



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.

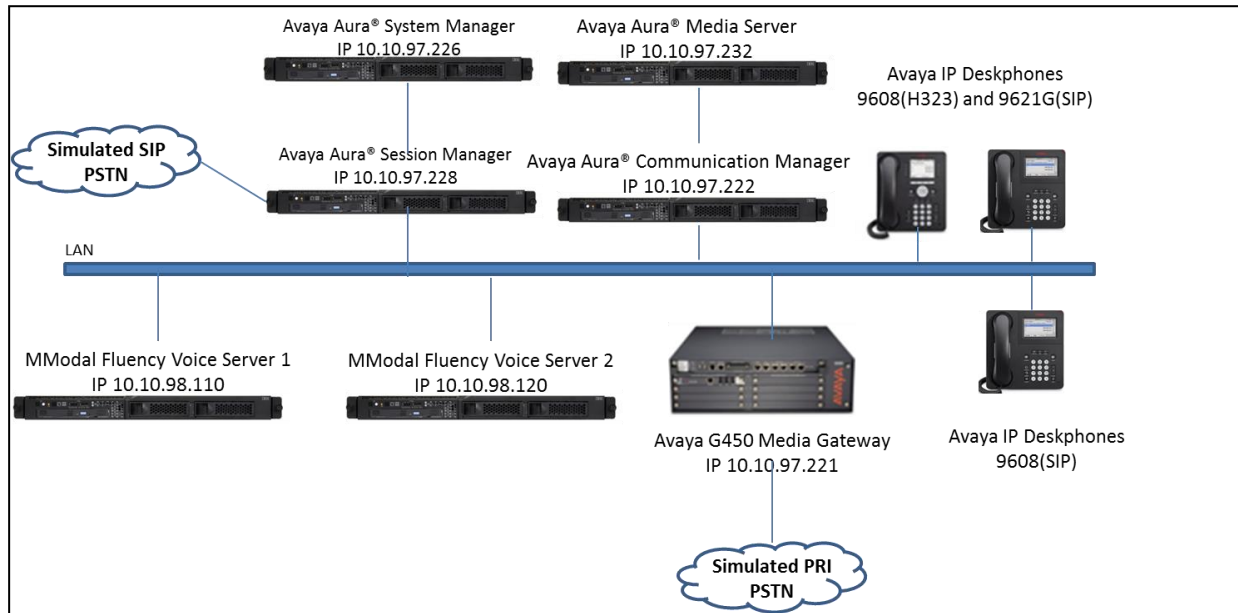


Figure 1: Compliance Testing Configuration

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

Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload             RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3    Direct IP-IP Audio Connections? n
Enable Layer 3 Test? y                IP Audio Hairpinning? 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 Pattern	Len	Del	Insert Digits	Net	Conv	Node 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						Percent Full: 2		
Location: all								
Dialed String	Total Min Max		Route Pattern	Call Type	Node Num	ANI 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.

AVAYA
Aura® System Manager 7.0

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

User ID:

Password:

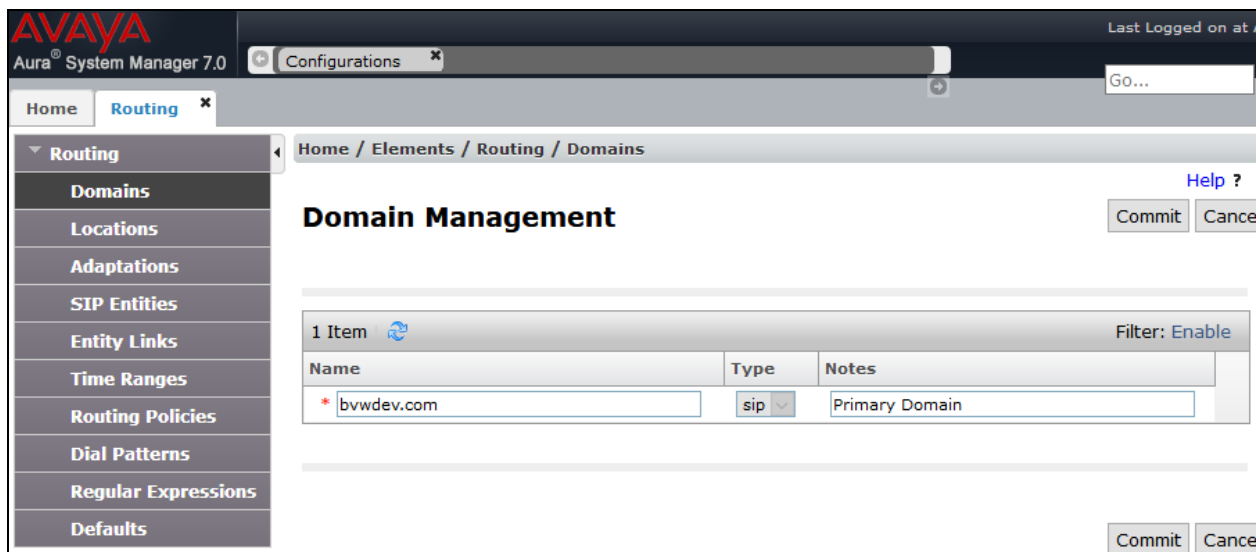
[Change Password](#)

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.

AVAYA
Aura® System Manager 7.0

Configurations

Last Logged on at April 2

Home / Elements / Routing / Locations

Location Details

General

* Name: Belleville

Notes: Belleville DevConnect Lab

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Commit Cancel

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.

* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

4 Items Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.33.5.*	Phones and Servers on private lab network
<input type="checkbox"/>	* 10.10.97.*	Lab PBX
<input type="checkbox"/>	* 10.10.98.*	
<input type="checkbox"/>	* 172.29.187.*	opentrade

Select : All, None

Commit Cancel

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.

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Aura® System Manager 7.0

Configurations

Last Logged on at April 18, 2017 12:00

Go... Log off admin

Home Routing

Home / Elements / Routing / Adaptations

Adaptation Details

Commit Cancel

General

* Adaptation Name: ForFVS1

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

Name	Value
iodstd	bvwdev.com
ioscrd	bvwdev.com
odstd	10.10.98.110

Select : All, None

Repeat the same step for FVS2 as display below:

Home / Elements / Routing / Adaptations

Adaptation Details

Commit Cancel

General

* Adaptation Name: ForFVS2

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

	Name	Value
<input type="checkbox"/>	iodstd	bvwdev.com
<input type="checkbox"/>	ioscrd	bvwdev.com
<input type="checkbox"/>	odstd	10.10.98.120

Select : All, None

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.

Domains	<div><h3>SIP Entity Details</h3><div>Commit Cancel</div><div>General<div><div>* Name: FVS_SIPTrunk1</div><div>* FQDN or IP Address: 10.10.98.110</div><div>Type: SIP Trunk</div><div>Notes:</div><div>Adaptation: ForFVS1</div><div>Location: Belleville</div><div>Time Zone: America/New_York</div><div>* SIP Timer B/F (in seconds): 4</div><div>Credential name:</div><div>Securable: <input type="checkbox"/></div><div>Call Detail Recording: egress</div></div><div>Loop Detection<div>Loop Detection Mode: On</div><div>Loop Count Threshold: 5</div><div>Loop Detection Interval (in msec): 90</div></div><div>SIP Link Monitoring<div>SIP Link Monitoring: Use Session Manager Configuration</div></div></div></div>
Locations	
Adaptations	
SIP Entities	
Entity Links	
Time Ranges	
Routing Policies	
Dial Patterns	
Regular Expressions	
Defaults	

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.

Entity Links

Override Port & Transport with DNS SRV: ☐

Add Remove

1 Item
Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* LinkToFVS1	DevvmSM	UDP	* 5060	FVS_SIPTrunk1	* 5060	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items
Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

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

SIP Entity Details

General

Commit

Cancel

* Name:

FVS_SIPTrunk2

* FQDN or IP Address:

10.10.98.120

Type:

SIP Trunk

Notes:

Adaptation:

ForFVS2

Location:

Belleville

Time Zone:

America/New_York

* SIP Timer B/F (in seconds):

4

Credential name:

Securable:

☐

Call Detail Recording:

egress

Loop Detection

Loop Detection Mode:

On

Loop Count Threshold:

5

Loop Detection Interval (in msec):

90

SIP Link Monitoring

SIP Link Monitoring:

Use Session Manager Configuration

Entity Links

Override Port & Transport with DNS SRV: ☐

Add

Remove

1 Item

Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* LinkToFVS2	DevvmSM	UDP	* 5060	FVS_SIPTrunk2	* 5060	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add

Remove

0 Items

Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit

Cancel

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.

Routing Policy DetailsCommitCancel

General

*** Name:**

Disabled: ☐

*** Retries:**

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
FVS_SIPTrunk1	10.10.98.110	SIP Trunk	

Repeat the same step for FVS2:

Routing Policy DetailsCommitCancel

General

*** Name:**

Disabled: ☐

*** Retries:**

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
FVS_SIPTrunk2	10.10.98.120	SIP Trunk	

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.

Dial Pattern DetailsCommitCancel

General

* Pattern:

721

* Min:

3

* Max:

10

Emergency Call:

☐

Emergency Priority:

1

Emergency Type:

SIP Domain:

bvwdev.com

Notes:

to FVS1 and FVS2

Originating Locations and Routing Policies

AddRemove

2 Items Filter: Enable

<input type="checkbox"/>	Originating Location Name ^	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab	RouteToFVS2	0	<input type="checkbox"/>	FVS_SIPTrunk2	
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab	RouteToFVS1	0	<input type="checkbox"/>	FVS_SIPTrunk1	

7. Configure MModal FVS

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

The screenshot shows the 'Fluency Voice Server Configuration' application window with the 'SIP' tab selected. The window has three tabs: 'General', 'SIP', and 'Advanced'. The 'SIP' tab contains two main sections: 'Registration' and 'Local'. The 'Registration' section includes fields for 'Register' (unchecked), 'Server Host:Port' (10.114.121.89:5060), 'User' (900), 'Alias' (900), 'Interval' (60), 'Password' (masked with asterisks), and 'Realm' (asterisk). The 'Local' section includes fields for 'IP Override' (empty), 'SIP Port' (5060), and 'Use TCP' (unchecked). To the right of the 'Local' section is a 'DTMF' section with 'Use RFC2833' checked and 'RFC2833 Payload Type' set to 101. At the bottom of the window are 'Save' and 'Close' buttons.

Section	Field	Value
Registration	Register	<input type="checkbox"/>
	Server Host:Port	10.114.121.89:5060
	User	900
	Alias	900
	Interval	60
	Password	*****
	Realm	asterisk
Local	IP Override	
	SIP Port	5060
	Use TCP	<input type="checkbox"/>
	DTMF	
Use RFC2833		<input checked="" type="checkbox"/>
RFC2833 Payload Type		101

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**.

SIP Entities Status for All Monitoring Session Manager Instances

Run Monitor

1 Items | Refresh Filter: Enable

Session Manager	Type	Monitored Entities					
		Down	Partially Up	Up	Not Monitored	Deny	Total
<input type="checkbox"/> DevvmsM	Core	11	0	14	1	0	26

Select: All, None

All Monitored SIP Entities

Run Monitor

25 Items | Refresh Filter: Enable

SIP Entity Name
<input type="checkbox"/> CS1K_Bottom
<input type="checkbox"/> FVS_SIPTrunk1

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

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes 'Home', 'Routing', and 'Session Manager'. The left sidebar contains a tree view with 'Session Manager' expanded, showing sub-items like 'Dashboard', 'Session Manager Administration', 'Communication Profile Editor', 'Network Configuration', 'Device and Location Configuration', 'Application Configuration', and 'System Status'. The 'System Status' section is further expanded to show 'SIP Entity Monitoring'. The main content area is titled 'SIP Entity, Entity Link Connection Status' and includes a breadcrumb trail: 'Home / Elements / Session Manager / System Status / SIP Entity Monitoring'. Below the title, there is a description: 'This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.' A link 'All Entity Links to SIP Entity: FVS_SIPTrunk1' is displayed, with 'FVS_SIPTrunk1' circled in red. A 'Summary View' button is present. Below this, a table shows the connection status for one item, 'DevvmSM'. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Deny, Conn. Status, Reason Code, and Link Status. The 'Conn. Status' and 'Link Status' for 'DevvmSM' are both 'UP'.

Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
DevvmSM	10.10.98.110	5060	UDP	FALSE	UP	200 OK	UP

Repeat the same step for FVS2:

The screenshot shows the Avaya Aura System Manager 7.0 interface, similar to the previous one. The main content area is titled 'SIP Entity, Entity Link Connection Status'. The link 'All Entity Links to SIP Entity: FVS_SIPTrunk2' is displayed, with 'FVS_SIPTrunk2' circled in red. The table shows the connection status for one item, 'DevvmSM'. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Deny, Conn. Status, Reason Code, and Link Status. The 'Conn. Status' and 'Link Status' for 'DevvmSM' are both 'UP'.

Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
DevvmSM	10.10.98.120	5060	UDP	FALSE	UP	200 OK	UP

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):

```

DevvmSM - traceSM V3.22 - FILTERED - Captured: 1014 Displayed: 31
-----
10.10.97.228      10.10.98.120
10.10.98.110
-----
13:03:54.897 |----BYE---->| | (17) sip:7219675800@ 10.10.98.110
13:03:55.397 |----BYE---->| | (17) sip:7219675800@ 10.10.98.110
13:03:55.952 |<--200 OK--| | (17) 200 OK (BYE)
13:04:41.394 |<----BYE----| | (21) sip: 10.10.97.222
13:04:41.399 |--200 OK-->| | (21) 200 OK (BYE)
13:07:05.562 |-----INVITE----->| | (89) T:7219675800 F:anonymous@anonymous U:7219675800 P:terminat
13:07:05.608 |<-----Ringing-----| | (89) 180 Ringing
13:07:05.610 |<-----200 OK-----| | (89) 200 OK (INVITE)
13:07:05.615 |-----ACK----->| | (89) sip:7219675800@ 10.10.98.120
13:07:05.655 |-----reINVITE----->| | (89) T:7219675800 F:anonymous@anonymous U:7219675800
13:07:05.659 |<-----Trying-----| | (89) 100 Trying
13:07:05.663 |<-----200 OK-----| | (89) 200 OK (INVITE)
13:07:05.666 |-----ACK----->| | (89) sip:7219675800@ 10.10.98.120
13:07:06.942 |-----INVITE----->| | (92) T:7219675800 F:anonymous@anonymous U:7219675800 P:terminat
13:07:06.993 |<-----Ringing-----| | (92) 180 Ringing
13:07:06.996 |<-----200 OK-----| | (92) 200 OK (INVITE)
13:07:07.001 |-----ACK----->| | (92) sip:7219675800@ 10.10.98.120
13:07:20.165 |-----OPTIONS----->| | (95) sip: 10.10.98.120
13:07:20.167 |<-----200 OK-----| | (95) 200 OK (OPTIONS)
-----
SIP ERR CallE TLS | s=Stop q=Quit ENTER=Details f=Filters w=Write a=ShowSM c=Clear i=Name r=RTP g=GoTo d=Call>

```

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:

Fluency Voice Server Diagnostic Utility - [Local Documents]

File View Window

VoiceQ Admin Monitor Upload Audio Local Documents

VoiceQ Communication

Channel: 1

☒ All Ports

Enable Disable Recycle

HangUp (Idle) HangUp (Now)

☐ Force Offline (Wait 1 Min to take effect)

Cache Manager Communication

Clear Cache [] Download Cache

Cache Message

Import Prompt Audio Download Only Prompts

Enter the Path to the Prompt Audio Files

[] Browse

VoiceQ Channel Status

ServerID 3030 #Channels 3 #Active 0 #Jobs 18 #Calls 30 ☒ Active On Top Reset

	#	Carc	Dev	Mic	Mc	Ac	Tir	Org	Doc#	Author	ID	Demo	Keys	Jol	Ca
1	1	Tel	Tel	Telephone										7	11
2	2	Tel	Tel	Telephone										6	10
3	3	Tel	Tel	Telephone									137,111	5	9

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:

Fluency Voice Server Diagnostic Utility - [Local Documents]

File View Window

VoiceQ Admin Monitor Upload Audio Local Documents

VoiceQ Communication

Channel: 1

☒ All Ports

Enable Disable Recycle

HangUp (Idle) HangUp (Now)

☐ Force Offline (Wait 1 Min to take effect)

Cache Manager Communication

Clear Cache [] Download Cache

Cache Message

Import Prompt Audio Download Only Prompts

Enter the Path to the Prompt Audio Files

[] Browse

VoiceQ Channel Status

ServerID 3030 #Channels 3 #Active 2 #Jobs 7 #Calls 11 ☒ Active On Top Reset

	#	Carc	Dev	Mic	Mode	Activity	Time	Org	Doc#	Author	ID	Demo	Keys	Jol	Ca
1	1	Tel	Tel	Telephone	Dictate	Record	00:00:12	600	000095	Tone - BongBailey	1111	Work Type...		1	1
2	2	Tel	Tel	Telephone	Dictate	Record	00:00:10	600	000096	Tone - BongBailey	1111	Work Type...		1	1
3	3	Tel	Tel	Telephone									137,111	5	9

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.

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