



**Application Notes for Raytheon JPS ARA-1 with Avaya
Communication Manager and Avaya SIP Enablement
Services – Issue 1.0**

Abstract

These Application Notes describe the configuration steps required for Raytheon JPS ARA-1 to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services.

The Raytheon JPS ARA-1 is a network device for interfacing radio equipment to SIP networks, thereby extending the coverage and capability of these networks. It is comparable to an analog telephone adapter, which allows a standard telephone to operate on a SIP network; the JPS ARA-1 provides the same capability to a radio.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Raytheon JPS ARA-1 which was compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services. The overall objective of the interoperability compliance testing was to verify Raytheon JPS ARA-1 features in an environment comprised of Avaya Communication Manager, Avaya SIP Enablement Services, various Avaya IP Telephones, and various Avaya SIP endpoints.

The JPS ARA-1 provides a seamless interface between a radio and an IP-based network using SIP. This brings to existing SIP networks all of the features inherent in a radio system, including the ability to reach otherwise inaccessible areas wirelessly. For example, a JPS ARA-1 can be used with a Land Mobile Radio (LMR) to extend the SIP Network into areas of rugged terrain, across bodies of water, or into tunnels.

Figure 1 provides the test configuration used for the compliance testing.

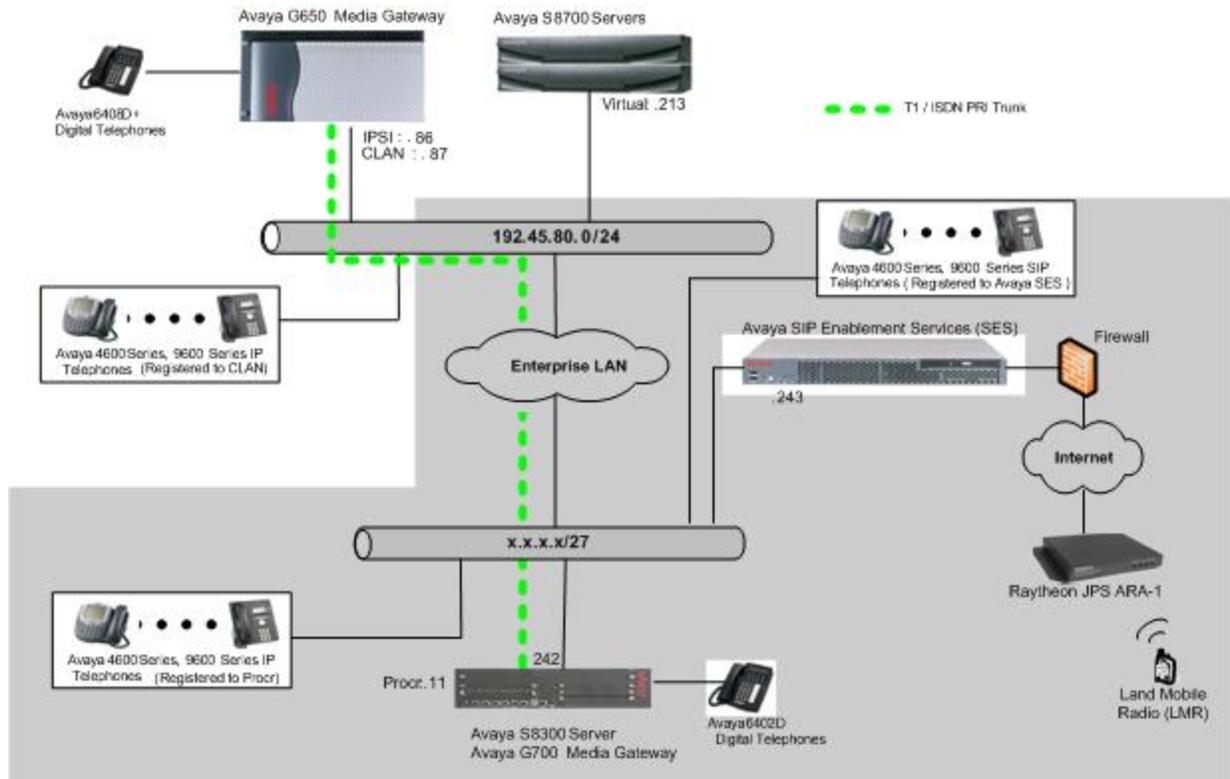


Figure 1: Test Configuration for Raytheon JPS ARA-1 with Avaya Communication Manager and Avaya SES

2. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya S8700 Servers		Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G650 Media Gateway		
	TN2312BP IP Server Interface	HW11 FW030
	TN799DP CLAN Interface	HW01 FW024
	TN2302AP IP Media Processor	HW20 FW117
Avaya S8300 Server with Avaya G700 Media Gateway		Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya S8500 Server		Avaya SIP Enablement Services 4.0 (Build 33.6)
Avaya 4600 Series IP Telephones		
	4620SW(H.323)	2.8
	4625SW(H.323)	2.8
	4610SW (SIP)	2.2.2
Avaya 9600 Series IP Telephones		
	9630 (H.323)	1.5
	9650 (H.323)	1.5
	9630 (SIP)	1.0.13.1
Avaya 6400D Series Digital Telephones		-
Raytheon JPS ARA-1		2.0.1

3. 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 a list of 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 during the compliance testing. 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 are configured as off-PBX telephones in Avaya Communication Manager.

3.1. 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

```
display system-parameters customer-options                               Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V14
Location: 1                      RFA System ID (SID): 1
Platform: 7                      RFA Module ID (MID): 1

                                USED
Platform Maximum Ports: 900     95
Maximum Stations: 450           17
Maximum XMOBILE Stations: 0     0
Maximum Off-PBX Telephones - EC500: 50 0
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.

```
display system-parameters customer-options                               Page 2 of 11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
Maximum Administered H.323 Trunks: 100 18
Maximum Concurrently Registered IP Stations: 50 3
Maximum Administered Remote Office Trunks: 0 0
Maximum Concurrently Registered Remote Office Stations: 0 0
Maximum Concurrently Registered IP eCons: 0 0
Max Concur Registered Unauthenticated H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 5 0
Maximum Video Capable IP Softphones: 5 0
Maximum Administered SIP Trunks: 100 50

Maximum Number of DS1 Boards with Echo Cancellation: 0 0
Maximum TN2501 VAL Boards: 0 0
Maximum Media Gateway VAL Sources: 0 0
Maximum TN2602 Boards with 80 VoIP Channels: 0 0
Maximum TN2602 Boards with 320 VoIP Channels: 0 0
Maximum Number of Expanded Meet-me Conference Ports: 0 0
```

3.2. 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 3.3** when configuring

an IP network region to specify which audio codecs may be used within and between network regions. The JPS ARA-1 only supports 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:
```

3.3. 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 4.1**. In the test configuration, **testroom.com** was used.
- Codec Set – Enter the IP codec set number as provisioned in **Section 3.2**.

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

                                IP NETWORK REGION

Region: 1
Location: Authoritative Domain: testroom.com
Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
Codec Set: 1                Inter-region IP-IP Direct Audio: no
UDP Port Min: 2048              IP Audio Hairpinning? n
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS        RTCP Reporting Enabled? y
Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46            Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5      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
```

3.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Avaya SES in Avaya Communication Manager running on an Avaya S8300 Server. Enter the **change node-names ip**

command, and add a node name for Avaya SES along with its IP address. The Processor-Ethernet (procr) board (or, in the case of an Avaya S8700-series Server, a CLAN board) 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
procr          12.176.170.242
SES           12.176.170.243
```

3.5. 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 **procr** as displayed in **Section 3.4**.
- Far-end Node Name - Set to the Avaya SES name configured in **Section 3.4**.
- Far-end Network Region - Set to the region configured in **Section 3.3**.
- Far-end Domain - This should match the SIP Domain value in **Section 4.1**. In the test configuration, **testroom.com** was used.

```
add signaling-group 1                                     Page 1 of 1
                SIGNALING GROUP
Group Number: 1          Group Type: sip
                        Transport Method: tls

Near-end Node Name: procr          Far-end Node Name: SES
Near-end Listen Port: 5061        Far-end Listen Port: 5061
Far-end Domain: testroom.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): 3
```

3.6. 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 3.5**.
- 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 3.1**).
- Service Type – Set to **tie**.

```
add trunk-group 1                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
Group Name: to SES                                COR: 1                 TN: 1             TAC: 115
Direction: two-way                               Outgoing Display? n
Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                                 Auth Code? n
                                                    Signaling Group: 1
                                                    Number of Members: 10
```

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

```
add trunk-group 1                                     Page 5 of 21
                                                    TRUNK GROUP
Administered Members (min/max): 1/10
GROUP MEMBER ASSIGNMENTS                          Total Administered Members: 10
Port      Name
1: T00001 to SES
2: T00002 to SES
3: T00003 to SES
4: T00004 to SES
5: T00005 to SES
6: T00006 to SES
7: T00007 to SES
8: T00008 to SES
9: T00009 to SES
10: T00010 to SES
```

3.7. 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 of Land Mobile

Radios (LMR). 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 20005                                     Page 1 of 5
                                     STATION
Extension: 20005                                     Lock Messages? n          BCC: 0
Type: 4620                                           Security Code:           TN: 1
Port: IP                                             Coverage Path 1:        COR: 1
Name: SIP 20005                                     Coverage Path 2:        COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                     Time of Day Lock Table:
Loss Group: 19                                     Personalized Ringing Pattern: 1
                                     Message Lamp Ext: 20005
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 – Set the extension of the OPS station as configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that the LMR 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 3.6**.

The following screen shows the OPS stations created for the compliance test.

```

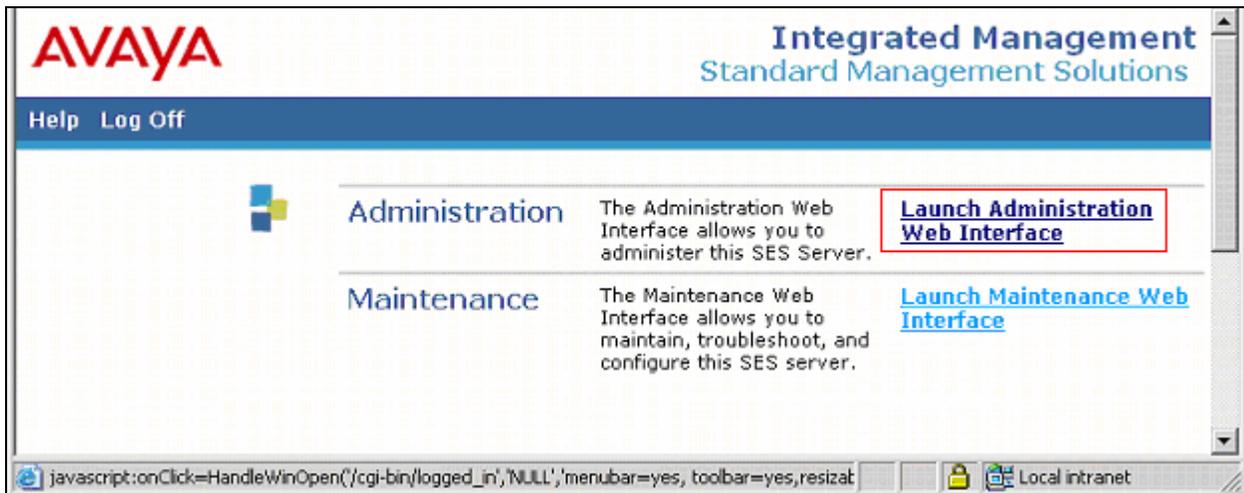
list off-pbx-telephone station-mapping
                                     STATION TO OFF-PBX TELEPHONE MAPPING
Station      Appl  CC  Phone Number  Config  Trunk  Mapping  Calls
Extension    Set    Set  Number       Set     Select Mode     Allowed
20001        OPS   20001 20001         1 / 1   1       both     all
20002        OPS   20002 20002         1 / 1   1       both     all
20003        OPS   20003 20003         1 / 1   1       both     all
20004        OPS   20004 20004         1 / 1   1       both     all
20005        OPS   20005 20005         1 / 1   1       both     all
20006        OPS   20006 20006         1 / 1   1       both     all
20007        OPS   20007 20007         1 / 1   1       both     all
20008        OPS   20008 20008         1 / 1   1       both     all
20009        OPS   20009 20009         1 / 1   1       both     all
20010        OPS   20010 20010         1 / 1   1       both     all
  
```

4. 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. The LMR 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.

4.1. 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 Administration Web Interface** link upon successful login.



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 3.5**. Click on the **Update** button if a field change was necessary.

AVAYA Integrated Management SIP Server Management
Server: 192.11.13.6

Help Exit

Top
Setup
Users
Conferences
Media Server Extensions
Emergency Contacts
Hosts
Media Servers
Address Map Priorities
Adjunct Systems
Trusted Hosts
Services
Server Configuration
System Properties
Admin Accounts
License
IM Log Settings
SNMP Configuration
Certificate Management
IM logs
Trace Logger
Export/Import to ProVision

View System Properties

SES_Version	SES-4.0.0.0-033.6
System Configuration	simplex
Host Type	home/edge

SIP Domain*

Note that the DNS domain is: testroom.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

License Host*	<input type="text" value="localhost"/>
Management System Access Login	<input type="text"/>
Management System Access Password	<input type="text"/>

DiffServ/TOS Parameters

Call Control PHB Value*	<input type="text" value="46"/>
-------------------------	---------------------------------

802.1 Parameters

Priority Value*	<input type="text" value="6"/>
-----------------	--------------------------------

Network Properties

Local IP	12.176.170.243
Local Name	SIPServer.testroom.com
Logical IP	12.176.170.243
Logical Name	SIPServer.testroom.com
Gateway IP Address	12.176.170.225

Redundant Properties

Management Device	SAMP
-------------------	------

Fields marked * are required.

4.2. 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 procr (or CLAN) IP interface that terminates the SIP link from Avaya SES (see **Section 3.4**).

Click **Add** when finished.

The screenshot displays the 'Add Media Server Interface' configuration page in the Avaya Integrated Management SIP Server Management interface. The page includes a sidebar with navigation options and a main content area with the following fields:

- Media Server Interface Name***: Text input field containing 'ACMS8300'.
- Host**: Drop-down menu showing '12.176.170.243'.
- SIP Trunk Link Type**: Radio buttons for 'TCP' and 'TLS', with 'TLS' selected.
- SIP Trunk IP Address***: Text input field containing '12.176.170.242'.
- Media Server Admin Address (see Help)**: Text input field containing '12.176.170.242'.
- Media Server Admin Login**: Text input field containing 'sipusers'.
- Media Server Admin Password**: Password input field with masked characters.
- Media Server Admin Password Confirm**: Password input field with masked characters.
- SMS Connection Type**: Radio buttons for 'SSH' and 'Telnet', with 'SSH' selected.

Fields marked * are required. An **Add** button is located at the bottom of the form.

4.3. 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 the LMR. This number was configured in **Section 3.7**.
- 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.
- 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.

The screenshot shows the 'Add User' form in the Avaya Integrated Management SIP Server Management interface. The form is titled 'Add User' and contains the following fields and options:

- Primary Handle*: 20005
- User ID: 20005
- Password*: [Redacted]
- Confirm Password*: [Redacted]
- Host*: 12.176.170.243
- First Name*: sip
- Last Name*: 20005
- Address 1: [Empty]
- Address 2: [Empty]
- Office: [Empty]
- City: [Empty]
- State: [Empty]
- Country: [Empty]
- Zip: [Empty]
- Add Media Server Extension:

Fields marked * are required.

The 'Add' button is highlighted with a red box.

At the next screen, enter the numeric telephone extension to be created in the database. Select the extension's media server from the drop-down list.

Click on the **Add** button.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes 'Help', 'Exit', and 'Update'. The server IP is '192.11.13.6'. The main content area is titled 'Add Media Server Extension' and contains the following text: 'Add Media Server extension for user 20005.' Below this, there is a form with 'Extension' set to '20005' and 'Media Server' set to 'ACMS8300'. A red box highlights the 'Add' button. A sidebar on the left lists navigation options: Top, Setup, Users (List, Add, Search, Edit, Delete, Password, Default Profile, Registered Users).

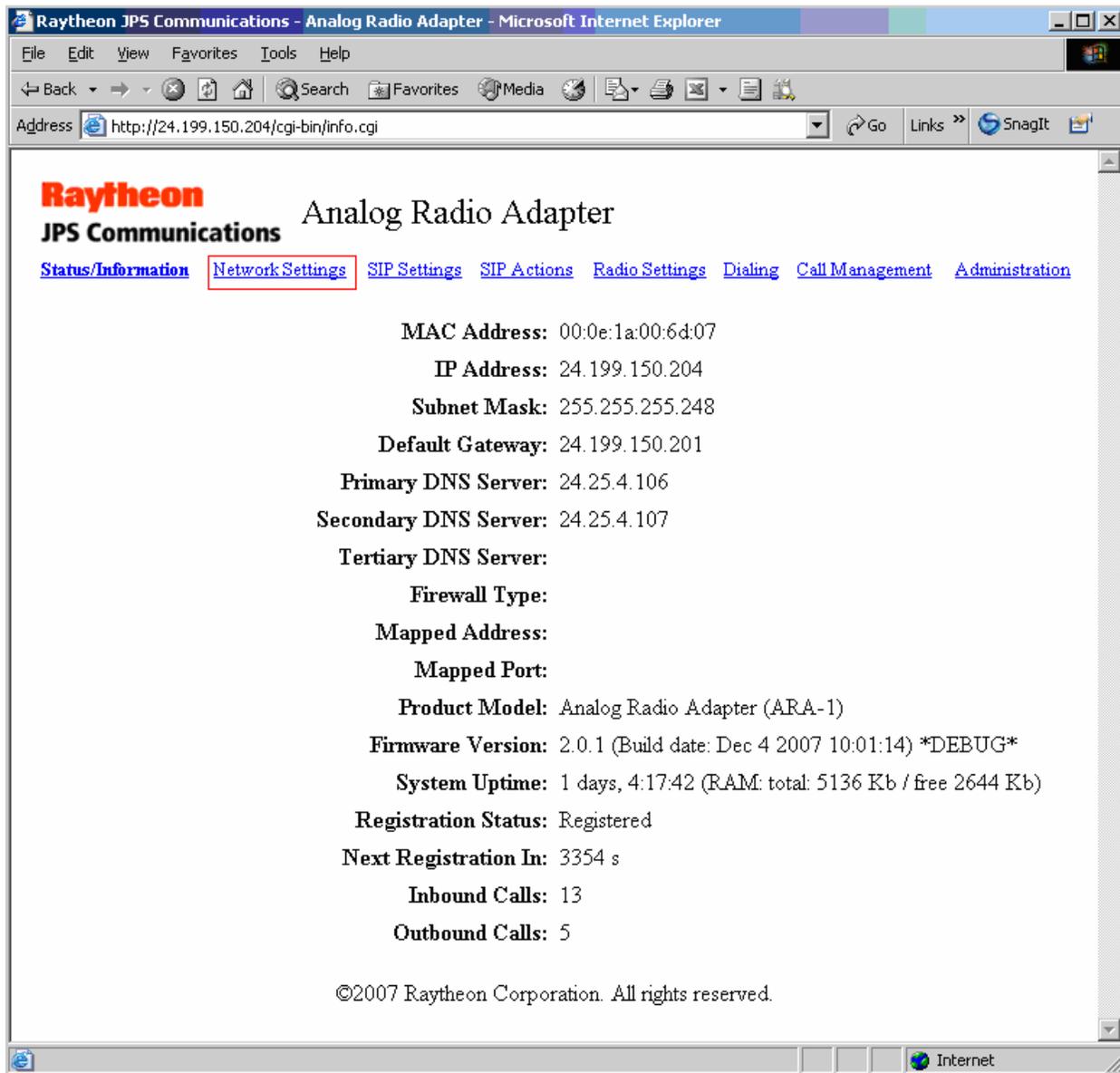
5. Configure Raytheon JPS ARA-1

This section describes the steps for configuring Raytheon JPS ARA-1. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. The LMR 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.

Launch a web browser, enter <http://<IP address of the JPS ARA-1>> in the URL, and log in with the appropriate credentials to access the Analog Radio Adapter page.

The screenshot shows a 'Enter Network Password' dialog box. It contains the following text: 'Please type your user name and password.' Below this, there are labels for 'Site:' (24.199.150.204) and 'Realm:' (24.199.150.203:80). There are input fields for 'User Name' and 'Password'. A checkbox labeled 'Save this password in your password list' is present. At the bottom, there are 'OK' and 'Cancel' buttons.

The following screen shows the Analog Radio Adapter's main page, which displays the Status/Information. To set or change the IP address of JPS ARA-1, click **Network Settings**.



The Network Settings page is utilized to set the IP address, Subnet Mask, and Default Gateway of the JPS ARA-1. The highlighted fields were configured for the compliance test.

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

Raytheon
JPS Communications Analog Radio Adapter

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

Boot Protocol:

IP Address:

Subnet Mask:

Default Gateway:

Primary DNS Server:

Secondary DNS Server:

Tertiary DNS Server:

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From the JPS ARA-1's main page, click **SIP Settings** to configure the interface settings to communicate with Avaya SES.

Raytheon
JPS Communications Analog Radio Adapter

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

Settings were saved successfully.

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From the SIP Settings page, configure the following fields:

- User Name – Enter one of the extension numbers configured in **Section 4.3**.
- Auth Password – Enter the corresponding extension password configured in **Section 4.3**.
- Domain – Enter the SIP domain configured in **Section 4.1**.
- Proxy – Enter the Avaya SES server IP address as specified in **Section 3.4**.
- Preferred Codec – Select G711u from the drop-down list.

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

Raytheon
JPS Communications Analog Radio Adapter

[Status/Information](#) [Network Settings](#) [SIP Settings](#) [SIP Actions](#) [Radio Settings](#) [Dialing](#) [Call Management](#) [Administration](#)

Display Name:

Username:

Auth ID:

Auth Password:

Domain:

Proxy:

Proxy Port:

Outbound Proxy:

Outbound Proxy Port:

Local Port:

Registration Expiration: (0-86400 s)

DTMF Mode:

Block DTMF In-Band:

Preferred Codec:

Silence Suppression:

Answer Incoming Calls:

Answer Incoming Delay: (0-30000 ms)

NAT Traversal:

STUN Server:

STUN Port:

Send Radio COR/AUX Status:

6. Interoperability Compliance Testing

The interoperability compliance testing included basic feature and serviceability testing. The feature testing evaluated the ability of Raytheon JPS ARA-1 to register, make outbound calls (to Avaya SIP endpoints and Avaya H.323 IP telephones), and receive inbound calls (from Avaya SIP endpoints and Avaya H.323 IP telephones). The serviceability testing introduced failure conditions to see if Raytheon JPS ARA-1 or SCM can resume its functions after failure recovery.

6.1. General Test Approach

All test cases were performed manually. The general approach was to register the JPS ARA-1 to Avaya SES, place outbound calls, and receive inbound calls. Serviceability failures were simulated by disconnecting cables, and circuit packs as well as resetting the Avaya S8300 Server.

6.2. Test Results

All test cases were executed and passed.

7. Verification Steps

This section provides the steps that can be performed to verify proper configuration of Avaya Communication Manager, Avaya SES, and Raytheon JPS ARA-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.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes 'Help', 'Exit', and 'Update' on the left, and 'Integrated Management SIP Server Management' and 'Server: 192.11.13.6' on the right. A left-hand navigation pane lists various system components, with 'Registered Users' selected. The main content area is titled 'Registered Users on 192.11.13.6' and contains a table of registered contacts. Below the table are two checkboxes for applying actions to all registered users and a 'Task' dropdown menu set to 'Reload-complete' with a 'Submit' button.

Handle and Name	Address
<input type="checkbox"/> 20001@testroom.com 20001, SIP	sip:20001@12.176.170.247:5060
<input type="checkbox"/> 20002@testroom.com 20002, sip	sip:20002@12.176.170.249;transport=tcp
<input type="checkbox"/> 20005@testroom.com 20005, sip	sip:20005@24.199.150.204:5060
<input type="checkbox"/> 20006@testroom.com 20006, sip	sip:20006@24.199.150.203:5060
<input type="checkbox"/> 20007@testroom.com 20007, sip	sip:20007@24.199.150.202

Using a network protocol analyzer, verify correct REGISTER messages are exchanged between Avaya SES and Raytheon JPS ARA-1.

No.	Time	Source	Destination	Protocol	Info
1050	159.667976	24.199.150.204	12.176.170.243	SIP	Request: REGISTER sip:12.176.170.243
1051	159.669432	12.176.170.243	24.199.150.204	SIP	Status: 401 Unauthorized (0 bindings)
1053	159.744090	24.199.150.204	12.176.170.243	SIP	Request: REGISTER sip:12.176.170.243
1054	159.755270	12.176.170.243	24.199.150.204	SIP	Status: 200 OK (0 bindings)

▶ Frame 1050 (430 bytes on wire, 430 bytes captured)
 ▶ Ethernet II, Src: 12.176.170.225 (00:07:eb:6a:ef:e0), Dst: 12.176.170.243 (00:11:25:ac:08:00)
 ▶ Internet Protocol, Src: 24.199.150.204 (24.199.150.204), Dst: 12.176.170.243 (12.176.170.243)
 ▶ User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
 ▶ Session Initiation Protocol

File: "H:\Avaya\JPS-1119\..."; P: 2116 D: 8 M: 0

8. Support

Technical support on the JPS ARA-1 can be obtained through the following:

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

9. Conclusion

These Application Notes describe the configuration steps required for Raytheon JPS ARA-1 to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services. All feature and serviceability test cases were completed.

10. Additional References

This section references the Avaya and Raytheon JPS product documentation that are relevant to these Application Notes.

[1] *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 3.1, February 2007, available at <http://support.avaya.com>.

[2] *Installation and Operational Manual ARA-1 Radio to SIP Interface*, Revision 1.2, September 2007.

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