



Avaya Solution & Interoperability Test Lab

Application Notes for Raytheon JPS ACU-2000IP with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Raytheon JPS ACU-2000IP Intelligent Interconnect System to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services. This integration enables customer calls and calls throughout the enterprise connected to Avaya Communication Manager, to be routed directly to field emergency responders like police, fire and EMT staff equipped with radios. In addition, radio-equipped responders can place calls to stations throughout the enterprise connected to Avaya Communication Manager as well as contacts outside the enterprise.

The Raytheon JPS ACU-2000IP Intelligent Interconnect System is a gateway that provides linking of radios over an IP network, control of interoperability systems via IP, change of remote channels over IP, and the ability to interface radios via SIP.

The Raytheon JPS ACU-2000IP is an integral part of the Avaya Public Safety Solution. These Application Notes present a sample configuration for a public safety environment.

Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab at the request of the Solutions Marketing Team.

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1. Introduction

These Application Notes describe the configuration steps required for Raytheon JPS ACU-2000IP Intelligent Interconnect System to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services. This integration enables customer calls and calls throughout the enterprise connected to Avaya Communication Manager, to be routed directly to field emergency responders like police, fire and EMT staff equipped with radios. In addition, radio-equipped responders can place calls to stations throughout the enterprise connected to Avaya Communication Manager as well as contacts outside the enterprise.

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The Raytheon JPS ACU-2000IP is an integral part of the Avaya Public Safety Solution. These Application Notes present a sample configuration for a public safety environment.

1.1. Public Safety Solution Overview

The Avaya Public Safety solution is designed to help government and private agencies responsible for the delivery of public safety services to enterprises and civilian populations. This includes:

- ✦ **Avaya Contact Center** applications such as expert agent selection to ensure the most qualified and most available resource rapidly attends to the case.
- ✦ **PlantCML Sentinel CM and Intelligent Work Station** integration which provides public safety community with call-center solutions designed to streamline emergency call-taking. Sentinel CM is a 911 incident management solution, and integration with Avaya Communication Manager is achieved through the Avaya Application Enablement Services (AES) Telephony Services Application Programming Interface (TSAPI) & Device, Media and Call Control Interface (DMCC) services.
- ✦ **Raytheon JPS ACU-2000IP Intelligent Interconnect System** integration which provides seamless communication across traditionally disparate communications such as Land-based Mobile Radio (LMR). Raytheon JPS ACU-2000IP is a radio IP/SIP gateway that allows IP-PBX stations to interface with radios. Multiple interface cards allow all radios to be a part of the IP/PBX system, acting as a SIP or VoIP station or device. Integration with Avaya Communication Manager is achieved through the SIP Enablement Services (SES).
- ✦ **NICE CLS/VoIP Logger** integration for secure recording of audio on the entire chain of service delivery from the conversations with the citizen, to command and control and dispatch, to resolution.

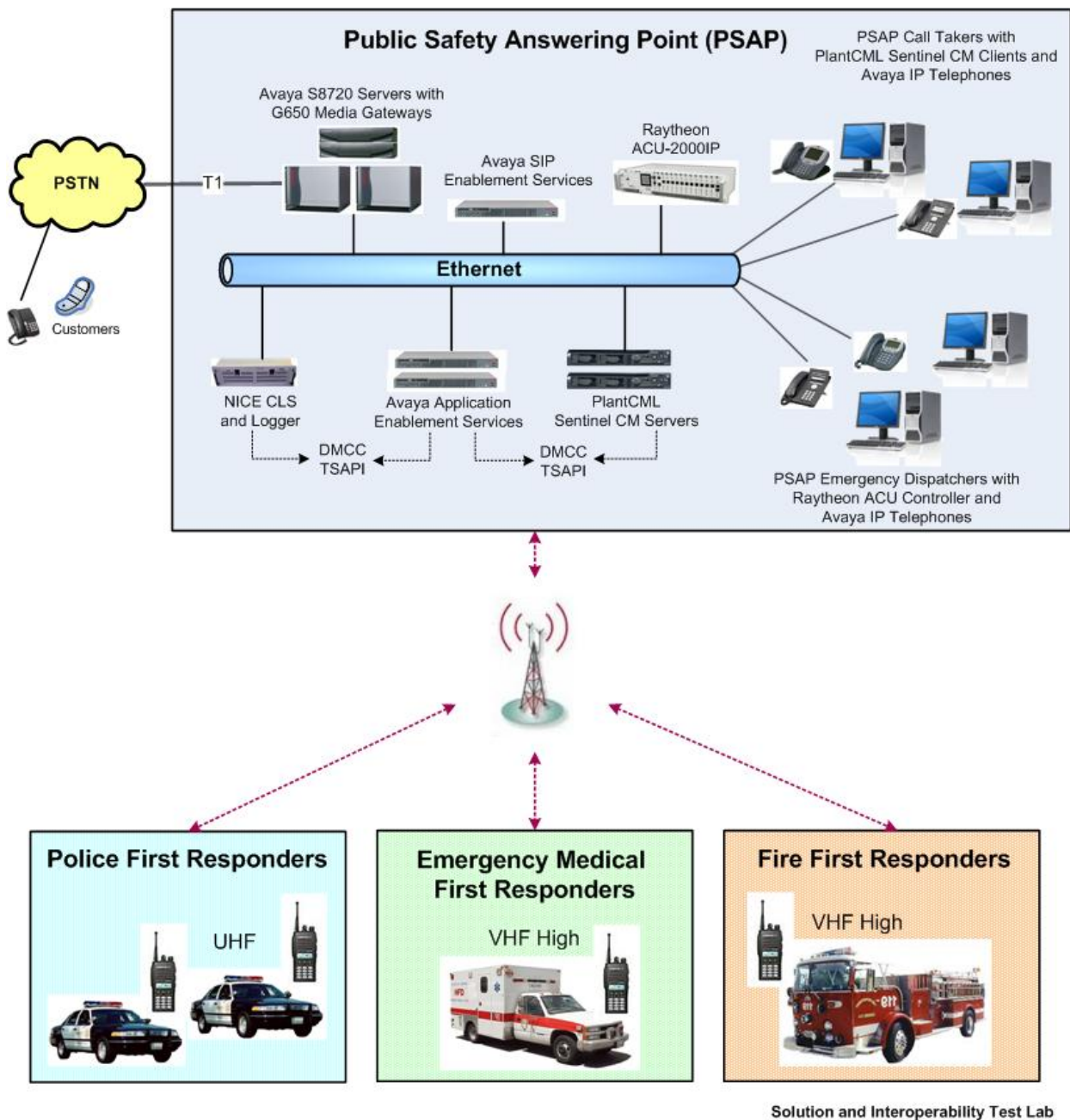


Figure 1: Avaya Public Safety Solution Reference Configuration

1.2. Raytheon JPS ACU-2000IP Intelligent Interconnect System Overview

The ACU-2000IP Intelligent Interconnect System includes:

- ACU-2000IP Gateway
- ACU-2000 Controller
- Radios

1.2.1. ACU-2000IP Gateway

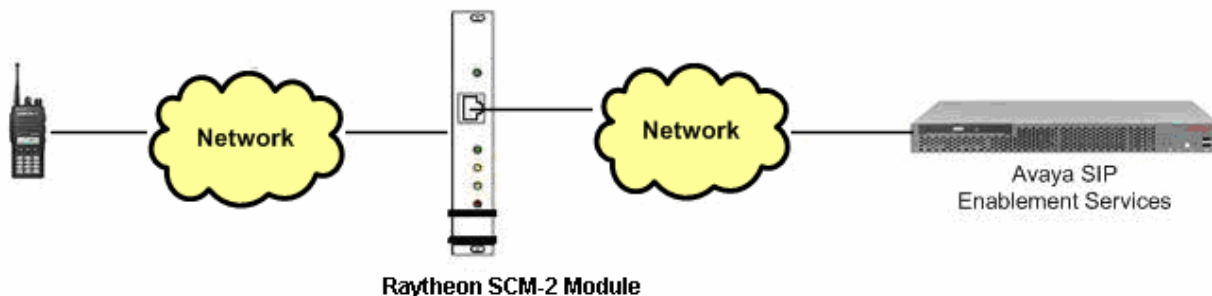
The ACU-2000IP gateway provides linking of radios over an IP network, and the ability to interface radios via SIP. The ACU-2000IP consists of a chassis with a number of plug-in modules. The essential modules reside on the left side of the chassis, and consist of the PSM, HSP, and CPM modules with the following functions:

- Power Supply Module (PSM) is the power supply for the chassis. This module is not duplicated. The PSM module does have the ability to switch over to DC operation if needed.
- Headset/Speaker/Prompt (HSP) Module provides a local interface to the system. The keypad, handset and speaker on this module allow a local operator to communicate with other system users like radios.
- Control Processor Module (CPM) controls the unit and relays status and control messages.

The rest of the chassis is occupied by up to 12 interface modules. These interface modules could be DSP or SCM which provide following functions:

- Digital Signal Processor (DSP) Module interfaces radios and other 4-wire devices to the ACU-2000IP. The DSP has a front panel RJ-45 connector that allows it to create a network Radio over IP (RoIP) link. It also has rear panel four-wire interface (15-pin D-sub connector) for radios.
- SIP Control Module-2 (SCM-2) creates an interface between a radio and an IP network, using the SIP protocol to initiate and manage connections. The SCM-2 modules have no provision to cross-connect with other modules within the chassis; they always interface their single associated radio to the IP network and any multi-party conferences that are put together by other systems in the network, such as Avaya Communication Manager.

Each SCM-2 interface module connects to a single radio device on the one end, and to Avaya SES on the other end as a SIP endpoint, as shown below. The ACU-2000IP will register the SCM-2 modules to Avaya SES at system start up. The ACU-2000IP supports the GSM and G.711MU codec, and only the G.711MU codec will be used in the public safety testing.



1.2.2. ACU-2000 Controller

The ACU-2000 Controller provides a GUI interface to control the ACU-2000IP. The main screen provides a clear overview of current system status with icons to identify the various system components. Cross-connections are made and disconnected by simple point-and-click procedures. The ACU-2000 Controller is a client application which can operate on any PC running Windows 2000, or Windows XP operating systems.

1.2.3. Radios

The ACU-2000IP supports UHF and VHF-High radios. These radio devices are half duplex and the ACU-2000IP gateway provides a cross-connection between the radios and the IP-PBX. There are several manufacturers of radios. Vertex UHF and VHF-High radios were used in this testing.

2. Network Topology

The network implemented for the reference configuration is shown in **Figure 2**. The Public Safety Answering Point location consists of Avaya S8720 Servers controlling G650 Media Gateways. The PSAP location is also equipped with Avaya SIP Enablement Services (SES) server, Avaya IP phones, Raytheon JPS ACU-2000IP gateway and ACU-Controller running on a Windows XP machine.

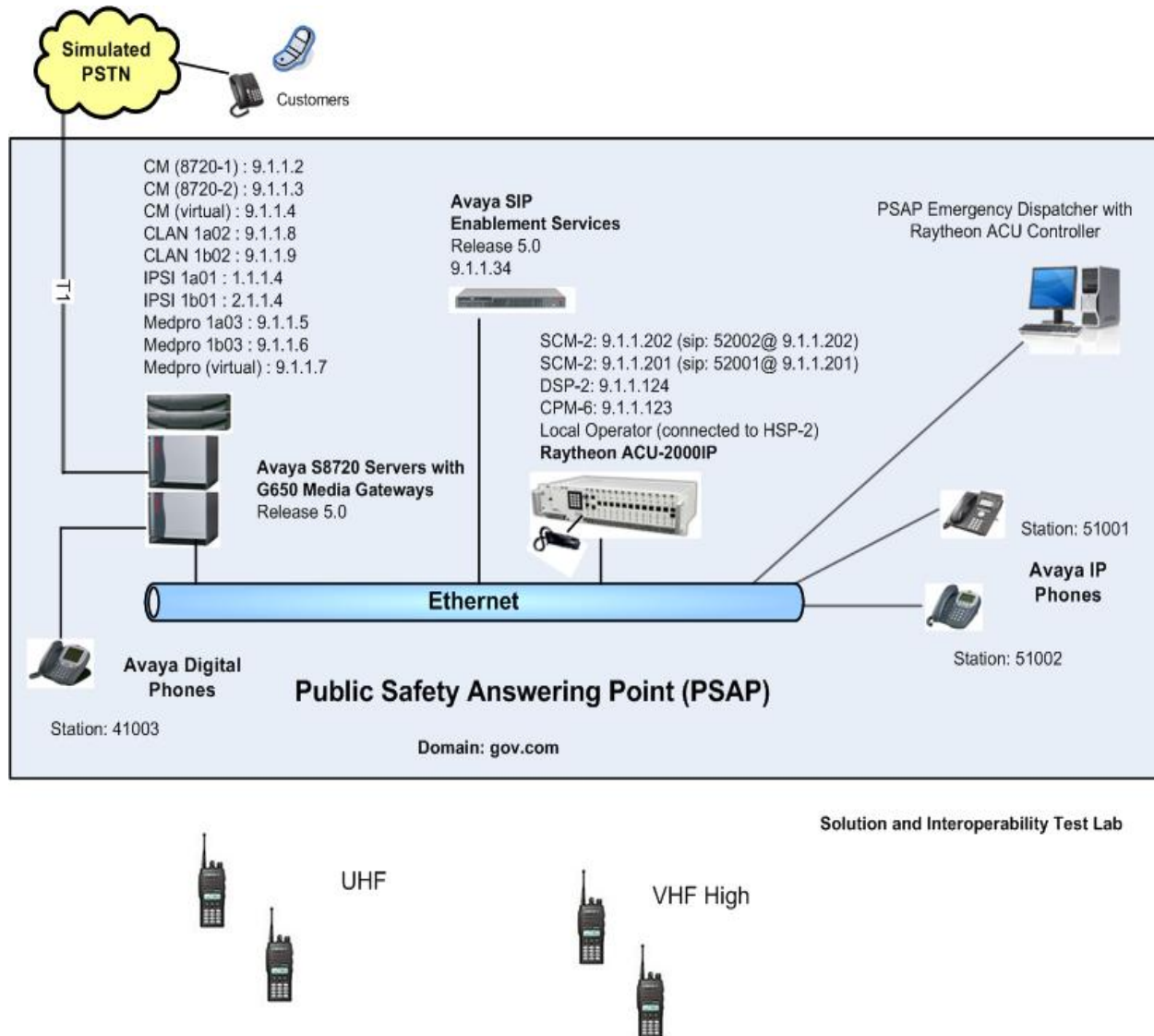


Figure 2: Network Diagram for JPS ACU-2000IP with Avaya Communication Manager and Avaya SES

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Device Description	Versions Tested
Avaya Communication Manager - S8720 Servers	Release 5.0 (R015x.00.0.825.4)
Avaya G650 Media Gateway - IPSI (TN2312BP) - CLAN (TN799DP) - MedPro (TN2602AP)	- HW15 FW039 - HW01 FW156 - HW02 FW033
Avaya SES (Combined Home-Edge)	Release 5.0 (825.31)
Avaya 4600 Series H.323 Telephones	R2.8
Avaya 9600 Series H.323 Telephones	R1.5
Avaya IP Softphone	R6.0
Avaya 6211 Analog Telephones	N/A
Avaya 2420 Digital Telephones	N/A
Raytheon JPS ACU-2000IP - CPM-6 - SCM-2 - DSP-6 - PSM-1 - HSP-2 - ACU-Controller	3.04 May 30 2007 2.0.1 (Build date: Dec 12 2007) V3.01 January-25-2007 (Hybrid DSP-1/NXU) N/A N/A 5.42 (Build 68R) OS for the ACU-2000 workstation is Windows XP Professional (Service Pack 2)
Vertex Radios - UHF - VHF High	N/A N/A

4. Configure Avaya Communication Manager

This section describes the procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES. The steps include setting up an IP codec set, an IP network region, an IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to set up an additional trunk. The highlights in the following screens indicate the values used. Default values may be used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. SIP telephones including the Raytheon JPS SCM-2 modules are configured as off-PBX telephones in Avaya Communication Manager.

4.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses. Each Raytheon JPS SCM-2 module will use one Off-PBX Telephone license.

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V15	Software Package : Standard		
Location: 1	RFA System ID (SID): 1		
Platform: 6	RFA Module ID (MID): 1		
			USED
Platform Maximum Ports:			44000 322
Maximum Stations:			36000 123
Maximum XMOBILE Stations:			0 0
Maximum Off-PBX Telephones - EC500:			50 3
Maximum Off-PBX Telephones - OPS:			100 10
Maximum Off-PBX Telephones - PBFMC:			0 0
Maximum Off-PBX Telephones - PVFMC:			0 0
Maximum Off-PBX Telephones - SCCAN:			0 0

On **Page 2**, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses. Each Raytheon JPS SCM-2 module will use one SIP Trunk license.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES			USED
Maximum Administered H.323 Trunks:			100 40
Maximum Concurrently Registered IP Stations:			1500 25
Maximum Administered Remote Office Trunks:			800 0
Maximum Concurrently Registered Remote Office Stations:			1500 0
Maximum Concurrently Registered IP eCons:			0 0
Max Concur Registered Unauthenticated H.323 Stations:			0 0
Maximum Video Capable H.323 Stations:			100 4
Maximum Video Capable IP Softphones:			100 4
Maximum Administered SIP Trunks:			100 20
Maximum Number of DS1 Boards with Echo Cancellation:			0 0
Maximum TN2501 VAL Boards:			10 1
Maximum Media Gateway VAL Sources:			0 0
Maximum TN2602 Boards with 80 VoIP Channels:			128 2
Maximum TN2602 Boards with 320 VoIP Channels:			128 0
Maximum Number of Expanded Meet-me Conference Ports:			0 0

4.2. IP Codec Set

This section describes the steps for administering an IP codec set in Avaya Communication Manager. This IP codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES. Enter the **change ip-codec-set <c>** command, where

c is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** when configuring an IP network region to specify which audio codecs may be used within and between network regions. In the sample configuration, only one network region is used.

The ACU-2000IP supports G.711MU and GSM. For integration with Avaya Communication Manager, enter G.711MU. Retain all other default field values.

```
change ip-codec-set 1                                     Page 1 of 2

                                IP Codec Set

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size(ms)
1: G.711MU      n          2          20
2:
3:

Media Encryption
1: none
2:
3:
```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – This should match the SIP Domain value on Avaya SES, in **Section 5.1**. In the test configuration, **gov.com** was used.
- Codec Set – Enter the IP codec set number as provisioned in **Section 4.2**.

```
change ip-network-region 1                               Page 1 of 19

                                IP NETWORK REGION

Region: 1
Location: 1      Authoritative Domain: gov.com
Name: Main Region - HQ
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
      Codec Set: 1      Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048      IP Audio Hairpinning? y
      UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS      RTCP Reporting Enabled? y
Call Control PHB Value: 48      RTCP MONITOR SERVER PARAMETERS
      Audio PHB Value: 48      Use Default Server Parameters? y
      Video PHB Value: 34
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 4      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

4.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager. Enter the **change node-names ip** command, and add a node name for Avaya SES along with its IP address. The CLAN board (in the case of an Avaya S8300 Server, Processor-Ethernet, procr) will be used as well in subsequent steps in these Application Notes.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
CLAN-01A02	9.1.1.8	
SES1	9.1.1.34	

4.5. Configure SIP Signaling Group

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add signaling-group <s>** command, where **s** is an available signaling group, and configure the following:

- Group Type – Set to **sip**.
- Near-end Node Name - Set to **CLAN-01A02** as displayed in **Section 4.4**.
- Far-end Node Name - Set to the Avaya SES name configured in **Section 4.4**.
- Far-end Network Region - Set to the region configured in **Section 4.3**.
- Far-end Domain - This should match the SIP Domain value in **Section 5.1**. In the test configuration, **gov.com** was used.

add signaling-group 2		Page 1 of 1	
SIGNALING GROUP			
Group Number: 2	Group Type: sip		
Transport Method: tls			
Near-end Node Name: CLAN-01A02	Far-end Node Name: SES1		
Near-end Listen Port: 5061	Far-end Listen Port: 5061		
Far-end Domain: gov.com	Far-end Network Region: 1		
Bypass If IP Threshold Exceeded? n			
DTMF over IP: rtp-payload	Direct IP-IP Audio Connections? n		
	IP Audio Hairpinning? n		
Enable Layer 3 Test? n			
Session Establishment Timer(min): 120			

4.6. Configure SIP Trunk Group

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group, and configure the following:

- Group Type – Set to **sip**.
- Group Name – Enter a descriptive name.
- TAC– Set to any available trunk access code that is valid in the provisioned dial plan.
- Signaling Group – Set to the Group Number field value configured in **Section 4.5**.
- Number of Members – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used, but still within the maximum number allowed (see **Section 4.1**).
- Service Type – Set to **tie**.

```
add trunk-group 2                                     Page 1 of 21
                                                    TRUNK GROUP

Group Number: 1                                     Group Type: sip          CDR Reports: y
Group Name: CM to SES                               COR: 1                 TN: 1                 TAC: 102
Direction: two-way                                Outgoing Display? n
Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                                  Auth Code? n
                                                    Signaling Group: 2
                                                    Number of Members: 20
```

On **Page 5** of the trunk-group form, verify that all trunk group members are assigned, as shown below.

```
display trunk-group 2                                Page 5 of 21
                                                    TRUNK GROUP
Administered Members (min/max): 1/20
GROUP MEMBER ASSIGNMENTS                        Total Administered Members: 20

Port      Name
1: T000024 CM to SES
2: T000025 CM to SES
3: T000026 CM to SES
4: T000027 CM to SES
5: T000028 CM to SES
6: T000029 CM to SES
7: T000030 CM to SES
8: T000031 CM to SES
9: T000032 CM to SES
10: T000033 CM to SES
```

4.7. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers. This OPS station extension will be used by the Raytheon JPS SCM-2 module to interconnect field radios to the enterprise connected to Avaya Communication Manager. This enables customer calls and calls throughout the enterprise to be routed directly to field emergency responders. In addition, radio-equipped responders can place calls to stations throughout the enterprise connected to Avaya Communication Manager as well as contacts outside the enterprise.

Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **4620**.
- Name – Enter a descriptive name

Repeat this step as necessary to configure additional SIP endpoint extensions.

add station 52001		Page 1 of 5
STATION		
Extension: 52001	Lock Messages? n	BCC: 0
Type: 4620	Security Code: 52001	TN: 1
Port: IP	Coverage Path 1:	COR: 1
Name: PSAP Dispatcher 1	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 52001	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Expansion Module? n	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	Customizable Labels? y	

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Enter the extension configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that SCM-2 will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- Config Set – Set to **1**, which contains the default values.
- Trunk Select – Set to the trunk group number configured in **Section 4.6**.

The following screen shows the OPS stations created during testing.

```
list off-pbx-telephone station-mapping
```


STATION TO OFF-PBX TELEPHONE MAPPING							
Station Extension	Appl	CC	Phone Number	Config Set	Trunk Select	Mapping Mode	Calls Allowed
52001	OPS		52001	1 /	2	both	all
52002	OPS		52002	1 /	2	both	all

5. Configure Avaya SES

This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Raytheon JPS SCM-2 modules will register with Avaya SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.


5.1. Configure SES Server Properties

Launch a web browser, enter <https://<IP address of SES server>/admin> in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login.



Integrated Management
Standard Management Solutions


Help Log Off



SES
Administration

The Administration Web Interface allows you to administer this SES server.

[Launch SES Administration Interface](#)



Maintenance

The Maintenance Web Interface allows you to maintain, troubleshoot, and configure the media server.

[Launch Maintenance Web Interface](#)

In the **Integrated Management SIP Server Management** page, select the **Server Configuration → System Properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group in Avaya Communication Manager in **Section 4.5**. Click on the **Update** button if a field change was necessary.

AVAYA

Help Exit

Top

- Setup
- Users
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- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
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- Media Server Extensions
- Server Configuration**
 - Admin Setup
 - IM Log Settings
 - License
 - SNMP Configuration
 - System Properties**
- SIP Phone Settings
- Survivable Call Processors
 - System Status
- Trace Logger
- Trusted Hosts

View System Properties

SES Version SES-5.0.0.0-825.31

System Configuration simplex

Host Type SES combined home-edge

SIP Domain* gov.com

Note that the DNS domain is Gov.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host* localhost

DiffServ/TOS Parameters

Call Control PHB Value* 46

802.1 Parameters

Priority Value* 6

Management System Access Login

Management System Access Password

DB Log Level disabled

Update

5.2. Configure Media Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the **Integrated Management SIP Server Management** page, select the **Media Servers → Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

- Media Server Interface Name – Enter a descriptive name for the media server interface.
- Host – From the drop-down list of IP addresses, select the IP address of the Avaya SES server to be associated with the Media Server interface.
- SIP Trunk Link Type – Select **TLS**.
- SIP Trunk IP Address – Enter the IP address for the media server's CLAN (or procr) IP interface that terminates the SIP link from Avaya SES (see **Section 4.4**).

Click **Add** when finished.

AVAYA

Help Exit

Top

- Setup
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- + Media Server Extensions
- + Server Configuration
- + SIP Phone Settings
- + Survivable Call Processors
- System Status
- + Trace Logger
- + Trusted Hosts

Add Media Server Interface

Media Server Interface Name* CLANA

Host 9.1.1.34

SIP Trunk

SIP Trunk Link Type ☐ TCP ☒ TLS

SIP Trunk IP Address* 9.1.1.8

Media Server

Media Server Admin Address (see Help)

Media Server Admin Port 5022

Media Server Admin Login

Media Server Admin Password

Media Server Admin Password Confirm

SMS Connection Type ☒ SSH ☐ Telnet ☐ Not Available

Note: Changing connection type to SSH connection type to Telnet resets media s

Fields marked * are required.

Add

5.3. Configure Users

This section provides steps to add users to be administered in the Avaya SES database. In the Integrated Management SIP Server Management page, select the **Users** → **Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- Primary Handle – Enter the phone number of SCM-2 modules. This number was configured in **Section 4.7**.
- User ID – Set to any descriptive name.
- Password / Confirm Password – Enter a password of at least 6 alphanumeric characters; both field entries must match exactly. Note the password entered in the screen below. This will be needed in **Section 6.3** for SCM-2 Auth Password field.
- Host – From the drop-down list of IP addresses, select the host serving the domain for this user. The IP address of the current server is selected by default.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Add Media Server Extension - Select this field to associate a new extension number with this user in the database. The Add MS Extension screen will be displayed next, after this user profile has been added.

Click **Add** when finished.

AVAYA

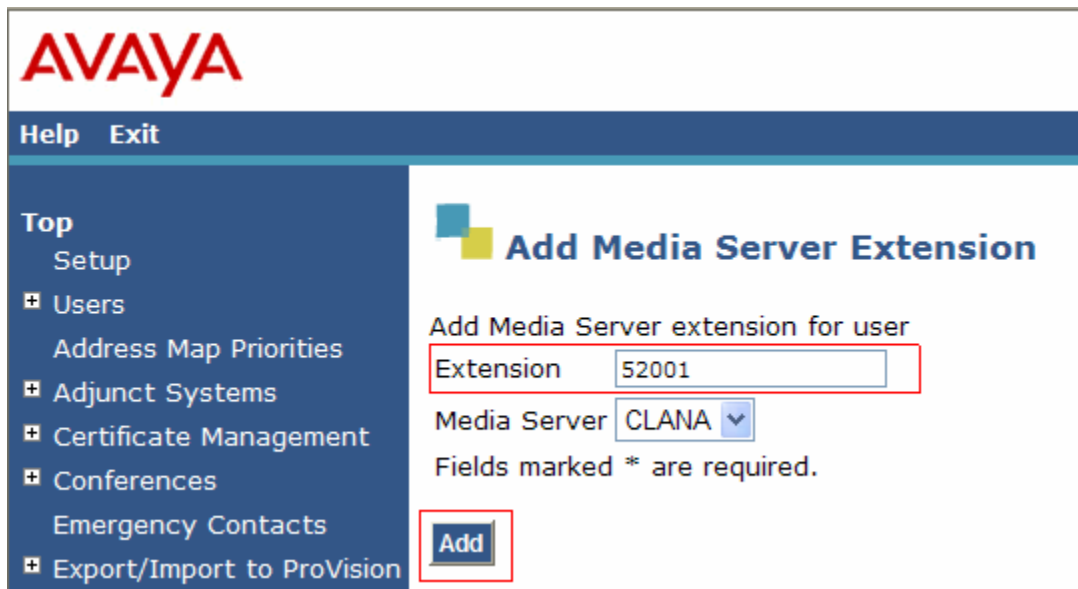
Help Exit

Top
Setup
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Hosts
IM logs
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List
Media Server Extensions
Server Configuration
Admin Setup
IM Log Settings
License
SNMP Configuration
System Properties
SIP Phone Settings
Survivable Call Processors
System Status
Trace Logger
Trusted Hosts

Add User

Primary Handle* 52001
User ID 52001
Password*
Confirm Password*
Host* 9.1.1.34
First Name* PSAP
Last Name* Dispatcher 1
Address 1
Address 2
Office
City
State
Country
Zip
Survivable Call Processor none
Add Media Server Extension ☒
Fields marked * are required.
Add

At the next screen, enter the numeric telephone extension to be created in the database. This should match the Phone Number entry on the off-pbx-telephone station-mapping form in **Section 4.7**. Select the extension's media server from the drop-down list. Click on the **Add** button.



The screenshot shows the Avaya web interface. At the top left is the Avaya logo. Below it is a navigation menu with 'Help' and 'Exit' at the top, followed by a list of options: 'Top', 'Setup', 'Users', 'Address Map Priorities', 'Adjunct Systems', 'Certificate Management', 'Conferences', 'Emergency Contacts', and 'Export/Import to ProVision'. The main content area is titled 'Add Media Server Extension'. It contains the text 'Add Media Server extension for user'. Below this are two input fields: 'Extension' with the value '52001' and 'Media Server' with a dropdown menu showing 'CLANA'. A note below the fields states 'Fields marked * are required.' At the bottom of the form is an 'Add' button. Red boxes highlight the 'Extension' field, the 'Media Server' dropdown, and the 'Add' button.

6. Configure Raytheon JPS ACU-2000IP

This section describes the steps for configuring the Raytheon JPS ACU-2000IP gateway. The Control Processor Module (CPM), DSP Module and SIP Control Module-2 (SCM-2) need to be configured.

6.1. Configure CPM Module

The default address of the CPM as shipped from the factory is 192.168.1.200. Follow the instructions in [3] to change the IP address to comply with the customer's network requirements.

Launch a web browser, enter <http://<IP address of CPM>> in the URL. The highlights in the following screens indicate the values used. Default values may be used for all other fields.

VOIP Management	Configuration	Connection Status	Module Names
CPM-6 Configuration			
My IP Address: 9 1 1 123 IP Gateway: 9 1 1 1		Subnet Mask: 255 255 255 0 Remote IP: 0 0 0 0	
Baud rate: 115200 Stop bits: 1 Chassis Configuration: Single Remote Control: Enabled Serial Sync Char: Disabled Chassis Extended Addr: Disabled		Serial port data bits: 8 Parity: None Connection Configuration: Auto Restore CPM-6 Name: JPS CPM-6 Chassis Reset Data: 00	
VOIP Configuration			
VOIP Port: 1221 Command Port: 23 Voice Compression: 1. GSM 13Kbps COR Inhibit Time (ms): 0 Duplex: Full Client Autoconnect: Yes		Unit is a: Server Remote Port: 1221 VOX Hangtime (ms): 500 COR Sense: Low active (H/W) Communications mode: Normal	
<input type="button" value="Save Changes"/> <input type="button" value="Clear Changes"/>			
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6.2. Configure DSP Module

The DSP Module interfaces radios and other 4-wire devices to the ACU-2000IP. The DSP has three interfaces:

- Front panel RJ-45 connector
- Rear panel four-wire interface (15-pin D-sub connector to radios)
- Chassis backplane for CPM control, for connections with other modules in the gateway.

The DSP module can use any two of its three interfaces, and the interface selection determines the operating mode. The operating modes for a DSP module are:

- **Standard Mode:** In this mode, the DSP module cross-connects audio and control signals from radios to other modules in the local ACU-2000IP gateway via the chassis backplane. This mode utilizes the rear panel four-wire connector and chassis backplane

interface. The front panel Ethernet port is used to provide network connectivity to the module, but there is no VoIP capability.

- **VoIP Standalone Mode:** This mode allows the DSP module to act as an independent, standalone, network-to-radio interface. This mode utilizes the front panel RJ45 and rear panel four-wire connector. This mode allows VoIP connections to other network-capable devices that use the JPS RoIP protocol. These devices include other DSP modules, CPM modules and Network Extension Units (NXU-2A), which is a separate product from Raytheon JPS.
- **VoIP Hybrid Mode:** This mode allows the DSP to function as an RoIP interface to the ACU-2000IP backplane, allowing remote cross-connection to take place over an IP network. In this mode, the DSP is visible to the CPM module. There is no connection to the associated rear panel D15 four-wire connector.

For integration with Avaya Communication Manager and Avaya SIP Enablement Services, the DSP module needs to be configured in the Standard Mode. This is the default configuration from the factory for the DSP module and no additional provisioning is required. Two DSP modules – one for UHF and one for VHF-High radio frequencies will be required for the Public Safety reference configuration as shown in **Figure 1** and **Figure 2**.

6.3. Configure SCM-2 Module

SIP Control Module-2 (SCM-2) creates an interface between a radio and an IP network, using the SIP protocol. The SIP user accounts configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension are used by the SCM-2 modules to register with Avaya SES. The ACU-2000IP gateway can have several SCM-2 modules. The steps detailed below need to be performed for each SCM-2 module that will register with Avaya SES.

The default address of the SCM-2 as shipped from the factory is 192.168.1.200. Follow the instructions in [3] to change the IP address to comply with the customer's network requirements.

1. Launch a web browser, enter <http://<IP address of SCM-2>> in the URL.

The following screen shows the SCM's main page, which displays the Status/Information. The highlights in the following screens indicate the values used. Default values may be used for all other fields.

Raytheon
JPS Communications SIP Control Module: SCM-2

[Status/Information](#) [Network Settings](#) [SIP Settings](#) [Network Squelch Settings](#)

MAC Address: 00:0e:1a:00:6a:5e

IP Address: 9.1.1.201

Subnet Mask: 255.255.255.0

Default Gateway: 9.1.1.1

Primary DNS Server:

Secondary DNS Server:

Tertiary DNS Server:

Firewall Type:

Mapped Address:

Mapped Port:

Product Model: SIP Control Module (SCM-2)

Firmware Version: 2.0.1 (Build date: Dec 12 2007 10:34:04)

System Uptime: 1 days, 4:08:25

Registration Status: Registered

Next Registration In: 3127 s

Inbound Calls: 1

Outbound Calls: 3

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2. From the SCM's main page, click **SIP Settings** to configure the interface settings to communicate with Avaya SES.

Raytheon
JPS Communications SIP Control Module: SCM-2

[Status/Information](#) [Network Settings](#) [SIP Settings](#) [Network Squelch Settings](#) [Call Management](#) [Administration](#)

3. From the SIP Settings page, configure the following fields:
- User Name – Enter one of the extension numbers configured in **Section 5.3**.
 - Auth Password – Enter the corresponding extension password configured in **Section 5.3**.
 - Domain – Enter the SIP domain configured in **Section 5.1**.
 - Proxy – Enter the Avaya SES server IP address as specified in **Section 4.4**.
 - Preferred Codec – Select G711u from the drop-down list.

Click on **Save** after the completion of the form.

Raytheon

JPS Communications

SIP Control Module: SCM-2

[Status/Information](#)

[Network Settings](#)

[SIP Settings](#)

[Network Squelch Settings](#)

Display Name:	<input type="text" value="Dispatcher #1"/>
Username:	<input type="text" value="52001"/>
Auth ID:	<input type="text"/>
Auth Password:	<input type="password" value="*****"/>
Domain:	<input type="text" value="gov.com"/>
Proxy:	<input type="text" value="9.1.1.34"/>
Proxy Port:	<input type="text" value="5060"/>
Outbound Proxy:	<input type="text"/>
Outbound Proxy Port:	<input type="text" value="5060"/>
Local Port:	<input type="text" value="5060"/>
Registration Expiration:	<input type="text" value="3600"/> (0-86400 s)
DTMF Mode:	<input type="text" value="In-band Only"/>
Block DTMF In-Band:	<input type="text" value="Disable"/>
Preferred Codec:	<input type="text" value="G711u"/>
Silence Suppression:	<input type="text" value="Disable"/>
Answer Incoming Calls:	<input type="text" value="Enable"/>
Answer Incoming Delay:	<input type="text" value="1000"/> (0-30000 ms)
NAT Traversal:	<input type="text" value="Disable"/>
STUN Server:	<input type="text"/>
STUN Port:	<input type="text" value="3478"/>
Send Radio COR/AUX Status:	<input type="text" value="Disable"/>
<input type="button" value="Save"/>	

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4. Repeat Steps 1-3 for every SCM-2 module that will register with Avaya SES.

6.4. Configure ACU-2000 Controller

This section describes the steps for configuring the Raytheon JPS ACU-Controller. Follow the instructions in [4] to install the ACU Controller on the workstation.

On the ACU Controller workstation, select **Start → Programs → ACUController → ACUController**.

At the top of the ACU Controller Main Screen, select **Sites → Select Network Sites...**

The **Select Network Site** dialog box will appear. In the **Add New Network Site**, enter the IP address of the CPM Module as configured in **Section 6.1**.

Click **Add and Connect** button. This will add the site to the list and then close the dialog box and connect to the new site immediately.

Select Network Site

Add New Network Site

IP Address: 9 1 1 123

Port (23 is default): 23

Description: ACU-2000IP

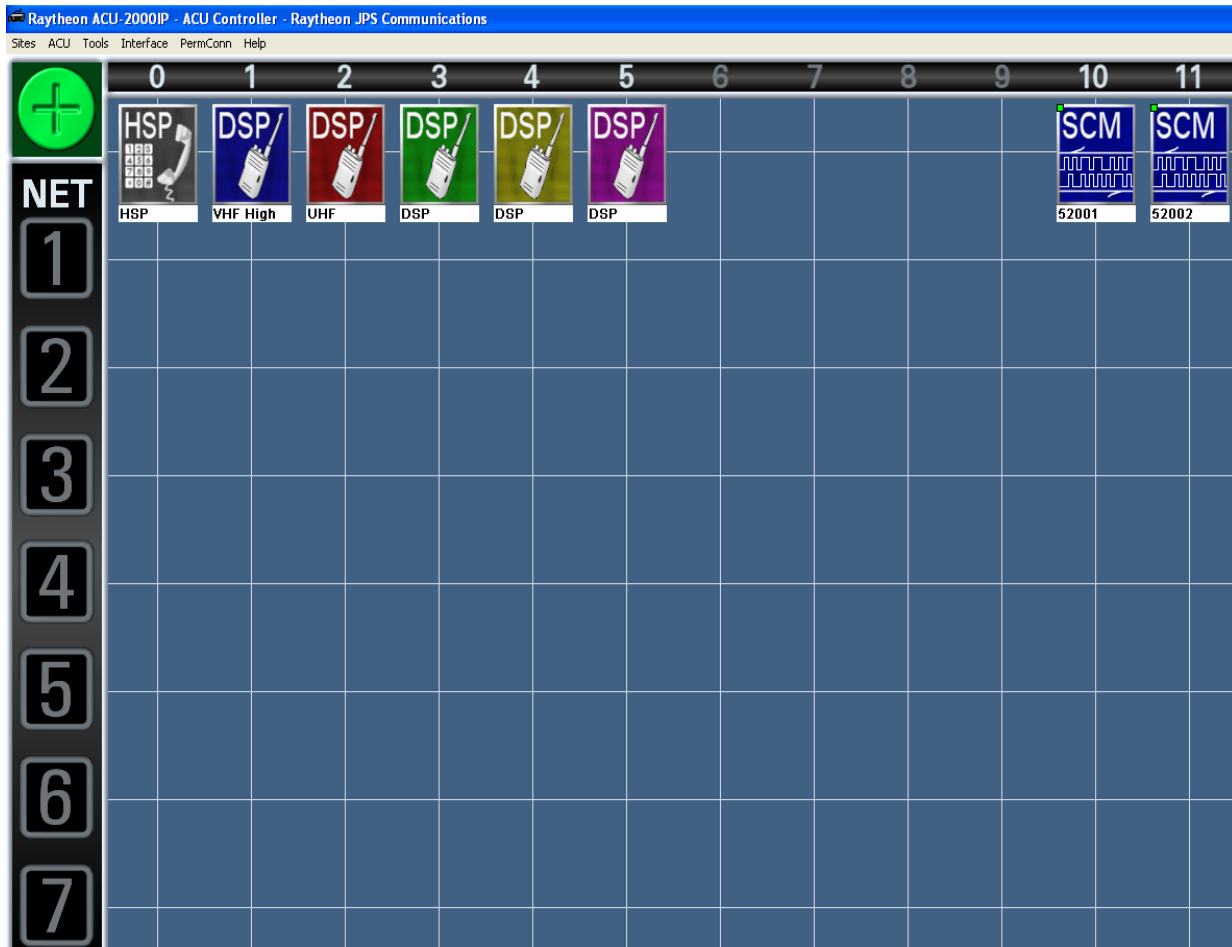
Add Add and Connect

Stored Network Sites

Connect Remove

Disconnect Close

The ACU-Controller will establish a connection with the ACU-2000IP gateway and the following Main Window will appear.



7. Interoperability Compliance Testing

The interoperability compliance testing included basic feature and serviceability testing.

The test scenarios for feature testing focused on the integration of Raytheon JPS ACU-2000IP with Avaya Communication Manager via Avaya SES. The specific tests include the following areas:

- Registration of SCM-2 to Avaya SES as SIP endpoints
- Feature tests
 - Basic calls with Avaya 4620/9630 IP telephones, with and without media shuffling
 - Hold and unhold from Avaya 4620/9630 IP telephones
 - Drop from the radio devices
 - Conference involving Avaya 4620/9630 IP telephones and SCM-2 modules

- Transfers from Avaya 4620/9630 IP telephones to SCM-2 modules or radios.

The serviceability testing focused on verifying the ability of Raytheon JPS ACU-2000IP to recover from adverse conditions, such as:

- Server interchanges / Reset
- Disconnect/reconnect of Ethernet cable to Avaya SES and Raytheon JPS ACU-2000IP
- Workstation Scenarios – Disconnect/reconnect Ethernet cables

7.1. General Test Approach

All test cases were performed manually. The general approach was to register the SCM to Avaya SES, place outbound calls, and receive inbound calls. Serviceability failures were simulated by disconnecting cables, and by executing reset system commands from the Avaya Communication Manager System Access Terminal interface.

7.2. Test Results

All test cases were executed and passed.

Few observations were made during testing which are noted below:

1. Disconnecting an active call from the ACU-2000IP gateway either from the radios or the local operator by pressing *# on the HSP keypad, does not release the SIP trunk immediately. SIP trunk is released after 3-4 minutes. The SCM-2 module does not send a “bye” message immediately.
2. It is recommended to restart the ACU-Controller application (Start → Programs → ACUController → ACUController) after the LAN connection is lost to the workstation. At times, ACU-Controller does not reflect the correct connect states after recovering from a LAN failure.

8. Verification and Troubleshooting

This section provides the tests that can be performed to verify proper configuration of Avaya Communication Manager, Avaya SES, and Raytheon JPS ACU-2000IP Intelligent Interconnect System:

1. In the Avaya SES **Integrated Management SIP Server Management** page, select the **Users → Registered Users** link from the left pane of the screen. Verify all SIP endpoints are registered.

Top

Setup

Users

Address Map Priorities

Adjunct Systems

Certificate Management

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License

SNMP Configuration

System Properties



Registered Users on 9.1.1.34

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Showing 1 to 2 of 2 registered contacts.

	Handle and Name	Address	Expires
<input type="checkbox"/>	52002@gov.com Dispatcher 2, PSAP	sip:52002@9.1.1.202:5060	Fri, 20 Jun 2008 15:28:51 EDT
<input type="checkbox"/>	52001@gov.com Dispatcher 1, PSAP	sip:52001@9.1.1.201:5060	Fri, 20 Jun 2008 15:28:52 EDT

☐ Apply to all registered users with compatible devices on this Home.

☐ Apply to all registered users with compatible devices on this page.

Task:

- Using a network protocol analyzer, verify correct REGISTER messages are exchanged between Avaya SES and ACU-2000IP.
- SCM Status Indicators – A SCM module's SIP connection status is shown on the ACU-Controller. These are small overlays on the module icons. The first icon indicates the mode – Registered is green, Unregistered is gray and Failed to Register is red. The second icon indicates the SIP connection status – Connected is green, Failed to Connect is red and no icon indicates no connection.

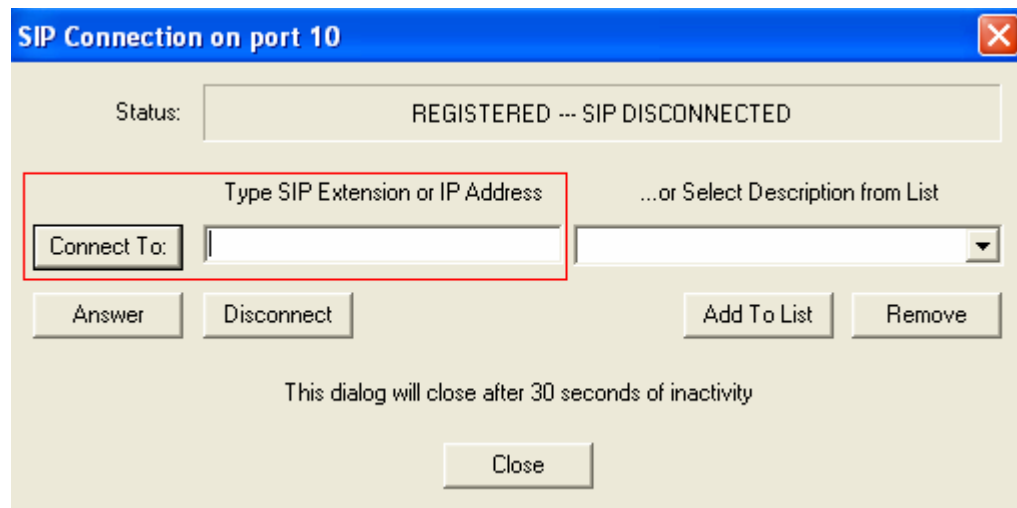
Registered and Disconnected



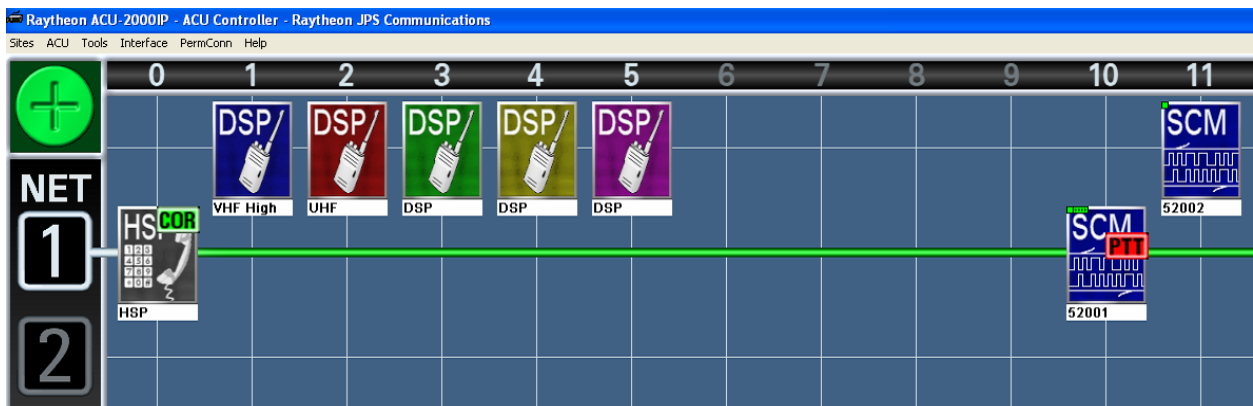
Registered and Connected



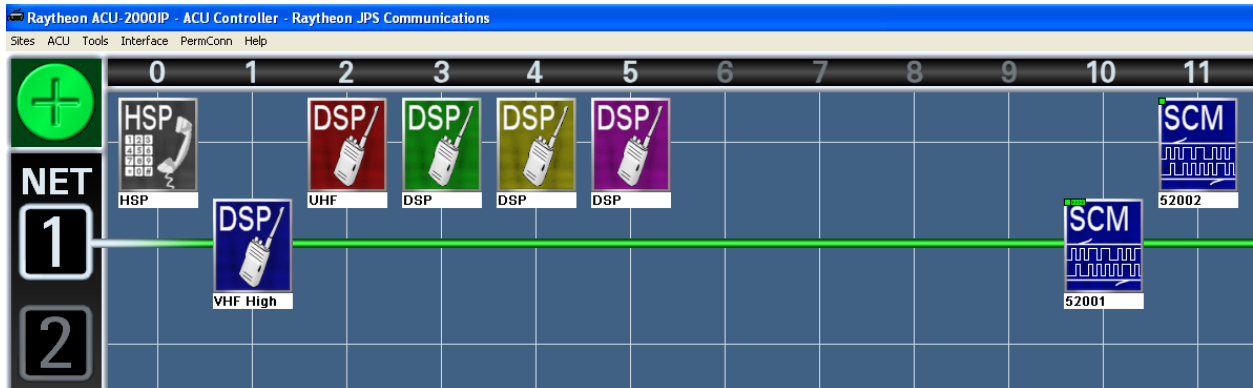
- Establish a call from the ACU-2000IP local operator and a user on the Avaya Communication Manager using the ACU-Controller. A connection is made by clicking the left mouse button on the HSP and SCM module icons. When establishing a connection with the SCM module, the **SIP Connection on port 10** dialog box will appear. Enter the phone number in the **Connect To:** field and press **Enter**.



Once the call is established, the ACU-Controller Main Screen will show the following:



- Establish a call from the radios using the ACU-Controller. A connection is made by clicking the left mouse button on the DSP and SCM module icons. When establishing a connection with the SCM module, the **SIP Connection on port 10** dialog box will appear. Enter the phone number in the **Connect To:** field and press **Enter**. Once the call is established, the ACU-Controller Main Screen will show the following:



6. COR Reporting and PTT Reporting - These are displayed on the ACU-Controller Main Screen as overlays indicating audio activities in the modules. For example, if the HSP operator is talking, the ACU-Controller Main Screen will show COR LED reporting for HSP and PTT LED reporting for the listening devices, VHF High radios, in this case.



9. Support

Technical support on the ACU-2000IP or SCM can be obtained through the following:

- **Phone:** (919) 790-1011 or (800) 498-3137
- **Web:** <http://www.jps.com/support>

10. Conclusion

These Application Notes describe the configuration steps required for Raytheon JPS ACU-2000IP to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services. All feature and serviceability test cases were completed.

11. Terminology

AES	Avaya Application Enablement Services
COR	Carrier Operated Relay. A module showing COR is receiving audio from another module.
CPM	Control Processor Module
DMCC	Device, Media and Call Control
HSP	Handset/Speaker/Prompt Module
NENA	National Emergency Number Association
PSAP	Public Safety Answering Point
PTT	Push-To-Talk. A module with an active PTT signal is receiving audio from another module.
RoIP	Radio over IP
SCM-2	SIP Control Module
SES	SIP Enablement Services
TSAPI	Telephony Services Application Programming Interface
UHF	Ultra High Frequency. Radio frequency range is from 300 MHz to 3 GHz
VHF	Very High Frequency. Radio frequency range is from 30 MHz to 300 MHz

12. Additional References

Avaya documentation can be located at <http://support.avaya.com>

[1] *Administrators Guide for Avaya Communication Manager*, Document 03-300509, Issue 4.0, Release 5.0, Jan 2008.

http://support.avaya.com/elmodocs2/comm_mgr/r5.0/03-300509_4.pdf

[2] *Installing, Adminstrating, Maintaining, and Troubleshooting SIP Enablement Services*, Document 03-600768, Jan 2008.

http://support.avaya.com/elmodocs2/sip/03_600768_5.pdf

JPS Communication documentation can be located at <http://www.jps.com/>

[3] *Installation and Operation Manual ACU-2000IP Intelligent Interconnect System*, JPS Communications, Inc, JPS P/N 5961-230200 Revision 1.0, and February 2007.

[4] *ACU-Controller Control Software for the ACU-2000 IP, ACU-1000, ACU-T and ACU-M Software Installation and Operation Manual*, JPS Communications, Inc, JPS P/N 5961-298200 Revision 5.4, April 2007.

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