Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.2 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.2, and Acme Packet Net-Net 3800 with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.2 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.2 and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. **Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service**. Avaya Aura® Session Manager and Avaya Aura® Communication Manager interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.2, and Acme Packet Net-Net 3800[1]with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.2 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.2 and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

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2. General Test Approach and Test Results

The test environment consisted of:

1. A simulated enterprise with System Manager, Session Manager, Communication Manager, Avaya phones, fax machines (Ventafax application), Acme Packet Session Border Controller (SBC), and Avaya Modular Messaging.
2. A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via **AVPN** or **MIS-PNT** transport.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

[1] Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

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2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for sample call flows) between Session Manager, Communication Manager, Acme Packet SBC and the AT&T IP Toll Free service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network. The following features were tested as part of this effort:

- SIP trunking
- G729 and G711 codecs
- AT&T IP Toll Free calls to Communication Manager stations/agents using Vector Directory Numbers (VDNs), and vectors
- Two-way talk path establishment between PSTN and Communication Manager VDNs/vectors and stations/agents, via the AT&T Toll Free service
- Passing of RFC2833 DTMF events and their recognition by navigating automated voice menus
- T.38 fax calls between Communication Manager and the AT&T IP Toll Free service/PSTN G3 and SG3 fax endpoints
- Inbound AT&T IP Toll Free service calls to Communication Manager that are directly routed to stations, and alternatively can be covered to Avaya Modular Messaging
- Legacy Transfer Connect
- Alternate Destination Routing
- IP Re-routing
- Long Duration Calls

2.2. Test Results and Known Limitations

1. G.726 codec is not supported between AT&T IP Toll Free service and Communication Manager.
2. If Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Communication Manager, then Communication Manager selects a codec according to the priority order specified in the Communication Manager codec set, not the priority order specified in the SDP offer. For example, if the AT&T IP Toll Free service offers G.711, G.729A, and G.729B in that order, but the Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Communication Manager selects G.729B, not G.711. The practical resolution is to provision the Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.
3. Although Session Manager 6.2 and Communication Manager 5.2.1 support the possibility of using SIP phones as valid telephone extensions, SIP phones were not tested as part of the configuration used to validate this solution.
4. G.711 faxing is not supported between Communication Manager and the AT&T IP Toll Free service. Communication Manager does not support the protocol negotiation that AT&T requires for G.711 fax calls. T.38 faxing is supported, as is Group 3 and Super Group 3 fax. Fax speeds are limited to 9600 bps in the configuration tested. In addition, Fax Error Correction Mode (ECM) is not supported by Communication Manager

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5. Communication Manager does not return a **488 Codec Mismatch** when the codec set offered by AT&T IP Toll Free service is not one of the codec's configured on Communication Manger. Communication Manager returns a **500 Server Internal Error** instead. This problem was detected in **Service Pack 11**. A patch was issued to resolve this problem which will be delivered in **Service Pack 15**.

The test objectives stated in **Section 2.1** with limitations noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

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3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communication services for a particular enterprise site. In the reference configuration, Communication Manager 5.2.1 runs on an Avaya S8720 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. The G650 contains system boards such as the Control LAN (C-LAN) and Media Processor (MedPro). This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" telephones are represented with Avaya 96x0 and 96x1 Series IP Telephones running H.323, Avaya 6408D Series Digital Telephone, Avaya Analog phone and Avaya one-X® Agent, a PC based softphone.
- The Acme Packet SBC 3800[2] provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network[3]. UDP transport protocol is used between the Acme Packet SBC and the AT&T Toll Free service.
- An existing Avaya Modular Messaging system provides the corporate voice messaging capabilities in the reference configuration. **The provisioning of Modular Messaging is beyond the scope of this document and is shown here for illustrative purposes only**.
- Inbound calls were placed from PSTN to the Customer Premises Equipment (CPE) via the AT&T IP Toll Free service, through the Acme Packet SBC, Session Manager, and Communication Manager. Communication Manager terminated the calls using appropriate phone or fax stations. The H.323 phones in the CPE registered to the Communication Manager C-LANs.

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[2] Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

[3] The AT&T IP Toll Free service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Acme Packet SBC in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme Packet SBC and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

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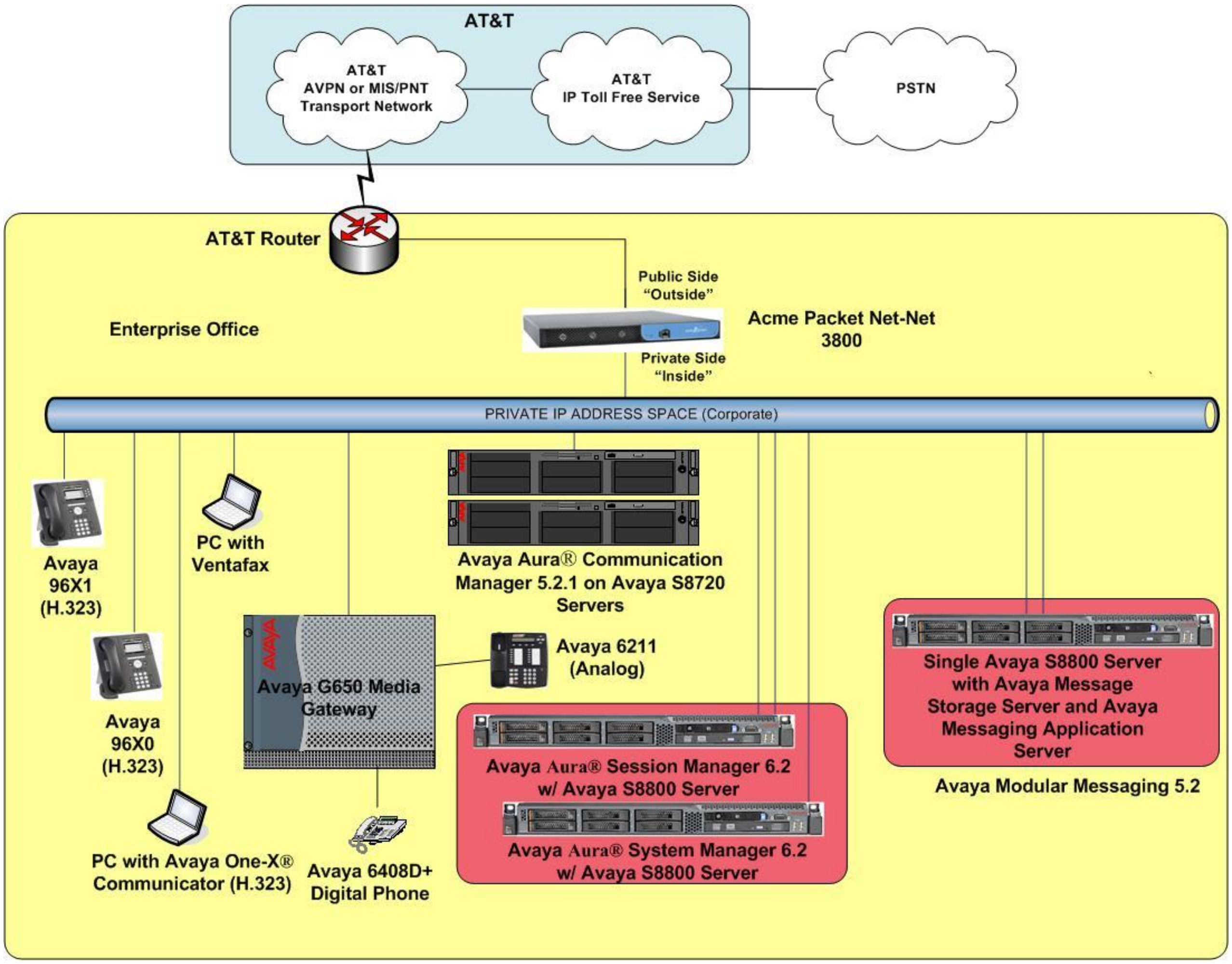

Figure 1: Reference Configuration

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3.1. Illustrative Configuration Information

The specific values used in this reference configuration listed in **Table 1** below are **for illustrative purposes only**. Customers must obtain and use the specific values for their configurations. For security purposes, real IP addresses and DNIS are not included.

Note - The AT&T IP Toll Free service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Toll Free service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® System Manager	
Management IP Address	10.80.150.209
Avaya Aura® Session Manager	
Management IP Address	10.80.150.210
Network IP Address	10.64.19.210
Avaya Aura® Communication Manager	
Control LAN (C-LAN) IP Address	10.80.130.206
Media Processor (MedPro) IP Address	10.80.130.207
Skill (Hunt Group) Extensions	53xxx
Agent Extensions	53xxx
Phone Extensions	50xxx
Announcement Extensions	33xxx
Vector Directory Numbers (VDN)	20xx
Acme Packet Session Border Controller	
IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Toll Free service)	192.168.62.50
IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager)	10.80.130.250
AT&T IP Toll Free service	
Border Element IP Address	192.242.225.210
DNIS Passed in Request URI used by Session Manager for routing	0000041535[0,1,7,8,9]10[5,6][0,1,7,8,9]

Table 1: Illustrative Values Used in this Reference Configuration

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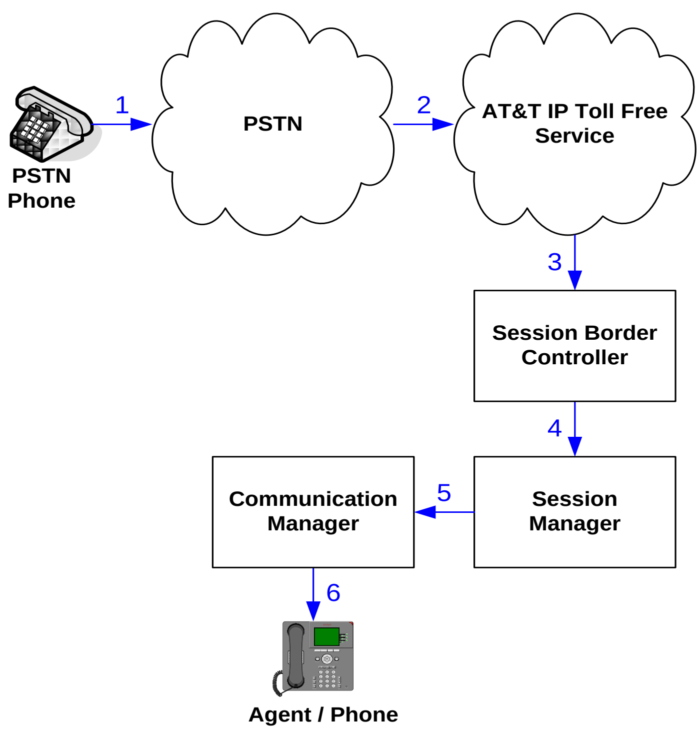
3.2. Call Flows

To understand how AT&T IP Toll Free service calls are handled by Session Manager and Communication Manager, two basic call flows are described in this section, however for brevity not all possible call flows are described.

3.2.1. Inbound Call to an Agent/Station using VDN/Vectors

The first call scenario illustrated in the figure below is an inbound AT&T IP Toll Free service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a phone, fax, or in some cases, a vector.

1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to Session Border Controller.
4. Session Border Controller performs any necessary SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines to where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to
   - A vector, which in turn, routes the call to an agent
   - Directly to an agent or a phone/fax extension.



Inbound AT&T IP Toll Free Call to VDN/Agent/Phone

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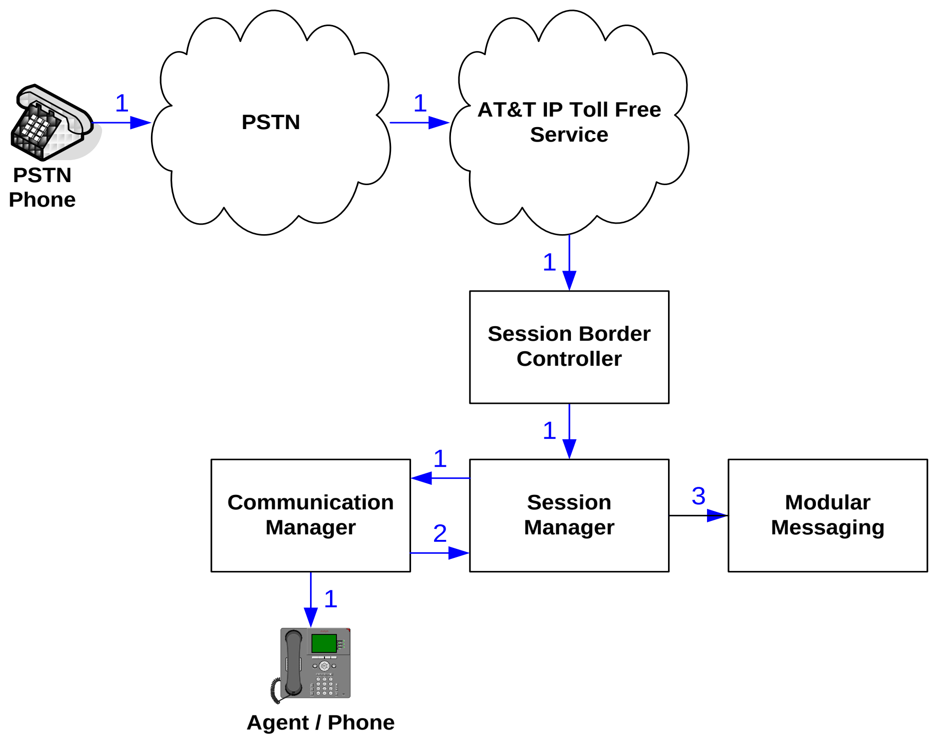
3.2.2. Coverage to Voicemail

The second call scenario illustrated in the figure below is an inbound call that is covered to voicemail. In this scenario, the voicemail system is a Modular Messaging system connected to Session Manager.

1. Same as call scenario in **Section 3.2.1**.
2. The agent or phone on Communication Manager does not answer the call, and the call covers to their voicemail which Communication Manager forwards to Session Manager.
3. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines it needs to route the call to Modular Messaging which answers the call and connects the caller to the called agent/phone voice mailbox. Note that the call continues to go through Communication Manager.



Inbound AT&T IP Toll Free Service Call to Agent/Phone Covered to Avaya Modular Messaging

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4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Equipment/Software	Release/Version
Avaya S8800 Server	Avaya Aura® System Manager 6.2 SP2 (6.2.0.0.15669-6.2.12.202) System Platform 6.2.1.0.9
Avaya S8800 Server	Avaya Aura® Session Manager 6.2 SP2 (6.2.2.0.62205)
Avaya S8720 Server	Avaya Aura® Communication Manager 5.2.1 SP13 (02.1.016.4-19880)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW15 FW054
TN799DP Control-LAN (C-LAN)	HW01 FW040
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW062
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW011
Avaya 9650 IP Telephone	H.323 Version S3.11b
Avaya 9620C IP Telephone	H.323 version S3.11b
Avaya 9641G IP Telephone	H.323 Version S6.2013U
Avaya one-X® Agent	2.5.00467.18
Avaya Digital Telephone 6408D+	
Avaya Analog phone	-
Fax device	Ventafax Home Version 6.1.59.144
Acme Packet Net-Net 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Toll Free service using AVPN/MIS-PNT transport service connection	VNI 23

Table 2: Equipment and Software Versions

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5. Configure Avaya Aura® Session Manager Release 6.2

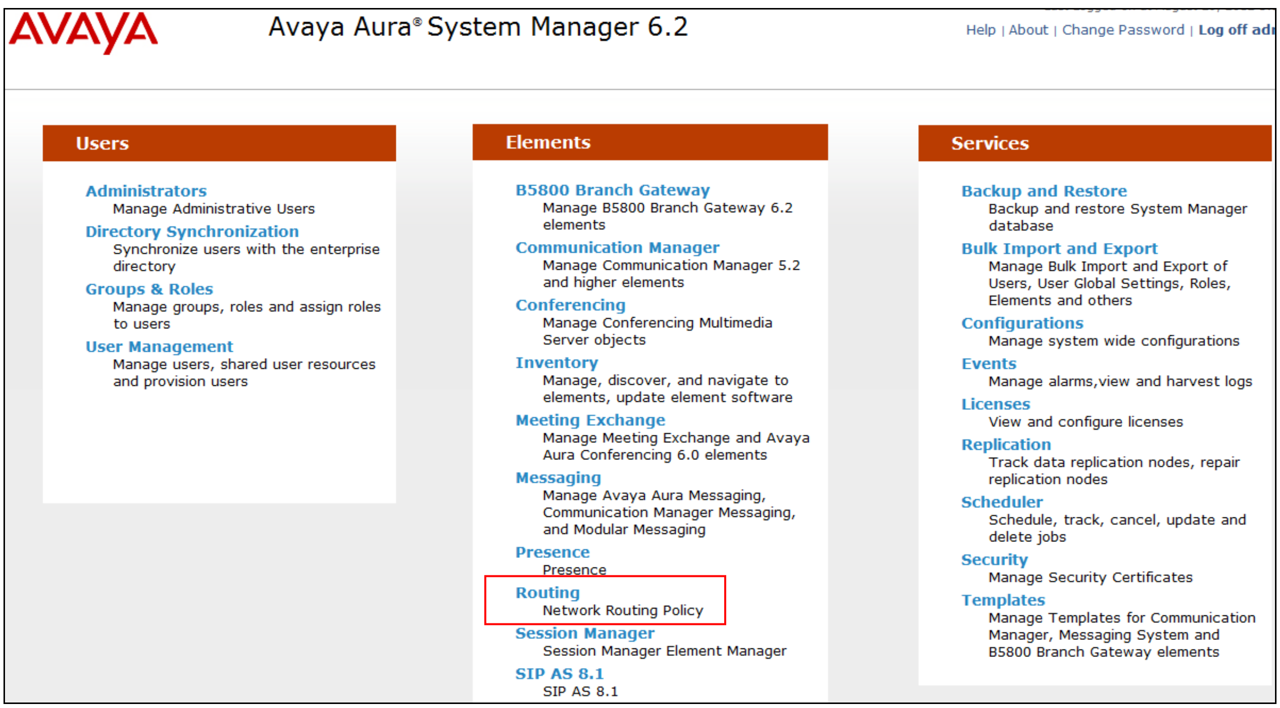
This section illustrates relevant aspects of the Session Manager configuration used in the verification of this compliance test solution for supporting AT&T IP Toll Free service.

Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Refer [**1**] to [**4**] for further details if necessary.

The following administration activities are described:

- Define SIP Domain
- Define Locations for routing purpose
- Configure the Adaptation Modules that are associated with various SIP Entities
- Define SIP Entities for Session Manager, Communication Manager, Acme Packet SBC, etc
- Define Entity Links between various SIP entities
- Define Routing Policies associated with Communication Manager, Acme Packet SBC, etc
- Define Dial Patterns which in conjunction with Routing Policies determine to which entity a call is routed to

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**http://<ip-address>**", where **<ip-address>** is the IP address of System Manager and logging in with the appropriate credentials. Once logged in, navigate to **Elements→Routing**.



System Manager Home Page

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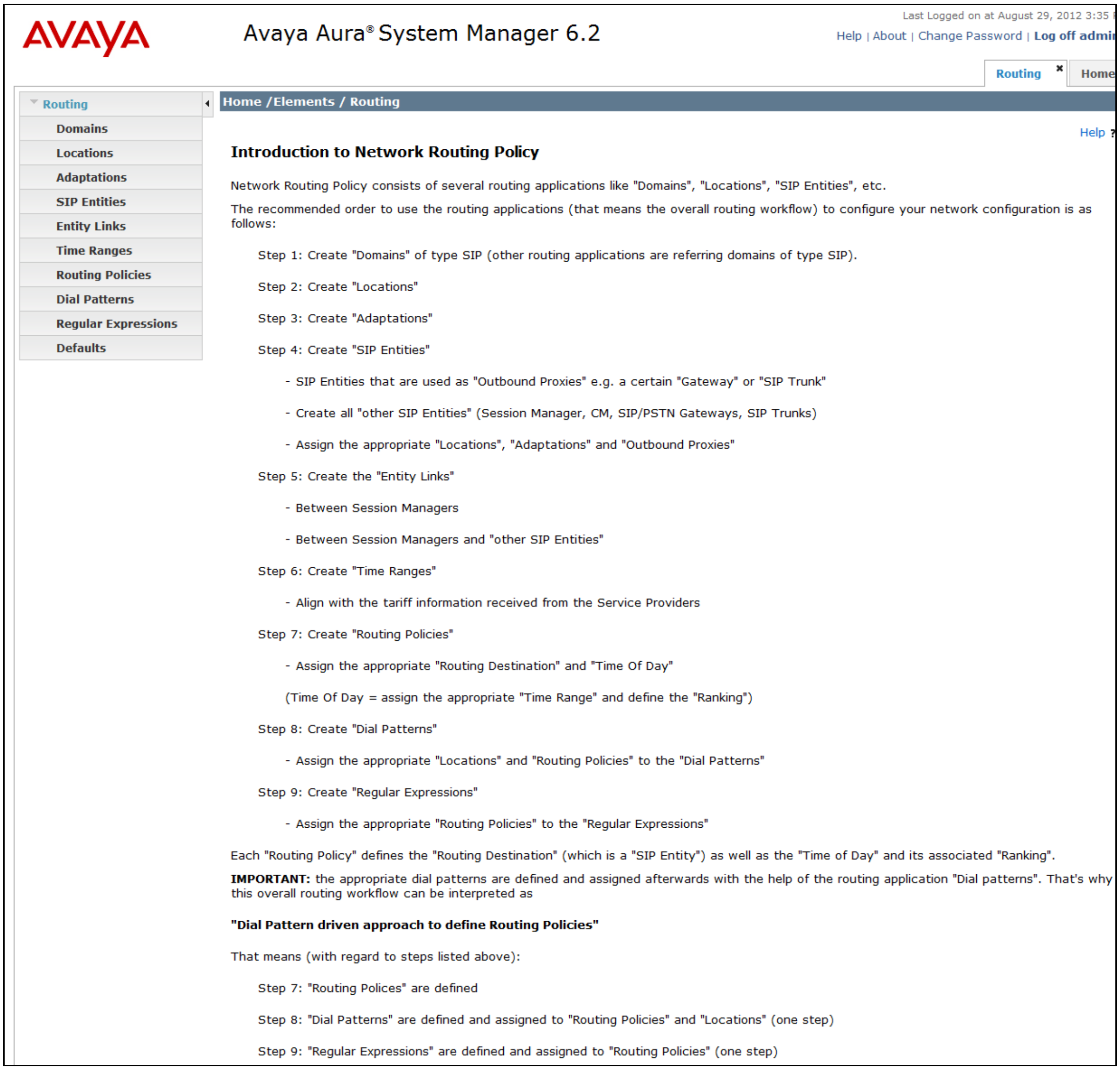
The screen below shows the various sub-headings with explanation of the left navigation menu that are referenced in this section.



Network Routing Policy Page

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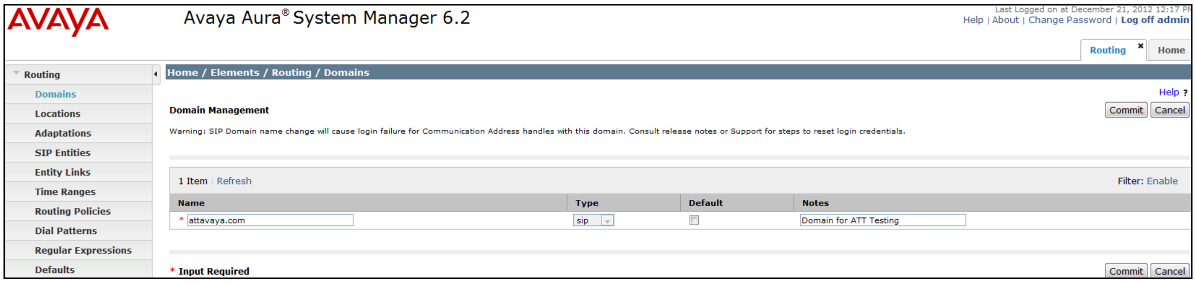
5.1. SIP Domain

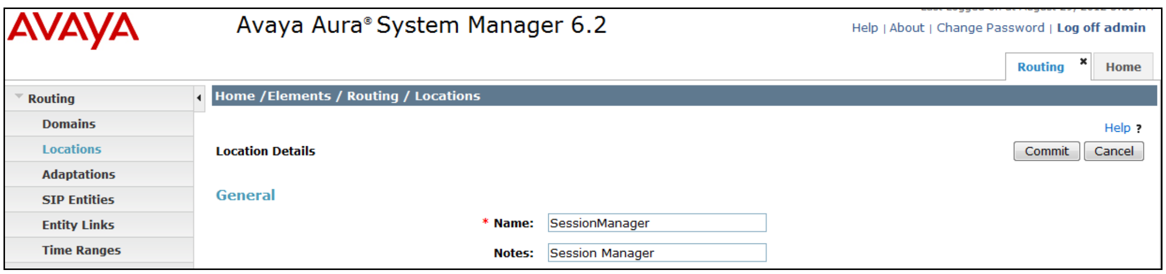
Navigate to **Routing→Domains** and click **New** (not shown). The following screen shows the domain used in this reference configuration.



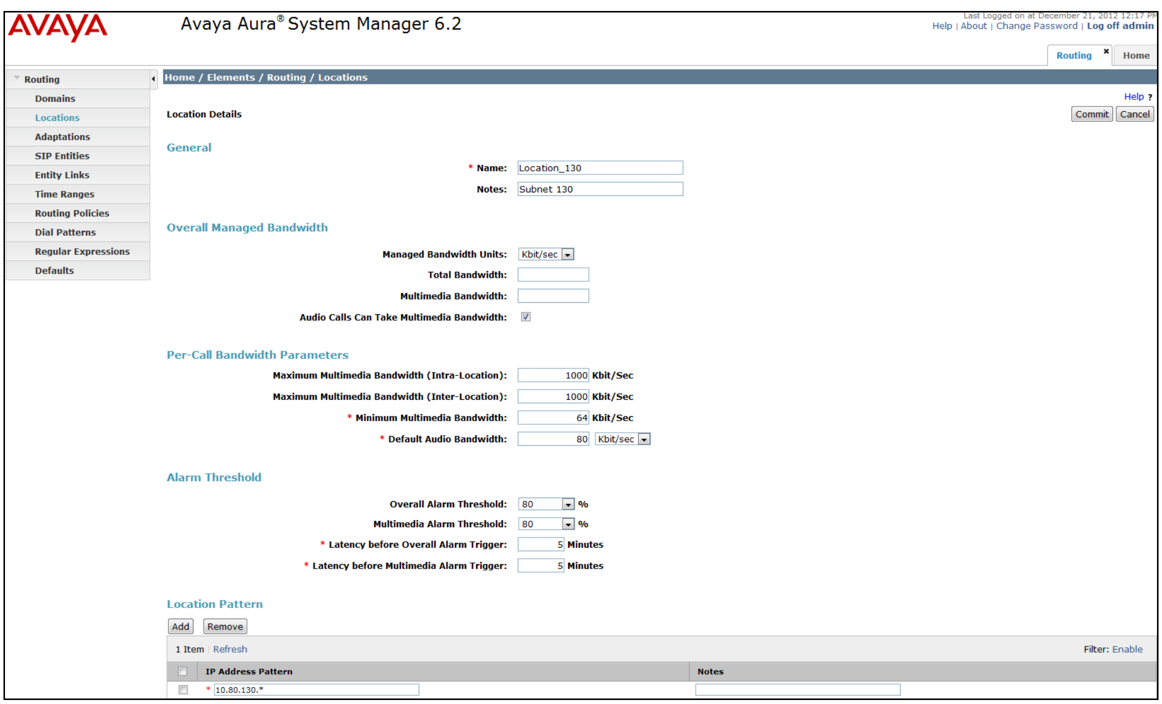
SIP Domains

5.2. Locations

Navigate to **Routing→Locations** and click **New** (not shown). The following screens show Location Details for various locations used in this AT&T IP Toll Free service testing.

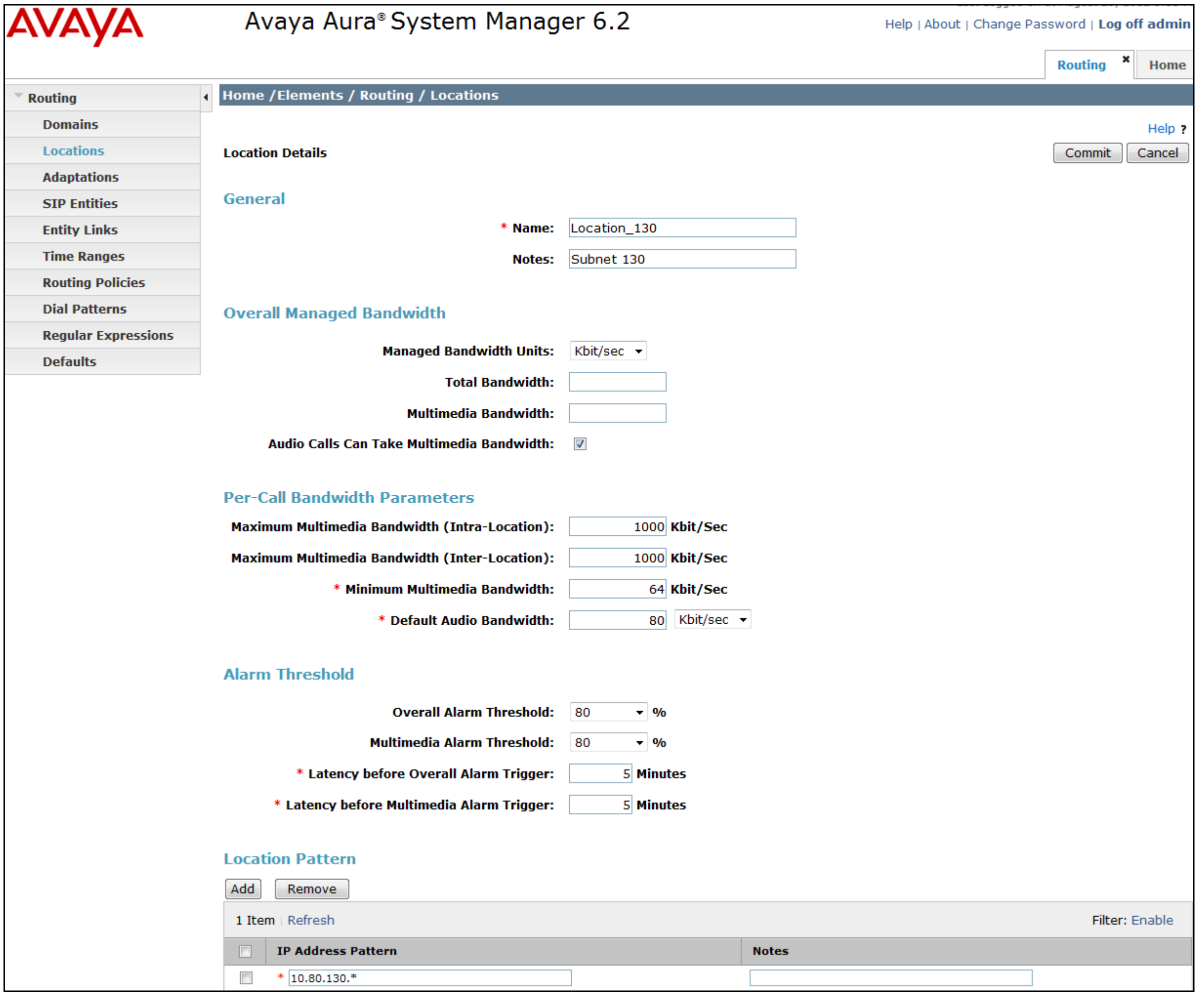


Session Manager Location Details



Acme Packet SBC Location Details

Location_130 shown below serves Communication Manager, Modular Messaging and other elements on the CPE.



Subnet 130 Location Details

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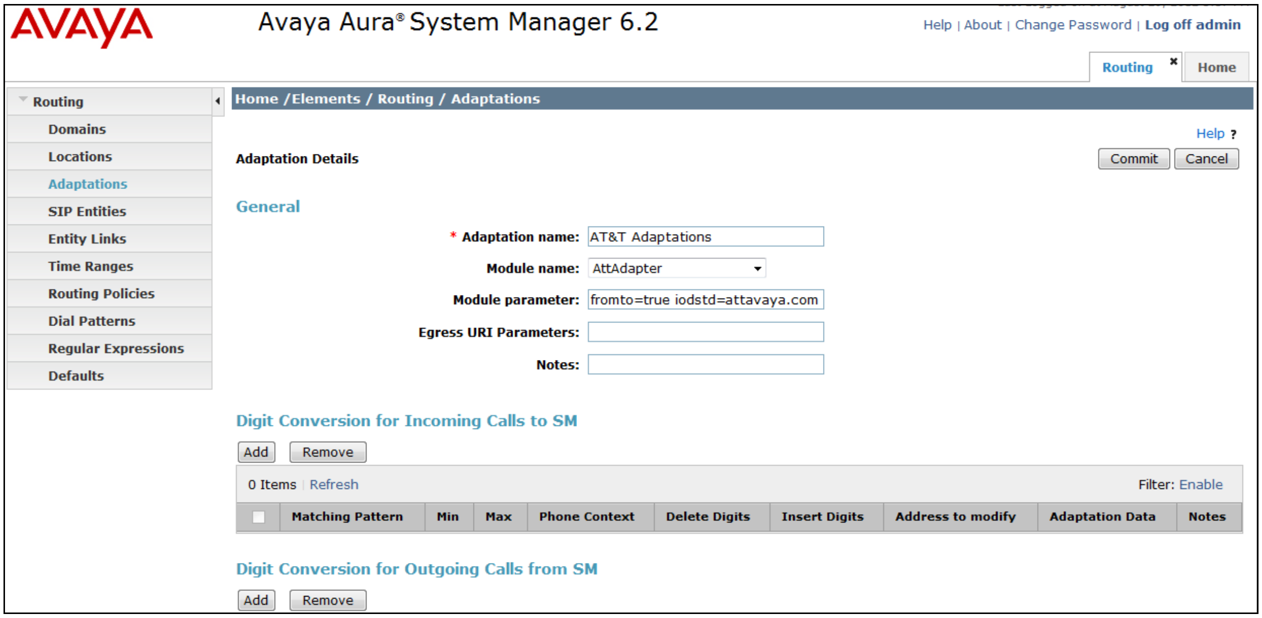
5.3. Configure Adaptations

The following screens display the adaptations used for calls between AT&T IP Toll Free Service entering Session Manager via Acme Packet SBC. This adaptation is applied to **AcmeSBCATT-5060** entity configured in **Section 5.4**.



Adaptation for Calls between AT&T IP Toll Free service and Session Manager

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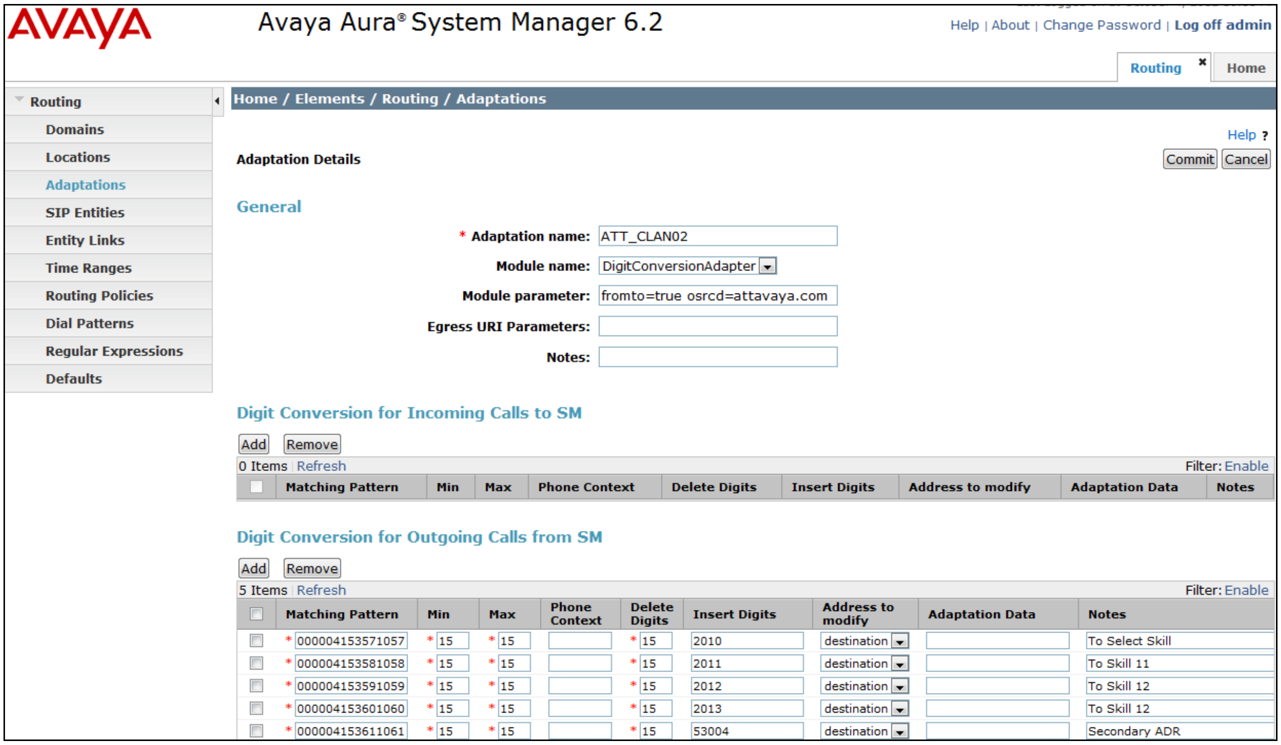
The following screens display the adaptations used for calls between Session Manager and Communication Manager. The following screen shows the adaptation used for **CM5.2CLAN1A02** configured in **Section 5.4**. In this case, digit conversion is done after routing decision has been made based upon the user part of the SIP URI.



Adaptation for calls between Session Manager and Communication Manager

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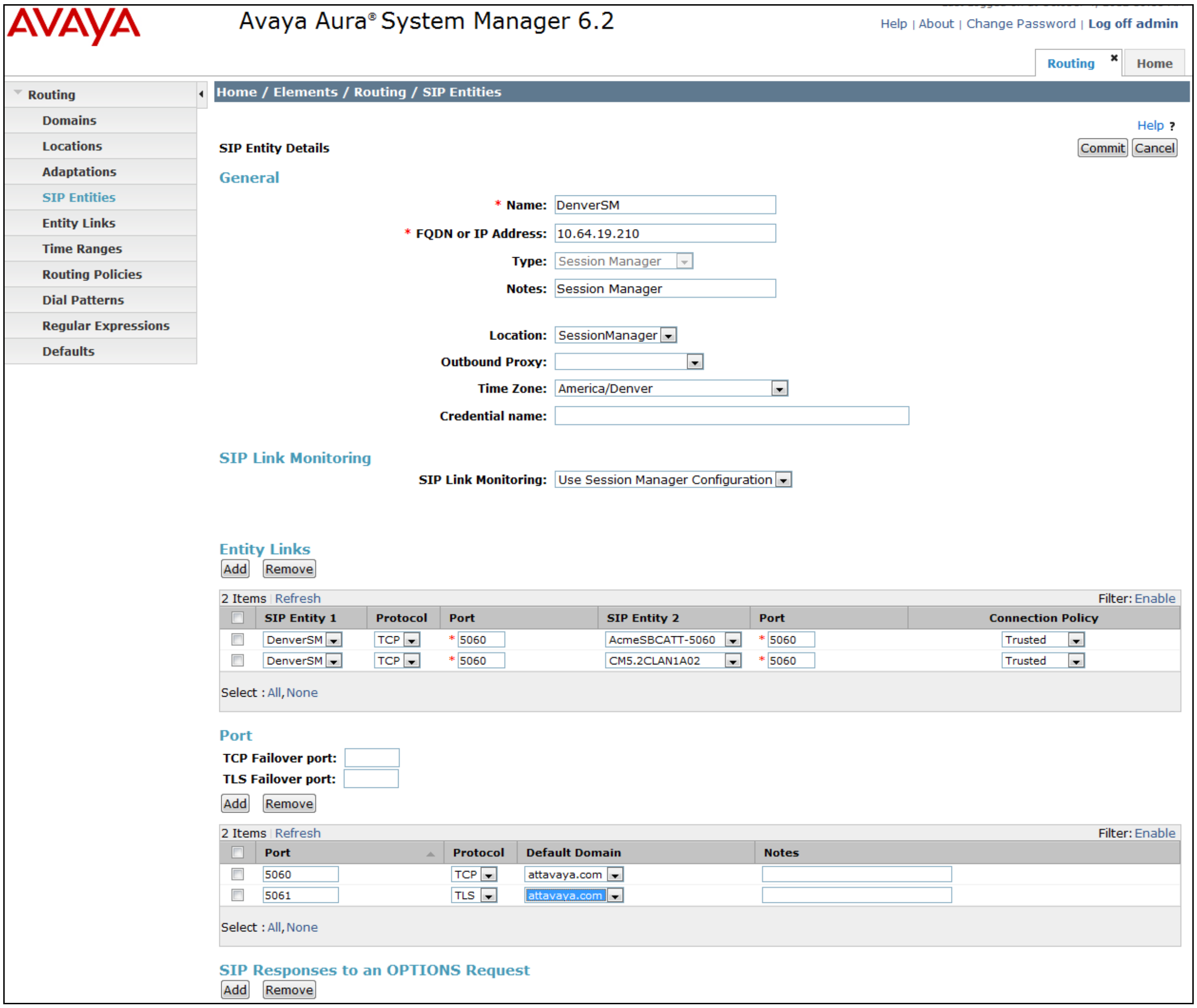
5.4. SIP Entities

The following screens show the entities along with Entity links configured for this reference configuration. See **Section 5.5** for Entity link configuration.

Note – In this reference configuration TCP is used as the transport protocol between Session Manager and all the SIP Entities including Communication Manager. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS to be used as transport protocol whenever possible.



Session Manager Entity

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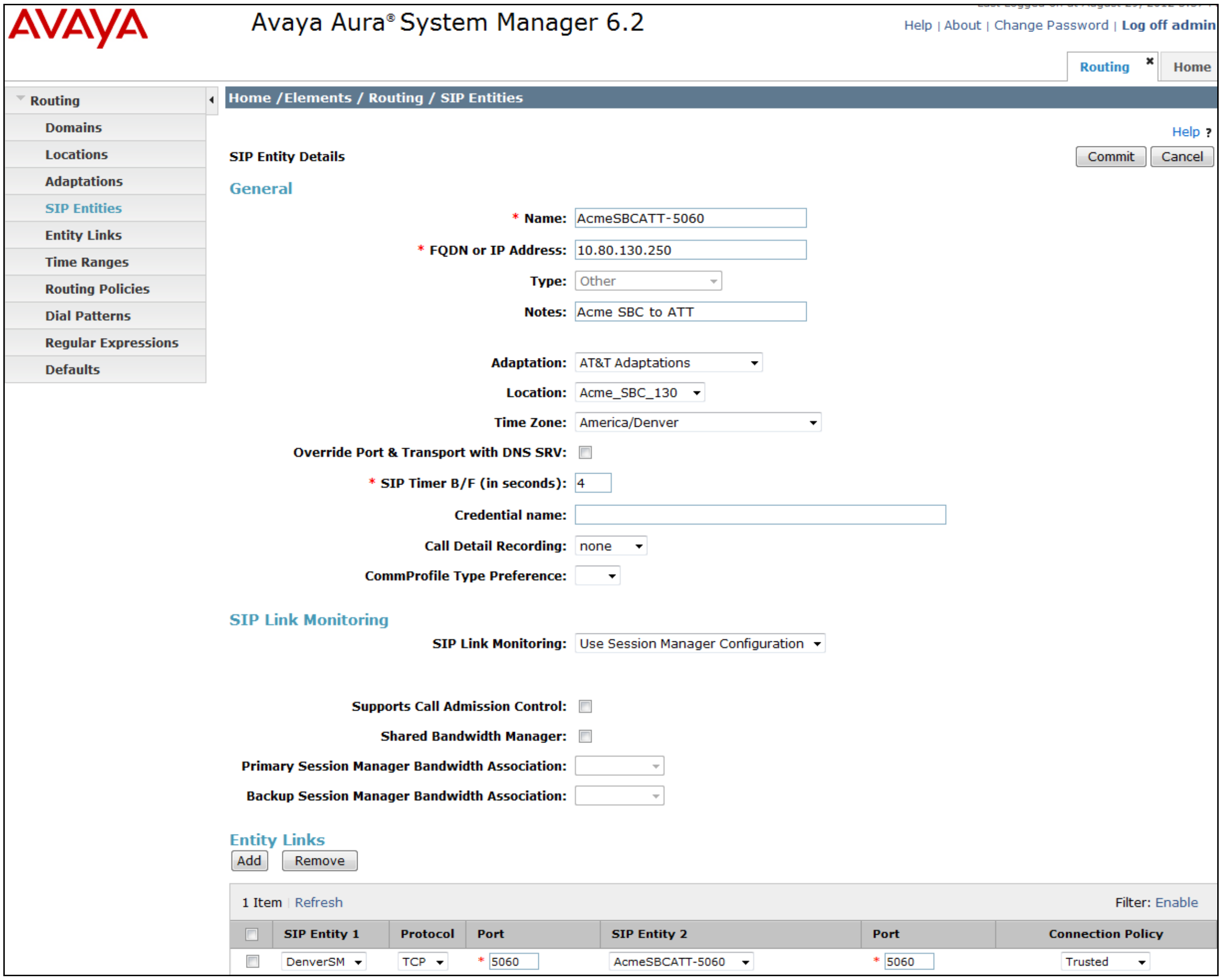

Acme Packet SBC Entity

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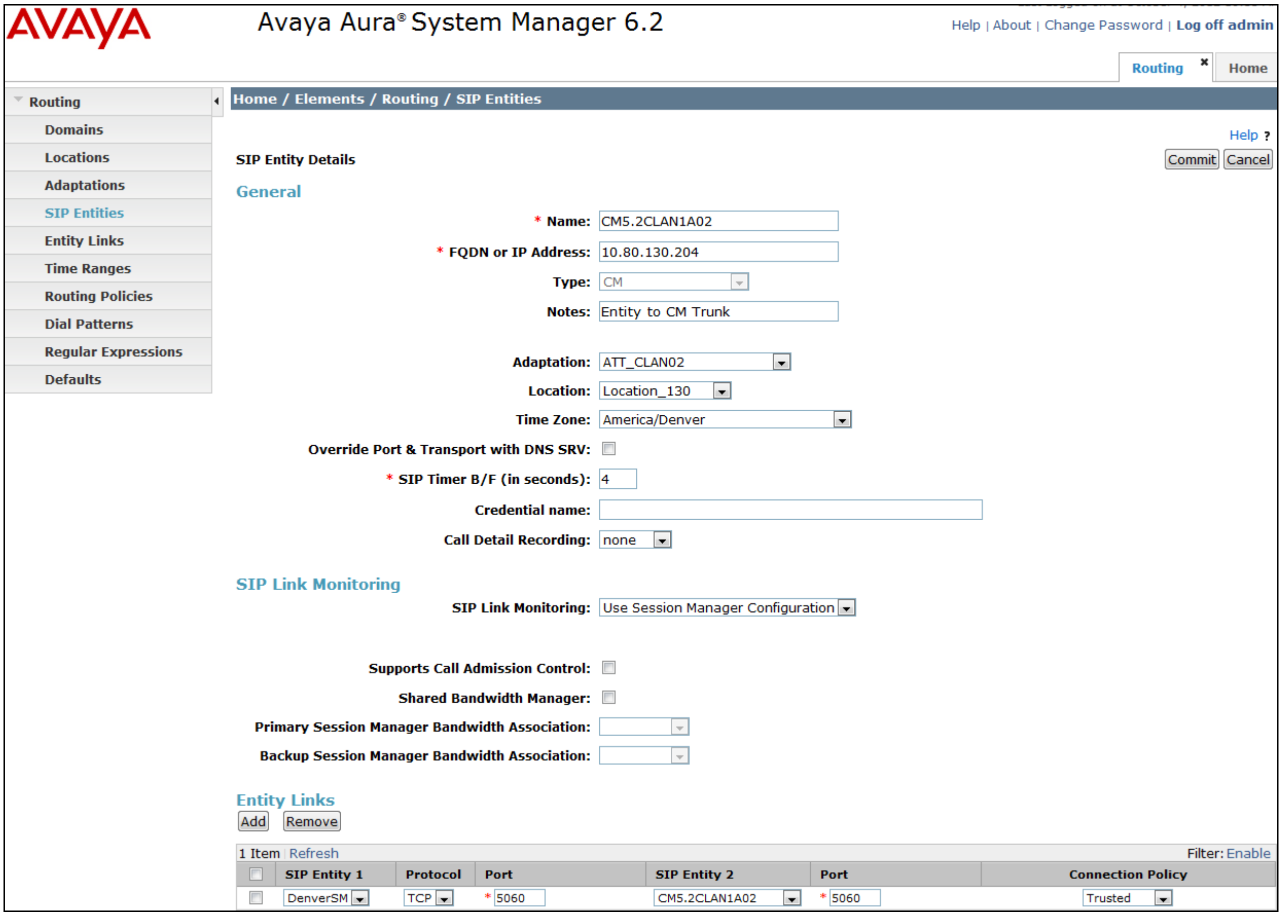

Communication Manager Entity

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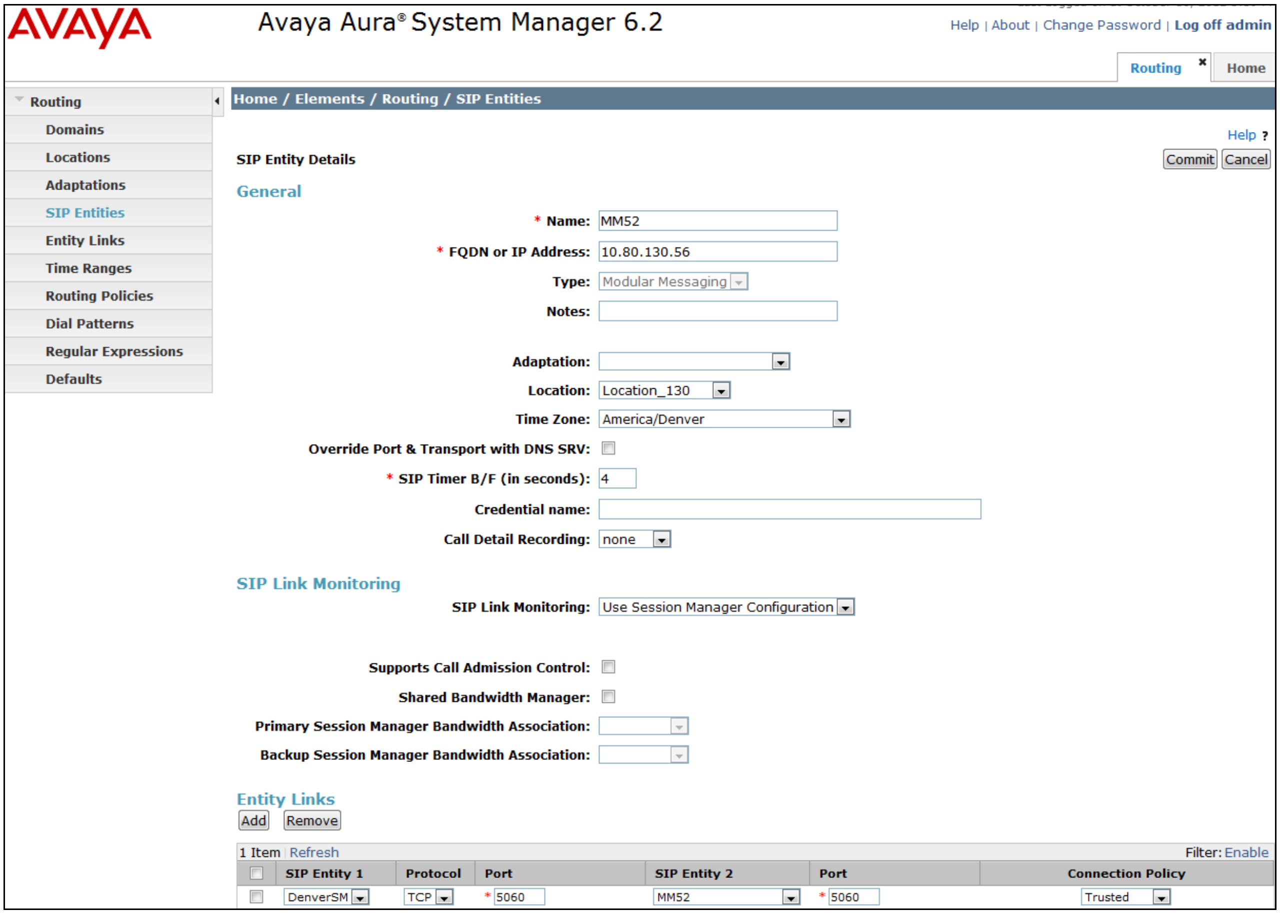
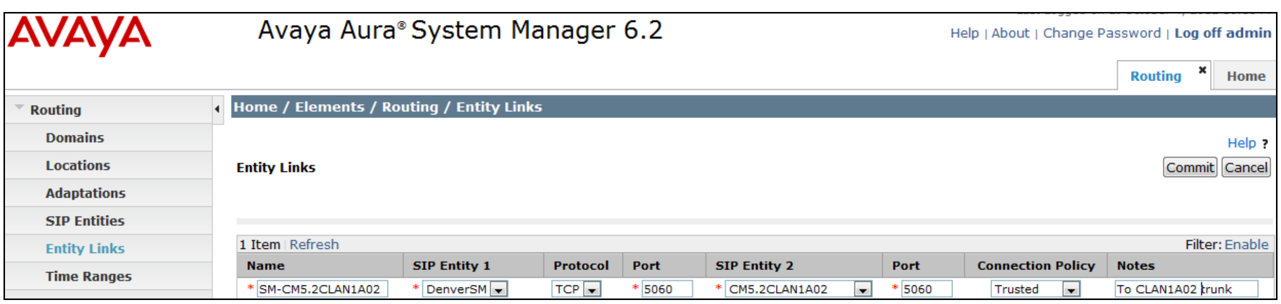


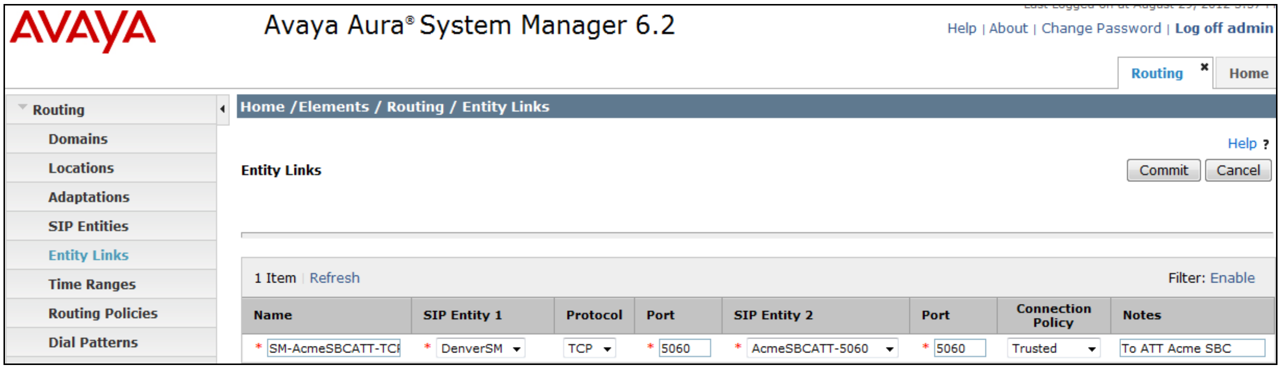
Modular Messaging Entity

5.5. Entity Links

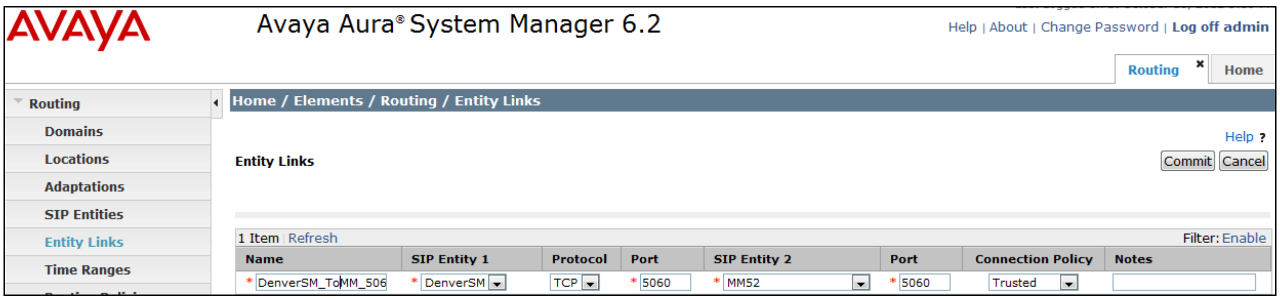
The following screens show the entity links configured for this reference configuration.



Entity link between Session Manager and Communication Manager



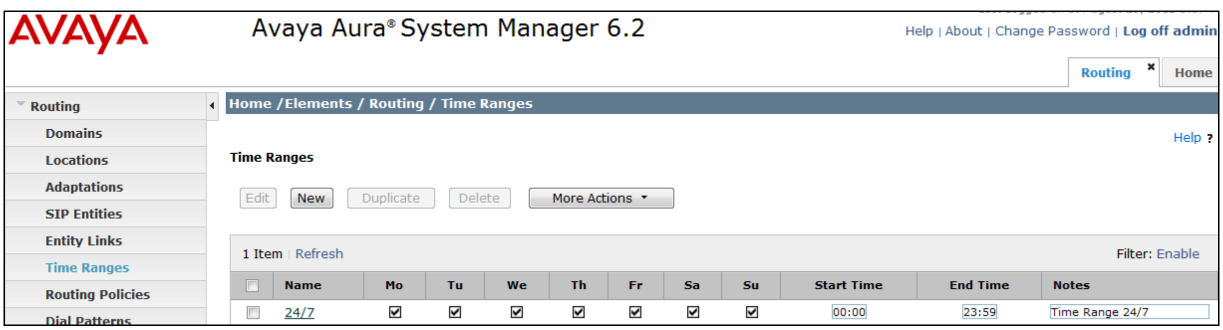
Entity link between Session Manager and Acme Packet SBC



Entity link between Session Manager and Modular Messaging

5.6. Time Ranges

The following screen shows the time range used for AT&T IP Toll Free service testing.



Time Range

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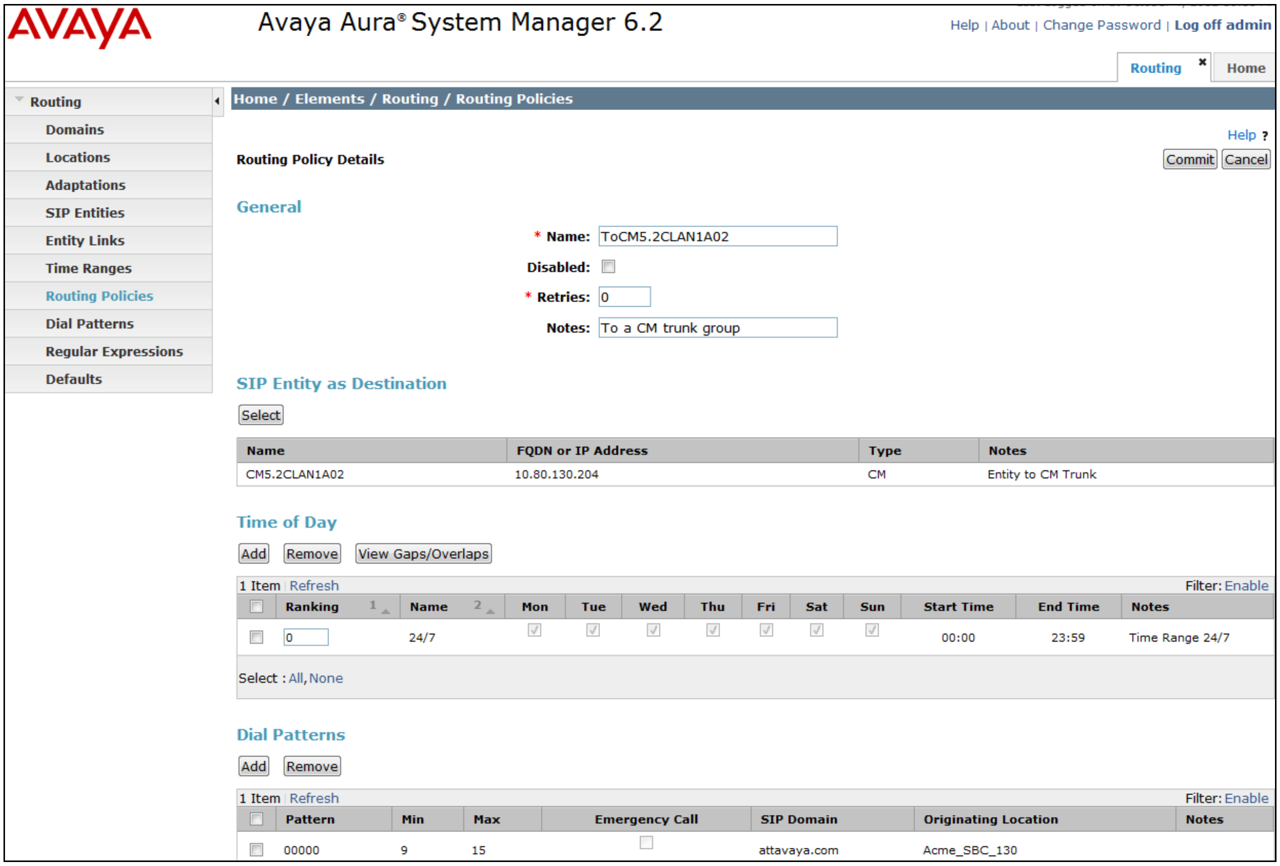
5.7. Routing Policies

The following screens show routing policy along with dial patterns defined for AT&T IP Toll Free service. See **Section 5.8** for dial pattern configuration.



Routing Policy to Communication Manager

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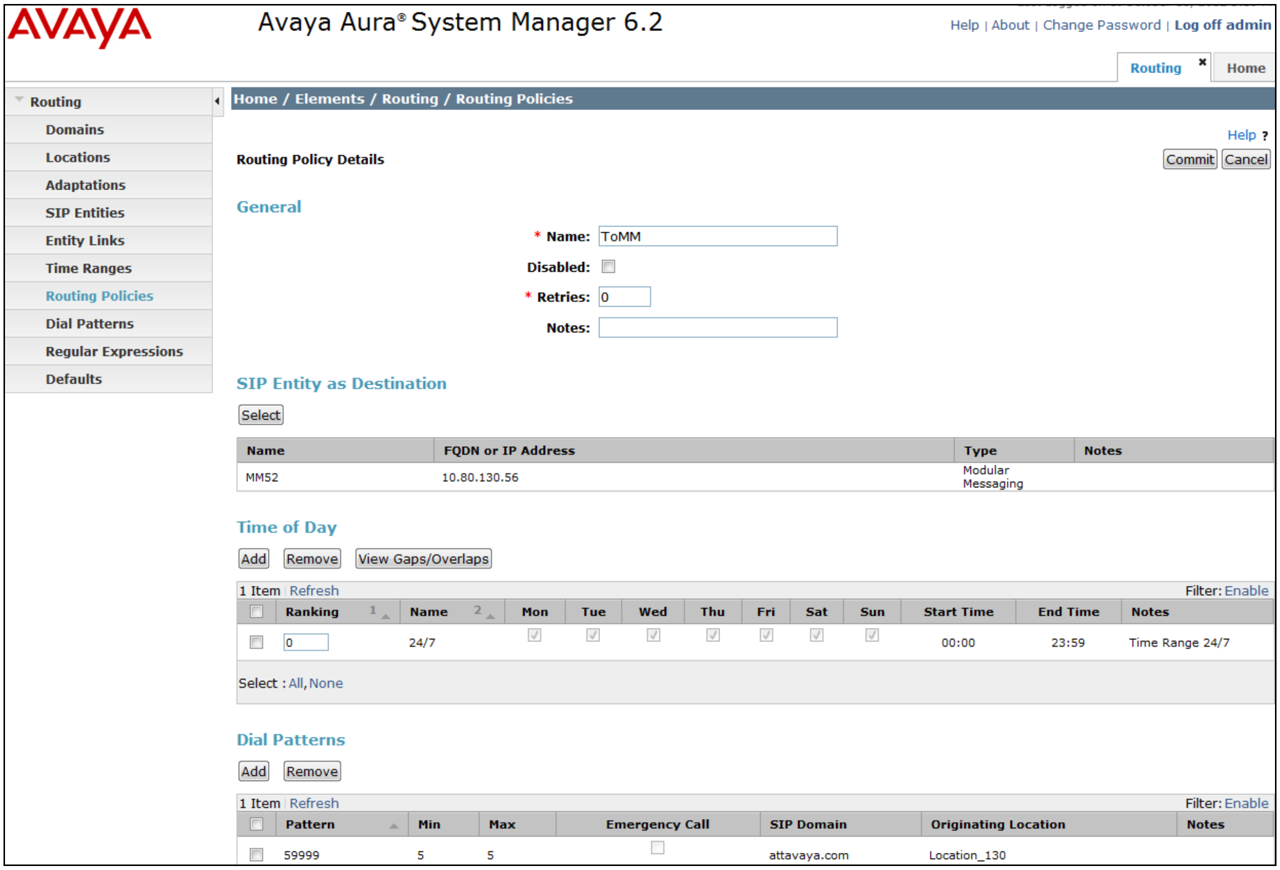

Routing Policy to Modular Messaging

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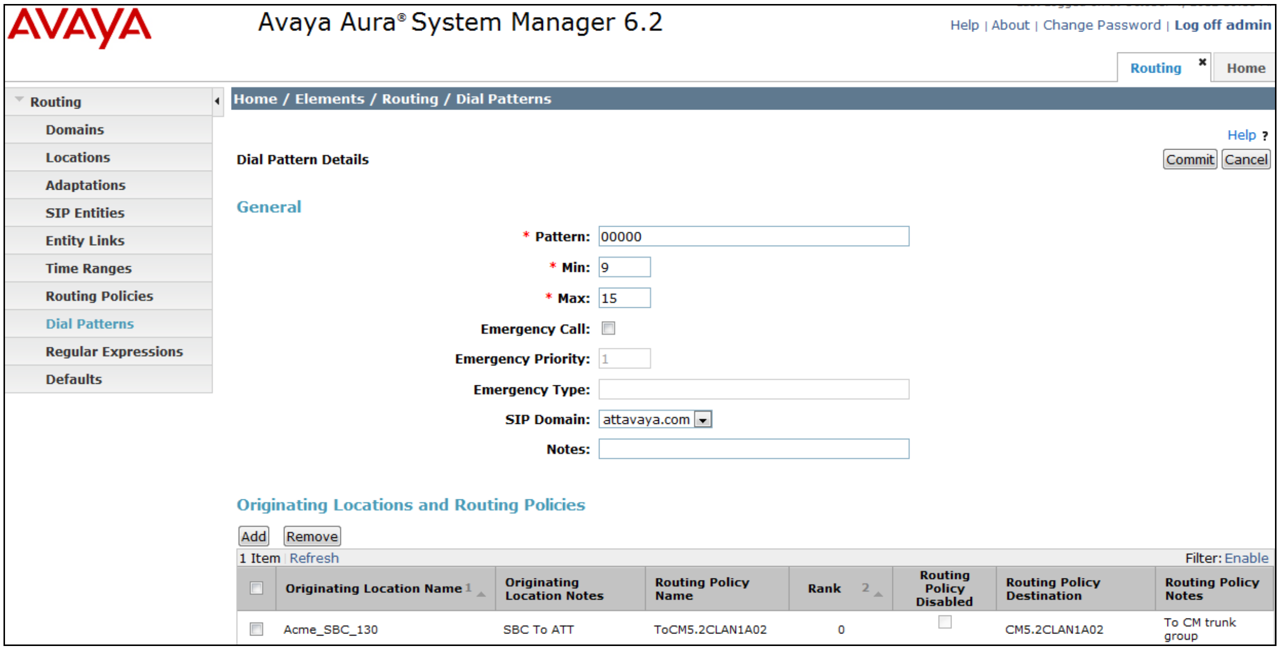
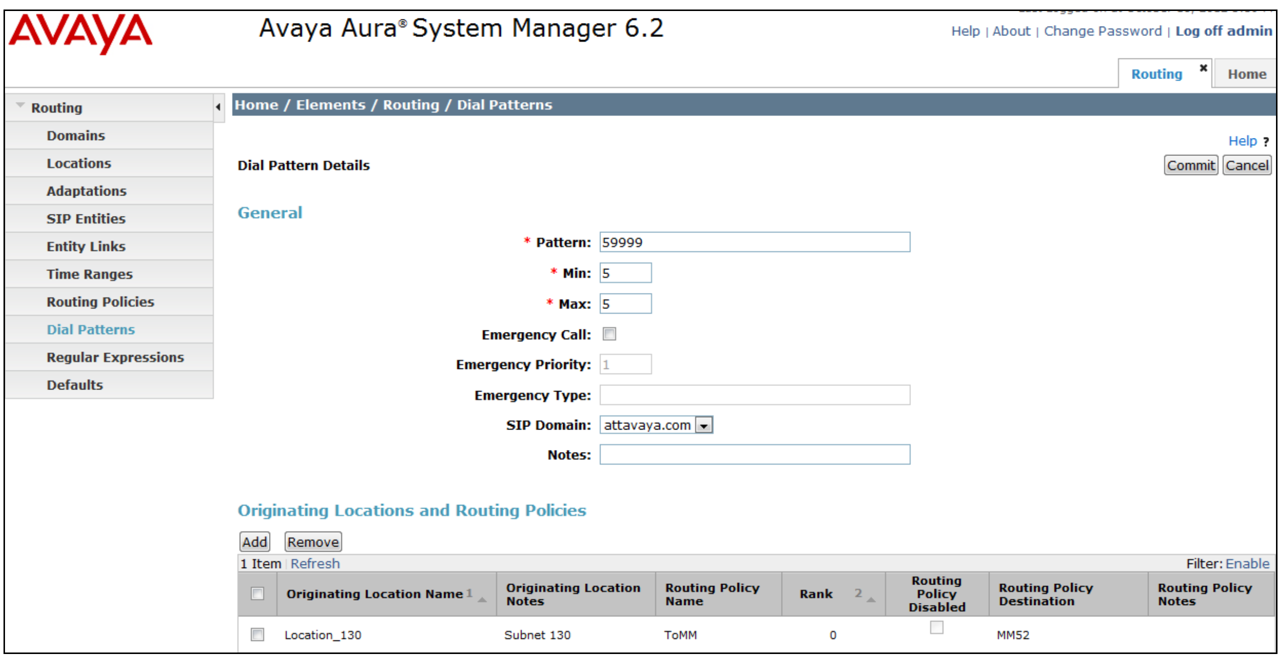
5.8. Dial Patterns

The following screens show dial patterns configured in this reference configuration.



Dial Pattern for Inbound Calls to Communication Manager



Dial Pattern for Covered Calls to Modular Messaging

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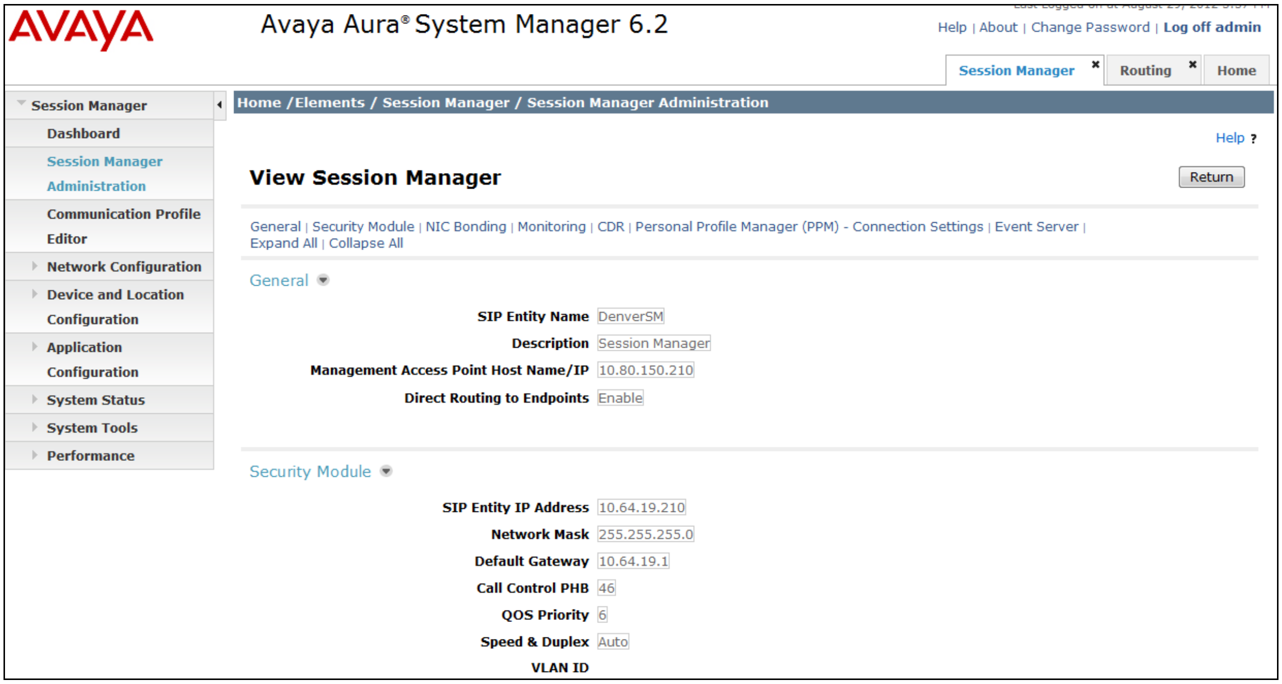
5.9. Session Manager Administration

Navigate to **Home→Elements→Session Manager→Session Manager Administration** and in **Session Manager Instances** select the appropriate Session Manager already configured. The following screen shows the Session Manager instance **DenverSM** used in this reference configuration.



View Session Manager (DenverSM)

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6. Configure Avaya Aura® Communication Manager 5.2.1

In this reference configuration Communication Manager 5.2.1 is provisioned in an Access Element configuration, supporting H.323 and Digital endpoints (SIP endpoints are not supported in this configuration). This section describes the administration steps for Communication Manager in support of the AT&T IP Toll Free service features listed in **Section 2**. These steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration, including stations, C-LAN, Media Processor, and announcement boards, etc., has already been performed. Consult [**5**] and [**6**] for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are specifically applicable to this Application Notes. Other parameter values may or may not match based on local configurations.

6.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On **Page 2** of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks (e.g. 5000).

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

IP PORT CAPACITIES                                           USED
                    Maximum Administered H.323 Trunks: 8000  0
          Maximum Concurrently Registered IP Stations: 18000 4
            Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
             Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                 Maximum Video Capable H.323 Stations: 0     0
                  Maximum Video Capable IP Softphones: 0     0
                      Maximum Administered SIP Trunks: 5000  250
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                            Maximum TN2501 VAL Boards: 10    1
                    Maximum Media Gateway VAL Sources: 0     0
          Maximum TN2602 Boards with 80 VoIP Channels: 128   0
         Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

        (NOTE: You must logoff & login to effect the permission changes.)
```


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2. On **Page 4** of the **system-parameters customer-options** and verify that the **IP Trunks** field in 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? y                         ISDN Feature Plus? y
                  Enhanced EC500? y      ISDN/SIP Network Call Redirection? n
    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? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

6.2. Dial Plan

The dial plan defines how the digit string will be used locally by Communication Manager. Note that the values shown below are examples used in the reference configuration. Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings:

- 3-digit Dial Access Codes (indicated with a **Call Type** of **dac**) beginning with the digit **1** (e.g. Trunk Access Codes, TACs, defined for trunk groups in this reference configuration conform to this format).
- 5-digit Extensions with a **Call Type** of **ext** beginning with the digits **5xxxx** (e.g. Local extensions for Communication Manager stations, agents, and Vector Directory Numbers, VDNs, in this reference configuration conform to this format).
- 1-digit Facilities Access Code (indicated with a **Call Type** of **fac**) (e.g. 8/**9** access code for AAR/ARS dialing).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1
    Dialed    Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String    Length Type     String   Length Type     String   Length Type
   1            3   dac
   5            5   ext
   8            1   fac
   9            1   fac
```


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6.3. IP Node Names

Following screen shows the node names used for AT&T IP Toll Free service provisioning.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
Gateway001          10.80.130.1
CLAN-1A02           10.80.130.204
SM62                10.64.19.210
```

6.4. IP Codec Parameters

Following screen shows the codec set used in this reference configuration.

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

                          IP Codec Set
   Codec Set: 2

   Audio        Silence      Frames   Packet
   Codec        Suppression  Per Pkt  Size(ms)
1: G.729B            n          3        30
2: G.729A            n          3        30
3: G.711MU           n          3        30
```

On **Page 2** of the **ip-codec-set** form, set **Mode - Fax** to **t.38-standard**.

```
change ip-codec-set 2                                           Page   2 of 2
                          IP Codec Set
                              Allow Direct-IP Multimedia? n
                   Mode                 Redundancy
    FAX            t.38-standard         0
    Modem          off                   0
    TDD/TTY        off                   0
    Clear-channel  n                     0
```


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6.5. IP Network Regions

Network Regions are used to group various Communication Manager Resources such as codecs, UDP port ranges, and inter-region communication. In this reference configuration only one network region was configured for all elements. Additional network regions can be defined if required. Enter **ip-network-region x**, where **x** is the number of an unused IP network region and configure as follows:

- **Name** - Enter any descriptive string.
- **Codec Set** – Set to Codec set configure in **Section 6.4**.
- **Intra and Inter IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible within the same region.
- **UDP Port Min**: - Set to **16384** (Required for AT&T IP Toll Free service)
- **UDP Port Max**: - Set to **32767** (Required for AT&T IP Toll Free service)

```
change ip-network-region 2                                  Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: attavaya.com
    Name: ATT Calls
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? y
   UDP Port Max: 32767
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
```

On **Page 3** of the form, verify that region 2 is using codec set 2 as specified on **Page 1** (this field is automatically populated). If additional regions are configured, this form can dictate what codec set to be used for communication with elements belonging to different network regions.

```
change ip-network-region 2                                   Page   3 of  19

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


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6.6. SIP Trunks

Following steps are used to configure SIP Trunk on Communication Manager:

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group as shown in the following screen.

```
add signaling-group  2                                            Page   1 of   1
                                SIGNALING GROUP

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

 Near-end Node Name: CLAN_1A02              Far-end Node Name: SM62
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 2
Far-end Domain: attavaya.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
```

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **2**).

```
add trunk-group 2                                                 Page   1 of  21
                               TRUNK GROUP

Group Number: 2                    Group Type: sip          CDR Reports: y
  Group Name: ATT                         COR: 1        TN: 1        TAC: 102
   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: 10
```

3. On **Page 2** of the **trunk-group** form set the **Preferred Minimum Session Refresh Interval (sec)** field to **900.** This entry will actually cause a value of 1800 to be generated in the SIP header.

```
add trunk-group 2                                                 Page   2 of  21
      Group Type: sip
TRUNK PARAMETERS
     Unicode Name: auto
                                        Redirect On OPTIM Failure: 5000
            SCCAN? n                              Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900
 Disconnect Supervision - In? y  Out? y
```


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4. On **Page 3** of the **trunk-group** form set **Numbering Format** field to **public**

```
add trunk-group 2                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y
                     Numbering Format: public
                                               UUI Treatment: service-provider
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
 Show ANSWERED BY on Display? y
```

5. On **Page 4** of the **trunk-group** form:

- Set **Support Request History?** field to **y**.
- Set **Telephone Event Payload Type** field to the RTP payload type required by the AT&T IP Toll Free service (e.g. **100**).

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS
                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 100
```

6.7. Public Unknown Numbering

This form is used to populate the **history-info** and user part of the**To** header with the Communication Manager extension. In this reference configuration, all extension of length **5** and prefixed by **5** entering Communication Manager on trunk group **2** will populate the required SIP headers with the correct telephone extension.

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext              Trk       CPN          CPN
Len Code             Grp(s)    Prefix       Len
 5                   2                      5         Total Administered: 3
 5  50002            3         7323680194   10           Maximum Entries: 9999
```


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6.8. Alternate Automated Routing (AAR) Table

The AAR table is selected based on the caller dialing the AAR access code (e.g. "**8**") as defined in **Section 6.2**. The access code is removed and the AAR table matches the remaining dialed digits and sends them to the designated route pattern configured in **Section 6.9**. Configure as follows:

- **Dialed String** – Set to Modular Messaging pilot number **59999**.
- **Min** and **Max** - Set to **5**, the minimum and maximum size the dialed string will assume.
- **Route Pattern** – Set to **2** as configured in **Section 6.9**.
- **Call Type** – Set to **unku**.

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

      Dialed           Total      Route     Call   Node   ANI
      String          Min  Max  Pattern     Type   Num    Reqd
      59999           5    5      2         unku          n
```

6.9. Route Pattern

This form defines the SIP trunk to be used based on the route pattern selected by the AAR table for local calls (see **Sections 5.4**).

- **Grp No** – Set to **2** i.e. the trunk group configured for Local Access.
- **FRL** – Set to **0** (zero).

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

  BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
  0 1 2 M 4 W     Request                                Dgts Format
                                                      Subaddress
 1: y y y y y n  n            rest                                      none
 2: y y y y y n  n            rest                                      none
 3: y y y y y n  n            rest                                      none
 4: y y y y y n  n            rest                                      none
 5: y y y y y n  n            rest                                      none
 6: y y y y y n  n            rest                                      none
```


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6.10. Optional Features

6.10.1. Call Center Provisioning

For provisioning the call center functionality, verify that the call center parameters are enabled as shown below. Verify that an agent login id is created with an appropriate skill. Verify the skill (hunt group) for that agent is in place. Make sure that a VDN as per the dial plan is in place along with the vector which lists the steps to be executed when an inbound call is received from AT&T IP Toll Free service.

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. Consult [**3**], [**4**], [**5**], and [**6**] for further details, if necessary. The configuration steps that follow are provided for reference purposes only.

```
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)? y              Service Level Maximizer? n
         BCMS/VuStats Service Level? y            Service Observing (Basic)? n
  BSR Local Treatment for IP & ISDN? n    Service Observing (Remote/By FAC)? n
                  Business Advocate? n           Service Observing (VDNs)? n
                    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? y            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)? n                   Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n     Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n               Vectoring (Variables)? n

        (NOTE: You must logoff & login to effect the permission changes.)
```

Call Center Optional Features Form

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In the reference configuration below, an inbound call from AT&I IP Toll Free service is handled using the **VDN 2010** which routes the call to **Vector 10** and based upon the digits specified by the caller, the call is directed to an appropriate skill. **Skill 11** is shown for reference purposes. Additional skills can be similarly added.

```
display agent-loginID 53001                                Page   1 of   2
                                 AGENT LOGINID

                Login ID: 6665611                                 AAS? n
                    Name: Agent1                                AUDIX? n
                      TN: 1                             LWC Reception: spe
                     COR: 1                    LWC Log External Calls? n
           Coverage Path: 2                  AUDIX Name for Messaging:
           Security Code:
                                         LoginID for ISDN/SIP Display? n
                                                             Password:
                                               Password (enter again):
                                                          Auto Answer:
station
                                                   MIA Across Skills: system
                                           ACW Agent Considered Idle: system
                                           Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :

     WARNING:  Agent must log in again before changes take effect
```

Agent Form – Page 1

```
display agent-loginID 53001                                  Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
Call Handling Preference: skill-level                Local Call Preference? n

    SN   RL SL          SN   RL SL          SN   RL SL          SN   RL SL
 1: 11      1        16:                 31:                 46:
 2:                  17:                 32:                 47:
 3:                  18:                 33:                 48:
```

Agent Form – Page 2

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```
display hunt-group 11                                           Page   1 of   3
                                  HUNT GROUP

            Group Number: 11                                    ACD? y
              Group Name: Skill-11                            Queue? y
         Group Extension: 666-5711                           Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                         MM Early Answer? n
           Security Code:                    Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

Skill (Hunt Group) Form – Page 1

```
display hunt-group 11                                          Page   2 of   3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none
     Supervisor Extension:

      Controlling Adjunct: none

Interruptible Aux Threshold: none
                                 Redirect on No Answer (rings):
                                              Redirect to VDN:
                   Forced Entry of Stroke Counts or Call Work Codes? n
```

Skill (Hunt Group) Form – Page 2

```
display vdn 2010                                            Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 666-5310
                                  Name: To SelectSkill
                           Destination: Vector Number        10

                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN#: 1
                              Measured: none

        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
* Follows VDN override rules
```

VDN (Vector Directory Number) Form

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```
display vector 10                                               Page   1 of   6
                                  CALL VECTOR
    Number: 10                 Name: RouteToSkill
                                              Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    2   secs hearing ringback
02 collect      1    digits after announcement 33002    for none
03 goto vector  11   @step 2 if digits           =      1
04 goto vector  12   @step 2 if digits           =      2
05 goto vector  13   @step 2 if digits           =      3
06
```

Vector (RouteToSkill) Form

```
display vector 11                                               Page   1 of   6
                                  CALL VECTOR
    Number: 11                 Name: Skill 11
                                              Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    2   secs hearing ringback
02 announcement 33003
03 queue-to     skill 11   pri m
04 announcement 33006
05 goto step    3             if unconditionally
06
```

Vector (Skill 11) Form

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6.10.2. Modular Messaging Coverage Path and Hunt Group

Hunt group 1 is used in the reference configuration to verify Modular Messaging coverage functionality. This hunt group is defined with the 5 digit Modular Messaging pilot number **59999**. The hunt group is associated with call **coverage path 1** in form below and the coverage path is assigned to a station (e.g., **50001**). Communication Manager will use the AAR access code **8** (defined in **Section 6.8**) to dial Modular Messaging (e.g. **859999**) as shown on **hunt-group** form below.

```
display coverage path 1                                          Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
       Don't Answer?            y              y        Number of Rings: 4
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n
COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1             Rng: 4  Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

Coverage Path Form

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

            Group Number: 1                                ACD? n
              Group Name: MM                             Queue? n
         Group Extension: 59999                         Vector? n
              Group Type: ucd-mia                Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Hunt Group Form – Page 1

```
display hunt-group 1                                             Page   2 of  60
                                  HUNT GROUP
                      Message Center: sip-adjunct
     Voice Mail Number        Voice Mail Handle        Routing Digits
                                                 (e.g., AAR/ARS Access Code)
     59999                    59999                    8
```

Hunt Group Form – Page 2

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```
display station 50001                                          Page   1 of   5
                                     STATION
Extension: 50001                         Lock Messages? n               BCC: 0
     Type: 9620                          Security Code: 123456           TN: 1
     Port: S00000                      Coverage Path 1: 1               COR: 1
     Name: H323-96XX-50001             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: 6665011
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                   Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
                                                          IP Video? n
                             Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y
```

Station Form

6.11. Saving Translations

To save all Communication Manager provisioning changes, enter the command **save translations.**

7. Avaya Modular Messaging

In this reference configuration, Avaya Modular Messaging is used to verify DTMF, Message Wait Indicator (MWI), as well as basic call coverage functionality. The Avaya Modular Messaging used in the reference configuration is provisioned for Multi-Site mode. Multi-Site mode allows Avaya Modular Messaging to serve subscribers in multiple locations. The administration for Modular Messaging is beyond the scope of these Application Notes. Consult [**7**], [**8**], [**9**], and [**10**] for further details.

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8. Configure Acme Packet Session Border Controller (SBC)

These Application Notes assume that basic Acme Packet SBC administration has already been performed. The Acme Packet SBC configuration used in the reference configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Use **putty** or similar tool to access Acme Packet SBC for configuration. Consult with Acme Packet Support [**11**] for further details and explanations on the configuration below.

ANNOTATION: The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Session Manager, Communication Manager, etc., reside to the AT&T IP services. The Session Agent Groups (**SAG**) defined here, and further down, provisioned under the session-groups **SP-PROXY** and **ENTERPRISE.** Note: Although **Enterprise** policy is not used for AT&T IP Toll Free service but is left in there in case the customer is using AT&T IP Flexible Reach service.

```
local-policy
        from-address
                                *
        to-address
                                *
        source-realm
                                Enterprise
        description
        activate-time           N/A
        deactivate-time         N/A
        state                   enabled
        policy-priority         none
        policy-attribute
                next-hop                sag:SP_PROXY
                realm                   ATT
                action                  none
                terminate-recursion     disabled
                carrier
                start-time              0000
                end-time                2400
                days-of-week            U-S
                cost                    0
                app-protocol
                state                   enabled
                methods
                media-profiles
                lookup                  single
                next-key
                eloc-str-lkup           disabled
                eloc-str-match
```


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ANNOTATION: The local policy below governs the routing of SIP messages from the AT&T IP services to Session Manager.

local-policy
  from-address
    *****
  to-address
    *****
  source-realm
    ATT
  description
  activate-time N/A
  deactivate-time N/A
  state enabled
  policy-priority none
  policy-attribute
    next-hop **10.64.19.210**
    realm **Enterprise**
    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
network-interface
  name wancom0
  sub-port-id 0
  description
  hostname
  ip-address 192.9.230.221
  pri-utility-addr
  sec-utility-addr
  netmask 255.255.255.0
  gateway 192.9.230.254
  sec-gateway
  gw-heartbeat

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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
ftp-address
icmp-address
snmp-address
telnet-address
ssh-address

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

network-interface
name **s0p0**
sub-port-id 0
description
hostname
ip-address **10.80.130.250**
pri-utility-addr
sec-utility-addr
netmask **255.255.255.0**
gateway **10.80.130.1**
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 attavaya.com
dns-timeout 11
hip-ip-list **10.80.130.250**
ftp-address
icmp-address **10.80.130.250**
snmp-address
telnet-address

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ssh-address

ANNOTATION: The network interface below defines the IP addresses on the interface connected to the network on which the AT&T IP services resides.

network-interface
name **s1p0**
sub-port-id 0
description
hostname
ip-address **192.168.62.51**
pri-utility-addr
sec-utility-addr
netmask **255.255.255.128**
gateway **192.168.62.1**
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 **192.168.62.51**
ftp-address
icmp-address **192.168.62.51**
snmp-address
telnet-address
ssh-address

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP services resides, and applies the SIP manipulation **modSendRecv**. Note that this manipulation is not used for AT&T IP Toll Free service but is kept in here for customers also subscribing to AT&T IP Flexible Reach service.

realm-config
identifier **ATT**
description
addr-prefix 0.0.0.0
network-interface **s1p0:0**
mm-in-realm enabled
mm-in-network enabled
mm-same-ip enabled

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mm-in-system enabled
bw-cac-non-mm disabled
msm-release disabled
generate-UDP-checksum disabled
max-bandwidth 0
fallback-bandwidth 0
max-priority-bandwidth 0
max-latency 0
max-jitter 0
max-packet-loss 0
observ-window-size 0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid modSendRecv
manipulation-string
manipulation-pattern
class-profile
average-rate-limit 0
access-control-trust-level none
invalid-signal-threshold 0
maximum-signal-threshold 0
untrusted-signal-threshold 0
nat-trust-threshold 0
deny-period 30
ext-policy-svr
diam-e2-address-realm
symmetric-latching disabled
pai-strip disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching none
restriction-mask 32
accounting-enable enabled
user-cac-mode none
user-cac-bandwidth 0
user-cac-sessions 0
icmp-detect-multiplier0
icmp-advertisement-interval 0
icmp-target-ip

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monthly-minutes 0
net-management-control disabled
delay-media-update disabled
refer-call-transfer disabled
dyn-refer-term disabled
codec-policy
codec-manip-in-realm disabled
constraint-name
call-recording-server-id
xnq-state xnq-unknown
hairpin-id 0
stun-enable disabled
stun-server-ip 0.0.0.0
stun-server-port 3478
stun-changed-ip 0.0.0.0
stun-changed-port 3479
match-media-profiles
qos-constraint
sip-profile
sip-isup-profile
block-rtcp disabled
hide-egress-media-update disabled

ANNOTATION: The realm configuration **Enterprise** below represents the internal network on which the Avaya elements reside.

realm-config
identifier **Enterprise**
description
addr-prefix 0.0.0.0
network-interfaces **s0p0:0**
mm-in-realm enabled
mm-in-network enabled
mm-same-ip enabled
mm-in-system enabled
bw-cac-non-mm disabled
msm-release disabled
generate-UDP-checksum disabled
max-bandwidth 0
fallback-bandwidth 0
max-priority-bandwidth 0
max-latency 0
max-jitter 0
max-packet-loss 0
observ-window-size 0
parent-realm
dns-realm

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media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit 0
access-control-trust-level none
invalid-signal-threshold 0
maximum-signal-threshold 0
untrusted-signal-threshold 0
nat-trust-threshold 0
deny-period 30
ext-policy-svr
diam-e2-address-realm
symmetric-latching disabled
pai-strip disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching none
restriction-mask 32
accounting-enable enabled
user-cac-mode none
user-cac-bandwidth 0
user-cac-sessions 0
icmp-detect-multiplier0
icmp-advertisement-interval 0
icmp-target-ip
monthly-minutes 0
net-management-control disabled
delay-media-update disabled
refer-call-transfer enabled
dyn-refer-term disabled
codec-policy
codec-manip-in-realm disabled
constraint-name
call-recording-server-id
xnq-state xnq-unknown
hairpin-id 0
stun-enable disabled
stun-server-ip 0.0.0.0

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stun-server-port 3478
stun-changed-ip 0.0.0.0
stun-changed-port 3479
match-media-profiles
qos-constraint
sip-profile
sip-isup-profile
block-rtcp disabled
hide-egress-media-update disabled

ANNOTATION: The session agent below represents the Session Manager used in this reference configuration.

session-agent
hostname SM62
ip-address 10.64.19.210
port 5060
state enabled
app-protocol SIP
app-type
transport-method UDP+TCP
realm-id Enterprise
egress-realm-id
description
carriers
allow-next-hop-lp enabled
constraints disabled
max-sessions 0
max-inbound-sessions0
max-outbound-sessions 0
max-burst-rate 0
max-inbound-burst-rate 0
max-outbound-burst-rate 0
max-sustain-rate 0
max-inbound-sustain-rate 0
max-outbound-sustain-rate 0
min-seizures 5
min-asr 0
time-to-resume 0
ttr-no-response 0
in-service-period 0
burst-rate-window 0
sustain-rate-window 0
req-uri-carrier-mode None
proxy-mode
redirect-action Proxy
loose-routing enabled

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```
send-media-session      enabled
response-map
ping-method             OPTIONS;hops=1
ping-interval           180
ping-send-mode          keep-alive
ping-all-addresses      disabled
ping-in-service-response-codes
out-service-response-codes
media-profiles
in-translationid
out-translationid
trust-me                enabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me             disabled
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate       0
early-media-allow
invalidate-registrations        disabled
rfc2833-mode            none
rfc2833-payload         0
codec-policy
enforcement-profile
refer-call-transfer     disabled
reuse-connections       TCP
tcp-keepalive           enabled
tcp-reconn-interval     10
max-register-burst-rate 0
register-burst-window   0
sip-profile
sip-isup-profile
```


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ANNOTATION: The session agent below represents the AT&T IP services border element. The Acme Packet SBC will attempt to send calls to the border element based on successful responses to the OPTIONS **ping-method**. The AT&T IP Toll Free service border element is also specified in the **session-group** section below.

```
session-agent
        hostname                135.242.225.210
        ip-address              135.242.225.210
        port                    5060
        state                   enabled
        app-protocol            SIP
        app-type
        transport-method        UDP
        realm-id                ATT
        egress-realm-id
        description
        carriers
        allow-next-hop-lp       enabled
        constraints             disabled
        max-sessions            0
        max-inbound-sessions    0
        max-outbound-sessions           0
        max-burst-rate          0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate        0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures            5
        min-asr                 0
        time-to-resume          0
        ttr-no-response         0
        in-service-period       0
        burst-rate-window       0
        sustain-rate-window     0
        req-uri-carrier-mode    None
        proxy-mode
        redirect-action
        loose-routing           enabled
        send-media-session      enabled
        response-map
        ping-method             OPTIONS;hops=70
        ping-interval           60
        ping-send-mode          keep-alive
        ping-all-addresses      disabled
        ping-in-service-response-codes
```


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out-service-response-codes
media-profiles
in-translationid
out-translationid
trust-me enabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me disabled
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate 0
early-media-allow
invalidate-registrations disabled
rfc2833-mode none
rfc2833-payload 0
codec-policy
enforcement-profile
refer-call-transfer disabled
reuse-connections NONE
tcp-keepalive none
tcp-reconn-interval 0
max-register-burst-rate 0
register-burst-window 0
sip-profile
sip-isup-profile

ANNOTATION: The session agent below is used for failover testing to ATT IP services. The state is changed to **enabled** when the testing is performed.

session-agent
hostname **1.1.1.1**
ip-address **1.1.1.1**
port **5060**
state **disabled**
app-protocol **SIP**
app-type
transport-method **UDP**
realm-id **ATT**
egress-realm-id
description **ATT-Failover**

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carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	**OPTIONS;hops=70**
ping-interval	**60**
ping-send-mode	**keep-alive**
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	

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trunk-group
max-register-sustain-rate 0
early-media-allow
invalidate-registrations disabled
rfc2833-mode none
rfc2833-payload 0
codec-policy
enforcement-profile
refer-call-transfer disabled
reuse-connections NONE
tcp-keepalive none
tcp-reconn-interval 0
max-register-burst-rate 0
register-burst-window 0
sip-profile
sip-isup-profile

> **ANNOTATION**: The **session group** below specifies the AT&T IP services border element.
>
> Note - Multiple session-agents may be specified in a session-group. The *strategy* parameter may be used to select how these multiple session-agents are used (e.g. *Hunt* and *RoundRobin*).

session-group
group-name SP_PROXY
description
state enabled
app-protocol SIP
strategy RoundRobin
dest
1.1.1.1
135.242.225.210
trunk-group
sag-recursion enabled
stop-sag-recurse 401,407

> **ANNOTATION**: The SIP interface below is used to communicate with the AT&T IP services.

sip-interface
state enabled
realm-id ATT
description
sip-port
address 192.168.62.51
port 5060
transport-protocol UDP

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```
            tls-profile
            allow-anonymous        all
            ims-aka-profile
    carriers
    trans-expire                0
    invite-expire               0
    max-redirect-contacts 0
    proxy-mode
    redirect-action
    contact-mode                none
    nat-traversal               none
    nat-interval                30
    tcp-nat-interval            90
    registration-caching        disabled
    min-reg-expire              300
    registration-interval       3600
    route-to-registrar          disabled
    secured-network             disabled
    teluri-scheme               disabled
    uri-fqdn-domain
    trust-mode                  all
    max-nat-interval            3600
    nat-int-increment           10
    nat-test-increment          30
    sip-dynamic-hnt             disabled
    stop-recurse                401,407
    port-map-start              0
    port-map-end                0
    in-manipulationid
    out-manipulationid
    manipulation-string
    manipulation-pattern
    sip-ims-feature             disabled
    operator-identifier
    anonymous-priority          none
    max-incoming-conns          0
    per-src-ip-max-incoming-conns 0
    inactive-conn-timeout 0
    untrusted-conn-timeout            0
    network-id
    ext-policy-server
    default-location-string
    charging-vector-mode pass
    charging-function-address-mode pass
    ccf-address
    ecf-address
```

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term-tgrp-mode none
implicit-service-route disabled
rfc2833-payload 101
rfc2833-mode transparent
constraint-name
response-map
local-response-map
ims-aka-feature disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive none
add-sdp-invite disabled
add-sdp-profiles
sip-profile
sip-isup-profile

ANNOTATION: The SIP interface below is used to communicate with the Avaya elements.

sip-interface

state enabled
realm-id **Enterprise**
description
sip-port
  address **10.80.130.250**
  port **5060**
  transport-protocol **TCP**
  tls-profile
  allow-anonymous all
  ims-aka-profile
carriers
trans-expire 0
invite-expire 0
max-redirect-contacts 0
proxy-mode
redirect-action
contact-mode none
nat-traversal none
nat-interval 30
tcp-nat-interval 90
registration-caching disabled
min-reg-expire 300
registration-interval 3600
route-to-registrar disabled
secured-network disabled
teluri-scheme disabled
uri-fqdn-domain

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```
trust-mode              all
max-nat-interval        3600
nat-int-increment       10
nat-test-increment      30
sip-dynamic-hnt         disabled
stop-recurse            401,407
port-map-start          0
port-map-end            0
in-manipulationid
out-manipulationid      rejectOptions
manipulation-string
manipulation-pattern
sip-ims-feature         disabled
operator-identifier
anonymous-priority      none
max-incoming-conns      0
per-src-ip-max-incoming-conns  0
inactive-conn-timeout   0
untrusted-conn-timeout  0
network-id
ext-policy-server
default-location-string
charging-vector-mode    pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode          none
implicit-service-route  disabled
rfc2833-payload         101
rfc2833-mode            transparent
constraint-name
response-map
local-response-map
ims-aka-feature         disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive           none
add-sdp-invite          disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```


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ANNOTATION: The SIP manipulation shown below are used for modifying the **sendonly** value in SDP to **sendrecv**. See **Section 2.2**, bullet **1** for further details.

sip-manipulation
name **modSendRecv**
description **Modify sendonly to sendrecv**
split-headers
join-headers
header-rule
name **modsendonly**
header-name **Content-type**
action **manipulate**
comparison-type **case-sensitive**
msg-type **any**
methods **INVITE**
match-value
new-value
element-rule
name **modmline**
parameter-name **application/sdp**
type **mime**
action **find-replace-all**
match-val-type **any**
comparison-type **case-sensitive**
match-value **sendonly**
new-value **sendrecv**

ANNOTATION: The SIP manipulation shown below intercepts the SIP OPTIONS message from AT&T Border Element and respond with Acme Packet alive message.

sip-manipulation
name **rejectOptions**
description
split-headers
join-headers
header-rule
name **RejectOpts**
header-name **From**
action **reject**
comparison-type **case-sensitive**
msg-type **request**
methods **OPTIONS**
match-value
new-value **405:"Acme Packet is alive, check back later"**

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> **ANNOTATION**: The steering pools below define the IP Addresses and RTP port ranges on the respective realms. The **ATT** realm IP Address will be used as the CPE media traffic IP Address to communicate with AT&T. The **ATT** realm RTP port range is an AT&T IP services requirement. Likewise, the IP Address and RTP port range defined for the **Enterprise** realm steering pool will be used to communicate with the Avaya elements. Please note that the **Enterprise** realm port range does not have to be within the range specified below.

```
steering-pool
        ip-address              192.168.62.51
        start-port              16384
        end-port                32767
        realm-id                ATT
steering-pool
        ip-address              10.80.130.250
        start-port              16384
        end-port                32767
        realm-id                Enterprise
system-config
        hostname                Enterprise-Acme
        description
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled            enabled
        enable-snmp-auth-traps          disabled
        enable-snmp-syslog-notify       disabled
        enable-snmp-monitor-traps       disabled
        enable-env-monitor-traps        disabled
        snmp-syslog-his-table-length    1
        snmp-syslog-level       WARNING
        system-log-level        WARNING
        process-log-level       NOTICE
        process-log-ip-address          0.0.0.0
        process-log-port        0
        collect
                sample-interval         5
                push-interval           15
                boot-state              disabled
                start-time              now
                end-time                never
                red-collect-state       disabled
                red-max-trans           1000
                red-sync-start-time     5000
                red-sync-comp-time      1000
                push-success-trap-state         disabled
```


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call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	**192.168.62.1**
restart	enabled
exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	disabled
cli-more	disabled
terminal-height	24
debug-timeout	0
trap-event-lifetime	0
default-v6-gateway	::
ipv6-support	disabled
cleanup-time-of-day	00:00

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9. Verification Steps

The following steps may be used to verify this reference configuration:

9.1. Avaya Aura® Communication Manager

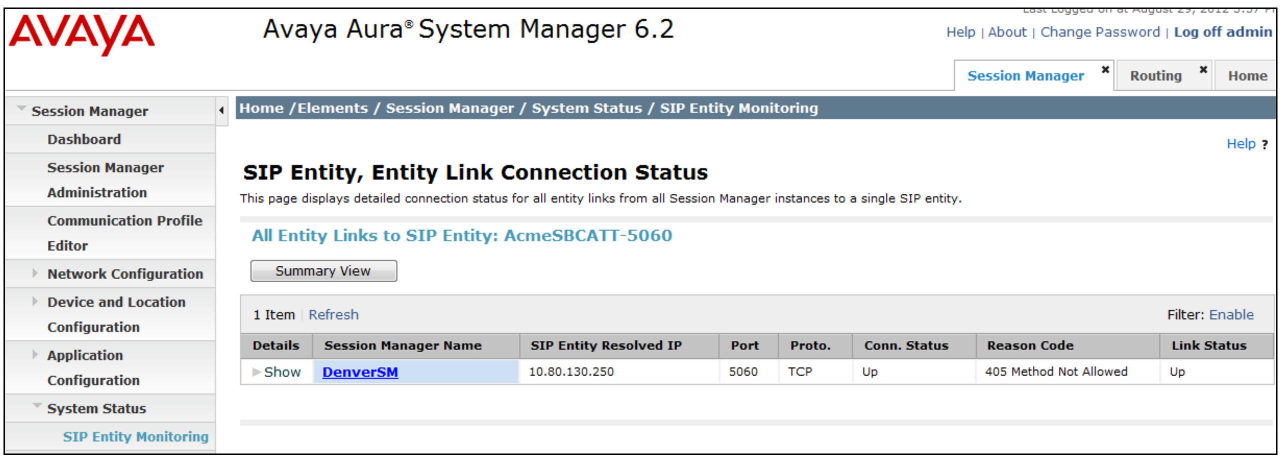
The following examples are only a few of the monitoring commands available on Communication Manager. See [**5**] and [**6**] for more information.

- From the Communication Manager console connection enter the command ***list trace tac xxx***, where ***xxx*** is a trunk access code to verify that the inbound or outbound calls are using the right trunk groups. Similarly, ***list trace station***, ***list trace vdn***, and ***list trace vector***, ***status trunk*** and ***status station*** commands can be used on Communication Manager.

9.2. Avaya Aura® Session Manager

Navigate to **Home→ Elements→ Session Manager→ System Status → SIP Entity Monitoring** and click on the SIP Entity for which the status is required. Following screen shows status for the entity link between Session Manager and Acme Packet SBC.

Note: The Reason Code column indicates that Session Manager has received a **SIP 405 Method Not Allowed** response (normal for this reference configuration) to the **SIP OPTIONS** it generated. This response is sufficient for SIP Link Monitoring to consider the link up.



9.3. AT&T IP Toll Free

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Verify basic call functions such as hold, transfer and conference.
3. Verify the use of DTMF signaling.

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10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Acme Packet SBC can be configured to interoperate successfully with the AT&T IP Toll Free service using either AVPN or MIS-PNT transport. This solution provides users of Avaya Aura® Communication Manager the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection. These Application Notes further demonstrated that the Avaya Aura® Session Manager Adaptation Module could be utilized to do digit manipulation for inbound calls.

Note: These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide **configuration guidance** to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

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11. References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

Avaya Aura® Session Manager/System Manager

[1] Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.2, July 2012

[2] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.2, Issue 2, August 2012

[3] Implementing Avaya Aura® System Manager, Doc ID 03-603473 Issue 1, July 2012

[4] Administering Avaya Aura® System Manager, Document Number 03-603324, Release 6.2, July 2012

Avaya Aura® Communication Manager

[5] Administering Avaya Aura® Communication Manager, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509

[6] Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Release 5.2, April 2009, Document Number 07-600780

Avaya Modular Messaging

[7] Modular Messaging Multi-Site Guide Release 5.1, June 2009

[8] Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades, June 2009

[9] Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades, June 2009

[10] Modular Messaging for IBM Lotus Domino 5.1 Installation and Upgrades, June 2009

Acme Packet Support (login required):

[11] http://www.acmepacket.com/support.htm

AT&T IP Toll Free Service Descriptions:

[12] *AT&T IP Toll Free* http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/

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