



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Experience Portal 6.0, Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service using MIS/PNT or AVPN Transport – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Experience Portal, Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Acme Packet Net-Net (models 3800, 4250, or 4500) with the AT&T IP Toll Free service using **MIS/PNT** or **AVPN** transport connection.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. Avaya Aura® Experience Portal is a speech-enabled Interactive Voice Response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner.

An Acme Packet Net-Net is the point of connection between Avaya Aura® Session Manager 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 signaling for interoperability. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Experience Portal, Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Acme Packet Net-Net (models 3800, 4250, or 4500) with the AT&T IP Toll Free service using **MIS/PNT** or **AVPN** transport connection. **Note that the configuration steps in these Application Notes are used for this reference configuration and not meant to be prescriptive.**

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. Avaya Aura® Experience Portal is a speech-enabled Interactive Voice Response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner.

An Acme Packet Net-Net is the point of connection between Avaya Aura® Session Manager 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 signaling for interoperability.

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

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site with Experience Portal, Communication Manager, Session Manager, System Manager, Avaya phones, an Acme Session Border Controller, an Apache Tomcat application server, and a speech server (Nuance Recognizer and Nuance Vocalizer).
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise site was connected via MIS/PNT or AVPN transport connection.

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

- Inbound calls to various Experience Portal applications.
- Inbound caller interaction with Experience Portal applications, including prompting, caller DTMF input, wait treatment (e.g., music on hold), Automatic Speech Recognition, and Text to Speech.
- Experience Portal applications canvassing of Communication Manager for skilled agent availability before transferring inbound calls to the skills.
- Experience Portal applications transferring of inbound calls to Communication Manager skilled agent regardless of agent's availability.
- Call and two-way talkpath establishment between callers and Communication Manager agents following transfers from Experience Portal.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729a and G.711 codec support.
- Inbound AT&T IP Toll Free calls to Experience Portal destined for agents/stations connected to Communication Manager, if unanswered, are covered to Messaging.
- Experience Portal applications sending DTMF to the AT&T IP Toll Free to invoke AT&T IP Toll Free Legacy Transfer Connect features (only those permitted for Voice Response Units) and processing the resulting DTMF responses from the AT&T IP Toll Free service.
- Inbound calls to a self service Experience Portal application which forwards the call to 8YY or any other PSTN number over AT&T IP Flex Reach network.
- Long duration calls.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for sample call flows) between Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya Aura® Experience Portal, Acme Packet Net-Net, 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
- Passing of DTMF events and their recognition by navigating automated voice menus
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer
- Legacy Transfer Connect
- Alternate Destination Routing

2.2. Known Limitations/Test Results

1. AT&T IP Transfer Connect option of the AT&T IP Toll Free service was not verified with Experience Portal 6.0 and hence not supported.
2. G.726 codec is not supported by Experience Portal 6.0.
3. 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, G.729A, and G.711 in that order, then Communication Manager selects G.729A, 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.
4. Although Experience Portal release 6.0 and Communication Manager 6.0.1 support the possibility of using SIP phones as valid telephone extensions, SIP phones were not tested as part of the configuration used to validate this solution.
5. A slight delay in ringback was observed on Calling Party telephone when the call is transferred from Experience Portal to an agent on Communication Manager.
6. For an outcall to an 8YY number from Experience Portal, the Experience Portal application needs to add a Diversion Header otherwise AT&T network will send a 403 Forbidden message back and the call will fail. This diversion header can also be added on the Acme SBC as shown in **Section 9**, and that was the way it was implemented in this reference configuration.

The test objectives stated in **Section 2** with limitations as 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 sample configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Experience Portal provides interactive voice response services to inbound callers. Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Manager (EPM) server. Single server was used for MPP and EPM for this reference configuration.
- Communication Manager provides the enterprise voice communications services. In this sample configuration, Communication Manager runs on an Avaya S8800 Server.
- Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Managers in an enterprise.
- The Avaya G650 Media Gateway provides the physical interfaces and resources for enterprise voice communications. This solution is extensible to other Avaya Media Gateways.
- Avaya phones are represented with Avaya 96xx Series IP Telephones running H.323 software. Additionally Avaya one-X® Agent and Analog and Digital phones were also used.
- The Acme Packet Net-Net Session Director (SD) 3800¹ provides SIP Session Border Controller (Acme SBC) functionality between the AT&T IP Toll Free service and the enterprise internal network². UDP transport protocol is used between the Acme Packet Net-Net SD and the AT&T IP Toll Free service.
- The Apache Tomcat Application Server hosts the VXML and CCXML applications that provide the directives for handling the inbound calls to Experience Portal which are referenced in Experience Portal.
- The Speech Server consists of Nuance Recognizer and Nuance Vocalizer. Experience Portal uses the Speech Server for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) capabilities.
- Messaging provides the corporate voice messaging capabilities in this reference configuration. The provisioning of Messaging is beyond the scope of this document.

¹ Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

² The AT&T IP Toll Free service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Acme Packet SBC in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme SBC and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

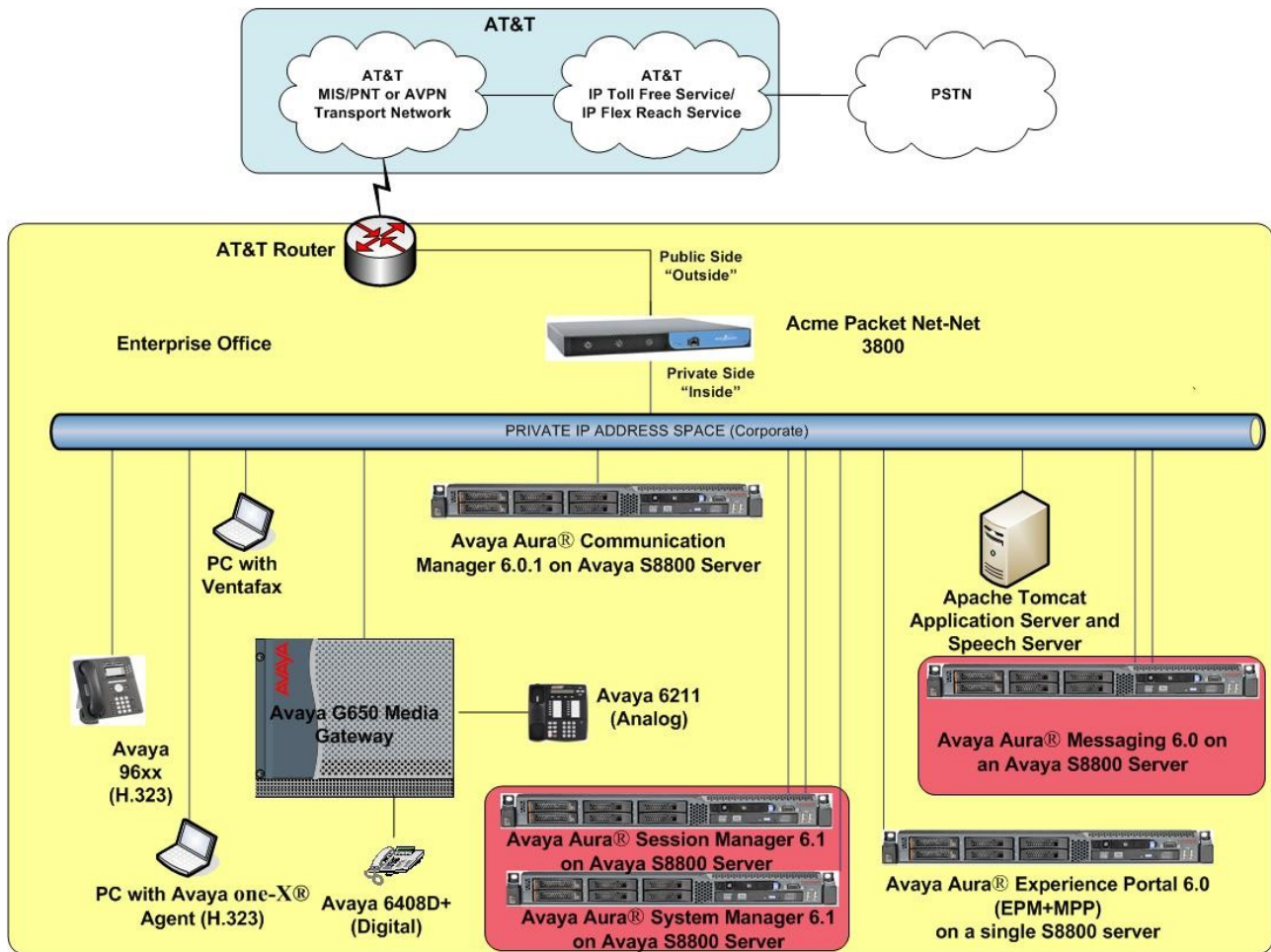


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table** below and in subsequent sections are used in this reference configuration, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Component	Illustrative Value in these Application Notes
Avaya Aura® Experience Portal	
EPM/MPP Servers IP Address	10.80.130.220
Automatic Speech Recognition and Text to Speech server IP Address	10.80.130.153
Avaya Aura® Communication Manager	
C-LAN IP Address	10.8.130.206
Vector Directory Number (VDN) Extensions	666-20xx
Skill (Hunt Group) Extensions	666-40xx
Agent Extensions	666-30xx
Phone Extensions	666-50xx
Announcement Extensions	666-10xx
Avaya Aura® Session Manager/System Manager	
System Manager IP Address	10.80.150.204
Session Manager Management IP Address	10.80.150.205
Session Manager Network IP Address	10.80.150.206
Acme Packet Session Border Controller	
IP Address of “Outside” Interface (connected to AT&T IP Toll Free Service)	192.168.62.50
IP Address of “Inside” Interface (connected to Avaya elements)	10.80.130.250
AT&T IP Toll Free Service	
Border Element IP Address	135.242.225.210
DNIS Passed in Request URI used by Session Manager for routing	00000[1,2,3,4,5]100[1,2,3,4,5]
Digits Passed in SIP “To” Header to Avaya Aura® Experience Portal	800555xxxx

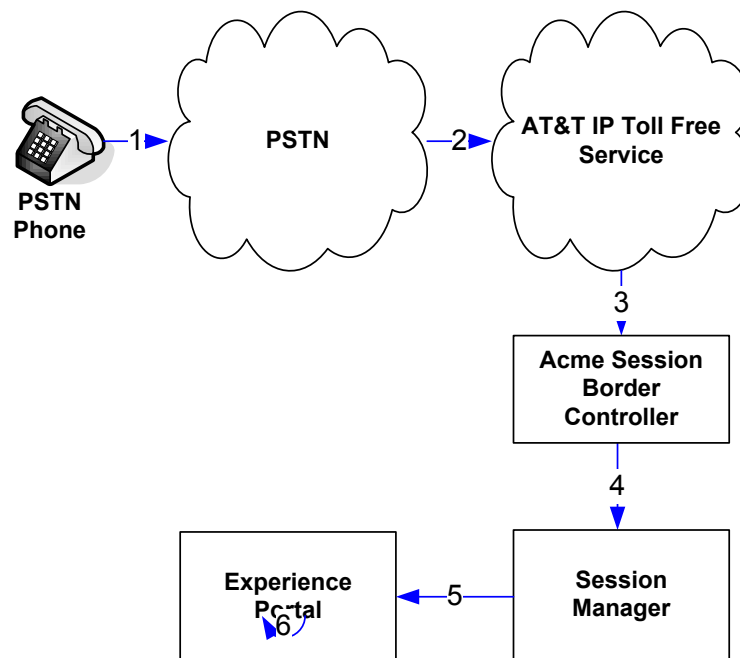
Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IP Toll Free calls are handled by Experience Portal, several call flows are described in this section.

The first call scenario illustrated below is an inbound call arriving and remaining on Experience Portal.

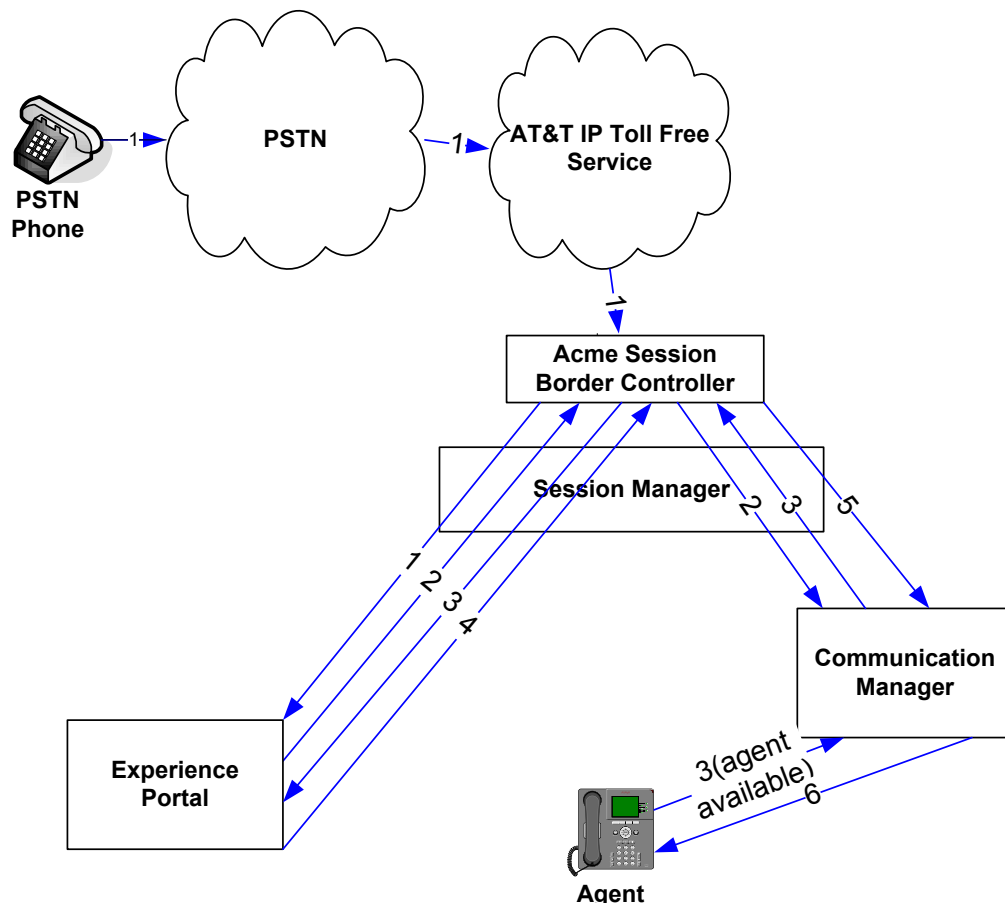
1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to the Acme SBC.
4. Acme SBC performs any necessary SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Experience Portal.
6. Experience Portal matches the called party number to a VXML and/or CCXML application, answers the call, and handles the call according to the directives specified in the application. In this scenario, the application sufficiently meets the caller's needs or requests, and thus the call does not need to be transferred to Communication Manager.



Inbound Call Handled Entirely by Experience Portal

The second call scenario illustrated below is an inbound call arriving on Experience Portal and transferred to Communication Manager only after an agent with appropriate skill becomes available on Communication Manager.

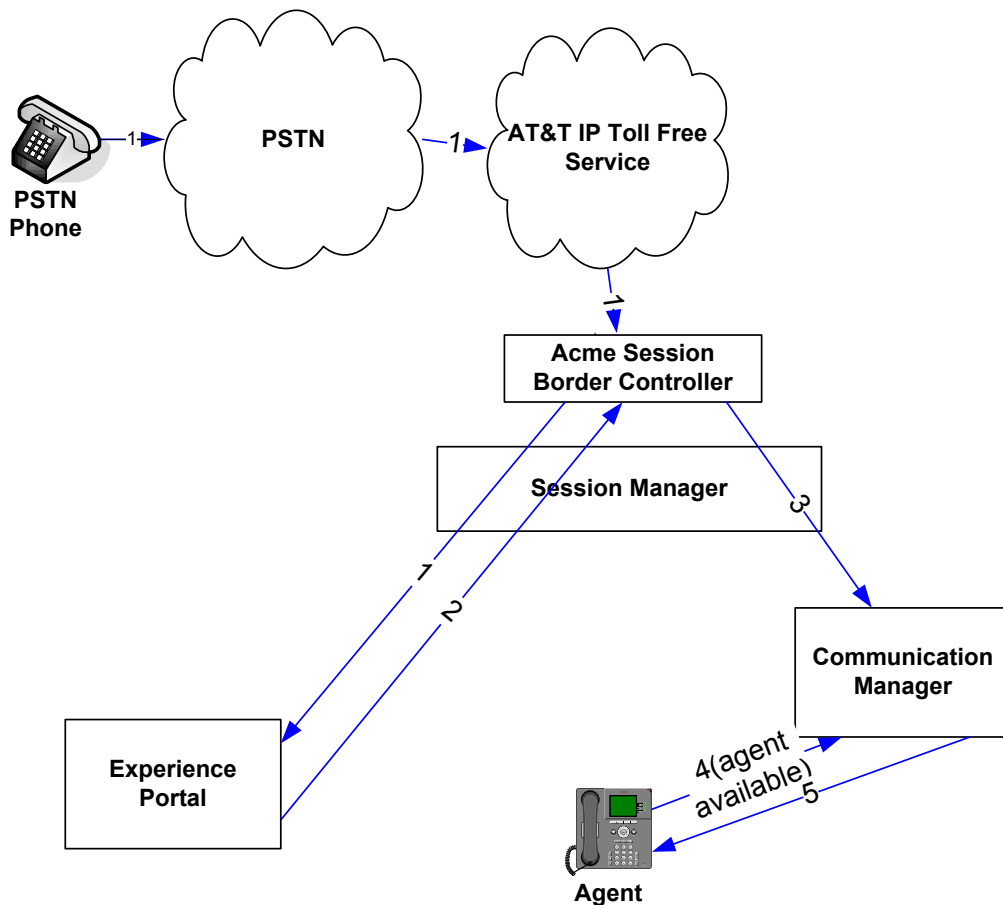
1. Same as the first five steps from the first call scenario.
2. In this scenario, the application is not sufficient to meet the caller's requests, and thus the call needs to be transferred to a Communication Manager agent. Experience Portal then puts the inbound call on hold and places a call to vector/skill for an agent on Communication Manager via Acme SBC/Session Manager. While the inbound call is on hold, Experience Portal may play music to the caller, prompt the caller for additional information, or otherwise interact with the caller.
3. Communication Manager informs Experience Portal when an agent in that skill becomes available.
4. Experience Portal instructs the Acme SBC to transfer the inbound call to that skill.
5. The Acme SBC transfers the inbound call to the aforementioned skill on Communication Manager.
6. Communication Manager routes the call to the agent.



Inbound Call Handled by Experience Portal and Transferred to Communication Manager upon Agent Availability

The third call scenario illustrated below is an inbound call arriving on Experience Portal and transferred to Communication Manager skill without determining whether an agent with required skill is available or not.

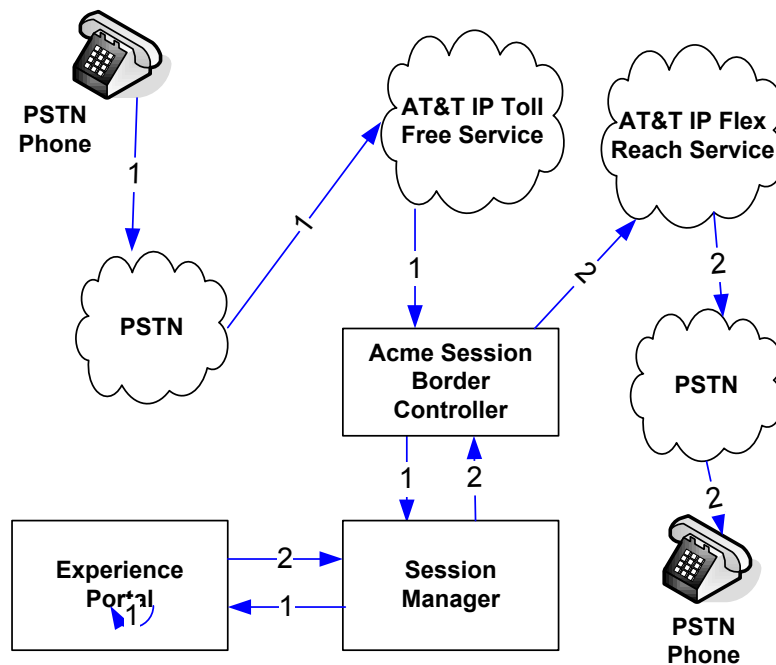
1. Same as the first five steps from the first call scenario.
2. In this scenario, the application on Experience Portal is not sufficient to meet the caller's needs or requests, and thus the call needs to be transferred to an agent/skill on Communication Manager. Experience Portal instructs the Acme SBC via Session Manager to transfer the inbound call to an agent/skill on Communication Manager without verifying that an agent with required skill is available on Communication Manager.
3. The Acme SBC transfers the inbound call to the required skill/agent on Communication Manager.
4. An agent becomes available on Communication Manager.
5. Communication Manager routes the call to the agent.



Inbound Call Transferred by Experience Portal to Communication Manager regardless of Agent Availability

The fourth call scenario illustrated below is an inbound call arriving on Experience Portal and forwarded to an 8YY number or any other PSTN number over AT&T Flex Reach network.

1. Same as the first six steps from the first call scenario.
2. In this scenario, the application is sufficient to meet the caller's requests, and thus the call needs to be forwarded to another PSTN number. Based upon the selection, Experience Portal forwards the call to an appropriate PSTN number which can be a regular PSTN number or an 8YY number.



Inbound Call forwarded by Experience Portal to another PSTN number

4. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura® Experience Portal 6.0
Experience Portal Management (EPM)	EPM 6.0.0.0.3306
Media Processing Platform (MPP)	MPP 6.0.0.0.3401
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 with Service Pack 5 (R016x.00.1.510.1)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW06 FW054
TN799DP Control-LAN (C-LAN)	HW01 FW040
TN2602AP IP Media Processor (MedPro)	HW02 FW061
TN2501AP VAL-ANNOUNCEMENT	HW02 FW018
Avaya S8800 Server	Avaya Aura® System Manager 6.1.0 (SP5)
Avaya S8800 Server	Avaya Aura® Session Manager 6.1.0 (SP5)
Avaya 9650 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.110b
Avaya 9611 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release S6.0.0
Avaya one-X® Agent	Release 2.5
Apache Tomcat Application Server	6.0.33
Nuance Recognizer	9.0
Nuance Recognizer English en-US Language Pack	9.0
Nuance Vocalizer	5.0.5
Nuance Vocalizer American English en-US Donna	5.0.2
Nuance MediaServer	5.0.5
Acme Packet Net-Net Session Director 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Toll Free Service	VNI 22

Table 1: Equipment and Software Versions

5. Avaya Aura® Session Manager

These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Consult [4] and [5] for further details if necessary. Configuration of Session Manager is performed from System Manager. To invoke the System Manager Common Console, launch a web browser. Enter `https://<IP address of the System Manager server>/` in the URL field, and log in with the appropriate credentials.

5.1. Background

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

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

5.2. Routing Policies

Routing Policies define how Session Manager will route calls between SIP network elements. Routing Policies are dependent on the administration of several inter-related items:

- **SIP Entities** – SIP Entities represent SIP network elements such as Session Managers, Communication Managers, Session Border Controllers, SIP gateways, SIP trunks, and other SIP network devices.
- **Entity Links** – Entity Links define the SIP trunk/link parameters, e.g., ports, protocol (UDP/TCP/TLS), and trust relationship, between Session Manager and other SIP Entities.
- **SIP Domains** – SIP Domains are the domains for which Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Session Manager applies Routing Policies to route those calls to SIP Entities. For calls to other domains, Session Manager routes those calls to another SIP proxy (either a pre-defined default SIP proxy or one discovered through DNS).
- **Locations** – Locations define the physical and/or logical locations in which SIP Entities reside. Call Admission Control (CAC) / bandwidth management may be administered for each location to limit the number of calls to and from a particular Location.

- Adaptations – Adaptations are used to apply any necessary protocol adaptations, e.g., modify SIP headers, and apply any necessary digit conversions for the purpose of inter-working with specific SIP Entities. As another example, basic “Digit Conversion” Adaptations are used in this reference configuration to convert digit strings in **destination** (e.g., Request-URI) and **origination** (e.g. P-Asserted Identity) type headers of SIP messages sent to and received from SIP Entities.
- Dial Patterns – A Dial Pattern specifies a set of criteria and a set of Routing Policies for routing calls that match the criteria. The criteria include the called party number and SIP domain in the Request-URI, and the Location from which the call originated. For example, if a call arrives at Session Manager and matches a certain Dial Pattern, then Session Manager selects one³ of the Routing Policies specified in the Dial Pattern. The selected Routing Policy in turn specifies the SIP Entity to which the call is to be routed. Note that Dial Patterns are matched after ingress Adaptations have already been applied.
- Time Ranges – Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Routing Policy may be associated with one or more Time Ranges during which the Routing Policy is in effect. For example, for a Dial Pattern administered with two Routing Policies, one Routing Policy can be in effect on weekday business hours and the other Routing Policy can be in effect on weekday off-hours and weekends. **In the reference configuration no restrictions were placed on calling times.**

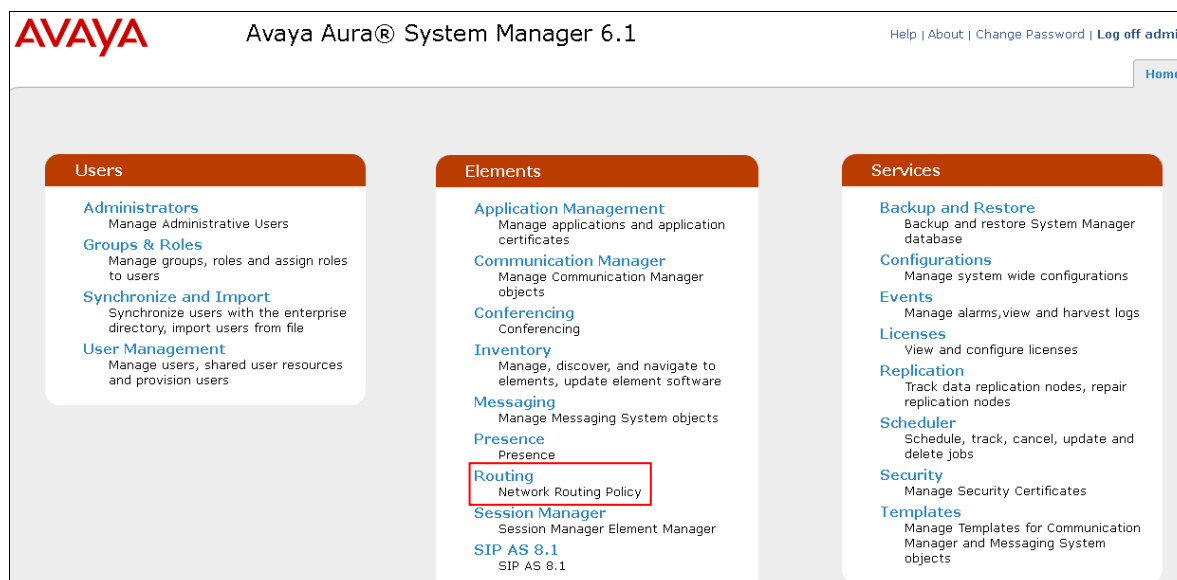
The general strategy employed in this reference configuration with regard to Called Party Number manipulation and matching, and call routing is as follows:

- Use common number formats and uniform numbers in matching called party numbers for routing decisions.
- On ingress, Session Manager may apply any called party number modifications necessary to **normalize** the number to a common format or uniform number as defined in the Dial Patterns.
- On egress, Session Manager may apply any called party number modifications necessary to conform to the expectations of the next-hop SIP Entity.

Of course, the items above are just several of many possible strategies that can be implemented with Session Manager.

³ The Routing Policy in effect at that time with highest ranking is attempted first. If that Routing Policy fails, then the Routing Policy with the next highest rankings is attempted, and so on.

To view the sequenced steps required for configuring network routing policies, click **Routing** on the System Manager Common Console (see below).

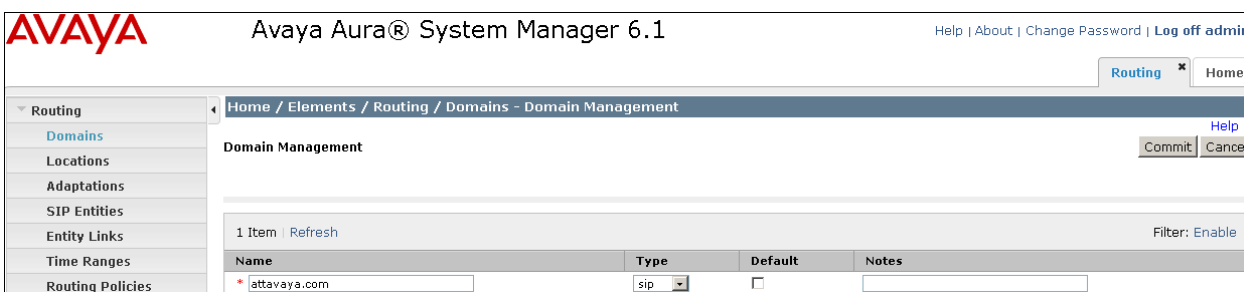


System Manager Common Console Page

5.3. SIP Domains

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

1. In the left pane under **Routing**, click **Domains**. On the **Domain Management** page, click on **New** [not shown] and configure as follows:
 - **Name** – Set to **attavaya.com** in this reference configuration. This domain is used in **Section 6.2, Step 4** and **Section 7.4, Step 1**.
 - **Type** – Set to **sip**.
 - **Notes** – Optional Field.
2. Click **Commit**.
3. Repeat above steps to add additional domains.



Domain Management Page

5.4. Locations

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

1. In the left pane under **Routing**, click on **Locations**. On the **Location** page [not shown] click **New**.
2. On the **Location Details** page, configure as follows:
 - **Name** – Enter any descriptive string.
 - **Notes** – [Optiona] Enter a description.
 - **Managed Bandwidth** and **Average Bandwidth per Call** – [Optiona] To limit the number of calls going to and from this location i.e., apply Call Admission Control.
 - **Location Pattern** – [Optiona] To identify IP addresses associated with this Location. In the reference configuration, the IP address of Acme SBC i.e. **10.80.130.250** was used.
3. Click **Commit**.
4. Repeat above steps to add any additional Locations (e.g. **Location_130** for **Experience Portal, Messaging and Communication Manager** matching on **IP Address Pattern** of **10.80.130.***, and **Location_150** for **Session Manager** matching on **IP Address Pattern** of **10.80.150.***) used in this reference configuration.

The screenshot shows the 'Location Details' page in the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with 'Routing' selected, and sub-items like 'Domains', 'Locations', 'Adaptations', etc. The main content area is titled 'Location Details' and includes a breadcrumb trail 'Home / Elements / Routing / Locations - Location Details'. Below the title, there's a 'General' section with fields for 'Name' (set to 'Acme_SBC_130') and 'Notes' (set to 'SBC to ATT'). There are 'Commit' and 'Cancel' buttons. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' as 'Kbit/sec' and 'Total Bandwidth' as an empty field. The 'Per-Call Bandwidth Parameters' section shows 'Default Audio Bandwidth' as '80 Kbit/sec'. The 'Location Pattern' section has 'Add' and 'Remove' buttons, and a table with one item: '10.80.130.250'. The table has columns for 'IP Address Pattern' and 'Notes'. At the bottom, there's a 'Select : All, None' option.

IP Address Pattern	Notes
10.80.130.250	

Location Details Page for Acme SBC

5.5. Adaptations

Adaptations on Session Manager are always between Session Manager and another entity. Adaptations could potentially be applied to both calls coming into Session Manager and going out from the Session Manager. In this section, Adaptations are administered for calls from AT&T to

Experience Portal (Section 5.5.1) and the calls forwarded from Experience Portal to Communication Manager (Section 5.5.2).

5.5.1. Adaptation for Calls to Avaya Aura® Experience Portal

This adaptation replaces the IP address of Session Manager in Request URI and **To** header with the Avaya CPE SIP domain **attavaya.com**.

1. In the left pane under **Routing**, click **Adaptations**. On the **Adaptations** page, click on **New** [not shown].
2. In the **Adaptation Details – General** section, configure as follows:
 - **Adaptation name** – Set to any descriptive string.
 - **Module name** - Select **DigitConversionAdapter** from the drop-down list; if no module name is present, select <click to add module> and enter **DigitConversionAdapter**.
 - **Module parameter** - Enter **fromto=true iodstd=attavaya.com odst=135.242.225.210**, which will replace the IP Address/Domain in the Request URI and **To** header with the Avaya CPE domain **attavaya.com** for egress to Experience Portal. Also, it replaces the domain in the calls originating from Experience Portal destined for Acme SBC to the IP Address of the AT&T Border element.
3. In the **Adaptation Detail - Digit Conversion for Incoming Calls to SM** section, configure as follows to remove a leading + sign in the user part of Request URI:
 - **Matching Pattern** – Set to match the first character user part of Request URI.
 - **Min** and **Max** – Set to **8**.
 - **Delete Digits** – Set to **1**.
 - **Address to modify** – Select **destination** from the drop-down list.
4. Click **Commit**.

Note: In the reference configuration no **Digit Conversation for Outgoing Calls from SM** are required.

The screenshot shows the Avaya Aura® System Manager 6.1 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations (highlighted), SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Adaptations - Adaptation Details'. Below this, the 'General' section is active, showing fields for 'Adaptation name' (AT&T Adaptations), 'Module name' (DigitConversionAdapter), and 'Module parameter' (fromto=true iodstd=attavaya.com odst=135.242.225.210). There are also fields for 'Egress URI Parameters' and 'Notes'. At the bottom, the 'Digit Conversion for Incoming Calls to SM' section is visible, featuring a table with columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, and Notes. The table contains one row with values: *x, 8, 8, , 1, , destination, and Remove +. The interface also includes a 'Commit' button and a 'Cancel' button.

Adaptation Details Page – Adaptation for Acme SBC

5.5.2. Adaptation for Calls to Avaya Aura® Communication Manager

This adaptation replaces the IP address of Session Manager with the Avaya CPE SIP domain **attavaya.com** in the **PAI** header.

1. In the left pane under **Routing**, click **Adaptations**. On the **Adaptations** page, click **New** [not shown].
2. On the **Adaptation Details – General** section, configure as follows:
 - **Adaptation name** – Set to any descriptive string.
 - **Module name** - Select **DigitConversionAdapter** from the drop-down list; if no module name is present, select **<click to add module>** and enter **DigitConversionAdapter**.
 - **Module parameter** - Enter **osrcd=attavaya.com**, which will replace the IP Address/Domain in the **PAI** header for egress to Communication Manager.
3. In the **Adaptation Detail - Digit Conversion for Incoming Calls to SM** section, configure as follows to remove a leading + sign in the user part of Request URI:
 - a. **Matching Pattern** – Set to match the first character user part of Request URI.
 - b. **Min** and **Max** – Set to **8**.
 - c. **Delete Digits** – Set to **1**.
 - d. **Address to modify** – Select **destination** from the drop-down list.
4. Click **Commit**.

Note: In the reference configuration no **Digit Conversation for Outgoing Calls from SM** are required.

The screenshot shows the Avaya Aura® System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and 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", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The main content area is titled "Adaptation Details" and has a breadcrumb trail: "Home / Elements / Routing / Adaptations - Adaptation Details". There are "Commit" and "Cancel" buttons in the top right. The "General" section contains the following fields:

- * Adaptation name: ATT CLAN
- Module name: DigitConversionAdapter (selected from a dropdown)
- Module parameter: osrcd=attavaya.com
- Egress URI Parameters: (empty field)
- Notes: (empty field)

The "Digit Conversion for Incoming Calls to SM" section has "Add" and "Remove" buttons. Below this is a table with 1 item. The table has columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, and Notes. The first row shows:

- Matching Pattern: *x
- Min: 8
- Max: 8
- Phone Context: (empty)
- Delete Digits: 1
- Insert Digits: (empty)
- Address to modify: destination (selected from a dropdown)
- Notes: Remove +

Adaptation Details Page – Adaptation for Communication Manager

5.6. SIP Entities

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

- Avaya Aura® Session Manager
- Avaya Aura® Experience Portal
- Avaya Aura® Communication Manager
- Acme Session Border Controller
- Avaya Aura® Messaging

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

5.6.1. Avaya Aura® Session Manager SIP Entity

1. In the left pane under **Routing**, click **SIP Entities**. In the **SIP Entities** page click **New** [not shown].
2. In the **General** section of the **SIP Entity Details** page, configure as follows:
 - **Name** – Enter a descriptive name for Session Manager.
 - **FQDN or IP Address** – Enter the IP address of the Session Manager network interface, (*not* the management interface), provisioned during installation. Set to **10.80.150.206** in this reference configuration.
 - **Type** – Select **Session Manager**.
 - **Location** – Select **Location_150_SM** as configured in **Section 5.4**.
 - **Outbound Proxy** – (Optional) Leave blank or select another SIP Entity. For calls to SIP domains for which Session Manager is not authoritative, Session Manager routes those calls to this **Outbound Proxy** or to another SIP proxy discovered through DNS if **Outbound Proxy** is not specified.
 - **Time Zone** – Select the time zone in which Session Manager resides.
3. In the **SIP Link Monitoring** section of the **SIP Entity Details** page select **Use Session Manager Configuration** for the **SIP Link Monitoring** field.
4. In the **Port** section of the **SIP Entity Details** page, click on **Add** and provision as follows:
 - **Port** – Enter **5060** (see note above).
 - **Protocol** – Select **TCP** (see note above).
 - **Default Domain** – (Optional) Select a SIP domain administered in **Section 5.3**.
 - Repeat this step to configure additional port entries.
5. The screen below also shows all the entity links configured for this entity. These Entity links are actually configured/displayed in **Section 5.7**.
6. Click **Commit**.

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

SIP Entity Details

[Help ?](#)
[Commit](#)
[Cancel](#)

General

* Name: ASM

* FQDN or IP Address: 10.80.150.206

Type: Session Manager

Notes: Session Manager

Location: Location_150_CM

Outbound Proxy:

Time Zone: America/Denver

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Entity Links

Add

Remove

4 Items

Refresh

Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	ASM	TCP	* 5060	CM6.0.1-ATT-CLAN1A02	* 5060	Trusted
<input type="checkbox"/>	ASM	TCP	* 5061	Messaging	* 5060	Trusted
<input type="checkbox"/>	ASM	TCP	* 5060	AEP6.0	* 5060	Trusted
<input type="checkbox"/>	ASM	TCP	* 5090	AcmeSBCATT-5090	* 5090	Trusted

Select : All, None

Port

Add

Remove

2 Items

Refresh

Filter: Enable

	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	attavaya.com	
<input type="checkbox"/>	5090	TCP	attavaya.com	

SIP Entity Details Page –Session Manager SIP Entity

5.6.2. Avaya Aura® Communication Manager SIP Entity

1. In the **SIP Entities** page, click **New** [not shown].
2. In the **General** section of the **SIP Entity Details** page, configure as follows:
 - **Name** – Enter any descriptive name for the Communication Manager Signaling Interface.
 - **FQDN or IP Address** – Enter the IP address of the Communication Manager C-LAN provisioned/displayed in **Section 7.3, Step 2**.
 - **Type** – Select **CM**.
 - **Adaptation** – Select the Adaptation administered in **Section 5.5.2**.
 - **Location** – Select a Location administered in **Section 5.4**.
 - **Time Zone** – Select the time zone in which Communication Manager resides.
 - In the **SIP Link Monitoring** section of the **SIP Entity Details** page select **Use Session Manager Configuration** for **SIP Link Monitoring** field.
3. The screen below shows the entity link configured for this entity. This Entity link is actually configured/displayed in **Section 5.7.1**.
4. Click **Commit**.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

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

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

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

General

* Name: CM6.0.1-ATT-CLAN1A02

* FQDN or IP Address: 10.80.130.102

Type: CM

Notes: CM6.0.1 for ATT on CLAN 1A02

Adaptation: ATT CLAN

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

Entity Links

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

1 Item [Refresh](#) Filter: Enable

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	ASM	TCP	* 5060	CM6.0.1-ATT-CLAN1A02	* 5060	Trusted

Select : All, None

SIP Entity Details Page –Communication Manager SIP Entity

5.6.3. Acme Session Border Controller SIP Entity

To configure the Session Border Controller Entity, repeat the Steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of the private (inside) interface configured in **Section 9** under **network interface** section and the **Type** field is set to **Other**. The entity link is configured/displayed in **Section 5.7.2**. See the screen below for the values used in this reference configuration.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities - SIP Entity Details". The left sidebar contains a menu with "Routing" selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and has tabs for "General" (selected) and "SIP Link Monitoring". The "General" tab contains the following fields: "Name" (AcmeSBCATT-5060), "FQDN or IP Address" (10.80.130.250), "Type" (Other), "Notes" (Acme SBC to ATT), "Adaptation" (AT&T Adaptations), "Location" (Acme_SBC_130), "Time Zone" (America/Denver), "Override Port & Transport with DNS SRV" (unchecked), "SIP Timer B/F (in seconds)" (4), "Credential name" (empty), and "Call Detail Recording" (none). The "SIP Link Monitoring" tab shows "SIP Link Monitoring" set to "Use Session Manager Configuration". Below the tabs is the "Entity Links" section with "Add" and "Remove" buttons. A table lists the entity link with 1 item. The table has columns: SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Connection Policy. The row shows: SIP Entity 1: ASM, Protocol: TCP, Port: 5060, SIP Entity 2: AcmeSBCATT-5060, Port: 5060, Connection Policy: Trusted. At the bottom, there is a "Select : All, None" option.

SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
ASM	TCP	5060	AcmeSBCATT-5060	5060	Trusted

SIP Entity Details Page – Session Border Controller SIP Entity

5.6.4. Avaya Aura® Experience Portal Entity

To configure the Experience Portal Entity, repeat the Steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of the Experience Portal and the **Type** field is set to **Voice Portal**. The entity link is configured/displayed in **Section 5.7.3**. See the screen below for the values used in this reference configuration.

The screenshot shows the Avaya Aura® System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". The left sidebar contains a menu with "Routing" selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and "General". It contains the following fields:

- Name:** AEP6.0
- FQDN or IP Address:** 10.80.130.220
- Type:** Voice Portal (dropdown)
- Notes:** Avaya Aura Experience Portal 6.0
- Adaptation:** (dropdown)
- Location:** Location_130 (dropdown)
- Time Zone:** America/Denver (dropdown)
- Override Port & Transport with DNS SRV:** ☐
- SIP Timer B/F (in seconds):** 4
- Credential name:** (text field)
- Call Detail Recording:** none (dropdown)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)

Below the "General" section is the "Entity Links" section, which includes "Add" and "Remove" buttons. It shows a table with 1 item:

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	ASM	TCP	* 5060	AEP6.0	* 5060	Trusted

At the bottom of the table, it says "Select : All, None".

SIP Entity Details Page –Experience Portal SIP Entity

5.6.5. Avaya Aura® Messaging SIP Entity

To configure the Messaging SIP Entity, repeat the steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of Messaging and the **Type** field is set to **Modular Messaging**. The entity link is configured/displayed in **Section 5.7.4**. See the screen below for the values used in this reference configuration.

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities - SIP Entity Details". The left sidebar contains a menu with "Routing" selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" and has "General" and "SIP Link Monitoring" tabs. The "General" tab is active, showing fields for: Name (Messaging), FQDN or IP Address (10.80.150.222), Type (Modular Messaging), Notes (Aura Messaging), Adaptation (empty), Location (Location_150_CM), Time Zone (America/Denver), Override Port & Transport with DNS SRV (unchecked), SIP Timer B/F (in seconds) (4), Credential name (empty), Call Detail Recording (none), and SIP Link Monitoring (Use Session Manager Configuration). Below the tabs is an "Entity Links" section with "Add" and "Remove" buttons. A table lists the configured entity link:

1 Item Refresh							Filter: Enable
	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	
<input type="checkbox"/>	ASM	TCP	* 5060	Messaging	* 5060	Trusted	

SIP Entity Details Page –Messaging SIP Entity

5.7. Entity Links

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

- Avaya Aura® Communication Manager
- Acme Session Border Controller
- Avaya Aura® Experience Portal
- Avaya Aura® Messaging

Note – In this reference configuration TCP (port 5060) is used as the transport protocol between Session Manager and all the SIP Entities including Communication Manager. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS (port 5061) to be used as transport protocol between Experience Portal/Communication Manager and Session Manager in customer environments.

5.7.1. Entity Link to Avaya Aura® Communication Manager

1. In the left pane under **Routing**, click **Entity Links**. In the **Entity Links** page click **New** [not shown].
2. On the **Entity Links** page, provision as follows:
 - **Name** – Enter a descriptive name for this link to Communication Manager.
 - **SIP Entity 1** – Select the SIP Entity administered in **Section 5.6.1** for the Session Manager. SIP Entity 1 must always be the Session Manager instance.
 - **SIP Entity 1 Port** – Enter **5060**.
 - **SIP Entity 2** – Select the SIP Entity administered in **Section 5.6.2** for Communication Manager.
 - **SIP Entity 2 Port** - Enter **5060**.
 - **Connection Policy** – Select **Trusted**.
 - **Protocol** – Select **TCP**.
3. Click **Commit**.

The screenshot displays the Avaya Aura® System Manager 6.1 interface. The left navigation pane shows 'Routing' expanded, with 'Entity Links' selected. The main content area is titled 'Entity Links' and contains a table with one row of configuration data. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The configuration shows a link between SM-CM6.0.1-CLAN1A and CM6.0.1-ATT-CLAN1A02 using TCP on port 5060 with a 'Trusted' connection policy.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM-CM6.0.1-CLAN1A	* ASM	TCP	* 5060	* CM6.0.1-ATT-CLAN1A02	* 5060	Trusted	SM to CM6.0.1

Entity Links Page – Entity Link to Communication Manager

5.7.2. Entity Link to Acme Session Border Controller

To configure the entity link between the Session Manager and Acme SBC SIP entity, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.3**. See the screen below for the values used in this reference configuration.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The row shows: SM-AcmeSBC, ASM, TCP, 5060, AcmeSBCATT-5060, 5060, Trusted, and SM to SBC to ATT. There are buttons for 'Commit' and 'Cancel' at the top right of the table area.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM-AcmeSBC	* ASM	TCP	* 5060	* AcmeSBCATT-5060	* 5060	Trusted	SM to SBC to ATT

Entity Links Page – Entity Link to Acme SBC SIP Entity

5.7.3. Entity Link to Avaya Aura® Experience Portal

To configure this entity link, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.4**. See the screen below for the values used in this reference configuration.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The row shows: SM-AEP6.0, ASM, TCP, 5060, AEP6.0, 5060, Trusted, and SM to AEP 6.0. There are buttons for 'Commit' and 'Cancel' at the top right of the table area.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM-AEP6.0	* ASM	TCP	* 5060	* AEP6.0	* 5060	Trusted	SM to AEP 6.0

Entity Links Page – Entity Link to Experience Portal SIP Entity

5.7.4. Entity Link to Avaya Aura® Messaging

To configure this entity link, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.5**. See the screen below for the values used in the reference configuration.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and shows a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The row shows: ASM_Messaging_506, ASM, TCP, 5060, Messaging, 5060, Trusted, and an empty Notes field. There are buttons for 'Commit' and 'Cancel' at the top right of the table area.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* ASM_Messaging_506	* ASM	TCP	* 5060	* Messaging	* 5060	Trusted	

Entity Links Page – Entity Link to Messaging SIP Entity

5.8. Time Ranges

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

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Time Ranges - Time Ranges

Time Ranges

Commit Cancel

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 Ranges Page

5.9. Routing Policies

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

- Routing Policy to Avaya Aura® Experience Portal
- Routing Policy to Acme Session Border Controller
- Routing Policy to Avaya Aura® Communication Manager for calls from AT&T IP Toll Free service
- Routing Policy to Avaya Aura® Messaging

5.9.1. Routing Policy to Avaya Aura® Experience Portal

1. In the left pane under **Routing**, click **Routing Policies**. On the **Routing Policies** page click **New** [not shown].
2. In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** (e.g. **ToAEP6.0**) for routing calls from AT&T IP Toll Free service via Acme SBC, and ensure that the **Disabled** checkbox is unchecked to activate this Routing Policy.
3. In the **SIP Entity as Destination** section of the **Routing Policy Details** page, click **Select**. A pop-up window is displayed [not shown] where Experience Portal entity configured in **Section 5.6.4** is selected. The result is displayed below in the **SIP Entity as Destination** section.
4. On the **Routing Policy Details** page shown below, click **Add** in the **Time of Day** section. In the **Time Range List** page [not shown], check the checkbox(s) corresponding to one or more Time Ranges administered in **Section 5.8**, and click **Select**. On the **Routing Policy Details** page show below, enter a **Ranking** (the lower the number, the higher the ranking) in the **Time of Day** section for each Time Range.
5. Any **Dial Patterns** that were previously defined will be displayed and entries may be added or removed here. Dial patterns for this reference configuration are provisioned in **Section 5.10.1**.
6. No **Regular Expressions** were used in this reference configuration.
7. Click **Commit**.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

Commit Cancel

General

Name: ToAEP6.0

Disabled: ☐

Notes: Routing to AEP 6.0

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AEP6.0	10.80.130.220	Voice Portal	Avaya Aura Experience Portal 6.0

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
1	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	10	<input type="checkbox"/>	attavaya.com	Acme_SBC_130	For Routing calls to Experience Portal

Routing Policy Details Page to Experience Portal

5.9.2. Routing Policy to Acme Session Border Controller

To configure routing policy to Acme SBC, repeat steps in **Section 5.9.1**. The following screen shows the routing policy configured for the calls to be routed to Acme SBC. Dial pattern/s for calls to be routed to Acme SBC are configured/displayed in **Section 5.10.2**.

AVAYA

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing

Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

Commit

Cancel

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

General

* Name: To_ATTAcme5060

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AcmeSBCATT-5060	10.80.130.250	Other	Acme SBC to ATT

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item | Refresh

Filter: Enable

<input type="checkbox"/>	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

4 Items | Refresh

Filter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	666	7	7	<input type="checkbox"/>	attavaya.com	Location_130	
<input type="checkbox"/>	303	10	10	<input type="checkbox"/>	-ALL-	Location_130	
<input type="checkbox"/>	314346	10	10	<input type="checkbox"/>	-ALL-	Location_130	
<input type="checkbox"/>	800	10	10	<input type="checkbox"/>	-ALL-	Location_130	

Routing Policy Details Page to Acme SBC

5.9.3. Routing Policy to Avaya Aura® Communication Manager

To configure routing policy to Communication Manager, repeat steps in **Section 5.9.1**. The following screen shows the routing policy configured for the calls to be routed to Communication Manager. Dial pattern/s for calls to be routed to Communication Manager are configured/displayed in **Section 5.10.3**.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

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

Home / Elements / Routing / Routing Policies - Routing Policy Details [Help ?](#)

Routing Policy Details [Commit](#) [Cancel](#)

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
CM6.0.1-ATT-CLAN1A02	10.80.130.102	CM	CM6.0.1 for ATT on CLAN 1A02

Time of Day

[Add](#) [Remove](#) [View Gaps/Overlaps](#)

1 Item [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	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](#)

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	666	7	7	<input type="checkbox"/>	attavaya.com	Acme_SBC_130	
<input type="checkbox"/>	666	7	7	<input type="checkbox"/>	attavaya.com	Location_150_CM	

Routing Policy Details Page to Communication Manager

5.9.4. Routing Policy to Avaya Aura® Messaging

To configure routing policy to Messaging, repeat steps in **Section 5.9.1**. The following screen shows the routing policy configured for the calls to be routed to Messaging. Dial pattern/s for calls to be routed to Messaging, is/are configured/displayed in **Section 5.10.4**.

The screenshot displays the Avaya Aura® System Manager 6.1 interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The 'General' tab is active, showing the policy name 'ToCM_Messaging', a 'Disabled' checkbox, and a note 'Routing to Messaging'. Below this is the 'SIP Entity as Destination' section with a 'Select' button and a table listing the destination 'Messaging' with FQDN '10.80.150.222' and type 'Modular Messaging'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, followed by a table with one item for '24/7' with checkboxes for all days of the week. The 'Dial Patterns' section has 'Add' and 'Remove' buttons and a table with two items: '6665000' for 'attavaya.com' and '6665000' for 'attavaya.com', both with 'Min' and 'Max' values of 7.

Routing Policy Details

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Messaging	10.80.150.222	Modular Messaging	Aura Messaging

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

2 Items Refresh Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
6665000	7	7	<input type="checkbox"/>	attavaya.com	Location_130	Dial Pattern for calls to Messaging System
6665000	7	7	<input type="checkbox"/>	attavaya.com	Acme_SBC_130	Dial Pattern for calls to Messaging System

Routing Policy Details Page to Messaging

5.10. Dial Patterns

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

- Inbound PSTN calls from AT&T IP Toll Free service destined for Avaya Aura® Experience Portal
- Dial Pattern for Acme SBC
- Calls transferred to Avaya Aura® Communication Manager
- Calls to Avaya Aura® Messaging pilot number

5.10.1. Matching Inbound Calls from AT&T IPTF Service to Avaya Aura® Experience Portal

In this example inbound calls from any PSTN number with the pattern 00000xxxxx are defined.

1. In the left pane under **Routing**, click on **Dial Patterns**. In the **Dial Patterns** page click on **New** [not shown].
2. In the **General** section of the **Dial Pattern Details** page, configure as follows:
 - **Pattern** – Enter matching patterns for inbound dialed digits. Set to **00000** for this reference configuration.
 - **Min** and **Max** – Enter **10**.
 - **SIP Domain** – Select one of the SIP Domains defined in **Section 5.3** or **-ALL-**, to select all of those administered SIP Domains. Only those calls with the same domain in the Request-URI as the selected SIP Domain (or any of the administered SIP Domains if **-ALL-** is selected) can match this Dial Pattern. Set to **attavaya.com** in this reference configuration.
 - **Notes** - [Optional] Add any notes if desired.
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click **Add**.
4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page [not shown], select the locations from where calls can originate to be routed to Experience Portal. Note that only those calls that originate from the selected Location(s), or all administered Locations if **-ALL-** is selected, can match this Dial Pattern. Originating location **Acme_SBC_130** configured in **Section 5.4** was selected in this reference configuration.
5. In the **Routing Policies** section of the **Originating Location and Routing Policy List** page [not shown], select the Routing Policy administered for routing calls to Experience Portal in **Section 5.9.1**.
6. In the Originating Location and Routing Policy section, the values selected are displayed.
7. Click **Commit** on **Dial Pattern Details** page.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Commit Cancel

General

* Pattern: 00000

* Min: 10

* Max: 10

Emergency Call: ☐

SIP Domain: attavaya.com

Notes: For Routing calls to Experience Portal

Originating Locations and Routing Policies

Add Remove

1 Item 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"/>	Acme_SBC_130	SBC to ATT	ToAEP6.0	0	<input type="checkbox"/>	AEP6.0	Routing to AEP 6.0

Dial Pattern Details Page - Matching Inbound Calls from AT&T to Experience Portal

5.10.2. Dial Pattern for Acme SBC

Repeat steps in **Section 5.10.1** to add additional dial patterns. The following screen shows the dial pattern configured for the calls to be routed to Acme SBC from Experience Portal in this reference configuration. Additional dial patterns **303xxxxxxx**, **314xxxxxxx** and **800xxxxxxx** were also configured. Calls from Experience Portal are always routed to Acme SBC first and Acme SBC then decides based upon the local policy whether to forward the calls to PSTN or send them back to Session Manager for delivery to Communication Manager. In this reference configuration, calls to **800**, **314** and **303** were forwarded to PSTN whereas calls to **666xxxx** were sent back to Session Manager for delivery to endpoints on Communication Manager.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

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

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#) - Dial Pattern Details [Help ?](#)

Dial Pattern Details [Commit](#) [Cancel](#)

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

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

1 Item [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	To_ATTAcme5060	0	<input type="checkbox"/>	AcmeSBCATT-5060	

Dial Pattern Details Page – Acme SBC

5.10.3. Matching Calls to Avaya Aura® Communication Manager

Repeat steps in **Section 5.10.1** to add additional dial patterns. The following screen shows the dial pattern configured for the calls to be routed to Communication Manager via Acme SBC from Experience Portal in this reference configuration. In this example, the calls from Experience Portal to **666xxxx** are routed to Communication Manager via Session Manager.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Commit Cancel

General

* Pattern: 666

* Min: 7

* Max: 7

Emergency Call: ☐

SIP Domain: attavaya.com

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item 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"/>	Acme_SBC_130	SBC to ATT	ToCM6.0.1-CLAN1A02	0	<input type="checkbox"/>	CM6.0.1-ATT-CLAN1A02	Routing to CM6.0.1 on CLAN 1A02

Dial Pattern Details Page – Matching calls from Experience Portal for Acme SBC

5.10.4. Matching Inbound Calls to Avaya Aura® Messaging Pilot Number

Communication Manager stations cover to Messaging using a pilot extension **6665000** in this reference configuration. Also, stations on Communication Manager may dial this number to retrieve messages or modify mailbox settings. To match dial pattern for the calls covered to Messaging, repeat the Steps in **Section 5.10.1**. Routing Policy configured in **Section 5.9.4** was used to route the call to Messaging.

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". Below the navigation bar, there's a breadcrumb trail: "Home / Elements / Routing / Dial Patterns - Dial Pattern Details". The left sidebar contains a menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns (selected), Regular Expressions, and Defaults. The main content area is titled "Dial Pattern Details" and has a "General" tab selected. The form fields are as follows:

- * Pattern: 6665000
- * Min: 7
- * Max: 7
- Emergency Call: ☐
- SIP Domain: attavaya.com
- Notes: Dial Pattern for calls to Messaging Pilot

Below the form, there's a section titled "Originating Locations and Routing Policies" with "Add" and "Remove" buttons. It shows "1 Item" and a "Refresh" button. A table lists the configuration:

<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	ToCM_Messaging	0	<input type="checkbox"/>	Messaging	Routing to Messaging

Dial Pattern Details – Coverage to Messaging

5.10.5. Dial Pattern for MWI (Message Waiting Indicator) for stations on Avaya Aura® Communication Manager

This pattern is used to match the stations on Communication Manager for MWI. Repeat the Steps in **Section 5.10.1**. Routing Policy configured in **Section 5.9.3** was used to route the MWI Notify to Communication Manager.

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns (selected), Regular Expressions, and Defaults. The main content area is titled 'Dial Pattern Details' and includes a 'General' tab. The form fields are as follows:

- * Pattern: 666
- * Min: 7
- * Max: 7
- Emergency Call: ☐
- SIP Domain: attavaya.com
- Notes: (empty text box)

Below the form is a section titled 'Originating Locations and Routing Policies' with 'Add' and 'Remove' buttons. It shows a table with 1 item, filtered by '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_150_CM	Communication Manager	ToCM6.0.1-CLAN1A02	0	<input type="checkbox"/>	CM6.0.1-ATT-CLAN1A02	Routing to CM6.0.1 on CLAN 1A02

Dial Pattern Details – MWI for Communication Manager stations

5.11. Avaya Aura® Session Manager Administration

1. On the screen shown in **Section 5.2**, click Session Manager.
2. In the left pane of **Session Manager** page, click **Session Manager Administration**. On the **Session Manager Administration** page [not shown] in the Session Manager Instances, click **Add** [not shown] to add a Session Manager instance.
3. The screen below shows the Session Manager instance configured for this reference configuration.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Session Manager](#) × [Routing](#) × [Home](#)

[Home / Elements / Session Manager](#) [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 Page

6. Avaya Aura® Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [1], [2] and [[3] for further details if necessary.

6.1. Background

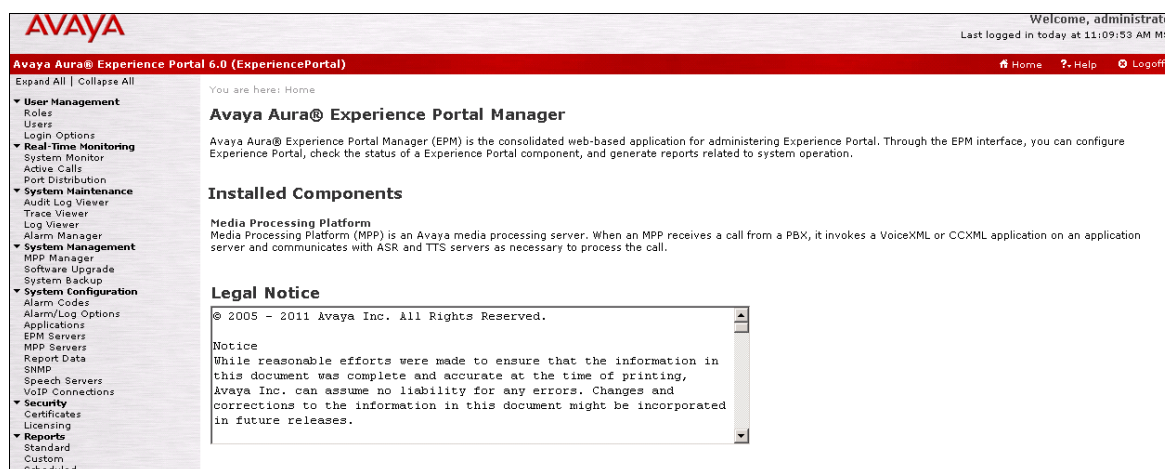
Experience Portal handles inbound calls according to the directives specified by Voice XML (VXML) and/or Call Control XML (CCXML) applications. These applications do not reside on Experience Portal, but on one or more separate application servers. References to these applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match⁴ is found, Experience Portal informs the caller that the call can not be handled, and disconnects the call.

For this reference configuration, VXML and CCXML applications were developed specifically to exercise SIP call flow scenarios expected to occur with the AT&T IP Toll Free service. In production, enterprises can develop their own VXML and/or CCXML applications to meet their specific customer self-service needs, or consult Avaya Professional Services and/or authorized Avaya Business Partners. The development and deployment of VXML and CCXML applications is beyond the scope of these Application Notes.

6.2. VoIP Connection

This section describes the steps on Experience Portal for administering a SIP connection to the Session Manager.

1. Launch a web browser, enter `http://<IP address of the Avaya EPM server>/` in the URL, log in with the appropriate credentials and the following screen is displayed.



Experience Portal Home Page

⁴ One application reference may be configured with “inbound default” as the called number to handle all inbound calls that do not match any other application references.

2. In the left pane, navigate to **Security**→**Licensing**. On the **Licensing** page, verify that Experience Portal is properly licensed. For required licenses that are not enabled, contact an authorized Avaya account representative to obtain the licenses

The screenshot shows the Avaya Aura Experience Portal 6.0 interface. The left navigation pane is expanded to the **Security** section, with **Licensing** selected. The main content area displays the **Licensing** page. At the top, it says "You are here: Home > Security > Licensing". Below this, there is a description: "This page displays the Experience Portal license information that is currently in effect. Experience Portal uses Avaya License Manager (WebLM) to control the number of telephony ports that are used." There are two main sections: **License Server Information** and **Licensed Products**. The **License Server Information** section shows the License Server URL as https://AEP60:8443/WebLM/LicenseServer, Last Updated as 10/20/11 2:19:41 PM MDT, and Last Successful Poll as 11/14/11 11:34:48 AM MST. The **Licensed Products** section shows a table of licenses for Experience Portal, including Announcement Ports (100), ASR Connections (100), Basic Ports for AACC (100), Enable Media Encryption (1), Enhanced Call Classification (100), SIP Signaling Connections (100), Telephony Ports (100), TTS Connections (100), and Video Server Connections (100). It also shows the Version as 6, Last Successful Poll as 11/14/11 11:34:48 AM MST, and Last Changed as 11/14/11 11:19:47 AM MST.

Experience Portal Licensing Page

3. In the left pane, navigate to **System Configuration**→**VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk. Note that only **ONE** SIP trunk can be active at any given time on Experience Portal.

The screenshot shows the Avaya Aura Experience Portal 6.0 interface. The left navigation pane is expanded to the **System Configuration** section, with **VoIP Connections** selected. The main content area displays the **VoIP Connections** page. At the top, it says "You are here: Home > System Configuration > VoIP Connections". Below this, there is a description: "This page displays a list of Voice over Internet Protocol (VoIP) servers that Experience Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time." There are two tabs: **H.323** and **SIP**. The **SIP** tab is selected. Below the tabs, it says "No SIP Connections are configured." There are three buttons: **Add**, **Delete**, and **Help**.

VoIP Connections Page

4. Configure the SIP connection as follows:
- **Name** – Set to a descriptive name.
 - **Enable** – Set to **Yes**.
 - **Proxy Transport** – Set to **TCP**.
 - **Proxy Server Address** – Set to the IP address of the Session Manager signaling interface.
 - **Proxy Server Port** – Set to **5060**.
 - **SIP Domain** – Set to SIP domain configured in **Section 5.3**.
 - **Consultative Transfer** – Select **REFER** radio button.
 - **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity. In this reference configuration a value of **10** was used for this field.
 - Set to the **All Calls can be either inbound or outbound** radio button.
 - Click **Save**.

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

- ▼ **User Management**
 - Roles
 - Users
 - Login Options
- ▼ **Real-Time Monitoring**
 - System Monitor
 - Active Calls
 - Port Distribution
- ▼ **System Maintenance**
 - Audit Log Viewer
 - Trace Viewer
 - Log Viewer
 - Alarm Manager
- ▼ **System Management**
 - MPP Manager
 - Software Upgrade
 - System Backup
- ▼ **System Configuration**
 - Alarm Codes
 - Alarm/Log Options
 - Applications
 - EPM Servers
 - MPP Servers
 - Report Data
 - SNMP
 - Speech Servers
 - VoIP Connections
- ▼ **Security**
 - Certificates
 - Licensing
- ▼ **Reports**
 - Standard
 - Custom
 - Scheduled

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > Add SIP Connection

Add SIP Connection

Use this page to add a new SIP connection.

Name:

Enable: ☒ Yes ☐ No

Proxy Transport:

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
10.80.150.206	5060	0	0	Remove

[Additional Proxy Server](#)

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☐ INVITE with REPLACES ☒ REFER

SIP Timers

T1: millisecond(s)

T2: millisecond(s)

B and F: millisecond(s)

Call Capacity

Maximum Simultaneous Calls:

☒ All Calls can be either inbound or outbound

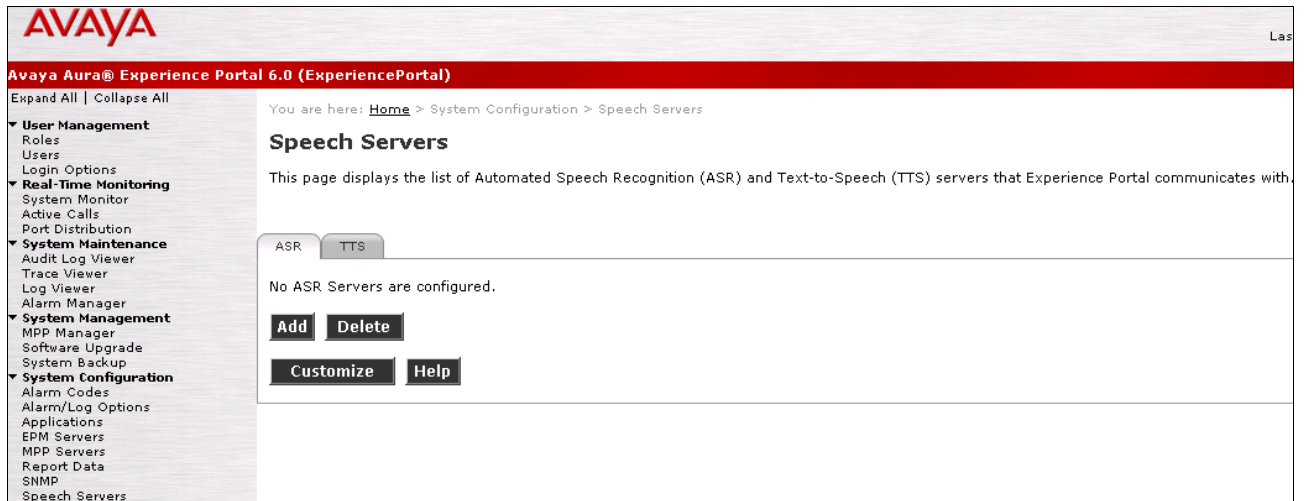
☐ Configure number of inbound and outbound calls allowed

Add SIP Connection Page

6.3. Speech Servers

The installation and administration of the Speech Servers is beyond the scope of this document.

1. To configure Experience Portal for communication with Speech Server, navigate to **System Configuration→Speech Servers** and the following screen is displayed. Click **ASR** and **Add** to add an ASR server.



Speech Server Page

2. On the **Add ASR Server** page, configure as follows:
- **Name** – Set to any descriptive name.
 - **Enable** – Select the **Yes** radio button.
 - **Engine Type** – Select **Nuance**.
 - **Network Address** – Set to the IP address of the ASR Server.
 - **Languages** – Select the appropriate value.
 - Click **Save**.

AVAYA

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Add ASR Server

Add ASR Server

Use this page to configure Experience Portal to communicate with a new ASR server.

Name:

Enable: ☒ Yes ☐ No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed ASR Resources:

New Connection per Session: ☐ Yes ☒ No

Languages:

MRCP

Ping Interval: second(s)

Response Timeout: second(s)

Protocol:

RTSP URL:

Save **Cancel** **Help**

Add ASR Server Page

3. Click **TTS** and **Add** on the screen shown in Step 1. On the **Add TTS Server** page, configure as follows:
- **Name** – Set to any descriptive name.
 - **Enable** – Select the **Yes** radio button.
 - **Engine Type** – Select **Nuance**.
 - **Network Address** – Set to the IP address of the ASR Server.
 - **Languages** – Select the appropriate value.
 - Click **Save**.

AVAYA

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [Speech Servers](#) > Add TTS Server

Add TTS Server

Use this page to configure Experience Portal to communicate with a new TTS server.

Name:

Enable: ☒ Yes ☐ No

Engine Type:

Network Address:

Base Port:

Total Number of Licensed TTS Resources:

New Connection per Session: ☐ Yes ☒ No

Voices:

- English(Irish) en-IE Moira F
- English(South_African) af-ZA Tessa F
- English(Scottish) en-SC Fiona F
- English(USA) en-US Donna F**
- English(USA) en-US Erica F
- English(USA) en-US Jennifer F

MRCP

Ping Interval: second(s)

Response Timeout: second(s)

Protocol:

RTSP URL:

Save **Cancel** **Help**

Add TTS Server Page

6.4. Application References

This section describes the steps on Experience Portal for administering a reference to a VXML and/or CCXML application residing on an application server.

1. In the left pane, navigate to **System Configuration** → **Applications**. On the **Applications** page [not shown], click on **Add** to add an application and configure as follows:
 - **Name** – Set to a descriptive name.
 - **Enable** – Set to **Yes**.
 - **MIME Type** – Set **CCXML/VoiceXML** for the application used in this reference configuration.
 - **VoiceXML** and/or **CCXML URL** – Set to the URL(s) to access the VXML and/or CCXML application(s) on the application server.
 - **Speech Servers ASR** and **TTS** – Set to **Nuance**.
 - **Languages** is set to **English (USA) en-US** and **Voices** is set to **English(USA) en-US Donna F**. This is as per Speech server settings in **Section 6.3**.
 - **Application Launch** – Set to **Inbound**.

Inbound AT&T IP Toll Free service calls with these called party numbers will be handled by this application defined in the following steps.

- Select the **Number** or **URI** radio button. URI is used where the called party number is a mix of numbers and characters.
- **Called Number** – Set to an inbound AT&T IP Toll Free service called party number specified in the **To** header of the inbound SIP INVITE message. Repeat to define additional AT&T IP Toll Free service called party numbers if necessary.

AVAYA

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

▼ User Management

Roles

Users

Login Options

▼ Real-Time Monitoring

System Monitor

Active Calls

Port Distribution

▼ System Maintenance

Audit Log Viewer

Trace Viewer

Log Viewer

Alarm Manager

▼ System Management

MPP Manager

Software Upgrade

System Backup

▼ System Configuration

Alarm Codes

Alarm/Log Options

Applications

EPM Servers

MPP Servers

Report Data

SNMP

Speech Servers

VoIP Connections

▼ Security

Certificates

Licensing

▼ Reports

Standard

Custom

Scheduled

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > Add Application

Add Application

Use this page to deploy and configure a new application on the Experience Portal system.

Name:

Enable: ☒ Yes ☐ No

Type:

URI

☒ Single ☐ Fail Over ☐ Load Balance

CCXML URL:

Verify

VoiceXML URL:

Verify

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

Speech Servers

ASR:

TTS:

English(USA) en-US

English(USA) en-US Donna F

Languages:

Voices:

Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

Add

8884575815

Remove

Add Application Page

- Repeat above step/s to administer additional applications.

AT; Reviewed
SPOC 3/21/2012

Solution & Interoperability Test Lab Application Notes
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E60C601S61APTF

6.5. Add MPP Server

1. In the left pane, navigate to **System Configuration**→**MPP Servers** and the following screen is displayed. Click **Add**.

The screenshot shows the Avaya Aura Experience Portal 6.0 interface. The left navigation pane is expanded to 'System Configuration' > 'MPP Servers'. The main content area is titled 'MPP Servers' and contains a table with columns: Name, Host Address, Network Address (VoIP), Network Address (MRCP), Network Address (AppSvr), Maximum Simultaneous Calls, and Trace Level. Below the table, it states 'No MPPs configured.' and has 'Add' and 'Delete' buttons. At the bottom, there are tabs for 'MPP Settings', 'Browser Settings', 'Event Handlers', 'Video Settings', 'VoIP Settings', and 'Help'.

MPP Servers Page

2. Enter any descriptive name in the **Name** field and IP address of the MPP server in the **Host Address** field and click **Continue**.

The screenshot shows the 'Add MPP Server' page in the Avaya Aura Experience Portal 6.0. The left navigation pane is expanded to 'System Configuration' > 'MPP Servers'. The main content area is titled 'Add MPP Server' and contains the text 'Use this page to add a new MPP server.' Below this, there are two input fields: 'Name' with the value 'MPP1' and 'Host Address' with the value '10.80.130.220'. At the bottom, there are three buttons: 'Continue', 'Cancel', and 'Help'.

Add MPP Servers Page

3. Check the **Trust this certificate** box and click **Save**.

AVAYA

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

▼ **User Management**

Roles

Users

Login Options

▼ **Real-Time Monitoring**

System Monitor

Active Calls

Port Distribution

▼ **System Maintenance**

Audit Log Viewer

Trace Viewer

Log Viewer

Alarm Manager

▼ **System Management**

MPP Manager

Software Upgrade

System Backup

▼ **System Configuration**

Alarm Codes

Alarm/Log Options

Applications

EPM Servers

MPP Servers

Report Data

SNMP

Speech Servers

VoIP Connections

▼ **Security**

Certificates

Licensing

▼ **Reports**

Standard

Custom

Scheduled

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > Add MPP Server

Add MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging T system has heavy call traffic. The system might experience performance issues if Trace Levels are set t system.

Name:	MPP1
Host Address:	<input type="text" value="10.80.130.220"/>
Network Address (VoIP):	<input type="text" value="<Default>"/>
Network Address (MRCP):	<input type="text" value="<Default>"/>
Network Address (AppSvr):	<input type="text" value="<Default>"/>
Maximum Simultaneous Calls:	<input type="text" value="10"/>
Restart Automatically:	<input checked="" type="radio"/> Yes <input type="radio"/> No

MPP Certificate

The following certificate was sent by the MPP for verification. The displayed certificate should be identical to the certificate established during the installation of the target MPP. Acceptance of the certificate will allow the MPP access to privileged services on the EPM. If the certificate does not match, ensure that the host address has been entered correctly.

```
Owner: CN=AREP60,0=Avaya,OU=EPM
Issuer: CN=AREP60,0=Avaya,OU=EPM
Serial Number: a5fd15294dcc7152
Valid from: October 20, 2011 2:00:30 PM MDT until October 17, 2021 2:00:30 PM MDT
Certificate fingerprints
    MD5: 04:36:22:ea:ea:59:69:7f:27:81:79:2c:47:e1:6a:7d
    SHA: 5d:b3:5f:e1:09:d5:09:01:b8:41:ef:ae:96:ff:84:fd:90:da:0a:4d
```

☒ Trust this certificate

Categories and Trace Levels ▶

Save

Cancel

Help

Add MPP Server Page - Continued

4. Click **VoIP Settings** tab on the screen displayed in **Step 1** and the following screen is displayed. Verify that TCP ports are in the range of **16384** and **32767** as required AT&T IP Toll Free service. Additionally set **Discontinuous Transmission** field under **Audio Codecs** to **No**.

AVAYA Welcome, administrator
Last logged in today at 11:09:53 AM

Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > [VoIP Settings](#)

VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

Port Ranges	Low	High
UDP:	23000	30999
TCP:	16384	32767
MRCP:	33000	33999
H.323 Station:	35000	50000

RTP Monitor Settings

Host Address:

Port:

VoIP Audio Formats

MPP Native Format:

Audio Codecs

Packet Time:

G729: ☒ Yes ☐ No

Reduced Complexity Encoder: ☒ Yes ☐ No

Discontinuous Transmission: ☐ Yes ☒ No

First Offered:

VoIP Settings Page

6.6. Configuring RFC2833 Event Value Offered by Avaya Aura® Experience Portal

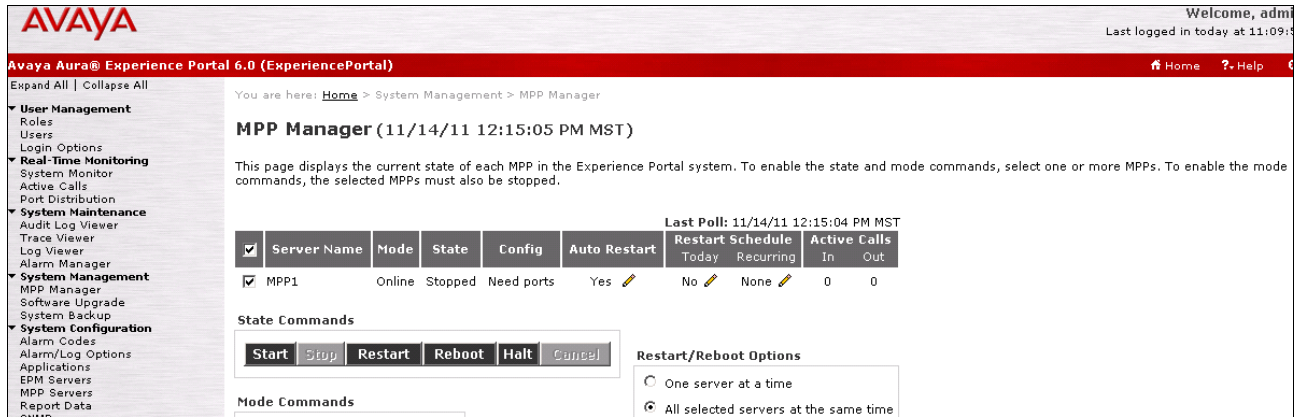
The configuration change example noted in this section is not required for any of the call flows illustrated in these Application Notes. For incoming calls from AT&T IP Toll Free service to Experience Portal, AT&T specifies the value 100 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Experience Portal answers, the SDP from Experience Portal matches this AT&T offered value. When Experience Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Experience Portal offers the SDP. By default, Experience Portal specifies the value 127 for the RFC2833 telephone-events. Optionally, the value that is offered by Experience Portal can be changed, and this section outlines the procedure that can be performed by an Avaya authorized representative.

- Access the Experience Portal via the command line interface and navigate to the `/opt/Avaya/Experience Portal/MPP/config` directory.
- Edit the file `mppconfig.xml`.
- Search for the parameter **mpp.sip.rfc2833.payload**.
- If the parameter is already specified in the file, simply edit the value assigned to the parameter. If there is no such parameter specified, add a line such as the following to the file, where 100 is the value to be used for the RFC2833 events.
 - `<parameter name="mpp.sip.rfc2833.payload">100</parameter>`

After saving the file with the change, restart the MPP server for the change to take effect as shown in **Section 6.7**.

6.7. MPP Manager

In the left pane, navigate to **System Maintenance** → **MPP Manager** and select the **MPP1**. Click **Restart** to make sure that the changes made in the above steps are effected. Note that all the configuration changes do not require restart of the MPP Manager.



Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Welcome, admin
Last logged in today at 11:09:35

Expand All | Collapse All

You are here: [Home](#) > System Management > MPP Manager

MPP Manager (11/14/11 12:15:05 PM MST)

This page displays the current state of each MPP in the Experience Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 11/14/11 12:15:04 PM MST

<input checked="" type="checkbox"/>	Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls		
						Today	Recurring	In	Out
<input checked="" type="checkbox"/>	MPP1	Online	Stopped	Need ports	Yes	No	None	0	0

State Commands

Restart/Reboot Options

☐ One server at a time

☒ All selected servers at the same time

Mode Commands

MPP Manager Page

7. Avaya Aura® Communication Manager

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. For any values not configured, defaults are used in this reference configuration. These Application Notes assume that basic Communication Manager administration has already been performed. Consult [6] and [7] for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to this reference configuration. Other parameter values may or may not match specific local configurations.

7.1. System Parameters

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

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

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks: 100		30	
Maximum Concurrently Registered IP Stations: 12000		6	
Maximum Administered Remote Office Trunks: 8000		0	
Maximum Concurrently Registered Remote Office Stations: 12000		0	
Maximum Concurrently Registered IP eCons: 0		0	
Max Concur Registered Unauthenticated H.323 Stations: 20		0	
Maximum Video Capable H.323 Stations: 20		0	
Maximum Video Capable IP Softphones: 20		0	
Maximum Administered SIP Trunks: 5000		30	
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: 5		0	
Maximum TN2602 Boards with 80 VoIP Channels: 128		0	
Maximum TN2602 Boards with 320 VoIP Channels: 128		1	
Maximum Number of Expanded Meet-me Conference Ports: 200		0	
NOTE: You must logoff & login to effect the permission changes.)			

System-Parameters Customer-Options Form – Page 2

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

```

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

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

```

System-Parameters Customer-Options Form – Page 4

7.2. Dial Plan

Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings were administered for this sample configuration:

- 3-digit dial access codes (indicated with a **Call Type** of **dac**) beginning with the digit **1** – Trunk Access Codes (TACs) defined for trunk groups in this sample configuration.
- 7-digit extensions with a **Call Type** of **ext** beginning with the digit **6661** – used for announcements, beginning with the digit **6662** – used for Vector Directory Numbers (VDN), beginning with the digit **6663** – used for agent login ids, beginning with the digit **6664** – used for hunt group extensions, and beginning with the digit **6665** – used for telephone extensions.

```

change dialplan analysis                                               Page 1 of 12
                                DIAL PLAN ANALYSIS TABLE
                                Location: all                          Percent Full: 1

Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
String   Length  Type  String   Length Type  String   Length Type
1         3       dac
6662      7       ext
6663      7       ext
6664      7       ext
6665      7       ext
8         1       fac
9         1       fac
*         3       fac
#         3       fac

```

Dialplan Analysis Form

7.3. IP Network Parameters

These Application Notes assume that the appropriate IP network regions and IP codec sets have already been administered to support internal calls, i.e., calls within the enterprise. For simplicity in this sample configuration, all Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within are assigned to a single IP network region. This section describes the steps for administering an additional IP network region and IP codec set to represent inbound calls from the AT&T IP Toll Free service to Experience Portal that are subsequently transferred to Communication Manager via Session Manager and Acme SBC. Note that the configuration steps in these application notes are used for this reference configuration and not meant to be prescriptive in nature.

1. Enter the **change ip-codec-set n** command, where **n** is the number of an unused IP codec set to be used for inbound calls. On **Page 1** of the **ip-codec-set** form, provision following codecs. AT&T IP Toll Free service uses **G.729A** as its preferred codec but also supports **G.711MU** and **G.726A-32K**. G.726 is supported by Communication Manager but not by Experience Portal.

change ip-codec-set 2		Page 1 of 2	
IP Codec Set			
Codec Set: 2			
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.729A	n	3	30
2: G.711MU	n	2	20
3: G.726A-32K	n	2	20

IP-Codec-Set Form for Inbound Calls – Page 1

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

change ip-codec-set 2		Page 2 of 2	
IP Codec Set			
Allow Direct-IP Multimedia? n			
FAX	Mode	Redundancy	
Modem	t.38 standard	0	
TDD/TTY	off	0	
Clear-channel	US	3	
	n	0	

IP-Codec-Set Form for Inbound Calls – Page 2

2. Enter the **change node-names ip** command, and add a node name and the IP address for the Session Manager. Also note the node name and IP address of a C-LAN board that is assigned to one of the IP network regions administered for local Communication Manager elements within the Avaya site. This C-LAN board will be used in **Section 7.4, Step 1** for administering a SIP trunk to the Session Manager.

change node-names ip		Page 1 of 2	
IP NODE NAMES			
Name	IP Address		
ASM	10.80.150.206		
CLAN-1A02	10.80.130.102		

Change Node-Names IP Form

- Enter the **change ip-network-region n**, where **n** is the number of an unused IP network region. This IP network region will be used to represent the AT&T IP Toll Free service. Note that the **Code Set** field is set to **2** which is configured in **Step 1**. Also, the port range is set between **16384** and **32767** as required by AT&T.

```

change ip-network-region 2                                     Page 1 of 20
                                IP NETWORK REGION
Region: 2
Location: Authoritative Domain: attavaya.com
Name:
MEDIA PARAMETERS                               Intra-region IP-IP Direct Audio: yes
Codec Set: 2                                   Inter-region IP-IP Direct Audio: yes
UDP Port Min: 16384                             IP Audio Hairpinning? n
UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS                       RTCP Reporting Enabled? y
Call Control PHB Value: 46                     RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46                           Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5                       AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                               RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5

```

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

On **Page 4** of the **ip-network-region** form, for each IP network region pair consisting of this IP network region as the **src rgn** and another IP network region as the **dst rgn**, provision the following:

- codec set** – Set to the codec set administered in **Step 1**.
- direct WAN** – Set to **y**.
- WAN-BW-limits** – Set to the maximum number of calls or bandwidth allowed between the two IP network regions. The setting shown below was used in this reference configuration.

In the example below, for all calls to elements in IP network region 1 will use codec set 2.

```

change ip-network-region 2                                     Page 4 of 19
                                Inter Network Region Connection Management
src dst codec direct  WAN-BW-limits  Video    Intervening  Dyn
rgn rgn set   WAN  Units   Total Norm  Prio Shr Regions  CAC IGAR AGL
2   1   2     y   NoLimit  Total Norm  Prio Shr Regions  n  all
2   2
2   3
2   4

```

IP-Network-Region Form for an IP Network Region Representing the AT&T IP Toll Free Service– Page 4

7.4. Inbound Calls

This section describes the steps for administering the SIP trunk from Communication Manager to Session Manager.

1. Enter the **add signaling-group s** command, where **s** is the number of an unused signaling group, and provision the following:
 - **Group Type** – Set to **sip**.
 - **Transport Method** – Set to **tcp**. Note that this is only the transport protocol used between Communication Manager and the Session Manager.
 - **Near-end Node Name** – Set to the node name of the C-LAN board noted in **Section 7.3, Step 2**.
 - **Far-end Node Name** – Set to the node name of the Session Manager as administered in **Section 7.3, Step 2**.
 - **Near-end Listen Port** and **Far-end Listen Port** – Set to **5060**.
 - **Far-end Network Region** – Set to the IP network region administered in **Section 7.3, Step 3** to represent the PSTN.
 - **Far-end Domain** – Set to **attavaya.com**. This domain matches the domain configured in **Section 5.3**.
 - **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF as per RFC 2833.
 - **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible.

add signaling-group 2		Page 1 of 1	
Group Number: 2		Group Type: sip	
		Transport Method: tcp	
Near-end Node Name: CLAN_1A02		Far-end Node Name: ASM	
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? n		IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n		Direct IP-IP Early Media? n	
		Alternate Route Timer(sec): 6	

Signaling-Group Form for Transferred Inbound Calls

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

add trunk-group 2		Page 1 of 21	
TRUNK GROUP			
Group Number: 2	Group Type: sip	CDR Reports: y	
Group Name: ATT IPTF	COR: 1	TN: 1	TAC: 102
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: public-ntwrk	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 2		
	Number of Members: 10		

Trunk-Group Form for Transferred Inbound Calls – Page 1

3. Enter the **change public-unknown-numbering 0** command to specify the connected party numbers sent on transferred inbound calls. In the **public-unknown-numbering** form, for each local extension range assigned to Communication Manager phones, agents, skills (hunt groups), and VDNs, provision an entry as follows:
 - **Ext Len** – Enter the total number of digits in the local extension range.
 - **Ext Code** – Enter enough leading digits to identify the local extension range.
 - **Trk Grp(s)** – Enter the number of the trunk group administered in **Step 2**.
 - **CPN Prefix** – If necessary, enter enough prefix digits to form the desired connected party number.
 - **CPN Len** – Enter the total length of the connected party number to be sent.

change public-unknown-numbering 0				Page 1 of 2	
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
7	666	2		7	Total Administered: 2 Maximum Entries: 9999
10	303	2		10	

Public-Unknown-Numbering Form

7.5. Optional Features

The reference configuration uses hunt groups, vectors, and Vector Directory Numbers (VDNs), to provide additional functionality during testing:

- Hunt Group 1 –Messaging coverage for Communication Manager extensions
- VDN 6662010/Vector 10 – VDN and vectors used to select the agent skill

Following VDN/Vectors were used for calls transferred to an agent/skill on Communication Manager without verifying the availability of an agent as described in third call scenario in **Section 3.2**.

- VDN 6662011/Vector 11/Hunt Group 11 – Route call to Agent with Skill 11
- VDN 6662012/Vector 12/Hunt Group 12 – Route call to Agent with Skill 12
- VDN 6662013/Vector 13/Hunt Group 13 – Route call to Agent with Skill 13

Following VDN/Vectors were used for calls anchored on Experience Portal and only transferred to an agent on Communication Manager once agent becomes available as described in second call scenario in **Section 3.2**.

- VDN 6662031/Vector 31/Hunt Group 31 – Route call to Agent with Skill 11
- VDN 6662032/Vector 32/Hunt Group 32 – Route call to Agent with Skill 12
- VDN 6662033/Vector 33/Hunt Group 33 – Route call to Agent with Skill 13

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. Refer to **[8]** and **[9]** for further details if necessary. The samples that follow are provided for reference purposes only.

7.5.1. Hunt Group for Station Coverage to Messaging

Hunt group **2** is used in the reference configuration to verify the coverage to voicemail. The hunt group **2** is defined with the 7 digit Messaging pilot number **666-5000**.

display hunt-group 2		Page 1 of 60
HUNT GROUP		
Group Number: 2	ACD? n	
Group Name: Messaging	Queue? n	
Group Extension: 666-5000	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:		

Hunt Group Form – Page 1

display hunt-group 2		Page 2 of 60
HUNT GROUP		
Message Center: sip-adjunct		
Voice Mail Number	Voice Mail Handle	Routing Digits
		(e.g., AAR/ARS Access Code)
6665000	6665000	8

Hunt Group Form – Page 2

The hunt group is associated with a coverage path **h2** and this coverage path is assigned to a station/agent.

display coverage path 2	
COVERAGE PATH	
Coverage Path Number: 2	
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
All?	n n
DND/SAC/Goto Cover?	y y
Holiday Coverage?	n n
COVERAGE POINTS	
Terminate to Coverage Pts. with Bridged Appearances? n	
Point1: h2	Rng: 4 Point2:
Point3:	Point4:
Point5:	Point6:

Coverage Path Form

7.5.2. 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 via Experience Portal.

In this reference configuration, an inbound call from AT&T IP Toll Free service is handled using the routing policy configured in **Section 5.9.3** and dial pattern configured in **Section 5.10.1**.

display system-parameters customer-options		Page 6 of 11
CALL CENTER OPTIONAL FEATURES		
Call Center Release: 5.0		
ACD? y	Reason Codes? n	
BCMS (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

display agent-loginID 6663011		Page 1 of 3
AGENT LOGINID		
Login ID: 6663011	AAS? n	
Name: Agent1	AUDIX? n	
TN: 1	LWC Reception: spe	
COR: 1	LWC Log External Calls? n	
Coverage Path: 2	AUDIX Name for Messaging:	
Security Code:	LoginID for ISDN/SIP Display? n	
	Password:	
	Password (enter again):	
	Auto Answer: station	
	MIA Across Skills: system	
	ACW Agent Considered Idle: system	
	Aux Work Reason Code Type: system	
	Logout Reason Code Type: system	
	Maximum time agent in ACW before logout (sec): system	
	Forced Agent Logout Time: :	
WARNING: Agent must log in again before changes take effect		

Agent Form – Page 1

display agent-loginID 6663011										Page 2 of 2	
AGENT LOGINID											
Direct Agent Skill:						Service Objective? n					
Call Handling Preference: skill-level						Local Call Preference? n					
SN	RL	SL	SN	RL	SL	SN	RL	SL	SN	RL	SL
1:	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: 6664011						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 4	
HUNT GROUP											
Skill? y						Expected Call Handling Time (sec): 180					
AAS? n											
Measured: none											
Supervisor Extension:											
Controlling Adjunct: none											
Multiple Call Handling: none											
Timed ACW Interval (sec):						After Xfer or Held Call Drops: n					

Skill (Hunt Group) Form – Page 2

display vdn 6662010 Page 1 of 3

VECTOR DIRECTORY NUMBER

Extension: 666-2010

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

SelectSkill VDN

display vector 10 Page 1 of 6

CALL VECTOR

Number: 10

Name: RouteToSkill

Meet-me Conf? n Lock? n

Basic? y EAS? n G3V4 Enhanced? y ANI/II-Digits? y ASAI Routing? y

Prompting? y LAI? n G3V4 Adv Route? n CINFO? n BSR? n Holidays? n

Variables? n 3.0 Enhanced? n

01 wait-time 2 secs hearing ringback

02 collect 1 digits after announcement 661002 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

RouteToSkill Vector⁵

display vector 11 Page 1 of 6

CALL VECTOR

Number: 11

Name: Skill 11

Meet-me Conf? n Lock? n

Basic? y EAS? n G3V4 Enhanced? y ANI/II-Digits? y ASAI Routing? y

Prompting? y LAI? n G3V4 Adv Route? n CINFO? n BSR? n Holidays? n

Variables? n 3.0 Enhanced? n

01 wait-time 2 secs hearing ringback

02 announcement 6661003

03 queue-to skill 11 pri m

04 announcement 6661006

05 goto step 3 if unconditionally

06

Skill-11 Vector

⁵ This vector was used for the call flow scenario where Experience Portal transfers the inbound call to an Communication Manager skill without checking whether an agent in that skill is available.

display vector 31

Page 1 of 6

CALL VECTOR

Number: 31 Name: VP Test Vector
Multimedia? n Attendant Vectoring? n Meet-me Conf? n Lock? n
Basic? y EAS? y 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 queue-to skill 11 pri m
02 stop
03

Sample Vector⁶

8. Avaya Aura® Messaging

The administration for Messaging is beyond the scope of these Application Notes. Refer to [10] and [11] for further details.

⁶ This vector was used for the call flow scenario where Experience Portal checks a Communication Manager skill for agent availability before transferring the inbound call to the skill.

9. Configure Acme Session Border Controller

The Acme SBC configuration used in the sample configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Consult with Acme Packet Support [12] for further details and explanations on the configuration below.

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

```
local-policy
  from-address          *
  to-address            *
  source-realm          IPTF-Enterprise
  description
  activate-time         N/A
  deactivate-time       N/A
  state                 enabled
  policy-priority       none
  last-modified-by      admin@console
  last-modified-date    2011-08-12 10:25:23
  policy-attribute
    next-hop            192.168.62.50
    realm                ATT
    action               none
    terminate-recursion disabled
    carrier
    start-time           0000
    end-time             2400
    days-of-week         U-S
    cost                 0
    app-protocol         SIP
    state                enabled
    methods
    media-profiles
```

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

```
local-policy
  from-address          *
  to-address            00000
                        666
                        +666
  source-realm          ATT
  description
  activate-time         N/A
  deactivate-time       N/A
```


state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-08-12 10:25:23
policy-attribute	
next-hop	10.80.150.206
realm	IPTF-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	

ANNOTATION: The local policy below governs the routing of SIP messages from the Experience Portal to Communication Manager via Session Manager

local-policy	
from-address	*
to-address	666
source-realm	IPTF-Enterprise
description	
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-08-12 10:25:23
policy-attribute	
next-hop	10.80.150.206
realm	IPTF-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	

media-manager	
state	enabled
latching	enabled
flow-time-limit	86400
initial-guard-timer	300
subsq-guard-timer	300

tcp-flow-time-limit	86400
tcp-initial-guard-timer	300
tcp-subsq-guard-timer	300
tcp-number-of-ports-per-flow	2
hnt-rtcp	disabled
algd-log-level	NOTICE
mbcd-log-level	NOTICE
red-flow-port	1985
red-mgcp-port	1986
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
media-policing	enabled
max-signaling-bandwidth	7752190
max-untrusted-signaling	80
min-untrusted-signaling	20
app-signaling-bandwidth	0
tolerance-window	30
rtcp-rate-limit	0
min-media-allocation	32000
min-trusted-allocation	60000
deny-allocation	32000
anonymous-sdp	disabled
arp-msg-bandwidth	32000
fragment-msg-bandwidth	0
rfc2833-timestamp	disabled
default-2833-duration	100
rfc2833-end-pkts-only-for-non-sig	enabled
translate-non-rfc2833-event	disabled
dnalg-server-failover	disabled
last-modified-by	admin@console
last-modified-date	2010-09-08 10:22:03

network-interface	
name	wancom0
sub-port-id	0
description	
hostname	
ip-address	135.9.230.221
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.0
gateway	135.9.230.254
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
last-modified-by      admin@console
last-modified-date    2011-08-12 10:21:39

```

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
  last-modified-by      admin@console
  last-modified-date    2011-08-12 14:58:25

```

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

```

network-interface
  name          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
last-modified-by                admin@console
last-modified-date              2011-08-12 10:24:07
ntp-config
  server                        192.9.1.2
  last-modified-by              admin@console
  last-modified-date            2009-03-12 10:20:46

phy-interface
  name                          wancom0
  operation-type                Control
  port                          2
  slot                          0
  virtual-mac
  wancom-health-score           9
  last-modified-by              admin@console
  last-modified-date            2011-08-12 10:21:30

phy-interface
  name                          s0p0
  operation-type                Media
  port                          0
  slot                          0
  virtual-mac                    00:08:25:a0:f3:68
  admin-state                    enabled
  auto-negotiation              enabled
  duplex-mode                    FULL
  speed                          100
  last-modified-by              admin@console
  last-modified-date            2011-08-13 15:29:00

phy-interface
  name                          s1p0
  operation-type                Media
  port                          0
  slot                          1
  virtual-mac                    00:08:25:a0:f3:6e
  admin-state                    enabled
  auto-negotiation              enabled
  duplex-mode                    FULL
  speed                          100
  last-modified-by              admin@console
  last-modified-date            2011-08-13 15:29:23

```

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP Toll Free service resides, and applies SIP manipulations **RemoveUPDATE**.

realm-config	
identifier	ATT
description	
addr-prefix	0.0.0.0
network-interfaces	
	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	
in-translationid	
out-translationid	
in-manipulationid	removeUpdAndmodifyPtime
out-manipulationid	NAT_IP
manipulation-string	
class-profile	
average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	4
maximum-signal-threshold	3000
untrusted-signal-threshold	10
nat-trust-threshold	0
deny-period	60
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0

net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2009-04-22 19:26:23

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

realm-config

identifier	IPTF-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	
in-translationid	
out-translationid	
in-manipulationid	AddDiversion
out-manipulationid	
manipulation-string	
class-profile	
average-rate-limit	0
access-control-trust-level	high
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
symmetric-latching	disabled

pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	enabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2011-08-12 19:50:37

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

session-agent	
hostname	Enterprise-IPTF
ip-address	10.80.150.206
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	UDP+TCP
realm-id	IPTF-Enterprise
egress-realm-id	
description	Session Manager
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0

max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS ;hops=0
ping-interval	180
ping-send-mode	keep-alive
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	enabled
reuse-connections	TCP
tcp-keepalive	enabled
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
last-modified-by	admin@console
last-modified-date	2011-08-17 17:36:26

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

session-agent	
hostname	135.242.225.200
ip-address	135.242.225.200
port	5060

state	enabled
app-protocol	SIP
app-type	
transport-method	UDP
realm-id	ATT
egress-realm-id	
description	AT&T Border Element
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS ;hops=70
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	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none

rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
last-modified-by	admin@console
last-modified-date	2011-08-17 17:36:20

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

sip-config

state	enabled
operation-mode	dialog
dialog-transparency	enabled
home-realm-id	IPTF-Enterprise
egress-realm-id	IPTF-Enterprise
nat-mode	None
registrar-domain	
registrar-host	
registrar-port	0
register-service-route	always
init-timer	500
max-timer	4000
trans-expire	32
invite-expire	180
inactive-dynamic-conn	32
enforcement-profile	
pac-method	
pac-interval	10
pac-strategy	PropDist
pac-load-weight	1
pac-session-weight	1
pac-route-weight	1
pac-callid-lifetime	600
pac-user-lifetime	3600
red-sip-port	1988
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
add-reason-header	disabled
sip-message-len	4096
enum-sag-match	disabled
extra-method-stats	enabled
registration-cache-limit	0
register-use-to-for-lp	disabled
options	max-udp-length=0 set-inv-exp-at-100-resp
add-ucid-header	disabled
last-modified-by	admin@console
last-modified-date	2011-08-12 10:22:04

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

```

sip-interface
state
realm-id
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
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	
last-modified-by	admin@console
last-modified-date	2009-04-22 18:14:23

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

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

sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
last-modified-by	admin@console
last-modified-date	2009-04-16 18:07:58

ANNOTATION: The SIP manipulation below removes **UPDATE** from the Allow header in SIP messages from the AT&T IP Toll Free service as **UPDATE** is not supported by Experience Portal. It also modifies the **maxptime** attribute to **ptime** as Experience Portal does not recognize **maxptime** attribute.

sip-manipulation

name	removeUpdateAndModifyMaxptime
description	Strip Update from Allow list, modify Ptime
header-rule	
name	ReplaceMaxptime
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	maxptime
new-value	ptime

header-rule	
name	EditAllow
header-name	Allow
action	manipulate
comparison-type	pattern-rule
match-value	
msg-type	any
new-value	
methods	
element-rule	
name	StripUPDATE
parameter-name	
type	header-value
action	find-replace-all
match-val-type	any
comparison-type	pattern-rule
match-value	(,\s*UPDATE UPDATE\s*,)
new-value	
last-modified-by	admin@console
last-modified-date	2011-10-22 19:25:08

ANNOTATION: The SIP manipulation below adds a **Diversion** header in SIP messages from the Experience Portal to AT&T Flex Reach service as **Diversion** header is not generated by Experience Portal. A valid DID is required for calls 8YY numbers otherwise the calls will fail. See **Section 2.2, Item 6** for further information. This manipulation rule was used in this reference configuration and is not intended to be prescriptive.

sip-manipulation	
name	AddDiverions
description	Add Diversion Header for 8YY calls
header-rule	
name	AddDiversionHdr
header-name	Diversion
action	add
comparison-type	boolean
match-value	
msg-type	request
methods	
new-value	"sip:7323204084@10.80.130.220"
last-modified-by	admin@console
last-modified-date	2011-08-22 19:25:08

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

steering-pool	
ip-address	192.168.62.50
start-port	16384
end-port	32767
realm-id	ATT
network-interface	
last-modified-by	admin@console
last-modified-date	2011-08-25 19:11:47
steering-pool	

ip-address	10.80.130.250
start-port	16384
end-port	32767
realm-id	IPTF-Enterprise
network-interface	
last-modified-by	admin@console
last-modified-date	2011-08-12 10:25:12
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	172.16.253.4
restart	enabled
exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	enabled
cli-more	disabled
terminal-height	24
debug-timeout	0
trap-event-lifetime	0
last-modified-by	admin@console
last-modified-date	2011-08-12 10:20:46

10. Verification Steps

10.1. General

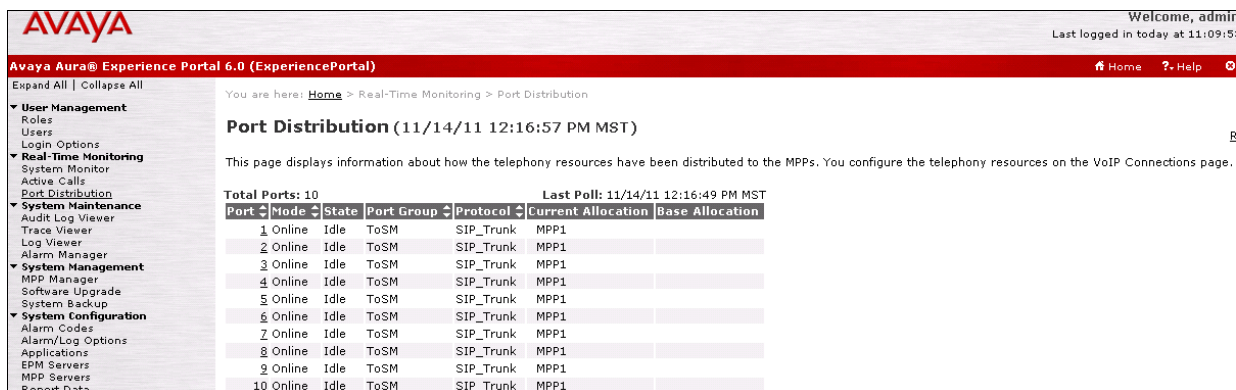
The following steps may be used to verify the configuration:

- Place an inbound call to Experience Portal application, and verify that two-way talkpath exists. Interact with the Experience Portal prompts and verify that the call remains stable for several minutes and disconnect properly.
- Place an inbound call to Experience Portal application that can canvass Communication Manager for skilled agent availability, and select the appropriate prompt(s) to request a transfer to an agent. Verify that when no agent in the skill is available, the caller hears wait treatment from the Experience Portal application while waiting to be transferred. Verify that when an agent in the skill becomes available, the call is successfully transferred to the agent and two-way talkpath exists between the caller and the agent.
- Place an inbound call to Experience Portal application that can transfer an inbound call to Communication Manager regardless of skilled agent availability, and select the appropriate prompt(s) to request a transfer to an agent. Verify that the transfer completes successfully. Verify that when no agent in the skill is available, the caller hears wait treatment from Communication Manager. Verify that when an agent in the skill becomes available, the call is successfully routed to the agent and two-way talkpath exists between the caller and the agent.

10.2. Avaya Aura® Experience Portal

The following commands are issued from the System Manager console.

1. Navigate to **Real-Time Monitoring**→**Port Distribution** to verify the SIP trunk on Experience Portal SIP Trunk has been properly configured as shown below:



Avaya Aura® Experience Portal 6.0 (ExperiencePortal)

Welcome, admin
Last logged in today at 11:09:5

Expand All | Collapse All

You are here: [Home](#) > Real-Time Monitoring > Port Distribution

Port Distribution (11/14/11 12:16:57 PM MST)

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

Total Ports: 10 Last Poll: 11/14/11 12:16:49 PM MST

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
1	Online	Idle	ToSM	SIP_Trunk	MPP1	
2	Online	Idle	ToSM	SIP_Trunk	MPP1	
3	Online	Idle	ToSM	SIP_Trunk	MPP1	
4	Online	Idle	ToSM	SIP_Trunk	MPP1	
5	Online	Idle	ToSM	SIP_Trunk	MPP1	
6	Online	Idle	ToSM	SIP_Trunk	MPP1	
7	Online	Idle	ToSM	SIP_Trunk	MPP1	
8	Online	Idle	ToSM	SIP_Trunk	MPP1	
9	Online	Idle	ToSM	SIP_Trunk	MPP1	
10	Online	Idle	ToSM	SIP_Trunk	MPP1	

2. Navigate to **Real-Time Monitoring**→**Active Calls** to verify the number of active calls, the trunk being used and the application running on Experience Portal:

AVAYA

Welcome, administrator
Last logged in today at 10:59:37 AM MDT

Voice Portal 5.1 (VoicePortal)

You are here: [Home](#) > Real-Time Monitoring > Active Calls

Active Calls (9/22/11 12:32:43 PM MDT)

This page displays the status of all the active calls being handled by the Voice Portal system.

Total Active Calls: 1

Last Poll: 9/22/11 12:32:44 PM MDT

Port	Port Group	Protocol	Call Type	MPP Server	Start Time	Calling Number/URI	Called Number/URI	Application	ASR Server	TTS Server
1 ToSM		SIP_Trunk	Inbound	MPP1	9/22/11 12:32:41 PM MDT	tel:3035381760;phone-context=private	tel:0000011001;phone-context=private	SelfService	SpeechSvr	

Help

10.3. Troubleshooting Tools

The logging and reporting functions within the Experience Portal web interface may be used to examine the details of Experience Portal calls.

The Communication Manager **list trace vector**, **list trace vdn**, **list trace tac**, and/or **status trunk trunk-group-no** commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The **traceSM** function within the Session Manager may be used to capture SIP traces between Session Manager and the AT&T IP Toll Free service. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

11. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya Aura® Experience Portal and the Acme Packet Net-Net can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of Avaya Aura® Experience Portal the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection. The test objectives stated in **Section 2** with limitations noted in **Section 2.2** were verified.

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

12. References

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

- [1] *Planning for Avaya Aura® Experience Portal*, August 2011
- [2] *Implementing Avaya Aura® Experience Portal on a single server*, August 2011
- [3] *Administering Aura® Experience Portal*, January 2011
- [4] *Installing and Configuring Avaya Aura® Session Manager*, April 2011.
- [5] *Administering Avaya Aura® Session Manager*, October 2011.
- [6] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509, August 2010
- [7] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document Id 555-245-205, August 2010
- [8] *Administering Avaya Aura® Call Center Features*, November 2010
- [9] *Programming Call Vectors in Avaya Aura® Call Center*, June 2010
- [10] *Administering Avaya Aura® Messaging*, December 2011
- [11] *Implementing Aura® Messaging*, October 2011

Acme Packet Support (login required):

- [12] <http://support.acmepacket.com>

AT&T IP Toll Free Service Descriptions:

- [13] *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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