

Avaya Solution & Interoperability Test Lab

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

Abstract

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

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya AuraTM Session Manager, Avaya AuraTM 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 AuraTM 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 AuraTM Session Manager and Avaya AuraTM 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 AuraTM Session Manager and then Avaya AuraTM 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 <u>http://support.avaya.com</u>. 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 <u>http://support.avaya.com</u>) to directly access specific support and consultation services based upon their Avaya support agreements.

1.3. Known Limitations

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

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

2. Reference Configuration

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

- Avaya Aura[™] Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya Aura[™] Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- Avaya AuraTM System Manager provides a common administration interface for centralized management of all Avaya AuraTM Session Manager instances in an enterprise.
- Avaya AuraTM Communication Manager provides the voice communications services for a particular enterprise site. In this sample configuration, Avaya AuraTM Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Avaya Aura[™] Communication Manager. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "office" phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software, as well as Avaya 6400 Series Digital Telephones.
- The Acme Packet Net-Net Session Director (SD) 3800 provides SIP Session Border Controller (SBC) functionality, including address translation and UDP/TCP protocol mediation¹, between the AT&T IP Toll Free service and the enterprise internal network. For brevity, the Acme Packet Net-Net SD 3800 will be referred to as the Acme Packet SBC through the remainder of these Application Notes.
- Avaya Modular Messaging (in MultiSite mode in this sample configuration) provides the corporate voice messaging capabilities for enterprise users.

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

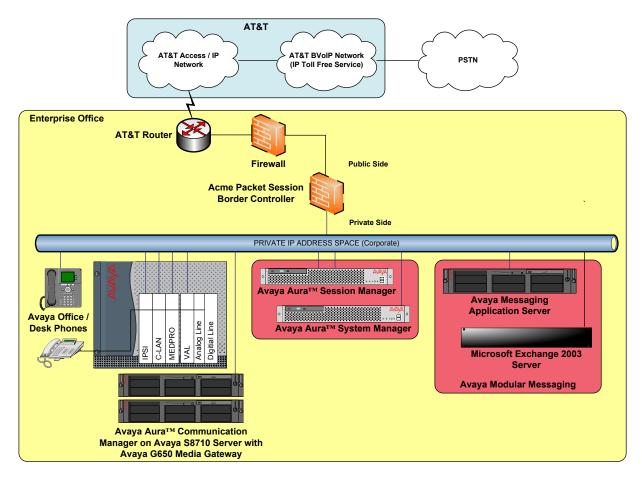


Figure 1: Sample Configuration

2.1. Illustrative Configuration Information

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

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

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

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by Avaya Aura[™] Session Manager and Avaya Aura[™] Communication Manager, two general call flows are described in this section.

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Toll Free service call that arrives on Avaya Aura[™] Session Manager and is subsequently routed to Avaya Aura[™] Communication Manager.

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

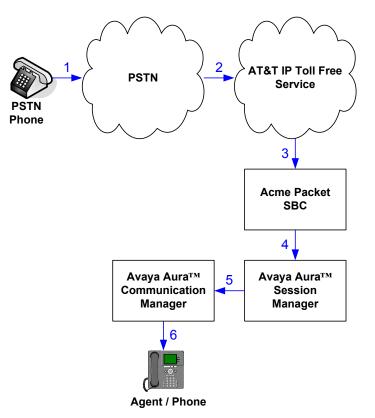


Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Phone

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Solution & Interoperability Test Lab Application Notes ©2009 Avaya Inc. All Rights Reserved. The second call scenario illustrated in **Figure 3** is an inbound call that is covered to voicemail. In this scenario, the voicemail system is an Avaya Modular Messaging system connected to Avaya AuraTM Session Manager. The Avaya Modular Messaging system is in MultiSite mode.

- 1. Same as the first five steps and Step 6b from the first call scenario.
- 2. The called Avaya Aura[™] Communication Manager agent or phone does not answer the call, and the call covers to the agent's or phone's voicemail. Avaya Aura[™] Communication Manager forwards² the call to Avaya Aura[™] Session Manager.
- 3. Avaya AuraTM Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines to where the call should be routed next. In this case, Avaya AuraTM Session Manager routes the call to Avaya Modular Messaging. Avaya Modular Messaging answers the call and connects the caller to the called agent's or phone's voice mailbox. Note that the call³ continues to go through Avaya AuraTM Communication Manager.

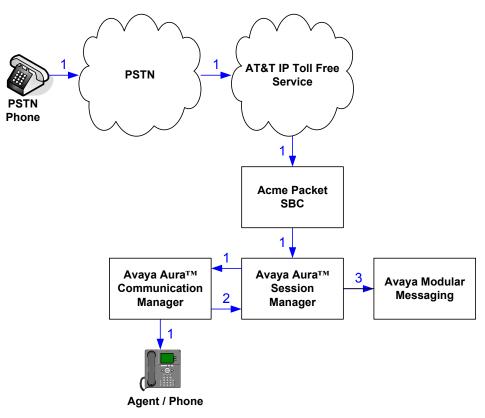


Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Phone Covered to Avaya Modular Messaging

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

³ The SIP signaling path still goes through Avaya AuraTM Communication Manager. In addition, since the inbound call and Avaya Modular Messaging use different codecs (G.729 and G.711, respectively), Avaya AuraTM Communication Manager performs the transcoding, and thus the RTP media path also goes through Avaya AuraTM 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 TM System Manager 1.1 SP1
	(1.1.4.0.111013)
Avaya S8510 Server	Avaya Aura TM Session Manager 1.1 SP1
	(1.1.4.0.111013)
SM100 Card	-
Avaya S8710 Server	Avaya Aura TM Communication Manager
	5.2 with Service Pack 1
	(R015x.02.0.947.3 with update 17294)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW046
TN799DP Control-LAN (C-LAN)	HW01 FW032
TN2302AP IP Media Processor	HW20 FW120
(MedPro)	
TN2602AP IP Media Resource 320	HW02 FW047
(MedPro)	
TN2501AP VAL-ANNOUNCEMENT	HW02 FW021
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	000005
Avaya 9630 IP Telephone	Avaya one-X [™] Deskphone Edition
	H.323 Release 3.0 Service Pack 1
Avaya 9640 IP Telephone	Avaya one-X [™] Deskphone Edition
	H.323 Release 3.0 Service Pack 1
Avaya 4610SW IP Telephone	2.9 SP1 (2.9.1)
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Server	Avaya Modular Messaging 5.1 with
	Patch 8 (9.0.370.18)
Microsoft Exchange 2003 Server on	6.5.7638.1
Microsoft Windows Server 2003 R2	
Enterprise Edition Service Pack 2	
Fax	-
Acme Packet Net-Net Session Director	SCX6.1.0 MR-1 Patch 1 (Build 277)
3800	
AT&T IP Toll Free Service	VNI 14

Table 2: Equipment and Software Versions

4. Avaya Aura™ Session Manager

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

4.1. Background

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

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

4.2. Network Routing Policies

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

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

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AuraTM Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, Avaya AuraTM 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 interworking with specific SIP Entities. For example, an AT&T-specific Adaptation is used in these Application Notes to remove SIP History-Info headers from SIP messages sent to the AT&T IP Toll Free service network. As another example, basic "Digit Conversion" Adaptations are used in this sample configuration to convert digit strings in "destination" and "origination" type headers, e.g., Request-URI and P-Asserted Identity, respectively, of SIP messages sent to and received from SIP Entities.
- Dial Patterns A Dial Pattern specifies a set of criteria and a set of Network Routing Policies for routing calls that match the criteria. The criteria include the called party number and SIP domain in the Request-URI, and the Location from which the call originated. For example, if a call arrives at Avaya AuraTM Session Manager and matches a certain Dial Pattern, then Avaya AuraTM Session Manager selects one⁴ of the Network Routing Policies specified in the Dial Pattern. The selected Network Routing Policy in turn specifies the SIP Entity to which the call is to be routed. Note that Dial Patterns are matched after ingress Adaptations have already been applied.
- Time Ranges Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Network Routing Policy may be associated with one or more Time Ranges during which the Network Routing Policy is in effect. For example, for a Dial Pattern administered with two Network Routing Policies, one Network Routing Policy can be in effect on weekday business hours and the other Network Routing Policy can be in effect on weekday off-hours and weekends.

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

- Use common number formats and uniform numbers in matching called party numbers for routing decisions.
- On ingress to Avaya Aura[™] Session Manager, apply any called party number modifications necessary to "normalize" the number to a common format or uniform number⁵. For example,

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

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

assume that there are three SIP Entities representing three different Avaya AuraTM Communication Manager systems, and a SIP Entity representing a centralized voicemail system, e.g., Avaya Modular Messaging in MultiSite mode. Further, assume that each Avaya AuraTM Session Manager system dials a different pilot extension to call Avaya Modular Messaging. To simplify the routing for such calls, in Avaya AuraTM Session Manager, modify the different called pilot extensions to a uniform pilot number. The uniform pilot number can then be used in routing decisions, thereby minimizing the number of Dial Patterns that need to be administered to match and route calls to Avaya Modular Messaging.

 On egress from SM, apply any called party number modifications necessary to conform to the expectations of the next-hop SIP Entity. For example, on egress from Avaya AuraTM Session Manager to Avaya AuraTM Communication Manager, modify the called party number such that the number is consistent with the dial plan on Avaya AuraTM Communication Manager.

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

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

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

AVAYA

Avaya Aura System Manager 1.0

Welcome, **admin** Last Logged on at Jul. 02, 2009 12:29 PM

Home / Network Routing Policy	
▶ Asset Management	Introduction to Network Routing Policy (NRP)
▶ User Management	Network Routing Policy consists of several NRP applications like "SIP Domains", "Locations", "SIP Entities", etc.
 Monitoring Network Routing Policy 	The recommended order to use the NRP applications (that means the overall NRP workflow) to configure your network configurationi follows:
SIP Domains Adaptations	Step 1: Create "SIP Domains"
Locations	Step 2: Create "Locations"
SIP Entities	Step 3: Create "Adaptations"
Entity Links	Step 4: Create "SIP Entities"
Time Ranges	
Routing Policies Dial Patterns	- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
Regular Expressions	- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
Personal Settings	- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
▶ Security	Step 5: Create the "Entity Links"
▶ Applications	
▶ Settings	- Between Session Managers
▶ Session Manager	- Between Session Managers and "other SIP Entities"
Shortcuts	Step 6: Create "Time Ranges"
Change Password	- Align with the tariff information received from the Service Providers
Landing Page	Step 7: Create "Routing Policies"
Help for Import All Data Help for Export All Data	- Assign the appropriate "Routing Destination" and "Time Of Day"
Help for Committing configuration changes	(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")
	Step 8: Create "Dial Pattern"
	- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Pattern"
	Step 9: Create "Regular Expressions"
	- Assign the appropriate "Routing Policies" to the "Regular Expressions"
	Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Rank
	IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of NRP application "Dial pattern". That's this overall NRP workflow can be interpreted as
	"Dial Pattern driven approach to define routing policies"
	That means (with regard to steps listed above):
	Step 7: "Routing Polices" are defined
	Step 8: "Dial Pattern" are defined and assigned to "Routing Policies" and "Locations" (one step)
	Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

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

4.3. SIP Domains

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

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

AVAYA	Avaya Aura System Manage	r 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 09:35 AM Help Log off
Home / Network Routing Policy / S	IP Domains		
 Asset Management User Management Monitoring 	SIP Domains		Commit Cancel
 Monitoring Network Routing Policy 			
SIP Domains	1 Item Refresh		Filter: Enable
Adaptations	Name	Notes	
Locations	• spdevcon.com	140(63	
SIP Entities	spuevcon.com		
Entity Links			
Time Ranges			
Routing Policies	* Input Required		Commit Cancel
Dial Patterns			
Regular Expressions			
Personal Settings			

Figure 5: SIP Domains Page

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

4.4. Locations

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

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

AVAYA	Avaya Aura System Manager 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 09:35 AM Help Log off
Home / Network Routing Policy / L	.ocations / Location Details	
 Asset Management User Management 	Location Details	Commit Cancel
Monitoring	General	
▼ Network Routing Policy	Name	Notes
SIP Domains	• Main	Main Site
Adaptations		
Locations	Managed Bandwidth: Kbit/sec 💌	
SIP Entities	* Average Bandwidth per Call: 80 Kbit/sec ⊻	
Entity Links	* Time to Live (secs): 3600	
Time Ranges		
Routing Policies	Location Pattern	
Dial Patterns	Add Remove	
Regular Expressions	0 Items Refresh	Filter: Enable
Personal Settings		
▶ Security	IP Address Pattern	Notes
▶ Applications		
Settings	* Input Required	Commit

Figure 6: Location Details Page – Main Site

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

Αναγα	Avaya Aura System Manager 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 09:35 AM Help Log off
Home / Network Routing Policy /	Locations / Location Details	
 Asset Management User Management Monitoring 	Location Details	Commit Cancel
▼Network Routing Policy	Name	Notes
SIP Domains	• Site 1	Site 1
Adaptations		
Locations	Managed Bandwidth: Kbit/sec 👻	
SIP Entities	* Average Bandwidth per Call: 80 Kbit/sec 💌	
Entity Links	* Time to Live (secs): 3600	
Time Ranges		
Routing Policies	Location Pattern	
Dial Patterns	Add Remove	
Regular Expressions	0 Items Refresh	Filter: Enable
Personal Settings		
▶ Security	IP Address Pattern	Notes
▶ Applications		
▶ Settings	* Input Required	Commit

Figure 7: Location Details Page – Site 1

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4.5. Adaptations

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

- Modification⁶ of SIP messages sent to the AT&T IP Toll Free service.
- Modification of digit strings in URIs of "origination" and "destination" type headers in SIP messages sent to and received from Avaya Aura[™] Communication Manager.
- Modification of digit strings in URIs of "origination" and "destination" type headers in SIP messages sent to and received from Avaya Modular Messaging.

4.5.1. Adaptation for AT&T

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

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

AVAVA	Avaya Aura System Manager 1.0 Welcome, admin Last Logged on at Jul. 2 AM						2, 2009 09:35	
								Help Log off
Home / Network Routing Policy / A	daptations / Adaptation Details							
▶ Asset Management	Adaptation Detai	s					Comr	nit Cancel
▶ User Management								
▶ Monitoring	General							
▼ Network Routing Policy	Name	Adaptatio	n Module		Egress URI Pa	rameters	Notes	
SIP Domains	AT&T Adaptation	AttAdapter						
Adaptations								
Locations	Digit Conversion for In	coming Cal	ls to SM					
SIP Entities	Add Remove							
Entity Links								
Time Ranges	0 Items Refresh						Fi	lter: Enable
Routing Policies	Matching Pattern	Min	Мах	Delete Digits	Insert Digits	Address to	modify	Notes
Dial Patterns								
Regular Expressions	Digit Conversion for Ou	itgoing Call	s from S	M				
Personal Settings	Add Remove							
▶ Security								
Applications	0 Items Refresh						Fi	lter: Enable
▶ Settings	Matching Pattern	Min	Мах	Delete Digits	Insert Digits	Address to	modify	Notes
Session Manager								
Shortcuts	* Input Required						Comr	nit Cancel

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

4.5.2. Adaptation for Avaya Aura™ Communication Manager

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

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

- On egress SIP messages to Avaya Aura[™] Communication Manager where the Request-URI contains an AT&T IP Toll Free service number, the Adaptation converts the number to an extension on Avaya Aura[™] Communication Manager.
- On egress SIP messages to Avaya Aura[™] Communication Manager where the Request-URI and/or Message-Account⁷ header contains a 10-digit number associated with an extension on Avaya Aura[™] Communication Manager, the Adaptation converts the number to the extension.
- On ingress SIP messages from Avaya Aura[™] Communication Manager where the P-Asserted-Identity⁸ header contains an extension on Avaya Aura[™] Communication Manager, the Adaptation converts the extension to a 10-digit number.
- On ingress SIP messages from Avaya Aura[™] Communication Manager where the Request-URI contains the Avaya Modular Messaging pilot extension (as dialed by Avaya Aura[™] Communication Manager), the Adaptation converts the pilot extension to a uniform 10-digit pilot number⁹.
 - 1. In the Adaptations page (not shown), click on "New".
 - 2. In the Adaptation Details page, enter a descriptive Name and "DigitConversionAdapter" for Adaptation Module.
 - 3. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to provision an entry for converting a range of AT&T IP Toll Free service numbers to extensions on Avaya Aura[™] Communication Manager. Provision the entry as follows:
 - Matching Pattern Enter enough leading digits to uniquely match the number range, specifically the range of numbers contained in the Request-URI of inbound SIP INVITE messages from the AT&T IP Toll Free service.
 - Min and Max Enter the total number of digits in the number range.
 - Delete Digits and Insert Digits If necessary, enter the number of leading digits that need to be deleted from the number range, and the specific leading digits that need to be prefixed to the number range, respectively, in order to match an extension range on Avaya AuraTM Communication Manager.
 - Address to modify Select "destination".

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

⁷ Present in SIP NOTIFY messages from Avaya Modular Messaging to indicate the voice mailbox number to which the message pertains.

⁸ Typically identifies the connected party number when sent by an answering party (or the calling party number when sent by the calling party).

⁹ With the assumption that the pilot extensions dialed by other SIP Entities, e.g., other Avaya Aura[™] Communication Manager systems connected to this Avaya Aura[™] Session Manager, will also be converted to the same 10-digit pilot number.

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

In the sample configuration, Avaya Modular Messaging sends a 10-digit string in the range 732853xxxx in the Request-URI and Message-Account header. Thus the second entry in the **Digit Conversion for Outgoing Calls from SM** table in **Figure 9** matches the number range 732853xxxx and deletes the leading five digits to match the extension range 3xxxx on Avaya AuraTM Communication Manager.

- Repeat Step 5 as necessary to provision additional entries to cover all expected ranges of 10-digit numbers associated with extensions on Avaya Aura[™] Communication Manager.
- In the Digit Conversion for Incoming Calls to SM section, click on "Add" to provision an entry for converting a range of extensions on Avaya AuraTM Communication Manager to the associated 10-digit numbers. Provision the entry as follows:
 - **Matching Pattern** Enter enough leading digits to uniquely match the extension range.
 - Min and Max Enter the total number of digits in the extension range.
 - **Delete Digits** and **Insert Digits** If necessary, enter the number of leading digits that need to be deleted from the extension range, and the specific leading digits that need to be prefixed to the extension range, respectively, in order to form the 10-digit number range.
 - Address to modify Select "origination".

In the sample configuration, Avaya Aura[™] Communication Manager sends a 5-digit string in the range 2xxxx, 3xxxx, or 5xxxx in the P-Asserted-Identity header. Thus the first, second, and fourth entries in the **Digit Conversion for Incoming Calls to SM** table in **Figure 9** match the number ranges 2xxxx, 3xxxx, and 5xxxx, respectively, and prefixes "73285" to those number ranges to form 10-digit numbers.

8. Repeat Step 7 as necessary to provision additional entries to cover all extension ranges on Avaya Aura[™] Communication Manager.

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

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

10. Click on "Commit".

Αναγα	Avaya Aura System Manager 1.0 Help Log					: Logged on at Aug. 10, 2009 09:52 Help Log off
Home / Network Routing Policy / Ad	aptations / Adaptation Details					
 Asset Management User Management 	Adaptation Detail	s				Commit
▶ Monitoring	General					
Network Routing Policy	Name	Adaptation Mo	dule		Egress URI Parameters	Notes
SIP Domains	 Site1 CM Digit Conver 	DigitConversio	nAdapter			
Adaptations						
Locations	Digit Conversion for Ind	coming Calls to	SM			
SIP Entities	Add Remove					
Entity Links	4 Items Refresh					Filter: Enable
Time Ranges		[Theorem Endote
Routing Policies	Matching Pattern ▲	Min Max	Delete Digits	Insert Digits	Address to modify	Notes
Dial Patterns	2	• 5 • 5	• 0	73285	origination 🖌	Site 1 PAI
Regular Expressions	• 3	• 5 • 5	• 0	73285	origination 💌	Site 1 PAI
Personal Settings	. 30900	• 5 • 5	• 5	9089530000	destination 💌	Site 1 Calls to Multi-Site MM
Security	• 5	· 5 · 5	• 0	73285	origination 💌	Site 1 PAI
Applications						
Settings	Select: All, None (0 of 4 Se	lected)				
Session Manager						
Shortcuts	Digit Conversion for Ou	tgoing Calls fr	om SM			
Change Password	Add Remove					
Help for Adaptation Details fields						
Help for Committing configuration	2 Items Refresh					Filter: Enable
changes	□ Matching Pattern ▲	Min Max	Delete Digits	Insert Digits	Address to modify	Notes
	• 0000010	• 9 • 9	• 5	3	destination 💌	IPTF Calls to Site 1
	• 73285	• 10 • 10	• 5		destination 💌	MM MWI Notify to Site 1
	Select: All, None (0 of 2 Se	lected)				

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

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4.5.3. Adaptation for Avaya Modular Messaging

The Adaptation administered in this section is used for digit conversion on SIP messages to and from Avaya Modular Messaging (in the Main Location) as follows:

- On egress SIP messages to Avaya Modular Messaging where the Request-URI contains the Avaya Modular Messaging uniform pilot number, the Adaptation prefixes a leading "1" to the number in order to conform to the numbering plan administered on Avaya Modular Messaging for this sample configuration¹⁰. Recall that in Section 4.5.2, on ingress SIP messages from Avaya AuraTM Communication Manager, the pilot extension contained in the Request-URI is converted to a uniform 10-digit pilot number. Here, a leading "1" is prefixed to that uniform pilot number in the Request-URI on egress to Avaya Modular Messaging.
- On egress SIP messages to Avaya Modular Messaging where the P-Asserted-Identity header contains a 10-digit number associated with an extension on Avaya Aura[™] Communication Manager, the Adaptation prefixes a leading "1" to the number in order to conform to the numbering plan administered on Avaya Modular Messaging for this sample configuration. Recall that in Section 4.5.2, on ingress SIP messages from Avaya Aura[™] Communication Manager, the extensions contained in the P-Asserted-Identity header are converted to 10-digit numbers. Here, a leading "1" is prefixed to those 10-digit numbers in the P-Asserted-Identity header on egress to Avaya Modular Messaging.
- On ingress SIP messages from Avaya Modular Messaging where the Request-URI and/or Message-Account header contains an 11-digit number with a leading "1", the Adaptation removes the leading "1".
 - 1. In the Adaptations page (not shown), click on "New".
 - 2. In the Adaptation Details page, enter a descriptive Name and "DigitConversionAdapter" for Adaptation Module.
 - 3. In the **Digit Conversion for Outgoing Calls from SM** section, click on "**Add**" to configure an entry for prefixing a leading "1" to the Avaya Modular Messaging uniform pilot number. Provision the entry as follows:
 - Matching Pattern Enter the Avaya Modular Messaging uniform pilot number.
 - Min and Max Enter "10".
 - **Delete Digits** Enter "**0**".
 - Insert Digits Enter "1".
 - Address to modify Select "destination".
 - 4. In the **Digit Conversion for Outgoing Calls from SM** section, click on "Add" to configure an entry for prefixing a leading "1" to a range of 10-digit numbers associated with extensions on Avaya Aura[™] Communication Manager. Provision the entry as follows:
 - **Matching Pattern** Enter enough leading digits to uniquely match the number range, specifically the range of 10-digit numbers contained in the P-Asserted-Identity header (converted from extensions to 10-digit numbers in Section 4.5.2

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

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

- Min and Max Enter "10".
- **Delete Digits** Enter "**0**".
- Insert Digits Enter "1".
- Address to modify Select "origination".
- Repeat Step 4 as necessary to provision additional entries to cover all expected ranges of 10-digit numbers associated with extensions on Avaya Aura[™] Communication Manager.
- 6. In the **Digit Conversion for Incoming Calls to SM** section, click on "**Add**" to configure an entry for removing the leading "1" from all 11-digit numbers in the Request-URI and/or Message-Account header of ingress SIP messages from Avaya Modular Messaging. Provision the entry as follows:
 - Matching Pattern Enter "1".
 - Min and Max Enter "11".
 - Delete Digits Enter "1".
 - Address to modify Select "destination".
- 7. Click on "Commit".

AVAYA	Ava	va Aura Sv	stem	Mana	aer 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 12:54 PM				n at Jul. 22, 2009 12:54
FULYE	Avaya Aura System Manager 1.0									Help Log off
Home / Network Routing Policy / A	Adaptations / Ada	ptation Details								
Asset Management	Adapta	tion Detail	s							Commit Cancel
▶ User Management										
▶ Monitoring	General									
▼ Network Routing Policy	Name		Adapta	ntion Mod	ule		E	gress URI Parameter	s N	otes
SIP Domains	• MM Digit	Conversion	DigitCo	nversionA	\dapter					
Adaptations										
Locations	Digit Conv	ersion for Inc	oming	Calls to	SM					
SIP Entities		nove	_							
Entity Links		1076								
Time Ranges	1 Item Re	efresh								Filter: Enable
Routing Policies	Mat	ching Pattern 🔺	Min	Мах	Delete	Insert	Diaits	Address to	Notes	
Dial Patterns		,			Digits			modify		
Regular Expressions	• 1		• 11	• 11	• 1			destination 💌		
Personal Settings	Select: All,	None (0 of 1 Sel	lected)							
▶ Security										
Applications	Digit Conv	ersion for Ou	taoina (alle fro	m SM					
▶ Settings			cyoing .	suns no						
▶ Session Manager	Add Rer	nove								
Charden de	2 Items F	Refresh								Filter: Enable
Shortcuts Change Password	Mat	ching Pattern 🔺	Min	Мах	Delete Digits	Insert	Digits	Address to modify	Notes	
Help for Adaptation Details fields	- 73	3285	• 10	• 10	• 0	1		origination 🗸		
Help for Committing configuration		18953000	• 10	• 10	• 0	1		destination 🗸		
changes			10	10	-	-			L]
	Select: All,	None (0 of 2 Sel	lected)							
	* Input Reg	uired								Commit Cancel

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

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4.6. SIP Entities

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

- Avaya AuraTM Session Manager
- Avaya AuraTM Communication Manager
- Acme Packet SBC
- Avaya Modular Messaging

4.6.1. Avaya Aura[™] Session Manager SIP Entity

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

This entry enables Avaya Aura[™] Session Manager to accept SIP requests on TLS port 5061. In addition, Avaya Aura[™] Session Manager will associate SIP requests received on this port that contain the IP address of the SM100 card on Avaya Aura[™] Session Manager in the host part of the Request-URI with the selected SIP **Default Domain**.

- 4. Repeat Step 3 to provision another similar entry, except with "5060" for Port and "TCP" for Protocol.
- 5. Repeat Step 3 as necessary to provision entries for other ports on which Avaya Aura[™] Session Manager is allowed to accept SIP requests.
- 6. Click on "**Commit**".

AVAYA	Avaya Aura Sys	tem Ma	Welcome, admin Las PM	t Logged on at Jul. 22, 2009 12:54 Help Log off	
Home / Network Routing Policy / SI	IP Entities / SIP Entity Details				
 ▶ Asset Management ▶ User Management ▶ Monitoring 	SIP Entity Details			Commit Cancel	
 Network Routing Policy 	Name	FQDN o	or IP Address	Туре	Notes
SIP Domains	* SM1	• 10.16	50.183.209	Session Manager 💌	Asset Board of S
Adaptations	Entity Links 🖲				
Locations	Adaptation:		v		
SIP Entities	Location:		Main 🔽 🕑		
Entity Links	Outbound Proxy:		······································		
Time Ranges	Time Zone:		America/New_York	*	
Routing Policies	Override Port & Transport with	DNS SRV:			
Dial Patterns	SIP Timer B/F (in seconds):		* 4		
Regular Expressions	Credential name:				
Personal Settings	Credential nume.				
▶ Security	SIP Link Monitoring				
Applications	SIP Link Monitoring: Use Ses	sion Manage	er Configuration 💌		
 Settings Session Manager 	Port Add Remove				
Shortcuts	2 Items Refresh				Filter: Enable
Change Password	·				
Help for SIP Entity Details fields			Default Domain	Notes	
Help for Committing configuration changes			spdevcon.com		
unanges	5061	TLS 🚩	spdevcon.com 💌		
	Select: All, None (0 of 2 Selec	ted)			
	* Input Required				Commit Cancel

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

4.6.2. Avaya Aura[™] Communication Manager SIP Entity

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

Αναγα	Avaya Aura	Welcome, admin PM	Last Logged on at Jul. 22, 2009 12:54 Help Log off	
Home / Network Routing Policy /	SIP Entities / SIP Entity Deta	ils		
 ▶ Asset Management ▶ User Management ▶ Monitoring 	SIP Entity Det General	ails		Commit Cancel
 Network Routing Policy 	Name	FQDN or IP Address	Туре	Notes
SIP Domains	Site1 CLAN1	• 10.160.179.110	CM	Site1 CM CLAN1
Adaptations	Entity Links 🕨			
Locations	Adaptation:	Site1 CM Digit Conversion 🛩]	
SIP Entities	Location:	Site 1 💌 🖲		
Entity Links	Time Zone:	America/New_York	~	
Time Ranges	Override Port & Transp	oort with DNS SRV:		
Routing Policies	SIP Timer B/F (in seco	nds): * 4		
Dial Patterns	Credential name:			
Regular Expressions	Call Detail Recording:	egress 💙		
Personal Settings				
▶ Security	SIP Link Monitoring	9		
▶ Applications	SIP Link Monitoring:	Link Monitoring Enabled	*	
▶ Settings	Proactive Monitoring In	nterval (in seconds): * 900		
Session Manager	Reactive Monitoring Int	terval (in seconds): * 120		
Shortcuts	Number of Retries:	* 1		
Change Password Help for SIP Entity Details fields	* Input Required			Commit

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

4.6.3. Acme Packet SBC SIP Entity

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

AVAYA	Avaya Aura S	ra System Manager 1.0 Welcome, admin Last Logged on at Jul. 22, 2009 PM Help Last Logged on at Jul. 22, 2009 PM				
Home / Network Routing Policy / Asset Management User Management	SIP Entities / SIP Entity Details SIP Entity Detail General	ils		Commit Cancel		
▶ Monitoring			_			
▼ Network Routing Policy	Name	FQDN or IP Address	Туре	Notes		
SIP Domains	AT&T IPTF via Acme SD	• 10.160.183.219	SBC	V IP is Acme SD IP		
Adaptations	Entity Links 🕑					
Locations	Adaptation:	AT&T Conversion	*			
SIP Entities	Location:	Main 💌 🕑				
Entity Links	Time Zone:	America/New_York	*			
Time Ranges	Override Port & Transport	t with DNS SRV:				
Routing Policies	SIP Timer B/F (in second	s): * 4				
Dial Patterns	Credential name:					
Regular Expressions	Call Detail Recording:	egress 🗸				
Personal Settings	cun becan keebruing.	CGICUS -				
▶ Security	SIP Link Monitoring					
▶ Applications	SIP Link Monitoring:	Link Monitoring Enabled	*			
▶ Settings	Proactive Monitoring Inter	rval (in seconds): * 900				
▶ Session Manager	Reactive Monitoring Inter	val (in seconds): * 120				
Shortcuts	Number of Retries:	* 1				
Change Password Help for SIP Entity Details fields	* Input Required			Commit Cancel		

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

4.6.4. Avaya Modular Messaging SIP Entity

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

Αναγα	Avaya Aura S	System Manager 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 12: PM Help Log (
Home / Network Routing Policy /	SIP Entities / SIP Entity Details							
 Asset Management User Management 	SIP Entity Detai	ls		Commit Cancel				
 Monitoring Network Routing Policy 	Name	FQDN or IP Address	Туре	Notes				
SIP Domains	ModularMessaging	• 10.160.183.220	Other	▼				
Adaptations Locations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns Regular Expressions Personal Settings	Entity Links Adaptation: Location: Time Zone: Override Port & Transport SIP Timer B/F (in seconds Credential name: Call Detail Recording:		v					
 Security Applications Settings Session Manager 	SIP Link Monitoring SIP Link Monitoring: Proactive Monitoring Inter Reactive Monitoring Interv Number of Retries:		~					
Change Password Help for SIP Entity Details fields	* Input Required			Commit				

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

4.7. Entity Links

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

- Avaya AuraTM Communication Manager
- Acme Packet SBC
- Avaya Modular Messaging

4.7.1. Entity Link to Avaya Aura™ Communication Manager

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

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- **Trusted** Check the checkbox.
- **Protocol** Select "**TLS**".
- 3. Click on "Commit".

AVAYA	Avaya Aur	a System	Mana	ger 1.0		Welc PM	ome, admin	Last Logged o	on at Jul. 22, 2009 12:54 Help Log off
Home / Network Routing Policy	/ Entity Links								
 ▶ Asset Management ▶ User Management ▶ Monitoring 	Entity Links								Commit Cancel
Network Routing Policy SIP Domains									
	1 Item Refresh								Filter: Enable
Adaptations	Name	SIP Entity	Port	SIP Entity 2		Port	Trusted	Protocol	Notes
Locations SIP Entities	Site1CLAN1	• SM1 💌	• 5061	Site1 CLAN1	~	• 5061		TLS 🔽	
Entity Links									
Time Ranges									
Routing Policies	* Input Required								Commit Cancel
Dial Patterns	Input Kequireu								

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

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

Repeat Section 4.7.1 with the following differences:

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

AVAYA	Avaya Aur	a System	Mana	ger 1.0	Welc PM	ome, admin	Last Logged c	n at Jul. 22, 2009 12:54 Help Log off
Home / Network Routing Policy	e / Entity Links							
▶ Asset Management	Entity Links							Commit Cance
▶ User Management								
▶ Monitoring								
Network Routing Policy								
SIP Domains	1 Item Refresh							Filter: Enable
Adaptations		SIP Entity						
Locations	Name	1	Port	SIP Entity 2	Port	Trusted	Protocol	Notes
SIP Entities	AT&T IPTF	• SM1 🔽	• 5060	• AT&T IPTF via Acme SD 💌	• 5060	~	ТСР 🔽	
Entity Links								
Time Ranges								
Routing Policies	* Input Required							Commit Cance
Diel Detterree	Inpac Kequileu							Canad

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

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4.7.3. Entity Link to Avaya Modular Messaging

Repeat Section 4.7.1 with the following differences:

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

AVAYA	Avaya Aura	System	Mana	ger 1.0		Welc PM	ome, admin	Last Logged o	n at Jul. 22, 2009 12:54
	,	,		5					Help Log off
Home / Network Routing Policy / Er	ntity Links								
▶ Asset Management	Entity Links								Commit Cancel
▶ User Management	-								
▶ Monitoring									
Network Routing Policy									
SIP Domains	1 Item Refresh								Filter: Enable
Adaptations		SIP Entity							
Locations	Name	1	Port	SIP Entity 2		Port	Trusted	Protocol	Notes
SIP Entities	 ModularMessaging 	• SM1 🔽	• 5060	 ModularMessaging 	*	• 5060	~	ТСР 💌	
Entity Links									
Time Ranges									
Routing Policies	* Input Required								Commit Cancel
Dial Patterns									

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

4.8. Time Ranges

- 1. In the left pane under **Network Routing Policy**, click on "**Time Ranges**". In the **Time Ranges** page (not shown), click on "**New**".
- 2. Continuing in the **Time Ranges** page, enter a descriptive **Name**, check the checkboxes for the desired day(s) of the week, and enter the desired **Start Time** and **End Time**.
- 3. Click on "Commit".
- 4. Repeat Steps 1 3 to provision additional time ranges.

AVAYA	Avaya Aura S	Avaya Aura System Manager 1.0							Welcome, admin Last Logged on at Jul. 22, 2009 12:54 PM Help Log off				
Home / Network Routing Policy	/ Time Ranges												
▶ Asset Management	Time Ranges										Commit Canc		
▶ User Management	-												
▶ Monitoring													
▼Network Routing Policy													
SIP Domains	1 Item Refresh										Filter: Enable		
Adaptations	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes		
Locations											Hotes		
SIP Entities	AllTimes		~	✓	~	~	~	✓	• 00:00	• 23:59			
Entity Links													
Time Ranges													
Routing Policies	* Input Required										Commit Canc		

Figure 18: Time Ranges Page

4.9. Routing Policies

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

- Avaya Aura[™] Communication Manager
- Avaya Modular Messaging

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

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

AVAYA	Avaya Aura	System Mana	ger 1.0)			Welcom PM	e, admin Last Logg		2009 13:46 lp Log off
Home / Network Routing Policy / Ro	outing Policies / Routing Polic	y Details								
▶ Asset Management ▶ User Management	Routing Policy	Details							Commit	Cancel
▶ Monitoring	General									
▼ Network Routing Policy	Name			Di	sabled		Notes			
SIP Domains	ToSite1CM									
Adaptations										
Locations	SIP Entity as Destin	ation								
SIP Entities	Select									
Entity Links									1	
Time Ranges	Name	FQDN or IP Address						Туре	Notes	
Routing Policies										
Dial Patterns	Time of Day									
Regular Expressions	Add Remove	View Gaps/Overlaps								
Personal Settings										
▶ Security	0 Items Refresh								Filte	r: Enable
Applications	Ranking Na	ime Mon Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
▶ Settings										

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

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

avaya		Avaya Aura Syster	n Manager 1.0	PM	Help Log o
lome / Network Routing Policy ,	/ Routing Po	licies / Routing Policy Details / S	IP Entity List		
Asset Management	SIF	PEntity List			Select Cano
▶ User Management					
▶ Monitoring					
▼Network Routing Policy					
SIP Domains	SIP	Entities			
Adaptations	4.74				citere carble
Locations	4 It	ems Refresh			Filter: Enable
SIP Entities		Name	FQDN or IP Address	Туре	Notes
Entity Links	0	AT&T IPTF via Acme SD	10.160.183.219	SBC	IP is Acme SD IP
Time Ranges	0	ModularMessaging	10.160.183.220	Other	
Routing Policies	۲	Site1 CLAN1	10.160.179.110	СМ	Site1 CM CLAN1
Dial Patterns	0	SM1	10.160.183.209	Session Manager	Asset Board of SM
Regular Expressions	Sel	ect: None			
Personal Settings	361				
Security					
Applications					
▶ Settings					Select Canc

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

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

Αναγα	Avaya Aura	a System Ma	anager 1.0)		Welco PM	me, admin La	st Logged on at Jul. 22, H	2009 13:46 elp Log off
Home / Network Routing Policy /	/ Routing Policies / Routing Pol	licy Details							
▶ Asset Management	Routing Policy	/ Details						Comm	it Cancel
▶ User Management									
▶ Monitoring	General								
▼ Network Routing Policy	Name			Dis	abled	Notes			
SIP Domains	ToSite1CM								
Adaptations									
Locations	SIP Entity as Desti	ination							
SIP Entities	Select								
Entity Links									
Time Ranges	Name	FQDN or IP	Address			Туре	No	otes	
Routing Policies	Site1 CLAN1	10.160.179.1	10			СМ	Site	91 CM CLAN1	
Dial Patterns									
Regular Expressions	Time of Day								
Personal Settings	Add Remove	View Gaps/Overlap	5						
▶ Security									
Applications	0 Items Refresh							Filt	er: Enable
▶ Settings	Ranking M	Name Mon	Tue Wed	Thu	Fri 9	iat Sur	n Start T	ime End Time	Notes
▶ Session Manager		T I	1			T			

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

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

AVAYA		Avaya Aura System Manager 1.0							Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM Help Log off						
lome / Network Routing Policy	/ Routing Poli	cies / Routin(g Policy Det	ails / Time	Range Lis	t									
 Asset Management User Management 	Tim	e Range	e List								Sel	ect Canc			
▶ Monitoring ▼ Network Routing Policy															
SIP Domains	Time	Ranges													
Adaptations															
Locations	1 Ite	m Refresh									Fi	lter: Enable			
SIP Entities		Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes			
Entity Links		AllTimes							Z	00:00	23:59				
Time Ranges															
Routing Policies	Selec	t: All, None (1 of 1 Se	lected)											
Dial Patterns															
Regular Expressions															
Personal Settings												ect] Cano			
Fersonal Settings											Sel	ect Can			

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

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

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avaya	Avaya Aur	a System Mana	ager 1	L.O			Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM Help Log off					
Home / Network Routing Policy ,	/ Routing Policies / Routing Po	licy Details									p Log on	
▶ Asset Management	Routing Polic	y Details								Commit	Cance	
▶ User Management												
Monitoring	General											
▼Network Routing Policy	Name		Not	es								
SIP Domains	• ToSite1CM											
Adaptations												
Locations	SIP Entity as Dest	ination										
SIP Entities	Select											
Entity Links												
Time Ranges	Name	FQDN or IP Ad	iress				Туре		Notes			
Routing Policies	Site1 CLAN1	10.160.179.110					СМ		Site1 CM CL	AN1		
Dial Patterns												
Regular Expressions	Time of Day											
Personal Settings	Add Remove	View Gaps/Overlaps										
 Security Applications 	1 Item Refresh									Filter	r: Enable	
 Settings 	D Desking (Norra D. Norr	Terr		Thu	E al	0-1	0	Obsist Times	n - d Timer		
 Session Manager 	Ranking 1		Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes	
P Session Manayer		AllTimes 🖂	×	¥	×	V	×	¥	00:00	23:59		
Shortcuts	Select: All, None (0	of 1 Selected)										
Change Password												

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

4.9.2. Routing Policy for Routing to Avaya Modular Messaging

Repeat Section 4.9.1 with the following differences:

- In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** for routing calls to Avaya Modular Messaging, and ensure that the **Disabled** checkbox is unchecked to activate this Network Routing Policy.
- In the **SIP Entity List** page, select the SIP Entity administered in Section 4.6.4 for Avaya Modular Messaging, and click on "**Select**".

AVAYA	Avaya Aura Sy	Avaya Aura System Manager 1.0									:009 13:46 p Log off
Home / Network Routing Policy	/ Routing Policies / Routing Policy De	tails									
▶ Asset Management	Routing Policy Det	tails								Commit	Cancel
 User Management Monitoring 	General										
▼Network Routing Policy	Name										
SIP Domains	• ToMM										
Adaptations											
Locations	SIP Entity as Destinatio	n									
SIP Entities	Select										
Entity Links	301000								F		
Time Ranges	Name	F	QDN or IP	Address					Туре	Notes	
Routing Policies	ModularMessaging	10	.160.183.2	20					Other		
Dial Patterns											
Regular Expressions	Time of Day										
Personal Settings	Add Remove View	Gaps/Overlaps									
 Security Applications 	1 Item Refresh									Filter	r: Enable
▶ Settings	□ Ranking 1 ▲ Nam	ne 2.⊾ Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
▶ Session Manager		nes 🗸	V	V	Image: A state of the state	V	V	V	00:00	23:59	
Shortcuts	Select: All, None (0 of 1 Sele	ected)									
Change Password											

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

4.10. Dial Patterns

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

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

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

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

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AVAYA	Avaya Aura S	System	n Manage	r 1.0	Welc AM	ome, adm i	i n Last Logged o	n at Aug. 10, 2009 09:52 Help Log off			
Home / Network Routing Policy / I	Dial Patterns / Dial Pattern Deta i	ils									
 Asset Management User Management Monitoring 	Dial Pattern Det	ails						Commit Cancel			
 Monitoring Network Routing Policy 		Min	Мах	Emergency Call	SIP Domain		Notes				
SIP Domains		9	• 9		-ALL-	~	Notes				
Adaptations	0000010	2	2								
Locations	Originating Locations	and Rou	tina Policies								
SIP Entities	Add Remove										
Entity Links											
Time Ranges	0 Items Refresh							Filter: Enable			
Routing Policies	Originating Locatio	on Orig	inating Locatio	on Routing Policy	Routing	Rout	ing Policy	Routing Policy			
Dial Patterns	Name	Note		Name	Policy Disabled	Dest	ination	Notes			
Regular Expressions											
Personal Settings	Denied Originating Lo	cations									
▶ Security	Add Remove										
Applications	0 Items Refresh							Filter: Enable			
Settings											
Session Manager	Originating Locatio	on					Notes				
Shortcuts Change Password	* Input Required							Commit Cancel			

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

- 4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page, check the checkbox corresponding to the Location to which the Acme Packet SBC is assigned (see Section 4.6.3 Step 2). Note that only those calls that originate from the selected Location(s), or all administered Locations if "-ALL-" is selected, can match this Dial Pattern.
- In the Routing Policies section of the Originating Location and Routing Policy List page, check the checkbox corresponding to the Routing Policy administered for routing calls to Avaya Aura[™] Communication Manager in Section 4.9.1.
- 6. In the Originating Location and Routing Policy List page, click on "Select".

AVAVA	Avaya Aura Syst	em Manager 1	.0	Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM
· ·				Help Log off
Home / Network Routing Policy / D	Dial Patterns / Dial Pattern Details / Lo	cations and Routing Pol	icy List	
Asset Management	Originating Location	n and Routing	Policy List	Select
▶ User Management		5	•	
▶ Monitoring				
Network Routing Policy				
SIP Domains	Originating Location			
Adaptations	O Marrie Defeath			Elhow English
Locations	3 Items Refresh			Filter: Enable
SIP Entities	Name		Notes	
Entity Links	-ALL-		Any Locations	
Time Ranges	Main		Main Site	
Routing Policies	Site 1		Site 1	
Dial Patterns	Select: All, None (1 of 3 Select	od)		
Regular Expressions	Select. All, None (1 of 3 Select	eu)		
Personal Settings				
▶ Security				
Applications	Routing Policies			
▶ Settings				
▶ Session Manager	2 Items Refresh			Filter: Enable
Shortcuts	Name	Disabled	Destination	Notes
Change Password	ТоММ		ModularMessaging	
Changerassword	✓ ToSite1CM		Site1 CLAN1	
	Select: All, None (1 of 2 Select	ed)		
	Scietti All, None (1 01 2 Seletti	,		
				Select

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

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

avaya	Avaya Au	ra System Ma	Welco AM	ome, admin Last Logged	l on at Aug. 10, 2009 09:52 Help Log off	
Home / Network Routing Policy / Dia	al Patterns / Dial Pattern	Details				
▶ Asset Management ▶ User Management ▶ Monitoring	Dial Pattern I General	Details				Commit Cancel
Network Routing Policy	Pattern	Min Max	Emergency Call	SIP Domain	Notes	
SIP Domains	• 0000010	• 9 • 9		-ALL-	~	
Adaptations				L		
Locations	Originating Locati	ons and Routing	Policies			
SIP Entities	Add Remove	2				
Entity Links						
Time Ranges	1 Item Refresh					Filter: Enable
Routing Policies	Originating La	cation Originatio	g Location Routing Poli	cy Routing	Routing Policy	Routing Policy
Dial Patterns	Name	Notes	Name	Disabled	Destination	Notes
Regular Expressions	Main	Main Site	ToSite1CM		Site1 CLAN1	
Personal Settings						
Security	Select: All, None (0	of 1 Selected)				
Applications						
▶ Settings	Denied Originating	g Locations				
Session Manager	Add Remove					
Shortcuts	0 Items Refresh					Filter: Enable
Change Password	Originating Lo	cation			Notes	
Help for Dial Pattern Details fields						
Help for Location and Routing Policy	* Input Required					Commit

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

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

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

Avaya Aur	r 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 13: PM					
Dial Patterns / Dial Pattern (Details						Help Log of
Dial Pattern D	etails						Commit Cance
General							
Pattern	Min	Мах	Emergency Call	SIP Domain		Notes	
• 73285				-ALI -	~		
7.0200	10	10		1 Hata			
Originating Locatio	ons and R	outing Policies					
Add Kenlove							
0 Items Refresh							Filter: Enable
Originating Lo	cation O	riginating Location	on Routing Policy	Routing	Routi	ing Policy	Routing Policy
Name			Name	Policy Disabled			Notes
Denied Originating	Location	s					
Add Remove							
0 Itoms Pofrash							Filter: Enable
o items Refresh						1	Filter: Enable
Originating Lo	cation					Notes	
* Input Required							Commit Cance
	Dial Patterns / Dial Pattern D Dial Pattern D General Pattern • 73285 Originating Location Add Remove 0 Items Refresh Denied Originating Lo Name 0 Items Refresh O Items Refresh O Items Refresh O Items Refresh O Items Refresh O Items Refresh O Items Refresh	Dial Patterns / Dial Pattern Details Dial Pattern Details General Pattern Min 73285 10 Originating Locations and R Add Remove 0 Items Refresh Denied Originating Location Add Remove 0 Items Refresh Ditems Refresh	Dial Patterns / Dial Pattern Details Dial Pattern Details General Pattern Min Max 73285 10 10 Originating Locations and Routing Policies Add Remove Interns Refresh Denied Originating Locations Add Remove Interns Refresh Driginating Locations Add Remove Interns Refresh Driginating Location	Dial Pattern Details General Pattern Min Max Emergency Call • 73285 • 10 • 10 • Originating Locations and Routing Policies Add Remove 0 Items Refresh • Originating Location Originating Location Name Originating Locations Add Remove • O Items Refresh • O Items Refresh • O Items Refresh • O Items Refresh • O Items Refresh	Dial Pattern Details Dial Pattern Details General Pattern Min Max Emergency Call SIP Domain 73285 + 10 + 10 - ALL- Originating Locations and Routing Policies Add Remove 0 Items Refresh Originating Locations Add Remove 0 Items Refresh Denied Originating Locations Add Remove 0 Items Refresh Driginating Locations Add Remove 0 Items Refresh Driginating Locations	Dial Pattern Details Dial Pattern Details General Pattern Min Max Emergency Call SIP Domain 73285 10 10	Dial Pattern Details General Pattern Min Max Emergency Call SIP Domain Notes * 73285 * 10 * 10 - ALL- • • Originating Locations and Routing Policies • • • • • Items Refresh • <td< td=""></td<>

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

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

Αναγα	Avaya Aura Sys	tem Manager 1	.0	Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM
		j		Help Log off
Home / Network Routing Policy /	Dial Patterns / Dial Pattern Details / L	ocations and Routing Po	licy List	
▶ Asset Management	Originating Locatio	on and Routing	Policy List	Select
▶ User Management		5	•	
▶ Monitoring				
▼ Network Routing Policy				
SIP Domains	Originating Location			
Adaptations				
Locations	3 Items Refresh			Filter: Enable
SIP Entities	Name		Notes	
Entity Links	-ALL-		Any Locations	
Time Ranges	Main		Main Site	
Routing Policies	Site 1		Site 1	
Dial Patterns				
Regular Expressions	Select: All, None (1 of 3 Selec	cted)		
Personal Settings				
▶ Security				
Applications	Routing Policies			
▶ Settings	-			
▶ Session Manager	2 Items Refresh			Filter: Enable
Shortcuts	Name	Disabled	Destination	Notes
Change Password	ТоММ		ModularMessaging	
Change Passworu	ToSite1CM		Site1 CLAN1	
	Select: All, None (1 of 2 Sele	atod)		
	Select. All, None (1 Of 2 Selec	licu)		
	[
				Select Cancel

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

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

avaya	Avaya Aura System Manager 1.0				Welcome, admin Last Logged on at Jul. 22, 2009 13: PM Help Log (
Home / Network Routing Policy / Di	ial Patterns / Dial Pattern I	Details							
▶ Asset Management	Dial Pattern D	Details						Commit Cance	
▶ User Management									
▶ Monitoring	General								
▼ Network Routing Policy	Pattern	Min	Max I	mergency Call	SIP Domain		Notes		
SIP Domains	• 73285	• 10	• 10		-ALL-	~			
Adaptations									
Locations	Originating Location	ons and R	outing Policies						
SIP Entities	Add Remove								
Entity Links									
Time Ranges	1 Item Refresh							Filter: Enable	
Routing Policies	Originating Lo	cation (riginating Location	Routing Polic	y Routing	Routi	ng Policy	Routing Policy	
Dial Patterns	□ Name	١	lotes	Name	y Policy Disabled	Desti	nation	Notes	
Regular Expressions	-ALL-	А	ny Locations	ToSite1CM		Site1 C	LAN1		
Personal Settings	Select: All, None(0	of 1 Coloctor	4.5						
▶ Security	Select. All, None (0	DI I Seleccei	,)						
Applications									
▶ Settings	Denied Originating	Location	S						
▶ Session Manager	Add Remove								
Shortcuts	0 Items Refresh							Filter: Enable	
Change Password	Originating Lo	cation					Notes		
Help for Dial Pattern Details fields									
Help for Location and Routing Policy Lists	* Input Required							Commit	

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

4.10.3. Matching Calls to Avaya Modular Messaging Pilot Number

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

AVAYA	Avaya Aur	Avaya Aura System Manager 1.0					Welcome, admin Last Logged on at Jul. 22, 2009 13:4 PM				
Home / Network Routing Policy /	/ Dial Patterns / Dial Pattern E	Details						Help Log off			
> Asset Management	Dial Pattern D	etails						Commit Cance			
🕨 User Management											
Monitoring	General										
Network Routing Policy	Pattern	Min	Мах	Emergency Call	SIP Domain		Notes				
SIP Domains	• 9089530000	• 10	• 10		-ALL-	~					
Adaptations											
Locations	Originating Locatio	ons and R	outing Policies								
SIP Entities	Add Remove		2								
Entity Links											
Time Ranges	0 Items Refresh							Filter: Enable			
Routing Policies	Originating Lo	cation O	riginating Locatio	n Routing Policy	Routing	Rout	ing Policy	Routing Policy			
Dial Patterns	Name		otes	Name	Policy Disabled		ination	Notes			
Regular Expressions											
Personal Settings	Denied Originating	Location	5								
Security	Add Remove										
Applications	O there a l Defende							Filter: Enable			
▶ Settings	0 Items Refresh							Hiter: Enable			
Session Manager	Originating Lo	cation					Notes				
Shortcuts	* Input Required							Commit Cance			
Change Password											

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

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

Αναγα	Avaya Aura Syst	em Manager 1	.0	Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM	
				Help Log off	
Home / Network Routing Policy /	Dial Patterns / Dial Pattern Details / Lo	cations and Routing Pol	licy List		
Asset Management	Originating Location	n and Routing	Policy List	Select Cancel	1
▶ User Management	5 5	5			
▶ Monitoring					
Network Routing Policy					1
SIP Domains	Originating Location				
Adaptations					
Locations	3 Items Refresh			Filter: Enable	
SIP Entities	□ Name		Notes		
Entity Links	-ALL-		Any Locations		1
Time Ranges	Main		Main Site		1
Routing Policies	Site 1		Site 1		
Dial Patterns	Select: All, None (1 of 3 Select	ad)			
Regular Expressions	Select: All, None (1 of 3 select	eu)			
Personal Settings					
▶ Security					2
▶ Applications	Routing Policies				
▶ Settings					1
Session Manager	2 Items Refresh			Filter: Enable	
Shortcuts	Name	Disabled	Destination	Notes	
Change Password	🔽 ТоММ		ModularMessaging		
Change Password	ToSite1CM		Site1 CLAN1		
	Select: All, None (1 of 2 Select	ed)			
	Select. All, None (1 612 Select	eu)			
				Select	

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

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

AVAYA	Avaya Aura System Manager 1.0				Welcome, admin Last Logged on at Jul. 22, 2009 13:4 PM Help Log o l				
Home / Network Routing Policy / Di	al Pattern	s / Dial Pattern I	Details						
▶ Asset Management	Dial	Pattern D)etails						Commit Cancel
▶ User Management									
▶ Monitoring	Gene	ral							
▼Network Routing Policy	Patte	ern	Min	Мах	Emergency Call	SIP Domain		Notes	
SIP Domains	• 90	89530000	• 10	• 10		-ALL-	~		
Adaptations									
Locations	Origii	nating Locatio	ons and R	outing Policies					
SIP Entities	Add	Remove							
Entity Links									
Time Ranges	1 Ite	m Refresh							Filter: Enable
Routing Policies		Originating Lo	cation O	riginating Locatio	n Routing Policy	Routing	Routi	ng Policy	Routing Policy
Dial Patterns		Name	N	otes	Name	Policy Disabled		nation	Notes
Regular Expressions		-ALL-	Ar	ny Locations	ToMM		Modula	irMessaging	
Personal Settings	Solo	t: All, None (0 (of 1 Coloctor	4.)					
▶ Security	36161	c. All, None (0 (JI I Selected	,,					
▶ Applications									
▶ Settings	Denie	ed Originating	Location	S					
▶ Session Manager	Add	Remove							
Shortcuts	0 Ite	ms Refresh							Filter: Enable
Change Password		Originating Lo	cation					Notes	
Help for Dial Pattern Details fields									
Help for Location and Routing Policy Lists	* Inpu	t Required							Commit Cance

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

4.11. Session Manager Administration

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

Αναγα	Avaya Aura System Mana	ger 1.0	Welcome, admin Last Logged on at Jul. 22, 2009 13:46 PM Help Log off
Home / Session Manager / Session	Manager Administration / New Session Manager		
 Asset Management User Management 	Add Session Manager		Cancel Save
 Monitoring Network Routing Policy Security 	General Security Module Monitoring CDF Expand All Collapse All	ξ	
▶ Applications	General 💌		
 Settings Session Manager Session Manager Administration System State Administration Security Module Status Data Replication Status 	* SIP Entity Name Description * Management Access Point Host Name/IP		
Local Host Name Resolution Maintenance Tests	Security Module 💌		
SIP Firewall Configuration	SIP Entity IP Address * Network Mask	10.160.183.209 255.255.255.224	
Tracer Configuration Trace Viewer Call Routing Test	* Default Gateway * Call Control PHB		
Managed Bandwidth Usage	* QOS Priority * Speed & Duplex		
Shortcuts Change Password	VLAN ID		

Figure 34: Add Session Manager Page

5. Avaya Aura[™] Communication Manager

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

5.1. System Parameters

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

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

display system-parameters customer-options		Page	2 of	11
OPTIONAL FEATURES				
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:	1000	41		
Maximum Concurrently Registered IP Stations:	18000	10		
Maximum Administered Remote Office Trunks:	0	0		
Maximum Concurrently Registered Remote Office Stations:	0	0		
Maximum Concurrently Registered IP eCons:	0	0		
Max Concur Registered Unauthenticated H.323 Stations:	5	0		
Maximum Video Capable H.323 Stations:	10	0		
Maximum Video Capable IP Softphones:	10	0		
Maximum Administered SIP Trunks:	1000	152		
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0		
Maximum Number of DS1 Boards with Echo Cancellation:	1	0		
Maximum TN2501 VAL Boards:	10	1		
Maximum Media Gateway VAL Sources:	50	0		
Maximum TN2602 Boards with 80 VoIP Channels:	128	1		
Maximum TN2602 Boards with 320 VoIP Channels:	128	0		
Maximum Number of Expanded Meet-me Conference Ports:	0	0		

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

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

```
display system-parameters customer-options
                                                                Page
                                                                       4 of
                                                                             11
                                 OPTIONAL FEATURES
  Emergency Access to Attendant? y
                                                                 IP Stations? y
          Enable 'dadmin' Login? y
           Enhanced Conferencing? y
                                                         ISDN Feature Plus? v
                                     ISDN/SIP Network Call Redirection? y
                 Enhanced EC500? y
   Enterprise Survivable Server? n
                                                             ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n
                                                                    ISDN-PRI? y
              ESS Administration? n
                                                Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y
                                                       Malicious Call Trace? n
    External Device Alarm Admin? n
                                                   Media Encryption Over IP? n
 Five Port Networks Max Per MCC? n Mode Code for Centralized Voice Mail? n
               Flexible Billing? n
  Forced Entry of Account Codes? n
                                                   Multifrequency Signaling? y
     Global Call Classification? n Multimedia Call Handling (Basic)? y
Hospitality (Basic)? y Multimedia Call Handling (Enhanced)? y
Hospitality (G3V3 Enhancements)? n Multimedia IP SIP Trunking? n
                       IP Trunks? y
           IP Attendant Consoles? n
```

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

5.2. Dial Plan

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

- 3-digit dial access codes (indicated with a **Call Type** of "**dac**") beginning with the digit "**1**" Trunk Access Codes (TACs) defined for trunk groups in this sample configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**3**" local extensions for Avaya Aura[™] Communication Manager stations, agents, and Vector Directory Numbers (VDNs) in this sample configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**5**" local extensions for Avaya AuraTM Communication Manager skills (hunt groups) and announcements in this sample configuration conform to this format.

change dialplan and	alysis			Pag	ge 1 of 12				
		DIAL PLAN ANALYSIS TABLE Location: all Percent Full: 1							
String Ler 1 3	tal Call ngth Type 3 dac 5 ext 5 ext	Dialed String	Total Call Length Type	Dialed String	Total Call Length Type				

Figure 37: Dialplan Analysis Form

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5.3. IP Network Parameters

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

 Enter the change ip-codec-set ci command, where ci is the number of an IP codec set used only for internal calls. On Page 1 of the ip-codec-set form, ensure that "G.711MU", "G.729B", and "G.729A" are included in the codec list as shown in Figure 38.

```
change ip-codec-set 1
                                                                   1 of
                                                                          2
                                                            Page
                         IP Codec Set
   Codec Set: 1
                           Frames
   Audio
                Silence
                                     Packet
                Suppression Per Pkt Size(ms)
   Codec
1: G.711MU
                              2
                                       20
                    n
                               2
2: G.729B
                     n
                                       20
                              2
3: G.729A
                                       20
                     n
```

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

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

2. Enter the **change ip-codec-set ce** command, where **ce** is the number of an unused IP codec set. This IP codec set will be used for inbound AT&T IP Toll Free calls. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown in **Figure 39**.

change ip	-codec-set 2			Page	1 of	2
Codec	Set: 2					
Audio	Silence	Frames	Packet			
Codec	Suppression	Per Pkt	Size(ms)			
1: G.729	B n	2	20			
2: G.729	A n	2	20			
3: G.711	MU n	2	20			

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

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

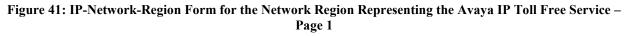
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change ip-codec-set	t 2		Page	2 of	2
5 1			-		
	IP Codec S	et			
	Allow	Direct-IP Multimedia? n			
	Mode	Redundancy			
FAX	t.38-standard	0			
Modem	off	0			
TDD/TTY	off	0			
Clear-channel	n	0			

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

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

```
change ip-network-region 61
                                                                 Page
                                                                        1 of 19
                                IP NETWORK REGION
  Region: 61
Location:
                 Authoritative Domain:
   Name:
MEDIA PARAMETERS
                                 Intra-region IP-IP Direct Audio: yes
     Codec Set: 2
                                Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048
                                            IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
                                          RTCP Reporting Enabled? y
Call Control PHB Value: 46
Audio PHB Value: 46
RTCP MONITOR SERVER PARAMETERS
Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
                                   AUDIO RESOURCE RESERVATION PARAMETERS
        Video 802.1p Priority: 5
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
```



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

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

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

change ip-network-region 61	Page	3	of	19
Source Region: 61 Inter Network Region Connection Manageme	nt	I G	A	M e
dst codec direct WAN-BW-limits Video Intervening	Dyn	A	G	а
rgn set WAN Units Total Norm Prio Shr Regions	CÂC			S
1 2 y NoLimit		n		
2				
3				
4				
5				
6				
7				
8				
9				
10				
11				
12				
13				
14				
15				

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

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

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

Figure 43: Change Node-Names IP Form

5.4. Inbound Calls

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

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

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- **Transport Method** Set to "**tls**". Note that this is only the transport protocol used between Avaya AuraTM Communication Manager and Avaya AuraTM Session Manager. The transport protocol used between Avaya AuraTM Session Manager and the Acme Packet SBC is TCP, and the transport protocol used between the Acme Packet SBC and the AT&T IP Toll Free service is UDP.
- Near-end Node Name Set to the node name of the C-LAN board noted in Section 5.3 Step 4.
- **Far-end Node Name** Set to the node name of Avaya Aura[™] Session Manager as administered in Section 5.3 Step 4.
- Near-end Listen Port and Far-end Listen Port set to "5061".
- **Far-end Network Region** Set to the IP network region administered in Section 5.3 Step 3 to represent the AT&T IP Toll Free service.
- Far-end Domain Leave blank.
- **DTMF over IP** Set to "**rtp-payload**" to enable Avaya Aura[™] Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** Set to "**y**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible.

add signaling-group 61 Page 1 of 1 Group Number: 61 Group Type: sip Transport Method: tls IMS Enabled? n Near-end Node Name: clan-01a09 Far-end Node Name: ASM1 Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Network Region: 61 Far-end Domain: Bypass If IP Threshold Exceeded? n Direct IP-IP Audio Connections? y DTMF over IP: rtp-payload Session Establishment Timer(min): 3 IP Audio Hairpinning? n Direct IP-IP Early Media? n Enable Layer 3 Test? n H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6

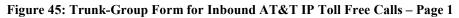
```
Figure 44: Signaling-Group Form for Inbound AT&T IP Toll Free Calls
```

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

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

add trunk-group 61	TRUNK GROUP		Pac	ge 1 of	21
Dial Access? n	Group Type: COR: cgoing Display? Auth Code?	1 n Night	CDR TN: 1 Service:	Reports: TAC:	-
Service Type: public-ntwrk				Group: 62 embers: 20	



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

 8 adds the appropriate prefix to form the appropriate connected party numbers, e.g., converts from extensions to complete numbers in the PAI header sent to the AT&T IP Toll Free service.
 - **CPN Len** Enter the total number of digits in the local extension range.

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

cha	nge public-unk	nown-numbe	ring O			Page	1 of	2
		NUMBE	RING - H	PUBLIC/UNKNOWN	FORMAT			
				Total				
Ext	Ext	Trk	CPN	CPN				
Len	Code	Grp(s)	Prefix	Len				
					Total	Administe	ered:	70
5	3	61		5	Max	imum Entr	cies:	9999
5	5	61		5				

Figure 46: Public-Unknown-Numbering Form

5.5. Call Center

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

display hunt-group 1001		Page 1 of 3
	HUNT	GROUP
Group Number:	1001	ACD? Y
Group Name:	IPTF Skill 1	Queue? y
Group Extension:	51001	Vector? y
Group Type:	ead-mia	
TN:	1	
COR:	1	MM Early Answer? n
Security Code:		Local Agent Preference? n
ISDN/SIP Caller Display:		
Queue Limit:	unlimited	
Calls Warning Threshold:	Port:	
Time Warning Threshold:	Port:	

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

display hunt-group 1001		Page	2 of	3
	HUNT GROUP			
Skill? y AAS? n Measured: none Supervisor Extension:	Expected Call Handling Time	(sec):	180	
Controlling Adjunct: none				
Interruptible Aux Threshold: none	Redirect on No Answer (rings			
Forced Entry o	Redirect to VDI f Stroke Counts or Call Work		n	
roiced Billy O	I SCIUNE COUNCE OF CAIL WOLK	coues:	11	

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

display hunt-group 1001 HUNT GROUP	Page	3 of	3
LWC Reception: none AUDIX	Name:		
Message Center: none			

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

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display agent-loginID 32001	Page	1 of 2
AGENT LOGINID		
Login ID: 32001	AAS?	
Name: Agent-61000	AUDIX?	
TN: 1 LWC I	Reception:	spe
COR: 1 LWC Log Exter:	nal Calls?	n
Coverage Path: AUDIX Name for 1	Messaging:	
Security Code:		
LoginID for ISDN/SI	P Display?	n
	Password:	
Password (ent	er again):	
Au	to Answer:	station
MIA Acro	ss Skills:	system
ACW Agent Conside	ered Idle:	system
Aux Work Reason	Code Type:	system
Logout Reason	Code Type:	system
Maximum time agent in ACW before log	out (sec):	system
Forced Agent Lo	gout Time:	:

Figure 50: Sample Agent Form – Page 1

display agent-	-loginID 32001				Page	2 of	2
	-	AGENT LOGI	NID		-		
Direct A	Agent Skill:			Serv	rice Obje	ective?	n
Call Handling	Preference: sk	cill-level		Local Ca	ll Prefe	erence?	n
		DI GI	CNI	DI GI	0.11	DI GI	
SN RL SI		RL SL	SN	RL SL	SN	RL SL	
1: 1001 1	16:	31:	:		46:		
2: 1002 2	17:	32:	:		47:		
3: 1003 3	18:	33:	:		48:		
4:	19:	34:			49:		
5:	20:	35:			50:		
6:	21:	36:			51:		
3: 7:	22:	37:			52:		
8:	23:	38:			53:		
9:	24:	39:			54:		
10:	25:	40:			55:		
11:	26:	41:	:		56:		
12:	27:	42:	:		57:		
13:	28:	43:			58:		
14:	29:	44:			59:		
15:	30:	45:			60:		

Figure 51: Sample Agent – Page 2

```
display vector 1001
                                                              Page 1 of
                                                                            6
                                  CALL VECTOR
   Number: 1001
                             Name: RouteToSkill1
Multimedia? n Attendant Vectoring? n Meet-me Conf? n
                                                                      Lock? y
    Basic? y EAS? y G3V4 Enhanced? y ANI/II-Digits? y ASAI Routing? y
Prompting? y LAI? n G3V4 Adv Route? y CINFO? n BSR? y Holidays? n
Variables? y 3.0 Enhanced? y
01 wait-time 2 secs hearing ringback
02 announcement 52101
03 queue-to skill 1001 pri m
04 wait-time 10 secs hearing music
05 announcement 52001
                    if unconditionally
06 goto step 3
07 stop08
09
10
11
12
```

Figure 52: Sample Vector

display vdn 31001		Page	1 of	3
VECTOR DIRECTO	ORY NUMBER			
Extension: 3	31001			
Name*: Si	kill 1001			
Destination: Ve	Vector Number 1	001		
Attendant Vectoring? n	1			
Meet-me Conferencing? n				
Allow VDN Override? n				
COR: 1				
TN*: 1				
Measured: no				
VDN of Origin Annc. Extension*:				
1st Skill*:				
2nd Skill*:				
3rd Skill*:				
JIG DAIII .				
* Follows VDN Override Rules				

Figure 53: Sample VDN

6. Avaya Modular Messaging

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

7. Configure Acme Packet SBC

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

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

local-policy

*
*
INSIDE-SM
N/A
N/A
enabled
none
admin@console
2009-05-26 17:55:28
10.242.225.200
OUTSIDE
none
disabled
0000
2400
U-S
0
SIP
enabled

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

local-policy

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from-address to-address source-realm OUTSIDE description activate-time N/A deactivate-time N/A state enabled policy-priority none last-modified-by admin@console last-modified-date 2009-06-09 15:56:00 policy-attribute next-hop 10.160.183.209 realmINSIDE-SM action none terminate-recursion disabled carrier start-time 0000 2400 end-time U-S days-of-week 0 cost 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-quard-timer 300 tcp-number-of-ports-per-flow 2 disabled hnt-rtcp 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 1000 red-sync-comp-time media-policing enabled max-signaling-bandwidth 7752190 max-untrusted-signaling 80 20 min-untrusted-signaling app-signaling-bandwidth 0 tolerance-window 30 rtcp-rate-limit 0 min-media-allocation 32000

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min-trusted-allocation

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60000

	<pre>deny-allocation anonymous-sdp arp-msg-bandwidth fragment-msg-bandwidth rfc2833-timestamp default-2833-duration rfc2833-end-pkts-only-for-non-stranslate-non-rfc2833-event dnsalg-server-failover last-modified-by last-modified-date</pre>	32000 disabled 32000 0 disabled 100 sig enabled disabled disabled admin@console 2009-03-12 10:22:03
netwo	rk-interface	
110 0 110	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	
		disabled
	state heartbeat	
		0
	retry-count	0
	retry-timeout	1
	health-score	0
	dns-ip-primary	
	dns-ip-backup1	
	dns-ip-backup2	
	dns-domain	
	dns-timeout	11
	hip-ip-list	
	ftp-address	
	icmp-address	
	snmp-address	
	telnet-address	
	last-modified-by	admin@console
	last-modified-date	2009-03-12 10:21:39
netwo	rk-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
		200.200.200.202
	gateway	
	sec-gateway	
	gw-heartbeat	
	state	disabled
	heartbeat	0
	retry-count	0

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

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

network-interface	
name	s0p0
sub-port-id	0
description	
hostname	
ip-address	10.160.177.210
pri-utility-addr	10.160.177.211
sec-utility-addr	10.160.177.212
netmask	255.255.255.224
gateway	10.160.177.193
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain dns-timeout	1 1
	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
Tast-mourreu-uale	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 sub-port-id	s0p1 0	
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description hostname 10.160.183.219 ip-address pri-utility-addr 10.160.183.217 sec-utility-addr 10.160.183.218 255.255.255.224 netmask gateway 10.160.183.193 sec-gateway gw-heartbeat state disabled heartbeat 0 retry-count 0 retry-timeout 1 health-score 0 dns-ip-primary dns-ip-backup1 dns-ip-backup2 dns-domain dns-timeout 11 hip-ip-list ftp-address icmp-address snmp-address telnet-address admin@console last-modified-by last-modified-date 2009-05-26 18:01:51 ntp-config server 10.152.6.12 last-modified-by admin@console last-modified-date 2009-03-12 10:20:46 phy-interface s0p0 name operation-type Media port 0 slot \cap virtual-mac 00:08:25:a0:f3:68 admin-state enabled auto-negotiation enabled duplex-mode FULL speed 100 last-modified-by admin@console last-modified-date 2009-05-13 15:29:00 phy-interface name s0p1 operation-type Media 1 port slot Ο virtual-mac 00:08:25:a0:f3:69 admin-state enabled auto-negotiation enabled FULL duplex-mode speed 100 last-modified-by admin@console last-modified-date 2009-05-26 14:51:45

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phy-interface	
name	s1p0
operation-type	Media
port	0
slot	1
virtual-mac	00:08:25:a0:f3:6e
admin-state	enabled
auto-negotiation	enabled
duplex-mode	FULL
speed	100
last-modified-by	admin@console
last-modified-date	2009-05-13 15:29:23
phy-interface	2009 03 13 13.29.23
name	slpl
operation-type	Media
port	1
slot	1
virtual-mac	00:08:25:a0:f3:6f
admin-state	disabled
auto-negotiation	enabled
duplex-mode	FULL
speed	100
last-modified-by	admin@console
last-modified-date	2009-05-13 15:29:37
phy-interface	2003 00 10 10.23.07
name	wancom1
operation-type	Control
port	1
slot	0
virtual-mac	Ĵ
wancom-health-score	8
last-modified-by	admin@console
last-modified-date	2009-03-12 10:21:30
phy-interface	
name	wancom2
operation-type	Control
port	2
slot	0
virtual-mac	-
wancom-health-score	9
last-modified-by	admin@console
last-modified-date	2009-03-12 10:21:30

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

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

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<pre>mm-same-ip mm-in-system bw-cac-non-mm msm-release generate-UDP-checksum max-bandwidth fallback-bandwidth max-priority-bandwidth max-latency max-jitter max-packet-loss observ-window-size parent-realm dns-realm</pre>	enabled enabled disabled disabled 0 0 0 0 0 0 0 0 0
media-policy in-translationid	
out-translationid	
in-manipulationid out-manipulationid	RemoveUPDATE
manipulation-string	NAT_IP
class-profile	
average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	4
maximum-signal-threshold	3000
untrusted-signal-threshold nat-trust-threshold	10 0
deny-period	60
ext-policy-svr	00
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
<pre>icmp-advertisement-interval icmp-target-ip</pre>	0
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
codec-policy	
codec-manip-in-realm constraint-name	disabled
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478

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stun-changed-ip	0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2009-04-22 19:26:23

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

realm-	config	
	identifier	INSIDE-SM
	description	
	addr-prefix	0.0.0.0
	network-interfaces	
		s0p1:0
	mm-in-realm	enabled
	mm-in-network	enabled
	mm-same-ip	enabled
	mm-in-system	enabled
	bw-cac-non-mm	disabled
	msm-release	disabled
	generate-UDP-checksum	disabled
	max-bandwidth	0
	fallback-bandwidth	0
	max-priority-bandwidth	0
	max-latency	0
	max-jitter	0
	max-packet-loss	0
	observ-window-size	0
	parent-realm	
	dns-realm	
	media-policy	
	in-translationid	
	out-translationid	
	in-manipulationid	
	out-manipulationid	
	manipulation-string	
	class-profile	0
	average-rate-limit access-control-trust-level	0 hish
	invalid-signal-threshold	high O
	5	0
	<pre>maximum-signal-threshold untrusted-signal-threshold</pre>	0
	nat-trust-threshold	0
	deny-period	30
	ext-policy-svr	50
	symmetric-latching	disabled
	pai-strip	disabled
	trunk-context	arbabica
	early-media-allow	
	enforcement-profile	
	additional-prefixes	
	restricted-latching	none
	restriction-mask	32

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	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	01/0	
	qos-constraint		
	last-modified-by	admin@consc	
	last-modified-date	2009-05-26	
madum		2009-05-20	13.08.13
reaun	dancy-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 thealth	0	
	media-if-peercheck-time	0	
	_	0	
	peer		ha nni
	name		bc-pri
	state	enabl	
	type	Prima	ary
	destination		
	address		169.254.1.1:9090
	network-interface		wancom1:0
	destination		
	address		169.254.2.1:9090
	network-interface		wancom2:0
	peer		

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

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

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

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

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

session-agent	
hostname	

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

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0 max-sustain-rate max-inbound-sustain-rate 0 max-outbound-sustain-rate 0 5 min-seizures 0 min-asr time-to-resume 0 ttr-no-response 0 in-service-period 0 burst-rate-window 0 sustain-rate-window 0 req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map ping-method OPTIONS ping-interval 300 ping-send-mode keep-alive ping-in-service-response-codes out-service-response-codes media-profiles in-translationid out-translationid trust-me disabled request-uri-headers stop-recurse local-response-map ping-to-user-part ping-from-user-part li-trust-me disabled in-manipulationid out-manipulationid manipulation-string p-asserted-id trunk-group max-register-sustain-rate 0 early-media-allow invalidate-registrations disabled rfc2833-mode none rfc2833-payload 0 codec-policy enforcement-profile refer-call-transfer disabled reuse-connections TCP tcp-keepalive none 10 tcp-reconn-interval 0 max-register-burst-rate register-burst-window 0 last-modified-by admin@console last-modified-date 2009-06-02 18:24:49 **ANNOTATION:** The sip-config defines global sip-parameters, including SIP timers, SIP options, which realm to send requests to if not specified elsewhere, and enabling the SD to collect statistics on requests other than REGISTERs and INVITES.

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

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

sip-interface	
state	enabled
realm-id	OUTSIDE
description	
sip-port	
address	10.160.177.210
port	5060
transport-protocol	UDP
tls-profile	
allow-anonymous	agents-only
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	
manipulation-string	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns per-src-ip-max-incoming-conns	0
inactive-conn-timeout	
	0
untrusted-conn-timeout network-id	0
ext-policy-server default-location-string	
_	2255
charging-vector-mode charging-function-address-mode	pass
ccf-address	Pass
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ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
refer-call-transfer	disabled
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
last-modified-by	admin@console
last-modified-date	2009-05-14 18:46:43

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

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

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0 port-map-start 0 port-map-end 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 disabled implicit-service-route rfc2833-payload 101 rfc2833-mode transparent constraint-name response-map local-response-map ims-aka-feature disabled enforcement-profile refer-call-transfer disabled route-unauthorized-calls tcp-keepalive none add-sdp-invite disabled add-sdp-profiles last-modified-by admin@console last-modified-date 2009-05-26 18:06:22

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

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

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uri-host type replace action match-val-type ip case-sensitive comparison-type match-value new-value \$LOCAL IP header-rule name manipTo header-name то action manipulate comparison-type case-sensitive match-value msg-type request new-value methods element-rule name то parameter-name uri-host type action replace match-val-type ip comp. match-vaiu. new-value comparison-type case-sensitive \$REMOTE IP last-modified-by admin@console 2009-03-12 10:22:14 last-modified-date

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

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

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

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

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

8. General Test Approach and Test Results

The test environment consisted of:

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

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

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

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

9. Verification Steps

9.1. Verification Tests

The following steps may be used to verify the configuration:

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

AVAYA	Avaya Aura System Manager 1.0	Welcome, admin Last Logged on at Jul. 23, 2009 12:53 PM
Furger		Help Log off
Home / Session Manager / Call Ro	uting Test	
▶ Asset Management	Call Routing Test	
▶ User Management	This page allows you to test SIP routing algorithms on Session Manager instances. Enter information about a SIP INVITE to learn how it will be routed	
▶ Monitoring	based on current administration.	
▶ Network Routing Policy	SIP INVITE Parameters	
▶ Security	Called Party URI	Calling Party Address
▶ Applications	sip:000001001@spdevcon.com	10.160.183.219
▶ Settings	Calling Party URI sip:7325551212@10.242.225.200	Session Manager Listen Port
▼ Session Manager	Day Of Week Time (UTC)	Transport Protocol
Session Manager Administration	Thursday V 18:38	TCP
System State Administration	Called Session Manager Instance	Execute Test
Security Module Status	SM1 💌	Language and the second s

Figure 54: Call Routing Test Page

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

 Security Applications Settings Session Manager Session Manager Administration System State Administration Security Module Status Data Replication Status Local Host Name Resolution 	Called Party URI Calling Party Address sip:00001001@spdevcon.com 10.160.183.219 Calling Party URI Session Manager Listen Port sip:7325551212@10.242.225.200 5060 Day Of Week Time (UTC) Transport Protocol Thursday 18:38 TCP ♥		
Maintenance Tests SIP Firewall Configuration SIP Monitoring Tracer Configuration	Routing Decisions Route < sip:31001@spdevcon.com > to SIP Entity Site1 CLAN1 (10.160.179.110). Terminating Location is Site 1.		
Trace Viewer Call Routing Test Managed Bandwidth Usage	Routing Decision Process		
Shortcuts	Originating Location is Main. Using digits < 000001001 > and nost < spdevcon.com > for routing.		
Change Password Help for Call Routing Testing Help for Page Fields			
	NRP Adaptations: DigitConversionAdapter applied. NRP Adaptations: Request-URI set to sip:31001@spdevcon.com Route < sip:31001@spdevcon.com > to SIP Entity Site1 CLAN1 (10.160.179.110), Terminating Location is Site 1.		

Figure 55: Call Routing Test Page – Test Results

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

9.2. Troubleshooting Tools

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

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

10. Conclusion

As illustrated in these Application Notes, Avaya AuraTM Session Manager, Avaya AuraTM 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 AuraTM 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 AuraTM Session Manager AT&T Adaptation Module could be utilized to remove History-Info header information on egress SIP messages to the AT&T IP Toll Free service.

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

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

11. References

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

- [1] Avaya Aura[™] Session Manager Overview, Issue 1, Release 1.1, May 2009, Document Number 03-603323
- [2] *Installing and Administering Avaya Aura™ Session Manager*, Issue 1.1, Release 1.1, June 2009, Document Number 03-603324
- [3] *Administering Avaya Aura*[™] *Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
- [4] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205
- [5] Avaya Aura[™] Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Release 5.2, April 2009, Document Number 07-600780
- [6] Avaya Aura[™] Call Center 5.2 Automatic Call Distribution Reference, Release 5.2, April 2009, Document Number 07-602568
- [7] Modular Messaging MultiSite Guide Release 5.1, June 2009
- [8] Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades, June 2009
- [9] Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades, June 2009
- [10] Modular Messaging for IBM Lotus Domino 5.1 Installation and Upgrades, June 2009

Acme Packet Support (login required):

[11] <u>http://support.acmepacket.com</u>

AT&T IP Toll Free Service Descriptions:

[12] AT&T IP Toll Free

http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/

12. Change History

Issue	Date	Reason
1.0		Initial issue.

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