IP Network Speaker

T-8807BMY

Description

This IP network speaker supports AI voice Q&A. It can read new textbooks aloud, helping students better engage in class. It can also assist teachers with interactive Q&A sessions, guiding students to deeper thinking. It can also meet campus bell and local sound reinforcement needs. It's suitable for primary and secondary school teaching scenarios.

Features

*The speaker is designed with high-density ABS material, which has the advantages of shockproof, durability, exquisite appearance, etc.
*It has two MIC input interfaces, one of which supports connecting a microphone to realize local paging and sound amplification functions and supports network volume adjustment; one of which can be connected to an external digital detector. The device has a built-in digital ambient sound detection algorithm to detect abnormal status of the playback speaker.
*Built-in microphone, supports audio detection, supports collecting and detecting audio frames, network packet loss rate, maximum frame spacing, link crossing points and other data, and analyzes playback status and audio recognition, and uploads it to the background, supports exporting reports.
*It has 1 line (AUX) input interface, supports network volume adjustment, supports local sound amplification function when disconnected from the network, and supports background accompaniment preset function.
*It has a short-circuit input interface and supports customized functions such as alarm triggering, local media library music playback, and volume adjustment.
*It has one RS-485 interface and supports external volume control panel.
*The dual network interface design and the terminal support redundant backup function effectively avoid the problem of single point failure causing the device to be unable to permanently connect to the system.
*It has a 100V constant voltage signal backup input interface, which switches to the backup channel when the machine is out of network, avoiding crosstalk between the local signal and the backup signal.
*Supports network and analog 100V primary and backup switching. In the event of a power outage or network disconnection, it automatically switches to the analog 100V constant-voltage backup line. The delay for switching to the hearing backup is less than 0.03 seconds, and the switching process is smooth, without lags or dropouts, without affecting normal broadcasts. When the network and power supply return to normal, it automatically switches to the primary channel in less than 0.03 seconds, with no delays, lags, or dropouts, without affecting normal broadcasts.
*The main speaker has a built-in 2×30W (MAX) dual-channel Class D digital power amplifier, one channel is connected to the sub-speaker, and adopts a high and low frequency division design; the sound quality is delicate and the power is strong; it has a network volume setting.
*Built-in network audio decoding module, supports mainstream audio formats such as MP3, WAV, FLAC, OGG, AAC, OPUS, and is compatible with the full sampling rate of 8kHz-48kHz.
*Built-in DSP audio processing, supports ultra-low latency digital mixing, and 10-band EQ equalization configuration.
*Built-in 3-level priority settings: (1) Network alarm signals take priority over local input signals and 100V analog backup. (2) Local input signals MIC, AUX and network background music are prioritized by the server; local input signals MIC and AUX are mixed at the same level. (3) 100V analog backup signals have the lowest priority.
*Supports remote firmware upgrades and device maintenance via the network, reducing staff workload.

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*The device is equipped with a U-band wireless handheld microphone and has local paging and sound amplification functions . It adopts a single-channel dual-antenna design, covers a frequency range of 640MHz~690MHz, and achieves frequency matching through infrared frequency matching technology .

*The device's microphone features a one-touch activation of AI voice interaction mode. The system connects to the AI big data teaching model via the Internet. This enables functions such as textbook reading, question analysis and response, and teacher-teaching interaction, making the classroom more engaging.

*The system uses a data redundancy encoding and decoding algorithm and supports anti-packet loss recovery. When the network packet loss rate reaches 37.5% , the audio playback is smooth.

*The end-to-end delay of the system playing collected audio is ≤ 5ms.

Specification

Network interface	Standard RJ45 × 2 input
Transfer rate	100Mbps
Supported agreement	TCP/IP, UDP, IGMP, ICMP
Audio format	Supports mainstream audio formats such as MP3, WAV, FLAC, OGG, AAC, OPUS, etc.
Audio mode	16-bit CD-quality sound
Sampling rate	8kHz-48kHz
Auxiliary line input level	350mV Industrial standard 3.81mm crimping terminal
Frequency response	80Hz-16kHz (+1dB/-3dB)
MIC input sensitivity (unbalanced)	120mV Industrial standard 3.81mm crimping terminal
MIC frequency response	200Hz-10kHz (+1dB/-3dB)
Harmonic distortion	≤1%
Signal-to-noise ratio	≥70dB
Output power	2×30W (MAX)
Maximum sound pressure level	99dB
Sensitivity	86dB
Power consumption of the whole machine	60W
Short-circuit input	Dry contact input industrial standard 3.81mm crimping terminal
Backup 100V input	Industrial standard 5.08mm crimping terminals
Wireless U-segment Microphone	Wireless handheld microphone, with single-channel dual antenna design, covers a frequency range of 640MHz~690MHz, and achieves frequency matching through infrared frequency matching technology
AI wake-up mode	Long press the button to start the question, release it to end the question
Working environment temperature	5℃~40℃
Working environment humidity	0% to 80% relative humidity, no condensation
Input power	~220V 50Hz
Main box net weight	2.8kg
Auxiliary box net weight	2.4kg
Dimensions (L×W×H)	165×140.2×305mm