Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Voice Portal with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Voice Portal with the AT&T IP Toll Free service. 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 Voice Portal is a speech-enabled interactive voice response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. Avaya Voice Portal interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya Voice Portal with the AT&T IP Toll Free service. 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 Voice Portal is a speech-enabled interactive voice response system that allows enterprises to provide multiple self- and assisted service resources to their customers in a flexible and customizable manner. **Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.** Avaya Voice Portal interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see Section 2.2 for descriptions) to Avaya Voice Portal and subsequent call transfers to Avaya Communication Manager skills and agents.

1.2. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. The "Connect with Avaya" section provides the worldwide support directory. 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.

1.3. Known Limitations

1. Avaya Voice Portal currently supports only G.711 codecs, so customers with the AT&T IP Toll Free service must use service profiles that include a G.711 codec.
2. Although Avaya Voice Portal release 5.0 and Avaya Communication Manager release 5.1.2 support the possibility of using SIP phones as agent stations, SIP phones were not tested as part of the configuration used to validate this solution.

2. Reference Configuration

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

- Avaya Voice Portal provides interactive voice response services to inbound callers. Avaya Voice Portal consists of one or more Media Processing Platform (MPP) servers and a Voice Portal Management System (VPMS) server.
- Avaya Communication Manager provides the enterprise voice communications services. In this sample configuration, Avaya Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers.

- The Avaya Media Gateway provides the physical interfaces and resources for enterprise voice communications. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "office" phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software.
- The Acme Packet Net-Net Session Director (SD) 3800 provides SIP Session Border Controller (SBC) functionality between the AT&T IP Toll Free service and the enterprise internal network. For brevity, the Acme Packet Net-Net SD 3800 will be referred to as the Acme Packet SBC through the remainder of these Application Notes.
- The Apache Tomcat Application Server[1] hosts the VXML and CCXML applications that provide the directives for handling the inbound calls to Avaya Voice Portal. Avaya Voice Portal references those applications.
- The Speech Server consists of Nuance OpenSpeech Recognizer and Nuance RealSpeak. Avaya Voice Portal uses the Speech Server for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) capabilities.

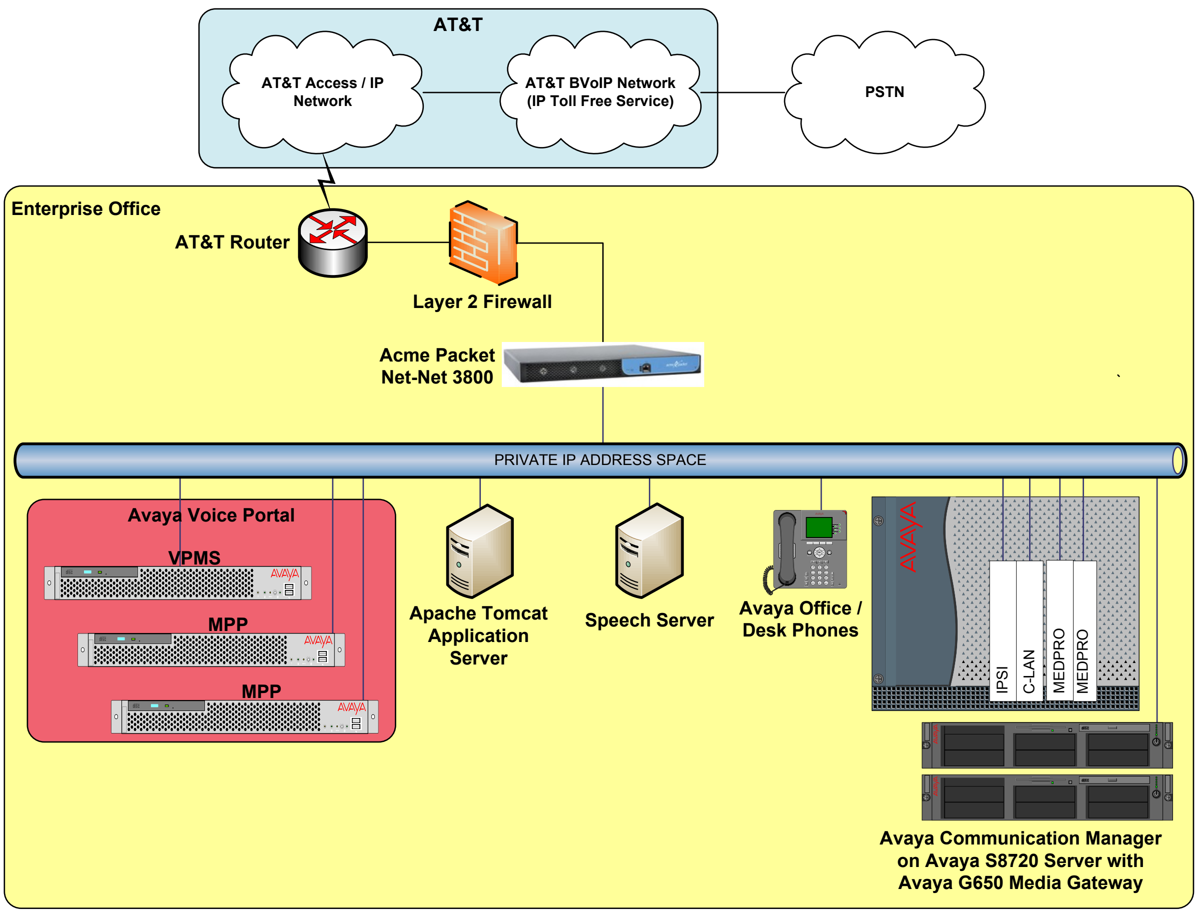

Figure 1: Sample Configuration

[1] For testing convenience only, the Apache Tomcat Application Server was installed on the Avaya VPMS. In production, the application server would be installed on a separate server.

2.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the sample configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Component	Illustrative Value in these Application Notes
Avaya Voice Portal	
MPP Servers IP Addresses	10.8.139.177 10.8.139.237
Avaya Communication Manager	
C-LAN IP Address	10.8.139.168
Vector Directory Number (VDN) Extensions	50xxx
Skill (Hunt Group) Extensions	60xxx
Agent Extensions	61xxx
Phone Extensions	51xxx
Announcement Extensions	52xxx
Acme Packet SBC	
IP Address of "Outside" Interface (connected to AT&T IP Toll Free Service)	10.160.177.210 (active) 10.160.177.211 (primary) 10.160.177.212 (secondary)
IP Address of "Inside" Interface (connected to Avaya elements)	10.8.139.240 (active) 10.8.139.241 (primary) 10.8.139.242 (secondary)
AT&T IP Toll Free Service	
Border Element IP Address	10.242.225.200
Digits Passed in SIP To Header to Avaya Voice Portal	00041530xxxxx

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

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

The first call scenario illustrated in **Figure 2** is an inbound call arriving and remaining on Avaya Voice Portal.

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

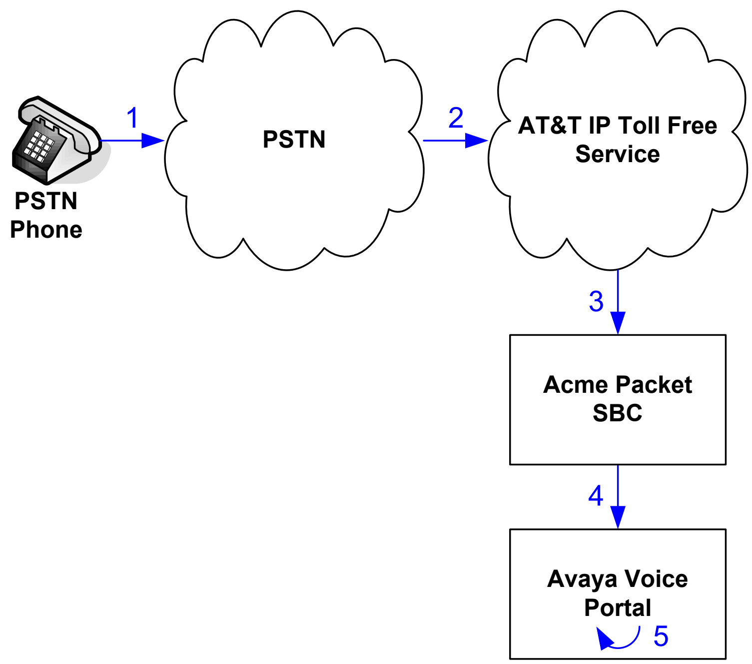

Figure 2: Inbound Call Handled Entirely by Avaya Voice Portal

The second call scenario illustrated in **Figure 3** is an inbound call arriving on Avaya Voice Portal and transferred to Avaya Communication Manager only after an Avaya Communication Manager skill has been canvassed for agent availability and an agent becomes available.

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 needs or requests, and thus the call needs to be transferred to an Avaya Communication Manager agent. Avaya Voice Portal then puts the inbound call on hold and canvasses a skill on Avaya Communication Manager by placing a call to a vector on Avaya Communication Manager. While the inbound call is on hold, Avaya Voice Portal can play music to the caller, prompt the caller for additional information, or otherwise interact with the caller.
3. Avaya Communication Manager informs Avaya Voice Portal when an agent in that skill becomes available.
4. Avaya Voice Portal instructs the Acme Packet SBC to transfer the inbound call to that skill.
5. The Acme Packet SBC transfers the inbound call to the aforementioned skill on Avaya Communication Manager.
6. Avaya Communication Manager routes the call to the agent.

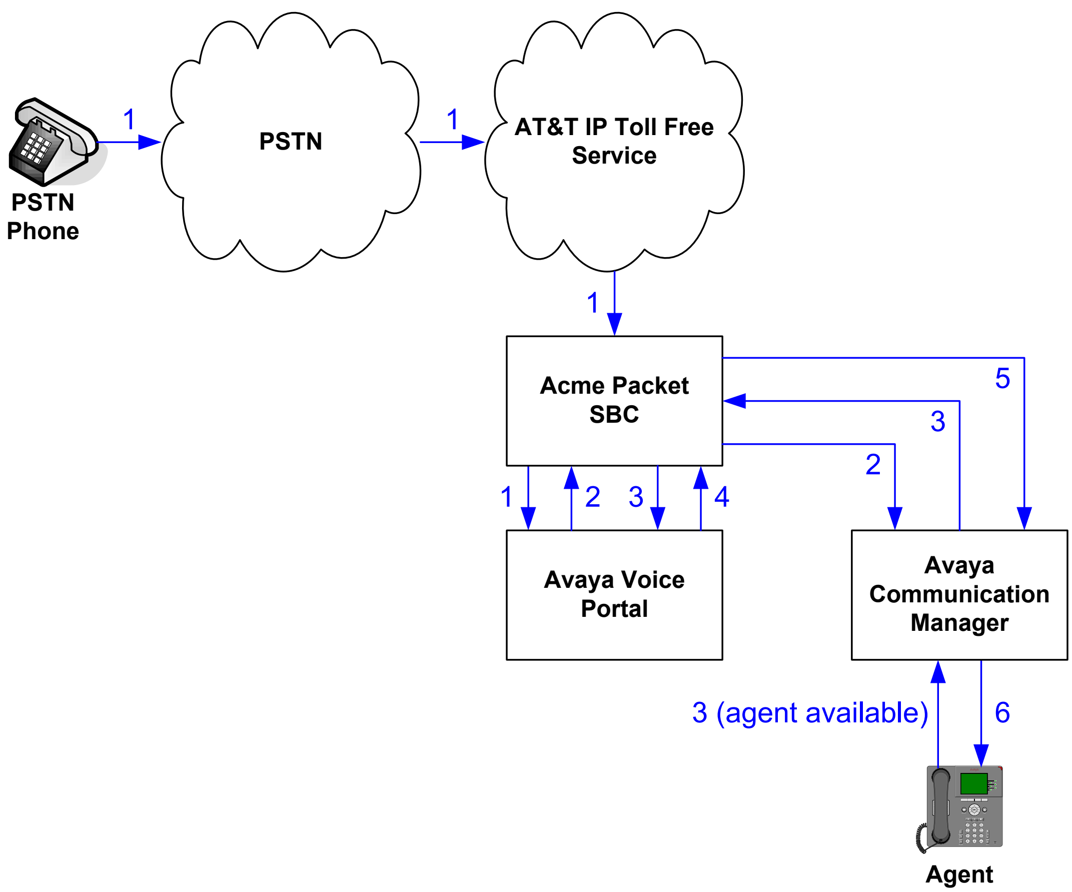

Figure 3: Inbound Call Handled Transferred by Avaya Voice Portal to Avaya Communication Manager Upon Available Agent

The third call scenario illustrated in **Figure 3** is an inbound call arriving on Avaya Voice Portal and transferred to an Avaya Communication Manager skill without canvassing that skill for agent availability, i.e., transferred to that skill regardless of whether an agent in that skill was available.

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 needs or requests, and thus the call needs to be transferred to an Avaya Communication Manager agent. Avaya Voice Portal instructs the Acme Packet SBC to transfer the inbound call to an Avaya Communication Manager skill.
3. The Acme Packet SBC transfers the inbound call to the aforementioned skill on Avaya Communication Manager.
4. An agent becomes available.
5. Avaya Communication Manager routes the call to the agent.

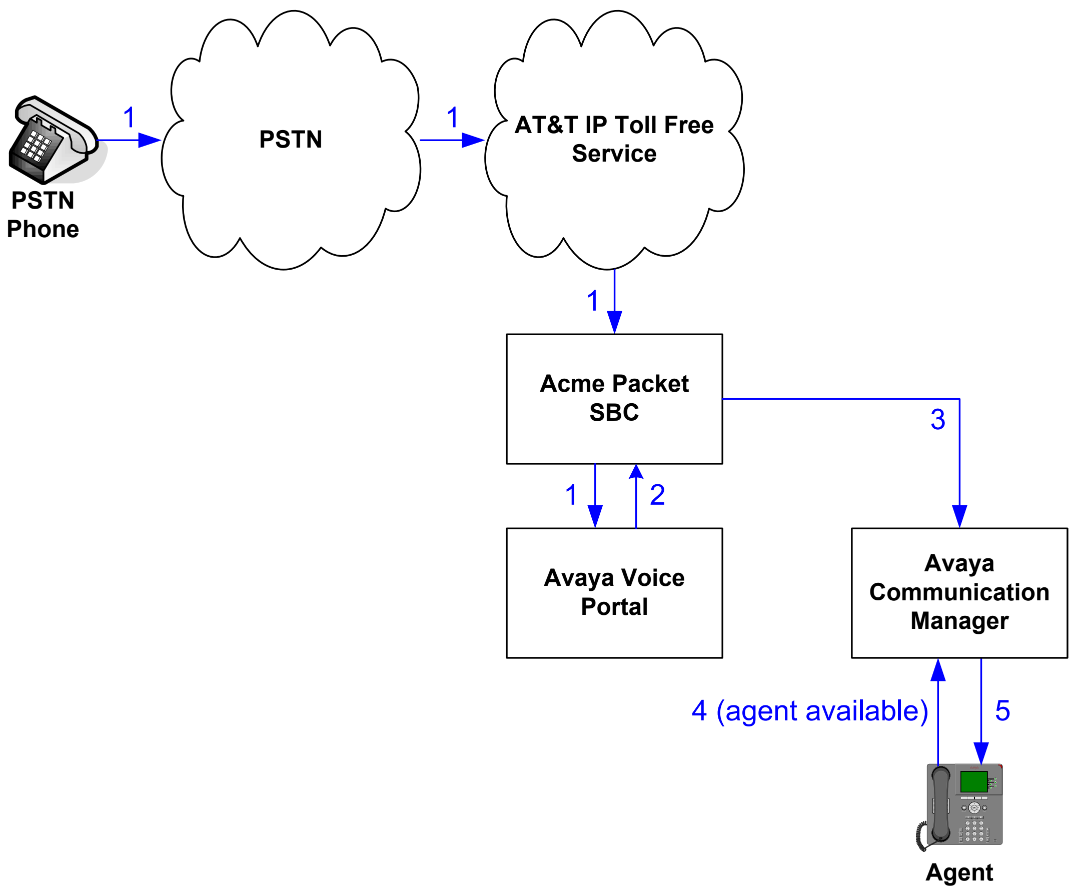

Figure 4: Inbound Call Handled Transferred by Avaya Voice Portal to Avaya Communication Manager Regardless of Agent Availability

3. Equipment and Software Validated

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

Component	Version
Avaya Voice Portal	5.0
Voice Portal Management System (VPMS)	5.0.0.0.4602
Media Processing Platform (MPP)	5.0.0.0.4603
Avaya S8720 Server	Avaya Communication Manager 5.1.2 with Service Pack 1 (R015x.01.2.416.4 with update 17067)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW12 FW045
TN799DP Control-LAN (C-LAN)	HW00 FW031
TN2302AP IP Media Processor (MedPro)	HW20 FW118
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW046
TN2501AP VAL-ANNOUNCEMENT	HW03 FW021
Avaya S8500B Server	Avaya SIP Enablement Services 5.1.2 with Service Pack 1 (SES-5.1.2.0-416.4b with update SES-01.2.416.4-SP1)
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0
Avaya 9650 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0
Avaya 4620SW IP Telephone	2.9.1
Apache Tomcat Application Server	6.0.18
Nuance OpenSpeech Recognizer	3.0.17
Nuance OpenSpeech Recognizer English en-US Language Pack	3.0.3
Nuance RealSpeak	4.0.12.0
Nuance RealSpeak American English en-US Jill	4.0.12.0.06187
Nuance SpeechWorks MediaServer	3.1.15
Acme Packet Net-Net Session Director 3800	SCX6.1.0 MR-1 Patch 1 (Build 277)
AT&T IP Toll Free Service	VNI 14

Table 2: Equipment and Software Versions

4. Avaya Voice Portal

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

4.1. Background

Avaya Voice Portal handles inbound calls according to the directives specified by Voice XML (VXML) and/or Call Control XML (CCXML) applications. The applications do not reside on Avaya Voice Portal, but rather on one or more separate application servers. References to those applications are administered on Avaya Voice Portal, along with one or more called numbers for each application reference. When an inbound call arrives at Avaya Voice Portal, the called party number is matched against those administered called numbers. If a match is found, then the corresponding application is accessed to handle the call. If no match[2] is found, Avaya Voice Portal informs the caller that the call can not be handled, and disconnects the call.

For the sample configuration described in these Application Notes, 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. Consult [1], [2], and [3] for further details if necessary.

4.2. VoIP Connection

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

1. Launch a web browser, enter http://<IP address of the Avaya VPMS server>/admin in the URL, and log in with the appropriate credentials. In the left pane, expand **System Configuration** and click on "**VoIP Connections**".

[2] 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.

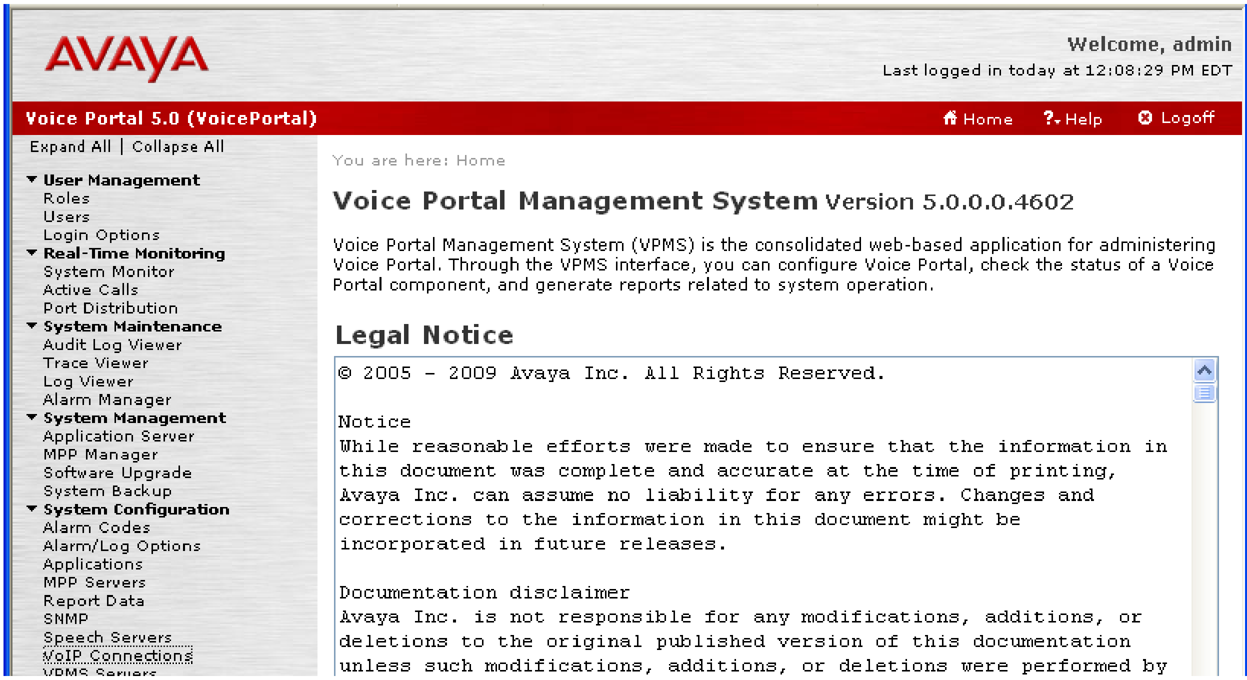

Figure 5: VPMS Home Page

2. In the **VoIP Connections** page, select the **SIP** tab and click on "**Add**".

Figure 6: VoIP Connections Page

3. In the **Add SIP Connection** page, provision the following and click on "**Continue**":
   - **Name** – Enter a descriptive name.

- **Proxy Transport** – Select "**TCP**".
- **Proxy Server Address** – Enter the IP address of the Acme Packet SBC interface on the enterprise internal network.
- **Proxy Server Port** – Enter "**5060**".

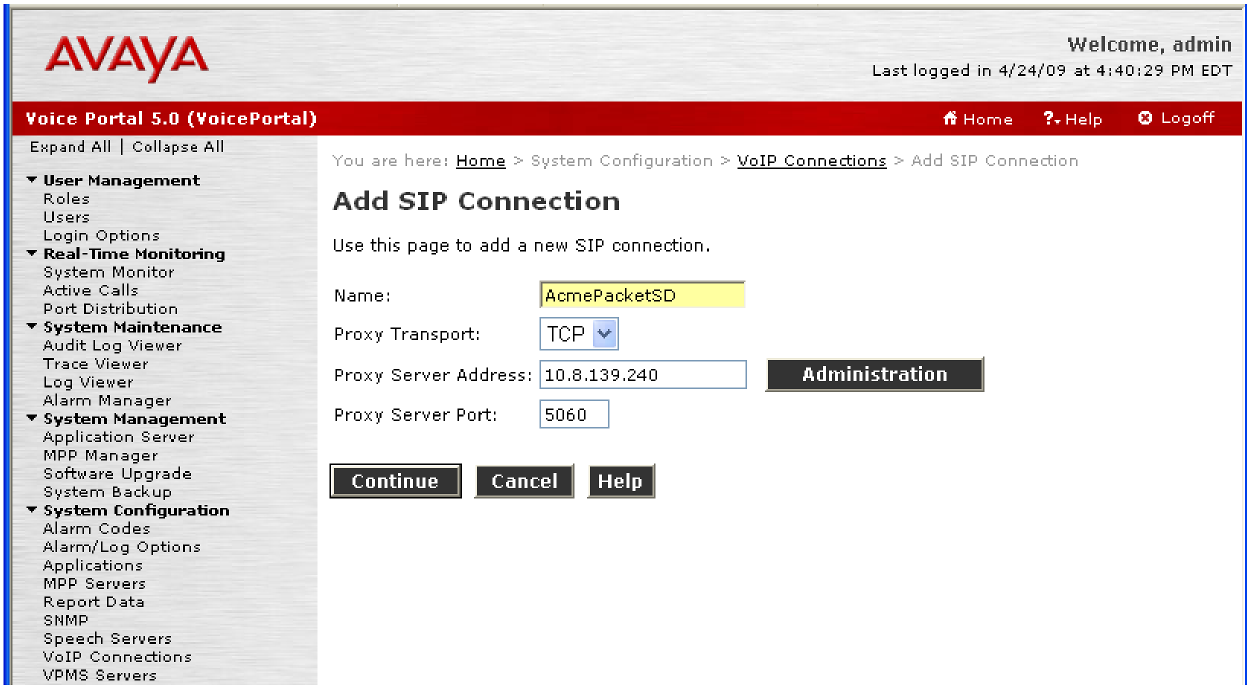

Figure 7: Add SIP Connection Page

4. Continuing in the **Add SIP Connection** page, provision the following and click on "**Save**":
   - **Enable** – Select "**Yes**".
   - **SIP Domain** – Enter the SIP domain used in the enterprise, for example, as noted in Section 5.3 Step 4.
   - **Call Capacity Maximum Simultaneous Calls** – Enter a number in accordance with call capacity needs and license allowances.
   - Select the **Call Capacity All Calls can be either inbound or outbound** radio button.

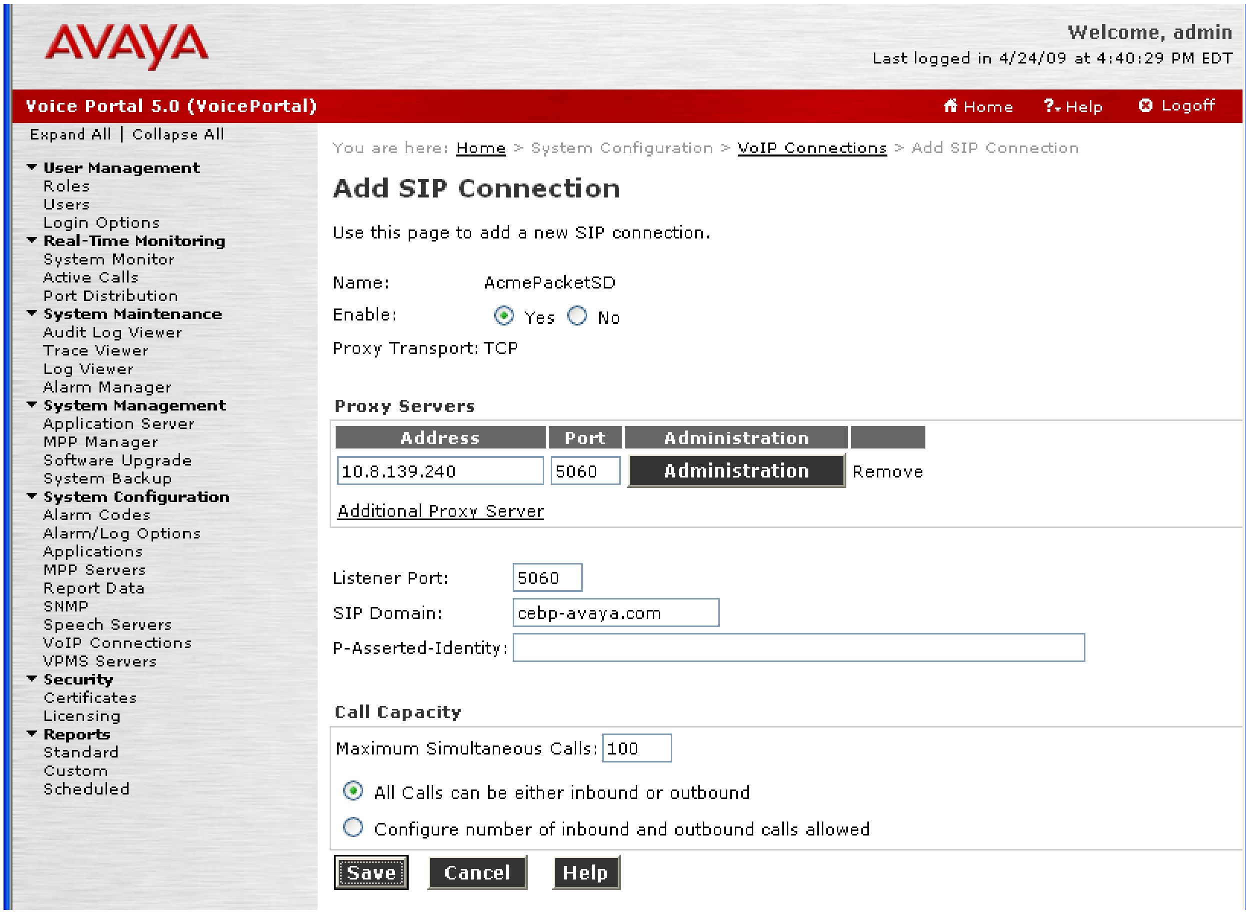

Figure 8: Add SIP Connection Page – Continued

4.3. Application References

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

1. In the left pane, expand **System Configuration** and click on "**Applications**". In the **Applications** page, click on "**Add**".

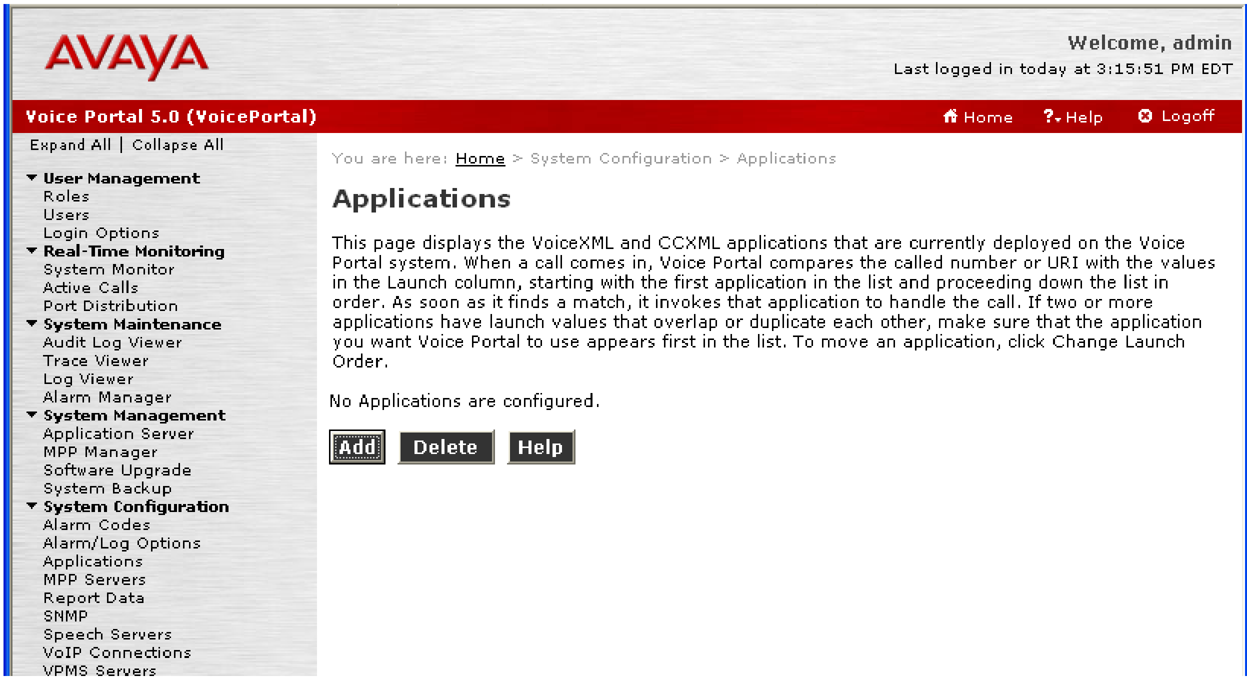

Figure 9: Applications Page

2. In the **Add Application** page, provision the following and click on "**Save**":
   - **Name** – Enter a descriptive name.
   - **Enable** – Select the "**Yes**".
   - **MIME Type** – Select "**VoiceXML**", "**CCXML**", or "**CCXML/VoiceXML**" according to the application type.
   - **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server.
   - **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
   - **Application Launch Type –** Select "**Inbound**".
   - Select the **Number** radio button.
   - **Called Number –** Enter an inbound AT&T IP Toll Free service called party number, specifically the number contained in the To header of the inbound SIP INVITE message, and click on "**Add**". Repeat to define additional AT&T IP Toll Free service called party numbers if necessary. Inbound AT&T IP Toll Free service calls with these called party numbers will be handled by the application defined in this Step.

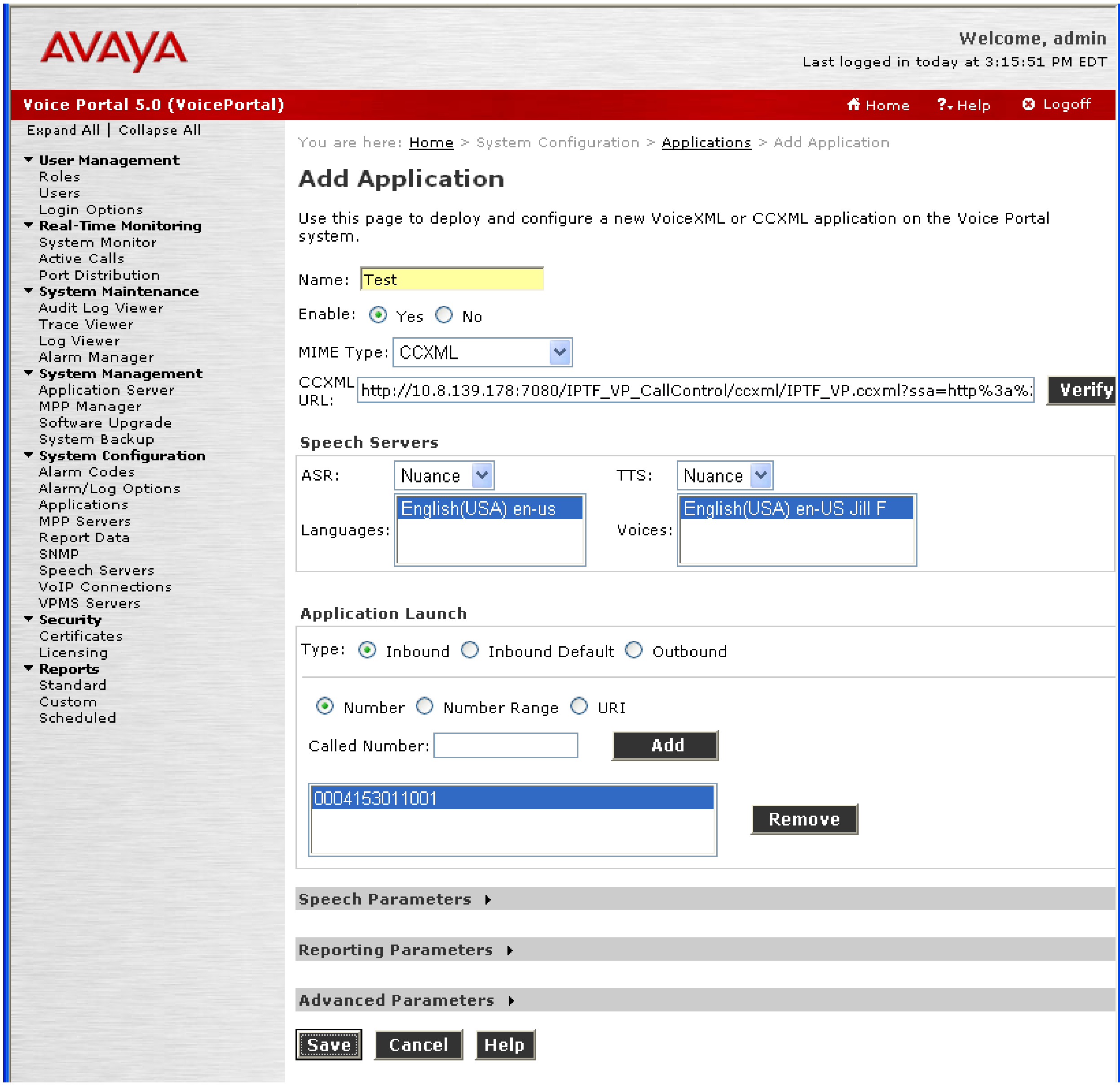

Figure 10: Add Application Page

3. Repeat Steps 1 - 2 to administer references to additional applications.

5. Avaya Communication Manager

This section describes the administration steps for Avaya Communication Manager in support of the sample configuration described in these Application Notes. The steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Avaya Communication Manager administration, including stations, C-LAN, Media Processor, and announcement boards, etc., has already been performed. Consult [4] and [5] for further details if necessary.

5.1. System Parameters

This section reviews the Avaya 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: 600   356
  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
```

Figure 11: 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
```

Figure 12: System-Parameters Customer-Options Form – Page 4

5.2. Dial Plan

Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings administered in **Figure 13**:

- 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 conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**5**" – local extensions for Avaya Communication Manager stations, Vector Directory Numbers (VDNs), and announcements, in this sample configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digit "**6**" – local extensions for Avaya Communication Manager agents and skills (hunt groups) in this sample configuration conform to this format.

```
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
     5            5    ext
     6            5    ext
```

Figure 13: Dialplan Analysis Form

5.3. 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 Avaya site. For simplicity in this sample configuration, all Avaya Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within the Avaya site are assigned to a single IP network region and all internal calls use a single IP codec set. This section describes the steps for administering an additional IP network region to represent the AT&T IP Toll Free service, and another IP codec set for transferred inbound calls, i.e., inbound calls from the AT&T IP Toll Free service to Avaya Voice Portal that are subsequently transferred to Avaya Communication Manager.

1. Enter the **change ip-codec-set ci** command, where **ci** is the number of an IP codec set used only for internal calls. On Page 1 of the **ip-codec-set** form, ensure that "**G.711MU**" is included in the codec list as shown in **Figure 14**.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
```

Figure 14: IP-Codec-Set Form for Internal Calls – Page 1

Repeat this step as necessary for each IP codec set used only for internal calls.

2. Enter the **change ip-codec-set ct** command, where **ct** is the number of an unused IP codec set. This IP codec set will be used for transferred inbound calls. On Page 1 of the **ip-codec-set** form, provision "**G.711MU**" as the only codec as shown in **Figure 15**.

```
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.711MU           n         2        20
 2:
 3:
```

Figure 15: IP-Codec-Set Form for External Calls – Page 1

3. Enter the **change node-names ip** command, and add a node name and the IP address for the Acme Packet 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 Avaya Communication Manager elements within the Avaya site. This C-LAN board will be used in Section 5.4 Step 1 for administering a SIP trunk to the Acme Packet SBC.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
AcmeSD                   10.8.139.240
clan-1a03                10.8.139.168
```

Figure 16: Change Node-Names IP Form

4. Enter the **display ip-network-region nrc**, where **nrc** is the number of the IP network region to which the C-LAN board in Step 3 is assigned. Note the value for **Authoritative Domain**.

```
display ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: cebp-avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

Figure 17: IP-Network-Region Form for the Network Region to which C-LAN board is Assigned – Page 1

5. 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 20                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 20
Location:         Authoritative Domain:
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

Figure 18: IP-Network-Region Form for the Network Region Representing the Avaya IP Toll Free Service – Page 1

On Page 3 of the **ip-network-region** form, for each IP network region pair consisting of this IP network region as the **src rgn** and the IP network region from Step 4 as the **dst rgn**, provision the following:

- **codec set** – Set to the codec set administered in Step 2.
- **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 in **Figure 19** was used for testing purposes only.

```
change ip-network-region 20                                     Page   3 of  19

                     Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening    Dyn
 rgn rgn  set   WAN  Units     Total Norm  Prio Shr Regions        CAC IGAR AGL
 20  1    2      y   NoLimit                                           n   all
 20  2
 20  3
 20  4
 20  5
 20  6
 20  7
 20  8
 20  9
 20  10
 20  11
 20  12
 20  13
 20  14
 20  15
```

Figure 19: IP-Network-Region Form for an IP Network Region Representing the AT&T IP Toll Free Service– Page 3

5.4. Inbound Calls

This section describes the steps for administering the SIP trunk to the Acme Packet 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 Avaya Communication Manager and the Acme Packet SBC. The transport protocol used between the Acme Packet SBC and the AT&T IP Toll Free service is UDP.
   - **Near-end Node Name** – Set to the node name of the C-LAN board noted in Section 5.3 Step 3.
   - **Far-end Node Name** – Set to the node name of the Acme Packet SBC as administered in Section 5.3 Step 3.
   - **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 5.3 Step 5 to represent the PSTN.
   - **Far-end Domain** – Leave blank.
   - **DTMF over IP** – Set to "**rtp-payload**" to enable Avaya Communication Manager to use DTMF according to 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 20                                          Page   1 of   1

 Group Number: 20              Group Type: sip
                         Transport Method: tcp


   Near-end Node Name: clan-1a03             Far-end Node Name: AcmeSD
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 20
       Far-end Domain:

                                       Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload        Direct IP-IP Audio Connections? y
                                                   IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3          Alternate Route Timer(sec): 6
```

Figure 20: 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 20                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 20                   Group Type: sip          CDR Reports: y
  Group Name: AcmePacketSBC               COR: 1        TN: 1        TAC: 120
   Direction: incoming      Outgoing Display? n
 Dial Access? n
Queue Length: 0
Service Type: public-ntwrk

                                                      Signaling Group: 20
                                                    Number of Members: 100
```

Figure 21: Trunk-Group Form for Transferred Inbound Calls – Page 1

3. Enter the **change public-unknown-numbering 0** command to specify the connected party numbers sent on transferred inbound calls. In the **public-unknown-numbering** form, for each local extension range assigned to Avaya 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.

   In **Figure 22**, for inbound calls to Avaya Communication Manager extensions 5xxxx and 6xxxx, 5-digit connected party numbers 5xxxx and 6xxxx are sent (i.e., the connected party's extension is sent without modification).

```
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
 5  5               20                         5           Maximum Entries: 9999
 5  6               20                         5
```

Figure 22: Public-Unknown-Numbering Form

5.5. Call Center

The administration of Avaya Communication Manager Call Center elements – agents, skills (hunt groups), vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Consult [4], [5], [6], and [7] for further details if necessary. The samples that follow are provided for reference purposes only.

```
display hunt-group 100                                          Page   1 of   3
                                  HUNT GROUP

            Group Number: 100                               ACD? y
              Group Name: Test Skill                      Queue? y
         Group Extension: 60100                          Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:

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

Figure 23: Sample Skill (Hunt Group) Form – Page 1

```
display hunt-group 100                                          Page   2 of   3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none             Service Objective (sec): 20
     Supervisor Extension:                 Service Level Supervisor? n


      Controlling Adjunct: none



 Timed ACW Interval (sec):                   Dynamic Queue Position? n
   Multiple Call Handling: none




                                 Redirect on No Answer (rings):
                                               Redirect to VDN:
                   Forced Entry of Stroke Counts or Call Work Codes? n
```

Figure 24: Sample Skill (Hunt Group) Form – Page 2

```
display hunt-group 100                                          Page   3 of   3
                                  HUNT GROUP

                       LWC Reception: none          AUDIX Name:




                      Message Center: none
```

Figure 25: Sample Skill (Hunt Group) Form – Page 3

```
display agent-loginID 61000                                     Page   1 of   2
                                AGENT LOGINID

                Login ID: 61000                                     AAS? n
                    Name: Agent-61000                             AUDIX? n
                      TN: 1                               LWC Reception: spe
                     COR: 1                      LWC Log External Calls? n
           Coverage Path:                      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:   :
```

Figure 26: Sample Agent Form – Page 1

```
display agent-loginID 61000                                     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: 100     1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
 4:                 19:                 34:                 49:
 5:                 20:                 35:                 50:
 6:                 21:                 36:                 51:
 7:                 22:                 37:                 52:
 8:                 23:                 38:                 53:
 9:                 24:                 39:                 54:
10:                 25:                 40:                 55:
11:                 26:                 41:                 56:
12:                 27:                 42:                 57:
13:                 28:                 43:                 58:
14:                 29:                 44:                 59:
15:                 30:                 45:                 60:
```

Figure 27: Sample Agent – Page 2

```
display vector 110                                          Page   1 of   6
                                  CALL VECTOR

    Number: 110              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 1st  pri m
02 stop
03
04
05
06
07
08
09
10
11
12
```

Figure 28: Sample Vector[3]

```
display vector 100                                          Page   1 of   6
                                  CALL VECTOR

    Number: 100              Name: 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 wait-time    2   secs hearing silence
02 announcement 52100
03 queue-to     skill 1st  pri m
04 wait-time    10  secs hearing music
05 announcement 52001
06 goto step    3             if unconditionally
07 stop
08
09
10
11
12
```

Figure 29: Sample Vector[4]

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

```
display vdn 50110                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 50110
                                 Name*: VP Test
                         Vector Number: 110
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none

               Service Objective (sec): 20
        VDN of Origin Annc. Extension*:
                            1st Skill*: 100
                            2nd Skill*:
                            3rd Skill*:



* Follows VDN Override Rules
```

Figure 30: Sample VDN

6. Configure Acme Packet SBC

The Acme Packet 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 [8] 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., Avaya Voice Portal, Avaya Communication Manager, etc., reside to the AT&T IP Toll Free service.

```
local-policy
      from-address
                                  *
      to-address
                                  *
      source-realm
                                  INSIDE
      description
      activate-time               N/A
      deactivate-time             N/A
      state                       enabled
      policy-priority             none
      last-modified-by            admin@console
      last-modified-date          2009-03-12 10:25:23
      policy-attribute
            next-hop                    10.242.225.200
            realm                       OUTSIDE
            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 Avaya Voice Portal.

```
local-policy
      from-address
                                  *
      to-address
                                  *
      source-realm
                                  OUTSIDE
      description
      activate-time               N/A
      deactivate-time             N/A
```

```
    state                           enabled
    policy-priority                 none
    last-modified-by                admin@console
    last-modified-date              2009-03-12 10:25:23
    policy-attribute
          next-hop                        SAG:AV_VOICE_PORTAL
          realm                           INSIDE
          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 aids the routing of SIP messages between Avaya Voice Portal and Avaya Communication Manager.

```
local-policy
    from-address
                                    *
    to-address
                                    50
                                    51
                                    60
                                    61
    source-realm
                                    OUTSIDE
    description
    activate-time                   N/A
    deactivate-time                 N/A
    state                           enabled
    policy-priority                 none
    last-modified-by                admin@console
    last-modified-date              2009-04-20 19:56:49
    policy-attribute
          next-hop                        10.8.139.168
          realm                           INSIDE
          action                          none
          terminate-recursion             enabled
          carrier
          start-time                      0000
          end-time                        2400
          days-of-week                    U-S
          cost                            0
          app-protocol
          state                           enabled
          methods
          media-profiles
local-policy
    from-address
```

```
                              *
      to-address
                              50
                              51
                              60
                              61
      source-realm
                              INSIDE
      description
      activate-time           N/A
      deactivate-time         N/A
      state                   enabled
      policy-priority         none
      last-modified-by        admin@console
      last-modified-date      2009-04-20 19:57:04
      policy-attribute
            next-hop                cebp-avaya.com
            realm                   INSIDE
            action                  none
            terminate-recursion     disabled
            carrier
            start-time              0000
            end-time                2400
            days-of-week            U-S
            cost                    0
            app-protocol
            state                   enabled
            methods
            media-profiles
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             2009-03-12 10:22:03
network-interface
      name                           wancom1
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.1.1
      sec-utility-addr               169.254.1.2
      netmask                        255.255.255.252
      gateway
      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             2009-03-12 10:21:39
network-interface
      name                           wancom2
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.2.1
      sec-utility-addr               169.254.2.2
      netmask                        255.255.255.252
      gateway
      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            2009-03-12 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                          s1p0
  sub-port-id                   0
  description
  hostname
  ip-address                    10.8.139.240
  pri-utility-addr              10.8.139.241
  sec-utility-addr              10.8.139.242
  netmask                       255.255.255.0
  gateway                       10.8.139.1
  sec-gateway
  gw-heartbeat
        state                         disabled
        heartbeat                     0
        retry-count                   0
        retry-timeout                 1
        health-score                  0
  dns-ip-primary                10.152.6.12
  dns-ip-backup1
  dns-ip-backup2
  dns-domain                    cebp-avaya.com
  dns-timeout                   11
    hip-ip-list
  ftp-address
    icmp-address                  10.8.139.240
  snmp-address
  telnet-address
  last-modified-by              admin@console
  last-modified-date            2009-03-13 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                          s0p0
```

```
      sub-port-id                     0
      description
      hostname
      ip-address                      10.160.177.210
      pri-utility-addr                10.160.177.211
      sec-utility-addr                10.160.177.212
      netmask                         255.255.255.224
      gateway                         10.160.177.193
      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              2009-03-12 10:24:07
ntp-config
      server                          10.152.6.12
      last-modified-by                admin@console
      last-modified-date              2009-03-12 10:20:46
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              2009-05-13 15:29:00
phy-interface
      name                            s0p1
      operation-type                  Media
      port                            1
      slot                            0
      virtual-mac                     00:08:25:a0:f3:69
      admin-state                     disabled
      auto-negotiation                enabled
      duplex-mode                     FULL
      speed                           100
      last-modified-by                admin@console
      last-modified-date              2009-05-13 15:29:12
```

```
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            2009-05-13 15:29:23
phy-interface
      name                          s1p1
      operation-type                Media
      port                          1
      slot                          1
      virtual-mac                   00:08:25:a0:f3:6f
      admin-state                   disabled
      auto-negotiation              enabled
      duplex-mode                   FULL
      speed                         100
      last-modified-by              admin@console
      last-modified-date            2009-05-13 15:29:37
phy-interface
      name                          wancom1
      operation-type                Control
      port                          1
      slot                          0
      virtual-mac
      wancom-health-score           8
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:21:30
phy-interface
      name                          wancom2
      operation-type                Control
      port                          2
      slot                          0
      virtual-mac
      wancom-health-score           9
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:21:30
```

ANNOTATION: The realm configuration "OUTSIDE" below represents the external network on which the AT&T IP Toll Free service resides, and applies two SIP manipulations (RemoveUPDATE and NAT_IP).

```
realm-config
      identifier                    OUTSIDE
      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               RemoveUPDATE
out-manipulationid              NAT_IP
manipulation-string
class-profile
average-rate-limit              0
access-control-trust-level      none
invalid-signal-threshold        4
maximum-signal-threshold        3000
untrusted-signal-threshold      10
nat-trust-threshold             0
deny-period                     60
ext-policy-svr
symmetric-latching              disabled
pai-strip                       disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching             none
restriction-mask                32
accounting-enable               enabled
user-cac-mode                   none
user-cac-bandwidth              0
user-cac-sessions               0
icmp-detect-multiplier          0
icmp-advertisement-interval     0
icmp-target-ip
monthly-minutes                 0
net-management-control          disabled
delay-media-update              disabled
refer-call-transfer             disabled
codec-policy
codec-manip-in-realm            disabled
constraint-name
call-recording-server-id
stun-enable                     disabled
stun-server-ip                  0.0.0.0
stun-server-port                3478
```

```
      stun-changed-ip                0.0.0.0
      stun-changed-port              3479
      match-media-profiles
      qos-constraint
      last-modified-by               admin@console
      last-modified-date             2009-04-22 19:26:23
```

ANNOTATION: The realm configuration "INSIDE" below represents the internal network on which the Avaya elements reside, and applies a SIP manipulation (AVP-manip).

```
realm-config
      identifier                     INSIDE
      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
      out-manipulationid             AVP-manip
      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             2009-03-12 19:50:37
redundancy-config
      state                          enabled
      log-level                      INFO
      health-threshold               75
      emergency-threshold            50
      port                           9090
      advertisement-time             500
      percent-drift                  210
      initial-time                   1250
      becoming-standby-time          180000
      becoming-active-time           100
      cfg-port                       1987
      cfg-max-trans                  10000
      cfg-sync-start-time            5000
      cfg-sync-comp-time             1000
      gateway-heartbeat-interval     0
      gateway-heartbeat-retry        0
      gateway-heartbeat-timeout      1
      gateway-heartbeat-health       0
      media-if-peercheck-time        0
      peer
            name                           acmesbc-pri
            state                          enabled
            type                           Primary
            destination
                  address                        169.254.1.1:9090
                  network-interface              wancom1:0
            destination
                  address                        169.254.2.1:9090
                  network-interface              wancom2:0
```

```
      peer
            name                              acmesbc-sec
            state                             enabled
            type                              Secondary
            destination
                  address                           169.254.1.2:9090
                  network-interface                 wancom1:0
            destination
                  address                           169.254.2.2:9090
                  network-interface                 wancom2:0
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:21:53
```

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

```
session-agent
      hostname                      10.242.225.200
      ip-address                    10.242.225.200
      port                          5060
      state                         enabled
      app-protocol                  SIP
      app-type
      transport-method              UDP
      realm-id                      OUTSIDE
      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                 300
      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              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               2009-03-17 17:36:20
```

ANNOTATION: The session agent below represents one of two Avaya Voice Portal MPP servers used in the sample configuration.

```
session-agent
      hostname                         10.8.139.177
      ip-address                       10.8.139.177
      port                             5060
      state                            enabled
      app-protocol                     SIP
      app-type
      transport-method                 StaticTCP
      realm-id                         INSIDE
      egress-realm-id
      description                      Voice Portal MPP1
      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                  300
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                  none
tcp-reconn-interval            0
max-register-burst-rate        0
register-burst-window          0
last-modified-by               admin@console
last-modified-date             2009-03-17 17:36:26
```

ANNOTATION: The session agent below represents the other Avaya Voice Portal MPP servers used in the sample configuration.

```
session-agent
      hostname                      10.8.139.237
      ip-address                    10.8.139.237
      port                          5060
      state                         enabled
      app-protocol                  SIP
      app-type
      transport-method              StaticTCP
      realm-id                      INSIDE
      egress-realm-id
      description                   Voice Portal MPP2
      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                 300
      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                  none
      tcp-reconn-interval            0
      max-register-burst-rate        0
      register-burst-window          0
      last-modified-by               admin@console
      last-modified-date             2009-03-17 17:36:53
```

ANNOTATION: The session agent below represents the Avaya Communication Manager C-LAN interface.

```
session-agent
      hostname                       cebp-avaya.com
      ip-address                     10.8.139.168
      port                           5060
      state                          enabled
      app-protocol                   SIP
      app-type
      transport-method               StaticTCP
      realm-id                       INSIDE
      egress-realm-id
      description                    CLAN 1
      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                   300
    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             disabled
    reuse-connections               TCP
    tcp-keepalive                   none
    tcp-reconn-interval             0
    max-register-burst-rate         0
    register-burst-window           0
    last-modified-by                admin@console
    last-modified-date              2009-03-17 17:44:57
```

ANNOTATION: The session group below consists of the two session agents defined earlier to represent the two Avaya Voice Portal MPP servers used in the sample configuration. The session group below load balances SIP requests to the Avaya Voice Portal MPP servers in a round robin fashion.

```
session-group
    group-name                      AV_VOICE_PORTAL
    description
    state                           enabled
    app-protocol                    SIP
    strategy                        RoundRobin
    dest
                                    10.8.139.177
                                    10.8.139.237
    trunk-group
```

```
sag-recursion                  disabled
stop-sag-recurse               401,407
last-modified-by               admin@console
last-modified-date             2009-03-12 10:24:50
```

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                  INSIDE
      egress-realm-id                INSIDE
      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             2009-03-12 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                       OUTSIDE
      description
      sip-port
            address                        10.160.177.210
            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
refer-call-transfer             disabled
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                        INSIDE
      description
      sip-port
            address                         10.8.139.240
            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
      refer-call-transfer            disabled
      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 performs address translation and topology hiding for SIP messages between the AT&T IP Toll Free services and the Avaya elements.

```
sip-manipulation
      name                           NAT_IP
      description                    Topology hiding for TO and FROM SIP
      header-rule
            name                           manipFrom
            header-name                    From
            action                         manipulate
            comparison-type                case-sensitive
            match-value
            msg-type                       request
            new-value
            methods
            element-rule
                  name                           FROM
                  parameter-name
```

```
            type                            uri-host
            action                          replace
            match-val-type                  ip
            comparison-type                 case-sensitive
            match-value
            new-value                       $LOCAL_IP
    header-rule
        name                            manipTo
        header-name                     To
        action                          manipulate
        comparison-type                 case-sensitive
        match-value
        msg-type                        request
        new-value
        methods
        element-rule
            name                            TO
            parameter-name
            type                            uri-host
            action                          replace
            match-val-type                  ip
            comparison-type                 case-sensitive
            match-value
            new-value                       $REMOTE_IP
    last-modified-by                admin@console
    last-modified-date              2009-03-12 10:22:14
```

ANNOTATION: The SIP manipulation below performs some modification of SIP headers in support of Avaya Voice Portal.

```
sip-manipulation
    name                            AVP-manip
    description                     Support Avaya VP Application
    header-rule
        name                            requri
        header-name                     request-uri
        action                          manipulate
        comparison-type                 case-sensitive
        match-value
        msg-type                        request
        new-value
        methods                         INVITE
        element-rule
            name                            getExpires
            parameter-name                  Expires
            type                            uri-header
            action                          store
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
        element-rule
            name                            getUUI
            parameter-name                  User-to-User
            type                            uri-header
```

```
            action                          store
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
      element-rule
            name                            getReplaces
            parameter-name                  Replaces
            type                            uri-header
            action                          store
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
header-rule
      name                            addExpires
      header-name                     Expires
      action                          add
      comparison-type                 boolean
      match-value                     $requri.$getExpires
      msg-type                        any
      new-value                       $requri.$getExpires.$0
      methods
header-rule
      name                            addUUI
      header-name                     User-to-User
      action                          add
      comparison-type                 boolean
      match-value                     $requri.$getUUI
      msg-type                        any
      new-value                       $requri.$getUUI.$0
      methods
header-rule
      name                            addReplaces
      header-name                     Replaces
      action                          add
      comparison-type                 boolean
      match-value                     $requri.$getReplaces
      msg-type                        any
      new-value                       $requri.$getReplaces.$0
      methods
header-rule
      name                            decodeUUIsemi
      header-name                     User-to-User
      action                          find-replace-all
      comparison-type                 pattern-rule
      match-value                     %3B
      msg-type                        any
      new-value                       ;
      methods
header-rule
      name                            decodeUUIequal
      header-name                     User-to-User
      action                          find-replace-all
      comparison-type                 pattern-rule
```

```
        match-value                     %3D
        msg-type                        any
        new-value                       =
        methods
  header-rule
        name                            decodeReplacesSemi
        header-name                     Replaces
        action                          find-replace-all
        comparison-type                 pattern-rule
        match-value                     %3B
        msg-type                        any
        new-value                       ;
        methods
  header-rule
        name                            decodeReplacesEqual
        header-name                     Replaces
        action                          find-replace-all
        comparison-type                 pattern-rule
        match-value                     %3D
        msg-type                        any
        new-value                       =
        methods
  header-rule
        name                            stripReqUri
        header-name                     request-uri
        action                          manipulate
        comparison-type                 case-sensitive
        match-value
        msg-type                        request
        new-value
        methods                         INVITE
        element-rule
              name                            delExpires
              parameter-name                  Expires
              type                            uri-header
              action                          delete-element
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value
        element-rule
              name                            delUUI
              parameter-name                  User-to-User
              type                            uri-header
              action                          delete-element
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value
        element-rule
              name                            delReplaces
              parameter-name                  Replaces
              type                            uri-header
              action                          delete-element
              match-val-type                  any
```

```
            comparison-type                 case-sensitive
            match-value
            new-value
    header-rule
        name                          stripTo
        header-name                   To
        action                        manipulate
        comparison-type               case-sensitive
        match-value
        msg-type                      request
        new-value
        methods                       INVITE
        element-rule
            name                            delExpires
            parameter-name                  Expires
            type                            uri-header
            action                          delete-element
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
        element-rule
            name                            delUUI
            parameter-name                  User-to-User
            type                            uri-header
            action                          delete-element
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
        element-rule
            name                            delReplaces
            parameter-name                  Replaces
            type                            uri-header
            action                          delete-element
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value
            new-value
    header-rule
        name                          manipFrom
        header-name                   From
        action                        manipulate
        comparison-type               case-sensitive
        match-value
        msg-type                      request
        new-value
        methods
        element-rule
            name                            FROM
            parameter-name
            type                            uri-host
            action                          replace
            match-val-type                  ip
            comparison-type                 case-sensitive
```

```
                match-value
                new-value                       $LOCAL_IP
      header-rule
            name                          manipTo
            header-name                   To
            action                        manipulate
            comparison-type               case-sensitive
            match-value
            msg-type                      request
            new-value
            methods
            element-rule
                name                            TO
                parameter-name
                type                            uri-host
                action                          replace
                match-val-type                  ip
                comparison-type                 case-sensitive
                match-value
                new-value                       $REMOTE_IP
      last-modified-by              admin@console
      last-modified-date            2009-03-12 10:23:30
```

ANNOTATION: The SIP manipulation below removes "UPDATE" from the Allow header in SIP messages from the AT&T IP Toll Free service.

```
sip-manipulation
      name                          RemoveUPDATE
      description                   Strip Update from Allow list
      header-rule
            name                          EditAllow
            header-name                   Allow
            action                        manipulate
            comparison-type               pattern-rule
            match-value
            msg-type                      any
            new-value
            methods
            element-rule
                name                            StripUPDATE
                parameter-name
                type                            header-value
                action                          find-replace-all
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value                     (,\s*UPDATE|UPDATE\s*,)
                new-value
      last-modified-by              admin@console
      last-modified-date            2009-04-22 19:25:08
```

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

```
steering-pool
      ip-address                    10.160.177.210
```

```
        start-port                      49152
        end-port                        65535
        realm-id                        OUTSIDE
        network-interface
        last-modified-by                admin@console
        last-modified-date              2009-03-25 19:11:47
steering-pool
        ip-address                      10.8.139.240
        start-port                      49152
        end-port                        65535
        realm-id                        INSIDE
        network-interface
        last-modified-by                admin@console
        last-modified-date              2009-03-12 10:25:12
system-config
        hostname                        acmesbc-pri
        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                 2009-03-12 10:20:46
```

7. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site with Avaya Voice Portal, Avaya Communication Manager, Avaya phones, an Acme Packet SBC, an Apache Tomcat application server, and a speech server (Nuance OpenSpeech Recognizer and Nuance RealSpeak).
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise sites was connected.

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

- Inbound calls to various Avaya Voice Portal applications.
- Inbound caller interaction with Avaya Voice Portal applications, including prompting, caller DTMF input, wait treatment (e.g., music on hold), speech recognition, and TTS.
- Avaya Voice Portal applications canvassing of Avaya Communication Manager skills for agent availability before transferring inbound calls to the skills.
- Avaya Voice Portal applications transferring of inbound calls to Avaya Communication Manager skills regardless of agent availability.
- Call and two-way talkpath establishment between callers and Avaya Communication Manager agents following transfers from Avaya Voice Portal.
- G.711 codec.
- Avaya Voice Portal applications sending DTMF to the AT&T IP Toll Free to invoke AT&T IP Toll Free Legacy Transfer Connect features (only those permitted for voice response units) and processing the resulting DTMF responses from the AT&T IP Toll Free service.

The above test objectives of Section 7 with limitations as noted in Section 1.3 were verified. Note that this configuration requires Update 17067 for Avaya Communication Manager release 5.1.2. Subsequent releases of Avaya Communication Manager are expected to incorporate this update within the primary release.

8. Verification Steps

8.1. Verification Tests

The following steps may be used to verify the configuration:

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

2. Place an inbound call to an Avaya Voice Portal application that can canvass an Avaya Communication Manager skill for 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 Avaya Voice Portal application while waiting to be transferred. Verify that when an agent in the skill becomes available, the call is successfully transferred to the agent and two-way talkpath exists between the caller and the agent.

3. Place an inbound call to an Avaya Voice Portal application that can transfer an inbound call to an Avaya Communication Manager skill regardless of 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 Avaya 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.

8.2. Troubleshooting Tools

The Avaya Communication Manager "list trace vector", "list trace vdn", "list trace tac", and/or "status trunk-group" commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The logging and reporting functions within the Avaya VPMS web interface may be used to examine the details of Avaya Voice Portal calls. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

9. Conclusion

These Application Notes described the steps for configuring Avaya Voice Portal with the AT&T IP Toll Free service. The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communication solution that provides toll-free services over SIP trunks. Avaya Voice Portal is a speech-enabled interactive voice response system that allows enterprises to provide multiple self and assisted service resources to their customers in a flexible and customizable manner.

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.

10. References

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

[1] *Planning for Voice Portal*, March 2009
[2] *Implementing Voice Portal on multiple servers*, March 2009
[3] *Administering Voice Portal*, March 2009
[4] *Administrator Guide for Avaya Communication Manager*, Issue 4.0, Release 5.0, January 2008, Document Number 03-300509
[5] *Feature Description and Implementation for Avaya Communication Manager*, Issue 6, January 2008, Document Number 555-245-205
[6] *Avaya Call Center Release 5.0 Call Vectoring and Expert Agent Selection (EAS) Guide*, Release 5.0, January 2008, Document Number 07-600780.
[7] *Avaya Call Center Release 5.0 Automatic Call Distribution (ACD) Guide*, Release 5.0, January 2008, Document Number 07-602568.

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

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

11. Change History

Issue	Date	Reason
1.0		Initial issue.

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.