Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.1, and Acme Packet Net-Net with Verizon Business IP Contact Center (IPCC) Services Suite – Issue 1.0

Abstract

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.1, and Acme Packet Net-Net Session Border Controller integration with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the VoIP Inbound, IP Contact Center, and IP-IVR SIP trunk service offers. This service suite provides toll free inbound calling via standards-based SIP trunks as well as re-routing of inbound toll free calls to alternate destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. The Communication Manager Network Call Redirection (NCR) and SIP User-to-User Information (UUI) features can be utilized together to transmit UUI within SIP signaling messages to alternate destinations via the Verizon network. These Application Notes supplement previously published Application Notes documenting integration with Verizon IPCC using different versions of Communication Manager and Session Manager.

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

Avaya Aura® SIP Solution using Avaya Aura® Communication Manager 5.2.1 has not been certified independently by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

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1. Introduction

These Application Notes describe a sample configuration of Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.1, and Acme Packet Net-Net Session Border Controller integration with Verizon Business IP Contact Center (IPCC) Services suite. The Verizon Business IPCC Services suite is comprised of the VoIP Inbound, IP Contact Center, and IP-IVR SIP trunk service offers. These Application Notes supplement previously published Application Notes [JF-VZIPCC and JRR-VZIPCC] documenting integration with Verizon IPCC using different versions of Communication Manager and Session Manager.

In the sample configuration, an Acme Packet 4250 Net-Net Session Border Controller 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 Avaya Aura® Communication Manager, the Acme Packet Net-Net Session Border Controller (SBC), and other applications such as Avaya Modular Messaging. Avaya Aura® Communication Manager SIP trunks and Acme Packet SBC "session-agents" are provisioned to terminate at Avaya Aura® Session Manager.

The Verizon Business IPCC Services suite described in these Application Notes is designed for business customers using Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The service provides inbound toll-free service via standards-based SIP trunks. Using SIP Network Call Redirection (NCR), trunk-to-trunk connections of certain inbound calls to Avaya Aura® Communication Manager can be avoided by requesting that the Verizon network transfer the inbound caller to an alternate destination. In addition, the Avaya Aura® Communication Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI data within SIP signaling messages to alternate destinations. This capability allows the service to transmit a limited amount of call-related data between call centers to enhance customer service and increase call center efficiency. Examples of UUI data might include a customer account number obtained during a database query or the best service routing data exchanged between sites using Avaya Aura® Communication Manager.

Verizon Business IPCC Services suite is a portfolio of IP Contact Center (IPCC) interaction services that includes VoIP Inbound and IP Interactive Voice Response (IP IVR). Access to these features may use Internet Dedicated Access (IDA) or Private IP (PIP). PIP was used for the sample configuration described in these Application Notes. 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.

Avaya Aura® SIP Solution using Avaya Communication 5.2.1 has not been certified independently by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows to Session Manager and Communication Manager, and subsequent redirection of inbound calls to Verizon for re-routing to alternate destinations. See **Section 2.2** for an overview of key call flows and **Section 9** for detailed verifications of key call flows. Additional test objectives are listed in **Section 8**.

1.2. Support

1.2.1 Avaya

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.

1.2.2 Verizon

For technical support, visit online support at http://www.verizonbusiness.com/us/customer/

1.3. Known Limitations

The following limitations are noted for the sample configuration described in these Application Notes:

- Verizon Business IPCC Services suite does not support fax.
- Verizon Business IPCC Services suite does not support History Info or Diversion Headers.
- Verizon Business IPCC Services suite does not support G.729B codec, and IP-IVR of the service suite supports G.711Mu only.
- Although the Verizon IPCC Services suite defines call flows that would allow a call to remain in Communication Manager vector processing upon failure of a vector-triggered REFER attempt, such call scenarios could not be verified on the production Verizon circuit used for testing. See **Section 2.2.3** for additional information.
- When Vector Directory Numbers (VDN's) are used on Communication Manager to trigger vector execution for SIP NCR and REFER, Verizon accepts the REFER from the enterprise site, then sends an INVITE message to the enterprise, indicating a network hold state with connection address "0.0.0.0". It happens sometimes that Communication Manager will also send an INVITE message to the Verizon network. Verizon will respond to this INVITE message with a "491 Request Pending" response, which will trigger another INVITE message from Communication Manager to the Verizon network. A series of INVITE/491 message exchanges will continue for several seconds in this fashion. These messages do not impact the completion of the call to the refer-to destination. The workaround documented in **Section 4.11** of [JRR-VZIPCC] for avoiding these extra messages does not work all the time with Communication Manager Release 5.2.1.
- With Communication Manager Release 5.2.1, an additional Communication Manager configured as a Feature Server is required to support enterprise SIP phones and their inter-working with endpoints of other types (i.e., H.323, digital and/or analog) controlled by the Communication Manager Access Element (this limitation was lifted in Communication Manager Release 6.0 and later releases). Since Communication Manager configured as an

Access Element plus a separate Communication Manager Feature Server in the Release 5.2.1 environment is not a typical deployment configuration, Communication Manager Feature Server (and therefore enterprise SIP phones) was not included in the sample configuration described in these Application Notes.

- Communications Manager Release 5.2.1 responds to INVITE requests for inbound calls with "180 Ringing" with SDP though Verizon Test Plan states that "IPCC customers cannot send 180 with SDP". No negative impact to call processing was observed. Communication Manager Release 6.0 and later releases support "183 Session Progress" messages with SDP as preferred by Verizon.

2. Reference Configuration

Figure 1 illustrates the sample configuration used for the DevConnect compliance testing. The configuration is comprised of the Avaya CPE location connected via a T1 Internet connection to the Verizon Business IPCC service node. The Avaya CPE location simulates a customer site. At the edge of the Avaya CPE location, an Acme Packet Net-Net SBC provides NAT functionality and SIP header manipulation. The Acme Packet SBC receives traffic from the Verizon Business IPCC Services on UDP port 5060 and sends traffic to the Verizon Business IPCC Services using destination UDP port 5072. The PIP service defines a secure MPLS connection between the Avaya CPE T1 connection and the Verizon IPCC service node.

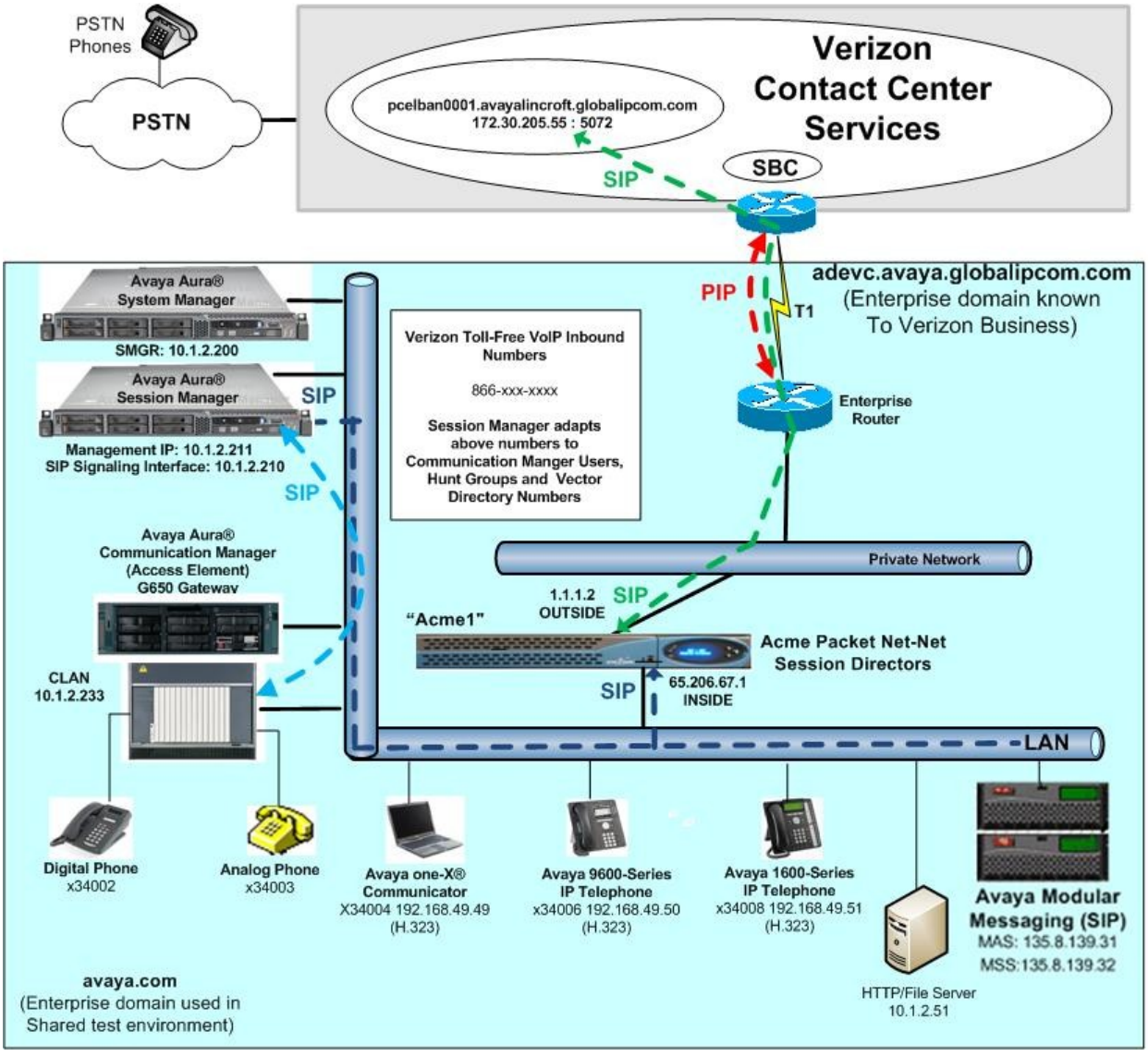


Figure 1: Avaya Solution & Interoperability Test Lab Configuration

The Verizon provided toll-free numbers were mapped by Avaya Aura® Session Manager or Avaya Aura® Communication Manager to various Communication Manager extensions. The extension mappings were varied during the testing to allow inbound toll-free calls to terminate directly on user extensions or indirectly, through hunt groups, vector directory numbers (VDNs) and vectors, to user extensions and contact center agents.

The Avaya CPE environment was known to Verizon Business IP Trunk Service as FQDN *adevc.avaya.globalipcom.com*. For efficiency, the Avaya CPE environment utilizing Session Manager Release 6.1 and Communication Manager Release 5.2.1 was shared among many ongoing test efforts at the Avaya Solution & Interoperability Test Lab. Access to the Verizon Business IPCC services was added to a configuration that already used domain "avaya.com" at the enterprise. As such, Session Manager or the SBC were used to adapt the domains as needed. These Application Notes indicate the configuration that would not be required in cases where the CPE domain in Communication Manager and Session Manager match the CPE domain known to Verizon.

The following summarizes Request URI contents as well as transport protocol and port for toll-free calls in the sample configuration:

- Verizon Business IPCC Services node sends the following to the SBC:
  - The CPE FQDN of ***adevc.avaya.globalipcom.com*** in the Request URI.
  - Sends the packet to Avaya CPE using destination port 5060 via UDP
- Acme Packet Net-Net SBC sends the following to Session Manager:
  - The Request URI contains ***10.1.2.210***, the IP Address of the SIP signaling interface of Avaya Aura® Session Manager
  - Sends the packet to Session Manager using destination port 5060 via TCP
- Avaya Aura® Session Manager sends the following to Communication Manager
  - The Request URI contains ***avaya.com***, to match the shared Avaya SIL test environment.
  - Sends the packet to Communication Manager using destination port 5067 via TCP to allow Communication Manager to distinguish Verizon traffic from other traffic arriving from the same instance of Session Manager

Note – The Fully Qualified Domain Names and IP addressing specified in these Application Notes apply only to the reference configuration shown in **Figure 1**. Verizon Business customers will use different FQDNs and IP addressing as required.

2.1. History Info and Diversion Headers

The Verizon Business IPCC Services suite does not support SIP History Info Headers or Diversion Headers. Therefore, Communication Manager was provisioned not to send History Info Headers or Diversion Headers.

2.2. Call Flows

To understand how inbound Verizon toll-free calls are handled by Session Manager and Communication Manager, key call flows are summarized in this section.

2.2.1 Inbound IP Toll Free Call with no Network Call Redirection

The first call scenario illustrated in **Figure 2** is an inbound Verizon IP Toll Free call that is routed to Communication Manager, which in turn routes the call to a vector, agent, or phone. No redirection is performed in this simple scenario. A detailed verification of such a call with Communication Manager and Wireshark traces can be found in **Section 9.1.1**.

1. A PSTN phone originates a call to a Verizon IP Toll Free number.
2. The PSTN routes the call to the Verizon IP Toll Free service network.
3. The Verizon IP Toll Free service routes the call to the Acme Packet SBC.
4. The Acme Packet SBC performs SIP Network Address Translation (NAT) and any necessary SIP header manipulations, and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to a) a hunt group or vector, which in turn routes the call to an agent or phone, or b) directly to a phone.

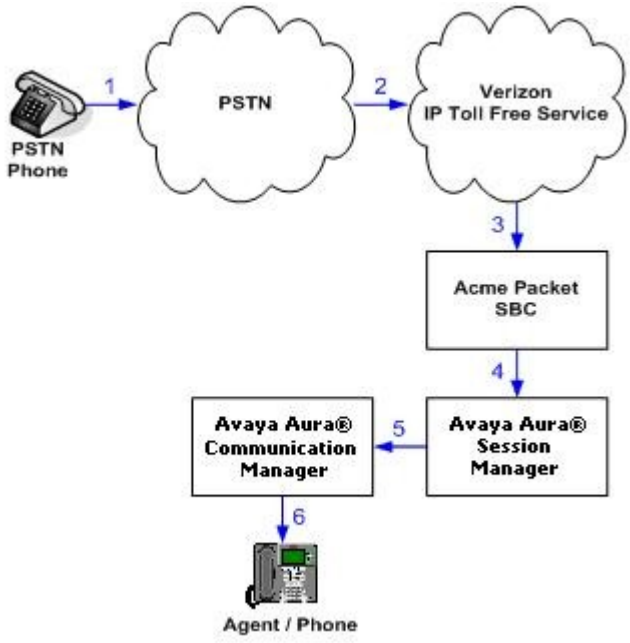


Figure 2: Inbound Verizon IP Toll Free Call – No Redirection

2.2.2 Inbound IP Toll Free Call with Post-Answer Network Call Redirection

The second call scenario illustrated in **Figure 3** is an inbound Verizon IP Toll Free call that is routed to a Communication Manager Vector Directory Number (VDN) to invoke call handling logic in a vector. The vector answers the call and then redirects the call back to the Verizon IP Toll Free service for routing to an alternate destination. Note that Verizon IP Toll Free service does not support redirecting a call before it is answered (using a SIP 302), and therefore the vector must include a step that results in answering the call, such as playing an announcement.

A detailed verification of such call with both Communication Manager and Wireshark traces can be found in **Section 9.1.2** for a PSTN destination and **Section 9.1.3** for a Verizon IP Toll Free SIP-connected alternate destination. In the latter case, the Verizon IP Toll Free service can be used to pass User to User Information (UUI) from the redirecting site to the alternate destination.

1. Same as the first five steps in **Figure 2**.
2. Communication Manager routes the call to a vector, which answers the call, plays an announcement, and attempts to redirect the call by sending a SIP REFER message out the SIP trunk upon which the inbound call arrived. The SIP REFER message specifies the alternate destination in the Refer-To header. The SIP REFER message passes back through Session Manager and the Acme Packet SBC to the Verizon IP Toll Free service network.
3. The Verizon IP Toll Free service places a call to the target party contained in the Refer-To header. Upon answer, the calling party is connected to the target party.
4. The Verizon IP Toll Free service clears the call on the redirecting/referring party (Communication Manager).

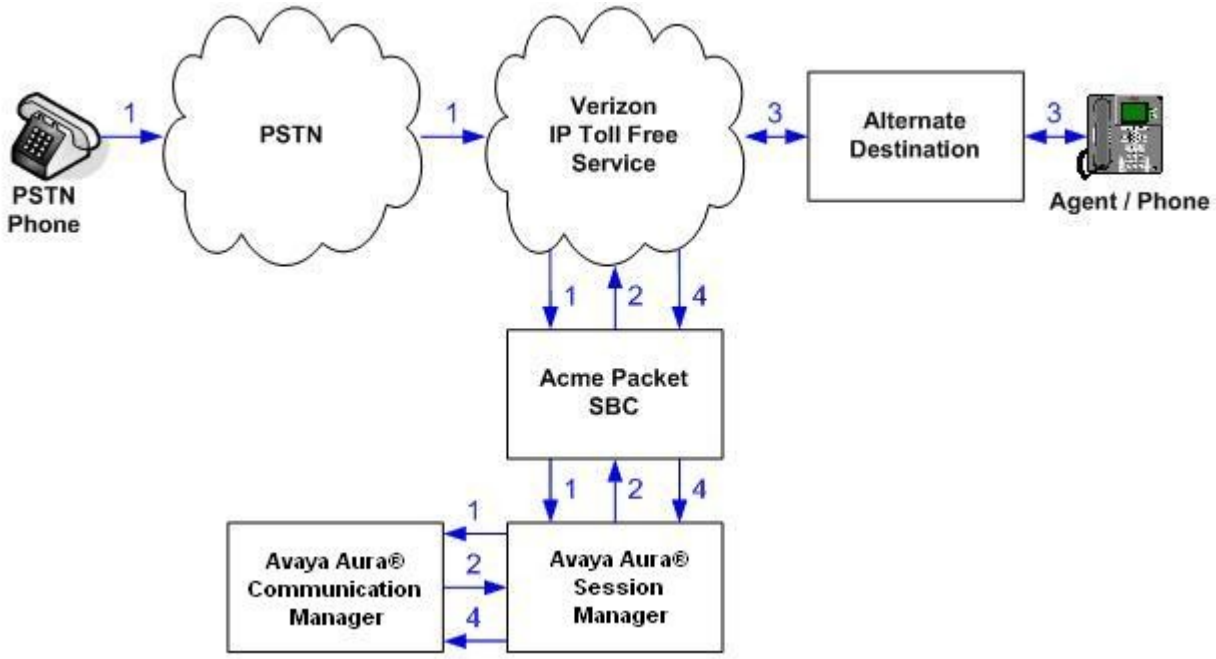


Figure 3: Inbound Verizon IP Toll Free Call – Post-Answer SIP REFER Redirection Successful

2.2.3 Inbound IP Toll Free Call with Unsuccessful Network Call Redirection

The next call scenario illustrated in

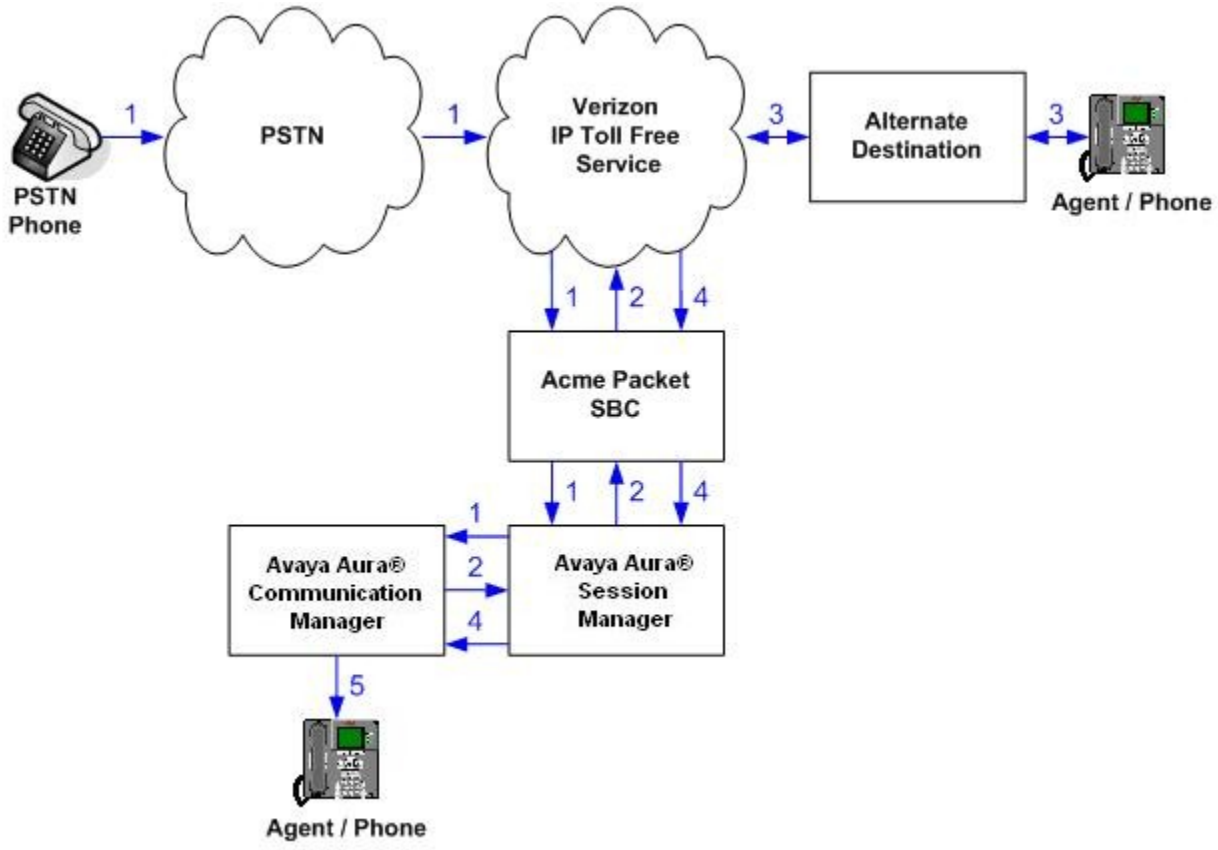


Figure 4 is similar to the previous call scenario, except that the redirection is unsuccessful due to the alternate destination being busy or otherwise unavailable. As a result, Communication Manager "takes the call back" and continues vector processing. For example, the call may route to an agent, phone, or announcement after unsuccessful NCR.

1. Same as **Figure 2**.
2. Same as **Figure 2**.
3. The Verizon IP Toll Free service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The Verizon IP Toll Free service notifies the redirecting/referring party (Communication Manager) of the error condition.
5. Communication Manager routes the call to a local agent, phone, or announcement.

Note: As noted in **Section 1.3**, except for egregious configuration errors, this "error handling" scenario could not be verified on the production Verizon circuit used for testing. On the production circuit, Verizon sends a SIP BYE which terminates Communication Manager vector processing for the call when the alternate destination is busy or otherwise unavailable. In cases where misconfiguration is introduced such that the Refer-To header is malformed or the REFER times out, Communication Manager can continue vector processing

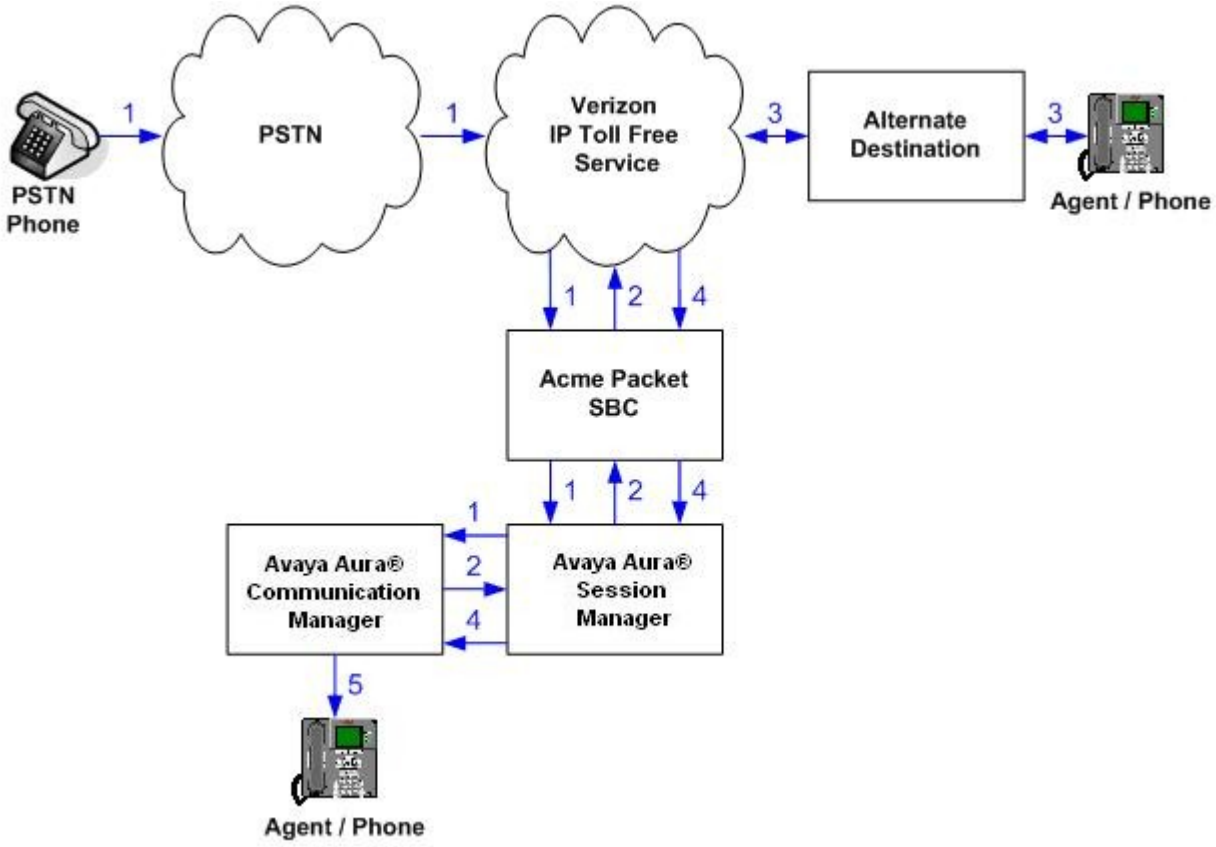


Figure 4: Inbound Verizon IP Toll Free Call – Post-Answer SIP REFER Redirection Unsuccessful

3. Equipment and Software Validated

The following equipment and software were used in the sample configuration.

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4 + patch 18576)
Avaya G650 Media Gateway – CONTROL-LAN (CLAN) – IP MEDIA PROCESSOR – IP SERVER INTFC (IPSI)	 TN799DP – HW01 FW032 TN2602AP – HW02 FW047 TN2312BP – HW15 FW046
Avaya S8800 Server	Avaya Aura® System Manager 6.1 Build 6.1.0.0.7345, Patch 6.1.5.2
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 6.1.0.0.610023
Avaya 9600-Series Telephone (H.323)	Avaya one-X® Deskphone Edition 3.1.1
Avaya 1600-Series Telephone (H.323)	Avaya one-X® Deskphone Value Edition 1.2.2
Avaya one-X® Communicator (H.323)	6.0 with SP1 (6.0.1.16)
Avaya 6408-D Digital Telephone	N/A
Avaya 6210 Analog Telephone	N/A
Avaya Modular Messaging (Application Server)	Avaya Modular Messaging (MAS) 5.2 SP6 Patch 2 (9.2.357.6022)
Avaya Modular Messaging (Storage Server)	Avaya Modular Messaging (MSS) 5.2 SP6 Patch 2
Acme Packet Net-Net 4250[1]	SC6.2.0 MR-3 Patch 5 (Build 687)

Table 1: Equipment and Software Used in the Sample Configuration

4. Configure Avaya Aura® Communication Manager Release 5.2.1

This section illustrates an example configuration allowing SIP signaling to Avaya Aura® Session Manager via an Avaya C-LAN in the Avaya G650 Media Gateway.

Note - The initial installation, configuration, and licensing of the Avaya servers and media gateways for Avaya Aura® Communication Manager are assumed to have been previously completed and are not discussed in these Application Notes.

All Communication Manager configuration is performed via the Communication Manager SAT interface of the Avaya S8800 Server. Screens are abridged for brevity in presentation.

[1] Although an Acme Net-Net 4250 was used in the sample configuration, the 3800, 4500, and 9200 platforms are also supported.

4.1. Verify Licensed Features

The Communication Manager license file controls customer options on the system. 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 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.

```
display system-parameters customer-options                    Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 800    200
           Maximum Concurrently Registered IP Stations: 18000  3
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                 Maximum Video Capable H.323 Stations: 0      0
                  Maximum Video Capable IP Softphones: 0      0
                     Maximum Administered SIP Trunks: 800    198
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                           Maximum TN2501 VAL Boards: 10     1
                    Maximum Media Gateway VAL Sources: 0      0
           Maximum TN2602 Boards with 80 VoIP Channels: 128    0
          Maximum TN2602 Boards with 320 VoIP Channels: 128    2
   Maximum Number of Expanded Meet-me Conference Ports: 0      0
```

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

```
display system-parameters customer-options                    Page   3 of  10
                                OPTIONAL FEATURES

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

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 will be required for the call flows as described in **Section 2.2**, verify that the **ISDN/SIP Network Call Redirection** feature is enabled.

```
display system-parameters customer-options                    Page   4 of  10
                              OPTIONAL FEATURES

   Emergency Access to Attendant? y                          IP Stations? y
            Enable 'dadmin' Login? y
            Enhanced Conferencing? y                    ISDN Feature Plus? y
                   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? n             Local Survivable Processor? n
          Extended Cvg/Fwd Admin? n                  Malicious Call Trace? n
     External Device Alarm Admin? n               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? n               Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? n
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n             Multimedia IP SIP Trunking? n
                       IP Trunks? y

            IP Attendant Consoles? n
```

On **Page 5** of the **System-Parameters Customer-Options** form, verify that the **Private Networking** is enabled.

```
display system-parameters customer-options                    Page   5 of  10
                              OPTIONAL FEATURES

                 Multinational Locations? n           Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? n
                      Multiple Locations? n
                                              System Management Data Transfer? n
          Personal Station Access (PSA)? n             Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? n
                  Port Network Support? y             Time of Day Routing? n
                       Posted Messages? n      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? n
                                                                  Wireless? n
                           Remote Office? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? 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 is used to refer calls to alternate destinations using SIP NCR. Vector variables are used to include User-User Information (UUI) with the referred calls.

```
display system-parameters customer-options                    Page   6 of  10
                        CALL CENTER OPTIONAL FEATURES

                         Call Center Release: 5.0

                                  ACD? y                        Reason Codes? y
                        BCMS (Basic)? y             Service Level Maximizer? y
          BCMS/VuStats Service Level? y           Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? y    Service Observing (Remote/By FAC)? y
                  Business Advocate? n            Service Observing (VDNs)? y
                    Call Work Codes? y                            Timed ACW? y
       DTMF Feedback Signals For VRU? y                  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? y    Vectoring (G3V4 Advanced Routing)? y
           Lookahead Interflow (LAI)? y                  Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y     Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? y                Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? y                Vectoring (Variables)? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

4.2. Dial Plan

In the sample configuration, the Avaya CPE environment uses five digit local extensions that start with 3, such as 3xxxx. Trunk Access Codes are of call type "dac" and are 3 digits in length beginning 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.

```
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
    1           3    dac
    2           5    ext
    3           5    ext
    4           5    ext
    5           5    ext
    6           5    ext
    7           5    ext
    8           1    fac
    9           1    fac
    *           3    fac
    #           3    fac
```

4.3. 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 "sm61" with IP address 10.1.2.210, and the node name for the CLAN in the Avaya G650 Media Gateway controlled by Communication Manager is "clan1" with IP address 10.1.2.233.

```
change node-names ip                                          Page   1 of  2
                              IP NODE NAMES
    Name              IP Address
sm61                 10.1.2.210
clan1                10.1.2.233
```

4.4. IP Interface

The ***add ip-interface*** or ***change ip-interface*** command can be used to configure the CLAN parameters. In the sample configuration, the CLAN board used is located in Cabinet 1 Carrier A Slot 2 of the G650 Media Gateway. The following screen shows the parameters used in the sample configuration. While the focus here is the use of the CLAN for SIP Trunk Signaling, observe that the CLAN will also be used for registrations from H.323 IP Telephones and H.248 gateways in the sample configuration.

```
change ip-interface 01a02                                     Page   1 of   3
                                  IP INTERFACES


                 Type: C-LAN
                 Slot: 01A02         Target socket load and Warning level: 400
          Code/Suffix: TN799  D            Receive Buffer TCP Window Size: 8320
      Enable Interface? y                          Allow H.323 Endpoints? y
                 VLAN: n                           Allow H.248 Gateways? y
       Network Region: 1                             Gatekeeper Priority: 5


                                IPV4 PARAMETERS
            Node Name: clan1
          Subnet Mask: /24
     Gateway Node Name: Gateway001

         Ethernet Link: 1
         Network uses 1's for Broadcast Addresses? y
```

4.5. Network Regions

Network regions provide a means to logically group resources. In the shared Communication Manager configuration used for the testing, the Avaya G650 Media Gateway is in region 1. To provide testing flexibility, network region 54 was associated with the Acme Packet Net-Net SBC and the CPE IP phones 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. IP telephones can be assigned a network region based on an IP address mapping. The network region can also associate the IP telephone to a

location for location-based routing decisions. 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.

The screen below shows that devices with IP addresses in the 10.1.2.0/24 subnet are assigned to network region 1. These include Communication Manager and Session Manager that were set up for shared test environment. Devices with IP addresses in the 65.206.67.0/24 subnet are assigned to network region 54. In the sample configuration, the Acme Packet Net-Net SBC with inside IP address 65.206.67.1 is therefore placed in network region 54. IP telephones used for the compliance test are all assigned to network region 54 with IP addresses in the 192.168.49.0/24 subnet. 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: 10.1.2.1                                  /24    1      n
   TO: 10.1.2.255
 FROM: 65.206.67.0                               /24    54     n
   TO: 65.206.67.255
 FROM: 192.168.49.0                              /24    54     n
   TO: 192.168.49.255
```

The following screen shows the **ip-network-region-54** configuration. In the shared test environment, network region 54 is used to allow unique behaviors for the Verizon test. In this example, codec set 4 will be used for calls within region 54. The shared Avaya Solution & Interoperability Test Lab environment uses the domain "avaya.com". 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 far-end of the Avaya signaling group (as will be shown in **Section 4.7**). Session Manager will be configured to adapt "avaya.com" to "adevc.avaya.globalipcom.com" in the PAI header.

```
change ip-network-region 54                                    Page   1 of  19
                            IP NETWORK REGION
  Region: 54
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? n
```

```
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The following screen shows the inter-network region connection configuration for network region 54. The bold row shows that network region 54 is directly connected to network region 1, and that codec set 4 will 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 1will also show codec set 4 for region 1 to region 54 connectivity.

```
change ip-network-region 54                                    Page   3 of  19

 Source Region: 54    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
```

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** setting. 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 be configured to adapt the domain from "avaya.com" to "adevc.avaya.globalipcom.com".

```
change ip-network-region 1                                    Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 10001
DIFFSERV/TOS PARAMETERS                RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The following screen shows the inter-network region connection configuration for network region 1. The bold row shows that network region 1 is directly connected to network region 54, and that codec set 4 will be used for any connections between region 54 and region 1.

```
change ip-network-region 1                                    Page   6 of  19

 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
 46
 47
 48
 49
 50
 51
 52
 53
 54   4     y    NoLimit                                          n       t
 55
```

4.6. IP Codec Sets

The following screen shows the configuration for **ip-codec-set 4**, the codec set configured to be used for calls within network region 54 and for calls between region 1 and region 54. In general, an IP codec set is a list of allowable codecs in priority order. Using the example configuration shown below, all calls via the SIP trunks configured for Verizon testing would use G.729A, since G.729A is preferred by both Verizon and the Avaya ip-codec-set. Any calls using this same codec set that are placed between devices capable of the G.722-64K codec (e.g., Avaya 9600-Series IP Telephone) can use G.722.

Note that if G.711MU is omitted from the list of allowed codecs in **ip-codec-set 4**, calls from Verizon that are answered by Avaya Modular Messaging will use VoIP resources on the Avaya G650 Media Gateway to convert from G.729A (facing Verizon) to G.711MU (facing Modular Messaging). If G.711MU is included in ip-codec-set 4, then calls from Verizon that are answered by Modular Messaging will not use G650 VoIP resources, but rather be "ip-direct" using G.711MU from Modular Messaging to the inside of the Acme Packet Net-Net SBC.

Also note that the IP-IVR service of the Verizon Business IPCC Services suite supports G.711MU only, therefore G.711MU should be specified in the ip-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:
```

On **Page 2** of the form:

- Configure the Fax **Mode** field to "off" since 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 used for Avaya Modular Messaging and other connections within network region 1.

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

                         IP Codec Set

   Codec Set: 1

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

4.7. 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 "clan1", and a **Far-end Node Name** of "sm61". 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 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 used for processing incoming PSTN calls from Verizon via Session Manager. The **Far-end Network Region** is configured to region 54. Port 5067 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 TCP port 5067. The use of different ports is one means to allow Communication Manager to distinguish different types of calls arriving from the same Session Manager, and vice-versa. Other parameters may be left at default values.

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

 Group Number: 67              Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n


   Near-end Node Name: clan1                 Far-end Node Name: sm61
 Near-end Listen Port: 5067               Far-end Listen Port: 5067
                                       Far-end Network Region: 54
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                 Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

The following screen shows signaling group 32, the signaling group to Session Manager that was in place prior to adding the Verizon SIP Trunking configuration to the shared Avaya Solution & Interoperability Test Lab configuration. This signaling group reflects configuration not specifically related to Verizon testing. For example, calls using Avaya SIP Telephones and calls routed to other Avaya applications, such as Avaya Modular Messaging, use this signaling group. Again, the **Near-end Node Name** is "clan1" and the **Far-end Node Name** is "sm61", the node name of the Session Manager. Unlike the signaling groups used for the Verizon signaling, the **Far-end Network Region** is "1". The **Far-end Domain** is set to "avaya.com" matching the configuration in place prior to adding the Verizon SIP Trunking configuration. Change setting for **Alternate Route Timer (sec)** to "15". This allows more time for outbound PSTN calls to complete.

```
display signaling-group 32
                                SIGNALING GROUP

 Group Number: 32              Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n



   Near-end Node Name: clan1                 Far-end Node Name: sm61
 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? n                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec):15
```

4.8. SIP Trunk Groups

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

NOTE: For Verizon Business customers utilizing either Verizon **IP Contact Center** or **IP-IVR** service offers, at least one **Elite Agent license is <u>required</u>** to support the ability to utilize the Network Call Redirection capabilities of those services with Communication Manager. This license is required to enable the **ISDN/SIP Network Call Redirection** feature. This licensed feature must be turned **ON** (as shown in **Section 4.1**) to support Network Call Redirection. Additional details on how to configure Network Call Redirection in Communication Manager can be found within the supporting text and figures contained within this 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-CPESBC-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: 10
```

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 refresh SIP call sessions.

```
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
```

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
```

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. Although not strictly

necessary, the **Telephone Event Payload Type** has been set to "101" to match Verizon's 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 the SIP REFER method 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 "y" to allow REFER to be exercised.

The Verizon IPCC Services do not support Diversion headers or History-Info headers, 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? y
                     Send Diversion Header? n
                   Support Request History? n
              Telephone Event Payload Type: 101
```

The following shows Page 1 for trunk group 32, the bi-directional "tie" trunk group to Session Manager that existed before adding the Verizon SIP Trunk configuration to the shared Avaya Solution & Interoperability Test Lab network. Recall that this trunk is used for communication with other Avaya applications, such as Avaya Modular Messaging, and does not reflect any unique Verizon configuration.

```
display trunk-group 32                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 32                    Group Type: sip           CDR Reports: y
  Group Name: To SM61                        COR: 1        TN: 1        TAC: 132
   Direction: two-way         Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                     Signaling Group: 32
                                                   Number of Members: 100
```

The following shows Page 3 for trunk group 32. Note that unlike the trunks associated with Verizon calls that use "public" numbering, this tie trunk group uses a "private" **Numbering Format**.

```
display trunk-group 32                                          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




 Show ANSWERED BY on Display? y
```

The following shows Page 4 for trunk group 32. 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.

```
display trunk-group 32                                          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:
```

4.9. Vector Directory Numbers (VDNs) and Vectors for SIP NCR

This section describes the basic commands used to configure Vector Directory Numbers (VDNs) and corresponding vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality. These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to individual needs. Call centers may also use customer hosts running applications used in conjunction with Application Enablement Services (AES) to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes.

4.9.1 Post-Answer Redirection to a PSTN Destination

This section provides an example configuration of a vector that will use post-answer redirection to a PSTN destination. A corresponding detailed verification is provided in **Section 9.1.2**. In this example, the inbound toll-free call is routed to **vdn 65033** shown in the following abridged screen. The originally dialed Verizon IP Toll Free number may be mapped to VDN 65033 by Session Manager digit conversion, or via the incoming call handling treatment for the inbound trunk group on Communication Manager.

```
display vdn 65033                                          Page   1 of   3
                          VECTOR DIRECTORY NUMBER

                            Extension: 65033
                                Name*: Refer-Vector
                           Destination: Vector Number        33

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

VDN 65033 is associated with **vector 33**, which is shown below. Vector 33 plays an announcement (step 02) to answer the call. After the announcement, the "route-to number" (step 03) includes "~r+17328953304" where the number 732-895-3304 is a PSTN destination. This step causes a REFER message to be sent where the Refer-To header includes "+17328953304" as the user portion. Note that Verizon IP Contact Center services require the "+" in the Refer-To header for this type of call redirection. If the REFER triggered by step 03 fails, an announcement will be played and the call disconnected (step 04).

```
display vector 33                                          Page   1 of   6
                                CALL VECTOR

    Number: 33               Name: Refer-to-PSTN
                                        Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 67008
03 route-to     number ~r+17328953304   with cov n if unconditionally
04 disconnect   after announcement 67030
05
06
```

4.9.2 Post-Answer Redirection With UUI to a SIP Destination

This section provides an example of post-answer redirection with UUI passed to a SIP destination. A corresponding detailed verification is provided in **Section 9.1.3**. In this example, the inbound call is routed to **vdn 65034** shown in the following abridged screen. The originally dialed Verizon toll-free number may be mapped to VDN 65034 by Session Manager digit conversion, or via the incoming call handling treatment for the inbound trunk group on Communication Manager.

```
display vdn 65034                                          Page   1 of   3
                          VECTOR DIRECTORY NUMBER

                            Extension: 65034
                                Name*: Refer-Vector-with-UUI
                           Destination: Vector Number        34

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

To facilitate testing of NCR with UUI, the following vector variables were defined.

```
change variables                                           Page   1 of  39
                            VARIABLES FOR VECTORS

Var Description                Type    Scope Length Start Assignment       VAC
A   Test1                      asaiuui L     16     1
B   Test2                      asaiuui L     16     17
C
D
```

VDN 65034 is associated with **vector 34**, which is shown below. Vector 34 sets data in the vector variables "A" and "B" (steps 01 and 02) and plays an announcement to answer the call (step 04). After the announcement, the "route-to" number step includes "~r+18668510107". This step causes a REFER message to be sent where the Refer-To header includes "+18668510107" as the user portion. The Refer-To header will also contain the UUI set in variables A and B. Verizon will include this UUI in the INVITE ultimately sent to the SIP-connected target of the REFER, which is toll-free number "18668510107". In the sample configuration, where only one location was used, 866-851-0107 is another toll-free number assigned to the same circuit as the original call. In practice, NCR with UUI allows Communication Manager to send the call or customer-related data along with the call to another contact center. If REFER triggered by step 05 fails, an announcement will be played and the call disconnected (step 06).

```
display vector 34                                              Page   1 of   6
                                  CALL VECTOR

    Number: 34               Name: Refer-with-UUI
                                          Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? n
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 set           A      = none   CATR  1234567890123456
02 set           B      = none   CATR  7890123456789012
03 wait-time     2   secs hearing ringback
04 announcement 67030
05 route-to      number ~r+18668510107   with cov n if unconditionally
06 disconnect    after announcement 67030
07
```

4.10. 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 screen below, a specific Communication Manager extension "34006" is mapped to a Verizon IPTF number "866-851-8119", when the call uses trunk group 67. In the course of the testing, multiple Verizon toll-free numbers were associated with different Communication Manager extensions and functions.

Note that no Vector Directory Numbers (VDN) associated with vectors that can issue a REFER (as illustrated in the prior section) are listed in the screen below. Making an entry such as this for each VDN will trigger unnecessary SIP messaging for toll-free calls to VDNs that use SIP NCR with REFER, as summarized in **Section 1.3**..

```
change public-unknown-numbering 0                              Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                        Total
Ext Ext           Trk       CPN         CPN
Len Code          Grp(s)    Prefix      Len
                                                 Total Administered: 18
 5  3                                    5          Maximum Entries: 9999
 5  6                                    5
 5  34006         67        8668518119  10
 5  34008         67        8668510107  10
```

4.11. 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. 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 "8668502380" to extension "34006".

```
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 8668502380      all 34006
```

4.12. Modular Messaging Hunt Group

Although not specifically related to Verizon, this section shows the hunt group used for access to Avaya Modular Messaging. In the sample configuration, users with voice mail have a coverage path containing **hunt group 32**. Users can dial extension "33000" to reach Modular Messaging (e.g., for message retrieval). The following screen shows Page 1 of hunt-group 32.

```
display hunt-group 32                                          Page   1 of  60
                                HUNT GROUP

           Group Number: 32                                 ACD? n
             Group Name: Modular Messaging                Queue? n
        Group Extension: 33000                           Vector? n
             Group Type: ucd-mia                  Coverage Path:
                     TN: 1            Night Service Destination:
                    COR: 1                      MM Early Answer? n
          Security Code:                  Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

The following screen shows Page 2 of hunt-group 32, which routes with the **AAR access code** "8" and **Voice Mail Number** "33000".

```
display hunt-group 32                                          Page   2 of  60
                                HUNT GROUP









                     Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                    (e.g., AAR/ARS Access Code)
     33000                    33000                     8
```

4.13. AAR Routing to Modular Messaging via Session Manager

Although not specifically related to Verizon, this section shows the AAR routing for the number used in the hunt group in the previous section. The bold row shows that calls to the number range 33xxx, which includes the Modular Messaging hunt group 33000, will use **Route Pattern** 32. 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 32.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    2                     5    5     32       aar          n
    222                   5    5     31       aar          n
    3                     5    5     32       aar          n
    30100                 5    5     32       aar          n
    305                   5    5     32       aar          n
    309                   5    5     100      aar          n
    31                    5    5     12       aar          n
    3100                  5    5     32       aar          n
    3101                  5    5     32       aar          n
    32                    5    5     22       aar          n
    320                   5    5     42       aar          n
    33                    5    5     32       aar          n
```

4.14. Uniform Dial Plan (UDP) Configuration

Although not specifically related to Verizon, this section shows the UDP configuration, with the bold row showing that calls of the form 33xxx will be routed via AAR.

```
display uniform-dialplan 3                                      Page   1 of   2
                      UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                    Insert              Node
  Pattern        Len Del      Digits     Net Conv Num
 3               5   0                   aar  n
 30100           5   0                   aar  n
 309             5   0                   aar  n
 31              5   0                   aar  n
 32              5   0                   aar  n
 33              5   0                   aar  n
 400             5   0                   aar  n
 420             5   0                   aar  n
 5               5   0                   aar  n
 502             5   0                   aar  n
 60              5   0                   aar  n
 7               7   0                   aar  n
```

4.15. Route Pattern for Internal Calls via Session Manager

Although not specifically related to Verizon, this section shows the AAR routing for the number used in the hunt group for Modular Messaging. **Route pattern 32** contains trunk group "32", the "private" tie trunk group to Session Manager.

```
change route-pattern 32                                          Page   1 of   3
                    Pattern Number: 32  Pattern Name: To ASM
                                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: 32   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
```

4.16. Private Numbering

Although not specifically related to Verizon, this section shows the private numbering configuration associated with the calls using trunk group 32. The bold row configures any five digit number beginning with "3" (i.e., 3xxxx) that uses trunk group "32" to retain the original 5 digit number (i.e., no digit manipulation is specified, and the **Total Len** is "5").

```
change private-numbering 3                                       Page   1 of   2
                          NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private         Total
Len Code           Grp(s)     Prefix          Len
 5  3              32                         5     Total Administered: 5
 5  30101          52         9133245977      10       Maximum Entries: 540
 5  34000          52         9133245981      10
 5  34001          52         9133245980      10
```

4.17. 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. **Coverage Path 1** is set to "32" to give this user coverage to Avaya Modular Messaging.

```
change station 34006                                             Page   1 of   5
                                    STATION

Extension: 34006                       Lock Messages? n                BCC: 0
     Type: 9630                        Security Code: 123456            TN: 1
     Port: S00532                    Coverage Path 1: 32               COR: 1
     Name: AllanH96xxH               Coverage Path 2:                  COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
             Loss Group: 19           Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 34006
```

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

```
change station 34006                                          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 - Send Calling Number and Name?
       Service Link Mode: as-needed                    EC500 State: enabled
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections?
y
  Emergency Location Ext: 34006         Always Use? n IP Audio Hairpinning? n
```

4.18. Coverage Path

This section illustrates an example coverage path for a station with a mailbox on Avaya Modular Messaging. Hunt group 32 ("h32"), the hunt group to Modular Messaging, is **Point1** in coverage path 32.

```
change coverage path 32                                       Page   1 of   1
                                COVERAGE PATH

                  Coverage Path Number: 32
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                      Next Path Number:           Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
       Don't Answer?             y               y         Number of Rings: 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: h32            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

4.19. Saving Communication Manager Configuration Changes

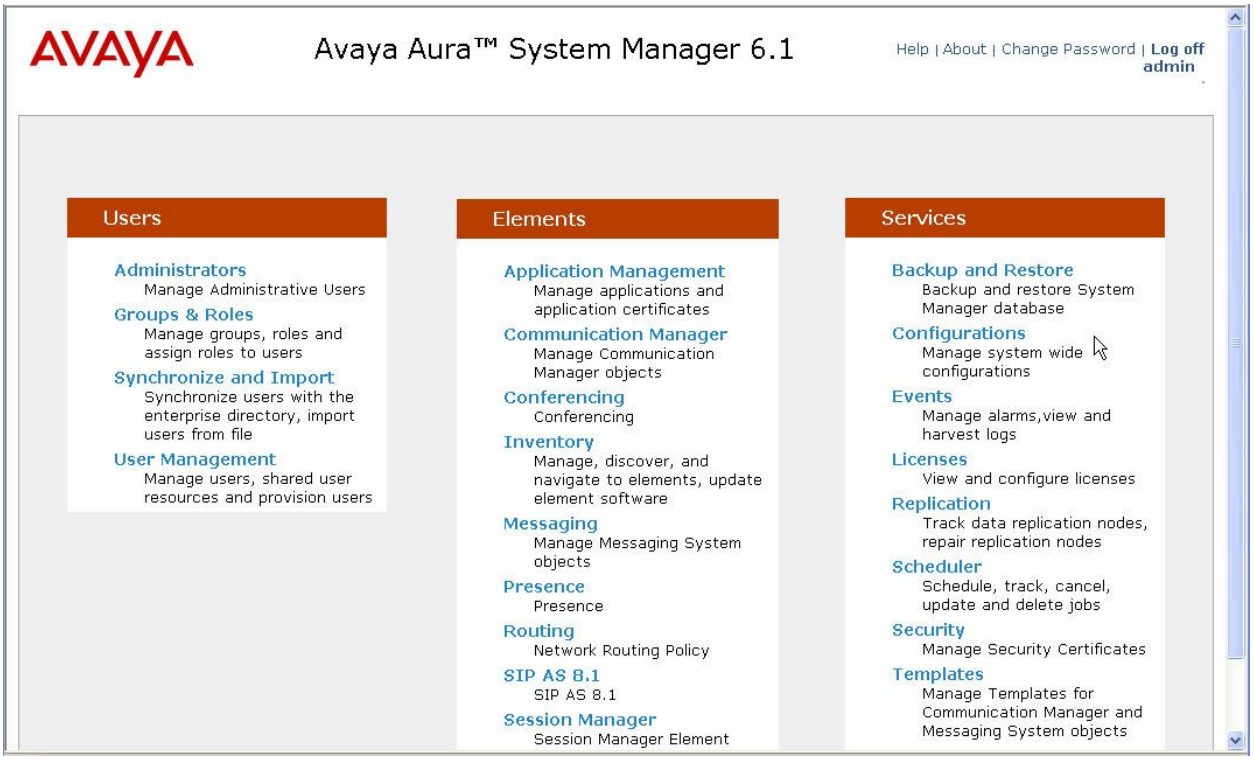
The command "save translation all" can be used to save the configuration.

5. Avaya Aura® Session Manager Provisioning

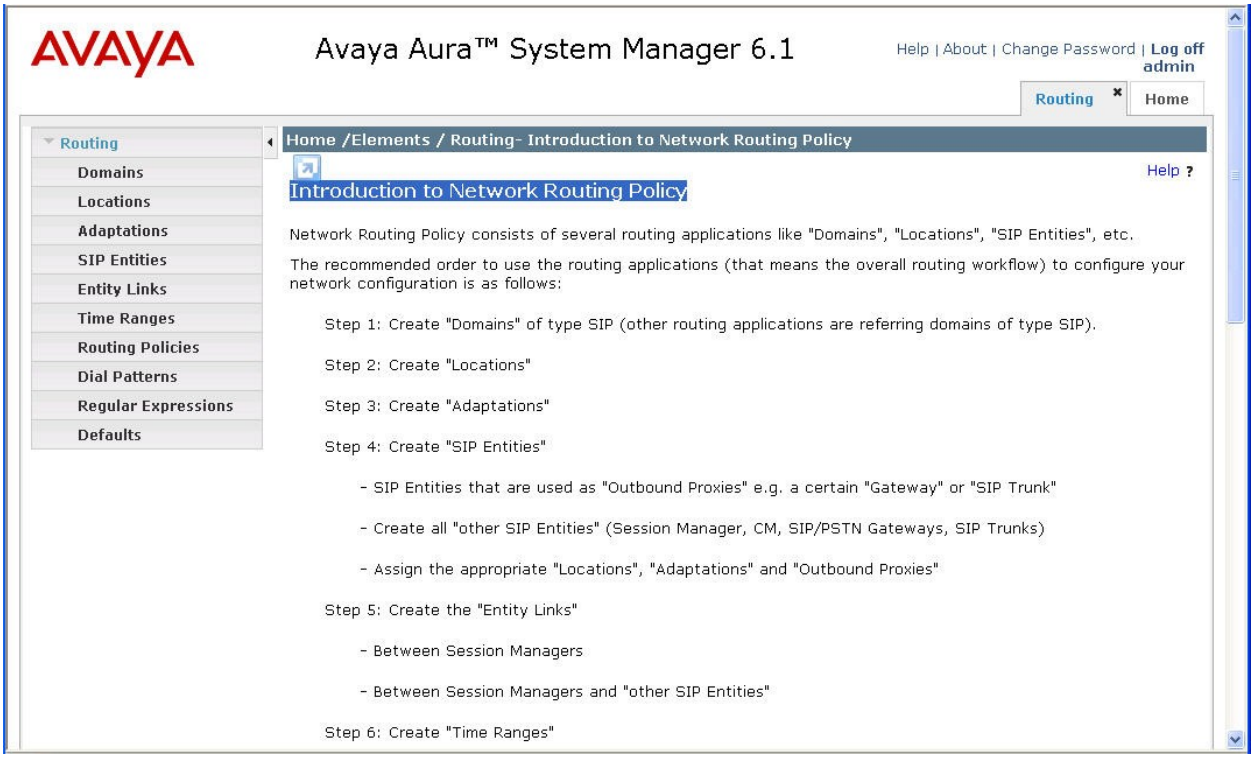
This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration used in the verification of these Application Notes.

Note – The following sections assume that Avaya Aura® Session Manager and Avaya Aura® System Manager have been installed and that network connectivity exists between the two. For more information on Avaya Aura® Session Manager see **[3]**.

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. At the **Session Manager Log On** screen, provide the appropriate credentials and click on **Login** (not shown). The initial screen shown below is then displayed.



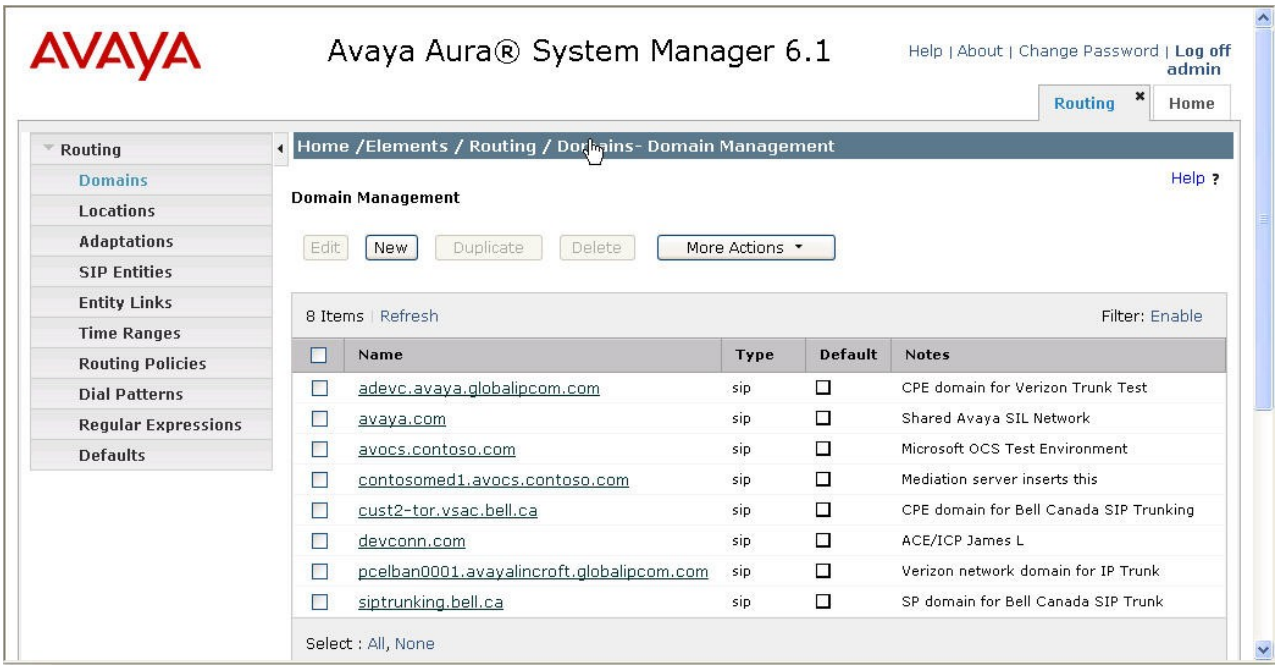
Most of the configuration items are performed in the Routing Element. Click on **Routing** in the **Elements** column to bring up the **Introduction to Network Routing Policy** screen. The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.



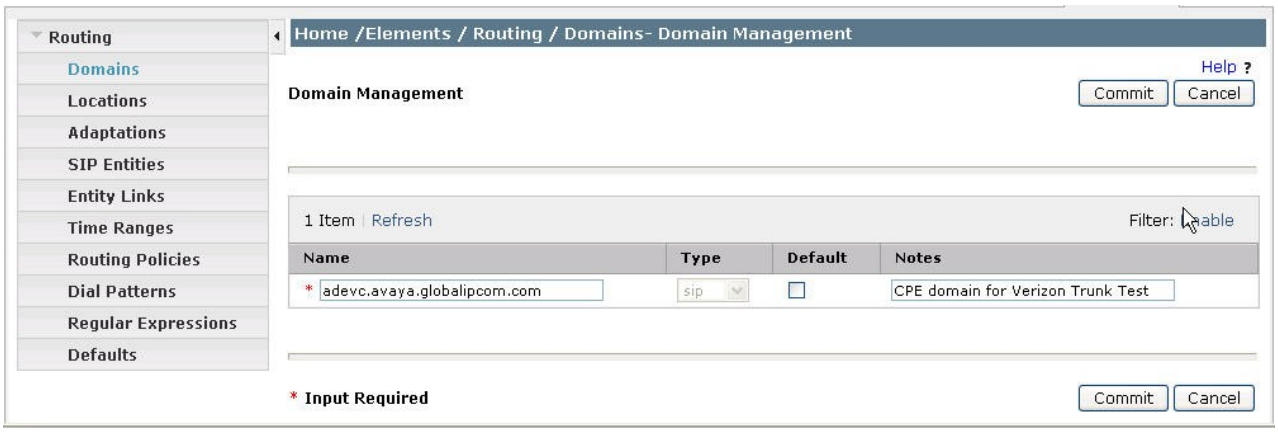
5.1. Domains

To change or add 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, including an Avaya Modular Messaging system with SIP integration to Session Manager. The domain "avaya.com" is not known to the Verizon production service.



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 for inbound toll-free calls.

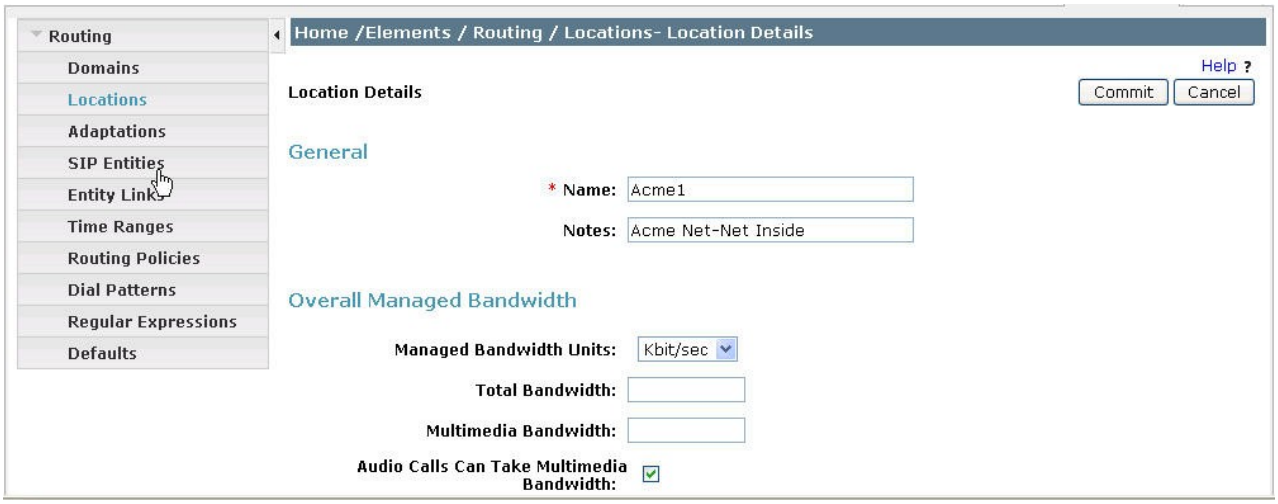


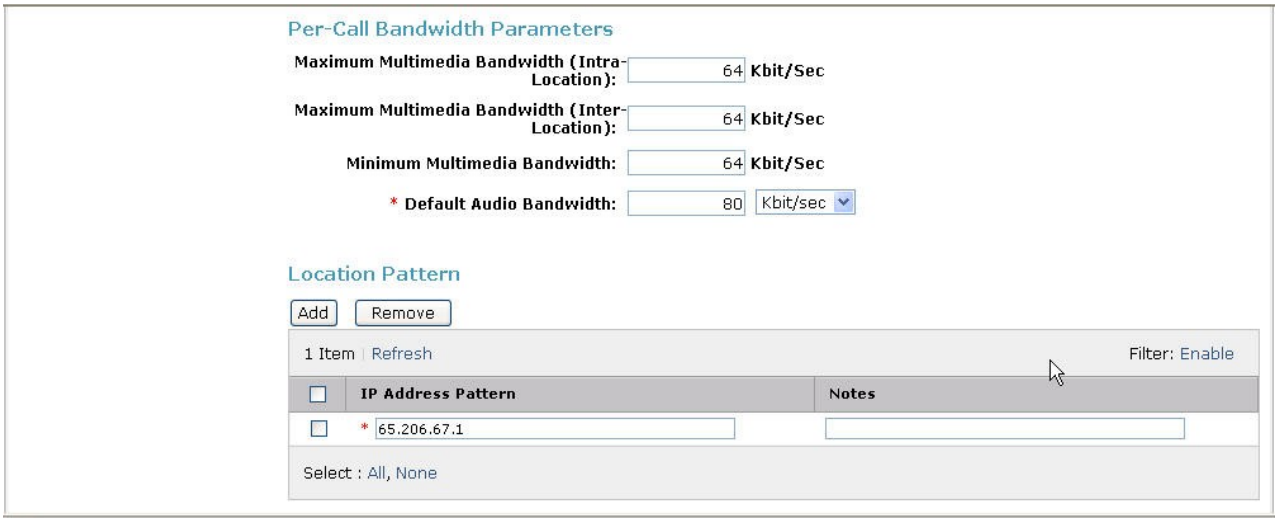
5.2. Locations

To change or add 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.

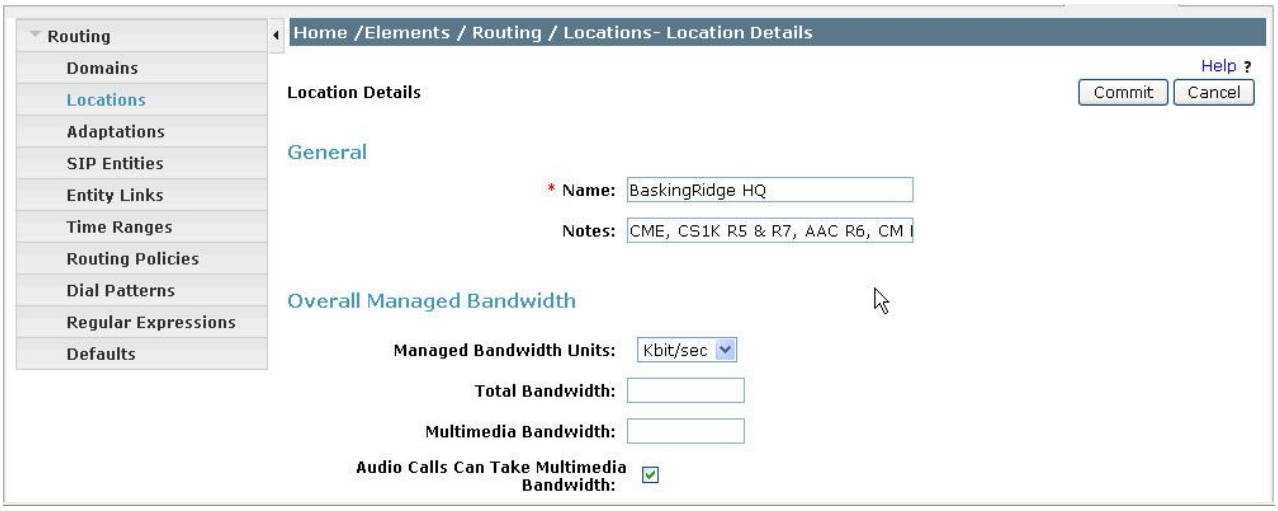
The following screenshots show upper and lower portions of the **Location Details** screen for the Location named "Acme1", corresponding to the primary Acme Packet Net-Net SBC. Later, the location with name "Acme1" will be assigned to the corresponding "Acme1" SIP Entity. The IP address 65.206.67.1 of the inside (private) interface of "Acme1" is entered in the **IP Address Pattern** field.

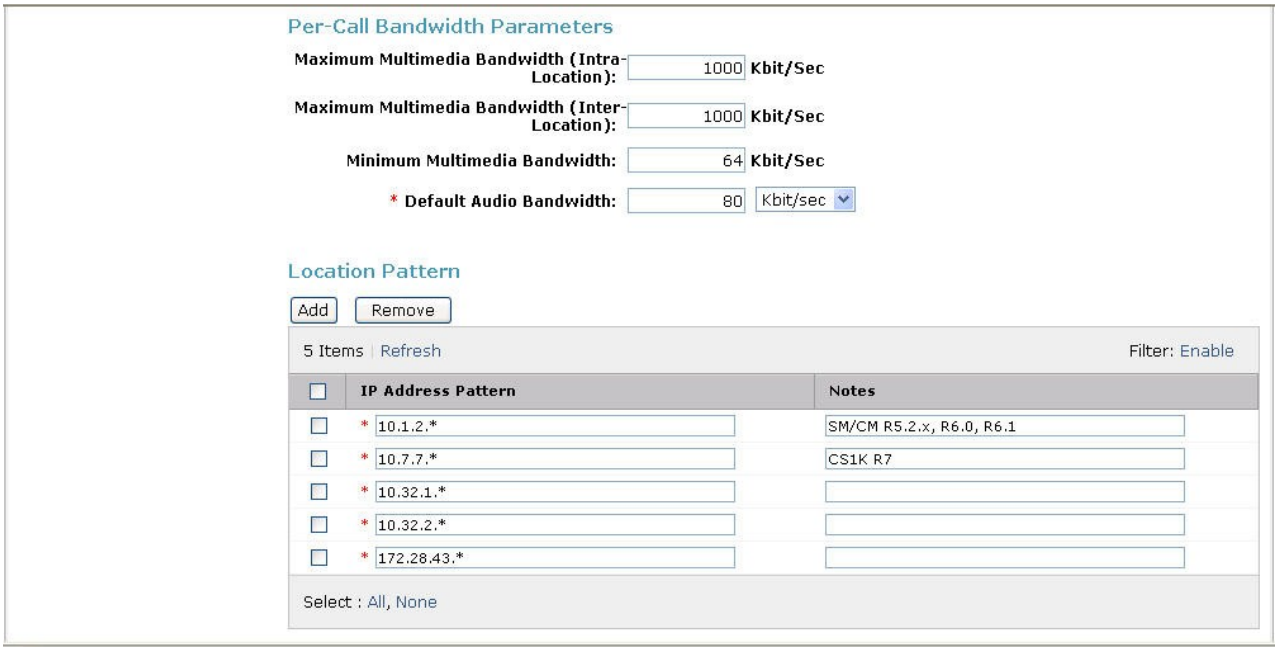
The screens also show various settings relating to enhanced Call Admission Control with the **Overall Managed Bandwidth** and **Per-Call bandwidth Parameters** sections. These were not used in the sample configuration.





The following screens show the upper and lower portions of the **Location Details** screen for the Location named "BaskingRidge HQ". The IP addresses administered for this location correspond to the shared components of the Avaya Solution & Interoperability Test Lab environment, such as Communication Manager Release 5.2.1, Session Manager Release 6.1, and other communication and application servers.

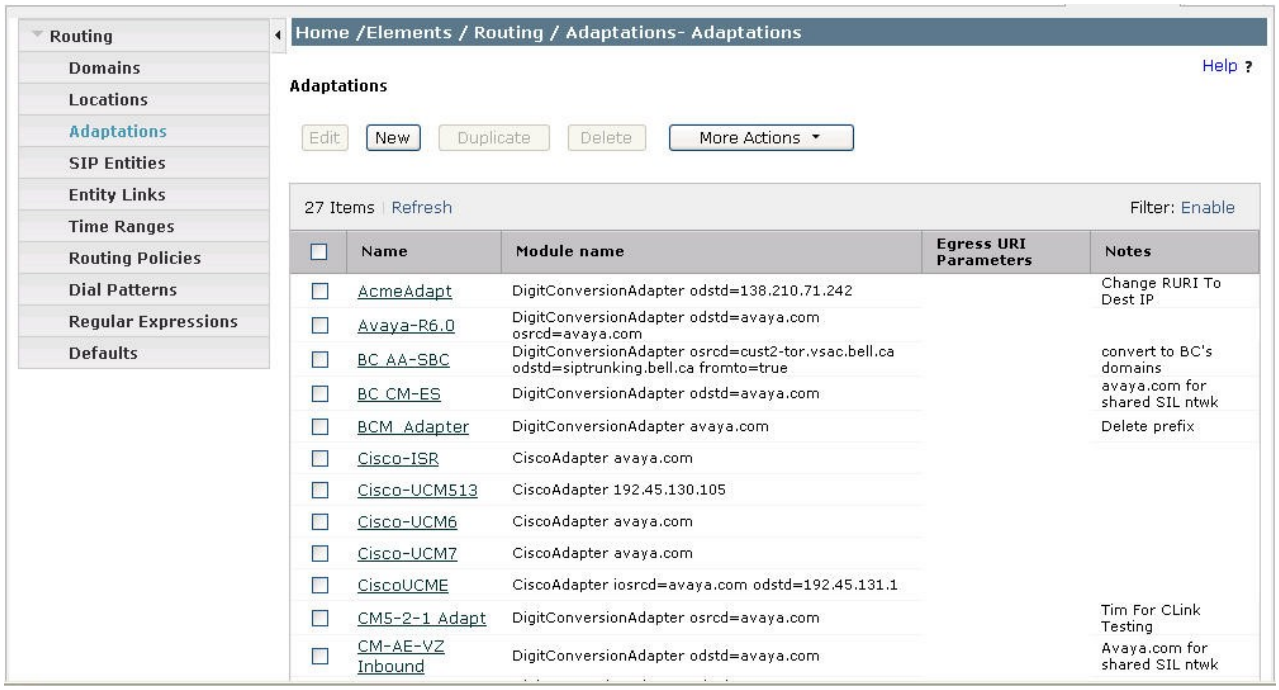




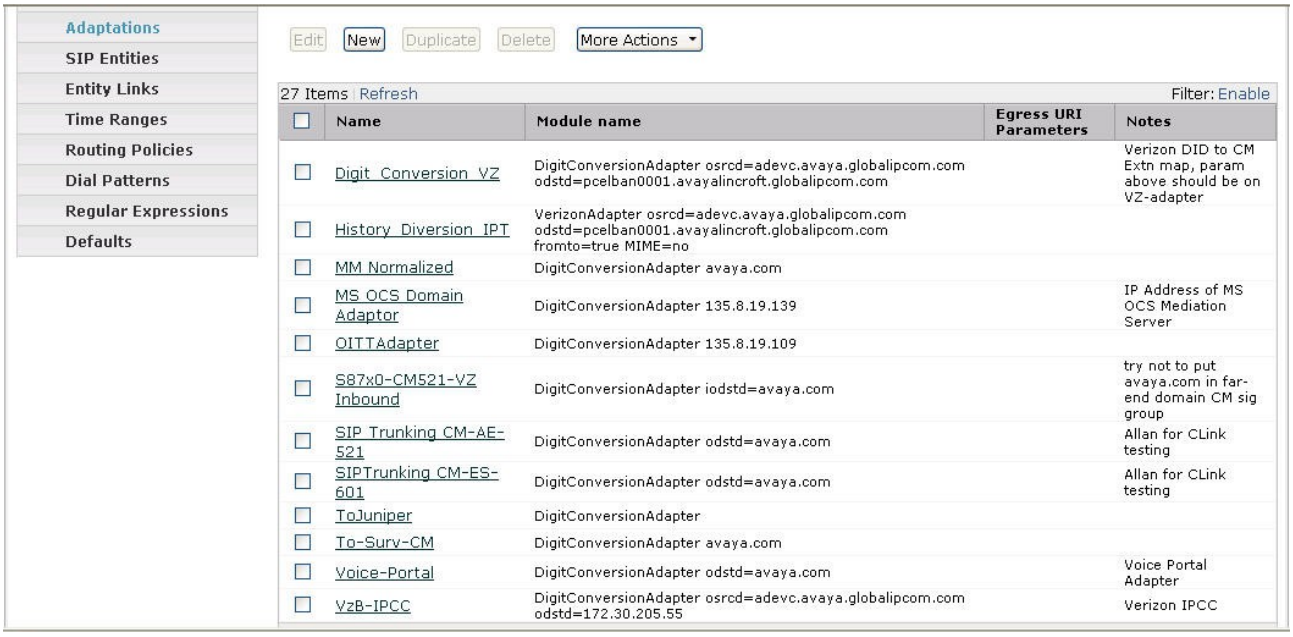
5.3. Adaptations

To change or add 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.



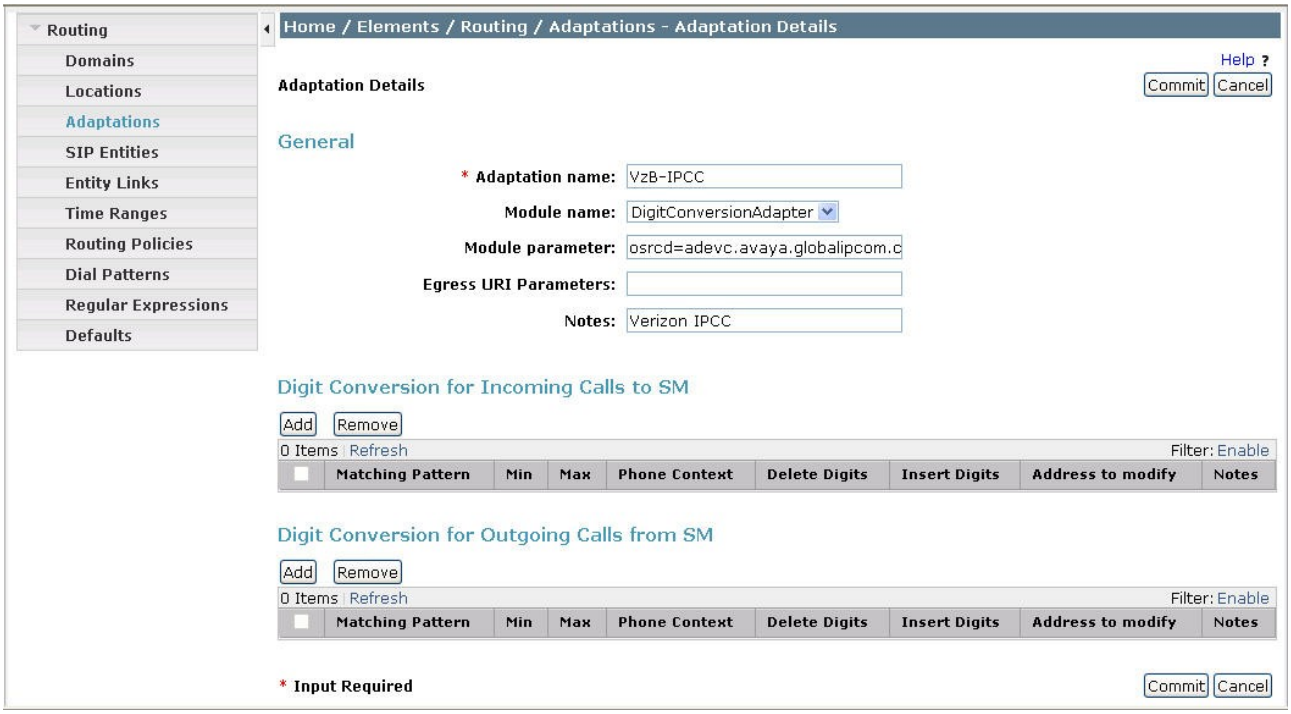
The following screen shows another portion of the list of adaptations in the sample configuration.



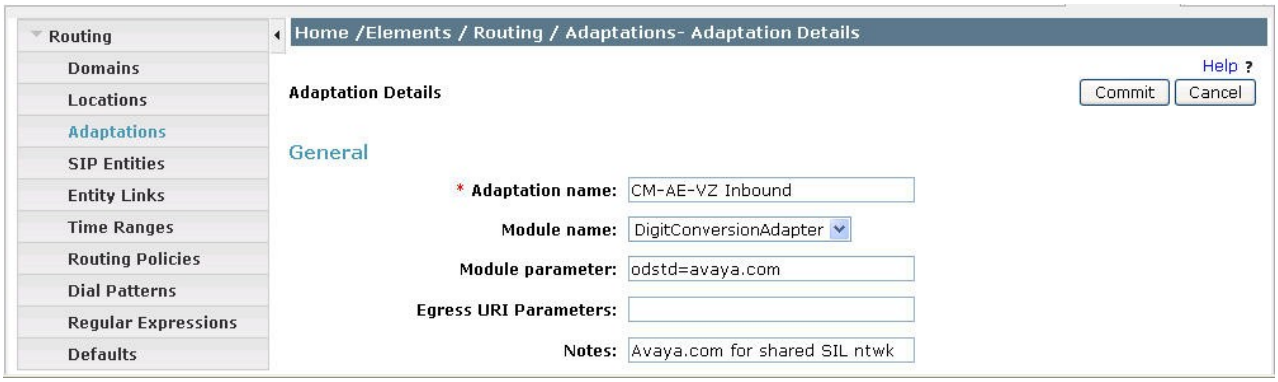
The adapter named "VzB-IPCC" shown in the 2nd screen above will later be assigned to the Acme Packet Net-Net SBC SIP Entity. This adaptation uses the "DigitConversionAdapter" and specifies two parameters that are used to adapt the FQDN to the domains expected by the Verizon network in the sample configuration:

- "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 Communication Manager 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 necessary 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. For example, the BYE message sent from the enterprise to Verizon to end a toll-free inbound call will use this IP Address as the host portion of the Request URI.

The following screen shows the complete **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.

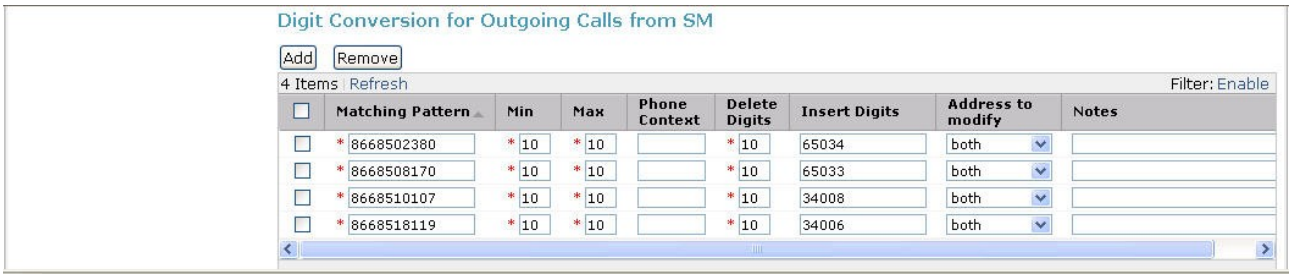


The adapter named "CM-AE-VZ Inbound" shown below will later be assigned to the Communication Manager SIP Entity for calls involving Verizon. This adaptation uses the "DigitConversionAdapter" and specifies the "odstd=avaya.com" **Module parameter** to adapt the domain to the domain expected by Communication 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 to the Avaya CPE, the Request-URI header sent to Communication Manager will contain "avaya.com" as expected by Communication Manager in the shared Avaya Solution & Interoperability Test Lab configuration. Depending on the Communication Manager configuration, it may not be necessary for Session Manager to adapt the domain in this fashion.



Scrolling down, the following screen shows a portion of the "CM-AE-VZ Inbound" adapter that can be used to convert digits between the extension numbers used on Communication Manager and the toll-free numbers assigned by Verizon. An example portion of the settings for **Digit Conversion for Incoming Calls to SM** is shown below. Since this adapter will be applied to the Communication Manager SIP Entity later on, the settings for incoming calls to SM correspond with outgoing calls from Communication Manager to Session Manger for egress to Verizon (via Acme Packet Net-Net SBC).

An example portion of the settings for **Digit Conversion for Outgoing Calls from SM** (i.e., inbound to Communication Manager) is shown below. During the testing, the digit conversion was varied to allow the same toll-free number to be used to test different Communication Manager call destinations.



In general, digit conversion such as this, that converts a Communication Manager extension (e.g., 65034, in this case, a VDN) to a corresponding toll-free number (e.g., 866-850-2380), can be performed in Communication Manager or in Session Manager. In the example shown above, if a user on the PSTN dials 866-850-2380, Session Manager will convert the number to extension 65034 before sending the SIP INVITE to Communication Manager. As such, it would not be necessary to use the incoming call handling table of the receiving Communication Manager trunk group to convert the toll-free number to its corresponding destination number.

5.4. SIP Entities

To change or add 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.

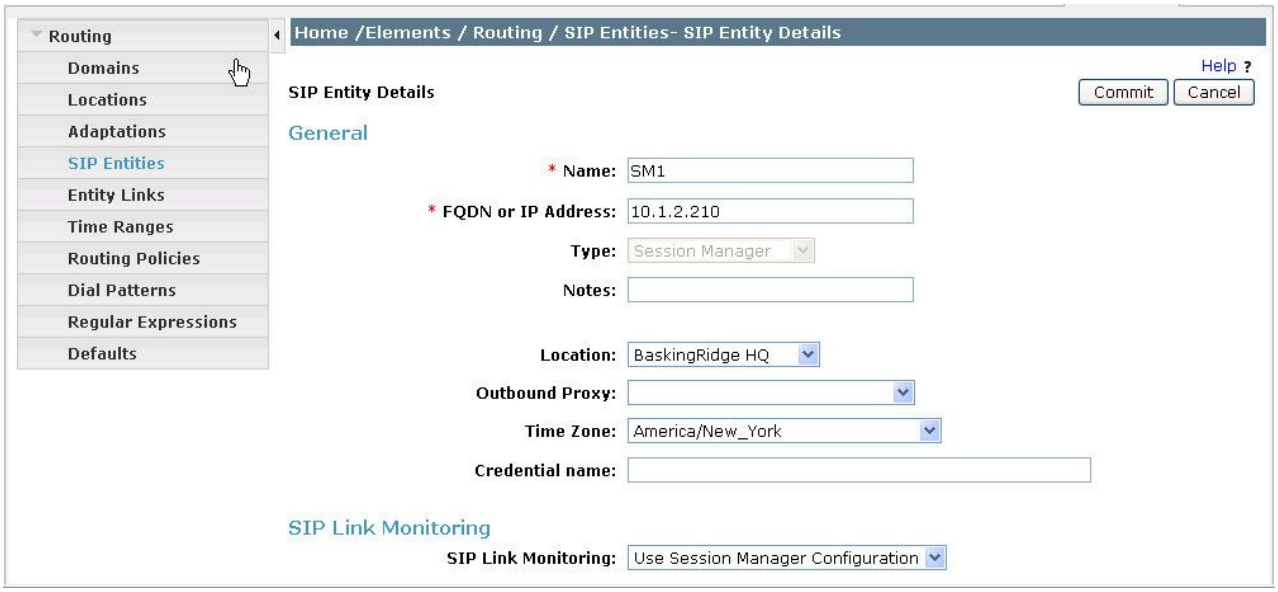
The following screen shows a portion of the list of configured SIP entities. In this screen, the SIP Entities named "Acme1 (corresponding to the Acme Packet Net-Net SBC) and "alpinemas1" (corresponding to Modular Messaging Application Server) are relevant to these Application Notes. Other relevant SIP Entities named "CM521-AE-clan1-5067" for Communication Manager (configured as Access Element) and "SM1" for Session Manager are listed in other pages of the SIP Entity list (not shown).

47 Items | Refresh Filter: Enable

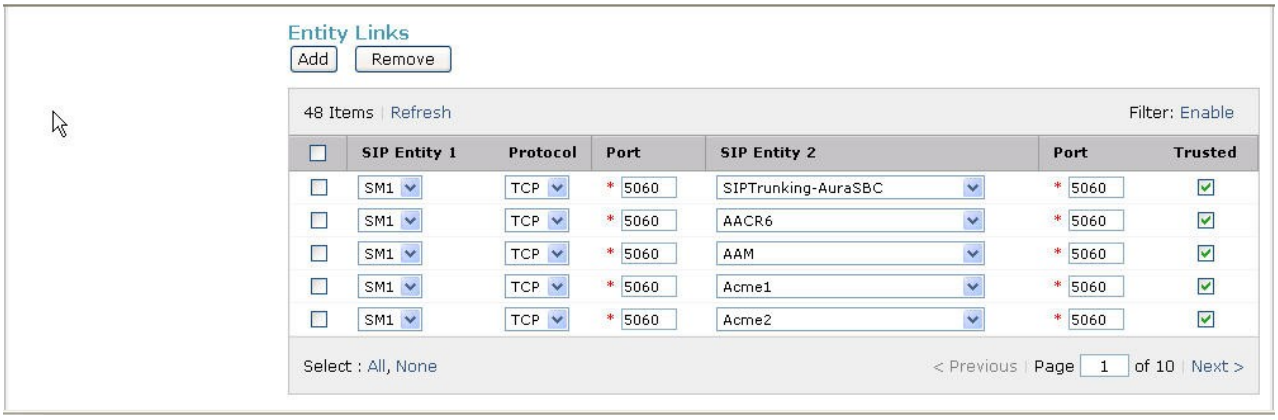
Name	FQDN or IP Address	Type	Notes
AACR6	10.7.7.185	Other	Avaya Aura Conferencing R6
AAM	135.8.139.136	Modular Messaging	For use by Tony M's group
ACE	10.32.48.26	SIP Trunk	ACE R2.2.3 James Liu
Acme1	65.206.67.1	Other	Inside IP Acme1
Acme2	65.206.67.21	Other	Acme2 Inside
AG2330	192.168.75.160	Other	
AllanC-S8300-G350	10.32.2.80	CM	For Survivability Test
alpinemas1	135.8.139.31	Modular Messaging	For use by Tony M's group
AudioCodes M1000	m1000.avaya.com	Other	QSIG/SIP GW for CS1000
AuraSBC	65.206.67.93	Other	Avaya Aura SBC Inside IP
BCM50 R6	10.7.7.221	Other	
BR2 AudioCodes MP114	192.168.75.110	Other	SIP Media Gateway
BR2 AudioCodes MP118	192.168.75.100	Other	SIP Media Gateway
CallCenter	10.1.2.233	CM	
Cisco 2921 SRST	120.1.1.1	Other	Branch 2

Select : All, None < Previous | Page 1 of 4 | Next >

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 signaling interface IP address (10.1.2.210), 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. Unless changed elsewhere, links from other SIP entities to this instance of Session Manager will use the default SIP Link Monitoring timers, configurable at the Session Manager level. If desired, these timers may be customized for each SIP entity.

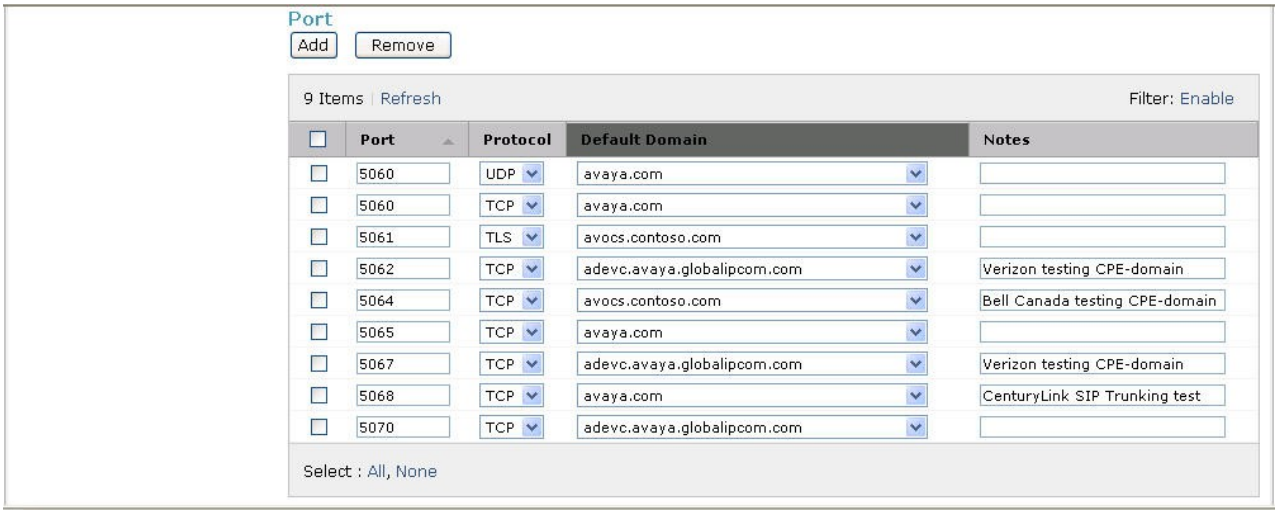


Scrolling down, the following screen shows the middle portion of the **SIP Entity Details**, a listing of the **Entity Links** previously configured for "SM1". The links relevant to these Application Notes are described in the following section.

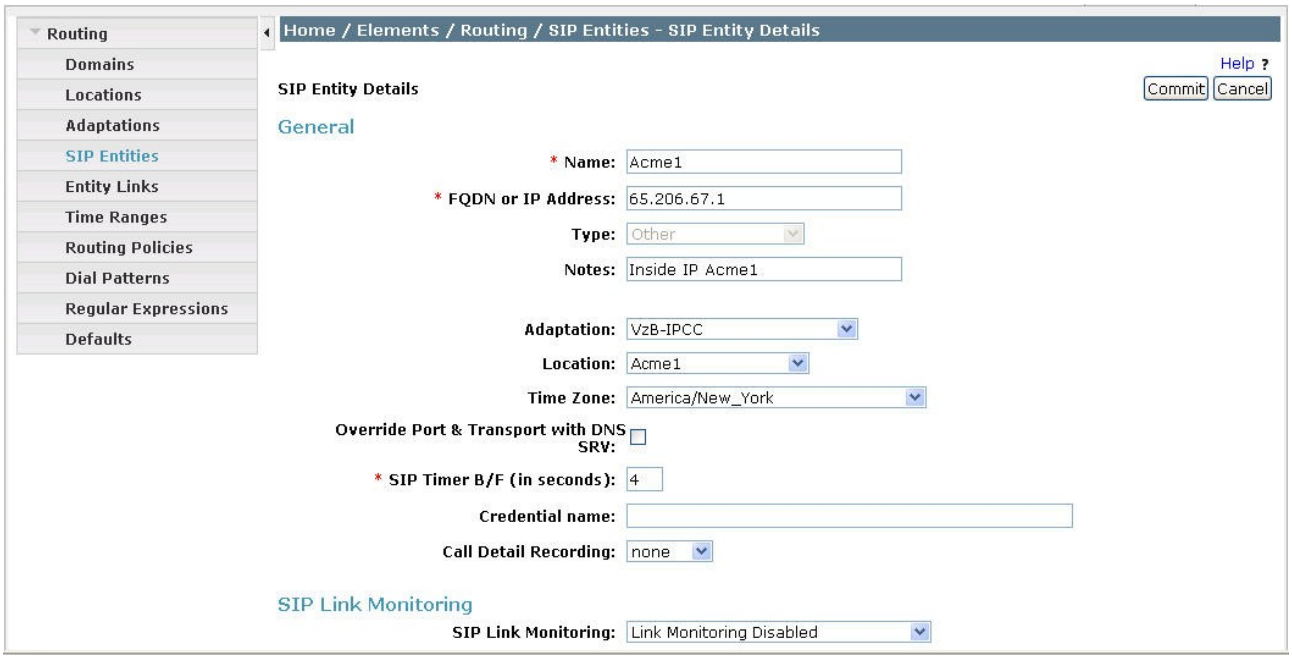


Scrolling down, the following screen shows the lower portion of the **SIP Entity Details**, a listing of the configured ports for "SM1". In the sample configuration, TCP port 5060 was already in place for the shared test environment, using **Default Domain** "avaya.com". To enable calls with Verizon

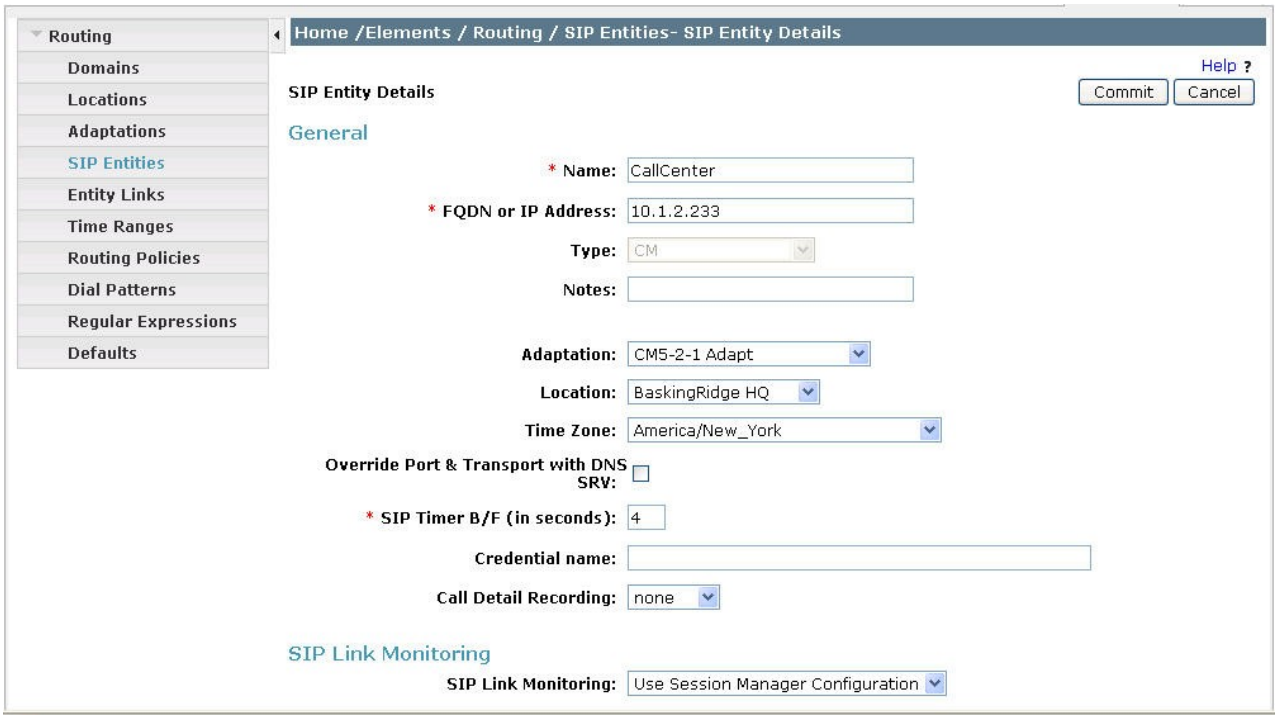
to be distinguished from other types of SIP calls using the same Session Manager, TCP port 5067 was added, with **Default Domain** "adevc.avaya.globalipcom.com". Click the **Add** button to configure a new port. TCP is used in the sample configuration for improved visibility during testing.



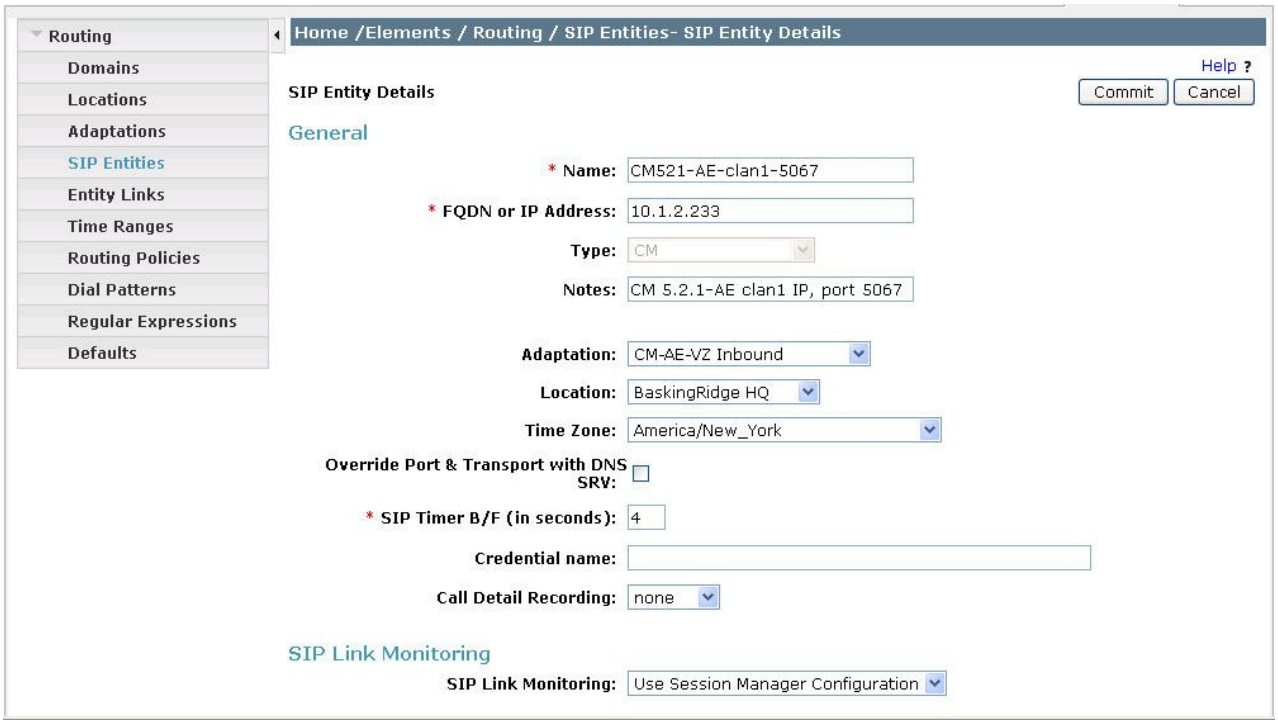
The following screen shows the **SIP Entity Details** corresponding to "Acme1". The **FQDN or IP Address** field is configured with the Acme Packet Net-Net SBC inside IP Address (65.206.67.1). "Other" is selected from the **Type** drop-down menu for SBC SIP Entities. This Acme Packet Net-Net SBC has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



The following screen shows a portion of the **SIP Entity Details** corresponding to a Communication Manager SIP Entity named "CallCenter" This is the SIP Entity that was already in place in the shared Avaya Solution & Interoperability Test Lab environment, prior to adding the Verizon IPCC trunk configuration. The **FQDN or IP Address** field contains the IP address of the CLAN card in the Avaya G650 Media Gateway controlled by Communication Manager. "CM" is selected from the **Type** drop-down menu. In the shared test environment, the **Adaptation** "CM5-2-1 Adapt" and **Location** "BaskingRidge HQ" had already been assigned to this Communication Manager SIP entity.



The following screen shows the **SIP Entity Details** for an entity named "CM521-AE-clan1-5067". This entity uses the same **FQDN or IP Address** (10.1.2.233) as the prior entity with name "CallCenter"; both correspond to the IP address of the CLAN card in the Avaya G650 Media Gateway controlled by Communication Manager. Later, a unique port, 5067, will be used for the Entity Link between Session Manager and Communication Manager named "CM521-AE-clan1-5067". Using a different port is one approach that will allow Communication Manager to distinguish traffic from Verizon from other SIP traffic arriving from the same IP address of Session Manager. The adapter "CM-AE-VZ Inbound" is applied to this SIP entity. Recall that this adapter is used to adapt the domain as well as map the Verizon 10 digit DID numbers to the corresponding Communication Manager extensions and other destination numbers.



5.5. Entity Links

To change or add 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.

Note – In the Entity Link configurations below (and in the Communication Manager SIP trunk configuration), TCP was selected as the transport protocol for the Avaya CPE in the sample configuration. TCP was used to facilitate trace analysis during network verification. The use of the TLS protocol is recommended by Avaya in customer deployments.

The following screen shows a partial list of configured links. In the screen below, the links named "Acme1", "CM-AE-clan1-5067", and "CallCenter" are relevant to these Application Notes. Each

of the links uses the SIP entity named “SM1” as **SIP Entity 1**, and the appropriate entity, such as “Acme1” or “Acme2” for **SIP Entity 2**.

Note that there are two SIP Entity Links, using different TCP ports, linking the same SM1 with the same Communication Manager. For one link, named “CallCenter”, both entities use port 5060. For the other, named “CM-AE-clan1-5067”, both entities use port 5067.

48 Items | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
AACR6	SM1	TCP	5060	AACR6	5060	☑	
AAM-SM1-TCP	SM1	TCP	5060	AAM	5060	☑	Between SM1 and AAM
Acme1	SM1	TCP	5060	Acme1	5060	☑	
Acme2	SM1	TCP	5060	Acme2	5060	☑	
AG2330	SM1	TCP	5060	AG2330	5080	☑	
AuraSBC	SM1	TCP	5060	AuraSBC	5060	☑	
Call Center	SM1	TCP	5060	CallCenter	5060	☑	
Cisco-UCM6	SM1	TCP	5060	Cisco-UCM6	5060	☑	
Cisco-UCM7	SM1	TCP	5060	Cisco-UCM7	5060	☑	
Cisco UCME 513	SM1	TCP	5060	CUCM-513	5060	☑	
CiscoUCME-Link	SM1	TCP	5060	CiscoUCME	5060	☑	
CiscoUCME-Link-UDP	SM1	UDP	5060	CiscoUCME	5060	☑	
CM521-AE-clan1-5067	SM1	TCP	5067	CM521-AE-clan1-5067	5067	☑	For Verizon IP Trunk testing
CM521-AE-clan1-5068	SM1	TCP	5068	CM521-AE-clan1-5068	5068	☑	
CM521-ForSPs	SM1	TLS	5066	CM5-2-1	5066	☑	Tim Link to CM for SPs

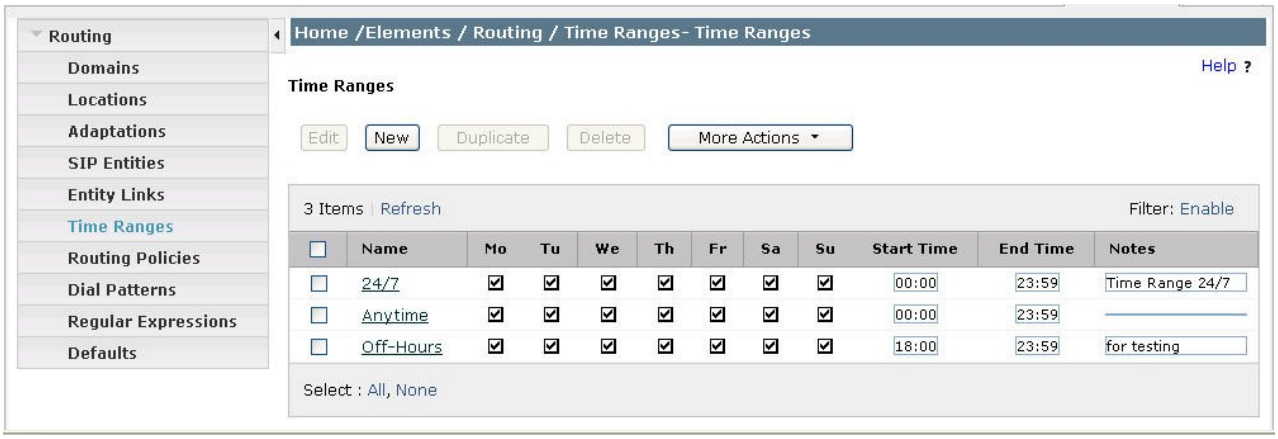
Select : All, None < Previous | Page 1 of 4 | Next >

The link named “CallCenter” existed in the shared configuration prior to adding the Verizon IP Trunk-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.

The link named “CM-AE-clan1-5067” also links Session Manager “SM1” with the same Communication Manager. However, this link uses port 5067 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.

5.6. Time Ranges

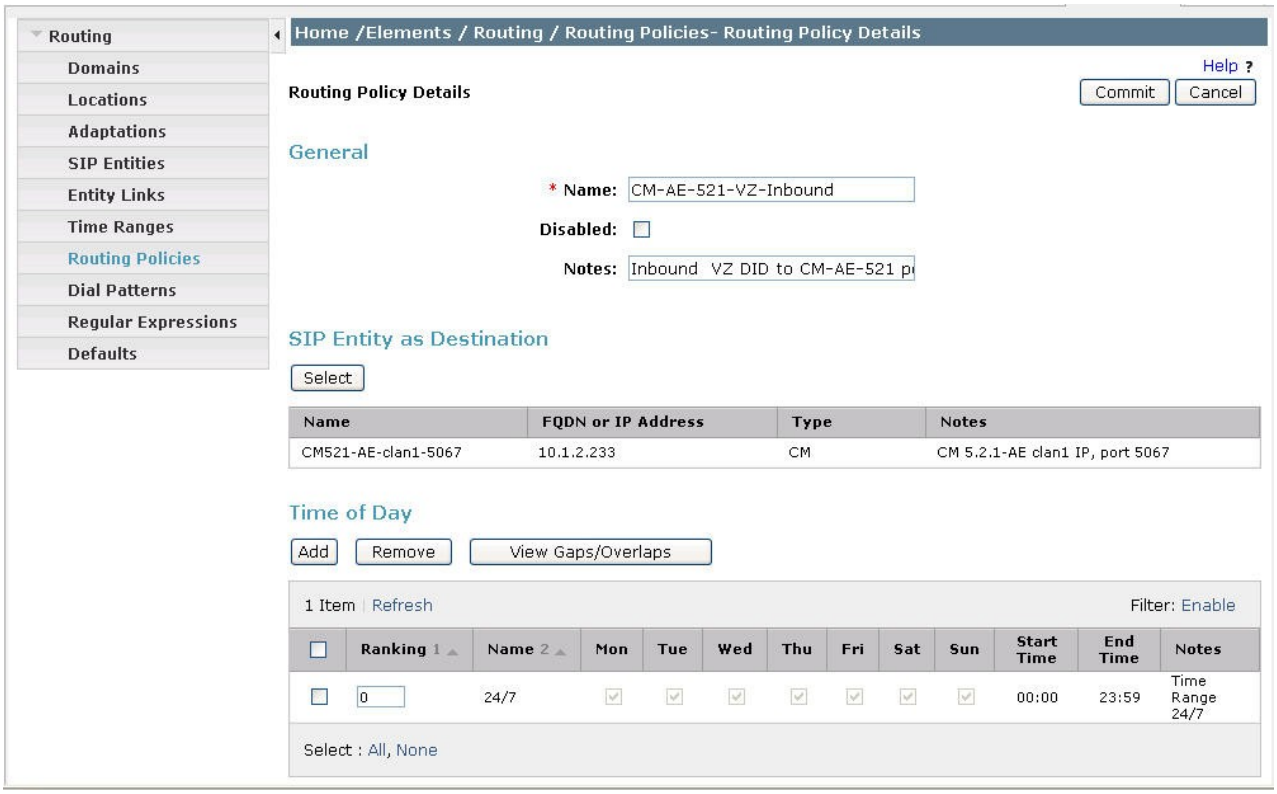
To view or change Time Ranges, select **Routing → Time Ranges**. The Routing Policies shown subsequently will use the "24/7" range since time-based routing was not the focus of these Application Notes. Click the **Commit** button after changes are completed.



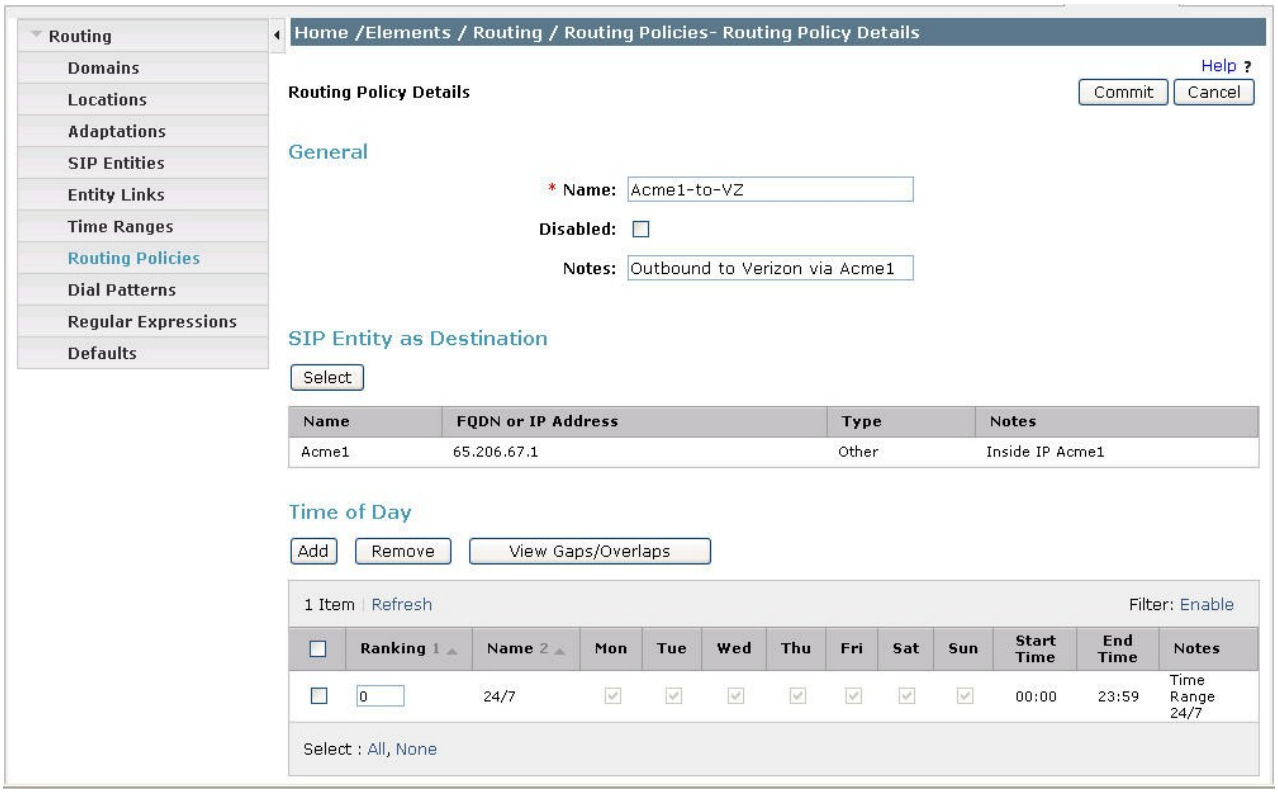
5.7. Routing Policies

To change or add 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.

The following screen shows the **Routing Policy Details** for the policy named "CM-AE-521-VZ-Inbound" associated with inbound IPTF calls from Verizon to Communication Manager. Observe the **SIP Entity as Destination** is the entity named "CM521-AE-clan1-5067". After dial patterns are assigned to use this routing policy, the lower portion of the screen will show the dial patterns using the routing policy.



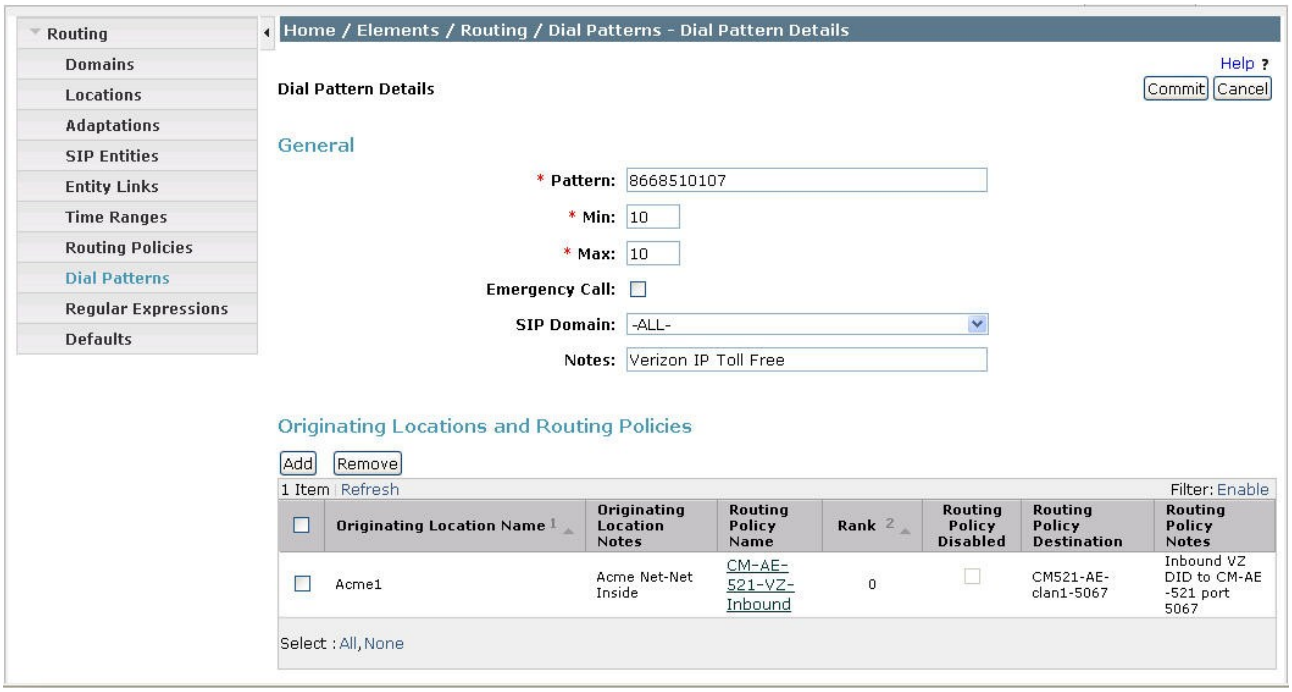
The following screen shows the **Routing Policy Details** for the policy named "Acme1-to-VZ" associated with outgoing calls from Communication Manager to the PSTN via Verizon through Acme1. Observe the **SIP Entity as Destination** is the entity named "Acme1". After dial patterns are assigned to use this routing policy, the lower portion of the screen will show the dial patterns using the routing policy.



5.8. Dial Patterns

To change or add 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.

The following screen illustrates an example dial pattern used to route an inbound IPTF call to the enterprise. When a user on the PSTN dials a toll-free number such as 866-851-0107, Verizon delivers the number to the enterprise, and the Acme Packet Net-Net SBC sends the call to Session Manager. The dial pattern below matches on 866-851-0107 specifically. Dial patterns can alternatively match on ranges of numbers. Under **Originating Locations and Routing Policies**, the routing policy named "CM-AE-521-VZ-Inbound" is selected, which sends the call to Communication Manager using the routing policy destination "CM521-AE-clan1-5067" as described previously. The **Originating Location Name** is "Acme1".



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 (not shown).

6. Acme Packet Net-Net Session Border Controller

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.

The Acme Packet Net-Net SBC configuration used in the verification of these Application Notes is similar to the configuration detailed in previously published Application Notes [JF-VZIPCC]. Therefore, this section will focus on differences from the configuration in the previously published

Application Notes, and new recommendations for the Acme Packet Net-Net SBC configuration due to the new releases of Avaya Aura® Session Manager and Avaya Aura® Communication Manager. See reference [JF-VZIPCC] for detailed configuration steps covering the Acme Packet Net-Net SBC.

6.1. Session Agent for Session Manager Release 6

Conceptually, the **session-agent** configured for Session Manager Release 6.1 is the same as the one configured in **Section 5.3.7.2** of reference [JF-VZIPCC], which defined a session agent for a prior release of Session Manager. The relevant part of the session agent configuration is included below, since the IP Address of Session Manager is different in these Application Notes.

```
session-agent
        hostname                10.1.2.210
        ip-address              10.1.2.210
        port                    5060
        state                   enabled
        app-protocol            SIP
        transport-method        StaticTCP
        realm-id                INSIDE
        description             Fred-SM61
        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
        reuse-connections       TCP
        tcp-keepalive           enabled
        tcp-reconn-interval     10
```

6.2. Session Agent for Verizon IPCC Network

Conceptually, the **session-agent** configured for the Verizon IPCC network is the same as the "outside session agent" configured in **Section 5.3.7.1** of reference [JF-VZIPCC]. The relevant part of the session agent configuration is included below, since the IP Address and port used by Verizon is different in these Application Notes.

```
session-agent
        hostname                172.30.205.55
        ip-address              172.30.205.55
        port                    5072
        state                   enabled
        app-protocol            SIP
        transport-method        UDP
        realm-id                OUTSIDE
        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

6.3. Session Agent Group for Session Manager Release 6.1

Conceptually, the **session–group** (session agent group) "ENTERPRISE" configured for the Avaya CPE is the same as the one configured in **Section 5.3.8.2** of reference [JF-VZIPCC], which defined a session agent group whose destination was the session agent corresponding to a prior release of Session Manager. The relevant portion of the configuration is included here, since the IP Address of the destination Session Manager is different in these Application Notes. When more than one instance of Session Manager is included in a configuration, the use of a **session-group** allows each of the Session Manager instances to be included in the session group. The Session Manager instance selected for a given call is based on the "strategy" parameter (e.g., "Hunt" or "RoundRobin"). In the sample configuration with only one Session Manager instance, the strategy is moot.

session-group

group-name	ENTERPRISE
state	enabled
app-protocol	SIP
strategy	RoundRobin
dest	10.1.2.210

6.4. Session Agent Group for Verizon IPCC

Conceptually, the **session-group** (session agent group) "SERV_PROVIDER" configured for the Verizon IPCC Network is the same as the one configured in **Section 5.3.8.1** of reference [JF-VZIPCC], which defined a session agent group whose destination was the Verizon IPCC session agent IP Address. The relevant portion of the configuration is included here, since the IP Address of the destination is different in these Application Notes.

session-group

group-name	SERV_PROVIDER
state	enabled
app-protocol	SIP
strategy	Hunt
dest	172.30.205.55

6.5. SIP Header Manipulation

In **Section 5.3.11** of reference [JF-VZIPCC], a SIP header manipulation is defined and applied to the "outside" realm towards Verizon. This sip-manipulation contains various header rules mainly to replace inside or private IP Addresses in headers with the appropriate outside or public IP Addresses in the SIP messages sent to Verizon. Since the rules defined in [JF-VZIPCC] use variables such as $REMOTE_IP to define replacement values, no change is required owing to the

different IP Address used by the Verizon IPCC service for the configuration in these Application Notes. This section recommends other changes to the SIP header manipulation.

6.5.1 P-Site Header Removal

Session Manager Release 6.1 inserts a P-Site header which contains the IP Address of System Manager as a parameter. Since there is no value in sending this header to Verizon in the sample configuration, the header can be stripped by the SBC. Calls can still be completed successfully if the configuration in this section is not performed and the P-Site header is sent to Verizon. This information is included to allow the reader to delete the P-Site header if desired so that the private IP address of System Manager is not revealed on the public side of the SBC.

To remove the P-Site header, an additional header rule is added to the existing header manipulations described in **Section 5.3.11** of reference [JF-VZIPCC]. This new **header-rule** to delete the P-Site header is shown below.

```
header-rule
                name              delPsite
                header-name       P-Site
                action            delete
                comparison-type   pattern-rule
                match-value
                msg-type          request
                new-value
                methods
```

With this header rule configured and activated, any P-Site header inserted by Session Manager will not be sent to Verizon.

6.5.2 P-Location Header Removal

For an outbound call from a Communication Manager user to the PSTN, Session Manager Release 6.1 inserts a P-Location header into the INVITE message sent to the SBC. For an inbound call from the PSTN to a Communication Manager user, Session Manager Release 6.1 inserts a P-Location header into the 200 OK message sent to the SBC when the call is answered. The presence of the P-Location header does not present a problem for calls to or from the Verizon IP Trunk Service. However, since there may be no value in sending this header to Verizon, and since tracing tools may flag this header as an unknown header, this section shows a sample SBC configuration to strip the P-Location header in the SBC so that Verizon does not receive it.

To remove the P-Site header, an additional header rule is added to the existing header manipulations described in **Section 5.3.11** of reference [JF-VZIPCC]. This new **header-rule** to delete the P-Location header is shown below.

```
header-rule
                name              delPLocation
                header-name       P-Location
                action            delete
```

```
comparison-type        pattern-rule
match-value
msg-type               any
new-value
methods
```

With this header rule configured and activated, the P-Location header inserted by Session Manager Release 6.1 will not be sent to Verizon.

6.5.3 REFER Header

In **Section 5.3.11.5** of reference [JF-VZIPCC], a manipulation is defined for the REFER method to change the host portion of the Refer-To header to "loc1.interoplab3.21sip.com". This host domain is not relevant for the Verizon service configuration used with these Application Notes. Therefore, the "new-value" parameter in **Section 5.3.11.5** of reference [JF-VZIPCC] can be changed from "loc1.interoplab3.21sip.com" to "$REMOTE_IP". With this changed header rule configured and activated, the host portion of a Refer-To header sent to Verizon will be "172.30.205.55", the Verizon IPCC address.

6.6. Access Control

In **Section 5.3.12.1** of reference [JF-VZIPCC], an **access-control** configuration is applied to the OUTSIDE realm to permit SIP UDP access from the source-address corresponding to the Verizon IPCC service node used in reference [JF-VZIPCC]. Since the Verizon IPCC service used for these Application Notes used different IP parameters, a different access-control configuration is required. The following shows the new access-control permitting SIP traffic from the Verizon IPCC service.

```
access-control
        realm-id                OUTSIDE
        description             Verizon-IPCC
        source-address          172.30.205.55:5072
        destination-address      0.0.0.0
        application-protocol    SIP
        transport-protocol      UDP
        access                  permit
```

7. 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 Solution & Interoperability Test 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.

7.1. Service Access Information

The following service access information (FQDN, IP addressing, ports, toll free numbers) was provided by Verizon for the sample configuration.

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

Toll Free Numbers
866-850-2380
866-850-8170
866-851-0107
866-851-8119
866-616-4250
866-616-4254

8. General Test Approach and Test Results

The test approach was manual testing of inbound and referred calls using the Verizon IPCC Services on a production Verizon PIP access circuit, as shown in **Figure 1**.

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

- Inbound Verizon toll-free calls to Avaya Aura® Communication Manager telephones and VDNs/Vectors
- Inbound private toll-free calls (e.g., PSTN caller uses *67 followed by the toll-free number)
- Inbound Verizon toll-free calls redirected using Avaya Aura® Communication Manager SIP NCR (via SIP REFER/Refer-To) to PSTN alternate destinations
- Inbound Verizon IP toll-free calls redirected using Avaya Aura® Communication Manager SIP NCR with UUI (via SIP REFER/Refer-To with UUI) to a SIP-connected destination
- Inbound toll-free voice calls can use G.711MU or G.729A codecs.
- Inbound toll-free voice calls can transmit DTMF tones using RFC 2833
- Inbound toll-free voice calls to Avaya Aura® Communication Manager stations can be covered to Avaya Modular Messaging.

Testing was successful, except as noted in the limitations described in **Section 1.3.**

Examples of representative verified call scenarios are detailed in **Section 9**.

9. Verification Steps

This section provides example verifications of the sample configuration illustrated in these Application Notes.

9.1. Communication Manager and Wireshark Verifications

This section illustrates verifications using Communication Manager and Wireshark to illustrate key SIP messaging.

9.1.1 Sample Incoming Call from PSTN via Verizon SIP Trunk

Incoming toll-free calls arrive from Verizon at the Acme Packet Net-Net SBC, which sends the call to Session Manager. Session Manager sends the call to Communication Manager via the configured entity link using port 5067. On Communication Manager, the incoming call arrives via signaling group 67 and trunk group 67.

The following Communication Manager ***list trace*** trace output shows a call incoming on trunk group 67. The PSTN telephone dialed 866-851-0107. Session Manager can map the number received from Verizon to the extension of a Communication Manager telephone (x34008), or the incoming call handling table for trunk group 67 can do the same. In the trace below, Session Manager had already mapped the Verizon number to the Communication Manager extension. Extension "34008" is an IP Telephone with IP Address 192.168.49.47 in Region 54. Initially, the media resources on G650 Media Gateway (10.1.2.236) were used, but once the call is answered, the final RTP media path is "ip-direct" from the IP Telephone (192.168.49.47) to the "inside" of an Acme Packet Net-Net SBC (65.206.67.1).

```
list trace tac 167                                                  Page   1

                                LIST TRACE

time            data

08:51:59 SIP<INVITE sip:34008@avaya.com:5060;transport=tcp  SIP/
08:51:59 SIP<2.0
08:51:59     active trunk-group 67 member 1  cid 0xa57
08:51:59 SIP>SIP/2.0 180 Ringing
08:51:59     dial 34008
08:51:59     ring station    34008 cid 0xa57
08:51:59     G729A ss:off ps:20
             rgn:54 [192.168.49.47]:2570
             rgn:1 [10.1.2.236]:7520
08:51:59     G729 ss:off ps:20
             rgn:54 [65.206.67.1]:49760
             rgn:1 [10.1.2.236]:7512
08:51:59     xoip options: fax:off modem:off tty:US  uid:0x5011d
             xoip ip: [10.1.2.236]:7512
08:52:05 SIP>SIP/2.0 200 OK
                                                                    Page   2

time            data
08:52:05     active station    34008 cid 0xa57
08:52:06 SIP<ACK sip:34008@10.1.2.233:5067;transport=tcp  SIP/2.
08:52:06 SIP<0
08:52:06 SIP>INVITE sip:+17324221876@65.206.67.1:5060;transport=
08:52:06 SIP>tcp SIP/2.0
08:52:06 SIP<SIP/2.0 100 Trying
08:52:06 SIP<SIP/2.0 200 OK
08:52:06 SIP>ACK sip:+17324221876@65.206.67.1:5060;transport=tcp
08:52:06 SIP> SIP/2.0
08:52:06     G729A ss:off ps:20
             rgn:54 [65.206.67.1]:49760
             rgn:54 [192.168.49.47]:2570
08:52:06     G729 ss:off ps:20
             rgn:54 [192.168.49.47]:2570
             rgn:54 [65.206.67.1]:49760
```

The following screen shows Page 2 of the output of the ***status trunk*** command pertaining to this same call. Note that the signaling uses port 5067 between Communication Manager and Session Manager. Note the media is "ip-direct" from the IP Telephone (192.168.49.47) to the inside IP Address of Acme1 (65.206.67.1) using G.729.

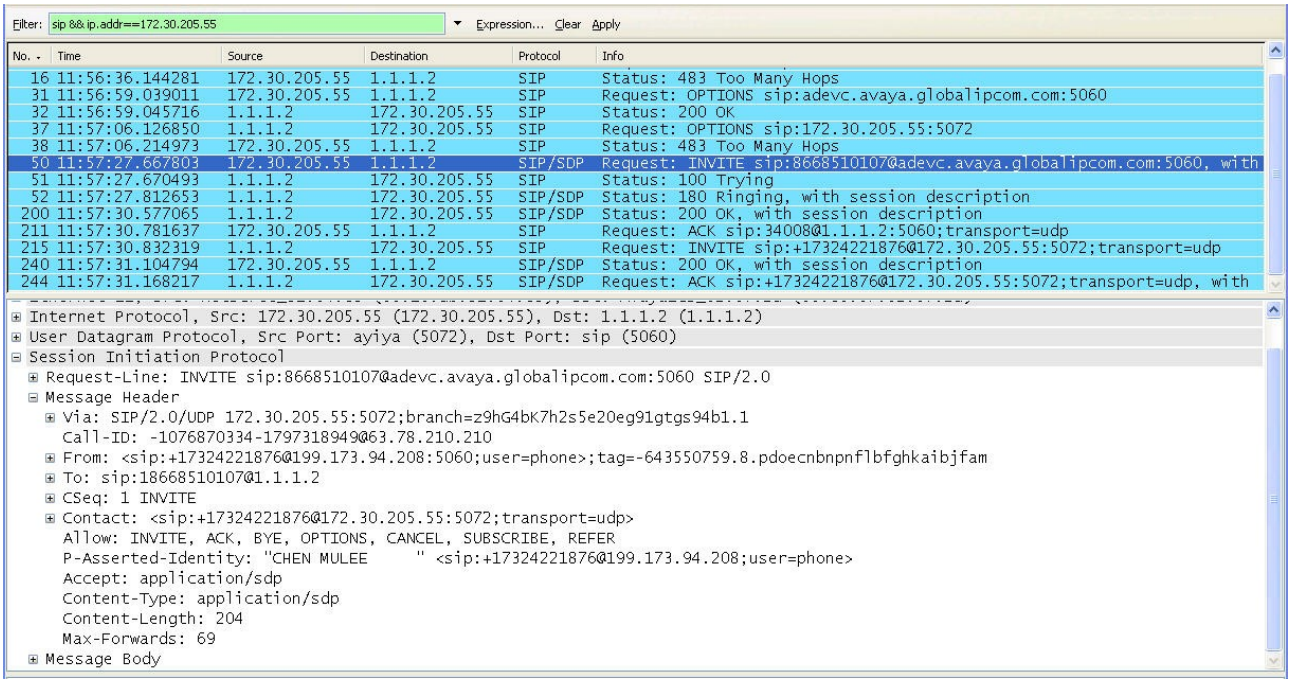
```
status trunk 67/1                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0217
  Signaling   IP Address                              Port
   Near-end:  10.1.2.233                            : 5067
    Far-end:  10.1.2.210                            : 5067
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.729
   Audio      IP Address                              Port
   Near-end:  192.168.49.47                         : 2570
    Far-end:  65.206.67.1                           : 49762
```

The following screen shows Page 3 of the output of the ***status trunk*** command pertaining to this same call. Here it can be observed that G.729a is used.

```
status trunk 67/1                                              Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T00285
T00285:TX:65.206.67.1:49762/g729/20ms
S00533:RX:192.168.49.47:2570/g729a/20ms

dst port: S00533
```

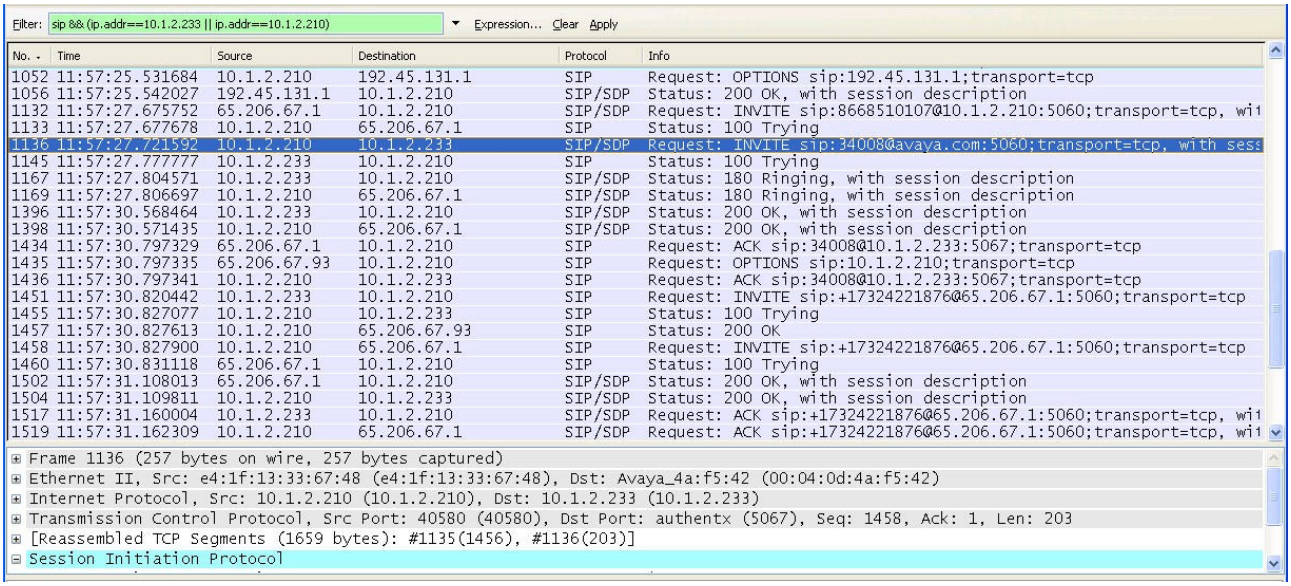
The following portion of a filtered Wireshark trace (tracing only SIP messages on the public interface on the "outside" of the SBC) shows an incoming PSTN call. In frame 50, Verizon sends the INVITE to the Acme Packet SBC (1.1.1.2). Frame 50 is selected and expanded to illustrate the contents of the various headers sent by Verizon. The trace shows that the SIP message uses UDP with source port 5072 and destination port 5060. The subsequent call processing of this call will be illustrated in the context of the "inside" trace analysis (private side of SBC) that follows.

Note that this trace also shows exchanges of SIP OPTIONS messages at the top in frames 31-38. In frame 31 Verizon sends OPTIONS, and the Acme Packet Net-Net SBC responds with 200 OK in frame 32. In frame 37, the Acme Packet Net-Net SBC sends OPTIONS, and Verizon responds with 483 Too Many Hops in frame 38. The 483 response from Verizon is both expected (since the Acme has been configured to set Max-Forwards to 0 in OPTIONS) and sufficient to keep the Acme session agent in-service.



The following portion of a filtered Wireshark trace (tracing SIP messages on the private inside interface of the SBC only) shows the same incoming PSTN call. In frame 1132, the inside interface of the Acme Packet SBC (65.206.67.1) sends an INVITE to Session Manager (10.1.2.210). In highlighted frame 1136, Session Manager sends the INVITE to Communication Manager (10.1.2.233). Observe that Session Manager has already adapted the Verizon toll-free number to its corresponding Communication Manager extension (34008). In the lower portion of the screen, observe the use of TCP and destination port 5067 on Communication Manager.

In frame 1167, Communication Manager sends a 180 Ringing with SDP. Note that enhancements in Communication Manager Release 6 allow a 183 with SDP to be configured and sent, as desired by Verizon. In frame 1396, Communication Manager sends 200 OK with SDP when the user answers the call. In frame 1451, Communication Manager sends the INVITE to begin the process of shuffling the media paths to "ip-direct" (note that this "shuffling" INVITE was without SDP), which concludes with the ACK in frame 1519.

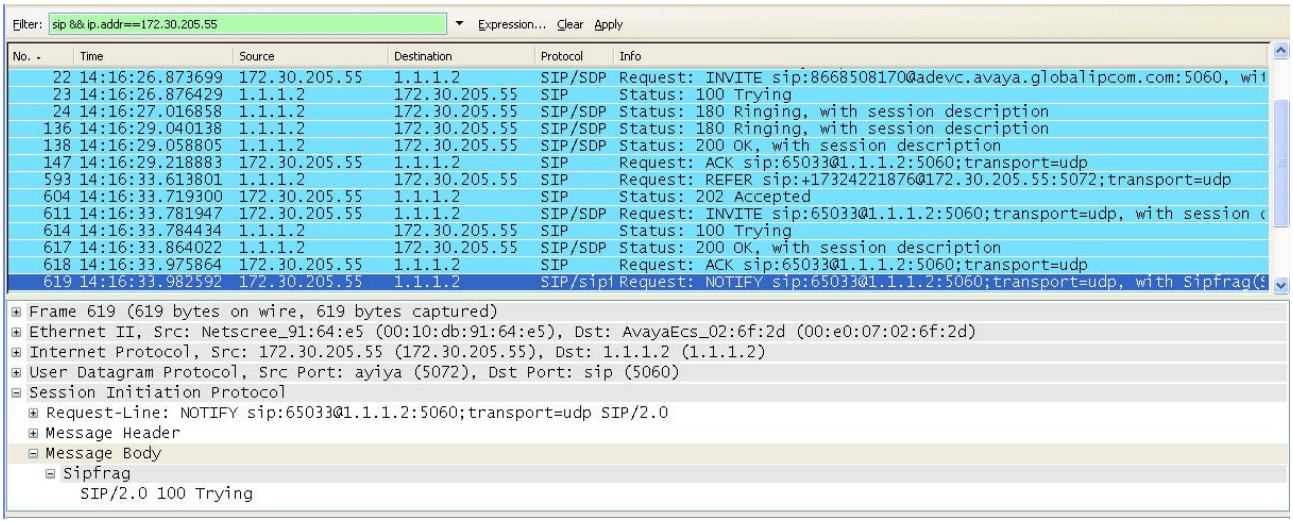


9.1.2 Sample Inbound Call Referred via Call Vector to PSTN Destination

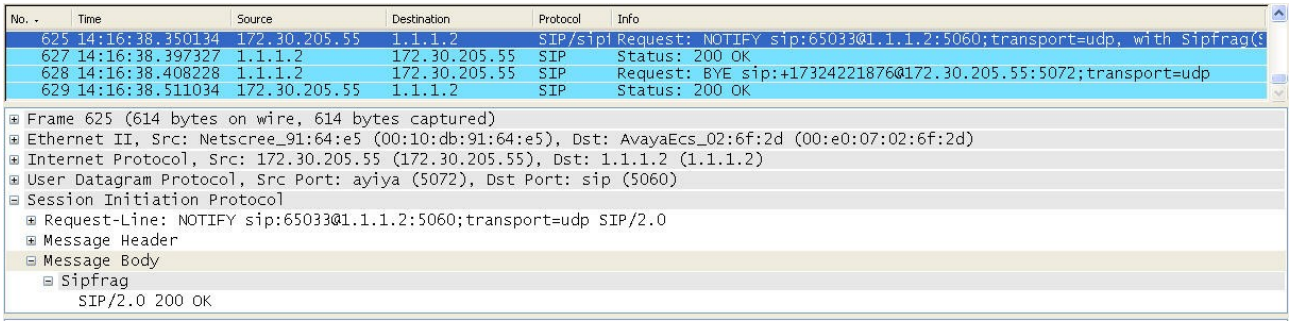
The following edited and annotated Communication Manager ***list trace*** output shows a call incoming on trunk group 67. The call was routed to the Communication Manager VDN (Vector Directory Number) "65033" associated with call vector 33 (see **Section 4.9.1**). The vector answers the call, plays an announcement to the caller, and then uses a "route-to" step to cause a REFER message to be sent with a Refer-To header containing the number configured in the vector "route-to" step). The PSTN telephone dialed 866-850-8170. In the trace below, Session Manager had already mapped the Verizon number to the Communication Manager VDN extension. The annotations in the edited trace highlight the call progression. At the conclusion, the PSTN caller that dialed the Verizon toll-free number is connected to the Referred-to PSTN destination, and no trunks (i.e., from trunk group 67 handling the call) are in use.

```
list trace tac 167                                                    Page   1
                              LIST TRACE
time           data

12:00:54 SIP<INVITE sip:65033@avaya.com:5060;transport=tcp  SIP/2.0
12:00:54     active trunk-group 67 member 1  cid 0xa5f
12:00:54   33  1 vdn e65033 bsr appl   0 strategy 1st-found override n
12:00:54   33  1 wait 2 secs hearing ringback
12:00:54 SIP>SIP/2.0 180 Ringing
12:00:54     dial 65033
12:00:54     ring vector 33    cid 0xa5f
12:00:54     G729 ss:off ps:20
             rgn:54 [65.206.67.1]:49770
             rgn:1 [10.1.2.236]:8080
12:00:56   33  2 announcement 67008
                                                                      Page   2
12:00:56 SIP>SIP/2.0 180 Ringing
12:00:56   33  2      announcement: board 01A13 ann ext: 67008
/** Call is answered by Communication Manager to play announcement **/
12:00:56 SIP>SIP/2.0 200 OK
12:00:56     active announcement    67008 cid 0xa5f
12:00:56     hear annc board 01A13 ext 67008 cid 0xa5f
12:00:56 SIP<ACK sip:65033@10.1.2.233:5067;transport=tcp  SIP/2.0
12:01:00     idle announcement     cid 0xa5f
/** Announcement completes and route-to step in vector follows **/
12:01:00   33  3 route-to number ~r +17328953304 cov n if unconditionally
12:01:00 SIP>REFER sip:+17324221876@65.206.67.1:5060;transport=tcp SIP/2.0
/** Verizon sends 202 Accepted to REFER **/
12:01:00 SIP<SIP/2.0 202 Accepted
/** Verizon sends INVITE to hold the call with the vector **/
/** while redirecting the call to the Refer-To Number     **/
12:01:01 SIP<INVITE sip:65033@10.1.2.233:5067;transport=tcp  SIP/2.0
12:01:01 SIP>SIP/2.0 100 Trying
12:01:01 SIP>SIP/2.0 200 OK
                                                                      Page   3
12:01:01 SIP<ACK sip:65033@10.1.2.233:5067;transport=tcp  SIP/2.0
/** Verizon sends NOTIFY about call being tried and answered by Refer-To Number **/
12:01:01 SIP<NOTIFY sip:65033@10.1.2.233:5067;transport=tcp  SIP2.0
12:01:01 SIP>SIP/2.0 200 OK
12:01:05 SIP<NOTIFY sip:65033@10.1.2.233:5067;transport=tcp  SIP/2.0
12:01:05 SIP>SIP/2.0 200 OK
12:01:05   33  3 LEAVING VECTOR PROCESSING cid 2655
/** Communication Manager terminates the original call from Verizon **/
12:01:05 SIP>BYE sip:+17324221876@65.206.67.1:5060;transport=tcp  SIP/2.0
```

The following portion of a filtered Wireshark trace (tracing SIP messages on the public outside interface of the SBC only) shows the same incoming PSTN call. The call vector answers the call (frame 138), plays a short announcement to the caller (note elapsed time between frames 138 and 593 when RTP carrying the announcement is flowing), and then uses a "route-to" step to cause a REFER message to be sent (frame 593) with a Refer-To header containing the number configured in the "route-to" step. In frame 604, Verizon sends a 202 Accepted message for the REFER. In highlighted frame 619, Verizon sends a NOTIFY message; the lower area of the screen illustrates the NOTIFY is for a "100 Trying".



Verizon routes the call to the number specified in the Route-To header (i.e., the number in the route-to step in the vector). Scrolling down in this same trace, when the PSTN destination answers, Verizon sends the NOTIFY message in highlighted frame 625; the lower area of the screen illustrates the NOTIFY is for a "200 OK". Observe the BYE messages clear the call to the enterprise site. Although the PSTN caller who dialed the IP Toll Free number is talking to the Referred-to destination, no trunks are in use to the enterprise site that received the call.



9.1.3 Sample Inbound Call Referred with UUI to Alternate SIP Destination

The following Communication Manager ***list trace vector*** trace output shows a different sample incoming Verizon toll-free call to 866-850-2380. The call was routed to the Communication Manager VDN 65034 associated with the call vector 34 (see **Section 4.9.2**). As in previous illustrations, this vector will answer the call, play an announcement to the caller, and then use a "route-to" step to cause a REFER message to be sent to Verizon. In this case, the Refer-To number will cause Verizon to route the call to another SIP-connected destination. In the sample configuration, where only one site is available, this was tested by including a different IP Toll Free number (866-851-0107) assigned to the same site in the Refer-To header. The vector also sets UUI data that will be included in the Refer-To header. When Verizon routes the call to the "alternate" destination, the INVITE message will contain a User-To-User header containing the UUI data sent in the Refer-To header. In practice, this would allow a Communication Manager at one site to pass call or customer-related data to another site via the Verizon network.

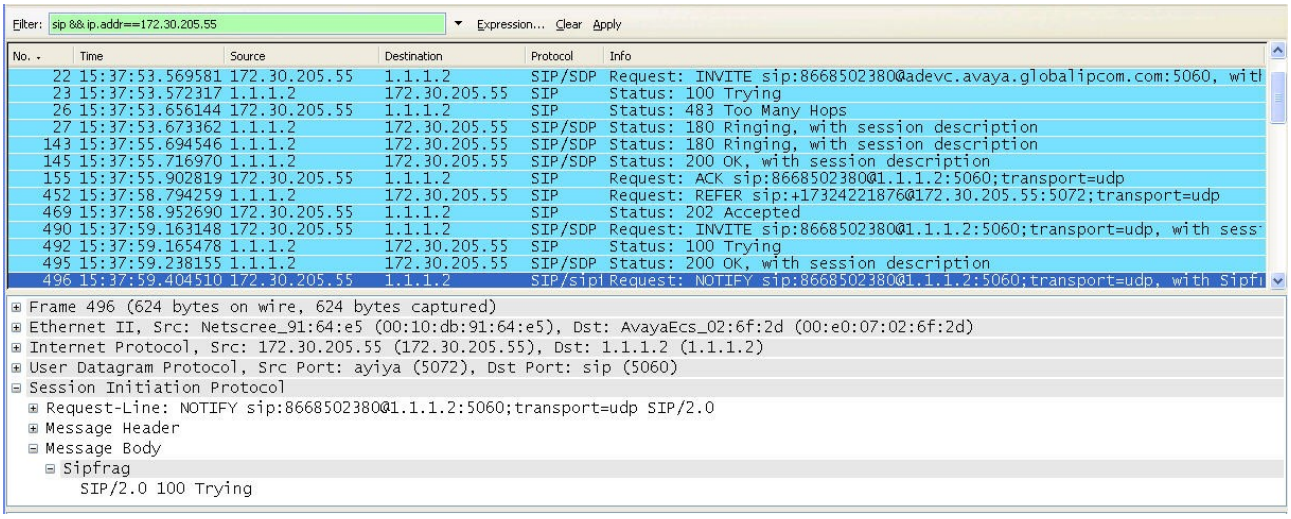
```
list trace tac 167                                                    Page   1
                              LIST TRACE
time           data

13:22:20 SIP<INVITE sip:65034@avaya.com:5060;transport=tcp  SIP/2.0
13:22:20     active trunk-group 67 member 1  cid 0xa63
13:22:20    0  0 ENTERING TRACE cid 2659
13:22:20   34  1 vdn e65034 bsr appl   0 strategy 1st-found override n
13:22:20   34  1 set A = none CATR 1234567890123456
13:22:20   34  1      operand    = []
13:22:20   34  1      operand    = [1234567890123456]
13:22:20   34  1      ========= CATR =========
13:22:20   34  1      variable A = [1234567890123456] asaiuui local
13:22:20   34  1      asaiuui chg from [] to [1234567890123456]
13:22:20   34  2 set B = none CATR 7890123456789012
13:22:20   34  2      operand    = []
13:22:20   34  2      operand    = [7890123456789012]
13:22:20   34  2      ========= CATR =========
13:22:20   34  2      variable B = [7890123456789012] asaiuui local
13:22:20   34  2      asaiuui chg from [] to [7890123456789012]
13:22:20   34  3 wait 2 secs hearing ringback
13:22:20 SIP>SIP/2.0 180 Ringing
13:22:20     dial 65034
13:22:20     ring vector 34    cid 0xa63
13:22:22   34  4 announcement 67030
13:22:22 SIP>SIP/2.0 180 Ringing
13:22:22   34  4      announcement: board 01A13 ann ext: 67030
13:22:22 SIP>SIP/2.0 200 OK
13:22:22     active announcement    67030 cid 0xa63
13:22:22     hear annc board 01A13 ext 67030 cid 0xa63
13:22:22 SIP<ACK sip:8668502380@10.1.2.233:5067;transport=tcp  S
13:22:22 SIP<IP/2.0
13:22:25     idle announcement     cid 0xa63
13:22:25   34  5 route-to number ~r +18668510107 cov n if unconditionally
13:22:25 SIP>REFER sip:+17324221876@65.206.67.1:5060;transport=t
13:22:25 SIP>cp SIP/2.0
13:22:25 SIP<SIP/2.0 202 Accepted
```

The following beginning of a filtered Wireshark trace (tracing SIP messages on the public outside interface of the SBC only) shows the same incoming PSTN call to the Verizon toll-free number 866-850-2380. At the start, the trace looks very similar to the one shown in the previous section.

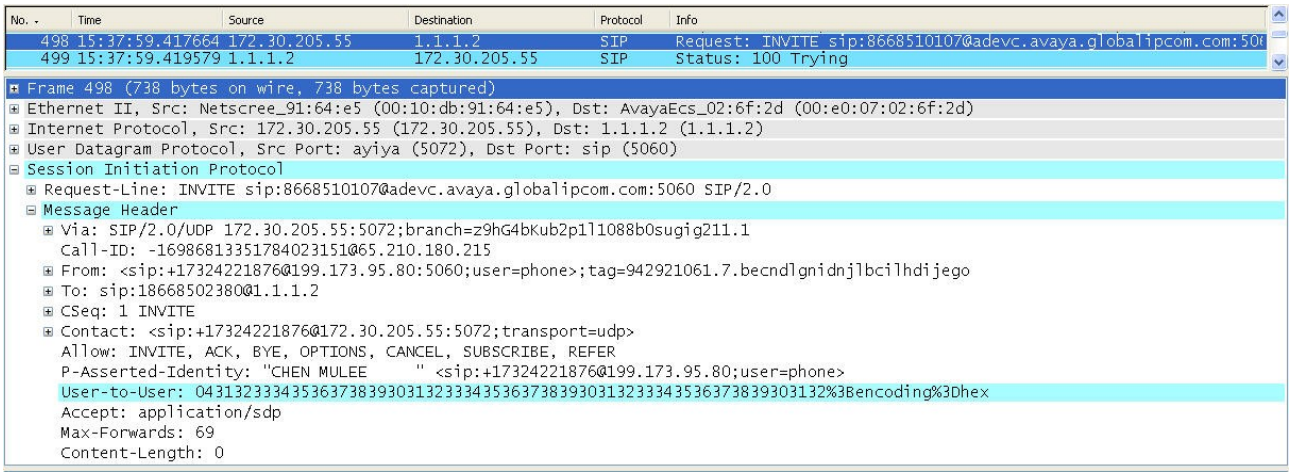
The user dials the same number 866-850-2380 and Session Manager has adapted the number to Communication Manager VDN 65034 associated with call vector 34. The call vector answers the call (frame 145), plays a short announcement to the caller (note elapsed time between frames 145 and 452), and then uses a "route-to" step to cause a REFER message to be sent (frame 452). The REFER includes a Refer-To header containing the number configured in the "route-to" step, which in this case contains another IP Toll Free number (866-851-0107). The REFER also contains the UUI data set in vector 34. Although not expanded in the wireshark trace below, the format of the Refer-To header will be like the following, where the host portion "Verizon-IPCC" can be manipulated by the Acme Packet Net-Net SBC as needed:

```
Refer-To: <sip:+18668510107@Verizon-IPCC?
User-to-User=043132333435363738393031323334353637383930313233343536373839303132
%3Bencoding%3Dhex>
```

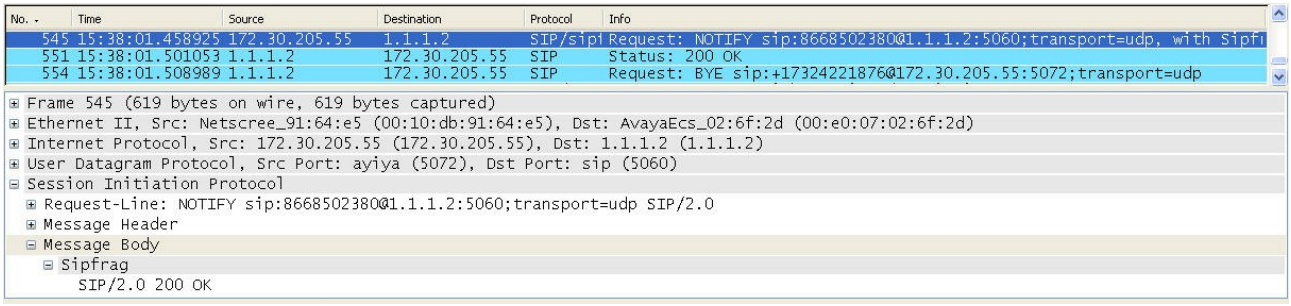
In frame 469, Verizon sends a 202 Accepted message for the REFER, and in the highlighted frame 496, Verizon sends a NOTIFY with "100 Trying" as illustrated previously.



Verizon then routes the call to the number specified in the Route-To header which in this case is another Verizon toll-free number assigned to this same site (i.e., in production, this would typically be used to route to an alternate site). Scrolling down in this same trace, frame 498 is selected below to show the INVITE from Verizon that was stimulated by the REFER/Refer-To. From the highlighted message summary, it can be observed that the R-URI contains 866-851-0107, the toll-free number used in the Refer-To step in the vector. In the lower portion of the screen, where details of the contents of the INVITE are shown, note that the PAI contains the original caller ID of the true PSTN caller (732-422-1876), and the User-to-User header contains the contents of the UUI previously sent by the Avaya CPE to Verizon in the Refer-To header in the REFER message. The reader may also observe that this INVITE from Verizon does not contain SDP.



Once the referred-to destination has answered, Verizon sends the NOTIFY containing the "200 OK" result in frame 545, which is highlighted and expanded. Communication Manager then clears the original call that stimulated the REFER with the BYE in frame 554. The PSTN caller and the answering party of the referred-to call are now talking. If the answering party of the referred-to call is a Communication Manager user who has a "uui-info" button, and the answering user's Class of Restriction (COR) allows "Station Button Display of UUI IE data", the answering user can see the UUI data on the display phone by pressing the "uui-info" button. In a multi-site contact center setting, a contact center agent answering a call at site B could see the UUI sent in the REFER from site A.

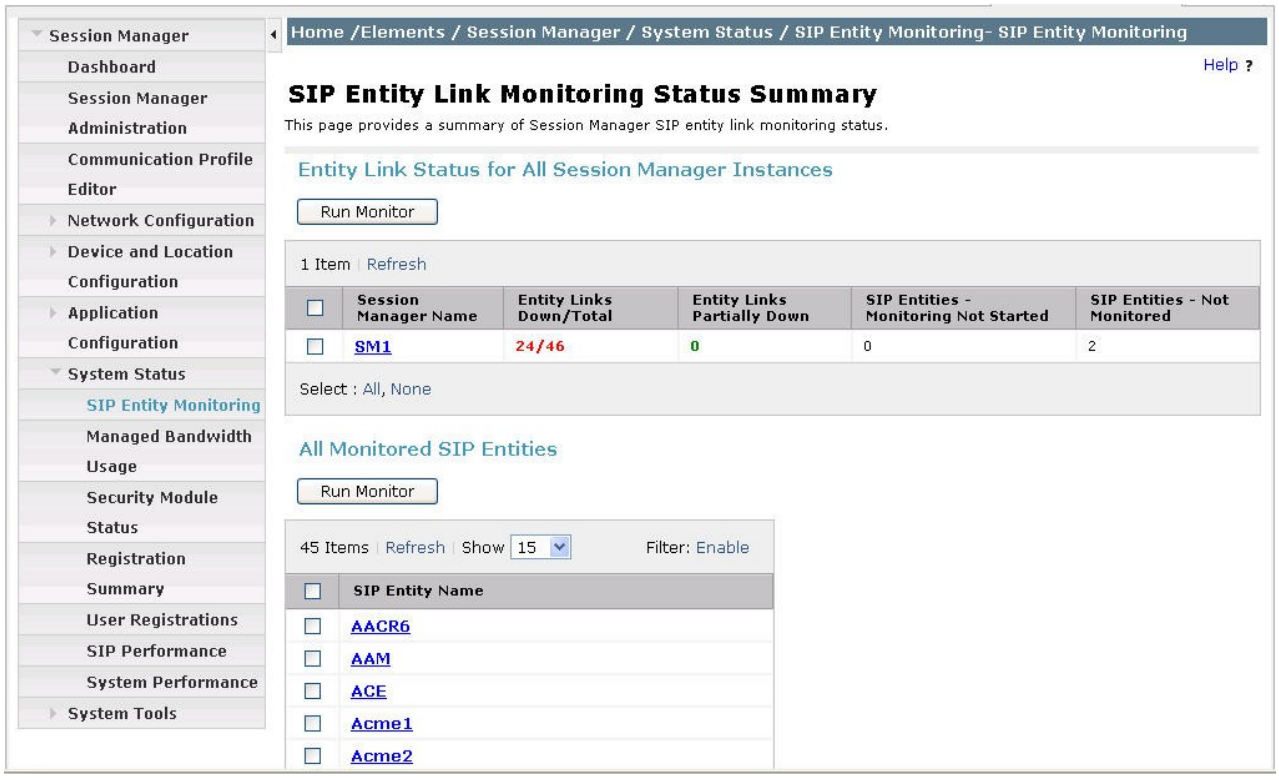


9.2. Avaya Aura® System Manager and Session Manager Verifications

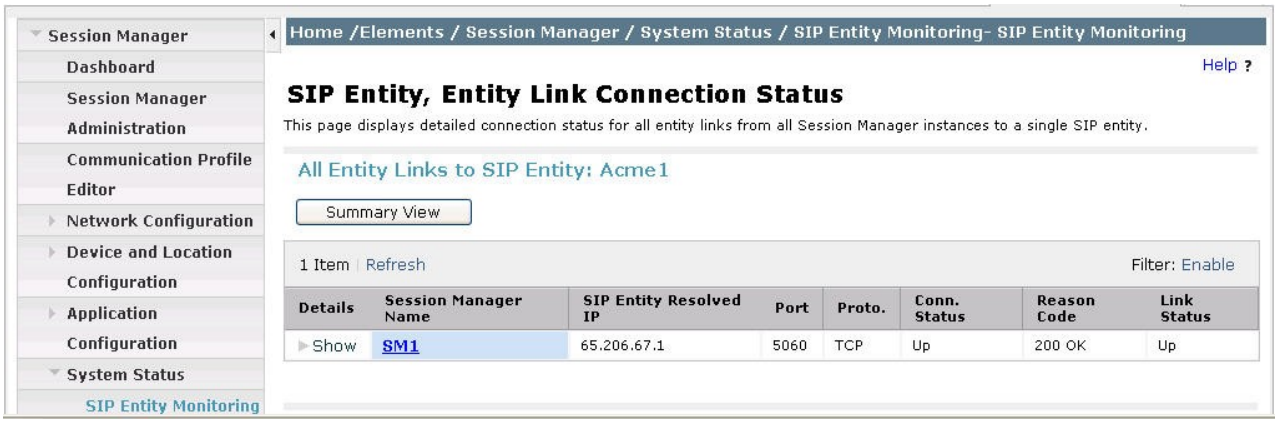
This section contains verification steps that may be performed using System Manager for Session Manager.

9.2.1 Verify SIP Entity Link Status

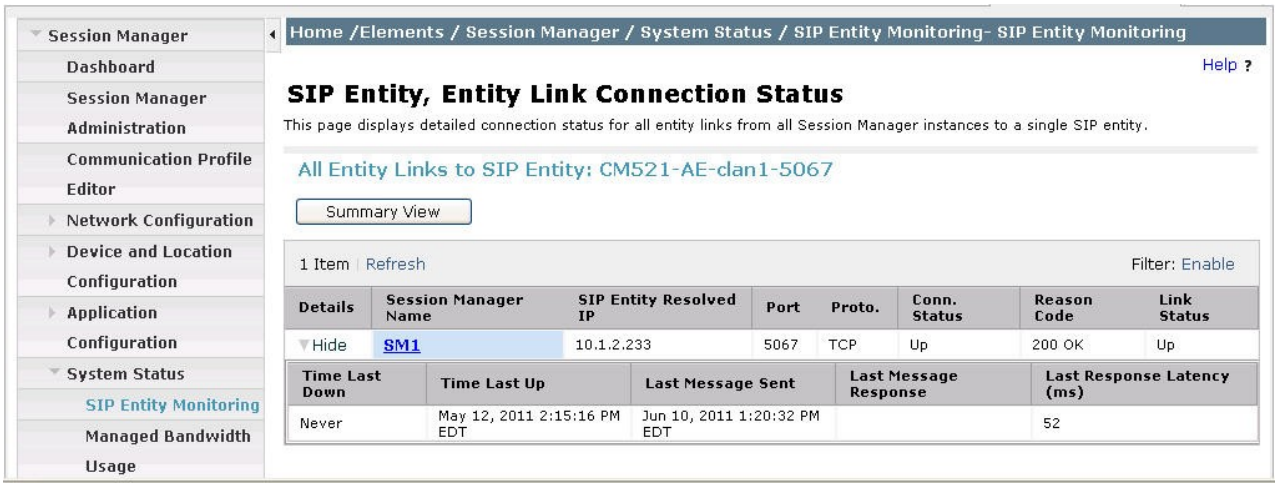
Log in to System Manager. Navigate to **Home → Elements → Session Manager → System Status → SIP Entity Monitoring**, as shown below.



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.

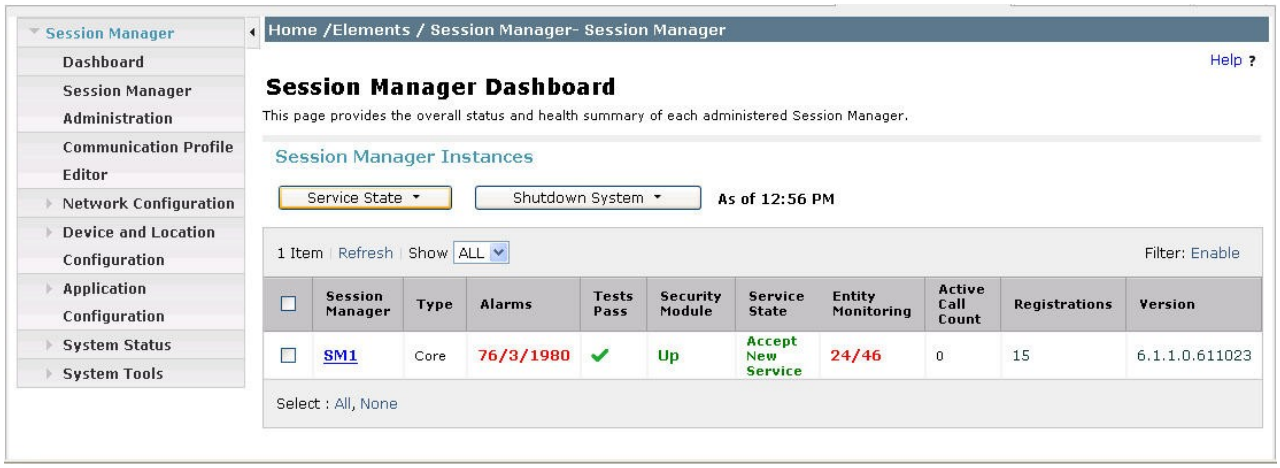


Return to the list of monitored entities, and select another entity of interest, such as "CM521-AE-clan1-5067". Under normal operating conditions, the **Link Status** should be "Up" as shown in the example screen below for "CM521-AE-clan1-5067". In this case, "Show" under **Details** was selected to view additional information. Note the use of port 5067.



9.2.2 Verify System State

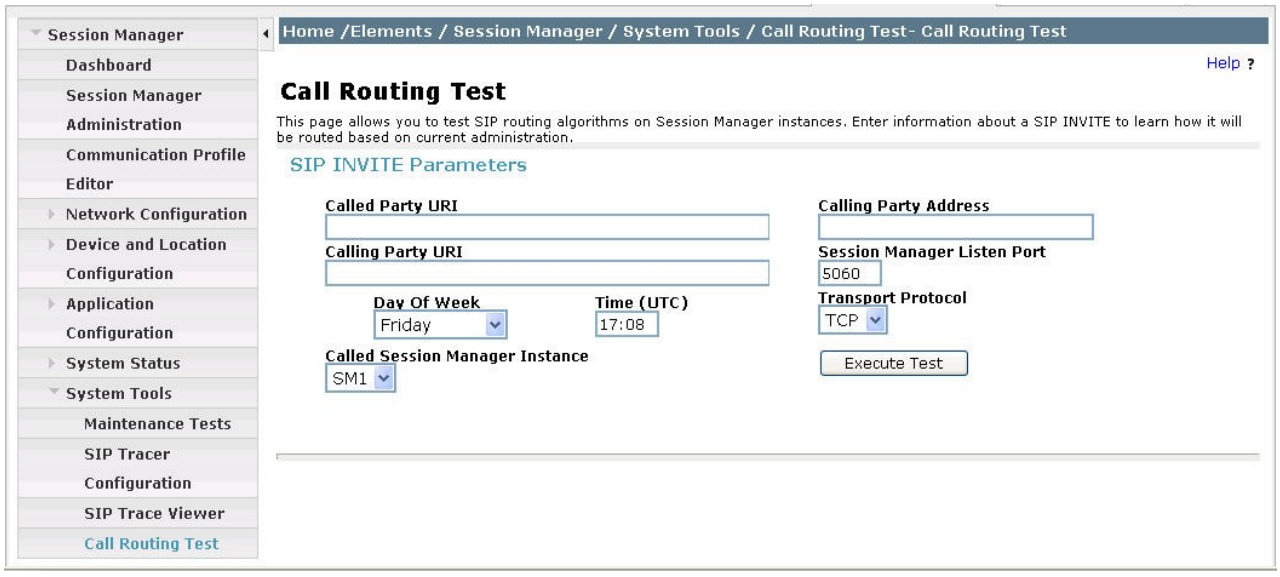
Navigate to **Home → Elements → Session Manager**, as shown below.



Verify that a green check mark is placed under **Tests Pass** and the **Service State** is "Accept New Service." The **Version** can also be observed.

9.2.3 Call Routing Test

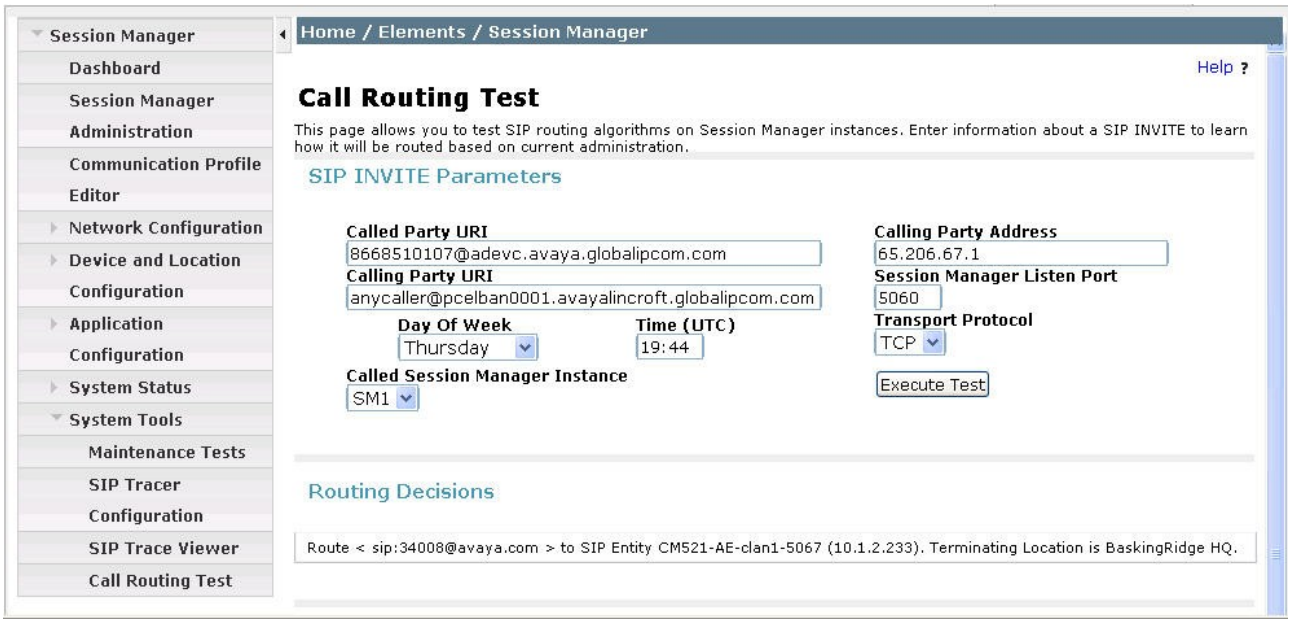
The **Call Routing Test** verifies the routing for a particular source and destination. To run the routing test, navigate to **Home → Elements → Session Manager → System Tools → Call Routing Test**, as shown below.



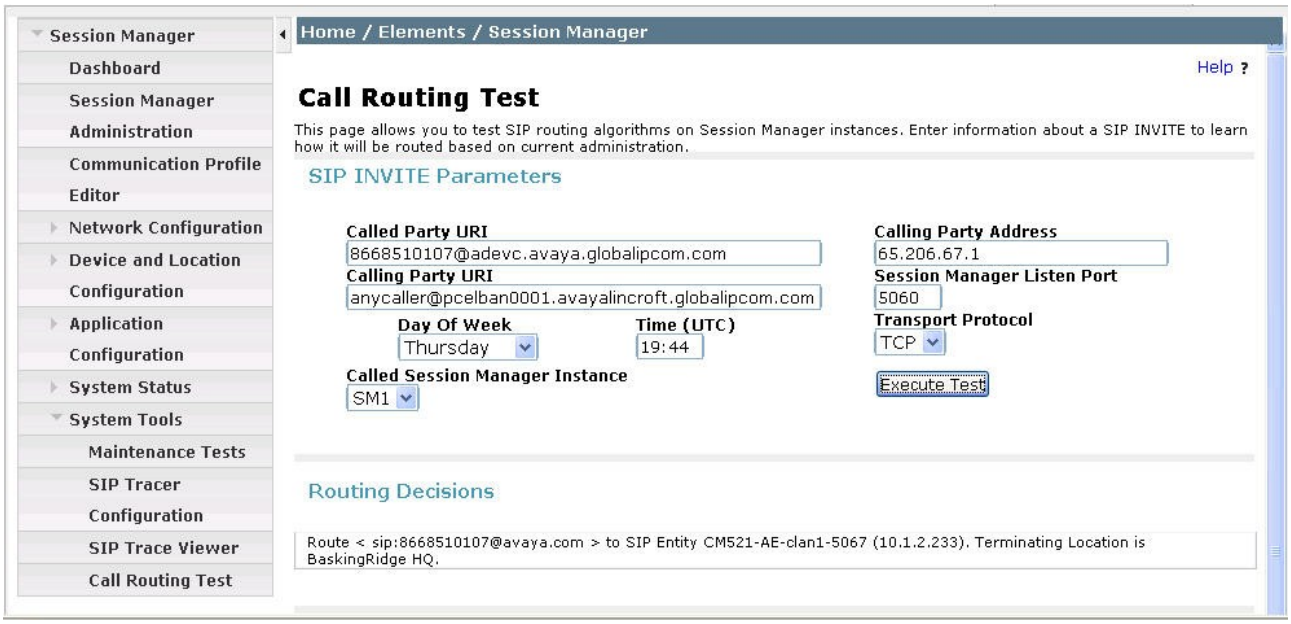
Populate the fields for the call parameters of interest and click **Execute Test**.

For example, the following shows a call routing test for an inbound toll-free call from the PSTN to the enterprise via Acme1 (65.206.67.1). Under **Routing Decisions**, observe that the call will route

to Communication Manager using the SIP entity named "CM521-AE-clan1-5067". The domain in the Request-URI is converted to "avaya.com", and the digits are manipulated such that the Verizon toll-free number (i.e., 866-851-0107) is converted to a Communication Manager extension (i.e., 34008) by the adapter assigned to the Communication Manager entity. Scroll down to inspect the details of the **Routing Decision Process** if desired (not shown).



After a configuration change that removed the Verizon toll-free number to Communication Manager extension digit manipulation from the Session Manager adapter, the following example shows a call routing test for an inbound call from the PSTN to the enterprise via Acme1. Under **Routing Decisions**, observe that the call will still route to Communication Manager using the SIP entity named "CM521-AE-clan1-5067", but the Request-URI now contains the full 10 digit toll-free number. With configuration like this, the incoming call handling table of the Communication Manager trunk group receiving the incoming call (i.e., trunk group 67 in the sample configuration) would need to map the Verizon toll-free number to a Communication Manager extension.



10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager Release 5.2.1, Avaya Aura® Session Manager Release 6.1, and an Acme Packet Net-Net Session Border Controller can be configured to interoperate successfully with Verizon Business IP Contact Center Services suite. This solution provides users of Avaya Aura® Communication Manager the ability to support inbound toll free calls over a Verizon Business VoIP Inbound SIP trunk service connection. In addition, these Application Notes further demonstrate that the Avaya Aura® Communication Manager implementation of SIP Network Call Redirection (SIP-NCR) can work in conjunction with Verizon's Business IP Contact Center services implementation of SIP-NCR to support call redirection over SIP trunks inclusive of passing User-User Information (UUI).

Please note that the sample configurations shown in these Application Notes are intended to provide configuration guidance to supplement other Avaya product documentation.

Avaya Aura® SIP Solution using Avaya Aura® Communication Manager Release 5.2.1 has not been independently certified by Verizon labs. These Application Notes can be used to facilitate customer engagements via the Verizon field trial process, pending Verizon labs independent certification.

11. Additional References

11.1. Avaya

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

[1] *Administering Avaya Aura® Communication Manager*, Release 5.2, May 2009, Document Number 03-300509, available at https://support.avaya.com/css/P8/documents/100059292

[2] *Administering Avaya Aura® Session Manager,* Release 6.1, November 2010, Document Number 03-603324, available at https://support.avaya.com/css/P8/documents/100121656

[3] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Number 03-603473, available at https://support.avaya.com/css/P8/documents/100120934

[4] *Maintaining and Troubleshooting Avaya Aura® Session Manager,* Doc ID 03-603325, Release 6.1, March 2011, available at https://support.avaya.com/css/P8/documents/100120937

[5] *Administering Avaya Aura® System Manager*, Release 6.1, November 2010, available at https://support.avaya.com/css/P8/documents/100120857

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

Application Notes Reference [JF-VZIPCC] documents Verizon IPCC Services with previous version of Avaya Aura® Communication Manager and Avaya Aura® 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-VZIPCC] documents Verizon IPCC Services with newer versions of Avaya Aura® Communication Manager.
[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 Center (IPCC) Services Suite – Issue 1.1 (including a declaration of support for Communication Manager Release 6.0.1 and Session Manager Release 6.1)
https://devconnect.avaya.com/public/download/dyn/SM6CM6_VzBIPCC.pdf

11.2. Verizon Business

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

[6] *Verizon Business IPCC Interoperability Test Plan, Revision 1.7, Aug 27, 2009*

[7] *Verizon Business IP Contact Center Trunk Interface Network Interface Specification, Document Version 2.2.1.9, Aug 25, 2009*

[8] *Additional information regarding Verizon Business IPCC Services suite offer can be found at http://www.verizonbusiness.com/products/contactcenter/ip/*