Avaya Solution & Interoperability Test Lab

Configuring Avaya one-X® Mobile SIP for iOS 6.2 as a Remote User with SRTP to Avaya Session Border Controller Advanced for Enterprise 6.2 Server with Avaya Aura® Midsize Enterprise 6.2 Server & Avaya Aura® Messaging 6.2 Server – Issue 1.1

Abstract

These Application Notes describe the configuration steps required to register the Avaya one-X® Mobile SIP for IOS as a Remote User with SRTP to the Avaya Session Border Controller Advanced for Enterprise Server with Avaya Aura® Solution for Midsize Enterprise Server and Avaya Aura® Messaging Server. The Application Notes also identifies how to configure SRTP from the Avaya one-X® Mobile SIP for IOS as a Remote User to the outside interface of the Avaya Session Border Controller Advanced for Enterprise Server and configure SRTP from the inside interface of the Avaya Session Border Controller Advanced for Enterprise Server to the Avaya Aura® Solution for Midsize Enterprise Server. The Application Note also describes how to administer Avaya Aura® Messaging Server to function with SRTP with the Avaya one-X® Mobile SIP for IOS as a Remote User with the Avaya Session Border Controller Advanced for Enterprise Server.

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1. Introduction

These Application Notes describe the configuration steps required to register the Avaya one-X® Mobile SIP for IOS 6.2.2.702 as a Remote User with SRTP to the Avaya Session Border Controller Advanced for Enterprise Server 6.2.0.Q40 with Avaya Aura® Solution for Midsize Enterprise Server 6.2 and Avaya Aura® Messaging Server 6.2. These Application Notes also identify how to configure SRTP from the Avaya one-X® Mobile SIP for IOS as a Remote User to the outside interface of the Avaya Session Border Controller Advanced for Enterprise Server and configure SRTP from the inside interface of the Avaya Session Border Controller Advanced for Enterprise Server to the Avaya Aura® Solution for Midsize Enterprise Server and Avaya Aura® Messaging Server. These Application Note also describe how to administer Avaya Aura® Messaging Server to function with SRTP with the Avaya one-X® Mobile SIP for IOS as a Remote User with the Avaya Session Border Controller Advanced for Enterprise Server.

2. Interoperability Tests

The following sections describe the test scenario used to verify the functionality of the Avaya one-X Mobile SIP for IOS with SRTP as a Remote User with SRTP with the Avaya Session Border Controller Advanced for Enterprise Server.

2.1. Test Description and Coverage

This section provides an overview of the test cases performed after the installation and configuration of the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP with the Avaya Session Border Controller Advanced for Enterprise Server.

2.1.1. Basic IP Telephony Features

The following Basic IP Telephony Features were tested:

- Basic Calls
- Codec Negotiation
- Direct IP-IP Media Shuffling
- PPM Download
- Hold
- Drop

2.1.2. Supplementary Features

The following Supplementary Features were tested:

- Call Forwarding
- Bridged Call Appearance
- Call Pickup with FAC
- TLS

2.1.3. Messaging

Following Messaging Features were verified for Remote Users with SRTP with Avaya Aura® Messaging 6.2 SP1

- Login and access to mailbox
- Leave/Retrieve Voice Mail Messages with proper MWI operation
- Call Sender
- Reply
- Forward Message

2.1.4. Test Results

All test cases passed. The following are the observations for the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server:

- Avaya one-X Mobile SIP for IOS as Remote User registered to the Avaya Session Border Controller Advanced for Enterprise Server uses SRTP for secure encryption of the audio.
- Avaya one-X Mobile SIP for IOS as Remote User registered to the Avaya Session Border Controller Advanced for Enterprise Server has added security as all communication uses TLS.
- The Avaya Session Border Controller Advanced for Enterprise Server is supported as an alternative to VPN in an untrusted network. The Avaya one-X Mobile SIP for IOS connects to Session Manager Server through the Avaya Session Border Controller Advanced for Enterprise Server thus making communication secure.

3. Reference Configuration

The configuration used in these Application Notes is shown in **Figure 1**. The Avaya Aura® Solution for Midsize Enterprise is installed on Avaya System Platform on a S8800 Server. Avaya Aura® Solution for Midsize Enterprise contains Avaya Aura® System Manager, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as virtual machines running with the Avaya Aura® Solution for Midsize Enterprise. Avaya Aura® Communication Manager running as an Evolution Server is used for Off-PBX Station Mapping (OPS). Avaya Aura® Messaging is a template installed on Avaya System Platform on an S8800 Server. The Avaya Session Border Controller Advanced for Enterprise software is installed and configured on Red Hat Linux 5.6 Operating System on an S8800 Server. The diagram indicates logical signaling connections. All components in the Corporate LAN are physically connected to a single Avaya Ethernet Routing Switch (ERS) 2550T-PWR, and are administered in subnet range 192.168.1.x. The Avaya one-X Mobile SIP for IOS Application was obtained from the iTunes App Store and installed on an Apple IPhone 4S. The Avaya one-X Mobile SIP for IOS with SRTP as a Remote User registers to the B1 external interface of the Avaya Session Border Controller Advanced for Enterprise Server.

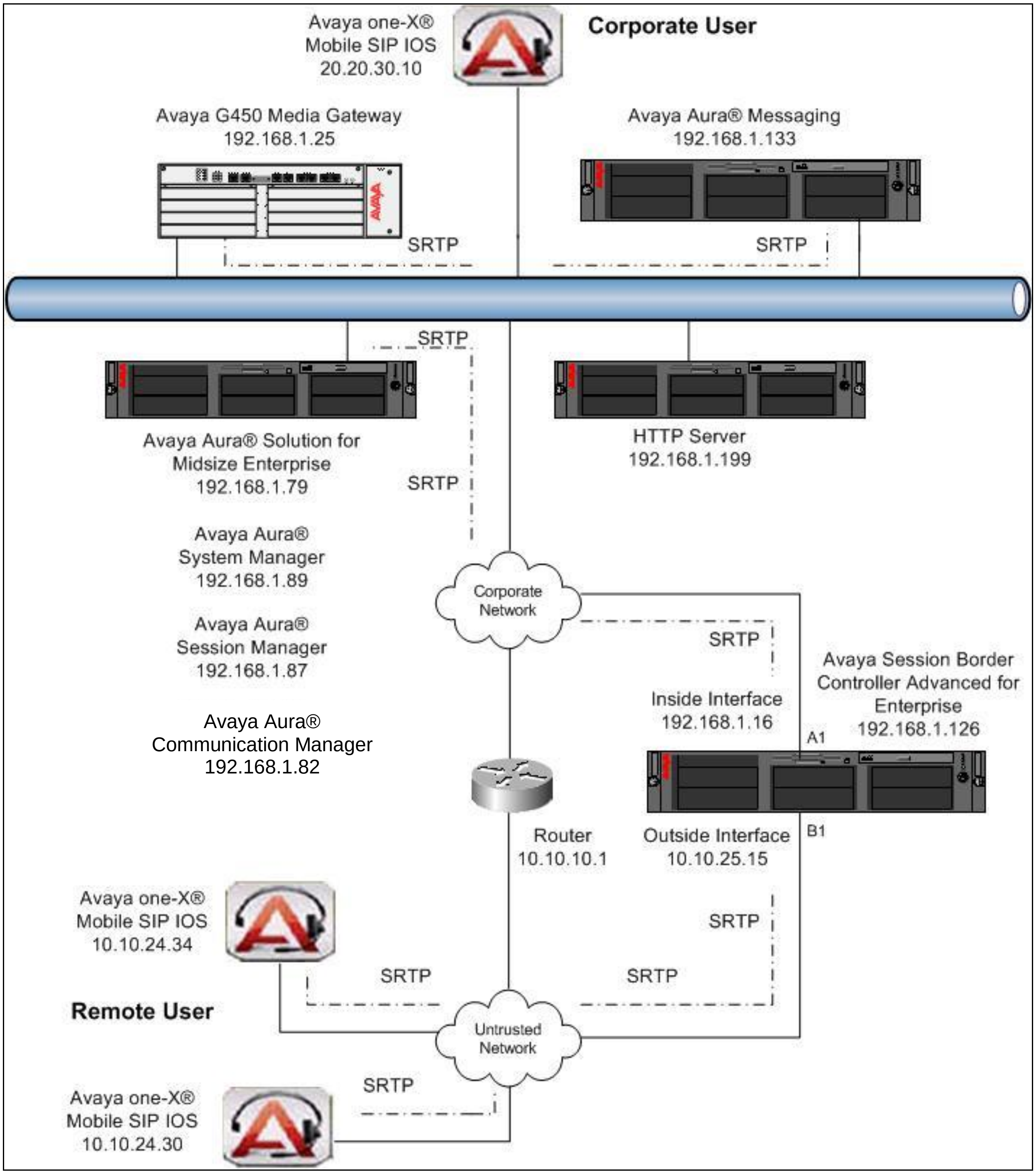


Figure 1 Avaya one-X Mobile SIP for IOS Remote User

4. Equipment and Software Validated

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

Avaya Aura®	Software
Avaya S8800 Server	Avaya Aura® Solution for Midsize Enterprise R6.2 Release 6.2.0.0.3105 Update: Service Pack 4
	Avaya Aura® System Manager R6.2 Release 6.2.16.1.1993 Update: Service Pack 4
	Avaya Aura® Session Manager R6.2 R6.2.4.0.624005 Update: Service Pack 4
	Avaya Aura® Communication Manager R16x.02.0.823.0.20199 Update: Service Pack 4
Avaya G450 Media Gateway	Avaya G450 Media Gateway Release 32.24.0
Avaya S8800 Server	Avaya Aura® Messaging R6.2 MSG 02.0.823.0-109_0102 Update: Service Pack 1
Avaya S8800 Server	Avaya Session Border Controller Advanced for Enterprise Release 6.2.0.Q40
Avaya one-X Mobile SIP iOS	Avaya one-X Mobile SIP iOS R6.2 App Release 6.2.2.702

5. Administer Avaya Aura® Communication Manager Server

This section highlights the important commands for defining the Avaya one-X Mobile SIP for IOS as a Remote User as an Off-PBX Station (OPS) and administering a SIP Trunk and Signaling Group to carry calls between the Avaya one-X Mobile SIP for IOS as a Remote User and the SIP endpoints registered to Session Manager on the Corporate LAN in Communication Manager Server. This section will also explain how to administer SRTP on Communication Manager so that SRTP can be used from the inside interface on the Session Border Controller Server to the Communication Manager Server on the Corporate LAN.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** has been set to the value that has been licensed, and that this value will accommodate addition of the SIP telephones. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to obtain additional capacity.

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

     G3 Version: V15                                 Software Package: Standard
       Location: 2                                RFA System ID (SID): 1
       Platform: 25                               RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 181
                                      Maximum Stations: 2400  9
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 2400  2
                    Maximum Off-PBX Telephones -   OPS: 2400  5
                    Maximum Off-PBX Telephones - PBFMC: 2400  2
                    Maximum Off-PBX Telephones - PVFMC: 2400  0
```

Verify that there are sufficient licenses to administer the SIP Trunk. This is the **Maximum Administered SIP Trunk** value on **Page 2** of System Parameter Customer-Options.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 8000  12
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 8000  0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 100   3
                       Maximum Administered SIP Trunks: 5000  160
  Maximum Administered Ad-hoc Video Conferencing Ports: 8000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer Dial Plan

This section describes the **Dial Plan Analysis** screen**.** This is Communication Manager's way of translating digits dialed by the user. The user can determine the beginning digits and total length for each type of call that Communication Manager needs to interpret. The **Dialed String** beginning with the number **4** and with a **Total Length** of **5** digits will be used to administer the **extension** range used for the one-X Mobile SIP for IOS device.

```
display dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all           Percent Full:    1

       Dialed    Total  Call    Dialed    Total  Call      Dialed    Total  Call
       String    Length Type    String    Length Type      String    Length Type
     1             3    dac
     2             5    aar
     3             5    ext
     35            5    aar
     4             5    ext
     60            4    aar
     7             5    aar
     8             6    aar
     *             2    fac
```

5.3. Administer IP Node-Name

This section describes **IP Node-Name.** This is where Communication Manager assigns the IP Address and node-name to Session Manager. The node-name is **SM** and the IP Address is **192.168.1.87** within Communication Manager Server. The Communication Manager Server automatically populates a processor node name to the IP Address of Communication Manager Server. This node name is **procr** with IP Address **192.168.1.82**.

```
list node-names all

                        NODE NAMES

Type     Name               IP Address
IP       EnterpriseCM       192.168.1.6
IP       SM                 192.168.1.87
IP       SessionM2          192.168.1.60
IP       BSM                192.168.1.157
IP       default            0.0.0.0
IP       procr              192.168.1.82
```

5.4. Administer Signaling Group

This section describes the **Signaling Group** screen. The **Group Type** was set to **sip** and the **Transport Method** was set to **tls**. Since the Avaya one-X Mobile SIP for IOS as a Remote User is using a Communication Manager Evolution Server for Off Pbx Station Mapping the **IMS Enabled** setting must be set to **n**. Since the SIP trunk is between Communication Manager Evolution Server and Session Manager the **Near-end Node Name** is the node name of the "procr" of the Communication Manager Server. The **Far-end Node Name** is the node name of the Session Manager Server. This is **SM**. The **Near-end Listen Port** and **Far-end Listen Port** are both set to port number **5061**. The **Far-end Network-Region** was set to **1.**

```
display signaling-group 120
                                SIGNALING GROUP

 Group Number: 120            Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM





   Near-end Node Name: procr                 Far-end Node Name: SM
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
Enable Layer 3 Test? n                            Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

5.5. Administer Trunk Group

This section describes the **Trunk Group** used to carry calls between the Avaya one-X Mobile SIP IOS as a Remote User. Trunk Group 120 was configured as a SIP Trunk with the **Group Type** set as **sip.** The trunk **Group Name** was set to **To SM**. The **Direction** of the calls was set to **two-way** as there will be calls to and from the Remote SIP Users registered to the Avaya Session Border Controller. The **Service Type** was set to **tie** as the trunk is an internal trunk between Communication Manager Evolution Server and Session Manager. The **Signaling Group** number assigned to this trunk is **120**. The **Number of Members** assigned to this trunk group is **100**. All other fields on this page are left as default.

```
display trunk-group 120                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 120                    Group Type: sip          CDR Reports: y
  Group Name: To SM                  COR: 1        TN: 1        TAC: 120
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? N
                                              Member Assignment Method: auto
                                                       Signaling Group: 120
                                                     Number of Members: 100
```

5.6. Administer Calling Party Number Information

Use the **change private-numbering 0** to add **an Extension Length** of **5** with **Extension code** of **4.** The **Total Length** of the CPN number was **5.** The **change private-numbering 0** command was also used to add an **Extension Code** of **8** for the Messaging hunt group number.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT
Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  4                                            5     Total Administered: 2
 5  8                                            5
```

5.7. Administer Route Selection

Use the **change aar a 5** to administer the automatic alternate route selection to route calls between via the SIP trunk to Session Manager. Calls to the number beginning with **5** that are a **minimum** of **5** digits and a **maximum** of **5** digits in length are sent to routing pattern 120. The **Call Type** was set to **unku.**

```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed           Total      Route     Call    Node  ANI
          String          Min  Max   Pattern    Type    Num   Reqd
    4                     5    5      120       unku          n
```

Use the **change route-pattern 120** to add **trunk group 120** to route pattern 120. Ensure the **Sesure SIP** value was set to **No**.

```
change route-pattern 120                                          Page   1 of   3
                    Pattern Number: 120 Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                 QSIG
                             Dgts                                        Intw
 1: 120  0                                                                n   user
 2:                                                                       n   user
```

5.8. Administer IP Network Region

This section describes **IP Network Region** screen**.** It was decided to place the Avaya one-X Mobile SIP for iOS as a Remote User in network region 1. The **Authoritative Domain** must mirror the domain name of Session Manager. This was **silstack.com**. The codec used on the SIP endpoints were placed in **Codec Set 1**. IP Shuffling was turned on so both **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** were set to **yes.**

```
display ip-network-region 1                                       Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: silstack.com
    Name:
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? 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
```

5.9. Administer IP Codec Set

This section describes the **IP Codec Set** screen. IP Codec **G.711MU, G.711A** and **G.729** were used for testing purposes with the Remote User SIP endpoints.

```
display 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: G.711A            n          2        20
 3: G.729             n          2        20
```

5.10. Verify Off PBX Telephone Station Mapping

This section show the **off-pbx-telephone station-mapping**. The Avaya one-X Mobile SIP for IOS as a Remote User extension **40040** uses off pbx **Application OPS** which is used for SIP enabled telephones. This information is populated in Communication Manager when the Avaya one-X Mobile SIP for IOS as a Remote User is administered in User Management in System Manager. The SIP **Trunk Selection** is set to **aar**. The **Config Set** which is the desired call treatment was set to **1**.

```
display off-pbx-telephone station-mapping 53177                     Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


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

 40040          OPS        40040               1  /   aar     both         all
 40050          OPS        40050               1  /   aar     both         all
 40060          OPS        40060               1  /   aar     both         all
```

The **Call Limit** is set to **6** as shown below. This is the maximum amount of simultaneous calls for extension 40040. The **Mapping Mode** field was set to **both** in this configuration setup. This is used to control the degree of integration between the Remote User SIP telephones. The **Calls Allowed** field was set to **all**. This identifies the call filter type for a SIP Phone. The **Bridged Calls** field was set to **none** as it was not needed for testing purposes.

```
display off-pbx-telephone station-mapping 53177                     Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping     Calls       Bridged       Location
 Extension     Name    Limit       Mode        Allowed     Calls
 40040         OPS      6        both          all         none
 40050         OPS      4        both          all         both
```

5.11. Administer Hunt Group

Hunt Group number **1** was administered and was assigned **Group Name Mango**. **Group Extension 80960** was assigned to hunt group 1. **ucd-mia** was assigned as the **Group Type**.

```
display hunt-group 1                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 1                               ACD? n
              Group Name: Mango                         Queue? n
         Group Extension: 80960                        Vector? n
              Group Type: ucd-mia               Coverage Path: 1
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Select **sip-adjunct** for **Message Center.** The **Voice Mail Handle** was set to 80960 the same value as the **Group Extension** on Page 1. The **Voice Mail Handle** was set to **80960.** The **Routing Digits *08** is used in the **Voice Mail Number** field as a Feature Access Code to access the SIP trunk the hunt group number goes out across.

```
display hunt-group 1                                            Page   2 of  60
                                  HUNT GROUP






                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle        Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     80960                     80960               *08
```

5.12. Administer Coverage Path

Configure a coverage path for the Message Application Subscriber. Use the command **add coverage path n** where **n** is the coverage path number to be assigned. Configure a coverage point, using value **hx** where **x** is the hunt group number defined in **Section 5.11**. In this case its hunt-group 1 or **h1** as shown below.

```
add coverage path n
                                 COVERAGE PATH

                   Coverage Path Number: 3
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n              n
               Busy?             y              y
       Don't Answer?             y              y        Number of Rings: 2
                All?             n              n
 DND/SAC/Goto Cover?             y              y
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1             Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

5.13. Administer Station Screen

This screen describes the **station** form setup for the Avaya one-X Mobile SIP IOS as a Remote User on Communication Manager. This information is populated on Communication Manager when user 40040 is administered in User Management in System Manager in **Section 6.14** The **Extension** used was **40040** with phone **Type 9640SIP**. **Coverage Path 1** was set to **1** as described in **Section 5.12**. The **Name** of the phone was set to **40040, 40040** and all other values on **Page 1** of the station form were left as default.

```
display station 40040                                          Page   1 of   5
                                     STATION

Extension: 40040                         Lock Messages? n               BCC: 0
     Type: 9640SIP                       Security Code:                  TN: 1
     Port: S00010                      Coverage Path 1: 1               COR: 1
     Name: 40040,40040                 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: 40040
            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

                                                         IP Video? n
```

The **SIP Trunk** value was set to **aar** on **Page 6** of the station form.

```
add station 40040                                              Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

5.14. Administer SRTP on Communication Manager

It was decided that SRTP would be administered from the inside interface of the Session Border controller to the Communication Manager Server. There are a number of settings on Communication Manager that need to be set in order for SRTP to function correctly. The **change system-parameters customer-option** command was used to set the **set media encryption over IP** setting on **Page 4** to **YES.**

```
change system-parameters customer-options                         Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                 IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                           ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                             ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                    ISDN-PRI? y
              ESS Administration? y                 Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                       Malicious Call Trace? y
     External Device Alarm Admin? y                     Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
```

The change **system-parameters ip-options** command was used to set the **Override ip-codec-set for SIP direct-media connections** setting on **Page 4** to **NO**.

```
change system-parameters ip-options                               Page   4 of   4
                          IP-OPTIONS SYSTEM PARAMETERS

 SYSLOG FROM TN BOARDS
   Local Facility #: local4

 Dest #1 IP address:                                              Port #: 514
 Dest #2 IP address:                                              Port #: 514
 Dest #3 IP address:                                              Port #: 514


 Override ip-codec-set for SIP direct-media connections? n
```

The change **system-parameters features** command was used to set the **SDP Capability Negotiation for SRTP** setting on **Page 19** to **YES.**

```
change system-parameters features                                 Page  19 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
                   Direct IP-IP Audio Connections? y
                             IP Audio Hairpinning? n
                          Synchronization over IP? n
              SDP Capability Negotiation for SRTP? y
                    SIP Endpoint Managed Transfer? y
```

The **change ip-codec 1** command was used to set the **Media Encryption** setting on **Page 1** of the ip-codec setting to **1-srtp-aescm128-hmac80.**

```
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: G.711A           n          2        20
 3: G.729            n          2        20
 4:
 5:
 6:
 7:


     Media Encryption
 1: 1-srtp-aescm128-hmac80
 2:
 3:
```

5.15. Save Translations

Use the **save translation** command to save these changes.

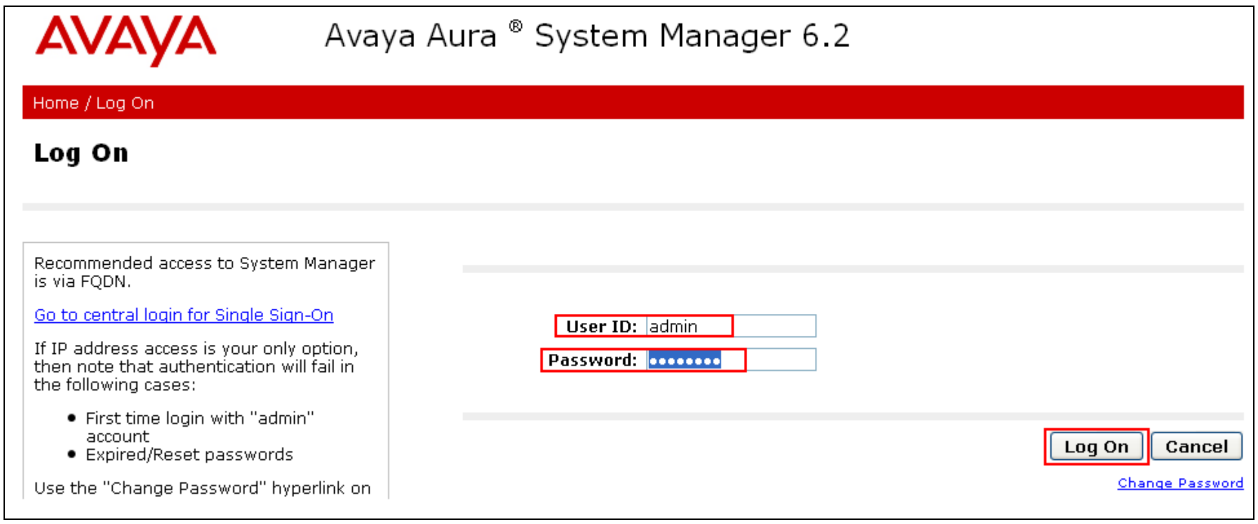
```
save translation

                               SAVE TRANSLATION

           Command Completion Status                          Error Code

           Success                                            0
```

6. Administer Avaya Aura® Session Manager

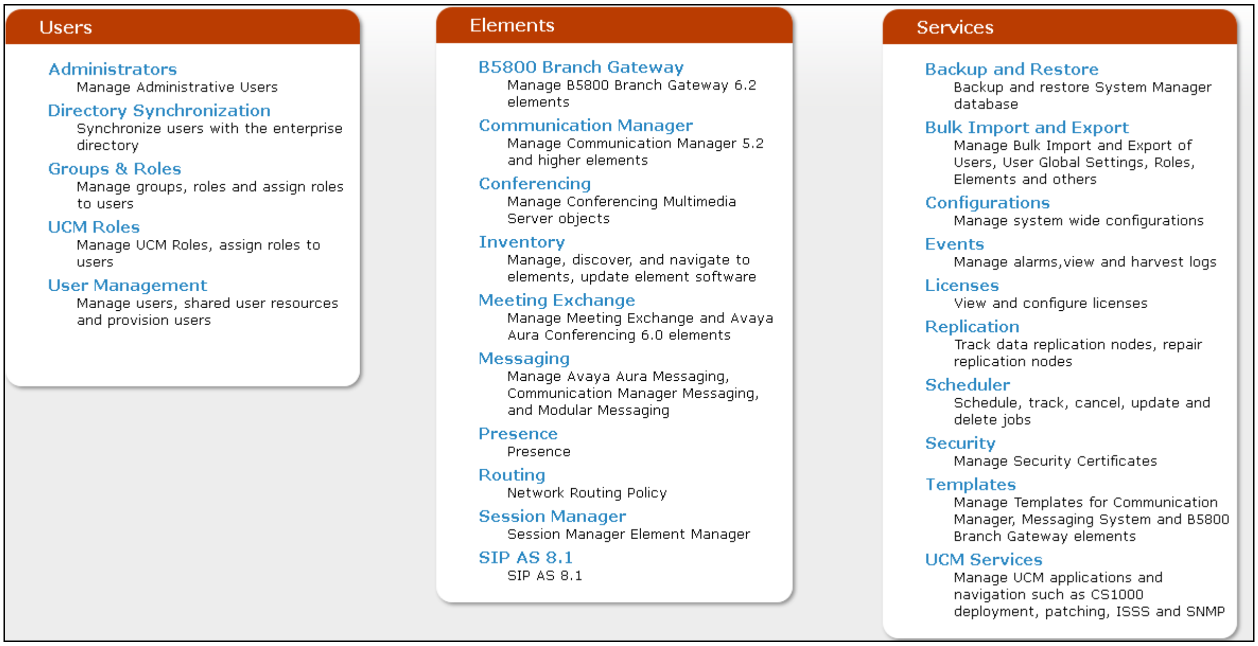
The following steps describe configuration of the Avaya one-X Mobile SIP for IOS as a Remote User with Session Manager. This section describes administering SIP Entities between Session Manager and the Communication Manager Server in order to establish a SIP Entity link between Session Manager and the Communication Manager Server. It also discusses administering the SIP Entities between Session Manager and the Messaging Server. Administering the Avaya one-X Mobile SIP for IOS as a Remote User in User Management to register to the Avaya Session Border Controller Advanced Enterprise Server with Session Manager is also discussed.

6.1. Access Avaya Aura® System Manager

Access the System Manager web interface, by entering **http://<ip-addr>/SMGR** as the URL in an Internet browser, where *<ip-addr>* is the IP address of the server running System Manager graphical user interface. Log in with the appropriate **User ID** and **Password** and press the **Log On** button to access System Manager.

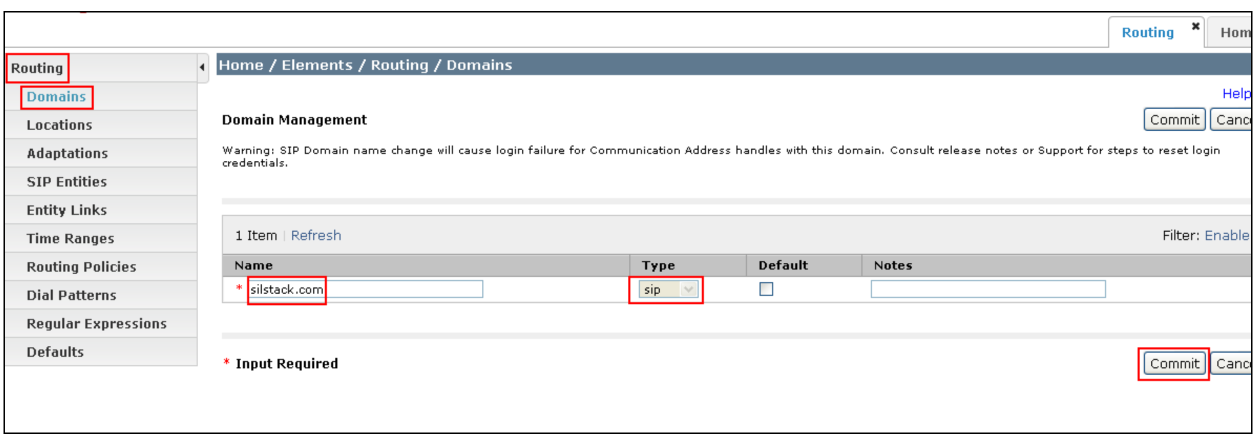


The **main menu** of the **System Manager Graphical User Interface** is displayed in the following screenshot.



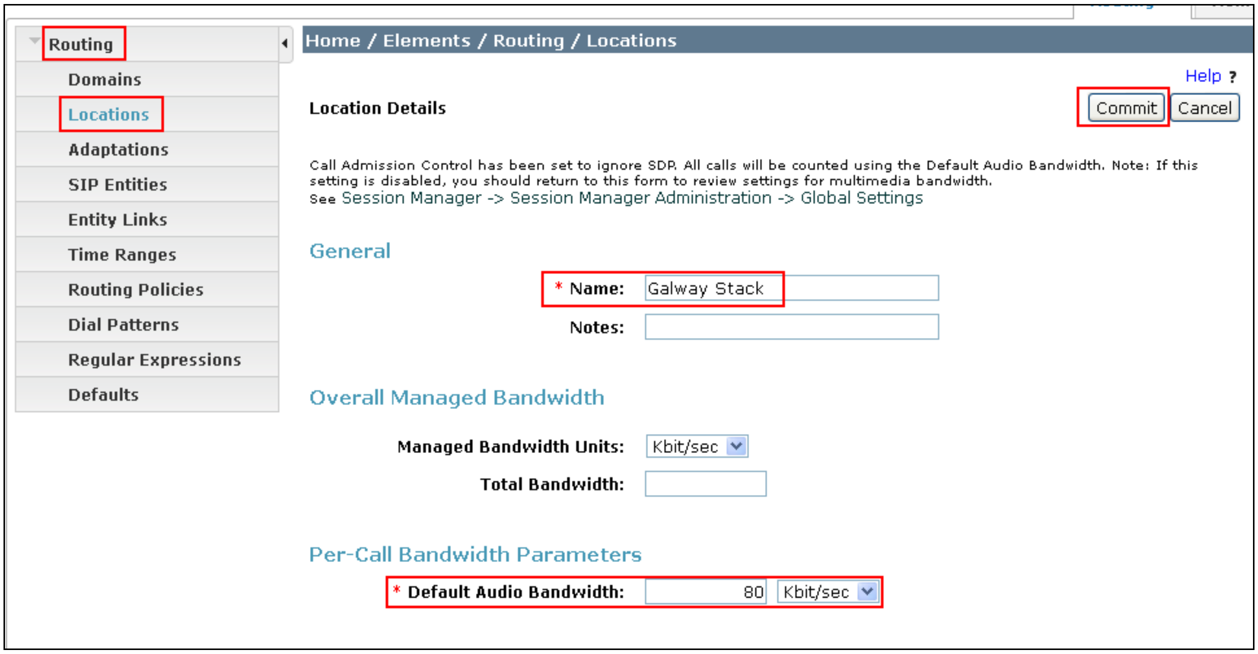
6.2. Administer SIP Domain

The following screenshot shows the configuration used to add a **SIP Domain**. Under the heading **Routing** on the left hand side of the system management interface of System Manager, access the sub heading **Domains**. The name of the SIP Domain used in Session Manager **silstack.com** was added. The Type was set to **sip**. Press the **Commit** button to add the SIP Domain.

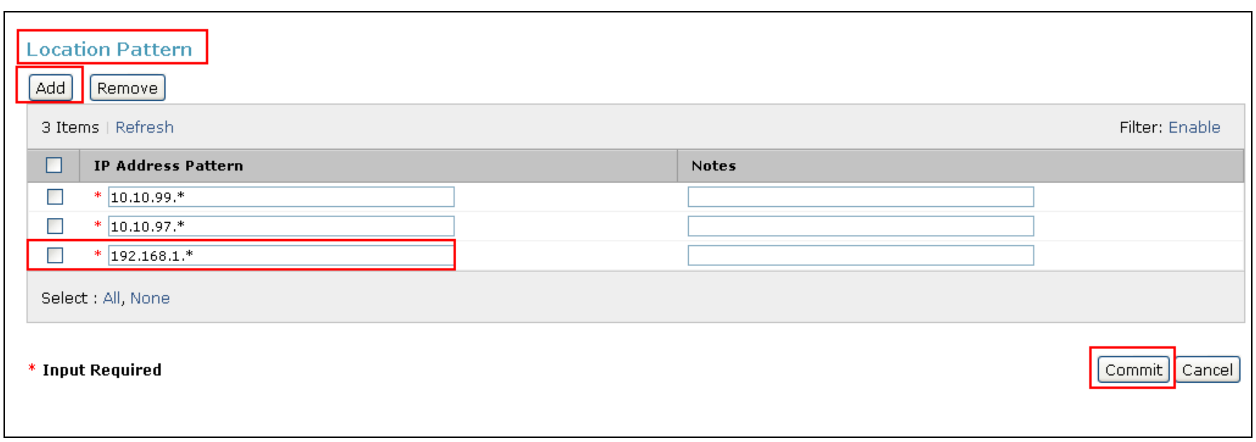


6.3. Administer Location

To add a new Location, click on **Routing** and access the **Locations** sub heading. A location **Name Galway Stack** was added to the Session Manager. The **Default Audio Bandwidth** was set to **80Kbit/sec**. The **Commit** button was pressed to confirm changes. Locations are used to identify logical and physical locations where SIP entities reside for the purposes of bandwidth management or location based routing.

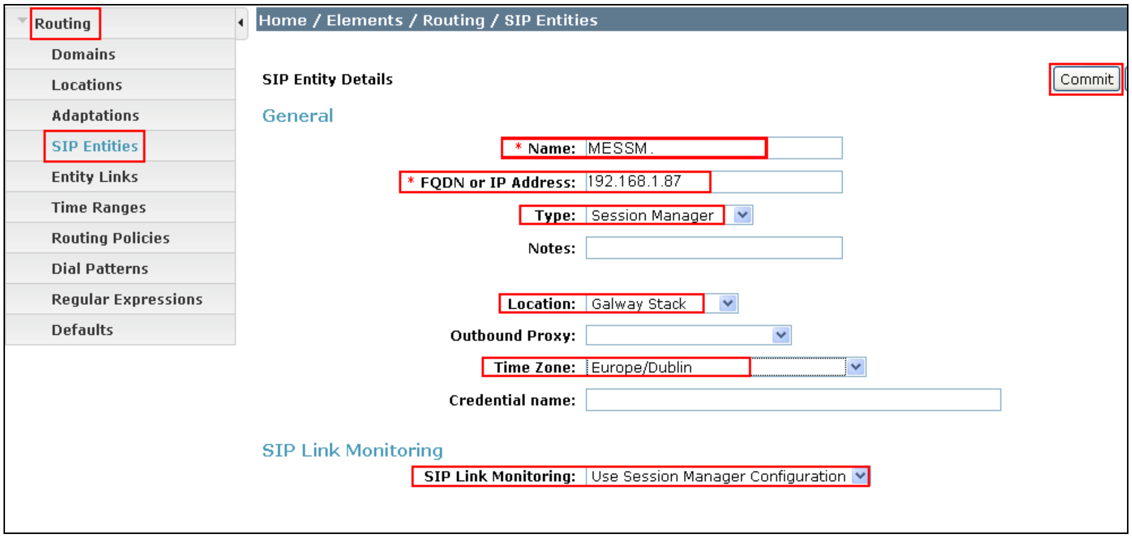


User **Location Pattern** an IP Address Pattern for **192.168.1.x** was added. The **Commit** button was pressed to add the IP Address Pattern to the Location.

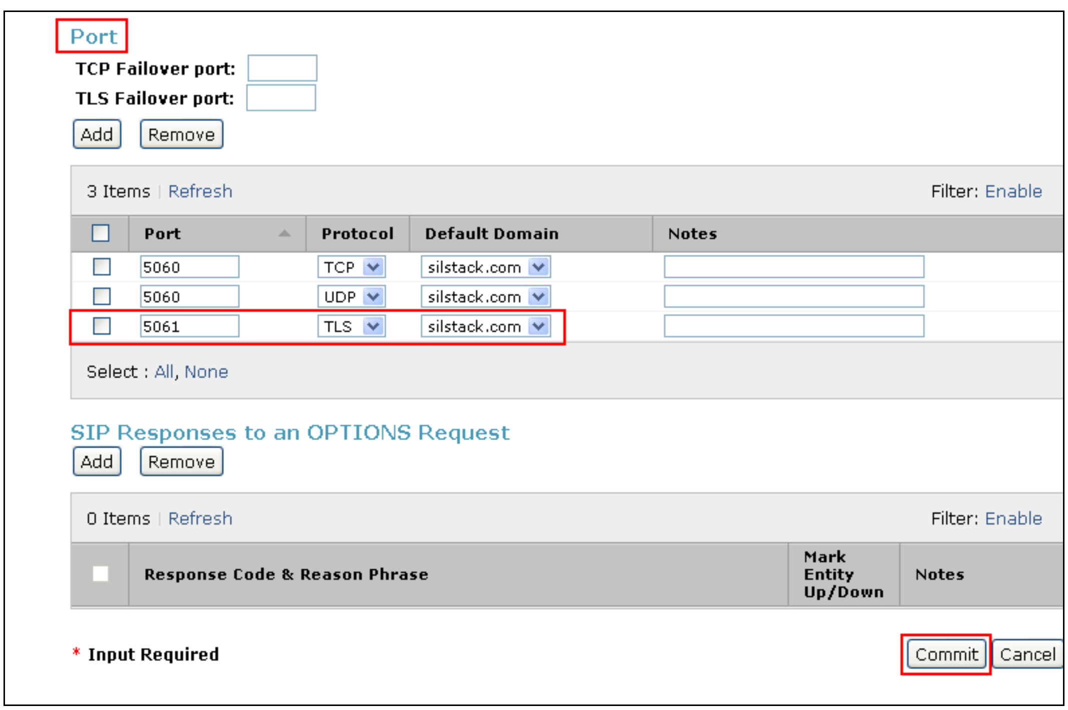


6.4. Administer Avaya Aura® Session Manager SIP Entity

This application note assumes that the basica installation steps for integrating Session Manager with System Manager have already been completed. The screenshot below shows what the completed Session Manager administration looks like in System Manager.

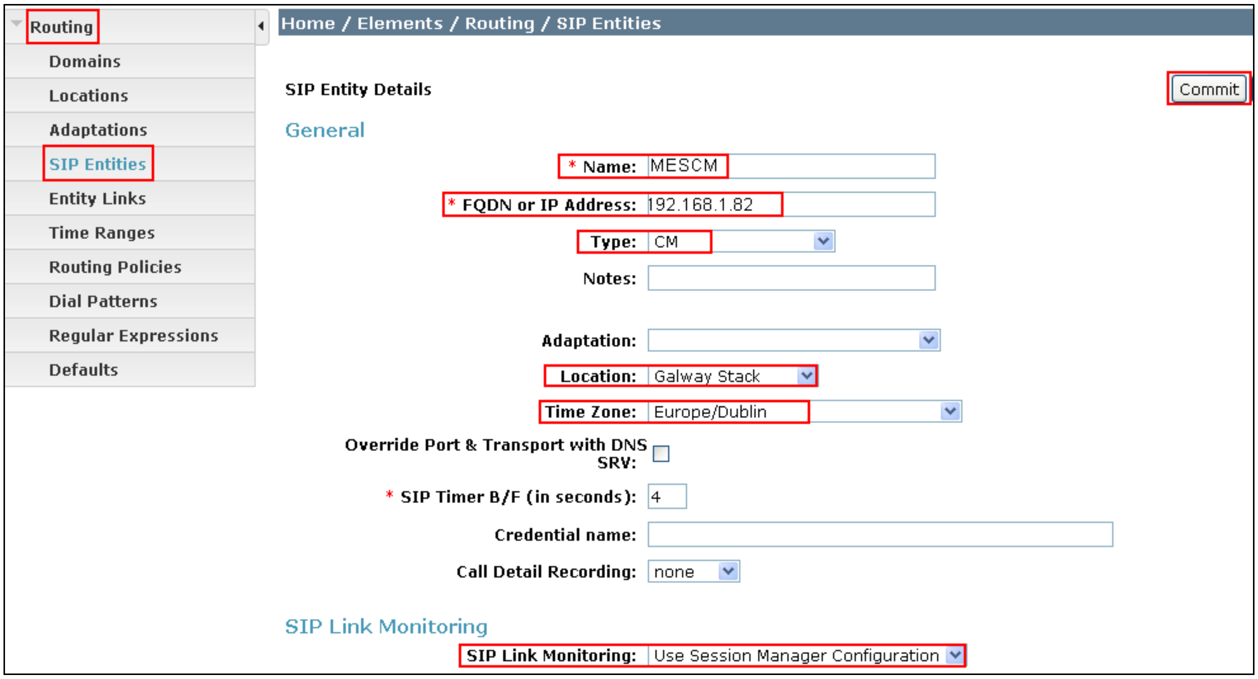


The following screenshot shows what **Port** settings need to be configured for the SIP Entity. With the signaling protocol being set to **TLS** port **5061** was used in the SIP Entity SIP trunk. Press the **Commit** button.



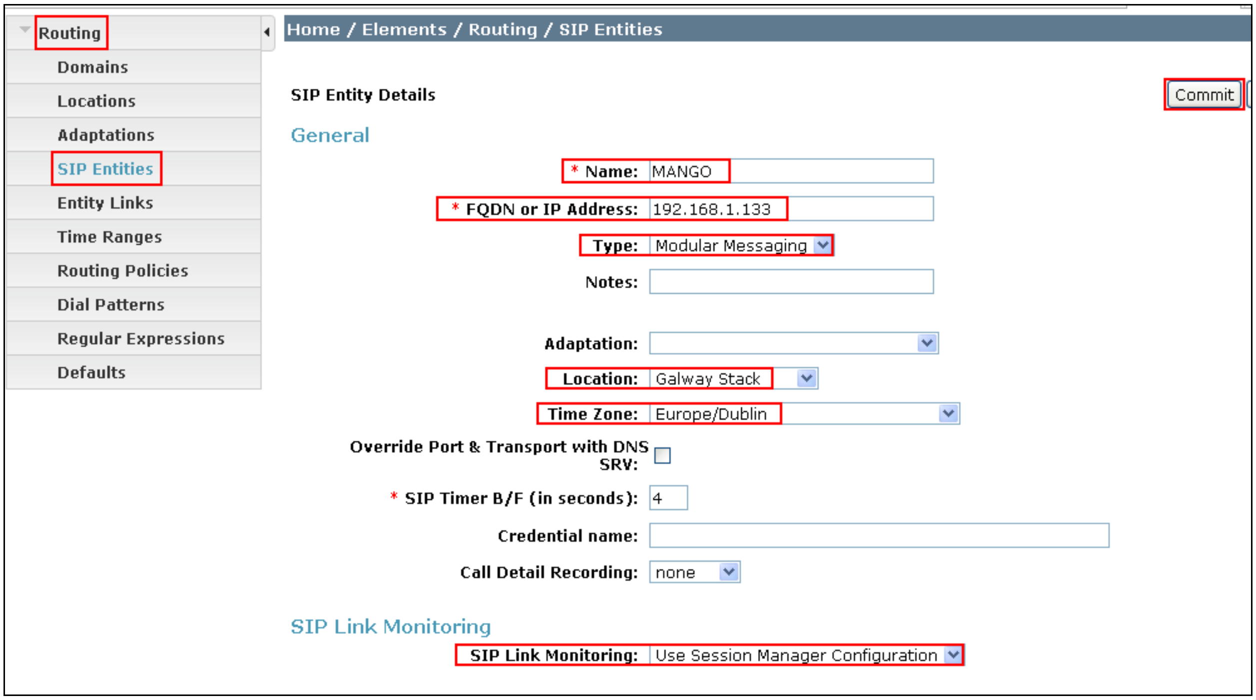
6.5. Administer Avaya Aura® Communication Manager Server SIP Entity

The Communication Server SIP Entity is the second part of the link between the Session Manager and the Communication Manager Server. The **Name** of the SIP Entity was **MESCM.** The **FQDN or IP Address** was set to **192.168.1.82** which was the IP Address of the Communication Manager Server. The **Type** was set to **CM** for Communication Manager**.** The Location was set to **Galway Stack** and the **SIP Link Monitoring** was set to **Use Session Manager Configuration**. Press the **Commit** button.



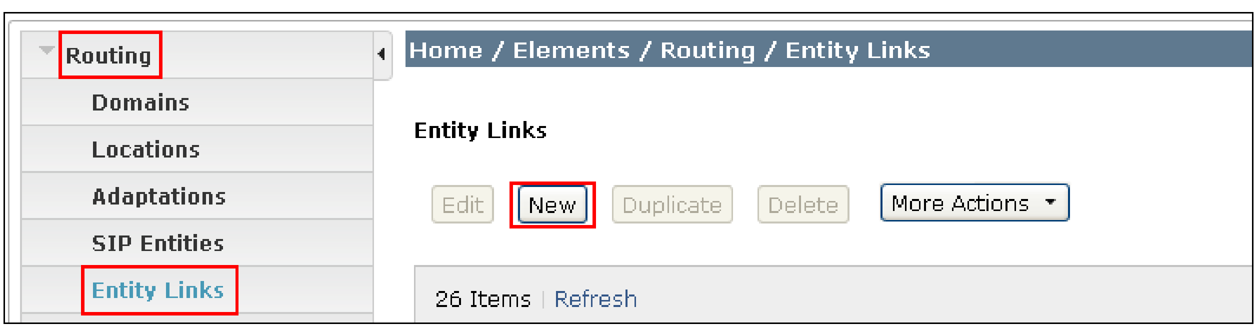
6.6. Administer Avaya Aura® Messaging SIP Entity

The following describes the Messaging SIP Entity to the Session Manager. The **Name** of the SIP Entity was **MANGO.** The **FQDN or IP Address** was set to **192.168.1.133** which was the IP Address of the Messaging Server. The **Type** was set to **Modular Messaging** for Messaging**.** The Location was set to **Galway Stack** and the **SIP Link Monitoring** was set to **Use Session Manager Configuration**. Press the **Commit** button.

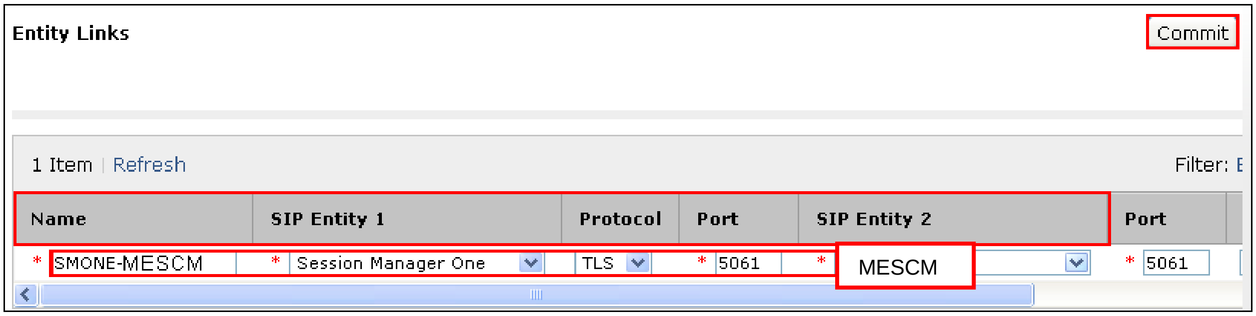


6.7. Administer SIP Entity Link

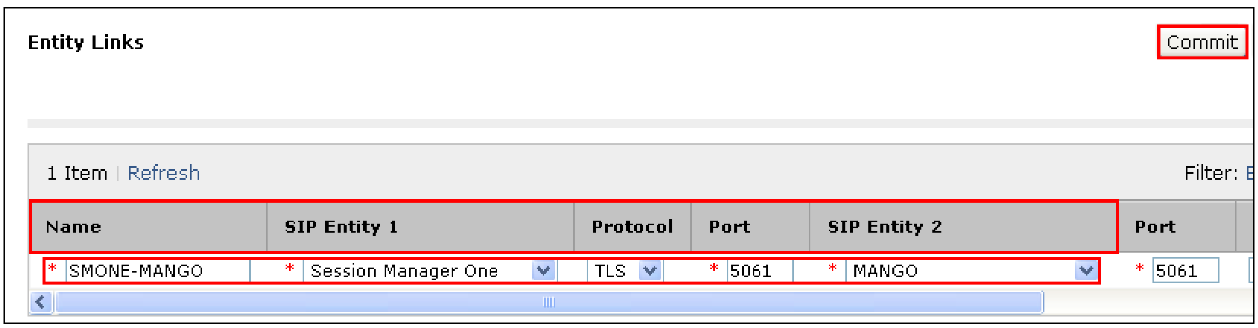
To administer the SIP Entity link access the sub heading **Routing → Entity Links** on the left hand side of the Session Manager GUI. Select the **New** button.



The SIP Entity Link shown below is the link between Session Manager and the Communication Manager Server. The Name of the **Entity Link** was **SMONE-MESCM**. The **SIP Entity 1** was set to **Session Manager One.** The **Protocol** was **TLS** and the **Port** was **5061**. The **SIP Entity 2** was set to **MESSM.**

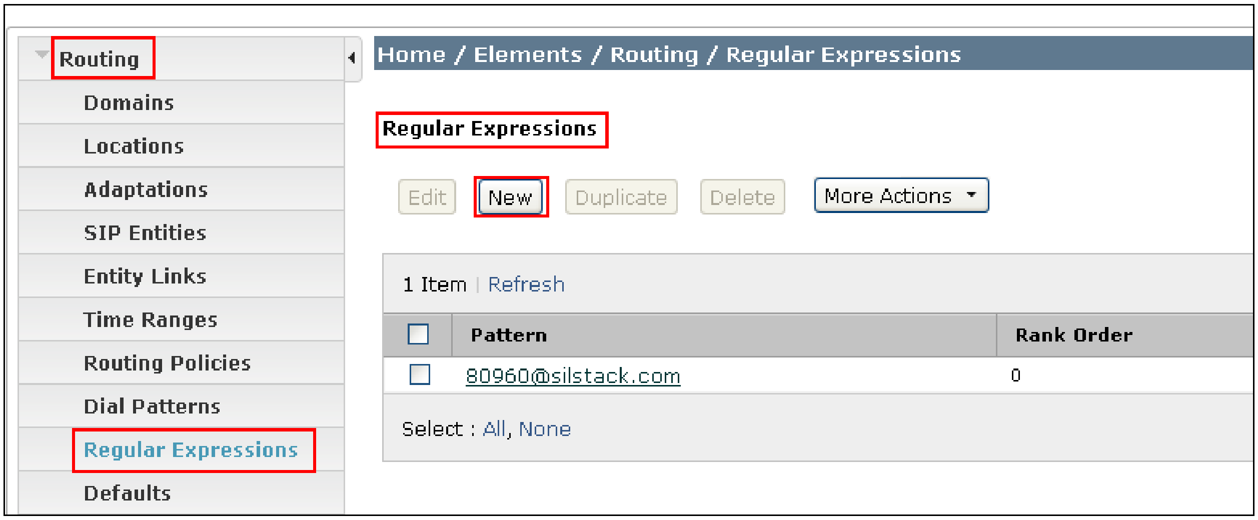


The SIP Entity Link shown below is the link between Session Manager and the Messaging Server. The Name of the **Entity Link** was **SMONE-MANGO**. The **SIP Entity 1** was set to **Session Manager One.** The **Protocol** was **TLS** and the **Port** was **5061**. The **SIP Entity 2** was set to **MANGO.**



6.8. Administer Regular Expression

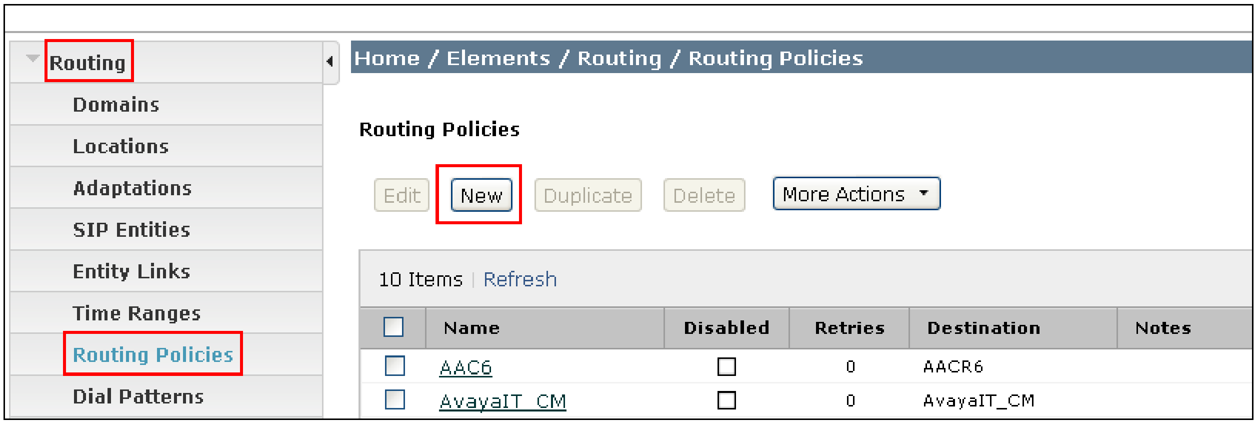
Select **Routing → Regular Expressions**. Under the **Regular Expressions** field select the **New** button.



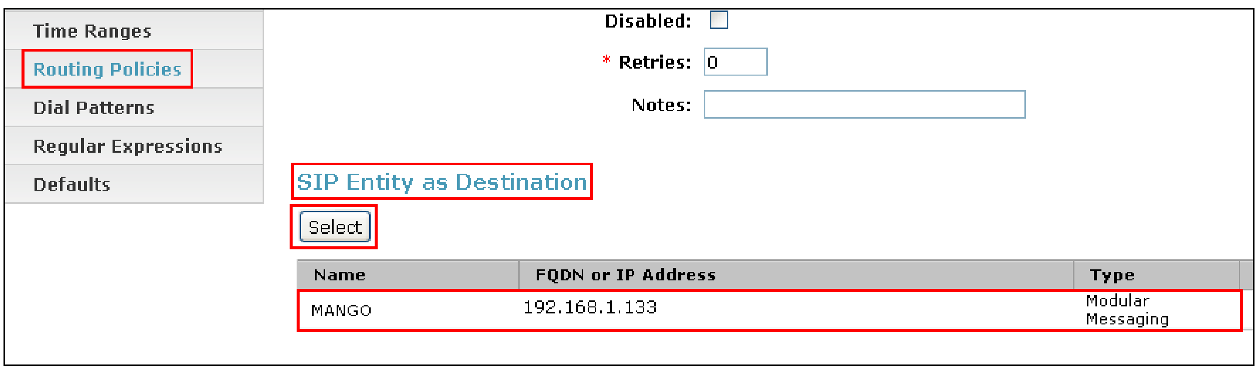
The **Pattern** was set to **80960@silstack.com**. The **Rank Order** was set to **0.** The **Commit** button was pressed to save the changes. This matches the Hunt Group Extension configured in **Section 5.11.**

6.9. Administer Routing Policy

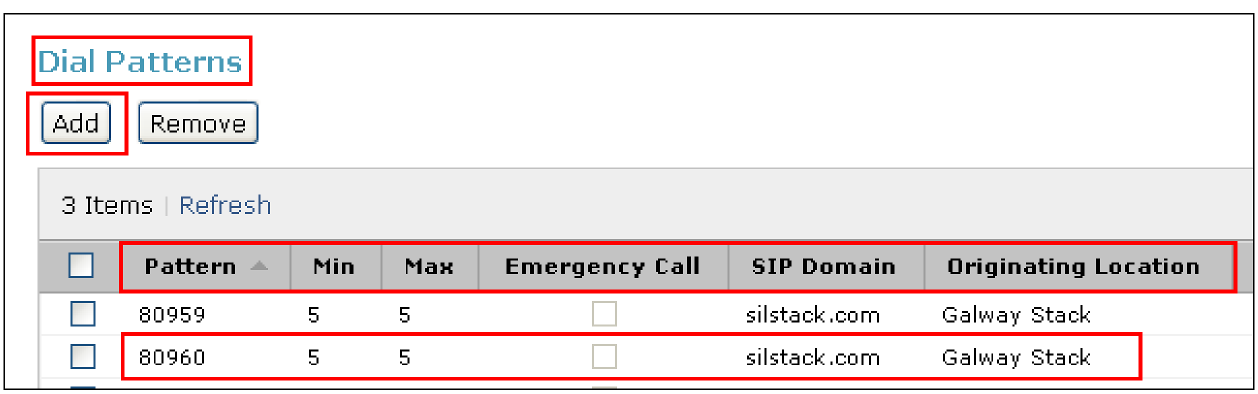
Select **Routing → Routing Policies**. Under the **Routing Policies** field select the **New** button.



Under **Routing Policies** the **SIP Entity as Destination** with the **Name** as **MANGO** and the **IP Address** as **192.168.1.133** and the **Type** set as **Modular Messaging** was **selected.**

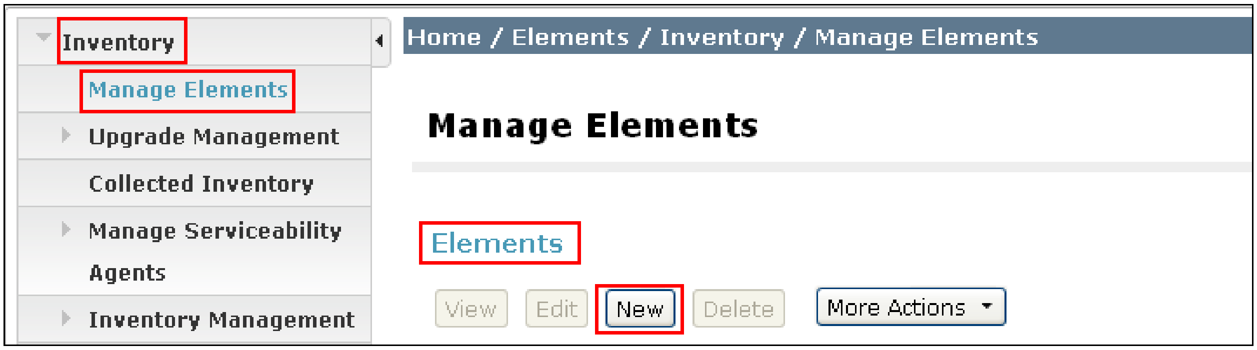


Under **Dial Patterns** the **Pattern** for **80960** with a **minimum** length of **5** digits a **maximum** length of **5** digits a **SIP Domain** as **silstack.com** and **Originating Location** as **Galway Stack** was **added.**

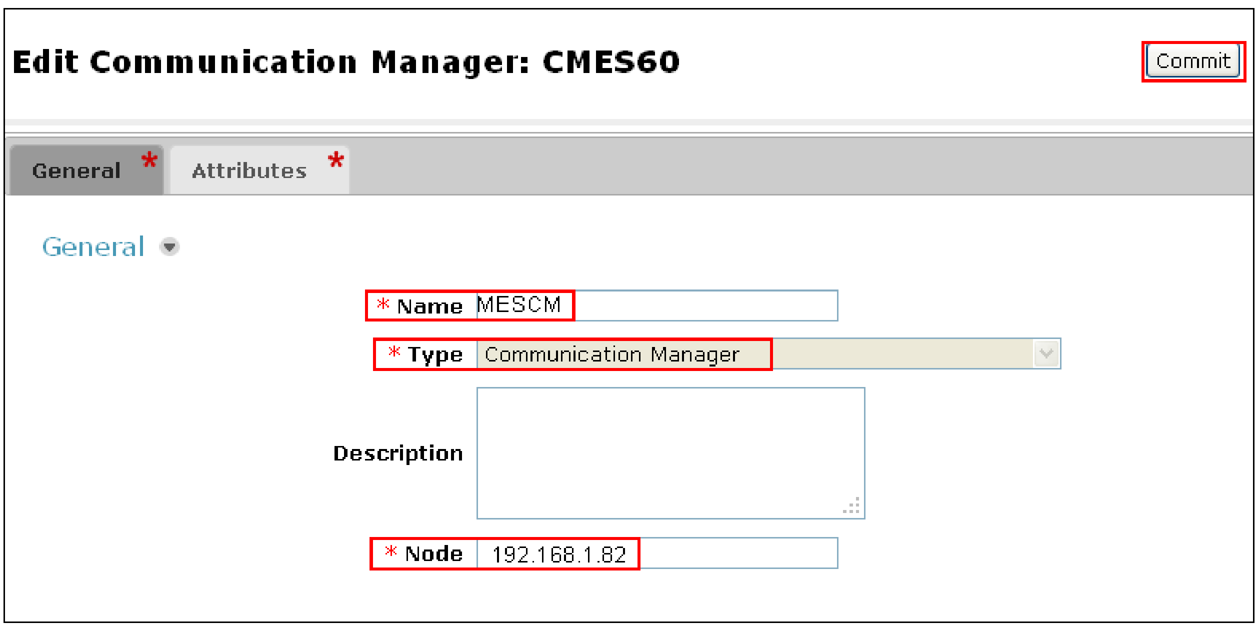


6.10. Administer Avaya Aura® Communication Manager as a Managed Element

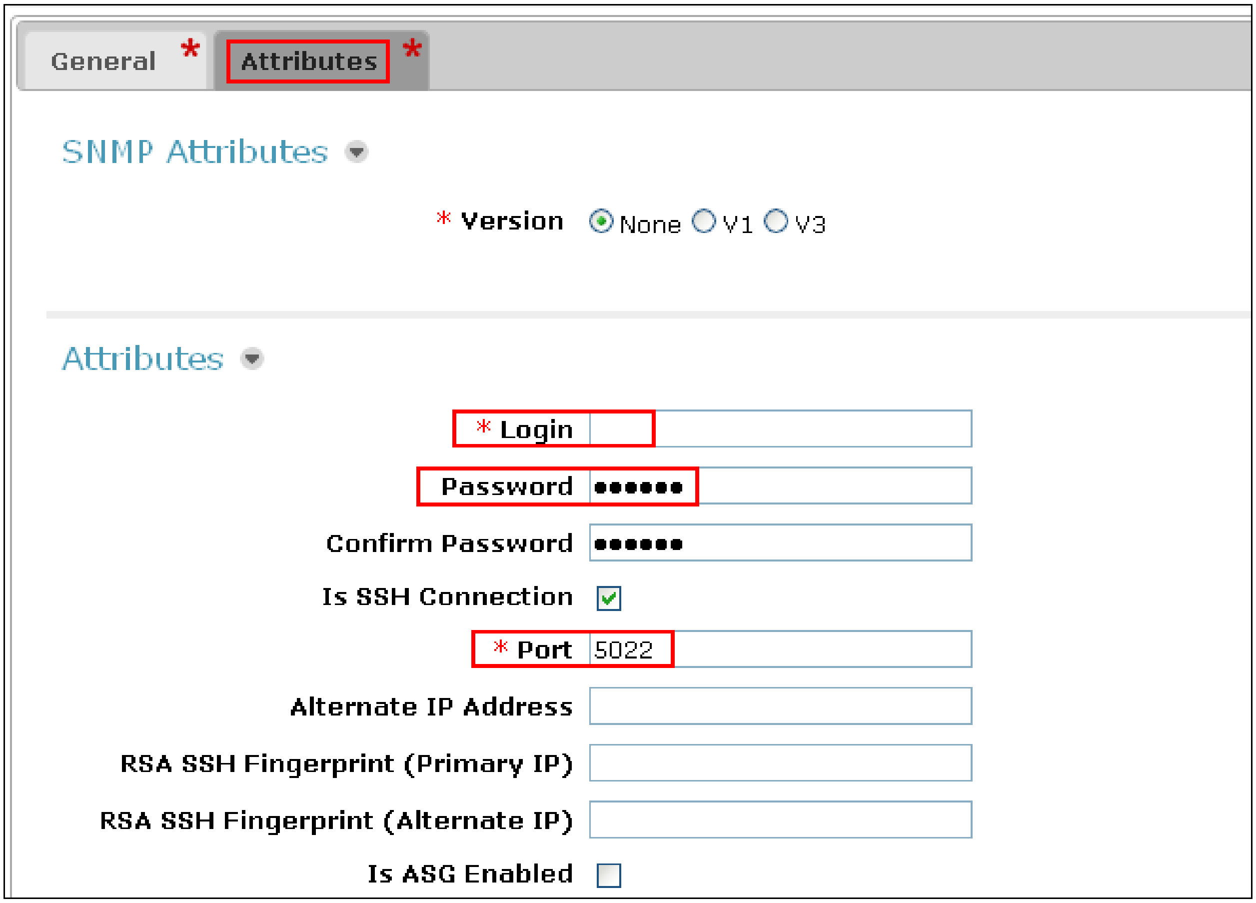
In order for Communication Manager to supply configuration and feature support to the Avaya one-X Mobile SIP for IOS as a Remote User when it registers to Session Manager, Communication Manager must be added as an application. Under the **Inventory** heading on the left hand side of the System Manager GUI access the **Manage Elements** sub heading. Under **Elements** select the **New** button.



The Manage Element **Name** was **MESCM**. The **Type** was set to **Communication Manager**. The **Node** IP Address was set to **192.168.1.82**.

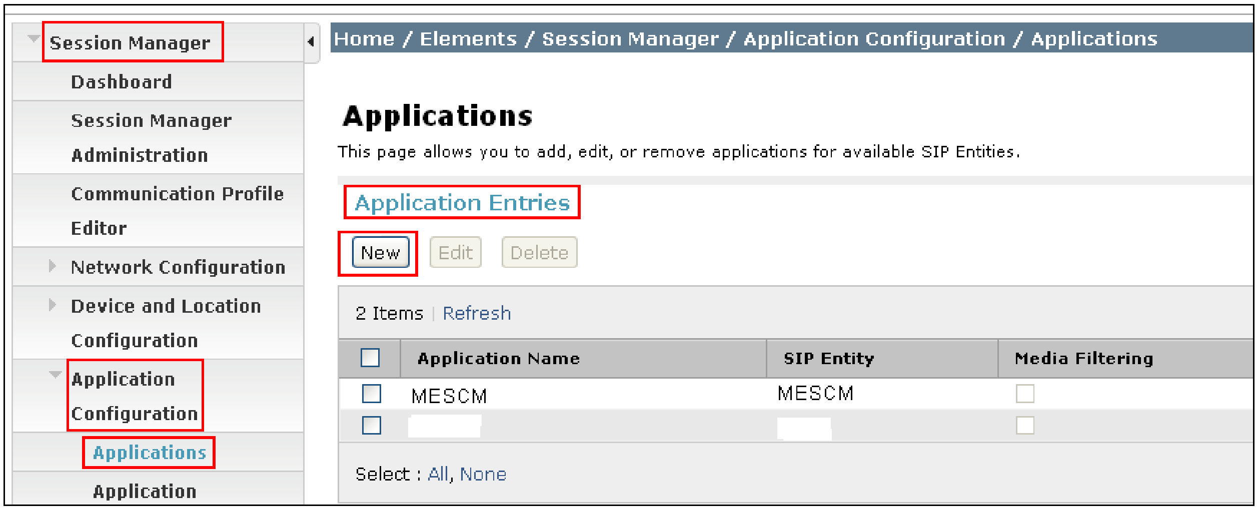


Access the **Attributes** section and set the **Login**. This was the login used to access the Communication Manager Evolution Server. The **Password** was set to the password used to access the Communication Manager Evolution Server. The **Port** was set to **5022.**

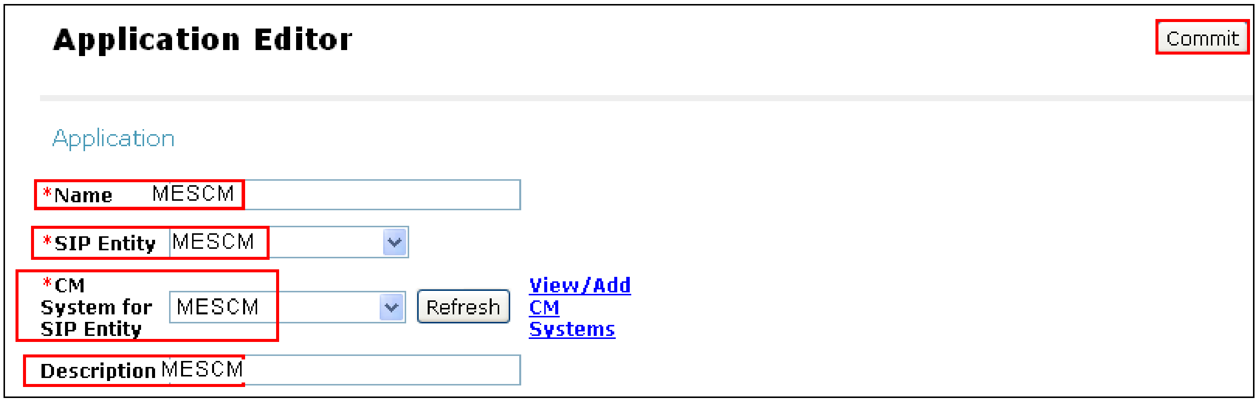


6.11. Administer Avaya Aura® Communication Manager Server Application

To configure the Communication Manager Evolution Server Application access **Session Manager→ Application Configuration → Applications.** Under **Application Entries,** select the **New** button.

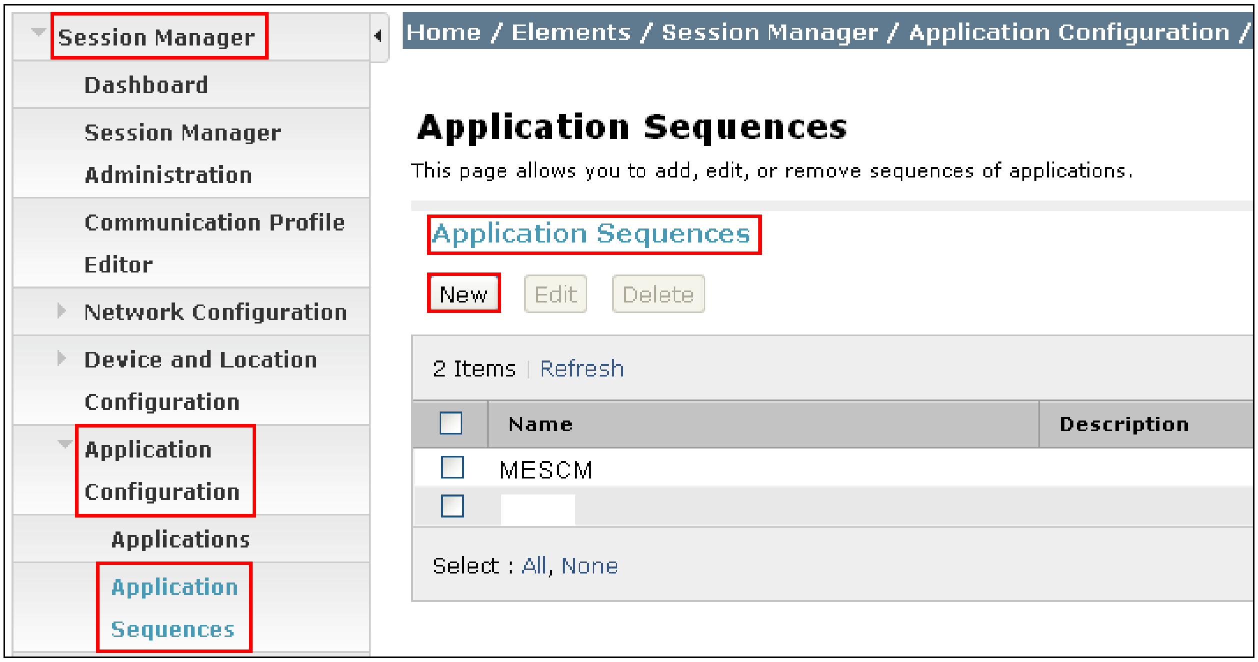


The **Name** of the Application was **MESCM**. The **SIP Entity** used was **MESCM**. The **CM System for SIP Entity** used was **MESCM**. The **Description** of the Application was **MESCM**.

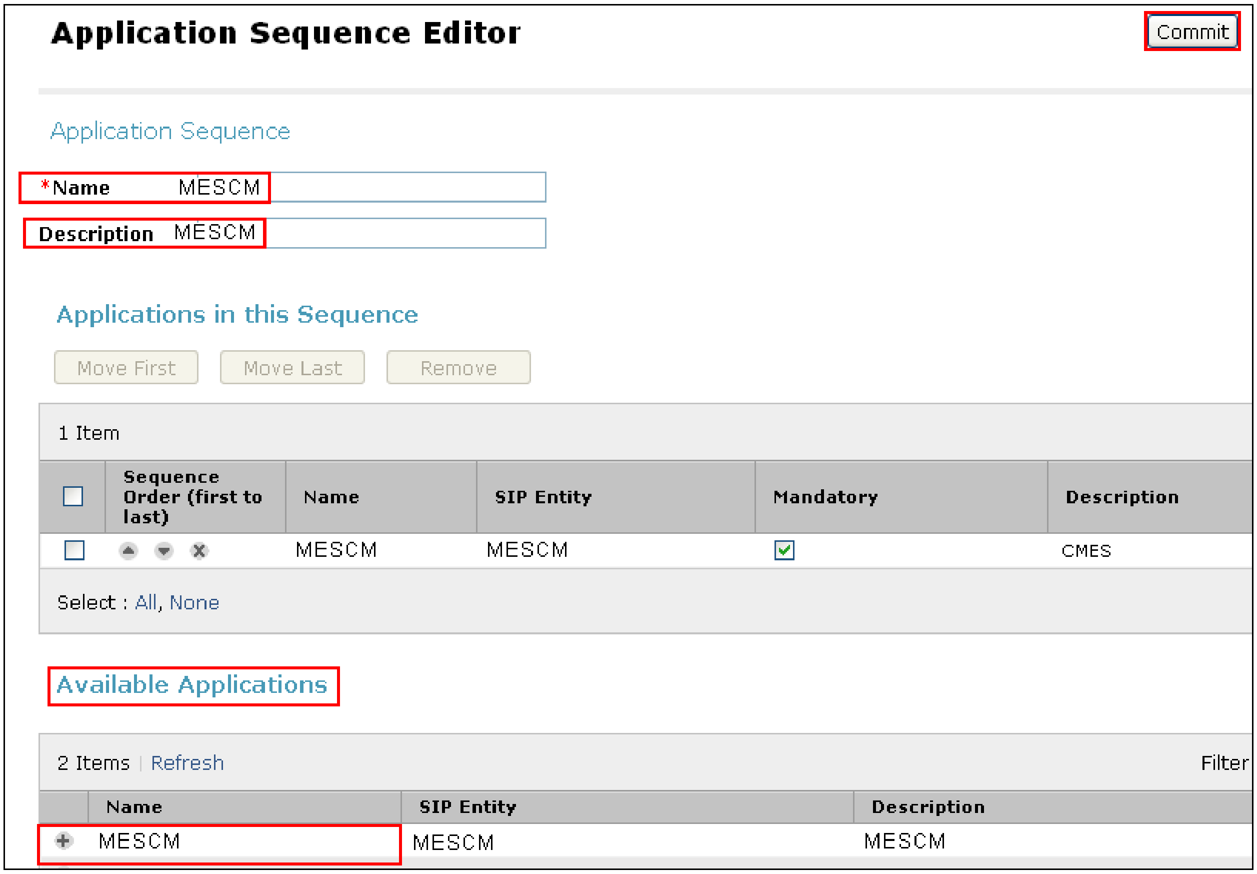


6.12. Administer Avaya Aura® Communication Manager Server Application Sequence

To configure the Communication Manager Evolution Server Application Sequence access the **Session Manager** heading on the left hand side System Manager GUI. Access the sub heading **Application Configuration → Application Sequences.**

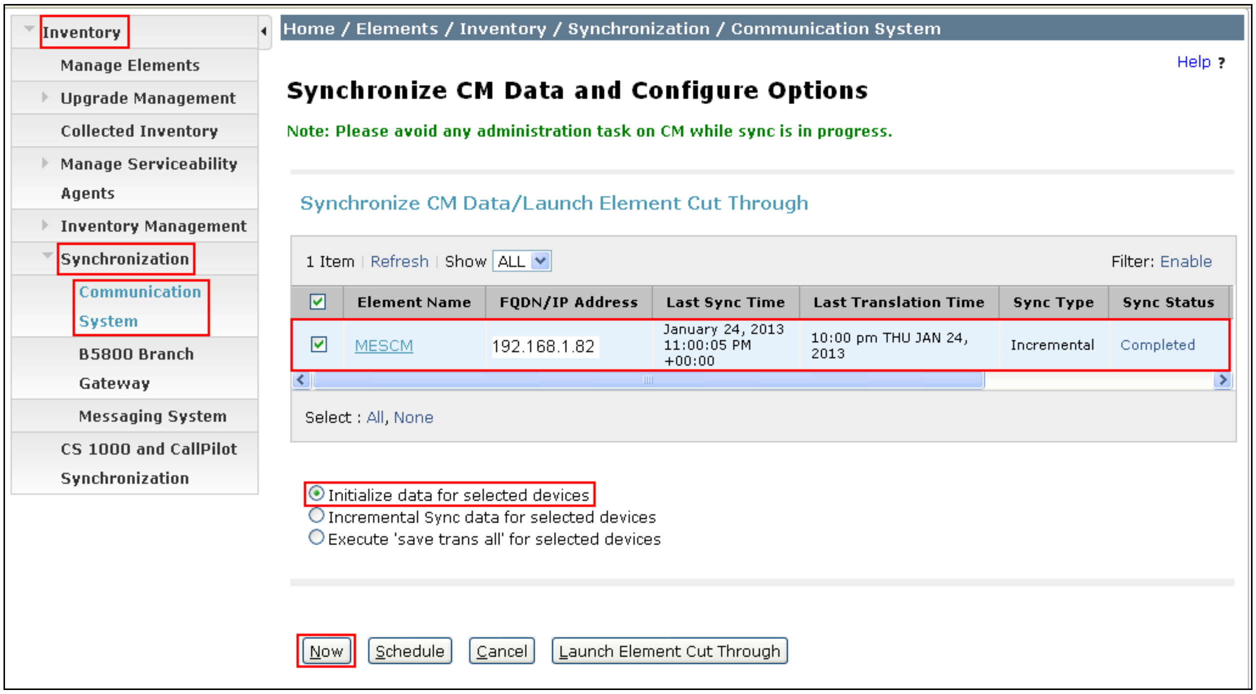


The Evolution Server Application Sequence **Name** was added as **MESCM.** The **Description** field was set to **MESCM**. Under the **Available Applications** field select the **plus** button to the left of the **MESCM** Name. This will then populate MESCM in the Application in this Sequence field. Select the **Commit** button to save the changes.



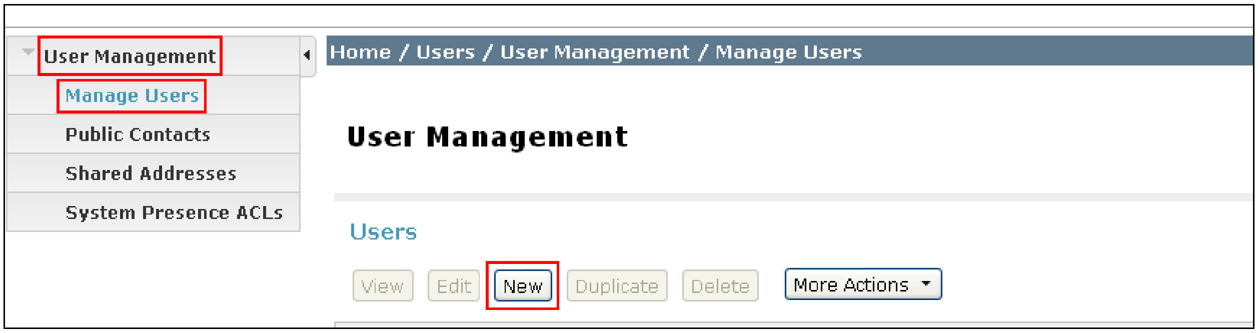
6.13. Synchronize Avaya Aura® Communication Manager Data

To synchronize the Communication Manager Data access **Inventory → Synchronization → Communication System** heading on the left hand side of the System Manager GUI. Access the sub heading **Communication System**. The following screenshot shows the MESCM, the Communication Manager Evolution Server synchronized to the Session Manager by highlighting the **Initialize data for the selected devices** option and selecting the **Now** key.

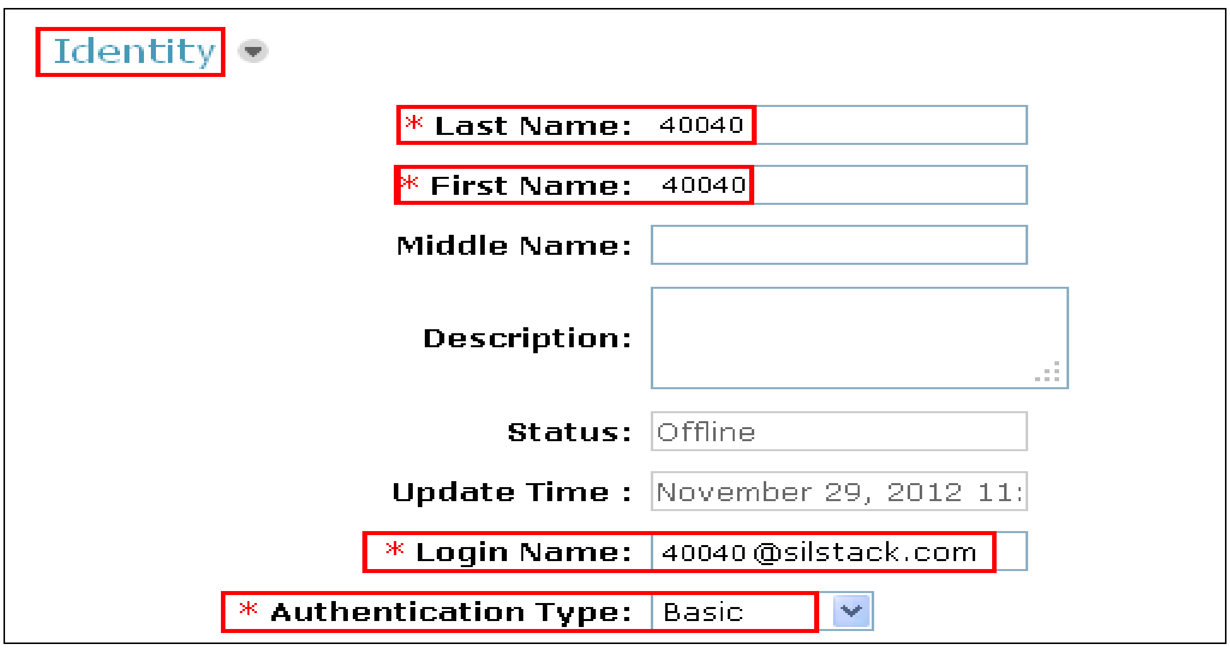


6.14. Administer SIP User

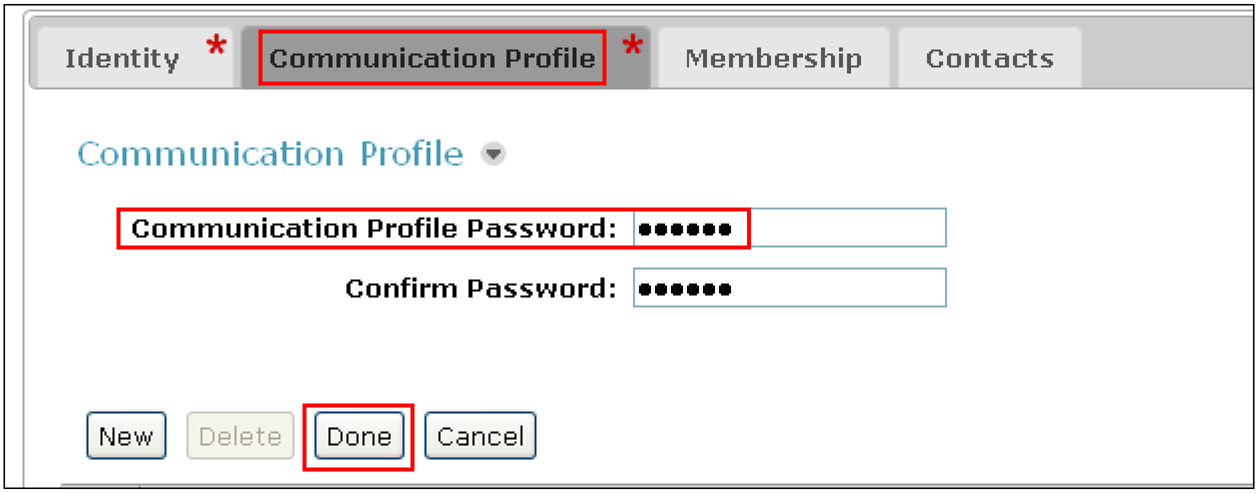
To add a SIP User to Session Manager, access the **User Management → Manage Users** heading on the left hand side of the System Manager GUI. Select the New button to add a new SIP User to Session Manager.



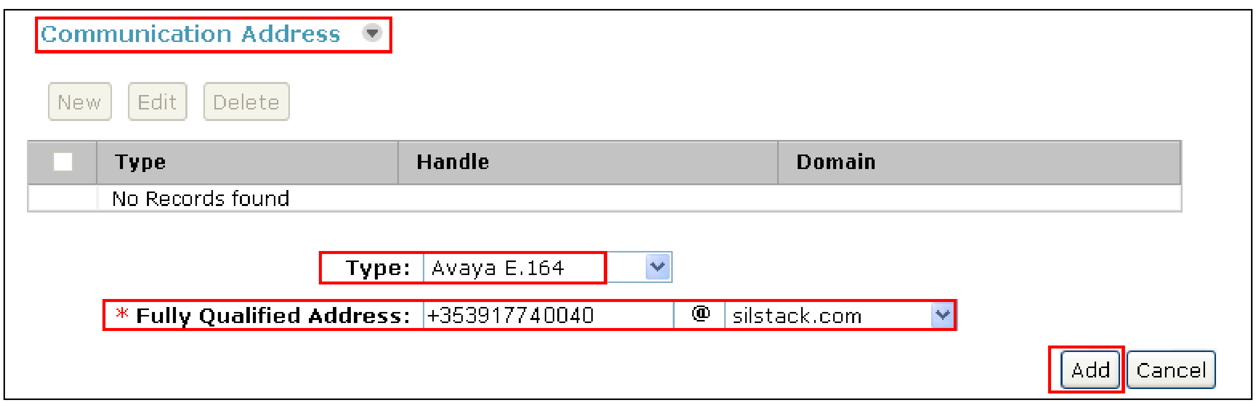
Select the **Identity** sub heading. The **Last Name** was set to **40040** and **First Name** was set to **40040.** The **Login Name** was set to **40040@silstack.com**. The **Authentication Type** was set to **Basic**.



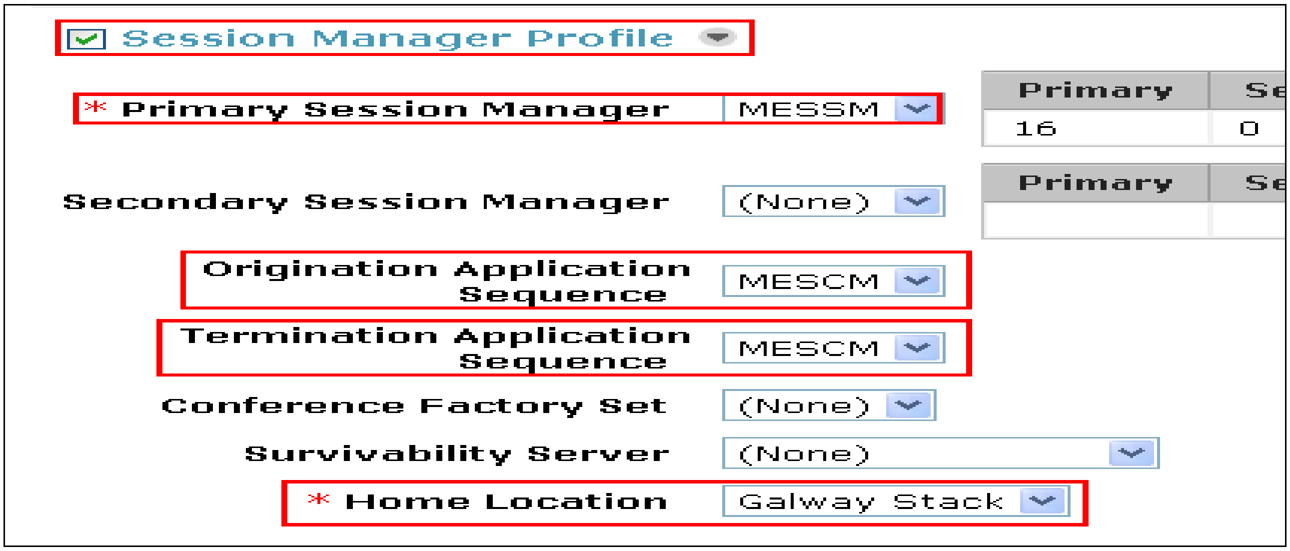
Next, click on the **Communication Profile** Tab. Select the **Communication Profile** sub heading. The **Communication Profile Password** was set. Select the **Done** button to save the changes.



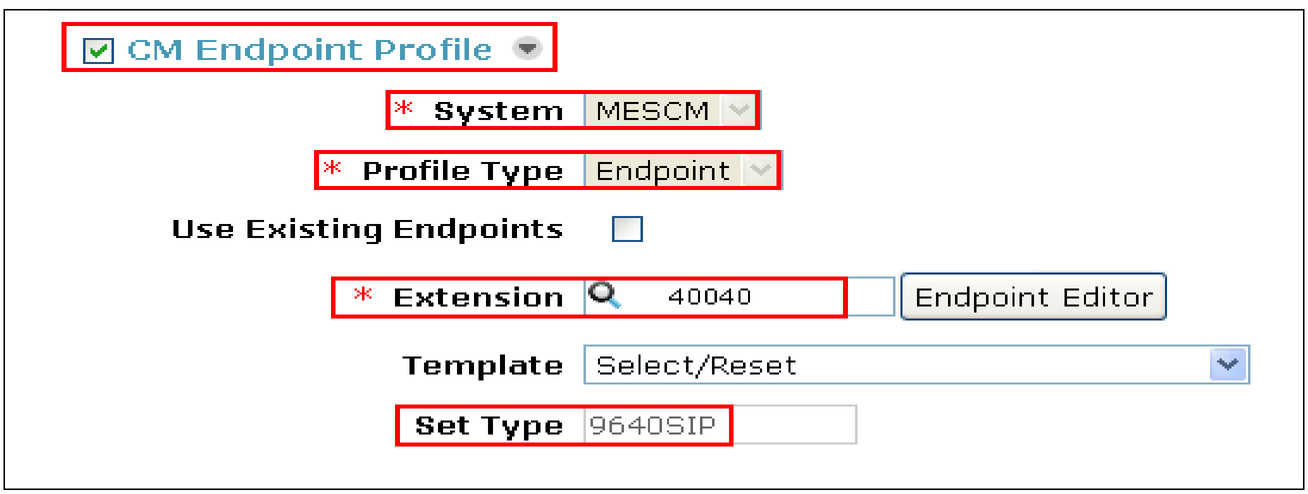
Select the **Communication Address** heading. The **Type** was set to **Avaya E.164**.The **Fully Qualified Address** was set to **+353917740040@silstack.com.** The **Add** button was pressed to save the changes.



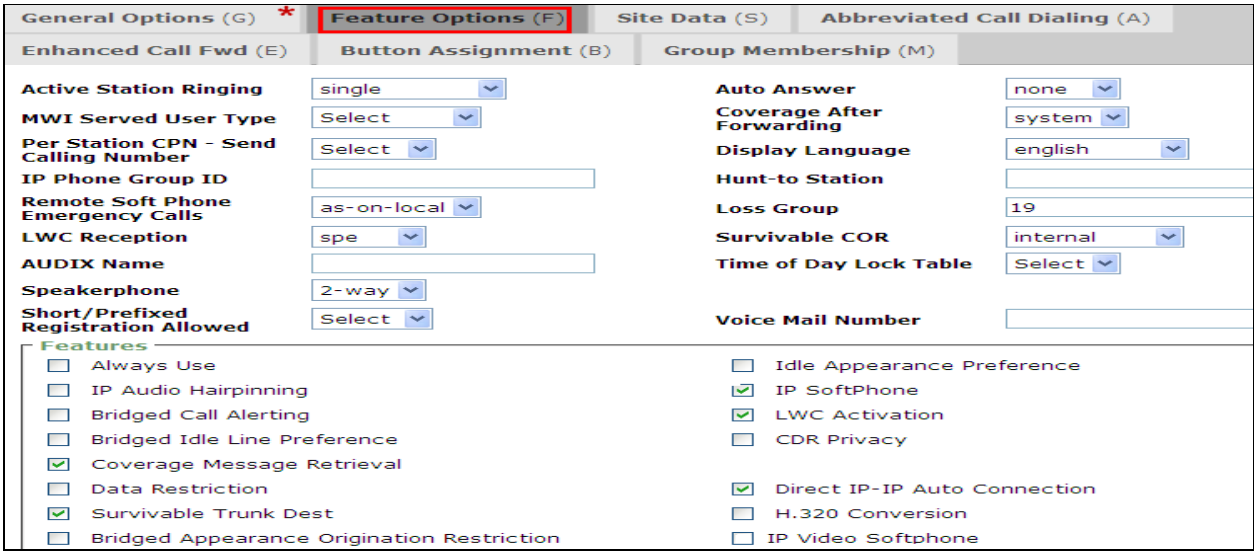
Select **Session Manager Profile** heading was selected. The **Primary Session Manager** was set to **MESSM**. This equates to the Session Manager SIP entity. The **Origination Application Sequence** was set to **MESCM**. The **Termination Application Sequence** was set to **MESCM.** The **Home Location** was set to **Galway Stack**.



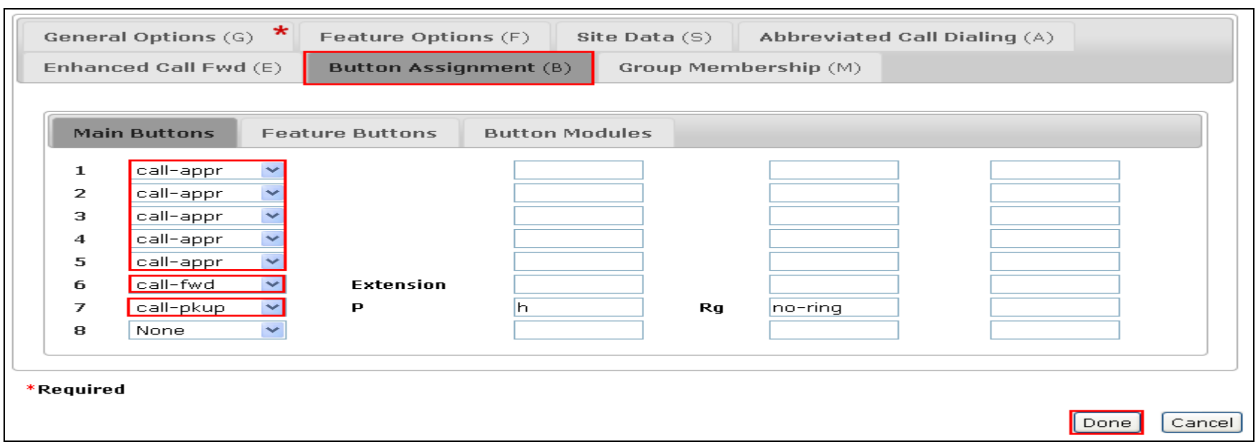
In order for the Station Profile template information to be pushed from the Session Manager down to the Communication Manager Evolution Server, **enable** the **CM Endpoint Profile** box. The **System** was set to **MESCM.** This is the Communication Manager Server Element Name. The **Extension** was set to **40040** and the **Template** and **Set Type** was set to **9640SIP.**



Click on **Endpoint Editor** and under **Feature Options** the settings were left as default**.**



Within **Button Assignments** a value of **5 call-appr** buttons were set. A **call-fwd** button and a **call-pkup** button were also assigned. The **Done** button was pressed



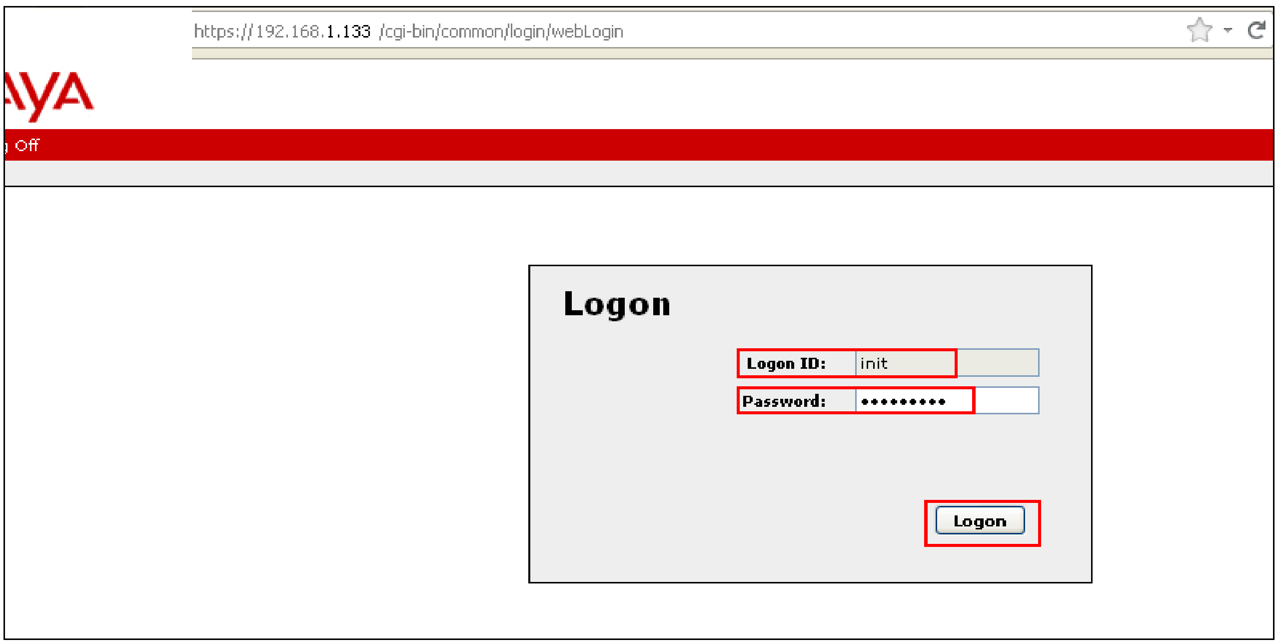
Press the **Commit** button to save the changes.

7. Administer Avaya Aura® Messaging Server

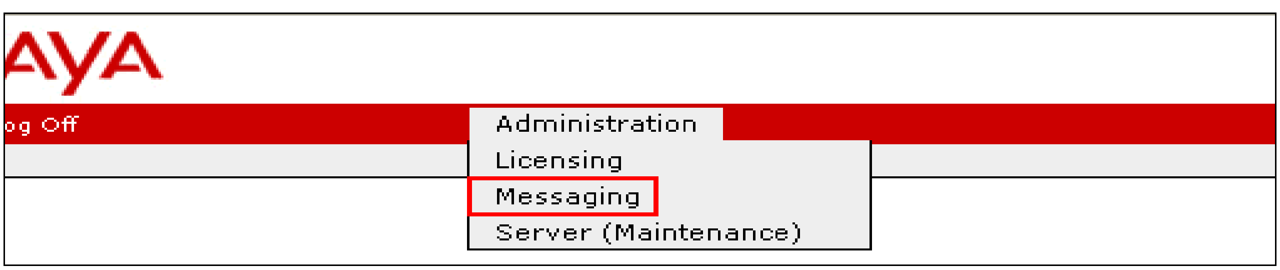
This section highlights the important commands for administering Avaya Aura Messaging to function correctly with SRTP and adding a subscriber for the Avaya one-X Mobile SIP for IOS as a Remote User to the Messaging Server.

7.1. Access Avaya Aura® Messaging

Access the Messaging web interface, by entering **http://<ip-addr>** as the URL in an Internet browser, where *<ip-addr>* is the IP address of the server running the Messaging graphical user interface. Log in with the appropriate **Login ID** and **Password** and press the **Logon** button to access the Messaging Server.



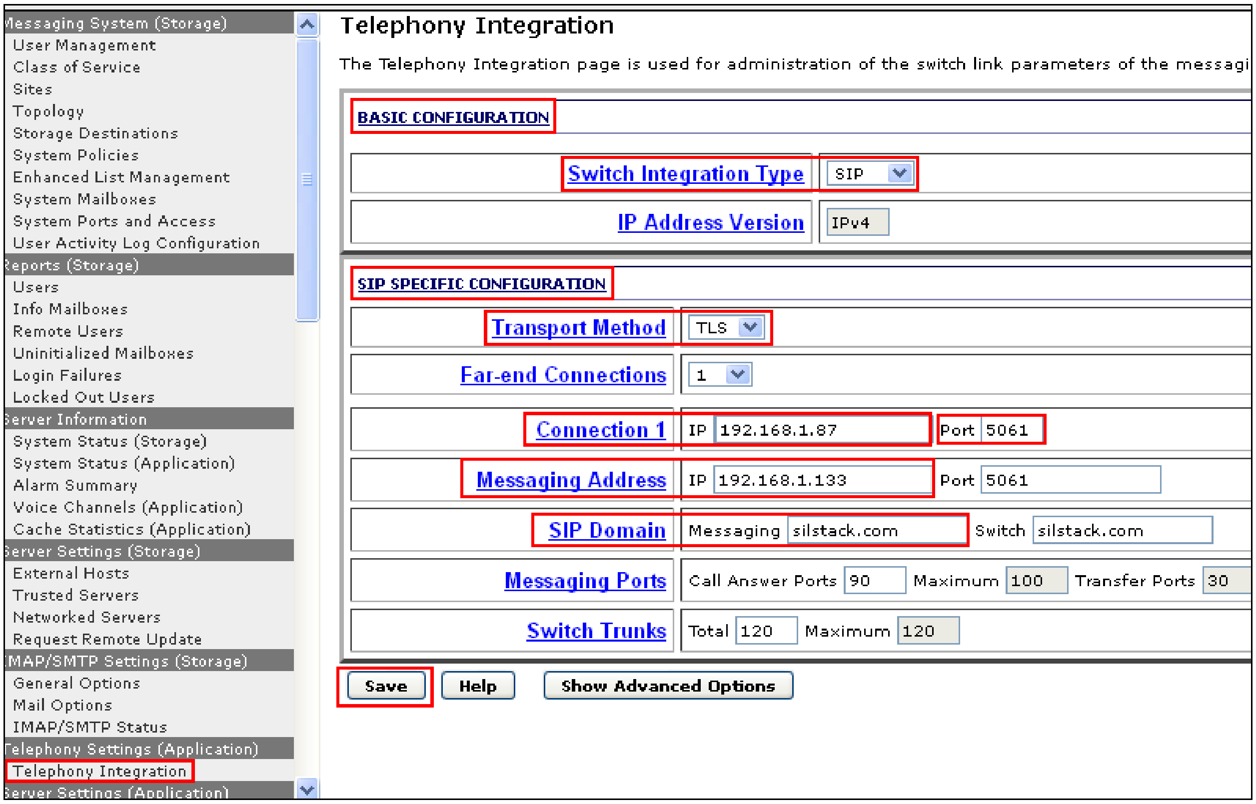
Under the Administration heading select **Messaging**.



7.2. Administer Telephony Integration with SRTP

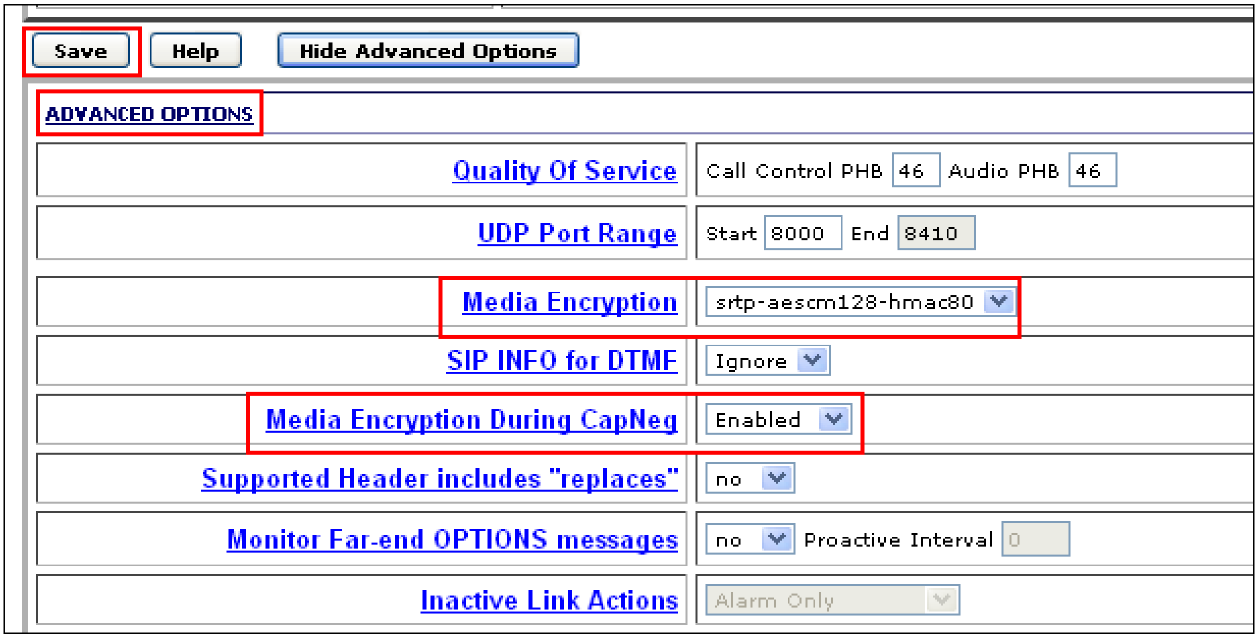
Select **Telephony Integration** under the Telephony Settings heading on the left hand side of the Messaging Graphical User Interface. Under **BASIC CONFIGURATION** the **Switch Integration Type** was set to **SIP**. Under **SIP SPECIFIC CONFIGURATION** the **Transport**

Method was set to **TLS**. The **Connection 1** setting was set to **192.168.1.87,** the IP Address of the Session Manager. The **Port** was set to **5061**. The **Messaging Address** was set to **192.168.1.133**. The **SIP Domain** was set to **silstack.com.** The **Save** button was selected to save the changes.



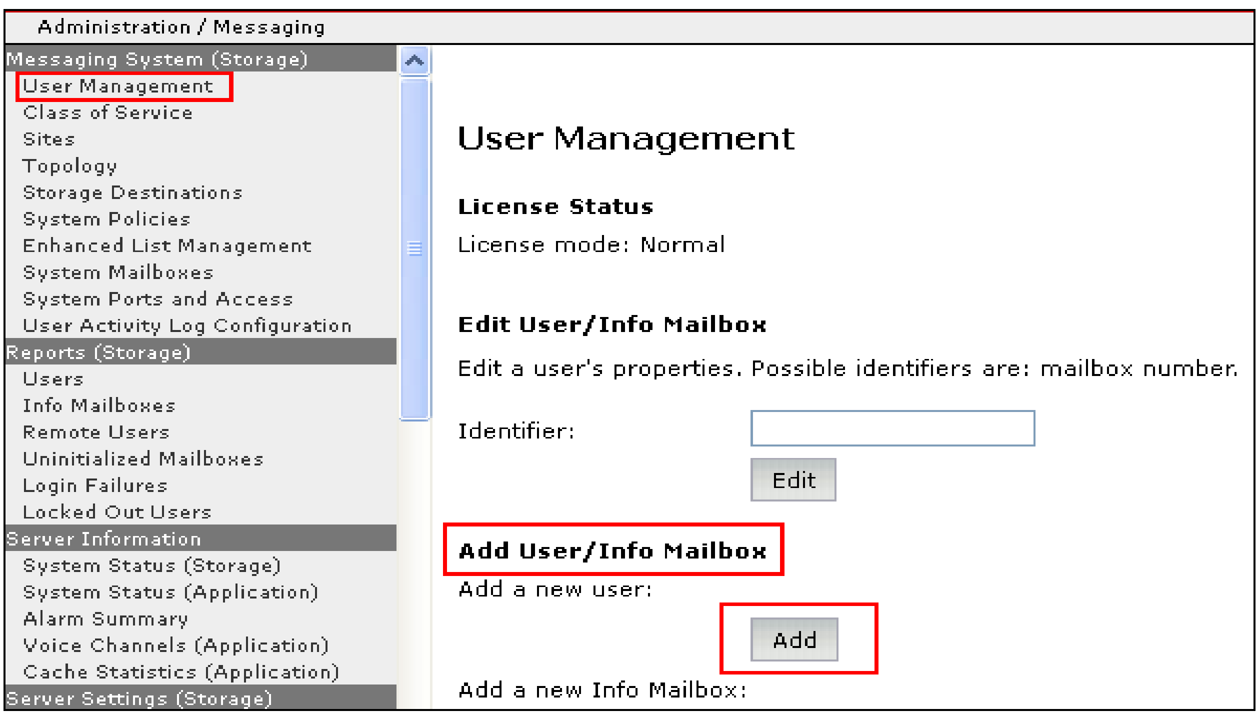
The **Show Advanced Options** setting was selected.

Under the **Advanced Options** settings the **Media Encryption** setting was set to **srtp-aescm128-hmac80.** The **Media Encryption During CapNeg** setting was **Enabled**. The **Save** button was selected to save these changes to the Messaging Server. Note that the Messaging Server needed to be stopped and started for these SRTP changes to take effect on the Messaging Server.

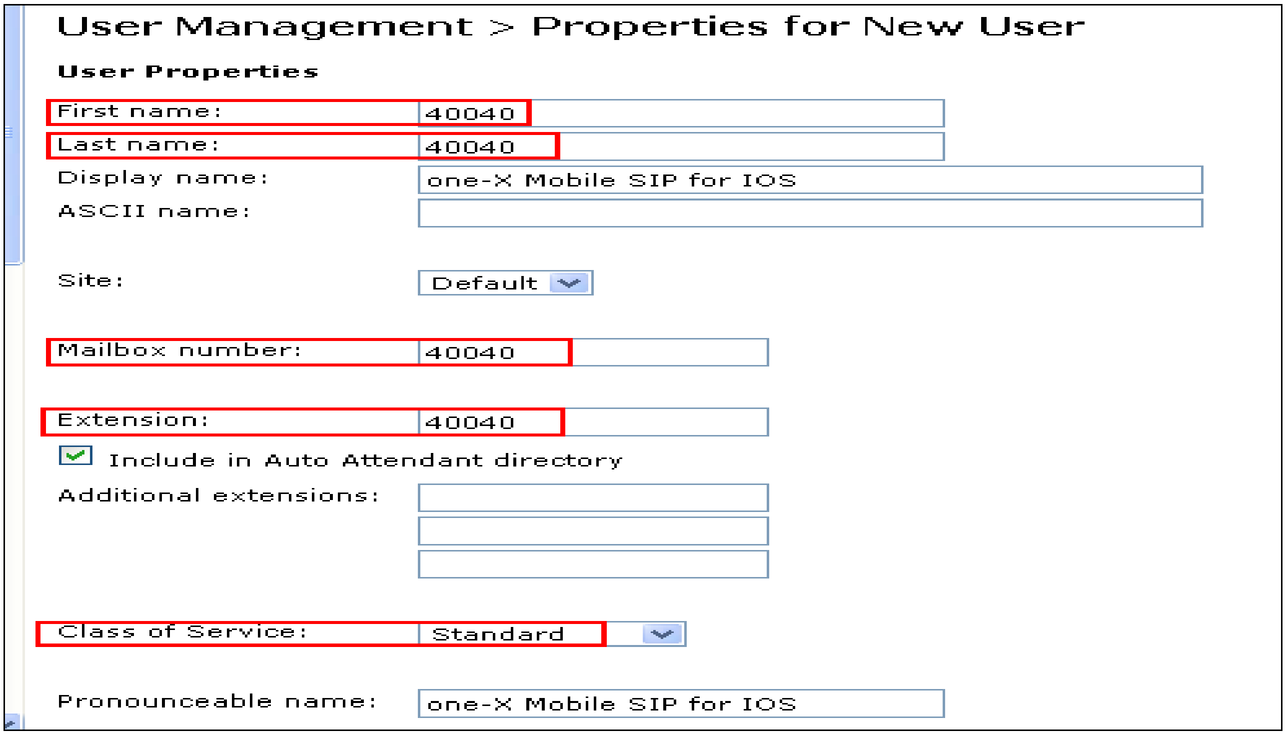


7.3. Administer Subscriber

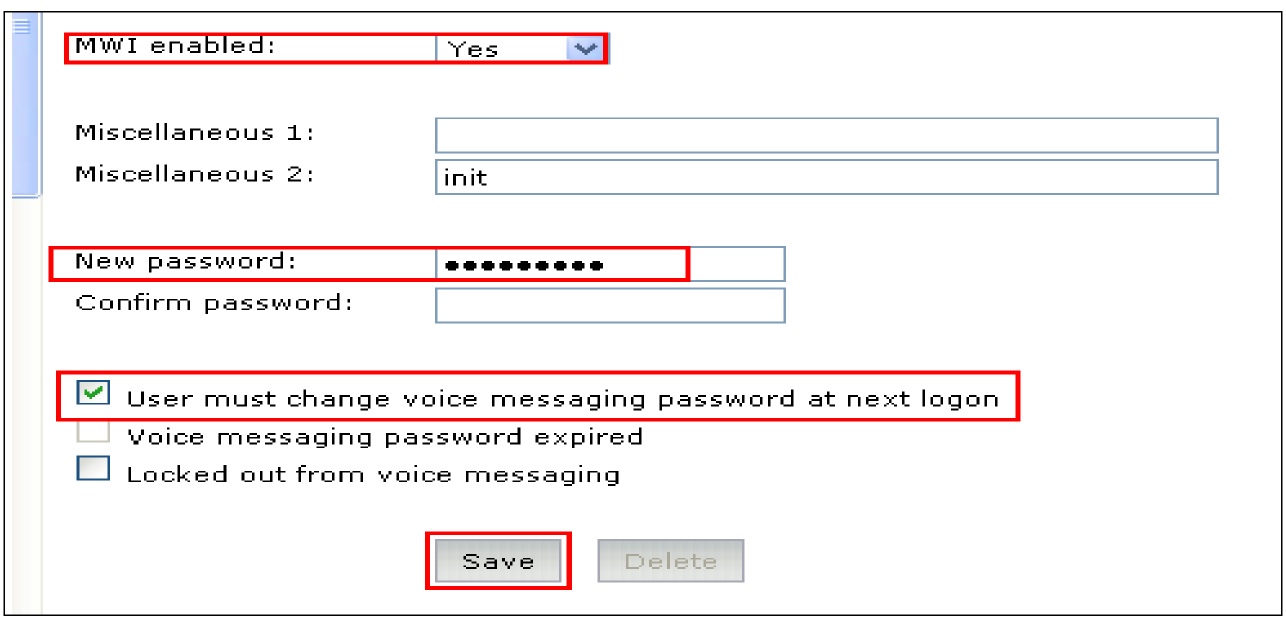
To add a subscriber to the Messaging Server select **User Management**. Under the **Add User/Info Mailbox** heading select the **Add** button.



The **First Name** was set to **40040**. The **Last Name** was set to **40040**. The **Mailbox Number** was set to **40040**. The **Extension** was set to **40040**. The **Class of Service** was set to **Standard.**



The **MWI enabled** setting was set to **Yes**. The **New password** was set to the password of the subscriber mailbox for the Avaya one-X Mobile SIP for IOS as a Remote User. The **User must change voice messaging password on next logon** setting was **enabled**. The **Save** button was pressed to save these changes.

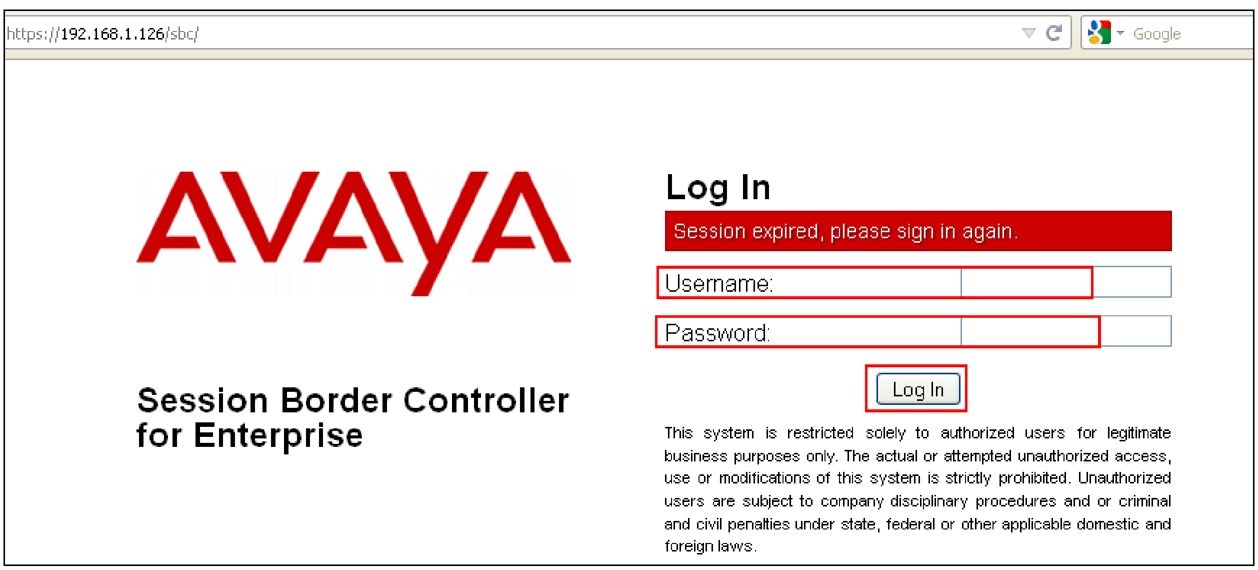


8. Administer Avaya Session Border Controller Advanced for Enterprise

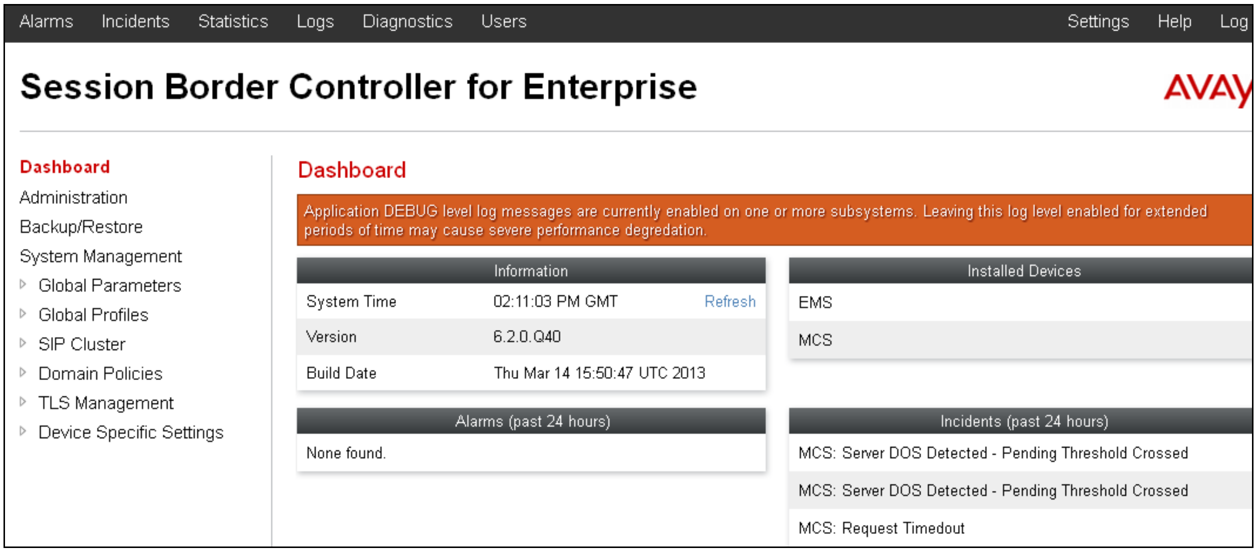
This section highlights the important steps for administering Avaya one-X Mobile SIP for IOS as a Remote User with SRTP to register to the Session Border Controller Server. It was decided that the Avaya one-X Mobile SIP for IOS as a Remote User would be administered with SRTP from the remote endpoint to the outside interface on the Session Border Controller and with SRTP from the inside interface to the Communication Manager Server. This section will document administering the media rule with SRTP to be used on the outside and inside interfaces on the Session Border Controller. It will also document the steps needed to administer signaling and media interfaces to the Session Manager and Remote User. It will highlight the steps required to configure Routing Profiles and End Point Policy Groups needed to be assigned to Subscriber and Server Flows within an End Point Flow. An asterisk **(*)** used in the option field for this section indicates that any or all choices for that parameter are acceptable. It is assumed that IP Addresses for all ports have been assigned during installation.

8.1. Access Avaya Session Border Controller Advanced for Enterprise

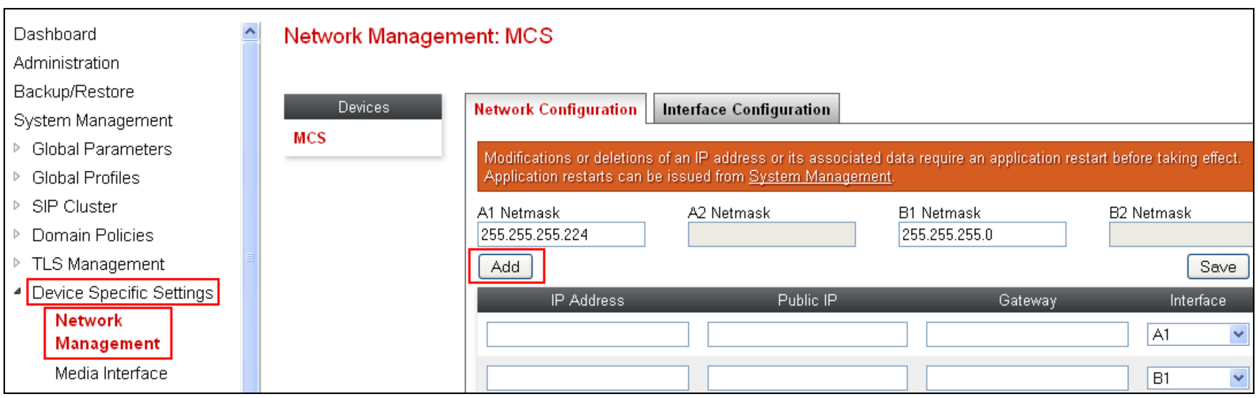
Access the Avaya Session Border Controller Advanced web interface, by entering **https://<ip-addr>** as the URL in an Internet browser, where *<ip-addr>* is the Management IP address of the server running the Avaya Avaya Session Border Controller Advanced graphical user interface. Log in with the appropriate **Username** and **Password** and press the **Log In** button to access the server.



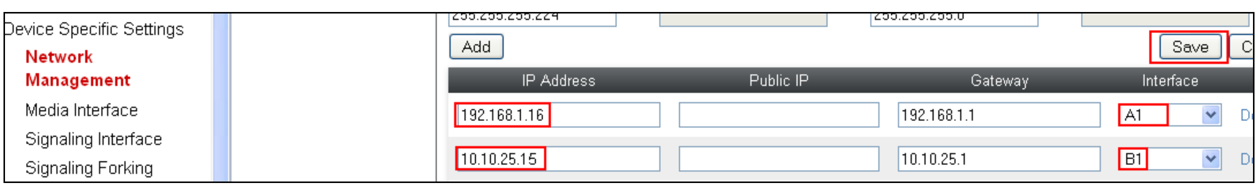
The following page is displayed.



Select **Device Specific Settings→Network Management→Add** to add the IP Address of the Session Border Controller Server interfaces.

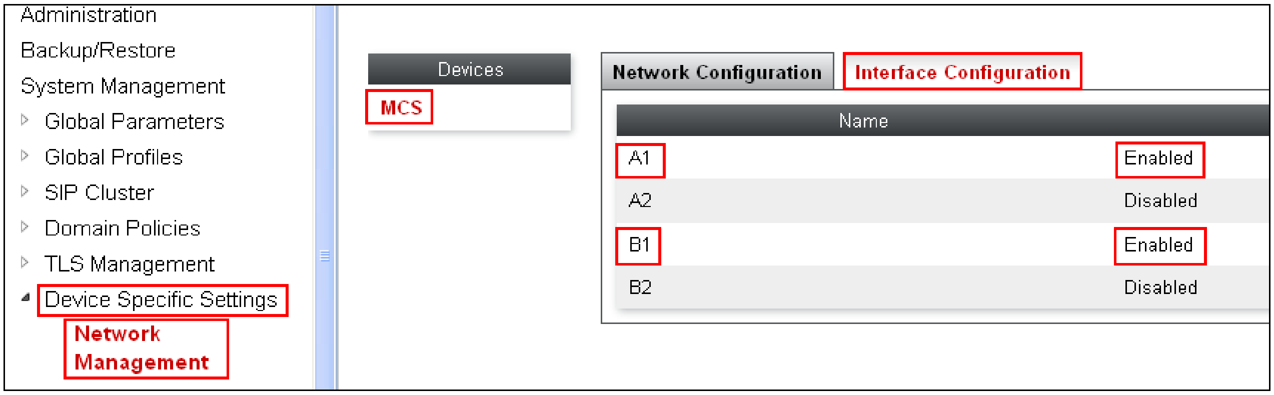


The IP Address of the **A1** inside interface was set to **192.168.1.16**. The IP Address of the **B1** outside interface was set to **10.10.25.15.**



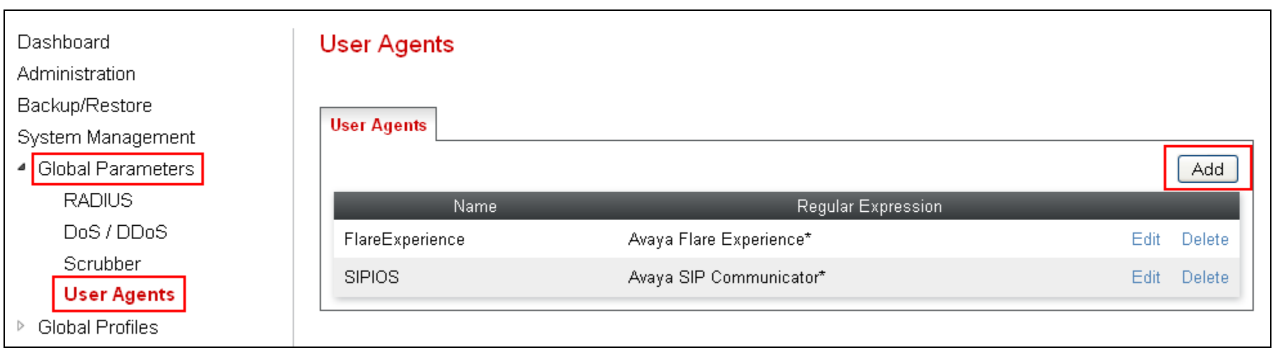
8.2. Enable Interfaces on the Avaya Session Border Controller Advanced for Enterprise

Select **Device Specific Settings→Network Management→Interface Configuration**. The **A1** internal interface was **Enabled** by selecting the **Toggle State** button. The **B1** external interface was also **Enabled** by selecting the **Toggle State** button**.**

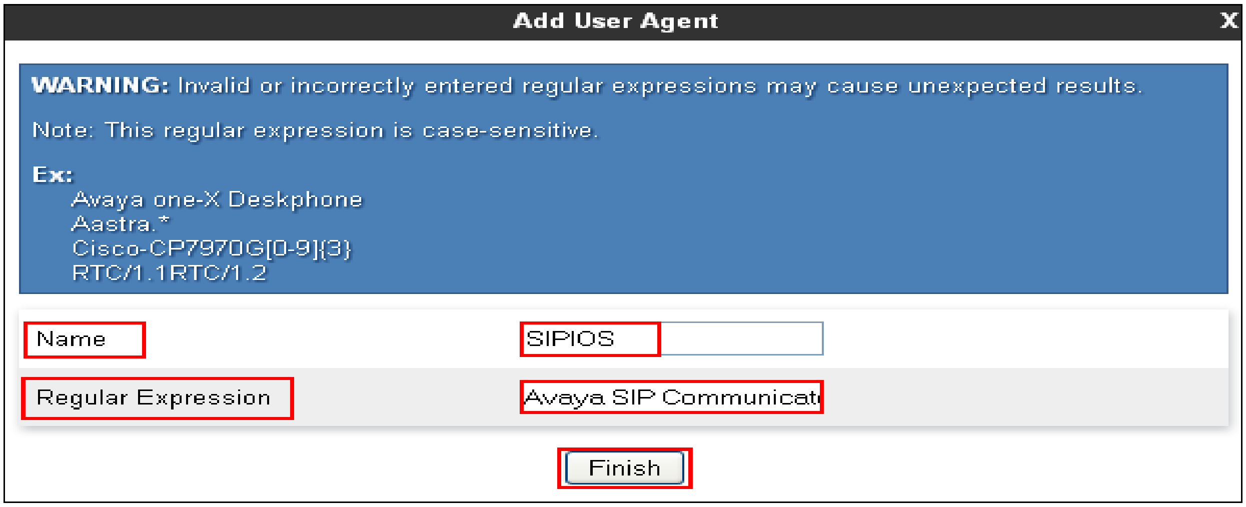


8.3. Administer User Agent

A User Agent was added for Avaya one-X Mobile SIP for IOS to allow the Avaya one-X Mobile SIP for IOS remote user access the network. To administer a User Agent for Avaya one-X Mobile SIP for IOS under **Global Parameters** select the **User Agents** heading. Select the **Add** button.

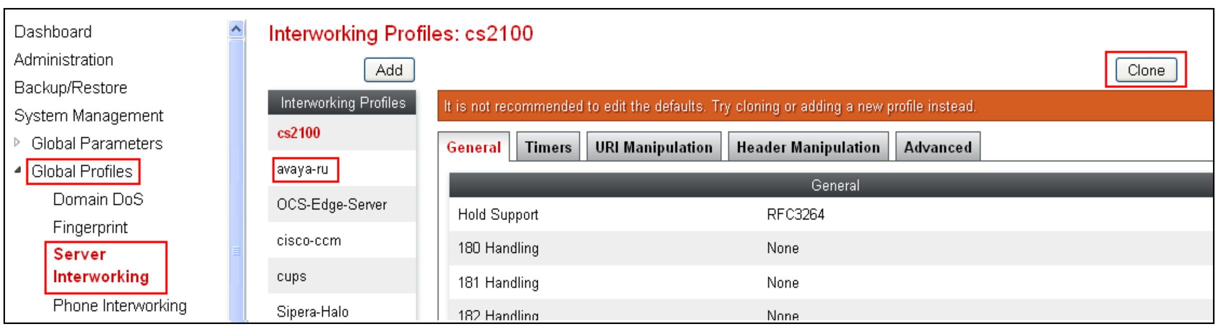


The **Name** was set to **SIPIOS** and the **Regular Expression** was set to **Avaya SIP Communicator***. The **Finish** button was selected to save the changes.

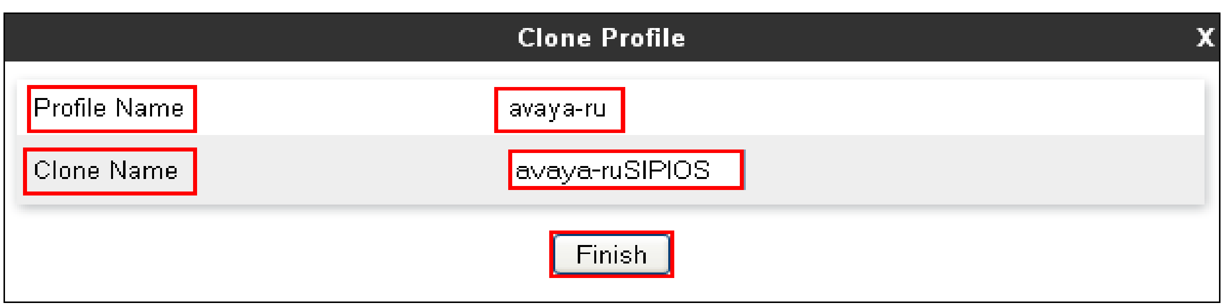


8.4. Administer Server Interworking

An Interworking Profile was used to manipulate headers for compatibility purposes. It was decided to use an existing Server Interworking Profile named **avaya-ru** and clone this Server Interworking Profile. To clone the Server Interworking select **Global Profiles→Server Interworking→avaya-ru→Clone**.



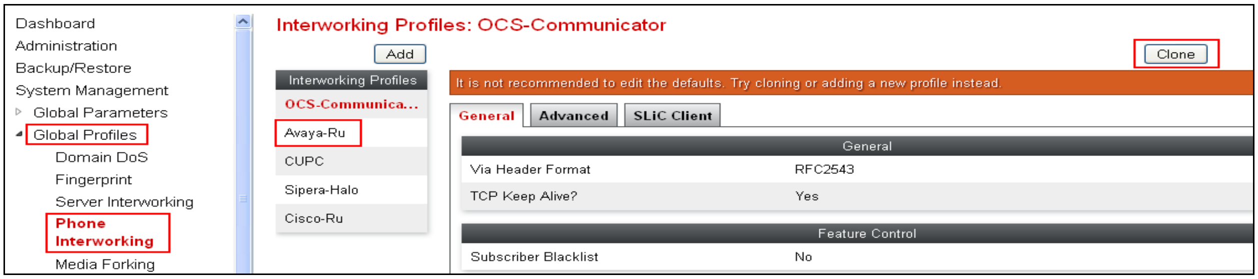
The **Profile Name** selected was **avaya-ru**. The **Clone Name** was set to **avaya-ruSIPIOS**. The **Finish** button was selected to save the changes.



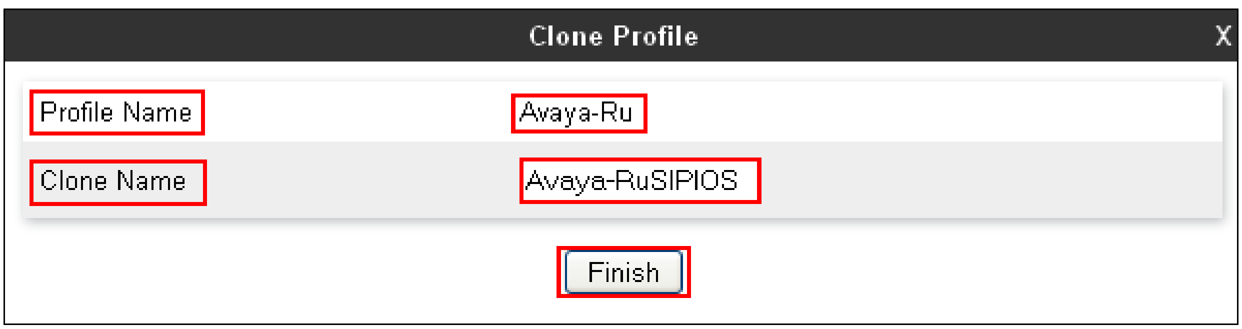
All values for the Server Interworking Profile were left as default.

8.5. Administer Phone Interworking

A Phone Interworking allows the user to edit specific SIP signaling message parameters to allow interoperability between endpoints and SIP implementations. It was decided to use an existing Phone Interworking Profile named **Avaya-Ru** and clone this Phone Interworking Profile. To clone the Phone Interworking select **Global Profiles→Phone Interworking→Avaya-Ru→Clone**.



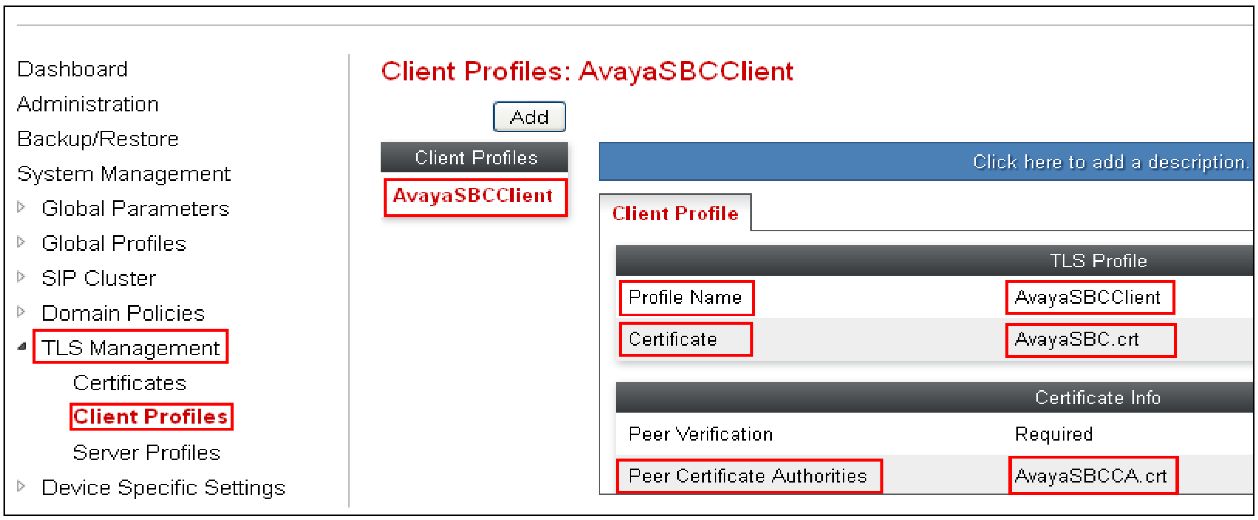
The **Profile Name** selected was **Avaya-Ru**. The **Clone Name** was set to **Avaya-RuSIPIOS**. The **Finish** button was selected to save the changes.



The settings for the Phone Interworking Profile were left as default.

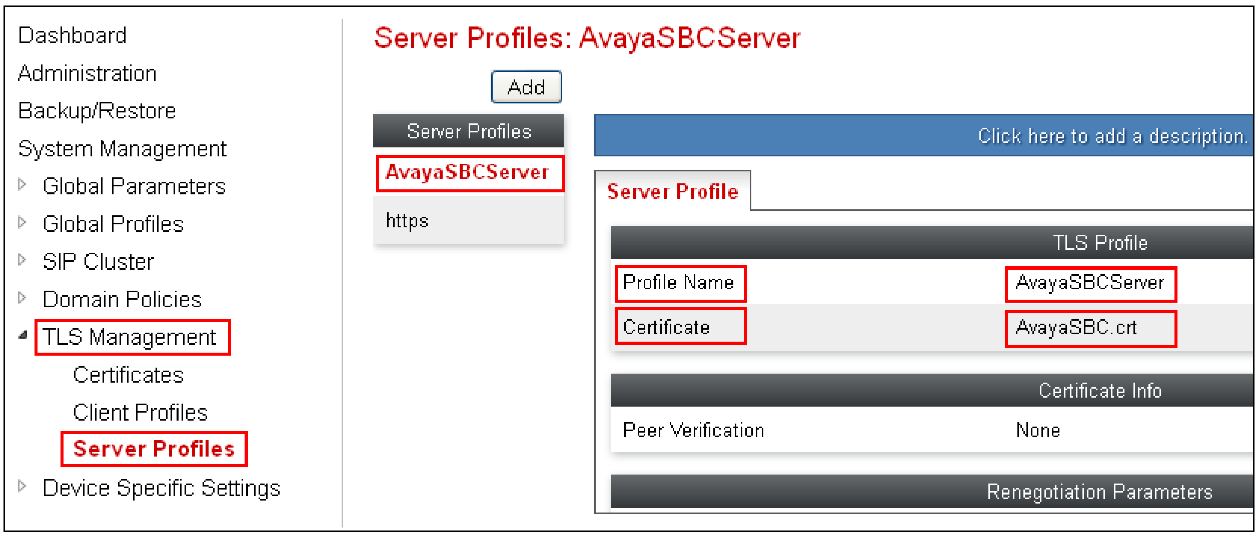
8.6. Verify TLS Client Profile

A Client Profile is needed to allow the Avaya one-X Mobile SIP for IOS as a Remote User to participate in a secure TLS session. The Session Border Controller has a pre installed Avaya client profile as part of the Session Border Controller software named AvayaSBCClient. It was decided to use this pre installed Avaya client profile for configuration purposes. Select **TLS Management→Client Profiles→AvayaSBCClient.** The **Profile Name** was **AvayaSBCClient**. The AvayaSBCClient profile contained the **Certificate** named **AvayaSBC.crt.** The AvayaSBCClient profile also contained the **Peer Certificate Autorities** root CA certificate named **AvayaSBCCA.crt.** These certificates are all Avaya signed certificates and trusted by other Avaya Servers.



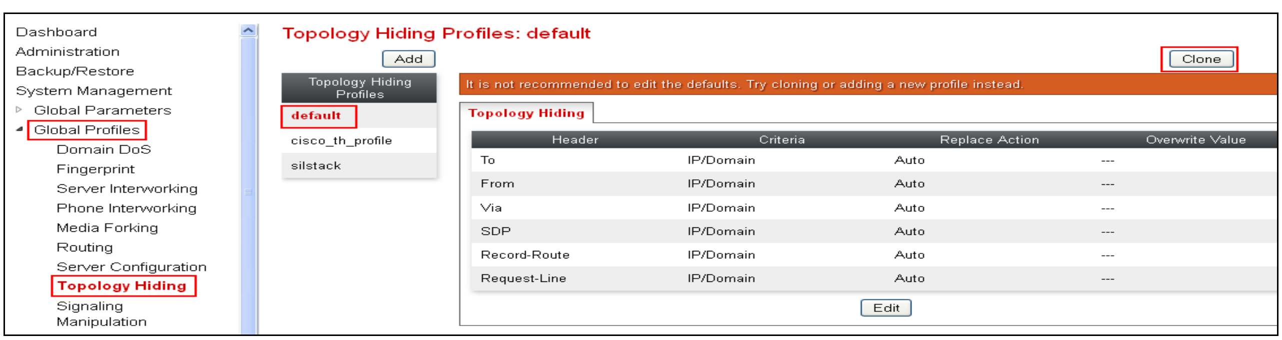
8.7. Verify TLS Server Profile

To allow the Avaya one-X Mobile SIP for IOS as a Remote User to participate in a secure TLS session a TLS Server Profile was also used. The Session Border Controller has a pre installed Avaya server profile as part of the Session Border Controller software. It was decided to use this pre installed Avaya server profile for configuration purposes. Select **TLS Management→Server Profiles→AvayaSBCServer.** The **Profile Name** was **AvayaSBCServer**. The AvayaSBCServer profile contained the **Certificate** named **AvayaSBC.crt.**

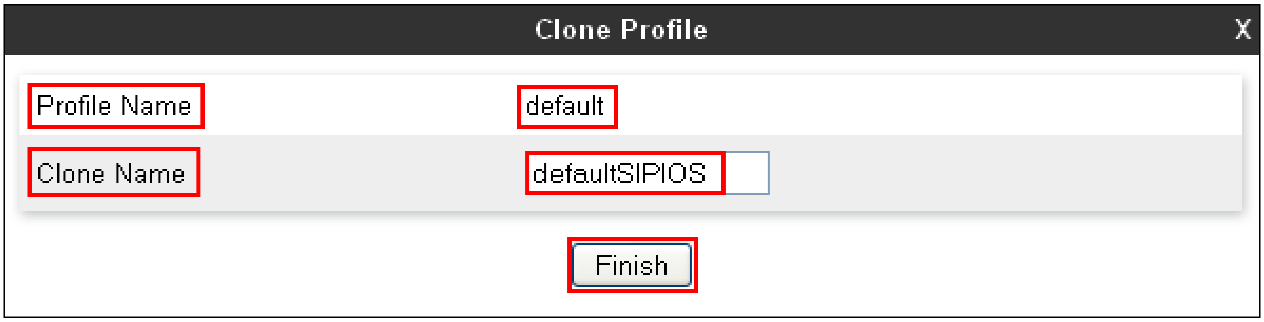


8.8. Administer Topology Hiding for Subscriber and Server Flow

Topology Hiding is a UC-Sec security feature which allows the user to change certain key SIP messages parameters to hide how the enterprise network map appears to the outside world. The Topology Hiding created will be applied to the Subscriber and Server flow. It was decided to use an existing topology hiding named **default** and clone this Topology hiding Profile. To clone the Topology Hiding Profile select **Global Profiles→Topology Hiding→default→Clone.**

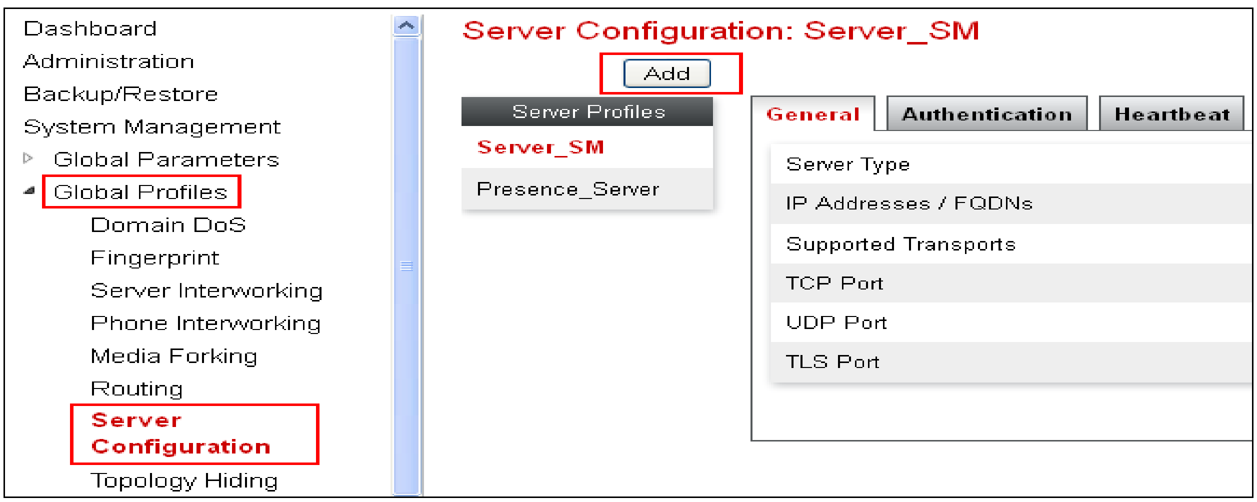


The **Profile Name** was set to **default.** The **Clone Name** was set to **defaultSIPIOS.** The **Finish** button was selected to save the changes.

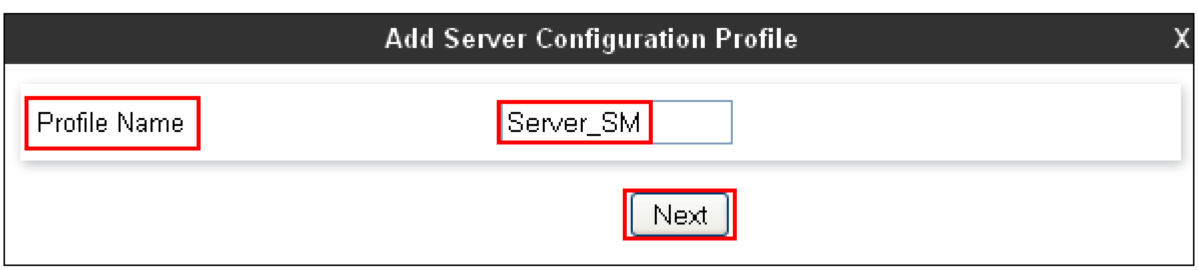


8.9. Administer Session Manager Server Configuration

This section describes creating a Call Server Profile for the Avaya Session Manager Server on the Avaya Session Border Controller. Select **Global Profiles→Server Configuration→Add**.

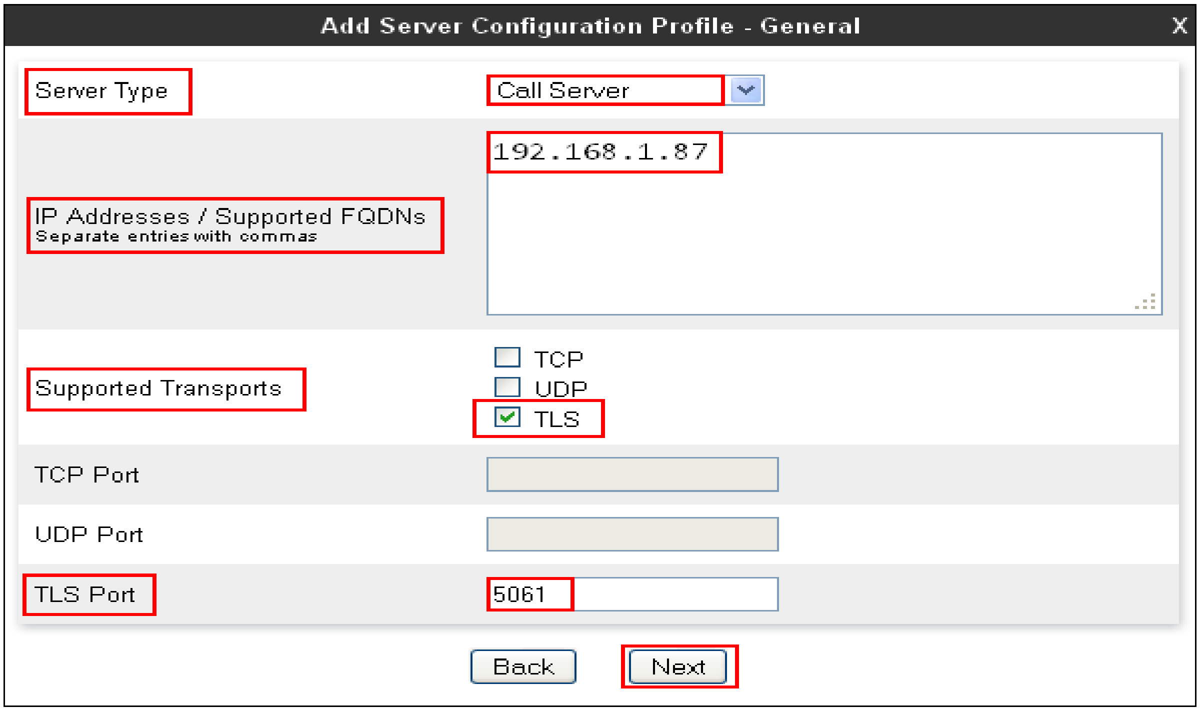


The **Profile Name** was set to **Server_SM**. The **Next** button was selected to continue to the next page.

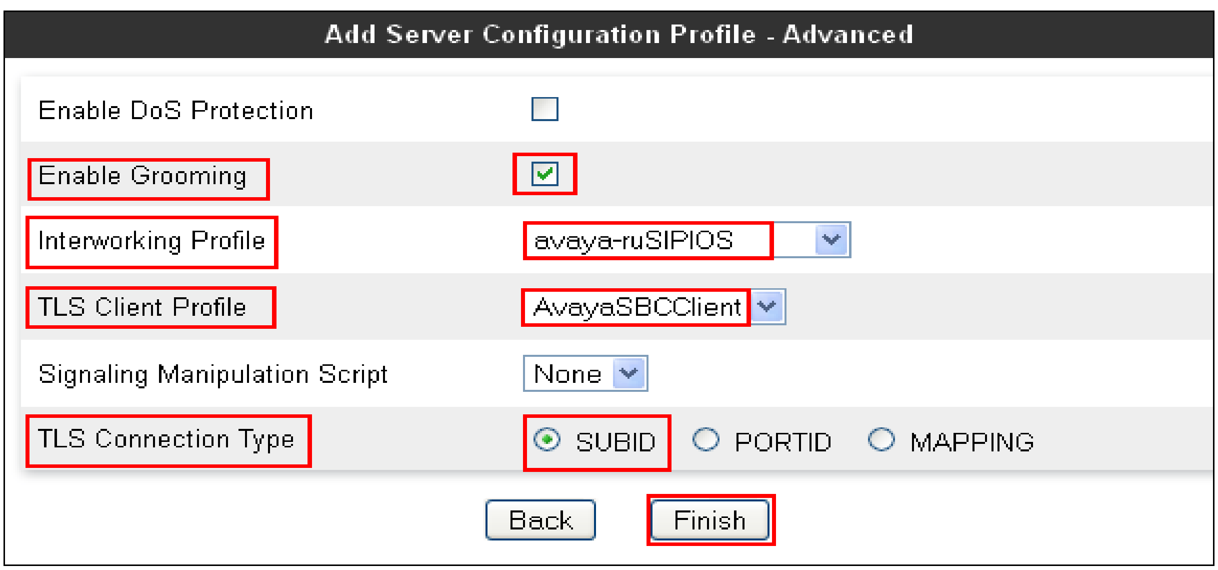


The Server Configuration Server_SM was administered in this section. The **Server Type** was set to **Call Server**. The **IP address** was set to **192.168.1.87**. This was the Signaling Interface of the

Session Manager. The **Supported Transports** was set to **TLS**. The **TLS Port** was set to **5061**. The **Next** button was selected to continue to the next page.

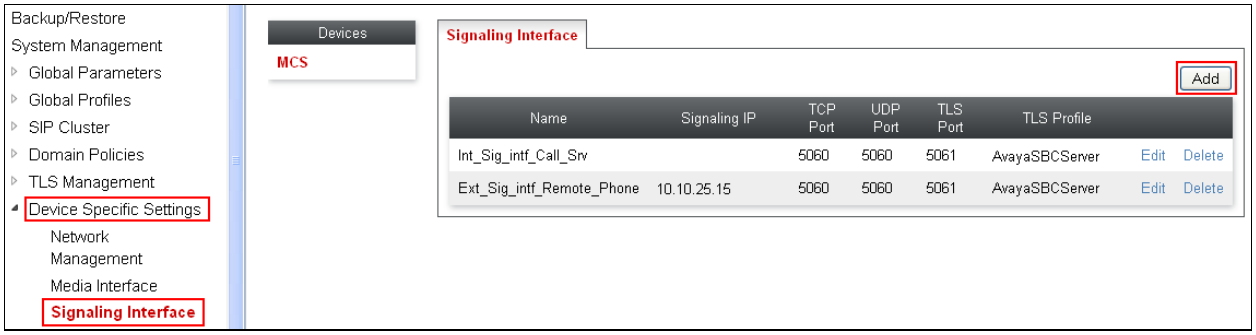


The **Enable Grooming** setting was also **Enabled.** The **Interworking Profile** value was set to **avaya-ruSIPIOS**. The **TLS Client Profile** was set to **AvayaSBCClient**. The **TLS Connection Type** was set to **SUBID**. The **Finish** button was selected.

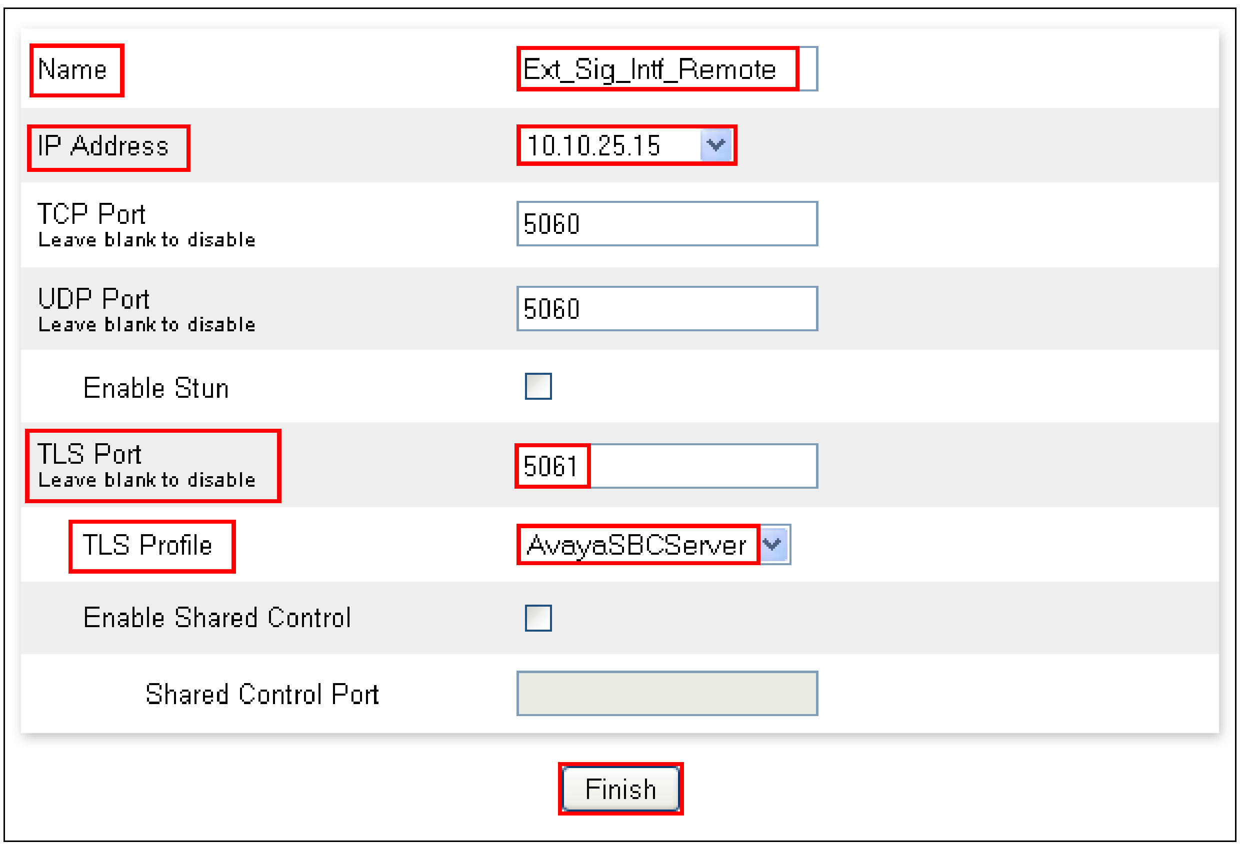


8.10. Administer External Signaling Interface Toward Remote User

The section explains administering a signaling interface to the Avaya one-X Mobile SIP for IOS as a Remote User endpoint. Select **Device Specific Settings→Signaling Interface→Add**.

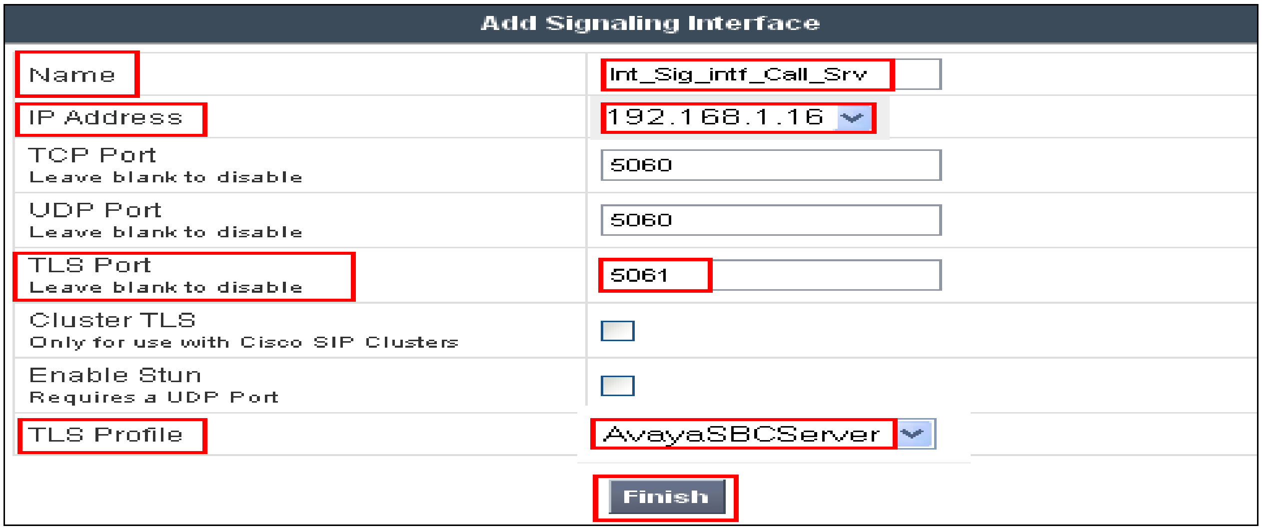


The **Name** was set **to Ext_Sig_intf_Remote _Phone**. The **IP Address** was set to **10.10.25.15.** This was the IP Address of the B1 external interface of the Avaya Avaya Session Border Controller. The **TLS Port** was set to **5061**. The **TLS Profile** was set to **AvayaSBCServer**. The **Finish** button was pressed to save the changes.



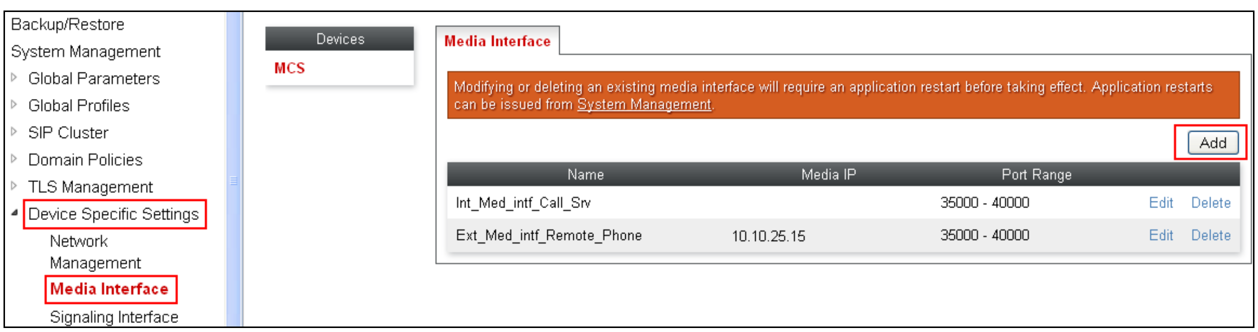
8.11. Administer Internal Signaling Interface toward Session Manager

This section explains administering a signaling interface to the Session Manager Server. Select **Device Specific Settings→Signaling Interface→Add.** The **Name** was set **to Int_Sig_intf_Call _Srv**. The **IP Address** was set to **192.168.1.16.** This was the IP Address of the A1 internal interface of the Avaya Session Border Controller The **TLS Port** was set to **5061**. The **TLS Profile** was set to **AvayaSBCServer**. The **Finish** button was pressed to save the changes.

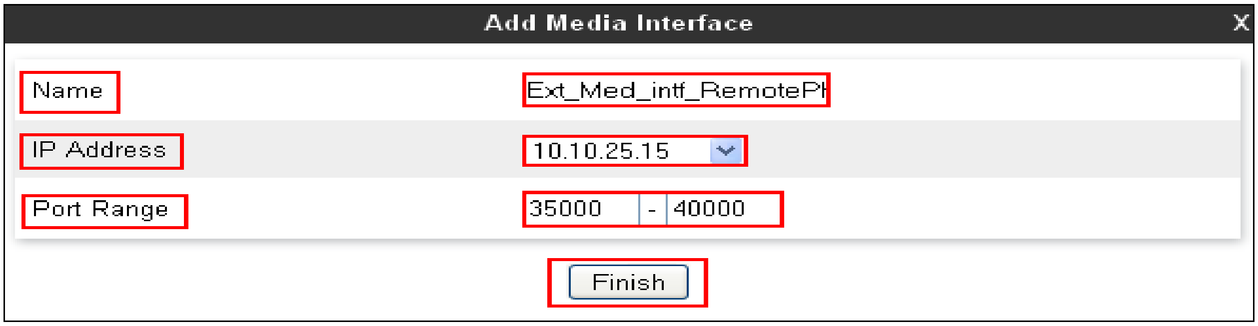


8.12. Administer External Media Interface Toward Remote User

The section explains administering a media interface to the Avaya one-X Mobile SIP for IOS as a Remote User. Select **Device Specific Settings→Media Interface→Add**.

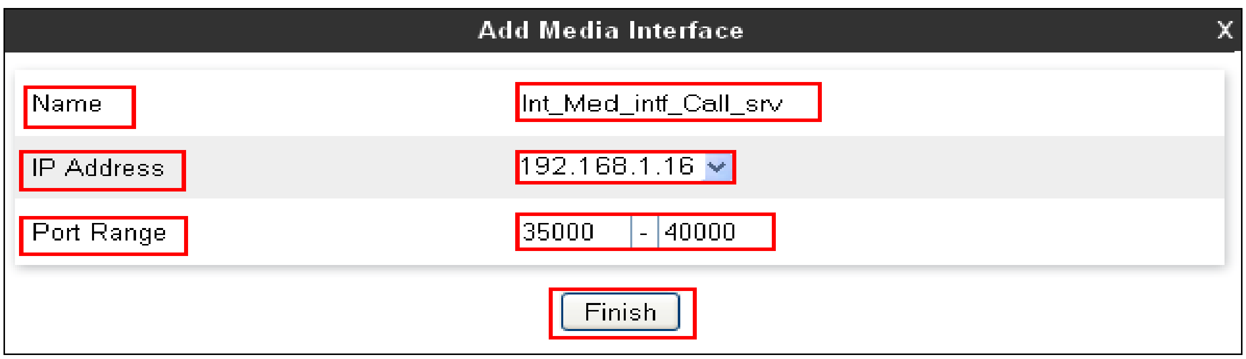


The **Name** was set to **Ext_Med_intr_Remote_Phone**. The **IP Address** was set to **10.10.25.15**. The **Port Range** was set to **35000 – 40000**. The **Finish** button was selected to save the changes.



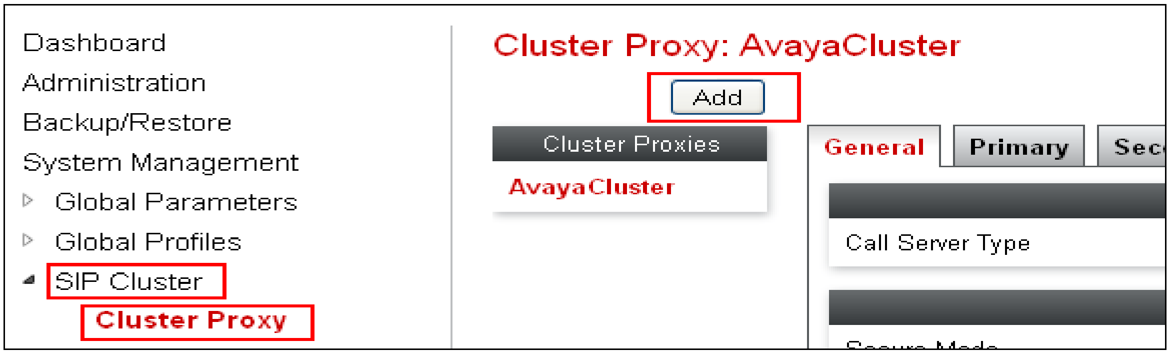
8.13. Administer Internal Media Interface Toward Session Manager

This section explains administering a media interface to the Session Manager Server. Select **Device Specific Settings→Media Interface→Add Media**. The **Name** was set to **Int_Med_intr_Call_Srv**. The **IP Address** was set to **192.168.1.16**. The **Port Range** was set to **35000 – 40000**. The **Finish** button was selected to save the changes.

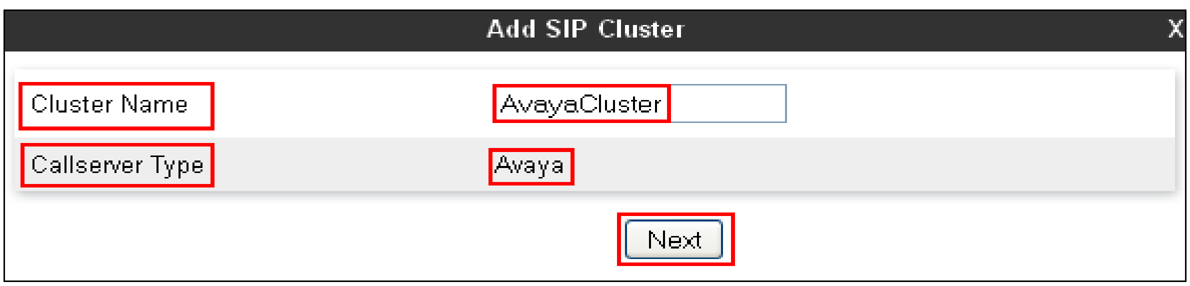


8.14. Administer SIP Cluster

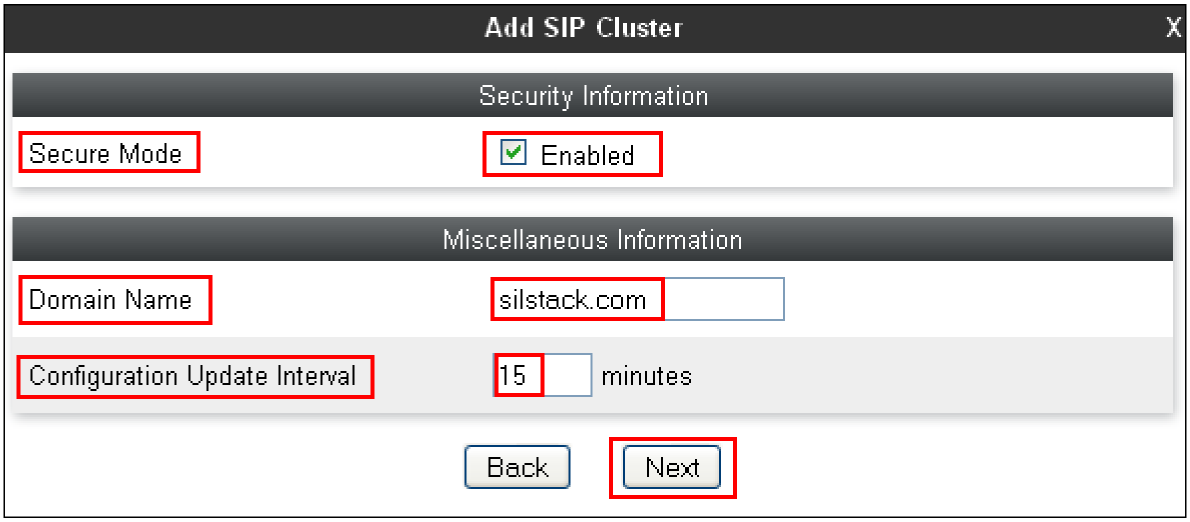
This section describes creating a SIP Cluster. The SIP Cluster will be administered with secure mode to use TLS for SIP. This allows the user to use https for the endpoint configuration download. A TLS Server Profile will be created to allow the user to download PPM information to the Avaya one-X Mobile SIP for IOS as a Remote User. Select **SIP Cluster→Cluster Proxy→Add.**



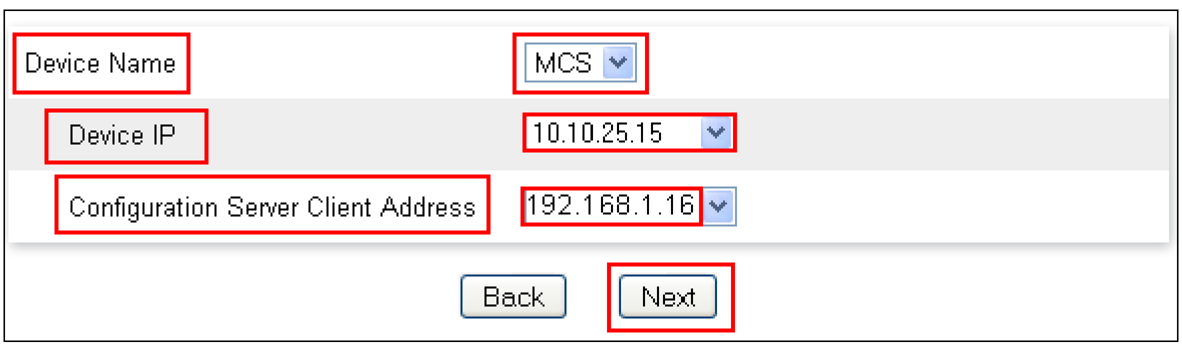
The **Cluster Name** was set to **AvayaCluster.** The **Callserver Type** was set to **Avaya.** The **Next** button was selected to continue to the next page.



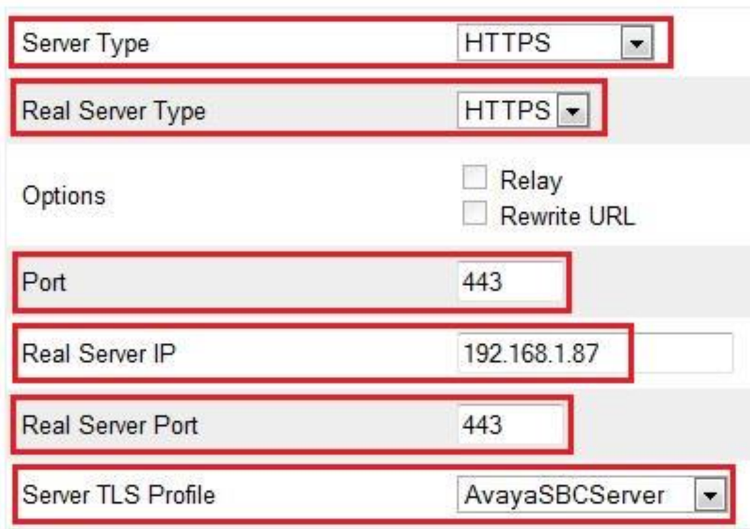
The **Secure Mode** was **Enabled** to allow TLS for SIP. The **Domain Name** was set to **silstack.com.** The **Finish** button was selected to save the changes.



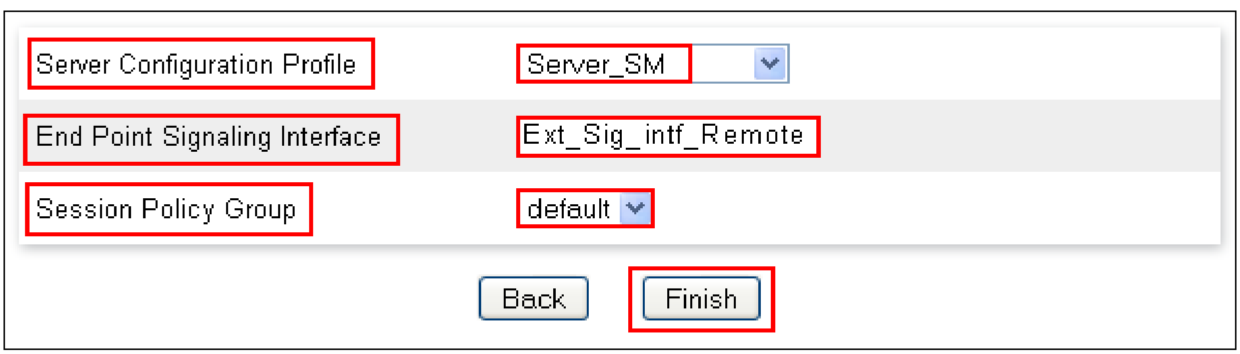
The **Device Name** was set to **MCS**. The **Device IP** was set to **10.10.25.15**. This was the B1 external interface of the Avaya Avaya Session Border Controller. The **Configuration Server Client Address** was set to **192.168.1.16.** This was the A1 internal interface of the Avaya Session Border Controller. The **Next** button was selected to continue to the next page.



The **Server Type** was set to **HTTPS.** The **Real Server Type** was set to **HTTPS**. The **UC-Sec Port** was set to **443**. The **Real Server IP** was set to **192.168.1.87**. This is the IP Address of Session Manager . The **Real Server Port** was set to **443**. The **Server TLS Profile** was set to **AvayaSBCServer.** The **Finish** button was selected to save the changes.

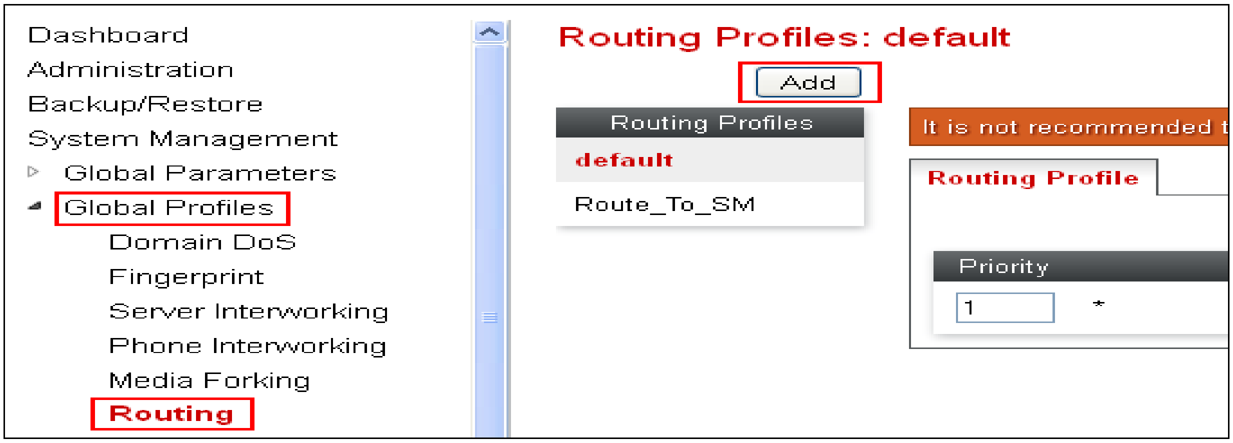


The Session Manager **Server Configuration Profile** was set to **Server_SM**. The **End Point Signaling Interface** was set to **Ext_Sig_intf_Remote_Phone**. The **Session Policy Group** was set to **default**. The **Finish** button was selected to save the changes.

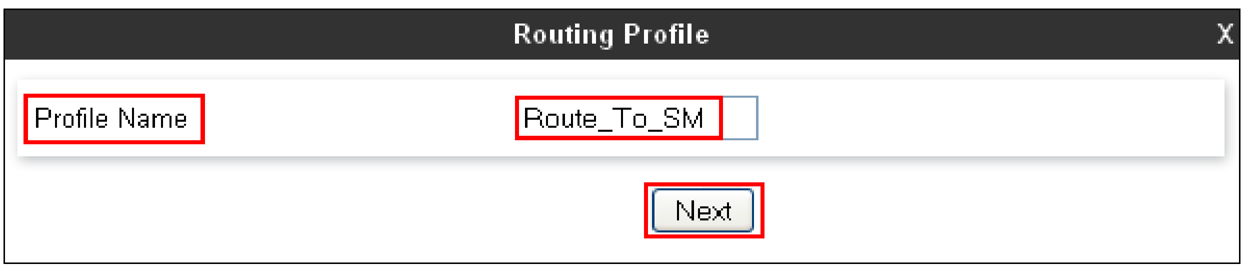


8.15. Administer Routing Profile Toward Session Manager for Subscriber Flow

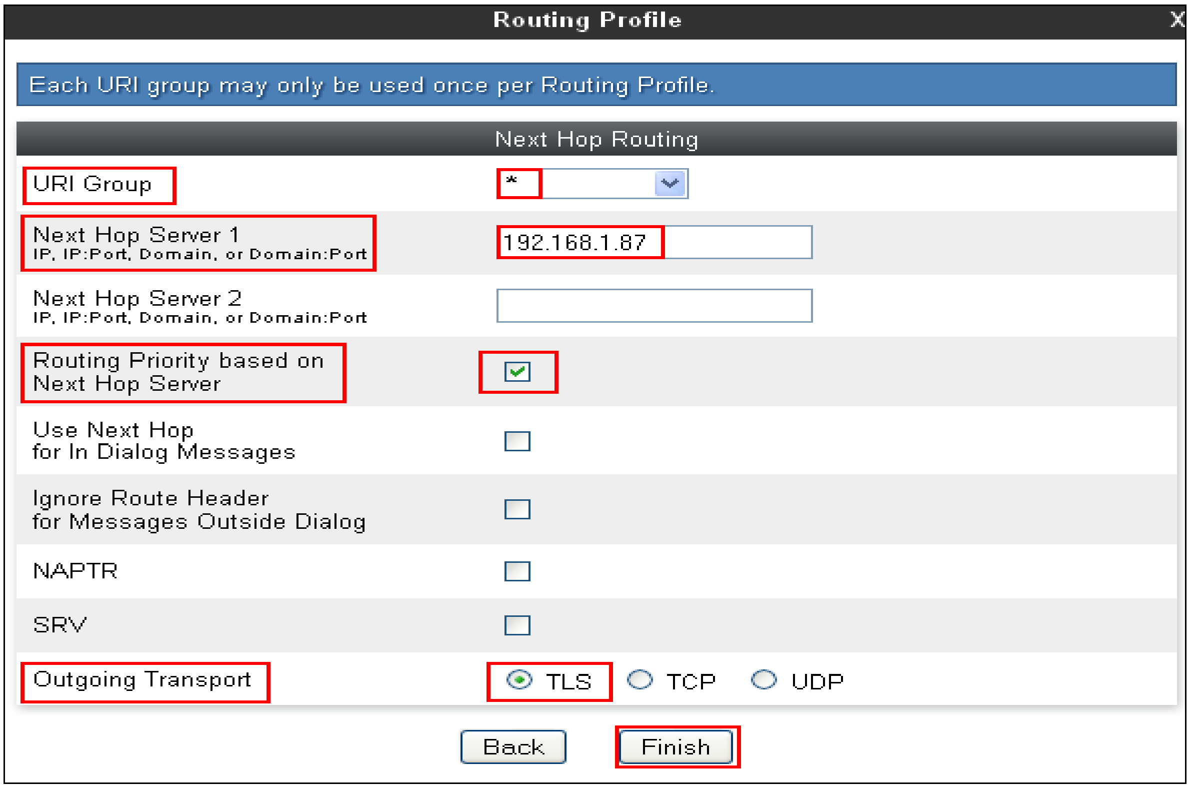
A Routing Profile is administered to the Session Manager and must be assigned to the Subscriber Flow. Select **Global Profiles→Routing→Add**.



The **Profile Name** was set to **Route_To_SM**. The **Next** button was selected to continue to the next page.

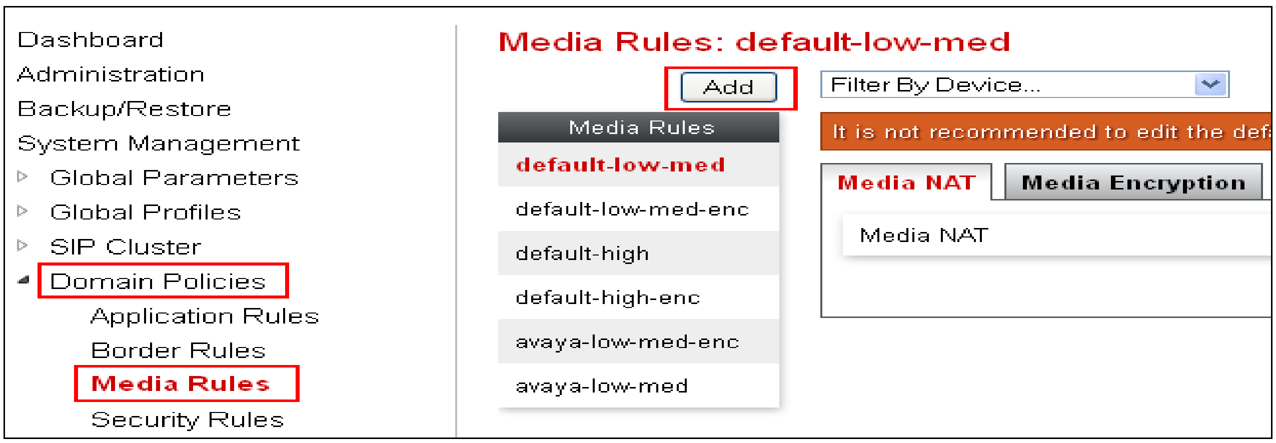


The **URI Group** was set to * to indicate all URI groups were acceptable. The **Next Hop Server 1** was set to **192.168.1.87**. This was the IP Address of the Signaling Interface of the Session Manager. The **Routing Priority based on Next Hop Server** value was **Enabled**. The **Outgoing Transport** was set to **TLS.** The **Finish** button was selected to save the changes.

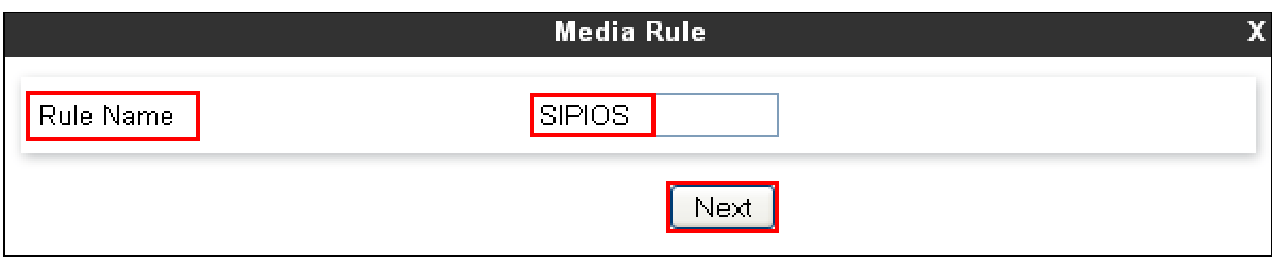


8.16. Administer SRTP Media Rule for the End Point Policy Group for Subscriber Flow and Server Flow

A specific Media Rule is administered and assigned to the End Point Policy Group which will be assigned to the Subscriber and Server Flows. Since the Avaya one-X Mobile SIP for iOS as a Remote User is registering to the Avaya Session Border Controller using TLS, it was decided that the Avaya one-X Mobile SIP for IOS as a Remote User would be administered to use SRTP. It was also decided that SRTP would be administered from inside interface of the Session Border controller to the Communication Manager Server. Therefore a media rule called SRTPSIP is administered with SRTP and assigned to the End Point Policy Group and then assigned to the Subscriber and Server Flows. This ensures the B1 outside interface to the Avaya one-X Mobile SIP for IOS as a Remote User will use SRTP media encryption and the A1 inside interface to the Session Manager and Communication Manager Servers will also use SRTP media encryption. To add a Media rule select **Domain Policies→Media Rules→Add.**

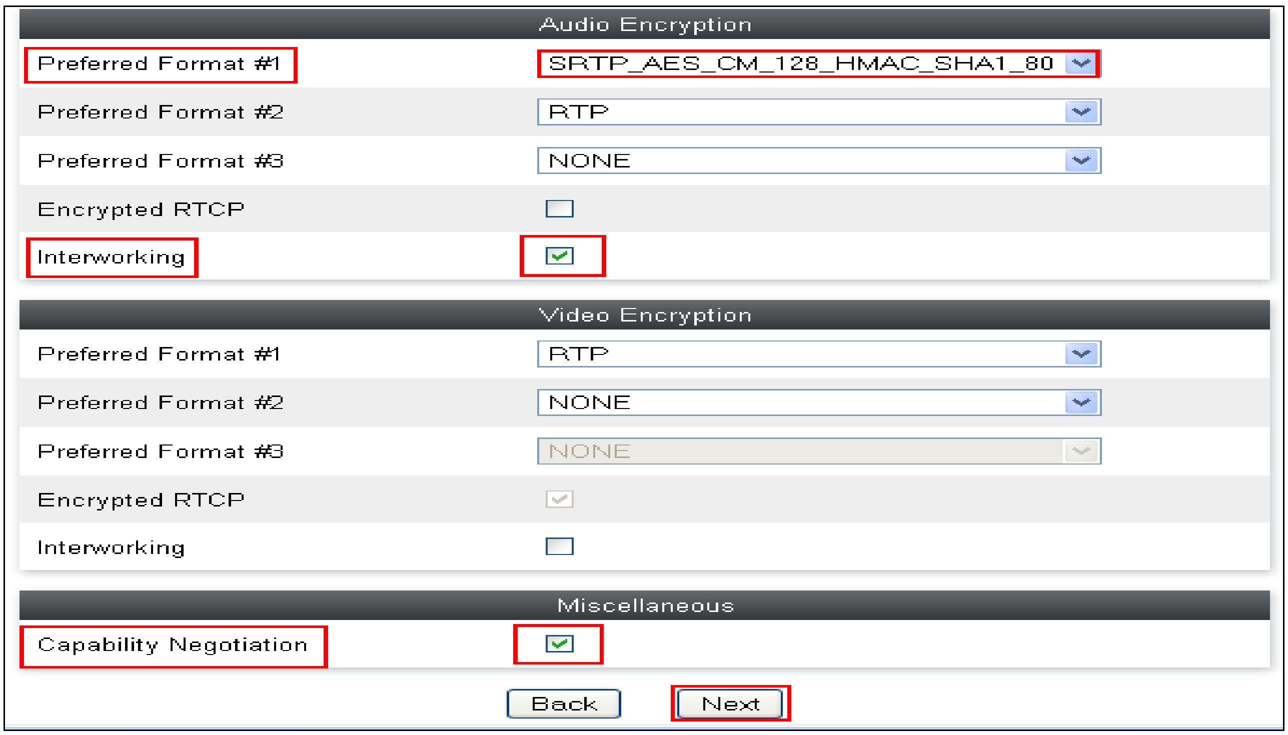


The **Rule Name** was set to **SIPIOS**. The **Next** button was selected to continue to the next page.

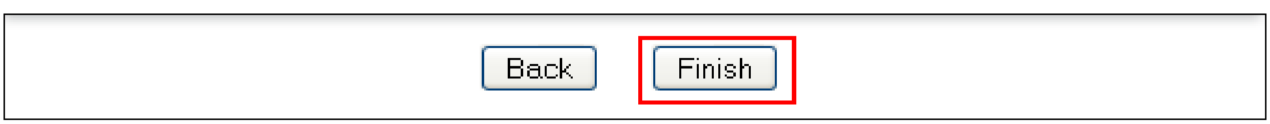


The **Media Nat** value had the **Learn Media IP dynamically** setting **Enabled**. The **Next** button was selected to continue to the next page.

The **Preferred Format #1** value was set to **SRTP_AES_CM_128_HMAC_SHA1_80**. The **Interworking** setting was **Enabled.** The **Capability Negotiation** setting was also **Enabled.** The **Next** button was selected to proceed to the next page.

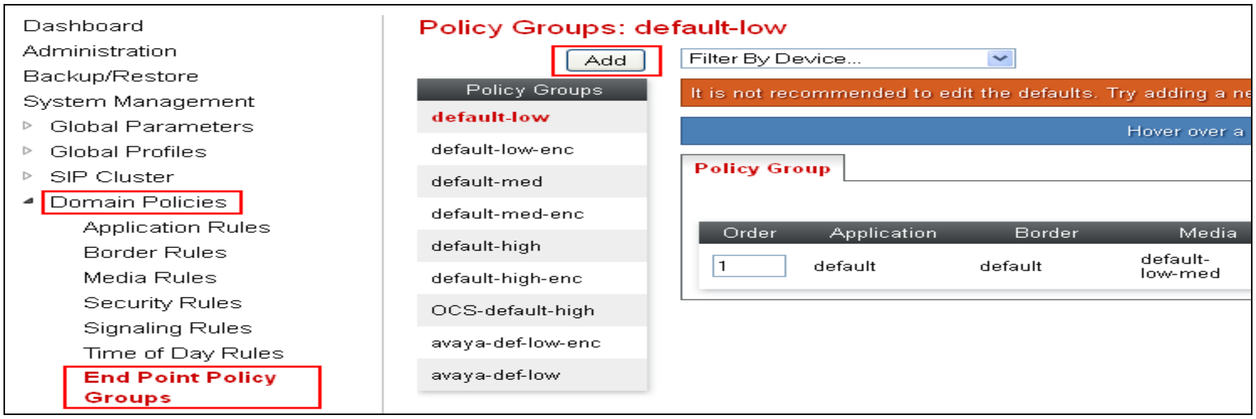


The Media Anomaly, the Media Silencing and the Media QoS settings were all disabled on the Media Rule. The **Finish** was selected to save the changes.



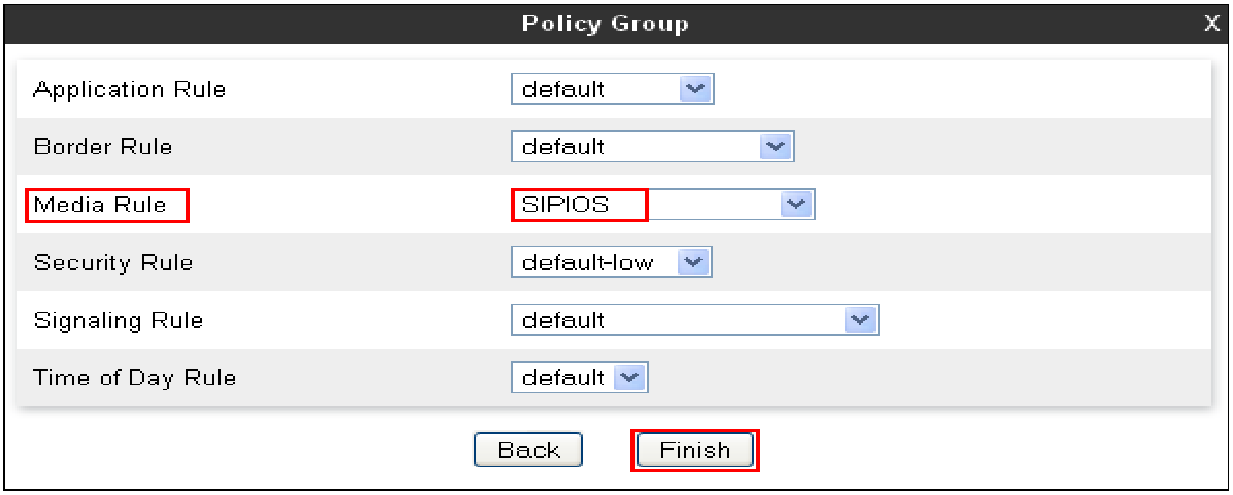
8.17. Administer End Point Policy Group for Subscriber Flow and Server Flow

An End Point Policy Group called SRTPSIPIOS was administered to be assigned to the Subscriber and Server Flow. To add an End Point Policy Group select **Domain Policies→End Point Policy Group→Add.**



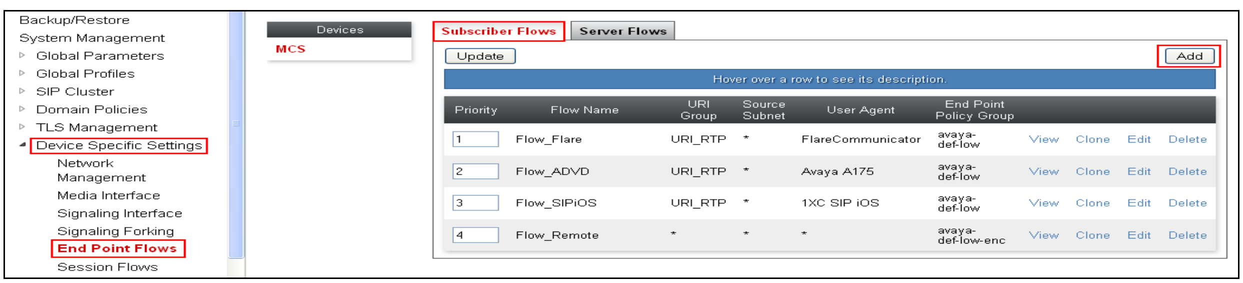
The **Group Name** was set to **SRTPSIPIOS**. The **Next** button was selected to continue to the Next page.

The **Media Rule** called **SIPIOS** was assigned to the End Point Policy Group called SRTPSIPIOS. All other settings where set as default. The **Finish** button was selected to save the changes.

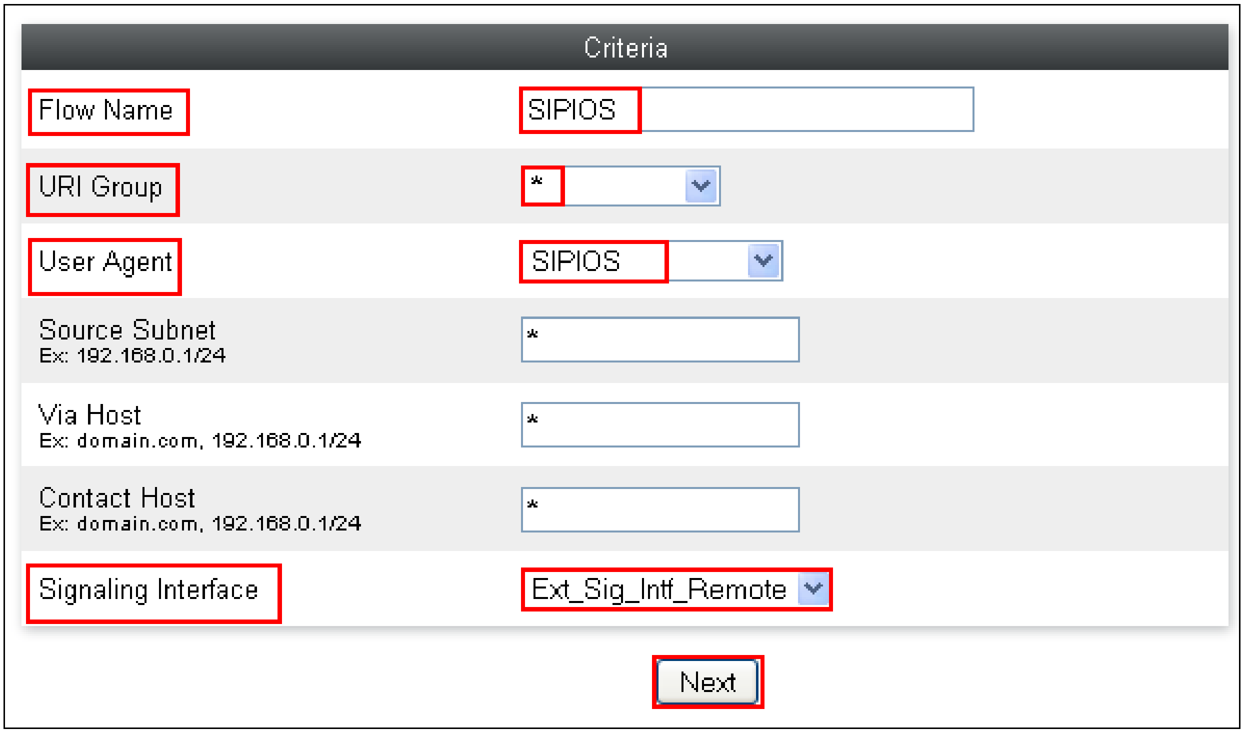


8.18. Administer End Point Flow with Subscriber Flow

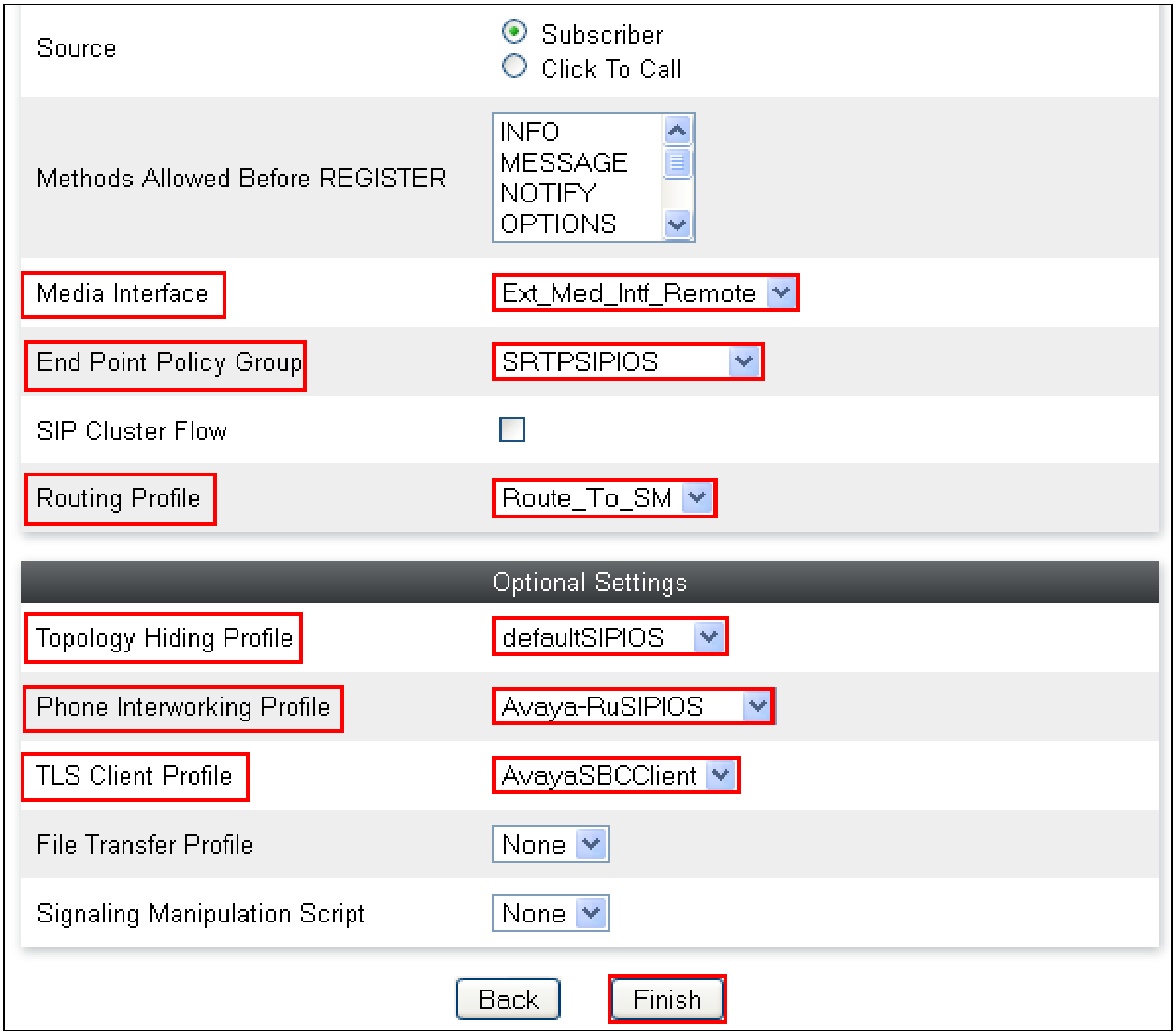
The End Point Flow allows the user to determine how calls will be handled on the Session Border Controller. Select **Device Specific Settings→End Point Flow→Subscriber Flow→Add.**



The **Flow Name** was set to **SIPIOS**. The **URI Group** was set to *. The **User Agent** was set to **SIPIOS.** The **Signaling Interface** was set to **Ext_Sig_intf_Remote_Phone**. The **Next** button was selected to continue to the next page.

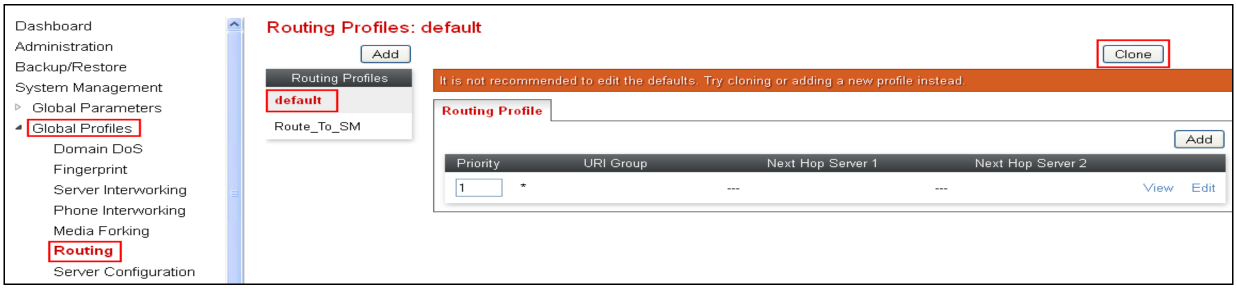


The **Media Interface** was set to **Ext_Med_intf_Remote_Phone**. The **End Point Policy Group** was set to **SRTPSIPIOS**. The **Routing Profile** was set to **Route_To_SM**. The **Topology Hiding Profile** was set to **defaultSIPIOS**. The **Phone Interworking Profile** was set to **Avaya_RuSIPIOS.** The **TLS Client Profile** was set to **AvayaSBCClient**. The **Finish** button was selected to save the changes.

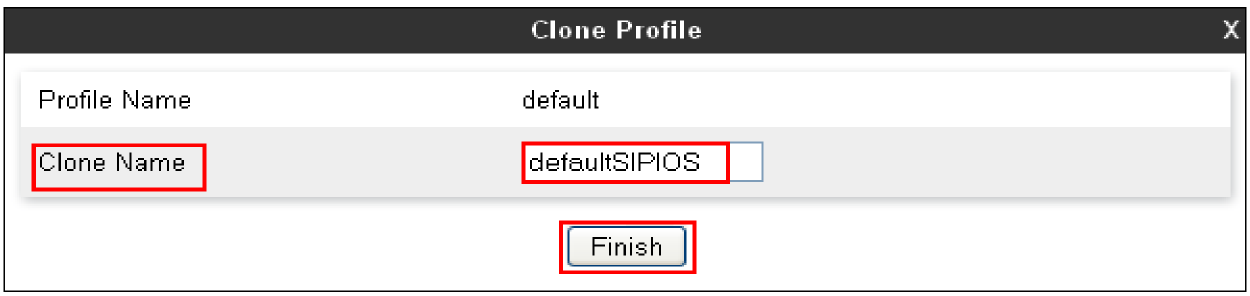


8.19. Administer Routing Profile Toward Remote User for Server Flow

A Routing Profile is administered to the Remote User and must be assigned to the Server Flow. It was decided to use an existing Routing Profile called **default** and clone this Routing Profile. To clone the Routing Profile select **Global Profiles→Routing→default→Clone.**



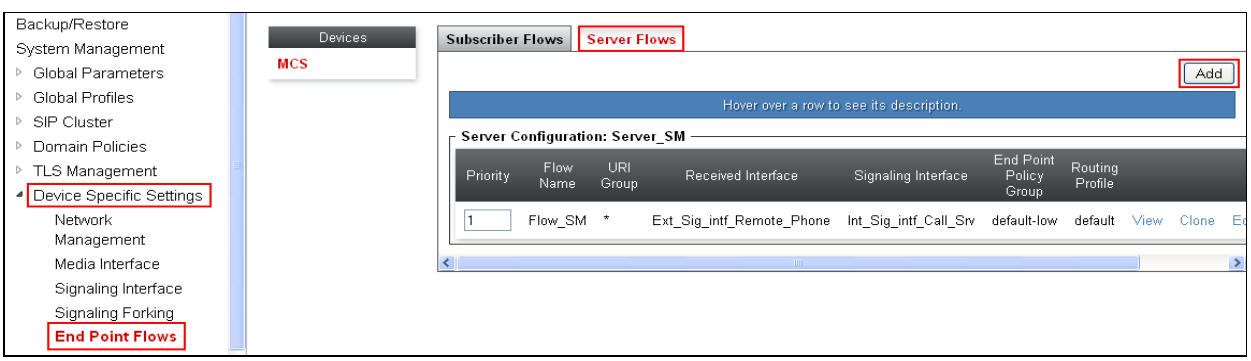
The **Profile Name** selected was **default**. The **Clone Name** was set to **defaultSIPIOS**. The **Finish** button was selected to save the changes.



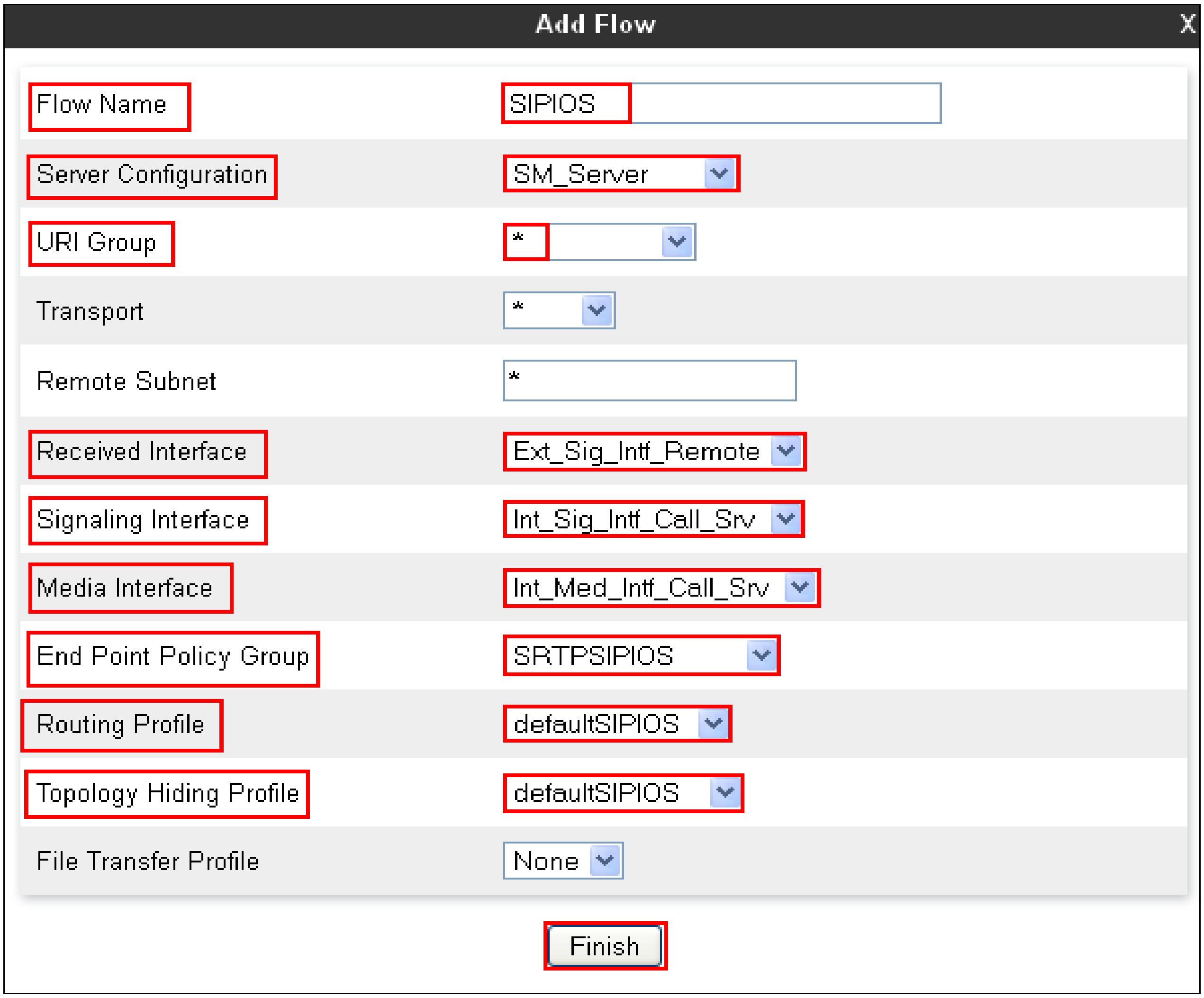
All values for the Routing Profile were left as default.

8.20. Administer End Point Flow with Server Flow

To administer the Server Flow to the End Point Flow select **Device Specific Settings→End Point Flow→Server Flow→Add.**



The **Flow Name** was set to **SIPIOS.** The **Server Configuration** was set to **Server_SM.** The **URI Group** was set to ***.** The **Received Interface** was set to **Ext_Sig_intf_Remote _Phone**. The **Signaling Interface** was set to **Int_sig_intf_Call_Srv**. The **Media Interface** was set to **Int_Med_intf_Call_Srv**. The **End Point Policy Group** was set to **SRTPSIPIOS**. The **Routing Policy** was set to **defaultSIPIOS.** The **Topology Hiding Profile** was set to **defaultSIPIOS**. The **Finish** button was selected to save the changes.



9. Administer Avaya one-X® Mobile SIP for IOS

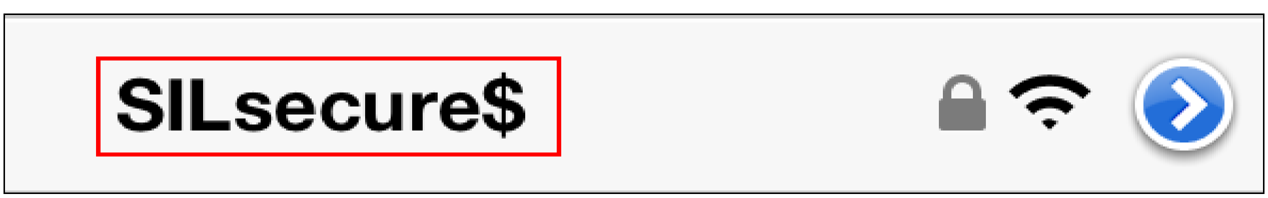
This section highlights the important commands for administering Avaya one-X® Mobile SIP for IOS to set up a sip account and register to the Session Border Controller Advanced for Enterprise Server. It also describes configuring the Avaya one-X® Mobile SIP for IOS to connect to the SILsecure$ wireless network. This Application Notes assumes that the Avaya one-X® Mobile SIP for IOS App has already been downloaded to an IPhone 4S handset.

9.1. Access Wireless Network

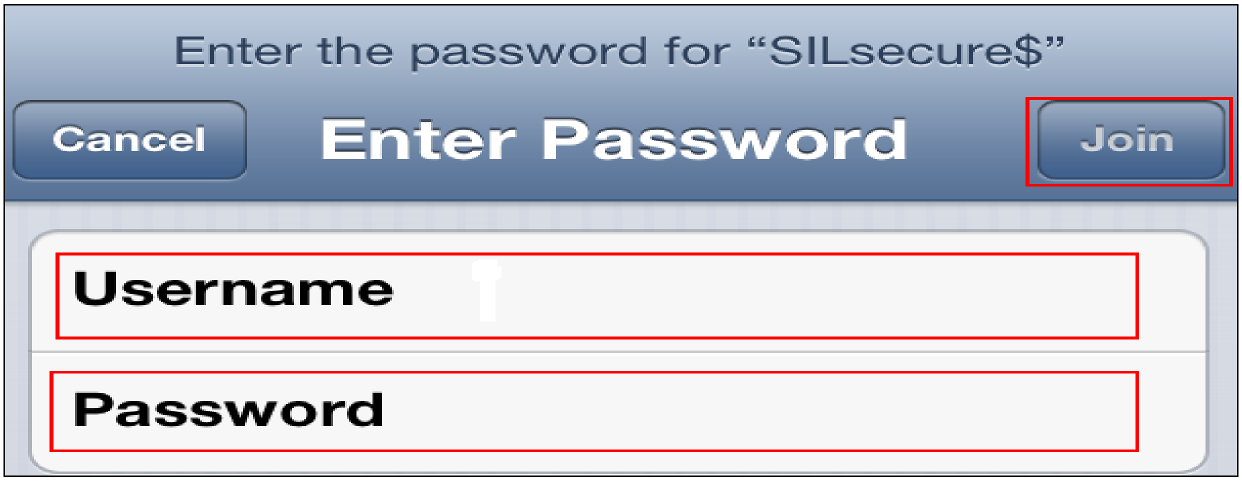
Access the **Settings** heading on the IPhone4S handset.

Under the **Wi-Fi Networks** and select the **Choose a Network** heading.

The **SILsecure$** wireless network was selected.



The **Username** and **Password** was entered and the **Join** button was selected.



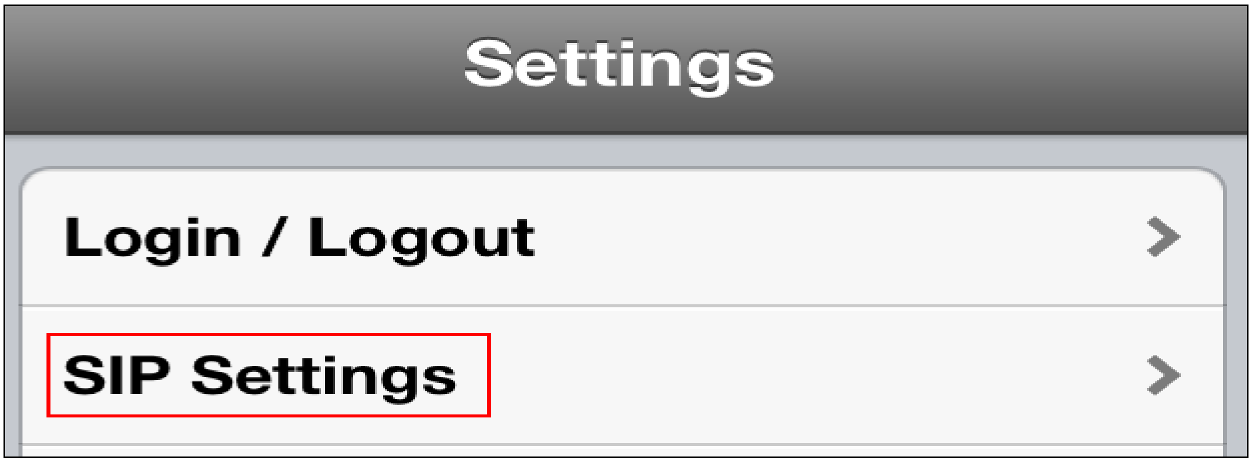
The Avaya one-X Mobile SIP for IOS had joined the **SILsecure$** wireless network.

9.2. Administering Avaya one-X® Mobile SIP Communicator for iOS

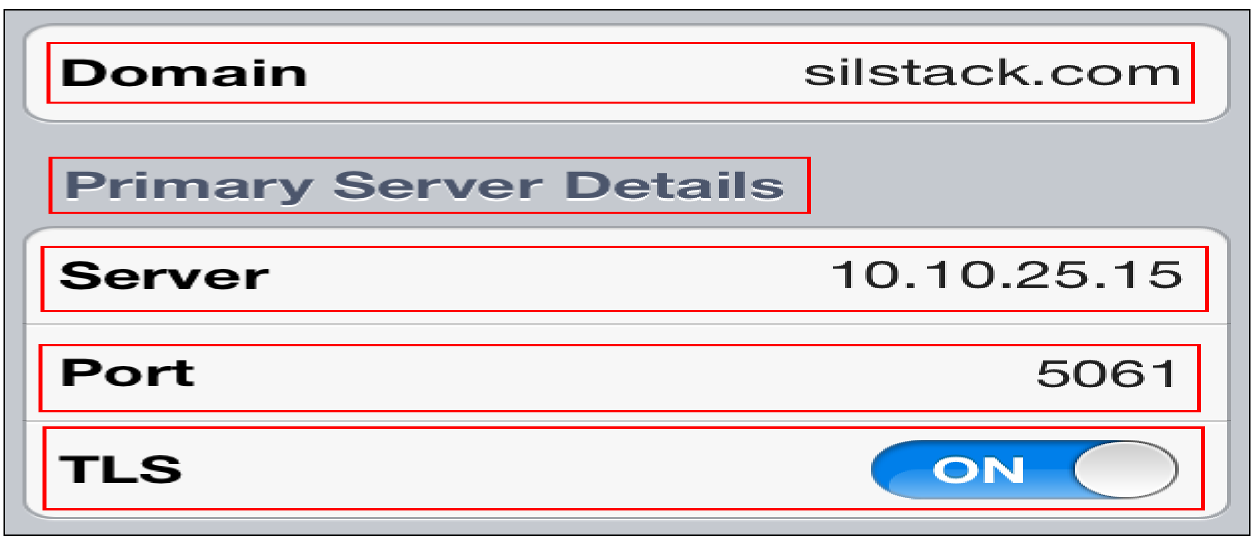
It is assumed that the Avaya one-X® Mobile SIP for IOS has already been downloaded to an IPhone 4S handset. Select the Avaya one-X® Mobile SIP for IOS heading on the IPhone 4S.

Select the **Settings** heading at the button of the screen.

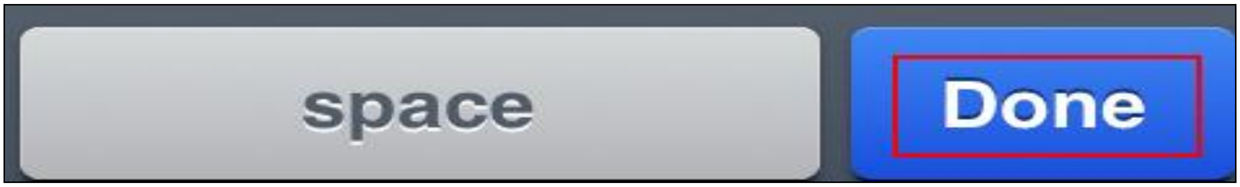
Select the heading **SIP Settings**.



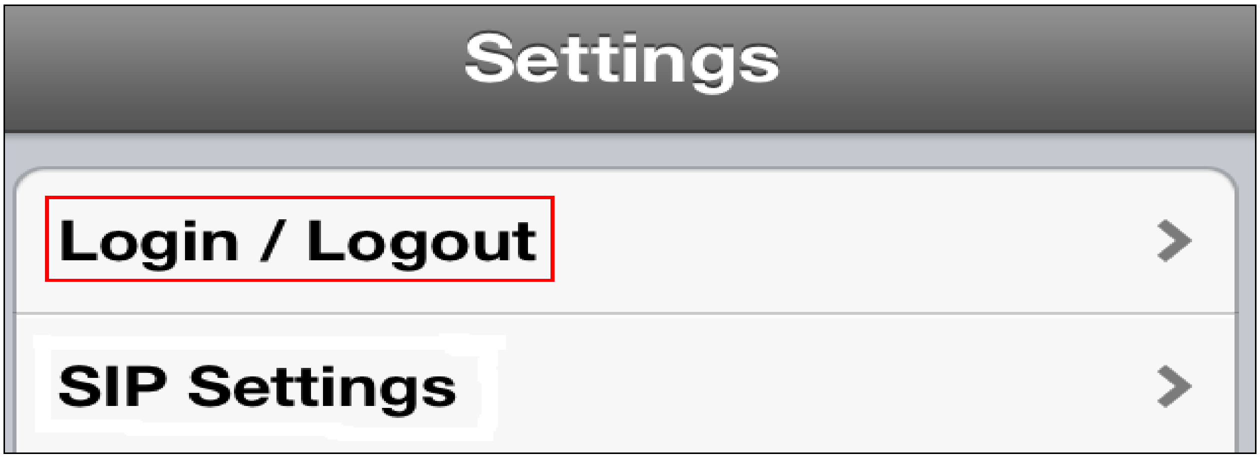
The **Domain** was set to **silstack.com.** Under the **Primary Server Details** the **Server** was set to **10.10.25.15**. The **Port** was set to **5061** and the protocol was set to **TLS.**



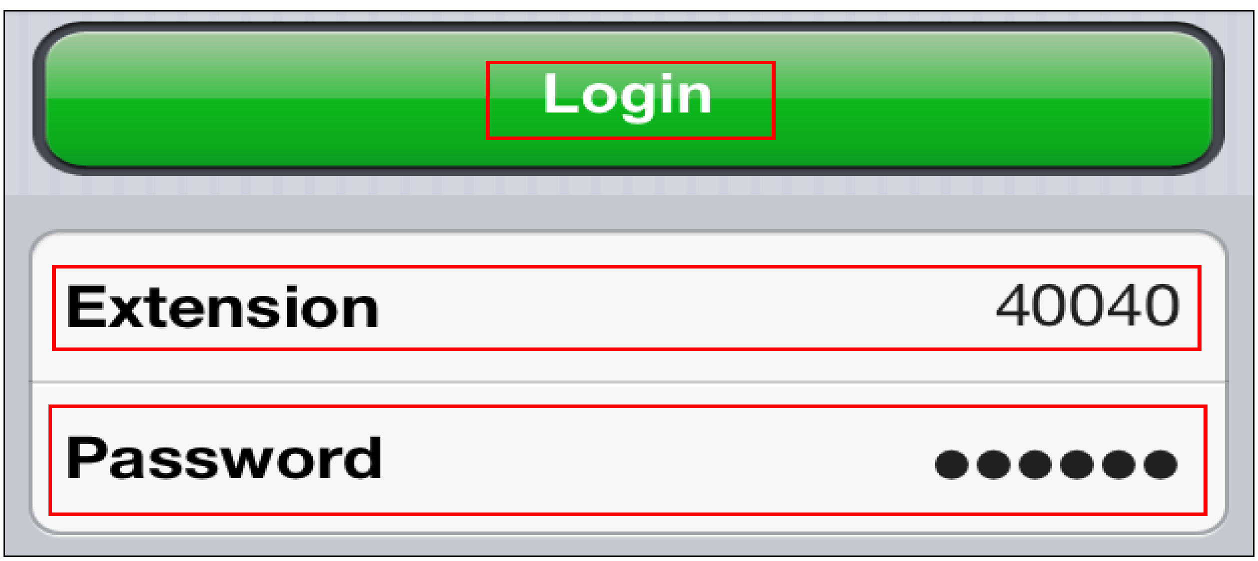
The **Done** button was selected.



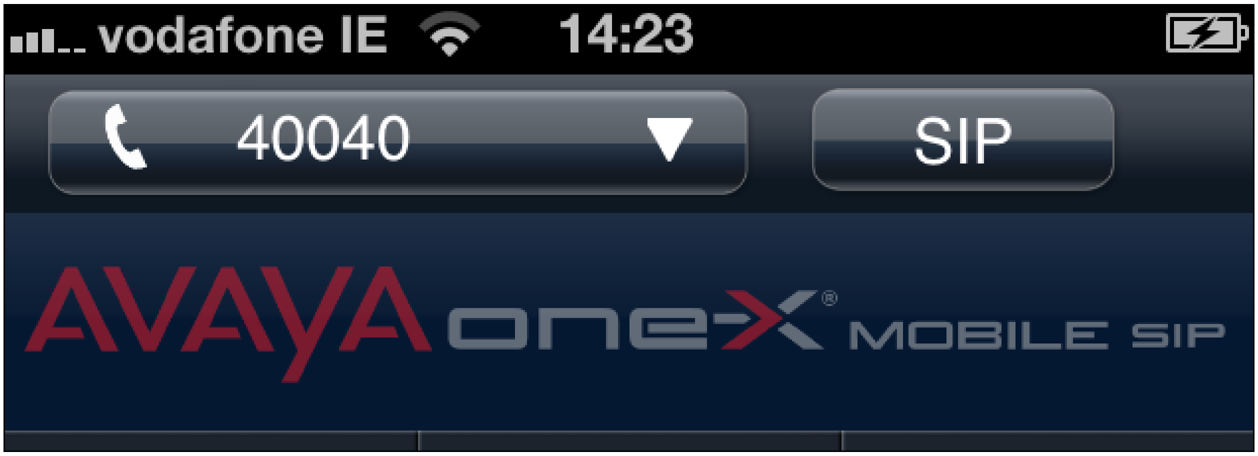
The **Login / Logout** heading was selected.



The **Extension** was set to **40040** and the **Password** was set. The **Login** button was selected.



The Avaya one-X Mobile SIP for IOS was seen to register to the Session Border Controller Server.



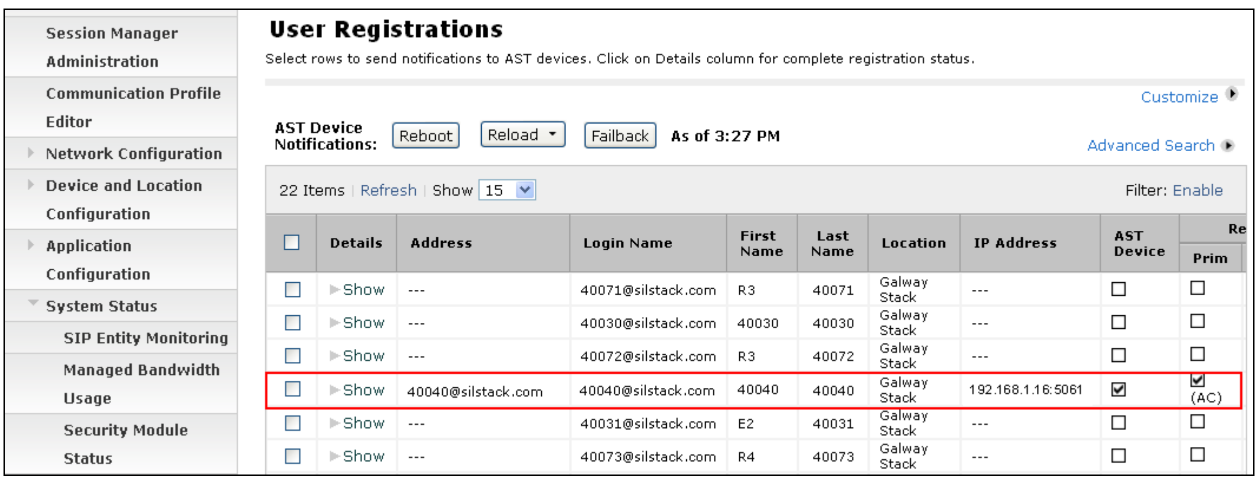
10. Verification Steps

The following six verification steps were tested using the sample configuration. The following steps can be used to verify installation in the field.

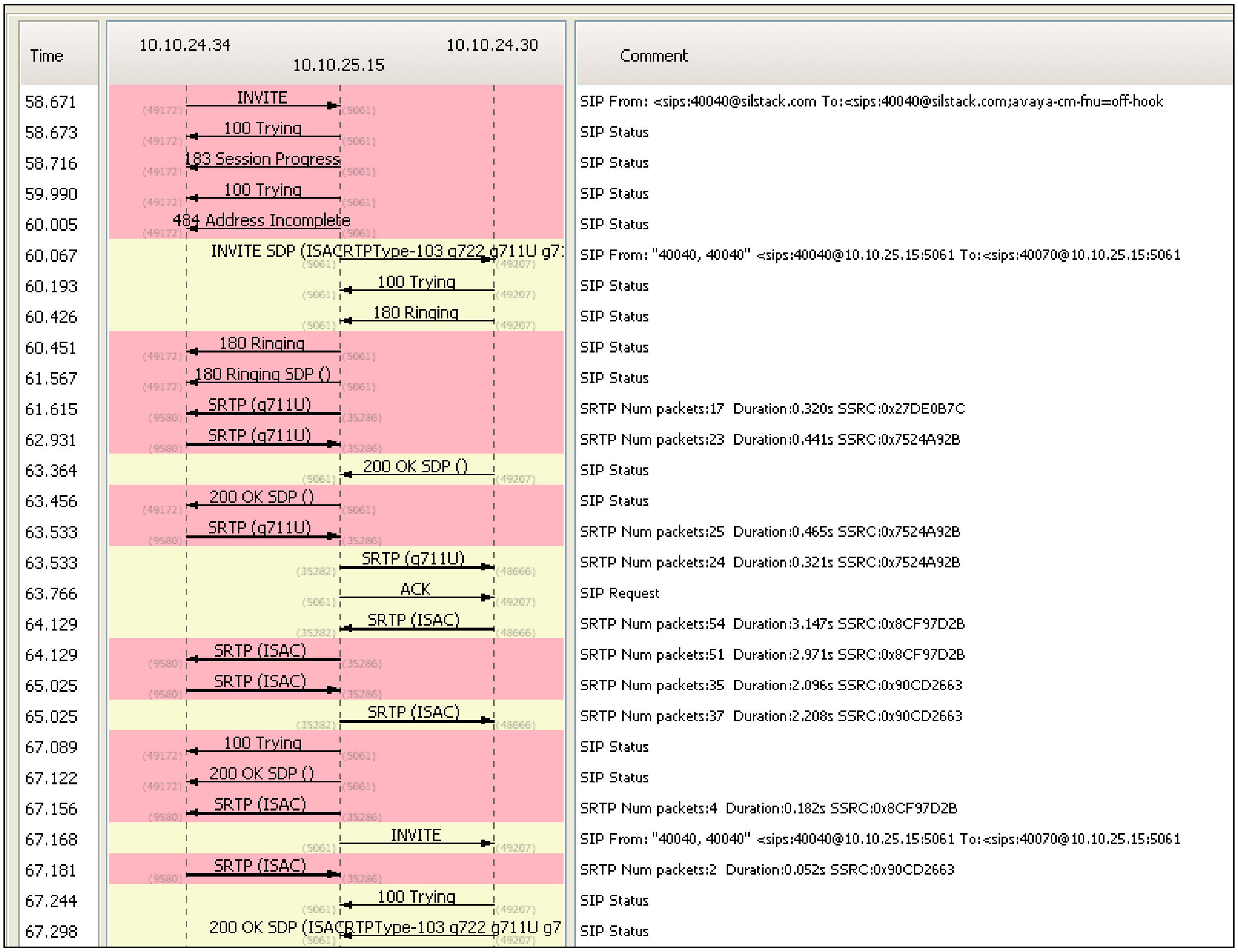
1. Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP obtained an IP Address from the SILsecure$ wireless network.
2. Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP was registered to the Avaya Session Border Controller Advanced for Enterprise Server.
3. Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server was seen to use SRTP from the Remote User to the outside interface of the Avaya Session Border Controller Advanced for Enterprise Server.
4. Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server was seen to use SRTP from the inside interface of the Avaya Session Border Controller Advanced for Enterprise Server to the Avaya Communication Manager Server.
5. Verified that a message could be left on the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server and that the MWI was seen to function correctly for Basic Messaging.
6. Verified the PPM button information was seen on the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server.

Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP obtained an IP Address from the SILsecure$ wireless network.

Verified that **extension 40040** was seen to register to the Avaya Session Border Controller Advanced for Enterprise Server.



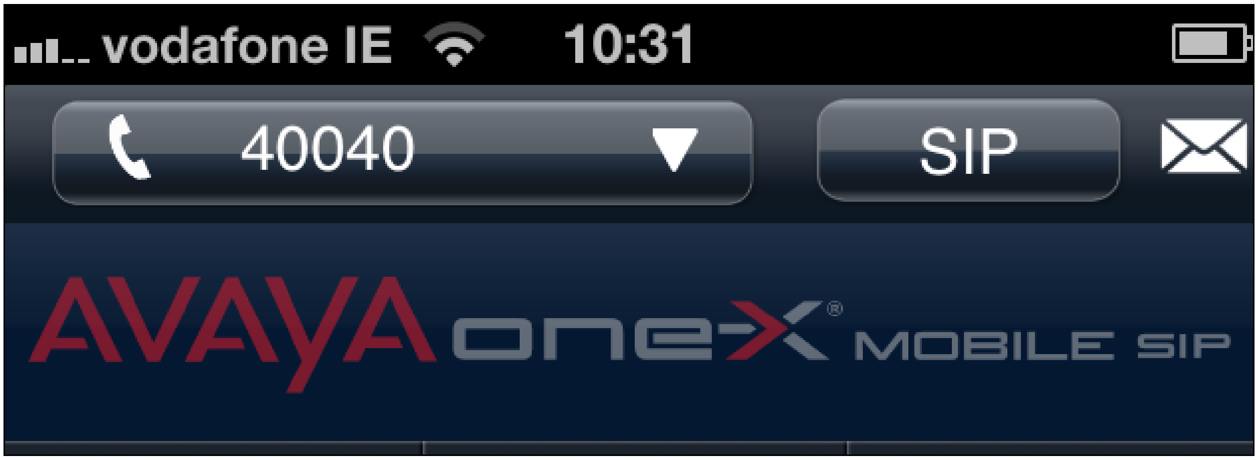
Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server was seen to use SRTP from the Remote User to the outside interface IP Address 10.10.25.15 of the Avaya Session Border Controller Server.



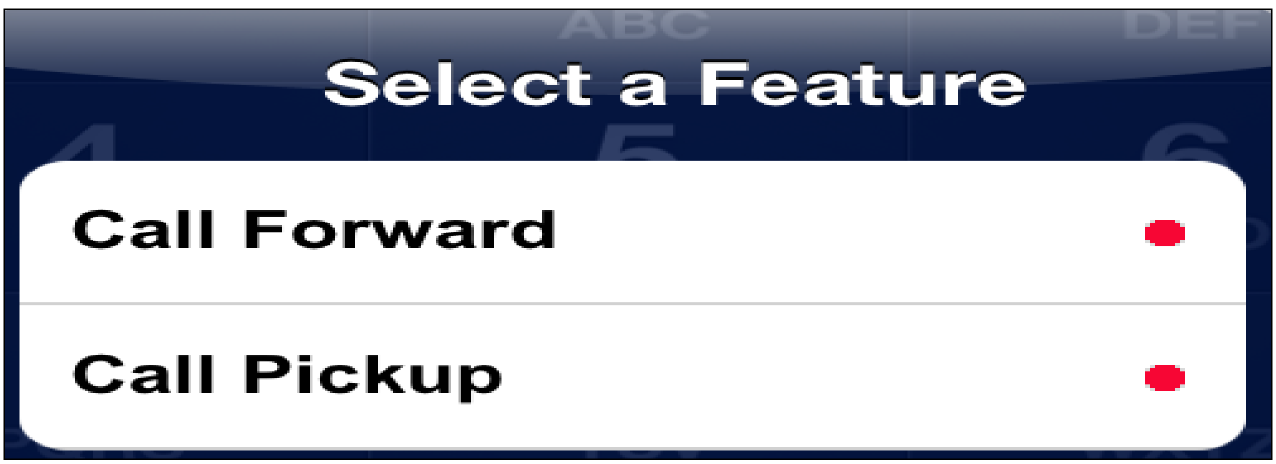
Verified the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server was seen to use SRTP from the inside interface IP Address 192.168.1.16 of the Avaya Session Border Controller Server to the Avaya Communication Manager Server.

```
status trunk 120/113                                            Page    3 of    3
                                  SRC PORT TO DEST PORT TALKPATH
src port: T00155
T00155:TX:192.168.1.16:35134/g722-64/20ms/1-srtp-aescm128-hmac80
T00291:RX:192.168.1.16:35132/g722-64/20ms/1-srtp-aescm128-hmac80
```

Verified that a message could be left on the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server and that the MWI was seen to function correctly for Basic Messaging.



Verified the PPM button information was seen on the Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server.



11. Conclusion

These Application Notes describe the configuration steps required to register the Avaya one-X® Mobile SIP for IOS as a Remote User with SRTP to the Avaya Session Border Controller Advanced for Enterprise Server with Avaya Aura® Solution for Midsize Enterprise Server and Avaya Aura® Messaging Server. These Application Notes also identify how to configure SRTP from the Avaya one-X® Mobile SIP for IOS as a Remote User to the outside interface of the Avaya Session Border Controller Advanced for Enterprise Server and configure SRTP from the inside interface of the Avaya Session Border Controller Advanced for Enterprise Server to the Avaya Aura® Solution for Midsize Enterprise Server and Avaya Aura® Messaging Server.

These Application Notes also describe how to administer Avaya Aura® Messaging Server to function with SRTP with the Avaya one-X® Mobile SIP for IOS as a Remote User with the Avaya Session Border Controller Advanced for Enterprise Server. Please refer to **Section 2.1.4** for the observations associated with Avaya one-X Mobile SIP for IOS as a Remote User with SRTP registered to the Avaya Session Border Controller Advanced for Enterprise Server.

12. Additional References

This section references Avaya documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] Administering Avaya Session Border Controller for Enterprise, January 2013, Release 6.2, Issue 2.

[2] Administering Avaya one-X® Mobile SIP for IOS, November 2012, Release 6.2, Issue 1.

[3] Administering Avaya Aura® System Manager, July 2012 Issue 2.0

[4] Administering Avaya Aura® Session Manager, February 2012, Document Number 03-603324

[5] Administering Avaya Aura® Communication Manager, February 2012, Document Number 03-603479

[6] Administering Avaya Aura® Messaging, December 2011

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com