impedance	>10KΩ
Balanced output impedance	470Ω
Unbalanced input impedance	>5KΩ
Unbalanced output impedance	470Ω
Static power consumption	7.1W
Output power consumption	≤30W
Display	2.2-inch TFT-LCD display, resolution 240*320 pixels
Operating temperature	-10℃~+60℃
Working humidity	20%~80% relative humidity, no condensation
Colour	Black
Net weight	3.2Kg
Dimension	484×258×44mm
Installation method	19-inch standard cabinet