ACU SIP-LMR™

Twelve Channel SIP-to-Radio Channel Bank

JPS Interoperability Solutions

The ACU SIP-LMR Channel Bank provides BSI compliant SIP on the network side, and on the radio side, all of the ease of setup and comprehenive interface features that made JPS the leader in voice communications interoperability.

Benefits

- SIP-to-Radio interfaces bring radio advantages to SIP networks or SIP to radio networks
- Interoperability is as simple as creating a conference call within a SIP PBX
- Supports 4 to 12 channels as desired
- Extends SIP based communication to rugged terrain areas where cell phone carriers do not provide coverage
- Supports calls made both with and without a proxy server
- Supports operations behind a NAT firewall
- Brings SIP PBX features into the radio arena – including call logging, forwarding and recording
- Web based control and configuration, including an intuitive help-rich Graphical User Interface; no need to load custom applications on control PCs
- Handset and speaker for local monitor and communications with SIP or radio interface; assists system setup

Overview

The ACU SIP-LMR Channel Bank provides a seamless interface between radios and an IP based network using the SIP protocol. This brings to existing SIP networks all of the features inherent in a radio system, including the ability to wirelessly reach otherwise inaccessible areas. For example, the ACU SIP-LMR can be used with an LMR system to extend the SIP Network into areas of rugged terrain, across bodies of water, or into tunnels. Radio networks can now benefit from all of the features available with SIP. These include interoperable communications among disparate radio systems that is as easy as creating a typical PBX conference call and also other PBX features such as Call Logging, Call Forwarding, and Call Recording.

Capable SIP and Radio Interfaces

Each channel on the SIP side of the ACU SIP-LMR assigns its associated radio a unique extension that can easily be dialed using any IP phone, softphone, or other voice communications device associated with the SIP PBX (Private Branch Exchange). Any number of radios, SIP phones, or other audio devices in the network can be conferenced together by the SIP PBX. Alternatively, the ACU SIP-LMR can assign an IP address to its associated radio for communications over any IP-based network or the Internet with another SIP-enabled device (such as a SIP phone or softphone). The radio side of the interface makes full use of the extensive suite of digital signal processor algorithms, hundreds of interface cables, and numerous problem-solving techniques that JPS has evolved during more than a decade as the market leader in radio interoperability.

Why SIP?

The main goals of modern communications system design include: convergence of voice, data, and video; standards-based, open protocols; and individual IP addresses for all end-devices. Session Initiation Protocol, SIP, is widely seen as the preferred pathway to achieving these goals. SIP is a signaling protocol used to create, manage, and terminate sessions in an IP-based network. A session could be a simple two-party call or a multimedia conference session. SIP focuses on the setup, modification, and termination of sessions - allowing maximum versatility of the format and content of data being shared.

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JPS's SIP to radio interfaces provide a standards-based, open-protocol method to seamlessly connect radios to SIP networks. Best of all, system users continue to use their existing devices, saving hardware and training costs.

Radio Interface Specifications

Radio RX Input: Balanced Hi-Z; accepts signals from -30 to +11 dBm, 10 to 3600 Hz

Radio Unsquelch Detection: Hardwired COR input line, DSP-based VOX

Radio TX Audio Output: Unbalanced Lo-Z; adjustable from -30 to +11 dBm, 10 to 3350Hz

Radio Transmit Control: Open drain PTT signal; with max sink current of 100 mA, max open circuit voltage of 60 VDC

Network Specifications

RFC 3261: SIP: Session Initiation Protocol

RFC 2976: The SIP INFO Method

RFC 3515: The Session Initiation Protocol (SIP) Refer Method

RFC 2327: SDP: Session Description Protocol

RFC 3264: An offer/answer model with Session Description Protocol (SDP)

RFC 1889: RTP: A transport protocol for real-time applications

RFC 1890: RTP protocol for audio and video conferences with minimal control

SIP Support Vocoders: G711a and G711u

General/Environmental

Programming/Configuration: HTTP (Web)

Network Interface Type: 10/100BASE-T Ethernet, 100Mbps; RJ-45 Connector

Input Power : +11 to +15 VDC @ 0.5A max, 12VDC Power Supply supplied

Power Connector : Coaxial jack, 205mm, ID, 5 to 5.5mm OD; Center Pin Positive; Reverse Polarity Protected

Size and Weight : 1.7"H x 6.75"W x 8.25"D (4.3 x 17.2 x 18.5cm) 1.1lbs. (2.4kg)

Operating Temperature : 0 to +40 degrees C

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