ARA-1™

Analog Radio Adaptor

JPS Interoperability Solutions

The ARA-1 extends the coverage and capability of an existing SIP PBX by allowing the interface of LMR radios to the system. It's comparable to an analog telephone adapter (ATA), which allows a standard telephone to operate on a SIP network; the ARA-1 provides the same capability to a radio.

Benefits

- SIP-to-Radio interface brings radio advantages to SIP networks and SIP capabilities to radio networks.
- Interoperability is as simple as creating a conference call within a SIP PBX.
- Extends SIP based communication to areas where cell phone carriers do not provide coverage (but are served by LMR).
- Enables communications between radios and network communications devices such as SIP phones and softphones.
- Brings SIP PBX features into the radio arena - including call logging, forwarding and recording.
- Supports calls made both with and without a proxy server.
- Supports operations behind a NAT firewall.
- Multiple codec and compression level options allow system optimization relative to available bandwidth.

Overview

Simply defined, the ARA-1 provides a seamless interface between a radio and an IP-based network using SIP. 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, an ARA-1 can be used with an LMR system to extend the SIP Network into areas of rugged terrain, across bodies of water, or into tunnels.

The ARA-1 also provides all of the features available with SIP to radio networks. 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.

SIP-to-Radio Interface

The SIP side of the ARA 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 ARA-1 can assign an IP address to its associated radio for communications over any IP-based network (including the Internet), with another SIP-enabled device such as a SIP phone, a softphone, or a second radio/ARA-1 pair.

The radio side of the interface makes full use of the extensive suite of digital signal processor algorithms, hundreds of custom radio 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. SIP focuses on the setup, modification and termination of sessions maintaining simplicity while allowing versatility of the format and content of the data being shared.

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JPS has taken interoperability to the next level. Need to communicate where telephones can't? In subways, areas of rugged terrain, or across bodies of water? The JPS Analog Radio Adaptor is an open protocol, standards-based system that connects radios to SIP networks seamlessly. Think of it as Analog Telephone Adaptor technology, but for radio

Radio 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 or VMR

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; multiple compression levels

General/Enviromental

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 Wallcube 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 17.5 x 21cm); 1.1lbs. (2.4kg)

Temperature : Operating: -20 to +60 degrees C, Storage: -40 to +85 degrees C

Humidity : Up to 95% @ 55 degrees C

Regulatory Compliance: FCC Part 15, CE, TUV (Safety for US & Canada)

JPS Interoperability Solutions
5800 Departure Drive
Raleigh, NC 27616
919.790.1011
919.865.1400 fax
24/7 Support Provided
www.jpsinterop.com

Sales Inquiries:
Sales@jpsinterop.com

Support Inquiries:
Support@jpsinterop.com

Media Inquiries:
media@jpsinterop.com

Facebook:
www.facebook.com/jpsinterop

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