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 to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services.

The ACU-2000IP offers a full suite of network capabilities including linking of radios over an IP network, control of large interoperability systems via IP, remote channel change over IP, and the ability to interface radios via SIP. The ACU-2000IP is modular, completely scalable, and field-configurable to meet customer needs. During the compliance test effort, only the SIP Control Module (SCM) was tested.

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 ACU-2000IP, which was compliance tested with Avaya Communication Manager and Avaya SIP Enablement Services. The overall objective of the interoperability compliance testing is to verify Raytheon JPS ACU-2000IP features in an environment comprised of Avaya Communication Manager, Avaya SIP Enablement Services, various Avaya IP Telephones, and various Avaya SIP endpoints.

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

During the compliance test effort, only the SIP Control Module (SCM) was tested. The SCM is a SIP proxy that acts as a SIP endpoint.

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

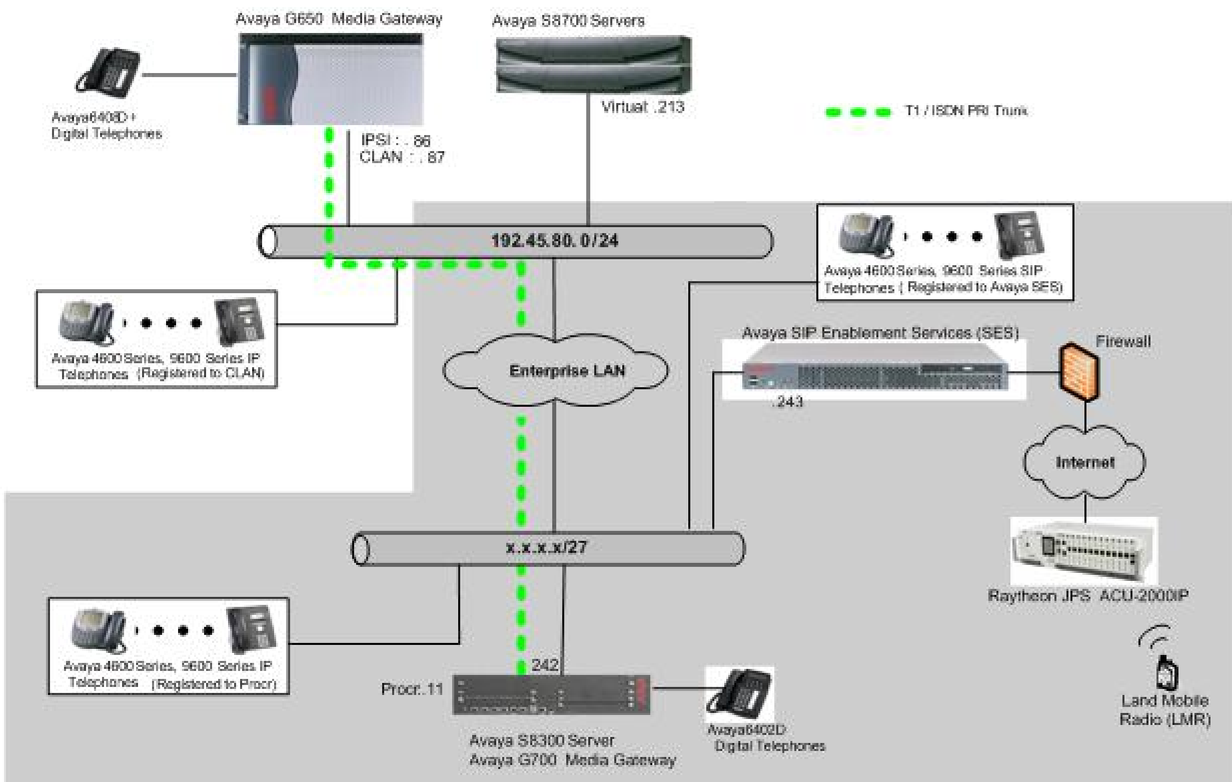


Figure 1: Test Configuration for JPS ACU-2000IP with Avaya Communication Manager and Avaya SES

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 TN799DP CLAN Interface TN2302AP IP Media Processor	HW11 FW030 HW01 FW024 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) 4625SW(H.323) 4610SW (SIP)	2.8 2.8 2.2.2
Avaya 9600 Series IP Telephones		
	9630 (H.323) 9650 (H.323) 9630 (SIP)	1.5 1.5 1.0.13.1
Avaya 6400D Series Digital Telephones		-
Raytheon JPS ACU-2000IP		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 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 ACU-2000IP 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 Network Region: 1
       Far-end Domain: testroom.com

                                             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 20001                                                  Page   1 of   5
                                     STATION

Extension: 20001                         Lock Messages? n               BCC: 0
     Type: 4620                          Security Code:                  TN: 1
     Port: IP                      Coverage Path 1:                 COR: 1
     Name: SIP 20001                   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: 20001
            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 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 during the compliance test.

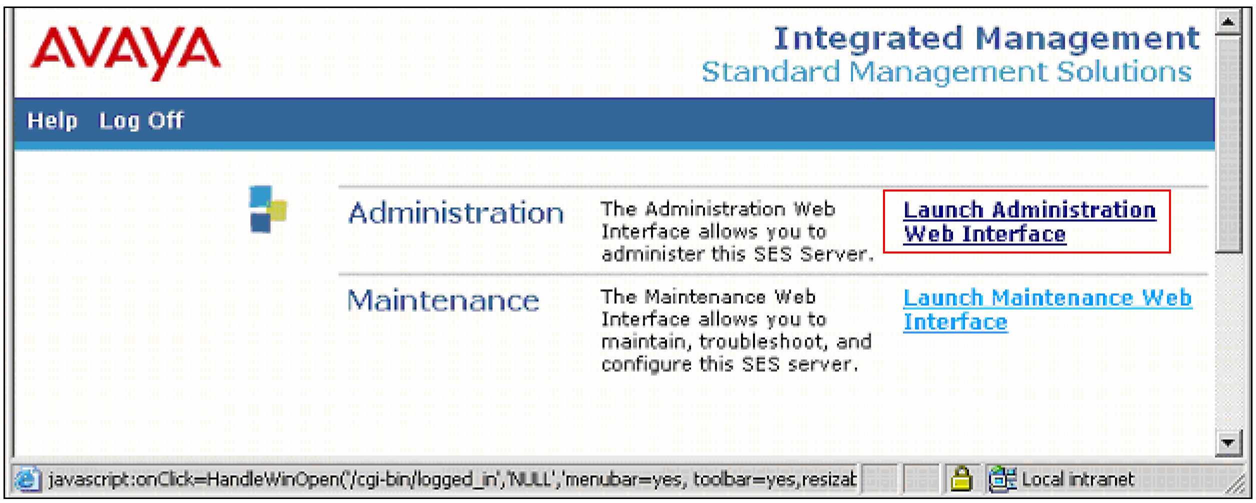
```
list off-pbx-telephone station-mapping

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station        Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                  Set    Select  Mode         Allowed

20001          OPS        20001            1  /   1       both         all
20002          OPS        20002            1  /   1       both         all
20003          OPS        20003            1  /   1       both         all
20004          OPS        20004            1  /   1       both         all
20005          OPS        20005            1  /   1       both         all
20006          OPS        20006            1  /   1       both         all
20007          OPS        20007            1  /   1       both         all
20008          OPS        20008            1  /   1       both         all
20009          OPS        20009            1  /   1       both         all
20010          OPS        20010            1  /   1       both         all
```

4. Configure Avaya SES

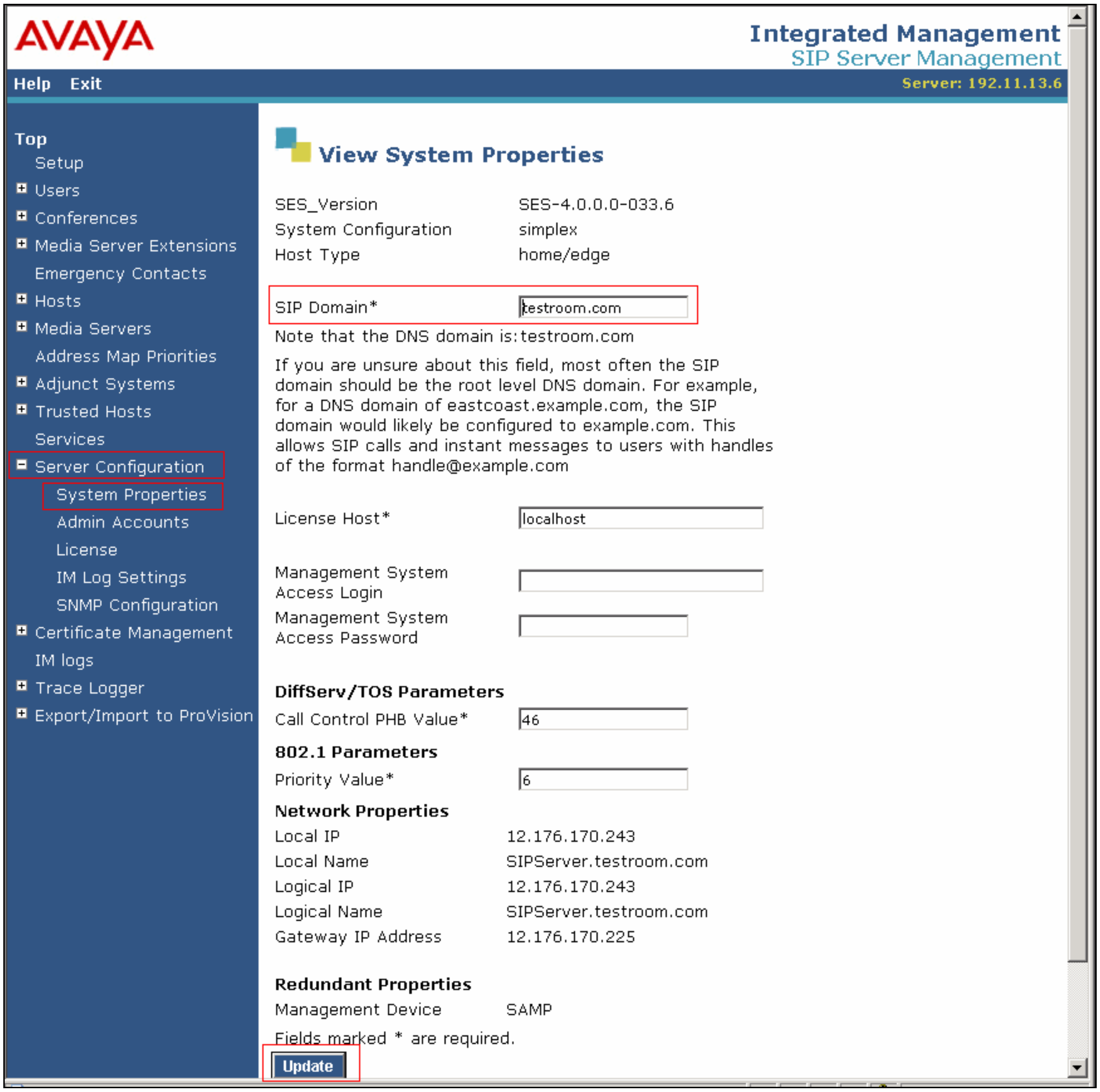
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. LMRs will register with Avaya SES using the SIP user accounts. The highlights 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.



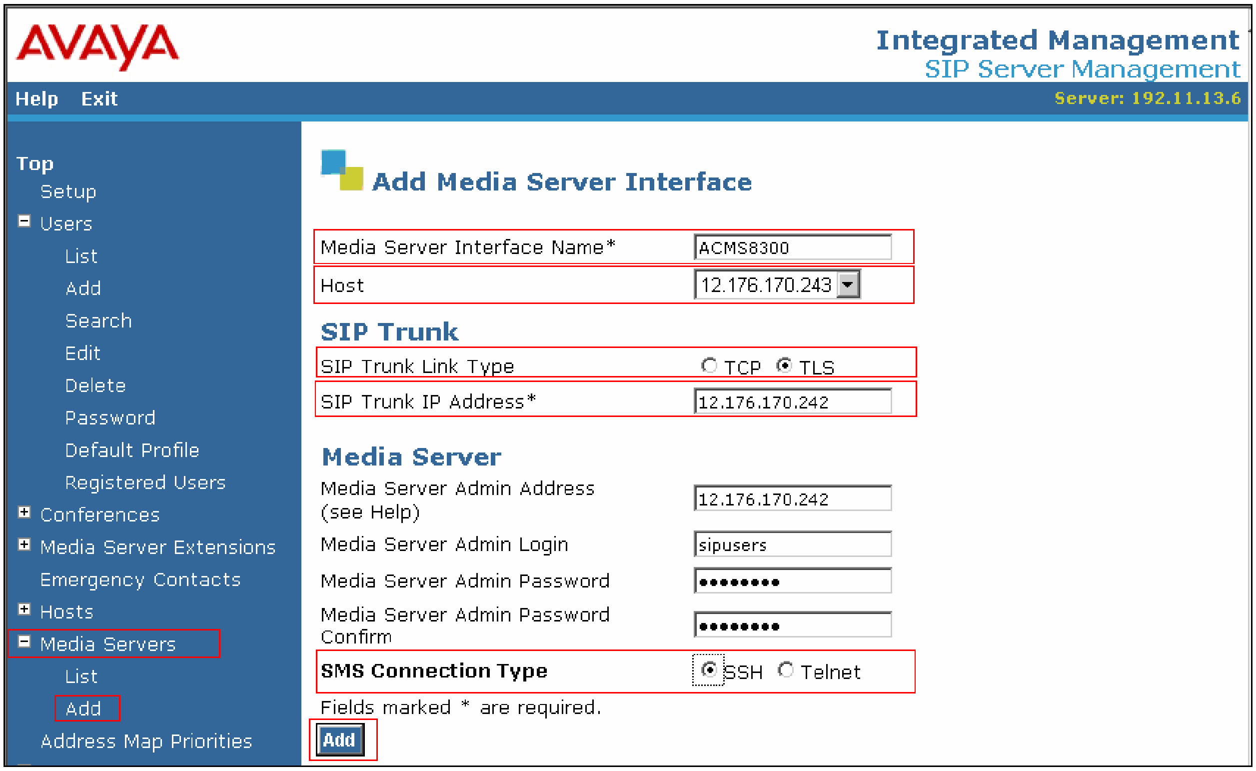
4.2. Configure Media Server Interface

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.



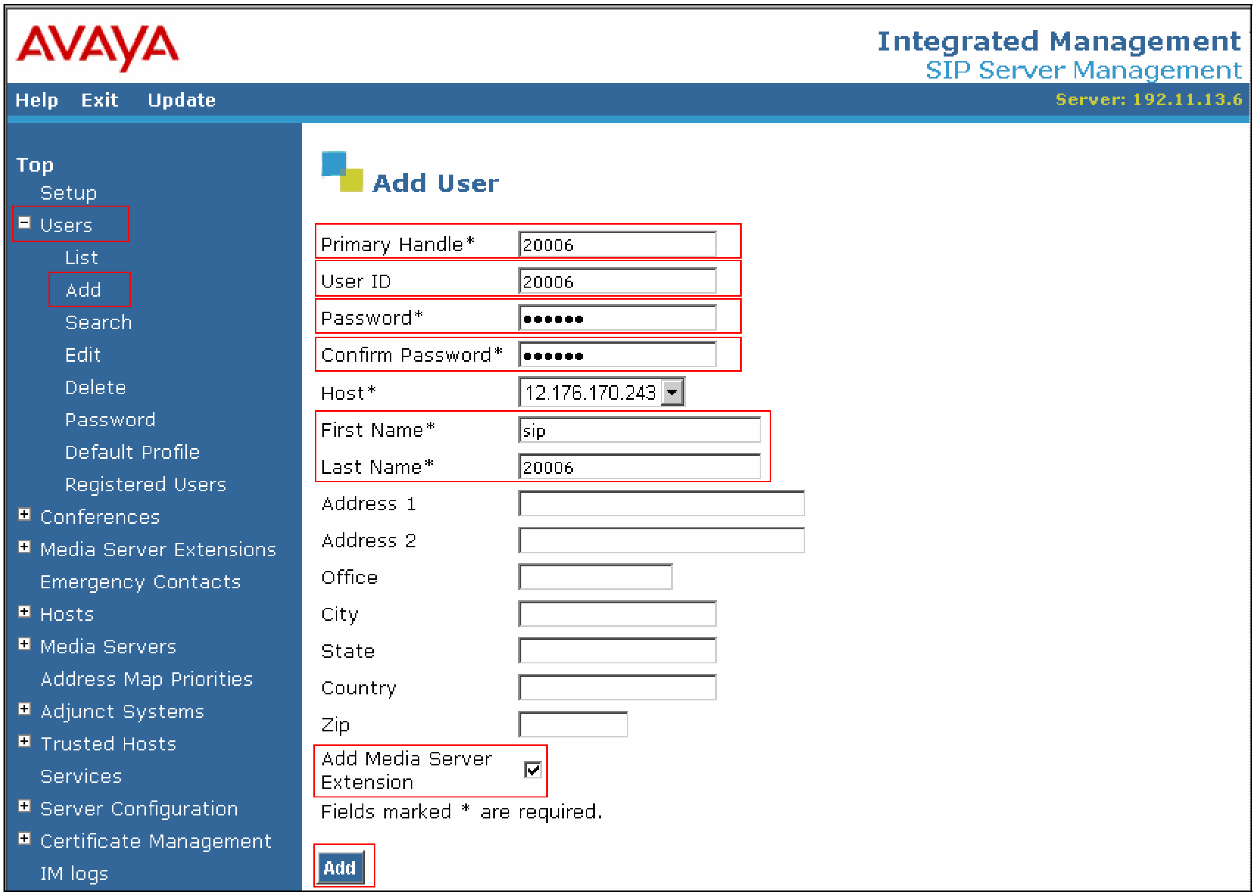
4.3. Configure Users

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

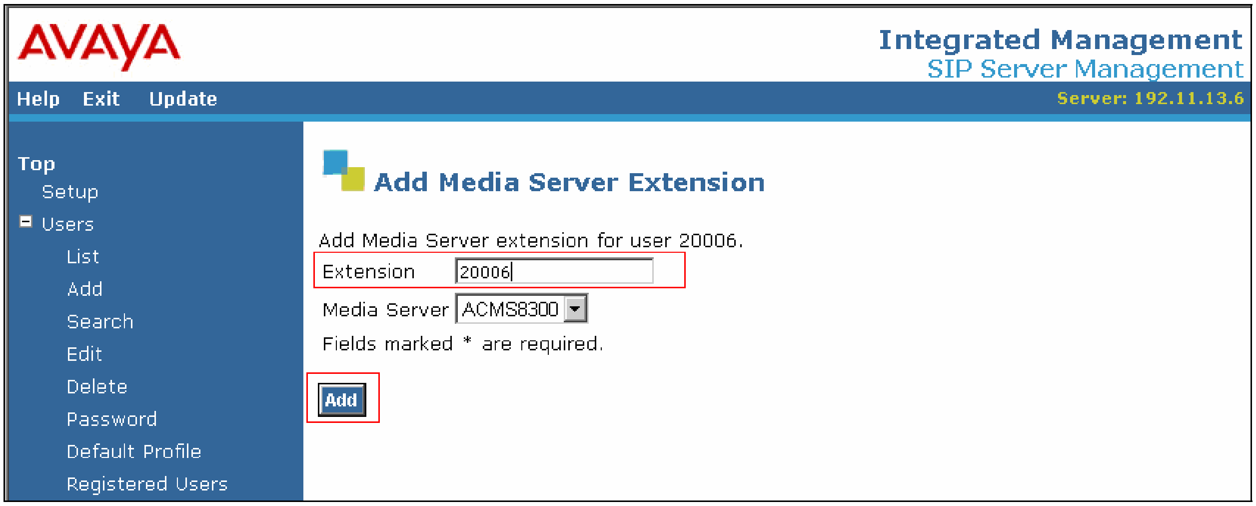
- Primary Handle – Enter the phone number of LMRs. This number was configured in **Section 3.7**.
- User ID – Set to any descriptive name.
- Password / Confirm Password – Enter a password of at least 6 alphanumeric characters; both field entries must match exactly.
- Host – From the drop-down list of 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.



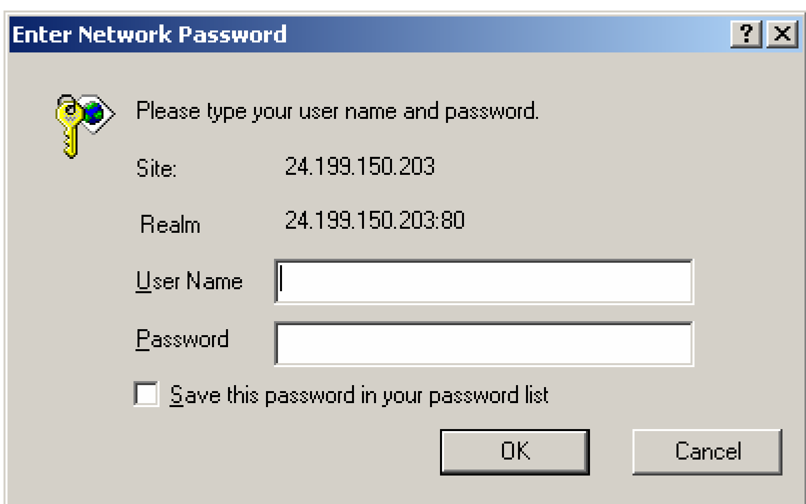
At the next screen, enter the numeric telephone extension to be created in the database. This should match the phone number entry on the off-pbx-telephone station-mapping form in **Section 3.7**. Select the extension's media server from the drop-down list. Click on the **Add** button.



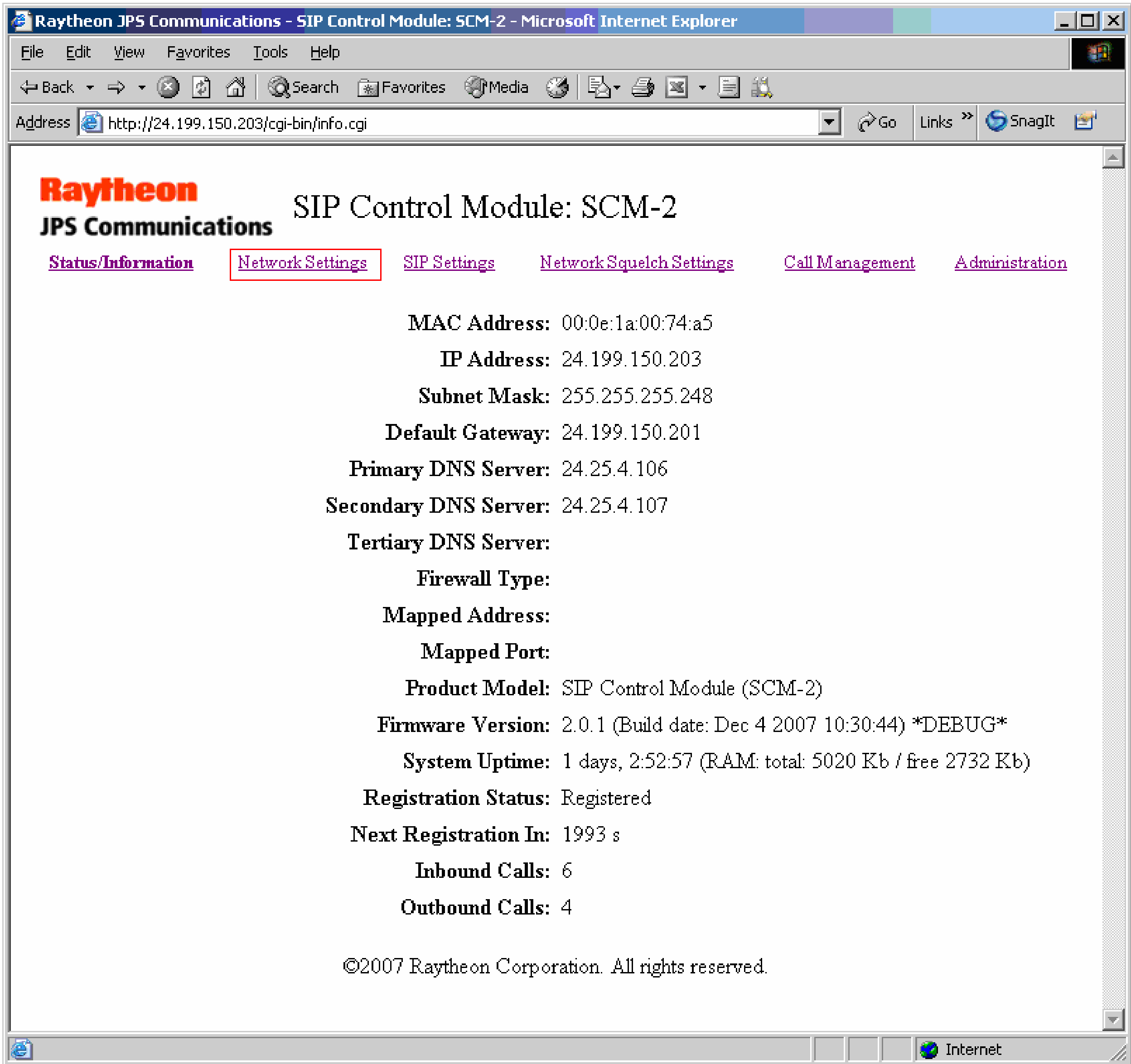
5. Configure Raytheon JPS ACU-2000IP

This section describes the steps for configuring the SIP Control Module (SCM) in the JPS ACU-2000IP unit. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. LMRs 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 SCM>** in the URL, and log in with the appropriate credentials to access the SIP Control Module (SCM) page.

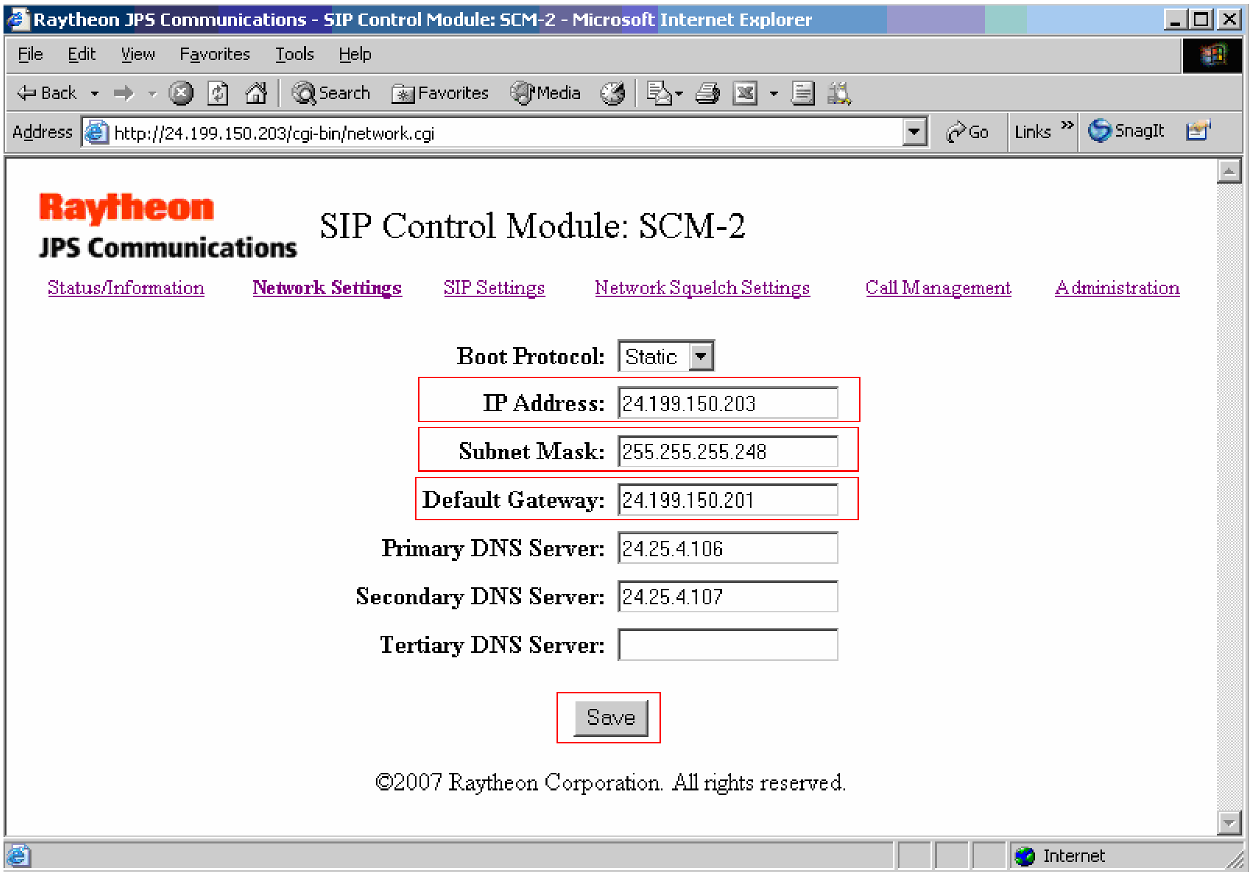


The following screen shows the SCM's main page, which displays the Status/Information. To set or change the IP address of SCM, click **Network Settings**.

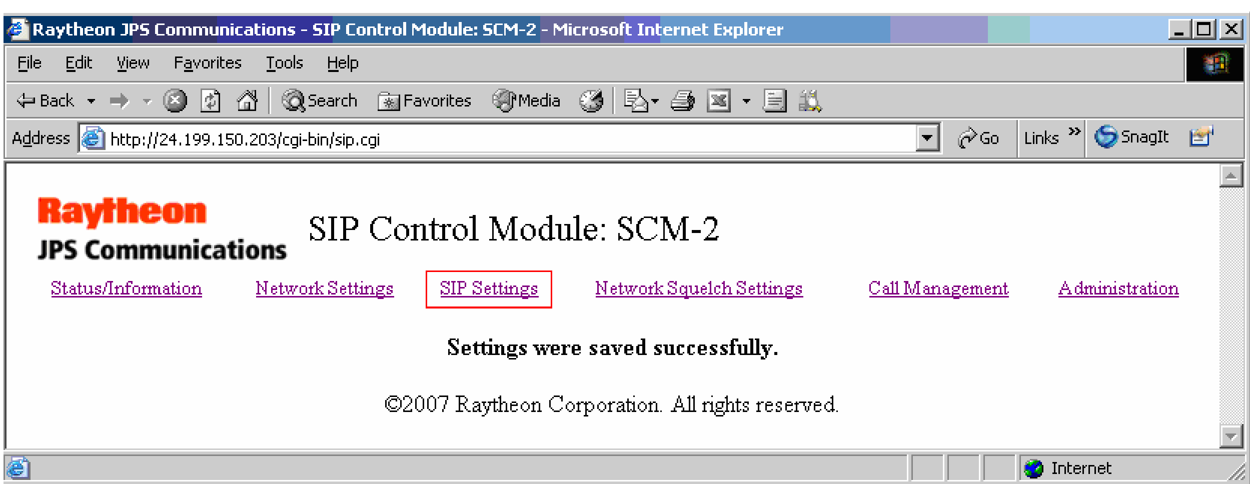


The Network Settings page is utilized to set the IP address, Subnet Mask, and Default Gateway of SCM. The highlighted fields were configured for the compliance test.

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



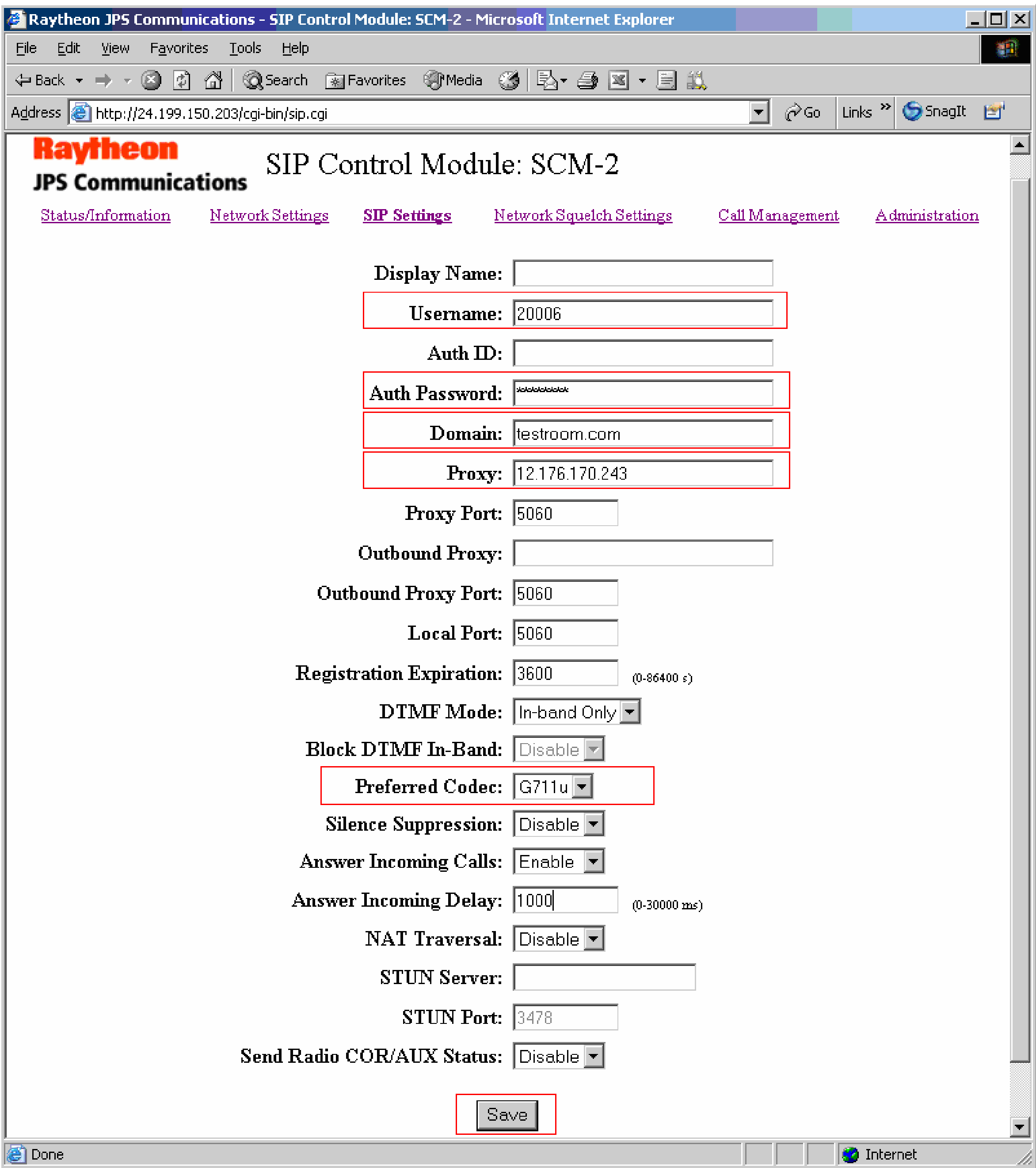
From the SCM's main page, click **SIP Settings** to configure the interface settings to communicate with Avaya SES.



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.



6. Interoperability Compliance Testing

The interoperability compliance testing included basic feature and serviceability testing. The feature testing evaluated the ability of JPS ACU-2000IP 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 JPS ACU-2000IP or the 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 SCM 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.

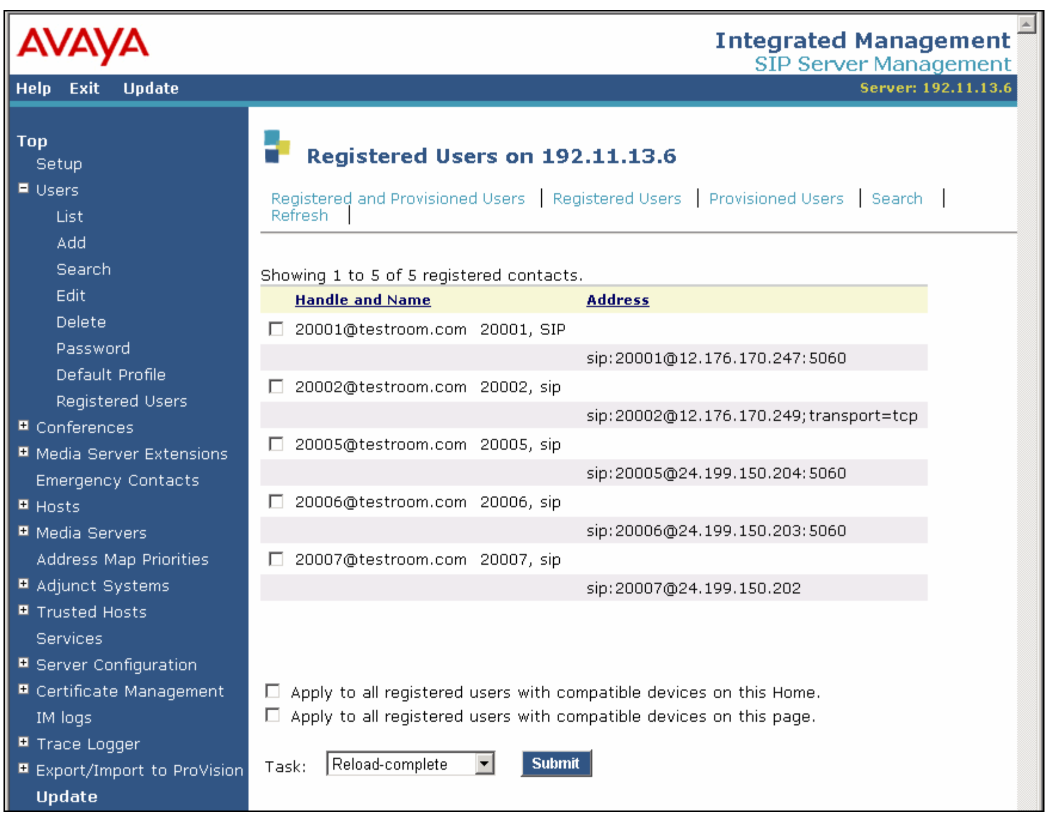
6.2. Test Results

All test cases were executed and passed.

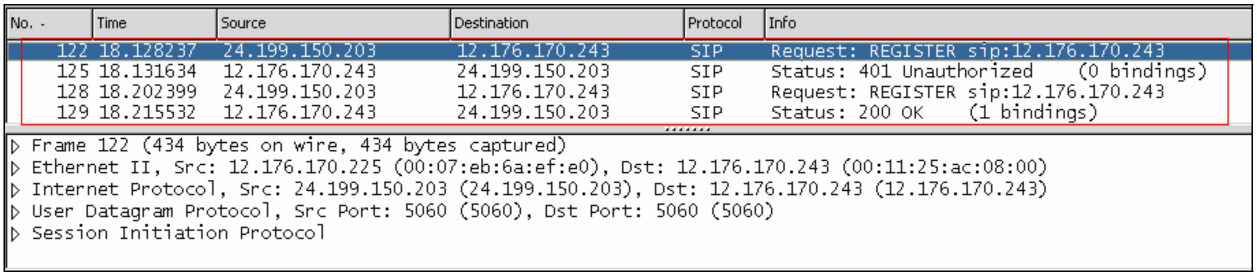
7. Verification Steps

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

- 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.



Using a network protocol analyzer, verify correct REGISTER messages are exchanged between Avaya SES and JPS ACU-2000IP.



8. 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

9. 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.

10. Additional References

This section references the Avaya and 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 ACU-2000IP Intelligent Interconnect System*, Revision 1.1, October 2007.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.