Avaya Solution & Interoperability Test Lab

Application Notes for VCSR (doing business as OpenMethods) OpenIVR with Avaya Aura® Communication Manager 6.2 and Avaya Aura® Session Manager 6.3 via a SIP Trunk with Shared User-to-User Treatment – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring VCSR (doing business as OpenMethods) OpenIVR with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. OpenIVR is a SIP based system that interfaces with Avaya Aura® Session Manager and utilizes a VXML Interpreter to provide menu driven IVR features. The overall objective of the interoperability compliance testing was to verify the basic functions of OpenIVR with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk with Shared User-to-User Treatment configured.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring VCSR (doing business as OpenMethods) OpenIVR with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. OpenIVR is a SIP based system that interfaces with Avaya Aura® Session Manager and utilizes a VXML Interpreter to provide menu driven IVR features. The overall objective of the interoperability compliance testing was to verify the basic functions of OpenIVR with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk with Shared User-to-User Treatment configured. OpenIVR is a SIP based FreeSwitch solution.

OpenMethods provides customer contact management solutions and services in the business process outsourcing arena, primarily in the communications, financial services, healthcare, technology and transportation and leisure industries.

2. General Test Approach and Test Results

This section describes the general test approach used to verify the interoperability of OpenIVR with Communication Manager and Session Manager using a SIP trunks with Shared User-to-User Treatment configured.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability tests. The general test approach was to place calls from Avaya SIP, H.323, digital and analog phones into OpenIVR to exercise the IVR prompts using DTMF digits and verify UUI data was passed in the SIP messaging.

The following items were verified during the test:

- SIP OPTIONS monitoring of the health of the SIP trunks was verified. Both the Avaya enterprise equipment and OpenIVR can monitor health using SIP OPTIONS.
- Outbound calls from Avaya endpoints containing UUI data to OpenIVR.
- IVR interaction using keystrokes to select one of two options, "To attach data press 1" or "To leave the data the same, press 2".
- Blind transfer from the IVR to agents on Communication Manager via VDN, containing UUI data was passed from IVR to the agents.
- Transfer of IVR calls containing UUI data between agents.
- Verify codec negotiation of codec G.711 ULAW.
- Proper disconnect and release of resources when the Avaya endpoint hangs up an active call.
- Proper disconnect and release of resources when IVR option was not selected and the IVR times out and disconnects the call.

Serviceability testing included verifying proper operation/recovery from network outages, unavailable resources, and Session Manager, Communication Manager and OpenIVR restarts.

2.2. Test Results

All functionality and serviceability test cases were completed successfully. The following observations were made:

- OpenMethods does not support media shuffling; therefore corresponding parameters must be disabled on the Avaya signaling group or network region.
- Speech recognition to navigate OpenIVR was not an available test option, only DTMF digits.
- While transferring a call from an agent with an Avaya SIP phone containing UUI data, it sent the data in the SIP INVITE. For a call transferred from an agent with an Avaya H.323 phone the data is sent in the SIP REFER.

2.3. Support

Technical support for OpenIVR can be obtained by contacting OpenMethods at:

- **Phone:** 1-816-283-VXML (8965)
- **Web:** http://www.openmethods.com/support.php

3. Reference Configuration

Figure 1 illustrates the configuration used to test the interoperability of the OpenIVR solution with Communication Manager and Session Manager. Endpoints include Avaya 96xx and 96x1 Series SIP and H.323 one-X® Deskphones**,** Avaya 1416 Digital phone and an Avaya 6211 Analog phone. OpenIVR is located in the OpenMethods' lab and connected to the Avaya lab via a SIP trunk over the Internet. For confidentiality and privacy purposes, any actual public IP addresses used in the compliance testing have been replaced with private IP address in the Application Notes.

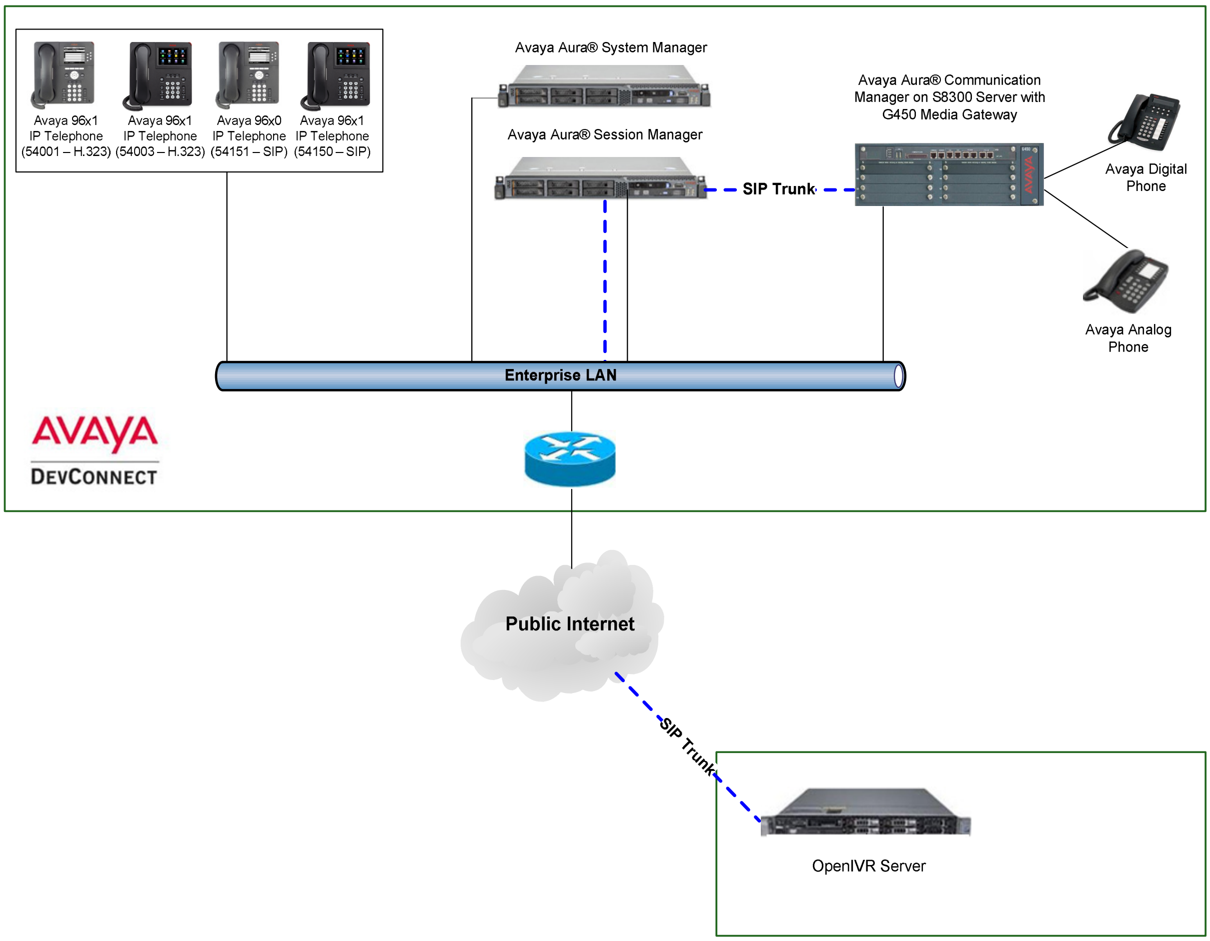


Figure 1: OpenIVR Test Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager on Avaya S8300D Server	R6.2 (R016x.02.0.823.0, Patch 20396)
G450 Media Gateway	32.26.0
Avaya Aura® System Manager on Avaya S8800 Server	6.3 (Build No. - 6.3.0.8.5682-6.3.8.860 Software Update Revision No: 6.3.1.9.1337)
Avaya Aura® Session Manager on Avaya S8800 Server	6.3 (6.3.1.0.631004)
Avaya one-X® Deskphones (SIP)	2.6.9 (96xx) 6.2.1 (96x1)
Avaya one-X® Deskphones (H.323)	3.2.0 (96xx) 6.2.3.13 (96x1)
Avaya 1416 Digital Phone	-
Avaya 6211 Analog Phone	-
OpenIVR Server running on Dell™ PowerEdge™ R610	2.2.0_v4

5. Configure Avaya Aura® Communication Manager

All the configuration changes in this section for Communication Manager are performed through the System Access Terminal (SAT) interface. For more information on configuring Communication Manager, refer to the Avaya product documentation, **Reference [1]**.

This section provides the procedures for configuring Communication Manager.
The procedures include the following areas:

- Verify System Capabilities and Licensing
- Administer IP Codec Set
- Administer IP Network Region
- Administer IP Node Names
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Administer Route Pattern
- Administer AAR Analysis
- Administer UUI Collect Vector
- Administer UUI Test VDN
- Administer Incoming Vector
- Administer Incoming VDN
- Administer Agent Skill Group
- Administer Agent ID
- Administer Feature Access Codes

5.1. Verify System Capabilities and Licensing

5.1.1. SIP Trunk Capacity

Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** have sufficient remaining capacity. The example shows that **4000** licenses are available and **20** are in use.

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  20
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  20
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.1.2. ISDN/SIP Network Call Redirection Check

Verify that **ISDN/SIP Network Call Redirection** is enabled on **Page 4** of system-parameters customer options.

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

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

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

5.1.3. Vector (Basic) Check

Verify that **Vectoring (Basic)** is enabled on **Page 6** of system-parameters customer options.

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

                          Call Center Release: 6.0

                                ACD? y                           Reason Codes? y
                       BCMS (Basic)? y                Service Level Maximizer? n
         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.)
```

5.1.4. Vectoring (Variables) Check

Verify that **Vectoring (Variables)** is enabled on **Page 6** of system-parameters customer options. **Note:** the Vectoring (Variables) is enabled in order to generate and populate simulated UUI contents for testing.

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

                          Call Center Release: 6.0

                                ACD? y                              Reason Codes? y
                       BCMS (Basic)? y                   Service Level Maximizer? n
         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.)
```

5.1.5. Expert Agent Selection (EAS) Check

Verify that **Expert Agent Selection (EAS)** is enabled on **Page 6** of system-parameters customer options.

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

                          Call Center Release: 6.0

                                ACD? y                              Reason Codes? y
                       BCMS (Basic)? y                   Service Level Maximizer? n
         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.)
```

5.1.6. Expert Agent Selection (EAS) Enabled Feature Check

Use the **display system-parameter features** command and on **Page 11**, verify that **Expert Agent Selection (EAS) Enabled** feature is enabled.

```
display system-parameters features                              Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                   Delay:
    Message Waiting Lamp Indicates Status For: station

  VECTORING
                    Converse First Data Delay: 0      Second Data Delay: 2
               Converse Signaling Tone (msec): 100         Pause (msec): 70
                     Prompting Timeout (secs): 10
                 Interflow-qpos EWT Threshold: 2
    Reverse Star/Pound Digit For Collect Step? n
          Available Agent Adjustments for BSR? n
                            BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
               Service Observing: Warning Tone? y     or Conference Tone? n
 Service Observing/SSC Allowed with Exclusion? n
         Allow Two Observers in Same Call? n
```

5.2. Administer IP Codec Set

Use the **change ip-codec-set command** to administer an IP codec set. IP codec set ***1*** was used during compliance testing. Multiple codecs can be listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The example below shows the values used during compliance testing. IP codec sets are used in **Section 5.3** for configuring IP network regions to specify which codec sets may be used within and between network regions.

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

                          IP Codec Set

    Codec Set: 1

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

5.3. Administer IP Network Region

Use the **change ip-network-region** command to administer the network region settings. The values shown below are the values used during compliance testing. Note that the **IP-IP Direct Audio** was disabled during the compliance test with OpenMethods.

- **Authoritative Domain**: ***avaya.com***
- **Name**: Any descriptive name may be used (if desired).
- **Intra-region IP-IP Direct Audio**: ***no***
  Inter-region IP-IP Direct Audio: ***no***
  IP-IP Direct Audio (media shuffling) can be further restricted at the trunk level on the **Signaling Group** form.
- **Codec Set**: ***1*** The codec set contains the list of codecs available for calls within this IP network region.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: SM_Public
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4. Administer IP Node Names

Use the **change node-names ip** command to create a node name and enter the IP address of Session Manager. Enter a descriptive name in the **Name** column and the Session Manager IP address in the **IP Address** column. Also note the node name of the processor (***procr***) as it will be used later to configure the SIP trunk between Communication Manager and Session Manager.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
aes                 192.168.62.108
default             0.0.0.0
msgsrvr             192.168.62.28
procr               192.168.62.28
procr6              ::
sm                  192.168.62.18
```

5.5. Administer SIP Signaling Group

Use the command **add signaling group** to create a signaling group between Communication Manager and Session Manage. This signaling group is used for inbound and outbound calls between OpenMethods and the Avaya enterprise. For the compliance test, signaling group ***1*** was configured using the parameters highlighted below. Default values may be used for all other fields.

- **Group Type:** ***sip***
- **Transport Method:** ***tls***
- **IMS Enabled:** ***n*** **(**This specifies the Communication Manager will function as an Evolution Server.)
- **Peer Detection Enabled**: ***y***
- **Peer Server:** Use default value, ***Others***. **Note:** default value is replaced with "***SM***" after SIP trunk to Session Manager is established.
- **Near-end Node Name:** Node name that maps to the IP address of the processor (i.e., ***procr***) from **Section 5.4**.
- **Far-end Node Name:** Session Manager node name from **Section 5.4**.
- **Near-end Listen Port:** ***5061***
- **Far-end Listen Port:** ***5061***
- **Far-end Network Region:** IP-network-region from **Section 5.3**.
- **Far-end Domain:** Authoritative Domain from **Section 5.3**.
- **Direct IP-IP Audio Connections:** ***n*** (This setting disables Media Shuffling on the trunk level.)

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others


   Near-end Node Name: procr                 Far-end Node Name: sm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
                                   Far-end Secondary Node Name:
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? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                    Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

Trunk group ***1*** was configured with the **add trunk-group** command using the parameters highlighted below. Default values may be used for all other fields.

- **Group Type:** ***sip***
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** ***tie***
- **Signaling Group:** The number of the signaling group added in **Section 5.5**.
- **Number of Members:** The number of simultaneous calls that can be routed to Session Manager.

```
add trunk-group 1                                                 Page   1 of  22
                                TRUNK GROUP

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: to_SM_Public                COR: 1        TN: 1        TAC: *001
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

On **Page 3**:

- **Numbering Format:** ***private***
- **UUI Treatment:** ***shared***
- **Maximum Size of UUI Contents:** ***128***

```
add trunk-group 1                                                 Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no
               Send UCID? n

 Show ANSWERED BY on Display? y

 DSN Term? n
```

On **Page 5**, set the following value:

- **Network Call Redirection:** *y*

```
display trunk-group 1                                              Page   5 of  22
                              PROTOCOL VARIATIONS

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


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

5.7. Administer Route Patten

Use the **change route-pattern** command to create a route pattern that will route calls to the SIP trunk that connects to Session Manager.

A descriptive name was entered for the **Pattern Name** field. The **Grp No** field was set to the trunk group created in **Section 5.6**. The Facility Restriction Level (**FRL**) field was set to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The **Numbering Format** was set to ***lev0-pvt***. The default values were used for all other fields.

```
change route-pattern 1                                             Page   1 of   3
                    Pattern Number: 1   Pattern Name: SM_62_18
                             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: 1    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                               lev0-pvt  none
```

5.8. Administer AAR Analysis

Automatic Alternate Routing (AAR) was used to route calls to OpenIVR via Session Manager. Use the **change aar analysis** command to create an entry in the AAR Digit Analysis Table for this purpose. The highlighted entry specifies that if the dialed number is ***2001*** and is ***4*** digits long, to use route pattern ***1***. Route pattern ***1*** routes calls to Session Manager.

```
change aar analysis 2                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 3

          Dialed            Total     Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    2001                  4    4      1         aar          n
```

5.9. Administer UUI Collect Vector

Administer the UUI Test vector to collect 7 digits and then route to 2001 (OpenIVR). This vector is associated to the Vector Directory Number (VDN) in **Section 5.10**.

Vector 5 does the following:

- Step 1 - Plays ringback for 2 seconds.
- Step 2 - Collects 7 digits.
- Step 3 - Sets the 7 collected digits to ASAI UUI variable UU.
- Step 4 – Routes to number 2001 (OpenIVR).

Using the command **change vector x** where **x** is the number of the vector to create. Add the basic steps below.

```
change vector 5                                                Page   1 of   6
                                  CALL VECTOR

    Number: 5                Name: UUI Test
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 collect      7    digits after announcement none     for none
03 set          UU     = digits CATL  none
04 route-to     number 2001             with cov n if unconditionally
```

Note: The parameters for ASAI UUI variable UU in the previous step and other vector variables are defined using the **change variables** command.

```
change variables                                                Page  32 of  39
                              VARIABLES FOR VECTORS

Var Description                     Type    Scope Length Start Assignment       VAC
UM
UN
UO
UP
UQ
UR
US
UT
UU  UUI Test                        asaiuui L     7      1
```

5.10. Administer UUI Test VDN

Administer the UUI Test VDN. This VDN uses vector ***5*** from **Section 5.9**.

Using the command **add vdn x** where **x** is the extension of the VDN to create and enter the following values:

- **Extension:** Enter the extension allowed by the dial plan.
- **Name:** Enter a descriptive name.
- **Destination:** Enter the Vector Number created in **Section 5.9**.

```
display vdn 54555                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 54555
                                 Name*: UUI Test
                           Destination: Vector Number        5
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none


        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

5.11. Administer Incoming Vector

Administer a vector that will queue an incoming call to a call center agent. This vector is associated with the incoming Vector Directory Number (VDN) to be configured in **Section 5.12**. After OpenIVR collects 7 digits and the caller presses "1" to attach data or "2" to leave the data the same, OpenIVR will transfer the call to the VDN.

Using the command **change vector x** where **x** is the number of the vector to create. Add the basic steps below. For step 1, use the skill group number from **Section 5.13**.

```
change vector 1                                                   Page   1 of   6
                                  CALL VECTOR

    Number: 1                Name: DevConnect Test
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 queue-to     skill 1    pri h
02 wait-time    30  secs hearing ringback
03 stop
```

5.12. Administer Incoming VDN

Administer the incoming VDN. This VDN uses the vector from **Section 5.11**.

Using the command **add vdn x** where **x** is the extension of the VDN to create and enter the following values:

- **Extension:** Enter the extension allowed by the dial plan.
- **Name:** Enter a descriptive name.
- **Destination:** Enter Vector Number created in **Section 5.11**.

```
add vdn 54888                                                     Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 54888
                                 Name*: DevConnect Test
                           Destination: Vector Number        1
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none


        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

5.13. Administer Agent Skill Group

Administer the agent skill group. Using the command **add hunt x** where **x** is an available skill group number and enter the following values:

- **Group Name:** Enter descriptive name for the agent skill.
- **Group Extension:** Enter an extension for the skill.
- **ACD:** ***y***
- **Queue:** ***y***
- **Vector:** ***y***

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

            Group Number: 1                                 ACD? y
              Group Name: DevConnect Test                 Queue? y
         Group Extension: 54899                          Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                        MM Early Answer? n
           Security Code:                   Local Agent Preference? n
 ISDN/SIP Caller Display:

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

On **Page 2**, set the following value:

- **Skill:** ***y***

```
display hunt-group 1                                          Page   2 of   4
                                  HUNT GROUP

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


      Controlling Adjunct: none




   Multiple Call Handling: none


 Timed ACW Interval (sec):          After Xfer or Held Call Drops? n
```

5.14. Administer Agent ID

For the sample configuration add an agent id that is associated with the skill group configured in **Section 5.13** in order to receive ACD calls.

Using the command **add agent x** where **x** is a valid extension number for an agent id defined in the system. Fill in the indicated fields. Enter the following values on **Page 1** of the **change agent-loginID** form. Default values may be used for all other fields.

- **Name:** Display name for agent.
- **Password:** Numeric password used when the agent logs into a station.

```
add agent-loginID 54777                                           Page   1 of   2
                                 AGENT LOGINID

                Login ID: 54777                                       AAS? n
                    Name: DevConnect Test 1                         AUDIX? n
                      TN: 1                                 LWC Reception: spe
                     COR: 1                        LWC Log External Calls? n
           Coverage Path:                        AUDIX Name for Messaging:
           Security Code:
                                             LoginID for ISDN/SIP Display? n
                                                                Password: 123456
                                                  Password (enter again): 123456
                                                              Auto Answer: station
                                                        MIA Across Skills: system
                                                ACW Agent Considered Idle: system
                                                Aux Work Reason Code Type: system
                                                  Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                                  Forced Agent Logout Time:   :

    WARNING:  Agent must log in again before changes take effect
```

On Page 2 associate the agent to the skill created in **Section 5.13**.
Note: SN is skill number and SL is skill level.

```
display agent-loginID 54777                                       Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                                   Service Objective? n
Call Handling Preference: skill-level                     Local Call Preference? n

    SN    RL SL          SN    RL SL
 1: 1        1       16:
```

5.15. Administer Feature Access Codes

Using the command **change feature-access-codes**, administer the following feature access codes-on **Page 5** of the form:

- **Auto-In Access Code:** FAC to staff in the agent ID to their respective skill.
- **Login Access Code:** FAC to login the agent ID to their respective skill.
- **Logout Access Code:** FAC to logout the agent ID from their skill.

```
change feature-access-codes                                    Page   5 of  11
                              FEATURE ACCESS CODE (FAC)

                                Call Center Features
  AGENT WORK MODES
                    After Call Work Access Code:
                             Assist Access Code:
                            Auto-In Access Code: *13
                           Aux Work Access Code: *12
                              Login Access Code: *11
                             Logout Access Code: *15
                          Manual-in Access Code: *14
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as shown in the reference configuration. All provisioning for Session Manager is performed via the System Manager web interface. This section assumes that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

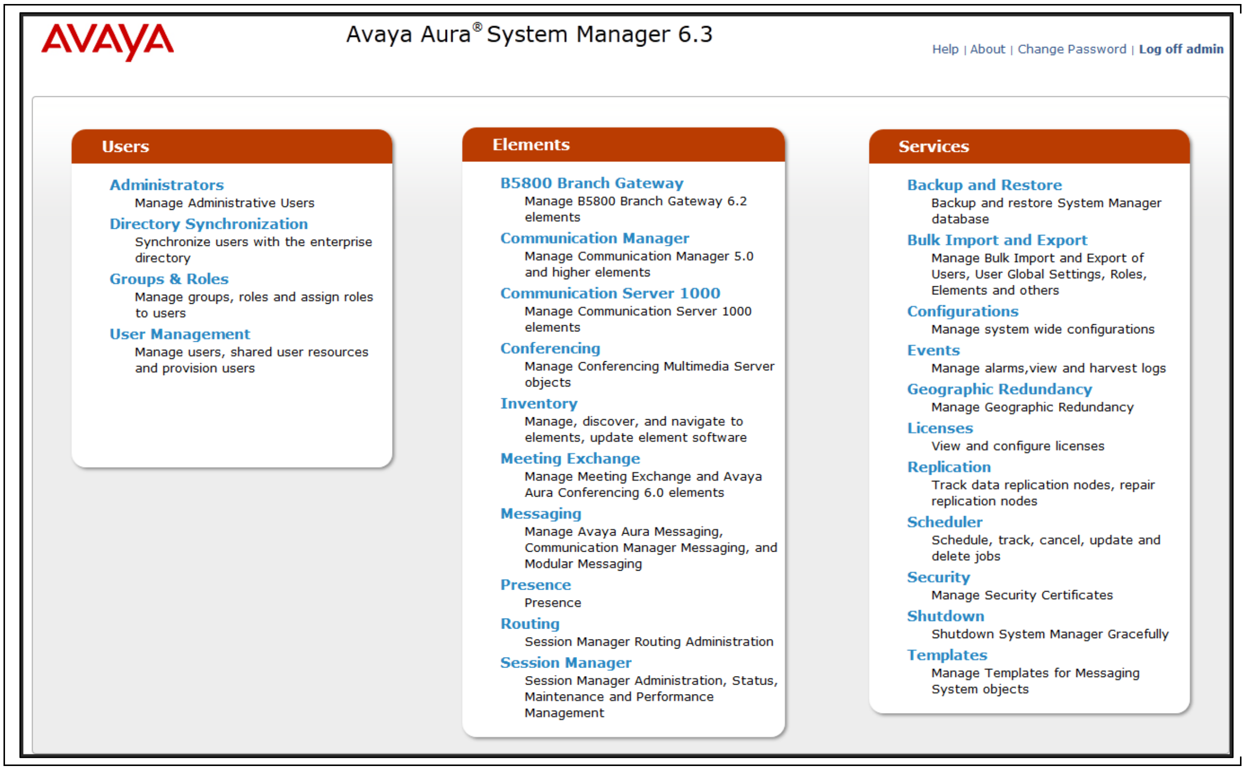
This section summarizes the configuration steps that are necessary for interoperating with OpenIVR. The test environment was previously configured to enable Communication Manager and Session Manager at each site to communicate with each other. Details of this configuration are not described in this document. Additional information can be obtained from **Reference [3]**.

The procedures described in this section include configurations for the following:

- Administer SIP Domains
- Administer Adaptation
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

Session Manager is managed via System Manager. Using a web browser, access ***https://<ip-addr of System Manager>/SMGR***.

Log in using appropriate credentials. The main page for the administrative interface is shown below.



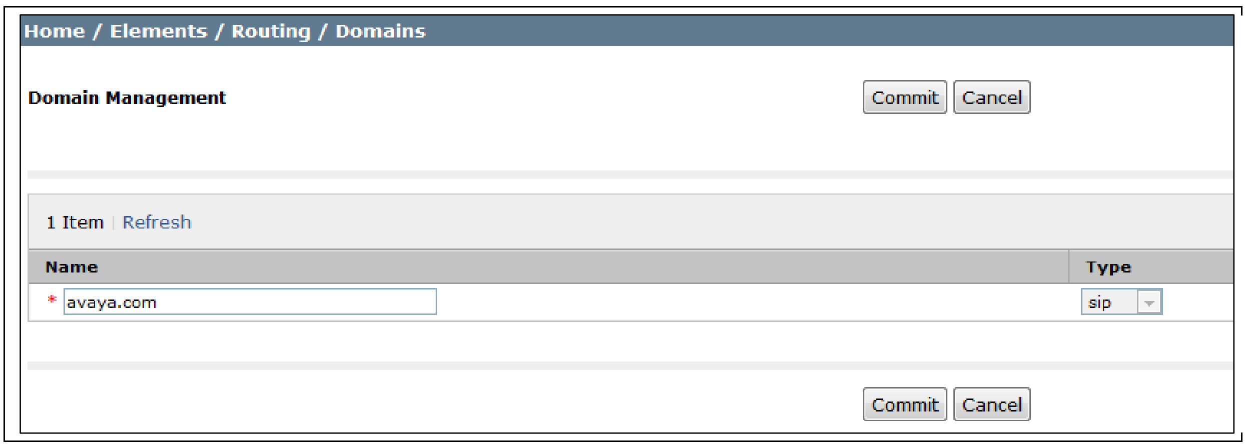
6.1. Administer SIP Domains

In the reference configuration, one SIP domain was used: avaya.com.

Navigate to **Element → Routing → Domains** and click the **New** button (not shown) to add a new SIP domain with the following:

- **Name**: ***avaya.com***
- **Type** : ***sip***
- **Notes**: Optional descriptive text.

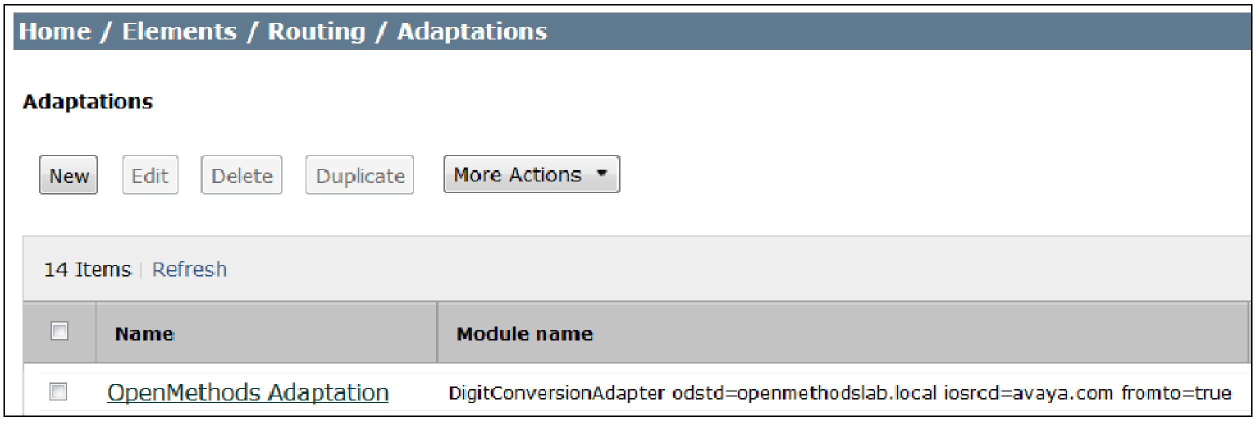
Click on the **Commit** button.

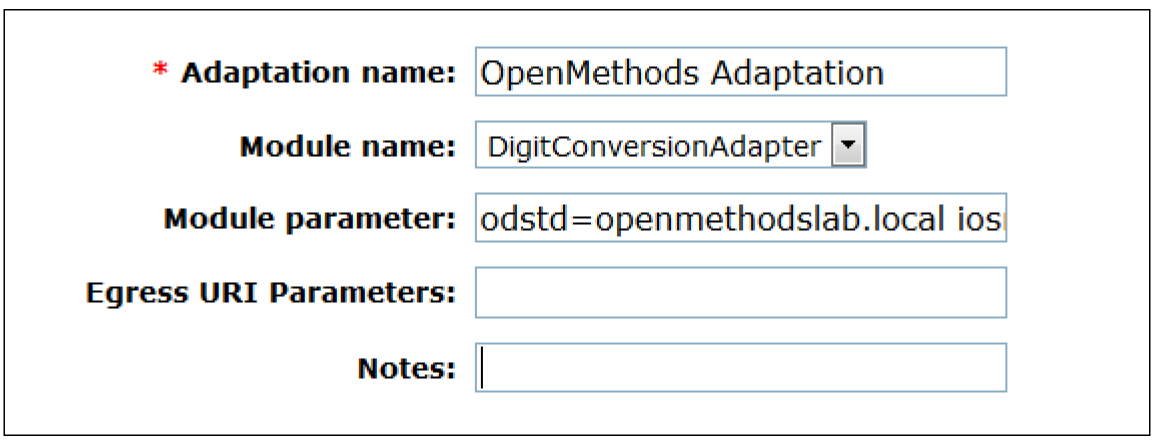


6.2. Administer Adaptation

Navigate to **Routing → Adaptations**. Click **New** to add a new Adaptation with the following:

- **Adaptation name:** Enter a descriptive name for the adaptation.
- **Module name:** Select ***DigitConversionAdapter***.
- **Module parameter:** During compliance testing, ***odstd=openmethodslab.local iosrcd=avaya.com fromto=true*** was used.





6.3. Administer SIP Entities

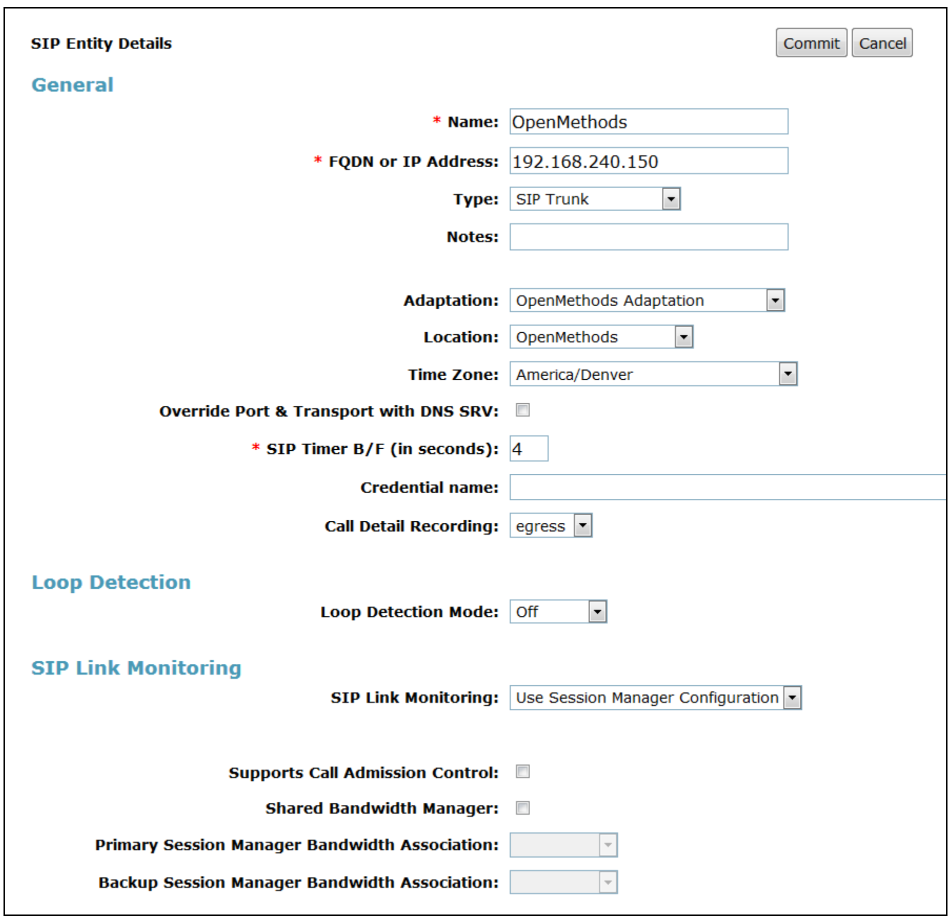
A SIP Entity must be added for the Session Manager and for each SIP-based telephony system supported by it using SIP trunks. During compliance testing, a SIP Entity was added for OpenIVR.

Navigate to **Routing → SIP Entities**, and click the **New** button (not shown) to add a SIP Entity. The configuration details for the SIP Entity defined for OpenIVR are as follows:

Under **General**:

- **Name**: A descriptive name.
- **FQDN or IP Address**: ***192.168.240.150***
- **Type**: ***SIP Trunk***
- **Adaptation:** ***OpenMethods Adaptation*** as configured in **Section 6.2**.
- **Location:** Select a previously configured location (the configuration of the location is not shown in this document).

Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity.



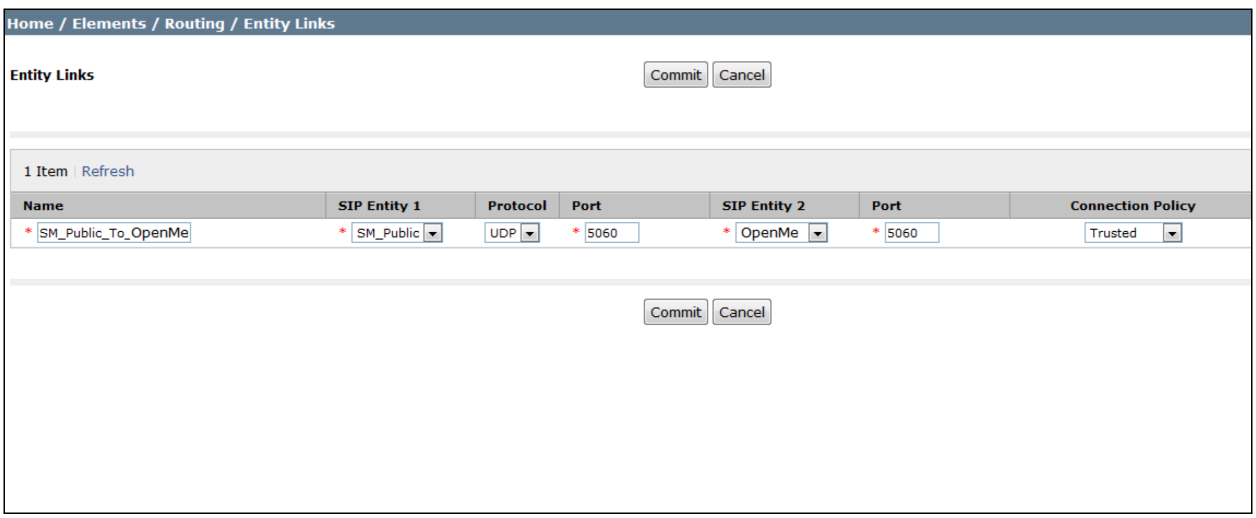
6.4. Administer Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. During compliance testing one Entity Link was created:

- Session Manager ←→ OpenIVR

Navigate to **Routing → Entity Links**, and click the **New** button (not shown) to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager to OpenMethods.

- **Name**: A descriptive name.
- **SIP Entity 1**: Select the Session Manager SIP Entity.
- **Protocol**: Select ***UDP*** as the transport protocol to match the protocol used by OpenMethods.
- **Port**: ***5060***. This is the port number to which the other system sends SIP requests.
- **SIP Entity 2**: Select the OpenMethods SIP Entity.
- **Port: *5060***. This is the port number on which the other system receives SIP requests.
- **Connection Policy**: Select ***Trusted***.
- **Notes**: Optional descriptive text.



Click **Commit** to save the configuration

6.5. Administer Routing Policies

Routing Policies were added for routing calls to OpenIVR.

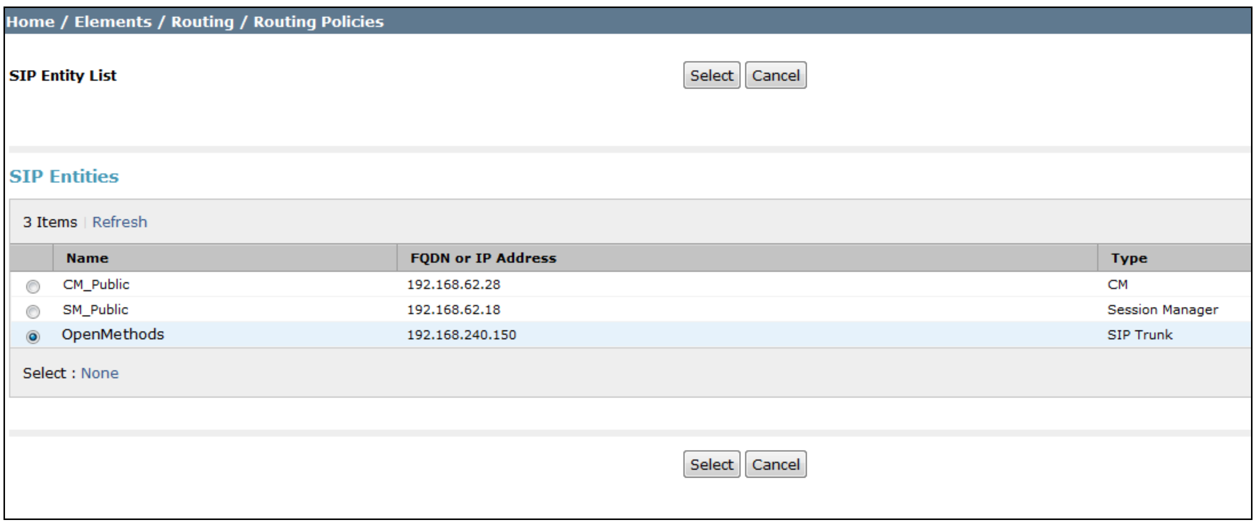
Navigate to **Routing** → **Routing Policies**, and click the **New** button (not shown) to add a new Routing Policy as follows.

Under **General**:

- **Name**: A descriptive name.
- **Notes**: Optional descriptive text.

Under **SIP Entity as Destination**

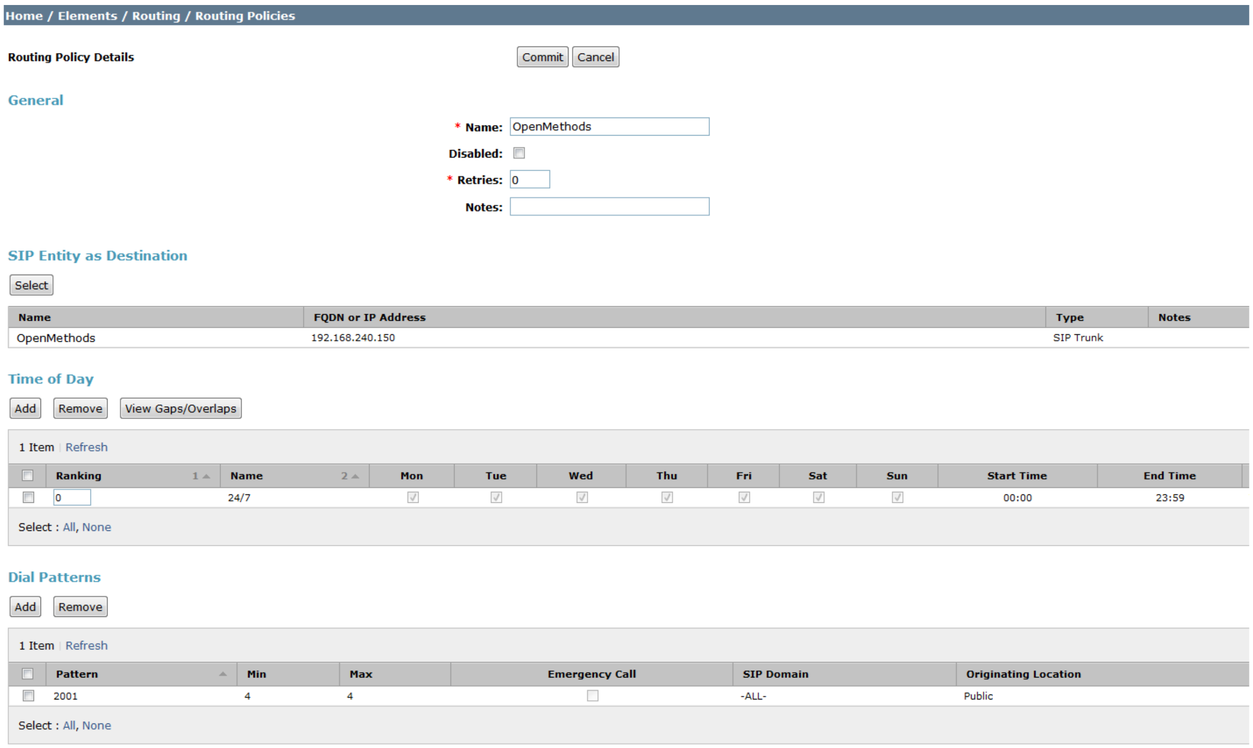
Click the **Select** button and the screen below is displayed. Select ***OpenMethods*** SIP Entity (defined in **Section 6.4**), to which the routing policy applies, and click the **Select** button to return to the previous screen.



Under **Time of Day**
Click **Add** to select a Time Range (not shown since the default time range of 24/7 was used during compliance testing).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration.

The screen below shows the routing policy used during compliance testing.



6.6. Administer Dial Pattern

Dial Patterns define digit strings to be matched against dialed numbers for directing calls to the appropriate SIP Entities. The number “2001” was routed to OpenIVR.

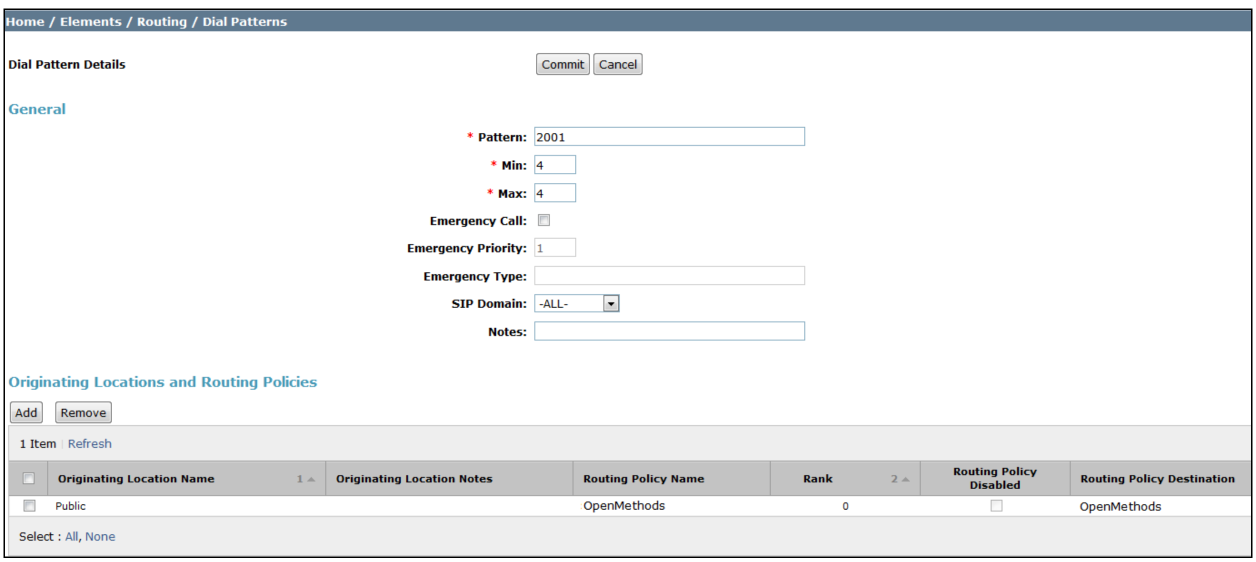
Navigate to **Routing → Dial Patterns**, click the **New** button (not shown) to add a new Dial Pattern as follows.

Under **General**:

- **Pattern**: Dialed number or prefix.
- **Min**: Minimum length of dialed number.
- **Max**: Maximum length of dialed number.
- **SIP Domain**: Select the SIP Domain created in **Section 6.1** (or select ***–ALL–*** to be less restrictive).
- **Notes**: Optional descriptive text.

Under **Originating Locations and Routing Policies**
Click **Add** to select the appropriate originating Location (e.g., ***Public***) and Routing Policy (e.g., ***OpenMethods***) from the list (not shown).

Default settings can be used for the remaining fields. Click **Commit** to save the configuration.



7. Configure OpenIVR

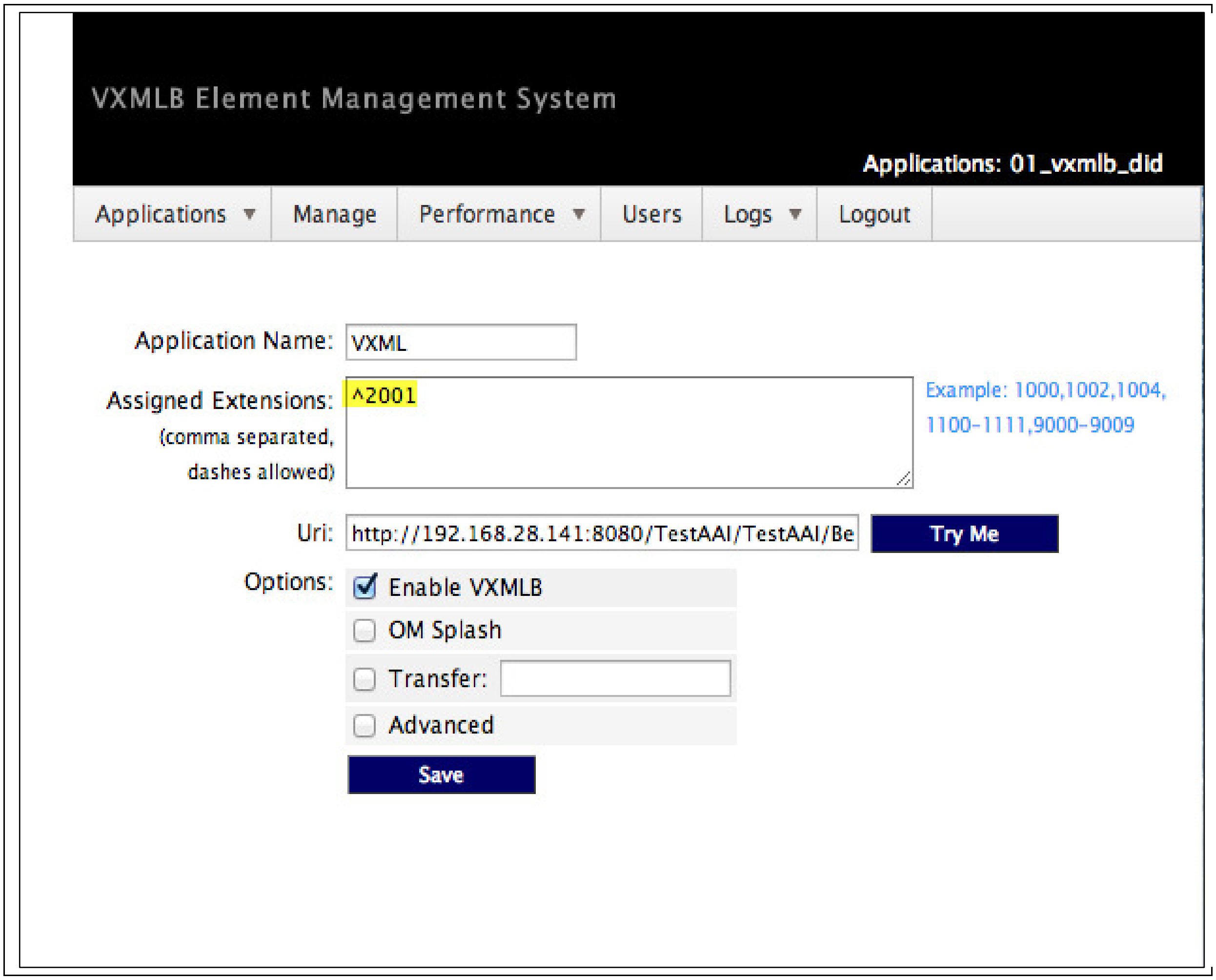
The following steps are required for OpenIVR to properly accept calls from Session Manager.

Note: OpenMethods performs all configurations for their equipment; therefore very minimal information is included in these application notes.

On the OpenIVR server open **acl.conf.xml** in a text editor. Under **<list name="domains" default="allow">**, add the IP address for Session Manager ***<node type="allow" domain="192.168.62.18:/>***. Save **acl.conf.xml**.

```
<configuration name="acl.conf" description="Network Lists">
  <network-lists>
    <!--
         These ACL's are automatically created on startup.

         rfc1918.auto  - RFC1918 Space
         nat.auto      - RFC1918 Excluding your local lan.
         localnet.auto - ACL for your local lan.
         loopback.auto - ACL for your local lan.
    -->

    <list name="lan" default="allow">
      <node type="deny" cidr="192.168.42.0/24"/>
      <node type="allow" cidr="192.168.42.42/32"/>
    </list>

    <!--
        This will traverse the directory adding all users
        with the cidr= tag to this ACL, when this ACL matches
        the users variables and params apply as if they
        digest authenticated.
    -->
    <list name="domains" default="allow">
      <!-- domain= is special it scans the domain from the directory to build the ACL -->
      <node type="allow" domain="$${domain}"/>
      <node type="allow" domain="<192.168.62.18"/>
      <!-- use cidr= if you wish to allow ip ranges to this domains acl. -->
      <!-- <node type="allow" cidr="192.168.0.0/24"/> -->
    </list>

  </network-lists>
```

The IVR extension used for compliance testing was 2001. Open the OpenMethods VXMLB Element Management System and select the application used in the compliance test. In the **Assigned Extensions** field add the IVR number ***2001*** with a prefix ^. Click on **Save**.



8. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands that can be used to troubleshoot the solution.

Verification Steps:

1. Verify that endpoints at the enterprise site can place calls to OpenIVR and that the call remains active for more than 35 seconds.
    a. Verify the passing of DTMF events and their recognition by navigating automated menus.
    b. Verify codec negotiation.
    c. Verify contents of the User-to-User header.
2. Verify that endpoints at the enterprise site can receive a transferred call from OpenIVR and that the call can remain active for more than 35 seconds.
    a. Verify codec negotiation.
    b. Verify contents of the User-to-User header.
3. Verify that the IVR will end an active call by timing out and trunk resources are released.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up and trunk resources are released.

Communication Manager commands:

- **list trace station** <extension number> - Traces calls to and from a specific station.
- **list trace tac** <trunk access code number> - Trace calls over a specific trunk group.
- **status station** <extension number> - Displays signaling and media information for an active call on a specific station.
- **status trunk** <trunk group number> - Displays trunk group information.

Session Manager commands:

- **traceSM** – Session Manager command line tool for traffic analysis. Log in to the CLI based Session Manager management interface to run this command.
- **SIP Entity Link Monitoring** – If monitoring is enabled, log in to the System Manager web console. Navigate to **Element**s → **Session Manager** → **System Status** → **SIP Entity Link Monitoring**. Verify all links are up.

9. Conclusion

These Application Notes describe the configuration steps required for OpenIVR to successfully interoperate with Avaya Aura® Session Manager 6.3 and Avaya Aura® Communication Manager 6.2. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya product documentation can be found at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7.0, December 2012, Document Number 03-300509.
[2] *Avaya Aura® Call Center Elite Feature Reference*, Release 6.2, Issue 2, December 2012.
[3] *Administering Avaya Aura® Session Manager*, Release 6.3, December 2012.

OpenMethods product documentation can be obtained by using the contact details in **Section 2.3**.

[4] *OpenIVR*, *Installation and User's Manual*, Version 2.2.0_v4.

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