Avaya Solution & Interoperability Test Lab

Application Notes for MModal Fluency Voice Server with Avaya Aura® Session Manager 7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for MModal Fluency Voice Server to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. MModal FVS is an Interactive voice response (IVR) that records dictations.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 configuration steps required for MModal Fluency Voice Server (FVS) to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks.

In the compliance testing, calls from internal and external callers were routed over SIP trunks to FVS. FVS played greeting announcements, used DTMF digits to determine the action such as enter User ID then a soft talkdown tone is played until user speak FVS start to record dictations, enter DTMF digit to interrupts, play, resume or end recording.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually from users on the PSTN and on Communication Manager to FVS. Speech and DTMF input were used from the callers for recording dictations, interrupts, play, resume or end recording.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to FVS.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and the MModal FVS is UDP. FVS does not utilize any capabilities of TLS.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included:

- G.711MU with shuffling option off.
- FVS receives an incoming call.
- Caller hangs up a call. FVS hangs up a call.
- Receiving a call with delayed offer (SDP in OK instead of INVITE).
- Receiving DTMF as RFC2833.
- Caller putting call on hold/resume call from hold.
- FVS responses to a re-INVITEs.
- FVS responds to OPTIONS ping.
- Load balancing between 2 FVSs.

The serviceability testing focused on verifying the ability of FVS to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to FVS or one FVS server is out of service and incoming call is routed to available FVS without any delay.

2.2. Test Results

All test cases were executed, and the following were observations on FVS:

- The application only supports the G.711MU codec, and does not support codec negotiation and media shuffling.

- Load balancing is not fully round robin. By design, Session Manager will randomly route calls to any available FVS.

2.3. Support

Technical support on FVS can be obtained through the following:

- **Phone:** 1-888-dictate

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Session Manager and FVS. A 10 digit Uniform Dial Plan (UDP) was used to facilitate routing with FVS.

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.

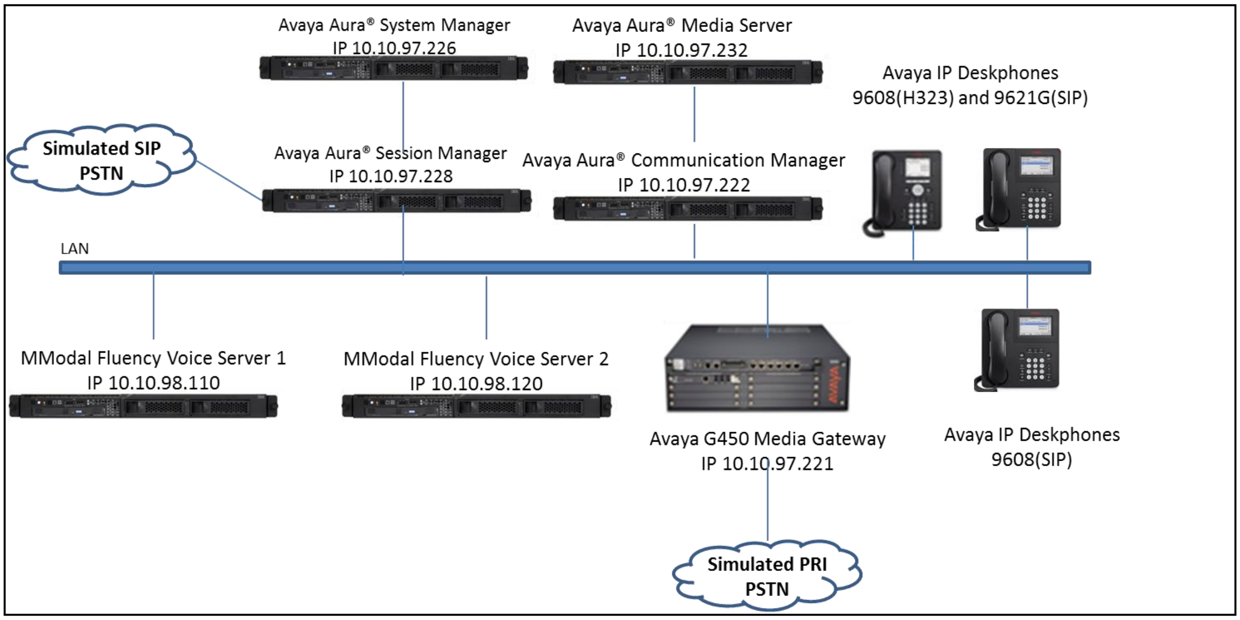


Figure 1: Compliance Testing Configuration

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 in Virtual Environment	7.0.1.2 SP2
Avaya G450 Media Gateway	37.41
Avaya Aura® Media Server in Virtual Environment	7.8
Avaya Aura® Session Manager in Virtual Environment	7.0.1.2
Avaya Aura® System Manager in Virtual Environment	7.0.1.2
Avaya 9608 IP Deskphone (H.323)	6.6.4
Avaya 9608 & 9621G IP Deskphones (SIP)	7.0.1.4
MModal FVS on Microsoft Windows Server 2012	3.6 R2 Standard 64 bit

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- Administer SIP signaling group
- Administer SIP trunk group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer uniform dial plan
- Administer AAR analysis

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with FVS.

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 1800  1
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 1
                   Maximum Video Capable IP Softphones: 24000 20
                       Maximum Administered SIP Trunks: 24000 54
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "1". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tls"
- **Near-end Node Name:** An existing C-LAN node name or "procr".
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with MModal.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with MModal.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** "n", FVS requires shuffling off.

```
display signaling-group 1                                       Page   1 of   3
                                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: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: SM-VM
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: bvwdev.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? y
         Enable Layer 3 Test? y
                                                   Alternate Route Timer(sec): 6
```

5.3. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "5". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"

```
add trunk-group 5                                             Page   1 of  21
                                TRUNK GROUP

Group Number: 5                  Group Type: sip          CDR Reports: y
  Group Name: ToFVS                     COR: 1        TN: 1        TAC: #005
   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: 20
```

Navigate to **Page 3**, and enter "private" for **Numbering Format**.

```
add trunk-group 5                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: internal
                                                   Maintenance Tests? y
   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                               Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
 DSN Term? n                      SIP ANAT Supported? n
```

5.4. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.2**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. For **Codec Set**, enter an available codec set number for integration with FVS.

```
change ip-network-region 1                                    Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: bvwdev.com
    Name: Region1                 Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with network regions used by Avaya endpoints and by the trunk to the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and by the trunk to the PSTN.

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

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1                                                            all
 2
 3
 4
 5
 6
 7
 8
```

5.5. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.4**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that FVS only supports the G.711 Mu-law codec variant. The codec shown below was used in the compliance testing.

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

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2:
 3:
 4:
 5:
```

5.6. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach FVS, in this case "5". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 5                                         Page   1 of   3
                    Pattern Number: 52  Pattern Name: MModal
                             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: 5   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
                                                                 unk-unk
 1: y y y y y n  n            rest                                        none
```

5.7. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed 10 digits 7219675800 to FVS. Note that other routing methods may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing of digits 721, as shown below.

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

  Matching                   Insert                Node
  Pattern       Len Del      Digits     Net Conv Num

 721            10   0                  aar  n
```

5.8. Administer AAR Analysis

Use the "change aar analysis 7" command, and add an entry to specify how to route calls to 721. In the example shown below, calls with digits 721 will be routed as an AAR call using route pattern "5" from **Section 5.6**.

```
change aar analysis 7                                            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
    721                    5   10      5       aar          n
```

6. Configure Avaya Aura® Session Manager

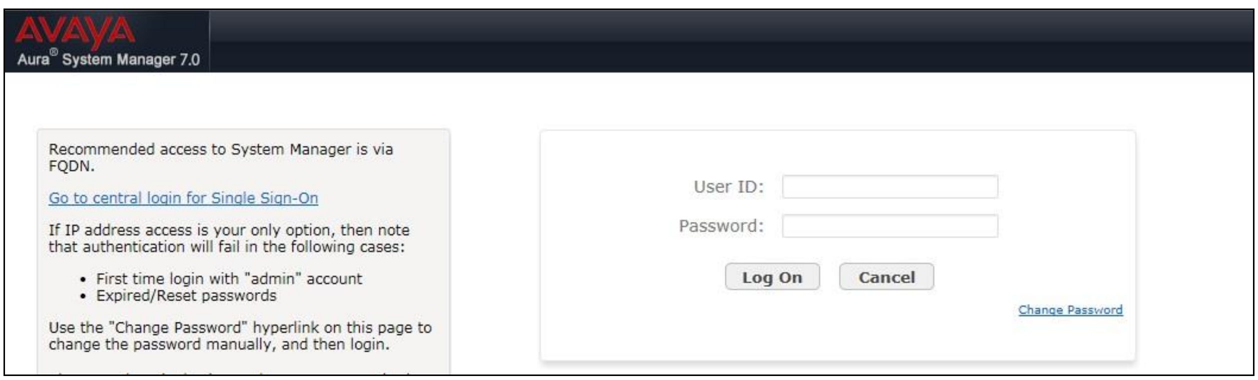
This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer Domains
- Administer Locations
- Administer Adaptations
- Administer SIP entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

Since the configuration was created during compliance test and the screenshots were capture after testing therefore the screenshot will display in modify mode instead of new create objects.

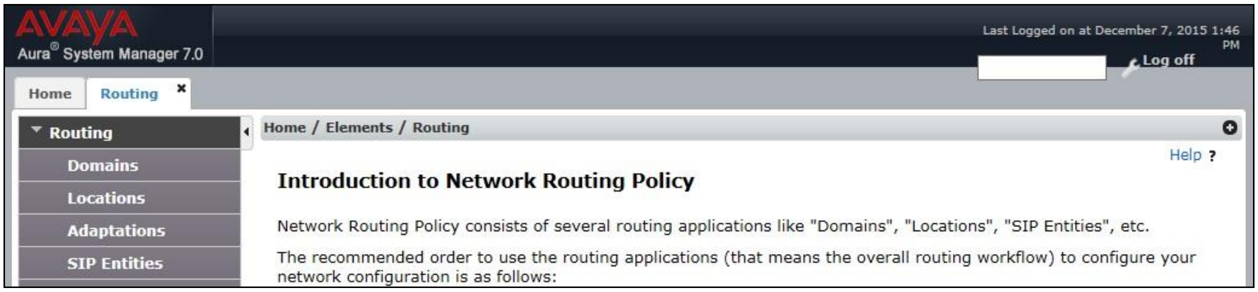
6.1. Launch System Manager

Access the System Manager web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of System Manager. Log in using the appropriate credentials.

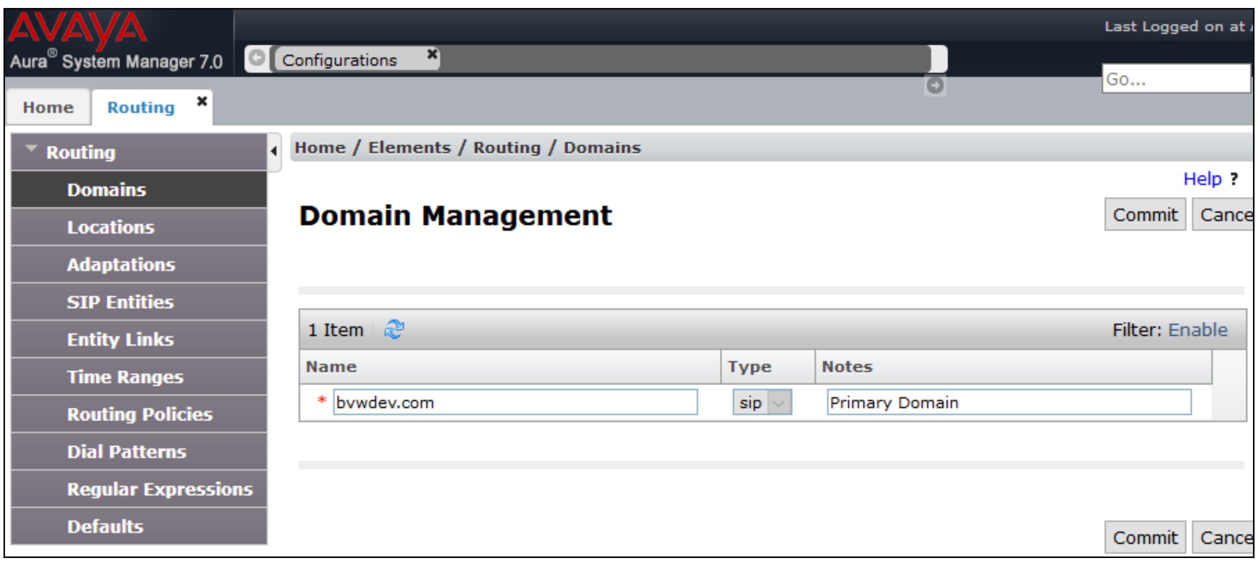


6.2. Administer Domains

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below.

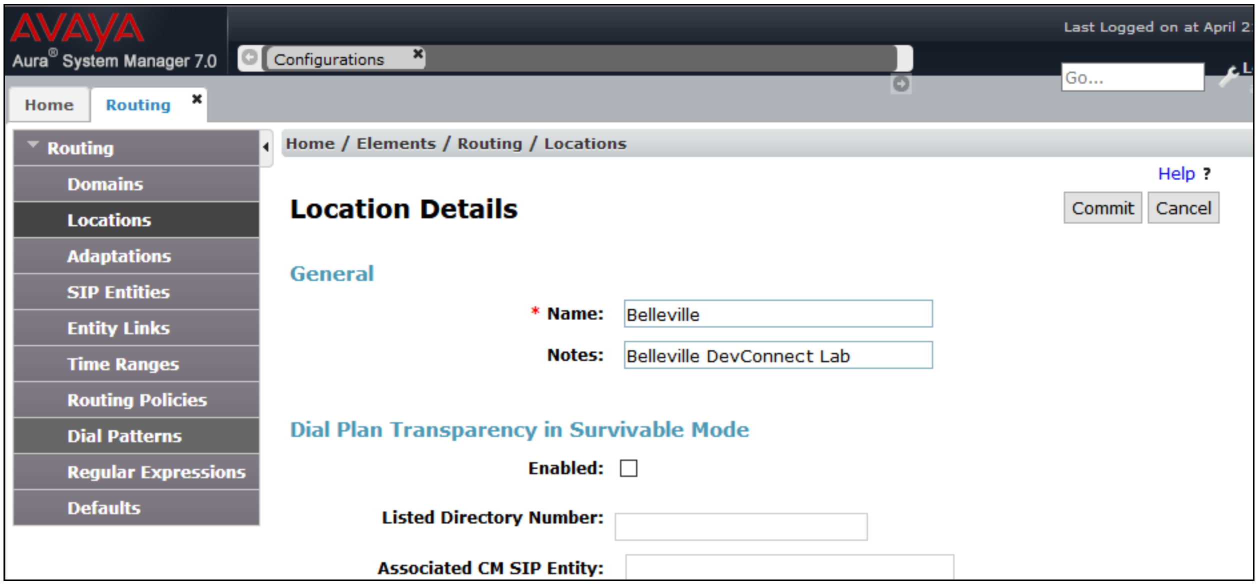


Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for FVS. The **Domain Management** screen is displayed. In the **Name**, enter a domain name used in **Section 5.2**, select **Type** and optional **Notes**.

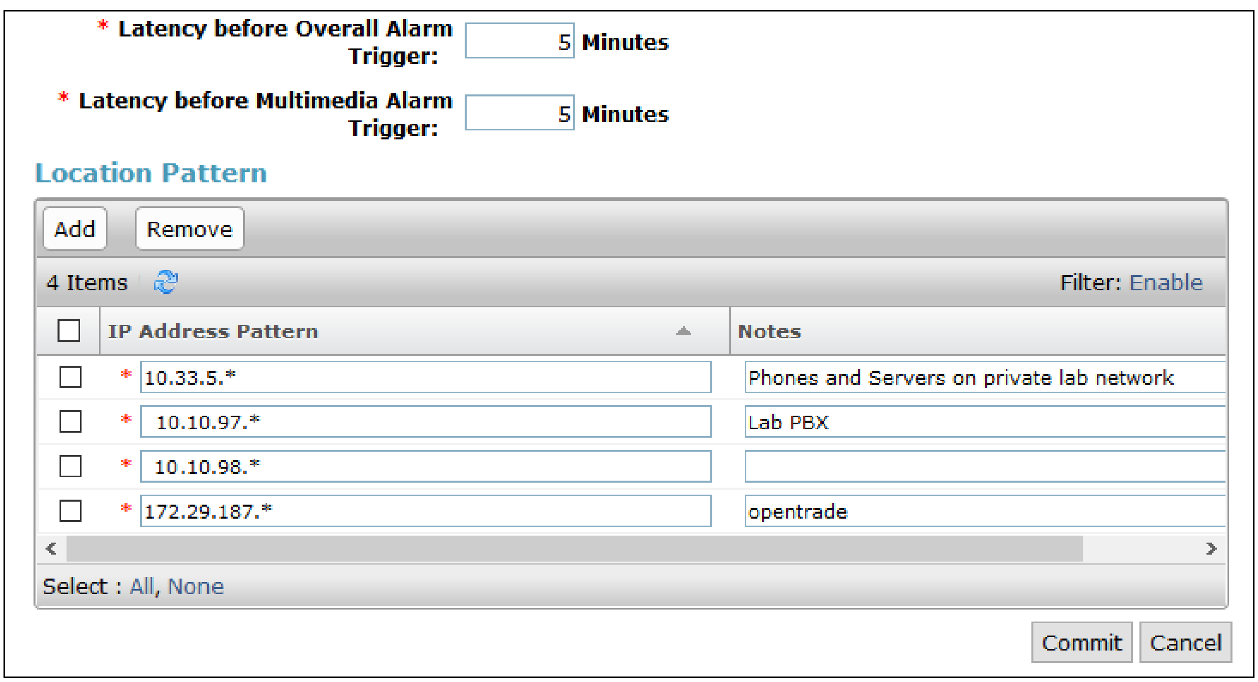


6.3. Administer Locations

Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for FVS. The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of FVS in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



6.4. Administer Adaptations

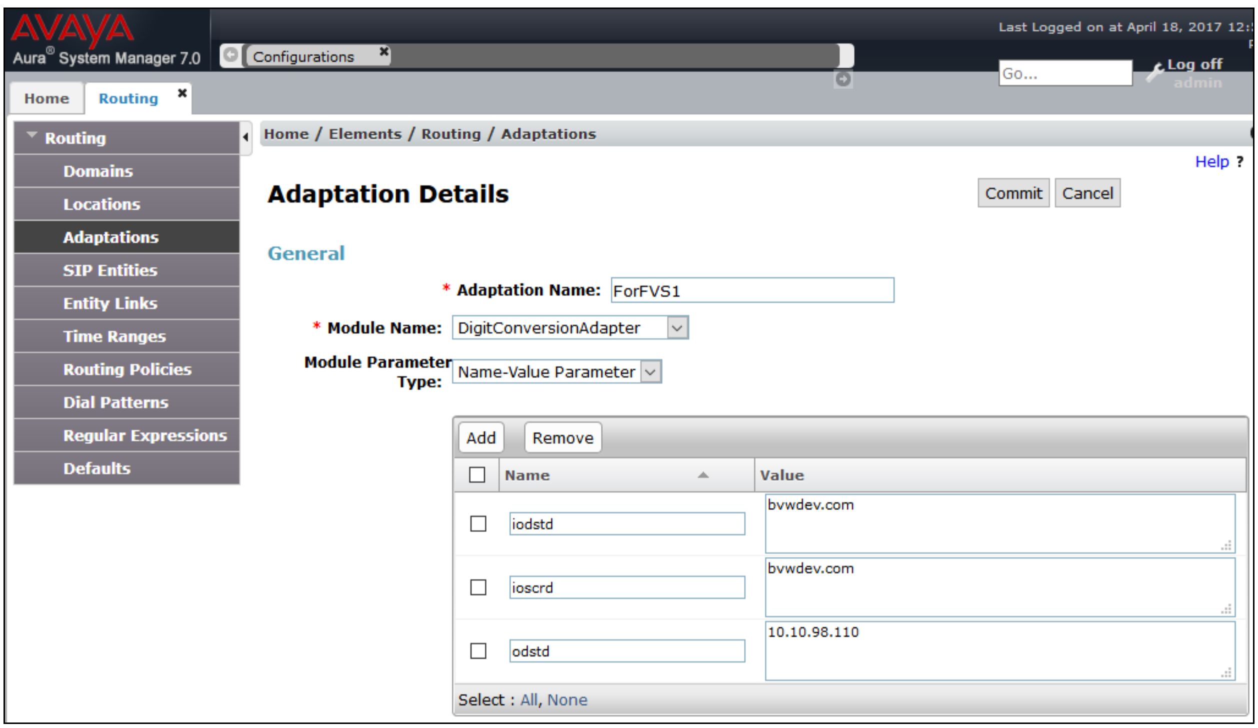
Add two new Adaptations, one for FVS1 and one for FVS2. Select **Routing →Adaptations** from the left panel, and click **New** in the subsequent screen (not shown) to add a new Adaptation for FVS.

The **Adaptation Details** screen is displayed. Enter the following values for specified fields and retain the default value for the remaining fields.

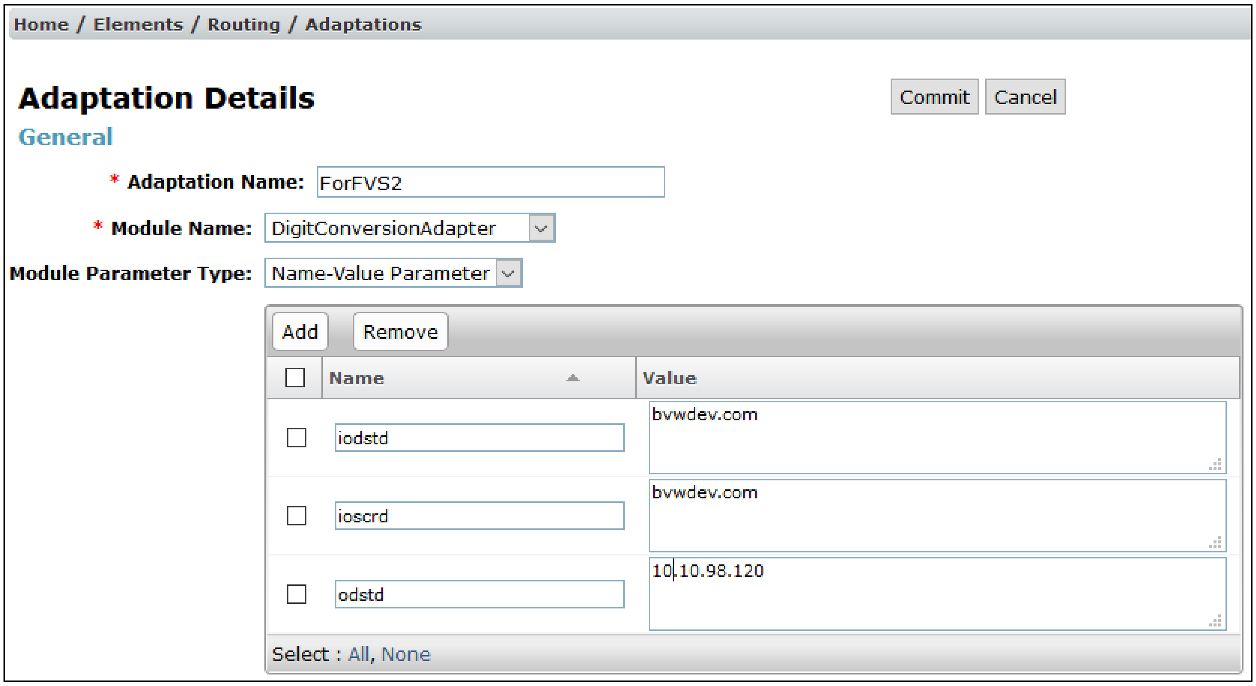
- **Adaptation Name:** A descriptive name**.**
- **Module Name:** Select DigitConversionAdapter.
- **Module Parameter Type:** Select Name-Value Parameter.

Click Add to add new item for parameter:

- **Name:** iodstd and **Value:** bvwdev.com.
- **Name:** ioscrd and **Value:** bvwdev.com.
- **Name:** odstd and **Value:** FVS's IP address, for example, 10.10.98.110.



Repeat the same step for FVS2 as display below:

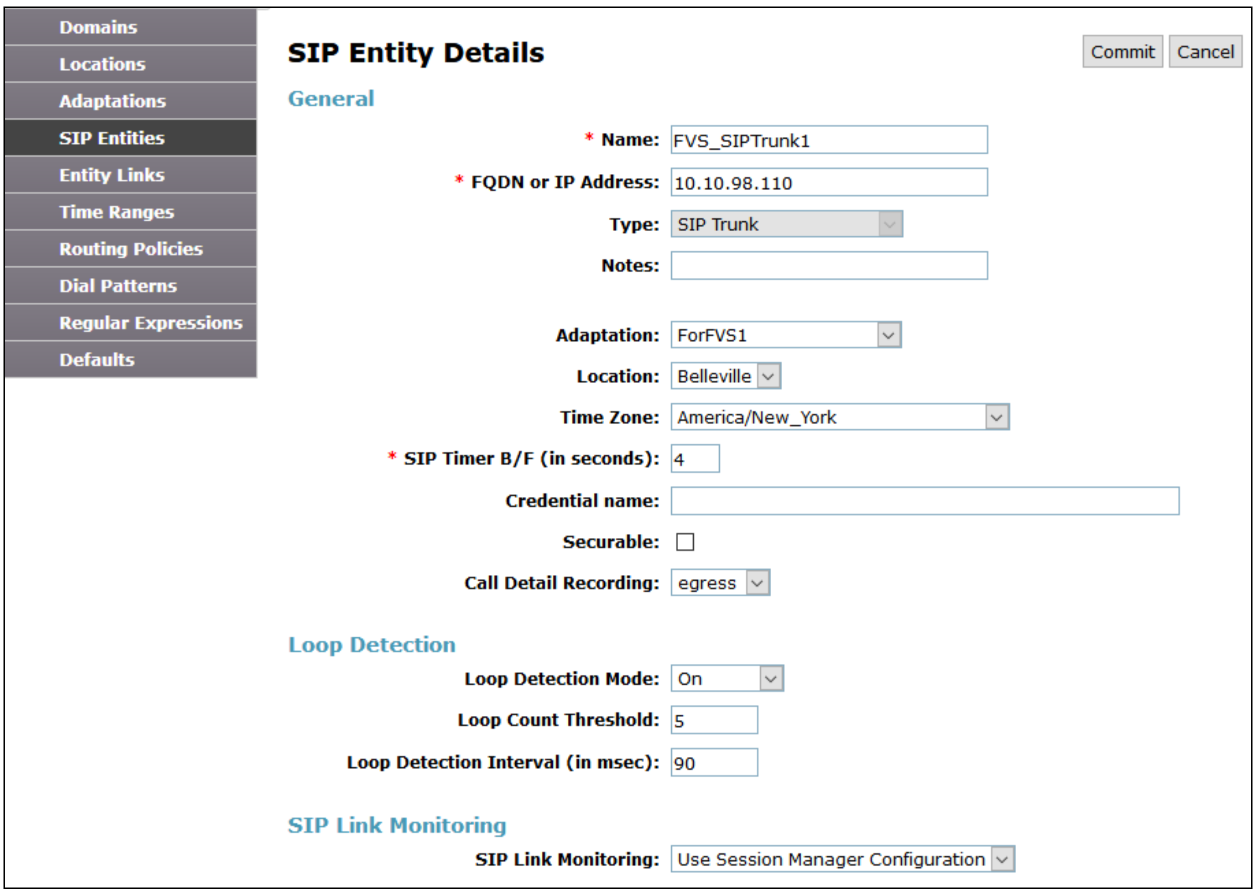


6.5. Administer SIP Entities

Add two new SIP entities, one for FVS1 and one for FVS2. Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for FVS.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

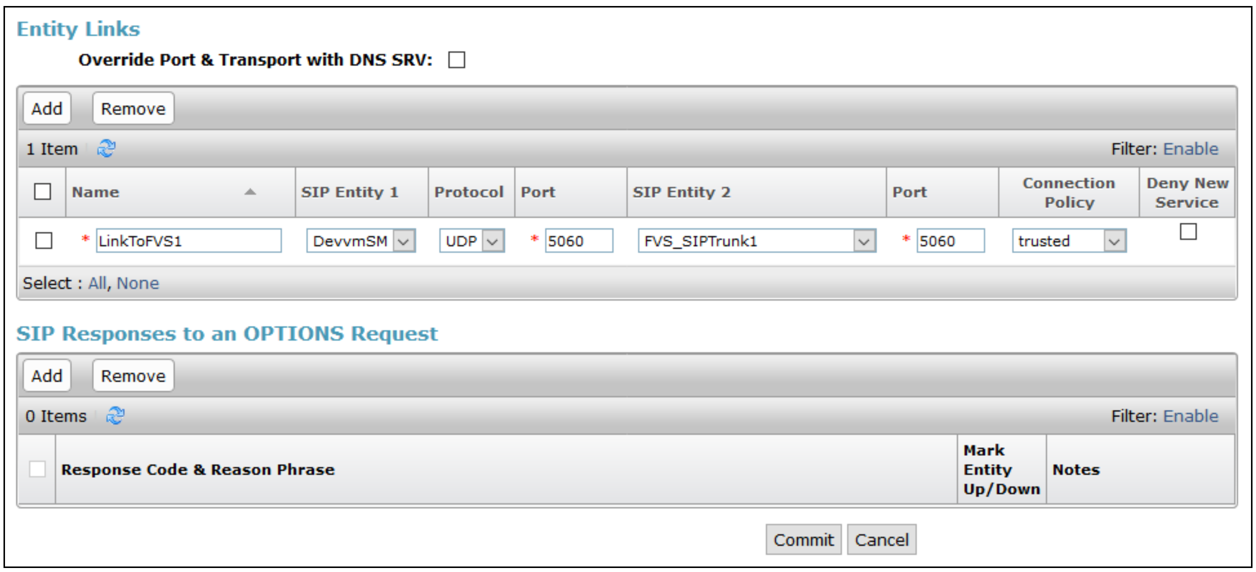
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the FVS1 server.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the FVS location name from **Section 6.3**.
- **Time Zone:** Select the applicable time zone.



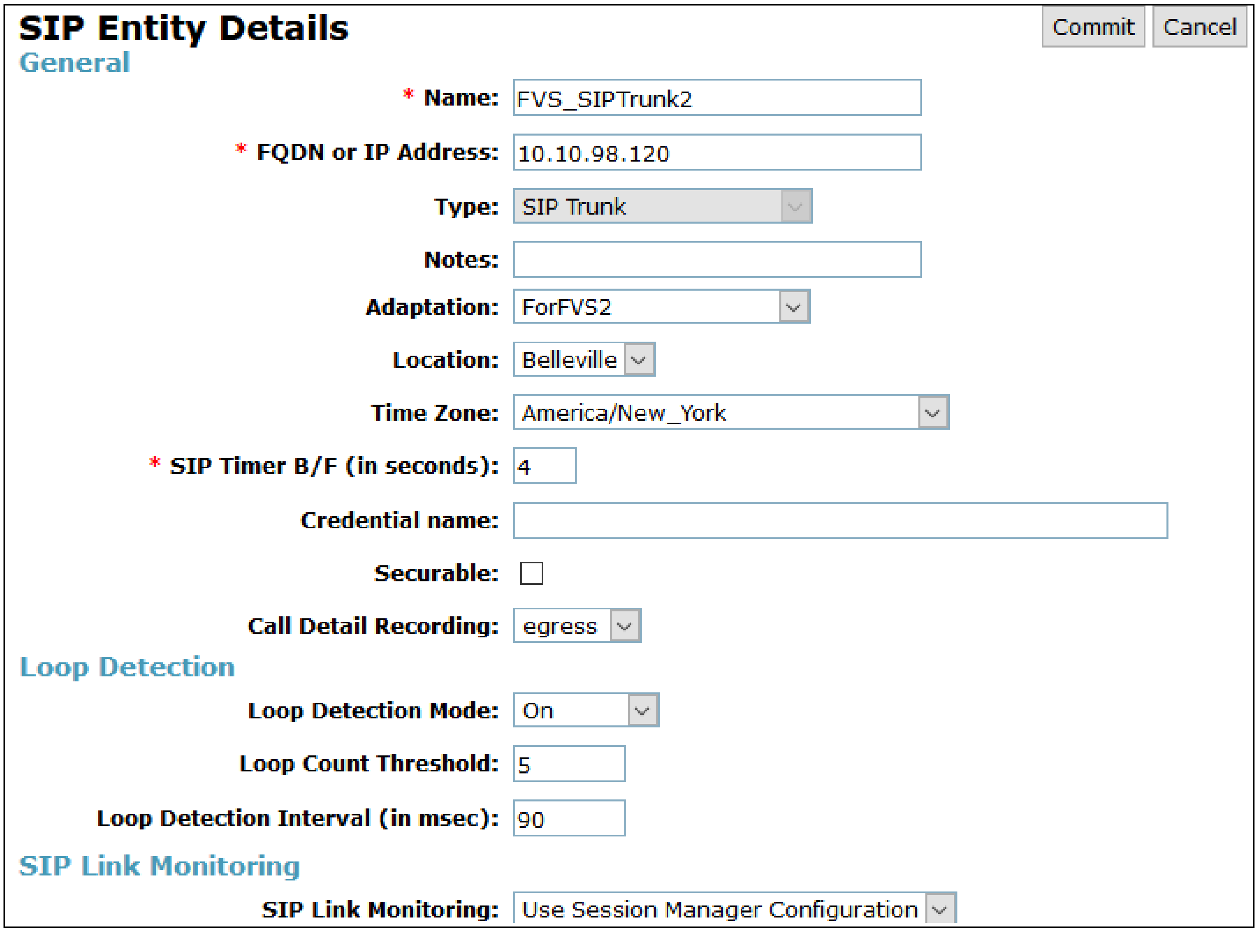
Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

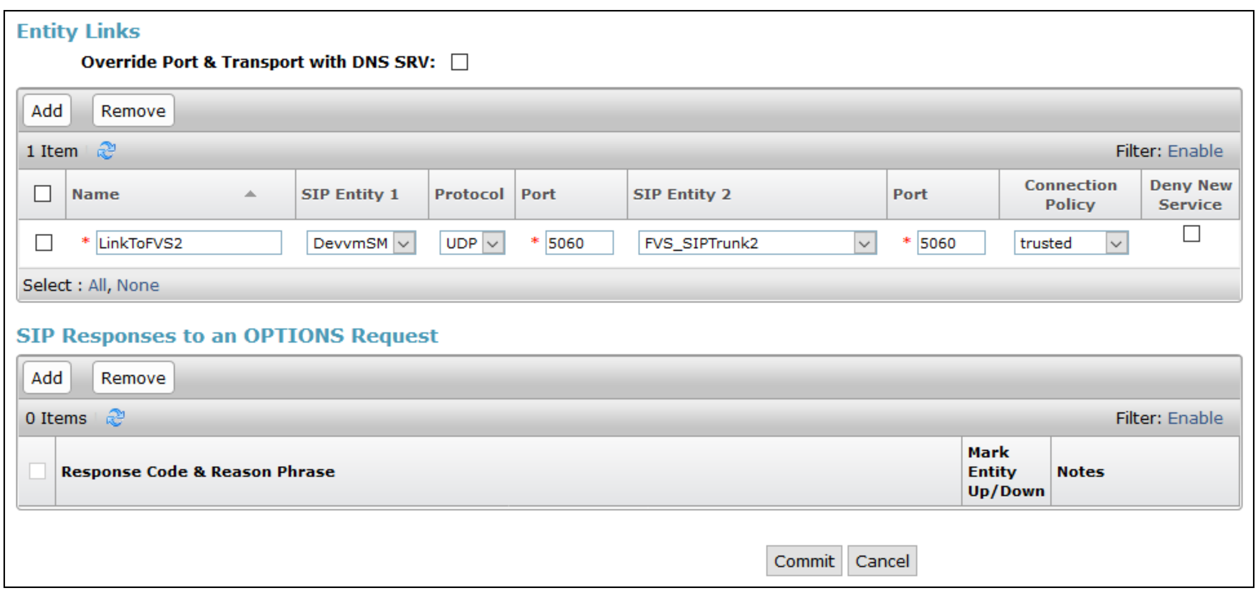
- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DevvmSM".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The FVS entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that FVS can support both UDP and TCP and the compliance testing used the UDP protocol.



Repeat same step for FVS2, below is the screenshot for FVS2 SIP Entity and Entity Link:



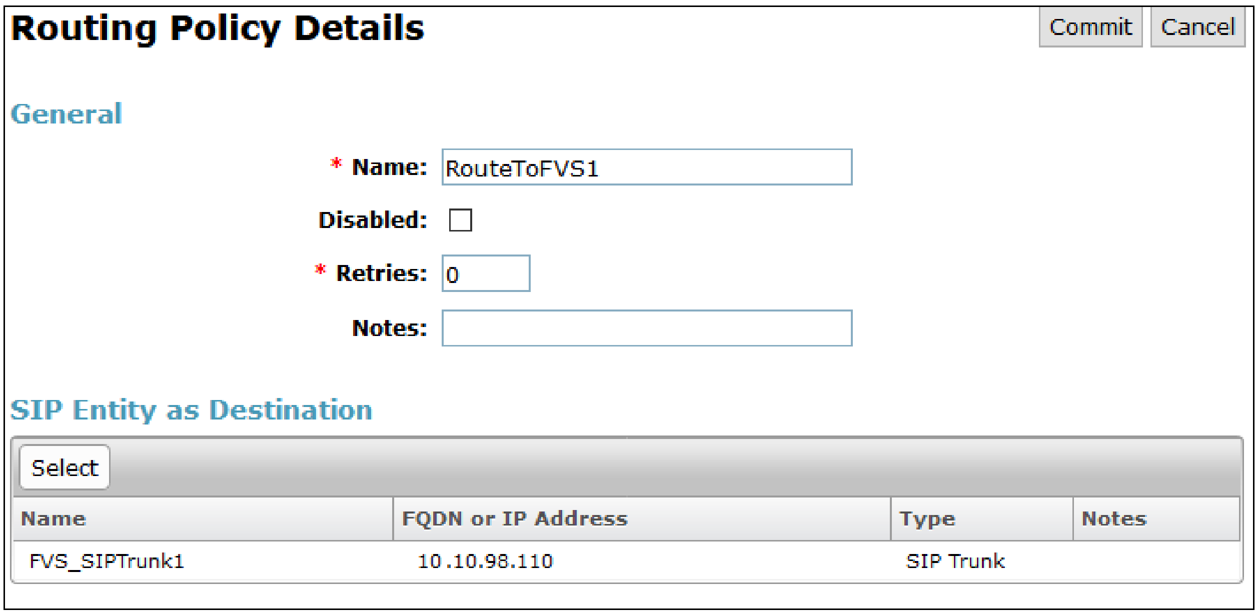


6.6. Administer Routing Policies

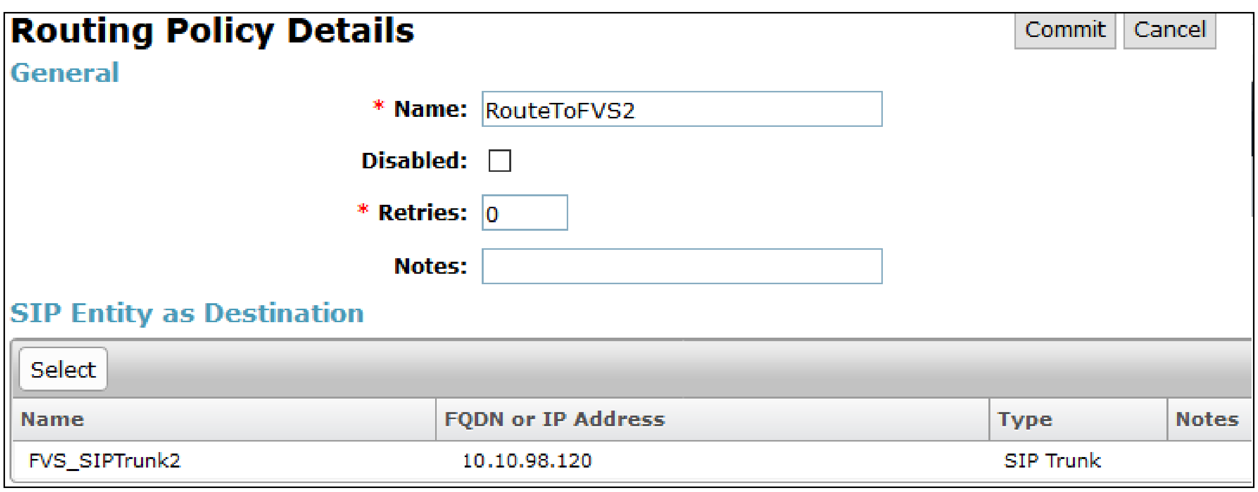
Add two new routing policies, one for FVS and one for the new SIP trunks with Communication Manager. Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for FVS.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the FVS entity name from **Section 6.5**. The screen below shows the result of the selection.



Repeat the same step for FVS2:

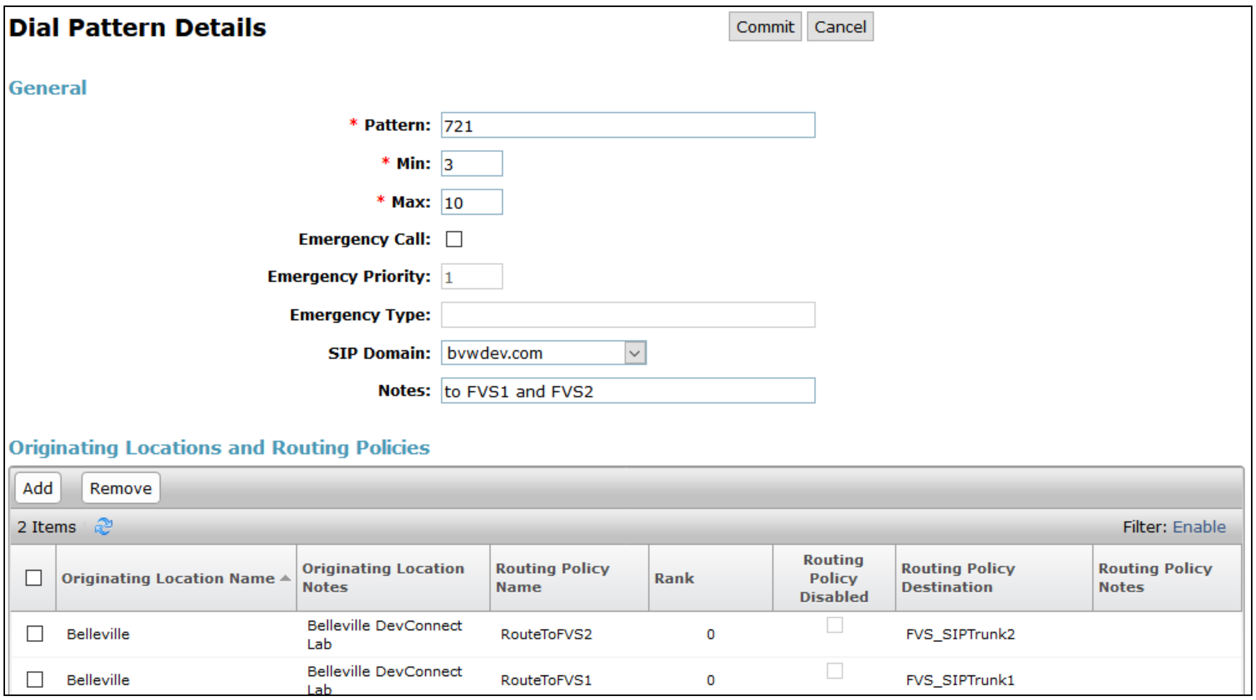


6.7. Administer Dial Patterns

Same dial pattern will be created for FVS1 and FVS2. Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach FVS. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

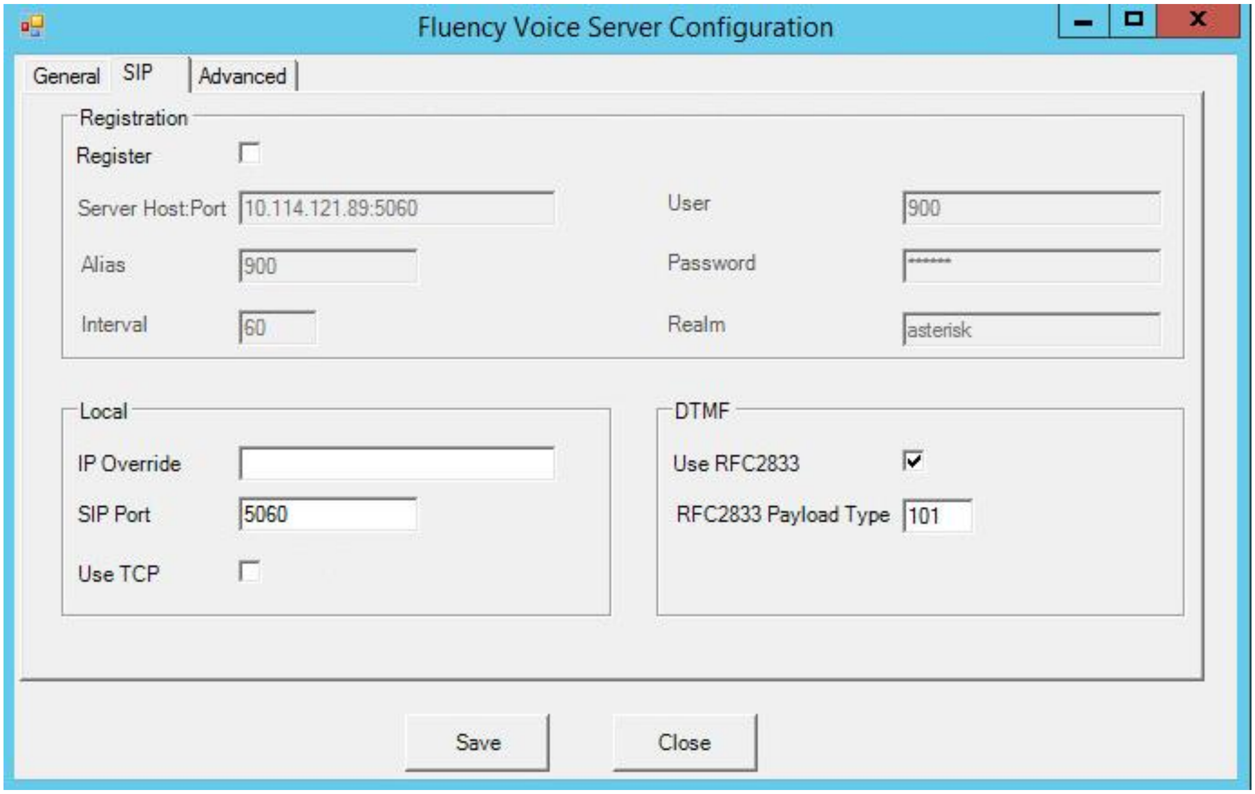
- **Pattern:** A dial pattern to match, in this case “721”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 5.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching FVS1 and FVS2. In the compliance testing, FVS routing policies from **Section 6.6** were selected as shown below.



7. Configure MModal FVS

This section provides the procedures for configuring FVS. It is assume that FVS already installed and operational. The procedures include configure DTMF use RFC2833. On FVS, launch **Fluency Voice Server Configuration** application, in **SIP** tab verify option **Use RFC2833** is checked as displayed below:



8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and FVS.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 5

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0052/001 T00146   in-service/idle    no
0052/002 T00147   in-service/idle    no
0052/003 T00148   in-service/idle    no
0052/004 T00149   in-service/idle    no
0052/005 T00150   in-service/idle    no
0052/006 T00151   in-service/idle    no
0052/007 T00152   in-service/idle    no
0052/008 T00153   in-service/idle    no
0052/009 T00154   in-service/idle    no
0052/010 T00155   in-service/idle    no
```

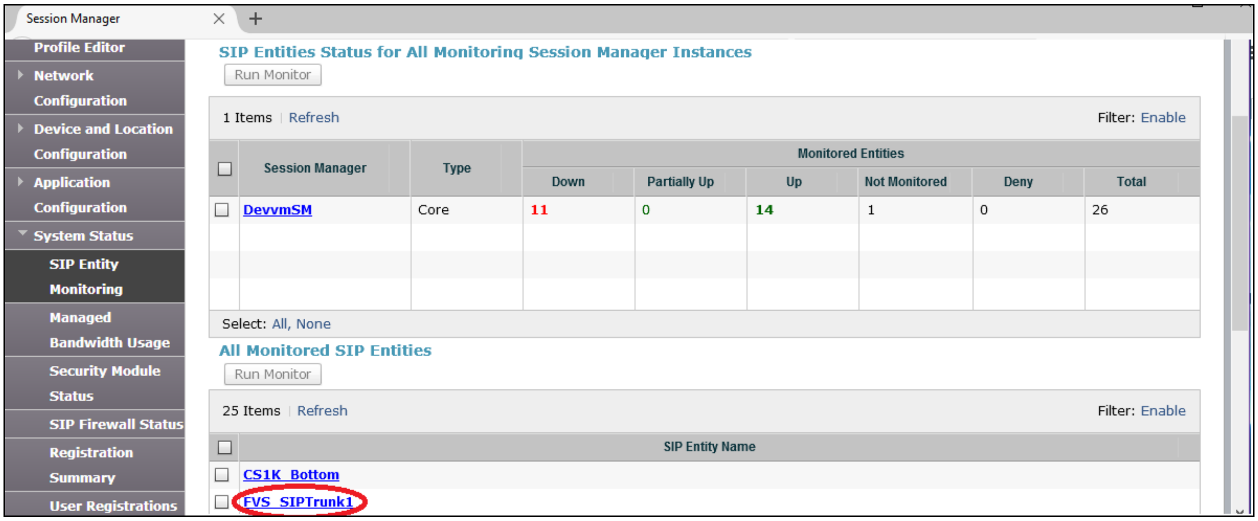
Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.2**. Verify that the **Group State** is "in-service", as shown below.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

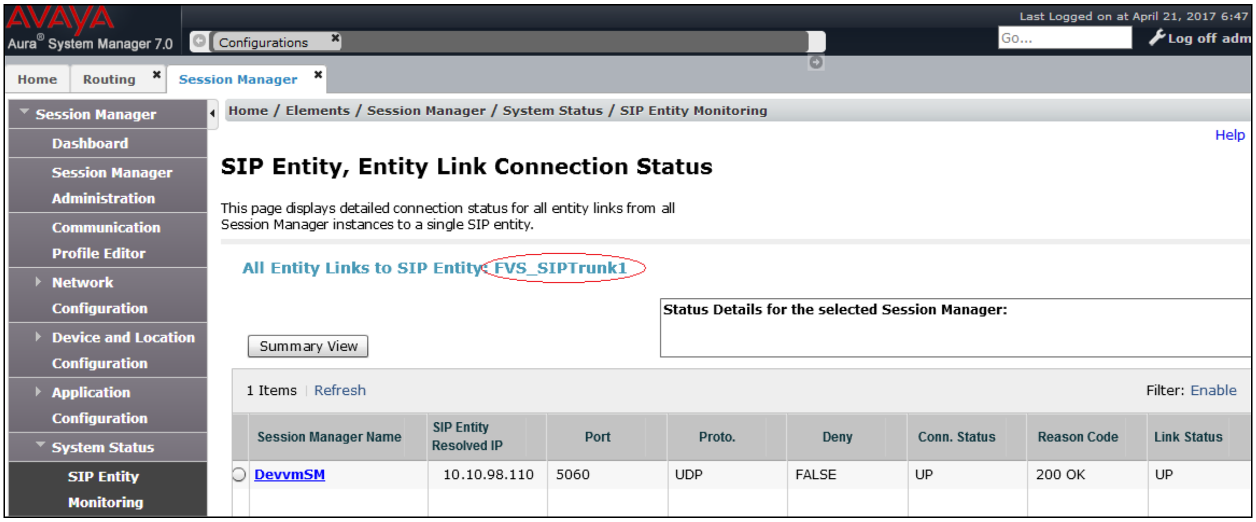
8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown).

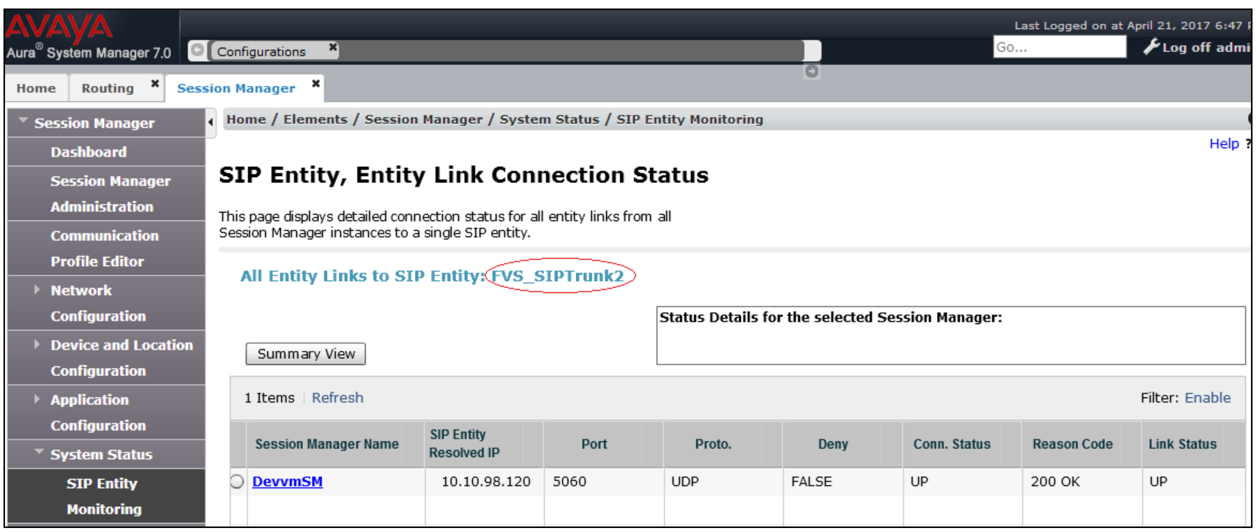
Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the FVS entity name from **Section 6.5**.



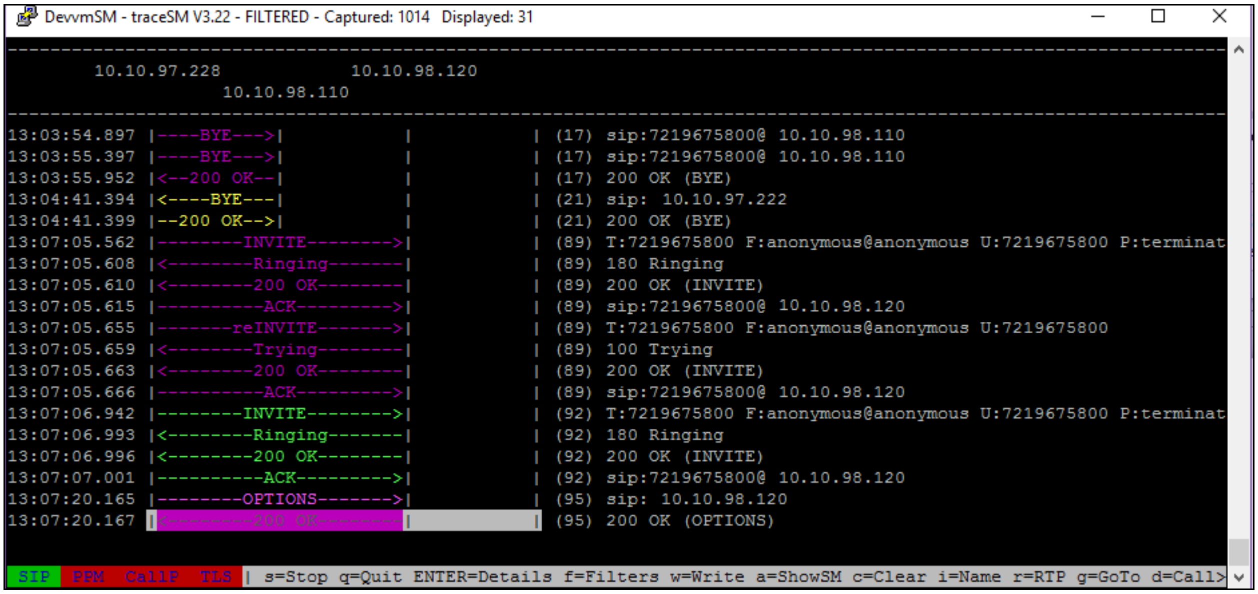
The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "UP", as shown below.



Repeat the same step for FVS2:

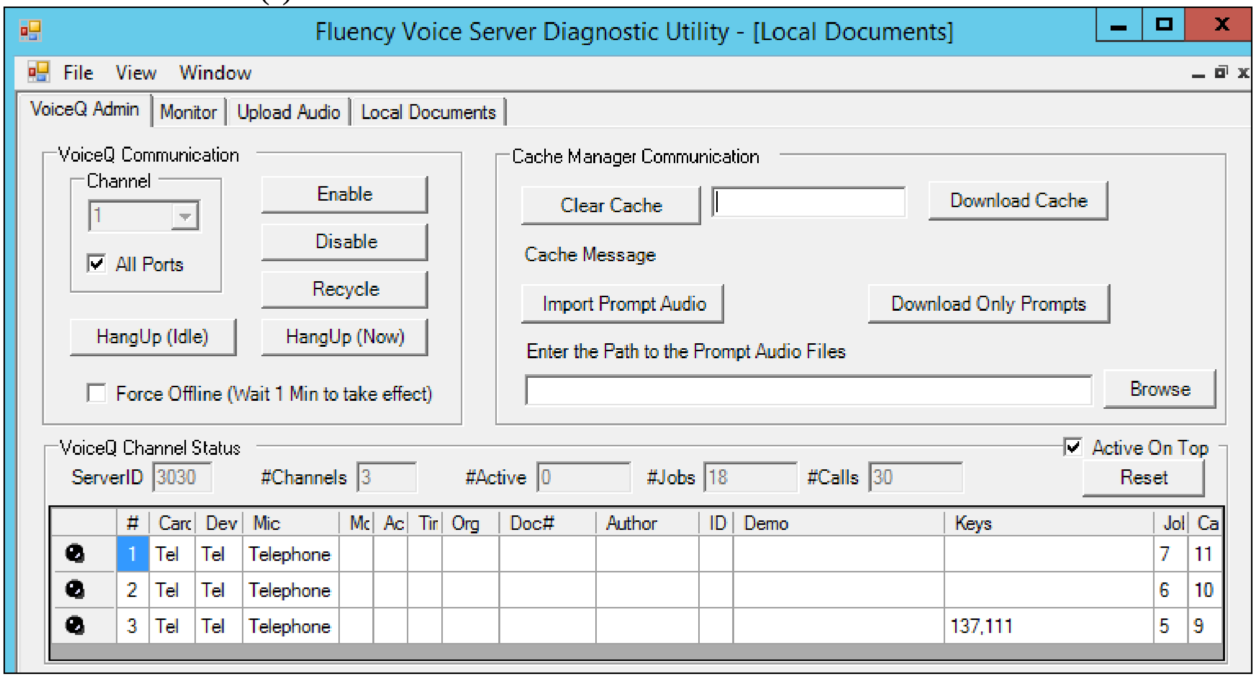


Make two phones call to FVS as mention in **Section 8.3**, verify the traces display detail of the calls as shown below (last 2 calls):

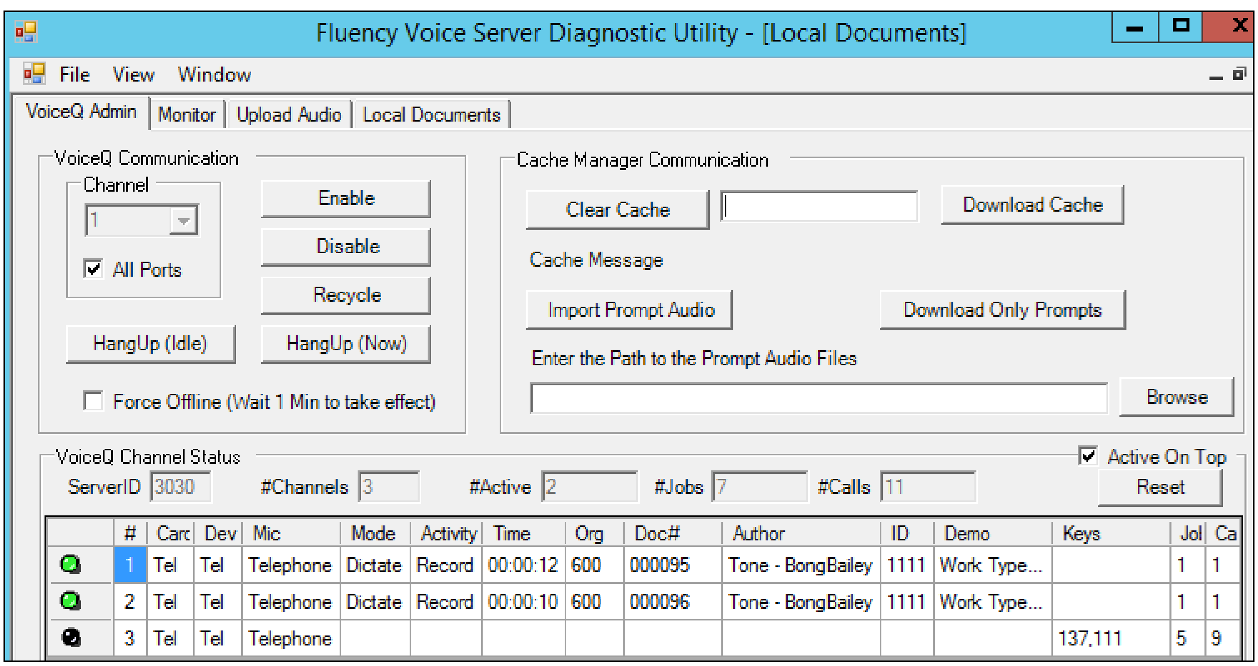


8.3. Verify MModal FVS

On Fluency Voice Server launch Fluency Voice Server Diagnostic Utility verify all voice channels are Enable(s) as shown below:



Make couples phone calls to FVS, for example, two call were made and connected to FVS2, below is the status of 2 voice channels are in the call:



Please reference back to **Section 8.2** for trace log of two calls on Session Manager.

9. Conclusion

These Application Notes describe the configuration steps required for MModal FVS to successfully interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. 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.

1. *Administering Avaya Aura® Communication Manager*, Release 7.0.1 555-245-205 Issue 3 October 2016.
2. *Administering Avaya Aura® Session Manager*, Release 7.0.1 Issue 2 May 2016.
3. *Administering Avaya Aura® System Manager*, Release 7.0.1

MModal document available upon request.

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.