Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Voice Portal 5.1, Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service using MIS/PNT Transport – Issue 1.1

Abstract

These Application Notes describe the steps for configuring Avaya Voice 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** transport connections.

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 Voice 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 Voice 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** transport connections. **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 Voice 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.

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2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site with Avaya Voice Portal, Avaya Aura® Communication Manager, Avaya Aura® Session Manger, Avaya Aura® 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 transport connections.

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

- Inbound calls to various Voice Portal applications.
- Inbound caller interaction with Voice Portal applications, including prompting, caller DTMF input, wait treatment (e.g., music on hold), Automatic Speech Recognition, and Text to Speech.
- Voice Portal applications canvassing of Communication Manager for skilled agent availability before transferring inbound calls to the skills.
- Voice 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 Voice 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 Voice Portal destined for agents/stations connected to Communication Manager, if unanswered, are covered to Modular Messaging.
- Voice 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 Voice 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 Voice 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 Avaya Voice Portal 5.1 and hence not supported.
2. Avaya Voice Portal 5.1 does not send DTMF digits using RFC2833 so the Legacy Transfer Connect using Voice Portal application could not be tested. Legacy Transfer Connect using the agent/telephone was successfully tested.
3. G.726 codec is not supported by Avaya Voice Portal 5.1.
4. If Avaya Aura® Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Avaya Aura® Communication Manager, then Avaya Aura® Communication Manager selects a codec according to the priority order specified in the Avaya Aura® Communication Manager codec set, not the priority order specified in the SDP offer. For example, if the AT&T IP Toll Free service offers G.711, G.729A, and G.729B in that order, but the Avaya Aura® Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Avaya Aura® Communication Manager selects G.729A, not G.711. The practical resolution is to provision the Avaya Aura® Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.
5. Although Avaya Voice Portal release 5.1 and Communication Manager 5.2.1 support the possibility of using SIP phones as valid telephone extensions, SIP phones were not tested as part of the configuration used to validate this solution.
6. For an outcall to an 8YY number from Voice Portal, the Voice 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:

- Avaya Voice Portal provides interactive voice response services to inbound callers. Avaya Voice Portal consists of one or more Media Processing Platform (MPP) servers and a Voice Portal Management System (VPMS) server. Single server was used for MPP and VPMS for this reference configuration.
- Avaya Aura® Communication Manager provides the enterprise voice communications services. In this sample configuration, Avaya Aura® Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers.
- Avaya Aura® Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya Aura® Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- Avaya Aura® System Manager provides a common administration interface for centralized management of all Avaya Aura® 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 46xx and 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) 3800[1] provides SIP Session Border Controller (Acme SBC) functionality between the AT&T IP Toll Free service and the enterprise internal network[2]. 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 Avaya Voice Portal which are referenced in Avaya Voice Portal.
- The Speech Server consists of Nuance Recognizer and Nuance Vocalizer. Avaya Voice Portal uses the Speech Server for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) capabilities.
- An existing Avaya Modular Messaging system (in Multi-Site mode in this reference configuration) provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Modular 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.

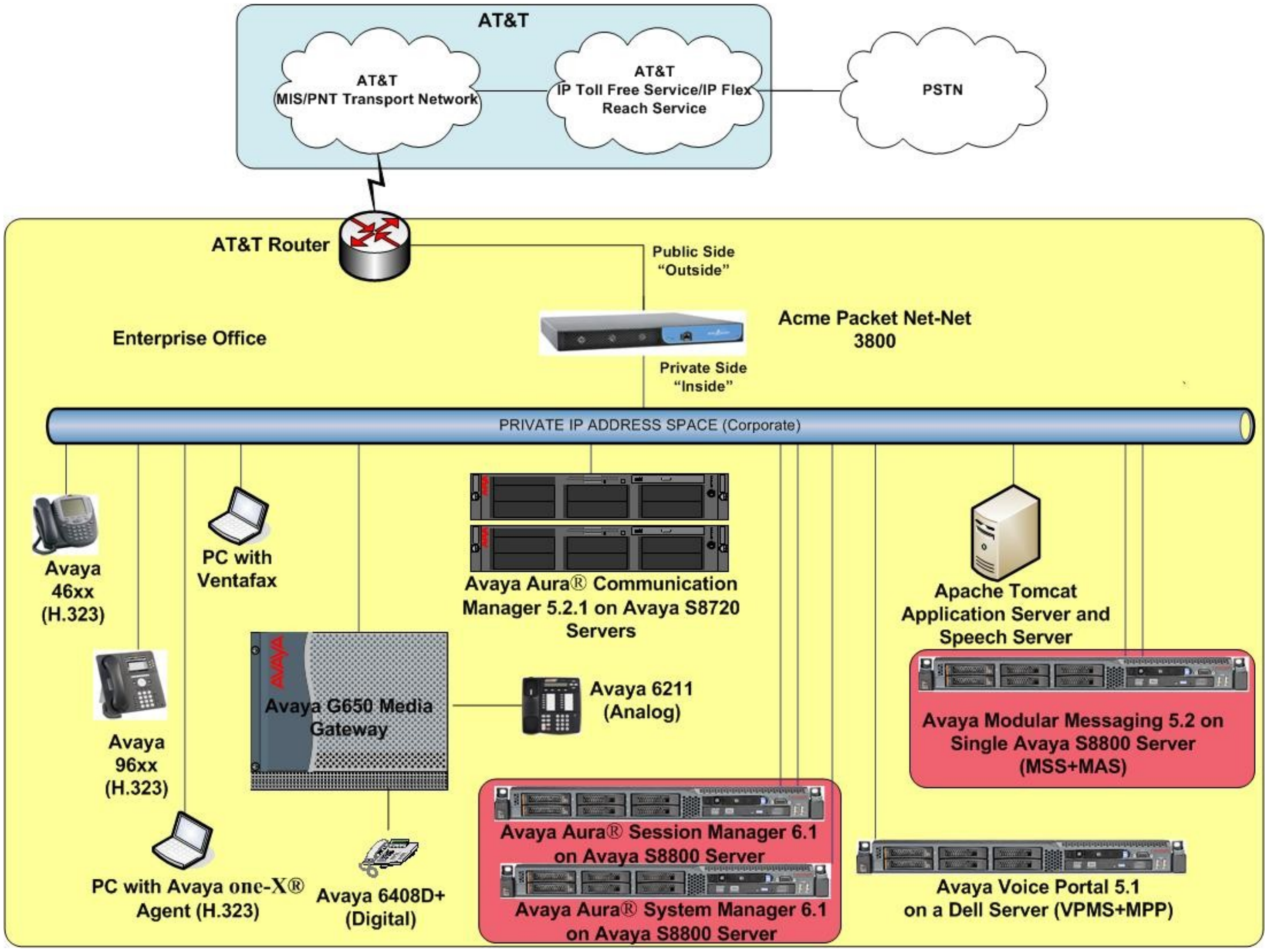


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** 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 Voice Portal	
VPMS/MPP Servers IP Address	10.80.100.54
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	20xx
Skill (Hunt Group) Extensions	51xxx
Agent Extensions	53xxx
Phone Extensions	50xxx
Announcement Extensions	33xxx
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	135.242.225.200
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 Voice Portal	00041530xxxxx

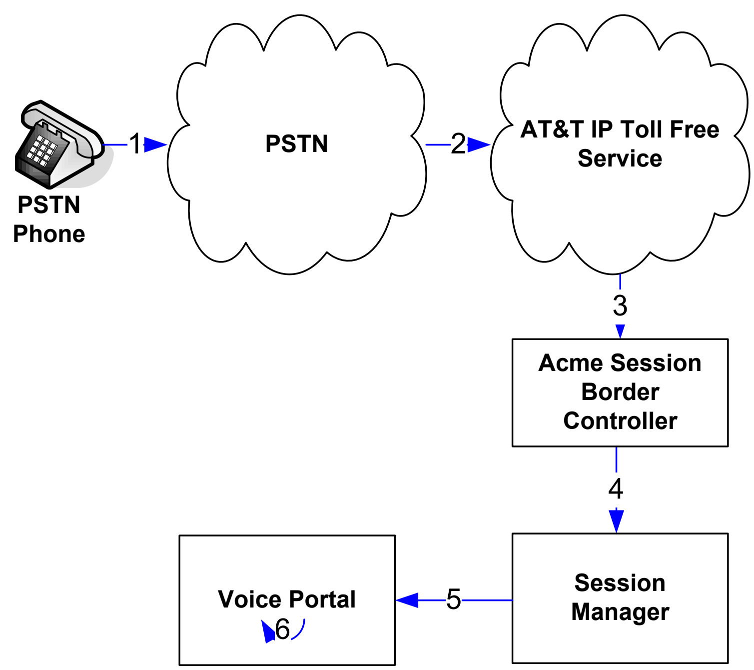
Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

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

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

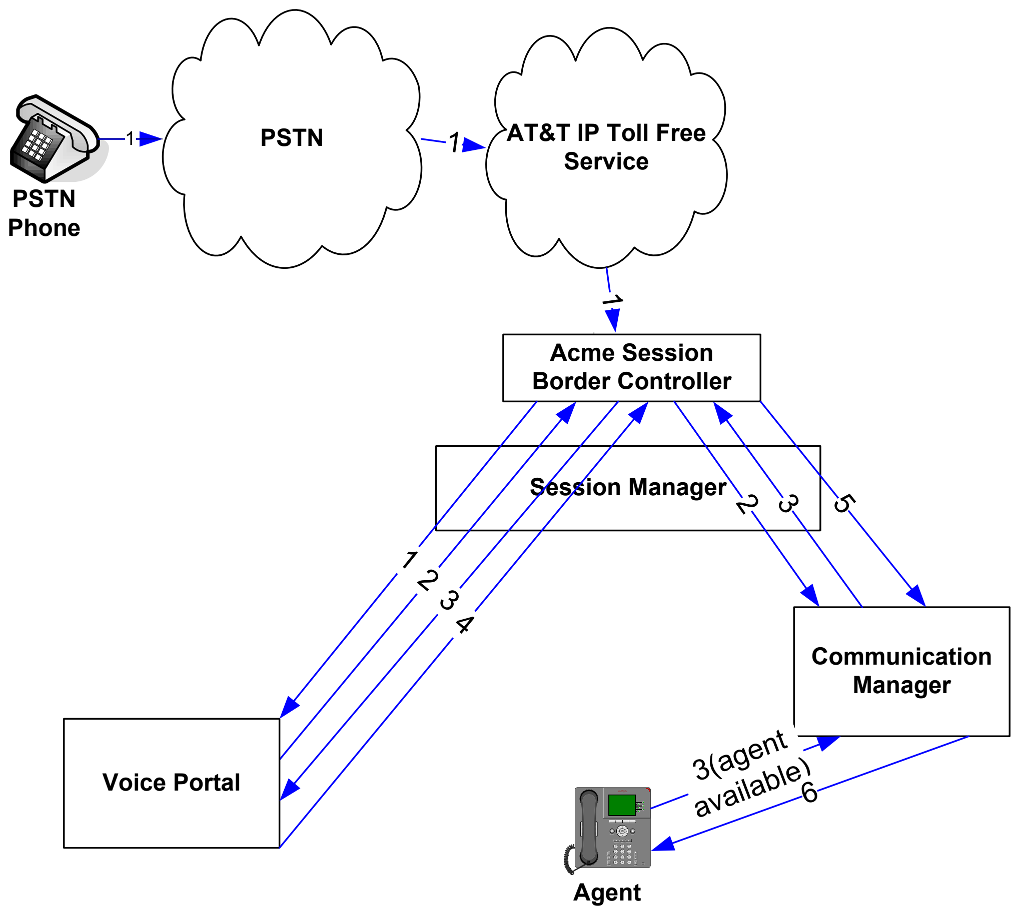
1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to the Acme SBC.
4. Acme SBC performs any necessary SIP header modifications, and routes the call to Session 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 Voice Portal.
6. Voice 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 Avaya Voice Portal

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

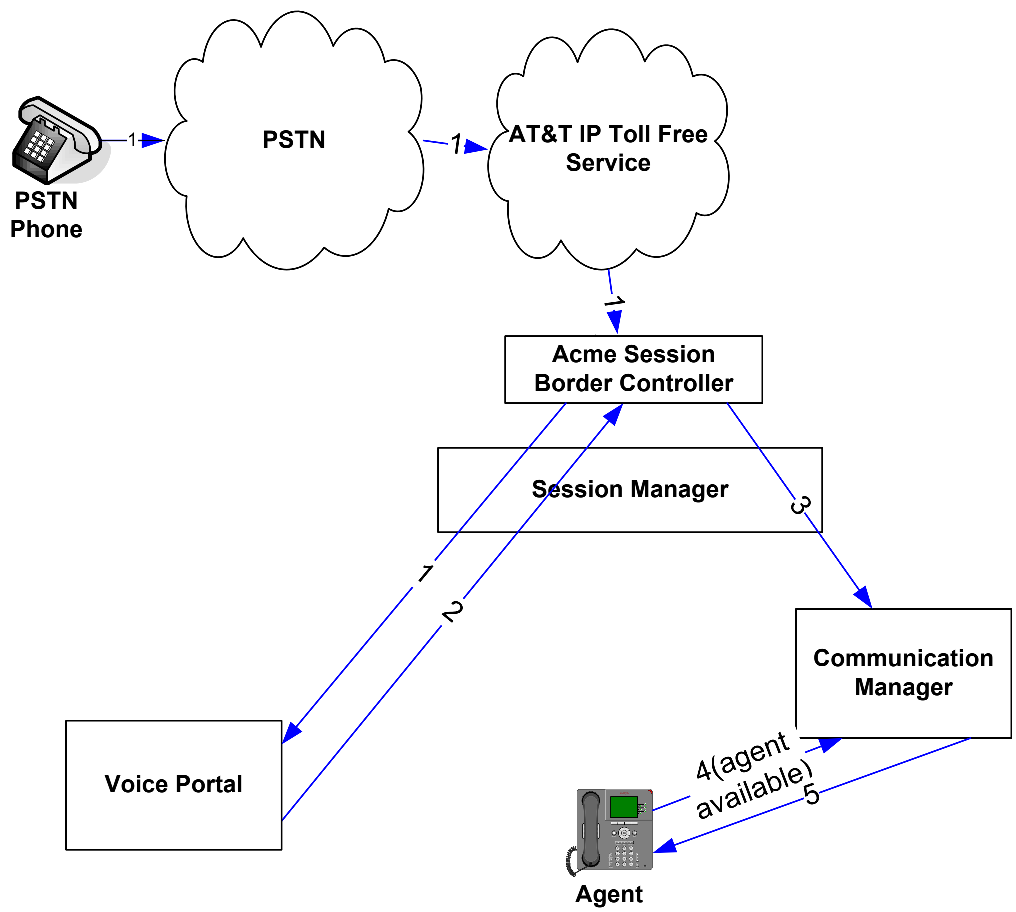
1. Same as the first five steps from the first call scenario.
2. In this scenario, the application is not sufficient to meet the caller's requests, and thus the call needs to be transferred to a Communication Manager agent. Voice Portal then puts the inbound call on hold and places a call to vector/skill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

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

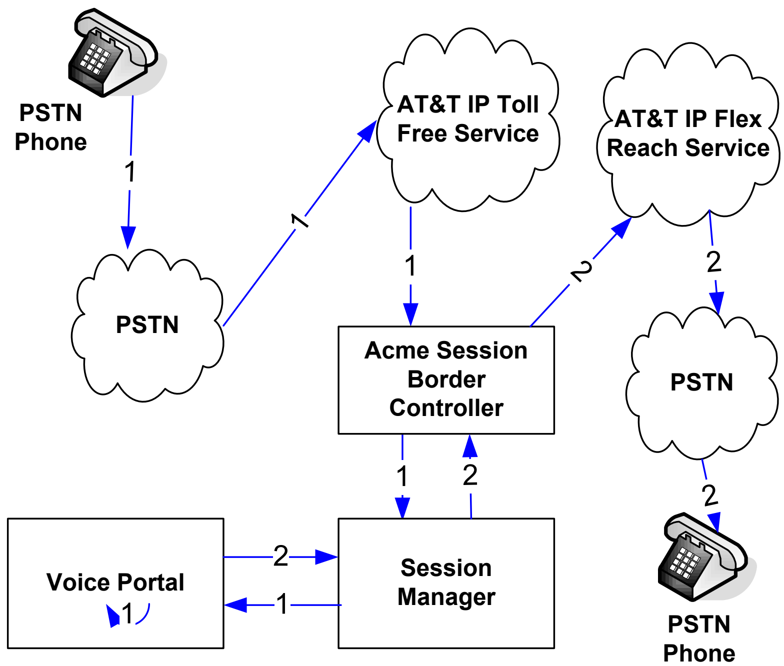
1. Same as the first five steps from the first call scenario.
2. In this scenario, the application on Voice Portal is not sufficient to meet the caller's needs or requests, and thus the call needs to be transferred to an agent/skill on Communication Manager. Voice Portal instructs the Acme SBC via Session 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 Voice Portal to Communication Manager regardless of Agent Availability

The fourth call scenario illustrated below is an inbound call arriving on Voice 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, Voice Portal forwards the call to an appropriate PSTN number which can be a regular PSTN number or an 8YY number.



Inbound Call forwarded by Voice 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 Voice Portal	5.1
Voice Portal Management System (VPMS)	5.1.0.1.1601
Media Processing Platform (MPP)	5.1.0.1.1601
Avaya S8720 Server	Avaya Aura® Communication Manager 5.2.1 with Service Pack 4 (R015x.02.1.016.4-18942)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW054
TN799DP Control-LAN (C-LAN)	HW00 FW040
TN2602AP IP Media Processor (MedPro)	HW02 FW060
TN2501AP VAL-ANNOUNCEMENT	HW02 FW018
Avaya S8800 Server	Avaya Aura® System Manager 6.1.0 (SP4)
Avaya S8800 Server	Avaya Aura® Session Manager 6.1.0 (SP4)
Avaya 9650 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.11
Avaya 9611 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.0
Avaya 4620SW IP Telephone	2.9.1
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 20

Table 2: 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 one[3] 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**.

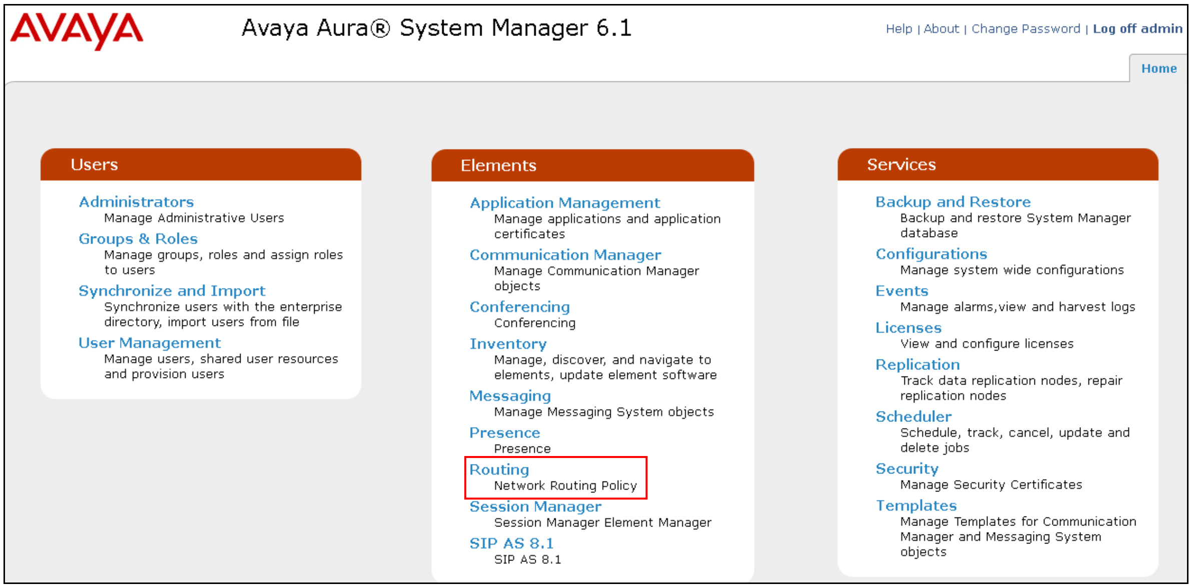
The general strategy employed in this reference configuration with regard to Called Party Number manipulation and matching, and call routing is as follows:

- 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).

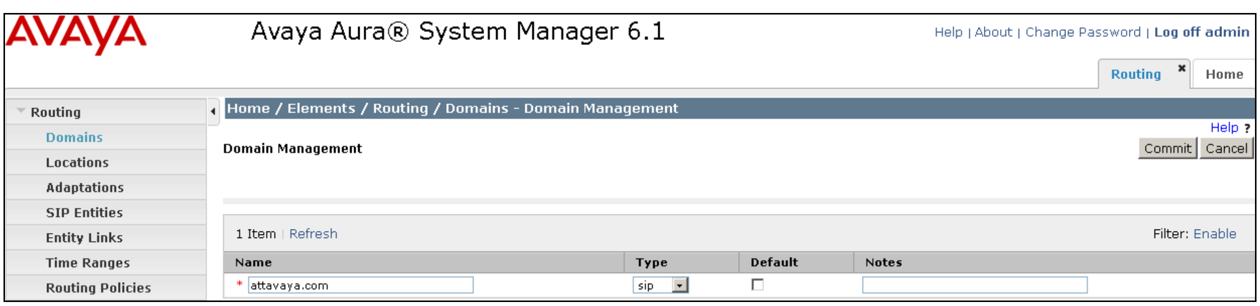


Main Routing Page

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.
   - **Type** – Set to **sip**.
   - **Notes** – Optional Field.
2. Click **Commit**.
3. Repeat above steps to add additional domains.

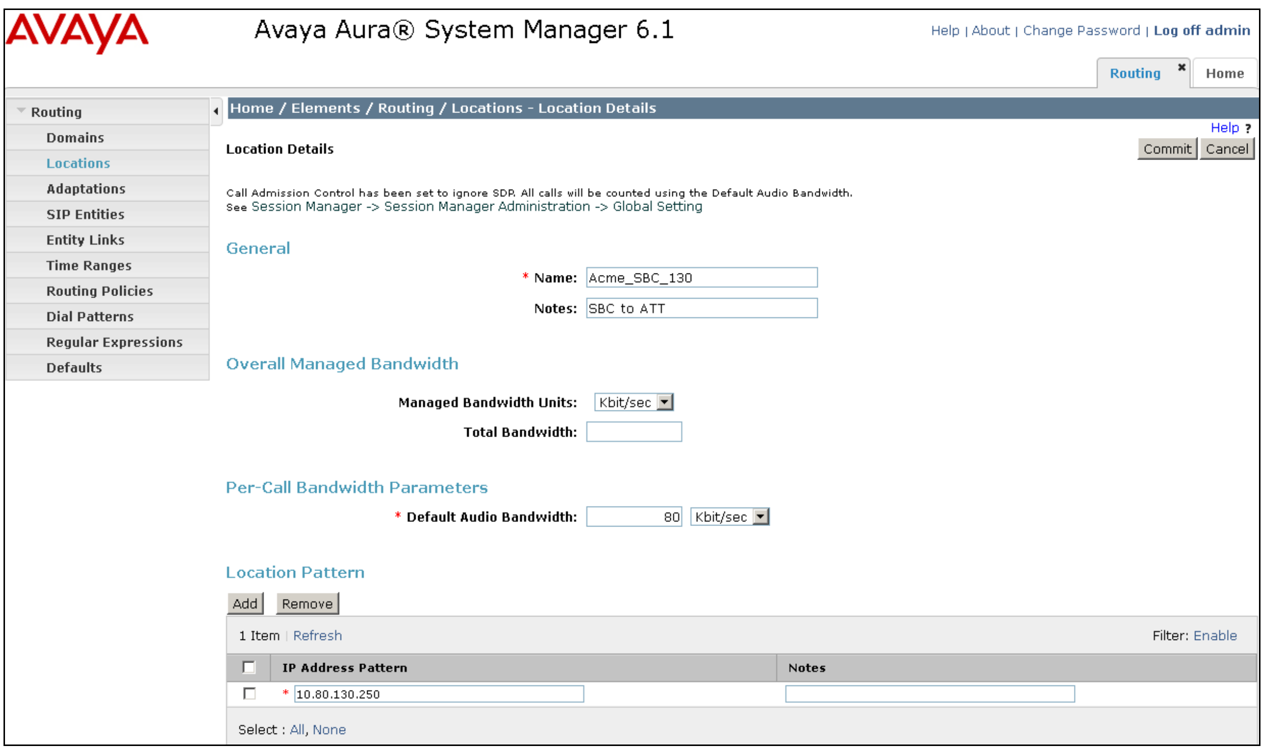


Domain Management Page

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** – (Optional) Enter a description.
   - **Managed Bandwidth** and **Average Bandwidth per Call** – [Optional] To limit the number of calls going to and from this location i.e., apply Call Admission Control.
   - **Location Pattern** - [Optional] 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. **Subnet 10.80.100.x** for **Voice Portal**, **10.80.150.x** for **Session Manager (Location_150_SM)** and **10.80.130.x** for **Communication Manager**) used in this reference configuration.



Location Details Page for Acme SBC

5.5. Adaptations

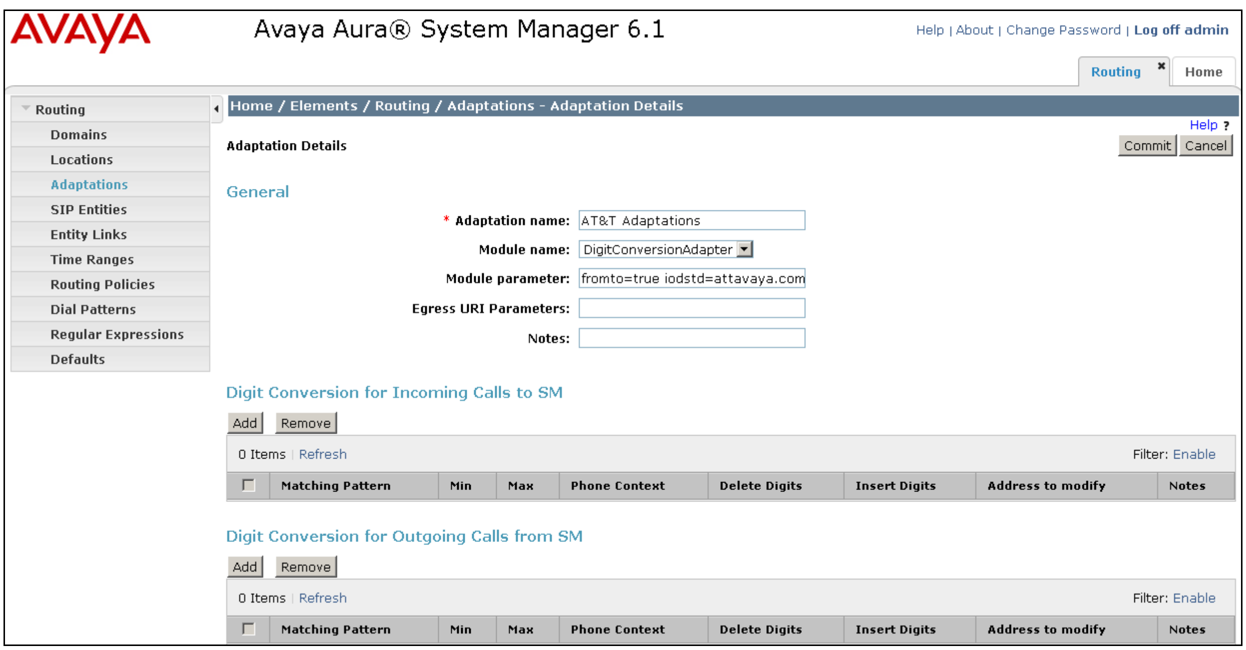
Adaptations on Session Manager are always between Session Manager and another entity. Adaptations could potentially be applied to both calls coming into Session Manager and going out from the Session Manager. In this section, Adaptations are administered for calls from AT&T to Voice Portal (**Section 5.5.1**) and the calls forwarded from Voice Portal to Communication Manager (**Section 5.5.2**).

5.5.1. Adaptation for Calls to Avaya Voice 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** page, 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 odstd=135.242.225.200**, which will replace the IP Address/Domain in the Request URI and **To** header with the Avaya CPE domain **attavaya.com** for egress to Voice Portal. Also, it replaces the domain in the calls originating from Voice Portal destined for Acme SBC to the IP Address of the AT&T Border element.
3. Click **Commit**.

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



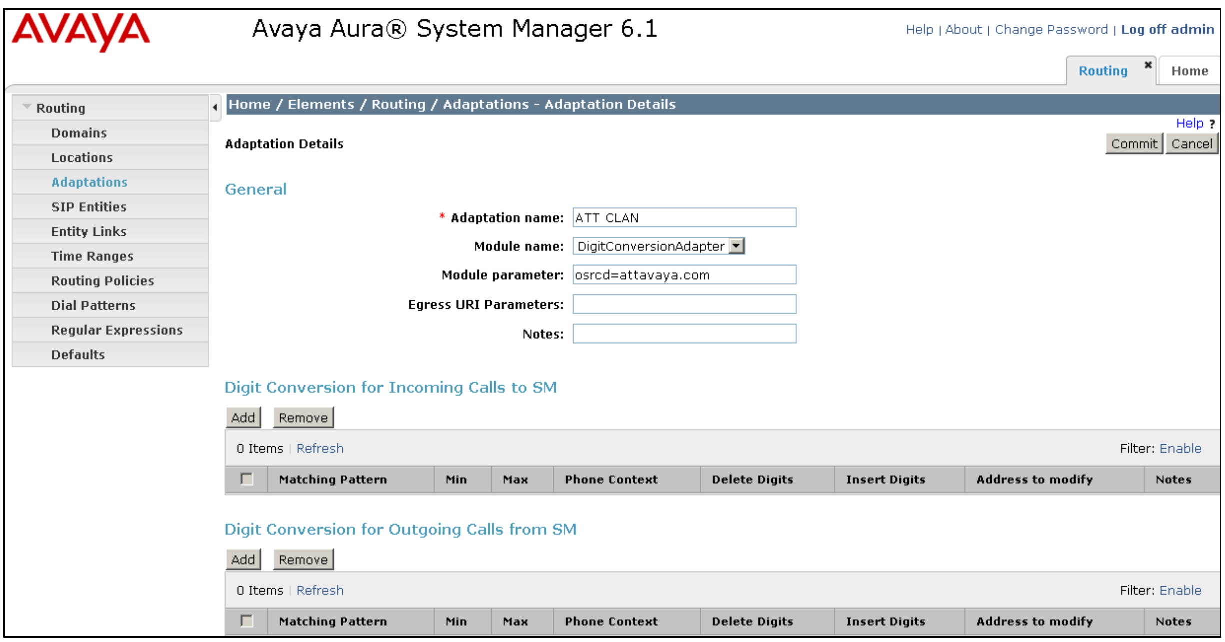
Adaptation Details Page – Adaptation for Voice Portal

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** page, 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. Click **Commit**.

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



Adaptation Details Page – Adaptation for Communication Manager

5.6. SIP Entities

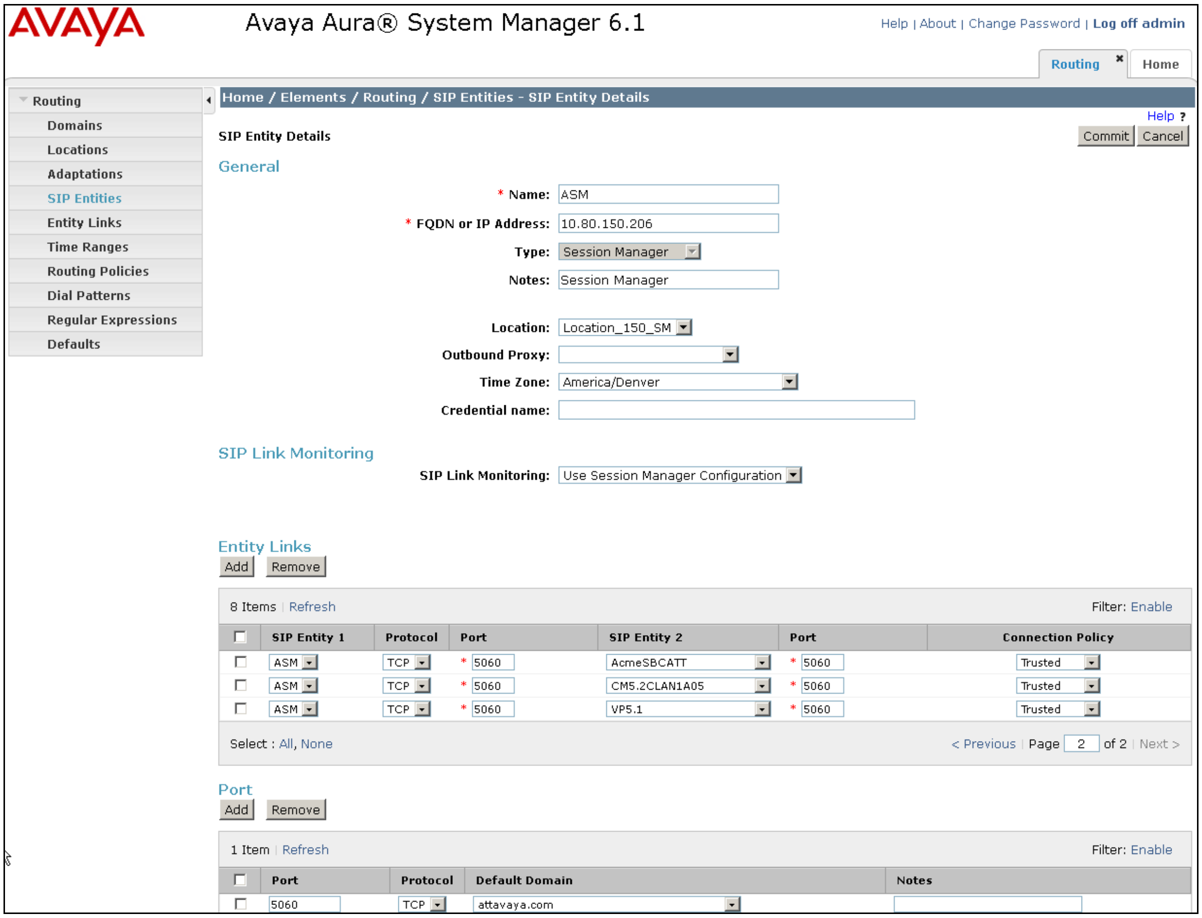
In this section, SIP Entities are administered for the following SIP network elements:

- Avaya Aura® Session Manager
- Avaya Voice Portal
- Avaya Aura® Communication Manager
- Acme Session Border Controller
- Avaya Modular 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

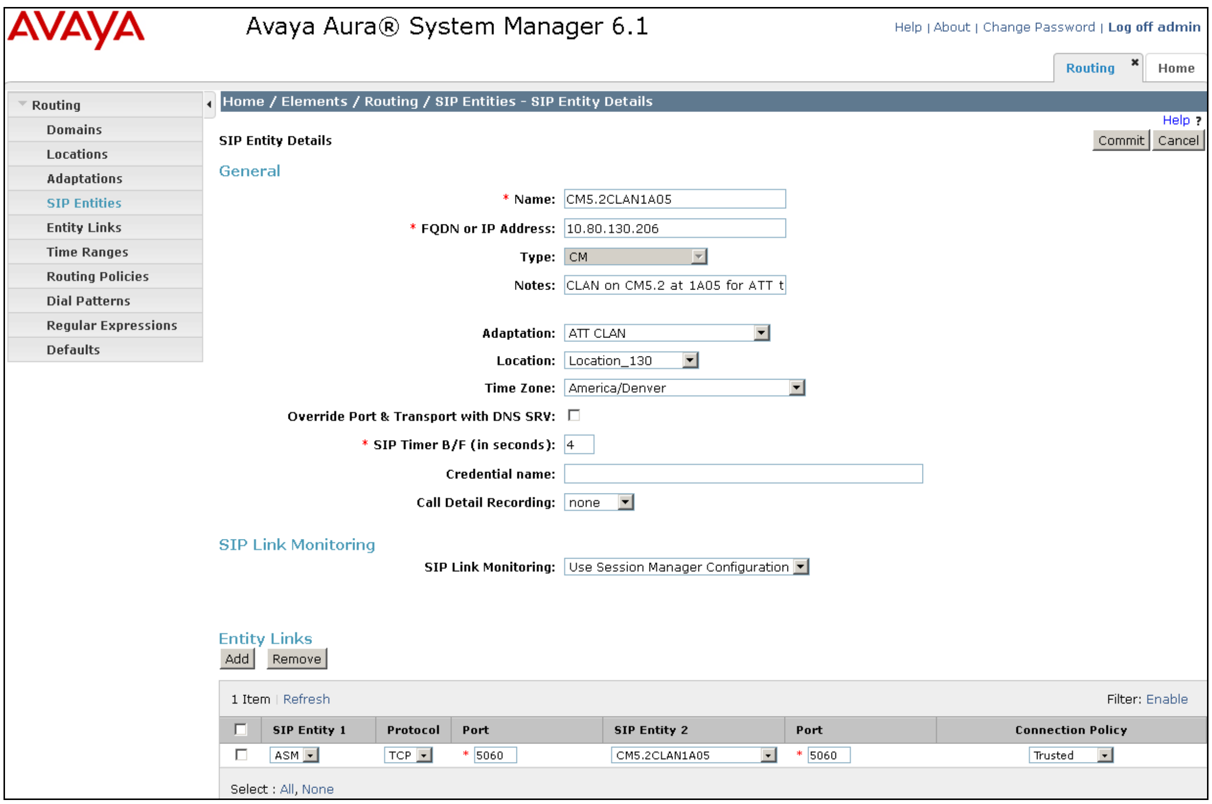
1. In the left pane under **Routing**, click **SIP Entities**. In the **SIP Entities** page click **New** [not shown].
2. In the **General** section of the **SIP Entity Details** page, configure as follows:
   - **Name –** Enter a descriptive name for Session Manager.
   - **FQDN or IP Address –** Enter 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**.
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**.



SIP Entity Details Page –Session Manager SIP Entity

5.6.2. Avaya Aura® Communication Manager SIP Entity

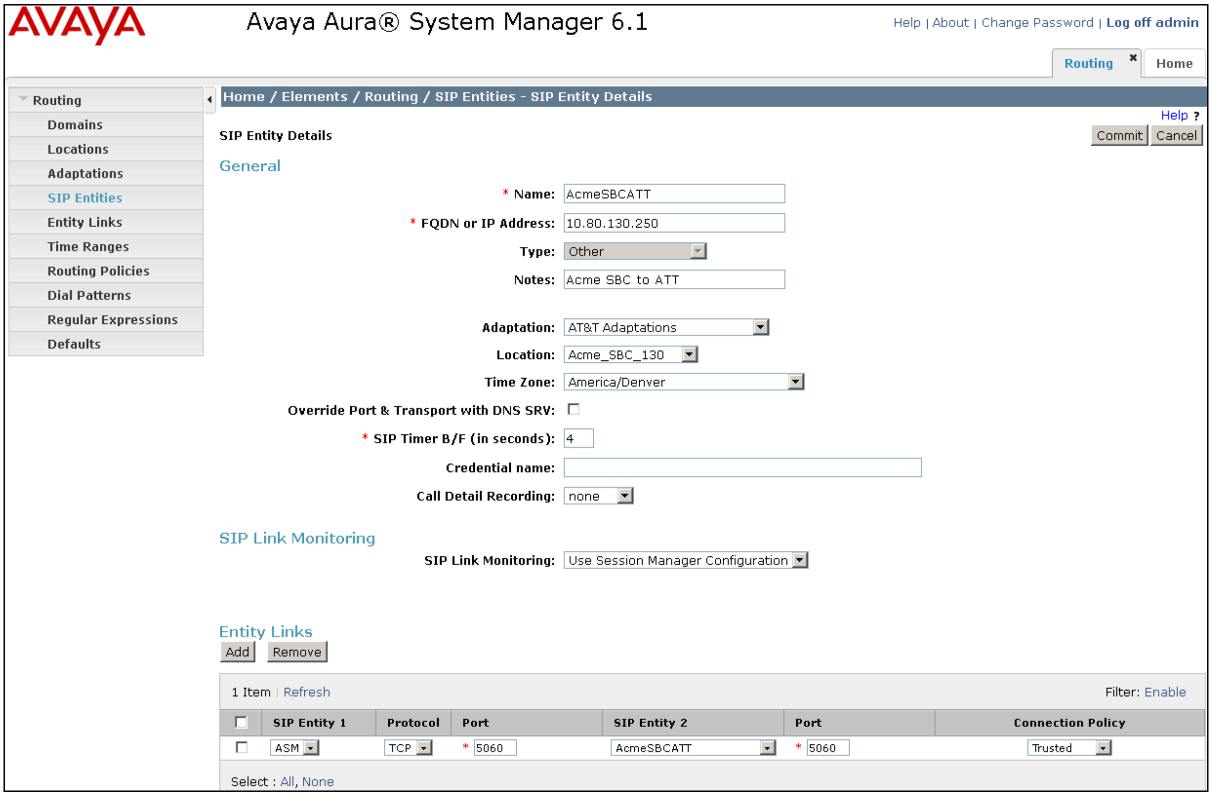
1. In the **SIP Entities** page, click **New** [not shown].
2. In the **General** section of the **SIP Entity Details** page, configure as follows:
   - **Name –** Enter any descriptive name for the Communication Manager Signaling Interface.
   - **FQDN or IP Address –** Enter the IP address of the Communication 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**.



SIP Entity Details Page –Communication Manager SIP Entity

5.6.3. Acme Session Border Controller SIP Entity

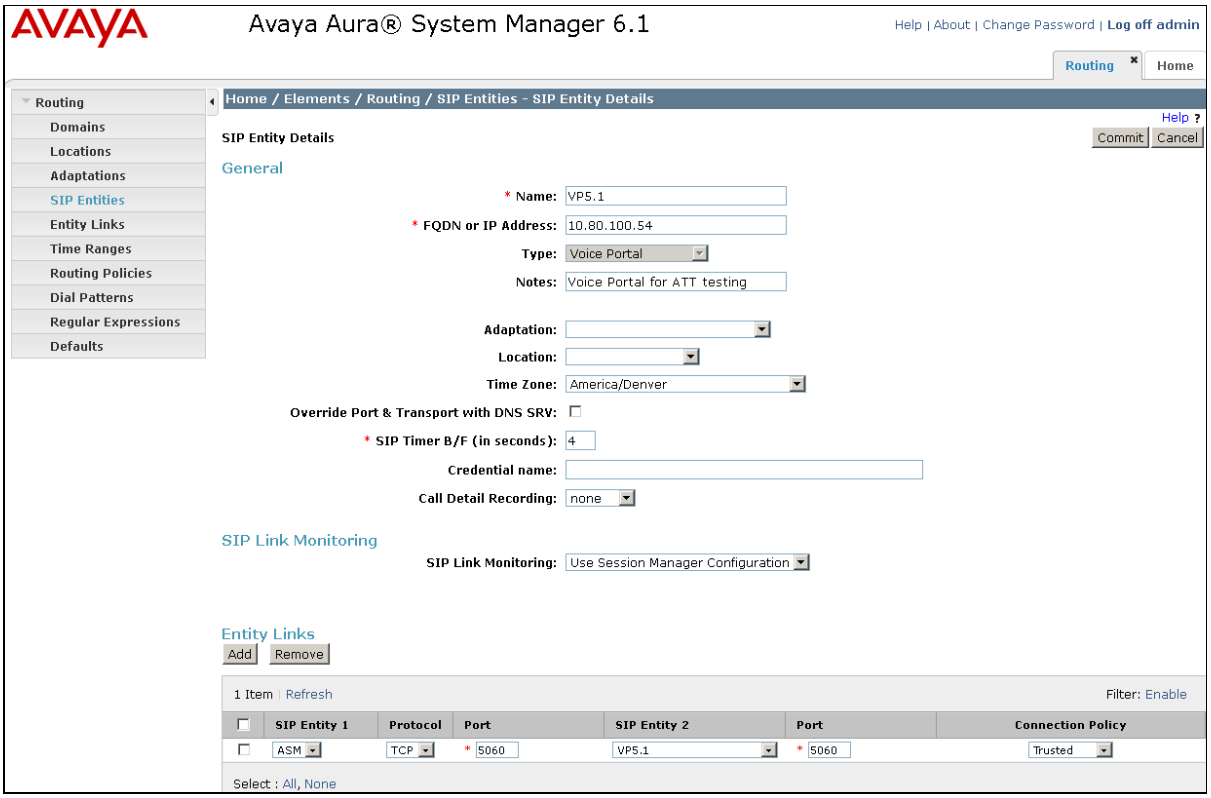
To configure the Session Border Controller Entity, repeat the Steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of the private (inside) interface configured in **Section 9** under **network interface** section and the **Type** field is set to **Other**. The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



SIP Entity Details Page – Session Border Controller SIP Entity

5.6.4. Avaya Voice Portal Entity

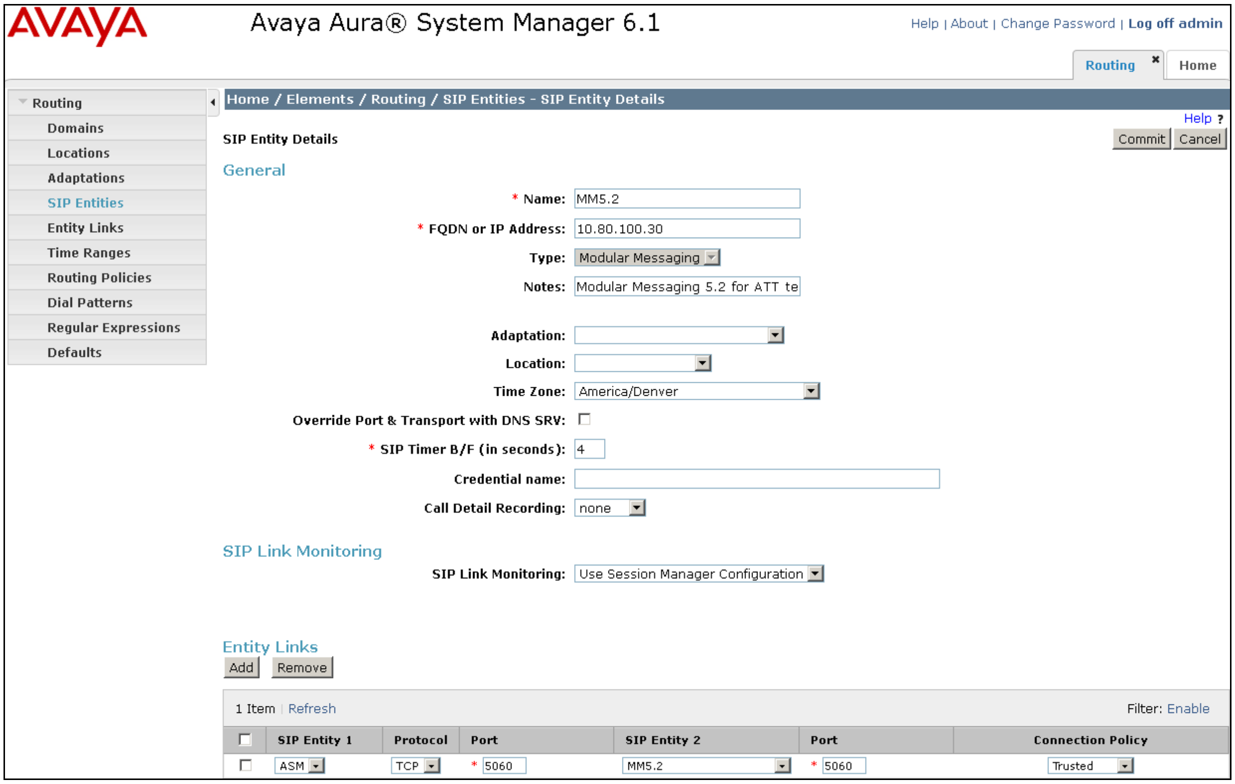
To configure the Voice Portal Entity, repeat the Steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of the Voice Portal and the **Type** field is set to **Voice Portal**. The entity link is configured/displayed in **Section 5.7.3**. See the screen below for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



SIP Entity Details Page –Voice Portal SIP Entity

5.6.5. Avaya Modular Messaging SIP Entity

To configure the Modular Messaging SIP Entity, repeat the steps in **Section 5.6.2**. The **FQDN or IP Address** field is populated with the IP address of the Modular Messaging Application Server (MAS) 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.



SIP Entity Details Page – Avaya Modular Messaging SIP Entity

5.7. Entity Links

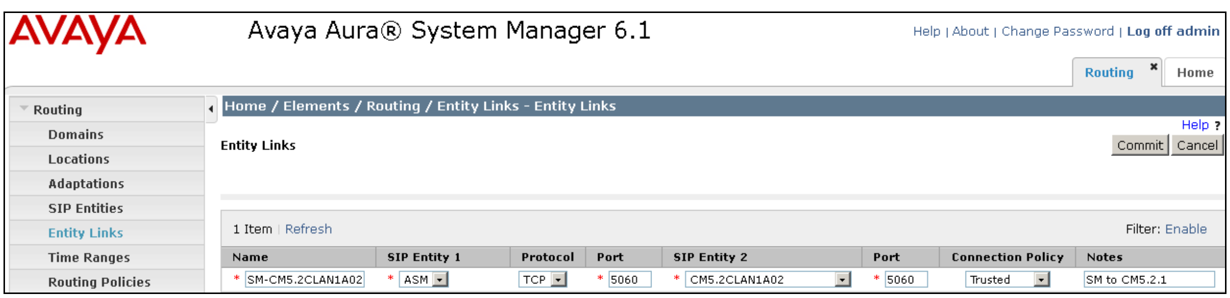
In this section, Entity Links are administered between Avaya Aura® Session Manager and the following SIP Entities:

- Avaya Aura® Communication Manager
- Acme Session Border Controller
- Avaya SIP Voice Portal
- Avaya Modular 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 VoicePortal/Communication Manager and Session Manager in customer environments.

5.7.1. Entity Link to Avaya Aura® Communication Manager

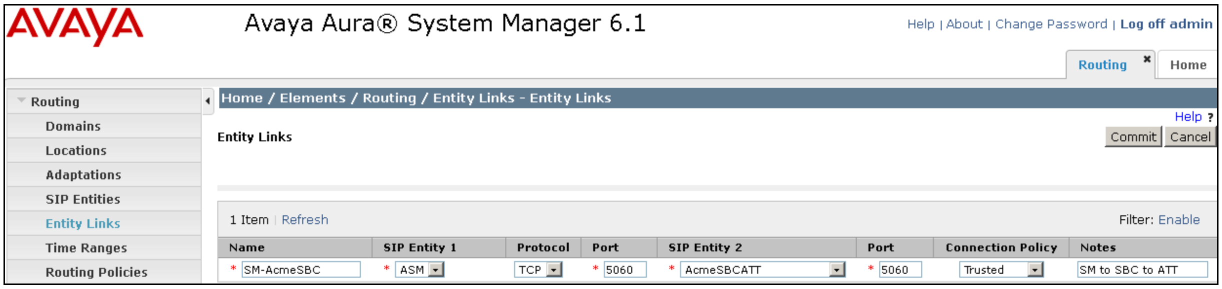
1. In the left pane under **Routing**, click **Entity Links**. In the **Entity Links** page click **New** [not shown].
2. On the **Entity Links** page, provision as follows:
   - **Name** – Enter a descriptive name for this link to Communication Manager.
   - **SIP Entity 1** – Select the SIP Entity 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**.



Entity Links Page – Entity Link to Communication Manager

5.7.2. Entity Link to Acme Session Border Controller

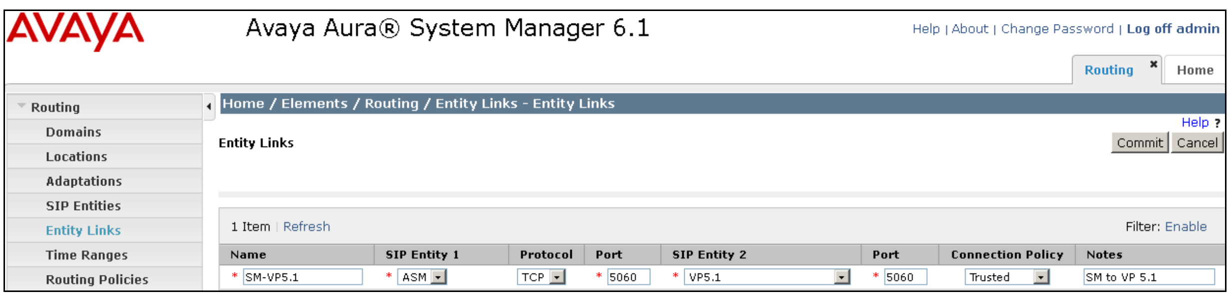
To configure the entity link between the Session Manager and Acme SBC SIP entity, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.3**. See the screem below for the values used in this reference configuration.



Entity Links Page – Entity Link to Acme SBC SIP Entity

5.7.3. Entity Link to Avaya Voice Portal

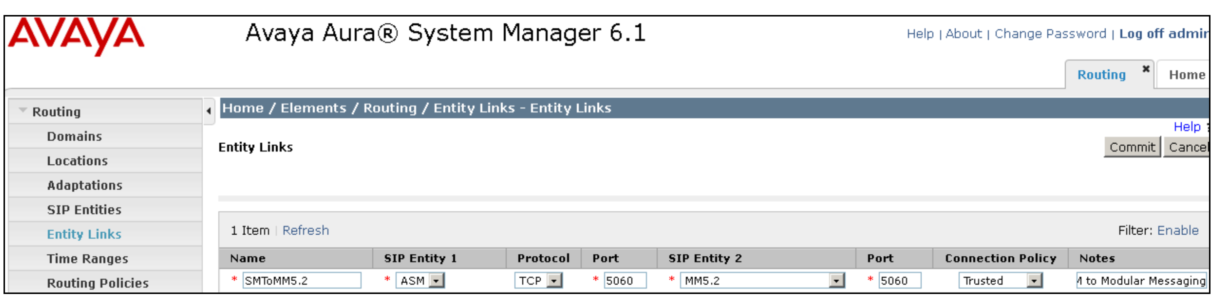
To configure this entity link, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.4**. See the screen below for the values used in this reference configuration.



Entity Links Page – Entity Link to Voice Portal SIP Entity

5.7.4. Entity Link to Avaya Modular Messaging

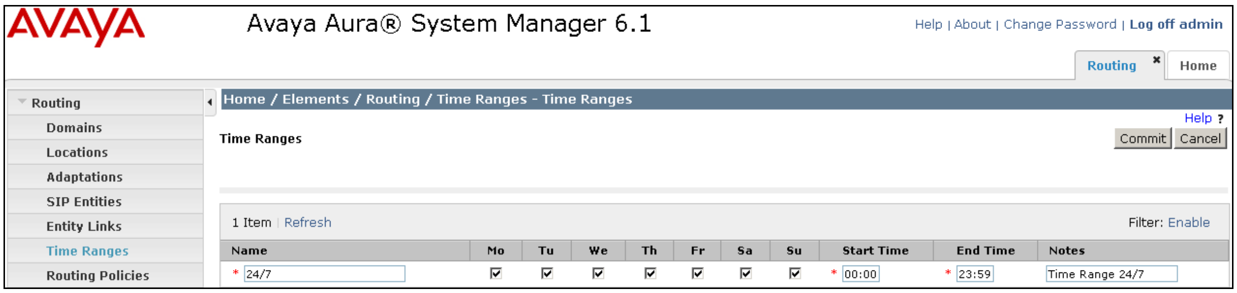
To configure this entity link, repeat the steps in **Section 5.7.1**. The **SIP Entity 2** field is populated with the SIP Entity configured in **Section 5.6.5**. See the screen below for the values used in the reference configuration.



Entity Links Page – Entity Link to Avaya Modular 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.



Time Ranges Page

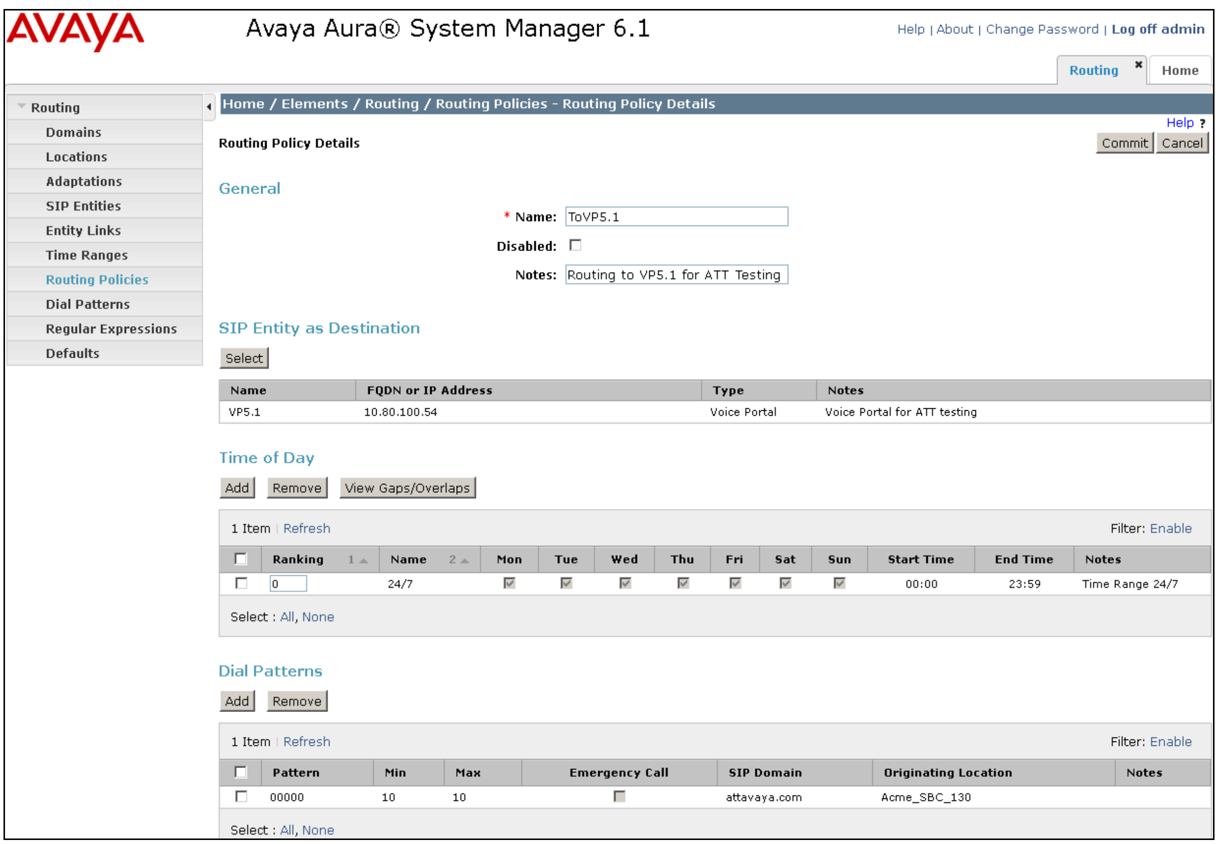
5.9. Routing Policies

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

- Routing Policy to Avaya Voice 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 Modular Messaging

5.9.1. Routing Policy to Avaya Voice Portal

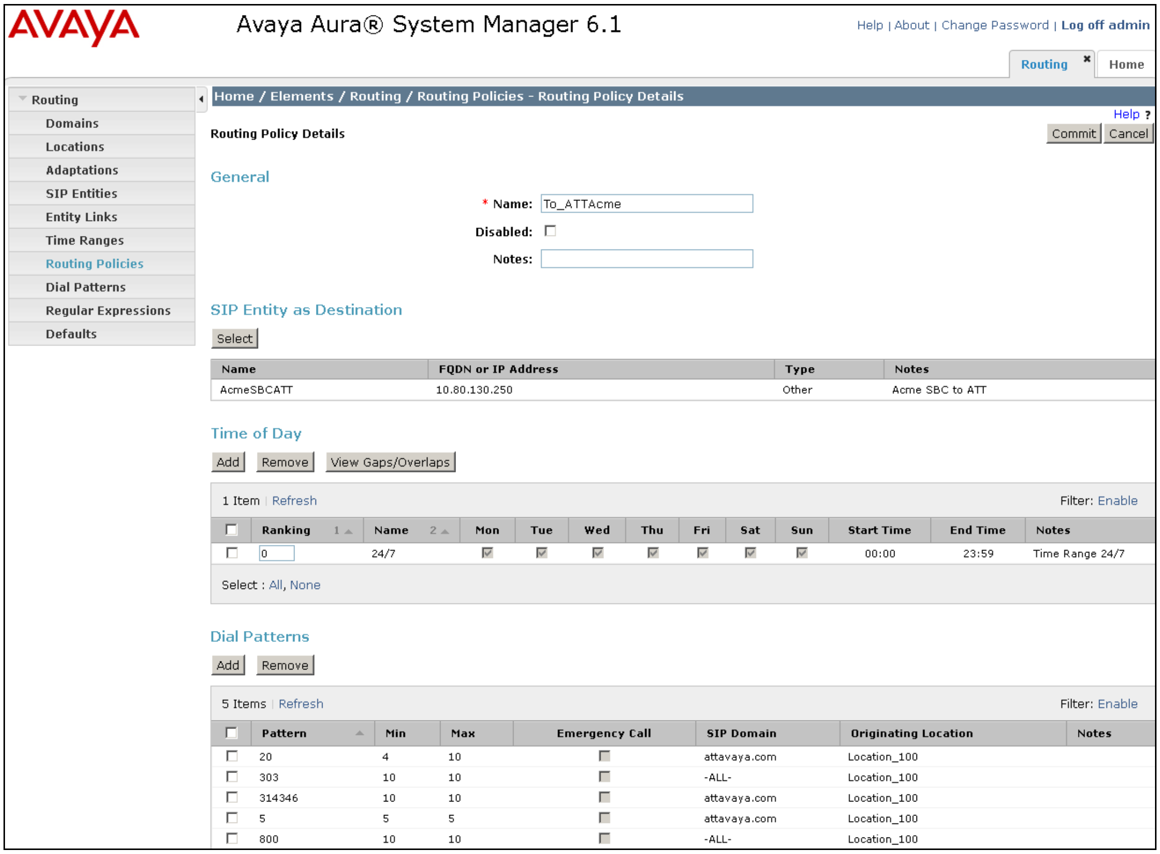
1. In the left pane under **Routing**, click **Routing Policies**. On the **Routing Policies** page click **New** [not shown].
2. In the **General** section of the **Routing Policy Details** page, enter a descriptive **Name** (e.g. **ToVP5.1**) for routing calls from AT&T IP Toll Free service via Acme 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 Voice 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**.



Routing Policy Details Page to Voice Portal

5.9.2. Routing Policy to Acme Session Border Controller

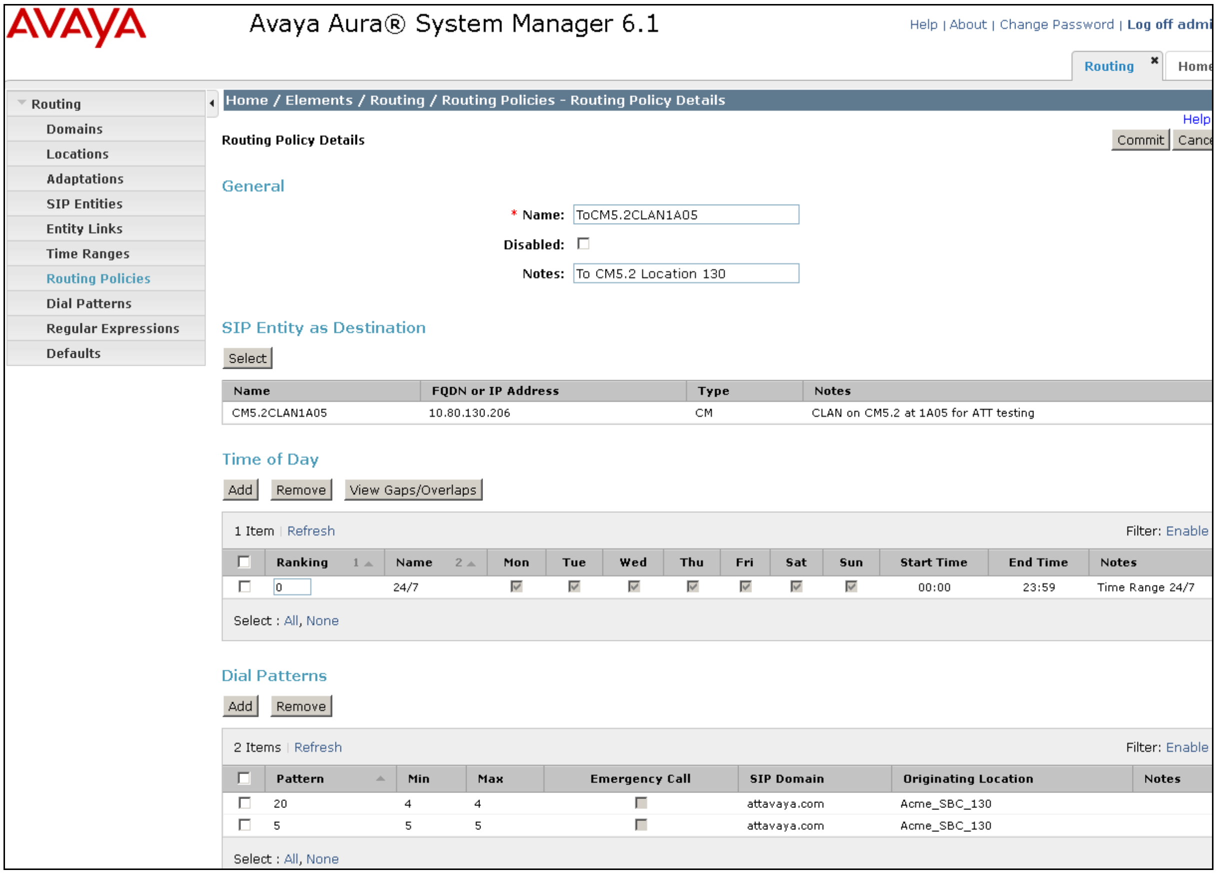
To configure routing policy to Acme SBC, repeat steps in **Section 5.9.1**. The following screen shows the routing policy configured for the calls to be routed to Acme SBC. Dial pattern/s for calls to be routed to Communication Manager are configured/displayed in **Section 5.10.2**.



Routing Policy Details Page to Acme SBC

5.9.3. Routing Policy to Aura® Communication Manager

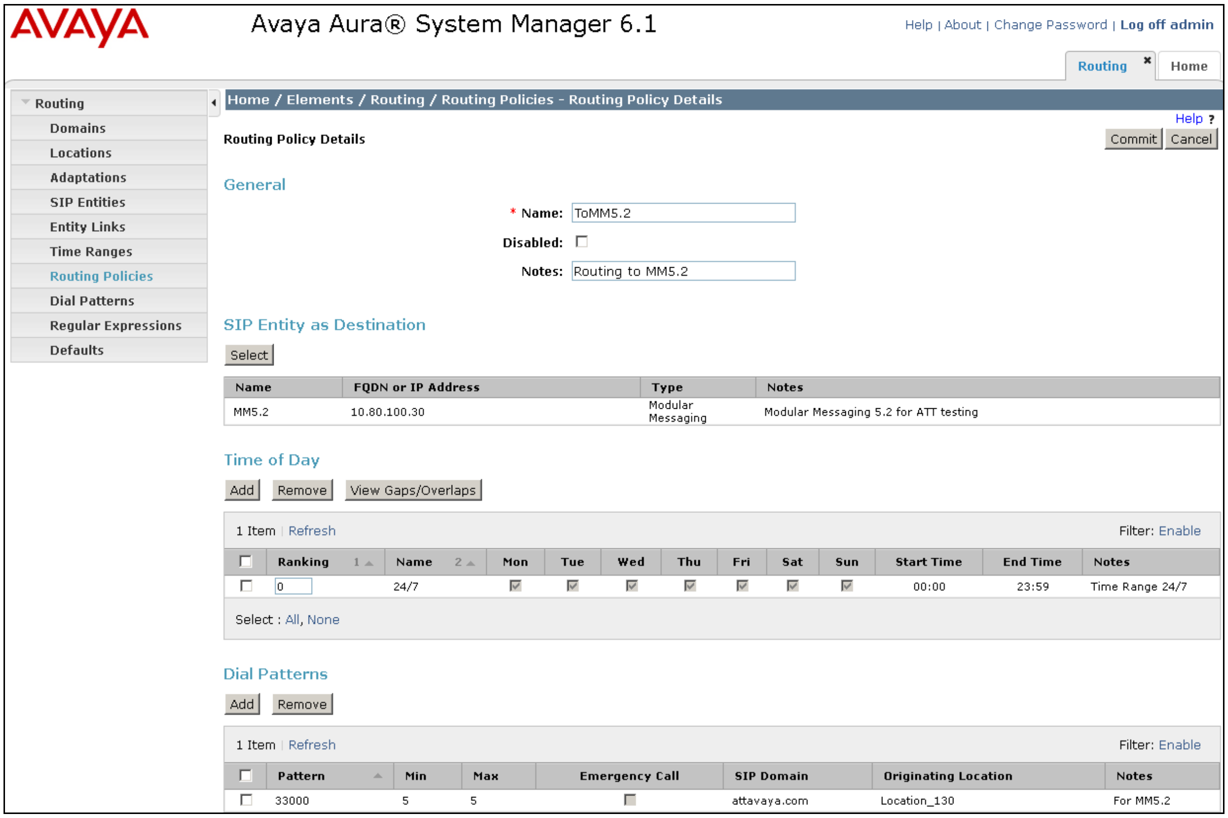
To configure routing policy to Comunication Manager, repeat steps in **Section 5.9.1**. The following screen shows the routing policy configured for the calls to be routed to Communication Manager. Dial pattern/s for calls to be routed to Communication Manager are configured/displayed in **Section 5.10.3**.



Routing Policy Details Page to Communication Manager

5.9.4. Routing Policy to Avaya Modular Messaging

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



Routing Policy Details Page to Avaya Modular 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 Voice Portal
- Dial Pattern for Acme SBC
- Calls transferred to Avaya Aura® Communication Manager
- Calls to Avaya Modular Messaging pilot number

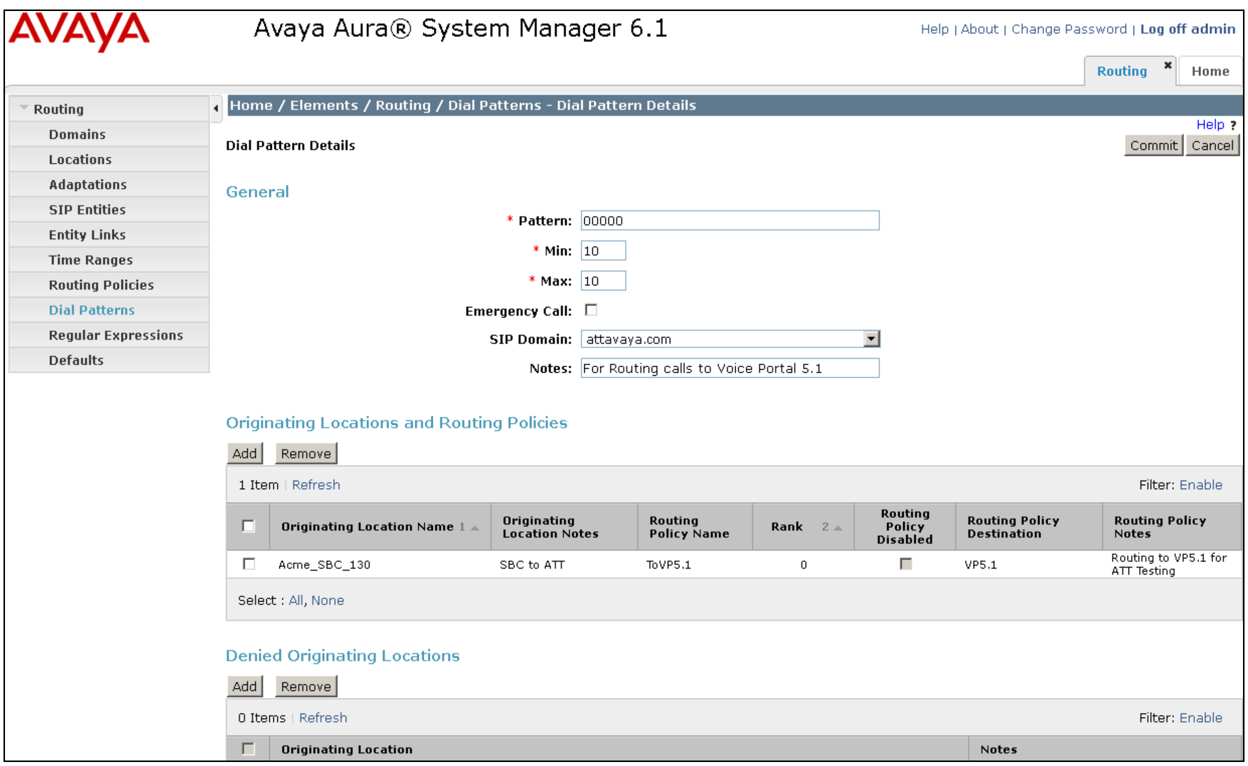
5.10.1. Matching Inbound Calls from AT&T IPTF Service to Avaya Voice Portal

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.

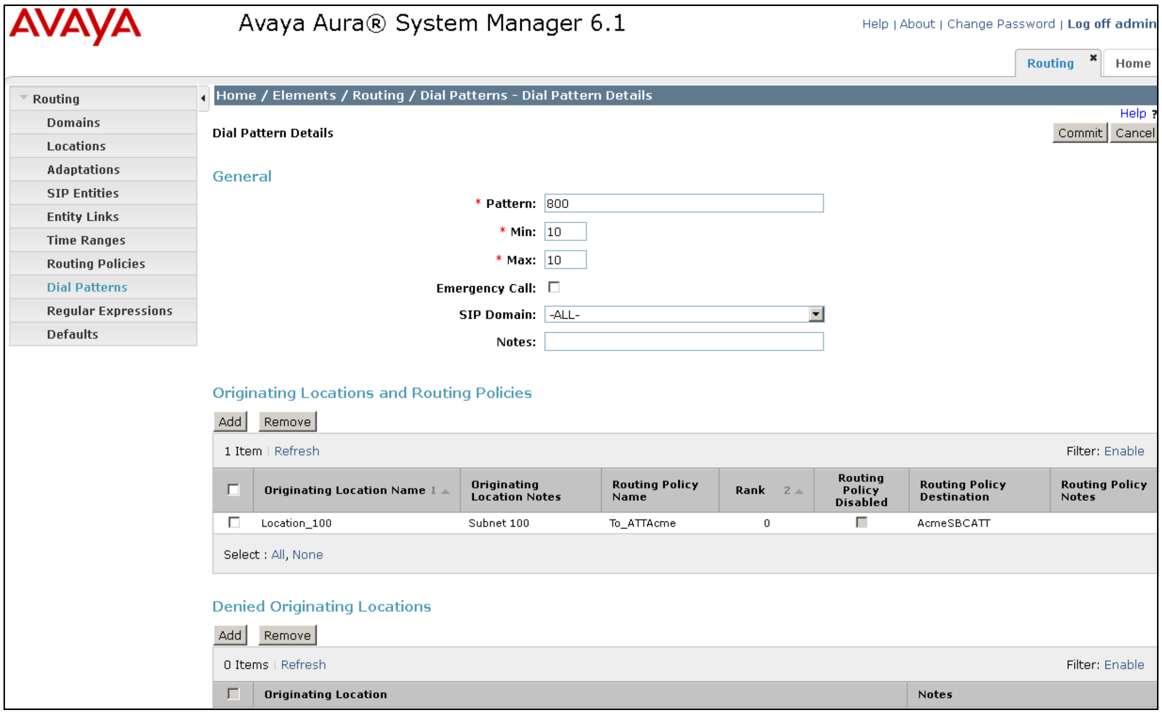
3. In the **Originating Locations and Routing Policies** section of the **Dial Pattern Details** page, click **Add**.
4. In the **Originating Location** section of the **Originating Location and Routing Policy List** page [not shown], select the locations from where calls can originate to be routed to Voice 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 Voice 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.



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

5.10.2. Dial Pattern for Acme SBC

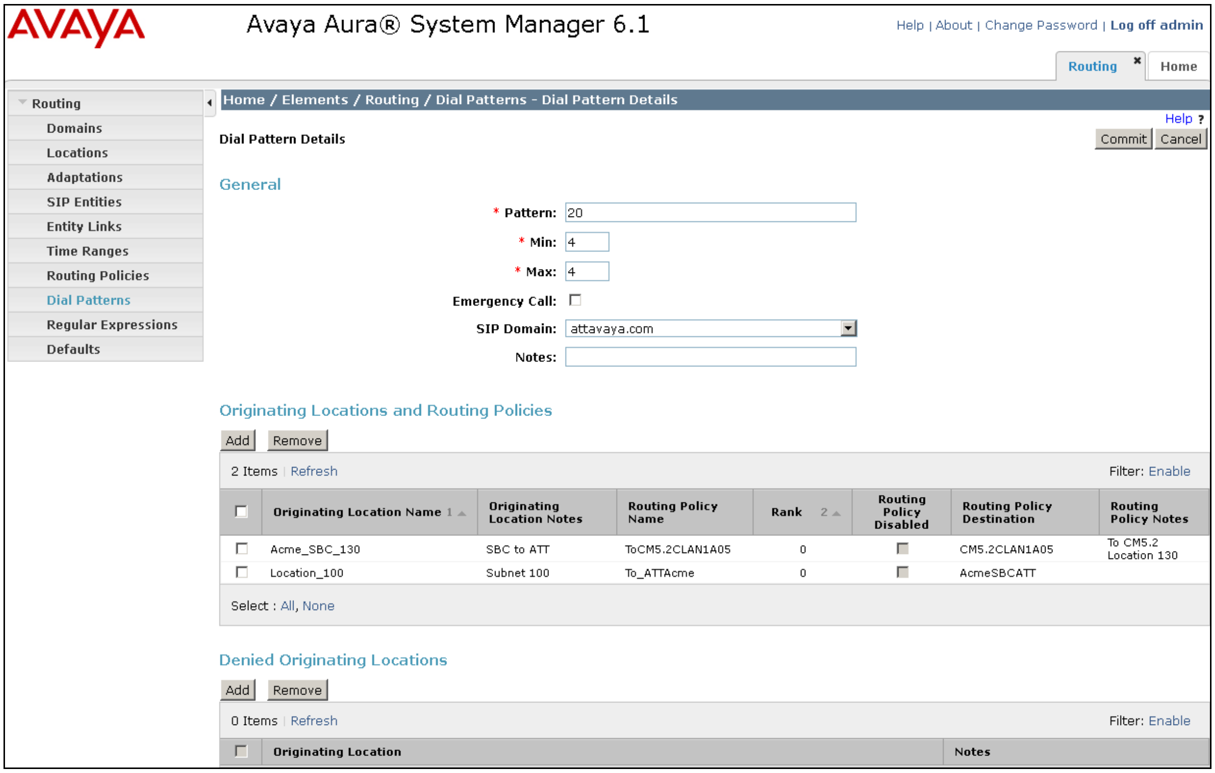
Repeat steps in **Section 5.10.1** to add additional dial patterns. The following screen shows the dial pattern configured for the calls to be routed to Acme SBC from Voice Portal in this reference configuration. Additional dial patterns **303xxxxxxx**, **20xxx** and **50xxx** were also configured. Calls from Voice 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** and **303** were forwarded to PSTN whereas calls to **20xxx** and **50xxx** were sent back to Session Manager for delivery to endpoints on Communication Manager.



Dial Pattern Details Page – for Acme SBC

5.10.3. Matching Calls to Avaya Aura® Communication Manager via Acme Session Border Controller

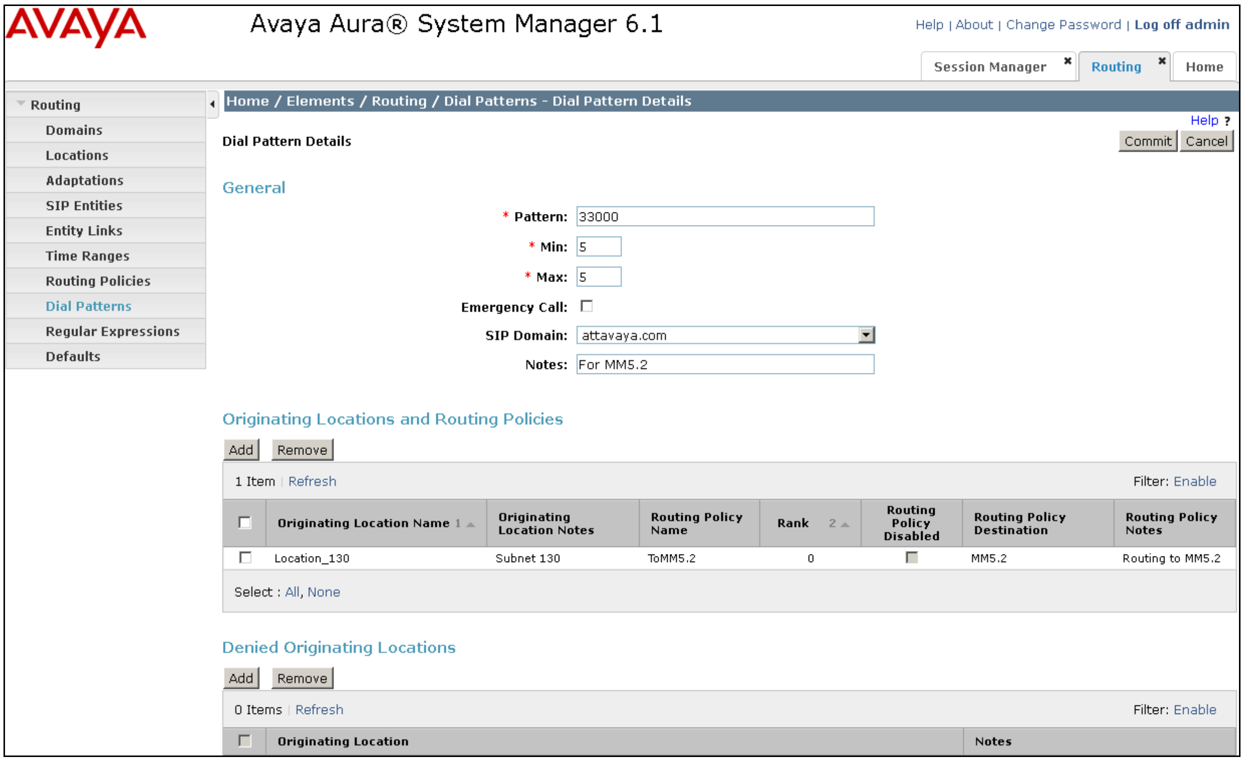
Repeat steps in **Section 5.10.1** to add additional dial patterns. The following screen shows the dial pattern configured for the calls to be routed to Communication Manager via Acme SBC from Voice Portal in this reference configuration. In this example, the calls from Voice Portal to **20xx** are first routed to Acme SBC which in turn routes the call back to Communication Manager via Session Manager. Another dial pattern was configured in this reference configuration for **5xxxx**.



Dial Pattern Details Page – Matching calls from Voice Portal for Acme SBC and Communication Manager

5.10.4. Matching Inbound Calls to Avaya Modular Messaging Pilot Number

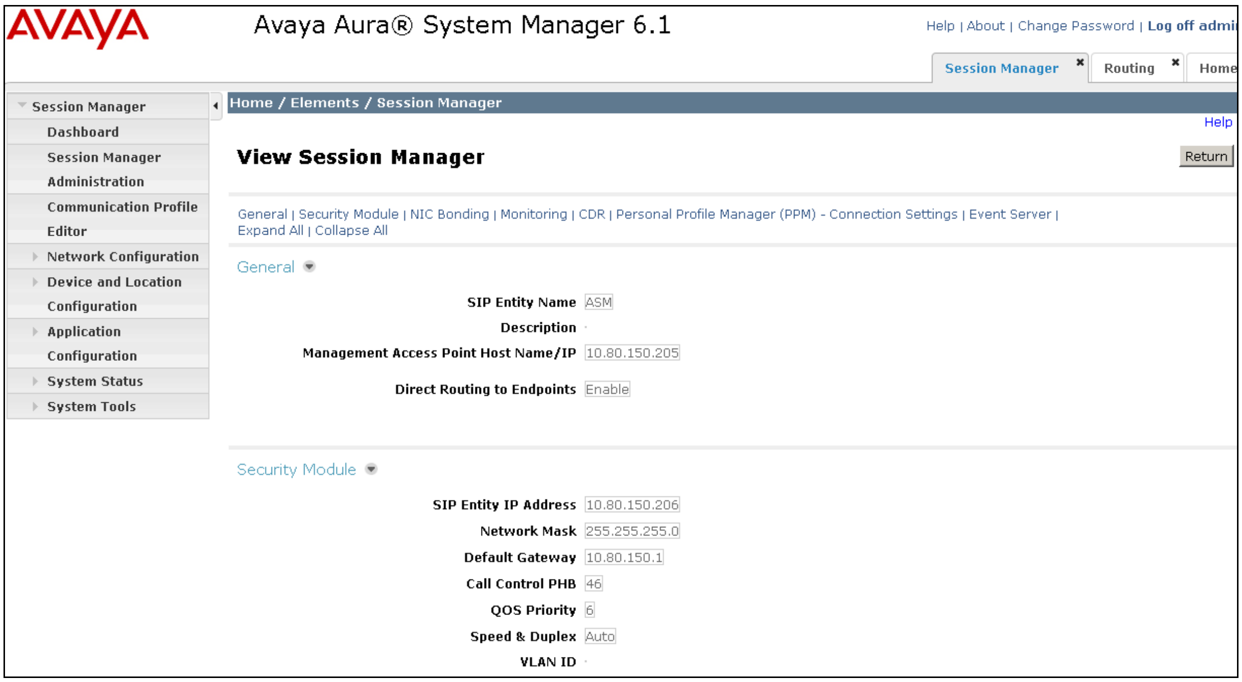
Communication Manager stations cover to Modular Messaging using a pilot extension **33000** in this reference configuration. Also, stations on Communication Manager may dial this number to retrieve messages or modify mailbox settings. To match dial pattern for the calls covered to Modular Messaging, repeat the Steps in **Section 5.10.1**. Routing Policy configured in **Section 5.9.4** was used to route the call to Modular Messaging.



Dial Pattern Details – Coverage to Modular Messaging

5.11. 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.



View Session Manager Page

6. Avaya Voice Portal

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

6.1. Background

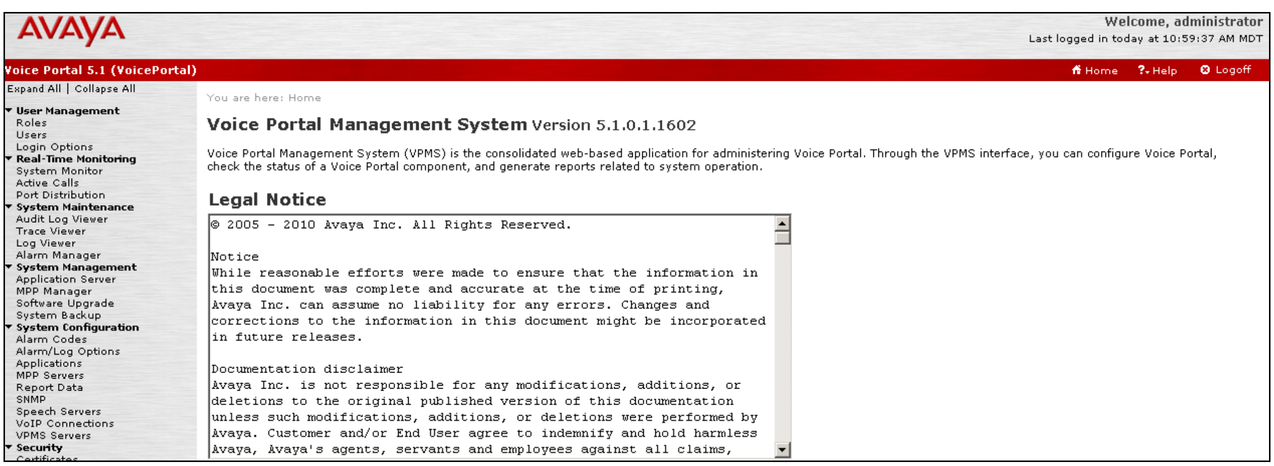
Voice Portal handles inbound calls according to the directives specified by Voice XML (VXML) and/or Call Control XML (CCXML) applications. The applications do not reside on Voice Portal, but rather on one or more separate application servers. References to these applications are administered on Voice Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Voice 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 match[4] is found, Voice 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 Voice Portal for administering a SIP connection to the Acme SBC.

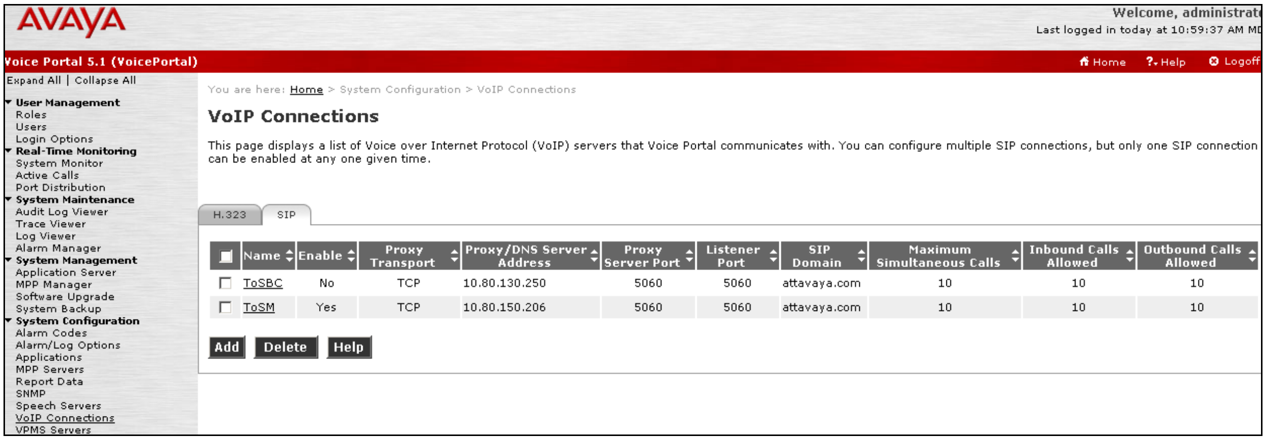
1. Launch a web browser, enter http://<IP address of the Avaya VPMS server>/ in the URL, and log in with the appropriate credentials.



VPMS Home Page

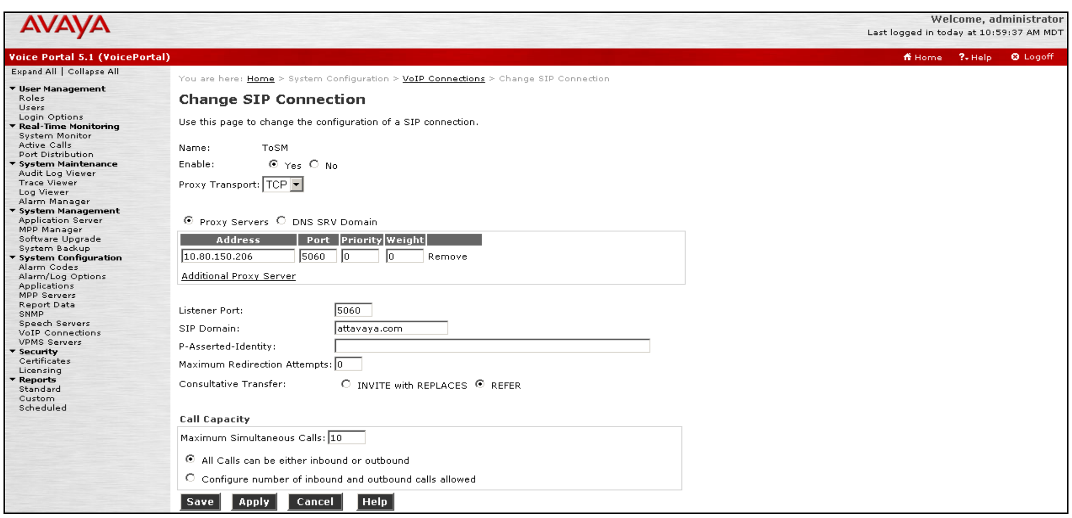
[4] One application reference may be configured with "inbound default" as the called number to handle all inbound calls that do not match any other application references.

2. In the left pane, navigate to **System Configuration→VoIP Connections**. On the **VoIP Connections** page, select the **SIP** tab and click **Add** to add a SIP trunk. The screen below shows the SIP trunks already configured on Voice Portal. Note that only **ONE** SIP trunk can be active at any given time on Voice Portal.



VoIP Connections Page

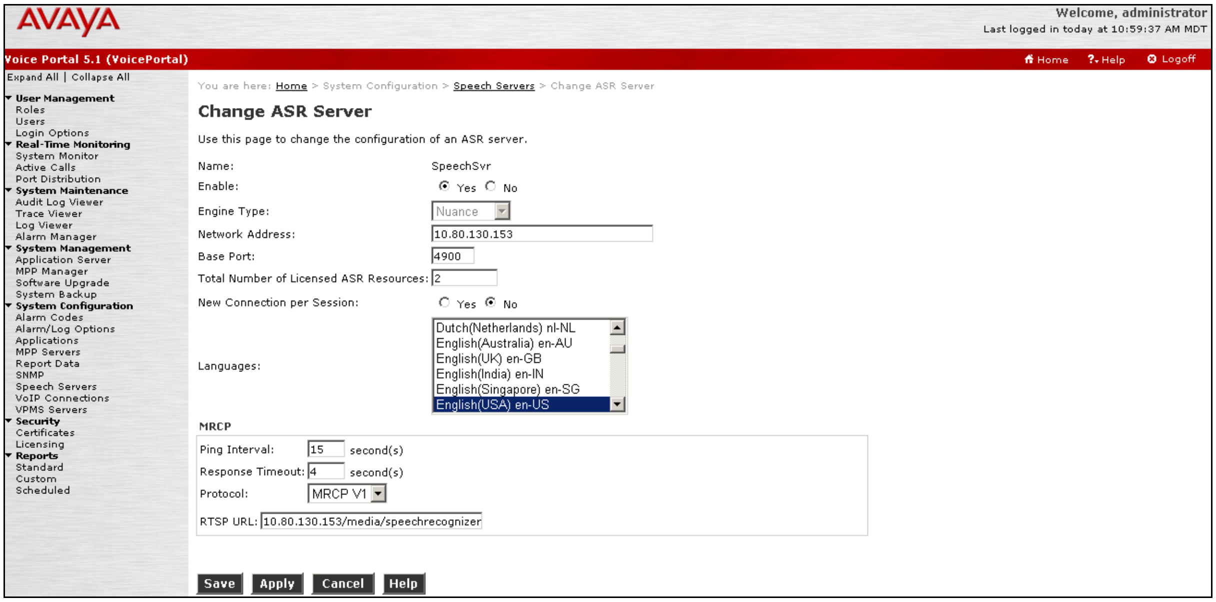
3. Click **ToSM** (SIP trunk already configured) and the following screen appears. Verify the following fields:
   - **Name** – Set to a descriptive name.
   - **Enable** – Set to **Yes**.
   - **Proxy Transport** – Set to **TCP**.
   - **Proxy Server Address** – Set to the IP address of the Session Manager signaling interface.
   - **Proxy Server Port** – Set to **5060**.
   - **SIP Domain** – Set to SIP domain configured in **Section 5.3**.
   - **Maximum Simultaneous Calls** – Set to a number in accordance with licensed capacity.
   - Set to the **All Calls can be either inbound or outbound** radio button.



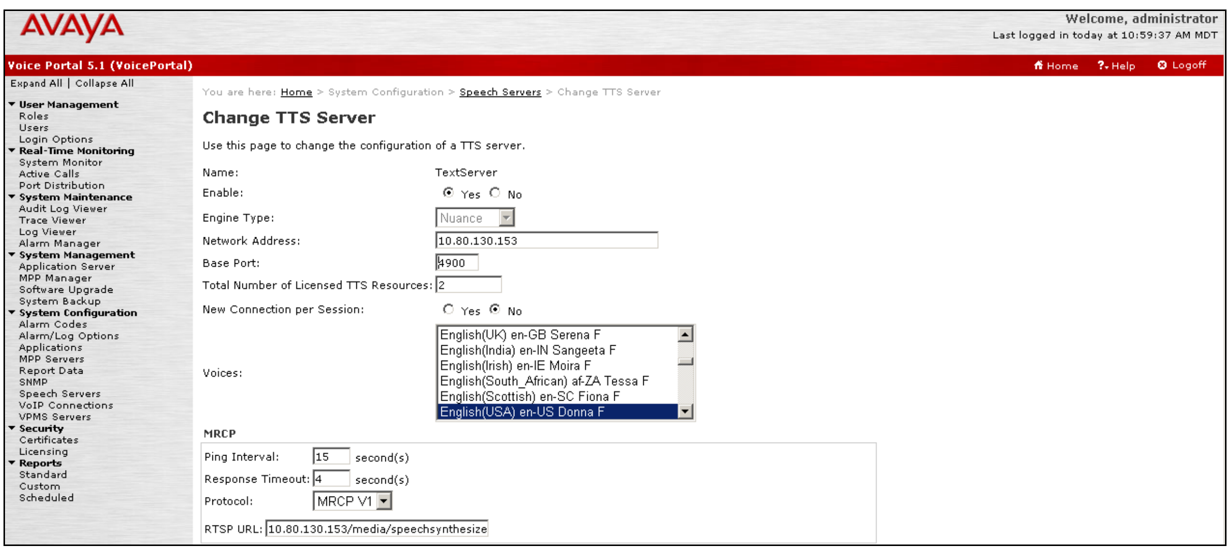
Change SIP Connection Page

6.3. Speech Servers

The configuration of the Speech Servers is beyond the scope of this document. To configure Speech Servers, navigate to **System Configuration→Speech Servers**. The following screens show the ASR and TTS server settings used in this reference configuration:



Change ASR Server Page



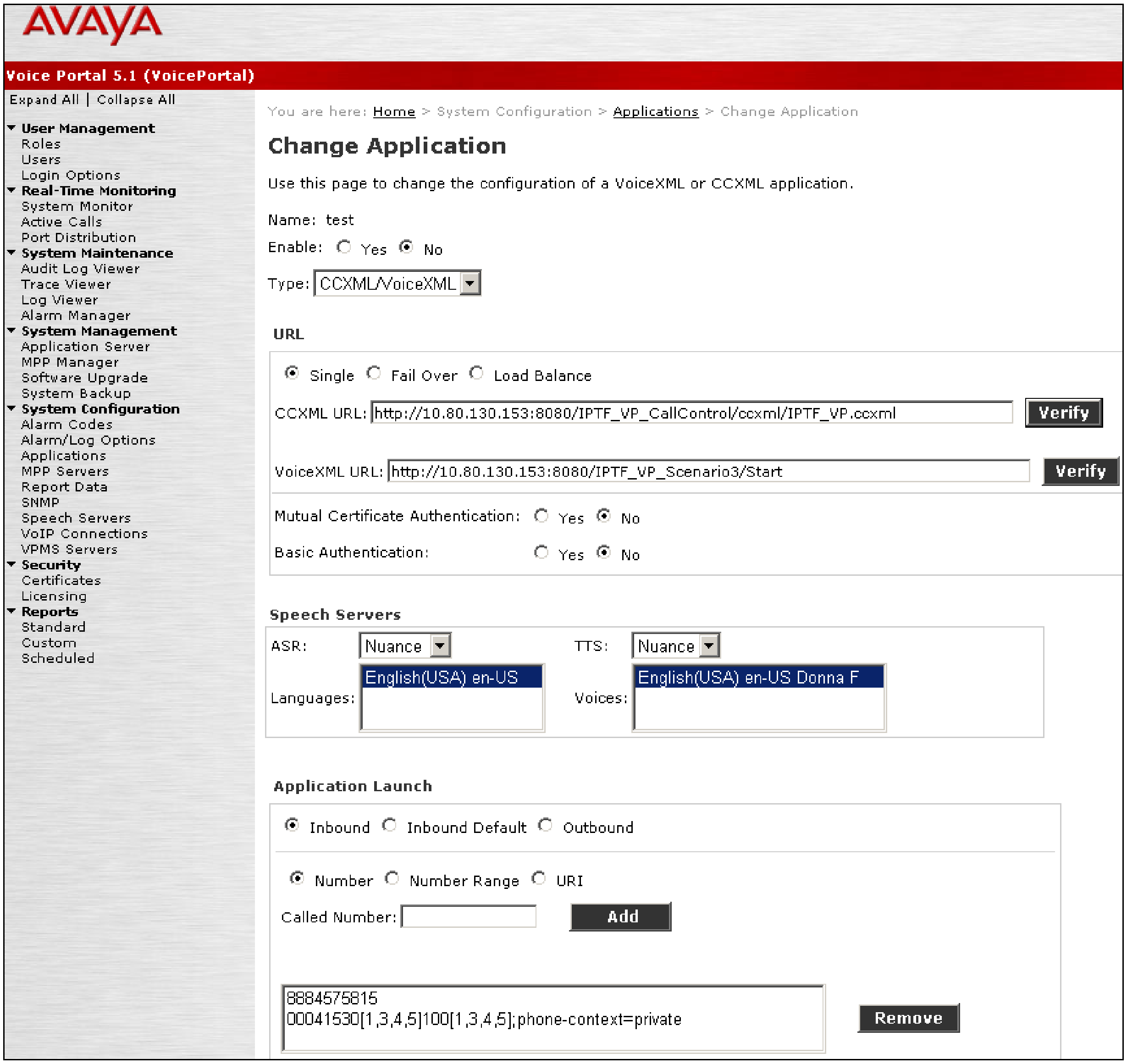
Change TTS Server Page

6.4. Application References

This section describes the steps on Voice 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. The screen below shows the application already configured on Voice Portal. Verify the following fields:
   - **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.

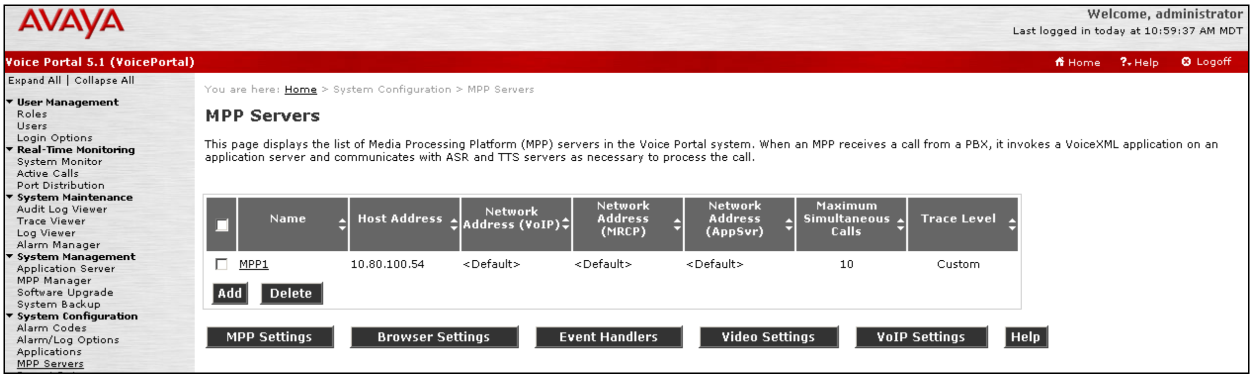


Change Application Page

2. Repeat above step/s to administer additional applications.

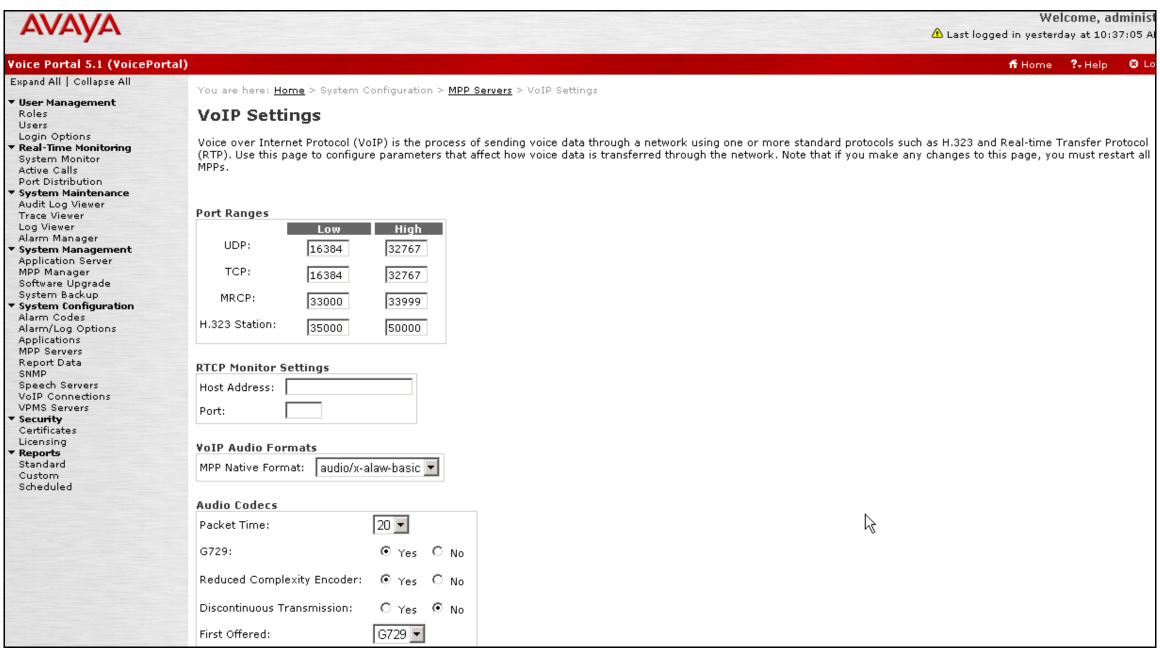
6.5. VOIP Settings

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



MPP Servers Page

Click **VoIP Settings** tab 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**.



VoIP Settings Page

6.6. Configuring RFC2833 Event Value Offered by Voice 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 Voice Portal, AT&T specifies the value 100 for the RFC2833 telephone-events that signal DTMF digits entered by the user.

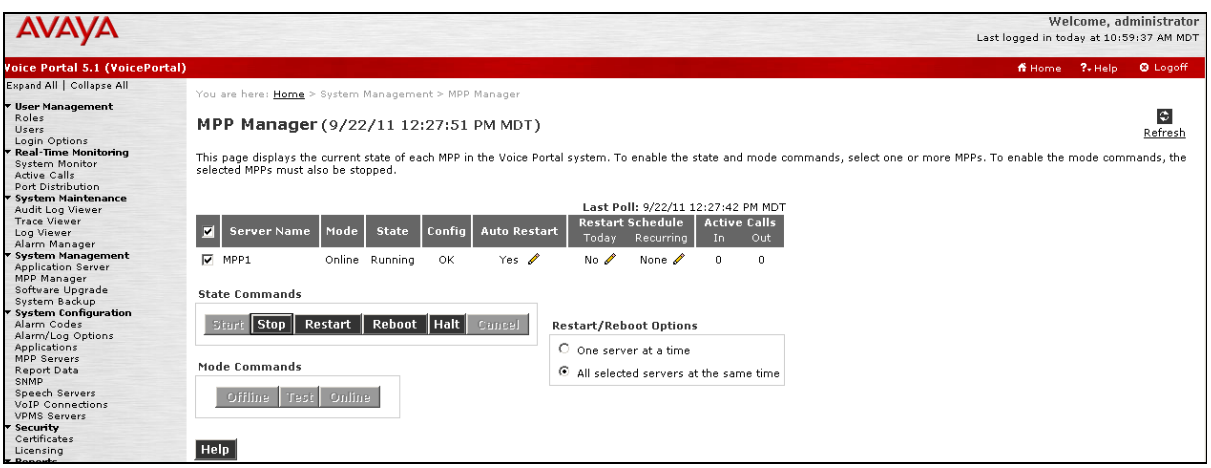
When Voice Portal answers, the SDP from Voice Portal matches this AT&T offered value. When Voice Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Voice Portal offers the SDP. By default, Voice Portal specifies the value 127 for the RFC2833 telephone-events. Optionally, the value that is offered by Voice Portal can be changed, and this section outlines the procedure that can be performed by an Avaya authorized representative.

- Access the Voice Portal via the command line interface and navigate to the /opt/Avaya/VoicePortal/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.



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  10
                                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  10
                                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.
- 4-digit extensions with a **Call Type** of **“ext**” beginning with the digit **“2**” – used for Vector Directory Numbers (VDN).
- 5-digit extensions with a **Call Type** of **“ext**” beginning with the digit **“3**” – Used for announcements.
- 5-digit extensions with a **Call Type** of “**ext**” beginning with the digit “**5**” – Used for local extensions for stations, agents and skills (hunt groups).

```
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
    2             4    ext
    3             5    ext
    5             5    ext
```

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 Voice 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 ct** command, where **ct** 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 it preferred codec but also supports **G.711MU** and **G.726A-32K**.

```
change ip-codec-set 2                                       Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A        n             2         20
 2: G.711MU       n             2        20
 3: G.726A-32K    n             2         20
```

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

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-1A05           10.80.130.206
```

Change Node-Names IP Form

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

```
change ip-network-region 2                                    Page   1 of  19
                               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 3** 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   3 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                                          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 3

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**.
   - **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 5                                         Page   1 of   1

 Group Number: 5              Group Type: sip
                        Transport Method: tcp


   Near-end Node Name: CLAN_1A05            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

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

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 "**Incoming**".
   - **Service Type** – Set to "**public-ntwrk**".
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1**.
   - **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group.

```
add trunk-group 5                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 5                   Group Type: sip           CDR Reports: y
  Group Name: ATT IPTF                   COR: 1        TN: 1        TAC: 105
   Direction: Incoming       Outgoing Display? n
 Dial Access? n                                              Night Service:

Service Type: public-ntwrk        Auth Code? n

                                                   Signaling Group: 5
                                                 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
                                                     Total
Ext Ext              Trk       CPN                    CPN
Len Code             Grp(s)    Prefix                 Len
                                                              Total Administered: 2
 4  2                 5                              4           Maximum Entries: 9999
 5  5                5                                5
```

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 – Modular Messaging coverage for Communication Manager extensions
- VDN 2010/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 2011/Vector 11/Hunt Group 11 – Route call to Agent with Skill 11
- VDN 2012/Vector 12/Hunt Group 12 – Route call to Agent with Skill 12
- VDN 2013/Vector 13/Hunt Group 13 – Route call to Agent with Skill 13

Followng VDN/Vectors were used for calls anchored on Voice 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 2031/Vector 31/Hunt Group 31 – Route call to Agent with Skill 11
- VDN 2032/Vector 32/Hunt Group 32 – Route call to Agent with Skill 12
- VDN 2033/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 Modular Messaging

Hunt group **2** is used in the reference configuration to verify the coverage to voicemail. The hunt group **2** is defined with the 5 digit Modular Messaging pilot number **33000**.

```
display hunt-group 2                                            Page   1 of  60
                                  HUNT GROUP
            Group Number: 2                                ACD? n
              Group Name: MM Voicemail                   Queue? n
         Group Extension: 33000                         Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

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)
     33000                    33000                     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        Number of Rings: 3
                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 Voice 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 53001                                        Page   1 of   2
                                AGENT LOGINID

                Login ID: 53001                                     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 53001                                     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:
```

Page 2

```
display hunt-group 11                                           Page   1 of   3
                                  HUNT GROUP

            Group Number: 11                            ACD? y
              Group Name: Skill-11                    Queue? y
         Group Extension: 53011                      Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                  MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

Skill (Hunt Group) Form – Page 1

```
display hunt-group 11                                           Page   2 of   3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none
    Supervisor Extension:

     Controlling Adjunct: none

Interruptible Aux Threshold: none
                                 Redirect on No Answer (rings):
                                               Redirect to VDN:
                    Forced Entry of Stroke Counts or Call Work Codes? n
```

Skill (Hunt Group) Form – Page 2

```
display vdn 2010                                          Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2010
                                  Name: To SelectSkill
                           Destination: Vector Number        10

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


                            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 33002    for none
03 goto vector  11   @step 2 if digits          =      1
04 goto vector  12   @step 2 if digits          =      2
05 goto vector  13   @step 2 if digits          =      3
06
```

RouteToSkill Vector[5]

```
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 33003
03 queue-to     skill 11   pri m
04 announcement 33006
05 goto step    3             if unconditionally
06
```

Skill-11 Vector

[5] This vector was used for the call flow scenario where Voice 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 Vector[6]

8. Avaya Modular Messaging

In this sample configuration, Avaya Modular Messaging is provisioned for Multi-Site mode. Multi-Site mode allows Avaya Modular Messaging to server subscribers in multiple locations. The administration for Modular Messaging is beyond the scope of these Application Notes. Refer to **[10]**, [**11**] and **[12]** for further details.

[6] This vector was used for the call flow scenario where Voice 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 [13] 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., Voice Portal, Communication Manager, etc., reside to the AT&T IP Toll Free service.

```
local-policy
      from-address
                               *
      to-address
                               *
      source-realm
                               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 Voice Portal via Session Manager.

```
local-policy
      from-address
                               *
      to-address
                               *
      source-realm
                               ATT
      description
```

```
      activate-time                  N/A
      deactivate-time                N/A
      state                          enabled
      policy-priority                none
      last-modified-by               admin@console
      last-modified-date             2011-08-12 10:25:23
      policy-attribute
            next-hop                       10.80.150.206
            realm                          Enterprise
            action                         none
            terminate-recursion            disabled
            carrier
            start-time                     0000
            end-time                       2400
            days-of-week                   U-S
            cost                           0
            app-protocol                   SIP
            state                          enabled
            methods
            media-profiles
```

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

```
local-policy
      from-address
                                     *
      to-address
                                     20 50
      source-realm
                                     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                          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
      dnsalg-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
                                      s1p0:0
      mm-in-realm                     enabled
      mm-in-network                   enabled
      mm-same-ip                      enabled
      mm-in-system                    enabled
      bw-cac-non-mm                   disabled
      msm-release                     disabled
      generate-UDP-checksum           disabled
      max-bandwidth                   0
      fallback-bandwidth              0
      max-priority-bandwidth          0
      max-latency                     0
      max-jitter                      0
      max-packet-loss                 0
      observ-window-size              0
      parent-realm
      dns-realm
      media-policy
      in-translationid
      out-translationid
      in-manipulationid               RemoveUPDATE
      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 **Enterprise** below represents the internal network on which the Avaya elements reside.

```
realm-config
      identifier                    Enterprise
      description
      addr-prefix                   0.0.0.0
      network-interfaces            s0p0:0
      mm-in-realm                   enabled
      mm-in-network                 enabled
      mm-same-ip                    enabled
      mm-in-system                  enabled
      bw-cac-non-mm                 disabled
      msm-release                   disabled
      generate-UDP-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                        10.80.150.206
     ip-address                      10.80.150.206
     port                            5060
     state                           enabled
     app-protocol                    SIP
     app-type
     transport-method                UDP+TCP
```

```
realm-id                        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=0
      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                 Enterprise
      egress-realm-id               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                           enabled
        realm-id                        ATT
        description
        sip-port
                address                         192.168.62.50
                port                            5060
                transport-protocol              UDP
                tls-profile
                allow-anonymous                 all
                ims-aka-profile
        carriers
        trans-expire                    0
        invite-expire                   0
        max-redirect-contacts           0
        proxy-mode
        redirect-action
        contact-mode                    none
        nat-traversal                   none
        nat-interval                    30
        tcp-nat-interval                90
        registration-caching            disabled
        min-reg-expire                  300
        registration-interval           3600
        route-to-registrar              disabled
        secured-network                 disabled
        teluri-scheme                   disabled
        uri-fqdn-domain
        trust-mode                      all
        max-nat-interval                3600
        nat-int-increment               10
        nat-test-increment              30
        sip-dynamic-hnt                 disabled
        stop-recurse                    401,407
        port-map-start                  0
        port-map-end                    0
        in-manipulationid
        out-manipulationid
        manipulation-string
        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                       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 Voice Portal.

```
sip-manipulation
      name                          RemoveUPDATE
      description                   Strip Update from Allow list
      header-rule
            name                          EditAllow
            header-name                   Allow
            action                        manipulate
            comparison-type               pattern-rule
            match-value
            msg-type                      any
            new-value
            methods
            element-rule
                  name                          StripUPDATE
                  parameter-name
                  type                          header-value
```

```
                action                          find-replace-all
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value                     (,\s*UPDATE|UPDATE\s*,)
                new-value
        last-modified-by                admin@console
        last-modified-date              2011-08-22 19:25:08
```

ANNOTATION: The SIP manipulation below adds a **Diversion** header in SIP messages from the Voice Portal to AT&T Flex Reach service as **Diversion** header is not generated by Voice 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@10.80.100.54"
        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                        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

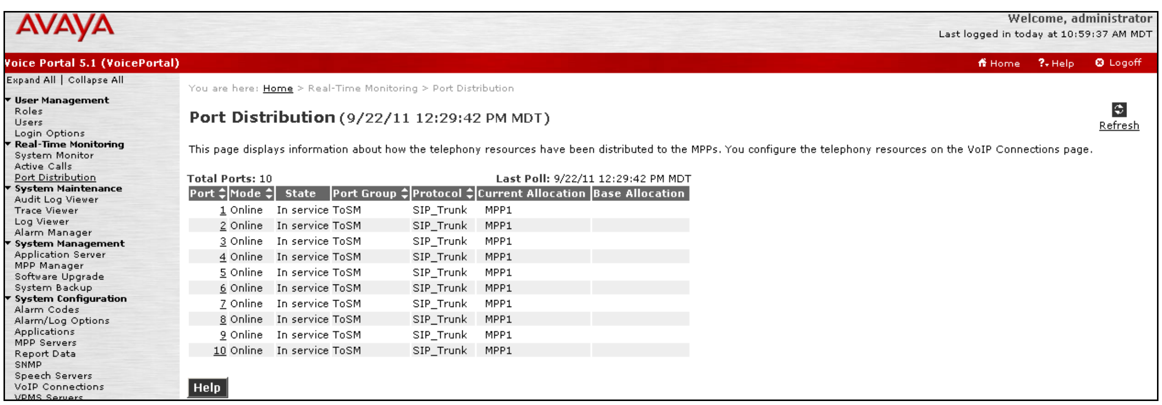
The following steps may be used to verify the configuration:

- Place an inbound call to Voice Portal application, and verify that two-way talkpath exists. Interact with the Voice Portal prompts and verify that the call remains stable for several minutes and disconnect properly.
- Place an inbound call to Voice 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 Voice 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 Voice 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.

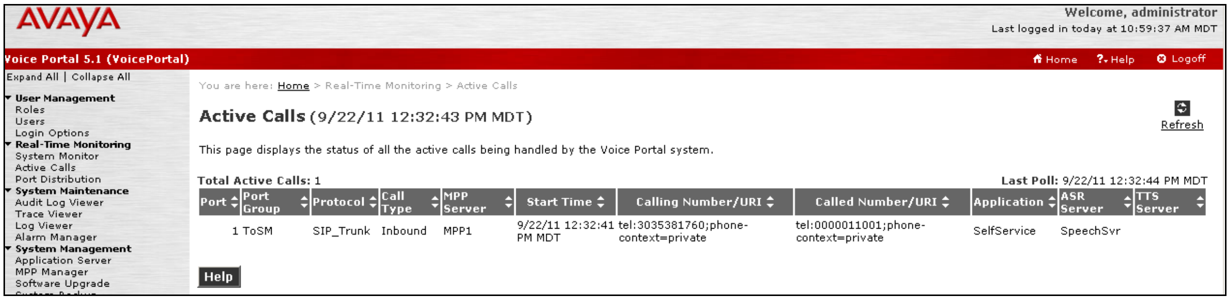
10.2. Avaya Voice Portal

The following commands are issued from the System Manager console.

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



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



10.3. Troubleshooting Tools

The logging and reporting functions within the Avaya VPMS web interface may be used to examine the details of Voice 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 **traceSES** function within the SES may be used to capture SIP traces between SES 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 Voice 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 Voice Portal the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection.

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 Voice Portal*, June 2010
[2] *Implementing Voice Portal on a single server*, June 2010
[3] *Administering Voice Portal*, January 2011
[4] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473, April 2011.
[5] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, May 2011.
[6] *Administering Avaya Aura® Communication Manager*, Document Number 03-300509, May 2009
[7] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.0, 555-245-205, May 2009
[8] *Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, Document Number 07-600780, April 2009
[9] *Avaya Aura® Call Center 5.2 Automatic Call Distribution Reference*, Document Number 07-602568, April 2009
[10] *Modular Messaging Multi-Site Guide Release 5.1*, June 2009
[11] *Modular Messaging Messaging Application Server (MAS) Administration Guide,* July 2011
[12] *Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades*, June 2009

Acme Packet Support (login required):
[13] http://support.acmepacket.com

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

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