SIP731T rack-mounted network playback terminal





SIP731T is a rackmount SIP playback terminal with 10/100M Ethernet interface, a set of AUX audio input and a set of AXU audio output on the rear panel, and a set of relay output of 220V AC, which can be directly connected to a pure post-amplifier, making the post-amplifier play audio from the network and local. Also has a 6.35mm interface microphone input, three volume adjustment knobs, and level indicator, as a local amplification and volume adjustment use.

SIP731T device is a SIP network broadcast decoding pre, only the network decoding function, its receive the audio data of the network, provide audio output. SIP731T and SIP server, SIP network microphone host supporting the use of the main control room to SIP731T terminal for timing ringing, real-time voice broadcast and emergency broadcast and other functions.

Installation and Configuration

The SIP731T is indoor rack mountable with RJ45 network interface and AC power interface. Upon power-up, the SIP731T will automatically obtain an IP address (BootP or DHCP) or find its IP address using Manager software. The unit can also be easily configured and controlled via a standard web browser.

Application Development and Integration

SIP731T supports various control and communication modes, and we provide development documentation and programming examples for playback as well as SIP intercom (off SIP server). Serial port control commands and network control commands are also available. Software developers can write their own audio applications with custom features.

Functional Features

- Standard RJ45 network interface, providing online remote upgrade of firmware;
- ARM+DSP-based architecture with high-speed industrial-grade chips, with a start-up time of ≤ 1 second;
- With volume adjustment potentiometer, you can realize the volume hardware adjustment
- With microphone input and line input, you can achieve the local audio source or microphone to achieve local amplification function.
- Support SIP v1 (RFC2543), v2 (RFC3261) and other network protocols
- Supports wideband G.722, narrowband G.711a/u and MP3 audio decoding protocols.
- Support RTP multicast playback protocol, support 10 priority levels, high priority audio can interrupt the low priority
- Support our private broadcast protocol, which can be broadcast shouting, timed ringing and other operations through our broadcast software
- Provide PC-based IP and other parameters configuration program manager
- Relay control: automatic control (with sound that is closed) / manual control
- Support firmware remote upgrade

Technical parameters

- Network interface: 10/100Base-TX adaptive network, RJ45 interface, send and receive data instructions
- Network protocols: support SIP v1 (RFC2543), v2 (RFC3261), TCP, UDP, RTP, ARP, ICMP, DHCP, DNS and IGMP, etc.
- Input Voltage : AC 220V

- MIC input: local microphone input interface, frequency range 70~12.K5Hz; typical amplitude 50mVpp, signal-to-noise ratio 68dB
- Line input: can be used as local amplification, typical amplitude 2000mVpp
- Line output: external active speaker or amplifier, typical amplitude 2200mVpp
- Relay output: one power relay output and one switching relay output
- Input Voltage : AC 220V
- Power consumption: static <300mW
- Operating temperature: -40℃～85℃
- Specification size: 485×225×65mm
- Panel: Aluminum alloy housing
- Weight: 3.2kg
- Installation method: rack-mounted