ACU-2000 IP™

IP-Centric Interoperability Gateway

JPS Interoperability Solutions

The ACU-2000 IP provides true convergence of Local Interoperability, IP communications and control, and SIP. Now, you can bring all the advantages of the open-standards SIP protocol to your radio systems and add radio functionality to your network.

Benefits

- All of the features of the ACU- 1000, plus the ability to interface with SIP devices and networks
- Connect SIP VoIP devices to radios
- Two-way radio users have access to features that have traditionally been available only to telephone users, including the ability to directly call telephone extensions, call forwarding, call logging, and call recording
- Control a large interoperability system via IP
- Connect radio systems at multiple sites across an IP network
- Remotely change radio channel or frequency over IP
- When used as part of a Wide Area Interoperability System (WAIS), the distributed design ensures continuity of local operations in the event of net- work failure

Overview

The ACU-2000 IP offers a full suite of network capabilities including linking of radios over an IP network, control of large interoperability systems via IP, remote channel change over IP, and the ability to interface radios via SIP. The ACU-2000 IP builds on the industry standard ACU-1000's ability to link disparate communications systems. These systems can be linked, monitored and controlled over an IP network, and the SIP capabilities allow SIP-based systems or individual SIP endpoints (such as SIP phones or softphones) to be included in the conferences. Like the ACU-1000, the ACU-2000 IP is modular, completely scalable, and field configurable to meet the customer needs.

Customers can employ the new SIP communication capabilities in either of two versions, depending on their requirements.

Local Interoperability Gateway with SIP Capability

This version of the ACU-2000 IP adds SIP phones, SIP PBXs and other SIP devices to the long list of communications media that can be included in an interoperability conference. The SIP capability allows radios to be operated from anywhere on the network with the ease of operating a telephone; the interface even includes a speed dial.

This version continues the ACU-1000 "distributed network" approach, with local interoperability links taking place within the unit itself, not relying on a network server and thus assuring continuity of local operations in the event of network failure.

Server-based, Highly Scalable Interoperability

The second version allows radios and other four-wire devices to be included in a server-based network topology using SIP to initiate and manage all cross-connections. Each radio is assigned its own IP address and the interoperability takes place in the IP realm.

Why SIP?

SIP is a standards-based open protocol used to create, manage and terminate sessions in an IP network. SIP enables the convergence of voice, data, and video, allowing equipment with varying media capabilities to be conferenced together. An essential component of the protocol is a determination of the services supported by each of the different types of communications equipment in the conference, so that any services held in common can be exchanged between them.

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When the situation is critical, your team needs integrated voice, data, and multi-media communications in conjunction with seamless interoperability. The ACU-2000IP from JPS provides a true SIP-based gateway to digitally converge existing radio systems with SIP telephones, networks, and devices. Now you can bring all of the advantages of the open-standards SIP protocol to your radio systems and add radio functionality to your network.

Radio/4W Interface Specifications

Audio Input: Balanced or Unbal 600 ohms or Hi-Z; -26dBm to +12dBm levels; 100Hz to 3200 Hz

Audio Output: Balanced or Unbal 600 ohms; -26dBm to +12dBm levels; 100Hz to 3200 Hz

Digital I/O: COR/Squelch and AUX inputs, PTT, and AUX outputs; E&M input/output

DSP Algorithms: VOX or VMR Voice Detection; TD-Mode Noise Reduction; DTMF; Audio Equalizer; TX and/or RX Audio Delay; Peak Limiter; COR Sampling; TX Keying Tones

SIP Network Interface

RFC Supported: 3261, 2976, 3515, 2327, 3264, 1889

SIP Support Vocoders: GSM and G711u

Telephone Line Interface

Phone Line: RJ-11 Connectors (2); -24 dBm to 0 dBm levels

DSP Algorithms: DTMF Detection and Generation; DSP Adaptive Hybrid, DSP VOX

RoIP (Radio over IP) Interface

Network Interface: RJ-45 Connector; 10/100Base-T Ethernet

Radio-Centric Features: Audio delay and jitter buffers to handle network latency; embedded COR, PTT, and RS-232 serial control; three audio vocoder types.

Programming/Configuration: HTTP (Password Protected Web)

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

General/Environmental

AC Input Power : 115 to 230 VAC +/- 15% 47-63Hz, 80VA Typical, 100 VA Maximum

DC Input Power : +11 to +15 VDC @4A Nominal

Battery Charger : 1A Output Maximum; Tapered charge circuitry for a lead-acid battery

Size and Weight : 5.25"H x 19"W x 11"D (13.3 x 48.3 x 28cm)

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

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