



DevConnect Program

Application Notes for Parlance Service Version 9 with Avaya Aura® Session Manager Release 10.1 and Avaya Aura® Communication Manager Release 10.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Parlance Service to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Parlance Service is a SIP-based solution that provides speech recognition as a managed service.

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.

1. Introduction

These Application Notes describe the configuration steps required for Parlance Service to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Parlance Service delivers speech recognition as a managed service for leading health systems, higher education institutions, and enterprises, transforming old fashioned IVRs and automated attendants into easy-to-navigate speech-driven tools.

In the compliance testing, calls from internal and external callers were routed over SIP trunks via Session Manager to Parlance for IVR functions. The unsupervised transfer feature was accomplished by Parlance Service via use of SIP REFER.

2. General Test Approach and Test Results

The feature test cases were performed manually. A call is placed that routes to the external menu of Parlance Service through Session Manager via SIP trunk. The caller is connected to the Parlance Service and hears “Parlance Corporation. Please say the first and last name of the person or the name of the department you'd like to reach”. The caller says “James Smith”, the Parlance Service responds, “Connecting to James Smith, please hold or say cancel”. The call is then transferred to the local extension that is assigned to the contact “James Smith”. The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to the Parlance servers and/or busy out SIP trunk group from Communication Manager.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and Parlance did not include use of any specific encryption features as requested by Parlance. Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

- SIP OPTIONS.
- Detecting Speech Barge-in.
- Keep-Alive and session refresh timer.
- Telephony features: hold, abandoned call, voicemail, call forward, blind transfer, conditional transfer and attended transfer.
- DTMF RFC2833.
- DNIS and ANI matching.
- Parlance Service SIP trunk redundancy.

The serviceability testing focused on verifying the ability of Parlance to recover from adverse conditions, that consists of SIP trunk disconnect and SIP trunk redundancy.

2.2. Test Results

All test cases were executed and verified with one observation below.

- For attended transfer, Parlance Service uses the IP address and transport number in SIP URI from the Contact header of the incoming call to make the outbound call to connect to the destination transferee number. The Contact header in the INVITE message of the incoming call to the Parlance Service is routed by Session Manager and always contains the information of the caller, who originally initiates the call; in this case it always comes from Communication Manager. The regular expression adaptation in Session Manager is configured in **Section 6.3** and applied on the Parlance SIP entity to modify the Communication Manager IP address and the transport 5061 to the Session Manager IP address and the transport 5060.
- Parlance Service only supports codec G.711 U-law.

2.3. Support

Technical support on Parlance can be obtained through the following:

- **Web:** www.ParlanceCorp.com
- **Email:** customersupport@parlancecorp.com
- **Phone:** +1.888.700.6263 and say “Support”

3. Reference Configuration

As shown in **Figure 1** SIP trunks were used between the Parlance Service call servers and Session Manager. A 4-digit Uniform Dial Plan was used to facilitate dialing with Parlance Service. Calls to extensions 52xx were routed over the SIP trunks to Parlance. Internal users on Communication Manager will dial 5201 to reach Parlance internal menu and PSTN user will dial an assigned DID number that is routed to the dial pattern 5200 to reach the external menu of Parlance Service.

The detailed administration of connectivity between Communication Manager, Session Manager and Avaya SBCE are not the focus of these Application Notes and will not be described.

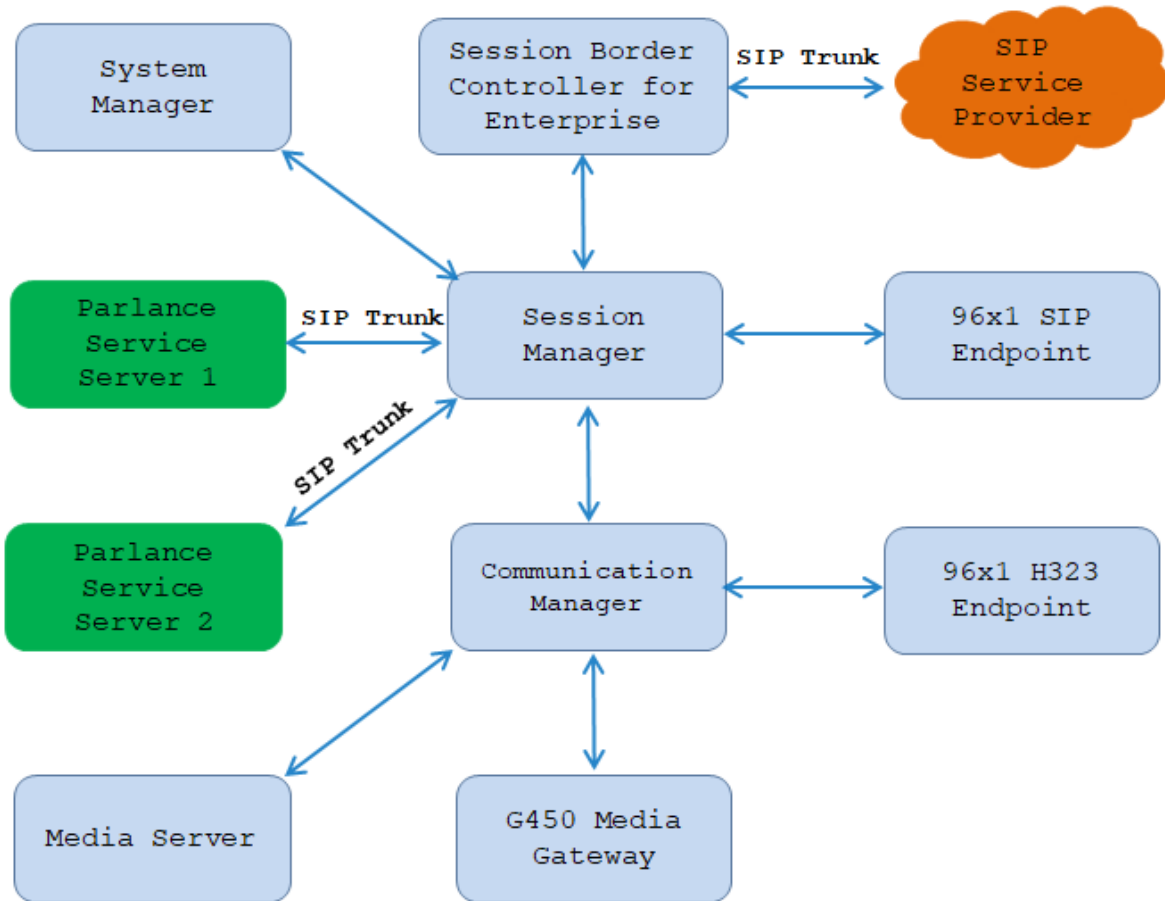


Figure 1: Compliance Testing Configuration

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 running on Virtual Environment	R020x.01.0.974.0 10.1.0.2.0.974.27607
Avaya Aura® Session Manager running on Virtual Environment	10.1.0.2.1010215
Avaya Aura® Media Server running on Virtual Environment	8.0.2.218
Avaya G450 Media Gateway	42.08
Avaya Aura® System Manager running on Virtual Environment	10.1.0.2 SP2 Software Update Revision No: 10.1.0.2.0715160 Hot Fix - 1010215160
Avaya Session Border Controller for Enterprise running on Virtual Environment	10.1.1.0-35-21872
Avaya IP Deskphones - J189 (H.323) - 9641GS (H.323) - 9611G (SIP)	6.8511 6.8511 7.1.9.0.8
Parlance Service Server running on Windows 2019	9

5. Configure Avaya Aura® Communication Manager

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration has already been performed. 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 private numbering
- Administer uniform dial plan
- Administer AAR analysis

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: 18000 4
    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 0
    Maximum Video Capable IP Softphones: 18000 0
    Maximum Administered SIP Trunks: 24000 30
    Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

5.2. Administer System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class of Restriction or Class of Service levels. Refer to [1] for more details.

```
change system-parameters features                               Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y
      Music/Tone on Hold: music Type: ext 1104
      Music (or Silence) on Transferred Trunk Calls? no
      DID/Tie/ISDN/SIP Intercept Treatment: attendant
      Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n

      Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
      Protocol for Caller ID Analog Terminals: Bellcore
      Display Calling Number for Room to Room Caller ID Calls? n
```

5.3. Administer SIP Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “52”. 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” in this case.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration with Parlance.
- **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 Parlance.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** enter “y”.

```
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? n
Peer Detection Enabled? n Peer Server: SM        Clustered? n
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: SM10
Near-end Listen Port: 5061                    Far-end Listen Port: 5061
                                                    Far-end Network Region: 1

Far-end Domain: avayalab.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
```


5.4. Administer SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “52”. 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 1                                     Page 1 of 22
                                     TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: Private Trunk                       COR: 1                 TN: 1           TAC: #01
  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: 14
```

Navigate to **Page 3** and enter “private” for **Numbering Format**.

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

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

                                                Hold/Unhold Notifications? y
  Modify Tandem Calling Number: no

  Send UCID? y

  Show ANSWERED BY on Display? y
```

5.6. 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 5.3**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “yes” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Parlance.

```

change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION
Region: 1              NR Group: 1
Location: 1           Authoritative Domain: avayalab.com
Name: Loc-1          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? 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
  
```

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

```

change ip-network-region 1                                     Page 4 of 20

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

5.7. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.5**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Parlance supports only codec G.711 U-Law. The compliance testing only covered the **G.711MU** codec.

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

                                IP MEDIA PARAMETERS

Codec Set: 1

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

5.8. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an available route pattern number to be used to reach Parlance, in this case “1”. 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 5.4**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 1 Page 1 of 3

                                Pattern Number: 1      Pattern Name: SIP-TLS-To-SM
SCCAN? n      Secure SIP? n      Used for SIP stations? n

Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/  IXC
No          Mrk Lmt List Del  Digits           QSIG
                                Dgts              Intw
1: 1      0
2:
3:
4:
5:
6:

                                BCC VALUE  TSC  CA-TSC  ITC  BCIE  Service/Feature  PARM  Sub  Numbering  LAR
                                0 1 2 M 4 W      Request          Dgts  Format
1: y y y y y n  n          rest          lev0-pvt  next
2: y y y y y n  n          rest          none
3: y y y y y n  n          rest          none
```

5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to Parlance. Add an entry for the trunk group defined in **Section 5.4**. In the example shown below, all calls originating from a 4-digit extension beginning with **33** and **34** routed to trunk group 1 will result in a 4-digit calling number. The calling party number will be in the SIP “From” header.

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

Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len
4	33	1		4
4	34	1		4

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits **52xx** to Parlance. 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 52xx, 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
52	4	0		aar	n

5.11. Administer AAR Analysis

Use the “change aar analysis 52” command and add an entry to specify how to route calls to 52xx. In the example shown below, calls with digits 52xx will be routed as an AAR call using route pattern “52” from **Section 5.8**.

```
change aar analysis 52                                       Page 1 of 2
                                AAR DIGIT ANALYSIS TABLE
                                Location: all
                                Percent Full: 2
```

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
52	4	4	1	aar		n

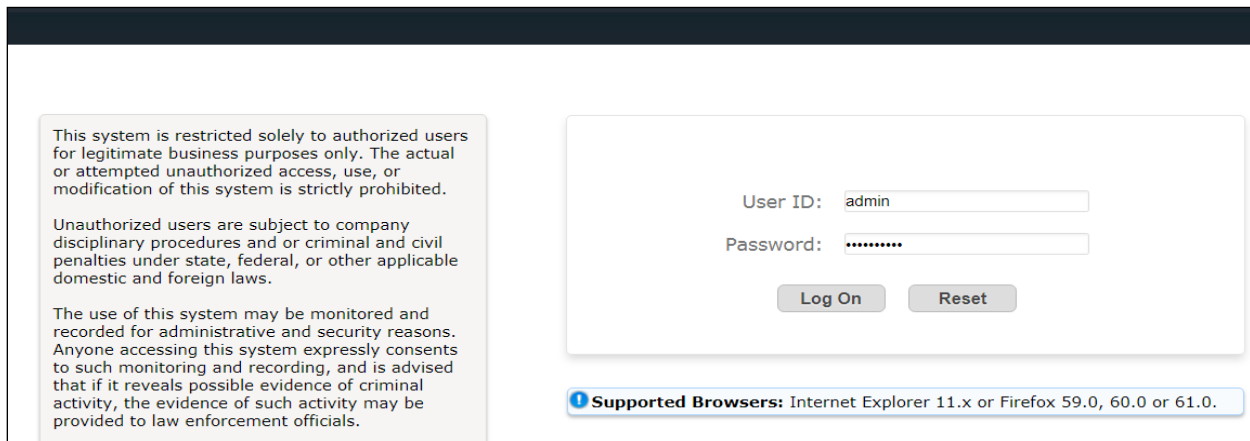
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer locations
- Administer Regular Expression Adaptation
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

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.



This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

User ID: admin

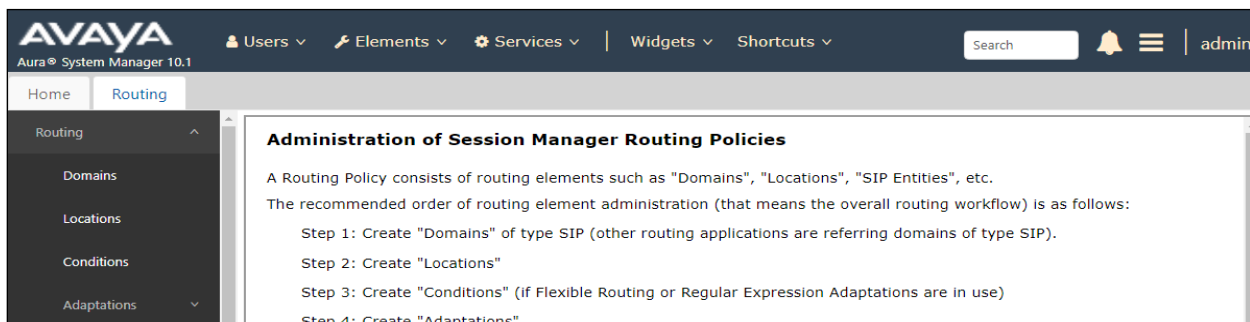
Password: *****

Log On Reset

Supported Browsers: Internet Explorer 11.x or Firefox 59.0, 60.0 or 61.0.

6.2. Administer Locations

In the subsequent screen (not shown), select **Elements** → **Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing** → **Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a new location for Parlance.



AVAYA Aura System Manager 10.1

Users Elements Services Widgets Shortcuts

Search admin

Home Routing

Routing

- Domains
- Locations
- Conditions
- Adaptations

Administration of Session Manager Routing Policies

A Routing Policy consists of routing elements such as "Domains", "Locations", "SIP Entities", etc.

The recommended order of routing element administration (that means the overall routing workflow) is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Conditions" (if Flexible Routing or Regular Expression Adaptations are in use)
- Step 4: Create "Adaptations"

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 10.1

Users | Elements | Services | Widgets | Shortcuts | Search | adm

Home | Routing

Location Details [Commit] [Cancel] [Help ?]

General

* **Name:** Parlance

Notes: Parlance Location

Dial Plan Transparency in Survivable Mode

Enabled:

Listed Directory Number: []

Associated CM SIP Entity: []

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

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

[Add] [Remove]

2 Items [Filter: Enable]

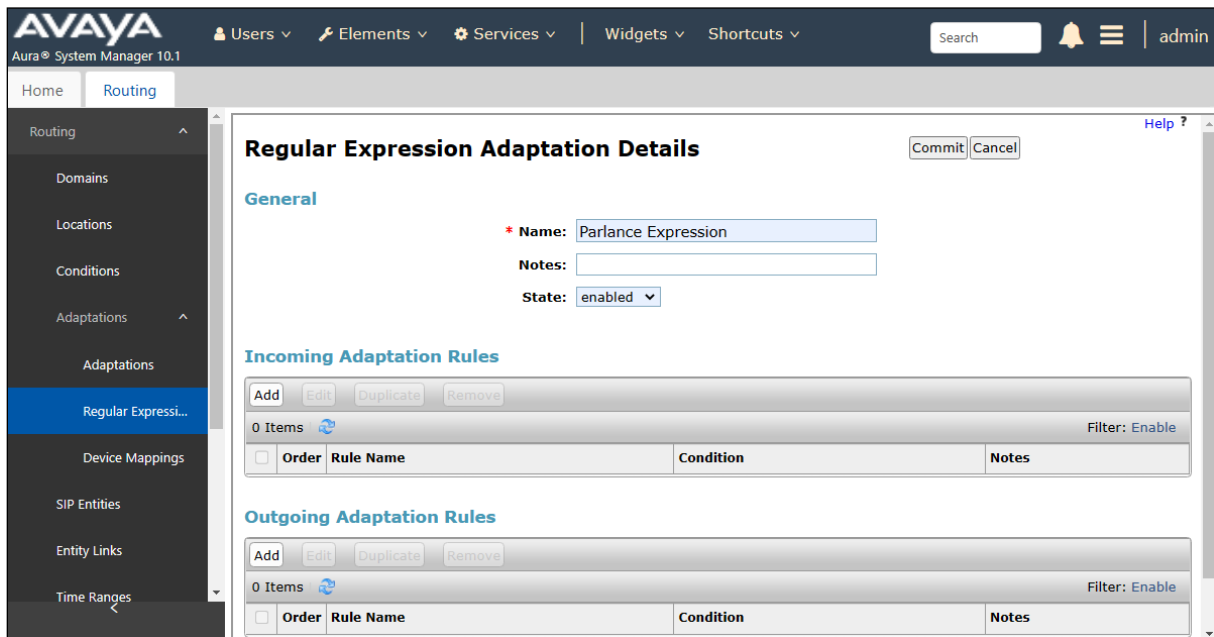
<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.33.1.62	Parlance Server 1
<input type="checkbox"/>	* 10.33.1.63	Parlance Server 2

Select : All, None

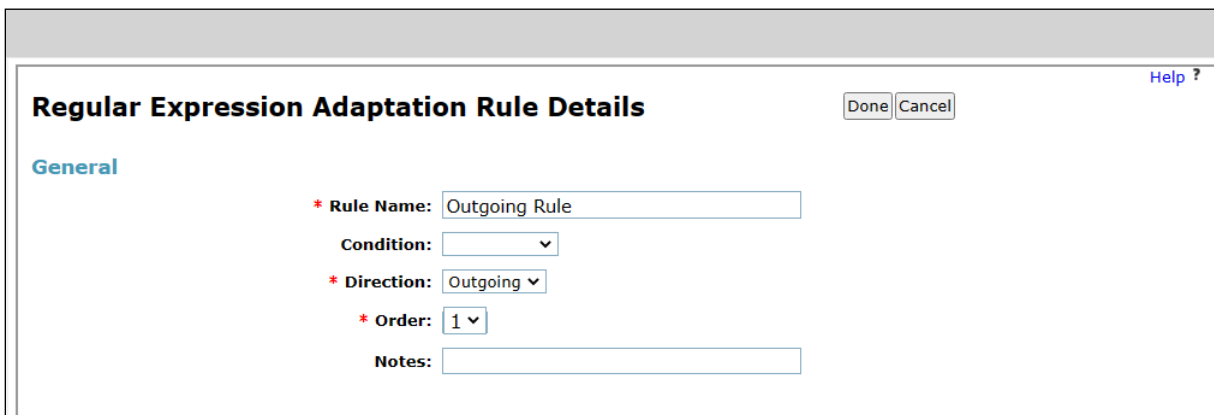
6.3. Administer Regular Expression Adaption

Session Manager can be configured to use Adaptation Modules to modify the Contact header of the incoming INVITE message sent from Session Manager to the Parlance Service server. See the observation noted in **Section 2.2** for more information.

To create a regular expression adaptation, navigate to **Routing → Adaptations → Regular Expression Adaptations**. The **Regular Expression Adaptations** displays in the right-hand side of the window, select **New** (not shown) to create a new regular expression adaptation. In the **Regular Expression Adaptation Details** page, enter a name in the **Name** field, select “**enabled**” in the **State** field. In the **Outgoing Adaptation Rules** section, select **Add** to add a new outgoing rule.



The **Regular Expression Adaptation Rule Details** page displays, enter a rule name in the **Rule Name** field, select “**Outgoing**” in the **Direction** field and leave other fields at default.



In the **Rule Actions** section, select **Add** to add a new rule as shown in the figure below. In this rule, the Contact header of the outgoing INVITE message from Session Manager to the Parlance Service server will be modified in the IP address and port of the SIP URI. For example, the SIP URI 3401@10.33.1.43:5061 will be modified to 3401@10.33.1.42:5060 as the INVITE message arrives at the Parlance Service server.

Rule Variables

Add Remove

0 Items Filter: Enable

	Variable Name	Source Type	Source	Instance	Match Expression	Notes
<input type="checkbox"/>						

Rule Actions

Add Remove

1 Item Filter: Enable

	Order	Source Type	Source	Instance	Operation	Match Expression	Replace / Add Ex
<input type="checkbox"/>	1	Header	Contact	any	modify	10.33.1.43:5061	10.33.1.42:5060

Select : All, None

6.4. Administer SIP Entities

In the SIP entity, add two new SIP entities for Parlance and one for Communication Manager.

6.4.1. SIP Entity for Parlance

Select **Routing** → **SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Parlance.

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

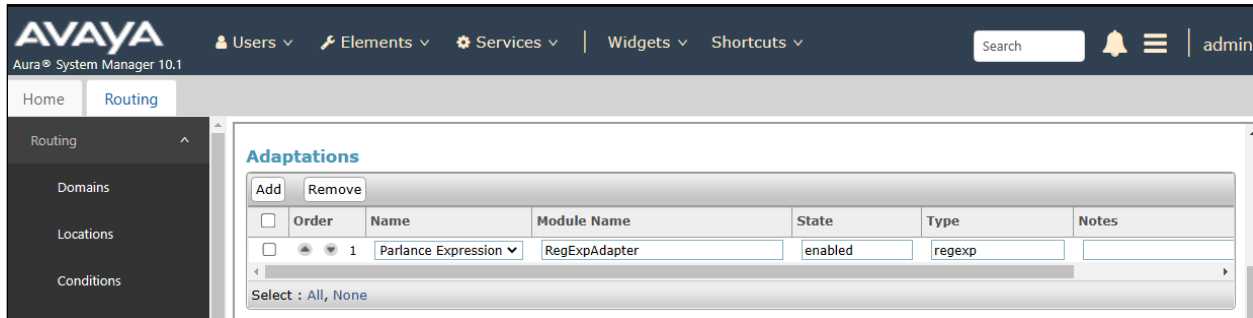
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Parlance server.
- **Type:** “SIP trunk”
- **Notes:** Any desired notes.
- **Location:** Select the Parlance location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile 'admi' are also visible. The left sidebar shows a navigation tree with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** ParlanceS1
- FQDN or IP Address:** 10.33.1.62
- Type:** SIP Trunk
- Notes:** (empty)
- Location:** Parlance
- Time Zone:** America/Denver
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- Securable:**
- Call Detail Recording:** none

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area.

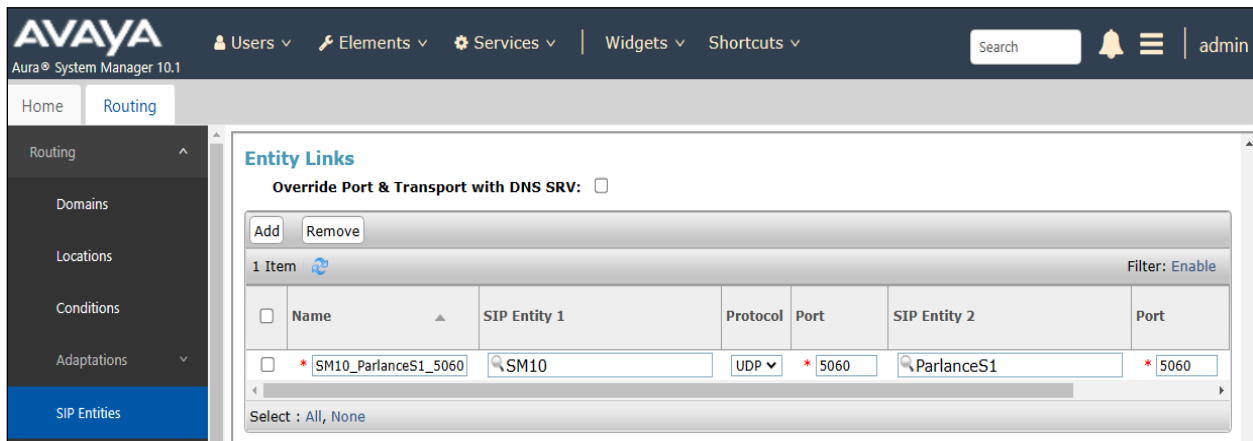
In the **Adaptations** section, select **Add** to add the Parlance Expression adaptation as configured in **Section 6.3**.



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 “SM10”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The Parlance entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that Parlance can support UDP and TCP, and the compliance testing used the UDP protocol.



Repeat the same procedures above to add second SIP entity **ParlanceS2** for another Parlance Service server.

6.4.2. SIP Entity for Communication Manager

Select **Routing** → **SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with Parlance.

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 an existing CLAN or the processor interface.
- **Type:** “CM”
- **Notes:** Any desired notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

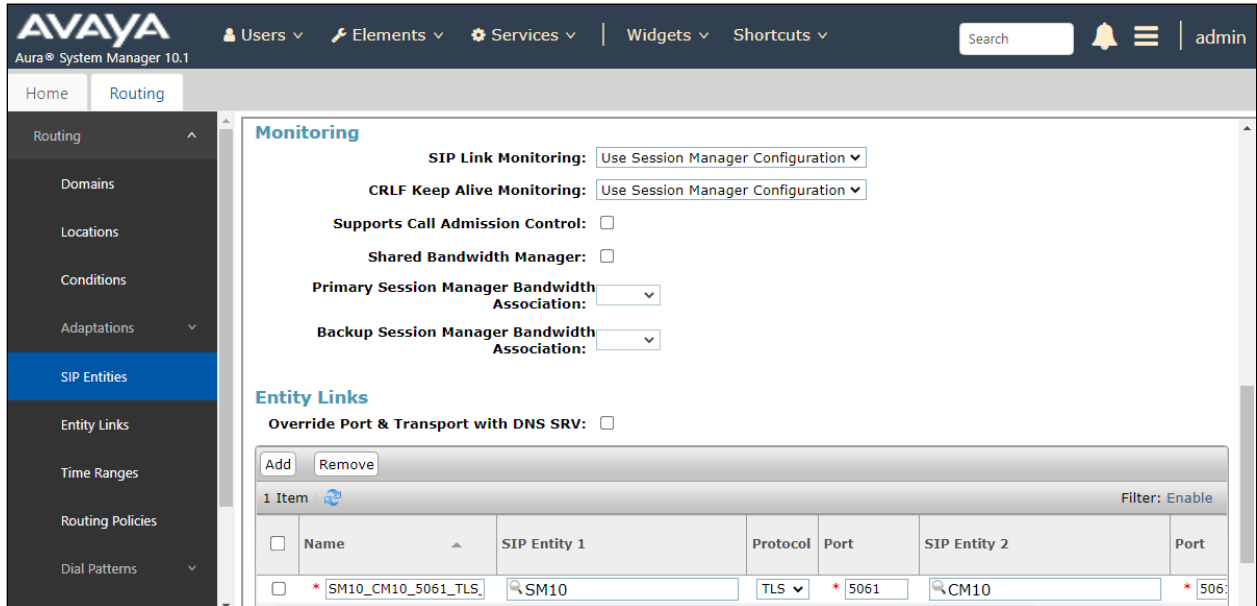
The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 10.1', and menu items for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile 'admin' are also visible. The left sidebar shows a navigation tree with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' section with the following fields:

- Name:** CM10
- FQDN or IP Address:** 10.33.1.43
- Type:** CM
- Notes:** (empty text box)
- Location:** Communication Manager
- Time Zone:** America/Denver
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty text box)
- Securable:**
- Call Detail Recording:** both

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area.

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 “SM10”.
- **Protocol:** Select TLS protocol.
- **Port:** Enter the TLS port 5061.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** Enter the TLS port 5061.
- **Connection Policy:** “trusted”



6.5. Administer Routing Policies

Add two new routing policies, one for Parlance and one for the new SIP trunks with Communication Manager.

6.5.1. Routing Policy for Parlance

Select **Routing** → **Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Parlance.

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the **ParlanceS1** entity name from **Section 6.4.1**. The screen below shows the result of the selection.

The screenshot displays the Avaya Aura System Manager 10.1 interface for configuring a Routing Policy. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Name:** To-ParlanceS1
- Disabled:**
- Retries:** 0
- Notes:** (empty text box)

The 'SIP Entity as Destination' section features a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
ParlanceS1	10.33.1.62	SIP Trunk	

The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows 1 item in a table with the following details:

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Repeat the same procedure above to add another routing policy for the SIP entity **ParlanceS2**. The screen below shows the routing policy **To-ParlanceS2**.

The screenshot shows the 'Routing Policy Details' configuration page in the Avaya DevConnect interface. The left sidebar is expanded to 'Routing Policies'. The main content area is titled 'Routing Policy Details' and includes a 'Commit' button and a 'Cancel' button. The configuration is divided into three sections:

- General:**
 - * Name: To-ParlanceS2
 - Disabled:
 - * Retries: 0
 - Notes: (empty text box)
- SIP Entity as Destination:**
 - Select: (dropdown menu)
 - Table with columns: Name, FQDN or IP Address, Type, Notes. One row is visible: Name: ParlanceS2, FQDN or IP Address: 10.33.1.63, Type: SIP Trunk, Notes: (empty).
- Time of Day:**
 - Buttons: Add, Remove, View Gaps/Overlaps
 - 1 Item (with refresh icon)
 - Filter: Enable
 - Table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, Notes. One row is visible: Ranking: 0, Name: 24/7, Mon: checked, Tue: checked, Wed: checked, Thu: checked, Fri: checked, Sat: checked, Sun: checked, Start Time: 00:00, End Time: 23:59, Notes: Time Range 24/7.

6.5.2. Routing Policy for 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 Communication Manager.

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager Entity name from **Section 6.4.2**. The screen below shows the result of the selection.

Routing Policy Details [Commit] [Cancel]

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM10	10.33.1.43	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/> 0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

6.6. Administer Dial Patterns

Add a new dial pattern for Parlance and update existing dial patterns for Communication Manager.

6.6.1. Dial Pattern for Parlance

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 the Parlance Service server. 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 “52”.
- **Min:** The minimum number of digits to match, in this case “4” was used.
- **Max:** The maximum number of digits to match, in this case “4” was used.
- **SIP Domain:** Select the applicable domain, in this case “All” selected.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching the Parlance Service server. In the compliance testing, the entry allowed for call originations from Communication Manager endpoint in locations “All”. Two routing policies **To-ParlanceS1** and **To-ParlanceS2** from **Section 6.5.1** were selected as shown below.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left navigation pane shows the 'Routing' menu expanded to 'Dial Patterns'. The main content area is titled 'Dial Pattern Details' and contains the following configuration fields:

- Pattern:** 52
- Min:** 4
- Max:** 4
- Emergency Call:**
- SIP Domain:** -ALL-
- Notes:** (empty)

Below the 'General' section is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with 2 items. The table has the following columns: Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy Disabled, Routing Policy Destination, and Routing Policy Notes.

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-		To-ParlanceS1	0	<input type="checkbox"/>	ParlanceS1	
-ALL-		To-ParlanceS2	0	<input type="checkbox"/>	ParlanceS2	

6.6.2. Dial Pattern for Communication Manager

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 Communication Manager. 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 “33”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select the applicable domain, in this case “All”.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for call originations from Communication Manager endpoint in locations “All”. The Communication Manager routing policy from **Section 6.5.2** was selected as shown below.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left navigation pane shows the 'Dial Patterns' menu item selected. The main content area is titled 'Dial Pattern Details' and includes a 'Commit' and 'Cancel' button. The 'General' section contains the following fields:

- * Pattern: 33
- * Min: 4
- * Max: 4
- Emergency Call:
- SIP Domain: -ALL-
- Notes: (empty text box)

The 'Originating Locations and Routing Policies' section features an 'Add' button and a table with one entry:

<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"/>	-ALL-	To-CM10	CM10	0	<input type="checkbox"/>	CM10	

7. Configure Parlance Service

Parlance Service is a Software as a Service (SaaS) product, deployed and installed within UC environment using either physical or virtual servers, and is an extension of voice infrastructure. It is remotely provisioned and managed by Parlance engineers, based on telephony and application requirements. Therefore, no configuration details are necessary for these Application Notes.

8. Verification Steps

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

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 5.4**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 1
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0001/001	T00001	in-service/idle	no
0001/002	T00002	in-service/idle	no
0001/003	T00003	in-service/idle	no
0001/004	T00004	in-service/idle	no
0001/005	T00005	in-service/idle	no
0001/006	T00006	in-service/idle	no
0001/007	T00007	in-service/idle	no
0001/008	T00008	in-service/idle	no
0001/009	T00009	in-service/idle	no
0001/010	T00010	in-service/idle	no
0001/011	T00011	in-service/idle	no
0001/012	T00012	in-service/idle	no
0001/013	T00013	in-service/idle	no
0001/014	T00014	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 (not shown). Click the **ParlanceS1** entity name. 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 10.1 interface. The left navigation pane is expanded to 'SIP Entity Monit...'. The main content area displays the 'SIP Entity, Entity Link Connection Status' page for the selected Session Manager 'ParlanceS1'. The page includes a 'Summary View' button and a table with the following data:

	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	SM10	IPv4	10.33.1.62	5060	UDP	FALSE	UP	200 OK	UP

The screenshot below shows the SIP status of the SIP entity ParlanceS2. Verify that the **Conn Status** and **Link Status** are “UP”, as shown below.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The left navigation pane is expanded to 'SIP Entity Monit...'. The main content area displays the 'SIP Entity, Entity Link Connection Status' page for the selected Session Manager 'ParlanceS2'. The page includes a 'Summary View' button and a table with the following data:

	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	SM10	IPv4	10.33.1.63	5060	UDP	FALSE	UP	200 OK	UP

8.3. Verify Parlance Service

From the PSTN, place a call that routes to the external menu of Parlance Service through Session Manager via SIP trunk. The PSTN user is connected to the Parlance Service and hears “*Parlance Corporation. Please say the first and last name of the person or the name of the department you'd like to reach*”.

The PSTN user says “**James Smith**”, the Parlance Service responds, “**Connecting to James Smith, please hold or say cancel**”. The call is then transferred to the local extension 3301 that is assigned to the contact “James Smith”. The local extension 3301 rings, answer the call on the extension 3301. The call is established between the PSTN user and the local extension with two-way audio.

9. Conclusion

These Application Notes describe the configuration steps required for Parlance Service Version 9 to successfully interoperate with Avaya Aura® Session Manager Release 10.1 and Avaya Aura® Communication Manager Release 10.1. All feature and serviceability test cases were completed successfully with the test results in **Section 2.2**.

10. Additional References

This section references the Avaya documentation that are relevant to these Application Notes. Product documentation for Avaya Aura® Session Manager, including the following, is available at: <http://support.avaya.com/>

- [1] Administering Avaya Aura® Session Manager, Document 03-300509, Issue 10, Release 10.1, August 2022
- [2] Administering Avaya Aura® System Manager, Issue 9.0, Release 10.1, August 2022
- [3] Administering Avaya Aura® Communication Manager, Document 03-300509, Issue 10, Release 10.1, August 2022
- [4] Avaya Aura® Communication Manager Feature Description and Implementation, Document 555-245-205, Issue 9.0, Release 10.1, May 2022

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