



# CS-71

## SIP Trading Device

The CS-71 is an economical solution for enabling a single channel of Push to Talk capability for Hoot n Holler or Ringdown calls using standard SIP protocol.

### All New Design

- Breakthrough Purpose build SIP Device
- (1) channel SIP call, PTT
- Standards based SIP stack
- Compact Design for Maximum Portability
- Simple to setup and use
- Microphone, handset or headset option
- 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



COMMUNICATE MORE, CARRY LESS

# Technical Specifications

## Channels

- (1) SIP line

## 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 without SIP Server
- Manual-Dial / Answer
- Manual - Answer Only
- Hands Free & Privacy

## Interfaces

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

## 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, RTP

## Dimensions

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

## Codec Options

- G.711 a/uLaw: 80kbit
- G.729: 20kbit
- Speex: 4-15kbit

## Signaling

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

