Installation and Operation Manual

ARA-1
Radio to SIP Interface

Designed and Manufactured by:

Raytheon Company

5800 Departure Drive

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24/7 Technical Support

For support, call 1-800-498-3137

P/N 5060-800200 **Revision 3.0** **November 2009**

FEDERAL COMMUNICATIONS COMMISSION

(FCC) COMPLIANCE NOTICE:

RADIO FREQUENCY INTERFERENCE NOTICE

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case users will be required to correct the interference at their own expense.

CAUTION

Changes or modifications to this equipment not expressly approved by Raytheon could void the user's authority to operate this equipment.

NOTICE

Raytheon reserves the right to make changes to the equipment and specifications without prior notice.

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Glossary		
ATA	Analog Telephone Adapter	A device that interfaces an analog telephone to a SIP network.
COR	Carrier Operated Relay	A signal from a receiver that indicates when a carrier or signal is being received and the receiver is unsquelched. Sometimes called COS (Carrier Operated Squelch).
COS	Carrier Operated Squelch	See COR.
DHCP	Dynamic Host Configuration Protocol	A method of automatically assigning a dynamic IP address to an Ethernet device at startup time. DHCP conserves IP addresses in networks among devices that do not stay permanently connected.
DSP	Digital Signal Processor	A type of microprocessor, optimized for signal processing functions.
IP	Internet Protocol	A protocol designed to allow communications between computers on different networks.
LAN	Local Area Network	A group of computers and associated devices that share a common communications line, typically within a small geographic area. Compare with ***WAN***.
PBX	Private Branch Exchange	A private telephone system serving an office, campus, or similar setting.
PTT	Push-to-talk	A signal to a radio that activates the transmit function and controls the actual transmission of radio frequency energy over the air. Also called a keyline.
SIP	Session Initiation Protocol	A protocol for initiating, modifying, and terminating multimedia sessions.
SIP Phone		A telephone that uses SIP as its signaling protocol.
Soft Phone		A computer program that provides SIP Phone capability.
TCP	Transmission Control Protocol	An additional layer to the ***Internet Protocol***, which ensures delivery of packets sent across the network. It can resolve situations such as lost packets or packets arriving out of order. Compare with ***UDP***.
UDP	User Datagram Protocol	An additional layer to the ***Internet Protocol***, which does not ensure delivery of packets, but which offers more speed and lower transmission overhead than ***TCP***.
URI	Universal Resource Identifier	A string of characters that identify (name) a resource to facilitate interactions over a network.
VoIP	Voice over Internet Protocol	A method of sending voice communications across a digital network. Also called VoP (Voice over Packet).
WAN	Wide Area Network	A network that is spread out over a wide geographic area, such as around a city or state. It may include other public or shared networks. Compare with ***LAN***.

1 General Information

1.1 Scope

This instruction manual provides the information necessary to install and operate the ARA-1 ™ Radio-to-SIP Interface.

1.2 Description

The Raytheon JPS ARA-1 Radio-to-SIP Interface is a network device used for interfacing radio equipment to SIP networks, thereby extending the coverage and capability of these networks. It is comparable to an ATA (analog telephone adapter), which allows a standard telephone to operate on a SIP network; the ARA-1 provides the same capability to a radio. The ARA-1 makes special provision for the differences between radios and telephones. In particular, the half-duplex nature of radios and the control signals they require are accommodated by the ARA-1.

Designed for years of continuous operation in mission-critical applications and remote locations, the ARA-1 has no moving parts and requires no periodic shutdown or maintenance. Start up upon power on is typically less than 10 seconds.

1.2.1 General

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 to radio networks all of the features available with SIP. These include interoperable communication 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.

1.2.2 SIP-to-Radio Interface

The SIP side of the ARA-1 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. 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 or the Internet with another SIP-enabled device (such as a SIP Phone, a softphone, or another radio/ARA-1 pair). 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.

Figure 1-1 ARA-1 Basic Block Diagram

1.2.3 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 versatility of the format and content of the data being shared. Since SIP is a standards-based, open protocol, SIP system operators can pick and choose among third-party vendors when selecting existing or future applications to add to their systems. This avoids the anti-competitive, single-vendor “lock-in” that occurs with closed proprietary protocols.

1.3 Network Details

The ARA-1 is a 10/100BASE-T Ethernet device, and each unit has a unique Ethernet address and an RJ-45 physical interface jack. A 10/100BASE-T device operates at either 10 or 100 Mbps and interconnects to an Ethernet hub or switch using standard CAT 5 twisted pair cable, also known as UTP. The maximum cable length between an ARA-1 and its hub port is 100 meters. With the right connective equipment (recommended or supplied by JPS), the ARA-1’s Ethernet port can be linked with virtually any LAN, WAN, or the Internet, no matter which topology or cabling system is in use.

1.4 Applications

The ARA-1 can be used either as part of a SIP PBX or without one. Often the radio associated with the ARA-1 will be part of a repeater system. The features of each of these configurations are explained in the following sections. It is important to read all three of the following sections to fully understand the application of radios to SIP networks.

1.4.1 Operation within a SIP PBX

Figure 1-2 illustrates a basic ARA-1 application within a SIP PBX network. The PBX (typically a software application running on a server) assigns extensions (associated to the IP addresses) to each of the communications devices within the system. There can be any number of the end-user devices (Sip Phones, softphones, analog phone/ATA pairs, or radio/ARA-1 pairs) in the PBX.

When a SIP Phone user wants to place a call to another SIP Phone, he or she can simply dial that phone’s extension. The same process is followed to place a call to the radio: simply dial the ARA-1’s extension. The SIP Phone user does not need to understand the esoterics of basic radio operation; this is handled by the ARA-1.

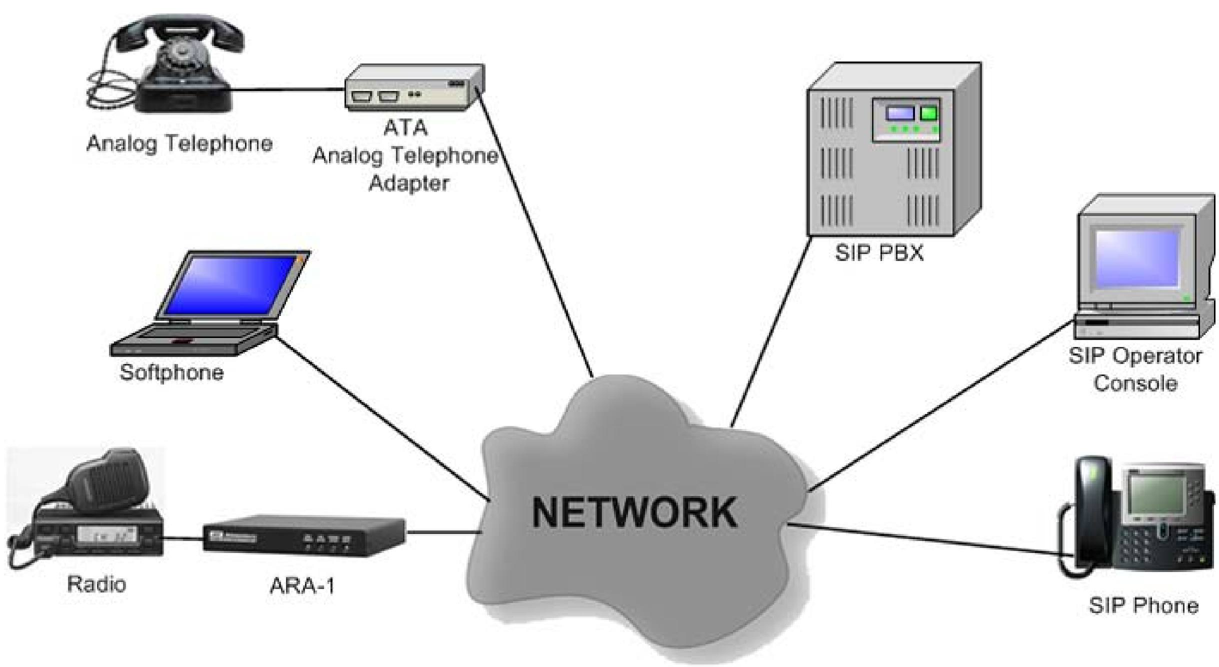

Figure 1-2 Example Of The ARA-1 in SIP PBX Network

Similarly, calls to the radio can be placed by the softphone or the analog telephone, interfaced by the ATA, simply by entering the extension assigned to the ARA-1.

The SIP PBX can provide a multitude of functions and features that expand and enhance the communications process. These include conferencing, voice mail, call logging, call forwarding, and essentially any other feature available with a commercial telephone service. Because the ARA-1 is based on the open-source SIP protocol, a wide range of PBX features are available from a wide range of sources.

NOTE: The PBX conferencing function, with multiple radio/ARA-1 pairs, provides network-based interoperability between disparate radio systems. For example, an 800 MHz trunked radio can be conferenced together with a P25 digital radio and a VHF conventional radio.

1.4.2 Operation Outside of a SIP PBX

The ARA-1 can also be used without a SIP PBX, but without the features provided by the PBX, so that only one-to-one connections are possible. Three variations are illustrated in Figure 1-3. The network employed can be any type of IP-based network, including the Internet.

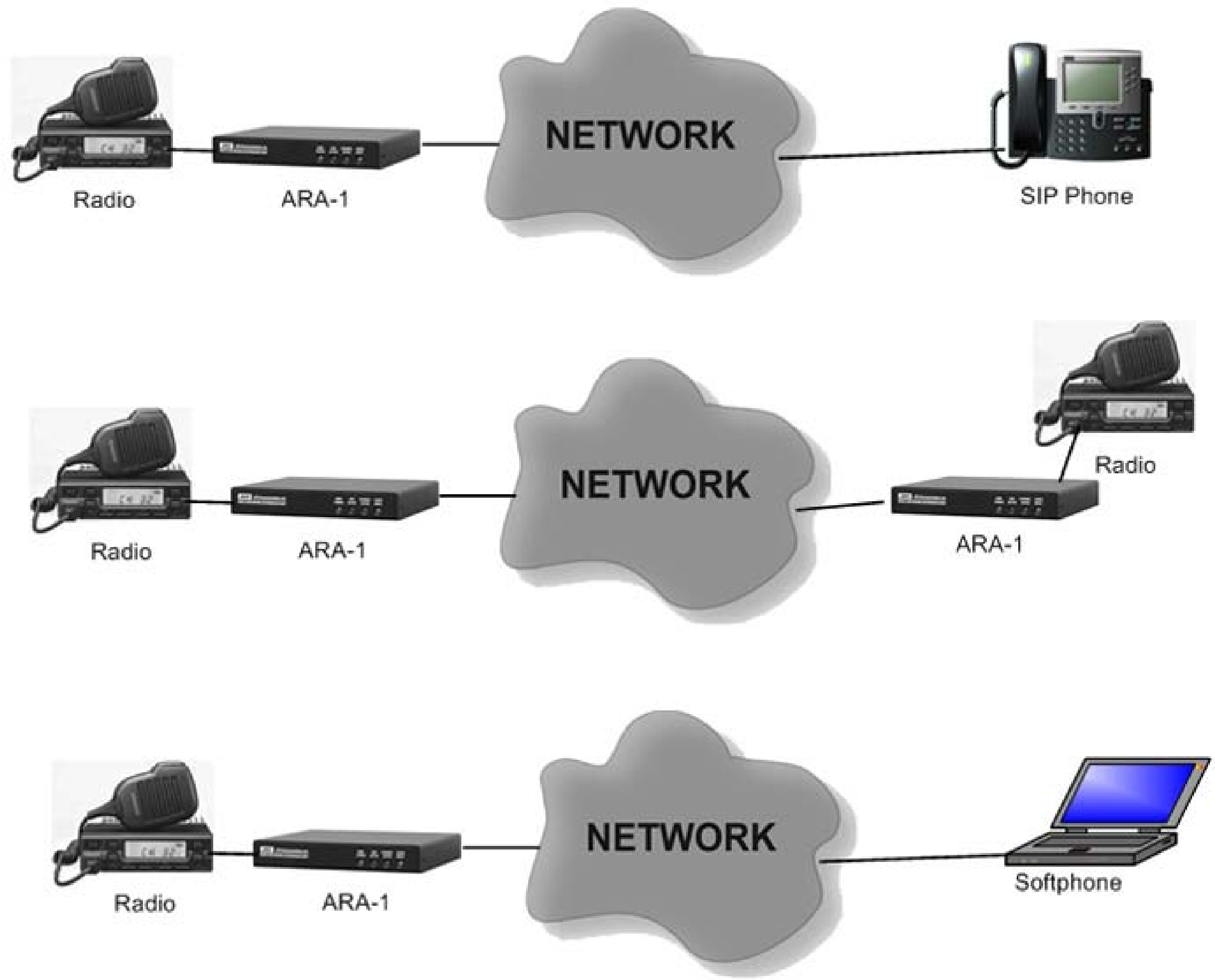

Figure 1-3 Examples Of ARA-1 Use Over Internet

The first setup in Figure 1-3 shows a connection between a SIP Phone and a radio. Without the extensions provided by the PBX, the SIP Phone "dials" the IP address of the ARA-1. See Section 1.6 for an overview of how to initiate connections via the ARA-1 rather than by the SIP Phone.

> **NOTE:** The radio shown in the first illustration in Figure 1-3 may be set as "receive-only" and used to monitor information and pass it over the network. For example, it may be receiving local weather or traffic reports or scanning frequencies set aside for public safety use.

The second setup in Figure 1-3 shows a pair of radio systems connected via the ARA-1 technology and the network. These radio systems may be on opposite sides of the country, or they may be collocated, but on different frequencies or bands.

The third setup shows a connection between a softphone and a radio through the ARA-1. This operation is identical to the SIP Phone application.

1.4.3 Use of the ARA-1 with a Repeater System

It is important to note that the ARA-1 makes a connection not to a single radio or single radio user, but to an entire *radio system*. All signals picked up by the radio are relayed through the ARA-1, and more importantly, transmissions through the ARA-1 are sent to all radios that can pick up this transmission.

In many cases, the ARA-1 and radio will be used within a repeater system. The radio associated with the ARA-1 will create an RF link to a repeater system as illustrated in Figure 1-4.

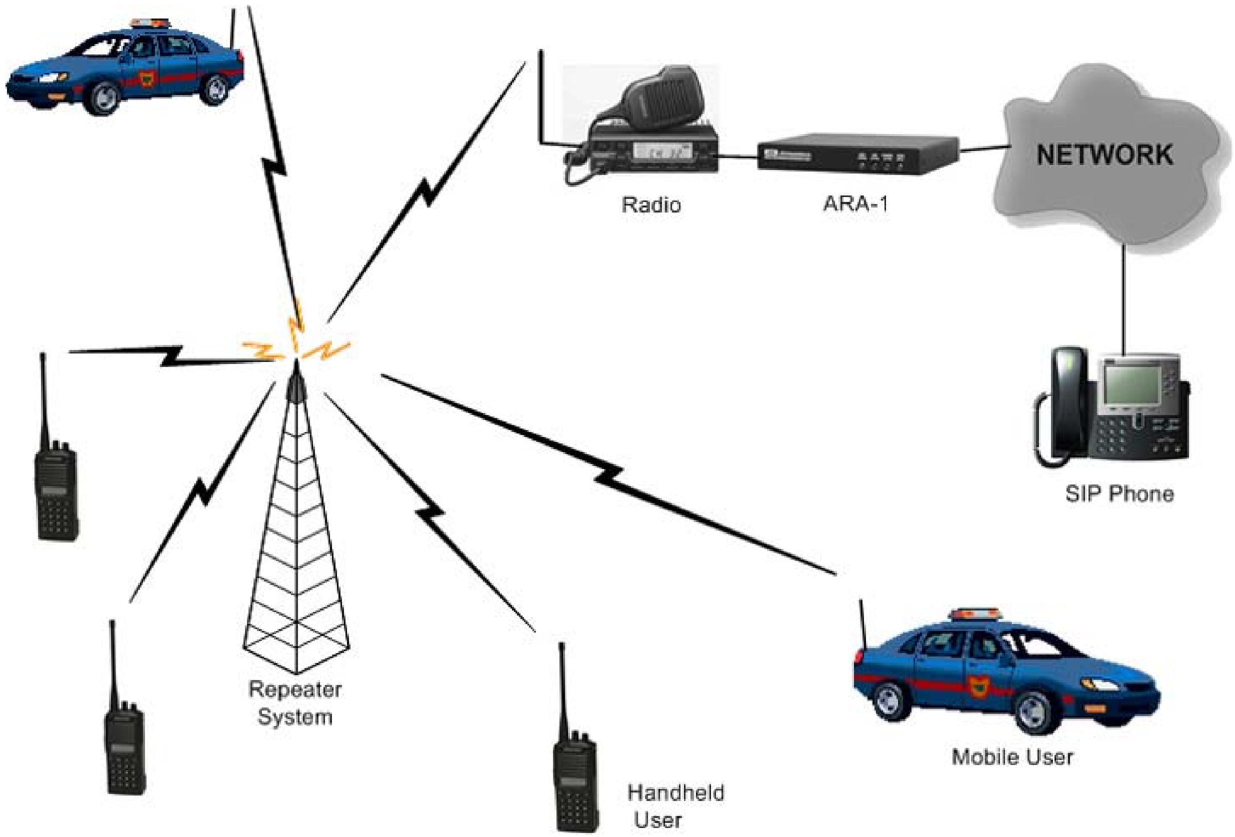

Figure 1-4 ARA-1 with a Repeater System

Whenever the SIP Phone user connected to the radio/ARA-1 pair speaks, the radio transmits. This transmission is picked up by the repeater system, which retransmits it to all of the mobile and handheld radio users in the system. Similarly, all of their transmissions are retransmitted by the repeater system to be received by all system users, including the radio associated with the ARA, so all of this traffic can be heard by the SIP Phone user.

This concept applies to both trunked and conventional repeater systems. Note that no changes to the repeater system are required, hence there is no installation or other downtime involved with using the ARA-1 to create a single SIP network connection to the system, or to use multiple ARA-1s to conference two or more radio systems together.

1.5 Connection to Devices Other Than a Radio

The ARA-1 contains all of the interface features necessary for a competent interface between a radio and a SIP network. These features also provide a seamless interface of other four-wire devices such as an audio console.

1.6 Initiating Connections via the ARA-1 and Associated Radio

So far calls to a radio through the ARA-1 have been described. There are three methods for initiating a call from the radio/ARA-1 end of the connection. Each of these methods is described in full detail in Section 4, *Operation*.

1.6.1 Using a Web Browser

Simply browse to the ARA-1's IP address, select the *Call Management* page, enter the SIP address or extension, and click *Connect*.

1.6.2 Using DTMF

If your radio has a DTMF keypad, there are two options that can be used to initiate connections to an IP address or a PBX extension. DTMF input not terminated by a pound (#) digit invokes an internal "speed dial" calling guide that is set up within the ARA-1; this calling guide associates incoming DTMF sequences with the destination IP address or extension.

Whenever the DTMF input is terminated by the pound digit, the ARA-1's *Pound Terminated Dialing* feature dials the received DTMF sequence (except for the pound digit). This allows a radio user with a DTMF keypad to dial any number or extension just as the user would dial a regular telephone.

1.6.3 Using Squelch Breaks

The ARA-1 can use the COR (unsquelched condition) input from its associated radio as a signal from radio system users that they want to make or end a call. The radio users in the field key and unkey their radios at a specific cadence (user-programmable). For example, the required cadence may be four key/unkey sequences at the specified rate (three or five will not trigger a response). The radio associated with the ARA-1 unsquelches and re-squelches at the same cadence, and passes this cadence on to the ARA-1. When the required cadence is detected, the ARA-1 initiates a call to a pre-defined SIP address or extension. A COR Cadence can also be used to terminate the call.

1.7 SIP Instructions

This manual does not attempt to familiarize the reader with SIP fundamentals. SIP is an open protocol and there are many references that explain how to best make use of it. The ARA-1 is fully compliant with the SIP protocol.

1.8 Specifications

Table 1-1 Specifications

Radio RX Audio Input	
Input Impedance	Balanced 47k ohms, transformer coupled
Input Level	Incoming signals adjustable from –30 to +11 dBm to set 0 dBm nominal input; +15 dBm clipping; +20 dB boost configurable
Frequency Response	10 Hz to 3600 Hz +/- 2dB
Radio TX Audio Output	
Output Impedance	Unbalanced 600 ohms, AC Coupled
Output Level	Adjustable from –30 to +11 dBm, 0 dBm nominal factory default; +9 dBm clipping into a 600 ohm load
Frequency Response	10 Hz to 3350 Hz +/- 2dBm
Distortion	0.5% or less (excepting Vocoder)

Table 1-1 Specifications

Radio COR and AUX Inputs	
Input Impedance	47k ohm pull-up to +5V
Polarity	COR: Selectable active low or active high; AUX Inputs: Active low
Threshold	+2.5V nominal
Protection	Up to + 100 VDC
Radio PTT and AUX Outputs	
Output Type	Open drain, 47k ohm pull-up to +5V
Maximum Sink Current	100 mA
Max Open Circuit Voltage	+60 VDC
Network Interface	
Interface Type	10/100BASE-T Ethernet, 10 or 100 Mbps; RJ-45 Connector
Protocols	SIP, SDP, RTP, STUN
Audio Vocoder	Selectable, 13 or 64 Kbps data rate
General/Environmental	
Programming/Configuration	Web, Telnet, or RS-232 Interface
Front Panel	Power, Link Active, Channel Active, and Audio Level LEDs
Rear Panel	Audio/Data, Serial, Network, and Power Connectors
Audio/Data Connector	DB-15 Female
Input Power (12 VDC Nom)	+11 to +15 VDC at 0.5A max. 12VDC (wall-cube supplied)
Power Connector	Coaxial Jack, 2.5 mm ID, 5 to 5.5 mm OD; Center Pin Positive; Reverse Polarity Protected
Size and Weight	1.3”H x 8.3”W x 6.7”D (3.3 x 21.1 x 17.0 cm); 1.1 lbs. (0.5kg)
Temperature	Operating: -20 to +60 degrees C; Storage: -40 to +85 degrees C
Humidity	Up to 95% at 55 degrees C
Shock	MIL-STD-810D, Method 516.3, Procedure VI
Vibration	MIL-STD-810D, Method 514.3, Category 1

1.9 Equipment and Accessories Supplied

Table 1-2 Equipment and Accessories Supplied 120 VAC Version

ARA-1 Shipping Level - JPS P/N 5060-800000		
Quantity	**Item**	**JPS P/N**
1	ARA-1 Final Assembly Includes the ARA-1 enclosure with the ARA-1 PCB Assy	5060-801000
1	DC Power Supply [100 to 240 VAC, 47-63 Hz to +12 VDC, 500 mA]	1620-120600
1	Operation and Maintenance Manual	5060-800200
1	Accessory Kit	5060-800150
	Consisting of: *Qty* *Part Number* *Description* 1 0313-070000 Network Cable, 6 ft. 1 0313-080515 Audio Crossover Adapter (for use with JPS radio interface cables)	

1.10 Optional Equipment: Not Supplied

Table 1-3 Optional Equipment - Not Supplied

Description	**JPS P/N**
Generic Radio Interface Cable; unterminated at radio end; specify length: 15, 30, or 50 feet	5961-291115- 15/30/50
Interface cables for a very wide range of commercial radios are available for purchase Consult the Raytheon website at the link below and look for the cable list link on the right. Radio Interface Cable List The url is http://www.raytheon.com/businesses/ncs/civilcomms/ics/index.html	

End of Section 1

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2 Installation

2.1 General

This section provides the instructions for unpacking, inspection, installation, and setup. Also included are directions for reshipment of damaged parts or equipment.

2.2 Unpacking and Inspection

After unpacking the unit, retain the carton and packing materials until the contents have been inspected and checked against the packing list. If there is a shortage or any evidence of damage, do not attempt to use the equipment. Contact the carrier and file a shipment damage claim. A full report of the damage should also be reported to the JPS Customer Service Department. The following information should be included in the report:

- Order Number
- Equipment Model and Serial Numbers
- Shipping Agency
- Date(s) of Shipment

The Customer Service Department can be reached by phone at (919) 790-1011, by fax at (919) 790-1456. Upon receipt of this information, JPS will arrange for repair or replacement of the equipment.

2.3 Reshipment of Equipment

If it is necessary to return the equipment to the manufacturer, an RMA (Returned Material Authorization) number must first be obtained from JPS. This number must be noted on the outside of the packing carton and on all accompanying documents. When packing the unit for reshipment, it is best to use the original packaging for the unit; if this is not possible, make sure that adequate packing material is used to prevent excessive shocks during transport and handling.

Shipment should be made prepaid consigned to:

JPS Communications

Customer Service Department

5800 Departure Drive

Raleigh, North Carolina 27616

USA

Plainly, mark with indelible ink all mailing documents as follows:

GOODS RETURNED FOR REPAIR

Mark all sides of the package:

FRAGILE - ELECTRONIC EQUIPMENT

Inspect the package prior to shipment to be sure it is properly marked and securely wrapped.

2.4 Installation Overview

NOTE: ARA-1 installation requires knowledge of Ethernet network fundamentals as well as a basic understanding of IP (Internet Protocol). As with any network-connected device, improperly configuring and installing the ARA-1 could disrupt proper network operation. Please seek the assistance of your network administrator or other knowledgeable person if you are unsure about how your network is configured.

Four steps are needed to properly install the ARA-1. These steps are:

1. Determine the desired IP address, subnet mask, and (if applicable) the gateway address for the unit. You may have to contact the network administrator for your organization to obtain this information.

NOTE: Operation of the ARA-1 is not possible without this information.

2. Provide the proper primary power for the unit.

NOTE: Use only the Class 2 power supply provided with the equipment.

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3. Interconnect the unit with the communications system via the ARA-1's rear panel connectors. J7 provides the audio and control lines necessary to interface the ARA-1 to your audio equipment. Radio interface cables for most common makes and models can be purchased from JPS Communications.

4. Configure the unit’s operational parameters per Sections 2.8.3 (rear panel audio level adjustments) and 3.2 (system configurations set by web browser).

2.5 Installation Considerations

Careful attention to the following installation suggestions should result in the best unit/system performance. Figure 2-1 provides overall unit dimensions.

The ARA-1 must be installed in a structure that provides both protection from the weather and assurance of ambient temperatures between -20 and +60 degrees C. Since the unit is neither splash proof nor corrosion resistant, it must be protected from exposure to salt spray. When the unit is mounted in a cabinet with other heat-generating equipment, the use of a rack blower is suggested to keep the cabinet interior temperature within specifications.

NOTE: If the ARA-1 is installed in a high RF environment such as a repeater system or other transmitter site, it is recommended that all cable assemblies be individually shielded, with the shield grounded to the ground pin on the terminal block for that module. For all D-subminiature connector cable assemblies, cable shields should be connected to connector shells so that they make contact with the grounded D-subminiature connector shells on the ARA-1.

NOTE: For the DC input, the plug is the equipment disconnect device.

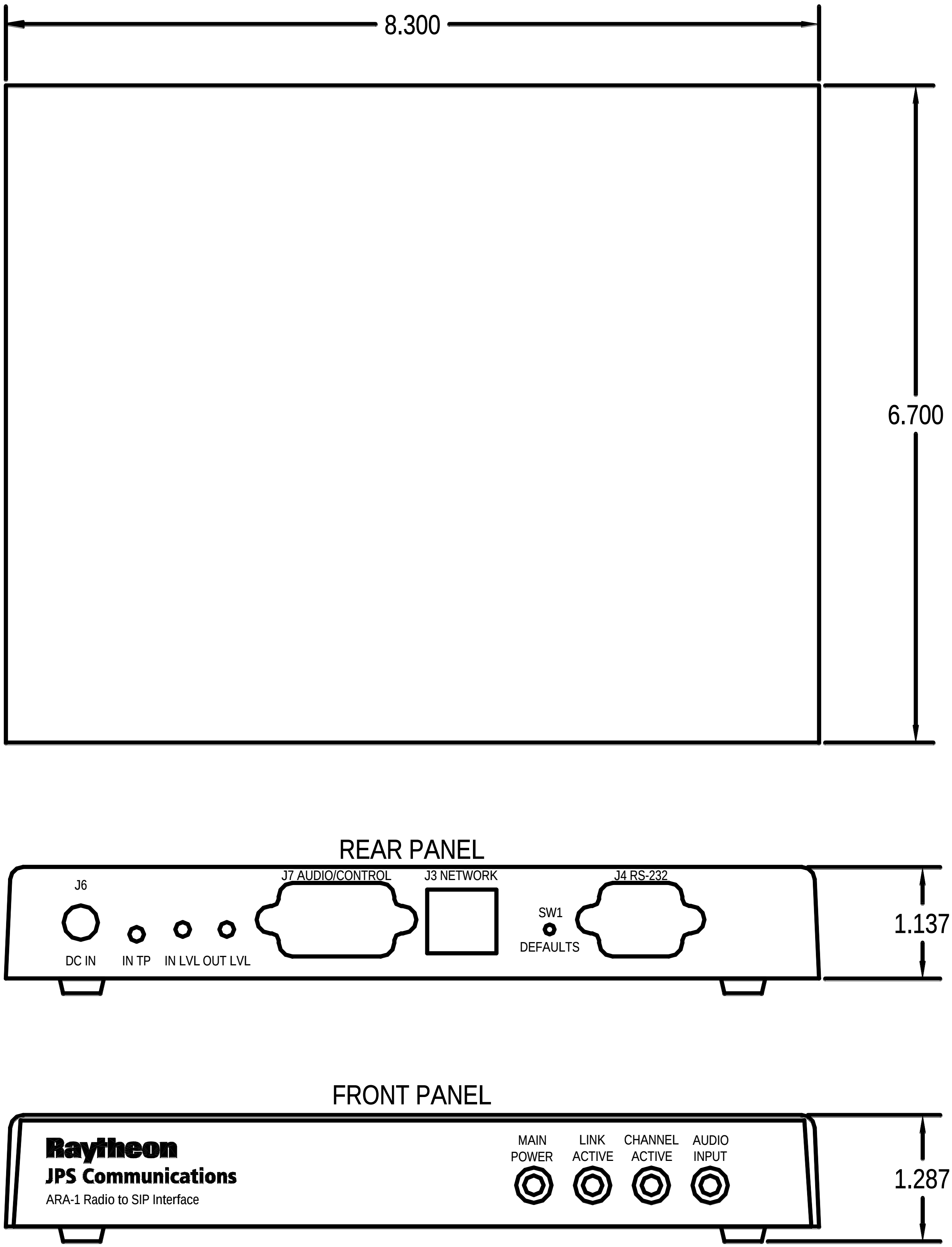

Figure 2-1 Outline Dimensions

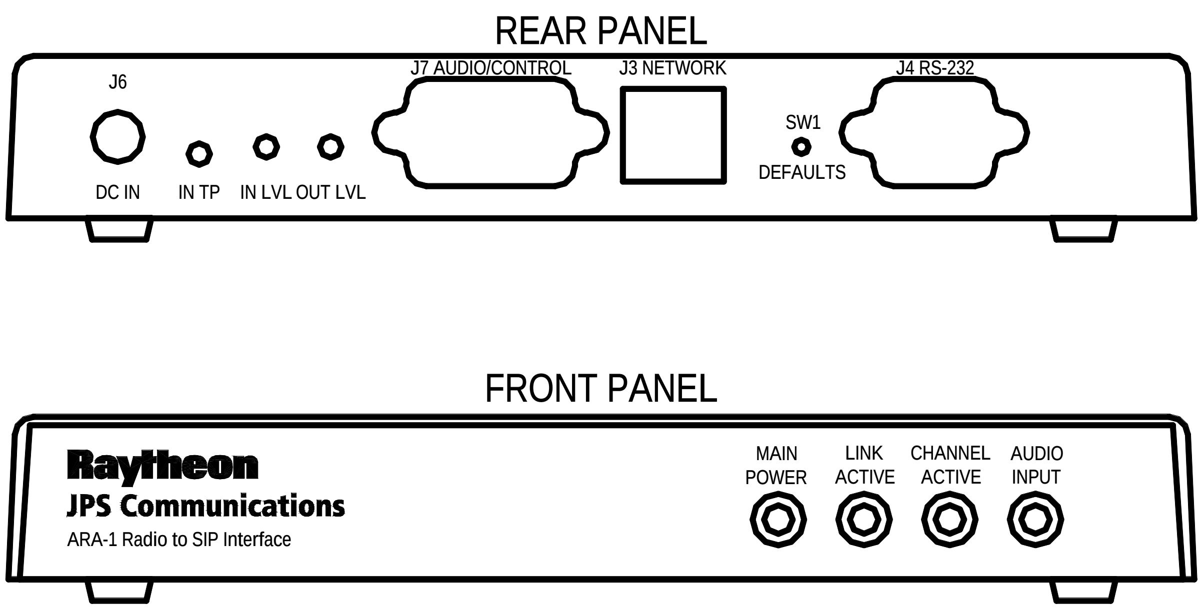

Figure 2-2 Front and Rear Panel Views

2.5.1 Internal Configuration

There are no internal user-configurable components, switches, or other controls. There should be no need to ever open the ARA-1 case.

2.6 Power Requirements

The ARA-1 is designed to operate from a nominal +12V DC supply. The unit will meet all of its specifications over a voltage range of +11 to +15 VDC and will be damaged by a DC source that delivers a constant (non-transient) DC voltage above this range. The DC power consumption is 500 mA maximum. The AC adapter provided with the unit meets these specifications.

NOTE: Use only the Class 2 power supply provided with the equipment.

The ARA-1 is a microprocessor-controlled device. As with any such equipment, a very short loss of power can cause operational problems and/or cause the unit to reset. The communications link will be inoperable during the reset period. JPS recommends the ARA-1 and associated equipment be connected to an AC power source that utilizes a UPS (uninterruptible power system). If the overall site does not have UPS capability, the ARA-1 should be plugged into a smaller UPS, such as those used for personal computer systems.

2.7 Installation Checklist

Table 2-1 Basic Installation Checklist

Determine ARA-1 network parameters such as IP address, subnet mask, and gateway address.	You must assign these values. If you are not sure how to determine these values, see the network administrator for your organization.
Provide suitable power for the device.	See Section 2.6.
Make interconnections.	See Section 2.8.
Adjust audio levels.	See Section 2.8.3.
Configure ARA-1 parameters.	See Section 3.

2.8 Rear Panel Adjustments and Connectors

Refer to Figure 2-2 for a view of the ARA-1 rear panel. All rear panel connectors and adjustment potentiometers are explained below, starting at the left side of the panel.

2.8.1 DC Input Connector (J6)

The ARA-1 operates on a nominal +12 VDC. The power is applied through J6 via the “wall cube” AC adapter provided with the unit.

2.8.2 Connection to Radio or Other Four-Wire Device (J7)

The interface between the ARA-1 and associated radio or other audio device is made via J7 (Audio/Control) on the rear panel. J7 is a female DB-15 connector.

Table 2-2 ARA-1 Pinout (J7)

PIN	Signal	Description
1	Ground	Ground connection
2	N/A	Not used
3	/AUX In 0	Auxiliary Input 0 - Active low
4	/AUX Out 0	Auxiliary Output 0 - Active low
5	Ground	Ground connection
6	Audio Input	Balanced audio input
7	Analog Ground	Analog ground
8	Audio Output	Unbalanced Audio output
9	N/A	Not used
10	/AUX In 1	Auxiliary Input 1 - Active low; general purpose
11	/AUX Out 1	Auxiliary Output 1 - Active low; general purpose
12	/COR Input	Input from radio COR, programmable active high or low
13	/PTT Out	Output to radio PTT, active low, open drain
14	Audio Input	Balanced audio input
15	Analog Ground	Analog ground

NOTE: To interface unbalanced "single-ended" audio, connect the audio to one of the two balanced audio inputs and ground the other. Interface cables purchased from JPS Communications handle the unbalanced/balanced audio issue.

2.8.3 Audio Level Adjustment Potentiometers and Input Test Point

The audio input level to the ARA-1 is set by adjusting the IN LVL control on the rear panel. With "normal radio receive audio" input applied at J7, adjust the IN LVL control until the AUDIO INPUT indicator flashes on voice peaks.

NOTE: "Normal radio receive audio" means the audio output that results when receiving a fully-quieted (on frequency) speech signal from someone talking at a typical speaking volume.

The OUT LVL control sets the audio output level from the ARA-1 and may be adjusted to the level suitable for the equipment connected to the unit. If necessary, the input audio level can be further boosted within the unit's software (see Section 3.3.6).

A test probe may be inserted into the test point to measure the level of the incoming audio. The proper audio input level may also be set by connecting an AC voltmeter to the test point TP1 on the real panel and adjusting the IN LVL control for an average audio level of about 0.2V or –12dBm.

NOTE: An Audio Crossover Adapter, part number 0313 080515, is included with the ARA-1. This DB-15 male to DB-15 female adapter allows the use of radio cables developed specifically for the JPS ACU-1000 Intelligent Interconnect system to be used with the ARA-1. It provides a crossover of transmit and receive audio, as well as COR and PTT control signals. You only need this adapter if you are planning to connect a radio to the ARA-1 using a JPS designed or manufactured radio interface cable. The adapter makes the ARA-1 audio connector pinout match the one found on the ACU-1000. If you are designing a cable based on Table 2-2, or if you are connecting the ARA-1 directly to an ACU-1000, then you do not need the Audio Crossover Adapter.

2.8.4 Network Connection (J3)

The ARA-1 is connected to the Ethernet network via rear panel connector J3 using a standard RJ-45 Ethernet Patch Cable (non-crossover). A six-foot long cable is included with the unit.

2.8.5 Serial Port Connection (J4)

J4 is a standard RS-232 DCE serial port. It is a female DB-9 connector, and can be interfaced to most PCs, typically standard DTE serial ports, using a DB-9 straight-through serial cable (not included with the ARA-1).

NOTE: This connector is used only during factory setup.

Table 2-3 J4 Serial Port Pinout

J4 pin	Description
2	TX data
3	RX data
5	Ground

End of Section 2

3 Configuration

3.1 General

This section explains all settings and level adjustments that configure the ARA-1 other than the rear panel potentiometer audio level adjustments described in Section 2.8.3. It is not necessary to remove the ARA-1 cover to configure the unit.

The instructions are broken down into two main sections:

- Interfacing the unit to the SIP network
- Interfacing the unit to a radio or other four-wire device

3.2 Configuration Details: Network Interface

Configuration is performed by connecting to the unit's IP address with a web browser.

The ARA-1 comes from the factory configured with the following default settings:

- IP Address: 192.168.1.200
- Subnet Mask: 255.255.255.0
- Gateway IP: 0.0.0.0

If these settings are compatible with your network, you may plug the ARA-1 into your Ethernet switch and proceed with the configuration. If you wish to configure the ARA-1 with a standalone computer, you should set your computer's network settings to allow communication with these defaults. See your network administrator if you need assistance with your computer settings.

NOTE: If you connect your computer directly to the ARA-1 (without an Ethernet switch), you will need to use an Ethernet crossover cable instead of a standard Ethernet cable.

NOTE: To restore the factory default conditions, depress the rear panel switch *SW1 (DEFAULTS)* for 5 seconds while unit power is on. All parameters (including the unit's IP address) will be returned to JPS factory defaults. ***Any previously assigned user parameters will be lost during this process.***

3.2.1 Basic Unit Status and Information

Apply power to the ARA-1, connect your Ethernet cable, and start a web browser on your computer. Enter 192.168.1.200 in the address field of your browser (or other IP address if the unit's configuration has been changed). If your settings and connection are correct, you should see the page shown in Figure 3-1.

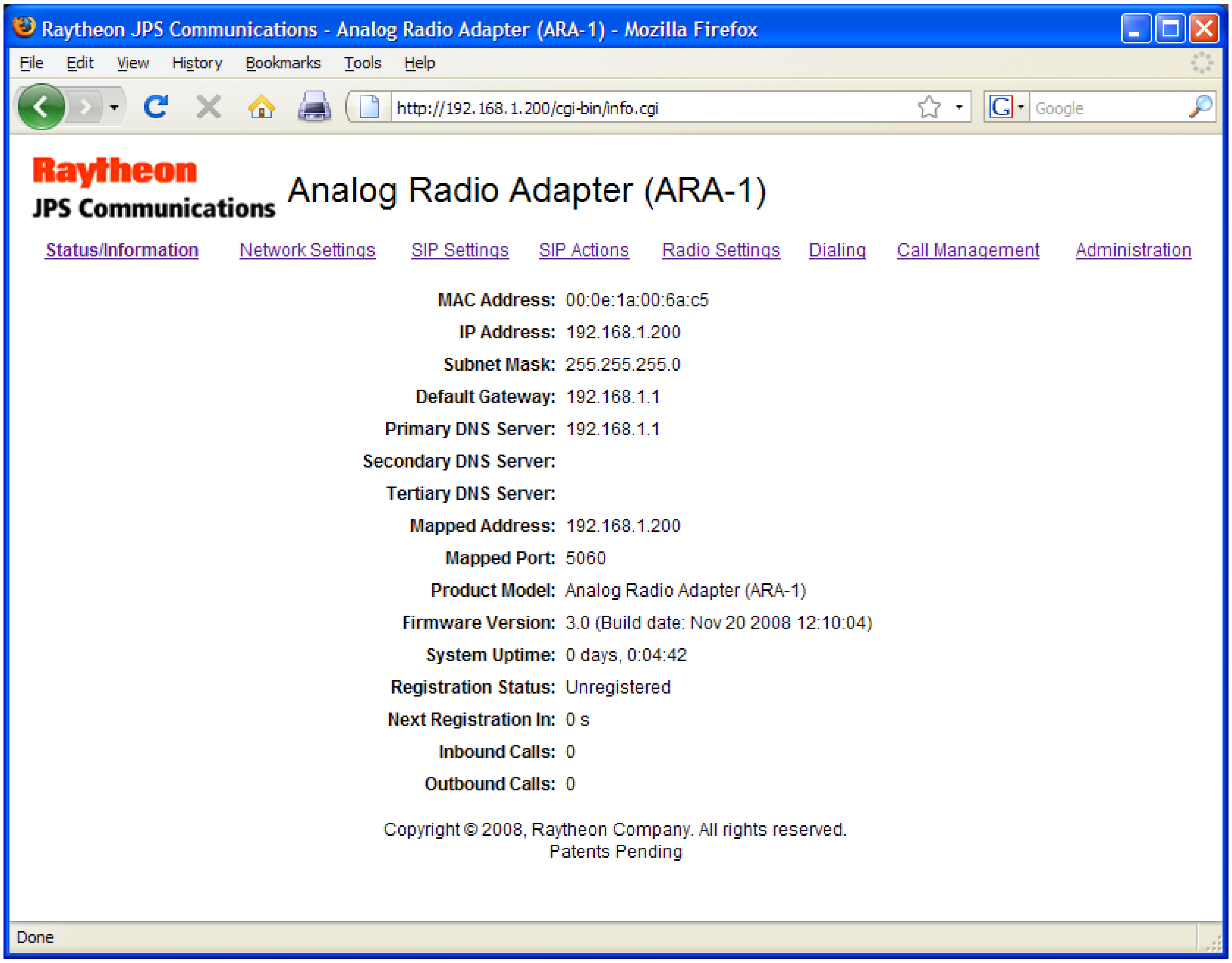

Figure 3-1 Status/Information Page

This is the status and information page for the ARA-1 and the page you are greeted with when you browse to the unit. It shows the current network settings as well as some other status information, such as the version of the firmware currently loaded in the ARA-1. Now click the *Network Settings* link to go to the next page.

3.2.2 Network Settings

The Network Settings page is where all network parameters are set.

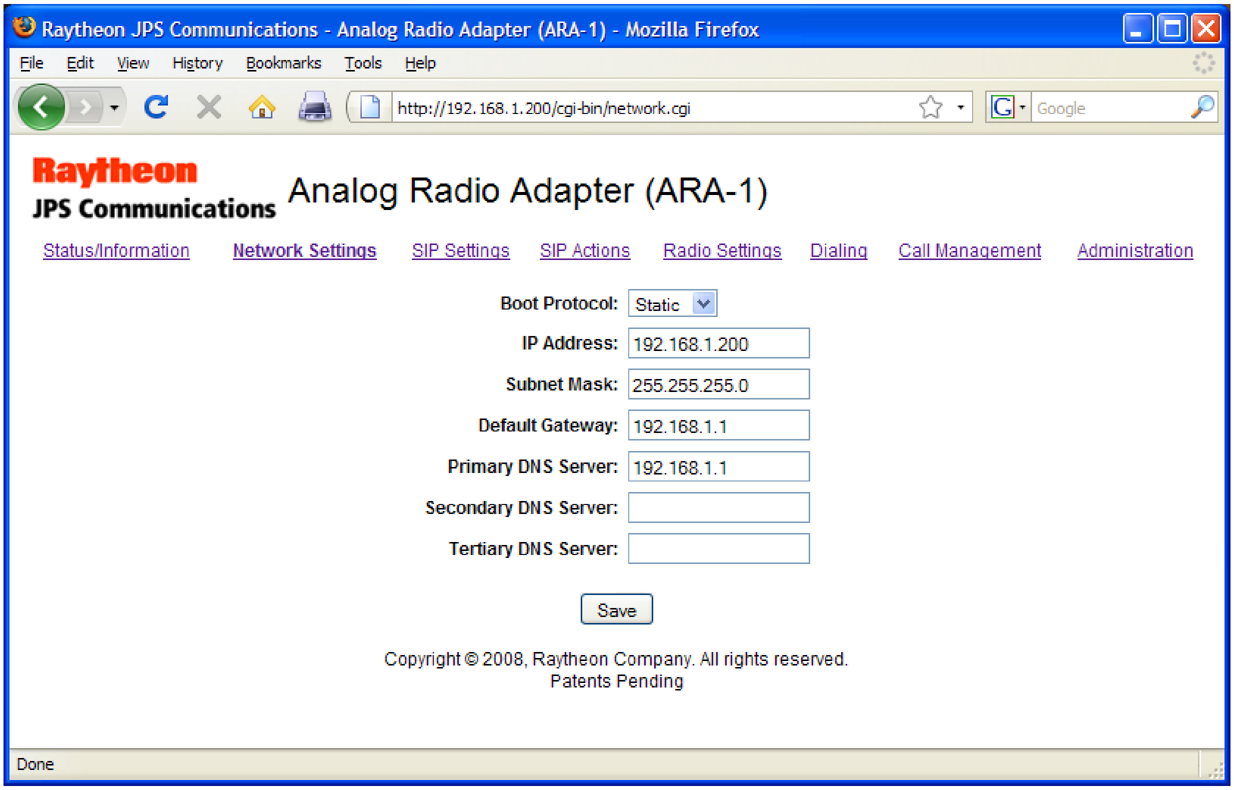

Figure 3-2 Network Settings Page

If you select *Static* from the Boot Protocol drop-down menu, then you must adjust the other settings to match your particular network. If you select *DHCP* as your boot protocol, then your local DHCP server will assign these values for you. When you have made any necessary changes, click *Save* at the bottom of the page. These settings are not actually applied until the unit is restarted, so you can continue to make other changes if necessary. Now click on *SIP Settings* for the next page.

3.2.3 SIP Settings

The SIP Settings page is where you will configure the SIP settings for this device. The ARA-1 can register with a SIP PBX, or it can operate as a standalone device.

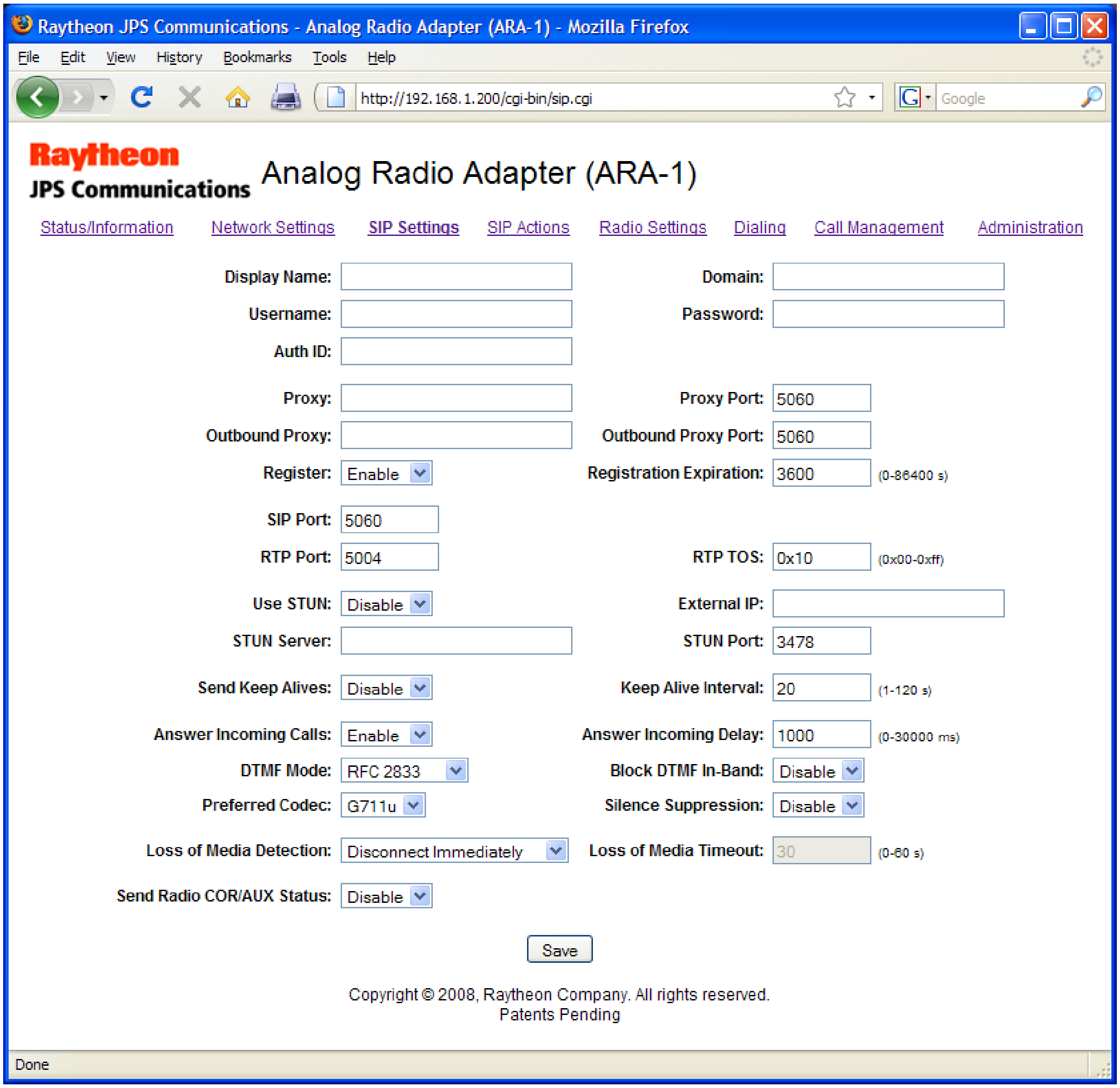

Figure 3-3 SIP Settings Page

The settings shown in Figure 3-3 are the same ones that would be set for any SIP endpoint such as a SIP Phone. An exception is the *Send Radio COR/AUX Status* setting. If set to *Enable*, the status of the COR and AUX Input pins on the ARA-1 rear panel will be sent across the IP link. This is useful if your radio has a hardware squelch line (COR) and you are linking to other ARA-1 units. Sending COR Status will tell the other units when the radio is unsquelched, and the other radios can assert their PTT control output lines if they are part of the connection. This is a more sure and timely method than the use of VOX or VMR as the network audio gating function at the other ARA-1 units. See also Section 3.3.5.

Table 3-1 SIP Settings Options	
Settings Option	**Description**
Display Name:	The name displayed on a remote SIP Phone when it connects to the ARA-1.
Domain:	The unit's SIP domain (if needed). The domain portion of the unit's URI.
Username:	The SIP user name or extension. The username portion of the unit's URI.
Password:	The password used for authentication when required.
Auth ID:	The user ID used for authentication when required and different from the username.
Proxy:	SIP proxy server address. Can be a name (e.g. mysip.com) or IP address.
Proxy Port:	The port number of the specified SIP proxy server.
Outbound Proxy:	SIP proxy server used for outbound calls if separate from the primary SIP proxy used for registration.
Outbound Proxy Port:	The port number of the specified outbound SIP proxy server.
Register:	Enable/disable registration with SIP proxy server.
Registration Expiration:	Time interval between successful registrations with the SIP proxy.
SIP Port:	The local port number for SIP packets. Usually same as the Proxy port.
RTP Port:	The local port number for RTP packets.
RTP TOS:	Value to set in RTP packet TOS IP header field for QOS applications.
Use STUN:	Enable/disable the use of STUN to discover the device's external IP address.
External IP:	Hard coded external IP address to use.
STUN Server:	The name or IP address of the STUN server to use.
STUN Port:	The port number of the STUN server.
Send Keep Alives:	Enable/disable the sending of SIP keep alive packets.
Keep Alive Interval:	Interval (in seconds) to send SIP keep alive packets.
Answer Incoming Calls:	Allows the unit to ignore incoming calls or answer them automatically.
Answer Incoming Delay:	Allows the unit to wait for the specified amount of time (0 to 30,000 msec) before answering an incoming call.
DTMF Mode	Mode to use for sending DTMF during a call.

Table 3-1 SIP Settings Options	
Block DTMF In-Band:	Block DTMF in the audio stream when using a DTMF Mode other than In-Band.
Preferred Codec:	The voice compression type the ARA-1 offers for outgoing calls. Available options: 13 kbps GSM or 64 kbps G.711u (default).
Silence Suppression:	If disabled, packets will be sent even during audio silence.
Loss of Media Detection:	Action to take when a loss of the media stream is detected during a call. Options are Disable (do nothing), Disconnect Immediately (hang up the call), or reINVITE then Disconnect (Send a reINVITE to try to reestablish the call and then hang up if that fails).
Loss of Media Timeout:	Number of seconds media is lost before performing the configured action.
Send Radio COR/AUX Status:	If enabled, COR/AUX input status will be sent via the RTP extension header.

When the SIP settings have been entered, click *Save* at the bottom of the page. Then click *SIP Actions* for the next page..

3.2.4 SIP Actions

SIP Actions is a mechanism by which the hardware outputs of the ARA-1 (PTT, AUXOUT0, and AUXOUT1) may be controlled by a remote SIP endpoint. If so configured, the proper DTMF sequence, when detected by the ARA-1's network interface, will turn these outputs on or off. Use the *SIP Actions* page (shown in Figure 3-4) to configure these DTMF sequences.

The *DTMF Digit Timeout* entry determines how the ARA-1 decides whether a detected DTMF digit is part of the current DTMF sequence or the start of a new one. If the time between the end of one digit and the start of the next is less than the DTMF Digit Timeout, that character will be considered part of the current DTMF sequence and appended to the digits already detected. As soon as a pause is measured that is longer than the timeout entry, the current DTMF sequence will be considered complete. The factory default setting—a duration of one second (1000 ms)—should work for most systems.

In order to control the PTT output of the ARA-1 using *SIP Actions*, the *Network COR Type* setting on the *Radio Settings* page (see Figure 3-5) must be set to *SIP Actions*.

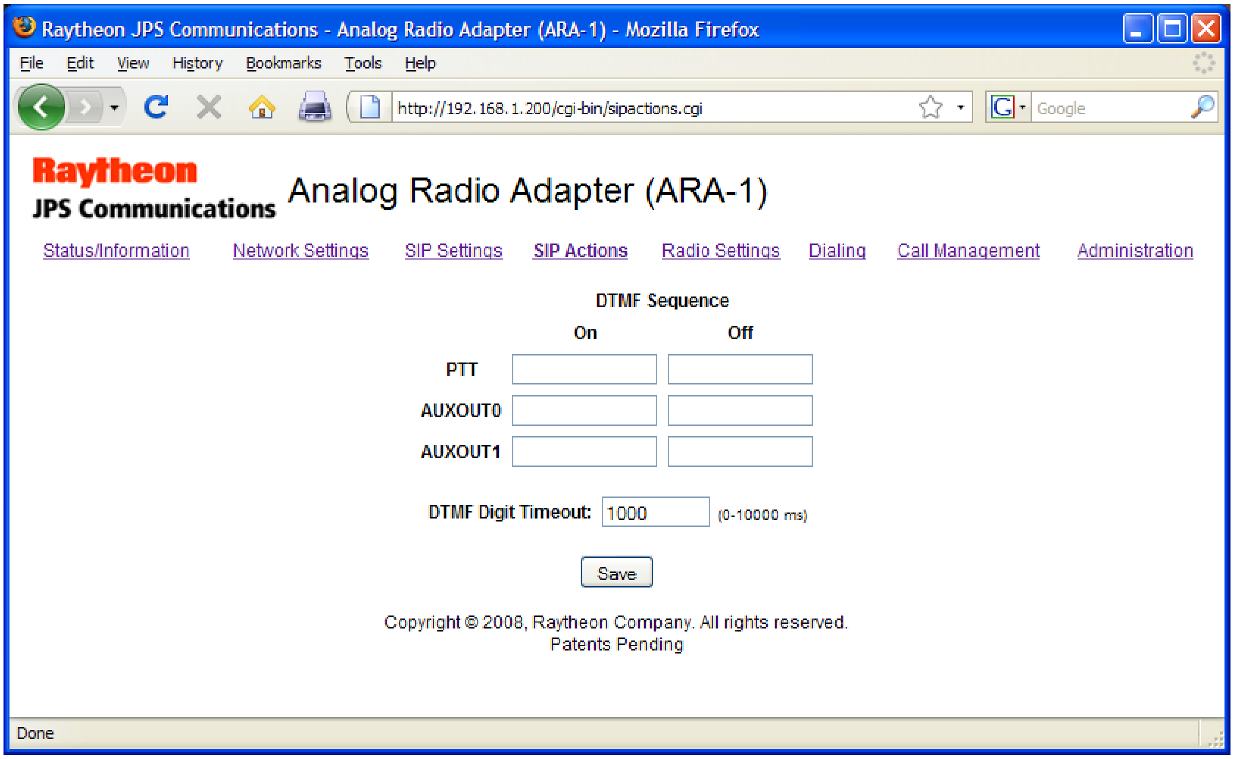

Figure 3-4 SIP Actions Page

3.3 Configuration Details: Radio Interface

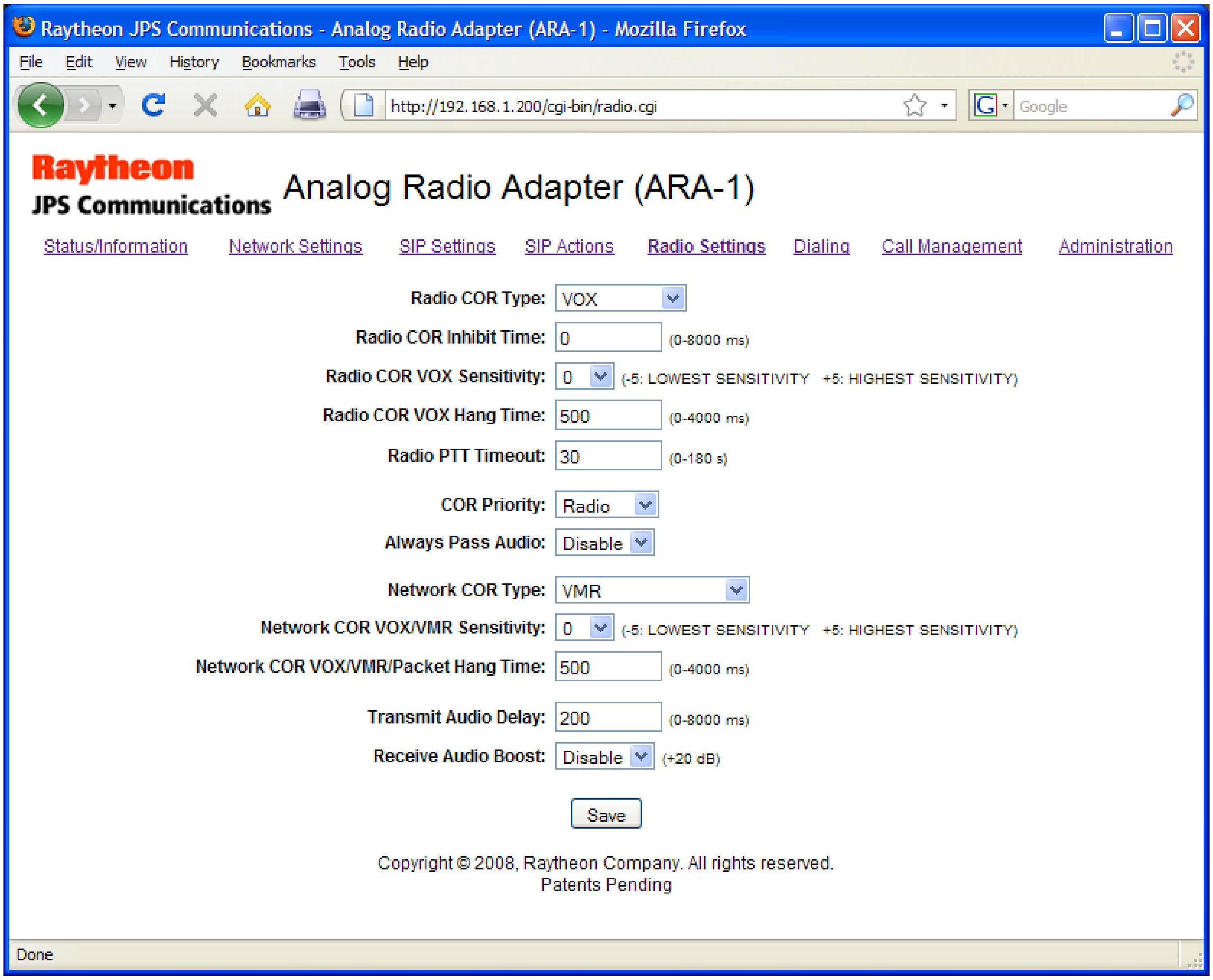

Figure 3-5 Radio Settings Page

Included on the *Radio Settings* page (shown in Figure 3-5) are all the settings that affect the interface to, and operation of, the radio cabled to the ARA-1. This page is used to configure and optimize the unit for best performance in a particular radio application. Each of the settings is explained in this section; the default settings are shown in Figure 3-5.

3.3.1 Radio COR Settings Options

The ARA-1 must know when its associated radio is receiving a valid signal; it uses this to determine when it should send whatever audio is present on the radio's receive audio output lines across the SIP network. The Radio COR settings all ensure that this function is optimized.

3.3.1.1 Radio COR Type

The *Radio COR Type* setting determines which method the ARA-1 will use to determine when the radio is receiving a valid signal, then the radio/ARA-1 pair will be put into the unsquelched mode [also called "open squelch"] and send the radio RX audio to the SIP network. The ARA-1 can either use the COR output signal from the radio or use VOX (see Section 3.3.1.3). The COR output is a control signal from the radio that activates when the squelch opens. If this line is available, connect it to the ARA-1 COR input and select *ACTIVE HIGH* or *ACTIVE LOW*, depending on whether this line asserts a voltage when the radio is receiving (active high) or pulls to ground when receiving (active low). This control line may also be called *COS* or simply *SQUELCHED* or *UNSQUELCHED*.

If the radio does not have a COR output control line, select *VOX*. When the ARA-1 is in VOX mode, it measures the volume of the sound available in the RX output from the radio. Whenever this audio exceeds a set threshold, the VOX trips, signaling the unsquelched condition (see *Radio COR VOX Sensitivity*, Section 3.3.1.3). When using the VOX mode, adjust the squelch on the radio so that no noise is produced unless the radio is actually receiving a signal. FM radios that are running at full "open squelch" output a high volume of noise when there is no carrier present, and this noise will inappropriately trip the VOX function.

> **NOTE:** The Applications Notes that are provided with radio interface cables purchased from JPS will identify whether a COR line is available, and if so, whether its sense is active high or active low.

3.3.1.2 Radio COR Inhibit Time

In some radios, the COR line activates momentarily when the radio reverts to receive from transmit. Even if a hardware COR line is not being used, the radio may produce a burst of audio when going from the transmit state to a squelched receive state. This "false COR" can cause problems in some applications, so the ARA-1 includes a provision to ignore the COR signal for a specified period of time. In many cases this provision is not needed.

The ARA-1 front panel *CHANNEL ACTIVE* LED is lit whenever the ARA-1 has detected active COR or its VOX function has been tripped. If this LED flashes whenever the radio drops out of transmit mode, raise the COR Inhibit time until this no longer occurs. See also *Ping Pong*, explained in Section 5.7.

3.3.1.3 Radio COR VOX Sensitivity

The *Radio COR VOX Sensitivity* setting adjusts the sensitivity of the audio-activated COR system, also called VOX (Voice Operated Switch). The sensitivity should be set to the lowest value that always causes the VOX to trip during speech signals from the radio. Setting to a higher sensitivity will increase the likelihood that the unit will "false," that is, unsquelch inappropriately due to noise or other invalid sounds. Make sure the radio RX audio level is set properly before you adjust the VOX Sensitivity.

3.3.1.4 Radio COR VOX Hang Time

When using VOX as the Radio COR Type, the system depends on the presence of audio to consider a signal present. Since speech is not continuous (there are pauses in it), the VOX system must “hang,” or wait for a certain period of time, before making the determination that the signal is no longer present, otherwise it will resquelch momentarily between syllables or during short pauses in speech. Set the *Radio COR VOX Hang Time* to the lowest level that does not create inappropriate resquelching. The ARA-1 front panel *CHANNEL ACTIVE* LED is lit whenever the ARA-1 has detected active COR from the radio or its Radio VOX function has been tripped. If this LED flashes during pauses in speech from the radio, the hang time must be increased.

3.3.2 Radio PTT Timeout

The *Radio PTT Timeout* option sets the maximum amount of time (in seconds) that the ARA-1 will continuously assert PTT. Its purpose is to protect the radio’s transmitter from damage as well as to prevent radio users from being locked out by a “hung” PTT. When the PTT timeout triggers, the original source of the PTT (Network COR) must clear (de-activate) before that source will be allowed to again activate the ARA-1’s PTT output to the radio.

For example, if a user is connected to an ARA-1 via a SIP Phone, and the ARA-1 is configured to use VMR as the Network COR Type, an overly sensitive SIP Phone microphone and/or loud background voices can cause the ARA-1 to key indefinitely, thereby preventing any return communications from radio users. The PTT Timeout will trigger after the set amount of time has passed, unkeying the associated radio regardless of network audio content. Furthermore, the ARA-1 will not allow network audio input to key the radio again until there is a break in the network COR signal generated by the VMR function.

3.3.3 COR Priority

Since radios are half-duplex devices (you can either talk or listen, but not both at the same time), the possibility exists that the radio may be receiving a signal at the same time a signal is being received from the SIP network. The *COR Priority* setting allows the user to select which one has priority. When set to *Radio Priority*, the radio RX audio takes precedence. That is, if the radio is unsquelched (COR active), audio from the network will not put the radio into transmit mode until the radio squelches (COR inactive). This means that people communicating via radios will have precedence over communications coming in via the SIP network.

When set to *Network Priority*, valid audio from the network will key the radio associated with the ARA-1 regardless of any RF signals being received by the radio.

For applications where full-duplex operation is desired, set this option to *Disable*. In this case, neither the radio COR nor network COR will take priority, allowing both to pass through unabated.

3.3.4 Always Pass Audio

By default, the ARA-1 will only pass audio from the radio interface to the network interface or vice versa when the appropriate COR is present. In some applications, it may be desirable to have audio pass through regardless of COR status, such as in full-duplex systems. Use the *Always Pass Audio* option to enable audio pass through.

3.3.5 Network COR Settings Options

Network COR settings define how and when the audio coming from the network is seen to be valid and, therefore, will cause the associated radio to "key up" and transmit this audio. Also affecting this is the *COR Priority* setting, which decides which has precedence—the radio or the network—when valid audio is being received from both simultaneously (see Section 3.3.3).

3.3.5.1 Network COR Type

The *Network COR Type* function is similar to *Radio COR Type* except that there are more options. This setting tells the ARA-1 how to determine when there is a signal coming from the SIP network, which will ultimately activate the attached transmitter. VOX senses the audio level, while VMR (Voice Modulation Recognition) looks specifically for human speech and ignores non-speech signals. VMR can help prevent false transmitter activation from background noise on the SIP connection (such as someone breathing heavily into a SIP Phone handset mouthpiece, or someone using a SIP Phone in a high ambient noise environment).

The *RTP Header*, *RTP Header + VOX*, and *RTP Header + VMR* options may be useful if multiple ARA-1 units are integrated in the SIP network. These COR types make use of the ARA-1 RTP extension header that sends the unit's COR status to other ARA-1s that they are conferenced with. This is helpful as these other ARA-1s will receive a positive indication of COR status that arrives coincidentally with the radio audio, and the ARA-1 will not have to derive the COR status using a VOX or VMR function. This is a quicker and surer way to determine when the linked ARA-1s should key their associated transmitters. For these settings to have any utility on the unit being set up, another ARA-1 that may link to this unit must have the SIP Settings *Send Radio COR/AUX Status* option enabled. See Section 3.2.3 for instructions about enabling this function.

To further clarify, when Radio A is linked to Radio B over a SIP network via a pair of ARA-1s, whenever Radio A is receiving a valid signal (and, therefore, has active COR), Radio B should have its PTT activated so that it can retransmit the audio received from Radio A. If the ARA-1 associated with Radio A has the *Send Radio COR/AUX Status* function enabled, it will send its COR status over the network as part of the RTP extension header. The ARA-1 associated with Radio B can make use of this information only if its Network COR Type is set to one of the following:

- *RTP Header*
- *RTP Header + VOX*
- *RTP Header + VMR*

Use the *RTP Header* setting if all end-devices on the system are connected via ARA-1s, for example, if there will only be two radios connected over the Internet (see the center diagram of Figure 1-3). When the Network COR Type is set to *RTP Header*, the only method used to validate network audio (and hence key the associated radio) is the COR information transferred in the RTP extension header. This means that the ARA-1 will ignore network audio from other devices not interfaced by an ARA-1 with its *Send Radio COR/AUX Status* function enabled (for example, a SIP Phone). If these other devices will be used, set the ARA-1 Network COR Type to either *RTP Header + VOX* or *RTP Header + VMR*.

Both *RTP Header + VOX* and *RTP Header + VMR* make use of the COR status information from linked ARA-1s, but also properly link with non-ARA-1 devices. When either of these modes is selected, the ARA-1 will use both functions (the COR status *or* VOX; the COR status *or* VMR) to validate network audio. Select between these two options using the same reasoning as you would selecting between VOX and VMR.

NOTE: The RTP extension header used to transfer COR/AUX status is not part of the full common standard and, therefore, there is a possibility that another SIP device may be using this header for some other purpose. Two conditions may result:

(1) An incompatible SIP device will misinterpret the COR/AUX extension header sent from the ARA-1. Most likely, the device will interpret the status information as audio. Clicking noises may result in the device's audio output.

(2) An incompatible SIP device will send a non-standard RTP extension header that is misinterpreted by the ARA-1. This may cause the ARA-1 to signal the associated radio to key inappropriately.

Use the following guidelines when deciding which Network COR types to use:

- If all end devices are radios interfaced by ARA-1s, use RTP Header as the Network COR type in all ARA-1s and enable the *Send Radio COR/AUX Status* SIP setting in all ARA-1s.

- If the network consists of mixed devices, for example, multiple ARA-1 units as well other SIP devices such as SIP Phones or softphones, and these devices have no incompatibilities with SIP extension headers use either *RTP Header + VOX* or *RTP Header + VMR* as the Network COR type in all ARA-1s and enable the *Send Radio COR/AUX Status* SIP setting in all ARA-1s.

- If the network consists of mixed devices, for example, multiple ARA-1 units as well other SIP devices such as SIP Phones or softphones, and these devices have SIP extension header incompatibilities, do not use any of the Network COR Types that include RTP. Instead use one of the other settings as the Network COR type in all ARA-1s and disable the *Send Radio COR/AUX Status* SIP setting in all ARA-1s.

The next setting—*SIP Actions*—should be selected when controlling the ARA-1's PTT output via DTMF over the network. See Section 3.2.4 for instructions on how to configure SIP Actions.

The next network COR type option is *Packet*. When *Packet* is selected, the ARA-1 will key the radio based upon the presence or absence of audio RTP packets. When the ARA-1 is receiving audio RTP packets, it will activate PTT and key the radio. When it stops receiving audio RTP packets, it will deactivate PTT thus unkeying the radio. This feature is often used in conjuction with silence suppression on the remote device.

When *Packet* COR is enabled, the *Network COR VOX/VMR/Packet Hang Time* setting described in Section 3.3.5.3 can be used to adjust how long PTT remains active after receiving the last audio RTP packet. This is used to prevent the same “drop outs” that can occur when using VOX or VMR detection.

When selected, the final setting option—*Disable*—never allows any audio received from the network to key the attached radio. This can be useful in situations where one intends only to monitor a radio’s receive signal.

3.3.5.2 Network COR VOX/VMR Sensitivity

The *Network COR VOX/VMR Sensitivity* setting performs the same function as the *Radio COR Sensitivity*, but applies to audio coming from the SIP network. It specifies the level of audio that is required to be considered “valid” and, therefore, to cause the radio to transmit the audio. If you are using VOX or VMR and the transmitter is not activating reliably you may need to increase the sensitivity. If the transmitter is being activated by background noise, you may need to decrease the sensitivity.

3.3.5.3 Network COR VOX/VMR Hang Time

The *Network COR VOX/VMR Hang Time* setting performs the same function as the *Radio COR Hang Time*, but applies to the audio coming from the SIP network. If the transmitter deactivates or “drops out” between words or syllables, you may need to increase the hang time. For the most natural conversation, do not set the hang time to any amount longer than necessary to remove the drop outs.

3.3.6 Audio Adjustments

3.3.6.1 Transmit Audio Delay

In some specific applications (such as interfacing to a trunked radio system) it may be necessary to delay the audio going to the radio. The amount of delay is set in the *Transmit Audio Delay* field. See Section 5.6 of the *System Troubleshooting* section for more information.

3.3.6.2 Receive Audio Boost

The audio input level is adjusted via a control on the rear panel. The instructions for setting this adjustment can be found in Section 2.8.3. If additional received audio gain is necessary, it may be selected here, with the additional gain added in the digital domain.

When the radio settings have been made, click *Save* to save them. Then click *Dialing* to go to the *Dialing* page (shown in Figure 3-6).

3.4 Outgoing Call Configuration

A radio user may initiate and terminate calls over the SIP network by one of two methods:

- If the user’s radio has a keypad, a pre-selected DTMF sequence may be transmitted. This is the easiest and surest way (and, therefore, the preferred method)

- If DTMF is not possible, then a specified COR Cadence (Squelch Break) sequence can be used. The *Dialing* page is where this capability is configured.

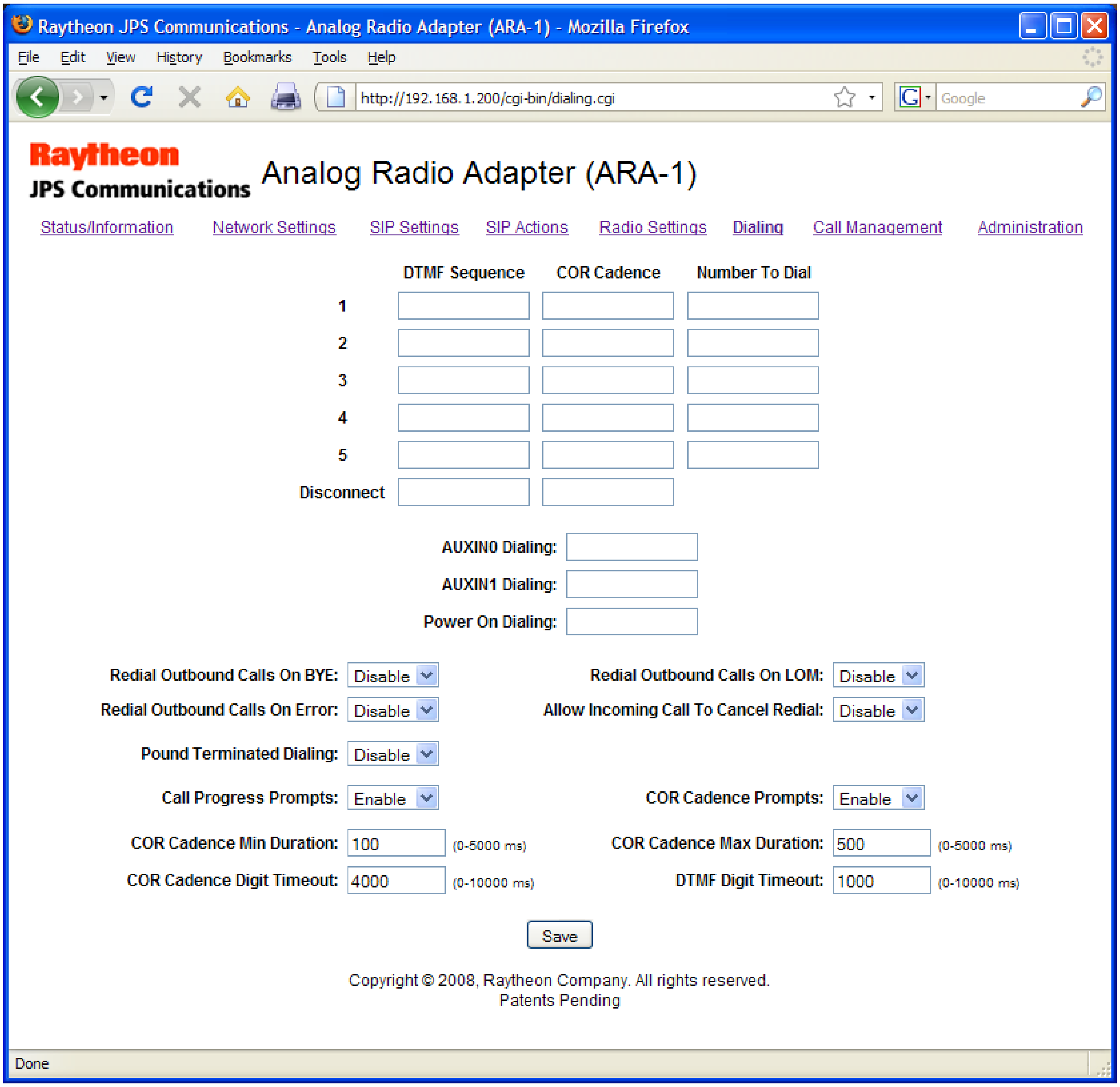

Figure 3-6 Dialing Page— Outgoing Call Configuration

3.4.1 Configuring Outgoing Call Initiation via DTMF

Enter a valid DTMF sequence that will be transmitted by the radio (such as *100, 999, or other valid DTMF, up to 10 characters) into one of the *DTMF Sequence* fields. Then enter the corresponding *Number To Dial*. This must be a valid SIP address, so what is entered here will depend on your system. It could be a SIP PBX extension number, a SIP PBX phone number, or an IP address (perhaps of another ARA-1). A full SIP URI may also be entered. When the radio cabled to the ARA-1 receives any of these preset DTMF sequences, the ARA-1 will attempt to initiate a connection to the end-user device associated with this *Number To Dial* setting.

NOTE: Make sure you also program an entry into the *Disconnect* field so you can terminate the call when you are finished. Something like ***, ###, or another unique sequence is appropriate and will not be confused with a number to dial.

3.4.1.1 Pound Terminated Dialing

The *Pound Terminated Dialing* function allows radio users with DTMF keypads to dial a number or extension as if they were using a telephone simply by pressing the pound (**#**) key after the dialing sequence. Whenever this is done, the ARA-1 simply initiates a call to the end user identified by the DTMF sequence entered (minus the pound). If there is no pound digit appended to the DTMF detected, the ARA-1 will instead follow the operation outlined in Section 3.4.1, comparing the detected DTMF to the pre-configured *DTMF Sequences* and then calling the associated *Number to Dial*.

3.4.1.2 Call Progress Prompts

The ARA-1 provides voice prompts (automated voice messages) that allow the radio user to keep track of the progress of the call (*connecting, disconnecting*, etc.). The factory default setting is *Enable*. If you do not want these prompts transmitted over the radio, set to *Disable*. This function operates with both the DTMF and the COR Cadence call initiation modes.

3.4.1.3 DTMF Digit Timeout

The *DTMF Digit Timeout* function specifies the maximum time allowed between DTMF digits during a call initiation sequence. As soon as a pause is measured longer than this setting, the ARA-1 will consider the DTMF sequence finished. The factory default setting of one second (1000 ms) should work for most systems.

3.4.2 Configuring Outgoing Call Initiation Via COR Cadence

Many radios do not have DTMF keypads, so the ARA-1 offers an alternative method to initiate a SIP call. *COR Cadence* is a feature of the ARA-1 whereby radio users can press the PTT button on their radios a specified number of times at a specified rate to initiate a pre-programmed connection.

COR (Carrier Operated Relay) is an indication that a receiver is detecting a carrier signal that is strong enough to open its squelch. Some radios may call this signal the *Squelch Output* (or similar) rather than the *COR Output*.

When a radio user in the field gives the radio's PTT switch five quick presses, it will transmit five quick carrier pulses. These will be picked up by the receiver cabled to the ARA-1, and it will activate its COR output with five corresponding pulses; this *COR Cadence* will be detected by the ARA-1. These pulses are also called *Squelch Breaks*. To minimize the possibility of falsing on random noise, pulses that fall out of a specified duration are ignored by the ARA-1.

> **NOTE:** This method only works if:
>
> (1) The radio cabled to the ARA-1 has a COR Output signal that is connected to the ARA-1's COR Input pin. The COR signal line must be properly configured via the *Radio Settings* page (Figure 3-5).
>
> (2) The radio PTT has full control of the transmit function of the radio. This means that whenever the PTT switch is pressed, the radio is transmitting, and it is not transmitting at any other time. Full control is not available with most trunking systems. When the PTT switch is initially depressed, the trunking controller function has temporary control of when the radio is actually transmitting a carrier.

3.4.2.1 COR Cadence

In the *COR Cadence* field, enter a one or two digit number. For a single-step cadence, enter a single number between *2* and *9*. This number will specify how many times the radio user must depress the PTT input "key clicks" to trigger the call initiation. For a two-step cadence, enter a pair of digits. For example, *35* means that the COR Cadence is made up of three key clicks followed by a pause, and then completed by five more key clicks. The key clicks must match the criteria specified in the *COR Cadence Min Duration*, *COR Cadence Max Duration*, and *COR Cadence Digit Timeout* fields.

In the *Number To Dial* field, enter the SIP Extension or IP address of the end-user device that you want to call whenever this COR Cadence is detected. Be sure to also add the cadence that you will use to terminate the call in the *Disconnect* field. This cadence must be different from any listed *Number To Dial* cadence.

3.4.2.2 Call Progress Prompts

The ARA-1 provides voice prompts (automated voice messages) that allow the radio user to keep track of the progress of the call (*connecting*, *disconnecting*, etc.). The factory default setting is *Enable*; if you do not want these prompts transmitted over the radio, set to *Disable*. This function operates with both the DTMF and the COR Cadence call initiation modes.

3.4.2.3 COR Cadence Prompts

The ARA-1 also provides voice prompting (automated voice messages) during the COR Cadence call initiation process. Whenever the ARA-1 detects one of the COR Cadences specified in the *COR Cadence* fields, a verification prompt is transmitted. For example, if the COR Cadence is six squelch breaks in a row, the voice message prompt will be the word *six*. If the COR Cadence is three squelch breaks followed by five more, the voice messages will occur after each portion of the sequence, that is two prompts, the word *three* and then the word *five*.

Set this option to *Disable* if you do not want these prompts to be transmitted.

3.4.2.4 COR Cadence Min Duration

The *COR Cadence Min Duration* setting is the minimum duration of the detected squelch break before an individual COR pulse is recognized. This usually corresponds with the amount of time the user must hold the PTT button depressed. This also specifies the minimum time that must elapse before the next squelch break pulse begins (how long the PTT switch must remain inactivated between presses).

The factory default setting of 100 milliseconds should work well with most systems. One way to determine if a change is needed is to use *COR Cadence Prompts* to provide quick feedback. The minimum duration can be reduced and/or the maximum duration can be extended if the ARA-1 does not always detect each of the squelch breaks attempted.

NOTE: Do not be concerned if the explanation of COR Cadence setup parameters seems overly complicated. The ability to modify the settings and the information that explains them is provided for the rare circumstances where changes are needed. The factory default settings provide considerable leeway and will work well with most systems and most users. If you feel that the settings are not working properly, it is easy to get feedback through the *COR Cadence Prompts* feature.

3.4.2.5 COR Cadence Max Duration

A detected squelch break pulse and the delay between individual pulses can be no longer than the set max duration. For a two-step cadence, the ARA-1 considers that any delay longer than the *COR Cadence Max Duration*, but less than the *COR Cadence Digit Timeout*, is a pause between the two steps.

The factory default setting of one-half second (500 milliseconds) should work well with most systems. One way to determine if a change is needed is to use *COR Cadence Prompts* to provide quick feedback. The minimum duration can be reduced and/or the maximum duration can be extended if the ARA-1 does not always detect each of the squelch breaks attempted. The more loosely the criteria are set, the more likely that the unit could false on noise-induced squelch breaks.

3.4.2.6 COR Cadence Digit Timeout

Once the ARA-1 detects a squelched condition that exceeds the set *COR Cadence Digit Timeout*, the unit will consider that the COR Cadence sequence is complete. For a two-step COR Cadence sequence, this is the maximum time you can wait between squelch break pulses.

3.4.3 Automated Dialing Methods

In addition to the methods described above that allow a radio user to initiate a call, the ARA-1 also supports methods to automate the initiation of a call.

3.4.3.1 Power On Dialing

When the *Power On Dialing* setting is configured with a valid extension, SIP URI, or IP address, the ARA-1 will automatically dial it at power up. If the ARA-1 is set to register with a SIP proxy, it will not dial until after it has successfully registered.

3.4.3.2 AUX Input Dialing

The ARA-1 can initiate and disconnect a call using the auxiliary inputs that are part of the radio DB15 interface connector. When configured, the ARA-1 will dial the configured extension, SIP URI, or IP address when the auxiliary input goes active and disconnect the call when the input goes inactive.

Use *AUXIN0 Dialing* to configure AUX input dialing for the first auxiliary input, and use *AUXIN1 Dialing* for the second auxiliary input.

3.4.4 Automatic Redialing

The ARA-1 can be configured to automatically redial a call under different circumstances. This feature will most often be used in conjunction with the automated calling methods described above, such as to permanently nail up a link between two sites.

To have the ARA-1 redial a call it initiated when it receives a BYE (the other end hangs up), set the *Redial Outbound Calls On BYE* setting to *Enable*.

To have the ARA-1 redial a call it initiated if an error occurs, such as the remote end is unavailable or a timeout occurred, set the *Redial Outbound Calls On Error* to *Enable*.

To have the ARA-1 redial a call it initiated that ends due to loss of media detection (see Table 3-1), set the *Redial Outbound Calls On LOM* to *Enable*.

The ARA-1 waits a short period of time after the redial-causing event occurs before it attempts to redial. During this time, it is possible that the remote end may try to reestablish the call, or a call may come in from a different endpoint. The *Allow Incoming Call to Cancel Redial* option can be used to prevent the incoming call from interrupting the redial attempt. If set to *Disable*, the ARA-1 will respond to the incoming call with a *Temporarily Unavailable* message. If set to *Enable*, the ARA-1 will cancel the redial and answer the incoming call.

Click *Save* to save the settings and then click *Call Management* to go to the *Call Management* page.

3.5 Call Management

The *Call Management* page (shown in Figure 3-7)allows the user to initiate a call from the ARA-1 via a web browser. It is actually an operations function page rather than a configuration page, but is included here for clarity.

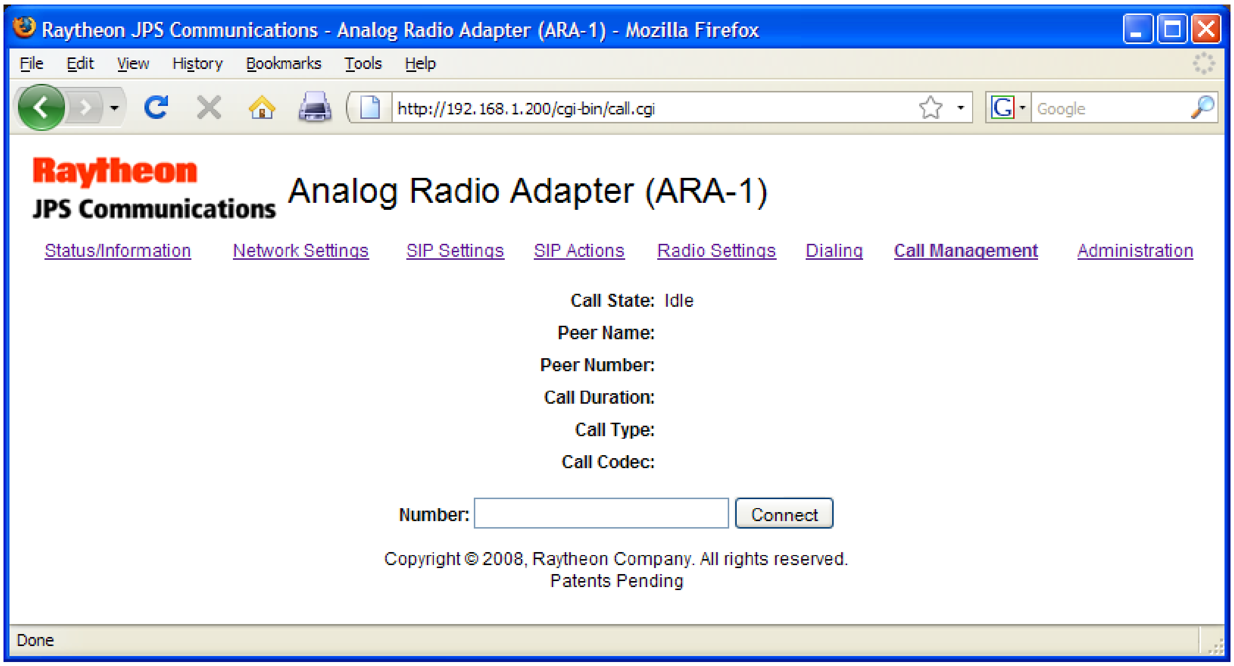

Figure 3-7 Call Management Page

The URI of the end user that the call is being initiated to is entered in the *Number* field. If a connection is already active, this page may be used to break the connection. Call progress information is also provided.

NOTE: This page does not automatically update. You must click the refresh button in your browser to see the results of the call request.

Finally, click on *Administration* to go to the final page.

3.6 Administrative Functions

The *Administration* page (shown in Figure 3-8) allows password protection of access to the ARA-1 web pages, facilitates upgrades to the unit's firmware, and provides a means to remotely reboot the ARA-1.

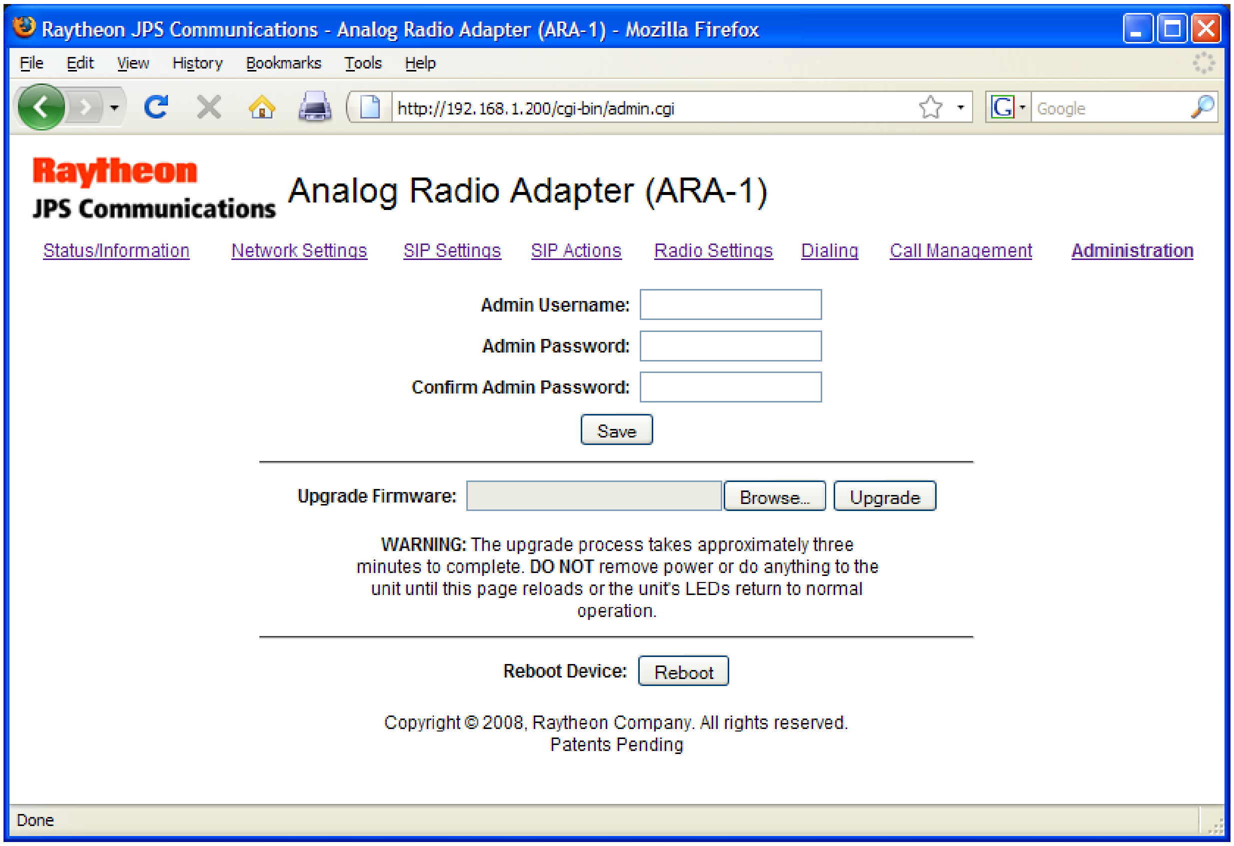

Figure 3-8 Administration Page

3.6.1 Password Protection

Enter a username and password and click *Save* to password protect the device. If you forget the password, the only way to restore access is to reset the unit to factory defaults (see Section 3.7).

3.6.2 Firmware Upgrade

The ARA-1 is designed to support firmware updates in the field. The website below will indicate the latest firmware version available for download, and supply a link to begin the process. Export regulations require that the form (supplied after a download request) be filled out prior to enabling the download.

http://raytheon.com/capabilities/products/ara1/index.html

On the right side of the page all ARA-1 related downloads are listed.

The *Status/Information* page of the ARA-1 will list the currently loaded firmware so that you can determine if an upgrade is needed. After you fill out the export regulations form, instructions will be emailed that will allow the download of the latest firmware. Download the new firmware file to your computer. Next, from the ARA-1's *Status/Information* page, use the *Browse* button to locate the file and click *Upgrade* to load the firmware into the ARA-1.

3.6.3 Remote Reboot of the ARA-1

Click the *Reboot* button if it becomes necessary to remotely reboot the unit.

3.7 Resetting the ARA-1 to Factory Defaults

If you find it necessary to reset the ARA-1 to its factory default state (for example, if you have forgotten the IP address or the password), you can reset all the settings to the factory default settings by pressing and holding *SW1 (Defaults)* on the rear panel for 5 seconds. The switch is recessed to prevent accidental use. Use a pointed object such as a pen or paper clip to press and hold the switch.

NOTE: All settings that you have entered will be lost and reset to factory defaults if you do this.

End of Section 3

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4 Operation

4.1 General

This section contains information and instructions required for proper operation of the ARA-1.

4.2 Front Panel Indicators

All front panel indicator LEDs are explained below, starting at the left side of the unit. Refer to Figure 2-2 for views of controls and connectors.

4.2.1 Power LED

The yellow Power LED is lit whenever DC power is applied to the unit.

4.2.2 Link Active LED

The green Link Active LED is illuminated whenever the ARA-1 has accepted a SIP invite or when a SIP invite from the ARA-1 has been accepted. The LED goes out when the SIP call is terminated.

4.2.3 Channel Active LED

The green Channel Active LED is lit whenever radio-side COR is active (the associated receiver is unsquelched or the audio from the receiver has tripped the ARA-1 VOX function). This LED also indicates that the ARA-1 is sending radio receive audio across the IP link.

4.2.4 Audio Input LED

The yellow Audio Input LED is provided as a visual aid in setting the proper input audio level for optimal operation. See Section 2.8.3 for instructions.

4.3 ARA-1 Operation

Basic operation and control of the ARA-1 is discussed in this section. These instructions assume the ARA-1 has already been correctly configured per Section 3.

4.3.1 Operation at Power-Up

When unit power is applied, all front panel LEDs will light for 10 to 15 seconds and then cycle on and off, one after the other. During this time, the ARA-1 will register with a SIP Proxy server if it has been configured to do so. Otherwise no action occurs and the unit will wait for the initiation of an outgoing call (from the radio end) or an incoming call (from the SIP network).

4.3.2 Basic Operation

The ARA-1, once fully and properly configured, appears as a SIP endpoint and will respond to SIP invites. If it has been programmed to register with a SIP proxy, it will attempt to do so. Once a connection is established, the LINK ACTIVE indicator on the front panel will light. When the unit receives audio via the SIP connection, it will key the transmitter via the PTT line, and the audio will be transmitted over the radio link. When the associated radio is unsquelched and causes the ARA to detect active COR, the CHANNEL ACTIVE indicator on the front panel will light, and the received audio will be sent over the SIP network.

> **NOTE:** What causes the ARA to detect active COR depends on the method used to detect that the radio is receiving a valid signal: VOX or a hardwired COR input from the radio. This is controlled by the *Radio COR Type* setting. See Section 3.3.1.1.

4.3.3 Outgoing Call Initiation

There are three methods to initiate a call from the radio side rather than the network side:

- Call Initiation via DTMF
- Call Initiation via COR Cadence
- Web browser

DTMF call initiation involves the use of a radio's DTMF keypad to transmit a preset sequence of digits that correspond to a SIP extension or the IP address of a SIP end user. The DTMF sequences and corresponding SIP destinations are pre-programmed per the procedure explained in Section 3.4.1. Another DTMF sequence terminates the call.

COR Cadence call initiation involves the use of preset squelch break sequences; that is, a remote radio user triggers the radio's PTT input a preset number of times at a defined rate. The radio cabled to the ARA-1 detects these transmit bursts and signals the ARA-1, which initiates a call to the preset SIP destination that corresponds with the detected cadence. The COR Cadences and SIP destinations are pre-programmed per the procedure explained in Section 3.4.2. Another COR Cadence terminates the call.

Calls can also be initiated by browsing to the ARA-1's IP address, selecting the *Call Management* page, entering the SIP destination (SIP PBX extension number or the end-device IP address) and pressing *Connect*. See Figure 4-1.

4.3.4 Call Progress Monitoring

The ARA-1 provides a pair of methods for monitoring the progress of ongoing calls:

- Call Progress voice prompts
- Call Management Information available via web browser

The ARA-1 provides voice prompts (automated voice messages) that allow the radio user to keep track of the progress of the call (*connecting, disconnecting,* etc.). The factory default setting is *Enable*. If you do not want these prompts transmitted over the radio, set to *Disable* (See Section 3.4.1.1). This function operates with both the DTMF and the COR Cadence call initiation modes.

Alternatively, the user can browse to the ARA-1's IP address and select the *Call Management* page to view current Call Progress and Call Status information. See Figure 4-1.

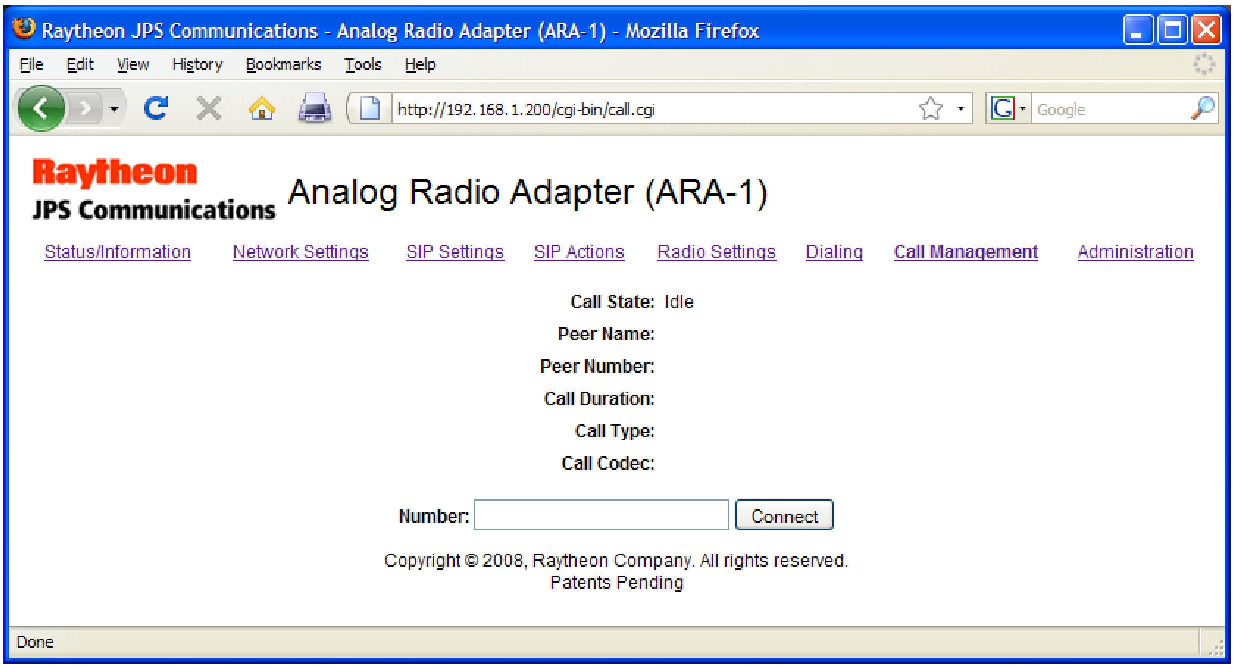

Figure 4-1 Call Management Page

4.3.5 System Information Prompts

The ARA-1 can "speak" the current network settings out over the radio port as an aid to setting up the units. This is particularly useful if you have set the unit to use DHCP and you do not know the assigned IP address.

The unit responds to specific DTMF or COR Cadence sequences and speaks the IP address, subnet mask, and gateway address as specified below:

Table 4-1 System Information Prompts

DTMF Sequence	COR Cadence	Spoken Value
* * 0 #	10	IP Address
* * 1 #	11	Subnet Mask
* * 2 #	12	Gateway Address

As an example, if a radio user transmits the DTMF characters ** * 0 #* to the radio associated with the ARA-1, the ARA-1 will key the associated radio and announce the IP address. Alternatively, the remote radio user may click the mic 10 times to get the same information (as long as the radio's COR output is connected to the ARA-1 and properly configured, see Section 3).

End of Section 4

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5 System Troubleshooting

5.1 System Troubleshooting

This section provides some hints to optimization of ARA-1 setup based on system operation symptoms.

These symptoms include:

- Missed first syllables
- Missed syllables in mid-conversation
- Ping pong (cross-connected radios key and unkey repeatedly after the end of an intended transmission)
- False keying (inappropriate keying due to noise)

5.2 Missed First Syllables: Radio Side

If the users on the radio-system end of an ARA-1 conversation regularly miss the initial syllables of messages from the network, it is likely that the TX audio delay setting of the ARA-1 must be increased. There are two probable causes:

- A trunked radio system
- Network VOX or VMR function is not optimized in a system that is not trunked

A trunked radio system is the most common and most obvious example of this condition because of the time it takes a trunked radio to acquire an open channel. See Section 5.6 for a full explanation.

If the radio system is not trunked, the missed syllables could be caused by a Network VOX or VMR function that is not optimized. In this case, it may still be that the TX audio delay should be increased in order to give the function enough time to detect the existence of valid audio. Alternatively, it could be that the Network VOX or VMR threshold needs to be made more sensitive. Adjustments may be required to both settings. If the sensitivity is set too high, it is likely that the radio listener will also hear dropouts in the middle of a message, rather than only at the beginning. See Section 5.3. Attempt to rectify first by optimizing the threshold setting.

NOTE: The *Radio TX Audio Delay* must never be set below 200 milliseconds when VOX or VMR are used to gate the audio coming in from the network side. Clipping of the first syllables of any spoken message is very likely to occur with lower delay settings.

5.3 Missed Syllables Mid-Conversation: Radio Side

The most likely cause of radio users noticing missed syllables in mid-conversation only is VOX or VMR dropout. If the Network VOX/VMR hang time is incorrectly adjusted, the VOX or VMR function may momentarily unkey the transmitter and then quickly rekey. Audio is lost during the unkey/rekey process. The solution is to increase the hang time. See Section 3.3.5.3.

It is also possible that the VOX/VMR threshold is set too high and/or the TX audio delay is too short, though this would likely also cause missed syllables at the start of a transmission. See Section 5.2.

5.4 Missed First Syllables: Network Side

If the users on the network end of an ARA-1 conversation miss the initial syllables of radio messages sent over the network, the *Radio COR Type* is probably set to *VOX*, and the *VOX* function is not properly optimized. In this case, it is likely that the VOX threshold should be made more sensitive. See Section 3.3.5 for configuration explanation and instructions.

If the threshold is too high, it is likely that the radio listener will also hear dropouts in the middle of a conversation, rather than only at the beginning. See Section 5.5. There is no audio delay setting that helps relieve this symptom.

5.5 Missed Syllables Mid-Conversation: Network Side

Radio VOX dropout is the most likely reason that users who are linked over the SIP network will notice missed syllables (mid-conversation only) of a radio transmission. If the Radio COR VOX hang time is incorrectly adjusted, the VOX may momentarily unkey the transmitter and then quickly rekey. Audio is lost during the unkey/rekey process. The solution is to increase the hang time. See Section 3.3.1.4.

It is also possible that the Radio COR VOX threshold is set too high, though this would likely also cause missed syllables at the start of a transmission. See Section 5.4.

5.6 Explanation: Trunked Channel Acquisition Delay

Trunked systems including 800 MHz Trunked Radio Systems are a very common public safety communications format. When trunked system users begin a transmission, their radios must first communicate with the Trunking Controller. The Trunking Controller has ultimate control of each radio's TX function. When a trunked system radio's PTT input is activated, the Trunking Controller first ensures that the user's radio is on an open channel, and then provides a tone to the user. This tone signals that it is now OK to begin speaking. This is an incomplete overview of Trunked Radio operation, but the concept essential to interoperability is the time gap between when a user activates a radio's PTT switch and when that user may begin speaking.

This gap poses a problem to any Interoperability System. When the trunked radio system is cross-connected to another radio, the operator of the other radio does not hear the "Channel Ready" acknowledgement tone (also called the "go ahead" tone), and may not even be aware that he is cross-connected to a trunked system. ***If this radio operator simply begins talking, the first syllables or words will be lost while the trunked radio is silent and waiting to acquire a free channel.*** This is simply not acceptable in the circumstance when interoperability is most frequently needed: during a disaster or other unusual event when clear communication is crucial.

The solution is to add delay to the audio that is being patched from other radios into the trunked system by increasing the *TX Audio Delay* setting of the associated DSP-2 module. This TX audio delay should match or exceed the channel acquisition time. This holds up the RX audio from cross-connected radios until the trunked radio is ready to begin transmitting.

Be sure to take into account the fact that channel acquisition times are increased when the trunked system is exceptionally busy. Since any type of incident that requires interoperability is likely to be very busy for all communications, the interoperability system must have the ability to add sufficient audio delay to compensate. Keep in mind that the ACU-1000 allows quick "on-the-fly" adjustment of the delay time either at the incident scene, or remotely using the ACU Controller or the WAIS Controller.

Also keep in mind that the ARA-1 allows "on-the-fly" adjustment of the delay time through the web browser interface.

Refer to Figure 5-1 and Figure 5-2 for an illustration of the problem and how it can be resolved.

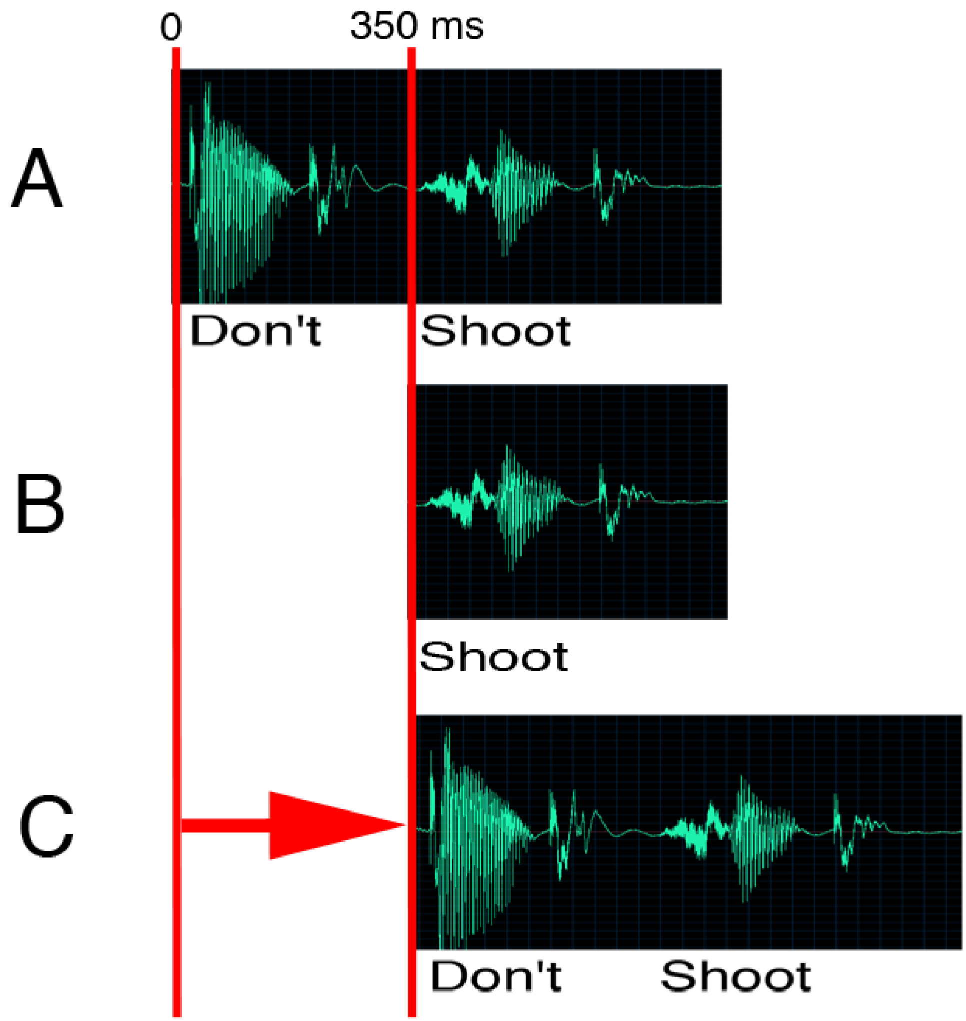

Figure 5-1 "Shoot" Versus "Don't Shoot"

Figure 5-1 illustrates the following three stages in a trunked channel acquisition delay scenario:

- A: The audio is being sent into the SIP PBX by radio #1. Radio #1 is conferenced with radio #2.
- B: Radio #2 is an 800 MHz trunked radio with a channel acquisition delay of 350 milliseconds. Therefore, radio #2 will not start transmitting the audio from radio #1 until 350 ms have past, and the first word of the message is clipped.
- C: If the ARA-1 delays the audio to radio #2 by at least as long as the channel acquisition delay, the entire message gets through.

Figure 5-2 shows the potential communication problems that can occur when the necessary delay is not provided, with messages clipped or lost entirely. The vertical lines signify various channel acquisition delays. Without corresponding TX Audio delays, all speech up until the channel is acquired will be clipped off of the beginning of the transmission (which could be an entire short, but vital, message). If the proper TX audio delay is present, no speech is lost.

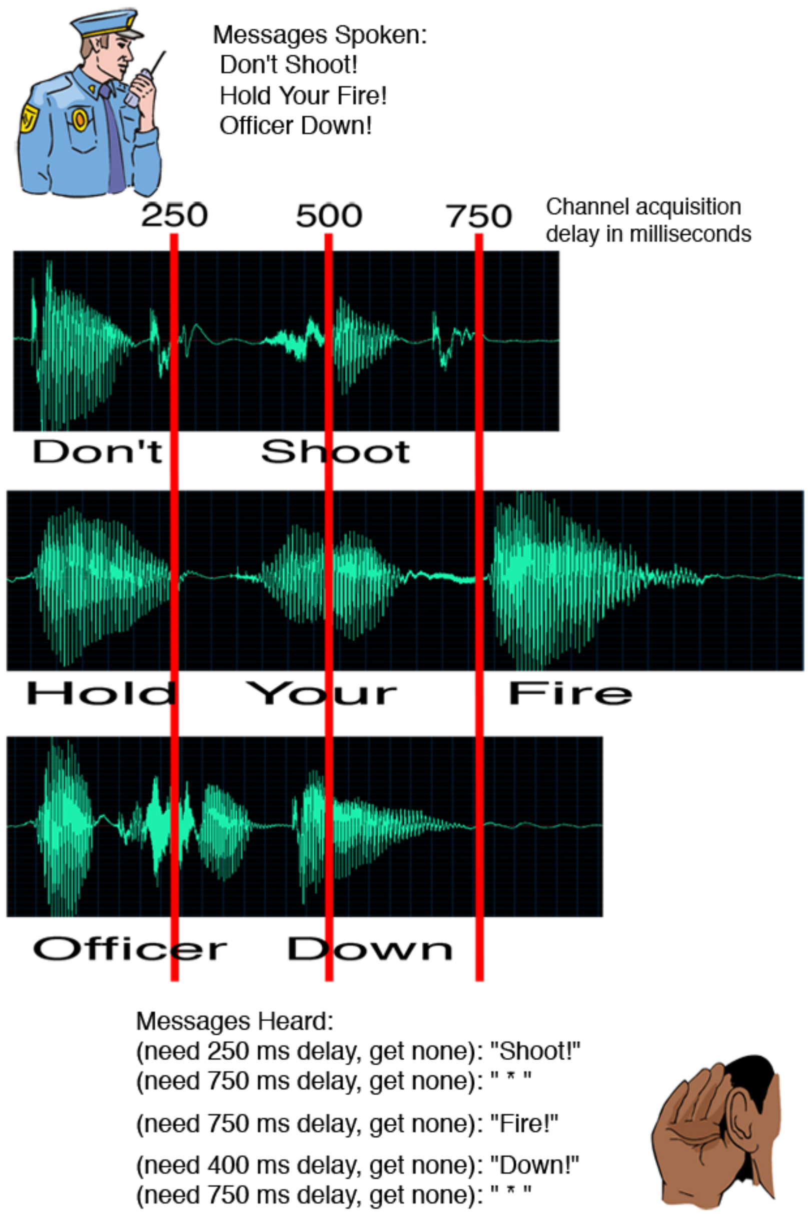

Figure 5-2 Why Audio Delay Is Crucial

5.7 Ping Pong

"Ping Pong" is a descriptive term for a very disruptive behavior that some types of radios exhibit: when cross-connected (conferenced within a SIP PBX), each causes the other to repeatedly key momentarily. The full activity may be difficult to diagnose when the radios are cross-connected a large distance apart over a network (allowing only one radio to be seen at a time), but the one visible radio will repeatedly key momentarily. The ARA-1 has a configuration feature that can prevent this disruptive behavior.

The root cause is the tendency of some radios to inappropriately unsquelch momentarily at the end of each transmission. That is, a radio receives a PTT input, putting it into the transmit mode, and when the PTT input ends, the radio not only drops out of the transmit mode but jumps momentarily to the unsquelched receive mode, even though there is no external RF input to cause the radio to unsquelch.

Remember that for any pair of cross-connected radios, whenever one radio is unsquelched, the other is keyed. If a radio of the pair exhibits the "momentary inappropriate unsquelch after PTT" behavior, any cross-connected radio will momentarily (and inappropriately) transmit. If both radios unsquelch momentarily at the end of each transmission, the system will repeatedly "ping pong," with first one radio keyed momentarily and then the other.

This effect can be experienced when the PTT inputs are activated by either a COR input or by VOX (as some radios will send out a noise burst after they unkey).

To prevent this, first use the ARA-1 front panel LEDs to determine if the particular radio make and model connected to the ARA-1 exhibits the "inappropriate COR after PTT" behavior. The Channel Active LED lights whenever the ARA-1 is detecting active COR. During the system configuration process, simply observe this LED after the radio keys. If it flashes on momentarily at the end of each key sequence, turn on the adjustable *Radio COR Inhibit Time* (see Section 3.3.1) and increase the time value until the condition no longer occurs. This function instructs the module to ignore any unsquelch detection (COR) that occurs immediately following the cessation of a transmit sequence. The duration of the timer is adjustable to optimize for different radios, which may exhibit the inappropriate unsquelch indication for times as short as 100 milliseconds, and as long as several seconds.

5.8 False Keying

The ARA-1 has a feature to prevent the radio cabled to the ARA-1 from keying up frequently (that is, going into TX mode) due to background noise coming in on the network audio (for example, from connected SIP Phones in high-noise locations or multiple SIP Phone users all conferenced together, with one or two users breathing heavily while they are supposed to be just listening). The ARA-1's *Network COR Type* can be switched to *VMR* mode. VMR stands for Voice Modulation Recognition; with this mode enabled, the radio will key only if there is actual speech detected in the audio from the network.

End of Section 5

6 ARA-1 FAQ (Frequently Asked Questions)

6.1 General

This section provides answers to some frequently asked questions about the installation and operation of the ARA-1.

How much network bandwidth does the ARA-1 consume?

The ARA-1 bandwidth usage varies depending on the application. When COR is inactive and silence suppression is enabled (or VOX is enabled and no audio is being transmitted across the link), the bandwidth usage is essentially zero. During an active call when COR is activated and the GSM voice compression method is used, audio will be sent across the link in each direction at approximately 35 Kbps. The G.711u vocoder will require approximately 86 Kbps. The unit uses the same bandwidth as a SIP Phone using the same vocoders.

Can I use the ARA-1 on a dialup connection?

No. The ARA-1’s network connection is via Ethernet. Most dialup (telephone line) modems only provide RS-232 connections, but even if an Ethernet connection to a dialup is available, the ARA-1 may not operate properly since the connection speed is not guaranteed on a dialup connection. Poor telephone lines or varying line conditions may cause a dialup modem to reduce its connection speed without the user being aware of this condition. For this and other reasons, dialup connections are not supported by the ARA-1.

My application does not provide a COR line. What should I do?

COR is usually obtained from a radio and indicates that a signal is being received. If you do not have a COR line, you can use *VOX* mode. *VOX* mode essentially derives a COR signal from the presence of incoming audio from the radio.

I have a 100 Mbps Ethernet. Can I connect the 10 Mbps port on the ARA-1 to my network?

Most 100 Mbps equipment (hubs, switches, routers) will work with either a 10 Mbps or 100 Mbps connection, so it should work without any problems. Check with your network administrator if you are unsure about your network equipment capabilities.

Where can I find a description of the VoIP protocols used by the ARA-1?

The ARA-1 uses the following open protocols: SIP, SDP, RTP, and STUN. These protocols are available from **http://www.ietf.org**.

Can I hook one ARA-1 directly to another via their Ethernet ports?

Yes, if you use a crossover Ethernet patch cable. A straight-through patch cable (such as the one supplied with the unit) can only be used to attach the ARA-1 to network interface equipment such as hubs, switches, or routers. You can directly connect a pair of ARA-1 units up to 100 meters apart using a CAT5 crossover cable.

What is the difference between a static IP address and a dynamic IP address?

On an IP network such as the Internet, the IP address is like a phone number. It is a unique number that identifies the network device, and it allows connections to be made between network devices. There are two kinds of IP addresses: those that are permanently assigned and those that are assigned temporarily. Permanently assigned IP addresses are called static IP addresses, while temporary IP addresses are called dynamic IP addresses. Static and dynamic IP addresses work the same way, but a dynamic IP address is like having a telephone number that nobody else knows; it effectively limits you to making outgoing calls only.

Why do we need static and dynamic IP addresses? Why not just assign addresses permanently like phone numbers?

One problem with IP addresses is there are not enough to go around. There are "only" about 4 billion IP addresses available. On the surface that would appear to be enough, but some companies use a lot of them, the military uses a lot of them, and all the millions of people using the Internet use a lot of them. To ease this problem, many Internet providers only assign an IP address to a computer when someone actually dials in to connect to the Internet. When they disconnect, the IP address goes back into the pool of addresses so someone else can use it. This makes sense in cases where computers might sit for a long time without needing to access the Internet, as there is no point in tying up an IP address when it is not being used. There are plans for an upgraded IP addressing system that will fix the problem of too few IP addresses, but for now it is a limitation we must live with.

What kind of IP address does the ARA-1 need?

The ARA-1 will function properly with either a static or dynamic IP address. Which is better depends upon your application. If the ARA-1 is being used with a SIP proxy, then a dynamic IP address is fine. If the ARA-1 is being used in a standalone application where other devices are issuing SIP invites to it, then a static IP address may be needed.

Who assigns IP addresses?

For Internet access, your ISP (Internet Service Provider) will assign IP addresses to you. If you are on a private network (like a WAN or LAN), then your network administrator will assign IP addresses. You should never program an IP address into an ARA-1 without checking with the ISP or network administrator first.

What happens if two ARA-1s have the same IP address (or an ARA-1 has the same address as some other device)?

The ARA-1s will not work properly. Both units will respond at the same time, causing network errors. The extreme case would be if an ARA-1 is programmed to have the same IP address as an existing computer on the same network. This would likely render that computer unusable for network functions, and could cause other users to have problems as well. The bottom line is that two devices on the same network cannot share the same IP address.

What is a firewall?

A firewall is a security device that prevents people outside of a network from accessing computers or devices inside the network. A company firewall prevents incoming network connections so no one outside the company can access the company's computers via the Internet.

How does a firewall affect the ARA-1?

A firewall may prevent a connection to an ARA-1 if it is behind the firewall and the SIP proxy or other connecting device is not. This problem can often be overcome through the use of STUN, which is described in the configuration section of this manual.

Should I use a firewall with the ARA-1?

Strictly speaking from the standpoint of the ARA-1, there is no reason to use a firewall. The ARA-1 is a standalone network device that poses minimal security risks. However, if the network the ARA-1 is on is shared with computers, servers, or other network devices, then a firewall may be advisable. Talk to your network administrator about such situations.

What is NAT?

NAT (Network Address Translation) is a scheme by which many network devices can share one IP address. The NAT router translates packets passed through it between the single public IP address it holds and the private IP addresses used by devices on its network. This means that no computers behind the NAT router are directly accessible from outside the network since none of them have public (or routable) IP addresses.

How does NAT affect the ARA-1?

Using NAT does not adversely affect the operation of the ARA-1. The use of the STUN protocol may be necessary when the unit is located behind a NAT device. Special considerations should be made in the event that multiple ARA-1 units will be located behind a NAT router. In that case, each unit will need to be assigned a different local VoIP port, and the device will need to be configured to use the STUN protocol.

Should I use NAT with the ARA-1?

The only time it is advisable to use NAT with the ARA-1 is when you wish to have multiple units on a network, but have only one public IP address. An example of this would be a bank of ARA-1s sharing a broadband type connection, such as DSL or cable modem, where only one IP address is allocated by your ISP. Again, it may be necessary to use different local SIP ports for each unit and to enable the use of STUN on each unit for proper operation.

Can I change the vocoder on the ARA-1?

Yes. This is the voice compression method used to convert the audio signal into a digital format. There are times when the user may want or need higher voice quality or transparency to certain kinds of signaling tones. The preferred vocoder may be selected from the *Web Configuration* page.

Can I pass modem tones over the voice port of the ARA-1?

While the ARA-1 is designed primarily for voice operation, it can pass many types of signaling tones if the 64 kbps vocoder is selected. However, due to timing and/or bandwidth considerations, proper operation of modem tones can not be assured. If you have an application that requires use of modem tones, consult JPS.

I've connected my radio to the ARA-1, I have the levels turned all the way up, and can still barely hear the audio. What's wrong?

The ARA-1 uses balanced audio on the input. If, instead of connecting to both balanced input lines, you connect your audio to one audio input and ground, you will get the effect described above. There is enough leakage in the transformer to get some audio through, but it will not work properly. The solution is to ground one side of the ARA-1 balanced input.

I've connected my radio to the ARA-1 and it works, but the audio sounds very "tinny" and doesn't have much bass.

Make sure you have not used the discriminator output, which is sometimes available on FM receivers and transceivers. Discriminator audio is pre-emphasized, and there is no de-emphasis circuitry in the ARA-1. You will need to use your regular audio output instead.

I've configured the settings on the ARA-1 incorrectly and would like to reset the unit to the factory default settings.

Locate an access hole on the rear panel of the ARA-1 marked *DEFAULTS*. Inside is SW1, the defaults switch. With the ARA-1 powered on, insert a small object into the hole and press and hold the switch inside for 5 seconds. Upon releasing the switch, the ARA-1 will restart and the unit will be set to factory defaults.

End of Section 6

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7 Index