

Avaya Solution & Interoperability Test Lab

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

## Abstract

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

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

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

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

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# 1. Introduction

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

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# 2. General Test Approach and Test Results

The test environment consisted of:

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

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

<sup>&</sup>lt;sup>1</sup> Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

# 2.1. Interoperability Compliance Testing

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

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

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

## 2.2. Test Results and Known Limitations

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

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

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

## 2.3. Support

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

Avaya customers may obtain documentation and support for Avaya products by visiting <u>http://support.avaya.com</u>. 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.

# 3. Reference Configuration

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

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

<sup>&</sup>lt;sup>2</sup> Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

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

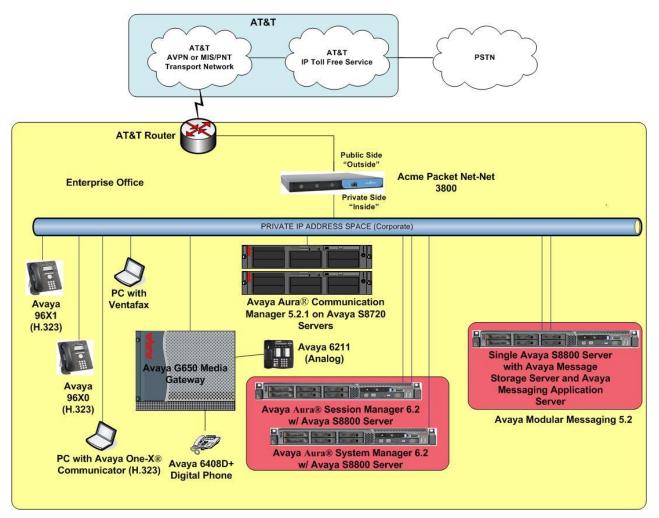


Figure 1: Reference Configuration

# **3.1. Illustrative Configuration Information**

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

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

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

Table 1: Illustrative Values Used in this Reference Configuration

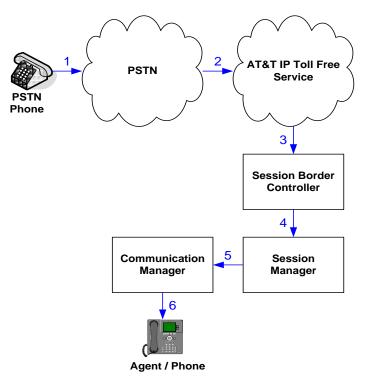
### 3.2. Call Flows

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

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

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

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

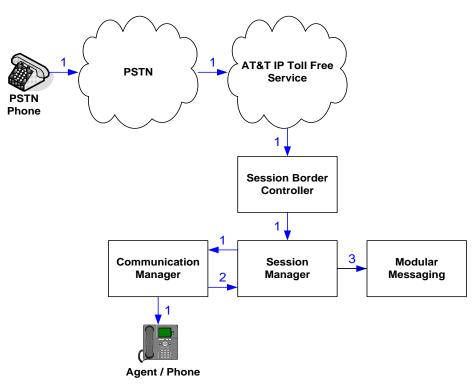


Inbound AT&T IP Toll Free Call to VDN/Agent/Phone

#### 3.2.2. Coverage to Voicemail

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

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



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

# 4. Equipment and Software Validated

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

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

 Table 2: Equipment and Software Versions

# 5. Configure Avaya Aura® Session Manager Release 6.2

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

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

The following administration activities are described:

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

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

Avaya Aura® Sys		ystem Manager 6.2	Help   About   Change Password   Log off ad	
Users		Elements	Services	
directory Groups & Roles Manage groups, to users User Management	nization 's with the enterprise roles and assign roles	B5800 Branch Gateway Manage B5800 Branch Gateway 6.2 elements Communication Manager Manage Communication Manager 5.2 and higher elements Conferencing Manage Conferencing Multimedia Server objects Inventory Manage, discover, and navigate to elements, update element software Meeting Exchange Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging Presence Presence Presence Routing Network Routing Policy Session Manager Element Manager SIP AS 8.1	<ul> <li>Backup and Restore <ul> <li>Backup and restore System Manager database</li> </ul> </li> <li>Buik Import and Export <ul> <li>Manage Buik Import and Export of Users, User Global Settings, Roles, Elements and others</li> </ul> </li> <li>Configurations <ul> <li>Manage system wide configurations</li> </ul> </li> <li>Events <ul> <li>Manage alarms, view and harvest logs</li> </ul> </li> <li>Dicenses <ul> <li>View and configure licenses</li> </ul> </li> <li>Replication <ul> <li>Track data replication nodes, repair replication nodes</li> </ul> </li> <li>Scheduler <ul> <li>Scheduler</li> <li>Schedule, track, cancel, update and delete jobs</li> </ul> </li> <li>Security <ul> <li>Manage Security Certificates</li> <li>Templates</li> <li>Manage Templates for Communication Manager, Messaging System and B5800 Branch Gateway elements</li> </ul> </li> </ul>	

System Manager Home Page

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

αναγα	Avaya Aura® System Manager 6.2 Help   About   Change Password   Log off a
* Routing	Home /Elements / Routing
Domains	
Locations	Introduction to Network Routing Policy
Adaptations	Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.
SIP Entities	The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is a
Entity Links	follows:
Time Ranges	Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
Routing Policies	Step 2: Create "Locations"
Dial Patterns	
Regular Expressions	Step 3: Create "Adaptations"
Defaults	Step 4: Create "SIP Entities"
	- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
	- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
	- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
	Step 5: Create the "Entity Links"
	- Between Session Managers
	- Between Session Managers and "other SIP Entities"
	Step 6: Create "Time Ranges"
	- Align with the tariff information received from the Service Providers
	Step 7: Create "Routing Policies"
	- Assign the appropriate "Routing Destination" and "Time Of Day"
	(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")
	Step 8: Create "Dial Patterns"
	- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"
	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 "Ranking".
	IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of the routing application "Dial patterns". That's this overall routing 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 Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)
	Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

**Network Routing Policy Page** 

# 5.1. SIP Domain

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

AVAYA	Avaya Aura® System Manager 6.2	2			Last Logged on at D Help   About   Change Pa	Jecember 21, 201 assword   Log o	12 12:17 P off admin
						Routing *	Home
<sup>™</sup> Routing	Home / Elements / Routing / Domains						
Domains							Help ?
Locations	Domain Management					Commit	Cancel
Adaptations	Warning: SIP Domain name change will cause login failure for Communic	ation Address handles with this domain. Consult rel-	ease notes or Support for	steps to reset login credentials.			
SIP Entities							
Entity Links	1 Item   Refresh					Filton	: Enable
Time Ranges						Filter.	chable
Routing Policies	Name	Туре	Default	Notes			
Dial Patterns	* attavaya.com	sip 👻		Domain for ATT Testing			
Regular Expressions							
Defaults	* Input Required					Commit	Cancel

**SIP Domains** 

#### 5.2. Locations

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

AVAYA	Avaya Aura® System Manager 6.2		Help   About   Change Password   Log off admin		
			Routing * Home		
The Routing	Home /Elements / Routing / Locations				
Domains			Help ?		
Locations	Location Details		Commit Cancel		
Adaptations					
SIP Entities	General				
Entity Links	* Name: Se	ssionManager			
Time Ranges	Notes: Se	ssion Manager			

**Session Manager Location Details** 

AVAYA	Avaya Aura® System Manager 6.2			Last Logged on at December 21, 2012 12:17 PP Help   About   Change Password   Log off admin
•				Routing × Home
* Routing	Home / Elements / Routing / Locations			
Domains				Help ?
Locations	Location Details			Commit Cancel
Adaptations				
SIP Entities	General			
Entity Links	* Name:	Location_130		
Time Ranges	Notes:	Subnet 130		
Routing Policies				
Dial Patterns	Overall Managed Bandwidth			
Regular Expressions	Managed Bandwidth Units:	Kbit/sec 💌		
Defaults	Total Bandwidth:			
	Multimedia Bandwidth:			
	Audio Calls Can Take Multimedia Bandwidth:	<b>V</b>		
	Per-Call Bandwidth Parameters			
	Maximum Multimedia Bandwidth (Intra-Location):	1000 Kbit/Sec		
	Maximum Multimedia Bandwidth (Inter-Location):	1000 Kbit/Sec		
	* Minimum Multimedia Bandwidth:	64 Kbit/Sec		
	* Default Audio Bandwidth:	80 Kbit/sec 📼		
	Alarm Threshold			
	Overall Alarm Threshold:	80 💌 %		
	Multimedia Alarm Threshold:	80 💌 %		
	* Latency before Overall Alarm Trigger:	5 Minutes		
	* Latency before Multimedia Alarm Trigger:	5 Minutes		
	Location Pattern			
	Add Remove			
	1 Item   Refresh			Filter: Enable
	IP Address Pattern		Notes	
	10.80.130.*			

Acme Packet SBC Location Details

Location\_130 shown below serves Communication Manager, Modular Messaging and other elements on the CPE.

	Avaya Aura® System Manag	Help   About	Change Password   Log off adm	
				Routing * Hom
Routing	Home / Elements / Routing / Locations			
Domains				Help
Locations	Location Details			Commit Cance
Adaptations				
SIP Entities	General			
Entity Links	* Name:	Location_130		
Time Ranges	Notes:	Subnet 130		
Routing Policies				
Dial Patterns	Overall Managed Bandwidth			
Regular Expressions		1/1-1-1		
Defaults	Managed Bandwidth Units:	Kbit/sec 🔻		
	Total Bandwidth:			
	Multimedia Bandwidth:			
	Audio Calls Can Take Multimedia Bandwidth:			
	Per-Call Bandwidth Parameters			
	Maximum Multimedia Bandwidth (Intra-Location):	1000 Kbit/Sec		
	Maximum Multimedia Bandwidth (Inter-Location):	1000 Kbit/Sec		
	* Minimum Multimedia Bandwidth:	64 Kbit/Sec		
	* Default Audio Bandwidth:	80 Kbit/sec 🔻		
	Alarm Threshold			
	Overall Alarm Threshold:	80 - %		
	Multimedia Alarm Threshold:	80 <b>~ %</b>		
	* Latency before Overall Alarm Trigger:	5 Minutes		
	* Latency before Multimedia Alarm Trigger:	5 Minutes		
	Location Pattern			
	Add Remove			
	1 Item   Refresh			Filter: Enable
	IP Address Pattern		Notes	
	* 10.80.130.*			

Subnet 130 Location Details

## **5.3. Configure Adaptations**

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

AVAYA	Avaya Aura® System Manager 6.2	Help   About   Change Password   Log off admi
-		Routing * Home
Routing	Home /Elements / Routing / Adaptations	
Domains		Help
Locations	Adaptation Details	Commit Cancel
Adaptations		
SIP Entities	General	
Entity Links	* Adaptation name: AT&T Adaptations	
Time Ranges	Module name: AttAdapter	<b>~</b>
Routing Policies	Module parameter: fromto=true iodstd=att	tavaya.com
Dial Patterns	Egress URI Parameters:	· · · · · · · · · · · · · · · · · · ·
Regular Expressions		
Defaults	Notes:	
	Digit Conversion for Incoming Calls to SM	
	Add Remove	
	0 Items   Refresh	Filter: Enable
	Matching Pattern Min Max Phone Context Delete Digit	its Insert Digits Address to modify Adaptation Data Notes
	Digit Conversion for Outgoing Calls from SM	

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

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

				unuge	r 6.2			Help   About   C	hange Password   Log	off admin
									Routing	× Home
Routing Home	/ Elements / Rout	ting / Ad	laptatio	ns						
Domains										Help ?
Locations Adapta	ation Details								Com	mit Cancel
Adaptations										
SIP Entities Gene	ral									
Entity Links		*	Adaptatio	on name: 🛛	ATT_CLAN	102				
Time Ranges			Modu	le name: 1	DigitConve	ersionAdapter 💌				
Routing Policies		M	lodule pa	rameter: f	romto=tri	ue osrcd=attava	va.com			
Dial Patterns			URI Par				Jereom			
Regular Expressions		cyress								
Defaults				Notes:						
Digit	Conversion for I	ncomir	ng Calls	to SM						
Add	Remove									
0 Item	IS Refresh									lter: Enable
	Matching Pattern	Min	Max	Phone Con	ntext	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
Digit Add	Conversion for ( Remove	Dutgoin	g Calls	from SM						
5 Item	s Refresh			Dhaw	D-L 1	- 1			Fil	lter: Enable
	Matching Pattern	Min	Max	Phone Context	Delete Digits		Address to modify	Adaptation Data	Notes	
	* 000004153571057	* 15	* 15		* 15	2010	destination 🖵		To Select Skill	
	* 000004153581058	* 15	* 15		* 15	2011	destination 👻		To Skill 11	
	* 000004153591059	* 15	* 15		* 15	2012	destination 💌		To Skill 12	
	* 000004153601060 * 000004153611061	* 15 * 15	* 15		* 15	2013 53004	destination -		To Skill 12 Secondary ADR	

Adaptation for calls between Session Manager and Communication Manager

#### 5.4. SIP Entities

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

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

Ανάγα	Avaya Aura®	System Manag	ger 6.2	Help   Abo	ut   Change Password   <b>Log off adm</b> i
•					Routing × Home
Routing	Home / Elements / Rou	uting / SIP Entities			
Domains					Help
Locations	SIP Entity Details				Commit Cance
Adaptations	General				
SIP Entities	General	* Name	: DenverSM		
Entity Links					
Time Ranges		* FQDN or IP Address			
Routing Policies		Туре	Session Manager 🔍		
Dial Patterns		Note	Session Manager		
Regular Expressions					
Defaults			: SessionManager 👻		
		Outbound Proxy	r: 🔹		
		Time Zone	America/Denver	•	
	Entity Links Add Remove				
	2 Items   Refresh				Filter: Enab
		Protocol Port	SIP Entity 2	Port	Connection Policy
		TCP 💌 * 5060 TCP 💌 * 5060	AcmeSBCATT-5060  CM5.2CLAN1A02	* 5060 * 5060	Trusted 💌
	Select : All, None				
	Port TCP Failover port: TLS Failover port: Add Remove				
	2 Items Refresh		e h.z		Filter: Enab
	<b>Port</b> 5060		tavaya.com 💌	Notes	7
	5061		tavaya.com		 ]
	Select : All, None				
	SIP Responses to an Add Remove	n OPTIONS Request			

**Session Manager Entity** 

AVAYA	Avaya Aura® System Manag	jer 6.2	Help   About	Change Password   Log off admi
				Routing × Home
Routing	Home /Elements / Routing / SIP Entities			
Domains				Help
Locations	SIP Entity Details			Commit Cancel
Adaptations	General			
SIP Entities		AcmeSBCATT-5060		
Entity Links				
Time Ranges	* FQDN or IP Address:	10.80.130.250		
Routing Policies	Туре:	Other 👻		
Dial Patterns	Notes:	Acme SBC to ATT		
Regular Expressions				
Defaults	Adaptation:	AT&T Adaptations -		
	Location:	Acme_SBC_130 -		
	Time Zone:	America/Denver 👻		
	Override Port & Transport with DNS SRV:			
	* SIP Timer B/F (in seconds):	4		
	Credential name:			
	Call Detail Recording:			
	CommProfile Type Preference:	-		
	SIP Link Monitoring			
	-	Use Session Manager Configuration 👻		
	Supports Call Admission Control:			
	Shared Bandwidth Manager:			
	Primary Session Manager Bandwidth Association:	<b></b>		
	Backup Session Manager Bandwidth Association:	T		
	Entity Links			
	Add Remove			
				-
	1 Item   Refresh			Filter: Enable
	SIP Entity 1 Protocol Port	SIP Entity 2	Port	Connection Policy

Acme Packet SBC Entity

AVAVA	Avaya Aura® System Manage	r 6.2	lelp   About   Change Password   Log off admin
			Routing × Home
Routing	Home / Elements / Routing / SIP Entities		
Domains			Help
Locations	SIP Entity Details		Commit Cance
Adaptations	General		
SIP Entities		CM5.2CLAN1A02	
Entity Links			
Time Ranges	* FQDN or IP Address:		
Routing Policies	Туре:		
Dial Patterns	Notes:	ntity to CM Trunk	
Regular Expressions	Adaptation:		
Defaults			
		Location_130	
		America/Denver	
	Override Port & Transport with DNS SRV:		
	* SIP Timer B/F (in seconds):	ł	
	Credential name:		
	Call Detail Recording:	none 💌	
	SIP Link Monitoring	Jse Session Manager Configuration 💌	
	STE Link Plointoining.		
	Supports Call Admission Control:	3	
	Shared Bandwidth Manager:		
	Primary Session Manager Bandwidth Association:	<b>v</b>	
	Backup Session Manager Bandwidth Association:	<b>*</b>	
	Entity Links Add Remove		
	1 Item Refresh		Filter: Enab
	SIP Entity 1 Protocol Port	SIP Entity 2 Port	Connection Policy
	DenverSM TCP 💌 * 5060	CM5.2CLAN1A02 💌 * 5060	Trusted

**Communication Manager Entity** 

<i>Α</i> \V <i>Α</i> \Y <i>Α</i> \	Avaya Aura® System Mana	ger 6.2			Help   About   Change Pa	ssword   Log	off admin
•						Routing	Home
Routing	Home / Elements / Routing / SIP Entities					-	
Domains							Help ?
Locations	SIP Entity Details					Comm	nit Cancel
Adaptations	General						
SIP Entities		e: MM52					
Entity Links							
Time Ranges	* FQDN or IP Addre						
Routing Policies		Modular Messaging	V				
Dial Patterns	Not	25:					
Regular Expressions	a						
Defaults	Adaptati		<b>•</b>				
		on: Location_130					
		America/Denver					
	Override Port & Transport with DNS S						
	* SIP Timer B/F (in second	5): 4					
	Credential nam	1e:					
	Call Detail Recordi	ng: none 💌					
	SIP Link Monitoring SIP Link Monitorin	ng: Use Session Manag	er Configural	tion 💌			
	Supports Call Admission Contr	ol:					
	Shared Bandwidth Manag	er:					
	Primary Session Manager Bandwidth Association	on: 👻					
	Backup Session Manager Bandwidth Associatio	on:					
	Entity Links						
	Add Remove						
	1 Item Refresh						ter: Enable
	SIP Entity 1 Protocol Port DenverSM TCP + 5060	SIP Entity	2	* 5060	Connec	tion Policy	

Modular Messaging Entity

## 5.5. Entity Links

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

avaya	Avaya Aura	<sup>®</sup> System M	lanager	6.2			Help   About   Change	Password   Log off admin
								Routing * Home
Routing	Home / Elements / Re	outing / Entity Lin	ks					
Domains								Help ?
Locations	Entity Links							Commit Cancel
Adaptations								
SIP Entities								
Entity Links	1 Item   Refresh							Filter: Enable
Time Ranges	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
rine Ranges	* SM-CM5.2CLAN1A02	* DenverSM 👻	TCP 👻	* 5060	* CM5.2CLAN1A02	\$ \$ 5060	Trusted	To CLAN1A02 trunk

Entity link between Session Manager and Communication Manager

AVAYA	Avaya Aura® System Manager 6.						Help   A	bout   Change Pa	assword   Log	off admin
									Routing	Home
Routing	4	Home /Elements / Rou	uting / Entity Links							
Domains										Help ?
Locations		Entity Links							Commit	Cancel
Adaptations										
SIP Entities										
Entity Links										
Time Ranges		1 Item   Refresh							Filter	Enable
Routing Policies		Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes	
Dial Patterns		* SM-AcmeSBCATT-TCF	* DenverSM 👻	TCP 🔻	* 5060	* AcmeSBCATT-5060 🔻	* 5060	Trusted 🔹	To ATT Acme	SBC

Entity link between Session Manager and Acme Packet SBC

AVAYA	Avaya Aura	Avaya Aura® System Manager 6.2					er 6.2 Help   About   Change Password					
Routing	Home / Elements / Re	outing / Entity Link	s									
Domains										Help ?		
Locations	Entity Links								C	ommit Cancel		
Adaptations												
SIP Entities												
Entity Links	1 Item   Refresh			1						Filter: Enable		
Time Ranges	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Por	rt	Connection Policy	Notes			
	* DenverSM_ToMM_506	* DenverSM 💌	TCP 💌	* 5060	* MM52	▼ * 50	060	Trusted 💌				

Entity link between Session Manager and Modular Messaging

# 5.6. Time Ranges

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

AVAYA	Avaya Aura® System Manager 6.2	Help   About   Change Password   Log off admir
		Routing × Home
Routing	Home /Elements / Routing / Time Ranges	
Domains		Help ?
Locations	Time Ranges	
Adaptations	Edit New Duplicate Delete More Actions •	
SIP Entities	Edit New Dupicate Delete More Actions	
Entity Links	1 Item   Refresh	Filter: Enable
Time Ranges		Filter: Enable
Routing Policies	Name Mo Tu We Th Fr Sa Su	Start Time End Time Notes
Dial Dattorns		00:00 23:59 Time Range 24/7

**Time Range** 

# 5.7. Routing Policies

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

Ανάγα	Avaya Aura® Sy	stem Ma	nager	6.2					Help	About   Change Pa	ssword   I	.og off adm
•											Routing	* Hom
Routing	Home / Elements / Routing	/ Routing Pol	cies									
Domains												Help
Locations	Routing Policy Details										Co	ommit Canc
Adaptations											_	
SIP Entities	General											
Entity Links		*	Name: To	CM5.2CL4	N1A02							
Time Ranges		Di	sabled: 📃									
Routing Policies		* 1	Retries: 0									
Dial Patterns			Notes: To	a CM tru	nk aroup							
Regular Expressions												
Defaults	SIP Entity as Destinatio	n										
	Select											
	Jelett											
	Name	-	N or IP Add	ress				Туре		Notes		
	CM5.2CLAN1A02	10.8	0.130.204					CM	E	Entity to CM Trunk		
	Time of Day											
	Add Remove View Gaps/C	worland										
		wenaps										
	1 Item Refresh	e 2 Mo	n Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes	Filter: Enab
	0 24/7	.u		<b>V</b>	1	1	<b>V</b>	7	00:00	23:59	Time Ra	24/7
	24/7								00.00	23:39	Time Ka	ige 24/7
	Select : All, None											
	Dial Patterns											
	Add Remove											
	1 Item Refresh											Filter: Enab
	Pattern Min	Max	Eme	ergency C	all	SIP	Domain		Originatin	g Location		Notes
	00000 9	15				attav	ava.com		Acme SBC	130		

**Routing Policy to Communication Manager** 

AVAVA	Avaya /	Aura® Sys	tem Man	ager	6.2					Help   Abo	ut   Change Pa	ssword   L	og off	admin
												Routing	×	Home
Routing	Home / Eleme	nts / Routing /	Routing Policie	25										
Domains														Help ?
Locations	Routing Policy D	etails										Co	mmit	
Adaptations														
SIP Entities	General													
Entity Links			* Na	me: To	MM									
Time Ranges			Disal	oled: 🔳										
<b>Routing Policies</b>			* Ret	ries: 0										
Dial Patterns			No	otes:										
Regular Expressions														
Defaults	SIP Entity as	Destination												
	Select Name MM52 Time of Day Add Removel		<b>QDN or IP Addre</b>	55						Type Modular Messagin	g	25		
		view Gaps/Ove	inaps											
	1 Item Refresh	1 Name	2 Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes	Filter: I	Enable
		24/7		1	7	1	1	<b>V</b>	1	00:00	23:59	Time Rar	nge 24/7	7
	Select : All, None													
	Dial Patterns													
	1 Item   Refresh												Filter	Feeb!-
	Pattern	🔺 Min	Max	E	mergency	Call	SI	o Domaii	n	Originating L	ocation		Filter: I	
	59999	5	5				atta	avaya.cor	m	Location_130				

**Routing Policy to Modular Messaging** 

### 5.8. Dial Patterns

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

AVAVA	Avaya Aura® System	Manager 6	.2		Help	About   Change Pas	sword   Log o	ff admi
							Routing *	Home
Routing	Home / Elements / Routing / Dial Pa	ntterns						
Domains								Help
Locations	Dial Pattern Details						Commi	Cance
Adaptations								
SIP Entities	General							
Entity Links		* Pattern: 0000	)					
Time Ranges		* Min: 9						
Routing Policies		* Max: 15						
Dial Patterns	Em	ergency Call: 📃						
<b>Regular Expressions</b>	Emerge	ency Priority: 1						
Defaults	Eme	rgency Type:						
		SIP Domain: attav	ava.com 🚽					
		Notes:			-			
		Notes.						
	Originating Locations and Routir	ng Policies						
	Add Remove	-						
	1 Item Refresh						Filte	r: Enabl
	Originating Location Name 1	Originating Location Note <del>s</del>	Routing Policy Name	Rank 2 🔔	Routing Policy Disabled	Routing Policy Destination	Routing Notes	g Policy
	Acme_SBC_130	SBC To ATT	ToCM5.2CLAN1A02	0		CM5.2CLAN1A02	To CM t group	runk

Dial Pattern for Inbound Calls to Communication Manager

AVAVA	Avaya Aura® System Manager 6.2		Help	About   Change Pass	sword   Log o	off admin
-					Routing *	Home
Routing	Home / Elements / Routing / Dial Patterns					
Domains						Help ?
Locations	Dial Pattern Details				Comm	t Cancel
Adaptations						
SIP Entities	General					
Entity Links	* Pattern: 59999					
Time Ranges	* Min: 5					
Routing Policies	* Max: 5					
Dial Patterns	Emergency Call:					
Regular Expressions	Emergency Priority: 1					
Defaults	Emergency Type:					
	SIP Domain: attavaya.com					
	Notes:					
	Originating Locations and Routing Policies					
	Add Remove					
	1 Item   Refresh				Filte	er: Enable
	Originating Location Name 1         Originating Location Notes         Routing Policy Name         Rank	2	Routing Policy Disabled	Routing Policy Destination	Routing Notes	Policy
	Location_130 Subnet 130 ToMM 0			MM52		

**Dial Pattern for Covered Calls to Modular Messaging** 

# 5.9. Session Manager Administration

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

AVAYA	Avaya Aura® System Manag	jer 6.2	Help   About   Change Pass	word   Log off admin
-			Session Manager ×	Routing × Home
▼ Session Manager	Home /Elements / Session Manager / Session I	Manager Administration		
Dashboard				Help ?
Session Manager	Miner Consider Manager			
Administration	View Session Manager			Return
Communication Profile Editor	General   Security Module   NIC Bonding   Monitoring   Expand All   Collapse All	CDR   Personal Profile Manager (PPM) - Connection	Settings   Event Server	
Network Configuration	General 💌			
Device and Location	General			
Configuration	SIP Entity Name	DenverSM		
Application		Session Manager		
Configuration	Management Access Point Host Name/IP	10.80.150.210		
> System Status	Direct Routing to Endpoints	Enable		
> System Tools				
Performance	Security Module 💌			
	SIP Entity IP Address	10.64.19.210		
	Network Mask	255.255.255.0		
	Default Gateway	10.64.19.1		
	Call Control PHB	46		
	QOS Priority	6		
	Speed & Duplex	Auto		
	VLAN ID			

View Session Manager (DenverSM)

# 6. Configure Avaya Aura® Communication Manager 5.2.1

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

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

# 6.1. System Parameters

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

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

display system-parameters customer-options		Page	<b>2</b> of	11
OPTIONAL FEATURES				
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:	8000	0		
Maximum Concurrently Registered IP Stations:	18000	4		
Maximum Administered Remote Office Trunks:	0	0		
Maximum Concurrently Registered Remote Office Stations:	0	0		
Maximum Concurrently Registered IP eCons:	0	0		
Max Concur Registered Unauthenticated H.323 Stations:	0	0		
Maximum Video Capable H.323 Stations:	0	0		
Maximum Video Capable IP Softphones:	0	0		
Maximum Administered SIP Trunks:	5000	250		
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0		
Maximum Number of DS1 Boards with Echo Cancellation:	0	0		
Maximum TN2501 VAL Boards:	10	1		
Maximum Media Gateway VAL Sources:	0	0		
Maximum TN2602 Boards with 80 VoIP Channels:	128	0		
Maximum TN2602 Boards with 320 VoIP Channels:	128	2		
Maximum Number of Expanded Meet-me Conference Ports:	0	0		
(NOTE: You must logoff & login to effect the pe	rmissi	on char	nges.)	

2. On **Page 4** of the **system-parameters customer-options** and verify that the **IP Trunks** field in is set to **y**.

```
display system-parameters customer-options
                                                                Page
                                                                       4 of
                                                                             11
                                 OPTIONAL FEATURES
  Emergency Access to Attendant? y
                                                                 IP Stations? y
          Enable 'dadmin' Login? y
           Enhanced Conferencing? y
                                                          ISDN Feature Plus? v
                                     ISDN/SIP Network Call Redirection? n
                  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
```

## 6.2. Dial Plan

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

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

change dial	plan ana	alysis					Page	<b>1</b> of 1	12
			DIAL PLA	N ANALYS	IS TAB	LE			
			Lo	cation:	all	Pe	ercent E	ull: 1	
Dialed	Total	Call	Dialed	Total	Call	Dialed	Total	Call	
String	Length	n Type	String	Length	Туре	String	Length	п Туре	
1	3	dac							
5	5	ext							
8	1	fac							
9	1	fac							

#### 6.3. IP Node Names

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

change node-names	ip				Page	<b>1</b> of	2
		ΙP	NODE	NAMES			
Name	IP Address						
Gateway001	10.80.130.1						
CLAN-1A02	10.80.130.204						
SM62	10.64.19.210						

#### 6.4. IP Codec Parameters

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

```
change ip-codec-set 2
                                                                         2
                                                           Page
                                                                  1 of
                        IP Codec Set
  Codec Set: 2
  Audio
               Silence
                            Frames
                                     Packet
  Codec
               Suppression Per Pkt Size(ms)
1: G.729B
                    n
                             3
                                       30
2: G.729A
                              3
                    n
                                       30
3: G.711MU
                              3
                                       30
                    n
```

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

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

### 6.5. IP Network Regions

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

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

```
change ip-network-region 2
                                                                    1 of
                                                                          19
                                                             Page
                              IP NETWORK REGION
 Region: 1
Location:
                Authoritative Domain: attavaya.com
   Name: ATT Calls
MEDIA PARAMETERS
                               Intra-region IP-IP Direct Audio: yes
     Codec Set: 2
                               Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 16384
                                           IP Audio Hairpinning? y
  UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
                                         RTCP Reporting Enabled? y
Call Control PHB Value: 46
                              RTCP MONITOR SERVER PARAMETERS
       Audio PHB Value: 46
                                 Use Default Server Parameters? v
       Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5
                                     AUDIO RESOURCE RESERVATION PARAMETERS
```

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

```
change ip-network-region 2
                                                                       19
                                                                 3 of
                                                           Page
Source Region: 2 Inter Network Region Connection Management
                                                                Т
                                                                        М
                                                                G A
                                                                        e
dst codec direct WAN-BW-limits Video
                                                          Dyn A G
                                            Intervening
                                                                        а
           WAN Units Total Norm Prio Shr Regions
                                                           CAC R L
ran set
                                                                        S
1
     2
2
3
```

#### 6.6. SIP Trunks

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

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

```
add signaling-group 2
                                                             Page
                                                                    1 of
                                                                           1
                               SIGNALING GROUP
Group Number: 2
                             Group Type: sip
                       Transport Method: tcp
 IMS Enabled? n
Near-end Node Name: CLAN 1A02
                                            Far-end Node Name: SM62
Near-end Listen Port: 5060
                                          Far-end Listen Port: 5060
                                       Far-end Network Region: 2
Far-end Domain: attavaya.com
                                           Bypass If IP Threshold Exceeded? n
                                                    RFC 3389 Comfort Noise? n
Incoming Dialog Loopbacks: eliminate
                                            Direct IP-IP Audio Connections? y
        DTMF over IP: rtp-payload
Session Establishment Timer(min): 3
                                                      IP Audio Hairpinning? n
        Enable Layer 3 Test? y
                                                  Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n
                                            Alternate Route Timer(sec): 6
```

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

add trunk-grou	ıp 2			Pa	. <b>ge 1</b> o:	E 21
		TRUNK GROUP				
Group Number:	2	Group Type:	sip	CDR	Reports	: V
Group Name:	ATT	COR:	_	TN: 1	TAC	: 102
Direction:	incoming	Outgoing Display?	n			
Dial Access?	n		Nigh	nt Service:		
Queue Length:	0					
Service Type:	public-ntwrk	Auth Code?	n			
			N	Signaling Number of M	-	

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

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

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

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

- 5. On Page 4 of the trunk-group form:
- Set **Support Request History?** field to y.
- Set **Telephone Event Payload Type** field to the RTP payload type required by the AT&T IP Toll Free service (e.g. **100**).

```
add trunk-group 2Page4 of21PROTOCOL VARIATIONSMark Users as Phone? nPrepend '+' to Calling Number? nSend Transferring Party Information? nNetwork Call Redirection? nSend Diversion Header? nSupport Request History? yTelephone Event Payload Type: 100
```

#### 6.7. Public Unknown Numbering

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

char	nge public-unkr	nown-number	ring O		Pa	ge 1	of	2
		NUMBER	RING - PUBLIC/UN	KNOWN FO	RMAT			
				Total				
Ext	Ext	Trk	CPN	CPN				
Len	Code	Grp(s)	Prefix	Len				
5		2		5	Total Admin	istered	l: 3	
5	50002	3	7323680194	10	Maximum 1	Entries	: 99	999

## 6.8. Alternate Automated Routing (AAR) Table

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

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

change aar analysis 0							Page	1 of	2
		AA	R DIGIT A	NALYSIS	TABLE				
			Locat	ion: all	L		Percent Fu	ıll: 1	
		- 1	Dente	0-11	<b>N</b> - 1 -				
Dialed	Tot	aı	Route	Call	Node	ANI			
String	Min	Max	Pattern	Type	Num	Reqd			
59999	5	5	2	unku		n			

### 6.9. Route Pattern

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

- Grp No Set to 2 i.e. the trunk group configured for Local Access.
- **FRL** Set to  $\mathbf{0}$  (zero).

change route-	-pattern 2				Page	<b>1</b> of	3
2	-	n Number:	2 Pattern Name	: ToMM	-		
		SCCAN?	n Secure SI	?? n			
Grp FRL N	NPA Pfx Hop To	ll No. Ir	nserted			DCS/	IXC
No	Mrk Lmt Li		lgits			QSIG	
		Dgts				Intw	
1:20						n	user
2:						n	user
3:						n	user
4:						n	user
5:						n	user
6:						n	user
		ITC BCIE	Service/Feature			2	LAR
012M4W	V Request			-	Format		
				Subaddr	ess		
1: УУУУУ	•	rest				1	none
2: ууууу	ynn	rest				1	none
3: ууууу	ynn	rest				]	none
4: y y y y y	ynn	rest				]	none
5: y y y y y	ynn	rest				]	none
6:ууууу	ynn	rest				1	none

# 6.10. Optional Features

# 6.10.1. Call Center Provisioning

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

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

display system-parameters customer-op	tions Po	age 6 of 11							
CALL CENTER									
Call Center	Release: 5.0								
ACD?	-								
BCMS (Basic)?	y Service Level Maximi	zer? n							
BCMS/VuStats Service Level?	y Service Observing (Bas.	LC)? n							
BSR Local Treatment for IP & ISDN?	n Service Observing (Remote/By Fi	AC)? n							
Business Advocate?	n Service Observing (VD	Js)? n							
Call Work Codes?	n Timed 2	ACW? n							
DTMF Feedback Signals For VRU?	n Vectoring (Bas	ic)?y							
Dynamic Advocate?	n Vectoring (Prompting	ng)? y							
Expert Agent Selection (EAS)?	y Vectoring (G3V4 Enhance	ed)? y							
EAS-PHD?	y Vectoring (3.0 Enhance	ed)? y							
Forced ACD Calls?	n Vectoring (ANI/II-Digits Routi:	ng)? y							
Least Occupied Agent?	n Vectoring (G3V4 Advanced Routi	ng)? y							
Lookahead Interflow (LAI)?	n Vectoring (CIN	70)? n							
Multiple Call Handling (On Request)?	n Vectoring (Best Service Routi:	ng)? n							
Multiple Call Handling (Forced)?	n Vectoring (Holida	ys)? n							
PASTE (Display PBX Data on Phone)?	n Vectoring (Variable	es)? n							
(NOTE: You must logoff & logi	n to effect the permission changes.)								

**Call Center Optional Features Form** 

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

display agent-loginID 53001	1	Page	<b>1</b> of	2
	AGENT	LOGINID		
Login ID: 6	6665611		AAS?	n
Name: A	Agent1		AUDIX?	n
TN: 1	1	LWC F	Reception:	spe
COR: 1	1	LWC Log Exterr	al Calls?	n
Coverage Path: 2	2	AUDIX Name for M	lessaging:	
Security Code:				
		LoginID for ISDN/SIE	<pre>P Display?</pre>	n
			Password:	
		Password (ente	er again):	
		Aut	co Answer:	
station				
		MIA Acros	ss Skills:	system
		ACW Agent Conside	ered Idle:	system
		Aux Work Reason (	Code Type:	system
		Logout Reason (		-
Maxi	imum time age	ent in ACW before logo	out (sec):	system
		Forced Agent Loc	gout Time:	:
WARNING: Agent must ]	log in again	before changes take e	effect	

#### Agent Form – Page 1

display agent-loginID	Pa	age 2 o	f 2			
AGENT LOGINID						
Direct Agent Skill:				Service Objective? n		
Call Handling Preference: skill-level				Local Call Preference? n		
SN RL SL	SN F	RL SL	SN	RL SL	SN	RL SL
1: <b>11 1</b> 1	.6 <b>:</b>		31:		46:	
2: 1	7:		32:		47:	
3: 1	.8:		33:		48:	

Agent Form – Page 2

display hunt-group 11		Page	<b>1</b> of 3
	HUNT	GROUP	
Group Number:	11	ACD3	? у
Group Name:	Skill-11	Queue?	У
Group Extension:	666-5711	Vector?	У
Group Type:	ead-mia		
TN:	1		
COR:	1	MM Early Answer?	n
Security Code:		Local Agent Preference?	n
ISDN/SIP Caller Display:			
Queue Limit:	unlimited		
Calls Warning Threshold:	Port:		
Time Warning Threshold:	Port:		

#### Skill (Hunt Group) Form – Page 1

display hunt-group 11	Page 2 of 3
	HUNT GROUP
<b>Skill? y</b> AAS? n Measured: none Supervisor Extension:	Expected Call Handling Time (sec): 180
Controlling Adjunct: none	
Interruptible Aux Threshold: none	
	Redirect on No Answer (rings):
	Redirect to VDN:
Forced Entry o	of Stroke Counts or Call Work Codes? n

#### Skill (Hunt Group) Form – Page 2

display vdn 2010	Page 1 of 3
VECTOR DIREC	CTORY NUMBER
	666-5310 To SelectSkill Vector Number 10
Meet-me Conferencing? Allow VDN Override? COR: TN#: Measured:	n 1 1
VDN of Origin Annc. Extension*: 1st Skill*: 2nd Skill*: 3rd Skill*: * Follows VDN override rules	

#### VDN (Vector Directory Number) Form

display vector	10	Page 1 of 6
	CALL VECTOR	
Number: 10	Name: RouteToSkill	
	Meet-me Conf? n	Lock? n
Basic? y	EAS? n G3V4 Enhanced? y ANI/II-Digits? y	ASAI Routing? y
Prompting? y	LAI? n G3V4 Adv Route? n CINFO? n BSR? n	Holidays? n
Variables? n	3.0 Enhanced? n	
01 wait-time	2 secs hearing ringback	
02 collect	1 digits after announcement 33002 for no	one
03 goto vector		
04 goto vector	12 @step 2 if digits = 2	
05 goto vector	13 @step 2 if digits = 3	
06		

Vector (RouteToSkill) Form

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

Vector (Skill 11) Form

### 6.10.2. Modular Messaging Coverage Path and Hunt Group

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

display coverage path 1			Page 1 of 1		
	COVERAGE	PATH			
Coverag	e Path Number: 1				
Cvg Enabled for VDN R	oute-To Party? n	Hunt	after Coverage? n		
Nex	t Path Number:	Linka	lge		
COVERAGE CRITERIA					
Station/Group Status	Inside Call	Outside Cal	.1		
Active?	n	n			
Busy?	У	У			
Don't Answer?	У	У	Number of Rings: 4		
All?	n	n			
DND/SAC/Goto Cover?	У	У			
Holiday Coverage?	n	n			
COVERAGE POINTS					
Terminate to Coverage Pts. with Bridged Appearances? n					
Point1: h1 R	ng: 4 Point2:				
Point3:	Point4:				
Point5:	Point6:				

Coverage	Path	Form
Coverage	I uuii	I UI III

display hunt-group 1			Page	<b>1</b> of	60
		HUNT GROUP			
Group Number:	1	ACD?	n		
Group Name:	MM	Queue?	n		
Group Extension:	59999	Vector?	n		
Group Type:	ucd-mia	Coverage Path:			
TN:	1	Night Service Destination:			
COR:	1	MM Early Answer?	n		
Security Code:		Local Agent Preference?	n		
ISDN/SIP Caller Display:	mbr-name	e			

#### Hunt Group Form – Page 1

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

Hunt Group Form – Page 2

```
display station 50001
                                                                       1 of
                                                                              5
                                                               Page
                                      STATION
Extension: 50001
                                                                          BCC: 0
                                          Lock Messages? n
                                       Security Code: 123456
Coverage Path 1: 1
Coverage Path 2:
    Type: 9620
                                                                          TN: 1
    Port: S00000
                                                                          COR: 1
    Name: H323-96XX-50001
                                                                         COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19 Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 6665011
       Speakerphone: 2-way
Display Language: english
                                             Mute Button Enabled? y
                                                   Button Modules: 0
Survivable GK Node Name:
         Survivable COR: internal
                                                Media Complex Ext:
   Survivable Trunk Dest? y
                                                      IP SoftPhone? n
                                                          IP Video? n
                              Short/Prefixed Registration Allowed: default
                                               Customizable Labels? y
```

### **Station Form**

### 6.11. Saving Translations

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

# 7. Avaya Modular Messaging

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

# 8. Configure Acme Packet Session Border Controller (SBC)

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

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

local-policy

oney	
from-address	
	*
to-address	
	*
source-realm	
	Enterprise
description	
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
policy-attribute	
next-hop	sag:SP_PROXY
realm	ATT
action	none
terminate-recu	rsion disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
app-protocol	
state	enabled
methods	
media-profiles	3
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	l

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

### local-policy

local-policy			
from-a	address		
		*	
to-add	lress	.1.	
		*	
source	e-realm		
daganin	ation	ATT	
descrip	e-time	N/A	
	vate-time	N/A N/A	
state	ale-time	enable	d
	-priority	none	u
	-attribute	none	
poney	next-hop		10.64.19.210
	realm		Enterprise
	action		none
	terminate-recu	rsion	disabled
	carrier		
	start-time		0000
	end-time		2400
	days-of-week		U-S
	cost		0
	app-protocol		SIP
	state		enabled
	methods		
	media-profiles	5	
	lookup		single
	next-key		
	eloc-str-lkup		disabled
. 1	eloc-str-match		
network-inter	lace		
name	ant id	wanco 0	IIIO
sub-po descriț		0	
hostna			
ip-add		192.9	230.221
1	lity-addr	172.7.	230.221
-	lity-addr		
netmas	•	255.25	5.255.0
gatewa			230.254
sec-ga	•		
-	artbeat		
-			

state heartbeat retry-count retry-timeout health-score dns-ip-primary dns-ip-backup1 dns-ip-backup2 dns-domain		disabled 0 0 1 0
dns-timeout hip-ip-list	11	
ftp-address icmp-address snmp-address telnet-address ssh-address		

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

network-interface	
name	s0p0
sub-port-id	0
description	
hostname	
ip-address	10.80.130.250
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.0
gateway	10.80.130.1
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	attavaya.com
dns-timeout	11
hip-ip-list	10.80.130.250
ftp-address	
icmp-address	10.80.130.250
snmp-address	
telnet-address	

ssh-address

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

network-interface	
name	s1p0
sub-port-id	0
description	
hostname	
ip-address	192.168.62.51
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.128
gateway	192.168.62.1
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	192.168.62.51
ftp-address	
icmp-address	192.168.62.51
snmp-address	
telnet-address	
ssh-address	

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

#### realm-config

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

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

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

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

### realm-config

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

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

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

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

session-agent

hostname	SM62
ip-address	10.64.19.210
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	UDP+TCP
realm-id	Enterprise
egress-realm-id	
description	
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-session	s0
max-outbound-sessio	ns 0
max-burst-rate	0
max-inbound-burst-ra	
max-outbound-burst-	rate 0
max-sustain-rate	0
max-inbound-sustain-	-rate 0
max-outbound-sustain	n-rate 0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	Proxy
loose-routing	enabled

send-media-session	enabled
response-map ping-method	<b>OPTIONS;hops=1</b>
ping-interval	180
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-respon	
out-service-response-	
media-profiles	loues
in-translationid	
out-translationid	
trust-me	enabled
request-uri-headers	chaolea
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	disdoled
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-r	ate 0
early-media-allow	ute o
invalidate-registration	s disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
reuse-connections	ТСР
tcp-keepalive	enabled
tcp-reconn-interval	10
max-register-burst-rat	e 0
register-burst-window	
sip-profile	
sip-isup-profile	

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

### session-agent

ip-address135.242.225.210port5060stateenabledapp-protocolSIPapp-typeSIPtransport-methodUDPrealm-idATTegress-realm-iddisableddescriptionenabledcarriersdisabledallow-next-hop-lpenabledanax-sessions0max-sessions0max-inbound-sessiors0max-outbound-sessiors0max-outbound-sessiors0max-sustain-rate0max-sustain-rate0max-sustain-rate0max-outbound-sustair-rate0max-sustain-rate0max-outbound-sustair-rate0max-sustain-rate0max-sustain-rate0min-seizures5min-asr0time-to-resume0sustain-rate-window0in-service-period0burst-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0sustain-rate-window0 <t< th=""><th>hostname</th><th>135.242.225.210</th></t<>	hostname	135.242.225.210
port5060stateenabledapp-protocolSIPapp-typeSIPtransport-methodUDPrealm-idATTegress-realm-iddescriptioncarriersallow-next-hop-lpallow-next-hop-lpenabledconstraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-outbound-sessions0max-sustain-rate0max-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0max-sustain-rate0max-sustain-rate0min-seizures5min-asr0time-to-resume0in-service-period0burst-rate-window0sustain-rate-window0sustain-rate-window0proxy-modeenabledsend-media-sessionenabledsend-media-sessionenabled		
stateenabledapp-protocolSIPapp-typeSIPtransport-methodUDPrealm-idATTegress-realm-iddescriptioncarriersallow-next-hop-lpallow-next-hop-lpenabledconstraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-session0max-outbound-session0max-sustain-rate0max-outbound-session0min-seizures5min-asr0time-to-resume0time-to-resume0service-period0burst-rate-window0sustain-rate-window0service-period0service-period0sustain-rate-window0requri-carrier-modeNoneproxy-modeenabledredirect-actionioose-routingloose-routingenabledsend-media-sessionenabled	-	
app-protocolSIPapp-typetransport-methodUDPrealm-idATTegress-realm-iddescriptioncarriersallow-next-hop-lpenabledconstraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-sessions0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0max-outbound-session0min-seizures5min-asr0ime-to-resume0in-service-period0sendin-rate-window0send-media-sessionenabledsend-media-sessionenabledsend-media-sessionenabled	-	
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realm-idATTegress-realm-idegress-realm-iddescriptioncarriersallow-next-hop-lpenabledconstraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-burst-rate0max-outbound-burst-rate0max-outbound-burst-rate0max-outbound-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-modeenabledredirect-actionenabledloose-routingenabledsend-media-sessionenabledresponse-mapinabled		UDP
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allow-next-hop-lp enabled constraints disabled max-sessions 0 max-inbound-sessions 0 max-outbound-sessions 0 max-burst-rate 0 max-burst-rate 0 max-outbound-burst-rate 0 max-outbound-burst-rate 0 max-sustain-rate 0 max-outbound-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 sustain-rate-window 0 req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled	-	
constraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-burst-rate0max-inbound-burst-rate0max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode	carriers	
constraintsdisabledmax-sessions0max-inbound-sessions0max-outbound-sessions0max-burst-rate0max-inbound-burst-rate0max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode	allow-next-hop-lp	enabled
max-inbound-sessions0max-outbound-sessions0max-burst-rate0max-inbound-burst-rate0max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0o1burst-rate-window0sustain-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode		disabled
max-outbound-sessions0max-burst-rate0max-inbound-burst-rate0max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	max-sessions	0
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max-inbound-burst-rate0max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-moderedirect-actionloose-routingenabledsend-media-sessionenabledresponse-map	max-outbound-sessio	ns 0
max-outbound-burst-rate0max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	max-burst-rate	0
max-sustain-rate0max-inbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	max-inbound-burst-ra	ate 0
max-inbound-sustain-rate0max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode	max-outbound-burst-	rate 0
max-outbound-sustain-rate0min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode	max-sustain-rate	0
min-seizures5min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	max-inbound-sustain-	-rate 0
min-asr0time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	max-outbound-sustain	n-rate 0
time-to-resume0ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	min-seizures	5
ttr-no-response0in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-moderedirect-actionloose-routingenabledsend-media-sessionenabledresponse-map	min-asr	0
in-service-period0burst-rate-window0sustain-rate-window0req-uri-carrier-modeNoneproxy-mode-redirect-action-loose-routingenabledsend-media-sessionenabledresponse-map-	time-to-resume	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	ttr-no-response	0
sustain-rate-window 0 req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map	in-service-period	0
req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map	burst-rate-window	0
proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map	sustain-rate-window	0
redirect-action loose-routing enabled send-media-session enabled response-map	req-uri-carrier-mode	None
loose-routing enabled send-media-session enabled response-map	proxy-mode	
send-media-session enabled response-map	redirect-action	
response-map	loose-routing	enabled
	send-media-session	enabled
ning-method OPTIONS-hons-70	response-map	
ping-include Of 1101(5,10ps=70	ping-method	<b>OPTIONS;hops=70</b>
ping-interval 60	ping-interval	60
ping-send-mode keep-alive		
ping-all-addresses disabled		
ping-in-service-response-codes	ping-in-service-respo	nse-codes

out-service-response-codes media-profiles in-translationid out-translationid enabled trust-me request-uri-headers stop-recurse local-response-map ping-to-user-part ping-from-user-part disabled li-trust-me in-manipulationid out-manipulationid manipulation-string manipulation-pattern p-asserted-id trunk-group max-register-sustain-rate 0 early-media-allow disabled invalidate-registrations 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-window0 sip-profile sip-isup-profile

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

session-agent

hostname	1.1.1.1
ip-address	1.1.1.1
port	5060
state	disabled
app-protocol	SIP
app-type	
transport-method	UDP
realm-id	ATT
egress-realm-id	
description	<b>ATT-Failover</b>

Solution & Interoperability Test Lab Application Notes ©2013 Avaya Inc. All Rights Reserved. carriers allow-next-hop-lp enabled constraints disabled max-sessions 0 max-inbound-sessions0 max-outbound-sessions 0 max-burst-rate 0 max-inbound-burst-rate 0 max-outbound-burst-rate 0 max-sustain-rate 0 0 max-inbound-sustain-rate max-outbound-sustain-rate 0 min-seizures 5 min-asr 0 0 time-to-resume 0 ttr-no-response in-service-period 0 burst-rate-window 0 sustain-rate-window 0 req-uri-carrier-mode None proxy-mode redirect-action loose-routing enabled send-media-session enabled response-map ping-method **OPTIONS;hops=70** ping-interval **60** ping-send-mode keep-alive ping-all-addresses disabled ping-in-service-response-codes out-service-response-codes media-profiles in-translationid out-translationid disabled trust-me request-uri-headers stop-recurse local-response-map ping-to-user-part ping-from-user-part li-trust-me disabled in-manipulationid out-manipulationid manipulation-string manipulation-pattern p-asserted-id

trunk-group max-register-sustain-rate 0 early-media-allow invalidate-registrations disabled rfc2833-mode none rfc2833-payload 0 codec-policy enforcement-profile refer-call-transfer disabled reuse-connections NONE tcp-keepalive none tcp-reconn-interval 0 max-register-burst-rate 0 register-burst-window0 sip-profile sip-isup-profile

ANNOTATION: The session group below specifies the AT&T IP services border element.

```
Note - Multiple session-agents may be specified in a session-group. The strategy parameter may be used to select how these multiple session-agents are used (e.g. Hunt and RoundRobin).
```

#### session-group

SP_PROXY
enabled
SIP
RoundRobin
1.1.1.1
135.242.225.210
enabled
401,407

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

sip-interface

state	enabled
realm-id	ATT
description	
sip-port	
address	192.168.62.51
port	5060
transport-pro	tocol UDP

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tls-profile			
allow-anonym	ous all		
ims-aka-profil			
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	disabled		
trust-mode	all		
max-nat-interval	3600		
nat-int-increment	10		
nat-test-increment	10 30		
sip-dynamic-hnt	disabled		
stop-recurse	401,407		
port-map-start	0		
port-map-end	0		
in-manipulationid			
out-manipulationid			
manipulation-string			
manipulation-pattern			
sip-ims-feature	disabled		
operator-identifier			
anonymous-priority	none		
max-incoming-conns 0			
per-src-ip-max-incom	ing-conns 0		
inactive-conn-timeout	:0		
untrusted-conn-timeo	ut O		
network-id			
ext-policy-server			
default-location-string	5		
charging-vector-mode	pass		
charging-function-add	-		
ccf-address	-		
ecf-address			

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

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

### sip-interface

state enabled		ed
realm-id Enterprise		prise
description		-
sip-port		
address		10.80.130.250
port		5060
transport-pro	otocol	ТСР
tls-profile		
allow-anonyn		all
ims-aka-profi	le	
carriers		
trans-expire	0	
invite-expire	0	
max-redirect-contacts	s 0	
proxy-mode		
redirect-action		
contact-mode	none	
nat-traversal	none	
nat-interval	30	
tcp-nat-interval	90	
registration-caching	disable	ed
min-reg-expire	300	
registration-interval	3600	
route-to-registrar	disable	
secured-network	disable	
teluri-scheme	disable	ed
uri-fqdn-domain		

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

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

sip-manipulation

impulat				
name		modSendRecv		
descri	ption	Modify se	endonly to sendrecv	
split-h	eaders			
join-he	eaders			
heade	r-rule			
	name	m	odsendonly	
	header-name		ontent-type	
	action		anipulate	
	comparison-ty		se-sensitive	
	msg-type	an		
methods			INVITE	
	match-value			
	new-value			
	element-rule			
	name		modmline	
		eter-name		
	type		mime	
	action		find-replace-all	
		val typo	-	
		val-type	any	
	-	rison-type		
	match-		sendonly	
	new-va	lue	sendrecv	

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

sip-manipulat	tion	
name	rejec	tOptions
descr	iption	
split-l	headers	
join-h	leaders	
heade	er-rule	
	name	RejectOpts
	header-name	From
	action	reject
	comparison-type	case-sensitive
	msg-type	request
	methods	OPTIONS
	match-value	
	new-value	405:"Acme Packet is alive, check back later"

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

### steering-pool

steering-pool	L			
ip-address		192.168.62.51		
start-port		16384		
end-p		32767		
realn	n-id	ATT		
steering-pool	l			
ip-ad	dress	10.80.	130.250	)
start	•port	16384		
end-p		32767	32767	
realn		Enter	prise	
system-confi				
hostn		Enter	prise-A	cme
	iption			
locati				
	system-contact			
	system-name			
	system-location			
1	-enabled	enable		
	e-snmp-auth-tra	-	disable	
	e-snmp-syslog-i	•		
	e-snmp-monitor	-		
	e-env-monitor-t	-	disable	ed
_	-syslog-his-table			
-	-syslog-level	WARI		
•	m-log-level	WARI		
-	ss-log-level	NOTI		
-	ss-log-ip-addres		0.0.0.0	)
-	ss-log-port	0		
collec				
	sample-interv	al	5	
	push-interval		15	
	boot-state		disable	ed
	start-time		now	
	end-time		never	
	red-collect-sta		disable	ed
	red-max-trans		1000	
	red-sync-start		5000	
	red-sync-com	1	1000	
	push-success-	trap-sta	te	disabled

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

# 9. Verification Steps

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

### 9.1. Avaya Aura® Communication Manager

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

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

### 9.2. Avaya Aura® Session Manager

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

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

AVAYA	Avaya Aura® System Manager 6.2 н					lelp   About   Change Password   Log off admir		
-							Session Manager * Ro	uting <sup>×</sup> Home
Session Manager	I Home /El	ements / Session Manage	r / System Status / SIP Er	ntity Mon	itoring			
Dashboard								Help
Session Manager	SIP E	ntity, Entity Link (	Connection Status					
Administration		isplays detailed connection status			instances to	a single SIP entity.		
<b>Communication Profile</b>	All Cost	e tisks to oth taking						
Editor	All Enti	ty Links to SIP Entity:	ACMESBCATT-5060					
Network Configuration	Sumn	nary View						
Device and Location Configuration	1 Item   Refresh							Filter: Enable
Application	Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Configuration	► Show	DenverSM	10.80.130.250	5060	TCP	Up	405 Method Not Allowed	Up
System Status								
SIP Entity Monitoring								

## 9.3. AT&T IP Toll Free

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.

- 2. Verify basic call functions such as hold, transfer and conference.
- 3. Verify the use of DTMF signaling.

# 10. Conclusion

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

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

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

# 11. References

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

### Avaya Aura® Session Manager/System Manager

- [1] Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.2, July 2012
- [2] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.2, Issue 2, August 2012
- [3] Implementing Avaya Aura® System Manager, Doc ID 03-603473 Issue 1, July 2012
- [4] Administering Avaya Aura® System Manager, Document Number 03-603324, Release 6.2, July 2012

### Avaya Aura® Communication Manager

- [5] Administering Avaya Aura® Communication Manager, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
- [6] Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Release 5.2, April 2009, Document Number 07-600780

### Avaya Modular Messaging

- [7] Modular Messaging Multi-Site Guide Release 5.1, June 2009
- [8] Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades, June 2009
- [9] Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades, June 2009
- [10] Modular Messaging for IBM Lotus Domino 5.1 Installation and Upgrades, June 2009

### Acme Packet Support (login required):

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

### **AT&T IP Toll Free Service Descriptions:**

[12] AT&T IP Toll Free

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

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Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at <u>devconnect@avaya.com</u>.