Embedded software: Digital IP network platform terminal embedded software V2.02



Description

The wall-mounted codec amplifier is suitable for places with sound amplification needs such as hearing test schools, office buildings, conference centers, alarm centers, station lounges, etc. It uses network digital transmission to amplify the timed ringing tasks or one-way calls set in the system. It also meets the local wireless connection sound amplification needs.

Feature

*The integrated double 86 wall-mounted structure has a simple and beautiful appearance.

- *It has a 2.8-inch color screen, supports network task status display, volume display, and real-time display of task execution status.
- *It has 1-channel MIC input interface, suitable for local microphone sound reinforcement.
- *The built-in microphone supports the one-key inspection function, which can effectively detect the terminal link of the device and support uploading the collected audio data to the server for subsequent detailed analysis and processing.
- *It has 1-channel line input interface, suitable for external signal input and local sound reinforcement.
- *It has 1-channel line output interface, suitable for external signal amplification equipment.
- *The dual network interface design and the terminal support redundant backup function effectively avoid the problem of the device being unable to permanently connect to the system due to a single point of failure.
- *It has 1-channel 100V constant voltage signal backup input interface, which switches to the backup channel when the machine is without network to avoid crosstalk between local signals and backup signals.
- *Supports network and analog 100V main and standby switching functions. Supports automatic switching to the analog 100V constant voltage backup line when power is off or the network is disconnected. The delay of the hearing backup switching is less than 0.03 seconds. There is no delay, no lag, no dropout during the switching process, and it does not affect the normal broadcast; when the network and power supply return to normal, it automatically switches to the main channel. The switching time is less than 0.03 seconds. There is no delay, no lag, no dropout during the switching the switching time is less than 0.03 seconds. There is no delay, no lag, no dropout during the switching process, and it does not affect the normal broadcast.
- *Built-in 2×30W (MAX) dual-channel Class D digital power amplifier, with delicate sound quality and strong power, can be connected to external speakers; and has network volume setting.
- *Built-in network audio decoding module, supports mainstream audio formats such as MP3, WAV, FLAC, OGG, AAC, OPUS, etc., 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) The priority of local input signals MIC, AUX and network background music is configured 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 equipment maintenance via the network to reduce staff workload.
- *The device is equipped with a U-band wireless handheld microphone, has local paging and sound amplification functions, adopts a single-channel dualantenna design, covers a frequency range of 640MHz~690MHz, and achieves frequency matching through infrared frequency matching technology.
- *The system adopts data redundancy encoding and decoding algorithm and supports anti-packet loss recovery function. The network packet loss is 37.5%, and the audio playback is smooth.

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Specification

Network interface	Standard RJ45×2 input
Transmission rate	100Mbps
Supported protocols	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
AUX input sensitivity	350mV (unbalanced) Industrial standard 2.54mm crimp terminal
MIC input sensitivity	120mV (unbalanced) Industrial standard 2.54mm crimp terminal
LINE OUT output frequency response	80Hz-16kHz (+1dB/-3dB)
LINE OUT output distortion	≤0.1%
LINE OUT output signal-to-noise ratio	≥73dB
LINE OUT output impedance	470Ω
Amplifier frequency response	80Hz-16kHz (+1dB/-3dB)
Amplifier Harmonic Distortion	≤1%
Amplifier signal-to-noise ratio	≥73dB
Amplifier output power	2×30W (MAX)
Power consumption	≤45W
Wireless U-band Microphone	The wireless handheld microphone adopts a single-channel dual-antenna design, with a frequency range
	of 640MHz~690MHz, and frequency matching is achieved through infrared frequency matching technology.
Working environment temperature	5°C~40°C
Working environment humidity	20%~80% relative humidity, no condensation
Input Power	~220V 50Hz
Weight	0.9kg
Dimensions (L x W x H)	85.1×45×171.95mm