Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Session Manager 5.2, Avaya Aura™ Communication Manager 5.2.1, and Acme Packet Net-Net Session Director 6.1.0 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.

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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 5.2.1 supports the possibility of using SIP phones as stations, SIP phones were not tested as part of the reference 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.729B 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. G.726 codec is not supported between Avaya Aura™ Communication Manager and 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:

- 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. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communications services for a particular enterprise site. In this sample configuration, 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 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.
- 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. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme Packet SBC and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

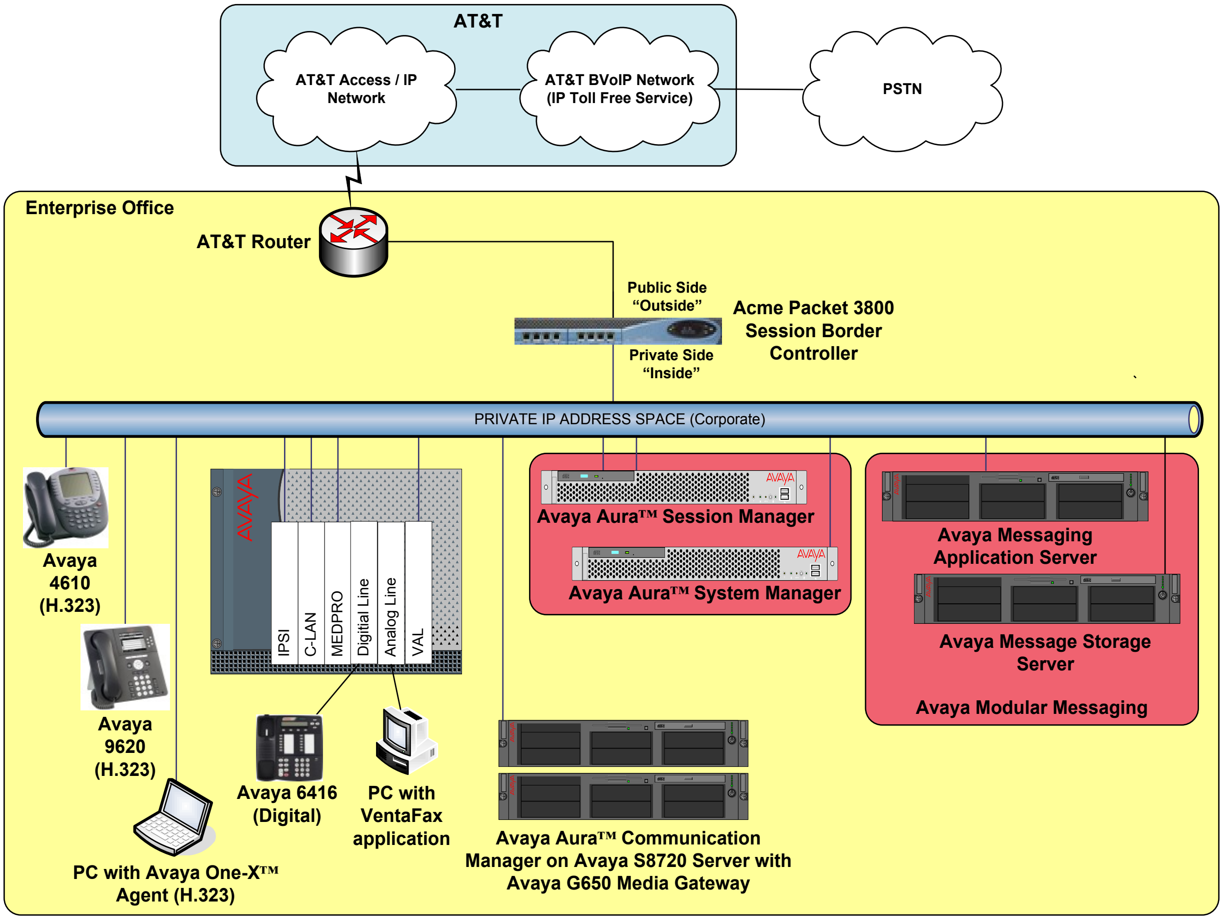

Figure 1: Reference Configuration

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.

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

Component	Illustrative Value in these Application Notes
Avaya Aura™ System Manager	
Management IP Address	192.168.67.135
Avaya Aura™ Session Manager	
Management IP Address	192.168.67.136
SM100 Card IP Address	192.168.67.137
Avaya Aura™ Communication Manager	
C-LAN IP Address	192.168.67.13
Vector Directory Number (VDN) Extension	26120
Skill (Hunt Group) Extensions	1002, 1003, 1004
Agent Extensions	26666, 26667, 26668
VDN Extensions	26112, 26113, 26114, 26116
Phone Extensions	26102, 26103, 26104
Voice Messaging Pilot Extension	26002
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Message Storage Server (MSS) IP Address	192.168.67.140
Pilot Number	17231126002
Acme Packet SBC	
IP Address of Public Interface (connected to AT&T IP Toll Free Service)	192.168.64.130
IP Address of Private Interface (connected to Avaya CPE)	192.168.67.130
AT&T IP Toll Free Service	
Border Element IP Address	135.25.29.74
Digits Passed in SIP Request-URI	00000104x

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

1. A PSTN 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 Session Manager.
5. 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, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to a) a vector, which in turn, routes the call to an agent, or b) directly to an agent or phone.

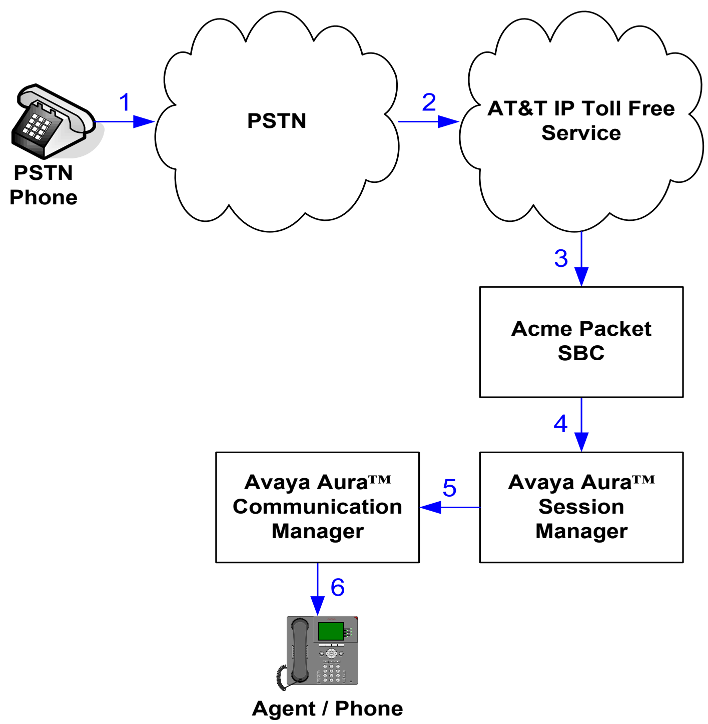

Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Phone

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

1. Same as the **Steps 1-5** and **Step 6b** from the first call scenario.
2. The called Communication Manager agent or phone does not answer the call, and the call covers to the agent's or phone's voicemail. Communication Manager forwards[2] the call to Session Manager.
3. 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, Session Manager routes the call to Modular Messaging. 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 Communication Manager.

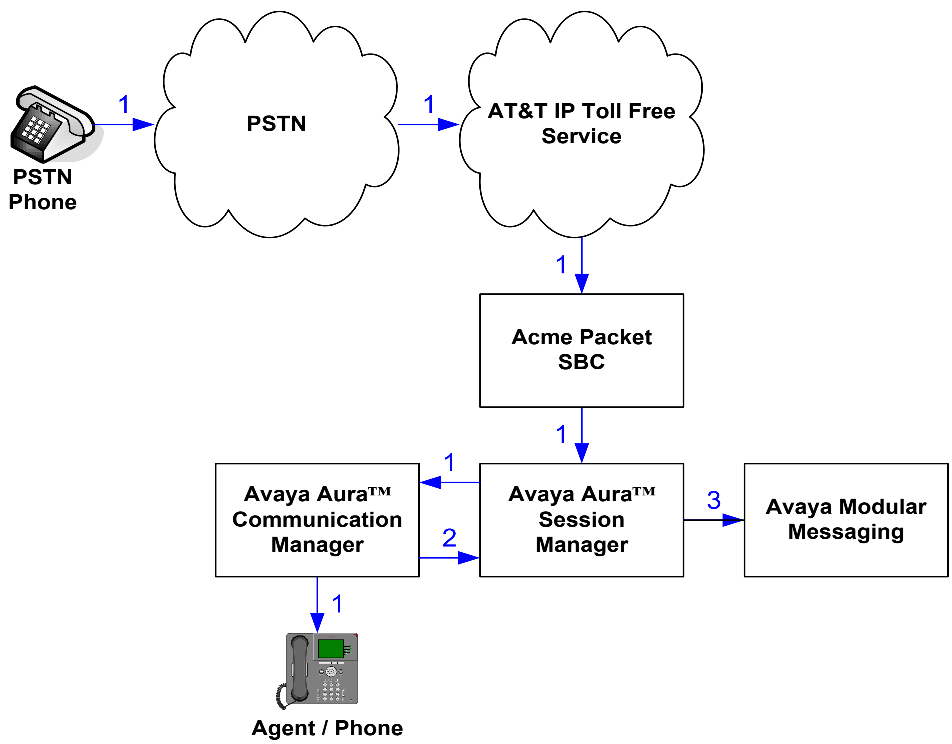

Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Phone Covered to Avaya Modular Messaging

[2] Communication Manager places a call to Modular Messaging, and then connects the inbound caller to Modular Messaging. SIP redirect methods, e.g., 302, are not used.

[3] The SIP signaling path still goes through Communication Manager. In addition, since the inbound call and Modular Messaging use different codecs (G.729 and G.711, respectively), Communication Manager performs the transcoding, and thus the RTP media path also goes through Communication Manager.

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™ Session Manager 5.2 – 5.2.1.1.521012
Avaya S8510 Server	Avaya Aura™ System Manager 5.2 – 5.2.1.0.521001
Avaya S8710 Server	Avaya Aura™ Communication Manager 5.2.1 - R015x.02.1.016.4 with Service Pack 1 - 02.1.016.4-17959
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface (IPSI)	HW15 FW050
TN799DP Control-LAN (C-LAN)	HW01 FW037
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW054
TN2501AP VAL-ANNOUNCEMENT	HW03 FW021
TN2224CP Digital Line	HW08 FW015
TN793CP Analog Line	HW05 FW010
Avaya 9630 IP Telephone	ha96xxua3_10.bin
Avaya 4610SW IP Telephone	a10d01b2_9_1.bin
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Server	Avaya Modular Messaging 5.2 with Service Pack 1- 520101
Fax	Ventafax 6.1.59.144
Acme Packet Net-Net Session Director 3800	SCX610m3p1
AT&T IP Toll Free Service	VNI16

Table 2: Equipment and Software Versions

4. Avaya Aura™ Session Manager

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

4.1. Background

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

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

4.2. Network Routing Policies

Network Routing Policies define how 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 Session Manager instances, 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 Session Manager instances and other SIP Entities.
- SIP Domains – SIP Domains are the domains for which Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, 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 may be used 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 Session Manager and matches a certain Dial Pattern, then 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 Session Manager, apply any called party number modifications necessary to "normalize" the number to a common format or uniform number. For example, assume that there are three SIP Entities representing three different Communication Manager systems, and a SIP Entity representing a centralized voicemail system, e.g., Modular Messaging in MultiSite mode. Further, assume that each Session Manager system dials a different pilot extension to call Modular Messaging. To simplify the routing for such calls, in 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 Modular Messaging.

[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.

- 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 Session Manager to Communication Manager, modify the called party number such that the number is consistent with the dial plan on Communication Manager.

For example, in the reference configuration DNIS digits (00000104x) specified in the Request URIs from the AT&T Toll Free service, are converted to Communication Manager local extensions/VDNs (261xx) by the Session Manager Adaptation *AT&T Adaptation* (see **Section 4.5**). These local extensions are then routed to Communication Manager by the associated dial pattern *261xx* (see **Section 4.10**). Of course, this is just one of many possible strategies that can be implemented with Session Manager to route calls.

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

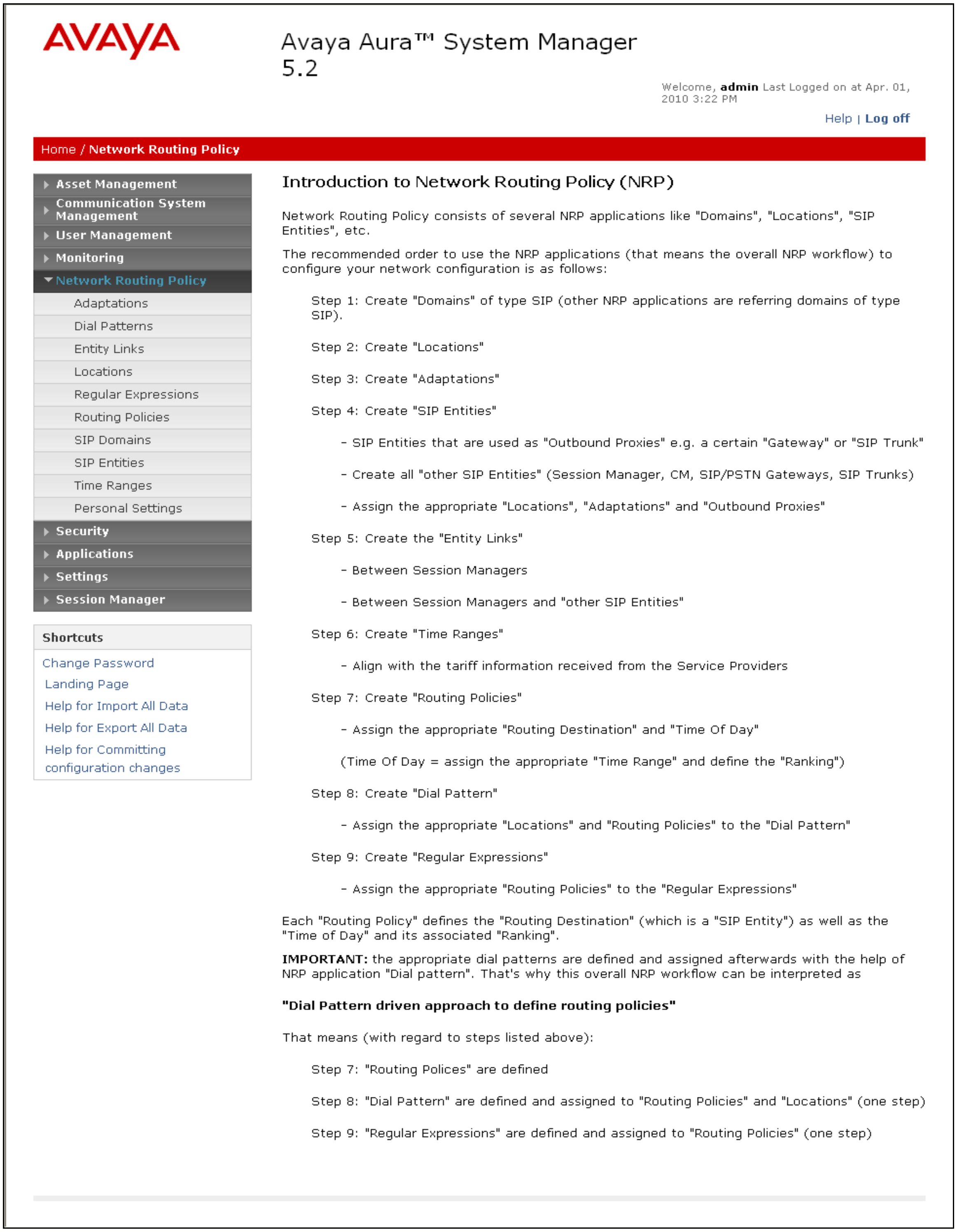

Figure 4: Introduction to Network Routing Policy (NRP) Page

4.3. SIP Domains

The steps in this section specify the SIP domains for which 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 **Domain Management** page, enter a SIP domain for **Name** (i.e. **customera.com**)and click on "**Commit**".

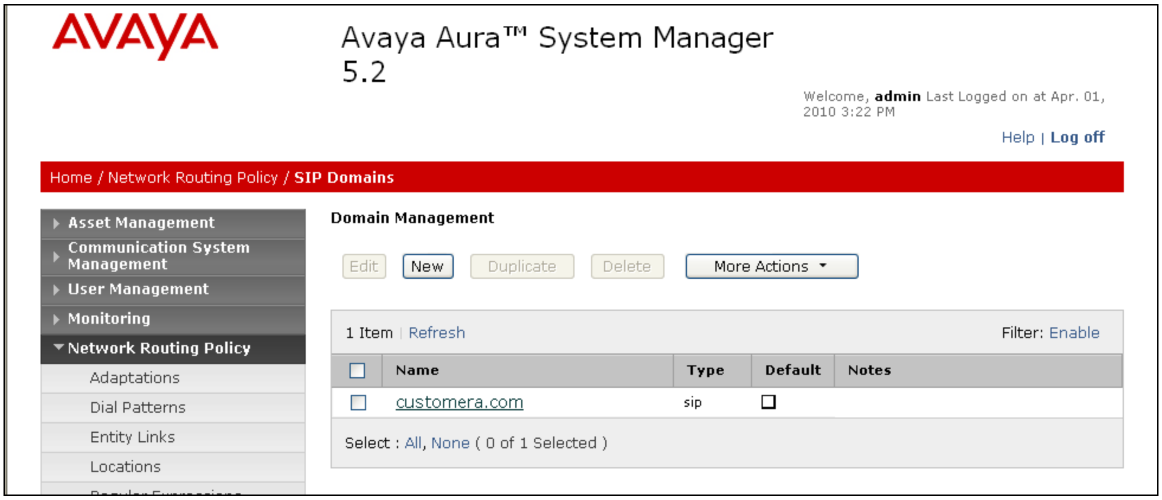

Figure 5: SIP Domains Page

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. In the reference configuration only one location is used for all the Avaya CPE.

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 Bandwidth** and **Average Bandwidth per Call**.
4. [Optional] To identify IP addresses associated with this Location, add **Location Pattern** entries accordingly.
5. Click on "**Commit**".

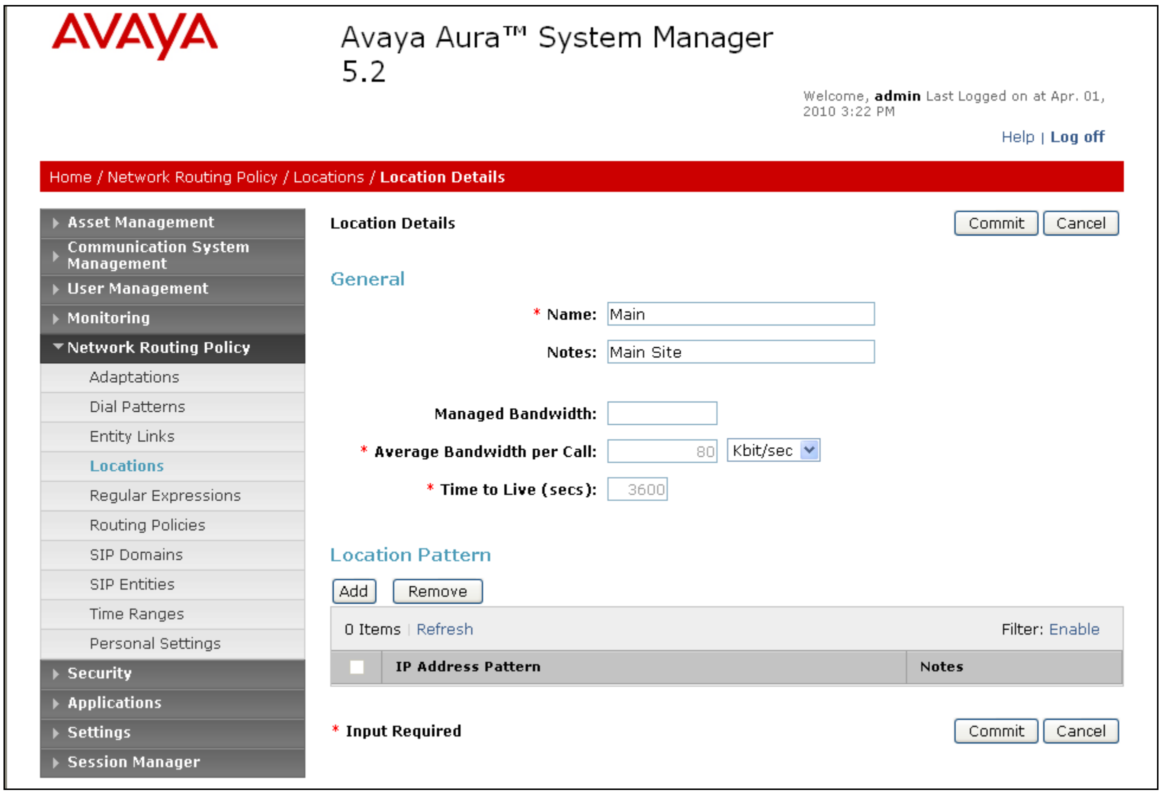

Figure 6: Location Details Page – Main Site

6. Repeat **Steps 1 - 5** to add any additional Locations.

4.5. Adaptations

As described in **Section 4.2**, Adaptations may be applied to "origination", "destination" or "both". In the reference configuration Adaptations were applied to destination" type headers. In this section, Adaptations are administered for the following purposes:

- Modification of digit strings in URIs of "destination" type headers in SIP messages received from the AT&T IP Toll Free service.
- Modification of digit strings in URIs of "destination" type headers in SIP messages received from Communication Manager.
- Modification of digit strings in URIs of "destination" type headers in SIP messages received from Modular Messaging (MWI).

The digit manipulations are performed by specifying the special "AttAdapter" Adaptation[5] or the basic "DigitConversionAdapter" Adaptation.

Note – As digit conversion (and their resulting call processing) may be performed on both incoming and outgoing calls between Session Manager and the various SIP Entities, the following sections should be viewed as examples of possible call processing. Other call processing solutions are possible.

4.5.1. Adaptation for AT&T

The Adaptation administered in this section (AttAdapter) is applied to SIP messages sent from the AT&T IP Toll Free service (by way of the Acme Packet SBC in the Main Location) to Communication Manager. Here inbound AT&T IP Toll Free service DNIS numbers (00000104x) are converted to their associated Communication Manager extensions (261xx) or the Modular Messaging pilot number (26000). Dial Patterns then route the calls to Communication Manager or Modular Messaging (Section 4.10).

DNIS numbers are mapped as follows:

- Calls are directed to Skill/Agent VDNs (26112, 26113, and 26114).
- Calls are directed to a DTMF digit manipulated option menu VDN 26116.
- Calls are directed to the Modular Messaging pilot number extension (26000).

[5] Currently, in addition to digit manipulation, the AT&T Adaptation removes any History-Info headers from SIP messages sent to AT&T.

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 in **Adaption name** and "**AttAdapter**" for Adaptation **Module name.**
3. In the **Matching Pattern fields** enter the DNIS digit strings sent by AT&T Toll Free inbound calls.
4. In the **Min** and **Max** fields enter the DNIS number lengths (in the reference configuration all DNIS numbers are **9** digits long**).**
5. In the **Insert Digits** fields enter the corresponding Communication Manager extension (station or skill VDN).
6. In the **Address to modify** fields enter **destination**.
7. In the **Notes** fields enter a description if desired and click on "**Commit**".

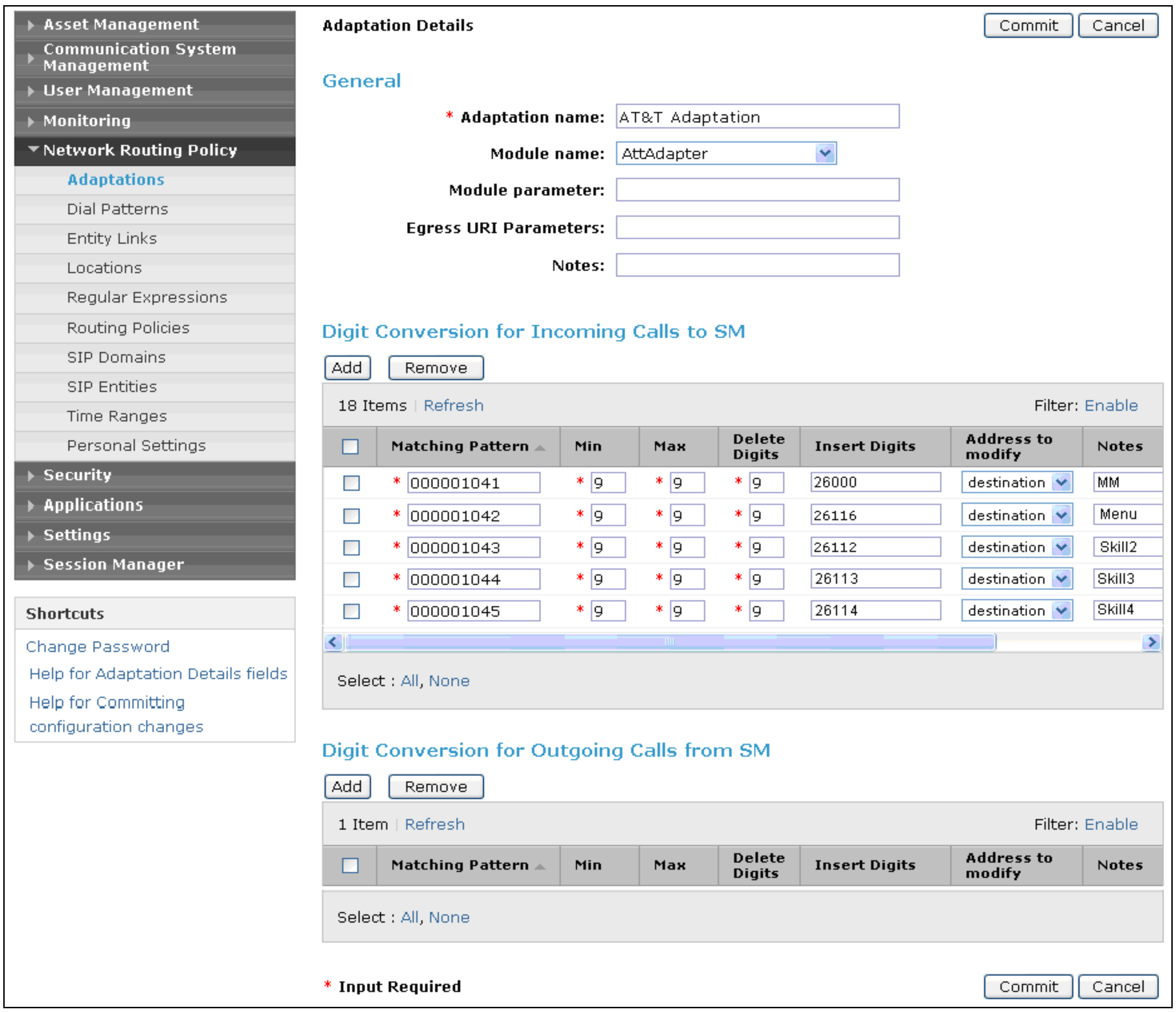

Figure 7: Adaptation Details Page – Adaptation for AT&T

4.5.2. Adaptation for Avaya Aura™ Communication Manager

The Adaptation administered in this section (DigitConversionAdapter) is used for digit conversion on SIP messages from Communication Manager as follows:

- On ingress SIP messages from Communication Manager where the Request-URI contains the Modular Messaging pilot extension (26000), the Adaptation converts the pilot extension to the 11-digit pilot number (17231126000)[6].
  1. In the **Adaptations** page (not shown), click on "**New**".
  2. In the **Adaptation Details** page, enter a descriptive **Adaptation name** and "**DigitConversionAdapter**" for Adaptation **Module name**.
  3. In the **Digit Conversion for Incoming Calls to SM** section, click on "**Add**" to provision an entry for converting the Modular Messaging Pilot extension (26000) on Communication Manager to the associated 11-digit Modular Messaging pilot number. Provision the entry as follows:
     - **Matching Pattern –** Enter enough leading digits to uniquely match the extension range (e.g. **26000**).
     - **Min** and **Max –** Enter the total number of digits in the extension range (e.g. **5**).
     - **Delete Digits –** If necessary, enter the number of leading digits that need to be deleted from the extension range. In the reference configuration this field was set to **0**.
     - **Insert Digits –** Enter the leading digits that need to be added to the extension range (e.g. **172311**) to match the corresponding Modular Messaging pilot number (e.g. 26000 becomes 17231126000).
     - **Address to modify –** Select "**destination**".
  4. Repeat **Steps 1-3** to add any additional digit conversions as necessary.
  5. Click on "**Commit**".

[6] With the assumption that the pilot extensions dialed by other SIP Entities, e.g., any other Avaya Aura™ Communication Manager systems connected to this Avaya Aura™ Session Manager, will also be converted to the same 11-digit pilot number.

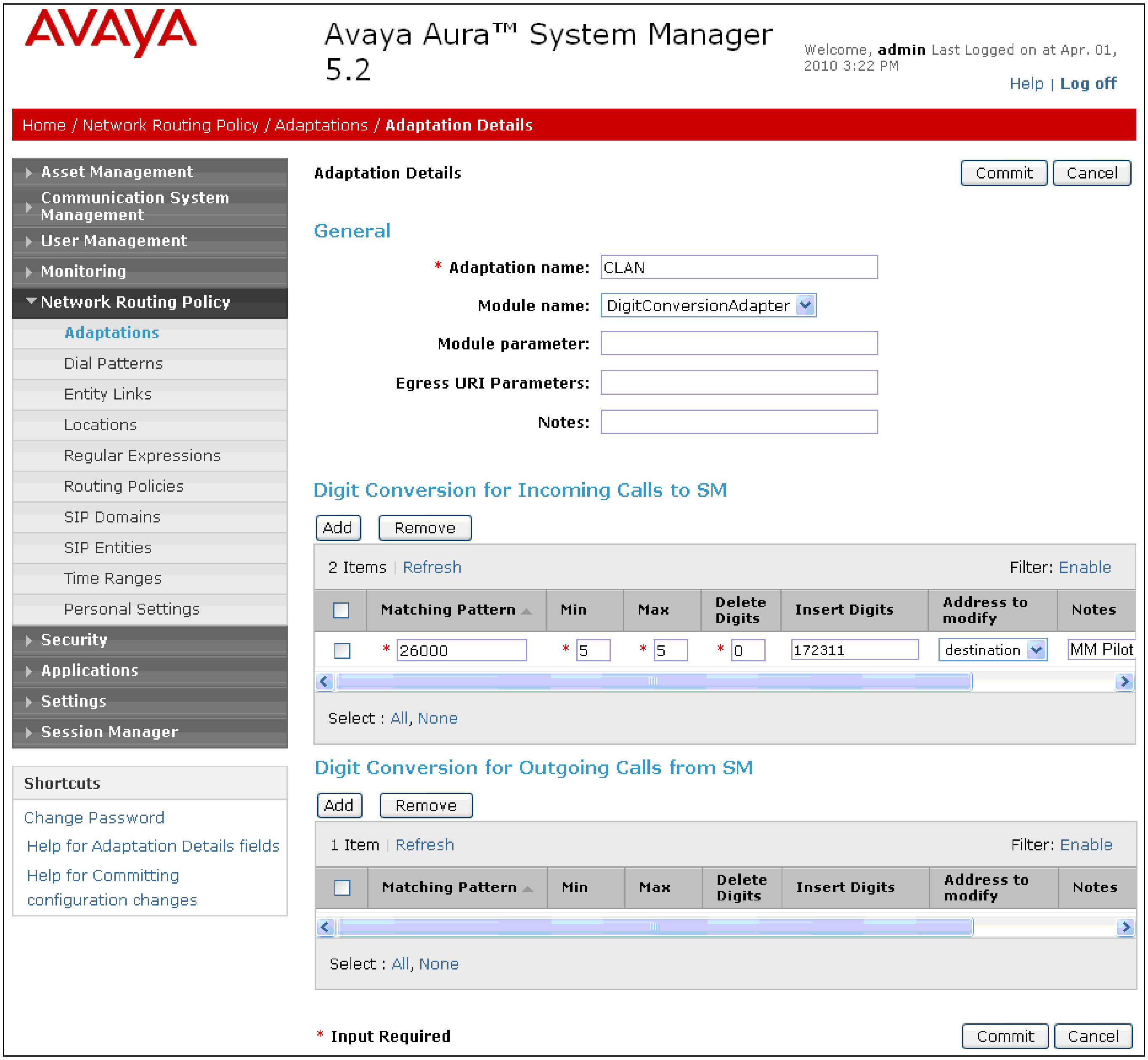

Figure 8: Adaptation Details Page – Adaptation for Avaya Aura™ Communication Manager

4.5.3. Adaptation for Avaya Modular Messaging

The Adaptation administered in this section (DigitConversionAdapter) is used for digit conversion on SIP NOTIFY messages (MWI) from Modular Messaging. On ingress SIP NOTIFY messages from Modular Messaging to Communication Manager where the Request-URI contains the Modular Messaging mailbox number (e.g.17231126101), the Adaptation converts the mailbox number to the associated Communication Manager 5 digit extension (e.g. **26101**).

1. In the **Adaptations** page (not shown), click on "**New**".
2. In the **Adaptation Details** page, enter a descriptive **Adaptation name** and "**DigitConversionAdapter**" for Adaptation **Module name**.
3. In the **Digit Conversion for Incoming Calls to SM** section, click on "**Add**" to provision an entry for converting the Modular Messaging mailbox numbers (172311xxxxx) to the associated 5-digit Communication Manager Modular Messaging extensions (261xx). Provision the entry as follows:
   - **Matching Pattern –** Enter enough leading digits to uniquely match the mailbox range (e.g. **172311**).
   - **Min** and **Max –** Enter the total number of digits in the extension range (e.g. **11**).
   - **Delete Digits –** If necessary, enter the number of leading digits that need to be deleted from the extension range. In the reference configuration this field was set to **6**.
   - **Insert Digits –** Leave blank
   - **Address to modify –** Select "**destination**".
4. Repeat **Steps 1-3** to add any additional digit conversions as necessary.
5. Click on "**Commit**".

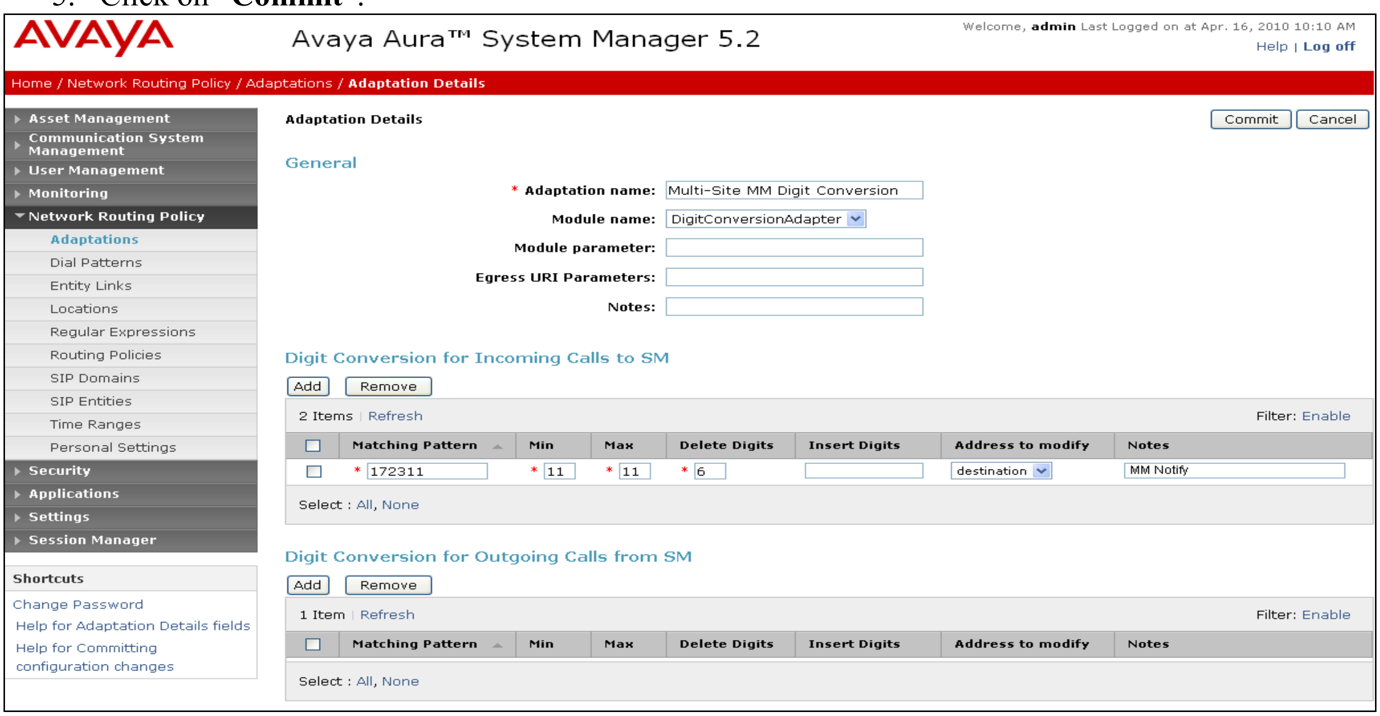

Figure 9: Adaptation for Avaya Aura™ Communication Manager

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

Note – In the reference configuration TCP (port 5060) is used as the transport protocol between Session Manager and all the SIP Entities including Communication Manager. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS (port 5061) to be used as transport protocol between Communication Manager and Session Manager in customer environments. Therefore both TCP and TLS transport were provisioned for the Session Manager SIP Entity (**Section 4.6.1**), although only TCP was used in the reference configuration.

4.6.1. Avaya Aura™ Session Manager SIP Entity

Note – As described above, TCP (port 5060) is used as the transport protocol between Session Manager and all the SIP Entities including Communication Manager. However, since Avaya best practices call for TLS (port 5061) to be used as transport protocol between Communication Manager and Session Manager, both TCP and TLS transport were provisioned for the Session Manager SIP Entity (**Section 4.6.1**), although only TCP was used in the reference configuration.

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 Session Manager.
   - **FQDN or IP Address –** Enter the IP address of the SM100 card on Session Manager.
   - **Type –** Select "**Session Manager**".
   - **Location** – Select a Location administered in **Section 4.4**. In the sample configuration, 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 Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
   - **Time Zone** – Select the time zone in which Session Manager resides.
3. In the **Port** section of the **SIP Entity Details** page, click on "**Add**" and provision an entry as follows:
   - **Port –** Enter "**5060**".
   - **Protocol –** Select "**TCP**".
   - **Default Domain –** (Optional) Select a SIP domain administered in **Section 4.3**.

   This entry enables Session Manager to accept SIP requests on TCP port 5060 (see note above). In addition, Session Manager will associate SIP requests received on this port that contain the IP address of the SM100 card on Session Manager in the host part of the

Request-URI with the selected SIP **Default Domain**. Only **Type: Session Manager** SIP Entities have the **Port** fields.

4. Repeat **Step 3** to provision another similar entry, except with "**5061**" for **Port** and "**TLS**" for **Protocol**.
5. Repeat **Step 3** as necessary to provision entries for other ports on which Session Manager is allowed to accept SIP requests.
6. Click on "**Commit**".

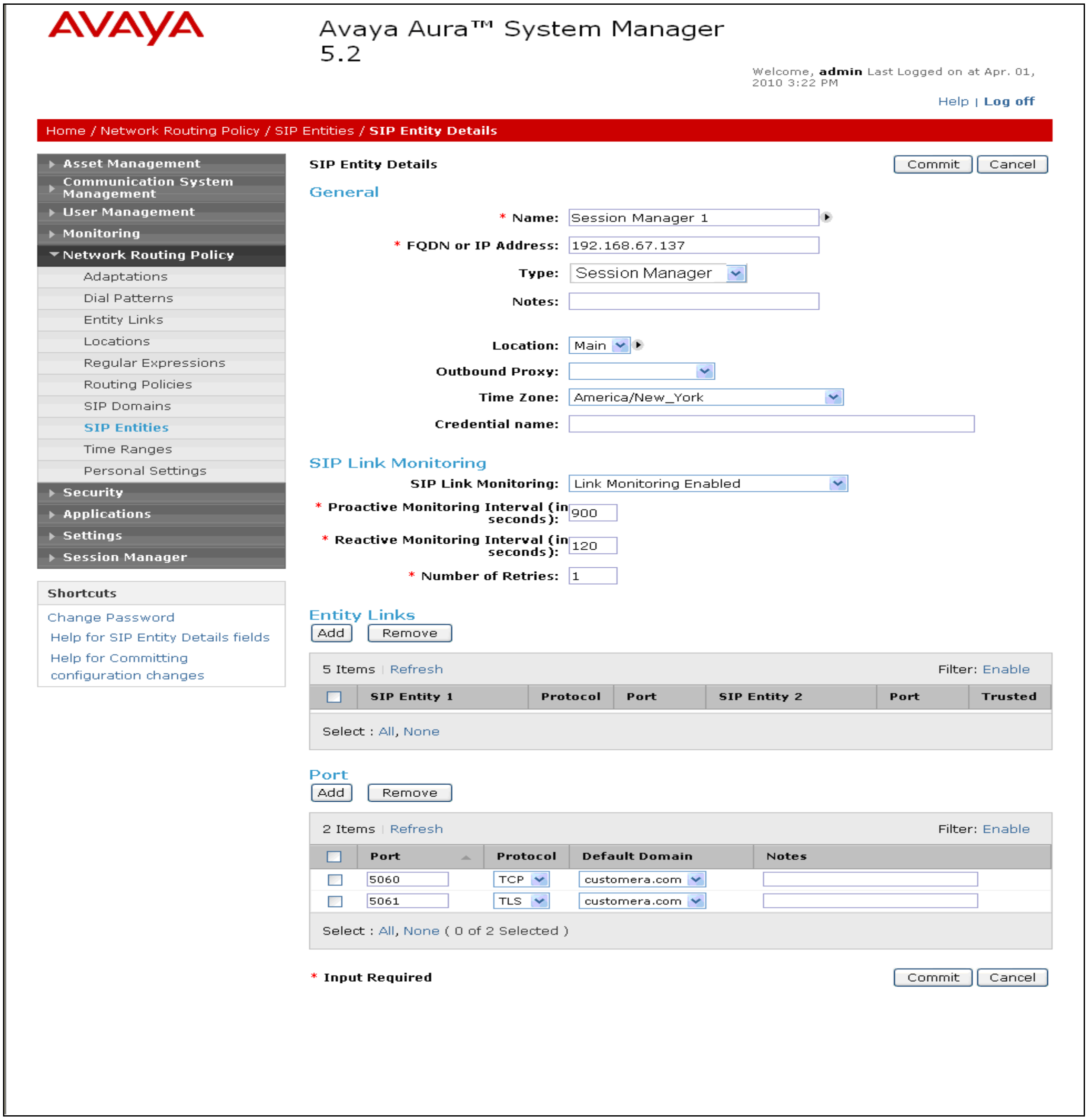

Figure 10: SIP Entity Details Page – Avaya Aura™ Session Manager SIP Entity

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 Communication Manager.
    - **FQDN or IP Address –** Enter the IP address of the 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, Communication Manager is assigned to the "Main" Location.
    - **Time Zone** – Select the time zone in which Communication Manager resides.
3. Click on "**Commit**".

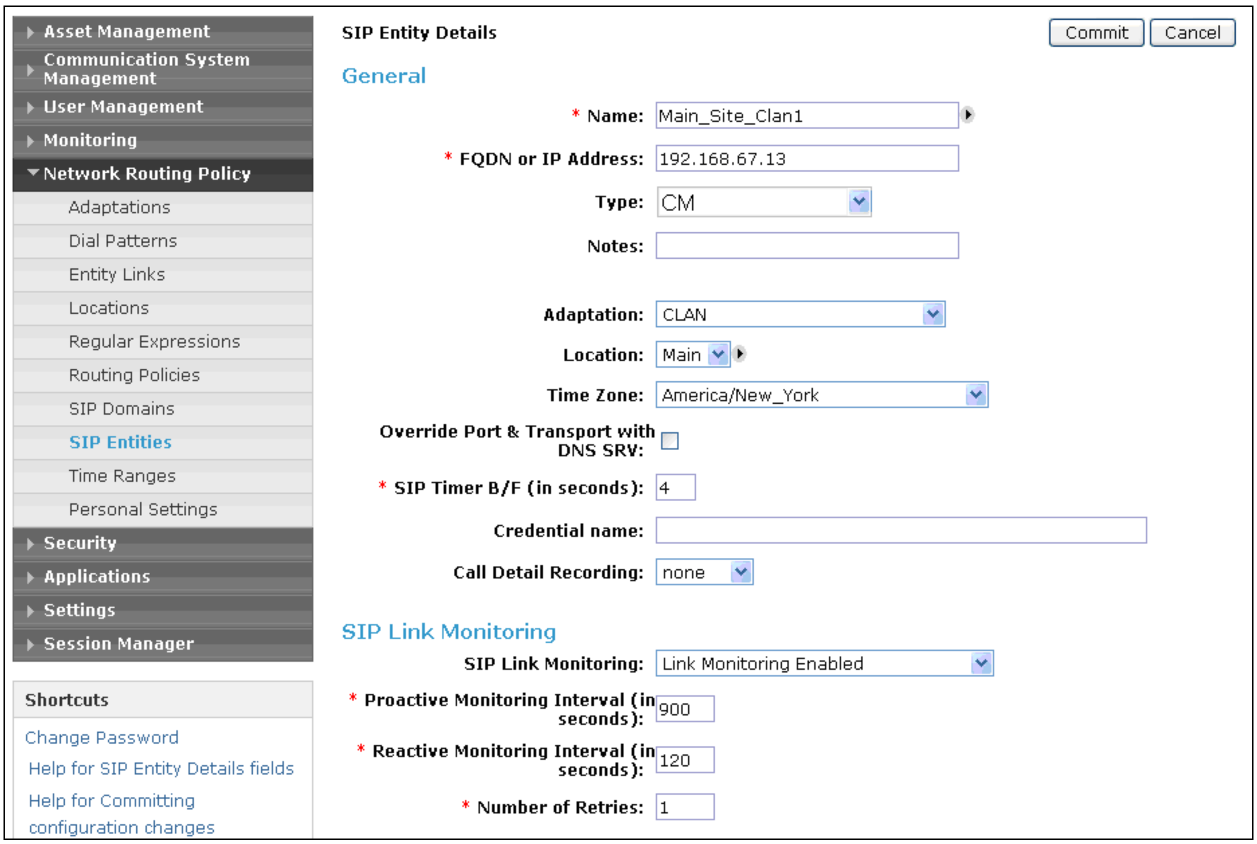

Figure 11: SIP Entity Details Page – Avaya Aura™ Communication Manager SIP Entity

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 "**Other**".
   - **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**".

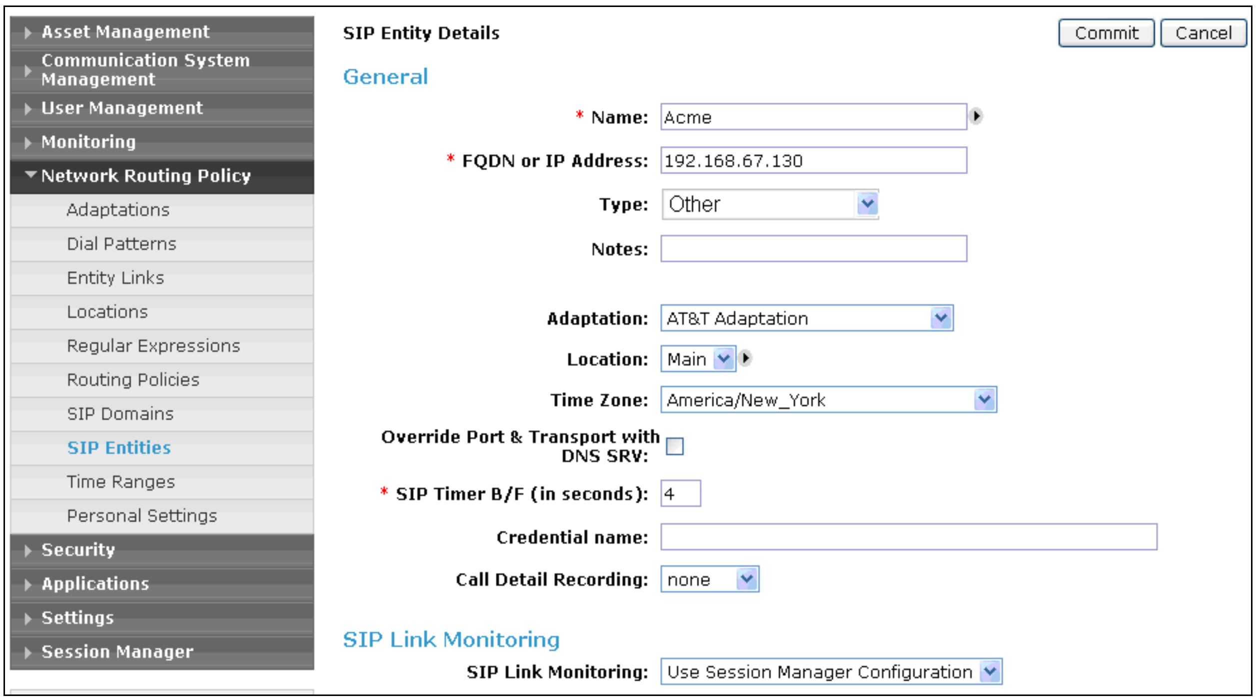

Figure 12: SIP Entity Details Page – Acme Packet SBC SIP Entity

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 Modular Messaging.
   - **FQDN or IP Address –** Enter the IP address of the Modular Messaging Application Server (MAS).
   - **Type –** Select "**Other**".
   - **Adaptation –** Leave blank.
   - **Location** – Select a Location administered in **Section 4.4.** In the sample configuration, Modular Messaging is assigned to the "Main" Location.
   - **Time Zone** – Select the time zone in which Modular Messaging resides.
3. Click on "**Commit**".

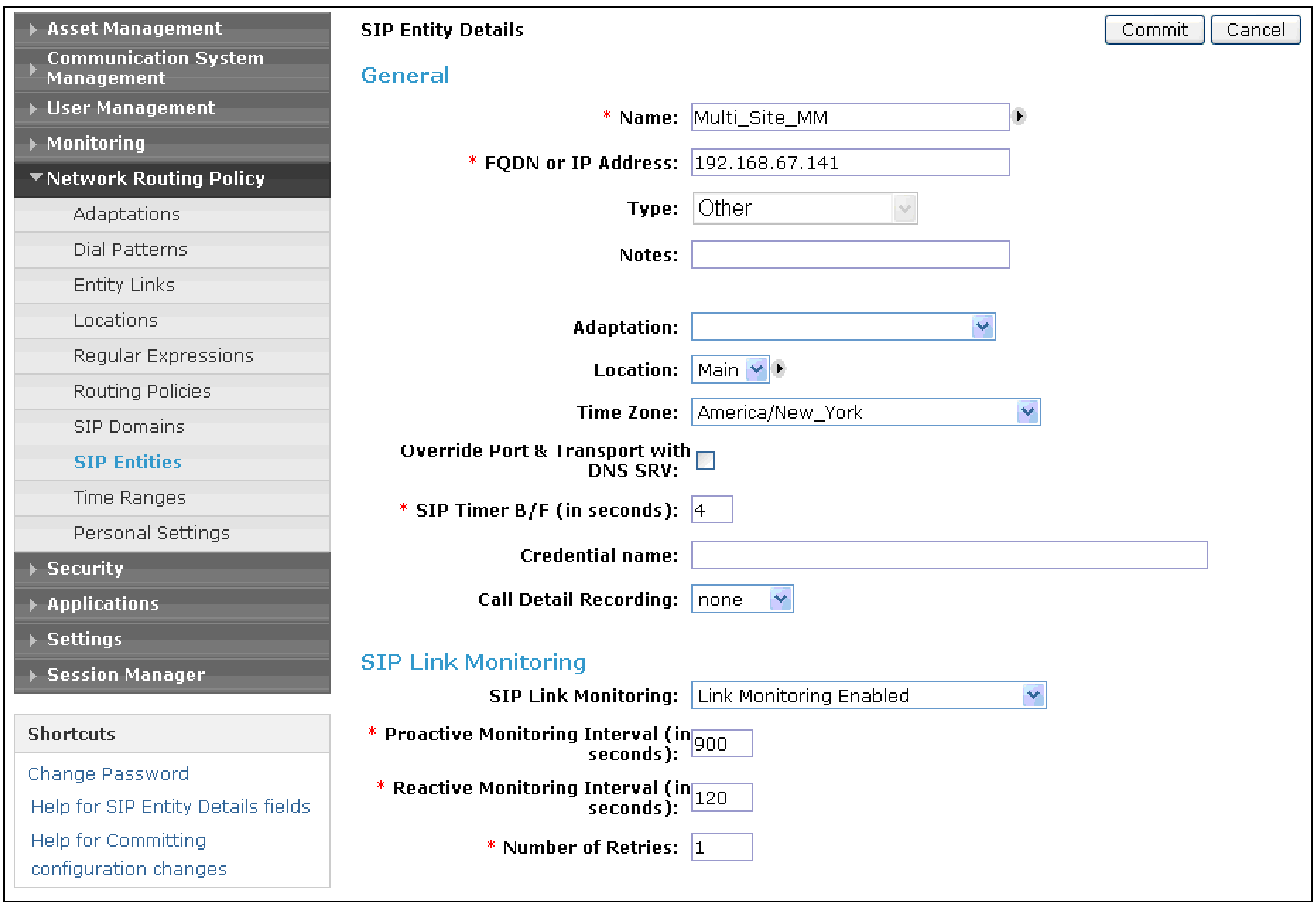

Figure 13: SIP Entity Details Page – Avaya Modular Messaging SIP Entity

4.7. Entity Links

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

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

See the note in Section 4.6 regarding transport protocols and ports used in the reference configuration.

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.
   - **SIP Entity 1** – Select the SIP Entity administered in **Section 4.6.1** for Session Manager. SIP Entity 1 must always be a Session Manager instance.
   - **Protocol –** Select "**TCP**".
   - **SIP Entity 1 Port** – Enter "**5060**".
   - **SIP Entity 2** –Select the SIP Entity administered in **Section 4.6.2** for Communication Manager.
   - **SIP Entity 2 Port** - Enter "**5060**".
   - **Trusted –** Check the checkbox.
3. Click on "**Commit**".

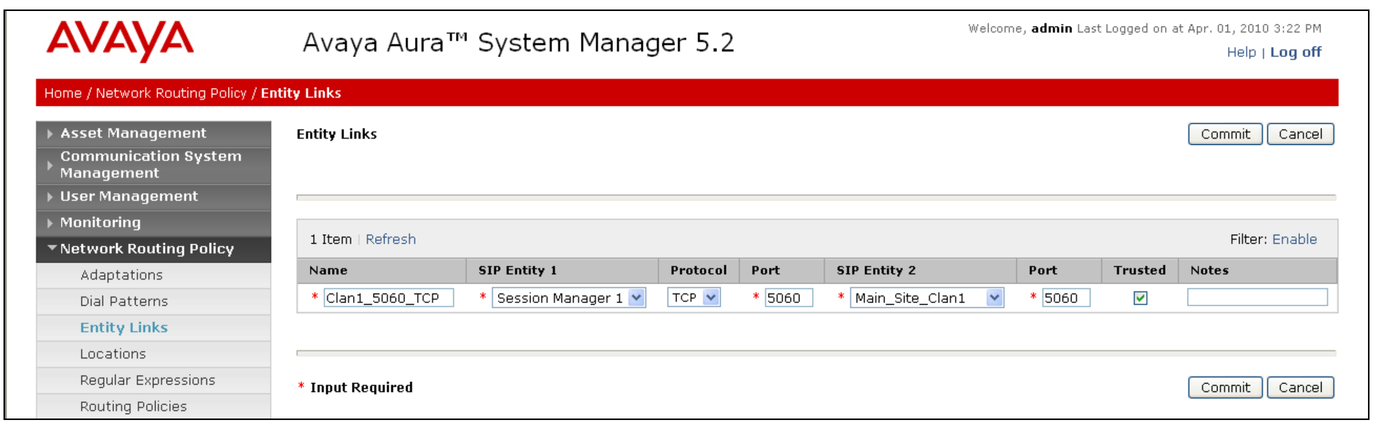

Figure 14: Entity Links Page – Entity Link to Avaya Aura™ Communication Manager

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:

- **SIP Entity 2** – Select the SIP Entity administered in **Section 4.6.3** for the Acme Packet SBC.

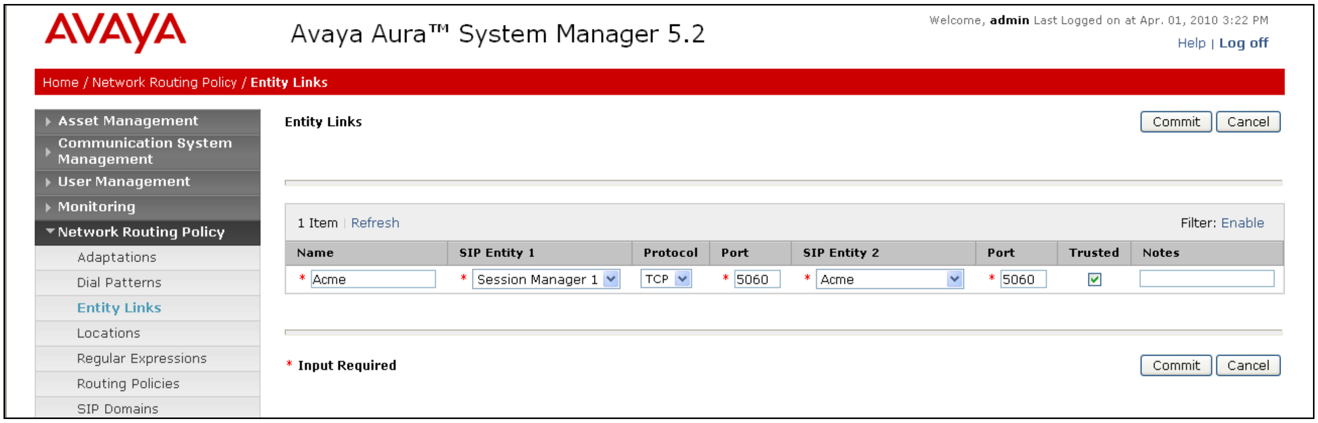

Figure 15: Entity Links Page – Entity Link to AT&T IP Toll Free Service via Acme Packet SBC

4.7.3. Entity Link to Avaya Modular Messaging

Repeat **Section 4.7.1** with the following differences:

- **SIP Entity 2** – Select the SIP Entity administered in **Section 4.6.4** for Modular Messaging.

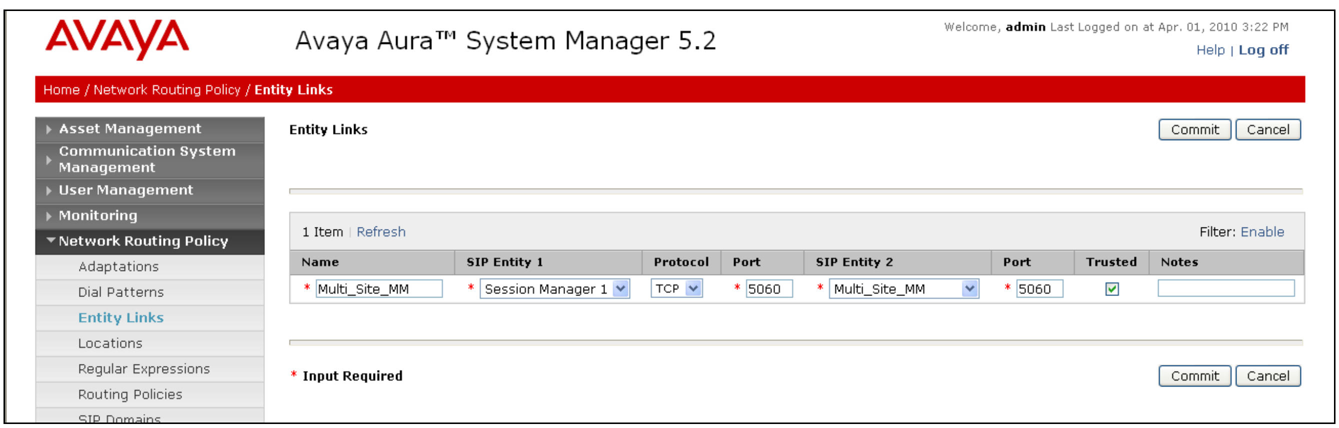

Figure 16: Entity Links Page – Entity Link to Avaya Modular Messaging

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.

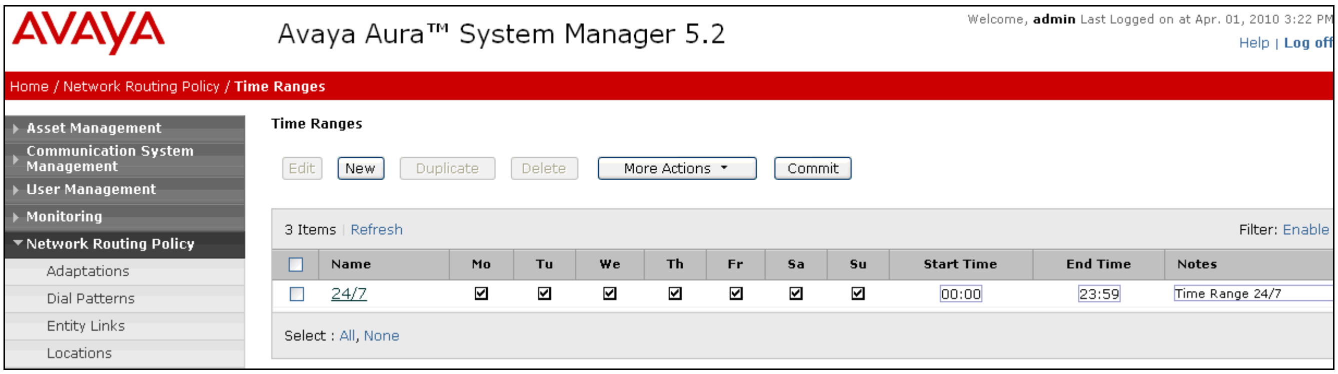

Figure 17: Time Ranges Page

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 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**".

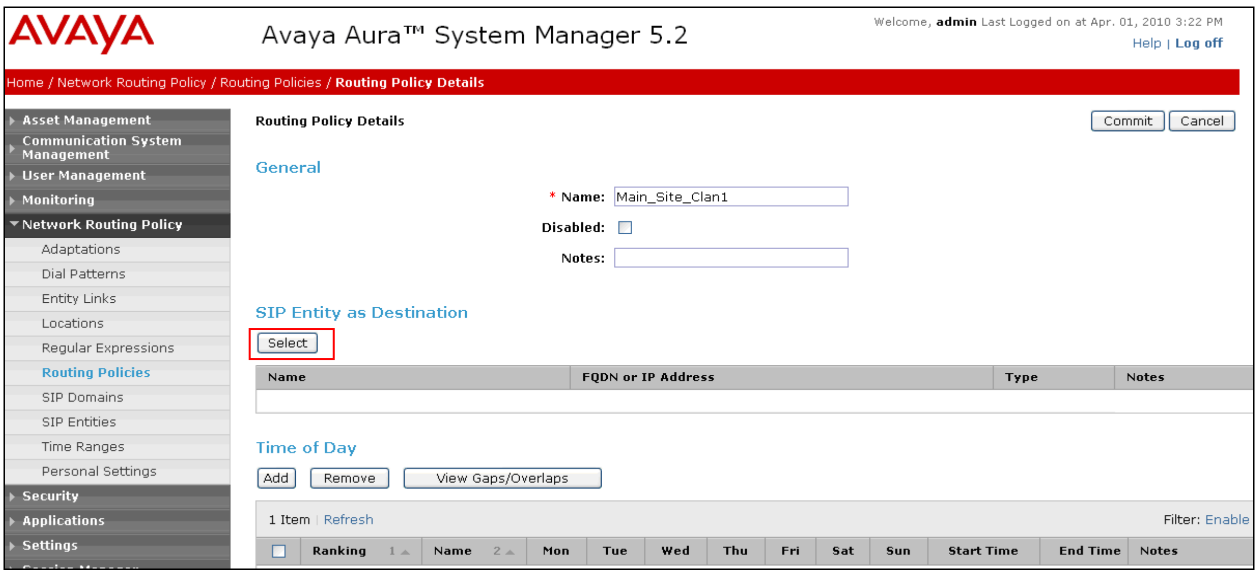

Figure 18: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager

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

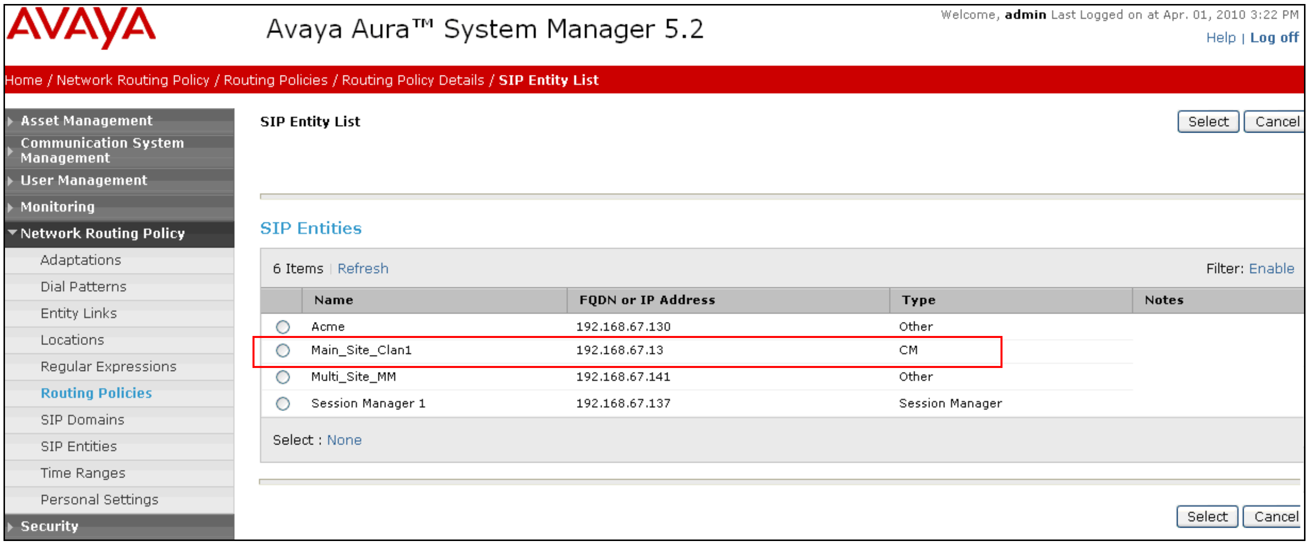

Figure 19: SIP Entity List Page - Routing to Avaya Aura™ Communication Manager

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

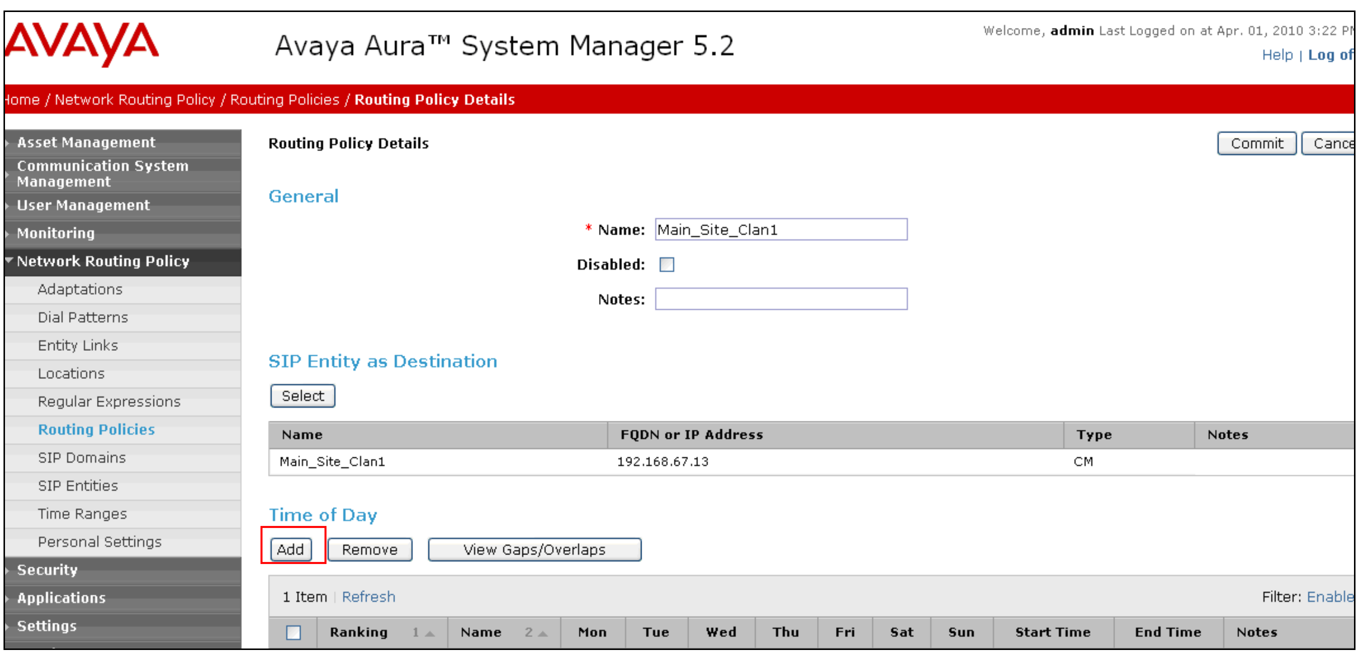

Figure 20: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager (Continued)

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

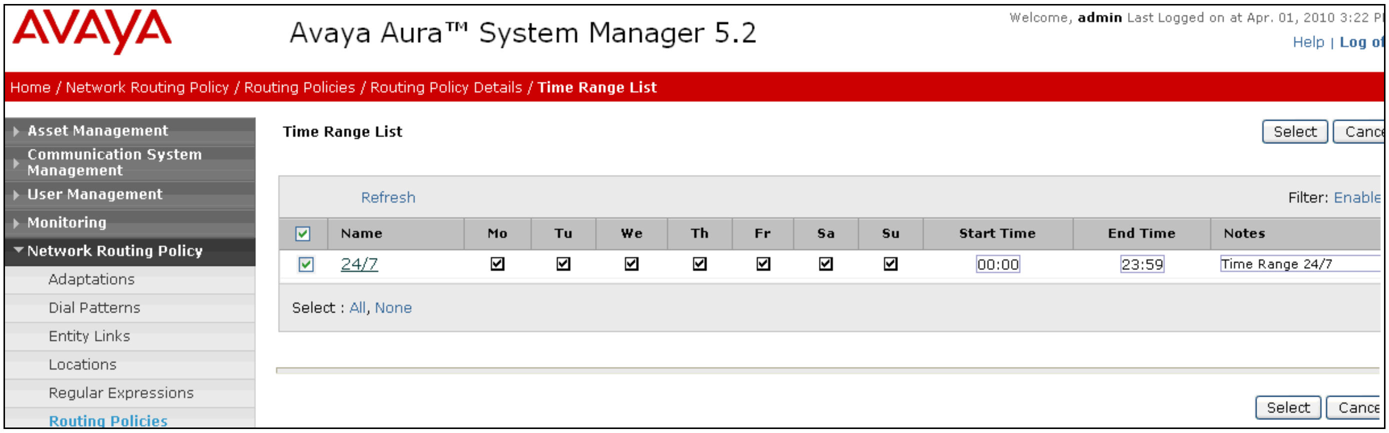

Figure 21: Time Range List Page - Routing to Avaya Aura™ Communication Manager

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**".

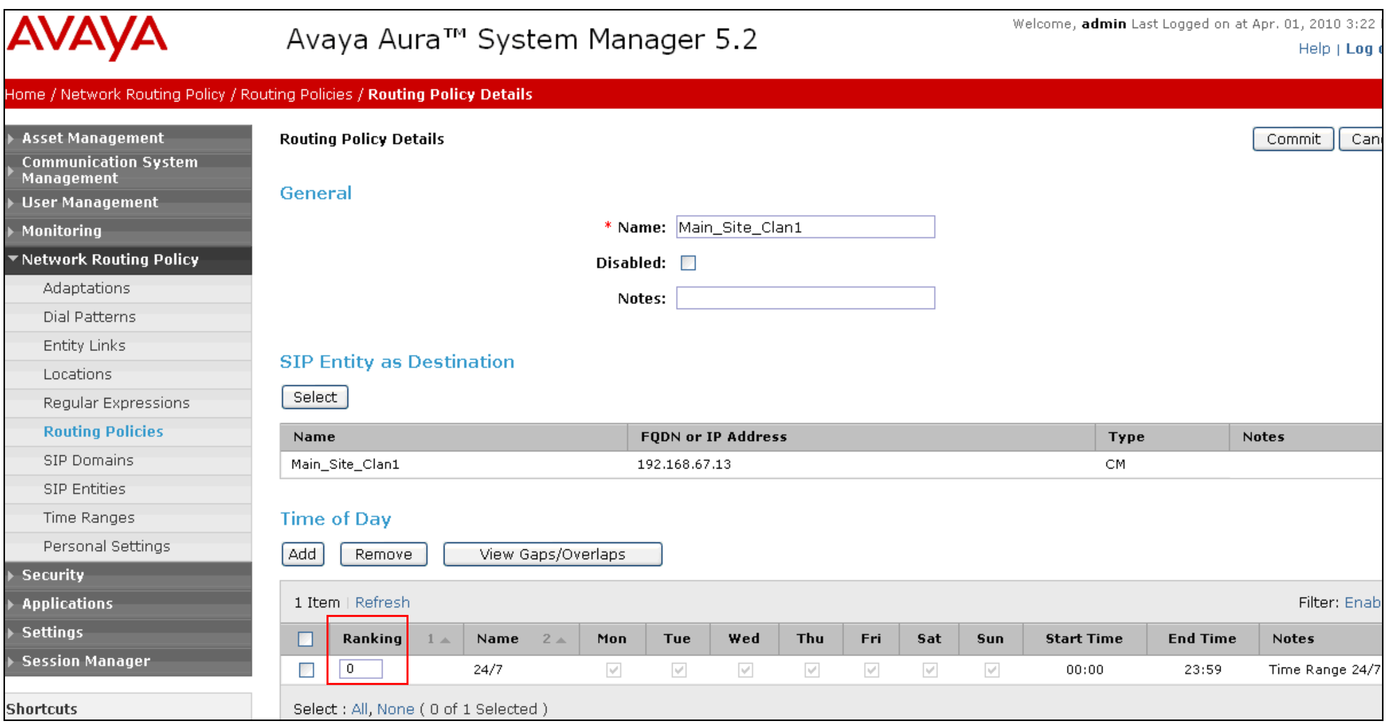

Figure 22: Routing Policy Details Page - Routing to Avaya Aura™ Communication Manager (Final)

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 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 Modular Messaging, and click on "**Select**".

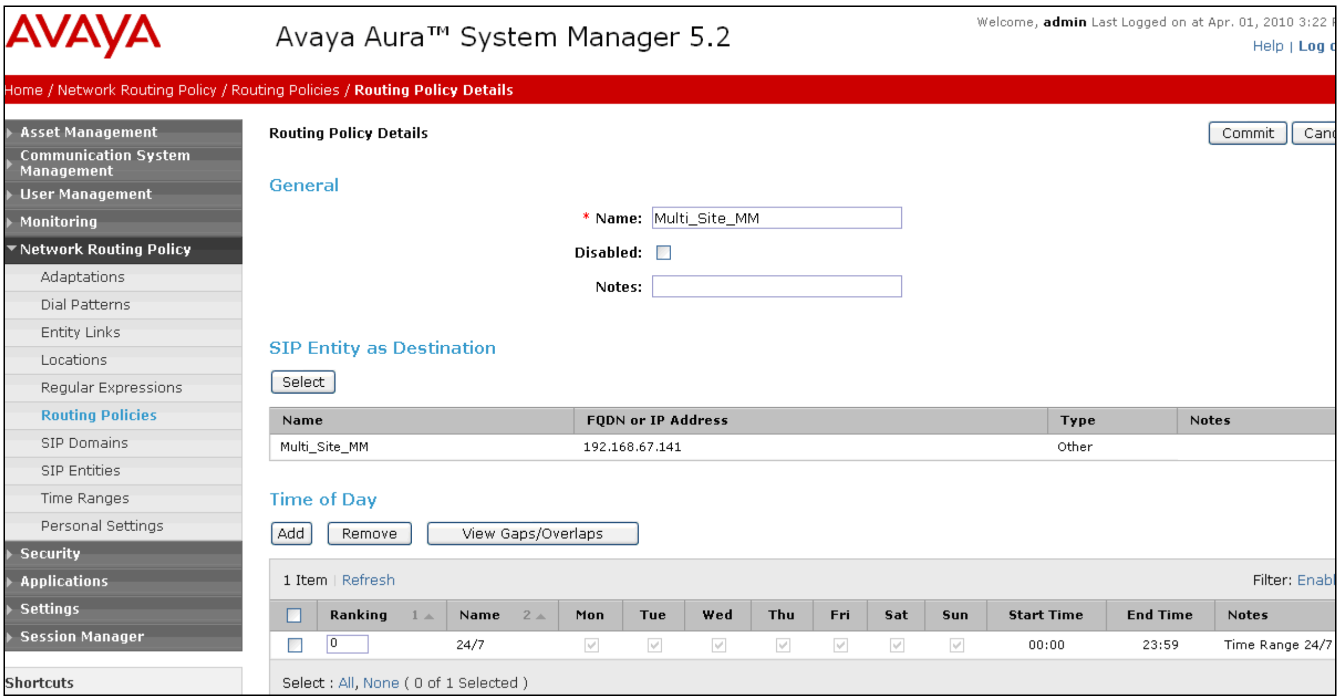

Figure 23: Routing Policy Details Page - Routing to Avaya Modular Messaging

4.10. Dial Patterns

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

- **Section 4.10.1** – Inbound AT&T IP Toll Free calls to numbers associated with stations/agents on Communication Manager (see **Section 4.5.1**).
- **Section 4.10.2** – Communication Manager calls to the Modular Messaging uniform pilot extension (see **Section 4.5.2**).
- **Section 4.10.3** - Modular Messaging NOTIFY (MWI) calls to Communication Manager stations/agents (see **Section 4.5.3**).

4.10.1. Matching Inbound AT&T IP Toll Free Service Calls to Avaya Aura™ Communication Manager Extensions.

After the inbound AT&T IP Toll Free Service called numbers are converted to Communication Manager extensions (described in **Section 4.5.1**), this section routes these calls to Communication Manager.

Note - If the call is for the Modular Messaging Pilot extension (26000), the call will be converted to the Modular Messaging Pilot number (17231126000) in **Section 4.10.2**.

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 the range of Communication Manager extensions (26xxx) created by the AttAdaptation (**Section 4.5.1**).
   - **Min** and **Max –** Enter the total number of digits in the number range (e.g. 5).
   - **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.
   - Add a description in the **Notes** field if desired.
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on "**Add**".

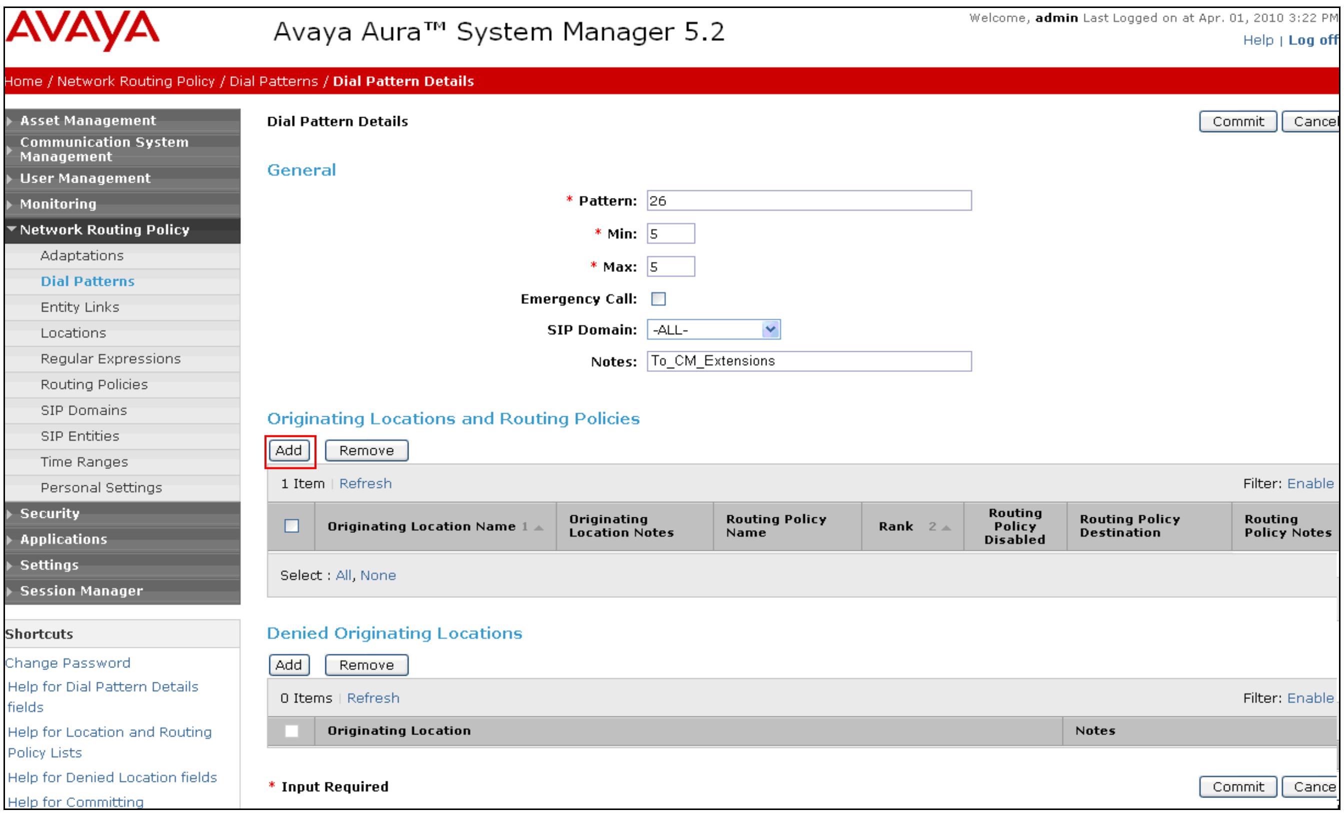

Figure 24: Dial Pattern Details Page - Matching Inbound AT&T IP Toll Free Service Calls to Avaya Aura™ Communication Manager extensions

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**), or select "**-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 Communication Manager in **Section 4.9.1.**
6. In the **Originating Location and Routing Policy List** page, click on "**Select**".

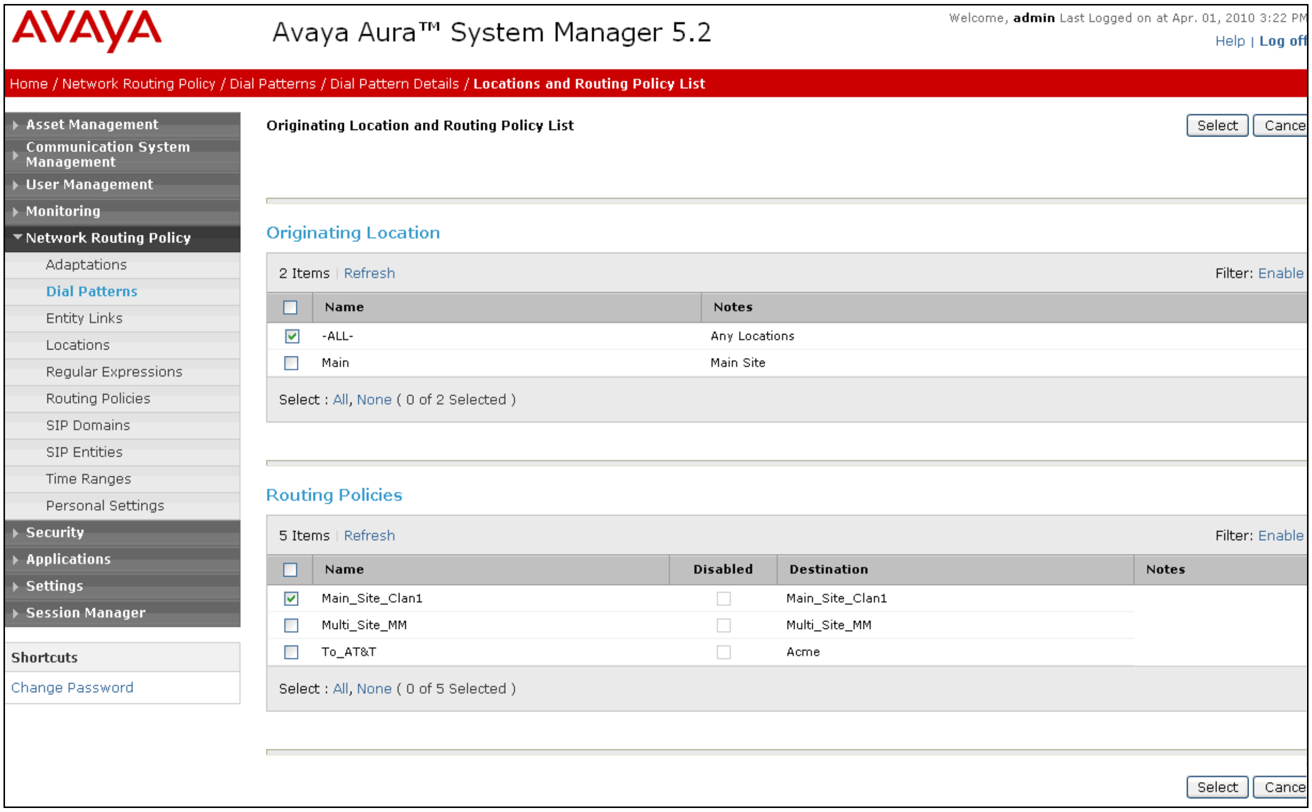

Figure 25: Originating Location and Routing Policy List Page - Matching Inbound AT&T IP Toll Free Service Calls to Avaya Aura™ Communication Manager extensions.

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

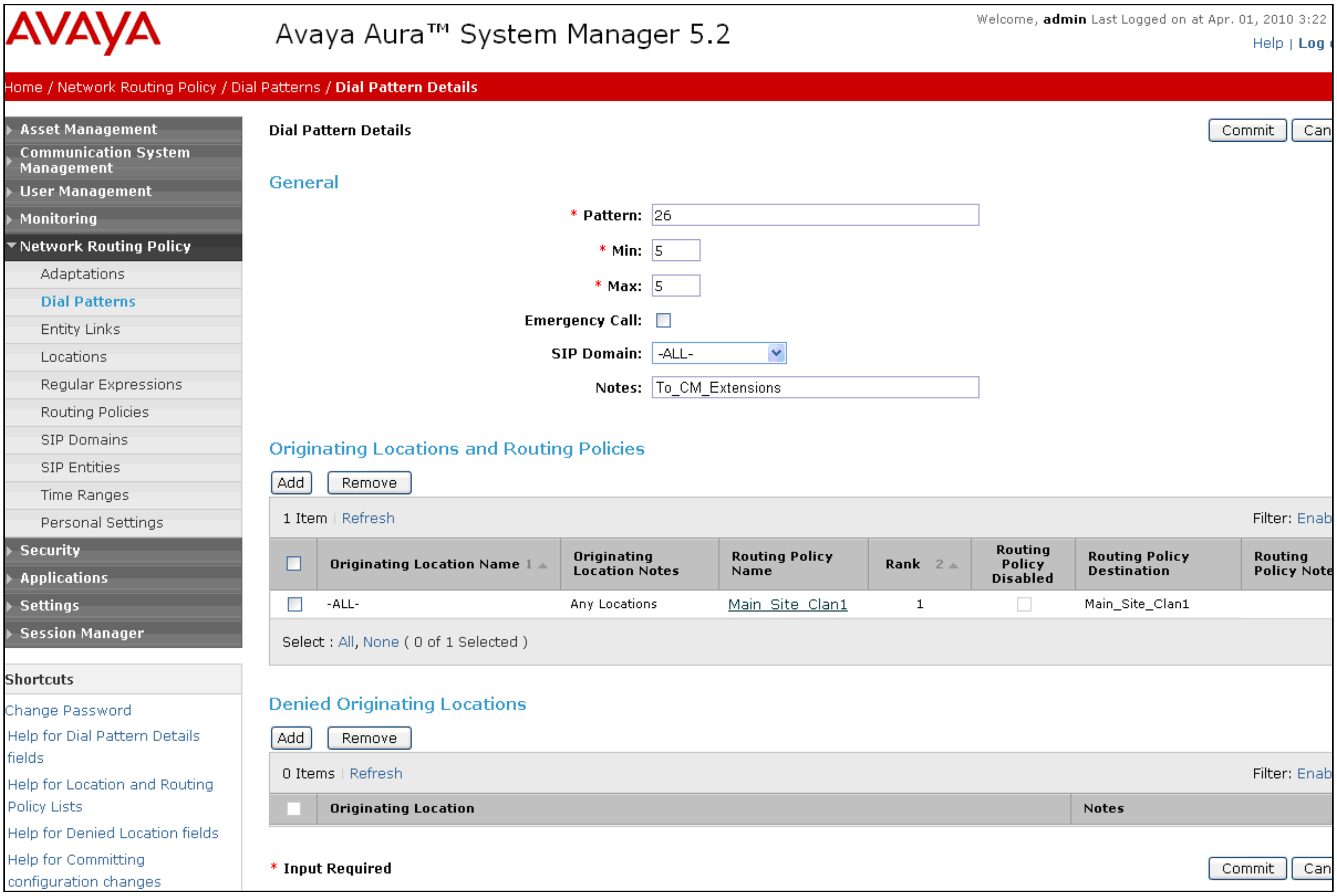

Figure 26: Dial Pattern Details - Matching Inbound Calls to Avaya Aura™ Communication Manager extensions (Final)

4.10.2. Matching Avaya Aura™ Communication Manager Calls to the Avaya Modular Messaging Pilot Number

After the Modular Messaging Pilot extension 26000 is called by a station retrieving messages, it is converted to the Pilot number 17231126000 (described in **Section 4.5.2**), and routes the call to Modular Messaging.

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 Modular Messaging pilot number (e.g. 17231126000).
   - **Min** and **Max** – Enter "**11**".
   - **SIP Domain** – Select "**-ALL-**".
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click on "**Add**".

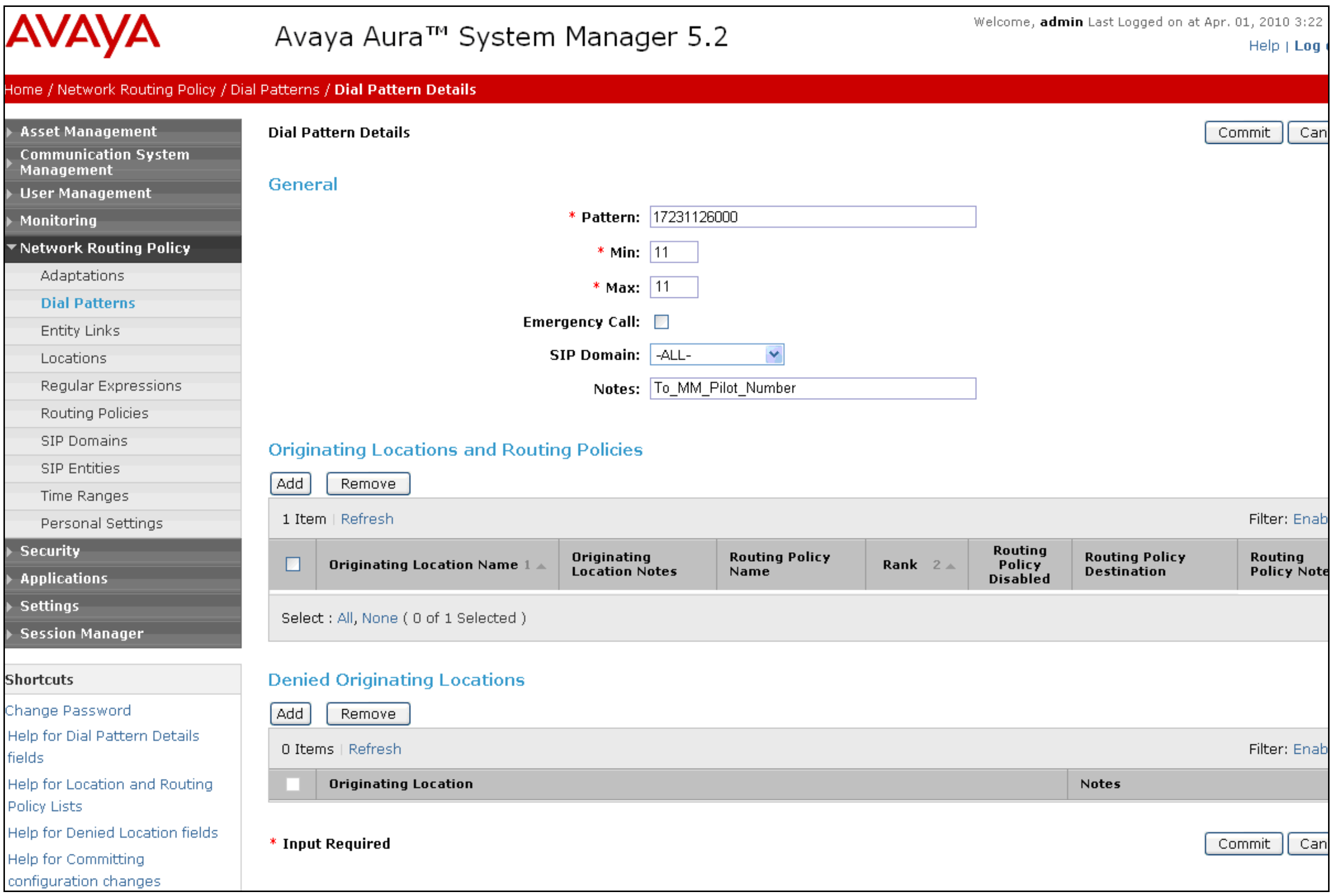

Figure 27: Dial Pattern Details Page - Matching Calls with 11-digit Called Party Numbers Associated with Avaya Modular Messaging Pilot Number.

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 Modular Messaging in **Section 4.9.2**.
6. In the **Originating Location and Routing Policy List** page, click on "**Select**".

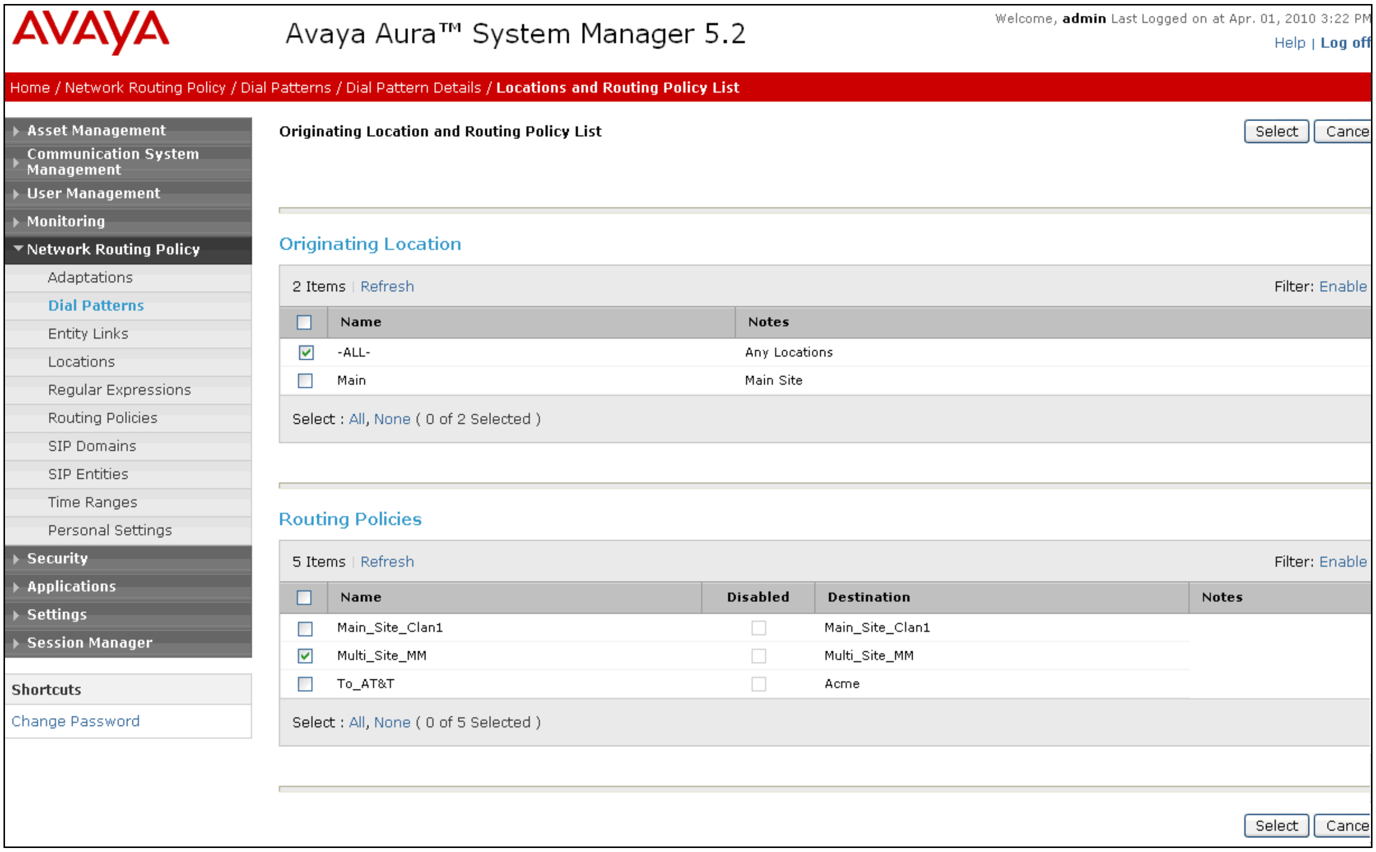

Figure 28: Originating Location and Routing Policy List Page - Matching Calls with 11-digit Called Party Numbers Associated with Avaya Modular Messaging Pilot Number.

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

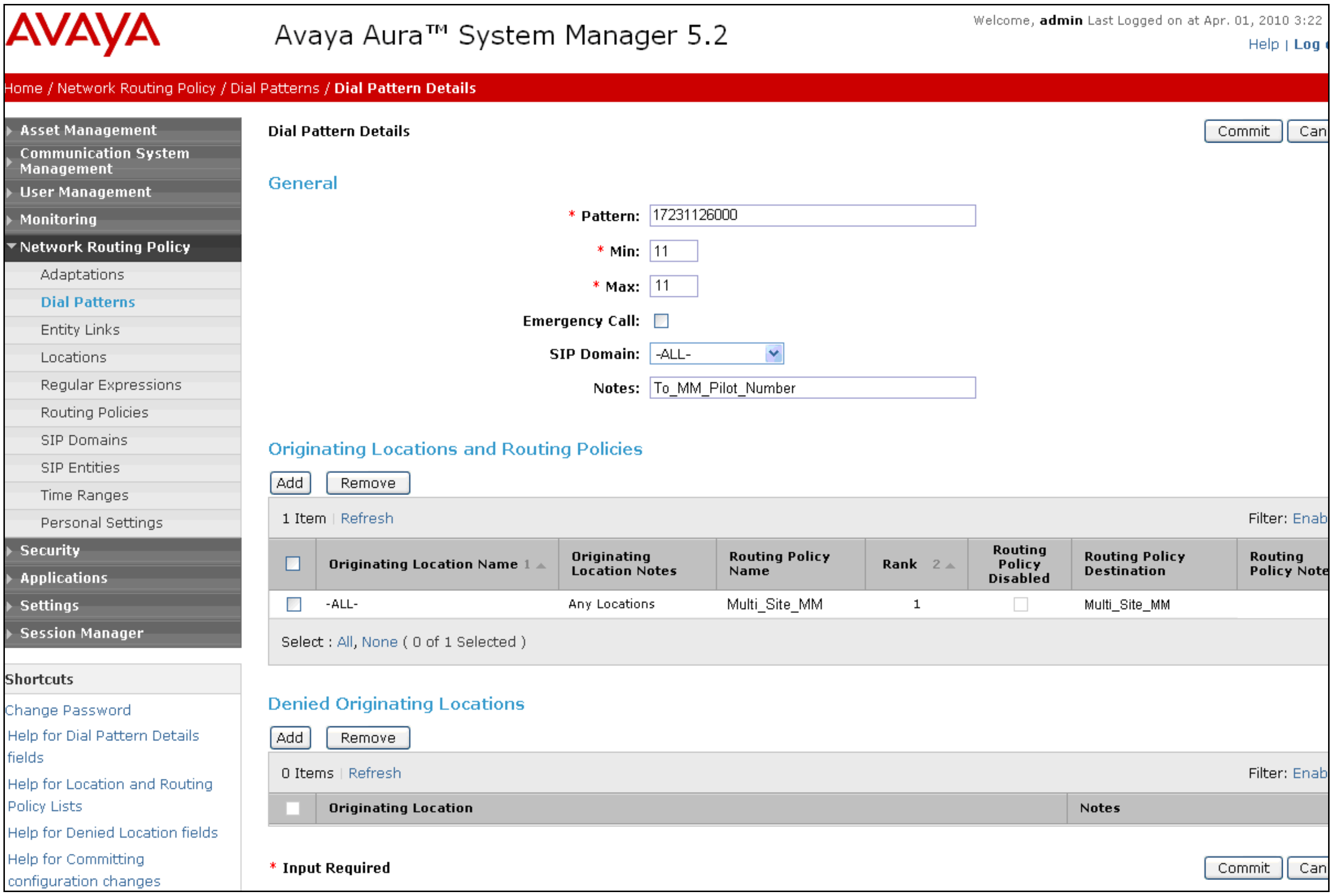

Figure 29: Dial Pattern Details Page - Matching Calls with 11-digit Called Party Numbers Associated with Avaya Modular Messaging Pilot number (Final).

4.10.3. Matching Avaya Modular Messaging NOTIFY (MWI) Calls to Avaya Aura™ Communication Manager Stations/Agents

If a message is left for a Modular Messaging subscriber mailbox, Modular Messaging will send a SIP NOTIFY message to the associated Communication Manager station/agent. The NOTIFY will be sent to the 11 digit mailbox number (e.g. 172311261xx) and the mailbox number is converted to the associated extension 261xx (described in **Section 4.5.3**).

Once the NOTIFY 11 digits are converted to the 5 digit extension, the same Dial Pattern used in **Section 4.10.1** will route the NOTIFY call to Communication Manager. When all completed, the Dial Pattern summary page will appear as follows:

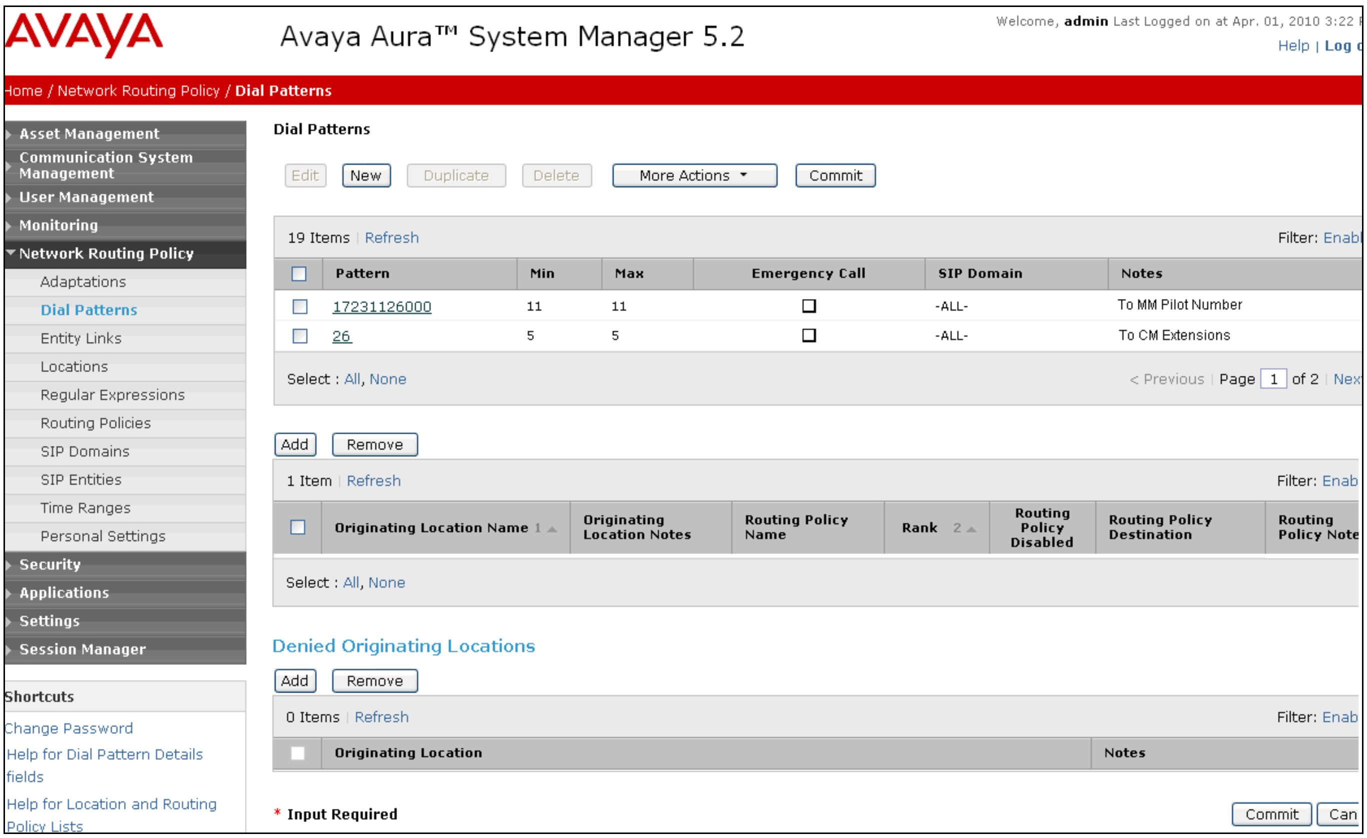

Figure 30: Dial Patterns Page - Final

4.11. Avaya Aura™ 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 Session Manager.
3. In the **Security Module** section of the **Add Session Manager** page, enter the **Network Mask** and **Default Gateway** for the SM100 card.
4. Verify that **Enable Monitoring** is checked under the **Monitoring** header.
5. Click on "**Commit**".

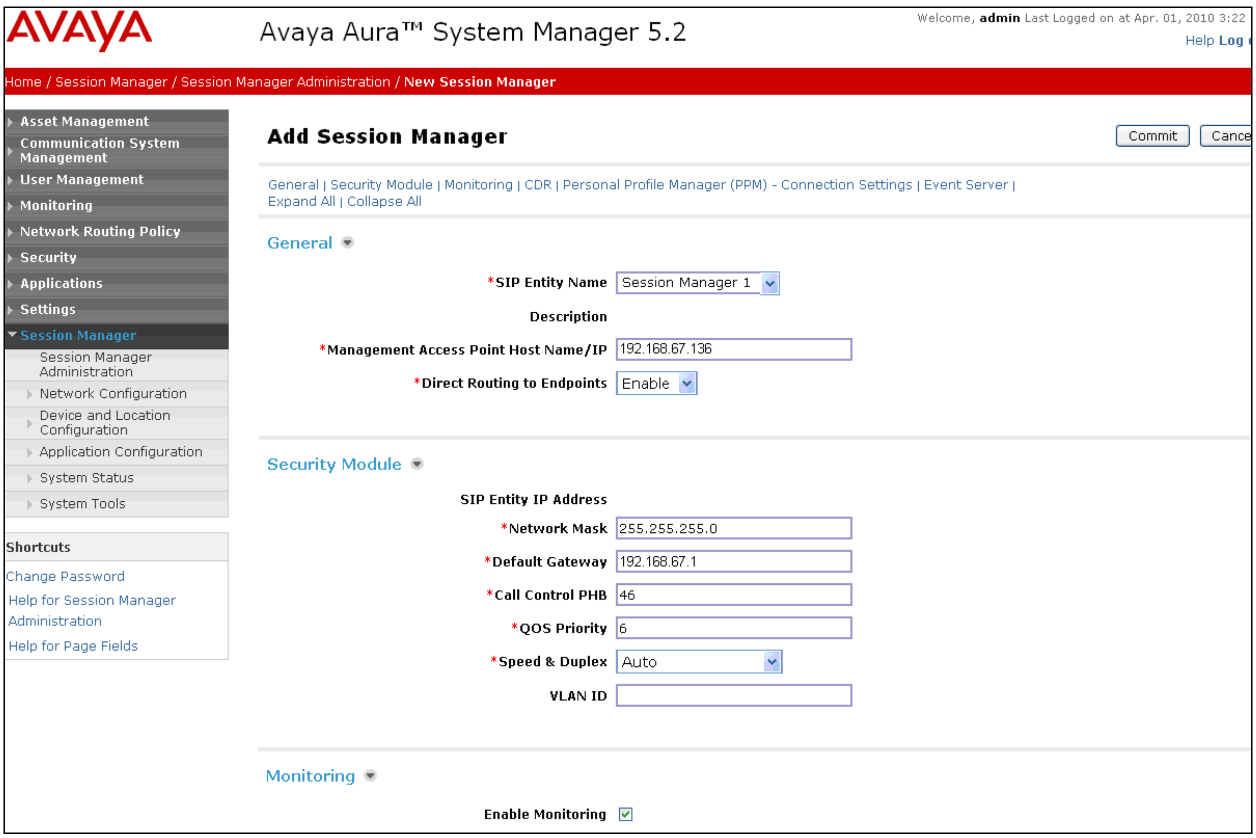

Figure 31: Add Session Manager Page

5. Avaya Aura™ Communication Manager

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

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

5.1. System Parameters

This section reviews the 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: 8000  0
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 5000  50
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 32: System-Parameters Customer-Options Form – Page 2

2. On **Page 4** of the **system-parameters customer-options** form, verify that **IP Trumks** 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? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? y               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? n
     External Device Alarm Admin? n                 Media Encryption Over IP? y
  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)? n
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n               Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? n
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 33: 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 34**:

- 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 digits "**26**" – local extensions for Communication Manager stations, agents, and Vector Directory Numbers (VDNs) 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
     26            5    ext
```

Figure 34: 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 Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within the Avaya site (including Modular Messaging) are assigned to a single IP network region (region 1) and all internal calls use a single IP codec set (codec set 1). 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 x** command, where **x** 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 35.**

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

Figure 35: 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 x** command, where **x** 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 36.**

```
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 36: 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 37: IP-Codec-Set Form for External Calls – Page 2

3. Enter the **change ip-network-region x**, where **x** is the number of an unused IP network region (e.g. **52**). This IP network region will be used to represent the AT&T IP Toll Free service. On **Page 1** of the form:
   - Set the **Codec Set:** to **2**
   - Set the **UDP Port Min:** value to **16384**
   - Set the **UDP Port Max:** value to **32767**

```
change ip-network-region 52                                 Page   1 of  19
                               IP NETWORK REGION
  Region: 52
Location: 1       Authoritative Domain:
    Name: Inbound
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS                  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 38: IP-Network-Region Form the Avaya IP Toll Free Service –Page 1

4. 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 (e.g. region 1) as the **dst rgn**, provision the following:
   - **codec set** – Set to the codec set **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.

```
change ip-network-region 52                                     Page   3 of  19

 Source Region: 52    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
```

Figure 39: IP-Network-Region Form for the AT&T IP Toll Free Service – Page 3

5. Enter the **change ip-network-region nr**, where **nr** is the number of an unused IP network region (e.g. 1). This IP network region will be used for Modular Messaging. On **Page 1** of the form:
   - Set the **Codec Set:** to **1**
   - Set the **UDP Port Min:** value to **16384**
   - Set the **UDP Port Max:** value to **32767**

```
change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain:
    Name: Inbound
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS                  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 40: IP-Network-Region Form for Avaya Modular Messaging – Page 1

6. On **Page 3** of the **ip-network-region** form, for each IP network region administered for local Communication Manager elements within the Avaya site (e.g. region **52**) as the **dst rgn**, provision the following:
   - **codec set** – Set to the codec set to **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.

```
change ip-network-region 1                                        Page   3 of  19

 Source Region: 1    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
 50
 51
 52   2     y    NoLimit                                             n
```

Figure 41: IP-Network-Region Form for Avaya Modular Messaging – Page 3

7. Enter the **change node-names ip** command, and add a node name and the IP address for the 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 Communication Manager elements within the Avaya site as described at the beginning of this section. This C-LAN board will be used in **Section 5.4** for administering a SIP trunk to Session Manager.

```
change node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
Gateway001            192.168.67.1
MainCLAN1A02          192.168.67.13
MainMP1A04            192.168.67.15
MainSM                192.168.67.137
MainVAL1A06           192.168.67.17
default               0.0.0.0
procr                 0.0.0.0



( 7 of 7   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Figure 42: Change Node-Names IP Form

5.4. Inbound Calls

This section describes the steps for administering the SIP trunk with Session Manager.

See the note in Section 4.6 regarding the transport protocols and ports used in the reference configuration.

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group, and provision the following:
   - **Group Type** – Set to "**sip**".
   - **Transport Method** – Set to "**tcp**". .
   - **Near-end Node Name** – Set to the node name of the C-LAN board noted in **Section 5.3**.
   - **Far-end Node Name** – Set to the node name of Avaya Aura™ Session Manager as administered in **Section 5.3**.
   - **Near-end Listen Port** and **Far-end Listen Port** – set to "**5060**".
   - **Far-end Network Region** – Set to the IP network region administered in **Section 5.3** to represent the AT&T IP Toll Free service.
   - **Far-end Domain** – Leave blank.
   - **DTMF over IP** – Set to "**rtp-payload**" to enable Communication Manager to use DTMF according to RFC 2833.
   - **Direct IP-IP Audio Connections** – Set to "**y**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible.

```
add signaling-group 52                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 52              Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n


   Near-end Node Name: MainCLAN1A02          Far-end Node Name: MainSM
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 52
Far-end Domain:

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

Figure 43: 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 52                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip          CDR Reports: y
  Group Name: Inbound                     COR: 1        TN: 1        TAC: 152
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                 Night Service:

Service Type: public-ntwrk          Auth Code? n
                                                       Signaling Group: 52
                                                     Number of Members: 10
```

Figure 44: 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.
   - For each local extension range assigned to Communication Manager phones, agents, skills (hunt groups), and VDNs, provision an entry as follows:
     i. **Ext Len** – Enter **5** (5 digit dial plan)
     ii. **Ext Code** – Enter **26** (for extension 26xxx)
     iii. **Trk Grp(s)** – Enter the number of the trunk group administered in **Step 2**
     iv. **CPN Prefix** – Leave blank
     v. **CPN Len** – Enter the total number of digits in the local extension range

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext           Trk       CPN           CPN
Len Code          Grp(s)    Prefix        Len
5  26            52                      5         Total Administered: 1
                                                   Maximum Entries: 9999
```

Figure 45: Public-Unknown-Numbering Form – Inbound Trunk Digits

5.5. Avaya Modular Messaging Calls

This section describes the steps for administering the SIP trunk to Modular Messaging. Note that the trunk for Modular Messaging calls still terminate at Session Manager. Session Manager passes the calls to Modular Messaging.

1. Enter the **add signaling-group s** command, where **s** is the number of an unused signaling group. Provision is the same as shown in **Section 5.4** except for the following:
   - **Far-end Domain** – Set to the Session Manager SIP Domain specified in **Section 4.3.**
   - **Far-end Network Region** – Set to region 1 as described in **Section 5.3**.

```
add signaling-group 50                                          Page   1 of   1
                                SIGNALING GROUP
 Group Number: 50             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
   Near-end Node Name: MainCLAN1A02          Far-end Node Name: MainSM
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Figure 46: Signaling-Group Form for Avaya Modular Messaging 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 50                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip           CDR Reports: y
  Group Name: MM                          COR: 1        TN: 1        TAC: 150
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                    Night Service:

Service Type: public-ntwrk          Auth Code? n
                                                        Signaling Group: 50
                                                      Number of Members: 10
```

Figure 47: Trunk-Group Form for Modular Messaging Calls – Page 1

3. Enter the **change public-unknown-numbering 0** command to specify the Communication Manager extensions passed in the PAI information for MWI.
   - For each local extension range assigned to Communication Manager that have corresponding mailboxes on Modular Messaging:
     i. **Ext Len** – Enter **5** (5 digit dial plan)
     ii. **Ext Code** – Enter **26** (for extensions 26xxx)
     iii. **Trk Grp(s)** – Enter the number of the trunk group administered in **Step 2**
     iv. **CPN Prefix** – Leave blank
     v. **CPN Len** – Enter the total number of digits in the local extension range

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext            Trk       CPN             CPN
Len Code           Grp(s)    Prefix          Len
5  26              50                        5          Total Administered: 1
                                                        Maximum Entries: 9999
```

Figure 48: Public-Unknown-Numbering Form – Avaya Modular Messaging Trunk digits

5.6. Call Center Provisioning

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

```
display agent-loginID 26666                                    Page   1 of   2
                                 AGENT LOGINID

                Login ID: 26666                                  AAS? n
                    Name: Agent_2_9630                         AUDIX? n
                      TN: 1                            LWC Reception: spe
                     COR: 1                   LWC Log External Calls? n
           Coverage Path: 1                 AUDIX Name for Messaging:
           Security Code:
                                        LoginID for ISDN/SIP Display? n
                                                            Password: 2580
                                              Password (enter again): 2580
                                                         Auto Answer: station
                                                   MIA Across Skills: system
                                           ACW Agent Considered Idle: system
                                           Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :

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

Figure 49: Sample Agent Form – Page 1

```
display agent-loginID 26666                                    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: 2       1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
```

Figure 50: Sample Agent Form – Page 2

```
display hunt-group 2                                           Page   1 of   3
                                  HUNT GROUP

            Group Number: 2                                    ACD? y
              Group Name: Skill2                             Queue? y
         Group Extension: 26002                             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 51: Sample Skill (Hunt Group) Form – Page 1

```
display hunt-group 2                                           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): 2
                                               Redirect to VDN: 20000
                   Forced Entry of Stroke Counts or Call Work Codes? n
```

Figure 52: Sample Skill (Hunt Group) Form – Page 2

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

                      LWC Reception: spe           AUDIX Name:


                     Message Center: none
```

Figure 53: Sample Skill (Hunt Group) Form – Page 3

```
display vector 1002                                            Page   1 of   6
                                  CALL VECTOR

    Number: 1002             Name: Skill2
                                              Meet-me Conf? n            Lock? n
     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? n   Holidays? n
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 26012
03 queue-to     skill 2    pri m
04 wait-time    10  secs hearing music
05 announcement 26015
06 goto step    3              if unconditionally
07 stop
08
09
```

Figure 54: Sample Skill Vector Form – Page 1

```
display vdn 26112                                              Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 26112
                                 Name*: skill2
                           Destination: Vector Number        1002

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

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

* Follows VDN Override Rules
```

Figure 55: Sample Skill VDN Form – Page 3

6. Avaya Modular Messaging

In this sample configuration, Modular Messaging is configured for MultiSite mode. MultiSite mode allows Modular Messaging to server subscribers in multiple locations. The administration for MultiSite mode is beyond the scope of these Application Notes. Consult **[7]** 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 **[8]** 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., Session Manager, Communication Manager, etc., reside to the AT&T IP Toll Free service. See the **session-group** sections for the parameters defined by the **next-hop → sag:** values.

```
local-policy
      from-address
                               *
      to-address
                               *
      source-realm
                               INSIDE
      description
      activate-time            N/A
      deactivate-time          N/A
      state                    enabled
      policy-priority          none
      last-modified-by         admin@console
      last-modified-date       2010-02-15 15:08:38
      policy-attribute
            next-hop                 sag:SP_PROXY
            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 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            2010-02-18 13:21:41
      policy-attribute
            next-hop                      SAG:ENTERPRISE
            realm                         INSIDE
            action                        none
            terminate-recursion           disabled
            carrier
            start-time                    0000
            end-time                      2400
            days-of-week                  U-S
            cost                          0
            app-protocol                  SIP
            state                         enabled
            methods
            media-profiles

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       775880
      max-untrusted-signaling       80
      min-untrusted-signaling       20
      app-signaling-bandwidth       0
      tolerance-window              30
      rtcp-rate-limit               0
      min-media-allocation          2000
      min-trusted-allocation        4000
      deny-allocation               64000
      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             2009-11-04 00:34:23
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             2009-11-04 00:33:51
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             2009-11-04 00:33:51
```

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                     192.168.64.130
      pri-utility-addr               192.168.64.131
      sec-utility-addr               192.168.64.132
      netmask                        255.255.255.0
      gateway                        192.168.64.1
      sec-gateway
      gw-heartbeat
            state                          disabled
            heartbeat                      0
            retry-count                    0
            retry-timeout                  1
            health-score                   0
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                    11
        hip-ip-list                    192.168.64.130
      ftp-address
        icmp-address                   192.168.64.130
      snmp-address
      telnet-address
      last-modified-by               admin@console
      last-modified-date             2009-11-06 13:33:09

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                  192.168.67.130
      pri-utility-addr            192.168.67.131
      sec-utility-addr            192.168.67.132
      netmask                     255.255.255.0
      gateway                     192.168.67.1
      sec-gateway
      gw-heartbeat
            state                       disabled
            heartbeat                   0
            retry-count                 0
            retry-timeout               1
            health-score                0
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                 11
        hip-ip-list                 192.168.67.130
      ftp-address                 192.168.67.130
        icmp-address                192.168.67.130
      snmp-address
      telnet-address
      last-modified-by            admin@console
      last-modified-date          2009-11-04 01:40:53

ntp-config
      server                      135.8.139.1
      last-modified-by            admin@console
```

```
      last-modified-date            2009-11-04 00:27:53
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            2009-11-04 00:24:39
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            2009-11-04 00:29:41
phy-interface
      name                          s1p0
      operation-type                Media
      port                          0
      slot                          1
      virtual-mac                   00:08:25:a0:f3:6e
      admin-state                   disabled
      auto-negotiation              enabled
      duplex-mode                   FULL
      speed                         100
      last-modified-by              admin@console
      last-modified-date            2009-11-04 00:33:23
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            2009-11-04 00:33:23
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             2009-11-04 00:33:51
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             2009-11-04 00:33:51
```

ANNOTATION: The realm configuration “OUTSIDE” below represents the external network on which the AT&T IP Toll Free service resides, and applies the SIP manipulation “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
      out-manipulationid             NAT_IP
      manipulation-string
      class-profile
      average-rate-limit             0
      access-control-trust-level     medium
      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               2010-03-30 17:11:15
```

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

```
realm-config
        identifier                       INSIDE
        description
        addr-prefix                      0.0.0.0
        network-interfaces
                                         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              2010-01-08 13:53:19

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             2009-11-04 00:34:07
```

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

```
session-agent
      hostname                       135.25.29.74
      ip-address                     135.25.29.74
      port                           5060
      state                          enabled
      app-protocol                   SIP
      app-type
      transport-method               UDP
      realm-id                       OUTSIDE
      egress-realm-id
      description                    AT&T_BE
```

```
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=20
ping-interval                  30
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             2009-12-01 14:51:04
```

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

```
session-agent
      hostname                       192.168.67.137
      ip-address                     192.168.67.137
      port                           5060
      state                          enabled
      app-protocol                   SIP
      app-type
      transport-method               StaticTCP
      realm-id                       INSIDE
      egress-realm-id
      description                    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                Proxy
      loose-routing                  enabled
      send-media-session             enabled
      response-map
      ping-method                    OPTIONS;hops=0
      ping-interval                  60
      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             0
        max-register-burst-rate         0
        register-burst-window           0
        last-modified-by                admin@console
        last-modified-date              2010-03-30 15:23:36
```

ANNOTATION: The session-groups define the destinations for the Inside and Outside local-policies (**next-hop → sag:** parameters). The **SP_PROXY** session-group specifies the IP address of the AT&T Toll Free service border element. The ENTERPRISE session-group specifies the IP address of the Session Manager.

```
session-group
        group-name                      SP_PROXY
        description
        state                           enabled
        app-protocol                    SIP
        strategy                        Hunt
        dest
                                        135.25.29.74
        trunk-group
        sag-recursion                   disabled
        stop-sag-recurse                401,407
        last-modified-by                admin@console
        last-modified-date              2009-12-04 20:10:41
session-group
        group-name                      ENTERPRISE
        description
        state                           enabled
        app-protocol                    SIP
        strategy                        Hunt
        dest
                                        192.168.67.137
```

```
      trunk-group
      sag-recursion                    disabled
      stop-sag-recurse                 401,407
      last-modified-by                 admin@console
      last-modified-date               2009-11-05 17:52:47
```

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
      egress-realm-id                  INSIDE
      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
      proxy-sub-events
      last-modified-by                 admin@console
      last-modified-date               2009-11-04 00:34:23
```

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                        192.168.64.130
            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             2009-11-04 00:49:24
```

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

```
sip-interface
        state                          enabled
        realm-id                       INSIDE
        description
        sip-port
                address                        192.168.67.130
                port                           5060
                transport-protocol             TCP
                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              2009-11-04 00:50:10
```

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 headers
      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                        any
                  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                        any
                  comparison-type                       case-sensitive
                  match-value
                  new-value                             $REMOTE_IP
      last-modified-by                    admin@console
      last-modified-date                  2010-01-08 13:41:08
```

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

```
steering-pool
      ip-address                          192.168.64.130
      start-port                          16384
      end-port                            32767
      realm-id                            OUTSIDE
      network-interface
      last-modified-by                    admin@console
      last-modified-date                  2010-01-06 20:34:25

steering-pool
      ip-address                          192.168.67.130
      start-port                          16384
      end-port                            32767
      realm-id                            INSIDE
      network-interface
      last-modified-by                    admin@console
      last-modified-date                  2010-03-26 19:08:37

system-config
```

```
hostname                        acmesbc
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                 135.8.139.1
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              2009-11-04 00:27:17
```

8. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with System Manager, Session Manager, Communication Manager, Avaya phones, fax machines, an Acme Packet SBC, and Modular Messaging.
- A laboratory version of the AT&T IP Toll Free service via MIS/PNT, 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 Communication Manager VDNs, agents, and phones.
- Call and two-way talkpath establishment between callers and 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.
- Communication Manager phones sending DTMF to the AT&T IP Toll Free to invoke AT&T IP Toll Free Legacy Transfer Connect features, and Communication Manager processing the resulting DTMF responses from the AT&T IP Toll Free service.
- Inbound AT&T IP Toll Free service calls to Communication Manager that are directly routed to agents and unanswered can be covered to 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. Call Verification Tests

The following steps may be used to verify the configuration:

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to voicemail.
3. Verify the call routing administration on Session Manager. In the left pane of the 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.
   a. **Figure 52** shows a sample routing test for an inbound call with the called URI information of the Acme Packet SBC "private" interface (192.168.67.130) with the called AT&T IP Toll Free DNIS number 000001041. The calling request URI contains the Acme Packet SBC "public" interface (192.168.64.130) with the calling PSTN number 7326712438.Click on "**Execute**".

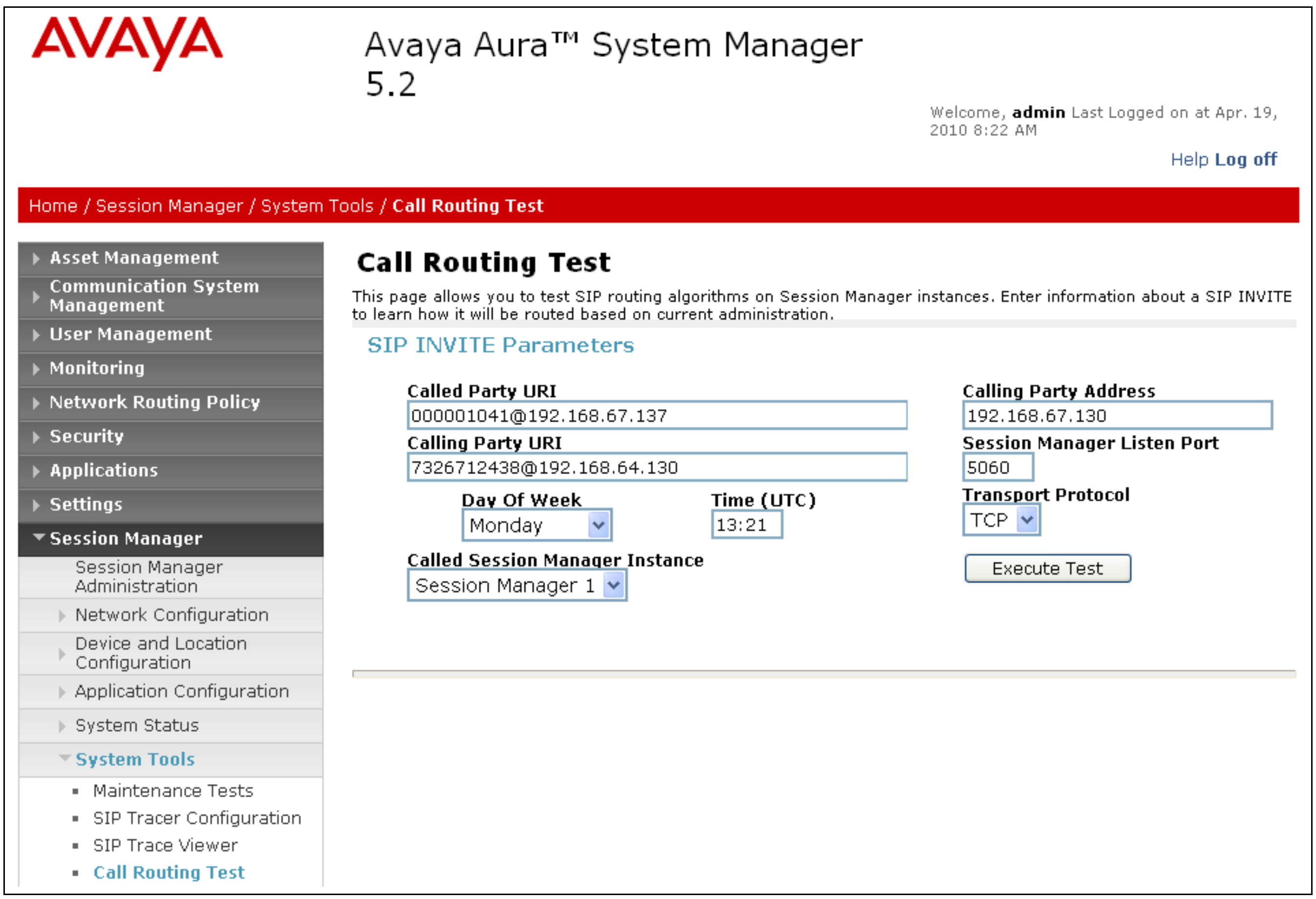

Figure 56: Call Routing Test Page

b. 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 Session Manager in **Section 4**. In the example the final call routing is displayed under the **Routing Decision** heading, with all the routing logic steps listed below under the **Routing Decision Process** heading.

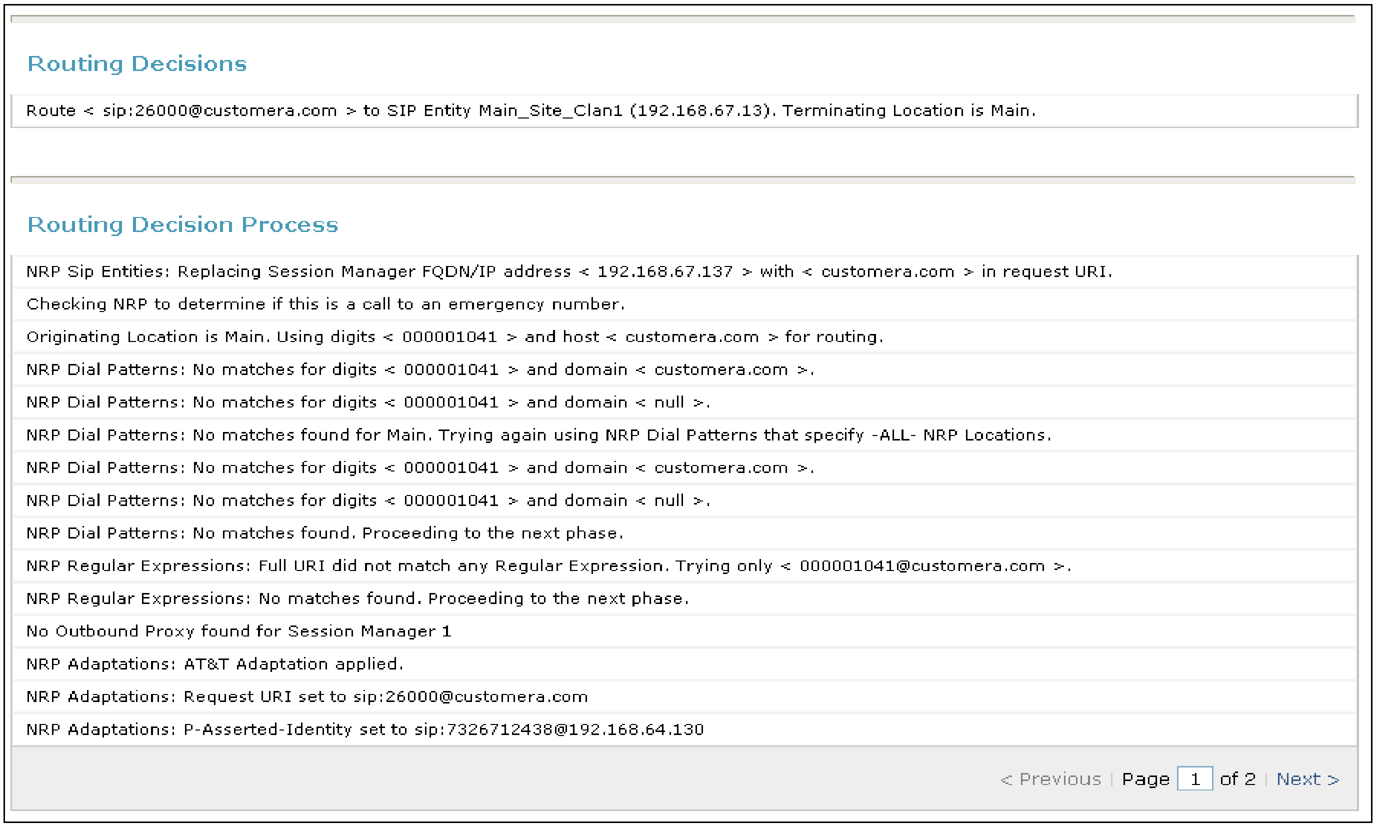

Figure 57: Call Routing Test Page – Test Results – Page 1

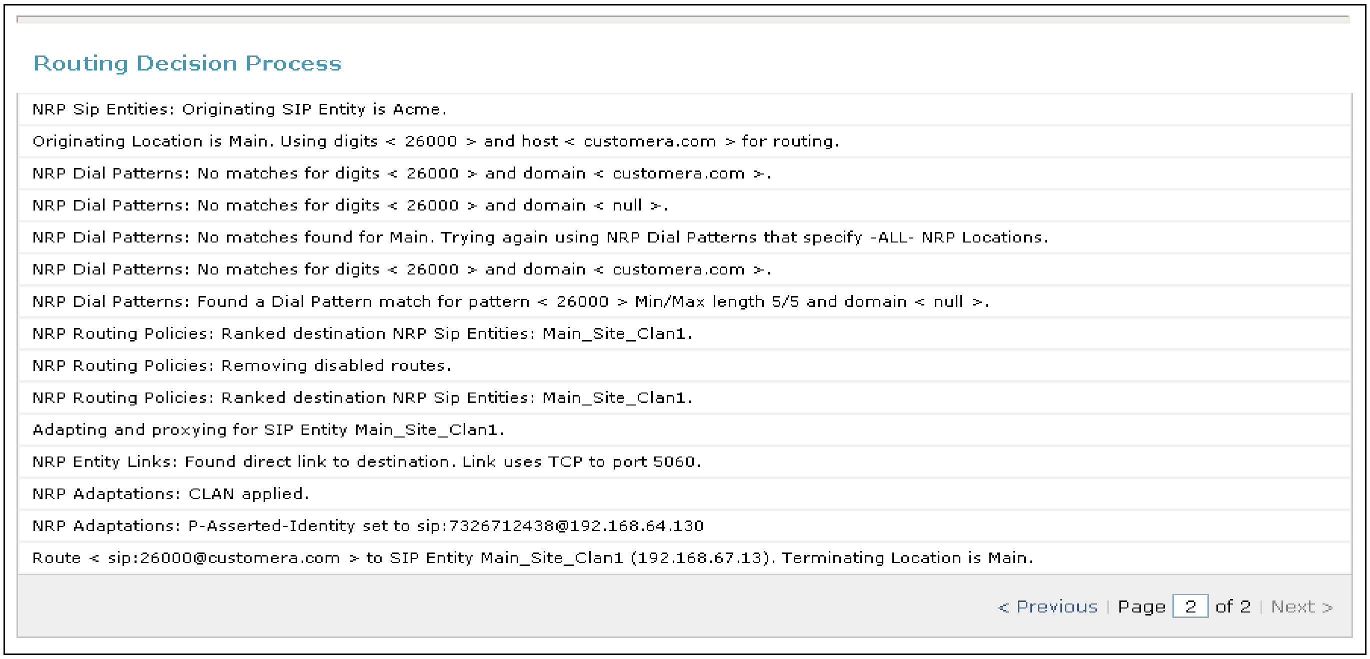

Figure 58: Call Routing Test Page – Test Results – Page 2

9.2. Troubleshooting Tools

The 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 System Manager Common Console may be used to examine the details of 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: 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 2, Release 5.2, November 2009, Document Number 03-603323

[2] *Administering Avaya Aura™ Session Manager*, Issue 2, Release 5.2, November 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 with Avaya Message Storage Server (MSS), Messaging Application Server (MAS) Administration Guide, November* 2009

Acme Packet Support (login required):

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

AT&T IP Toll Free Service Descriptions:

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

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.