Avaya Solution & Interoperability Test Lab

Configuring SIP Trunks among Avaya Aura® Session Manager R6.0 SP1, Avaya Aura® Communication Manager R6.0 SP2, and Cisco Unified Communications Manager Express R8.1 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses Avaya Aura® Session Manager to connect Avaya Aura® Communication Manager and Cisco Unified Communications Manager Express using SIP trunks.

For the sample configuration, Avaya Aura® Session Manager runs on an Avaya S8800 Server, Avaya Aura® Communication Manager runs on an Avaya S8800 Server with Avaya G450 Media Gateway, and Cisco Unified Communications Manager Express runs on a Cisco 3825 Integrated Services Router (ISR). The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura® Communication Manager.

1 Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura® Session Manager R6.0 Service Pack 1 to connect Avaya Aura® Communication Manager R6.0 Service Pack 2 and Cisco Unified Communications Manager Express (Cisco UCME) R8.1 using SIP trunks.

As shown in **Figure 1**, the Avaya 9630 IP Telephone (H.323), Avaya 9620C IP Telephone (SIP), and 2420 Digital Telephone are supported by Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Cisco 7965G IP Telephone (SIP) and the Cisco 7975G IP Telephone (SCCP) are supported by the Cisco UCME. SIP trunks are used to connect these two systems to Avaya Aura® Session Manager. All inter-system calls are carried over these SIP trunks. Avaya Aura® Session Manager can support flexible inter-system call routing based on dialed number, calling number and system location, and can also provide protocol adaptation to allow multi-vendor systems to interoperate. It is managed by a separate Avaya Aura® System Manager, which can manage multiple Avaya Aura® Session Managers by communicating with their management network interfaces. Avaya Modular Messaging expands the capabilities and features by providing centralized voicemail services to subscribers at the Cisco and Avaya sites. The Avaya Modular Messaging configuration is outside of the scope of these Application Notes.

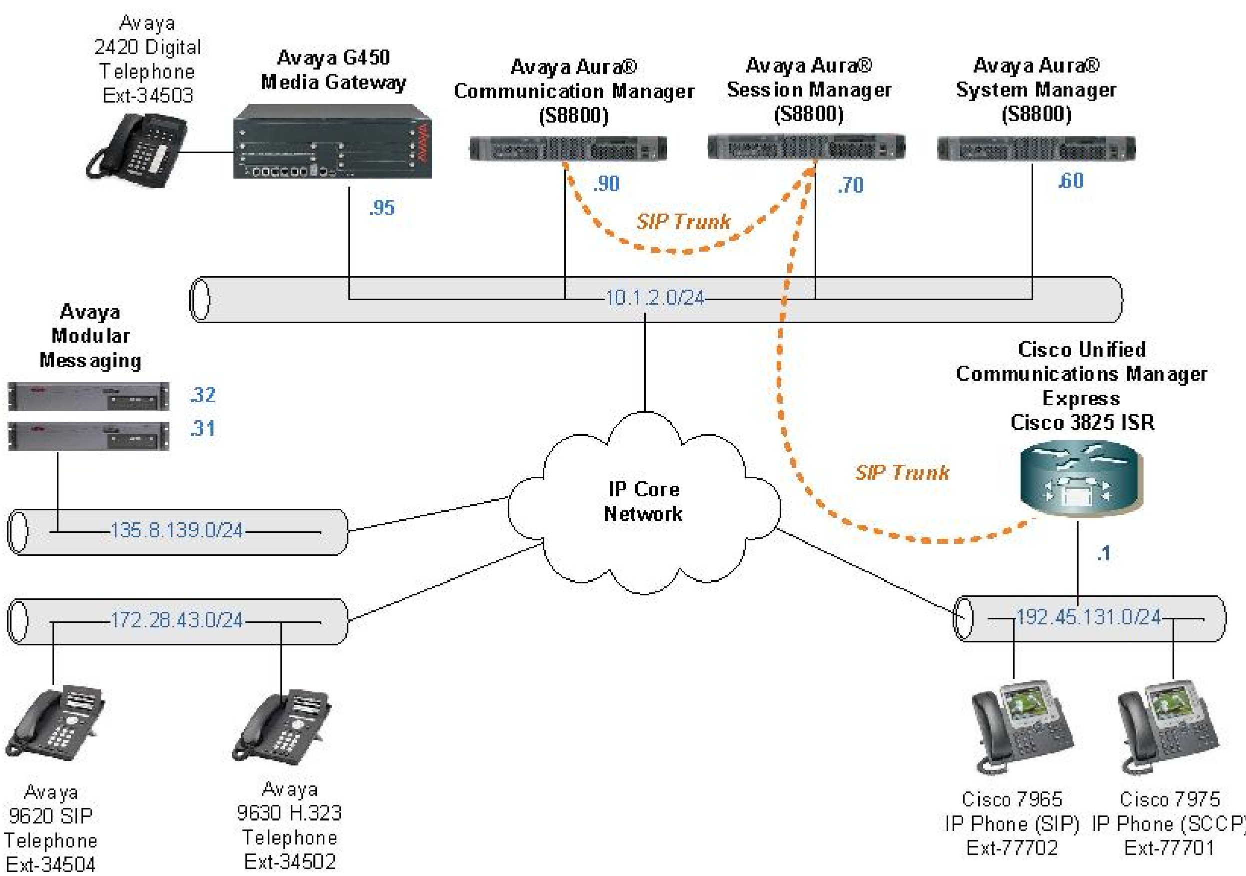


Figure 1 – Sample Configuration

For the sample configuration, Avaya Aura® Session Manager runs on an Avaya S8800 Server, Avaya Aura® Communication Manager runs on an Avaya S8800 Server with Avaya G450 Media Gateway, and Cisco Unified Communications Manager Express runs on Cisco 3825 Integrated Services Router (ISR). The results in these Application Notes should be applicable to other Avaya Aura® servers and Media Gateways.

A five digit Uniform Dial Plan (UDP) is used for dialing between systems. Unique extension ranges are associated with Avaya Aura® Communication Manager R6.0 (345xx) and Cisco UCME R8.1 (777xx).

These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones will not be described (see the appropriate documentation listed in **Section 9**).

2 Equipment and Software Validated

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

DEVICE DESCRIPTION	VERSION TESTED
Avaya Aura® Communication Manager Running on an Avaya S8800 Server with an Avaya G450 Media Gateway	6.0 (R016x.00.0.345.0) with SP2 (00.0.345.0-18567)[1]
Avaya Aura® System Manager Running on an Avaya S8800 Server	6.0 SP1 (Build No. 6.0.0.0.668-3.0.7.0)
Avaya Aura® Session Manager Running on an Avaya S8800 Server	6.0 SP1 (6.0.1.0.601009)
Avaya 9630 IP Telephone (H.323)	3.101S
Avaya 9620 IP Telephone (SIP)	2.6.3
Avaya 2420 Digital Telephone	-
Avaya Modular Messaging Messaging Application Server (MAS) Messaging Storage Server (MSS)	5.2, SP5 (Patch 1)
Cisco Unified Communications Manager Express Running on a Cisco 3825 ISR	8.1 IOS 15.1(2)T1 (ED)
Cisco 7965G Unified IP Phone (SIP)	SIP45.8-5-4S
Cisco 7975G Unified IP Phone (SCCP)	SCCP75.8-5-4S

[1] Testing start with CM R6.0 SP1 (00.0.345.0-18444) and finished with CM R6.0 SP2 (00.0.345.0-18567).

3 Configure Avaya Aura® Communication Manager

This section illustrates relevant configuration for Communication Manager SIP Trunking to Session Manager. The configuration in this section uses the System Access Terminal (SAT) interface, and screens may be abridged for brevity in presentation. For further information on Communication Manager, please consult references **[4]** and **[5]**. The configuration procedures include the following areas:

- Verify Communication Manager License
- Administer System Parameters Features
- Administer IP Node Names
- Administer IP Network Regions
- Administer IP Codec Sets
- Administer SIP trunk Group and Signaling Group
- Administer Private Numbering
- Administer Uniform Dial Plan
- Administer AAR Analysis
- Administer Route Patterns
- Save Transactions

3.1 Verify Avaya Aura® Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 100
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 156
```

3.2 Configure System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers. This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch For simplicity, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Submit the change.

Note: This feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 9** for more details.

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
              Automatic Callback with Called Party Queuing? y
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y
```

3.3 IP Node Names

Node names are mappings of names to IP Addresses that can be used in various screens. The following abridged screen shows the relevant node-names used in the sample configuration. **Name** "SM1" and **IP Address** "10.1.2.70" are entered for Session Manager. The IP Address of the S8800 processor Ethernet interface named "procr" is configured via the Web administration of the S8800 Server. Here, it can be observed that "procr" and "10.1.2.90" are the **Name** and **IP Address** for Communication Manager running on the Avaya S8800 Server. For other system types, where an Avaya C-LAN card is used as the SIP signaling interface, the node name and IP Address of the C-LAN card would be entered here.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AS5400              10.3.3.40
Edge                10.3.3.60
Home1               10.3.3.50
Home2               10.3.3.41
SES                 10.3.3.50
SM1                 10.1.2.70
SurvCM              10.32.2.80
default             0.0.0.0
msgserver           10.3.3.14
procr               10.1.2.90
procr6              ::
```

3.4 Network Regions

Network regions provide a means to logically group resources. Non-IP telephones (e.g., analog, digital) derive network region and location configuration from the Avaya gateway to which the device is connected. The following display command ("display media-gateway 1") shows that media gateway 1 is an Avaya G450 Media Gateway configured for **Network Region 1**.

```
display media-gateway 1                                           Page   1 of   2
                             MEDIA GATEWAY 1

                    Type: g450
                    Name: G450 Evolution Srvr
               Serial No: 08IS43202588
            Encrypt Link? y                   Enable CF? n
          Network Region: 1                    Location: 1
                                              Site Data:
           Recovery Rule: none

              Registered?  y
   FW Version/HW Vintage: 30 .12 .1  /1
        MGP IPV4 Address: 10.1.2.95
        MGP IPV6 Address:
   Controller IP Address: 10.1.2.90
             MAC Address: 00:1b:4f:03:57:b0
```

Scroll down to **Page 2** to obtain a list of the modules installed on the Avaya G450 Media Gateway.

```
display media-gateway 1                                        Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g450

Slot   Module Type             Name                    DSP Type  FW/HW version
 V1:                                                   MP80      44   3
 V2:
 V3:   MM712                   DCP MM
 V4:
 V5:   MM714                   ANA MM
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

IP telephones can be assigned a network region based on an IP address mapping. The following screen illustrates a subset of the IP network map configuration. When the IP address of a registering IP telephone is in the ip-network-map, the phone is assigned the network region assigned by the form shown below. Strictly speaking, this ip-network-map configuration is not necessary, since the Avaya IP Telephones are assigned to the IP Network Region of the CLAN or PROCR interfaces they register to.

```
change ip-network-map                                          Page   1 of  63
                               IP ADDRESS MAPPING
                                              Subnet Network      Emergency
 IP Address                                   Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM: 172.28.43.0                            /      1      n
   TO: 172.28.43.255
```

The following screen shows IP Network Region 1 configuration. Connections within network region 1 use codec set 1 by virtue of the **Codec Set** configuration shown on **Page 1** below. For the **Authoritative Domain** field, enter the SIP domain configured for this enterprise. Optionally, a descriptive **Name** can be configured. To enable direct media connections for calls between the Avaya devices in network region 1, ensure that the **Intra-region IP-IP Direct Audio** is set to "yes". To permit direct media connections to other regions (unless otherwise prohibited by the other region), set the **Inter-region IP-IP Direct Audio** field to "yes".

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: Avaya devices
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

The following screen shows the inter-network region connection configuration for region 1. The bold row shows that network region 1 is directly connected to network region 3, and that codec set 3 will be used for connections between region 1 and region 3. Later, when the SIP signaling group is defined, the "far-end region" will be set to network region 3. Having different network regions for the local Avaya devices and the far-end of a SIP trunk allows different codec parameters for intra-region connections (e.g., using codec set 1 for Avaya connections) and inter-region connections (e.g., using codec set 3 for Avaya-Cisco connections in the sample configuration). Once submitted, the configuration becomes symmetric, meaning that network region 3, **Page 3** will also show codec set 3 for region 3 – region 1 connectivity.

```
change ip-network-region 1                                      Page   4 of  20

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    1                                                               all
 2    2     y    NoLimit                                             n
 3    3     y    NoLimit                                             n
```

The following screen shows **Page 1** of the IP Network Region 3 configuration. Observe that the **Intra-region IP-IP Direct Audio** and the **Inter-region IP-IP Direct Audio** fields are both set to "yes".

```
change ip-network-region 3                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain:
    Name: Far-end-SIP
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

The following screen shows **Page 3** of the IP Network Region 3 configuration. The bolded row illustrates the symmetric configuration of the region 3-1 connectivity, using codec set 3.

```
change ip-network-region 3                                      Page   3 of  19

 Source Region: 3     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    3     y    NoLimit                                             n
 2
 3    3                                                                 all
```

3.5 IP Codec Sets

The following screens show the configuration for codec sets 1 and 3. In general, an IP codec set is a list of allowable codecs in priority order. In the sample configuration, all connections among the Avaya devices use codec set 1, preferentially using “G.711MU” with “SRTP” encryption, as shown below.

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

    Media Encryption
1: 1-srtp-aescm128-hmac80
2: aes
3: none
```

In the sample configuration, all connections between the Avaya devices and the Cisco devices will use codec set 3, specified for inter-region connections between region 1 and region 3. During the testing, the codec parameters for codec set 3 were varied, with successful calls using “G.711MU”, and variants of G.729, each with no encryption (e.g. “G.729”, “G.729AB”).

```
change ip-codec-set 3                                             Page   1 of   2
                         IP Codec Set
   Codec Set: 3

   Audio        Silence      Frames   Packet
   Codec        Suppression  Per Pkt  Size(ms)
1: G.729             n          2        20
2: G.729AB           n          2        20
3: G.711MU           n          2        20
4:
5:
6:
7:

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

3.6 Configure SIP Signaling Group and Trunk Group

3.6.1 SIP Signaling Group

This section illustrates the configuration of the SIP Signaling Group to Session Manager. The signaling group has a **Group Type** of “sip”, and a **Near-end Node Name** of “procr”, the S8880 Server. For the Communication Manager Evolution Server configuration, **IMS Enabled** should be set to “n” and **Peer Detection Enabled** to ”y”. The **Peer Server** field will be automatically populated as a result of the enabled peer detection. The **Far-end Node Name** is the node name “SM1” for Session Manager. The **Transport Method** is “tls”, and the **Near-End Listen Port** and **Far-End Listen Port** use port “5065”. Since an adaptation module will be defined in Session Manager to set the domain for all incoming calls to “avaya.com”, this value can be put in the **Far-end Domain**, and all outgoing and incoming calls to/from Session Manager will use this single trunk. This eliminates the need for a separate trunk for incoming calls from Cisco UCME which use the IP address of Session Manager instead of the SIP domain. The **Far-end Network Region** has been configured to be “3”, to allow different behaviors, such as codec selection, for intra-region and inter-region calls. Although not required, the **Enable Layer 3 Test** parameter is enabled to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Other fields can be left at default values, including **DTMF over IP** set to “rtp-payload” which corresponds to RFC 2833.

```
add signaling-group 26                                          Page   1 of   1

                                SIGNALING GROUP

 Group Number: 26              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: procr                Far-end Node Name: SM1
 Near-end Listen Port: 5065               Far-end Listen Port: 5065
                                       Far-end Network Region: 3

Far-end Domain: avaya.com
                                             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? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

This section illustrates the configuration of the SIP Trunk Group 26 to Session Manager. The trunk group has a **Group Type** of "sip" and a **Service Type** of "tie". An appropriate Trunk Access Code (**TAC**) and **Group Name** are configured. Trunk group 26 is associated with **Signaling Group** "26", and the **Number of Members** field is "10", indicating that this trunk group can support ten simultaneous calls.

```
add trunk-group 26                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 26                   Group Type: sip          CDR Reports: y
  Group Name: To ASM                      COR: 1        TN: 1        TAC: 126
   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: 26
                                                     Number of Members: 10
```

The following shows **Page 2** for trunk group 26. All parameters shown are default values, except for the **Preferred Minimum Session Refresh Interval**, which has been changed from the default value 600 to 900 to avoid unnecessary SIP messaging with Cisco UCME to negotiate to a higher refresh interval during call establishment.

```
add trunk-group 26                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900


                                     Delay Call Setup When Accessed Via IGAR? n
```

The following shows **Page 3** for trunk group 26. All parameters shown are at default values, with the exception of the bold fields, which optionally allow an Avaya-configured display string to appear on display-equipped telephones in the event that an anonymous or restricted incoming call is received from this trunk group. (The replacement display strings can be configured on Page 9 of the "change system-features" form, not shown). In the sample configuration, a "private" numbering format is used.

```
add trunk-group 26                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                Modify Tandem Calling Number: no


 Show ANSWERED BY on Display? y
```

The following shows **Page 4** for trunk group 26. All parameters shown are at default values, with the exception of the **Telephone Event Payload Type** associated with DTMF signaling, which has been set to the value "101".

```
add trunk-group 26                                              Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 101


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
                             Enable Q-SIP? n
```

3.7 Private Numbering

The "change private-numbering" command may be used to define the format of numbers such as the "calling party number" to be sent to Cisco UCME. In the bolded row shown in the abridged output below, all calls originating from a 5-digit extension beginning with 345 (i.e., 345xx) will not have any number prefixed, but rather a 5 digit calling party number will be sent in the SIP "From" and "P-Asserted-Identity" headers. In the sample configuration, this allows the Avaya user's five digit telephone extension to appear on the display of the Cisco telephones. In a production environment, other rows in this table may be used to ensure that an appropriate calling party number is sent for calls routed via trunks to the PSTN.

```
change private-numbering 0                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  345              26                          5     Total Administered: 1
                                                          Maximum Entries: 540
```

3.8 Uniform Dial Plan

The Uniform Dial Plan (UDP) is configured such that calls matching the 777xx extension range of Cisco telephones are part of the overall UDP configuration. The following screen shows a sample UDP configuration using the "change uniform-dialplan 7" command. When a user dials a 5 digit extension beginning with 777 (i.e., 777xx), the call will use Automated Alternate Routing (AAR) for further analysis. Although not shown, please note that 777 needs to be added to the Dial Plan Analysis table prior to configuring this form.

```
change uniform-dialplan 7                                           Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                  Percent Full: 0
  Matching                     Insert              Node
  Pattern       Len Del        Digits     Net Conv Num
 777             5   0                    aar   n
```

3.9 AAR Analysis

The AAR Analysis table is configured such that calls matching the 777xx extension range of Cisco telephones are routed to **Route Pattern** "26", as shown below.

```
change aar analysis 777                                         Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern     Type   Num   Reqd
    777                   5    5      26        unku         n
```

3.10 Route Pattern

Route pattern 26 is configured to include trunk group 26, the SIP trunk group to Session Manager, as shown below. Configure this route pattern to route calls to trunk group number "26" configured in **Section 3.6.2.** Assign the lowest **FRL** (facility restriction level) to allow all callers to use this route pattern. For **LAR** in row number (1) corresponding to the first trunk group entry, enter "next". This will ensure that for calls (SIP INVITEs) for which Communication Manager receives no response, the shorter **Alternate Route Timer** will be used instead of the much longer **Session Establishment Timer**, minimizing the time before the caller hears reorder tone. See **Section 3.6.1** for these parameters.

```
change route-pattern 26                                         Page   1 of   3
                    Pattern Number: 26  Pattern Name: To ASM
                             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: 26   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                        next
```

3.11 Save Translations

Configuration of Communication Manager is complete. Use the "save Translations" command to save these changes.

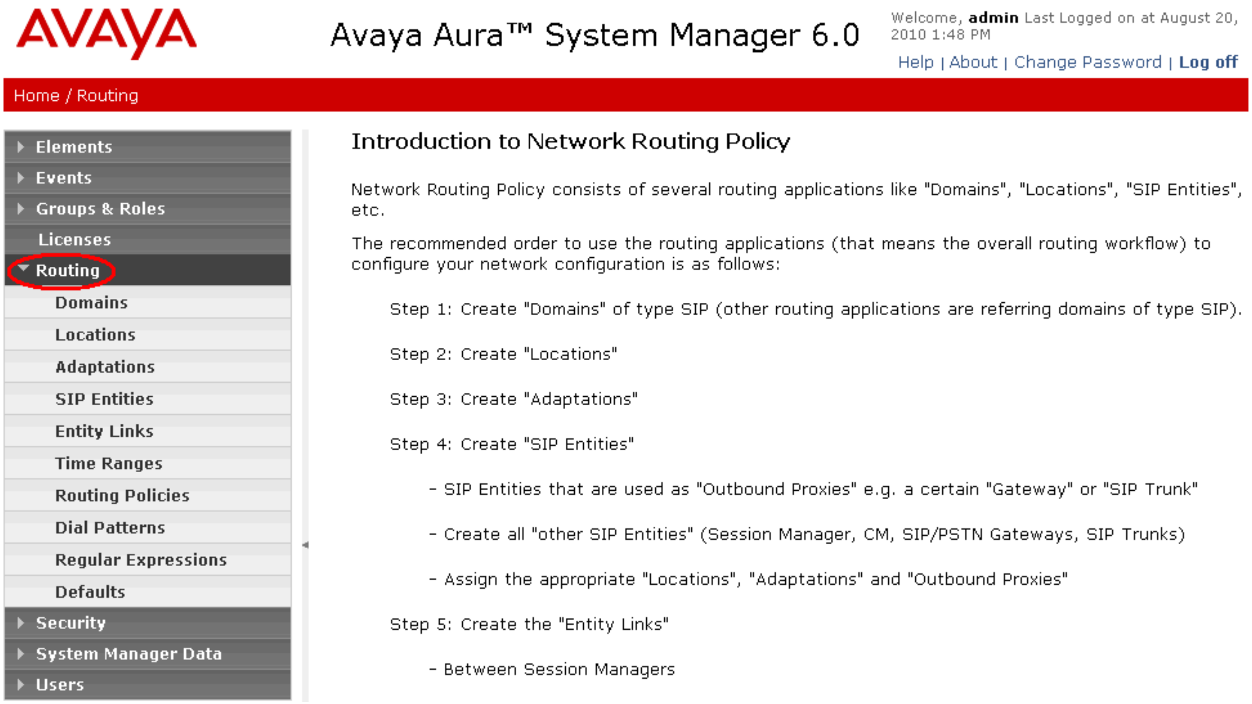
4 Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. For further info on Session Manager, see **[1-3]**. The configuration procedures include the following areas:

- SIP Domains – the domains for which Session Manager is authoritative for routing SIP calls
- Locations – the logical or physical location of a SIP entity, which can be used for location-based routing or bandwidth management and call admission control
- Adaptations – SIP protocol adaptations (e.g., SIP header manipulations) can be used to improve and simplify interoperability with other SIP entities. Digit conversion adaptations can be used to modify digit strings on ingress/egress to Session Manager to normalize and simplify configuration of a common dial plan among systems that may have disparate dial plans
- SIP Entities – SIP entities correspond to the SIP telephony systems and Session Manager instances.
- Entity Links - define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Time Ranges - allow time-based criteria for call routing
- Routing Policies - configurable call routing between the SIP Entities
- Dial Patterns – configurable criteria for call routing (e.g., called party number pattern matching) and routing policies to be used when criteria are met
- Configure Session Manager
- Add Communication Manager as an Evolution Server
- Add Users for SIP Telephones

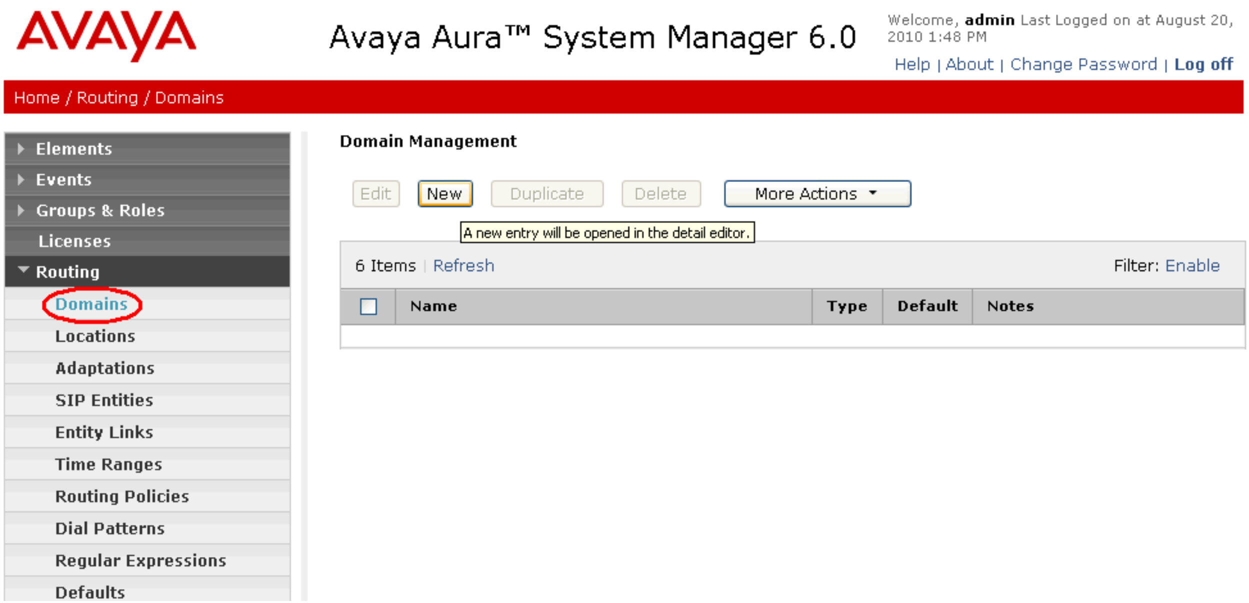
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “http://<ip-address>/SMGR, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice.

The menu shown below is displayed. Expand the **Routing** Link on the left side as shown. The sub-menus displayed in the left column below will be used to configure all but the last three items mentioned earlier (**Sections 4.1** through **4.8**).

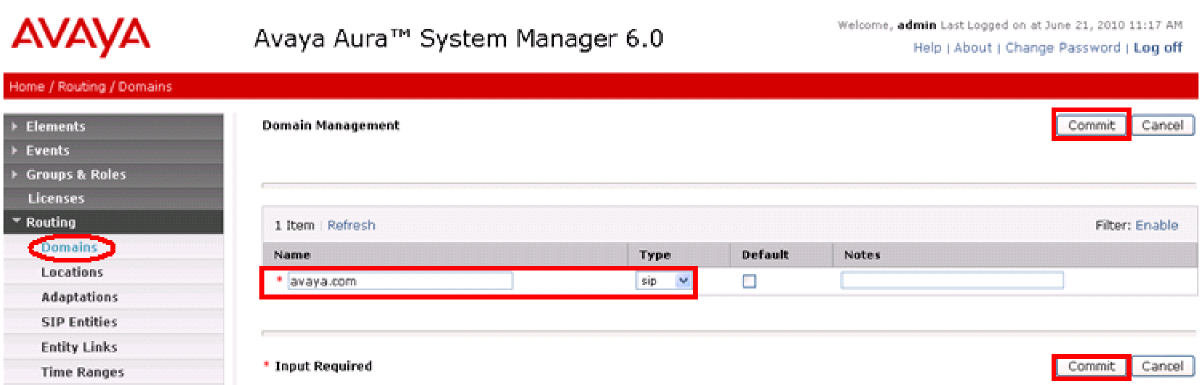


4.1 Configure the SIP Domain

To add the SIP domain for which the communications infrastructure will be authoritative, select **Routing → Domains** on the left as shown below.



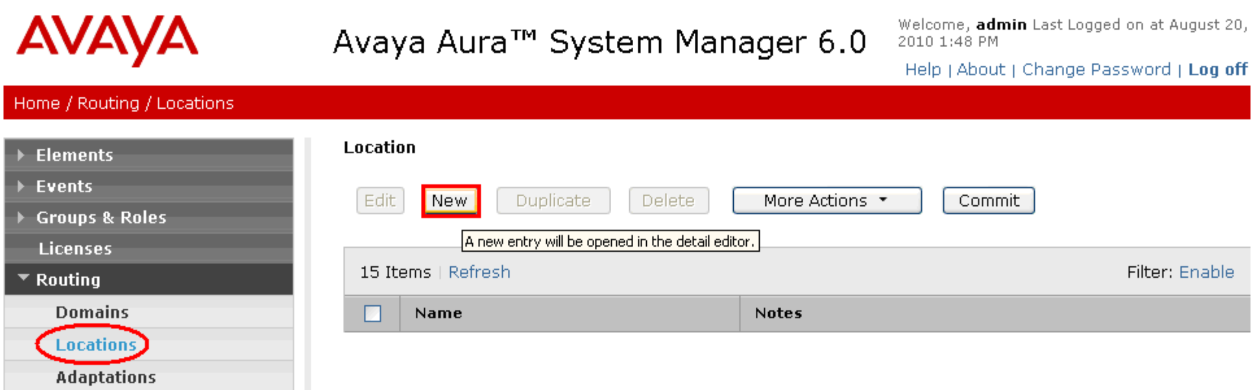
Click the **New** button. On the screen shown below, enter the authoritative domain name (e.g., "avaya.com") in the **Name** field. Optionally, enter descriptive text in the **Notes** field. Click the **Commit** button.



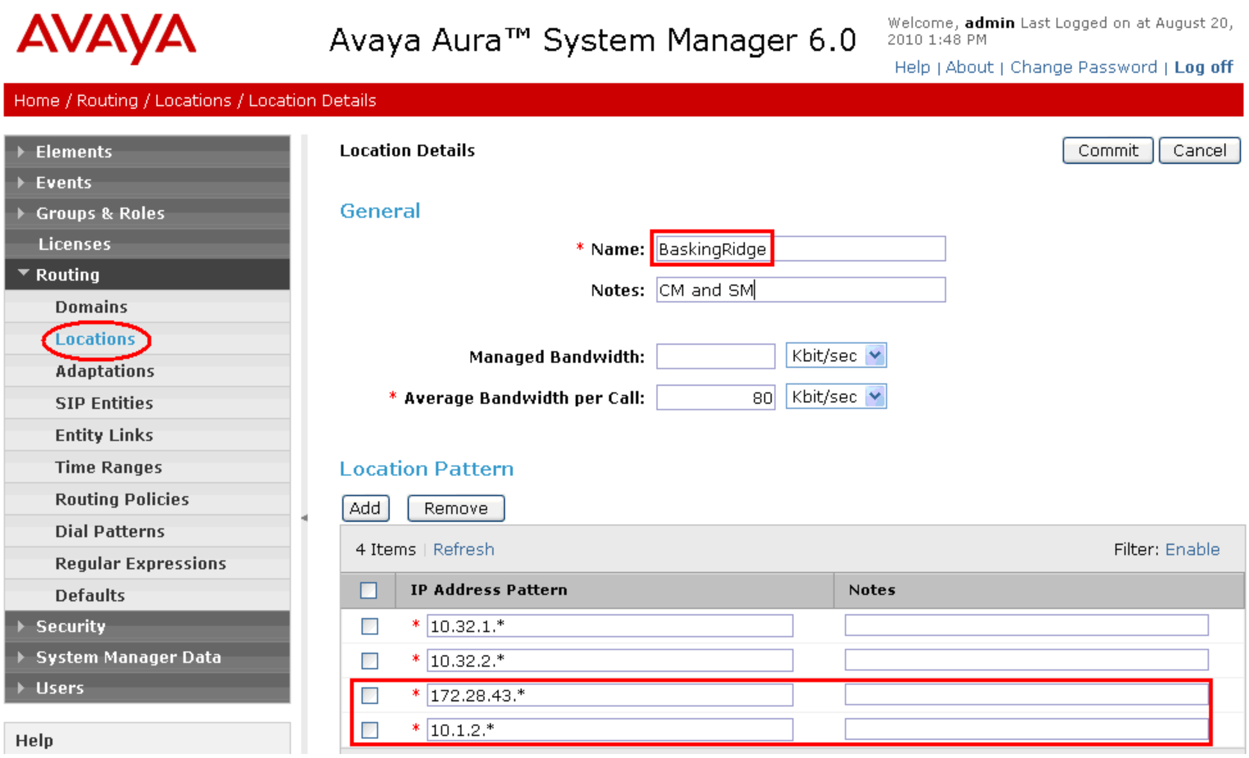
4.2 Configure Locations

Locations can be used to identify logical or physical locations where SIP entities reside. If desired, the location of the originator of a call can be used as a routing criterion or for bandwidth management purposes. The screens associated with locations are illustrated below, although routing decisions in the sample configuration are not determined by the location, and bandwidth management techniques are not illustrated.

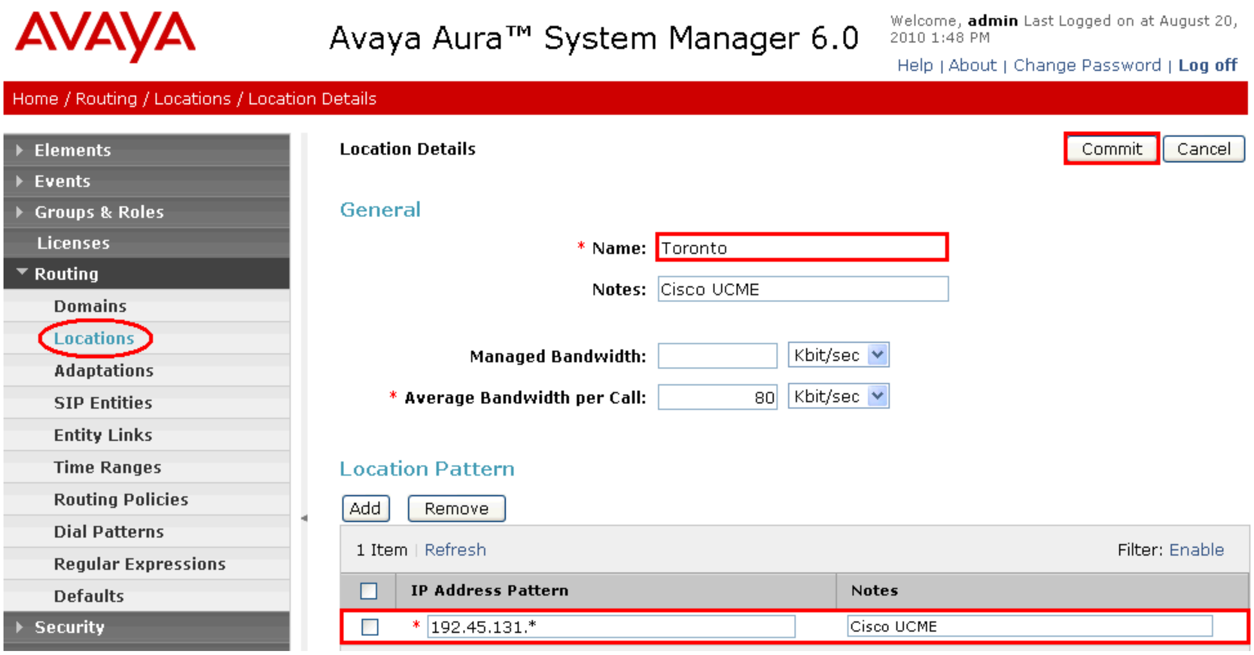
To configure locations, select **Routing → Locations**, as shown below. To add a new location, click **New**, or select a location from the list of existing locations.



The following screen shows the location whose **Name** is “BaskingRidge”. In the sample configuration, Communication Manager and Session Manager are configured for the “BaskingRidge” location. The **IP Address Pattern** “10.1.2.*” corresponds to IP Addresses used for Session Manager, and “172.28.43.*” corresponds to IP Addresses used for Communication Manager. Click **Commit** to save each Location definition.



The following screen shows the location whose **Name** is “Toronto”. In the sample configuration, Cisco UCME is configured for the “Toronto” location. As shown in **Figure 1**, the **IP Address Pattern** “192.45.131.*” corresponds to the IP Addresses used for Cisco UCME and the associated Cisco IP Telephones.



4.3 Configure Adaptations

Two Adaptations need to be created: One for calls from/to Communication Manager called "DigitConversionAdapter" and the other for calls from/to Cisco UCME called "CiscoAdapter".

The "DigitConversionAdapter" will adapt SIP request and SIP response messages. It uses the following pieces of information to perform digit adaptation on various SIP headers:

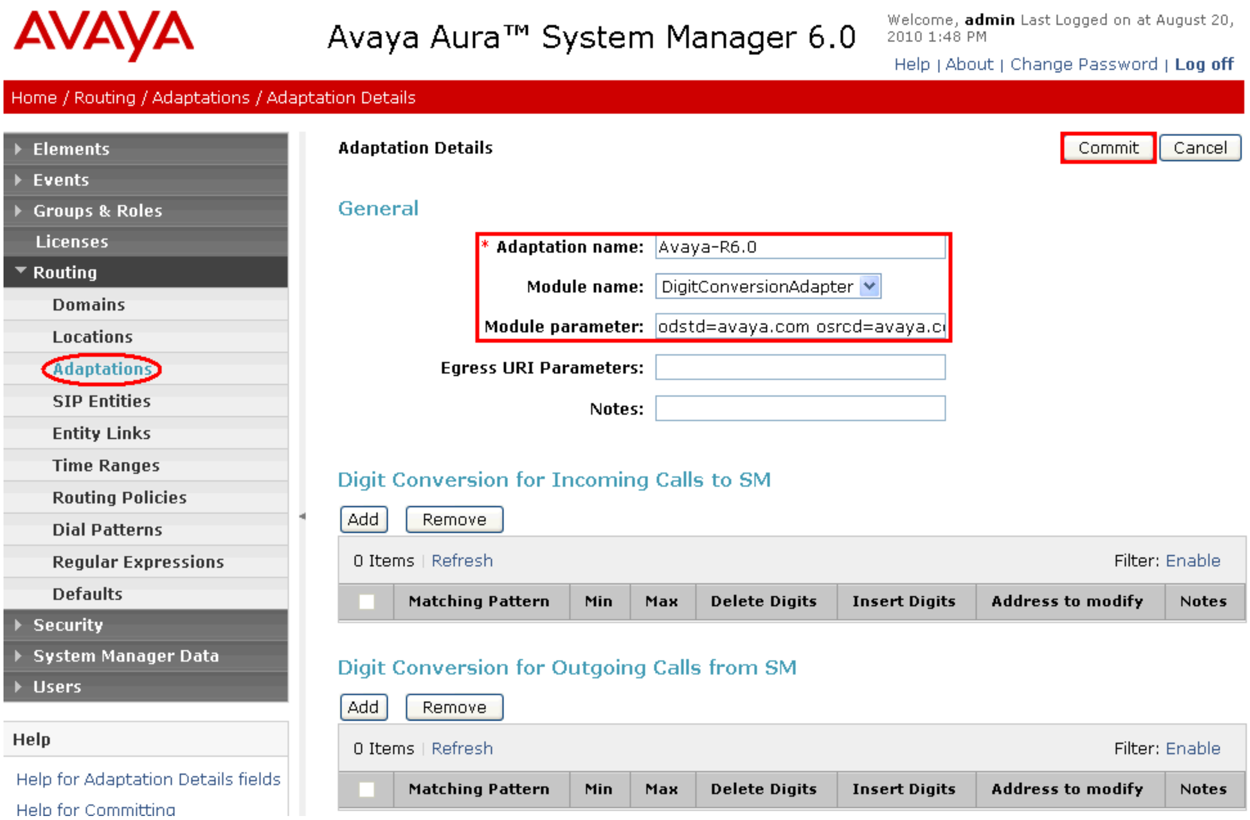
- Adaptation direction (incoming/ingress or outgoing/egress)
- Matching digit pattern and corresponding digits to remove/insert
- Domain name change for source components and destination components

The "CiscoAdapter" provides two basic header manipulations: converting between Diversion and History-Info headers and converting between P-Asserted-Id and Remote-Party-Id headers. The Diversion and Remote-Party-Id headers have not been accepted by the IETF. They are replaced by History-Info and P-Asserted-Identity respectively, but are still used in the Cisco products. The Cisco Adapter will also perform all the conversions available by the "DigitConversionAdapter".

For the Communication Manager adaptation, enter the following information.

Adaptation name	An informative name for the adaptation (e.g., “Avaya-R6.0”)
Module name	Select **DigitConversionAdapter**.
Module parameter	The parameter “odstd=avaya.com” specifies that the domain in the SIP Request-URI and NOTIFY/message-summary body of messages sent by Session Manager to that SIP Entity will be overridden with “avaya.com”. The parameter “osrcd=avaya.com” specifies that the domain in the P-Asserted-Identity header and the calling part of the History-Info header of messages sent by Session Manager will be overridden with “avaya.com”.

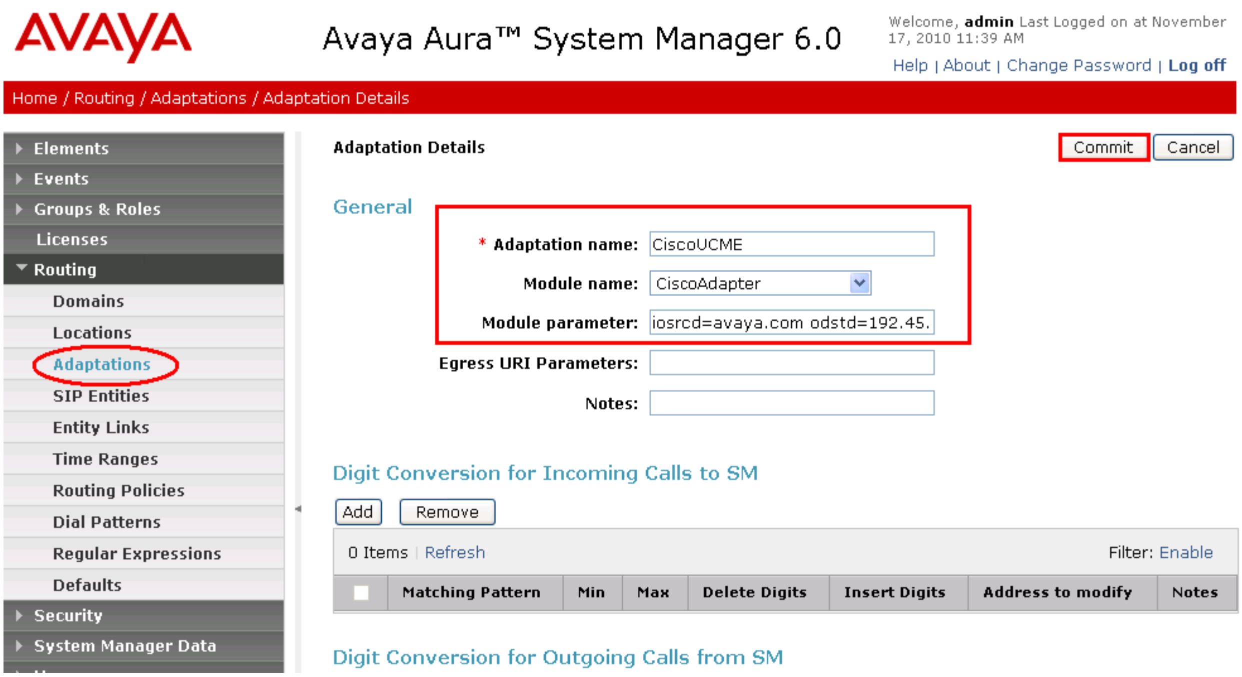
Since no digit conversions are required, the remaining fields can be left at their defaults.



For the Cisco UCME adaptation, enter the following information.

Adaptation name	"CiscoUCME", an informative name for the adaptation
Module name	Select **CiscoAdapter**.
Module parameter	Enter "iosrcd=avaya.com" to specify the Session Manager source SIP domain. Enter "odstd=192.45.131.1" where "192.45.131.1" is the IP address for Cisco UCME.

Since no digit conversions are required, the remaining fields can be left at their defaults.



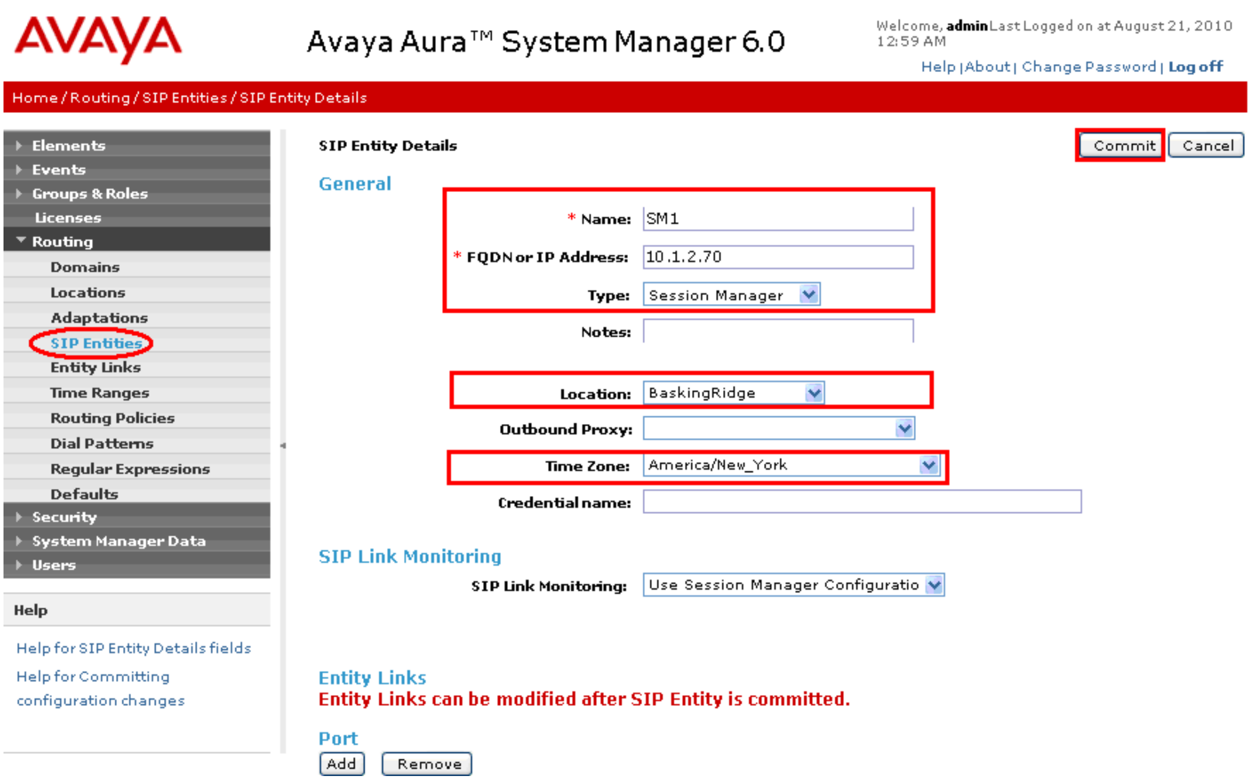
4.4 Configure SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration a SIP Entity is added for Session Manager and Cisco UCME. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The screen is displayed as shown on the next page. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the Session Manager or the signaling interface on the telephony system.
- **Type:** "Session Manager" for Session Manager, **"CM"** for Communication Manager, and **"Oth**er" f**o**r Cisco UCME.
- **Adaptation:** Select appropriate adaptation (Note: Not needed for SM1).
- **Location:** Select one of the locations defined in **Section 4.2**.
- **Time Zone:** Time zone for this location.

The following screen shows addition of Session Manager.



Under *Port* for the Session Manager Entity, click **Add**, and then edit the fields in the resulting new row as shown below:

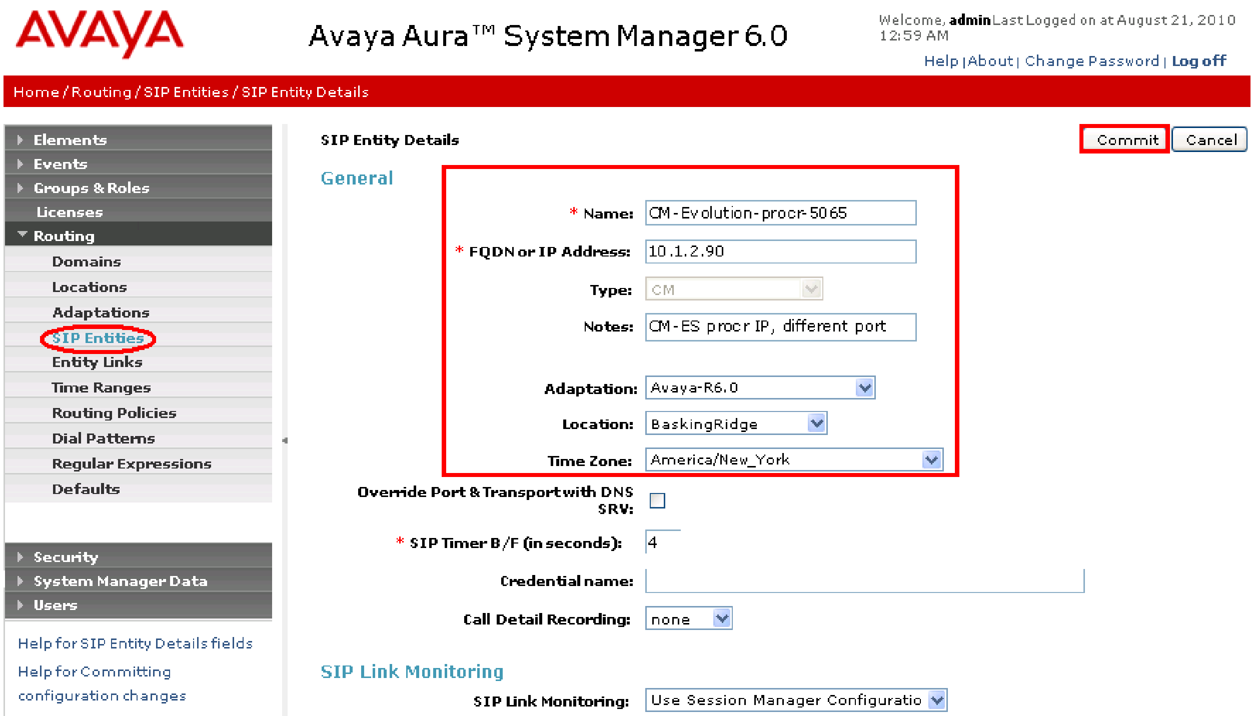
- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., "avaya.com").

4 Items | Refresh Filter: Enable

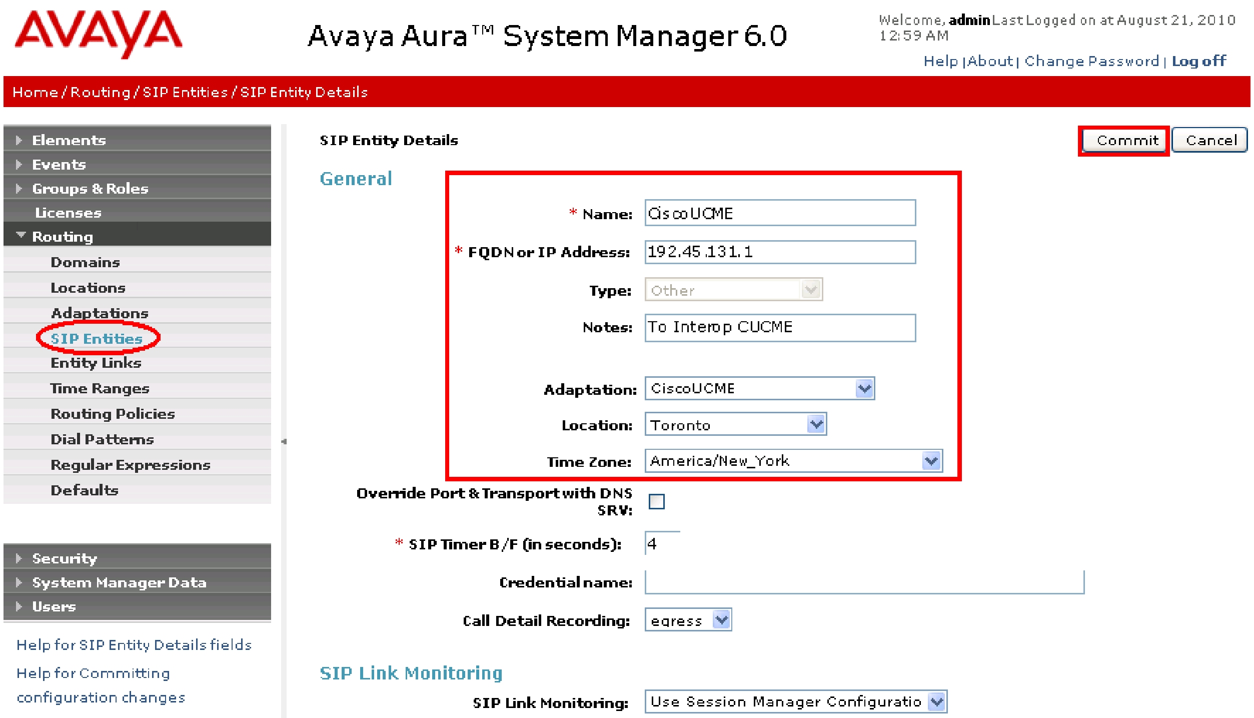
	Port	Protocol	Default Domain	Notes
☐	5060	TCP	avaya.com	
☐	5060	UDP	avaya.com	
☐	5061	TLS	avaya.com	
☐	5065	TLS	avaya.com	

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The following screen shows addition of Communication Manager. The IP address used is that of the Avaya S8800 server.



The following screen shows addition of Cisco UCME. The IP address used is that of the Cisco UCME ethernet interface.



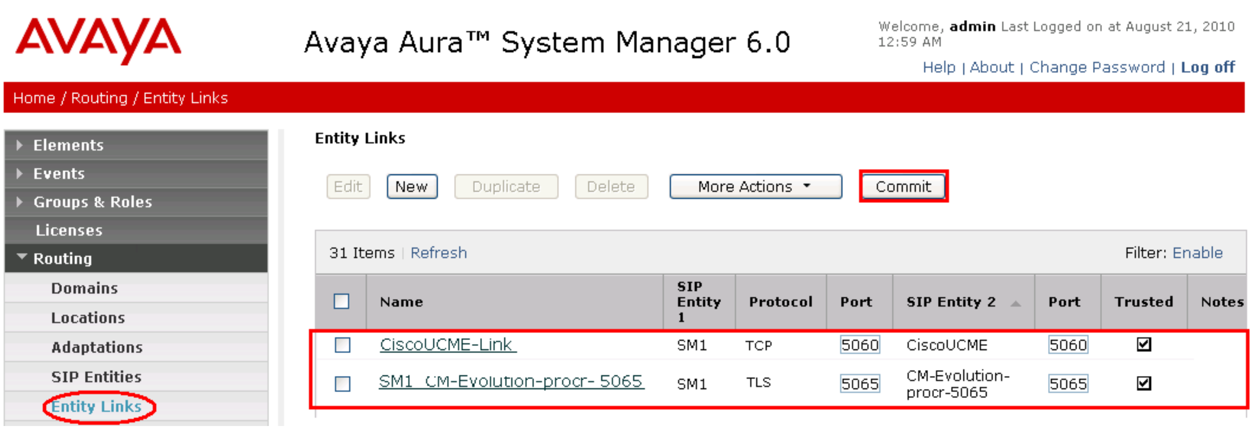
4.5 Configure Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Transport protocol for communication between entities.
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box. ***Note:*** *If this box is not checked, calls from the associated SIP Entity specified in* ***Section 4.4*** *will be denied.*

Click **Commit** to save each Entity Link definition. The following screen shows the result of adding the two Entity Links for Communication Manager and Cisco UCME.

Note: A third entity link between Cisco UCME and Session Manager was added using UDP port 5060. This entity link was needed because under certain call scenarios, Cisco UCME sent SIP traffic to Session Manager using UDP transport even though the Cisco UCME dial-peer configuration was set to use TCP transport.

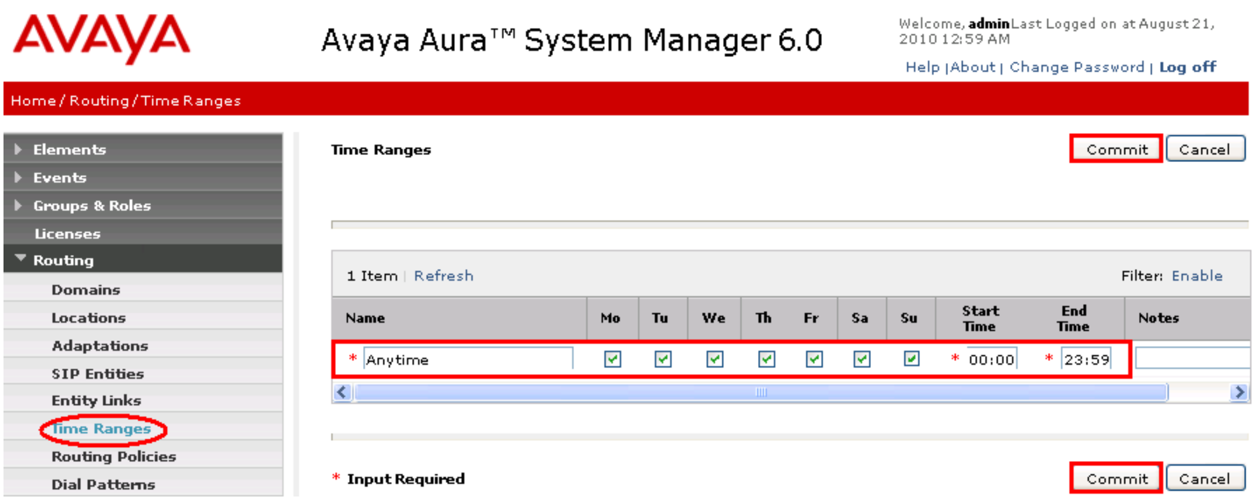


4.6 Configure Time Ranges

Before adding routing policies (see **Section 4.7**), time ranges must be defined during which the policies will be active. In the sample configuration, one policy was defined that would allow routing to occur at anytime. To add this time range, select **Time Ranges** on the left and click on the **New** button on the right. Fill in the following:

- **Name:** A descriptive name (e.g., "Anytime").
- **Mo** through **Su** Check the box under each of these headings
- **Start Time** Enter "00:00".
- **End Time** Enter "23:59"

Click **Commit** to save this time range.



4.7 Configure Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 4.4**. Two routing policies must be added – one for Communication Manager and one for Cisco UCME. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The screen shown on the next page is displayed. Fill in the following:

Under *General*:

- Enter a descriptive name in **Name**.

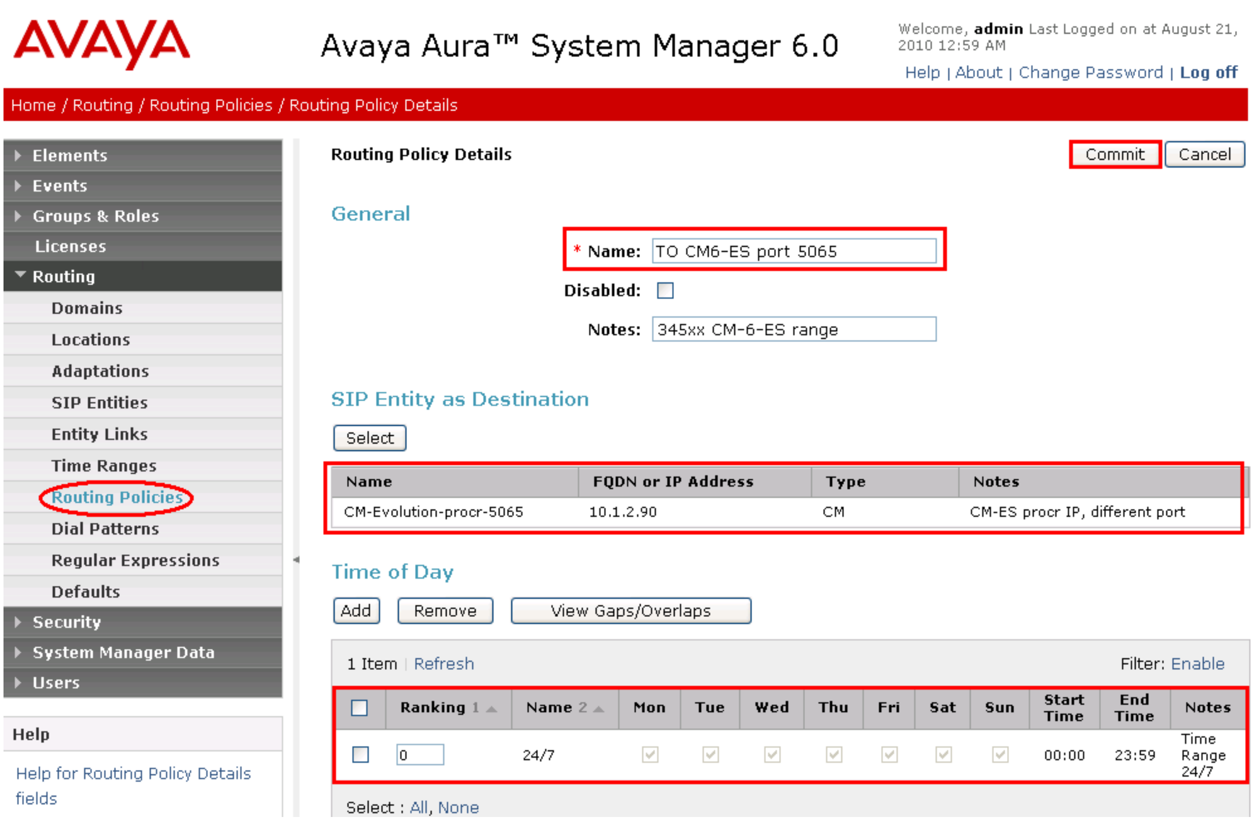
Under *SIP Entity as Destination*:

- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

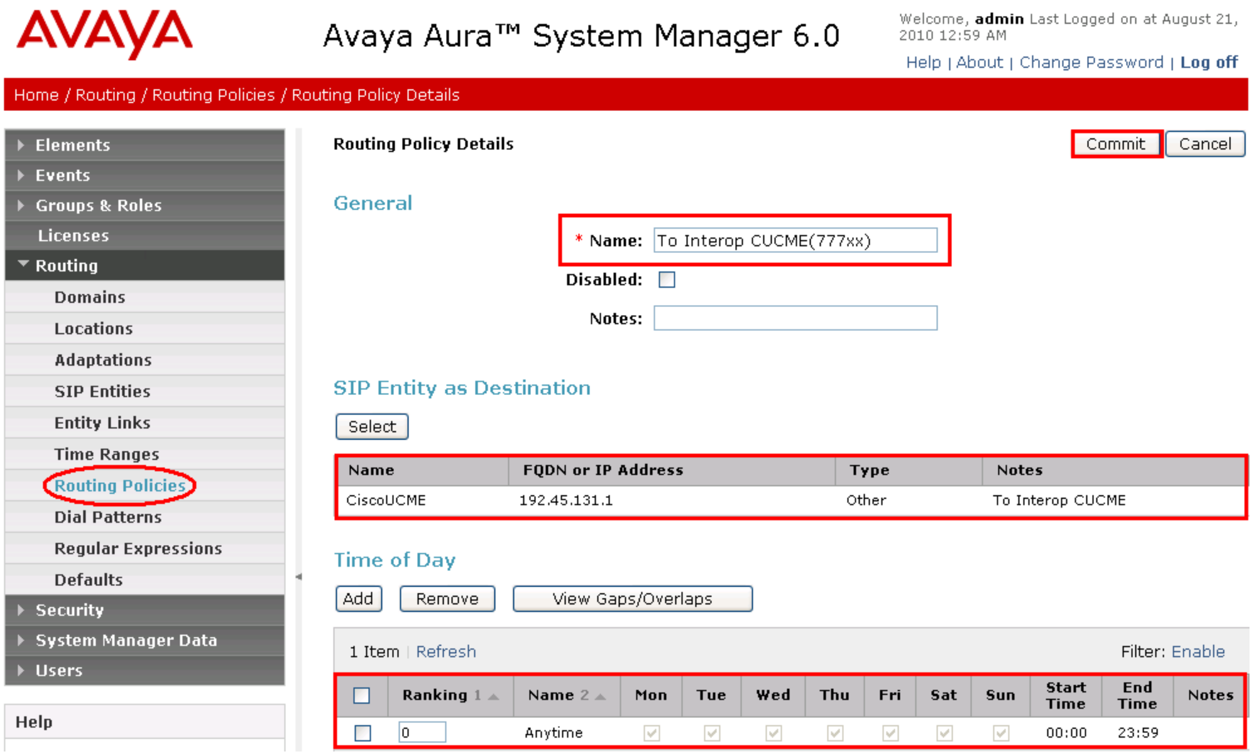
Under *Time of Day*:

- Click **Add**, and select the time range configured in **Section 4.6**.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for Cisco UCME.



4.8 Configure Dial Patterns

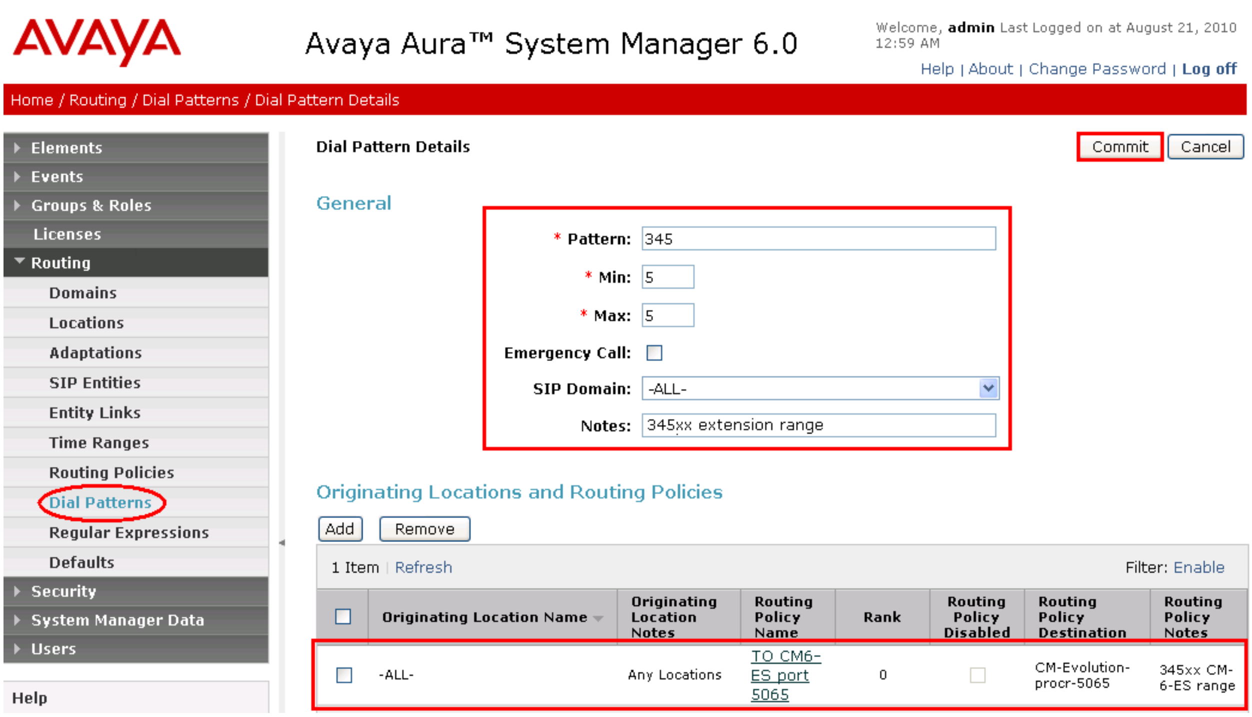
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with "345" reside on Communication Manager, and 5-digit extensions beginning with "777" reside on Cisco UCME. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen later in this section, which corresponds to the dial pattern for routing calls to Communication Manager:

Under *General*:

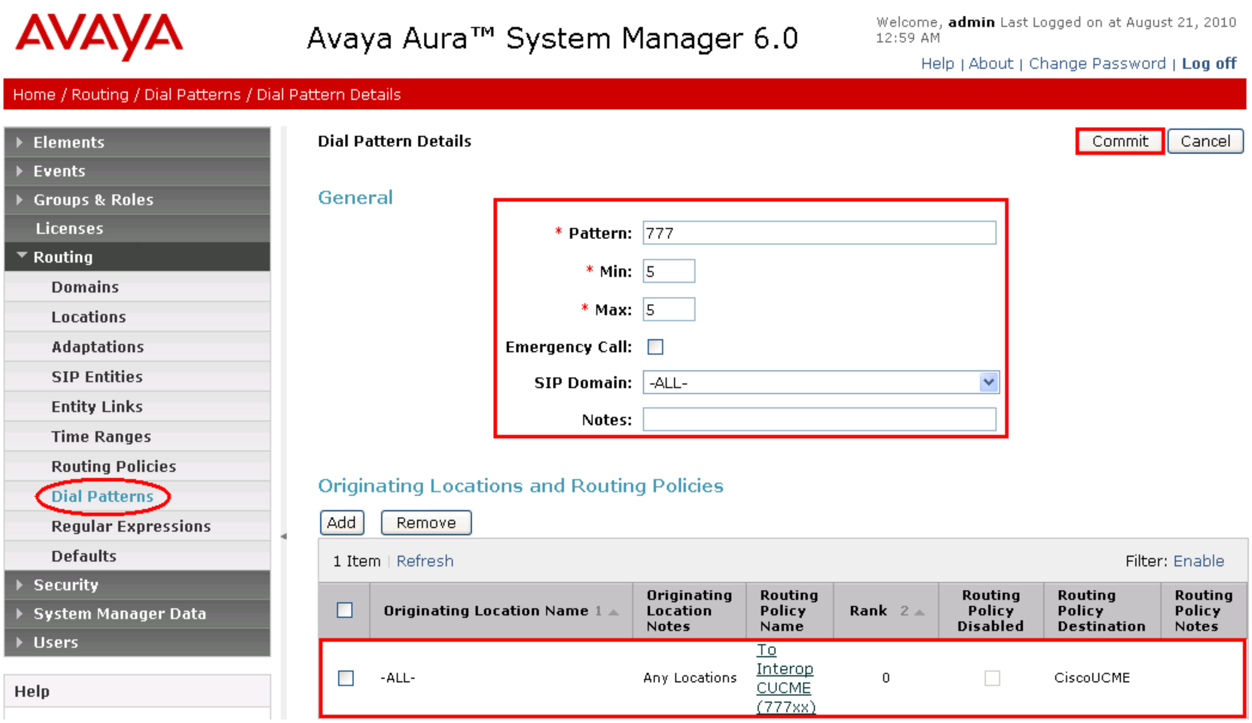
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** Select **-ALL-**
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for Communication Manager.



The following screen shows the dial pattern definitions for Cisco UCME.



4.9 Configure Avaya Aura® Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Navigate to **Elements → Session Manager → Session Manager Administration** in the panel menu on the left. Then on the right, under **Session Manager Instances**, click **New** (not shown) and fill in the fields as described below:

Under **General:**

- **SIP Entity Name** — Select the name of the SIP Entity added for Session Manager, here **SM1**
- **Description** — Descriptive comment (optional)
- **Management Access Point Host Name/IP** — Enter the IP address of the Session Manager management interface

Under **Security Module:**

- **Network Mask** — Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** — Enter the IP address of the default gateway for Session Manager
- **SIP Entity IP Address** — Will be automatically filled in based on the selected **SIP Entity Name**.

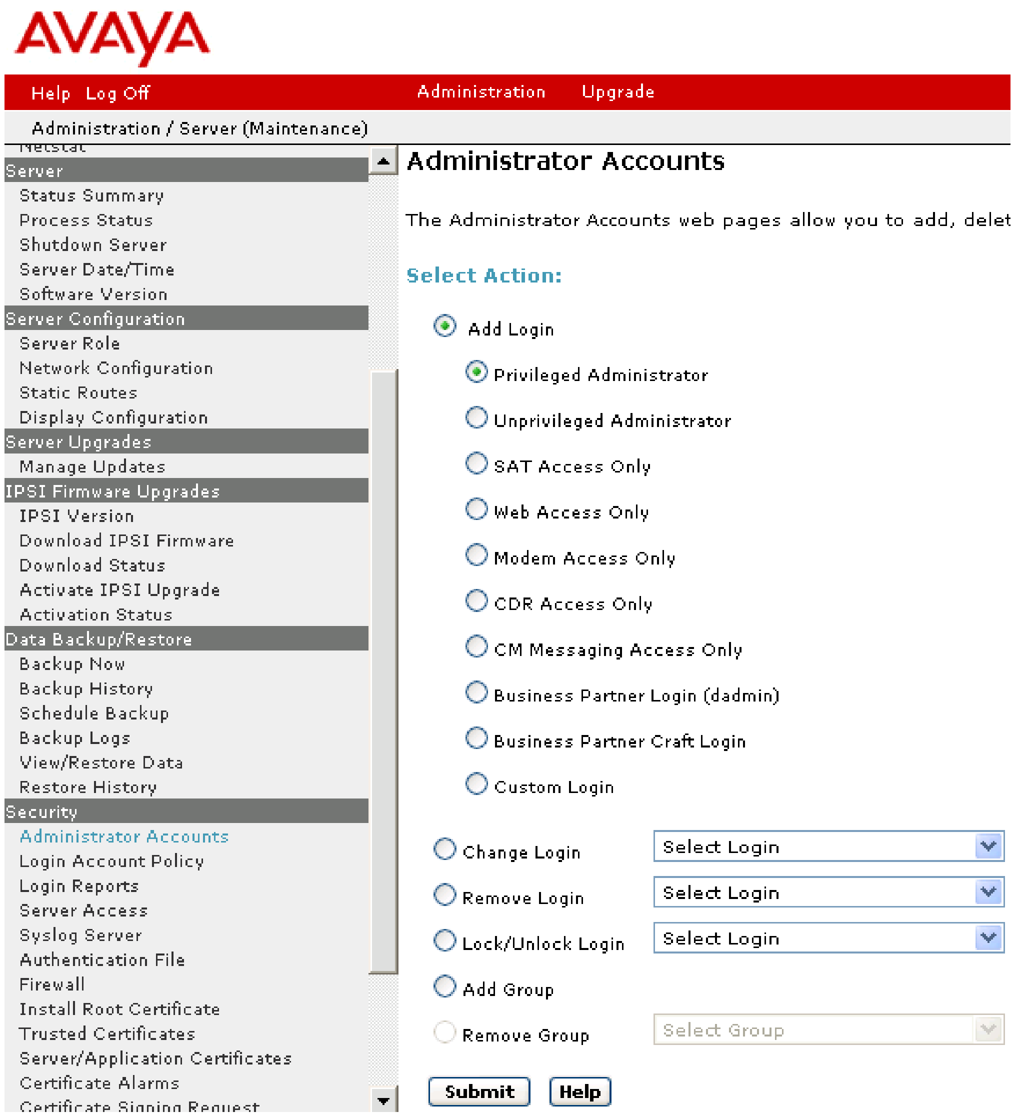
Use default values for the remaining fields. Click **Commit** to add this Session Manager. The following screen shows the resulting Session Manager.

4.10 Add Avaya Aura® Communication Manager as an Evolution Server

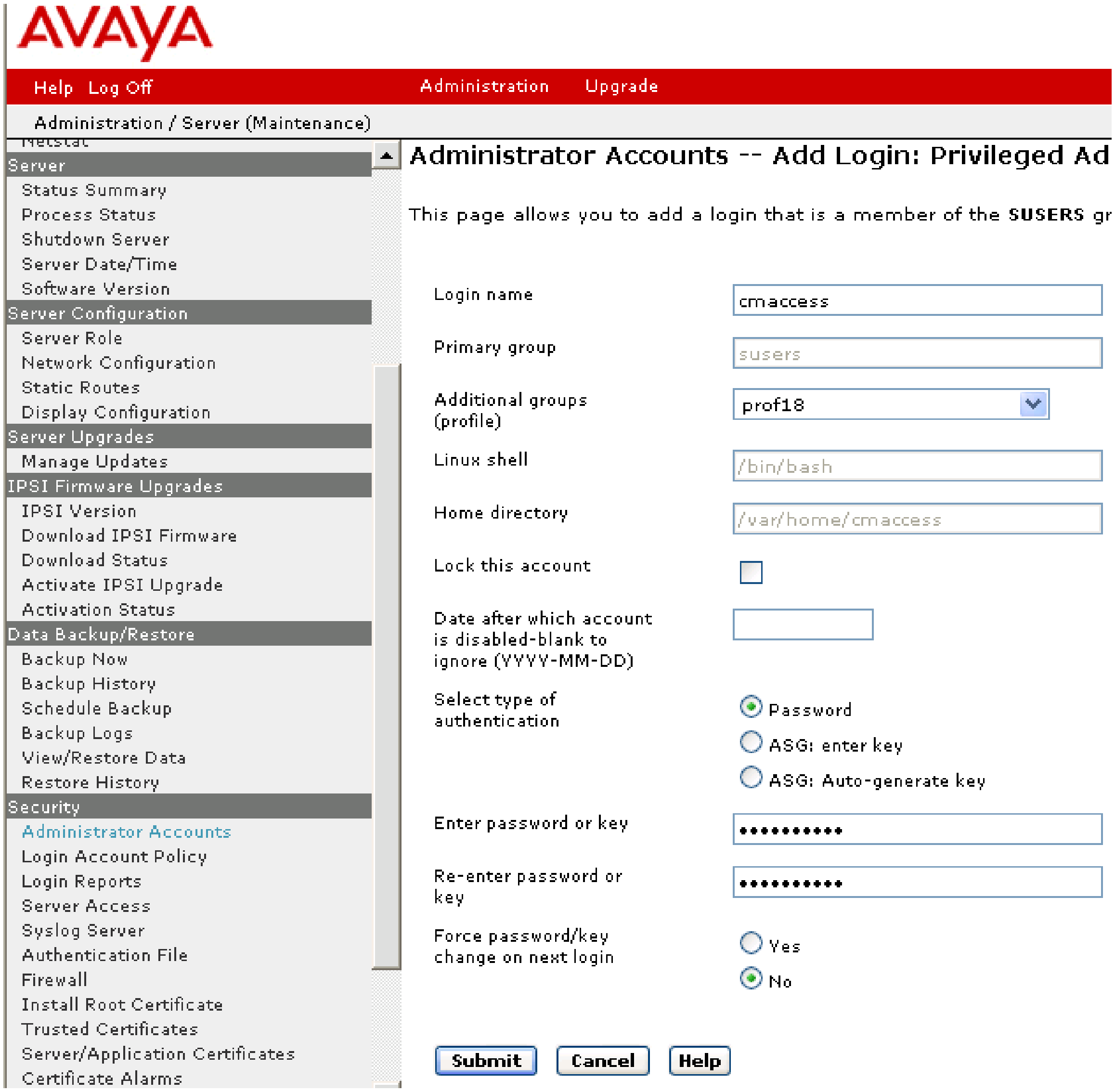
In order for Communication Manager to provide configuration and Evolution Server support to telephones, Communication Manager must be added as an application in Session Manager. This comprises a two step procedure. First, an access login must be configured on Communication Manager for the purpose of data synchronization with System Manager. Then the Application Element for that Communication Manager can be added via System Manager.

4.10.1 Create a Login on the Avaya Aura®Communication Manager Server

Use a web browser to access the Communication Manager maintenance web interface, and navigate to **Security → Administrator Accounts** on the left menu. Select **Add Login** and **Privileged Administrator**, as shown below. Click on **Submit**.



On the next screen, enter a **Login name** and a password in the **Enter password or key** and **Re-enter password or key** fields, and click **Submit**.

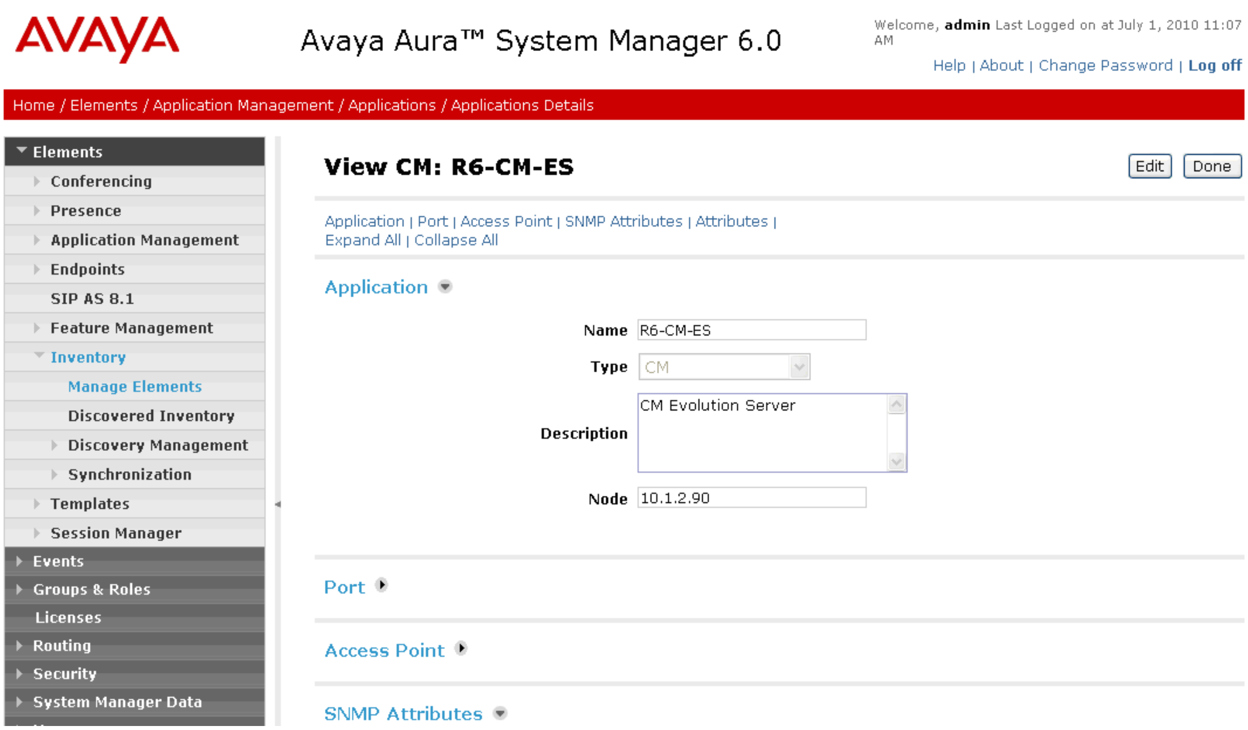


4.10.2 Create an Application Element

Return to System Manager and select **Elements** → **Inventory** → **Manage Elements** on the left. Click on **New** (not shown). Enter the following fields and use defaults for the remaining fields:

Under **Application**:

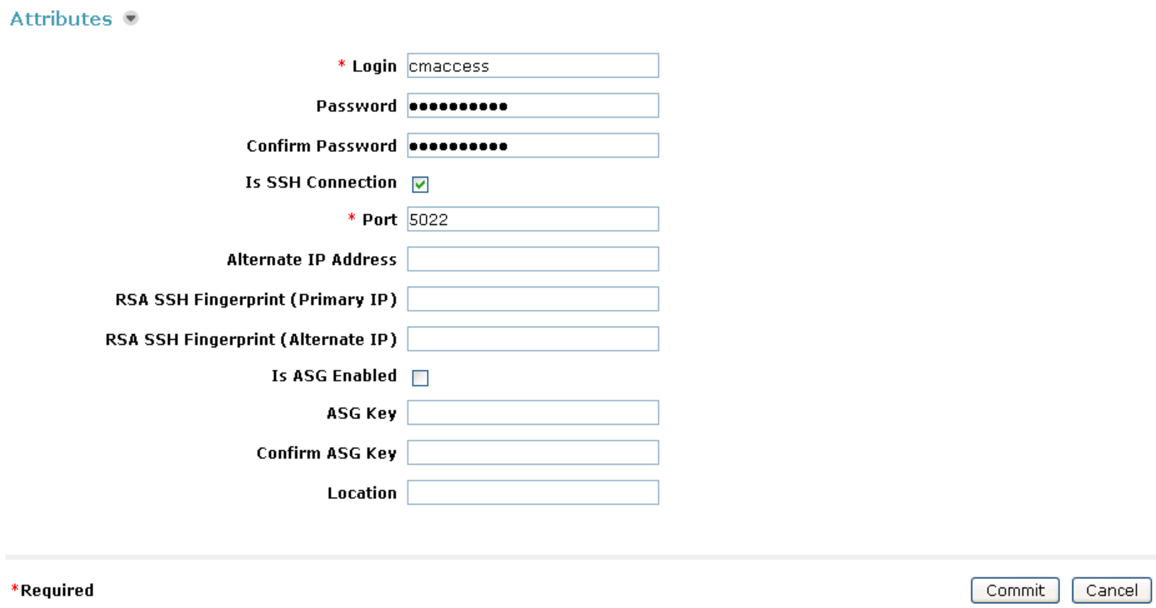
- **Name** A descriptive name
- **Type** Select **CM**
- **Node** Enter the IP address for Communication Manager SAT access



Under **Attributes**:

- **Login** Login created in the previous section
- **Password** Password created in the previous section
- **Confirm Password** Password created in the previous section

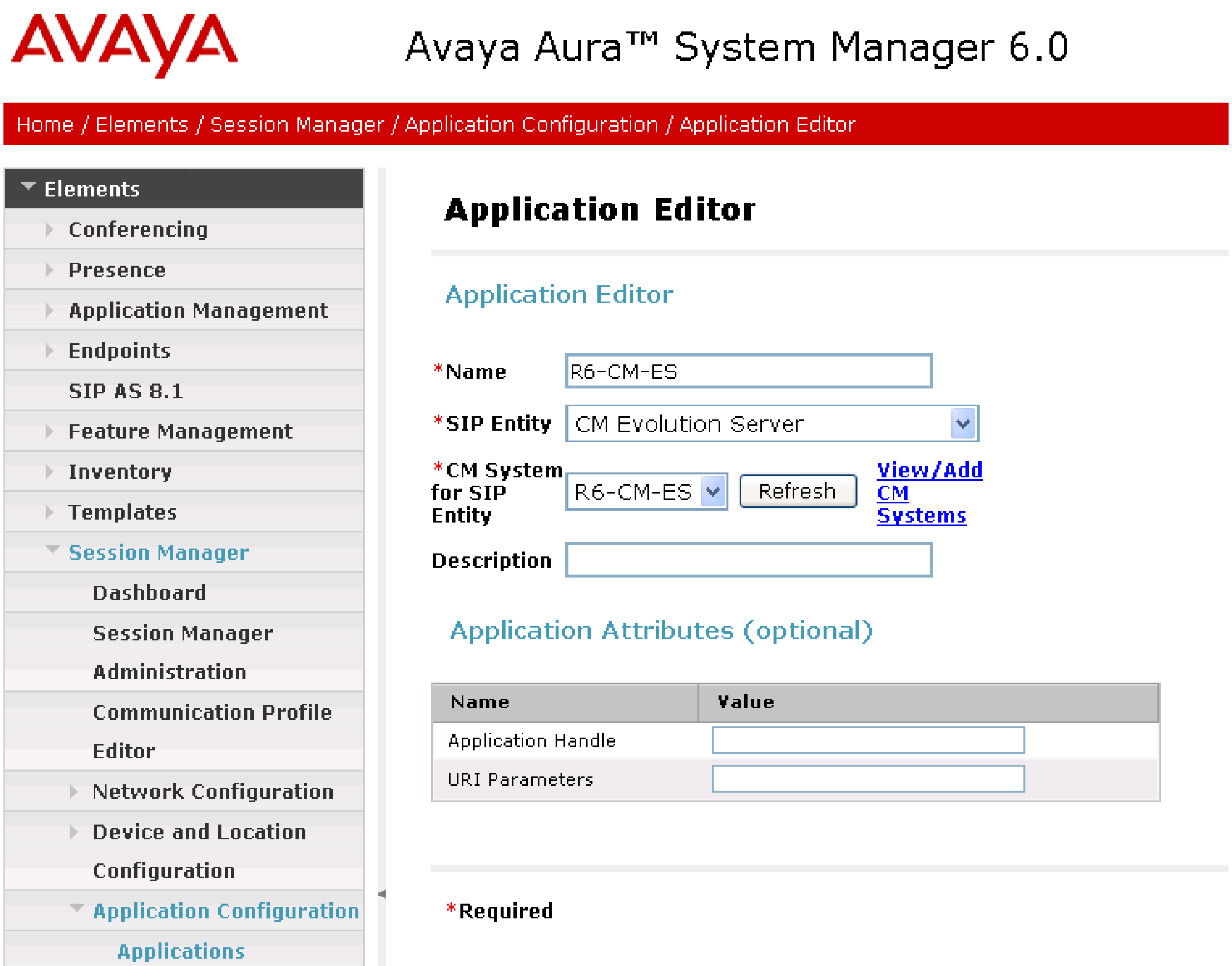
Click on **Commit** to save.



4.10.3 Create an Application

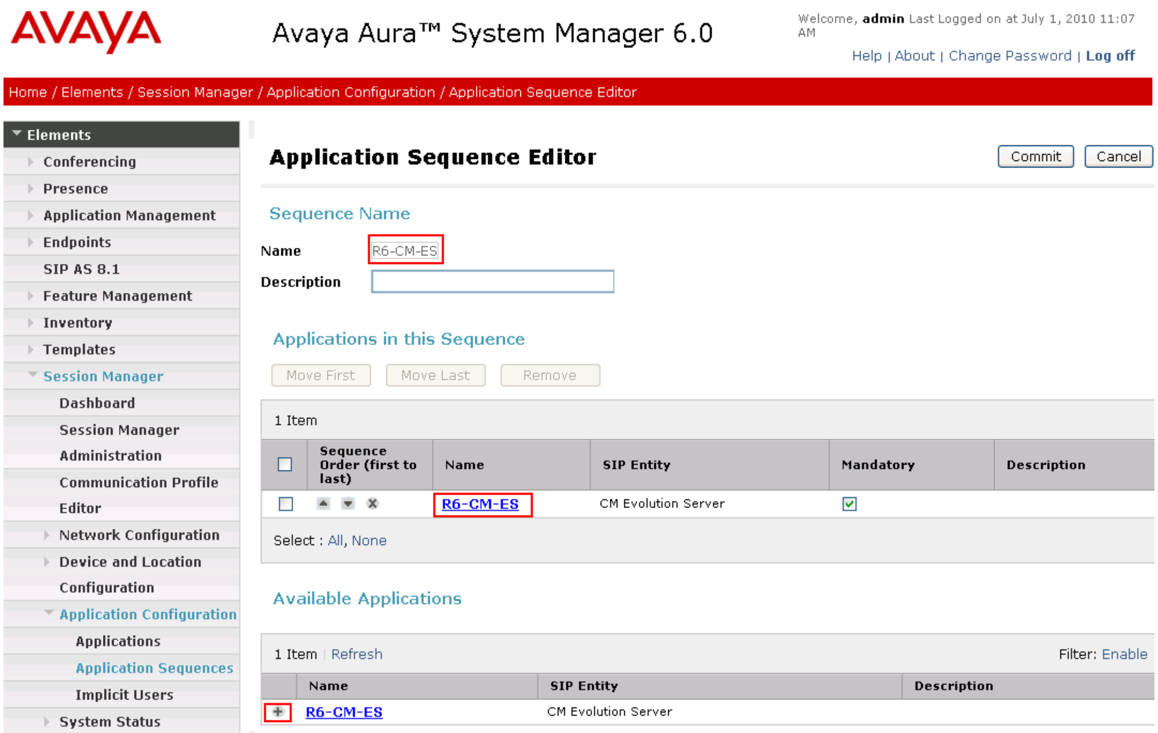
Select **Elements → Session Manager → Application Configuration → Applications** on the left. Click on **New** (not shown). Enter following fields and use defaults for the remaining fields and click on **Commit** to save.

- **Name** — A descriptive name
- **SIP Entity** — Select the CM SIP Entity defined in **Section 4.4**
- **CM System for SIP Entity** — Select the CM System for SIP Entity defined in **Section 4.10.2**



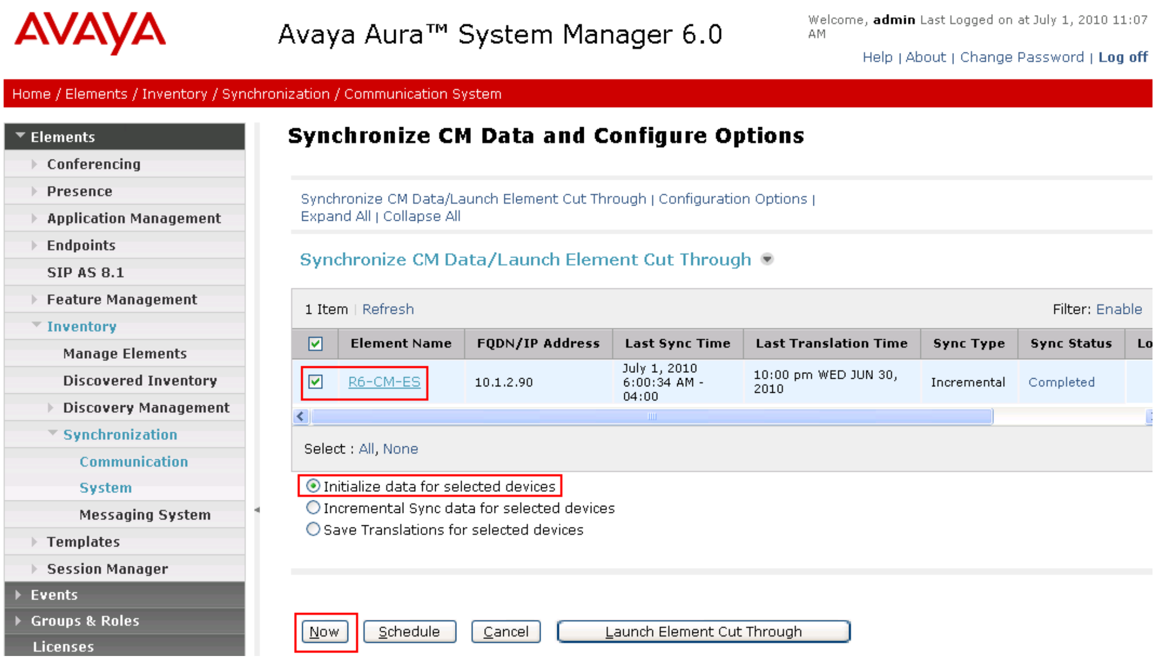
4.10.4 Create an Application Sequence

Select **Elements → Session Manager → Application Configuration → Application Sequences** on the left. Click on **New** (not shown). Enter a descriptive **Name**. Click on the + sign next to the appropriate **Available Applications** and they will move up to the **Applications in this Sequence** section. Click on **Commit** to save.



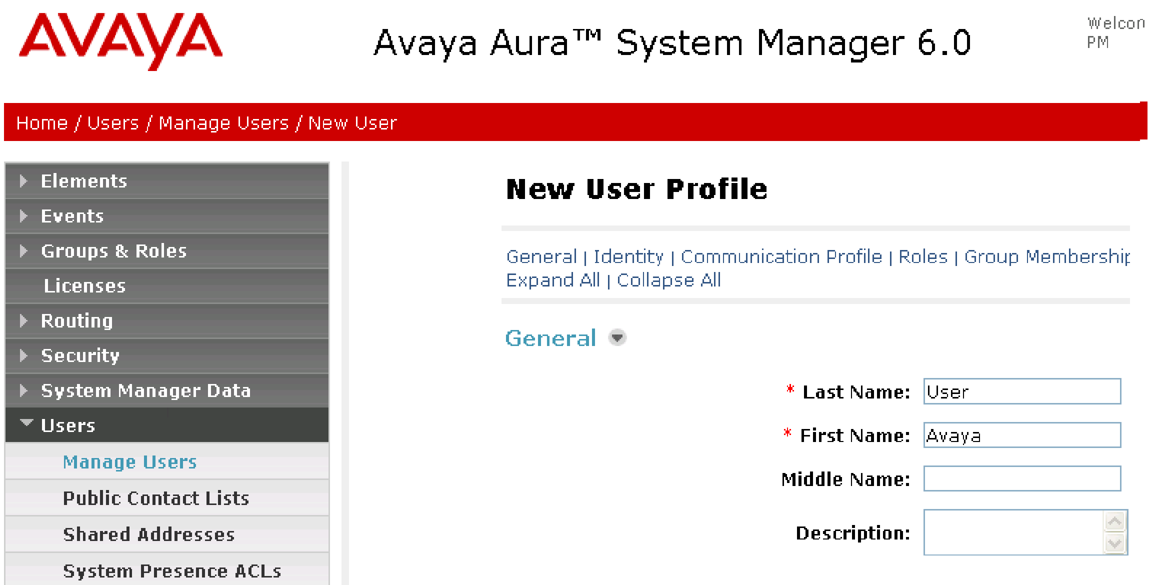
4.10.5 Synchronize Avaya Aura® Communication Manager Data

Select **Elements → Inventory → Synchronization → Communication System** on the left. Select the appropriate **Element Name**. Select **Initialize data for selected devices**. Then click on **Now**. This may take some time. Use the menus on the left under **System Manager Data → Scheduler** to determine when the task is complete.



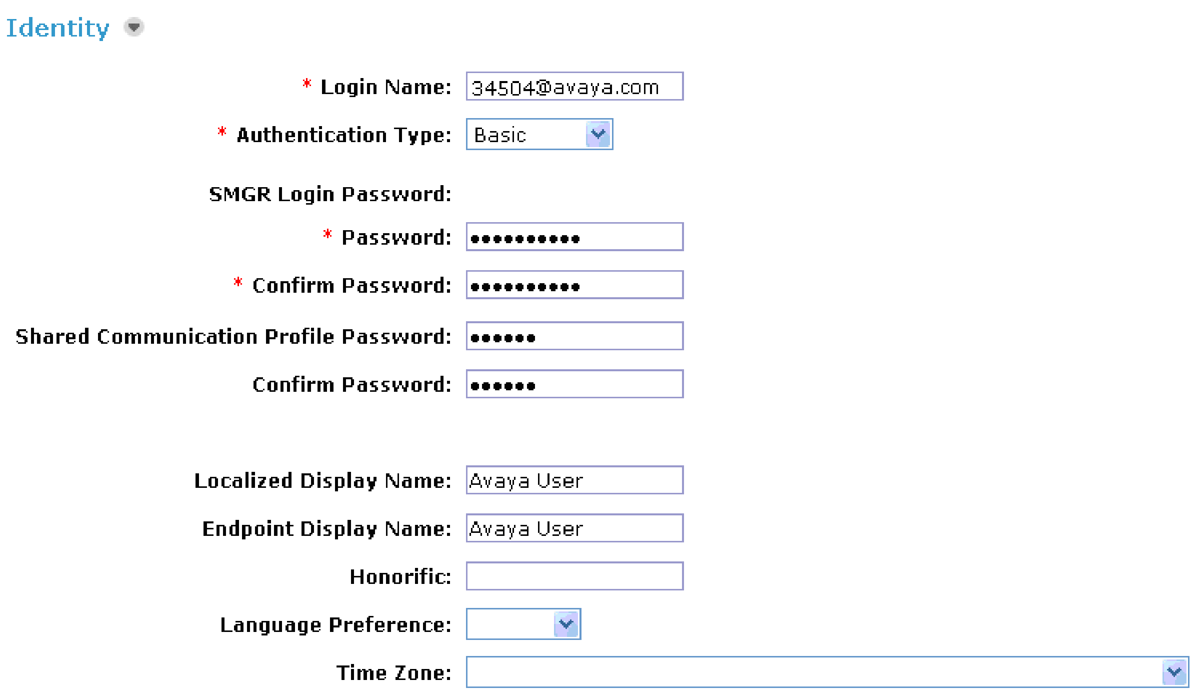
4.11 Add Users for SIP Telephones

SIP telephone users must be added to Session Manager. Select **Users → Manage Users** on the left. Then click on **New** (not shown). Enter a **First Name** and **Last Name**.



Under **Identity**:

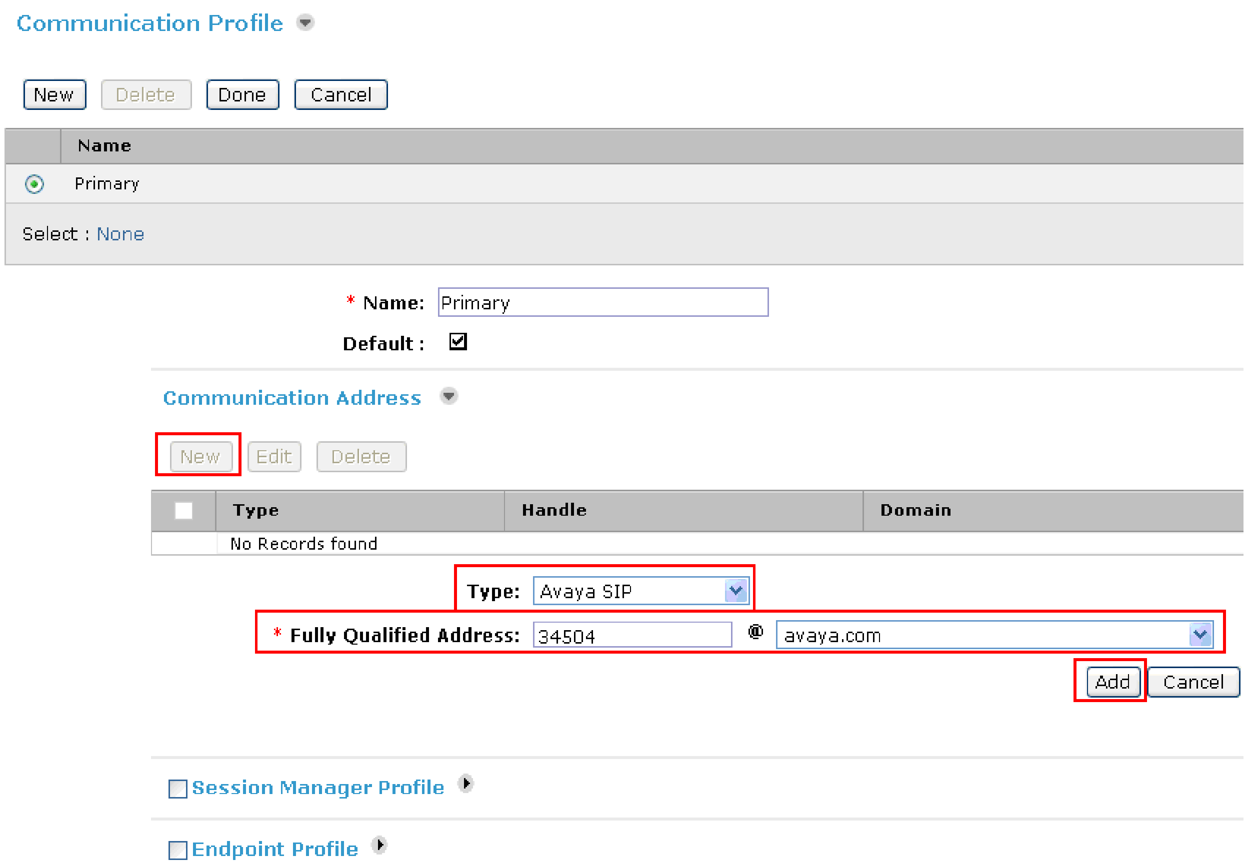
- **Login Name** The desired phone extension number@domain.com where domain was defined in **Section 4.1**
- **Password** Password for user to log into System Manager (SMGR)
- **Shared Communication Profile Password** Password to be entered by the user when logging into the phone.
- **Localized Display Name** The name to be used as calling party
- **Endpoint Display Name** The name to be used as calling party



Navigate to and click on **Communication Profile** section to expand. Then click on **Communication Address** to expand that section. Click on **New** and enter the following and defaults for the remaining fields:

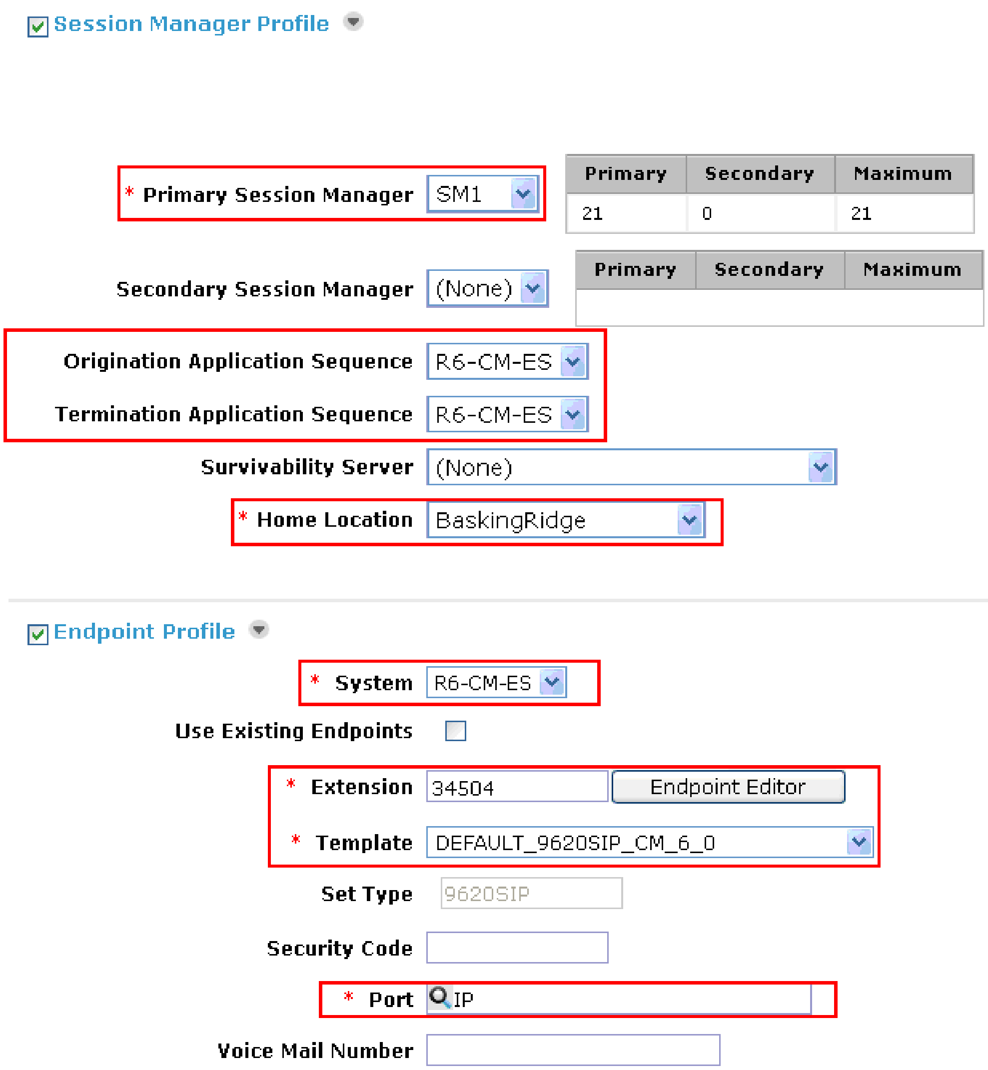
- **Type** Select **Avaya SIP**
- **Fully Qualified Address** Enter the extension number
- **@** Select the domain defined in **Section 4.1**

Click on **Add**.



Navigate to and click on **Session Manager Profile** to expand. Select the appropriate Session Manager server for **Primary Session Manager**. For **Origination Application Sequence** and **Termination Application Sequence** select the application sequence created in **Section 4.10.4**. Select the location defined in **Section 4.2** for **Home Location**. Click on **Endpoint Profile** to expand that section. Enter the following fields and use defaults for the remaining fields. Make sure to check the **Session Manager Profile** and **Endpoint Profile** checkboxes. Click on **Commit** to save (not shown).[2]

- **System** Select the CM Entity
- **Extension** Enter a desired extension number
- **Template** Select a telephone type template
- **Port** Select **IP**



[2] Note that when **Use Existing Endpoints** is not checked, Session Manager will automatically create station and off-pbx station-mapping forms in Communication Manager. This section should not be completed until the data synchronization task created in **Section 4.10.5** has completed.

5 Configure Cisco Unified Communications Manager Express

Cisco Unified Communications Manager Express (Cisco UCME) is a call-processing application in Cisco IOS software that enables Cisco routers to deliver key-system or hybrid Private Branch Exchange (PBX) functionality for enterprise branch offices or small businesses. It supports H.323 and SIP trunk operation to other IP PBX systems.

This section illustrates the relevant Cisco UCME configuration for SIP trunking to Communication Manager via Session Manager. A VoIP dial peer “trunk” is configured in the UCME to connect to the Session Manager for communication with Communication Manager.

With the Cisco IOS 15.1(2)T1 used in this configuration, Cisco 7965G SIP and 7975G SCCP telephones require the following firmware to work with the Cisco UCME.

Cisco 7965G SIP Telephone:

- SIP45.8-5-4S.loads
- term45.default.loads
- term65.default.loads
- apps45.8-5-4TH1-6.sbn
- cnu45.8-5-4TH1-6.sbn
- cvm45sip.8-5-4TH1-6.sbn
- dsp45.8-5-4TH1-6.sbn
- jar45sip.8-5-4TH1-6.sbn

Cisco 7975G SCCP Telephone:

- SCCP75.8-5-4S.loads
- term75.default.loads
- jar75sccp.8-5-4TH1-6.sbn
- cvm75sccp.8-5-4TH1-6.sbn
- apps75.8-5-4TH1-6.sbn
- cnu75.8-5-4TH1-6.sbn
- dsp75.8-5-4TH1-6.sbn

This section focuses on the VoIP related configuration (in bold) on the Cisco 3825 router.

```
version 15.1
no service pad
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
service password-encryption
service sequence-numbers
```

```
!
hostname CME-3825
!
boot-start-marker
boot system flash c3825-ipvoicek9-mz.151-2.T1.bin      --- Boot image
boot-end-marker
!
! card type command needed for slot/vwic-slot 1/1
security authentication failure rate 3 log
security passwords min-length 6
logging buffered 51200
logging console critical
enable secret 5 $1$vrfA$TvozCsgK1j/m.gohuDw7Q1
!
no aaa new-model
clock timezone edt -5 0
clock summer-time ESTime date Apr 6 2003 2:00 Oct 26 2003 2:00
clock calendar-valid
no network-clock-participate slot 1
!
dot11 syslog
no ip source-route
!
ip cef
!
no ip dhcp use vrf connected
ip dhcp excluded-address 192.45.131.1 192.45.131.9
ip dhcp excluded-address 192.45.131.100 192.45.131.254
!
ip dhcp pool ucme                          --- DHCP server configuration
  import all
  network 192.45.131.0 255.255.255.0       --- Network/subnet configuration
  default-router 192.45.131.2              --- Default router configuration
  option 150 ip 192.45.131.1               --- Use option 150 to set UCME as TFTP server
!
no ip bootp server
no ip domain lookup
ip domain name interoplab.local
ip name-server 192.45.132.182
no ipv6 cef
multilink bundle-name authenticated
!
voice-card 0                               --- Enable card to share DSP resources
 dsp services dspfarm
!
```

```
voice call carrier capacity active
!
voice service voip                                          --- Voice Class and Service VoIP configuration
 allow-connections sip to sip                               --- Enable B2BUA and allow SIP-SIP connections
 redirect ip2ip                                             --- Allow SIP calls to be hairpinned through UCME
 no supplementary-service sip moved-temporarily  --- Disable sending 302[3]
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco  --- Set T.38 as fax protocol
 sip
  registrar server                                          --- Enable SIP Registrar service for SIP endpoints
  g729 annexb-all                                           --- Enable UCME to accept all G729 codec flavors
!
voice class codec 1                                         --- Voice Class Codec configuration for SIP trunks
 codec preference 1 g729r8                                  --- Configure G.729 as 1st preferred codec
 codec preference 2 g729br8                                 --- Configure G.729B as 2nd preferred codec
 codec preference 3 g711ulaw                                --- Configure G.711 u-law as 3rd preferred codec
!
voice class codec 2                                         --- Voice Class Codec configuration for SIP phones
 codec preference 1 g711ulaw                                --- Configure G.711 u-law as 1st preferred codec
 codec preference 2 g729r8                                  --- Configure G.729[4] as 2nd preferred codec
!
voice register global                          --- SIP global settings for SIP phone registration
 mode cme                                      --- Enable mode for provisioning SIP phones on CM
 source-address 192.45.131.1 port 5060         --- Enable UCME router to receive SIP messages
 max-dn 50                                     --- Define max dn number supported on UCME
 max-pool 20                                   --- Limited the # of SIP phones supported by UCME.
 load 7965 SIP45.8-5-4S                        --- Associate a 7965  phone type with a phone firmware file
 timezone 12                                   --- Configure timezone
 dialplan-pattern 1 777.. extension-length 5 --- Define dialplan-pattern for the Cisco UCME stations
 dialplan-pattern 2 3.... extension-length 5  --- Define dialplan-pattern for the Avaya stations & voicemail
 external-ring bellcore-dr3                    --- Define external voicemail access number
 voicemail 33000                               --- Define Modular Messaging voicemail access number
 create profile                                --- Create a profile on UCME
!
voice register dn  2                           --- SIP phone directory number settings
 number 77702                                  --- Define directory number (extension)
 call-forward b2bua mailbox 77702              --- Define call forward mailbox number
 call-forward b2bua noan 33000 timeout 15  --- Define call forward on no answer
 allow watch
 name Maria                                    --- Define directory name
 label Maria                                   --- Define directory label
 mwi                                           --- Enable MWI
!
```

[3] Avaya Aura® Communication Manager supports SIP "302" messages; however due to interoperability issues observed during call forwards, this supplementary services was disabled on Cisco UCME.
[4] The Cisco 7965 SIP Telephone does not support the G.729B codec (annexb=yes).

```
voice register dialplan  1                       --- Create dial plan 1 for Cisco UCME SIP phones
 type 7940-7960-others                           --- Define a phone type for the SIP dial plan
 pattern 1 777..                                 --- Define a dial pattern for Cisco UCME extensions
 pattern 2 3....                                 --- Define a dial pattern for Avaya extensions & voicemail
!
voice register pool  2                           --- SIP phone settings
 id mac 001E.4A34.D081                           --- Enter SIP phone MAC address
 type 7965                                       --- Define phone type
 number 1 dn 2                                   --- Assign directory number 2 to phone line 1
 dialplan 1                                      --- Assign dial plan to this phone pool
 presence call-list
 dtmf-relay rtp-nte                              --- Configure dtmf-relay as rtp-nte (RFC 2833)
 voice-class codec 2                             --- Assign voice codec class 2 to the phone
 speed-dial 1 34503  label "Avaya Digital - 34503"
 speed-dial 2 34502 label "Avaya 9630 IP - 34502"
 speed-dial 3 34504 label "Avaya 9620 SIP - 34504"
 blf-speed-dial 4 77701 label "Tony"
!
interface GigabitEthernet0/1
 ip address 192.45.131.1 255.255.255.0           --- IP Address assigned to Cisco UCME gigabit interface
 no ip redirects
 no ip unreachables
 no ip proxy-arp
 ip route-cache flow
 duplex auto
 speed auto
 media-type rj45
 negotiation auto
 no mop enabled
!
router eigrp 1
 network 192.45.131.0
 no auto-summary
!
ip default-gateway 192.45.131.2
ip forward-protocol nd
!
ip http server
ip http authentication local
no ip http secure-server
ip http path flash:
!
logging trap debugging
snmp-server community public RO
snmp-server location SIL
```

```
snmp-server contact x@xxxx.com
!
!--- Enable TFTP server & have these files available for 7965/7945 SIP & 7975 SCCP phones to download
tftp-server flash:term65.default.loads
tftp-server flash:term45.default.loads
tftp-server flash:SIP45.8-5-4S.loads
tftp-server flash:jar45sip.8-5-4TH1-6.sbn
tftp-server flash:cvm45sip.8-5-4TH1-6.sbn
tftp-server flash:apps45.8-5-4TH1-6.sbn
tftp-server flash:cnu45.8-5-4TH1-6.sbn
tftp-server flash:dsp45.8-5-4TH1-6.sbn
tftp-server flash:term75.default.loads
tftp-server flash:SCCP75.8-5-4S.loads
tftp-server flash:jar75sccp.8-5-4TH1-6.sbn
tftp-server flash:cvm75sccp.8-5-4TH1-6.sbn
tftp-server flash:apps75.8-5-4TH1-6.sbn
tftp-server flash:cnu75.8-5-4TH1-6.sbn
tftp-server flash:dsp75.8-5-4TH1-6.sbn
!
control-plane
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
sccp local GigabitEthernet0/1      --- Set local interface that SCCP applications use to register with UCME
sccp ccm 192.45.131.1 identifier 1 version 7.0  --- Enable UCME to register SCCP applications
sccp                                            --- Enable SCCP and its related applications
!
sccp ccm group 1                                --- Create UCME SCCP group
 description UCME-GROUP                         --- Create UCME SCCP group description
 bind interface GigabitEthernet0/1              --- Bind GigabitEthernet0/1 interface to SCCP group
 associate ccm 1 priority 1                     --- Associate priority 1 to UCME
 associate profile 1 register UCME-3825         --- Associates a DSP farm profile with UCME group
!
dspfarm profile 1 transcode                     --- Define an application profile for DSP farm services.
 codec g711ulaw                                 --- Specify G.711 u-law codec
 codec g711alaw                                 --- Specify G.711 a-law codec
 codec g729ar8                                  --- Specify G.729A codec
 codec g729abr8                                 --- Specify G.729AB codec
 codec g729r8                                   --- Specify G.729 codec
 codec g729br8                                  --- Specify G.729B codec
 maximum sessions 10                            --- Specify maximum number of sessions
 associate application SCCP                     --- Associate SCCP with the DSP farm profile
!
```

```
dial-peer voice 3 voip                          --- Create a VoIP dial-peer "SIP Trunk" to connect to Avaya
 description "Out to Avaya SM/CM"               --- Configure Description
 destination-pattern 3....                      --- Configure destination-pattern 3.... for calls to 345.. & 33...
 voice-class codec 1                            --- Assign voice codec class 1 to the dial-peer
 session protocol sipv2                         --- Set Session Protocol SIP Version 2
 session target ipv4:10.1.2.70                  --- Configure Avaya Aura(R) Session Manager as session target
 session transport tcp                          --- Configure SIP session transport to TCP
 dtmf-relay rtp-nte                             --- Configure dtmf-relay as rtp-nte (RFC 2833)
 no vad
!
dial-peer voice 777 voip                        --- Create an incoming VoIP dial-peer "SIP Trunk"
 description "Incoming dial-peer"               --- Configure Description
 voice-class codec 1                            --- Assign voice codec class 1 to the dial-peer
 session protocol sipv2                         --- Set Session Protocol SIP Version 2
 session transport tcp                          --- Configure SIP session transport to TCP
 incoming called-number 777..                   --- Configure incoming called 777..
 dtmf-relay rtp-nte                             --- Configure dtmf-relay as rtp-nte (RFC 2833)
 no vad
!
presence
 presence call-list
 max-subscription 120
!
gateway
 timer receive-rtp 1200
!
sip-ua                                          --- Configure SIP User Agent
 keepalive target ipv4:10.1.2.70                --- Configure Avaya SM as  SIP OPTIONS target
! --- Configure Avaya Modular Messaging as MWI server and enable support for unsolicited NOTIFY messages
 mwi-server ipv4:135.8.139.31 expires 3600 port 5060 transport tcp unsolicited
 xfer target dial-peer                          --- Hidden command to use the dial-peer as the transfer target
 presence enable
!
telephony-service                               --- SCCP global telephony-service settings for SCCP phones
 sdspfarm units 5                               --- Configure maximum # of DSP farms allowed to register
 sdspfarm transcode sessions 8                  --- Configure maximum # of G.729 transcoder sessions
 sdspfarm tag 1 mtp001D45E95F20
 max-ephones 24                                 --- Set maximum number of phones that can register to UCME
 max-dn 72                                      --- Set maximum number of directory numbers
 ip source-address 192.45.131.1 port 2000 --- Set IP address and port #  for UCME phone registration
 system message SIL UCME                        --- Configure a message for display on SCCP phones
 load 7975 SCCP75.8-5-4S                        --- Associate a 7975 SCCP phone type with a firmware file
 time-zone 12                                   --- Configure time zone
 voicemail 33000                                --- Define Modular Messaging voicemail access number
 max-conferences 12 gain -6                     --- Set maximum number of simultaneous 3-party conferences
```

```
 call-forward pattern .....                    --- Configure call forward pattern
 moh music-on-hold.au                          --- Configure Music-On-Hold (MOH) file
 web admin system name interop secret 5 $1$Vtit$9esw1diEw.JuzfAgcLnd71
 web admin customer name DE password bear
 dn-webedit
 time-webedit
 transfer-system full-consult                  --- Configure transfers using consultation, if available
 transfer-pattern .T                           --- Configure transfer pattern
 secondary-dialtone 9                          --- Configure secondary dial tone for outside line
 create cnf-files                              --- Create XML configuration files required for SCCP Phones
!
ephone-dn  11  dual-line                       --- SCCP phone directory number settings
 number 77701                                  --- Define directory number (extension)
 label Tony                                    --- Define directory label
 name Tony                                     --- Define directory name
 allow watch
 call-forward busy 33000                       --- Define call forward busy
 call-forward noan 33000 timeout 10            --- Define call forward on no answer
 mwi sip                                       --- Enable SIP MWI

!
ephone  11                                     --- SCCP phone settings
 mac-address 001D.45E9.5F20                    --- Enter SCCP phone MAC address
 username "tony" password 1234                 --- Set username and password
 presence call-list
 blf-speed-dial 1 77710 label "Fred"
 blf-speed-dial 2 77702 label "Maria"
 speed-dial 1 34503  label "Avaya Digital - 34503"
 speed-dial 2 34502 label "Avaya 9630 IP - 34502"
 speed-dial 3 34504 label "Avaya 9620 SIP - 34504"
 type 7975                                     --- Define phone type
 mwi-line 1                                    --- Enable MWI for line 1
 keep-conference endcall             --- Configure conference initiator to exit & leave other parties connected
 button  1:11                                  --- Assign directory number 11 to button 1
 pin 1234                                      --- Set username and password
!
banner login ^CAuthorized access only!
 Disconnect IMMEDIATELY if you are not an authorized user!^C
!
line con 0
 login local
 transport output telnet
line aux 0
!
line con 0
```

```
 login local
 transport output telnet
line aux 0
 login local
 transport output telnet
line 130
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output all
line vty 0 4
 privilege level 15
 login local
 transport input all
line vty 5 15
 privilege level 15
 login local
!
scheduler allocate 20000 1000
end
```

After the configuration steps are complete, use the following commands to reset all SIP and SCCP telephones to force them to load the configuration file.

```
configure t
voice register global
  reset
  exit
telephony-service
 reset all
```

6 Verification Steps

This section provides the tests that can be performed on Communication Manager, Session Manager, and Cisco UCME to verify their proper configuration.

6.1 Verify Avaya Aura® Communication Manager

This section presents screens from Communication Manager that can be used to verify or troubleshoot the configuration.

6.1.1 SIP Signaling Group and Trunk Group Status

The SIP Signaling Group and SIP Trunk Group to Session Manager should be in-service. The following screen shows the "status trunk 26" screen, showing all trunks are in-service and idle.

```
status trunk 26
                             TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy
0026/001 T00017   in-service/idle    no
0026/002 T00018   in-service/idle    no
0026/003 T00019   in-service/idle    no
0026/004 T00020   in-service/idle    no
0026/005 T00021   in-service/idle    no
0026/006 T00022   in-service/idle    no
0026/007 T00023   in-service/idle    no
0026/008 T00024   in-service/idle    no
0026/009 T00025   in-service/idle    no
0026/010 T00026   in-service/idle    no
```

If the trunk group is not in-service, check the SIP Signaling Group status. The following screen shows the "status signaling-group 26" screen, showing that the signaling group is in-service.

```
status signaling-group 26
                         STATUS SIGNALING GROUP

        Group ID: 26                             Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

If the signaling group is in a "bypass" state, check the **Enable Layer 3 Test** parameter on the signaling group screen. If the **Enable Layer 3 Test** for the signaling group is set to "n", Communication Manager will use an "ICMP ping" test to verify that the far-end of the signaling group is reachable. Some networks may not pass ICMP ping, which is a possible cause for the signaling group to be marked for "bypass" and the corresponding trunk group to be marked "Out-of-Service/Far-end". In this state, Communication Manager would not use the trunk for outbound calls, but would allow an incoming call. In the sample configuration, the **Enable Layer 3 Test** has been set to "y", meaning that Communication Manager will use a SIP OPTIONS message to the far-end (Session Manager in this case) to verify connectivity. If the signaling group is marked for "bypass", and the SIP OPTIONS method is used, verify that the far-end node name (and corresponding IP Address) correctly refers to Session Manager. Verify that Session Manager is on-line and configured properly for a SIP Entity to Communication Manager. The Session Manager SIP Entity representing Communication Manager should specify the IP Address corresponding to the node name at the "near-end" of the Communication Manager signaling group (i.e., in this case, the S8800 "procr" IP Address).

Note: For greater detail, the traces shown below were captured with "Shuffling" enabled on the Communication Manager SIP signaling group to Session Manager.

6.1.2 Avaya Telephone Calls Cisco Telephone

This section has example calls where an Avaya H.323 telephone calls Cisco SIP and SCCP telephones. Greater detail is included in the initial illustrations, since the results including displays and connection topology are independent of the called telephone type in the sample configuration.

6.1.2.1 Avaya H.323 Telephone Calls Cisco SIP Telephone

The following "list trace station" output illustrates a call from the Avaya IP Telephone with extension 34502 to Cisco SIP Telephone extension 77702. The Avaya telephone, with IP Address 172.28.43.2 in network region 1, dials 77702. The call is routed using UDP and AAR to route pattern 26 containing trunk group 26. When the Cisco telephone is ringing, the Cisco telephone's display will show "From Tom Avaya (34502)" which correspond to the name and extension of the Avaya calling telephone. Similarly, the Avaya telephone will display "Maria 77702", which correspond to the Alerting Name and number configured for the called Cisco telephone. Upon answer by the called Cisco user, the displays are unchanged. The "far-end" region is region 3, and therefore the media connection is between region 1 and region 3. Codec set 3 governs this connectivity, and the final connection uses G.729, which was specified in ip-codec-set 3 at the time of this call. The initial media path connects the Cisco UCME with IP Address 192.45.131.1 in network region 3 to the Avaya G450 VoIP resources, at 10.1.2.95. After "shuffling" occurs, the final media path connects the Cisco UCME with IP Address 192.45.131.1 in network region 3 to the Avaya IP Telephone, at 172.28.43.2. With "Shuffling" or "Direct IP-IP Audio Connections" disabled on the Communication Manager SIP signaling group, the media path would stay between the Cisco UCME and the Avaya G450 VoIP resource.

```
list trace station 34502                                                Page   1

                                LIST TRACE

time            data

11:40:53 TRACE STARTED 08/24/2010 CM Release String cold-00.0.345.0-18567
11:41:05     active station    34502 cid 0x3e
11:41:05     G711MU ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:1 [10.1.2.95]:2050
11:41:05     dial 77702 route:UDP|AAR
11:41:05     term trunk-group 26    cid 0x15
11:41:05     dial 77702 route:UDP|AAR
11:41:05     route-pattern  26 preference 1  cid 0x15
11:41:05     seize trunk-group 26 member 2  cid 0x15
11:41:05     Calling Number & Name NO-CPNumber NO-CPName
11:41:05 SIP>INVITE sip:77702@avaya.com SIP/2.0
11:41:05     Setup digits 77702
11:41:05     Calling Number & Name 34502 Tom Avaya
11:41:05 SIP<SIP/2.0 100 Trying
11:41:05     Proceed trunk-group 26 member 2  cid 0x15
11:41:07 SIP<SIP/2.0 180 Ringing
11:41:07 SIP>PRACK sip:77702@192.45.131.1:5060;transport=tcp SIP
11:41:07 SIP>/2.0
11:41:07     Alert trunk-group 26 member 2  cid 0x15
11:41:07 SIP<SIP/2.0 200 OK
11:41:10 SIP<SIP/2.0 200 OK
11:41:10 SIP>ACK sip:77702@192.45.131.1:5060;transport=tcp SIP/2
11:41:10 SIP>.0
11:41:10     active trunk-group 26 member 2  cid 0x15
11:41:10     G729 ss:off ps:20
             rgn:3 [192.45.131.1]:18462
             rgn:1 [10.1.2.95]:2054
11:41:10     xoip options: fax:T38 modem:off tty:US  uid:0x500f8
             xoip ip: [10.1.2.95]:2050
11:41:10 SIP>INVITE sip:77702@192.45.131.1:5060;transport=tcp SI  ! Shuffling INVITE
11:41:10 SIP>P/2.0
11:41:10 SIP<SIP/2.0 100 Trying
11:41:10 SI11:41:10 SIP<SIP/2.0 200 OK
11:41:10     G729 ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:3 [192.45.131.1]:18462
11:41:10 SIP>ACK sip:77702@192.45.131.1:5060;transport=tcp SIP/2
11:41:10 SIP>.0
11:41:10     G729A ss:off ps:20
             rgn:3 [192.45.131.1]:18462
             rgn:1 [172.28.43.2]:63878
```

The "status trunk" command can also be used, as shown below for this same call, while active. **Page 2** is shown below. The near-end and far-end signaling IP Addresses and Ports can be observed for the TLS connection between Communication Manager and Session Manager. The media connection information shows that the call is "ip-direct" between the Avaya IP Telephone and Cisco UCME. With "Shuffling" or "Direct IP-IP Audio Connections" disabled, the "Audio Connection Type" would display "ip-tdm".

```
status trunk 26/1                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                               Port
   Near-end:  10.1.2.90                              : 5065
    Far-end:  10.1.2.70                              : 5065
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729
   Audio      IP Address                               Port
   Near-end:  172.28.43.2                            : 63878
    Far-end:  192.45.131.1                           : 18246

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:               Video Far-end Codec:
```

On **Page 3**, further details can be observed.

```
status trunk 26/1                                              Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00249
T00249:TX:192.45.131.1:18246/g729/20ms
S00106:RX:172.28.43.2:63878/g729a/20ms
```

If the Cisco telephone holds the call, music on hold from Cisco UCME is heard by the Avaya telephone.

If the Avaya telephone holds the call, the media path must move from the Avaya IP telephone to the Avaya G450 announcement capability playing the music. The following is an example status screen taken when the Avaya phone had held the call, and the Cisco telephone user was listening to music from the Avaya G450 announcement capability.

```
status trunk 26/1                                             Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  10.1.2.90                             : 5065
    Far-end:  10.1.2.70                             : 5065
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-tdm        Authentication Type: None
    Near-end Audio Loc: MG1                    Codec Type: G.729
   Audio      IP Address                              Port
   Near-end:  10.1.2.95                             : 2064
    Far-end:  192.45.131.1                          : 18246

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:              Video Far-end Codec:
```

If the Avaya IP telephone resumes the held call, the media path moves off Avaya G450 Media Gateway back to the Avaya IP telephone. That is, the connection topology returns to the status before the call was held.

If the Cisco SIP telephone transfers the call to the Cisco SCCP telephone, the transfer is successful, and the final connection topology has the Avaya IP Telephone communicating with Cisco UCME. Post transfer, the display on the transferred-to telephone is “From Tom Avaya (34502)”, the name and number of the Avaya telephone. The display on the Avaya telephone updates to “Tony 77701”, the name and number of the transferred-to Cisco SCCP telephone.

6.1.2.2 Avaya H.323 Telephone Calls Cisco SCCP Telephone

The following “list trace station” output illustrates a call from the Avaya IP Telephone with extension 34502 to Cisco SCCP Telephone extension 77701. The Avaya telephone, with IP Address 172.28.43.2 in network region 1, dials 77701. The call is routed using UDP and AAR to route pattern 26 containing trunk group 26. When the Cisco telephone is ringing, the Cisco telephone’s display will show “From Tom Avaya (34502)” which correspond to the name and extension of the Avaya calling telephone. Similarly, the Avaya telephone will display “Tony 77701”, which correspond to the Alerting Name and number configured for the called Cisco telephone. Upon answer by the called Cisco user, the displays are unchanged. The “far-end” region is region 3, and therefore the media connection is between region 1 and region 3. Codec set 3 governs this connectivity, and the final connection uses G.729, which was specified in ip-codec-set 3 at the time of this call. The initial media path connects the Cisco UCME with IP Address 192.45.131.1 in network region 3 to the Avaya G450 VoIP resources, at 10.1.2.95. After “shuffling” occurs, the final media path connects the Cisco UCME with IP Address

192.45.131.1 in network region 3 to the Avaya IP Telephone, at 172.28.43.2. As stated previously, with “Shuffling” disabled, the media path would stay between the Cisco UCME and the Avaya G450 VoIP resource.

```
list trace station 34502                                                Page   1

                                LIST TRACE
time            data

09:21:55 TRACE STARTED 08/24/2010 CM Release String cold-00.0.345.0-18567
09:22:06     active station    34502 cid 0x3e
09:22:06     G711MU ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:1 [10.1.2.95]:2050
09:22:06     dial 77701 route:UDP|AAR
09:22:06     term trunk-group 26    cid 0x1c
09:22:06     dial 77701 route:UDP|AAR
09:22:06     route-pattern  26 preference 1  cid 0x1c
09:22:06     seize trunk-group 26 member 3  cid 0x1c
09:22:06     Calling Number & Name NO-CPNumber NO-CPName
09:22:06 SIP>INVITE sip:77701@avaya.com SIP/2.0
09:22:06     Setup digits 77701
09:22:06     Calling Number & Name 34502 Tom Avaya
09:22:06 SIP<SIP/2.0 100 Trying
09:22:06     Proceed trunk-group 26 member 3  cid 0x1c
09:22:06 SIP<SIP/2.0 180 Ringing
09:22:06 SIP>PRACK sip:77701@192.45.131.1:5060;transport=tcp SIP
09:22:06 SIP>/2.0
09:22:06     Alert trunk-group 26 member 3  cid 0x1c
09:22:06 SIP<SIP/2.0 200 OK
09:22:10 SIP<SIP/2.0 200 OK
09:22:10 SIP>ACK sip:77701@192.45.131.1:5060;transport=tcp SIP/2
09:22:10 SIP>.0
09:22:10     active trunk-group 26 member 3  cid 0x1c
09:22:10     G729 ss:off ps:20
             rgn:3 [192.45.131.1]:16502
             rgn:1 [10.1.2.95]:2054
09:22:10     xoip options: fax:T38 modem:off tty:US  uid:0x500f9
             xoip ip: [10.1.2.95]:2056
09:22:10 SIP>INVITE sip:77701@192.45.131.1:5060;transport=tcp SI    ! Shuffling INVITE
09:22:10 SIP>P/2.0
09:22:10 SIP<SIP/2.0 100 Trying
09:22:11 SIP<SIP/2.0 200 OK
09:22:11     G729 ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:3 [192.45.131.1]:16502
09:22:11 SIP>ACK sip:77701@192.45.131.1:5060;transport=tcp SIP/2
09:22:11 SIP>.0
09:22:11     G729A ss:off ps:20
             rgn:3 [192.45.131.1]:16502
             rgn:1 [172.28.43.2]:63878
```

The “status trunk” command can also be used, with similar output to that already presented in the prior section. Rather than repeat, more detailed information is provided for a Cisco held call. If the Cisco telephone holds the call, music on hold from Cisco UCME is heard by the Avaya telephone. The following screen illustrates the connection while on hold at the Cisco side.

```
status trunk 26/10                                                Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00249
T00249:TX:192.45.131.1:16502/g729/20ms
S00106:RX:172.28.43.2:63878/g729a/20ms
```

Once the call is resumed, two-way audio is restored properly.

If the Avaya IP telephone transfers the call to the Avaya digital telephone, the transfer is successful, and the final connection is between the Avaya G450 VoIP resource and Cisco UCME. Post transfer, the display on the transferred-to Avaya telephone will show “Tony 77701”, the name and number of the connected Cisco telephone. The display on the connected Cisco telephone updates to “From Digital Avaya (34511)”, the name and number of the transferred-to Avaya telephone.

If the Cisco SCCP telephone (77701) transfers the call to the Cisco SIP telephone (77702), the transfer is successful, and the final connection is between the Avaya G450 VoIP resource and Cisco UCME. Post transfer, the display on the Avaya telephone will show “Answered by 77702”. The display on the transferred-to Cisco SIP telephone will show “From 77701”, the name and number of the original Cisco SCCP telephone that completed the transfer instead of the name and number of the connected Avaya digital telephone.

6.1.3 Cisco Telephone Calls Avaya Telephone

This section has example calls where Cisco SIP and SCCP telephones call the Avaya IP telephone.

6.1.3.1 Cisco SIP Telephone calls Avaya H.323 Telephone

The following “list trace tac” output illustrates an incoming call from the SIP trunk to Session Manager for a call from Cisco SIP Telephone extension 77702 to Avaya IP Telephone extension 34502. When the Avaya telephone is ringing, the Cisco telephone’s display will show “To 34502” which correspond to the number of the called Avaya telephone. The name is not displayed because Cisco UCME upon receipt of a 180 RINGING message from Avaya sends a “183 Session Progress” message to the Cisco SIP Telephone with no Called Party Name in the “Remote-Party-ID” header (e.g. Remote-Party-ID: <sip:34502@192.45.131.1>;party=called;screen=no;privacy=off). The Avaya IP Telephone will display “Maria 77702”, which correspond to the name and number configured for the calling Cisco SIP Telephone. Upon answer by the called Avaya user, the Avaya telephone display is unchanged, however Cisco SIP phone display is updated correctly “To Tom Avaya (34502)”. (Do not be deceived by the trace output below showing no calling number and name. The number and name of the Cisco caller do appear on the Avaya telephone’s display).

Similar to the calls from Avaya to Cisco, the final media path is between Cisco UCME (192.45.131.1) and the Avaya IP Telephone (172.28.43.2). Again, please note that with "Shuffling" disabled, the media path would stay between the Cisco UCME and the Avaya G450 VoIP resource.

```
list trace tac 126                                                  Page   1

                                LIST TRACE

time            data
11:14:17 TRACE STARTED 08/24/2010 CM Release String cold-00.0.345.0-18567
11:14:25 SIP<INVITE sip:34502@avaya.com:5060 SIP/2.0
11:14:25     active trunk-group 26 member 1  cid 0x14
11:14:25 SIP>SIP/2.0 180 Ringing
11:14:25     dial 34502
11:14:25     ring station    34502 cid 0x14
11:14:25     G711MU ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:1 [10.1.2.95]:2056
11:14:25     G729 ss:off ps:20
             rgn:3 [192.45.131.1]:17786
             rgn:1 [10.1.2.95]:2058
11:14:25     xoip options: fax:T38 modem:off tty:US  uid:0x500f7
             xoip ip: [10.1.2.95]:2058
11:14:25 SIP<PRACK sip:34502@10.1.2.90:5065;transport=tcp SIP/2.
11:14:25 SIP<0
11:14:25 SIP>SIP/2.0 200 OK
11:14:28 SIP>SIP/2.0 200 OK
11:14:28     active station    34502 cid 0x14
11:14:28 SIP<ACK sip:34502@10.1.2.90:5065;transport=tcp SIP/2.0
11:14:28 SIP>INVITE sip:77702@192.45.131.1:5060;transport=tcp SI  ! Shuffling INVITE
11:14:28 SIP>P/2.0
11:14:28 SIP<SIP/2.0 100 Trying
11:14:29 SIP<SIP/2.0 200 OK
11:14:29 SIP>ACK sip:77702@192.45.131.1:5060;transport=tcp SIP/2
11:14:29 SIP>.0
11:14:29     G729A ss:off ps:20
             rgn:3 [192.45.131.1]:17786
             rgn:1 [172.28.43.2]:63878
11:14:29     G729 ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:3 [192.45.131.1]:17786
```

Hold/resume and transfer scenarios from both the Avaya telephone and Cisco telephone were verified and work properly as described previously. Screen details would be redundant and reveal no new information.

6.1.3.2 Cisco SCCP Telephone calls Avaya H.323 Telephone

The following "list trace tac" output illustrates an incoming call from the SIP trunk to Session Manager for a call from Cisco SCCP Telephone extension 77701 to Avaya IP Telephone extension 34502. When the Avaya telephone is ringing, the Cisco telephone's display will show "To 34502" which correspond to the number of the called Avaya telephone. The Avaya IP telephone will display "Tony 77701", which correspond to the Name and number configured for the calling Cisco telephone. Upon answer by the called Avaya user, the Avaya telephone display is unchanged, however the Cisco SIP phone display is updated correctly "To Tom Avaya (34502)".

Similar to the corresponding calls from Avaya to Cisco, the final media path is between Cisco UCME (192.45.131.1) and the Avaya IP Telephone (172.28.43.2). With "Shuffling" disabled, the media path would stay between the Cisco UCME and the Avaya G450 VoIP resource.

```
list trace tac 126                                                    Page   1
                                LIST TRACE
time            data
13:15:26 TRACE STARTED 08/24/2010 CM Release String cold-00.0.345.0-18567
13:15:49 SIP<INVITE sip:34502@avaya.com:5060 SIP/2.0
13:15:49     active trunk-group 26 member 1  cid 0x47
13:15:49 SIP>SIP/2.0 180 Ringing
13:15:49     dial 34502
13:15:49     ring station    34502 cid 0x47
13:15:49     G711MU ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:1 [10.1.2.95]:2090
13:15:49     G729 ss:off ps:20
             rgn:3 [192.45.131.1]:18074
             rgn:1 [10.1.2.95]:2092
13:15:49     xoip options: fax:T38 modem:off tty:US  uid:0x500f7
             xoip ip: [10.1.2.95]:2092
13:15:49 SIP<PRACK sip:34502@10.1.2.90:5065;transport=tcp SIP/2.
13:15:49 SIP<0
13:15:49 SIP>SIP/2.0 200 OK
13:15:53 SIP>SIP/2.0 200 OK
13:15:53     active station    34502 cid 0x47
13:15:53 SIP<ACK sip:34502@10.1.2.90:5065;transport=tcp SIP/2.0
13:15:53 SIP>INVITE sip:77701@192.45.131.1:5060;transport=tcp SI ! Shuffling INVITE
13:15:53 SIP>P/2.0
13:15:53 SIP<SIP/2.0 100 Trying
13:15:54 SIP<SIP/2.0 200 OK
13:15:54 SIP>ACK sip:77701@192.45.131.1:5060;transport=tcp SIP/2
13:15:54 SIP>.0
13:15:54     G729A ss:off ps:20
             rgn:3 [192.45.131.1]:18074
             rgn:1 [172.28.43.2]:63878
13:15:54     G729 ss:off ps:20
             rgn:1 [172.28.43.2]:63878
             rgn:3 [192.45.131.1]:18074
```

Hold/resume from both the Avaya telephone and Cisco telephone were verified and work properly as described previously. Screen details would be redundant and reveal no new information.

If the Cisco SCCP telephone (77701) transfers the call to the Cisco SIP telephone (77702), the transfer is successful, and the final connection is between the Avaya IP Telephone and Cisco UCME. Post transfer, the display on the transferred-to Cisco SIP telephone will incorrectly show "From Tony (34502)", the name of the Cisco SCCP Telephone with the number of the connected Avaya IP telephone.   The display on the connected Avaya telephone will show "Answered by 77702".

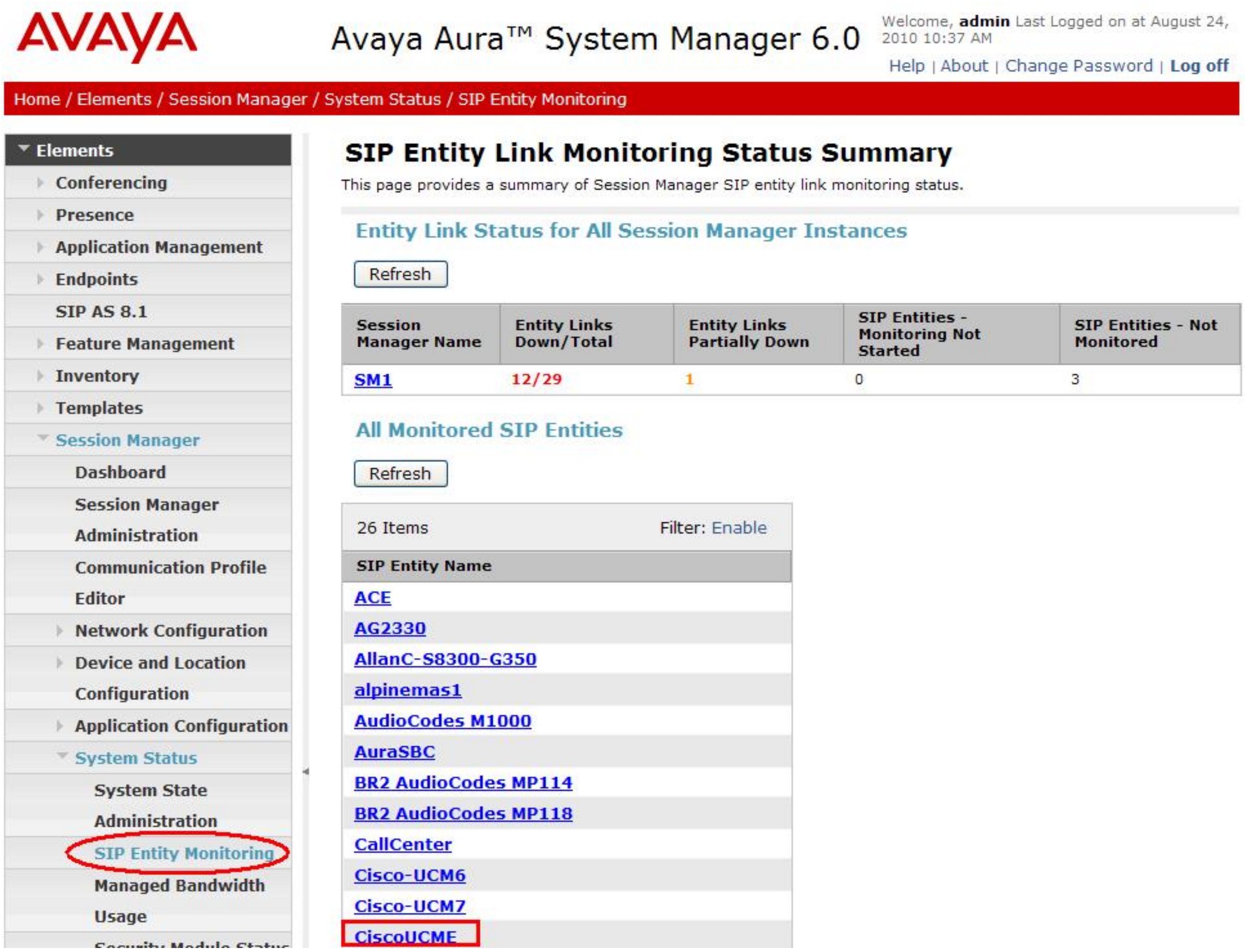
If the Avaya IP telephone (34502) transfers the call to the Avaya digital telephone (34503), the transfer is successful, and the final connection is between the Avaya G450 VoIP resource and Cisco UCME. Post transfer, the display on the Avaya digital telephone will show "Answered by 77702", the name and number of the connected Cisco telephone. The display on the connected Cisco SCCP telephone does not update and will show "From Tony (34502)", the name of the Cisco SCCP Telephone with the number of the connected Avaya IP telephone.

6.2 Verify Avaya Aura® Session Manager

Session Manager includes SIP monitoring and routing test capabilities that can aid in verifying proper configuration and operation.

6.2.1 SIP Monitoring

Select **Elements → Session Manager → System Status → SIP Entity Monitoring** as shown below.



Select the name of the relevant SIP entity from the list of monitored SIP entities. The following screen shows a sample result when the "CiscoUCME" SIP Entity was selected. Observe that the connection is up. Cisco UCME is responding to the SIP OPTIONS message from Session Manager with a "200 OK".

1 Item Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SM1	192.45.131.1	5060	TCP	Up	200 OK	Up

Under the **Details** column, **Show** can be clicked to obtain further information, which may be particularly relevant if there is a problem. In this case, **Show** reveals the following:

1 Item Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Hide	SM1	192.45.131.1	5060	TCP	Up	200 OK	Up

Time Last Down	Time Last Up	Last Message Sent	Last Response Latency (ms)
Aug 12, 2010 11:47:14 PM EDT	Aug 12, 2010 11:49:06 PM EDT	Aug 24, 2010 1:45:31 PM EDT	10

Similarly, information about the status of the link between Session Manager and Communication Manager can be obtained by selecting **Elements → Session Manager → System Status → SIP Entity Monitoring** and clicking on the link named "CM-Evolution-procr-5065". As can be seen in the screen below, the connection is "Up". Communication Manager is also responding with a "200 OK" to SIP OPTIONS sourced by Session Manager.

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: CM-Evolution-procr-5065

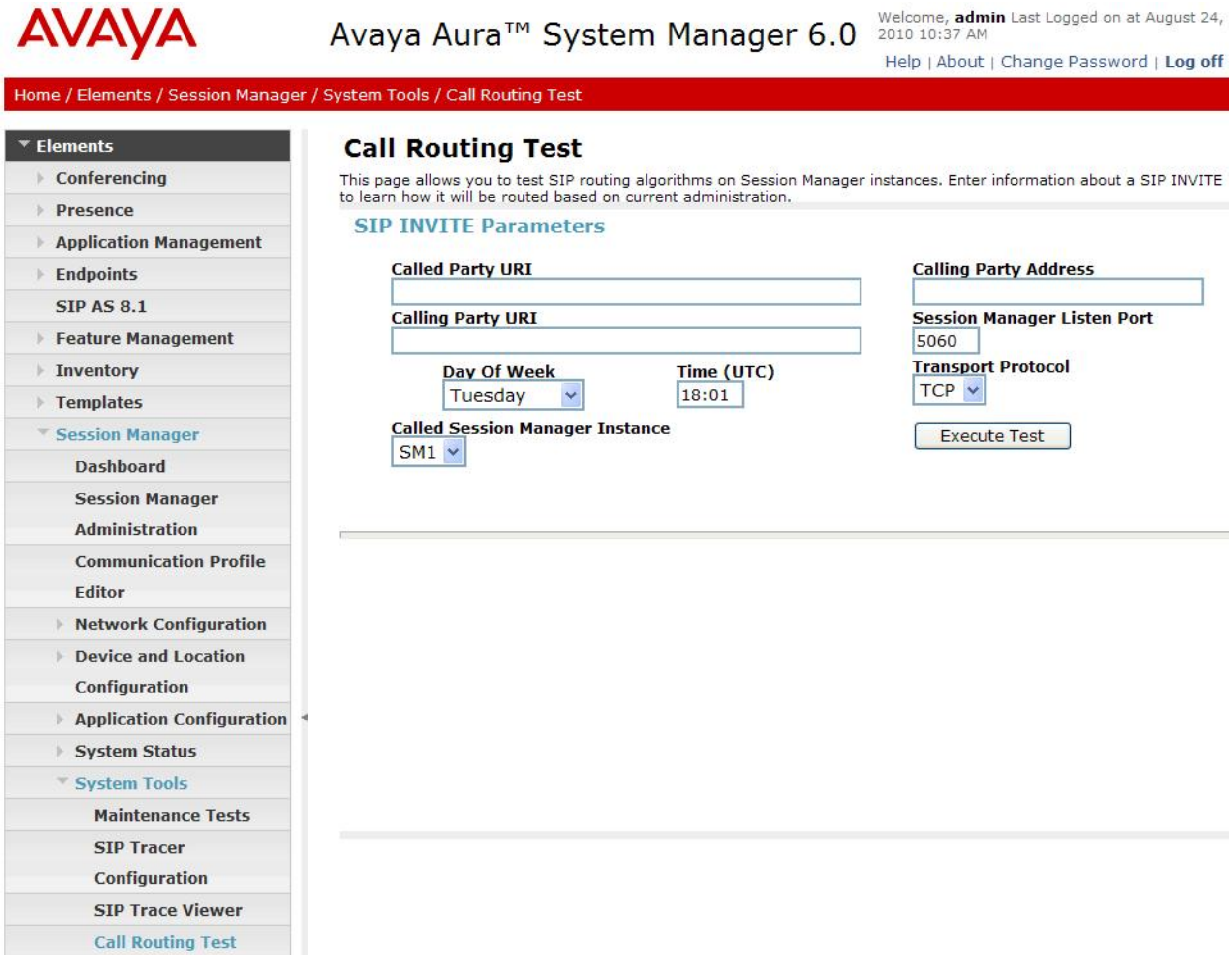
Refresh Summary View

1 Item Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SM1	10.1.2.90	5065	TCP	Up	200 OK	Up

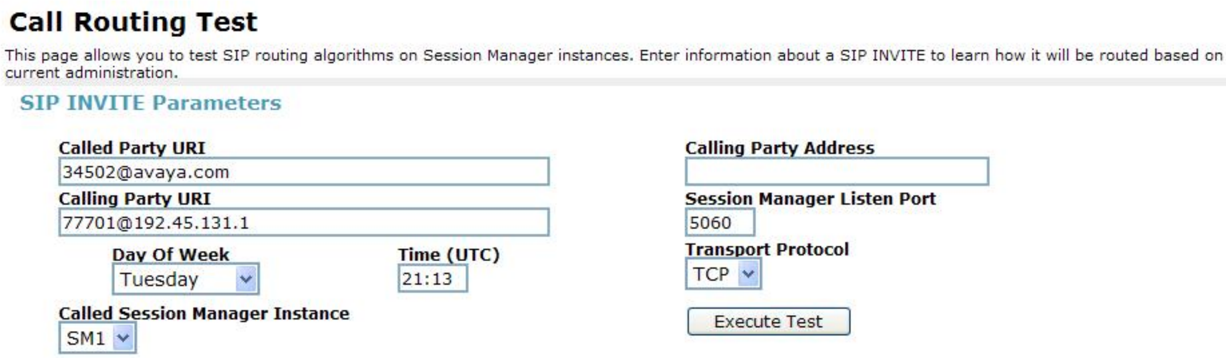
6.2.2 Call Routing Test

To check that the configured Network Routing Policy will result in the expected routing between systems, select **Elements → Session Manager → System Tools → Call Routing Test.** The following screen is presented.

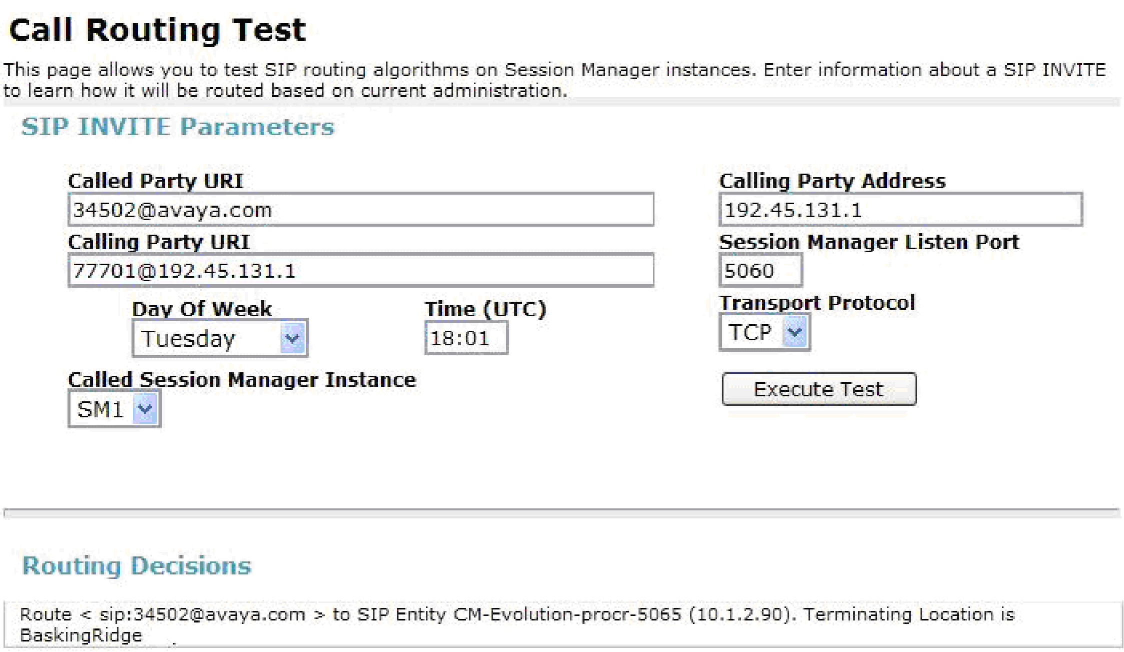


6.2.2.1 Cisco Telephone Calls Avaya Telephone

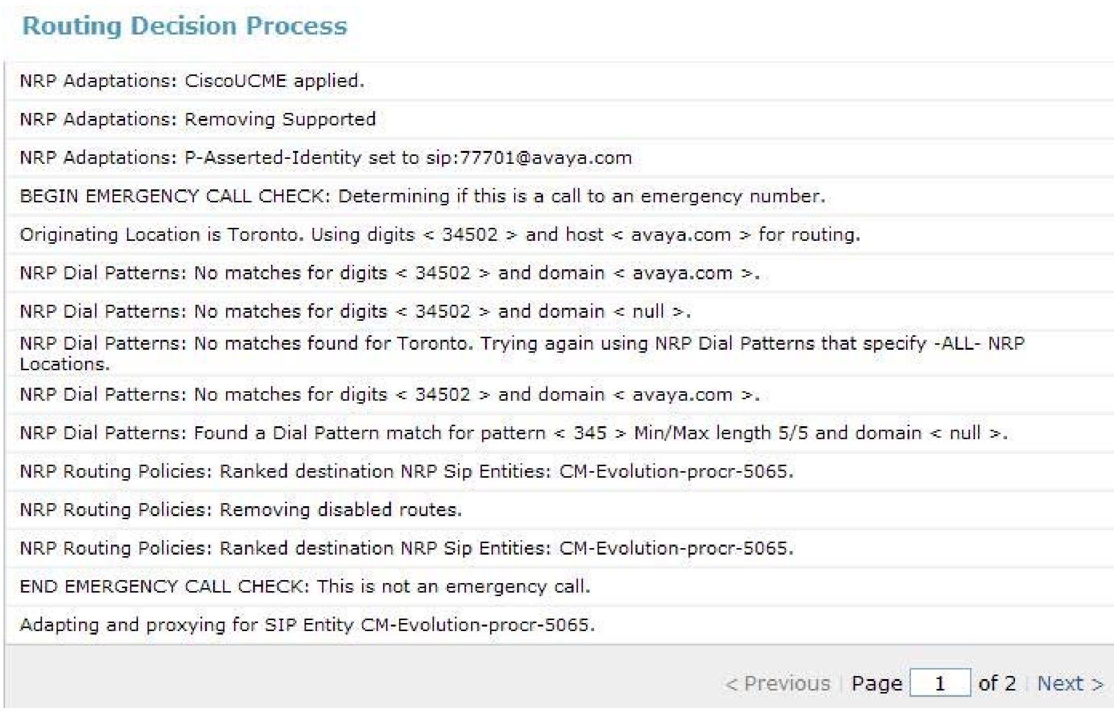
The following screen shows an example of a routing test for a Cisco telephone (77701) calling an Avaya telephone (34502). The self-explanatory **Called Party URI** and **Calling Party URI** fields are populated for a routing query.



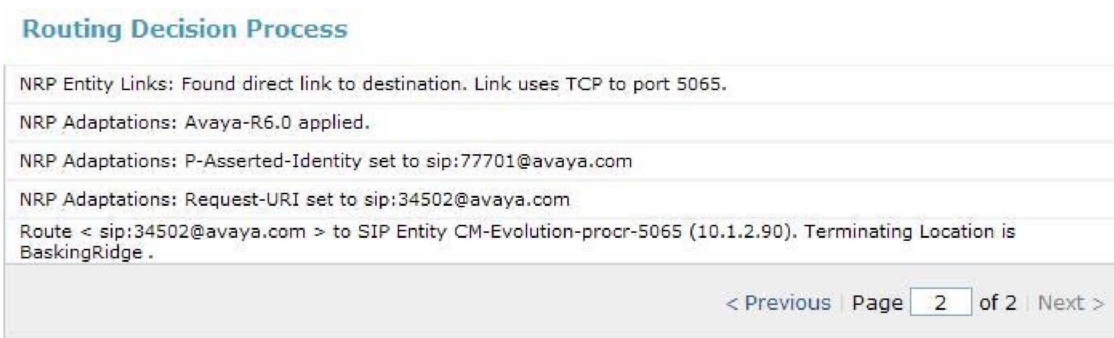
After typing in the **Calling Party Address** with the IP Address of Cisco UCME, the **Execute Test** button is pressed. The following screen illustrates the summary result, under the heading **Routing Decisions**. If the caller is extension 77701, and the call comes from Cisco UCME using TCP port 5060, and arrives Tuesday at 18:01 (or "Anytime" in the sample configuration), and the called party is 34502, the call will be routed to SIP Entity "CM-Evolution-procr-5065" at terminating location "BaskingRidge". This is the expected result from the configuration presented in **Section 6**.



Scrolling down below the **Routing Decisions** heading, additional information is available that may reinforce understanding of the configuration and decision process. For example, from the following series of screen captures, it can be observed that the originating SIP entity is recognized as “CiscoUCME” in location “Toronto”. The CiscoAdapter is invoked to set, and the P-Asserted-Identity (PAI) is populated with the calling party number. (For an actual call that contained the caller’s name in the Remote-Party-ID field, Session Manager would also copy the calling party name). No location-specific routing entry has been configured, but an “ALL” locations entry matches. The call ultimately is routed to SIP Entity “CM-Evolution-procr-5065”.

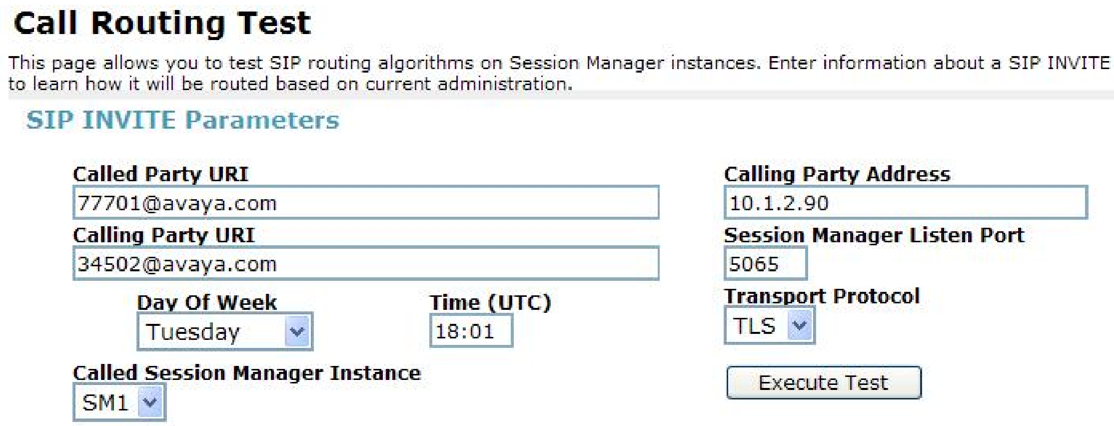


Additional information follows on Page 2.

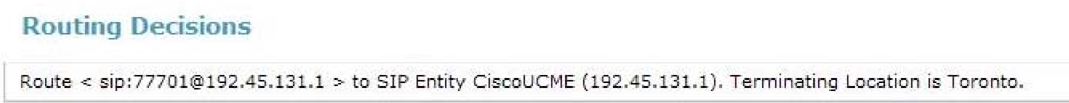


6.2.2.2 Avaya Telephone Calls Cisco Telephone

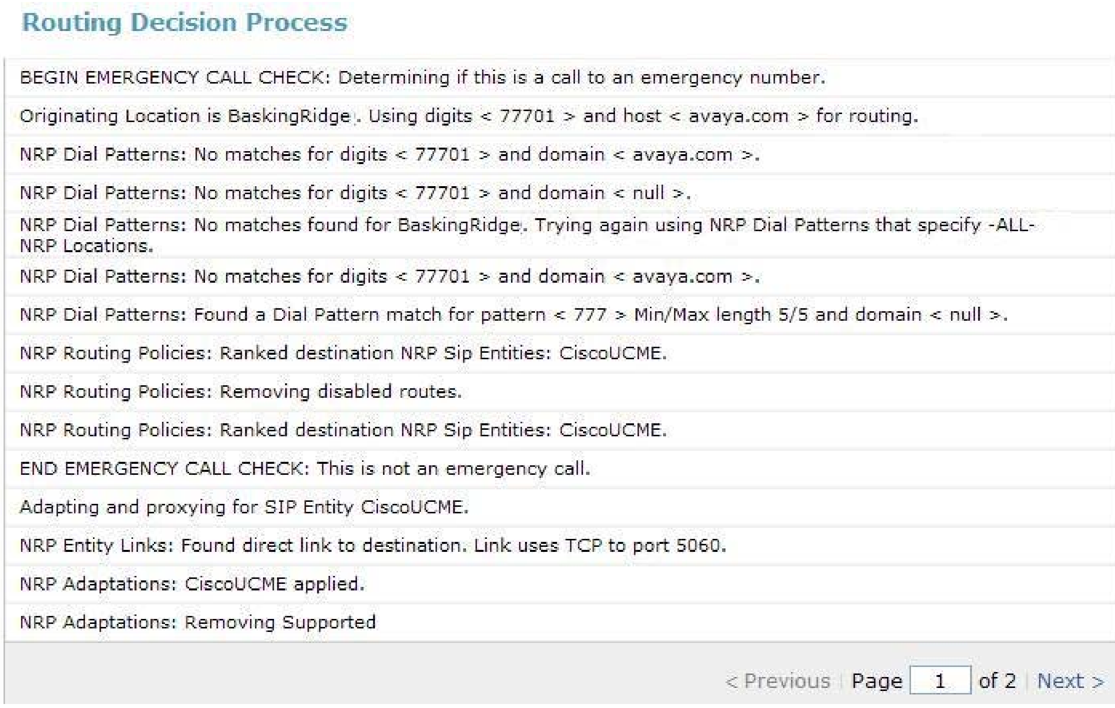
The following screen shows an example of a routing test for an Avaya telephone (34502) calling a Cisco telephone (77701). The Calling Party Address is the IP Address of the Avaya S8800 server running Communication Manager. In this case, TLS and port 5065 is selected.



The following screen shows the summary result. The call will be routed to Cisco UCME at IP Address 192.45.131.1, in terminating location “Toronto”.



Scrolling down below the **Routing Decisions** heading, the originating SIP entity is recognized as "CM-Evolution-procr-5065" in location "BaskingRidge". No location-specific routing entry is configured for "BaskingRidge", but the "ALL" locations configuration matches. The call is routed to SIP Entity "CiscoUCME" using TCP and port 5060.

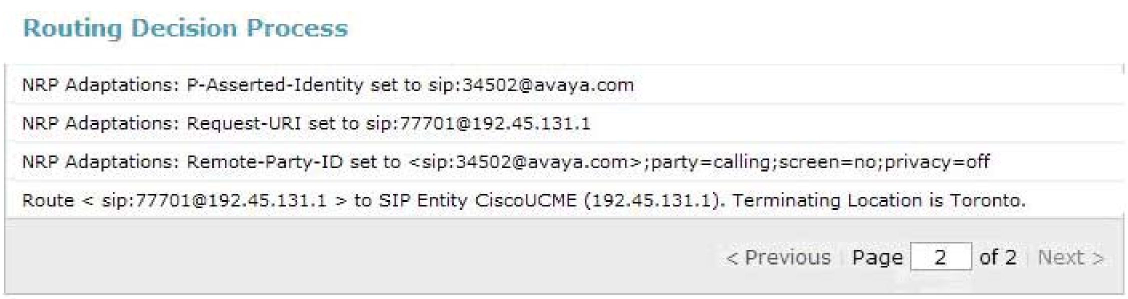


Routing Decision Process

BEGIN EMERGENCY CALL CHECK: Determining if this is a call to an emergency number.
Originating Location is BaskingRidge. Using digits < 77701 > and host < avaya.com > for routing.
NRP Dial Patterns: No matches for digits < 77701 > and domain < avaya.com >.
NRP Dial Patterns: No matches for digits < 77701 > and domain < null >.
NRP Dial Patterns: No matches found for BaskingRidge. Trying again using NRP Dial Patterns that specify -ALL- NRP Locations.
NRP Dial Patterns: No matches for digits < 77701 > and domain < avaya.com >.
NRP Dial Patterns: Found a Dial Pattern match for pattern < 777 > Min/Max length 5/5 and domain < null >.
NRP Routing Policies: Ranked destination NRP Sip Entities: CiscoUCME.
NRP Routing Policies: Removing disabled routes.
NRP Routing Policies: Ranked destination NRP Sip Entities: CiscoUCME.
END EMERGENCY CALL CHECK: This is not an emergency call.
Adapting and proxying for SIP Entity CiscoUCME.
NRP Entity Links: Found direct link to destination. Link uses TCP to port 5060.
NRP Adaptations: CiscoUCME applied.
NRP Adaptations: Removing Supported

< Previous | Page 1 of 2 | Next >

Additional details can be found on Page 2, including information on how the "Remote-Party-ID" is populated.



Routing Decision Process

NRP Adaptations: P-Asserted-Identity set to sip:34502@avaya.com
NRP Adaptations: Request-URI set to sip:77701@192.45.131.1
NRP Adaptations: Remote-Party-ID set to <sip:34502@avaya.com>;party=calling;screen=no;privacy=off
Route < sip:77701@192.45.131.1 > to SIP Entity CiscoUCME (192.45.131.1). Terminating Location is Toronto.

< Previous | Page 2 of 2 | Next >

6.2.3 CiscoAdapter Summary for Improved Display Interoperability

Section 7.1.2 and Section 7.1.3 provide a summary of expected displays for basic calls and transferred calls. The CiscoAdapter of Session Manager plays an important role in providing display interoperability. For example, Cisco UCME sends and processes display information that appears in the "Remote-Party-ID". The Session Manager CiscoAdapter can extract information from standard SIP elements and populate the "Remote-Party-ID" for Cisco UCME consumption. Similarly, the Session Manager CiscoAdapter can extract information from the

"Remote-Party-ID" and populate standard SIP elements for proper processing by Communication Manager.

6.2.3.1 Avaya Telephone Calls Cisco Telephone

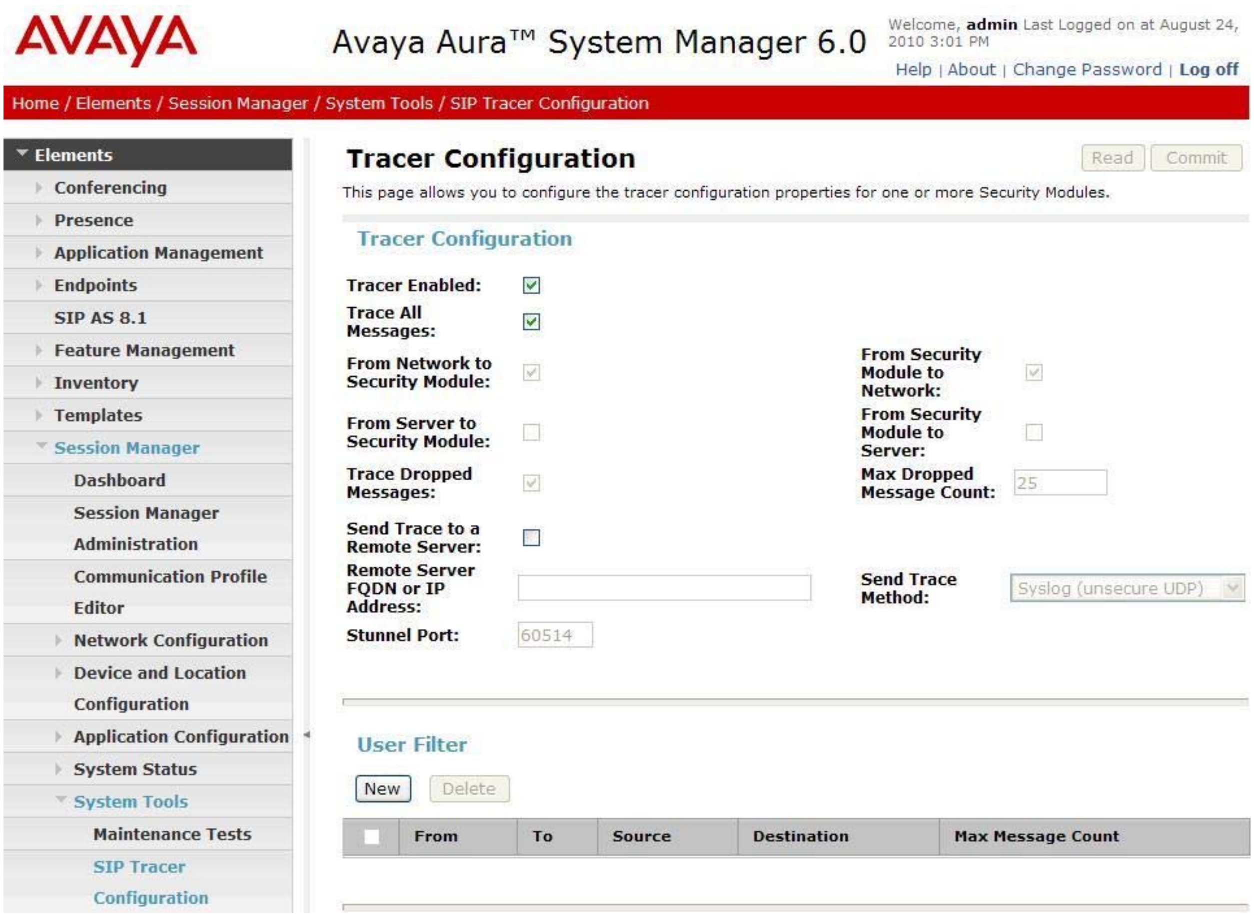
When an Avaya telephone calls a Cisco telephone, the SIP INVITE message sent from Communication Manager to Session Manager will include standard SIP information about the caller (e.g., in the From header and P-Asserted-Identity or PAI). As the call passes through Session Manager, Session Manager inserts the Remote-Party-ID containing the name and number of the Avaya caller. The Cisco telephone displays the caller's information. When the Cisco telephone alerts, Cisco UCME sends a "180 RINGING" SIP message to Session Manager with the Remote-Party-ID containing the "Alerting Name" and number of the ringing telephone. Session Manager extracts the information from the Remote-Party-ID and populates the PAI in the 180 RINGING sent to Communication Manager. Communication Manager displays the name and number of the alerting Cisco user on the calling party's display. A similar adaptation is performed on the 200 OK message when the Cisco telephone answers the call.

6.2.3.2 Cisco Telephone Calls Avaya Telephone

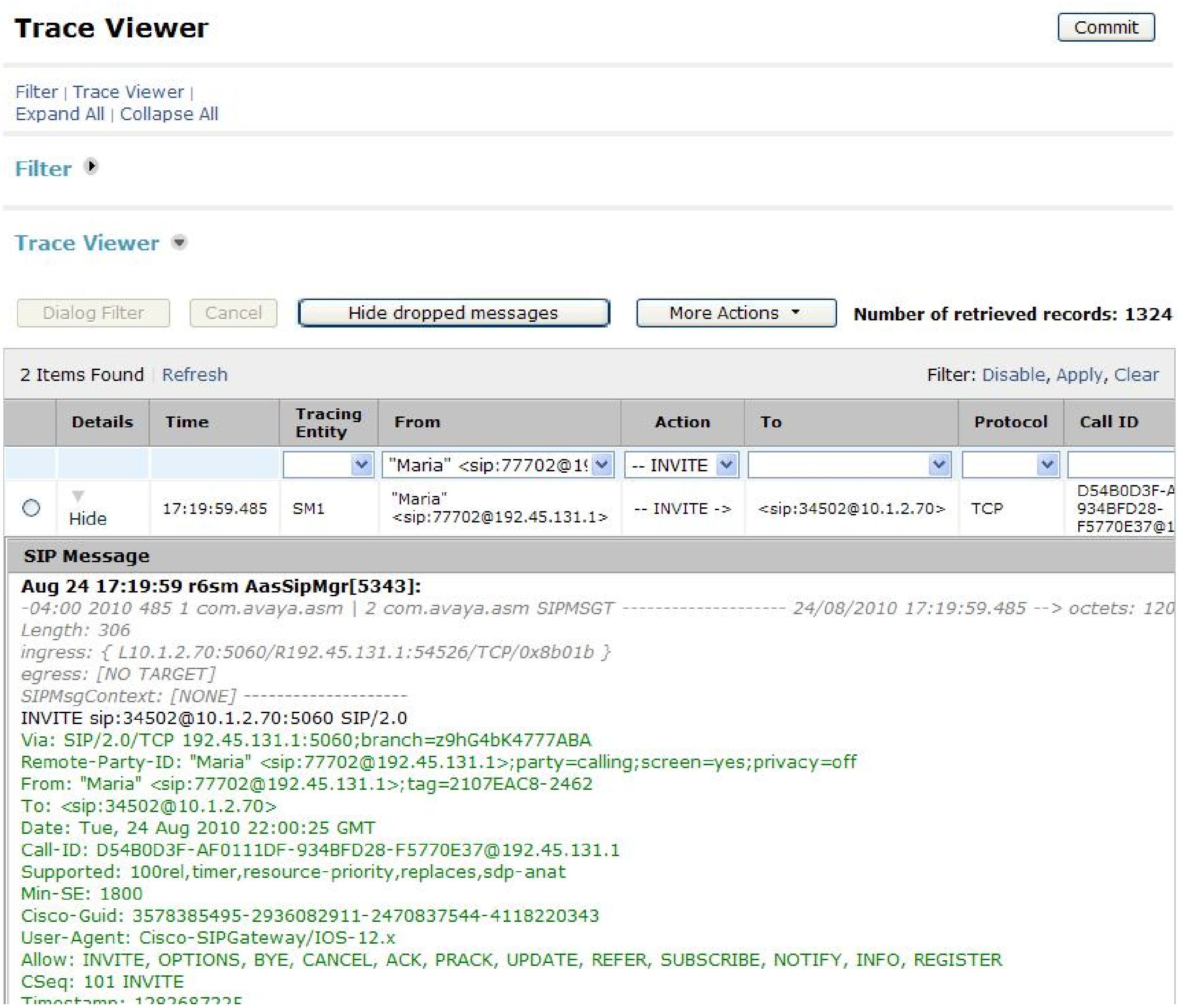
When a Cisco telephone calls an Avaya telephone, the SIP INVITE message sent from Cisco UCME to Session Manager can include the caller's name and number in the Remote-Party-ID. As the call passes through Session Manager, Session Manager extracts the caller's information from the Remote-Party-ID and populates standard SIP elements (e.g., PAI) in the SIP INVITE toward Communication Manager, which displays the caller's information on the alerting Avaya phone. When the Avaya telephone rings, Communication Manager sends a "180 RINGING" SIP message to Session Manager with the name and number of the ringing user in standard SIP elements (e.g., Contact, PAI). Session Manager extracts the alerting party information and populates the Remote-Party-ID for the 180 RINGING back to Cisco UCME. Cisco UCME gets the name and number of the alerting Avaya; however Cisco UCME then sends a "183 Session Progress" message to the Cisco SIP Telephone with no Called Party Name in the "Remote-Party-ID" header (e.g. Remote-Party-ID: <sip:34502@192.45.131.1>;party=called;screen=no;privacy=off). Similar results were observed with Cisco SCCP telephones. A similar adaptation is performed on the 200 OK message when the Avaya telephone answers the call. The Cisco Telephone updates the display with the name and number received in the "200 OK" message (e.g. Remote-Party-ID: "Tom Avaya" <sip:34502@192.45.131.1>;party=called;screen=no;privacy=off) when the call is answered.

6.2.4 SIP Message Tracing

This section provides examples of Session Manager SIP message traces using the sample configuration. To configure tracing, select **Elements → Session Manager → System Tools → SIP Tracer Configuration** as shown below. **Section 9** of reference **[2]** provides details on the available SIP tracing and filtering options available via this screen.



Once the tracer configuration has been established, SIP message traces can be viewed by selecting **Elements → Session Manager → System Tools → SIP Trace Viewer**. The following screen shows an example of an expanded SIP INVITE message sent by Communication Manager to Session Manager. Note that SIP message tracing visibility via Session Manager is still possible when TLS is used between Communication Manager and Session Manager. That is, it is not necessary to change the transport to TCP in order to have visibility into the SIP messages as is typically the case using a line monitoring tool.



6.3 Verify Cisco Unified Communications Manager Express

The following commands can be used to troubleshoot calls over SIP trunks:

Show commands:

- **show ephone registered** - verifies ephone registration.
- **show voice register all** – displays all SIP configuration and register information.
- **show call active voice brief** - displays active call information for voice calls.

- **show voip rtp connection** - displays RTP named-event packet information (e.g. caller-ID number, IP Address, and ports).
- **show sip-ua call** - displays active call SIP user agent information.

Debug commands:

- **debug ccsip message** - displays all SIP messages.
- **debug ccsip calls** - displays SIP call trace information.
- **debug sccp message** - displays the sequence of the SCCP messages.
- **debug voip rtp session named – events** - enables debugging for RTP named events packets.

7. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager R6.0 SP 2 can interoperate with Cisco Unified Communications Manager Express R8.1 "IOS 15.1(2)T1" using SIP trunks via Avaya Aura® Session Manager R6.0 SP1. The following is a list of interoperability issues to note:

- Cisco SIP Telephones could not blind transfer active calls with Avaya Telephones to Cisco SCCP Telephones. All attempts to perform such operation failed, causing Cisco UCME to display memory allocation failure (MALLOCFAIL) messages. Attended transfer scenarios did not have this issue.
- During testing, Cisco UCME did not "shuffle" audio directly between the Avaya IP telephones and the Cisco IP telephones. All RTP traffic went through Cisco UCME.
- Calling and Called Party Name and Number displays may not be consistent in some cases for calls involving transfers, conferences, and call forwarding.
- Restricted presentation of display information is either off, (i.e., both name and number appear on the display), or privacy is full, where neither name nor number are presented on the display. That is, it is not possible to restrict only the number but display the name, or restrict only the name, and display the number.
- Privacy calls between Avaya telephones and Cisco <u>SIP</u> telephones did not work as expected:
    - Invoking privacy on calls between Avaya SIP telephones and Cisco SIP telephones resulted in privacy being invoked on both the calling and called parties. Cisco UCME returns a Remote-Party-ID header with "privacy=full" in the "180 RINGING" message, therefore restricting the presentation of display information on the Avaya telephones.
    - Invoking privacy on calls between Cisco SIP telephones and Avaya telephones, Cisco SIP telephones and Cisco SCCP telephones, or just between Cisco SIP telephones resulted in privacy being invoked on both the calling and called parties. Such calls include the proper SIP messaging between the Avaya and Cisco systems; however Cisco UCME sends a Remote-Party-ID header with "privacy=full" in the final "200OK" message to the Cisco SIP Telephone restricting the presentation of display information on the Cisco SIP Telephone.
    - Privacy calls between Avaya telephones and Cisco SCCP telephones worked as expected.

8. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Aura® Session Manager:

[1] Avaya Aura® Session Manager Overview, Doc ID 03-603323 (Issue 3) Release 6.0, available at http://support.avaya.com.

[2] Administering Avaya Aura® Session Manager, Doc ID 03-603324 (Issue 3) Release 6.0, available at http://support.avaya.com.

[3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325 (Issue 1.0) Release 6.0, available at http://support.avaya.com.

Avaya Aura® Communication Manager:

[4] *SIP Support in Avaya Aura® Communication Manager Running on Avaya S8xxx Servers*, Doc ID 555-245-206 (Issue 9), May, 2009, available at http://support.avaya.com.

[5] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509 (Issue 6.0), June 2010, available at http://support.avaya.com.

Product documentation for Cisco Systems products may be found at http://www.cisco.com

[6] *Cisco Unified Communications Manager Express System Administrator Guide*, September 29, 2010, Part Number: OL-10663-02

[7] *Cisco Unified Communications Manager Express Command Reference Guide*, February 27, 2009, Part Number: OL-10894-01

[8] *Cisco Call Manager Express (CME) SIP Trunking Configuration example*, November 16, 2007, Document ID: 91535

[9] *Cisco Unified CME Solution Reference Solution Design Guide*, Release 7.0(1), Part Number: OL-1062101-01

[10] *Cisco Unified Communications Manager Express: SIP Implementation Guide*, November 9, 2007, Document ID: 99946

[11] *Release Notes for Cisco IOS Release 15.1T*, Part Number: OL-22146-03 - http://www.cisco.com/en/US/docs/ios/15_1/release/notes/151TRN.pdf

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