Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and Acme Packet Net-Net Session Director with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and the Acme Packet Net-Net Session Director with the AT&T IP Toll Free service. The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura™ Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

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

TABLE OF CONTENTS

1. Introduction

These Application Notes describe the steps for configuring Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and the Acme Packet Net-Net Session Director with the AT&T IP Toll Free service. The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura™ Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. **Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.** Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see Section 2.2 for descriptions) to the Acme Packet Net-Net Session Director and subsequent routing to Avaya Aura™ Session Manager and then Avaya Aura™ Communication Manager skills and agents/phones.

1.2. 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. The "Connect with Avaya" section provides the worldwide support directory. 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.

1.3. Known Limitations

1. Although Avaya Aura™ Communication Manager release 5.2 supports the possibility of using SIP phones as agent stations, SIP phones were not tested as part of the configuration used to validate this solution.
2. If Avaya Aura™ Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Avaya Aura™ Communication Manager, then Avaya Aura™ Communication Manager selects a codec according to the priority order specified in the Avaya Aura™ 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.726 in that order, but the Avaya Aura™ Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Avaya Aura™ Communication Manager selects G.729A, not G.711. The practical resolution is to provision the Avaya Aura™ Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.

3. Avaya Aura™ Communication Manager does not support G.726 codec with the AT&T IP Toll Free service.

2. Reference Configuration

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

- Avaya Aura™ Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya Aura™ 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.
- Avaya Aura™ System Manager provides a common administration interface for centralized management of all Avaya Aura™ Session Manager instances in an enterprise.
- Avaya Aura™ Communication Manager provides the voice communications services for a particular enterprise site. In this sample configuration, Avaya Aura™ Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Avaya Aura™ Communication Manager. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "office" phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software, as well as Avaya 6400 Series Digital Telephones.
- The Acme Packet Net-Net Session Director (SD) 3800 provides SIP Session Border Controller (SBC) functionality, including address translation and UDP/TCP protocol mediation[1], between the AT&T IP Toll Free service and the enterprise internal network. For brevity, the Acme Packet Net-Net SD 3800 will be referred to as the Acme Packet SBC through the remainder of these Application Notes.
- Avaya Modular Messaging (in MultiSite mode in this sample configuration) provides the corporate voice messaging capabilities for enterprise users.

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

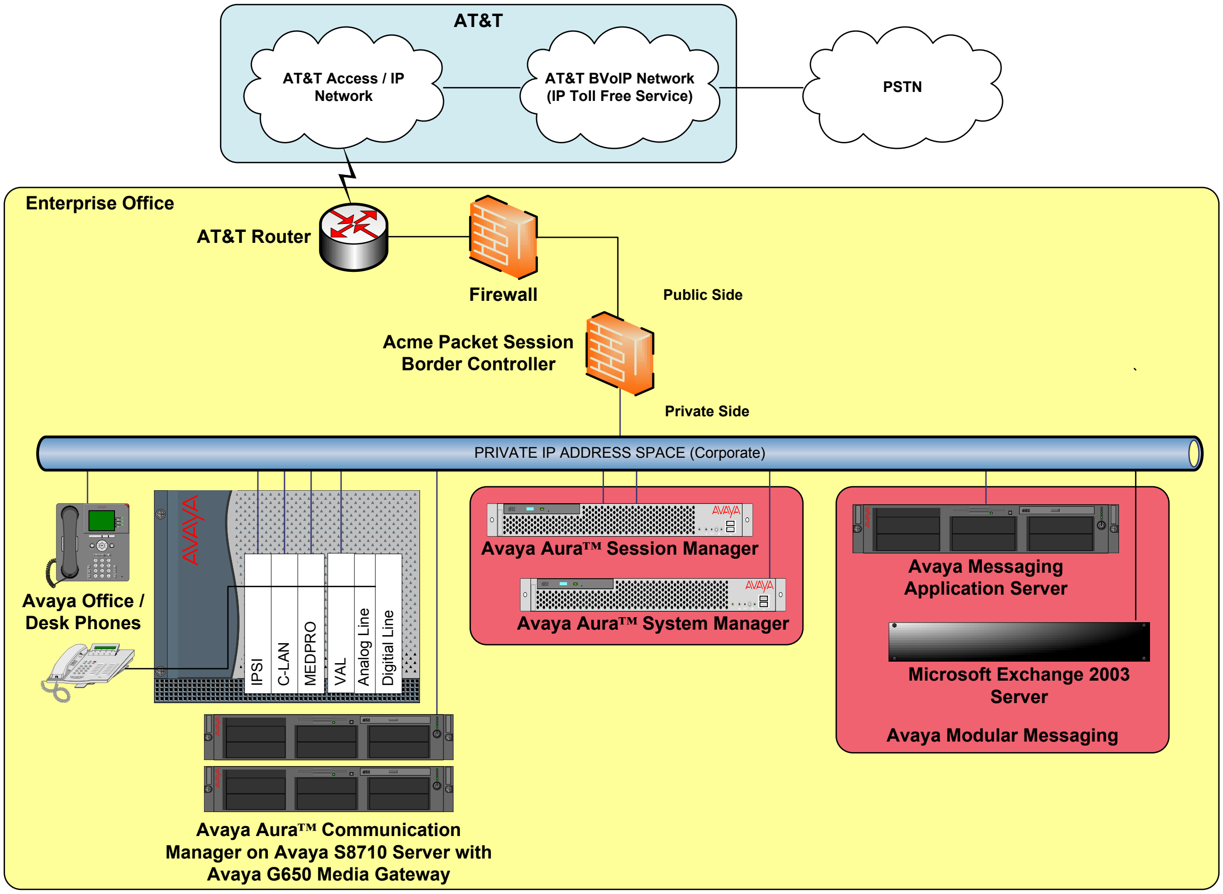

Figure 1: Sample Configuration

2.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the sample 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.

Component	Illustrative Value in these Application Notes
Avaya Aura™ System Manager	
Management IP Address	10.160.183.197
Avaya Aura™ Session Manager	
Management IP Address	10.160.183.207
SM100 Card IP Address	10.160.183.209
Avaya Aura™ Communication Manager	
C-LAN IP Address	10.160.179.110
Vector Directory Number (VDN) Extensions / Associated 10-digit Numbers	31xxx / 7328531xxx
Skill (Hunt Group) Extensions / Associated 10-	51xxx / 7328551xxx

Component	Illustrative Value in these Application Notes
digit Numbers	
Agent Extensions / Associated 10-digit Numbers	32xxx / 7328532xxx
Phone Extensions / Associated 10-digit Numbers	30xxx / 7328530xxx
Announcement Extensions / Associated 10-digit Numbers	52xxx / 7328552xxx
Voice Messaging Pilot Extension	30900
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	10.160.183.220
Microsoft Exchange 2003 Server	10.160.183.222
Pilot Number	9089530000
Acme Packet SBC	
IP Address of "Outside" (Public) Interface (connected to AT&T IP Toll Free Service)	10.160.177.210 (active) 10.160.177.211 (primary) 10.160.177.212 (secondary)
IP Address of "Inside" (Private) Interface (connected to Avaya elements)	10.160.183.219 (active) 10.160.183.217 (primary) 10.160.183.218 (secondary)
AT&T IP Toll Free Service	
Border Element IP Address	10.242.225.200
Digits Passed in SIP Request-URI	0000010xx

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by Avaya Aura™ Session Manager and Avaya Aura™ 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 Avaya Aura™ Session Manager and is subsequently routed to Avaya Aura™ Communication Manager.

1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to the Acme Packet SBC.
4. The Acme Packet SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Avaya Aura™ Session Manager.
5. Avaya Aura™ Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines to where the call should be routed next. In this case, Avaya Aura™ Session Manager routes the call to Avaya Aura™ Communication Manager.
6. Depending on the called number, Avaya Aura™ 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 phone.

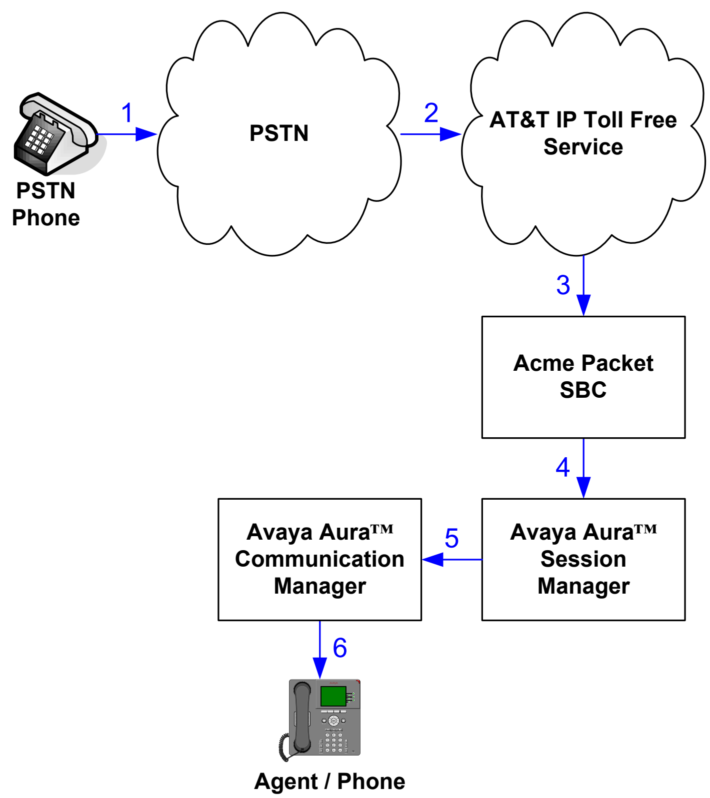

Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Phone

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

1. Same as the first five steps and Step 6b from the first call scenario.
2. The called Avaya Aura™ Communication Manager agent or phone does not answer the call, and the call covers to the agent's or phone's voicemail. Avaya Aura™ Communication Manager forwards[2] the call to Avaya Aura™ Session Manager.
3. Avaya Aura™ Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines to where the call should be routed next. In this case, Avaya Aura™ Session Manager routes the call to Avaya Modular Messaging. Avaya Modular Messaging answers the call and connects the caller to the called agent's or phone's voice mailbox. Note that the call[3] continues to go through Avaya Aura™ Communication Manager.

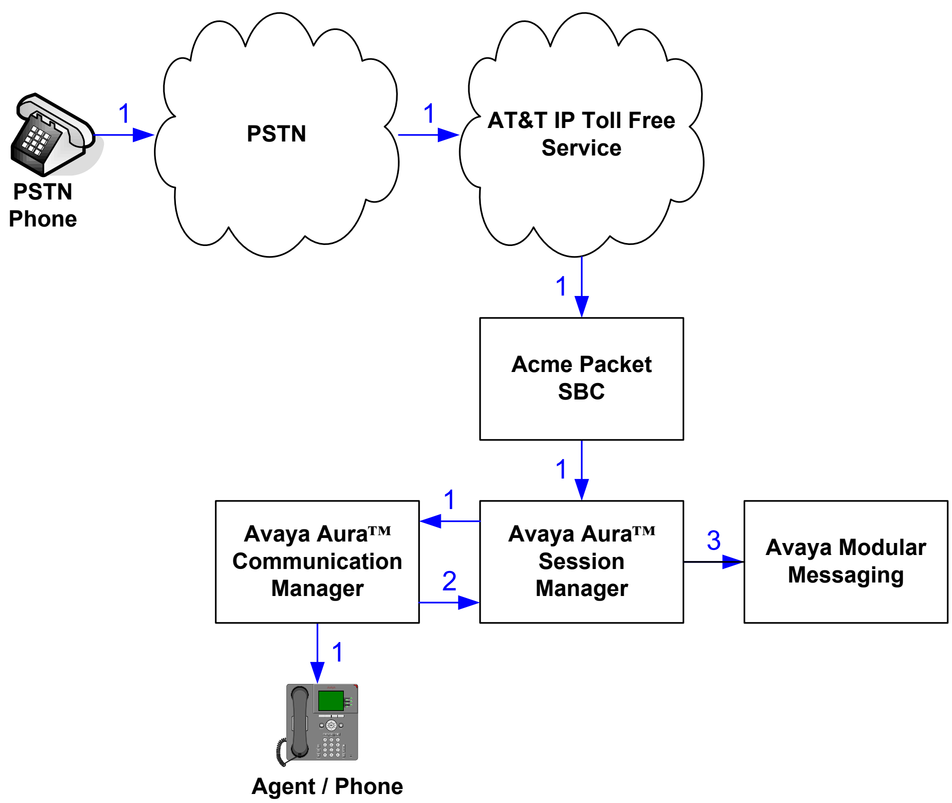

Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Phone Covered to Avaya Modular Messaging

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

3. Equipment and Software Validated

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

Component	Version
Avaya S8510 Server	Avaya Aura™ System Manager 1.1 SP1 (1.1.4.0.111013)
Avaya S8510 Server	Avaya Aura™ Session Manager 1.1 SP1 (1.1.4.0.111013)
SM100 Card	-
Avaya S8710 Server	Avaya Aura™ Communication Manager 5.2 with Service Pack 1 (R015x.02.0.947.3 with update 17294)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW046
TN799DP Control-LAN (C-LAN)	HW01 FW032
TN2302AP IP Media Processor (MedPro)	HW20 FW120
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW047
TN2501AP VAL-ANNOUNCEMENT	HW02 FW021
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	000005
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0 Service Pack 1
Avaya 9640 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0 Service Pack 1
Avaya 4610SW IP Telephone	2.9 SP1 (2.9.1)
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Server	Avaya Modular Messaging 5.1 with Patch 8 (9.0.370.18)
Microsoft Exchange 2003 Server on Microsoft Windows Server 2003 R2 Enterprise Edition Service Pack 2	6.5.7638.1
Fax	-
Acme Packet Net-Net Session Director 3800	SCX6.1.0 MR-1 Patch 1 (Build 277)
AT&T IP Toll Free Service	VNI 14

Table 2: Equipment and Software Versions

4. Avaya Aura™ Session Manager

These Application Notes assume that basic Avaya Aura™ System Manager and Avaya Aura™ Session Manager administration has already been performed. Consult [1] and [2] for further details if necessary. Configuration of Avaya Aura™ Session Manager is performed from Avaya Aura™ System Manager. To invoke the Avaya Aura™ System Manager Common Console, launch a web browser, enter http://<IP address of the Avaya Aura™ System Manager server>/IMSM in the URL, and log in with the appropriate credentials.

4.1. Background

Avaya Aura™ Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. Avaya Aura™ 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 Avaya Aura™ 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 Avaya Aura™ Session Manager and relies on Avaya Aura™ 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 Avaya Aura™ System Manager.

When calls arrive at Avaya Aura™ Session Manager from a SIP Entity, Avaya Aura™ 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 Avaya Aura™ Session Manager. Avaya Aura™ Session Manager then matches the calls against certain criteria embodied in profiles termed "Dial Patterns", and determines the destination SIP Entities based on "Network Routing Policies" specified in the matching Dial Patterns. Lastly, before the calls are routed to the respective destinations, Avaya Aura™ 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.

4.2. Network Routing Policies

Network Routing Policies define how Avaya Aura™ Session Manager routes calls between SIP network elements. A Network Routing Policy is dependent on the administration of several inter-related items:

- SIP Entities – SIP Entities represent SIP network elements such as Avaya Aura™ Session Manager instances, Avaya Aura™ Communication Manager systems, Session Border Controllers, SIP gateways, SIP trunks, and other SIP network devices.
- Entity Links – Entity Links define the SIP trunk/link parameters, e.g., ports, protocol (UDP/TCP/TLS), and trust relationship, between Avaya Aura™ Session Manager instances and other SIP Entities.
- SIP Domains – SIP Domains are the domains for which Avaya Aura™ Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Avaya

Aura™ Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, Avaya Aura™ Session Manager routes those calls to another SIP proxy (either a pre-defined default SIP proxy or one discovered through DNS).

- Locations – Locations define the physical and/or logical locations in which SIP Entities reside. Call Admission Control (CAC) / bandwidth management may be administered for each location to limit the number of calls to and from a particular Location.
- Adaptations – Adaptations are used to apply any necessary protocol adaptations, e.g., modify SIP headers, and apply any necessary digit conversions for the purpose of inter-working with specific SIP Entities. For example, an AT&T-specific Adaptation is used in these Application Notes to remove SIP History-Info headers from SIP messages sent to the AT&T IP Toll Free service network. As another example, basic "Digit Conversion" Adaptations are used in this sample configuration to convert digit strings in "destination" and "origination" type headers, e.g., Request-URI and P-Asserted Identity, respectively, of SIP messages sent to and received from SIP Entities.
- Dial Patterns – A Dial Pattern specifies a set of criteria and a set of Network Routing Policies for routing calls that match the criteria. The criteria include the called party number and SIP domain in the Request-URI, and the Location from which the call originated. For example, if a call arrives at Avaya Aura™ Session Manager and matches a certain Dial Pattern, then Avaya Aura™ Session Manager selects one[4] of the Network Routing Policies specified in the Dial Pattern. The selected Network Routing Policy in turn specifies the SIP Entity to which the call is to be routed. Note that Dial Patterns are matched after ingress Adaptations have already been applied.
- Time Ranges – Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Network Routing Policy may be associated with one or more Time Ranges during which the Network Routing Policy is in effect. For example, for a Dial Pattern administered with two Network Routing Policies, one Network Routing Policy can be in effect on weekday business hours and the other Network Routing Policy can be in effect on weekday off-hours and weekends.

The general strategy employed in this sample configuration with regard to Called Party Number manipulation and matching, and call routing is as follows:

- Use common number formats and uniform numbers in matching called party numbers for routing decisions.
- On ingress to Avaya Aura™ Session Manager, apply any called party number modifications necessary to "normalize" the number to a common format or uniform number[5]. For example,

---

[4] The Network Routing Policy in effect at that time with highest ranking is attempted first. If that Network Routing Policy fails, then the Network Routing Policy with the next highest rankings is attempted, and so on.

[5] This sample configuration deviates from this general strategy for inbound AT&T IP Toll Free service calls – the AT&T IP Toll Free service called party numbers are not modified on ingress to Avaya Aura™ Session Manager, but rather on egress from Avaya Aura™ Session Manager. The main reason for the deviation is only to distinctly illustrate the path in Avaya Aura™ Session Manager when routing AT&T IP Toll Free service calls (as opposed to

assume that there are three SIP Entities representing three different Avaya Aura™ Communication Manager systems, and a SIP Entity representing a centralized voicemail system, e.g., Avaya Modular Messaging in MultiSite mode. Further, assume that each Avaya Aura™ Session Manager system dials a different pilot extension to call Avaya Modular Messaging. To simplify the routing for such calls, in Avaya Aura™ Session Manager, modify the different called pilot extensions to a uniform pilot number. The uniform pilot number can then be used in routing decisions, thereby minimizing the number of Dial Patterns that need to be administered to match and route calls to Avaya Modular Messaging.
- On egress from SM, apply any called party number modifications necessary to conform to the expectations of the next-hop SIP Entity. For example, on egress from Avaya Aura™ Session Manager to Avaya Aura™ Communication Manager, modify the called party number such that the number is consistent with the dial plan on Avaya Aura™ Communication Manager.

Of course, the above is just one of many possible strategies that can be implemented with Avaya Aura™ Session Manager.

To view the sequenced steps required for configuring network routing policies, click on "**Network Routing Policy**" in the left pane of the Avaya Aura™ System Manager Common Console.

routing other calls). An administrator who is familiar with Avaya Aura™ Session Manager digit conversion capabilities, however, could implement the general strategy instead for inbound AT&T IP Toll Free service calls.

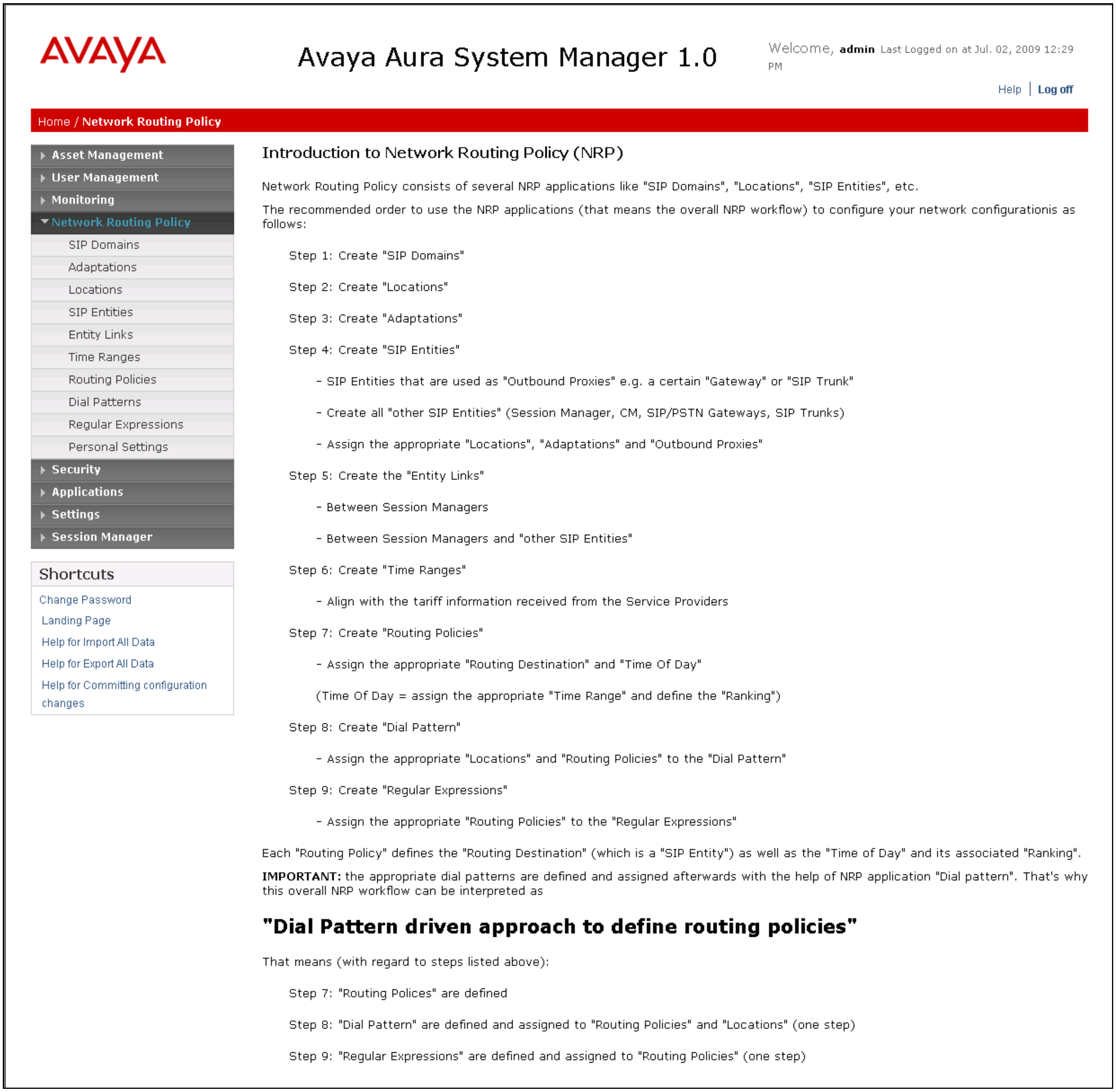

Figure 4: Introduction to Network Routing Policy (NRP) Page

4.3. SIP Domains

The steps in this section specify the SIP domains for which Avaya Aura™ Session Manager is authoritative.

1. In the left pane under **Network Routing Policy**, click on "**SIP Domains**". In the **SIP Domains** page (not shown), click on "**New**".
2. Continuing in the **SIP Domains** page, enter a SIP domain for **Name** and click on "**Commit**".

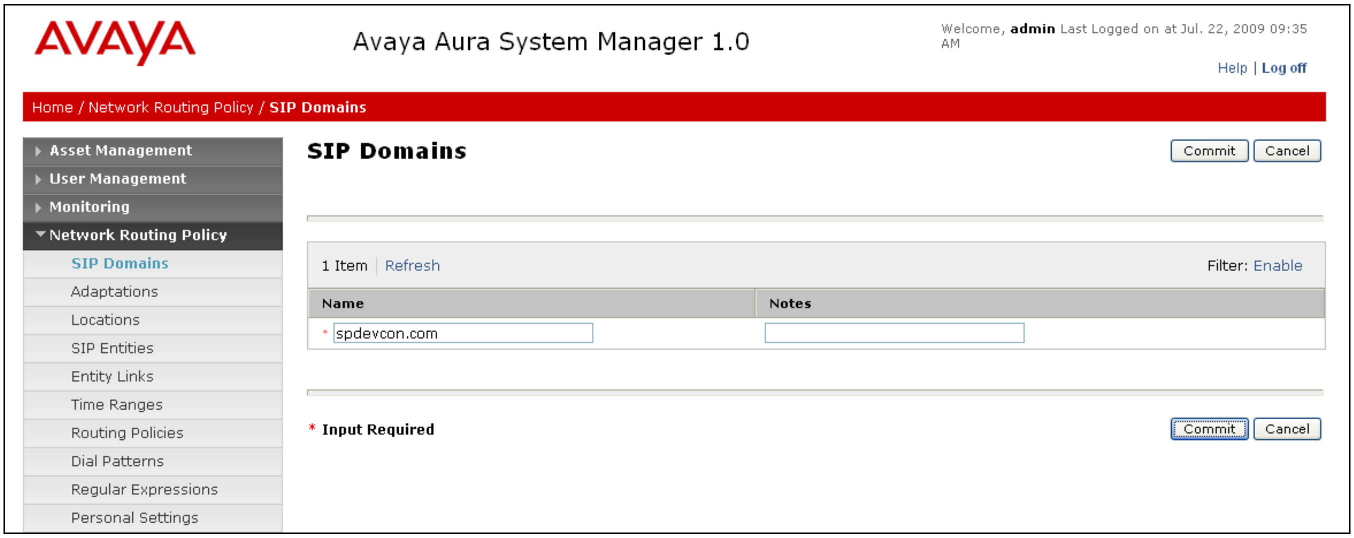

Figure 5: SIP Domains Page

3. Repeat Steps 1 - 2 to add any additional SIP domains.

4.4. Locations

The steps in this section define the physical and/or logical locations in which SIP Entities reside.

1. In the left pane under **Network Routing Policy**, click on "**Locations**". In the **Location** page (not shown), click on "**New**".
2. In the **Location Details** page, enter a descriptive **Name**.
3. [Optional] To limit the number of calls going to and from this Location, i.e., apply CAC, specify the **Managed Bandwith** and **Average Bandwidth per Call**.
4. [Optional] To identify IP addresses associated with this Location, add **Location Pattern** entries accordingly.
5. Click on "**Commit**".

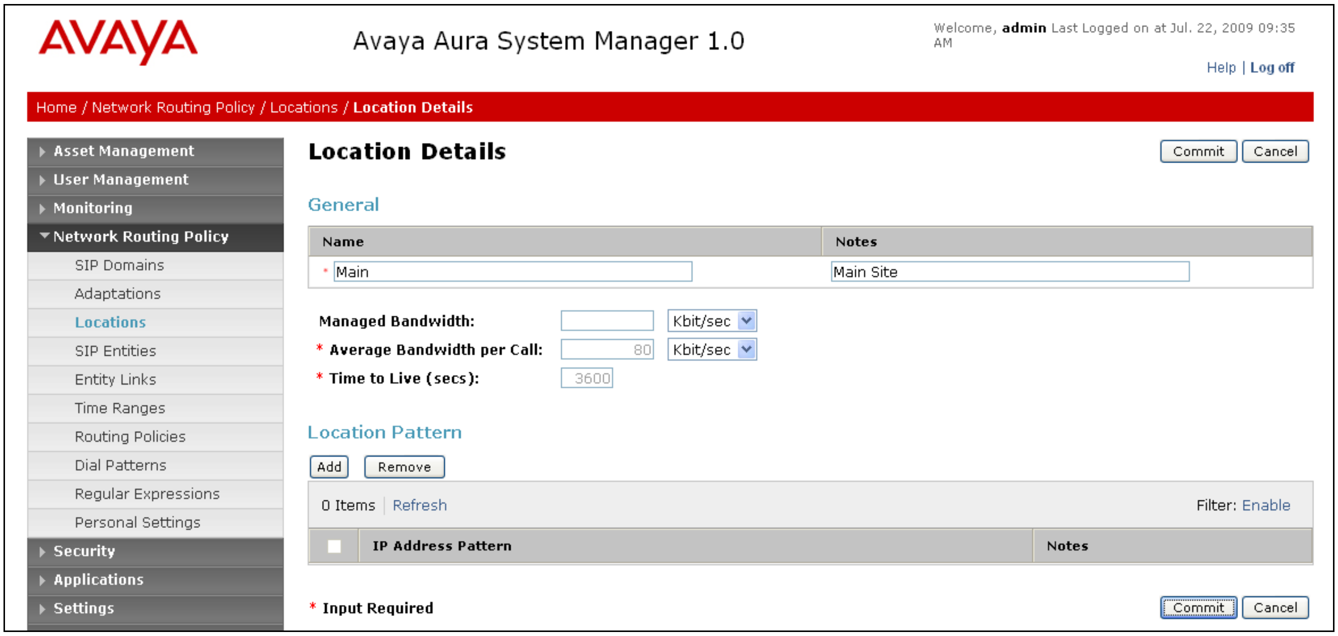

Figure 6: Location Details Page – Main Site

6. Repeat Steps 1 - 5 to add any additional Locations. In this sample configuration, two Locations named "Main" and "Site 1" are defined. As described in subsequent sections, Avaya Aura™ Session Manager, the Acme Packet SBC, and Avaya Modular Messaging are assigned to the "Main" Location, while Avaya Aura™ Communication Manager is assigned to the "Site 1" Location.

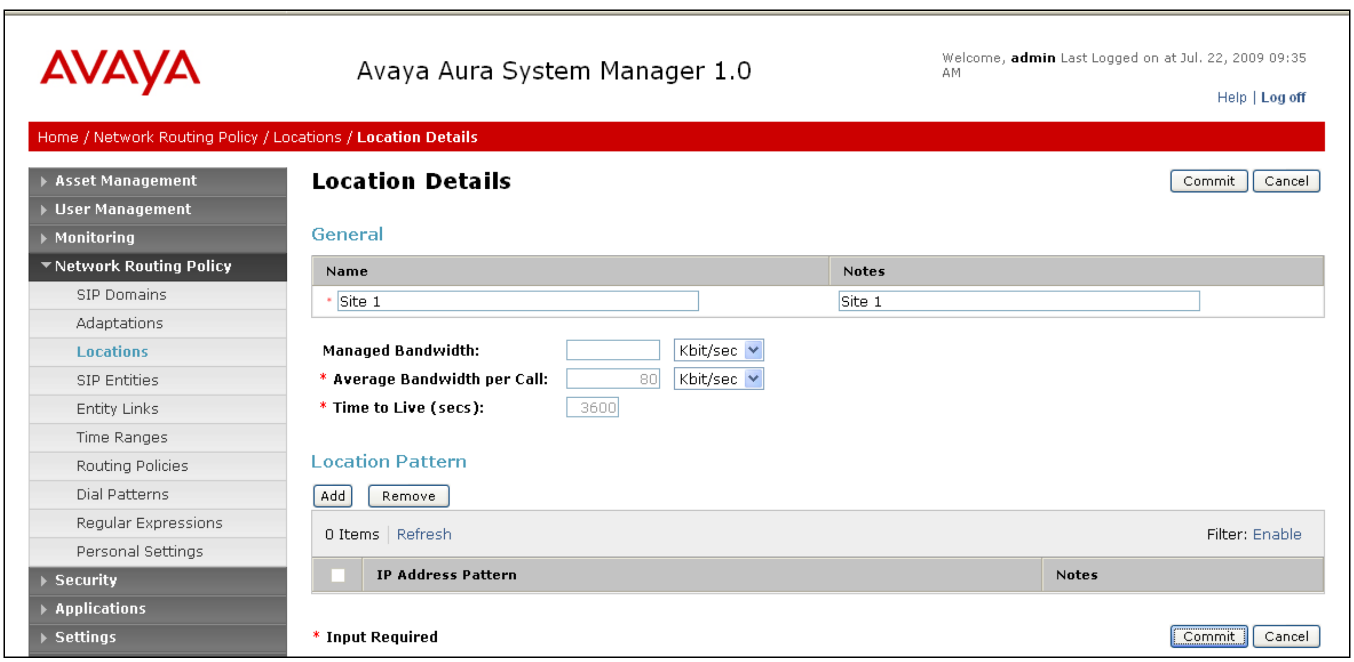

Figure 7: Location Details Page – Site 1

4.5. Adaptations

In this section, Adaptations are administered for the following purposes:

- Modification[6] of SIP messages sent to the AT&T IP Toll Free service.
- Modification of digit strings in URIs of "origination" and "destination" type headers in SIP messages sent to and received from Avaya Aura™ Communication Manager.
- Modification of digit strings in URIs of "origination" and "destination" type headers in SIP messages sent to and received from Avaya Modular Messaging.

4.5.1. Adaptation for AT&T

The Adaptation administered in this section is applied to SIP messages sent to the AT&T IP Toll Free service (by way of the Acme Packet SBC in the Main Location).

1. In the left pane under **Network Routing Policy**, click on "**Adaptations**". In the **Adaptations** page (not shown), click on "**New**".
2. In the **Adaptation Details** page, enter a descriptive **Name** and "**AttAdapter**" for **Adaptation Module,** and click on "**Commit**".

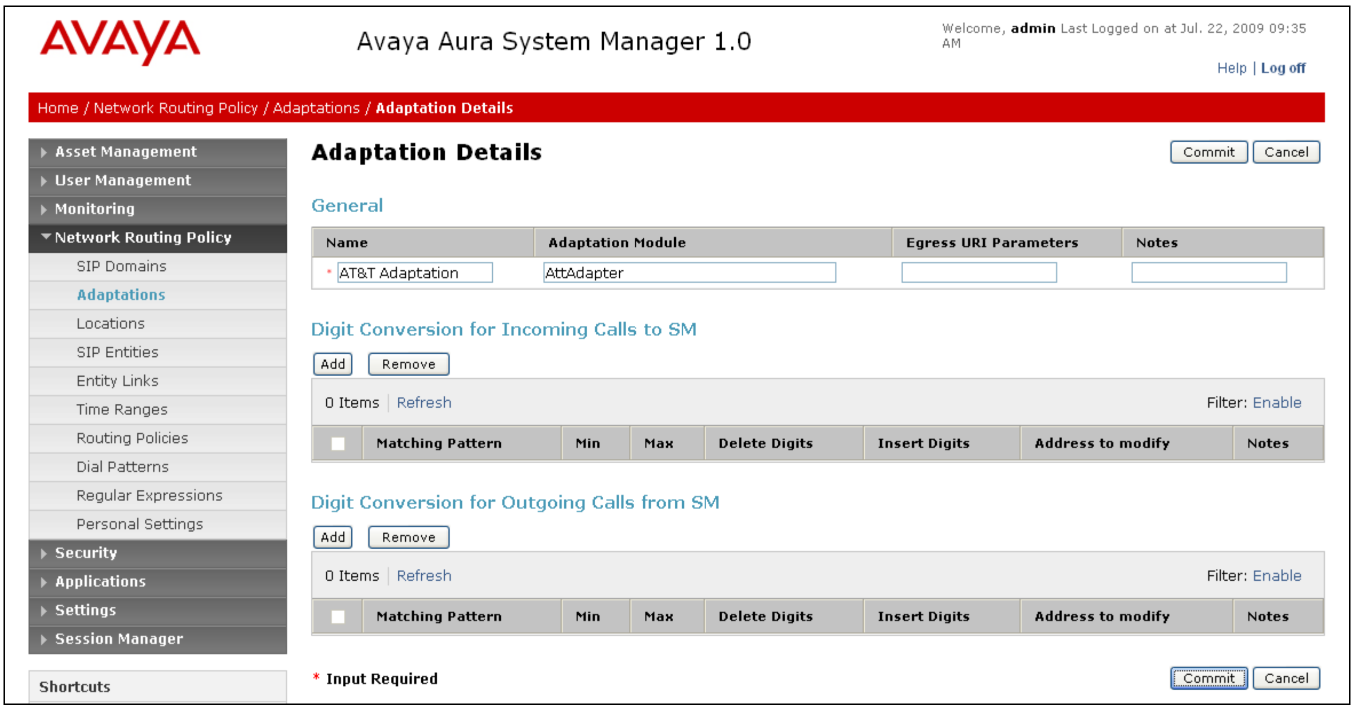

Figure 8: Adaptation Details Page – Adaptation for AT&T

4.5.2. Adaptation for Avaya Aura™ Communication Manager

The Adaptation administered in this section is used for digit conversion on SIP messages to and from Avaya Aura™ Communication Manager (in the Site 1 Location) as follows:

[6] Currently, the Adaptation removes the History-Info header.

- On egress SIP messages to Avaya Aura™ Communication Manager where the Request-URI contains an AT&T IP Toll Free service number, the Adaptation converts the number to an extension on Avaya Aura™ Communication Manager.
- On egress SIP messages to Avaya Aura™ Communication Manager where the Request-URI and/or Message-Account[7] header contains a 10-digit number associated with an extension on Avaya Aura™ Communication Manager, the Adaptation converts the number to the extension.
- On ingress SIP messages from Avaya Aura™ Communication Manager where the P-Asserted-Identity[8] header contains an extension on Avaya Aura™ Communication Manager, the Adaptation converts the extension to a 10-digit number.
- On ingress SIP messages from Avaya Aura™ Communication Manager where the Request-URI contains the Avaya Modular Messaging pilot extension (as dialed by Avaya Aura™ Communication Manager), the Adaptation converts the pilot extension to a uniform 10-digit pilot number[9].

1. In the **Adaptations** page (not shown), click on "**New**".
2. In the **Adaptation Details** page, enter a descriptive **Name** and "**DigitConversionAdapter**" for **Adaptation Module**.
3. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to provision an entry for converting a range of AT&T IP Toll Free service numbers to extensions on Avaya Aura™ Communication Manager. Provision the entry as follows:
   - **Matching Pattern –** Enter enough leading digits to uniquely match the number range, specifically the range of numbers contained in the Request-URI of inbound SIP INVITE messages from the AT&T IP Toll Free service.
   - **Min** and **Max –** Enter the total number of digits in the number range.
   - **Delete Digits** and **Insert Digits –** If necessary, enter the number of leading digits that need to be deleted from the number range, and the specific leading digits that need to be prefixed to the number range, respectively, in order to match an extension range on Avaya Aura™ Communication Manager.
   - **Address to modify –** Select "**destination**".

   In the sample configuration, the AT&T IP Toll Free service sends a 9-digit string in the range 0000010xx in the Request-URI. Thus the first entry in the **Digit Conversion for Outgoing Calls from SM** table in **Figure 9** matches the number range 0000010xx, deletes the leading five digits, and prefixes a leading "**3**" to the resulting number range, to match the extension range 310xx on Avaya Aura™ Communication Manager.

---

[7] Present in SIP NOTIFY messages from Avaya Modular Messaging to indicate the voice mailbox number to which the message pertains.
[8] Typically identifies the connected party number when sent by an answering party (or the calling party number when sent by the calling party).
[9] With the assumption that the pilot extensions dialed by other SIP Entities, e.g., other Avaya Aura™ Communication Manager systems connected to this Avaya Aura™ Session Manager, will also be converted to the same 10-digit pilot number.

4. Repeat Step 3 as necessary to provision additional entries to cover all expected ranges of AT&T IP Toll Free service numbers.
5. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to provision an entry for converting a range of 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager to those extensions. Provision the entry as follows:
   - **Matching Pattern –** Enter enough leading digits to uniquely match the number range, specifically the range of numbers contained in the Request-URI of inbound SIP INVITE messages, and the Request-URI and Message-Account header of SIP NOTIFY messages from Avaya Modular Messaging.
   - **Min** and **Max –** Enter "**10**".
   - **Delete Digits** and **Insert Digits –** If necessary, enter the number of leading digits that need to be deleted from the number range, and the specific leading digits that need to be prefixed to the number range, respectively, in order to match an extension range on Avaya Aura™ Communication Manager.
   - **Address to modify –** Select "**destination**".

   In the sample configuration, Avaya Modular Messaging sends a 10-digit string in the range 732853xxxx in the Request-URI and Message-Account header. Thus the second entry in the **Digit Conversion for Outgoing Calls from SM** table in **Figure 9** matches the number range 732853xxxx and deletes the leading five digits to match the extension range 3xxxx on Avaya Aura™ Communication Manager.
6. Repeat Step 5 as necessary to provision additional entries to cover all expected ranges of 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager.
7. In the **Digit Conversion for Incoming Calls to SM** section, click on "**Add**" to provision an entry for converting a range of extensions on Avaya Aura™ Communication Manager to the associated 10-digit numbers. Provision the entry as follows:
   - **Matching Pattern –** Enter enough leading digits to uniquely match the extension range.
   - **Min** and **Max –** Enter the total number of digits in the extension range.
   - **Delete Digits** and **Insert Digits –** If necessary, enter the number of leading digits that need to be deleted from the extension range, and the specific leading digits that need to be prefixed to the extension range, respectively, in order to form the 10-digit number range.
   - **Address to modify –** Select "**origination**".

   In the sample configuration, Avaya Aura™ Communication Manager sends a 5-digit string in the range 2xxxx, 3xxxx, or 5xxxx in the P-Asserted-Identity header. Thus the first, second, and fourth entries in the **Digit Conversion for Incoming Calls to SM** table in **Figure 9** match the number ranges 2xxxx, 3xxxx, and 5xxxx, respectively, and prefixes "**73285"** to those number ranges to form 10-digit numbers.
8. Repeat Step 7 as necessary to provision additional entries to cover all extension ranges on Avaya Aura™ Communication Manager.

9. In the **Digit Conversion for Incoming Calls to SM** section, click on “**Add**” to provision an entry for converting the Avaya Modular Messaging pilot extension as dialed by Avaya Aura™ Communication Manager to a uniform 10-digit pilot number. Provision the entry as follows:
   - **Matching Pattern** – Enter the pilot extension.
   - **Min** and **Max** – Enter the total number of digits in the pilot extension.
   - **Delete Digits** and **Insert Digits** – If necessary, enter the number of leading digits that need to be deleted from the extension range, and the specific leading digits that need to be prefixed to the pilot extension, respectively, in order to form the uniform 10-digit pilot number.
   - **Address to modify** – Select “**destination**”.

   In the sample configuration, Avaya Aura™ Communication Manager sends “**30900**” in the Request-URI. Thus the third entry in the **Digit Conversion for Incoming Calls to SM** table in **Figure 9** matches “**30900**”, deletes all five digits, and inserts “**9089530000**”.
10. Click on “**Commit**”.

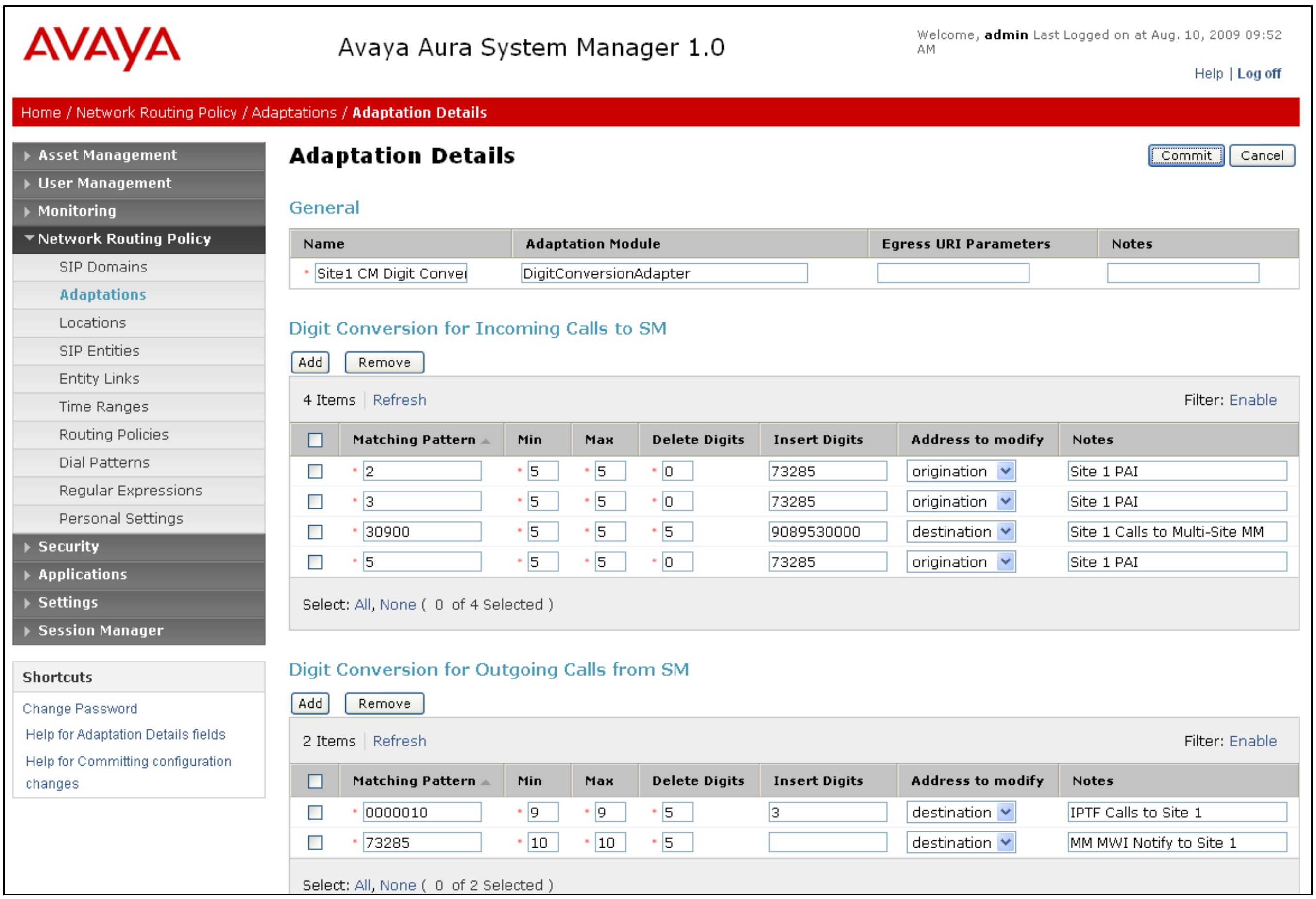

Figure 9: Adaptation Details Page – Adaptation for Avaya Aura™ Communication Manager

4.5.3. 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 (in the Main Location) as follows:

- On egress SIP messages to Avaya Modular Messaging where the Request-URI contains the Avaya Modular Messaging uniform pilot number, the Adaptation prefixes a leading "1" to the number in order to conform to the numbering plan administered on Avaya Modular Messaging for this sample configuration[10]. Recall that in Section 4.5.2, on ingress SIP messages from Avaya Aura™ Communication Manager, the pilot extension contained in the Request-URI is converted to a uniform 10-digit pilot number. Here, a leading "1" is prefixed to that uniform pilot number in the Request-URI on egress to Avaya Modular Messaging.
- On egress SIP messages to Avaya Modular Messaging where the P-Asserted-Identity header contains a 10-digit number associated with an extension on Avaya Aura™ Communication Manager, the Adaptation prefixes a leading "1" to the number in order to conform to the numbering plan administered on Avaya Modular Messaging for this sample configuration. Recall that in Section 4.5.2, on ingress SIP messages from Avaya Aura™ Communication Manager, the extensions contained in the P-Asserted-Identity header are converted to 10-digit numbers. Here, a leading "1" is prefixed to those 10-digit numbers in the P-Asserted-Identity header on egress to Avaya Modular Messaging.
- On ingress SIP messages from Avaya Modular Messaging where the Request-URI and/or Message-Account header contains an 11-digit number with a leading "1", the Adaptation removes the leading "1".

    1. In the **Adaptations** page (not shown), click on "**New**".
    2. In the **Adaptation Details** page, enter a descriptive **Name** and "**DigitConversionAdapter**" for **Adaptation Module**.
    3. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to configure an entry for prefixing a leading "1" to the Avaya Modular Messaging uniform pilot number. Provision the entry as follows:
        - **Matching Pattern –** Enter the Avaya Modular Messaging uniform pilot number.
        - **Min** and **Max –** Enter "**10**".
        - **Delete Digits –** Enter "**0**".
        - **Insert Digits –** Enter "**1**".
        - **Address to modify –** Select "**destination**".
    4. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to configure an entry for prefixing a leading "1" to a range of 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager. Provision the entry as follows:
        - **Matching Pattern –** Enter enough leading digits to uniquely match the number range, specifically the range of 10-digit numbers contained in the P-Asserted-Identity header (converted from extensions to 10-digit numbers in Section 4.5.2

[10] The Avaya Modular Messaging Multi-Site numbering plan in this sample configuration actually requires E.164 numbering, so prefixing/removing a leading "+" is also necessary and is done in Avaya Modular Messaging. This topic is beyond the scope of these Application Notes. Consult [7], [8], [9], and [10] for further information.

Steps 7 - 8) of inbound SIP INVITE messages from Avaya Aura™ Communication Manager.

- **Min** and **Max** – Enter "**10**".
- **Delete Digits** – Enter "**0**".
- **Insert Digits** – Enter "**1**".
- **Address to modify** – Select "**origination**".

5. Repeat Step 4 as necessary to provision additional entries to cover all expected ranges of 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager.
6. In the **Digit Conversion for Incoming Calls to SM** section, click on "**Add**" to configure an entry for removing the leading "1" from all 11-digit numbers in the Request-URI and/or Message-Account header of ingress SIP messages from Avaya Modular Messaging. Provision the entry as follows:
   - **Matching Pattern** – Enter "**1**".
   - **Min** and **Max** – Enter "**11**".
   - **Delete Digits** – Enter "**1**".
   - **Address to modify** – Select "**destination**".
7. Click on "**Commit**".

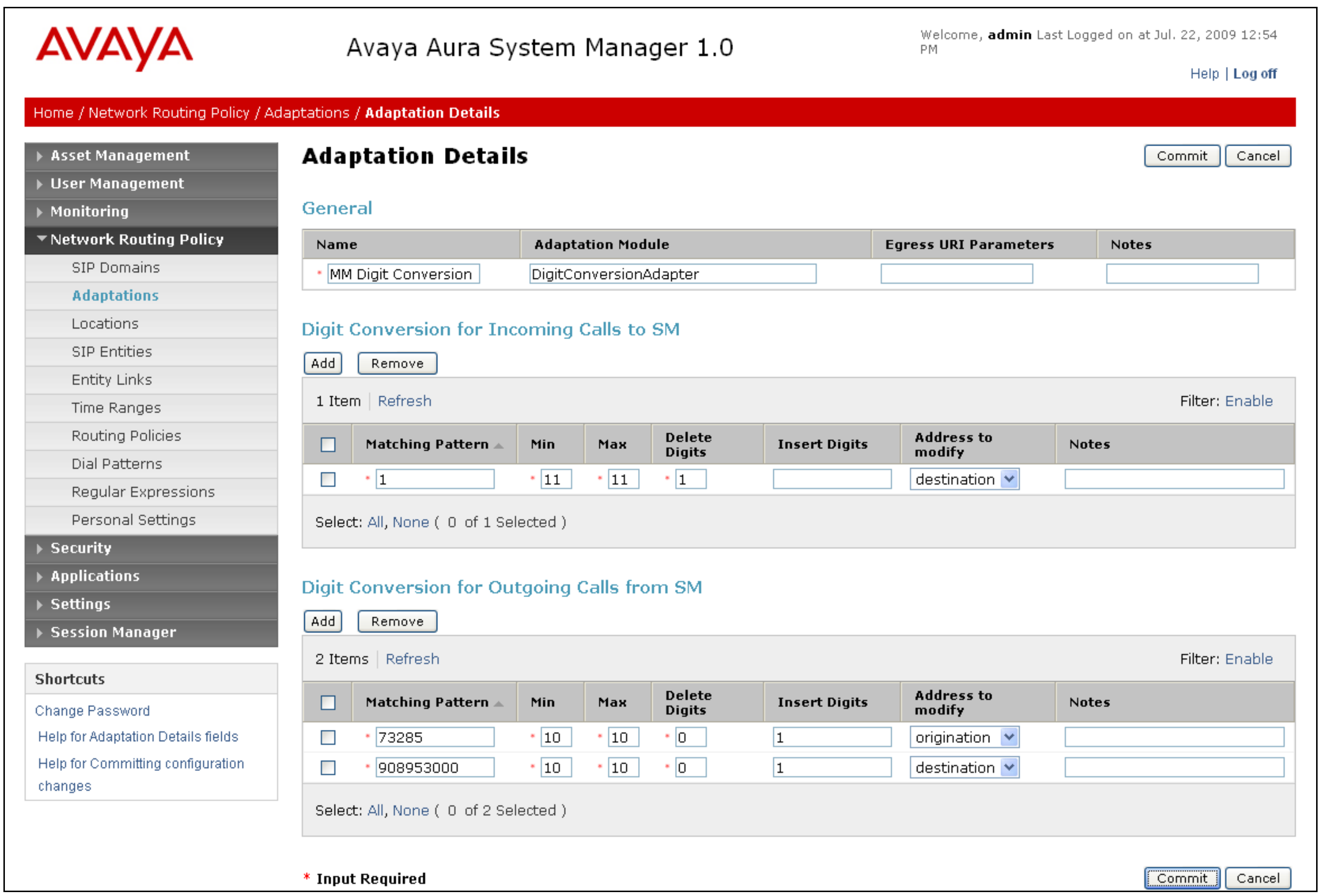

Figure 10: Adaptation Details Page – Adaptation for Avaya Modular Messaging

4.6. SIP Entities

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

- Avaya Aura™ Session Manager
- Avaya Aura™ Communication Manager
- Acme Packet SBC
- Avaya Modular Messaging

4.6.1. Avaya Aura™ Session Manager SIP Entity

1. In the left pane under **Network Routing Policy**, click on "**SIP Entities**". In the **SIP Entities** page (not shown), click on "**New**".
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name for Avaya Aura™ Session Manager.
   - **FQDN or IP Address –** Enter the IP address of the SM100 card on Avaya Aura™ Session Manager.
   - **Type –** Select "**Session Manager**".
   - **Location** – Select a Location administered in Section 4.4. In the sample configuration, Avaya Aura™ Session Manager is assigned to the "Main" Location.
   - **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Avaya Aura™ Session Manager is not authoritative, Avaya Aura™ 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 Avaya Aura™ Session Manager resides.
3. In the **Port** section of the **SIP Entity Details** page, click on "**Add**" and provision an entry as follows:
   - **Port –** Enter "**5061**".
   - **Protocol –** Select "**TLS**".
   - **Default Domain –** (Optional) Select a SIP domain administered in Section 4.3.

   This entry enables Avaya Aura™ Session Manager to accept SIP requests on TLS port 5061. In addition, Avaya Aura™ Session Manager will associate SIP requests received on this port that contain the IP address of the SM100 card on Avaya Aura™ Session Manager in the host part of the Request-URI with the selected SIP **Default Domain**.
4. Repeat Step 3 to provision another similar entry, except with "**5060**" for **Port** and "**TCP**" for **Protocol**.
5. Repeat Step 3 as necessary to provision entries for other ports on which Avaya Aura™ Session Manager is allowed to accept SIP requests.
6. Click on "**Commit**".

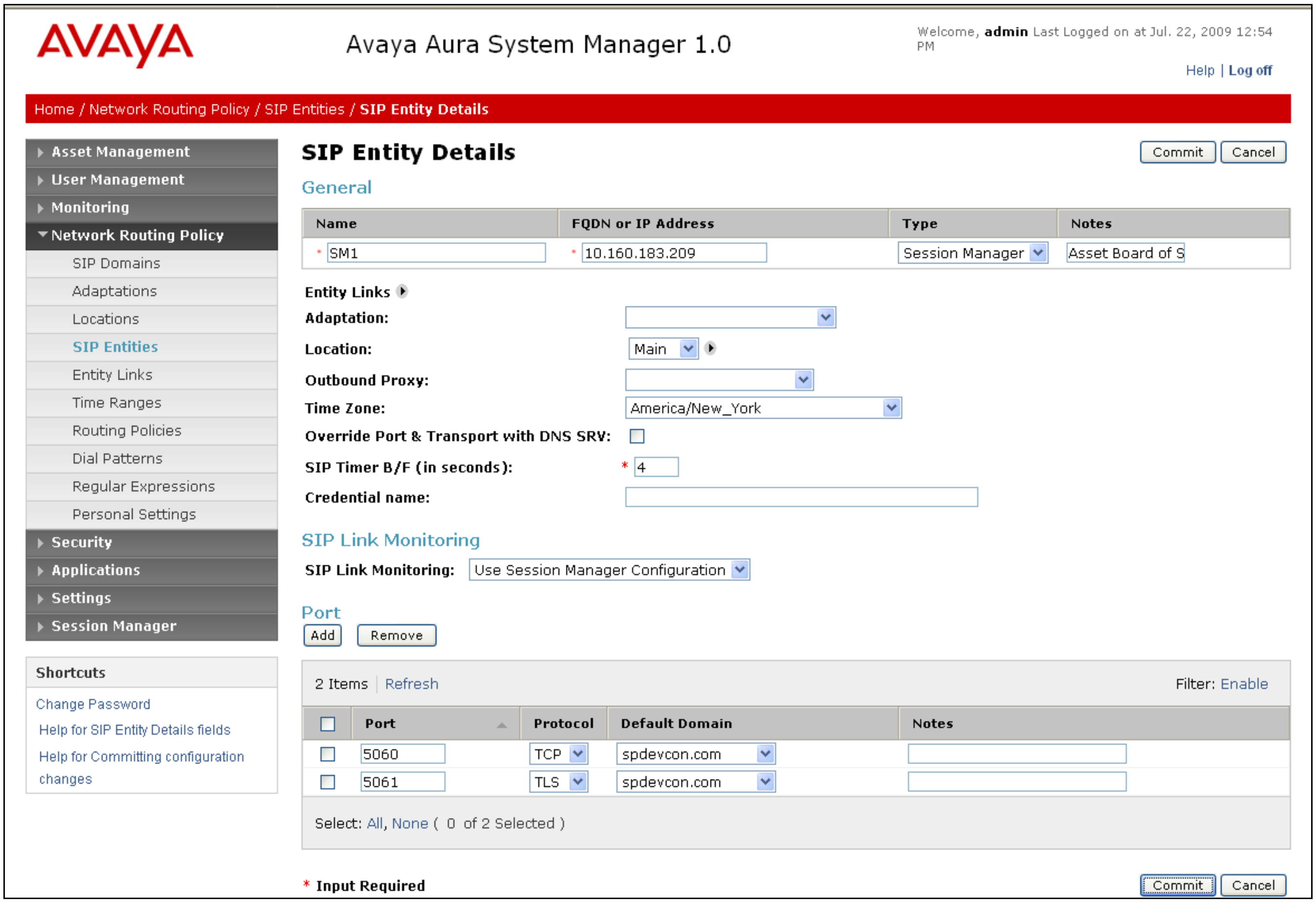

Figure 11: SIP Entity Details Page – Avaya Aura™ Session Manager SIP Entity

4.6.2. Avaya Aura™ Communication Manager SIP Entity

1. In the **SIP Entities** page, click on "**New**".
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name for Avaya Aura™ Communication Manager.
   - **FQDN or IP Address –** Enter the IP address of the Avaya Aura™ Communication Manager C-LAN board noted in Section 5.3 Step 4.
   - **Type –** Select "**CM**".
   - **Adaptation –** Select the Adaptation administered in Section 4.5.2.
   - **Location** – Select a Location administered in Section 4.4. In the sample configuration, Avaya Aura™ Communication Manager is assigned to the "Site 1" Location.
   - **Time Zone** – Select the time zone in which Avaya Aura™ Communication Manager resides.
3. Click on "**Commit**".

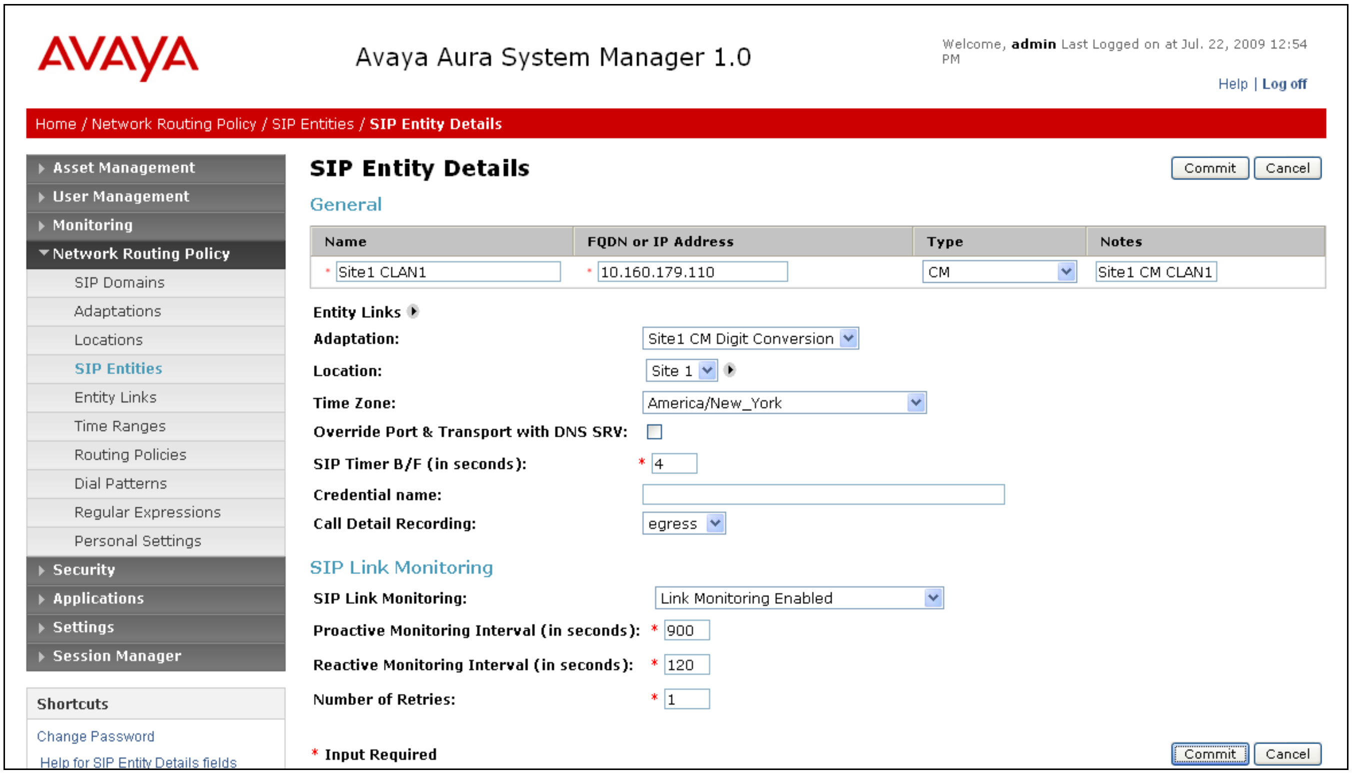

Figure 12: SIP Entity Details Page – Avaya Aura™ Communication Manager SIP Entity

4.6.3. Acme Packet SBC SIP Entity

1. In the **SIP Entities** page (not shown), click on "**New**".
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name for the Acme Packet SBC.
   - **FQDN or IP Address –** Enter the IP address of the "Inside" (Private) Interface of the Acme Packet SBC.
   - **Type** – Select "**SBC**".
   - **Adaptation –** Select the Adaptation administered in Section 4.5.1.
   - **Location** – Select a Location administered in Section 4.4. In the sample configuration, the Acme Packet SBC is assigned to the "Main" Location.
   - **Time Zone** – Select the time zone in which the Acme Packet SBC resides.
3. Click on "**Commit**".

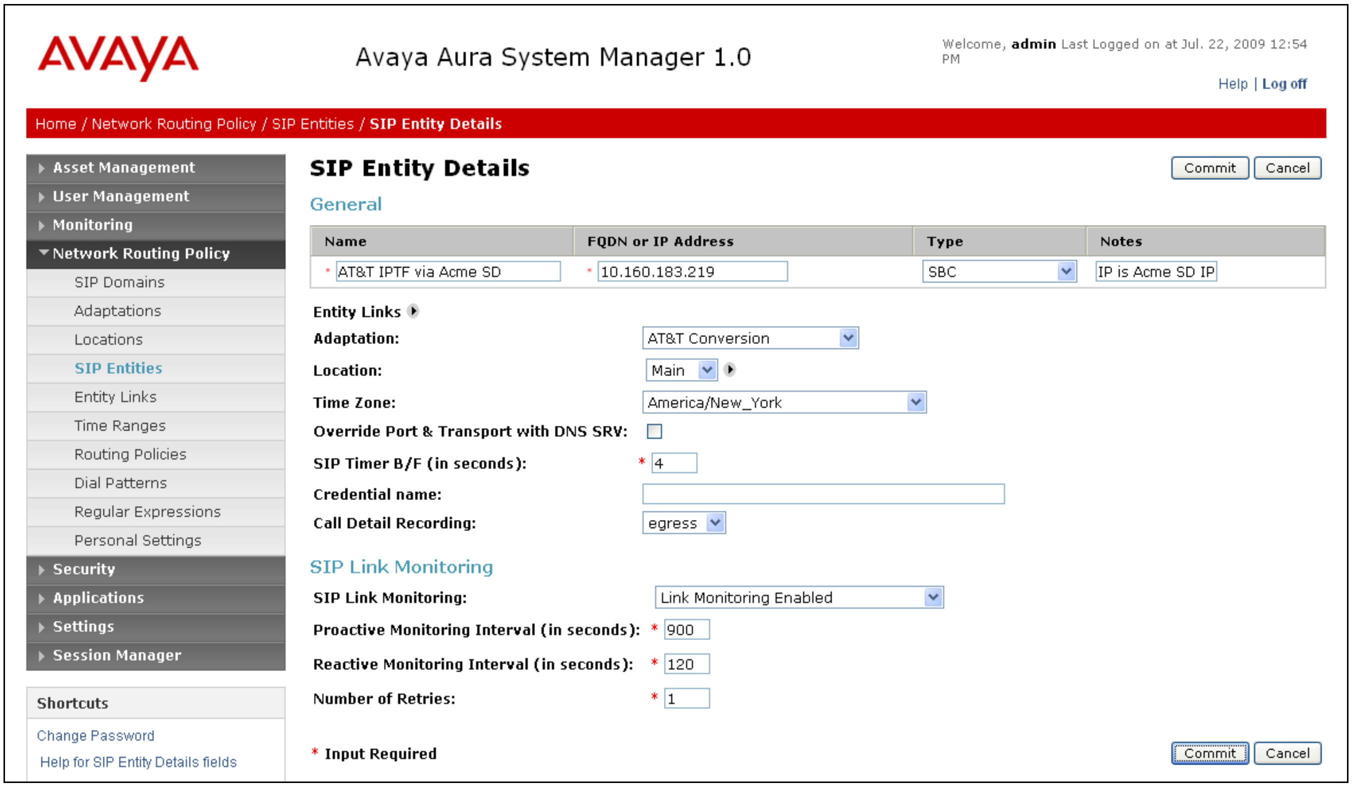

Figure 13: SIP Entity Details Page – Acme Packet SBC SIP Entity

4.6.4. Avaya Modular Messaging SIP Entity

1. In the **SIP Entities** page, click on "**New**".
2. In the **General** section of the **SIP Entity Details** page, provision the following:
   - **Name –** Enter a descriptive name for Avaya Modular Messaging.
   - **FQDN or IP Address –** Enter the IP address of the Avaya Modular Messaging Messaging Application Server (MAS).
   - **Type –** Select "**Other**".
   - **Adaptation –** Select the Adaptation administered in Section 4.5.3.
   - **Location** – Select a Location administered in Section 4.4. In the sample configuration, Avaya Modular Messaging is assigned to the "Main" Location.
   - **Time Zone** – Select the time zone in which Avaya Modular Messaging resides.
3. Click on "**Commit**".

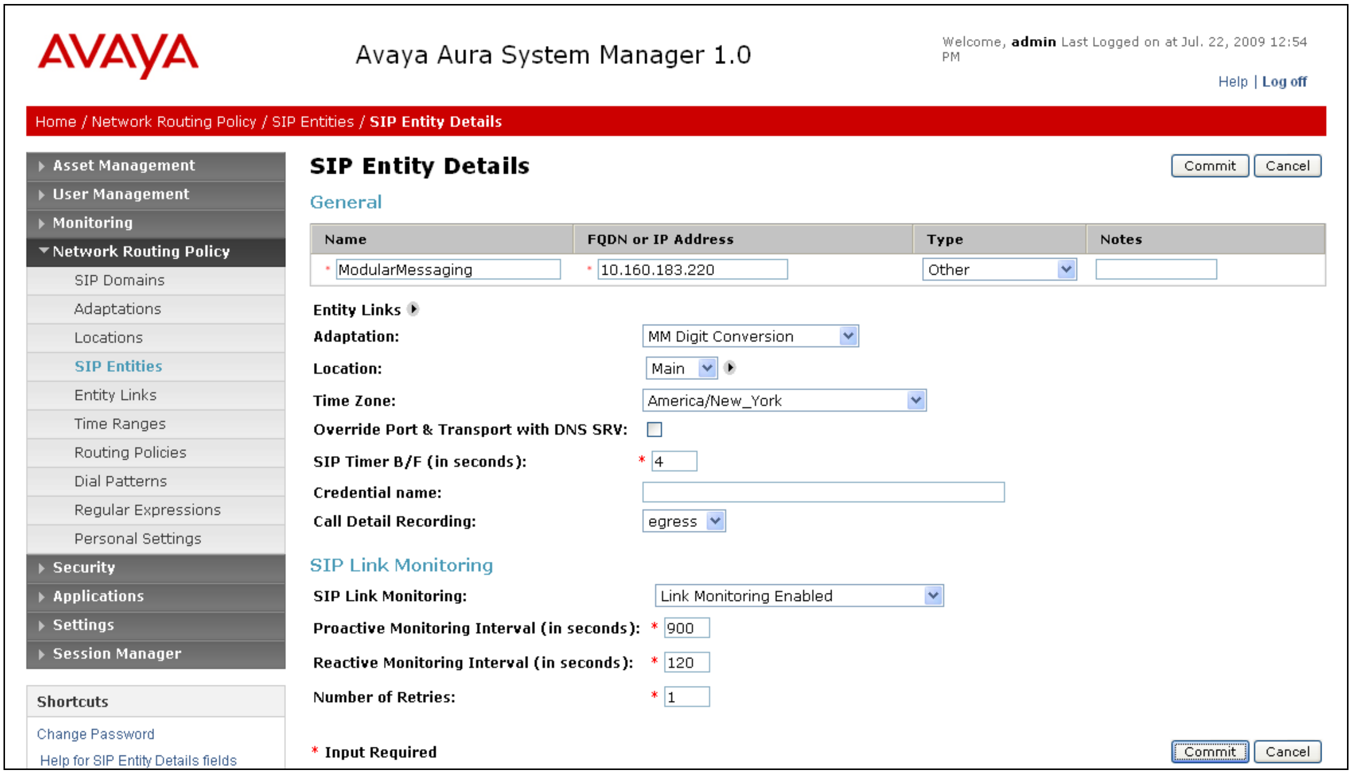

Figure 14: SIP Entity Details Page – Avaya Modular Messaging SIP Entity

4.7. Entity Links

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

- Avaya Aura™ Communication Manager
- Acme Packet SBC
- Avaya Modular Messaging

4.7.1. Entity Link to Avaya Aura™ Communication Manager

1. In the left pane under **Network Routing Policy**, click on "**Entity Links**". In the **Entity Links** page (not shown), click on "**New**".
2. Continuing in the **Entity Links** page, provision the following:
   - **Name** – Enter a descriptive name for the link to Avaya Aura™ Communication Manager.
   - **SIP Entity 1** – Select the SIP Entity administered in Section 4.6.1 for Avaya Aura™ Session Manager. SIP Entity 1 must always be an Avaya Aura™ Session Manager instance.
   - **SIP Entity 1 Port** – Enter "**5061**".
   - **SIP Entity 2** –Select the SIP Entity administered in Section 4.6.2 for Avaya Aura™ Communication Manager.
   - **SIP Entity 2 Port** - Enter "**5061**".

- **Trusted** – Check the checkbox.
- **Protocol** – Select "**TLS**".

3. Click on "**Commit**".

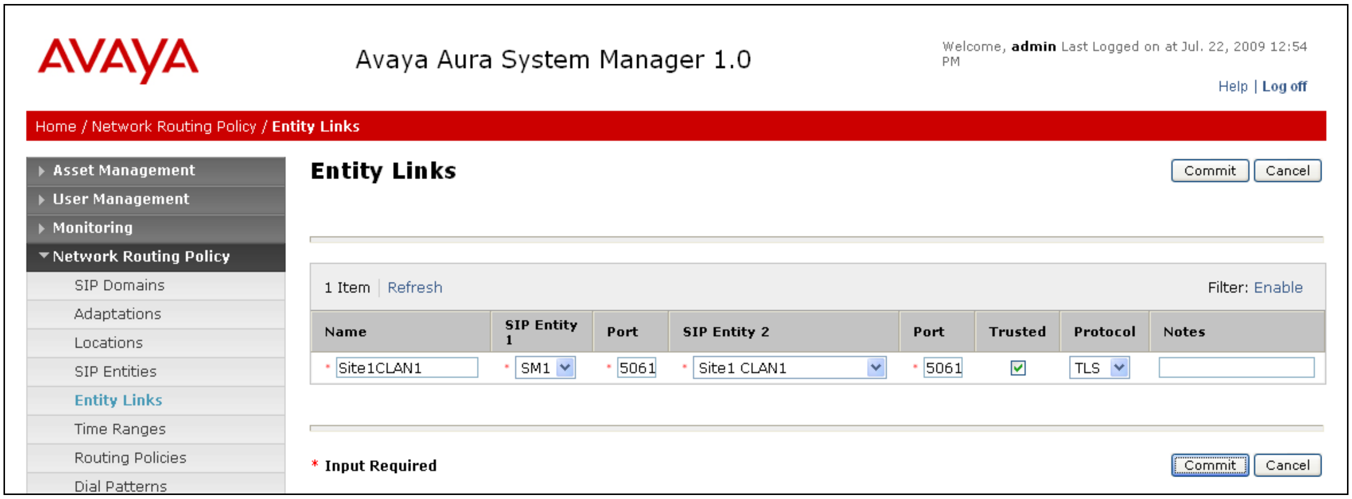

Figure 15: Entity Links Page – Entity Link to Avaya Aura™ Communication Manager

4.7.2. Entity Link to AT&T IP Toll Free Service via Acme Packet SBC

Repeat Section 4.7.1 with the following differences:

- **Name** – Enter a descriptive name for the link to the AT&T IP Toll Free service, by way of the Acme Packet SBC.
- **SIP Entity 1 Port** – Enter "**5060**".
- **SIP Entity 2** – Select the SIP Entity administered in Section 4.6.3 for the Acme Packet SBC.
- **SIP Entity 2 Port** - Enter "**5060**".
- **Protocol** – Select "**TCP**".

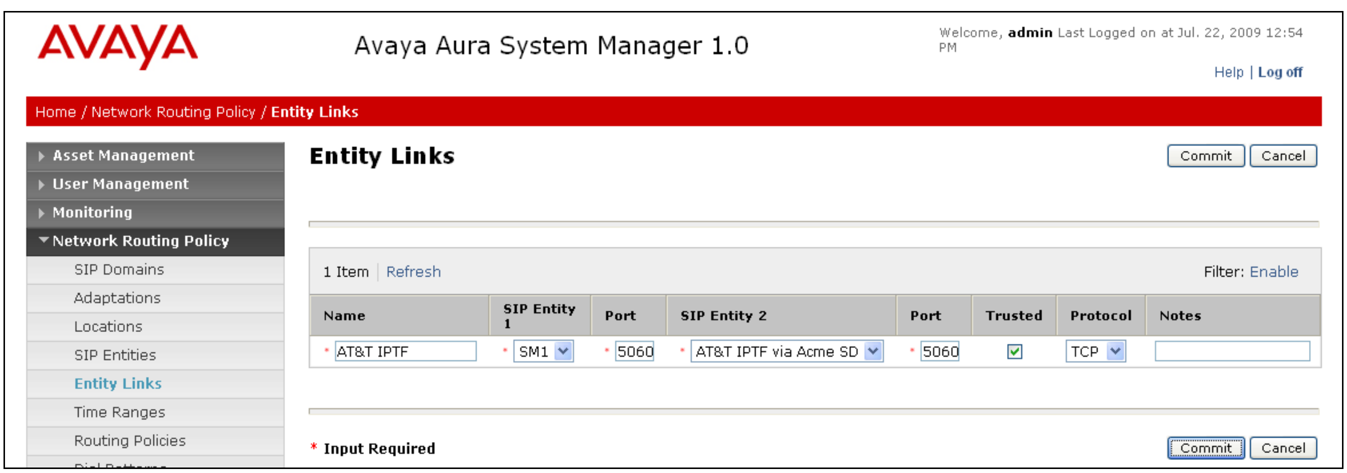

Figure 16: Entity Links Page – Entity Link to AT&T IP Toll Free Service via Acme Packet SBC

4.7.3. Entity Link to Avaya Modular Messaging

Repeat Section 4.7.1 with the following differences:

- **Name** – Enter a descriptive name for the link to Avaya Modular Messaging.
- **SIP Entity 1 Port** – Enter "**5060**".
- **SIP Entity 2** – Select the SIP Entity administered in Section 4.6.4 for Avaya Modular Messaging.
- **SIP Entity 2 Port** - Enter "**5060**".
- **Protocol** – Select "**TCP**".

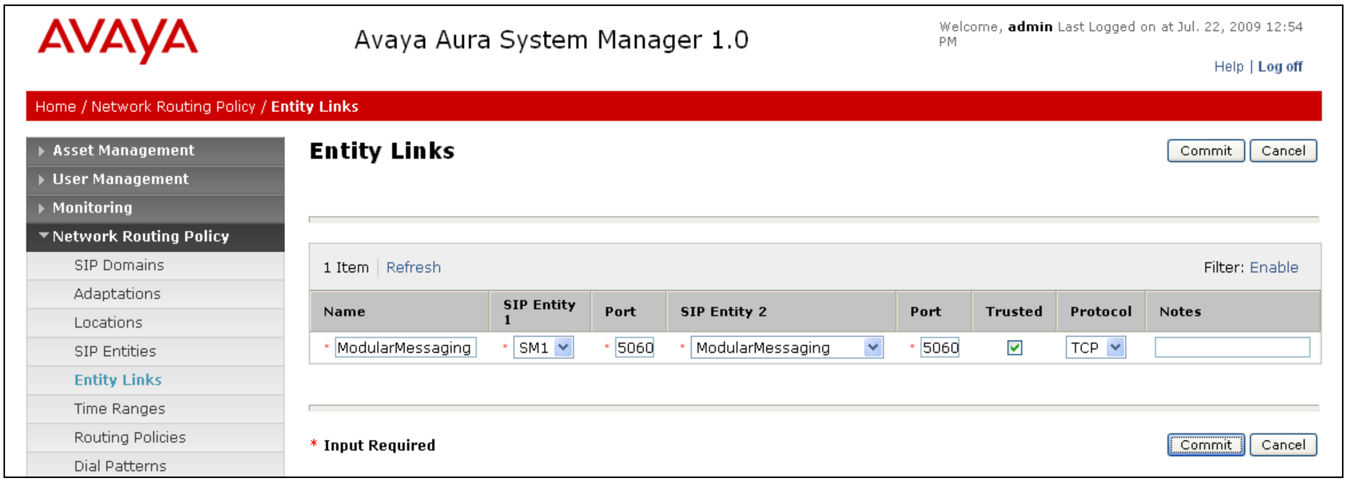

Figure 17: Entity Links Page – Entity Link to Avaya Modular Messaging

4.8. Time Ranges

1. In the left pane under **Network Routing Policy**, click on "**Time Ranges**". In the **Time Ranges** page (not shown), click on "**New**".
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**.
3. Click on "**Commit**".
4. Repeat Steps 1 – 3 to provision additional time ranges.

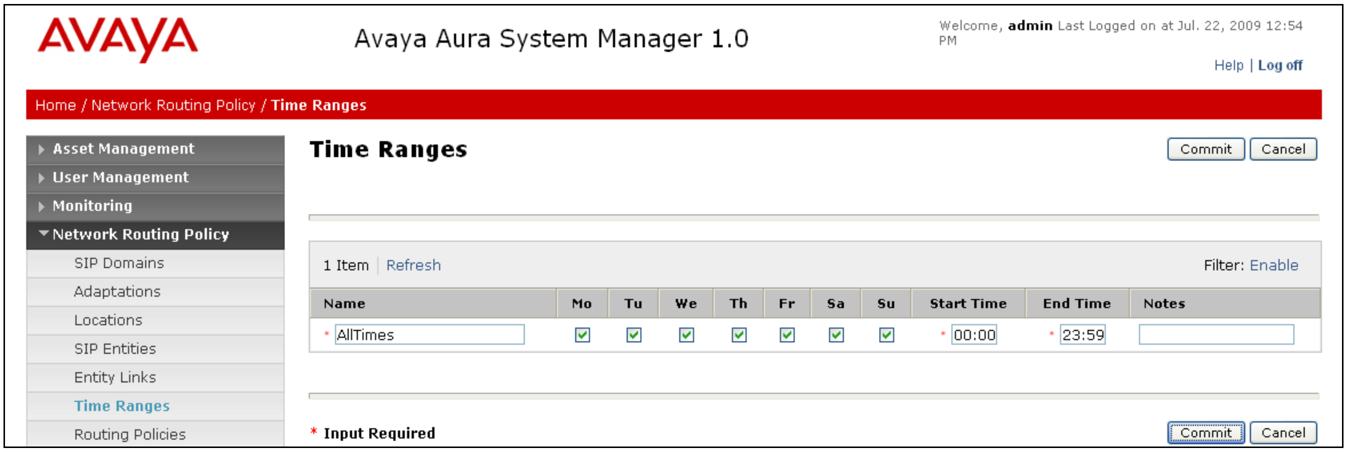

Figure 18: Time Ranges Page

4.9. Routing Policies

In this section, Routing Policies are administered for routing calls to the following SIP Entities:

- Avaya Aura™ Communication Manager
- Avaya Modular Messaging

4.9.1. Routing Policy for Routing to Avaya Aura™ Communication Manager

1. In the left pane under **Network Routing Policy**, click on "**Routing Policies**". In the **Routing Policies** page (not shown), click on "**New**".
2. In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing calls to Avaya Aura™ Communication Manager, and ensure that the **Disabled** checkbox is unchecked to activate this Network Routing Policy.
3. In the **SIP Entity as Destination** section of the **Routing Policy Details** page, click on "**Select**".

Figure 19: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager

4. In the **SIP Entity List** page, select the SIP Entity administered in Section 4.6.2 for Avaya Aura™ Communication Manager, and click on "**Select**".

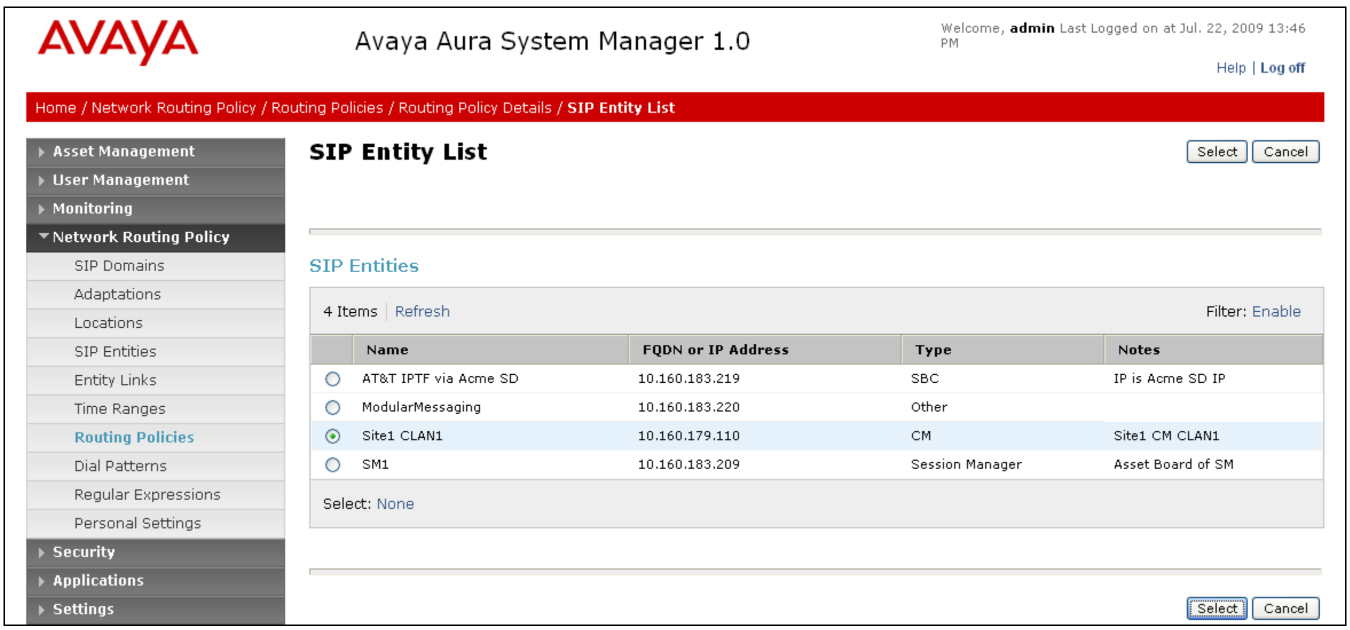

Figure 20: SIP Entity List Page - Routing to Avaya Aura™ Communication Manager

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

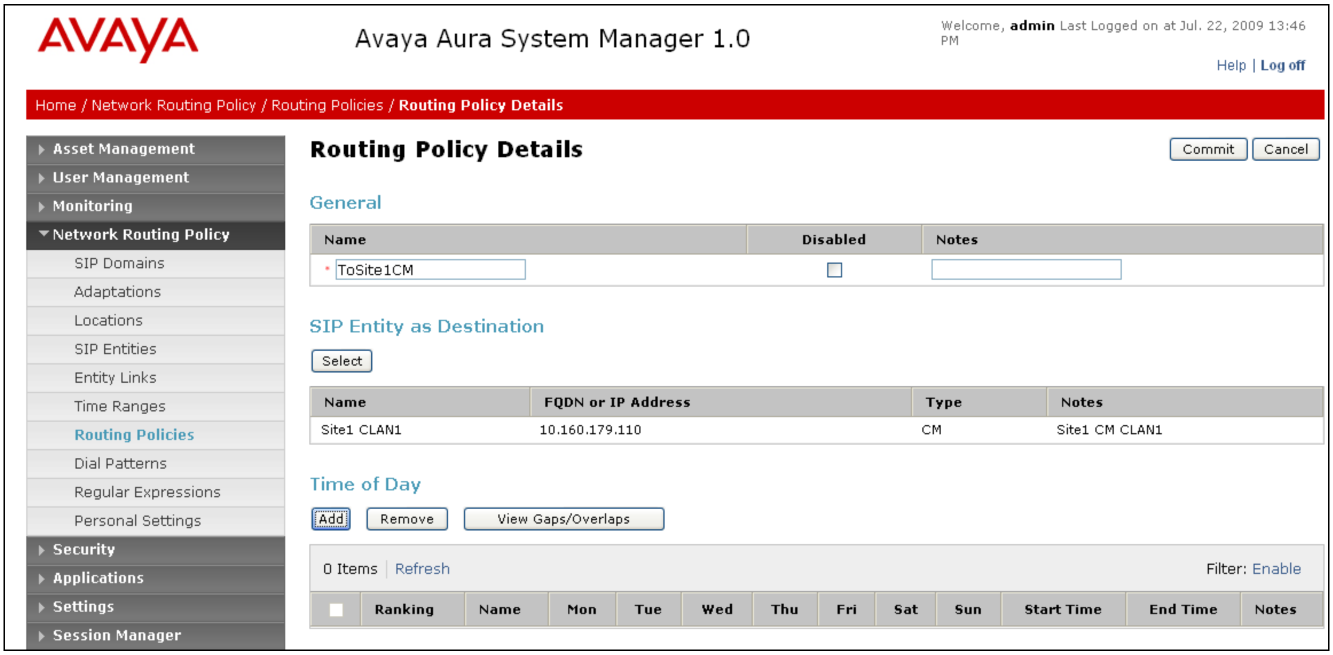

Figure 21: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager (Continued)

6. In the **Time Range List** page, check the checkbox(es) corresponding to one or more Time Ranges administered in Section 4.8, and click on "**Select**".

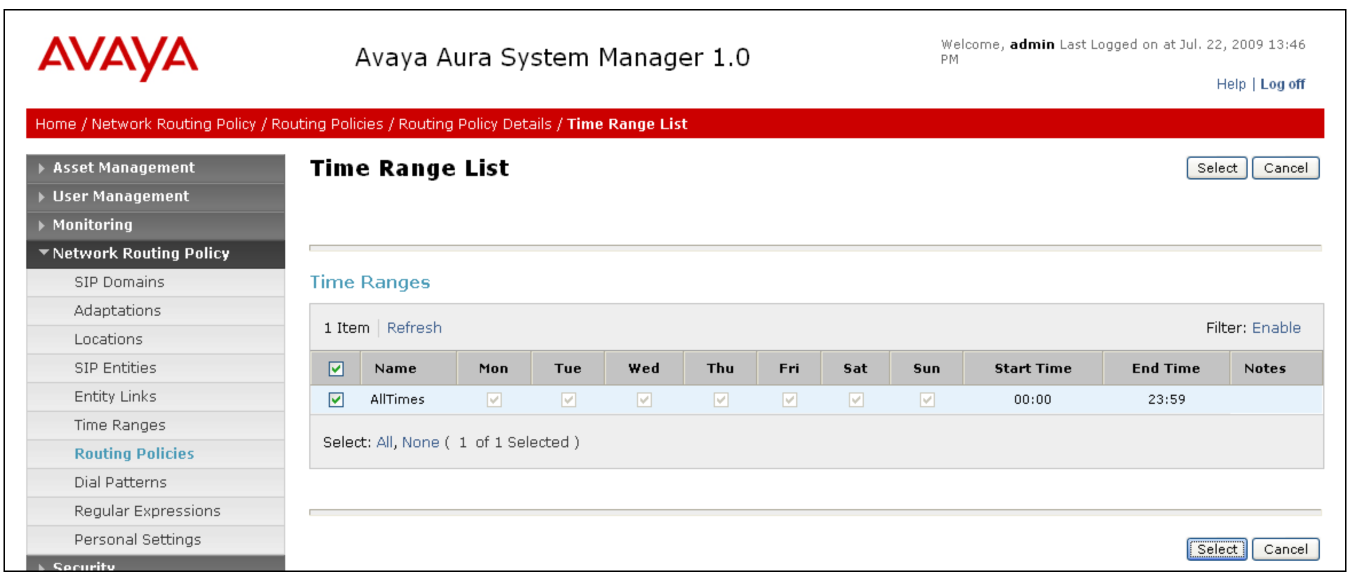

Figure 22: Time Range List Page - Routing to Avaya Aura™ Communication Manager

7. Returning to the **Routing Policy Details** page, in the **Time of Day** section, enter a **Ranking** (the lower the number, the higher the ranking) for each Time Range, and click on "**Commit**".

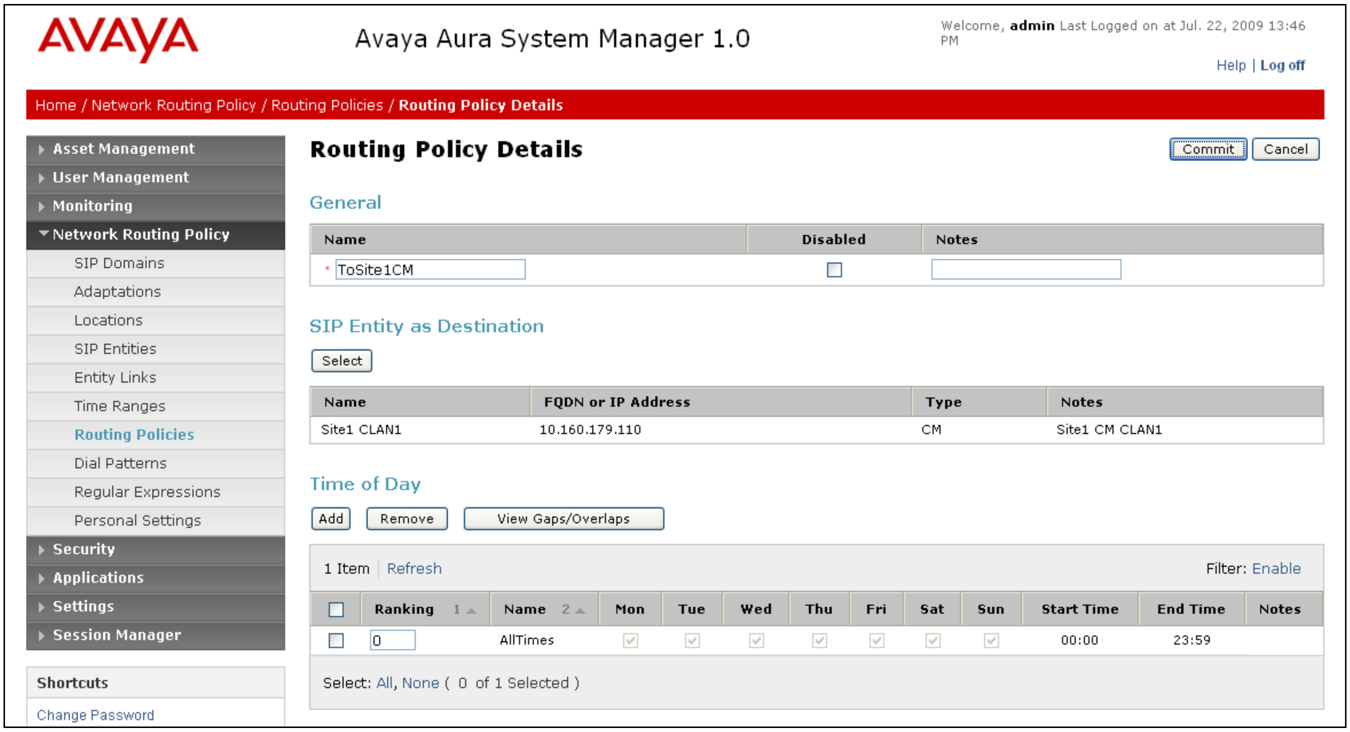

Figure 23: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager (Final)

4.9.2. Routing Policy for Routing to Avaya Modular Messaging

Repeat Section 4.9.1 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, and ensure that the **Disabled** checkbox is unchecked to activate this Network Routing Policy.
- In the **SIP Entity List** page, select the SIP Entity administered in Section 4.6.4 for Avaya Modular Messaging, and click on "**Select**".

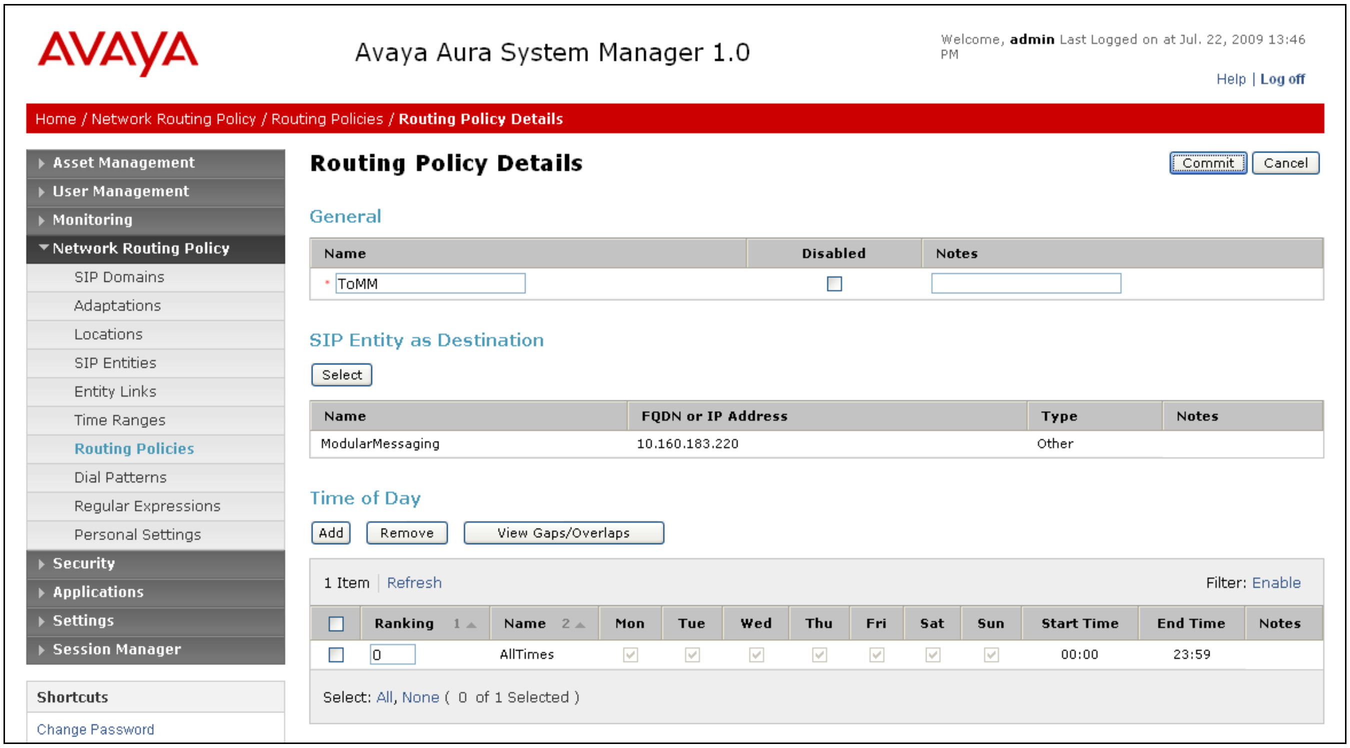

Figure 24: Routing Policy Details Page - Routing to Avaya Modular Messaging

4.10. Dial Patterns

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

- Inbound AT&T IP Toll Free service calls
- Calls to 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager
- Calls to the Avaya Modular Messaging uniform pilot number

4.10.1. Matching Inbound AT&T IP Toll Free Service Calls

1. In the left pane under **Network Routing Policy**, click on "**Dial Patterns**". In the **Dial Patterns** page (not shown), click on "**New**".
2. In the **General** section of the **Dial Pattern Details** page, provision the following:
   - **Pattern** – Enter enough leading digits to uniquely match a range of AT&T IP Toll Free service numbers, specifically the numbers contained in the Request-URI of inbound SIP INVITE messages from the AT&T IP Toll Free service.
   - **Min** and **Max –** Enter the total number of digits in the number range.
   - **SIP Domain** – Select one of the SIP Domains administered in Section 4.3 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.
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on "**Add**".

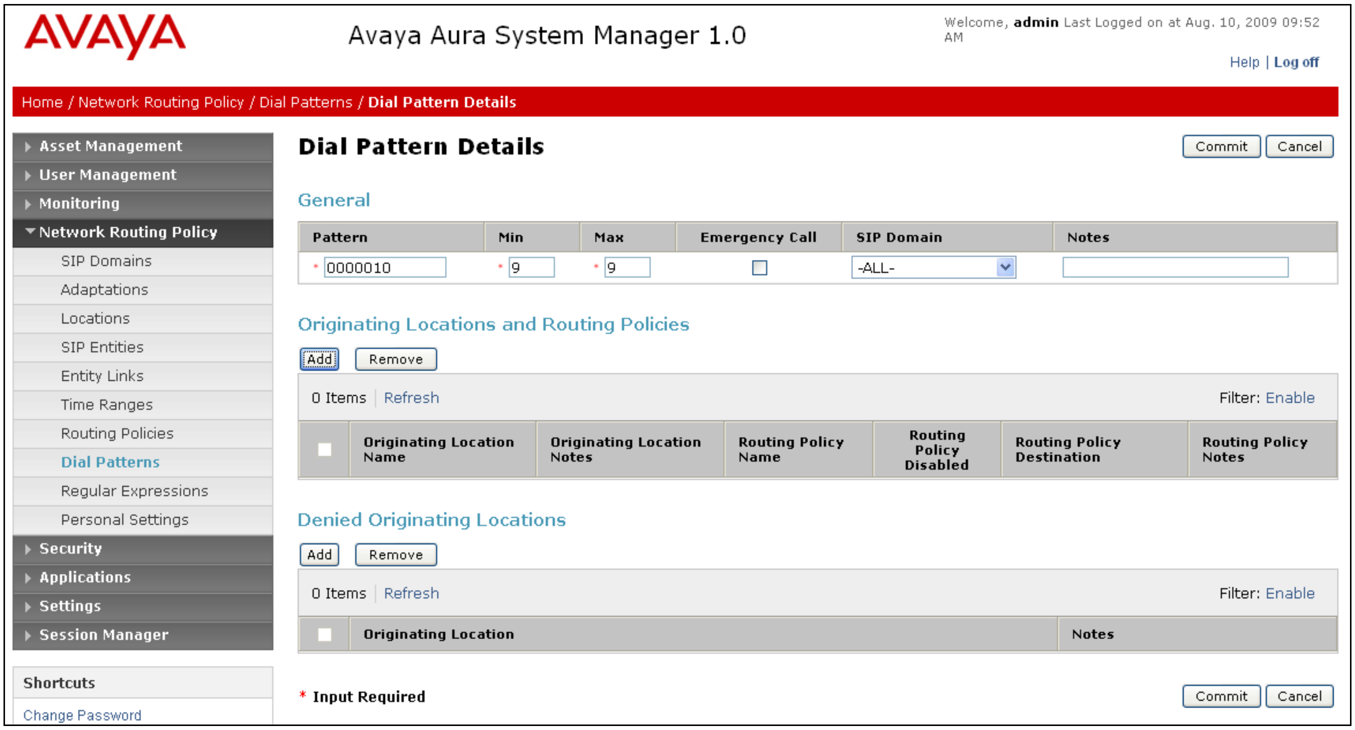

Figure 25: Dial Pattern Details Page - Matching Inbound AT&T IP Toll Free Service Calls

4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Location to which the Acme Packet SBC is assigned (see Section 4.6.3 Step 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.
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 Avaya Aura™ Communication Manager in Section 4.9.1.
6. In the **Originating Location and Routing Policy List** page, click on "**Select**".

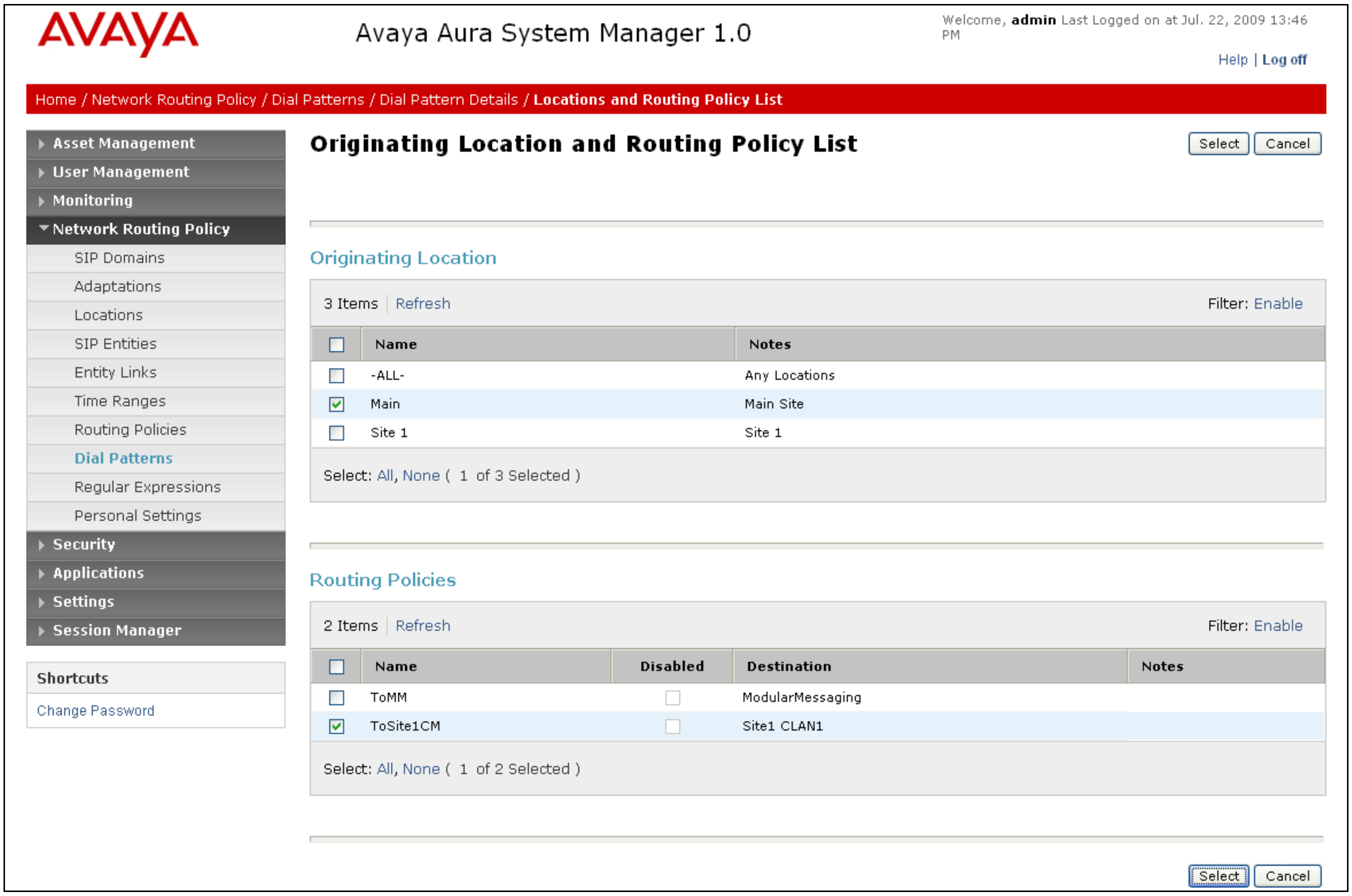

Figure 26: Originating Location and Routing Policy List Page - Matching Inbound AT&T IP Toll Free Service Calls

7. Returning to the **Dial Pattern Details** page, click on "**Commit**".

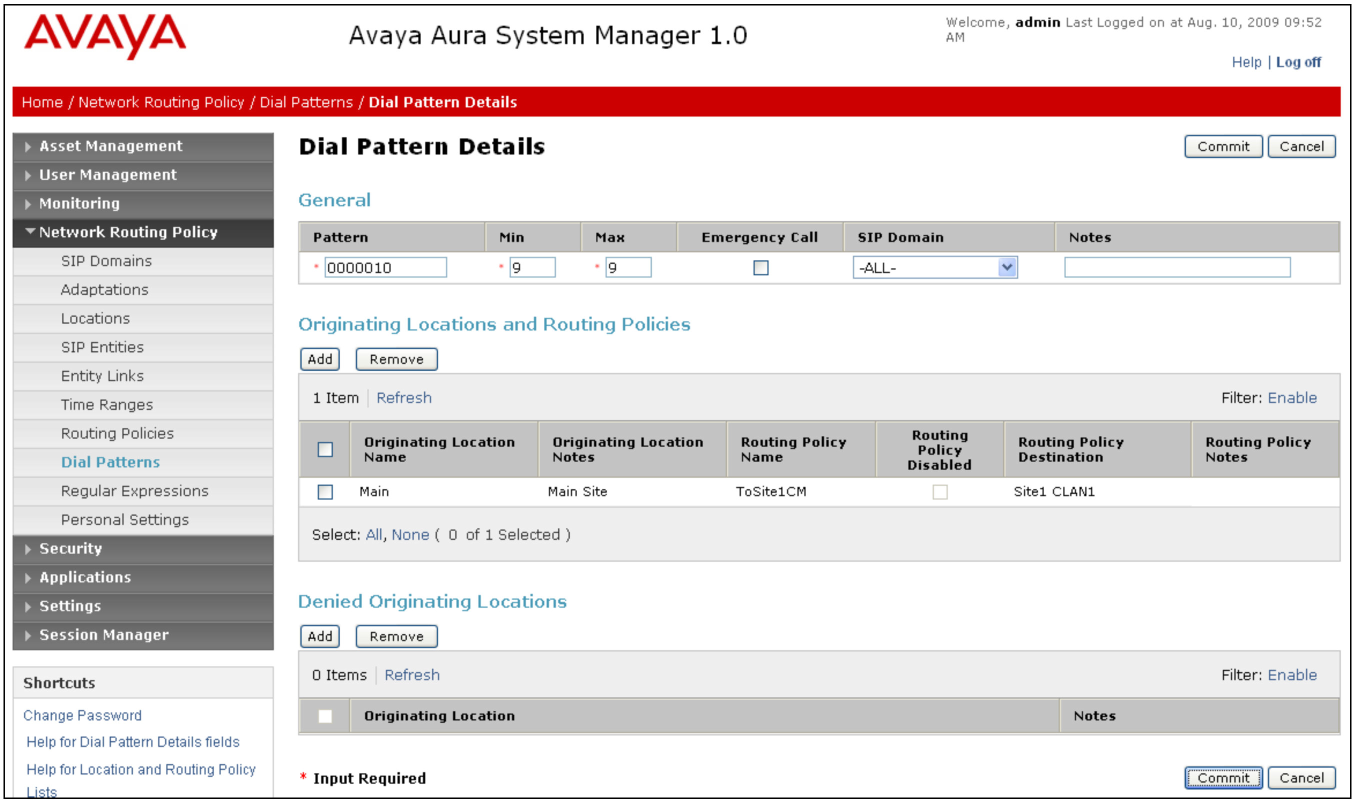

Figure 27: Dial Pattern Details - Matching Inbound AT&T IP Toll Free Service Calls (Final)

4.10.2. Matching Calls with 10-digit Called Party Numbers Associated with Extensions on Avaya Aura™ Communication Manager

1. In the **Dial Patterns** page, click on "**New**".
2. In the **General** section of the **Dial Pattern Details** page, provision the following:
   - **Pattern** – Enter enough leading digits to uniquely match a range of 10-digit numbers associated with extensions on Avaya Aura™ Communication Manager.
   - **Min** and **Max** – Enter "**10**".
   - **SIP Domain** – Select "**-ALL-**".
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on "**Add**".

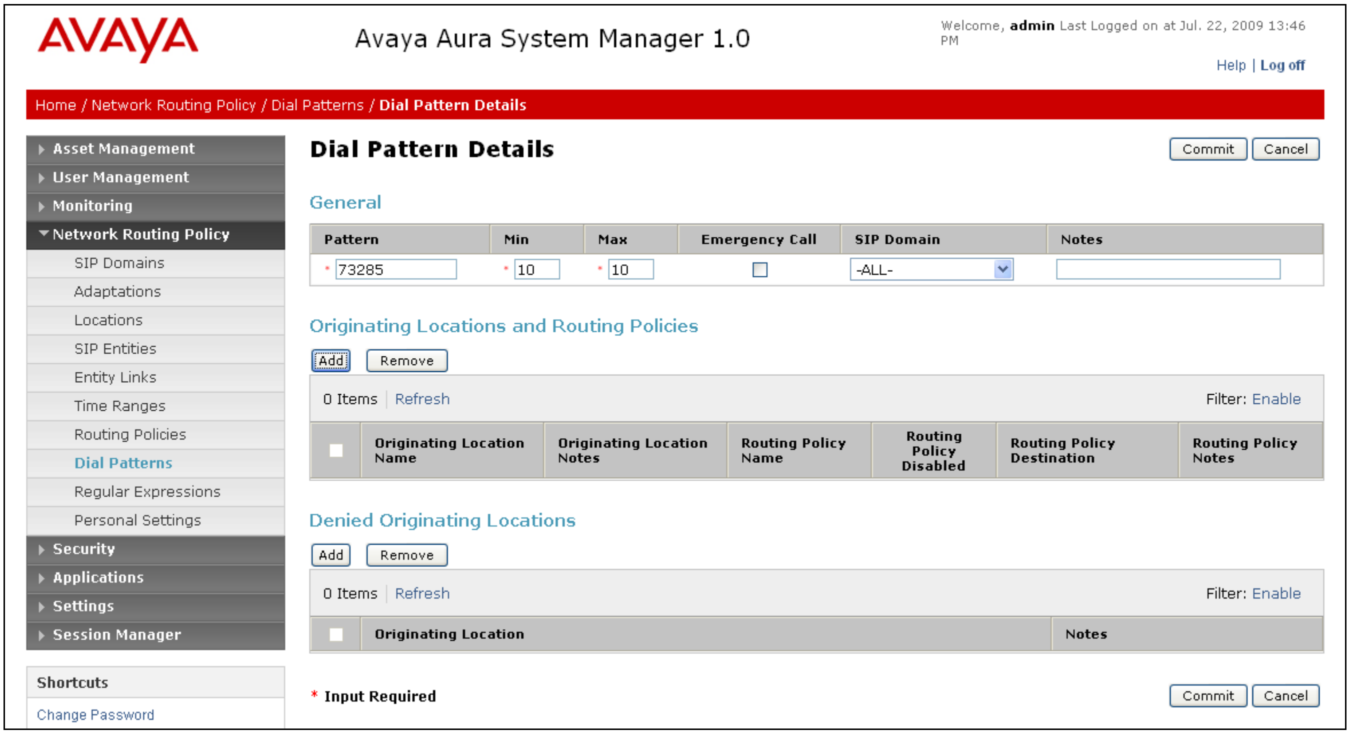

Figure 28: Dial Pattern Details Page - Matching Calls with 10-digit Called Party Numbers Associated with Extensions on Avaya Aura™ Communication Manager

4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to "**-ALL-**".
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 Avaya Aura™ Communication Manager in Section 4.9.1.
6. In the **Originating Location and Routing Policy List** page, click on "**Select**".

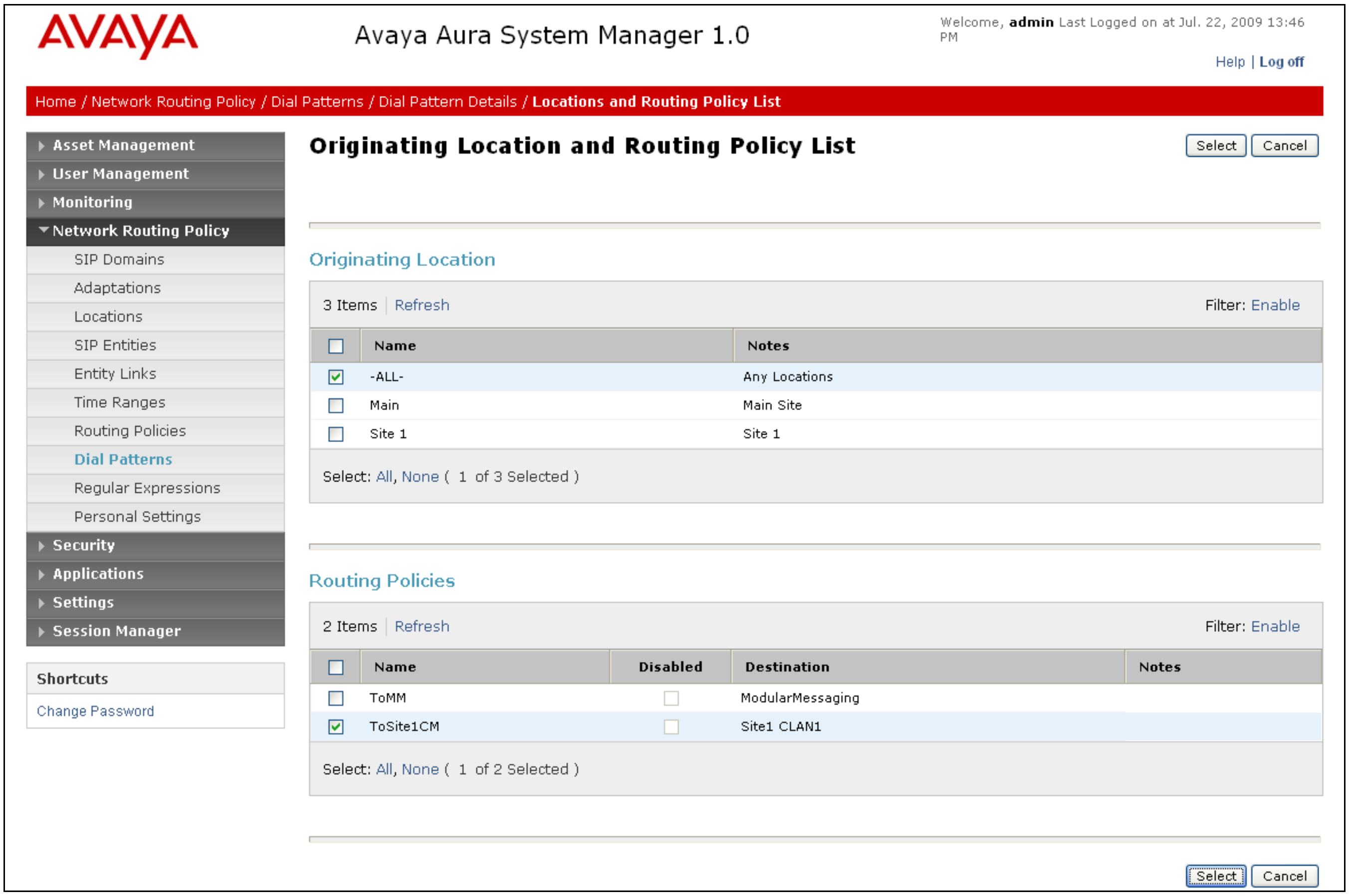

Figure 29: Originating Location and Routing Policy List Page - Matching Calls with 10-digit Called Party Numbers Associated with Extensions on Avaya Aura™ Communication Manager

7. Returning to the **Dial Pattern Details** page, click on "**Commit**".

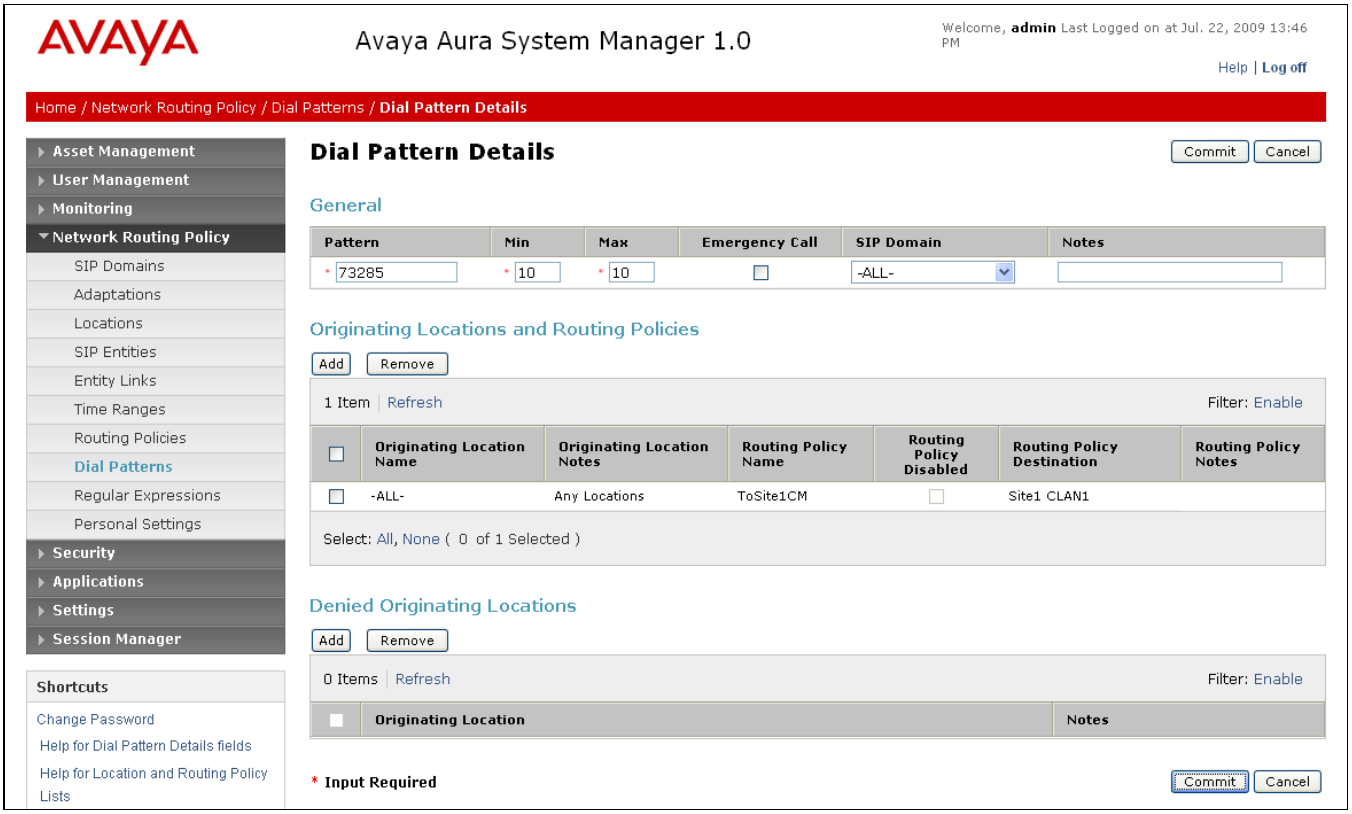

Figure 30: Dial Pattern Details Page - Matching Calls with 10-digit Called Party Numbers Associated with Extensions on Avaya Aura™ Communication Manager (Final)

4.10.3. Matching Calls to Avaya Modular Messaging Pilot Number

1. In the **Dial Patterns** page, click on "**New**".
2. In the **General** section of the **Dial Pattern Details** page, provision the following:
   - **Pattern** – Enter the Avaya Modular Messaging uniform pilot number.
   - **Min** and **Max** – Enter "**10**".
   - **SIP Domain** – Select "**-ALL-**".
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on "**Add**".

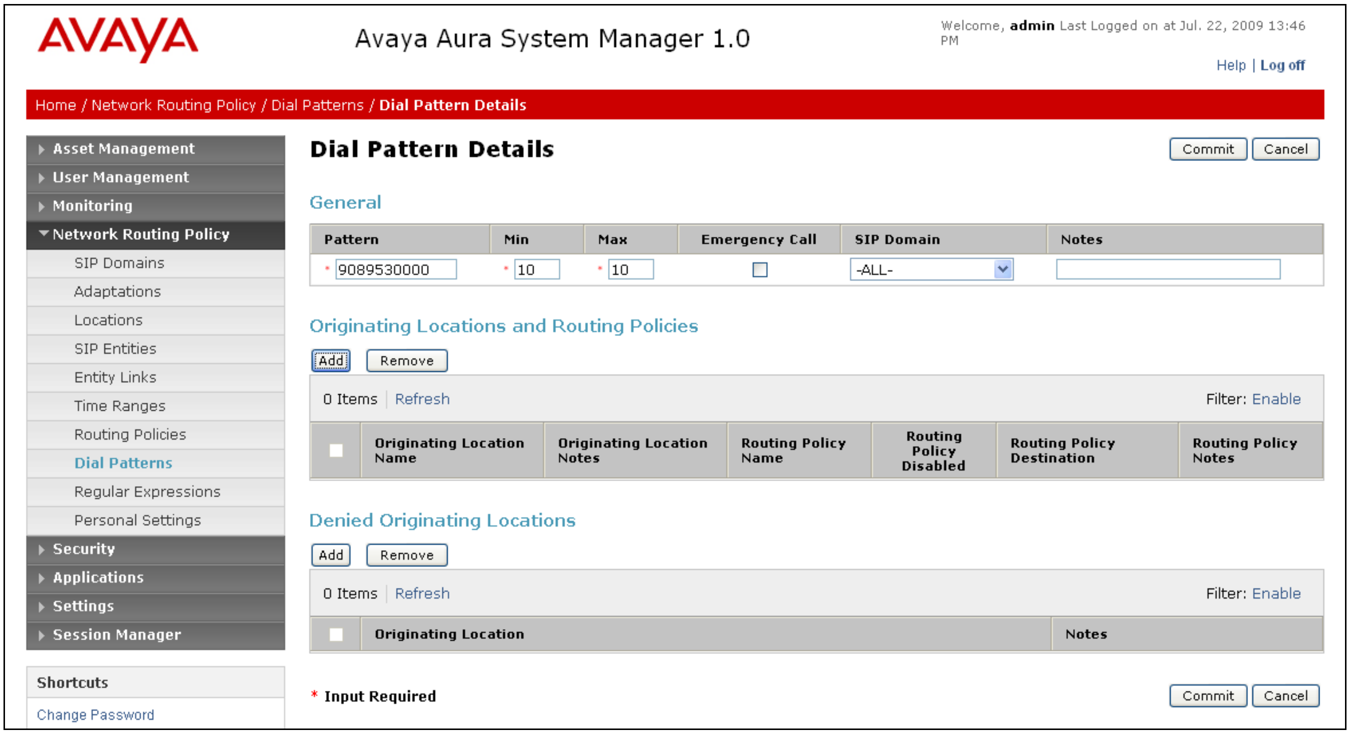

Figure 31: Dial Pattern Details Page - Matching Calls to Avaya Modular Messaging

4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to "**-ALL-**".
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 Avaya Modular Messaging in Section 4.9.2.
6. In the **Originating Location and Routing Policy List** page, click on "**Select**".

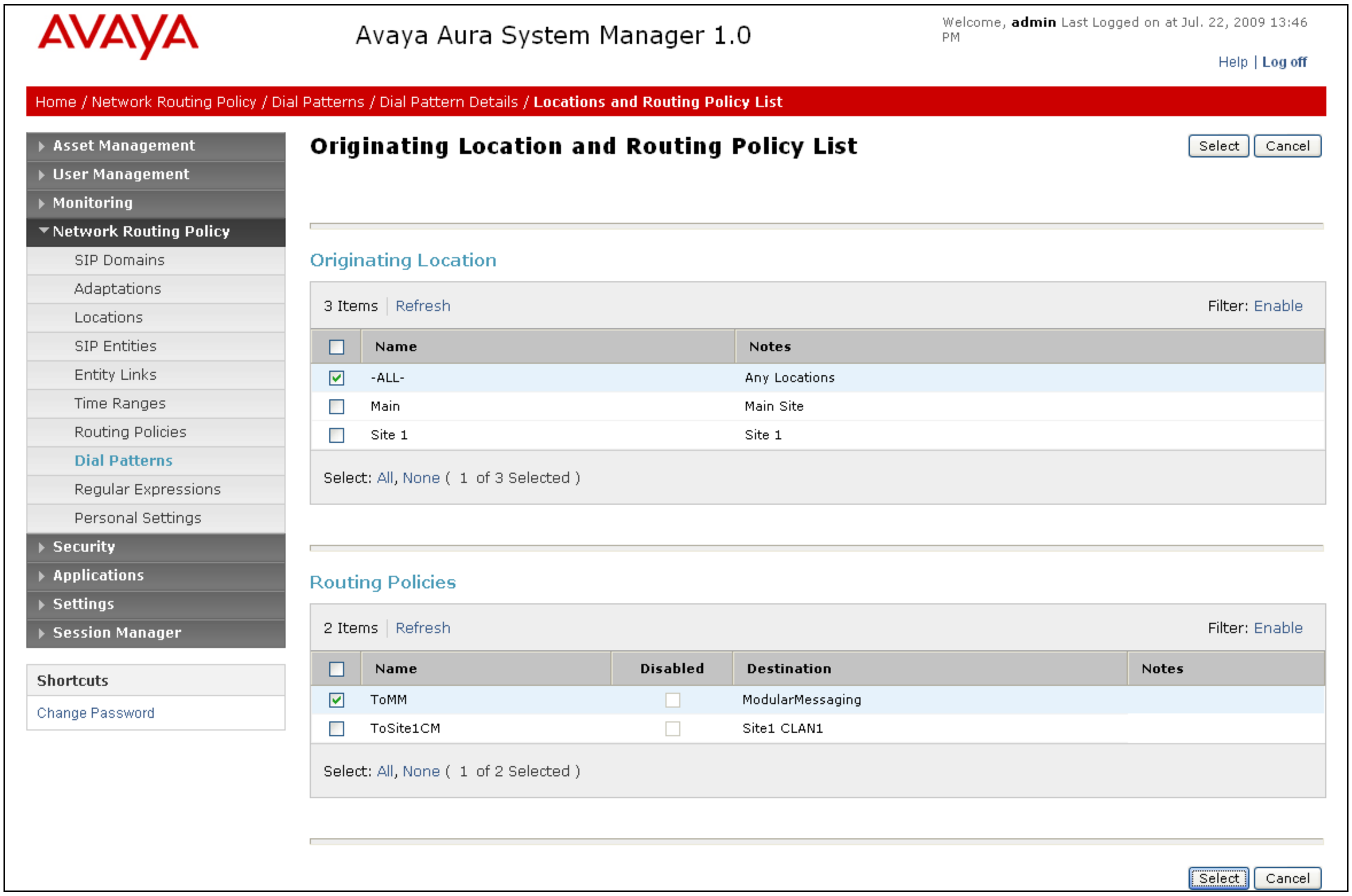

Figure 32: Originating Location and Routing Policy List Page - Matching Calls to Avaya Modular Messaging

7. Returning to the **Dial Pattern Details** page, click on "**Commit**".

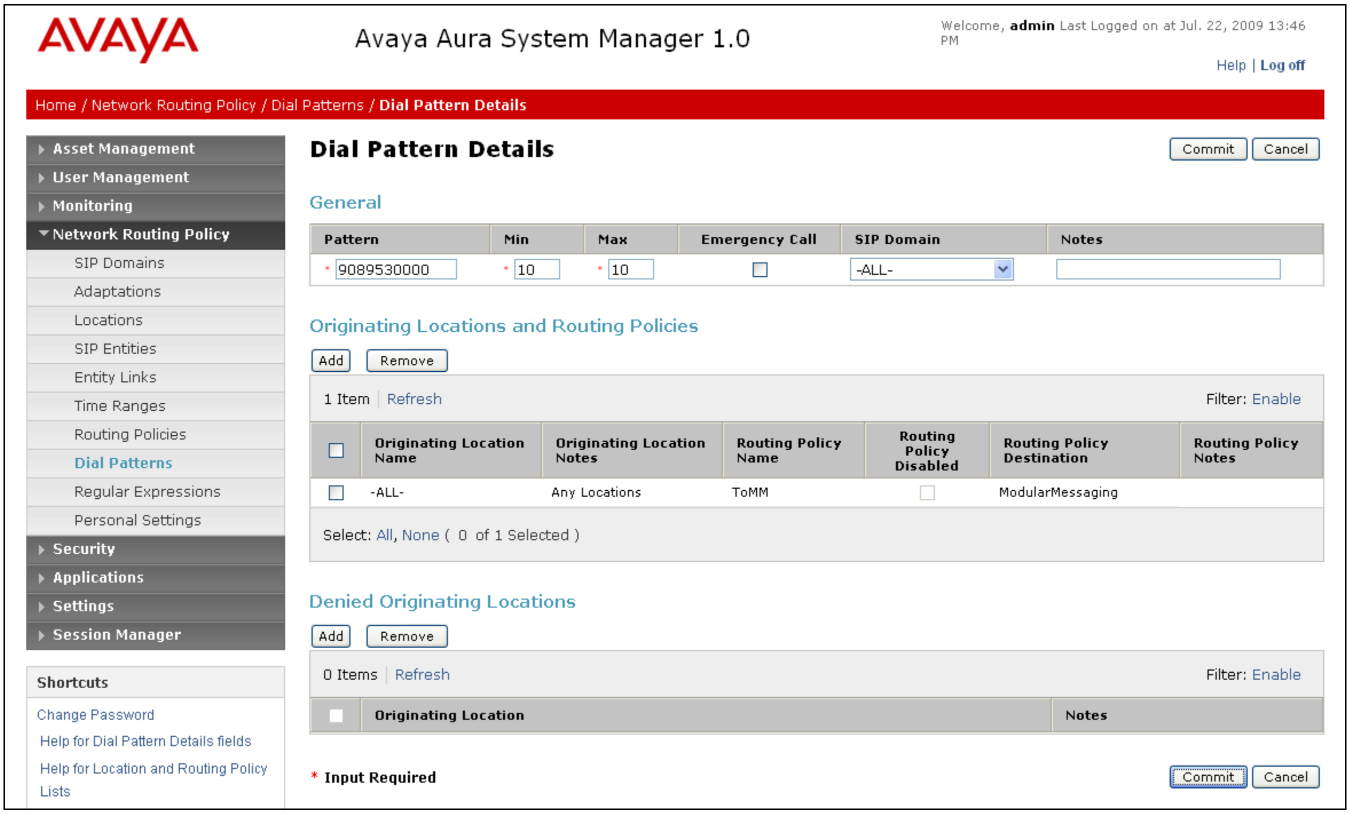

Figure 33: Dial Pattern Details Page - Matching Calls to Avaya Modular Messaging (Final)

4.11. Session Manager Administration

1. In the left pane under **Session Manager**, click on "**Session Manager Administration**". In the **Session Manager Administration** page (not shown), click on "**New**".
2. In the **General** section of the **Add Session Manager** page, provision the following:
   - **SIP Entity Name –** Select the SIP Entity administered for Avaya Aura™ Session Manager in Section 4.6.1.
   - **Management Access Point Host Name/IP –** Enter the IP address of the management interface on Avaya Aura™ Session Manager.
3. In the **Security Module** section of the **Add Session Manager** page, enter the **Network Mask** and **Default Gateway** of the SM100 card.
4. Click on "**Save**".

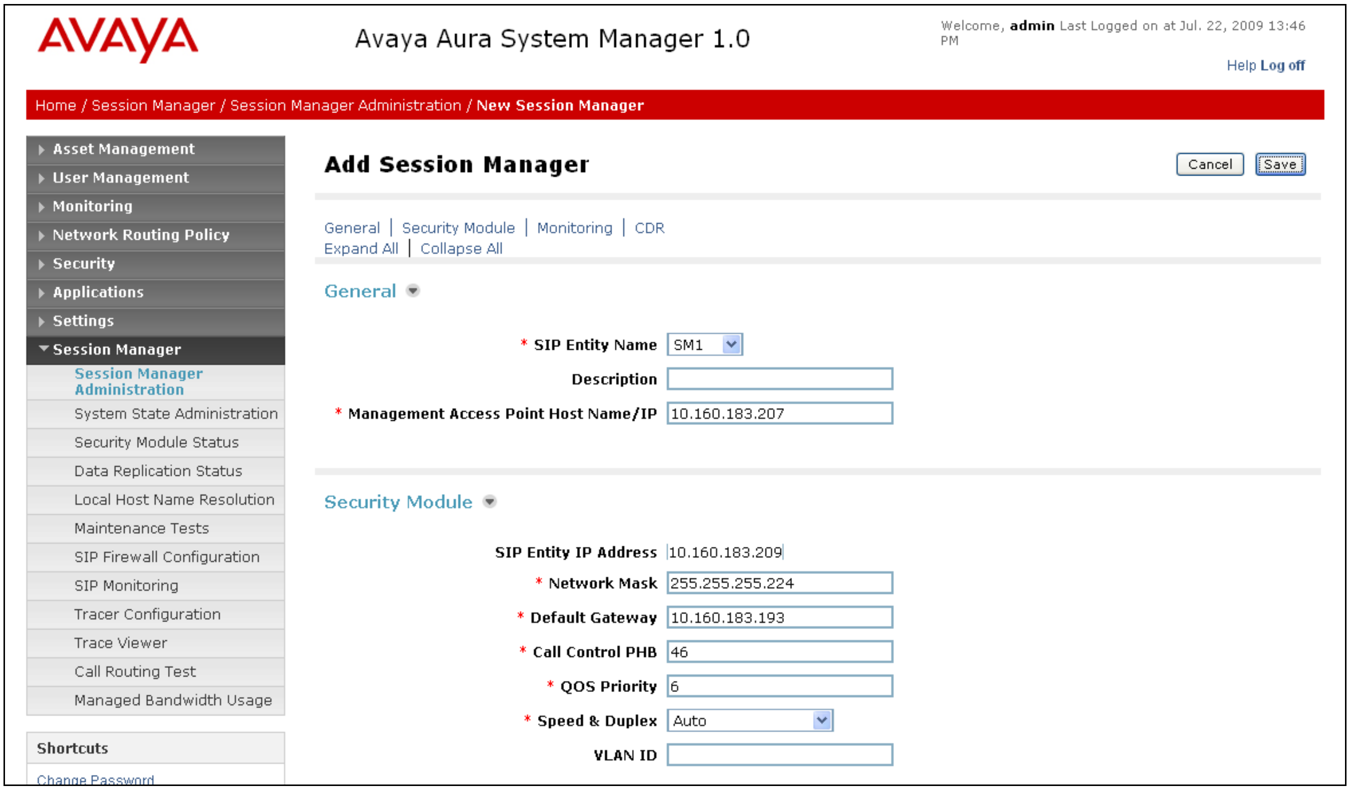

Figure 34: Add Session Manager Page

5. Avaya Aura™ Communication Manager

This section describes the administration steps for Avaya Aura™ Communication Manager in support of the sample configuration described in these Application Notes. The steps are performed from the Avaya Aura™ Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Avaya Aura™ Communication Manager administration, including stations, C-LAN, Media Processor, and announcement boards, etc., has already been performed. Consult [3] and [4] for further details if necessary.

5.1. System Parameters

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

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

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 1000  41
           Maximum Concurrently Registered IP Stations: 18000 10
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 5     0
                  Maximum Video Capable H.323 Stations: 10    0
                   Maximum Video Capable IP Softphones: 10    0
                       Maximum Administered SIP Trunks: 1000  152
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 1     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   1
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 35: System-Parameters Customer-Options Form – Page 2

2. On Page 4 of the **system-parameters customer-options** form, verify that the bolded field in the following screenshot is set to "**y**".

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

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

           IP Attendant Consoles? n
```

Figure 36: System-Parameters Customer-Options Form – Page 4

5.2. Dial Plan

Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings administered in **Figure 37**:

- 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 sample configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**3**" – local extensions for Avaya Aura™ Communication Manager stations, agents, and Vector Directory Numbers (VDNs) in this sample configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**5**" – local extensions for Avaya Aura™ Communication Manager skills (hunt groups) and announcements in this sample configuration conform to this format.

```
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
     3            5    ext
     5            5    ext
```

Figure 37: Dialplan Analysis Form

5.3. IP Network Parameters

These Application Notes assume that the appropriate IP network regions and IP codec sets have already been administered to support internal calls, i.e., calls within the Avaya site. For simplicity in this sample configuration, all Avaya Aura™ Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within the Avaya site are assigned to a single IP network region and all internal calls use a single IP codec set. This section describes the steps for administering an additional IP network region to represent the AT&T IP Toll Free service, and another IP codec set for external calls, i.e., inbound AT&T IP Toll Free calls.

1. Enter the **change ip-codec-set ci** command, where **ci** is the number of an IP codec set used only 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 as shown in **Figure 38**.

```
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
```

Figure 38: IP-Codec-Set Form for Internal Calls – Page 1

Repeat this step as necessary for each IP codec set used only for internal calls.

2. Enter the **change ip-codec-set ce** command, where **ce** is the number of an unused IP codec set. 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 in **Figure 39**.

```
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         2        20
 2: G.729A             n         2        20
 3: G.711MU            n         2        20
```

Figure 39: IP-Codec-Set Form for External Calls – Page 1

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
```

Figure 40: IP-Codec-Set Form for External Calls – Page 2

3. Enter the **change ip-network-region nr**, where **nr** is the number of an unused IP network region. This IP network region will be used to represent the AT&T IP Toll Free service.

```
change ip-network-region 61                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 61
Location:         Authoritative Domain:
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                   RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 41: IP-Network-Region Form for the Network Region Representing the Avaya IP Toll Free Service – Page 1

On Page 3 of the **ip-network-region** form, for each IP network region administered for local Avaya Aura™ Communication Manager elements within the Avaya site as the **dst rgn**, provision the following:

- **codec set** – Set to the codec set administered in Step 2.
- **direct WAN** – Set to "**y**".

- **WAN-BW-limits** – Set to the maximum number of calls or bandwidth allowed between the two IP network regions. The setting shown in **Figure 42** was used for testing purposes only.

```
change ip-network-region 61                                    Page   3 of  19

Source Region: 61    Inter Network Region Connection Management     I       M
                                                                    G  A    e
dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
1    2     y    NoLimit                                             n
2
3
4
5
6
7
8
9
10
11
12
13
14
15
```

Figure 42: IP-Network-Region Form for an IP Network Region Representing the AT&T IP Toll Free Service– Page 3

4. Enter the **change node-names ip** command, and add a node name and the IP address for the Avaya Aura™ Session Manager SM100 card. Also note the node name and IP address of a C-LAN board that is assigned to one of the IP network regions administered for local Avaya Aura™ Communication Manager elements within the Avaya site as described in Step 3. This C-LAN board will be used in Section 5.4 Step 1 for administering a SIP trunk to Avaya Aura™ Session Manager.

```
change node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ASM1                10.160.183.209
clan-01a09          10.160.179.110
```

Figure 43: Change Node-Names IP Form

5.4. Inbound Calls

This section describes the steps for administering the SIP trunk to Avaya Aura™ Session Manager.

1. Enter the **add signaling-group s** command, where **s** is the number of an unused signaling group, and provision the following:
   - **Group Type** – Set to “**sip**”.

- **Transport Method** – Set to "**tls**". Note that this is only the transport protocol used between Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. The transport protocol used between Avaya Aura™ Session Manager and the Acme Packet SBC is TCP, and the transport protocol used between the Acme Packet SBC and the AT&T IP Toll Free service is UDP.
- **Near-end Node Name** – Set to the node name of the C-LAN board noted in Section 5.3 Step 4.
- **Far-end Node Name** – Set to the node name of Avaya Aura™ Session Manager as administered in Section 5.3 Step 4.
- **Near-end Listen Port** and **Far-end Listen Port** – set to "**5061**".
- **Far-end Network Region** – Set to the IP network region administered in Section 5.3 Step 3 to represent the AT&T IP Toll Free service.
- **Far-end Domain** – Leave blank.
- **DTMF over IP** – Set to "**rtp-payload**" to enable Avaya Aura™ 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 MedPro resources when possible.

```
add signaling-group 61                                          Page   1 of   1

 Group Number: 61             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n


   Near-end Node Name: clan-01a09          Far-end Node Name: ASM1
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 61
Far-end Domain:

                                         Bypass If IP Threshold Exceeded? 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? n                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

Figure 44: Signaling-Group Form for Inbound AT&T IP Toll Free Calls

2. Enter the **add trunk-group t** command, where **t** is the number of an unused trunk group. On Page 1 of the **trunk-group** form, provision the following:
   - **Group Type** – Set to "**sip**".
   - **Group Name** – Enter a descriptive name.
   - **TAC** – Enter a trunk access code that is consistent with the dial plan.
   - **Direction** – Set to "**incoming**".
   - **Service Type** – Set to "**public-ntwrk**".
   - **Signaling Group** – Set to the number of the signaling group administered in Step 1.

- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group.

```
add trunk-group 61                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 61                       Group Type: sip          CDR Reports: y
  Group Name: ASM - AT&T IPTF                 COR: 1        TN: 1        TAC: 161
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                   Night Service:
                                     Auth Code? n
Service Type: public-ntwrk

                                                      Signaling Group: 61
                                                    Number of Members: 20
```

Figure 45: Trunk-Group Form for Inbound AT&T IP Toll Free Calls – Page 1

3. Enter the **change public-unknown-numbering 0** command to specify that connected party numbers are to be returned to the PSTN for inbound AT&T IP Toll Free service calls. In the **public-unknown-numbering** form, for each local extension range assigned to Avaya Aura™ Communication Manager phones, agents, skills (hunt groups), and VDNs, provision an entry as follows:
   - **Ext Len** – Enter the total number of digits in the local extension range.
   - **Ext Code –** Enter enough leading digits to identify the local extension range.
   - **Trk Grp(s)** – Enter the number of the trunk group administered in Step 2.
   - **CPN Prefix** – Leave blank. Avaya Aura™ Session Manager in Section 4.5.2 Steps 7 – 8 adds the appropriate prefix to form the appropriate connected party numbers, e.g., converts from extensions to complete numbers in the PAI header sent to the AT&T IP Toll Free service.
   - **CPN Len** – Enter the total number of digits in the local extension range.

   In **Figure 46**, for inbound calls to Avaya Aura™ Communication Manager extensions 3xxxx and 5xxxx, 5-digit connected party numbers 3xxxx and 5xxxx are sent (i.e., the connected party's extension is sent without modification).

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN           CPN
Len Code            Grp(s)    Prefix        Len
                                                      Total Administered: 70
 5  3               61                      5            Maximum Entries: 9999
 5  5               61                      5
```

Figure 46: Public-Unknown-Numbering Form

5.5. Call Center

The administration of Avaya Aura™ Communication Manager Call Center elements – agents, skills (hunt groups), vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Consult [3], [4], [5], and [6] for further details if necessary. The samples that follow are provided for reference purposes only.

```
display hunt-group 1001                                       Page   1 of   3
                                  HUNT GROUP

            Group Number: 1001                             ACD? y
              Group Name: IPTF Skill 1                   Queue? y
         Group Extension: 51001                         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:
```

Figure 47: Sample Skill (Hunt Group) Form – Page 1

```
display hunt-group 1001                                       Page   2 of   3
                                  HUNT GROUP

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


      Controlling Adjunct: none



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

Figure 48: Sample Skill (Hunt Group) Form – Page 2

```
display hunt-group 1001                                       Page   3 of   3
                                  HUNT GROUP

                       LWC Reception: none          AUDIX Name:


                      Message Center: none
```

Figure 49: Sample Skill (Hunt Group) Form – Page 3

```
display agent-loginID 32001                                     Page   1 of   2
                                AGENT LOGINID

                Login ID: 32001                                       AAS? n
                    Name: Agent-61000                              AUDIX? n
                      TN: 1                                 LWC Reception: spe
                     COR: 1                        LWC Log External Calls? n
           Coverage Path:                        AUDIX Name for Messaging:
           Security Code:
                                             LoginID for ISDN/SIP Display? n
                                                                 Password:
                                                    Password (enter again):
                                                               Auto Answer: station
                                                         MIA Across Skills: system
                                                 ACW Agent Considered Idle: system
                                                 Aux Work Reason Code Type: system
                                                   Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                                  Forced Agent Logout Time:   :
```

Figure 50: Sample Agent Form – Page 1

```
display agent-loginID 32001                                     Page   2 of   2
                                AGENT LOGINID
      Direct Agent Skill:                                  Service Objective? n
Call Handling Preference: skill-level                  Local Call Preference? n

    SN   RL SL           SN   RL SL           SN   RL SL           SN   RL SL
 1: 1001    1       16:                 31:                 46:
 2: 1002    2       17:                 32:                 47:
 3: 1003    3       18:                 33:                 48:
 4:                 19:                 34:                 49:
 5:                 20:                 35:                 50:
 6:                 21:                 36:                 51:
 7:                 22:                 37:                 52:
 8:                 23:                 38:                 53:
 9:                 24:                 39:                 54:
10:                 25:                 40:                 55:
11:                 26:                 41:                 56:
12:                 27:                 42:                 57:
13:                 28:                 43:                 58:
14:                 29:                 44:                 59:
15:                 30:                 45:                 60:
```

Figure 51: Sample Agent – Page 2

```
display vector 1001                                          Page   1 of   6
                                  CALL VECTOR

    Number: 1001            Name: RouteToSkill1
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? y
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? n   BSR? y  Holidays? n
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 52101
03 queue-to     skill 1001 pri m
04 wait-time    10  secs hearing music
05 announcement 52001
06 goto step    3             if unconditionally
07 stop08
09
10
11
12
```

Figure 52: Sample Vector

```
display vdn 31001                                            Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 31001
                                 Name*: Skill 1001
                           Destination: Vector Number        1001
                   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
```

Figure 53: Sample VDN

6. Avaya Modular Messaging

In this sample configuration, Avaya Modular Messaging is configured for MultiSite mode. MultiSite mode allows Avaya Modular Messaging to server subscribers in multiple locations. The administration for MultiSite mode is beyond the scope of these Application Notes. Consult [7], [8], [9], and [10] for further details.

7. Configure Acme Packet SBC

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

ANNOTATION: The local policy below governs the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, etc., reside to the AT&T IP Toll Free service.

```
local-policy
      from-address
                               *
      to-address
                               *
      source-realm
                               INSIDE-SM
      description
      activate-time            N/A
      deactivate-time          N/A
      state                    enabled
      policy-priority          none
      last-modified-by         admin@console
      last-modified-date       2009-05-26 17:55:28
      policy-attribute
            next-hop                 10.242.225.200
            realm                    OUTSIDE
            action                   none
            terminate-recursion      disabled
            carrier
            start-time               0000
            end-time                 2400
            days-of-week             U-S
            cost                     0
            app-protocol             SIP
            state                    enabled
            methods
            media-profiles
```

ANNOTATION: The local policy below governs the routing of SIP messages from the AT&T IP Toll Free service to Avaya Aura™ Session Manager.

```
local-policy
```

```
      from-address
                                    *
      to-address
                                    *
      source-realm
                                    OUTSIDE
      description
      activate-time                 N/A
      deactivate-time               N/A
      state                         enabled
      policy-priority               none
      last-modified-by              admin@console
      last-modified-date            2009-06-09 15:56:00
      policy-attribute
            next-hop                      10.160.183.209
            realm                         INSIDE-SM
            action                        none
            terminate-recursion           disabled
            carrier
            start-time                    0000
            end-time                      2400
            days-of-week                  U-S
            cost                          0
            app-protocol                  SIP
            state                         enabled
            methods
            media-profiles

media-manager
      state                         enabled
      latching                      enabled
      flow-time-limit               86400
      initial-guard-timer           300
      subsq-guard-timer             300
      tcp-flow-time-limit           86400
      tcp-initial-guard-timer       300
      tcp-subsq-guard-timer         300
      tcp-number-of-ports-per-flow  2
      hnt-rtcp                      disabled
      algd-log-level                NOTICE
      mbcd-log-level                NOTICE
      red-flow-port                 1985
      red-mgcp-port                 1986
      red-max-trans                 10000
      red-sync-start-time           5000
      red-sync-comp-time            1000
      media-policing                enabled
      max-signaling-bandwidth       7752190
      max-untrusted-signaling       80
      min-untrusted-signaling       20
      app-signaling-bandwidth       0
      tolerance-window              30
      rtcp-rate-limit               0
      min-media-allocation          32000
      min-trusted-allocation        60000
```

```
      deny-allocation                32000
      anonymous-sdp                  disabled
      arp-msg-bandwidth              32000
      fragment-msg-bandwidth         0
      rfc2833-timestamp              disabled
      default-2833-duration          100
      rfc2833-end-pkts-only-for-non-sig enabled
      translate-non-rfc2833-event    disabled
      dnsalg-server-failover         disabled
      last-modified-by               admin@console
      last-modified-date             2009-03-12 10:22:03
network-interface
      name                           wancom1
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.1.1
      sec-utility-addr               169.254.1.2
      netmask                        255.255.255.252
      gateway
      sec-gateway
      gw-heartbeat
            state                          disabled
            heartbeat                      0
            retry-count                    0
            retry-timeout                  1
            health-score                   0
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                    11
        hip-ip-list
      ftp-address
        icmp-address
      snmp-address
      telnet-address
      last-modified-by               admin@console
      last-modified-date             2009-03-12 10:21:39
network-interface
      name                           wancom2
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.2.1
      sec-utility-addr               169.254.2.2
      netmask                        255.255.255.252
      gateway
      sec-gateway
      gw-heartbeat
            state                          disabled
            heartbeat                      0
            retry-count                    0
```

```
            retry-timeout                  1
            health-score                   0
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                     11
        hip-ip-list
      ftp-address
        icmp-address
      snmp-address
      telnet-address
      last-modified-by                admin@console
      last-modified-date              2009-03-12 10:21:39
```

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

```
network-interface
      name                            s0p0
      sub-port-id                     0
      description
      hostname
      ip-address                      10.160.177.210
      pri-utility-addr                10.160.177.211
      sec-utility-addr                10.160.177.212
      netmask                         255.255.255.224
      gateway                         10.160.177.193
      sec-gateway
      gw-heartbeat
            state                          disabled
            heartbeat                      0
            retry-count                    0
            retry-timeout                  1
            health-score                   0
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                     11
      hip-ip-list
      ftp-address
      icmp-address
      snmp-address
      telnet-address
      last-modified-by                admin@console
      last-modified-date              2009-03-12 10:24:07
```

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

```
network-interface
      name                            s0p1
      sub-port-id                     0
```

```
        description
        hostname
        ip-address                      10.160.183.219
        pri-utility-addr                10.160.183.217
        sec-utility-addr                10.160.183.218
        netmask                         255.255.255.224
        gateway                         10.160.183.193
        sec-gateway
        gw-heartbeat
              state                           disabled
              heartbeat                       0
              retry-count                     0
              retry-timeout                   1
              health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        last-modified-by                admin@console
        last-modified-date              2009-05-26 18:01:51

ntp-config
        server                          10.152.6.12
        last-modified-by                admin@console
        last-modified-date              2009-03-12 10:20:46
phy-interface
        name                            s0p0
        operation-type                  Media
        port                            0
        slot                            0
        virtual-mac                     00:08:25:a0:f3:68
        admin-state                     enabled
        auto-negotiation                enabled
        duplex-mode                     FULL
        speed                           100
        last-modified-by                admin@console
        last-modified-date              2009-05-13 15:29:00
phy-interface
        name                            s0p1
        operation-type                  Media
        port                            1
        slot                            0
        virtual-mac                     00:08:25:a0:f3:69
        admin-state                     enabled
        auto-negotiation                enabled
        duplex-mode                     FULL
        speed                           100
        last-modified-by                admin@console
        last-modified-date              2009-05-26 14:51:45
```

```
phy-interface
      name                          s1p0
      operation-type                Media
      port                          0
      slot                          1
      virtual-mac                   00:08:25:a0:f3:6e
      admin-state                   enabled
      auto-negotiation              enabled
      duplex-mode                   FULL
      speed                         100
      last-modified-by              admin@console
      last-modified-date            2009-05-13 15:29:23
phy-interface
      name                          s1p1
      operation-type                Media
      port                          1
      slot                          1
      virtual-mac                   00:08:25:a0:f3:6f
      admin-state                   disabled
      auto-negotiation              enabled
      duplex-mode                   FULL
      speed                         100
      last-modified-by              admin@console
      last-modified-date            2009-05-13 15:29:37
phy-interface
      name                          wancom1
      operation-type                Control
      port                          1
      slot                          0
      virtual-mac
      wancom-health-score           8
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:21:30
phy-interface
      name                          wancom2
      operation-type                Control
      port                          2
      slot                          0
      virtual-mac
      wancom-health-score           9
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:21:30
```

ANNOTATION: The realm configuration “OUTSIDE” below represents the external network on which the AT&T IP Toll Free service resides, and applies two SIP manipulations (RemoveUPDATE and NAT_IP).

```
realm-config
      identifier                    OUTSIDE
      description
      addr-prefix                   0.0.0.0
      network-interfaces
                                    s0p0:0
      mm-in-realm                   enabled
      mm-in-network                 enabled
```

```
mm-same-ip                      enabled
mm-in-system                    enabled
bw-cac-non-mm                   disabled
msm-release                     disabled
generate-UDP-checksum           disabled
max-bandwidth                   0
fallback-bandwidth              0
max-priority-bandwidth          0
max-latency                     0
max-jitter                      0
max-packet-loss                 0
observ-window-size              0
parent-realm
dns-realm
media-policy
in-translationid
out-translationid
in-manipulationid               RemoveUPDATE
out-manipulationid              NAT_IP
manipulation-string
class-profile
average-rate-limit              0
access-control-trust-level      none
invalid-signal-threshold        4
maximum-signal-threshold        3000
untrusted-signal-threshold      10
nat-trust-threshold             0
deny-period                     60
ext-policy-svr
symmetric-latching              disabled
pai-strip                       disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching             none
restriction-mask                32
accounting-enable               enabled
user-cac-mode                   none
user-cac-bandwidth              0
user-cac-sessions               0
icmp-detect-multiplier          0
icmp-advertisement-interval     0
icmp-target-ip
monthly-minutes                 0
net-management-control          disabled
delay-media-update              disabled
refer-call-transfer             disabled
codec-policy
codec-manip-in-realm            disabled
constraint-name
call-recording-server-id
stun-enable                     disabled
stun-server-ip                  0.0.0.0
stun-server-port                3478
```

```
      stun-changed-ip                0.0.0.0
      stun-changed-port              3479
      match-media-profiles
      qos-constraint
      last-modified-by               admin@console
      last-modified-date             2009-04-22 19:26:23
```

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

```
realm-config
      identifier                     INSIDE-SM
      description
      addr-prefix                    0.0.0.0
      network-interfaces
                                     s0p1:0
      mm-in-realm                    enabled
      mm-in-network                  enabled
      mm-same-ip                     enabled
      mm-in-system                   enabled
      bw-cac-non-mm                  disabled
      msm-release                    disabled
      generate-UDP-checksum          disabled
      max-bandwidth                  0
      fallback-bandwidth             0
      max-priority-bandwidth         0
      max-latency                    0
      max-jitter                     0
      max-packet-loss                0
      observ-window-size             0
      parent-realm
      dns-realm
      media-policy
      in-translationid
      out-translationid
      in-manipulationid
      out-manipulationid
      manipulation-string
      class-profile
      average-rate-limit             0
      access-control-trust-level     high
      invalid-signal-threshold       0
      maximum-signal-threshold       0
      untrusted-signal-threshold     0
      nat-trust-threshold            0
      deny-period                    30
      ext-policy-svr
      symmetric-latching             disabled
      pai-strip                      disabled
      trunk-context
      early-media-allow
      enforcement-profile
      additional-prefixes
      restricted-latching            none
      restriction-mask               32
```

```
      accounting-enable               enabled
      user-cac-mode                   none
      user-cac-bandwidth              0
      user-cac-sessions               0
      icmp-detect-multiplier          0
      icmp-advertisement-interval     0
      icmp-target-ip
      monthly-minutes                 0
      net-management-control          disabled
      delay-media-update              disabled
      refer-call-transfer             disabled
      codec-policy
      codec-manip-in-realm            disabled
      constraint-name
      call-recording-server-id
      stun-enable                     disabled
      stun-server-ip                  0.0.0.0
      stun-server-port                3478
      stun-changed-ip                 0.0.0.0
      stun-changed-port               3479
      match-media-profiles
      qos-constraint
      last-modified-by                admin@console
      last-modified-date              2009-05-26 15:08:15
redundancy-config
      state                           enabled
      log-level                       INFO
      health-threshold                75
      emergency-threshold             50
      port                            9090
      advertisement-time              500
      percent-drift                   210
      initial-time                    1250
      becoming-standby-time           180000
      becoming-active-time            100
      cfg-port                        1987
      cfg-max-trans                   10000
      cfg-sync-start-time             5000
      cfg-sync-comp-time              1000
      gateway-heartbeat-interval      0
      gateway-heartbeat-retry         0
      gateway-heartbeat-timeout       1
      gateway-heartbeat-health        0
      media-if-peercheck-time         0
      peer
            name                            acmesbc-pri
            state                           enabled
            type                            Primary
            destination
                  address                         169.254.1.1:9090
                  network-interface               wancom1:0
            destination
                  address                         169.254.2.1:9090
                  network-interface               wancom2:0
      peer
```

```
        name                            acmesbc-sec
        state                           enabled
        type                            Secondary
        destination
              address                         169.254.1.2:9090
              network-interface               wancom1:0
        destination
              address                         169.254.2.2:9090
              network-interface               wancom2:0
  last-modified-by               admin@console
  last-modified-date             2009-03-12 10:21:53
```

ANNOTATION: The session agent below represents the AT&T IP Toll Free service border element.

```
session-agent
      hostname                    10.242.225.200
      ip-address                  10.242.225.200
      port                        5060
      state                       enabled
      app-protocol                SIP
      app-type
      transport-method            UDP
      realm-id                    OUTSIDE
      egress-realm-id
      description                 AT&T Border Element
      carriers
      allow-next-hop-lp           enabled
      constraints                 disabled
      max-sessions                0
      max-inbound-sessions        0
      max-outbound-sessions       0
      max-burst-rate              0
      max-inbound-burst-rate      0
      max-outbound-burst-rate     0
      max-sustain-rate            0
      max-inbound-sustain-rate    0
      max-outbound-sustain-rate   0
      min-seizures                5
      min-asr                     0
      time-to-resume              0
      ttr-no-response             0
      in-service-period           0
      burst-rate-window           0
      sustain-rate-window         0
      req-uri-carrier-mode        None
      proxy-mode
      redirect-action
      loose-routing               enabled
      send-media-session          enabled
      response-map
      ping-method                 OPTIONS;hops=0
      ping-interval               300
      ping-send-mode              keep-alive
      ping-in-service-response-codes
```

```
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid
        manipulation-string
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy
        enforcement-profile
        refer-call-transfer             disabled
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        last-modified-by                admin@console
        last-modified-date              2009-05-20 21:31:18
```

ANNOTATION: The session agent below represents the Avaya Aura™ Session Manager used in the sample configuration.

```
session-agent
        hostname                        10.160.183.209
        ip-address                      10.160.183.209
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                StaticTCP
        realm-id                        INSIDE-SM
        egress-realm-id
        description                     Avaya Session Manager
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
```

```
max-sustain-rate               0
max-inbound-sustain-rate       0
max-outbound-sustain-rate      0
min-seizures                   5
min-asr                        0
time-to-resume                 0
ttr-no-response                0
in-service-period              0
burst-rate-window              0
sustain-rate-window            0
req-uri-carrier-mode           None
proxy-mode
redirect-action
loose-routing                  enabled
send-media-session             enabled
response-map
ping-method                    OPTIONS
ping-interval                  300
ping-send-mode                 keep-alive
ping-in-service-response-codes
out-service-response-codes
media-profiles
in-translationid
out-translationid
trust-me                       disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me                    disabled
in-manipulationid
out-manipulationid
manipulation-string
p-asserted-id
trunk-group
max-register-sustain-rate      0
early-media-allow
invalidate-registrations       disabled
rfc2833-mode                   none
rfc2833-payload                0
codec-policy
enforcement-profile
refer-call-transfer            disabled
reuse-connections              TCP
tcp-keepalive                  none
tcp-reconn-interval            10
max-register-burst-rate        0
register-burst-window          0
last-modified-by               admin@console
last-modified-date             2009-06-02 18:24:49
```

ANNOTATION: The sip-config defines global sip-parameters, including SIP timers, SIP options, which realm to send requests to if not specified elsewhere, and enabling the SD to collect statistics on requests other than REGISTERs and INVITEs.

```
sip-config
      state                           enabled
      operation-mode                  dialog
      dialog-transparency             enabled
      home-realm-id                   INSIDE-SM
      egress-realm-id                 INSIDE-SM
      nat-mode                        None
      registrar-domain
      registrar-host
      registrar-port                  0
      register-service-route          always
      init-timer                      500
      max-timer                       4000
      trans-expire                    32
      invite-expire                   180
      inactive-dynamic-conn           32
      enforcement-profile
      pac-method
      pac-interval                    10
      pac-strategy                    PropDist
      pac-load-weight                 1
      pac-session-weight              1
      pac-route-weight                1
      pac-callid-lifetime             600
      pac-user-lifetime               3600
      red-sip-port                    1988
      red-max-trans                   10000
      red-sync-start-time             5000
      red-sync-comp-time              1000
      add-reason-header               disabled
      sip-message-len                 4096
      enum-sag-match                  disabled
      extra-method-stats              enabled
      registration-cache-limit        0
      register-use-to-for-lp          disabled
      options                         max-udp-length=0
                                      set-inv-exp-at-100-resp
      add-ucid-header                 disabled
      last-modified-by                admin@console
      last-modified-date              2009-05-26 19:33:56
```

ANNOTATION: The SIP interface below is used to communicate with the AT&T IP Toll Free service.

```
sip-interface
      state                          enabled
      realm-id                       OUTSIDE
      description
      sip-port
            address                        10.160.177.210
            port                           5060
            transport-protocol             UDP
            tls-profile
            allow-anonymous                agents-only
            ims-aka-profile
      carriers
      trans-expire                   0
      invite-expire                  0
      max-redirect-contacts          0
      proxy-mode
      redirect-action
      contact-mode                   none
      nat-traversal                  none
      nat-interval                   30
      tcp-nat-interval               90
      registration-caching           disabled
      min-reg-expire                 300
      registration-interval          3600
      route-to-registrar             disabled
      secured-network                disabled
      teluri-scheme                  disabled
      uri-fqdn-domain
      trust-mode                     all
      max-nat-interval               3600
      nat-int-increment              10
      nat-test-increment             30
      sip-dynamic-hnt                disabled
      stop-recurse                   401,407
      port-map-start                 0
      port-map-end                   0
      in-manipulationid
      out-manipulationid
      manipulation-string
      sip-ims-feature                disabled
      operator-identifier
      anonymous-priority             none
      max-incoming-conns             0
      per-src-ip-max-incoming-conns  0
      inactive-conn-timeout          0
      untrusted-conn-timeout         0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode           pass
      charging-function-address-mode pass
      ccf-address
```

```
      ecf-address
      term-tgrp-mode                 none
      implicit-service-route         disabled
      rfc2833-payload                101
      rfc2833-mode                   transparent
      constraint-name
      response-map
      local-response-map
      ims-aka-feature                disabled
      enforcement-profile
      refer-call-transfer            disabled
      route-unauthorized-calls
      tcp-keepalive                  none
      add-sdp-invite                 disabled
      add-sdp-profiles
      last-modified-by               admin@console
      last-modified-date             2009-05-14 18:46:43
```

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

```
sip-interface
      state                          enabled
      realm-id                       INSIDE-SM
      description
      sip-port
            address                        10.160.183.219
            port                           5060
            transport-protocol             TCP
            tls-profile
            allow-anonymous                agents-only
            ims-aka-profile
      carriers
      trans-expire                   30
      invite-expire                  0
      max-redirect-contacts          0
      proxy-mode
      redirect-action
      contact-mode                   none
      nat-traversal                  none
      nat-interval                   30
      tcp-nat-interval               90
      registration-caching           disabled
      min-reg-expire                 300
      registration-interval          3600
      route-to-registrar             disabled
      secured-network                disabled
      teluri-scheme                  disabled
      uri-fqdn-domain
      trust-mode                     all
      max-nat-interval               3600
      nat-int-increment              10
      nat-test-increment             30
      sip-dynamic-hnt                disabled
      stop-recurse                   401,407
```

```
      port-map-start                 0
      port-map-end                   0
      in-manipulationid
      out-manipulationid
      manipulation-string
      sip-ims-feature                disabled
      operator-identifier
      anonymous-priority             none
      max-incoming-conns             0
      per-src-ip-max-incoming-conns  0
      inactive-conn-timeout          0
      untrusted-conn-timeout         0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode           pass
      charging-function-address-mode pass
      ccf-address
      ecf-address
      term-tgrp-mode                 none
      implicit-service-route         disabled
      rfc2833-payload                101
      rfc2833-mode                   transparent
      constraint-name
      response-map
      local-response-map
      ims-aka-feature                disabled
      enforcement-profile
      refer-call-transfer            disabled
      route-unauthorized-calls
      tcp-keepalive                  none
      add-sdp-invite                 disabled
      add-sdp-profiles
      last-modified-by               admin@console
      last-modified-date             2009-05-26 18:06:22
```

ANNOTATION: The SIP manipulation below performs address translation and topology hiding for SIP messages between the AT&T IP Toll Free services and the Avaya elements.

```
sip-manipulation
      name                           NAT_IP
      description                    Topology hiding for TO and FROM SIP
      header-rule
            name                           manipFrom
            header-name                    From
            action                         manipulate
            comparison-type                case-sensitive
            match-value
            msg-type                       request
            new-value
            methods
            element-rule
                  name                           FROM
                  parameter-name
```

```
            type                            uri-host
            action                          replace
            match-val-type                  ip
            comparison-type                 case-sensitive
            match-value
            new-value                       $LOCAL_IP
    header-rule
        name                            manipTo
        header-name                     To
        action                          manipulate
        comparison-type                 case-sensitive
        match-value
        msg-type                        request
        new-value
        methods
        element-rule
            name                            TO
            parameter-name
            type                            uri-host
            action                          replace
            match-val-type                  ip
            comparison-type                 case-sensitive
            match-value
            new-value                       $REMOTE_IP
    last-modified-by                admin@console
    last-modified-date              2009-03-12 10:22:14
```

ANNOTATION: The SIP manipulation below removes "UPDATE" from the Allow header in SIP messages from the AT&T IP Toll Free service.

```
sip-manipulation
    name                            RemoveUPDATE
    description                     Strip Update from Allow list
    header-rule
        name                            EditAllow
        header-name                     Allow
        action                          manipulate
        comparison-type                 pattern-rule
        match-value
        msg-type                        any
        new-value
        methods
        element-rule
            name                            StripUPDATE
            parameter-name
            type                            header-value
            action                          find-replace-all
            match-val-type                  any
            comparison-type                 pattern-rule
            match-value                     (,\s*UPDATE|UPDATE\s*,)
            new-value
    last-modified-by                admin@console
    last-modified-date              2009-04-22 19:25:08
```

ANNOTATION: The steering pools below define the RTP port range on the respective realms.

```
steering-pool
      ip-address                    10.160.177.210
      start-port                    49152
      end-port                      65535
      realm-id                      OUTSIDE
      network-interface
      last-modified-by              admin@console
      last-modified-date            2009-03-25 19:11:47
steering-pool
      ip-address                    10.160.183.219
      start-port                    49152
      end-port                      65535
      realm-id                      INSIDE-SM
      network-interface
      last-modified-by              admin@console
      last-modified-date            2009-05-26 18:08:01
system-config
      hostname                      acmesbc-pri
      description
      location
      mib-system-contact
      mib-system-name
      mib-system-location
      snmp-enabled                  enabled
      enable-snmp-auth-traps        disabled
      enable-snmp-syslog-notify     disabled
      enable-snmp-monitor-traps     disabled
      enable-env-monitor-traps      disabled
      snmp-syslog-his-table-length  1
      snmp-syslog-level             WARNING
      system-log-level              WARNING
      process-log-level             NOTICE
      process-log-ip-address        0.0.0.0
      process-log-port              0
      collect
            sample-interval               5
            push-interval                 15
            boot-state                    disabled
            start-time                    now
            end-time                      never
            red-collect-state             disabled
            red-max-trans                 1000
            red-sync-start-time           5000
            red-sync-comp-time            1000
            push-success-trap-state       disabled
      call-trace                    disabled
      internal-trace                disabled
      log-filter                    all
      default-gateway               172.16.253.4
      restart                       enabled
      exceptions
      telnet-timeout                0
```

```
console-timeout                0
remote-control                 enabled
cli-audit-trail                enabled
link-redundancy-state          disabled
source-routing                 enabled
cli-more                       disabled
terminal-height                24
debug-timeout                  0
trap-event-lifetime            0
last-modified-by               admin@console
last-modified-date             2009-03-12 10:20:46
```

8. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura™ System Manager, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, Avaya phones, fax machines, an Acme Packet SBC, and Avaya Modular Messaging.
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected.

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

- Inbound AT&T IP Toll Free service calls to Avaya Aura™ Communication Manager VDNs, agents, and phones.
- Call and two-way talkpath establishment between callers and Avaya Aura™ Communication Manager agents/phones.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs.
- T.38 for inbound fax calls from the AT&T IP Toll Free service with G3 and SG3 fax endpoints.
- DTMF tone transmission using RFC 2833 in both directions.
- Avaya Aura™ Communication Manager phones sending DTMF to the AT&T IP Toll Free to invoke AT&T IP Toll Free Legacy Transfer Connect features, and Avaya Aura™ Communication Manager processing the resulting DTMF responses from the AT&T IP Toll Free service.
- Inbound AT&T IP Toll Free service calls to Avaya Aura™ Communication Manager that are directly routed to agents and unanswered can be covered to Avaya Modular Messaging.
- Long duration calls.

The above test objectives of Section 8 with limitations as noted in Section 1.3 were verified.

9. Verification Steps

9.1. Verification Tests

The following steps may be used to verify the configuration:

1. Verify the call routing administration on Avaya Aura™ Session Manager. In the left pane of the Avaya Aura™ System Manager Common Console, under **Session Manager**, click on "**Call Routing Test**". In the **Call Routing Test** page, enter the appropriate parameters of the test call. **Figure 54** shows a routing test for an inbound call arriving from the Acme Packet SBC (note the IP address "**10.160.183.219**" in the **Calling Party Address** field) to the number "**000001001**" in the SIP domain "**spdevcon.com**" (note the **Called Party URI** field). Click on "**Execute**".

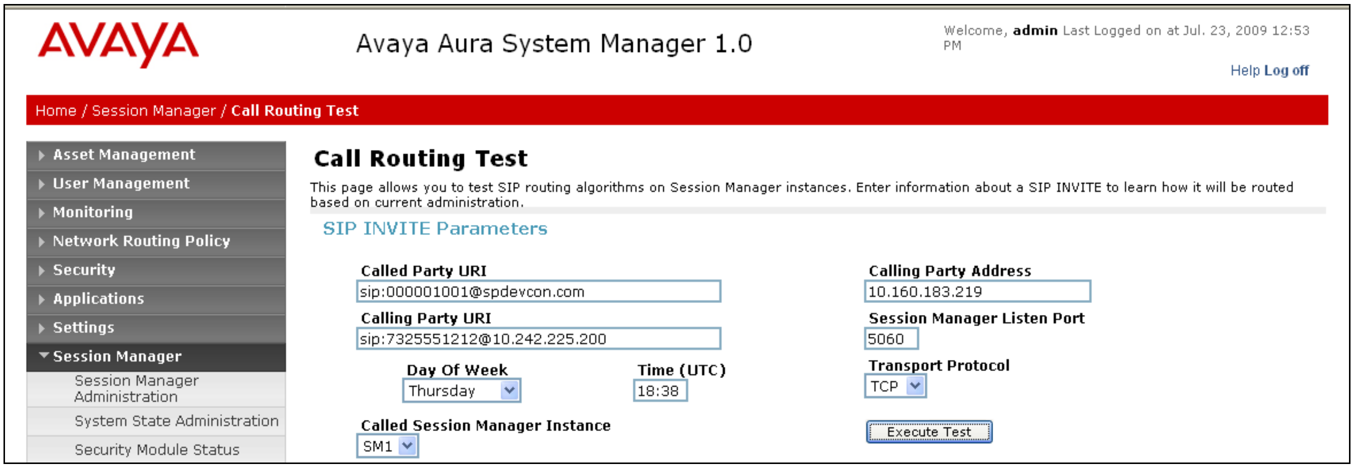

Figure 54: Call Routing Test Page

2. Verify that the test results in the **Routing Decisions** and **Routing Decision Process** are consistent with the expected results of the routing administration administered on Avaya Aura™ Session Manager in Section 4.

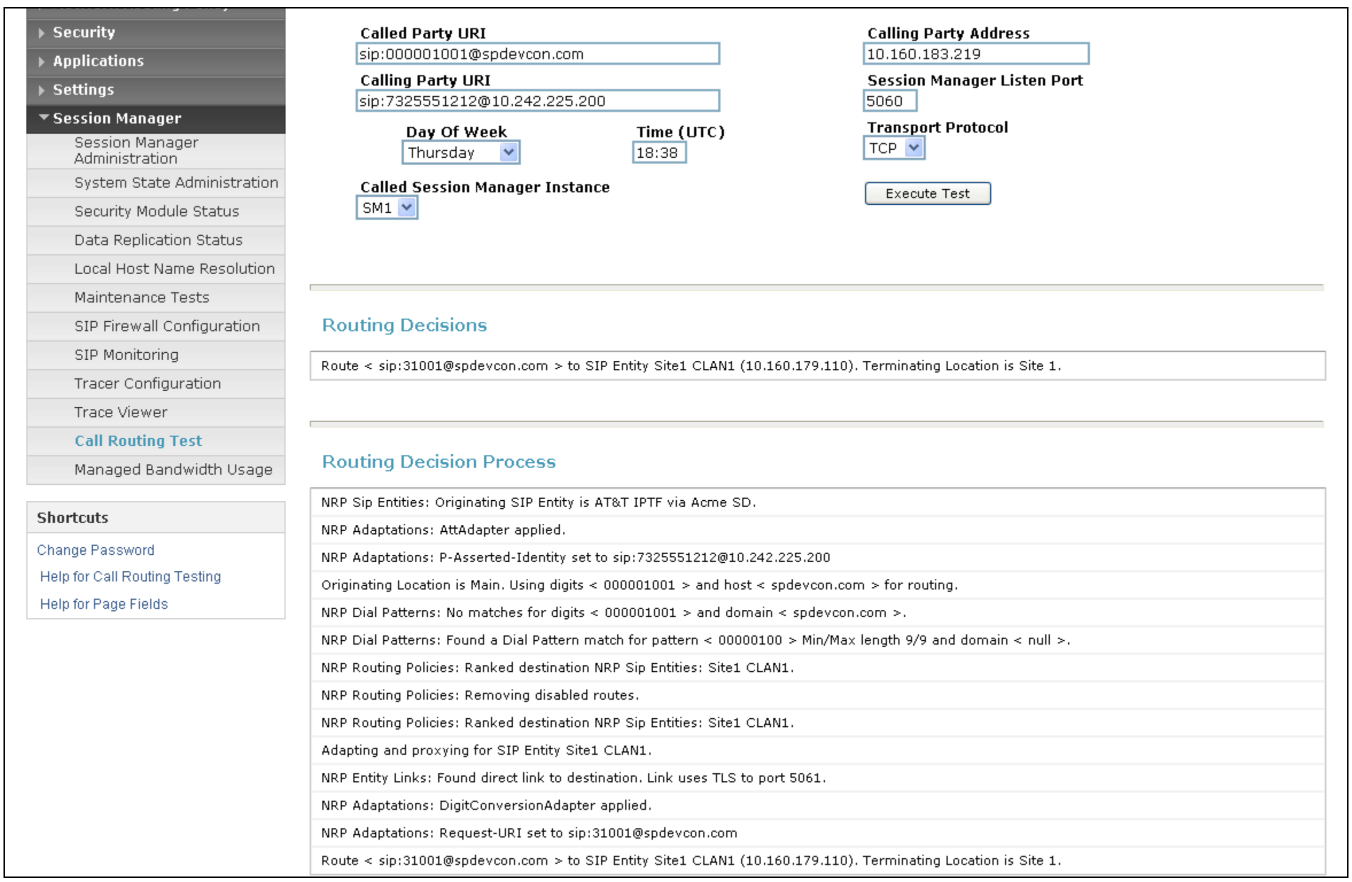

Figure 55: Call Routing Test Page – Test Results

3. Place an inbound call, answer the call, and verify that two-way talkpath exists. Verify that the call remains stable for several minutes and disconnect properly.
4. Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to voicemail.

9.2. Troubleshooting Tools

The Avaya Aura™ Communication Manager "list trace vector", "list trace vdn", "list trace tac", and/or "status trunk-group" commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The logging and reporting functions within the Avaya Aura™ System Manager Common Console may be used to examine the details of Avaya Aura™ Session Manager calls. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

10. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and the Acme Packet Net-Net Session Director 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. These Application Notes further demonstrated that the Avaya Aura™ Session Manager AT&T Adaptation Module could be utilized to remove History-Info header information on egress SIP messages to the AT&T IP Toll Free service.

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

The sample 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.

[1] *Avaya Aura™ Session Manager Overview*, Issue 1, Release 1.1, May 2009, Document Number 03-603323

[2] *Installing and Administering Avaya Aura™ Session Manager*, Issue 1.1, Release 1.1, June 2009, Document Number 03-603324

[3] *Administering Avaya Aura™ Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509

[4] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205

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

[6] *Avaya Aura™ Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568

[7] *Modular Messaging MultiSite Guide Release 5.1*, June 2009

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

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

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

Acme Packet Support (login required):

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

AT&T IP Toll Free Service Descriptions:

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

12. Change History

Issue	Date	Reason
1.0		Initial issue.

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.