INTEROPERABILITY NOW

Installation and Operation Manual

ACU-2000
Intelligent
Interconnect
System

Designed and Manufactured by:

JPS Interoperability Solutions

5800 Departure Drive

Raleigh, NC 27616

919-790-1011

Email: sales@jpsinterop.com / support@jpsinterop.com

www.jpsinterop.com

JPS P/N 5961-230200 **Revision 2.2** **June 2016**

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 JPS Interoperability Solutions could void the user's authority to operate this equipment.

NOTICE

JPS Interoperability Solutions reserves the right to make changes to the equipment and specifications without prior notice.

PROPRIETARY STATEMENT

JPS Interoperability Solutions
Phone: (919) 790-1011
Fax: (919) 865-1400
E-mail: sales@jpsinterop.com / support@jpsinterop.com
5800 Departure Drive
Raleigh, NC 27616

Table of Contents

List of Figures

List of Tables

Glossary

ACU Controller	Intuitive, icon based control program for operating an ACU-2000 interoperability system, either locally or over an IP connection. Provides full control & configuration capabilities.
ARA-1	JPS device that provides a SIP interface to an individual radio or other four-wire device
CSAP	Communications System Access Point – Any entry into the Interoperability System (UHF radio, telephone line, 8000 MHz trunked talk group, Nextel radio, audio console, etc.)
COR	Carrier Operated Relay - A receiver signal that gives a positive indication a carrier or signal is being received and the receiver is unsquelched. Same as COS.
COS	Carrier Operated Squelch - See COR.
CPM-6	Control Processor Module - This ACU module controls all aspects of system operation.
Cross-Connection	A link made between two communications systems interfaced to a single ACU-2000 chassis, or between systems interfaced over a network to two or more ACU-2000 systems.
CTCSS	Continuous Tone Controlled Squelch System. A squelch system using EIA Standardized sub-audible tones in the 67Hz to 250Hz frequency range. An FM squelch that opens only when the proper sub-audible tone is present.
DIP Switch	Dual In-Line Package Switch ("dipswitch") - A multi-unit switch that fits into a standard DIP integrated circuit footprint. It usually contains eight or ten individual switches.
DTMF	Dual Tone Multi Frequency - The standard touch-tone telephone dialing method sends DTMF characters over the PSTN line.
DSP	Digital Signal Processing (or Processor).
DSP-2IP	The Digital Signal Processor Module - The main radio interface of the ACU-2000 system. DSP algorithms provided include VOX, VMR, Audio Delay, Noise Reduction, and Tone Keying. The DSP-2 can instead provide an Ethernet interface to an ACU system.
EIA	Electronic Industries Association.
Extension	Each ACU-2000 interface module is given an extension number, based on which chassis slot it's plugged into. The extension number is used to define and track cross-connections between communications systems interfaced to the ACU-2000.
Hangtime	A system with hangtime will remain in the transmit mode for the duration of the set hangtime beyond the time indicated by any keying inputs. The hangtime prevents transmitter unkey during brief pauses in the transmission.
HSP-2IP	The ACU-2000 Handset/Speaker/Prompt Module - provides a local operator interface via a keypad, handset & speaker. System voice prompt circuitry also resides on this module.
Key	To key a transmitter means to cause it to transmit.
LED	Light Emitting Diode.
LMR	Land Mobile Radio.
LP-2IP	The Local Phone Module - interfaces a standard telephone set to the ACU-2000 system.
Mute	To quiet or inhibit audio.
Network Talkpath	A **RoIP**/VoIP link creating a cross-connection between JPS interoperability Solutions devices over an IP-based network.
NXU-2A	Network Extension Unit. A JPS device that interfaces a radio or other 4-wire device to other communications devices over an IP network. An NXU-2A uses RoIP.
PBX	Private Branch Exchange. A telephone system that owned and operated by a private company (such as a manufacturing business) rather than by a telephone company such as ATT.
PCB	Printed Circuit Board.
Port	The ACU rear panel connectors P1 through P12 provide *Communications Ports* to interface with other communications equipment.
PSTN-2IP	Public Switched Telephone Network Module - interfaces a telephone system to the ACU-2000.
PTT	Push-to-Talk. An active PTT signal causes a transmitter to key.
RoIP™	Radio over Internet Protocol. JPS Interoperability Solutions proprietary protocol which sends

Glossary

	voice plus radio control signals over an IP network,
RX	Receiver or Receiving.
SCM-1	One version of SIP Channel Module – Provides a SIP interface to an individual radio or other four-wire device. It does not communicate with other modules in the ACU chassis. Compare with **SCM-2**.
SCM-2	One version of SIP Channel Module – Provides a SIP interface to the ACU-2000 that can then be cross-connected to other modules in the ACU chassis, creating cross-connections to other communications devices in the ACU system. Compare with **SCM-1**.
SIP	Session Initiation Protocol – A flexible, standards-based, open protocol that initiates and manages communications between devices over an IP network.
Slot	A physical location in the ACU-2000 chassis where a module can be inserted.
SNR	Signal-to-Noise Ratio.
Squelch	A means of detecting audio and causing some action when it is present, such as keying a transmitter or unmuting an audio path.
TX	Transmit or Transmitter.
VMR	Voice Modulation Recognition. A type of squelch, which is activated only by spoken words and not by tones, noise, or other audio information.
VOX	Voice Operated Xmit (Transmit). A circuit or algorithm, which causes a transmitter to key or some other action when voice is present. This squelch type is activated by any audio signal, and is not restricted to voice only.
WAIS Controller	Intuitive, icon based control program for operating a Wide Area Interoperability System, with multiple JPS ACU and NXU devices interfaced by an IP Network. Provides full control & configuration capabilities. Compare with **ACU Controller**.

End of Section One.

1 Introduction

1.1 Scope

This instruction manual provides the information necessary to install, configure and operate the ACU-2000 Intelligent Interconnect System. The ACU Controller, a PC-based system control program is provided free of charge on a CD included with each ACU-2000 chassis; this program is mentioned in this manual but has its own full instruction manual. This manual does not attempt to serve as a tutorial on the open-source SIP protocol.

1.2 What Is Interoperability?

The ACU-2000 allows existing, disparate communications systems to communicate with each other. For example, within an Interoperability System, a conventional VHF radio system can communicate with an 800 MHz trunked system, an APCO 25 radio user can talk with a PSTN or SATCOM user, etc., or any number of these users can be conferenced together. For simplicity, within this manual these communications assets are called CSAPs (Communications System Access Points).

> ***Interoperability: The ability of the users of disparate communication systems to communicate with each other (for example, a patrolman can use his UHF radio to talk to a firefighter who is using her 800 MHz radio).***
>
> ***CSAP: Communications System Access Point. Any entry point into the Interoperability System. For example, a conventional VHF radio channel, an 800 MHz trunked talkgroup, a satellite phone system, Nextel handset, SIP Phone, or other type of communications system.***

Another definition of interoperability is: *The ability of any public service official to talk to whomever they need to, whenever they need to, when properly authorized*. The purpose of the ACU-2000 and related JPS Interoperability Solutions products is to help make this happen in an easy and efficient manner. Our Interoperability Solution ties together existing communications systems with minimal additional equipment and minimal disruption to ongoing communications.

The ACU-2000 provides interoperability locally or via an IP network

1.3 Local and Wide Area Interoperability

The need for interoperability usually arises during a disaster or other unusual event (as existing systems are set up to handle normal communications without interoperability). The majority of interoperability cross-connections are required between CSAPs located at the incident site.

Accordingly, the JPS Wide Area Interoperability Solution consists of a network of local systems that can communicate with each other as needed.

The local connections mainly take place with an ACU-2000 system, while the wide area connections take place over an IP-based network using the open-standards SIP protocol or using JPS RoIP/VoIP technology.

The area covered by an individual LIS (defined below) typically encompasses a political region such as a city, county or group of counties. This simplifies jurisdictional concerns that could otherwise impede the quick decisions and actions required during a disaster or other emergency situation.

Any individual LIS (as well as the entire wide area system) can be controlled and monitored from any location on the network by a dispatcher using the JPS ACU Controller or WAIS Controller software. The Windows-based ACU WAIS Controller provides full control of multiple Local Interoperability Systems, from the local level to statewide, or across any distance connected by an IP-based network. It allows interoperability connections to be made between any two (or more) CSAPs by a simple point & click procedure.

LIS – Local Interoperability System: An Interoperability System intended to service a single political or geographical region. The basic means of connecting the communications systems that an LIS services are ACU-2000 audio links.

WAIS – Wide Area Interoperability System: An Interoperability System that uses an IP-based network to provide interoperable communications among Local Interoperability Systems and with any number of individual users (such as system dispatchers or isolated radio systems).

Network Talkpath – An RoIP/VoIP communications link between elements of a Wide Area Interoperability System.

RoIP™ – Radio over Internet Protocol: A JPS Interoperability Solutions proprietary protocol that uses a standard IP-based network to transfer a VoIP channel and accompanying radio control signals including PTT, COR, and RS-232 serial control.

1.4 Local Interoperability Via The ACU-2000

The ACU-2000 is a main building block of the JPS overall interoperability capability. It consists of a chassis with a number of plug-in modules. The essential modules reside at the left side of each chassis. The PSM Module is the unit's power supply. The HSP Module provides a local interface to the system (via a keypad, handset and speaker). It also allows local setup and control, and houses the unit's voice prompting circuitry. The CPM Module controls the unit and relays status and control messages between the ACU-2000 and either of the system control programs, the ACU Controller or the WAIS Controller.

The rest of the unit is occupied by a number of *interface modules* (up to 12 in a single chassis). These modules interface the unit to the various CSAPs of the Local Interoperability System, or provide network talkpaths to the WAIS. Different modules may be installed, depending on the type of CSAP being interfaced. They are the DSP, PSTN, SCM, and LP modules.

The DSP Modules interface radio systems, network talkpaths, and some types of satellite phones, and iDEN systems. DSP modules also provide RoIP links. The PSTN Module interfaces telephone lines and some satellite phones or cellular phones. SCM-1 and SCM-2 modules provide links to SIP end-devices over an IP network. The LP (local phone module) allows a standard telephone set to be interfaced to the ACU-2000 so that a nearby user can communicate with the rest of the system, wired to the LP by a standard telephone cable with RJ-11 connectors.

Each interface module takes the audio and control signals of its associated CSAP and converts them to signals that can be understood by the rest of the system.

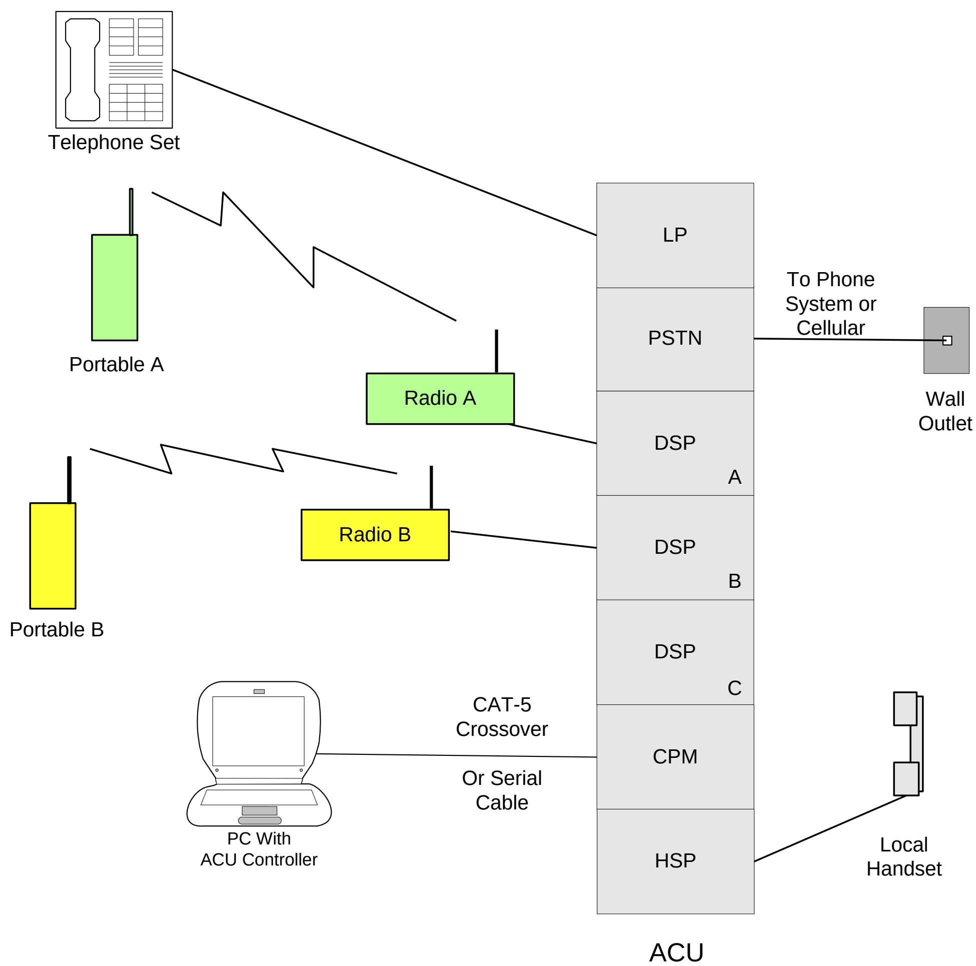


Figure 1-1 Local Interoperability with ACU-2000 Plug-In Modules

Figure 1-1 illustrates the local (non-networked) communications capabilities of the ACU-2000 plug-in modules. (Not shown: the PSM-1 Power Supply Module.)

- HSP Module: The keypad, handset and speaker on this module allow a local operator to communicate with other system users. For example, if the HSP module is cross-connected to DSP module A, the person holding the handset could converse with the person operating the portable labeled *Radio A*.
- CPM Module: The Control Processor Module receives control commands from the unit's operator. The commands may be entered via the HSP's keypad, or (more efficiently) by either the ACU Controller or WAIS Controller programs. The ACU Controller may be connected directly to an individual ACU-2000 or it can monitor and control the unit over an Ethernet network. The WAIS Controller is intended to monitor and control multiple ACU-2000s and other communications assets that make up a Wide Area Interoperability System, all connected to an IP-based network.
- DSP Module: This module interfaces radios and other 4-wire devices to the ACU-2000. The DSP has a front panel RJ-45 connector that allows it to create a network RoIP link (see Section 1.5 for information about the unit's IP capabilities). The DSP modules labeled A and B are shown interfacing radios to the system. These radios can be cross-connected with each other, or with any or all of the communications devices interfaced with the ACU-2000.
- PSTN Module: Interfaces the ACU-2000 to telephone or satellite phone systems.
- LP Module: Allows a standard telephone set to be interfaced to the system using a standard phone cord. The telephone handset becomes an audio interface and the telephone's keypad can be used for system control.

Note the difference between the LP and the PSTN. While the LP module interfaces an individual *telephone set*, the PSTN interfaces an *entire system* such as a landline with service or a cellular system. Connections to each are made with standard phone cords with RJ-11 connectors. The phone cord of the LP is plugged into a telephone set, while the PSTN phone cord plugs into the telephone wall outlet. The PSTN module can also be used to interface cellular phones and some satellite phones.

Radios, audio consoles, and similar equipment are 4-wire devices and interface to the ACU-2000 via the DSP module. 2-wire telephone systems equipment that *provide* loop current and ringer voltage such as a PBX or telephone central office interface via the PSTN module. 2-wire devices that *require* loop current and ringer voltage to operate (such as a telephone deskset), interface via the LP module. The LP module, in effect, simulates the telephone system to the target 2-wire device by providing loop current and ringer voltage.

A 4-wire device is one that has separate lines for transmit and receive audio signals. One pair for TX, and another for RX, totaling four wires.

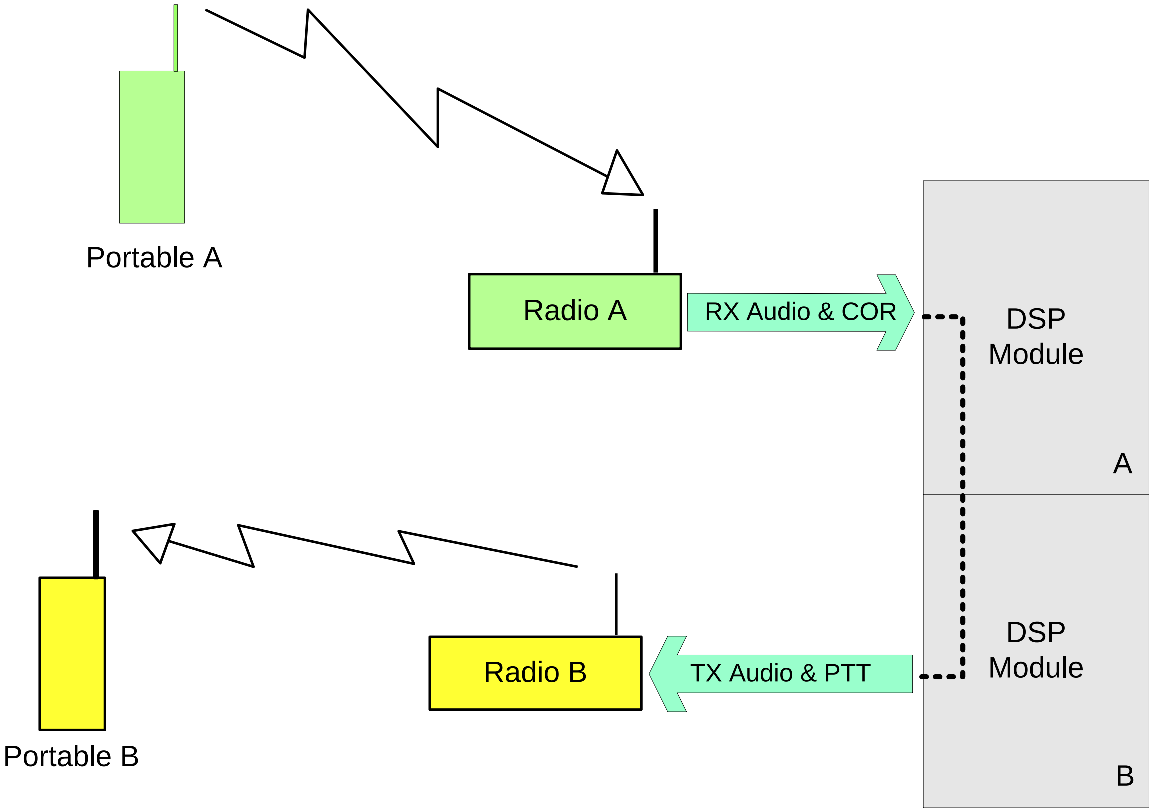
A 2-wire device carries both the transmit signal and the receive signal on the same pair of wires.

1.4.1 Local Cross-Connection Basic Explanation

When an operator uses the ACU Controller or the WAIS Controller to cross-connect two CSAPs at an ACU-2000, a command is issued to the associated ACU-2000 CPM module. The command instructs the CPM to tie the appropriate interface modules together as in the figures that follow. Only after the command is successfully completed (and a status message returned to any active control programs) does the Controller screen show the change in cross-connection configuration.

1.4.1.1 Radio-to-Radio Cross-Connections

Figure 1-2 shows two different types of radios patched together. Normally a Type A radio can communicate with any other Type A radio that’s in the system RF coverage range, but the signals from the Type A radios are completely ignored by the Type B radios, and vice-versa. The ACU-2000’s DSP Modules take the Type A radio’s audio and control signals, translate them for the Type B radio, and send them on to the Type B radio.



RX Audio: What is being spoken into Portable A
COR: This signal is active while the talking is occurring

TX Audio: This same audio from A being sent back out Radio B
PTT: This signal is active when COR is active

Figure 1-2 Pictorial Overview – Cross-Connections Using the ACU-2000

Essentially, the information required to create the cross-connection between the two radio systems via DSP modules A and B can be broken down into the following:

- Radio A, associated with the person talking into Portable A, provides the following information to the ACU-2000:
    - The person's speech. (The RX Audio in the figure.)
    - A control signal that indicates when this person is talking. (The COR signal in the figure.)
- The ACU-2000 provides the following information to Radio B, which is associated with to the person listening on Portable B:
    - The speech signals of the person talking. (The TX Audio in the figure.)
    - A control signal that tells when that person is talking. (The PTT Signal in the figure.)

During a conversation, these roles switch back and forth as each person moves between being the talker and being the listener.

Most radio systems are either simplex or half duplex; the important aspect to remember is that only one person can be heard at a time. With full duplex systems, all parties to a conversation may be heard simultaneously. A telephone system is a good example of a full duplex system. The ACU-2000 can accommodate both types of systems, but ***both*** parties of a conversation must be using full duplex equipment for either party to be able to simultaneously talk and listen.

COR: A signal that tells when a radio (or other communications device) is receiving a valid signal.

PTT: A signal that tells a radio (or other communications device) that a valid signal is being sent to be transmitted.

Full Duplex: System Users can simultaneously talk to and listen to other parties of the cross-connection.

Simplex or Half Duplex: Only one system user can be heard at a time.

A cross-connection can be a bit more complicated for some communications devices other than radios, but the job of the ACU-2000 is the same: to take the incoming audio and control signals from one communications medium, translate these signals into the proper outgoing control signals for another communications medium, and send out these signals along with the original received audio. In order for this to be easy and efficient, the ACU may be called upon to perform a variety of other tasks, such as discern between desired and unwanted audio, decide who gets to be heard when multiple people speak at the same time, or shift the timing of audio as required so that it can be heard.

1.4.1.2 Radio-to-Telephone Cross-Connections

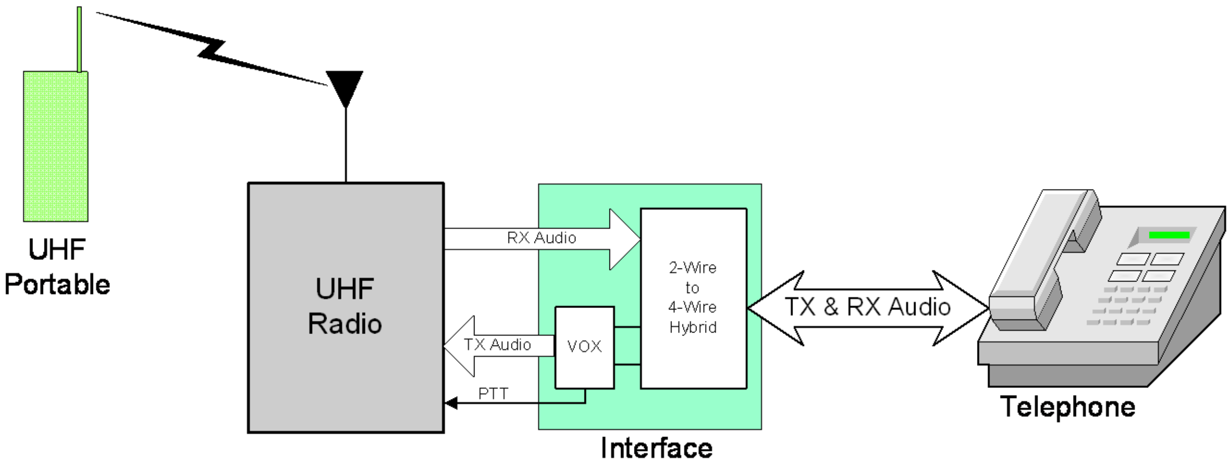


Figure 1-3 Basic Phone Patch

Figure 1-3 shows a basic connection between a 4-wire device (such as a radio) and a 2-wire device (such as a telephone line). 2-wire devices carry audio in both directions, simultaneously, on a single pair of wires. An interface (commonly referred to as ***Phone Patch)*** is required between these two disparate devices. In actuality, the phone patch 2-wire connection does not interface directly to a telephone, but instead to the ***telephone system***. The phone patch would most likely be connected to a phone jack on the wall by a standard telephone cable with RJ-11 connectors. To talk over the radio via the phone patch, you would use your telephone to call the number associated with that phone jack on the wall.

With most standard 2-wire devices (such as a telephone), there are no accompanying control signals such as PTT or COR. Because of its ability to carry both send & receive audio at the same time, these control signals do not benefit the telephone system. Therefore, a 2-wire to 4-wire radio connection requires that a VOX function be provided to derive the COR signal from the incoming phone line audio and supply the associated PTT output signal to the radio. The VOX (Voice Operated Xmit) triggers the PTT when a large enough signal is detected coming from the telephone system to indicate that the end-user is talking.

A VOX output is activated by the detection of audio that exceeds the set VOX threshold.

A Phone Patch interfaces the 2-wire telephone system to a 4-wire radio.

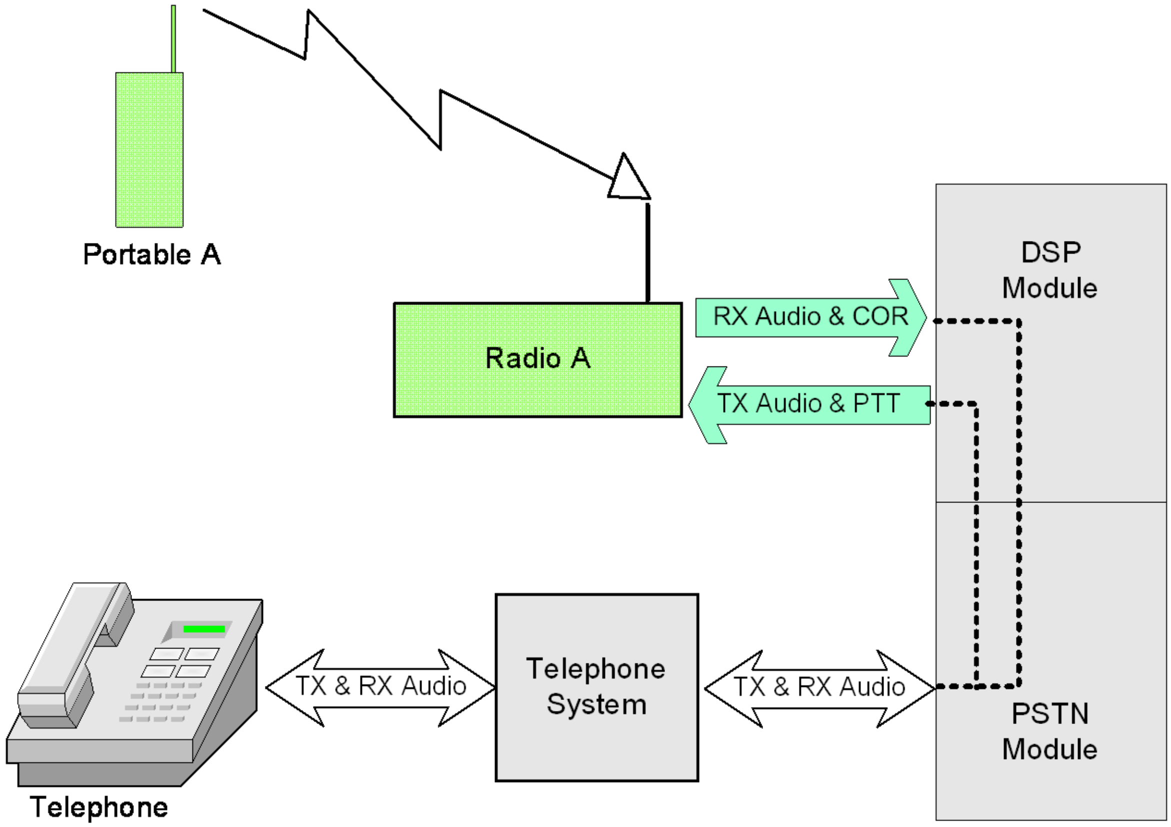


Figure 1-4 Telephone Cross-Connection

Within the ACU-2000, the PSTN module performs the 2-wire to 4-wire conversion. For a two-way conversation between a telephone caller and a radio user, the telephone system would be interfaced to a PSTN module. Within the ACU-2000, that PSTN module would be cross-connected to a DSP module, which provides the radio interface.

In Figure 1-4 above, the external interface of the ACU-2000's PSTN module is simple a pair of wires carrying 2-wire audio. The PSTN module VOX function detects any audio being spoken over the phone and causes a PTT signal to be sent out by any cross-connected DSP modules, along with the phone input audio. If the portable radio A is transmitting, the associated radio interfaced to the DSP module receives this audio and sends it to all cross-connected modules. The PSTN module places this audio on the 2-wire pair and sends if off to the telephone system.

In the ACU-2000, the incoming VOX or COR inputs are reported to the CPM module. The CPM module then commands any cross-connected modules to activate their PTT outputs. Incoming audio is placed on a rear panel audio bus, where it is picked off by any cross-connected modules for retransmission.

When a call is initiated by a telephone user (who calls the number of the phone line that has been connected to the PSTN module), the caller is prompted to use the phone's DTMF keypad to instruct the system which other party of the ACU interoperability system a connection is desired with.

If the connection is initiated from the ACU interoperability system (either by another system user, such as the operator of Radio A in Figure 1-4, or by one of the system control programs) the requester is prompted to provide the outside phone number that should be dialed.

1.4.1.3 Multiple Cross-Connections

The ACU-2000 is not limited to one-to-one conversations. Cross-connections can be made that create a large conference call with multiple users. This could include several different radio systems, a caller on a telephone line, a satellite phone or iDEN user, and a system operator using a dispatch console. All of these different users will be able to converse with each other.

These multiple cross-connections can take place either at the local level (up to 8 simultaneous two-way conversations within any ACU-2000) or over the wide area network (where the only limit is practical… how many people can engage in an effective conversation at the same time).

1.4.1.4 Monitor Connections

Any ACU-2000 module can be set to monitor any other module in that ACU-2000. When Module A monitors Module B, it hears any receive audio coming into module B from an outside source. However, module A does not send any audio to module B or otherwise affect its operation.

1.4.2 ACU Controller

The ACU-Controller provides an intuitive Graphical User Interface for easy control of an ACU-2000. The main screen, shown in Figure 1-5 provides a clear overview of current system status with icons to identify the various system components.

Cross-connections are made and dissolved by simple point-and-click procedures. Other features include the ability to store specific system configurations for later recall, a log file of system activity, and the ability for both local and remote (via an IP-based network) control.

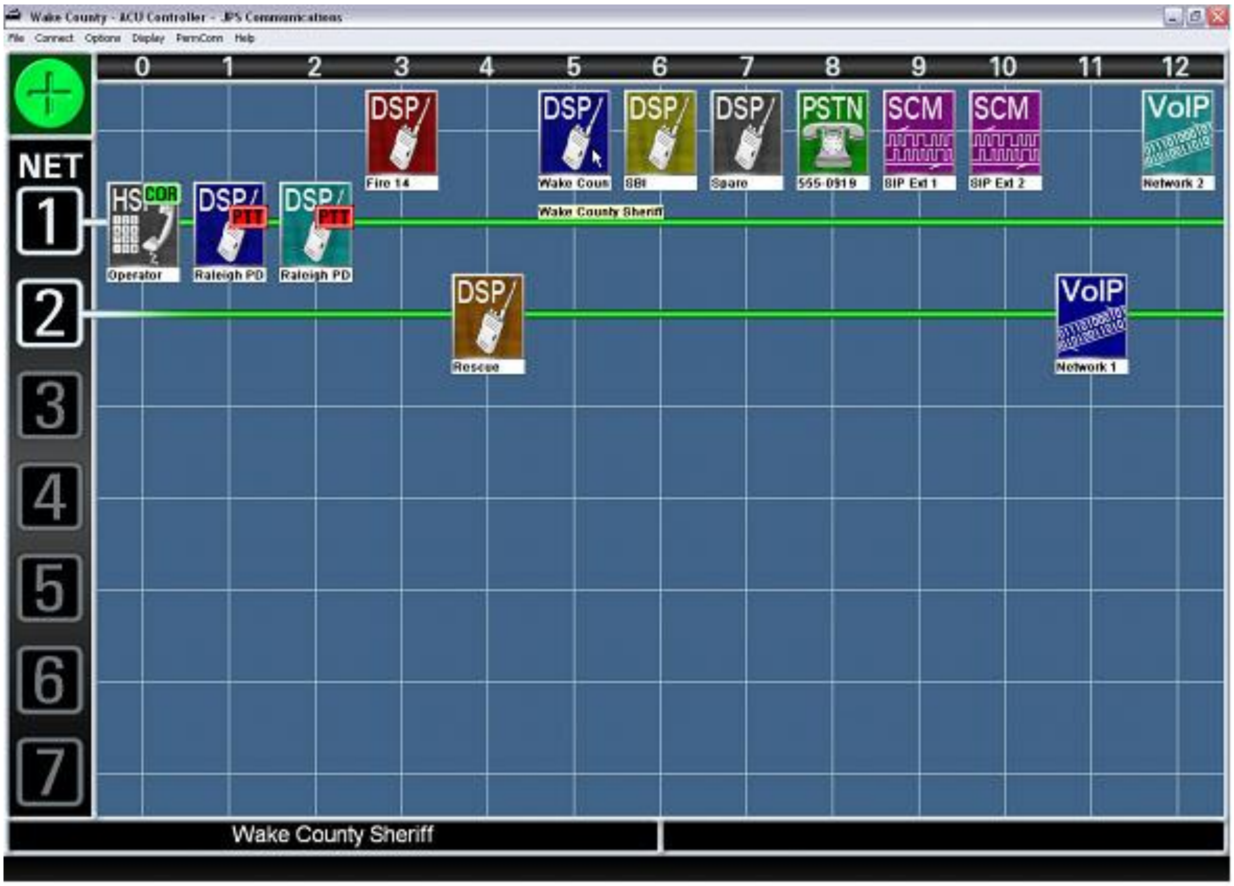


Figure 1-5 ACU Controller Main Screen

The figure above shows two cross-connections (referred to as *nets*); Net 1 is a three-way conversation among the Local Operator and radios interfaced to modules 1 & 2. The COR indication on the HSP icon denotes that the Local Operator is currently speaking into the HSP handset. Conversely, the two radios in the net are keyed up, transmitting the operator’s speech (as indicated by the PTT indicators on the extension 1 and extension 2 icons).

Net 2 is consisting of the radio at extension 4 connected via IP to a remote site via the module at extension 11.

The ACU Controller’s cursor can be seen on top of the DSP module icon at extension 5. When configured to do so, extra information is provided (near an icon or in the message area at the lower left portion of the screen) when the cursor hovers over an icon. In this case, it’s the full name given to the extension 5 radio system, *Wake County Sherriff*.

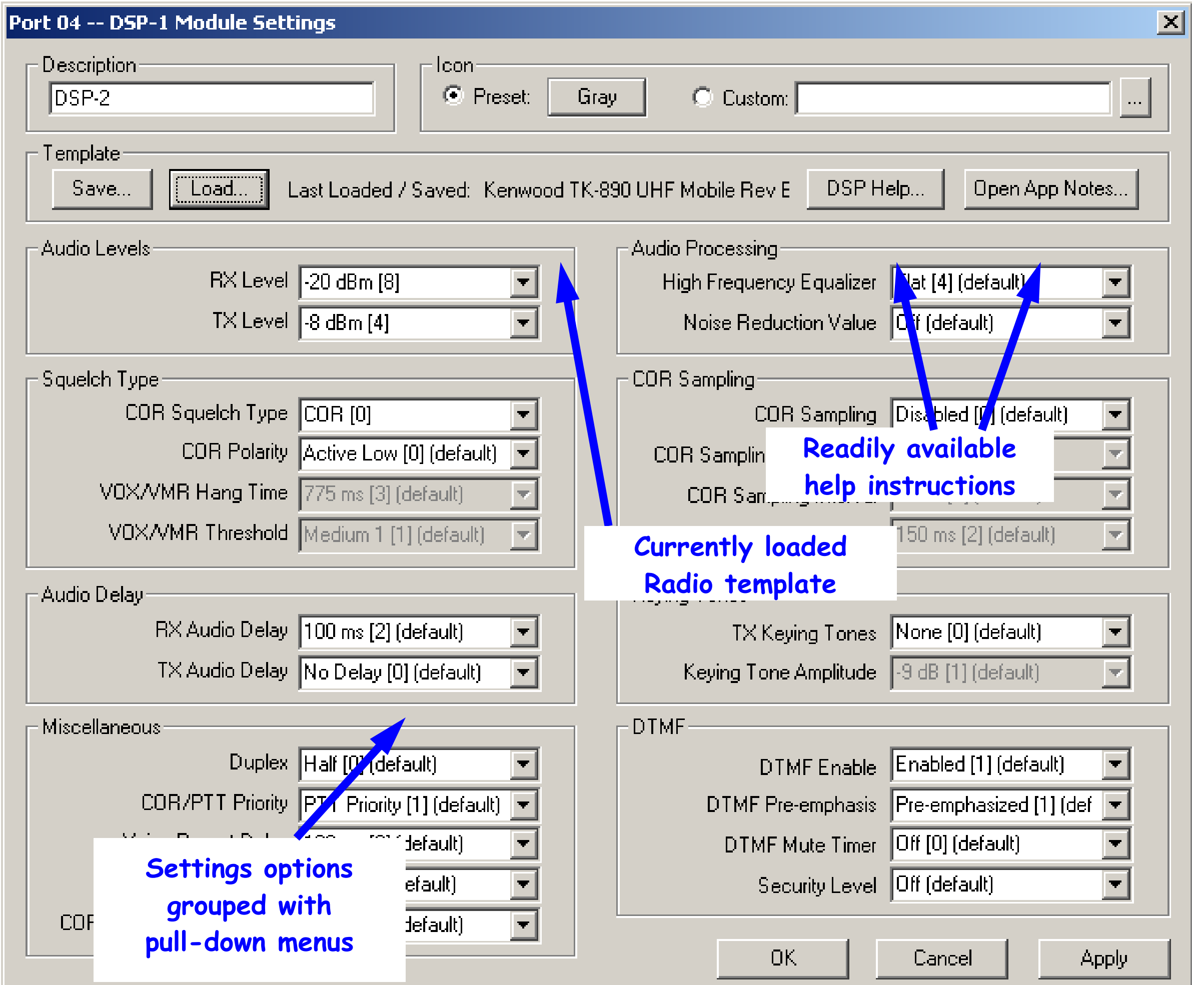


Figure 1-6 Interface Settings Screen

The ACU Controller Interface Settings Screens (one for each port, customized per the type of interface module installed) allow quick and easy adjustments of all of the interface setup and optimization parameters discussed in this manual. Optimization of a radio interfaced to an ACU-2000 is simplified further by the availability of stored radio templates. These radio templates allow instant application of any of the program's extensive library of setups for commonly used radio models.

The ACU Controller is provided free of charge with every ACU-2000; the ACU Controller manual is supplied within the ACU-2000 manual tucked into the front inside pocket. Additional copies of both the ACU Controller and its Installation and Operation Manual can be downloaded free of charge from the website as listed below.

http://www.jpsinterop.com

Please refer to the ACU Controller manual for more complete information.

1.5 IP Capabilities of the ACU-2000

The ACU-2000 has a wide variety of capabilities that are enabled by IP networks. All are explained in this section, which also introduces the two SIP Channel Modules, the SCM-1 and the SCM-2. The IP capabilities include:

- Interface of remote radios (using the IP network as an interface cable extender) via the DSP module's Ethernet port and JPS NXU devices.
- Use the provided ACU Controller software to configure, monitor, and control an ACU-2000 via an IP connection and the CPM module's Ethernet connector.
- Use Remote Radio Control Software to change radio channels and groups via an IP connection and the DSP module's Ethernet connector.
- Interface a number radios to a SIP PBX via the radio interface and Ethernet connectors of SCM-1 modules.
- Interface SIP end-user devices (such as SIP Phones or softphones) to an ACU-2000 Local Interoperability System via the SCM-2 module's Ethernet connector.
- Network together & provide centralized control to a number of ACU-2000 Local Interoperability Systems to create a WAIS (for *Wide Area Interoperability System*). Uses the intuitive, icon-based WAIS Controller software to monitor the system and manage connections. Uses the CPM's Ethernet connector for system control and the Ethernet connectors of DSP modules for RoIP (Radio over IP) communications.
- Provide an IP audio connection for a WAIS Controller operator into a Wide Area Interoperability System. There are two ways this can be done: either with the provided PCNXU software, which uses the audio and network capabilities of the computer housing the WAIS Controller software, or by an audio interface unit (called a *Local Extension*) networked by a JPS NXU-2A unit. The Local Extension unit may be customer furnished equipment or purchased from JPS Interoperability Solutions.

Figure 1-7 shows both local and networked interoperability. Compare with Figure 1-1.

DSP module C provides an RoIP network talkpath from the ACU chassis to another device (or conference of devices) elsewhere on the network. This could be a DSP module in another ACU chassis, a PC running the PCNXU software, or an NXU-2A unit that interfaces a single radio, an audio console, or some other four-wire device. WAIS Controller software is required to manage network connections among multiple ACU Local Interoperability Systems. See Section 1.7.

The SCM-2 module can interface a SIP end-point, such as a SIP Phone or a conference put together in a SIP PBX. SCM-1 modules (not shown in the figure) provide individual SIP interfaces to radios or other four-wire devices. See Section 1.6 for more information.

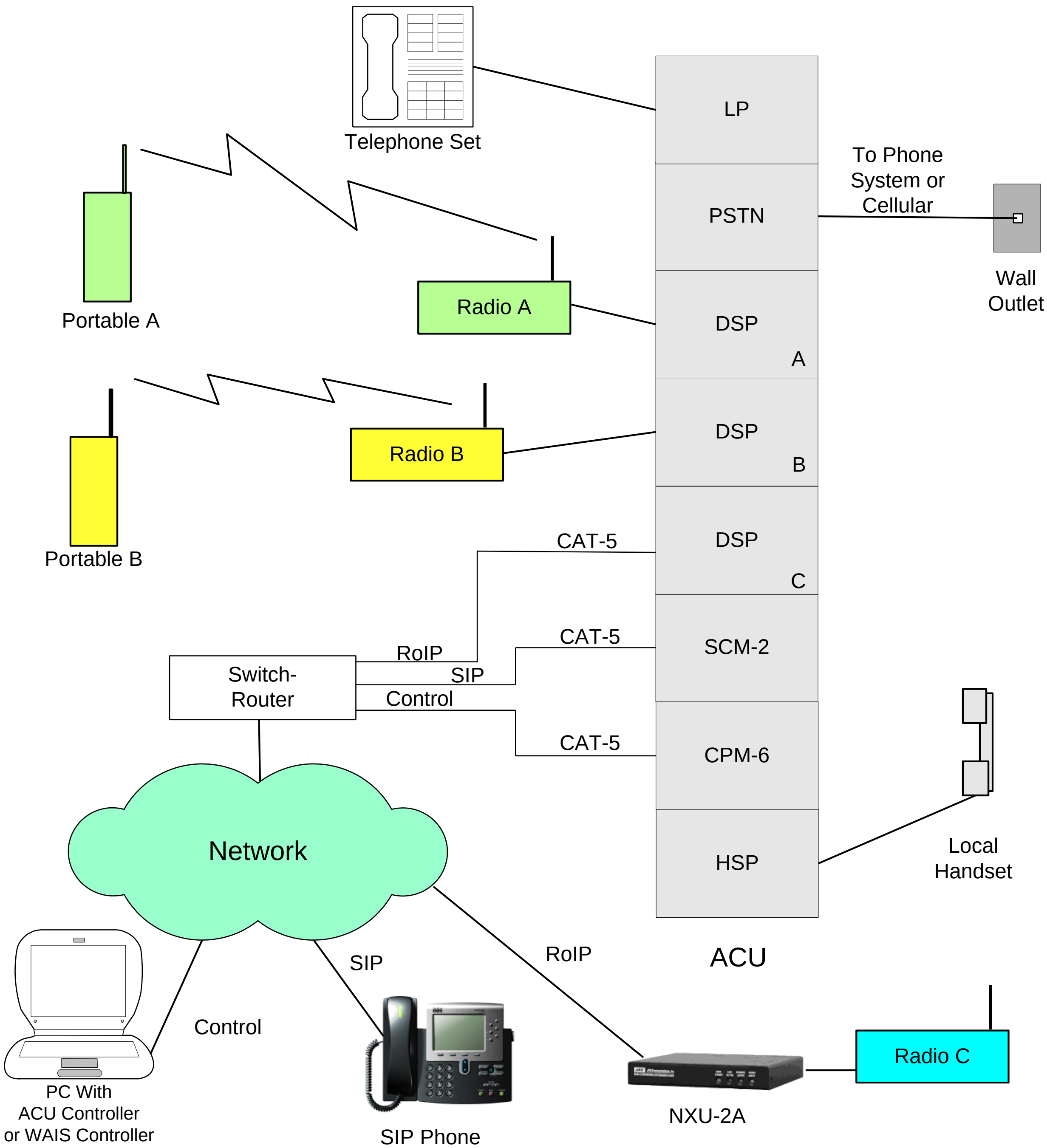


Figure 1-7 ACU-2000 Network Capabilities

The ACU Controller software is capable of ACU Local Interoperability System control either locally (as in Figure 1-1) or over an IP network. The ACU Controller is provided with each ACU-2000, along with a manual.

The purpose of the WAIS Controller software is to manage multiple Local Interoperability Systems that are interfaced by an IP network to create a Wide Area Interoperability System. This program can be purchased from JPS.

1.6 SIP Capabilities of the ACU-2000

Two different plug-in modules with SIP capability can be installed in the ACU-2000. One, the SCM-2, interfaces SIP devices to an ACU-based local interoperability system, similar to how PSTN module of Figure 1-4 interfaces a remote telephone to the system.

The second SIP-capable module, the SCM-1, interfaces a Radio to a SIP device but is never involved with any cross-connections within an ACU-2000. The SCM-1 essentially gives the radio an IP address and a SIP interface; if the SCM-1 and the radio it’s paired with are involved in any conferencing, this conferencing is done by some other device in the network, such as a SIP PBX.

Any IP-based network can be used- for example a private LAN or WAN, or the Internet. A sample application involves a system supervisor, while traveling, using his or her computer as a softphone to monitor the ACU-2000 system over the Internet and be cross-connected into the system when needed. The ACU-2000 includes PIN (personal ID number) protection features to prevent unauthorized entry into the system.

The operations of these two modules are further explained in this section.

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 SCM-1 and SCM-2 are fully compliant with the SIP protocol

1.6.1 SIP Cross-Connections With The SCM-2 Module

Figure 1-8 shows a basic connection between a 4-wire device (such as a radio) and a SIP-enabled device, in this case a SIP Telephone. This could also be another type of SIP device, such as a softphone (computer whose microphone and speaker act as a voice communications device).

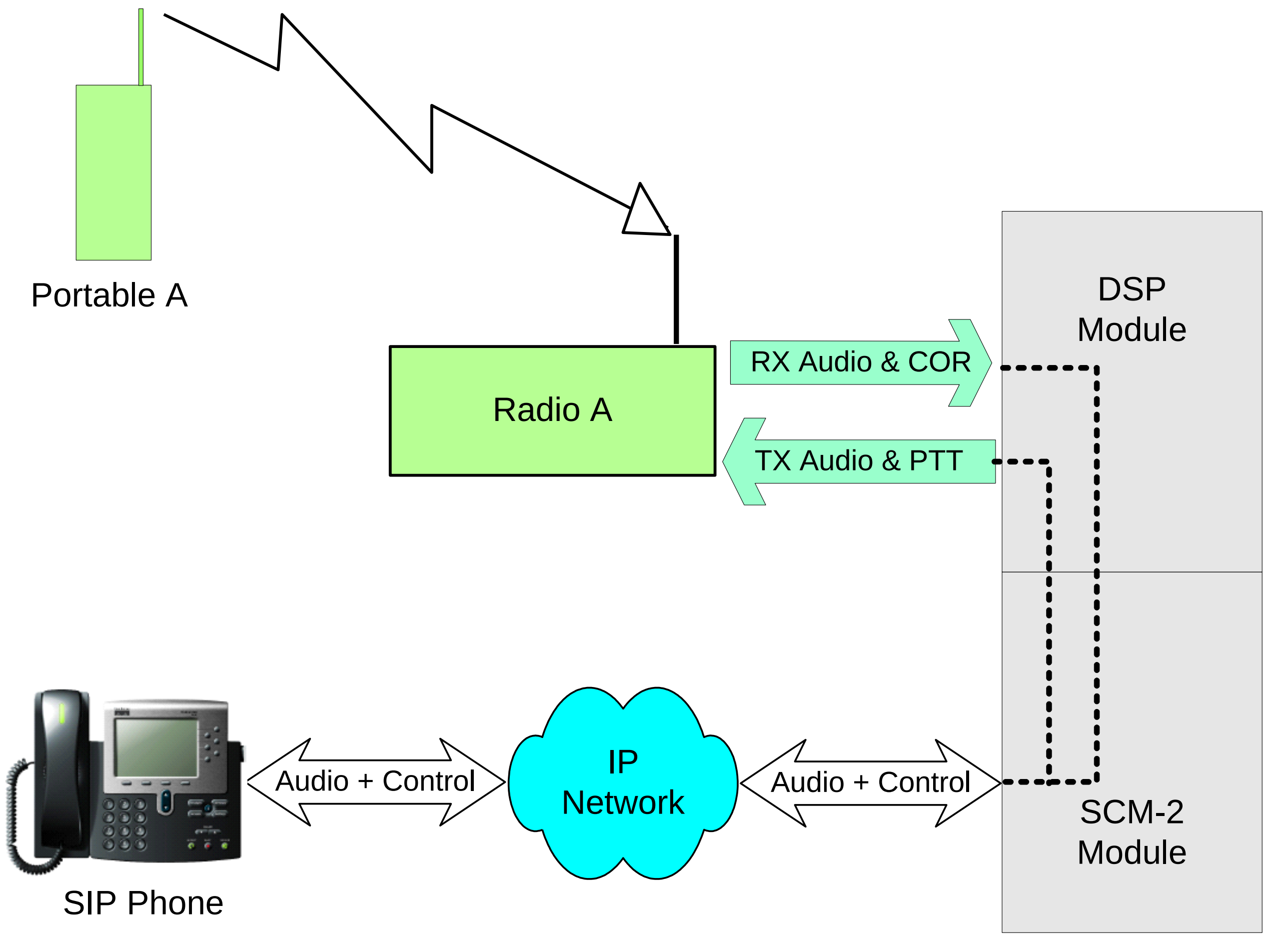


Figure 1-8 SCM-2/Radio Connection Over IP Network To A SIP Phone

The SCM-2 can interface a single ACU system radio (as shown in Figure 1-8) or any of the other CSAPs of an ACU system (the local handset operator, telephones, satellite phones, etc). Through cross-connections on the ACU backplane, communications conferences are possible that include multiple CSAPs. The SCM-2 module can engage in a one-to-one connection to any single SIP device (as shown in Figure 1-8) or, if registered as a member of a SIP PBX, the SCM-2 can interface multiple SIP devices that are conferenced within the SIP PBX.

To assist a system user, voice prompts are provided that tell what steps must be taken to initiate cross-connections and also to inform users of call progress.

1.6.2 SCM-1 As Individual Radio-To SIP Interface

Each SCM-1 module plugged into an ACU chassis creates an interface between a radio (or other four-wire device) and an IP network, using the SIP protocol to initiate and manage connections. The SCM-1 modules have no provision to cross-connect with other modules within the ACU chassis; they always interface their single associated radio to the IP network and any multi-party conferences are put together somewhere else in the network, such as in a SIP PBX or in a SIP Dispatch Console.

Figure 1-9 shows the individual SIP connections that can be made between a 4-wire device (such as a radio) and a SIP-Phone, softphone or another radio.

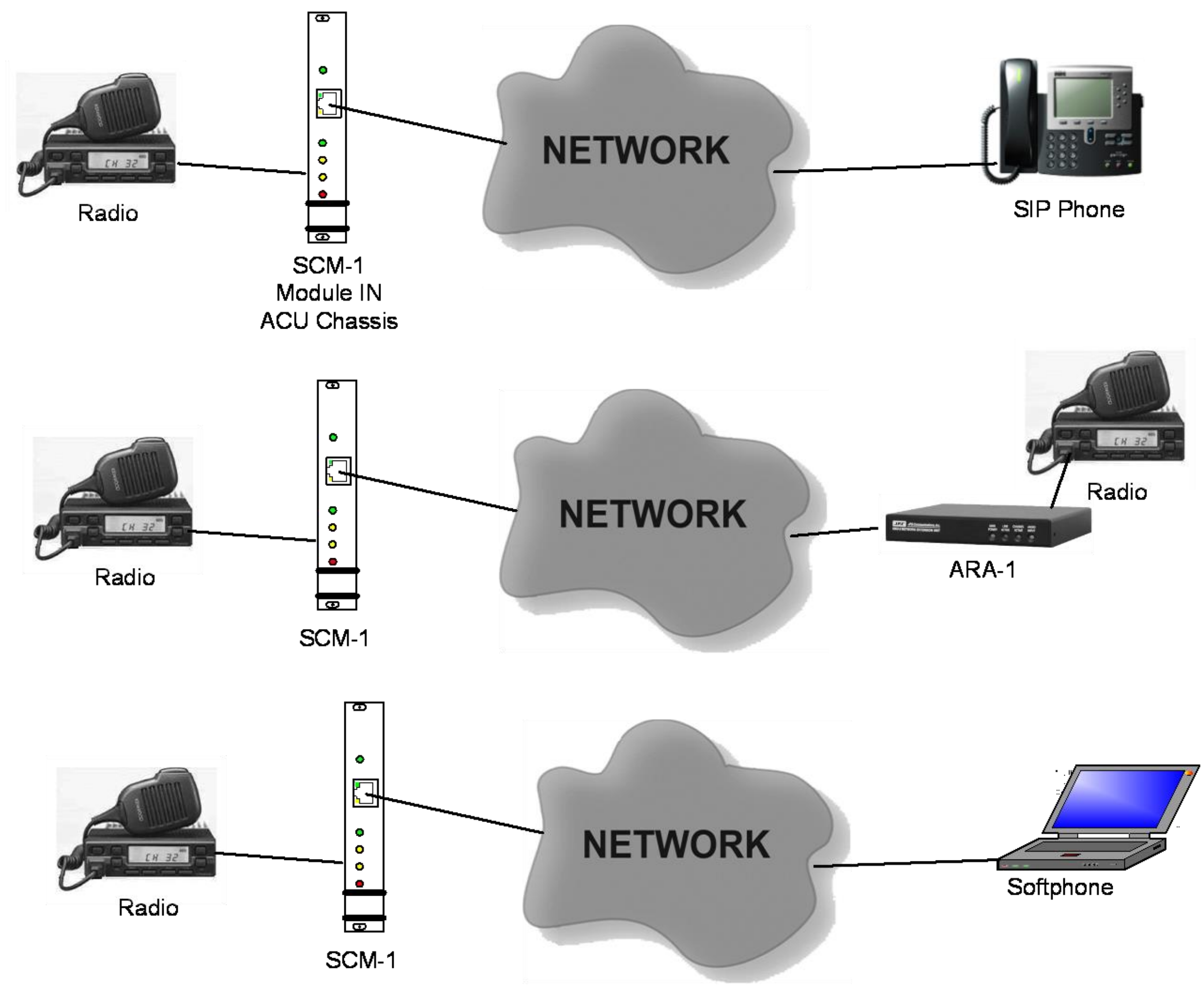


Figure 1-9 SCM-1 Interfaces Individual Radios

Note: JPS also sells a standalone unit called the ARA-1 [Analog Radio Adapter]. This unit functions just like a self-contained SCM-1, with an enclosure and external power supply. The ARA-1 may be a better choice for locations with only small number of radios to interface.

One likely use of the SCM-1 is as the radio interface of a SIP PBX or SIP Dispatch Console. For these server-based applications, the SCM-1 gives each radio an IP address and a SIP interface. The server can treat each interfaced radio just like it treats any other SIP device. In this application, all cross-connections take place in the IP realm within the server.

Figure 1-10 is a block diagram of a SIP-based PBX system. Each of the end devices is registered with the SIP PBX and given an extension number. Each member can initiate a call to any other by simply dialing the associated extension number. Conferencing of multiple end-users takes place within the PBX. The PBX is likely to also provide such features as call logging and recording, call forwarding, voice mail, etcetera.

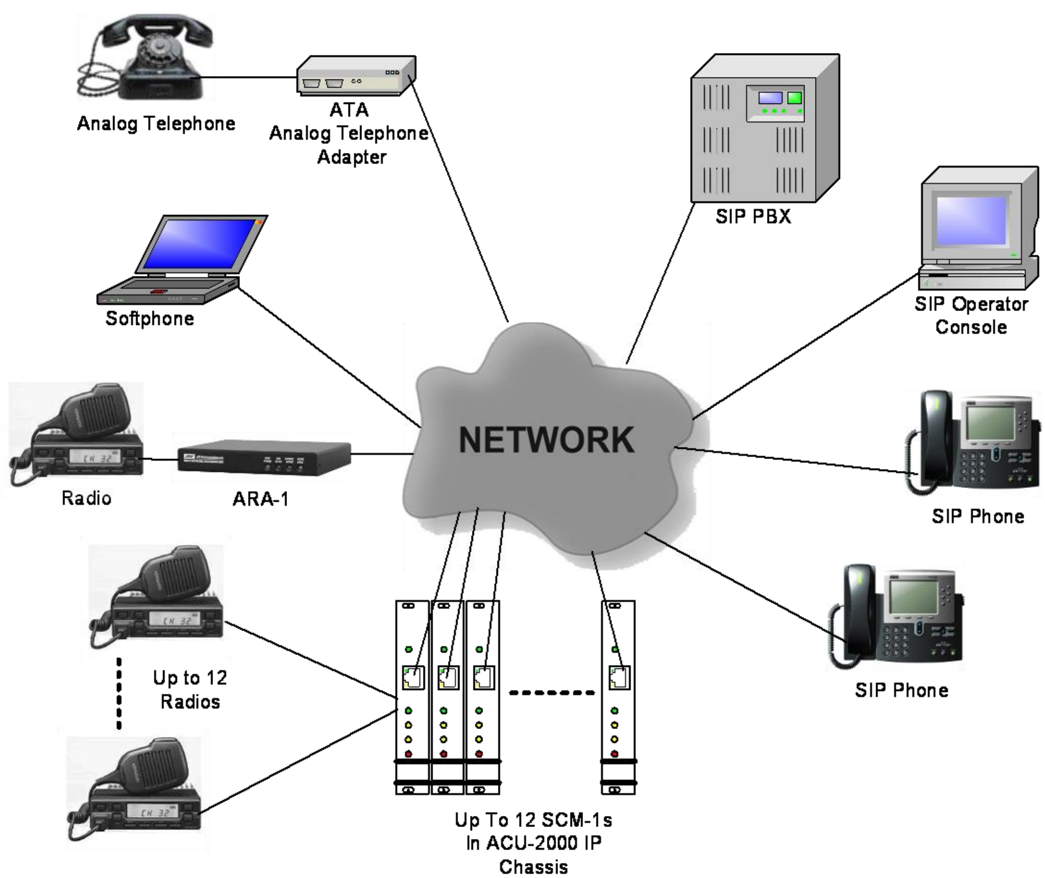


Figure 1-10 SCM-1 As An Extension Of A SIP PBX

A variety of SIP devices are shown- any (and any number) of these devices may be included in the SIP PBX:

- A SIP Phone
- A Softphone
- An analog phone interfaced by an ATA. This is a device that provides an IP address and SIP interfaces a standard analog phone (just like a JPS SCM-1 or ARA-1 gives an IP address and SIP interface to a radio or other four-wire device.
- Several radios interfaced by SCM-1 modules in an ACU chassis
- An individual radio interfaced by a JPS ARA-1 unit.

1.7 Wide Area Interoperability via IP Networks

Individual ACU-2000 based Local Interoperability Systems can be networked together to form a WAIS, or *Wide Area Interoperability System* that can encompass a large geographical region. These systems are useful for coordinating communications during a moving event such as a car chase or ambulance transport that crosses through multiple jurisdictions and/or communications systems. Hurricanes, floods, and tornados typically can produce widespread damage over multi-county areas. A wide area regional or statewide system can be beneficial in coordinating all of the different agencies that are typically involved in managing disasters of this magnitude.

Wide area interoperability over an IP network is made simple through the use of WAIS Controller software running on PCs at as many control points as desired. The WAIS Controller software manages the IP network talkpaths and ensures the most efficient use of resources when creating them. It is important to keep in mind that during a disaster, IP infrastructure could be affected and backup connectivity and routing methods should be considered.

1.7.1 The WAIS Controller

The WAIS Controller software provides an icon-based GUI interface to facilitate the control and monitoring of Wide Area Interoperability Systems. The WAIS Controller has a variety of views that allow its operators to create and dissolve cross-connections that take place within an individual Local Interoperability System as well between local systems over a network.

Multiple operators can simultaneously control or monitor a system. Each operator has a set of permissions that may limit the types of operations that the operator is allowed to perform, or the system components that the operator is allowed to control.

Figure 1-11 below shows the Overview for a wide area system that has eight different WAIS nodes (identified in the Site List at the left). The system depicted has many current cross-connections, identified by Network Group Letters, Local Group Numbers, and Module Icons. See the WAIS Controller Manual for further information.

Figure 1-11 WAIS Controller Overview Screen

1.7.1.1 WAIS Controller Dispatch Capabilities with DSP-3

The new WAIS Controller 2 has an additional capability that uses the PC's sound card to let an operator communicate with system end-users via DSP-3 modules. Any radio interfaced with a DSP-3 is available for direct communication with the Controller and appears, along with any interconnected modules, in a special screen called the Dispatch View (see Figure 1-12 below). The operator can either monitor or select (form a two-way conversation with) a DSP-3 and in either case has the option of including any other modules that are connected to the DSP-3. Multiple DSP-3s can be monitored and/or selected simultaneously, the number limited only by network bandwidth. When talking to dispatch modules, the operator has several choices for activating TX mode, including a footswitch.

Figure 1-12 WAIS Controller 2 Dispatch Screen

1.8 Related Equipment

This section introduces other equipment that can be used in conjunction with the ACU-2000 to monitor and control the unit, or to extend its range to an IP based network. Also described is the ACU-T, a version of the ACU-2000 in a Tactical Package.

1.8.1 NXU-2A

The NXU-2A Network Extension Unit is a separate product. It interfaces many types of four-wire communications devices (such as radios or dispatch terminals) to an IP network. The NXU-2A provides an RoIP link, which includes radio control signals (PTT, COR, RS-232) along with a VoIP audio path. A variety of vocoders can be configured; this allows the VoIP linked to be optimized based on its purpose and the bandwidth available. The NXU-2A is designed to operate over TCP/IP networks in conjunction with standard networking equipment- switches, hubs, routers etc.

Figure 1-13 shows a simple NXU-2A application were the unit is basically acting as a cable extender allowing the audio console to operate the radio with the radio located anywhere on the network. The audio console sends TX audio and a PTT signal to the radio, and the radio sends a COR (unsquelched indication) as well as RX audio back to the console. The NXU-2As can also pass RS-232 to control radio functions such as frequency or power level. Once the network connection between the NXU-2As is created, it remains in place until intentionally dissolved. When the console & radio are both idle (no audio being streamed across the network) the connection uses minimal bandwidth (an occasional keep-alive data packet is exchanged to maintain the connection).

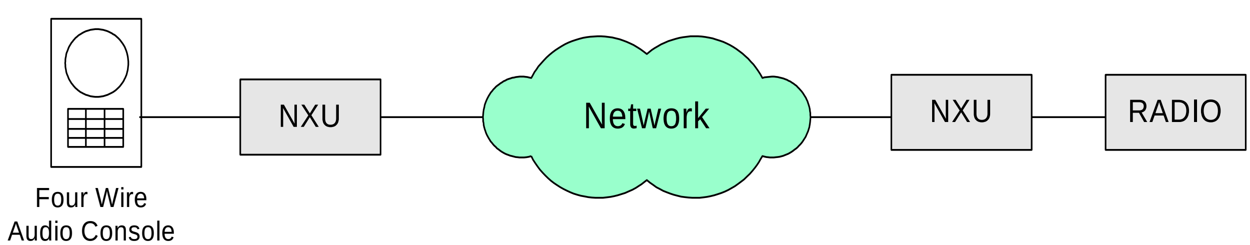


Figure 1-13 NXU-2A as Cable Extender

A similar use for the NXU-2A is to remotely connect a radio or other device (such as a four-wire audio console) to the ACU system as shown in Figure 1-14. Both configurations can be set up as a permanent connection, or the network connections can be controlled using the WAIS Controller software.

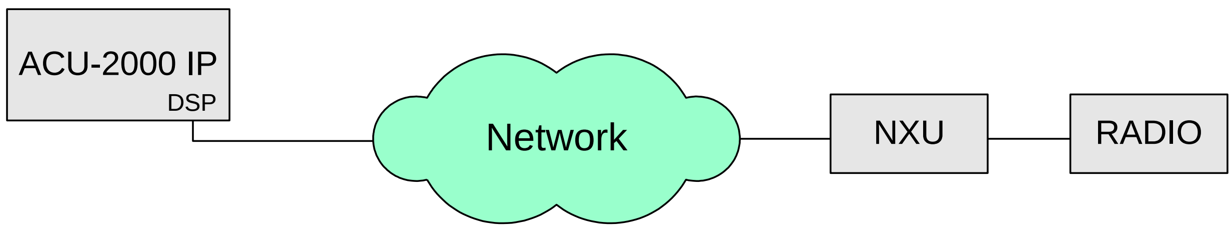


Figure 1-14 NXU-2A as ACU-2000 DSP- Module Network Interface

Figure 1-15 shows a range of NXU-2A applications, all interfacing the same network. In the type of system depicted, the WAIS Controller would normally be used to dynamically make and break connections between the various elements.

For example, in some situations, the radio in the lower right of the diagram can be connected to the 4-wire console; at other times to any one of the ACU-2000s. Note that an ACU-2000 can have multiple links to the network.

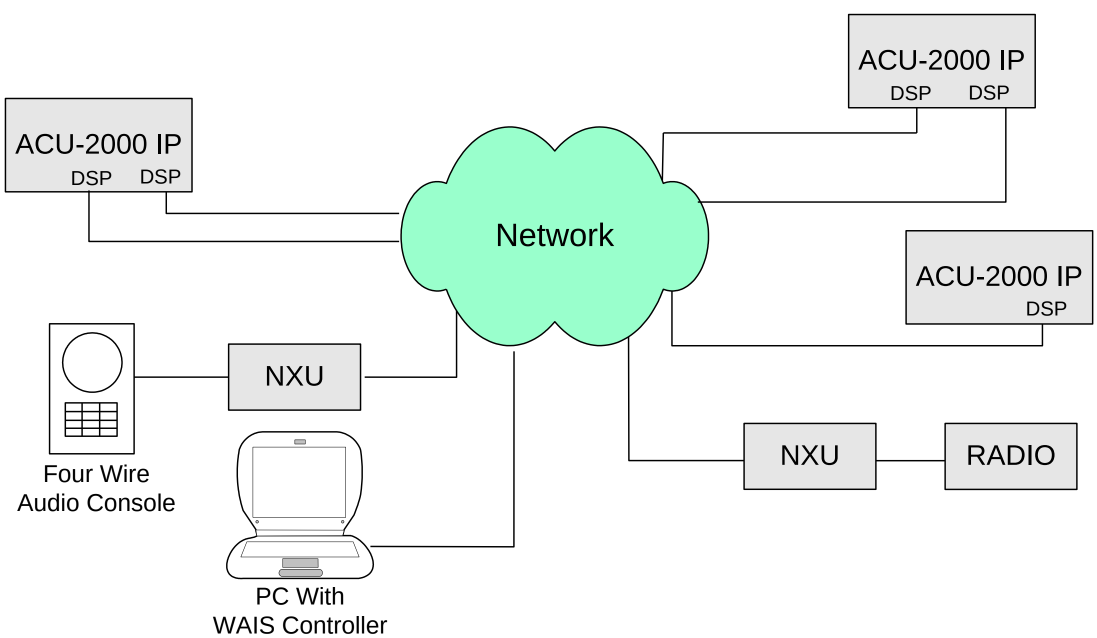


Figure 1-15 NXU-2A Applications

1.8.2 Local Extensions: LE-10 and LE-20

The LE-10 and LE-20 local control extensions provide a desktop console that can be used for communicating with any devices connected to an ACU-based Interoperability System. These consoles come equipped with a 15' cable that can plug directly into the connectors on the back of the ACU-2000 (for linking to the system via a DSP module) or to an NXU-2A to create an RoIP link to the system via an IP network. They can serve as a network control point in conjunction with a WAIS Controller-equipped PC.

The LE-10 is equipped with a handset and also includes a PTT indicator, DTMF keypad, monitor speaker, and parallel console indicator which shares PTT indication with multiple local extensions. It is configured as a 4-wire audio device. The LE-20 has similar features, but is equipped with a desk microphone rather than a handset. Photos of each are provided below.

Figure 1-16 LE-10

Figure 1-17 LE-20

1.9 Equipment and Accessories Supplied

There are two basic versions of the ACU-2000:

- A version that creates a Local Interoperability System, with local cross-connections within the chassis controlled by the CPM-6 modules. An HSP-2IP module provides a local control and monitoring. Up to 12 interface modules can be installed as needed to link devices such as radios, telephones, iDEN phones, or satellite phones to the system. This local system can be interfaced via an IP network with software control programs, remote radios, SIP devices, and other Local Interoperability Systems. See Figure 1-18 and Table 1-1.
- A version that provides as multiple radio interfaces to a SIP PBX or other SIP based server platform. This version can contain up to 12 SCM-1 modules in a chassis, along with one PSM-1IP Power Supply module. No other modules are needed or may be used in this version. See Figure 1-19 and Table 1-2.

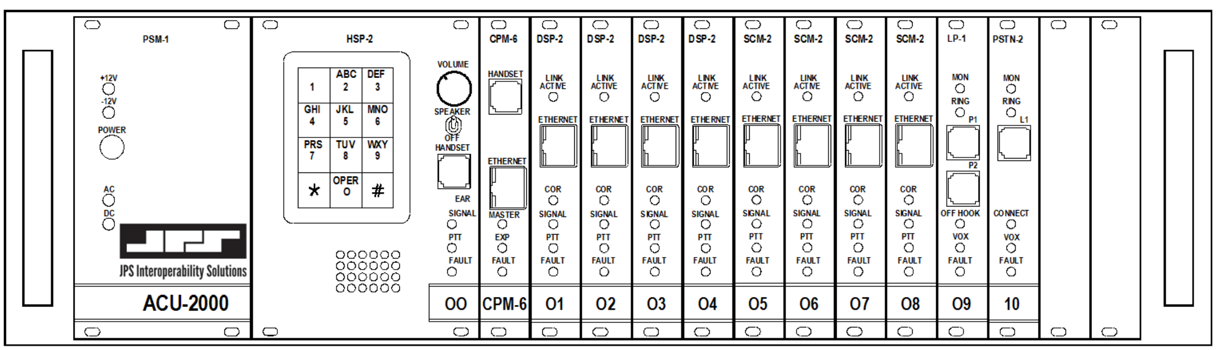


Figure 1-18 Local Interoperability System

The ACU-2000 Local Interoperability System depicted in Figure 1-18 is made up of the *Required Equipment Bundle* (The chassis/card cage with the PSM-1IP, HSP-2IP, and CPM-6 modules) and these interface modules: four DSP-2IP, four SCM-2, one LP-2IP, and one PSTN-2IP. Any grouping of interface modules may be installed, up to twelve total. There are ten interface modules in the figure; vacant slots are covered by blank plates.

It's also possible to install SCM-1 modules in the chassis, but the SCM-1s always act independently of the other modules and are never included in any cross-connections within the chassis.

Table 1-1 ACU-2000 Local Interoperability System Bundle		
Quantity	**Item**	**JPS P/N**
1	ACU-2000 Required Equipment Bundle: A single chassis ACU-2000 system consists of a chassis, PSM-1 IP, HSP-2 IP, and CPM-6 modules (the Required Equipment Bundle), plus up to 12 interface modules in any combination. Expanded systems are made of two chassis connected by an Expansion cable.	5961-230000
All items below are included within P/N 5961-230000		
1	Chassis, 19" rack mount, 3U (5.25") high	5961-231000
1	PSM-1 IP Power Supply Module	5951-833000
1	HSP-2 IP Handset/Speaker Module	5040-632200
1	CPM-6 Control Processor Module	5961-233000
1	Operation & Maintenance Manual	5961-230200
1	ACU-2000 Resource CD (Includes ACU Controller Software)	1096-129100
1	Accessory Kit	5961-230150
Interface Modules		
Quantity	**Item**	**JPS P/N**
As Required	DSP-2 IP Module Main Interface module for linking radios and other 4-wire devices to the ACU system. Can also function as RoIP network link.	5961-838000
A/R	SCM-1 Module SIP-to-Radio Interface Module (no local cross-connections).	5061-300000
A/R	SCM-2 SIP Interface Module – links SIP devices to the ACU system	5061-330000
A/R	PSTN-2 IP Module Interface module for linking to a PSTN, or linking a SATCOM Terminal, Cellular Phone or other similar 2-wire devices to the ACU system.	5050-330000
A/R	LP-2 IP Module Interface module for linking to 2-wire devices such as a local telephone set to the ACU system.	5070-430000

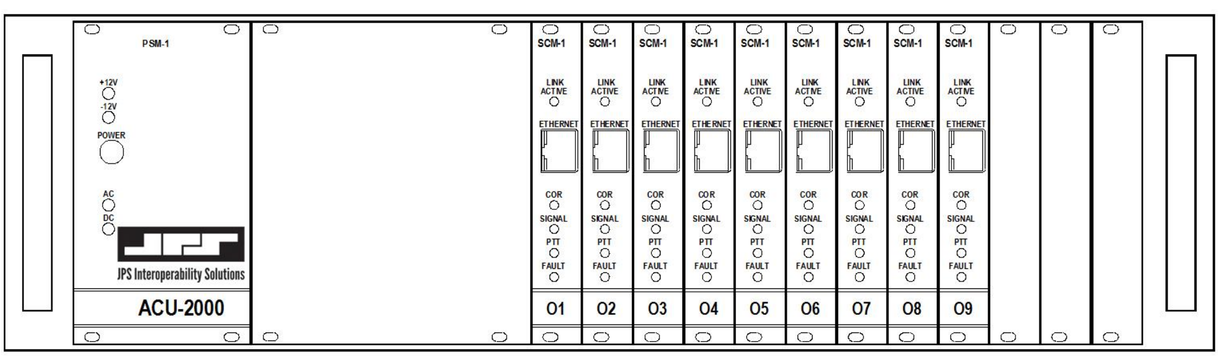


Figure 1-19 Server-Based System

This version of the ACU-2000 consists of multiple SCM-1 SIP-to-Radio interfaces housed in a chassis, powered by a PSM-1 IP power supply module. Figure 1-19 shows the Server-Based System Bundle (the chassis & power supply module) with nine SCM-1 modules installed. A total of twelve SCM-1s may be installed; empty slots are covered by blank plates.

Table 1-2 ACU-2000 Server-Based System Bundle

Quantity	Item	JPS P/N
1	ACU-2000 Server-Based System Bundle: A single chassis ACU-2000 system consists of a chassis and PSM-1 IP module. No need for linking chassis to create expanded systems.	5961-240000
All items below are included within P/N 5961-240000		
1	Chassis, 19" rack mount, 3U (5.25") high	5961-231000
1	PSM-1 IP Power Supply Module	5951-833000
1	Operation & Maintenance Manual	5961-230200
1	ACU-2000 Resource CD (Includes ACU Controller Software)	1096-129100
1	Accessory Kit	5961-230150
Interface Modules		
Quantity	**Item**	**JPS P/N**
As Required	SCM-1 Module SIP-to-Radio Interface Module (no local cross-connections).	5061-300000

Table 1-3 Accessory Kit

Quantity	Item	JPS P/N
1	Accessory Kit- Consisting of:	5961-200150

Qty	Part Number	Description
1	0313-037770	Line Cord
3	0360-015100	Conn, cable, DB-15 plug
2	0650-010250	Fuse, 3AG, 10A, 250V, fast-acting, fuses low voltage DC bus (F1)
2	0640-016100	Fuse, 5x20mm, 1.6A, 250V, time delay, for 230 VAC operation (F2)
2	0640-030100	Fuse, 5x20mm, 3A, 250V, time delay, for 115 VAC operation (F2)
2	0650-200200	Fuse, 3AG, 20A, 32V, fast-acting, for DC operation (F3)
3	0827-000004	Cable clamp for DB-15 connector
3	0853-044001	Screwlock set, female, 4-40 x 5/16"
5	0837-103200	Truss head screw, 10-32, 3/8" for rack mounting
5	0848-100001	Nylon washer, #10 for rack mounting
1	2010-200350	Screwdriver, pocket
1	5951-707000	Extender card assembly
1	0150-200000	Handset
1	0313-060000	Handset coil cord
1	5961-295300	External Speaker with DB-15 cable
1	0313-070000	Cable, CAT-5, Standard, 6', RJ-45
1	0314-000024	Cable, CAT-5, Crossover, Red, 6', RJ-45

Table 1-4 Optional Equipment - Not Supplied	
Item	**JPS P/N**
WAIS Controller (Control Software for PCs, Wide Area Interoperability Systems)	Consult JPS
STU-III Phone Option (use with DSP-2IP module for STU-III interface)	5961-295000
ACU-Terminal Block, 1.75" (19" rack panel for multiple STU-III Option Power)	5960-708000
ACU-Terminal Block, 3.50" (19" rack panel for multiple STU-III Option Power)	5960-707000
Battery Backup Option	5961-296000
Expansion Option Cable and number kit	5961-200160
LE-10 4-Wire Audio Remote with Handset & Speaker	5961-299000
LE-20 4-Wire Audio Remote with Desktop Mic and Speaker	5961-299001
LE-30 Remote Station	5961-299002
LE-40 Remote Speaker Microphone Assembly	5961-299005
Extended Rear Panel *Panel & cables to bring ACU-2000 rear panel connectors to front or rear of 19" rack.*	5020-400200
Null Modem Cable - 6 ft. (Connect computer directly to CPM-6 RJ-45)	0313-080100
Handset Holder Kit with Positive Retention (Recommended for Mobile Applications)	0827-080805

End of Section 2

2 Installation

2.1 General

This section provides the instructions for unpacking, inspection, installation and set-up. 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 be reported to the JPS Interoperability Solutions Customer Service Department. The following information should be included in the report:

1. Order Number
2. Equipment Model and Serial Numbers
3. Shipping Agency
4. Date(s) of Shipment

The JPS Interoperability Solutions Customer Service Department can be reached by phone at (919) 790-1011, or by FAX at (919) 865-1400, or email at support@jpsinterop.com. Upon receipt of this information, JPS Interoperability Solutions 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, a Returned Material Authorization (RMA) number must first be obtained from JPS Interoperability Solutions. 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, special attention should be given to providing adequate packing material around connectors and other protrusions, such as front panel controls. Rigid cardboard should be placed at the corners of the unit to protect against corner damage during shipment. Failure to protect the corners of the front panel causes the most common type of shipping damage experienced on returned equipment.

Shipment should be made prepaid consigned to:

JPS Interoperability Solutions

Customer Service Department

5800 Departure Drive

Raleigh, North Carolina 27616

USA

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

U.S. 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

Four steps are needed to properly install the ACU-2000. These steps are:

1. Provide mechanical mounting for the unit. See Section 2.5 for instructions regarding air circulation requirements and other mechanical mounting considerations.
2. Provide the proper primary power for the unit. See Section 2.6 and 2.7.
3. Interconnect the unit with the communications system via the unit's rear panel connectors. See Sections 2.10 and Figure 2-2.
4. Check all internal set-ups and adjustments per Sections 2.10 through 2.17.

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 ACU-2000 must be installed in a structure, which 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 rise to a minimum.

2.5.1 Mounting

For applications such as mobile command centers or transportable cases or any other application where some degree of shock and vibration is expected, the ACU-2000 must be mounted with rear support brackets in addition to the front mounting screws. The rear support brackets must not block airflow to the unit.

For fixed applications such as floor mounted cabinets or racks in a fixed equipment room, rear supports are recommended, but not required. Screws are provided in the accessory kit for securing the unit to a rack via the front panel.

2.5.2 Cooling

The ACU-2000 depends on natural convection for its cooling, therefore it must be mounted in a way that allows for sufficient air circulation or unacceptably high internal temperatures may result. There must be at least one inch of air space above and below the ACU-2000 to allow air to flow through the perforated metal top and bottom covers. It may not be set on a flat surface without provisions made for air to flow through the unit.

A fully loaded ACU-2000 (with twelve modules installed) dissipates approximately 100 watts. Consider other heat sources installed along with the ACU-2000 in a 19" rack or other type of cabinet. Do not install other heat generating devices below the ACU-2000. Use forced air-cooling in the cabinet if necessary.

> ***Note: When the ACU-2000 is installed in a high RF environment such as repeater site, it is recommended cable assemblies to each module should be individually shielded. The cable shields should be connected to the connector shells so they make contact with the grounded D-subminiature connector shells or Pin1 of P1-P13 on the backplane board, and not grounded at the opposite end of the cable.***
>
> ***Interface cables purchased from JPS Interoperability Solutions are shielded.***

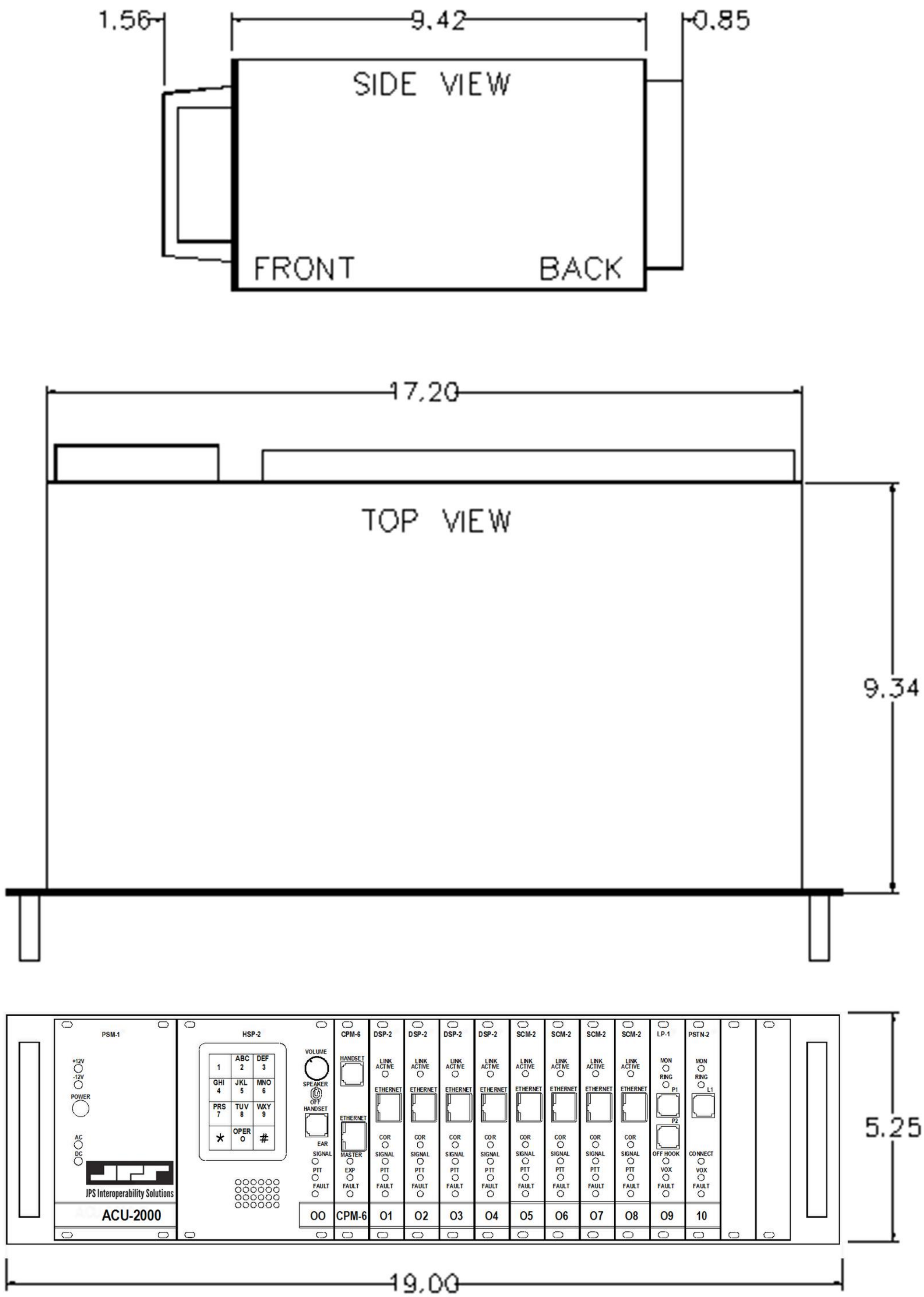


Figure 2-1 Outline Dimensions

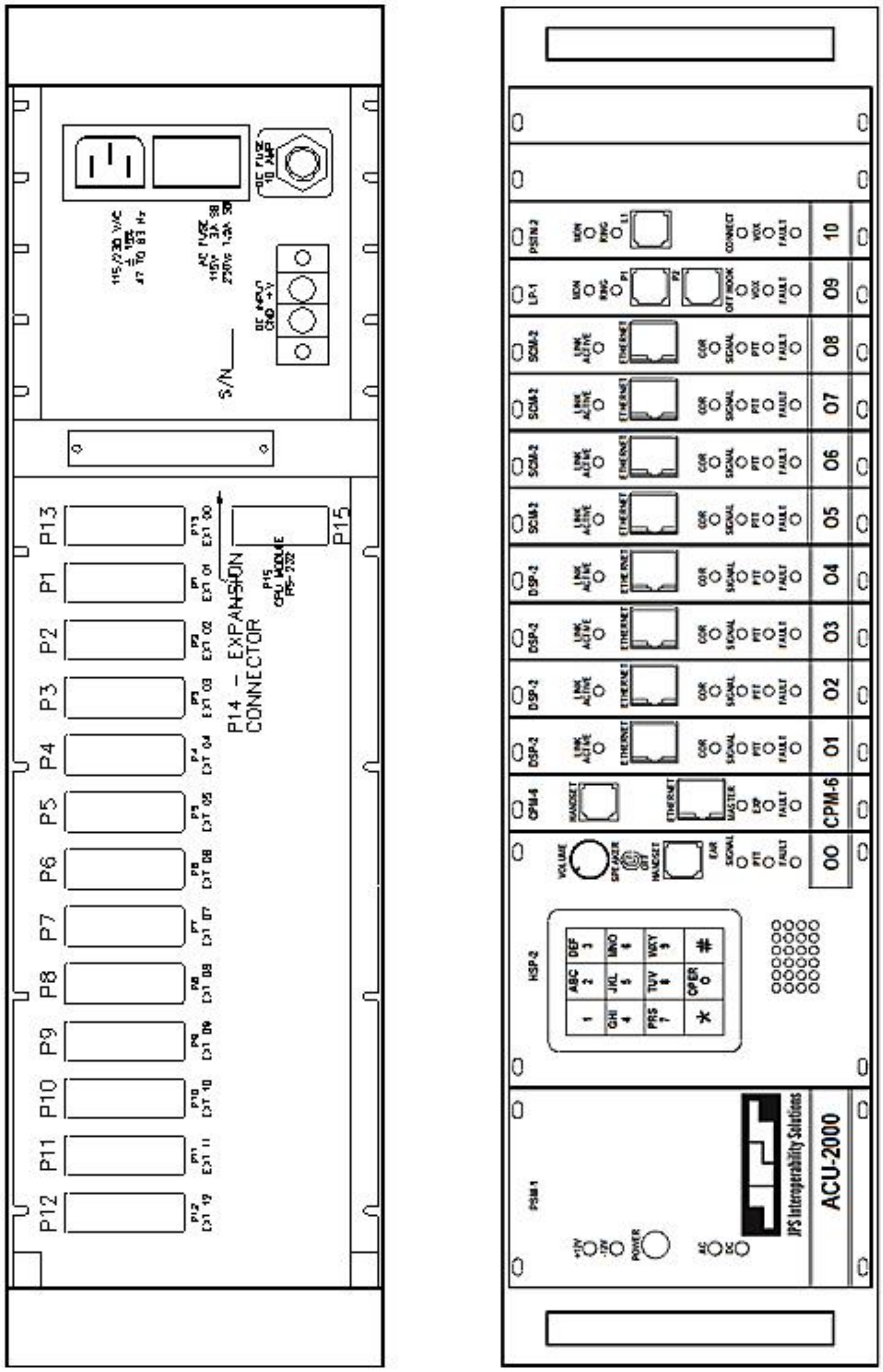


Figure 2-2 Control and Connector Locations

2.6 AC Power Requirements

The ACU-2000 is designed to operate from 115V or 230V, 47 to 63 Hz, single phase AC power source. The unit will meet all of its specifications over a voltage range of +/- 15% from nominal. The AC power consumption is 80 VA typical, 100 VA maximum.

The ACU-2000 is a microprocessor-controlled device. As with any such equipment, a very short loss of AC Power can cause operational problems and/or cause the unit to reset. JPS Interoperability Solutions recommends that the ACU-2000 be connected to an AC power source that utilizes an uninterruptible power system (UPS). If the overall site does not have UPS protection, the unit should be plugged into a smaller UPS, such as those used for personal computer systems

2.6.1 AC Line Voltage Selection

> ***CAUTION: To prevent damage to the unit, check the AC power line voltage selection before applying power. Also be certain that the unit is connected to a grounded outlet.***

As shipped from the factory, the ACU-2000 is normally set for the 115 VAC, but if stipulated on the Purchase order, the unit will instead be configured for 230 VAC. However, the voltage selection should always be checked before initial operation. The number visible at the bottom of the AC Power Input Module (located on the rear panel – See Figure 2-2) indicates the nominal line voltage range in the following manner:

- 110 position: nominal 115V Operation
- 220 position: nominal 230V operation

Note that if the AC Voltage selection is changed, the AC fuse must also be changed. To change the voltage selection, first remove the line power cord, and then use a small flat blade screwdriver to slide the fuse assembly out. A tab on the fuse assembly prevents its removal unless the power cord is disconnected, and the slot that's used when sliding the assembly out is only accessible when the cord is disconnected. Remove the fuse from the base of the assembly and replace with the correct fuse. Now use the screwdriver to push open the drawer in the fuse assembly and replace the spare fuse with a spare that corresponds with the AC line voltage. Slide the fuse assembly back into the AC Power Input Module, making sure that it is fully seated.

Finally, use the screwdriver to switch the line voltage selection to the correct position and reconnect the AC power cord.

- Nominal 115V Operation- Use 250V, 3 amp, T (time delay)
- Nominal 230V Operation- Use 250V, 1.6 amp, T (time delay)

To replace a blown fuse, follow the same procedure using the spare fuse in the drawer.

2.7 DC Power Requirements

The ACU-2000 will operate on +11 to +15 VDC, and the power supplies will automatically switch over to DC operation if the AC input voltage sags too low. Actual power consumption will depend on the number of interface modules installed. The DC power input characteristic of the unit is essentially constant power, i.e., the input power requirement is constant so the input current varies with the input voltage and number of modules installed. A fully loaded chassis consumes 56 Watts when run at a nominal 12V DC.

To find the input current given the input voltage, divide the input power by the voltage:

- 56W / 12V = 4.66A at 12V input.

To find the power consumption for less than a fully-loaded unit, use the following formula:

- Power Consumption (W) = 7.5W + (3.85W times number of modules).

2.7.1 DC Voltage Operation

The PSM-1A will automatically switch over to DC operation if AC line voltages drop too low. The PSM-1A operates with a nominal +12 VDC input only; it does not contain any provisions for +24 VDC.

CAUTION: Always disconnect both the AC and DC input power cabling from the ACU-2000 prior to servicing the unit.

Note: Any DC power supply connected to the ACU-2000 DC input must be Safety Extra Low Voltage (SELV) certified.

2.7.2 Battery Power for the ACU-2000

The ACU-2000 may also be connected to a 12V battery to provide back-up power if the AC mains fail. When powered by a +12V battery at the DC input, the ACU-2000 current consumption is the following: 0.623A + (0.32A * # interface modules). In other words, the basic chassis with PSM, CPM, and HSP modules draws 0.623 Amps, and each interface module draws an additional 0.32 Amp. So the current consumption would be shown as Load Current in **Error! Reference source not found.** below. The actual Amp-hours (AH) at a 20-hour rate are also shown for reference.

Table 2-1 Battery Sizing Chart						
Basic Chassis	**Nominal Load Current**	**Nominal 20 Hour**	**Nominal Battery Capacity Required**			
			Nominal Standby Time Desired			
# of Modules	**Amps**	**AH Rating**	**2 Hrs**	**5 Hrs**	**10 Hrs**	**20 Hrs**
2	1.26	25	7.2 AH Min@ 2.5 A Discharge Rate (STD option)	17 AH Min @ 2.5 A Discharge Rate	30 AH Min @ 2.5 A Discharge Rate	55 AH Min @ 2.5 A Discharge Rate
3	1.58	32				
4	1.9	38				
5	2.22	44				
6	2.54	51				
7	2.86	57	17 AH Min @ 4.5 A Discharge Rate	33 AH Min @ 4.5 A Discharge Rate	55 AH Min @ 4.5 A Discharge Rate	100 AH Min @ 4.5 A Discharge Rate
8	3.18	64				
9	3.5	70				
10	3.82	76				
11	4.14	83				
12	4.46	89				

To select a properly-sized Sealed Lead Acid (SLA) or GEL type battery, determine the number of modules in the shelf, and select the AH capacity with the specified discharge rate. The numbers are based on nominal discharge rates and generally available vendor sizes.

Some vendors that typically have product available to meet these needs:

- Enersys
- Panasonic
- Universal Power Group
- Yuasa

> ***Note: For a battery back-up system intended to provide extended backup time, external chargers may be needed, as the charger built into the ACU-2000 supplies only 1A trickle charge. For most of the examples above, this would amount to a trickle charge and would not be sufficient to effectively charge a battery that was allowed to fully discharge.***

2.7.3 External Chargers

If an external charger is required, it is recommended that a two-step constant-voltage constant-current type charger be used as it is fastest and most efficient, and will not overcharge the battery. Many chargers allow adjustment to optimize for cyclic use or float use. A series or parallel grouping of batteries is not recommended due to the problems in properly charging multiple batteries. When more standby time is required, simply getting a larger battery is the preferred solution.

2.7.4 Charge Switch

The PSM module Charge Switch, SW3, should be set to CHARGE only if a properly-sized Sealed Lead Acid (SLA) or GEL type battery is connected to the ACU's rear panel DC input connector. The charge current drain must not exceed 1 ampere.

2.7.5 Fuse Information

There are 3 fuses used in the ACU-2000 chassis.

F1 Common, unfiltered DC bus voltage

F2 AC input fuse

F3 DC input fuse

F1 fuses the unfiltered low-level DC bus voltage from the PSM; this bus powers the +5V DC switching supplies found on each of the other modules in the chassis. F1 prevents damage to the PSM if a short circuit or other unusual load is applied to this bus.

The PSM module must be removed from the chassis to access F1. Be sure to remove all AC & DC input cabling prior to removing or servicing the PSM power supply module.

All other fuses are located on the rear panel; they fuse input power.

Figure 2-3 provides a troubleshooting chart that helps isolate a blown fuse.

Table 2-2 ACU-2000 Fuses

F1	10AF 250V, 3AG	DC Bus	Low voltage DC to each ACU-2000 module; supplies +5V regulators
F2	3AT 250V, 5x20mm	AC Input	AC Line fuse (115 VAC nominal)
	1.6AT 250V, 5x20mm	AC Input	AC Line Fuse (230 VAC nominal)
F3	20AF 32V, 3AG	DC Input	DC Power Input Fuse (12 VDC nominal)

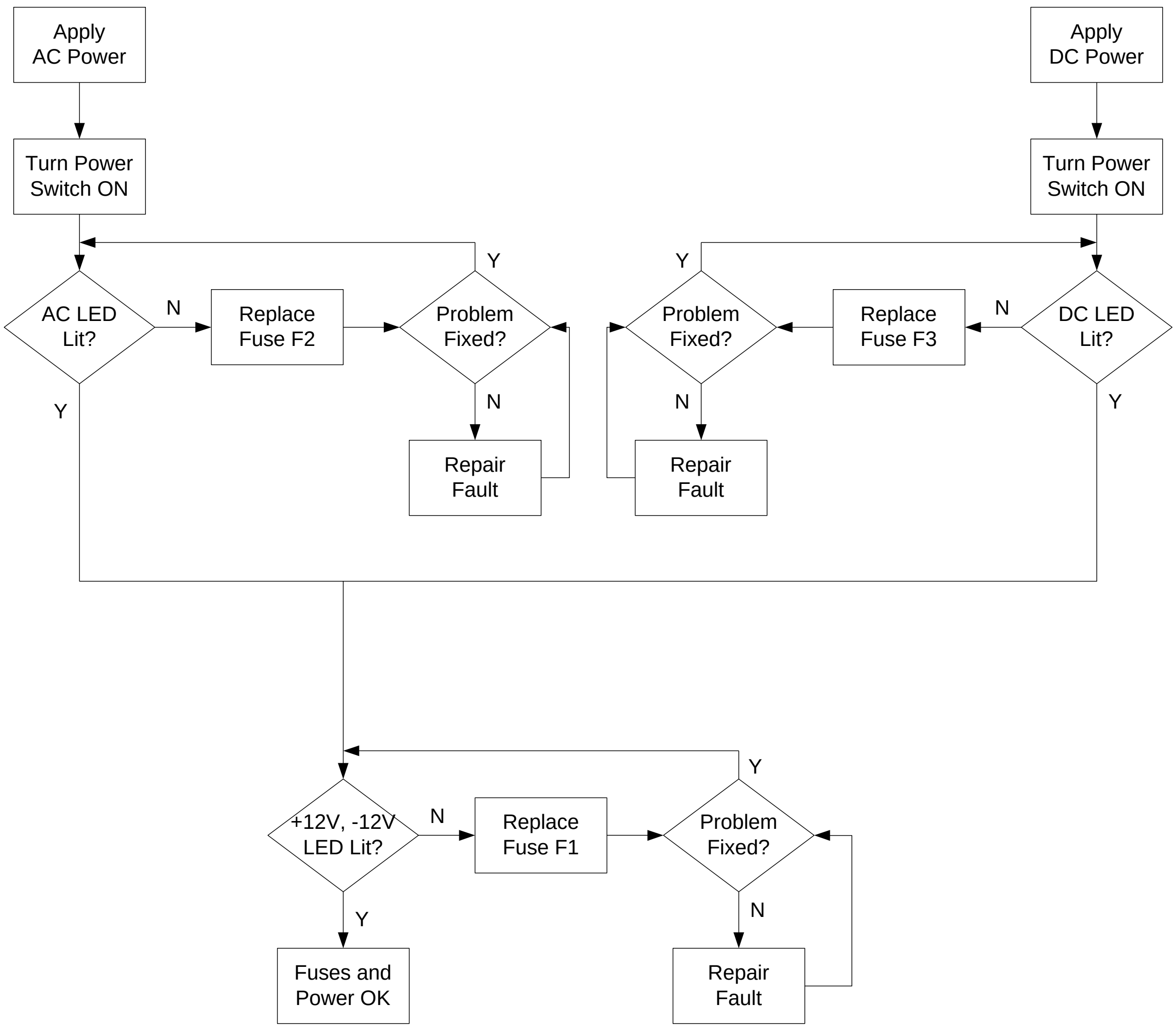


Figure 2-3 Blown Fuse Troubleshooting Chart

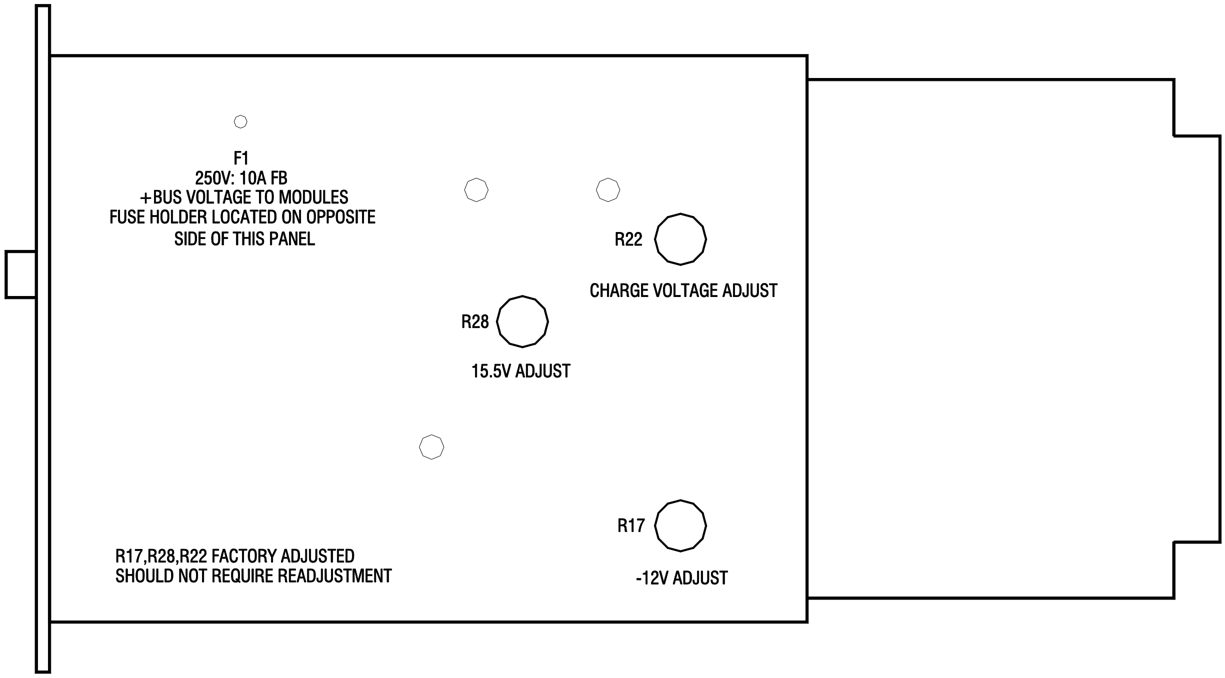


Figure 2-4 Side View of PSM Module

This is a simplified side view of the PSM. To replace Bus Voltage Fuse F1, first turn off the *Main Power* pushbutton and remove main power cabling from the unit. Loosen the four captive front panel screws and carefully slide the PSM from the chassis. To completely remove the PSM, the attached cable assembly must be disconnected (though this is not necessary to replace the fuse).

Fuse F1 is installed behind the heatsink panel as indicated above. Simply snap out the blown fuse and snap in a new one. Reverse disassembly procedures to reinstall the PSM.

2.8 Installation Checklist

Table 2-3 Installation Checklist	
Provide suitable Mounting and Cooling.	See Section 2.5.
Check AC Line voltage selection.	See Section 2.6.
DC Operation needed?	See Section 2.7.
Battery Backup needed?	See Section 2.7.2 and 2.7.4.
Make Connections to external radios or other communications equipment.	See Section 2.9 for External Interconnect Information.
Are radio interfaces properly configured?	See Configuration items for the DSP-2 in Table 2-11.
Are radio interfaces optimized?	See Configuration items for the DSP-2 in Table 2-11.
Set Telephone Line Level (if necessary).	See Section 2.13.4
Numerous other configuration options available but not included in this checklist. See Sections 2.11 to 2.17.5.2.	

2.9 Chassis Slots, Extensions, and Connectors

This section explains the ACU-2000 external connectors.

Up to 15 modules may be plugged into the ACU-2000 chassis. The left-most slot is reserved for the PSM Power Supply Module. The HSP Handset/Speaker/Prompt Module resides next to it, and the third slot is reserved for the CPM Control Processor Module. The 12 remaining slots may be occupied by any of the various ACU-2000 interface modules (DSP, PSTN, LP, SCM-1 or SCM-2).

Each of the interface module slots (and the HSP slot) has an associated D-15 connector on the backplane. The pin connections for each module depend on the type of interface module installed in the slot. The connectors for the 12 interface module slots are labeled P1-P12 on the backplane. P1 is associated with the module plugged into the slot adjacent to the CPM-4 Module, and the P12 is the connector for the module plugged into the right-most slot. Pin 1 is at the bottom of each connector. The connector for the HSP slot is labeled P13.

To reference the modules and system users, the 12 slots the interface modules plug into are called "extensions." Extensions 01 through 12 are associated with backplane connectors P1 through P12. The HSP module is identified by the extension *00* (think of this as similar to "O" for "Operator"). In an Expanded System where two chassis are connected together to provide more extensions, P1-P12 in the Master Chassis are still associated with extensions 01 to 12, and extensions 13 to 24 are associated with P1-P12 in the Expansion Chassis. The HSP slot in the Expansion Chassis is extension 25.

System users who employ the front panel keypad or DTMF from a telephone handset or radio keypad to initiate cross-connections also use these extension numbers to identify the connections they want to create. The extension numbers are part of the DTMF signal input entered by the remote user. See Sections 3.5 and 3.6 for full instructions for front panel and remote DTMF operation.

System operators who use the ACU Controller or WAIS Controller programs to monitor and control an Interoperability System will also note that the ACU-2000 interface modules are denoted by their extension numbers.

Table 2-4 Chassis Slots, Extensions, Connectors, and Modules (“Expansion” refers to the second chassis of a dual-chassis system)				
Chassis Slot	**Extension**		**Rear Panel Connector**	**Module Type**
	Master	**Expansion**		
PSM	None	None	None	PSM-
HSP	00	25	P13	HSP
CPM	None	None	None	CPM
1	01	13	P1	Various *
2	02	14	P2	Various
3	03	15	P3	Various
4	04	16	P4	Various
5	05	17	P5	Various
6	06	18	P6	Various
7	07	19	P7	Various
8	08	20	P8	Various
9	09	21	P9	Various
10	10	22	P10	Various
11	11	23	P11	Various
12	12	24	P12	Various
* Various: Any of the interface modules: DSP, SCM-1, SCM-2, LP, or PSTN modules.				

Note that in the second chassis of a dual-chassis Master/Expansion system, the HSP module has extension #25, and the interface modules are assigned extensions 13 through 24.

2.10 External Interconnect Information

2.10.1 DC Input Connector

This two-pin terminal block is mounted on the rear panel. The terminals for the ground (GND) and positive DC input voltage (+V) are clearly marked.

2.10.2 Serial Remote Connector – P15

This female 9-pin D-sub connector provides a serial RS-232 interface with the CPM module. The connector is labeled P15 on the backplane. Standard DCE pinout is used. This connector should not be used simultaneously with the RJ-45 Ethernet connector on the CPM front panel.

Table 2-5 Serial Remote Connections- P15	
PIN	**Signal**
2	TX Data
3	RX Data
5	Ground

2.10.3 Expansion Connector - P14

This connector carries parallel control signals and audio between the Expansion Connectors of the two ACU-2000 chassis of a dual-chassis system. An expansion cable may be purchased from JPS Interoperability Solutions; see Table 1-4. These connections are not intended (and must not be used) for any other purpose.

2.10.4 HSP Module Connections - P13

The HSP module must be plugged into the first slot to the right of the PSM module; this is extension 00 in the card cage, and connects it to P13 on the backplane.

An external speaker that can be plugged directly onto P13 is provided in the ACU-2000 Accessory Kit. To use the external speaker, the jumper plugs for JP1 and JP10 on the PCB must be properly configured. Set JP1 for EXT, and jumper JP10 pins 1&2.

Table 2-6 HSP-2A Module Connections- P13

PIN	Signal	Description
1	Ground	Ground connection.
2	N/C	No connection.
3	/AUX Out 1	Auxiliary Output 1- Active low; used for special functions only.
4	/AUX In 1	Auxiliary Input 1- Active low; used for special functions only.
5	Ground	Ground connection.
6	External Speaker – or – Ground for single ended TX audio- TX Out B	External Speaker output- Use JP1 to enable. Configure via JP10; pins 1-2 for Ext Spkr, pins 2-3 for TX audio ground. Ground used to allow use of standard JPS Interoperability Solutions radio and NXU-2A interface cables.
7	Audio Ground	Ground connection for audio input.
8	RXA Audio Line In	0 dBm line level audio input; 22k-100K ohm impedance via gain select jumpers JP7-JP9. (RXA)
9	N/C	No connection.
10	/AUX Out 2	Auxiliary Output 2- Active low; used for special functions only.
11	/AUX In 2	Auxiliary Input 2- Active low; used for special functions only.
12	/PTT Out	Active Low PTT output to a transmitter.
13	COR IN (/AUX In 3)	COR Input from a receiver; active low. May also be configured as Auxiliary Input 3- Active low; used for special functions only.
14	TX Out A (unbalanced)	Same audio as fed to the speaker except at 0 dBm line level from a 600 ohm source. Level is not affected by the front panel volume control.
15	Ground for single-ended RX Audio - RXB	Ground connection for Line In (RXB)

2.10.5 DSP Module Connections – P1 through P12

A DSP module may be plugged into any or all of the slots 1 through 12 in an ACU-2000 chassis. External connections are made via:

- When used to interface a radio or other four-wire device to the ACU backplane (Standard Mode), the associated rear panel D15 connector is used (e.g. a DSP module plugged into slot 5 uses the rear panel P5 interface connector).

- When the DSP module is used to create an RoIP network talkpath to the ACU backplane (VoIP hybrid mode) only its front panel RJ-45 Ethernet connector is used.
- When the DSP-2 module is to interface a radio or other four-wire device directly to an IP network (RoIP talkpath), with no connection to the ACU backplane (VoIP Standalone Mode), the associated rear panel D15 connector is used for the four-wire interface and the front panel RJ-45 connector is used for the IP connection.
- When a DSP-3 module is in Standard Mode, it may also take part in WAIS Controller enabled dispatch-related communications via its front panel RJ-45 Ethernet connector.

> ***Note: Interface cables for several hundred different radio makes and models may be purchased from JPS for use with SCM-1 and DSP modules. These cables, in conjunction with their associated Applications Notes and ACU Controller radio templates, simplify set up and configuration.***

Table 2-7 DSP-2and DSP-3 Module Connections- P1 through P12

PIN	Signal	Description
1	Ground	Ground connection.
2	RXD	RX Data; used for special functions only.
3	/AUX Out 1	Auxiliary Output 1- Active low; used for special functions only.
4	/AUX In 1	Auxiliary Input 1- Active low; used for special functions only.
5	Ground	Ground connection.
6	TX Out B	Balanced transmit audio output.
7	Audio Ground	Audio ground connection for unbalanced inputs/outputs.
8	RX In A	Balanced receive audio input.
9	TXD	TX Data; used for special functions only.
10	/AUX Out 2	Auxiliary Output 2- Active low; used for special functions only.
11	/AUX In 2	Auxiliary Input 2- Active low; used for special functions only.
12	/PTT Out	Active low PTT output to a transmitter.
13	/COR In	COR input from a receiver, active low.
14	TX Out A	Balanced transmit audio output.
15	RX In B	Balanced receive audio input.
Note: For unbalanced TX audio, ground “B” pin of audio pair; connect unbalanced audio to “A” pin.		

2.10.6 SCM-1 Module Connections – P1 through P12

An SCM module may be plugged into slots 1 through 12 in an ACU-2000 chassis. External connections are made via:

- Analog interface for radio or other four-wire devices- the associated rear panel D15 connector is used (e.g. an SCM-1 plugged into slot 8 uses the rear panel P8 connector). The rear panel connections are the same as those for the DSP-2 module and the SCM-1 can use the same radio interface cables
- IP network interface uses a front panel RJ-45 connector (standard CAT5 Ethernet cable).
- The SCM-1 module has no connection with the ACU backplane.

Table 2-8 SCM-1 Module Connections- P1 through P12

PIN	Signal	Description
1	Ground	Ground connection.
2	RXD	RX Data; used for special functions only.
3	/AUX Out 1	Auxiliary Output 1- Active low; used for special functions only.
4	/AUX In 1	Auxiliary Input 1- Active low; used for special functions only.
5	Ground	Ground connection.
6	TX Out B	Balanced transmit audio output.
7	Audio Ground	Audio ground connection for unbalanced inputs/outputs.
8	RX In A	Balanced receive audio input.
9	TXD	TX Data; used for special functions only.
10	/AUX Out 2	Auxiliary Output 2- Active low; used for special functions only.
11	/AUX In 2	Auxiliary Input 2- Active low; used for special functions only.
12	/PTT Out	Active low PTT output to a transmitter.
13	/COR In	COR input from a receiver, active low.
14	TX Out A	Balanced transmit audio output.
15	RX In B	Balanced receive audio input.
Note: For unbalanced TX audio, ground “B” pin of audio pair; connect unbalanced audio to “A” pin.		

2.10.7 SCM-2 Module Connections – P1 through P12

An SCM-2 module may be plugged into slots 1 through 12 in an ACU-2000 chassis. External connections are made to an IP network via a front panel RJ-45 connector (standard CAT5 Ethernet cable). This module interfaces only the IP network and the ACU backplane - no rear panel connections.

2.10.8 PSTN Module Connections – P1 through P12

A PSTN module may be plugged into slots 1 through 12 in an ACU-2000 chassis. External connections are made via a front panel RJ-11 connector (standard telephone cord).

Table 2-9 PSTN-2 Module Connections- P1 through P12

PIN	Signal	Description
1	Ground	Ground connection.
2	Ground	Ground connection.
3	/AUX In 2	Auxiliary Input 2- Active low; used for special functions only.
4	/AUX Out 2	Auxiliary Output 2- Active low; used for special functions only.
5	Ground	Ground connection.
6	NC	No Connection
7	Audio Ground	Audio ground connection for unbalanced inputs/outputs.
8	NC	No Connection
9	Ground	Ground connection.
10	NC	No Connection
11	/AUX In 1	Auxiliary Input 1- Active low; used for special functions only.
12	/AUX Out 1	Auxiliary Output 1- Active low; used for special functions only.
13	NC	No Connection
14	NC	No Connection
15	NC	No Connection

2.10.9 LP Module Connections – P1 through P12

An LP module may be plugged into slots 1 through 12 in an ACU-2000 chassis. External connections are typically made via an RJ-11 connector (standard telephone cord) on the front panel.

Table 2-10 LP-2 Module Connections- P1 through P12

PIN	Signal	Description
1	Ground	Ground connection.
2	NC	No Connection
3	/AUX In 2	Auxiliary Input 2- Active low; used for special functions only.
4	/AUX Out 2	Auxiliary Output 2- Active low; used for special functions only.
5	Ground	Ground connection.
6	Tel Line 1 Tip	Telephone Line 1 Tip Connection.
7	Audio Ground	Audio ground connection for unbalanced inputs/outputs.
8	NC	No Connection
9	NC	No Connection
10	/VOX	VOX Output- Active low; used for special functions only
11	/AUX In 1	Auxiliary Input 1- Active low; used for special functions only.
12	/AUX Out 1	Auxiliary Output 1- Active low; used for special functions only.
13	/PTT In	Input - Active low; used for special functions only
14	Tel Line 1 Ring	Telephone Line 1 Ring Connection.
15	NC	No Connection

2.11 Hardware Configuration Settings

In the ACU-2000, there are two types of configuration settings: Hardware Settings and Programming Settings. Changing physical pots, jumpers, or switches on some of module adjust *hardware settings*. In general, the hardware settings are done once at installation and need not be changed unless the system configuration changes (if, for example, a different radio is interfaced to the ACU-2000 rear panel), while the programming configuration items are more likely to be set once after installation to optimize system performance, and sometimes revised later if local conditions change. This section explains all hardware configuration switch and jumper settings for each of the modules in a system. See Section 2.13 for a full explanation of Programming Configuration Settings.

To access the potentiometers, jumpers and switches, on a live module, use the Extender Card found in the Accessory Kit. Remove the module to be adjusted and install the Extender Card in its place. Insert the Extender Card with its connector on the right side of the card (the Extender Card connector must be on the same side of the extender card as the module components). The Extender Card can’t be plugged into the Power Supply Module slot. All modules except the power supply module can be "hot-plugged" (removed and re-inserted with the unit's power on) without damage, but interruptions to unit operation may occur, particularly if the CPM module, which controls unit operation, is removed.

Table 2-11 ACU-2000 Hardware Configuration Settings		
Main Chassis Rear Panel	**Designator**	**Factory Setting**
AC Line Voltage 110V/220V AC nominal	AC Line Input Module	Normally 110; Set for 220V if specified on Purchase Order,
Power Supply Module	**Designator**	**Factory Setting**
Charger On/Off	SW3	Off
HSP Module Configuration	**Designator**	**Factory Setting**
Internal/External Speaker Selection	JP1	Internal Speaker Enabled
Not used with ACU-2000	JP3	N/A
Not used with ACU-2000	R123	N/A
Microphone In level	JP4 to JP6	Hi- JP6
Line In level	JP7 to JP9	Normal-JP8
P13 Pin 6 configured for Speaker Out or Ground	JP10	Speaker Out
CPM-6 Module Configuration	**Designator**	**Factory Setting**
Reset module to Factory Default Configuration	J23	Do not reset unless needed
SCM-2 Module Configuration	**Designator**	**Factory Setting**
Reset module to Factory Default Configuration	J23	Do not reset
Reserved for future use	JP23	Pins 2&3
Not used with ACU-2000	JP1	N/A
Not used with ACU-2000	JP2	N/A
Not used with ACU-2000	J15	N/A
DSP-2 & DSP-3 Module Configuration	**Designator**	**Factory Setting**
RX Input. Low or High Impedance	JP1	Low (600 ohms)
RX Input: Balanced or Single-Ended	JP2	Balanced
RX Input: AC Coupled or DC Blocked*	J15	AC Coupled
Reserved for future use	JP23	Pins 2&3
Reset module to Factory Default Configuration	J22	Do not reset unless needed
LP-2 Module Configuration	**Designator**	**Factory Setting**
Handset Speaker Volume	JP1	Norm -6 dBm
Microphone Input Level	JP2	Norm – 0dBm
Loop Current	JP5	20 mA pins 1&2
PSTN-2 Module Configuration	**Designator**	**Factory Setting**
Ringer Volume	R61	Mid-Range
Line Hold Voltage – DO NOT CHANGE	R107	Factory Set – NO CHANGE
SCM-1 Module Configuration	**Designator**	**Factory Setting**
RX Input. Low or High Impedance	JP1	Low (600 ohms)
RX Input: Balanced or Single-Ended	JP2	Balanced
RX Input: AC Coupled or DC Blocked*	J15	AC Coupled
Reserved for future use	JP23	Pins 2&3
Reset module to Factory Default Configuration	J22	Do not reset unless needed

2.11.1 Power Supply Module Settings (Charger Switch)

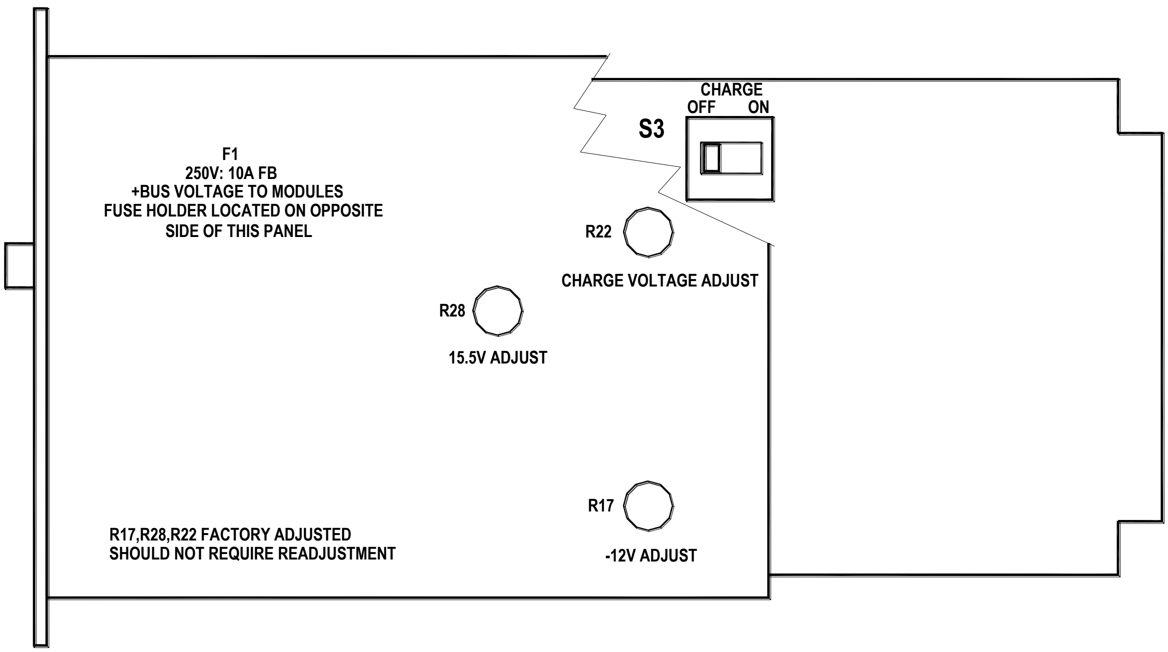


Figure 2-5 Power Supply Module Showing Charger Switch Location

The PSM module has only one user-configurable setting, S3, which turns the back-up battery charge circuitry off and on. The voltage adjustments are performed in the factory and should not require any field readjustments. Improper adjustment could cause faulty operation or damage the equipment.

The charge switch should be left off unless a back-up battery is attached to the rear panel DC power input and charging of this battery is required.

2.11.2 HSP-2IP Jumper Settings

The HSP Module has a variety of jumpers performing the functions detailed in. Default Settings are marked with an asterisk *.

Table 2-12 HSP Jumpers

JUMPER (As labeled)	POSITION	POSITION	POSITION
JP1- SPKR	**1-2 [Internal] ***	2-3 [External]	N/A
JP3- VOX Hangtime	**1-2 [Short] *** Not used with ACU-2000	2-3 [Long]	N/A
MIC Level	JP4 [-6dB gain]	**JP5 [0dB gain] ***	JP6 [+6dB gain]
Line Level	JP7 [-6dB gain]	**JP8 [0dB gain] ***	JP9 [+6dB gain]
JP-10 - P13 Configuration	**1-2 [External Speaker on pin 6 of P13] ***	2-3 [TXB/GND on pin 6 of P13]	N/A

Notes:

To use the external speaker, the SPKR jumper must be set to *External* and JP-10 must also be set to position 1-2 to bring this signal to rear panel connector P13.

MIC Level jumpers JP4, JP5, and JP6 set the gain of the MIC input. When using the supplied handset, the MIC level jumper should be set to JP5.

The Line Level jumpers JP7, JP8, and JP9 set the gain of the audio input available at the P13 RXA terminal.

JP3 and R123 (VOX hangtime & sensitivity) not used with ACU-2000.

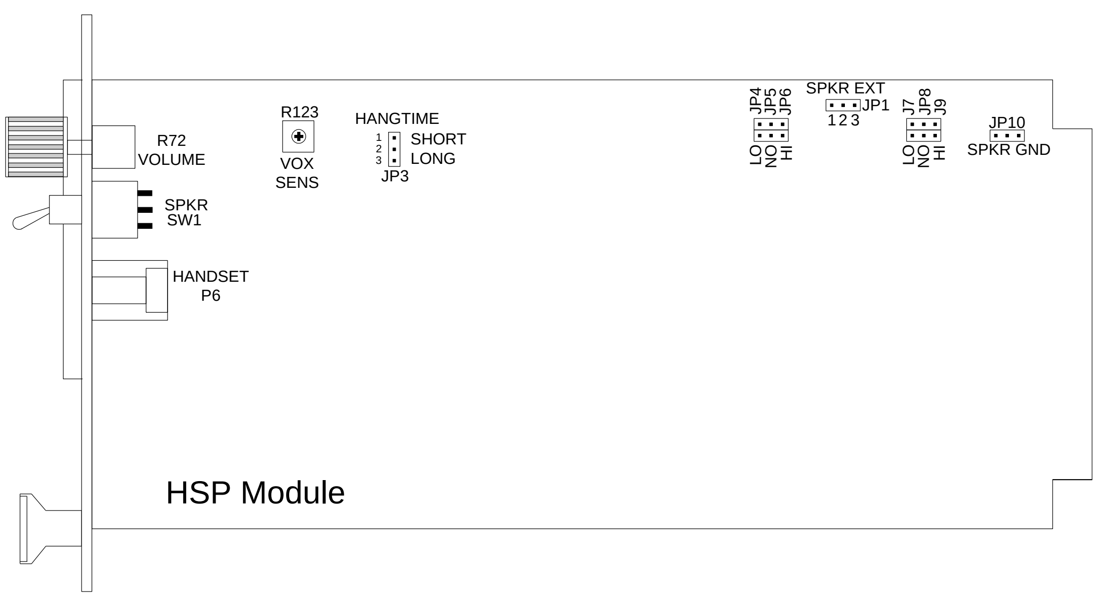


Figure 2-6 HSP Module Showing Jumper Locations

2.11.3 CPM-6 Jumper Settings

The CPM module has a single jumper; it may be used to reset the module back to factory default settings if this becomes necessary. The following settings will be reset back to factory default.

- All module settings
- All module names
- Chassis configuration
- Serial baud rate and serial port data bits, stop bit, and parity
- All IP related settings such as IP address, subnet mask, gateway
- All VoIP settings

Table 2-13 Restore Factory Default – J16	
Restore Factory Defaults	J16 1-2
Do not restore Factory Defaults	J16 2-3 *
* denotes factory default setting	

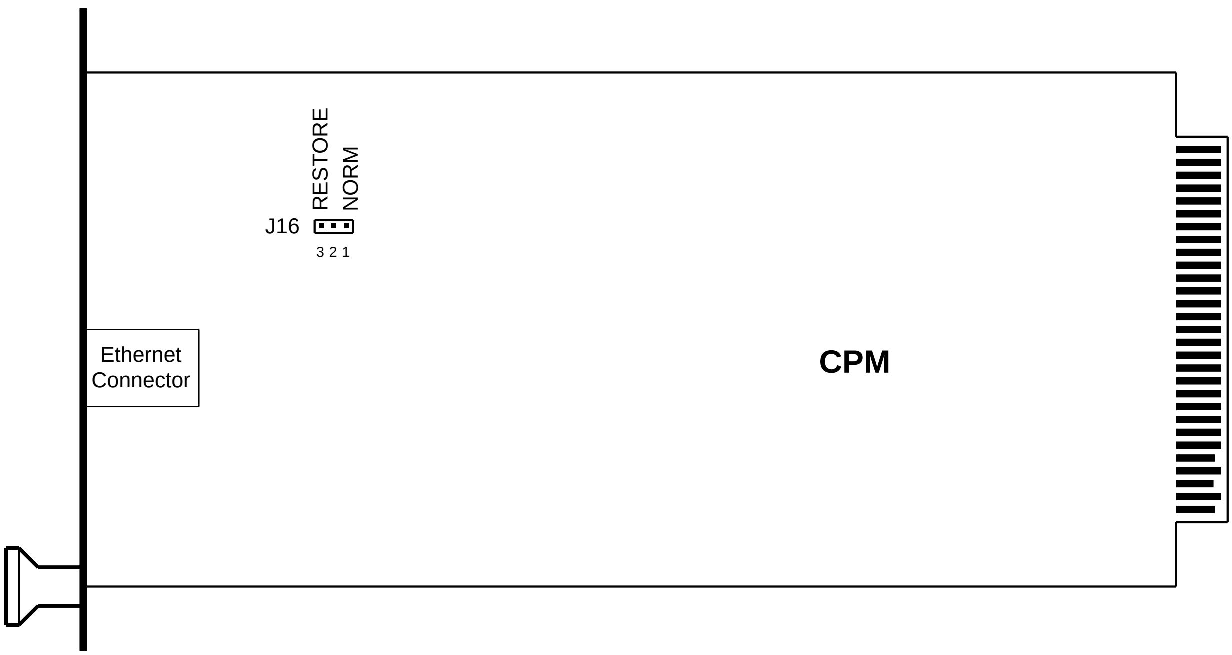


Figure 2-7 CPM Module Showing Jumper Location

2.11.4 DSP-2IP Jumper Settings

Figure 2-8 shows the locations of jumpers on the DSP module that set the configuration of the Receive Audio Input and another jumper that allows the VoIP features of the module configuration to be quickly reset to factory default settings.

Table 2-14 RX Audio Input Configuration Jumpers			
RX Input Configuration		JP1	JP2
Balanced *	**600 ohms ***	*Lo*	*Bal*
Balanced	High	Hi	Bal
Unbalanced	600 ohms	Not Available	Not Available
Unbalanced	High	Hi	UnBal

The DSP module's JP1 and JP2 jumpers set the audio input configuration for either 600 ohms balanced; 600 ohms high impedance, or single-ended (unbalanced) high impedance as listed in Table 2-14. The factory default settings are Balanced audio, 600 ohms. This is the proper configuration for most interface cables purchased from JPS. If another configuration is required, it will be stipulated on the Applications Note included with the cable (an available in the ACU Controller program).

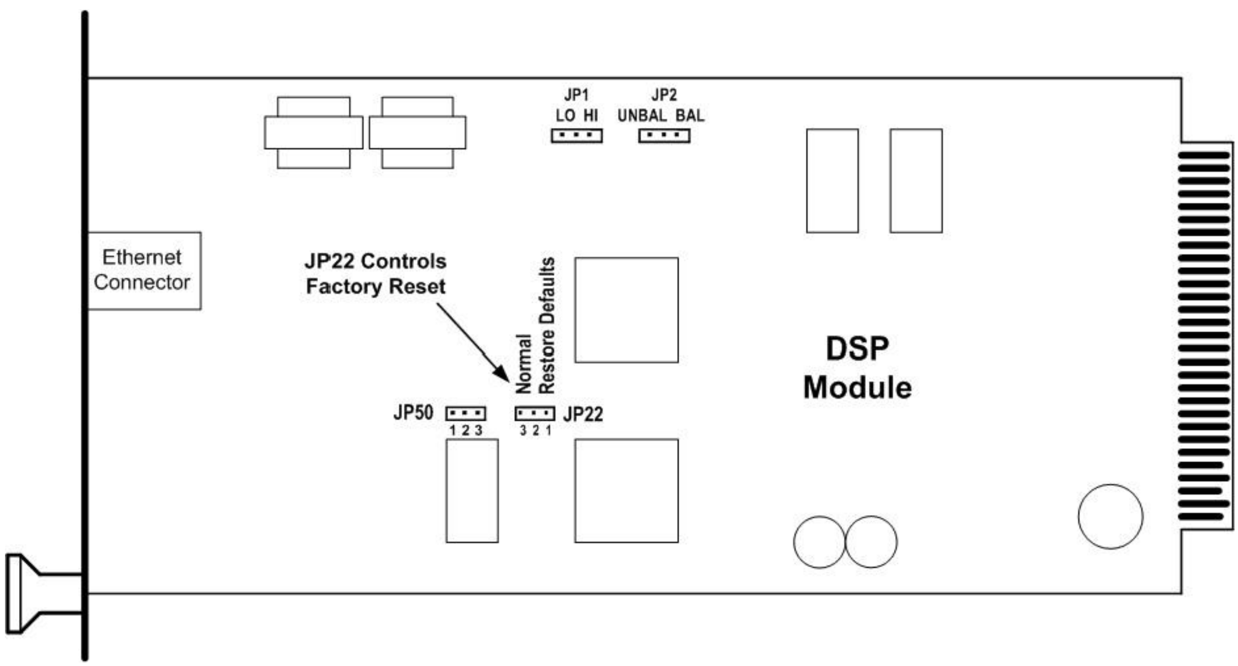


Figure 2-8 DSP Module Showing Jumper Locations

Jumper J22 restores the VoIP functions of the DSP module to factory defaults. It has no affect on the DSP's radio interface configuration items. It does not affect the jumper-selectable RX Input configuration, or the settings that apply to the module's four-wire radio interface.

Table 2-15 Restore Factory Default – J22	
Restore Factory Defaults	J22 1-2
Do not restore Factory Defaults	J22 2-3 *
* denotes factory default setting	

2.11.5 DSP-3IP Settings

Identical to those of the DSP-2IP.

2.11.6 PSTN-2IP Settings

The PSTN-2IP module has no jumpers, but there are two potentiometers as defined in the table below. The Ringer volume potentiometer is set to midrange in the factory and may be modified as desired. DO NOT MODIFY the setting of R107. Poor performance may result.

Table 2-16 Restore Factory Default – J22	
Ringer Volume	R61
Line Hold Voltage	R107
NOTE: Do not alter the setting of R107; set at initial factory test.	

2.11.7 LP-2IP Jumper Settings

The LP-2IP module has several jumper settings. The factory defaults should work well for the majority of standard telephone sets.

Table 2-17 Jumper Settings LP-2

Input/Output	Definition	Setting	Level	Jumper
Microphone In	2 Wire Input	High	-6 dBm	JP3
		Norm	**-9dBm ***	JP3
		Low	-12 dBm	JP3
Handset Speaker	2 Wire Output	High	-3 dBm	JP1
		Norm	**-6 dBm ***	JP1
		Low	-9 dBm	JP1
	Loop Current	Pins 2-3	50 mA	JP5
		Pins 1-2	**20 mA ***	JP5
* denotes factory default setting – Should work well for most telephone sets				

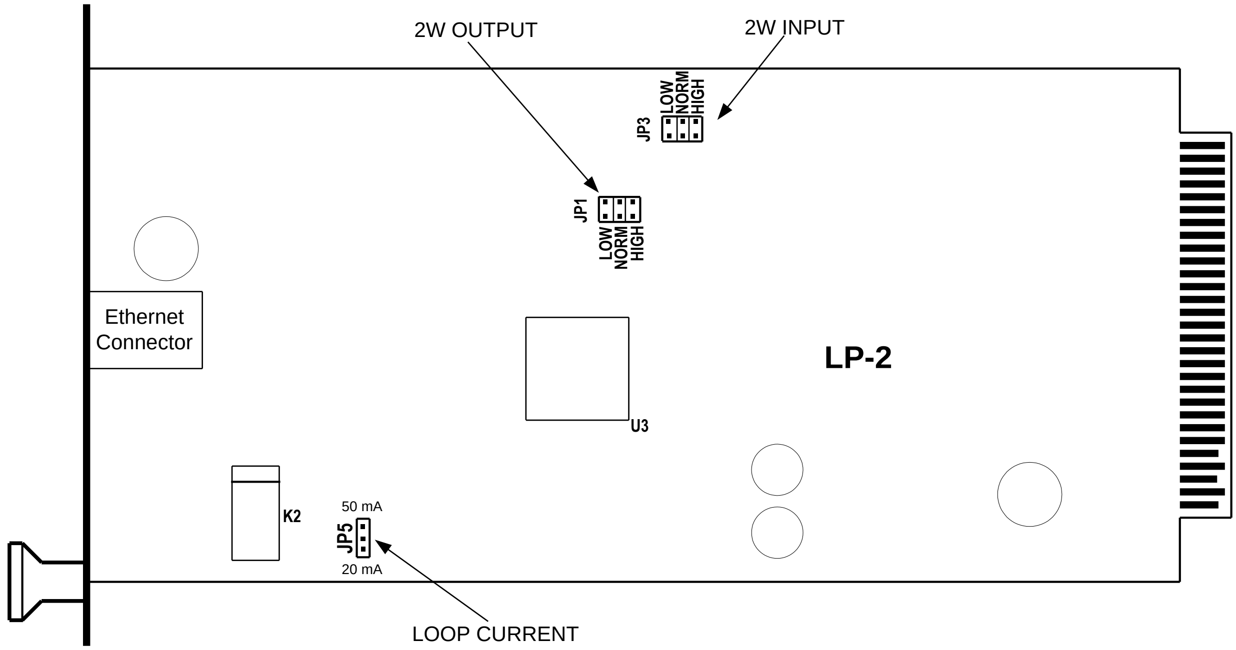


Figure 2-9 LP Module Showing Jumper Locations

2.12 Configuration Overview

There are a variety of ways to configure the individual ACU-2000 modules. They include the ACU Controller Software that is provide free of charge. The ACU Controller is the simplest and easiest method for configuring all of the settings not directly related to the unit's various Ethernet interfaces. Configuration of the Ethernet interface related items are likely done only once and is best performed by browsing to the module's IP address. If the ACU-2000 is part of a Wide Area Interoperability System, much of the configuration can be done using the WAIS Controller software. Much of the configuration of DSP. LP, and PSTN modules may also be performed using the HSP keypad.

2.12.1 Modules Configurable Via the ACU Controller

The preferred method for programming the non-IP network related configuration items of the ACU-2000 interface modules (the HSP, DSP, LP, SCM-2, and PSTN) is via the ACU Controller program. See the ACU Controller Manual for full instructions, or Section 1.4.2 of this manual for an overview. Refer also to Section 2.13 for an explanation of each programming configuration item. The programming configuration items that relate to a unit's RJ45 Ethernet port should be set by browsing to the module's IP address.

The SCM-1 module does not interact with the ACU-2000 backplane or any other modules in the chassis, or the ACU Controller. Therefore, it can be configured only by a web browser.

2.12.2 Modules Configurable Via Browsing to its Front Panel Ethernet Port

All of the features of the CPM can be configured via a browser. See Section 2.17.1.

The DSP module's VoIP features must be configured via a browser. See Section 2.17.2. The DSP-2s other features (related to radio interfacing, etc.) are best configured by the ACU Controller or WAIS Controller programs.

Both versions of the SIP Channel Module, the SCM-1 and SCM-2 can be configured by a web browser. The SCM-1 can be configured only by a web browser; some of the SCM-2 configuration programming items (particularly those that an operator may want to modify on-the-fly) may also be configured by the ACU Controller.

2.12.3 Modules Configurable Via the WAIS Controller

The WAIS Controller software may be used to configure some features of the DSP, PSTN, LP and SCM-2 interface modules. See the WAIS Controller Manual for details. If a system contains multiple WAIS Controllers, be aware that changes will show up on all WAIS Controllers in the system, but are updated only when the configuration page is opened. Change control can be password limited to properly trained personnel.

2.12.4 Modules Configurable via the HSP Keypad

Some of the ACU-2000 interface modules (DSP, LP and PSTN) can be configured via the HSP keypad as well as via the ACU Controller or the WAIS Controller. See information in Section 5.1 of the Appendix.

2.12.5 Changing the CPM Chassis Configuration Setting Via Serial Connection

If the CPM is being installed in an Expanded system (a system with more than one chassis) it may be necessary to change the CPM ***chassis*** configuration. The CPM chassis configuration may be changed by browsing to the module's Ethernet port. If this is not practical, an RS-232 serial connection may be employed instead, using a computer with a terminal program such as MTTTY or Hyperterminal. The chassis configuration will either be **Single** (a single chassis), **Master** (the Master chassis in a two chassis system) or **Expanded** (the Expansion chassis in a two chassis system).

1. Connect to the ACU-2000 serial port using a terminal program.
2. Enter the command **CONFIG** .
3. The ACU should return with *OK*.
4. Enter the desired configuration using the CHASSIS command. The example below changes the chassis configuration to "Single". Valid configurations are SINGLE, MASTER and EXPANSION.
5. The ACU will show the new chassis configuration setting.
6. Enter the command *SAVE.*
7. The ACU will reboot.

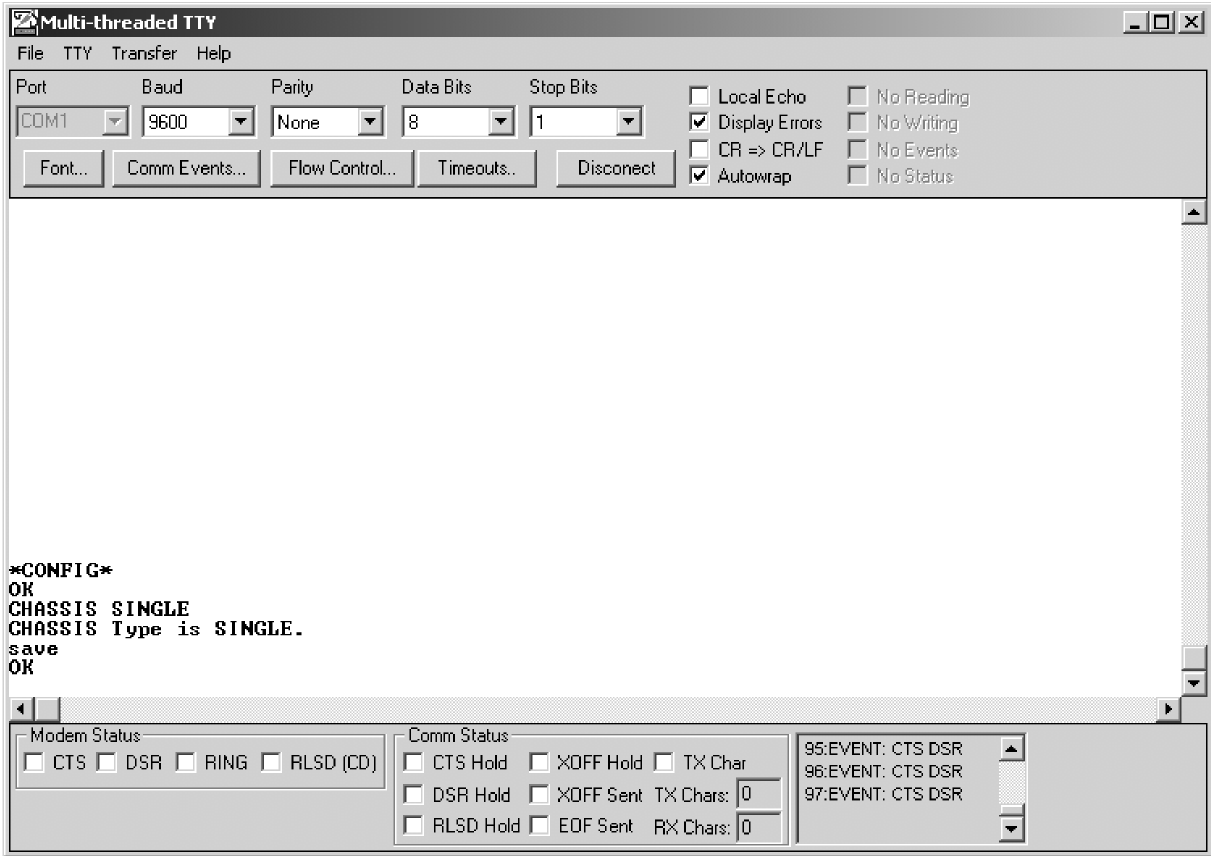


Figure 2-10 Chassis Configuration Setup

2.13 Description of Configuration Items

This section describes all configuration settings options that may be set by the ACU Controller or WAIS Controller software. Many of these configuration items can also be set via the HSP module's keypad. Not included in this section are descriptions of the settings accessed only via a web browser; these are covered in Section 2.17.

2.13.1 Generic Configuration Items

There are a several functions that apply to all interface modules (DSP, LP, PSTN, and SCM-2) modules. Many of these are apply to operations that allow system end-users to control the system via DTMF. This type of operation is fully explained in Section 3.6. If system operation via DTMF is not desired, Disable DTMF and the other DTMF settings need not be considered. For DSP modules, this applies also to the PIN Security and the Module Security Level Settings, as the ACU system requires that a proper PIN be entered only after a DTMF input is used to request system access.

2.13.1.1 DTMF Enable/Disable

This configuration item determines whether an ACU-2000 module considers any DTMF characters present in its input audio to be commands meant for that module.

When set to *DTMF Disabled* (the factory default setting) DTMF characters in the module's RX cannot be used to control the system.

When set to *DTMF Enabled*, the receiving module assumes that all incoming DTMF characters are commands and responds accordingly.

If the DTMF Mute Timer is enabled, the system mutes DTMF characters that are detected in the RX input. If the DTMF Mute Timer is disabled, any incoming DTMF is simply passed through along with the rest of the program audio.

A likely reason for setting a module to the DTMF Command Disable mode would be to prevent any outside users from connecting to the ACU-2000 system via DTMF. This is especially likely if an operator using the ACU Controller software normally controls the system, or authorized system users do not have DTMF keypads on their radios.

See also the DTMF Mute Timer adjustment.

2.13.1.2 DTMF Mute Timer

When the DTMF Mute Timer is enabled and properly set, DTMF signals in a module's incoming RX audio are detected and "Muted;" that is, they are not passed on to the ACU-2000 internal audio bus to be routed to another module. This ensures that they are not sent back out in any module's TX audio.

A module cannot instantaneously mute a DTMF signal; some time is required to detect its presence. Therefore, when a DTMF signal first becomes present in the receive audio; a short burst is passed through. The DTMF mute timer ensures that if a string of DTMF characters are present in the receive input a short burst of only the first DTMF character is passed through.

This is accomplished by muting the audio as soon as the first character is detected, and then keeping the audio muted until the first character is complete, and until enough of the next character has been received so it is detected. Each time a new DTMF character is detected, the timer is reset. When the timer expires (because no new DTMF character is detected in the receive input), the audio is no longer muted.

The factory default is DTMF Mute Timer Disabled, as the majority of ACU-2000 systems do not employ DTMF control. This setting allows the DTMF signals to be passed through the system like all other audio. It also prevents inevitable occasional "falsing" on voice signals that are similar to a DTMF character; this falsing would momentarily mute throughput audio.

If DTMF control is used, it may be desirable to mute the DTMF characters as they may be annoying to other system users, and a timer setting 1 second works well in most cases. If the setting is too low (because some system users transmit DTMF characters slowly), a short burst of DTMF will be passed through at the start of each DTMF character. The timer should be set to a value that is longer than the maximum time elapsed between the end of one DTMF character and the start of the next.

2.13.1.2.1 Data Mode & Command Mode

There are some circumstances when it's important that incoming DTMF not be interpreted by the ACU-2000 as control input, but instead must be passed on to other external equipment. The Data & Command Modes are not configuration items, but if these modes will be used, the DTMF Mute Timer configuration item must be properly set, so they are explained here.

The default mode is *Command Mode*; in this mode (as long as DTMF is enabled), DTMF characters detected in a module's RX input are interpreted to be ACU system commands. If a module is switched to *Data Mode*, this incoming DTMF is instead interpreted as commands intended for external equipment.

The system settings used are different depending on whether the module that's passing on the DTMF is a PSTN module (which regenerates DTMF) or any other type of module (which does not) as explained in the following sections.

2.13.1.2.2 Using Data Mode To Transmit DTMF via a PSTN Module

If the system will be used, for example, to allow a radio user to control a telephone answering machine via DTMF, the following will occur:

A radio with a DTMF keypad will transmit to a radio interfaced by cable to a DSP module. This DSP will be cross-connected to a PSTN module that is wired to a phone line. If the radio user creates the cross-connection using the radio's DTMF keypad, the DSP module must be in the *Command Mode* when he does so, since when in this mode, DTMF coming into the DSP module is interpreted as system command input. He must then use the keypad to put the DSP module into the *Data Mode*, so that subsequent incoming DTMF is instead interpreted as control characters intended for other equipment. While in *Data Mode* all DTMF (other than the specific DTMF sequence that signals the module to return to the *Command Mode*) is not interpreted as control commands.

While in the *Data Mode*, DTMF from the radio is detected by the DSP module and relayed via the CPM module to the cross-connected PSTN module as serial data. The PSTN regenerates and transmits the DTMF characters into the phone line. This regeneration cleans up the DTMF, so any noise or frequency-response related distortion of the DTMF characters (caused by radio transmission of the DTMF) is not passed on to the phone line.

The DTMF input characters ** 8 0* toggle a module in and out of the *Data Mode*. All modules begin operation in the default *Command Mode*. When the ** 8 0* Data/Command Mode sequence is received, the module switches to the *Data mode* and returns the voice prompt *Data Mode*. Whenever the ** 8 0* command is again received, the module responds with the *Command Mode* prompt and reverts to default operation.

Continuing with the example of a radio user controlling a telephone answering machine using the radio's DTMF keypad: First the radio user sends the appropriate DTMF to connect to a PSTN module. Following the voice prompts provided, the radio user then enters his home telephone number. If he continues to press the HSP keypad after the call is answered, no

DTMF will be transmitted until he enters the ** 8 0* command. Once he does, he may enter the password needed to gain access to his answering machine. All subsequent keypad entries will result in the transmission of the DTMF characters until he either toggles out of the Data Mode with another ** 8 0* entry or disconnects via the ** #* sequence, (which also returns the module to the *Command Mode*).

It is important to note that only the PSTN module regenerates DTMF in this way. This mode was created mainly to allow an ACU-2000 user to make a connection to a PSTN module and be able to control equipment that is connected to the phone line and uses DTMF signaling (voice mail systems, answering machines, etc.

When the system is as described here, the DSP module must have its DTMF Mute Timer turned on, so that only the regenerated DTMF will be sent via the phone line.

2.13.1.2.3 Using Data Mode To Transmit DTMF via a Module other than the PSTN

If it is desired that DTMF characters be passed through the system, *being sent out by a module other than a PSTN,* the DTMF mute timer must be turned off. This is required because only the PSTN regenerates DTMF; the others will just pass it through like any other program audio. The DTMF mute timer must be set to off on the module that receives the DTMF or it can not be passed on to the cross-connected module or modules.

Please note that it is possible, but not advisable, to retransmit DTMF, bringing DTMF into a DSP module from radio A and retransmitting the DTMF on a second radio via a cross-connected DSP module. The normal FM noise that will accompany the DTMF, along with frequency response related distortion (caused by pre-emphasis & de-emphasis in the radio audio circuits) can have an adverse effect on DTMF signal quality and detection.

2.13.1.3 PIN Security (Module Security)

The ACU-2000 Module Security feature requires that a correct PIN (Personal Identification Number) be entered before a user can make a connection and thereby gain access to the system via DTMF. This section explains the two different PIN security modes (Priority Operation and Exclusive Operation) and lists how to set-up and use both of these modes.

The Module Security Level Selection sets a module’s security level. Where n is the security level, with the security level set by n defined as: 0 = not secure (no PIN required), 1 = least secure, up to 9 = most secure.

Note that this security applies to the module that a cross-connection is being requested to. To clarify: Assume that a radio user is working on the radio channel of the system radio of ACU extension 5. This person wants to use DTMF to cross-connect with the PSTN module of extension 11 in order to make a phone patch. If PIN Security is enabled, the radio user will be requested to enter the appropriate PIN to gain access to the PSTN Module at extension 11, not to gain access to the DSP module at extension 5.

2.13.1.3.1 How PIN Security Works

- ACU-2000 extensions may be programmed for various security levels from 0 to 9. An extension set to security level 0 is available to all system users, without regard to PIN numbers. The ACU-2000 will not ask the user to input a PIN when a connection is requested to an extension set to security level 0.

- When PIN numbers are entered into the ACU-2000 database using the ACU Controller, they are assigned a security level. This security level corresponds to the extension security levels and identifies which extensions can be accessed via the PIN number.
- When a user tries to connect via DTMF to a secure extension (an extension that has a security level above 0) the user will be prompted *ENTER ID*. The user's assigned PIN must be entered at this time. If the security level of the PIN is not correct to provide access, the *SECURITY VIOLATION* voice prompt will be heard, and the connection will not be made. If the security level is correct, the connection will be allowed and the ACU-2000 will make the requested connection; when successful, the normal prompt will be heard: *n CONNECTED*, where *n* is the requested extension number.

2.13.1.3.2 PIN Security Modes

There are two modes of security operation, Priority and Exclusive. The ACU-2000 is either fully in the Priority mode, or fully in the Exclusive mode. It is not allowed to have some modules set to Priority, and some set to Exclusive.

2.13.1.3.3 Priority Operation Mode

Access is granted if the PIN security level ***is equal to or higher than*** the security level of the extension. Higher security levels are more secure; a PIN at security level 6 can access any extension set to level 6 and lower, but may not access extensions set at security levels 7, 8, or 9.

2.13.1.3.4 Exclusive Operation Mode

Access is granted ***only*** if the PIN security level ***is equal to*** the security level of the extension the user is requesting to be connected. A PIN at security level 6 will allow the user to make a connection with an extension set to level 6, but not to any extension with security levels from 1 to 5 or 7 to 9. Access to extensions set to security level 0 does not require PIN numbers.

2.13.1.4 Voice Prompt Initiation Delay

A delay can be added to the onset of system voice prompts. Different delays can be added to any of the interface modules. This variable delay is mainly used to compensate for slow-to-react equipment associated with a module. For example, if a local radio associated with a DSP module has a long settling time after its PTT is activated, it may be necessary to delay all voice prompts transmitted via the DSP. When additional delay is required, the distant radio user will not hear the beginnings of system voice prompts.

The factory default for DSP modules is 100 ms; for other interface modules the default is no delay.

2.13.2 HSP Module Configuration Items

The only configuration item for the HSP module is the generic Voice Prompt Initiation Delay.

2.13.3 DSP Module Configuration Items

See also the Generic Configuration Items that explain Voice Prompt Initiation Delay and settings related to DTMF control of the ACU by system end-users; this includes the PIN Security feature.

2.13.3.1 Receive Audio Level

This configuration item adjusts the audio receive level for a selected DSP. A correct receive level setting is required to ensure proper operation. Too high a level may cause flat-topping and distortion, while too low a level won't provide adequate audio volume. The front panel SIGNAL LED is provided as a guide to setting the level; raise the receive volume until the SIGNAL LED flashes momentarily on voice peaks. If the LED never lights, the level is too low; if the LED is on nearly continuously, the level is set too high. The following procedure is suggested:

- Connect the normal audio source to the module.
- Set the module at its least-sensitive setting. If the SIGNAL LED is flashing on voice peaks, this setting is correct. If the LED is now on continuously, the incoming audio level is higher than can be accommodated by the DSP module, and must be attenuated before reaching the module. If the SIGNAL LED is not flashing at this lowest setting, the module's gain must be increased; proceed to the next step.
- Raise the modules receive gain one step at a time until the SIGNAL LED is flashing on voice peaks (until the LED is flashing during voice peaks but is not lit continuously).
- Among the DSP module's algorithms is a Peak Limiter to prevent clipping of loud audio peaks. This limiter has a fast attack time and slow recovery rate.

2.13.3.2 Transmit Audio Level

This configuration item allows a modules transmit (output) audio level to be programmed. The transmit level must be set correctly to insure proper operation of radios or other equipment connected to this output. Too high a level may cause flat topping, distortion, or over-modulation of a connected radio, while too low a level won't provide adequate audio volume or modulation level. If the actual audio level requirement of the radio or other connected equipment is known, select this level from available settings. These levels assume a 600-ohm termination. If the required level is not known, the following procedure is suggested:

- An input audio source for the module is required (so this audio can be sent back out through the transmit audio port). Create a cross-connection between the module to be programmed with a second module (which is providing audio), and prepare to modify the TX level of the first module.
- Determine the proper input level to the connected equipment (output level from the module). It may be necessary to monitor the module's output audio level at the connected equipment's input port with an audio voltmeter or other means.
- Start with the TX output at its lowest level.

- Raise the output level one step at a time until the proper level is reached.

2.13.3.3 COR Polarity

This configuration item allows the module's hardwired COR input to work with either an active low or an active high COR input. If the radio's COR output goes low when a signal is being received, set the input to active low; if the radio's COR output goes high when a signal is being received, set the input to active high. This configuration parameter does not need to be programmed unless the hardwired COR input will be used.

2.13.3.4 Full/Half Duplex

This configuration item configures the module for either full duplex or half duplex operation. Set to full duplex if the connected radios or equipment can transmit and receive at the same time, and can be cross-connected to other full-duplex equipment in the ACU system. Set to half duplex otherwise.

2.13.3.5 COR Type, VOX/VMR Threshold, Hangtime, and Audio Delay

The DSP modules must have positive knowledge that a valid input audio signal is present. This validated audio input is retransmitted via any other modules that are currently cross-connected to the DSP module valid input audio. This valid input indication is called COR (for Carrier Operated Relay, sometimes referred to as COS Carrier Operated Squelch). The DSP module can use an external hardwire COR line, an internal VMR (Voice Modulation Recognition) algorithm, or a VOX Squelch. In a full duplex connection, it may be desirable to ignore COR activity altogether and never mute the incoming audio. The correct selections depend on the type of radio or other equipment that is connected to the DSP modules receive audio input.

- **FM Radios-** For best reliability, use a hardwire COR signal, if one is available from the radio's own squelch circuit. If no hardwire COR signal is available, and the radio has a squelch circuit, properly adjust the radio's squelch, and use radio line or speaker audio in conjunction with VOX mode. VMR should be used for FM radios that must be operated with an open squelch (receiver noise is present when there is no signal). The VOX cannot be used in this condition because it will open on receiver noise, but the VMR opens only on speech, not on receiver noise. When used in this mode, the VMR threshold must be set to Med2 or High to avoid falsing on white noise from the FM discriminator.

- **AM Aircraft Radios-** Again, the best choice is a hardwired COR line from the radio, if one is available. If this isn't an option, VMR should be used. VMR thresholds of Low or Med1 may be most appropriate for this application.

- **HF SSB Radios-** The only reliable choice for HF radios is VMR. VMR thresholds of Low or Med1 may be most appropriate for this application.

- **Non-Radio Applications-** The choice for these applications is hardwire COR, if this signal is available. If not, use VOX if the audio is relatively noise-free; use VMR for noisy signals.

Whenever VMR or VOX are selected, the DSP will switch to default audio delay and hangtime settings that work well for each of these COR types. These default settings are recommended, but not mandatory except as explained below. When VMR or VOX is first selected, the defaults are set. The user may then make a change in these settings if any are necessary.

The Audio Muted When Squelched configuration item may be turned off so the module ignores COR and does not mute incoming audio when COR is active. This may be desirable for full-duplex setups. This may also be desirable when an operator is monitoring the activity of one or more of the system modules.

To set up the DSP for best operation with each of the COR choices:

- **Hardwire COR-** The only parameter that needs to be set is the COR polarity. If the radio's COR output goes low when a signal is being received, set the input to active low; if the radio's COR output goes high when a signal is being received, set the input to active high.

- **VOX-** The VOX algorithm will signal COR present whenever the incoming audio exceeds a set threshold. The signal can be tones, voice or noise. The VOX algorithm is looking for any audio signal above the set threshold. Three parameters determine how the VOX algorithm functions: threshold, hangtime, and delay (See definitions below). VOX and VMR use the same programming commands to set hangtime and threshold.

> ***Note: When VOX is selected, the DSP module will default to an RX audio delay setting of 60 ms and a hangtime setting of 775 ms. Other times may then be set.***

- **VMR-** The VMR algorithm is designed to detect speech in a wide range of input audio SNRs. Three parameters determine the performance of the VMR algorithm: threshold, hangtime, and delay.

> ***Note: VMR and VOX use the same programming commands to set hangtime and threshold. About default settings: when VMR is selected, the DSP will default to an audio delay setting of 220 ms and a hangtime setting of 775 ms. It's possible to reset these parameters to different values, but RX audio delay cannot be set below 220 ms, and hangtime cannot be set below 775 ms. These minimum settings are required to ensure proper VMR operation.***

Threshold: The VOX threshold is signal amplitude related: the higher the threshold, the louder the input must be to trip the VOX and open the squelch. However, the VMR threshold is not amplitude related; instead, it specifies how stringent the VMR algorithm is when deciding whether a signal contains speech or noise. Because of the statistical nature of speech and noise, the VMR algorithm is not perfect and a performance tradeoff occurs at different threshold settings: at Low threshold, the unit is least likely to fail to detect speech, but most likely to false on noise. When the Threshold is set to the High setting, the unit is least likely to false on noise, but will fail to detect some speech. The correct setting will depend on aspects of the incoming signal and the requirements of the system. A lower threshold should be used if the input noise is not excessive, such as from an AM or HF SSB radio. A higher threshold is necessary for use with an open-squelch FM radio, where full noise is present when no signal is present. The standard factory setting of Med2 should be suitable for most situations and signal types.

The threshold configuration programming command varies the threshold for both VOX and VMR. One settings option disables VOX/VMR entirely, so no level of input audio will cause the VOX or VMR to be tripped. This setting is useful only for system testing.

Hangtime: Hangtime keeps the audio path enabled for an adjustable duration after the moment when speech is no longer detected, preventing the audio from being muted between syllables or during pauses in speech.

Delay The DSP can add an adjustable delay to the module's input audio, output audio, or both. Using either VOX or VMR necessitates some amount of RX delay as described above and in the following section.

When either the VOX or VMR COR Type is selected, the ACU sets the hangtime and the RX audio to the default settings as described above. These settings can then be altered if needed.

2.13.3.5.1 Receive (Input) Audio Delay

When speech first appears at the RX audio input, some time passes before it can be detected. The adjustable audio delay prevents the loss of the audio that is received before the detection takes place and audio gates can be opened to send this audio on to other modules. The amount of delay needed depends on the type of COR detection in use, as the different methods require different processing times. When hardwire COR is used, the default delay is 20 ms, because an external COR signal normally arrives before its associated audio, so only minimum delay is needed. The VMR algorithm has a minimum speech detection time of about 100 ms, and its default delay is 220 ms, which allows time for speech to be reliably detected under most conditions. The VOX detection time is normally just a few tens of milliseconds, so its default delay time is 60 ms. The delays should be kept at the default values unless some system requirement dictates a change, such as the use of slow-to-key radios.

Be advised of the following important characteristics of the RX audio delay:

- There can never be a true zero delay for receive audio passing through the DSP module; an inherent processing delay is always present.

- The RX audio delay does not delay the handling of the COR signal. This means that if two radios are cross-connected through a pair of DSP modules, an active COR at one module will immediately key the other module. The incoming RX audio will then be delayed from being retransmitted by the connected module for the set RX audio delay time. The duration of the active COR time span (and corresponding PTT signal) will be extended by the set RX audio delay time.

- Keep the delay set as low as possible for clarity of conversation. If either the VOX or the VMR modes is selected and the first syllable or part of the first syllable is lost after a message is passed through the unit (and only the first part), most likely the audio delay should be increased. If there are dropouts throughout a transmission, check for proper input level setting as well as proper threshold & hangtime.

- Due to the longer processing time required for VMR mode delay settings below 220 ms are not advised; longer input audio delays of 260 and 300 ms can be set if low intelligibility of the received signal sometimes causes initial speech to be missed.

2.13.3.5.2 Transmit (Output) Audio Delay

The DSP can also add delay to the audio output of the module. Transmit Audio Delay is mainly used when the 4-wire device associated with the DSP is a trunked radio. When a user makes a trunked system transmission, there is a delay between when the radio's PTT is activated and when a channel is assigned so that communication may begin. Most trunk systems signal this ready status by a confirmation tone. There is no means to transfer this tone to ACU-2000 system users who are cross-connected to the trunked system. Instead, the DSP Output Audio Delay should be set to a duration that holds the TX audio until the channel has been selected so that the first syllable is not clipped.

If more than 800ms TX Audio Delay is required, contact JPS for recommendations.

2.13.3.6 COR Sampling

When a radio connected to the DSP module is operating in half-duplex mode, it cannot receive while it is transmitting. This means as long as the radio is in the TX mode, the remote radio user who's communicating to the ACU-2000 system through this radio will be locked out and unable to send any commands to the system. To make sure this condition does not last for extended periods, the module will drop PTT momentarily to allow it to check for an active COR input, which would indicate the remote user is trying to communicate with the system. If COR is detected during this sampling window, the module will hold the local radio unkeyed for at least five seconds so the remote user has time to speak or enter a DTMF command. There are three variable parameters plus ON/OFF functions associated with COR sampling: initial delay time, sampling interval, and sampling window width. In general, it is desirable to keep the sampling interval as long as is feasible and the window width as short as is feasible because each time a sample is taken, it creates a gap is put in the transmit audio, and syllables or words can be missed.

- **COR Sampling ON/OFF-** The factory default setting for COR Sampling is OFF (disabled), so no COR sampling will occur unless it's enabled (ON).

- **Initial Delay Time-** This configuration item sets how long after the start of the user-initiated PTT that the first sample window occurs. If the PTT goes inactive before the initial delay time expires, the initial delay time is reset, and starts running again at the onset of PTT. Note this time is set separately from the sampling interval, allowing it to be set longer than the sampling interval. The factory default for the initial delay is 10 seconds.

- **Sampling Interval-** The first COR sample takes place when the initial delay expires. The module momentarily ignores the system input holding it in the transmit mode, drops PTT and samples for an incoming COR. If COR is not detected, and the system PTT input remains active, the module re-asserts PTT and maintains it for a time less than or equal to the sampling interval. While PTT is continuously active, samples will continue to be taken at this interval. The factory default setting is 5 seconds. A shorter interval will allow quicker take-over of the system by the radio user, but will disrupt transmit audio more often.

- **Sampling Window Width-** This sets how long the local radio stays unkeyed and searching for COR from the local radio. The proper value depends on how quickly this radio can switch from transmit to receive and how fast COR can be detected. For Hardwire COR signals this depends on how quickly the local radio can output a COR signal in response to a received carrier. For DSP modules using VOX or VMR based COR, it depends on how long it takes the DSP module to detect a valid signal the RX audio from the radio. Keep the sampling window width as short as possible, because a gap is put into the transmit audio during this time; but not too short or COR sampling will be ineffective because the system does not have sufficient time to respond. The factory set value of 150 ms is the minimum practical value for most radios, while some radios require a window of 250 ms or more.

2.13.3.7 Noise Reduction Value

The DSP module uses time domain mode noise reduction, designed to peak up any correlated information (such as speech), in the audio passband. It reduces noise by forming dynamic bandpass filters around correlated information, thus automatically reducing the bandwidth to the minimum necessary to pass the information. This type of noise reduction is most effective on purely random noise, such as white or pink noise, and less effective on impulse noises. The noise reduction value allows the amount of noise reduction to be set in ten steps from off to maximum. Increasing the level provides more actual noise reduction, but may give a surging quality to the recovered audio depending on its frequency content. Reducing the level lowers the noise reduction but may provide the best sounding audio in some cases. The best setting in a particular application depends on the noise level and represents a balance between noise reduction amount and ultimate audio quality.

The factory default is Off.

2.13.3.8 Audio Muted when Squelched

This selection determines whether the module’s audio output to the ACU-2000 internal bus (and therefore, to other modules in the system) is muted when the module is not detecting active COR. The default setting mutes the audio when squelched, but sometimes other system requirements (such as the need for full-time monitoring of an input signal) may dictate the audio be not muted.

Default setting is muted.

2.13.3.9 Transmit Keying Tones/Keying Tone Amplitudes

The DSP can mix keying tones with its transmit audio output. This allows the DSP to signal a connected transmitter to key using only the audio output lines, eliminating the need for an extra wire to carry the PTT output. Keying tone types include a 1950 Hz continuous tone and the EIA Keying Sequence (see below). Factory default is No Keying Tones.

The Keying Tone Amplitude configuration item command pertains only to the amplitude of the 1950 Hz continuous keying tone relative to the transmit audio output. The selections range from -6 dB to -15dB. The default setting is –9 dB.

The DSP can also produce the EIA tone keying sequence using function tone F1. The EIA tone keying sequence has three tones, produced in succession:

- Alert Tone, 2175 Hz tone for 125 ms @ +10dB relative to program audio (the Alert Tone is sometimes called the *High Guard Tone*).
- F1 Function Tone, 1950 Hz for 40 ms @ 0dB.
- Hold Tone, 2175 Hz @ -20dB, mixed with the TX program audio as long as PTT is enabled (also called the *Low Guard Tone*).

The levels of the EIA keying tones are expressed relative to the normal TX program audio, the level of normal speech content. When the DSP is set for default TX audio (0 dBm) into a terminated 600 Ohm load, the levels are +10 dBm for the High Guard Tone, 0 dBm for the Function Tone, and –20 dBm for the Low Guard Tone. Since the maximum output capability of the DSP module is approximately +10 dBm, and the Alert Tone level is 10 dB above the program audio, the program audio must not be set higher than 0 dBm if EIA Tone Keying is used or distortion of the Alert Tone can result, and may interfere with detection at the transmitter. The Keying Tone Amplitude configuration item does not affect the EIA keying sequence tone amplitudes.

2.13.3.10 COR Inhibit Time after PTT

Some types of radios produce momentary (and unwanted) COR outputs just after their PTT inputs are de-activated. If a local radio connected by interface cable to the ACU-2000 exhibits this behavior, the COR Inhibit feature causes this COR signal to be ignored. Since a received signal on a system module is retransmitted on all cross-connected modules, this improper COR signal can cause cross-connected extensions to momentarily key. If both radios of a cross-connected pair exhibit this behavior, they can *ping-pong;* that is, following a desired transmission, the receiving radio will cause the transmitting radio to key again momentarily, this radio returns the favor and the process can continue indefinitely. The COR inhibit timer must be set high enough so that no COR occurs following the cessation of PTT. Observing the DSP module’s front panel COR & PPT indicators can detect the existence of this problem and confirm when it’s been resolved. If the COR LED always flashes when the PTT LED goes out, the COR Inhibit time should be increased.

The factory default is 100 ms.

2.13.3.11 PTT or COR Priority (Transmit or Dispatch Priority)

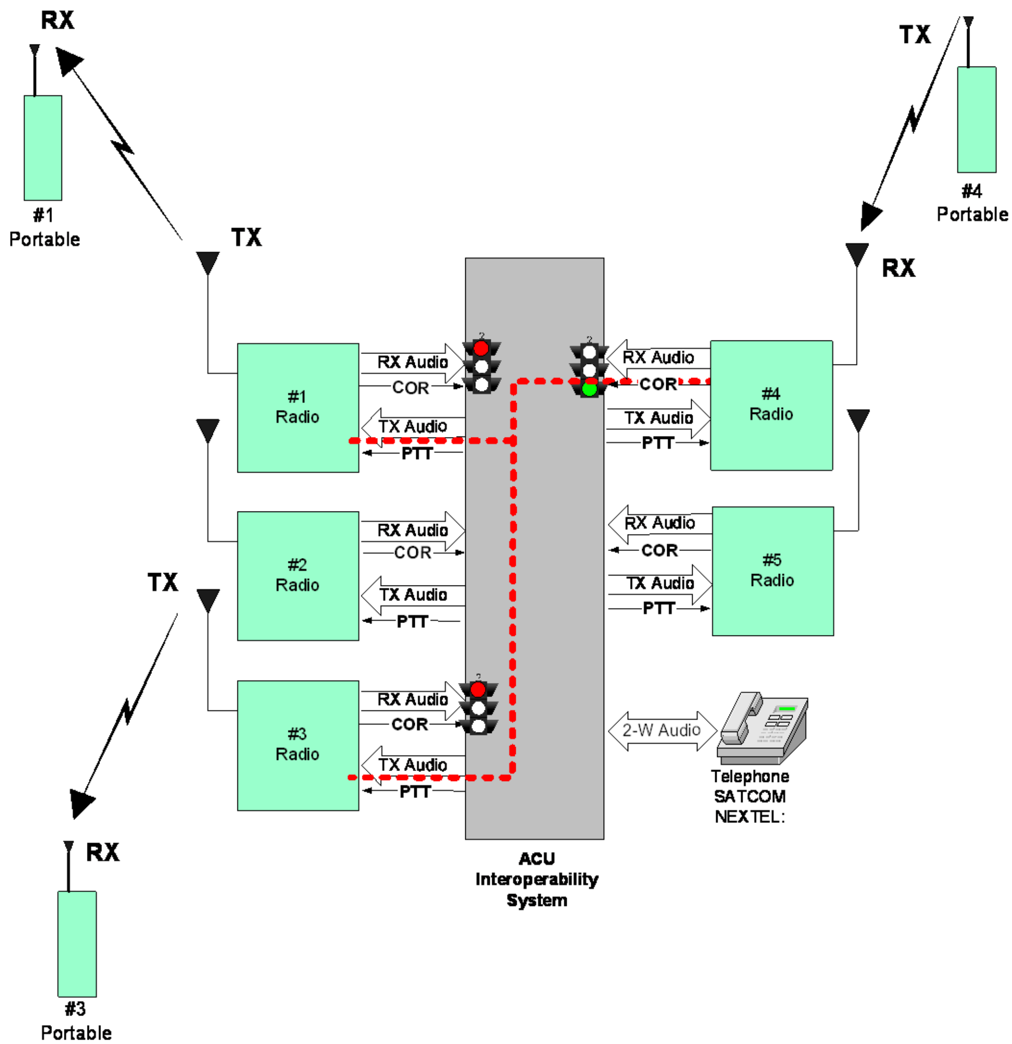
This feature determines whether any system user currently accessing the system will always maintain control until this user stops talking (stops keying his radio) or if another user can also access the system.

PTT (TX) or COR (Dispatch) priority only matter if the radio or other equipment connected to the DSP is half duplex (cannot transmit and receive simultaneously). If a full duplex radio is used, PTT and COR can occur simultaneously, so there is no reason to set to COR priority, and the module should be left in the PTT priority default.

PTT Priority is the factory default setting.

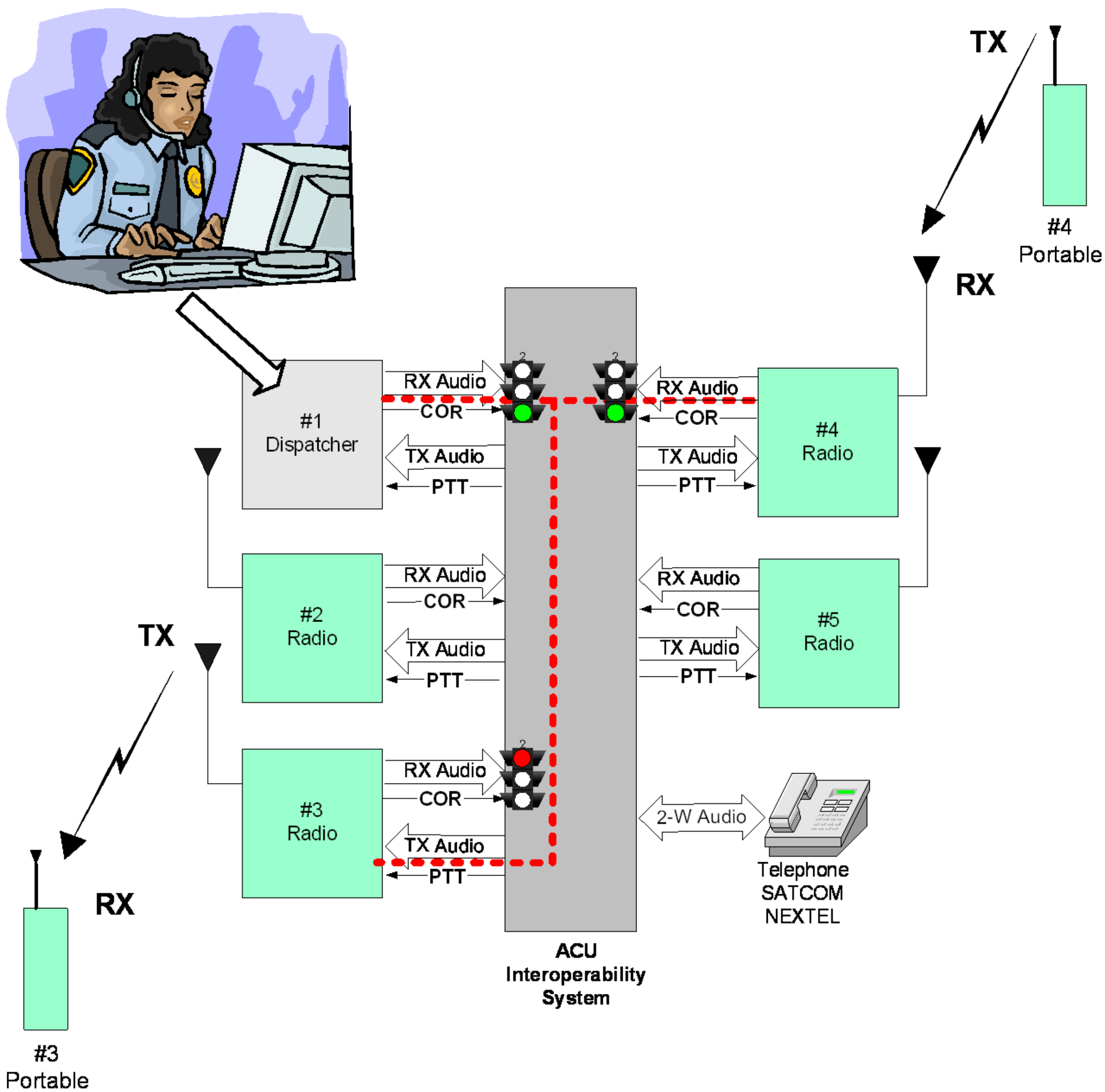
Normally all interfaces are set to PTT Priority. This means that if two or more radios or other 4-wire devices (e.g. a dispatch console) are cross-connected, whoever talks first is in control and no one else can be heard until this person stops talking (and releases the radio PTT).

If the DSP module associated with one user is set to Dispatch (COR) Priority, an unsquelch condition received at the Interoperability System from this console will override the other user's control of the system. The dispatcher's audio will be transmitted instead, or will be mixed with existing incoming audio from any other DSP module also set to Dispatch Priority.



TX Priority

In the figure above, a cross-connection has been made to link radios at extensions #1, #3, and #4. At the moment depicted, a portable of the extension 4 radio system has keyed (TX) first, so its audio is being retransmitted to the ext 1 and 3 radio systems. The stoplights signify that, until the #4 portable unkeys, only the audio from the #4 radio will be allowed. If the #1 or #3 portables transmit before #4 stops transmitting, the active COR signals of the ACU system receivers will be ignored by the ACU.



Dispatch Priority

The figure above shows a dispatch center interfaced to ACU extension #1. This extension has been configured for Dispatch Priority, and any active COR signal at this extension will not be blocked and will instead cause the dispatcher's audio to be retransmitted, even if the #4 portable stays in the TX mode. The dispatcher's audio will be transmitted instead, or will be mixed with existing incoming audio from any other DSP module also set to COR (Dispatch) Priority.

Another scenario when COR (Dispatch) Priority may be helpful is during a cross-connection to a telephone or other VOX-activated device. If there is much background noise at the device, this noise may trip the VOX inappropriately. COR priority at other system extension will allow these users to continue to access the system despite this inappropriate VOX operation.

2.13.3.12 High Frequency Equalizer

The DSP module can reshape the high frequency response of its receive audio input. Equalization can have two effects:

- Improved DTMF detection when using radios with a nonlinear response, and,
- Better-sounding audio for some radios. The high frequency response can be either cut or boosted by up to 5 dB.

A flat frequency response is the factory default setting.

2.13.3.13 DTMF Pre-emphasis

FM radios (VHF, UHF, 800 MHz) use pre-emphasis in the transmitted audio and de-emphasis in the received audio. Pre-emphasis and de-emphasis alter, and then restore, the audio frequency response in order to improve the quality of the received signal with respect to high frequency noise. In most FM transmitters that have built-in DTMF signaling, the DTMF characters are added after the pre-emphasis circuitry. When detected in an FM receiver, the DTMF characters are taken from the discriminator audio, prior to the de-emphasis circuitry.

In the ACU-2000, DTMF detection is performed by the DSP module rather than by the associated receiver. When the DSP receives line audio or speaker audio from an FM receiver (rather than discriminator audio), any received DTMF characters have been inappropriately de-emphasized. This incorrect shaping of the frequency response of the DTMF characters impedes proper DTMF detection. The DSP can add pre-emphasis to the DTMF detection algorithm (leaving the received audio flat) for improved detection.

Note that these settings do not affect the audio that is passed on by the DSP to be cross-connected via another module in the system. The factory default adds pre-emphasis and should be used any time the DSP audio input source is the line audio or speaker audio output of an FM receiver. The no pre-emphasis selection should be used if the audio source is the discriminator output of an FM receiver or from a source other than an FM receiver.

2.13.3.14 Auxiliary Output Control

The standard configuration for DSP is that the AUX-1 output changes logic state when the module is cross-connected. Contact JPS if other configurations are required.

2.13.4 PSTN Configuration Items

See also the Generic Configuration Items that explain Voice Prompt Initiation Delay and settings related to DTMF control of the ACU by system end-users; this includes the module security feature.

2.13.4.1 Telephone Line Level

This configuration item programs the PSTN for different telephone line levels. The selections range from 0 dBm to -24 dBm in 3 dBm steps. It simultaneously sets the telephone send and receive levels. The default setting is -9 dBm, which is the maximum level allowed into U.S. (and most foreign) telephone networks at the subscriber end. Many PABX units require a level of -12 dBm. Higher levels should only be selected for use into field wire or private networks that are known to accommodate higher levels. If the telephone receive audio volume is too low; first use the Telephone Receive Level Boost setting to increase the level.

2.13.4.2 Telephone Receive Level Boost

This configuration item provides additional volume to the PSTN receive input. The Telephone Line Level item explained above sets the correct audio levels for proper hybrid operation and correct levels on the PSTN line. This configuration item is used to boost the PSTN receive audio at the output of the hybrid. When necessary, use this command to increase the volume of audio coming into the PSTN so it matches the volume level of other audio signals in the ACU-2000 system. The factory default of a 6 dB boost works for most systems, but if a different level is required, the options for range from 0 to 12 dB.

2.13.4.3 PSTN Type

This configuration item allows the PSTN to be programmed for either a normal telephone system or a SATCOM terminal. The SATCOM terminal requires a **#** be appended to the entered telephone number. When the PSTN is programmed for SATCOM operation, this is done automatically, so the # need not be entered by the user. The factory default is for a regular telephone line; the only other selection is for SATCOM use.

2.13.4.4 Dial Mode

The PSTN module can use either DTMF or Pulse Dialing. In most systems, DTMF is used, but some older systems may still require the use of Pulse Dialing. When in the Pulse Dialing mode, all digits after the initial telephone number are sent not as pulses, but as DTMF. This allows the use of an answering machine, etc., which require DTMF command input after a connection is made on the pulse dial system.

2.13.4.5 RX Audio Delay

An adjustable delay can be added to the PSTN's input audio. When speech first appears in the input audio, some time passes before it can be detected by the VOX algorithm. The adjustable audio delay prevents the loss of the audio that is received before the detection takes place and audio gates can be opened to send this audio on to other modules.

The factory default is 35 ms.

2.13.4.6 VOX Threshold

This command setting determines the sensitivity of the PSTN VOX. To be sure to avoid missing speech, the factory default setting is Low Threshold, which provides maximum sensitivity. There may be instances where less sensitivity is desired (for example if excessive if background noise is present).

The factory default setting is for Low Threshold.

2.13.4.7 VOX Hang Time

VOX hang time determines how long the VOX stays active after speech disappears. This keeps radios that are communicating with the PSTN from unkeying between words spoken by the telephone caller. If too short a hangtime is set, the radios will unkey frequently and syllables may be missed during the time it takes the transmitter to key again. Too long a hangtime causes the party at the other end to wait unnecessarily long for the VOX to unkey before beginning their response during the conversation.

The factory default is 1 second.

2.13.4.8 2-Wire

The PSTN can only be configured for standard 2-wire hybrid operation.

2.13.4.9 Outgoing Ring Timer

The length of time the PSTN allows the phone being called to ring is set by this configuration item. The selection options are: no ring, a 30 second ring (factory default), a 1-minute ring, or a continuous ring. When set for 30 seconds or one minute, the call will time-out if the call is not answered before the ringing time elapses. If Outgoing Ring is set to *no ring*, no outgoing calls are allowed. This feature is useful, for example, if the PSTN is connected to a satellite terminal and, while it's important to receive incoming calls, the system operator does not want to pay for outgoing calls. If set for continuous ring, the incoming call does not time out and the phone will continue to ring until the call is terminated by the originator.

2.13.4.10 Inactivity Disconnect Timer

The Inactivity Disconnect Timer disconnects a PSTN module from the ACU-2000 system if no activity is detected on the line for a set time duration. This prevents the connection from being tied up if a telephone user forgets to give the *Disconnect* command or the connection is otherwise lost without notification to the system. If there's insufficient activity to trip the VOX before this timer expires, the connection is terminated by the ACU. The timer is reset whenever the VOX is tripped.

Timer selections include: None (the connection will not be terminated due to inactivity no matter how long), 30 seconds, 1 minute, 2 minutes (factory default), 5 minutes, and 10 minutes.

2.13.5 LP Module Configuration Items

See also the Generic Configuration Items that explain Voice Prompt Initiation Delay and settings related to DTMF control of the ACU by system end-users; this includes the module security feature.

2.13.5.1 RX Audio Delay

An adjustable delay can be added to the LP module's input audio. When speech first appears in the input audio, some time passes before the VOX algorithm can detect it. The adjustable audio delay prevents the loss of the audio that is received before the detection takes place and audio gates can be opened to send this audio on to other modules.

2.13.5.2 VOX Threshold

This command setting determines the sensitivity of the LP module VOX function. There are three levels (Low, Med. High) and Off.

2.13.5.3 VOX Hang Time

VOX hang time determines how long the VOX stays active after speech disappears. This keeps the module COR output active during pauses in the speech of the person operating the telephone set. This prevents cross-connected modules from unkeying during these pauses. If too short a hang time is set, the radios will unkey frequently and syllables may be missed during the time it takes the transmitter to rekey. Too long a hang time causes the party at the other end to wait unnecessarily long for the VOX to unkey before beginning their response during the conversation.

2.13.5.4 Dial & Busy Tone Style

The Local Phone Module generates the dial tone and busy tone sent to the phone that is plugged into it. When the handset of the local phone is picked up, this dial tone is heard in the earpiece. After a request to make a connection is made, the user will hear the ring cadence as set in Section 2.13.5.5 while the ACU-2000 attempts to make the connection. If the extension is busy, a busy tone will be heard. The factory default generates the standard US tones, which is currently the only style offered

2.13.5.5 Ring Cadence

The cadence of the ring the LP sends to the associated telephone set is selected by this configuration item. Two styles are offered; the standard US style (factory default), and European style. This ring cadence applies both to the tone heard in the earpiece of the telephone set cabled to the LP module when a call is being made and to the ring that is produced and sent to the caller when a call is placed to the local phone (ringback).

2.13.5.6 Dial Tone Enable

The LP can generate a dial tone whenever a caller picks up the handset of the associated telephone set in order to place a call (that is, make a ACU system connection). If desired, this dial tone can be disabled.

2.13.5.7 Ringback Enable

When a call is placed to the extension occupied by the LP module, the LP can signal this call has been made via its ringback signal. For example, if an ACU-2000 user attempts to make a connection with his UHF radio to a system extension occupied by an LP module, he will hear this ringback signal as he waits for the handset at the LP module to be picked up. It's possible to disable this ringback signal; in this case the UHF radio user will not hear anything as he waits for his call to go through.

2.13.5.8 Outgoing Ring Timer

The length of time the LP module causes its associated phone set to ring when a call is received is adjusted by this configuration item. The selection options are: no ring, a 30 second ring (factory default), and a 1-minute ring; or alternatively a continuous ring. When set for either 30 seconds or one minute, the call will time-out if the call is not answered before the Ringing Time elapses. When set to *no ring,* no outgoing calls are allowed. If set for continuous ring, the incoming call does not time out and the phone will continue to ring until the call is answered or terminated by the originator.

2.13.5.9 Aux Output Control

The standard configuration for the LP module's AUX-1 output is to change logic state when cross connected. Contact JPS if other configurations are required.

2.13.6 SCM-2 Module Configuration Items

See also the Generic Configuration Items that explain Voice Prompt Initiation Delay and settings related to DTMF control of the ACU by system end users; this includes the module security feature.

The following configuration items, except for Connection Type, can be adjusted via the ACU Controller. Squelch Type, VOX Threshold, VOX Hang Time, and Connection Type can be adjusted via the HSP keypad. SIP network settings can be set by browsing to the SCM-2 module's IP address.

2.13.6.1 Squelch Type

This setting (also called Network COR Type) tells the SCM-2 how to determine when there is a signal coming from the SIP network that will ultimately activate a transmitter of a radio associated with a DSP module that's cross-connected with the SCM-2. 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 next three settings may be useful if the SCM-2 will be linked over the network to other JPS SIP interfaces (ARA-1 units, SCM-1 modules, or other SCM-2 modules). These COR types make use of a RTP extension header that includes the unit's COR status. This is helpful as these other units will receive a positive indication of COR status that arrives coincident with the radio audio and they 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 units should key their associated transmitters. For these settings to have any utility on the unit being set up, the other SIP interface that the SCM-2 may link to must have its SIP Settings option *Send Radio COR/AUX Status* enabled.

To further clarify: When Radio A is linked to Radio B over a SIP network via a pair of JPS SIP interface devices, 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 remote SIP interface 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 SCM-2 associated with Radio B can make use of this information only if its network COR Type is set to one of the following:

- RTP
- RTP + VOX
- RTP + VMR

Use the *RTP* setting if the all end-devices on the system are connected via JPS SIP interface devices. When the network COR Type is set to RTP, 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 SCM-2 will ignore network audio from other devices not interfaced by a SIP interface that has its *Send Radio COR/AUX Status* function enabled (for example, from a standard SIP Phone). If these other devices will be used, set the SCM-2 Network COR Type to either *RTP + VOX* or *RTP + VMR.*

Both *RTP + VOX* and *RTP + VMR* make use of the COR status information from linked JPS SIP interface devices but will also properly link with non-JPS SIP devices. When either of these modes is selected, the SCM-2 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 to select 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 some other SIP device may be using this header for some other purpose. Two conditions may result:***
>
> ***(1) An incompatible SIP device misinterprets the COR/AUX extension header sent from the SCM-2. 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 SCM-2. This may cause the SCM-2 to inappropriately detect active COR.***

Use the following guidelines when deciding how best to apply the use of the RTP Header:

- When all end devices are radios interfaced by JPS SIP devices that support this COR/AUX transfer extension, use RTP as the Network COR type in all devices and enable the *Send Radio COR/AUX Status* SIP setting in all devices.
- If the network consists of mixed devices (multiple JPS SIP devices, as well other SIP devices, such as SIP Phones or softphones), and these devices have no incompatibilities with SIP extension headers, use either *RTP + COR* or *RTP + VMR* as the Network COR type in all JPS devices and enable the *Send Radio COR/AUX Status* SIP setting.
- When the network consists of mixed devices (multiple JPS SIP devices, 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 JPS SIP devices and also disable the *Send Radio COR/AUX Status* SIP setting.

The next network COR type option is *Packet*. When *Packet* is selected, the SCM-2 will detect active COR based upon the presence or absence of audio RTP packets. When the SCM-2 is receiving audio RTP packets, it will report active COR. When it stops receiving audio RTP packets, it will report no active COR. This feature is often used in conjunction with silence suppression on the remote device.

When *Packet* COR is enabled, the *VOX Hang Time* setting described below can be used to adjust how long the SCM-2 keeps COR 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.

The final setting option—*Disable*—completely disables the detection of network COR. This can be useful in situations where one intends only to monitor a radios receive signal.

2.13.6.1.1 VOX Threshold

This setting specifies what level the audio coming into the SCM-2 from the network must attain before it is considered valid and, therefore, will cause the module to signal any connected radio to transmit this audio. If you are using VOX or VMR and the associated transmitter is not activating reliably you may need to increase the sensitivity. If a cross-connected transmitter is being activated by background noise you may need to decrease the sensitivity.

2.13.6.1.2 VOX Hang Time

When using VOX as the Network 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 hang time to the lowest level that does not create inappropriate resquelching. Hangtime is also required for VMR, as again there are pauses in speech, but also there are parts of speech that cannot be differentiated from non-speech.

The SCM-2 front panel *COR* LED is lit whenever the module’s Network VOX or VMR has detected the presence of valid audio signals from the IP network. If this LED flashes during pauses in speech from the radio, the hangtime must be increased. For the most natural conversation, do not set any longer than necessary to remove the drop outs.

2.13.6.2 Auto Answer

When Auto Answer is set to on, the ACU responds to incoming calls (SIP session requests) by sending a voice prompt to the SCM-2 caller. The prompt asks the caller to send a DTMF sequence that instructs the ACU to create a cross-connection between the SCM-2 module and another module.

If operation by remote DTMF is desired, be sure that DTMF Enable is also set to Enable for the SCM-2, or the caller's DTMF input will be ignored. System operation via DTMF is explained in Section 3.6.

When Auto Answer is set to off, the ACU will not send any voice prompts to the caller, but instead the associated SCM-2 module icon will ring on an ACU Controller interfaced to the ACU. If there is no ACU Controller interfaced to the unit, incoming SIP session requests will be ignored.

2.13.6.3 Connection Type

The *Connection Type* setting determines how the SCM-2 will respond after answering an incoming SIP call or when a backplane connection is made with another module. Four modes are available:

- Mode 0 (Normal): In this mode, the SCM-2 behaves like a PSTN module. When a backplane connection is made to the SCM-2, the ACU user is prompted to enter a phone number for the SCM-2 to call. When a SIP call is received, the SIP caller is prompted to enter the extension of a module to connect to. In this mode, backplane connections and SIP calls are coupled and cannot exist without each other. When one is disconnected, the other is also disconnected.

- Mode 1 (SIP Direct): In this mode, the ACU user is never prompted to enter a phone number when a backplane connection is made to the SCM-2. Instead, the ACU user is simply connected to the SCM-2, and the ACU assumes that the SIP connection takes care of itself. Incoming SIP callers are still prompted to enter an extension, however.

- Mode 2 (Backplane Direct): In this mode, when a call is received, the SIP caller is never prompted to enter an extension. Instead, the SIP caller simply hears *Connected* when the SCM-2 answers. The SCM-2 assumes that the backplane connection is made by something or someone other than the SIP caller. In this mode, all DTMF commands received from the SIP caller by the SCM-2 are ignored. When a backplane connection is made to the SCM-2, the ACU user is still prompted to enter a phone number, however.

- Mode 3 (Both): This mode is a combination of the SIP Direct and Backplane Direct modes. Neither ACU users nor SIP callers are ever prompted. In this mode, backplane connections and SIP calls are completely decoupled.

See Section 5 for instructions on how to program the SCM-2's connection mode via the HSP keypad.

2.14 Local Radio Interface & Optimization Procedures

There are three major steps to interfacing radios and other four-wire devices to a DSP module:

- Connect the communications devices to ACU-2000 rear panel D15 connectors using properly designed interface cables (either purchased from JPS or designed/built by the customer).
- Optimize each individual interface/ACU extension.
- Optimize the operation of each extension when cross-connected with the other extensions in the system.

This section explains how best to perform these steps. Acceptable communications links are possible without following all of the steps described, but following the procedures outlined will provide seamless communications in a wide variety of difficult communications environments. These procedures may appear more complicated than those required for less sophisticated interoperability devices. This is largely because this other equipment does not provide many of the features that the ACU-2000 employs to resolve complex interoperability problems.

JPS has simplified the interface process by:

- Proper design of interface cables
- Coordination of the use of these cables with the ACU Controller software
- Applications Notes describing the peculiarities associated with individual communications devices
- Use of templates to load settings that have been optimized for each individual device
- Flow charts to guide the installer through the final optimization steps

All of these are explained in the following sub-sections, using the Universal Unterminated Cable Application Notes as a guideline.

2.14.1 Interface Cables for Radios and Other Communications Devices

The systems engineers at JPS Interoperability Solutions have created interface cables and accompanying Application Notes for a large number of radios and other communications devices. These notes are posted on the JPS Website and available by email at:

http://www.jpsinterop.com / support@jpsinterop.com

2.14.2 Radio Device Set Up Using Applications Notes and Radio Templates

The details of interfacing a radio to the ACU-2000 are contained in Application Notes that are specific for each radio or other device. These notes are posted on the JPS Interoperability Solutions Website and available by email at:

http://www.jpsinterop.com / support@jpsinterop.com

Newly created notes may be found there. These notes are also available via the ACU Controller configuration page. In addition to radio specific cable Application Notes, there is also a generic cable Application Note that helps an installer design a radio cables and optimize the interface.

Note: The cable designed for any particular radio will work with the 1-pin D-sub interface connector any of these JPS Interoperability Devices: ACU-2000, ACU-1000, ACU-M, NXU-2A, ARA-1, and ACU-T. The ACU-T requires a short adapter cable.

The following sub-sections give additional information to further explain the interfacing information presented in the Application Notes documents.

2.14.2.1 Drawing Number and Revision:

Important entries in the document header include the Drawing Number and the Revision. The radio model number is not used as the drawing name because some Application Notes apply to several different radio models. The drawing number corresponds to the P/N of the interface cable. Both the P/N and revision are marked on each cable. It's important that the revision number of Application Notes document matches that of the cable itself.

The revision letters and numbers denote the following:

Letter change (A to B or B to C): A change has been made that will materially affect the use of the Application Notes by the customer. Example: Change in attenuation network or change in recommended DSP module programming settings.

Number change (B to B1 or B1 to B2) Denotes a documentation-only change has been made that will not affect use of the Application Notes by the customer. Example: Correcting a misspelling or adding to (but not materially changing) an explanation.

The cable information on the JPS Website lists the current revisions of the Application Notes and the cable drawings. If you are in possession of an Application Note of a previous revision, download the newer revision if there has been a Revision Letter change; this isn't necessary if only the number has changed or appended.

2.14.2.2 APPLIES TO:

Make sure that the Applications Note is the correct one for the radio being interfaced. If you are not sure, recheck the cable list on the JPS web site or contact JPS Customer Service. If a cable does not exist for a particular radio, then either use the universal cable to make a cable, or request that a new cable be made by contacting JPS Customer Service.

2.14.2.3 RADIO MODIFICATIONS:

Sometimes it's necessary to make a physical modification to the radio. Usually this is required because needed signals (such as a PTT input or a TX Audio input (MIC Audio) are not available without the change. Sometimes these signals can be made accessible through radio programming changes rather than hardware modifications. The Application Notes always endeavor to provide the easiest possible interface method.

2.14.2.4 RADIO PROGRAMMING:

Al necessary or recommended changes to the radio programming are listed.

JPS recommends that radios interfaced to the ACU-2000 rear panel be set for low transmit power. The reason is that many Local Interoperability Systems have a number of radios operating in close proximity, this increases the chance of radio front-end desensitization and other problems likely a complex RF environment. Higher power exacerbates these problems. This is particularly true of Tactical, Transportable, and Mobile Applications where quick set up or a quickly changing RF environment is likely. If careful attention is paid to antenna placement, the TX and RX frequencies used, as well as proper grounding and shielding practices, it's certainly possible to operate radios at a higher power setting, but stay aware of the potential for problems.

Radio Programming functions that may be required are:

- Make signals accessible at the radio I/O (PTT, Mic In, COR, etc.)
- Set Channel Frequency
- Set power output for lowest reliable level
- Set CTCSS or DCS
- Squelch/COR pinout
- COR output: Active Hi or Active Low
- Program Squelch/COR output for carrier plus tone
- Set Minimum audio level
- Disable any off-hook setting
- Enable cable ignition sense

2.14.2.5 RADIO CONTROLS:

Any necessary or recommended radio control adjustments are provided. Many radios do not have a constant volume Line Out, and the RX audio provided to the ACU-2000 varies with the volume setting. If so, the proper setting is listed.

2.14.2.6 CABLE CONNECTIONS:

The various cables covered by the Application Notes are listed, as well as the type of RF connector found on the associated radio.

2.14.2.7 DSP JUMPERS:

The proper DSP module audio input configuration jumper settings are listed. See 2.11.4 for more information regarding the DSP module audio input configuration.

2.14.2.8 RADIO INTERFACE OPTIMIZATION:

This section lists all of the relevant DSP module options settings. These are the same settings that are automatically configured when the corresponding radio template is loaded via the ACU Controller. Note that the DSP module has a variety of features that are either not directly related to the interface cabling or are dependent on the particular system. Some of these features are covered by the Individual Extension Optimization Procedures explained in Section 2.14.4.

The settings listed in the Application Notes are optimum starting points based on actual radio tests performed by JPS Systems Engineering. Because of radio-to-radio variations and different operational situations, these are starting points only. Be sure to follow through with the optimization procedures that follow. The ACU Controller and WAIS Controller also allow on-the-fly changes of an individual radio interface when dynamic communications environments require it.

For Example:

- Trunked system channel acquisition delays increase dramatically due to stepped up system usage.

- A poorly-performing radio in the field must be communicated with, necessitating a change in RX audio levels or the introduction of Noise Reduction.

- Providing Dispatch Priority to an important system user.

2.14.2.9 NOTES:

Any special notes related to this radio are provided, as well as any setup hints pertaining to some likely operational scenarios.

2.14.3 Request for Creation of New Cable Designs

If a customer would like to request that JPS Interoperability Solutions create a new cable design, below is a list of the basic steps in the process.

- Contact Customer Service (919) 790-1011 with a request for a new cable design.
- Be prepared to furnish the target radio and one other radio (portable or mobile) that can communicate with it.
- The target radios must be set up and tuned so that they communicate out of the box.

- All accessories, a full service manual, programming instructions, and a programming box with proper cables must be supplied.
- If target radio is a satellite phone, the phone must be commissioned, tested, and ready for use.
- Type of cable required (ACU-2000, ACU-1000, ACU-T, or Radio Tray) and application.
- Permission to open the radio to modify or program as required.
- Technical customer contact person with phone numbers.
- List of materials and equipment supplied.
- Contact Customer Service at (919) 790-1011 for an RMA# prior to shipping any material or radios.
- JPS Interoperability Solutions will do the following:
    - Design the interface cable
    - Determine any radio and programming changes.
    - Optimize the Interface module setup.
    - Draw a cable schematic.
    - Prepare a detailed Application Note that describes the optimal settings.
    - Add the new cable and Application Note to the Interface Cable Database.

2.14.4 Optimization of Individual ACU-2000 Extensions

2.14.4.1 Connections to DSP Module D15 Connectors

Once the radio, satellite phone, or other communications device is connected to the ACU-2000 rear panel using the proper interface cable, the interface must be optimized to fit the individual communications device (usually a radio) and the circumstances of its use. A flow chart is provided to assist with this task. This flowchart assumes that the ACU Controller is being used to adjust the DSP module configuration, but the same steps may be performed using the WAIS Controller or the HSP keypad.

The detailed notes that follow each flow chart provide additional explanations when necessary. If the installer has had some experience setting up an ACU-2000 interface it may prove sufficient to simply follow the flow chart. The notes are tagged by the number of each question posed in the flow chart (e.g. QA3 is the third question in flow chart A, and *QA3* is noted at the top of the question diamond).

Flow Chart A is the typical Overall Setup procedure and should apply in most circumstances. Charts B & C are used only these conditions exist:

- Use Flow Chart B only if Application Notes are not available

- Use Flow Chart C only if the radio being interfaced is part of a trunked system

Flow Chart B guides the installer through a simplified version of the steps taken by JPS Systems Engineering when a new radio interface cable is designed.

Flow Chart C provides a systematic procedure for determining the proper amount of TX Audio Delay to compensate for the Trunked System's channel acquisition delay.

After an individual extension has been optimized, that port's operation with the other parts of the ACU-2000 system must be checked and optimized if necessary.

Following the flow charts and explanations is a description of a procedure for operational testing and final adjustment. See Section 2.14.4.5.

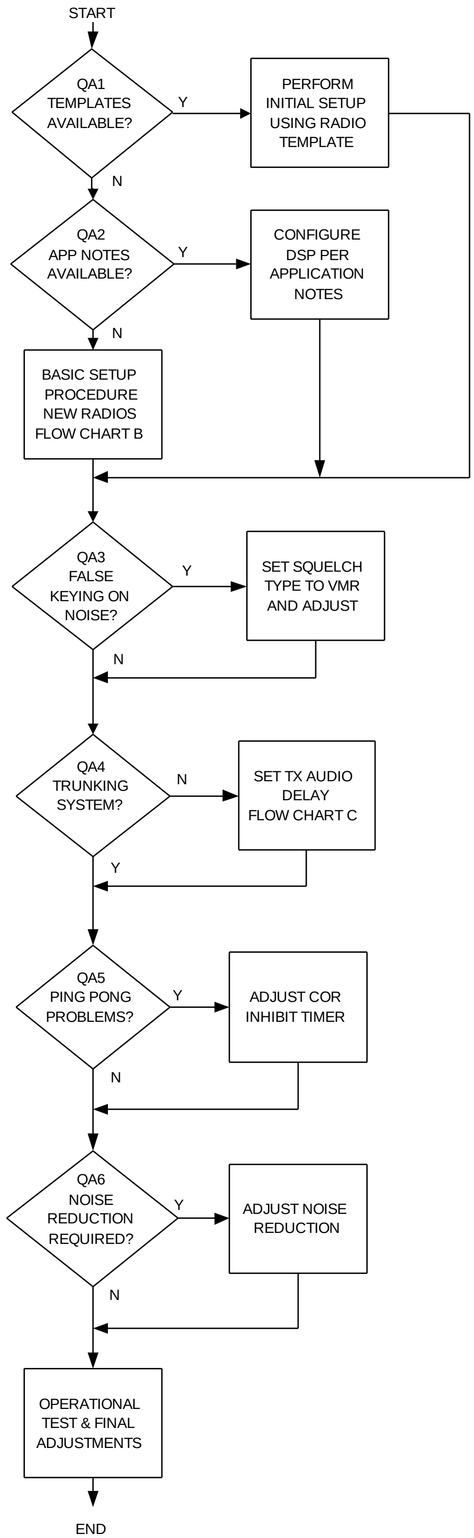


Figure 2-11 Setup Flowchart A, Overall Instructions

2.14.4.2 Flow Chart A Details – DSP Module Setup Procedure

QA1 Are Templates Available?

To determine if there is a template for the radio model being interfaced, double-click on any DSP module icon to bring up its setting screen. Next, click on the *Load* button to bring up the list of radio templates.

(QA1 Yes) Initial Set Up Using Radio Template

> ***If you are not sure if the radio and cable are configured per the JPS Application Notes, do not click on the Template file to load it. Cancel and click on the Open App Notes button to pull up the proper Application Notes. If the provisions of the Application Notes can't be followed, jump ahead to Flow Chart B.***

- Click on the template file that has the same name as the model of radio being interfaced. All of the DSP settings will change to those listed on the Application Notes. The Last Loaded / Saved field will list the name of the Template File.

Jump to QA3.

(QA1 No) Jump to QA2

QA2 Are Application Notes Available?

Application Notes are provided with radio interface cables purchased from JPS or may be obtained from the Customer Service Department. If you do not currently have the proper radio template file, but do have Application Notes, proceed to step QA2 Yes. Otherwise jump to flowchart B

(QA2 Yes) Application Notes but no Template File

- Ensure that the radio and cable are configured per the Application Notes.
- Set the DSP module to its default settings:
- If each ACU Controller setting is marked [default], no action is necessary. Otherwise, use the ACU Controller to set all to the Default position. Alternatively, all settings can be quickly set to their default settings by loading the DSP Default Template.
- Use the ACU Controller to change all DSP settings to those listed on the Application Notes.
- Click *Apply* to save the settings and stay in the Settings Mode.

Jump to Q3

(QA2 No) Jump to Flow Chart B

QA3 False Keying on Noise?

If the radio Squelch Type is either COR or VOX, and the channel is noisy, the radio may unsquelch inappropriately due to this RF noise. When the radio is cross-connected to another radio via the ACU-2000 or ACU-T, the cross-connected radio will transmit every time the noisy radio unsquelches.

If a radio has a tendency to key on noise, change the Squelch Type to VMR (Voice Modulation Recognition). The DSP Module will unsquelch only when human speech is detected in the receive signal.

[Inappropriate unsquelch of the radio can’t be resolved by changing the VOX Threshold of the DSP Module]

(QA3 Yes) Set Squelch Type To VMR Mode

- On the DSP Module Settings Screen, select VMR from the Squelch Type options.
- Next determine the proper threshold. Listen while the radio receives a speech signal. The default setting is Med1. If the radio does not break squelch for all received speech, the threshold is too high; adjust to Low. If the radio breaks squelch on all speech signals and also on some noise input, increase the threshold to the Med 2 setting, and if necessary to High. (Note, for extremely noisy signals it may not be possible to find a threshold setting that will unsquelch for all speech signals and also always stay squelched during periods of high noise).
- Now adjust the hang time. The intent of VMR hang time is to keep the system unsquelched during pauses in speech. The default (and minimum) setting is 775 milliseconds. If the radio squelches inappropriately during the reception of speech, raise the hang time in one step increments until proper operation is reached.
- Click *Apply* to save the settings.

Jump to QA4

(QA3 No) Jump to QA4

QA4 Is The Radio Part of a Trunking System?

Trunked radio systems allow efficient use of multiple channel systems. When a user requests access, the system automatically switches the user’s radio to a free (unused) channel. Trunked systems users, when keying their radios, must wait for a tone that signals that a free channel has been acquired before beginning a conversation, while conventional (non-trunked) system users can begin talking as soon as they key their radios.

The Channel Acquired Tone that signals trunked radio users that they may begin speaking is not available to system users on cross-connected radios, so the trunked radio’s TX audio must be delayed, following assertion of PTT, until the normal channel acquisition time has passed.

(QA4 Yes) Jump to Flow Chart C

(QA4 No) Jump to QA5

QA5 Ping-Pong Problems?

Some radios have a tendency to unsquelch momentarily at the end of each transmission. In an ACU system, when two radios are cross-connected, whenever one radio is unsquelched, the other is keyed. If a cross-connected radio exhibits the momentary unsquelch after TX behavior, the cross-connected radio will inappropriately transmit. If both radios unsquelch at the end of each transmission, the system will "ping-pong", with first one radio keyed momentarily and then the other. This effect can be experienced with the DSP module set to either the COR or VOX Squelch Types.

To ensure that a radio being interfaced will not create a ping-pong behavior, key the radio and check for signs of a momentary unsquelch at the end of the transmission. This can be done by cross-connecting the HSP module with the radio. Key the HSP handset and see if the HSP's PTT LED comes on momentarily after the transmission is ended. If the PTT LED comes on, the ACU should be set to ignore this inappropriate COR signal by enabling the COR Inhibit after PTT feature.

> ***Note: While the COR Inhibit Timer will prevent ping-pong and inappropriate keying of cross-connected radios; it will also prevent the radio from receiving a legitimate signal until the timer expires. This is not usually a problem because the radio is normally producing a burst of noise during this time. If system performance with the COR Inhibit Timer enabled is unsatisfactory, switch to VMR mode.***
>
> ***See (QA3 Yes) Set Squelch Type to VMR Mode.***

(QA5 Yes) Adjust COR Inhibit Timer

- The default setting for the COR Inhibit Time after PTT is 100 ms. If this does not prevent the DSP front panel COR indicator from lighting momentarily at the end of a transmission, set to the next highest time and repeat the test.

> ***Note: The Signal LED may still light.***

- Continue to raise the time setting one step higher than the time required to prevent the inappropriate COR indication.
- It may be necessary to check system performance with two radios cross-connected (rather than using the HSP) to ensure the optimum COR Inhibit Timer Setting.
- Click *Apply* to save the setting.

(QA5 No) Jump to QA6

QA6 Is Noise Reduction Required? [Affects RX Audio Only]

The DSP module has a Digital Noise Reduction Mode that can be used to clean up noisy received signal input. This affects noise that is mixed with the speech signals, not RF noise that unsquelches a radio, creating a loud noise burst. The only method to find the correct amount of Noise Reduction to apply is to listen to the received signal as the level is changed; this is best done using the HSP Handset so that you can be sure that all noise heard is from the radio's received signal. Do not use the HSP speaker or a cross-connected radio. A little Noise Reduction goes a long way, and too much will give the received signal a fuzzy, artificial sound. It may be advantageous to attempt to improve the signal quality by other means (such as improving antenna placement) before adding Noise Reduction.

(QA6 Yes) Adjust Noise Reduction

- The default setting for Noise Reduction is Off (no reduction). While listening to the received signal, increase the Noise Reduction setting one step at a time until the best signal quality is reached.
- If possible, listen to the receive signal from several different sources and determine the Noise Reduction setting that works for most.
- If the signal quality is later improved, revisit the Noise Reduction setting.
- Click *Apply* to save the setting.

Jump to *Operational Test & Final Adjustments*.

(QA6 No) Jump to *Operational Test & Final Adjustments*

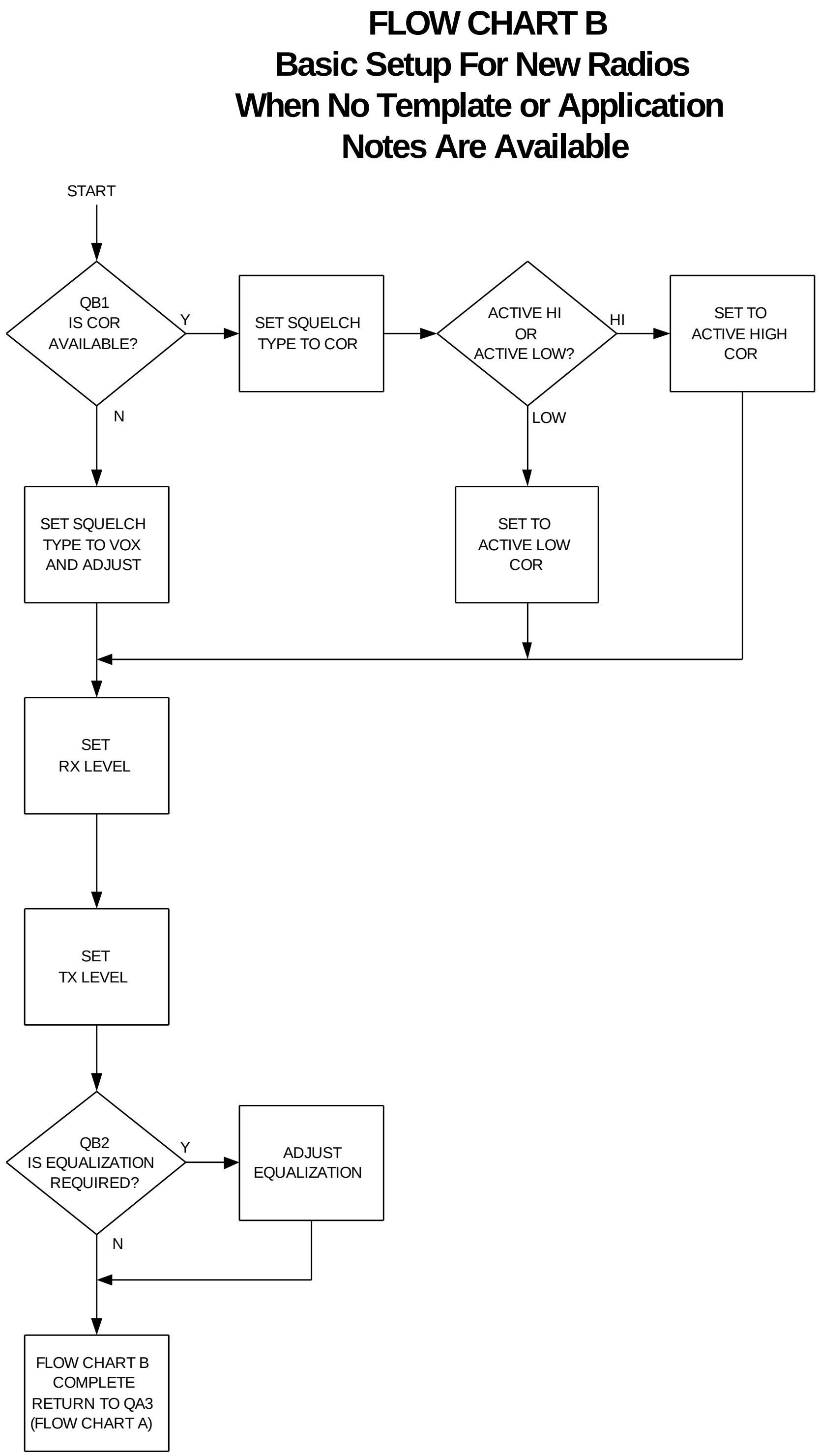


Figure 2-12 Setup Flowchart B, Basic Initial Setup

2.14.4.3 Flow Chart B Details – Basic Setup

DSP Setup With No Templates & No Application Notes

QB1 Is a COR Signal Available?

The default Squelch Type is VOX because it will function with all radios. However, if a COR (unsquelched) signal line is available, this is usually a better choice. Determine if the radio has an output that will go either high when the radio unsquelches (Active High COR), or low (Active Low COR). It may be necessary to change the radio programming to enable the COR output signal.

(QB1 Yes) Set COR Squelch Type & COR Polarity

- On the DSP Settings Screen, change the Squelch Type To COR.
- The default COR Polarity is Active Low. If the COR signal is actually Active High, change the COR Polarity setting to Active High.

> ***Note: A radio with active high COR, when connected to a DSP Module set for Active Low, will keep the DSP's COR LED on except when the radio unsquelches.***

- Click *Apply* to save the settings.

Jump to Set RX Level

(QB1 No) Jump to Set RX Level

Set RX Level

The RX (Receive) level must be optimized to allow best system operation. First of all, conversations, especially conference calls, will be more intelligible if all voices are at the same volume level. Second, VOX and VMR work best at the proper RX level.

- Monitor the front panel of the DSP module while the radio is receiving a voice signal at a normal speaking volume level.
- Watch the DSP front panel SIGNAL light. It should flicker with the incoming speech. If the level is too high, the LED will be on constantly during received speech. If too low, the LED will never come on, or will flicker only occasionally.
- Adjust the RX Level until the Signal LED flickers with incoming speech.
- Click *Apply* to save the setting.

Note: If the interface is using speaker audio from the radio, the level will vary depending on the radio's volume control setting. Set the RX level in the DSP to 0 dBm, and then vary the radio volume level until the proper Signal LED indication is achieved. Note the setting, and keep the volume control at this setting.

Set TX Level

The proper TX level is required to fully modulate the transmitter, but not over modulate it. Most radios have an audio limiter prior to the transmitter to prevent over modulation. Even with the limiter, some radios will still over modulate and some even shut off the TX signal when the input is too high. When the level is set too low the audio of the radio receiving the signal will be lower than normal, requiring that its volume control be turned up to an abnormal position. When the audio is too hot the audio will sound squashed or forced, and if the radio does not have a TX audio limiter the audio will sound distorted and over modulated.

- Cross-connect the HSP Module to the DSP Module being adjusted, and use the HSP Handset to key the radio while speaking at a normal volume level.
- Monitor to the TX audio of the interfaced radio on a receiver set to the radio's TX frequency.
- The quickest way to set the TX audio level is to use the ACU Controller to set the DSP Module's TX level to its lowest setting. Increase the TX level until the audio in the monitoring radio stops increasing in level. This is the threshold point where the limiter is preventing the TX level from going any higher. Leave the DSP Module's TX level at this threshold value.
- You may also follow the radio's recommended TX input audio setting procedure.
- Click *Apply* to save the TX Level setting.

QB2 Is Equalization Required?

High Frequency Equalization either boosts or rolls off the high end of the RX audio spectrum. This adjustment compensates for poor RX audio quality. The best way to determine the proper High Frequency Equalization Setting is to listen to the received audio in the HSP handset (not the HSP speaker, unless a high-quality external speaker is connected).

(QB2 Yes) Adjust Equalization

- Monitor the RX Audio in the HSP handset.
- If the audio sounds like it is lacking treble, the high frequencies can be increased (boost).
- If the signal sounds too bright or harsh, the high frequencies can be attenuated (cut).
- When the best-sounding audio is attained, click *Apply* to save the setting.

Note: Most Motorola mobiles and portables sound best with a boost of at least 3.5 dB.

(QB2 No) Flow Chart B Complete - Jump to QA3

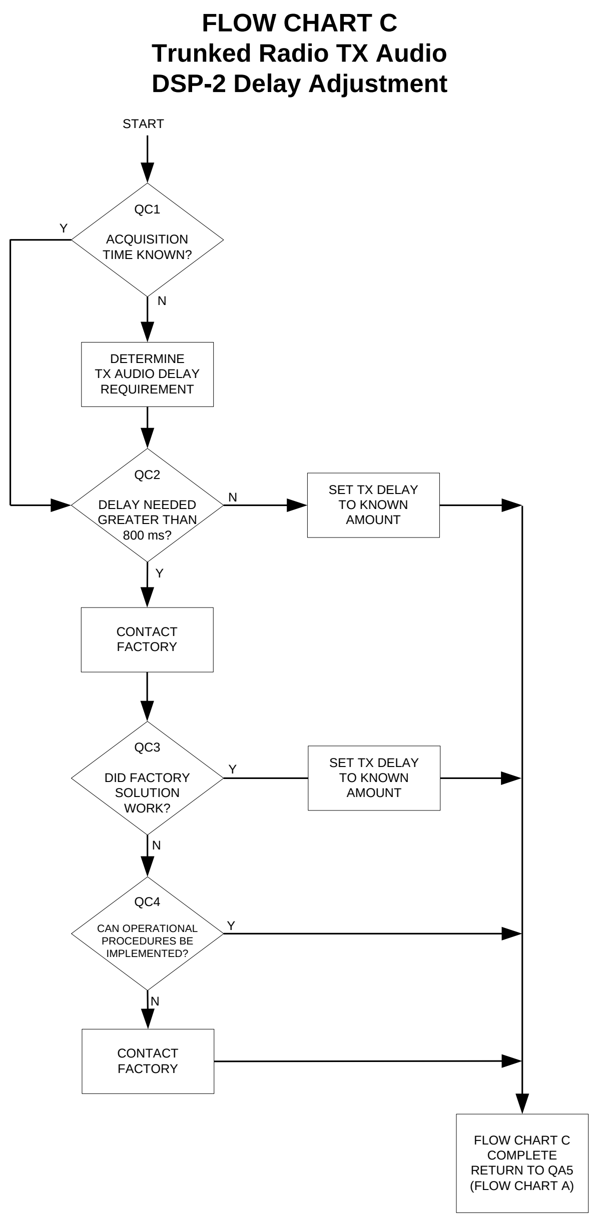


Figure 2-13 Setup Flowchart C, Trunked System Audio Delay Adjustment

2.14.4.4 Flow Chart C Details - Trunked Radio TX Audio Delay Adjustment Procedure

QC1 Is Acquisition Time Known?

Trunked Systems require TX audio delay that matches the normal channel acquisition time. This delay holds up the RX audio from cross-connected radios until the trunked radio is ready to begin transmitting. If the Channel Acquisition Time is known, the task is simply to add this amount of TX audio delay. If it is not known, a procedure must be followed to determine the proper TX Audio Delay Setting.

(QC1 No) Determine TX Audio Delay Requirement

- Set the TX Audio Delay to its maximum setting of 800 ms.
- Cross-connect the HSP Module to the DSP associated with the trunked radio.
- Transmit via the HSP while monitoring the signal via a receiver that is part of the trunked system.
- If the first syllable of the transmissions is being cut, more than 800 ms delay is needed. Contact Systems and Applications Engineering at the JPS factory for recommendations.
- If the first syllable is not being cut, lower the delay setting until it is, then raise the setting up one step.
- Click *Apply* to save the TX Audio Delay setting.

Jump to QA5. Flow Chart C Complete.

> ***The TX Audio Delay Setting may be a compromise between comfortable conversation and safety. When it is imperative that absolutely no first syllables are ever lost, the TX audio delay must be at least as long as the longest channel acquisition time (normally longest during times of heaviest system use). This may result in longer than needed delays during periods of low or normal system traffic. If not sure, leave at max TX Audio Delay.***

(QC1 Yes) Jump to QC2

QC2 More Than 800 ms Delay Needed?

The DSP module can provide up to 800ms TX audio delay. Contact JPS for recommendations if delay requirement exceeds 800 ms.

(QC2 No) Set Known Delay

- Set the TX Audio Delay to the setting equal to (or just above) the known channel acquisition time.
- Click on *Apply* to save the TX Audio Delay setting.

Jump to QA5. Flow Chart C Complete.

(QC2 Yes) Contact JPS for recommendations.

QC3 Did Factory Solution Work?

(QC3 Yes) Implement Factory Solution Recommendation

- Set TX Audio Delay

Jump to QA5. Flow Chart C is Complete.

(QC3 No) Jump to QC4

QC4 Can Operational Procedures be Implemented?

(QC4 Yes) Implement Operational Procedures

When long channel acquisition times cannot be completely compensated for by adding TX Audio Delay, system users must follow proper procedures to ensure that the entire message is being heard. This can mean simply waiting for the necessary time after keying their microphones before beginning to speak, repeating important communications, or getting a response from the listener. Alternatively, there may be adjustments or modifications that can be made to the trunked radio system to decrease the channel acquisition times. Contact the system supplier to inquire about reducing acquisition times.

(QC4 No)

Contact JPS for further recommendations.

Flow Chart C Complete - Jump to QA5

2.14.4.5 Operational Test and Final Adjustment

This procedure accounts for radio-to-radio variations and verifies proper cross-connection operation of the system. It will be necessary to create separate cross-connections of the radio being interfaced with every other radio in the ACU-2000 system, and perform each of the following checks. The associated DSP settings can be varied during the operational test to check for the optimum level. Use the ACU Controller to move the setting up or down by one level while communicating to check whether overall operation is degraded or improved, except for correcting TX & RX levels. If these settings need modification, jump back to Flowchart B and QB1 and perform the entire settings procedure.

If necessary, refer to the flow charts for more information regarding these quick checks.

- Verify proper TX and RX levels. If problems are noted jump back to QB1 of the flow chart. Do not modify individual module levels, or you may raise an RX level when what is really needed is a lower TX level.
- Check for ping-pong problems.
- Listen for noise problems.
- Listen to RX signal quality to determine if equalization is required.
- Verify that the proper Squelch Type has been set.
- For Trunked radios, listen for audio delay issues (missed first syllables or too much delay before first syllable is heard).
- Repeat the above steps as necessary.

When completed, there are two options:

If a template already exists for the radio being interfaced, click *OK* to save the settings and exit the DSP settings screen.

If you are interfacing a radio that does not already have a stored template and you want to create one to apply to other radios in the system (or to save these exact settings for later recall), click *Save* and store the settings under the Template name of your choice.

2.15 PSTN Simplified Setup Procedure

These instructions assist in the basic setup of a PSTN module to the phone system it's connected to. This step-by-step process lists the steps in the best order to quickly achieve the optimal results.

Note that the phone line's send and receive level settings work inversely proportionate to one another. Setting the PSTN to -9 dBm means the incoming audio in boosted nine decibels before it is put on the ACU backplane and the outgoing is attenuated 9 decibels before it hits the phone line. Since the audio level within the ACU chassis is 0 dBm, this will present a -9 dBm signal to the phone system.

Note the default setting of -9 dBm is the maximum allowed into most U.S. and foreign telephone networks at the subscriber end. Some PBXs may only allow -12 dBm. Ensure that the line levels are correct for the network being used and ***do not exceed maximum allowed levels***.

- Cross-connect the HSP handset to the PSTN and place a call from the PSTN to the phone number of an associate who will help with the setup. Refer to Section 2.13.4.1, Telephone Line Level. Within the guidelines presented, the line level may be adjusted in 3 dB steps to present the proper level to your associate at the remote telephone.
- Now listen as the associate speaks and verify that the VOX LED illuminates on the PSTN even for softly spoken speech. If the VOX doesn't always trip, raise the telephone RX

Level Boost setting until it does, but no higher. 6 dB of boost is the default setting and a good place to start.

- Continue the conversation while monitoring the VOX LED. If problems persist with failure to VOX or if false VOXing due to background noise at the distant phone occurs, adjust the VOX threshold. There are only two settings, Low & High. The default setting is Low threshold. If the VOX does not always trip, and the setting is currently at High, lower the threshold to the Low setting. If the threshold is already set to Low the VOX cannot be made more sensitive, so instead increase the RX Level Boost. If the VOX sometimes falses on background noise and the threshold is set to low, move the threshold to the high setting.

Note: The VOX is expected to trip on loud background noises; lower the threshold only if the VOX is activated for background sounds that are below the volume of normal speech.

- Now have your associate on the distant phone count from one to twenty at a slow, conversational rate. The VOX should remain active throughout. If the VOX drops in and out, raise the VOX hangtime just until this no longer occurs. The default setting is one second; if the VOX continually drops out between words, increase it to 1.5 seconds or, if necessary, to two seconds.

- With the HSP handset, talk with a normal conversation level and have your associate verify that you are being received at an acceptable level. If not, and you are outputting the maximum legal level to the line contact JPS for further suggestions.

- Interconnecting multiple active PSTNs together on the same net can sometimes cause hybrid unbalance issues. Please contact JPS for further suggestions and guidance if this is required as there are complex tradeoffs required.

JPS 24/7 Technical Assistance Phone Number is 1-800-498-3137. Press 1 for ACU Help.

2.16 Network Connections

All network connections can be made via standard CAT-5 Ethernet cables and the front panel RJ-45 connectors of the individual CPM, DSP, SCM-1, or SCM-2 interface modules. The NXU-2A unit may be included in the system as explained in its Operations Manual and in Section 1.8 of this manual. All of these devices follow the IP protocol and may be interfaced by standard off-the-shelf network equipment, switches, NAT Routers, hubs, etc.

2.17 Configuration Programming Via A Browser

The CPM-6, DSP-2IP, SCM-1 and SCM-2 modules have front panel Ethernet ports that may be used for configuration. Browse to the IP address of the module and view or change configuration settings as described in the sections that follow.

2.17.1 CPM-6 Configuration Programming via Browser

The CPM module's Ethernet port allows the ACU-2000 to be connected directly to an IP-Based network for remote control by the ACU Controller program, the WAIS Controller, or to be configured by a web browser program as explained in this section.

To change the CPM system configuration settings, the user must connect the CPM to the user's Ethernet LAN via the front panel RJ-45 Ethernet connector, via a standard "straight" Ethernet CAT5 cable to a switch or router which is also connected to a PC with network access to the same switch or router. Alternatively, the user may connect the Ethernet port of the CPM module directly to the computer's Ethernet port via an Ethernet "crossover" cable.

The user must browse (using a web browser such as the *Internet Explorer*) to the IP address of the CPM. The default address of the CPM as shipped from the factory is 192.168.1.200. This IP address may be changed to comply with the user's network requirements.

2.17.1.1 Information Page

Upon successfully browsing to the CPM, a screen similar to Figure 2-14 will appear. This page provides a summary of the current CPM module operating status and configuration.

Items in the top section are relevant to system configuration while the items in the lower section pertain to VOIP operation.

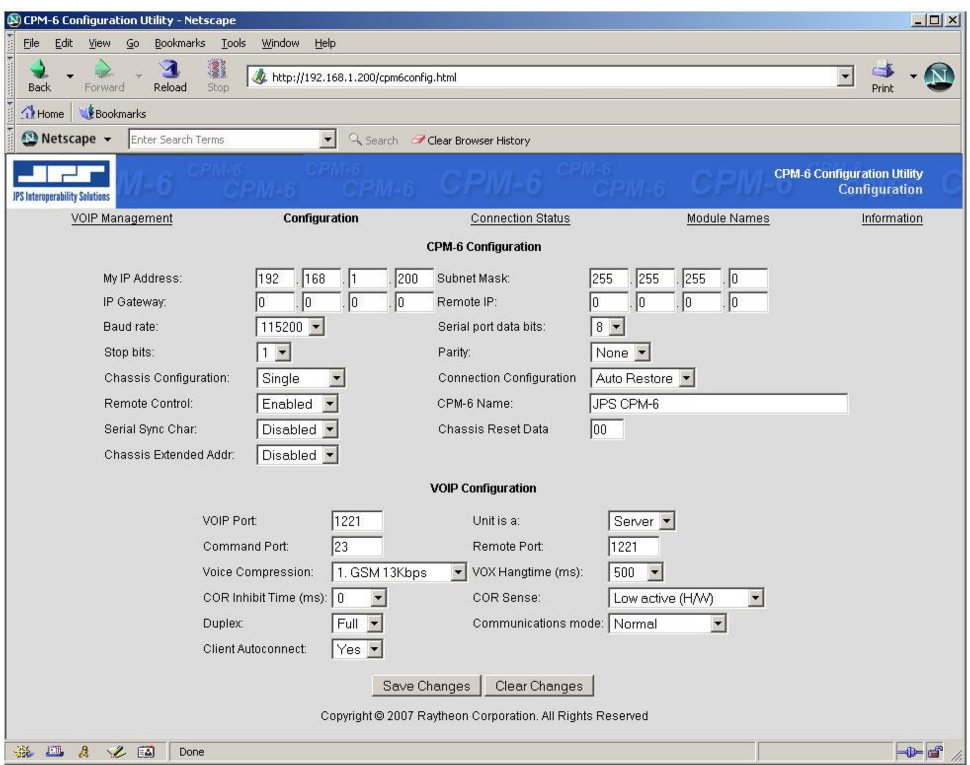


Figure 2-14 CPM Information Page

To modify any of the CPM operating parameters, select the **Configuration** link that appears at the top of the page. The Configuration Page will appear. See Figure 2-15 on the next page.

2.17.1.2 Configuration Page

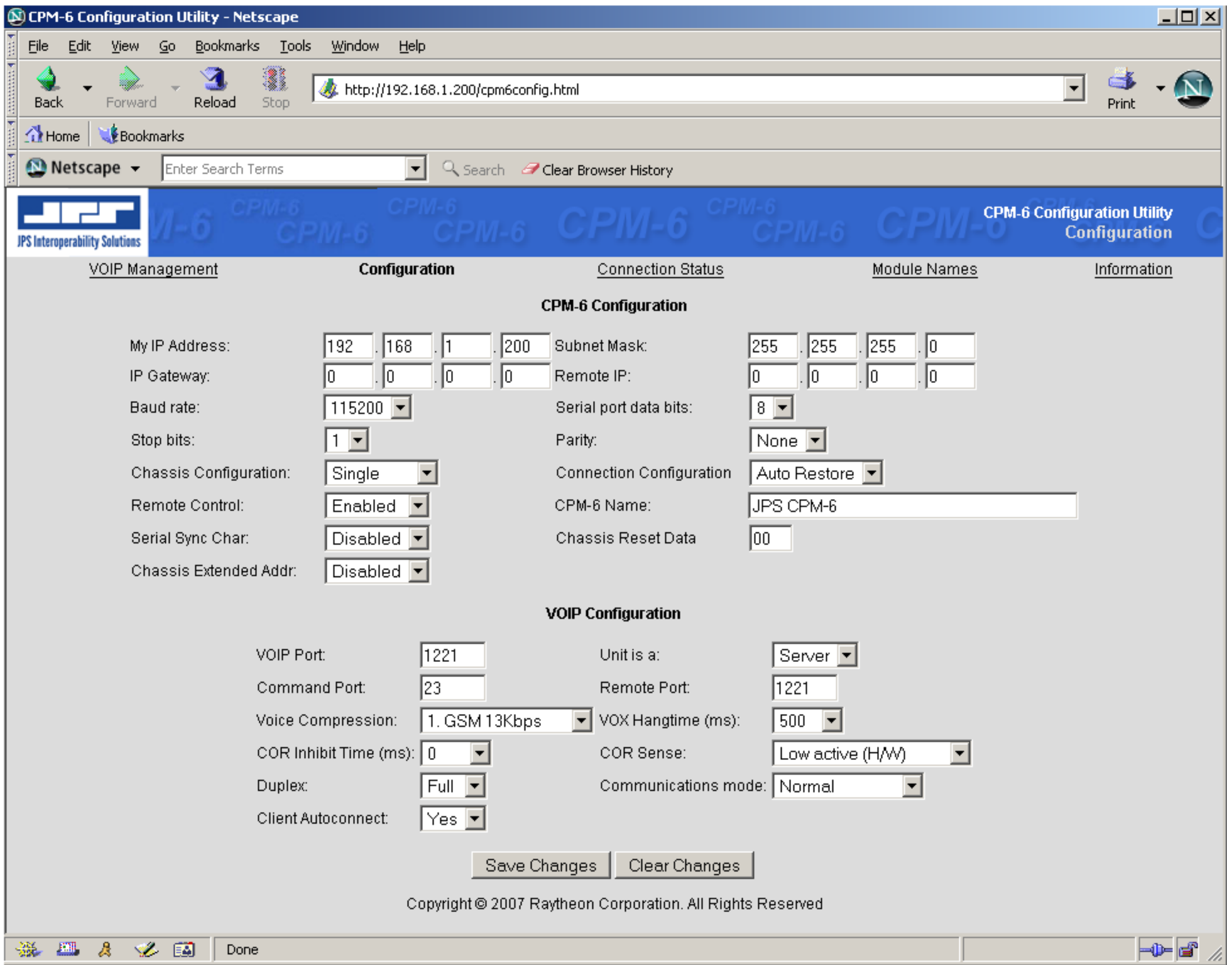


Figure 2-15 CPM Configuration Page

Configuration changes to the CPM are made either by selecting a field and entering text, or by making a selection from pull-down menus. Each of these fields and its accompanying options are described in the following sections.

> ***Note: For any operational changes to take effect, you must SAVE CHANGES via the Save Changes button. This button may be viewed by scrolling down to the bottom of the page.***

2.17.1.2.1 Configuration Page Field Descriptions

My IP Address:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the unit.

> ***Note: upon saving the changes, the user will need to browse to the new address to continue configuration.***

Subnet Mask:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol mask of the unit.

> ***Note: upon saving the changes, the user will need to browse to the new address to continue configuration.***

IP Gateway:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the gateway address that the unit will use for resolving external network accesses.

Remote IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the remote VoIP unit that is paired with this unit. When the CPM is set up as a client, this is where the server's IP address is entered. [This option relates to the VoIP capability of the CPM module.]

Baud Rate:

The pull-down menu allows the user to configure the baud rate for the RS-232 serial port located on the rear panel of the ACU-2000. Nine baud rates are available: 300, 1200, 2400, 4800, 9600, 19200, 38400, 57600, and 115200.

Serial Port Data Bits:

The pull-down menu allows the user to configure the number of data bits for the serial port. Two choices are available: 7 or 8 bits. The ACU Controller serial mode uses 8 bits.

Stop Bits:

The pull-down menu allows the user to configure the number of stop bits for the serial port. Two choices are available: 1 or 2 bits. The ACU Controller serial mode uses 1 bit.

Parity:

The pull-down menu allows the user to configure the parity for the serial port. Three choices are available: even, odd, or none. The ACU Controller serial mode uses *none*.

Chassis Configuration:

The pull-down menu allows the ACU-2000 chassis configuration to be set to one of the following:

- **Single**, for a single chassis.
- **Master**, for a multiple chassis system where this is the main chassis.
- **Expanded**, for a multiple chassis system where this is the Expansion chassis.

Connection Configuration:

The ACU-2000 can be configured to automatically restore the module cross-connections to the user programmed preset configuration upon power-up. See the *Stored Connections Auto Restore* feature of Section 5.1.

- **Auto Restore**, puts the module connections in the user programmed configuration on power-up.
- **No Restore**, leaves all modules unconnected on power-up.

Remote Control:

This pull-down menu allows the ACU-2000 RS-232 serial port on the rear panel to be **Enabled** or **Disabled**.

CPM Name:

The user may enter text in this field that identifies this CPM. The name should uniquely differentiate this unit from all others on the network. This name will appear on the WAIS Controller to identify this site.

Serial Sync. Char:

This pull-down menu is used to **Enable** or **Disable** the RS-232 remote control synchronizing character feature. The ACU Controller serial mode uses disable.

Chassis Reset Data:

The user may enter a 2-digit number in this field that will be used by the chassis reset (system reset) command that is available to remote DTMF users.

The System Reset Feature allows users in the field with DTMF capability (and the proper code) to reset the ACU-2000 to its initial power-up state. This means all current cross-connections will be lost, and the unit will return to any connections stored last by the * 3 6 command. In order to prevent any inadvertent or unauthorized use of this powerful feature, the System Reset Feature can only be used after first being enabled by entering a code other than 00 as the Chassis reset data field of the CPM Configuration Page (or via the HSP keypad). Users accessing the system via DTMF input must then enter this code in order to implement this feature.

The ACU-2000 factory default for this feature is *disabled*. To enable System Reset capability, enter any number other than 0 0 in the Chassis Reset Data Field. The feature is then enabled, and *n n* is the system reset code. If a user enters the DTMF command ** 9 0 n n***,** the system will be reset. If ** 9 0* and any digits other than the system reset code are entered, the system will not be reset.

To disable this feature, enter *0 0* in this field. This feature may also be enabled or disabled by the HSP keypad.

VoIP Port:

The user may enter numeric text in this field (1-65535) defining which port to use for the VoIP traffic.

Unit is a:

The pull-down menu allows the user to choose either *server* or *client* mode. VoIP operation requires pairing of clients and servers.

Command Port:

The user may enter numeric text in this field (1-65535) defining which port to use for the remote control traffic.

Remote Port:

The user may enter numeric text in this field (1-65535) defining which port to use when connecting to its paired unit for VoIP traffic.

Voice Compression:

The pull-down menu allows the user to configure the voice compression (vocoder) that is used for VoIP audio traffic. Five choices are available: GSM at 13Kbps, ADPCM at 16Kbps, ADPCM at 24Kbps, ADPCM at 32Kbps, and PCM at 64Kbps. If the unit is a server, adaptation to the incoming client vocoder is automatically selected to match the client request.

VOX Hang time (ms):

The pull-down menu allows the user to configure the VoIP audio VOX hangtime in milliseconds. Five hangtimes are available: 500ms, 1000ms, 2000ms, 3000ms, and 4000ms.

COR Inhibit Time (ms):

The pull-down menu allows the user to configure the COR inhibit time. Six options are available: 0ms, 500ms, 1000ms, 2000ms, 3000ms, 4000ms. WHAT DO THESE HAVE TO DO WITH VoIP?

COR Sense:

The pull-down menu allows the user to configure the COR sense. Three options are available: active low, active high, and VOX.

Duplex:

The pull-down menu allows the user to configure the VoIP channel for either full or half duplex operation.

Communications Mode:

The pull-down menu allows the user to configure for either *normal*, *broadcast*, *connectionless*, or *multicast* modes of VoIP communications. Except for special applications this setting should be left as *normal*.

Client Autoconnect:

The CPM can be configured to automatically attempt to establish connection with a given Server if the CPM is configured as a Client and a valid IP address has been entered for the Server.

2.17.1.3 VoIP Connection Management

Client VOIP sessions may be managed by browsing to the *VoIP Management* link at the top of any of the CPM's web pages. Figure 2-16 shows the *VoIP Management* page.

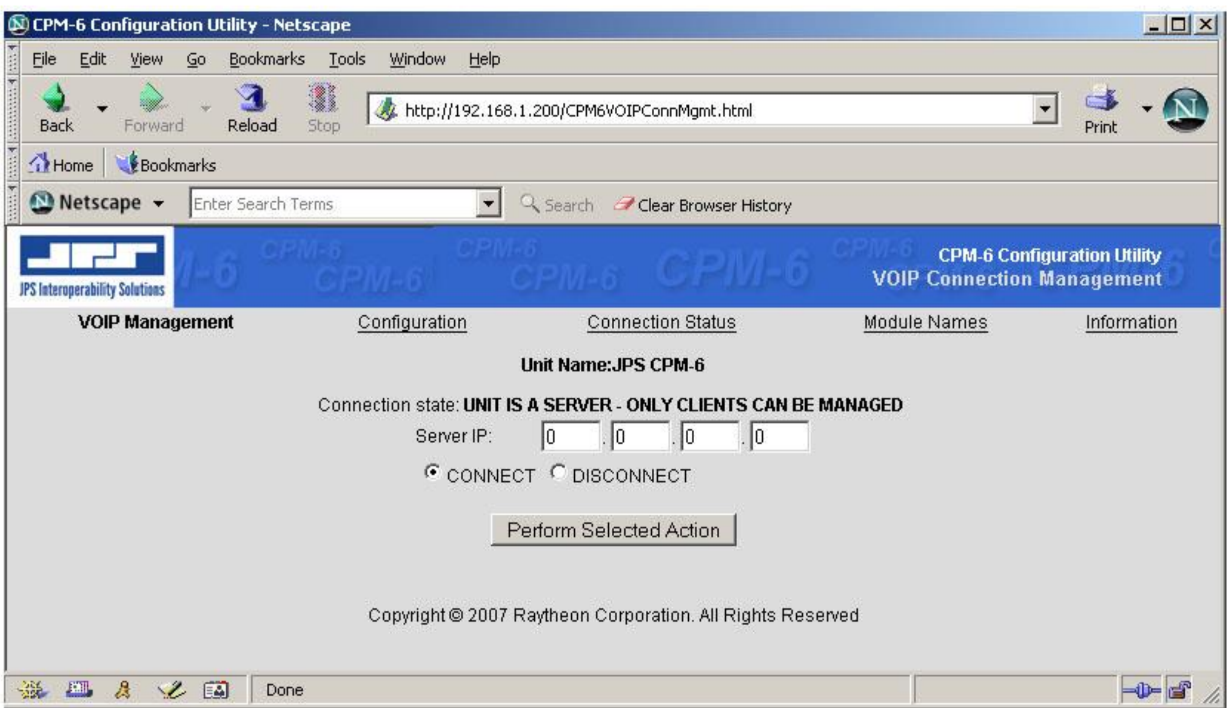


Figure 2-16 CPM VoIP Connection Management Page

This page is used to force the use of a new server, or for connecting or disconnecting an existing link to a server.

Server IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the server unit that is paired with this unit.

Connect / Disconnect:

The user may request that a connection to the server be established by activating the *CONNECT* button.

The user may request that the connection to the server be broken by activating the *DISCONNECT* button.

Note: Requests will be processed only after activating the* Perform Selected Action *button.

2.17.1.4 Connection Status

VOIP session status may be monitored by browsing to the **Connection Status** link at the top of any of the unit's web pages. A variety of statistics for the session are presented.

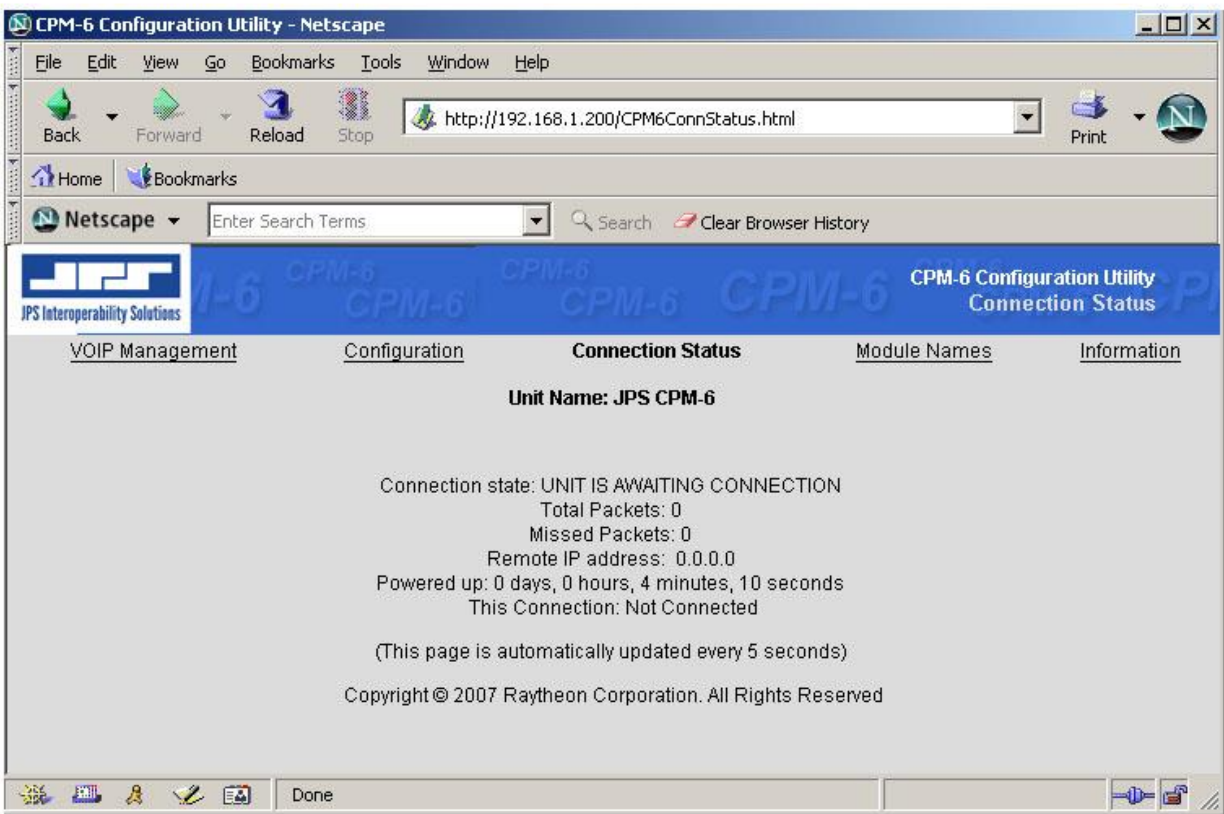


Figure 2-17 CPM Connection Status Page

2.17.1.5 Module Identifier Page

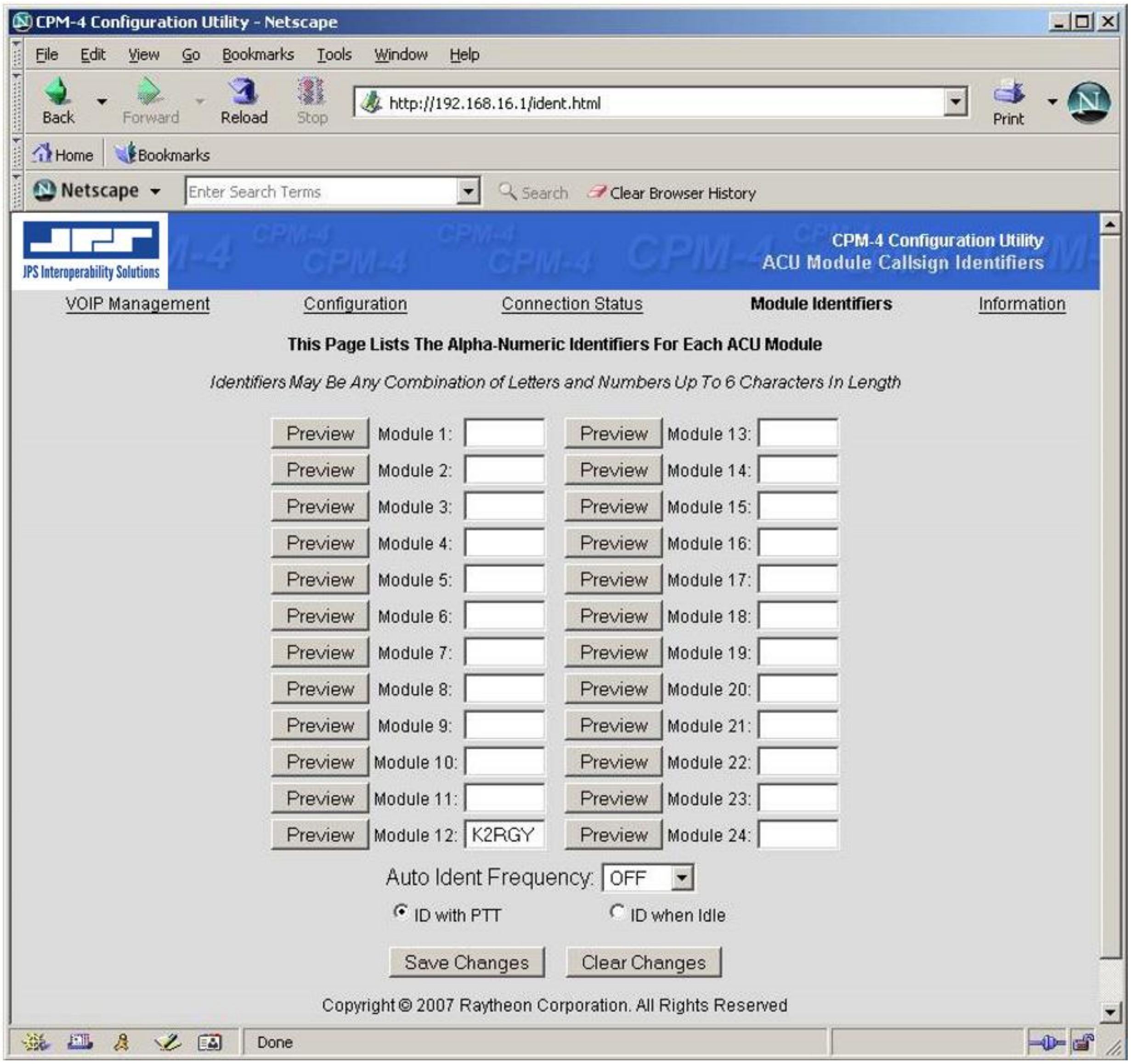


Figure 2-18 CPM Module Identifier Page

The ACU-2000's Automatic Unit ID feature allows customer-specified alphanumeric call signs to be broadcast at specified times following system transmissions. This allows local officials to identify which system(s) are currently in operation and can be a valuable aid in finding duplicate or otherwise inappropriate cross-patches.

The Auto-ID function is triggered when a module's PTT is applied. This starts a timer that is user specified to be 5, 10, 15, 30, or 60 minutes. When it is time for the ID to be broadcast, the module will automatically apply PTT and transmit a voice prompt consisting of the user-entered alphanumeric call-sign data.

The ACU-2000 system must have a CPM-6 module running version 3.05 (or later) software. Only DSP-2IP or DSP-3IP modules can be used with the Auto-ID feature. The DSP modules must be running version 3.03 (or later) software. The DSP modules must also be in *Standard* mode (Hybrid and VoIP Standalone modes do not have this feature).

The Auto-ID feature is set up by first browsing to the CPM-6. This is usually done by entering the IP address of the module in the browser's URL window. The user then selects the *Module Identifiers* link.

The *Module Identifier* page shows a list of 24 modules, one for each possible slot in an ACU system. Any slot where a DSP module is not installed or otherwise cannot be used to generate Auto-ID will have a grayed-out *Preview* button. In the CPM Module Identifier Page screen shot only modules 7 and 9 are "active." Both also have call signs already configured.

Each module has a text entry window for the user to enter alphanumeric text for the call sign. Only letters A through Z and digits 0 through 9 are accepted as call-sign text.

After the user has entered the call-sign information for each applicable module, the *Save Changes* button is used to commit the data to permanent memory. There is also a *Clear Changes* button that will automatically clear out any pending user changes to call-sign data.

Each module has a *Preview* button. This is used to preview the call-sign announcement on the ACU HSP module speaker. This does not play the call-sign prompt over the air.

The *Auto Ident Rate* pull-down selection allows the user to select Auto-ID times of 5, 10, 15, 30, or 60 minutes. Selecting *OFF* will disable the Auto-ID feature. The Auto Ident Rate is the time the system will wait, following the activation of PTT, before it will automatically broadcast the call-sign announcement.

There are two mutually exclusive selections titled *ID with PTT* and *ID when Idle*. Assume the *Auto Ident Rate* is set for 5 minutes. When the DSP module first asserts its PTT output, the timer begins. After 5 minutes, the module is eligible to transmit the call-sign announcement.

If the *ID when Idle* option is selected, the ACU will announce the call sign when the timer elapses as long as there are no COR or PTT conditions currently present on the module (the module is not receiving or transmitting). The prompt is transmitted once, and the timer does not start counting again until the next active PTT session for the module. If an active PTT or COR condition is present when the timer elapses, the call sign prompt is transmitted as soon as the condition ends.

If *ID with PTT* is selected, the feature never creates a PTT condition on its own, but instead sends out the call sign only at the end of a transmission that's caused elsewhere within the ACU system. That is, the unit will not send out the call sign as soon as the timer elapses but will wait until the next PTT occurs following the elapsed time, and transmit the call-sign prompt as soon as PTT is deactivated. The timer is restarted and the prompt will be transmitted again following the first PTT after the timer again expires.

There is a *Broadcast ID to Net Members* box on the *Module Identifiers* screen. Typically, call-sign announcements are only transmitted to the radio connected to the DSP module. When *Broadcast to Net Members* is selected, the call-sign announcement is instead transmitted by all modules currently in the same “interoperability net” as the DSP-2IP or DSP-3IP.

2.17.1.6 Module Names Page

The CPM has the ability to store and retrieve user-defined names for each of the modules installed in the chassis. The module names may be programmed by browsing to the **Module Names** link at the top of the web pages, and then typing in the name alongside the associated module extension number.

These module names are only needed when using the ACU-2000 in a WAIS (Wide Area Interoperability System). See the WAIS Controller manual for details. These names will show up on the WAIS Controller screen.

> ***Note: Module 0 corresponds with the HSP module of a single Chassis ACU-2000 or Master ACU-2000 of a dual chassis system. Numbers 1 through 12 are associated with the unit’s interface module extension numbers. For dual chassis systems, Module 25 is the Expansion Chassis HSP module, while numbers 13 through 24 correspond with this chassis’ interface extensions. See Figure 2-19.***

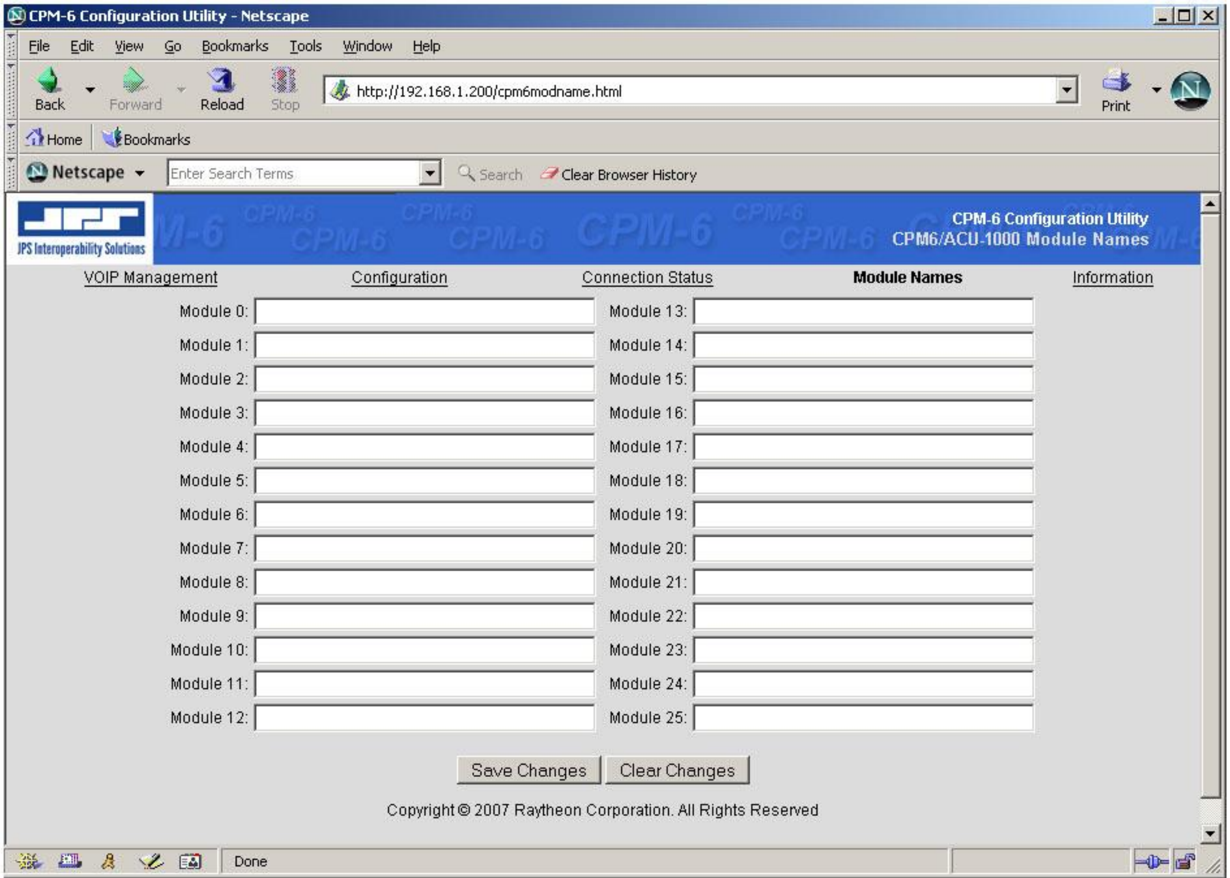


Figure 2-19 CPM Module Names Page

2.17.1.7 Restoring Factory Defaults

In rare circumstances, there may be the need to completely restore the CPM to the original configuration that was established when the module was manufactured. The procedure for doing this follows.

Equipment Required:

- Extender Card (supplied in the ACU-2000 Accessory Kit).
- ACU-2000 (ACU-T)

Procedure:

1. Power down the ACU-2000.
2. Remove the CPM from the rack.
3. Install the extender card into the now empty slot.
4. Install the CPM into the extender card.
5. Configure the *Restore Factory Defaults* jumper J16 pins 2&3 [left to center]
6. Power up the ACU-2000.
7. Wait 15 seconds. (The modules front panel LEDs will sequence.)
8. Power off the ACU-2000.

9. Remove the CPM from the extender card.
10. Remove the extender card from the ACU-2000.
11. Re-install the bridging block on jumper J16 pins 1&2 [right to center].
12. Re-install the CPM back in the ACU-2000 and you are finished.

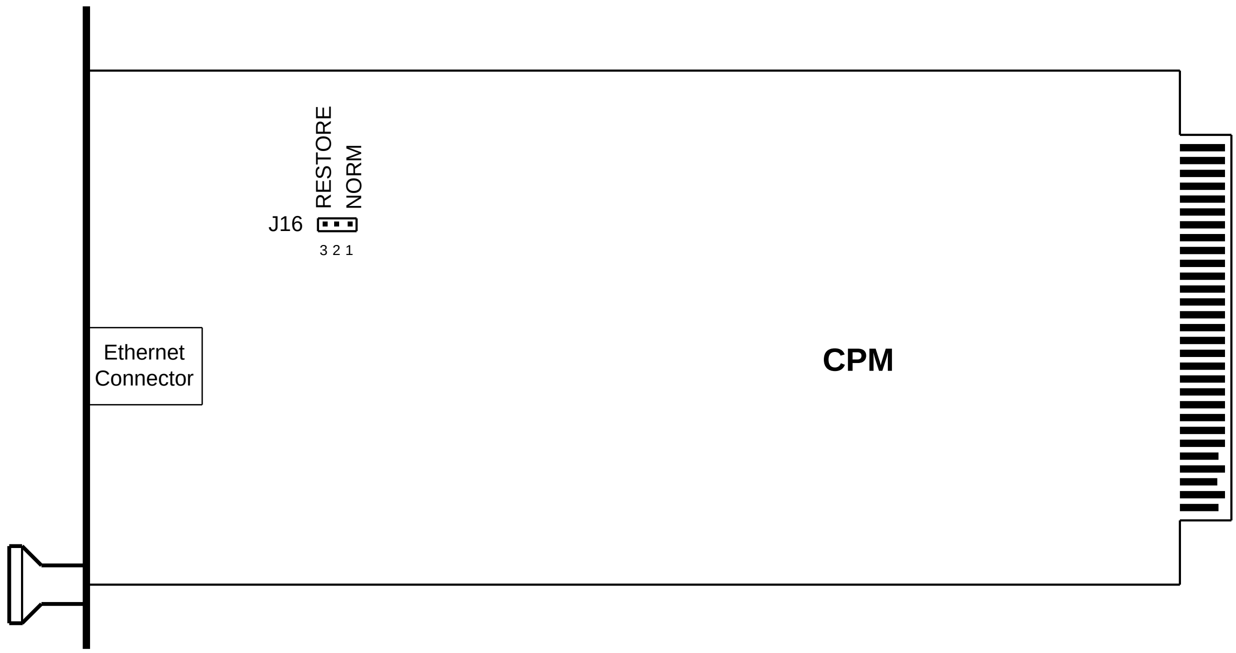


Figure 2-20 CPM Restore Defaults Jumper Location

The completion of the above procedure will re-establish the original factory configuration to the CPM. In summary, they are shown in Figure 2-21.

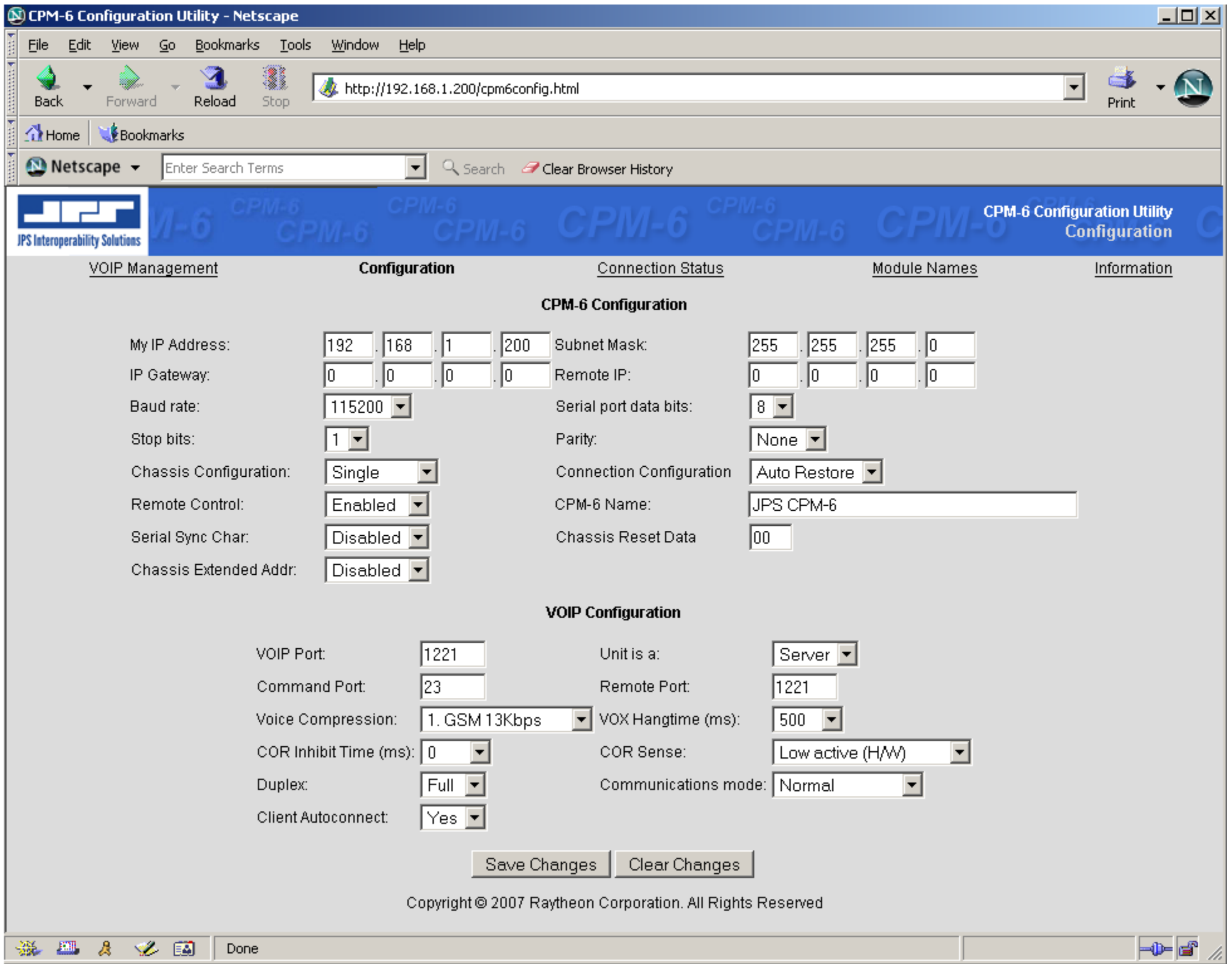


Figure 2-21 CPM Restored Factory Defaults

2.17.1.8 CPM Software Updates

The CPM is designed to support software updates in the field. Should it become necessary to install updates to the software, the following process should be followed. Instructions and the update software are available on the web at:

http://www.jpsinterop.com

Or by email at:

support@jpsinterop.com

On the page all ACU-2000 related downloads are listed. This includes this manual, the ACU Controller, and latest versions of software for the ACU-2000 modules.

Equipment Required:

1. PC with Internet network access via a browser (Internet Explorer).
2. Ethernet access via an Ethernet switch (for the CPM connection).
3. Ethernet cable.
4. CPM installed in the ACU-2000.
5. CPM software and installation software (available from the web site listed above).
6. The IP address of the networked CPM that will be updated.

Procedure:

1. Use the PC connected to the Internet to browse to the web site listed above.
2. Download the CPM software by right clicking on *cpm6_update.zip*, and choose *save target as*. Browse to a local folder on your computer to deposit it, then click *save*.
3. Unzip the files in the zip archive.
4. Connect the CPM to the LAN via the switch and Ethernet cable.
5. Launch the autoupdate software by navigating to the folder where it was unzipped / saved, and double clicking on the file *autoupdate.exe*.
6. A dialog box similar to the one shown below will appear:

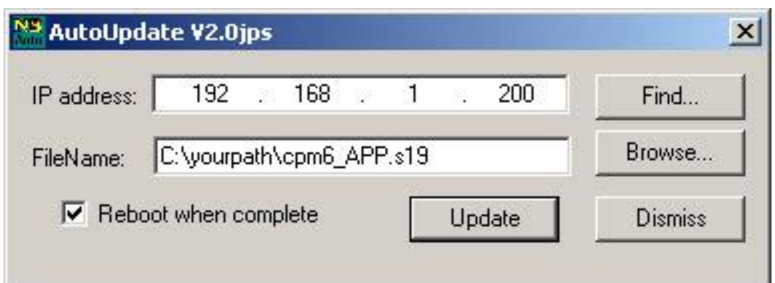


Figure 2-22 AutoUpdate Dialog

7. Enter the IP address of the CPM, the path to *cpm6_app.s19* (the update software), and insure that the *Reboot when complete* checkbox is checked.
8. Click the *Update* button, and a status bar will appear, showing the update progress.
9. After a short delay (10-15 seconds), the following dialog will appear: the CPM will be reset and restarted with the new software activated.

Figure 2-23 Successful Programming Announcement

10. Click OK to close the *AutoUpdate* dialog.

11. Verify the new version of the software has been loaded correctly by browsing to the IP address of the CPM-6, and validating the Firmware version listed on the Information Page matches the latest release CPM SW Version (per web site).

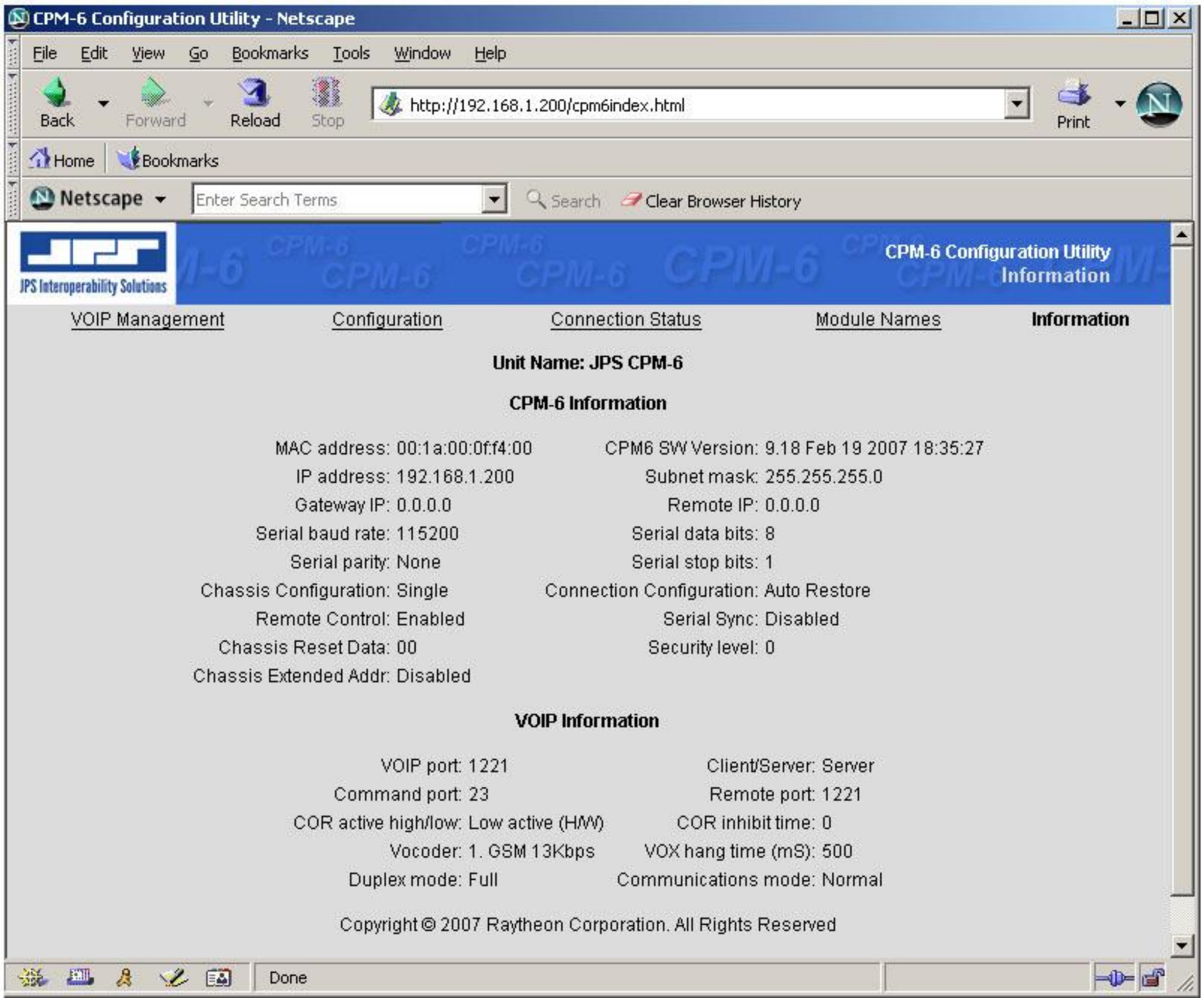


Figure 2-24 CPM Information Page

12. This completes the process for updating the CPM software.

2.17.2 DSP-2IP Configuration Programming via Browser

This section addresses the configuration of the features enabled by the module's Ethernet port. Either of three operating modes may be selected. The DSP module can use any two of its three interfaces, and the interface selection determines the operating mode.

The interfaces are:

- Front panel RJ45 Ethernet connector
- Rear panel four-wire interface (15-pin D-sub connector to radios or other devices)

- Chassis backplane for CPM control and for connections with other modules

The operating modes are that these interfaces can produce are

- Standard Mode: Rear panel four-wire & chassis backplane
- VoIP Standalone Mode: Front panel RJ45 and rear panel four-wire connector
- VoIP Hybrid Mode: Front panel RJ45 and chassis backplane

The VoIP features of the two VoIP modes are configured by browsing to the front panel RJ45 connector. The non-VoIP features of all modes are set either by browsing to the RJ45 or via the ACU Controller. Please note that once a unit is configured by a browser to VoIP Standalone Mode, it can no longer communicate with the CPM module or the ACU Controller. The ACU Controller includes various features (such as a library of stored radio configuration templates) that assist with optimizing the four-wire interface, so the ACU Controller is the recommended setup method for this interface in all modes other than VoIP Standalone Mode.

See 2.17.3 for minor variations in programming the DSP-3 module, which allows VoIP communications with a WAIS Controller 2 operator in conjunction with the Standard Mode operation.

2.17.2.1 Standard Mode

When in the Standard mode, the DSP module cross-connects audio and control signals from radios to other modules in the local ACU system via the chassis backplane. It’s still possible to browse to the module via its front panel RJ-45 Ethernet port, but there is no VoIP capability. A more precise explanation includes the facts that:

- The local ACU system includes any modules of an Expansion Chassis
- Besides its primary function as a radio interface, the DSP module will interface any type of four-wire device

2.17.2.2 VoIP Standalone Mode

This mode allows the DSP module to act as an independent, standalone, network-to-radio interface. The ACU chassis provide a rear panel radio connector and power & ground signals to the module but has no other interactions with it. In this mode, the DSP module will be ignored by the ACU Controller and WAIS Controller programs, as well as by the CPM module and any other modules in the ACU chassis. The module’s Ethernet connector allows VOIP connections to other network-capable devices on the network that use the JPS RoIP protocol. These devices include other DSP modules, CPM modules, NXU-2A units, as well as PCs running the PCNXU software. The DSP in Standalone Mode functions exactly like an NXU unit, and multiple NXU-2 units may be replaced by multiple DPSs in Standalone Mode.

> ***Note: If you are unfamiliar with the capabilities and functionality of the NXU-2A, information and free downloads (including the full NXU-2A manual), are available at:***
>
> ***http://www.jpsinterop.com***

2.17.2.3 VoIP Hybrid Mode

This mode allows the DSP to function as an RoIP interface to the ACU backplane, allowing remote cross-connection to take place over an IP network. In this mode, the DSP is visible to the ACU's CPM module and the ACU Controller and WAIS Controller software. There is no connection to the associated rear panel D15 four-wire connector, as the DSP module's two operating interfaces in this mode are the front panel RJ-45 and the chassis backplane.

2.17.2.4 Provisioning

As shipped from the factory, the default configuration of the DSP is Standard mode, and no network provisioning is required. When used in either the VoIP Standalone or VoIP Hybrid modes, the DSP must be configured for VOIP operations. Configuration is accomplished via PC / Browser / Ethernet web access. Configuration is best done using the ACU Controller Software; most of the DSP module's configuration options in the Standard mode are not accessible via the browser interface.

To activate either of the two VoIP modes, first connect the DSP to an Ethernet LAN via the front panel RJ-45 Ethernet connector. Use a "straight-through" CAT5 Ethernet cable to a switch or router which is also connected to a PC with network access to the same switch or router.

The user must browse (using a web browser such as Internet Explorer) to the DSP module's IP address. The default address as shipped from the factory is 192.168.1.200. (After provisioning, this IP address may be changed to suit user requirements).

A direct connection between a PC and the module may be used when:

- A CAT 5 *crossover cable* is used rather than a *straight-through cable*
- The PC is preconfigured with a static IP address

2.17.2.5 DSP-2 Information Page

Upon successfully browsing to the DSP module, a page similar to Figure 2-25 will appear. This page contains a summary of the module's current operating status and configuration.

Items on the left side are relevant to VoIP / networking issues, while items listed on the right side mainly report other operational options.

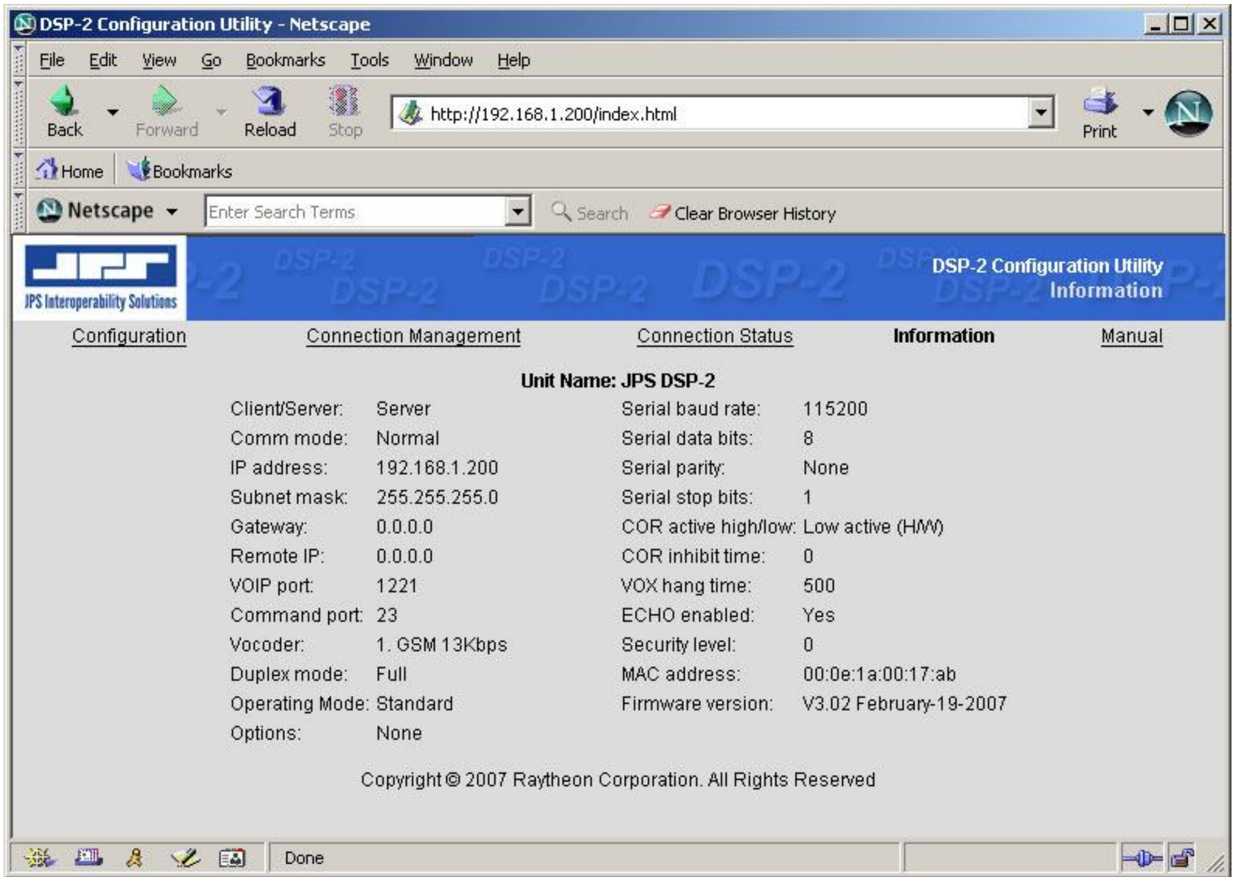


Figure 2-25 DSP Information Page

2.17.2.6 DSP-2 Configuration Page

To provision the DSP module for VoIP operation, select the *Configuration* link that appears at the top of the page (see Figure 2-26).

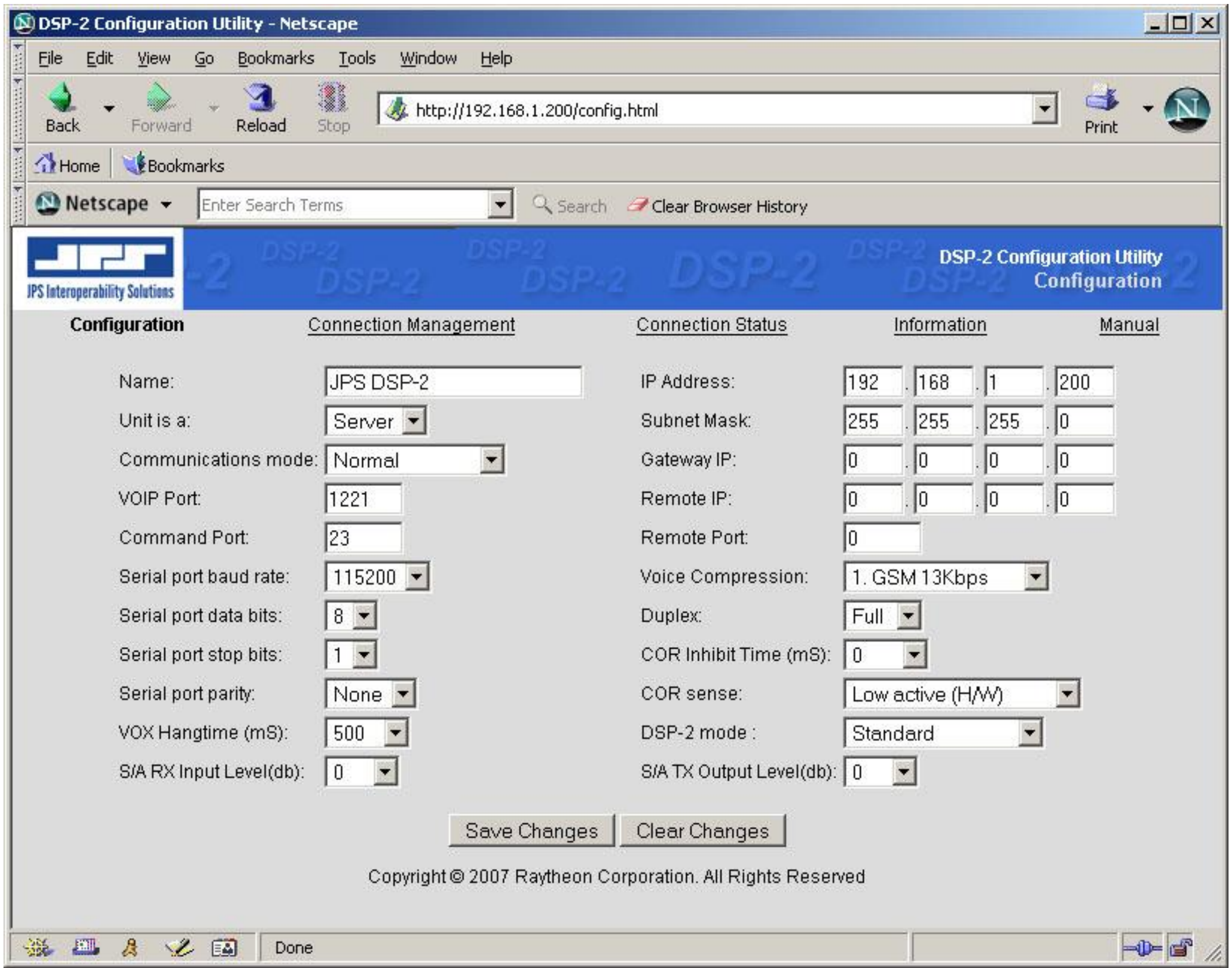


Figure 2-26 DSP Configuration Page

Configuration changes to the DSP module are made either by selecting a field and entering text, or by making a selection from pull-down boxes. Descriptions of each of these fields, and the options each contains, are presented in the next section.

> ***Note: For any operational changes to take effect, you must save changes via the Save Changes button.***

2.17.2.6.1 Configuration Page Field Descriptions:

Name:

The user may enter text in this field that identifies this DSP module. The name should uniquely identify this unit, differentiating it from all other units on the network.

Unit is a:

The pull-down menu allows the user to choose either *server* or *client* mode. VoIP operation requires pairing of clients and servers.

Communications Mode:

The pull-down menu allows the user to configure for either *normal*, *broadcast*, *connectionless*, or *multicast* modes of VoIP communications.

VOIP Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use for the VoIP traffic.

Command Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use for network command traffic.

Serial Port Baud Rate:

The pull-down menu allows the user to configure the baud rate for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Nine baud rates are available: 300, 1200, 2400, 4800, 9600, 19200, 38400, 57600, and 115200.

Serial Port Data Bits:

The pull-down menu allows the user to configure the number of data bits for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Two choices are available: 7 or 8 bits.

Serial Port Stop Bits:

The pull-down menu allows the user to configure the number of stop bits for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Two choices are available: 1 or 2 bits.

Serial Port Parity:

The pull-down menu allows the user to configure the parity for the serial communication channel on the associated rear-panel DB-15 connector. Three choices are available: even, odd, or none.

> ***Note: Configuration options that affect the ACU rear panel D15 connector four-wire interface (for radios or other four-wire devices) are best set by the ACU Controller, unless this isn't possible, for example, when the DSP module is in the Standalone mode and therefore has no communications with a CPM module and hence the ACU Controller.***

VOX Hangtime (ms):

The pull-down menu allows the user to configure the VOX hangtime in milliseconds. Five hangtimes are available: 500ms, 1000ms, 2000ms, 3000ms, and 4000ms. VoIP or 4W interface?

IP Address:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the unit.

> ***Note: Upon saving the changes, the user will need to browse to the new address to continue configuration.***

Subnet Mask:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol mask of the unit.

> ***Note: Upon saving the changes, the user will need to browse to the new address to continue configuration.***

Gateway IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the gateway address that the unit will use for resolving external network accesses.

Remote IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the remote VoIP unit that is paired with this unit.

Remote Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use when connecting to its paired unit for VoIP traffic.

Voice Compression:

The pull-down menu allows the user to configure the voice compression (vocoder) that is used for VOIP audio traffic. Five choices are available: GSM at 13Kbps, ADPCM at 16Kbps, ADPCM at 24Kbps, ADPCM at 32Kbps, and PCM at 64Kbps. If the unit is a server, adaptation to the incoming client vocoder is automatically selected to match the client request.

Duplex:

The pull-down menu allows the user to configure the module for either full or half duplex operation. For Hybrid Mode (see the DSP Mode option setting) the ACU Controller is the preferred method for DSP module audio interface adjustment.

OR Inhibit Time (ms):

The pull-down menu allows the user to configure the COR inhibit time. Six options are available: 0ms, 500ms, 1000ms, 2000ms, 3000ms and 4000ms; 4W interface. This setting applies to the unit's four-wire audio interface and is included for adjustment when the module is being used in the standalone mode. For Hybrid Mode (see the DSP Mode option setting) the ACU Controller is the preferred method for DSP module audio interface adjustments.

COR Sense:

The pull-down menu allows the user to configure the COR sense. Three options are available: active low, active high, and VOX. This setting applies to the unit's four-wire audio interface and is included for adjustment when the module is being used in the standalone mode. For Hybrid Mode (see the DSP Mode option setting) the ACU Controller is the preferred method for DSP module audio interface adjustments.

DSP Mode:

The pull-down menu allows the user to configure the operating mode of the DSP. Four modes are supported (see the beginning of Section 2.17.2 for an explanation of the modes):

	Mode	Interfaces Used
1.	Standard Mode:	Rear panel four-wire & chassis backplane
2.	VoIP Standalone Mode:	Front panel RJ45 and rear panel four-wire connector
3.	VoIP Hybrid Mode:	Front panel RJ45 and chassis backplane
4.	Test:	Test mode for factory use only with loopback fixture.

RX Input Level:

The pull-down menu allows the user to adjust the receive input level for receiver audio. Five levels are supported: +12db, +4db, 0db, -4db, and –12db; 4W interface. This setting applies to the unit's four-wire audio interface and is included for adjustment when the module is being used in the standalone mode. For Hybrid Mode (see the DSP Mode option setting) the ACU Controller is the preferred method for DSP module audio interface adjustments.

TX Output Level:

The pull-down menu allows the user to adjust the transmit output level for transmit audio. Five levels are supported: +12db, +4db, 0db, -4db, and –12db; 4W interface. This setting applies to the unit's four-wire audio interface and is included for adjustment when the module is being used in the standalone mode. For Hybrid Mode (see the DSP Mode option setting) the ACU Controller is the preferred method for DSP module audio interface adjustments.

2.17.2.7 DSP-2 Connection Management Page

Client VOIP sessions may be managed by browsing to the **Connection Management** link at the top of any of the unit's web pages. Figure 2-27 shows the Connection Management page.

This page may be used force the use of a different server, or for connecting or disconnecting an existing link to a server.

Server IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the server unit that is paired with this unit.

Connect / Disconnect:

The user may request that a connection to the server be established by activating the *CONNECT* button.

The user may request that the connection to the server be broken by activating the *DISCONNECT* button.

Note: Requests will be processed only after activating the Perform Selected Action button.

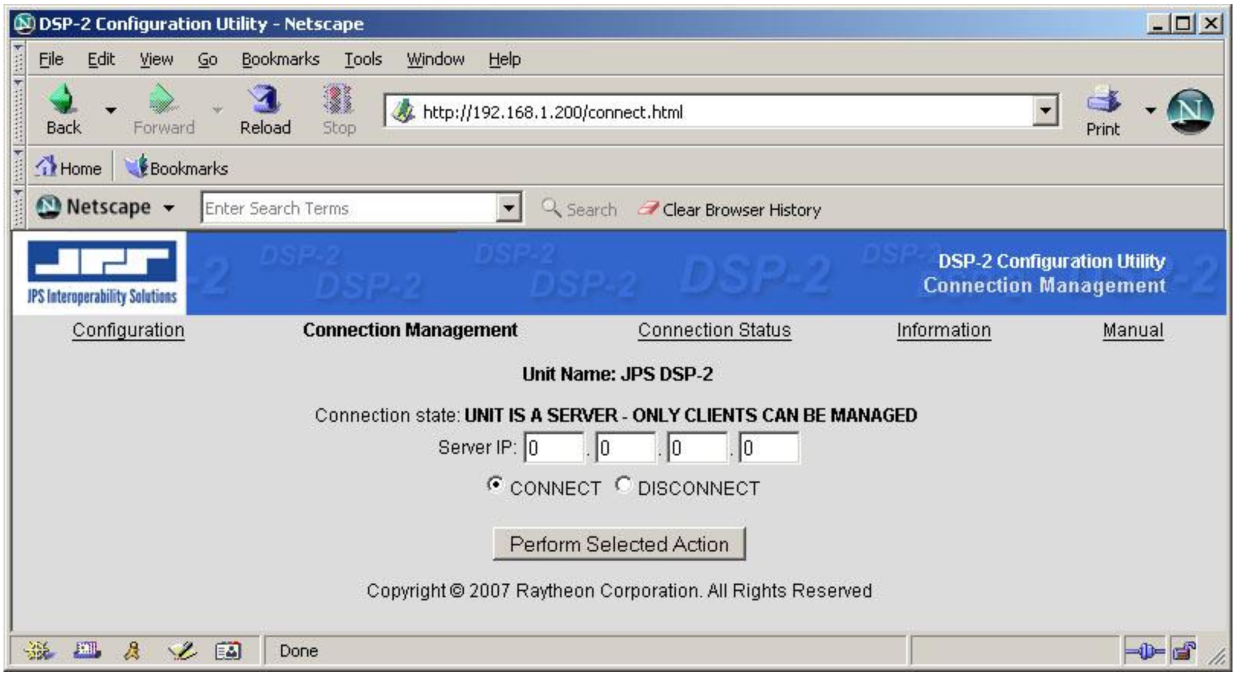


Figure 2-27 DSP Connection Management Page

2.17.2.8 DSP-2 Connection Status Page

VOIP session status may be monitored by browsing to the **Connection Status** link. A variety of statistics for the session are presented (see Figure 2-28).

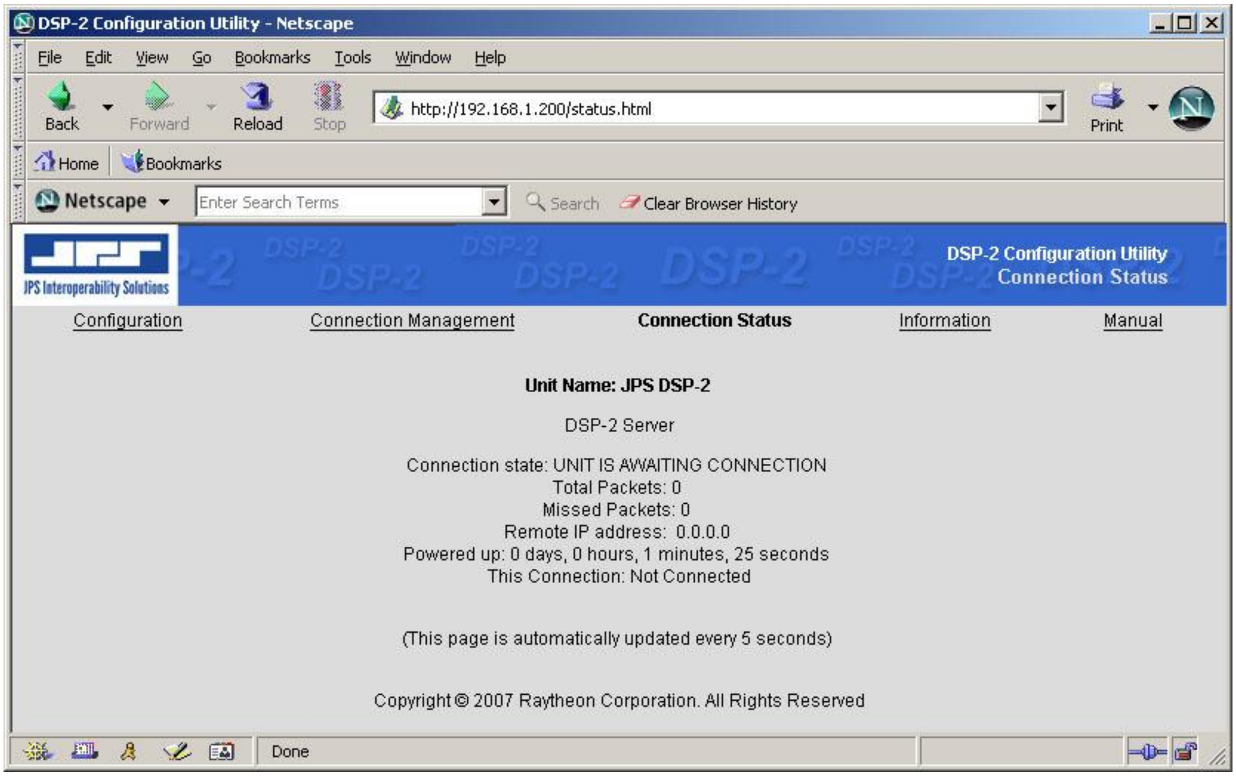


Figure 2-28 DSP Connection Status Page

2.17.2.9 DSP-2 Manual Page

A subset of the ACU-2000 manual regarding DSP configuration issues may be viewed online by browsing to the *Manual* link at the top of any of the unit's web pages. Full ACU-2000 related documentation is available for download on the web page at:

http://www.jpsinterop.com

2.17.2.10 Restoring Factory Defaults

In rare circumstances, there may be the need to completely restore the DSP to the original configuration that was set when the module was manufactured. The procedure for doing this follows. (Does not apply to four-wire interface configuration options).

Note: Jumper locations are shown in Figure 2-8

Procedure:

1. Power down the ACU-2000.
2. Remove the DSP from the rack.
3. Install the extender car d (supplied in the Accessory Kit) into the now empty slot.
4. Install the DSP into the extender card.
5. Configure the *Restore Factory Defaults* jumper **JP22 [center to right].**
6. Power up the ACU-2000.
7. Wait 10 seconds. (The LEDs will repeatedly sequence…)
8. Remove the bridging block from **JP22 [while powered]**.
9. Wait 15 seconds. (The DSP will complete the reset sequence)
10. Power off the ACU-2000.
11. Remove the DSP and extender card from the ACU-2000.
12. Re-install the bridging block on jumper **JP22 [left to center].**
13. Install the DSP back into the vacant slot in the ACU-2000.
14. Power up the ACU-2000.
15. Finished

Results: The completion of the above procedure will re-establish the original factory configuration to the DSP. The factory defaults as of the release of this manual are shown in Figure 2-29.

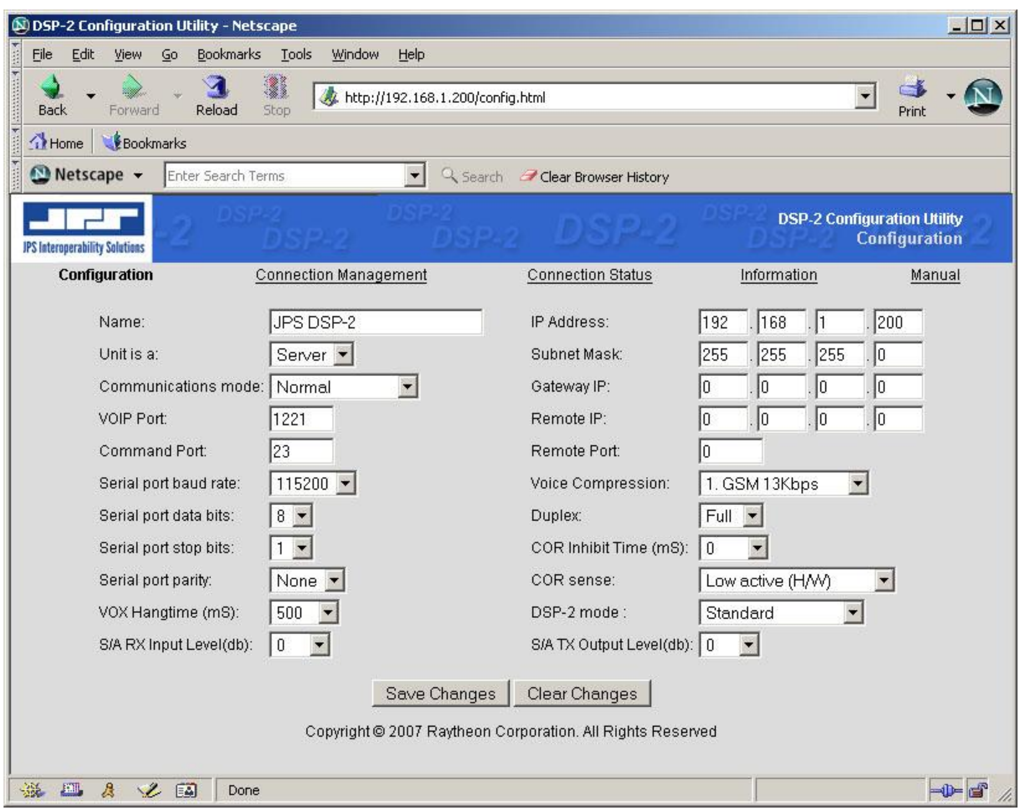


Figure 2-29 DSP Module Restored Factory Defaults

2.17.2.11 Software Updates

The DSP is designed to support software updates in the field. Should it become necessary to install updates to the software, the following process should be followed. Instructions and the update software are available on the web at:

http://www.jpsinterop.com

Or via email at: support@jpsinterop.com.

Equipment Required:

- PC with Internet network access via a browser (e.g. Internet Explorer).
- Ethernet access via an Ethernet switch (for the connection to the DSP module's front panel Ethernet connector).
- Ethernet cable.
- DSP installed in the ACU-2000.
- DSP software and installation software available from the web site listed above.

The IP addr **Procedure:**

- ess of the networked DSP that will be updated.

1. Use the PC connected to the Internet to browse to the web site listed above
2. Download the DSP software by right clicking on *dsp2_update.zip,* and choose *save target as*. Browse to a local folder on your computer to deposit it, then click *save.*
3. Unzip the files in the zip archive.
4. Connect the DSP to the LAN via the switch and Ethernet cable.
5. Launch the autoupdate software by navigating to the folder where it was unzipped / saved, and double clicking on the file *autoupdate.exe*.
6. A dialog box similar to the one shown below will appear:

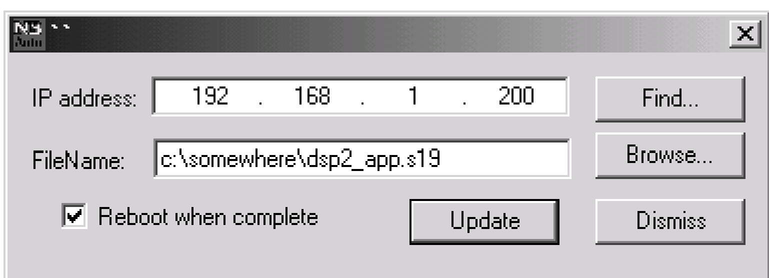


Figure 2-30 DSP Module Software Update Dialog

7. Enter the IP address of the DSP, the path to *dsp2_app.s19* (the update software), and insure that the *Reboot when complete* checkbox is checked.
8. Click the *Update* button, and a status bar will appear, showing the update progress.

9. After a short delay (10-15 seconds), the following dialog will appear: the DSP will be reset and restarted with the new software activated.

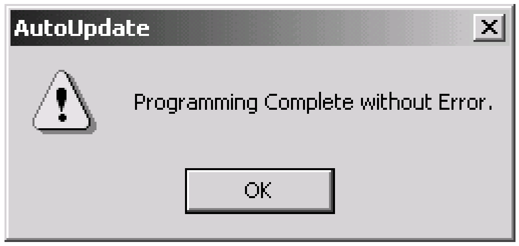


Figure 2-31 Successful Programming Announcement

10. Click OK to close the *AutoUpdate* dialog.
11. Verify the new version of the software has been loaded correctly by browsing to the IP address of the DSP, and validating the Firmware version matches the latest release (per the website).

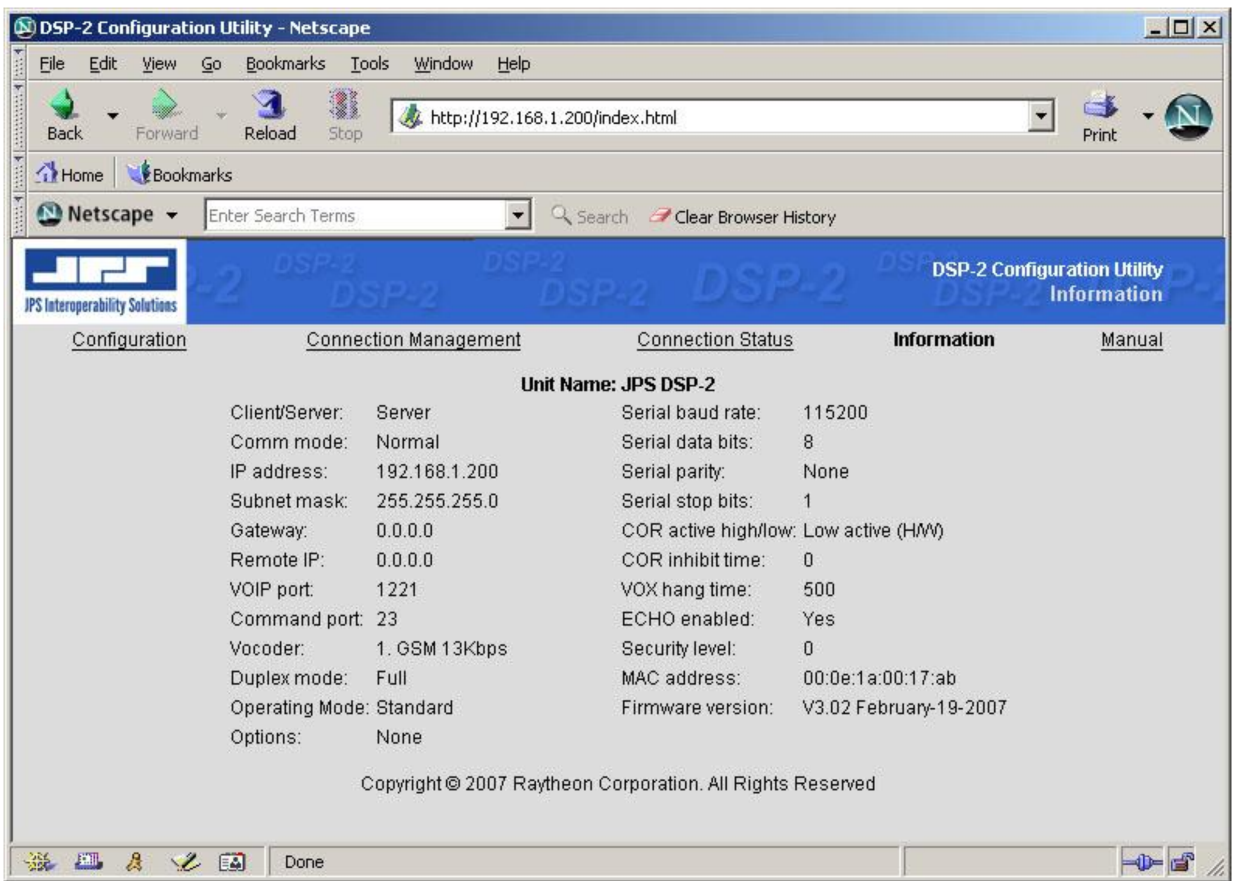


Figure 2-32 DSP-2 Module Information Page

12. This completes the process for updating the DSP Module software.

2.17.3 DSP-3 Configuration Programming via Browser

The DSP-3 configuration is the same as for the DSP-2 except for the minor variations as noted below.

2.17.3.1 DSP-3 Information Page

Upon successfully browsing to the DSP module, a page similar to Figure 2-33 will appear. This page contains a summary of the module's current operating status and configuration.

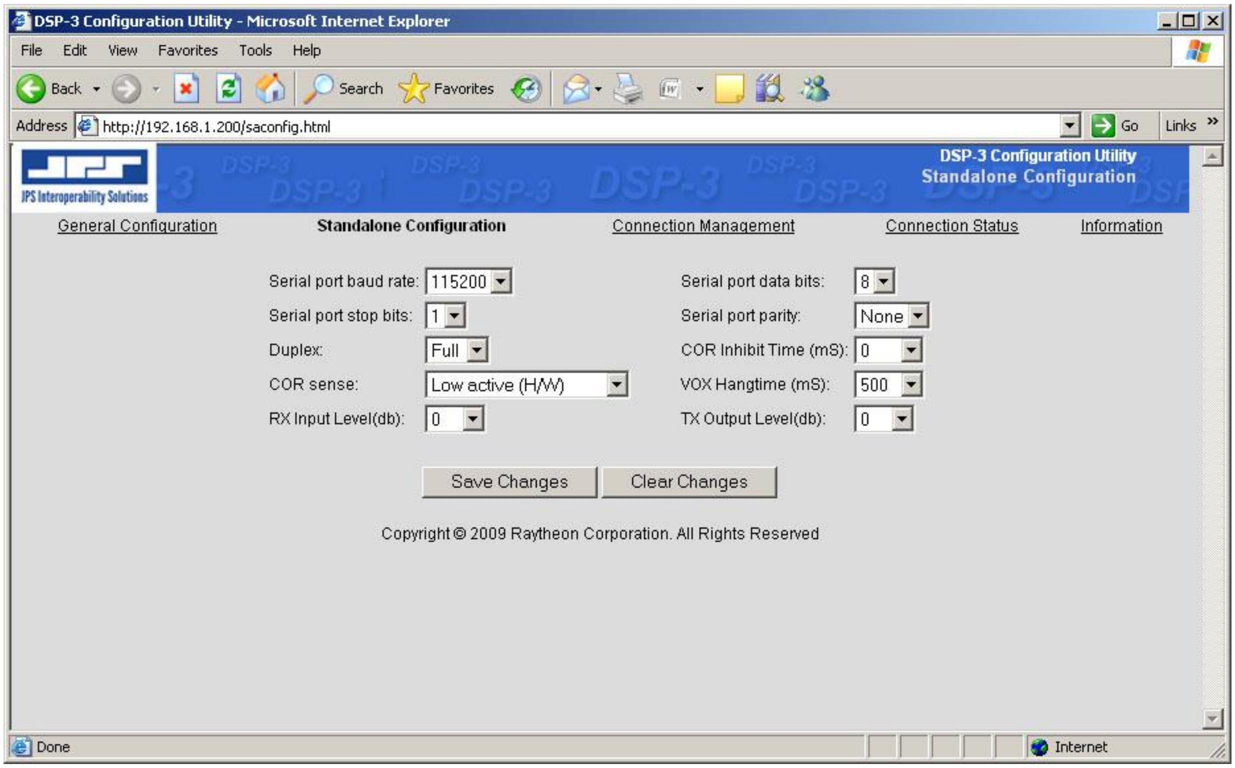


Figure 2-33 DSP-3 Information Page

2.17.3.2 DSP-3 Configuration Screens

The DSP-3 General Configuration Screen (Figure 2-34) includes all configuration items that are not dependent on the DSP-3 mode. Those configuration items that apply only to the Standalone Mode are found on a separate page as shown in Figure 2-35.

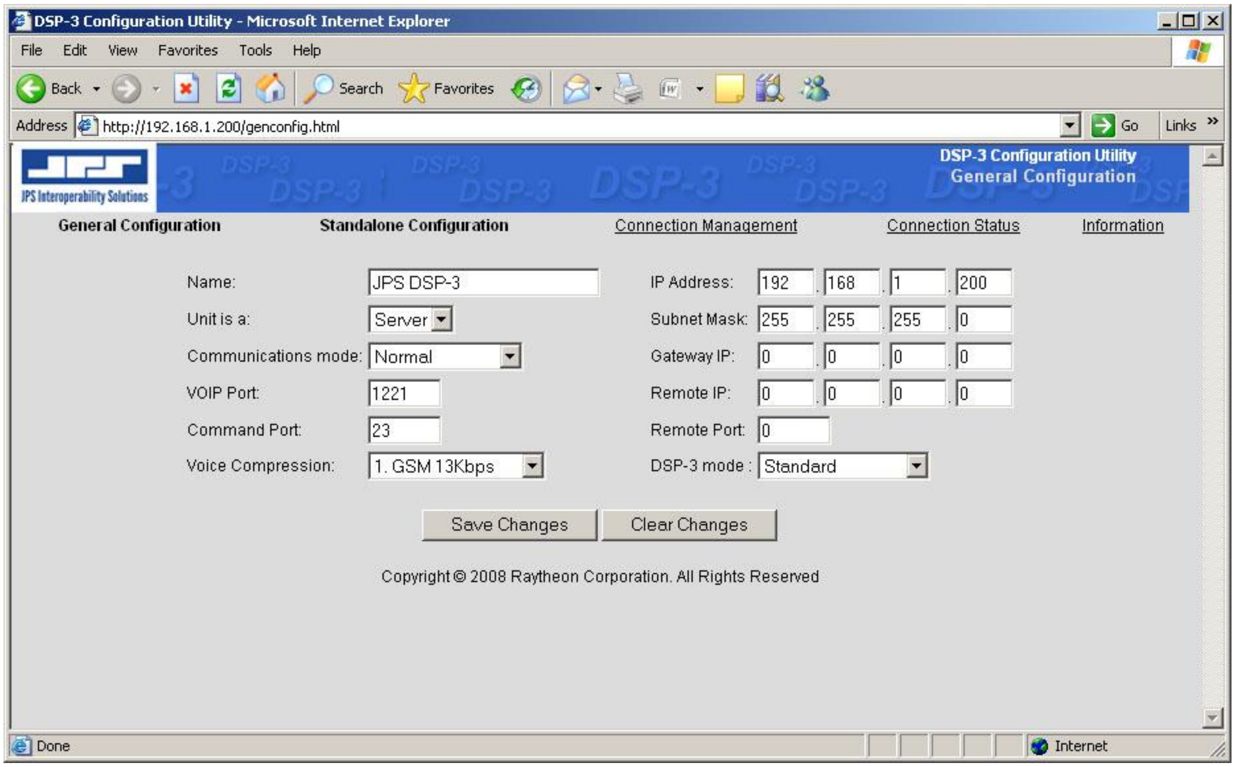


Figure 2-34 DSP-3 General Configuration Page

2.17.3.2.1 DSP-3 General Configuration Page Field Descriptions:

Name:

The user may enter text in this field that identifies this DSP module. The name should uniquely identify this unit, differentiating it from all other units on the network.

Unit is a:

The pull-down menu allows the user to choose either *server* or *client* mode. VoIP operation requires pairing of clients and servers.

Communications Mode:

The pull-down menu allows the user to configure for either *normal*, *broadcast*, *connectionless*, or *multicast* modes of VoIP communications.

VOIP Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use for the VoIP traffic.

Command Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use for network command traffic.

Voice Compression:

The pull-down menu allows the user to configure the voice compression (vocoder) that is used for VOIP audio traffic. Five choices are available: GSM at 13Kbps, ADPCM at 16Kbps, ADPCM at 24Kbps, ADPCM at 32Kbps, and PCM at 64Kbps. If the unit is a server, adaptation to the incoming client vocoder is automatically selected to match the client request.

IP Address:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the unit.

Note: Upon saving the changes, the user will need to browse to the new address to continue configuration.

Subnet Mask:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol mask of the unit.

Note: Upon saving the changes, the user will need to browse to the new address to continue configuration.

Gateway IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the gateway address that the unit will use for resolving external network accesses.

Remote IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the remote VoIP unit that is paired with this unit.

Remote Port:

The user may enter numeric text in this field (1-65535 decimal) defining which port to use when connecting to its paired unit for VoIP traffic.

DSP Mode:

Four modes are supported (see the beginning of Section 2.17.2 for an explanation):

	Mode	Interfaces Used
1.	Standard Mode:	Rear panel four-wire & chassis backplane
2.	VoIP Standalone Mode:	Front panel RJ45 and rear panel four-wire connector
3.	VoIP Hybrid Mode:	Front panel RJ45 and chassis backplane
4.	Test:	test mode: for factory use only with loopback fixture.

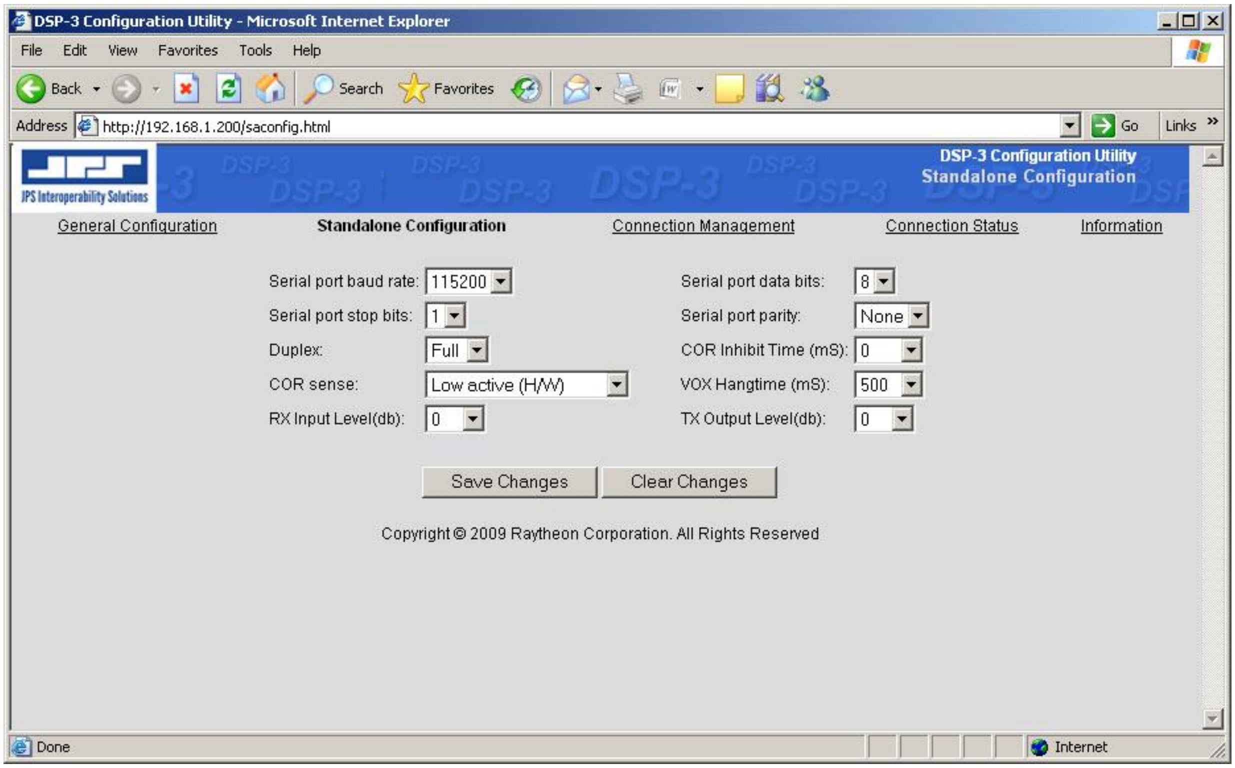


Figure 2-35 DSP-3 Standalone Mode Configuration Page

2.17.3.2.2 DSP-3Standalone Mode Configuration Page Field Descriptions:

These Configuration items apply to VoIP Standalone mode only. The method for configuration of these parameters when the module is in the Standard and VoIP Hybrid modes is via the ACU Controller, which includes additional tools to assist proper set up.

Serial Port Baud Rate:

The pull-down menu allows the user to configure the baud rate for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Nine baud rates are available: 300, 1200, 2400, 4800, 9600, 19200, 38400, 57600, and 115200.

Serial Port Stop Bits:

The pull-down menu allows the user to configure the number of stop bits for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Two choices are available: 1 or 2 bits.

Duplex:

The pull-down menu allows the user to configure the radio interface for either full or half duplex operation.

COR Sense:

The pull-down menu allows the user to configure the radio COR sense. Three options are available: active low, active high, and VOX.

RX Input Level:

The pull-down menu allows the user to adjust the input level for radio receive audio. Five levels are supported: +12db, +4db, 0db, -4db, and –12d.

Serial Port Data Bits:

The pull-down menu allows the user to configure the number of data bits for the serial communication channel on the DB-15 connector of the ACU-2000 slot. Two choices are available: 7 or 8 bits.

Serial Port Parity:

The pull-down menu allows the user to configure the parity for the serial communication channel on the associated rear-panel DB-15 connector. Three choices are available: even, odd, or none.

COR Inhibit Time (ms):

The pull-down menu allows the user to configure the radio interface COR inhibit time. Six options are available: 0ms, 500ms, 1000ms, 2000ms, 3000ms and 4000ms.

VOX Hangtime (ms):

The pull-down menu allows the user to configure the VOX hangtime in milliseconds. Five hangtimes are available: 500ms, 1000ms, 2000ms, 3000ms, and 4000ms.The hangtime is applied to the radio RX audio.

TX Output Level:

The pull-down menu allows the user to adjust the transmit output level for the radio transmit audio. Five levels are supported: +12db, +4db, 0db, -4db, and –12db.

2.17.3.3 DSP-3 Connection Management Page

Client VOIP sessions may be managed by browsing to the **Connection Management** link at the top of any of the unit's web pages. Figure 2-36 shows the Connection Management page. Functions identically to the DSP-2 Connection Management feature.

This page may be used force the use of a different server, or for connecting or disconnecting an existing link to a server.

Server IP:

The user may enter numeric text (0-255 decimal) into each of the four fields that define the unique Internet Protocol address of the server unit that is paired with this unit.

Connect / Disconnect:

The user may request that a connection to the server be established by activating the *CONNECT* button.

The user may request that the connection to the server be broken by activating the *DISCONNECT* button.

Note: Requests will be processed only after activating the Perform Selected Action button.

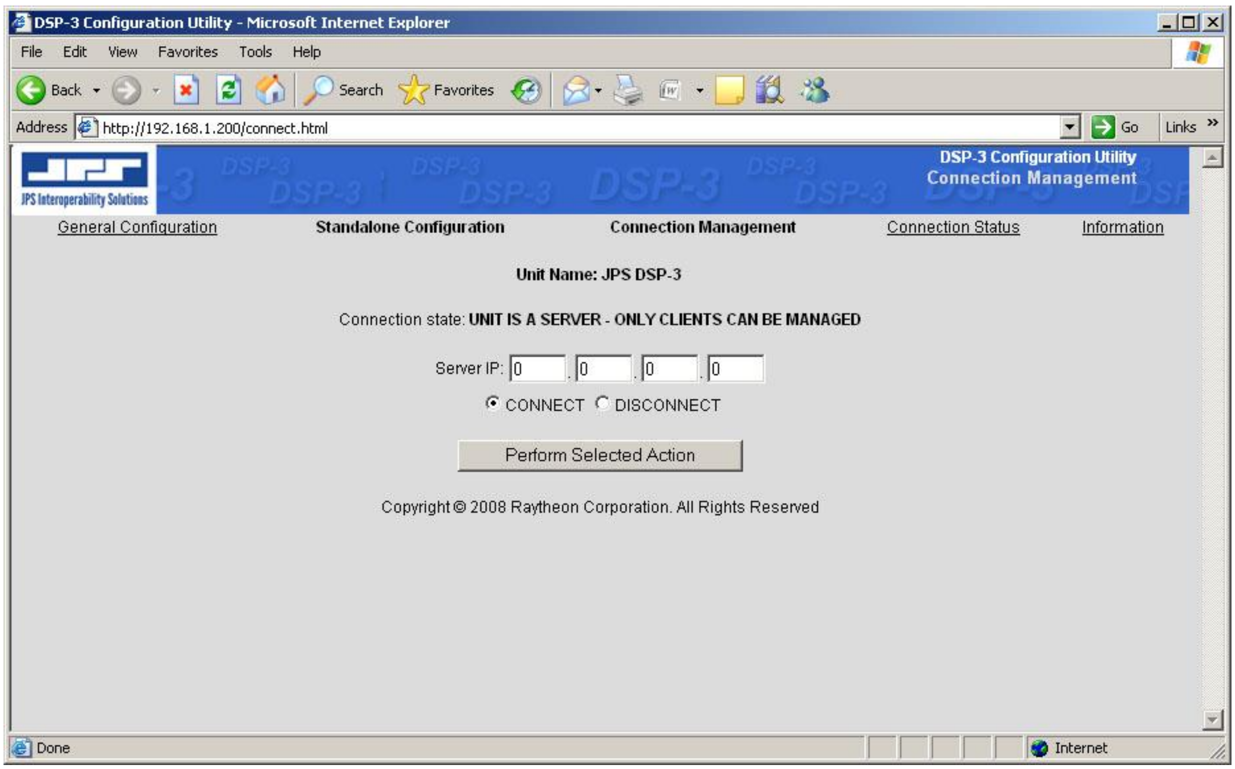


Figure 2-36 DSP-3 Connection Management Page

2.17.3.4 Connection Status Page

VOIP session status may be monitored by browsing to the **Connection Status** link. A variety of statistics for the session are presented (see Figure 2-37).

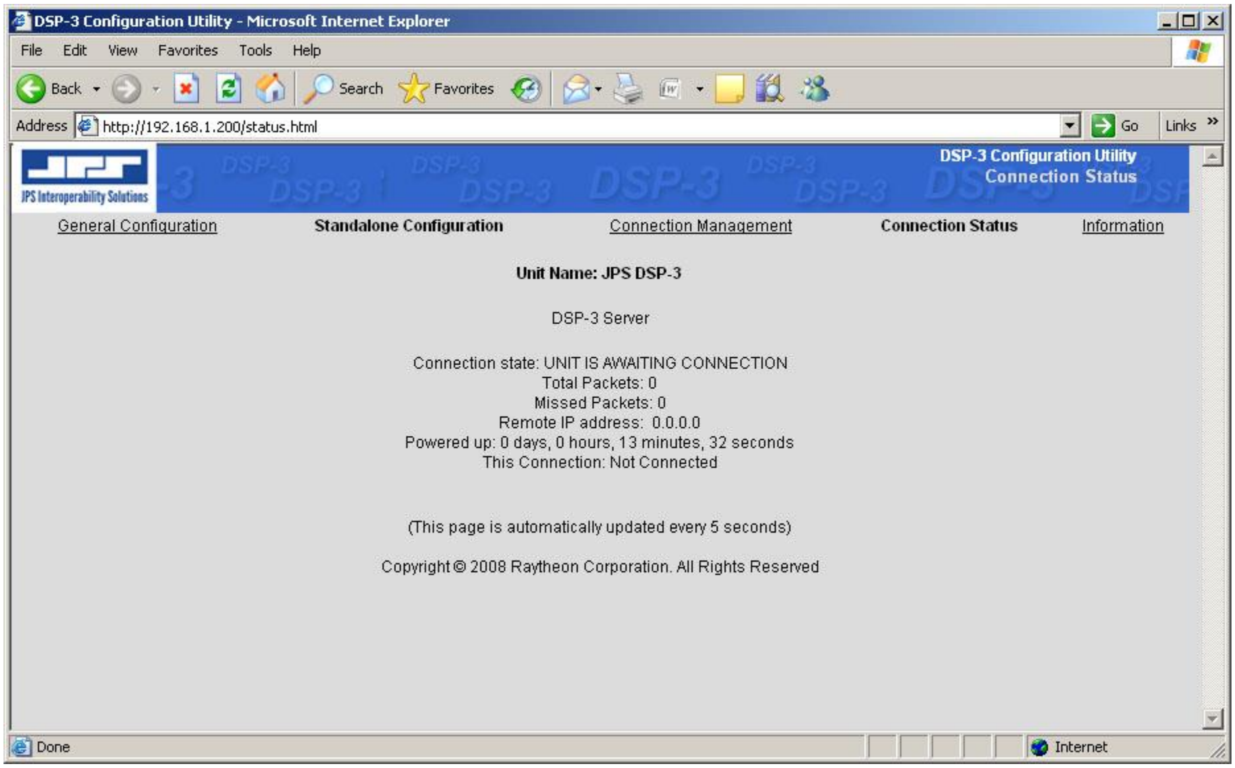


Figure 2-37 DSP-3 Connection Status Page

2.17.3.5 Restoring Factory Defaults

In rare circumstances, there may be the need to completely restore the DSP-3 to the original configuration that was set when the module was manufactured. The procedure for doing this is described in Section 2.17.2.10. The same procedure applies to the DSP-2 (Does not apply to four-wire interface configuration options).

2.17.3.6 Software Updates

The DSP-3 is designed to support software updates in the field. Should it become necessary to install updates to the software, the process described in Section 2.17.2.11 should be followed. If DSP-2 software is updated into the DSP-3 module, it will no longer provide the DSP-3 to WAIS Controller 2 Dispatch communications capability. DSP-3 software cannot be uploaded into a DSP-2 module.

2.17.4 SCM-2 Configuration Programming via Browser

This section addresses the configuration of the module's Ethernet port. Configuration is accomplished via PC / Browser / Ethernet web access. To configure connect the SCM-2 to the user's Ethernet LAN via the front panel RJ-45 Ethernet connector, via a "straight" Ethernet cable to a switch or router which is also connected to a PC with network access to the same switch or router. A PC connected directly to the SCM-2's Ethernet port will also work, but the PC must be pre-configured to have a static IP address.

Next browse (using a web browser such as *Internet Explorer*) to the module's IP address. The default address as shipped from the factory is 192.168.1.200. (After provisioning, this IP address may be changed to comply with user requirements).

2.17.4.1 SCM-2 Configuration - Status/Information Page

Upon successfully browsing to the module, a Status/Information page similar to Figure 2-38 will appear. It shows network settings and other status information.

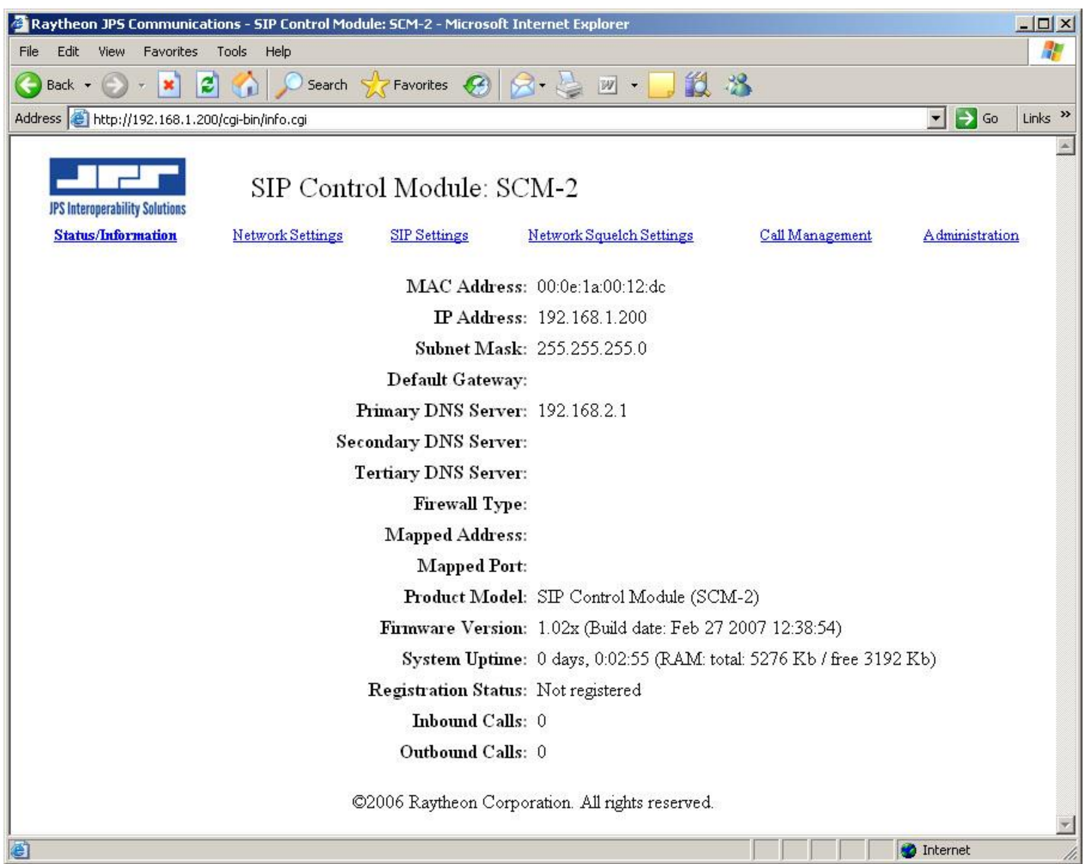


Figure 2-38 SCM-2 Status/Information Page

2.17.4.2 SCM-2 Configuration - Network Settings Page

To next step is to configure the module's Network Settings by selecting the appropriate link at the top of the page (see Figure 2-39).

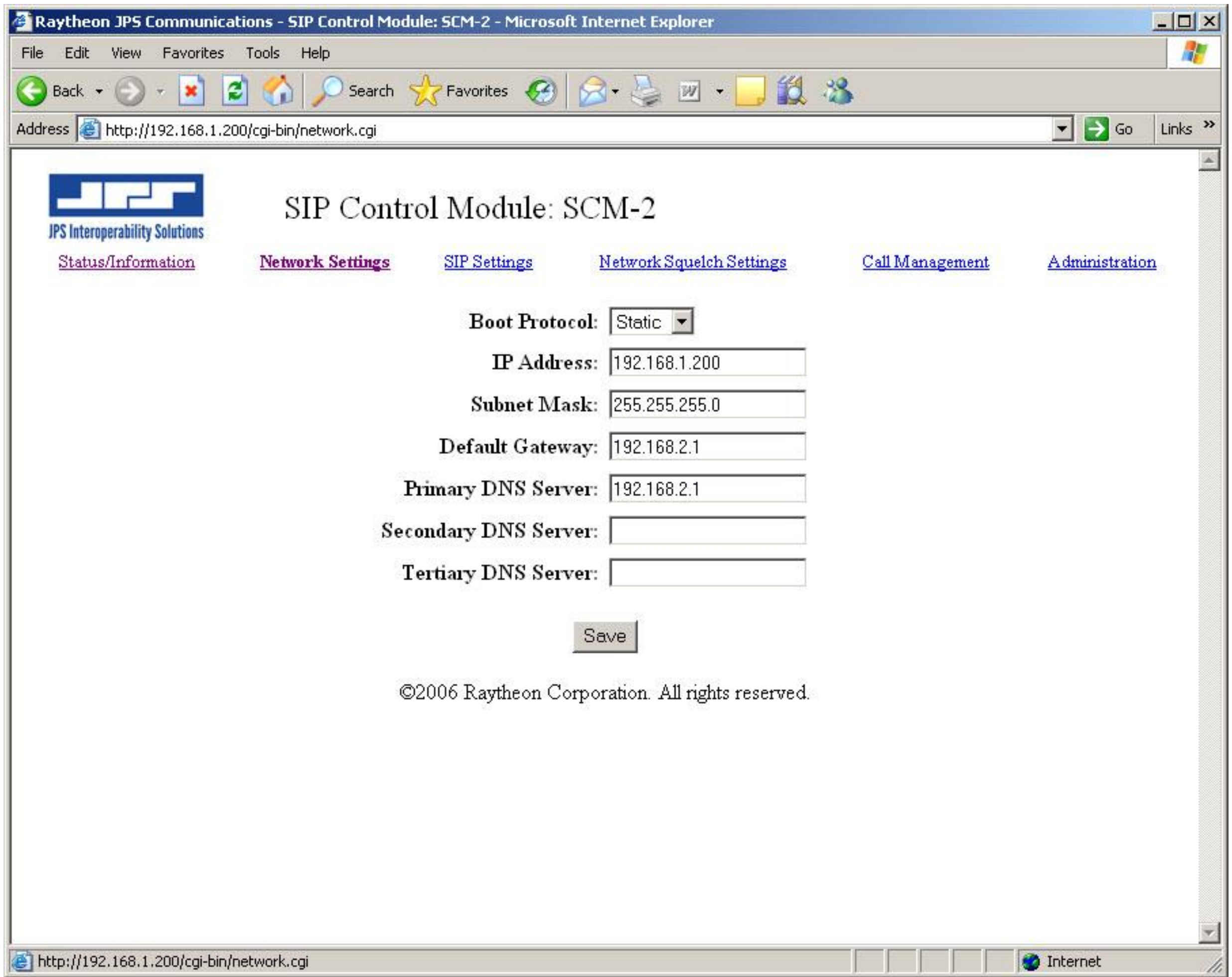


Figure 2-39 SCM-2 Network Settings Page

If you select *Static* for the Boot Protocol then you must set the other settings to match your particular network. If you use *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.

> ***Note: For any operational changes to take effect, you must save changes via the* Save *button.***

2.17.4.3 SCM-2 Configuration - SIP Settings

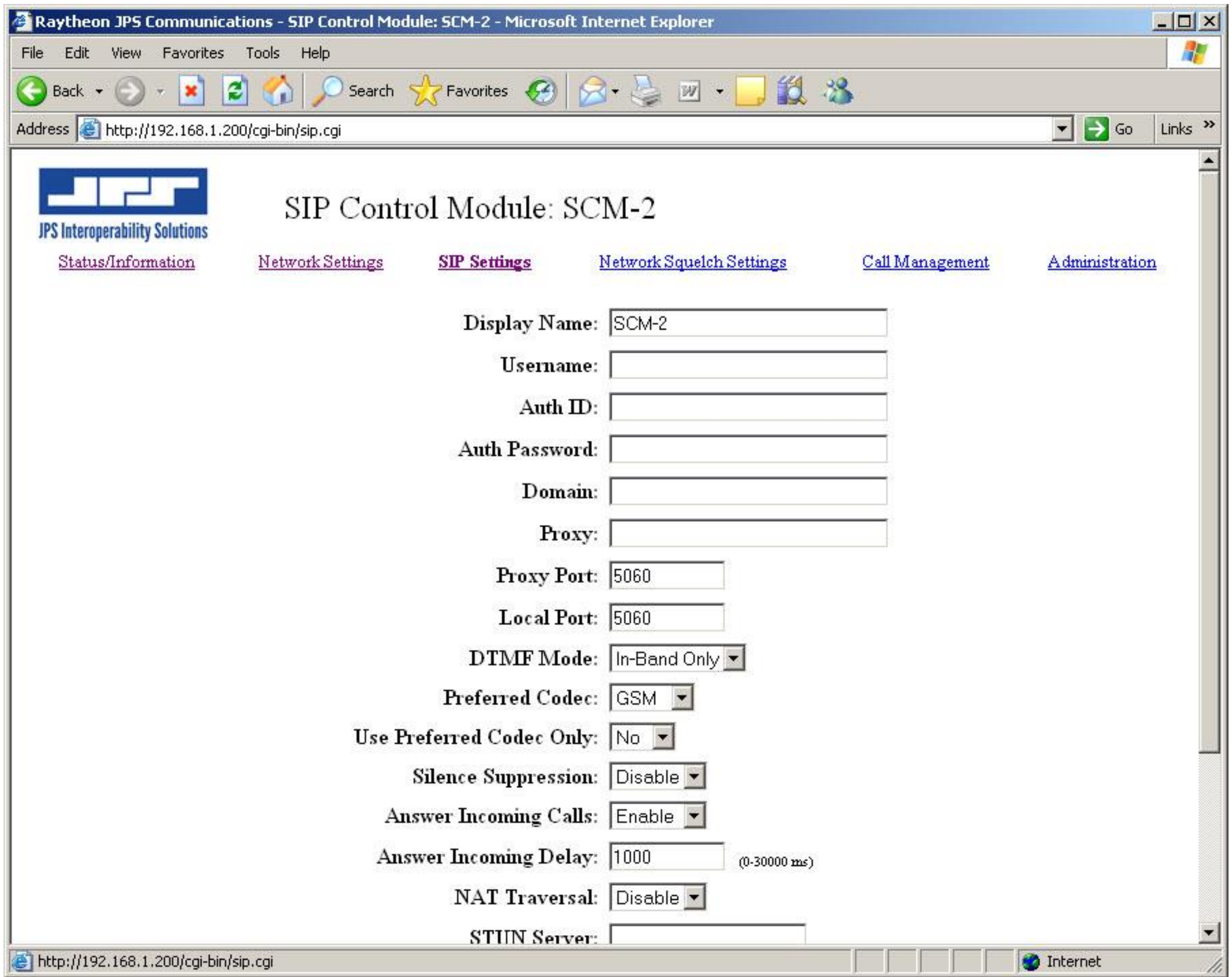


Figure 2-40 SCM-2 SIP Settings Page

The SIP Settings page is used to configure the module's SIP settings. The SCM-2 can register with a SIP proxy or can create a one-to-one SIP session by IP direct dialing.

This page contains the same settings that would be set for any SIP endpoint such as a SIP Phone. A major exception is the Send Radio COR/AUX Status setting. If you enable this feature the status of the COR and AUX Input pins on the rear panel of a DSP module cross-connected to the SCM-2 module 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 JPS SIP interfaces, such as other SCM modules or 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 robust and timely method than the use of VOX or VMR as the network audio gating function at the other SIP interface units.

Table 2-18 SIP Settings Options	
Settings Option	**Description**
Display Name:	The name displayed on a remote SIP Phone when it connects to the SCM-2.
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 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:	Enables/disables 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:	Enables/disables 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:	Enables/disables the sending of SIP keep-alive packets.
Keep Alive Interval:	Interval in seconds to send SIP keep-alive packets.
DTMF Mode:	Mode to use for sending DTMF during a call.
Block DTMF In-Band:	Blocks DTMF in the audio stream when using a DTMF Mode other than *In-Band*.
Preferred Codec:	The voice compression type the SCM-2 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 be sure to click *Save* at the bottom of the page.

2.17.4.4 SCM-2 Configuration - Call Management

This page allows the user to initiate a call from via a web browser or check ongoing call status. It's actually more of an operations function page that can be called up regularly to manage calls, rather than a configuration page that is typically used only at initial system configuration. It is included here for clarity.

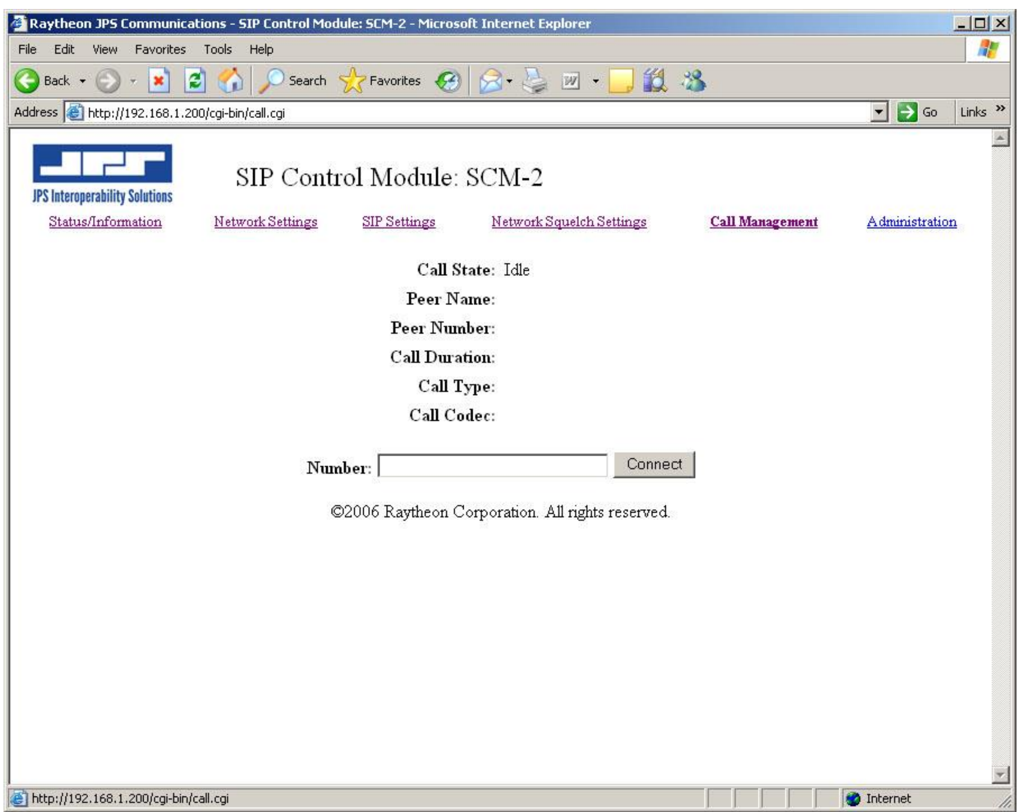


Figure 2-41 SCM-2 Call Management Page

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 refresh in your browser to see the results of the call request.***

2.17.4.5 SCM-2 Configuration - Administrative Functions

This page allows password protection of access to the SCM-2 module's web pages, facilitates upgrades to the module's firmware, and provides a means to remotely reboot the module.

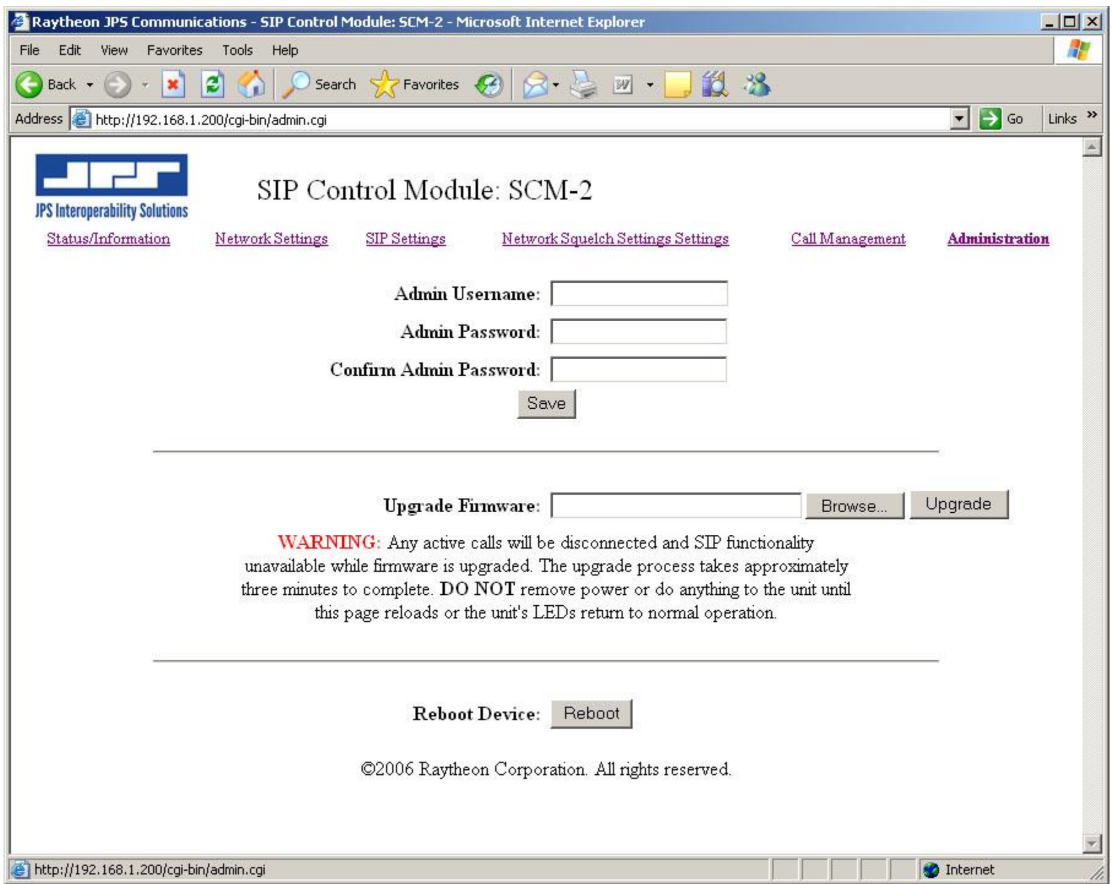


Figure 2-42 SCM-2 Administration Page

2.17.4.5.1 Password Protection

Enter a username and password and click *Save* to password protect the SCM-2's web pages; once a password is entered, access is denied unless the proper password is entered. If you forget the password the only way to restore access is to reset the unit to factory defaults (see Section 2.17.4.5.2 which follows).

2.17.4.5.2 Restoring Factory Defaults

In rare circumstances, there may be the need to completely restore the SCM module to the original configuration that was set when the module was manufactured. The procedure for doing this follows.

Procedure:

1. Power down the ACU-2000.
2. Remove the SCM module from the chassis.
3. Install the extender car d (supplied in the Accessory Kit) into the now empty slot.
4. Install the SCM into the extender card.
5. Power up the ACU.
6. Configure the *Restore Factory Defaults* jumper **JP22 [center to right].**
7. Wait 10 seconds.
8. Restore the *Restore Factory Defaults* jumper JP22 [left to center]
9. Wait 15 seconds (the SCM module will complete the reset sequence).
10. Power off the ACU.
11. Remove the SCM from the extender card.
12. Remove the extender card from the ACU chassis.
13. Install the SCM back into the vacant slot in the ACU.
14. Power up the ACU and you are finished.

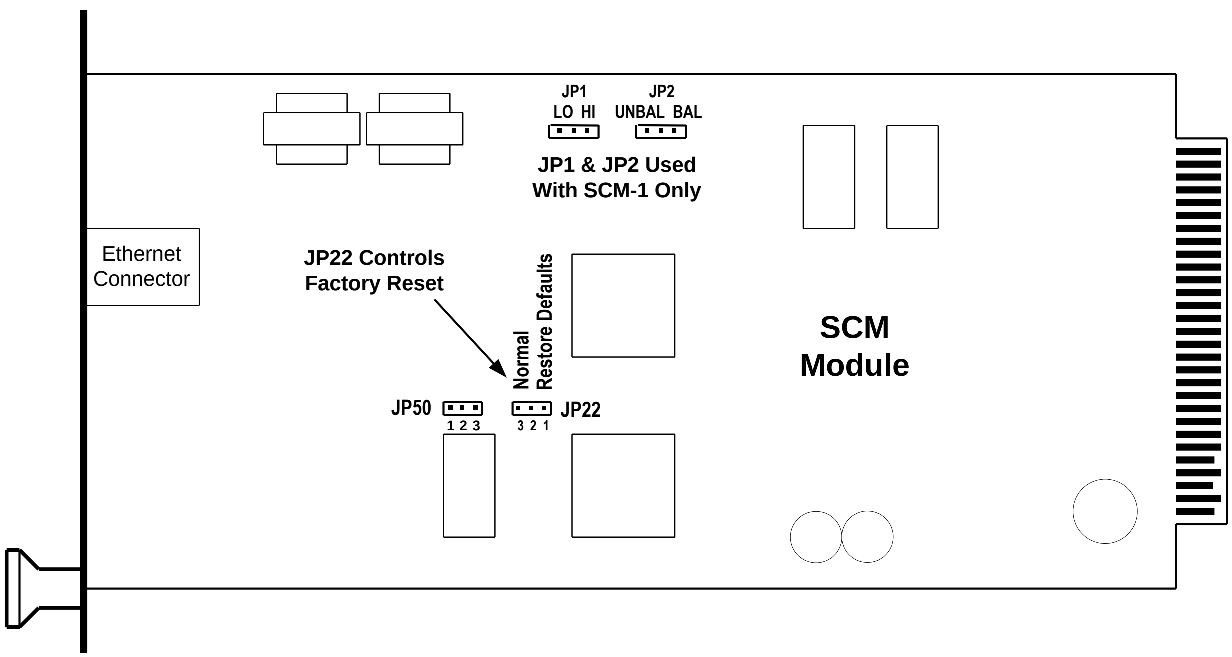


Figure 2-43 Jumper for Factory Default Restoration

2.17.4.5.3 Software Update

The SCM is designed to support software updates in the field. Should it become necessary to install updates to the software, the following process should be followed. Instructions and the update software are available on the web at:

http://www.jpsinterop.com

or via email at: support@jpsinterop.com.

On the page all ACU-2000 related downloads are listed. This includes this manual, the ACU Controller, and latest versions of software for the ACU-2000 modules. Export regulations require that the form supplied on the website after a download request be filled out prior to enabling the download.

For further instructions, see Section 2.17.2.11 for instructions on updating the DSP module, which covers the software update process in more detail. The SCM module update procedure is identical.

2.17.4.5.4 Remote Reboot of the SCM-2 Module

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

2.17.5 SCM-1 Configuration Programming via Browser

Unlike the SCM-2 and all other ACU interface modules, SCM-1 operation does not include any interoperability between ACU modules. This section therefore addresses the configuration of the module's two external interfaces: front panel Ethernet connector and its rear panel D-sub radio connector. Configuration is accomplished via PC / Browser / Ethernet web access. To configure, connect the SCM-1 to the user's Ethernet LAN via a "straight-through" CAT 5 cable to a switch or router which is also connected to a PC with network access to the same switch or router. A direct Ethernet connection to a PC via crossover cable will also work as long as the PC has been preconfigured to have a static IP address.

Next browse (using a web browser such as *Internet Explorer*) to the module's IP address. The default address as shipped from the factory is 192.168.1.200. After provisioning, this IP address may be changed to comply with user requirements.

2.17.5.1 SCM-1 Configuration - Status/Information Page

Successfully browsing to the module will bring a page similar to Figure 2-44.It shows the current network settings as well as some other status information, such as the version of the firmware currently loaded in the SCM-1.

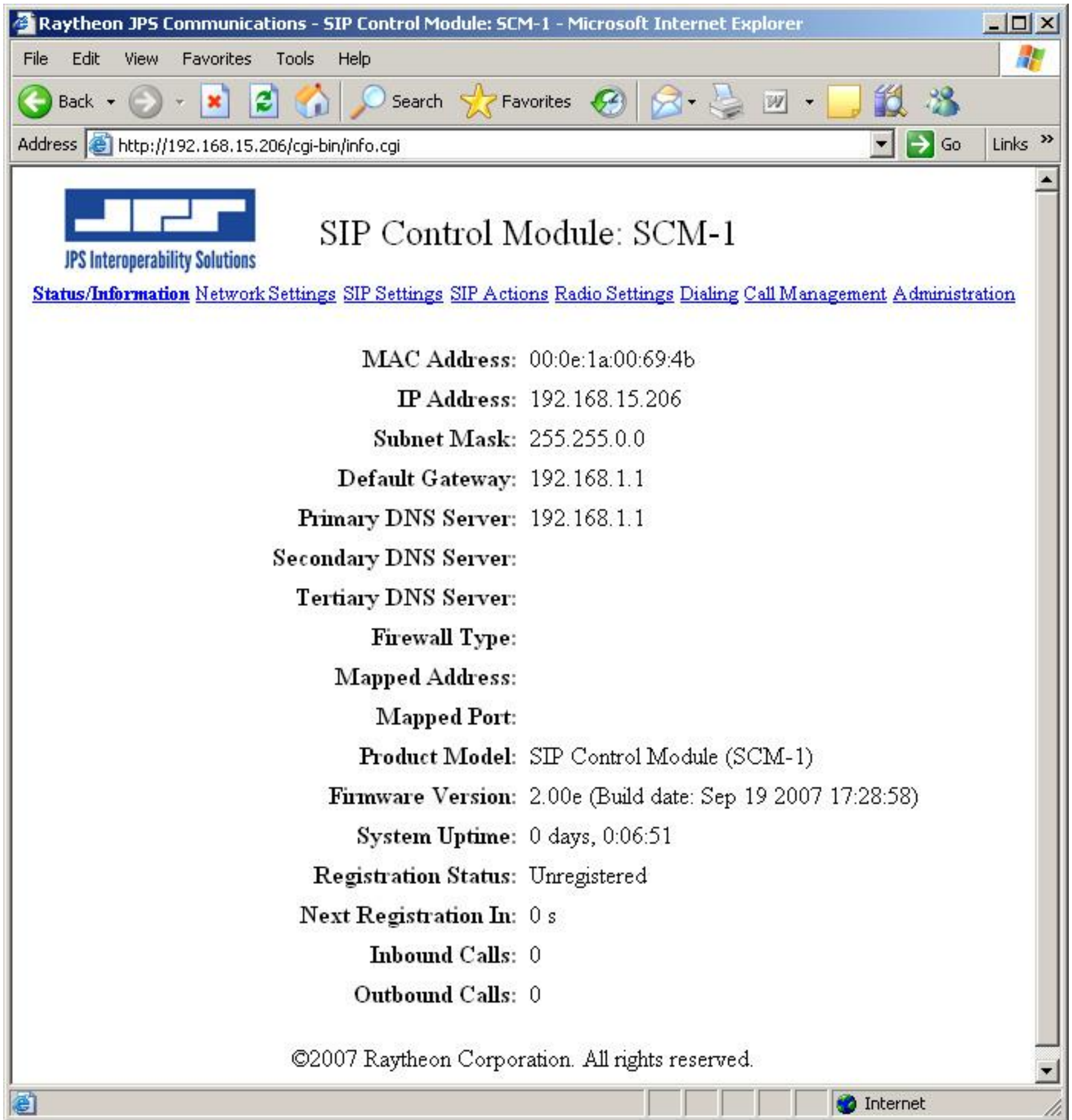


Figure 2-44 SCM-1 Status/Information Page

2.17.5.2 SCM-1 Configuration - Network Settings Page

To next step is to configure the module's Network Settings by selecting the appropriate link at the top of the page.

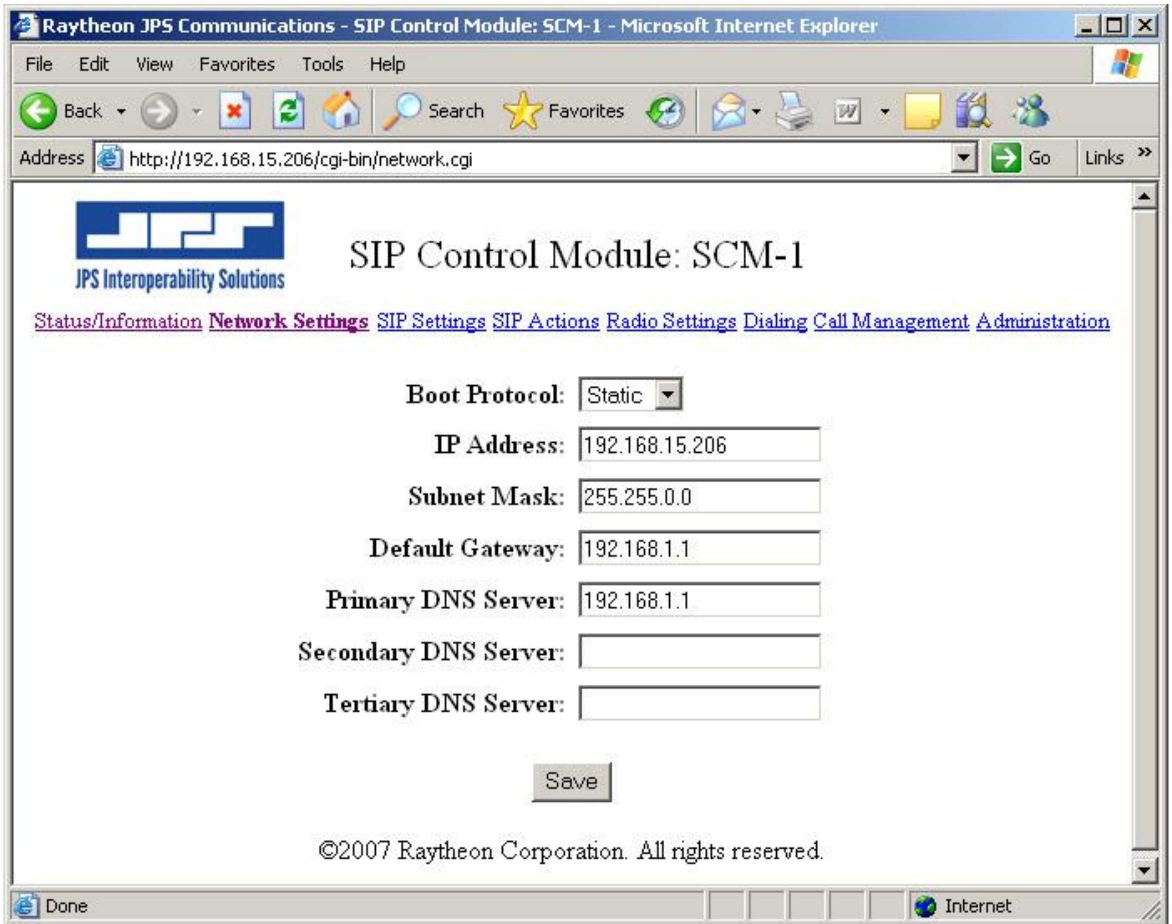


Figure 2-45 SCM-1 Network Settings Page

If you select *Static* for the Boot Protocol then you must set the other settings to match your particular network. If you use *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.

Note: For any operational changes to take effect, you must save changes via the Save button.

2.17.5.3 SCM-1 Configuration - SIP Settings

This page allows SIP settings configuration. The SCM module can register with a SIP proxy or can use IP direct dialing to create an individual SIP-to-SIP connection.

The page contains the same settings that would be set for any SIP endpoint such as a SIP Phone. There's one exception, and that's the COR/AUX Status setting. If you enable this feature the status of the COR input that of the modules AUX Input pins will be sent across the IP link. This is useful if the SCM links to another JPS SIP interface (such as other SCM modules or ARA-1 units) that interfaces a remote radio. Sending COR Status will assert the PTT control output lines to that remote radio. This is a more robust and timely method than the use of VOX or VMR as the network audio gating function at the other SIP-radio interface.

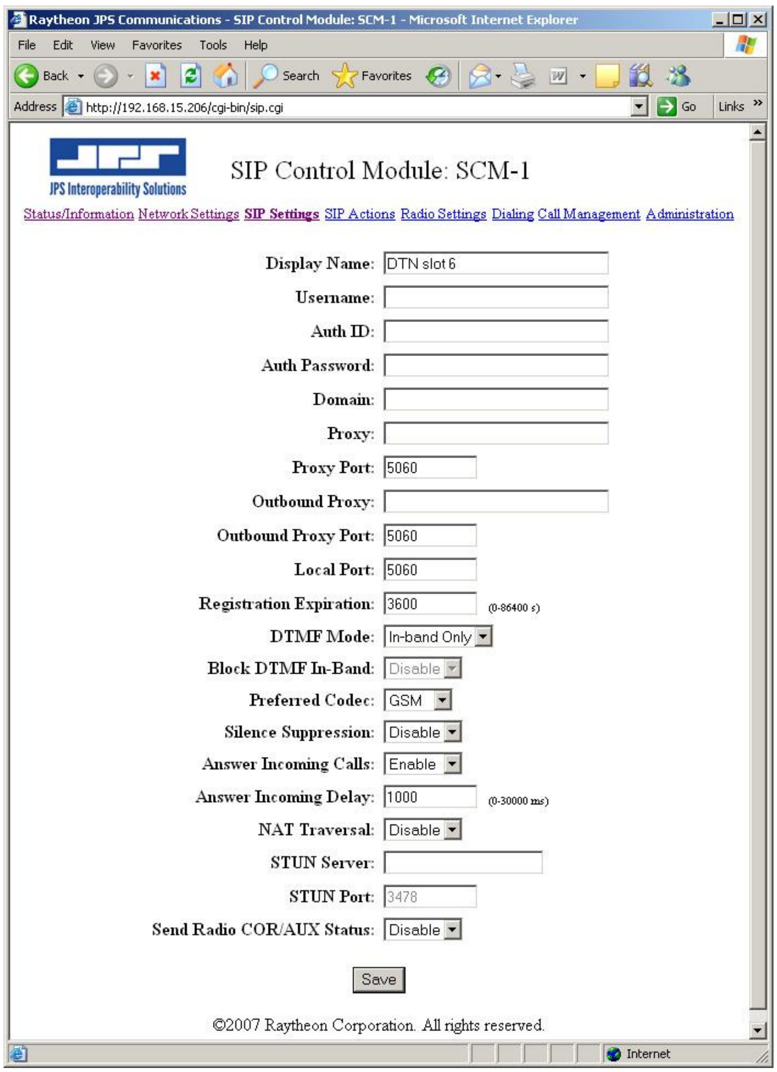


Figure 2-46 SCM-1 SIP Settings Page

Table 2-12 SIP Settings Options	
Settings Option	**Description**
Display Name:	The name displayed on a remote SIP Phone when it connects to the SCM-1.
Domain:	The unit's SIP domain (if needed). The domain portion of the unit's URI.
Username:	The SIP username 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 is 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:	Enables/disables 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:	Enables/disables 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.

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 SCM-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.

2.17.5.4 SCM-1 Configuration – SIP Actions

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

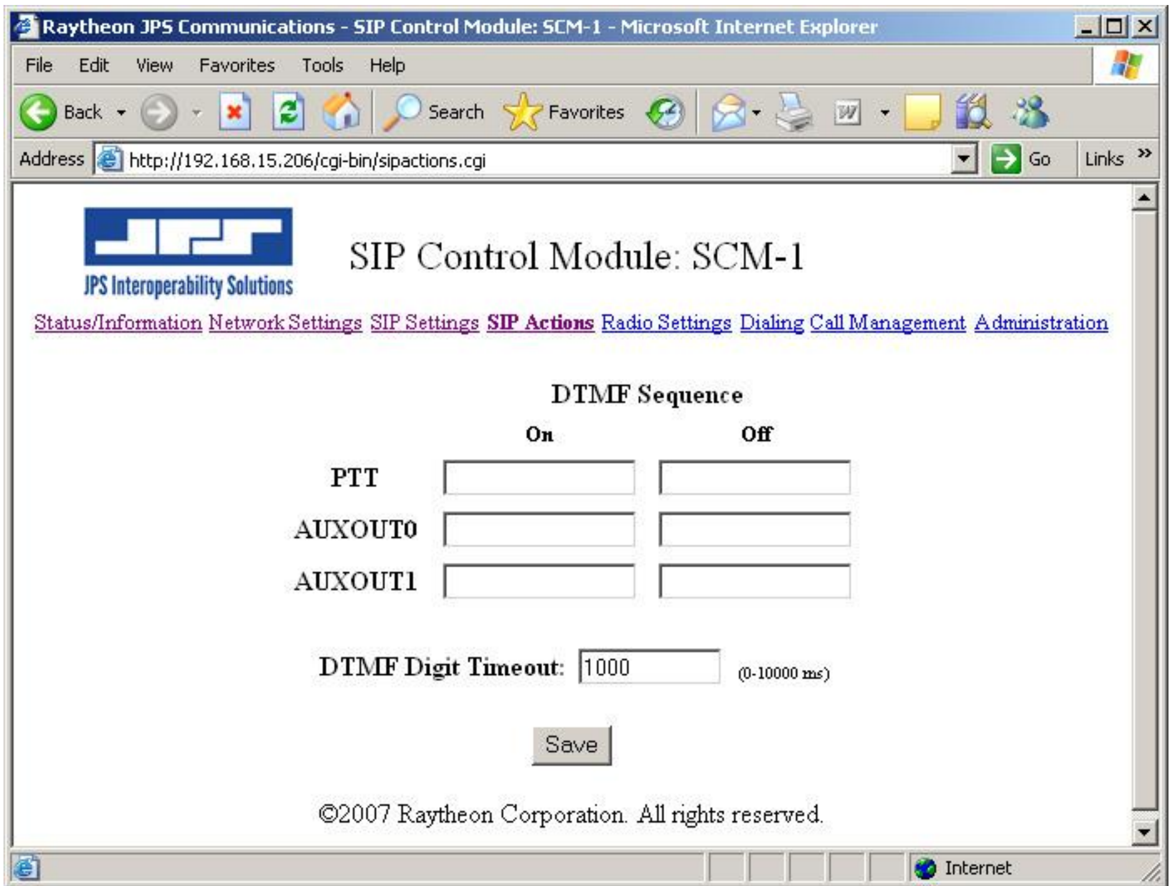


Figure 2-47 SIP Actions Page

The DTMF Digit Timeout entry determines how the SCM-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's longer than the timeout entry, the current DTMF sequence will be considered complete. The factory default setting - one second duration (1000 ms) - should work for most systems.

In order to control the PTT output of the SCM-1 using SIP Actions, the *Network COR Type* setting on the *Radio Settings* page must be set to *SIP Actions*.

2.17.5.5 SCM-1 Configuration - Radio Interface

The *Radio Settings* page is used to configure the SCM-1's radio interface. Included in the SCM-1's Radio Settings are all of the settings that affect the interface to, and operation of, the radio cabled to the ACU rear panel. This page is used to configure and optimize the unit for best performance in a particular radio application. Each of the settings is explained below; the default settings are shown in the figure.

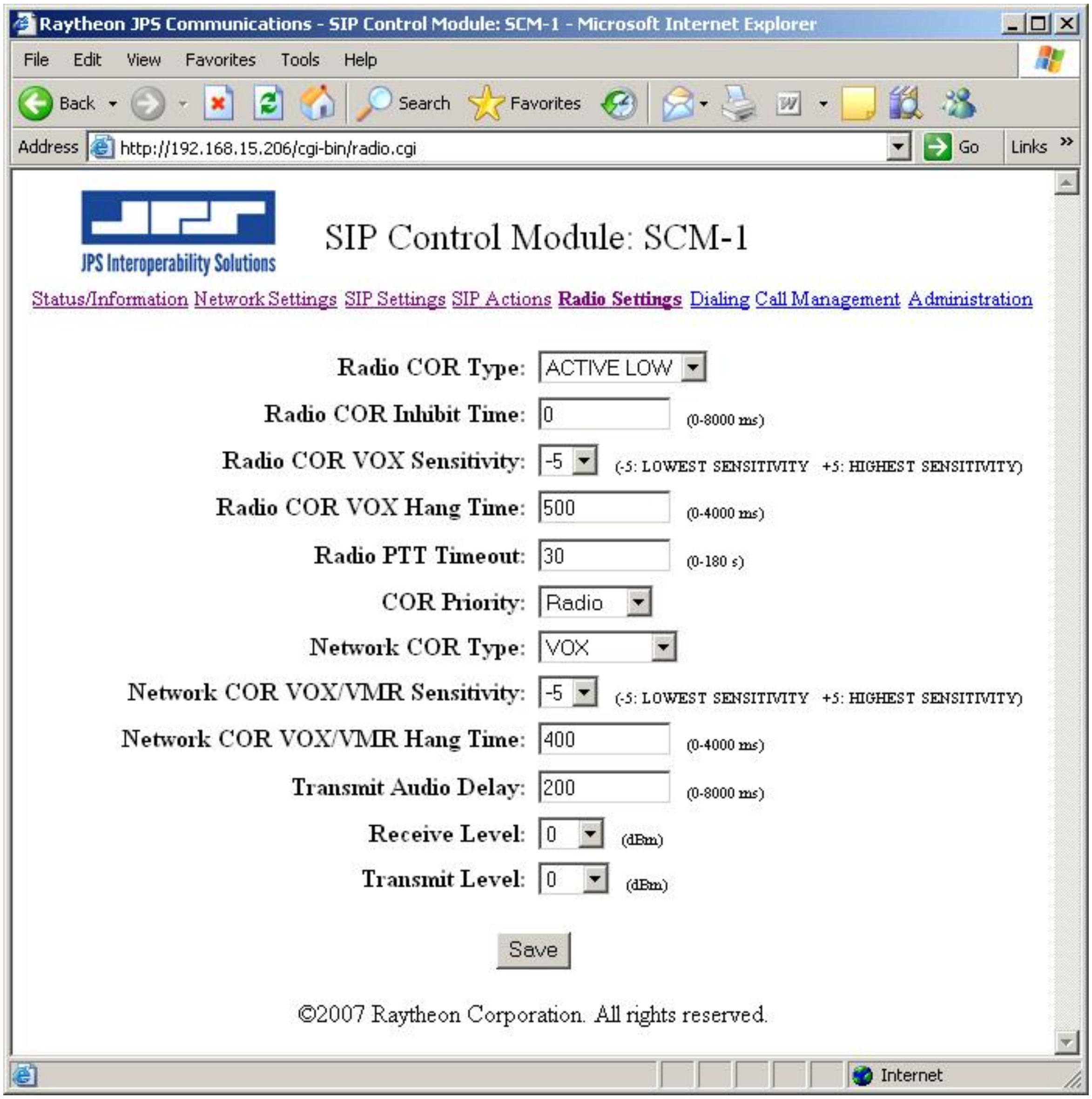


Figure 2-48 Radio Settings Page

2.17.5.5.1 Radio COR Settings Options

The SCM-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.

2.17.5.5.2 Radio COR Type

This determines which method the SCM-1 will use to determine when the radio is receiving a valid signal and the radio/SCM-1 pair will be put into the unsquelched mode [also called "open squelch"] and send the radio RX audio to the SIP network. The SCM-1 can either use the COR output signal from the radio, or use VOX (see below). The COR output is a control signal from the radio which activates when the squelch opens. If this line is available, connect it to the SCM-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 doesn't have a COR output control line, select VOX. When the SCM-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 below). 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 from the radio, and if so, whether its sense is active high or active low. Sometimes programming or other changes are required to activate this signal line- if so, this is indicated in the Application Notes.

2.17.5.5.3 Radio COR Inhibit Time

In some radios the COR line activates momentarily when the radio reverts to receive mode from transmit mode. 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 SCM-1 includes a provision to ignore the COR signal for a specified period of time following the cessation of a TX sequence. In many cases it's not needed.

The SCM-1 front panel's *COR* LED is lit whenever the SCM-1 has detected active COR from the radio 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 the troubleshooting information of Section 4.5.

2.17.5.5.4 Radio COR VOX Sensitivity

This 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. See Section 2.17.5.6 regarding proper audio level adjustment.

2.17.5.5.5 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 hang time to the lowest level that does not create inappropriate resquelching. The SCM-1 front panel *COR* LED is lit whenever the SCM-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 hangtime must be increased.

2.17.5.5.6 Radio PTT Timeout

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

For example, if a user is connected to an SCM-1 with a SIP Phone and the SCM-1 is configured to use VMR as the Network COR Type, an overly sensitive SIP Phone microphone and loud background voices can cause the SCM-1 to key indefinitely preventing communications back from radio users. The PTT Timeout will then trigger after the set amount of time has passed. Furthermore, the SCM-1 will not let PTT the radio again due to SIP audio until there is a break in the network COR signal generated by VMR.

2.17.5.5.7 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. This 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 SCM-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.

2.17.5.5.8 Always Pass Audio

By default, the SCM-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 full-duplex systems. Use this option to enable audio pass through.

2.17.5.5.9 Network COR Settings Options

These settings define how & 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 **Error! Reference source not found.**).

2.17.5.5.10 Network COR Type

This function is similar to *Radio COR Type* except that there are more options. This setting tells the SCM-1 how to determine when there is a signal coming from the SIP network that 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 next three settings may be useful if multiple JPS SIP interfaces (ARA-1 units or SCM-1/SCM-2 modules) are integrated in the SIP network. These additional COR types make use of an RTP extension header that sends the unit's COR status to other JPS SIP Interfaces . This is helpful as these other interfaces will receive a positive indication of COR status that arrives coincident with the radio audio and they will not have to derive the COR status using a VOX or VMR function. A received COR input is translated into a PTT output to the associated radio, and is a quicker and more robust method way to determine when to key a transmitter. For these settings to have any utility on the unit being set up, the JPS SIP interface device that the SCM-1 is linked to must have its SIP Settings option *Send Radio COR/AUX Status* enabled.

To further clarify: When Radio A is linked to Radio B over a SIP network via a pair of JPS SIP interface devices, 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 SIP interface device associated with Radio A has the *Send Radio COR/AUX Status* function enabled, it will sends its COR status over the network as part of the RTP extension header. The SCM-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 JPS SIP interface devices. 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 SCM-1 will ignore network audio from other devices not interfaced by a SIP interface that has its *Send Radio COR/AUX Status* function enabled (for example, from a standard SIP Phone). If these other devices will be used, set the SCM-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 SIP interfaces and also properly link with devices that don't send this information. When either of these modes is selected, the SCM-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 to select 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 some other SIP device may be using this header for some other purpose. Two conditions may result:

(1) An incompatible SIP device misinterprets the COR/AUX extension header sent from the SCM-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 sends a non-standard RTP extension header that is misinterpreted by the SCM-1. This causes the SCM-1 to signal the radio to key inappropriately.

Use the following guidelines when deciding how best to apply the use of the RTP Header:

- When all SIP interfaces are JPS devices, use RTP Header as the Network COR type in all SCM-1s and enable the *Send Radio COR/AUX Status* SIP setting in all system SCM-1s, SCM-2s, and ARA-1s.

- When the network consists of mixed devices (multiple JPS SIP interfaces, 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 SCM-1s and enable the *Send Radio COR/AUX Status* SIP setting in all SCM-1s, SCM-2s, and ARA-1s.

- When the network consists of mixed devices (JPS SIP interfaces, 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 JPS SIP interfaces and disable the *Send Radio COR/AUX Status* SIP setting in all devices.

The next setting —*SIP Actions* —should be selected when controlling the SCM-1's PTT output via DTMF over the network. See Section 2.17.5.4 for instructions on how to configure SIP Actions.

The next network COR type option is *Packet*. When *Packet* is selected, the SCM-1 will key the radio based upon the presence or absence of audio RTP packets. When the SCM-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 conjunction with silence suppression on the remote device.

When *Packet* COR is enabled, the *Network COR VOX/VMR/Packet Hang Time* setting described below 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.

The final setting option of *Disable*, when set, will disallow any audio received from the network from keying the attached radio. This can be useful in a situation where a radio is attached to a SIP network in a *listen-only* mode.

2.17.5.5.11 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 incoming audio that is required to be considered valid and, therefore, cause the associated 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.

2.17.5.5.12 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 any longer than necessary to remove the drop outs.

2.17.5.6 SCM-1 Audio Adjustments

2.17.5.6.1 Transmit Audio Delay

In some applications (such as interfacing to a trunked radio system) it may be necessary to delay the audio going to the radio transmitter. The amount of delay is set here, ranging from 0 to 8000 milliseconds, Changes do not take effect until the *Save* button is pressed. See system troubleshooting guide, Section 4, for more information about the need for TX audio delay.

2.17.5.6.2 Receive Audio Level

This adjusts the audio into the SCM-1 from its associated receiver. A correct receive level is required to ensure proper operation. Too high a level may cause flat-topping and distortion, while too low a level won't provide adequate audio volume. The front panel SIGNAL LED is provided as a guide to setting the level; raise the receive volume until the SIGNAL LED flashes momentarily on voice peaks. If the LED never lights, the level is too low; if the LED is on nearly continuously, the level is set too high. The following procedure is suggested:

- Transmit on the proper frequency so that the radio associated with the SCM-1 unsquelches and speak in a normal volume.
- Set the SCM-1 module at its lowest (least-sensitive) volume setting. If the module's SIGNAL LED is flashing on voice peaks, this setting is correct. If it's on continuously, the incoming audio level is higher than can be accommodated by the SCM-1 module, and must be attenuated before reaching the module. If the SIGNAL LED is not flashing at this lowest setting, the module's gain must be increased; proceed to the next step.
- Raise the modules receive gain one step at a time until the SIGNAL LED is flashing on voice peaks (until the LED is flashing during voice peaks but is not lit continuously).
- If the LED doesn't flash even on the highest setting, add some Receive Audio Boost.

2.17.5.6.3 Transmit Audio Level

This adjusts the audio from the SCM-1 module to its associated transmitter. The transmit level must be set correctly to insure proper operation of radios or other equipment connected to this output. Too high a level may cause flat topping, distortion, or over-modulation of a connected radio, while too low a level won't provide adequate audio volume or modulation level. If the actual audio level requirement of the radio or other connected equipment is known, select this level from available settings. These levels assume a 600-ohm termination. If the required level is not known, the following procedure is suggested:

- An input audio source for the module is required (so this audio can be sent back out through the transmit audio port). Create a cross-connection with a SIP Phone or other device so that audio can be sent through the SCM-1 to the transmitter.
- Determine the proper input level to the connected equipment (output level from the module). It may be necessary to monitor the module's output audio level at the connected equipment's input port with an audio voltmeter.
- Start with TX audio at its lowest setting; speak in a normal volume into the SIP Phone.
- Raise the Transmit Audio Level one step at a time until the proper level is reached

> ***Note: When the radio settings have been made, click Save to save them.***

2.17.5.7 SCM-1 Configuration – Dialing Page (Outgoing Call Configuration)

A radio user may initiate and terminate calls to the SCM-1 module by either 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), but 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.

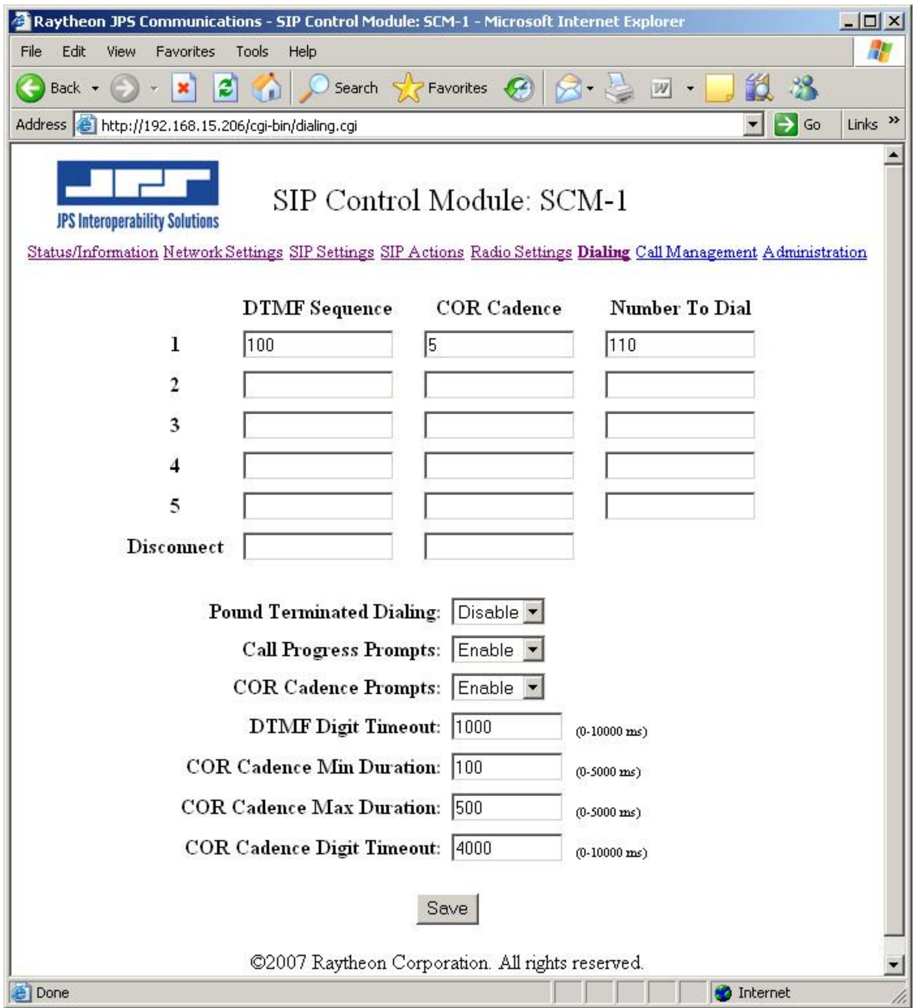


Figure 2-49 Dialing Page; Outgoing Call Configuration

2.17.5.7.1 Configure Outgoing Call Initiation Via DTMF

Enter a valid DTMF sequence that will be transmitted by the radio (such as *100 or 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. A full SIP URI may also be entered. When the radio receives any of these preset

DTMF sequences, the SCM-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 ### or other unique sequence is appropriate and won't be confused with a number to dial.***

2.17.5.7.2 Pound Terminated Dialing

This 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 SCM-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 SCM-1 will instead follow the operation outlined above in Section 2.17.4.1, comparing the detected DTMF to the pre-configured *DTMF Sequences* and then calling the associated *Number to Dial*.

2.17.5.7.3 Call Progress Prompts

The SCM-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 don't want these prompts transmitted over the radio, set to *Disable*. This function operates with both the DTMF and the COR Cadence call initiation modes.

2.17.5.7.4 DTMF Digit Timeout

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

2.17.5.8 Configure Outgoing Call Initiation Via COR Cadence

Many radios do not have DTMF keypads, so the SCM-1 offers an alternative method to initiate a SIP call. COR Cadence is a feature of the SCM-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" stands for "Carrier Operated Relay." It is an indication that a receiver is detecting a carrier signal that's 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 FM 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 SCM-1, and it will activate its COR output with five corresponding pulses, and this *COR Cadence* will be detected by the SCM-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 SCM-1.

Note: This method only works if:

(1) The radio cabled to the SCM-1 has a COR output signal that is connected to the SCM-1's COR input pin. The COR signal line must be properly configured via the Radio Settings page.

(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's 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.

2.17.5.8.1 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 nine. 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.

2.17.5.8.2 Call Progress Prompts

The SCM-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 don't want these prompts transmitted over the radio, set to *Disable*. This function operates with both the DTMF and the COR Cadence call initiation modes.

2.17.5.8.3 COR Cadence Prompts

The SCM-1 also provides voice prompting (automated voice messages) during the COR Cadence call initiation process. Whenever the SCM-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 6 squelch breaks in a row, the voice message prompt will be the word *six*. If the COR Cadence is 3 squelch breaks followed by 5 more, the voice messages will occur after each portion of the sequence, two prompts, *three* and then *five*.

Set this option to *Disable* if you don't want these prompts to be transmitted.

2.17.5.8.4 COR Cadence Min Duration

This 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 is 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 SCM-1 does not always detect each of the squelch breaks attempted.

> ***Note: Don't 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 aren't working properly, it's easy to get feedback through the COR Cadence Prompts feature.***

2.17.5.8.5 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 SCM-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 is 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 SCM Module does not always detect each of the squelch breaks attempted. The looser the criteria are set, the more likely that the unit could false on noise-induced squelch breaks.

2.17.5.8.6 COR Cadence Digit Timeout

Once the SCM-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.

2.17.5.8.7 Automated Dialing Methods

2.17.5.8.7.1 Power On Dialing

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

2.17.5.8.7.2 AUX Input Dialing

The SCM-1 can initiate and disconnect a call using the auxiliary inputs that are part of the radio DB15 interface connector. When configured, the SCM-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 the *AUXIN0 Dialing* field to configure AUX input dialing for the first auxiliary input and use the *AUXIN1 Dialing* field for the second auxiliary input.

2.17.5.8.8 Automatic Redialing

The SCM-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 create a link between two sites.

To have the SCM-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 SCM-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 SCM-1 redial a call it initiated that ends due loss of media detection (see Table 2-12), set the *Redial Outbound Calls On LOM* to *Enable*.

The SCM-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 SCM-1 will respond to the incoming call with a *Temporarily Unavailable* message. If set to *Enable*, the SCM-1 will cancel the redial and answer the incoming call.

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

2.17.5.9 SCM Configuration - Call Management

This page allows the user to initiate a call from via a web browser. It's actually an Operations function page that may be accessed at any time to manage calls, rather than a Configuration page that is usually accessed only once at initial radio installation and set up. It's included here along with the other pages for clarity.

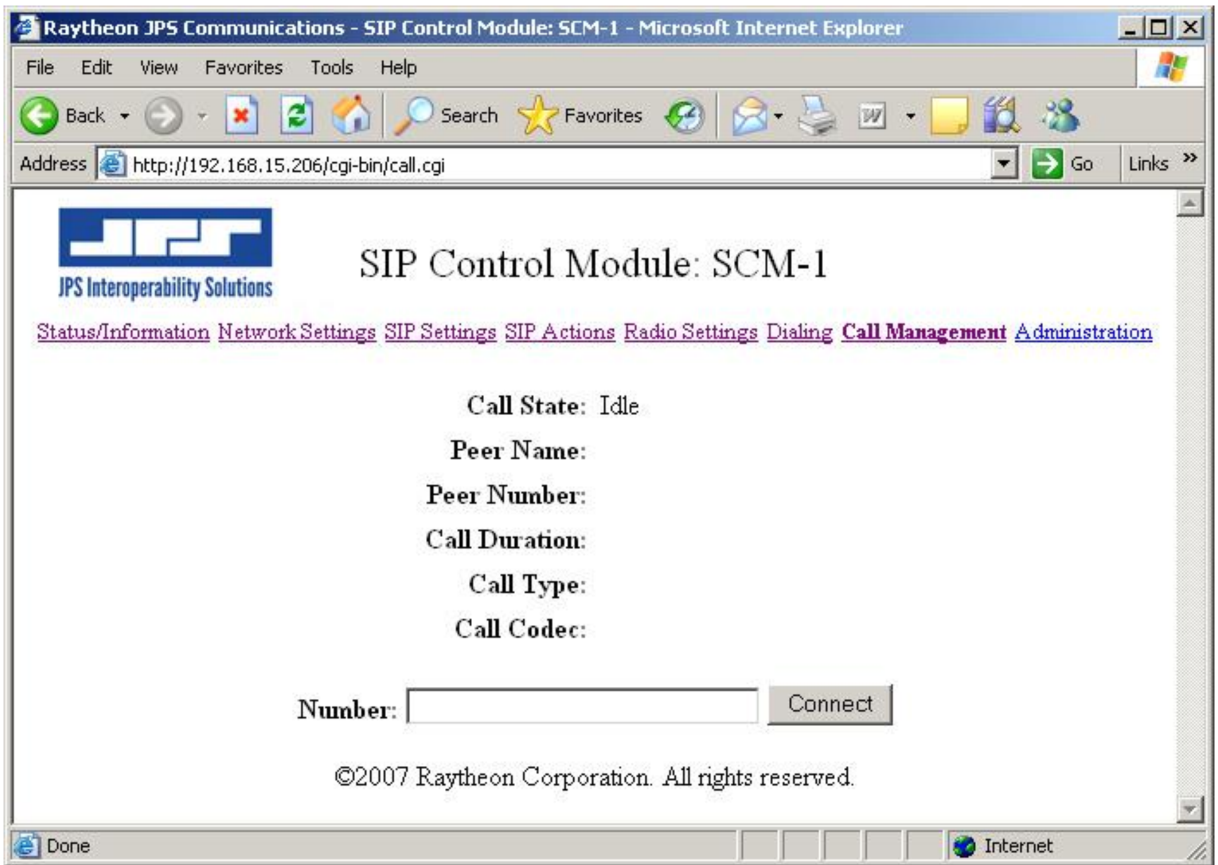


Figure 2-50 Call Management Page

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 refresh in your browser to see the results of the call request.***

2.17.5.10 SCM-1 Configuration - Administrative Functions

This page allows password protection of access to the SCM Module's web pages, facilitates upgrades to the module's firmware, and provides a means to remotely reboot the module.

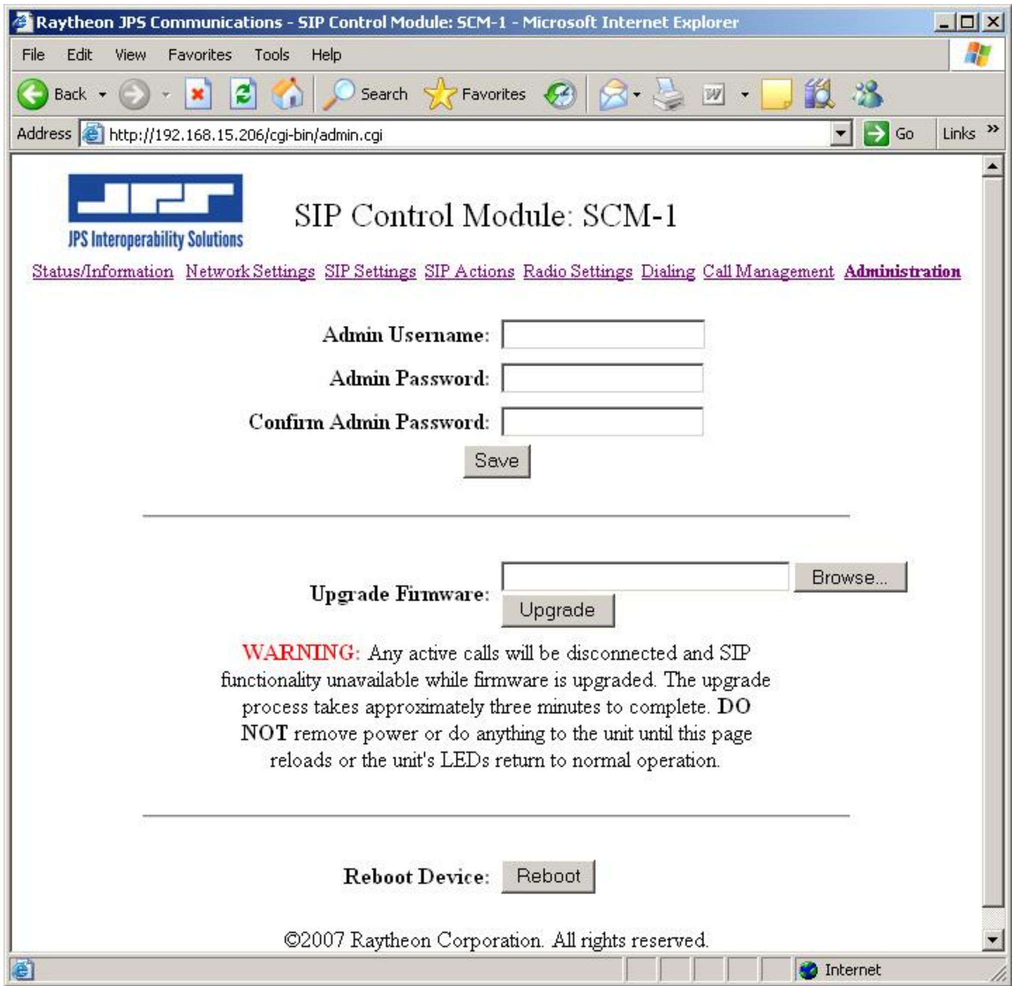


Figure 2-51 Administration Page

2.17.5.10.1 Password Protection

Enter a username and password and click *Save* to password protect the device. After a password has been entered, it will be requested whenever an attempt is made to browse to the module; access to view or change settings will be denied unless the proper password is entered. If you forget the password, the only way to restore access is to reset the unit to factory defaults (see next Section 2.17.5.10.2).

2.17.5.10.2 Restoring Factory Defaults

In rare circumstances, it may be necessary to completely restore the SCM module to the original configuration that was set when the module was manufactured. The procedure for doing this follows.

Procedure:

1. Power down the ACU-2000.
2. Remove the SCM module from the chassis.
3. Install the extender card (supplied in the Accessory Kit) into the now empty slot.
4. Install the SCM into the extender card.
5. Power up the ACU.
6. Configure the *Restore Factory Defaults* jumper JP22 [center to right].
7. Wait 10 seconds.
8. Restore the *Restore Factory Defaults* jumper JP22 [left to center]
9. Wait 15 seconds. (The SCM module will complete the reset sequence)
10. Power off the ACU.
11. Remove the SCM from the extender card.
12. Remove the extender card from the ACU chassis.
13. Install the SCM back into the vacant slot in the ACU.
14. Power up the ACU, and you are now finished

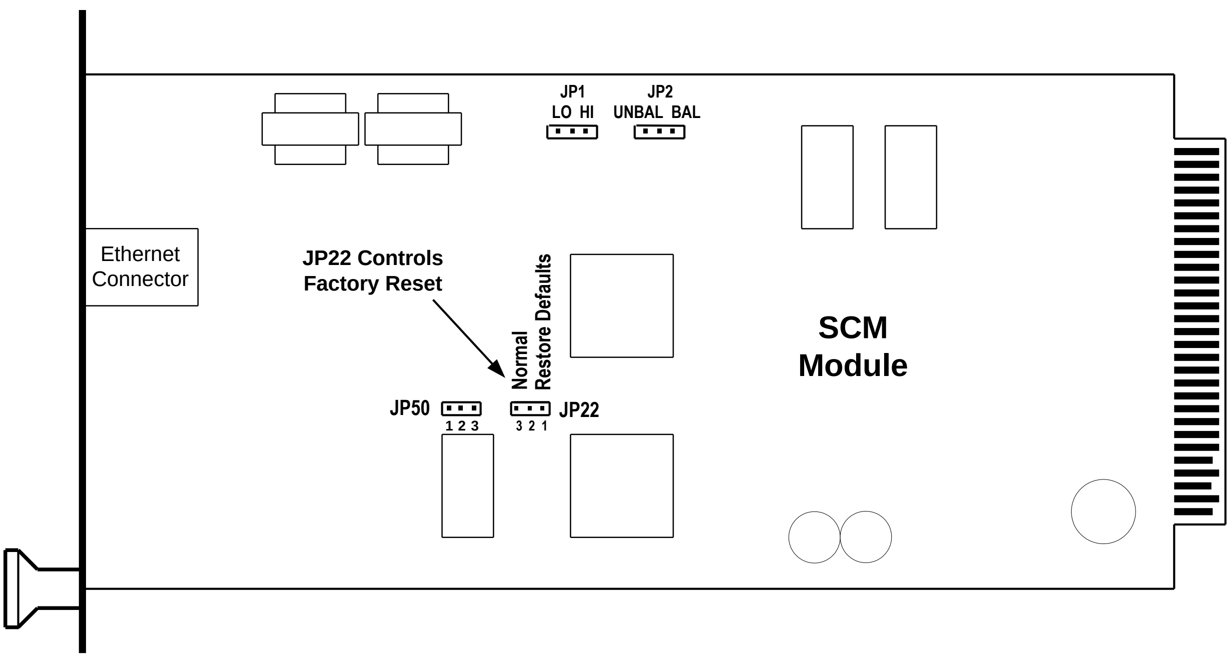


Figure 2-52 Jumper For Factory Default Restoration

2.17.5.10.3 Firmware Upgrade

The SCM is designed to support software updates in the field. Should it become necessary to install updates to the software, the following process should be followed. Instructions and the update software are available on the web at:

http://www.jpsinterop.com

On the right side of the page all ACU-2000 related downloads are listed. This includes this manual, the ACU Controller, and latest versions of software for the ACU-2000 modules. Export regulations require that the form supplied on the website after a download request be filled out prior to enabling the download.

See Section 2.17.2.10 for further instructions on updating the DSP module. The SCM module update procedure is identical.

2.17.5.10.4 Remote Reboot of the SCM-1 Module

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

End of Section Two

3 Operation

3.1 General

Just as the ACU-2000 is capable of many different applications, from local, tactical uses to Wide Area Interoperability Systems, the ACU-2000 also has a corresponding variety of available control methods, including:

- Local Control via a PC with the ACU Controller cabled directly to the ACU-2000. [Serial or Ethernet control]
- Remote Control via ACU Controller (or multiple ACU Controllers) connected via a network. [Ethernet control]
- Remote control via the WAIS Controller – For Wide Area Interoperability Systems capable of connecting multiple ACU-2000 systems together over a network. [Ethernet control]
- Local manual control via the HSP handset, keypad, and speaker.
- Remote manual control via DTMF.

3.2 Operation Via ACU Controller

The ACU Controller program for PCs running Windows is the primary method to control a single ACU-2000. The PC can be connected directly to the ACU-2000 by either a CAT5 cross-over network cable plugged into the CPM front panel Ethernet Jack, or by an RS-232 cable attached to the ACU-2000 rear panel DB-9 serial port. Alternatively, the ACU Controller can run the ACU-2000 over any Ethernet (IP-based) network that is connected to both the computer and the ACU-2000 (in this case, a standard CAT5 cable is used, not the cross-over cable used with a direct PC to CPM connection). This allows control of an ACU-2000 by multiple operators at different computers on the network. The CPM arbitrates the incoming commands, giving priority to the first received, and provides status messages so that all ACU Controllers on the network are kept up-to-date.

For optimum operation of multiple ACU-2000 systems that can be cross-connected to each other over a network (forming a Wide Area Interoperability System) the WAIS Controller program is the best choice for system control and monitoring. See Section 1.7.1.

Note: Only a broad overview of ACU Controller operation is presented here. See the ACU Controller Manual for full details regarding system setup & operation via the ACU Controller. An ACU Controller CD and manual are included with every ACU-2000 chassis. Program downloads and the current manual revision (in PDF format) are available on the website:

www.jpsinterop.com

The ACU Controller Main Screen (see Figure 3-1) shows all modules in the ACU-2000 chassis in the form of individual icons that may be customized to depict the type of communications system they are interfaced to.

The top row of the Main Screen shows all of the ACU-2000's *idle modules*. These icons represent all communications systems not currently engaged in cross-connections.

The rest of the screen displays ongoing cross-connections (nets). A cross-connection between any two communications systems is created by a simple point and click procedure with the cursor placed over each of the desired module icons. The ACU Controller commands the ACU-2000 to make the cross-connection, and after it has done so, it reports back to the ACU Controller and the ACU Controller then displays the net. In this example two nets are ongoing: Net 1 is made up of modules 2 & 7, and Net 2 includes modules 4, 6, & 9.

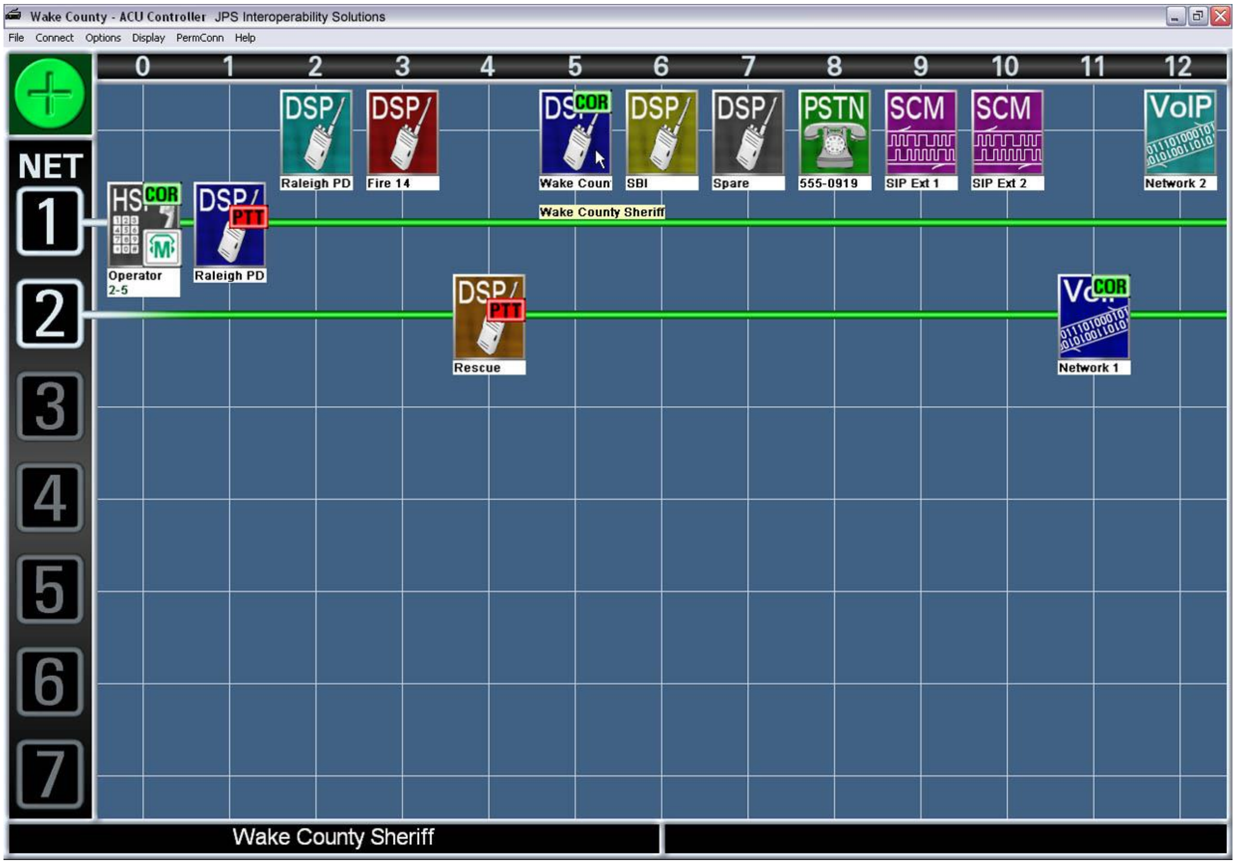


Figure 3-1 ACU Controller Main Screen

Note that if several ACU Controllers are controlling an ACU-2000 via a network, all operators will see, on their Main Screens, the changes made by each of the other operators. The Main Screen changes based on the ACU-2000's response to a request from the ACU Controller, and all ACU Controllers linked to this ACU-2000 over the network receive these responses.

A number of special connection modes can be performed, including temporarily tying Net 1 and Net 2 together. See the ACU Controller manual for details.

Two message areas at the bottom of the Main Screen help guide the operator in controlling the system and understanding its status.

The ACU Controller is also an excellent tool for system setup and configuration. See Section 2.12.1 of this manual and the ACU Controller Manual for details.

3.3 Operation Via WAIS Controller

Wide Area Interoperability (WAIS) Systems tie multiple ACU-2000 systems, as well as other communication assets, together over an IP-based network. The purposes of the WAIS Controller program are to present, in a clear and simple format, the large amount of information inherent in a Wide Area Interoperability System and give the program operator the means to control the system. The WAIS Controller has different views that provide the operator a close-up look at an individual ACU-2000 system or a broad overview of the entire system.

Figure 3-2 WAIS Controller Overview Screen

> ***Note: A description of WAIS Controller operation is beyond the scope of this manual. See the WAIS Controller Manual for full detail. Full information, including a simulator program available by free download, is available on the website:***
>
> ***http://www.jpsinterop.com***

3.4 Local Operation Considerations

3.4.1 Unit Power-Up

Prior to initial power-up, ensure the ACU-2000 is correctly configured for the AC or DC power source being used. If using 220 VAC, make sure the unit is not configured for 110 VAC, or damage may result. Depress the Main Power Switch on the PSM module. Either the AC or the DC LED below the power switch should light and the green +12V and -12V LEDs should light. The ACU-2000 will run internal start-up tests, and then begin operation.

3.4.2 Removal and Replacement of Modules

The ACU-2000 Modules, with the exception of the power supply, can be "Hot-Swapped." That is, they can be removed and inserted while the unit's main power is on without resulting in damage. If a module that is presently communicating with a second module is removed, that link will be lost. If the CPM module is removed, all system operation ceases. The CPM will not be damaged by Hot-Swapping; however, it is advised to turn the power off when replacing the CPM.

CAUTION: Turn Power OFF and disconnect the ACU-2000 from its AC and DC power sources before removing or replacing the PSM Power Supply Module.

3.4.3 Front Panel Controls and Indicators

All front panel controls and indicator LEDs are explained below, starting at the left side of the unit. The module that contains the control or indicator is listed in parenthesis. Refer to Figure 2-2 for Front and Rear panel views

3.4.3.1 Power Switch

The Power switch controls the AC line and DC power to the unit.

3.4.3.2 +12V and -12V Indicators

These green LEDs are driven from the corresponding power supply output voltages and are illuminated whenever the unit's main power is on. An unlit -12V or +12V LED when the AC or DC indicators are on could indicate a failed power supply voltage.

3.4.3.3 AC and DC Indicators

These LEDs light when the corresponding power source is applied at the unit’s rear panel. If both AC and DC power are currently applied, both LEDs will be illuminated.

3.4.3.4 Speaker Switch

This switch turns the front panel speaker on and off. It does not affect the headphone audio or the external speaker driver audio signal available at the rear panel.

3.4.3.5 Headphones Output Jack

This stereo jack accepts a stereo or mono 3.5-mm (called 1/8") headphone jack. The monaural headphone audio signal is supplied to both sections of a stereo jack. Headphone volume, along with the volume of the speaker and handset earpiece, is controlled by the front panel volume potentiometer.

3.4.3.6 Volume Control

This potentiometer adjusts the volume to the speaker, handset, and headphones.

3.4.3.7 Fault LEDs

The red FAULT LEDs will be illuminated whenever the associated module's built-in-test circuitry detects a fault condition for that module.

3.4.3.8 Master/Expansion LEDs

The Master and Expansion LEDs are illuminated only when a pair of ACU-2000's are daisy-chained together to create an Expanded System. These LEDs indicate the status of each unit in the configuration. The Master chassis contains modules corresponding to extensions 0 through 12, and the Expansion chassis houses extensions 13 through 25.

3.4.3.9 Mon (Monitor) LED

The Monitor LED of any module is lit whenever that module is being monitored by another module.

3.4.3.10 Signal LED

The signal LED gives an indication of the proper audio level entering the module from the outside world. This LED lights when the audio level is correct for the module. The input audio level should be adjusted so the signal LED just flashes on voice peaks. If the LED never lights, the audio level is too low; if the LED stays lit nearly continuously, the audio level is too high for best system operation. See Section 1 for installation and setup instructions.

3.4.3.11 PTT LED, VOX LED

The PTT LED lights to indicate that PTT output is asserted along with TX (output) audio being sent from the module. For an SCM-1 module or DSP module in either the standard or VoIP standalone mode, this means that the PTT output of the associated rear panel D15 connector is being asserted. For an SCM-2 or DSP module in VoIP hybrid mode, the PTT LED indicates a PTT indication and audio being sent out over the network via the module's front panel RJ-45 connector. The HSP module's PTT LED is lit when audio is being sent to the HSP's speaker and earpiece.

3.4.3.12 COR LED, VOX LED

The COR and VOX LEDs provide an indication showing that the module has detected what it considers to be valid input audio. The COR LED on a DSP or SCM-1 module indicates when the module has detected valid audio from the radio or other device cabled to the associated rear panel connector. The VOX LED of a PSTN or LP module, or the COR LED of the SCM-2 module indicates a valid input signal at the front panel RJ-11 or RJ-45 connectors.

Note that (unless multiple modules are simultaneously detecting valid input audio), an active COR or VOX means that any cross-connected modules will assert their PTT outputs.

3.4.3.13 Ring LED

The PSTN Ring LED lights while the module is receiving a ring signal from the telephone line. The LP Ring LED is lit whenever it is causing the associated telephone set to ring.

3.4.3.14 Connect LED

The Connect LED lights when the PSTN module is actually connected to a telephone line, either in response to automatic answer of an incoming ring, or after dialing out.

3.4.3.15 Off Hook LED

The LP module's Off Hook LED lights whenever the local phone set is taken off hook (that is, the handset is not hung up. Telephones are *off hook* during a call and *on hook* otherwise.

3.5 Local Operation Via HSP

This section explains how the ACU-2000 may be locally controlled using the HSP module's handset, keypad and speaker. Local operation via the HSP may take place in conjunction with operation via the ACU Controller or the WAIS Controller. When connections are made or broken by a Local Operator using the HSP, the changes are reported to any active, connected control programs by the ACU-2000 and show up immediately.

Note, however, that when the ACU-2000 is being controlled remotely via either the WAIS Controller or the ACU Controller, these programs will prevent HSP configuration programming. It's possible to override this lockout feature, see the *Regain Control* Operational Command Item of Table 3-1.

3.5.1 HSP Local Operation Capabilities

The HSP Module provides a means to locally monitor, control and configure an ACU-2000 system. The user can monitor audio via the handset or an internal speaker, or plug in external headphones or an external speaker. The handset includes a PTT switch, which must be depressed for the user's voice to be transmitted. Control is via a 3x4 keypad (standard telephone layout), which enables the user to select a module and enter control/configuration data. If the system contains a PSTN module, the user may place telephone calls manually using the HSP keypad and handset. The module also has an external connector for connection of 4 wire devices. The HSP module houses the system voice prompt generator. These voice prompts allow the ACU system to respond with English messages (optionally, other languages) that make it easier to configure and operate the system.

Table 3-1 and the subsequent text explain how to use the HSP as an operational control for the ACU-2000 system. A system user at the HSP is considered the local operator of the system, and has the extension number *0 0*. In an Expanded System, if there is an HSP in the expansion chassis, it has the extension number *2 5*.

The HSP can link the handset with any one of the system's interface modules, or with a number of them simultaneously in a conference call. A local operator can use the HSP to perform operations that remote users are unable to perform via DTMF input. These include storing the current state of connections for later recall, and removing a user other than oneself from a current cross-connection.

Table 3-1 HSP Operational Command Items			
Command Item	**Command**	**Description**	**Factory Default**
Make A Connection	* n n	Connect HSP to extension *nn*.	N/A
Break the Current Connections	* #	Terminate all connections the HSP is currently participating in.	N/A
Attention Command	* * *	ACU-2000 responds by identifying the extension number of the HSP module being queried.	N/A
Report Connections	* 3 0	Voice Prompts list all current connections.	N/A
Disconnect Another Extension	* 3 3 n n	Terminate all connections extension *nn* is currently participating in.	N/A
Monitor Function	* 3 4 n	*n* = Extension to be monitored. ** 3 4 n* toggles between Monitor On and Monitor Off.	Monitor Off
Store Connections	* 3 6	Store the current connection configuration for automatic recall at power-up.	None Stored
Regain Control	* 3 7	Regain system configuration programming control from a connected Controller Program.	N/A
Data/Command Mode	* 8 0	Toggles between *Data Mode* and *Command Mode*.	Command Mode
System Reset	* 9 0 n n	If nn is any series of digits other than 00, the *System Reset* feature is enabled, and *nn* is the system reset code. If nn is *00*, the feature is disabled.	Disabled

3.5.1.1 HSP Keypad – Make a Connection

When the command *** n n** is entered at the HSP keypad, a connection is made between the local operator at the HSP and the communications medium at extension n n. For example, if a VHF radio is wired to a DSP module installed at extension #5, the command *** 0 5** will connect the local operator to the VHF radio. The local operator will hear the distant radio user's transmit audio via the radio attached to the DSP module. When the PTT switch on the HSP handset is depressed, the local operator's speech will be transmitted to the radio user. If a PSTN module is installed at extension #8, the command *** 0 8** will connect the local operator to the telephone line wired to that PSTN module. Voice prompts guide the local operator at the HSP to enter the telephone number to be dialed, and provide other useful information, such as informing the user if a module is busy or otherwise unavailable.

To make a conference call among several modules, add a second connection after the first is made. For example, to make a conference call between the local operator at the HSP, the VHF radio at extension 05, and to a PSTN subscriber via the PSTN module at extension 08, first connect to the VHF radio as described above, then at any time add the PSTN subscriber by entering *** 0 8** and following the voice prompt instructions provided.

To make a connection between a pair of modules, neither of which is the HSP, the local operator must first make a connection between the HSP and one of these other modules, then create a conference call with the remaining module, and finally enter the disconnect command *** #** to remove the HSP from the call. If for some reason the two modules still engaged in the link cannot break the connection themselves (for instance, if both are HF radios so neither can reliably transmit DTMF commands), the local operator must use the HSP to break the connection. The local operator may use the *Monitor Function* command to listen to the conversation between the two HF radio users, and enter the *Disconnect Other Extensions* command to terminate the link when the conversation is complete.

3.5.1.2 HSP Keypad – Break a Connection

To disconnect the HSP from any connection, enter the disconnect command * # (star – pound). If the HSP is currently engaged in a conference call between the HSP, the VHF radio at extension 5 and a PSTN subscriber using the PSTN at extension 8, the * # sequence entered at the HSP keypad will remove the HSP from the call, but leave the VHF radio user and the PSTN subscriber connected.

3.5.1.3 HSP Keypad – Attention Command

When the Attention Command *** * *** (star – star – star) is entered through the HSP, a voice prompt will be returned identifying the extension number of the HSP module receiving the command. In non-expanded systems, this will result in the voice prompt *EXTENSION 00*. The Attention Command can be given whether or not the HSP module is currently connected to another module. This allows quick system operation verification without having to make a connection.

3.5.1.4 HSP Keypad – Report Connections

Whenever the Report Command, *** 3 0**, is entered at the HSP keypad, the ACU will list all existing connections. They can be heard at the front panel speaker or the HSP handset. This command cannot be made via any of the interface modules; it's valid only when entered by the HSP keypad.

3.5.1.5 HSP Keypad – Disconnect another Extension

This command allows the local operator to use the HSP keypad to select any system module and remove that extension from any cross-connections that it's currently participating in. When the operational command *** 3 3 nn**, is entered, extension nn immediately removed from all existing cross-connections. Note that this command cannot be made via any of the interface modules; it's valid only when entered by the HSP keypad.

3.5.1.6 HSP Keypad – Monitor Function

This command enables the HSP to monitor the receive audio being sent to any other module or group of modules. Enter *** 3 4 n**, where n is the extension to be monitored. To discontinue monitoring, re-enter *** 3 4 n**, where n is the extension currently being monitored. This command toggles the selected module, first turning the module function on, and then back off. When the monitor function is enabled, the local HSP operator will only hear the receive audio of the module being monitored; the Monitor Function does not give the HSP operator any control over the monitored module, and it will not receive any audio from the HSP handset. The monitoring module will still be capable of making connections and perform other functions if desired. The voice prompts; *MONITORING XX* and *XX DISCONNECTING,* inform the handset operator of the operational changes made.

3.5.1.7 HSP Keypad – Store Connection Table in Memory

Use the command *** 3 6** to store a table of all current module connections in non-volatile memory. After this has been done, every time power is re-applied to the ACU-2000, these connections will be automatically restored.

The factory default connectivity state is *no connections*, so until the first time the *Store Connection Table* command is used, the ACU-2000 will be initialized upon the re-application of power with no connections made. To ***change*** the stored connections, make all the desired connections, then enter *** 3 6**. To ***clear*** stored connections, terminate all connections, then enter *** 3 6**. This command cannot be made via any of the interface modules; it's valid only when entered by the HSP keypad.

3.5.1.8 HSP Keypad – Regain Control from Controller Program

The HSP is normally not allowed to enter ***configuration commands*** while either the WAIS or ACU Controller program is remotely controlling the ACU-2000, as these programs can set configuration much easier and multiple sources of commands can create confusion. If it is ever necessary to pull back the configuration control from one of these programs (for example, if the computer is remotely located and locks up), enter the command *** 3 7** to return control to the HSP.

3.5.1.9 HSP Keypad – Data / Command Modes

The HSP module receives its input audio from the handset, so it does not have a DTMF detector. Its keypad functions as its DTMF input. If the HSP is in the Command Mode, the keypad entries are interpreted as commands to the system. If the HSP is in the Data Mode, keypad entries are not interpreted as commands, but instead the keypad entries are sent as parallel data to a cross-connected PSTN module. The PSTN regenerates the DTMF and sends it out on the phone line. The ** 8 0* command toggles a module in and out of the Data Mode. All modules begin operation in the default Command Mode. When Data/Command Mode command is received, the module switches to the Data mode and returns the prompt *Data Mode*. Whenever the ** 8 0* command is again received, the module responds with the *Command Mode* prompt and reverts to normal operation.

A simple example of the use of this command assumes the local operator at the ACU-2000 wants to call home and check the messages on his answering machine. The HSP starts out in the Command Mode. The local operator connects to a PSTN module. Following the voice prompts provided, he then enters his home telephone number. If he continues to press the HSP keypad after the call is answered, no DTMF will be transmitted until he enters the *** 8 0** command. Once he does, he may enter the password to gain access to his answering machine. All subsequent keypad entries will result in the transmission of the DTMF characters until he either toggles out of the Data Mode with another *** 8 0** entry or disconnects with a *** #** entry.

It is important to note that only the PSTN module regenerates DTMF in this way. This mode was created mainly to allow an ACU-2000 user to make a connection to a PSTN module and be able to control equipment that is connected to the phone line and uses DTMF signaling (voice mail systems, answering machines, etc.

3.5.1.10 HSP Keypad – System Reset Feature

The System Reset Feature allows the local operator to reset the ACU-2000 to its initial power-up state. This means all current connections will be lost, and the unit will return to any connections stored last by the * 3 6 command. In order to prevent any inadvertent or unauthorized use of this powerful feature, the System Reset Feature can only be used after first being enabled at the HSP keypad. In addition, a system reset code is entered, and DSP or PSTN users must then enter this code in order to implement this feature.

The ACU-2000 factory default for this feature is disabled. To enable System Reset capability, enter * *9 0 n n*, where *nn* is any pair of digits other than 00. The feature is now enabled, and *nn* is the system reset code. If a DSP, RDI-1, or PSTN user (who is currently connected to the system) enters the DTMF command * *9 0 n n*, the system will be reset. If * *9 0* and any digits other than the system reset code are entered, the system will not be reset.

To disable this feature, enter * *9 0 0 0* at the HSP keypad. To re-enable, once again enter * *9 0 n n* , where *nn* can be either the previous system reset code or an entirely new code.

3.6 Operation Via Remote DTMF

An ACU-2000 may be controlled via DTMF from the field. This may be from the DTMF keypads of radios or other 4-wire devices, remote telephones connected via PSTN, or local telephone sets interfaced to an LP-2 module. Operation is similar to what can be done by a local operator via the HSP keypad, though the command set is more limited.

Operation via DTMF may take place in conjunction with operation via the ACU Controller or the WAIS Controller. When connections are made or broken by DTMF, the changes are reported to any control programs by the ACU-2000 and show up immediately.

Note that a module must have DTMF enabled or it will ignore DTMF input. It may also have PIN Security enabled. If so, DTMF will be ignored unless the proper PIN is entered.

Table 3-2 Operational Commands Via Remote DTMF

Command Item	Command	Description	Factory Default
Make A Connection	* n n	Connect the module to extension nn	No Connections
Break the Current Connections	* #	Terminate all connections the module is currently participating in.	N/A
Attention Command	* * *	ACU responds with the extension number of the module being queried.	N/A
Monitor Function	* 3 4 n	n = Extension to be monitored. * 3 4 n toggles between Monitor Mode and Normal (non-monitoring) Mode.	Disabled (Normal Mode)
Data /Command Mode	* 8 0	Toggles between Data Mode and Command Modes.	Command Mode
System Reset	* 9 0 n n	Performs system reset. *nn* is system reset code set via HSP keypad. (See Section 3.5.1.10)	Feature Disabled

3.6.1.1 Remote DTMF – Make a Connection

When the command *** n n** is detected in the audio input of any interface module, a connection is made between that module and the communications medium at extension **n n**. For example, if a VHF radio is wired to a DSP module installed at extension #5, the command *** 0 5,** entered into the keypad of a telephone set associated with an LP-2 module at extension #11, will cross-connect the local phone set to the VHF radio system. If a PSTN module is installed at extension #8, the command *** 0 8** will cross-connect the LP-2 user to the telephone line wired to that PSTN module. Voice prompts guide the LP-2 operator to enter the telephone number that will be called, and provide other useful information, such as informing the user if the line is busy. To make a conference call among several modules, add a second connection after the first is made. For example, make a conference call between the LP-2 user at extension 11, the VHF radio at extension 05, and to a PSTN subscriber via the PSTN module at extension 08. First make a connection between the LP-2 and the VHF radio as described above, then add the PSTN subscriber by entering *** 0 8** and following the voice prompt instructions provided.

3.6.1.2 Remote DTMF – Break a Connection

To disconnect an interface module from any connection, enter the DTMF disconnect command *** #** (star - pound). If the interface module is currently engaged in a conference call between several different modules, the *** #** sequence entered at the keypad of a radio associated with extension #8 will remove extension #8 from the call, but leave the remaining connections intact. Because of the way a telephone set is directly wired to the ACU-2000 chassis, it's possible to terminate an LP-2 connection by hanging up the LP-2 telephone handset. The only local means for any other modules to terminate their connection is the *** #** entry.

3.6.1.3 Remote DTMF – Attention Command

When the DTMF Attention Command *** * *** (star – star - star) is sent to any module, a voice prompt will be returned identifying the extension number of the module receiving the Attention Command. This can be done whether or not the module being queried is currently connected, so it can be used to check if the system is operational without having to make a connection.

3.6.1.4 Remote DTMF – Monitor Function

This command enables any module to monitor all audio from any other module. Enter *** 3 4 n**, where **n** is the extension to be monitored. To discontinue monitoring, re-enter *** 3 4 n**, where again **n** is the extension that is currently being monitored. This command toggles the selected module, turning the monitor function on, and then cutting it off. When the monitor function is enabled, the monitoring module will hear only the receive audio of the module being monitored. The Monitor Function does not allow the monitoring module to assert any control over the monitored module, nor can it send TX audio to the monitored module. Both the monitoring and the monitored module will still be capable of making connections and perform other functions if desired.

3.6.1.5 Remote DTMF – Data / Command Modes

When an ACU-2000 interface module detects DTMF in its receive audio, it can either interpret the DTMF tone as a command input (Command Mode) or transmit DTMF via a connected PSTN module so this DTMF can control other equipment (Data Mode). When in the Data mode, the DTMF input is detected and interpreted. This information is passed to the connected PSTN module, which then regenerates the DTMF and inserts it into the PSTN audio output. This regenerated DTMF is free of any FM noise or the frequency response related distortion that often results from the pre-emphasis and de-emphasis of FM audio circuits. The *** 8 0** command toggles a module in and out of the Data Mode. All modules begin operation in the default Command Mode. When the Data/Command operational command is received, the module switches to the Data mode and returns the prompt *Data Mode*. Whenever the *** 8 0** command is again received, the module responds with the *Command Mode* prompt and reverts to normal operation.

A simple example of the use of this command assumes the radio user wants to call home and check the messages on his answering machine. The user makes a connection from the DSP module associated with his radio to the PSTN module at extension 01. Following the voice prompts provided, he then enters the telephone number at his home. After the call is successfully made, the user then enters the *** 8 0** command. Once he does so, he may enter the password to gain access to his home answering machine. All subsequent DTMF entries will result in the regeneration of the DTMF characters until he either toggles back out of the Data Mode with another *** 8 0** entry or disconnects with a *** #** entry. The module that's receiving the DTMF should also have its DTMF Mute timer enabled (see the DTMF Mute Timer explanations in Section 2.13.1.2.1).

It is important to note only the PSTN module regenerates DTMF in this way. The Data Mode was created mainly to allow an ACU-2000 user to gain access to the system through any type of module, make a connection to a PSTN module and be able to control equipment that commonly uses DTMF signaling (voice mail systems, answering machines, etc). It is possible to use the Data Mode (with the DTMF Mute Timer disabled) to cause input DTMF to be repeated through the ACU-2000. This is not advisable because of the FM noise that will be mixed with the DTMF, along with the frequency response related distortion that will be compounded by the fact that the DTMF will be repeated through two radios.

3.6.1.6 Remote DTMF – System Reset Feature

The System Reset Feature allows a user who is interfaced with the system via a DSP, LP, or PSTN module to reset the ACU-2000 to its initial power-up state. This means all current connections will be lost, and the unit will return to any connections stored by the * 3 6 command (See Table 3-1 and Section 3.5.1.10). In order to prevent any inadvertent or unauthorized use of this powerful feature, the System Reset Feature can only be used after first being enabled at the HSP keypad. In addition, a system reset code is entered via the HSP, and remote users accessing the system via DTMF users must enter this code in order to implement the System Reset feature.

The ACU-2000 factory default for this feature is disabled. To enable System Reset capability, use the HSP to enter ** 9 0 n n*, where *n n* is any pair of digits other than 00. The feature is then enabled, and *n n* is the system reset code. If a DSP, or PSTN user (who is currently connected to the system) enters the DTMF command ** 9 0 n n*, the system will be reset. If ** 9 0* and any digits other than the system reset code are entered, the system will not be reset.

To disable this feature, enter ** 9 0 0 0* at the HSP front panel. To re-enable, once again enter the ** 9 0 n n* sequence, where *n n* can be either the previous system reset code or an entirely new code.

3.6.2 Basic Local HSP & Remote DTMF Operation Scenarios

Examples of the operation of the ACU-2000 ***without using the ACU Controller or WAIS Controller*** are discussed in the following paragraphs. This means operation through the use of commands entered either via the HSP keypad or the DTMF tones received by a communications system connected to the ACU-2000. The DTMF tones can be generated by a user of this same communications system provided they have a DTMF keypad. These examples and instructions assume the ACU-2000 has already been correctly configured. Scenarios assume most configuration options are at default settings with DTMF enabled. Otherwise, some changes in descriptions of scenario progress will result.

The scenarios may seem a bit redundant; this is a valid assessment as the ACU-2000 operational steps are intentionally consistent. Each scenario will add only a few new bits of information or operational aspects that are particular to a individual interface module.

Refer to Figure 3-3 below when reviewing the operational scenarios.

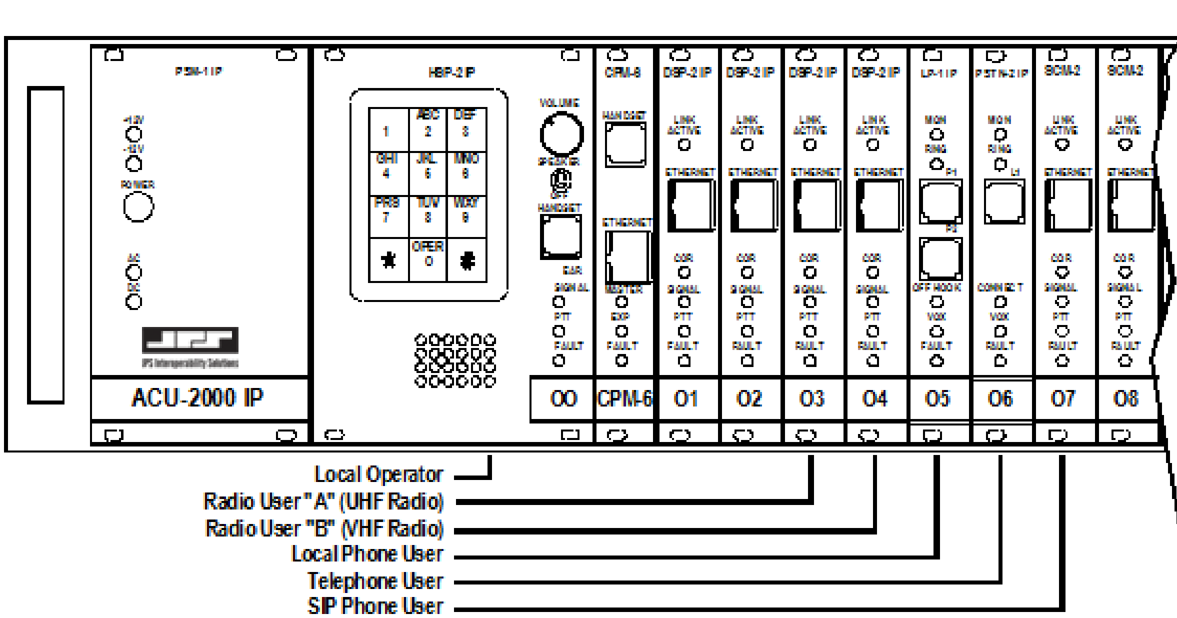


Figure 3-3 Pictorial Layout for Operating Scenarios

3.6.2.1 Radio to Radio

3.6.2.1.1 Conditions:

1) User A (with hand held UHF radio) wants to talk to VHF hand-held radio User B.
2) User B’s radio is not currently transmitting.

3.6.2.1.2 Operation Steps:

1) User A enters *** 0 4** to establish a connection with the User B’s radio (associated to the DSP module at extension 04).
2) The ACU-2000 enables the connection between the two radios and issues a voice prompt indicating the connection has been successfully completed.
3) Users A and B are now free to talk. At the end of conversation, ***either*** user can terminate the link by entering *** #**.

Note: For nets involving two modules, when either is removed from the net, the net is terminated. For nets involving more than two modules, removal of one user removes only that user. To dissolve the full net, all parties must terminate the link by using * #

3.6.2.2 PSTN to Radio

3.6.2.2.1 Conditions:

1) PSTN user wants to talk to hand-held radio User A.

3.6.2.2.2 Operation Steps:

1) PSTN user calling from distant phone dials the phone number associated with the ACU-2000's PSTN module.

2) ACU-2000 greets the PSTN caller and prompts him to enter “star” (*) and the two-digit number of the extension that caller would like to converse with. In our example, User A resides at extension 03, so the PSTN caller enters * *0 3*.

3) The ACU-2000 enables the connection to the radio and issues the voice prompt *03 CONNECTED* to the PSTN caller indicating the connection has been successfully completed.

4) PSTN caller and User A talk. At the end of conversation, ***either*** user can terminate the link by entering * *#*. In links involving more than two modules, all users should terminate the link by using * *#*.

3.6.2.3 Local Operator to Radio

3.6.2.3.1 Conditions:

1) Local operator wants to talk over band 1 to hand-held radio User A.

3.6.2.3.2 Operation Steps:

1) The Local Operator uses HSP handset and keypad to enter * *0 3* to link with User A.

2) The ACU-2000 makes the connection between the local operator and the band 1 radio, and then issues the voice prompt *03 CONNECTED*.

3) Local Operator and User A talk. At the end of conversation, ***either*** user can terminate the link by entering * *#***.** In links involving more than two modules, all users should terminate the link by using * *#*.

3.6.2.4 Radio to Local Operator

3.6.2.4.1 Conditions:

1) Hand-held radio User A wants to talk to local operator at the ACU-2000.

3.6.2.4.2 Operation Steps:

1) The Local Operator's Extension number is '00' (the HSP module), so hand-held radio User A enters *** 0 0** on his DTMF keypad.

2) The ACU-2000 enables the connection between the radio the local operator and issues the *00 CONNECTED* voice prompt to User A indicating the connection has been successfully completed.

3) User A can be heard at the HSP speaker and handset. User A and local operator talk. At the end of conversation, ***either*** user can terminate the link by entering * #.

3.6.2.5 Radio to PSTN

3.6.2.5.1 Conditions:

1) Hand-held radio User A wants to be connected to PSTN and make a call to 555-1234.

2) PSTN module is not busy, and remote phone at 555-1234 is not busy.

3.6.2.5.2 Operation Steps:

1) PSTN Module Extension number is '06', so User A enters *** 0 6** on keypad.

2) The ACU-2000 prompts the user A to *ENTER PHONE NUMBER*.

> ***Note: All phone number entries must be terminated by the # key, so the ACU-2000 can determine the end of the number.***

3) User enters **5 5 5 1 2 1 2 #**, and the ACU-2000 PSTN module initiates the call. Prompts advise User A of the progress in making the call. When the link is established, the phone at 555-1212 rings (just like any other phone call).

4) User A and the person called on telephone now talk. At the end of conversation, ***either*** user can terminate the link by entering *** #**. In links involving more than two modules, all users should terminate the link by using * #.

> ***Note: If no one answers the telephone, the attempt to connect will be terminated when the set Ringing time expires (Factory default setting is 30 seconds).***

3.6.2.6 Local Operator to SIP Extension

Conditions:

1) Local Operator wants to communicate with someone at extension 134 of a SIP PBX.

2) The SCM-2 module at the ACU-2000 is registered with the SIP PBX.

3) SIP PBX extension 134 is assigned to a SIP Phone that is not currently busy.

Operation Steps:

1) The Local Operator (using HSP handset and keypad) enters *** 0 7** to link with the SCM-2 module.

2) The ACU-2000 makes the internal cross-connection between the local operator and the SCM-2, then issues the voice prompt *ENTER PHONE NUMBER*.

3) The local operator enters the desired extension number via the HSP keypad *1 3 4 #* .

> ***Note: All SIP connection IP addresses and Extension Number entries must be terminated by the # key, so the ACU-2000 can determine the end of the entry.***

4) The SCM-2 module sends the proper data over the IP network to request a SIP session with the end user at extension 134 of the PBX that the SCM-2 is registered with. When the session is successfully initiated, the SIP Phone at extension 134 rings and is answered by the user. If the SIP Phone at extension 134 had been busy, the Local Operator would have heard the prompt: *DISCONNECTING* and the call attempt would have been terminated.

5) Local Operator and the person at extension 134 talk. At the end of the conversation, either user can terminate the call by entering ** #*. The SIP Phone user can also simply hang up.

3.6.2.7 Remote SIP Phone User (not a PBX member) to Local Phone

Conditions:

1) ACU system administrator is away on business and needs to contact a user on the ACU2000 local phone system from the broad-band Internet connection in her hotel room.

2) LP (Local Phone) module at extension 05 is not busy.

3) The SCM-2 is set to auto-answer.

Operation Steps:

1) The remote SIP Phone user dials the IP address of the SCM-2 from the SIP Phone.

2) As the SCM-2 is set to auto-answer, the call is answered and the remote user receives the voice prompt to enter * and a two digit extension to connect to another module in the ACU2000.

3) The SIP Phone user enters *** 0 5** to establish a link with the local phone module at extension 05.

4) The ACU-2000 makes the internal cross-connection between the SCM-2 and the local phone module. The Local Phone rings and is answered.

> ***Note: If no one answers the telephone, the attempt to connect will be terminated when the set Ringing time expires (Factory default setting is 30 seconds).***

5) SIP Phone user and the person at the local phone talk. At the end of conversation, ***either*** user can terminate the call by entering * # or by simply hanging up**.**

3.6.2.8 Radio to SIP Extension

Conditions:

1) Hand-held radio User A wants to talk to someone at extension 134 of SIP PBX.

Operation Steps:

1) The SCM-2's extension number is '07', so hand-held radio User A enters *** 0 7** on his DTMF keypad.

2) The ACU-2000 enables the connection between the radio and the SCM-2 and issues the *07 CONNECTED, ENTER PHONE NUMBER* voice prompt to radio User A indicating the connection has been successfully established.

3) Hand-held radio User A enters **1 3 4 #** on his DTMF keypad.

4) The SCM-2 module sends the proper data over the IP network to request a SIP session with the end user at extension 134 of the PBX that the SCM-2 is registered with. When the session is successfully initiated, the SIP Phone at extension 134 rings and can be answered by the user. If the SIP Phone at extension 134 had been busy, the Local Operator would hear the prompt: *DISCONNECTING*

5) Radio user and the person at extension 134 talk. At the end of conversation, ***either*** user can terminate the call by entering *** #.** The SIP Phone user can also simply hang up.

3.6.2.9 Local Phone to Radio

Conditions:

1) Local Phone user wants to talk to hand-held radio User A.

Operation Steps:

1) Local Phone user picks up handset and hears dial tone. User A resides at extension 03, so Local Phone user enters *** 0 3**.

2) The ACU-2000 enables the connection to the radio and issues the *03 CONNECTED* voice prompt to the Local Phone user. Local Phone User and User A talk.

3) At the end of the conversation, ***either*** user can terminate the link by entering *** #**. The Local Phone user can also simply hang up..

3.6.2.10 Conference Call

3.6.2.10.1 Conditions:

1) Local Operator has already established a conversation with hand-held radio user A. He now wants to turn the conversation into a conference call with a third party located at telephone number 555-1234.

2) PSTN module is not busy, and phone at 555-1234 is not busy.

3.6.2.10.2 Operation Steps:

1) PSTN Module Extension is '06', so local operator enters *** 0 6** on the HSP keypad.

2) The ACU-2000 then prompts user A to *ENTER PHONE NUMBER*.

> ***Note: All phone number entries must be terminated by the # key, so the ACU-2000 can determine the end of the number.***

3) User enters **5 5 5 1 2 3 4 #**, and the ACU-2000 PSTN module initiates the call. Prompts are given to User A advising him of the progress in making the call. (For example if the line is busy or there is no answer, the caller is informed.) When the link is established, ACU-2000 sends ringback audio to Local Operator and User A until the phone is answered.

4) All users can talk.

5) Additional parties can be added to the call if required.

6) Any user can terminate his connection to the link by entering *** #**; all other connections will be maintained. When only two parties remain in the conference call, and either of these users enter *** #,** the link will be terminated.

End of Section Three

4 System Troubleshooting

4.1 System Troubleshooting Overview

This section provides some hints to optimization of ACU-2000 setup based on system operation symptoms.

These symptoms include:

- Missed First Syllables
- Missed Syllables in Mid-Conversation
- Stuck Channel (system is locked up by one transmitter)
- Ping-Pong (cross-connected radios key & unkey repeatedly after end of intended transmission)
- False Keying (inappropriate keying due to RFI)
- Inability Of Dispatcher To Gain System Control
- Poor Audio Quality
  - Incompatible Volume Levels
  - Noisy Received Signals
  - Audio Shaping To Improve Sound Quality
- Unintended Consequences (symptoms resulting from inappropriate settings)

4.2 Missed First Syllables

There are several possible causes for the initial syllables of system messages being missed. To narrow down the source of the problem, first determine if one of the ACU-2000 extensions is missing initial syllables from messages from all other extensions, or more likely, that all extensions are missing the first syllables of messages from a particular extension.

The first example is rare and points to a slow to respond link at the one extension reporting the problem. This could be caused by a slow-to-key transmitter. This is best resolved by adding transmit audio delay on that extension to buffer the audio until the transmitter is fully active.

Note: If audio is lost during a transmission as well as at the beginning of a transmission, first consider the remedies explained in Section 4.3, Missed Syllables in Mid-Conversation

If the users of one of the radio systems connected to the ACU-2000 regularly miss the initial syllables of messages from all other radio systems (or other interoperability system members, such as dispatchers or telephone users), the DSP associated with the system missing the initial syllables needs to have its TX Audio delay setting increased. 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.

4.2.1 Trunked Channel Acquisition Delay

800 MHz Trunked Radio Systems (and other trunked 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 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's 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's being patched from other radios into the trunked system by increasing the TX Audio Delay setting of the associated DSP 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-2000 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.

Refer to Figure 4-1 and Figure 4-2 on the following pages for an illustration of the problem and how it can be resolved.

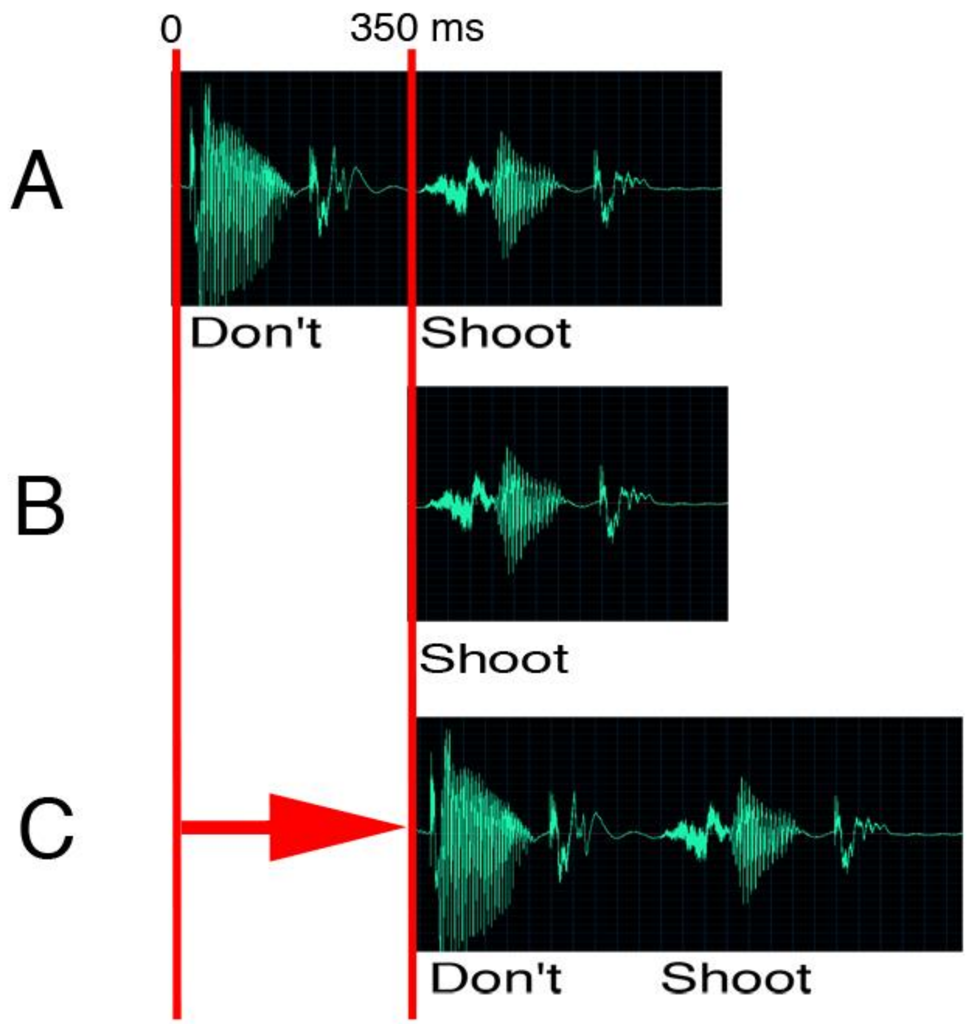


Figure 4-1 "Shoot" Versus "Don't Shoot"

A: The audio being sent into the Interoperability System by radio #1. Radio #1 is cross-connected to radio #2.

B: Radio #2 is an 800 MHz trunked radio with a Channel Acquisition Delay of 350 milliseconds. Therefore, radio #2 won't start transmitting the audio from radio #1 until 350 ms have past, and the first word of the message is clipped.

C: If the Interoperability System delays the audio to radio #2 by at least as long as the channel acquisition delay, the entire message gets through.

Figure 4-2 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.

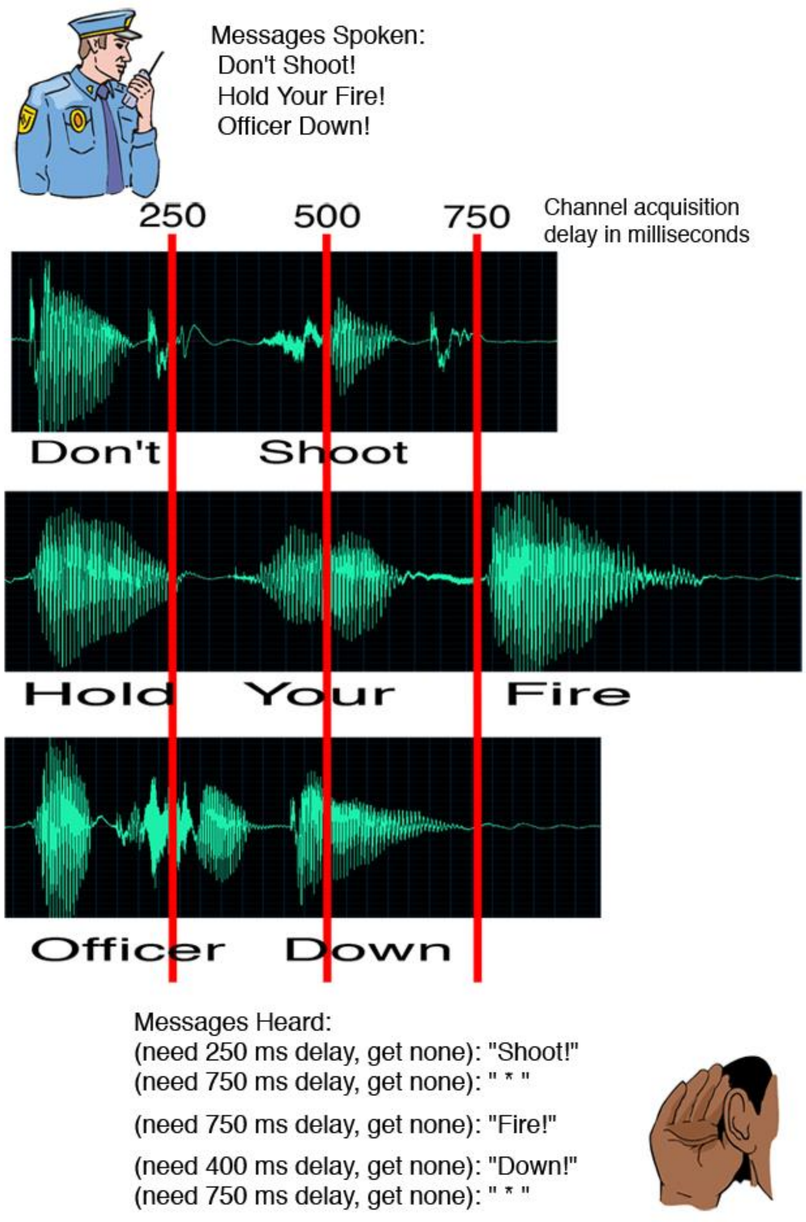


Figure 4-2 Why Audio Delay Is Crucial

4.3 Missed Syllables Mid-Conversation

The most likely causes of missed syllables in mid-conversation are VOX/VMR dropout or COR sampling. If the VOX/VMR hangtime is incorrectly adjusted, the VOX/VMR will momentarily unkey the transmitter and then quickly rekey. The solution is to increase the hangtime to be slightly longer than the speech inter-word time. This is discussed in Section 2.13.3.5.

COR sampling requires that PTT be dropped for a period of time every so many seconds. When PTT is dropped, there will be a hole in the audio. See Section 2.13.3.6 for a detailed explanation of COR sampling. This problem might be recognized by a regular pattern of interruptions on a long speech sample. Turning this function off temporarily would help identify if it was the cause of the problem.

4.4 Stuck Channel

Consider again the basic concept of radio-to-radio cross-connections. When one of the radios cabled to the ACU-2000 is unsquelched, all cross-connected system radios are transmitting. To the system users, this means that if a single radio operator keys his radio, all other radios in the cross-connected systems will be listening, and unable to access the system until the first user unkeys.

Now consider what happens if an ACU-2000 system radio is inappropriately unsquelched and stays unsquelched for an extended period. Possible causes are a problem with the radio, interference on its frequency, or a radio in the field that's stuck in the key-down mode, etc. It could simply be someone who is terribly long-winded and won't let the other system users break in. Whatever the reason, any system radios cross-connected to the problem radio will be stuck in the transmit mode, and the associated system users will not be able to access the Interoperability System. Refer to Figure 4-3 and consider what happens to the system if the #4 portable fails to unkey.

This problem is also referred to as "stuck mic".

The best solution is COR Sampling (also referred to as "COR Sniffing"). With COR Sampling, the COR inputs of other radios in the connection will be occasionally sampled, and if one is active, it will be given control of the system. This provides an opportunity for another user to break in and take over the control of the system. This may give that user a chance to alert the system's operator that there is a problem (if there is an operator monitoring ongoing voice traffic), or if DTMF control is available, this user can disconnect his radio from the system.

An effective COR Sampling function should have the ability to set how long any channel is stuck before the sampling begins. This is important because the stuck channel radio must be cut off momentarily for the function to operate, and it's important that this does not happen inappropriately.

Another way to deal with a stuck channel is for a system operator to constantly monitor all system activity and disconnect any offending radios. In practice, this is probably too much to ask of a busy operator.

4.5 Ping Pong

Some radios have a tendency to unsquelch momentarily at the end of each transmission. Remember that for any pair of cross-connected radios, whenever one radio is unsquelched, the other is keyed. If a radio interfaced to an Interoperability System exhibits the *momentary unsquelch* 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.

Only the local radios cabled to the ACU chassis can cause ping-pong. It can occur if the associated DSP modules are activated by either a COR input signal or a by VOX.

There are two ways to prevent this. First, turn on the adjustable *COR (unsquelch) Inhibit Timer after PTT*. 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. Invoke the feature for each DSP module that shows a short burst of COR just after a keying sequence; this can best be seen by observing the DSP-2 module front panel LEDs.

Another way to prevent this is to use neither COR nor VOX, but instead use VMR. Since Voice Modulation Recognition will not trip unless human speech is actually present, these momentary (and inappropriate) unsquelch conditions will simply be ignored by the system.

4.6 False Keying

When a radio is installed in an environment with lots of RF emissions near the receiver's frequency, these emissions may cause the radio to unsquelch inappropriately. Some radios have a greater tendency for this problem than others. When the inappropriate unsquelch occurs, any radios cross-connected with the offending radio will momentarily transmit a loud burst of noise.

If any radio has a tendency to key on noise (and it's not possible to rectify by reducing the RFI or altering antenna placement), the best solution is to change that radio's system interface to VMR Mode rather than to use either COR or VOX. In VMR Mode, the Interoperability System will ignore these inappropriate noise bursts because the VMR will trip only when human speech is detected in the receive signal.

Any incident scene is likely to be a volatile RF environment because of the wide range of communications devices being deployed. This makes the on-the-fly optimization capability of the ACU-2000 very beneficial. A quick switchover to VMR mode can easily be made by the ACU Controller or WAIS Controller when changing conditions warrant it.

4.7 Inability of Dispatcher to Gain System Control

If all system users have equal priority, the user that transmits first is in control until this person ceases transmitting and gives someone else a chance. It may be beneficial (or absolutely necessary) to give priority to one or two important users. Remember that an Interoperability System is tying together entire radio systems, not just individual radios.

This can be accomplished by being able to assign either TX Priority or Dispatch Priority to all system interfaces. (Also called PTT Priority or COR Priority).

Normally all interfaces are set to TX Priority. This means that if two or more radios or other 4-wire devices (e.g. a dispatch console) are cross-connected, whoever talks first is in control and no one else can be heard until this person stops talking (and releases the radio PTT).

If the DSP module associated with one user is set to Dispatch (COR) Priority, an unsquelch condition received at the Interoperability System from this console will override the other user's control of the system. The dispatcher's audio will be transmitted instead, or will be mixed with existing incoming audio from any other DSP module also set to Dispatch Priority.

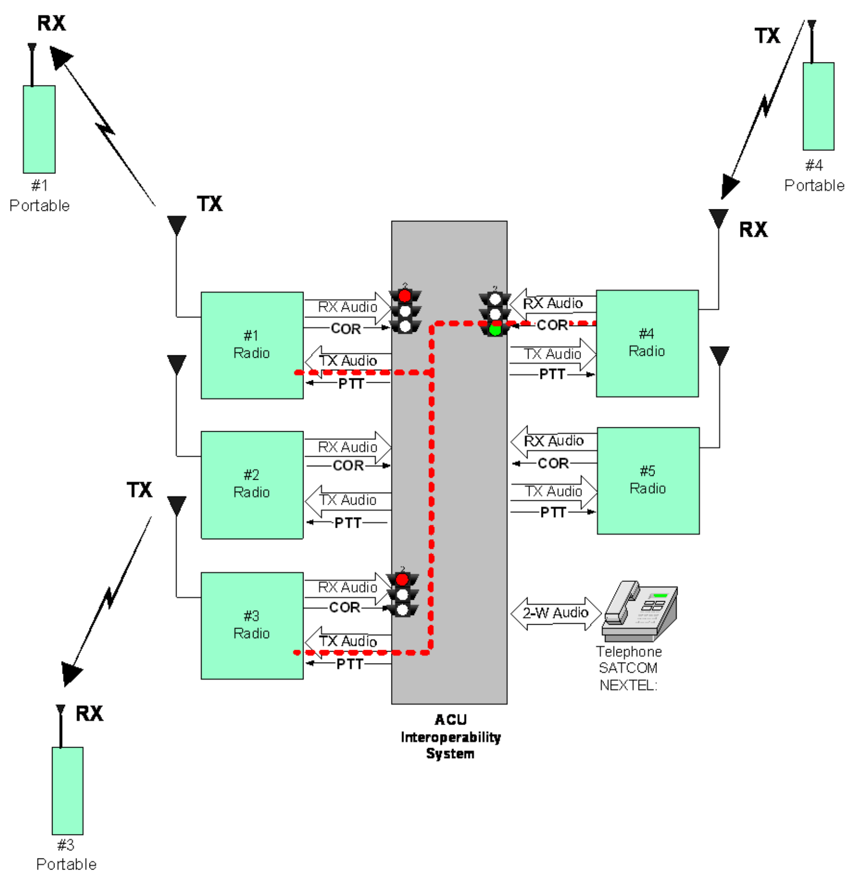


Figure 4-3 TX Priority

In Figure 4-3 above, a cross-connection has been made to link radios #1, #3, and #4. At the moment depicted, a portable of the #4 radio system has keyed (TX) first, so its audio is being retransmitted to the #1 and #3 radio systems. The stoplights signify that, until the #4 portable unkeys, only the audio from the #4 radio will be allowed. If the #1 or #3 portables transmit before #4 stops transmitting, the active COR signals of the ACU system receivers will be ignored by the system.

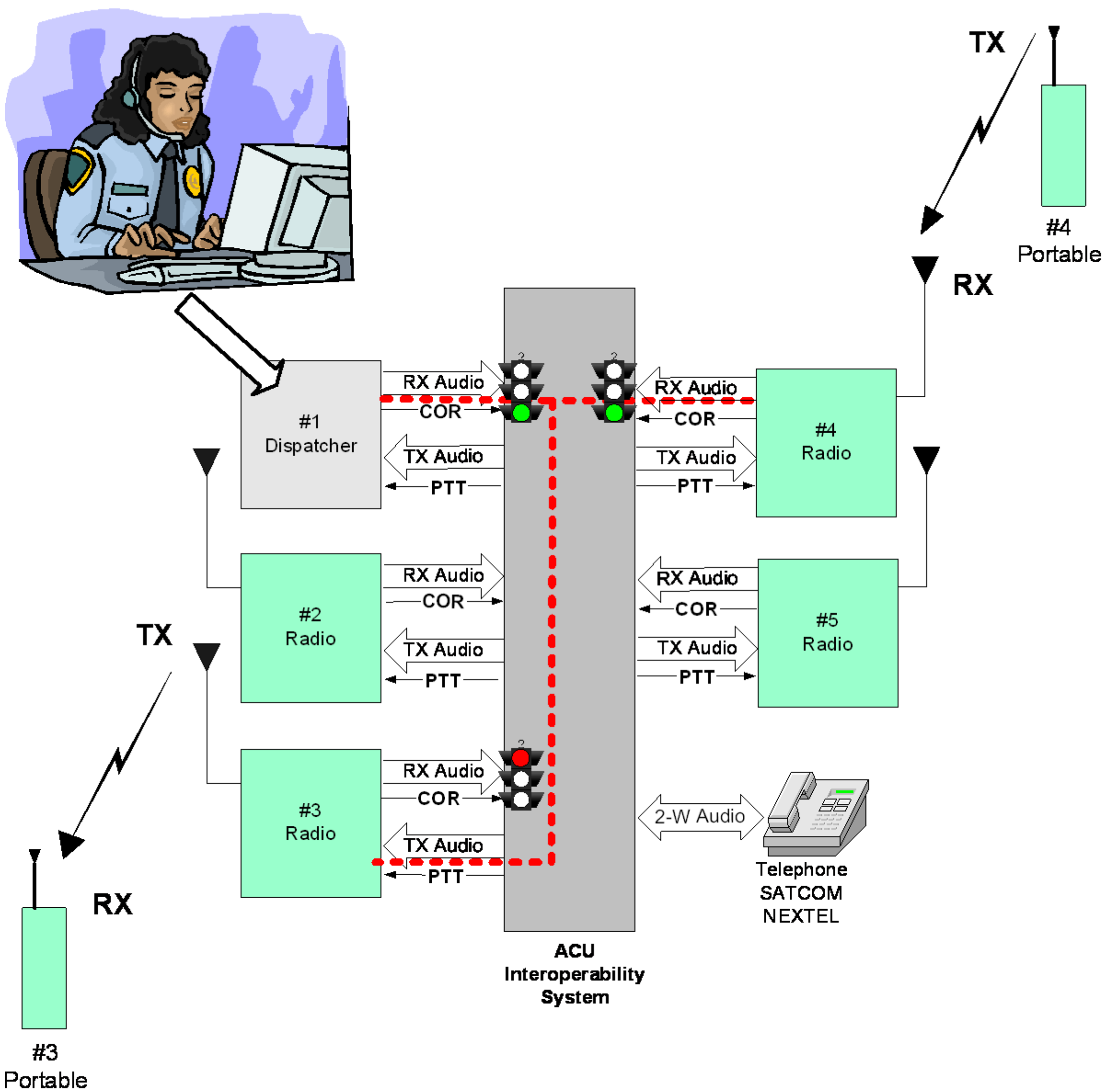


Figure 4-4 Dispatch Priority

Figure 4-4 shows a dispatch center interfaced to the #1 extension of the sample Interoperability System. The #1 extension has been configured for Dispatch Priority, and an active COR signal at this extension will not be blocked and will instead cause the dispatcher’s audio to be retransmitted, even if the #4 portable stays in the TX mode. The dispatcher’s audio will be transmitted instead, or will be mixed with existing incoming audio from any other DSP module also set to COR (Dispatch) Priority.

4.8 Poor Audio Quality

Clear communication is vital during an emergency. This section describes ways to optimize the clarity of spoken messages so that Interoperability System users are heard, heard correctly, and heard the first time, and not asked to repeat themselves. The ACU-2000 has a variety of options to improve audio quality.

4.8.1 Incompatible Audio Levels

When a conversation is taking place, especially a conference call between three or more people, clear communication is enhanced when all parties are heard at the same volume.

The ACU Controller's module settings screens provide a quick and simple means to set proper audio levels for each extension. It is preferable to adjust all extensions during initial setup rather than to try to modify individual interfaces later. This is because an interoperability cross-connection involves a variety of audio levels and it may be difficult to determine which level is at fault when not following a systematic process.

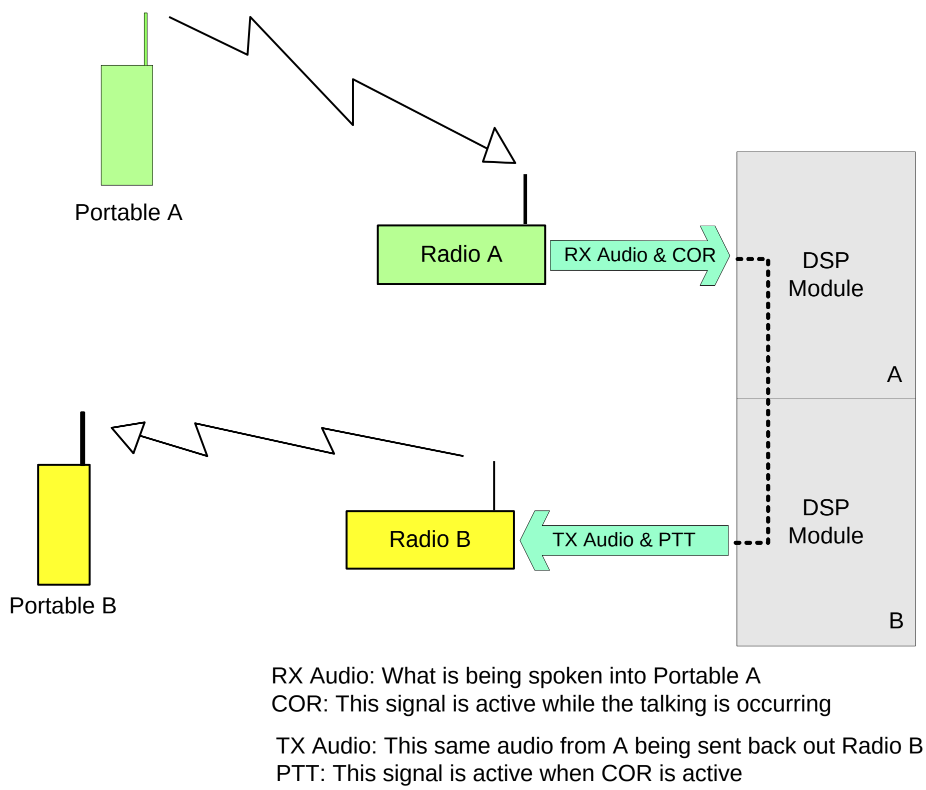


Figure 4-5 Audio Levels Involved With Each Cross-Connection

In the simple cross-connection depicted in Figure 4-5 the various volume levels to contend with are:

- How loud the person using Portable Radio A is talking
- The modulation limiting level of the Portable Radio A transmitter
- The RX audio output level of System Radio A (less problematic if a line level out is used; more so if the level is controlled by the speaker volume knob)
- DSP-2 Module A's RX audio level setting
- DSP-2 Module B's TX audio level setting
- The modulation limiting level of the System Radio B transmitter
- Portable Radio B's volume control

The above list is just for when the user of portable radio A is doing the talking. When the user of portable radio B replies, a second set of volume levels comes into play.

This procedure will focus on what is under the control of the ACU-2000 operator: the RX and TX audio levels of the DSP modules, and secondarily, the RX audio output level of the system radios.

For the cross-connect audio levels to be optimal, the internal audio levels on the audio busses are maintained at 0 dBm for both input and output. In general, each of the different types of modules has the means to adjust its respective input or output so that the module is interfacing the audio bus at a constant level of 0 dBm. For incoming, external audio, the SIGNAL LED on the front panel is the best indicator of the proper level for optimal performance. For outgoing transmit audio, the ACU-2000 operator only has control of the signal level presented to the transmitter and usually does not have the ability to make technical adjustments to the modulation controls.

Set RX Level Procedure

The RX (Receive) level must be optimized to allow best system operation. First of all, conversations, especially conference calls, will be more intelligible if all voices are at the same volume level. Second, VOX and VMR work best at the proper RX level.

- Cross-connect the HSP module to the DSP module being adjusted.
- Monitor the front panel of the DSP module while the radio is receiving a voice signal at a normal speaking volume level. Listen to the RX audio of the interfaced radio using the HSP handset.
- Watch the DSP front panel SIGNAL light. It should flicker with the incoming speech. If the level is too high, the LED will be on constantly during received speech. If too low, the LED will never come on, or will flicker only occasionally.
- Adjust the RX Level until the Signal LED flickers with incoming speech.

- Click *Apply* to save the setting.
- Is equalization or frequency shaping required? See Section 2.13.3.12.

> ***Note: If the interface is using speaker audio from the radio, the level will vary depending on the radio's volume control setting. Set the RX level in the DSP to 0 dBm, and then vary the radio volume level until the proper Signal LED indication is achieved. Note the setting, and keep the volume control at this setting.***

Set TX Level Procedure

The proper TX level is required to fully modulate the transmitter, but not over modulate it. Most radios have an audio limiter prior to the transmitter to prevent over modulation. Even with the limiter, some radios will still over modulate and some even shut off the TX signal when the input is too high. When the level is set too low the audio of the radio receiving the signal will be lower than normal, requiring that its volume control be turned up to an abnormal position. When the audio is too hot, the audio will sound squashed, or forced, and if the radio does not have a TX audio limiter, the audio will sound distorted and over-modulated.

- Cross-connect the HSP Module to the DSP Module being adjusted, and use the HSP Handset to key the radio while speaking at a normal volume level.
- Monitor to the TX audio of the interfaced radio on a receiver set to the radio's TX frequency.
- The quickest way to set the TX audio level is to use the ACU Controller to set the DSP Module's TX level to its lowest setting. Increase the TX level until the audio in the monitoring radio stops increasing in level. This is the threshold point where the limiter is preventing the TX level from going any higher. Leave the DSP Module's TX level at this threshold value.
- You may also follow the radio's recommended TX input audio setting procedure.
- Click *Apply* to save the TX Level setting.

4.8.1.1 Telephone Connection Audio Levels

If telephone audio levels do not seem to be optimum, perform the PSTN Simplified Setup Procedure carefully as described in Section 2.15. Verify that the telephone line losses and impedance are within standard limits. If problems still persist, please contact the JPS factory for assistance.

4.8.2 Noisy Received Signals

Certain types of radios such as HF, and AM typically contain a lot of noise in the demodulated signal. Even FM radios operating near their sensitivity limit will contain noise in the signal. The DSP-2 includes a noise reduction feature that should be considered if a particular channel is inherently noisy on a continuous basis.

The DSP-2 uses time domain mode noise reduction, designed to peak up any correlated information (such as speech), in the audio passband. It reduces noise by forming dynamic bandpass filters around correlated information, thus automatically reducing the bandwidth to the minimum necessary to pass the information. This type of noise reduction is most effective on purely random noise, such as white or pink noise, and less effective on impulse noises. The noise reduction value allows the amount of noise reduction to be set in ten steps from off to maximum. Increasing the level provides more actual noise reduction, but may give a surging quality to the recovered audio depending on its frequency content. Reducing the level lowers the noise reduction but may provide the best sounding audio in some cases. The best setting in a particular application depends on the noise level and represents a balance between noise reduction amount and ultimate audio quality.

The only method to find the correct amount of Noise Reduction to apply is to listen to the received signal as the level is changed; this is best done using the HSP Handset so that you can be sure that all noise heard is from the radio's received signal. Do not use the HSP speaker or a cross-connected radio. A little Noise Reduction goes a long way, and too much will give the received signal a fuzzy, artificial sound. It may be advantageous to attempt to improve the signal quality by other means (such as improving antenna placement) before adding Noise Reduction.

Noise Reduction Procedure

- The default setting for Noise Reduction is Off (no reduction). While listening to the received signal, increase the Noise Reduction setting one step at a time until the best signal quality is reached.
- If possible, listen to the receive signal from several different sources and determine the Noise Reduction setting that works for most.
- If the signal quality is later improved, revisit the Noise Reduction setting.
- Click *Apply* to save the setting.

4.8.3 Audio Equalization

For a given communications link, audio frequency shaping takes place in many different places. Typically, the microphone, associated amplifiers, modulation filters such as bandpass and preemphasis, receive deemphasis, receive audio amplifiers, and finally, the speaker or handset device. In general, the communications channel typically tries to optimize the 300 to 3000Hz range as this is where most of the information in speech is contained. In FM radios, in order to reduce noise, the higher frequencies are preemphasized, or boosted prior to transmission and then deemphasized, or rolled off after the discriminator audio output. The deemphasis not only brings the frequency shaping back to normal, but it has the advantage of rolling off any high frequency noise picked up during transmission.

Typically, the audio takeoff point for interface of an FM type receiver into a DSP module would either be at the discriminator output or the speaker output. If taken off at the discriminator output, it will still have the high frequency preemphasis which will give it a very tinny sound. Conversely, if taken off at the speaker, the audio shaping of a particular brand of radio may produce a more muffled sound that is hard to understand because it is lacking in high frequency information.

The DSP module includes an Audio Equalization feature that can either boost or roll off the high end of the RX audio spectrum. This adjustment can compensate for poor RX audio quality. Follow the procedure below to determine the proper Audio Equalization Setting.

Audio Equalization Procedure

The best way is to listen to the received audio in the HSP handset (not the HSP speaker, unless a high-quality external speaker is connected).

- Monitor the RX Audio in the HSP handset.
- If the audio sounds like it lacks treble, the high frequencies can be increased (boost).
- If the signal sounds too bright or harsh, the high frequencies can be attenuated (cut).
- There are 3 steps of boost and 3 steps of cut plus the default flat setting. Move the adjustment 1 step at a time and recheck for best sound.

4.9 Unintended Consequences

4.9.1 Unwanted Connections

The primary cause of unwanted connections that were not initiated by the operator or authorized user is DTMF received via the radio channels connected to the ACU-2000. It is possible that legitimate users are using DTMF for selective call, paging, or other functions within their own system and the sequence of numbers accidentally corresponds to an ACU-2000 command. DTMF can also be falsely detected as a result of intermodulation or other co-channel interference; for this reason, all commands include at least two DTMF characters detected within a short time period. In many cases, legitimate users of the system will be unaware that the use of their DTMF keypads is causing problems in the ACU-2000 system.

This is easily prevented by simply setting all modules in the ACU-2000 that have a DTMF decoder function to *DTMF disable*.

If it is essential for field personnel to have the ability to control the system via DTMF over radio, then a measure of protection can be gained by enabling DTMF detection only on the modules that the authorized people will access, and assign suitable security levels and PIN numbers.

End of Section Four

5 Appendix

5.1 Programming Configuration Settings via the HSP Keypad

The preferred setup method is via a computer using the ACU Controller software, but some system configuration can be performed using the HSP module keypad.

5.1.1 Keypad Programming Instructions:

1. Enter the programming mode by pressing *** 9 9** on the HSP keypad. The ACU-2000 responds with the voice prompt *Setup Mode*. Each time a user successfully enters one of the programming commands the ACU responds with *Ready*. The ACU-2000 stays in programming mode until the user exits this mode by entering the *** #** (star pound) sequence. The user does not have to enter *** #** until all programming is complete. The configuration changes are not entered into non-volatile memory until the *** #** sequence is entered. All programming configuration changes are then automatically stored in non-volatile memory. These settings are retained unless new settings stored or a *Reset to Factory Settings* command is entered.

> ***Note: If the ACU-2000 is currently operating under remote control via either the JPS ACU Controller or WAIS Controller programs, programming via the HSP is disabled and the Invalid Entry voice prompt will be heard. To override console control, enter * 3 7 and wait for the Ready prompt. Programming via the HSP can then begin. Any configuration changes made will not be seen via either controller program until it executes the Retrieve Current Configuration command from the File pull-down menu, or until it is terminated and restarted.***

2. When in the programming mode, select the individual module you wish to program by entering ** 0 1 n n*, where *n n* is the two-digit extension number of the module to be programmed. The extension numbers will be from 01 through 12 in a single chassis system and 01 through 24 in an expanded system, as the Master Unit in an expanded system contains the programming configuration for the Expansion Unit. For example, to set a parameter on a DSP module that is installed in extension 05, the user first enters the *Select Module to Program* command ** 0 1 0 5*. Once this command is given user may then enter as many configuration items as desired for the selected module. If the user mistakenly selects an extension that is empty, or there is another reason why the selected module is not valid for programming, the ACU-2000 will respond with an error message. There are no configuration items for the HSP module.

> ***Note: The System Programming Items at the start of Table 5-1 are system-wide programming commands that do not require the selection of a module. To execute these items, enter the programming mode, but do not select a particular module.***

3. Now that a module is selected, begin actual programming. Enter the desired programming command, following the format described in Table 5-1.

   Continuing the example from step 2 above:

   To set the receive level to 0 dBm on a DSP module located in extension 05, the user first selects this module by entering ** 0 1 0 5*, the ACU then responds with *05 Ready*, the user then sets the receive level to 0 dBm by entering ** 0 2 3*. The ACU responds with *Ready.*

4. When all parameters for a selected module are complete, another module can be chosen for programming as in step 2 above, or the user can exit the programming mode at this time, which will store all settings. This is accomplished by entering ** #***.** The ACU responds with *Saving configuration* followed by *Configuration has been saved.*

> ***Note: If the programming mode is not exited by pressing * # before the power to the ACU is turned off, none of the new configuration settings made will be saved.***

Table 5-1 ACU-2000 System Programming items

System Programming Item	Command	N = Selection	Factory
Enter Programming Mode	* 9 9	None	N/A
Store Current Connections	*36	None	N/A
Console Override	* 3 7	None	N/A
Select Module to Program	* 0 1 n n	n n = slot extension (two digits must be entered)	N/A
Exit Programming Mode	* #	None	N/A
Reset Modules to Factory Settings	* 9 9 9 9	None	N/A
Enable System PIN numbers	* 2 9 n	0 = Disable PIN numbers, 1 = Enable PIN numbers in *Priority* operation, 2 = Enable PIN numbers in *Exclusive* operation	Disabled
Program PIN numbers	* 3 0 nnnnx	nnnn is the four digit PIN, x is the security level from 0 to 9, 0 = not secure (PIN not required), 1=least secure, 9 = most secure.	PIN Database Cleared
Delete PIN numbers	* 3 1 nnnn	nnnn is the four digit PIN	N/A
Set CPM RS-232 Baud Rate	* 5 4 n	n signifies the Baud Rate (see text)	115200
Stored Connections Auto Restore at Power-up	* 5 5 n	0 = Feature is disabled 1 = Auto Restore is enabled.	Enabled

5.1.2 System Programming Items

This section explains system programming items of Table 5-1; they are used only to assist in setting the configuration items of Table 5-2. The DSP-2IP, PSTN-2IP, and LP-2IP modules can be configured by the HSP keypad following the procedure described.

5.1.3 Enter Programming Mode

Use the command * *9 9* to enter the programming mode. Once in this mode, the programming of each individual module's configuration items is possible. Unless otherwise specified, none of the programming items listed in Table 5-1 or Table 5-2 can be entered unless the ACU-2000 is in the Programming Mode.

5.1.3.1 Controller Override

If the ACU-2000 is currently under remote control using the ACU Controller software, programming via the HSP is disabled. To override the console software and re-enable programming via the HSP, enter * *3 7* at the HSP keypad. Configuration changes made during console override will not be available at the console screen until the *Retrieve Current Configuration* option is selected from the *File* pulldown menu.

5.1.3.2 Select a Module to Program

After entering the programming mode, use the command * *0 1 n n* to select the slot number of the module you wish to program, where *nn* is the two-digit slot number. For example, to program the module in slot 5, enter * *0 1 0 5*. After all programming commands are complete for one module; the programming of another module may begin after using this command to select that module. It is not necessary to exit and re-enter the programming mode each time a new module is programmed.

5.1.3.3 Exit Programming Mode

When programming is complete, use the command * # to exit the programming mode and store the configuration in non-volatile memory.

5.1.3.4 Reset Modules to Factory Settings

The command * *9 9 9 9* causes all modules in the ACU-2000 chassis to be reset to the factory settings. Be careful, as using this command will erase all custom configuration programming. This command can only be issued from programming mode.

5.1.4 PIN Security

This command configures PIN operation for the ACU-2000. PIN numbers *(Personal Identification Numbers)* are used to control DTMF access to the ACU-2000 system. Enable PIN numbers in the *Priority Mode* by entering ** 2 9 1*, or in the *Exclusive Mode* by entering ** 2 9 2*. To Disable PIN numbers enter ** 2 9 0*. When either PIN mode is enabled, users attempting to access the system will be prompted by the ACU-2000 to enter their PIN. When in the *Priority PIN Mode* the user's password security level must be equal to or higher than the security level of the module to gain access. *Exclusive PIN Mode* operation requires the user's password security level is identical to the security level of the extension the user is attempting to access. Extension security levels are set for all types of interface modules using the ** 3 2 n* configuration item. Instructions included with each module's list of configuration items.

5.1.4.1 Program PIN Numbers

The security level of each PIN is entered into the ACU-2000 database by this command. Up to 20 different PIN numbers may be entered. To enter a PIN into the database and/or set the level for the PIN, enter ** 3 0 n n n n x*, where *n n n n* is the four digit PIN and *x* is the security level to be associated with this PIN. There are nine available security levels, ranging from 1 to 9, where 1 is least secure and 9 is most secure. The security level 0 is essentially meaningless as an extension set to security level 0 does not require a PIN to gain access. PIN programming is a global command, meaning it's not necessary to select a module to program when entering PIN numbers or setting PIN security levels.

5.1.4.2 Delete PIN Numbers

To delete a particular PIN, enter ** 3 1 n n n n*, where n n n n is the four digit PIN. This is a global command. Note it is not necessary to input the security level when deleting a PIN.

5.1.4.3 Module Security Level Selection

This command sets a module's security level. Enter ** 3 2 n*, where *n* is the security level, with 0 = not secure (no PIN required), 1 = least secure, up to 9 = most secure. This is not a global command, each module's security level must be separately set. To set the security level for a particular module, first use the *Select Module to Program* command.

5.1.5 Chassis Baud Rate

The **54* command allows the Baud rate of the CPM RS-232 serial interface to be set using the HSP-2 keypad. The Baud rate can also be set using a web browser; this keypad method has been added for instances where an Ethernet Connection is inconvenient. Once the setup mode has been entered, simply enter **54n* to set the Baud rate, with n defined as:

***540**	300 Baud
***541**	1200 Baud
***542**	2400 Baud
***543**	4800 Baud
***544**	9600 Baud
***545**	19200 Baud
***546**	38400 Baud
***547**	57600 Baud
***548**	115200 Baud (factory default Baud rate).

5.1.6 Store Current Connections and Auto-Restore of Stored Connections at Power Up

Whenever the *Stored Connections Auto-Restore* feature is enabled, the ACU-2000 will automatically restore the cross-connection status of all modules to a user-programmed setting whenever the ACU power is cycled. These Stored Connections are saved by use of the **36* HSP command.

To store connections, simply set the ACU-2000 to the connection state that it should automatically return to after each power cycle and press *36 on the HSP keypad. Do not enter programming mode first. The unit will respond with a pair of voice prompts; *"Saving configuration"* when the process starts, and *Setup has been saved"* when the process is complete. If it ever becomes necessary to modify the stored connections, simply repeat this process. If the preference is to have no connections at power up, simply break all connections prior to the *36 keypad entry.

It's also possible (but not necessary) to use the **55* programming item. It allows the user to program the ACU-2000 to Enable or Disable the *Store Current Connections* feature via the HSP keypad.

The user has two options:

- ***550** Auto-restore is disabled.
- ***551** Auto-restore is enabled. (factory default setting)

The default stored connection status is *no cross-connections*, so unless a set of stored cross-connections has been entered using the **36* command, the **55*commands will have no effect, as every power cycle will bring the unit back with no connections in place. If set to disabled, again the unit will power up with no connections in place, despite any stored connections set by the *36 command.

5.1.7 Extended Addressing Chassis Number

This feature is used to set the Chassis Number for the ACU. This feature is useful when several ACU chassis are in a system, some of which have modules that are connected to radios on the same frequency. This feature is very handy when you have multiple ACUs that are in areas where the radio coverage overlaps. The user can cross connect radios by using the cross connect command utilizing DTMF, but obviously all radios on that same frequency would hear it, and there would be multiple ACUs accepting the command.

In this situation, a user selectable 2-digit chassis address can be used so that, when enabled, the user actually enters 4 digits when requesting a cross connect. The first two digits are the chassis number and the last two digits are the port number.

The Chassis Number can range from 40 to 79. Setting the Chassis Number to 00 will disable this feature. User options are:

- ***5600** Disabled
- ***56nn** Set to *nn* if *nn* is in the range for 40 to 79

The factory default is for Extended Addressing to be disabled.

5.1.8 SCM-2 Direct Connection Module

This feature allows the SCM-2 module to be automatically connected to another module whenever it receives a call. Enable this feature by entering **58nn*, where *nn* is the slot number of the module to connect to. This feature can be disabled by entering **58#*. The factory default is no direct connection module.

Table 5-2 Configuration Items

DSP Configuration Item	Command	N = Selection	Factory
Receive Audio Level	* 0 2 n	0 = 12dBm, 1 = 8dBm, 2 = 4dBm, 3 = 0dBm 4 = -4dBm, 5 = -8dBm, 6 = -12dBm, 7 = -16dBm, 8 = -20dBm, 9 = -26dBm	0dBm
Transmit Audio Level	* 0 3 n	0 = -26dBm, 1 = -20dBm, 2 = -16dBm, 3 = -12dBm, 4 = -8dBm, 5 = -4dBm, 6 = 0dBm, 7 = 4dBm, 8 = 8dBm, 9 = 12dBm	0dBm
COR Polarity	* 0 4 n	0 = Active Low, 1 = Active High	Active Low
Full/Half Duplex	* 0 8 n	0 = Full, 1 = Half	Half
DTMF Mute Timer Value	* 0 9 n	0 = Off, 1 = 0.5 Sec, 2 = 1 Sec, 3 = 1.5 sec, 4 = 2 sec, 5 = 2.5 sec, 6 = 3 sec, 7 = 3.5 s, 8 = 4 s, 9 = 4.5 sec	Off
Audio Delay H/W COR Mode	* 1 0 n	0 = 20 ms, 1 = 60 ms, 2 = 100 ms, 3 = 140 ms, 4 = 180 ms, 5 = 220 ms, 6 = 260 ms, 7 = 300 ms	20 ms
Audio Delay VOX Mode	* 1 0 n	0 = 20 ms, 1 = 60 ms, 2 = 100 ms, 3 = 140 ms, 4 = 180 ms, 5 = 220 ms, 6 = 260 ms, 7 = 300 ms	60 ms
Audio Delay VMR Mode	* 1 0 n	Less than 220 ms not allowed. 0,1,2,3,4,5 = 220 ms, 6 = 260 ms, 7 = 300 ms	220 ms
VMR/VOX Threshold	* 1 1 n	0 = Low (Highest Sensitivity), 1 = Med1, 2 = Med2, 3 = High (Lowest Sensitivity), 4-9 = Reserved for special applications – do not use	Med1
VOX Hang Time	* 1 2 n	0 = 175 ms, 1 = 375 ms, 2 = 575 ms, 3 = 775 ms,	775 ms

Table 5-2 Configuration Items			
DSP Configuration Item	**Command**	**N = Selection**	**Factory**
		4 = 975 ms, 5 = 1.175 sec, 6 = 1.375 s, 7 = 1.575 s	
VMR Hang Time	* 1 2 n	Less than 775 not allowed, 1, 2, 3 = 775 ms, 4 = 975 ms, 5 = 1.175 sec, 6 = 1.375 sec, 7 = 1.575 sec	775 ms
COR (squelch) Type	* 1 4 n	0 = COR, 1 = VMR, 2 = Reserved, 3 = VOX	VOX
COR Sampling On/Off	* 1 8 n	0 = Disabled, 1 = Enabled	Disabled
COR Sampling Initial Delay Time	* 1 9 n	0 = 2 sec, 1 = 4 sec, 2 = 6 sec, 3 = 8 sec, 4 = 10 sec, 5 = 12 sec, 6 = 14 sec, 7 = 16 sec, 8 = 18 sec, 9 = 20 sec	10 sec
COR Sampling Interval	* 2 0 n	0 = 1 sec, 1 = 2 sec, 2 = 3 sec, 3 = 4 sec, 4 = 5 sec, 5 = 6 sec, 6 = 7 sec, 7 = 8 sec, 8 = 9 sec, 9 = 10 sec	5 sec
COR Sampling Window Width	* 2 1 n	0 = 50 ms, 1 = 100 ms, 2 = 150 ms, 3 = 200 ms, 4 = 250 ms, 5 = 300 ms, 6 = 350 ms, 7 = 400 ms, 8 = 450 ms, 9 = 500 ms	150 ms
Noise Reduction Value (Peaker Value)	* 2 2 n	0 = Off, 1 = Minimum… 9 = Maximum	Off
Audio Muted when Squelched	* 2 3 n	0 = Muted, 1 = Not Muted	Muted
Transmit Keying Tones	* 2 5 n	0 = None, 1 = 1950 Hz Continuous, 2 = EIA Sequence (F1 function tone, 1950 Hz)	None
COR Inhibit Time after PTT	* 2 6 n	0 = None, 1 = 100 ms, 2 = 200 ms, 3 = 400 ms, 4 = 800 ms, 5 = 1 sec, 6 = 2 sec, 7 = 3 sec, 8 = 4 sec, 9 = 5 sec	100 ms
PTT or COR Priority (Half Duplex only)	* 2 7 n	0 = COR Priority, 1 = PTT Priority.	PTT Priority
Keying Tone Amplitude	* 2 8 n	0 = -6 dB, 1 = -9 dB, 2 = -12 dB, 3 = -15 dB Does not apply to EIA Keying	-9 dB
Module security level	* 3 2 n	0 = Not Secure, 1 =Least Secure, 9 = Most Secure	Not Secure
DTMF Enable	* 3 8 n	0 = Disabled, 1 = Enabled	Enabled
DSP Configuration Item (continued)	**Command**	**N = Selection**	**Factory**
High Frequency Equalizer	* 3 9 n	0 = Reserved, 1 = 5 dB cut, 2 = 3.5 dB cut, 3 = 2 dB cut, 4 = Flat, 5 = 2 dB boost, 6 = 3.5 dB boost, 7 = 5 dB boost, 8 and 9 = Reserved.	Flat
DTMF Pre-emphasis (HSP keypad only)	* 4 0 n	0 = DTMF Pre-emphasized 1 = DTMF Not Pre-emphasized	Pre-emphasis
Auxiliary Output Control	* 4 1 n	0 = Future option 1 = Local control by the module	Local Control
TX Audio Delay (was "Radio Type Selection")	* 4 3 n	0 = No Delay, 1 = 200 ms, 2 = 400 ms, 3 = 600 ms, 4 = 800 ms 5 through 9 reserved for future use	No Delay
Voice Prompt Initiation Delay	* 4 4 n	0 = No Delay, 1 = 50 ms, 2 = 100 ms, 3 = 500 ms, 4 = 750 ms, 5 = 1 sec, 6 = 2 sec, 7 = 3 sec, 8 = 4 sec, 9 = 5 seconds	100 ms
PSTN Configuration Item	**Command**	**n = Selection**	**Factory**
Telephone Line Levels	* 0 2 n	0 = 0dBm, 1 = -3dBm, 2 = -6dBm, 3 = -9dBm, 4 = -12dBm, 5 = -15dBm, 6 = -18dBm, 7 = -21dBm, 8 = -24dBm	-9dBm
Telephone RX Level Boost	* 0 3 n	0 = 0 dB, 1 = 2.5 dB, 2 = 4.5 dB, 3 = 6 dB, 4= 7.4 dB, 5= 8.5 dB, 6= 9.5 dB, 7 = 10. 5 dB,	6 dB

Table 5-2 Configuration Items			
DSP Configuration Item	**Command**	**N = Selection**	**Factory**
		8= 11.3 dB, 9 = 12 dB	
PSTN Type	* 0 5 n	0 = Normal, 1 = Satcom	Normal
PSTN Dialing Mode	* 0 6 n	0 = DTMF, 1 = Pulse	DTMF
DTMF Mute Timer	* 0 9 n	0 = Off, 1 = 0.5 sec, 2 = 1 sec, 3 = 1.5 sec, 4 = 2 sec, 5 = 2.5 sec, 6 = 3 sec, 7 = 3.5 s, 8 = 4 s, 9 = 4.5 s	1 sec
Audio delay Time	* 1 0 n	0 = 10 ms, 1 = 22 ms, 2 = 35 ms, 3 = 47 ms, 4 = 60 ms, 5 = 72 ms, 6 = 85 ms, 7 = 97 ms	35 ms
VOX Threshold	* 1 1 n	0 = VOX Off, 1 & 2 = Low, 3 = High, 9 = VOX Off	Low
VOX Hang Time	* 1 2 n	0 = 500 ms, 1 = 1 sec, S, 2 = 1.5 sec, 3 = 2.0 sec	1 sec
Two Wire Operation	* 2 4 n	0=2-Wire	2-Wire
Module security level	* 3 2 n	0 = Not Secure, 1 =Least Secure, 9 = Most Secure	Not Secure
Outgoing Ring Time	* 3 7 n	0 = No ring, 1 = 30 sec, 2 = 60 sec, 3 = Continuous	30 seconds
DTMF Enable	* 3 8 n	0 = Disabled, 1 = Enabled	Enabled
Auxiliary Output Control	* 4 1 n	0 = Future option 1 = Local control by the module	Local Control
Inactivity Disconnect Timer	* 4 2 n	0 = None, 1 = 30 sec, 2 = 1 min, 3 = 2 min, 4 = 5 min, 5 = 10 min, 6, 7, 8 & 9 = Reserved	2 Minutes
Voice Prompt Initiation Delay	* 4 4 n	0 = No Delay, 1 = 50 ms, 2 = 100 ms, 3 = 500 ms, 4 = 750 ms, 5 = 1 sec, 6 = 2 sec, 7 = 3 sec, 8 = 4 sec, 9 = 5 seconds	No Delay
LP Configuration Item	**Command**	**n = Selection**	**Factory**
DTMF Mute Timer	* 0 9 n	0 = Off, 1 = 0.5 sec, 2 = 1 sec, 3 = 1.5 sec, 4 = 2 sec, 5 = 2.5 sec, 6 = 3 sec, 7 = 3.5 s, 8 = 4 s, 9 = 5 s	1 second
Audio Delay Time	* 1 0 n	0 = 10 ms, 1 = 35 ms, 2 = 60 ms, 3 = 85 ms, 4 = 110 ms, 5 = 135 ms, 6 = 160 ms, 7 = 185 ms	60 ms
VOX Threshold	* 1 1 n	0 = VOX Off, 1 = Low, 2 = Med, 3 = High, 9 = Off	Med
VOX Hang Time	* 1 2 n	0 = 10 ms, 1 = 750 ms, 2 = 1.5 sec, 3 = 2.25 sec	750 ms
Module security level	* 3 2 n	0 = Not Secure, 1 =Least Secure, 9 = Most Secure	Not Secure
Dial and Busy Tone Style	* 3 3 n	0 = USA Style, 1 – 9 = Reserved	USA
Ring Cadence	* 3 4 n	0 = USA Style, 1 = European Style, 2 – 9 = Reserved	USA
Dial Tone Enable	* 3 5 n	0 = Dial Tone Disabled, 1 = Dial Tone Enabled	Enabled
Ringback Enable	* 3 6 n	0 = Ringback Disabled, 1 = Ringback Enabled	Enabled
Outgoing Ring Time	* 3 7 n	0 = No ring, 1 = 30 sec, 2 = 60 sec, 3 = Continuous	30 seconds
DTMF Enable	* 3 8 n	0 = DTMF Disabled, 1 = DTMF Enabled	Enabled
Auxiliary Output Control	* 4 1 n	0 = Future option 1 = Local control by the module	Local Control
Voice Prompt Initiation Delay	* 4 4 n	0 = No Delay, 1 = 50 ms, 2 = 100 ms, 3 = 500 ms, 4 = 750 ms, 5 = 1 sec, 6 = 2 sec, 7 = 3 sec, 8 = 4 sec, 9 = 5 seconds	No Delay
SCM-2 Configuration Item	**Command**	**n = Selection**	**Factory**
VMR/VOX Threshold	* 1 1 n	0 = Low (Highest Sensitivity), 1 = Medium, 2 = High (Lowest Sensitivity), 3-9 = Reserved for special applications – do not use	Medium
VMR/VOX Hang Time	* 1 2 n	0 = 0 ms, 1 = 200 ms, 2 = 400 ms, 3 = 600 ms, 4 = 800 ms, 5 = 1 sec, 6 = 1.2 s, 7 = 1.4 s, 8 = 1.6 a, 9 = 1.8 s	200 ms
COR (squelch) Type	* 1 4 n	0 = RTP+VOX, 1 = VMR, 2 = RTP, 3 = VOX, 4 = RTP+VMR, 5 = DISABLE, 6 = Reserved, 7 = PACKET	VOX

Table 5-2 Configuration Items

DSP Configuration Item	Command	N = Selection	Factory
Connection Type	* 5 7 n	0 = Normal, 1 = SIP Direct, 2 = Backplane Direct, 3 = Both	Normal
Direct Connection Module	*5 8 nn	This feature allows the SCM-2 module to be automatically connected to another module whenever it receives a call. Enable this feature by entering **58nn*, where *nn* is the slot number of the module to connect to. This feature can be disabled by entering **58#*.	The factory default is no direct connection module.

End of Manual

6 Index

End of Manual