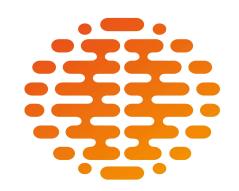
CS-71 SIP Mini Radio Console

At A Glance

SIP RoIP Device - The CS-71 is an economical solution for enabling a single channel of Push to Talk capability for LMR applications using standard SIP protocol.





- Breakthrough Purpose built SIP Device
- (1) channel SIP call, PTT
- Standards based SIP Stack
- Compact Design for Maximum Portability
- Simple to Set up & Use
- Microphone, handset or headset options
- Rotary volume control for ease of use
- PoE & Local power options

Technologically Advanced

- Native 1-channel SIP PTT Device
- Supported Codecs: G.711 A/uLaw, G.729 and Speex
- Static / DHCP / Remote Configuration
- Browser based administration
- Single Button Conference Setup/Access







Technical Specifications

Channels

■ (1) SIP Lines

Call Types

- SIP Radio Conferencing
- SIP Private Line (Point to Point/Ringdown)
- Group Calls/point to point intercom
- Push to Talk Broadcast

Mode of Operation

- Auto Answer / Auto Dial
- Point to Point w/o SIP Server
- Manual—Dial/Answer
- Manual—Answer only
- Hands Free & Privacy

Signaling

- SIP (compatible with RFC 3261)
- Audible ring tone for incoming calls

Interfaces

- 12" Gooseneck Microphone
- Handset w/PTT (RJ-25)
- Mono Headset (RJ-25)
- (1) Ethernet Port

Dimensions

- Width 4" / 102mm
- Depth 5.5" / 140mm
- Height 5.5" / 140mm
- Weight 1.1 lbs / 525 g
- 12" / 305mm Gooseneck microphone

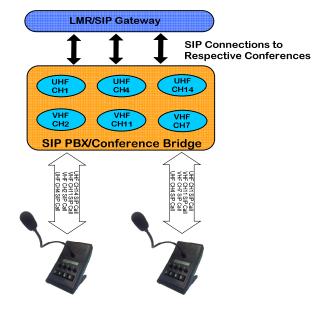
Codec Options (bandwidth including overhead)

■ G.711 a/u-law
■ G.729
■ Speex
4-15 kbit









Management

- Browser based, Internet Explorer, Google Chrome
- Upgradeable application firmware via file upload

Power

- 48 VDC, 1/2 A, External PoE Injector
- 48 VDC, IEEE 802.3af, Alt A & B, Power over Ethernet

Thermal

- 3 Watts
- 10 BTU/hr
- Cooling Ambient air

Network Requirements

- 100 Base T (full duplex)
- Protocols SIP, UDP, DHCP, DNS, Syslog, NTP, SSH, TCP, RTCP