SIP804T network intercom help microphone

SIP804T is a four-button help-to-talk microphone with 10/100M Ethernet interface, supporting G.711 audio codec, which receives audio data from SIP network, decodes and plays in real time, and also configures microphone input and speaker output.

SIP804T allows point-to-point SIP intercom and sub-regional RTP multicast to SIP terminals via microphone or local line input.

SIP804T, as SIP intercom terminal, supports full-duplex two-way intercom, G.722, G.711a/u codec and MP3 audio decoding, and has good echo suppression function. We provide the standard sip method, which can be used in other standard SIP systems and compatible with standard SIP terminal equipment.

Installation and Configuration

The desktop microphone SIP804T with RJ45 network interface and DC5.5-2.1 power interface will automatically get an IP address (BootP or DHCP) when powered on, or find its IP address using Manager software. This terminal can also be easily configured and controlled through a standard web browser.

Application Development and Integration

SIP804T 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

☑ Support standard SIP protocol, can be completely free of computer software support, set up directly and SIP intercom two-way talk or broadcast shouting

☑ You can use the key to initiate RTP multicast, send the local audio through the network with other RTP multicast-enabled devices with speakers or line output to send out

☑ Supports our private broadcast protocol, which can be broadcast shouting and timed ringing through our broadcast software

☑ Fully compatible with SIP711 intercom broadcast terminal, as well as compatible with other SIP intercom terminal, you can freely choose to build SIP intercom system

☑ You can call the designated SIP phone host by pressing the key, or call the SIP804T terminal through any phone host in the SIP system. Full duplex mode is adopted during intercom, and Echo suppression and cancellation function is provided.

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, IGMP, etc.
- MIC input: self-contained high-performance microphone, frequency range 70 ~ 12.K5Hz; typical amplitude 50mVpp, signal-to-noise ratio 68dB
- Audio output: with its own waterproof 3W speakers
- Intercom mode: support quiet mode and noisy mode, key answer and automatic answer, on-site hang-up and remote hang-up, etc.
- Panel: brushed aluminum alloy shell
- Operating temperature: -40℃～85℃
- Input voltage: DC 9~24V

- Power consumption: static <300mW
- Specification size: 162×105×40mm (excluding microphone pole)
- Weight: 0.6kg (including microphone pole)