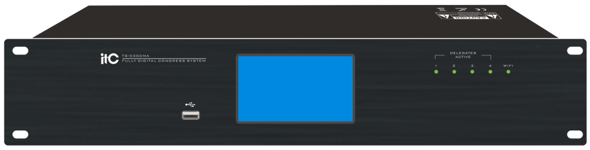
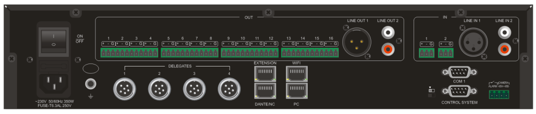
Conference Controller
TS-0300MA





Feature:

* Original clock synchronization and transmission technology, audio delay of less than 5ms, sampling rate 48K uncompressed audio transmission. STP, CAT5E are used to ensure long-distance and reliable transmission of conference information while provide 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.
* It supports 16-channel output function, and it can be flexibly configured as role separation output mode, simultaneous interpretation output mode, and phase control output mode. Each output channel can adjust * EQ, volume, delay and other parameters.
* The 16-channel role separation output mode enables the wired or wireless unit to output independently according to the ID number, which can be used by recording or voice transcription equipment. And the number of output channels can be extended by external equipment.
* The 16-channel simultaneous interpretation output mode allows interpretation audio to be output independently according to the channel number, which can be used by recording or monitoring equipment. * And the number of output channels can be extended by external equipment.
16-channel phase-controlled output mode; based on the original conference matrix technology, it can achieve 16-channel grouping output function with built-in nx16 audio matrix processor. Any input source (including all input sources and online microphones) can be output to any channel according to any volume ratio.
* The conference controller adopts TCP/IP network protocol. It supports C/S, B/S architecture at the same time, which can be controlled by PC software or browser.
* Control audio matrix parameters through WEB(including EQ, volume, delay, microphone sensitivity, etc.), 16-channel output mode switching, switching microphone synchronization, four languages switching between Chinese, English, Russian and French, and control role separation from controller.
* Large system capacity, the system supports up to 4096 wired conference units and 300 wireless conference units. The maximum number of speeches in the system is 8 wired microphones and 6 wireless microphones.
* The ring-shaped hand-in-hand function can be realized to ensure that the meeting can work normally when one of the network cables is disconnected or the unit has a problem.
* The WIFI network interface has a POE power supply function, which can directly connect to wireless AP; it can also extend the number of wireless AP by connecting to a POE network switch to provide a larger wireless coverage.
* Support for arbitrary switching of Chinese, English, Russian and French languages.
* We can check the battery power, WiFi signal and other information status of the online wireless unit by PC software;Support one key to turn off all/one wireless unit(s).
* Support simultaneous interpretation function, the system can simultaneously transmit up to 63+1 channels.
* With a fire alarm linkage trigger interface, it can provide fire alarm information and remind the venue personnel to evacuate urgently for safety.
* Support PELCO-D, VISCA camera control protocol, and it can work with high-definition camera tracking controller to realize automatic camera tracking.
* Four microphone management modes: FIFO (first in first out), NORMAL (normal mode), VOICE (voice control mode), APPLY (application mode).
* The system has functions such as initiating conference sign-in, voting, election, rating, satisfaction, and customization.
* It has a 4.3-inch full-color touch screen, which can set or view parameters and perform any touch operation.
* Powerful ID editing function, which can edit ID for wired unit, wireless unit, interpreter unit, and role separation from controller.
* With USB recording function, it can record and play meeting records.
* Support integration with paperless systems to realize the unified lifting function of paperless lifting microphones.
* Support 10-segment EQ adjustment function, 16 multi-function output channels and 2 LINEOUT output channels have 10-segment EQ adjustment function.
* The DANTE network interface can easily interact with external devices under the DANTE protocol.
* Support automatic gain function (AGC). It makes the gain of the amplifying circuit automatically adjusted with the signal strength. That is, it optimizes the sound difference caused by the different distance from the speaker.
* Adaptive feedback cancellation function(AFC)

Specification:

Model	TS-0300MA
Microphone capacity	Wired microphone ≤4096; wireless microphone ≤300
Simultaneous interpretation channel	63+1 channels
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 output 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	Connect conference system extension equipment
DANTE/NC port	Connect to external devices with DANTE protocol
WIFI network port	Connect to wireless AP
PC network port	Connect to the computer
Static power	30W
Output power consumption	320W
Wired microphone connection method	Special cable (6 PIN)
Standard	IEC-60914
Touch screen control	4.3 inch full color touch screen
Colour	Black
Net weight	5.6Kg
Dimension (LxWxH)	484x303x88mm
Installation method	19 inch standard cabinet