Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Avaya Aura® Session Border Controller 6.0.2 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Avaya Aura® Session Border Controller with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 6.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 6.0.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Avaya Aura® Session Border Controller 6.0.2 is the point of connection between Avaya Aura® Session Manager 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.

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.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Avaya Aura® Session Border Controller with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

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The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing AVPN or MIS/PNT[1] transport.

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

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with System Manager, Avaya Aura® Session Manager, Communication Manager, Avaya phones, fax machines (Ventafax application), Session Border Controller, and Avaya Modular Messaging.
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via AVPN transport.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for examples) between Session Manager, Communication Manager, Session Border Controller, 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.
- T.38 Fax.
- Passing of DTMF events and their recognition by navigating automated menus.
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer. Legacy Transfer Connect and Alternate Destination Routing features were also tested.

[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP..

2.2. Test Results

The main test objectives were to verify the following features and functionality:

- Inbound AT&T IP Toll Free service calls to Communication Manager telephones and VDNs/Vectors.
- Call and two-way talk path establishment between PSTN and Communication Manager telephones via the AT&T Toll Free service.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- T.38 fax calls between Communication Manager the AT&T IP Toll Free service/PSTN G3 and SG3 fax endpoints.
- DTMF tone transmission using RFC 2833 between Communication Manager and the AT&T IP Toll Free service/PSTN automated access systems.
- Inbound AT&T IP Toll Free service calls to Communication Manager that are directly routed to stations, and if unanswered, can be covered to Avaya Modular Messaging.
- Long duration calls.

The test objectives stated in **Section 2.1** with limitations as noted in **Section 2.2.1,** were verified.

2.2.1. Known Limitations

1. 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.729A, 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.
2. G.726 codec is not supported between Communication Manager and the AT&T IP Toll Free service.
3. 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 to have G.711 fax calls work. T.38 faxing is supported, as is Group 3 and Super Group 3 fax. Fax speeds are limited to 9600 in the configuration tested. In addition, Fax Error Correction Mode (ECM) is not supported by Communication Manager.
4. Avaya SIP telephones currently send RTP with a fixed 20ms packet interval even though an interval of 30 is specified in the SIP signaling sent by Communication Manager (see **Section 6.6**). This could cause reduced customer busy-hour bandwidth for AVPN based transport. A fix for this issue is currently being developed by Avaya.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 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.

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 communications services for a particular enterprise site. In the reference configuration, Communication Manager runs on an Avaya S8800 Server in a Processor Ethernet (Procr) 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 G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya desk telephones are represented with Avaya A175, 46x0, 96x0, and 96x1 Series IP Telephones running H.323 or SIP software, Avaya 6211 Series Analog Telephones, as well as Avaya one-X® Agent. Note that agent telephones are H.323.
- The Session Border Controller provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network[2]. UDP transport protocol is used between the Avaya Aura® SBC and the AT&T IP Toll Free service.
- An existing Avaya Modular Messaging system (in Multi-Site mode in this reference configuration) provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Modular Messaging is beyond the scope of this document.
- Inbound calls were placed from PSTN via the AT&T IP Toll Free service, through the Avaya Aura® SBC to the Session Manager which routed the call to Communication

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

Manager. Communication Manager terminated the call to the appropriate agent/phone or fax extension. The H.323 telephones on the enterprise side registered to the Communication Manager Procr. The SIP telephones registered to Session Manager.

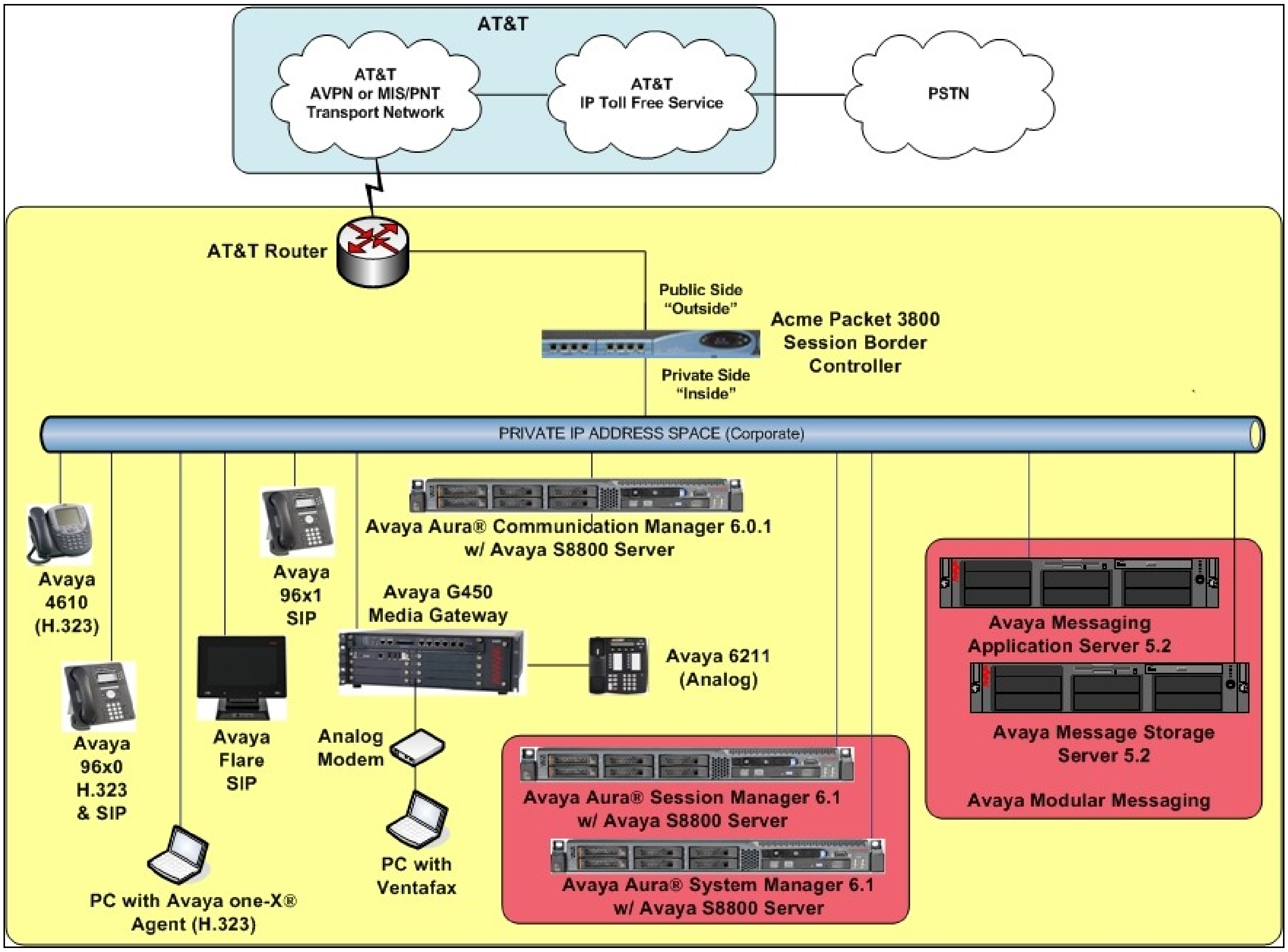


Figure 1: Reference configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

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	192.168.67.207
Avaya Aura® Session Manager	
Management IP Address	192.168.67.209
Network IP Address	192.168.67.210
Avaya Aura® Communication Manager	
Procr IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	40xxx = H323 and Analog 41xxx = SIP
Avaya CPE local dial plan	4xxxx
Voice Messaging Pilot Extension	46000
Avaya Aura® Session Border Controller	
IP Address of Outside (Public) Interface (connected to AT&T Access Router/IP Toll Free Service)	192.168.64.130 (active)
IP Address of Inside (Private) Interface (connected to Avaya Aura® Session Manager)	192.168.67.125 (active)
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Messaging Server (MSS) IP Address	192.168.67.140
Modular Messaging Dial Plan	1723114xxxx
AT&T IP Toll Free Service	
Border Element IP Address	135.25.29.74
AT&T Access router interface (to Avaya Aura® outside)	192.168.64.254
AT&T Access Router NAT address (Avaya Aura® outside address)	135.16.170.55

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by Session Manager and Communication Manager, two general call flows are described in this section. The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Toll Free service call that arrives on Session Manager and is subsequently routed to Communication Manager.

1. A PSTN telephone 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 the Avaya Aura® SBC.

4. The Avaya Aura® SBC performs SIP Network Address Translation (NAT) and 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 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) a vector, which in turn, routes the call to an agent, or b) directly to an agent or telephone.

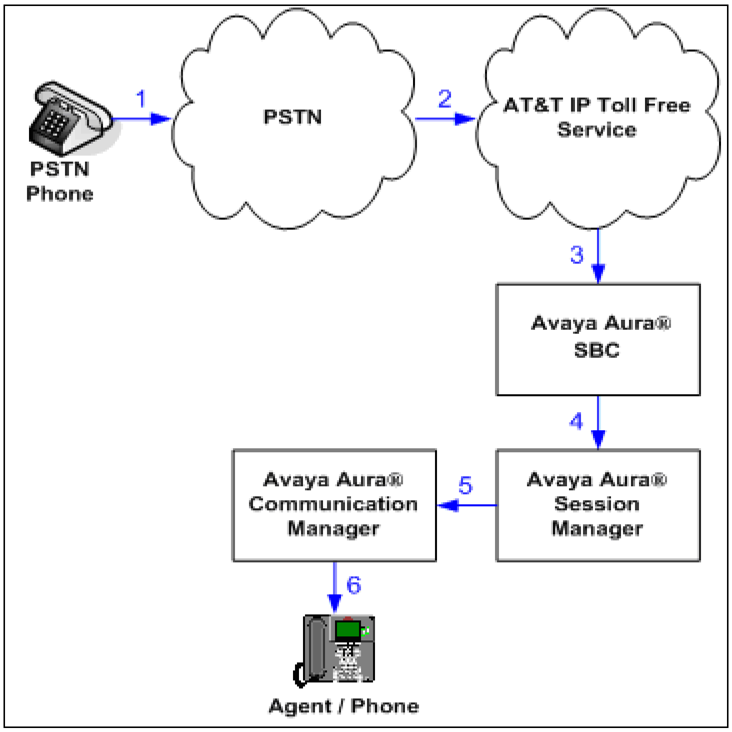


Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Telephone

The second call scenario illustrated in **Figure 3** is an inbound call that is covered to voicemail. In this scenario, the voicemail system is a Modular Messaging system connected to Session Manager. The Modular Messaging system is in MultiSite mode.

1. Same as the **Steps 1-5** and **Step 6b** from the first call scenario.
2. The called Communication Manager agent or telephone does not answer the call, and the call covers to the agent's or telephone's voicemail. Communication Manager forwards[3] the call to Session Manager.
3. 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 Modular Messaging. Modular Messaging answers the call and connects the caller to the called agent's or telephone's voice mailbox. Note that the call[4] continues to go through Communication Manager.

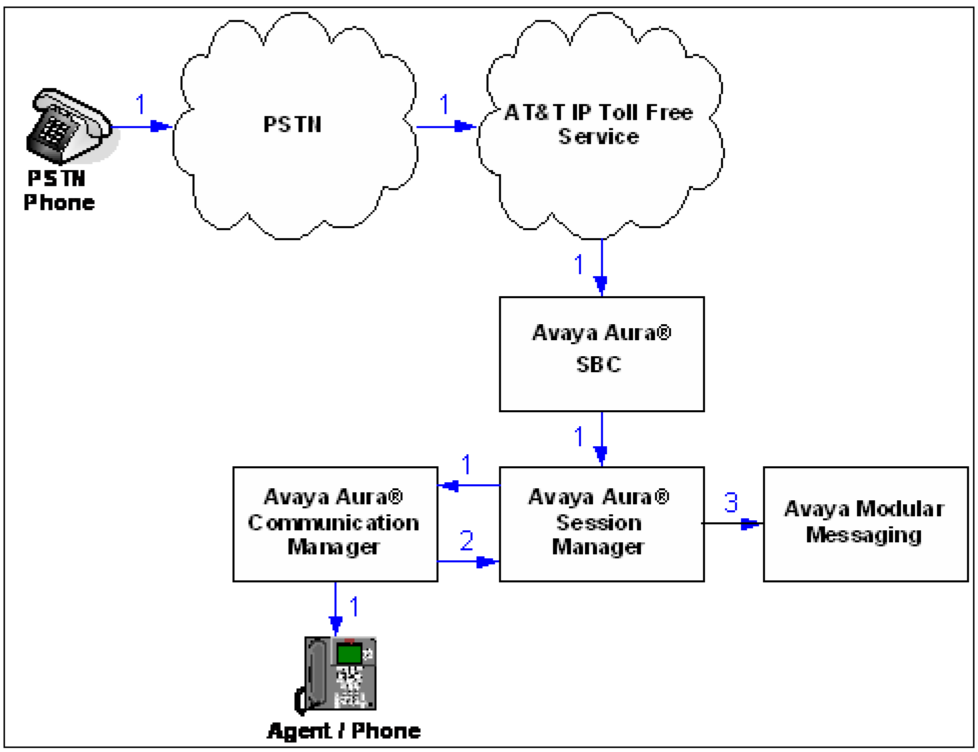


Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Telephone Covered to Avaya Modular Messaging

[3] Communication Manager places a call to Modular Messaging, and then connects the inbound caller to Modular Messaging. SIP redirect methods, e.g., 302, are not used.
[4] The SIP signaling path still goes through Communication Manager. In addition, since the inbound call and Modular Messaging use different codecs (G.729 and G.711, respectively), Communication Manager performs the transcoding, and thus the RTP media path also goes through Communication Manager.

4. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura® System Manager 6.1 SP4 (6.1.0.0.7345-6.1.5.112 update 6.1.8.1.1455) System Platform 6.0.3.1.3
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.4.0.614005)
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 SP3 (00.1.510.1-19009) System Platform 6.0.3.1.3
Avaya G450 Media Gateway	31.19.2
MM711 Analog card	HW31 FW094
Avaya S8800 Server	Avaya Aura® Session Border Controller 6.0.2.0.3
Avaya 9630 IP Telephone	H.323 Version S3.110b (ha96xxua3_11.bin) SIP Version 2.6.4 (SIP96xx_2_6_4_0.bin)
Avaya 9621 IP Telephone	SIP Version 6.0.1 (S96x1_SALBR6_0_1_V452)
Avaya A175 Flare™ Desktop Video Device (SIP telephone function)	SIP Version 1.0.3 (SIP_A175_1_0_3_000011)
Avaya one-X® Agent	2.5.00467.09
Avaya 4610SW IP Telephone	H323 Version 2.9.1 (a10d01b2_9_1.bin)
Avaya 6211 Analog telephone	-
Avaya Modular Messaging (MAS and MSS) on Avaya S3500 Servers	Release 5.2 – SP5 with Patch 1 (9.0.350.5019)
Fax device	Ventafax Home Version 6.1.59.144
AT&T IP Toll Free Service using AVPN/MIS-PNT transport service connection	VNI 20 & VNI 21

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release 6.1

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

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

This section provides the procedures for configuring Session Manager to receive calls from and route calls to the SIP trunk between Communication Manager and Session Manager, and the SIP trunk between Session Manager and the Avaya Aura® SBC. In addition, provisioning for calls to Modular Messaging are described.

Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. Session Manager connects and normalizes disparate SIP network components and provides a central point for external SIP trunking to the PSTN. The various SIP network components are represented as SIP Entities and the connections/trunks between Session Manager and those components are represented as Entity Links. Thus, rather than connecting to every other SIP Entity in the enterprise, each SIP Entity simply connects to Session Manager and relies on Session Manager to route calls to the correct destination. This approach reduces the dial plan and trunking administration needed on each SIP Entity, and consolidates said administration in a central place, namely System Manager.

When calls arrive at Session Manager from a SIP Entity, Session Manager applies SIP protocol and numbering modifications to the calls. These modifications, referred to as Adaptations, are sometimes necessary to resolve SIP protocol differences between disparate SIP Entities, and also serve the purpose of normalizing the calls to a common or uniform numbering format, which allows for simpler administration of routing rules in Session Manager. Session Manager then matches the calls against certain criteria embodied in profiles termed Dial Patterns, and determines the destination SIP Entities based on Routing Policies specified in the matching Dial Patterns. Lastly, before the calls are routed to the respective destinations, Session Manager again applies Adaptations in order to bring the calls into conformance with the SIP protocol interpretation and numbering formats expected by the destination SIP Entities.

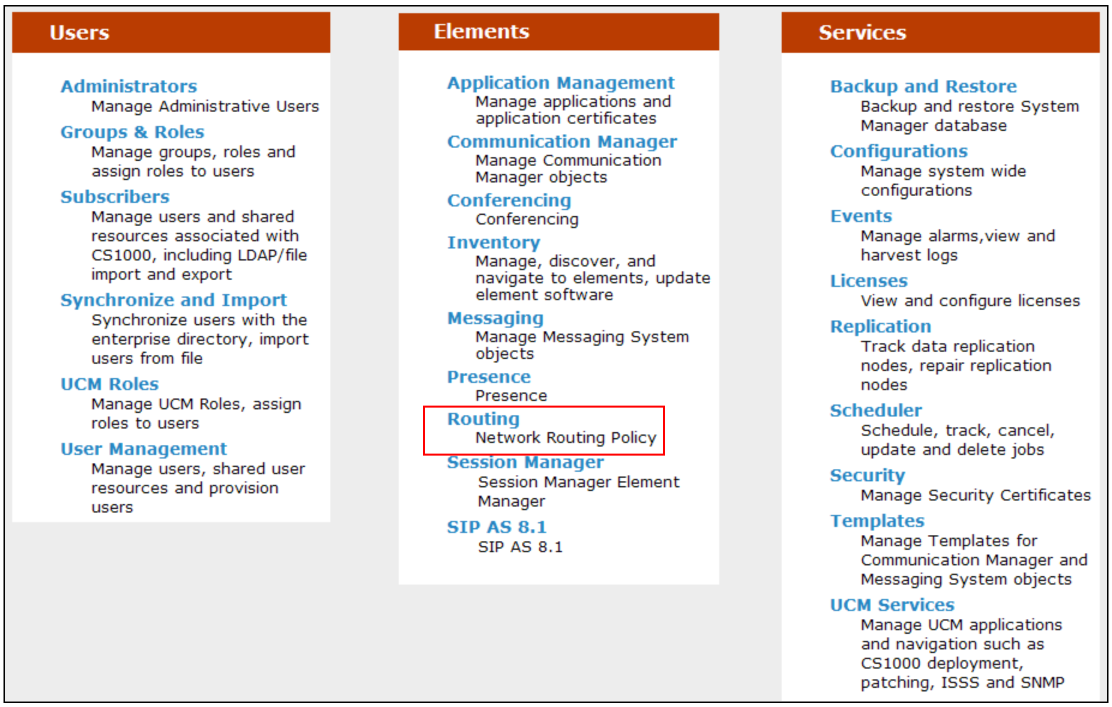
The following administration activities will be described:

- Define SIP Domain
- Define Locations for Communication Manager, the Avaya Aura® SBC, and Modular Messaging.
- Configure the Adaptation Modules that will be associated with the SIP Entities for Communication Manager, the Avaya Aura® SBC, and Modular Messaging.
- Define SIP Entities corresponding to Communication Manager, the Avaya Aura® SBC, and Modular Messaging.
- Define Entity Links describing the SIP trunk between Communication Manager and Session Manager, the SIP Trunk between Session Manager and the Avaya Aura® SBC, and the SIP trunk between Session manager and Modular Messaging.

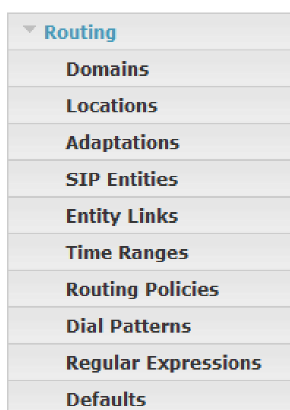
- Define Routing Policies associated with the Communication Manager, the Avaya Aura® SBC and Modular Messaging.
- Define Dial Patterns, which govern which routing policy will be selected for call routing.

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager.

In the **Log On** screen (not shown), enter appropriate **User ID** and **Password** and press the **Log On** button. Once logged in, a Release 6.1 **Home** screen like the following is displayed. From the **Home** screen below, under the **Elements** heading in the center, select **Routing**.



The screen shown below shows the various sub-headings of the left navigation menu that will be referenced in this section.

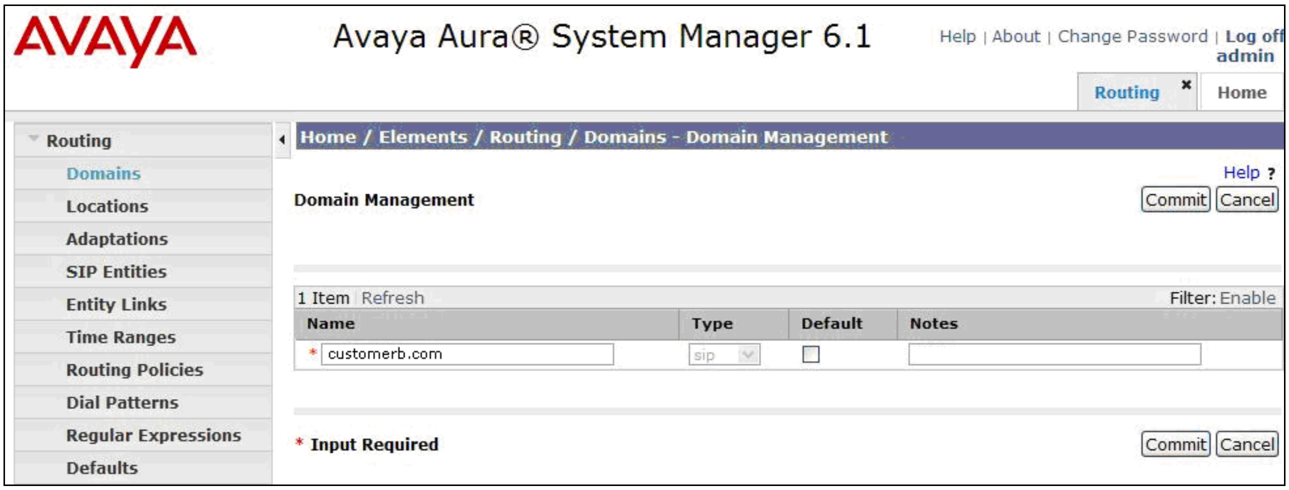


5.1. SIP Domain

Step 1 - Select **Domains** from the left navigation menu. In the reference configuration domain **customerb.com** was defined.

Step 2 - Click **New** (not shown)**.** Enter the following values and use default values for remaining fields**.**

- **Name** Enter the enterprise SIP Domain Name. In the sample screen below, **customerb.com** is shown.
- **Type** Verify **sip** is selected.
- **Notes** Add a brief description. [Optional]



Step 3 - Click **Commit** to save.

Note - Multiple SIP Domains may be defined if required.

5.2. Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. Location identifiers can be defined in a broad scope (e.g. 192.168.67.x for all devices on a particular subnet), or individual devices (e.g. 192.168.67.202 for a device's IP address). In the reference configuration Communication Manager, Modular Messaging, and the Avaya Aura® SBC were each defined as individual Locations.

5.2.1. Location for Avaya Aura® Communication Manager

Step 1 - Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section**,** enter the following values and use default values for remaining fields**.**

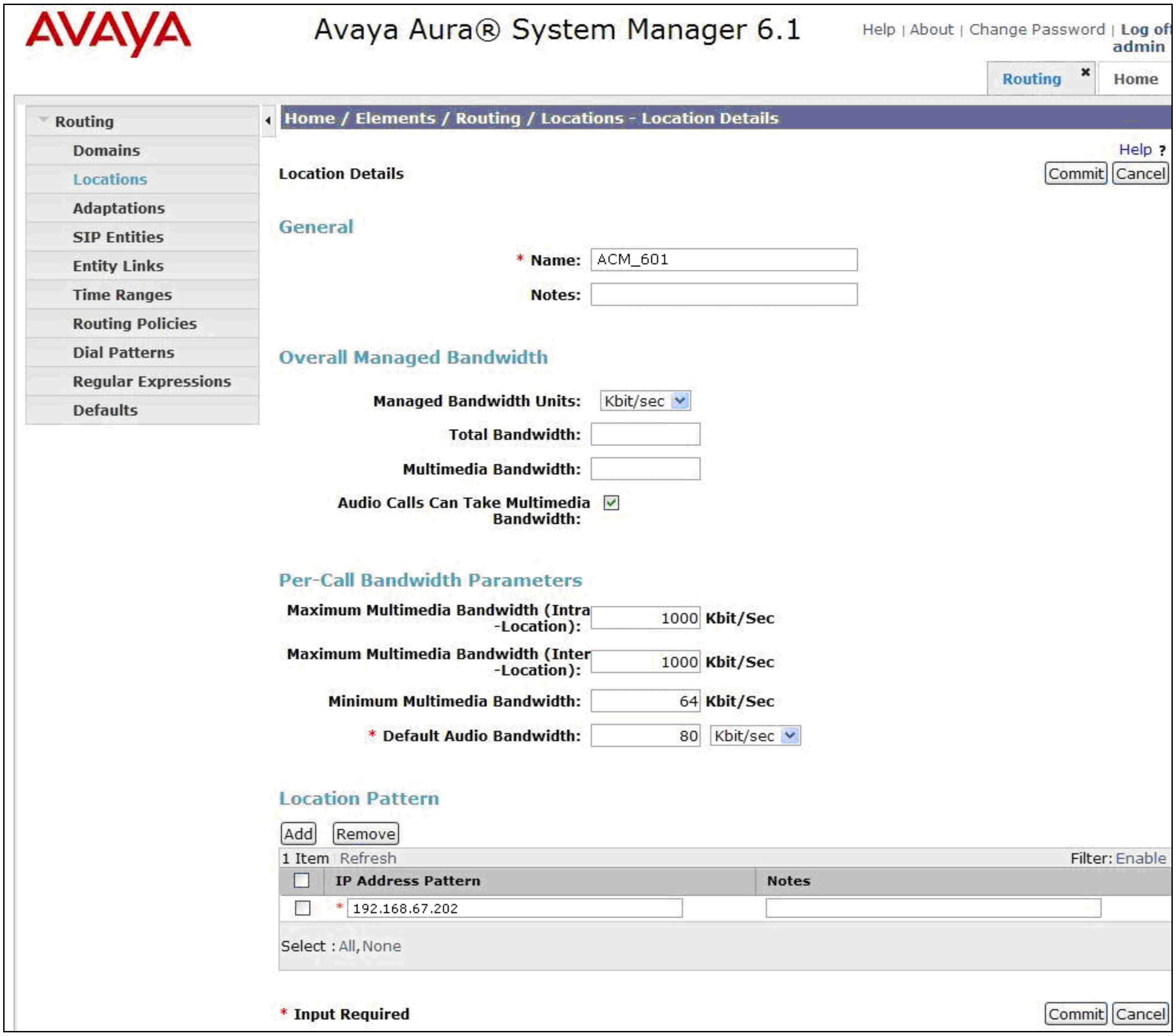
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the IP Address used to identify the Communication Manager location (e.g. **192.168.67.202**).
- **Notes** Add a brief description. [Optional]

Step 3 - Click **Commit** to save.

The screen below shows the screen for the Location defined for Communication Manager.



5.2.2. Location for the Avaya Aura® Session Border Controller

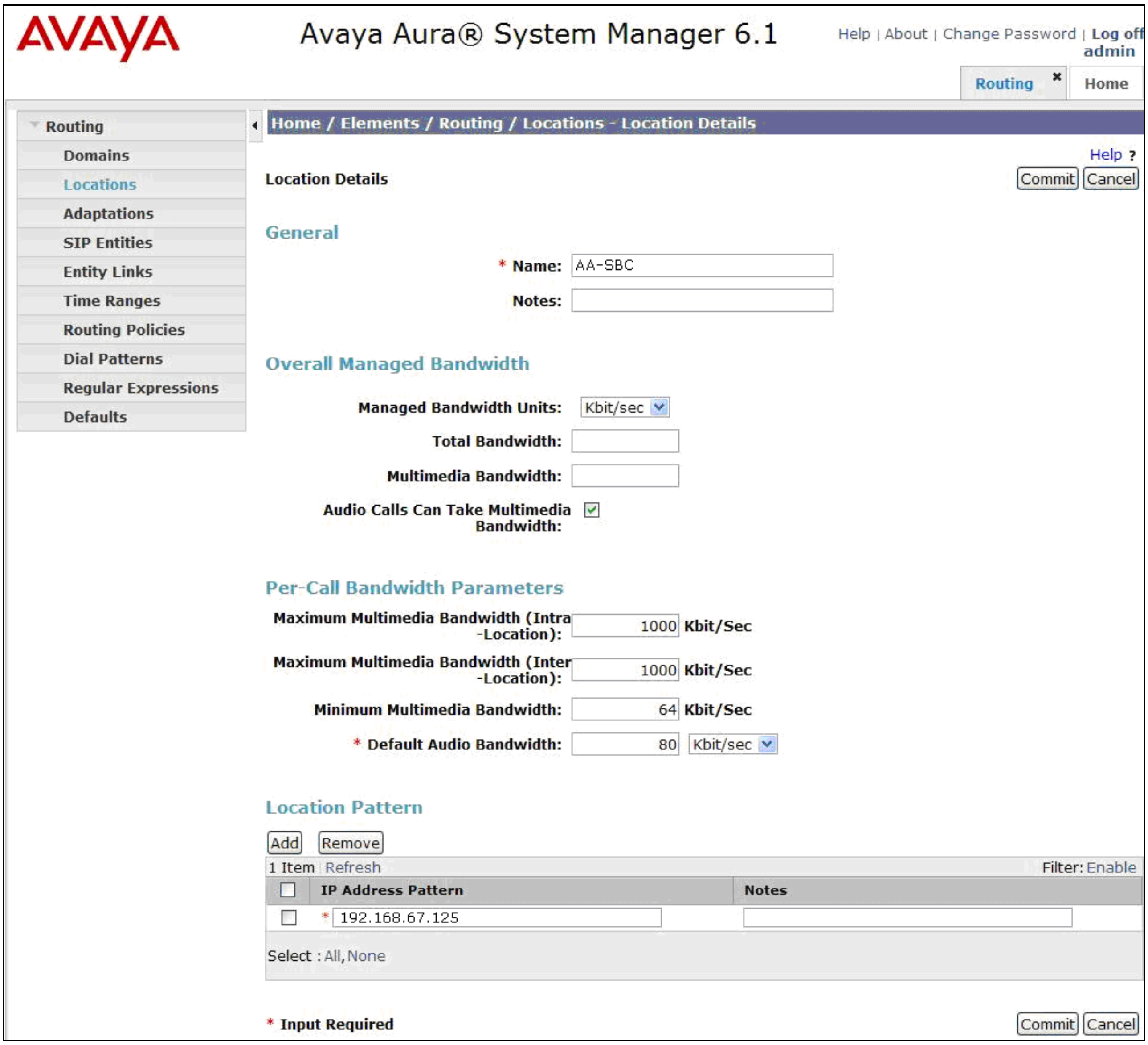
Step 1 - Select **Locations** from the left navigational menu and click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern:** Enter the IP Address or IP Address pattern used to identify the SBC location (e.g. **192.168.67.125**).
- **Notes:** Add a brief description. [Optional]

Step 3 - Click **Commit** to save.



5.2.3. Location for Modular Messaging

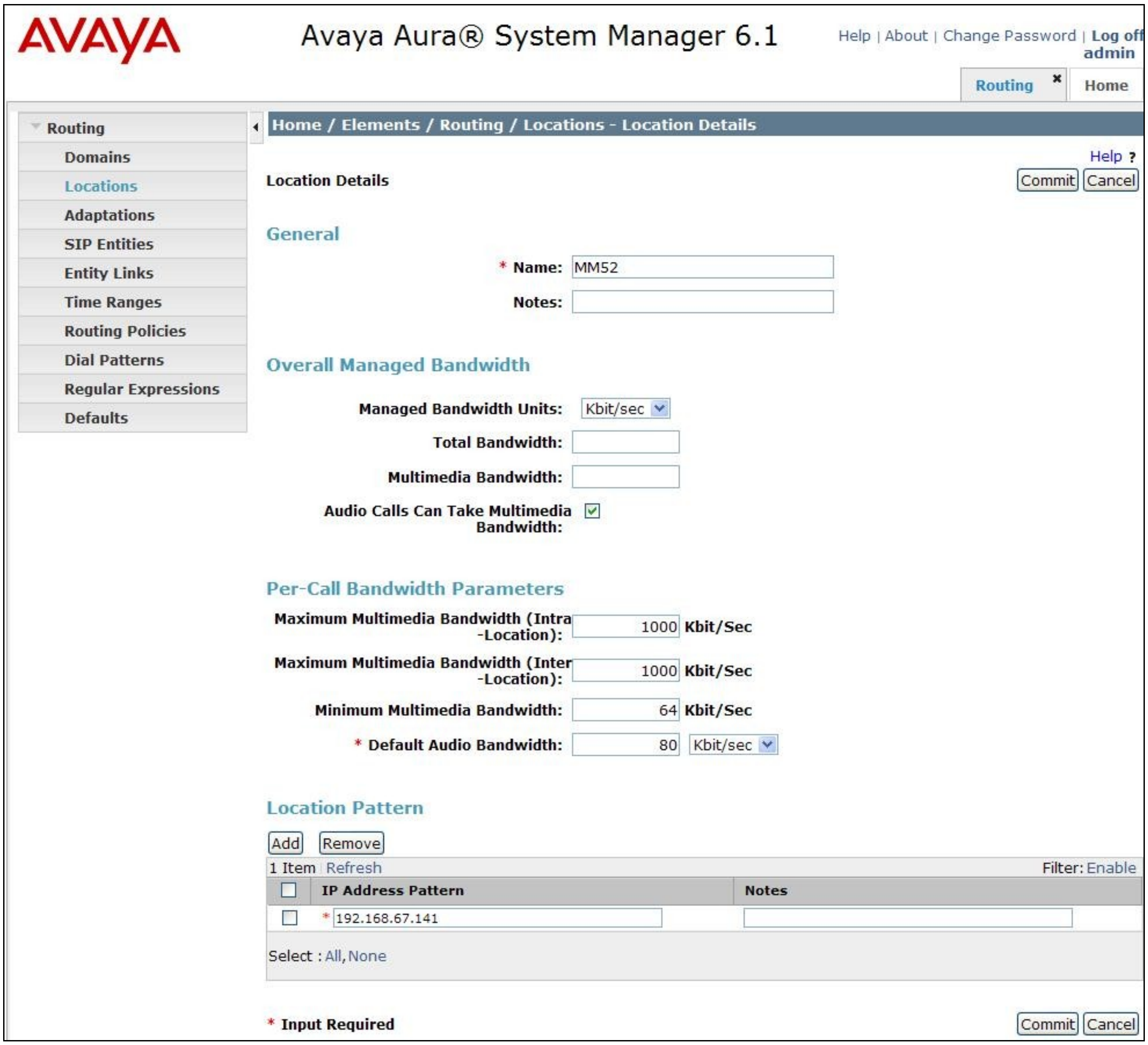
Step 1 - Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern:** Enter the IP Address used to identify the Modular Messaging MAS location (e.g. **192.168.67.141**).
- **Notes:** Add a brief description. [Optional]

Step 3 - Click **Commit** to save.



5.2.4. Location for Other CPE Devices

The location **main** is used as a wild card for any other devices in the CPE that may source traffic to Session Manager. In the Reference configuration Session Manager was assigned to this location. Note that a specific location like those described in the previous sections could have been used as well.

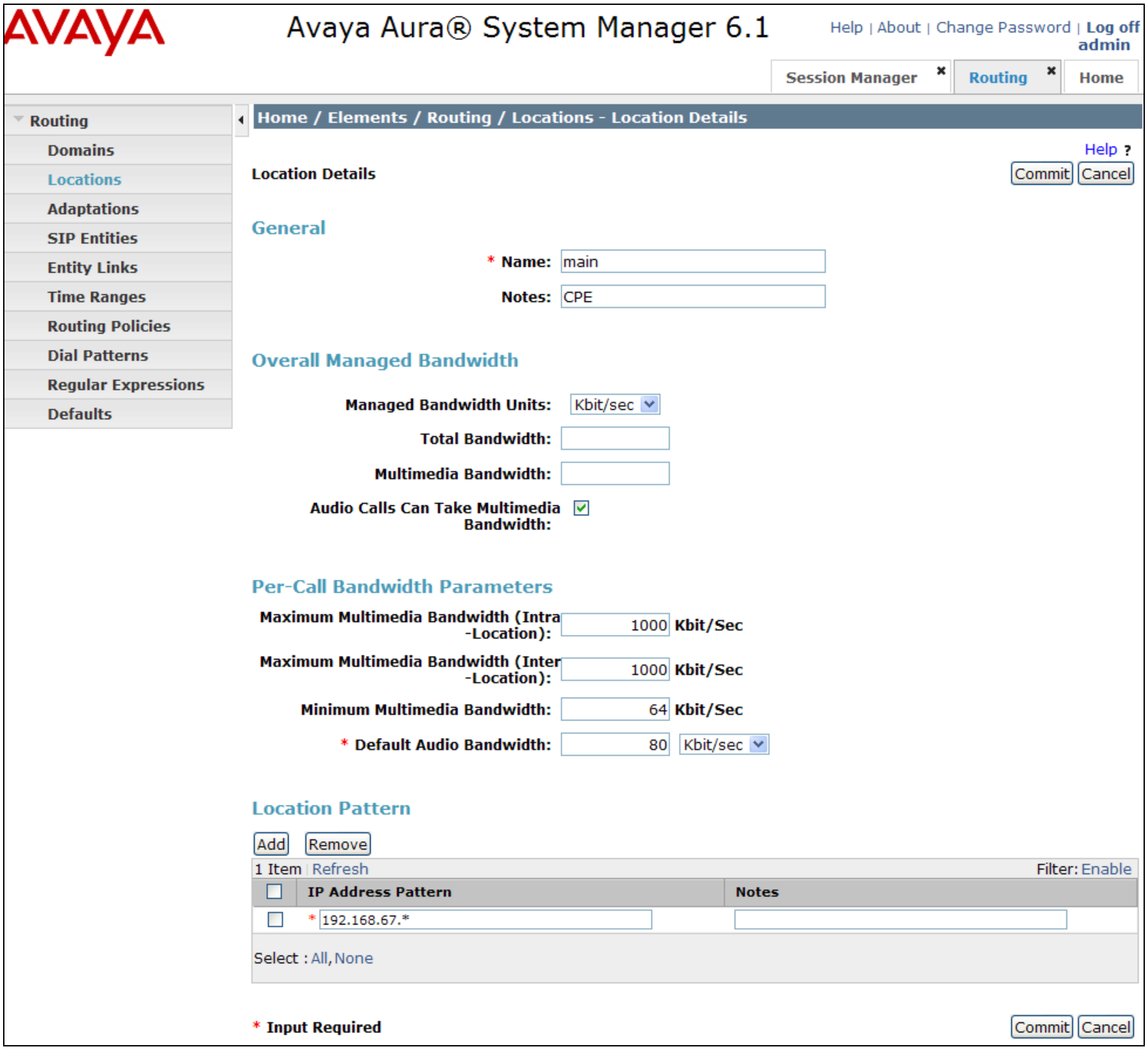
Step 1 - Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern:** Enter the IP address of the CPE subnet (e.g. **192.168.67.***).
- **Notes:** Add a brief description. [Optional]

Step 3 - Click **Commit** to save.



5.3. Configure Adaptations

Session Manager can be configured to use an Adaptation Modules to convert SIP headers in messages sent by AT&T to Communication Manager, and between Communication Manager and Modular Messaging. In the reference configuration the following adaptations were used.

In the reference configuration, Adaptations are administered for the following purposes:

- Calls from AT&T (**Section 5.3.1**) - Modification of SIP messages sent to Communication Manager.
    - The IP address of Session Manager (192.168.67.210) is replaced with the Avaya CPE SIP domain (**customerb.com**) in the Request URI.

- The AT&T called number digit strings in the Request URI are replaced with their associated Communication Manager extensions/VDNs.
- Calls to/from Modular Messaging (**Section 5.3.2**) - Modification of SIP messages sent to and received from Avaya Modular Messaging.
    - From MM (**Section 5.3.2**) – Modular Messaging 11 digit mailbox numbers are converted to the associated Communication Manager 5 digit extensions.

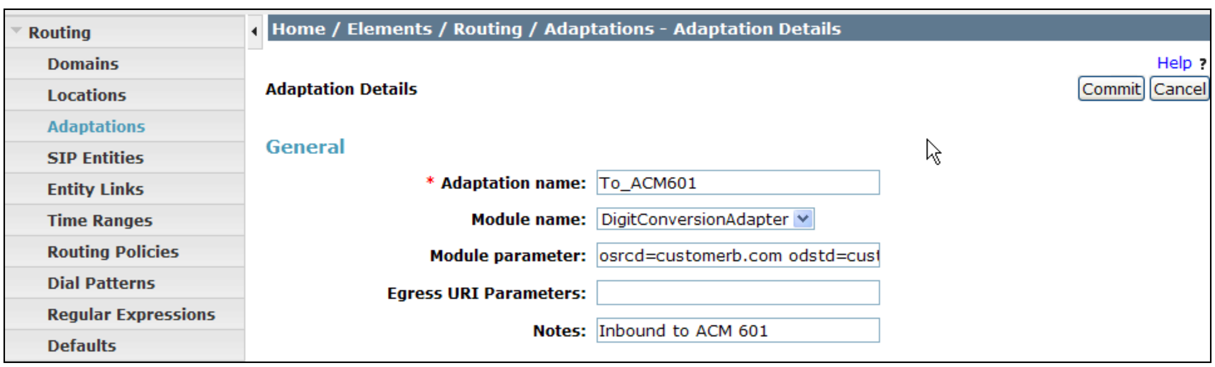
5.3.1. Adaptation for calls to Avaya Aura® Communication Manager

The Adaptation administered in this section is used for modification of SIP messages to Communication Manager from AT&T.

Step 1 - In the left pane under **Routing**, click on **Adaptations**. In the **Adaptations** page, click on **New** (not shown).

Step 2 - In the **Adaptation Details** page, enter:

- A descriptive **Name**, (e.g. **To_ACM601**).
- Select **DigitConversionAdapter** from the **Module Name** drop down menu (if no module name is present, select <click to add module> and enter **DigitConversionAdapter**).
- In the **Module parameter** field enter **odstd=customerb.com osrcd=customerb.com.** The odstd parameter will replace the IP address of Session Manager (*192.168.67.210*) with *customerb.com* in the *inbound* Request URI, and the osrcd parameter will replace the AT&T border element IP address (*135.25.29.74*) with *customerb.com* in the PAI header.



Step 3 – Scroll down to the **Digit Conversion for Outgoing Calls from SM** section (the *inbound* digits from AT&T that need to be replaced with their associated Communication Manager extensions before being sent to Communication Manager).

- Example 1: 0000001049 is a digit string sent in the Request URI by AT&T Toll Free service that is associated with Communication Manager extension 40002.
    - Enter **0000001049** in the **Matching Pattern** column.
    - Enter **10** in the **Min/Max** columns.
    - Enter **10** in the **Delete Digits** column.

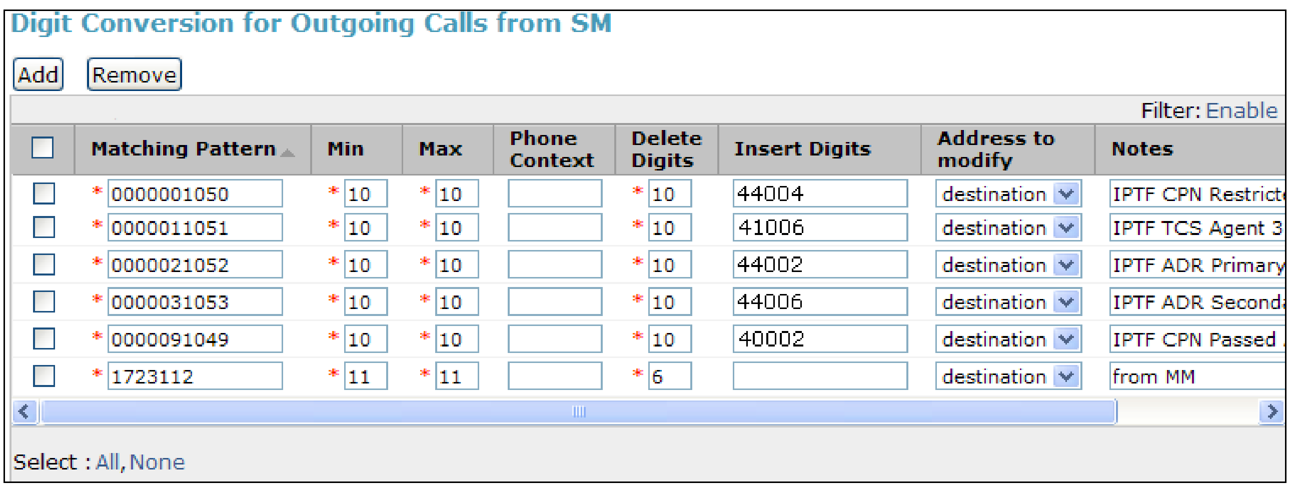
- Enter **40002** string in the **Insert Digits** column.
- Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
- Enter any desired notes.

- Example 2: 1723114xxxx is the format of the mailboxes sent by Avaya Modular messaging in Notify messages (MWI) to Communication Manager. These mailboxes must be converted to their associated Communication Manager extensions by deleting the first six digits.
  - Enter **1723114** in the **Matching Pattern** column.
  - Enter **11** in the **Min/Max** columns.
  - Enter **6** in the **Delete Digits** column.
  - Leave the **Insert Digits** column blank.
  - Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
  - Enter any desired notes.

Step 4 – Repeat **Step 3** for all additional AT&T DID numbers and/or Modular Messaging mailboxes.

Step 5 - Click on **Commit** (not shown).

Note - In the reference configuration no **Digit Conversion for Incoming Calls to SM** were required.


Digit Conversion for Outgoing Calls from SM

Add Remove

Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
	* 0000001050	* 10	* 10		* 10	44004	destination	IPTF CPN Restrict
	* 0000011051	* 10	* 10		* 10	41006	destination	IPTF TCS Agent 3
	* 0000021052	* 10	* 10		* 10	44002	destination	IPTF ADR Primary
	* 0000031053	* 10	* 10		* 10	44006	destination	IPTF ADR Second
	* 0000091049	* 10	* 10		* 10	40002	destination	IPTF CPN Passed
	* 1723112	* 11	* 11		* 6		destination	from MM

Select : All, None

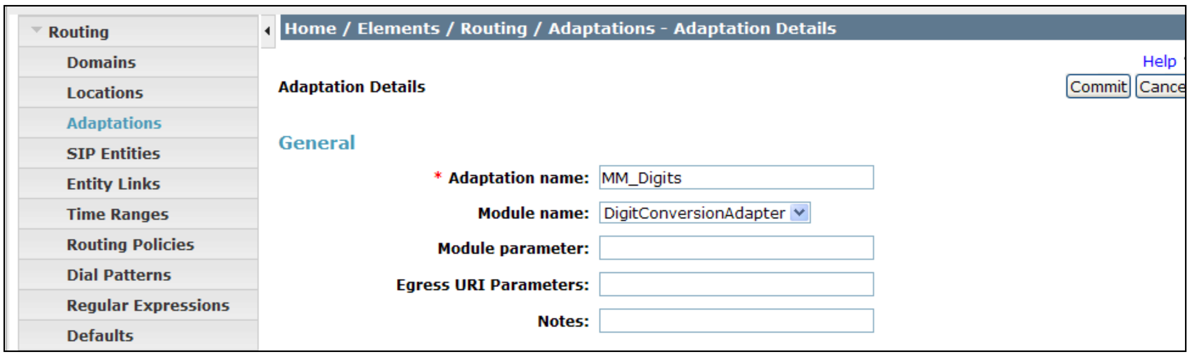
5.3.2. Adaptation for Avaya Modular Messaging

The Adaptation administered in this section is used for digit conversion on SIP messages to and from Avaya Modular Messaging.

Step 1 - In the left pane under **Routing**, click on **Adaptations**. In the **Adaptations** page, click on **New** (not shown).

Step 2 - In the **Adaptation Details** page, enter:

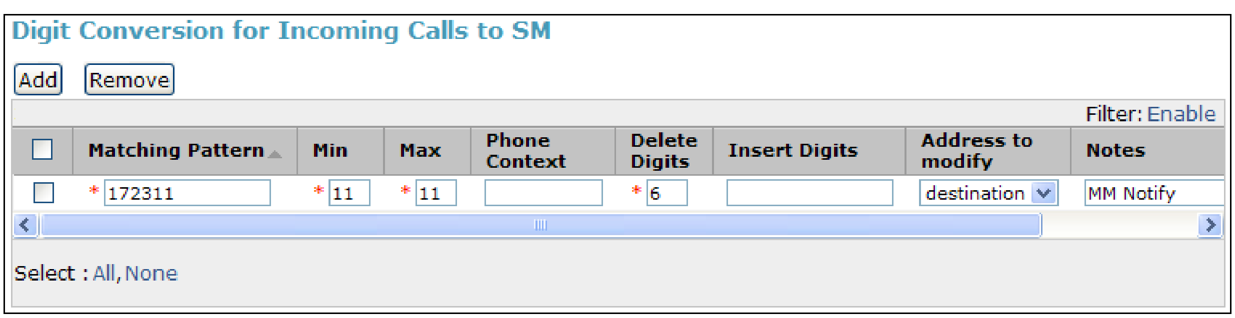
- A descriptive **Name**, (e.g. **MM_Digits**).
- Select **DigitConversionAdapter** from the **Module Name** drop down menu (if no module name is present, select <click to add module> and enter **DigitConversionAdapter**).



Step 3 – Scroll down to the **Digit Conversion for Incoming Calls to SM** section. These are the *inbound* NOTIFY digits Modular Messaging sends to Communication Manager to signal MWI. In the reference configuration, Modular Messaging used 11 digit mailbox numbers (the station extension with a 172311 prefix). This prefix is removed by Session Manager before sending the NOTIFY to Communication Manager.

- Example: 17231140002 is a digit string sent in the NOTIFY by Modular Messaging that is associated with Communication Manager extension 40002.
    - Enter **172311** in the **Matching Pattern** column.
    - Enter **11** in the **Min/Max** columns.
    - Enter **6** in the **Delete Digits** column.
    - Leave the **Insert Digits** column blank.
    - Specify that this should be applied to the SIP **destination** headers in the **Address to modify** column.
    - Enter any desired notes.

Step 4 - Click on **Commit** (not shown).



Note - In the reference configuration no **Digit Conversion for Outgoing Calls from SM** were required.

5.4. SIP Entities

In this section, SIP Entities are administered for the following SIP network elements:

- Session Manager (**Section 5.4.1**).
- Communication Manager, Local and Public access. Two entities are defined to allow two different SIP trunks (public and private) to be defined on Communication Manager. This permits different numbering plans to be administered on each so that the assigned AT&T IP Toll Free DID numbers are presented in the called number fields on the public trunk to AT&T, and local extensions are presented in the called number fields on the local trunk (e.g. coverage to Modular Messaging. See **Section 6.7** for the associated Communication Manager trunk provisioning). In addition, SIP telephones will use the local trunk for intra site calls as well as status signaling to Session Manager.
  - Communication Manager for AT&T access (**Section 5.4.2**) – This entity, and its associated entity link (using port 5080), is for calls from AT&T to Communication Manager via the Avaya Aura® SBC. Note that port 5080 is only used between Communication Manager and Session Manager.
  - Communication Manager for local access (**Section 5.4.3**) – This entity, and associated link (using port 5060), is for communication between Avaya SIP telephones and Communication Manager.
- Avaya Aura® SBC (**Section 5.4.4**) - This entity, and its associated entity link (using port 5060), is for inbound calls from the AT&T IP Toll Free service via the Avaya Aura® SBC.
- Avaya Modular Messaging (**Section 5.4.5**) – This entity, and its associated entity link (using port 5060), is for local calls from Modular Messaging to Communication Manager**.**

Note – In the 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 (port 5061) to be used as transport protocol when possible.

5.4.1. Avaya Aura® Session Manager SIP Entity

Step 1- In the left pane under **Routing**, click on **SIP Entities**. In the **SIP Entities** page click on **New** (not shown).

Step 2 - In the **General** section of the **SIP Entity Details** page, provision the following:

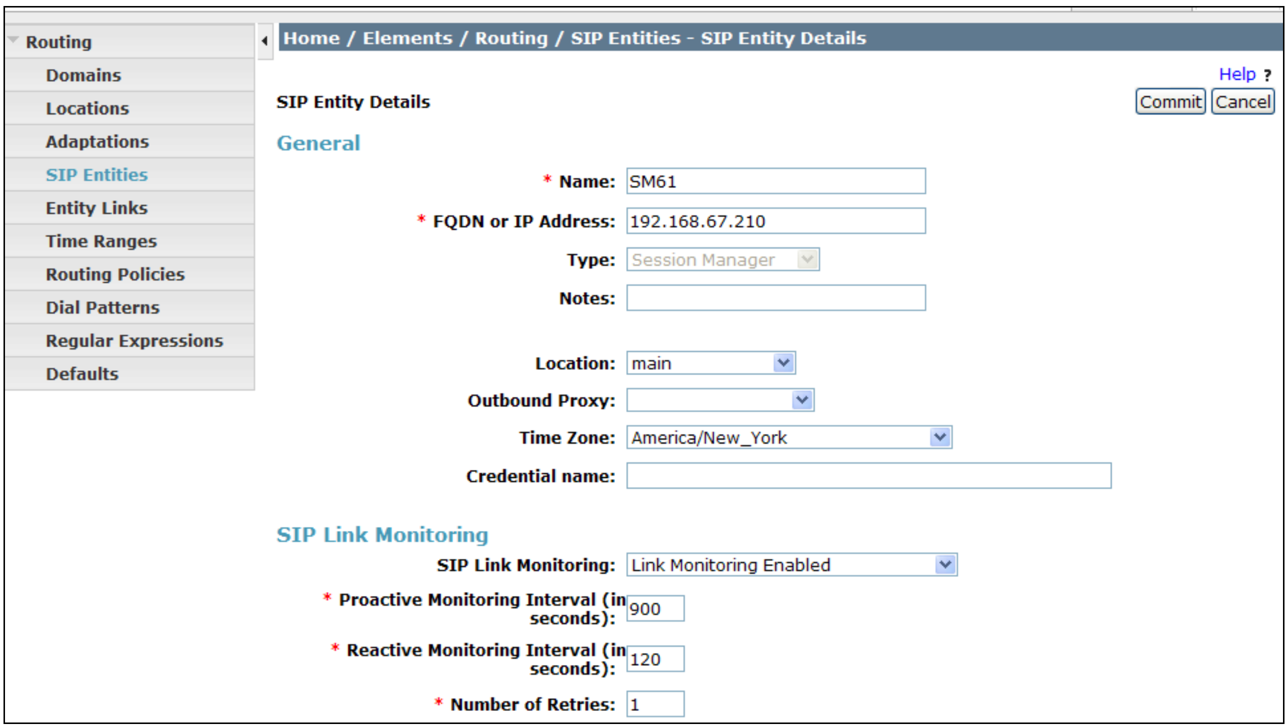
- **Name –** Enter a descriptive name for Session Manager (e.g. **SM61**).
- **FQDN or IP Address –** Enter the IP address of the Session Manager network interface, (*not* the management interface), provisioned during installation (e.g. **192.168.67.210**).
- **Type –** Verify **Session Manager** is selected.
- **Location** – Select location **main** (**Section 5.2.4)**.

- **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
- **Time Zone** – Select the time zone in which Session Manager resides.

Step 3 - In the **SIP Monitoring** section of the **SIP Entity Details** page configure as follows:

- Select **Link Monitoring Enabled** for **SIP Link Monitoring** field.
- Use the default values for the remaining parameters.

These entries enable Session Manager to accept SIP requests on the specified ports/protocols. In addition, Session Manager will accept SIP requests containing the IP address of Session Manager (192.168.67.210) in the host part of the Request-URI.



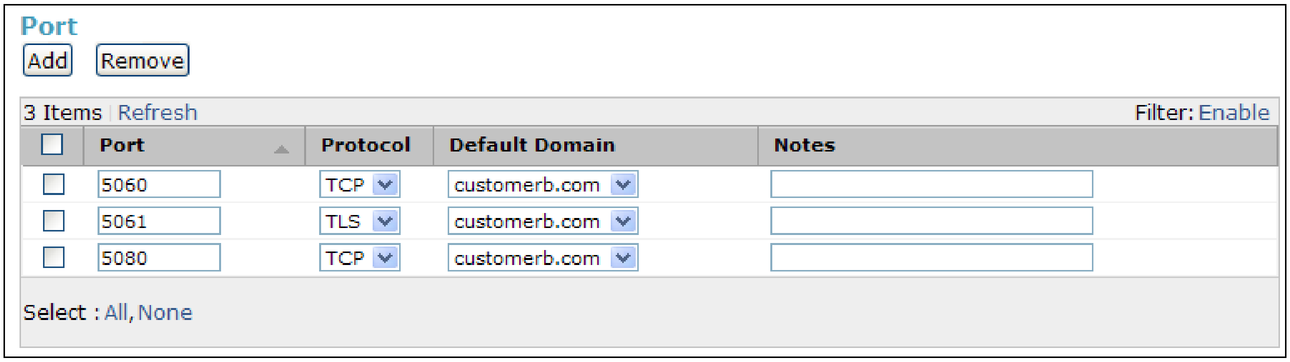
Step 4 - In the **Port** section of the **SIP Entity Details** page, click on **Add** and provision an entry as follows:

- **Port** – Enter **5080** (see note above).
- **Protocol** – Select **TCP** (see note above).
- **Default Domain** – (Optional) Select a SIP domain administered in **Section 5.1** for the selected **Default Domain** field (e.g. **customerb.com**)

Step 5 - Repeat **Step 4** to provision another entry, with **5060** for **Port** and **TCP** for **Protocol**. This is for local calls from the Avaya SIP telephones (and Modular Messaging), to Communication Manager.

Step 6 – Repeat **Step 4** to provision another entry, with **5061** for **Port** and **TLS** for **Protocol.** Although TLS was not used in the reference configuration (see the note at the beginning of this section), the addition of TLS is shown for completeness.

Step 7 - Click on **Commit** (not shown).



Note that the **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**.

5.4.2. Avaya Aura® Communication Manager SIP Entity - Public

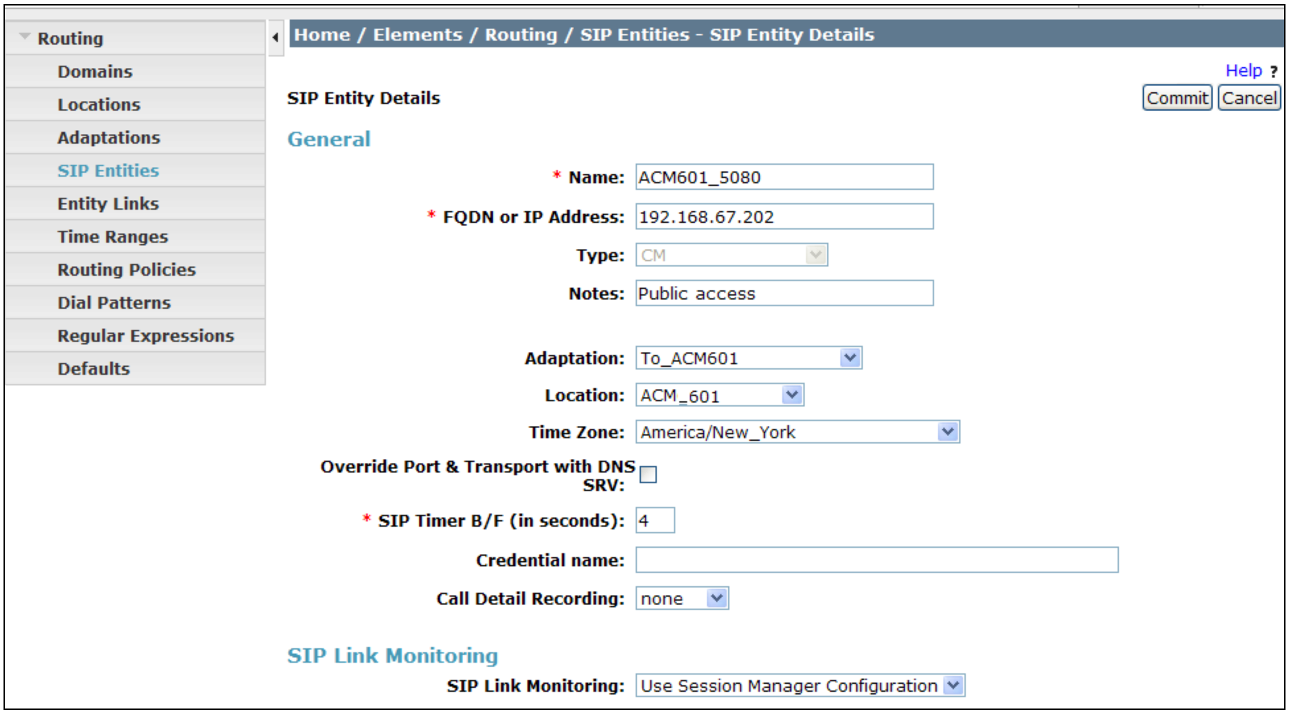
Step 1 - In the **SIP Entities** page, click on **New** (not shown).

Step 2 - In the **General** section of the **SIP Entity Details** page, provision the following:

- **Name –** Enter a descriptive name for the Communication Manager public trunk.
- **FQDN or IP Address –** Enter the IP address of the Communication Manager Processor Ethernet (procr) described in **Section 6.3**.
- **Type** – Select **CM**.
- **Adaptation –** Select the Adaptation administered in **Section 5.3.1**.
- **Location** – Select a Location administered in **Section 5.2.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Monitoring** section of the **SIP Entity Details** page select:
  - Select **Link Monitoring Enabled** for **SIP Link Monitoring** field.
  - Use the default values for the remaining parameters.

Step 3 - Click on **Commit**.

Note that the **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**.



5.4.3. Avaya Aura® Communication Manager SIP Entity – Local.

Configuration for this entity is similar to the entity configured in **Section 5.4.2**.

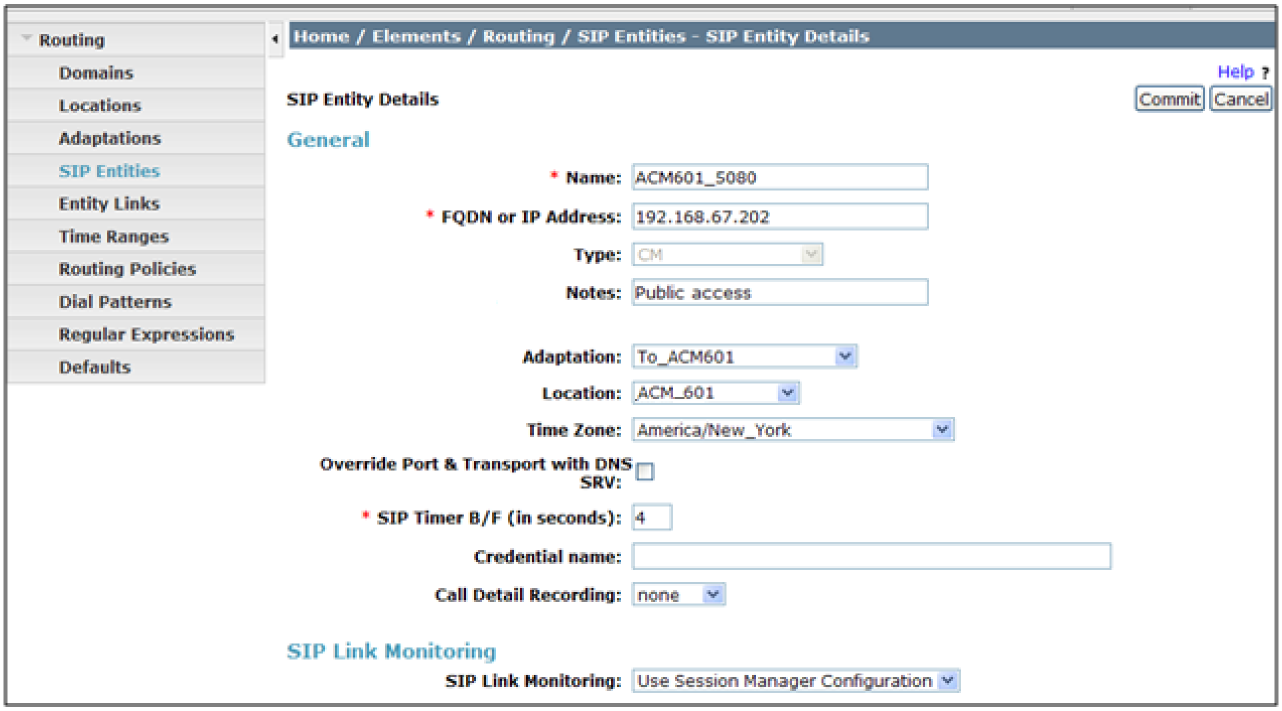
Step 1 - In the **SIP Entities** page, click on **New** (not shown).

Step 2 - In the **General** section of the **SIP Entity Details** page, provision the following:

- **Name –** Enter a descriptive name for the Communication Manager local trunk.
- **FQDN or IP Address –** Enter the IP address of the Communication Manager Processor Ethernet (procr) provisioned in **Section 6.3**.
- **Type –** Select **CM**.
- **Adaptation –** Select the Adaptation administered in **Section 5.3.1**.
- **Location** – Select a Location administered in **Section 5.2.1**.
- **Time Zone** – Select the time zone in which Communication Manager resides.
- In the **SIP Monitoring** section of the **SIP Entity Details** pageprovision the following:
  - Select **Link Monitoring Enabled** for **SIP Link Monitoring** field.
  - Use the default values for the remaining parameters.

Step 3 - Click on **Commit**.

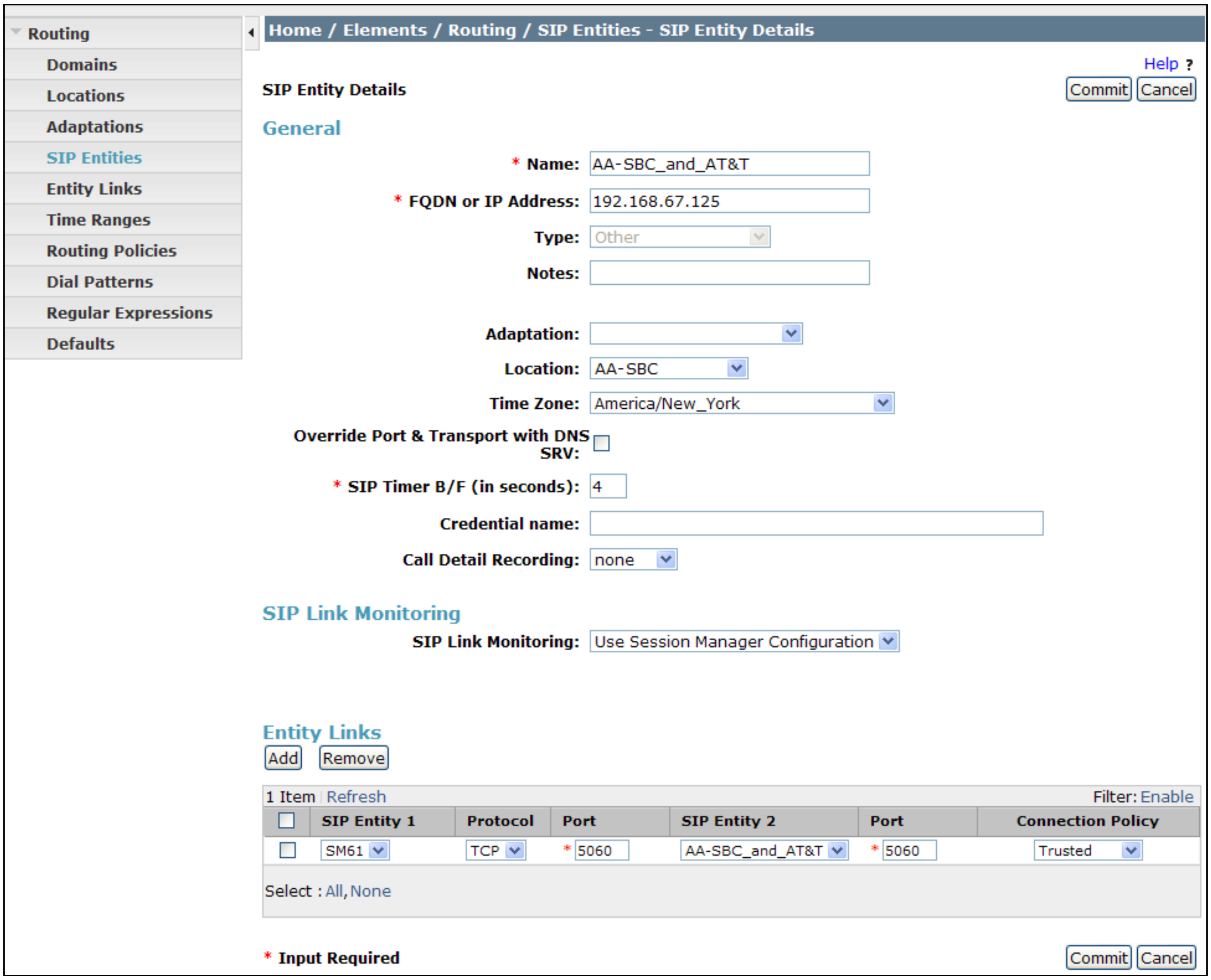
Note that the **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**.



5.4.4. Avaya Aura® SBC SIP Entity

To configure the Avaya Aura® SBC entity, repeat the steps in **Section 5.4.2**. The F**QDN or IP Address** field is populated with the IP address of the private (inside) Avaya Aura® SBC interface configured in **Section 8** and the **Type** field is set to **Other**. See the figure below for the values used in the reference configuration.

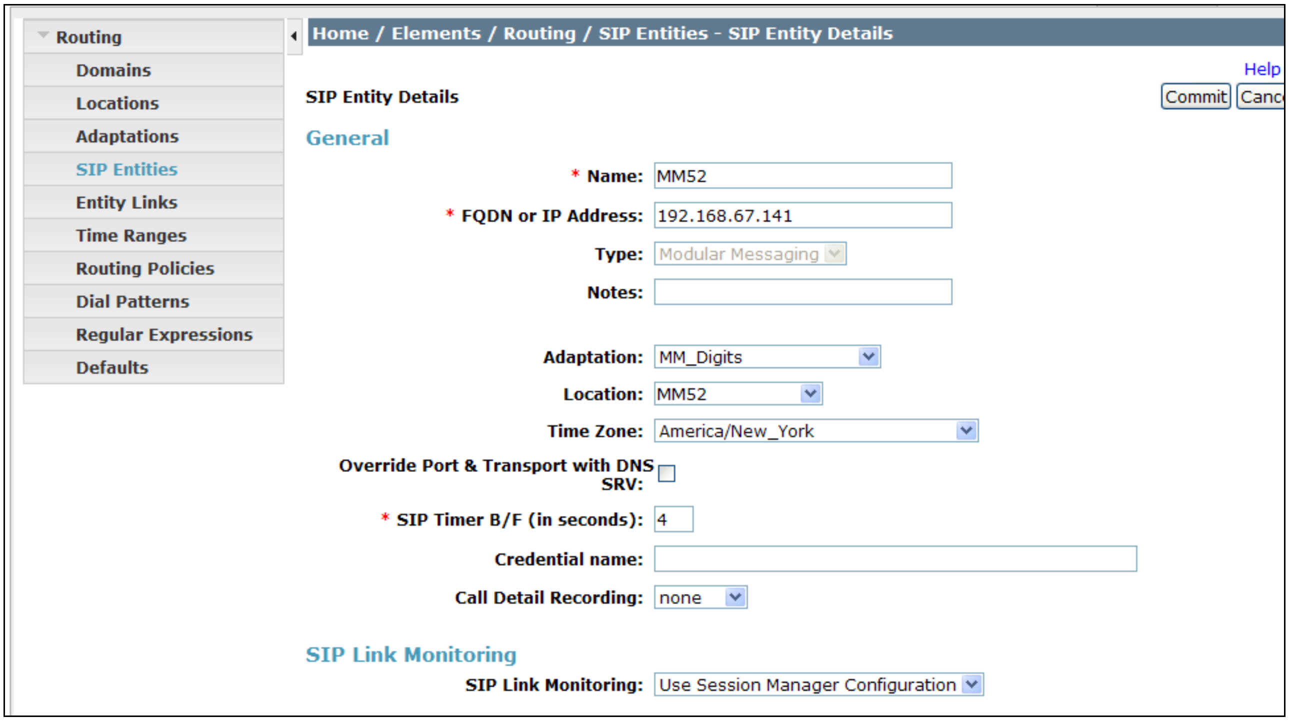
Note that the **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**.



5.4.5. Avaya Modular Messaging SIP Entity

To configure the Modular Messaging SIP entity, repeat the steps in **Section 5.4.2**. The **FQDN or IP Address** field is populated with the IP address of the Modular Messaging Application Server (MAS) and the **Type** field is set to **Other**. See the figure below for the values used in the reference configuration.

Note that the **Entity Links** section of the form (not shown) will be automatically populated when the Entity Links are defined in **Section 5.5**.



5.5. Entity Links

In this section, Entity Links are administered between Session Manager and the following SIP Entities:

- Avaya Aura® Communication Manager – Public (**Section 5.5.1**).
- Avaya Aura® Communication Manager - Local (**Section 5.5.2**).
- Avaya Aura® SBC (**Section 5.5.3**).
- Avaya Modular Messaging (**Section 5.5.4**).

Note – Once the Entity Links have been committed, the link information will also appear on the associated SIP Entity pages configured in **Section 5.4**.

Note – In the reference configuration TCP (port 5060) 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 (port 5061) to be used as transport protocol when possible.

5.5.1. Entity Link to Avaya Aura® Communication Manager - Public

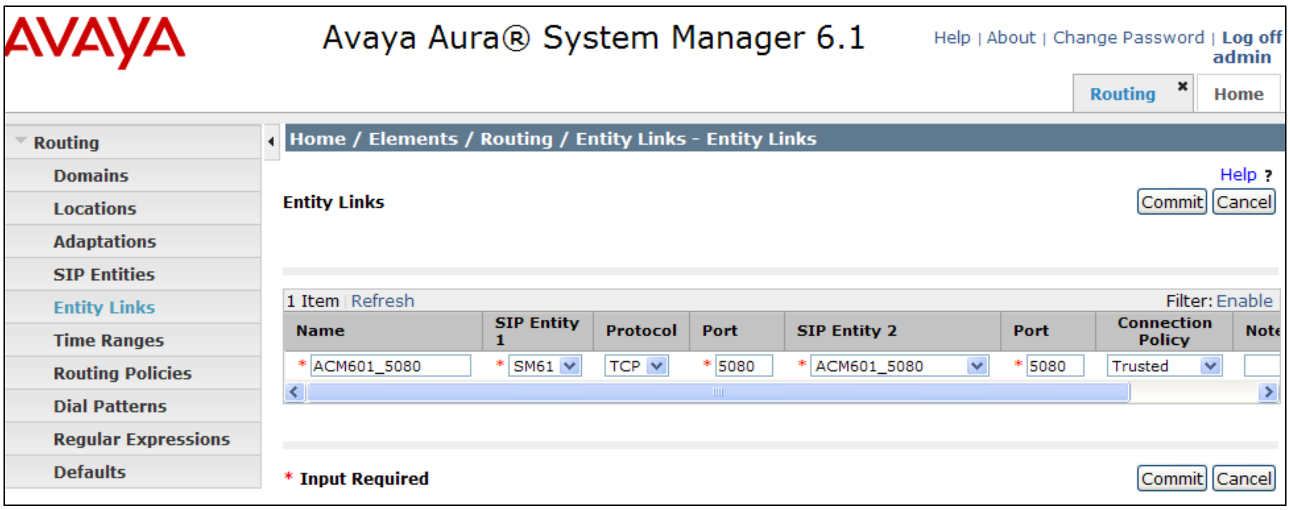
Step 1 - In the left pane under **Routing**, click on **Entity Links**. In the **Entity Links** page click on **New** (not shown).

Step 2 - Continuing in the **Entity Links** page, provision the following:

- **Name** – Enter a descriptive name for this link to Communication Manager (e.g. **ACM601_5080**).

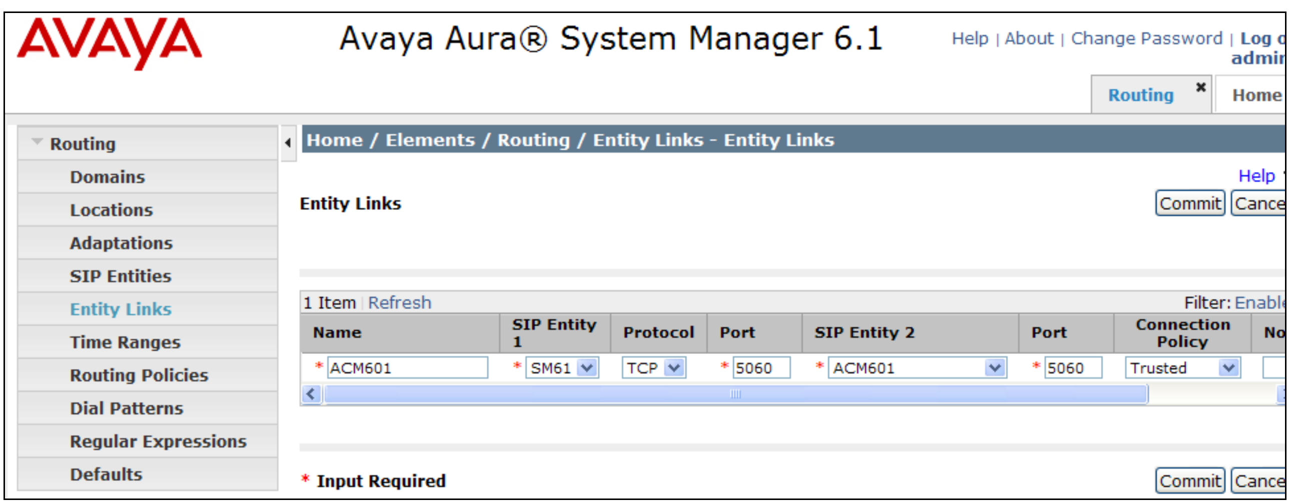
- **SIP Entity 1** – Select the SIP Entity administered in **Section 5.4.1** for Session Manager. SIP Entity 1 must always be a Session Manager instance.
- **SIP Entity 1 Port** – Enter **5080**.
- **SIP Entity 2** –Select the SIP Entity administered in **Section 5.4.2** for the Communication Manager public entity.
- **SIP Entity 2 Port** - Enter **5080**.
- **Trusted** – Select **Trusted**.
- **Protocol** – Select **TCP**.

Step 3 - Click on **Commit**.



5.5.2. Entity Link to Avaya Aura® Communication Manager Entity - Local

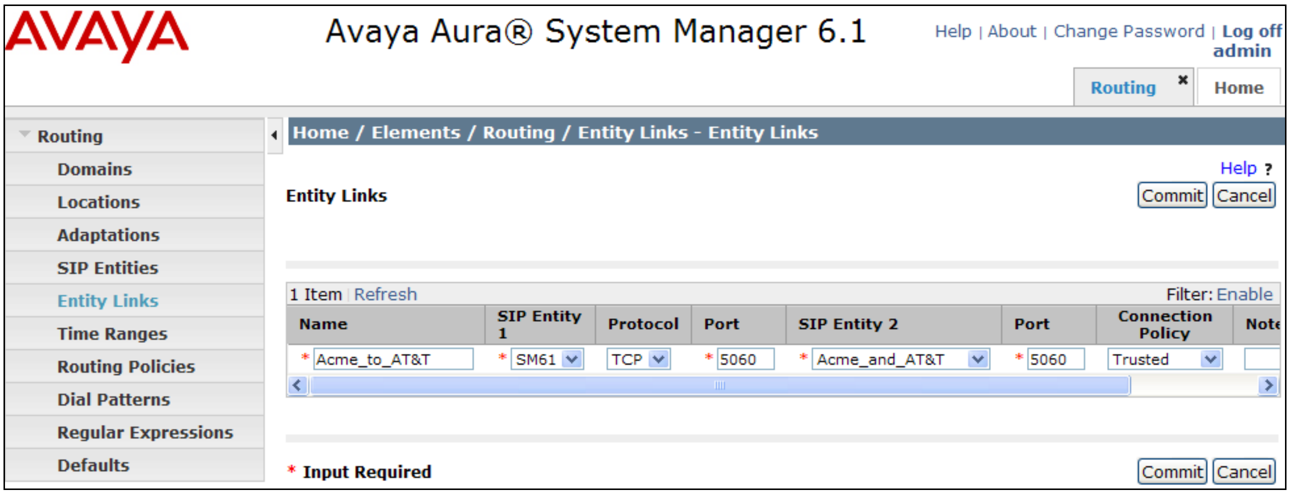
To configure this entity link, repeat the steps in **Section 5.5.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.4.3** for Communication Manager local Entity (e.g. **ACM601**). Note that the **Port** fields are populated with **5060**. See the figure below for the values used in the reference configuration.



5.5.3. Entity Link to AT&T IP Toll Free Service via Avaya Aura® SBC

Repeat the steps in **Section 5.5.1** with the following differences:

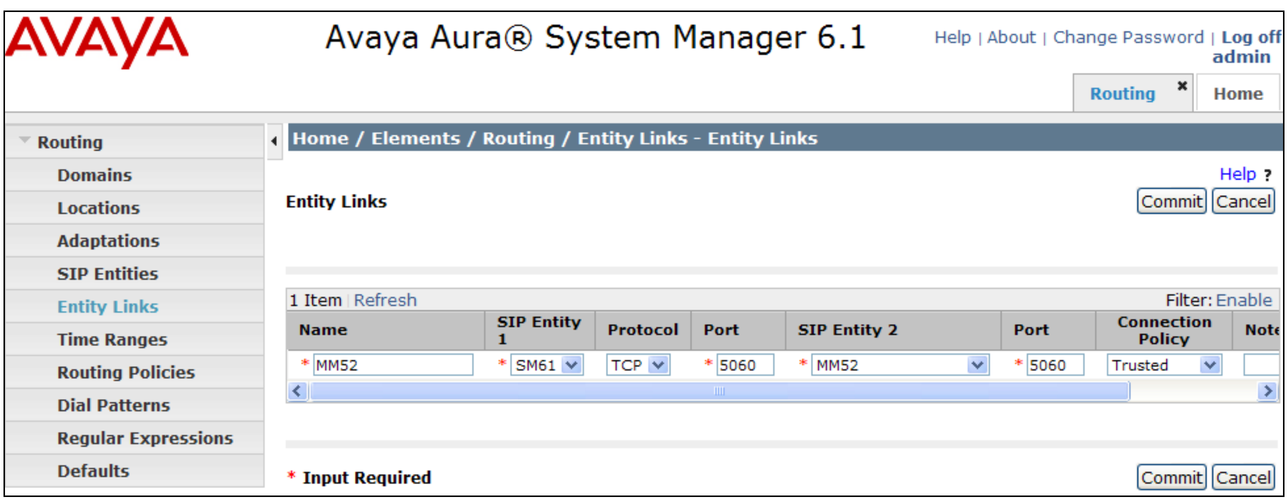
- **Name** – Enter a descriptive name for the link to the AT&T IP Toll Free service, by way of the Avaya Aura® SBC.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.4** for the Avaya Aura® SBC.



5.5.4. Entity Link to Avaya Modular Messaging

Repeat the steps in **5.5.1** with the following differences:

- **Name** – Enter a descriptive name for the link to Avaya Modular Messaging.
- **SIP Entity 2** – Select the SIP Entity administered in **Section 5.4.5** for Avaya Modular Messaging.



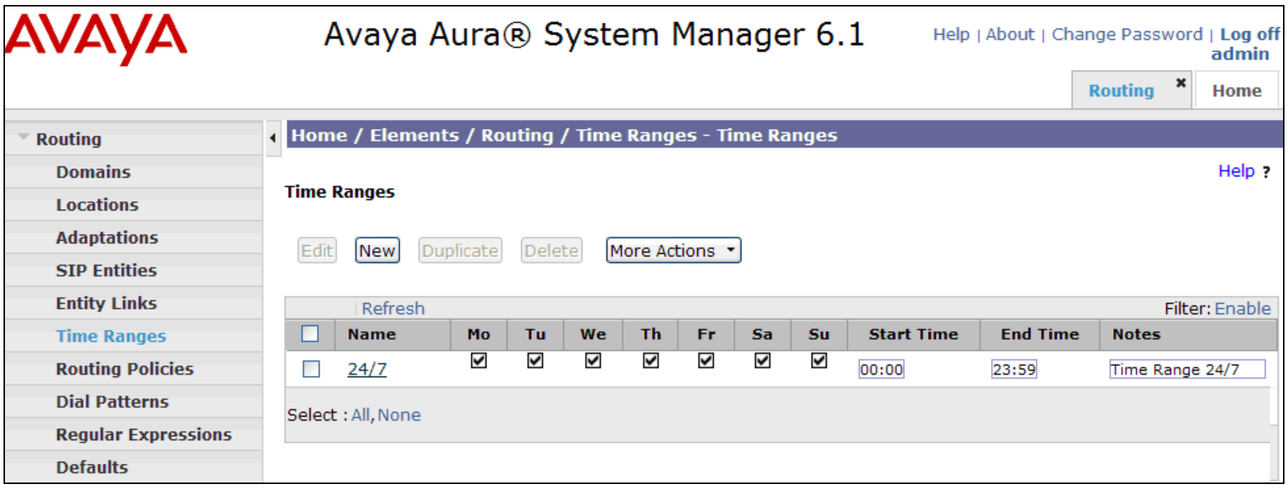
5.6. Time Ranges

Step 1 - In the left pane under **Routing**, click on **Time Ranges**. In the **Time Ranges** page click on **New** (not shown).

Step 2 - Continuing in the **Time Ranges** page, enter a descriptive **Name**, check the checkboxes for the desired day(s) of the week, and enter the desired **Start Time** and **End Time**.

Step 3 - Click on **Commit**.

Step 4 - Repeat **Steps 1 – 3** to provision additional time ranges.



5.7. Routing Policies

In this section, the following Routing Policies are administered:

- AT&T calls to Avaya Aura® Communication Manager (**Section 5.7.1**).
- Avaya Modular Messaging MWI notification to Avaya Aura® Communication Manager (**Section 5.7.2**).
- Avaya Aura® Communication Manager calls to Avaya Modular Messaging for call coverage (**Section 5.7.3**)

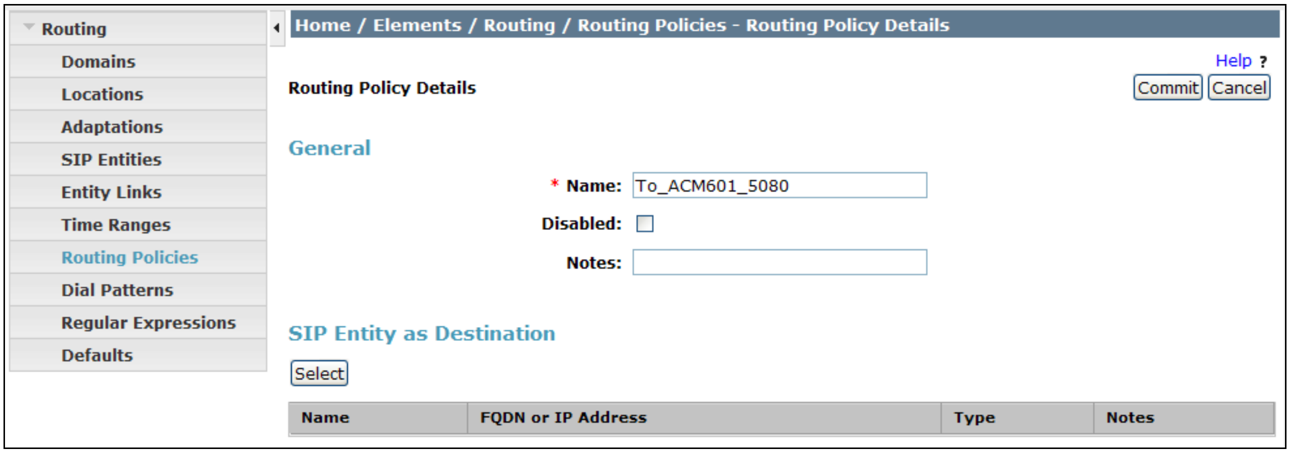
5.7.1. Routing Policy for Routing to Avaya Aura® Communication Manager from AT&T

Note that this routing policy will use the public SIP Entity **ACM601_5080**.

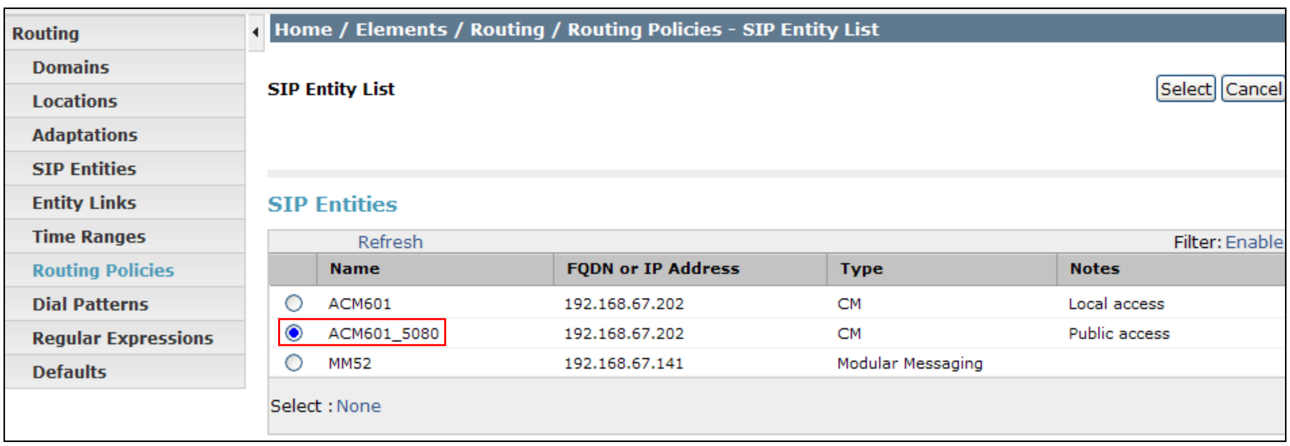
Step 1 - In the left pane under **Routing**, click on **Routing Policies**. In the **Routing Policies** page click on **New** (not shown).

Step 2 - In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing AT&T calls to Communication Manager (e.g. **To_ACM601_5080**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

Step 3 - In the **SIP Entity as Destination** section of the **Routing Policy Details** page, click on **Select** and the SIP Entity list page will open.



Step 4 - In the **SIP Entity List** page, select the SIP Entity administered in **Section 5.4.2** for Communication Manager (**ACM601_5080**), and click on **Select**.



Step 5 - Returning to the Routing Policy Details page in the Time of Day section, click on Add.

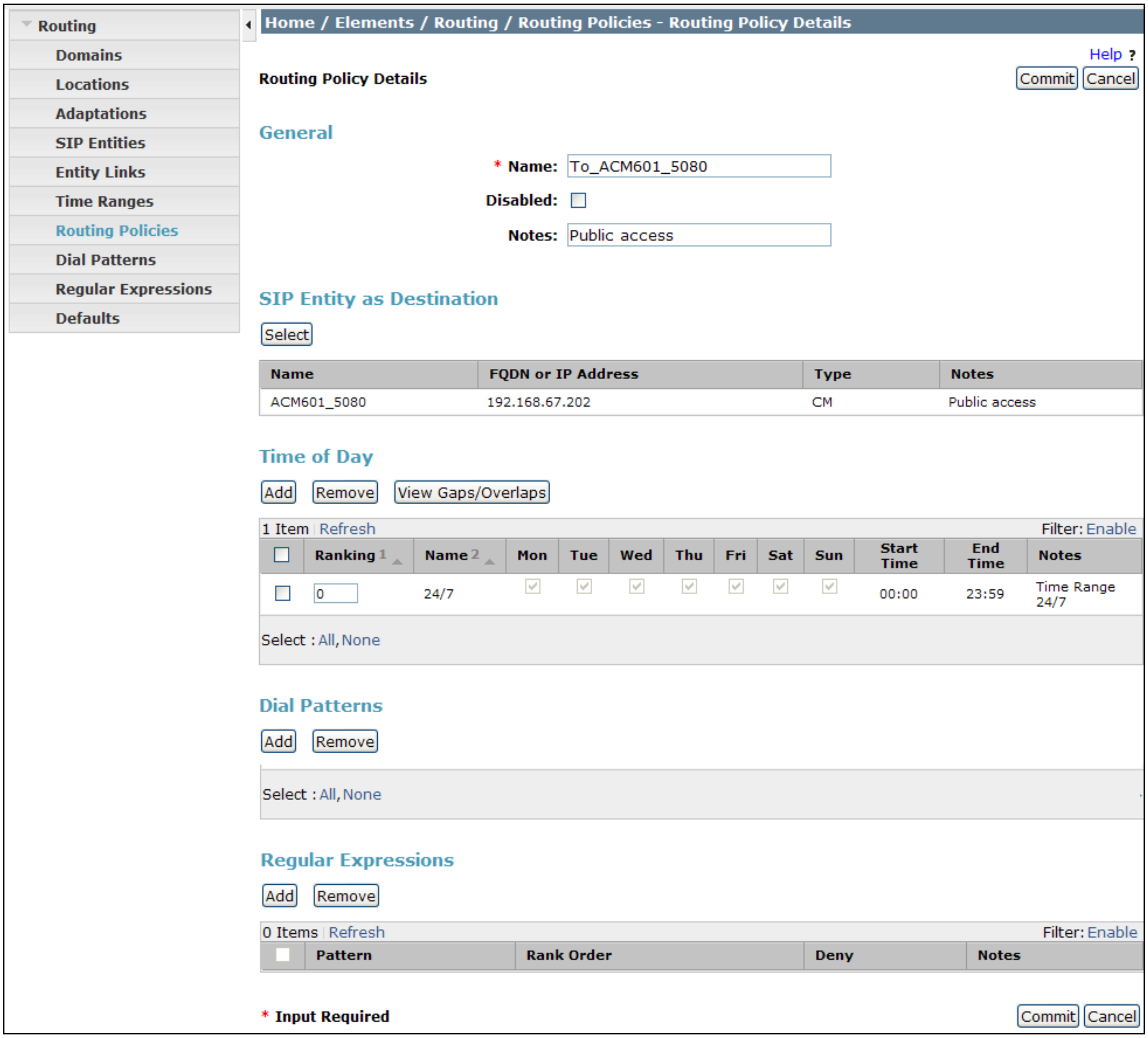
Step 6 - In the **Time Range List** page (not shown), check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 5.6**, and click on **Select**.

Step 7 - Returning to the **Routing Policy Details** page in the **Time of Day** section, if multiple Time Ranges were selected, user may enter a **Ranking** (the lower the number, the higher the ranking) for each Time Range, and click on **Commit**.

Step 8 - Note that once the **Dial Patterns** are defined (**Section 5.8**) they will appear in the **Dial Pattern** section of this form.

Step 9 - No **Regular Expressions** were used in the reference configuration.

Step 10 - Click on **Commit**.



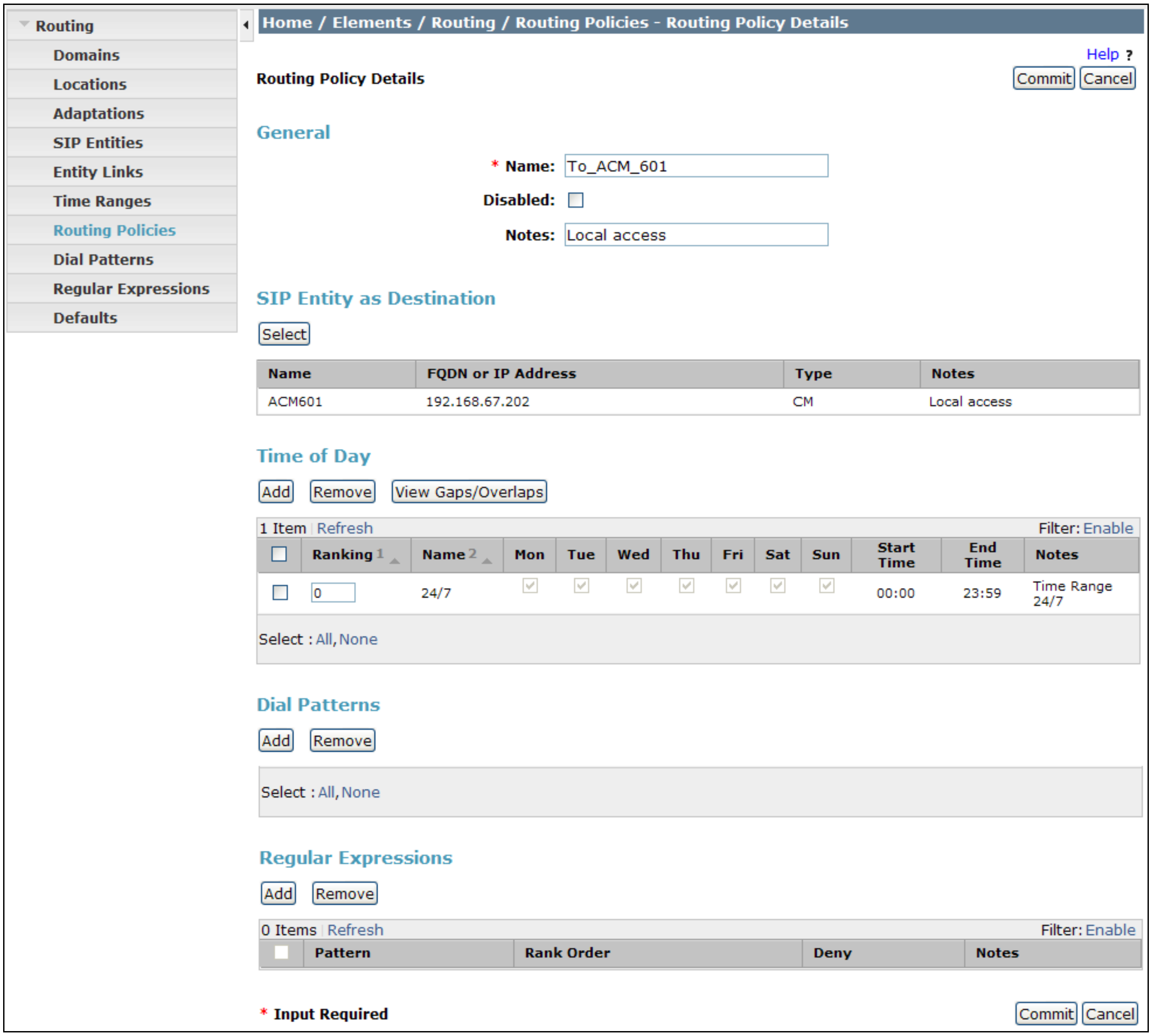
5.7.2. Routing Policy for Routing from Avaya Modular Messaging (MWI) to Avaya Aura® Communication Manager

Note that this routing policy will use the local SIP Entity **ACM601**.

Repeat the steps in **Section 5.7.1** with the following differences:

- In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing local calls to Communication Manager (**To_ACM610**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

- In the **SIP Entity List** page, select the SIP Entity administered in **Section 5.4.3** for Communication Manager (**ACM610**), and click on **Select**.
- Note that once the **Dial Patterns** are defined (**Section 5.8**), they will appear in the **Dial Pattern** section.

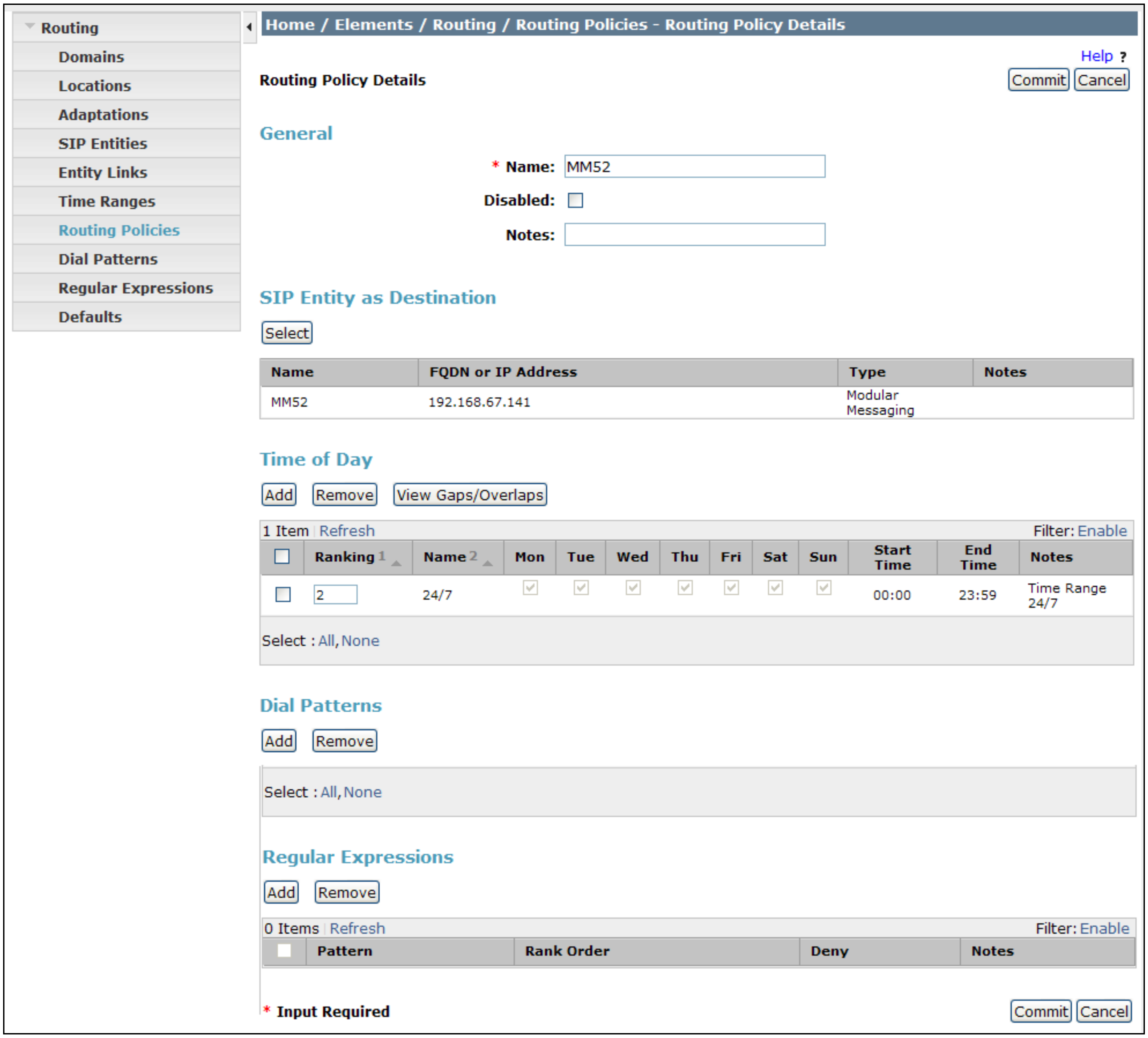


5.7.3. Routing Policy for Routing to Avaya Modular Messaging (Call Coverage) from Avaya Aura® Communication Manager

Repeat **Section 5.7.1** with the following differences:

- In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing calls to Avaya Modular Messaging (**MM52**), and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.

- In the **SIP Entity List** page, select the SIP Entity administered in **Section 5.4.5** for Avaya Modular Messaging (**MM52**), and click on **Select**.
- Note that once the **Dial Patterns** are defined (**Section 5.8**), they will appear in the **Dial Pattern** section.



5.8. Dial Patterns

In this section, Dial Patterns are administered matching the following calls:

- Inbound PSTN calls via AT&T IP Toll Free service to Communication Manager.
- Call Coverage/retrieval calls to Modular Messaging from Communication Manager to the Modular Messaging pilot number.

- Notifications from Avaya Modular Messaging (MWI) to Communications Manager 5 digit local extensions.

5.8.1. Matching Inbound PSTN Calls to Avaya Aura® Communication Manager

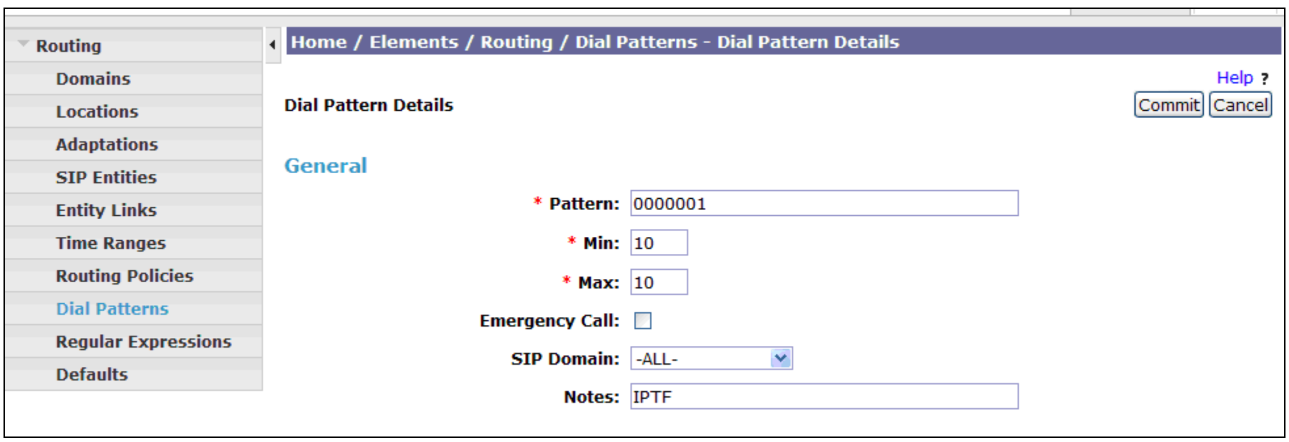
In the reference configuration inbound calls from the AT&T IP Toll Free service used the called digit pattern 0000001xxx in the SIP Request URI. This pattern is matched for further call processing.

Note – Be sure to match on the digit string specified in the Request URI, not the digit string that was dialed. They may be different.

Step 1 - In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** (not shown).

Step 2 - In the **General** section of the **Dial Pattern Details** page, provision the following:

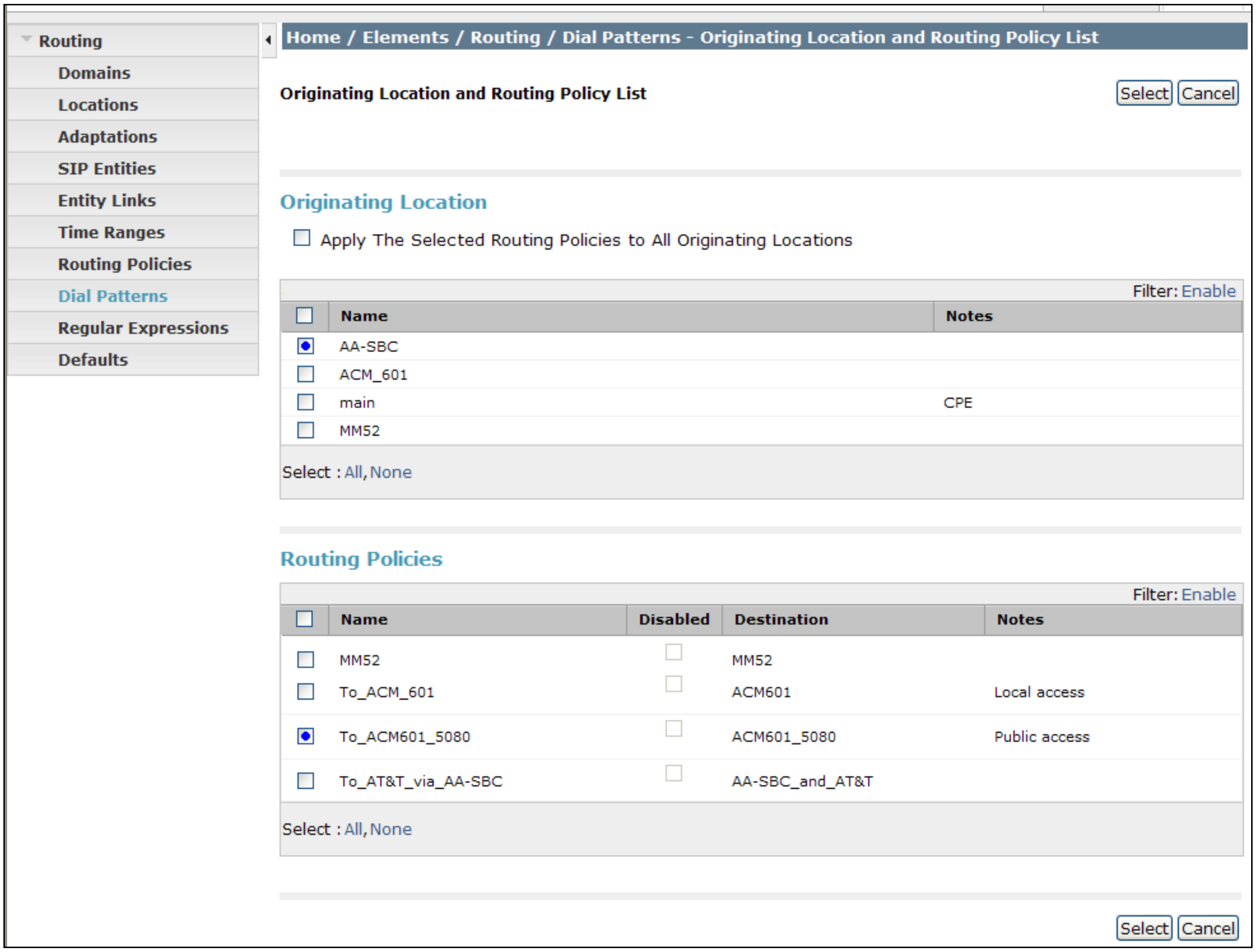
- **Pattern** – In the reference configuration, AT&T sends a 10 digit number in the Request URI with the format 0000001xxx. Enter **0000001**. Note - The adaptation defined for Communication Manager in **Section 5.3.1** will convert the various 0000001xxx numbers into their corresponding extensions.
- **Min** and **Max** – Enter **10**.
- **SIP Domain** – Select one of the SIP Domains defined in **Section 5.1** or **-ALL-**, to select all of those administered SIP Domains. Only those calls with the same domain in the Request-URI as the selected SIP Domain (or all administered SIP Domains if **-ALL-** is selected) can match this Dial Pattern.



Step 3 - In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page (not shown), click on **Add**.

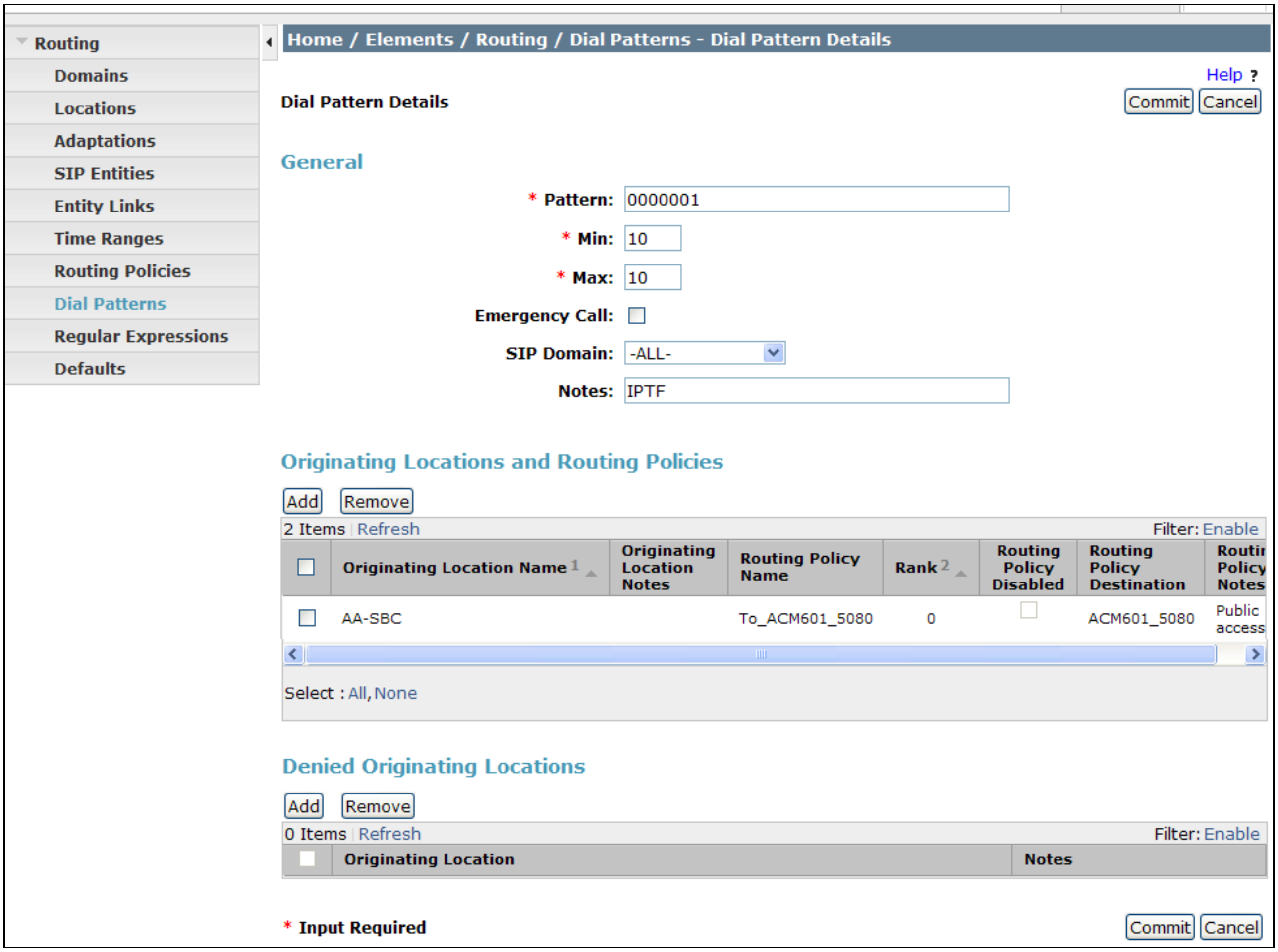
Step 4 - In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Location **AA-SBC** (see **Section 5.2.2**). Note that only those calls that originate from the selected Location(s), or all administered Locations if **-ALL-** is selected, can match this Dial Pattern.

Step 5 - In the **Routing Policies** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Routing Policy administered for routing calls to the Communication Manager Public trunk in **Section 5.7.1**.



Step 6 - In the **Originating Location and Routing Policy List** page, click on **Select**.

Step 7 - Returning to the **Dial Pattern Details** page click on **Commit**.



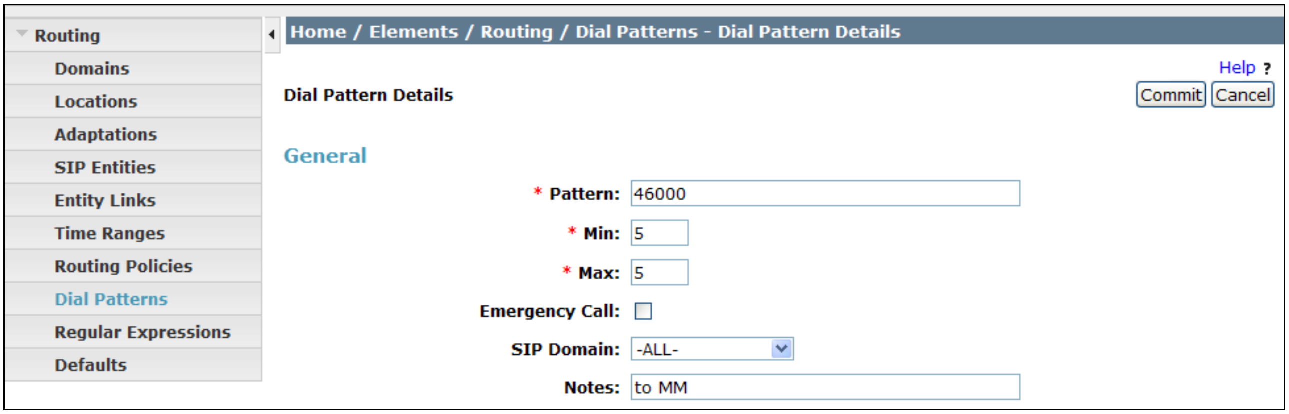
5.8.2. Matching Inbound Calls to Avaya Modular Messaging Pilot Number via Avaya Aura® Communication Manager

Communication Manager stations cover to Avaya Modular Messaging using a pilot extension (46000 in the reference configuration). Additionally stations may dial this pilot extension to retrieve messages or modify mailbox settings.

Step 1 - In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** (not shown).

Step 2 - In the **General** section of the **Dial Pattern Details** page, provision the following:

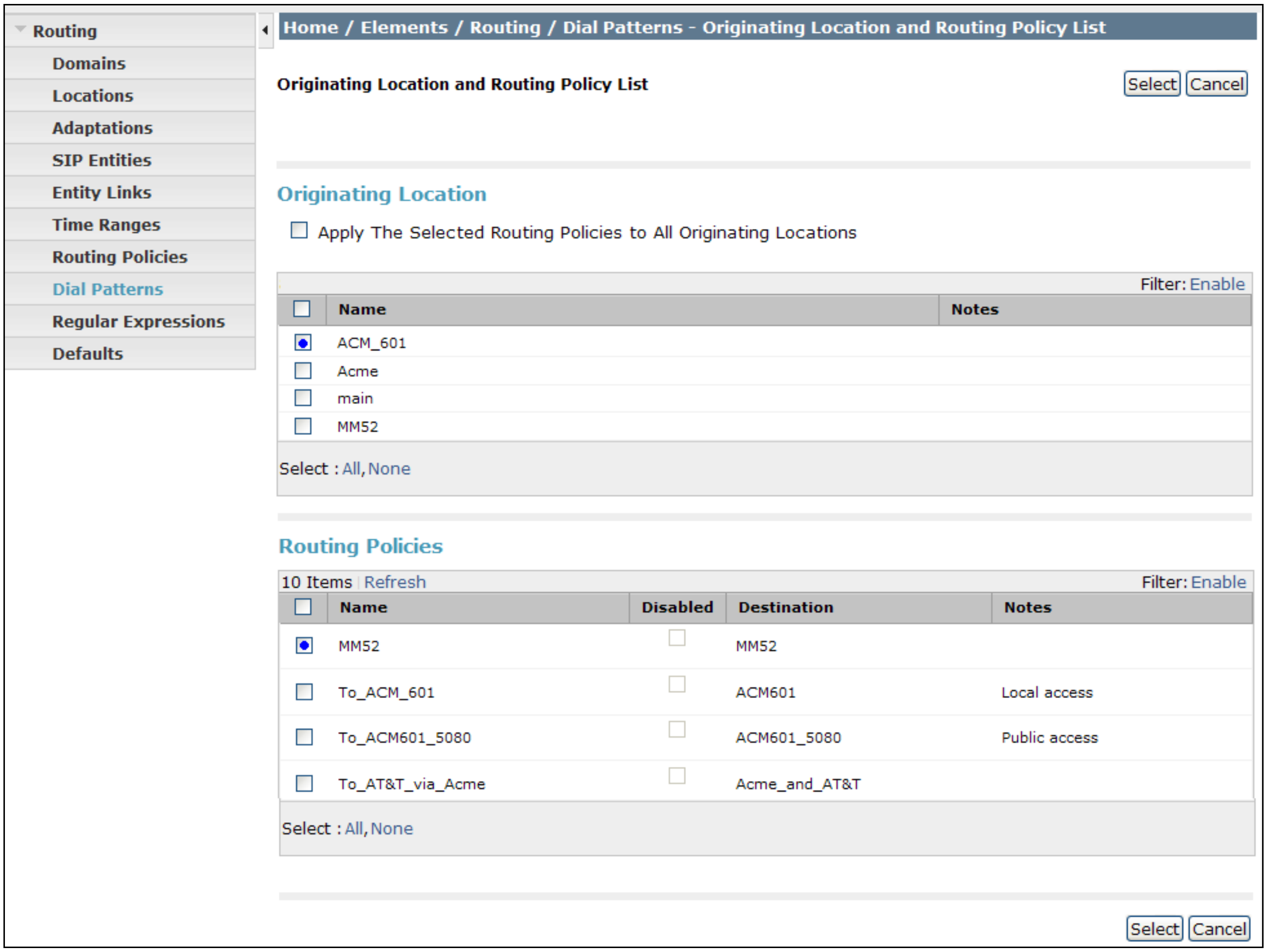
- **Pattern** – Enter the Avaya Modular Messaging pilot extension (e.g. **46000**)
- **Min** and **Max** – Enter **5**.
- **SIP Domain** – Select one of the SIP Domains defined in **Section 5.1** or **-ALL-**, to select all of those administered SIP Domains. Only those calls with the same domain in the Request-URI as the selected SIP Domain (or all administered SIP Domains if **-ALL-** is selected) can match this Dial Pattern.



Step 3 - In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page (not shown), click on **Add**.

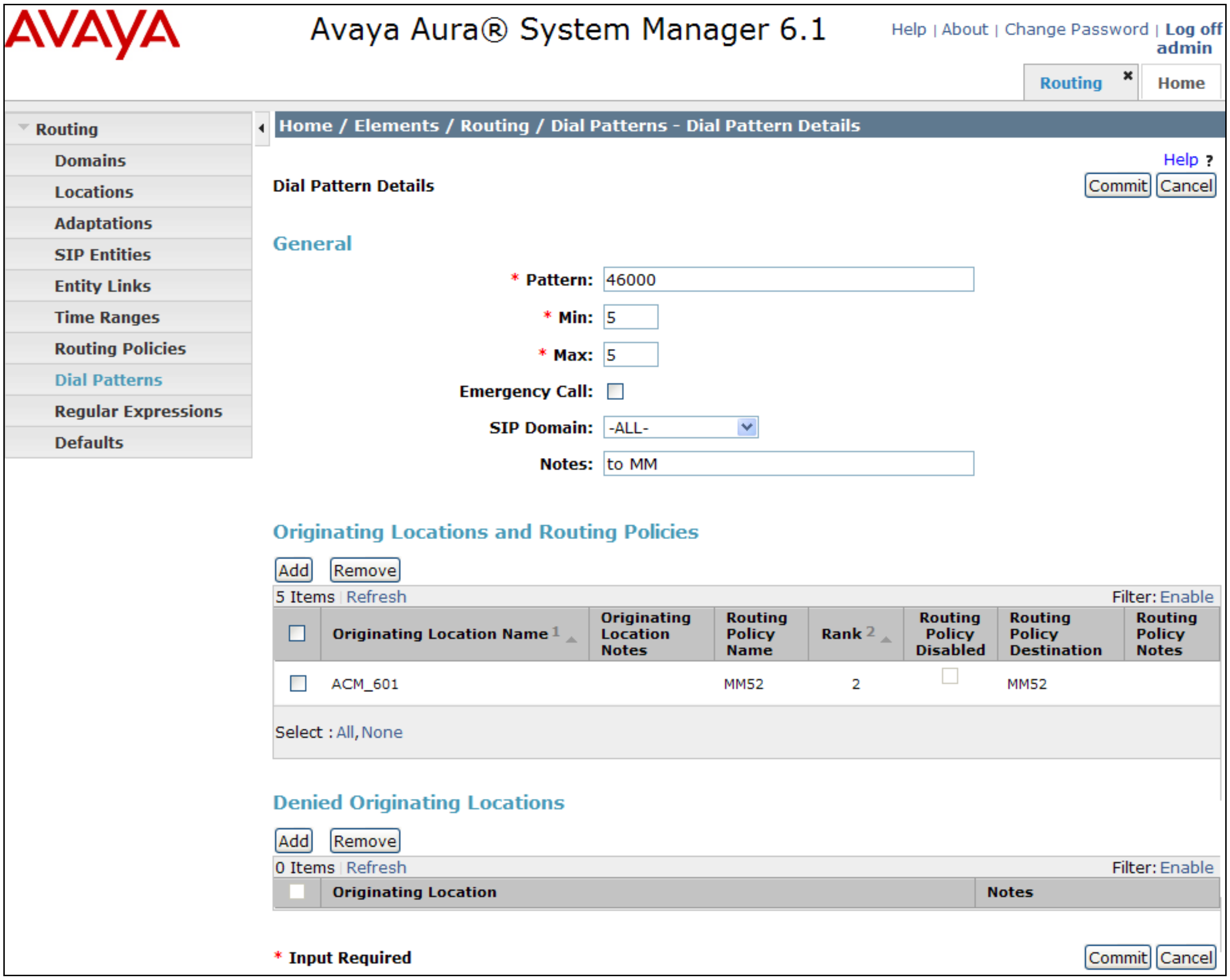
Step 4 - In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Location **ACM_601** (see **Section 5.2.1**). Note that only the calls that originate from the selected Location(s), or all administered Locations if **-ALL-** is selected, can match this Dial Pattern.

Step 5 - In the **Routing Policies** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Routing Policy **MM52** administered for routing calls to Modular Messaging in **Section 5.7.3**.



Step 6 - In the **Originating Location and Routing Policy List** page, click on **Select**.

Step 7 - Returning to the **Dial Pattern Details** page click on **Commit**.



5.8.3. Matching Inbound Calls to Avaya Aura® Communication Manager from Avaya Modular Messaging (MWI Notify).

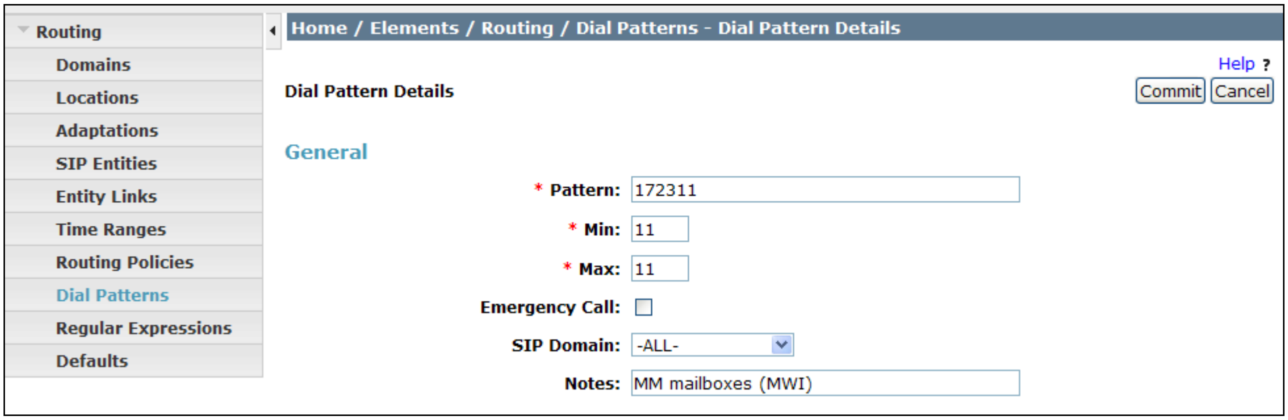
Avaya Modular Messaging will send SIP Notify messages to Communication Manager stations to indicate waiting messages (MWI). In the reference configuration, Modular Messaging uses 11 digit mailboxes. These 11 digit mailboxs use the format **172311xxxxx** where xxxxx is the Communication Manager extension. Note that these 11 digits are converted to the Communication Manager 5 digit extension in the Modular Messaging Adaptation defined in **Section 5.3.2**.

Step 1 - In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** (not shown).

Step 2 - In the **General** section of the **Dial Pattern Details** page, provision the following:

- **Pattern** – Enter the first 6 digits of the Modular Messaging mailbox number format (e.g. 1**72311**)
- **Min** and **Max** – Enter **11**.

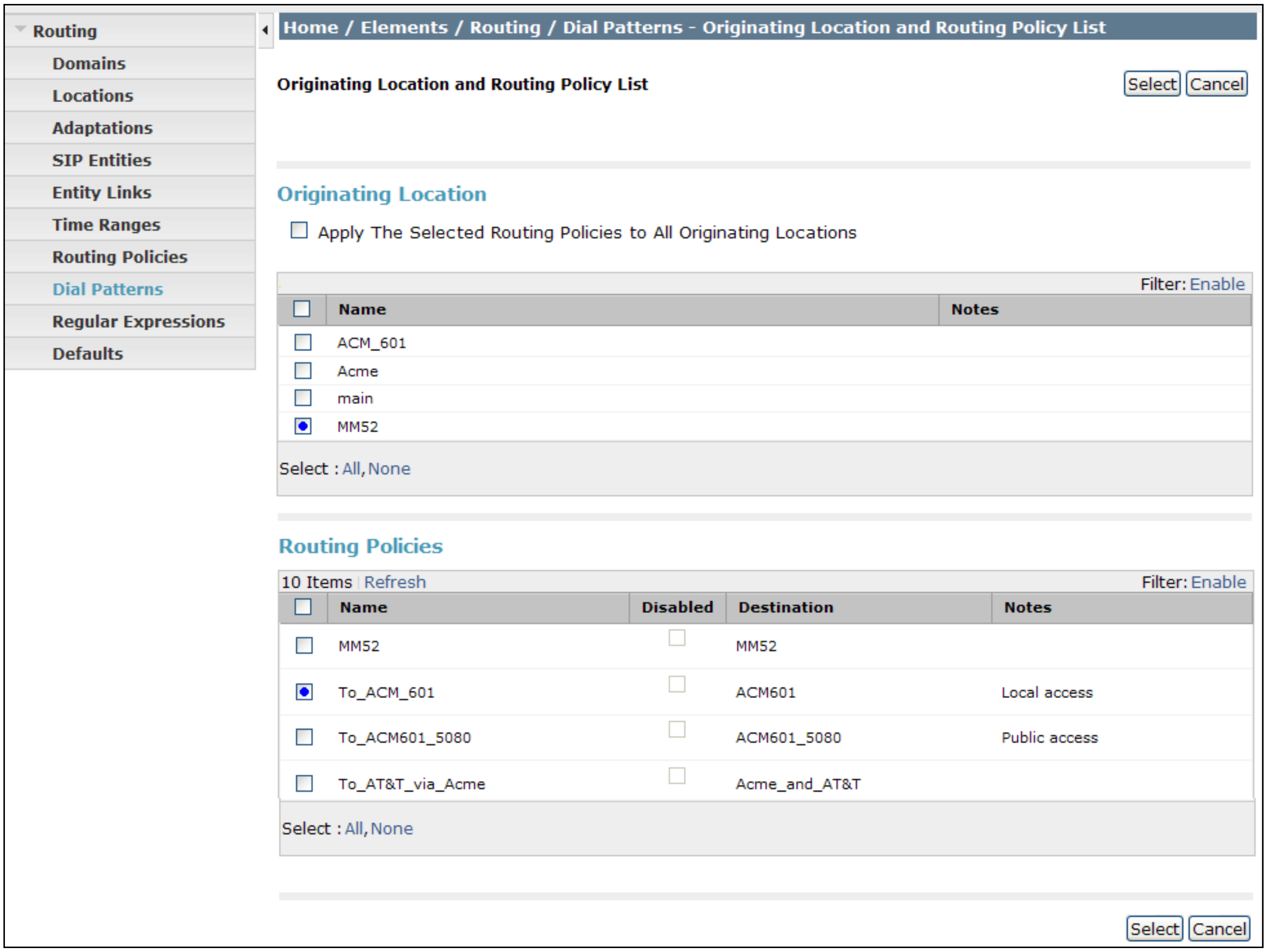
- **SIP Domain** – Select one of the SIP Domains defined in **Section 5.1** or **-ALL-**, to select all of those administered SIP Domains. Only the calls with the same domain in the Request-URI as the selected SIP Domain (or all administered SIP Domains if **-ALL-** is selected) can match this Dial Pattern.



Step 3 - In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page (not shown), click on **Add**.

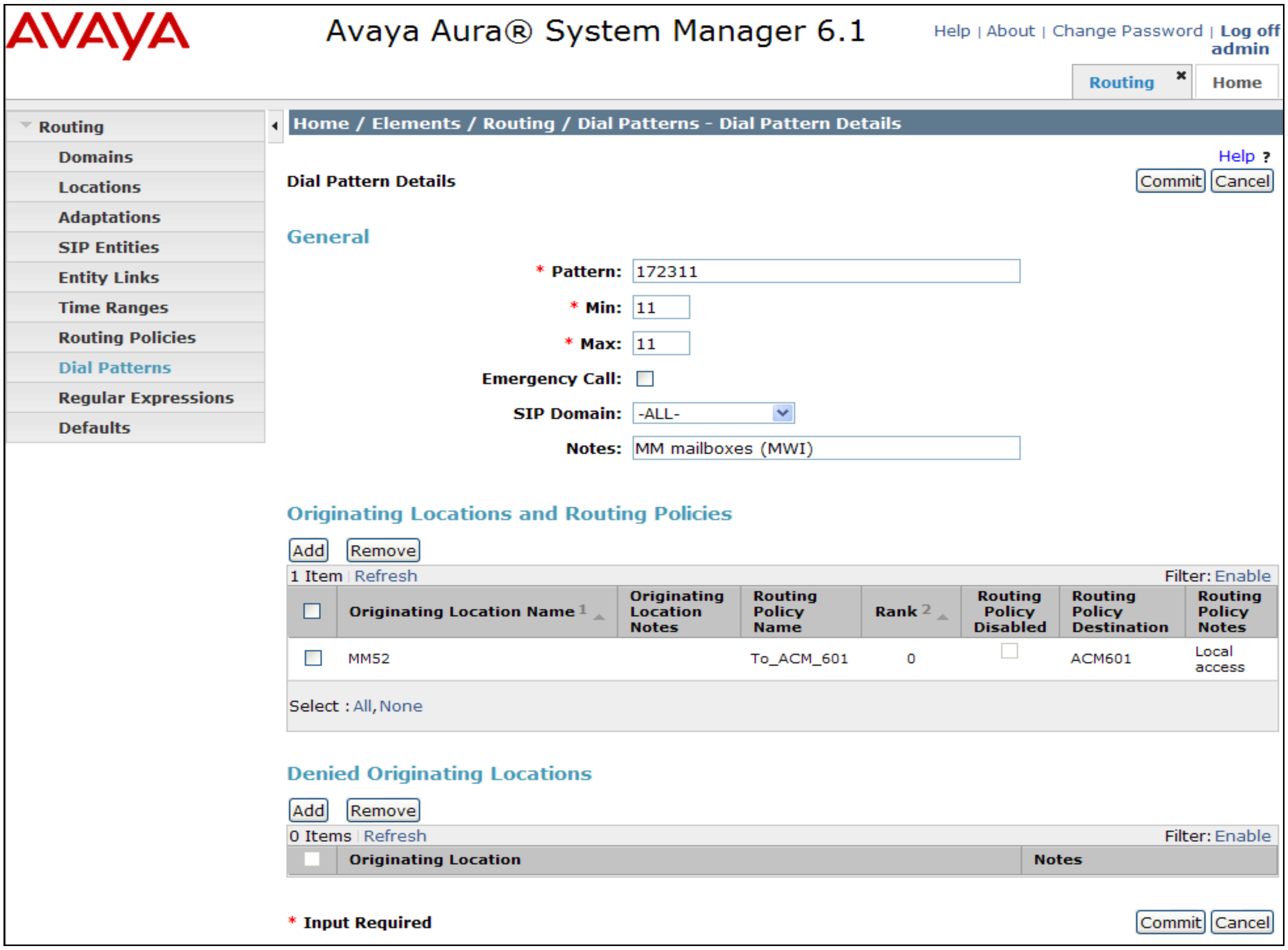
Step 4 - In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Location **MM52** (see **Section 5.2.3**). Note that only the calls that originate from the selected Location(s), or all administered Locations if **-ALL-** is selected, can match this Dial Pattern.

Step 5 - In the **Routing Policies** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Routing Policy **To_ACM_601** administered for routing calls to the Communication Manager Local trunk. in **Section 5.7.2**.



Step 6 - In the **Originating Location and Routing Policy List** page, click on **Select**.

Step 7 - Returning to the **Dial Pattern Details** page click on **Commit**.



6. Avaya Aura® Communication Manager

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration 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 applicable to these 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.

Step 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.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 2
                       Maximum Administered SIP Trunks: 24000 24
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

Step 2 - On **Page 3** of the **System-Parameters Customer-Options** form, verify that the **ARS** feature is enabled.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y           Audible Message Waiting? y
        Access Security Gateway (ASG)? y               Authorization Codes? y
        Analog Trunk Incoming Call ID? y                        CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                          CAS Main? n
Answer Supervision by Call Classifier? y                 Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                       DCS (Basic)? y
          ASAI Link Core Capabilities? y                 DCS Call Coverage? y
          ASAI Link Plus Capabilities? y                DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                           DS1 MSP? y
                                 ATMS? y             DS1 Echo Cancellation? y
                  Attendant Vectoring? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

Step 3 - On **Page 4** of the **system-parameters customer-options** form:

a. Verify that the **Enhanced EC500?** , the **IP Stations?, ISDN-PRI?** and the **IP Trunks?** fields are 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? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                              ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                     ISDN-PRI? y
              ESS Administration? y                Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y                  Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                  Multifrequency Signaling? y
      Global Call Classification? y          Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y                Multimedia IP SIP Trunking? y
                       IP Trunks? y
           IP Attendant Consoles? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

Step 5 - On **Page 5** of the **System-Parameters Customer-Options** form, verify that the **Private Networking** and **Processor Ethernet** fields are set to **y**.

```
display system-parameters customer-options                          Page   5 of  11
                                OPTIONAL FEATURES
                Multinational Locations? n              Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? y       Station as Virtual Extension? y
                     Multiple Locations? n
                                              System Management Data Transfer? n
          Personal Station Access (PSA)? y                 Tenant Partitioning? y
                        PNC Duplication? n         Terminal Trans. Init. (TTI)? y
                    Port Network Support? y                Time of Day Routing? y
                        Posted Messages? y         TN2501 VAL Maximum Capacity? y
                                                          Uniform Dialing Plan? y
                     Private Networking? y      Usage Allocation Enhancements? y
                 Processor and System MSP? y
                     Processor Ethernet? y                  Wideband Switching? y
                          Remote Office? y                            Wireless? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

6.2. Dial Plan

The dial plan defines how digit string will be used locally by Communication manager.

Step 1 - 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.** 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 **4xxxx**,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. access code **8** for outbound AAR dialing).
- 1-digit facilities access code (indicated with a **Call Type** of **fac**) (e.g. access code **9** for outbound ARS dialing).
  - Note – ARS is typically used for outbound dialing, which the AT&T IP Toll Free service does not support. It is shown here for informational purposes.
- 3-digit facilities access codes beginning with* and # for Agent logon/logoff (e.g.*66 or #76).

```
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
   4           5   ext
   8           1   fac
   9           1   fac
   *           3   fac
   #           3   fac
```

6.3. IP Node Names

Node names define IP addresses to various Avaya components in the enterprise.

Step 1 - Enter the **change node-names ip** command, and add a node name and the IP address for the Session Manager network interface (e.g. **ASM61**)

Step 2 – Repeat **Step 1** to add node names for the Avaya Aura® SBC and for Modular Messaging.

Step 3 - A Processor Ethernet (procr) based Communication Manager platform is used in the reference configuration. Make note of the Processor Ethernet node name and IP Address (**procr** & **192.168.67.202**). These entries appear automatically based on the address defined during Communication Manager installation.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ASM61               192.168.67.210
AA-SBC              192.168.67.125
MM52                192.168.67.141
default             0.0.0.0
procr               192.168.67.202
procr6              ::
```

6.4. IP Interface for procr

The display ip-interface procr command can be used to verify the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the reference configuration.

- Verify that **Enable Interface?**, **Allow H.323 Endpoints?**, and **Allow H248 Gateways?** Fields are set to **y**.
- Assign a network region (e.g. **1**).
- Use default values for the remaining parameters.

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


                  Type: PROCR
                                                      Target socket load:
19660

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


                                 IPV4 PARAMETERS
             Node Name: procr                    IP Address: 192.168.67.202
           Subnet Mask: /24
```

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 the reference configuration two network regions are used, one for local calls and one for AT&T calls.

6.5.1. IP Network Region 1 – Local Region

In the reference configuration local Communication Manager elements (e.g. procr) as well as other local Avaya devices (e.g. IP telephones, Modular Messaging) are assigned to ip-network-region 1.

Step 1 – Enter **change ip-network-region 1 x**, where **x** is the number of an unused IP network region (e.g. region 1). This IP network region will be used to represent the local CPE. Populate the form with the following values:

- Enter a descriptive name (e.g. **LOCAL**).
- Enter **customerb.com** in the **Authoritative Domain** field.
- Enter **1** for the **Codec Set** parameter.
- **Intra IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible within the same region.
- **Inter IP-IP Audio Connections** – Set to **yes**, indicating that the RTP paths should be optimized to reduce the use of media resources when possible between regions.
- **UDP Port Min**: - Set to **16384** (**AT&T requirement**).

- **UDP Port Max**: - Set to **32767** (**AT&T requirement**).

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: customerb.com
    Name: LOCAL
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step 2 - On **page 4** of the form:

- Verify that next to region **1** in the **dst rgn** column, the codec set is **1**.
- Next to region 2 in the **dst rgn** column, enter **2** (this means region 1 is permitted to talk to region 2 and it will use codec set 2 to do so). The **WAN** and **Units** columns will self populate with **y** and **No Limit**.
- Let all other values default for this form.

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

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

6.5.2. IP Network Region 2 – AT&T Trunk Region

In the reference configuration AT&T SIP trunk calls are assigned to ip-network-region **2**.

Step 1 - Repeat the steps in **Section 6.5.1** with the following changes:

- **Page 1**
  - Enter a descriptive name (e.g. **AT&T**)
  - Enter **2** for the **Codec Set** parameter.

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: customerb.com
    Name: AT&T
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? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step 2 – On **Page 4** of the form:

- Verify that codec **2** is listed for **dst rgn 1** and **2**

```
change ip-network-region 2                                      Page   4 of  20

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

6.6. IP Codec Parameters

6.6.1. Codecs for IP Network Region 1 (local calls)

In the reference configuration IP Network Region 1 uses codec set 1.

Step 1 - Enter the **change ip-codec-set x** command, where **x** is the number of an IP codec set used for internal calls. On **Page 1** of the **ip-codec-set** form, ensure that **G.711MU**, **G.729B**, and **G.729A** are included in the codec list. Note that the packet interval size will default to 20ms.

```
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.729B            n          2        20
 3: G.729A            n          2        20
```

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

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia:   384:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:   384:Kbits

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

6.6.2. Codecs for IP Network Region 2

In the reference configuration IP Network Region **2** uses codec set **2** for calls from AT&T.

Step 1 - Enter the **change ip-codec-set x** command, where **x** is the number of an unused IP codec set (e.g. **2**). This IP codec set will be used for inbound AT&T IP Toll Free calls. On **Page 1** of the **ip-codec-set** form, provision the codecs in the order shown. For G729B and G729A set **3** for the **Frames Per Pkt**. This will automatically populate **30** for the Packet Size(ms). Let G711MU default to **20**.

Note – See **Section 2.2.1**, **Item 4** for an issue regarding SIP telephone packet sizes.

```
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          2        20
```

Step 2 - On **Page 2** of the **ip-codec-set** form, set **FAX Mode** 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
```

6.7. SIP Trunks

Two SIP trunks are defined on Communication Manager in the reference configuration:

- AT&T access – SIP Trunk 2
  - Note that this trunk will use TCP port 5080 as described in **Section 5.5.1**.
- Local for Modular Messaging and Avaya SIP telephone access – SIP Trunk 1
  - Note that this trunk will use TCP port 5060 as described in **Section 5.5.2**.

SIP trunks are defined on Communication Manager by provisioning a Signaling Group and a corresponding Trunk Group.

Note – In the reference configuration TCP (port 5060 or 5080) 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 (port 5061) to be used as transport protocol in customer environments whenever possible.

6.7.1. SIP Trunk for AT&T IP Toll Free calls

This section describes the steps for administering the SIP trunk used for AT&T IP Toll Free calls. This trunk corresponds to the **ACM601_5080** Entity defined in **Section 5.4.2**.

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g. **2**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tcp**. Note – Although TCP is used as the transport protocol between the Avaya CPE components, the transport protocol used between the Avaya Aura® SBC and the AT&T IP Toll Free service is UDP.
- Verify the **IMS Enabled?** Is set to **n**.
- Verify that **Peer Detection Enabled** is **y** and that **Peer Server** is **SM**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 6.3**
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 6.3** (e.g. **ASM61**).
- **Near-end Listen Port** and **Far-end Listen Port** – set to **5080** (see Transport Method note above).
- **Far-end Network Region** – Set the IP network region to **2**, as set in **Section 6.5.2**.

- **Far-end Domain** – Enter **customerb.com**. This is the domain provisioned for Session Manager in **Section 5.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized to reduce the use of media resources on the Avaya Media Gateway when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This initiates Communication Manager to sends OPTIONS pings to Session Manager to provide link status.

```
add signaling-group 2                                               Page   1 of   1
                                SIGNALING GROUP
 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
   Near-end Node Name: procr                   Far-end Node Name: ASM61
 Near-end Listen Port: 5080                  Far-end Listen Port: 5080
                                          Far-end Network Region: 2
                                   Far-end Secondary Node Name:
Far-end Domain: customerb.com                  Bypass If IP Threshold Exceeded? n
                                               Incoming Dialog Loopbacks:
eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                        IP Audio Hairpinning? n
         Enable Layer 3 Test? Y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

Step 2 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **2**). On **Page 1** of the **trunk-group** form, provision the following:

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g. **ATT**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g. **102**).
- **Direction** – Set to **incoming**.
- **Service Type** – Set to **public-ntwrk**.
- **Signaling Group** – Set to the number of the signaling group administered in **Step 1** (e.g. **2**).
- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group (e.g. **20**).

```
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
                                              Member Assignment Method: auto
                                              Signaling Group: 2
                                              Number of Members: 20
```

Step 3 - On **Page 2** of the **Trunk Group** form:

- Set the **Preferred Minimum Session Refresh Interval(sec):** 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
             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Step 4 - On **Page 3** of the **Trunk Group** form:

- Set N**umbering Format:** 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
                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
 DSN Term? n
```

Step 5 - On **Page 4** of the **Trunk Group** form:

- Set **Telephone Event Payload Type** to the RTP payload type required by the AT&T IP Toll Free service (e.g. **100**).
- Use default for all other values.

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

6.7.2. Local SIP Trunk (Modular Messaging and Avaya SIP Telephones)

This section describes the steps for administering the local SIP trunk for Avaya Modular Messaging and Avaya SIP station calls. This trunk corresponds to the **ACM601** Entity defined in **Section 5.4.3**.

Step 1 - Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g. **1**), and provision the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tcp**. Note – Although TCP is used as the transport protocol between the Avaya CPE components, the transport protocol used between the Avaya Aura® SBC and the AT&T IP Toll Free service is UDP.
- Verify the **IMS Enabled?** Is set to **n**.
- Verify that **Peer Detection Enabled** is **y** and that **Peer Server** is **SM**.
- **Near-end Node Name** – Set to the node name of the **procr** noted in **Section 6.3**
- **Far-end Node Name** – Set to the node name of Session Manager as administered in **Section 6.3** (e.g. **ASM61**).
- **Near-end Listen Port** and **Far-end Listen Port** – set to **5060** (see Transport Method note above).
- **Far-end Network Region** – Set to the IP network region **1**, as defined in **Section 6.5.1**.
- **Far-end Domain** – Enter **customerb.com**. This is the domain used by Session Manager in **Section 5.1**.
- **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized to reduce the use of media resources on the Avaya Media Gateway when possible (known as shuffling).
- **Enable Layer 3 Test** – Set to **y**. This initiates Communication Manager to sends OPTIONS pings to Session Manager to provide link status.

```
add signaling-group 1                                          Page   1 of   1
                                SIGNALING GROUP
 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
   Near-end Node Name: procr                   Far-end Node Name: ASM61
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain: customerb.com
                                               Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                       IP Audio Hairpinning? n
         Enable Layer 3 Test? y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

Step 2 - Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **1**). On **Page 1** of the **trunk-group** form, provision the following:

- **Group Type** – Set to **sip**.
- **Group Name** – Enter a descriptive name (e.g. **Local**).
- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g. **101**).
- **Direction** – Set to **two-way**.
- **Service Type** – Set to **tie**.
- **Signaling Group** – Set to the number of the signaling group administered in **Step 1** (e.g. **1**).
- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group (e.g. **20**).

```
add trunk-group 1                                              Page   1 of  21
                                TRUNK GROUP
Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: Local                       COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? N
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 20
```

Step 3 - On **Page 2** of the **Trunk Group** form:

- Set the **Preferred Minimum Session Refresh Interval(sec):** to **900.** This entry will actually cause a value of 1800 to be generated in the SIP header.

```
add trunk-group 1                                               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
             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Step 4 - On **Page 3** of the **Trunk Group** form:

- Set N**umbering Format:** to **private**

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y
                     Numbering Format: private
                                                 UUI Treatment: service-provider
                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n
                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
 DSN Term? n
```

Step 5 - On **Page 4** of the **Trunk Group** form:

- Set **Telephone Event Payload Type** to the RTP payload type required by the AT&T IP Toll Free service (e.g. **100**).
- Use default for all other values.

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

6.8. Public Unknown Numbering

In the public unknown numbering form, Communication Manager local extensions are converted to AT&T Toll Free numbers (previously identified by AT&T) and directed to the public trunk defined in **Section 6.7.1**.

Step 1 - Using the **change public-unknown-numbering 0** command, enter:

- **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
- **Ext Code –** Enter the Communication Manager extension (e.g. **40001**).
- **Trk Grp(s)** – Enter the number of the AT&T trunk group (e.g. **2**).
- **CPN Prefix** – Enter the corresponding AT&T P Toll Free number (e.g. **7325554050**).
- **CPN Len** – Enter the total number of digits after the digit conversion (e.g. **10**).

Step 2 – Repeat **Step 1** for all corresponding AT&T IP Toll Free number/Communication Manager extensions.

```
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  40001           2         7325554050    10        Total Administered: 3
 5  40002           2         7325554051    10           Maximum Entries: 9999
 5  41001           2         7325554052    10
```

6.9. Private Numbering

The private-numbering form is used to direct calls to Avaya SIP telephones and calls to Modular Messaging (call coverage/retrieval) to the local trunk defined in **Section 6.7.2**.

Step 1 - Using the **change private-numbering 0** command, enter the Modular Messaging pilot number 46000.

- **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
- **Ext Code –** Enter the Communication Manager extension (e.g. **46000**).assigned to the Modular Messaging coverage hunt group defined in **Section 6.12**.
- **Trk Grp(s)** – Enter the number of the Local trunk group (e.g. **1**).
- **Total Len** – Enter the total number of digits after the digit conversion (e.g. **5**).

Step 2 – Repeat **Step 1** to direct calls to Avaya SIP telephones (the Communication Manager extension range 41xxx was used for SIP telephones) to the local trunk.

- **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
- **Ext Code –** Enter the Avaya SIP telephone extension range (e.g. **41**).
- **Trk Grp(s)** – Enter the number of the Local trunk group (e.g. **1**).
- **Total Len** – Enter the total number of digits after the digit conversion (e.g. **5**).

```
change private-numbering 0                                       Page   1 of   2

                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk         Private          Total
Len Code             Grp(s)      Prefix           Len
 5  46000            1                            5     Total Administered: 2
 5  41               1                            5       Maximum Entries: 540
```

6.10. Route Patterns

The AT&T IP Toll Free service does not support outbound dialing, so a route pattern is not required to direct calls to the public trunk. However a route pattern is used to direct calls to the local trunk.

6.10.1. Route Pattern for Modular Messaging and Avaya SIP Telephones

This form defines the local SIP trunk, based on the route-pattern selected by the AAR table in **Section 6.11** (e.g. calls to the Modular messaging pilot number 46000 or calls to Avaya SIP telephones 41xxx).

Step 1 – Enter the **change route-pattern 1** command and enter the following:

- In the **Grp No** column enter **1** for SIP trunk 1 (local trunk).
- In the **FRL** column enter **0** (zero).
- In the Numbering Format column, across from line **1:**, enter **unk-unk**.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 1    0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user
     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                               unk-unk  next
 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
```

6.11. AAR Dialing

Automatic Alternate Routing (AAR) is used to direct coverage calls for Modular Messaging (46000) or calls to the Avaya SIP telephones (41xxx) to the route pattern defined in **Section 6.10**.

Step 1 – Enter the change **aar analysis 0** command and for the SIP telephone extensions enter the following:

- **Dialed String** enter **41**
- **Min** & **Max** enter **5**
- **Route Pattern** enter **1**
- **Call Type** enter **aar**

Step 2 – For the Modular Messaging coverage hunt group extension enter the following:

- **Dialed String** enter **46000**
- **Min** & **Max** enter **5**
- **Route Pattern** enter **1**
- **Call Type** enter **aar**

```
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
    41                     5    5      1        aar           n
    46000                  5    5      1        aar           n
```

6.12. Provisioning for Coverage to Modular Messaging

To provide coverage to Modular Messaging for Communication Manager extensions, a hunt group is defined using the Modular Messaging pilot number (e.g. **46000**), as well as a coverage path that is defined to the various stations

6.12.1. Hunt Group for Station Coverage to Modular Messaging

Step 1 – Enter the command **add hunt-group x**, where **x** is an available hunt group (e.g. **1**), and on **Page 1** of the form enter the following:

- **Group Name** – Enter a descriptive name (e.g. **MM**).
- **Group Extension** – Enter an available extension (e.g. **46000**). Note that the hunt group extension need *not* be the same as the Modular Messaging pilot number.
- **ISDN/SIP Caller Display** – Enter **mbr-name**.
- Let all other fields default.

```
add hunt-group 1                                           Page   1 of   60
                                  HUNT GROUP
            Group Number: 1                                   ACD? n
              Group Name: MM                                Queue? n
         Group Extension: 46000                            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
```

Step 2 – On **Page 2** of the form enter the following:

- **Message Center –** Enter **sip-adjunct**.
- **Voice Mail Number** – Enter the Modular Messaging pilot number (e.g. **46000**).
- **Voice Mail Handle** - Enter the Modular Messaging pilot number (e.g. **46000**).
- **Routing Digits** – Enter the AAR access code defined in **Section 6.2** (e.g. **8**).

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

                    Message Center: sip-adjunct        Routing Digits
    Voice Mail Number        Voice Mail Handle     (e.g., AAR/ARS Access Code)
         46000                    46000                     8
```

6.12.2. Coverage Path for Station Coverage to Modular Messaging

After the coverage hunt group is provisioned, it is associated with a coverage path.

Step 1 – Enter the command **add coverage path x**, where **x** is an available coverage path (e.g. **1**), and on **Page 1** of the form enter the following:

- **Point1** – Specify the hunt group defined in the previous section (e.g. **h1**).
- **Rng** – Enter the number of rings before the stations go to coverage (e.g. **4**).
- Let all other fields default.

```
add 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:
```

6.12.3. Station Coverage Path to Modular Messaging

The coverage path configured in the previous section, is then defined on the stations.

Step 1 – Enter the command **change station xxxxx**, where **xxxxx** is a previously defined station or agent extension (e.g. Agent **47002**), and on **Page 1** of the form enter the following:

- **Coverage path** – Specify the coverage path defined in **Section 6.12.2**. Note that the coverage path field will appear at different positions on the form depending on whether agent or station extensions are being provisioned.

```
change agent-loginID 47002                                      Page   1 of   3
                                AGENT LOGINID
                Login ID: 47002                                       AAS? n
                    Name: Agent2                                    AUDIX? n
                      TN: 1                                 LWC Reception: spe
                     COR: 1                        LWC Log External Calls? n
           Coverage Path: 1                      AUDIX Name for Messaging:
           Security Code:
                                              LoginID for ISDN/SIP Display? n
                                                                  Password: 2580
                                                    Password (enter again): 2580
                                                               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
```

6.13. Call Center Provisioning

The administration of Communication Manager Call Center elements – agents, skills (hunt groups), vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Consult [5], and [6] for further details if necessary. The samples that follow are provided for reference purposes only.

- Agent form – **Page 1**

```
display agent-loginID 47002                                     Page   1 of   3
                                AGENT LOGINID
                Login ID: 47002                                       AAS? n
                    Name: Agent2                                    AUDIX? n
                      TN: 1                                 LWC Reception: spe
                     COR: 1                        LWC Log External Calls? n
           Coverage Path: 1                      AUDIX Name for Messaging:
           Security Code:             LoginID for ISDN/SIP Display? n
                                                                  Password: 2580
                                                    Password (enter again): 2580
                                                               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:   :
```

- Agent form – **Page 2**

```
display agent-loginID 47002                                     Page   2 of   3
                                 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: 2       1
 2:
```

- Skill 2 Hunt Group form – **Page 1**

```
display hunt-group 2                                             Page   1 of   4
                                  HUNT GROUP
            Group Number: 2                                    ACD? y
              Group Name: Skill2                             Queue? y
         Group Extension: 43002                             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 2 VDN form – **Page 1**

```
display vdn 44002                                                Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 44002
                                 Name*: Skill2
                           Destination: Vector Number        2
                   Attendant Vectoring? n
                  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
```

- Skill 2 Vector form – **Page 1**

```
display vector 2                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 2               Name: Skill2
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 42002
03 queue-to     skill 2    pri m
04 wait-time    10  secs hearing music
05 announcement 42005
06 goto step    3            if unconditionally
07 stop
08
```

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] and [8] for further details.

8. Configure Avaya Aura® Session Border Controller (SBC)

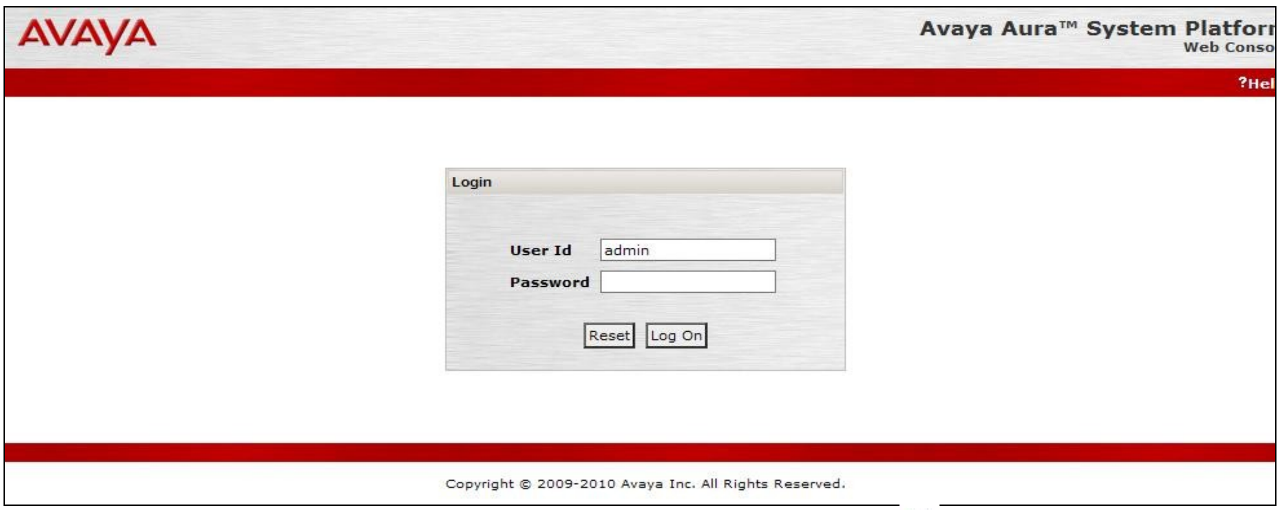
This section illustrates an example configuration of the Avaya Aura® SBC. In the sample configuration, the Avaya Aura® SBC resides on its own S8800 Server as an application template running on System Platform operating system. The application template defines basic functionality for the SBC such as IP addressing, SIP domains, etc. The installation of the System Platform and application template is assumed to have been previously completed (see the Avaya Aura® SBC references [9] and [10]) for additional information on the Avaya Aura® SBC installation.

Note - The AT&T IP Toll Free service border element IP addresses shown in this document are examples. AT&T Customer Care will provide the actual IP addresses as part of the IP Toll Free provisioning process.

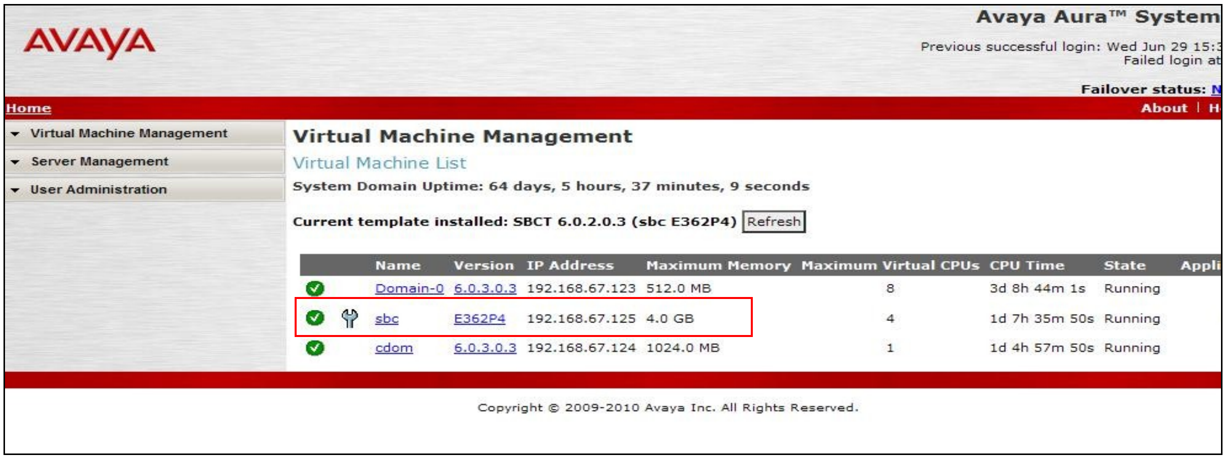
8.1. Logging into the Avaya Session Border Controller

Log in to the System Platform console domain by entering https://<ip-addr>/webconsole as shown in the example screen below. In the reference configuration, the console domain uses the IP Address 192.168.67.124. Enter an appropriate **User Id** and press the **Continue** button.

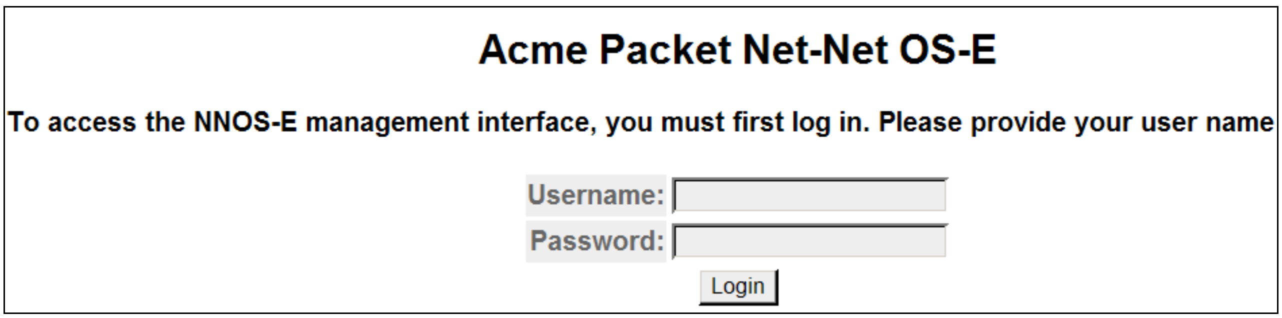
On the subsequent screen, enter the appropriate **Password** and click the **Log On** button.



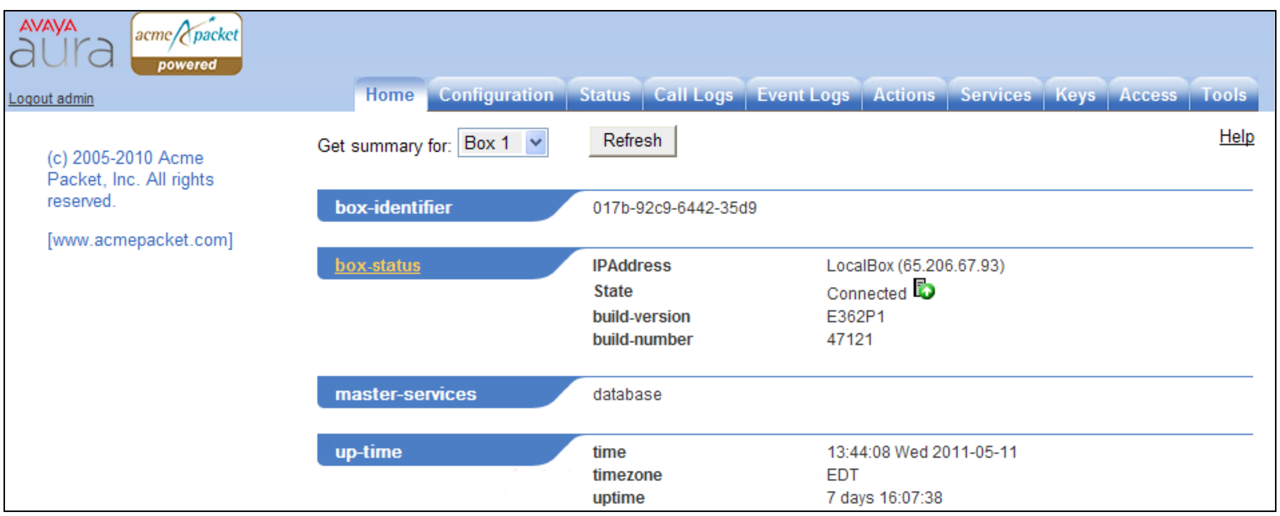
The **Virtual Machine List** will show the SBC Template. Click on the to access the Avaya SBC GUI interface.



Enter appropriate **Username** and **Password** and click **Login**.



The following shows an abridged **Home** screen after logging in. Note the tabs at the top.



8.2. Network Configuration

As described previously much of the network information is defined during installation of the SBC application template. However there may be occasions where these parameters need to be modified. Therefore these values are described below.

In the reference configuration, the Avaya S8800 Server has four physical network interfaces, labeled 1 through 4. The port labeled 1 (virtual eth0) is used for the management and private (inside) network interface of the SBC (toward the customer equipment). The port labeled 4 (virtual eth2) is used for the public (outside) network interface of the SBC (toward AT&T). These can be verified by checking the interface eth0 and interface eth2 settings (see **Section 8.2.1**).

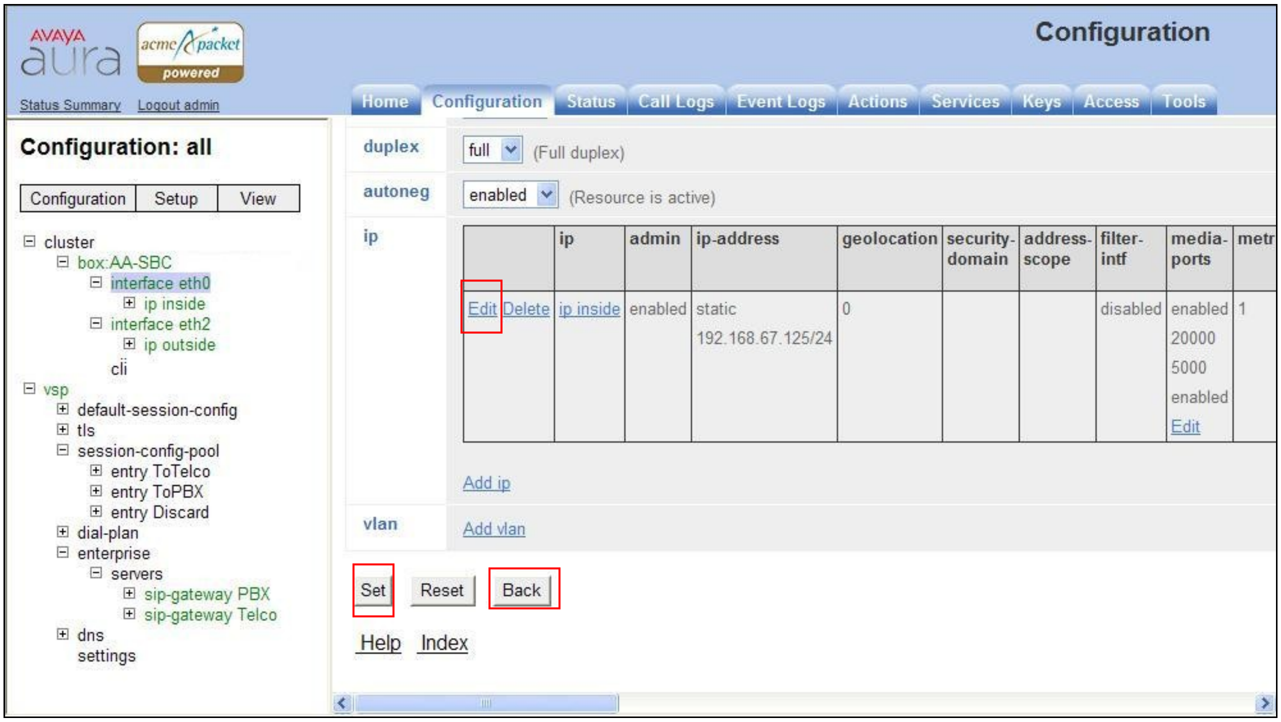
The AT&T AVPN transport service requires that RTP media traffic use UDP port range 16384-32767. This range is defined as part of interface eth2 (see **Section 8.2.3**).

SIP-Gateways are defined for corresponding to the private and public interfaces. In the reference configuration the private interface is defined as PBX and the public interface is defined as **Telco1** (see **Section 8.2.4**).

8.2.1. Verify IP Addressing

Step 1 - From the **Configuration** tab, select **cluster → box <name defined during install>** (e.g. **AA-SBC**). The **interface eth0** and **interface eth2** will be displayed. Click on **ip inside** (eth0) or **ip outside** (eth2) to display the interface configuration. Note that AT&T may require the eth2 IP address as part of the IP Toll Free service provisioning.

Step 2 - The configuration may be modified by clicking the **Edit** button. If changes are made, click on the **Set** button. To cancel changes or to go to a previous screen, click on **Back**.



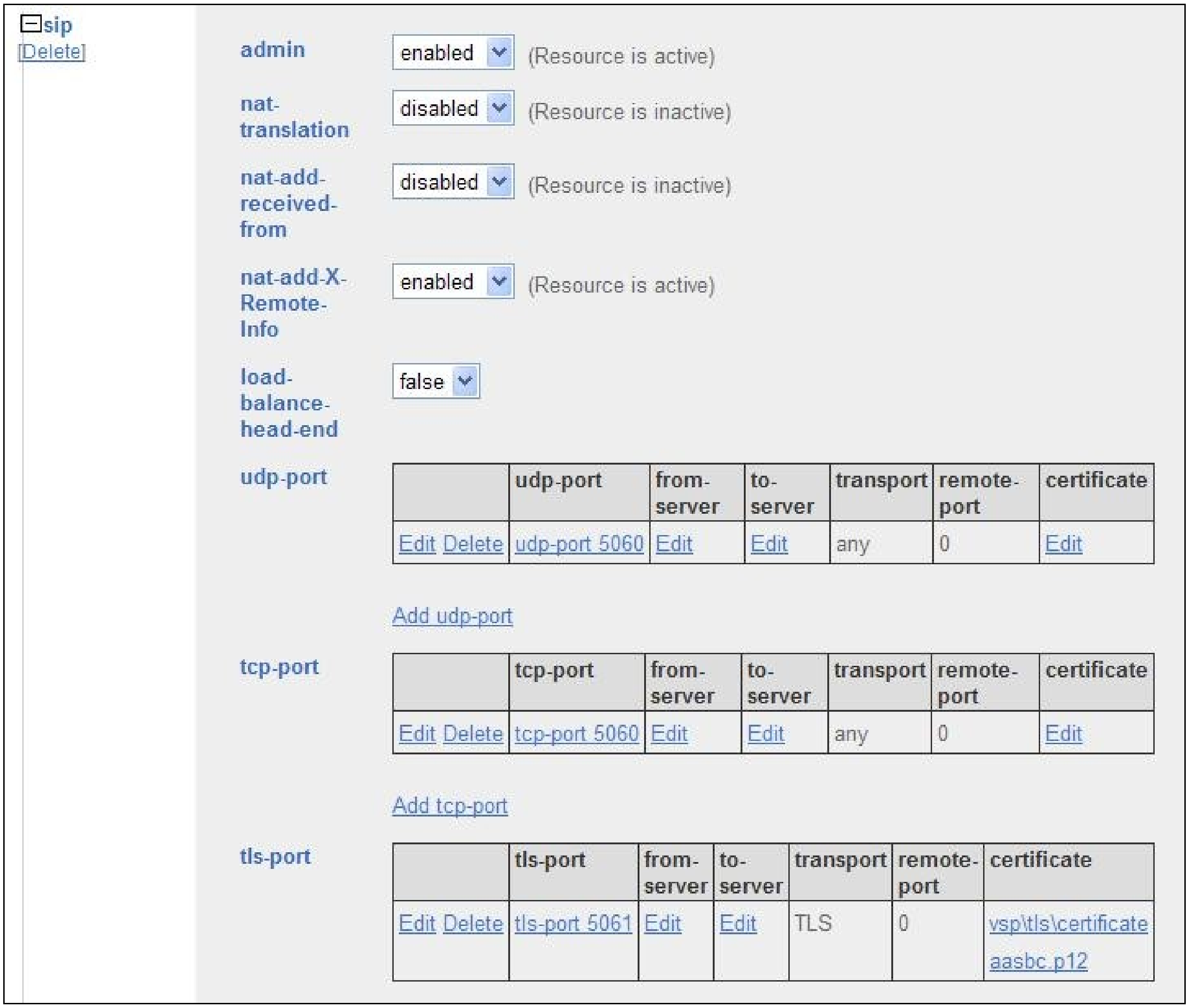
8.2.2. Transport Protocols

8.2.2.1 Private Interface – Eth0

The private interface, eth0, was provisioned to support UDP, TCP, and TLS transport protocols. However, TCP (port 5060) was used in the reference configuration for the connection to Session Manager (see **Section 5.4.4** and **5.5.3**). This can be displayed by the following:

Step 1 – Navigate to **cluster → box <name defined during install> → interface eth0 → ip inside.**

Step 2 – Scroll down to, and click on the **SIP** heading. The UDP, TCP, and TLS supported protocols are displayed.



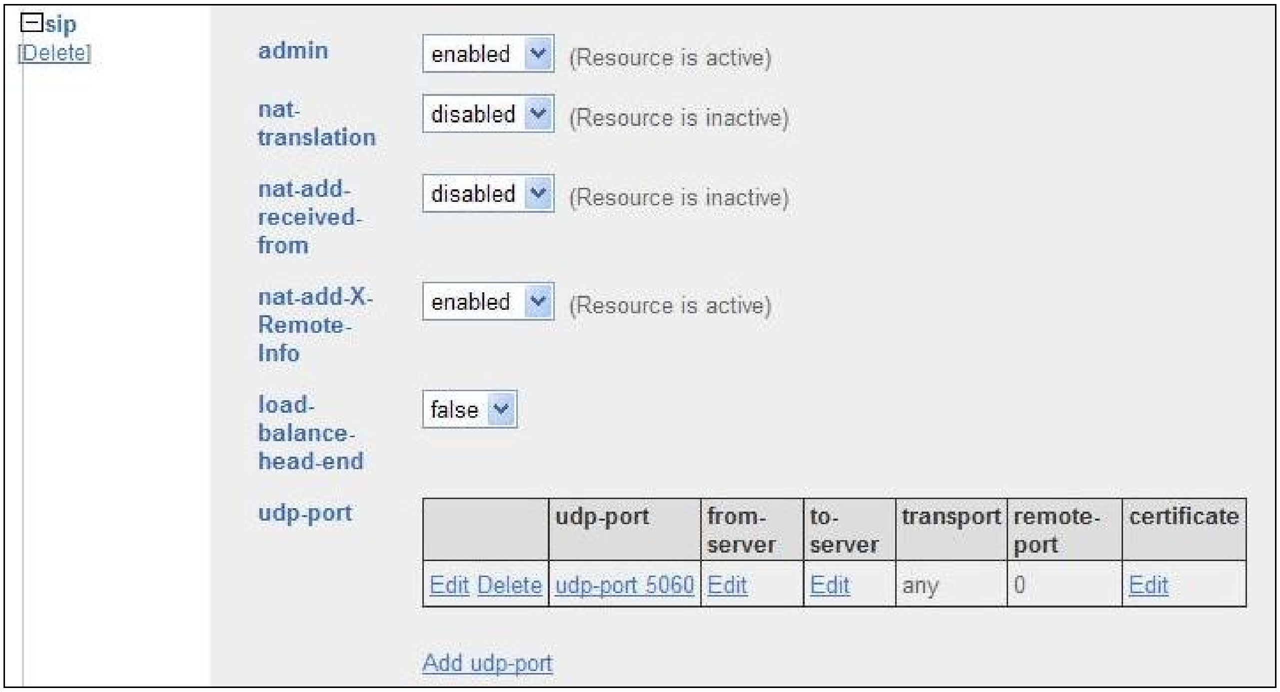
Step 3 - The configuration may be modified by clicking the **Edit** buttons. If changes are made, click on the **Set** button (not shown). To cancel changes or to go to a previous screen, click on **Back** (not shown).

8.2.2.2 Public Interface – Eth2

The AT&T IP Toll Free service requires UDP transport protocol between the Avaya SBC and the AT&T IP Toll Free service border element. Therefore, the public interface, eth2, was provisioned to support UDP transport protocol only. This can be displayed by the following:

Step 1 – Navigate to **cluster → box <name defined during install> → interface eth2 → ip outside.**

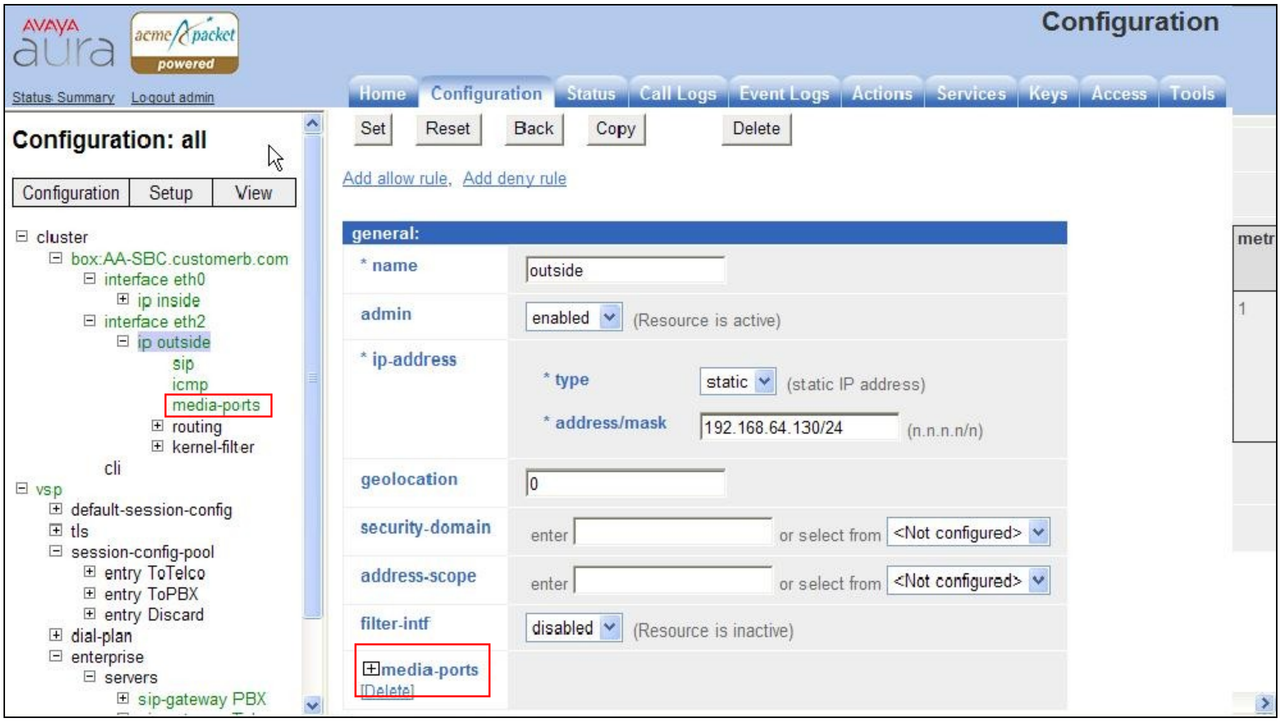
Step 2 – Scroll down to, and click on the **SIP** heading. The UDP (port 5060) transport protocol is displayed.



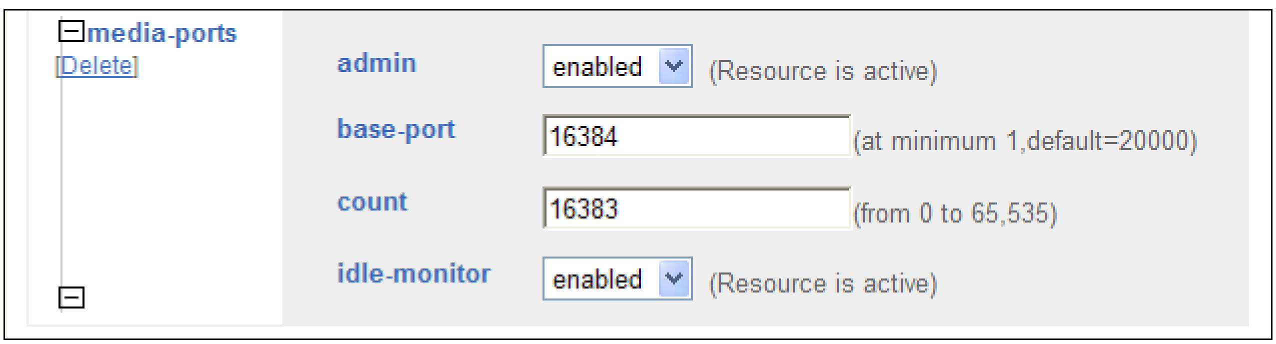
Step 3 - The configuration may be modified by clicking the **Edit** buttons. If changes are made, click on the **Set** button (not shown). To cancel changes or to go to a previous screen, click on **Back** (not shown).

8.2.3. Setting the RTP Port Range on Eth2

Step 1 - Go to **cluster → box <name defined during install> → interface eth2 → ip outside** to display the eth2 configuration toward AT&T. Select media-ports from either the menu or from the display.



Step 2 - The media port section will be displayed. Enter **16384** in the **base-port** field and **16383** in the **count** field.



Step 3 - Click on the **Set** button (not shown) to save.

Step 4 - Proceed to save and activate the configuration as described in **Section 8.3**.

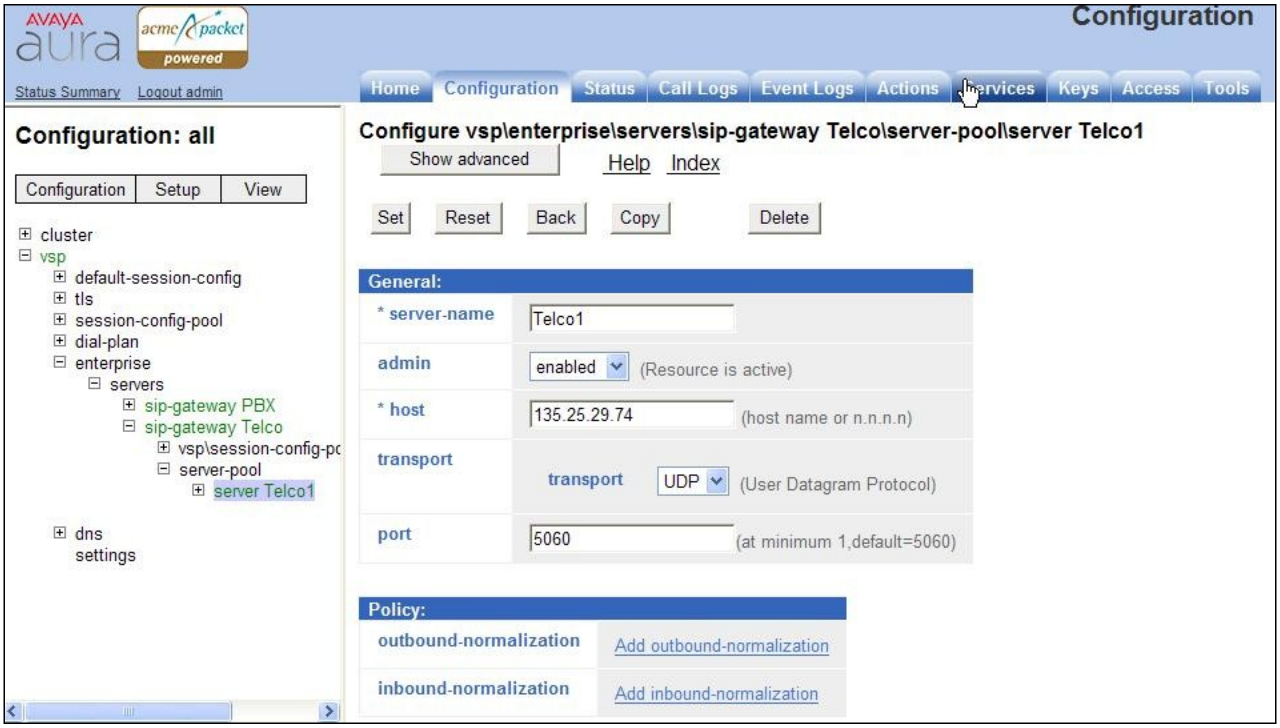
8.2.4. Configuring the SIP-Gateways

In the reference configuration, a sip-gateway was defined to AT&T (the IP Toll Free border element) and to the customer site (Session Manager). The AT&T gateway was defined as Telco1 and customer gateway was defined as PBX.

8.2.4.1 Telco1

Step 1 - Go to **vsp → enterprise → servers** and any previously defined sip-gateways will be displayed. In the reference configuration sip-gateways **PBX** and **Telco1** were defined.

Step 2 - Click on **sip-gateway Telco → servers → server-pool → server Telco1** and the Telco1 sip-gateway configuration will be displayed.



Step 3 - Verify the following:

- admin state is **enabled**.
- host address is the IP address of the AT&T IP Toll Free border element (e.g. **135.25.29.74**).
- transport protocol is **UDP**.
- port is **5060**.

Step 4 - Click on the **Set** button to save any changes or **Back** if no changes are required.

Step 5 - Proceed to save and activate the configuration as described in **Section 8.3**.

8.2.4.2 PBX

Repeat the steps in **Section 8.2.4.1** and verify the following:

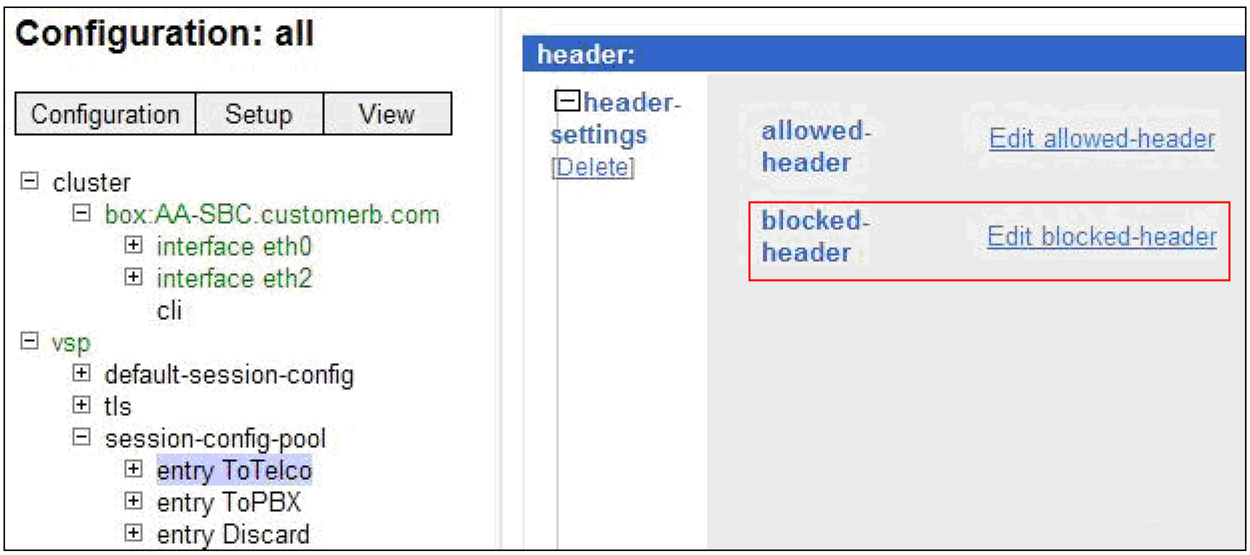
- admin state is **enabled**.
- host address is the IP address of Session Manager(e.g. **192.168.67.210**).
- transport protocol is **TCP**. Note that TCP was used in the reference configuration to facilitate protocol trace verification and troubleshooting. TLS may be used as well.
- port is **5060**.

8.2.5. Stripping SIP Headers (Optional)

The Avaya SBC can be used to strip SIP headers that are not required or supported by AT&T. For headers that have relevance only within the enterprise, it may be desirable to prevent the header from being sent to the public SIP Service Provider. For example, Session Manager Release 6.1 may insert the P-Location headers. The following procedures may be used to strip such headers that AT&T does not process.

Undesired headers may be removed via the session-config-pool. For example, during installation, two session-config-pools were created, To-Telco and To-PBX. First the headers are removed session-config-pool **To-Telco**. This will remove the specified headers for calls sent by the customer location to AT&T.

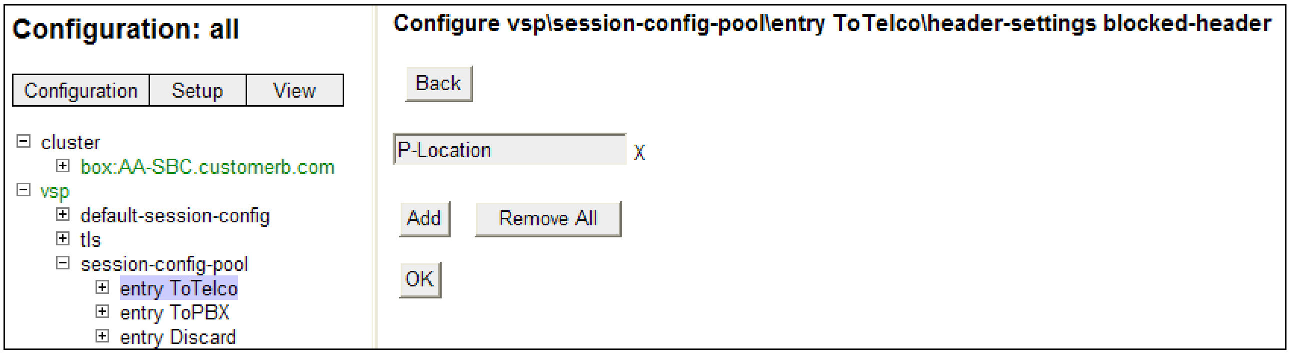
Step 1 - Navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. In the resultant screen, click **Edit blocked-header.**



Step 2 – Enter **P-Location** into the selection box.

Step 3 – If additional headers need to be blocked, click on the **Add** button.

Step 4 – When all headers are entered, click on **OK**.

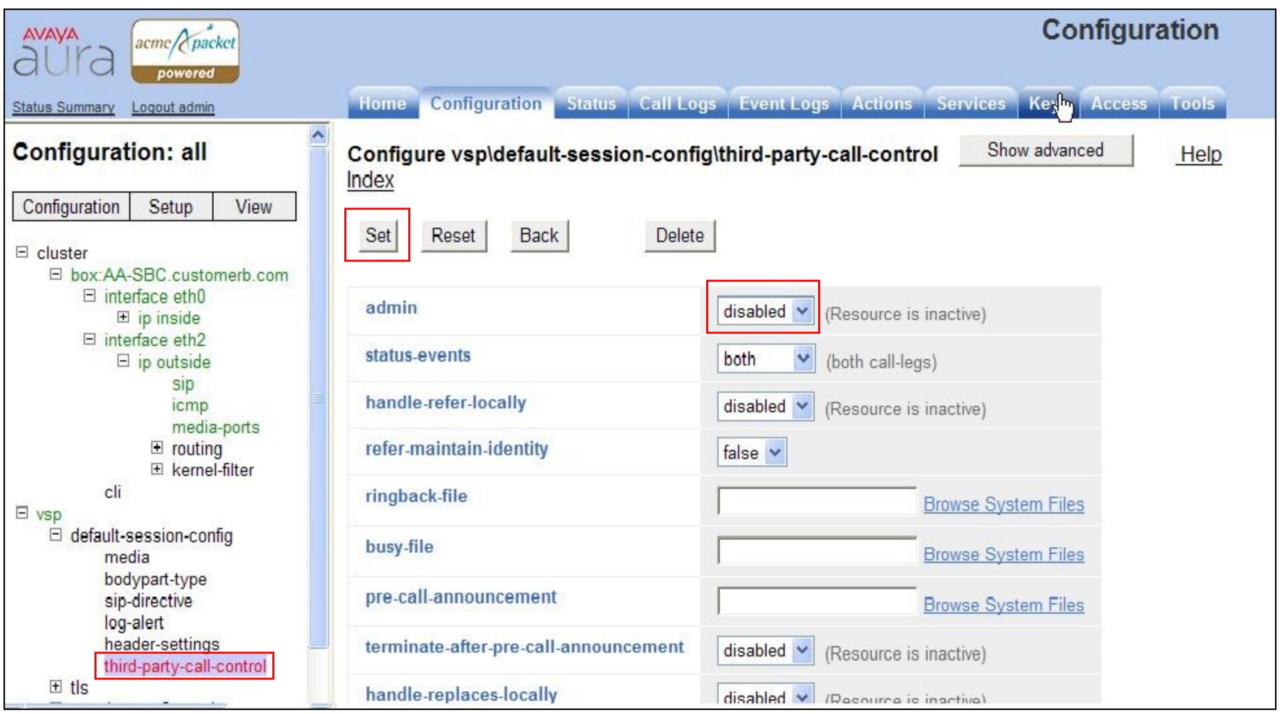


Step 5 - Proceed to save and activate the configuration as described in **Section 8.3**.

8.2.6. Disable Third Party Call Control

Step 1 - Navigate to **vsp → default-session-config → third-party-call-control**. To disable third-party-call-control, select **disabled** from the **admin** drop-down. Note - After disabling, the third-party-call-control link becomes red as shown below.

Step 2 - click **Set** as shown below.



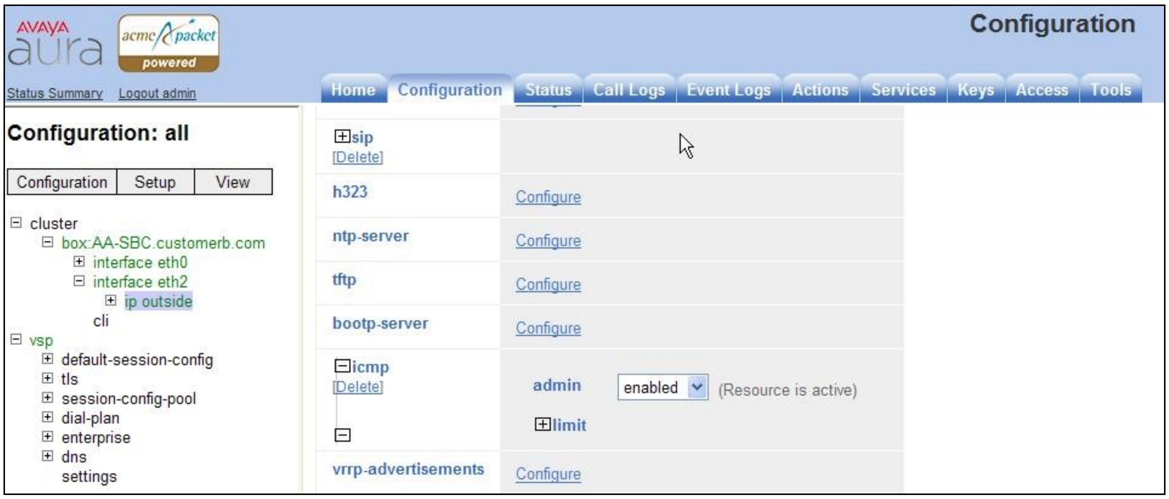
Step 3 - Proceed to save and activate the configuration as described in **Section 8.3**.

8.2.7. SIP OPTIONS Messages for AT&T Network Status

In the reference configuration the Avaya SBC sent SIP OPTIONS messages to the AT&T IP Toll Free border element to verify the state of the network connection. The AT&T response to the OPTIONS is 405 Method Not Allowed. Although this appears to be an error, in fact the arrival of the message assures the Avaya SBC that the network connection is up.

Step 1 - Navigate to **cluster → box:AvayaSBC → interface eth2 → ip outside**. Scroll down to, and click on, the **icmp** option.

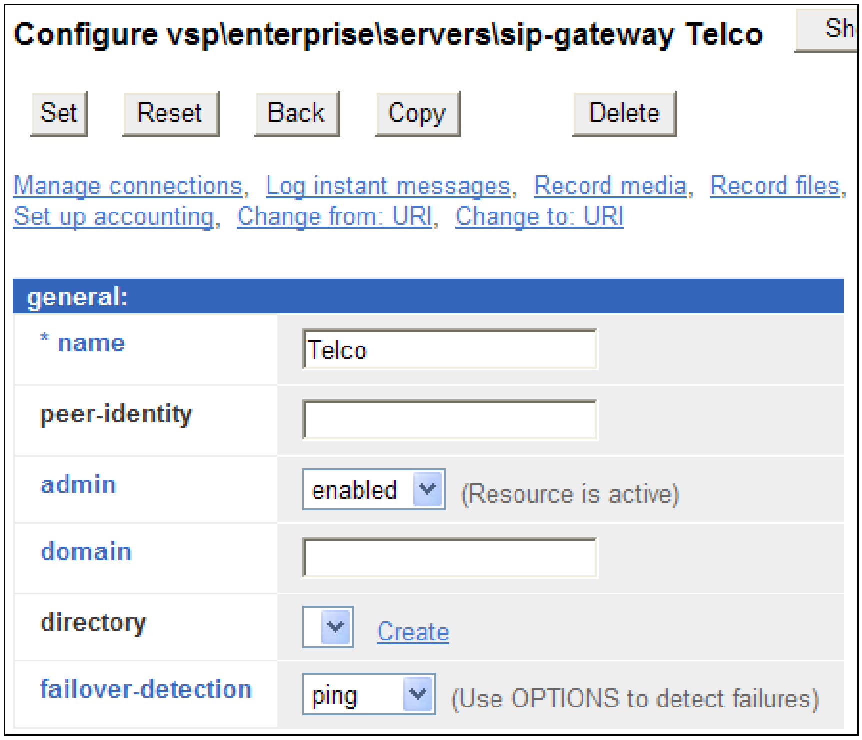
Step 2 - Set the **admin** option to **enabled**.



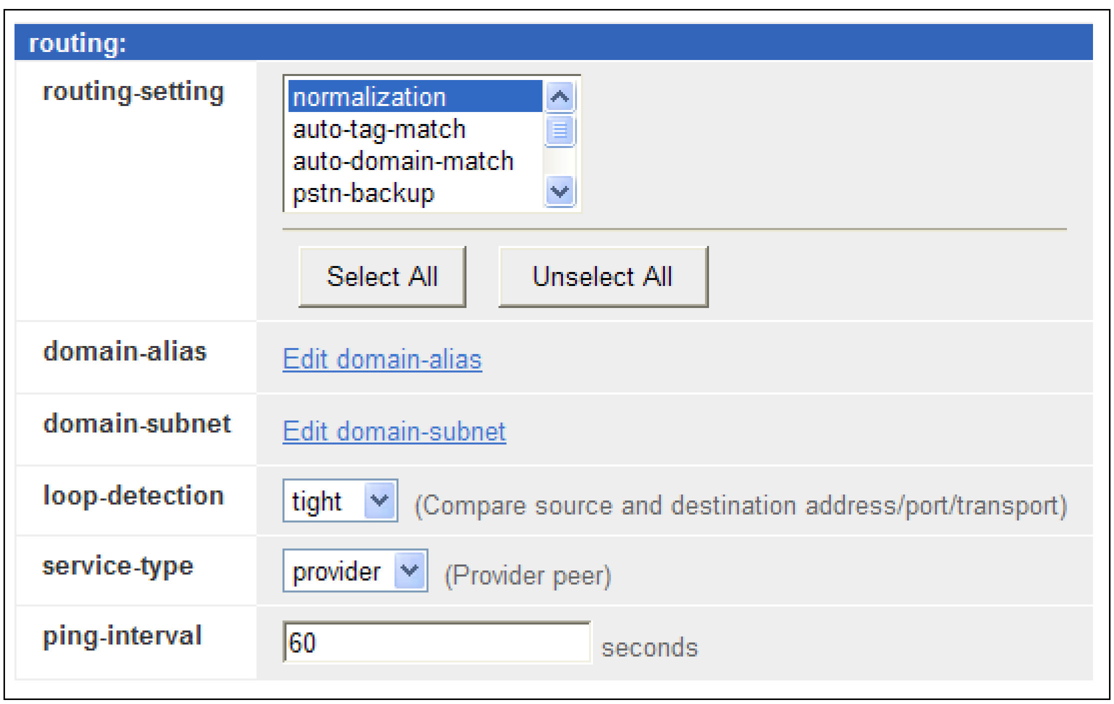
Step 3 - Scroll to the bottom of the screen and click **Set**.

Step 4 - Navigate to **vsp → enterprise → servers → sip-gateway Telco**. Click on the **Show Advanced** button at the top of the page (not shown).

Step 5 – In the **general:** section set **failover-detection** and select **ping** from the menu.



Step 6 – Scroll down to the **routing**: section and set the **ping-interval** as desired (e.g. **60**).

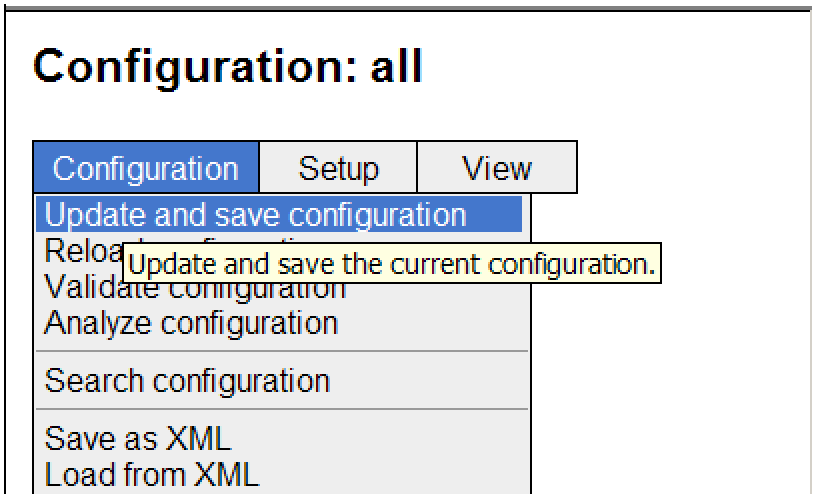


Step 7 - Scroll to the bottom of the screen and click **Set**.

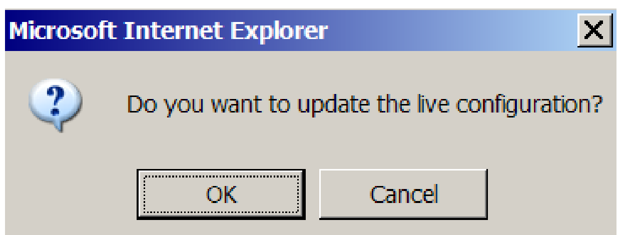
Step 8 - Proceed to save and activate the configuration as described in **Section 8.3**.

8.3. Saving and Activating Configuration Changes

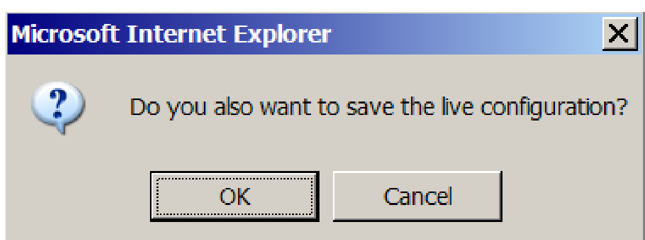
Step 1 - To save and activate configuration changes, select **Configuration → Update and save configuration** from the upper left hand side of the user interface, as shown below.



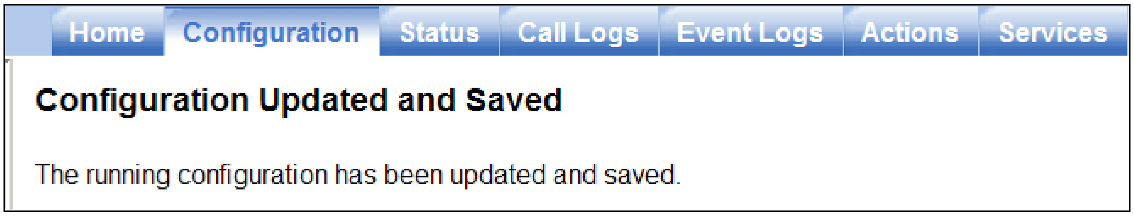
Step 2 - Click **OK** to update the live configuration.



Step 3 - Click **OK** to save the live configuration.



A screen that includes the following should appear.



9. Verification Steps

The following steps may be used to verify the configuration:

9.1. General

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. Place an inbound call to an agent or telephone, but do not answer the call. Verify that the call covers to Modular Messaging voicemail. Retrieve the message from Modular Messaging.

9.2. Avaya Aura® Communication Manager

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

1. From the Communication Manager console connection enter the command ***list trace tac xxx***, where xxx is a trunk access code defined for the SIP trunk to AT&T (e.g. 101). Note that Session Manager has previously converted the AT&T IP Toll Free number included in the Request URI to the Communication Manager extension 40002, before sending the INVITE to Communication Manager.

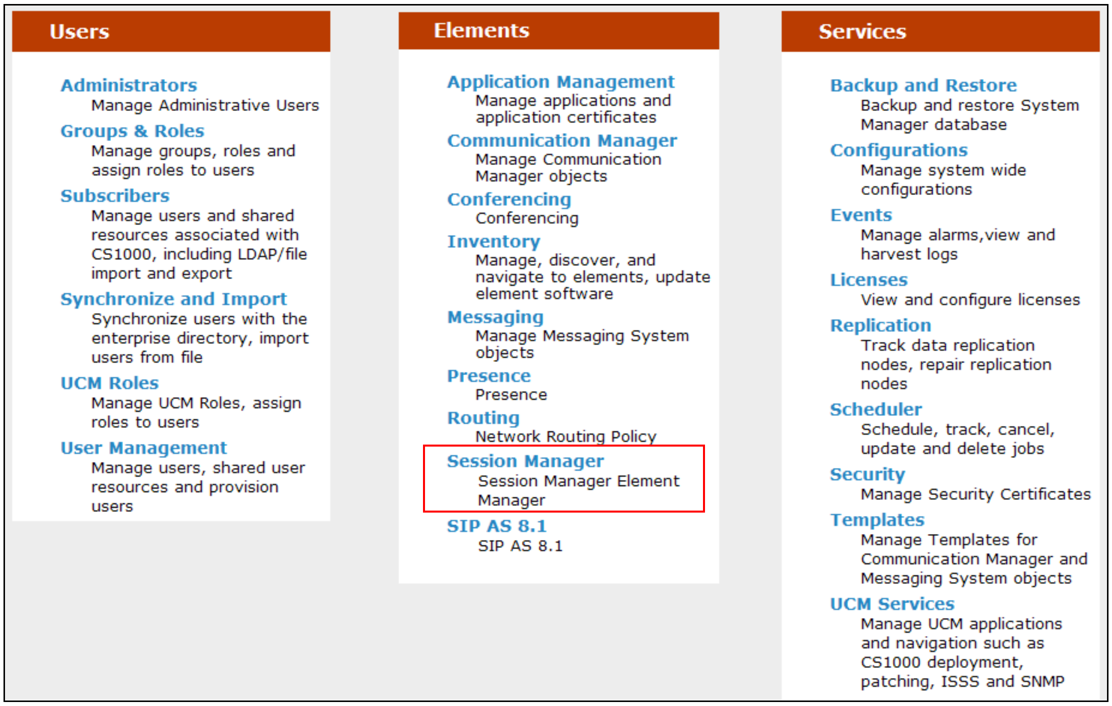
```
list trace tac 101                                Page   1

                    LIST TRACE

time         data

10:50:35 TRACE STARTED 07/19/2010 CM Release String cold-00.0.345.0-18246
10:50:49 SIP<INVITE sip:40002@customerb.com:5060;transport=tcp S
10:50:49 SIP<IP/2.0
10:50:49     active trunk-group 1 member 1  cid 0x270
10:50:49 SIP>SIP/2.0 183 Session Progress
10:50:49     dial 40002
10:50:49     ring station    40002 cid 0x270
10:50:49     G711MU ss:off ps:20
        rgn:1 [192.168.67.80]:17382
        rgn:1 [192.168.67.203]:16390
```

```
10:50:49     G729B ss:off ps:20
             rgn:2 [192.168.67.130]:16480
             rgn:1 [192.168.67.203]:16386
10:50:49     xoip options: fax:T38 modem:off tty:US  uid:0x50001
             xoip ip: [192.168.67.203]:16386
10:50:50 SIP>SIP/2.0 200 OK
10:50:50     active station    40002 cid 0x270
10:50:50 SIP<ACK sip:7323204384@192.168.67.202;transport=tcp SIP
10:50:50 SIP</2.0
10:50:50 SIP>INVITE sip:7326712438@192.168.67.130:5060;transport
10:50:50 SIP>=tcp SIP/2.0
10:50:50 SIP<SIP/2.0 100 Trying
10:50:51 SIP<SIP/2.0 200 OK
10:50:51 SIP>ACK sip:7326712438@192.168.67.130:5060;transport=tc
10:50:51 SIP>p SIP/2.0
10:50:51     G729AB ss:off ps:20
             rgn:2 [192.168.67.130]:16480
             rgn:1 [192.168.67.80]:17382
10:50:51     G729B ss:off ps:20
             rgn:1 [192.168.67.80]:17382
             rgn:2 [192.168.67.130]:16480
10:50:54 SIP>BYE sip:7326712438@192.168.67.130:5060;transport=tc
10:50:54 SIP>p SIP/2.0
10:50:54     idle station    40002 cid 0x270
```

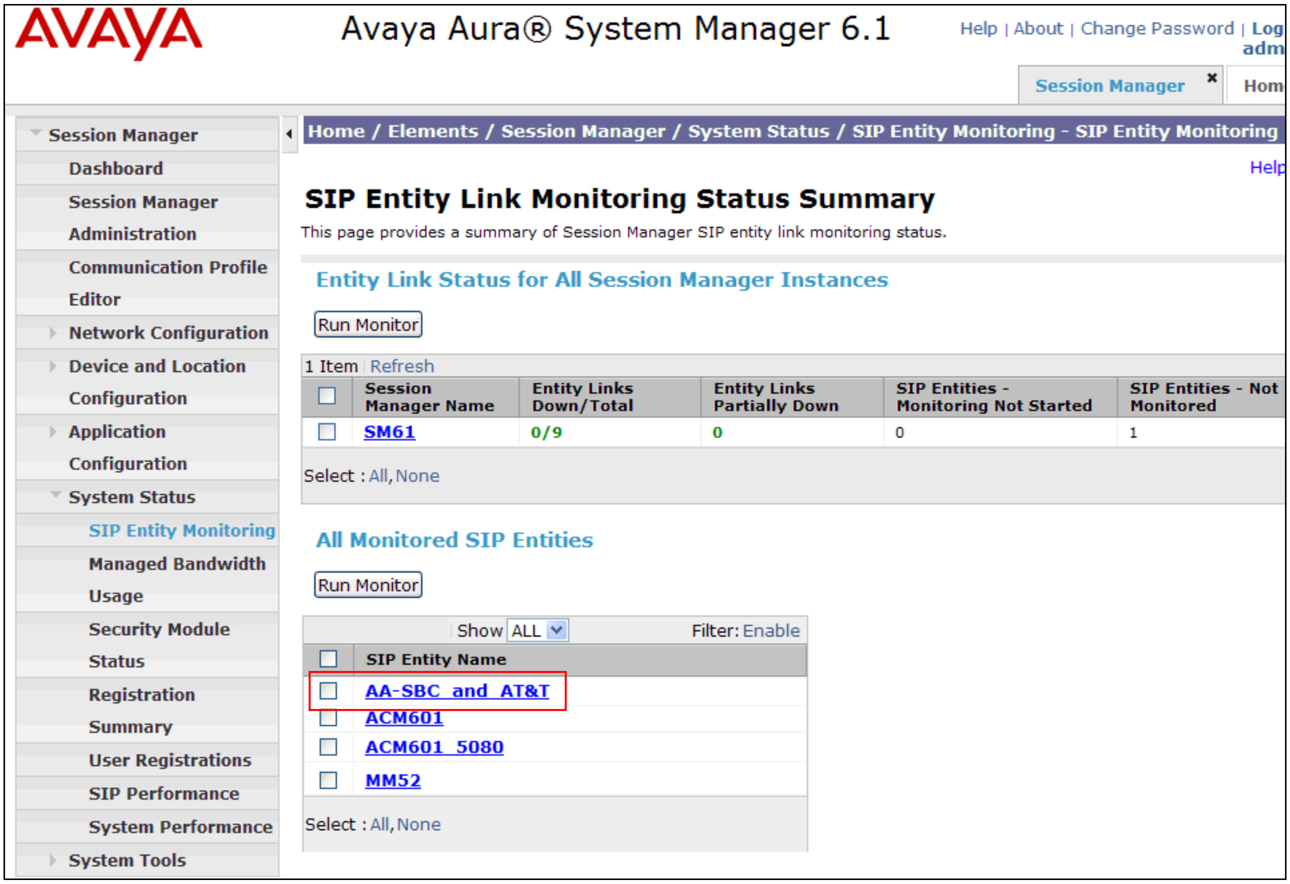
2. Similar Communication Manager commands are, ***list trace station***, ***list trace vdn***, and ***list trace vector***. Other useful commands are ***status trunk*** and ***status station***.

9.3. Avaya Aura® Session Manager

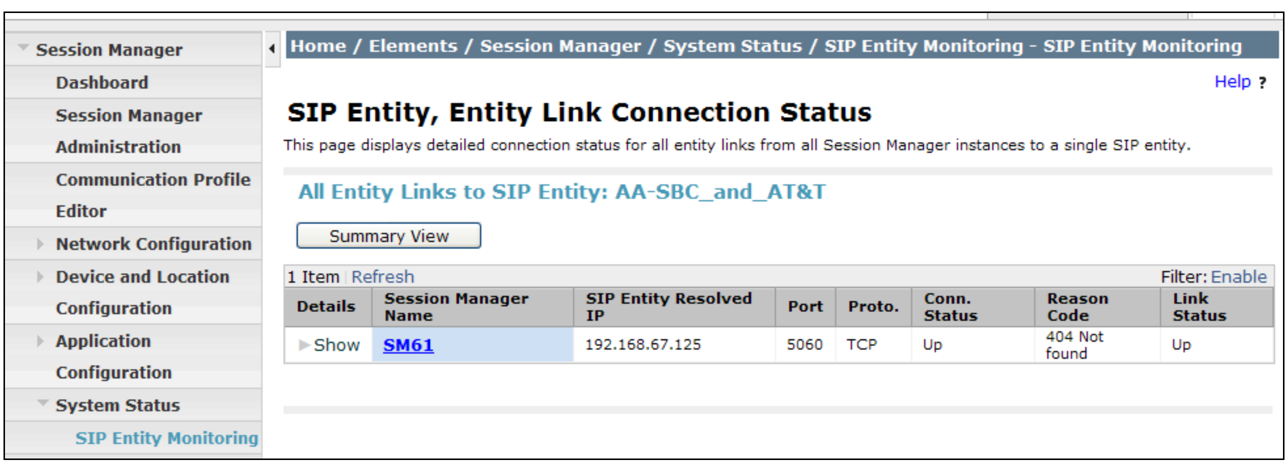
Step 1 - Access the System Manager GUI, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials. Once logged in, a Release 6.1 **Home** screen like the following is displayed. From the **Home** screen below, under the **Elements** heading in the center, select **Session Manager**.



Step 2 - Expand **System Status → SIP Entity Monitoring**.



Step – 3 From the list of monitored entities, select an entity of interest, such as **AA-SBC_and_AT&T**. Under normal operating conditions, the **Link Status** should be Up as shown in the example screen below. The **Reason Code** column indicates that the SBC has responded to SIP OPTIONS from Session Manager with a SIP 404 message (normal for the Avaya Aura® SBC to AT&T environment), which is sufficient for SIP Link Monitoring to consider the link up.



9.3.1. Call Routing Test

The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, expand **Elements → Session Manager → System Tools → Call Routing Test**. The following example shows an inbound call to Communication Manager from the AT&T IP Toll Free service. Note that the Request URI called number was 0000001050 and Session Manager converts this to Communication Manager extension 41006 before routing the call to Communication Manager

Step 1 – **Called Party URI** field = the information passed in the Request URI sent by the Avaya Aura® SBC (e.g. **0000001050@customerb.com**)

Step 2 – **Calling Party Address** field = the IP address of the inside interface of the Avaya Aura® (e.g. **192.168.67.130**).

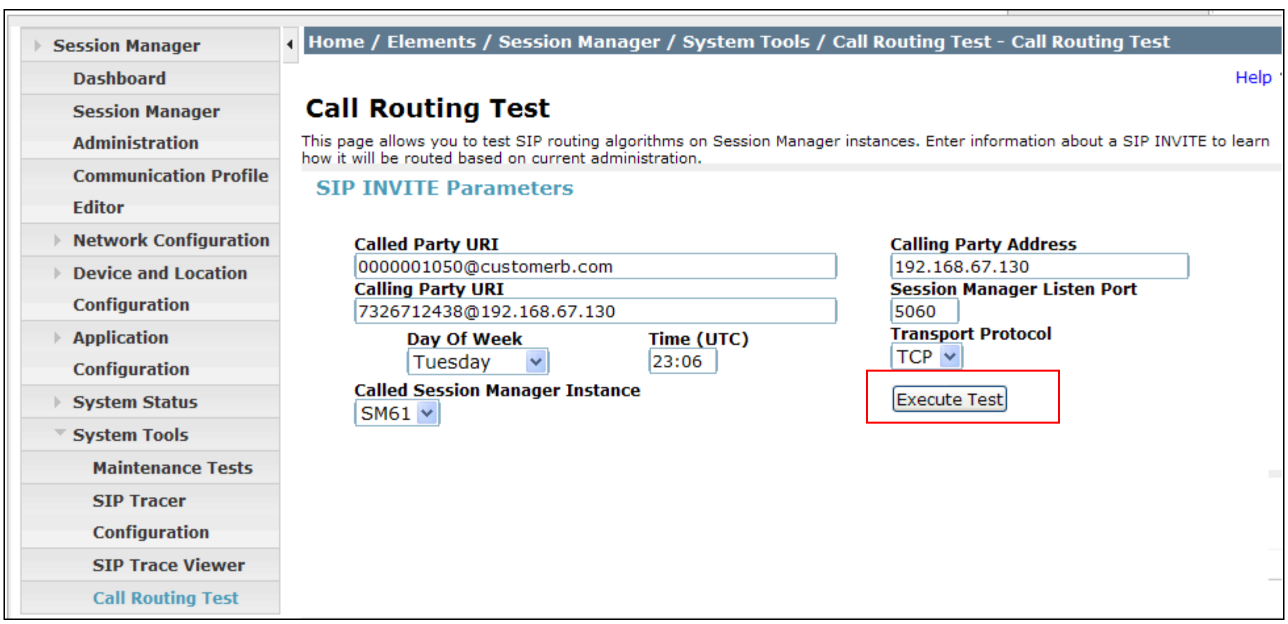
Step 3 – **Calling Party URI** field = The contents of the From header (e.g **7326712438@192.168.67.130**).

Step 4 – **Session Manager Listening Port** = **5060** and **Transport protocol** = **TCP** (see the note in **Section 5.5** regarding the use of TCP).

Step 5 – Populate the **Day of Week** and **Time (UTC)** fields, or let them default to current.

Step 6 – Verify that the **Called Session Manager** instance is correct (if multiple ones are defined).

Step 7 - Click on **Execute Test**.



The results of the test are shown below**.** The ultimate routing decision is displayed under the heading **Routing Decisions.** The example shows that a PSTN call to AT&T IP Toll Free service, delivering 0000001050 in the Request URI, is sent to Communication Manager extension **41006**. Further down, the **Routing Decision Process** steps are displayed (depending on the complexity of the routing, multiple pages may be generated).Verify that the test results are consistent with the expected results of the routing administered on Session Manager in **Section 5**.

Routing Decisions

Route < sip:41006@customerb.com > to SIP Entity ACM601_5080 (192.168.67.202). Terminating Location is main.

Routing Decision Process

NRP Adaptations: ACM_AT&T_Acme applied.
BEGIN EMERGENCY CALL CHECK: Determining if this is a call to an emergency number.
Originating Location is Acme. Using digits < 0000001050 > and host < customerb.com > for routing.
NRP Dial Patterns: No matches for digits < 0000001050 > and domain < customerb.com >.
NRP Dial Patterns: Found a Dial Pattern match for pattern < 0000001050 > Min/Max length 10/10 and domain < null >.
NRP Routing Policies: Ranked destination NRP Sip Entities: ACM601_5080.
NRP Routing Policies: Removing disabled routes.
NRP Routing Policies: Ranked destination NRP Sip Entities: ACM601_5080.
END EMERGENCY CALL CHECK: This is not an emergency call.
Adapting and proxying for SIP Entity ACM601_5080.
NRP Entity Links: Found direct link to destination. Link uses TCP to port 5080.
NRP Adaptations: To_ACM601 applied.
NRP Adaptations: P-Asserted-Identity set to sip:7326712438@customerb.com
NRP Adaptations: Request-URI set to sip:41006@customerb.com
NRP Adaptations: Request URI set to sip:41006@customerb.com

9.4. Protocol Traces

Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Avaya Aura® SBC public outside interface connection to the AT&T IP Toll Free service.

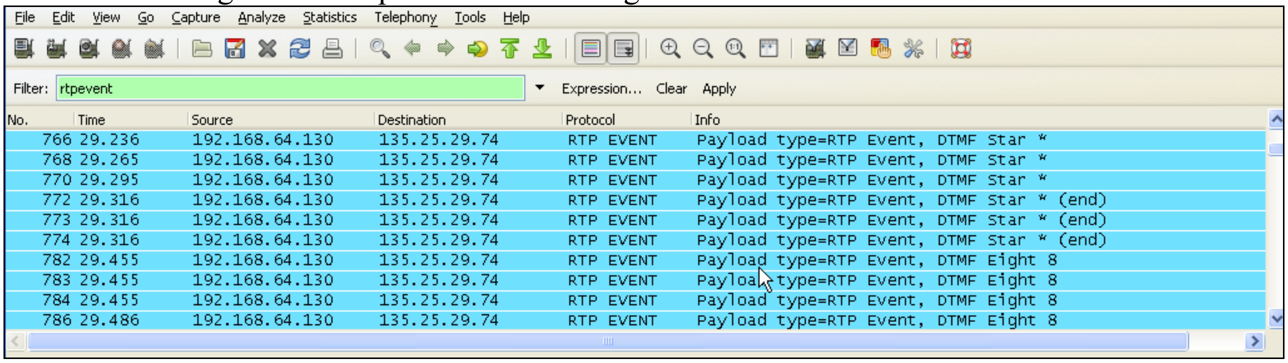
The following are examples of calls filtering on the SIP protocol.

File Edit View Go Capture Analyze Statistics Telephony Tools Help

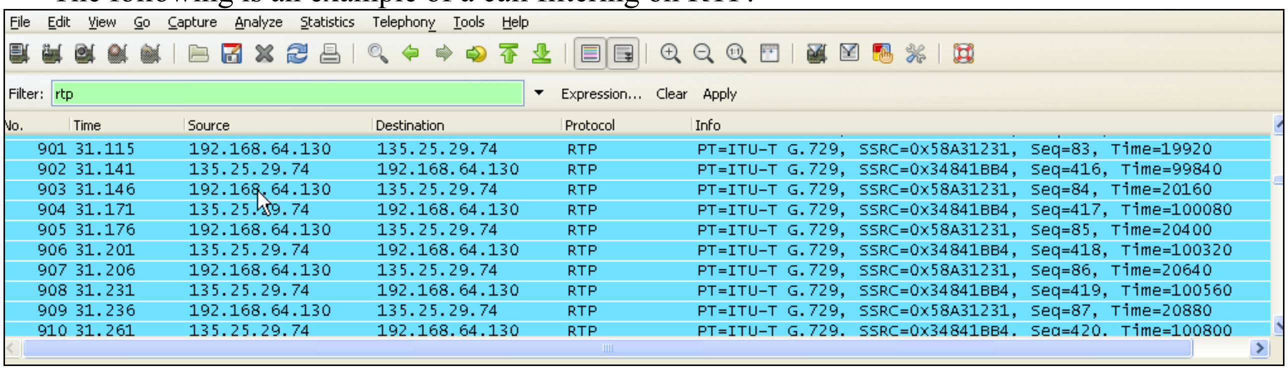
Filter: sip Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
25	18.493	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:0000011051@192.168.64.130:5060, with
26	18.495	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
27	18.573	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
168	20.562	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
170	20.572	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
178	20.672	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
433	24.398	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:7326712438@135.25.29.74:5060;transpor
436	24.433	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
441	24.484	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
442	24.495	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:7326712438@135.25.29.74:5060;transport=u

The following is an example of a call filtering on DTMF.



The following is an example of a call filtering on RTP.



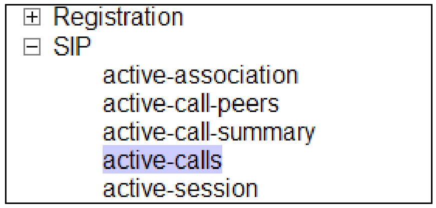
9.5. Avaya Aura® Session Border Controller Verification

This section contains verification steps that may be performed using the Avaya Aura® Session Border Controller.

9.5.1. Status Tab

Avaya SBC status information is available via the **Status** tab.

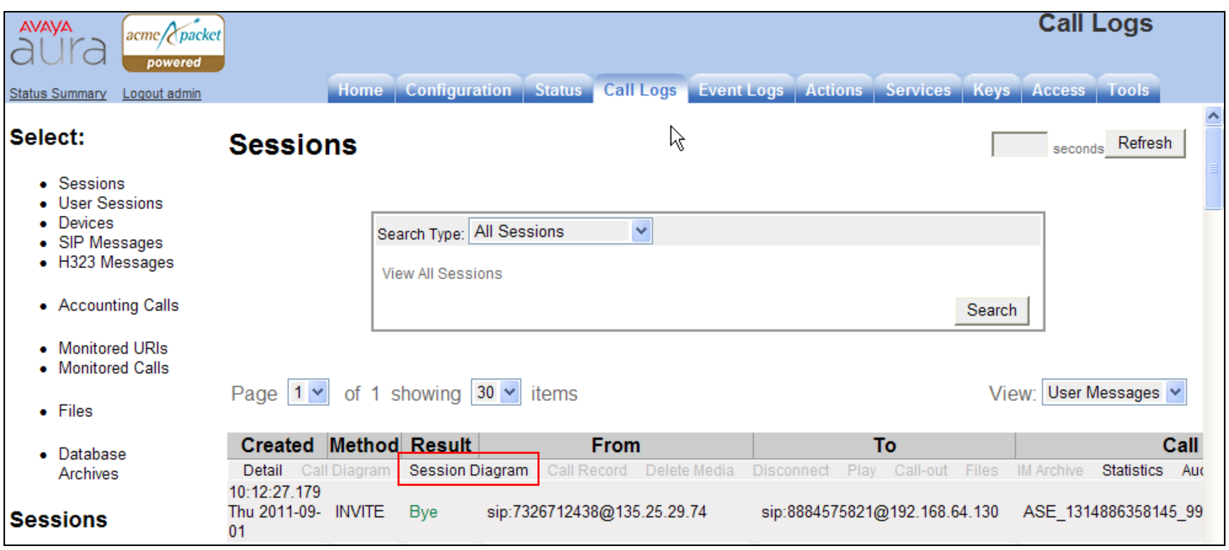
For example, there is a SIP heading on the left menu that can be expanded as shown below.



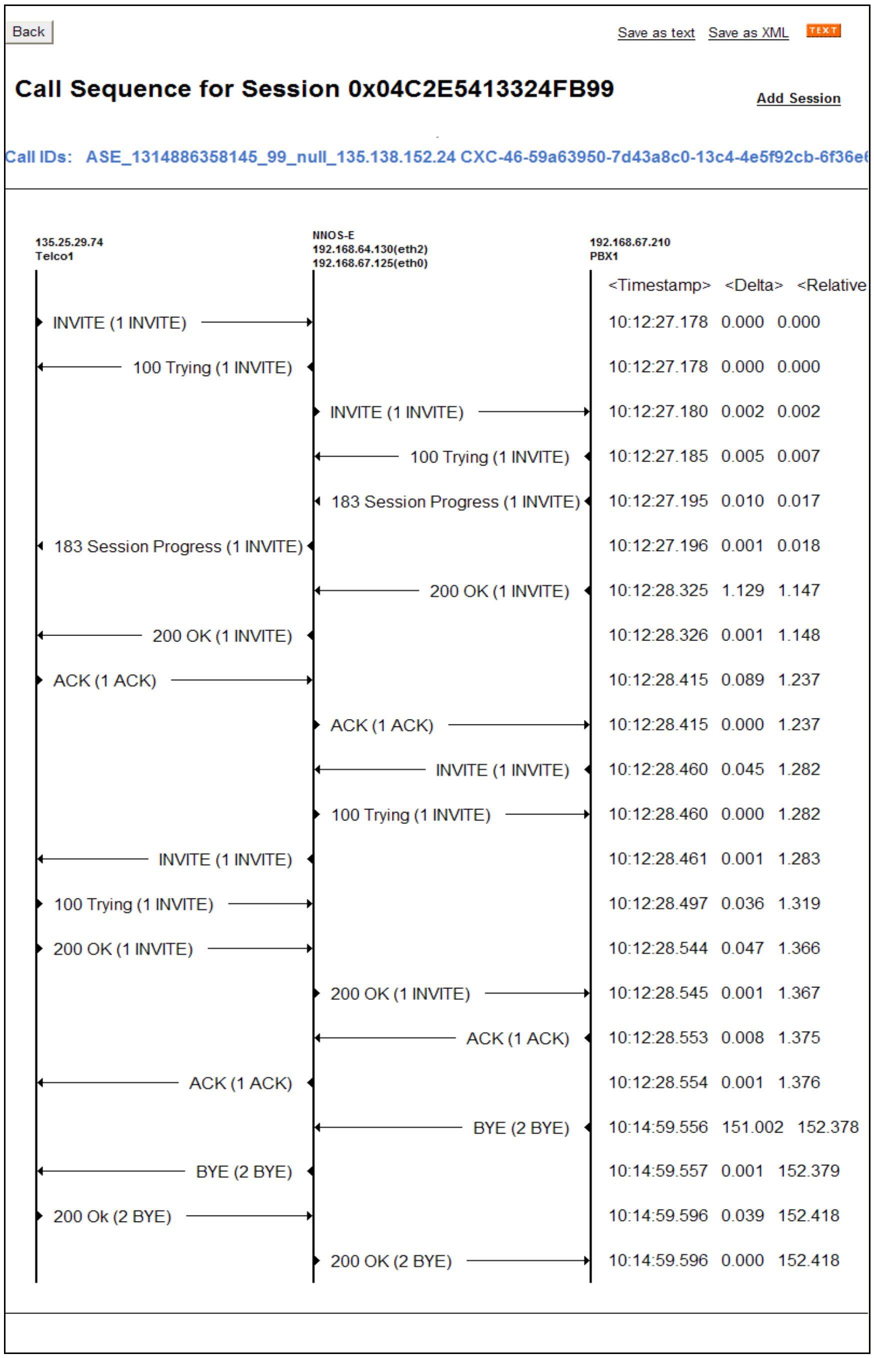
In the example below, **active-calls** was selected from the left, revealing details about an active inbound call from PSTN. Additional information about the call is available by moving the bottom scroll bar to the right (not shown).

9.5.2. Call Logs

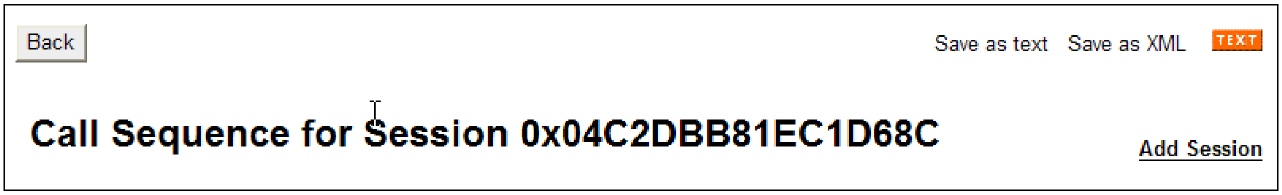
The **Call Logs** tab can provide useful diagnostic or troubleshooting information. In the following screen, the **SIP Messages** search capability can be observed. The following screen shows a portion of the **Call Logs** tab selected after an inbound call from PSTN.



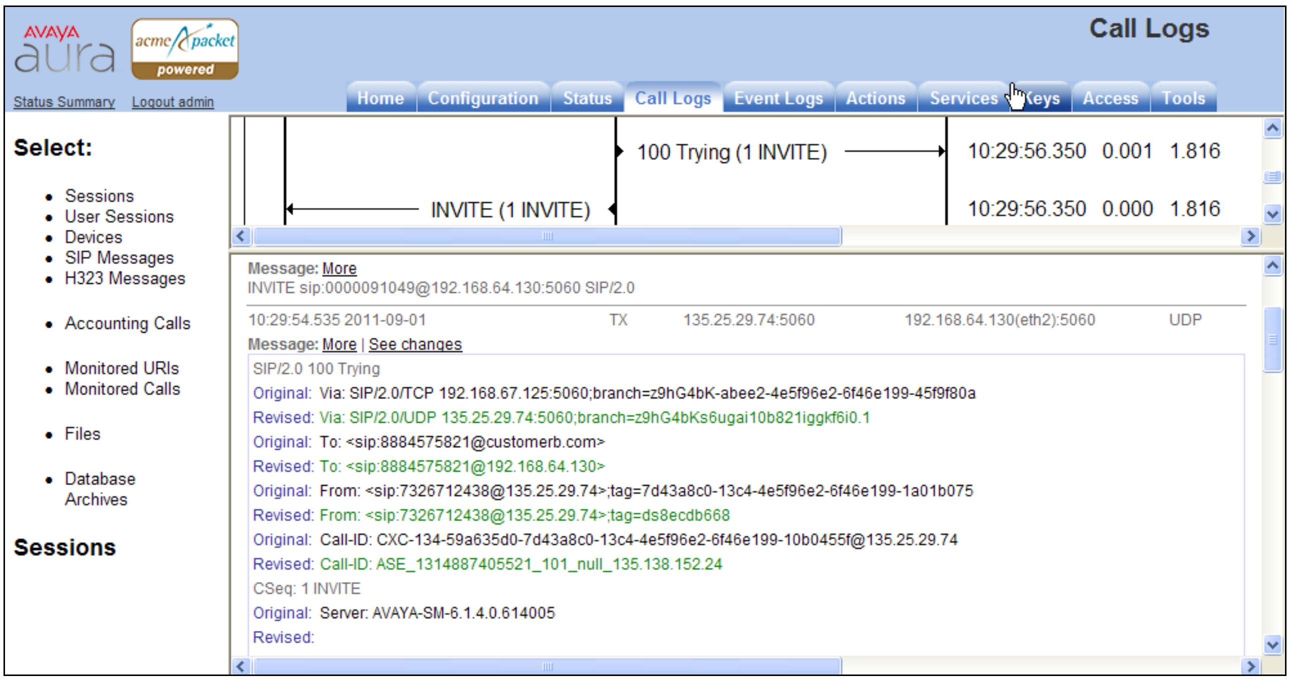
As shown below, to view a ladder diagram for the session, select the **Session Diagram** link. When the session window opens, expand the upper portion of the screen under the Call Sequence heading to display the ladder diagram. The following screen shows the ladder diagram for the inbound call. Note that the activity for both the inside private and outside public side of the SBC can be seen.



At the top right of the screen, the session may be saved as a text or XML file. If the session is saved as an XML file, using the **Save as XML** link, the xml file can be provided to support personnel that can open the session on another Avaya SBC for analysis.



The **Call Logs** tab also provides the capability to see modifications made to SIP headers by the SBC. Below the ladder diagram area is another screen section. Using the same Session Diagram as shown above, Scrolling down to the INVITE message sent by the SBC to AT&T. The **More** and **See changes** links was selected to expand the SIP message display and enable observation of the changes made by the SBC to the **Revised** message, as compared to the **Original** INVITE received from Session Manager.



10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Avaya Aura® SBC can be configured to interoperate successfully with the AT&T IP Toll Free service. 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.

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.

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, Issue 4, May 2011
[2] Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473 Issue 2.2, April 2011
[3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Issue 4.1, March 2011
[4] Administering Avaya Aura® System Manager, Document Number 03-603324, June 2010

Avaya Aura® Communication Manager

[5] Administering Avaya Aura® Communication Manager, Release 6.003-300509, Issue 6.0, June 2010*Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010
[6] Programming Call Vectors in Avaya Aura® Call Center, 6.0, June 2010

Avaya Modular Messaging

[7] Modular Messaging Multi-Site Guide Release 5.1, June 2009
[8] Modular Messaging Messaging Application Server (MAS) Administration Guide, July 2011

Avaya Aura™ Session Border Controller

[9] Installing and Configuring Avaya Aura® Session Border Controller, Release 6.0.1, November 2010 available at: http://support.avaya.com/css/P8/documents/100134970
[10] Avaya Aura® SBC System Administration Guide, V.6.0, 2010 available at: http://support.avaya.com/css/P8/documents/100111137

AT&T IP Toll Free Service Descriptions:

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

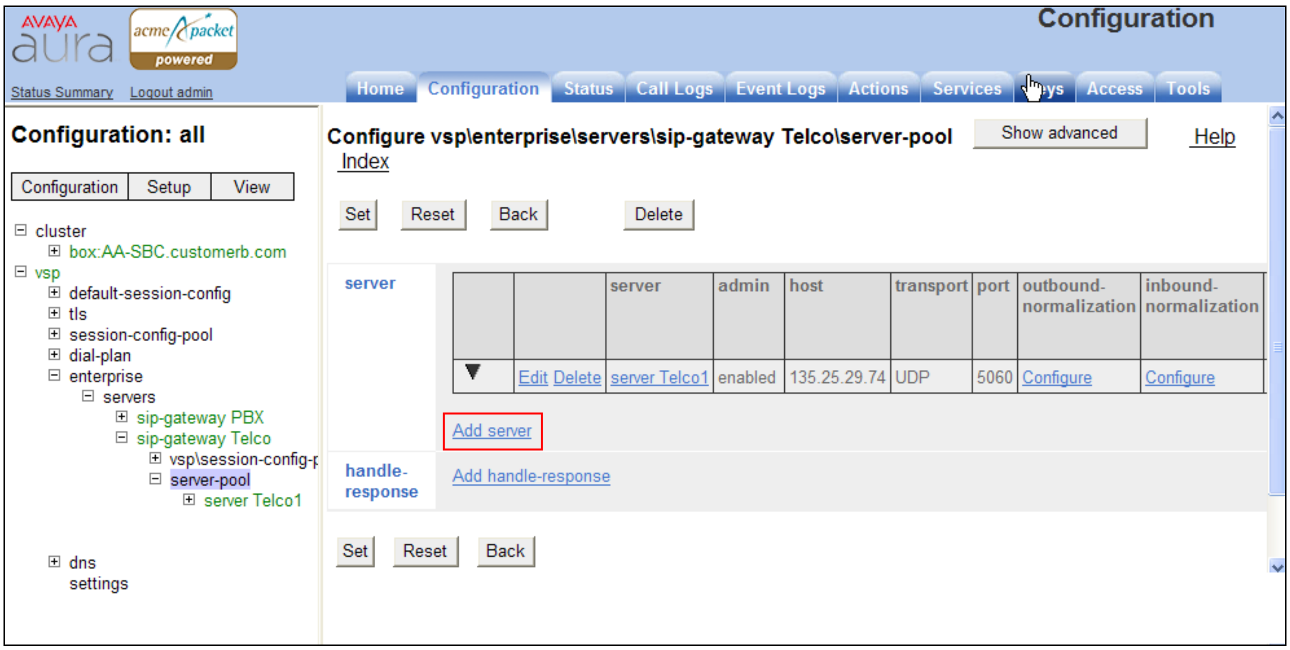
12. Addendum 1 – Avaya Aura® Session Border Controller Redundancy to Multiple AT&T Border Elements

AT&T may provide multiple network border elements for redundancy purposes. The Avaya Aura® SBC can be provisioned to support this redundant configuration.

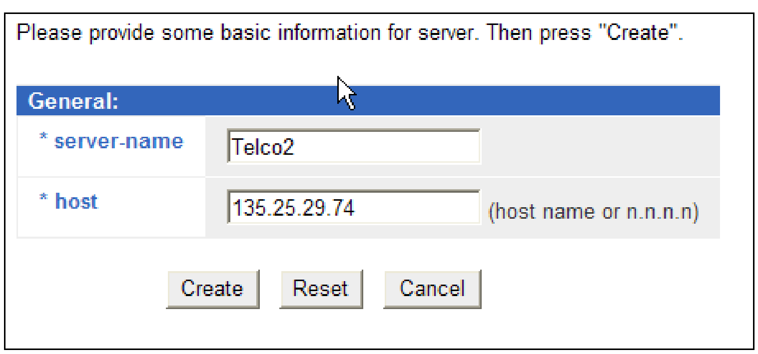
Given two AT&T border elements **135.25.29.74** and **135.25.29.75**, and building on the sip gateway configuration shown in **Section 8.2.4.1**, the Avaya Aura® SBC is provisioned as follows.

Step 1 - Go to **vsp → enterprise → servers → sip-gatewayTelco → server-pool** and the previously defined sip-gateway **Telco1** defined in **Section 8.2.4.1** will be displayed.

Step 2 – Click on **Add server**.

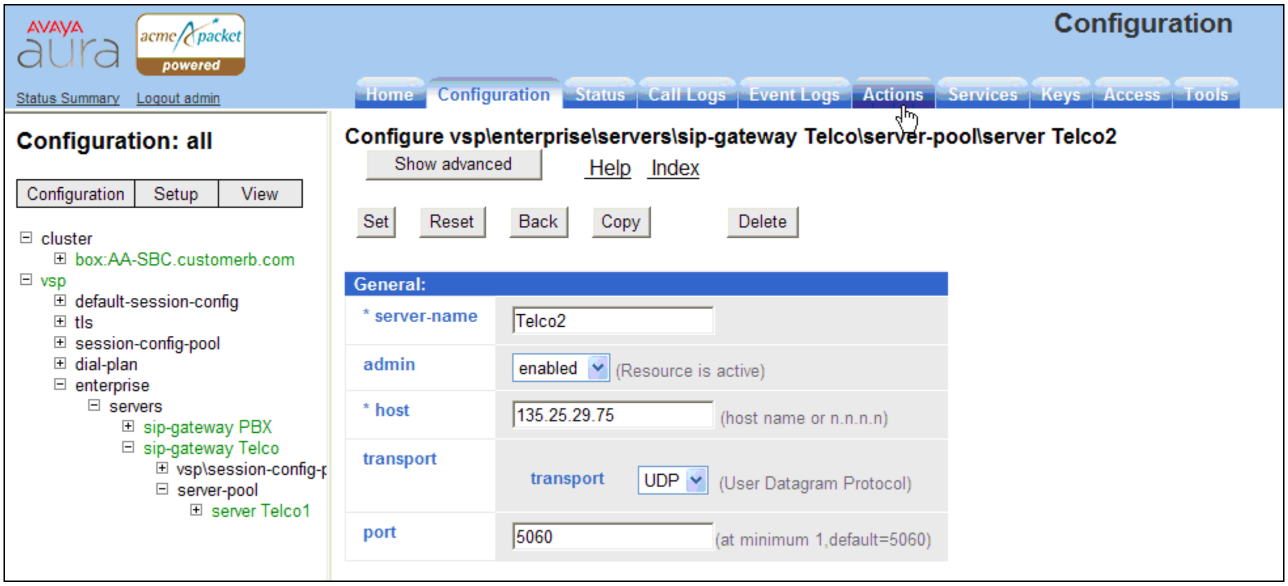


Step 3 – Enter a name in the server-name field (e.g **Telco2**) and enter the second AT&T boreder element IP address in the host field (e.g.**135.25.29.75**). Click on **Create**.

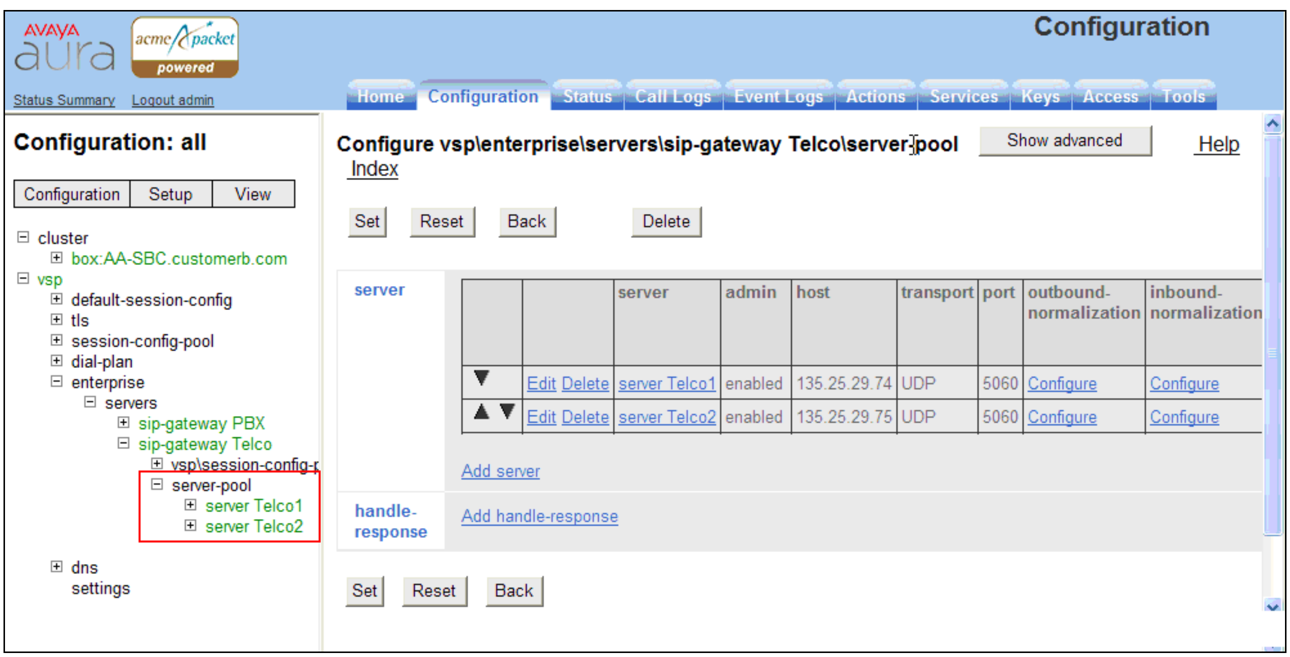


Step 4 – Enter the following:

- Admin is **enabled**.
- Transport protocol is **UDP**.
- Port is **5060**.



Step 5 - Click on the **Set** button to save. **Telco1**and **Telco2** will be displayed in the server-pool.



Step 5 - Proceed to save and activate the configuration as described in **Section 8.3**.

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.