Avaya Solution & Interoperability Test Lab

Application Notes for Skit Virtual Intelligent Voice Assistant (VIVA) solution 1.0 with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication Manager 8.1. - Issue 1.0

Abstract

These Application Notes describe the configuration steps for Skit Virtual Intelligent Voice Assistant (VIVA) solution 1.0 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3. Skit VIVA solution is an IVR application that connects to Session Manager as a SIP Entity.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1**, as well as 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 for Skit VIVA solution 1.0 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3 using Transport Control Protocol (TCP) and Real-time Transport Protocol (RTP).

Skit VIVA solution is a natural language understanding virtual assistant that can engage in true end-to-end conversations in natural language. VIVA solution main components:

- VIVA SIP, an engine connected to the Session Manager via SIP trunking and a pre-defined route pattern for incoming calls.
- VIVA AI/IVR on Cloud as Automated Speech Recognition and Natural Language Understanding Engines.

2. General Test Approach and Test Results

The general test approach was to configure the Skit VIVA solution to communicate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Interoperability testing contained functional tests manually mentioned in **Section 2.1**. The serviceability test cases were performed manually by disconnecting/reconnecting the SIP trunk connectivity to the VIVA solution.

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 the Skit VIVA Solution did not include use of any specific encryption features as requested by Skit.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution

and the third-party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third-party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g., jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another and may affect the reliability or performance of the overall solution. Different network elements (e.g., session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third-party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. Feature testing included the validation of the following:

- **Basic Inbound calls** – Tests inbound calls to Skit VIVA Solution.
- G.711A, G.711U codecs support and negotiation. (without shuffling)
- **Call Forward** from Avaya Endpoints to Skit VIVA Solution.
- **Call Hold** – Tests held calls to/from Skit VIVA Solution.
- **Call Transfer** – Tests transferred calls to/from Skit VIVA Solution.
- **IVR/AI Functionality and Proper transmissions of DTMF.**
- **Failover/Service** – Tests the behaviour of Skit VIVA when there are certain failed conditions.

2.2. Test Results

All functionality and serviceability test cases were completed successfully.

2.3. Support

Support for Skit VIVA products can be obtained as follows:

Email : hello@skit.com
Phone: +1 212 089795 64068

3. Reference Configuration

Figure 1 illustrates a sample configuration that consists of Avaya products and the Skit VIVA SIP.

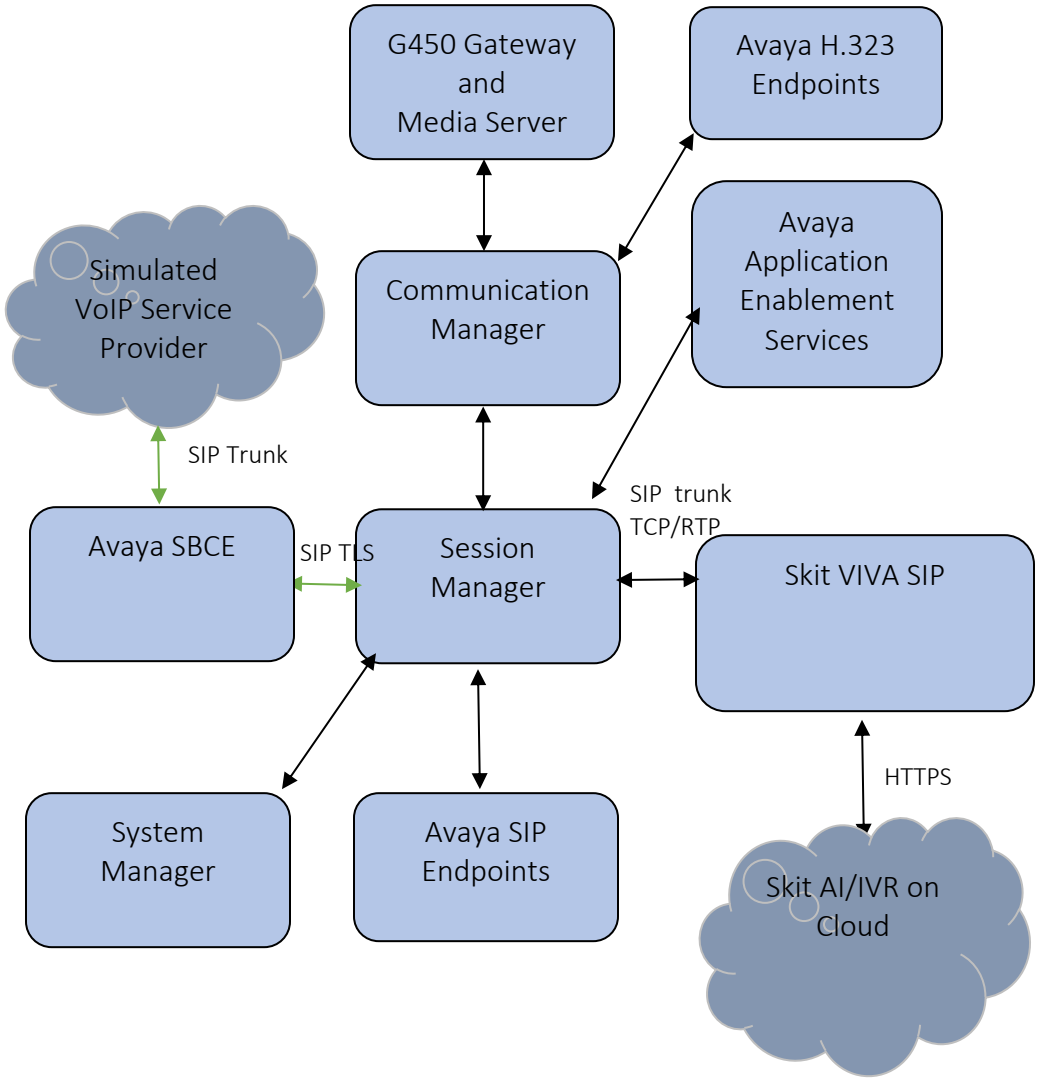


Figure 1: Test Configuration for Skit VIVA Solution and the Avaya Platform.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager in Virtual Environment	8.1.3.3
Avaya Aura® Session Manager in Virtual Environment	8.1.3.3
Avaya Aura® Communication Manager in Virtual Environment	8.1.3.3
Avaya G450 Media Gateway	41.16.30
Avaya Aura® Media Server in Virtual Environment	8.0.2.43
Avaya Session Border Controller for Enterprise in Virtual Environment	8.1.3.0
Avaya 9608G & 9641G IP Deskphone (H.323)	6.8
Avaya IX Workplace	3.21.0
Avaya 9641 & 9621 IP Deskphone (SIP)	7.1.9
Avaya J159, J179 & J189 SIP Deskphone	4.0.10
Avaya K175 & Avaya K155	3.1.0.0
Skit VIVA SIP	1.0.0

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

It is assumed that the general installation and configuration of Avaya Aura® environment and simulated PSTN SIP Trunk have been previously completed and is not discussed here.

The configuration operations described in this section can be summarized as follows:

- Verify System Parameters Customer Options.
- System Features and Access Codes.
- Configure Network Region and IP Codec.
- Configure SIP Signalling Group and Trunk Group.
- Administer Dial Plan.
- Administer Route Selection for VIVA SIP calls.

5.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that **Maximum Administered SIP Trunks** has sufficient capacity. Each call that receives IVR treatment from VIVA SIPuses a minimum of one SIP trunk. Calls that are routed back to stations commissioned on Communication Manager or calls that are routed back to Communication Manager to access the PSTN, use 2 SIP trunks.

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

IP PORT CAPACITIES                                        USED
                 Maximum Administered H.323 Trunks:  4000      0
       Maximum Concurrently Registered IP Stations:  1000      2
         Maximum Administered Remote Office Trunks:  4000      0
Max Concurrently Registered Remote Office Stations:  1000      0
          Maximum Concurrently Registered IP eCons:   68       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                   Maximum Video Capable Stations:   2400     0
               Maximum Video Capable IP Softphones:   1000    41
                   Maximum Administered SIP Trunks:  4000    305
  Max Administered Ad-hoc Video Conferencing Ports:  4000      0
   Max Number of DS1 Boards with Echo Cancellation: 80     0
```

On **Page 4**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                      Page   4 of  12
                               OPTIONAL FEATURES

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



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

On **Page 6**, ensure that **Uniform Dialing Plan** is set to **y**.

```
display system-parameters customer-options                      Page   6 of  12
                               OPTIONAL FEATURES

                Multinational Locations? n                Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? y      Station as Virtual Extension? y
                     Multiple Locations? n
               No-License Mode Disabled? y   System Management Data Transfer? n
          Personal Station Access (PSA)? y                Tenant Partitioning? y
                       PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                  Port Network Support? y                 Time of Day Routing? y
                       Posted Messages? y       TN2501 VAL Maximum Capacity? y
                                                        Uniform Dialing Plan? y
                    Private Networking? y     Usage Allocation Enhancements? y
              Processor and System MSP? y
                    Processor Ethernet? y                 Wideband Switching? y
                                                                  Wireless? n
                         Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y


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

5.2. System Features and Access Codes

For the testing, **Trunk-to Trunk Transfer** was set to **all** on **page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 10** for supporting documentation.

```
display 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 (or Silence) on Transferred Trunk Calls? all
              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
```

Use the **display feature-access-codes** command to verify that a FAC (feature access code) has been defined for both AAR and ARS. Note that ***50** is used for AAR and **9** for ARS routing.

```
 display feature-access-codes                                    Page   1 of  12
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                       Announcement Access Code:
                         Answer Back Access Code:
                           Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *50
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2: *51
                 Automatic Callback Activation: *52    Deactivation: *53
```

5.3. Configure Network Region and IP Codec.

In the Node Names IP form, note the IP Address of the **procr** and Session Manager (**smsip92**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
aes95               10.30.5.95
ams94               10.30.5.94
default             0.0.0.0
procr               10.30.5.93
procr6              ::
smsip92             10.30.5.92









( 7  of 7    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.2**. In this configuration, the domain name is **devconnect.com**. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session manager as **ip-network region 1** is specified in the SIP signaling group.

```
change ip-network-region 1                                    Page   1 of  20
                           IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: devconnect.com
    Name: SaiGon                Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              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
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk to VIVA SIP. The form is accessed via the **change ip-codec-set n** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the **ip-codec-set 1** example below includes **G.711A** (a-law) and **G.711MU** which are supported by VIVA SIP. The **Media Encryption** have **none** as an option, **Media Encryption** from Communication Manager by using a codec-set that doesn't have **'none'** as an option for calls between network regions with encryption forced. By adding none, if an unsecure call comes in, the call can still be processed as far as TCP or TLS (with certificates deployed) is concerned, but also if TLS protocol is set, non-secure SRTP calls will also be sent.

```
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.722-64K                    2        20
3: G.729              n         2        20
4: OPUS-WB20K                   1        20
5: G.711A             n         2        20
6:
7:

    Media Encryption                    Encrypted SRTCP: best-effort
1: 10-srtp-aescm256-hmac80
2: 1-srtp-aescm128-hmac80
3: none
3:
4:
5:
```

5.4. Configure Signaling Group and Trunk Group.

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager This signaling group and trunk group is used for internal calls between Avaya endpoints and used for calls to and from VIVA SIP. For the compliance test, signaling group 1 was used and was configured using the parameters highlighted below, shown on the screen on the next page:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the desired transport method, for compliance testing this was set to **tls**.
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node**

Name field, respectively. These values are taken from the **IP Node Names** form shown above.

- Set the **Near-end Node Name** to **procr**. This value is taken from the **IP Node Names** form shown above.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **smsip92**), as per **Section 5.5**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured above**.** This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- **Far-end Domain** was set to the domain used during compliance testing.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- The default values for the other fields may be used.

```
change 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? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  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? y
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: smsip92
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                    Far-end Network Region: 1

Far-end Domain: devconnect.com
                                          Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate               RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                  IP Audio Hairpinning? y
         Enable Layer 3 Test? y                  Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y           Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager dial plan. Set the **Service Type** field to **tie**. Specify the signaling group associated with this trunk group in the **Signaling Group** field and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```
change trunk-group 1                                          Page   1 of   4
                               TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: InternalCalls             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: 50
```

Settings on **Page 3** can be left as default. However, the **Numbering Format** in the example below is set to **private**.

```
change trunk-group 1                                          Page   3 of   4
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                            UUI Treatment: service-provider

                                             Replace Restricted Numbers? y
                                            Replace Unavailable Numbers? y


                               Modify Tandem Calling Number: no



 Show ANSWERED BY on Display? y

 DSN Term? n
```

5.5. Administer Dial Plan

It was decided for the compliance testing that all calls beginning with 4 and a total length of 5 digits were to be sent across the SIP trunk to Session Manager and therefore to VIVA SIP. In order to achieve this, automatic alternate routing (aar) would be used to route the calls. The dial plan and aar routing analysis need to be changed to allow this.

Type **change dialplan analysis**, in order to make changes to the dial plan. Ensure that **4** is added with a **Total Length** of **5** and a **Call Type** of uniform dialing plan (**udp)** table.

```
change dialplan analysis                                        Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                  Location: all            Percent Full: 2

   Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
   String   Length Type     String   Length Type     String   Length Type
 0             10 udp
 3             5  udp
 4             5 udp
 7             5  ext
 8             5  ext
 9             1  fac
 *             3  fac
 #             3  dac
```

5.6. Administer Route Selection for VIVA SIPCalls.

As digits **4**xxxx were defined in the dial plan as udp (**Section 5.5**) use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to numbers beginning with **4** that are **5** digits in length will be matched. No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

```
change uniform-dialplan 4                                       Page   1 of   2
                      UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                      Insert                Node
  Pattern           Len Del     Digits       Net Conv Num

 4                   5  0                     aar n
```

Use the **change aar analysis** x command to further configure the routing of the dialed digits. Calls to VIVA SIPbegin with **4** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 1**, which contains the SIP Trunk Group with Session Manager.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     10   10    4        lev0         n
    4                     5    5     1        lev0         n
    6                     5    5     1        lev0         n
    7                     5    5     1        lev0         n
    8                     5    5     1        lev0         n
    899                   5    5     1        lev0         n
```

Use the **change route-pattern *n*** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, **Route Pattern Number 1** is used to route calls to trunk group **(Grp No) 1**, this is the SIP Trunk with Session Manager.

```
change route-pattern 1                                         Page   1 of   4
                  Pattern Number: 1      Pattern Name: DevC-Int
   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                                                            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 Sub  Numbering LAR
   0 1 2 M 4 W     Request                                Dgts Format
 1: y y y y y n  n           rest                                lev0-pvt  none
 2: y y y y y n  n           rest                                          none
 3: y y y y y n  n           rest                                          none
 4: y y y y y n  n           rest                                          none
 5: y y y y y n  n           rest                                          none
 6: y y y y y n  n           rest                                          none
```

6. Configure Avaya Aura® Session Manager

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

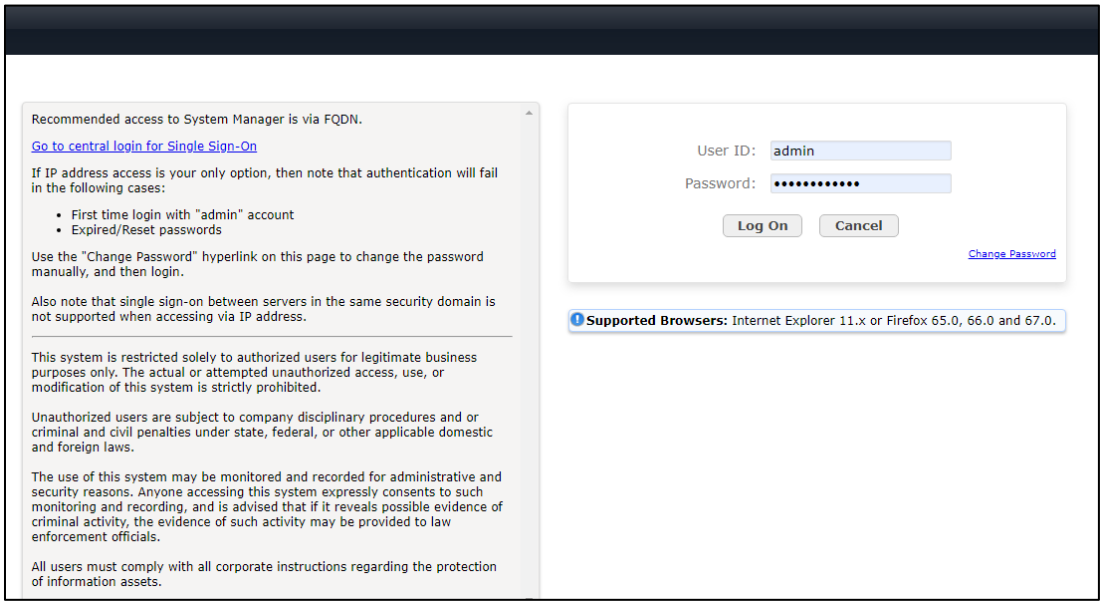
- Configure SIP Entities
- Configure Routing Policies
- Configure Dial Patterns

6.1. Configure SIP Entities

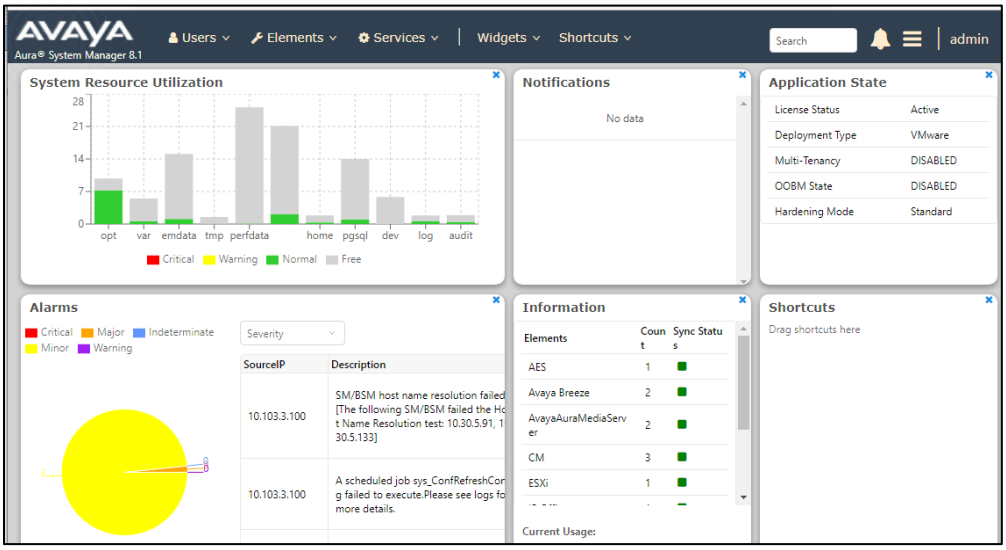
This section provides steps to add sip entities and sip trunks between Session Manager and VIVA SIP as well as between Session Manager and Communication Manager.

6.1.1. Configure SIP Entity for VIVA SIP

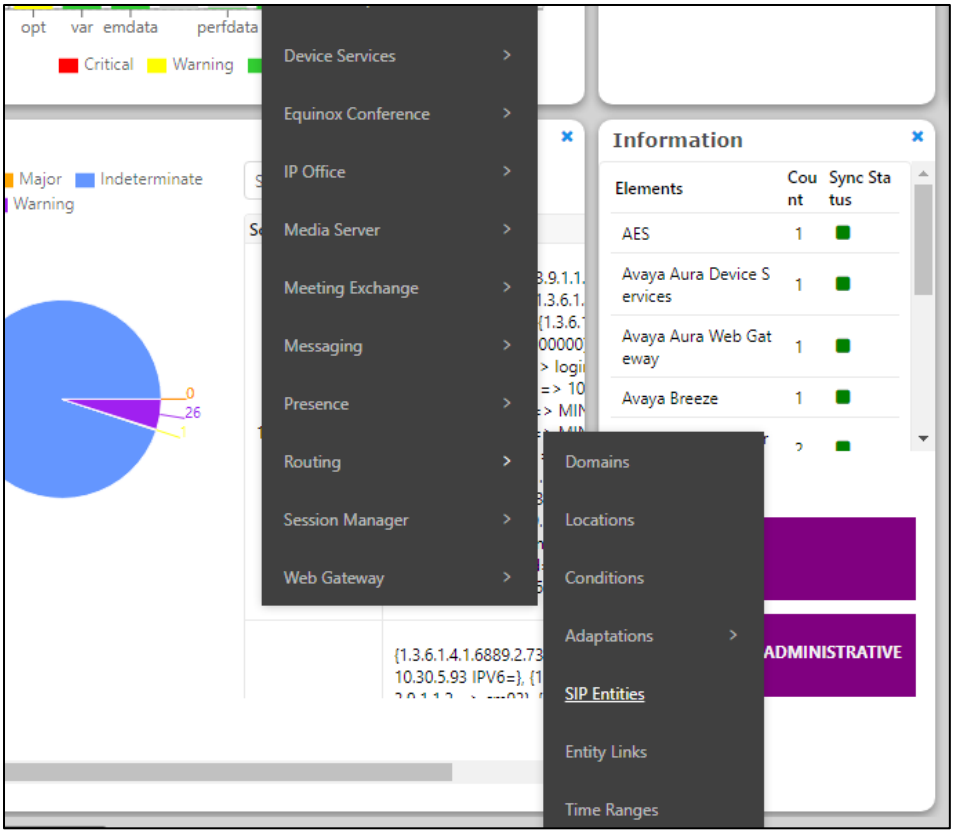
Configuration of SIP Entities is performed via Avaya Aura® System Manager. Access the System Manager Administration web interface by entering the System Manager (SMGR) URL in a web browser. Log in using appropriate credentials.



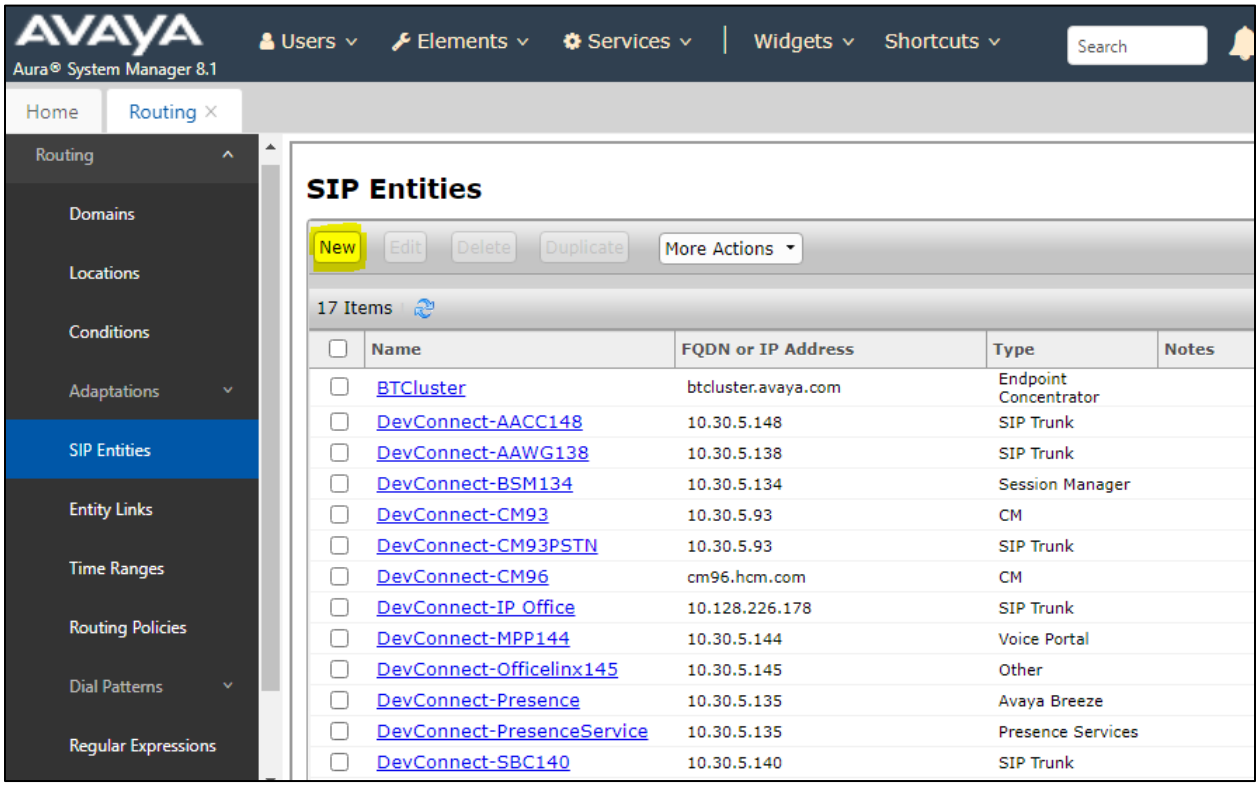
Once logged in, the following screen is displayed.



Select **Elements → Routing → SIP Entities**.

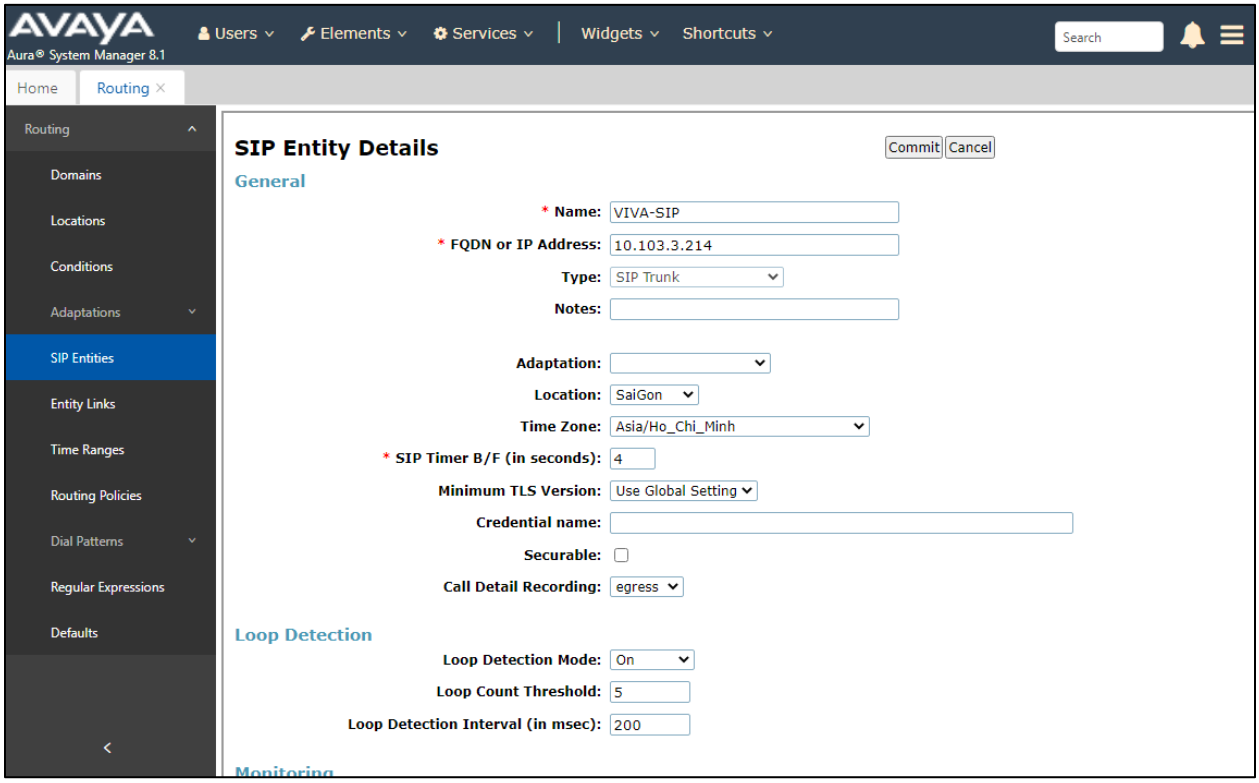


On **SIP Entities** page, press **New** to create new **SIP Entity**.



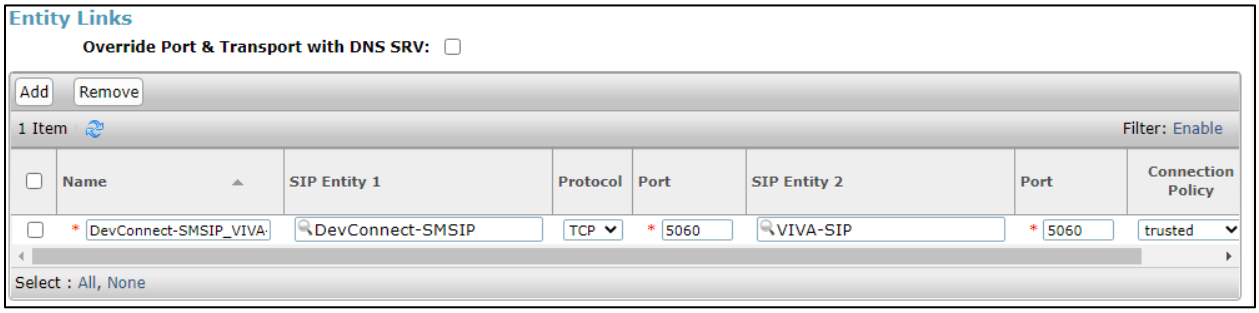
Enter a suitable **Name** and ensure that the correct **Location** and **Time Zone** are entered correctly, click on **Commit** to save the new entity.

Note: The setup of a Location is specific to each site, this can be added by clicking on **Locations** on the left panel on the screen shot below, the setup of the location for this site has not been documented as part of this setup as it would be already setup as part of the site installation.



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 "**DevConnect-SMSIP**".
- **Protocol:** "TCP"
- **Port:** "5060"
- **SIP Entity 2:** The Skit VIVA entity name from this section, in this case **"VIVA-SIP"**
- **Port:** "5060"
- **Connection Policy:** "trusted"

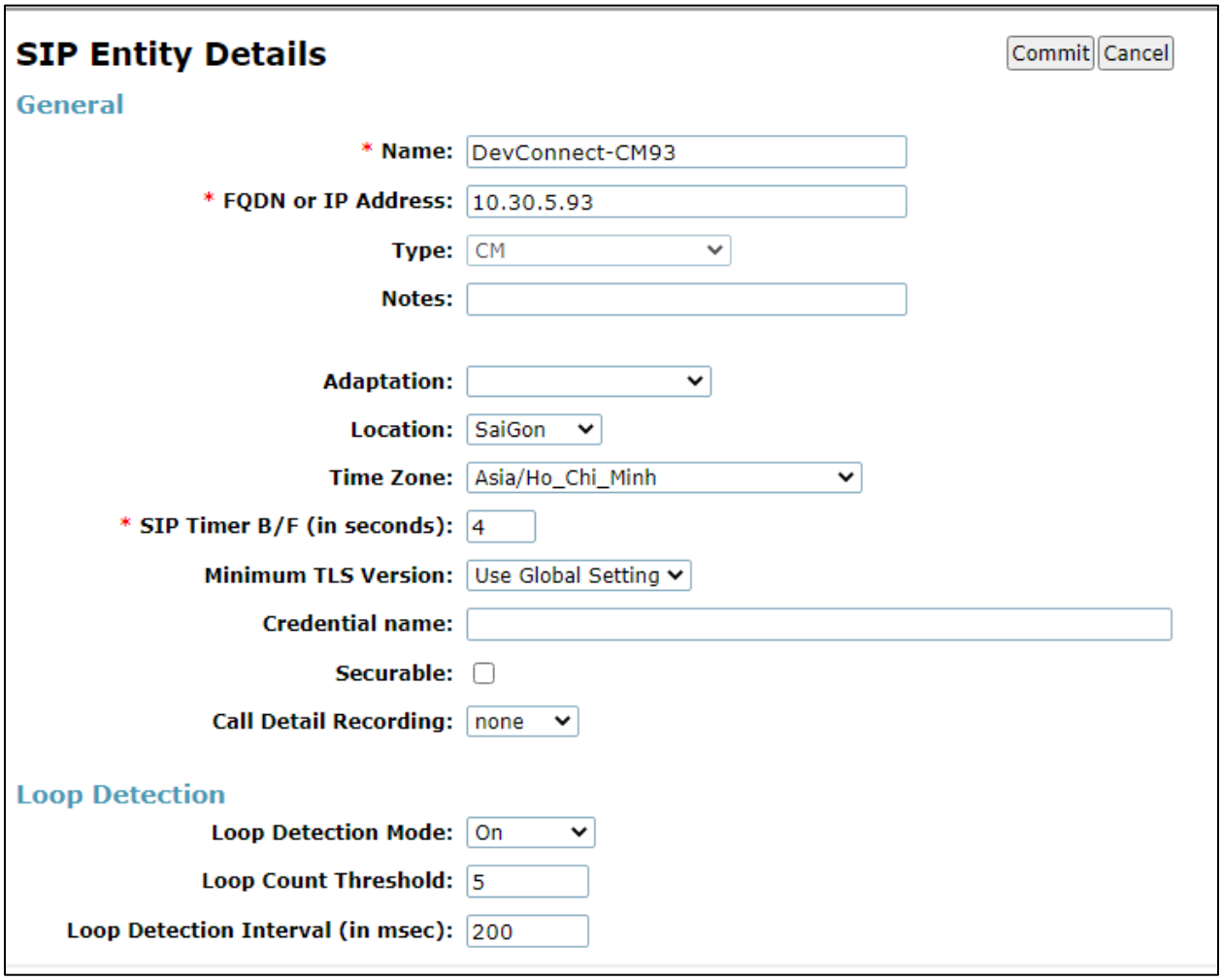


6.1.2. Configure SIP Entity for Communication Manager

Add new SIP entity for Avaya 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 Avaya Communication Manager.

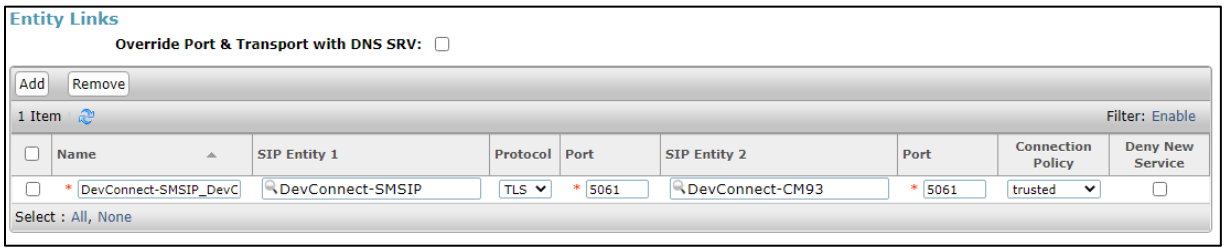
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name, example "DevConnect-CM93"
- **FQDN or IP Address:** The internal SIP IP address of Avaya CM.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the applicable location.
- **Time Zone:** Select the applicable time zone.



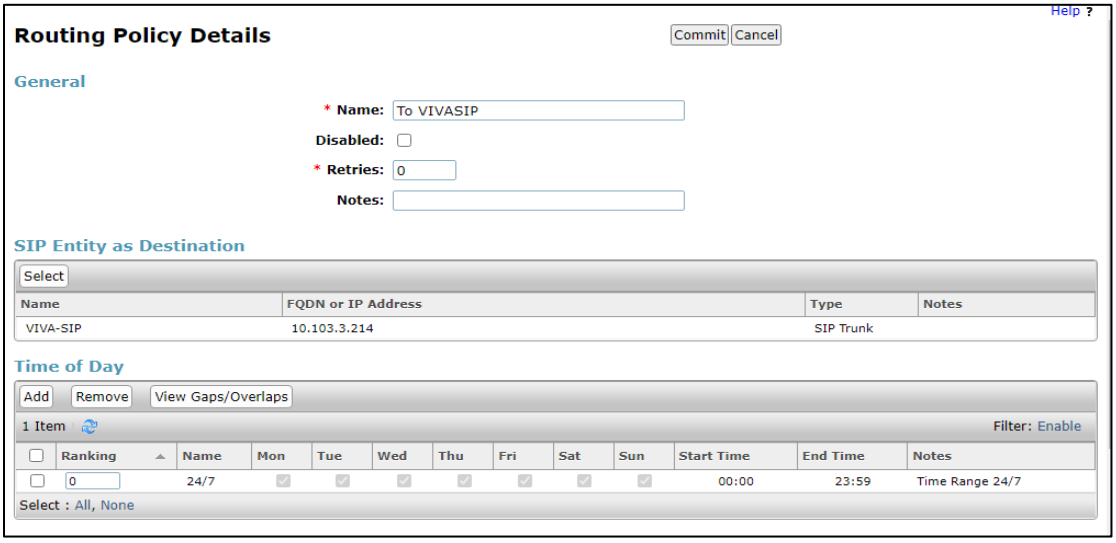
Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "**DevConnect-SMSIP**".
- **Protocol:** "TLS"
- **Port:** "5061"
- **SIP Entity 2:** The Avaya CM entity name from this section, in this case **"DevConnect-CM93"**
- **Port:** "5061"
- **Connection Policy:** "trusted"



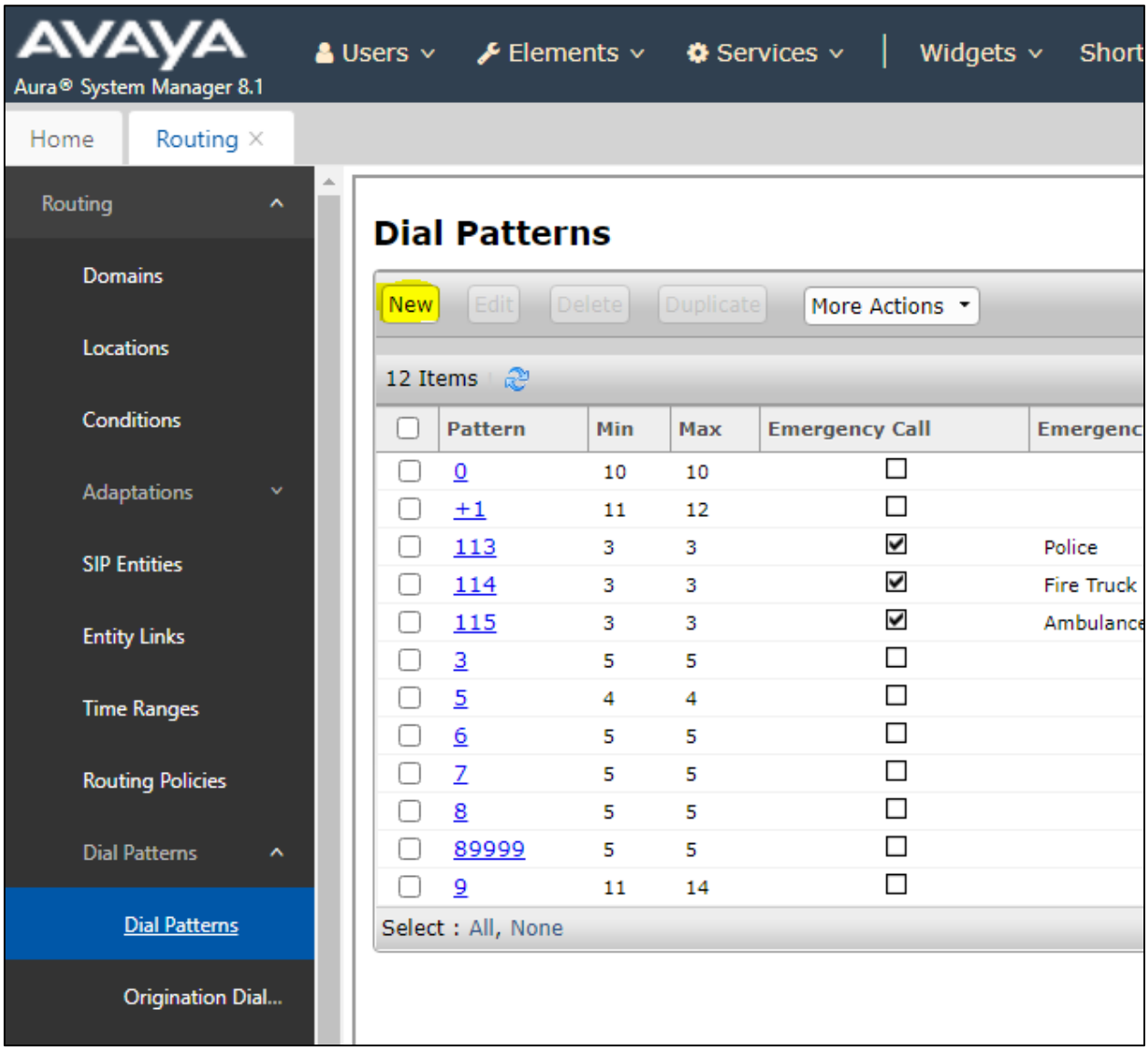
6.2. Configure Routing Policy for Skit VIVA SIP

This section to add a new routing policy for routing calls to VIVA SIP. Select **Routing** → **Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy to Communication Manager. The **Routing Policy Details** screen is displayed. In the **General** sub-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 VIVA SIP entity name from **Section 6.1**.

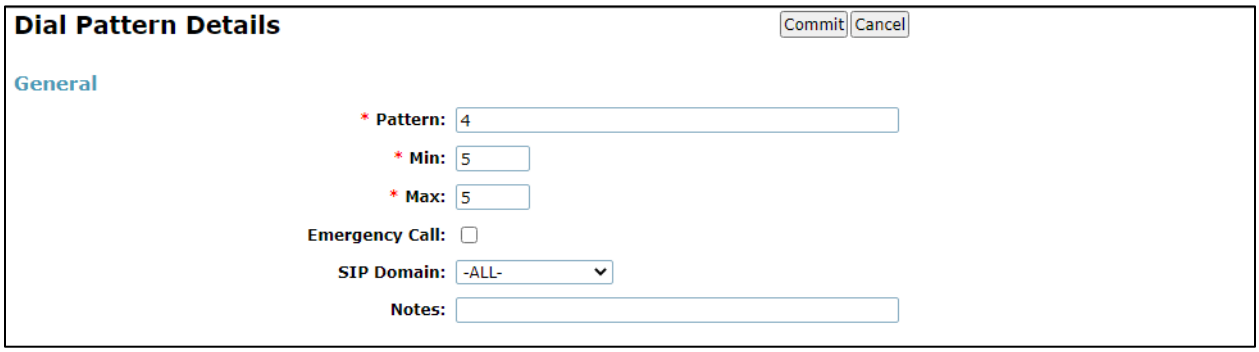


6.3. Configure Dial Pattern for Skit VIVA SIP

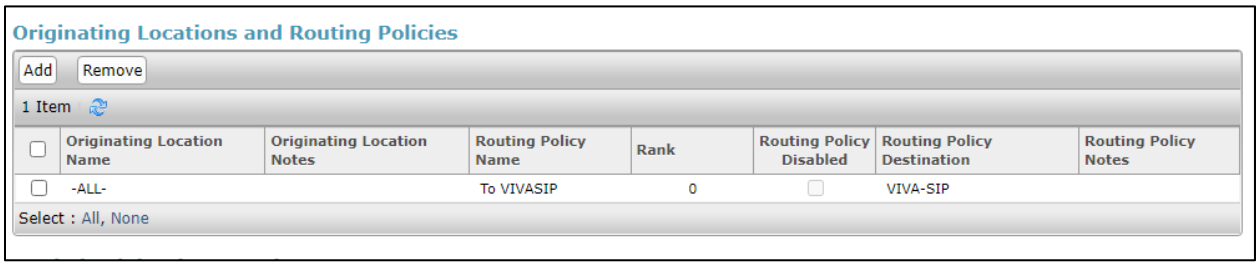
In order to route calls to the VIVA SIP a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.



The **Dial Pattern Details** screen is displayed. Enter the number to be routed noting this will be the same number outlined in **Section 5.4**. In the **Originating Locations and Routing Policies** sub-section, click **Add**.



Select a preconfigured **Originating Location** and select the **Routing Polices** created in previous **Section 6.2** (not shown). The configuration below shows calls to **4xxxx** were routed to VIVA SIP. Click on **Commit** as shown below to save configuration.



7. Configuration of Skit VIVA 1.0

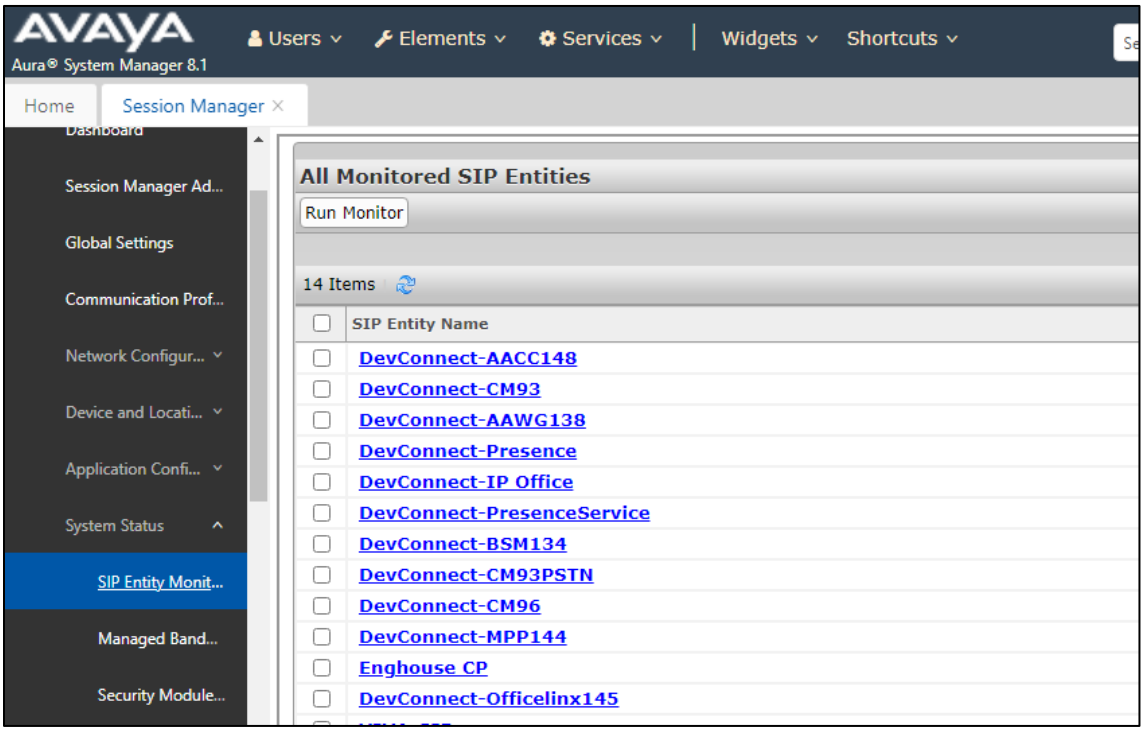
All configuration related to Skit VIVA is performed by to Skit engineers and, thus, is not documented.

8. Verification Steps

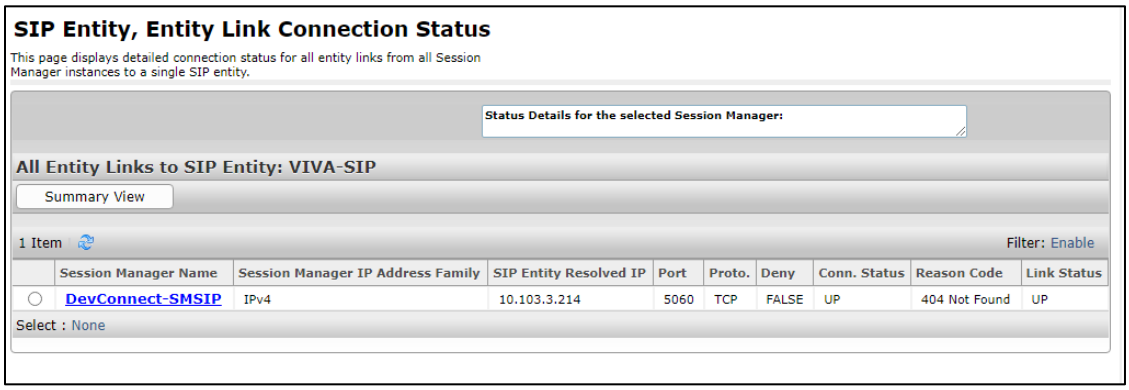
To verify a successful configuration of Skit VIVA SIP and Session Manager/Communication Manager a call is placed from a Communication Manager telephone to the VIVA SIP with the caller getting answered successfully hearing clear and audible speech.

8.1. Verify Entity Link between Session Manager and VIVA SIP

To verify SIP connectivity to VIVA SIP, via System Manager, navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring.** Under the **All Monitored SIP Entities,** select the **VIVA-SIP** Entity.

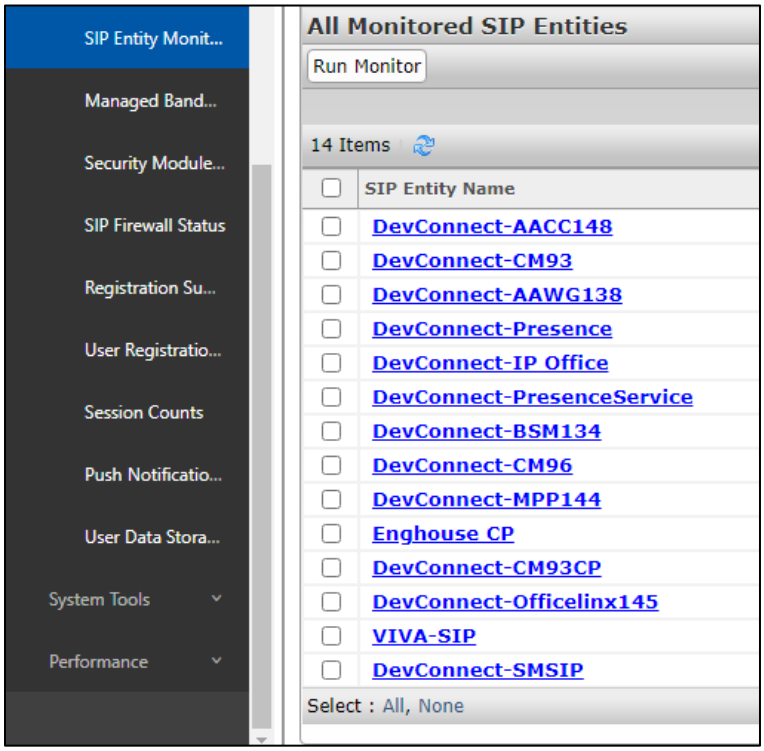


Verify **Conn. Status** is **UP**.

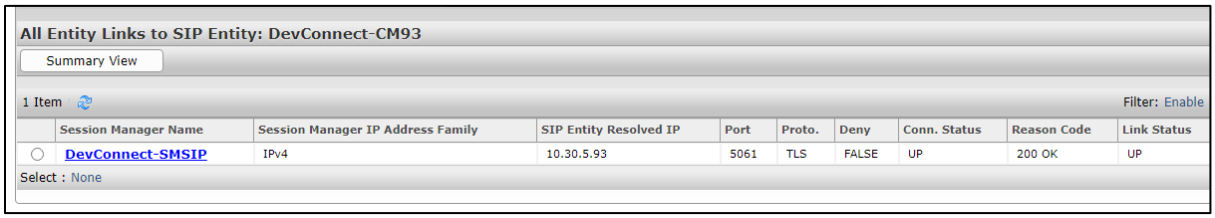


8.2. Verify Entity Links between Session Manager and Communication Manager

To verify SIP connectivity to VIVA SIP, via System Manager, navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring.** Under the **All Monitored SIP Entities**, select the **DevConnect-CM93** Entity.



Verify **Conn. Status** is **UP**.



8.3. Verify IVR

Place a call from the Avaya endpoints/PSTN to Skit VIVA with call number 4xxxx, ensure the call can be answered by VIVA SIP.

9. Conclusion

These Application Notes describe the configuration steps required for Skit Virtual Intelligent Voice Assistant (VIVA) solution to successfully interoperate with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication Manager 8.1. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2**.

10. Additional References

Documentation related to Avaya can be obtained from https://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 8, Nov 2020
[2] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 8, Feb 2021
[3] *Administering the Avaya Aura® Web Gateway*, Release 3.8 Issue 2, July 2020
[4] *Administering Avaya Aura® Application Enablement Services*, Release 8.1.x, Issue 8, Feb 2021
[5] *Administering Avaya Aura® Device Services*, Release 8.0.2, Issue 4, June 2020

Product documentation for Skit VIVA can be obtained by visiting the following website https://skit.ai/

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.