



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3 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 R6.2, Avaya Aura® Session Manager R6.3, 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 R6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 6.2 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 R6.3 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.

TABLE OF CONTENTS

1.	Introduction.....	3
2.	General Test Approach and Test Results.....	3
2.1.	Interoperability Compliance Testing.....	4
2.2.	Test Results and Known Limitations	4
2.3.	Support	5
3.	Reference Configuration.....	5
3.1.	Illustrative Configuration Information	7
3.2.	Call Flows	8
3.2.1.	Inbound Call to an Agent/Station using VDN/Vectors.....	8
3.2.2.	Coverage to Voicemail	9
4.	Equipment and Software Validated	10
5.	Configure Avaya Aura® Session Manager Release 6.3	11
5.1.	SIP Domain	12
5.2.	Locations	13
5.3.	Configure Adaptations	15
5.4.	SIP Entities.....	17
5.5.	Entity Links.....	21
5.6.	Time Ranges.....	22
5.7.	Routing Policies	23
5.8.	Dial Patterns	26
5.9.	Session Manager Administration	27
6.	Configure Avaya Aura® Communication Manager 6.2.....	28
6.1.	System Parameters	28
6.2.	Dial Plan.....	30
6.3.	IP Node Names.....	30
6.4.	IP Codec Parameters	30
6.5.	IP Network Regions	31
6.6.	SIP Trunks.....	32
6.6.1.	SIP Trunk for Inbound Calls with AT&T IP Toll Free service	32
6.6.2.	SIP Trunk for CM Messaging and SIP Endpoints	34
6.7.	Public Unknown Numbering.....	35
6.8.	Alternate Automated Routing (AAR) Table	35
6.9.	Route Pattern.....	36
6.10.	Optional Features.....	37
6.10.1.	Call Center Provisioning	37
6.10.2.	CM Messaging Coverage Path and Hunt Group.....	41
6.11.	Saving Translations	42
7.	Configure Acme Packet Session Border Controller	43
8.	Verification Steps.....	60
	60
8.1.	Avaya Aura® Communication Manager	60
8.2.	Avaya Aura® Session Manager	60
8.3.	AT&T IP Toll Free.....	60
9.	Conclusion	61
10.	References.....	62

1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3, and Acme Packet Net-Net 3800¹ with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

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The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. **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.

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 Aura® Communication Manager Messaging (CM 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.

¹ 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 CM Messaging
- Legacy Transfer Connect
- Alternate Destination Routing
- IP Re-routing
- Long Duration Calls

2.2. Test Results and Known Limitations

1. When the call is put on hold on Communication Manager, SDP with **a=sendonly** is sent to AT&T IP Flexible Reach service but it sends **a=inactive** in response which results in no Music-on-Hold being sent to PSTN. A Header Manipulation Rule was provided as shown in **Section 7** to send **a=sendrecv** to resolve this situation.
2. G.726 codec is not supported between AT&T IP Toll Free service and Communication Manager.
3. **Alternate Destination Routing/Ring No Answer (ADR/RNA)** only works if the call is destined for an agent. When AT&T IP Toll Free service receives a **181 Call Being Forwarded** message, it is able to process ADR/RNA properly which happens when the call is destined for an agent logged in at CPE. When call is destined for an endpoint on CPE, AT&T IP Toll Free service receives a **180 Ringing** and is unable to process ADR/RNA properly.
4. 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.

5. 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.

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 <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

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

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communication services for a particular enterprise site. In the reference configuration, Communication Manager 6.2 runs on an Avaya S8800 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya 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 and SIP, Avaya 6408D Series Digital Telephone, Avaya Analog phone and Avaya one-X® Communicator (SIP/H323), a PC based softphone.
- The Acme Packet SBC 3800² provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and

² Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

the enterprise internal network³. UDP transport protocol is used between the Acme Packet SBC and the AT&T Toll Free service.

- CM Messaging system provides the corporate voice messaging capabilities in the reference configuration. **The provisioning of CM Messaging is beyond the scope of this document.**
- 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 at the CPE are registered to the Communication Manager C-LANs and the SIP phones are registered to Session Manager.

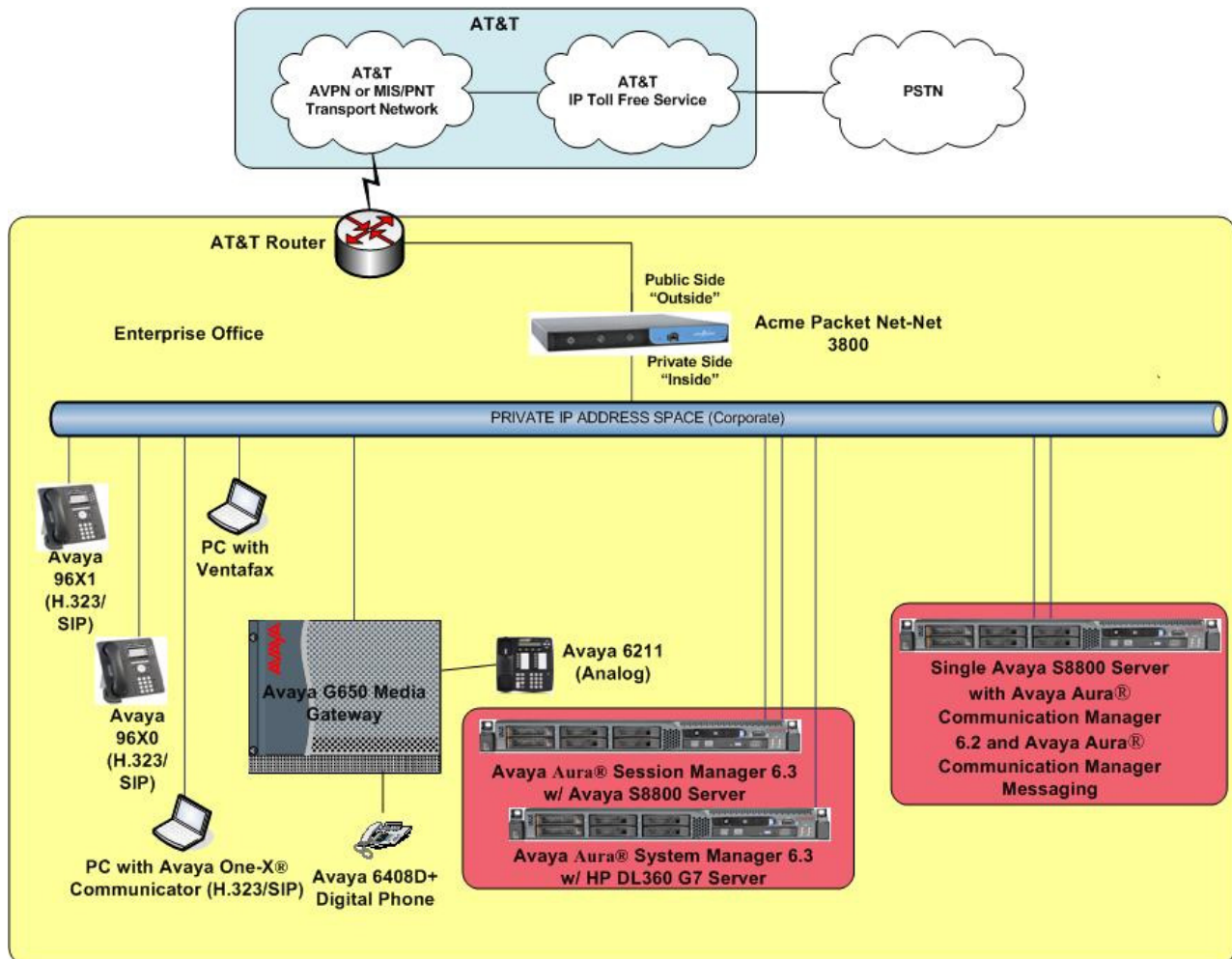


Figure 1: Reference Configuration

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

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	10.80.130.120
Avaya Aura® Session Manager	
Management IP Address	10.80.130.121
Network IP Address	10.80.130.122
Avaya Aura® Communication Manager	
Control LAN (C-LAN) IP Address	10.80.130.102
Media Processor (MedPro) IP Address	10.80.130.103
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 (connected to AT&T Access Router/IP Toll Free service)	192.168.62.50
IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager)	10.80.130.250
AT&T IP Toll Free service	
Border Element IP Address	192.242.225.210
DNIS Passed in Request URI used by Session Manager for routing	0000041535[0,1,7,8,9]10[5,6][0,1,7,8,9]

Table 1: Illustrative Values Used in this Reference Configuration

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 6.3 (6.3.0.8.923) System Platform 6.2.2.06002.0
Avaya S8800 Server	Avaya Aura® Session Manager 6.3 SP2 (6.3.0.0.630039)
Avaya S8800 Server	Avaya Aura® Communication Manager R6.2 SP5 (02.0.823.0-20396) System Platform 6.2.2.08001.0
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW06 FW057
TN799DP Control-LAN (C-LAN)	HW01 FW041
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW063
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW011
Avaya 9650 IP Telephone	H.323 R3.1.5
Avaya 9641G IP Telephone	H.323 R6.2.3.12
Avaya 9608 IP Telephone	SIP R6.2.1
Avaya one-X® Communicator (H323/SIP)	6.1.7.04-SP7-39506
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 AVPN/MIS-PNT transport service connection	VNI 26

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release 6.3

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**.



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.

Avaya Aura® System Manager 6.3

Last Logged on at March 27, 2013 1:07 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing × **Home**

Home / Elements / Routing

Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- 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 why 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 Policies" 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 Aura® System Manager 6.3

Last Logged on at March 27, 2013 1:07 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing × **Home**

Home / Elements / Routing / Domains

Domain Management Commit Cancel

1 Item Refresh Filter: Enable

Name	Type	Notes
* attavaya.com	sip	SIP domain for ATT

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.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.3", and a user status bar indicating "Last Logged on at March 27, 2013 1:07 PM" with links for "Help", "About", "Change Password", and "Log off admin". The left sidebar contains a tree view with "Routing" selected, and sub-items: "Domains", "Locations", "Adaptations", "SIP Entities", and "Entity Links". The main content area shows the breadcrumb "Home / Elements / Routing / Locations" and the "Location Details" form for "Session Manager". The form has "Commit" and "Cancel" buttons and a "Help ?" link. The "General" tab is active, showing fields for "Name" (Session Manager) and "Notes" (Session Manager).

Session Manager Location Details

The screenshot shows the Avaya Aura System Manager 6.3 interface for the "Acme Packet SBC" location. The top navigation bar and left sidebar are identical to the previous screen. The main content area shows the breadcrumb "Home / Elements / Routing / Locations" and the "Location Details" form for "Acme_SBC_130". The form has "Commit" and "Cancel" buttons and a "Help ?" link. The "General" tab is active, showing fields for "Name" (Acme_SBC_130) and "Notes" (SBC To ATT). Below the "General" tab, the "Overall Managed Bandwidth" section includes "Managed Bandwidth Units" (Kbit/sec), "Total Bandwidth", "Multimedia Bandwidth", and a checked checkbox for "Audio Calls Can Take Multimedia Bandwidth". The "Per-Call Bandwidth Parameters" section includes "Maximum Multimedia Bandwidth (Intra-Location)" (1000 Kbit/Sec), "Maximum Multimedia Bandwidth (Inter-Location)" (1000 Kbit/Sec), "Minimum Multimedia Bandwidth" (64 Kbit/Sec), and "Default Audio Bandwidth" (80 Kbit/sec). The "Alarm Threshold" section includes "Overall Alarm Threshold" (80 %), "Multimedia Alarm Threshold" (80 %), "Latency before Overall Alarm Trigger" (5 Minutes), and "Latency before Multimedia Alarm Trigger" (5 Minutes). The "Location Pattern" section includes "Add", "Remove", and "Refresh" buttons, a table with 1 item, and a "Filter: Enable" link. The table has columns for "IP Address Pattern" and "Notes", with one entry: "10.80.130.250" and "ATT Acme SBC internal address".

Acme Packet SBC Location Details

Location_130 shown below serves Communication Manager, CM Messaging and other elements on the CPE.

AVAYA

Avaya Aura® System Manager 6.3

Last Logged on at March 27, 2013 1:07 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Locations

Commit
Cancel

Help ?

Location Details

General

* Name:
Location_130

Notes:
Subnet 130

Overall Managed Bandwidth

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 %

* Latency before Overall Alarm Trigger:
5 Minutes

* Latency before Multimedia Alarm Trigger:
5 Minutes

Location Pattern

Add
Remove

1 Item
Refresh

☐
IP Address Pattern

☒
* 10.80.130.*

Notes

Filter: Enable

Subnet 130 Location Details

5.3. Configure Adaptations

The following screen displays the adaptation used for calls between AT&T IP Toll Free Service and Session Manager via Acme Packet SBC. The **Module parameter** field is set to **fromto=true iodstd=attavaya.com osrcd=192.168.62.50** (IP Address of the external interface of Acme Packet SBC) **odstd=135.242.225.210** (IP Address of AT&T IP Flexible Reach Border Element). This adaptation is applied to **AcmeSBCATT-5060** entity configured in **Section 5.4**.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.3', and a user status bar indicating 'Last Logged on at March 27, 2013 1:07 PM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar contains a menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations (highlighted), SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Home / Elements / Routing / Adaptations' and shows 'Adaptation Details' for 'AT&T Adaptations'. The 'General' tab is active, displaying fields for 'Adaptation name' (AT&T Adaptations), 'Module name' (AttAdapter), 'Module parameter' (fromto=true iodstd=attavaya.com), 'Egress URI Parameters' (empty), and 'Notes' (fromto=true iodstd=attavaya.com). Below this, there are two sections for 'Digit Conversion'. The first section, 'Digit Conversion for Incoming Calls to SM', has an 'Add' button and a table with 0 items. The second section, 'Digit Conversion for Outgoing Calls from SM', also has an 'Add' button and a table with 0 items. Both tables have columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, Adaptation Data, and Notes. The interface also includes 'Commit' and 'Cancel' buttons at the top right of the details section.

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

The following screen displays the adaptation used for calls between Session Manager and Communication Manager. The following screen shows the adaptation used for **CM62_CLAN1A02-5060** configured in **Section 5.4**. In this case, digit conversion is done before and after routing decision has been made based upon the user part of the SIP URI.

AVAYA
Avaya Aura® System Manager 6.3

Last Logged on at May 29, 2013 5:06 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing
Home

Home / Elements / Routing / Adaptations

Help ?

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Adaptation Details

Commit
Cancel

General

* Adaptation name:
ATT_CLAN02

Module name:
DigitConversionAdapter

Module parameter:
fromto=true osrcd=attavaya.com

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

Add
Remove

1 Item Refresh

Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*+	*1	*36		*1		origination		

Select : All, None

Digit Conversion for Outgoing Calls from SM

Add
Remove

5 Items Refresh

Filter: Enable

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*000004153571057	*15	*15		*15	2010	destination		
<input type="checkbox"/>	*000004153581058	*15	*15		*15	2011	destination		
<input type="checkbox"/>	*000004153591059	*15	*15		*15	53054	destination		
<input type="checkbox"/>	*000004153601060	*15	*15		*15	53053	destination		
<input type="checkbox"/>	*000004153611061	*15	*15		*15	53001	destination		

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.

AVAYA

Avaya Aura® System Manager 6.3

Last Logged on at May 29, 2013 5:06 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / SIP Entities

SIP Entity Details

CommitCancel

Help ?

General

* Name: SM63

* FQDN or IP Address: 10.80.130.122

Type: Session Manager

Notes:

Location: Session Manager

Outbound Proxy:

Time Zone: America/Denver

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled

* Proactive Monitoring Interval (in seconds): 900

* Reactive Monitoring Interval (in seconds): 120

* Number of Retries: 1

Entity Links

AddRemove

4 Items Refresh

Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	SM63	TCP	* 5060	CM Messaging	* 5080	Trusted	<input type="checkbox"/>
<input type="checkbox"/>	SM63	TCP	* 5060	AcmeSBCATT	* 5060	Trusted	<input type="checkbox"/>
<input type="checkbox"/>	SM63	TCP	* 5070	CM62_CLAN1A02-5070	* 5070	Trusted	<input type="checkbox"/>
<input type="checkbox"/>	SM63	TCP	* 5060	CM62_CLAN1A02-5060	* 5060	Trusted	<input type="checkbox"/>

Select : All, None

Port

TCP Failover port:

TLS Failover port:

AddRemove

4 Items Refresh

Filter: Enable

	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	attavaya.com	
<input type="checkbox"/>	5061	TLS	attavaya.com	
<input type="checkbox"/>	5070	TCP	attavaya.com	
<input type="checkbox"/>	5080	TCP	attavaya.com	

Session Manager Entity

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Routing

Home

Home / Elements / Routing / SIP Entities

Help ?

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

SIP Entity Details

Commit Cancel

General

* Name:

AcmeSBCATT-5060

* FQDN or IP Address:

10.80.130.250

Type:

Other

Notes:

SIP Trunk to Acme SBC for ATT

Adaptation:

AT&T Adaptations

Location:

Acme_SBC_130

Time Zone:

America/Denver

Override Port & Transport with DNS SRV:

☐

* SIP Timer D/T (in seconds):

4

Credential name:

Call Detail Recording:

none

CommProfile Type Preference:

SIP Link Monitoring

SIP Link Monitoring:

Use Session Manager Configuration

Supports Call Admission Control:

☐

Shared Bandwidth Manager:

☐

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

Add Remove

1 Item Refresh

Filter: Enable

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	SM63	TCP	* 5060	AcmeSBCATT-5060	* 5060	Trusted	<input type="checkbox"/>

Acme Packet SBC Entity

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Routing

Home

Home / Elements / Routing / SIP Entities

Commit

Cancel

Help ?

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

SIP Entity Details

General

* Name:

CM62_CLAN1A02-5060

* FQDN or IP Address:

10.80.130.102

Type:

CM

Notes:

Inbound Calls to CM SIP Trunk

Adaptation:

ATT_CLAN02

Location:

Location_130

Time Zone:

America/Denver

Override Port & Transport with DNS SRV:

☐

* SIP Timer B/F (in seconds):

4

Credential name:

Call Detail Recording:

none

SIP Link Monitoring

SIP Link Monitoring:

Use Session Manager Configuration

Supports Call Admission Control:

☐

Shared Bandwidth Manager:

☐

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

Add

Remove

1 Item

Refresh

Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	SM63	TCP	* 5060	CM62_CLAN1A02-5060	* 5060	Trusted	<input type="checkbox"/>

Communication Manager Entity for Inbound Calls (CM6.2CLAN1A02-5060)

AT:Reviewed
SPOC 7/29/2013

Solution & Interoperability Test Lab Application Notes
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19 of 63
CM62SM63APIPTF

The following screen shows SIP Entity configured for the Communication Manager trunk group used for CM Messaging and SIP endpoints. See **Section 6.6.2** for the trunk configuration from Communication Manager to Session Manager to support the CM Messaging and SIP endpoints.

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[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) * [Home](#)

Home / Elements / Routing / SIP Entities [Help ?](#)

SIP Entity Details [Commit](#) [Cancel](#)

General

* Name: CM62_CLAN1A02-5080

* FQDN or IP Address: 10.80.130.102

Type: CM

Notes: CM Messaging and SIP Endpoints

Adaptation:

Location: Location_130

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: ☐

Shared Bandwidth Manager: ☐

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

[Add](#) [Remove](#)

1 Item Refresh		Filter: Enable				
SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
SM63	TCP	*5080	CM62_CLAN1A02-5080	*5080	Trusted	<input type="checkbox"/>

Communication Manager Entity for SIP endpoints and coverage to CM Messaging (CM6.2CLAN1A02-5080)

The following screen shows SIP Entity configured for the CM Messaging which is installed and configured on Communication Manager platform. Installation and configuration of CM Messaging is beyond the scope of this document. The **FQDN or IP Address** field is set to the Communication Manager server IP address (**10.80.130.100**).

The screenshot displays the Avaya Aura System Manager 6.3 interface. The left sidebar shows the navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The configuration fields are as follows:

- Name:** CM Messaging
- FQDN or IP Address:** 10.80.130.100
- Type:** Modular Messaging
- Notes:** CM Messaging
- Adaptation:** (dropdown menu)
- Location:** Location_130
- Time Zone:** America/Denver
- Override Port & Transport with DNS SRV:** (checkbox, unchecked)
- SIP Timer B/F (in seconds):** 4
- Credential name:** (text field)
- Call Detail Recording:** none
- SIP Link Monitoring:** Use Session Manager Configuration
- Supports Call Admission Control:** (checkbox, unchecked)
- Shared Bandwidth Manager:** (checkbox, unchecked)
- Primary Session Manager Bandwidth Association:** (dropdown menu)
- Backup Session Manager Bandwidth Association:** (dropdown menu)
- Entity Links:** Add, Remove buttons

Below the configuration fields is a table showing the entity link configuration:

SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
SM63	TCP	*5060	CM Messaging	*5080	Trusted	<input type="checkbox"/>

CM Messaging Entity

5.5. Entity Links

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

The screenshot displays the Avaya Aura System Manager 6.3 interface for the 'Entity Links' configuration. The left sidebar shows the navigation menu with 'Entity Links' selected. The main content area is titled 'Entity Links' and includes a table showing the entity link configuration:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
*SM63_CM62_CLAN1A1	*SM63	TCP	*5060	*CM62_CLAN1A02-5060	*5060	Trusted	<input type="checkbox"/>	Link for Inbound SIP Tr

Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5060)

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Help | About | Change Password | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Entity Links [Help ?](#)

Entity Links [Commit](#) [Cancel](#)

1 Item [Refresh](#) Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
*SM63_CM62_CLAN1A	*SM63	TCP	*5080	*CM62_CLAN1A02-5080	*5080	Trusted	<input type="checkbox"/>	SIP Endpoint and CMM

Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5080)

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Help | About | Change Password | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Entity Links [Help ?](#)

Entity Links [Commit](#) [Cancel](#)

1 Item [Refresh](#) Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
*SM63_AcmeSBCATT-5060	*SM63	TCP	*5060	*AcmeSBCATT-5060	*5060	Trusted	<input type="checkbox"/>	Link to SBC-ATT

Entity link between Session Manager and Acme Packet SBC

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Help | About | Change Password | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Entity Links [Help ?](#)

Entity Links [Commit](#) [Cancel](#)

1 Item [Refresh](#) Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
*SM63_CMM_5080_TC	*SM63	TCP	*5060	*CM Messaging	*5080	Trusted	<input type="checkbox"/>	Link to CM Messaging

Entity link between Session Manager and CM Messaging

5.6. Time Ranges

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

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Help | About | Change Password | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Time Ranges [Help ?](#)

Time Ranges [New](#) [Edit](#) [Delete](#) [Duplicate](#) [More Actions](#)

1 Item [Refresh](#) Filter: Enable

Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	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.

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Avaya Aura® System Manager 6.3

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Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit

Cancel

General

* Name:

ToCM62CLAN1A02-5060

Disabled:

☐

* Retries:

0

Notes:

To CM Trunk

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM62_CLAN1A02-5060	10.80.130.102	CM	Inbound Calls to CM SIP Trunk

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item

Refresh

Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Dial Patterns

Add

Remove

1 Item

Refresh

Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
00000	9	21	<input type="checkbox"/>	attavaya.com	Acme_SBC_130	DNIS for IPTF/IPTC calls

Routing Policy for Communication Manager (CLAN1A02-5060)

Avaya Aura® System Manager 6.3

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[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Routing Policies

Commit
Cancel
Help ?

Routing Policy Details

General

* Name:

ToCM62CLAN1A02-5080

Disabled:

☐

* Retries:

0

Notes:

To Trunk Group for Messaging and

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM62_CLAN1A02-5080	10.80.130.102	CM	CM Messaging and SIP Endpoints

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item

Refresh

Filter: Enable

	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Dial Patterns

Add

Remove

2 Items

Refresh

Filter: Enable

	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	5	5	5	<input type="checkbox"/>	attavaya.com	Location_130	SIP Extensions/CM Messaging MWI
<input type="checkbox"/>	5	5	5	<input type="checkbox"/>	attavaya.com	Session Manager	SIP Extensions/CM Messaging MWI

Routing Policy for Communication Manager (CLAN1A02-5080)

[Routing](#) × [Home](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit

Cancel

Help ?

General

* Name:

To CM Messaging

Disabled:

☐

* Retries:

0

Notes:

To CM Messaging System

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM Messaging	10.80.130.100	Modular Messaging	

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item

Refresh

Filter: Enable

<input type="checkbox"/>	Ranking	1	Name	2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0		24/7		<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Dial Patterns

Add

Remove

2 Items

Refresh

Filter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	55000	5	5	<input type="checkbox"/>	attavaya.com	Acme_SBC_130	CM Messaging Pilot Number
<input type="checkbox"/>	55000	5	5	<input type="checkbox"/>	attavaya.com	Location_130	CM Messaging Pilot Number

Routing Policy for CM Messaging Pilot Number

5.8. Dial Patterns

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

AVAYA

Avaya Aura® System Manager 6.3

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Help | About | Change Password | Log off admin

Routing

Home

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns

Help ?

Dial Pattern Details

Commit

Cancel

General

* Pattern: 00000

* Min: 9

* Max: 21

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: attavaya.com

Notes: DNIS for IPTF/IPTC calls

Originating Locations and Routing Policies

Add

Remove

1 Item Refresh

Filter: Enable

	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Acme_SBC_130	Acme SBC To ATT	ToCM62CLAN1A02-5060	0	<input type="checkbox"/>	CM62_CLAN1A02-5060	To CM Trunk

Dial Pattern for Inbound Calls to Communication Manager (CLAN1A02-5060)

AVAYA

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Routing

Home

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns

Help ?

Dial Pattern Details

Commit

Cancel

General

* Pattern: 55000

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: attavaya.com

Notes: CM Messaging Pilot Number

Originating Locations and Routing Policies

Add

Remove

2 Items Refresh

Filter: Enable

	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Acme_SBC_130	Acme SBC To ATT	To CM Messaging	0	<input type="checkbox"/>	CM Messaging	To CM Messaging System
<input type="checkbox"/>	Location_130	Subnet 130	To CM Messaging	0	<input type="checkbox"/>	CM Messaging	To CM Messaging System

Dial Pattern for Covered Calls to CM Messaging

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Help | About | Change Password | Log off admin

Routing × Home

Home / Elements / Routing / Dial Patterns Help ?

Dial Pattern Details Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

2 Items Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Location_130	Subnet 130	ToCM62CLAN1A02-5080	0	<input type="checkbox"/>	CM62_CLAN1A02-5080	To Trunk Group for Messaging and SIP Endpoints
<input type="checkbox"/>	Session Manager	Session Manager	ToCM62CLAN1A02-5080	0	<input type="checkbox"/>	CM62_CLAN1A02-5080	To Trunk Group for Messaging and SIP Endpoints

Dial Pattern for SIP Extension and CM Messaging MWI

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 **SM63** used in this reference configuration.

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Help | About | Change Password | Log off admin

Session Manager × Routing × Home

Home / Elements / Session Manager / Session Manager Administration Help ?

View Session Manager Return

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

General

SIP Entity Name:

Description:

Management Access Point Host Name/IP:

Direct Routing to Endpoints:

Security Module

SIP Entity IP Address:

Network Mask:

Default Gateway:

Call Control PHB:

QOS Priority:

Speed & Duplex:

VLAN ID:

View Session Manager (SM63)

6. Configure Avaya Aura® Communication Manager 6.2

In this reference configuration Communication Manager 6.2 is provisioned in an Access Element configuration, supporting H.323, SIP and Digital endpoints. 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	2	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 permission changes.)					

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? y
Enhanced EC500? y	ISDN/SIP Network Call Redirection? n
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	

3. On **Page 11** of the **system-parameters features** and verify that the **Message Waiting Lamp Indicates Status For** field in is set to **loginID** to enable Message Waiting Indicator on the station where an agent is logged in.

display system-parameters features	Page 11 of 19
FEATURE-RELATED SYSTEM PARAMETERS	
CALL CENTER SYSTEM PARAMETERS	
EAS	
Expert Agent Selection (EAS) Enabled? y	
Minimum Agent-LoginID Password Length:	
Direct Agent Announcement Extension:	
Message Waiting Lamp Indicates Status For: loginID	
VECTORIZING	
Converse First Data Delay: 0	Second Data Delay: 2
Converse Signaling Tone (msec): 100	
Prompting Timeout (secs): 10	
Interflow-qpos EWT Threshold: 2	
Reverse Star/Pound Digit For Collect Step? n	
Available Agent Adjustments for BSR? n	
BSR Tie Strategy: 1st-found	
Store VDN name in Station's Local Call Log? n	
SERVICE OBSERVING	
Service Observing: Warning Tone? y	or Conference Tone? n
Service Observing/SSC Allowed with Exclusion? n	
Allow Two Observers in Same Call? 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).
- 4 or 5-digit Extensions with a **Call Type** of **ext** beginning with the digits **3xxxx**, **4xxx** or **5xxxx** (e.g. Announcements, Local extensions for Communication Manager stations, agents, Vector Directory Numbers, 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 dialplan analysis						Page 1 of 12		
DIAL PLAN ANALYSIS TABLE								
Location: all						Percent Full: 1		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
1	3	dac						
2	4	fac						
3	5	ext						
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	1 of	2
IP NODE NAMES				
Name	IP Address			
Gateway001	10.80.130.1			
CLAN-1A02	10.80.130.102			
SM63	10.80.130.122			

6.4. IP Codec Parameters

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

change ip-codec-set 2

Page1 of 2

IP Codec Set

Codec Set: 2

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.729B	n	3	30
2: G.729A	n	3	30
3: G.711MU	n	3	30

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

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

6.5. 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		Page 1 of 20
IP NETWORK REGION		
Region: 2		
Location: Authoritative Domain: attavaya.com		
Name: ATT Calls		
MEDIA PARAMETERS		
Codec Set: 2		Intra-region IP-IP Direct Audio: yes
UDP Port Min: 16384		Inter-region IP-IP Direct Audio: yes
UDP Port Max: 32767		IP Audio Hairpinning? y
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		RTCP Reporting Enabled? y
Audio PHB Value: 46		RTCP MONITOR SERVER PARAMETERS
Video PHB Value: 26		Use Default Server Parameters? y
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		Page 4 of 20
Source Region: 2		Inter Network Region Connection Management
		I M
		G A t
dst codec direct	WAN-BW-limits	Video Intervening Dyn A G c
rgn set WAN Units	Total Norm Prio Shr Regions	CAC R L e
1 2 y NoLimit		n t
2 2		all

6.6. SIP Trunks

Two trunks are configured for testing in this reference configuration.

- Trunk group to handle all inbound calls from AT&T IP Toll Free service
- Trunk group to handle CM Messaging and SIP extension registered with Session Manager

6.6.1. SIP Trunk for Inbound Calls with AT&T IP Toll Free service

This trunk is used to handle all the inbound calls from AT&T IP Toll Free service. 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 1		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: CLAN_1A02	Far-end Node Name: SM63	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 2	
Far-end Domain: attavaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group.

add trunk-group 1		Page 1 of 21
TRUNK GROUP		
Group Number: 1	Group Type: sip	CDR Reports: y
Group Name: ATT	COR: 1	TN: 1 TAC: 101
Direction: incoming	Outgoing Display? n	
Dial Access? n	Night Service:	
Queue Length: 0		
Service Type: public-ntwrk	Auth Code? n	
	Member Assignment Method: auto	
	Signaling Group: 1	
	Number of Members: 10	

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 1	Page 2 of 21
Group Type: sip	
TRUNK PARAMETERS	
Unicode Name: auto	
SCCAN? n	Redirect On OPTIM Failure: 5000
	Digital Loss Group: 18
Preferred Minimum Session Refresh Interval(sec): 900	
Disconnect Supervision - In? y Out? y	
XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n	

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

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

5. On **Page 4** of the **trunk-group** form:

- Set **Support Request History?** field to **n**.
- 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 1	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? n	
Support Request History? n	
Telephone Event Payload Type: 100	
Convert 180 to 183 For Early Media? n	
Always Use re-INIVIT for Display Updates? n	
Identity for Calling Party Display? P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Enable Q-SIP? n	

6.6.2. SIP Trunk for CM Messaging and SIP Endpoints

This SIP trunk is used for coverage to CM Messaging and SIP Endpoints. Configuration for this trunk is similar to the trunk group configured in **Section 6.6.1** with the differences shown in the screens below:

add signaling-group 3		Page 1 of 1
SIGNALING GROUP		
Group Number: 3	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: CLAN_1A02	Far-end Node Name: SM63	
Near-end Listen Port: 5080	Far-end Listen Port: 5080	
	Far-end Network Region: 2	
	Far-end Domain: attavaya.com	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early Media? n	
	Alternate Route Timer(sec): 6	

add trunk-group 3		Page 1 of 21
TRUNK GROUP		
Group Number: 3	Group Type: sip	CDR Reports: y
Group Name: CM Messaging/SIP Endpoints	COR: 1	TN: 1
Direction: two-way	Outgoing Display? n	TAC: 103
Dial Access? n	Night Service:	
Queue Length: 0		
Service Type: public-ntwrk	Auth Code? n	
	Member Assignment Method: auto	
	Signaling Group: 3	
	Number of Members: 10	

On **Page 3** of the trunk-group form set **Numbering Format** field to **private**

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

On **Page 4** of the **trunk-group** form, make sure that **Support Request History?** field is set to **y** [default].

add trunk-group 3	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 100	
Convert 180 to 183 For Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display? P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Enable Q-SIP? n	

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 **3** will populate the required SIP headers with the correct telephone extension. Additionally, this form is used for inbound calls to populate the user part in **Contact** and **PAI** headers.

change public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext	Ext	Trk	CPN	Total	
Len	Code	Grp(s)	Prefix	CPN	
5		3		5	Total Administered: 2
5	50001	3	73232	10	Maximum Entries: 9999

6.8. Alternate Automated Routing (AAR) Table

Automatic Alternate Routing (AAR) is used to direct calls to CM Messaging and SIP Endpoints registered with Session Manager via the route pattern defined in **Section 6.9**. In the following screen, **5005** string used for calls to SIP endpoints and **55000** is the pilot number used for coverage to CM Messaging via Session Manager using trunk configured in **Section 6.6.2**.

change ars analysis 1							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 15
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
	5005	5	5	3	aar		n
	55000	5	5	3	aar		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 calls destined to CM Messaging and SIP endpoints registered with Session Manager (see **Section 6.6.2**).

- **Grp No** – Set to **3** i.e. the trunk group configured in **Section 6.6.2**.
- **FRL** – Set to **0** (zero).

change route-pattern 3															Page 1 of 3			
Pattern Number: 2 Pattern Name: ToCMM																		
SCCAN? n Secure SIP? n																		
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted								DCS/	IXC		
No				Mrk	Lmt	List	Del	Digits								QSIG		
									Dgts								Intw	
1:	3	0														n	user	
2:																n	user	
3:																n	user	
4:																n	user	
5:																n	user	
6:																n	user	
BCC		VALUE		TSC	CA-TSC		ITC		BCIE	Service/Feature				PARM	No. Numbering		LAR	
0 1 2 M 4 W					Request										Dgts Format			
															Subaddress			
1:	y	y	y	y	y	n	n			rest							none	
2:	y	y	y	y	y	n	n			rest							none	
3:	y	y	y	y	y	n	n			rest							none	
4:	y	y	y	y	y	n	n			rest							none	
5:	y	y	y	y	y	n	n			rest							none	
6:	y	y	y	y	y	n	n			rest							none	

6.10. 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. If necessary, consult [5] and [6] for further details. The configuration steps that follow are provided for reference purposes only.

display system-parameters customer-options	Page 6 of 11
CALL CENTER OPTIONAL FEATURES	
Call Center Release: 5.0	
ACD? y	Reason Codes? n
CMS (Basic)? y	Service Level Maximizer? n
BCMS/VuStats Service Level? y	Service Observing (Basic)? n
BSR Local Treatment for IP & ISDN? n	Service Observing (Remote/By FAC)? n
Business Advocate? n	Service Observing (VDNs)? n
Call Work Codes? n	Timed ACW? n
DTMF Feedback Signals For VRU? n	Vectoring (Basic)? y
Dynamic Advocate? n	Vectoring (Prompting)? y
Expert Agent Selection (EAS)? y	Vectoring (G3V4 Enhanced)? y
EAS-PHD? y	Vectoring (3.0 Enhanced)? y
Forced ACD Calls? n	Vectoring (ANI/II-Digits Routing)? y
Least Occupied Agent? n	Vectoring (G3V4 Advanced Routing)? y
Lookahead Interflow (LAI)? n	Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n	Vectoring (Best Service Routing)? n
Multiple Call Handling (Forced)? n	Vectoring (Holidays)? n
PASTE (Display PBX Data on Phone)? n	Vectoring (Variables)? n
(NOTE: You must logoff & login 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		Page 1 of 3
AGENT LOGINID		
Login ID: 53001	AAS? n	
Name: 53001-9650-Agent	AUDIX? n	
TN: 1	LWC Reception: spe	
COR: 1	LWC Log External Calls? n	
Coverage Path: 1	AUDIX Name for Messaging:	
Security Code:	LoginID for ISDN/SIP Display? n	
	Password:	
	Password (enter again):	
	Auto Answer: station	
	MIA Across Skills: system	
	CW Agent Considered Idle: system	
	Aux Work Reason Code Type: system	
	Logout Reason Code Type: system	
	Maximum time agent in ACW before logout (sec): system	
	Forced Agent Logout Time: :	
WARNING: Agent must log in again before changes take effect		

Agent Form – Page 1

display agent-loginID 53001		Page 2 of 3
AGENT LOGINID		
Direct Agent Skill:	Service Objective? n	
Call Handling Preference: skill-level	Local Call Preference? n	
SN RL SL	SN RL SL	
1: 11 1	16: 31: 46:	
2:	17: 32: 47:	
3:	18: 33: 48:	

Agent Form – Page 2

display hunt-group 11		Page 1 of 4
HUNT GROUP		
Group Number: 11	ACD? y	
Group Name: Skill-11	Queue? y	
Group Extension: 53011	Vector? y	
Group Type: ead-mia		
TN: 1		
COR: 1	MM Early Answer? n	
Security Code:	Local Agent Preference? n	
ISDN/SIP Caller Display:		
Queue Limit: unlimited		
Calls Warning Threshold:	Port:	
Time Warning Threshold:	Port:	

Skill (Hunt Group) Form – Page 1

display hunt-group 11		Page 2 of 3
HUNT GROUP		
Skill? y	Expected Call Handling Time (sec): 180	
AAS? n		
Measured: none		
Supervisor Extension:		
Controlling Adjunct: none		
Multiple Call Handling: none		
Time ACW Interval (sec):	After Xfer or Held Call Drops? n	

Skill (Hunt Group) Form – Page 2

display vdn 2010		Page 1 of 3
VECTOR DIRECTORY NUMBER		
Extension: 2010		
Name: To SelectSkill		
Destination: Vector Number	10	
Attendant Vectoring? n		
Meet-me Conferencing? n		
Allow VDN Override? n		
COR: 1		
TN#: 1		
Measured: none		
VDN of Origin Annc. Extension*:		
1st Skill*:		
2nd Skill*:		
3rd Skill*:		
* Follows VDN override rules		

VDN (Vector Directory Number) Form

display vector 10

Page 1 of 6

```

                                CALL VECTOR
                                Name: RouteToSkill
Number: 10
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
Basic? y      EAS? n      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
Variables? y      3.0 Enhanced? y
01 wait-time      2      secs hearing ringback
02 collect      1      digits after announcement 33002      for none
03 goto vector    11      @step 2 if digits      =      1
04 goto vector    12      @step 2 if digits      =      2
05 goto vector    13      @step 2 if digits      =      3
06 goto step      2      if unconditionally
```

Vector (RouteToSkill) Form

display vector 11

Page 1 of 6

```

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

Vector (Skill 11) Form

6.10.2. CM Messaging Coverage Path and Hunt Group

Hunt group 1 is used in the reference configuration to verify CM Messaging coverage functionality. This hunt group is defined with the 5 digit CM Messaging pilot number **55000**. 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 6.2**) to dial CM Messaging (e.g. **855000**) as shown on **hunt-group** form below.

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

Coverage Path Form

display hunt-group 1		Page 1 of 60
HUNT GROUP		
Group Number: 1		ACD? n
Group Name: CM Messaging		Queue? n
Group Extension: 55000		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: grp-name		

Hunt Group Form – Page 1

display hunt-group 1		Page 2 of 60
HUNT GROUP		
Message Center: sip-adjunct		
Voice Mail Number	Voice Mail Handle	Routing Digits
		(e.g., AAR/ARS Access Code)
55000	55000	8

Hunt Group Form – Page 2

display station 50001		Page 1 of 5
STATION		
Extension: 50001	Lock Messages? n	BCC: 0
Type: 9620	Security Code: 123456	TN: 1
Port: S00000	Coverage Path 1: 1	COR: 1
Name: H323-96XX-50001	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
Speakerphone: 2-way	Message Lamp Ext: 50001	
Display Language: english	Mute Button Enabled? y	
Survivable GK Node Name:	Button Modules: 0	
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. Configure Acme Packet Session Border Controller

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 [7] for further details and explanations on the configuration below.

ANNOTATION: The local policies below governs the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Session Manager, Communication Manager, etc., reside to the AT&T IP services. The Session Agent Groups (**SAG**) defined here, and further down, provisioned under the session-groups **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
  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-recursion disabled
    carrier
    start-time       0000
    end-time          2400
    days-of-week      U-S
    cost              0
    app-protocol
    state            enabled
    methods
    media-profiles
    lookup            single
    next-key
    eloc-str-lkup     disabled
    eloc-str-match
```

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

```
local-policy
```

from-address	*
to-address	*
source-realm	ATT
description	
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
policy-attribute	
next-hop	10.80.130.122
realm	Enterprise
action	none
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
app-protocol	SIP
state	enabled
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	
network-interface	
name	wancom0
sub-port-id	0
description	
hostname	
ip-address	192.9.230.221
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.0
gateway	192.9.230.254
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	
ftp-address	
icmp-address	
snmp-address	
telnet-address	

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 reside.

```
network-interface
  name          s1p0
  sub-port-id   0
  description
  hostname
  ip-address     192.168.62.50
  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	192.168.62.50
ftp-address	
icmp-address	192.168.62.50
snmp-address	
telnet-address	
ssh-address	

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP services reside, and applies the SIP manipulation **NAT_IP**. 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	slp0:0
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	NAT_IP
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-multiplier	0
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
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

<p>ANNOTATION: The realm configuration Enterprise below represents the internal network on which the Avaya elements reside.</p>

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-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	
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-multiplier	0
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          SM63
  ip-address        10.80.130.122
  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-sessions 0
  max-outbound-sessions 0
  max-burst-rate     0
  max-inbound-burst-rate 0
  max-outbound-burst-rate 0
  max-sustain-rate   0
  max-inbound-sustain-rate 0
  max-outbound-sustain-rate 0
  min-seizures       5
  min-asr             0
  time-to-resume      0
  ttr-no-response     0
  in-service-period   0
  burst-rate-window   0
  sustain-rate-window 0
  req-uri-carrier-mode None
  proxy-mode
  redirect-action     Proxy
  loose-routing       enabled
  send-media-session  enabled
  response-map
  ping-method         OPTIONS;hops=1
  ping-interval       180
  ping-send-mode      keep-alive
  ping-all-addresses disabled
  ping-in-service-response-codes
  out-service-response-codes
  media-profiles
  in-translationid
  out-translationid
  trust-me            enabled
  
```

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	TCP
tcp-keepalive	enabled
tcp-reconn-interval	10
max-register-burst-rate	0
register-burst-window	0
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	
hostname	135.242.225.210
ip-address	135.242.225.210
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	UDP
realm-id	ATT
egress-realm-id	
description	
carriers	
allow-next-hop-ip	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0

max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;hops=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	
trust-me	enabled
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-window	0
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	ATT-Failover
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;hops=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	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	

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-window	0
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	
group-name	SP_PROXY
description	
state	enabled
app-protocol	SIP
strategy	RoundRobin
dest	1.1.1.1
	135.242.225.210
trunk-group	
sag-recursion	enabled
stop-sag-recurse	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	192.168.62.50
port	5060
transport-protocol	UDP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	0

invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	
manipulation-string	
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 interface below is used to communicate with the Avaya elements.

sip-interface

state	enabled
realm-id	Enterprise
description	
sip-port	
address	10.80.130.250
port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	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 manipulations shown below are used for modifying several headers (To, From and Contact) to hide the CPE topology.

sip-manipulation

name	NAT_IP
description	Topology hiding for To, From headers
split-headers	
join-headers	
header-rule	
name	manipFrom
header-name	From
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	
match-value	
new-value	
element-rule	
name	FROM
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	\$LOCAL_IP
header-rule	
name	manipTo
header-name	To
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	
match-value	
new-value	
element-rule	
name	TO
parameter-name	
type	uri-host

action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	\$REMOTE_IP

header-rule

name	modContactPlus
header-name	Contact
action	manipulate
comparison-type	pattern-rule
msg-type	any
methods	INVITE
match-value	
new-value	

element-rule

name	modVal
parameter-name	
type	uri-user
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	\+(.*)
new-value	\$modContactPlus.\$modVal.\$1

ANNOTATION: The SIP header manipulation shown below modifies the **sendonly** value in SDP to **sendrecv** using header rule **modsendonly**. See **Section 2.2, Item 1** for further details.

header-rule

name	modsendonly
header-name	Content-type
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	

element-rule

name	modmline
parameter-name	application/sdp
type	mime
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	sendonly
new-value	sendrecv

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

sip-manipulation

name	rejectOptions
description	
split-headers	
join-headers	
header-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	
ip-address	192.168.62.50
start-port	16384
end-port	32767
realm-id	ATT
steering-pool	
ip-address	10.80.130.250
start-port	16384
end-port	32767
realm-id	Enterprise
system-config	
hostname	Enterprise-Acme
description	
location	
mib-system-contact	
mib-system-name	
mib-system-location	
snmp-enabled	enabled
enable-snmp-auth-traps	disabled
enable-snmp-syslog-notify	disabled
enable-snmp-monitor-traps	disabled
enable-env-monitor-traps	disabled
snmp-syslog-his-table-length	1
snmp-syslog-level	WARNING
system-log-level	WARNING
process-log-level	NOTICE
process-log-ip-address	0.0.0.0
process-log-port	0
collect	
sample-interval	5
push-interval	15
boot-state	disabled
start-time	now
end-time	never
red-collect-state	disabled
red-max-trans	1000
red-sync-start-time	5000
red-sync-comp-time	1000

push-success-trap-state	disabled
call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	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

8. Verification Steps

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

8.1. 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.

8.2. Avaya Aura® Session Manager

Navigate to **Home**→**Elements**→**Session Manager**→**System Status**→**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.

The screenshot shows the Avaya Aura System Manager 6.2 interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, and System Status. The main content area is titled 'SIP Entity, Entity Link Connection Status' and shows a table of entity links. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto, Conn. Status, Reason Code, and Link Status. The first row shows 'DenverSM' with IP '10.80.130.250', Port '5060', Proto 'TCP', Conn. Status 'Up', Reason Code '405 Method Not Allowed', and Link Status 'Up'.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	DenverSM	10.80.130.250	5060	TCP	Up	405 Method Not Allowed	Up

8.3. AT&T IP Toll Free

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

9. 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.

10. References

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

Avaya Aura® Session Manager/System Manager

- [1] Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.3, December 2012
- [2] Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473 Issue 2, November 2010
- [3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.3, December 2012
- [4] Administering Avaya Aura® System Manager, Release 6.3, Issue 1.0, December 2012

Avaya Aura® Communication Manager

- [5] Administering Avaya Aura® Communication Manager, Issue 7.0, Release 6.2, December 2012, 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

Acme Packet Support (login required):

- [7] <http://www.acmepacket.com/support.htm>

AT&T IP Toll Free Service Descriptions:

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

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