Avaya Solution & Interoperability Test Lab

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

Abstract

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

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura® Experience Portal is a speech-enabled Interactive Voice Response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner. Avaya Aura® Communication Manager is a telephony application server.

An Acme Packet Net-Net is the point of connection between Avaya Aura® Experience Portal and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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

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1. Introduction

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

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Avaya Aura® Experience Portal is a speech-enabled Interactive Voice Response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner. Avaya Aura® Communication Manager is a telephony application server.

An Acme Packet Net-Net is the point of connection between Avaya Aura® Experience Portal and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

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

2. General Test Approach and Test Results

The test environment consisted of:

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

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

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

2.1. Interoperability Compliance Testing

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

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

- SIP trunking
- Passing of DTMF events and their recognition by navigating automated voice menus
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer
- Legacy Transfer Connect

- Alternate Destination Routing

2.2. Known Limitations/Test Results

1. AT&T IP Transfer Connect option of the AT&T IP Toll Free service was not verified with Experience Portal 6.0 and hence not supported.
2. Experience Portal 6.0 does not send DTMF digits using RFC2833 so the Legacy Transfer Connect using Experience Portal application could not be tested. Legacy Transfer Connect using the agent/telephone was successfully tested.
3. G.726 codec is not supported by Experience Portal 6.0.
4. If Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Communication Manager, then Communication Manager selects a codec according to the priority order specified in the Communication Manager codec set, not the priority order specified in the SDP offer. For example, if the AT&T IP Toll Free service offers G.711, G.729A, and G.729B in that order, but the Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Communication Manager selects G.729A, not G.711. The practical resolution is to provision the Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.
5. Although Experience Portal release 6.0 and Communication Manager 6.0.1 support the possibility of using SIP phones as valid telephone extensions, SIP telephones were not tested as part of the configuration used to validate this solution.
6. For an outcall to an 8YY number from Experience Portal, the Experience Portal application needs to add a Diversion Header otherwise AT&T network will send a 403 Forbidden message back and the call will fail. This diversion header can also be added on the Acme SBC as shown in **Section 9**, and that was the way it was implemented in this reference configuration.
7. Experience Portal 6.0 does not recognize maxptime as packet size and always defaults to a ptime of 20 msecs. A workaround was implemented on Acme SBC to convert the **maxptime** attribute to **ptime** as illustrated in **Section 8**.
8. A slight delay in ringback was observed on Calling Party telephone when the call is transferred from Experience Portal to an agent on Communication Manager.

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 Aura® Experience Portal provides Interactive Voice Response services to inbound callers. Avaya Aura® Experience Portal consists of one or more Media Processing Platform (MPP) servers and an Experience Portal Management System (EPM) server. Single server was used for MPP and EPM 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 S8800 Server.
- The Avaya G650 Media Gateway provides the physical interfaces and resources for enterprise voice communications. This solution is extensible to other Avaya Media Gateways.
- Avaya phones are represented with Avaya 96xx Series IP Telephones running H.323 software. Additionally Avaya one-X® Agent and Analog and Digital phones were also used.
- The Acme Packet Net-Net Session Director (SD) 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 Aura® Experience Portal which are referenced in Avaya Aura® Experience Portal.
- The Speech Server consists of Nuance Recognizer and Nuance Vocalizer. Avaya Aura® Experience Portal uses the Speech Server for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) capabilities.
- Avaya Aura® Messaging provides the corporate voice messaging capabilities in this reference configuration. The provisioning of Avaya Aura® Messaging is beyond the scope of this document.

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[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 reference configuration. Communication Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme SBC. In the reference configuration, Communication Manager uses SIP over TCP to communicate with the Acme Packet SBC.

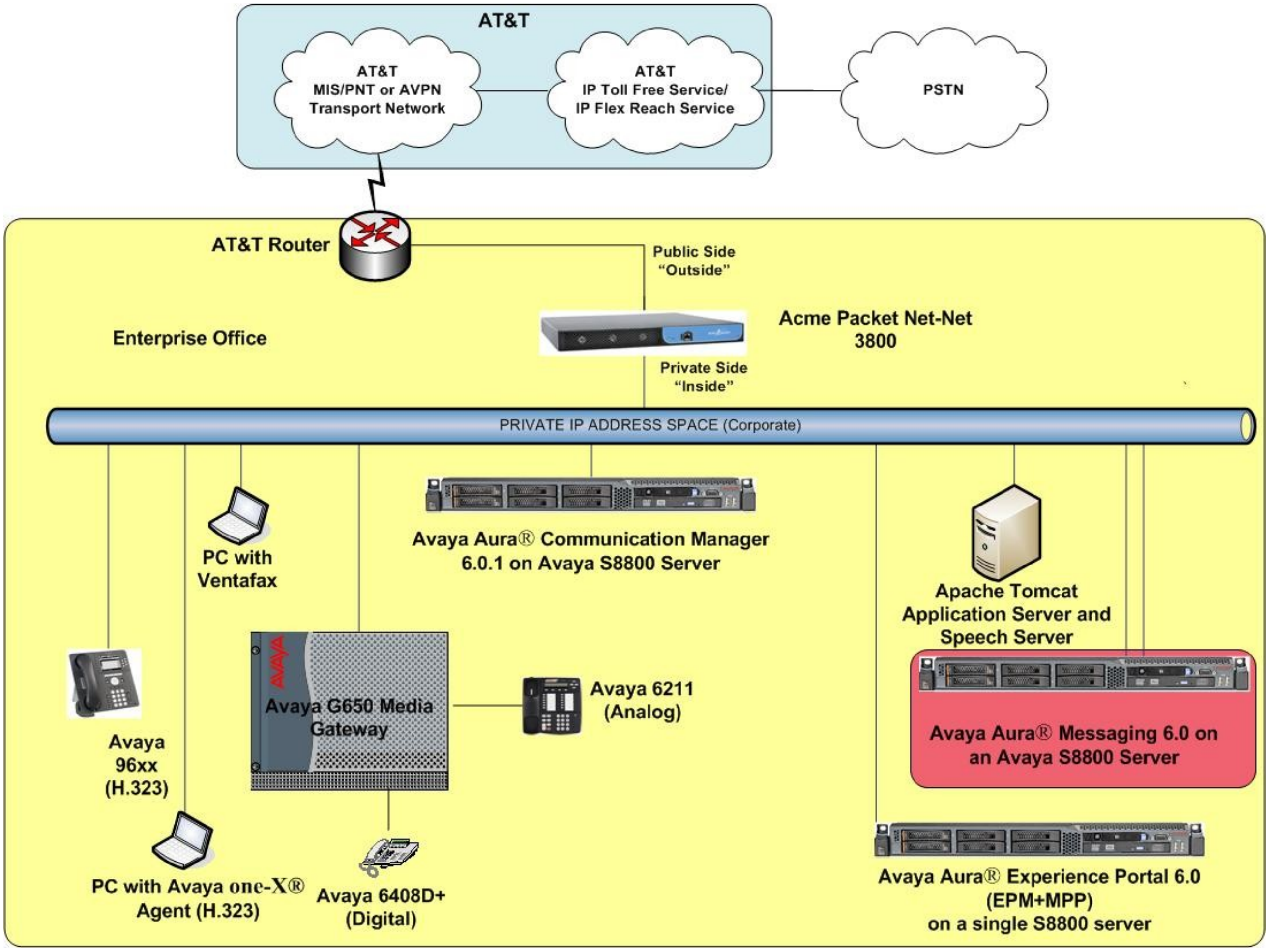


Figure 1: Reference Configuration

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 Aura® Experience Portal	
VPMS/MPP Servers IP Address	10.80.130.220
Automatic Speech Recognition and Text to Speech server IP Address	10.80.130.153
Avaya Aura® Communication Manager	
C-LAN IP Address	10.8.130.102
Vector Directory Number (VDN) Extensions	666-20xx
Skill (Hunt Group) Extensions	666-40xx
Agent Extensions	666-30xx
Phone Extensions	666-50xx
Announcement Extensions	666-10xx
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
Digits Passed in SIP "To" Header to Avaya Aura® Experience Portal	xxxxxx5815

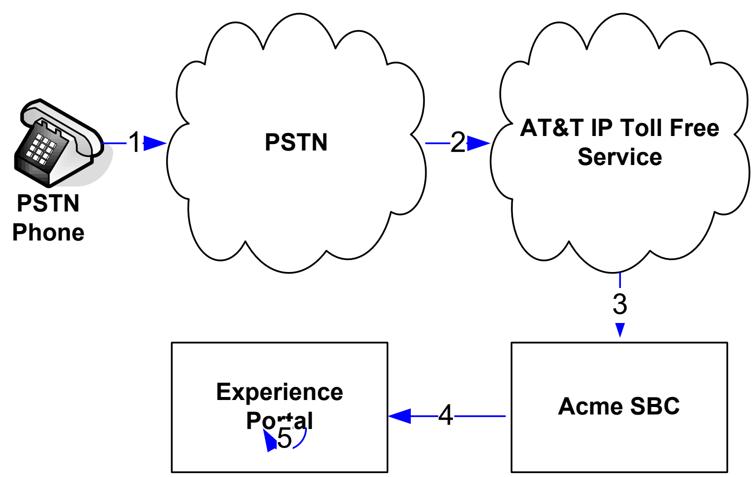
Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

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

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

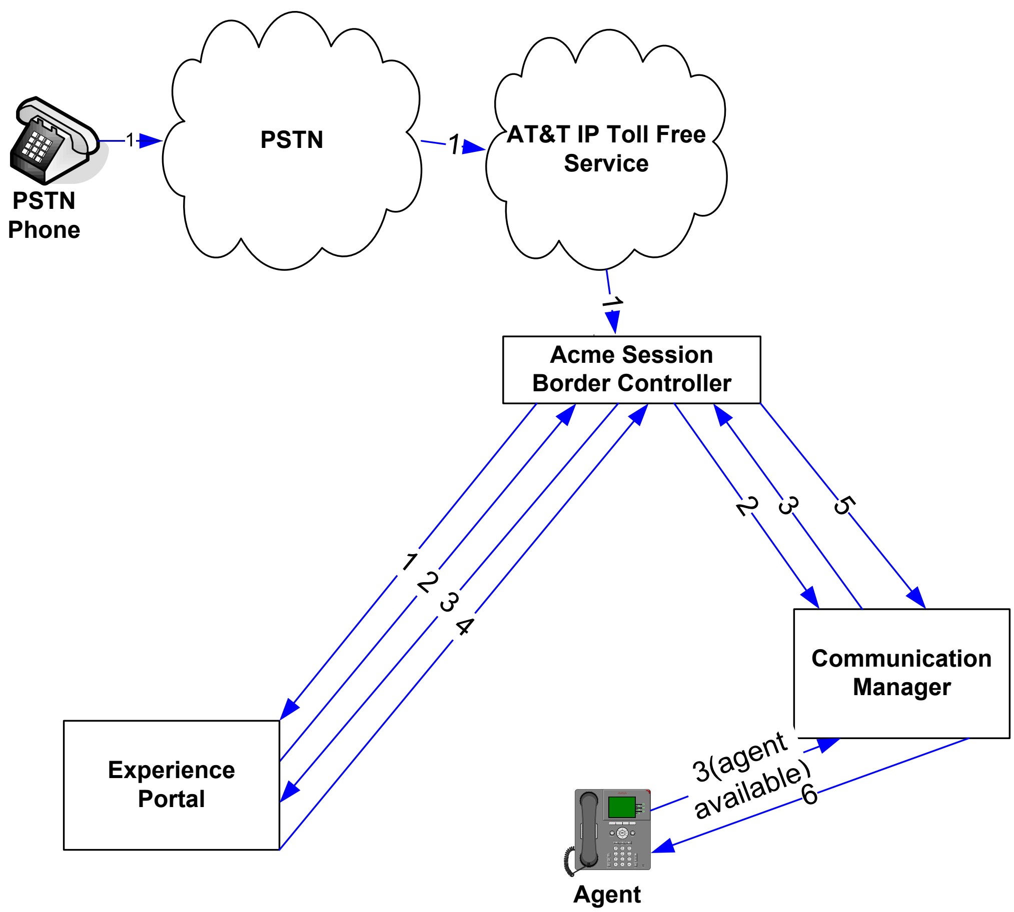
1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to the Acme SBC.
4. Acme SBC performs any necessary SIP header modifications, and routes the call to Experience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



Inbound Call Handled Entirely by Avaya Aura® Experience Portal

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

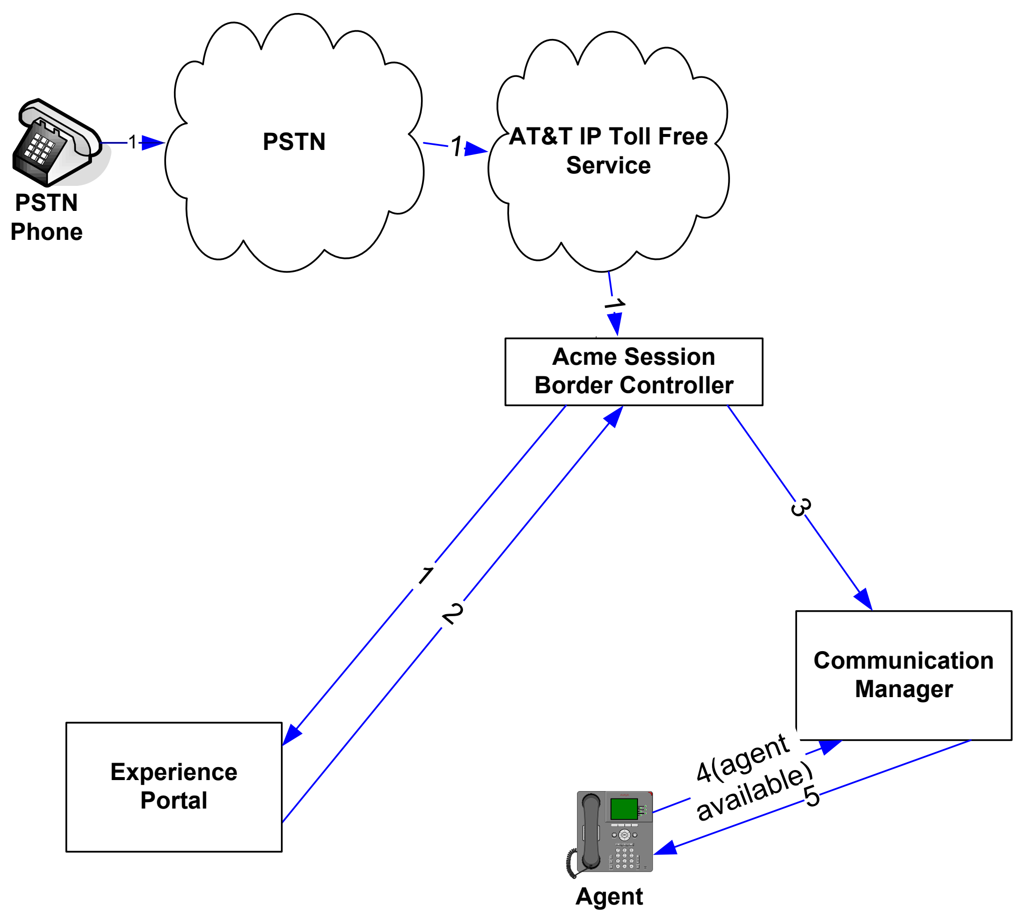
1. Same as the first four steps from the first call scenario.
2. In this scenario, the application is not sufficient to meet the caller's requests, and thus the call needs to be transferred to a Communication Manager agent. Experience Portal then puts the inbound call on hold and places a call to vector/skill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



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

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

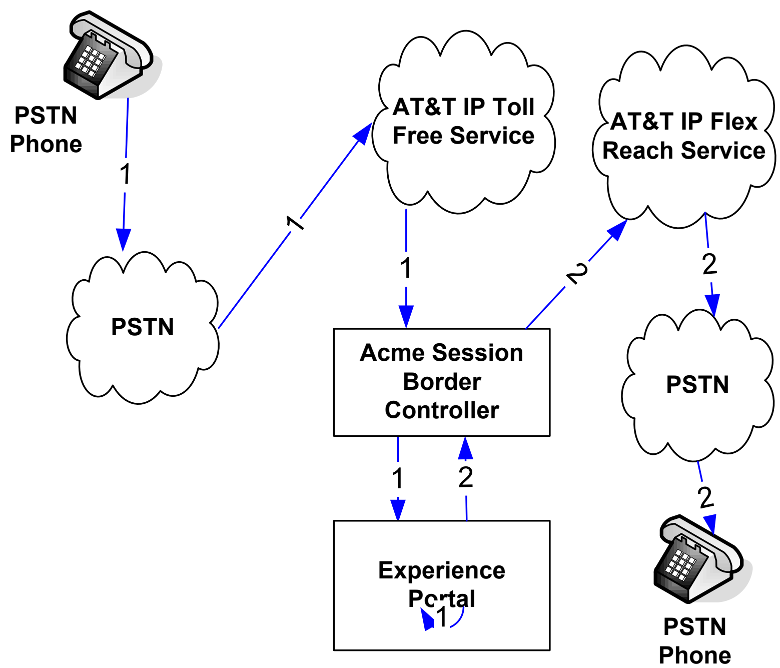
1. Same as the first four steps from the first call scenario.
2. In this scenario, the application on Experience Portal is not sufficient to meet the caller's needs or requests, and thus the call needs to be transferred to an agent/skill on Communication Manager. Experience Portal instructs the Acme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



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

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

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



Inbound Call forwarded by Experience Portal to another PSTN number

4. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura® Experience Portal 6.0
Experience Portal Management (EPM)	6.0.0.0.3306
Media Processing Platform (MPP)	6.0.0.0.3401
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 with Service Pack 5 (R016x.00.1.510.1)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW054
TN799DP Control-LAN (C-LAN)	HW00 FW040
TN2602AP IP Media Processor (MedPro)	HW02 FW061
TN2501AP VAL-ANNOUNCEMENT	HW02 FW018
Avaya 9650 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.110b
Avaya 9611 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release S6.0.0
Avaya 9620C IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.110b
Avaya one-X® Agent	Release 2.5
Apache Tomcat Application Server	6.0.33
Nuance Recognizer	9.0
Nuance Recognizer English en-US Language Pack	9.0
Nuance Vocalizer	5.0.5
Nuance Vocalizer American English en-US Donna	5.0.2
Nuance MediaServer	5.0.5
Acme Packet Net-Net Session Director 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Toll Free Service	VNI 22

Table 2: Equipment and Software Versions

5. Avaya Aura® Experience Portal

These Application Notes assume that the necessary Experience Portal licenses have been installed and basic Experience Portal administration has already been performed. Consult [1], **Error! Reference source not found.** and **Error! Reference source not found.** for further details if necessary.

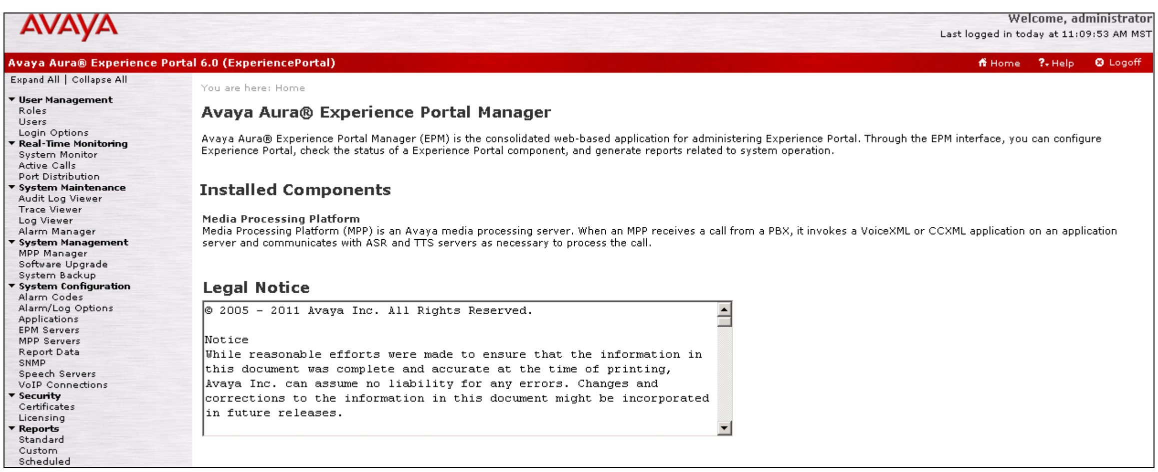
5.1. Background

Experience 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 Experience Portal, but rather on one or more separate application servers. References to these applications are administered on Experience Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Experience Portal, the called party number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match[3] is found, Experience Portal informs the caller that the call can not be handled, and disconnects the call. For this reference configuration, VXML and CCXML applications were developed specifically to exercise SIP call flow scenarios expected to occur with the AT&T IP Toll Free service. In production, enterprises can develop their own VXML and/or CCXML applications to meet their specific customer self-service needs, or consult Avaya Professional Services and/or authorized Avaya Business Partners. The development and deployment of VXML and CCXML applications is beyond the scope of these Application Notes.

5.2. VoIP Connection

This section describes the steps on Experience Portal for administering a SIP connection to the Acme SBC.

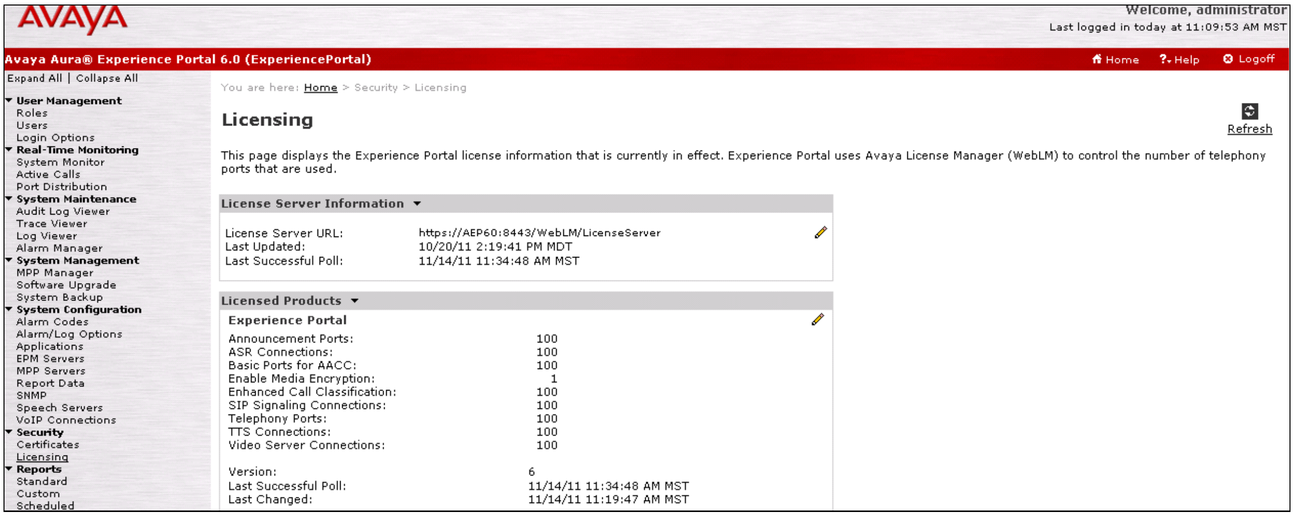
1. Launch a web browser, enter http://<IP address of the Avaya EPM server>/ in the URL, log in with the appropriate credentials and the following screen is displayed.



[3] 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.

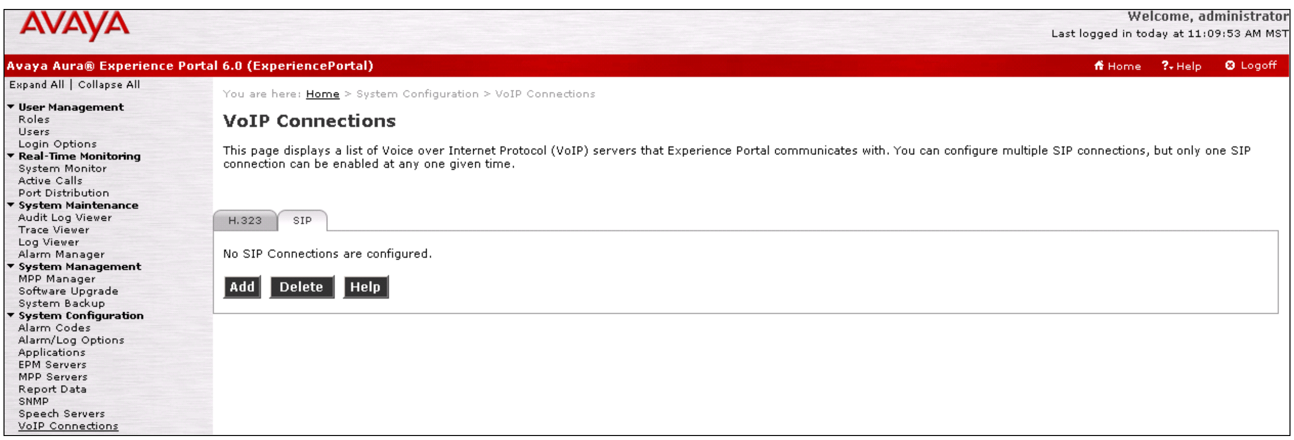
Experience Portal Home Page

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



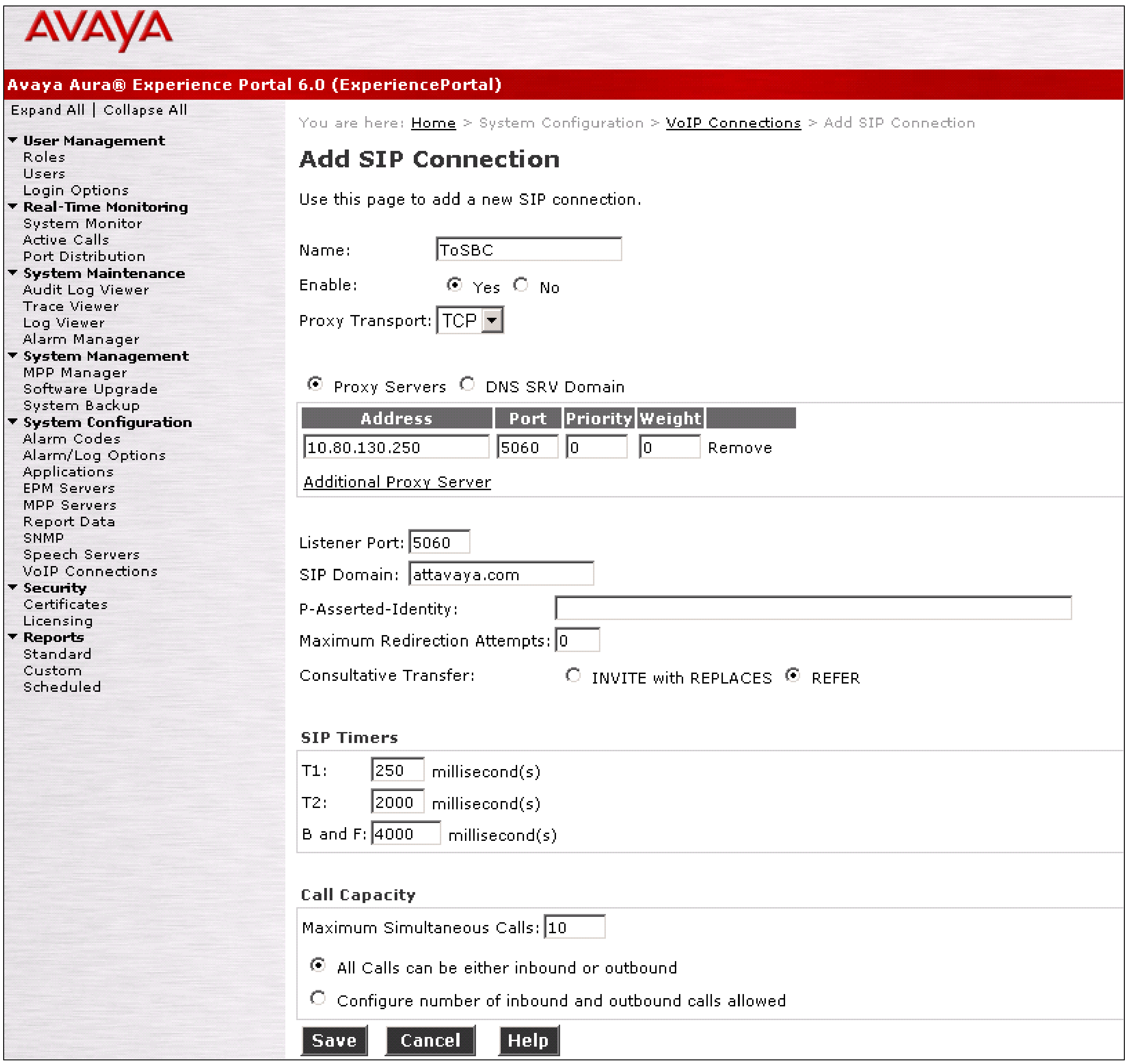
Experience Portal Licensing Page

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



VoIP Connections Page

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

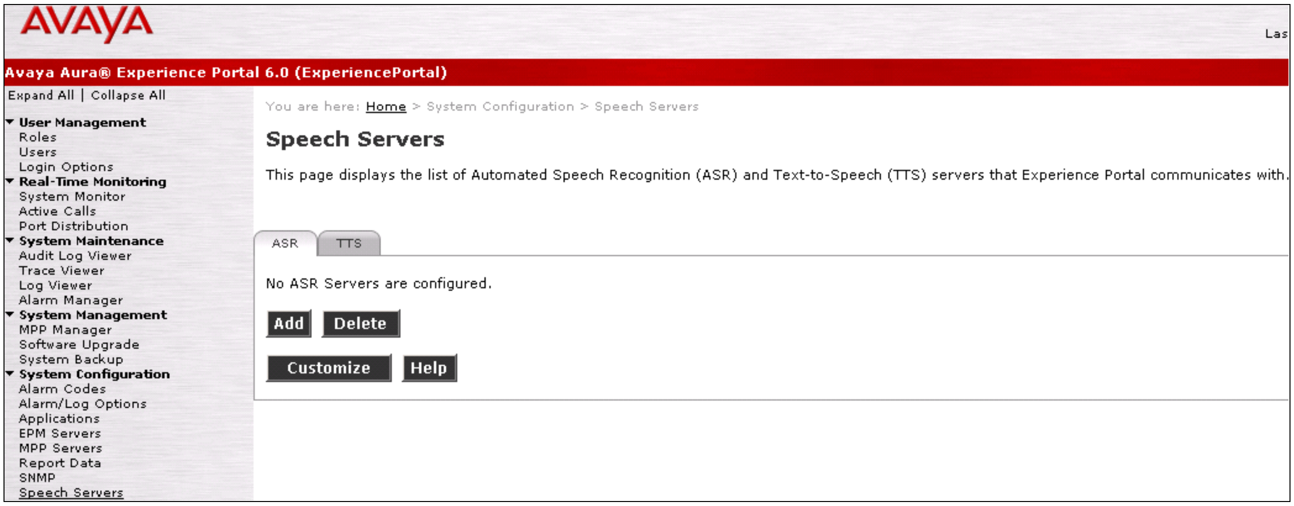


Add SIP Connection Page

5.3. Speech Servers

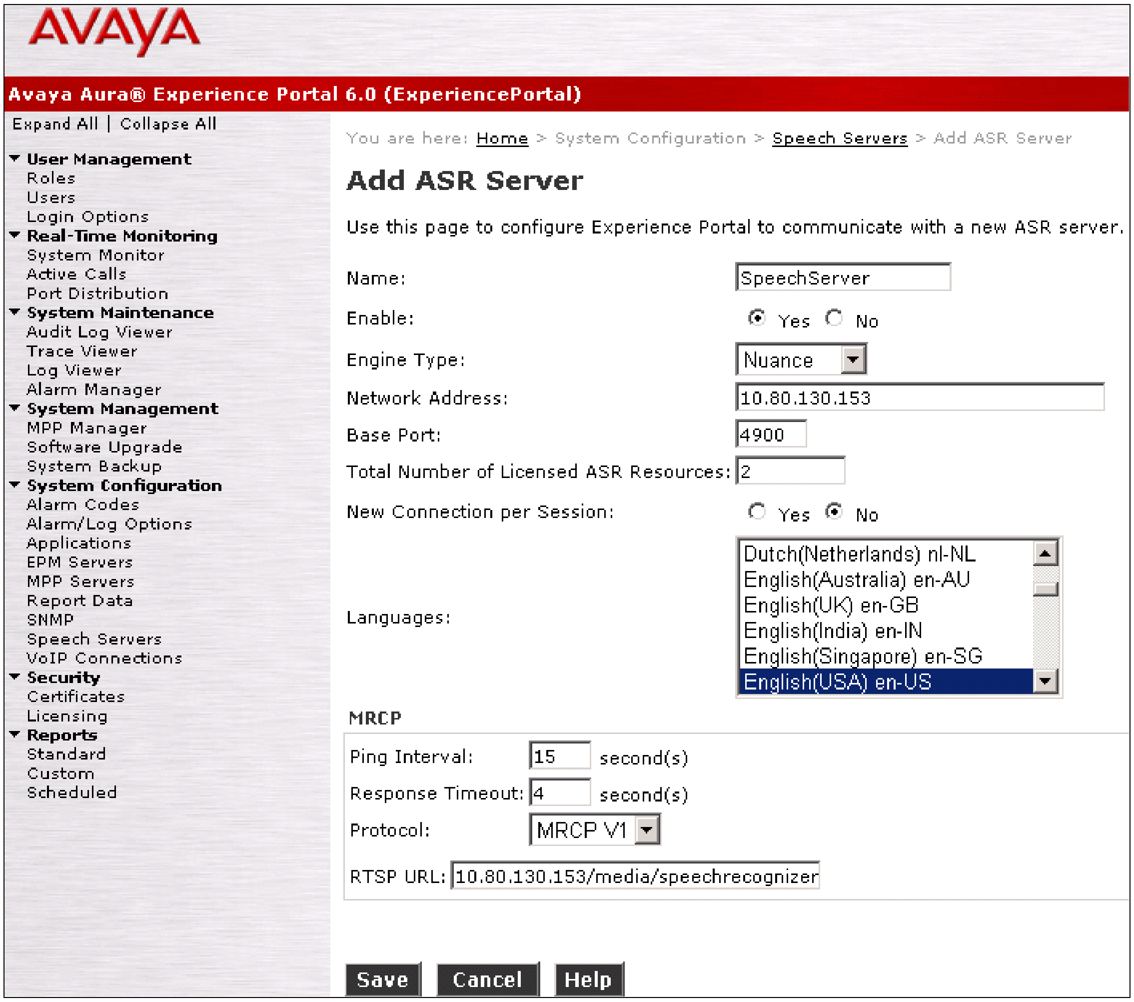
The installation and administration of the Speech Servers is beyond the scope of this document.

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



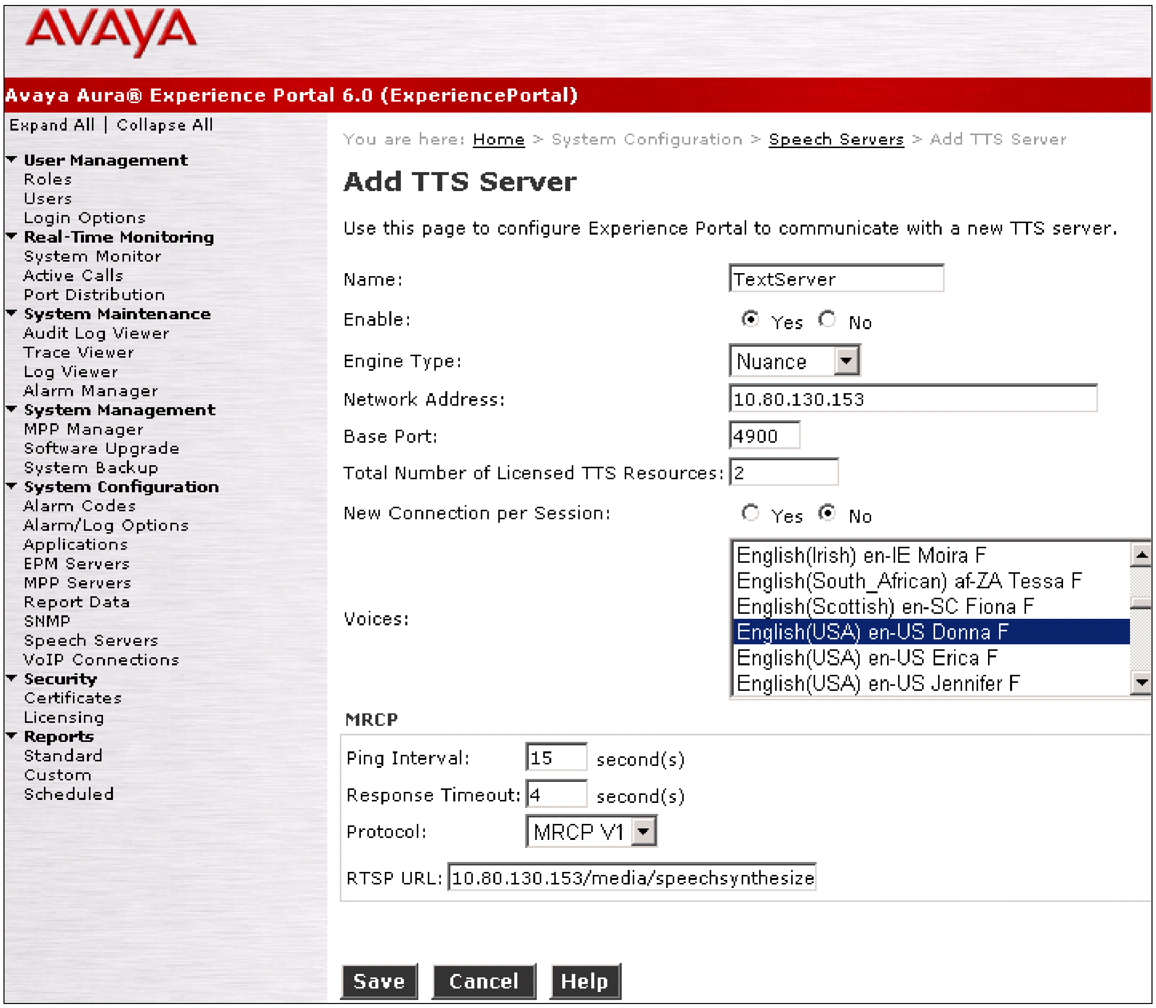
Speech Server Page

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



Add ASR Server Page

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



Add TTS Server Page

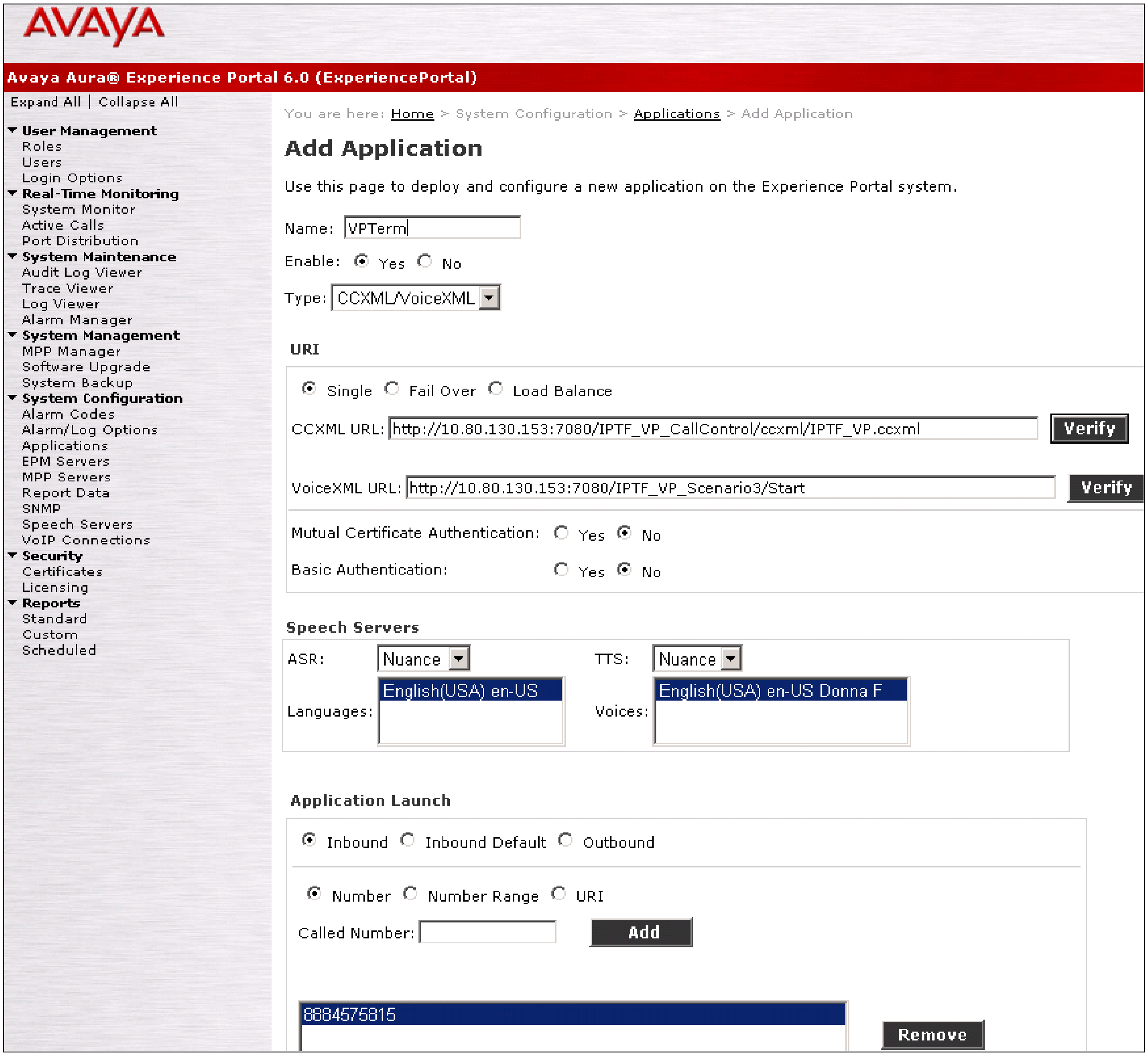
5.4. Application References

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

1. In the left pane, navigate to **System Configuration→Applications**. On the **Applications** page [not shown], click on **Add** to add an application and configure as follows:
   - **Name** – Set to a descriptive name.
   - **Enable** – Set to **Yes**.
   - **MIME Type** – Set **CCXML/VoiceXML** for the application used in this reference configuration.
   - **VoiceXML** and/or **CCXML URL** – Set to the URL(s) to access the VXML and/or CCXML application(s) on the application server.
   - **Speech Servers ASR** and **TTS** – Set to **Nuance**.
   - **Language** 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.

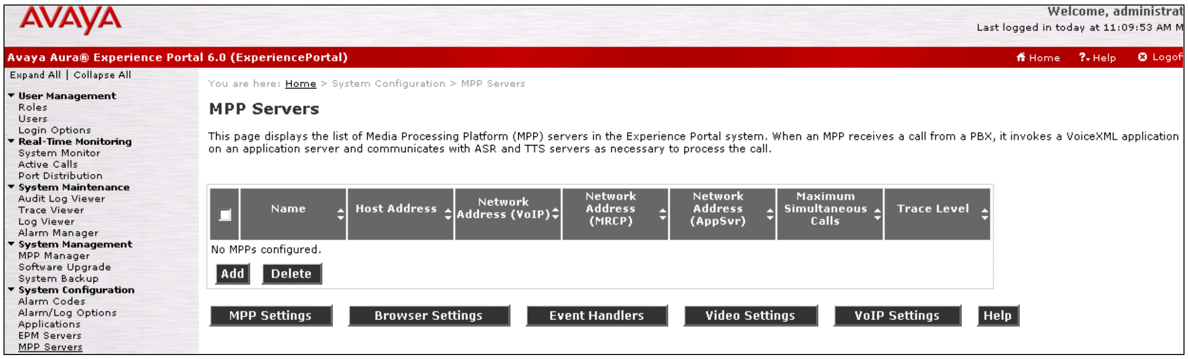


Add Application Page

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

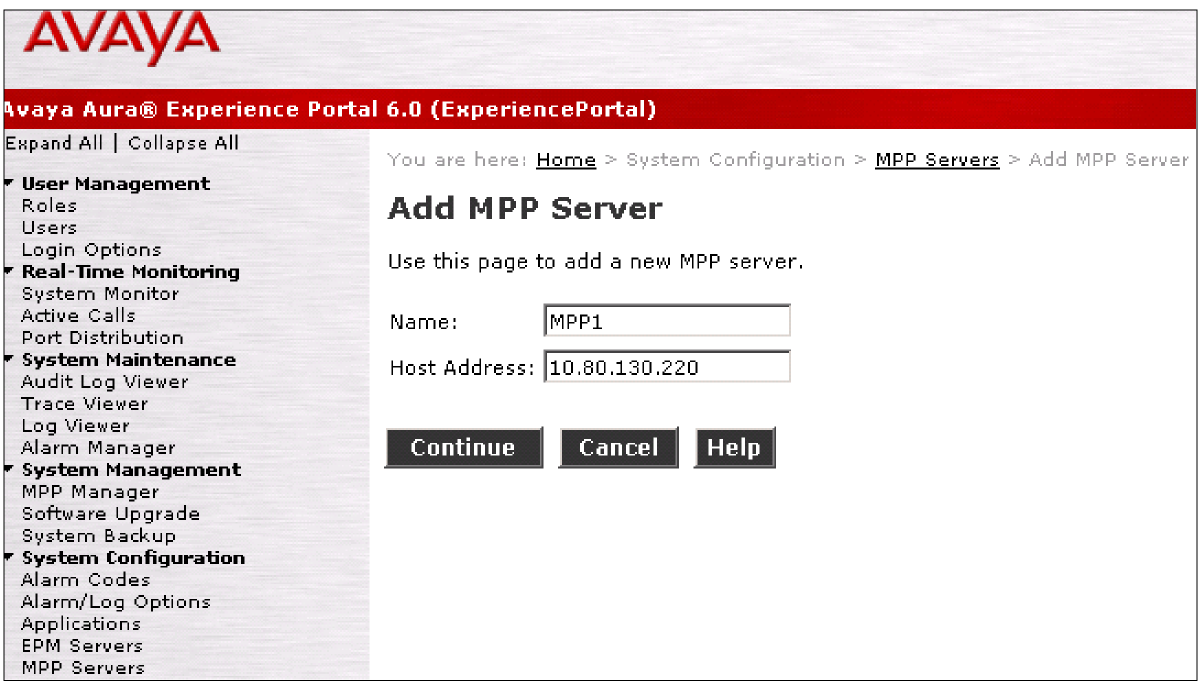
5.5. Add MPP Server

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



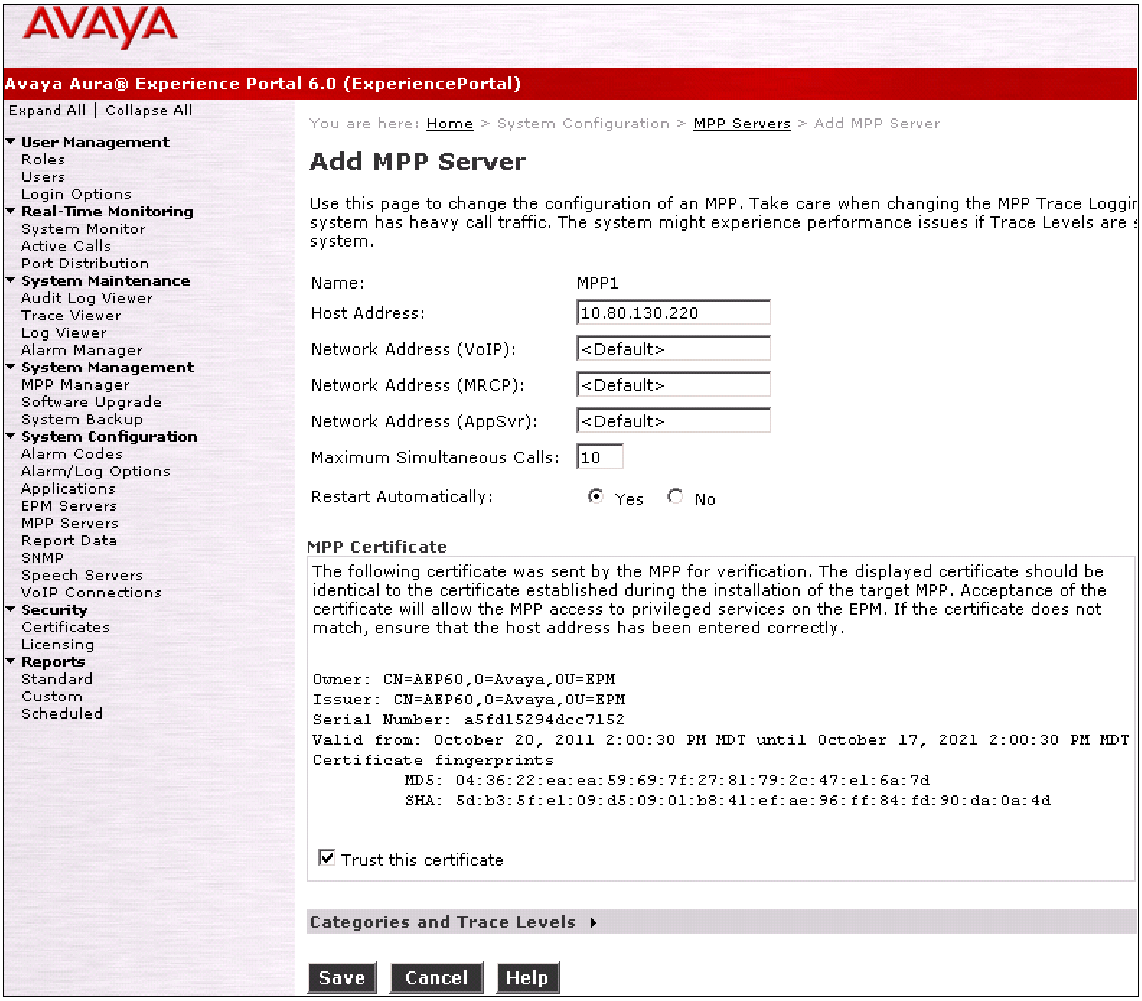
MPP Servers Page

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



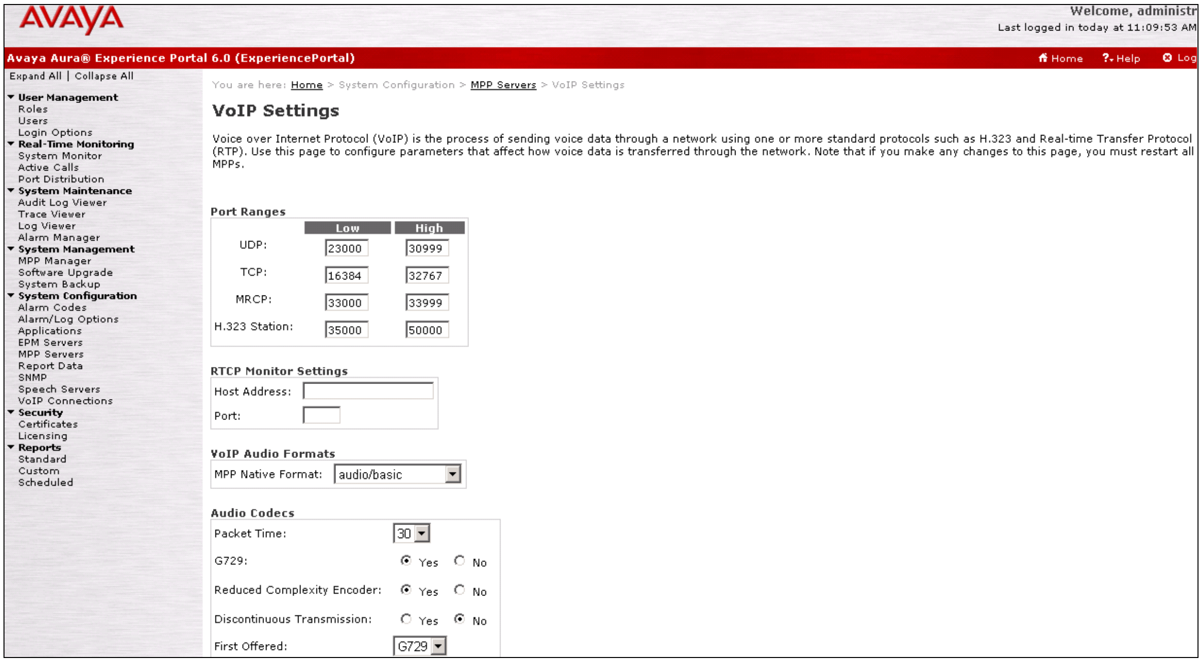
Add MPP Servers Page

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



Add MPP Server Page - Continued

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



VoIP Settings Page

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

The configuration change example noted in this section is not required for any of the call flows illustrated in these Application Notes. For incoming calls from AT&T IP Toll Free service to Experience Portal, AT&T specifies the value 100 for the RFC2833 telephone-events that signal DTMF digits entered by the user.

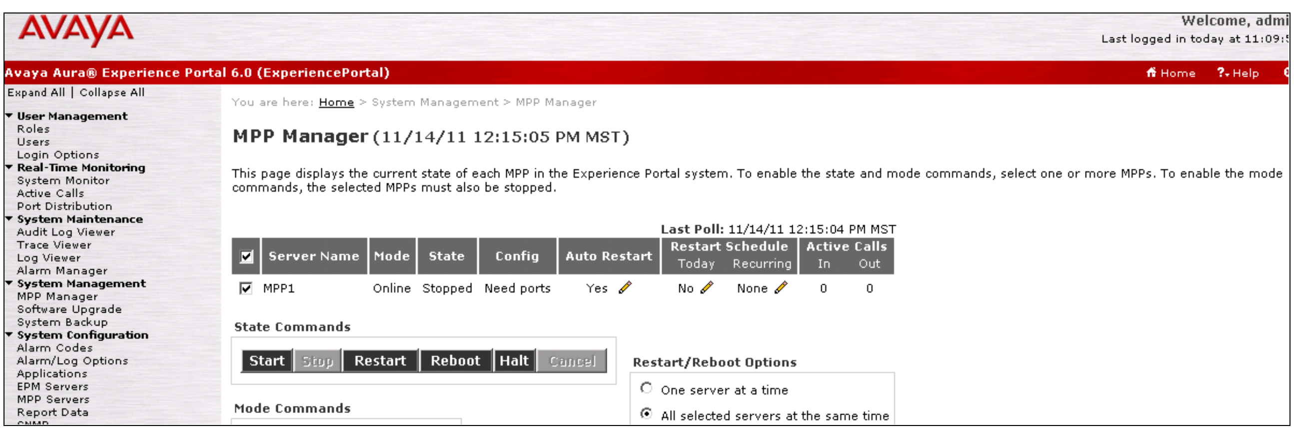
When Experience Portal answers, the SDP from Experience Portal matches this AT&T offered value. When Experience Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Experience Portal offers the SDP. By default, Experience Portal specifies the value 127 for the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

- Access the Experience Portal via the command line interface and navigate to the /opt/Avaya/ExperiencePortal/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**.

5.7. 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

6. 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 [4] and [5] 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.

6.1. System Parameters

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

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

```
display system-parameters customer-options                    Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   30
           Maximum Concurrently Registered IP Stations: 12000 6
             Maximum Administered Remote Office Trunks: 8000  0
Maximum Concurrently Registered Remote Office Stations: 12000 0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 20    0
                  Maximum Video Capable H.323 Stations: 20    0
                   Maximum Video Capable IP Softphones: 20    0
                       Maximum Administered SIP Trunks: 5000  30
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 5     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 200   0
      NOTE: You must logoff & login to effect the permission changes.)
```

System-Parameters Customer-Options Form – Page 2

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

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

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

           IP Attendant Consoles? N
        (NOTE: You must logoff & login to effect the permission changes.)
```

System-Parameters Customer-Options Form – Page 4

6.2. Dial Plan

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

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

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

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

Dialplan Analysis Form

6.3. IP Network Parameters

These Application Notes assume that the appropriate IP network regions and IP codec sets have already been administered to support internal calls, i.e., calls within the enterprise. For simplicity in this sample configuration, all Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within are assigned to a single IP network region. This section describes the steps for administering an additional IP network region and IP codec set to represent inbound calls from the AT&T IP Toll Free service to Experience Portal that are subsequently transferred to Communication Manager via 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**. G.726 is supported by Communication Manager but not by Experience Portal.

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

                          IP Codec Set

    Codec Set: 2

    Audio        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

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

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

                          IP Codec Set
                              Allow Direct-IP Multimedia? n

                      Mode                  Redundancy
 FAX                  t.38 standard          0
 Modem                off                    0
 TDD/TTY              US                     3
 Clear-channel        n                      0
```

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

2. Enter the **change node-names ip** command, and add a node name and the IP address for the Acme SBC. 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. These node names are used in **Section 6.4** and **Section 6.5**.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AcmeSBC              10.80.130.250
Messaging            10.80.130.222
CLAN-1A05            10.80.130.102
```

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

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

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

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

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

```
change ip-network-region 2                                    Page   4 of  20

                   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 4

6.4. Inbound Calls

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

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 Acme SBC.
   - **Near-end Node Name** – Set to the node name of the C-LAN board noted in **Section 6.3**, **Step 2**.
   - **Far-end Node Name** – Set to the node name of the Acme SBC as administered in **Section 6.3**, **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 6.3**, **Step 3**.
   - **Far-end Domain** – This field was left blank in this reference configuration to accept calls from all Domains/IP Addresses. Alternatively, Header Manipulation Rules can be implemented on Acme SBC to change the **From** header.
   - **DTMF over IP** – Set to **rtp-payload** to enable Communication Manager to use DTMF as per RFC 2833.
   - **Direct IP-IP Audio Connections** – Set to **y**, indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible.

```
add signaling-group 2                                          Page   1 of   1

 Group Number: 2              Group Type: sip
                        Transport Method: tcp


   Near-end Node Name: CLAN_1A02            Far-end Node Name: ASM
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 2
       Far-end Domain:

                                          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 2                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: ATT IPTF                    COR: 1        TN: 1        TAC: 102
   Direction: Incoming       Outgoing Display? n
 Dial Access? n                                              Night Service:
Queue Length: 0
Service Type: public-ntwrk         Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 10
```

Trunk-Group Form for Transferred Inbound Calls – Page 1

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

```
add trunk-group 5                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n                   Measured: none
                                                       Maintenance Tests? Y
              Numbering Format: public
                                        UUI Treatment: service-provider
                                         Replace Restricted Numbers? n
                                        Replace Unavailable Numbers? n
Show ANSWERED BY on Display? y
```

Trunk-Group Form for Transferred Inbound Calls – Page 3

4. On **Page 4** of the **trunk-group** form, provision the following:
   - **Support Request History** – Set to **n** as required by AT&T.
   - **Telepone Event Payload Type** – Set to **100** in this reference configuration.

```
add trunk-group 5                                              Page   4 of  21
                                PROTOCOL VARIATIONS
                     Mark Users as Phone? n
           Prepend '+' to Calling Number? n
    S end Transferring Party Information? n
                Network Call Redirection? n
                   Send Diversion Header? n
                 Support Request History? n
            Telephone Event Payload Type: 100
      Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? n
      Identity for Calling Party Display: P-Asserted-Identity
                            Enable Q-SIP? n
```

Trunk-Group Form for Transferred Inbound Calls – Page 4

5. 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
 7  666            2                        7           Maximum Entries: 9999
 10 303           2                        10
```

Public-Unknown-Numbering Form

6.5. Avaya Aura® Mesaging Traffic (Coverage and MWI)

Calls from AT&T IP Toll Free service after being transferred to an agent may be covered to the agent's mailbox. This section describes the steps for administering the SIP trunk from Communication Manager to Messaging.

1. Repeat the steps to in **Section 6.4**, **Step 1** to configure a signaling group to Messaging. Following screen shows the values used.

```
add signaling-group 3                                          Page   1 of   1

 Group Number: 3             Group Type: sip
                        Transport Method: tcp


   Near-end Node Name: CLAN_1A02           Far-end Node Name: Messaging
 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 Messaging Calls

2. Repeat the **Section 6.4**, **Steps 2-4** to configure a trunk group to Messaging. Following screens shows the values used.

```
add trunk-group 3                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: Messaging                   COR: 1        TN: 1        TAC: 103
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                              Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 3
                                                   Number of Members: 10
```

Trunk-Group Form for Messaging Calls – Page 1

```
add trunk-group 3                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n                    Measured: none
                                                        Maintenance Tests? Y
              Numbering Format: private
                                        UUI Treatment: service-provider
                                         Replace Restricted Numbers? n
                                       Replace Unavailable Numbers? n
                       Modify Tandem Calling Number: no
```

Trunk-Group Form for Messaging Calls – Page 3

```
add trunk-group 3                                              Page   4 of  21
                                PROTOCOL VARIATIONS
                     Mark Users as Phone? n
           Prepend '+' to Calling Number? n
    S end Transferring Party Information? n
                Network Call Redirection? n
                   Send Diversion Header? n
                 Support Request History? y
            Telephone Event Payload Type: 100


      Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? n
      Identity for Calling Party Display: P-Asserted-Identity
                           Enable Q-SIP? n
```

Trunk-Group Form for Messaging Calls – Page 4

3. Enter the **change private-numbering 0** command to specify the connected party numbers sent on transferred inbound calls. In the **private-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**.
   - **Private Prefix** – If necessary, enter enough prefix digits to form the desired connected party number.
   - **Total Len** – Enter the total length of the connected party number to be sent.

```
change private-numbering 0                                     Page   1 of   2
                          NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext Ext           Trk       Private         Total
Len Code          Grp(s)    Prefix          Len
                                                      Total Administered: 1
 4  7             3                         7            Maximum Entries: 9999
```

Private-Numbering Form

6.6. Optional Features

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

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

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

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

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

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

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. Refer to **[6]** and **[7]** for further details if necessary. The samples that follow are provided for reference purposes only.

6.6.1. Hunt Group for Station Coverage to Avaya Aura® Messaging

Hunt group **2** is used in the reference configuration to verify the coverage to voicemail. The hunt group **2** is defined with the 7 digit Messaging pilot number **6665000**.

```
display hunt-group 2                                            Page   1 of  60
                                  HUNT GROUP
            Group Number: 2                                 ACD? n
              Group Name: MM Voicemail                    Queue? n
         Group Extension: 666-5000                       Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Hunt Group Form – Page 1

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

Hunt Group Form – Page 2

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

```
display coverage path 2                                         Page   1 of  1
                                 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

6.6.2. Call Center Provisioning

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

```
display system-parameters customer-options                        Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                ACD? y                          Reason Codes? n
                       BCMS (Basic)? y               Service Level Maximizer? n
         BCMS/VuStats Service Level? y             Service Observing (Basic)? n
  BSR Local Treatment for IP & ISDN? n     Service Observing (Remote/By FAC)? n
                  Business Advocate? n             Service Observing (VDNs)? n
                    Call Work Codes? n                            Timed ACW? n
      DTMF Feedback Signals For VRU? n                   Vectoring (Basic)? y
                   Dynamic Advocate? n               Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y            Vectoring (G3V4 Enhanced)? y
                            EAS-PHD? y             Vectoring (3.0 Enhanced)? y
                   Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
               Least Occupied Agent? n     Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? n                   Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n      Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n                Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n               Vectoring (Variables)? n

        (NOTE: You must logoff & login to effect the permission changes.)
```

Call Center Optional Features Form

```
display agent-loginID 6663011                              Page   1 of   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 6663011                               Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                              Service Objective? n
Call Handling Preference: skill-level              Local Call Preference? n

    SN   RL SL          SN   RL SL          SN   RL SL          SN   RL SL
 1: 11      1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
```

Agent Form - Page 2

```
display hunt-group 11                                       Page   1 of   4
                                  HUNT GROUP

            Group Number: 11                              ACD? y
              Group Name: Skill-11                      Queue? y
         Group Extension: 666-4011                     Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                  MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:

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

Skill (Hunt Group) Form – Page 1

```
display hunt-group 11                                       Page   2 of   4
                                  HUNT GROUP

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

   Multiple Call Handling: none
 Timed ACW Interval (sec):     After Xfer or Held Call Drops:  n
```

Skill (Hunt Group) Form – Page 2

```
display vdn 6662010                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2010
                                  Name: To SelectSkill
                           Destination: Vector Number        10
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN#: 1
                              Measured: none

       VDN of Origin Annc. Extensions*:
                            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 6661002    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[4]

```
display vector 11                                               Page   1 of   6
                                  CALL VECTOR
    Number: 11              Name: Skill 11
                                             Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    2   secs hearing ringback
02 announcement 6661003
03 queue-to     skill 11   pri m
04 announcement 6661006
05 goto step    3             if unconditionally
06
```

Skill-11 Vector

```
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[5]

7. Avaya Aura® Messaging

The administration for Messaging is beyond the scope of these Application Notes. Refer to **[8]** and **[9]** for further details.

[4] This vector was used for the call flow scenario where Experience Portal transfers the inbound call to an Communication Manager skill without checking whether an agent in that skill is available.
[5] This vector was used for the call flow scenario where Experience Portal checks a Communication Manager skill for agent availability before transferring the inbound call to the skill.

8. 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 [**11**] for further details and explanations on the configuration below.

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

```
local-policy
      from-address
                                  *
      to-address
                                  *
      source-realm
                                  IPTF-Enterprise
      description
      activate-time               N/A
      deactivate-time             N/A
      state                       enabled
      policy-priority             none
      last-modified-by            admin@console
      last-modified-date          2011-08-12 10:25:23
      policy-attribute
            next-hop                    135.242.225.210
            realm                       ATT
            action                      none
            terminate-recursion         disabled
            carrier
            start-time                  0000
            end-time                    2400
            days-of-week                U-S
            cost                        0
            app-protocol                SIP
            state                       enabled
            methods
            media-profiles
```

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

```
local-policy
      from-address
                                  *
      to-address
                                  00000
      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                       10.80.100.220
            realm                          IPTF-Enterprise
            action                         none
            terminate-recursion            disabled
            carrier
            start-time                     0000
            end-time                       2400
            days-of-week                   U-S
            cost                           0
            app-protocol                   SIP
            state                          enabled
            methods
            media-profiles
```

ANNOTATION: The two local policies below aid the routing of SIP messages from the Experience Portal to Communication Manager.

```
local-policy
      from-address
                                     *
      to-address
                                     666
      source-realm
                                     IPTF-Enterprise
      description
      activate-time                  N/A
      deactivate-time                N/A
      state                          enabled
      policy-priority                none
      last-modified-by               admin@console
      last-modified-date             2011-08-12 10:25:23
      policy-attribute
            next-hop                       10.80.130.102
            realm                          IPTF-Enterprise
            action                         none
            terminate-recursion            disabled
            carrier
            start-time                     0000
            end-time                       2400
            days-of-week                   U-S
            cost                           0
            app-protocol                   SIP
            state                          enabled
            methods
            media-profiles
```

```
local-policy
      from-address
                                  *
      to-address
                                  666
      source-realm
                                  ATT
      description
      activate-time               N/A
      deactivate-time             N/A
      state                       enabled
      policy-priority             none
      last-modified-by            admin@console
      last-modified-date          2011-08-12 10:25:23
      policy-attribute
            next-hop                    10.80.130.206
            realm                       IPTF-Enterprise
            action                      none
            terminate-recursion         disabled
            carrier
            start-time                  0000
            end-time                    2400
            days-of-week                U-S
            cost                        0
            app-protocol                SIP
            state                       enabled
            methods
            media-profiles


media-manager
      state                       enabled
      latching                    enabled
      flow-time-limit             86400
      initial-guard-timer         300
      subsq-guard-timer           300
      tcp-flow-time-limit         86400
      tcp-initial-guard-timer     300
      tcp-subsq-guard-timer       300
      tcp-number-of-ports-per-flow  2
      hnt-rtcp                    disabled
      algd-log-level              NOTICE
      mbcd-log-level              NOTICE
      red-flow-port               1985
      red-mgcp-port               1986
      red-max-trans               10000
      red-sync-start-time         5000
      red-sync-comp-time          1000
      media-policing              enabled
      max-signaling-bandwidth     7752190
      max-untrusted-signaling     80
      min-untrusted-signaling     20
      app-signaling-bandwidth     0
      tolerance-window            30
```

```
      rtcp-rate-limit                0
      min-media-allocation           32000
      min-trusted-allocation         60000
      deny-allocation                32000
      anonymous-sdp                  disabled
      arp-msg-bandwidth              32000
      fragment-msg-bandwidth         0
      rfc2833-timestamp              disabled
      default-2833-duration          100
      rfc2833-end-pkts-only-for-non-sig enabled
      translate-non-rfc2833-event    disabled
      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 **removeUpdateAndModifyMaxptime**.

```
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              removeUpdateAndModifyMaxptime
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 Experience Portal used in this reference configuration.

```
session-agent
      hostname                       Enterprise-IPTF
      ip-address                     10.80.100.220
      port                           5060
      state                          enabled
      app-protocol                   SIP
      app-type
      transport-method               UDP+TCP
      realm-id                       Enterprise
      egress-realm-id
      description                    Experience Portal
      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                       135.242.225.210
      ip-address                     135.242.225.210
      port                           5060
      state                          enabled
      app-protocol                   SIP
      app-type
      transport-method               UDP
      realm-id                       ATT
      egress-realm-id
```

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

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

```
sip-interface
      state                         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                        IPTF-Enterprise
    description
    sip-port
        address                         10.80.130.250
        port                            5060
        transport-protocol              TCP
        tls-profile
        allow-anonymous                 all
        ims-aka-profile
    carriers
    trans-expire                    30
    invite-expire                   0
    max-redirect-contacts           0
    proxy-mode
    redirect-action
    contact-mode                    none
    nat-traversal                   none
    nat-interval                    30
    tcp-nat-interval                90
    registration-caching            disabled
    min-reg-expire                  300
    registration-interval           3600
    route-to-registrar              disabled
    secured-network                 disabled
    teluri-scheme                   disabled
    uri-fqdn-domain
    trust-mode                      all
    max-nat-interval                3600
    nat-int-increment               10
    nat-test-increment              30
    sip-dynamic-hnt                 disabled
    stop-recurse                    401,407
    port-map-start                  0
```

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

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

```
sip-manipulation
      name                      removeUpdateAndModifyMaxptime
      description               Strip Update from Allow list, modify Ptime
      header-rule
            name                      ReplaceMaxptime
            header-name               Content-Type
            action                    manipulate
            comparison-type           case-sensitive
            msg-type                  any
            methods                   INVITE
            match-value
            new-value
            element-rule
                  name                      modmline
                  parameter-name            application/sdp
```

```
            type                          mime
            action                        find-replace-all
            match-val-type                any
            comparison-type               case-sensitive
            match-value                   maxptime
            new-value                     ptime

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

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

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

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

steering-pool

```
      ip-address                      192.168.62.50
      start-port                      16384
      end-port                        32767
      realm-id                        ATT
      network-interface
      last-modified-by                admin@console
      last-modified-date              2011-08-25 19:11:47
steering-pool
      ip-address                      10.80.130.250
      start-port                      16384
      end-port                        32767
      realm-id                        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
```

9. Verification Steps

9.1. General

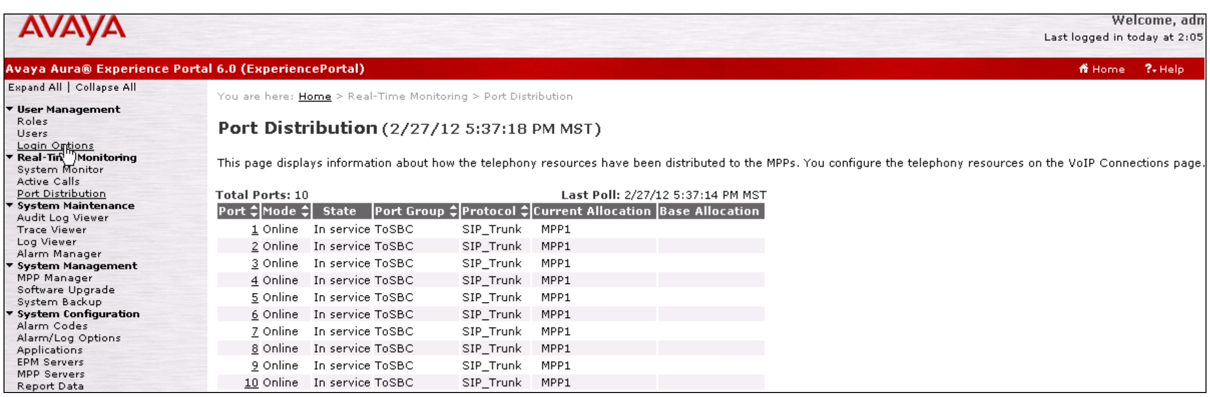
The following steps may be used to verify the configuration:

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

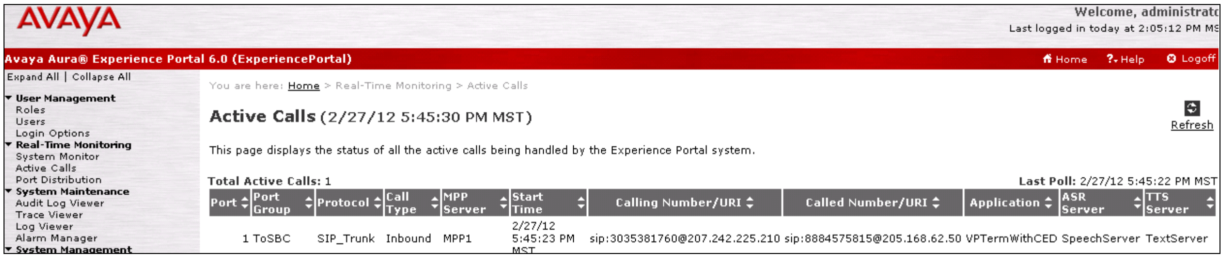
9.2. Avaya Aura® Experience Portal

The following commands are issued from the System Manager console.

1. Navigate to **Real-Time Monitoring→Port Distribution** to verify the SIP trunk on Experience Portal 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 Experience Portal:



9.3. Troubleshooting Tools

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

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

10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager, Avaya Aura® Experience Portal and the Acme Packet Net-Net can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of Avaya Aura® Experience Portal the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection.

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

11. References

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

[1] *Planning for Avaya Aura® Experience Portal*, August 2011
[2] *Implementing Avaya Aura® Experience Portal on a single server*, August 2011
[3] *Administering Aura® Experience Portal*, January 2011
[4] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509, August 2010
[5] *Avaya Aura® Communication Manager Feature Description and Implementation,* Document Id 555-245-205, August 2010
[6] *Administering Avaya Aura® Call Center Features*, November 2010
[7] *Programming Call Vectors in Avaya Aura® Call Center*, June 2010
[8] *Administering Avaya Aura® Messaging*, December 2011
[9] *Implementing Avaya Aura® Messaging*, October 2011
[10] *Application Notes for Avaya Aura® Experience Portal 6.0, Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service using MIS/PNT or AVPN Transport*

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

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

12. Addendum 1 – Configuration with Avaya Aura Session Manager

The alternate configuration tested here was with the calls referred to Communication Manager via Session Manager. This configuration was tested to simulate the hosted solution of Experience Portal deployment where the Experience Portal routes the call to an agent on Communication Manager via Session Manager. The configuration steps for the Session Manager are not listed here. Consult **Section 5** of [11] to configure Session Manager. Changes in call flow are illustrated in **Section 12.1**.

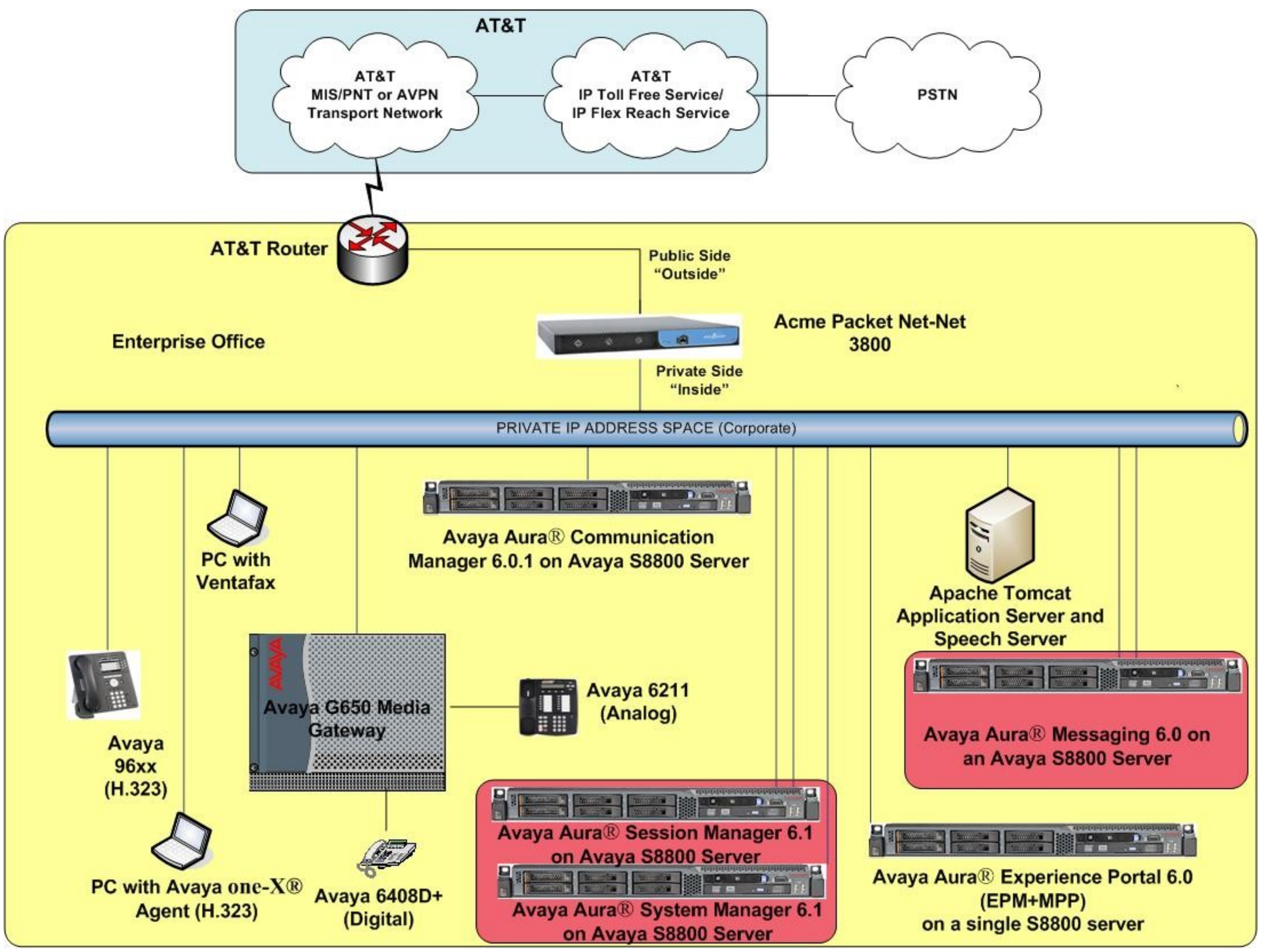


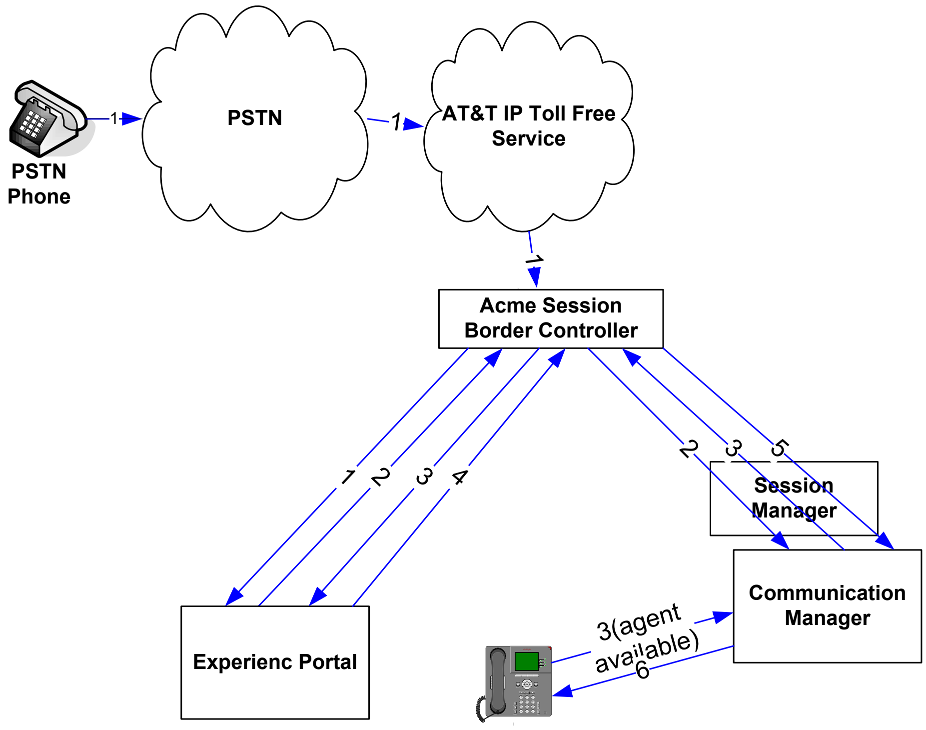
Figure 2: Alternate Reference Configuration

12.1. Call Flows

The first and fourth call scenarios illustrated in **Section 3.2** do not change with the introduction of Session Manager but for the second and third call flow, the call is routed to the Communication Manager via Session Manager.

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

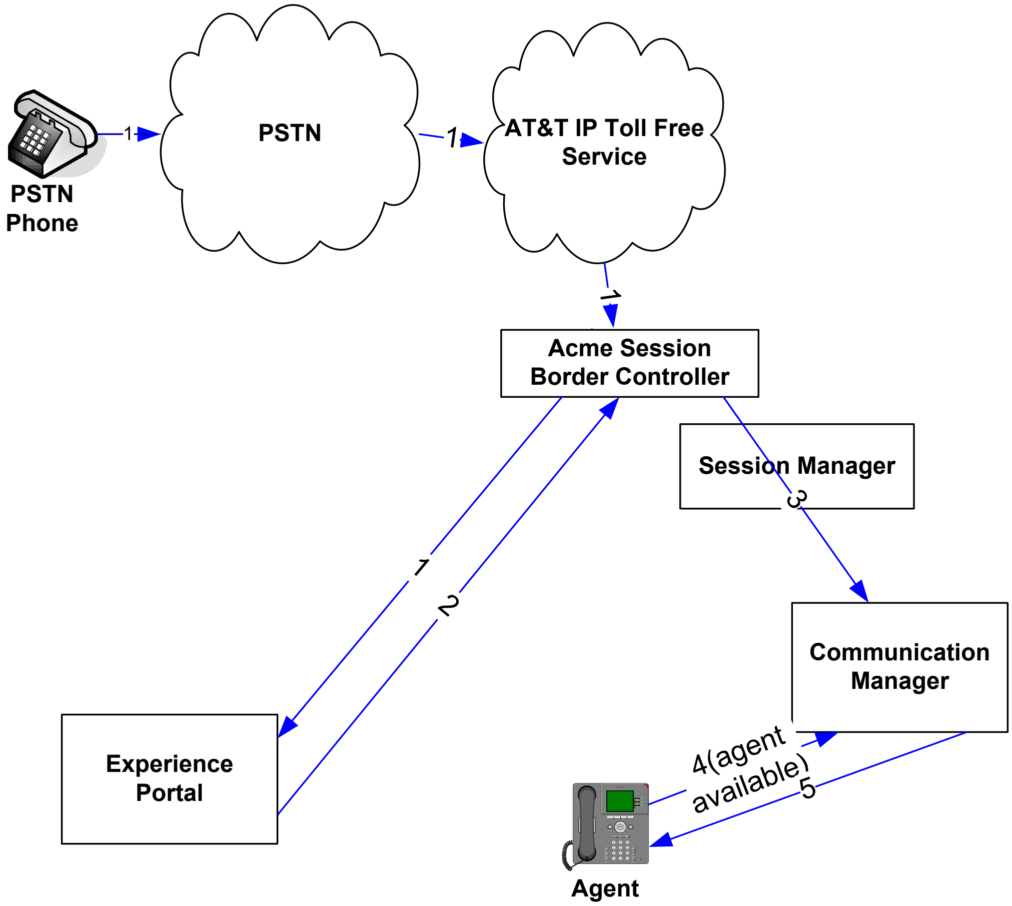
1. Same as the first four steps from the first call scenario.
2. In this scenario, the application is not sufficient to meet the caller's requests, and thus the call needs to be transferred to a Communication Manager agent. Experience Portal then puts the inbound call on hold and places a call to vector/skill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



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

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

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



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

12.2. Acme SBC Configuration

The only change to Acme configuration in **Section 8** for calls to be routed to Communication Manager through Session Manger is to the local policy where the next-hop will have the IP address of the Session Manager instead of the Communication Manager. The following local policies illustrate the corresponding change on the Acme SBC.

ANNOTATION: The two local policies below aid the routing of SIP messages from the Experience Portal to Communication Manager via Session Manager.

```
local-policy
      from-address
                               *
      to-address
                               666
                               666+
      source-realm
                               IPTF-Enterprise
      description
      activate-time            N/A
      deactivate-time          N/A
      state                    enabled
      policy-priority          none
      last-modified-by         admin@console
      last-modified-date       2011-08-12 10:25:23
      policy-attribute
            next-hop                 10.80.150.206
            realm                    IPTF-Enterprise
            action                   none
            terminate-recursion      disabled
            carrier
            start-time               0000
            end-time                 2400
            days-of-week             U-S
            cost                     0
            app-protocol             SIP
            state                    enabled
            methods
            media-profiles
```

```
local-policy
      from-address
                              *
      to-address
                              666
      source-realm
                              ATT
      description
      activate-time           N/A
      deactivate-time         N/A
      state                   enabled
      policy-priority         none
      last-modified-by        admin@console
      last-modified-date      2011-08-12 10:25:23
      policy-attribute
            next-hop                10.80.150.206
            realm                   IPTF-Enterprise
            action                  none
            terminate-recursion     disabled
            carrier
            start-time              0000
            end-time                2400
            days-of-week            U-S
            cost                    0
            app-protocol            SIP
            state                   enabled
            methods
            media-profiles
```

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.