Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Voice Portal, Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Acme Packet Net-Net SBC with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.1

Abstract

These Application Notes describe a sample configuration consisting of Avaya Voice Portal 5.1, Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0, and an Acme Packet Net-Net Session Border Controller. The enterprise equipment is integrated with the Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks. Using the sample configuration, Verizon toll-free calls can be delivered to Voice Portal self-service applications, which can transfer the callers to Communication Manager agents if necessary.

Verizon Business is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted in the Avaya Interoperability Test Lab, utilizing a Verizon Business Private IP (PIP) circuit connection to the production Verizon Business IPCC Services.

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1. Introduction

These Application Notes describe a sample configuration consisting of Avaya Voice Portal 5.1, Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0, and an Acme Packet Net-Net Session Border Controller. The enterprise equipment is integrated with the Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the IP Toll Free VoIP Inbound and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks. Using the sample configuration, Verizon toll-free calls can be delivered to Voice Portal self-service applications, which can transfer the callers to Communication Manager agents if necessary.

Access to the IPCC Services suite may use Internet Dedicated Access (IDA) or Private IP (PIP). PIP was used for the sample configuration described in these Application Notes. IP Toll Free VoIP Inbound is the base service offering that offers core call routing and termination features. IP IVR is an enhanced service offering that includes features such as menu-routing, custom transfer, and additional media capabilities.

In the sample configuration, an Acme Packet 4250 Net-Net Session Border Controller (SBC) is used as an edge device between the Avaya CPE and Verizon Business. The Acme Packet 3800 or 4500 SBC platforms may be used with similar configuration. The Acme Packet SBC performs SIP header manipulation and provides Network Address Translation (NAT) functionality to convert the private Avaya CPE IP addressing to IP addressing appropriate for the Verizon access method.

Avaya Aura® Session Manager is used as the Avaya SIP trunking "hub" connecting to Voice Portal, Communication Manager, the Acme Packet SBC, and other applications such as Avaya Modular Messaging. Avaya Voice Portal SIP Connections, Communication Manager SIP trunks, and Acme Packet SBC "session-agents" are configured to terminate at Session Manager.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows to Avaya Voice Portal and subsequent Voice Portal call transfers to Communication Manager skills and agents. Additional test objectives are listed in **Section 8**. See **Section 2.2** for an overview of key call flows and **Section 11** for detailed Wireshark verifications of key call flows.

1.2. Support

Verizon Business customers may obtain support for the Verizon Business IPCC service by visiting online support at http://www.verizonbusiness.com/us/customer/.

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

1.3. Known Limitations

- Avaya Voice Portal 5.1 requires the host portion of the To header of an inbound SIP INVITE message match the SIP domain configured in Voice Portal. Since Verizon IP Toll Free sends the outside (public) IP Address of the SBC in the To header, the Acme Packet SBC can be used to manipulate the To header so that the To header received by Voice Portal contains the SIP domain configured in Voice Portal. While the use of this SIP header manipulation does not present a problem, it is listed here to highlight the necessity of the SIP header manipulation shown in **Section 7**.
- For Avaya Voice Portal 5.1 to preserve the original caller id received from Verizon for an INVITE-based transfer to Communication Manager (e.g., bridged transfer, consultative transfer using INVITE with Replaces), the SBC can be used to strip the "+" that precedes the calling number. A product change request has been entered against Voice Portal so that future versions of Voice Portal can preserve the caller ID for INVITE-based transfers, even if the original caller ID begins with a "+", as is the case for Verizon IP Toll Free and Verizon IP IVR services. While the use of this SIP header manipulation does not present a problem, it is listed here to highlight the necessity of the SIP header manipulation shown in **Section 7**.
- After Avaya Voice Portal answers an inbound toll-free call with a self-service application, Voice Portal can transfer the call to a Communication Manager resource such as a skill and contact center agent, if desired by the caller or application. Depending on the configuration, after the transfer, the PSTN caller may not hear ring back tone while the transferred-to target (e.g., an agent) is ringing. (Acme Packet is investigating this issue via PD00017338.) To ensure that ring back tone is heard, the call vector associated with the transferred-to Vector Directory Number (VDN) can include an announcement step prior to the "queue-to skill" step as described in **Section 5.10**. Again, this announcement configuration does not present a problem, but is listed here to call attention to a benefit of having the transferred-to vector answer the call prior to ringing an agent.
- An intermittent problem was initially observed with Verizon IP IVR Service (i.e., problem was never observed with Verizon IP Toll Free). Later in the test cycle, after a Verizon network component upgrade, the problem could not be reproduced. The problem originally seen was as follows. If an Avaya Voice Portal transfer (occurring on the inside interface of the SBC) results in two consecutive INVITE messages sent to the Verizon IP IVR Service, the Verizon IP IVR Service at times responded to the second INVITE with a 500 Server Internal Error. If this response is seen, the call will remain connected for a time, but will be disconnected by Verizon within 2 minutes. This problem can be avoided if Avaya Voice Portal transfers the call to a VDN that includes an announcement step prior to the "queue-to skill" step as described in **Section 5.10**. For consultative transfers to a VDN involving IP IVR Service, the Voice Portal configurable behavior for Consultative Transfer can be set to REFER to avoid this problem, as shown in **Section 4.4**.
- While it is possible to disable the use of G.729 for outbound calls from Voice Portal, incoming calls to Voice Portal that list G.729 as the first audio codec will be answered by Voice Portal using G.729. Verizon IP Toll Free always lists G.729a as the first audio codec on the production circuit used for testing. Therefore, with the configuration shown in these Application Notes, when using Verizon IP Toll Free service, G.729a will always be used for

incoming calls, while the call is connected to Voice Portal. Verizon IP IVR offers only G.711MU on the production circuit used for testing, and G.711MU was tested with Verizon IP IVR. An enhancement request to Voice Portal (wi00830274) has been entered to ask for greater configurability in the codec selection for an incoming call to Voice Portal. Although not shown in these Application Notes, if it is desired that inbound calls from Verizon use G.711 while in Voice Portal, the SBC can be used to strip G.729a from the SDP of the INVITE from Verizon, so that Voice Portal receives only G.711 in the SDP offer.
- Verizon Business IPCC Services suite does not support fax.
- Verizon Business IPCC Services suite does not support History Info or Diversion Headers.

2. Sample 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 can consist of one or more Media Processing Platform (MPP) servers and a Voice Portal Management System (VPMS) server. Avaya Voice Portal also supports MPP and VPMS co-resident on the same server. The co-resident MPP and VPMS configuration was used in the sample configuration.
- Avaya Aura® Communication Manager provides the enterprise voice communications services. In the sample configuration, Communication Manager runs on an Avaya S8800 Server. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for enterprise voice communications. For example, announcements and call progress tones can be sourced from the media gateway. In the sample configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "office" phones are represented with Avaya 9600 Series IP Telephones running H.323 and SIP software. Avaya 9600 Series IP Telephones running H.323 firmware are used as contact center agents to staff Communication Manager skills.
- Avaya Aura® Session Manager routes SIP traffic within the enterprise. In the sample configuration, Session Manager runs on an Avaya S8800 Server.
- Avaya Aura® System Manager manages Session Manager and Communication Manager. In the sample configuration, System Manager runs on an Avaya S8800 Server.
- The Acme Packet Net-Net provides SIP Session Border Controller (SBC) functionality between the Verizon Business IPCC service suite and the enterprise internal network. The Acme Packet Net-Net 4250 will be referred to as the Acme Packet SBC in these Application Notes. The solution is extensible to other Acme Packet Net-Net models, including the 3800 and 4500.
- 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.

[1] During testing, the Apache Tomcat Application Server, Avaya MPP and VPMS software were installed on the same server. Separate servers may be used.

- Optionally, a Speech Server may be used for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) capabilities. The focus of these Application Notes is protocol compatibility, and a Speech Server was not utilized.

Figure 1 illustrates the sample configuration. As also noted in **Section 8.1**, the Verizon IP Toll Free and IP IVR numbers listed with a "*" are the numbers illustrated via screens and traces in these Application Notes. The sample configuration is similar to the configuration previously documented in reference [JRR-VZIPCC], with Avaya Voice Portal added.

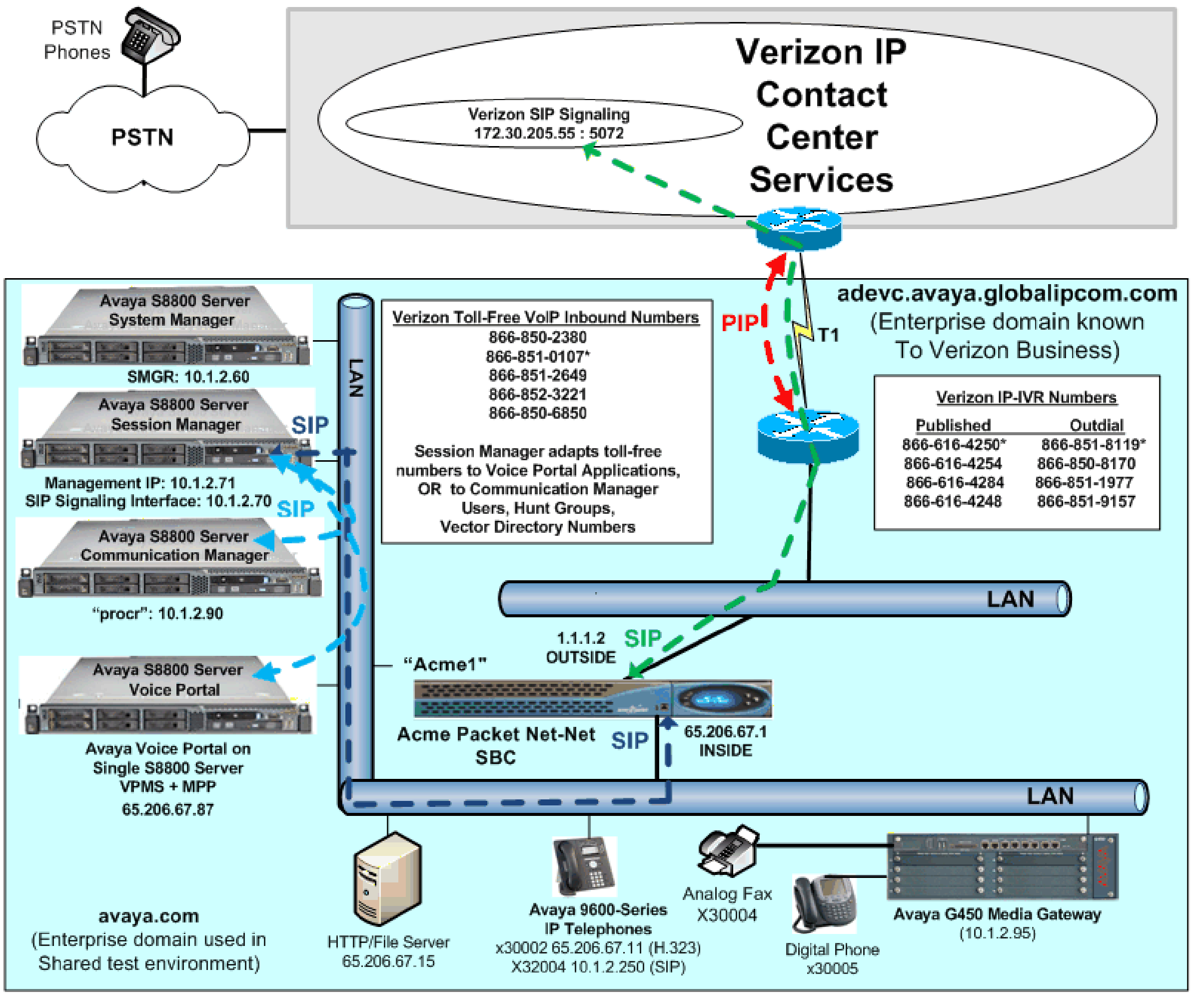


Figure 1: Sample Configuration

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 values appropriate for their specific configurations.

Component	Illustrative Value in these Application Notes
Avaya Voice Portal	
MPP Server IP Address	65.206.67.87
Number Triggering Voice Portal Application	44000 as illustrated. Other number to application mappings can be configured similarly.
Avaya Aura® Session Manager	
Signaling Interface	10.1.2.70
Avaya Aura® Communication Manager	
Processor Ethernet IP Address	10.1.2.90
Avaya G450 Media Gateway	10.1.2.95
Vector Directory Number (VDN) Extensions	36880 illustrated, others as needed
Skill (Hunt Group) Extensions	36680 illustrated, others as needed
Agent Extensions (Agent Login-Ids)	46880 illustrated, others as needed
Telephone Extensions	3000x
Acme Packet Net-Net SBC	
IP Address of "Outside" Interface (towards Verizon Business)	1.1.1.2
IP Address of "Inside" Interface (towards Avaya elements)	65.206.67.1
Verizon Business IP Contact Center Service	
Border Element IP Address and Port	172.30.205.55:5072
Digits Passed to Avaya Voice Portal	As illustrated, 866-851-0107 (IP Toll Free) and 866-851-8119 (IP IVR) mapped to 44000 by Session Manager Adaptation. Other mappings of Verizon numbers to Voice Portal application triggers can be configured similarly.

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how inbound Verizon Business IPCC calls are handled by Avaya Voice Portal, several call flows are described in this section. These call flows are illustrated via detailed Wireshark analysis in **Section 11** for both Verizon IP Toll Free and Verizon IP IVR.

One type of 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 a Verizon Business IPCC service number.
2. The PSTN routes the call to the Verizon Business IPCC service network.
3. The Verizon Business IPCC 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 Session Manager.
5. Session Manager routes the call to Avaya Voice Portal.
6. 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.
7. In this scenario, the application sufficiently meets the caller's needs or requests, and thus the call does not need to be transferred to Communication Manager.

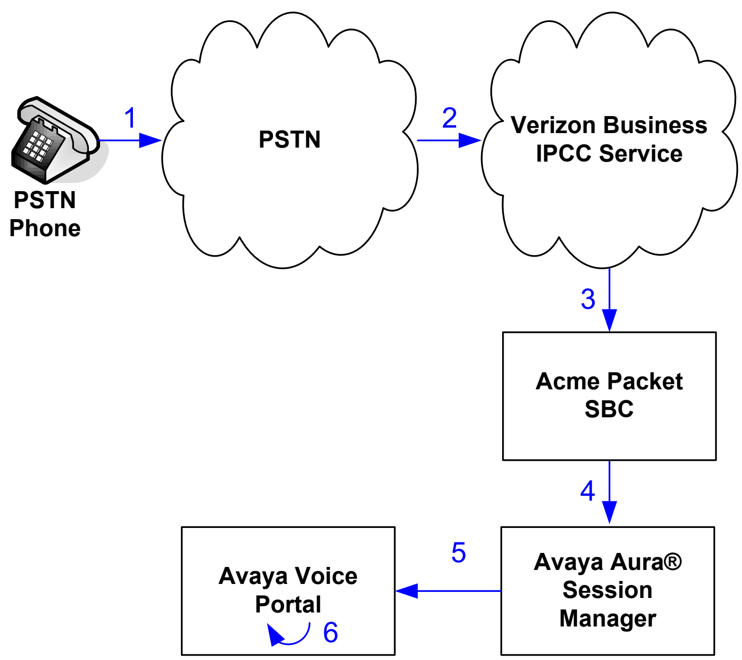


Figure 2: Inbound Call Handled Entirely by Avaya Voice Portal

Another type of call scenario, illustrated in **Figure 3**, is an inbound call arriving on Avaya Voice Portal and subsequently transferred by Voice Portal to a Communication Manager skill via a Communication Manager Vector Directory Number (VDN). Although the Voice Portal application logic can in general check for agent availability prior to transferring the call, in this case, assume that Voice Portal transfers the call to the Communication Manager VDN without first checking whether agents are available.

The steps depicted below are intended to be a high-level representation of the call flow. For detailed Wireshark illustrations of the SIP signaling for both Voice Portal blind and consultative transfer scenarios, see **Section 11**.

1. Same as the first five steps from the first call scenario.
2. In this scenario, the application is not sufficient to meet the caller's needs or requests, and thus the call needs to be transferred to a Communication Manager agent via Session Manager. Avaya Voice Portal instructs the Acme Packet SBC to transfer the inbound call to a Communication Manager skill.
3. The Acme Packet SBC transfers the inbound call to the aforementioned skill on Communication Manager via Session Manager.
4. An agent becomes available.
5. Communication Manager routes the call to the agent.

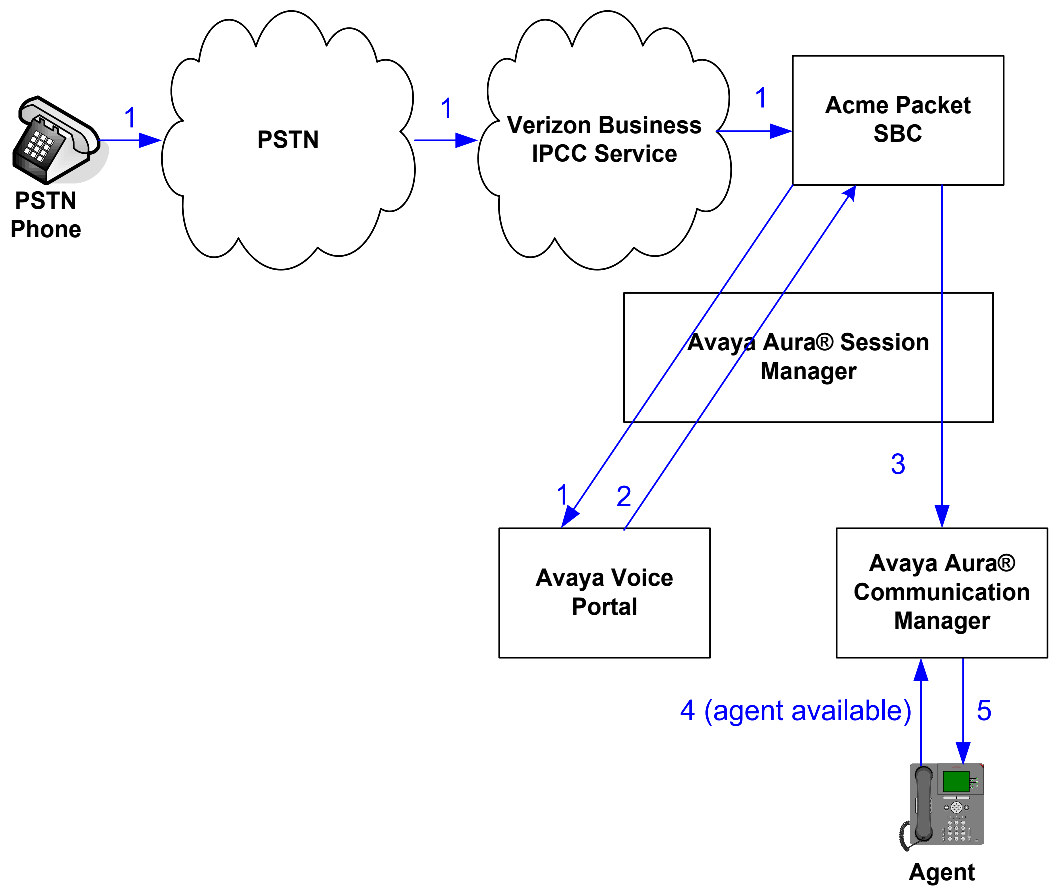


Figure 3: Inbound Call Transferred by Avaya Voice Portal to Communication Manager

3. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Voice Portal 5.1 Voice Portal Management System (VPMS) 5.1.0.0.4201 Media Processing Platform (MPP) Application Server 5.1.0.0.4206
Avaya S8800 Server	Avaya Aura® Communication Manager R6.0 (345.0 + patch 18444)
Avaya S8800 Server	Avaya Aura® System Manager R6.0 SP1
Avaya S8800 Server	Avaya Aura® Session Manager (6.0.1.0.601013)
Avaya G450 Media Gateway	30.13.2
Avaya 9630 IP Telephone (H.323)	3.1.1
Avaya 2420 Digital Telephone	---
Apache Tomcat Application Server	6.0.18
Acme Packet Net-Net 4250[2]	SC6.2.0m3p5.xz

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], [3], and [4] 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. References to these 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 (URI) is matched against the administered called numbers. If a match is found, then the corresponding application is accessed to handle the call.

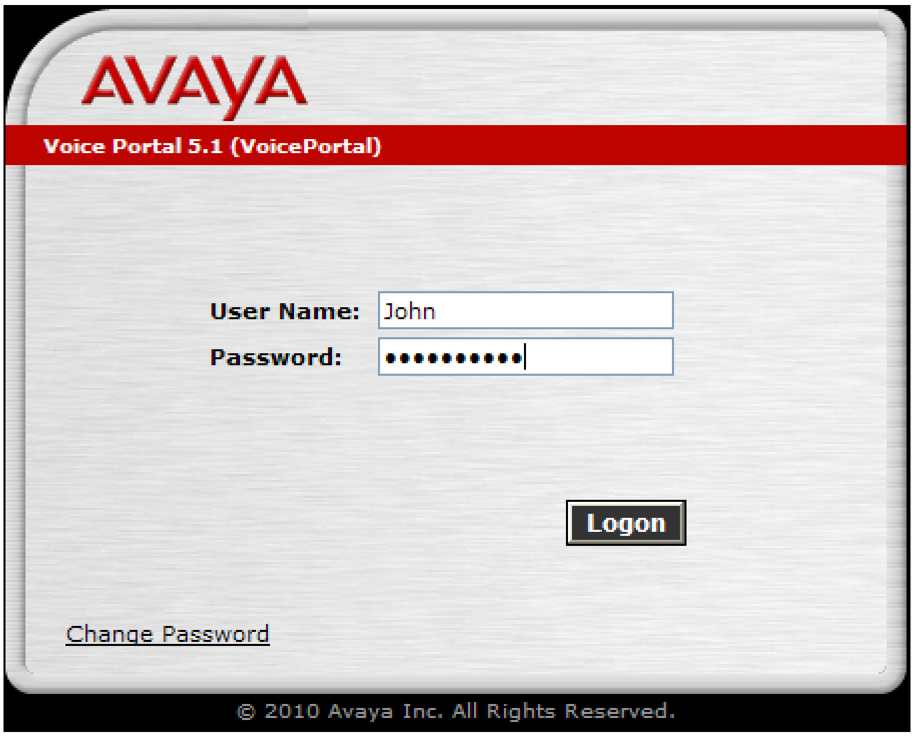
For the sample configuration described in these Application Notes, a simple VXML test application was used to exercise various SIP call flow scenarios with the Verizon Business IPCC services. In production, enterprises can develop their own VXML and/or CCXML applications

[2] Although an Acme Packet Net-Net 4250 was used in the sample configuration, the 3800, 4500, and 9200 platforms may also be used.

to meet 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 [4] for further details if necessary.

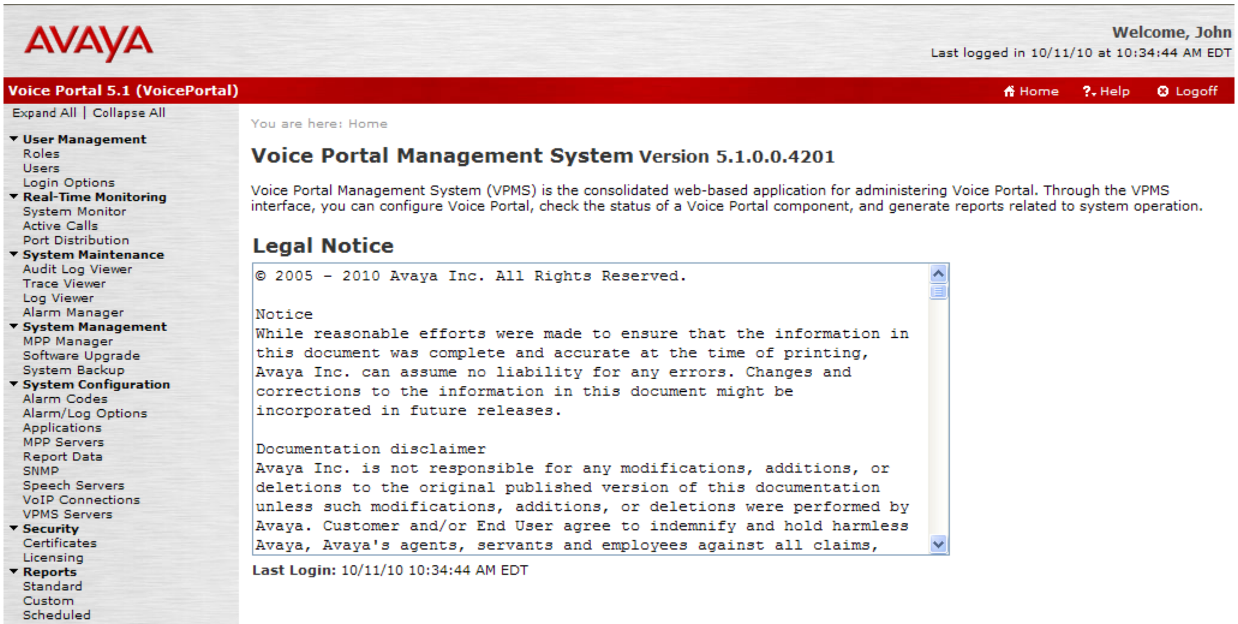
4.2. Log In to Voice Portal

Launch a web browser, enter https://65.206.67.87/VoicePortal in the URL, where 65.206.67.87 is the IP Address of Voice Portal Management System (VPMS) in the sample configuration. Enter the appropriate credentials as shown in the example screen below. Click **Logon**.



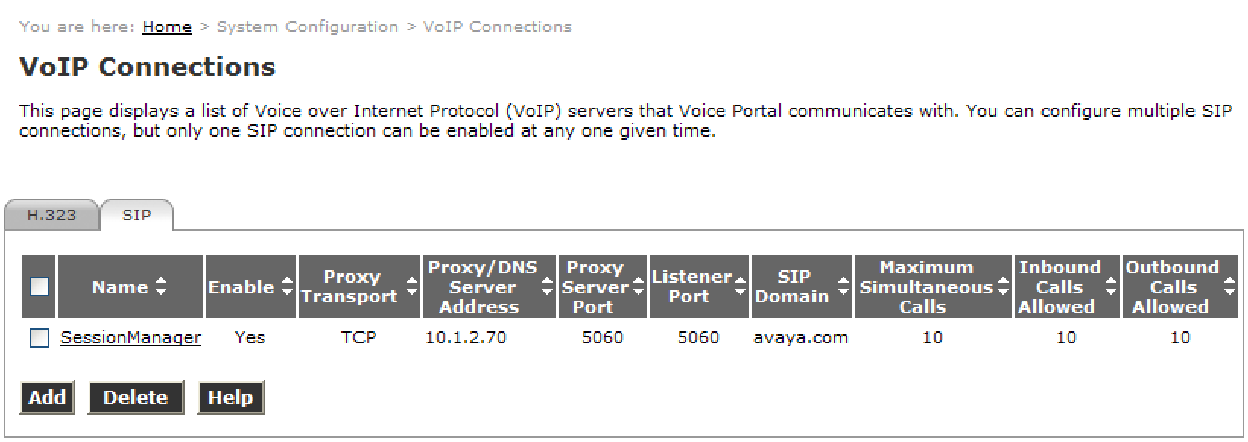
4.3. Voice Portal Home Screen

After logging in successfully, the **Home** screen will appear, as shown below.



4.4. VoIP Connection

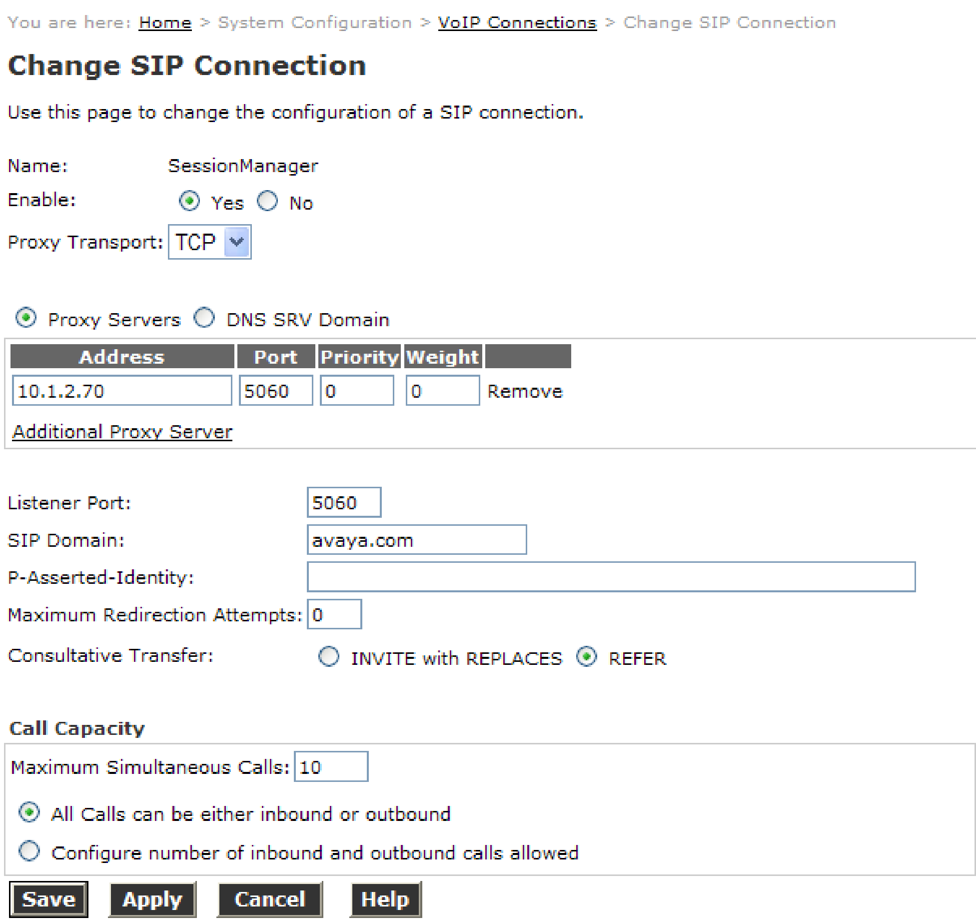
This section illustrates the procedure for administering a SIP connection from Voice Portal to Session Manager. From the left pane of the **Home** screen shown in **Section 4.3**, expand **System Configuration → VoIP Connections**. In the resultant screen, select the **SIP** tab. To add a new SIP connection, click **Add**. To view or edit an existing connection, click the **Name** of the connection. In the example screen shown below, a SIP connection named "SessionManager" had been added previously.



	Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls	Inbound Calls Allowed	Outbound Calls Allowed
	SessionManager	Yes	TCP	10.1.2.70	5060	5060	avaya.com	10	10	10

After clicking **Add** to add a new connection, or the name of an existing connection, a screen like the following is displayed. Configure the SIP connection to Session Manager as follows, and then click **Save**:

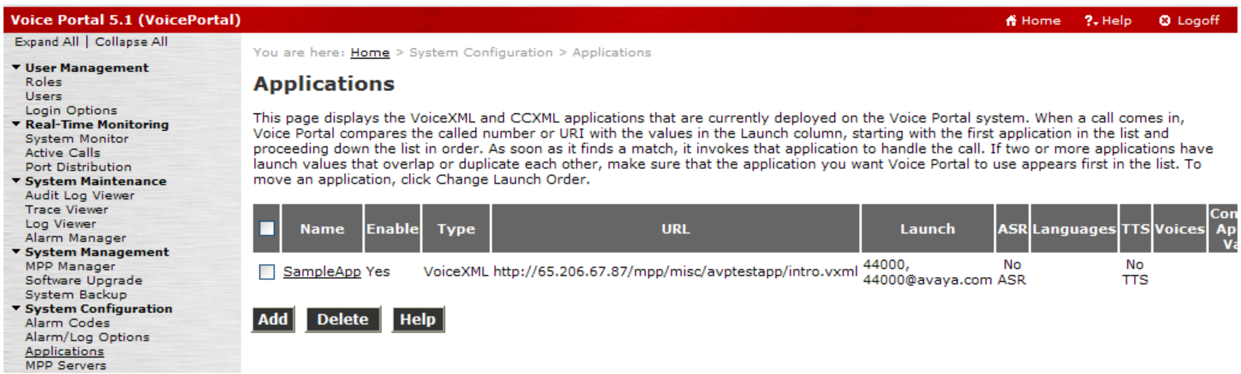
- **Name** – When adding, enter a descriptive name, such as "SessionManager".
- **Enable** – Select "**Yes**".
- **Proxy Transport** – Select "**TCP**".
- Click the **Proxy Servers** radio button**.**
- **Proxy Server Address** – Enter the IP address of the Session Manager SIP signaling interface, such as "10.1.2.70" in the sample configuration.
- **Proxy Server Port** – Enter "**5060**".
- **Listener Port** – Enter "**5060**"
- **SIP Domain** – Enter the enterprise SIP domain, such as "avaya.com"
- **Maximum Simultaneous Calls** – Enter the number of calls this SIP connection can handle, taking capacity and license considerations into account.
- **Consultative Transfer**: "INVITE with REPLACES" and "REFER" were tested successfully with Verizon IP Toll Free Service. The screen shows "REFER" selected because issues were observed when INVITE with REPLACES was used with the Verizon IP IVR Service. See **Section 1.3** and **Section 11**.



4.5. Application References

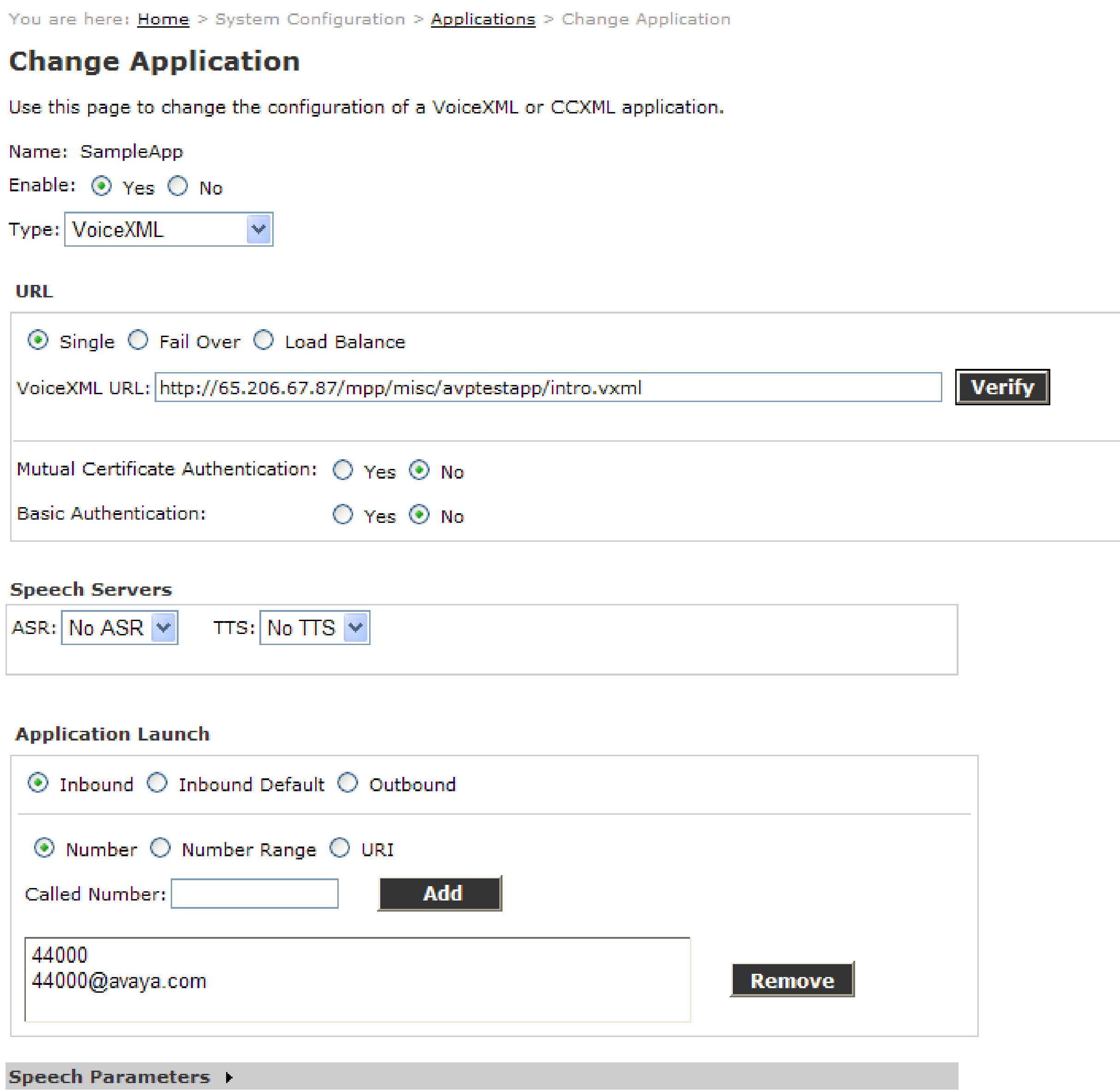
This section illustrates the procedure for administering a reference to a VXML and/or CCXML application residing on an application server. In the sample configuration, the applications were co-resident on one Voice Portal server, with IP Address 65.206.67.87.

From the left pane of the **Home** screen shown in **Section 4.3**, expand **System Configuration → Applications**. To add a new application, click **Add**. To view or edit an existing application, click the **Name** of the application. In the example screen shown below, a sample application named "SampleApp" had been added previously.



After clicking **Add** to add a new application, or the name of an existing application, a screen like the following is displayed. Configure an application as follows, and then click **Save.**

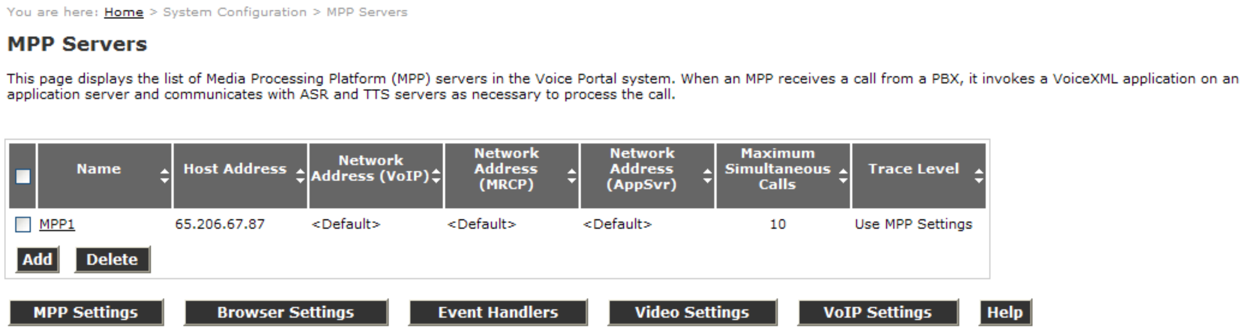
- **Name** – Enter a descriptive name.
- **Enable** – Select "**Yes**".
- **Type** – Select "**VoiceXML**", "**CCXML**", or "**CCXML/VoiceXML**" according to the application type.
- **VoiceXML** and/or **CCXML URL** – Enter the necessary URL(s) to access the VXML and/or CCXML application(s) on the application server. In the sample screen below, the Voice Portal test application on the single server Voice Portal is referenced.
- **Speech Servers ASR** and **TTS** – Select the appropriate ASR and/or TTS servers as necessary.
- **Application Launch** – Select "**Inbound**".
- Select the **Number** radio button to add a number or the **URI** radio button to add a URI.
- **Called Number** – Enter the number to match against an inbound SIP INVITE message, and click on "**Add**". In the sample configuration illustrated in these Application Notes, the dialed Verizon IP Toll Free number 866-851-0107 and the Verizon IP IVR outdial number 866-851-8119 were adapted by Session Manager to 44000. Repeat to define additional called party numbers as needed. Inbound Verizon Business toll-free service calls with these called party numbers will be handled by the application defined in this section.



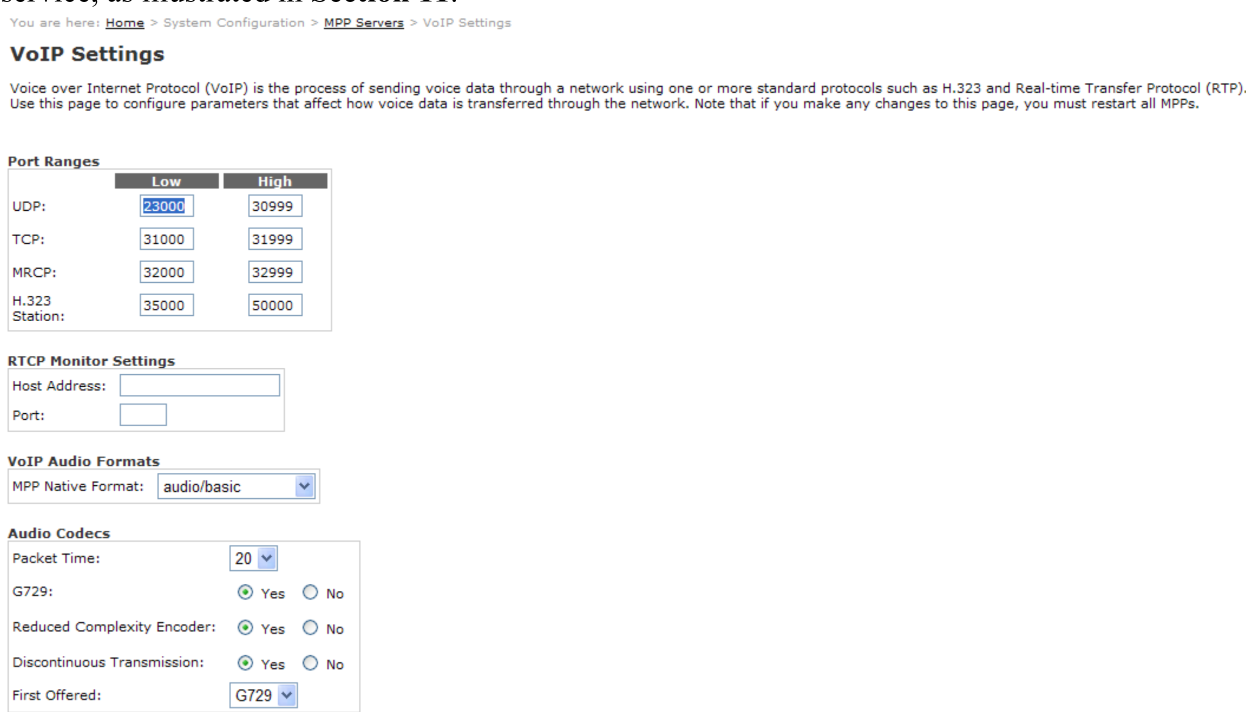
Add additional application references using the procedures in this section as needed.

4.6. MPP Servers and VoIP Settings

This section illustrates the procedure for viewing or changing the MPP Settings. From the left pane of the **Home** screen shown in **Section 4.3**, expand **System Configuration → MPP Servers.** In the sample configuration, the MPP Server is co-resident on a single server with the Voice Portal Management System (VPMS), and therefore the same **Host Address** "65.206.67.87" is used for both the MPP and VPMS.



Click the **VoIP Settings** button to view or change Voice over IP parameters. The following screen illustrates the default configuration retained in the sample configuration. With this configuration, inbound calls from Verizon IP Toll Free service that remain in Voice Portal for self-service will use the G.729a codec, as illustrated in **Section 11**. Inbound calls from Verizon IP Toll Free to Voice Portal that are subsequently transferred to Communication Manager may use G.729a or G.711MU, depending on the Communication Manager codec set configuration. On the production circuit used to verify these Application Notes, inbound calls from Verizon IP IVR service always use the G.711MU codec, since G.729a was not offered by Verizon IP IVR service, as illustrated in **Section 11**.



4.7. Configuring RFC2833 Event Value Offered by Voice Portal

The configuration change example noted in this section is not required for any of the call flows illustrated in these Application Notes. For incoming calls from Verizon services to Voice Portal, Verizon specifies the value 101 for the RFC2833 telephone-events that signal DTMF digits entered by the user. When Voice Portal answers, the SDP from Voice Portal matches this Verizon offered value.

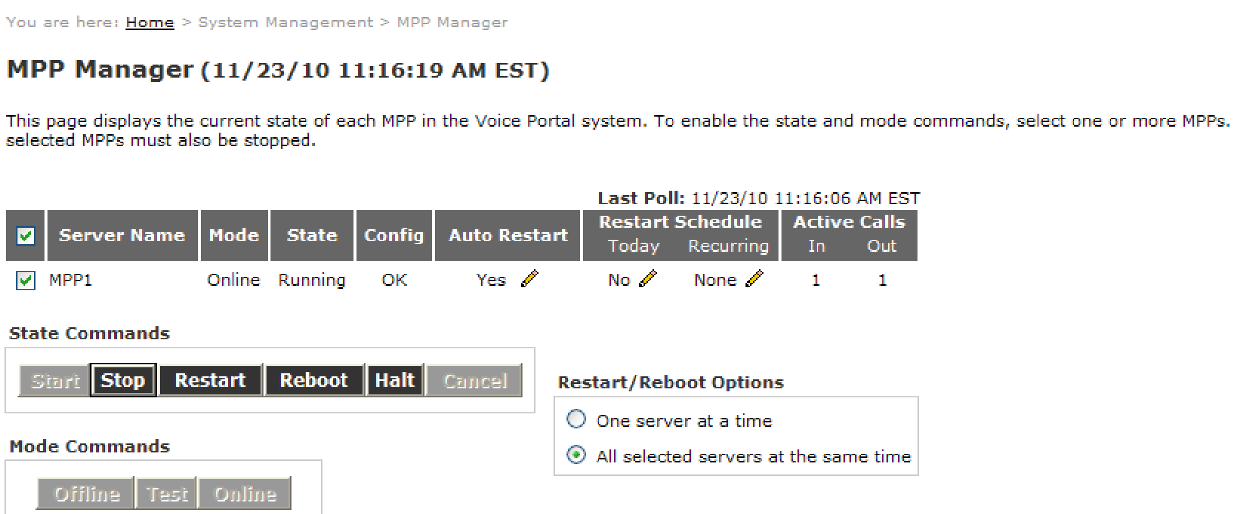
When Voice Portal sends an INVITE with SDP as part of an INVITE-based transfer (e.g., bridged transfer), Voice Portal offers the SDP. By default, Voice Portal specifies the value 127 for the RFC2833 telephone-events. Optionally, the value that is offered by Voice Portal can be changed, and this section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

Access Voice Portal via the command line interface.
Navigate to the following directory: /opt/Avaya/VoicePortal/MPP/config
Edit the file mppconfig.xml.
Search for the parameter "mpp.sip.rfc2833.payload". If there is no such parameter specified, add a line such as the following to the file, where the value 101 is the value to be used for the RFC2833 events. If the parameter is already specified in the file, simply edit the value assigned to the parameter.

```
<parameter name="mpp.sip.rfc2833.payload">101</parameter>
```

In the verification of these Application Notes, the line was added directly above the line where the sip.session.expires parameter is configured.

After saving the file with the change, restart the MPP server for the change to take effect. As shown below, the MPP may be restarted using the **Restart** button available via the Voice Portal GUI at **System Management → MPP Manager.**



5. Avaya Aura® Communication Manager

This section illustrates the Communication Manager configuration used in the verification of these Application Notes. The example configuration uses SIP signaling via the "Processor Ethernet" of the Avaya S8800 Servers to Session Manager. In configurations that use an Avaya G650 Media Gateway, it is also possible to use an Avaya C-LAN in an Avaya G650 Media Gateway for SIP signaling to Session Manager.

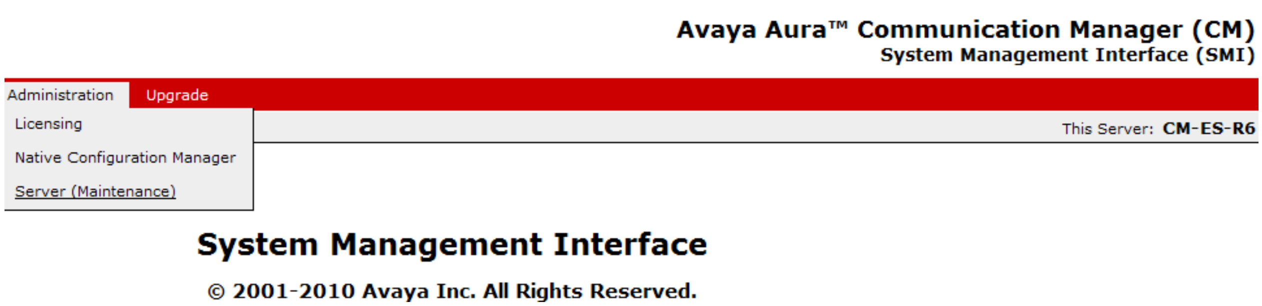
These Application Notes assume that basic Communication Manager administration has already been performed.

The Communication Manager configuration shown in this section builds off the Communication Manager configuration previously documented in reference [JRR-VZIPCC], with modest changes and additions.

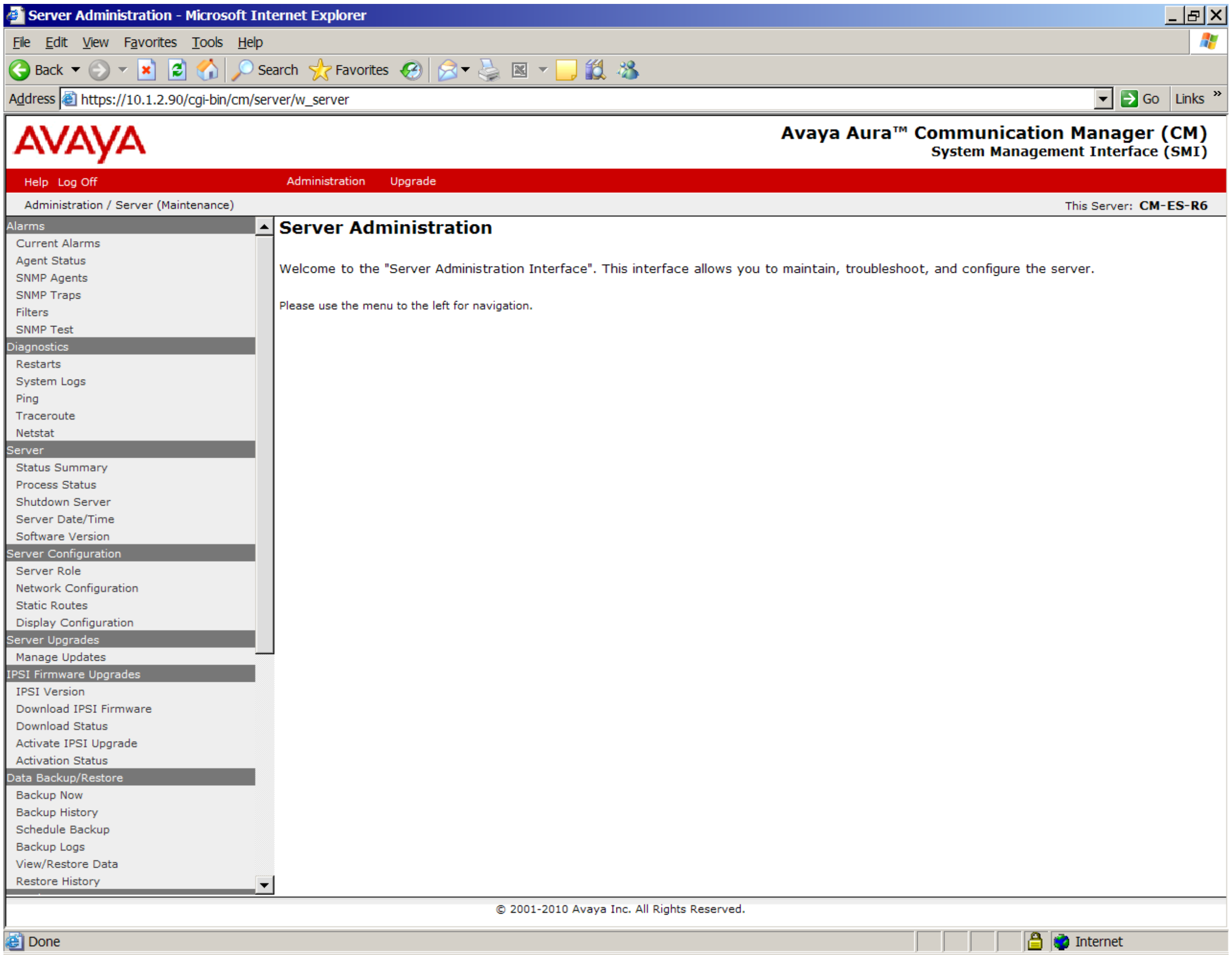
Except for the web configuration shown in **Section 5.1**, all remaining configuration is performed via the Communication Manager SAT interface of the Avaya S8800 Server. Screens are abridged for brevity in presentation.

5.1. Processor Ethernet Configuration on S8800 Server

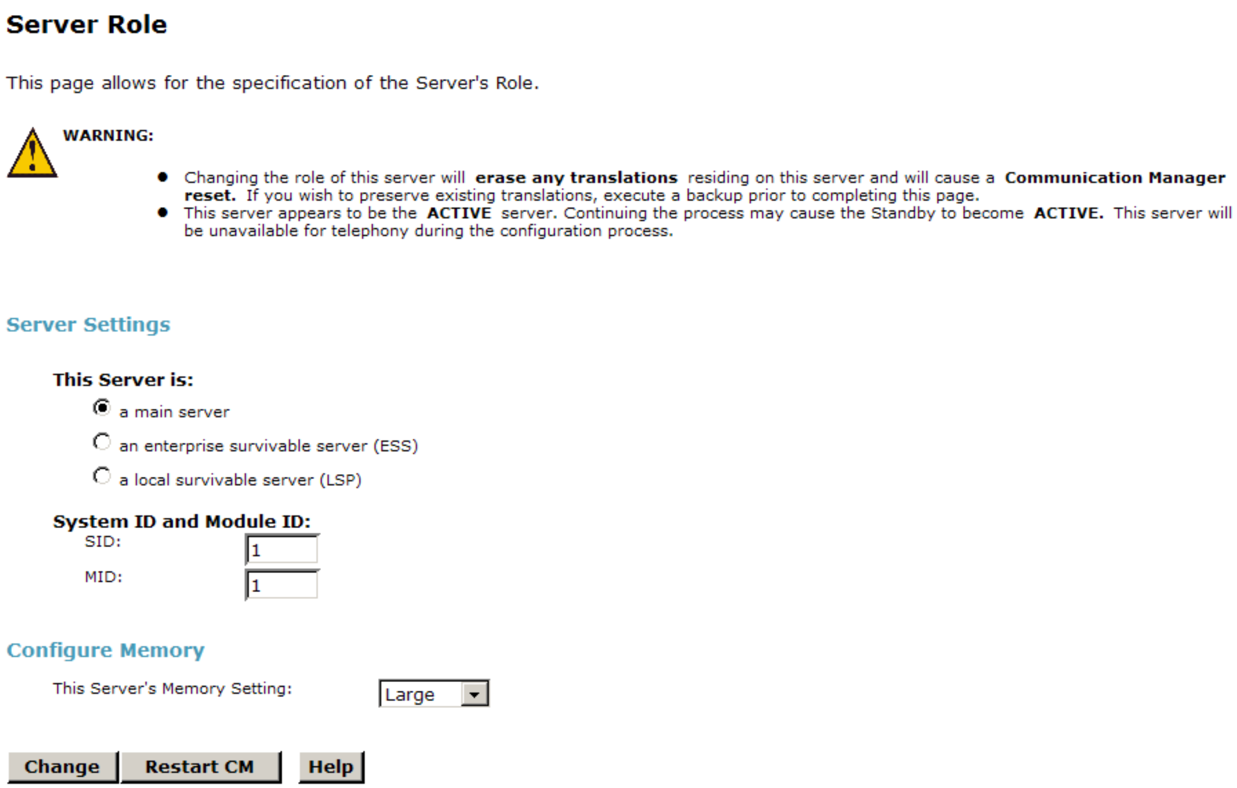
The screens in this section illustrate a previously completed configuration. Consult product documentation for further procedural guidance.
The S8800 Server can be accessed via a web interface in an internet browser. In the sample configuration, enter http://10.1.2.90 and log in with appropriate credentials (not shown). From the System Management Interface screen, select **Administration → Server (Maintenance)** as shown below.



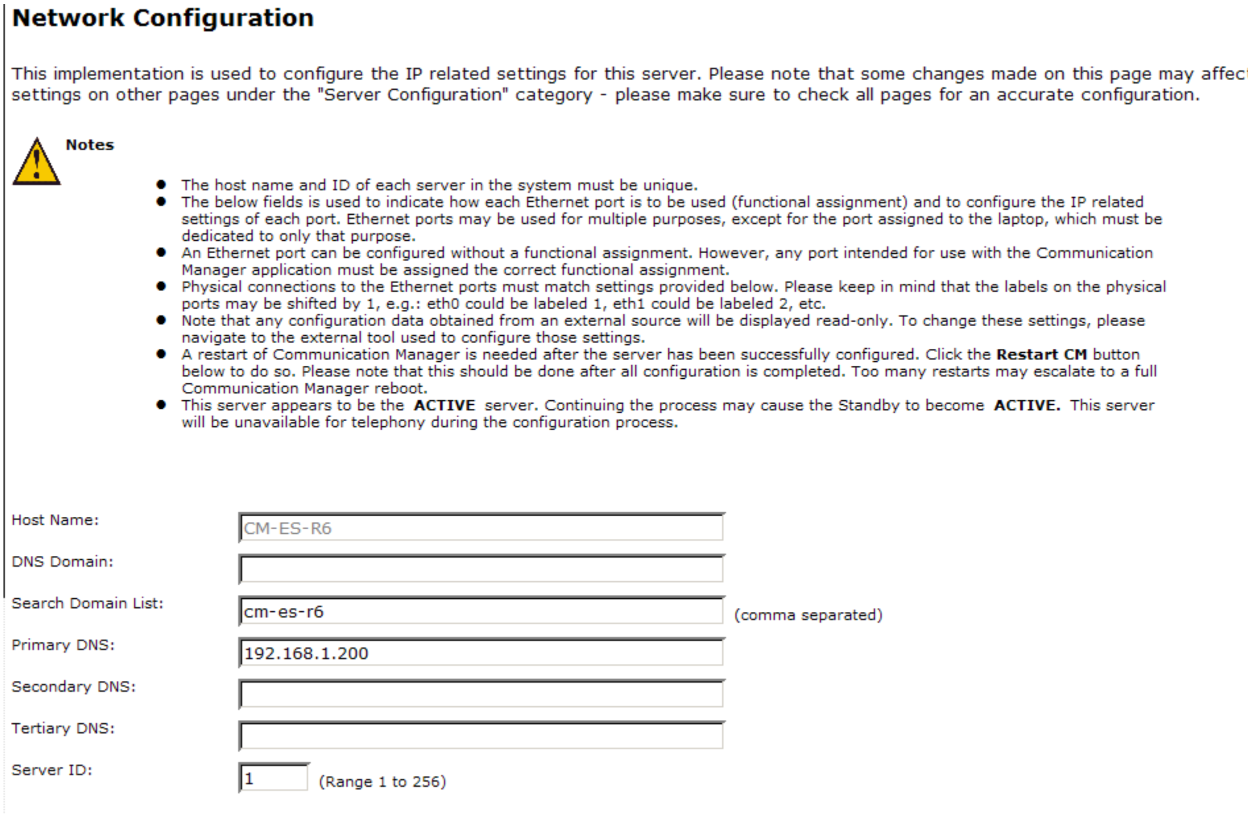
The resulting **Server Administration** screen is shown below.



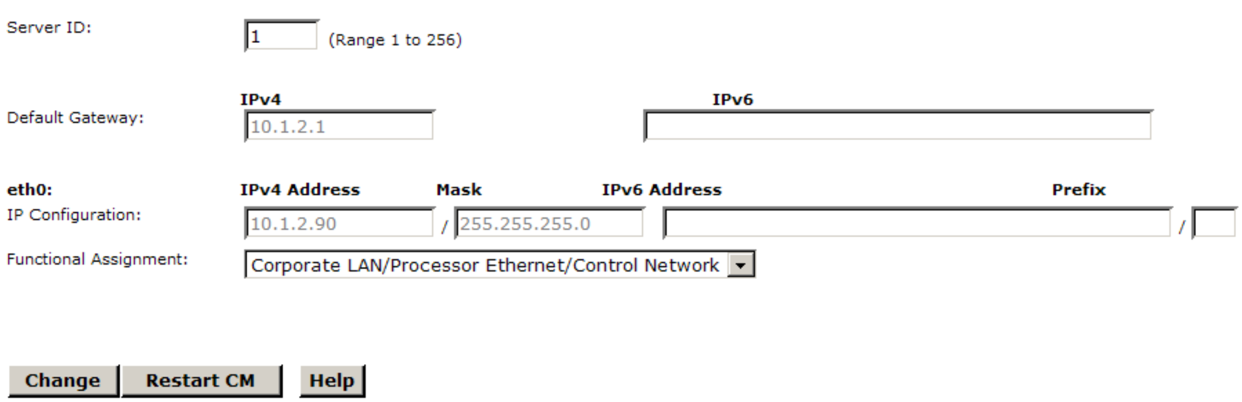
Under Server Configuration, select **Server Role** to view or configure the server role. In the sample configuration, the Avaya S8800 server is **a main server**, as shown below.



Under Server Configuration, select **Network Configuration** to view the network configuration. The following screen shows the upper portion of the **Network Configuration**.



Scrolling down, the following screen shows the lower portion of the Network Configuration. Note that the **IPv4 Address** of the server is 10.1.2.90, and that the **Functional Assignment** drop-down has assigned the **Corporate LAN/Processor Ethernet/Control Network** to the same "eth0" interface.



5.2. Verify Licensed Features

The Communication Manager license file controls customer capabilities. Contact an authorized Avaya representative for assistance if a required feature needs to be enabled.

On **Page 2** of the ***display system-parameters customer-options*** form, verify that the **Maximum Administered SIP Trunks** is sufficient for the combination of trunks to the Verizon Business IPCC Services and any other SIP applications. Each call from the Verizon Business IPCC Services to a non-SIP endpoint on Communication Manager uses one SIP trunk for the duration of the call. Each call from Verizon Business IPCC Services to a SIP endpoint uses two SIP trunks for the duration of the call. Of course, a call from the Verizon Business IPCC Service that is delivered from Session Manager to Voice Portal, and remains in Voice Portal for self-service, does not consume a Communication Manager SIP trunk.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 100
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 146
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
```

On **Page 3** of the **System-Parameters Customer-Options** form, verify that **ARS** is enabled.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y           Audible Message Waiting? y
        Access Security Gateway (ASG)? n               Authorization Codes? y
        Analog Trunk Incoming Call ID? y                        CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                          CAS Main? n
Answer Supervision by Call Classifier? y                 Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                       DCS (Basic)? y
          ASAI Link Core Capabilities? n                 DCS Call Coverage? y
          ASAI Link Plus Capabilities? n                DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                           DS1 MSP? y
                                ATMS? y               DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

On **Page 4** of the **System-Parameters Customer-Options** form, verify that **IP Trunks**, **IP Stations**, and **ISDN-PRI** features are enabled. If the use of SIP REFER messaging by Communication Manager will be required for the call flows described in **Section 2.2** of reference [JRR-VZIPCC], verify that the **ISDN/SIP Network Call Redirection** feature is enabled. In these Application Notes, Voice Portal will generate SIP REFER messages for inbound calls from Verizon IPCC that are transferred to Communication Manager, but Communication Manager will not generate SIP REFER messages in vector processing as was the case in reference [JRR-VZIPCC].

```
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? y
       Enterprise Wide Licensing? n                              ISDN-PRI? y
              ESS Administration? y           Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                  Malicious Call Trace? y
     External Device Alarm Admin? y              Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y             Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

On **Page 5** of the **System-Parameters Customer-Options** form, verify that the **Private Networking** and **Processor Ethernet** features are enabled if these features will be used, as is the case in the sample configuration.

```
display system-parameters customer-options                    Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                     Multiple Locations? n
                                           System Management Data Transfer? n
          Personal Station Access (PSA)? y              Tenant Partitioning? y
                        PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? y               Time of Day Routing? y
                       Posted Messages? y      TN2501 VAL Maximum Capacity? y
                                                       Uniform Dialing Plan? y
                    Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? y
                    Processor Ethernet? y                Wideband Switching? y
```

On **Page 6** of the **System-Parameters Customer-Options** form, verify that any required call center features are enabled. In the sample configuration, **vectoring** and **Expert Agent Selection** are used.

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 5.0

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

5.3. Dial Plan

In the sample configuration, the Avaya CPE environment uses five digit local extensions, such as 3xxxx and 4xxxx. Trunk Access Codes (TAC) are 3 digits in length and begin with 1. The Feature Access Code (FAC) to access ARS is the single digit 9. The Feature Access Code (FAC) to access AAR is the single digit 8. The dial plan illustrated here is not intended to be prescriptive; any valid dial plan may be used. The dial plan is modified with the **change dialplan analysis** command as shown below.

```
change dialplan analysis                                        Page   1 of  12
                          DIAL PLAN ANALYSIS TABLE
                                Location: all            Percent Full: 2

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   0           3   fac
   1           3   dac
   2           5   ext
   3           5   ext
   4           4   ext
   5           5   ext
   6           3   fac
   60          5   ext
   7           5   ext
   8           1   fac
   9           1   fac
   *           2   fac
   #           2   fac
```

5.4. Node Names

Node names are mappings of names to IP Addresses that can be used in various screens. The following abridged "change node-names ip" output shows relevant node-names in the sample configuration. As shown in bold, the node name for Session Manager is "SM1" with IP Address 10.1.2.70. The node name and IP Address (10.1.2.90) for the Processor Ethernet "procr" appears automatically due to the web configuration in **Section 5.1**.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM1                 10.1.2.70
procr               10.1.2.90
```

5.5. IP Interface for procr

The "add ip-interface procr" or "change ip-interface procr" command can be used to configure the Processor Ethernet (PE) parameters. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the PE for SIP Trunk Signaling, observe that the Processor Ethernet will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

```
change ip-interface procr                                         Page   1 of   2
                                  IP INTERFACES

                  Type: PROCR
                                                       Target socket load: 1700

      Enable Interface? y                          Allow H.323 Endpoints? y
                                                    Allow H.248 Gateways? y
        Network Region: 1                            Gatekeeper Priority: 5

                                 IPV4 PARAMETERS
             Node Name: procr                    IP Address: 10.1.2.90

           Subnet Mask: /24
```

5.6. Network Regions for Gateway, Telephones

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G450 Media Gateway is in region 1. To provide testing flexibility, network region 4 was associated with other components used specifically for the Verizon testing.

Non-IP telephones (e.g., analog, digital) derive network region and location configuration from the Avaya gateway to which the device is connected. The following display command shows that media gateway 1 is an Avaya G450 Media Gateway configured for network region 1. It can also be observed that the "Controller IP Address" is the Avaya S8800 processor Ethernet (10.1.2.90), and that the gateway IP Address is 10.1.2.95. These fields are not configured in this screen, but rather simply display the current information for the gateway.

```
change media-gateway 1                                          Page   1 of   2
                             MEDIA GATEWAY 1

                    Type: g450
                    Name: G450 Evolution Srvr
               Serial No: 08IS43202588
            Encrypt Link? y                  Enable CF? n
          Network Region: 1                   Location: 1
                                             Site Data:
           Recovery Rule: none

              Registered?  y
   FW Version/HW Vintage: 30 .13 .2  /1
       MGP IPV4 Address: 10.1.2.95
       MGP IPV6 Address:
   Controller IP Address: 10.1.2.90
             MAC Address: 00:1b:4f:03:57:b0
```

The following screen shows page 2 for media gateway 1. The gateway has an **MM712** media module supporting Avaya digital phones in slot **v3**, an **MM714** supporting analog devices in slot **v5**, and the capability to provide announcements and music on hold via "gateway-announcements" in logical slot **v9**.

```
change media-gateway 1                                          Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g450

Slot   Module Type            Name                     DSP Type  FW/HW version
 V1:                                                   MP80      45   3
 V2:
 V3:   MM712                  DCP MM
 V4:
 V5:   MM714                  ANA MM
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

IP telephones can be assigned a network region based on an IP address mapping. The following screen illustrates a subset of the IP network map configuration used to verify these Application Notes. If the IP address of a registering IP Telephone does not appear in the ip-network-map, the phone is assigned the network region of the "gatekeeper" (e.g., CLAN or PE) to which it registers. When the IP address of a registering IP telephone is in the ip-network-map, the phone is assigned the network region assigned by the form shown below. For example, the IP address 65.206.67.11 would be mapped to network region 4, based on the bold configuration below. In production environments, different sites will typically be on different networks, and ranges of IP Addresses assigned by the DHCP scope serving the site can be entered as one entry in the network map, to assign all telephones in a range to a specific network region.

```
change ip-network-map                                          Page   1 of  63
                               IP ADDRESS MAPPING

                                               Subnet Network     Emergency
 IP Address                                    Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM: 10.1.2.0                                /24    1      n
   TO: 10.1.2.255
 FROM: 65.206.67.0                             /24    4      n
   TO: 65.206.67.255
```

The following screen shows IP Network Region 4 configuration. In the shared test environment, network region 4 is used to allow unique behaviors for the Verizon test environment. In this example, codec set 4 will be used for calls within region 4. The shared Avaya Interoperability Test Lab environment uses the domain "avaya.com" (i.e., for network region 1 including the region of the processor ethernet "procr"). However, to illustrate the more typical case where the Communication Manager domain matches the enterprise CPE domain known to Verizon, the **Authoritative Domain** in the following screen is "adevc.avaya.globalipcom.com", the domain known to Verizon, as shown in **Figure 1**. Even with this configuration, note that the domain in the PAI header sent by Communication Manager to Session Manager will contain "avaya.com", the domain of the near-end of the Avaya signaling group. Session Manager will adapt "avaya.com" to "adevc.avaya.globalipcom.com" in the PAI header as needed.

```
change ip-network-region 4                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 4
Location:          Authoritative Domain: adevc.avaya.globalipcom.com
    Name: Verizon testing
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 4              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

The following screen shows the inter-network region connection configuration for region 4. The first bold row shows that network region 4 is directly connected to network region 1, and that codec set 4 will also be used for any connections between region 4 and region 1. For configurations where multiple remote gateways are used, each gateway will typically be configured for a different region, and this screen can be used to specify unique codec or call admission control parameters for the pairs of regions. If a different codec should be used for inter-region connectivity than for intra-region connectivity, a different codec set can be entered in the **codec set** column for the appropriate row in the screen shown below. Once submitted, the configuration becomes symmetric, meaning that network region 1, Page 4, will also show codec set 4 for region 4 to region 1 connectivity.

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

 Source Region: 4     Inter Network Region Connection Management     I       M
                                                                      G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    4     y    NoLimit                                             n       t
 2    4     y    NoLimit                                             n       t
 3    4     y    NoLimit                                             n       t
 4    4                                                                all
```

The following screen shows IP Network Region 1 configuration. In this example, codec set 1 will be used for calls within region 1 due to the **Codec Set** parameter on Page 1, but codec set 4 will be used for connections between region 1 and region 4 as noted previously. In the shared test environment, network region 1 was in place prior to adding the Verizon test environment and already used **Authoritative Domain** "avaya.com". Where necessary, Session Manager or the Acme Packet Net-Net SBC will adapt the domain.

```
change ip-network-region 1                                    Page   1 of  20
                              IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name: HQ CM and SIP Phones
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? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

The following screen shows the inter-network region connection configuration for region 1. The bold row shows that network region 1 is directly connected to network region 4, and that codec set 4 will be used for any connections between region 4 and region 1.

```
change ip-network-region 1                                      Page   4 of  20

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                                all
 2    2     y    NoLimit                                             n       t
 3    3     y    NoLimit                                             n       t
 4    4     y    NoLimit                                             n       t
```

5.7. IP Codec Sets

The following screen shows the configuration for codec set 4, the codec set configured to be used for calls within region 4 and for calls between region 1 and region 4. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, all calls with the PSTN via the SIP trunks would prefer to use **G.729A**, but also be capable of using **G.711MU**. Any calls using this same codec set that are between devices capable of the **G.722-64K** codec (e.g., Avaya 9600-Series IP Telephone) can use G.722. The specification of G.722 as the first choice is not required. That is, G.722 may be omitted from the codec set.

```
change ip-codec-set 4                                           Page   1 of   2
                          IP Codec Set
    Codec Set: 4

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.722-64K                    2        20
 2: G.729A            n          2        20
 3: G.711MU           n          2        20
 4:
 5:
 6:
 7:
```

On **Page 2** of the form:

- Configure the **Fax Mode** field to **off.** Verizon does not support T.38 fax.
- Configure the **Fax Redundancy** field to **0.**

```
change ip-codec-set 4                                           Page   2 of   2
                          IP Codec Set

                              Allow Direct-IP Multimedia? n

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

The following screen shows the configuration for codec set 1. The configuration for codec set 1 prefers **G.711MU** but also allows **G.729A**. Codec set 1 is used for Avaya Modular Messaging and other local Avaya CPE connections within region 1.

```
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: G.729A            n         2        20
 3:
 4:
 5:
 6:
 7:
```

5.8. SIP Signaling Groups

This section illustrates the configuration of the SIP Signaling Groups. Each signaling group has a **Group Type** of "sip", a **Near-end Node Name** of "procr", and a **Far-end Node Name** of "SM1". In the example screens, the **Transport Method** for all signaling groups is "tcp". In production, TLS transport between Communication Manager and Session Manager can be used. The **Enable Layer 3 Test** field is enabled on each of the signaling groups to allow Communication Manager to maintain the signaling group using the SIP OPTIONS method. Fields that are not referenced in the text below can be left at default values, including **DTMF over IP** set to "rtp-payload", which corresponds to RFC 2833.

The following screen shows signaling group 67. Signaling group 67 will be used for processing incoming PSTN calls from Verizon IPCC Services via Session Manager. This includes any Verizon toll-free calls that are directed by Session Manager directly to Communication Manager, as well as any Verizon toll-free calls that are initially directed to Voice Portal, and then transferred by Voice Portal using SIP REFER to Communication Manager. To enable calls transferred from Voice Portal to Communication Manager using SIP REFER to use signaling group 67, the Session Manager Dial Pattern configuration covering the transferred-to Communication Manager extensions must send calls from the Acme Packet SBC originating "location" to the SIP Entity corresponding to signaling group 67, as shown in **Section 6.8.2**. The **Far-end Network Region** is configured to region 4. Port 5062 has been configured as both the **Near-end Listen Port** and **Far-end Listen Port**. Session Manager will be configured to direct calls arriving from the PSTN with Verizon toll-free numbers to a route policy that uses a SIP entity link to Communication Manager specifying port 5062. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager. In the sample configuration, the **Peer Detection Enabled** field was set to "n". Other parameters may be left at default values.

```
change signaling-group 67                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 67             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others

   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5062                Far-end Listen Port: 5062
                                        Far-end Network Region: 4

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

The following screen shows signaling group 60, the signaling group to Session Manager that was in place prior to adding the Verizon configuration to the shared Avaya Solutions and Interoperability Test Lab configuration. This signaling group reflects configuration not specifically related to Verizon trunking. For example, calls using Avaya SIP Telephones and calls routed to other Avaya applications, such as Avaya Modular Messaging, use this signaling group. Calls involving Avaya Voice Portal, Session Manager and Communication Manager that do not involve Verizon may also use signaling group 60. For example, if Voice Portal transfers a non-Verizon call to a Communication Manager user, the call can use trunk group 60. Moreover, if Voice Portal uses the "bridged transfer" option to bridge an inbound PSTN call from Verizon to Communication Manager (i.e., rather than the "blind transfer" or "consultative transfer" options), trunk group 60 will be used. A Voice Portal bridged transfer "bridges" the inbound Verizon call to the Communication Manager leg of the call within Voice Portal (i.e., Voice Portal uses INVITE, and does not use SIP REFER to effect transfer).

As with signaling group 67, the **Near-end Node Name** is "procr" and the **Far-end Node Name** is "SM1", the node name of the Session Manager. Unlike the signaling groups used for the Verizon signaling, the **Far-end Network Region** is **1**. The **Peer Detection Enabled** field is set to "y" and a peer Session Manager has been previously detected. The **Far-end Domain** is set to "avaya.com" matching the configuration in place prior to adding the Verizon SIP Trunking configuration.

```
change signaling-group 60                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 10
```

5.9. SIP Trunk Groups

This section illustrates the configuration of the SIP Trunks Groups corresponding to the SIP signaling groups from the previous section.

The following shows page 1 for trunk group 67, which will be used for incoming toll-free calls from Verizon. The **Number of Members** field defines how many simultaneous calls are permitted for the trunk group. The **Service Type** field should be set to "public-ntwrk" for the trunks that will handle calls with Verizon. Although not strictly necessary, the **Direction** has been configured to "incoming" to emphasize that trunk group 67 is used for incoming calls only in the sample configuration.

```
change trunk-group 67                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 67                   Group Type: sip           CDR Reports: y
  Group Name: From-SM-Acme-VZ             COR: 1        TN: 1        TAC: 167
   Direction: incoming      Outgoing Display? n
 Dial Access? n                                Night Service:

Service Type: public-ntwrk          Auth Code? n
                                                    Signaling Group: 67
                                                  Number of Members: 6
```

The following shows Page 2 for trunk group 67. All parameters shown are default values, except for the **Preferred Minimum Session Refresh Interval**, which has been changed from the default 600 to 900. Although not strictly necessary, some SIP products prefer a higher session refresh interval than the Communication Manager default value, which can result in unnecessary SIP messages to re-establish a higher refresh interval for each call.

```
change trunk-group 67                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS
     Unicode Name: auto
                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

                                     Delay Call Setup When Accessed Via IGAR? n
```

The following shows Page 3 for trunk group 67. All parameters except those in bold are default values. Optionally, replacement text strings can be configured using the "system-parameters features" screen, such that incoming "private" (anonymous) or "restricted" calls can display an Avaya-configured text string on called party telephones.

```
change trunk-group 67                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                                         Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
 Show ANSWERED BY on Display? y
```

The following shows Page 4 for trunk group 67. The **PROTOCOL VARIATIONS** page is one reason why it can be advantageous to configure incoming calls from Verizon to arrive on specific signaling groups and trunk groups. The bold fields have non-default values. The **Convert 180 to 183 for Early Media** field is new in Communication Manager Release 6. Verizon recommends that inbound calls to the enterprise result in a 183 with SDP rather than a 180 with SDP, and setting this field to "y" for the trunk group handling inbound calls from Verizon produces this result. Note that Voice Portal also sends 183 with SDP for inbound Verizon calls directed from Session Manager to Voice Portal for self-service. Although not strictly necessary, the **Telephone Event Payload Type** has been set to 101 to match Verizon configuration. Setting the **Network Call Redirection** flag to "y" enables advanced services associated with the use of the REFER message, while also implicitly enabling Communication Manager to signal "send-only" media conditions for calls placed on hold at the enterprise site. If neither REFER signaling nor "send-only" media signaling is required, this field may be left at the default "n" value. In the testing associated with these Application Notes, the **Network Call Redirection** flag was set to "n". Voice Portal will send SIP REFER messages to transfer inbound calls from Verizon to Communication Manager, but Communication Manager will not use SIP REFER in vector processing in the sample configuration.

The Verizon IPCC Services do not support the Diversion header or the History-Info header, and therefore both **Support Request History** and **Send Diversion Header** are set to "n".

```
change trunk-group 67                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? n
             Telephone Event Payload Type: 101

       Convert 180 to 183 for Early Media? y
 Always Use re-INVITE for Display Updates? n
                            Enable Q-SIP? n
```

The following shows Page 1 for trunk group 60, the bi-directional "tie" trunk group to Session Manager that existed before adding the Verizon SIP Trunk configuration to the shared Avaya Solutions and Interoperability Test Lab network. Recall that this trunk is used for communication with other Avaya applications, such as Avaya Modular Messaging and internal calls with Voice Portal, and does not reflect any unique Verizon configuration.

```
change trunk-group 60                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 60                  Group Type: sip           CDR Reports: y
  Group Name: SM1                        COR: 1        TN: 1        TAC: 160
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                                      Signaling Group: 60
                                                    Number of Members: 100
```

The following shows Page 3 for trunk group 60. Note that unlike the trunks associated with Verizon calls that use "public" numbering, this tie trunk group uses a "private" **Numbering Format**.

```
change trunk-group 60                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y

                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

The following shows Page 4 for trunk group 60. Note that unlike the trunks associated with Verizon calls that have non-default "protocol variations", this trunk group maintains all default values. **Support Request History** must remain set to the default “y” to support proper subscriber mailbox identification by Avaya Modular Messaging.

```
change trunk-group 60                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                        Mark Users as Phone? n
              Prepend '+' to Calling Number? n
        Send Transferring Party Information? n
                   Network Call Redirection? n
                      Send Diversion Header? n
                    Support Request History? y
             Telephone Event Payload Type:

         Convert 180 to 183 for Early Media? n
   Always Use re-INVITE for Display Updates? n
                              Enable Q-SIP? n
```

5.10. Contact Center Configuration

This section describes the basic commands used to configure Vector Directory Numbers (VDNs) and corresponding vectors, announcements, skills, and agents used to verify call flows. These Application Notes provide rudimentary contact center configuration to illustrate and test the processing of Verizon IPCC calls transferred by Voice Portal to a Communication Manager VDN. In general, call centers will use more complex vector functionality tailored to individual needs.

This section provides an example configuration of a VDN and corresponding vector that will be used to verify Voice Portal blind transfer, consult transfer, and bridged transfer to a VDN. In this example, Voice Portal will not canvas Communication Manager for agent availability prior to transferring the call to the VDN. The inbound toll-free call is transferred to VDN 36880 shown in the following screen.

```
display vdn 36880                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 36880
                                 Name*: Route-To-Skill-80
                           Destination: Vector Number        80
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
```

VDN 36880 is associated with vector number 80, which is shown below. Vector 80 plays an announcement (step 02). In the sample configuration, the announcement with extension 31881 was a brief announcement (“e.g., thank you for choosing Avaya”) that enabled Communication Manager to answer the call. For an inbound call from Verizon to Voice Portal that is subsequently transferred by Voice Portal using SIP REFER, answering the call via the announcement step in the vector serves to complete the REFER call processing, allowing the

Acme Packet Net-Net SBC to provide new SDP attributes to Verizon on the public side of the SBC. Without this announcement step, pre-answer media treatments from Communication Manager, such as ring back while an agent is ringing, may not be heard by the PSTN caller. Next, the "queue-to skill 80" (step 03) is used. If an agent in skill 80 is available, the call will ring the agent, and the caller will hear ring back until the agent answers. If an agent in skill 80 is not available at the time the call is queued to skill 80, the announcement with extension 31880 (step 04) will be heard. In this simple vector configuration, this announcement is a recurring announcement that will repeat until the call is answered.

```
display vector 80                                               Page   1 of   6
                                  CALL VECTOR

    Number: 80              Name: Route-skill-80
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
02 announcement 31881
03 queue-to     skill 80   pri m
04 announcement 31880
05 stop
06
```

The following screens illustrate the hunt group / skill configuration for hunt-group / skill 80. On Page 1, observe the bold parameters.

```
change hunt-group 80                                            Page   1 of   4
                                  HUNT GROUP
            Group Number: 80                                 ACD? y
              Group Name: ACD-Hunt-80                      Queue? y
         Group Extension: 36680                           Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:
```

On Page 2, observe the bold parameters.

```
change hunt-group 80                                            Page   2 of   4
                                  HUNT GROUP

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

The following screens illustrate the announcement extensions used in the call vector steps. All announcements use the G450 Media Gateway.

Recall that announcement 31881 was used to answer a call in the call vector to complete SIP REFER call processing, prior to queuing the call to a skill.

```
display announcement 31881                                 Page   1 of   1
                       ANNOUNCEMENTS/AUDIO SOURCES
  Extension: 31881                        COR: 1
  Annc Name: Call-entering-queue-annc      TN: 1
  Annc Type: integrated                 Queue? y
Group/Board: 001V9
  Protected? n                           Rate: 64
```

Recall that announcement 31880 was used as a repeating announcement heard by a caller while a call was in queue to a skill (i.e., call arrives to a vector and no agent was immediately available).

```
display announcement 31880                                   Page   1 of   1
                       ANNOUNCEMENTS/AUDIO SOURCES
  Extension: 31880                        COR: 1
  Annc Name: Recurring-in-Queue-80-annc    TN: 1
  Annc Type: integ-rep                  Queue? y
Group/Board: 001V9
  Protected? n                           Rate: 64
```

The following screen illustrates an example agent login-ID.   This agent will staff skill 80.

```
change agent-loginID 46880                                     Page   1 of   3
                                 AGENT LOGINID

                Login ID: 46880                                    AAS? n
                    Name: Joey Skillful                          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: 1234
                                                 Password (enter again): 1234
                                                           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:   :
```

The following screen illustrates Page 2 for the example agent login-ID.   Observe Skill Number (SN) 80 is used.

```
change agent-loginID 46880                                    Page   2 of   3
                                 AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
Call Handling Preference: skill-level                  Local Call Preference? n

    SN   RL SL           SN   RL SL           SN   RL SL           SN   RL SL
 1: 80      1       16:                  31:                  46:
 2:                 17:                  32:                  47:
 3:                 18:                  33:                  48:
 4:                 19:                  34:
```

After logging in agent-loginID 46880 from the telephone with station user extension 30002, the “list agent-loginID” command can be used to verify the status.

```
list agent-loginID

                               AGENT LOGINID
Login ID     Name           Extension     Dir Agt  AAS/AUD        COR Ag Pr
SO
               Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv
Skil/Lv

46880        Joey Skillful  30002                                  1   lvl
                80/01     /       /       /       /       /       /       /
```

5.11. Public Numbering

The "change public-unknown-numbering" command may be used to define the format of numbers sent to Verizon in SIP headers such as the "From" and "PAI" headers.

In the first bolded row shown in the example abridged output below, a specific Communication Manager extension (x30002) is mapped to a Verizon number (866-851-2649), when the call uses trunk group 67. In the other bolded rows shown in the example abridged output below, entries are made for the specific Communication Manager Vector Directory Numbers (VDN) illustrated in the prior section. In the course of the testing, this configuration was varied, with different Verizon toll-free numbers associated with various Communication Manager extensions and functions.

```
change public-unknown-numbering 5                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk          CPN          CPN
Len Code          Grp(s)       Prefix       Len
                                                  Total Administered: 3
 5  3             60                        5        Maximum Entries: 9999
 5  556                                     5
 5  30002         67           8668512649   10
 5  36880         67           8668510107   10
```

5.12. Incoming Call Handling Treatment for Incoming Calls

In general, the "incoming call handling treatment" for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. For incoming Verizon toll-free calls that are delivered by Session Manager to Communication Manager rather than Voice Portal, if the toll-free number sent by Verizon is unchanged by Session Manager, then the number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. As an example, the following screen illustrates a conversion of toll-free number 8668512649 to extension 30002.

```
change inc-call-handling-trmt trunk-group 67                    Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 8668512649      all 30002
```

For calls that are transferred by Voice Portal to Communication Manager using SIP REFER, the incoming call handling table can be used to manipulate the number in the INVITE message sent by the Acme Packet SBC. In the sample configuration, no such configuration was necessary. For example, for a transfer by Voice Portal to VDN 36880, Voice Portal included 36880 in the Refer-To header. The Acme Packet SBC extracted 36880 from the Refer-To header, and sent 36880 in the INVITE towards Session Manager and Communication Manager.

5.13. Uniform Dial Plan (UDP) and Automatic Alternate Routing (AAR)

Although not specifically related to Verizon, this section shows a portion of the UDP and AAR configuration. In the UDP table shown below, calls of the form 44xxx will be routed via AAR. The bold row corresponding to pattern "44xxx" allows calls dialed by Communication Manager users to be routed via AAR to the Voice Portal self-service applications via Session Manager.

```
change uniform-dialplan 3                                          Page   1 of   2

                       UNIFORM DIAL PLAN TABLE
                                                               Percent Full: 0
  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num

 31              5   0                  aar  n
 3100            5   0                  aar  n
 33              5   0                  aar  n
 3400            5   0                  aar  n
 44              5   0                  aar  n
```

In the AAR table shown below, the bold row shows that calls to the number range 44xxx, which includes the Voice Portal application number 44000, will use **Route Pattern** 60. As can be observed from the other rows, various other dial strings also route to other internal destinations (i.e., not to Verizon) via route pattern 60.

```
change aar analysis 0                                              Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd

    3100                  5    5     60       unku         n
    32                    5    5     60       unku         n
    33                    5    5     60       unku         n
    3400                  5    5     60       unku         n
    44                    5    5     60       unku         n
```

5.14. Route Pattern for Internal Calls via Session Manager

Although not specifically related to Verizon, this section shows an example AAR routing to trunk group 60 to Session Manager. If a Communication Manager user dials a Voice Portal application (e.g., 44000), the call will use this route pattern. As shown in reference [JRR-VZIPCC], calls to Avaya Modular Messaging also use this route pattern. Route pattern 60 contains trunk group 60, the "private" tie trunk group to Session Manager.

```
change route-pattern 60                                         Page   1 of   3
                    Pattern Number: 60  Pattern Name: SM FS
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 60   0                    0                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.15. Private Numbering

Although not specifically related to Verizon, this section shows the private numbering configuration associated with the calls using trunk group 60. The bold rows configure any five digit number beginning with 3 or 4 (i.e., 3xxxx, 4xxxx) that uses trunk group 60 to retain the original 5 digit number (i.e., no digit manipulation is specified, and the **Total Len** is 5).

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  2                                             5     Total Administered: 5
 5  3                60                           5        Maximum Entries: 540
 5  4                60                           5
 5  5                                             5
```

5.16. Communication Manager Stations

In the sample configuration, five digit station extensions were used with the format 3xxxx. The following abbreviated screen shows an example extension for an Avaya H.323 IP telephone. As documented in reference [JRR-VZIPCC], coverage path 60 is assigned to give this user coverage to Avaya Modular Messaging.

```
change station 30002                                            Page   1 of   5
                                     STATION
Extension: 30002                        Lock Messages? n               BCC: 0
     Type: 9620                         Security Code: *                TN: 1
     Port: S00038                     Coverage Path 1: 60              COR: 1
     Name: Joey Votto                 Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
```

On Page 2, the **MWI Served User Type** is set to “sip-adjunct” for the SIP integration to Avaya Modular Messaging.

```
change station 30002                                            Page   2 of
5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer:
none
             CDR Privacy? n                             Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n                     Restrict Last Appearance? y
  Active Station Ringing: single
                                                           EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                         EC500 State: enabled
         Multimedia Mode: enhanced                   Audible Message Waiting? n
    MWI Served User Type: sip-adjunct             Display Client Redirection? n
                                                 Select Last Used Appearance? n
                                                   Coverage After Forwarding? s
                                                     Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
```

5.17. Coverage Path

This section illustrates an example coverage path for a station with a mailbox on Avaya Modular Messaging. Hunt group 60, the hunt group to Modular Messaging, is **Point1** in coverage path 60. See reference [JRR-VZIPCC] for additional information on coverage to Avaya Modular Messaging.

```
change coverage path 60                                      Page   1 of   1
                                 COVERAGE PATH
                   Coverage Path Number: 60
     Cvg Enabled for VDN Route-To Party? y        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n              n
               Busy?             y              y
       Don't Answer?             y              y        Number of Rings: 2
                All?             n              n
 DND/SAC/Goto Cover?             y              y
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h60            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

5.18. Saving Communication Manager Configuration Changes

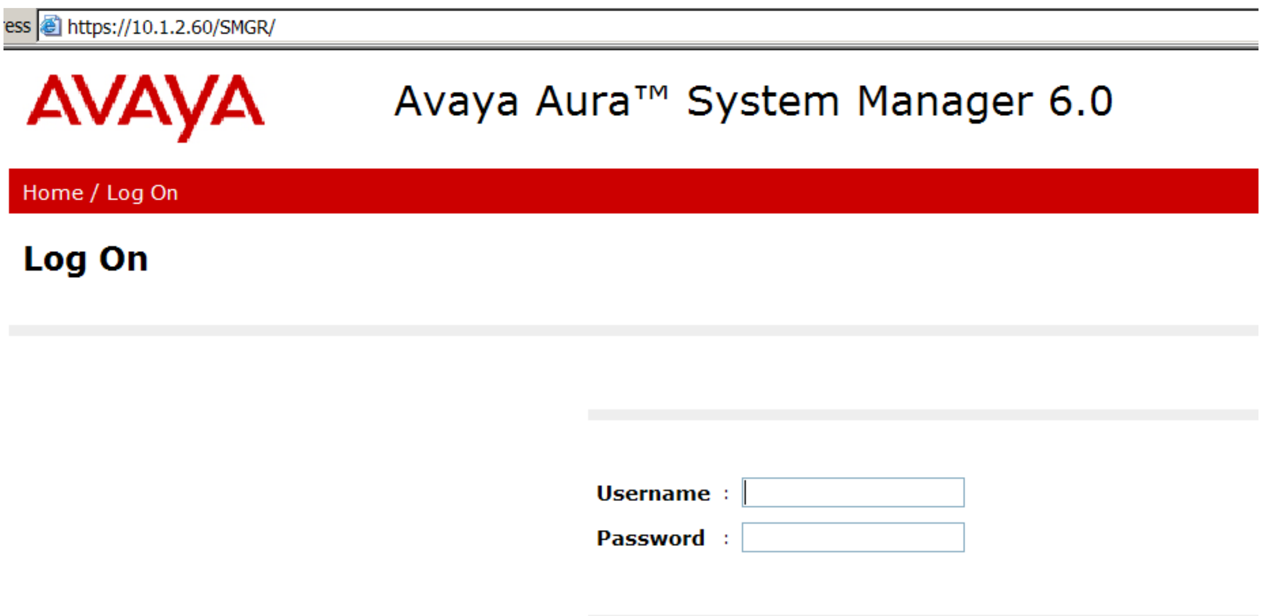
The command "save translation all" can be used to save the configuration.

6. Avaya Aura® Session Manager Configuration

This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

Note – The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two.

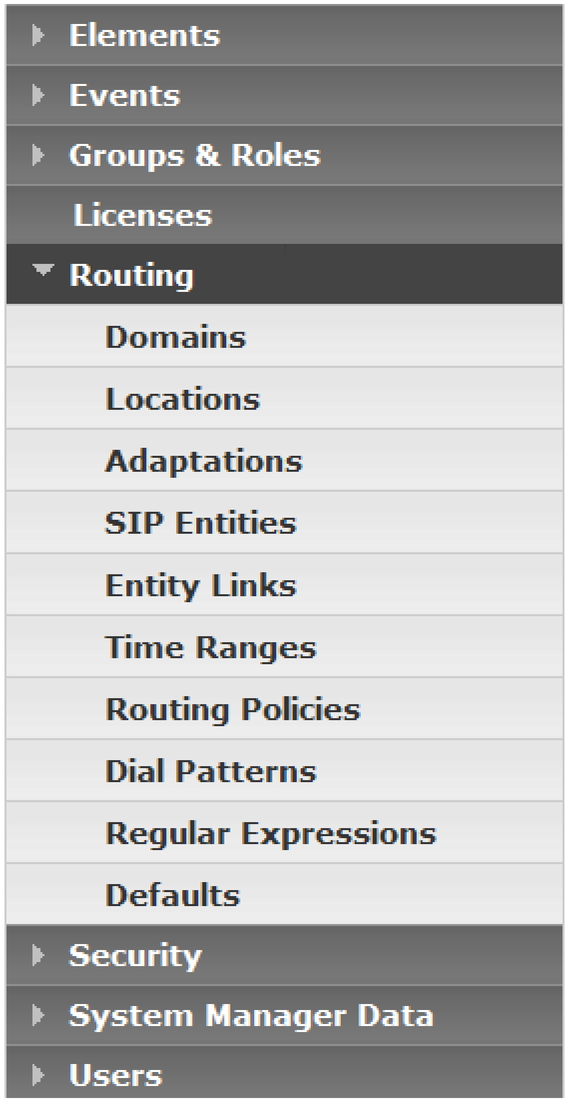
Session Manager is managed via System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **Username** and **Password** and press the **Log On** button (not shown).



Once logged in, a **Home Screen** is displayed. An abridged **Home Screen** is shown below.


Address https://10.1.2.60/SMGR/
Go
AVAYA
Avaya Aura™ System Manager 6.0
Welcome, admin Last Logged on at April 29, 2010 5:07 PM
Help | About | Change Password | Log off
Elements
Events
Groups & Roles
Licenses
Routing
Security
System Manager Data
Users
Help
Home Screen
Sub Pages
Action
Description
Help
Elements
This section provides various functionality related to elements. Some functionality is implemented by SMGR generic services and some are provided by product specific element managers.
Help for RTS
Events
Event Management section of the System Manager Console. This part of SMGR lets you view and administer logs and alrms related to the indivual domains of SMGR.
Help to manage events like logs and alarms
Groups & Roles
Groups and Roles administration section of System Manager Console. This part of SMGR lets you create and manage groups , roles and permissions.
Help to manage groups and roles
Licence Administration section of the system Manager Console. This
Help to administer

For readers familiar with prior releases of Session Manager, the configurable items under **Routing** in Release 6 were located under a heading called **Network Routing Policy** in prior releases. Select **Routing.** The screen shown below shows the various sub-headings.



When Routing is selected, the right side outlines a series of steps. The sub-sections that follow are in the same order as the steps outlined under **Introduction to Network Routing Policy (NRP)** in the abridged screen shown below.

Introduction to Network Routing Policy

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

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

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Scroll down to review additional steps if desired as shown below. In these Application Notes, all steps are illustrated with the exception of Step 9, since "Regular Expressions" were not used.

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"

(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 8: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

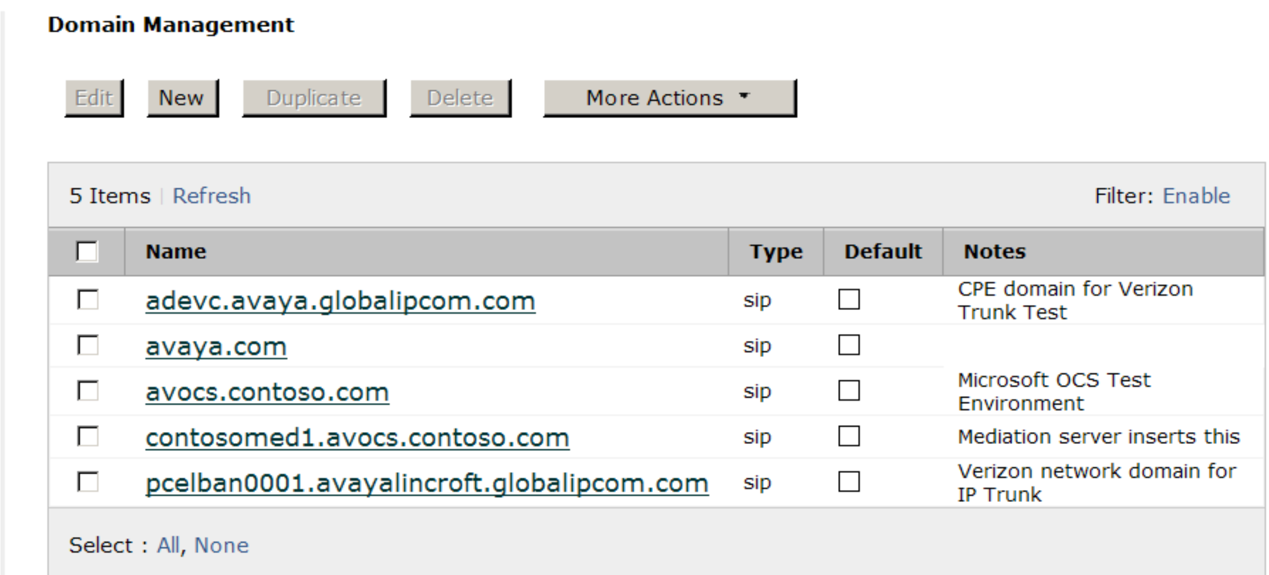
Step 9: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

6.1. Domains

To view or change SIP domains, select **Routing → Domains**. Click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button after changes are completed. The following screen shows a list of configured SIP domains. The Session Manager used in the verification of these Application Notes was shared among many Avaya interoperability test efforts. The domain "avaya.com" was already being used for communication among a number of Avaya systems and applications. The domain "avaya.com" is not known to the Verizon production service.


Domain Management

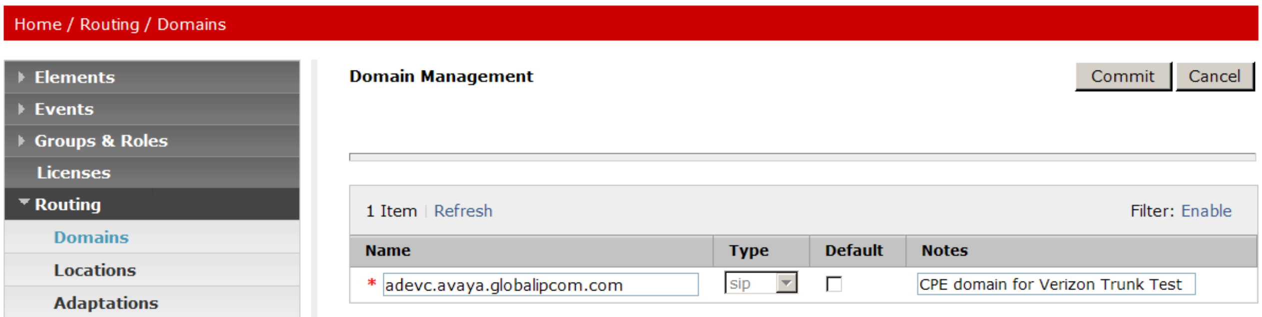
Edit | New | Duplicate | Delete | More Actions

5 Items | Refresh Filter: Enable

	Name	Type	Default	Notes
☐	adevc.avaya.globalipcom.com	sip	☐	CPE domain for Verizon Trunk Test
☐	avaya.com	sip	☐	
☐	avocs.contoso.com	sip	☐	Microsoft OCS Test Environment
☐	contosomed1.avocs.contoso.com	sip	☐	Mediation server inserts this
☐	pcelban0001.avayalincroft.globalipcom.com	sip	☐	Verizon network domain for IP Trunk

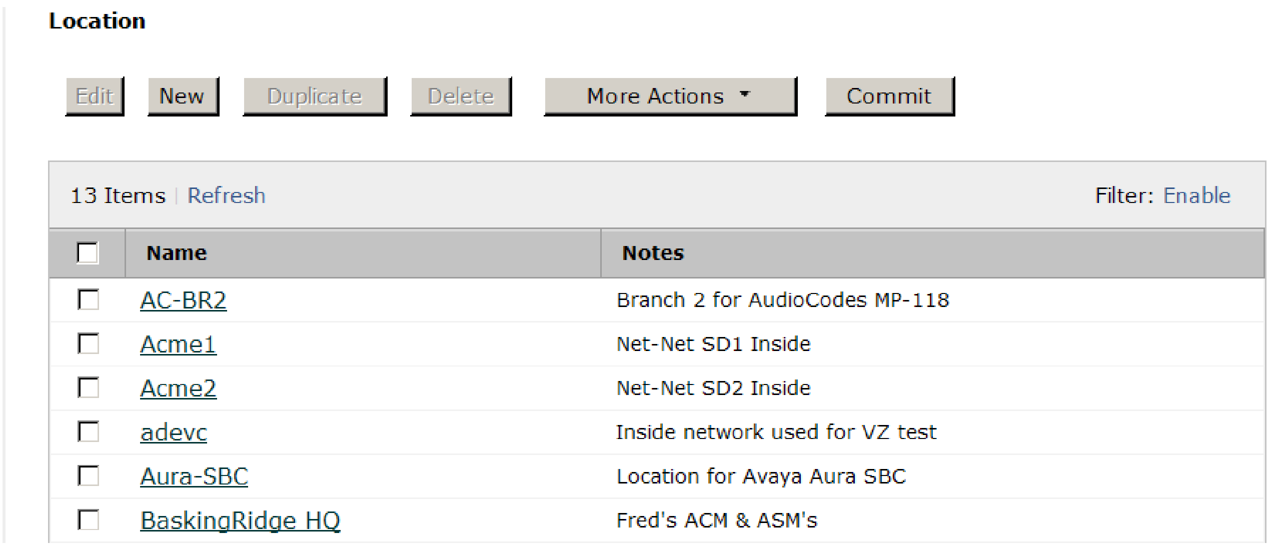
Select : All, None

The domain “adevc.avaya.globalipcom.com” is the domain known to Verizon as the enterprise SIP domain. In the sample configuration, Verizon included this domain as the host portion of the Request-URI in the SIP INVITE for inbound toll-free calls. Verizon set the host port of the To header with the outside (public) IP Address of the SBC (1.1.1.2).



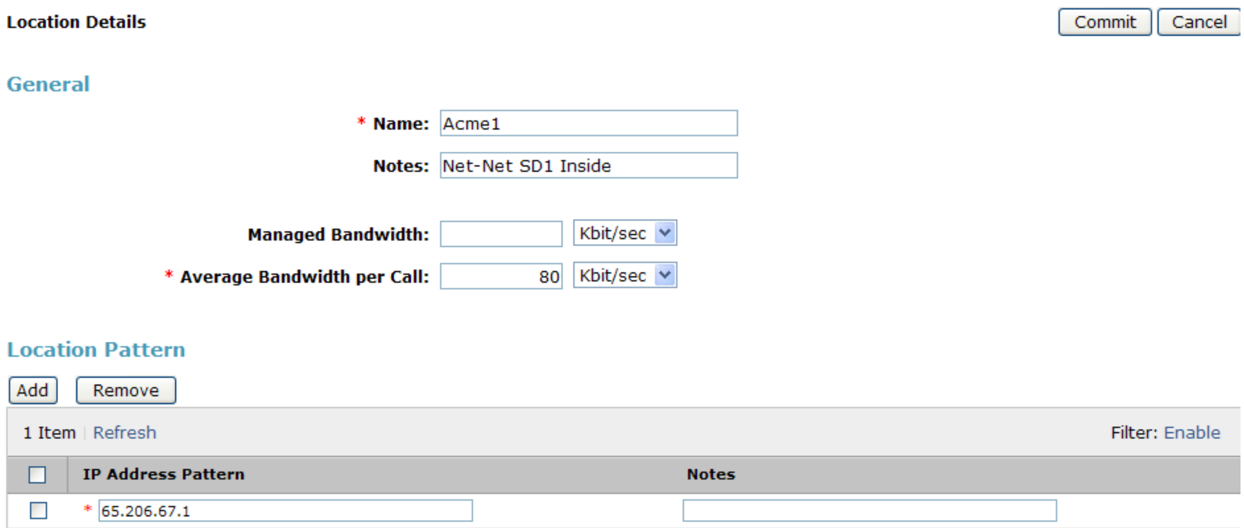
6.2. Locations

To view or change locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and **Edit** to edit an existing location, or the **New** button to add a location. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

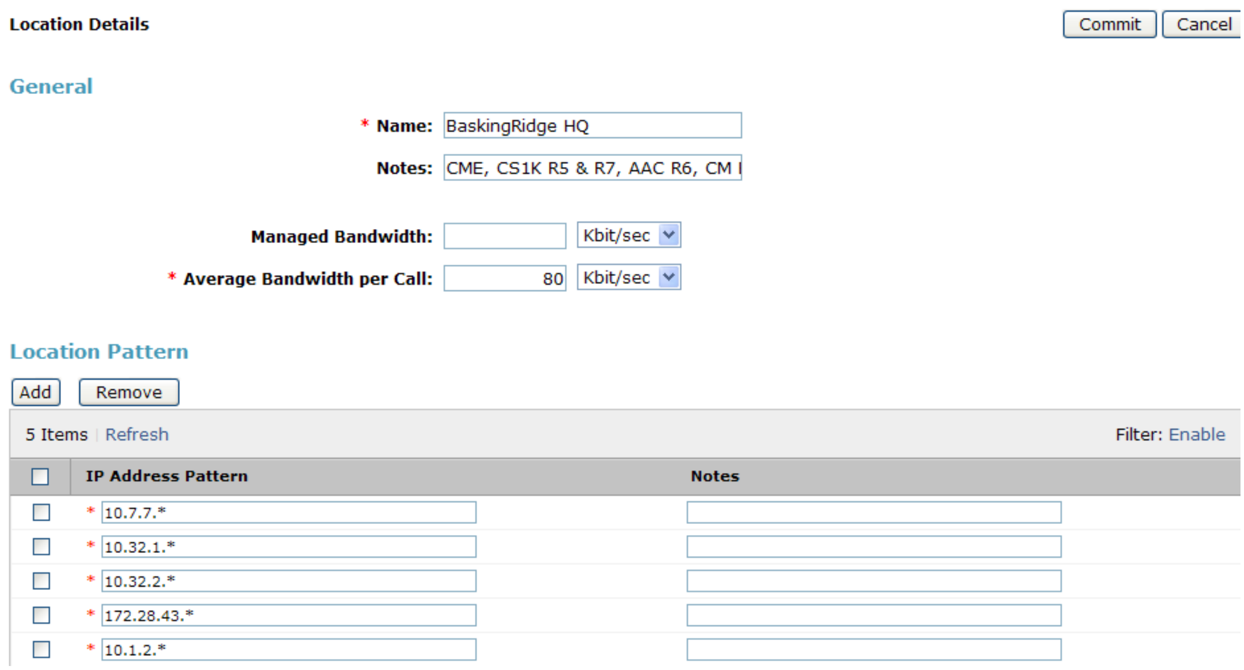


	Name	Notes
☐	AC-BR2	Branch 2 for AudioCodes MP-118
☐	Acme1	Net-Net SD1 Inside
☐	Acme2	Net-Net SD2 Inside
☐	adevc	Inside network used for VZ test
☐	Aura-SBC	Location for Avaya Aura SBC
☐	BaskingRidge HQ	Fred's ACM & ASM's

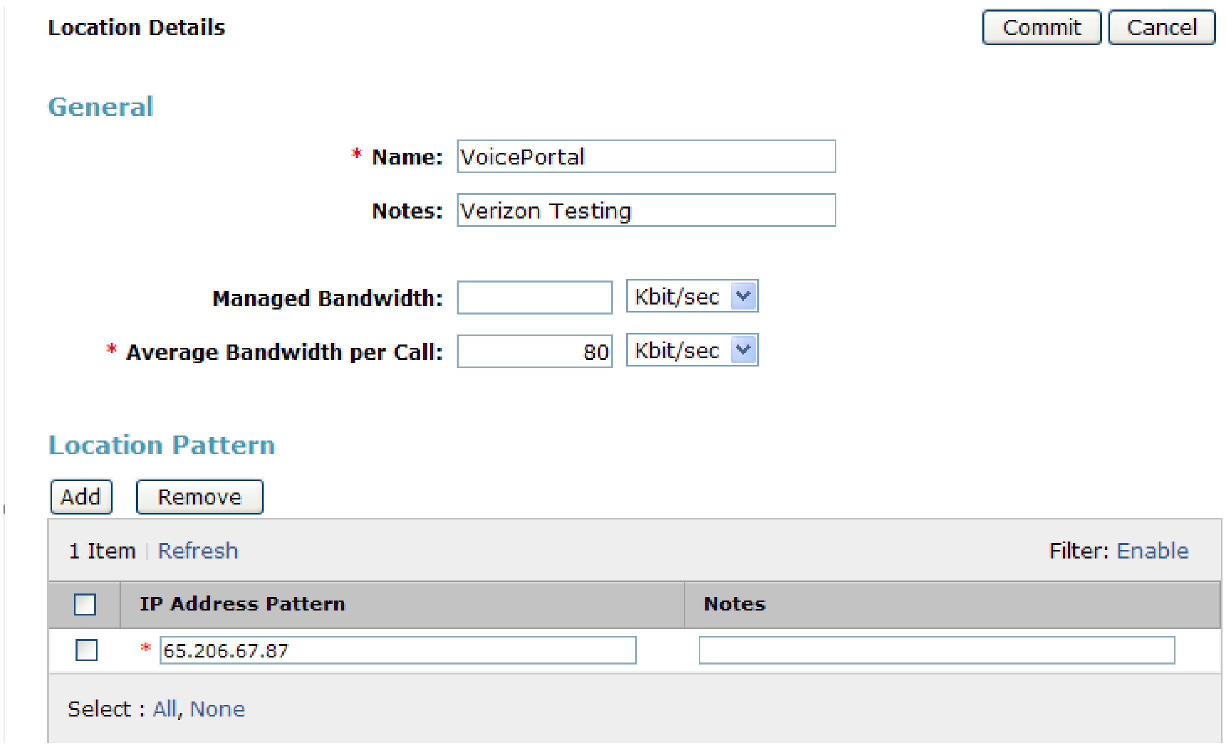
The following screen shows the **Location Details** for the location named "Acme1", corresponding to the Acme Packet Net-Net SBC relevant to these Application Notes. Later, the location with name "Acme1" will be assigned to the corresponding SIP Entity. The IP Address 65.206.67.1 of the inside (private) interface of "Acme1" is entered in the **IP Address Pattern** field.



The following screen shows the **Location Details** for the location named "BaskingRidge HQ". The SIP Entities and associated IP Addresses for this location correspond to the shared components of the Avaya Interoperability Lab test environment, such as Communication Manager Release 6 and Session Manager Release 6.



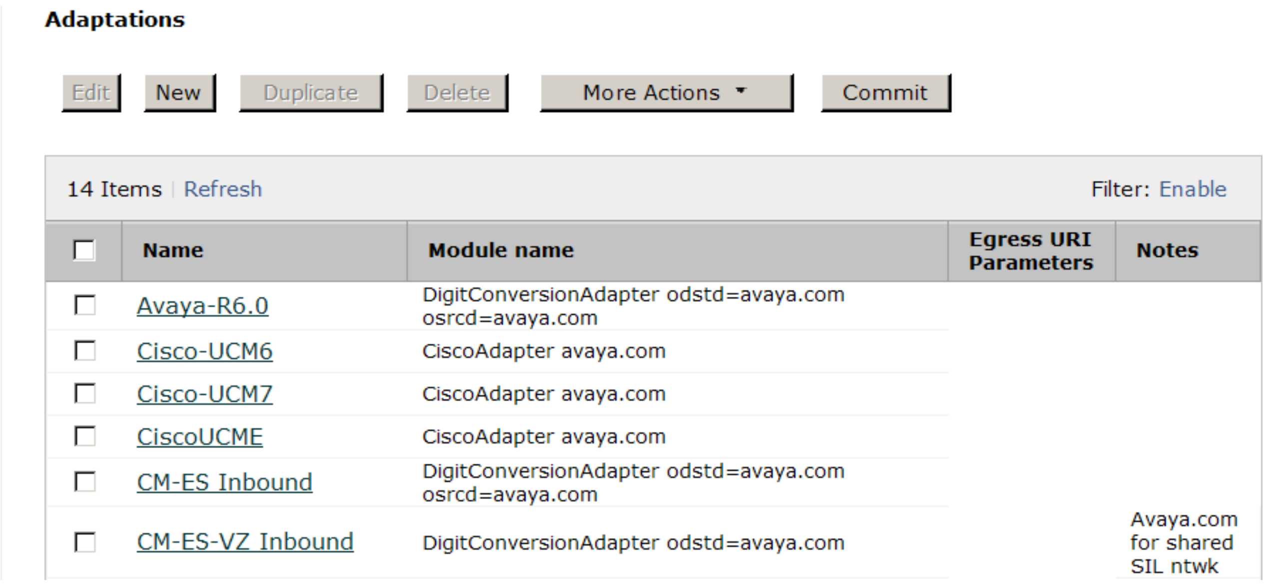
The following screen shows the **Location Details** for the location named "VoicePortal". The IP Address Pattern contains the IP Address 65.206.67.87 of the Voice Portal single server system used in the sample configuration.



6.3. Adaptations

To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed.

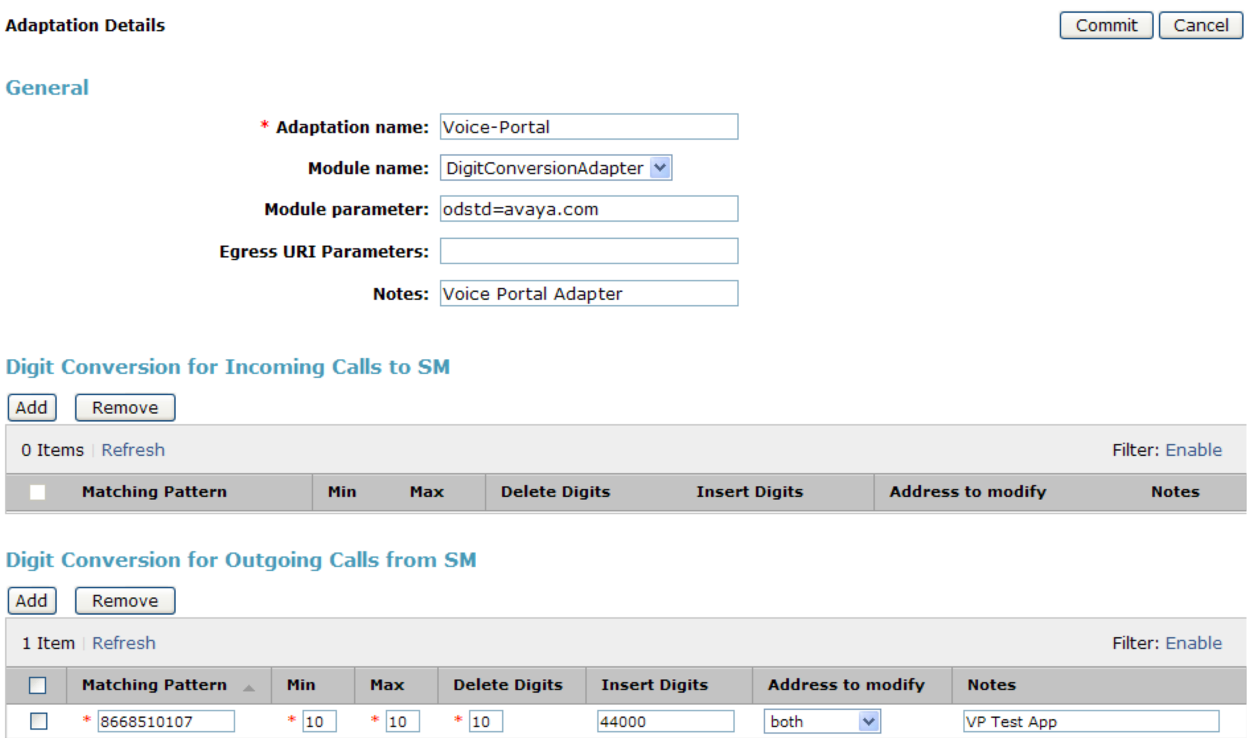
The following screen shows a portion of the list of adaptations in the sample configuration.



6.3.1. Adaptation for Session Manager to Voice Portal

The adapter named "Voice-Portal" shown below will later be assigned to the SIP Entity for Voice Portal. This adaptation uses the "DigitConversionAdapter" and specifies the "odstd=avaya.com" parameter to adapt the domain in the Request URI to the domain expected by Voice Portal in the sample configuration. For example, for inbound toll-free calls from Verizon to the Avaya CPE, the Request-URI header sent to Voice Portal will contain "avaya.com" as expected by Voice Portal. However, this is not sufficient. Voice Portal 5.1 also expects the To header to contain the configured domain. Since Session Manager 6.0 does not adapt the To header, the Acme Packet Net-Net SBC is used to adapt the To head to "avaya.com" as detailed in **Section 7**.

In the example screen shown below, the Verizon IP Toll Free number 866-851-0107 is adapted to the number 44000. In the course of testing, other Verizon numbers, such as Verizon IP IVR outdial number 866-851-8119 were adapted in similar fashion. In the sample configuration, Voice Portal will match 44000 to a Voice Portal application.



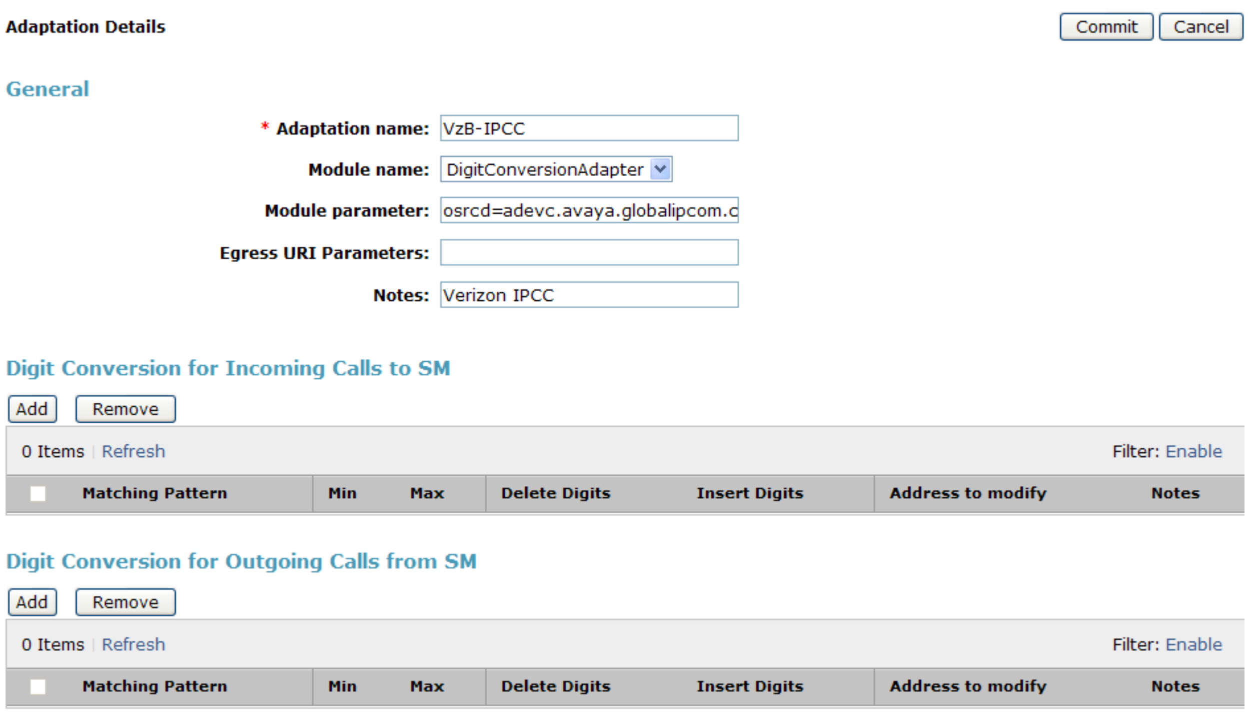
6.3.2. Adaptation for Session Manager to Acme Packet Net-Net SBC

The adapter named "VzB-IPCC" shown below will later be assigned to the Acme Packet Net-Net SIP Entity. The adapter is configured to apply two parameters:

- "osrcd=adevc.avaya.globalipcom.com". This configuration enables the source domain to be overwritten with "adevc.avaya.globalipcom.com". For example, for inbound toll-free calls from Verizon, the PAI header sent to Verizon in the 200 OK will contain "adevc.avaya.globalipcom.com". Depending on the enterprise domain configuration, it may not be necessary for Session Manager to adapt the domain in this fashion. In the sample configuration, where "avaya.com" was already in use in a shared Avaya environment, it was appropriate for Session Manager to adapt the domain from "avaya.com" to "adevc.avaya.globalipcom.com" where the latter is the CPE domain known to Verizon.

- "odstd=172.30.205.55" This configuration enables the destination domain to be overwritten with "172.30.205.55", the Verizon IPCC service node IP Address. The similar configuration including rationale is provided in Section 4.3.2.2 of reference [JF-VZIPCC].

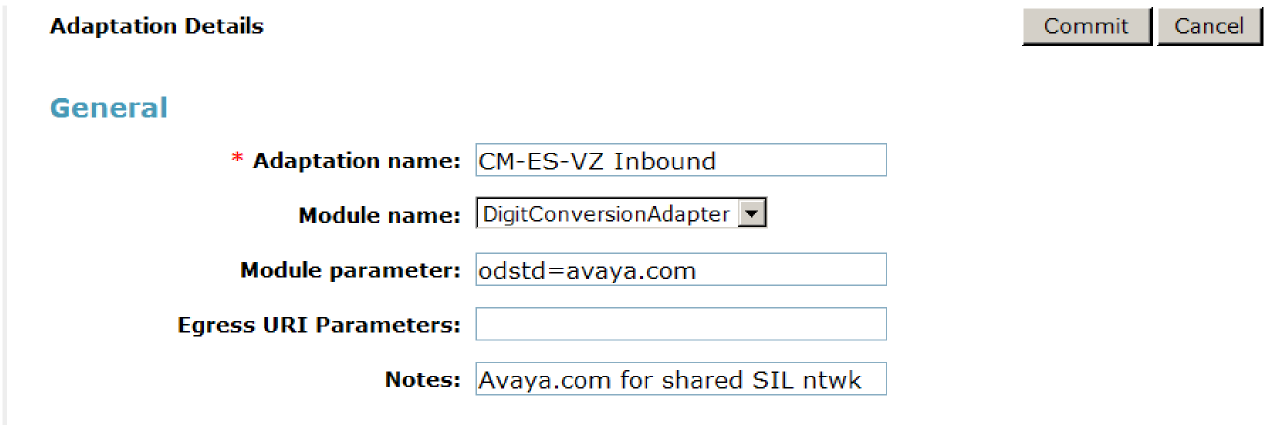
The following screen shows the adaptation details. Although the "DigitConversionAdapter" is used, no conversion of digits is required. The adapter is used to apply the module parameters, and not for true digit manipulation.



6.3.3. Adaptation for Session Manager to Communication Manager

The adapter described here is common to reference [JRR-VZIPCC]. This adapter is necessary only if certain calls from Verizon will be routed by Session Manager directly to Communication Manager, rather than first routing to Voice Portal for self-service.

The adapter named "CM-ES-VZ Inbound" shown below will later be assigned to the SIP Entity linking Session Manager to Communication Manager for calls involving Verizon. This adaptation uses the "DigitConversionAdapter" and specifies the "odstd=avaya.com" parameter to adapt the domain to the domain expected by Communication Manager in the sample configuration. More specifically, this configuration enables the destination domain to be overwritten with "avaya.com" for calls that egress to a SIP entity using this adapter. For example, for inbound toll-free calls from Verizon directly to Communication Manager, the Request-URI header sent to Communication Manager will contain “avaya.com” as expected by Communication Manager in the shared Avaya Interoperability Lab configuration. Depending on the Communication Manager configuration, it may not be necessary for Session Manager to adapt the domain in this fashion.



In the testing associated with these Application Notes, it is not necessary for this adapter to perform digit conversion. If desired, consult [JRR-VZIPCC] for examples of digit conversion for inbound toll-free calls routed by Session Manager to Communication Manager, rather than Voice Portal.

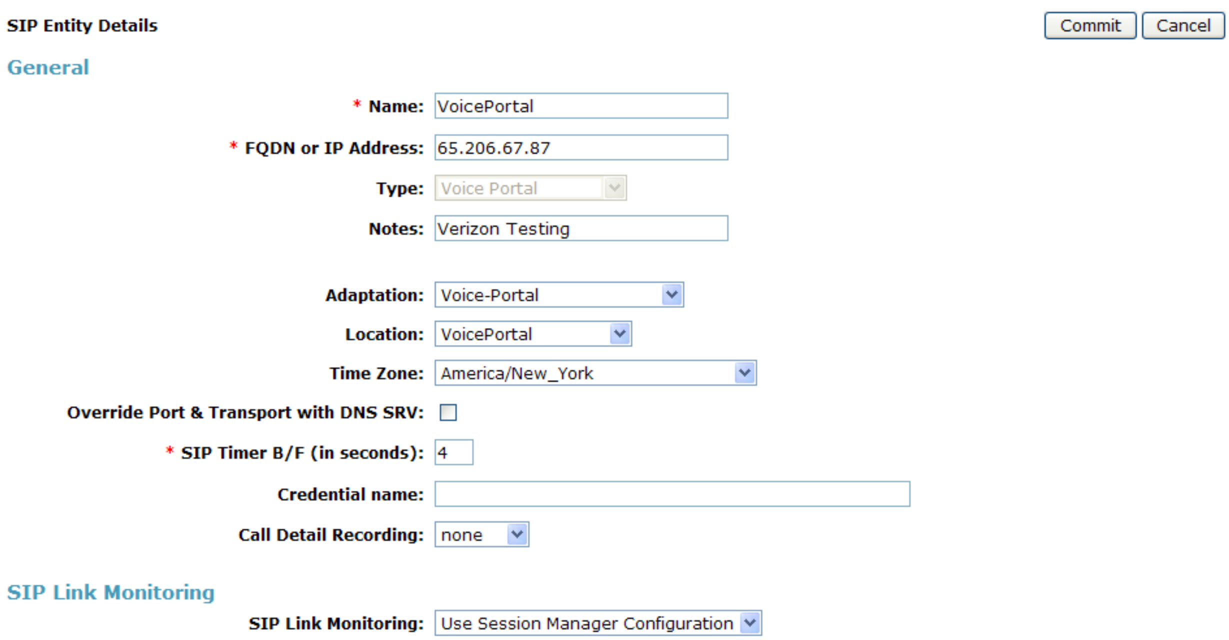
6.4. SIP Entities

To view or change SIP entities, select **Routing → SIP Entities**. Click the checkbox corresponding to the name of an entity and **Edit** to edit an existing entity, or the **New** button to add an entity. Click the **Commit** button after changes are completed.

Compared with the configuration documented in reference [JRR-VZIPCC], the only new SIP Entity is for Voice Portal. Other SIP Entities are illustrated below for completeness.

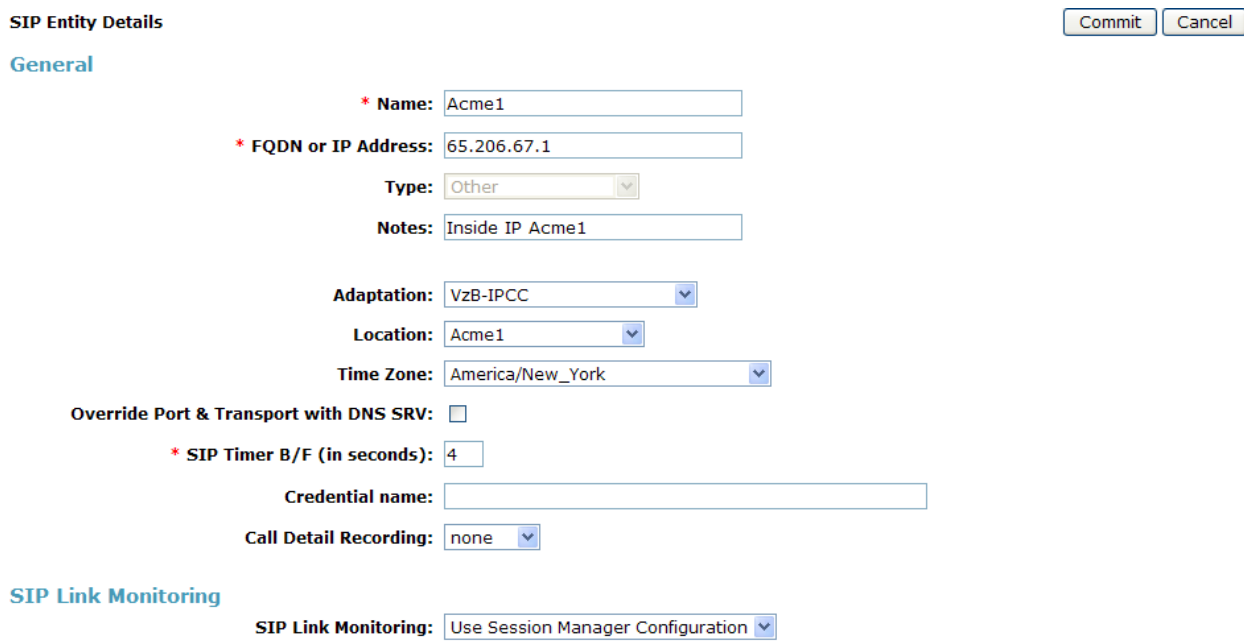
6.4.1. SIP Entity for Voice Portal

The SIP Entity named "VoicePortal" is shown below. The **FQDN or IP Address** is set to "65.206.67.87", the IP Address of the single server Voice Portal used in the sample configuration. If necessary, see reference [ICR] for an example using a common FQDN to distribute calls among several VoicePortal servers. The **Type** of entity is "Voice Portal". The **Location** is set to "VoicePortal", the location configured in **Section 6.2**. The **Adaptation** is set to "Voice-Portal", the adaptation configured in **Section 6.3**. Default parameters can be retained in other fields or modified to suit as desired.



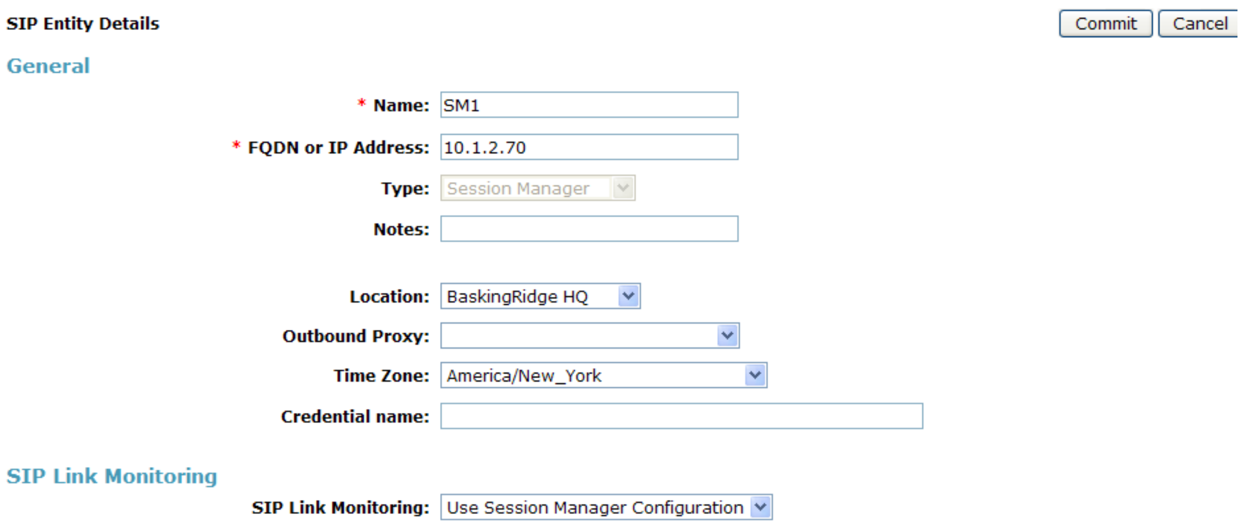
6.4.2. SIP Entity for Acme Packet Net-Net SBC

The following screen shows the **SIP Entity Details** corresponding to “Acme1”. The **FQDN or IP Address** field is configured with the Acme Packet Net-Net SBC inside IP Address (65.206.67.1). “Other” is selected from the **Type** drop-down menu for SBC SIP Entities. This Acme Packet Net-Net SBC has been assigned to **Location** “Acme1”, shown in **Section 6.2,** and the “VzB-IPCC” adapter shown in **Section 6.3** is applied.



6.4.3. SIP Entity for Session Manager

The following screen shows the upper portion of the **SIP Entity Details** corresponding to "SM1". The **FQDN or IP Address** field for "SM1" is the Session Manager Security Module IP Address (10.1.2.70), which is used for SIP signaling with other networked SIP entities. The **Type** for this SIP entity is "Session Manager". Select an appropriate location for the Session Manager from the **Location** drop-down menu. In the shared test environment, the Session Manager used location "BaskingRidge HQ". The default **SIP Link Monitoring** parameters may be used. If desired, these timers may be customized for each entity.



Scrolling down, the following screen shows the middle portion of the **SIP Entity Details**, a listing of the **Entity Links** previously configured for “SM1”. Use the **Next** button to reveal additional links (not shown). The links relevant to these Application Notes are described in the following section.

Entity Links

Add | Remove

34 Items | Refresh — Filter: Enable

☐	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
☐	SM1	TCP	* 5060	AACR6	* 5060	☑
☐	SM1	TCP	* 5060	Acme1	* 5060	☑
☐	SM1	TCP	* 5060	Acme2	* 5060	☑
☐	SM1	TCP	* 5060	AG2330	* 5080	☑
☐	SM1	TCP	* 5060	AuraSBC	* 5060	☑
☐	SM1	TCP	* 5060	CallCenter	* 5060	☑
☐	SM1	TCP	* 5060	Cisco-UCM6	* 5060	☑
☐	SM1	TCP	* 5060	Cisco-UCM7	* 5060	☑
☐	SM1	TCP	* 5060	CiscoUCME	* 5060	☑
☐	SM1	UDP	* 5060	CiscoUCME	* 5060	☑
☐	SM1	TCP	* 5060	CM Evolution Server	* 5060	☑
☐	SM1	TCP	* 5062	CM-Evolution-procr-5062	* 5062	☑
☐	SM1	TCP	* 5060	CS1K R5.5	* 5060	☑
☐	SM1	TCP	* 5060	CS1K R7	* 5060	☑
☐	SM1	TCP	* 5060	Denver Nortel CS1000e	* 5060	☑

Select : All, None — < Previous | Page 1 of 3 | Next >

Scrolling down, the following screen shows the lower portion of the **SIP Entity Details**, a listing of the configured ports for “SM1”. In the sample configuration, TCP port 5060 was already in place for the shared test environment, using **Default Domain** “avaya.com”. To enable calls with Verizon to be distinguished from other types of SIP calls using the same Session Manager, TCP port 5062 was added, with **Default Domain** “adevc.avaya.globalipcom.com”. Click the **Add** button to configure a new port. TCP is used in the sample configuration for improved visibility during testing.

Port

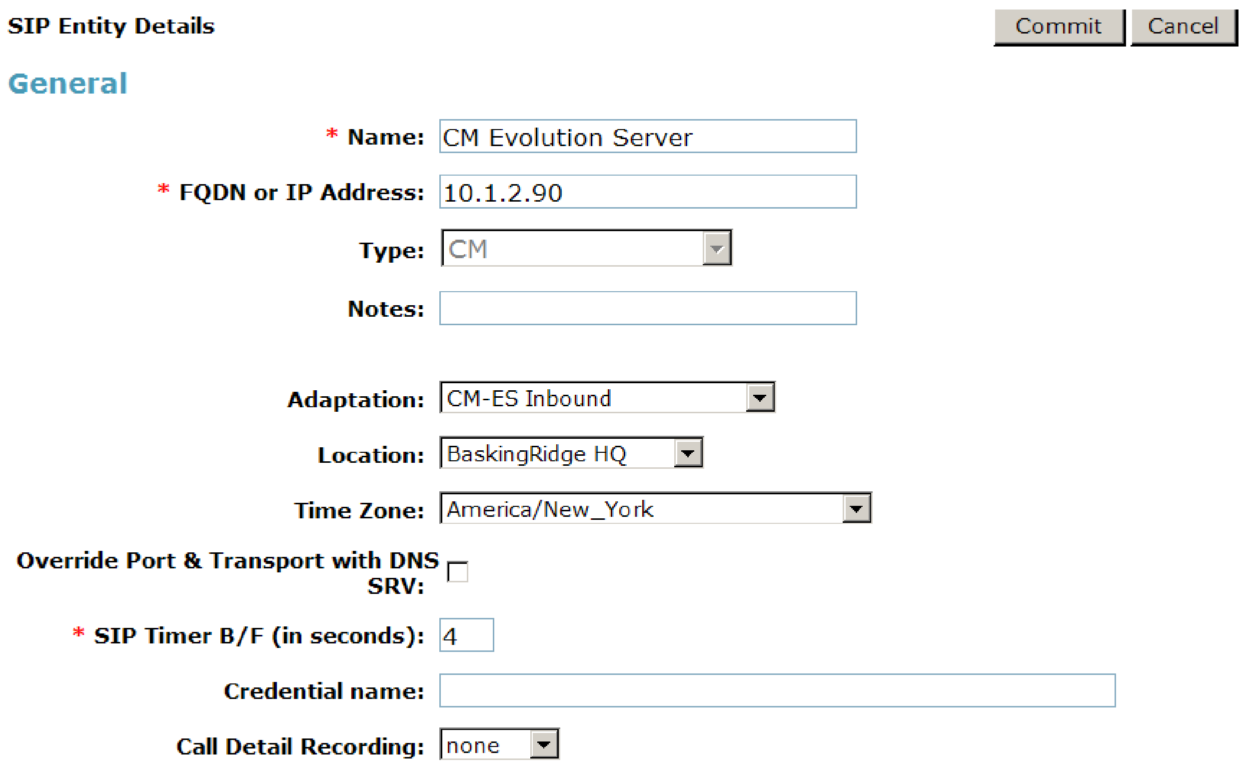
Add | Remove

5 Items | Refresh — Filter: Enable

☐	Port	Protocol	Default Domain	Notes
☐	5060	TCP	avaya.com	
☐	5060	UDP	avaya.com	
☐	5061	TLS	avaya.com	
☐	5062	TCP	adevc.avaya.globalipcom.com	Verizon testing CPE-domain
☐	5070	TCP	avocs.contoso.com	

6.4.4. SIP Entity for Communication Manager (Not Specific to Verizon)

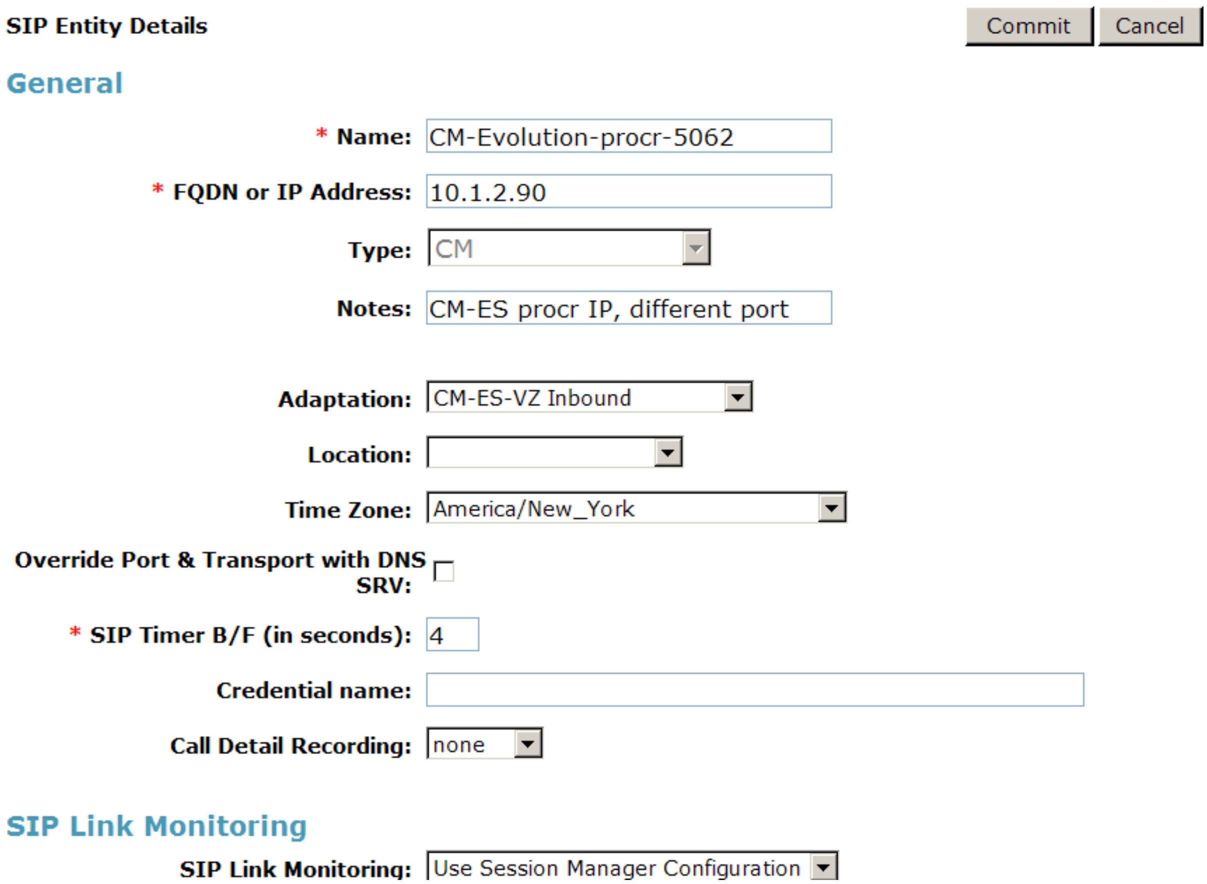
The following screen shows a portion of the **SIP Entity Details** corresponding to an Communication Manager SIP Entity named "CM Evolution Server" This is the SIP Entity that was already in place in the shared Avaya Interoperability Test Lab environment, prior to adding the Verizon IP Trunk configuration. The **FQDN or IP Address** field contains the IP Address of the "processor ethernet" (10.1.2.90). In systems with Avaya G650 Media Gateways containing C-LAN cards, C-LAN cards may also be used as SIP entities, instead of, or in addition to, the "processor ethernet". "CM" is selected from the **Type** drop-down menu. In the shared test environment, the **Adaptation** "CM-ES Inbound" and **Location** "BaskingRidge HQ" had already been assigned to the Communication Manager SIP entity. Reference [JRR-VZIPCC] can be consulted for additional details.



6.4.5. SIP Entity for Communication Manager (Specific to Verizon)

The following screen shows the **SIP Entity Details** for an entity named “CM-Evolution-procr-5062”. This entity uses the same **FQDN or IP Address** (10.1.2.90) as the prior entity with name “CM Evolution Server”; both correspond to the S8800 Processor Ethernet. Later, a unique port, 5062, will be used for the Entity Link to “CM-Evolution-procr-5062”. Using a different port is one approach that will allow Communication Manager to distinguish traffic originally from Verizon from other SIP traffic arriving from the same IP Address of Session Manager, such as SIP traffic generated by Avaya SIP Telephones or Avaya Modular Messaging. The adapter “CM-ES-VZ Inbound” is applied to this SIP entity. Recall that this adapter can be used to adapt the domain as well as map Verizon toll-free numbers to the corresponding Communication Manager extensions, if necessary. If desired, a location can be assigned if location-based routing criteria will be used.

Depending on Session Manager configuration, calls that are transferred by Voice Portal to a Communication Manager extension may use this SIP Entity. For example, using the sample configuration, if a call to a Verizon toll-free number is routed by Session Manager to Voice Portal, and the Voice Portal application requests a blind transfer to a Communication Manager extension, the transferred call will use this SIP Entity and its corresponding Communication Manager signaling group (e.g., 67) and trunk group (e.g., 67). When the Acme Packet SBC receives the REFER triggered by the Voice Portal blind transfer, the SBC will send an INVITE to Session Manager, and Session Manager will choose this SIP Entity to route the transferred call.



SIP Entity Details
Commit
Cancel
General
* Name: CM-Evolution-procr-5062
* FQDN or IP Address: 10.1.2.90
Type: CM
Notes: CM-ES procr IP, different port
Adaptation: CM-ES-VZ Inbound
Location:
Time Zone: America/New_York
Override Port & Transport with DNS SRV:
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration

6.5. Entity Links

To view or change Entity Links, select **Routing → Entity Links**. Click on the checkbox corresponding to the name of a link and **Edit** to edit an existing link, or the **New** button to add a link. Click the **Commit** button after changes are completed.

Compared with the configuration documented in reference [JRR-VZIPCC], the only new Entity Link is for Voice Portal. Other Entity Links are illustrated below for completeness.

Note – In the Entity Link configurations below (and in the Communication Manager SIP trunk configuration), TCP was selected as the transport protocol between Communication Manager and Session Manager in the sample configuration. TCP was used to facilitate trace analysis during network verification. TLS may be used in customer deployments.

6.5.1. Entity Link for Session Manager and Voice Portal

Compared with the configuration documented in reference [JRR-VZIPCC], the entity link shown below is the only addition. The Entity Link named "VoicePortal" represents the link between Session Manager and Voice Portal using the TCP protocol and port 5060.

Entity Links Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* VoicePortal	* SM1	TCP	* 5060	* VoicePortal	* 5060	☑	Verizon Testing

6.5.2. Entity Link for Session Manager and Acme Packet Net-Net

The following screen shows the entity link named "Acme1" linking Session Manager with the Acme Packet Net-Net SBC, using TCP and port 5060.

Entity Links Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* Acme1	* SM1	TCP	* 5060	* Acme1	* 5060	☑	

6.5.3. Entity Links for Session Manager to Communication Manager

As in [JRR-VZIPCC], two SIP Entity Links, using different TCP ports, link the same SM1 with Communication Manager. For one link, named "CM Evolution Server", both entities use port 5060. For the other, named "CM-ES-VZ-5062", both entities use port 5062.

The link named "CM Evolution Server" shown below links Session Manager "SM1" with the Communication Manager processor ethernet. This link existed in the shared configuration prior to adding the Verizon-related configuration. This link, using port 5060, can carry traffic between Session Manager and Communication Manager that is not necessarily related to calls with Verizon, such as traffic related to SIP Telephones registered to Session Manager, or traffic related to Avaya Modular Messaging, which has SIP integration to Session Manager.

Entity Links

Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* CM Evolution Server	* SM1	TCP	* 5060	* CM Evolution Server	* 5060	☑	

The link named "CM-ES-VZ-5062" shown below also links Session Manager "SM1" with the Communication Manager processor ethernet. However, this link uses port 5062 for both entities in the link. This link was created to allow Communication Manager to distinguish calls from Verizon from other calls that arrive from the same Session Manager. Other methods of distinguishing traffic could be used, if desired.

Entity Links

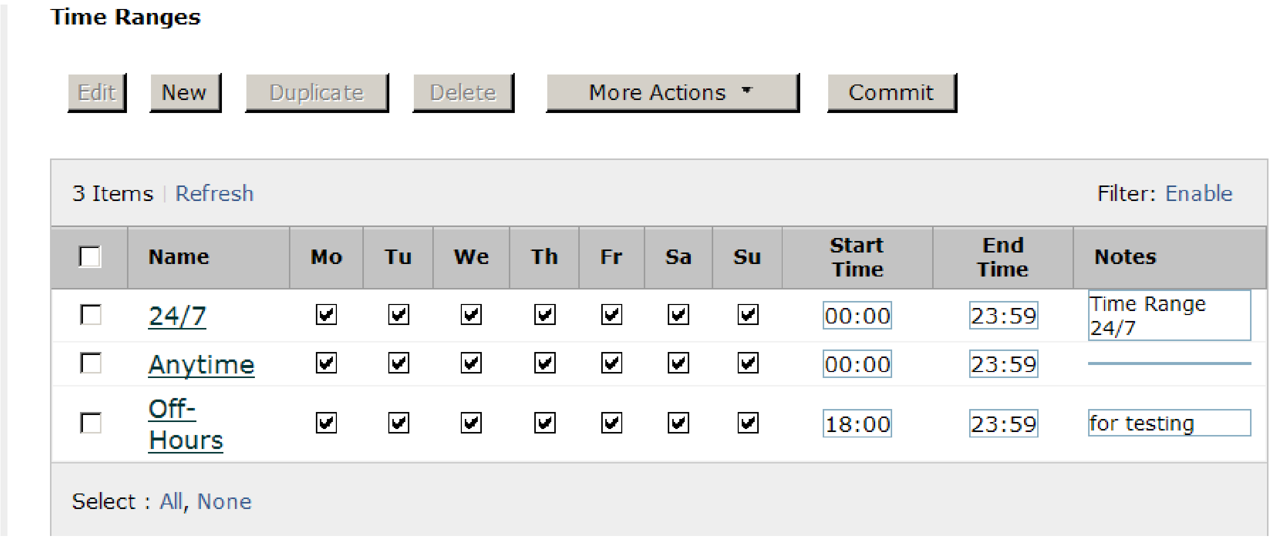
Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* CM-ES-VZ-5062	* SM1	TCP	* 5062	* CM-Evolution-procr-5062	* 5062	☑	Same IP, diff port

6.6. Time Ranges

To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the “24/7” range since time-based routing was not the focus of these Application Notes. Click the **Commit** button after changes are completed.



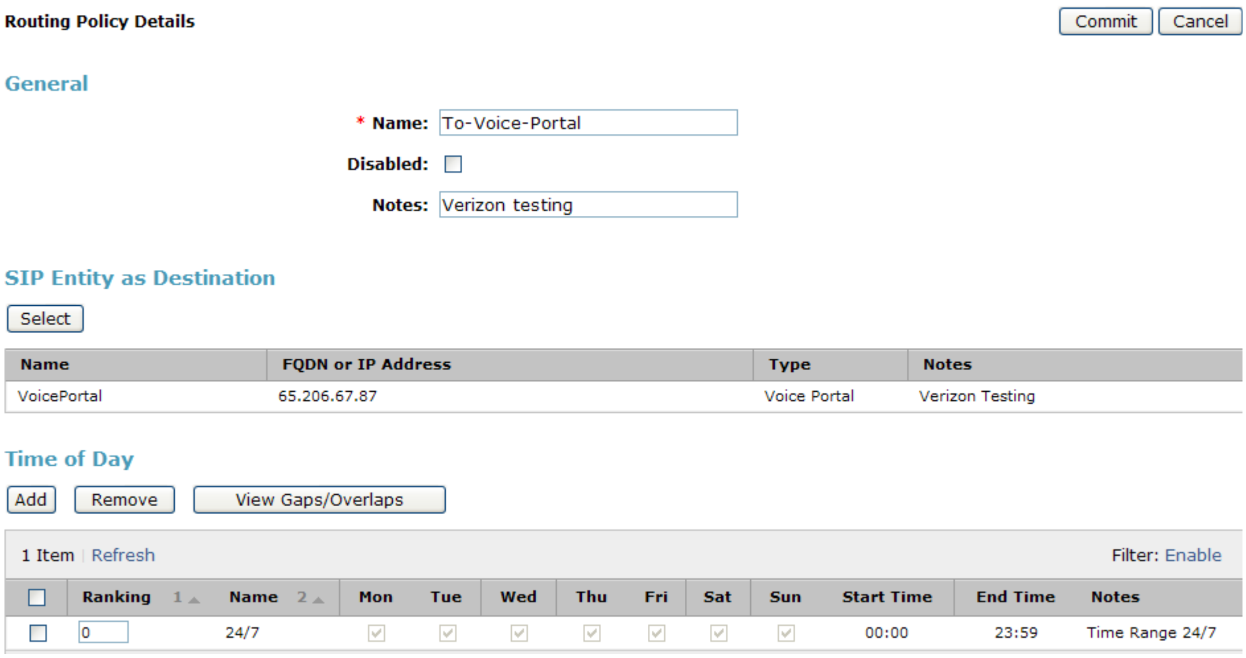
6.7. Routing Policies

To view or change routing policies, select **Routing → Policies**. Click on the checkbox corresponding to the name of a policy and **Edit** to edit an existing policy, or **New** to add a policy. Click the **Commit** button after changes are completed.

Compared with the configuration documented in reference [JRR-VZIPCC], the only new routing policy is for Voice Portal. Other routing policies are illustrated below for completeness.

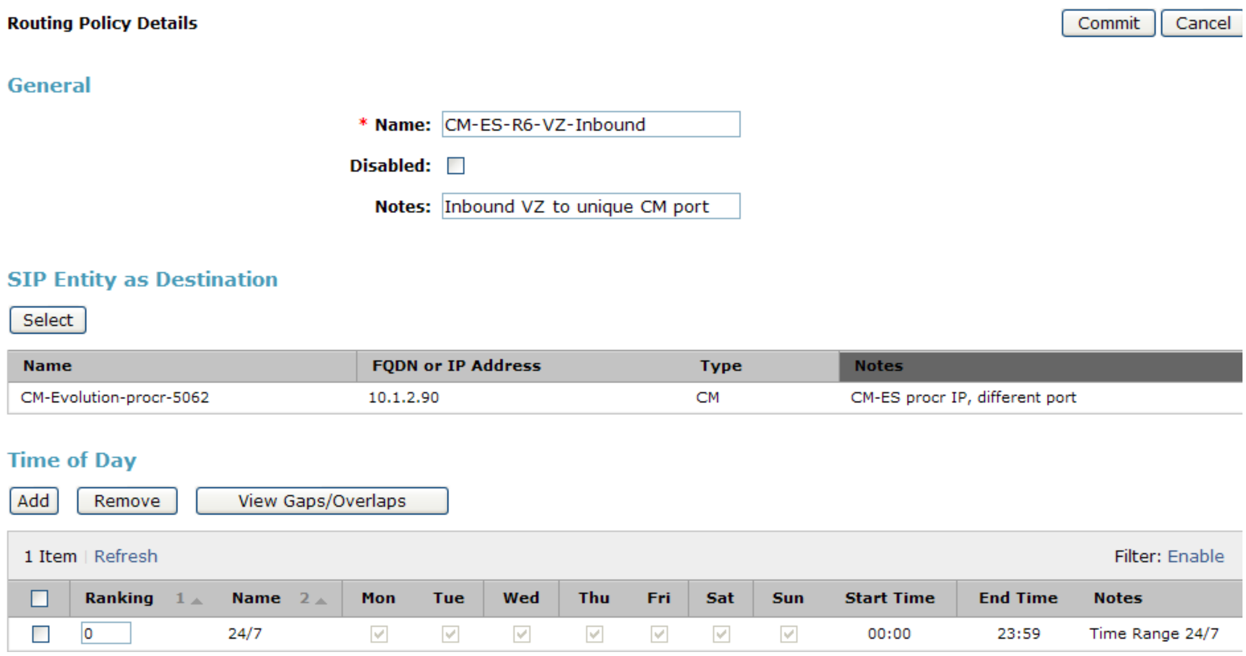
6.7.1. Routing Policy for Voice Portal

The following screen shows the **Routing Policy Details** for the policy named “To-Voice-Portal”. Note that the SIP Entity as Destination contains the SIP Entity “VoicePortal” configured in **Section 6.4.1**.



6.7.2. Verizon-Specific Routing Policy for Communication Manager

The following screen shows the **Routing Policy Details** for the policy named "CM-ES-R6-VZ-Inbound". This is the routing policy that directs calls to the Verizon-specific SIP entity, using a unique port 5062. Later, dial patterns will be defined for calls to be sent to this routing policy. The dial patterns to be sent to this routing policy will include calls transferred out of Voice Portal using REFER.



Routing Policy Details [Commit] [Cancel]

General

* Name: CM-ES-R6-VZ-Inbound

Disabled: ☐

Notes: Inbound VZ to unique CM port

SIP Entity as Destination

[Select]

Name	FQDN or IP Address	Type	Notes
CM-Evolution-procr-5062	10.1.2.90	CM	CM-ES procr IP, different port

Time of Day

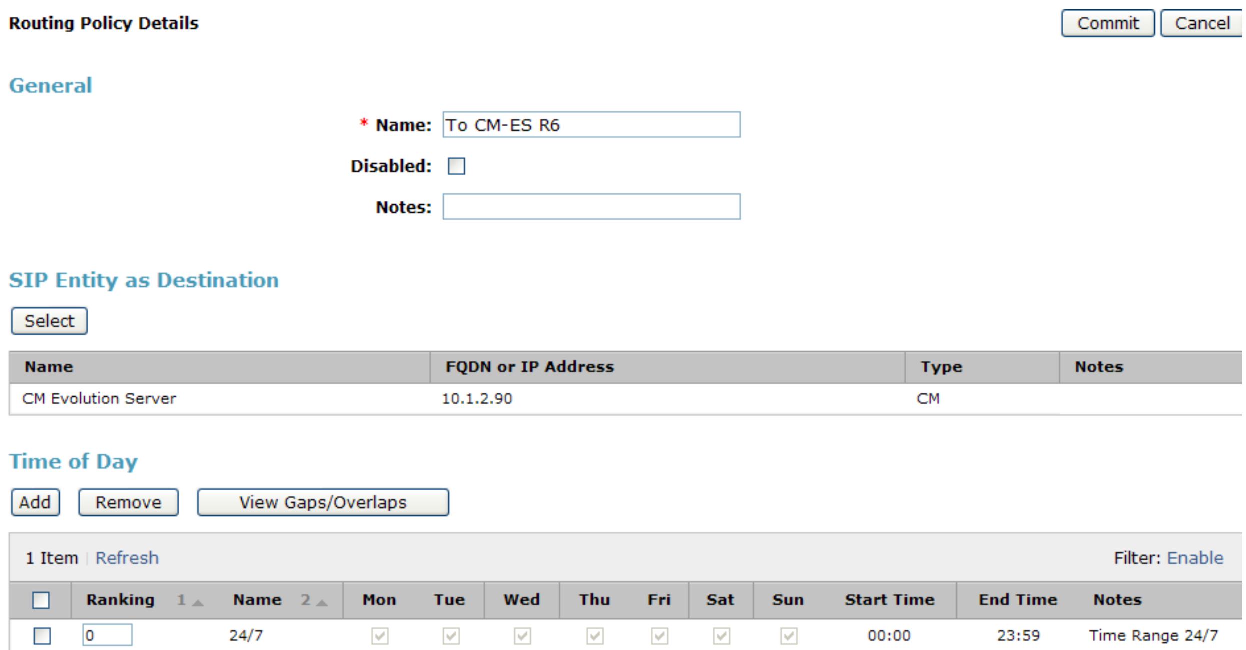
[Add] [Remove] [View Gaps/Overlaps]

1 Item | Refresh — Filter: Enable

☐	Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	0	24/7	✓	✓	✓	✓	✓	✓	✓	00:00	23:59	Time Range 24/7

6.7.3. General Routing Policy for Communication Manager

The following screen shows the **Routing Policy Details** for the policy named "To CM-ES R6". This is the routing policy that was in place prior to adding the Verizon-specific configuration. This routing policy will be associated with dial patterns for calls that are not related to Verizon, such as calls from Modular Messaging to Communication Manager. In the sample configuration, this policy may also be used for Verizon calls transferred out of Voice Portal using the "bridged transfer" method, since such a "transfer" is really a bridged call that remains in Voice Portal and does not result in a REFER being sent to the SBC.



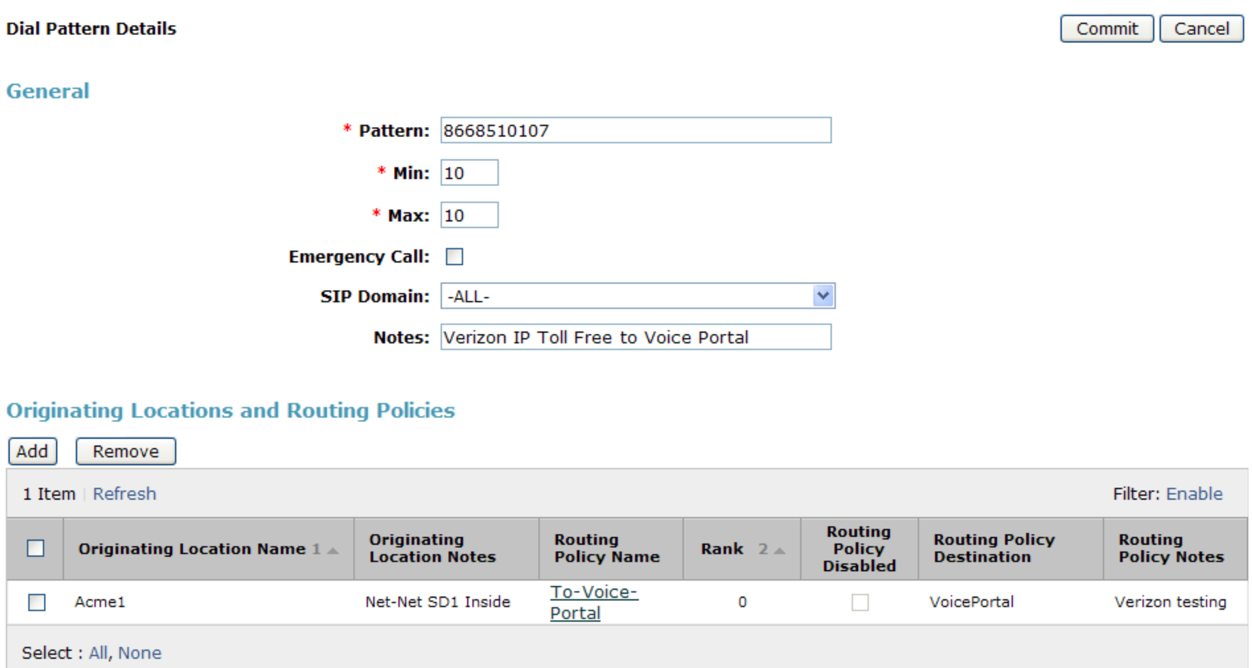
6.8. Dial Patterns

To view or change dial patterns, select **Routing → Dial Patterns**. Click on the checkbox corresponding to the name of a pattern and **Edit** to edit an existing pattern, or **New** to add a pattern. Click the **Commit** button after changes are completed.

Once Dial Patterns are configured that associate dialed numbers with routing policies, a return to the routing policy screen will list the Dial Patterns associated with the policy.

6.8.1. Dial Pattern for Calls from Verizon Directly to Voice Portal

The following screen illustrates an example dial pattern for a Verizon IP toll-free number that Session Manager will route to Voice Portal. In this case, the Pattern specifies "8668510107", a toll-free number designated for a Voice Portal self-service application. Under **Originating Locations and Routing Policies**, the **Originating Location Name** is "Acme1", the name of the location configured in **Section 6.2** and assigned to the Acme Packet Net-Net SIP entity in **Section 6.3**. The **Routing Policy Name** is "To-Voice-Portal", the name of the routing policy configured in **Section 6.7.1**.



Similar dial patterns can be defined for each Verizon IP Toll Free and Verizon IP IVR outdial number. Of course, if numbers are contiguous, one dial pattern can cover multiple numbers.

6.8.2. Dial Pattern for Calls Transferred by Voice Portal to a Communication Manager Extension

The following screen illustrates a dial pattern corresponding to a Communication Manager extension to which Voice Portal can transfer inbound calls from Verizon. In this case, the Pattern specifies "36880", a Vector Directory Number (VDN). Similar dial patterns can be specified covering other Communication Manager extensions to which Voice Portal can transfer inbound calls from Verizon. Under **Originating Locations and Routing Policies**, the **Originating Location Name** is "Acme1", the name of the location configured in **Section 6.2** and assigned to the Acme Packet Net-Net SIP entity in **Section 6.3**. The originating location is the SBC because the SBC is configured such that REFER messages generated by Voice Portal cause the SBC to send a new INVITE message to the Refer-To destination of the REFER, which is the target of the transfer (e.g., the VDN). The **Routing Policy Name** is "CM-ES-R6-VZ-Inbound", the name of the routing policy configured in **Section 6.7.2**. In sum, this dial pattern is used to allow calls originally routed from Verizon directly to Voice Portal to be transferred to Communication Manager extensions using the same signaling group and trunk group (i.e., 67) used for calls routed directly from Verizon to Communication Manager, as in reference [JRR-VZIPCC].

Dial Pattern Details

Commit Cancel

General

* **Pattern:** 36880
* **Min:** 5
* **Max:** 5
Emergency Call: ☐
SIP Domain: -ALL-
Notes: Transfer Out of VP to VDN 36880

Originating Locations and Routing Policies

Add Remove

1 Item | Refresh   Filter: Enable

☐	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	Acme1	Net-Net SD1 Inside	CM-ES-R6-VZ-Inbound	0	☐	CM-Evolution-procr-5062	Inbound VZ to unique CM port

6.8.3. Dial Patterns for Calls from Verizon Directly to Communication Manager

Consult reference [JRR-VZIPCC] for example dial patterns for any inbound Verizon toll-free calls that Session Manager should route directly to Communication Manager, rather than first routing to Voice Portal for a self-service opportunity.

7. Acme Packet Net-Net SBC

The Acme Packet Net-Net SBC configuration used in the verification of these Application Notes is the same as the configuration previously documented in [JRR-VZIPCC], except as noted in this section. Reference [JRR-VZIPCC] documents a similar configuration where the Acme Packet SBC communicates with the same Verizon IPCC Services on the public side of the SBC, and communicates with Session Manager on the private side of the SBC (i.e., as a hub for SIP communication with Communication Manager and Modular Messaging). In reference [JRR-VZIPCC], Avaya Voice Portal was not present in the configuration. In these Application Notes, Avaya Voice Portal is added to the configuration shown in [JRR-VZIPCC], and Voice Portal becomes the focus of the testing. Therefore, this section highlights the changes and additions to the Acme Packet SBC configuration required to execute the Voice Portal testing summarized in **Section 9**.

In the sample configuration, an Acme Packet 4250 Net-Net Session Border Controller is used as the edge device between the Avaya CPE and Verizon Business. Using similar configuration, the Acme Packet 3800 or 4500 platforms may be used. Reference [JF-VZIPCC] also includes detailed configuration steps covering an Acme Packet Net-Net SBC with Verizon IPCC Services.

7.1. Session Agent Change for Session Manager Release 6

The session agent configured for Session Manager Release 6 in Section 6.1 of reference [JRR-VZIPCC] is re-used in these Application Notes. Since the test objectives for these Application Notes include the testing of Voice Portal REFER-based transfers of Verizon toll-free calls to Communication Manager destinations, but do not include REFER-based transfers towards Verizon PSTN destinations, the "refer-call-transfer" parameter for this session agent is changed to "enabled". The relevant part of the session agent configuration is included below.

```
session-agent
        hostname                10.1.2.70
        ip-address              10.1.2.70
        port                    5060
        state                   enabled
        app-protocol            SIP
        transport-method        StaticTCP
        realm-id                INSIDE
        description             Session-Manager-R6
        allow-next-hop-lp       enabled
        loose-routing           enabled
        send-media-session      enabled
        ping-method             OPTIONS;hops=0
        ping-interval           60
        ping-send-mode          keep-alive
        options                 trans-timeouts=1
        refer-call-transfer     enabled
```

```
reuse-connections        TCP
tcp-keepalive            enabled
tcp-reconn-interval      10
```

7.2. SIP Manipulation For To Header Towards Voice Portal

As noted in **Section 1.3**, Voice Portal 5.1 requires that the host portion of the To header match the configured domain in Voice Portal. Since Session Manager 6.0 can not adapt the domain in the To header, the SBC is used to ensure that the domain in the To header matches the "avaya.com" domain shown in Voice Portal in **Section 4.4.**

The following header-rule is added to the SIP "out-manipulationid" applied to the "inside" realm towards Session Manager. The "new-value" in the element-rule is set to "avaya.com", as shown in bold below. In this case, "avaya.com" replaces "10.1.2.70" in the host portion of the To header, but the matching value shown below is not required nor intended to be prescriptive.

```
header-rule
        name                    Inbound_To
        header-name             To
        action                  manipulate
        comparison-type         case-sensitive
        msg-type                request
        methods
        match-value
        new-value
        element-rule
            name                    To
            parameter-name
            type                    uri-host
            action                  replace
            match-val-type          any
            comparison-type         case-sensitive
            match-value             10.1.2.70
            new-value               avaya.com
```

7.3. SIP Manipulation For From, Contact, PAI Headers Toward Voice Portal

As noted in **Section 1.3**, Verizon IP Toll Free and Verizon IP IVR services send a "+" preceding the calling party number in the From, PAI, and Contact headers of the SIP INVITE message. This "+" can be stripped by the SBC before reaching Voice Portal 5.1. If the "+" is not stripped from the From header, the Voice Portal will not send the original caller ID received from Verizon when a call is transferred using an INVITE-based transfer method (e.g., bridged transfer, or consultative transfer when INVITE with Replaces is used).

The following header-rules are added to the SIP "out-manipulationid" applied to the "inside" realm towards Session Manager. In each header-rule, the new-value of the header is the same as the old value, except that the "+" has been removed.

```
header-rule
                name                    modPAI
                header-name             P-Asserted-Identity
                action                  manipulate
                comparison-type         pattern-rule
                msg-type                any
                methods                 INVITE
                match-value
                new-value
                element-rule
                        name                    modVal
                        parameter-name
                        type                    uri-user
                        action                  find-replace-all
                        match-val-type          any
                        comparison-type         case-sensitive
                        match-value             \+(.*)
                        new-value               $modPAI.$modVal.$1
        header-rule
                name                    modContact
                header-name             Contact
                action                  manipulate
                comparison-type         pattern-rule
                msg-type                any
                methods                 INVITE
                match-value
                new-value
                element-rule
                        name                    modVal
                        parameter-name
                        type                    uri-user
                        action                  find-replace-all
                        match-val-type          any
                        comparison-type         case-sensitive
                        match-value             \+(.*)
                        new-value               $modContact.$modVal.$1
        header-rule
                name                    modFrom
                header-name             From
                action                  manipulate
                comparison-type         pattern-rule
```

```
        msg-type              any
        methods               INVITE
        match-value
        new-value
        element-rule
                name                  modVal
                parameter-name
                type                  uri-user
                action                find-replace-all
                match-val-type        any
                comparison-type       case-sensitive
                match-value           \+(.*)
                new-value             $modFrom.$modVal.$1
```

7.4. SIP Manipulation to Preserve User-to-User Information for REFER-based Transfers to Communication Manager

If User-to-User Information should be passed from Voice Portal to Communication Manager for REFER-based transfers (e.g., blind transfer, and consultative transfer using REFER), the following header-rules can be added to the configuration.

The following header rule should be added to the SIP "in-manipulationid" applied to the "inside" realm towards Session Manager. The approach is to store the contents of any "User-to-User" portion of the Refer-To header of a REFER message from Voice Portal after the literal text "UUI". Later, the SIP "out-manipulationid" will key off the literal text UUI so that the SBC can include the user to user information in the INVITE message generated when the refer-call-transfer option is enabled.

```
header-rule
        name                  requri
        header-name           Refer-To
        action                manipulate
        comparison-type       case-sensitive
        msg-type              request
        methods               REFER
        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                appenduriuser
        parameter-name
        type                uri-user
        action              replace
        match-val-type      any
        comparison-type     boolean
        match-value         $requri.$getUUI
        new-value           $ORIGINAL+UUI+$requri.$getUUI.$0
```

The following header rules should be added to the SIP "out-manipulationid" applied to the "inside" realm towards Session Manager. The header rules named "get_UUI", "get_UUI_To", and "get_UUI_Route" restore the original URI sans the literal text "UUI" and any user to user information. By specifying the match-value "(.*)(UUI)(.*)", the original URI is located in "$1", the information before the literal text "UUI", when UUI is present. The actual user to user information is in "$3" after the literal text "UUI".

```
header-rule
    name                get_UUI
    header-name         Request-URI
    action              manipulate
    comparison-type     case-sensitive
    msg-type            request
    methods             INVITE
    match-value
    new-value
    element-rule
        name                store_UUI
        parameter-name
        type                uri-user
        action              store
        match-val-type      any
        comparison-type     case-sensitive
        match-value         (.*)(UUI)(.*)
        new-value
    element-rule
        name                get_UUI
        parameter-name
        type                uri-user
        action              find-replace-all
        match-val-type      any
        comparison-type     case-sensitive
        match-value         (.*)(UUI)(.*)
        new-value           $get_UUI.$store_UUI.$1
```

```
header-rule
        name                    get_UUI_To
        header-name             To
        action                  manipulate
        comparison-type         case-sensitive
        msg-type                request
        methods                 INVITE
        match-value
        new-value
        element-rule
                name                    store_UUI
                parameter-name
                type                    uri-user
                action                  store
                match-val-type          any
                comparison-type         case-sensitive
                match-value             (.*)(UUI)(.*)
                new-value
        element-rule
                name                    get_UUI
                parameter-name
                type                    uri-user
                action                  find-replace-all
                match-val-type          any
                comparison-type         case-sensitive
                match-value             (.*)(UUI)(.*)
                new-value               $get_UUI_To.$store_UUI.$1

header-rule
        name                    get_UUI_Route
        header-name             Route
        action                  manipulate
        comparison-type         case-sensitive
        msg-type                any
        methods                 INVITE
        match-value
        new-value
        element-rule
                name                    store_UUI
                parameter-name
                type                    uri-user
                action                  store
                match-val-type          any
                comparison-type         case-sensitive
```

match-value (.*)(UUI)(.*)
new-value

element-rule
name get_UUI
parameter-name
type uri-user
action find-replace-all
match-val-type any
comparison-type case-sensitive
match-value (.*)(UUI)(.*)
new-value $get_UUI_Route.$store_UUI.$1

The following header rule inserts the User-to-User header into the INVITE message when UUI is present. The "$3" represents the actual user to user information.

header-rule
name add_UUI
header-name User-to-User
action add
comparison-type boolean
msg-type request
methods INVITE
match-value $get_UUI.$store_UUI
new-value $get_UUI.$store_UUI.$3

8. Verizon Business IPCC Services Suite Configuration

Information regarding Verizon Business IPCC Services suite offer can be found at http://www.verizonbusiness.com/products/contactcenter/ip/ or by contacting a Verizon Business sales representative.

The reference configuration described in these Application Notes was located in the Avaya Solutions and Interoperability Lab. Access to the Verizon Business IPCC Services suite was via a Verizon Private IP (PIP) T1 connection. Verizon Business provided all of the necessary service provisioning.

8.1. Service access information

The following service access information (FQDN, IP addressing, ports, toll free numbers) was provided by Verizon for the sample configuration. The numbers with the "*" are the numbers that are illustrated in the sample configuration in these Application Notes.

CPE (Avaya)	Verizon Network
adevc.avaya.globalipcom.com ***UDP port 5060***	***172.30.205.55*** ***UDP Port 5072***

IP Toll Free Numbers
866-850-2380
866-851-0107*
866-851-2649
866-852-3221
866-850-6850

IP IVR Published Number	IP IVR Outdial Number
866-616-4250*	866-851-8119*
866-616-4254	866-850-8170
866-616-4284	866-851-1977
866-616-4248	866-851-9157

9. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site with Avaya Voice Portal, Communication Manager, Session Manager, Avaya phones, and an Acme Packet Net-Net SBC.
- The production Verizon Business IPCC service, to which the simulated enterprise site was connected.

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

- Inbound Verizon IP Toll Free and IP IVR calls to Avaya Voice Portal applications.
- Caller interaction with Avaya Voice Portal applications, including caller DTMF input.
- Avaya Voice Portal applications transferring of inbound calls to Communication Manager skills / agents using blind transfer, consult transfer, and bridged transfer
- Transfer of User-to-User Information from Voice Portal to Communication Manager as part of transfer scenarios
- Call and two-way talk path establishment between callers and Communication Manager agents following transfers from Avaya Voice Portal.
- Use of G.729a codec (Verizon IP Toll Free only) and G.711 MU codec.

The above test objectives with limitations as noted in **Section 1.3** were verified successfully.

10. Verification Steps

10.1. 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 talk path exists. Interact with the Avaya Voice Portal prompts and verify that the call remains stable for several minutes and can be disconnected properly.

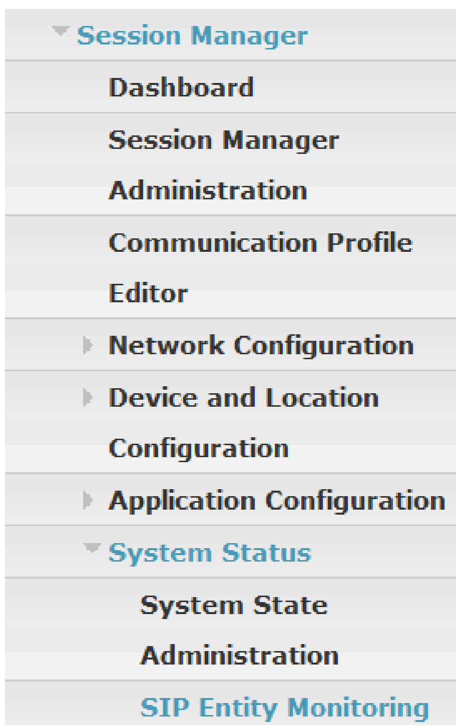
2. Place an inbound call to an Avaya Voice Portal application that can transfer an inbound call to a Communication Manager skill, and select the appropriate prompt(s) to request a transfer to an agent. Verify that the transfer completes successfully. Verify that when no agent in the skill is available, the caller hears appropriate wait treatment (e.g., an announcement). Verify that when an agent in the skill becomes available, the call is successfully routed to the agent and two-way talk path exists between the caller and the agent.

10.2. System Manager and Session Manager Verifications

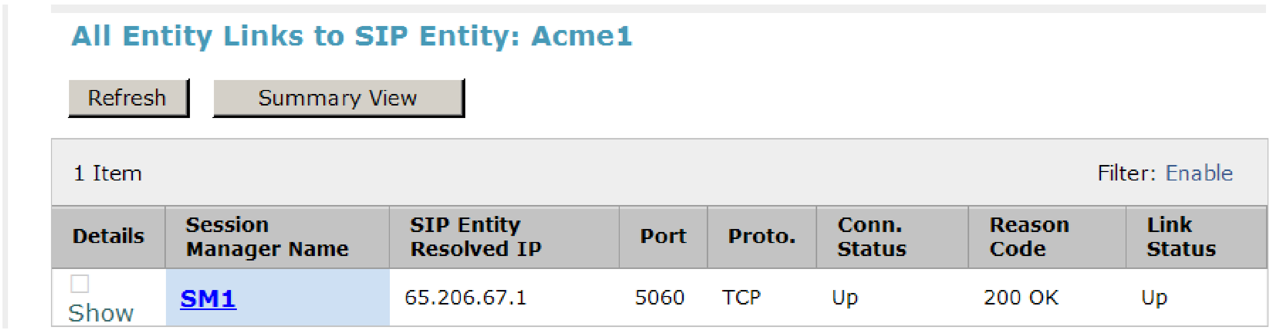
This section contains verification steps that may be performed using System Manager for Session Manager.

10.2.1. Verify SIP Entity Link Status

Log in to System Manager. Expand **Elements → Session Manager → System Status → SIP Entity Monitoring**, as shown below.



From the list of monitored entities, select an entity of interest, such as “Acme1”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below.



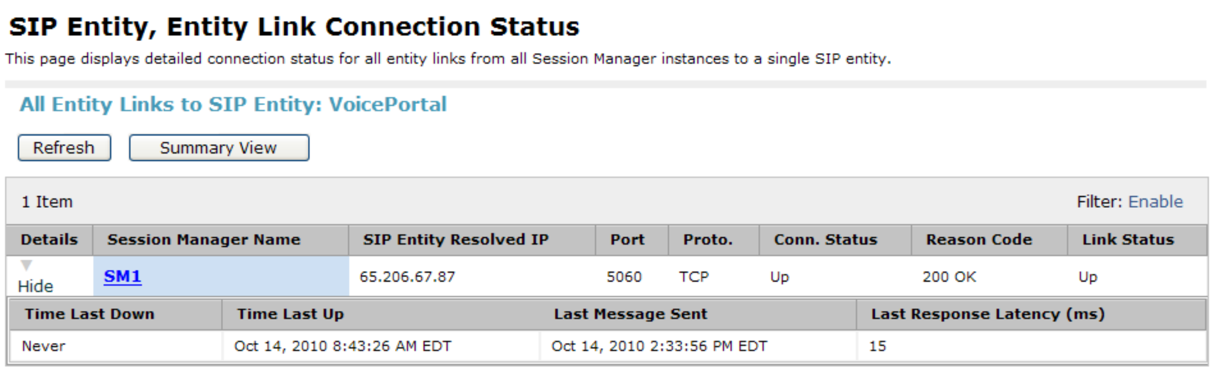
All Entity Links to SIP Entity: Acme1

Refresh | Summary View

1 Item — Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	SM1	65.206.67.1	5060	TCP	Up	200 OK	Up

Return to the list of monitored entities, and select another entity of interest, such as “VoicePortal”. Under normal operating conditions, the **Link Status** should be “Up” as shown in the example screen below. In this case, “Show” under Details was selected to view additional information.



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: VoicePortal

Refresh | Summary View

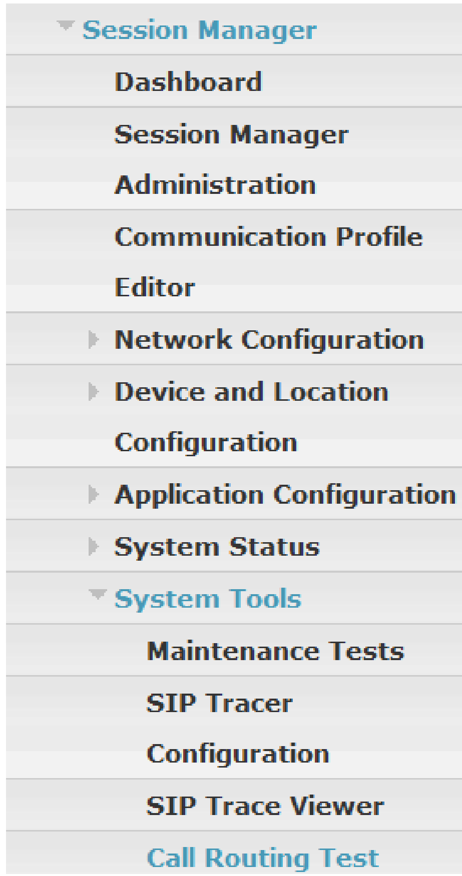
1 Item — Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Hide	SM1	65.206.67.87	5060	TCP	Up	200 OK	Up

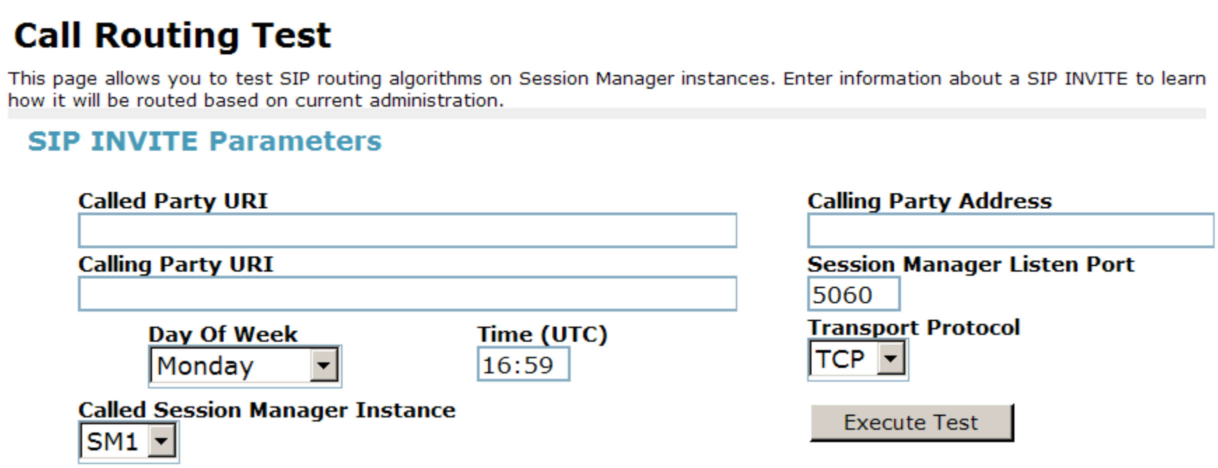
Time Last Down	Time Last Up	Last Message Sent	Last Response Latency (ms)
Never	Oct 14, 2010 8:43:26 AM EDT	Oct 14, 2010 2:33:56 PM EDT	15

10.2.2. Call Routing Test

The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, expand **Elements → Session Manager → System Tools → Call Routing Test**, as shown below.

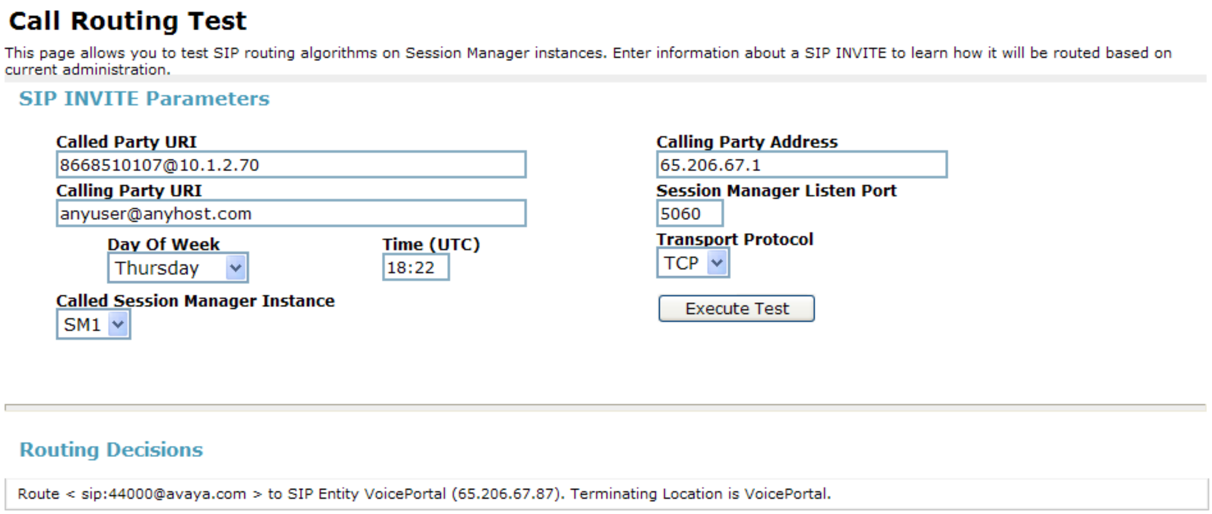


A screen such as the following is displayed.

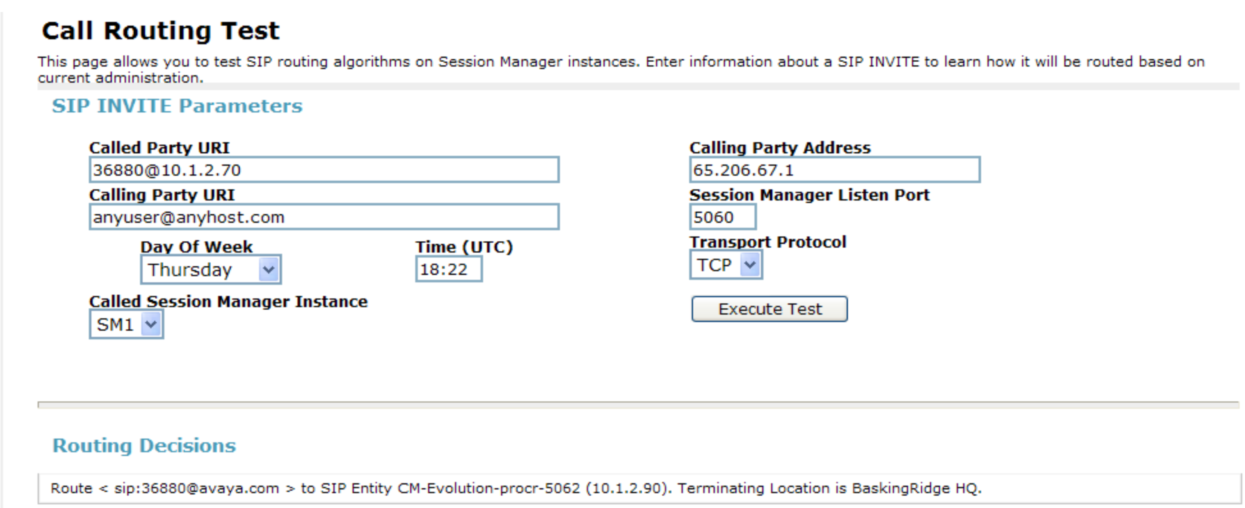


Populate the fields for the call parameters of interest and click **Execute Test**.

For example, the following shows a call routing test for an inbound toll-free call from the PSTN to Voice Portal via Acme1 (65.206.67.1). Under **Routing Decisions**, observe that the call will route to Voice Portal (65.206.67.87) using the SIP entity named "VoicePortal". The domain in the Request-URI is converted to "avaya.com", and the digits are manipulated such that the Verizon IP toll-free number (i.e., 866-851-0107) is converted to a number that will trigger a Voice Portal application (i.e., 44000). This adaptation is performed by the adapter assigned to the Voice Portal entity. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).

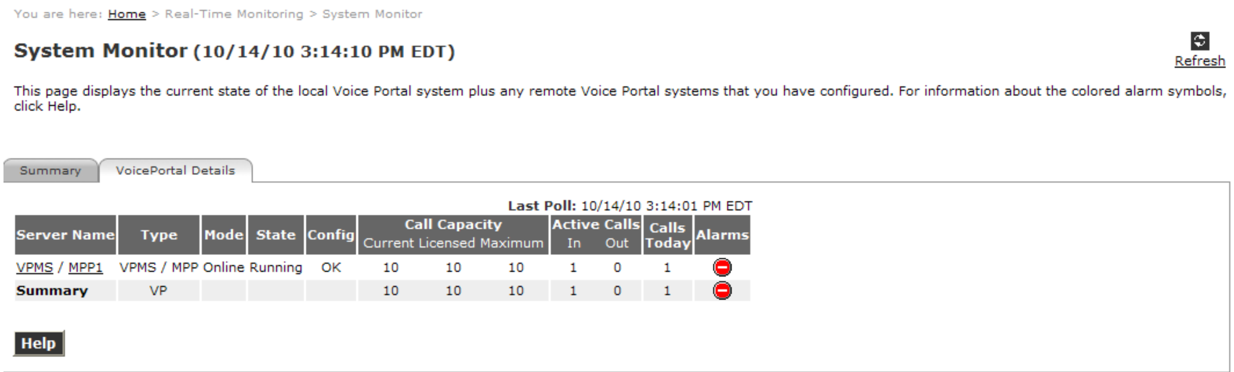


The following screen shows another example of a call routing test. In this case, the call routing test shows the results when a call is launched by the Acme Packet Net-Net SBC in response to a REFER from Voice Portal to Communication Manager Vector Directory Number (VDN) 36880. That is, when Voice Portal transfers the Verizon toll-free caller to VDN 36880 using REFER, Session Manager will receive an INVITE from the SBC with the number 36880. Session Manager will route this call to the SIP Entity "CM-Evolution-procr-5062" associated with the Verizon-specific trunk group 67 on Communication Manager.

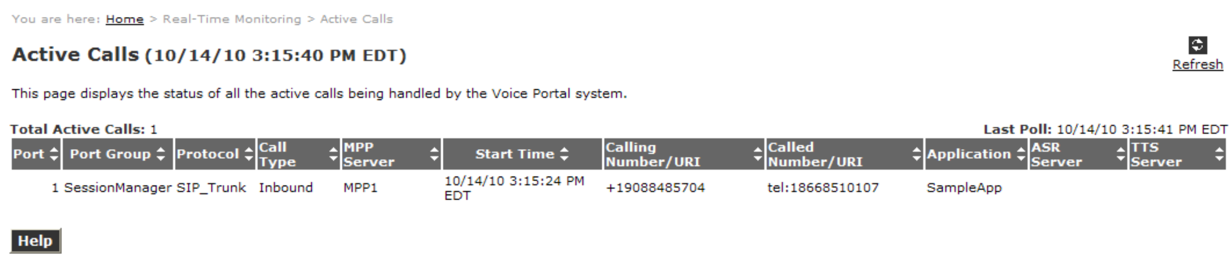


10.3. Voice Portal Verifications

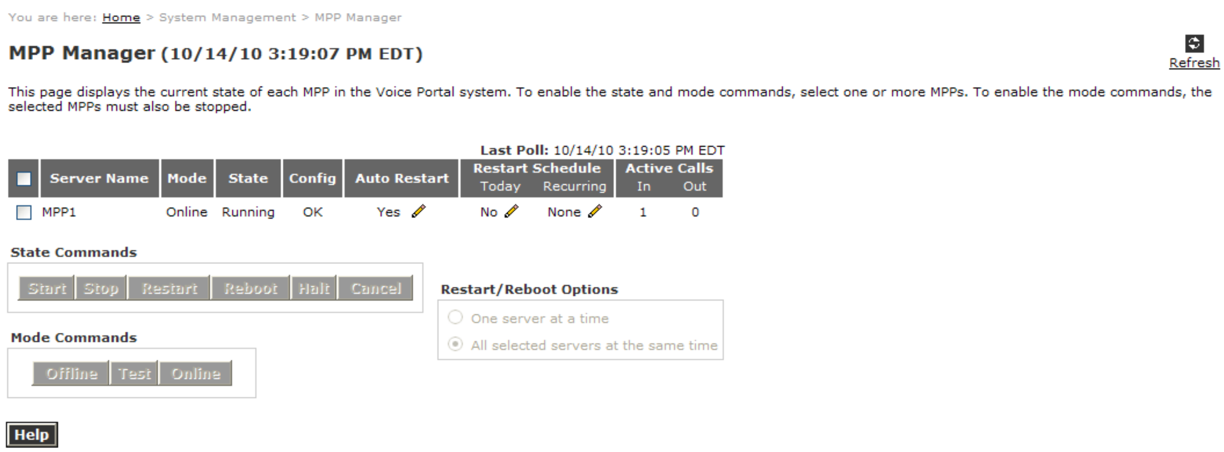
From the Voice Portal **Home** screen shown in **Section 4.3**, select **Real-Time Monitoring → System Monitor**. The following screen was captured while Voice Portal was processing a call.



From the Voice Portal **Home** screen shown in **Section 4.3**, select **Real-Time Monitoring → Active Calls**. The following screen was captured while Voice Portal was processing a call. As can be observed, the call was from PSTN caller 908-848-5704 to Verizon IP toll-free number 866-851-0107. The call is being processed by the Application "SampleApp" configured in **Section 4**.**5**. (Note that this screen was captured before the Acme SIP header manipulation shown in Section 7 to remove the + from the calling party number was applied).



From the Voice Portal **Home** screen shown in **Section 4.3**, select **System Management → MPP Manager**. In the resultant screen, the state of the MPP server(s) can be verified or managed, as shown below.



10.4. Troubleshooting Tools

The 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 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 can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

11. Wireshark Illustrations for Representative Calls

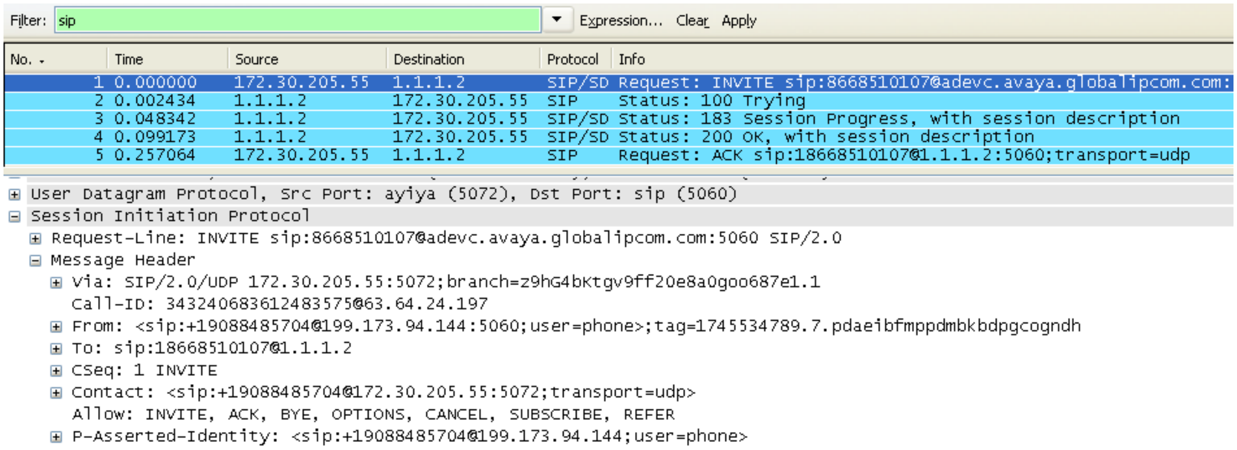
The following sub-sections illustrate the expected operation of the configuration shown in these Application Notes for a representative sampling of calls.

11.1. Inbound Verizon IPCC Calls that Remain in Voice Portal

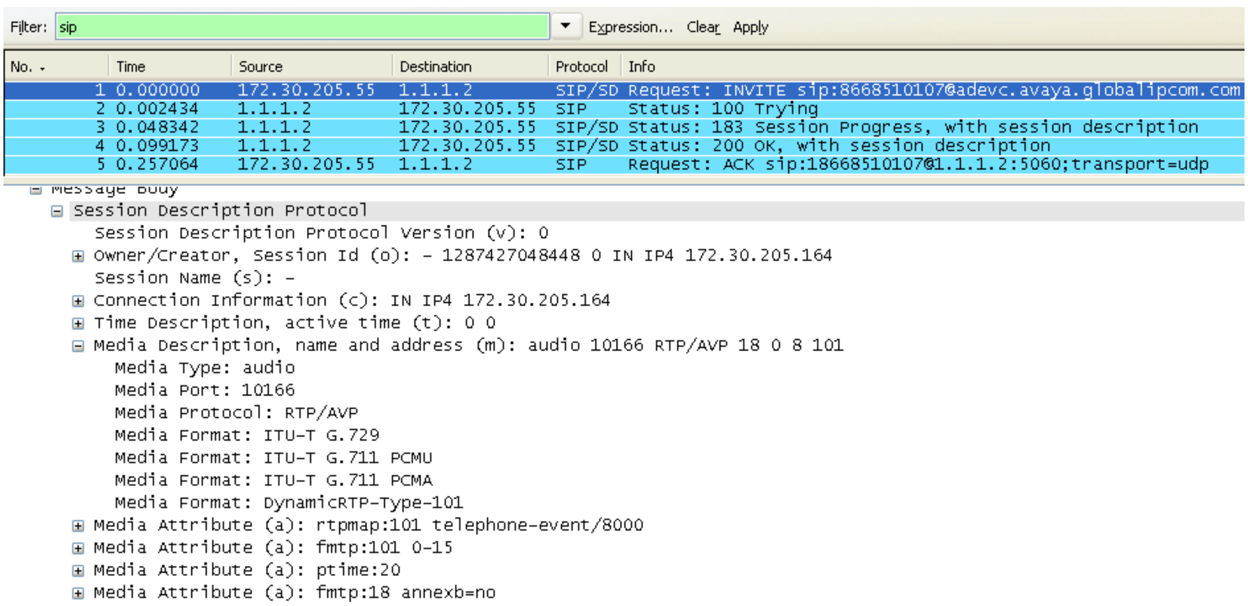
The traces in this section illustrate the handling of Verizon IPCC calls that are directed to Voice Portal for self-service. Subsequent sections that illustrate calls transferred by Voice Portal also use the call flows shown in this section, prior to the transfer request.

11.1.1. Verizon IP Toll Free Call to Voice Portal

The following portion of a filtered Wireshark trace taken on the public side of the Acme SBC shows the relevant SIP messaging. The caller dialed IP Toll Free number 866-851-0107. In highlighted frame 1, note that the INVITE message from Verizon IPCC (172.30.205.55) to the Acme Packet SBC (1.1.1.2) contains the enterprise domain known to Verizon "adevc.avaya.globalipcom.com" in the Request-URI. The To header contains the outside IP Address (1.1.1.2) of the SBC.



In this same trace, scrolling down in the center area for highlighted frame 1, the SDP information sent by Verizon for this IP Toll Free call can be observed. Note that Verizon offers G.729a as the first choice. The call will proceed using G.729a.



In this same trace, frame 4 is highlighted below to reveal the 200 OK sent when Voice Portal answers the call on the inside interface. Note that the call uses G.729A.

No.	Time	Source	Destination	Protocol	Info
1	0.000000	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668510107@adevc.avaya.globalipcom.com
2	0.002434	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
3	0.048342	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
4	0.099173	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
5	0.257064	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:18668510107@1.1.1.2:5060;transport=udp

```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 1.1.1.2
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 1.1.1.2
    ⊞ Time Description, active time (t): 0 0
    ⊟ Media Description, name and address (m): audio 49366 RTP/AVP 18 101
        Media Type: audio
        Media Port: 49366
        Media Protocol: RTP/AVP
        Media Format: ITU-T G.729
        Media Format: DynamicRTP-Type-101
    ⊞ Media Attribute (a): rtpmap:18 G729/8000
    ⊞ Media Attribute (a): fmtp:18 annexb=no
    ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

In this same trace, filtered differently, RTP can be observed. The call uses G.729a, and in the highlighted frame 464, an example RFC 2833 digit "5" can also be observed. In this case, the caller, upon being prompted for input from Voice Portal, entered the digit 5.

Filter: sip || rtp   Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
458	4.752619	172.30.205.164	1.1.1.2	RTP	PT=ITU-T G.729, SSRC=0x666AD467, Seq=16111, Time=27563904
459	4.767896	1.1.1.2	172.30.205.164	RTP	PT=ITU-T G.729, SSRC=0x32211CB2, Seq=31663, Time=689135230
460	4.772682	172.30.205.164	1.1.1.2	RTP	PT=ITU-T G.729, SSRC=0x666AD467, Seq=16112, Time=27564064
461	4.787902	1.1.1.2	172.30.205.164	RTP	PT=ITU-T G.729, SSRC=0x32211CB2, Seq=31664, Time=689135390
462	4.792317	172.30.205.164	1.1.1.2	RTP	PT=ITU-T G.729, SSRC=0x666AD467, Seq=16113, Time=27564224
463	4.807884	1.1.1.2	172.30.205.164	RTP	PT=ITU-T G.729, SSRC=0x32211CB2, Seq=31665, Time=689135550
464	4.812289	172.30.205.164	1.1.1.2	RTP EV	Payload type=RTP Event, DTMF Five 5
465	4.822407	172.30.205.164	1.1.1.2	RTP EV	Payload type=RTP Event, DTMF Five 5
466	4.827892	1.1.1.2	172.30.205.164	RTP	PT=ITU-T G.729, SSRC=0x32211CB2, Seq=31666, Time=689135710
467	4.832393	172.30.205.164	1.1.1.2	RTP EV	Payload type=RTP Event, DTMF Five 5

```
⊟ RFC 2833 RTP Event
    Event ID: DTMF Five 5 (5)
    0... .... = End of Event: False
    .0.. .... = Reserved: False
    ..00 1000 = Volume: 8
    Event Duration: 240
```

The following portion of a filtered Wireshark trace taken on the enterprise side of the Acme SBC shows the relevant SIP messaging. In highlighted frame 130, the SBC (65.206.67.1) sends the SIP INVITE message to Session Manager (10.1.2.70). In the center area of the trace, note that the To header contains "18668510107@avaya.com", since the SBC has manipulated the To header to ensure that the host part contained the "avaya.com" domain configured in Voice Portal. In frame 133, Session Manager sends the INVITE to Voice Portal (65.206.67.87). Note that Session Manager has adapted the Request-URI from the Verizon IP Toll Free number (866-851-0107) to the Voice Portal number 44000 triggering the sample application. The call proceeds with Voice Portal sending 183 Session Progress with SDP (frame 139) and 200 OK with SDP (frame 144). (Note that this trace was captured before the Acme SIP header manipulation shown in **Section 7** to remove the + from the calling party number was applied).

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1)   Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
130	4.938240	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:8668510107@10.1.2.70:5060;transport=tc
131	4.940114	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
133	4.983546	10.1.2.70	65.206.67.87	SIP/SD	Request: INVITE sip:44000@avaya.com:5060;transport=tcp, wi
136	4.989630	65.206.67.87	10.1.2.70	SIP	Status: 100 Trying
139	5.004297	65.206.67.87	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
142	5.006444	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
144	5.009672	65.206.67.87	10.1.2.70	SIP/SD	Status: 200 OK, with session description
150	5.083793	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
159	5.319206	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp
160	5.321259	10.1.2.70	65.206.67.87	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp

```
⊞ Request-Line: INVITE sip:8668510107@10.1.2.70:5060;transport=tcp SIP/2.0
⊟ Message Header
  ⊞ Via: SIP/2.0/TCP 65.206.67.1:5060;branch=z9hG4bKb2o8kr003grgues885g0.1
    Call-ID: 1404347059830231533@65.210.180.212
  ⊞ From: <sip:+19088485704@65.206.67.1:5060;user=phone>;tag=942921061.7.becndldnhjlnbleapoaimkdp
  ⊞ To: sip:18668510107@avaya.com
  ⊞ CSeq: 1 INVITE
  ⊞ Contact: <sip:+19088485704@65.206.67.1:5060;transport=tcp>
    Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
  ⊞ P-Asserted-Identity: <sip:+19088485704@65.206.67.1;user=phone>
```

In this same trace, frame 144 is highlighted to show the G.729a SDP information sent by Voice Portal.

No.	Time	Source	Destination	Protocol	Info
130	4.938240	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:8668510107@10.1.2.70:5060;transport=tcp
131	4.940114	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
133	4.983546	10.1.2.70	65.206.67.87	SIP/SD	Request: INVITE sip:44000@avaya.com:5060;transport=tcp, wit
136	4.989630	65.206.67.87	10.1.2.70	SIP	Status: 100 Trying
139	5.004297	65.206.67.87	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
142	5.006444	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
144	5.009672	65.206.67.87	10.1.2.70	SIP/SD	Status: 200 OK, with session description
150	5.083793	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
159	5.319206	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp
160	5.321259	10.1.2.70	65.206.67.87	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp

```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 65.206.67.87
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 65.206.67.87
    ⊞ Time Description, active time (t): 0 0
    ⊞ Media Description, name and address (m): audio 23032 RTP/AVP 18 101
    ⊞ Media Attribute (a): rtpmap:18 G729/8000
    ⊞ Media Attribute (a): fmtp:18 annexb=no
    ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

11.1.2. Verizon IP IVR Call to Voice Portal

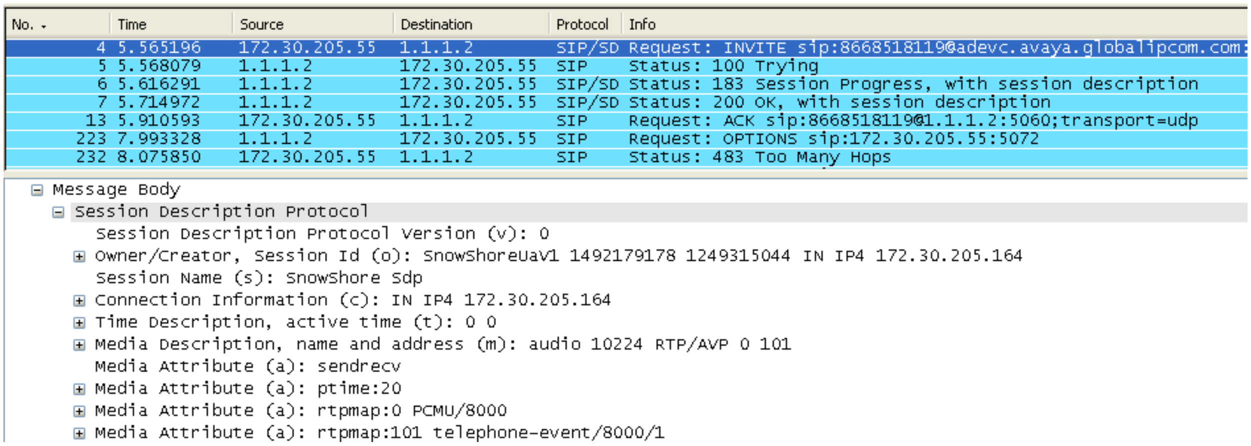
The following portion of a filtered Wireshark trace taken on the public side of the Acme SBC shows the relevant SIP messaging. The caller dialed IP-IVR published number 866-616-4250 and the call was delivered to the corresponding IP-IVR outdial number 8668518119. In the highlighted frame 4, note that the INVITE message from Verizon IPCC (172.30.205.55) to the Acme Packet SBC (1.1.1.2) contains the enterprise domain known to Verizon "adevc.avaya.globalipcom.com" in both the Request-URI and the To headers.

Filter: sip   Expression... Clear Apply

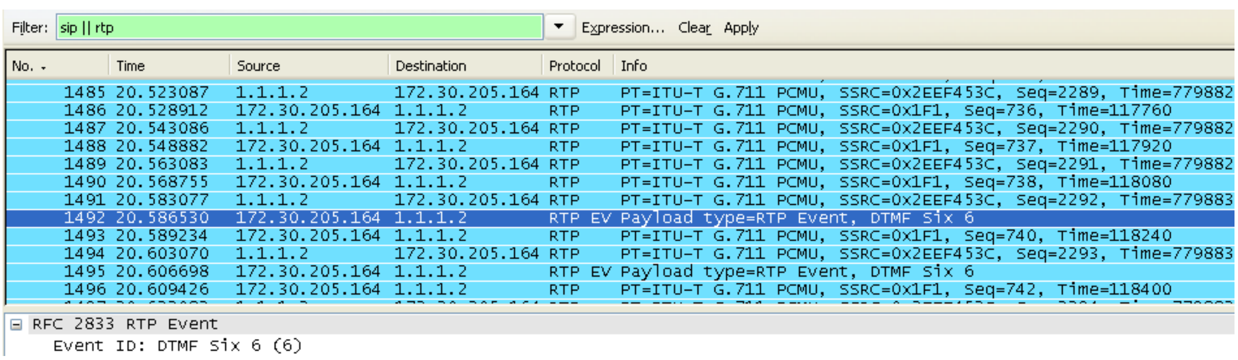
No.	Time	Source	Destination	Protocol	Info
4	5.565196	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668518119@adevc.avaya.globalipcom.com
5	5.568079	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
6	5.616291	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
7	5.714972	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
13	5.910593	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:8668518119@1.1.1.2:5060;transport=udp
223	7.993328	1.1.1.2	172.30.205.55	SIP	Request: OPTIONS sip:172.30.205.55:5072
232	8.075850	172.30.205.55	1.1.1.2	SIP	Status: 483 Too Many Hops

```
⊟ Session Initiation Protocol
  ⊞ Request-Line: INVITE sip:8668518119@adevc.avaya.globalipcom.com:5060 SIP/2.0
  ⊟ Message Header
    ⊞ Via: SIP/2.0/UDP 172.30.205.55:5072;branch=z9hG4bKqr523b1048phup8ev5p1.1
    ⊞ To: <sip:8668518119@adevc.avaya.globalipcom.com>
    ⊞ From: <sip:+19088485704@199.173.94.16:5060;user=phone>;tag=2528c7fe
```

In this same trace, scrolling down in the center area for highlighted frame 4, the SDP information sent by Verizon for this IP IVR call can be observed. In this case, only G.711MU is offered. The call will proceed using G.711MU.



In this same trace, filtered differently, RTP can be observed. The call uses G.711MU, and in the highlighted frame 1492, an example RFC 2833 digit "6" can also be observed. In this case, the caller, upon being prompted for input from Voice Portal, entered the digit 6.



The following portion of a filtered Wireshark trace taken on the enterprise side of the Acme SBC shows the relevant SIP messaging. In highlighted frame 1865, the SBC (65.206.67.1) sends the SIP INVITE message to Session Manager (10.1.2.70). In the center area of the trace, note that the To header contains "8668518119@avaya.com", since the SBC has manipulated the To header to ensure that the host part contained the "avaya.com" domain configured in Voice Portal. In frame 1869, Session Manager sends the INVITE to Voice Portal (65.206.67.87). Note that Session Manager has adapted the Request-URI from the Verizon IP IVR outdial number (866-851-8119) to the Voice Portal number 44000 triggering the sample application. The call proceeds with Voice Portal sending 183 Session Progress with SDP (frame 1877) and 200 OK with SDP (frame 1896). (Note that this screen was captured before the Acme SIP header manipulation shown in Section 7 to remove the + from the calling party number was applied).

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
1865	15.303977	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:8668518119@10.1.2.70:5060;transport=tcp,
1866	15.306826	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
1869	15.311280	10.1.2.70	65.206.67.87	SIP/SD	Request: INVITE sip:44000@avaya.com:5060;transport=tcp, with
1871	15.322500	65.206.67.87	10.1.2.70	SIP	Status: 100 Trying
1877	15.335063	65.206.67.87	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
1879	15.340739	65.206.67.87	10.1.2.70	SIP/SD	Status: 200 OK, with session description
1882	15.340972	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
1896	15.441000	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
1931	15.643038	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:8668518119@65.206.67.87;transport=tcp
1932	15.645548	10.1.2.70	65.206.67.87	SIP	Request: ACK sip:8668518119@65.206.67.87;transport=tcp

```
⊞ To: <sip:8668518119@avaya.com>
⊞ From: <sip:+19088485704@65.206.67.1:5060;user=phone>;tag=2528c7fe
```

11.2. Inbound Verizon IPCC Calls Blind Transferred by Voice Portal to a VDN on Communication Manager

The call flows in this section assume that the call flow shown in **Section 11.1** has already been completed. After Voice Portal answers and the self-service application interacts with the caller, Voice Portal uses blind transfer to transfer the call to Communication Manager Vector Directory Number 36880, whose corresponding vector programming is shown in **Section 5.10**.

11.2.1. Verizon IP Toll Free Call Blind Transferred by Voice Portal

The following portion of a filtered Wireshark trace taken on the enterprise side of the Acme SBC shows the relevant SIP messaging, picking up after the initial call had been answered by Voice Portal. Voice Portal requests a blind transfer to Communication Manager VDN 36880.

In highlighted frame 899, Voice Portal sends a SIP REFER to Session Manager. In the center area of the trace, note that the Refer-To header contains 36880@avaya.com. In frame 900, Session Manager sends the REFER message to the Acme Packet Net-Net SBC, from which the call initially arrived from Verizon. In frame 903, the Acme Packet SBC sends a SIP INVITE message whose Request-URI contains the number 36880, extracted from the Refer-To header.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
899	40.536645	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=t
900	40.539680	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=t
901	40.548244	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
903	40.549714	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transpor

```
    User-Agent: Avaya-VoicePortal/5.1.0.0.4206
  ⊟ Contact: <sip:18668510107@65.206.67.87;transport=tcp>
    ⊟ Contact Binding: <sip:18668510107@65.206.67.87;transport=tcp>
      ⊟ URI: <sip:18668510107@65.206.67.87;transport=tcp>
          SIP contact address: sip:18668510107@65.206.67.87
    Refer-To: <sip:36880@avaya.com;user=phone>
```

In this same trace, frame 903 is highlighted below, where the center area illustrates portions of the SIP INVITE sent by the Acme Packet SBC. Note that the From and PAI headers contain the PSTN caller's identity (908-848-5704), allowing Communication Manager access to the true caller id. (Note that this trace was captured before the Acme SIP header manipulation shown in Section 7 to remove the + from the calling party number was applied).

No.	Time	Source	Destination	Protocol	Info
899	40.536645	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=
900	40.539680	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=
901	40.548244	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
903	40.549714	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transpo

```
⊞ From: <sip:+19088485704@65.206.67.1:5060;user=phone>;tag=1745534789.1.pdaeibmmojhgkngclfnmoocn
⊞ To: <sip:36880@avaya.com;user=phone>
⊞ CSeq: 2 INVITE
⊞ Contact: <sip:+19088485704@65.206.67.1:5060;transport=tcp>
  Allow: INVITE, ACK, BYE, OPTIONS, CANCEL, SUBSCRIBE, REFER
⊞ P-Asserted-Identity: <sip:+19088485704@65.206.67.1;user=phone>
```

Changing the filtering options to include Communication Manager, the following portion of the same trace shows that Session Manager sends the INVITE to Communication Manager using the SIP Entity to Communication Manager used for calls from Verizon (i.e., using port 5062, associated with Communication Manager trunk group 67). This can be observed in highlighted frame 909, where the center area shows the INVITE sent to destination port 5062. Session Manager also adapts the domain in the Request-URI to avaya.com.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) || (sip Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
903	40.549714	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transpo
905	40.549728	10.1.2.70	65.206.67.87	SIP	Status: 202 Accepted
907	40.551554	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
909	40.557721	10.1.2.70	10.1.2.90	SIP/SD	Request: INVITE sip:36880@avaya.com:5060;user=phone;transpo

```
⊟ Transmission Control Protocol, Src Port: 51075 (51075), Dst Port: 5062 (5062), Seq: 1462, Ack: 1, Len: 145
```

Scrolling down in this same trace, highlighted frame 918 shows Communication Manager answering the call due to the announcement step in the call vector (80) associated with VDN 36880. In the center area, note that the SDP refers to the gateway (10.1.2.95) that plays the announcement. The media for the call can still use G.729a.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) || (sip Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
914	40.560481	10.1.2.90	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
918	40.561146	10.1.2.90	10.1.2.70	SIP/SD	Status: 200 OK, with session description
921	40.591172	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
923	40.595856	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
924	40.600690	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
925	40.603227	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,

```
⊟ Session Description Protocol
    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 10.1.2.90
    Session Name (s): -
  ⊞ Connection Information (c): IN IP4 10.1.2.95
  ⊞ Bandwidth Information (b): AS:64
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 2058 RTP/AVP 18 101
  ⊞ Media Attribute (a): rtpmap:18 G729/8000
  ⊞ Media Attribute (a): fmtp:18 annexb=no
  ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

Scrolling down in this same trace, in frame 923, the SBC receives the 200 OK answering the transferred call. In highlighted frame 924, the SBC sends a NOTIFY message where the center area shows this is for a 200 OK. In frame 925, Voice Portal receives this NOTIFY, and in frame 928, Voice Portal sends a BYE.

```
No.  Time       Source        Destination    Protocol Info
921  40.591172  10.1.2.70     65.206.67.1    SIP/SD   Status: 183 Session Progress, with session description
923  40.595856  10.1.2.70     65.206.67.1    SIP/SD   Status: 200 OK, with session description
924  40.600690  65.206.67.1   10.1.2.70      SIP/si   Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp
925  40.603227  10.1.2.70     65.206.67.87   SIP/si   Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp
927  40.606425  65.206.67.87  10.1.2.70      SIP      Status: 200 OK
928  40.606434  65.206.67.87  10.1.2.70      SIP      Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
930  40.607890  10.1.2.70     65.206.67.1    SIP      Status: 200 OK

⊟ Message Body
  ⊟ Sipfrag
      SIP/2.0 200 OK
```

Scrolling down in the same trace, after the Communication Manager agent answers the call, Communication Manager “shuffles” the media paths such that the final media path is direct from the answering agent to the inside IP address of the Acme Packet SBC. In highlighted frame 1178 below, note that the SDP contains the IP address of the answering agent, 65.206.67.11.

```
No.   Time       Source        Destination   Protocol Info
1158  50.014097  10.1.2.90     10.1.2.70     SIP      Request: INVITE sip:+19088485704@65.206.67.1:5060;transport
1160  50.016231  10.1.2.70     65.206.67.1   SIP      Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc
1164  50.055088  10.1.2.70     10.1.2.90     SIP      Status: 100 Trying
1167  50.095888  10.1.2.70     65.206.67.1   SIP      Request: INVITE sip:+19088485704@65.206.67.1:5060;transport
1169  50.099125  65.206.67.1   10.1.2.70     SIP      Status: 100 Trying
1171  50.367994  65.206.67.1   10.1.2.70     SIP/SD   Status: 200 OK, with session description
1173  50.369574  10.1.2.70     10.1.2.90     SIP/SD   Status: 200 OK, with session description
1176  50.370680  10.1.2.90     10.1.2.70     SIP/SD   Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc
1178  50.373044  10.1.2.70     65.206.67.1   SIP/SD   Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc

    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): - 1 3 IN IP4 10.1.2.90
    Session Name (s): -
  ⊞ Connection Information (c): IN IP4 65.206.67.11
  ⊞ Bandwidth Information (b): AS:384
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 2504 RTP/AVP 18 101
  ⊞ Media Attribute (a): rtpmap:18 G729/8000
  ⊞ Media Attribute (a): fmtp:18 annexb=no
  ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

The following trace taken on the outside or public side of the SBC reveals the SIP signaling with Verizon. In frame 2, Verizon sends the INVITE to the enterprise for the call to the IP Toll Free number. In highlighted frame 5, the 200 OK is illustrated, with the center area showing the SDP using source port 49300.

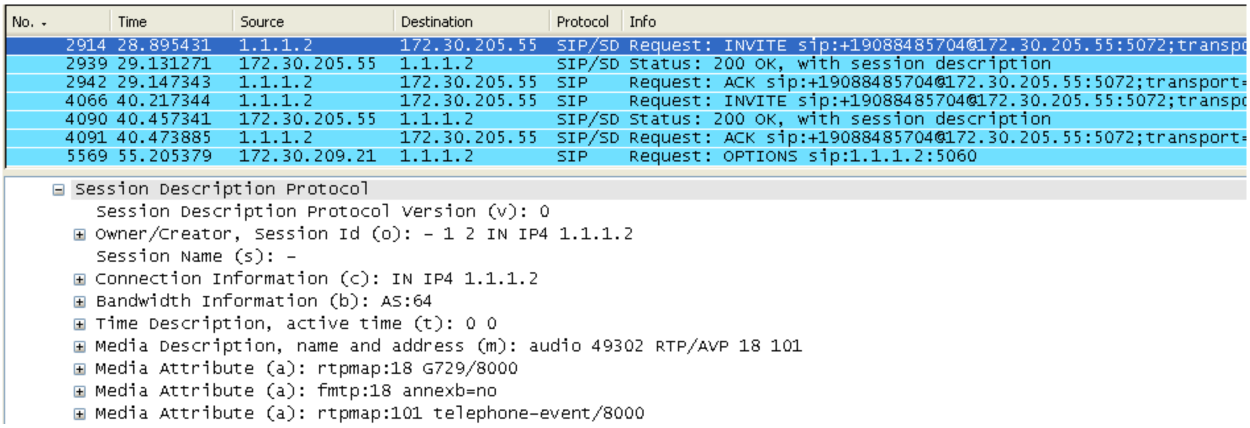
```
Filter: sip                                  Expression... Clear Apply

No.  Time      Source          Destination     Protocol Info
2    0.161012  172.30.205.55   1.1.1.2         SIP/SD   Request: INVITE sip:8668510107@adevc.avaya.globalipcom.com
3    0.163578  1.1.1.2         172.30.205.55   SIP      Status: 100 Trying
4    0.250985  1.1.1.2         172.30.205.55   SIP/SD   Status: 183 Session Progress, with session description
5    0.294089  1.1.1.2         172.30.205.55   SIP/SD   Status: 200 OK, with session description
6    0.497138  172.30.205.55   1.1.1.2         SIP      Request: ACK sip:18668510107@1.1.1.2:5060;transport=udp

⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 1.1.1.2
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 1.1.1.2
    ⊞ Time Description, active time (t): 0 0
    ⊞ Media Description, name and address (m): audio 49300 RTP/AVP 18 101
    ⊞ Media Attribute (a): rtpmap:18 G729/8000
    ⊞ Media Attribute (a): fmtp:18 annexb=no
    ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

Scrolling down in this same trace, after the transfer is completed and the announcement answers in the Communication Manager call vector, in highlighted frame 2914, the SBC sends an

INVITE, and the center area shows the SDP updated to use port 49302. Media containing the announcement from the Communication Manager gateway is now flowing to Verizon. When the call subsequently rings at a Communication Manager agent phone, ringback tone can be heard by the caller. When the Communication Manager agent answers, Communication Manager can “shuffle” the media for the call so that the final media path is “ip-direct” from the agent telephone to the inside interface of the SBC. This “shuffle INVITE” corresponds to frame 4066 in the trace below. Note that no REFER message is sent to Verizon.



No.	Time	Source	Destination	Protocol	Info
2914	28.895431	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
2939	29.131271	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
2942	29.147343	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
4066	40.217344	1.1.1.2	172.30.205.55	SIP	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
4090	40.457341	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
4091	40.473885	1.1.1.2	172.30.205.55	SIP/SD	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
5569	55.205379	172.30.209.21	1.1.1.2	SIP	Request: OPTIONS sip:1.1.1.2:5060

```
⊟ Session Description Protocol
    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 1.1.1.2
    Session Name (s): -
  ⊞ Connection Information (c): IN IP4 1.1.1.2
  ⊞ Bandwidth Information (b): AS:64
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 49302 RTP/AVP 18 101
  ⊞ Media Attribute (a): rtpmap:18 G729/8000
  ⊞ Media Attribute (a): fmtp:18 annexb=no
  ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

11.2.2. Verizon IP IVR Call Blind Transferred by Voice Portal

The following portion of a filtered Wireshark trace taken on the enterprise side of the Acme SBC shows the relevant SIP messaging, picking up after the initial call had been answered by Voice Portal. Voice Portal requests a blind transfer to Communication Manager VDN 36880.

In highlighted frame 5086, Voice Portal sends a SIP REFER to Session Manager. In the center area of the trace, note that the Refer-To header contains 36880@avaya.com. In frame 5097, Session Manager sends the REFER message to the Acme Packet Net-Net SBC, from which the call initially arrived from Verizon. In frame 5101, the Acme Packet SBC sends a SIP INVITE message whose Request-URI contains the number 36880, extracted from the Refer-To header. (Note that this trace was captured before the Acme SIP header manipulation shown in Section 7 to remove the + from the calling party number was applied).

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
5096	42.908139	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=tc
5097	42.911323	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=tc
5098	42.919150	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
5101	42.920628	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transport
5103	42.924252	10.1.2.70	65.206.67.87	SIP	Status: 202 Accepted
5105	42.926015	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
5128	42.966237	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
5136	43.028897	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
5137	43.033861	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp, wi
5138	43.036614	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp, wi

```
⊞ Contact: <sip:8668518119@65.206.67.87;transport=tcp>
  Refer-To: <sip:36880@avaya.com;user=phone>
```

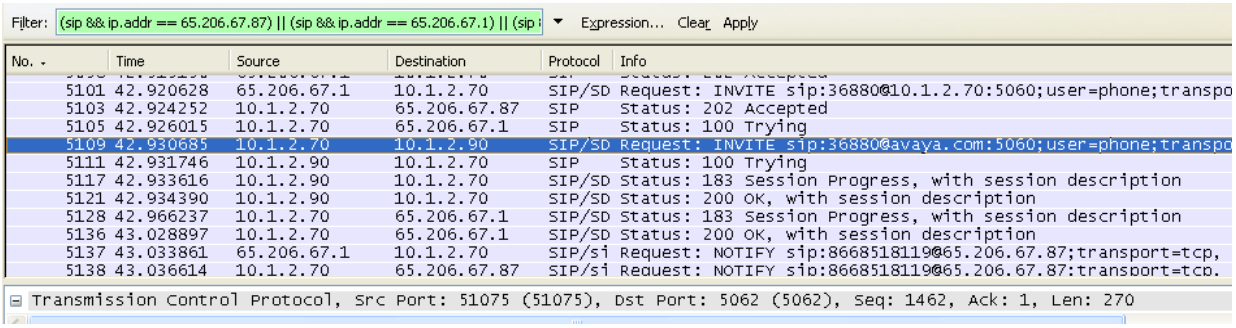
In this same trace, frame 5101 is highlighted below, where the center area illustrates portions of the SIP INVITE sent by the Acme Packet SBC. Note that the From and PAI headers contain the PSTN caller's identity (908-848-5704), allowing Communication Manager access to the true caller id.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) Expression... Clear Apply

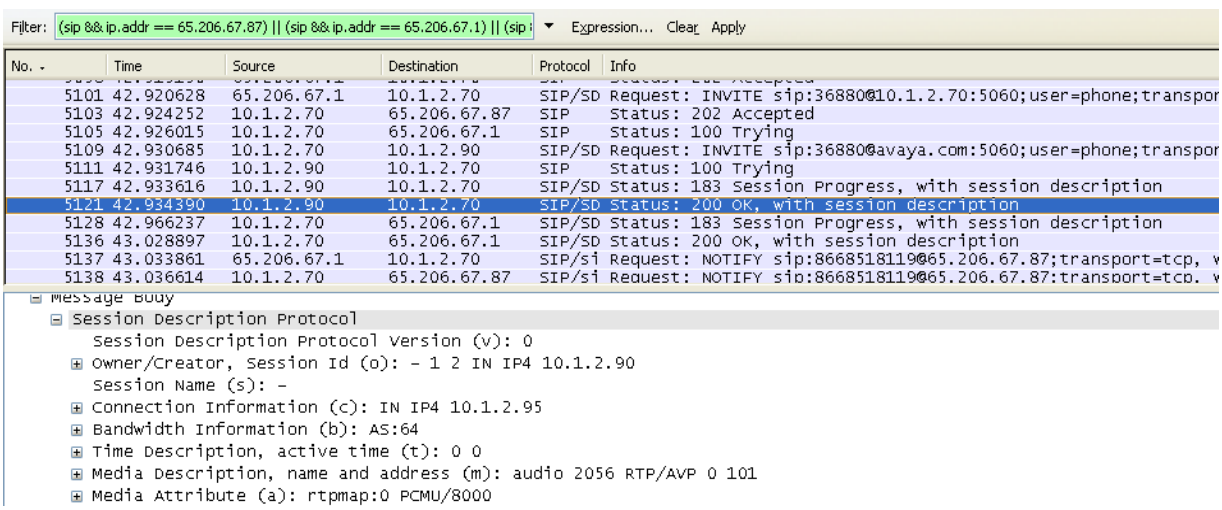
No.	Time	Source	Destination	Protocol	Info
5096	42.908139	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=t
5097	42.911323	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=t
5098	42.919150	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
5101	42.920628	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transpor

```
⊞ Request-Line: INVITE sip:36880@10.1.2.70:5060;user=phone;transport=tcp SIP/2.0
⊟ Message Header
  ⊞ Via: SIP/2.0/TCP 65.206.67.1:5060;branch=z9hG4bK8320ha20b061ses8m2o0.1
  ⊞ To: <sip:36880@avaya.com;user=phone>
  ⊞ From: <sip:+19088485704@65.206.67.1:5060;user=phone>;tag=647df55c
    Accept: application/sdp
    Accept-Encoding:
    Accept-Language: en
    Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, PRACK, REFER, SUBSCRIBE
    Allow-Events: refer
  ⊞ Contact: <sip:+19088485704@65.206.67.1:5060;transport=tcp>
    Date: Wed, 20 Oct 2010 19:39:39 GMT
    Max-Forwards: 64
  ⊞ P-Asserted-Identity: <sip:+19088485704@65.206.67.1;user=phone>
```

Changing the filtering options to include Communication Manager, the following portion of the same trace shows that Session Manager sends the INVITE to Communication Manager using the SIP Entity to Communication Manager used for calls from Verizon (i.e., using port 5062, associated with Communication Manager trunk group 67). This can be observed in highlighted frame 5109, where the center area shows the INVITE sent to destination port 5062. Session Manager also adapts the domain in the Request-URI to avaya.com.



Scrolling down in this same trace, highlighted frame 5121 shows Communication Manager answering the call due to the announcement step in the call vector (80) associated with VDN 36880. In the center area, note that the SDP refers to the gateway (10.1.2.95) that plays the announcement.



Scrolling down in this same trace, in frame 5136, the SBC receives the 200 OK answering the transferred call. In highlighted frame 5137, the SBC sends a NOTIFY message where the center area shows this is for a 200 OK. In frame 5138, Voice Portal receives this NOTIFY, and in frame 5173, Voice Portal sends a BYE.

No.	Time	Source	Destination	Protocol	Info
5136	43.028897	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
5137	43.033861	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp,
5138	43.036614	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp,
5143	43.045030	65.206.67.87	10.1.2.70	SIP	Status: 200 OK
5145	43.046734	10.1.2.70	65.206.67.1	SIP	Status: 200 OK
5146	43.048038	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:18668512649@10.1.2.90:5062;transport=tcp
5147	43.050276	10.1.2.70	10.1.2.90	SIP	Request: ACK sip:18668512649@10.1.2.90:5062;transport=tcp
5173	43.248904	65.206.67.87	10.1.2.70	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
5174	43.251193	10.1.2.70	65.206.67.1	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
5175	43.254362	65.206.67.1	10.1.2.70	SIP	Status: 200 OK
5177	43.255690	10.1.2.70	65.206.67.87	SIP	Status: 200 OK

```
⊟ Message Body
  ⊟ Sipfrag
      SIP/2.0 200 OK
```

Scrolling down in the same trace, after the Communication Manager agent answers the call, Communication Manager "shuffles" the media paths such that the final media path is direct from the answering agent to the inside IP address of the Acme Packet SBC. In highlighted frame 6435 below, note that the SDP contains the IP address of the answering agent, 65.206.67.11.

No.	Time	Source	Destination	Protocol	Info
6392	52.900470	10.1.2.70	65.206.67.1	SIP	Request: INVITE sip:+19088485704@65.206.67.1:5060;transpor
6394	52.903764	65.206.67.1	10.1.2.70	SIP	Status: 100 Trying
6431	53.166692	65.206.67.1	10.1.2.70	SIP/SD	Status: 200 Ok, with session description
6433	53.168573	10.1.2.70	10.1.2.90	SIP/SD	Status: 200 Ok, with session description
6435	53.169931	10.1.2.90	10.1.2.70	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=t
6442	53.208201	10.1.2.70	65.206.67.1	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=t

```
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 1 3 IN IP4 10.1.2.90
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 65.206.67.11
    ⊞ Bandwidth Information (b): AS:384
    ⊞ Time Description, active time (t): 0 0
    ⊞ Media Description, name and address (m): audio 2504 RTP/AVP 0 101
    ⊞ Media Attribute (a): rtpmap:0 PCMU/8000
```

The following trace taken on the outside or public side of the SBC reveals the SIP signaling with Verizon. In frame 9, the Verizon sends the INVITE to the enterprise for the call to the IP IVR outdial number. In highlighted frame 12, the 200 OK is illustrated, with the center area showing the SDP using source port 49216.

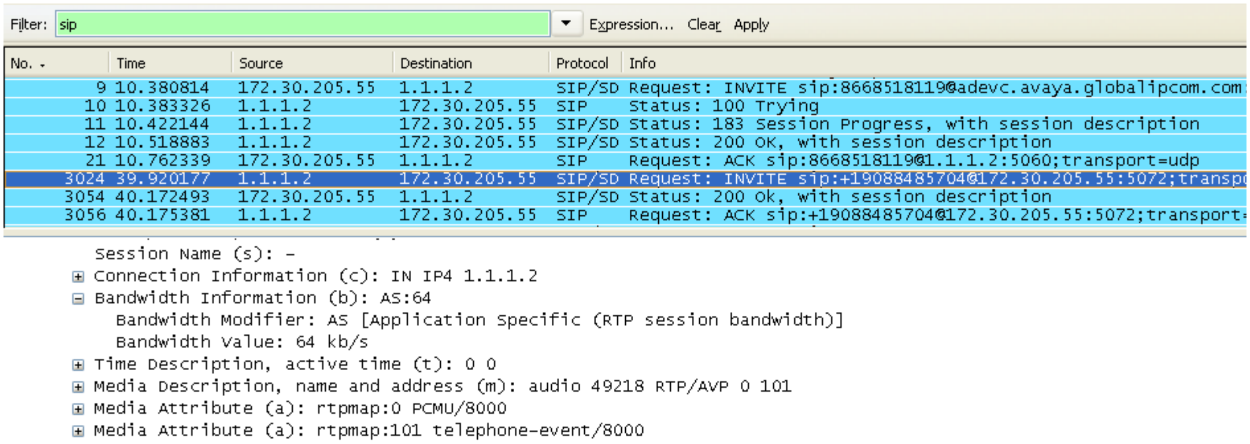
Filter: sip　Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
9	10.380814	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668518119@adevc.avaya.globalipcom.com:
10	10.383326	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
11	10.422144	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
12	10.518883	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
21	10.762339	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:8668518119@1.1.1.2:5060;transport=udp
3024	39.920177	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3054	40.172493	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 Ok, with session description
3056	40.175381	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=

```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 1 2 IN IP4 1.1.1.2
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 1.1.1.2
    ⊞ Time Description, active time (t): 0 0
    ⊞ Media Description, name and address (m): audio 49216 RTP/AVP 0 101
    ⊞ Media Attribute (a): rtpmap:0 PCMU/8000
```

After the transfer is completed and the announcement answers in the Communication Manager call vector, in highlighted frame 3024, the SBC sends an INVITE, and the center area shows the

SDP updated to use source port 49218. Media containing the announcement from the Communication Manager gateway is now flowing to Verizon.

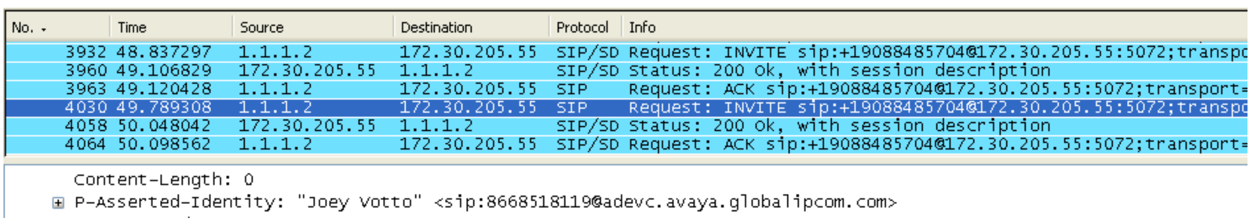


Filter: sip Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
9	10.380814	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668518119@adevc.avaya.globalipcom.com
10	10.383326	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
11	10.422144	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
12	10.518883	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
21	10.762339	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:8668518119@1.1.1.2:5060;transport=udp
3024	39.920177	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3054	40.172493	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3056	40.175381	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=

```
  Session Name (s): -
⊞ Connection Information (c): IN IP4 1.1.1.2
⊟ Bandwidth Information (b): AS:64
    Bandwidth Modifier: AS [Application Specific (RTP session bandwidth)]
    Bandwidth Value: 64 kb/s
⊞ Time Description, active time (t): 0 0
⊞ Media Description, name and address (m): audio 49218 RTP/AVP 0 101
⊞ Media Attribute (a): rtpmap:0 PCMU/8000
⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

Scrolling down in the same trace, highlighted frame 4030 corresponds to the Communication Manager “shuffle INVITE” (no SDP, Content-Length: 0), where the PAI contains the identity of the answering agent.



No.	Time	Source	Destination	Protocol	Info
3932	48.837297	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3960	49.106829	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3963	49.120428	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
4030	49.789308	1.1.1.2	172.30.205.55	SIP	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
4058	50.048042	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
4064	50.098562	1.1.1.2	172.30.205.55	SIP/SD	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=

```
  Content-Length: 0
⊞ P-Asserted-Identity: "Joey Votto" <sip:8668518119@adevc.avaya.globalipcom.com>
```

Note that no REFER message is sent to Verizon.

11.3. Inbound Verizon IPCC Calls Consultative Transferred by Voice Portal to a VDN on Communication Manager using REFER

The traces in this section illustrate Consultative Transfer using the "REFER" method, as shown in the configuration screen in **Section 4.4**.

11.3.1. Verizon IP Toll Free Call Consult Transferred by Voice Portal

The following is an example trace taken from the outside interface. Since this trace does not reveal new behaviors compared with traces illustrated previously, this outside trace is summarized, but the detailed view into individual messages is not provided. Frame 1 is the initial inbound IP Toll Free call from Verizon. This call is answered by Voice Portal in frame 4, and the caller interacts with the Voice Portal application. On the inside interface, Voice Portal initiates a consult transfer using REFER to VDN 36880, which answers the call with an announcement. When the announcement answers, the SBC sends the INVITE in frame 2803. Once the agent is selected and answers, Communication Manager "shuffles" the call to "ip-direct" on the inside interface, ultimately resulting in the "shuffle INVITE" in frame 3619.

Filter: sip — Expression... Clear Apply

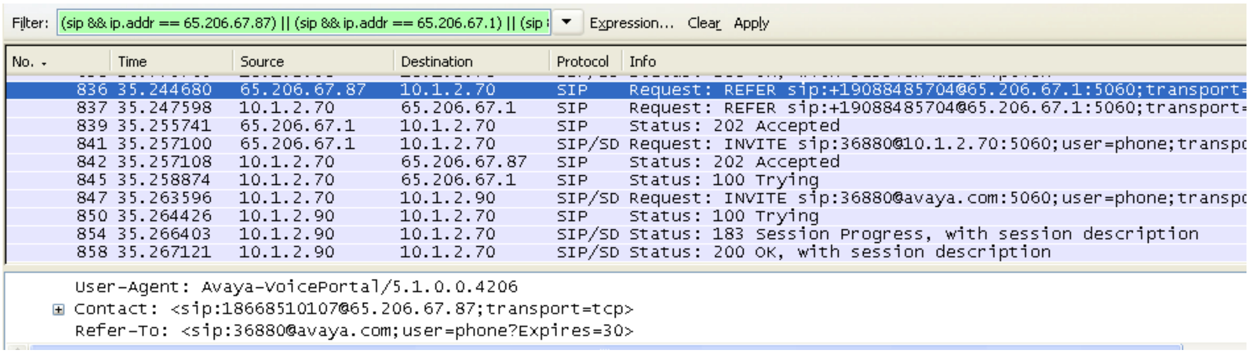
No.	Time	Source	Destination	Protocol	Info
1	0.000000	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668510107@adevc.avaya.globalipcom.com:
2	0.002498	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
3	0.052592	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
4	0.070141	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
5	0.260857	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:18668510107@1.1.1.2:5060;transport=udp
2780	27.397629	1.1.1.2	172.30.205.55	SIP	Request: OPTIONS sip:172.30.205.55:5072
2789	27.480852	172.30.205.55	1.1.1.2	SIP	Status: 483 Too Many Hops
2803	27.571804	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
2828	27.769355	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
2829	27.772173	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
3590	35.244500	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3611	35.434322	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3613	35.447928	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
3619	35.490122	1.1.1.2	172.30.205.55	SIP	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3643	35.754322	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3644	35.770242	1.1.1.2	172.30.205.55	SIP/SD	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=

The following portion of a filtered Wireshark trace taken on the inside interface shows the arrival of the call from Verizon through the answer of the call by Voice Portal. Since this is the same as prior illustrations, no elaboration is provided.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) — Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
194	7.792525	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:8668510107@10.1.2.70:5060;transport=tcp
195	7.794982	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
197	7.800882	10.1.2.70	65.206.67.87	SIP/SD	Request: INVITE sip:44000@avaya.com:5060;transport=tcp, wit
200	7.809383	65.206.67.87	10.1.2.70	SIP	Status: 100 Trying
204	7.823024	65.206.67.87	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
207	7.828430	65.206.67.87	10.1.2.70	SIP/SD	Status: 200 OK, with session description
210	7.830915	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
213	7.850898	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
217	8.048091	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp
218	8.050589	10.1.2.70	65.206.67.87	SIP	Request: ACK sip:18668510107@65.206.67.87;transport=tcp

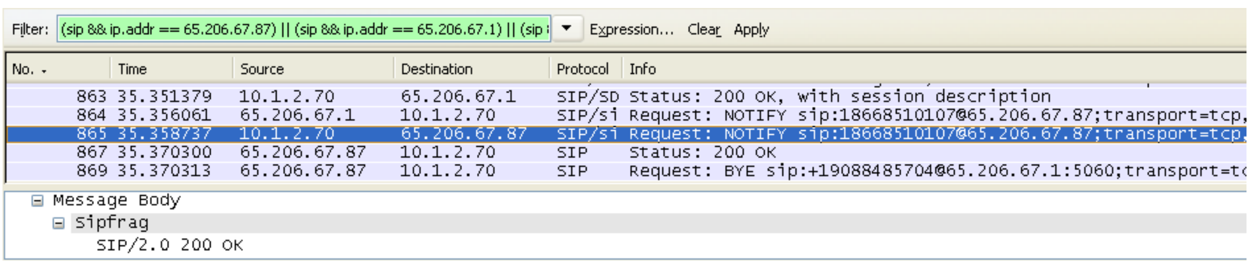
Scrolling down in this same trace, highlighted frame 836 shows Voice Portal sending the REFER, since in this case, Voice Portal was configured to use REFER for Consult Transfer. Since Voice Portal is using REFER for Consult Transfer, this trace is not substantially different from the one previously illustrated for blind transfer, so fewer details are provided. In the center area, note that the Refer-To contains 36880, the VDN to which Voice Portal is transferring the call. In frame 841, the SBC sends the INVITE to 36880. In frame 847, Session Manager sends the INVITE to Communication Manager using the SIP Entity corresponding to the Verizon-specific behaviors on Communication Manager, since the originating entity is the SBC and a dial pattern exists to sends calls from the SBC to the VDN to Communication Manager using the Verizon-specific entity. In frame 854, Communication Manager sends 183 with SDP, evidence that the Verizon-specific entity has been used. In frame 858, Communication Manager answers the call, since the vector corresponding to this VDN begins with an announcement step.



No.	Time	Source	Destination	Protocol	Info
836	35.244680	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=
837	35.247598	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport=
839	35.255741	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
841	35.257100	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transpo
842	35.257108	10.1.2.70	65.206.67.87	SIP	Status: 202 Accepted
845	35.258874	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
847	35.263596	10.1.2.70	10.1.2.90	SIP/SD	Request: INVITE sip:36880@avaya.com:5060;user=phone;transpo
850	35.264426	10.1.2.90	10.1.2.70	SIP	Status: 100 Trying
854	35.266403	10.1.2.90	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
858	35.267121	10.1.2.90	10.1.2.70	SIP/SD	Status: 200 OK, with session description

```
User-Agent: Avaya-VoicePortal/5.1.0.0.4206
Contact: <sip:18668510107@65.206.67.87;transport=tcp>
Refer-To: <sip:36880@avaya.com;user=phone?Expires=30>
```

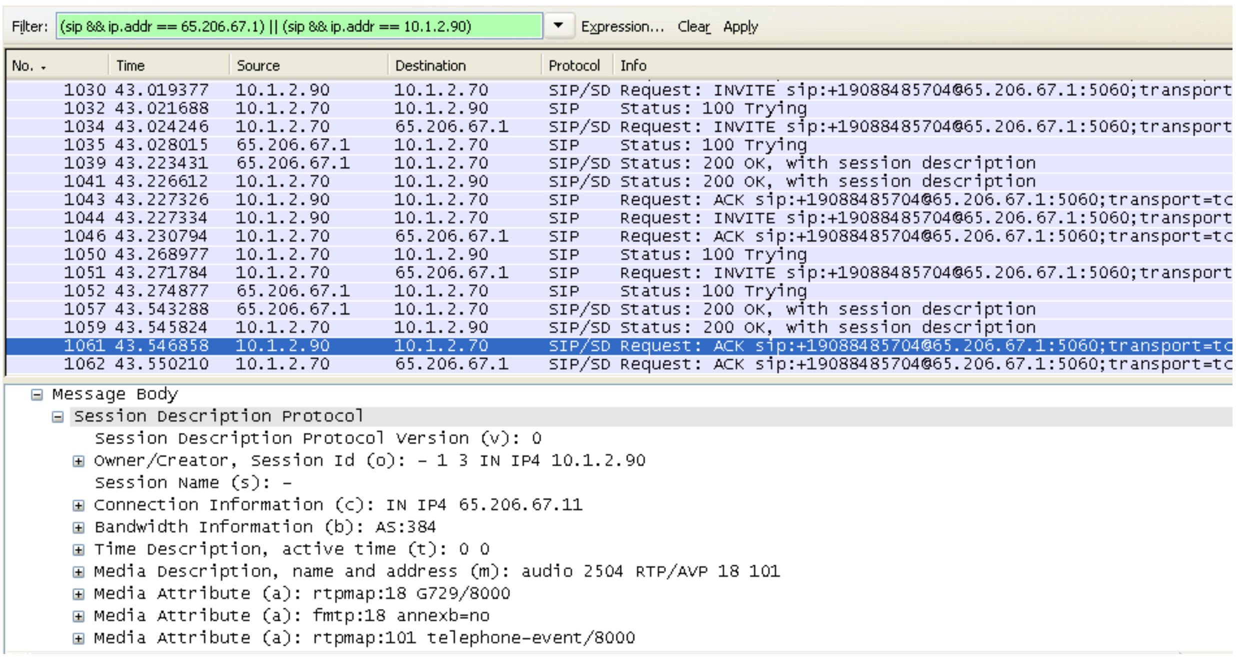
Scrolling down in this same trace, in frame 863, the SBC receives the 200 OK answering the transferred call, and in frame 864 the SBC sends the NOTIFY. Highlighted frame 865 shows the NOTIFY sent to Voice Portal contains “200 OK”. In frame 869, Voice Portal sends a BYE.



No.	Time	Source	Destination	Protocol	Info
863	35.351379	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
864	35.356061	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
865	35.358737	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
867	35.370300	65.206.67.87	10.1.2.70	SIP	Status: 200 OK
869	35.370313	65.206.67.87	10.1.2.70	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=tc

```
Message Body
  Sipfrag
    SIP/2.0 200 OK
```

Scrolling down in this same trace, when an agent on Communication Manager answers the call, Communication Manager "shuffles" the call away from the media gateway providing announcement and ringback services to "ip-direct", so that the final media path flows from the IP Telephone to the inside IP Address of the SBC. The ACK in frame 1061 is highlighted to reveal the SDP, which uses the IP Address (65.206.67.11) of the answering agent's telephone.



11.3.2. Verizon IP IVR Call Consult Transferred by Voice Portal

The traces for an IP IVR call for which Voice Portal uses consult transfer using REFER to a Communication Manager VDN that answers the call with an announcement step, and subsequently queues the call to a skill, is not substantially different from the trace analysis from the prior section, but is summarized here for completeness.

The following is an example trace taken from the outside interface. Frame 4 is the initial inbound IP IVR call from Verizon. This call is answered by Voice Portal in frame 7, and the caller interacts with the Voice Portal application. On the inside interface, Voice Portal initiates a consult transfer using REFER to VDN 36880, which answers the call with an announcement. After the announcement answers, the SBC sends the INVITE in frame 2668. Once the agent is selected and answers, Communication Manager "shuffles" the call to "ip-direct" on the inside interface, ultimately resulting in the "shuffle INVITE" that is highlighted in frame 3357.

Filter: sip

No.	Time	Source	Destination	Protocol	Info
4	9.382712	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668518119@adevc.avaya.globalipcom.com
5	9.385213	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
6	9.466724	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
7	9.554132	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
14	9.768748	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:8668518119@1.1.1.2:5060;transport=udp
2668	35.455015	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transp
2694	35.672671	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
2695	35.675585	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport
3326	41.877733	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transp
3348	42.089707	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3351	42.101195	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport
3357	42.153086	1.1.1.2	172.30.205.55	SIP	Request: INVITE sip:+19088485704@172.30.205.55:5072;transp
3379	42.369188	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3382	42.383406	1.1.1.2	172.30.205.55	SIP/SD	Request: ACK sip:+19088485704@172.30.205.55:5072;transport

Content-Length: 0
P-Asserted-Identity: "Joey Votto" <sip:8668518119@adevc.avaya.globalipcom.com>

The following portion of a filtered Wireshark trace taken on the inside interface shows the arrival of the call from Verizon through the answer of the call by Voice Portal. Since this is the same as prior illustrations, no elaboration is provided.

Filter: sip

No.	Time	Source	Destination	Protocol	Info
136	6.341167	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:8668518119@10.1.2.70:5060;transport=tcp
137	6.343156	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
140	6.386262	10.1.2.70	65.206.67.87	SIP/SD	Request: INVITE sip:44000@avaya.com:5060;transport=tcp, wit
142	6.394708	65.206.67.87	10.1.2.70	SIP	Status: 100 Trying
145	6.409406	65.206.67.87	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
147	6.411744	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
148	6.414718	65.206.67.87	10.1.2.70	SIP/SD	Status: 200 OK, with session description
154	6.500476	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
158	6.721804	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:8668518119@65.206.67.87;transport=tcp
159	6.724145	10.1.2.70	65.206.67.87	SIP	Request: ACK sip:8668518119@65.206.67.87;transport=tcp

Scrolling down in this same trace, highlighted frame 698 shows Voice Portal sending the REFER, since in this case, Voice Portal was configured to use REFER for Consult Transfer. Since Voice Portal is using REFER for Consult Transfer, this trace is not substantially different from the one previously illustrated for blind transfer, so fewer details are provided. In the center area, note that the Refer-To contains 36880, the VDN to which Voice Portal is transferring the call. In frame 701, the SBC sends the INVITE to 36880. In frame 708, Session Manager sends the INVITE to Communication Manager using the SIP Entity corresponding to the Verizon-specific behaviors on Communication Manager, since the originating entity is the SBC and a dial pattern exists to sends calls from the SBC to the VDN to Communication Manager using the Verizon-specific entity. In frame 714, Communication Manager sends 183 with SDP, evidence that the Verizon-specific trunk group has been used. In frame 719, Communication Manager answers the call, since the vector corresponding to this VDN begins with an announcement step.

Filter: (sip && ip.addr == 65.206.67.87) || (sip && ip.addr == 65.206.67.1) || (sip

No.	Time	Source	Destination	Protocol	Info
698	32.322832	65.206.67.87	10.1.2.70	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport
699	32.326549	10.1.2.70	65.206.67.1	SIP	Request: REFER sip:+19088485704@65.206.67.1:5060;transport
700	32.334478	65.206.67.1	10.1.2.70	SIP	Status: 202 Accepted
701	32.335860	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transp
702	32.335868	10.1.2.70	65.206.67.87	SIP	Status: 202 Accepted
705	32.337631	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
708	32.382111	10.1.2.70	10.1.2.90	SIP/SD	Request: INVITE sip:36880@avaya.com:5060;user=phone;transp
710	32.382999	10.1.2.90	10.1.2.70	SIP	Status: 100 Trying
714	32.384835	10.1.2.90	10.1.2.70	SIP/SD	Status: 183 Session Progress, with session description
719	32.385507	10.1.2.90	10.1.2.70	SIP/SD	Status: 200 OK, with session description

User-Agent: Avaya-VoicePortal/5.1.0.0.4206
Contact: <sip:8668518119@65.206.67.87;transport=tcp>
Refer-To: <sip:36880@avaya.com;user=phone?Expires=30>

Scrolling down in this same trace, in frame 726, the SBC receives the 200 OK answering the transferred call, and in frame 727 the SBC sends the NOTIFY. Highlighted frame 728 shows the NOTIFY sent to Voice Portal contains "200 OK". In frame 732, Voice Portal sends a BYE.

No.	Time	Source	Destination	Protocol	Info
722	32.388197	10.1.2.70	65.206.67.1	SIP/SD	Status: 183 Session Progress, with session description
726	32.400783	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
727	32.405085	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp,
728	32.407637	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:8668518119@65.206.67.87;transport=tcp,
730	32.408668	65.206.67.87	10.1.2.70	SIP	Status: 200 OK
732	32.408684	65.206.67.87	10.1.2.70	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
733	32.410374	10.1.2.70	65.206.67.1	SIP	Status: 200 OK
734	32.411712	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:18668512649@10.1.2.90:5062;transport=tcp
735	32.412352	10.1.2.70	65.206.67.1	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
737	32.415465	65.206.67.1	10.1.2.70	SIP	Status: 200 OK
739	32.416861	10.1.2.70	65.206.67.87	SIP	Status: 200 OK
742	32.418974	10.1.2.70	10.1.2.90	SIP	Request: ACK sip:18668512649@10.1.2.90:5062;transport=tcp

Message Body
Sipfrag
SIP/2.0 200 OK

Scrolling down in this same trace, when an agent on Communication Manager answers the call, Communication Manager "shuffles" the call away from the media gateway providing announcement and ring back services to "ip-direct", so that the final media path flows from the IP Telephone to the inside IP Address of the SBC. The ACK in frame 916 is highlighted to reveal the SDP, which uses the IP Address (65.206.67.11) of the answering agent's telephone. Note that G.711MU is used, since Verizon IP IVR only indicates support for G.711MU.

No.	Time	Source	Destination	Protocol	Info
897	39.047806	10.1.2.90	10.1.2.70	SIP	Request: INVITE sip:+19088485704@65.206.67.1:5060;transport
899	39.050042	10.1.2.70	65.206.67.1	SIP	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc
902	39.089733	10.1.2.70	10.1.2.90	SIP	Status: 100 Trying
904	39.100789	10.1.2.70	65.206.67.1	SIP	Request: INVITE sip:+19088485704@65.206.67.1:5060;transport
905	39.104022	65.206.67.1	10.1.2.70	SIP	Status: 100 Trying
912	39.324171	65.206.67.1	10.1.2.70	SIP/SD	Status: 200 Ok, with session description
914	39.325892	10.1.2.70	10.1.2.90	SIP/SD	Status: 200 Ok, with session description
916	39.327083	10.1.2.90	10.1.2.70	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc
918	39.329513	10.1.2.70	65.206.67.1	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tc

Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1 3 IN IP4 10.1.2.90
Session Name (s): -
Connection Information (c): IN IP4 65.206.67.11
Bandwidth Information (b): AS:384
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 2504 RTP/AVP 0 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:101 telephone-event/8000

11.4. Inbound Verizon IPCC Calls Consult Transferred by Voice Portal to a VDN on Communication Manager using INVITE with Replaces

The traces in this section illustrate Consultative Transfer using "INVITE with Replaces". Since intermittent problems were initially observed when using the "INVITE with Replaces" Consultative Transfer approach using the IP-IVR service, the configuration screen in **Section 4.4** shows the "REFER" method for Consult Transfer. Nevertheless, for completeness, this section presents an example using the INVITE with Replaces alternative for Verizon IP Toll Free, where no user visible problem was observed with this alternative. If IP IVR service will not be used, it may be preferable to configure Voice Portal to use the INVITE with Replaces Consultative Transfer approach, since this method allows Voice Portal to receive additional feedback from Communication Manager (e.g., 182 Queued) that may benefit Voice Portal application logic.

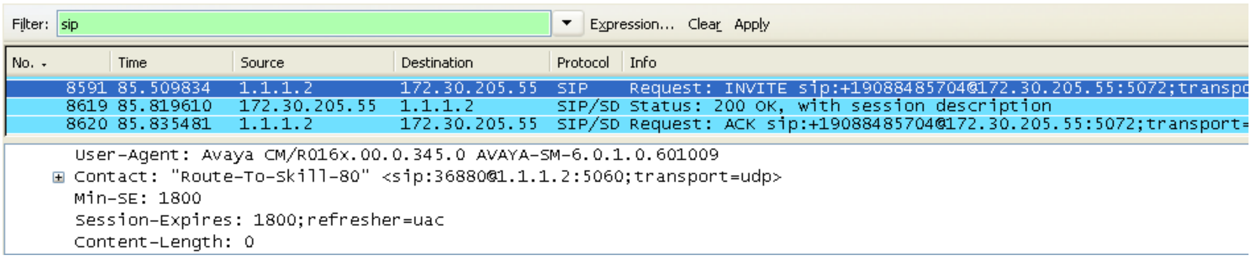
11.4.1. Verizon IP Toll Free Call Consult Transferred by Voice Portal using INVITE with Replaces

The following is an example trace taken from the outside interface of the SBC. Frame 2 is the initial inbound IP Toll Free call from Verizon. This call is answered by Voice Portal in frame 6, and the caller interacts with the Voice Portal application. On the inside interface, Voice Portal initiates a consult transfer using INVITE with Replaces to VDN 36880, which answers the call with an announcement. When the announcement answers, the SBC sends the INVITE in frame 3930. In this scenario, the Verizon network sees another INVITE in frame 3933. The INVITE in frame 3933 is the result of an INVITE sent from Communication Manager to the Acme SBC inside interface. In the sample trace below, Verizon IP Toll Free sends a 200 OK to the first INVITE (frame 3961) and a 491 Request Pending to the second INVITE (frame 3954). The back to back INVITE messages that Verizon will receive when Voice Portal uses Consultative Transfer with the INVITE with Replaces approach do not cause a user-visible problem with Verizon IP Toll Free.

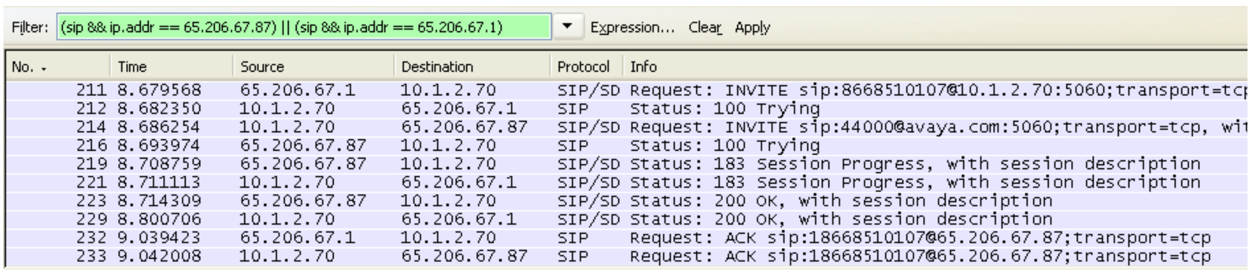
Filter: sip   Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
2	0.924282	172.30.205.55	1.1.1.2	SIP/SD	Request: INVITE sip:8668510107@adevc.avaya.globalipcom.com:
3	0.926790	1.1.1.2	172.30.205.55	SIP	Status: 100 Trying
5	0.969442	1.1.1.2	172.30.205.55	SIP/SD	Status: 183 Session Progress, with session description
6	1.057724	1.1.1.2	172.30.205.55	SIP/SD	Status: 200 OK, with session description
7	1.289315	172.30.205.55	1.1.1.2	SIP	Request: ACK sip:18668510107@1.1.1.2:5060;transport=udp
3930	39.868241	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3933	39.893748	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3954	40.061802	172.30.205.55	1.1.1.2	SIP	Status: 491 Request Pending
3955	40.062646	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
3961	40.108645	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
3963	40.111189	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=
3968	40.157822	1.1.1.2	172.30.205.55	SIP/SD	Request: INVITE sip:+19088485704@172.30.205.55:5072;transpo
3995	40.367906	172.30.205.55	1.1.1.2	SIP/SD	Status: 200 OK, with session description
4002	40.409827	1.1.1.2	172.30.205.55	SIP	Request: ACK sip:+19088485704@172.30.205.55:5072;transport=

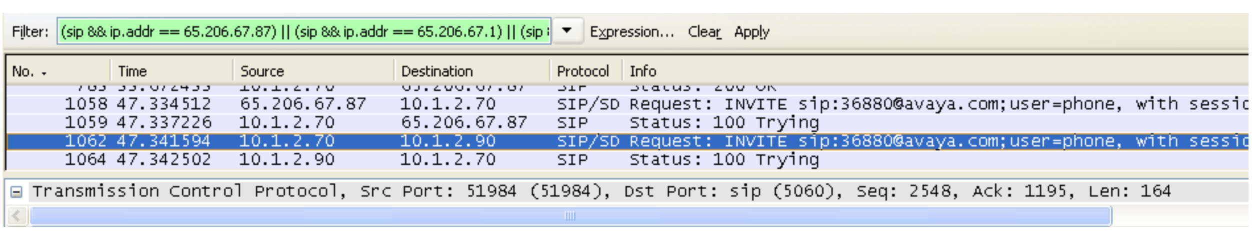
Scrolling down in this same trace, when the Communication Manager agent answers, Communication Manager sends a "shuffle INVITE" on the inside interface. This can be seen on the outside interface in highlighted frame 8591 below. The connection is stable with two-way talk path between the IP Toll Free caller and the Communication Manager answering agent.



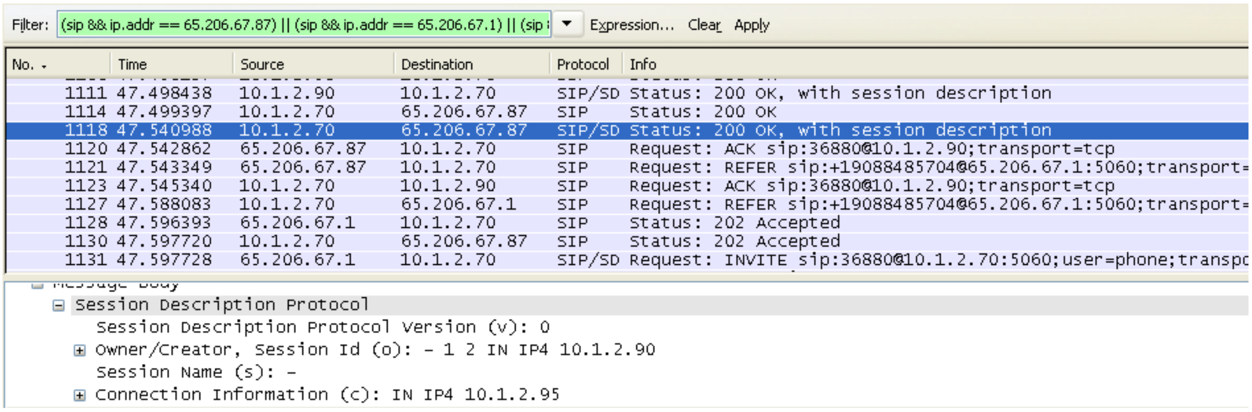
The following portion of a filtered Wireshark trace taken on the inside interface shows the arrival of the call from Verizon through the answer of the call by Voice Portal. Since this is the same as prior illustrations, no elaboration is provided.



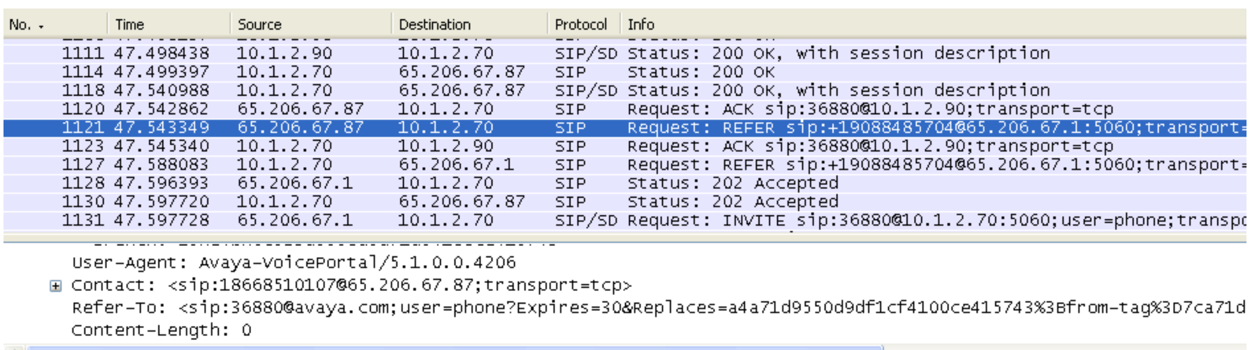
Scrolling down in this same trace, frame 1058 shows Voice Portal sending the INVITE containing the transferred-to VDN 36880, since in this case, Voice Portal was configured to use "INVITE with Replaces" for Consult Transfer. In highlighted frame 1062, Session Manager sends the INVITE to Communication Manager. This frame is highlighted to show that in this case, the transferred call does not initially use the "Verizon specific" SIP entity, since this leg of the call is an internal call from Voice Portal to Communication Manager (i.e., INVITE is not from the SBC). In the center area, it can be observed that the destination port on Communication Manager is 5060, associated with signaling group 60, used for internal enterprise communications.



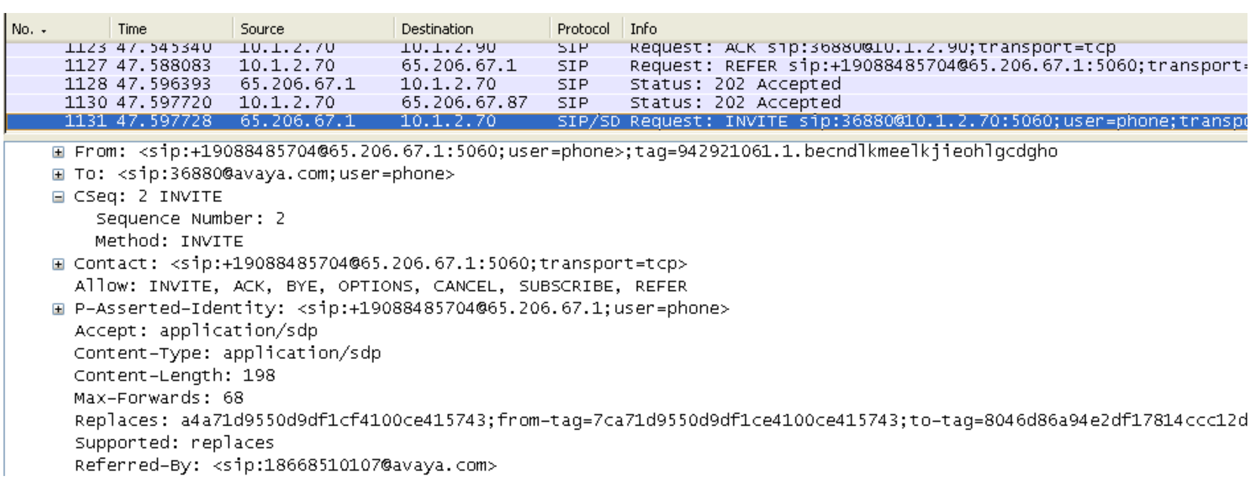
Scrolling down in this same trace, in frame 1111, Communication Manager answers the call from Voice Portal. In highlighted frame 1118, Voice Portal receives the 200 OK, and the center area shows that the Communication Manager gateway IP Address 10.1.2.95 in the SDP (e.g., an announcement).



Shown below is the same area of the trace, with a different frame highlighted. In highlighted frame 1121, Voice Portal sends a REFER to Session Manager for the original inbound call from Verizon. The center area shows that the Refer-To contains the VDN 36880 as well as the "Replaces" information. Session Manager sends this REFER to the SBC in frame 1127.



Scrolling down slightly, in highlighted frame 1131, the Acme Packet SBC sends an INVITE to the number extracted from the Refer-To header, and also includes a Replaces header in the INVITE. Note that the Replaces information in the INVITE matches the information from the Refer-To header shown above.



Scrolling down slightly, in highlighted frame 1136, note that Session Manager sends this INVITE to Communication Manager using the Verizon-specific trunk, since this INVITE came from the SBC. Although the original INVITE for the transferred leg of the call was processed on Communication Manager using trunk group 60, the final trunk used for the call will be the Verizon specific trunk 67 (i.e., note use of destination port 5062).

No.	Time	Source	Destination	Protocol	Info
1131	47.597728	65.206.67.1	10.1.2.70	SIP/SD	Request: INVITE sip:36880@10.1.2.70:5060;user=phone;transp
1133	47.599716	10.1.2.70	65.206.67.1	SIP	Status: 100 Trying
1136	47.604462	10.1.2.70	10.1.2.90	SIP/SD	Request: INVITE sip:36880@avaya.com:5060;user=phone;transp
1138	47.605352	10.1.2.90	10.1.2.70	SIP	Status: 100 Trying

⊟ Transmission Control Protocol, Src Port: 37567 (37567), Dst Port: 5062 (5062), Seq: 1462, Ack: 1, Len: 280

Scrolling down, Communication Manager sends a BYE in frame 1140, dropping the call from Voice Portal. Session Manager sends this BYE to Voice Portal in frame 1146. In frame 1143, Communication Manager sends the 200 OK answering the inbound call with the “INVITE with Replaces” from the SBC. In frame 1147, this 200 OK is received by the SBC, which then sends a NOTIFY message in frame 1150.

No.	Time	Source	Destination	Protocol	Info
1136	47.604462	10.1.2.70	10.1.2.90	SIP/SD	Request: INVITE sip:36880@avaya.com:5060;user=phone;transpo
1138	47.605352	10.1.2.90	10.1.2.70	SIP	Status: 100 Trying
1140	47.606156	10.1.2.90	10.1.2.70	SIP	Request: BYE sip:65.206.67.87;transport=tcp
1143	47.606615	10.1.2.90	10.1.2.70	SIP/SD	Status: 200 OK, with session description
1146	47.608860	10.1.2.70	65.206.67.87	SIP	Request: BYE sip:65.206.67.87;transport=tcp
1147	47.609462	10.1.2.70	65.206.67.1	SIP/SD	Status: 200 OK, with session description
1148	47.614838	65.206.67.87	10.1.2.70	SIP	Status: 200 OK
1150	47.615657	65.206.67.1	10.1.2.70	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
1151	47.616438	10.1.2.70	10.1.2.70	SIP	Status: 200 OK
1152	47.618241	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
1153	47.626850	65.206.67.87	10.1.2.70	SIP	Status: 200 OK

Scrolling down, Session Manager sends the NOTIFY (i.e., “200 OK” to the REFER) to Voice Portal in highlighted frame 1152. Voice Portal then sends a BYE in frame 1154, and Session Manager sends the BYE to the Acme SBC in frame 1156.

No.	Time	Source	Destination	Protocol	Info
1151	47.616438	10.1.2.70	10.1.2.90	SIP	Status: 200 OK
1152	47.618241	10.1.2.70	65.206.67.87	SIP/si	Request: NOTIFY sip:18668510107@65.206.67.87;transport=tcp,
1153	47.626850	65.206.67.87	10.1.2.70	SIP	Status: 200 OK
1154	47.626859	65.206.67.87	10.1.2.70	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
1155	47.628688	10.1.2.70	65.206.67.1	SIP	Status: 200 OK
1156	47.629593	10.1.2.70	65.206.67.1	SIP	Request: BYE sip:+19088485704@65.206.67.1:5060;transport=t
1157	47.630028	65.206.67.1	10.1.2.70	SIP	Request: ACK sip:36880@10.1.2.90:5062;transport=tcp
1158	47.632479	10.1.2.70	10.1.2.90	SIP	Request: ACK sip:36880@10.1.2.90:5062;transport=tcp

⊟ Message Body
⊟ Sipfrag
SIP/2.0 200 OK

Scrolling down, Communication Manager sends an INVITE in frame 1160, which ultimately results in a 491 Request Pending back from Verizon IP Toll Free on the outside interface, since the Acme Packet SBC had also recently sent an INVITE to Verizon IP Toll Free on the outside interface.

No.	Time	Source	Destination	Protocol	Info
1160	47.632978	10.1.2.90	10.1.2.70	SIP/SD	Request: INVITE sip:+19088485704@65.206.67.1:5060;transport=
1161	47.634233	10.1.2.70	65.206.67.87	SIP	Status: 200 OK
1162	47.636155	10.1.2.70	10.1.2.90	SIP	Status: 100 Trying
1164	47.637212	10.1.2.70	65.206.67.1	SIP/SD	Request: INVITE sip:+19088485704@65.206.67.1:5060;transport=
1165	47.640913	65.206.67.1	10.1.2.70	SIP	Status: 100 Trying
1175	47.812111	65.206.67.1	10.1.2.70	SIP	Status: 491 Request Pending
1177	47.813232	10.1.2.70	65.206.67.1	SIP	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tcp
1178	47.813491	10.1.2.70	10.1.2.90	SIP	Status: 491 Request Pending
1180	47.814113	10.1.2.90	10.1.2.70	SIP	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=tcp

Scrolling down in this same trace, when an agent on Communication Manager answers the call, Communication Manager "shuffles" the call away from the media gateway providing announcement and ringback services to "ip-direct", so that the final media path flows from the IP Telephone to the inside IP Address of the SBC. The ACK in frame 2655 is highlighted to reveal the SDP, which uses the IP Address (65.206.67.11) of the answering agent's telephone.

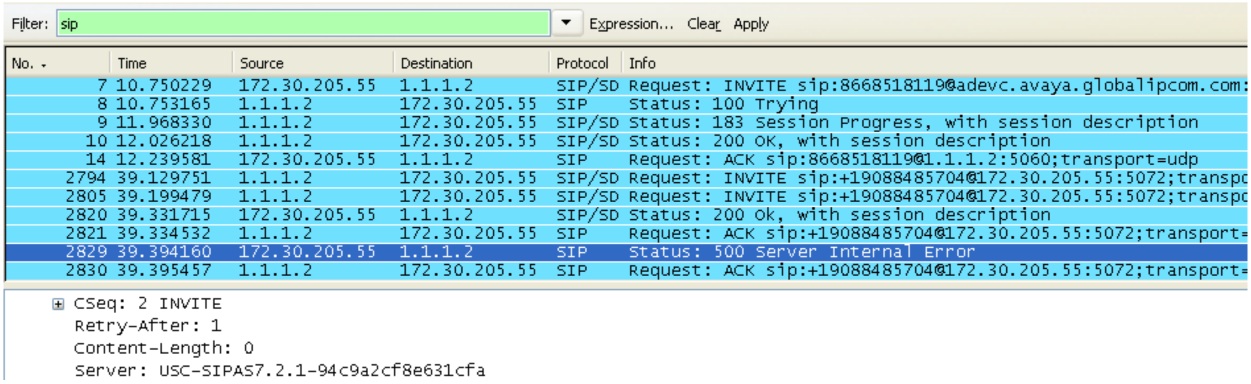
No.	Time	Source	Destination	Protocol	Info
2640	93.251335	10.1.2.90	10.1.2.70	SIP	Request: INVITE sip:+19088485704@65.206.67.1:5060;transpor
2642	93.253893	10.1.2.70	10.1.2.90	SIP	Status: 100 Trying
2643	93.255660	10.1.2.70	65.206.67.1	SIP	Request: INVITE sip:+19088485704@65.206.67.1:5060;transpor
2645	93.258817	65.206.67.1	10.1.2.70	SIP	Status: 100 Trying
2651	93.573642	65.206.67.1	10.1.2.70	SIP/SD	Status: 200 OK, with session description
2653	93.575489	10.1.2.70	10.1.2.90	SIP/SD	Status: 200 OK, with session description
2655	93.576862	10.1.2.90	10.1.2.70	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=t
2656	93.579525	10.1.2.70	65.206.67.1	SIP/SD	Request: ACK sip:+19088485704@65.206.67.1:5060;transport=t

```
⊟ Session Description Protocol
    Session Description Protocol Version (v): 0
  ⊞ Owner/Creator, Session Id (o): - 1 3 IN IP4 10.1.2.90
    Session Name (s): -
  ⊞ Connection Information (c): IN IP4 65.206.67.11
  ⊞ Bandwidth Information (b): AS:384
  ⊞ Time Description, active time (t): 0 0
  ⊞ Media Description, name and address (m): audio 2504 RTP/AVP 18 101
  ⊞ Media Attribute (a): rtpmap:18 G729/8000
  ⊞ Media Attribute (a): fmtp:18 annexb=no
  ⊞ Media Attribute (a): rtpmap:101 telephone-event/8000
```

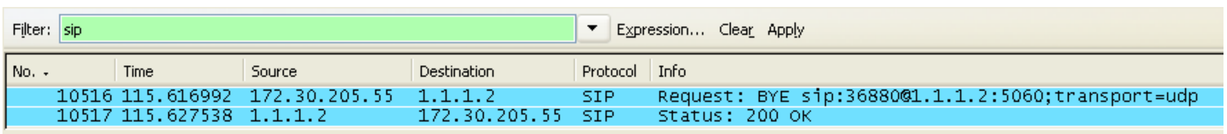
11.4.2. Verizon IP IVR Call Consult Transferred by Voice Portal using INVITE with Replaces

As noted in **Section 1.3**, if IP IVR service will be used, it is recommended to set the Voice Portal Consult Transfer option to REFER rather than INVITE with Replaces. If Voice Portal is configured for Consult Transfer using INVITE with Replaces, dropped calls may be experienced intermittently (i.e., the problem may or may not be observed).

The following trace is from the outside interface of the SBC. As also illustrated in the prior section with Verizon IP Toll Free, when INVITE with Replaces is used by Voice Portal for consult transfer, two consecutive INVITE messages can be seen by the Verizon service. Verizon IP Toll Free tolerates the consecutive INVITE messages, sending 491 messages that do not impact call stability. However, for some IP IVR calls using the production circuit used to verify these Application Notes, it has been observed that Verizon responds to the second INVITE with a 500 Server Internal Error, as shown in highlighted frame 2829 below.



Although the call can appear to be stable for a time, with the Communication Manager agent talking to the PSTN caller, once the 500 Error is observed, the call will ultimately be dropped by Verizon. Scrolling down in this same trace, frame 10516 shows the BYE.



This problem was seen early in the testing cycle, but could not be reproduced after Verizon network equipment was upgraded. To prevent the consecutive INVITEs that trigger this issue with IP IVR, Voice Portal consult transfer should use REFER, and Voice Portal should transfer calls to a VDN that answers the call (e.g., with a brief announcement) prior to queueing the call to a skill.

12. Conclusion

These Application Notes describe a sample configuration of Avaya Voice Portal with the Verizon Business IPCC service. The Verizon Business IPCC 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.

13. References

13.1. Avaya

Avaya product documentation, including the following, is available at http://support.avaya.com

[1] *Planning for Voice Portal*, June 2010
http://support.avaya.com/css/P8/documents/100089470
[2] *Implementing Voice Portal on a single server*, June 2010
http://support.avaya.com/css/P8/documents/100089465
[3] *Implementing Voice Portal on multiple servers*, June 2010
http://support.avaya.com/css/P8/documents/100089466
[4] *Administering Voice Portal*, June 2010
http://support.avaya.com/css/P8/documents/100089113
[5] *Installing and Configuring Avaya Aura™ Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010 available at http://support.avaya.com/css/P8/documents/100089133
[6] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, Issue 6.0 June 2010 available at http://support.avaya.com/css/P8/documents/100089333
[7] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100082630
[8] *Installing and Configuring Avaya Aura™ Session Manager,* Doc ID 03-603473 Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089152
[9] *Maintaining and Troubleshooting Avaya Aura™ Session Manager,* Doc ID 03-603325, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089154
[10] *Administering Avaya Aura™ System Manager*, Document Number 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089681

Avaya Application Notes, including the following, are also available at http://support.avaya.com

Application Notes Reference [JRR-VZIPCC] documents Verizon IPCC Services with Communication Manager Release 6 and Session Manager Release 6. The configuration documented in [JRR-VZIPCC] served as the starting point for the configuration in these

Application Notes. That is, Avaya Voice Portal was added to the configuration documented in [JRR-VZIPCC].
[JRR-VZIPCC] Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Acme Packet Net-Net with Verizon Business IP Contact Centers Services Suite – Issue 1.0
https://devconnect.avaya.com/public/download/dyn/SM6CM6_VzBIPCC.pdf

Application Notes Reference [JF-VZIPCC] documents Verizon IPCC Services with previous versions of Communication Manager and Session Manager.
[JF-VZIPCC] Application Notes for Avaya Aura™ Communication Manager 5.2, Avaya Aura™ Session Manager 1.1, and Acme Packet 3800 Net-Net Session Director with Verizon Business IP Contact Centers Services Suite – Issue 1.2
https://devconnect.avaya.com/public/download/dyn/AvayaSM_VzBIPCC.pdf

Application Notes Reference [JRR-VZIPT] documents Verizon IP Trunk Service with a similar configuration of the enterprise equipment shown in these Application Notes.
[JRR-VZIPT] Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0, and Acme Packet Net-Net SBC with Verizon Business IP Trunk SIP Trunk Service – Issue 1.0
https://devconnect.avaya.com/public/download/dyn/SM6Acme_VzB_IPT.pdf

Application Notes Reference [ICR] documents an Intelligent Customer Routing configuration using Voice Portal.
[ICR] Intelligent Customer Routing with Acme Packet Session Border Controller using Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0 and Avaya Voice Portal 5.1 – Issue 1.0
http://support.avaya.com/css/P8/documents/100113293

13.2. Acme Packet

Acme Packet Support (login required):
http://www.acmepacket.com/support.htm

13.3. Verizon Business

Information in the following documents was also used for these Application Notes:

[1] *Verizon Business IPCC Interoperability Test Plan, Revision 1.7, Aug 27, 2009*
[2] *Verizon Business IP Contact Center Trunk Interface Network Interface Specification, Document Version 2.2.1.9, Aug 25, 2009*
[3] *Additional information regarding Verizon Business IPCC Services suite offer can be found at http://www.verizonbusiness.com/products/contactcenter/ip/*

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.