Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1, with COLT VoIP Access SIP Service – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.1, Avaya Aura® Communication Manager Release 6.0.1, and the Acme Packet 3820 Session Border Controller release 6.1.0 with the COLT VoIP Access SIP Service.

The COLT VoIP Access SIP Service offer referenced within these Application Notes is designed for business customers with an Avaya SIP trunk solution. The service provides local and/or long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab., utilizing a COLT VoIP Access SIP circuit connection to the production COLT VoIP Access SIP Service.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.1, Avaya Aura® Communication Manager Release 6.0.1, and the Acme Packet 3820 Session Border Controller (SBC) release 6.1.0with the COLT VoIP Access SIP Service. The COLT VoIP Access SIP Service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

1.1. Interoperability Compliance Testing

Compliance testing scenarios for the configuration described in these Application Notes included the following:

- Inbound and outbound voice calls between telephones controlled by Avaya Aura® Communication Manager and the PSTN can be made using G.711A or G.729A codecs.
- Direct IP-to-IP Media (also known as "Shuffling") when applicable.
- DTMF using RFC 2833
    - Outbound call to PSTN application requiring post-answer DTMF (e.g., an IVR or voice mail system)
    - Inbound call from PSTN to Avaya CPE application requiring post-answer DTMF (Avaya vector digit collection steps)
- Hold / Retrieve with music on hold
- Call transfer.
- Conference calls
- SIP Diversion Header for call redirection
    - Call Forwarding
    - EC500
- Long hold time calls
- Inbound/outbound Fax using T.38

1.2. Support

1.2.1 Avaya

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com

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1.2.2 COLT

Telephone support is available on a national basis as shown in the following table, which shows the hotline number for each country where support is available, as well as a toll-free number if available.

Country	Hot Line	Toll-Free
Austria	(+43) 1 20 500 500	0800 880 990
Belgium	(+32) 2 790 16 29	0800 50701
Switzerland	(+41) 44 560 0720	0800 560 560
Denmark	(+45) 70 27 35 59	
France	(+33) 1 70995600	0800 948 888
Germany	(+49) 69 56606 3115	0800 855 4444
Ireland	(+34) 9355 02568	1800 944040
Italy	(+39) 0230 329 550	0800 909 377
Netherlands	(+31) 20 888 2433	0800 265 8023
Portugal	(+351) 211 200 222	808 780 222
Spain	(+34) 913 206018	901 888400
Sweden	(+46) 8781 8333	
UK	(+44) 203 140 2023	0800 136 166

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1.3. Known Limitations

No limitations were found during testing

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2. Reference Configuration

Figure 1 shows the sample configuration. As shown in **Figure 1**, the Acme Packet 3820 SBC receives traffic from the COLT VoIP Access SIP Service. The COLT VoIP Access SIP Service provided Direct Inward Dial (DID) 10 digit numbers. These DID numbers were mapped by Avaya Aura® Communication Manager to Avaya telephone four-digit extensions by deletion of all but the four trailing digits.

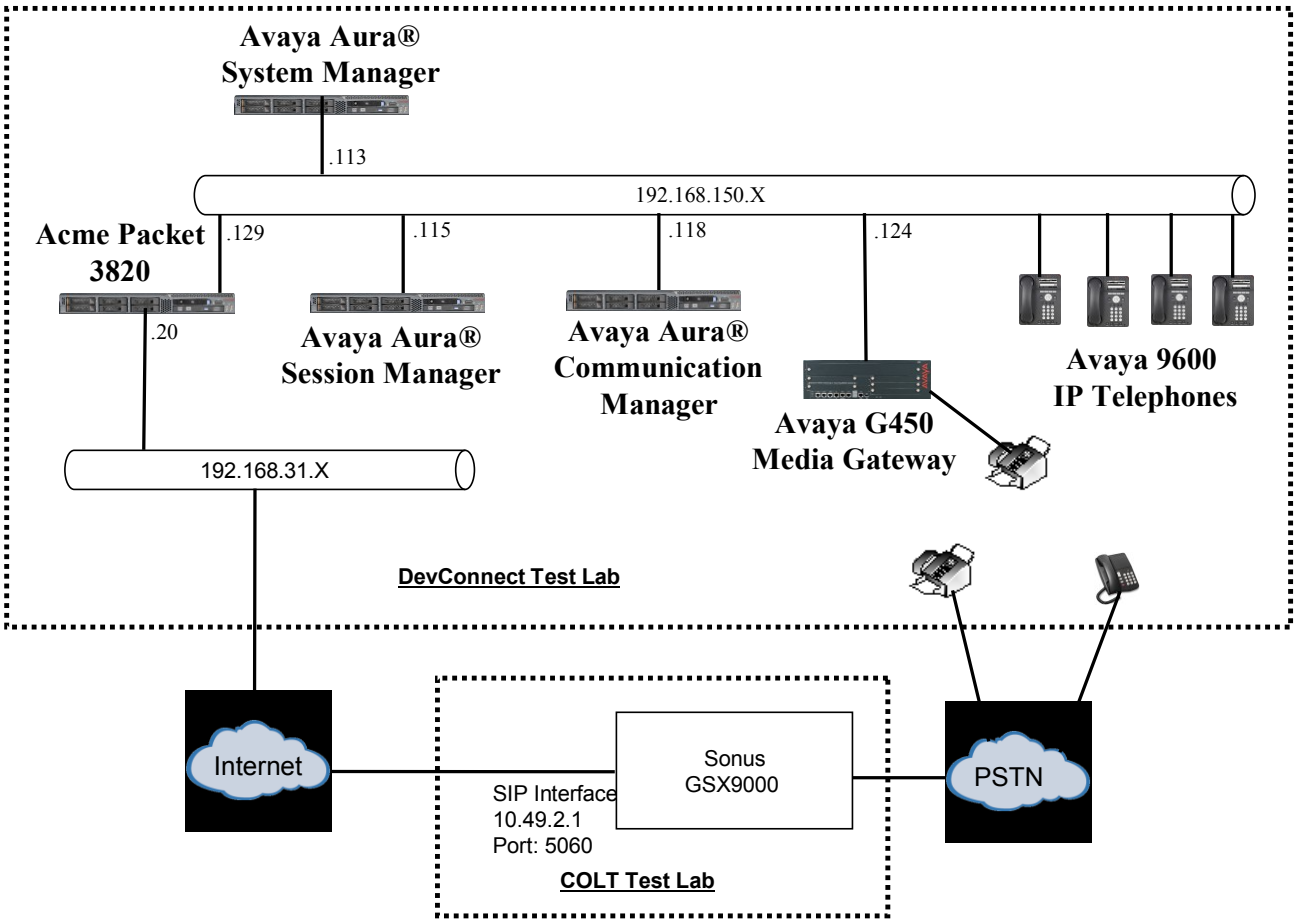

Figure 1: Avaya Interoperability Test Lab Configuration

The following components were used in the sample configuration:

- Sonus GSX
- Acme Packet 3820 Session Border Controller (SBC)
- Avaya Aura® Communication Manager Release 6
- Avaya Aura® Session Manager Release 6
- Avaya Aura® System Manager Release 6
- Avaya G450 Media Gateway
- Avaya 9600 Series IP telephones using the H.323 software bundle
- Avaya 9600 Series IP telephones using the SIP software bundle

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Some components described within this document use network names defined by a DNS server at address 192.168.200.2 (not shown in above diagram). This server defines the following DNS names used within the "inside" network used within this document:

Item	Hostname	IP Addr
System Manager domain0	smgrdom0.aura.dcffm	192.168.150.111
System Manager console	smgrcon.aura.dcffm	192.168.150.112
System Manager server	smgr.aura.dcffm	192.168.150.113
Session Manager server	sm.aura.dcffm	192.168.150.114
Session Manager asset	asset.aura.dcffm	192.168.150.115
Communication Manager domain0	cm1dom0.aura.dcffm	192.168.150.116
Communication Manager console	cm1con.aura.dcffm	192.168.150.117
Communication Manager server	cm1.aura.dcffm	192.168.150.118
G450 Media gateway	cm1gw.aura.dcffm	192.168.150.124
Acme 3820 SBC	sbc.aura.dcffm	192.168.150.129

Note – These Application Notes describe the provisioning used for the sample configuration shown in **Figure 1**. Other configurations may require modifications to the provisioning described in this document.

2.1. History Info and Diversion Headers

The COLT VoIP Access SIP Service does not support SIP History Info Headers. Instead, the COLT VoIP Access SIP Service requires that SIP Diversion Header be sent for redirected calls. The Avaya Aura® Communication Manager SIP trunk group form provides options for specifying whether History Info Headers or Diversion Headers are sent.

The Avaya Aura® Communication Manager call forwarding or Extension to Cellular (EC500) features may be used for call scenarios testing Diversion Header.

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3. Equipment and Software Validated

The following Avaya components were used run on Avaya S8800 servers.

Software	System Platform	Template
Avaya Aura® Communication Manager	6.0.2.0.5	6.1.0.0.2350 Patch: 00.1.510.1-18621
Avaya Aura® Session Manager	N/A	6.1.0.0.610023
Avaya Aura® System Manager	6.0.2.0.5	6.1.4.0 + SP0.r873

Table 1: Equipment and Software Used in the Sample Configuration

The following equipment and software were used in the sample configuration.

Equipment	Version
Avaya G450 Media Gateway	31.17.1
Avaya 9600-Series Telephones (H.323)	3.1.1
Avaya 9600-Series Telephones (SIP)	2.6.4
Acme Packet 3820 Session Border Controller	Firmware SCX6.1.0 MR-4 Patch 2 (Build 682)
Sonus PSX	7.3.3F6
Sonus GSX	7.3.3R1
Sonus DSI	7.3.3R1
Sonus EMS	7.3.3R2

Table 1: Equipment and Software Used in the Sample Configuration

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4. Avaya Aura® Communication Manager Configuration

4.1. Verify Licensed Features

The Communication Manager license file controls customer capabilities. Contact an authorized Avaya representative for assistance if a required feature needs to be enabled.

On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** is sufficient for the combination of trunks to the COLT VoIP Access SIP Service offer and any other SIP applications. Each call from a non-SIP endpoint to the COLT VoIP Access SIP Service uses one SIP trunk for the duration of the call.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
          Maximum Concurrently Registered IP Stations: 18000 0
            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: 1000  0
                      Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                            Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```


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On **Page 3** of the ***display system-parameters customer-options*** form, verify that the **ARS** feature is enabled.

```
display system-parameters customer-options                    Page   3 of  11
                              OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                      CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                        CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y  Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                      DCS (Basic)? y
          ASAI Link Core Capabilities? y                DCS Call Coverage? y
          ASAI Link Plus Capabilities? y               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                        DS1 MSP? y
                                ATMS? y            DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

On **Page 4** of the ***display system-parameters customer-options*** form, verify that the **Enhanced EC500**, **IP Trunks**, **IP Stations**, and **ISDN-PRI** features are enabled. If the use of SIP REFER messaging or send-only SDP attributes will be required (see also Section 4.9), verify that the **ISDN/SIP Network Call Redirection** feature is enabled.

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

   Emergency Access to Attendant? y                          IP Stations? y
            Enable 'dadmin' Login? y
            Enhanced Conferencing? y                   ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                      ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                            ISDN-PRI? y
              ESS Administration? y            Local Survivable Processor? n
         Extended Cvg/Fwd Admin? y                 Malicious Call Trace? y
     External Device Alarm Admin? y              Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
               Flexible Billing? n
  Forced Entry of Account Codes? y              Multifrequency Signaling? y
     Global Call Classification? y        Multimedia Call Handling (Basic)? y
            Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y              Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```


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On **Page 5** of the ***display system-parameters customer-options*** form, verify that the **Private Networking** and **Processor Ethernet** features are enabled.

```
display system-parameters customer-options                    Page   5 of  11
                              OPTIONAL FEATURES

                Multinational Locations? n           Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                     Multiple Locations? n
                                                System Management Data Transfer? n
          Personal Station Access (PSA)? y             Tenant Partitioning? y
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                 Port Network Support? y             Time of Day Routing? y
                      Posted Messages? y      TN2501 VAL Maximum Capacity? y
                                                     Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
               Processor and System MSP? y
                    Processor Ethernet? y               Wideband Switching? y
                                                                Wireless? n
                         Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

4.2. Dial Plan

In the sample configuration the Avaya CPE environment uses four digit local extensions, such as 2xxx. Trunk Access Codes (TAC) are 4 digits in length and begin with *8. The Feature Access Code (FAC) to access ARS is the single digit 0. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used.

The dial plan is modified with the ***change dialplan analysis*** command.

```
change dialplan analysis                                      Page   1 of  12
                           DIAL PLAN ANALYSIS TABLE
                                 Location: all            Percent Full: 2

   Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
   String   Length Type     String   Length Type     String   Length Type
  0          1   fac
  1          5   ext
  2          4   ext
  3          4   ext
  4          4   ext
  5          4   ext
  *2         4   fac
  *4         3   dac
  *7         3   fac
  *8         4   dac
  *9         2   dac
```


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4.3. Node Names

Node names are mappings of names to IP Addresses that can be used in various screens. The following abridged "change node-names ip" output shows relevant node-names in the sample configuration. As shown in bold, the node name for Session Manager is "asset" with IP Address 192.168.150.115. The node name and IP Address (192.168.150.118) for the Processor Ethernet "procr" appears automatically due to the web configuration in Section 4.1.

```
change node-names ip                                                  Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
asset               192.168.150.115
default             0.0.0.0
procr               192.168.150.118
procr6              ::
```

4.4. IP Interface for procr

The "add ip-interface procr" or "change ip-interface procr" command can be used to configure the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the PE for SIP Trunk Signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

```
change ip-interface procr                                             Page   1 of   2
                                  IP INTERFACES


                 Type: PROCR
                                                        Target socket load: 19660

      Enable Interface? y                            Allow H.323 Endpoints? y
                                                      Allow H.248 Gateways? y
        Network Region: 1                              Gatekeeper Priority: 5


                                 IPV4 PARAMETERS
             Node Name: procr                     IP Address: 192.168.150.118


           Subnet Mask: /24
```


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4.5. Network Regions for Gateway, Telephones

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway is in region 1. The MM711 Analog interface serves as an interface to the Fax device.

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

                                Type: g450

Slot   Module Type            Name                    DSP Type  FW/HW version
 V1:   S8300                  ICC MM                  MP80      45   6
 V2:   MM712                  DCP MM
 V3:   MM711                  ANA MM
 V4:
 V5:
 V6:
 V7:
 V8:                                                Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

The following screen shows IP Network Region 1 configuration. The "Authoritative Domain" is set to the enterprise SIP domain "aura.dcffm". This should match the value used in **Section 5.1**.

```
change ip-network-region 1                                      Page   1 of  20
                             IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: aura.dcffm
    Name: local
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 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
```


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4.6. IP Codec Sets

The following screen shows the configuration for codec set to be used for local and external calls. In general, an IP codec set is a list of allowable codecs in priority order.

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

On **Page 2** of the form configure the **FAX Mode** field to T.38**,** which is supported by COLT.

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

                         IP Codec Set

                             Allow Direct-IP Multimedia? n



                   Mode               Redundancy
   FAX             t.38-standard       0
   Modem           off                 0
   TDD/TTY         off                 3
   Clear-channel   n                   0
```

4.7. SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups that use the Processor Ethernet. Each signaling group has a **Group Type** of "sip", a **Near-end Node Name** of "procr", and a **Far-end Node Name** of "asset". In the example screens, the **Transport Method** for all signaling groups is "tcp". In production, TLS transport between Communication Manager and Session Manager may be used. The **Enable Layer 3 Test** field is enabled to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to "rtp-payload", which corresponds to RFC 2833.

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```
change signaling-group 1                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n         Transport Method: tcp
       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: asset
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain: aura.dcffm
                                              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
```

4.8. SIP Trunk Group

This section illustrates the configuration of the SIP Trunks Groups corresponding to the SIP signaling groups from the previous section.

The following shows **Page 1** for trunk group 1, which will be used for PSTN calls to and from COLT. The **Number of Members** field defines how many simultaneous calls are permitted for the trunk group. The **Service Type** field should be set to "public-ntwrk" for the trunks that will handle calls with COLT.

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Local-to-CM               COR: 1        TN: 1        TAC: *801
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                          Member Assignment Method: auto
                                                    Signaling Group: 1
                                                  Number of Members: 10
```

The following shows **Page 2** for trunk group 1. All parameters shown are default values, except for the **Preferred Minimum Session Refresh Interval**, which has been changed from the default 600 to 900.

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```
add trunk-group 1                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                        Redirect On OPTIM Failure: 9000

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




             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

The following shows **Page 3** for trunk group 1. All parameters except those in bold are default values. The **Numbering Format** field must be set to **private**.

```
add trunk-group 1                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                    Maintenance Tests? y



                     Numbering Format: private
                                           UUI Treatment: service-provider

                                            Replace Restricted Numbers? n
                                           Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no



 Show ANSWERED BY on Display? y
```

The following shows **Page 4** for trunk group 1. The **PROTOCOL VARIATIONS** page is one reason why it can be advantageous to configure incoming calls from COLT to arrive on specific signaling groups and trunk groups. Although not strictly necessary, the **Telephone Event Payload Type** has been set to 101 to match COLT configuration. In the testing associated with these Application Notes, transfer testing using INVITE was successfully completed with the **Network Call Redirection** flag set to default value "n".

For redirected calls, COLT supports the Diversion header, but not the History-Info header. Communication Manager can send only the Diversion header by marking **Send Diversion Header** to "y" and **Support Request History** to "n".

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```
add trunk-group 1                                              Page   4 of  21
                              PROTOCOL VARIATIONS

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


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                              Enable Q-SIP? n
```

4.9. Route Pattern Directing Outbound Calls to COLT

Route pattern 1 will be used for calls destined for the PSTN via the COLT VoIP Access SIP Service. Digit manipulation can be performed on the called number, if needed, using the **No. Del Dgts** and **Inserted Digits** parameters. Digit manipulation can also be performed by Session Manager.

If desired, one or more alternate Communication Manager trunks can be listed in the route pattern so that the Look-Ahead Routing (**LAR**) "next" setting can route-advance to attempt to compete the call using alternate trunks should there be no response or an error response from the far-end. Examples are provided in references [PE] and [LAR]. The **Numbering Format** should be set to **unk-unk**.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: PSTN
                                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: 1    0                        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                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```


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4.10. Public Numbering

The "change public-unknown-numbering" command may be used to define the format of numbers sent to COLT in SIP headers such as the "From" and "PAI" headers. In general, the mappings of internal extensions to COLT DID numbers may be done in Session Manager (via Digit Conversion in adaptations) or in Communication Manager (via public-unknown-numbering, and incoming call handling treatment for the inbound trunk group).

In the bolded rows shown in the example abridged output below, all Communication Manager extensions are mapped to a DID numbers by adding the sequence 44111111 to the beginning of the number, when the call uses trunk group 1. Alternatively, Communication Manager can send the extension to Session Manager by leaving the **CPN Prefix** field blank and setting the **CPN Len** to 4 (i.e., similar to the first row in the screen), and Session Manager can adapt the number to the COLT DID.

```
change public-unknown-numbering 4                              Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext           Trk       CPN            CPN
Len Code          Grp(s)    Prefix         Len
                                                 Total Administered: 1
 4  2             1         44111111       12       Maximum Entries: 9999

                                                 Note: If an entry applies to
                                                 a SIP connection to Avaya
                                                 Aura(tm) Session Manager,
                                                 the resulting number must
                                                 be a complete E.164 number.
```

4.11. ARS Routing For Outbound Calls

Although not illustrated in these Application Notes, location-based routing may be configured so that users at different locations that dial the same telephone number can have calls choose different route-patterns. Various example scenarios for a multi-location network with failover routing are provided in reference [PE]. In these Application Notes, the ARS "all locations" table directs ARS calls to specific SIP Trunks to Session Manager. Appropriate ARS entries can be added to match the various dial patterns (e.g., long distance, service numbers, etc.) to be sent to COLT.

The following screen shows a sample specific ARS configuration for a number that will be dialed in the Verification section of these Application Notes. If a user dials the ARS access code defined in **Section 4.2**, followed by the number beginning with "0" with a length between 7 and 15 digits, the call will select route pattern 1.

```
change ars analysis 0                                          Page   1 of   2
                          ARS DIGIT ANALYSIS TABLE
                                Location: all            Percent Full: 0

         Dialed           Total     Route    Call   Node  ANI
         String          Min  Max  Pattern   Type   Num   Reqd
    0                    7    15    1        pubu         n
```


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4.12. Incoming Call Handling Treatment for Incoming Calls

In general, the "incoming call handling treatment" for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by COLT is unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. As an example, the following screen illustrates a conversion of DID number 0044111111xxxx to extension xxxx. Both Session Manager digit conversion and Communication Manager incoming call handling treatment methods were tested successfully.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of  30
                        INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             14 0044111111       10
```

4.13. Public Unknown Numbering

Although not specifically related to COLT, this section shows the public unknown numbering configuration associated with the calls using trunk group 1. The bold row configures any four digit number beginning with 2 that uses trunk group 1 to add the prefix shown to calls made via the COLT SIP trunk.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                Total Administered: 2
 4  2             1         0044111111      14       Maximum Entries: 9999
```


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4.14. Avaya Aura® Communication Manager Stations

In the sample configuration, four digit station extensions were used with the format 2xxx. The following abbreviated screen shows an example extension for an Avaya H.323 IP telephone.

```
add station 2372                                                Page   1 of   5
                                        STATION

Extension: 2372                            Lock Messages? n              BCC: 0
     Type: 9640                            Security Code: 123456          TN: 1
     Port: S00014                        Coverage Path 1:                COR: 1
     Name: extn 2372                     Coverage Path 2:                COS: 1
                                         Hunt-to Station:
STATION OPTIONS
                                              Time of Day Lock Table:
              Loss Group: 19          Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 2372
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                   Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? y

                                                 IP Video Softphone? n
                                Short/Prefixed Registration Allowed: default

                                                Customizable Labels? y
```

4.15. EC500 Configuration for Diversion Header Testing

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 30002. Use the command **change off-pbx-telephone station mapping x** where *x* is the Communication Manager station (e.g. **2373**).

- **Station Extension** – This field will automatically populate
- **Application** – Enter **EC500**
- **Dial Prefix** – Enter a prefix (e.g., 1) if required by the routing configuration
- **Phone Number** – Enter the phone that will also be called (e.g., **492222222222**)
- **Trunk Selection** – Enter **ARS**. This means ARS will be used to determine how Communication Manager will route to the **Phone Number** destination.
- **Config Set** – Enter **1**
- Other parameters can retain default values

```
change off-pbx-telephone station-mapping 2373                    Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2373             EC500    0    -     492222222222     ars         1
```


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4.16. Saving Avaya Aura® Communication Manager Configuration Changes

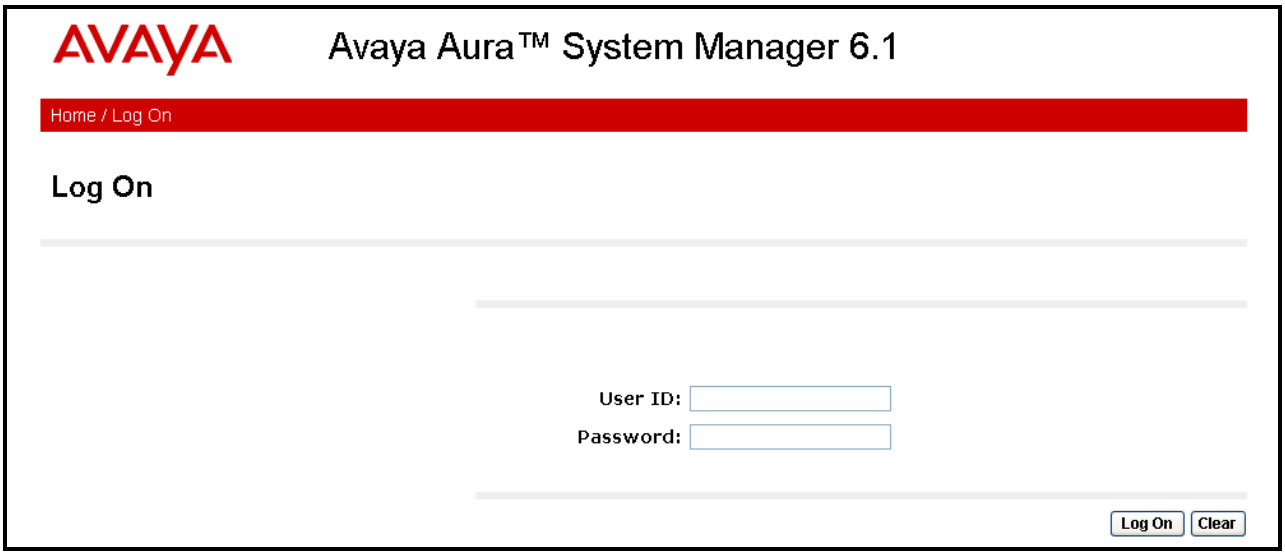
The command "save translation all" can be used to save the configuration.

5. Configure Avaya Aura® Session Manager Release 6.1

This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

Note – The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two. For more information on Session Manager see **[3]**.

Session Manager is managed via System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **Username** and **Password** and press the **Log On** button (not shown).



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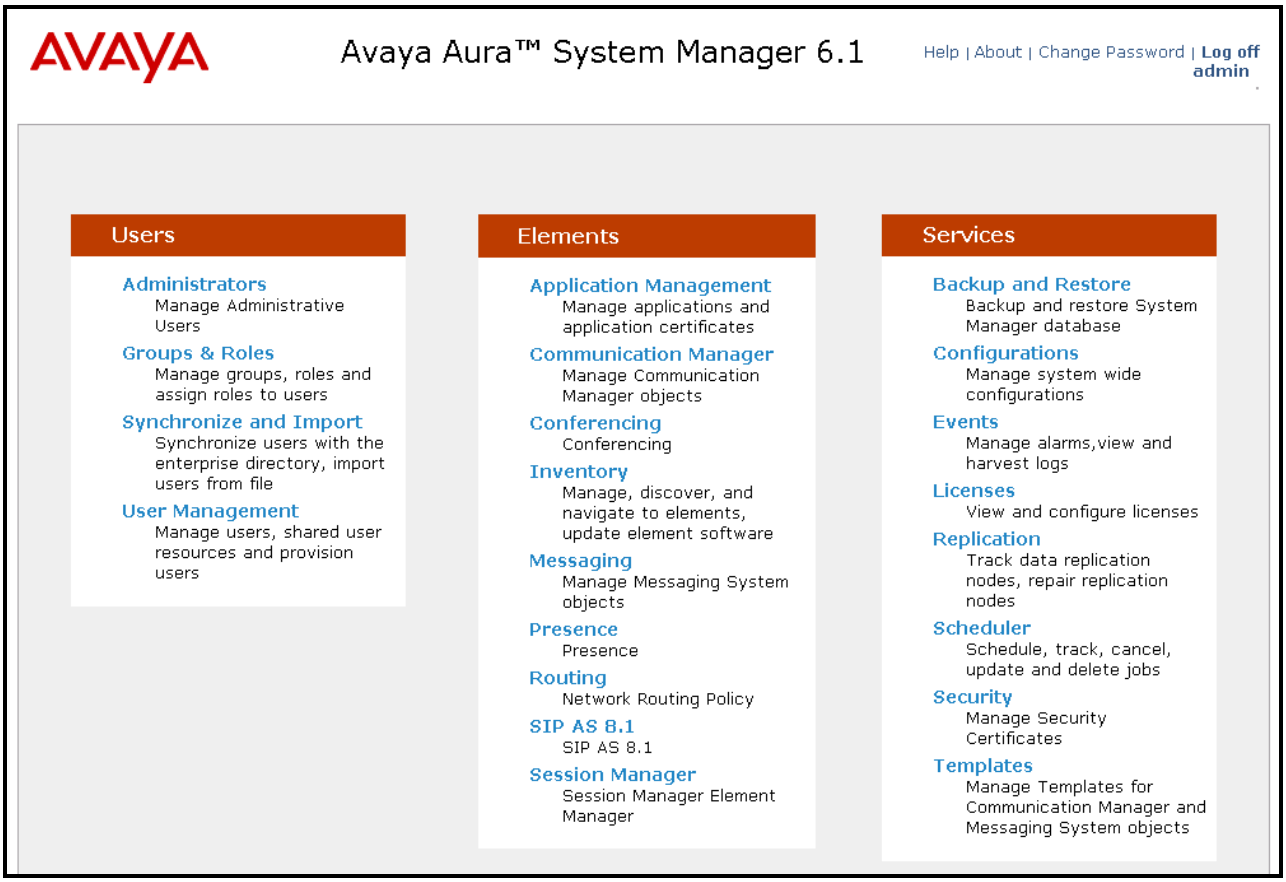
Once logged in, a **Home Screen** is displayed.



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When Routing is selected, the right side outlines a series of steps.

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The sub-sections that follow are in the same order as the steps outlined under **Introduction to Network Routing Policy (NRP)** in the abridged screen shown below. In these Application Notes, all these steps are illustrated with the exception of Step 9, since "Regular Expressions" were not used.

Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"

(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 8: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

Step 9: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of the routing application "Dial patterns". That's why this overall routing workflow can be interpreted as

"Dial Pattern driven approach to define Routing Policies"

That means (with regard to steps listed above):

Step 7: "Routing Polices" are defined

Step 8: "Dial Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)

Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

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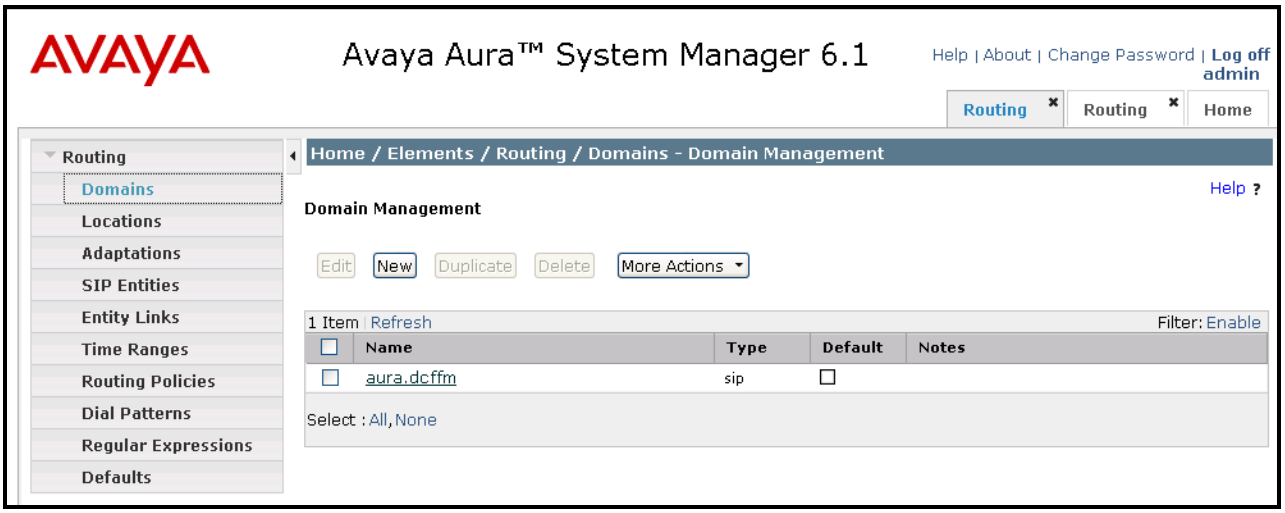
5.1. Domains

To view or change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button after changes are completed.

The following screen shows the list of configured SIP domains. The domain "aura.dcffm" is not known to the COLT production service. The domain name should match the one used in the ip-network-region described in **Section 4.5**.



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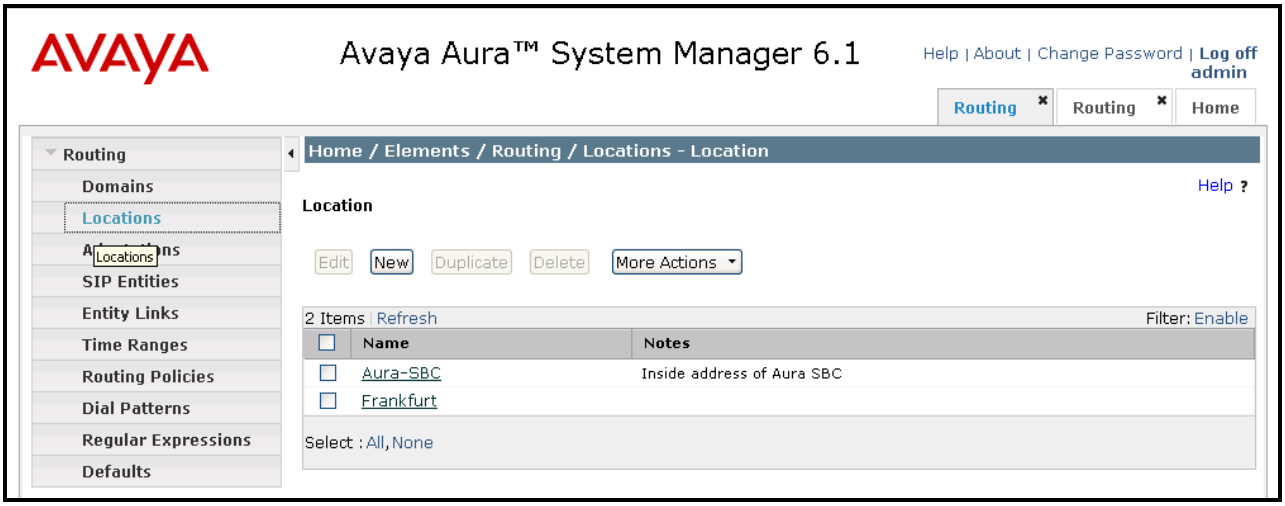
5.2. Locations

To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.



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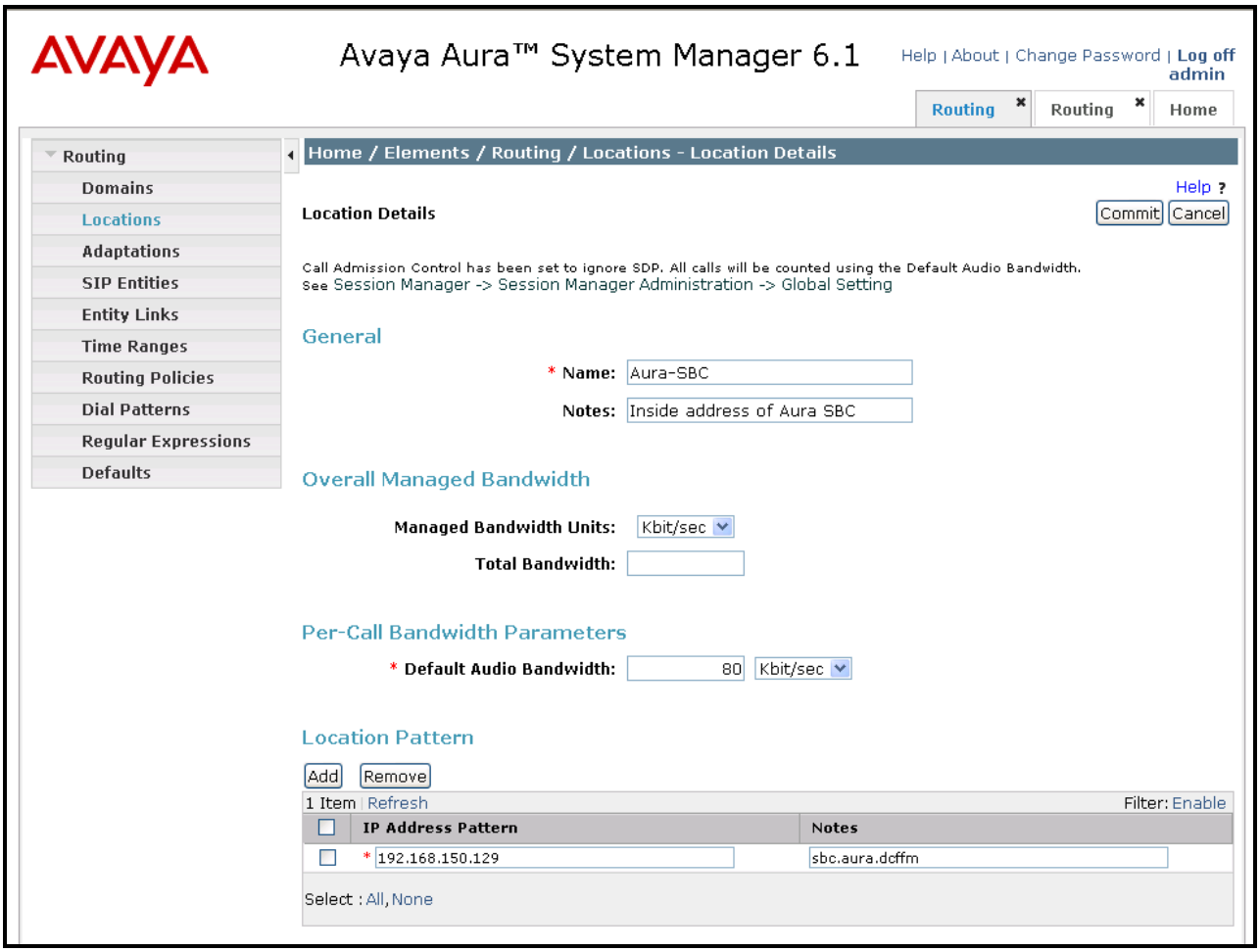
The following screen shows the location details for the location named "Aura-SBC", corresponding to the Acme Packet 3820 Session Border Controller. Later, the location with name "Aura-SBC" will be assigned to the corresponding SIP Entity. The IP Address 192.168.150.129 of the inside (private) interface of the SBC is entered in the **IP Address Pattern** field.



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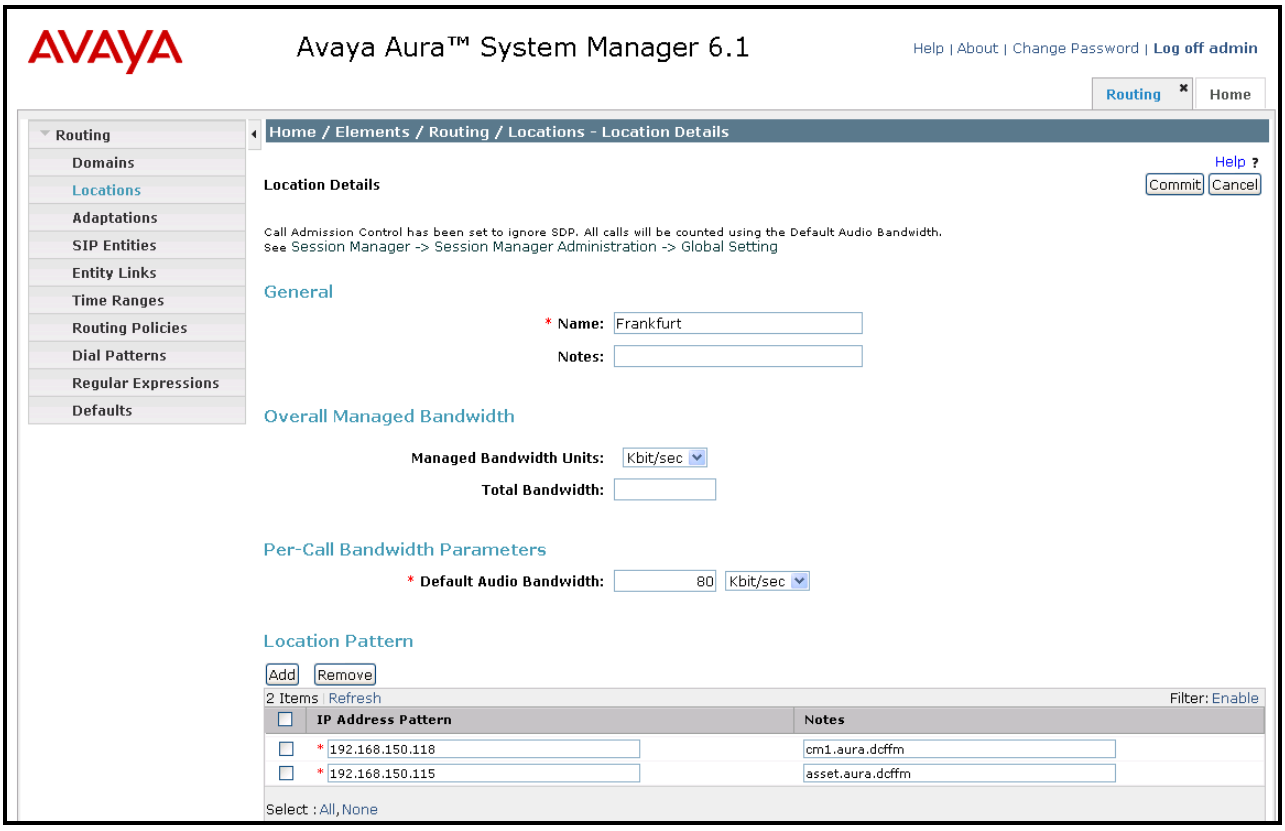
The following screen shows the location details for the location named “Frankfurt”. The SIP Entities and associated IP Addresses for this location correspond to the shared components of the Avaya Interoperability Lab test environment, such as Communication Manager Release 6.0.1, Session Manager Release 6.1.



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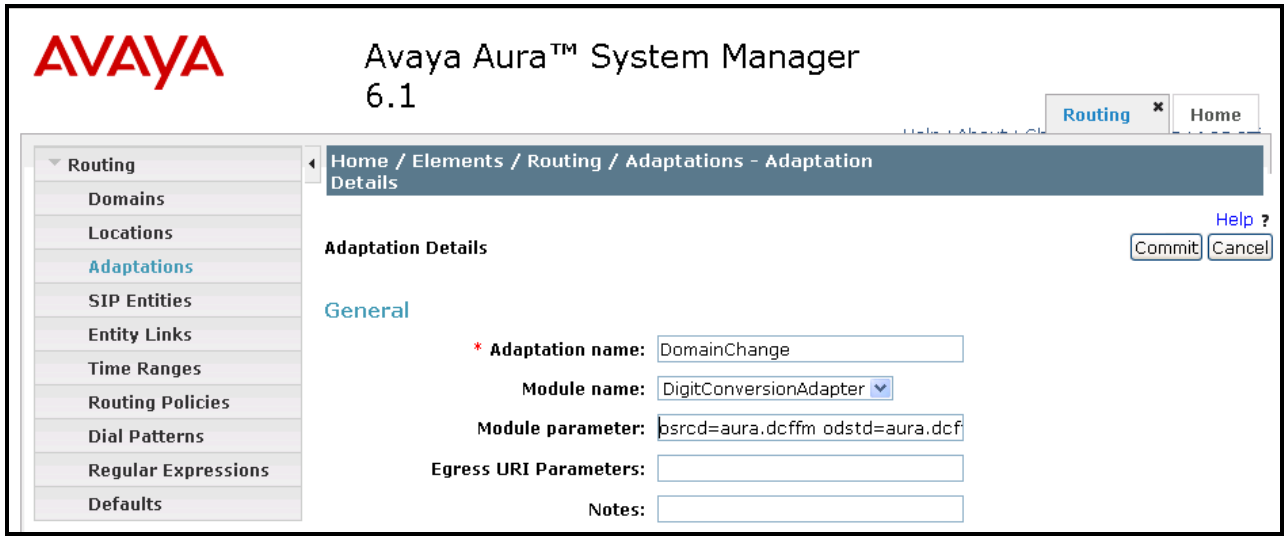
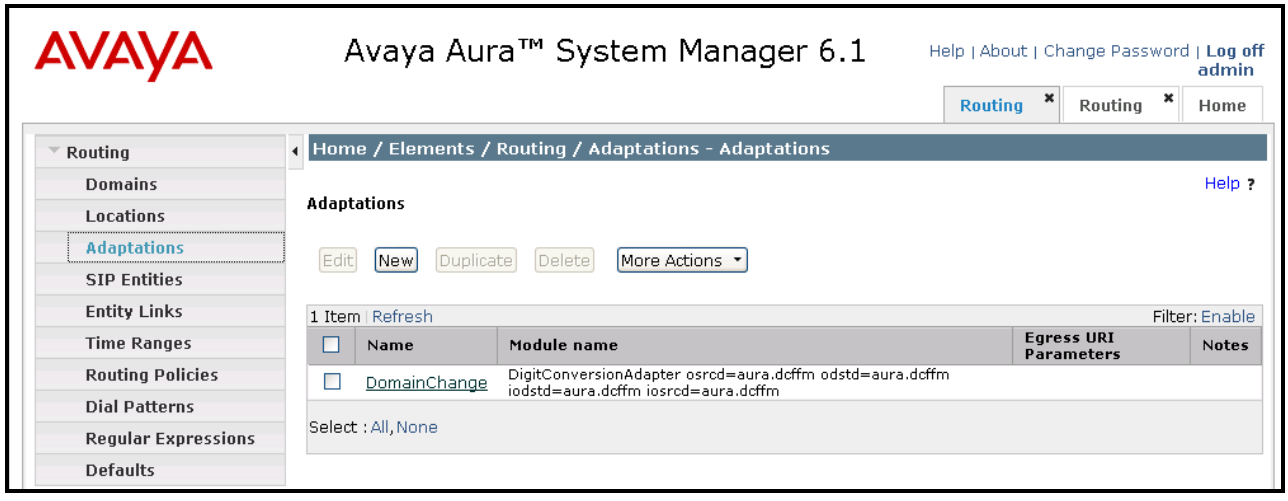
5.3. Adaptations

To view or change adaptations, select **Routing → Adaptations**. Click the **New** button to add an adaptation. Enter an appropriate "Adaptation name" to identify the adaptation. Select "DigitConversionAdapter" from the "Module name" drop-down menu. For the "Module parameter" field, the "osrcd", "odstd", "iodstd", and "iosrcd" parameters should each be assigned to the domain name defined in **Section 5.1**. Click the **Commit** button after changes are completed.



The following screen shows the list of adaptations in the sample configuration. For the tested configuration, all domain names for both incoming and outgoing messages are converted to "aura.dcffm".



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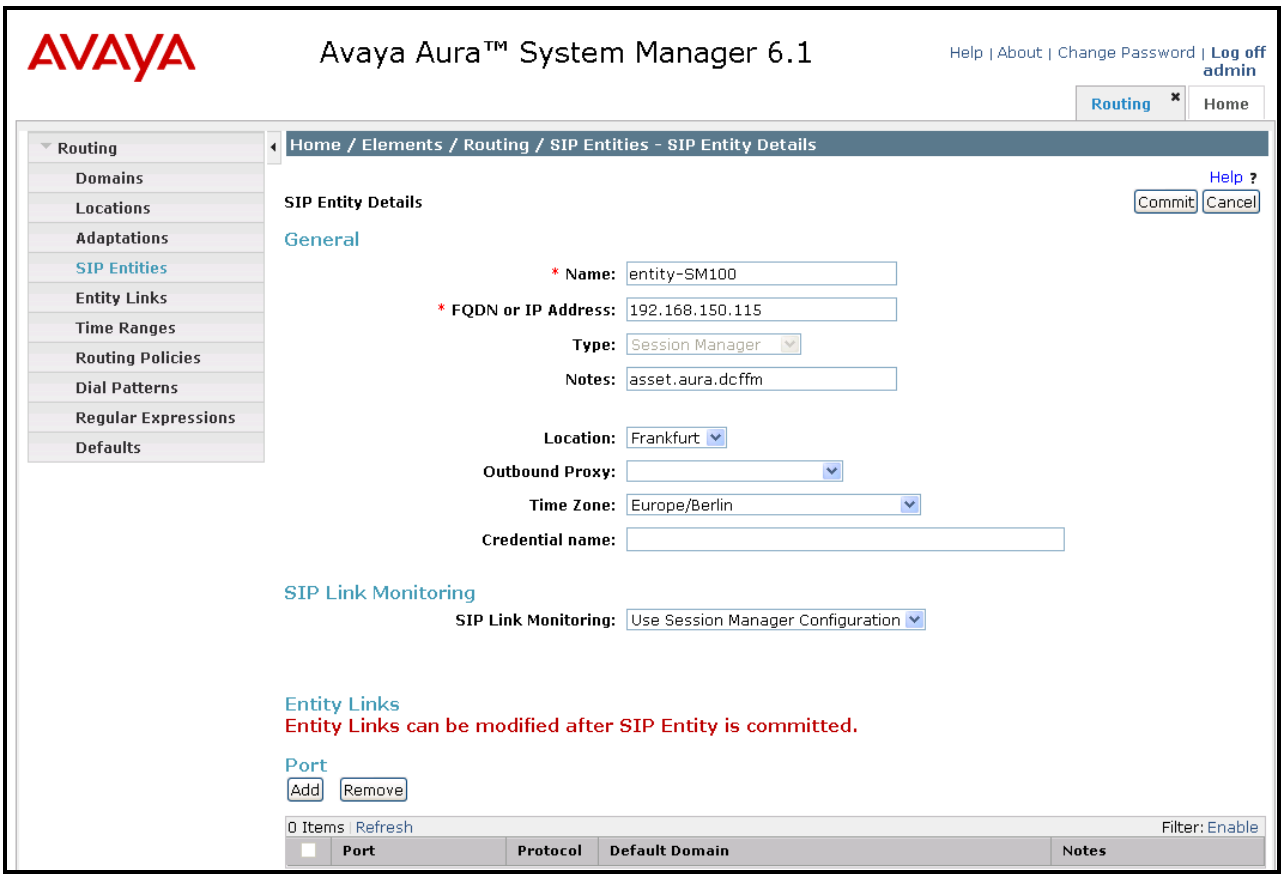
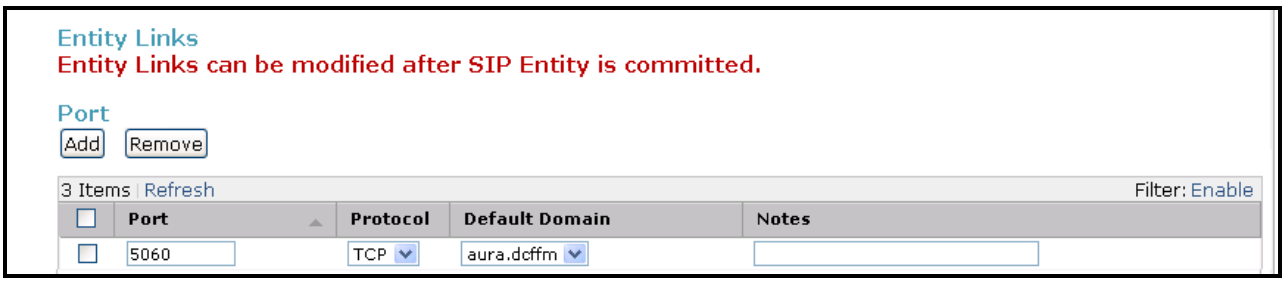
5.4. SIP Entities

To view or change SIP elements, select **Routing → SIP Entities**. For the **Name** parameter, enter an identifier to be assigned to the Session Manager interface. For the **FDQN or IP Address** parameter, enter the address value to be assigned to the Session Manager interface. For **Type**, select "Session Manager" from the drop-down menu. For **Location**, select the value assigned to the Session Manager in **Section 5.2**. Select the appropriate **Time Zone** for the Session Manager from the drop-down menu.



Click the **Add** button to configure a new port. TCP is used in the sample configuration for improved visibility during testing; TLS may be used in production.



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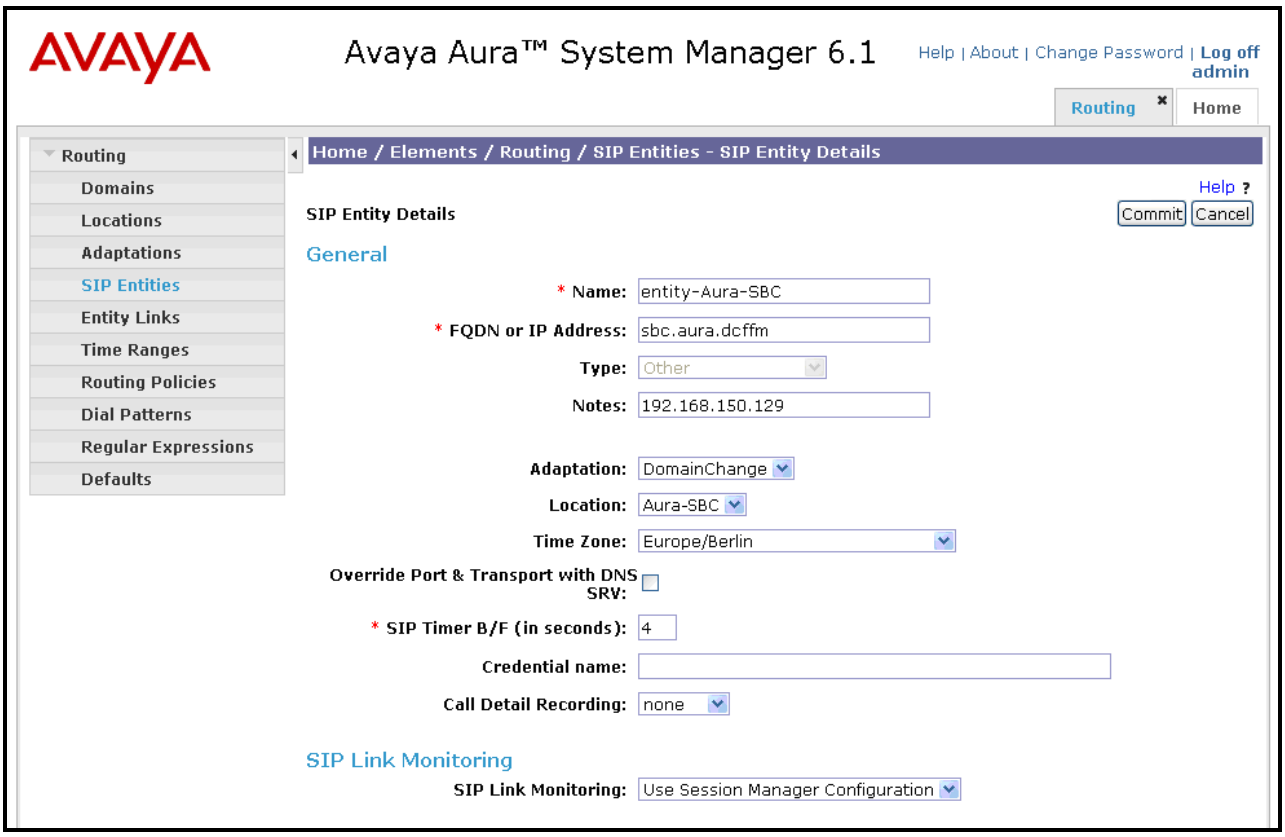
The following screen shows the **SIP Entity Details** for the Acme Packet 3820 SBC. Enter a descriptive **Name** to identify this entity. The **FQDN or IP Address** field is configured with the Acme Packet 3820 SBC inside IP Address (192.168.150.129). "Other" is selected from the **Type** drop-down menu for SBC SIP Entities. The SBC has been assigned to **Location** "Aura-SBC" shown in Section 5.2, and the "DomainChange" adapter shown in Section 5.3 is applied.



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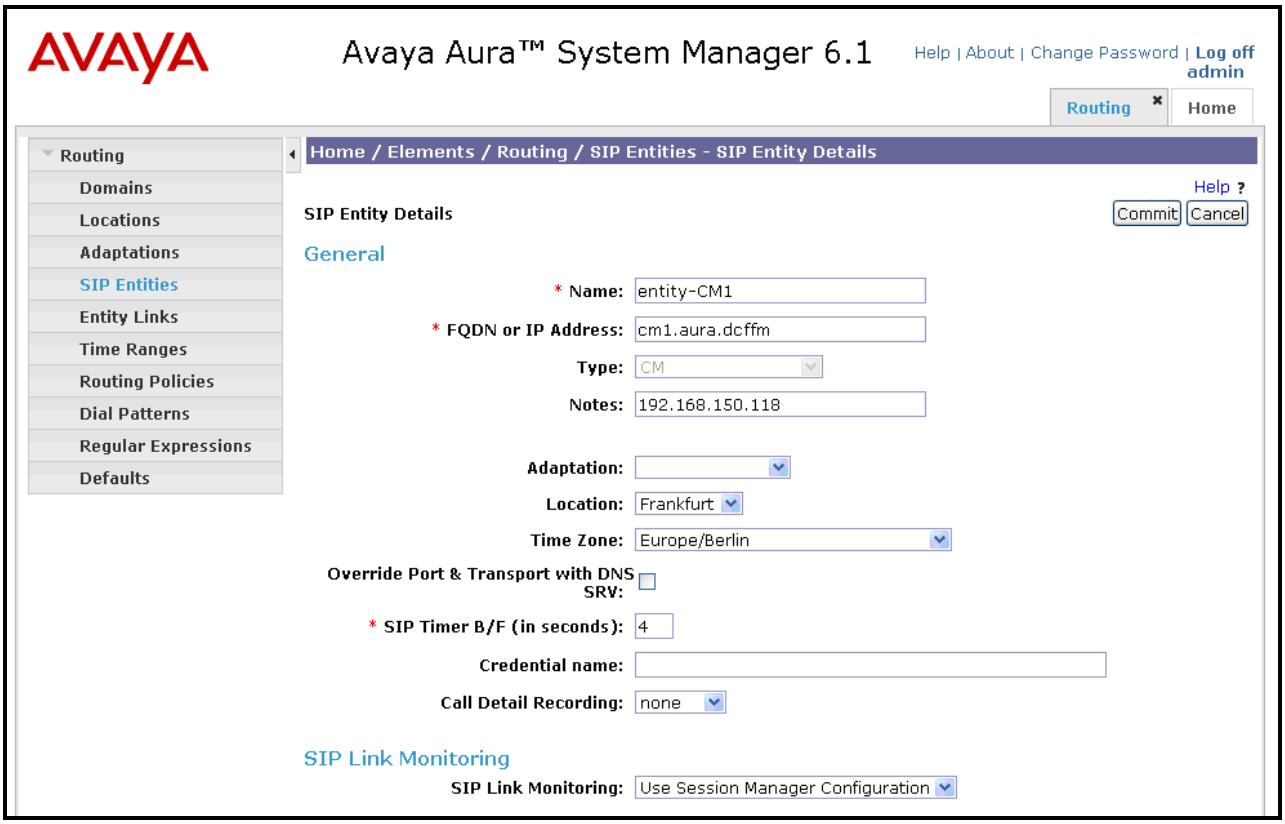
The following screen shows a portion of the **SIP Entity Details** corresponding to an Avaya Aura® Communication Manager SIP Entity named "entity-CM1" The **FQDN or IP Address** field contains the IP Address of the "processor ethernet" (192.168.150.118). In systems with Avaya G650 Media Gateways containing C-LAN cards, C-LAN cards may also be used as SIP entities, instead of, or in addition to, the "processor ethernet". "CM" is selected from the **Type** drop-down menu.



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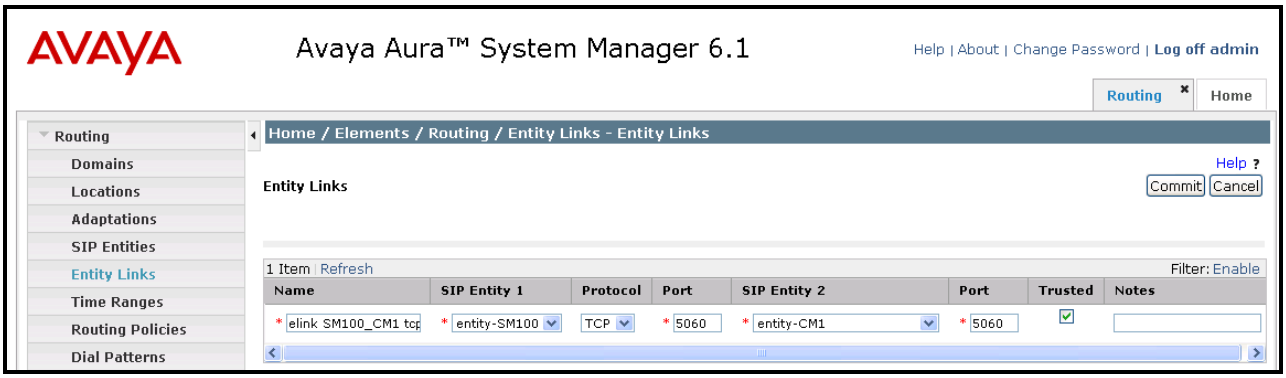
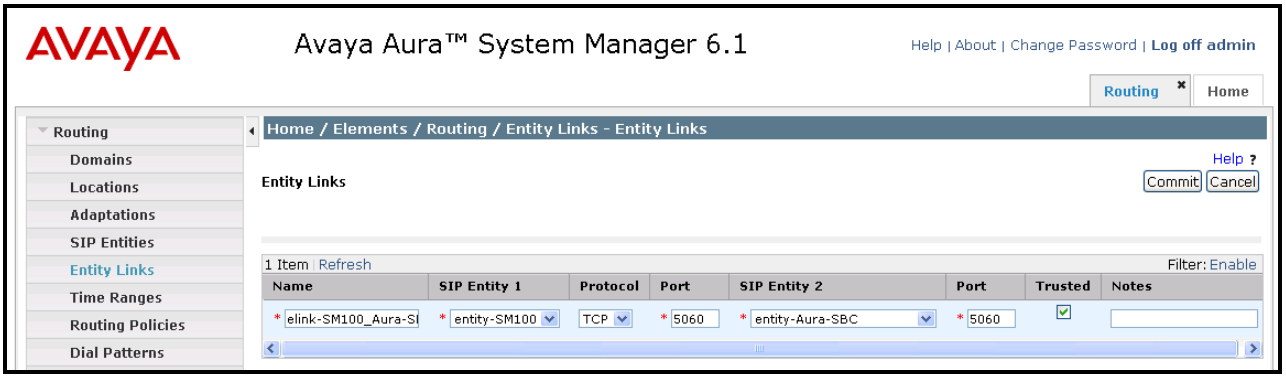
5.5. Entity Links

To view or change Entity Links, select **Routing → Entity Links**. Click the **New** button to add a link for Communication Manager. Assign an appropriate Name, and select the Session Manager entity as **SIP Entity 1,** and the Communication Manager entity as **SIP Entity 2**. Assign the **Protocol** as "TCP", select **Port** 5060, and click "Commit".



Click the **New** button to add a link for the Acme Packet 3820 SBC. Assign an appropriate Name, and select the Session Manager entity as **SIP Entity 1,** and the SBC entity as **SIP Entity 2**. Assign the **Protocol** as "TCP", select **Port** 5060, and click "Commit".



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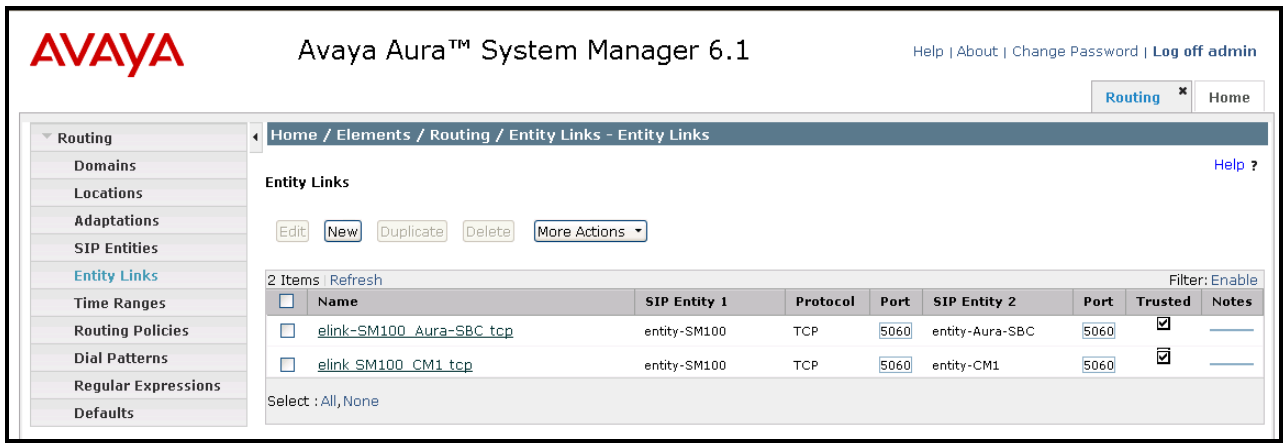
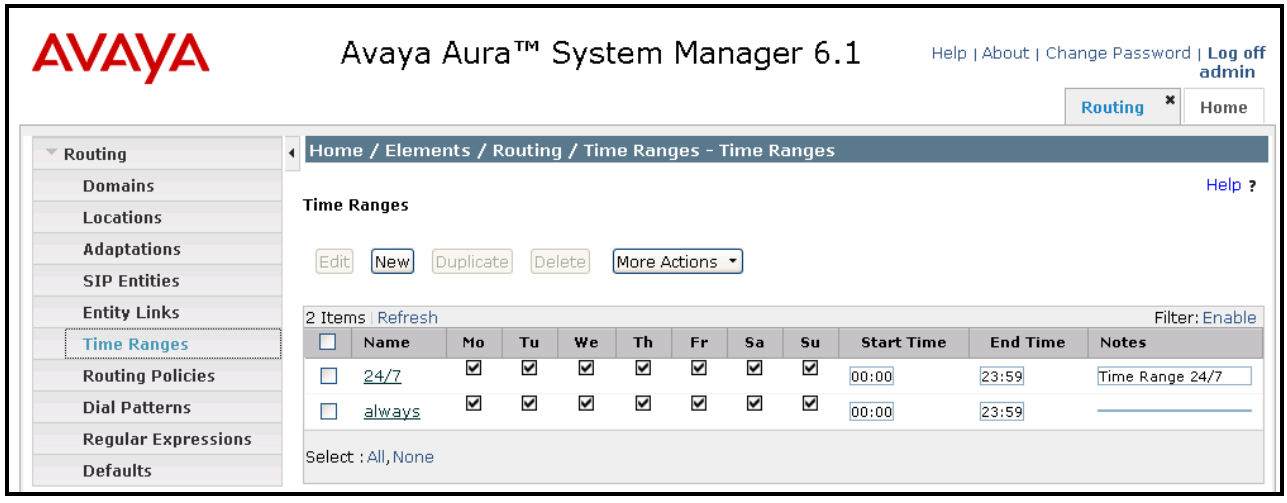
The following screen shows the list of configured links. Each of the links uses the entity named "entity-SM100" as **SIP Entity 1**, and the appropriate entity, such as "entity-Aura-SBC tcp" for **SIP Entity 2**.



5.6. Time Ranges

To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the "24/7" range since time-based routing was not the focus of these Application Notes.



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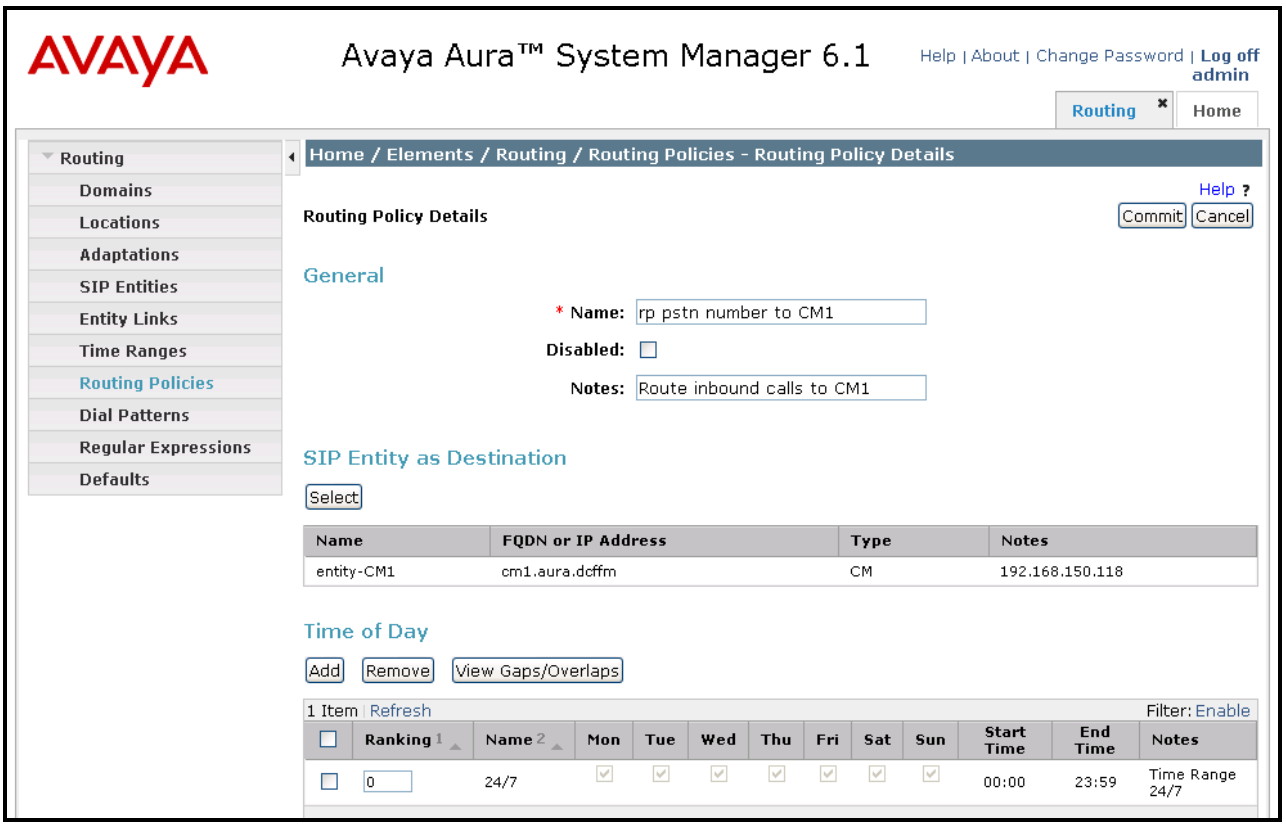
5.7. Routing Policies

To view or change routing policies, select **Routing → Routing Policies**. Click on the checkbox corresponding to the name of a policy and **Edit** to edit an existing policy, or **New** to add a policy. Click the **Commit** button after changes are completed.

The following screen shows the **Routing Policy Details** for the policy named "rp pstn number to CM1" associated with incoming PSTN calls from COLT to Communication Manager, using the Avaya S8800 PE. Observe the **SIP Entity as Destination** is the entity named "entity-CM1".



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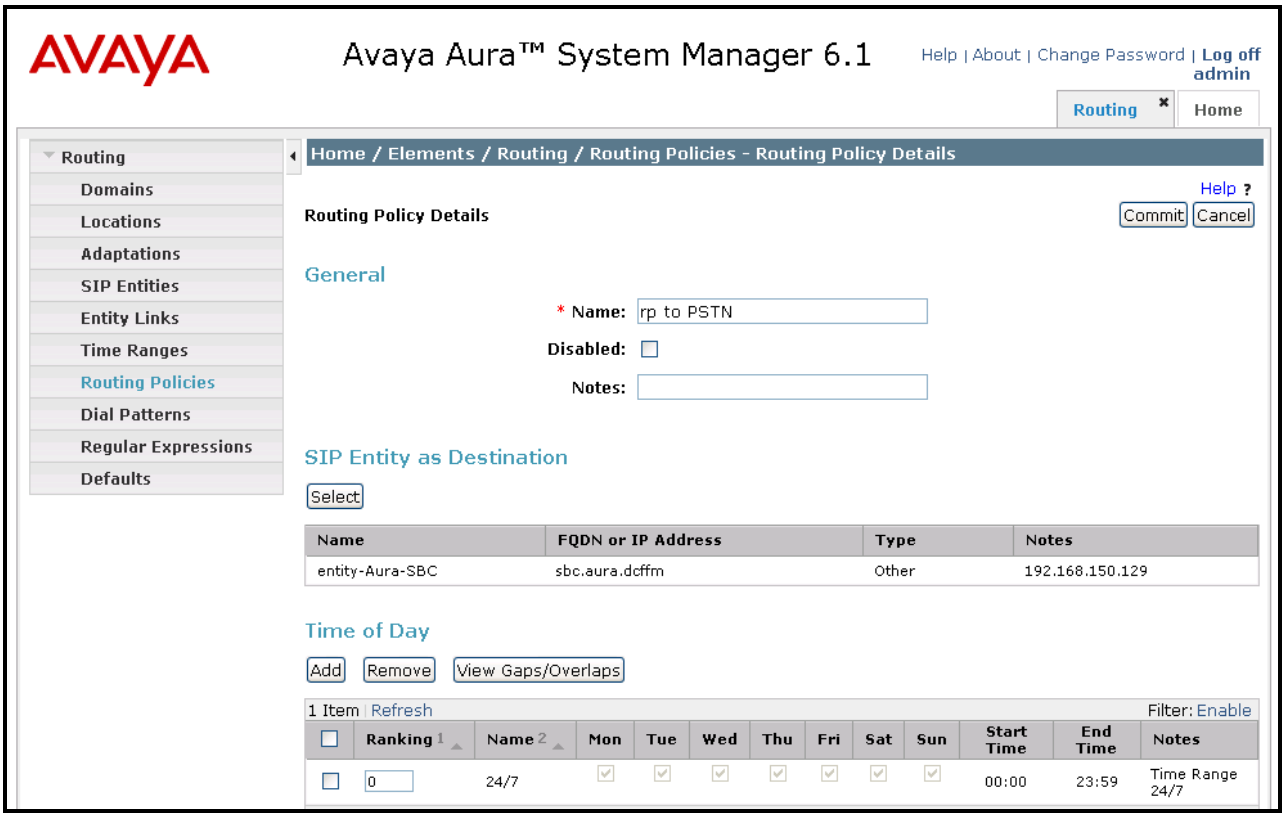
The following screen shows the **Routing Policy Details** for the policy named "rp to PSTN" associated with outgoing calls from Communication Manager to the PSTN via COLT through the Acme Packet 3820 SBC. Observe the **SIP Entity as Destination** is the entity named "entity-Aura-SBC".



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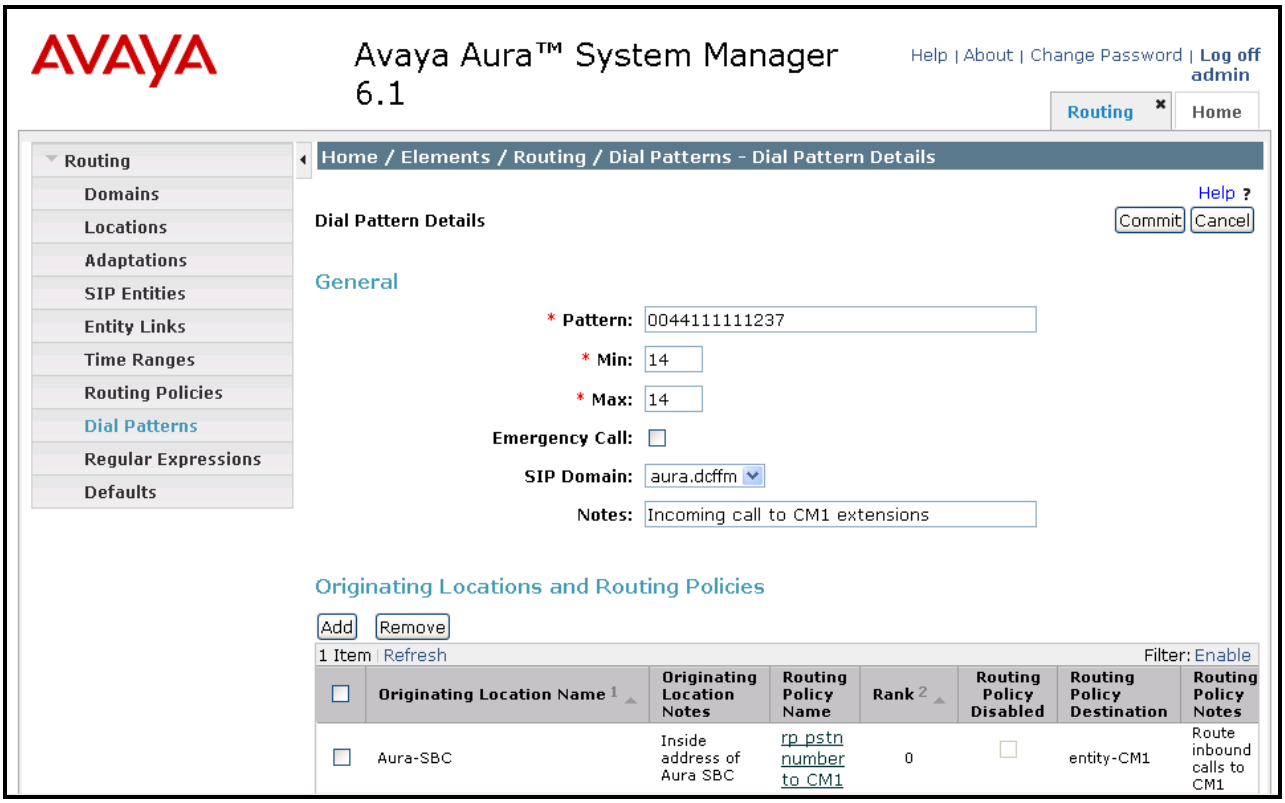
5.8. Dial Patterns

To view or change dial patterns, select **Routing → Dial Patterns**. Click on the checkbox corresponding to the name of a pattern and **Edit** to edit an existing pattern, or **New** to add a pattern. Click the **Commit** button after changes are completed.

The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the COLT VoIP Access SIP Service, such as 00441111112371, COLT delivers the number to the enterprise, and the Acme Packet 3820 SBC sends the call to Session Manager. The pattern below matches on 0044111111237 specifically. Dial patterns can alternatively match on ranges of numbers (e.g., a DID block). Under **Originating Locations and Routing Policies**, the routing policy named "rp pstn number to CM" is selected, which sends the call to Communication Manager as described previously.



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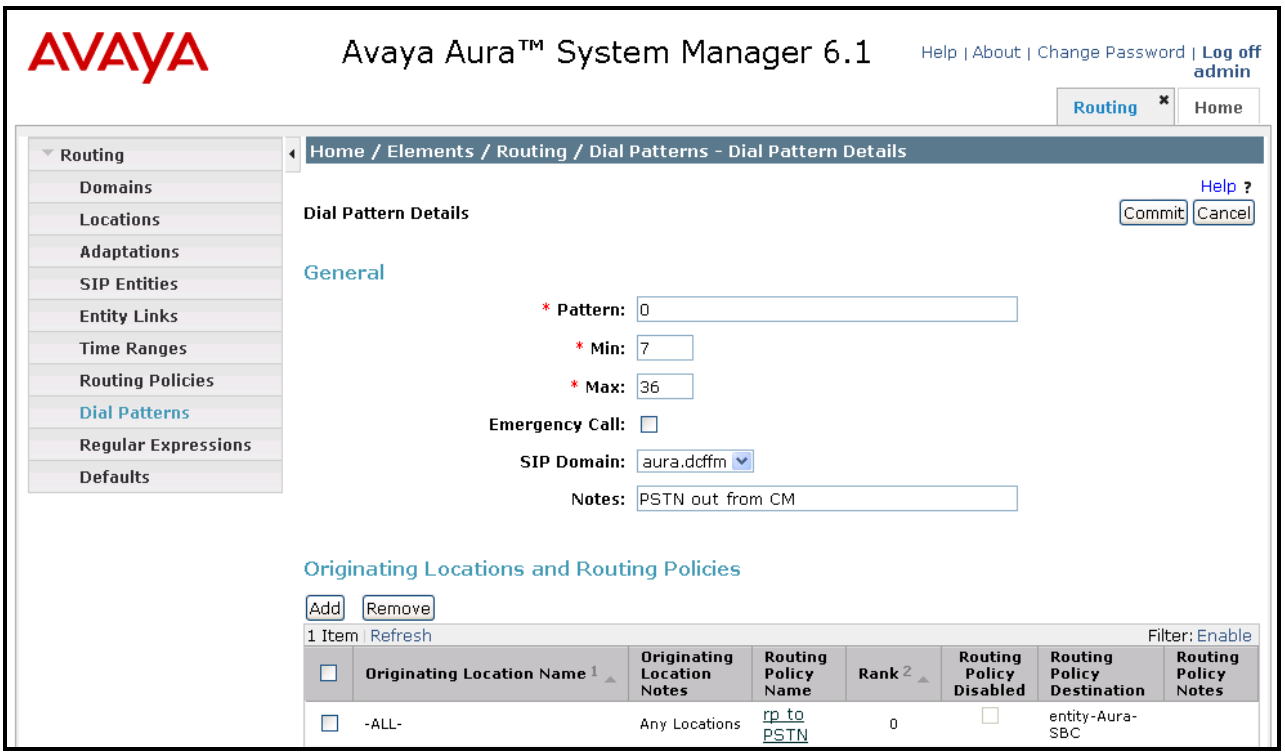
The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Manager user dials a PSTN number such as 001-908-848-5704, Communication Manager sends the call to Session Manager, via the S8800 PE. Session Manager will match the dial pattern shown below and send the call to the Acme Packet 3820 SBC via the **Routing Policy Name** "rp to PSTN".



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6. Configure Acme Packet 3820 Session Border Controller

This section illustrates an example configuration of the Acme Packet 3820 SBC. The configuration information for this component is via an xml file which can be edited by a text editor. Only those portions of the file which must be changed for the tested environment are shown. Comments have been added to describe each of the entries.

```
<?xml version='1.0' standalone='yes'?>
<config lastObjectId='48'>
        <!-- default gateway of the "outside" network -->
        <systemConfig hostname='AcmePacketSBC'
                defaultGW='192.168.31.254'
        </systemConfig>
        <!-- SBC address, net mask, and default gateway within "outside" network -->
        <networkInterface name='M00'
                ipAddress='192.168.31.20'
                netmask='255.255.255.0'
                gateway='192.168.31.1'
        </networkInterface>
        <!-- SBC address, net mask, and default gateway within "inside" network -->
        <networkInterface name='M10'
                ipAddress='192.168.150.129'
                netmask='255.255.255.0'
                gateway='192.168.150.254'
        </networkInterface>
        <!-- Realm description of "outside" network -->
        <realmConfig id=Colt-trunk'
                description='Colt SIP Trunk'
                outManipulationId='NAT_IP'
                symmetricLatching='disabled'
        </realmConfig>
        <!-- Realm description of "inside" network -->
        <realmConfig id='core'
                description='Devconnect Core'
                outManipulationId='NAT_IP'
                <networkInterfaceId name='M10'
                        subPortId='0'/>
        </realmConfig>
        <!-- Port addresses for "outside" network -->
        <steeringPool ipAddress='192.168.31.20'
                startPort='10000'
                endPort='19999'
                realmID='Colt-trunk'
        </steeringPool>
        <!-- Port addresses for "inside" network -->
        <steeringPool ipAddress='192.168.150.129'
                startPort='10000'
                endPort='19999'
                realmID='core'
        </steeringPool>
        <!-- Interface state for "outside" network -->
        <sipInterface state='enabled'
                RealmID=Colt-trunk'
                <sipPort address='192.168.31.20'
                        port='5060'
                        transProtocol='UDP'
                        imsAkaProfile=''/>
        </sipInterface>
        <!-- Interface state for "inside" network -->
        <sipInterface state='enabled'
                RealmID='core'
                <sipPort address='192.168.150.129'
                        port='5060'
                        transProtocol='TCP'
                        imsAkaProfile=''/>
```


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```
        </sipInterface>
        <!-- Hostname and IP address for Session Manager "asset" interface -->
        <sessionAgent hostname='192.168.150.115'
                ipAddress='192.168.150.115'
        </sessionAgent>
        <!-- List of Session Managers served by this SBC -->
        <sessionAgentGroup name='avaya-SM'
                description='Avaya Session Manager'
                strategy='RoundRobin'
                <sipDest name='192.168.150.115'/>
        </sessionAgentGroup>
        <!-- "Outside" address this SBC  -->
        <sessionAgent hostname='192.168.31.20'
                ipAddress='192.168.31.20'
        </sessionAgent>
        <!-- "Outside" address of Sonus GSX used as COLT SBC -->
        <sessionAgent hostname='10.49.2.2'
                ipAddress='10.49.2.2'
                description='Colt '
        </sessionAgent>
</config>
```

7. COLT VoIP Access SIP Service Offer Configuration

Information regarding COLT VoIP Access SIP Service offer can be found at http://www.colt.net or by contacting a COLT VoIP Access SIP Service sales representative.

The sample configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. COLT VoIP Access SIP Service provided all of the necessary service provisioning.

7.1. Fully Qualified Domain Name (FQDN)s

The COLT network uses IP addresses instead of domain names.

8. General Test Approach and Test Results

The test approach was manual testing of inbound and outbound calls using the COLT VoIP Access SIP Service on a production COLT VoIP access circuit, as shown in **Figure 1**. The test requirements specified in section 1.1 were tested successfully. Examples of the verified call scenarios are detailed in Section 9.

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9. Verification Steps

This section provides example verifications of the Avaya configuration with COLT VoIP Access SIP Service. Verification scenarios for the configuration described in these Application Notes included the following:

- Inbound and outbound voice calls between telephones controlled by Avaya Aura® Communication Manager and the PSTN can be made using G.711MU or G.729A codecs.
- Inbound and outbound Fax calls between Avaya Aura® Communication Manager and PSTN can be made in both directions using T.38 .
- Direct IP-to-IP Media (also known as "Shuffling") when applicable.
- DTMF Tone Support as specified by RCF2833
    - Outbound call to PSTN application requiring post-answer DTMF (e.g., an IVR or voice mail system)
    - Inbound call from PSTN to Avaya CPE application requiring post-answer DTMF (e.g., Avaya Modular Messaging, Avaya vector digit collection steps)
- Additional PSTN numbering plans (e.g. International, operator assist, 411)
- Hold / Retrieve with music on hold
- Call transfer
- Conference calls
- Voicemail Coverage and Retrieval
- SIP Diversion Header for call re-direction
    - Call Forwarding
    - EC500
- Long hold time calls

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9.1. Avaya Aura® Communication Manager Verifications

This section illustrates verifications from Avaya Aura® Communication Manager.

9.1.1 Example Incoming Call from PSTN via COLT SIP Trunk

Incoming PSTN calls arrive from COLT at the Acme Packet 3820 SBC, which sends the call to Session Manager. Session Manager sends the call to Communication Manager via the entity link corresponding to the Avaya S8800 PE. On Communication Manager, the incoming call arrives via signaling group 1 and trunk group 1.

The following Communication Manager "list trace" trace output shows a call incoming on trunk group 1. The PSTN telephone dialed 000441111112371. The incoming call handling table for trunk group 1 converted the number to 2371. Extension 2371 is a SIP Telephone with IP Address 192.168.150.86. Initially, the G450 Media Gateway (192.168.150.124) is used, but as can be seen in the final trace output, once the call is answered, the final RTP media path is "ip-direct" from the IP Telephone (192.168.150.86) to the "inside" of the Acme Packet 3820 SBC (192.168.150.129).

```
list trace tac *801                                                    Page   1
                                LIST TRACE
time            data

15:38:42 TRACE STARTED 03/15/2011 CM Release String cold-00.1.510.1-18621
15:38:54 SIP<INVITE sip:00441111112371@aura.dcffm:5060 SIP/2.0
15:38:54     active trunk-group 1 member 1  cid 0x87f
15:38:54     dial 2371
15:38:54     term station    2371 cid 0x87f
15:38:55 SIP>SIP/2.0 180 Ringing
15:38:55     G711A ss:off ps:20
             rgn:1 [192.168.150.129]:22946
             rgn:1 [192.168.150.124]:2058
15:38:55     xoip options: fax:T38 modem:off tty:off  uid:0x50001
             xoip ip: [192.168.150.124]:2058
15:38:55 SIP<PRACK sip:00441111112371@aura.dcffm:5060 SIP/2.0
15:38:55 SIP>SIP/2.0 200 OK
15:38:58 SIP>SIP/2.0 200 OK
15:38:58     active station    2371 cid 0x87f
15:38:58 SIP<ACK sip:00441111112371@aura.dcffm:5060 SIP/2.0
15:38:58 SIP>INVITE sip:496975050@10.49.2.1:5060;transport=tcp;m
15:38:58 SIP>addr=192.168.150.129 SIP/2.0
15:38:58 SIP<SIP/2.0 100 Trying
15:38:58 SIP<SIP/2.0 200 OK
15:38:58     G711A ss:off ps:20
             rgn:1 [192.168.150.86]:5004
             rgn:1 [192.168.150.129]:22946
15:38:58 SIP>ACK sip:496975050@10.49.2.1:5060;transport=tcp;madd
15:38:58 SIP>r=192.168.150.129 SIP/2.0
15:38:58     G711A ss:off ps:20
             rgn:1 [192.168.150.129]:22946
             rgn:1 [192.168.150.86]:5004
```


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The following screen shows **Page 2** of the output of the "status trunk" command pertaining to this same call. Note the media is "ip-direct" from the IP Telephone (192.168.150.86) to the inside IP address of the Acme 3820 SBC (192.168.150.129) using G.711A.

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

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  192.168.150.118                       : 5060
    Far-end:  192.168.150.115                       : 5060
 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.711A
   Audio      IP Address                              Port
   Near-end:  192.168.150.86                        : 5004
    Far-end:  192.168.150.129                       : 22946

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

The following screen shows **Page 3** of the output of the "status trunk" command pertaining to this same call. Here it can be observed that G.711A is used.

```
status trunk 1/1                                                Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00001
T00001:TX:192.168.150.129:22946/g711a/20ms
T00007:RX:192.168.150.86:5004/g711a/20ms
```


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9.1.2 Example Outgoing Call to PSTN via COLT SIP Trunk

The following trace shows an outbound ARS call from IP Telephone x2371 to the PSTN number 00496975056645. The call is routed to route pattern 1 and trunk group 1. The call initially uses the gateway (192.168.150.124), but after the call is answered, the call is "shuffled" to become an "ip-direct" connection between the IP Telephone (192.168.150.86) and the "inside" of the Acme 3820 SBC (192.168.150.129). In this case, the mapping from the caller's extension to the full DID number was performed in Session Manager.

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```
list trace tac *801                                                    Page   1

                                LIST TRACE

time            data

10:06:44 TRACE STARTED 03/16/2011 CM Release String cold-00.1.510.1-18621
10:06:55 SIP<INVITE sip:2371@aura.dcffm;avaya-cm-fnu=off-hook SI
10:06:55 SIP<P/2.0
10:06:55 SIP>SIP/2.0 183 Session Progress
10:06:55 SIP>SIP/2.0 484 Address Incomplete
10:06:55 SIP<INVITE sip:00496975056645@aura.dcffm SIP/2.0
10:06:55 SIP>SIP/2.0 100 Trying
10:06:55     dial 00 route:ARS
10:06:55     term trunk-group 1    cid 0x8af
10:06:55     dial 00496975056645# route:ARS
10:06:55     route-pattern  1 preference 1  cid 0x8af
10:06:55     seize trunk-group 1 member 4  cid 0x8af
10:06:55     Calling Number & Name NO-CPNumber NO-CPName
10:06:55     Calling Number & Name NO-CPNumber NO-CPName
10:06:55 SIP<ACK sip:2371@aura.dcffm;avaya-cm-fnu=off-hook;route
10:06:55 SIP<info=0-0-0;nrindex=0 SIP/2.0
10:06:55     Proceed trunk-group 1 member 4  cid 0x8af
10:06:59 SIP>SIP/2.0 183 Session Progress
10:06:59     G711A ss:off ps:20
             rgn:1 [192.168.150.129]:20274
             rgn:1 [192.168.150.124]:2062
10:06:59     xoip options: fax:T38 modem:off tty:off  uid:0x50004
             xoip ip: [192.168.150.124]:2062
10:06:59     G711A ss:off ps:20
             rgn:1 [192.168.150.86]:5004
             rgn:1 [192.168.150.124]:2058
10:06:59     xoip options: fax:T38 modem:off tty:off  uid:0x50001
             xoip ip: [192.168.150.124]:2058
10:06:59 SIP<PRACK sip:00496975056645@192.168.150.118;transport=
10:06:59 SIP<tcp SIP/2.0
10:06:59 SIP>SIP/2.0 200 OK
10:07:01 SIP>SIP/2.0 200 OK
10:07:01     active trunk-group 1 member 4  cid 0x8af
10:07:01 SIP<ACK sip:00496975056645@192.168.150.118;transport=tc
10:07:01 SIP<p SIP/2.0
10:07:01 SIP>INVITE sip:2371@192.168.150.86;transport=tcp SIP/2.
10:07:01 SIP>0
10:07:01 SIP<SIP/2.0 100 Trying
10:07:01 SIP<SIP/2.0 200 OK
10:07:01 SIP>ACK sip:2371@192.168.150.86;transport=tcp SIP/2.0
10:07:01     G711A ss:off ps:20
             rgn:1 [192.168.150.86]:5004
             rgn:1 [192.168.150.129]:20274
10:07:01     G711A ss:off ps:20
             rgn:1 [192.168.150.129]:20274
             rgn:1 [192.168.150.86]:5004
```


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The following screen shows **Page 2** of the output of the "status trunk" command pertaining to this same call. Note the media is "ip-direct" from the IP Telephone (192.168.150.86) to the inside IP Address of the Acme Packet 3820 SBC (192.168.150.129) using G.711.

```
status trunk 1/4                                                Page   2 of   3
                              CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling   IP Address                              Port
   Near-end:  192.168.150.118                       : 5060
    Far-end:  192.168.150.115                       : 5060
 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.711A
   Audio      IP Address                              Port
   Near-end:  192.168.150.86                        : 5004
    Far-end:  192.168.150.129                       : 20274
```

The following screen shows **Page 3** of the output of the "status trunk" command pertaining to this same call. Here it can be observed that G.711A is used.

```
status trunk 1/4                                                Page   3 of   3
                       SRC PORT TO DEST PORT TALKPATH
src port: T00004
T00004:TX:192.168.150.129:20274/g711a/20ms
T00001:RX:192.168.150.86:5004/g711a/20ms
```


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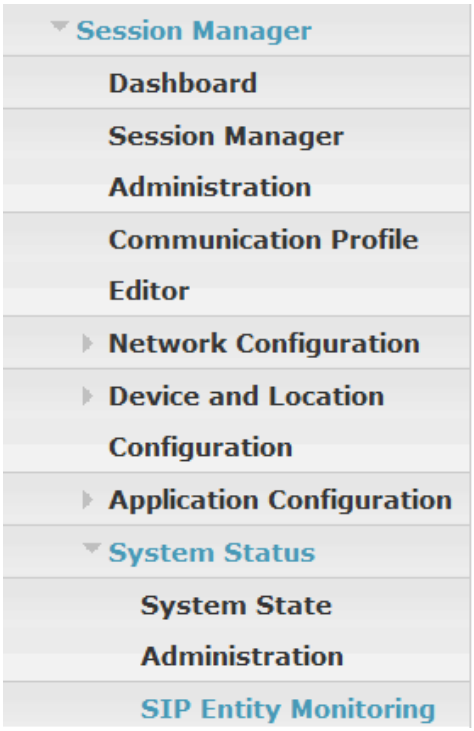
9.2. Avaya Aura® System Manager and Session Manager Verification

This section contains verification steps that may be performed using Avaya Aura® System Manager for Avaya Aura® Session Manager.

9.2.1 Verify SIP Entity Link Status

Log in to System Manager. Expand **Elements → Session Manager → System Status → SIP Entity Monitoring**, as shown below.



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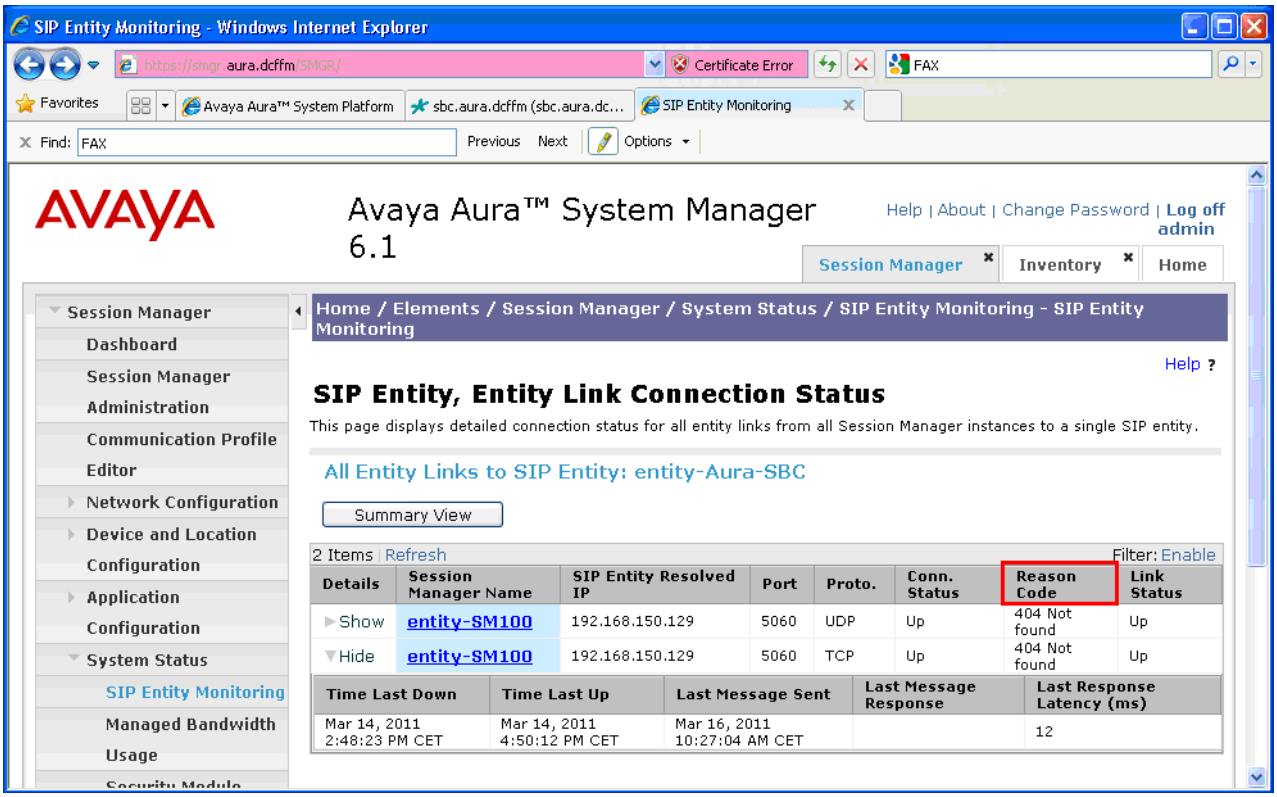
From the list of monitored entities, select an entity of interest, such as “entity-Aura-SBC”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. The **Reason Code** column indicates that the Aura SBC has responded to SIP OPTIONS from Session Manager with a SIP 404 message, which is sufficient for SIP Link Monitoring to consider the link up.



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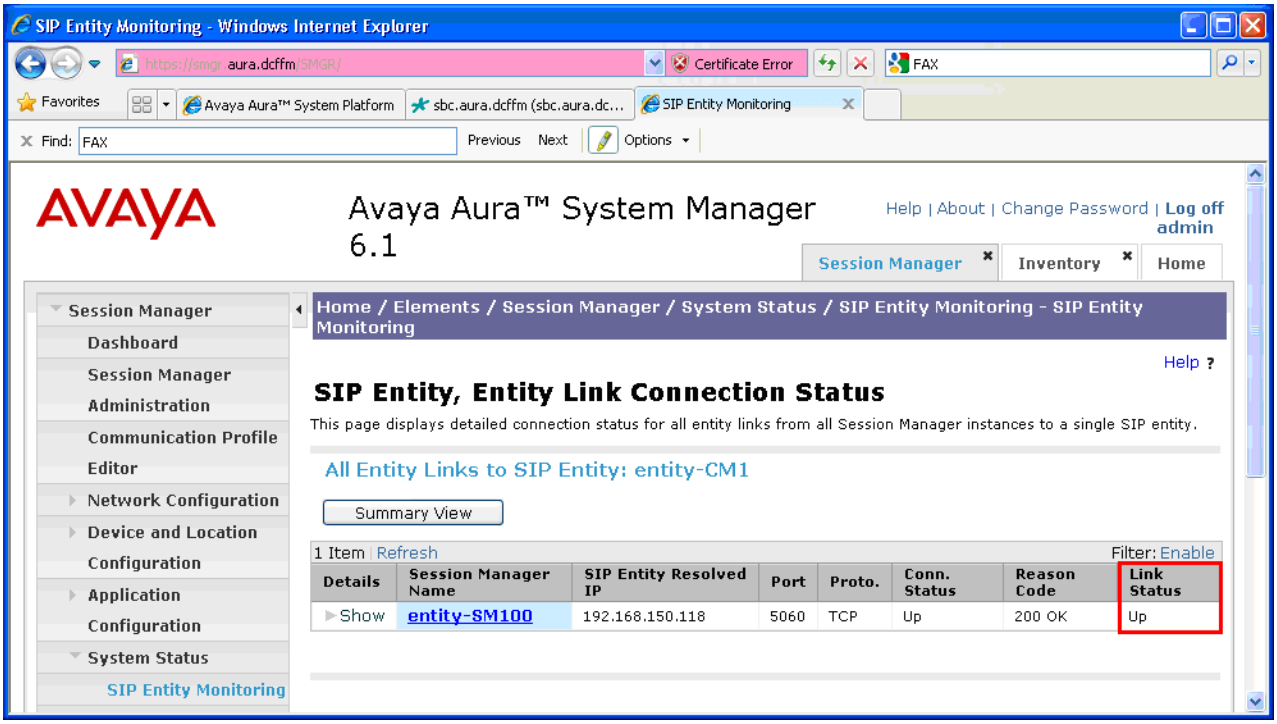
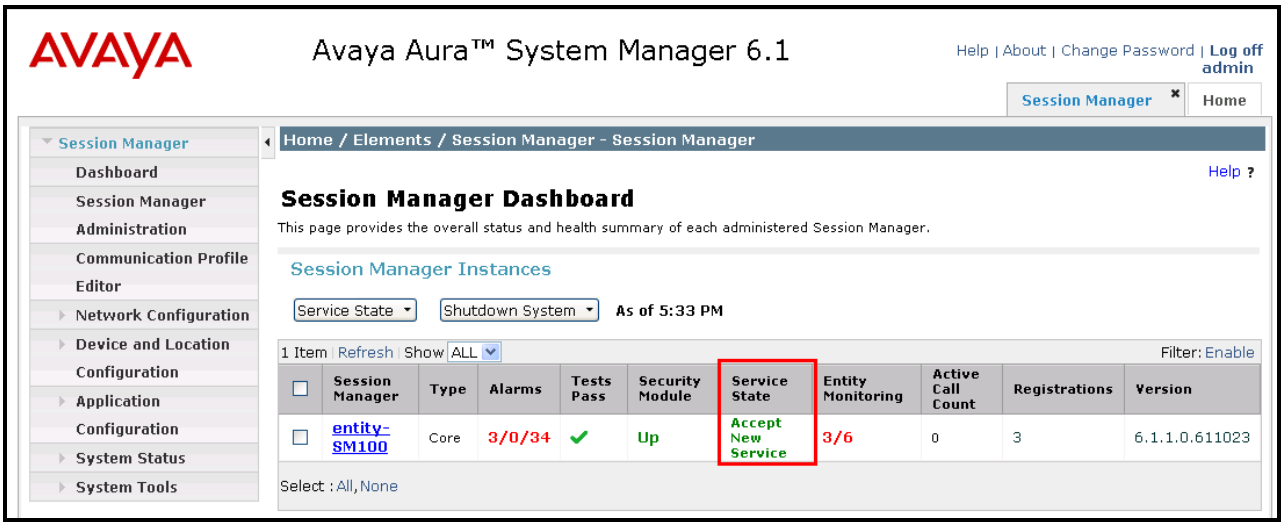
Return to the list of monitored entities, and select another entity of interest, such as "entity-CM1". Under normal operating conditions, the **Link Status** should be "Up" as shown in the example screen below. In this case, "Show" under Details was selected to view additional information.



9.2.2 Verify System State

Expand **Elements → Session Manager → Dashboard.** Verify that the Service State is "Accept New Service."



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10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0, and the Acme Packet 3820 SBC can be configured to interoperate successfully with COLT VoIP Access SIP Service. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager customers access to the PSTN using a COLT VoIP Access SIP Service IP Trunk public SIP trunk service connection.

11. Additional References

This section references documentation relevant to these Applications. Avaya product documentation, including the following, is available at http://support.avaya.com

[1] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010 available at http://support.avaya.com/css/P8/documents/100089133

[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509, Issue 6.0 June 2010 available at http://support.avaya.com/css/P8/documents/100089333

[3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100082630

[4] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473 Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089152

[5] *Maintaining and Troubleshooting Avaya Aura® Session Manager,* Doc ID 03-603325, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089154

[6] *Administering Avaya Aura® System Manager*, Document Number 03-603324, Release 5.2, November 2009 available at http://support.avaya.com/css/P8/documents/100089681

Avaya Application Notes, including the following, are also available at http://support.avaya.com

Application Notes Reference [PE] documents a configuration with testing results using Processor Ethernet on a main Communication Manager and an ESS for survivable SIP Trunking. The verifications in this document illustrate additional survivability considerations.

[PE] Sample Configuration Illustrating Avaya Aura® Communication Manager SIP Trunking Using Processor Ethernet and Acme Packet Net-Net 4500 Session Director – Issue 1.0
https://devconnect.avaya.com/public/flink.do?f=/public/download/interop/CM-PE-NN4500.pdf

Application Notes Reference [CLAN] documents a similar configuration to [PE] using survivable SIP Trunks signaled from C-LAN interfaces rather than processor Ethernet.

[CLAN] Sample Configuration Illustrating Avaya Aura® Communication Manager SIP Trunk Survivability with Enterprise Survivable Server and Acme Packet Net-Net 4500 Session Director, Issue 1.0
https://devconnect.avaya.com/public/flink.do?f=/public/download/interop/CM-ESS-NN4500.pdf

Application Notes Reference [LAR] contains additional information on Communication Manager Look-Ahead Routing.

[LAR] Sample Configuration for SIP Private Networking and SIP Look-Ahead Routing Using Avaya Communication Manager, Issue 1.0
http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/sip-pvt-lar.pdf

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