Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager Evolution Server R5.2.1, Avaya Aura® Session Manager R6.1, and Acme Packet 3800 Session Border Controller R6.2 with Star Telecom – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Avaya Aura® Session Manager Release 6.1, Avaya Aura® Communication Manager Release 5.2.1, and the Acme Packet 3800 Session Border Controller Release 6.2 with the Star Telecom system.

The Star Telecom 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 Lab, utilizing a Star Telecom circuit connection to the production Star Telecom 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 5.2.1, and the Acme Packet 3800 Session Border Controller (SBC) Release 6.2 with the Star Telecom system. The Star Telecom Service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

2. General Test Approach and Test Results

Communication Manager connects to the ACME 3800 Session Border Controller via Session Manager using a SIP connection. The ACME 3800 connects to the Star Telecom system using SIP signaling. Various call types were made from Communication Manager to and from the Star Telecom system to verify the interoperability.

2.1. Interoperability Compliance Testing

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

- General call processing between Communication Manager and Star Telecom systems including:
    - Codec/ptime (G.711 u-law / 20ms, G.729 / 20ms)
    - Hold/Retrieve on both ends
    - CLID, CPND displayed
    - Ring-back tone
    - Speech path
    - Dialing plan support
    - Advanced features (Call on Mute, Call Park, Call Waiting)
    - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- UUI (User to User Information)- Call Redirect Capability
- DTMF in both directions
- SIP Transport UDP, TCP
- Early Media Transmission

The following assumptions were made for this lab test configuration:

- Avaya Aura® Communication Manager R5.2.1 software and implementation of latest service packs.
- Star Telecom provides support to setup, configure and troubleshoot on carrier switch during testing execution.

During testing, the following activities were made to each test scenario:

- Calls were checked for the correct call progress tones and cadences.
- During the ringing state the ring back tone and destination ringing were checked.
- Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
- Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
- The display(s) of the sets/clients involved were checked for consistent and expected CLID and redirection information both prior to answer and after call establishment.
- The speech path was observed for timely and quality End to End tone audio path generation and application responses.
- The call server maintenance terminal window was open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.
- Speech path was checked before and after calls were put on/off hold from each end.
- Calls were checked to ensure that all resources such as Virtual trunks, Sets and VGWs are released when a call scenario ends.

2.2. Known Limitations

No limitations were found during testing.

2.3. Support

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

For technical support on Star Telecom system, please contact Star Telecom technical support at:

- Toll Free: 1-855-STAR-TEL (1-855-782-7835)
- http://www.startelecom.ca

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between Communication Manager and Star Telecom systems. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked out and replaced with fictitious IP addresses throughout the document.

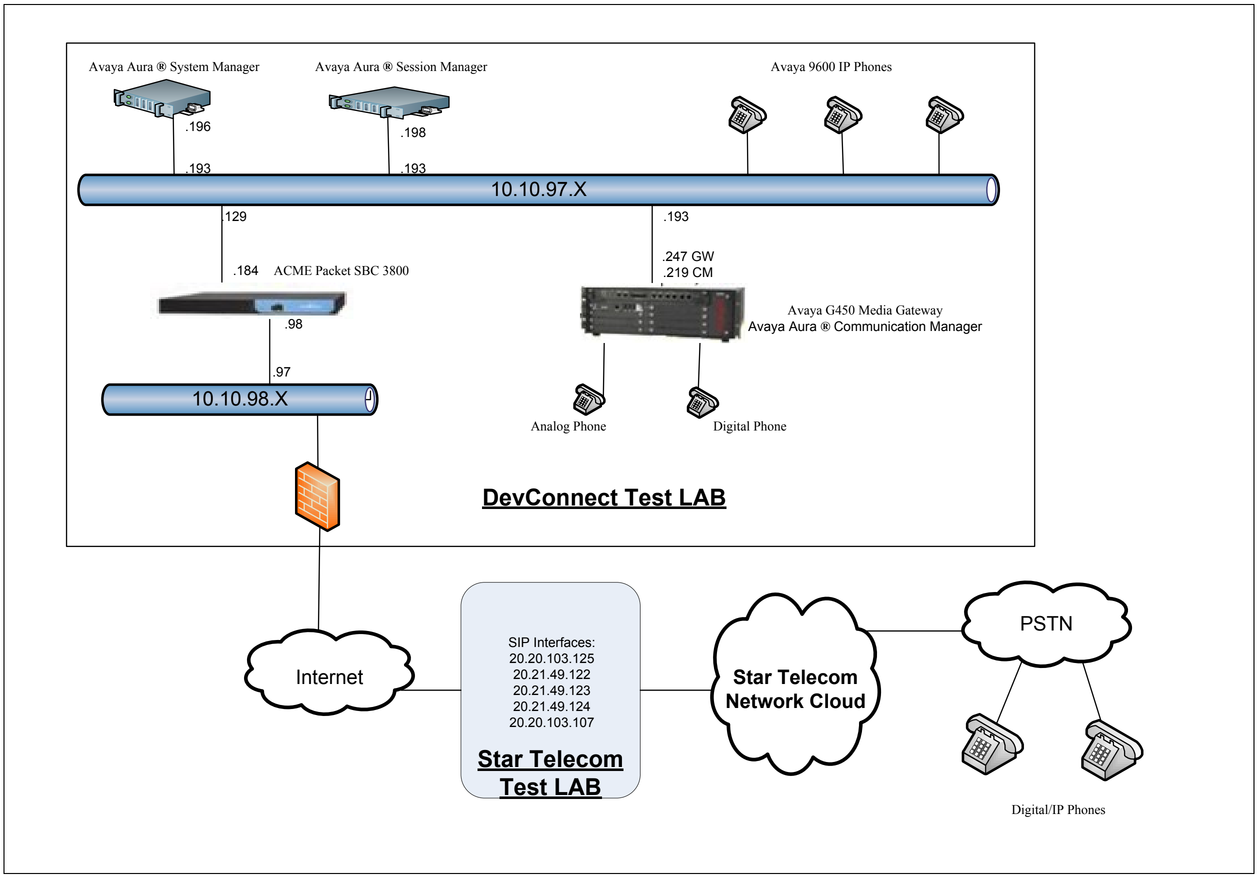


Figure 1- Network diagram for Avaya Aura® Communication Manager and Star Telecom System

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8300 Server	Avaya Aura® Communication Manager R015x.02.1.016.4 Patch: 02.1.016.4-18942
Avaya G450 Media Gateway MM711 Analog MM712 Digital	 HW31 FW91 HW05 FW09
Avaya S8800 Server	Avaya Aura® Session Manager R6.1.1.0.611023
Avaya S8800 Server	Avaya Aura ®System Manager R6.1.4.0 + SP0.r873
Avaya 9611 Phone (H323)	3.11
Avaya 9404 Digital Phone	N/A
Analog Phone	N/A
ACME Net-Net 3800	Firmware SCX6.2.0 MR-4 Patch 3 (Build 754)
Star Telecom Free Switch	Free Switch R3.2

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP signaling. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signaling associated with Star Telecom SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from Star Telecom via Session Border Controller and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to Session Manager. Session Manager directs the outbound SIP messages to Star Telecom network via Session Border Controller. Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

5.1. Verify Licensed Features

The Communication Manager license file controls the maximum values for these attributes. Contact an authorized Avaya sales representative for assistance if a required feature needs to be enabled or there is insufficient capacity.

Use the **display system-parameters customer-options** command and on **Page 2,** verify that the **Maximum Administered SIP Trunks** are sufficient for the combination of trunks to the Star Telecom offer and any other SIP applications. Each call from a non-SIP endpoint to the Star Telecom 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: 450    0
           Maximum Concurrently Registered IP Stations: 450    2
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                        Maximum Video Capable Stations: 0      0
                   Maximum Video Capable IP Softphones: 0      0
                       Maximum Administered SIP Trunks: 450    75
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 0      0
                     Maximum Media Gateway VAL Sources: 0      0
           Maximum TN2602 Boards with 80 VoIP Channels: 0      0
          Maximum TN2602 Boards with 320 VoIP Channels: 0      0
   Maximum Number of Expanded Meet-me Conference Ports: 0      0
```

On **Page 3**, verify that **ARS** is set to **y**.

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

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

On **Page 5**, verify that **Private Networking** and **Processor Ethernet** are set to **y**.

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

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

5.2. Configure Dial Plan

In the sample configuration, the Avaya CPE environment uses **4** digits to dial the local extensions (**ext**), such as **20**xx. For outbound calls via SIP trunk to Star Telecom, the feature access code (**fac**) **9** is used to access the Automatic Route Selection (ARS) table. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used.

Use the **change dialplan analysis** command to make changes the following values:

- Enter the **Dialed String 20** with **Total Length 4**
- Enter the **Dialed String 9** with **Total Length 1**

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call      Dialed   Total  Call
       String   Length Type    String   Length Type      String   Length Type
          0            3     fac
          1            3     fac
          20           4     ext
          3            5     ext
          4            10    ext
          6            10    ext
          8            1     fac
          9            1     fac
          *            2     fac
          *            3     fac
          *            4     dac
          #            2     fac
          #            3     fac
```

5.3. Configure IP Node Names

The 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. The node name for Session Manager is **DevASM** with IP Address **10.10.97.198**. The node name and IP Address for the Processor Ethernet (procr) are **procr** and **10.10.97.219.** The **procr** is the interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
change node-names ip
                                  IP NODE NAMES
    Name              IP Address
DevASM                10.10.97.198
default               0.0.0.0
procr                 10.10.97.219
```

5.4. Configure IP Interface for procr

Use the **change ip-interface procr** command to change the Processor Ethernet (procr) parameters. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the procr for SIP Trunk signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones. Ensure **Enable Interface** is **y** and **Network Region** is **1**

```
change ip-interface procr
                                  IP INTERFACES

                  Type: PROCR
                                                     Target socket load: 1700

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

                                 IPV4 PARAMETERS
             Node Name: procr
           Subnet Mask: /26
```

5.5. Configure IP 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 **Network Region 1** and **Location 1**. The **MM711** Analog interface serves as an interface to Analog phone. The **MM712** DCP interface serves as an interface to Digital phone.

```
display media-gateway 1
                              MEDIA GATEWAY
         Number: 1                                Registered?  y
           Type: g450              FW Version/HW Vintage: 28 .22 .0  /1
           Name: Media Gateway 1          MGP IP Address: 10.10 .97 .247
      Serial No: 08IS38199691      Controller IP Address: 10.10 .97 .219
   Encrypt Link? y                           MAC Address: 00:1b:4f:03:51:08
 Network Region: 1    Location: 1
                                               Site Data:
  Recovery Rule: none

Slot   Module Type            Name                    DSP Type  FW/HW version
 V1:   S8300                  ICC MM                  MP80      15   2
 V2:   MM711                  ANA MM
 V3:   MM712                  DCP MM
 V4:
 V5:
 V6:
 V7:
 V8:                                                Max Survivable IP Ext: 8
 V9: gateway-announcements   ANN VMM
```

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 7.1**. In this configuration, the domain name is **bvwdev7.com**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is set to **yes** to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources.The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** was used.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bvwdev7.com
    Name: procr
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.6. Configure IP Codec Set

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.

Use the **change ip-codec-set** command for the codec set specified in the **IP Network Region 1** form above. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test, the codec's supported by Star Telecom were configured, namely **G.711A**, **G.711MU** and **G.729.**

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

                          IP Codec Set

   Codec Set: 1

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

5.7. Configure SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups that will be used for inbound and outbound PSTN calls to Star Telecom Trunk Service.

Use the **add signaling-group x (where x is the signaling-group number)** command to set the following values:

- **Group Type** is set to **sip**
- **Transport Method** is set to **tcp**
- **IMS Enabled** is set to **n**
- **Near-end Node Name** is set to **procr.** This value is taken from the IP Node Name form shown in **Section 5.3**
- **Far-end Node Name** is set to **DevASM** (Node name of the Session Manager entered in **Section 5.3**)
- **Near-end Listen Port** is set to **5060**
- **Far-end Listen Port** is set to **5060**
- **Far-end Network Region** is set to 1 (The IP Network Region is configured in **Section 5.5**)
- **Far-end domain** is set to **bvwdev7.com** (domain name as added in **Section 5.5**) in **signaling group 1** for outbound calls and set to **blank** in **signaling group 2** for inbound calls
- **DTMF over IP** is set to **rtp-payload.** This value enables Communication Manager to send DTMF transmissions using RFC 2833
- **Enable Layer 3 Test** is set to **y** to allow Communication Manager to maintain the signaling group using the SIP OPTION method

```
add signaling-group 1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: procr                    Far-end Node Name: DevASM
   Near-end Listen Port: 5060                          Far-end Listen Port: 5060
                                                    Far-end Network Region: 1
Far-end Domain: bvwdev7.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                           Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

```
add signaling-group 2
                                 SIGNALING GROUP

 Group Number: 2              Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: procr                  Far-end Node Name: DevASM
   Near-end Listen Port: 5060                    Far-end Listen Port: 5060
                                                 Far-end Network Region: 1
Far-end Domain:

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

5.8. Configure SIP Trunk Groups

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

Use the **change trunk group x (where x is the trunk group number)** command to set the following values on **Page 1**:

- Choose a descriptive **Group Name**
- Specify a trunk access code (**TAC**) consistent with the dial plan, i.e. ***010, *011**
- The **Direction** is set to **outgoing** to allow outgoing calls and set to **incoming** to allow incoming calls.
- The **Service Type** field should be set to **public-ntwrk** for the trunks that will handle calls with Star Telecom
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**
- Specify the **Number of Members** supported by this SIP trunk group

```
change trunk-group 10                                          Page   1 of  21
                                 TRUNK GROUP

Group Number: 10                  Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: *010
   Direction: outgoing       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 50
```

Use the **change trunk group 10** command and on **Page 3** to set **Numbering Format** field as **private.**

```
change trunk-group 10                                           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
```

Use the **change trunk group 11** command to set the values of trunk group which will be used for PSTN calls to Star Telecom. Trunk group 11 is associated with **Signaling Group 2.**

```
change trunk-group 11                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 11                   Group Type: sip           CDR Reports: y
  Group Name: INSIDE CALL                COR: 1        TN: 1        TAC: *011
   Direction: incoming        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 2
                                                     Number of Members: 15
```

Use the **change display trunk group 11** command on **Page3** to set **Numbering Format** as **private.**

```
change trunk-group 11                                           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
```

5.9. Configure Route Pattern

Use the **change route-pattern 1** command to assign the SIP trunk groups described in **Section 5.8**. This allows route pattern 1 to destine the calls between the PSTN and the Star Telecom Service by using the SIP trunk group **10** or **11**. 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.

```
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: 10   0                                                              n   user
 2: 11   0                                                              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                                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
```

5.10. Configure Public Numbering

Use the **change public-unknown-numbering** command to define the format of numbers sent to Star Telecom in SIP headers such as the From and PAI headers. In general, the mappings of internal extensions to Star Telecom 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 **647725** to the beginning of the number, when the call uses trunk group **10**. Alternatively, Communication Manager can send the extension to Session Manager by leaving the CPN Prefix field blank and setting the CPN Len to 4 and Session Manager can adapt the number to the Star Telecom DID.

```
change public-unknown-numbering 1                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext      Ext               Trk       CPN             CPN
Len      Code           Grp(s)    Prefix           Len
                                                                          Total
Administered: 8
 4       2050            10        647725           10          Maximum Entries: 240
 4       2051            10        647725           10
 4       2052            10        647725           10
 4       2053            10        647725           10
 4       2054            10        647725           10
 4       2055            10        647725           10
 4       2057            10        647725           10
 5       2002            10                         4
```

5.11. Configure 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 Star Telecom.

Use the **change ars analysis 0** command to specify ARS configuration for a number that will be dialed in the Verification Steps (**Section 9**) of these Application Notes. If a user dials the ARS access code defined in **Section 5.2**, followed by the number beginning with **Dialed String 1** with a length**11** digits and **Call Type** as **pubu**, 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    18           1          pubu                n
           011            13   24           1          intl                n
           1              11   11           1          pubu                n
           3              5     5           3          pubu                n
           4              10   10           1          pubu                n
           6              10   10           1          pubu                n
```

5.12. Configure Incoming Call Handling Treatment

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 Star Telecom 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 11. As an example, use the **change inc-call-handling-trmt trunk-group 11** to convert incoming DID numbers **647725**xxxx to 4 digit extension xxxx by deleting **6** of the incoming digit.

```
change inc-call-handling-trmt trunk-group 11                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/            Number   Number          Del  Insert
 Feature                Len        Digits
 public-ntwrk        10          647725            6
```

5.13. Configure Avaya Aura® Communication Manager Stations

In the sample configuration, four digit station extensions were used with the format 2xxx. Use the **add station 2051** command to add an Avaya H.323 IP telephone

```
add station 2051                                                     Page   1 of   5
                                     STATION

Extension: 2051                          Lock Messages? n               BCC: 0
     Type: 9620                          Security Code: 1234             TN: 1
     Port: S00021                      Coverage Path 1: 1               COR: 1
     Name: IP_2051                     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: 2051
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n


                                            Customizable Labels? Y
```

5.14. Save Avaya Aura® Communication Manager Configuration Changes

Use the **save translation all** command to save the configuration.

6. Avaya Aura® Communication Manager Configuration for UUI-Call Redirect Capability

This section describes the additional administration steps on Communication Manager necessary for supporting interaction with the Star Telecom Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface.

Note: In the following sections, only the highlighted parameters are applicable to these Application Notes. Other parameters shown should be considered informational.

6.1. Configure System Parameters

This section reviews the additional Communication Manager licenses and features that are required for supporting the interaction with the Star Telecom Transfer Connect service. For required parameters that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the necessary licenses.

Enter the **display system-parameters customer-options** command. On **Page 4** of the system-parameters customer-options form, verify that the **ISDN/SIP Network Call Redirection?** is set to **y**.

```
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? n                        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? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                    Malicious Call Trace? n
     External Device Alarm Admin? n                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? n                Multifrequency Signaling? y
      Global Call Classification? n         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n              Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
        (NOTE: You must logoff & login to effect the permission
```

On **Page 6** of the **system-parameters customer-options** form, verify that the vectoring features outlined below are set to **y.**

```
display system-parameters customer-options                    Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                ACD? y                          Reason Codes? n
                       BCMS (Basic)? n                Service Level Maximizer? n
         BCMS/VuStats Service Level? n              Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n      Service Observing (Remote/By FAC)? n
                  Business Advocate? n              Service Observing (VDNs)? y
                    Call Work Codes? n                             Timed ACW? n
      DTMF Feedback Signals For VRU? n                    Vectoring (Basic)? y
                   Dynamic Advocate? n                Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y            Vectoring (G3V4 Enhanced)? y
                            EAS-PHD? n             Vectoring (3.0 Enhanced)? y
                   Forced ACD Calls? n      Vectoring (ANI/II-Digits Routing)? y
               Least Occupied Agent? n      Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? y                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? n       Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? n                  Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? n                 Vectoring (Variables)? y
```

6.2. Configure SIP Trunks

This section describes the steps for modifying the SIP trunk to the Acme Packet SBC to support the interaction with the Star Telecom Transfer Connect service.

Enter the **change trunk-group 11** command, where **11** is the number of the trunk group administered in **section 5.8** for inbound Star Telecom service calls. On **Page 4** of the trunk-group form, set **Network Call Redirection** to **y**.
Verify **Support Request History?** is set to **n**.
Verify **Telephone Event Payload Type** is set to **100.**

```
change trunk-group 11                                         Page   4 of  21
                              PROTOCOL VARIATIONS

                       Mark Users as Phone? n
             Prepend '+' to Calling Number? n
       Send Transferring Party Information? y
                  Network Call Redirection? y
                     Send Diversion Header? y
                   Support Request History? n
              Telephone Event Payload Type: 100
```

6.3. Configure Inbound Call Routing

This section describes the steps for routing inbound Star Telecom Transfer Connect service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality (see **Section 6.2** above). Two different inbound call routing scenarios are described in these Application Notes:

1. Pre-Answer Redirection - An inbound Star Telecom Transfer Connect service call that invokes SIP NCR (using a SIP 302 message) prior to the call being answered.
2. Post-Answer Redirection - An inbound Star Telecom Transfer Connect service call that invokes SIP NCR (using a SIP REFER message) after the call has been answered by a vector.

These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Application Enablement Services (AES) to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes.

6.3.1. Pre-Answer Redirection

This section provides an example of Pre-Answer Redirection. The following screen shots show how to route inbound Star Telecom Transfer Connect service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vector. The vector instructs Communication Manager to redirect the call to a designed number. In the example, the inbound call is routed to the **vdn 2056**, which invokes the **vector 22**.

```
change vdn 2056                                                Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2056
                                 Name*: 302
                           Destination: Vector Number        22


                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none

                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

Note: The parameters for ASAI UUI variables A and B, and other vector variables are defined using the **change variables** command.

```
change variables                                                Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                     Type    Scope Length Start Assignment       VAC
A   UuiTest1                        asaiuui L     16     1
B   UuiTest2                        asaiuui L     16     17
C
```

The **vector 22** does the following:

- Plays ringback for 3 seconds (vector step **02**).
- Assigns the data **1234567890123456** to ASAI UUI variable **B** (vector step **05**).
- Redirects the call to the number **6477252057** (vector step **08**). Note that since this vector did not answer the call, the presence of the **~r** in the **route-to number** instructs Communication Manager to send a SIP 302 message with the number 6477252057 in the user part of the Contact header URI, e.g., 6477252057@<host/domain>, to the Star Telecom Transfer Connect service (via the Acme Packet SBC).

```
change vector 22                                              Page   1 of   6
                                  CALL VECTOR

    Number: 22              Name: 302RingUUI
Multimedia? n                                                         Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    Ringing
02 wait-time    3   secs hearing ringback
03
04 #    Define UUI variable
05 set          B      = none   CATR  1234567890123456
06
07 #    Redirect
08 route-to     number ~r6477252057   with cov n if unconditionally
09 stop
10
11
12
```

6.3.2. Post-Answer Redirection

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the **vdn 2059**, which invokes the **vector 15**.

```
change vdn 2059                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2059
                                 Name*: REFER
                           Destination: Vector Number        15


                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none



                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

The **vector 15** does the following:

- Assigns the data **1234567890123456** to ASAI UUI variable **A** (vector step **02**).
- Answers the call to play an **announcement 3003** (vector step **05**). Attempts to redirect the call to the number **6477252057** (vector step **08**). Note that since this vector answered the call, the presence of the **~r** in the **route-to number** instructs Communication Manager to send a SIP REFER message with the number 6477252057 in the user part of the Refeer-header URI, e.g., 6477252057@<host/domain> to the Star Telecom Transfer Connect service (via the Acme Packet SBC).
- If the redirection fails (e.g. network denies the call), then **announcement 3004** (vector step **10**) is played to the caller.

```
change vector 15                                               Page   1 of   6
                                  CALL VECTOR

    Number: 15               Name: Refer_UUI
Multimedia? n                                                          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 #    Generate UUI
02 set          A      = none   CATR  1234567890123456
03
04 #    Play Refer announcement
05 announcement 3003
06
07 #    Refer occurs since this is post answer
08 route-to     number  ~r6477252057           with cov n if unconditionally
09 #    If Refer fails play announcement and disconnect
10 disconnect   after announcement 3004
11
12
```

6.3.3. Provision Station to display UUI

In order to display the UUI information defined in the **Sections 6.3.1** and **6.3.2** above, the Agent's station must have a UUI display button defined via the Communication Manager ***change station x*** form, where ***x*** is a station extension associated with the Agent. On **page 4** of the **change station 2057** form, add the **uui-info** feature to any available button appearance (e.g. button appearance **4**).

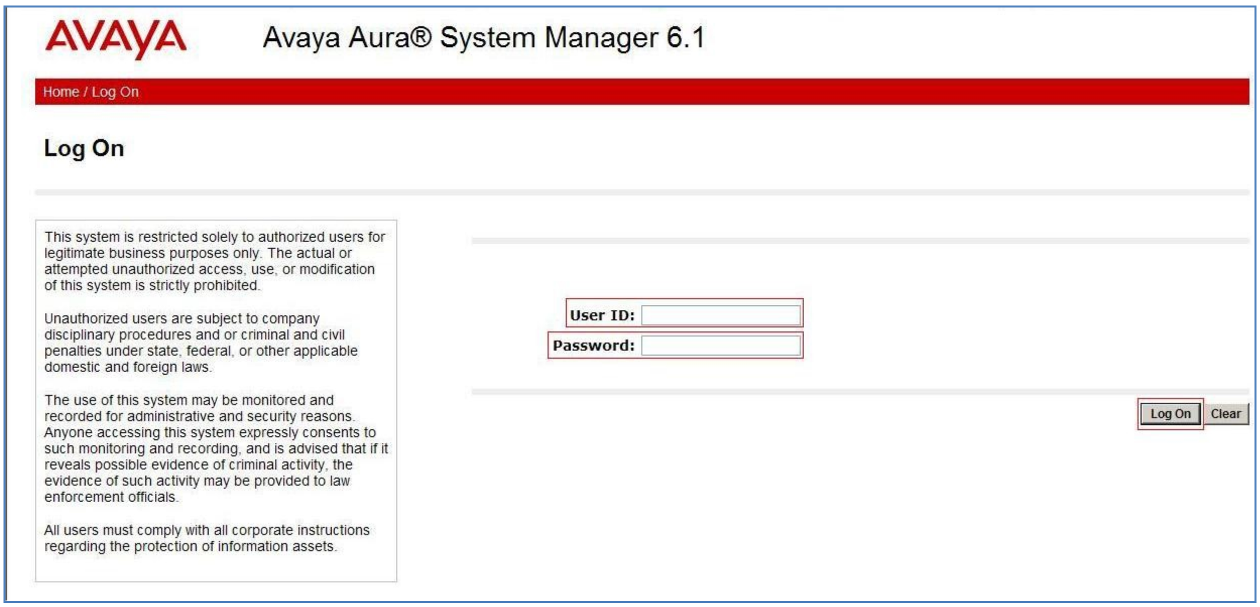
```
change station 2057                                          Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                              Headset? n
       Jack:                                              Speaker? n
      Cable:                                             Mounting: d
      Floor:                                          Cord Length: 0
   Building:                                            Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                          4: uui-info
 2: call-appr                          5:
 3: call-appr                          6:
```

7. Configure Avaya Aura® Session Manager Release R6.1

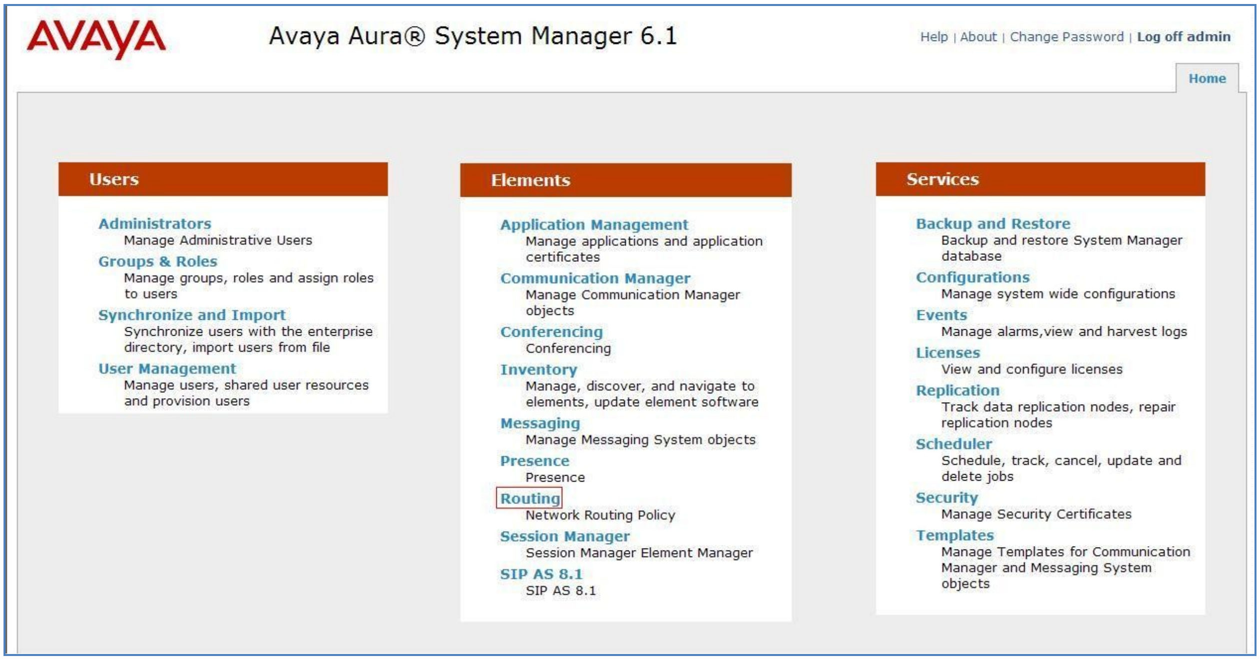
This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

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 **Section 11** of these Application Notes.

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 **User ID** and **Password** and press the **Log On** button



Once logged in, a Home Screen is displayed as below:



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

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

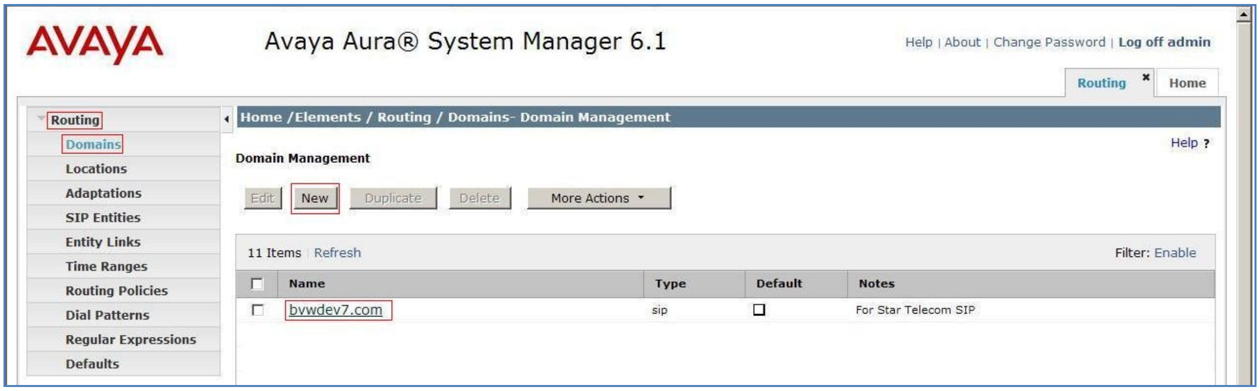
Step 8: "Dial Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)

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

7.1. Configure Domains

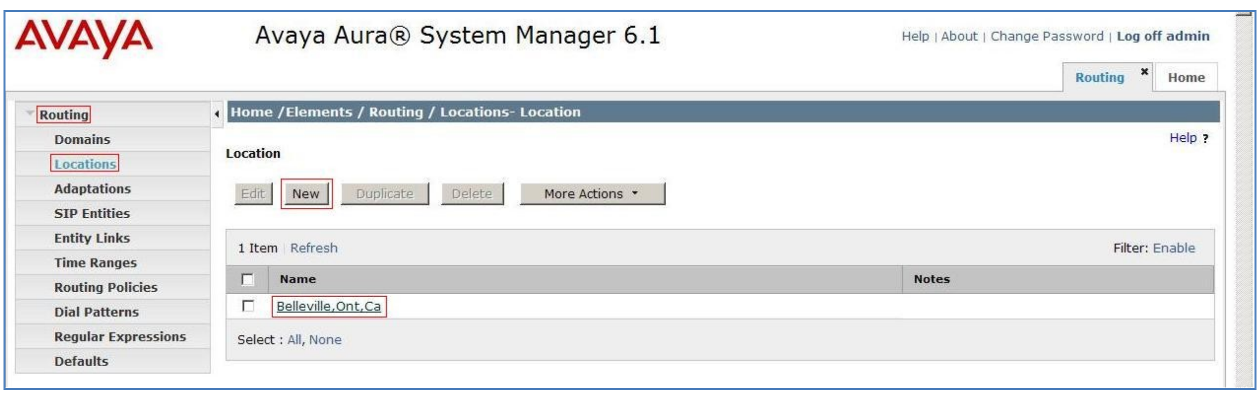
To add SIP domains that will be used with Session Manager, select **Routing → Domains**. Click the **New** button to add a new SIP domain entry. Click the Commit button after changes are completed.

The following screen shows the list of configured SIP domains. The domain **bvwdev7.com** is not known to the Star Telecom production service. The domain name should match the one used in the **ip-network-region** described in **Section 5.5.**

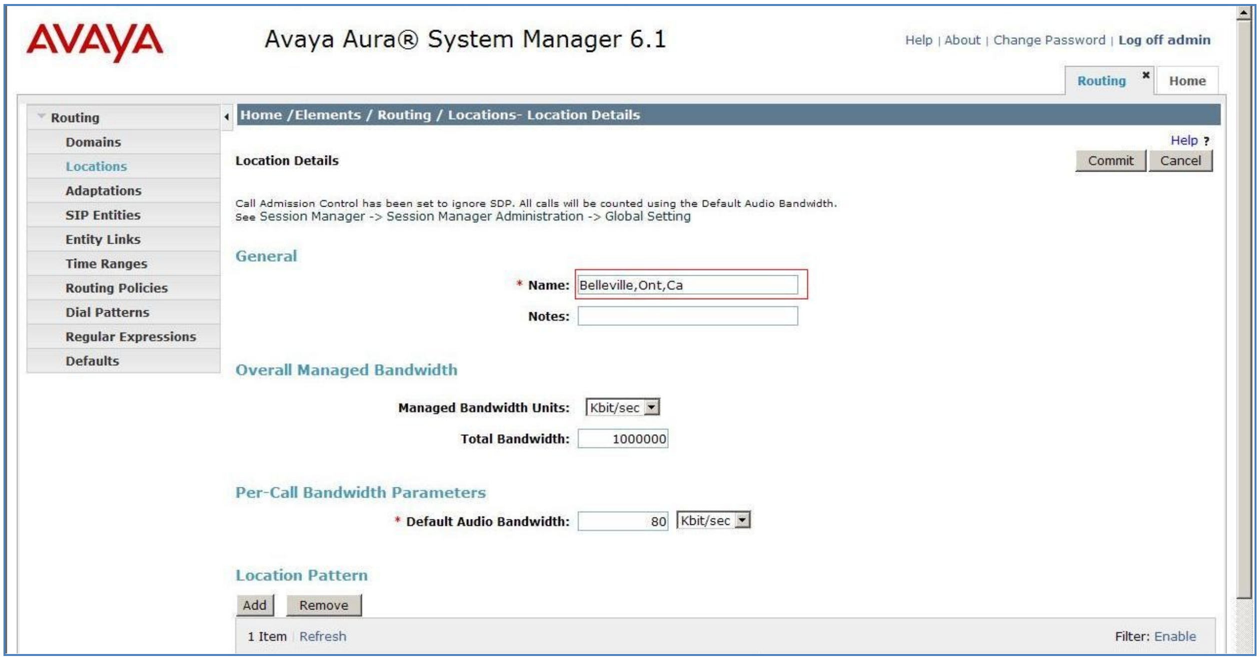


7.2. Configure Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for the enterprise SIP entities. To add 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 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.

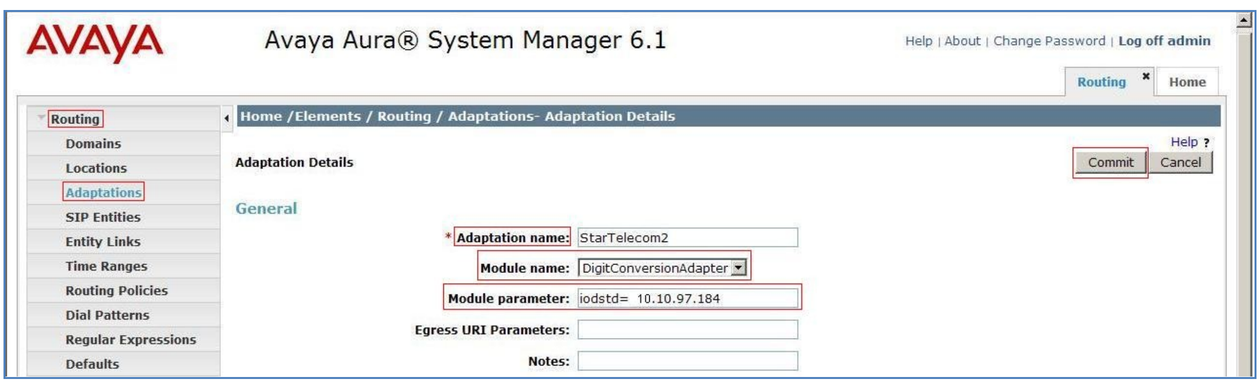


The following screen shows the location details for the location named **Belleville,Ont,Ca**, corresponding to the Acme Packet 3800 Session Border Controller. Later, the location with name Belleville,Ont,Ca will be assigned to the corresponding SIP Entity.



7.3. Configure Adaptations

Adaptation is configured to adapt the destinated IP address of Acme SBC. In order to add a new adaptation, select **Routing** →**Adaptations**. Click the New button to add an adaptation (not shown). Enter an appropriate **Adaptation name** to identify the adaptation. Select **DigitConversionAdapter** from the **Module name** drop-down menu. For the **Module parameter** field, the **iodstd** parameters should be assigned to IP address of internal interface of Acme SBC. Click the **Commit** button after changes are completed.



7.4. Configure SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **Routing → SIP Entities** and then click on the New button to add an adaptation (not shown). The following will need to be entered for each SIP Entity.

Under **General:**

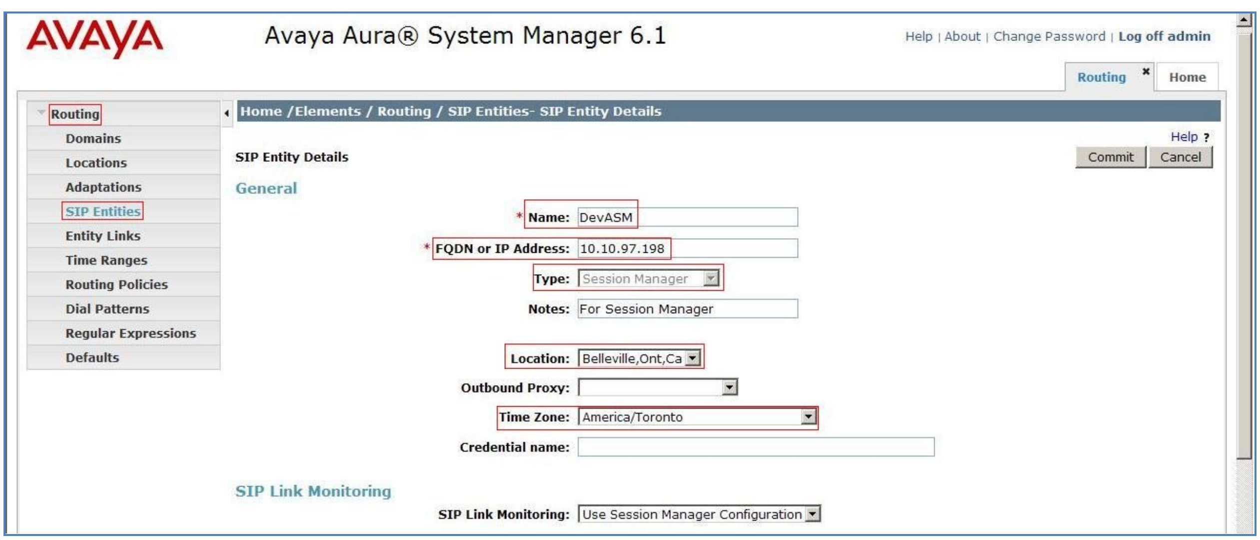
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Other** for the SBC SIP entity
- In the **Location** field select the appropriate location (Configured in **section 7.2**) from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration, there are three SIP Entities.

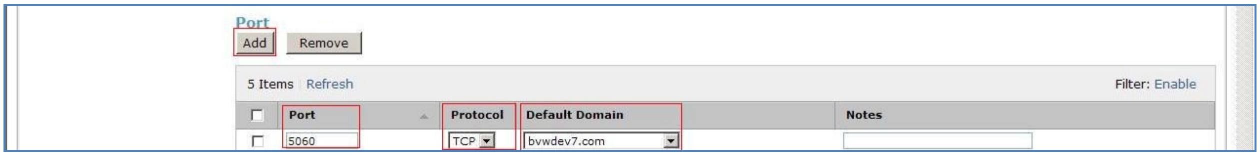
- Session Manager SIP Entity
- Communication Manager SIP Entity
- Session Border Controller SIP Entity

7.4.1. Configure Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager named **DevASM**. The **IP Address** field is set to the IP address **10.10.97.198** of the Session Manager SIP signaling interface.

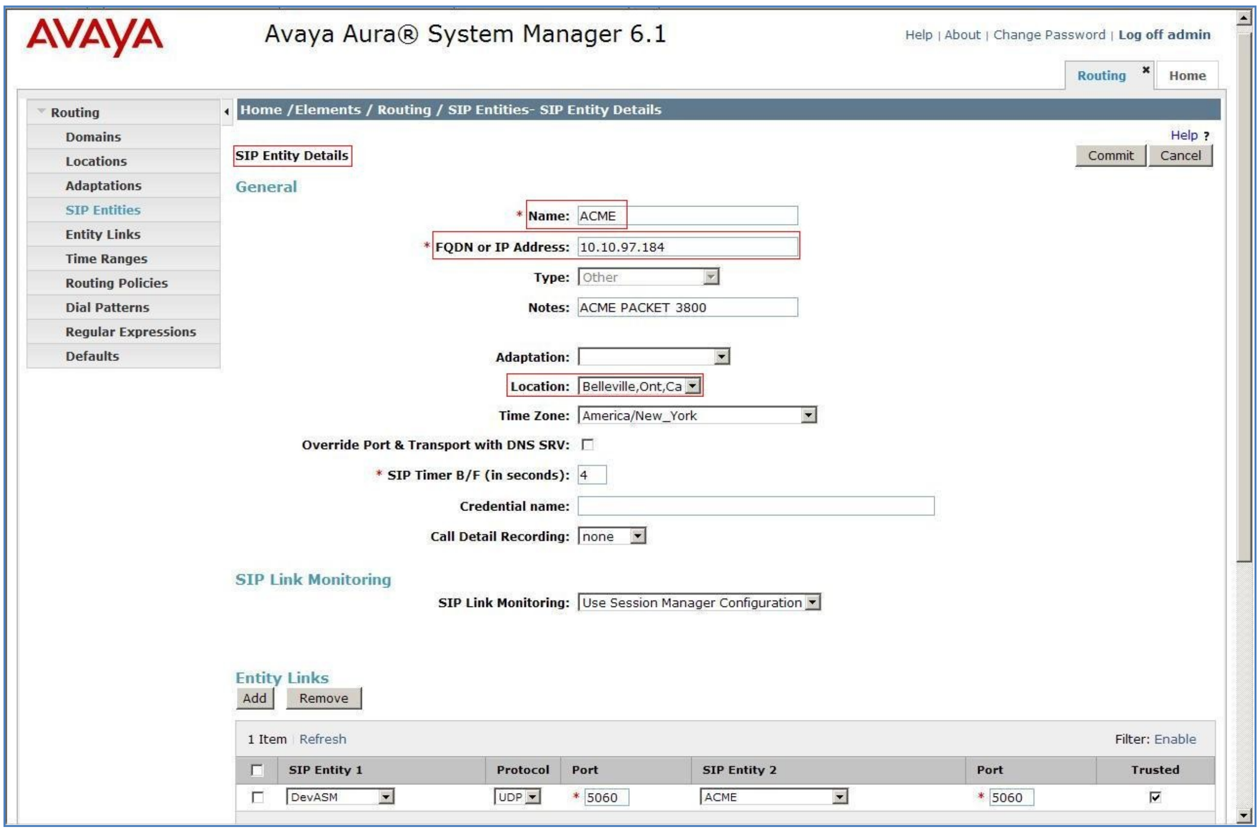


Click the **Add** button to configure a new port. **Protocol TCP** is used in the sample configuration for improved visibility during testing. **Port** is **5060** and **Default Domain** is **bvwdev7.com**.



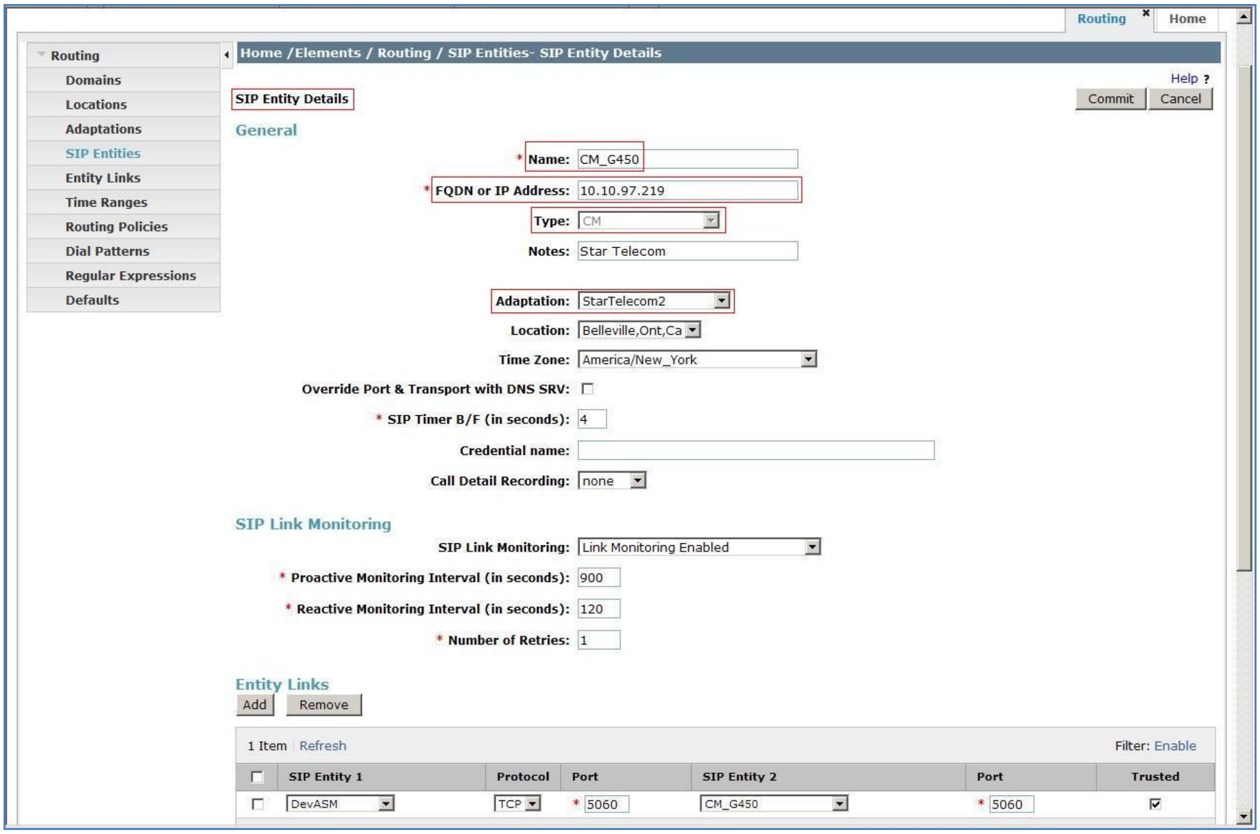
7.4.2. Configure Acme Packet Session Border Controller SIP Entity

The following screen shows the **SIP Entity Details** for the Acme Packet 3800 SBC named **ACME**. The **IP Address** field is configured with the Acme Packet 3800 SBC inside IP Address (**10.10.97.184**). The **Location** is **Belleville, Ont,Ca**.



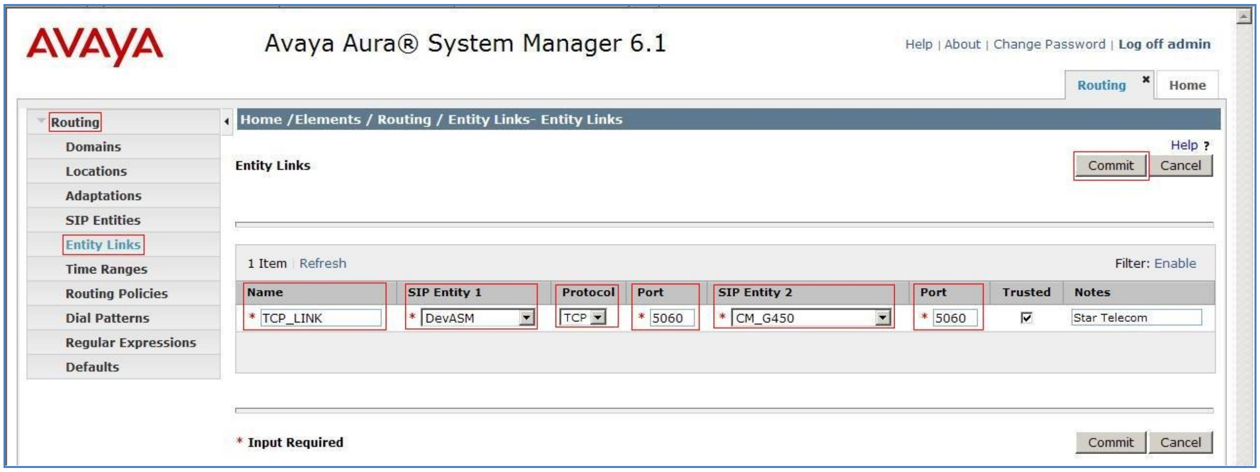
7.4.3. Configure Avaya Aura® Communication Manager SIP Entity

The following screen shows a portion of the **SIP Entity Details** corresponding to an Avaya Aura® Communication Manager SIP Entity named **CM_G450**. The **IP Address** field contains the IP Address of the processor ethernet (**10.10.97.219**). The **Type** field is set as **CM.** The **Adaptation StarTelecom2** is in used.

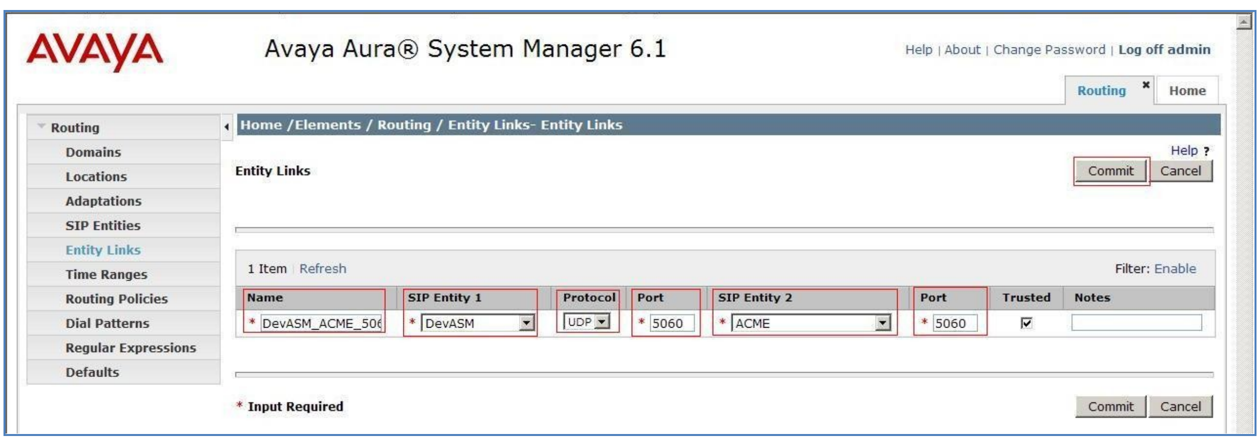


7.5. Configure Entity Links

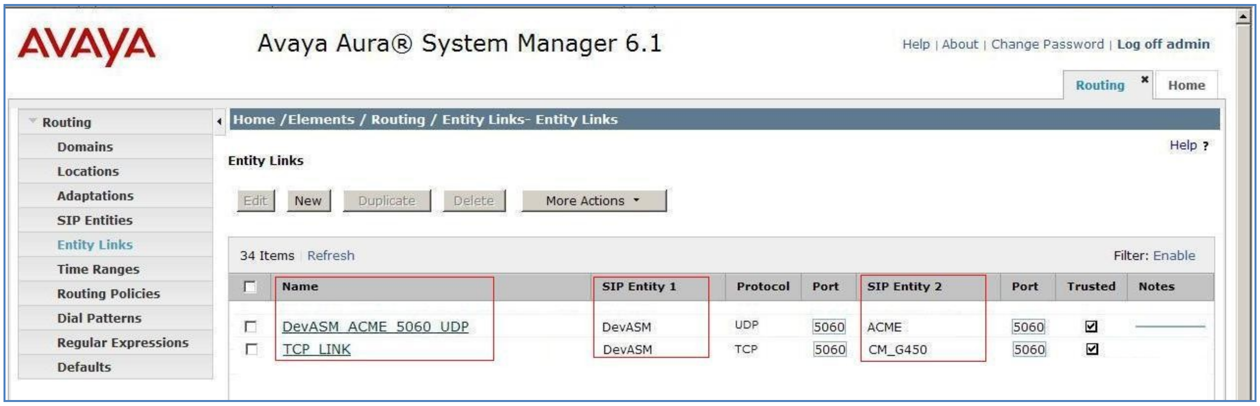
A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Routing →Entity Links**. Click the New button to add a link for Communication Manager (not shown). Assign an appropriate **Name**, and select the Session Manager entity as **SIP Entity 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 3800 SBC (not shown). 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 **UDP**, select **Port 5060**, and click **Commit**.

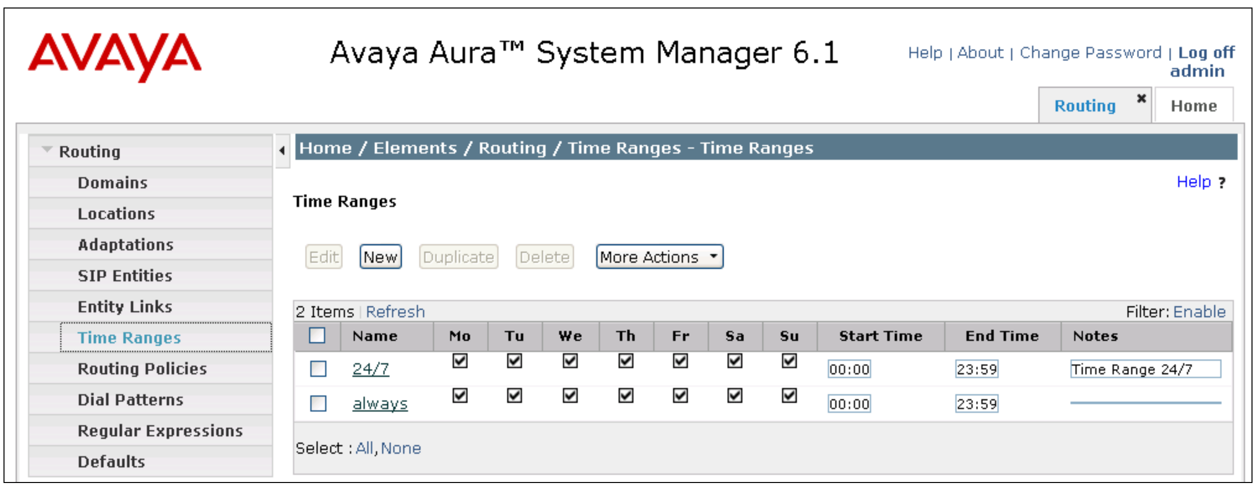


The following screen shows the list of configured links. Each of the links uses the entity named DevASM as SIP Entity 1, and the appropriate entity, such as CM_G450, ACME for SIP Entity 2.



7.6. Configure Time Ranges

Time Ranges is configured for time-based-routing. In order to add a Time Ranges, select **Routing → Time Ranges** and then click **New** button. The Routing Policies shown subsequently will use the 24/7 range since time-based routing was not the focus of these Application Notes.



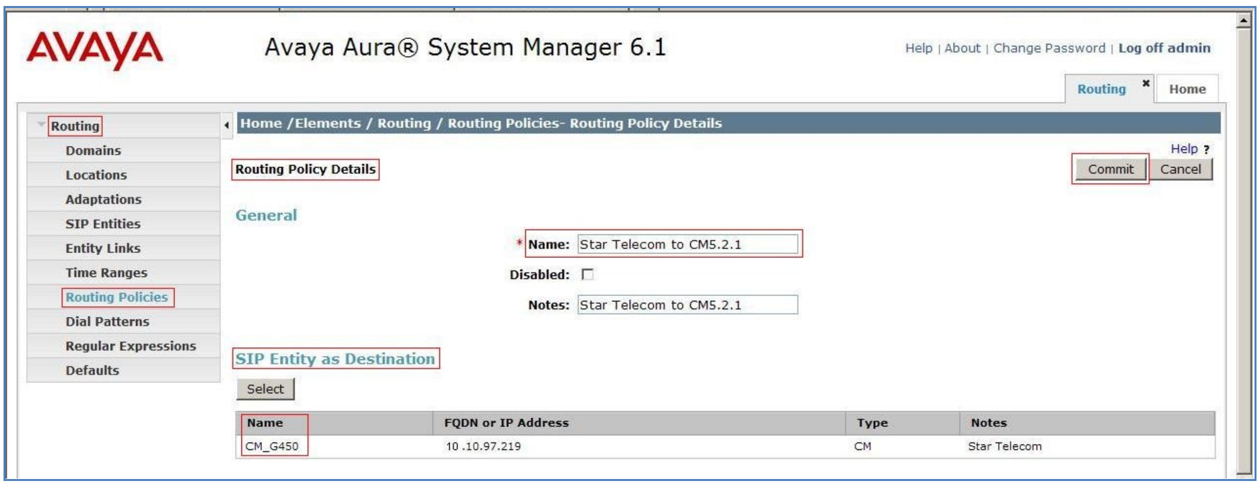
7.7. Configure Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a new routing policy, select **Routing → Routing Policies** and then click on the New button to create a routing policy (not shown).

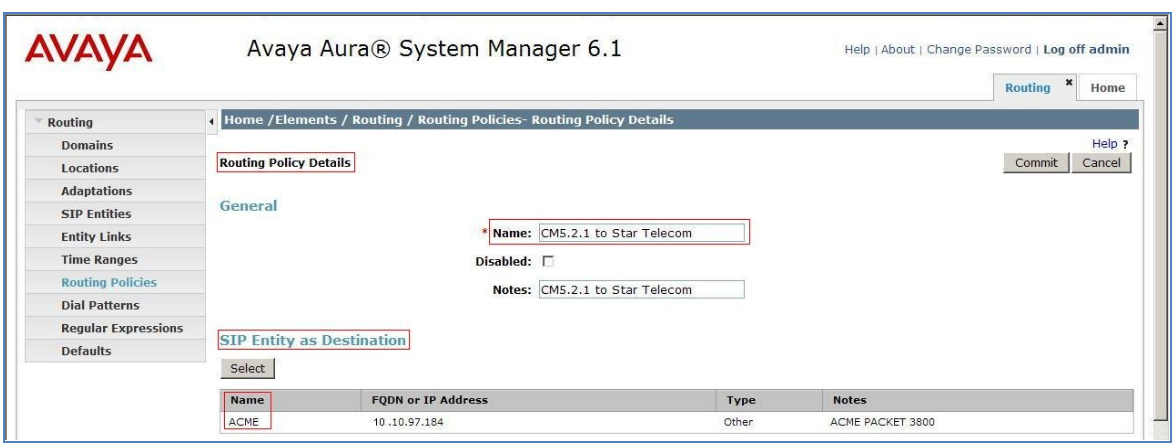
Under **General:**

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies

The following screen shows the **Routing Policy Details** for the policy named **Star Telecom to CM5.2.1** associated with incoming PSTN calls from Star Telecom to Communication Manager. Observe the **SIP Entity as Destination** is the entity named **CM_G450**.



The following screen shows the **Routing Policy Details** for the policy named **CM5.2.1 to Star Telecom** associated with outgoing calls from Communication Manager to the PSTN via Star Telecom through the Acme Packet 3800 SBC. Observe the **SIP Entity as Destination** is the entity named **ACME**.

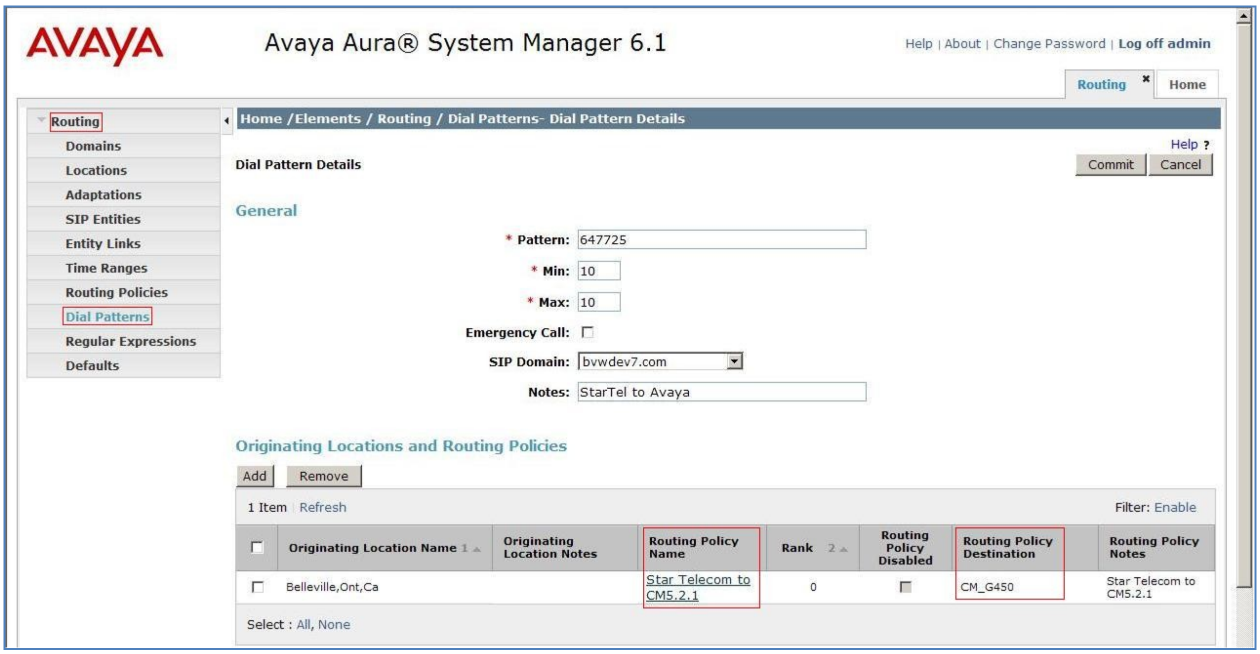


7.8. Configure Dial Patterns

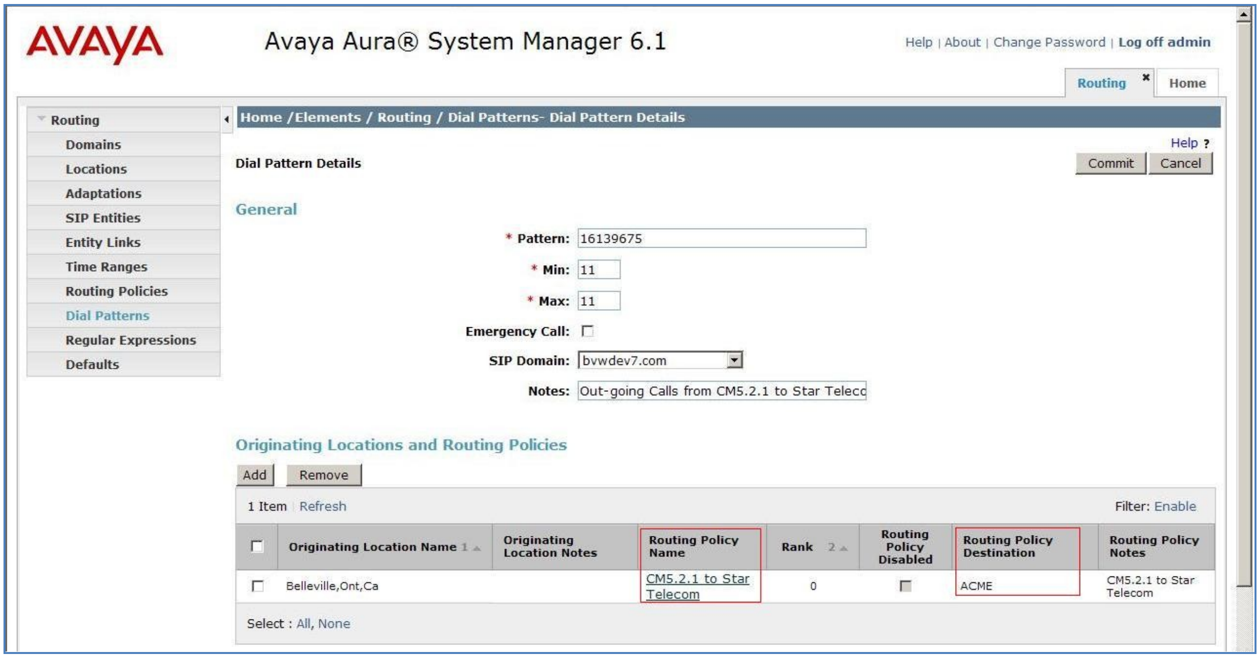
A dial pattern must be defined to direct calls to the appropriate telephony system. To add a new dial pattern, select **Routing → Dial Patterns** and then click on the New button to create a dial pattern (not shown). Under **General:**

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialed number
- In the **Max** field enter the maximum length of the dialed number
- In the **SIP Domain** field select the domain configured in **Section 7.1**

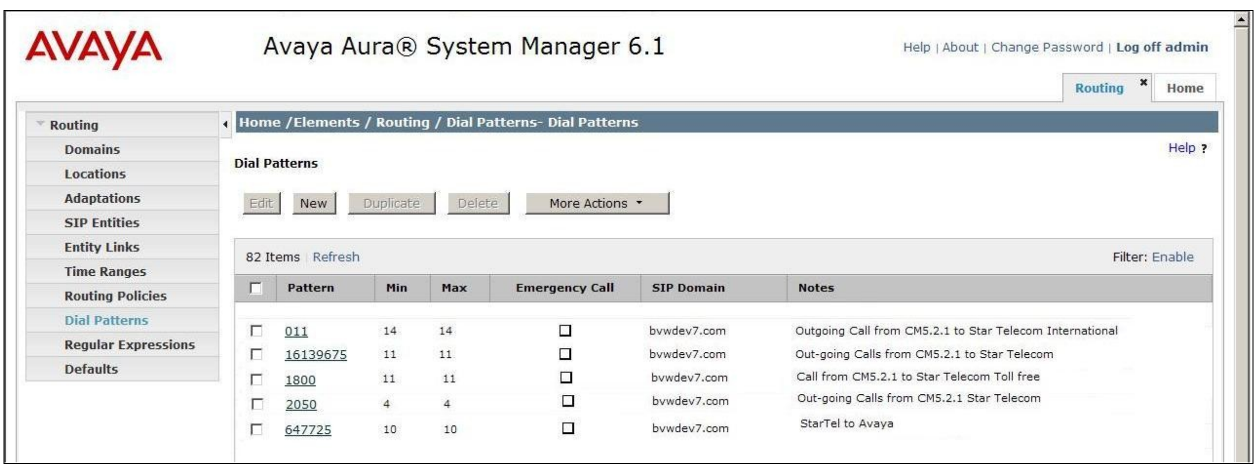
The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the Star Telecom Service, such as 6477252051, Star Telecom delivers the number to the enterprise, and the Acme Packet 3800 SBC sends the call to Session Manager. Under **Originating Locations and Routing Policies**, the **Routing Policy Name Star Telecom to CM5.2.1** is selected, which sends the call to Communication Manager as described previously and **Routing Policy Destination** is set as **CM_G450**.



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 1-613-967-5206, Communication Manager sends the call to Session Manager. Session Manager will match the dial pattern shown below and send the call to the Acme Packet 3800 SBC via the **Routing Policy Name CM5.2.1 to Star Telecom**. The **Routing Policy Destination** is set as **ACME**.



The following screen illustrates an example dial pattern used to verify inbound and outbound calls between the enterprise and the PSTN.



8. Configure Acme Packet Net-Net 3800

This section describes the configuration of the Acme Packet Net-Net 3800 necessary for interoperability with the Avaya Communication Manager and Star Telecom systems. The Net-Net 3800 was configured via the Acme Packet Command Line Interface (ACLI). This section assumes the reader is familiar with accessing and configuring the Acme Packet products.

This section will not attempt to describe each component in its entirety but instead will highlight critical fields in each component which relates to the functionality in these Application Notes. The remaining fields are generally the default/standard value used by the Net-Net 3800 for that field. In this testing, according to the configuration reference in **Figure 1**, the Avaya elements reside on the Private side and the Star Telecom elements reside on the Public side of the network.

8.1. Acme Packet Command Line Interface Summary

Note: Net-Net 3800 provisioning applicable to the reference configuration is shown in **bold** text. Other parameters and setting are shown for informational purposes.

The Net-Net 3800 is configured using the Acme Packet Command Line Interface (ACLI). The following are the generic ACLI steps for configuring various elements.

1. Access the console port of the Net-Net 3800 using a PC and a terminal emulation program such as HyperTerminal (use the RJ-45 to DB9 adapter as packaged with the Net-Net 3800 server for cable connection). Use the following settings for the serial port on the PC.
    - Bits per second: 115200
    - Data bits: 8
    - Parity: None
    - Stop bits: 1
    - Flow control: None
2. Log in to the Net-Net 3800 with the user password.
3. Enable the Super-user mode by entering the **enable** command and then the super user password. The command prompt will change to include a # instead of a > while in Super user mode. This level of system access (i.e. at the **acmesystem#** prompt) will be referred to as the **main** level of the ACLI. Specific sub-levels of the ACLI will then be accessed to configure specific **elements** and specific **parameters** of those elements.
4. In Super-user mode, enter the **configure terminal** command. The **configure terminal** command is used to access the system level where all operating and system elements may be configured. This level of system access will be referred to as the ***configuration*** level.
5. Enter the name of an element to be configured (e.g., **system**).
6. Enter the name of a sub-element, if any (e.g., **phy-interface).**
7. Enter the name of an element parameter followed by its value (e.g., **name INSIDE**).
8. Enter **done** to save changes to the element. Use of the **done** command causes the system to save and display the settings for the current element.
9. Enter **exit** as many times as necessary to return to the configuration level.
10. Repeat **Steps 5 - 9** to configure all the elements.

11. Enter **exit** to return to the main level.
12. Type **save-config** to save the entire configuration.
13. Type **activate-config** to activate the entire configuration.

After accessing different levels of the ACLI to configure elements and parameters, it is necessary to return to the main level in order to run certain tasks such as saving the configuration, activating the configuration, and rebooting the system.

8.2. Configure Physical and Network Interfaces

As part of the compliance test, the Ethernet slot 0/port 0 was connected to the internal corporate LAN. The Ethernet interface slot 1/port 0 was connected to the external un-trusted network. A network interface was defined for each physical interface to assign it a routable IP address.

The physical interface below defines the ports on the interface connected to the network on which the Avaya elements reside.

```
phy-interface
        name                                    INSIDE
        operation-type                   Media
        port                                    0
        slot                                    0
        virtual-mac
        admin-state                             enabled
        auto-negotiation                 enabled
        duplex-mode                      FULL
        speed                                   100
        overload-protection              disabled
        last-modified-by                  admin@console
        last-modified-date               2011-20-10 10:11:20
```

The physical interface below defines the ports on the interface connected to the network on which the Star Telecom elements reside.

```
phy-interface
        name                                    OUTSIDE
        operation-type                   Media
        port                                    0
        slot                                    1
        virtual-mac
        admin-state                             enabled
        auto-negotiation                 enabled
        duplex-mode                      FULL
        speed                                   100
        overload-protection              disabled
        last-modified-by                 admin@console
        last-modified-date               2011-20-10 10:11:30
```

The network interface below defines the IP addresses on the interface connected to the network on which the Avaya elements reside.

```
network-interface
        name                                    INSIDE
        sub-port-id                             0
        description
        hostname
        ip-address                              10.10.97.184
        pri-utility-addr
        sec-utility-addr
        netmask                                 255.255.255.192
        gateway                                 10.10.97.129
        sec-gateway
        gw-heartbeat
                state                                   disabled
                heartbeat                               0
                retry-count                             0
                retry-timeout                           1
                health-score                            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                             11
        hip-ip-list                             10.10.97.184
        ftp-address
        icmp-address                     10.10.97.184
        snmp-address
        telnet-address
        ssh-address
        last-modified-by                 admin@console
        last-modified-date               2011-20-10 10:20:11
```

The network interface below defines the IP addresses on the interface connected to the network on which the Star Telecom elements reside.

```
network-interface
        name                                    OUTSIDE
        sub-port-id                             0
        description
        hostname
        ip-address                              10.10.98.98
        pri-utility-addr
        sec-utility-addr
        netmask                                 255.255.255.224
        gateway                                 10.10.98.97
        sec-gateway
        gw-heartbeat
                state                                   disabled
                heartbeat                               0
                retry-count                             0
                retry-timeout                           1
                health-score                            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                             11
        hip-ip-list                             10.10.98.98
        ftp-address
        icmp-address                     10.10.98.98
        snmp-address
        telnet-address
        ssh-address
        last-modified-by                 admin@console
        last-modified-date               2011-20-10 15:22:28
```

8.3. Configure Realm

A realm represents a group of related Net-Net 3800 components. Two realms were defined for the compliance test. The realm configuration **INSIDE** below represents the internal network on which the Avaya elements reside.

```
realm-config
        identifier                                    INSIDE
        description
        addr-prefix                                   0.0.0.0
        network-interfaces
                                               INSIDE:0
        mm-in-realm                            disabled

       <Text removed for brevity>
```

The realm configuration OUTSIDE below represents the external network on which the Star Telecom system resides.

```
realm-config
        identifier                                    OUTSIDE
        description
        addr-prefix                                   0.0.0.0
        network-interfaces
                                              OUTSIDE:0
        mm-in-realm                            disabled

       <Text removed for brevity>
```

8.4. Configure Session Agent

A session agent defines the characteristics of a signaling peer to the Net-Net 3800.
The **session agent** below represents the Star Telecom border element. For redundancy purposes, Star Telecom uses multiple border elements, therefore the session agents will be defined accordingly (In below example, defined session agent with IP address as **20.20.103.125**). The Acme will attempt to send calls to the border element. The **in-manipulationid** and **out-manipulationid** define the SIP header manipulation applying to the OUTSIDE realm.

```
session-agent
        hostname                                20.20.103.125
        ip-address                              20.20.103.125
        port                                    5060
        state                                   enabled
        app-protocol                     SIP
        app-type
        transport-method                 UDP
        realm-id                                OUTSIDE
        egress-realm-id
        description                             StarTel_Avaya
        carriers
        allow-next-hop-lp                enabled
        constraints                             disabled

       <Text removed for brevity>

        ping-interval                           0
        ping-send-mode                    keep-alive

       <Text removed for brevity>

        ping-from-user-part
        li-trust-me                             disabled
        in-manipulationid                StarTel_TO_Avaya_NAT_IP
        out-manipulationid               Avaya_TO_StarTel_NAT_IP
        manipulation-string
```

The **session agent** below represents the configuration for inside interface to connect to Session Manager mentioned in **Section 7.4.**

```
session-agent
        hostname                                10.10.97.198
        ip-address                              10.10.97.198
        port                                    5060
        state                                   enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                                INSIDE
        egress-realm-id
        description                             StarTel_Avaya
        carriers
        allow-next-hop-lp               enabled
        constraints                             disabled

       <Text removed for brevity>
```

8.5. Configure SIP

The SIP configuration (***sip-config***) defines the global system-wide SIP parameters. The key SIP configuration (***sip-config***) fields are:

- **home-realm-id**: The name of the realm on the private side of the Net-Net 3800.
- **egress-realm-id**: The name of the realm on the private side of the Net-Net 3800.

```
sip-config
        state                                   enabled
        operation-mode                  dialog
        dialog-transparency             enabled
        home-realm-id                   INSIDE
        egress-realm-id                 INSIDE
        nat-mode                                None

       <Text removed for brevity>
```

8.6. Configure SIP Interface

The SIP interface (***sip-interface***) defines the receiving characteristics of the SIP interfaces on the Net-Net 3800. Two SIP interfaces were defined; one for each realm.
The SIP interface below is used to communicate with the Avaya Communication Manager system.

```
sip-interface
        state                                  enabled
        realm-id                               INSIDE
        description
        sip-port
                address                                     10.10.97.184
                port                                        5060
                transport-protocol                    UDP
                tls-profile
                allow-anonymous                       all

       <Text removed for brevity>
```

The SIP interface below is used to communicate with the Star Telecom system.

```
sip-interface
        state                                  enabled
        realm-id                               OUTSIDE
        description
        sip-port
                address                                     10.10.98.98
                port                                        5060
                transport-protocol                    UDP
                tls-profile
                allow-anonymous                       all

       <Text removed for brevity>
```

8.7. Configure SIP Manipulation

SIP manipulations are rules used to modify the SIP messages (if necessary) for interoperability. The following **sip-manipulation Avaya_TO_StarTel_NAT_IP** is applied to **OUTSIDE** realm ***out-manipulationid***. These rules perform the following:

- The header rule **manipTo** performs address translation and topology hiding for SIP messages between the Star Telecom and the Avaya elements.

```
sip-manipulation
        name                                      Avaya_TO_StarTel_NAT_IP
        description
        split-headers
        join-headers
                header-rule
                name                                           manipTo
                header-name                                    To
                action                                         manipulate
                comparison-type                         case-sensitive
                msg-type                                       any
                methods
                match-value
                new-value
                element-rule
                        name                                         To
                        parameter-name
                        type                                                uri-host
                        action                                              replace
                        match-val-type                               any
                        comparison-type                              case-sensitive
                        match-value
                        new-value                                    $REMOTE_IP
        header-rule
                name                                           HistRegex
                header-name                                    History-Info
                action                                         store
                comparison-type                                pattern-rule
                msg-type                                       request
                methods                                        INVITE
                match-value                                    ()
                new-value
                element-rule
                        name                                         GetUser
                        parameter-name
                        type                                                uri-user
                        action                                              store
                        match-val-type                               any
                        comparison-type                              pattern-rule
                        match-value
                        new-value
                element-rule
                        name                                         GetHost
                        parameter-name
                        type                                                uri-host
                        action                                              store
                        match-val-type                               any
                        comparison-type                              pattern-rule
                        match-value
                        new-value
```

```
            element-rule
                    name                                                GetUserReason1
                    parameter-name
                    type                                                       header-value
                    action                                                     store
                    match-val-type                                      any
                    comparison-type                                     pattern-rule
                    match-value                                         (.*)(Moved)(.*)
                    new-value
            element-rule
                    name                                                GetUserReason2
                    parameter-name
                    type                                                       header-value
                    action                                                     store
                    match-val-type                                      any
                    comparison-type                                     pattern-rule
                    match-value                                         (.*)(Busy)(.*)
                    new-value
            element-rule
                    name                                                GetUserReason3
                    parameter-name
                    type                                                       header-value
                    action                                                     store
                    match-val-type                                      any
                    comparison-type                                     pattern-rule
                    match-value
(.*)(Unavailable)(.*)
                    new-value
    header-rule
            name                                              AddDiversion1
            header-name                                       Diversion
            action                                            add
            comparison-type                           boolean
            msg-type                                          request
            methods                                           INVITE
            match-value
$HistRegex[0].$GetUserReason1
            new-value                     <sip:+$HistRegex[0].$GetUser.$0+@+
                                  $HistRegex[0].$GetHost.$0+>;privacy=off;
                                  reason=unconditional;screen=no

    header-rule
            name                                              AddDiversion2
            header-name                                       Diversion
            action                                            add
            comparison-type                           boolean
            msg-type                                          request
            methods                                           INVITE
            match-value
$HistRegex[0].$GetUserReason2
            new-value
                                          <sip:+$HistRegex[0].$GetUser.$0+@+
                                  $HistRegex[0].$GetHost.$0+>;privacy=off;
                                  reason=user\-busy;screen=no
    header-rule
            name                                              AddDiversion3
            header-name                                       Diversion
            action                                            add
            comparison-type                           boolean
            msg-type                                          request
            methods                                           INVITE
            match-value
$HistRegex[0].$GetUserReason3
```

```
            new-value
       <sip:+$HistRegex[0].$GetUser.$0+@+

$HistRegex[0].$GetHost.$0+>;privacy=off;
                                                                reason=no\-
answer;screen=no
        header-rule
            name                                  delHistInfo
            header-name                           History-Info
            action                                delete
            comparison-type                  case-sensitive
            msg-type                              any
            methods                               INVITE
            match-value
            new-value
        header-rule
            name                                  manipFrom
            header-name                           From
            action                                manipulate
            comparison-type                  case-sensitive
            msg-type                              any
            methods
            match-value
            new-value
            element-rule
                name                                  From
                parameter-name
                type                                       uri-host
                action                                     replace
                match-val-type                        any
                comparison-type                       case-sensitive
                match-value
                new-value                             10.10.98.98

       header-rule
            name                                  modReferTo
            header-name                           Refer-To
            action                                manipulate
            comparison-type                  case-sensitive
            msg-type                              any
            methods                          REFER
            match-value
            new-value
            element-rule
                name                                  modmline
                parameter-name
                type                                       uri-host
                action                                     find-replace-
all
                match-val-type                        any
                comparison-type                       case-sensitive
                match-value
                new-value                             $LOCAL_IP

       header-rule
            name                                  mod302
            header-name                           Contact
            action                                manipulate
            comparison-type                  case-sensitive
            msg-type                              Reply
            methods                          INVITE
            match-value
```

```
            new-value
            element-rule
                    name                                         modmline
                    parameter-name
                    type                                                uri-host
                    action                                              find-replace-
all
                    match-val-type                               any
                    comparison-type                              case-sensitive
                    match-value
                    new-value                                    $LOCAL_IP
        last-modified-by                admin@console
        last-modified-date              2011-20-10 21:42:22
```

The following **sip-manipulation StarTel_TO_Avaya_NAT_IP,** ***in-manipulationid,*** is applied to **OUTSIDE** realm and translates the SIP header information for Avaya Communication Manager to understand. These rules perform the following:

- The header rules **manipRURI** changes IP address to the Avaya Communication Manager Domain Name in the Request URI headers sent to the Avaya Communication Manager elements.

```
sip-manipulation
        name                                    StarTel_TO_Avaya_NAT_IP
        description
        split-headers
        join-headers
        header-rule
                name                                          manipRURI
                header-name                                   request-uri
                action                                        manipulate
                comparison-type                        case-sensitive
                msg-type                                      any
                methods                                       INVITE
                match-value
                new-value
                element-rule
                        name                                        modRURI
                        parameter-name
                        type                                               uri-host
                        action                                             replace
                        match-val-type                              any
                        comparison-type                             case-sensitive
                        match-value
                        new-value                                   bvwdev7.com
        header-rule
                name                                          manipTo
                header-name                                   To
                action                                        manipulate
                comparison-type                        case-sensitive
                msg-type                                      any
                methods
                match-value
                new-value
                element-rule
                        name                                        To
                        parameter-name
                        type                                               uri-host
```

```
                    action                                             replace
                    match-val-type                                     any
                    comparison-type                                    case-sensitive
                    match-value
                    new-value                                          bvwdev7.com
        last-modified-by               admin@console
        last-modified-date             2011-20-10 12:52:23
```

8.8. Configure Steering Pools

Steering pools define the range of ports to be used for the RTP voice stream. Two steering pools were defined, one for each realm.
The key steering pool (***steering-pool***) fields are:

- **ip-address:** The address of the interface on the Net-Net 3800.
- **start-port:** An even number of the port that begins the range.
- **end-port:** An odd number of the port that ends the range.
- **realm-id:** The realm to which this steering pool is assigned.

```
steering-pool
        ip-address                            10.10.98.98
        start-port                            20000
        end-port                              40000
        realm-id                              OUTSIDE
        network-interface
        last-modified-by               admin@console
        last-modified-date             2011-20-10 22:20:07
steering-pool
        ip-address                            10.10.97.184
        start-port                            20000
        end-port                              40000
        realm-id                              INSIDE
        network-interface
        last-modified-by               admin@console
        last-modified-date             2011-20-10 22:20:22
```

8.9. Configure Local Policy

The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, reside to the Star Telecom system and vice versa.

```
local-policy
        from-address
                                *
        to-address
                                  6477252050
                                  6477252051
                                6477252052
                                6477252053
                                6477252054
                                        6477252055
                                6477252056
                                6477252057
                                6477252058
                                6477252059

        source-realm
                                OUTSIDE
        description             StarTel_TO_Avaya
        activate-time           N/A
        deactivate-time        N/A
        state                   enabled
        policy-priority         none
        last-modified-by       admin@console
        last-modified-date     2011-20-10 14:44:50
        policy-attribute
                next-hop                        10.10.97.198
                realm                           INSIDE
                action                          none
                terminate-recursion       disabled
                carrier
                start-time                      0000
                end-time                        2400
                days-of-week                    U-S
                cost                            0
                app-protocol                    SIP
                state                           enabled
                methods
                media-profiles
                lookup                          single
                next-key
                eloc-str-lkup                   disabled
                eloc-str-match
```

```
local-policy
        from-address
                                bvwdev7.com
        to-address
                                *
        source-realm
                                INSIDE
        description             Avaya_TO_StarTel
```

```
    activate-time                  N/A
    deactivate-time               N/A
    state                          enabled
    policy-priority                none
    last-modified-by            admin@console
    last-modified-date          2011-20-10 20:25:30
    policy-attribute
            next-hop                                  20.20.103.125
            realm                                     OUTSIDE
            action                                    none
            terminate-recursion            disabled
            carrier
            start-time                                0000
            end-time                                  2400
            days-of-week                              U-S
            cost                                      0
            app-protocol                              SIP
            state                                     enabled
            methods
            media-profiles
            lookup                                    single
            next-key
            eloc-str-lkup                             disabled
            eloc-str-match
```

9. Verification Steps

The following steps may be used to verify the configuration.

9.1. General

Place an inbound or outbound call between a PSTN phone and an internal Avaya phone, answer the call, and verify that two-way speech path exists. Verify that the call remains stable for several minutes and disconnects properly.

9.1.1. Example for Inbound Call from PSTN via Star Telecom SIP Trunk

Incoming PSTN calls arrive from Star Telecom at the Acme Packet 3800 SBC, which sends the call to Session Manager. Session Manager sends the call to Communication Manager via the entity link corresponding to the Avaya S8300 PE. On Communication Manager, the incoming call arrives via signaling group 2 and trunk group 11.

The following Communication Manager **list trace** output shows a call incoming on trunk group 11. The PSTN telephone dialed 6477252057. The incoming call handling table for trunk group 11 converted the number to 2057. Extension 2057 is a H323 Telephone with IP Address 10.10.97.137. Initially, the G450 Media Gateway (10.10.97.247) 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 (10.10.97.137) to the inside of the Acme Packet 3800 SBC (10.10.97.184).

```
list trace tac *011                                                 Page   1

                                LIST TRACEtime            data
16:25:51 SIP<INVITE sip:6477252057@bvwdev7.com  SIP/2.0
16:25:51     Call-ID: 658200e7-7533-122f-6b8e-00259010ee66
16:25:51     active trunk-group 11 member 1    cid 0xff
16:25:51 SIP>SIP/2.0 180 Ringing
16:25:51     Call-ID: 658200e7-7533-122f-6b8e-00259010ee66
16:25:51     dial 2057
16:25:51     ring station      2057 cid 0xff
16:25:51     G711MU ss:off ps:20
             rgn:1 [10.10.97.137]:2752
             rgn:1 [10.10.97.247]:2052
16:25:51     G711MU ss:off ps:20
             rgn:1 [10.10.97.184]:20414
             rgn:1 [10.10.97.247]:2054
16:25:51     xoip options: fax:Relay modem:PT tty:US  uid:0x50033
             xoip ip: [10.10.97.247]:2054


list trace tac *011                                                 Page   2

                                LIST TRACE

time            data
16:25:52 SIP>SIP/2.0 200 OK
16:25:52     Call-ID: 658200e7-7533-122f-6b8e-00259010ee66
16:25:52     active station      2057 cid 0xff
16:25:53 SIP>SIP/2.0 200 OK
16:25:53     Call-ID: 658200e7-7533-122f-6b8e-00259010ee66
16:25:54 SIP>SIP/2.0 200 OK
```

```
16:25:54     Call-ID: 658200e7-7533-122f-6b8e-00259010ee66
16:25:56     idle station      2057 cid 0xff
```

9.1.2. Example for Outbound Call to PSTN via Star Telecom SIP Trunk

The following trace shows an outbound ARS call from IP Telephone x2057 to the PSTN number 6139675206. The call is routed to route pattern 1 and trunk group 10. The call initially uses the gateway (10.10.97.247), but after the call is answered, the call is shuffled to become an ip-direct connection between the IP Telephone (10.10.97.137) and the inside of the Acme 3800 SBC (10.10.97.184).

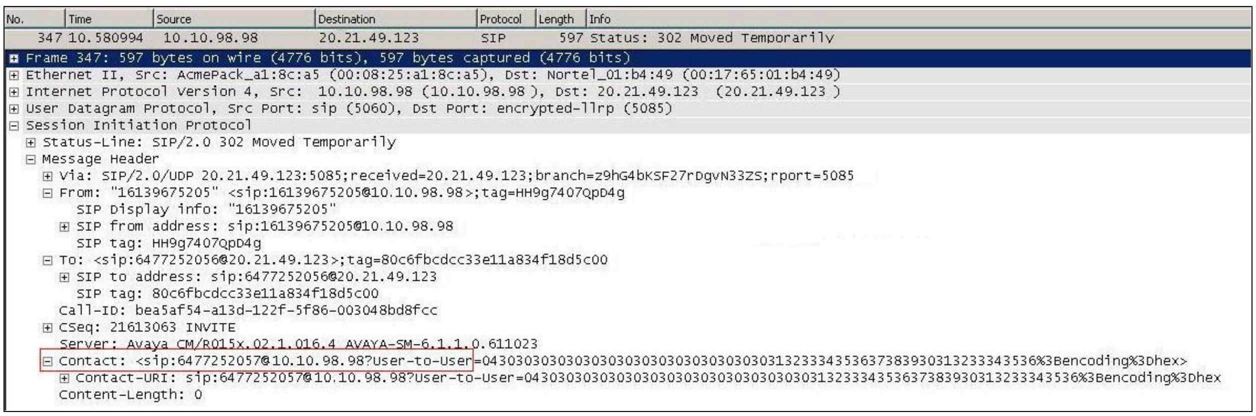
```
list trace tac *010                                                  Page   1

                                LIST TRACE

time            data

16:27:18     dial 916139675206 route:ARS
16:27:18     route-pattern  1 preference 1 location 1/ALL  cid 0x101
16:27:18     seize trunk-group 10 member 13    cid 0x101
16:27:18     Calling Number & Name 2057 IP_2057
16:27:18 SIP>INVITE sip:16139675206@bvwdev7.com SIP/2.0
16:27:18     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:18     Setup digits 16139675206
16:27:18     Calling Number & Name 6477252057 IP_2057
16:27:18 SIP<SIP/2.0 100 Trying
16:27:18     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:18     Proceed trunk-group 10 member 13    cid 0x101
16:27:19 SIP<SIP/2.0 183 Session Progress
16:27:19     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:19     G711MU ss:off ps:20
             rgn:1 [10.10.97.184]:20418
list trace tac *010                                                  Page   2

                  LIST TRACE

time            data
             rgn:1 [10.10.97.247]:2050
16:27:19     xoip options: fax:Relay modem:PT tty:US  uid:0x5000d
             xoip ip: [10.10.97.247]:2050
16:27:20 SIP<SIP/2.0 200 OK
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:20 SIP>ACK sip:16139675206@135.10.97.184:5060;transport=udp SI
16:27:20 SIP>P/2.0
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:20     active trunk-group 10 member 13    cid 0x101
16:27:20 SIP>INVITE sip:16139675206@135.10.97.184:5060;transport=udp
16:27:20 SIP> SIP/2.0
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:20 SIP<SIP/2.0 100 Trying
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:20 SIP<SIP/2.0 200 OK
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
list trace tac *010                                                  Page   3

                                LIST TRACE
```

```
time             data
16:27:20     G711MU ss:off ps:20
             rgn:1 [10.10.97.137]:2752
             rgn:1 [10.10.97.184]:20418
16:27:20 SIP>ACK sip:16139675206@135.10.97.184:5060;transport=udp SI
16:27:20 SIP>P/2.0
16:27:20     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:20     G711MU ss:off ps:20
             rgn:1 [10.10.97.184]:20418
             rgn:1 [10.10.97.137]:2752
16:27:23 SIP>BYE sip:16139675206@135.10.97.184:5060;transport=udp SI
16:27:23 SIP>P/2.0
16:27:23     Call-ID: 04aa07143e115ec4eafd7dd00
16:27:23     idle station      2057 cid 0x101
```

9.1.3. Redirection Verification Tests

1. Place an inbound call to a Star Telecom Transfer Connect service number enabled with Redirect features (302 redirection).

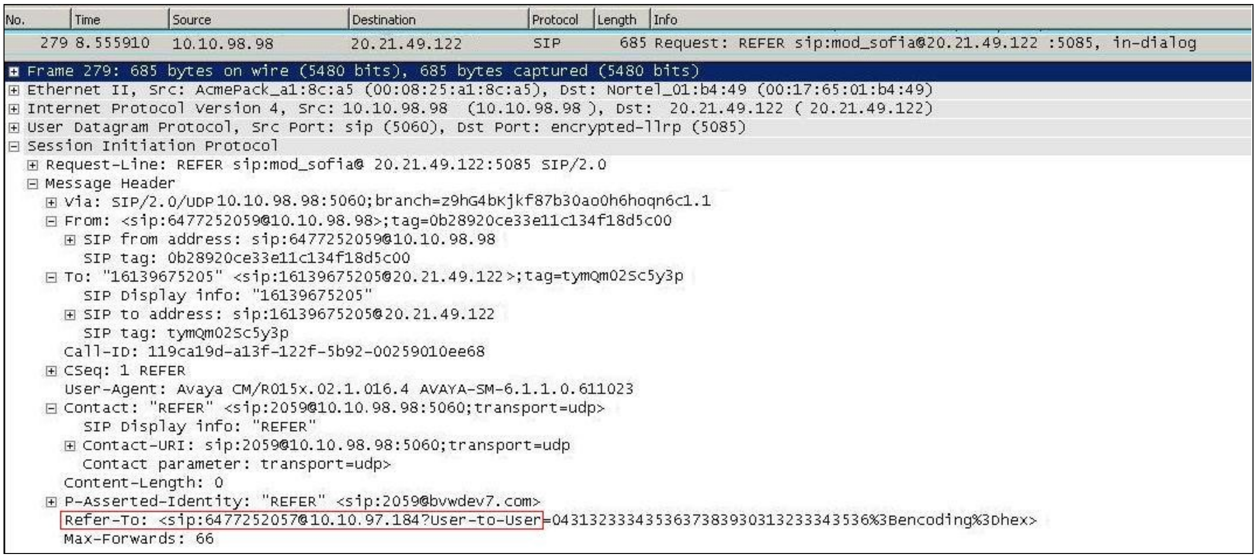
    - Verify that an appropriate Communication Manager vector immediately redirects the call back to the Star Telecom Transfer Connect service for redirection to the alternate destination. On Communication Manager enter the command *list trace vector x*, where *x* is an extension assigned to the associated vector. This will display the vector as it executes. Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet SBC public outside interface connection to the Star Telecom service. Verify that a 302 Moved Temporarily packet is sent and that it contains the alternate destination Star Telecom Transfer Connect service access number programmed in the vector, e.g 6477252057



    - When the redirection is complete, verify two way talk path.

2. Place an inbound call to a Star Telecom Transfer Connect service number enabled with IP Courtesy Transfer features (REFER redirection).

    - Verify that an appropriate Communication Manager vector immediately redirects the call back to the Star Telecom Transfer Connect service for redirection to the alternate destination.
    - On Communication Manager enter the command *list trace vector x*, where *x* is an extension assigned to the associated vector. This will display the vector as it executes.
    - Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet SBC public outside interface connection to the Star Telecom service. Verify that a REFER packet is sent and that it contains the alternate destination Star Telecom Transfer Connect service access number programmed in the vector, e.g. 6477252057.



- When the redirection is complete, verify two way talk path.

3. Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path, and that any defined user-to-user information (UUI) is displayed on the answering station (see **Section 6.3.3**).

10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.1, and the Acme Packet 3800 SBC R6.2 can be configured to interoperate successfully with Star Telecom. This solution allows Avaya Aura® Communication Manager and Avaya Aura® Session Manager customers access to the PSTN using a Star Telecom public SIP trunk service connection.

11. Additional References

Product documentation for ACME Packet may be found at:
http://www.acmepacket.com/support.htm

Product documentation for Avaya, 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

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