Avaya Solution & Interoperability Test Lab

Application Notes for Computer Instruments eONE with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Computer Instruments eONE to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. Computer Instruments eONE is an IVR development platform that includes a number of self-service IVR and Web applications. In the compliance testing, Computer Instruments eONE used SIP trunks to Avaya Aura® Session Manager to support inbound and outbound IVR applications.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 Computer Instruments eONE to interoperate with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager) using SIP trunks. Computer Instruments eONE (eONE) is an IVR development platform that includes a number of self-service IVR and Web applications. In the compliance testing, Computer Instruments eONE used SIP trunks to Avaya Aura® Session Manager to support inbound and outbound IVR applications.

The Computer Instruments eONE server used in the testing was deployed on cloud.

2. General Test Approach and Test Results

The feature test cases were performed manually. The eONE inbound application was tested by manually placing calls from users on the PSTN and on Communication Manager to the eONE inbound application. The associated eONE inbound application played greeting announcements and collected DTMF input from the caller to decide on the feature to provide, such as transfer to internal or external destinations. eONE outbound application to PSTN and Communication Manager were also tested.

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

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.

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.

The feature testing included G.711MU, codec negotiation, media shuffling, session refresh, hold/reconnect, inbound DTMF, invalid number, busy destination, and outgoing call screening.

The serviceability testing focused on verifying the ability of eONE to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to eONE.

2.2. Test Results

All test cases were executed and passed.

2.3. Support

Technical support on eONE can be obtained through the following:

- **Phone:** (888) 451-0851
- **Email:** support@instruments.com
- **WEB:** http://instruments.com/support/email_form.html (monitored 24x7)

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between eONE and Session Manager, and the applicable domain name used was "avaya.com". 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.

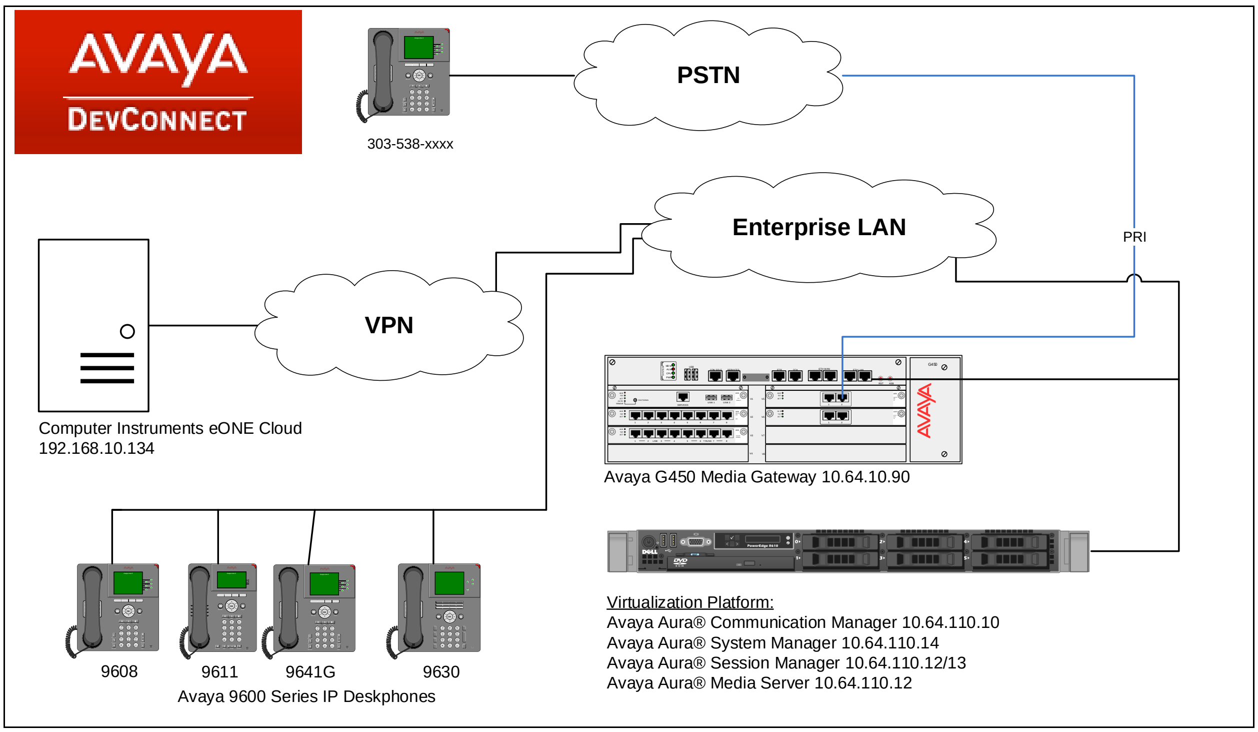


Figure 1: Computer Instruments eONE with Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager on Avaya S8300D Server with Avaya G450 Media Gateway	7.1.2.0.0.532.24184 37.19.0
Avaya Aura® Session Manager	7.1.1.0.711008
Avaya Aura® System Manager	7.1.1.0.046931
Avaya 96x0 IP Deskphone (H.323)	3.2.8
Avaya 96x1 IP Deskphone (H.323)	6.6.6
Avaya 96x0 IP Deskphone (SIP)	2.6.17
Avaya 96x1G IP Deskphone (SIP)	7.1.1.0
Computer Instruments eONE	6.1.5

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 trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer PSTN trunk group
- Administer tandem calling party number

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

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 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 0
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 12000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   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 (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
```

5.3. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "92". This trunk group is used between Communication Manager and Session Manager. 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: asm                        COR: 1        TN: 1        TAC: 101
   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:
                                                 Number of Members:
```

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

```
add 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? n
                                               Replace Unavailable Numbers? n

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

5.4. 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" in this case.
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Communication Manager.
- **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 eONE.
- **Far-end Domain:** The applicable domain name for the network. The empty Far-end Domain indicates "any" domain.

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: 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: asm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain:
                                             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): 65                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

5.5. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case “10”.

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: asm                        COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

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 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**, if desired. 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 eONE.

```
change ip-network-region 1                                            Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                       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
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.7. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that eONE only supports the G.711 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: 1

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

5.8. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach eONE via Session Manager, 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 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.
- **Numbering Format:** "lev0-pvt"

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name:
    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
```

5.9. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to eONE. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 5 and routed to any trunk group will result in a 5-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 Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  5                                             5     Total Administered: 1
                                                           Maximum Entries: 540
```

5.10. Administer AAR Analysis

Use the "change aar analysis 511" command, and add an entry to specify how to route calls to 51111. In the example shown below, calls with digits 51111 will be routed as an aar call type using route pattern "1" from **Section 0**.

```
change aar analysis 511                                             Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed           Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    51111                5    5      1        aar          n
```

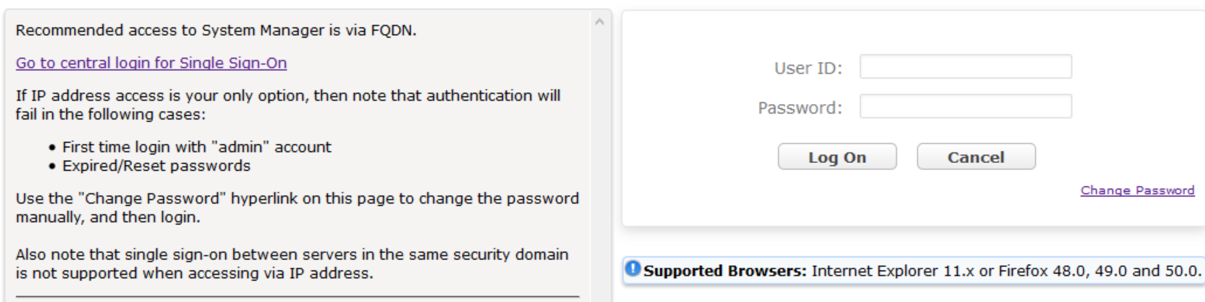
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer 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.



6.2. Administer SIP Entities

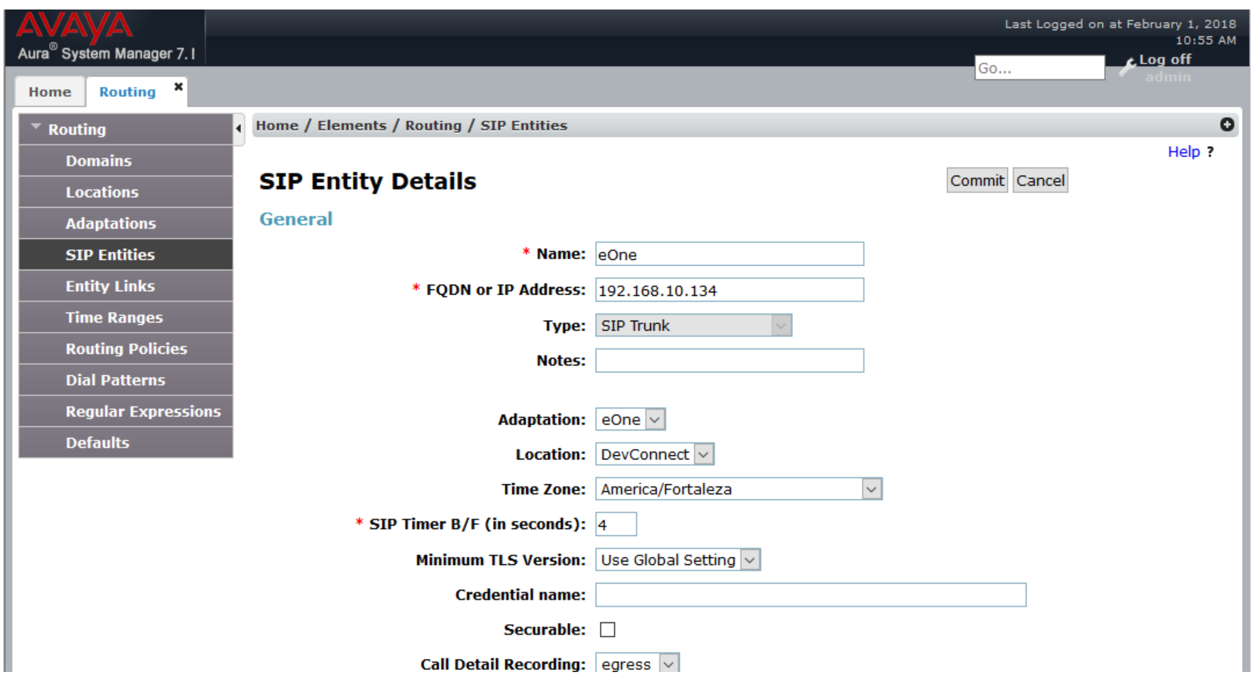
Add two new SIP entities, one for eONE and one for the new SIP trunks with Communication Manager.

6.2.1. SIP Entity for eONE

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

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 eONE SIP interface.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the eONE location name.
- **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.

- **SIP Entity 1:** The Session Manager entity name, in this case "asm".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The eONE entity name from this section.
- **Port:** "5060"

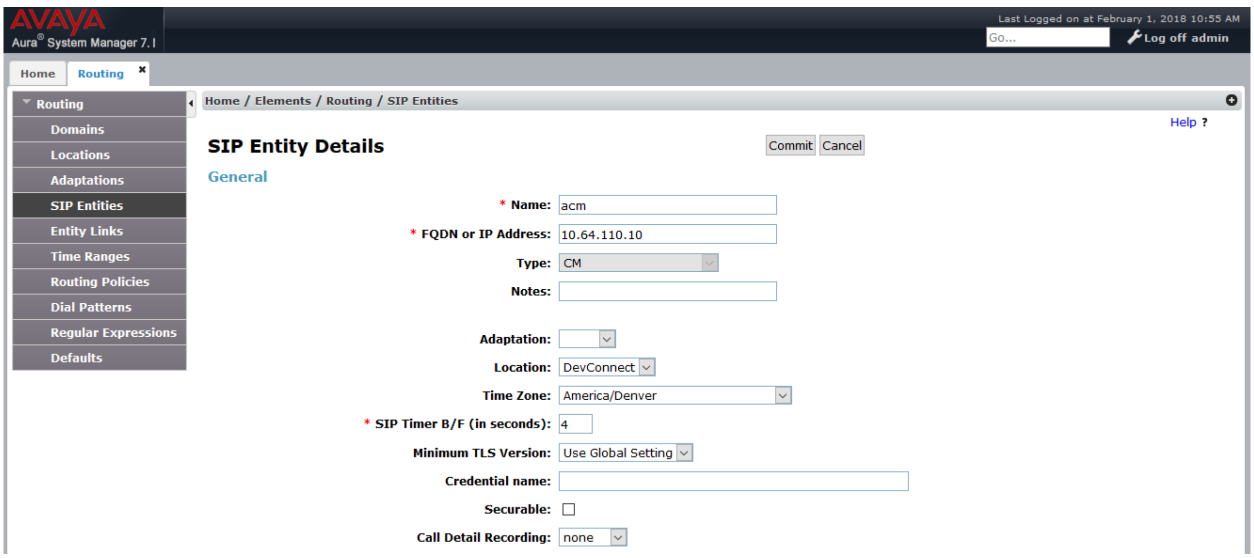
Note that eONE can support UDP and TCP, but during the compliance testing used the UDP protocol.

6.2.2. 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 eONE.

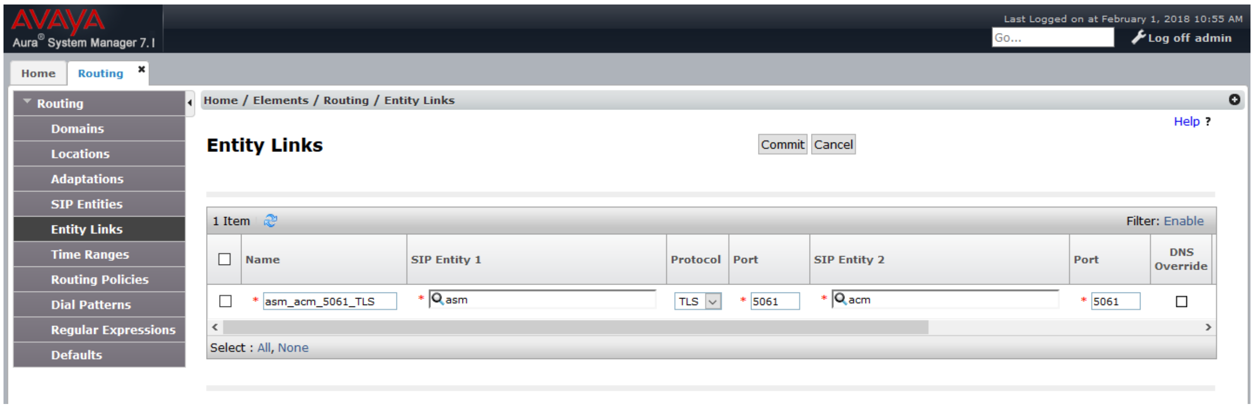
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **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.



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.

- **SIP Entity 1:** The Session Manager entity name, in this case "asm".
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port number from **Section 5.4**.



6.3. Administer Routing Policies

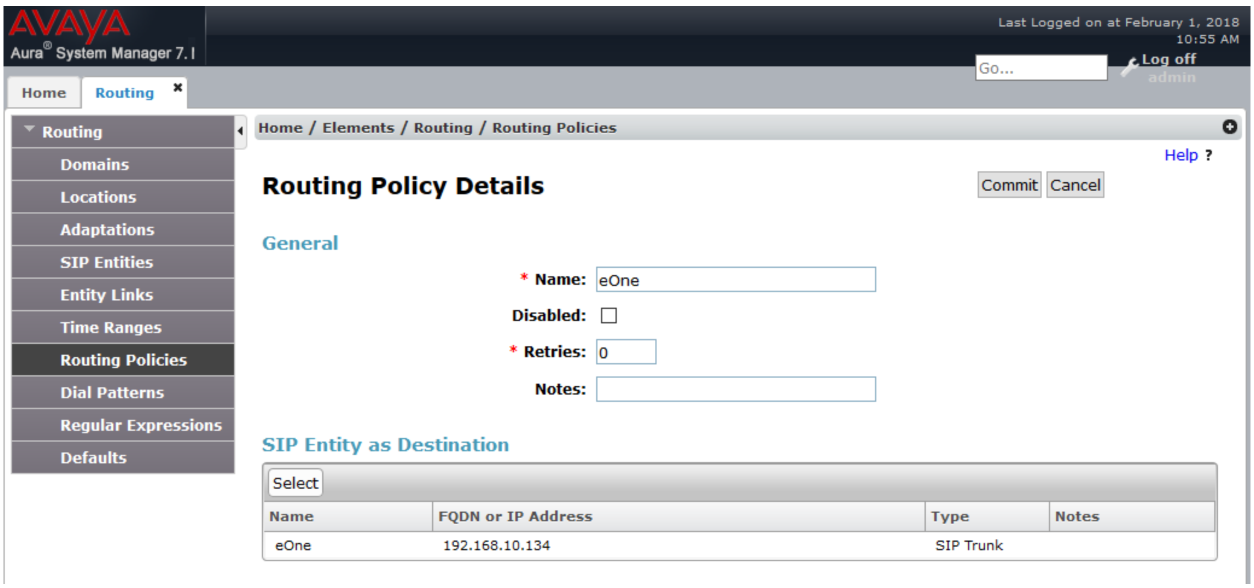
Add two new routing policies, one for eONE and one for the new SIP trunks with Communication Manager.

6.3.1. Routing Policy for eONE

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

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 eONE entity name from **Section 6.2.1**. The screen below shows the result of the selection.

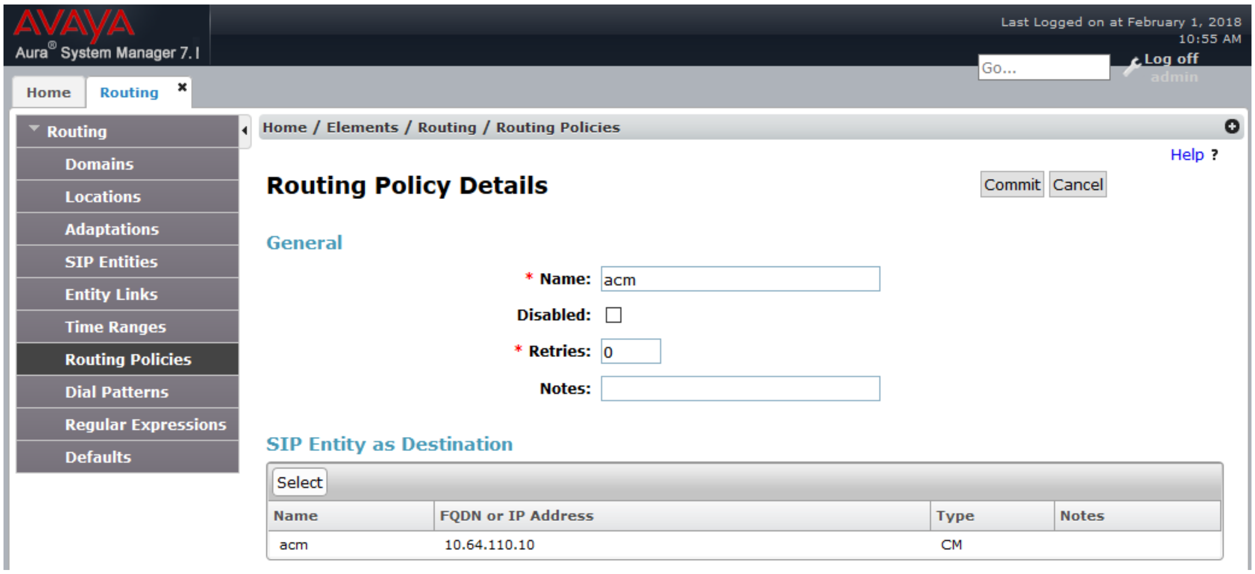


6.3.2. Routing Policy for Communication Manager

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** 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 Communication Manager entity name from **Section 6.2.2**. The screen below shows the result of the selection.



6.4. Administer Dial Patterns

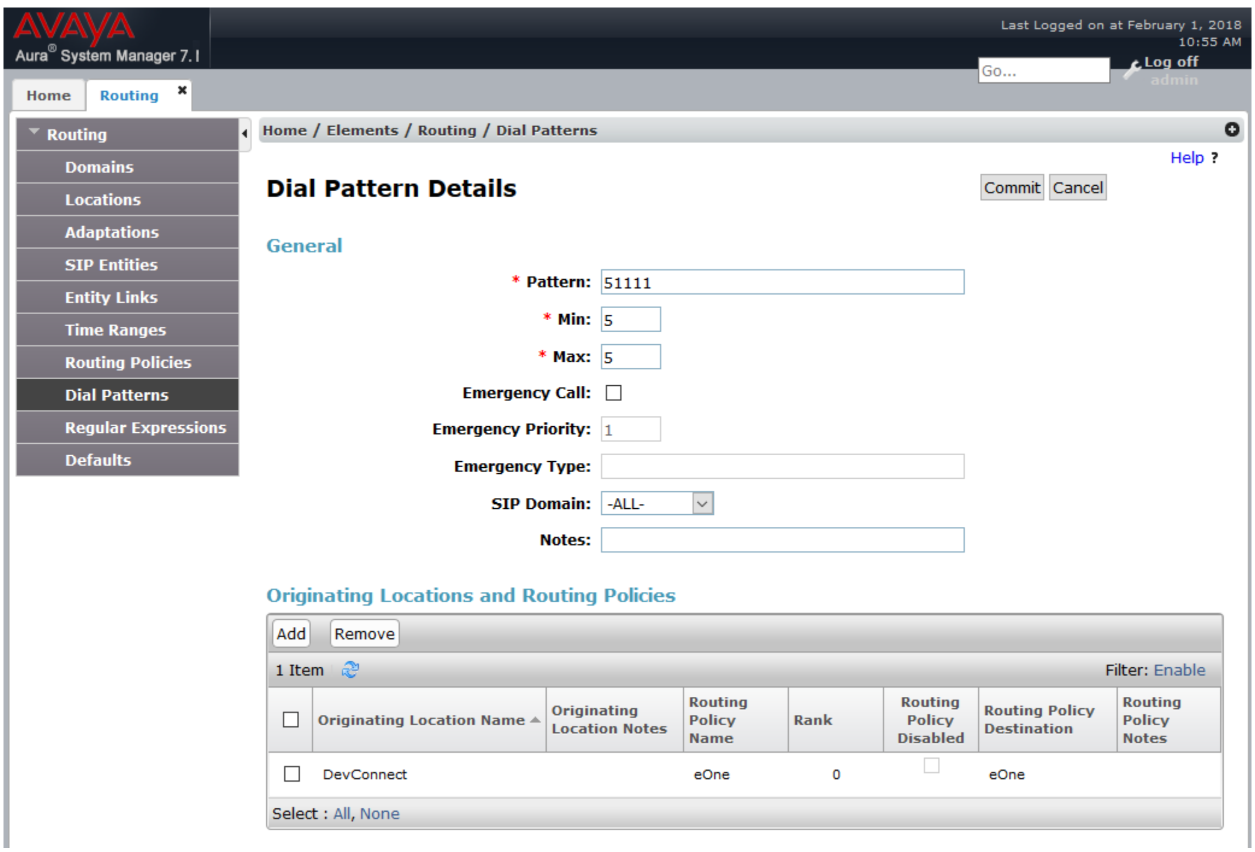
Add a new dial pattern for eONE, and update existing dial patterns for Communication Manager.

6.4.1. Dial Pattern for eONE

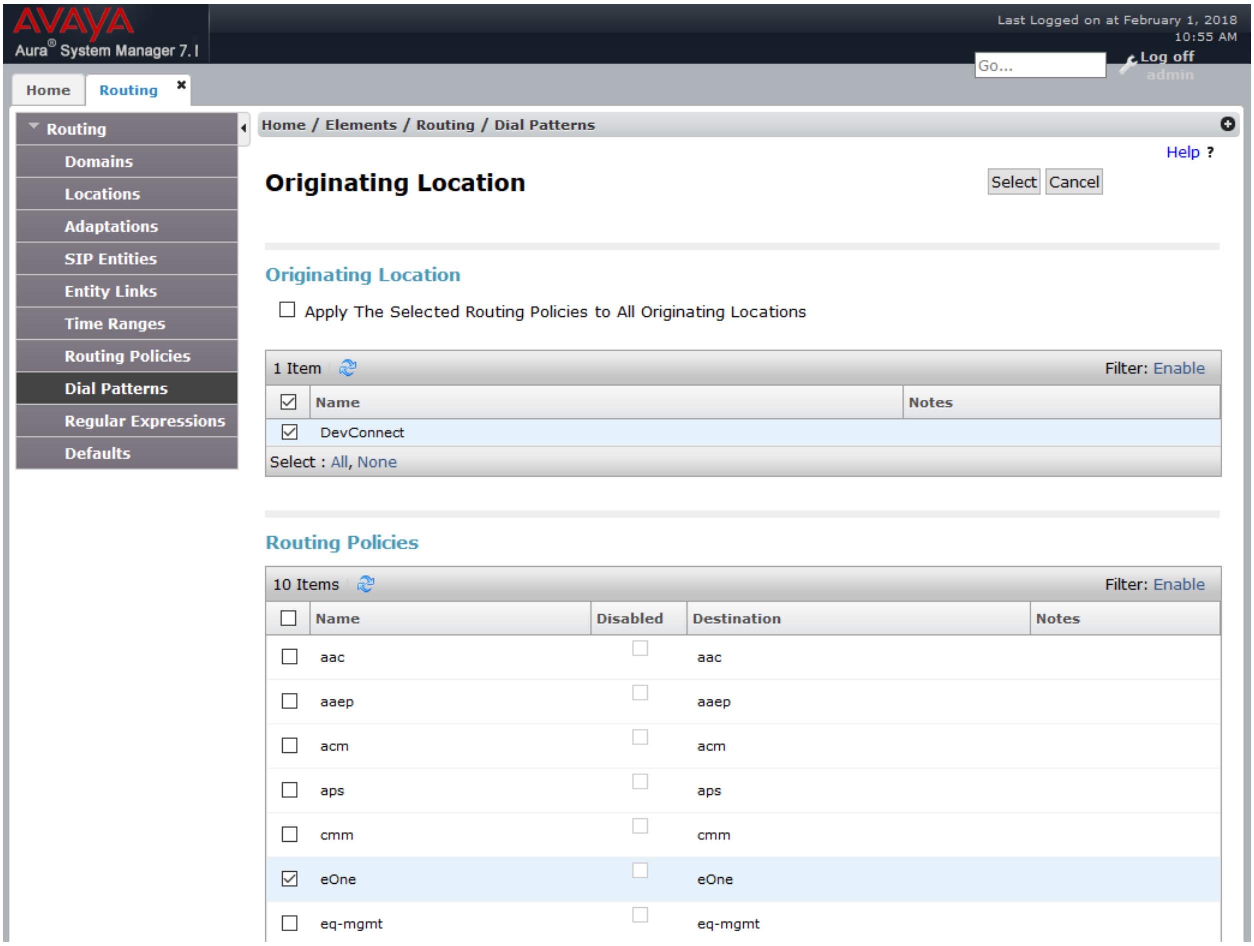
Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach eONE. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the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

- **Pattern:** A dial pattern to match, in this case "51111".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and select the routing policy for reaching eONE.



In the compliance testing, the policy allowed for call origination from “DevConnect”, and the eONE routing policy from **Section 6.3.1** was selected as shown below.

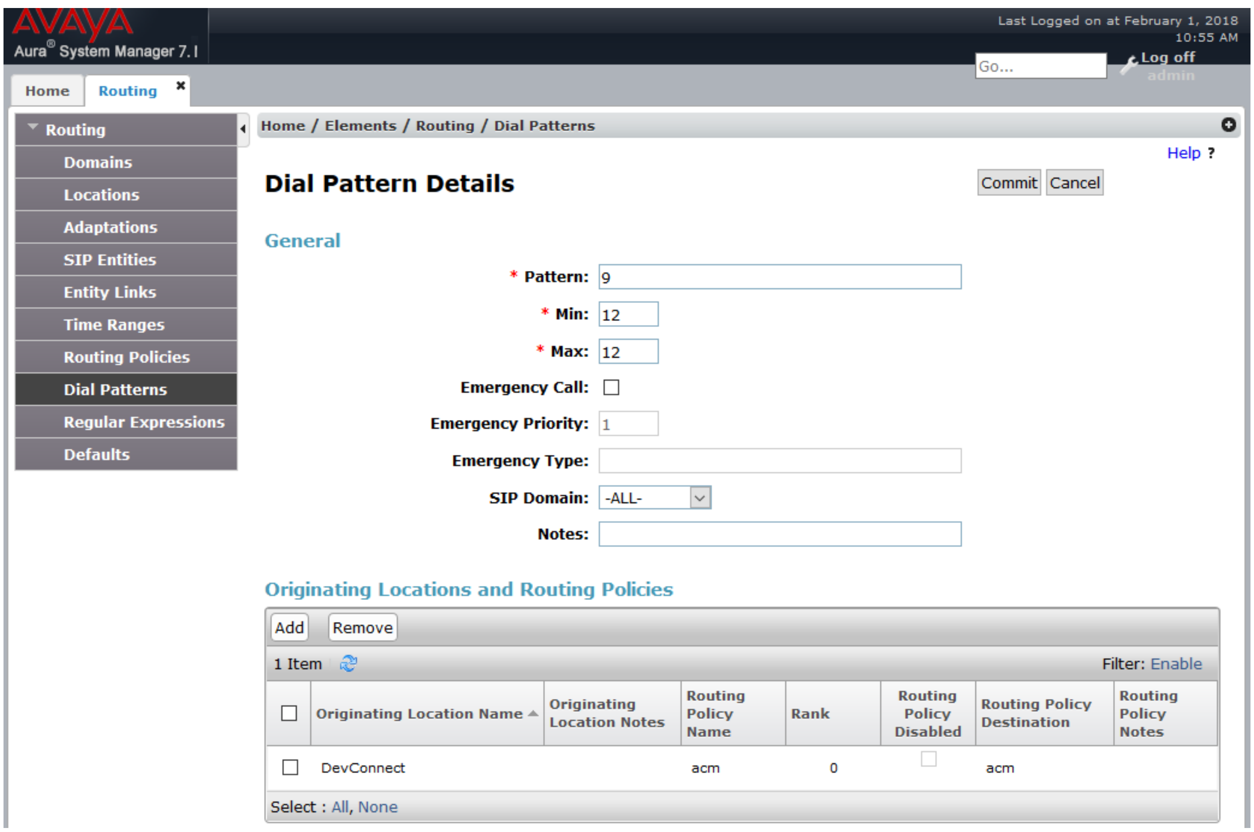


6.4.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "7200" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from eONE. In the compliance testing, the new policy allowed for call origination from the eONE location from **Section** Error! Reference source not found., and the Communication Manager routing policy from **Section 6.3.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to applicable Communication Manager dial patterns to reach the PSTN. In the compliance testing, eONE will add the prefix "9" for outbound calls to the PSTN, and therefore the existing dial pattern for "9" was also changed (not shown below).



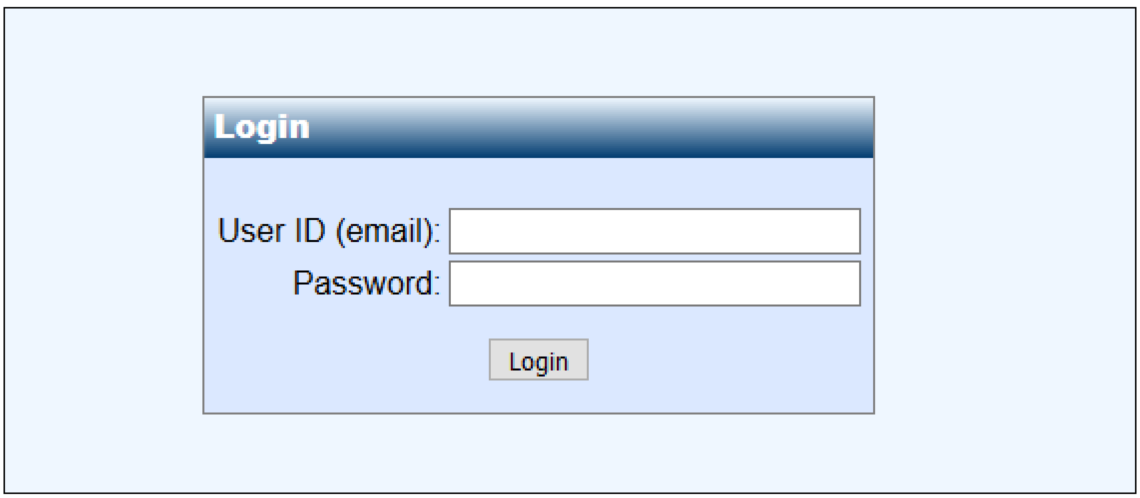
7. Configure Computer Instruments eONE

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

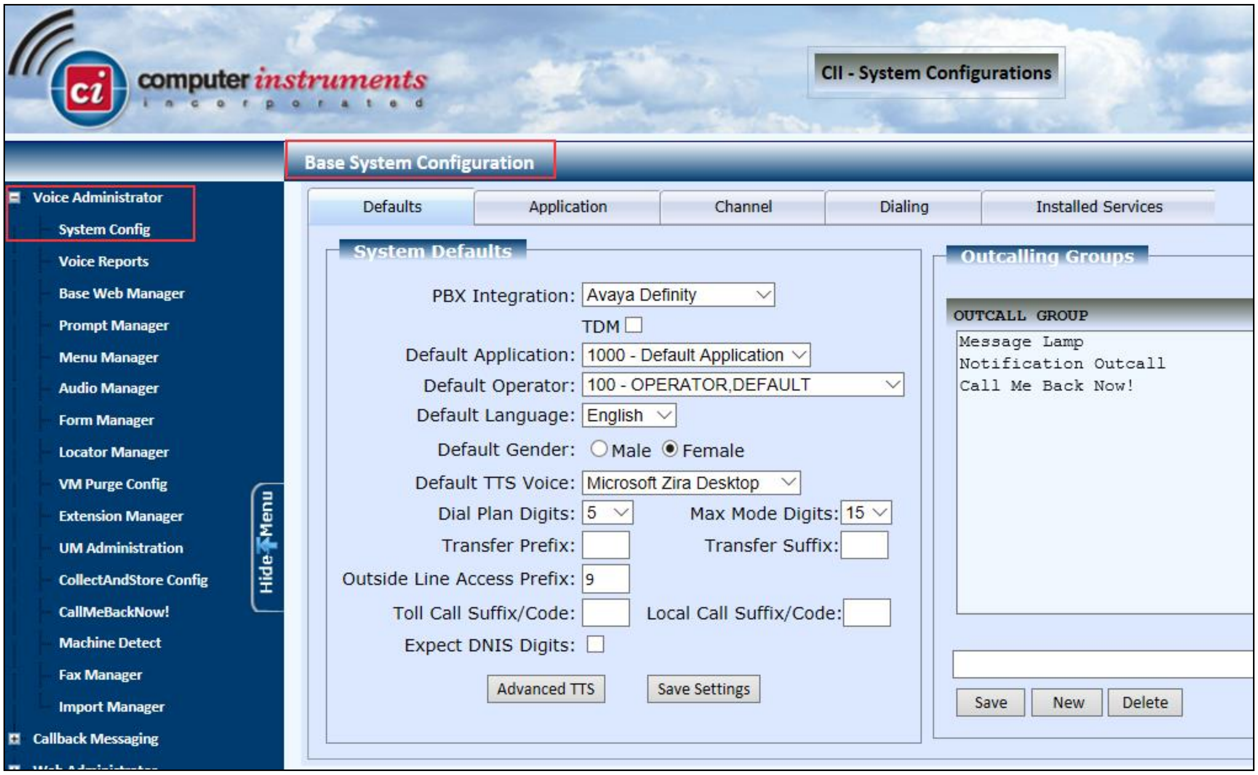
- Administer system config
- Administer EIVR.ini
- Restart service

7.1. Administer System Config

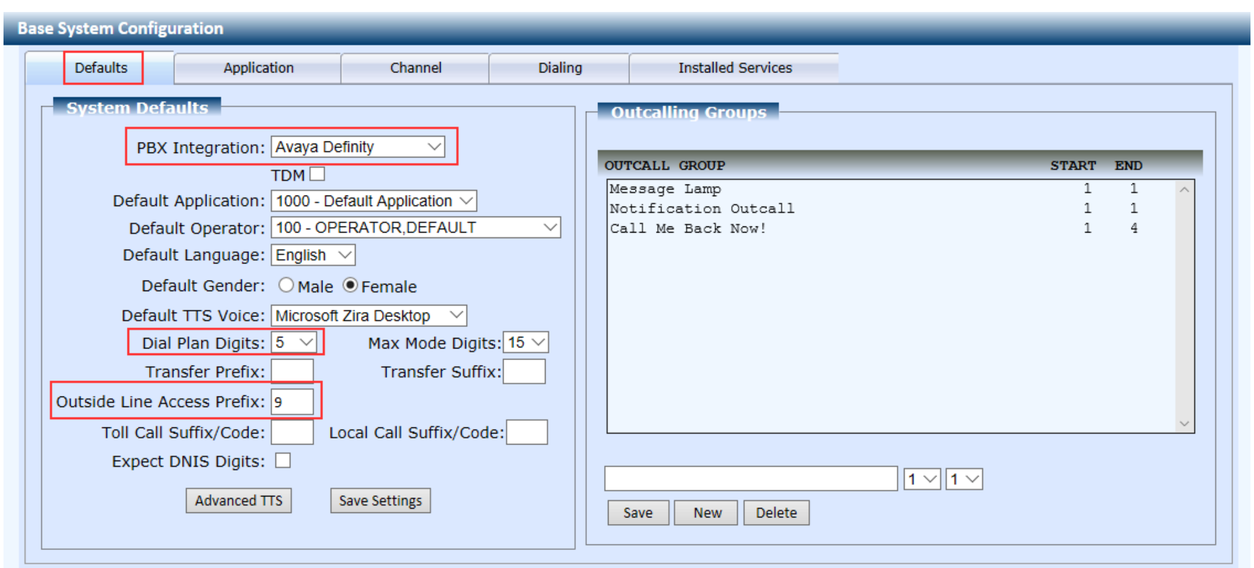
Computer Instruments engineers installed/licensed/configured eONE cloud IVR. This section shows what was configured by the Computer Instruments engineers. For more information, please contact the Computer Instruments support, mentioned in **Section 2.3**.
To access the **System Config** page, navigate to:
http://<ip-address>/eCI/VoiceAdmin/Default.aspx, where <ip-address> is the ip address of eONE server.
Provide appropriate credentials on the **Login** page.



In the **CII-Voice Administrator** page, select **Voice Administrator → System Config** in the left pane to display the **Base System Configuration** screen.



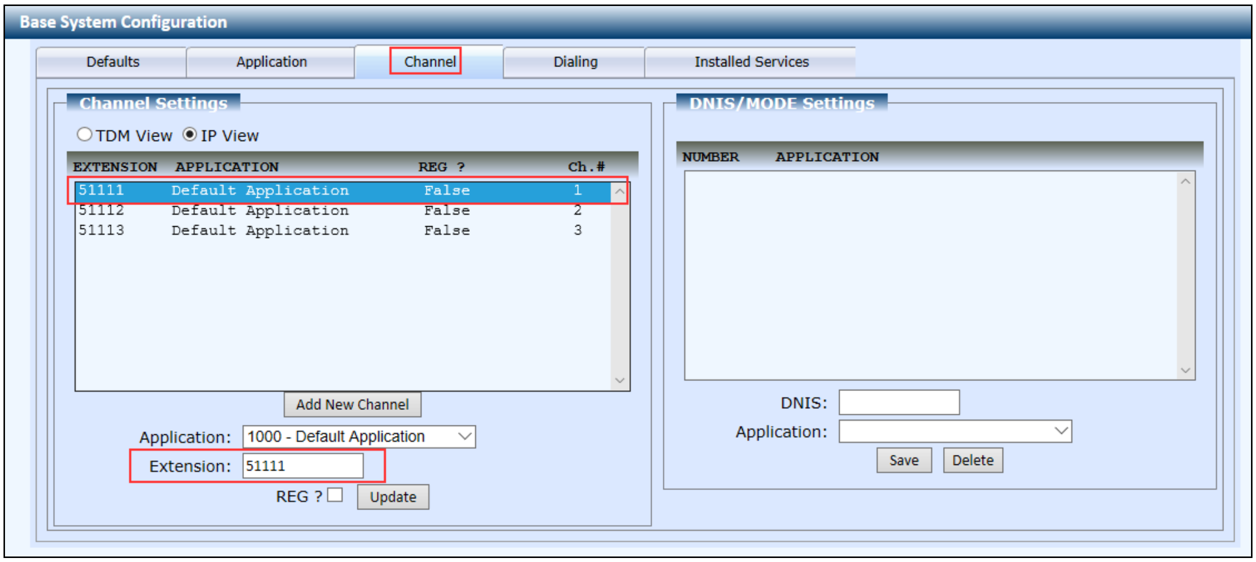
Select the **Defaults** tab from the top of the **Base System Configuration** screen. Select "Avaya Definity" for **PBX Integration**. For **Dial Plan Digits**, enter the maximum length of internal extensions on Avaya IP Office. For **Outside Line Access Prefix**, enter the applicable prefix for calls to the PSTN via Avaya IP Office.



Select the **Channel** tab from the top of the **Base System Configuration** pop-up screen.

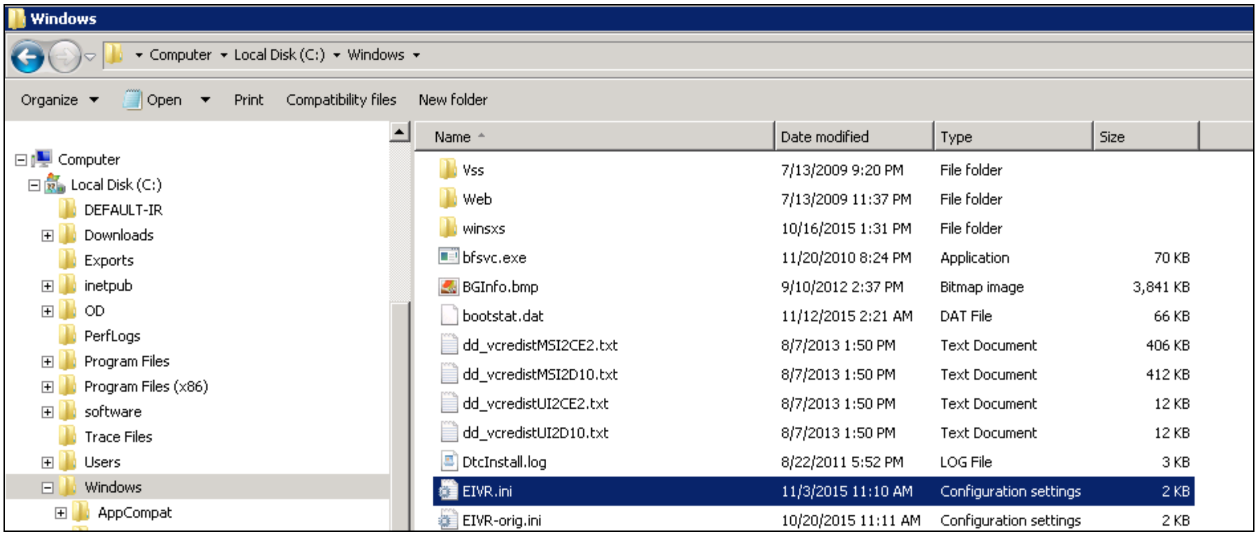
In the **Channel Setting** sub-section, select the first channel entry. For **Extension**, enter the applicable extension used for the inbound application, in this case "51111". By default, all third party channel resources are used for inbound applications unless otherwise specified. Select **Update** to update the extension value.

In the compliance testing, only one inbound application was used, and therefore only the first channel resource needed the extension mapping.

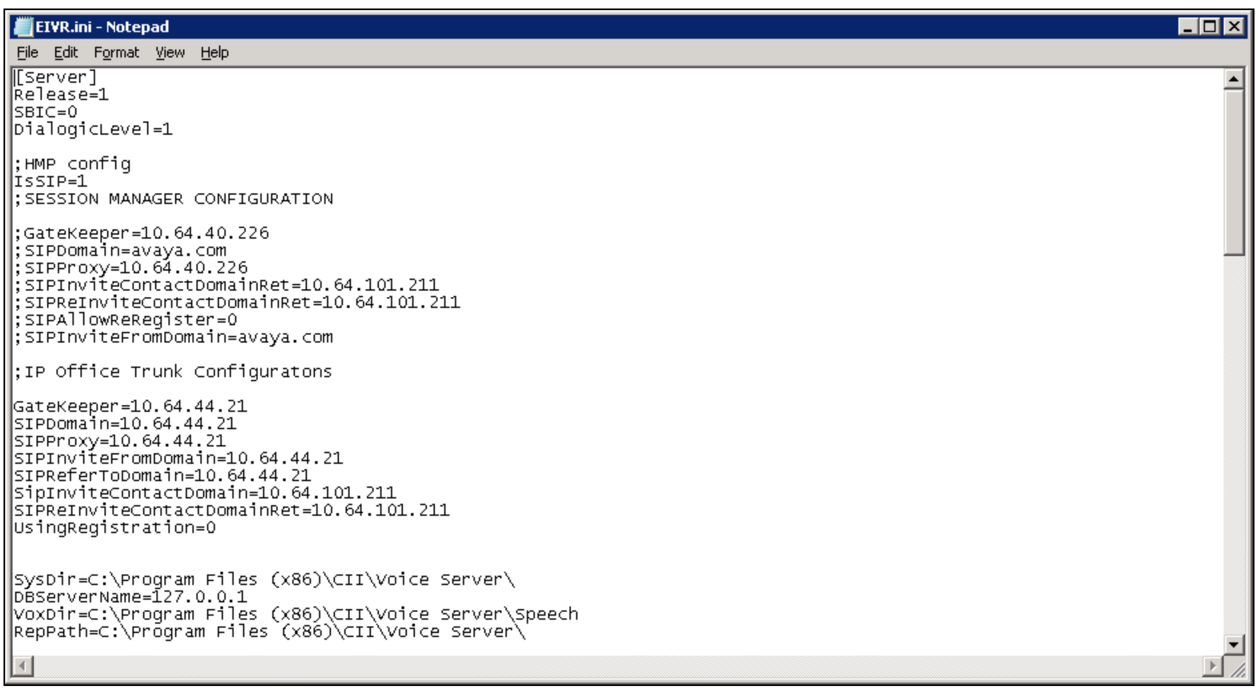


7.2. Administer EIVR.ini

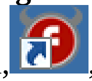
From the eONE server, navigate to the **C:\Windows** directory to locate the **EIVR.ini** file shown below.

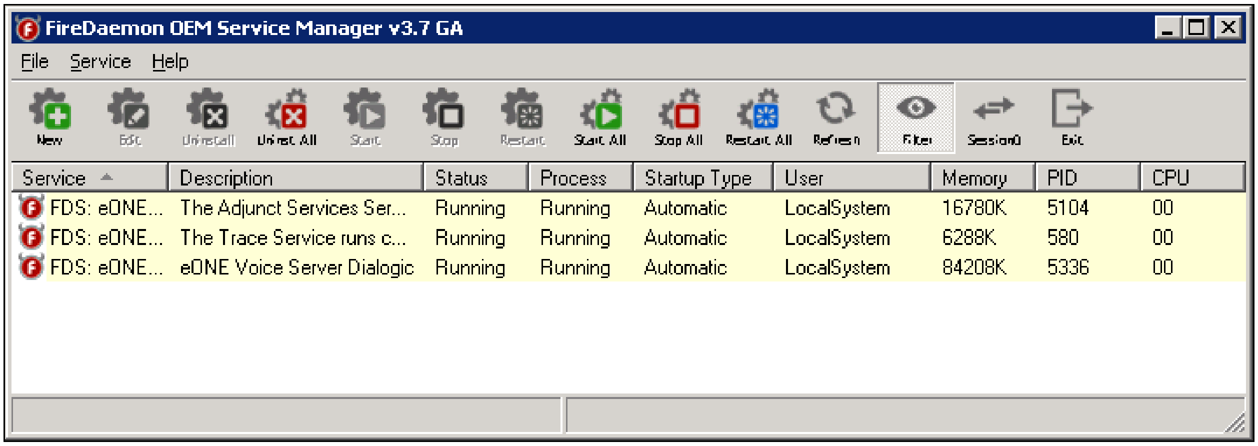


Open the **EIVR.ini** file with the Notepad application. Configure the parameters as shown below, where “10.64.110.65” is the IP address of Session Manager, “192.168.10.134” is the IP address of the eONE server, and “avaya.com” is the domain name. During the compliance test, the domain name is converted to IP address in the hosts file.



7.3. Restart Service

Run the **C:\Program Files (x86)\FireDaemon OEM\FireDaemonUI.exe** or select the **Service. Manager** icon,, from Desktop to display the screen below. Restart the **eONE Voice Server Dialogic** service and verify that the **Status** is *Running* as shown below.



8. Verification Steps

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

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 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
```

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 5.4**. Verify that the **Group State** is "in-service", as shown below.

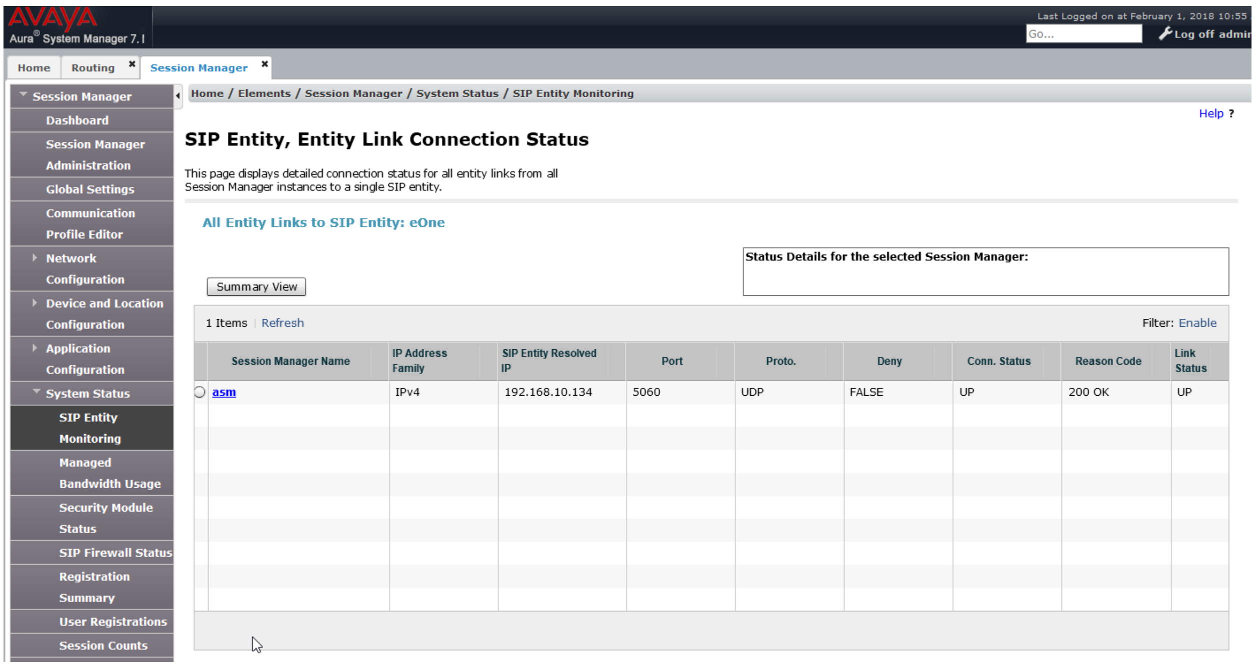
```
status signaling-group 1
                         STATUS SIGNALING GROUP

        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).

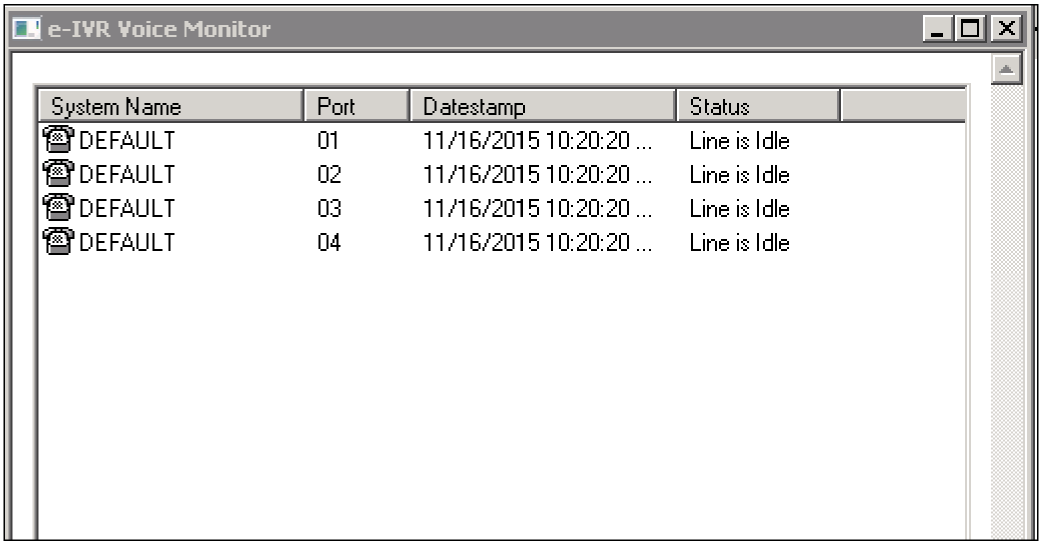
Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the eONE entity name from **Section 6.2.1**.

The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "Up".



8.3. Verify Computer Instruments eONE

Select the **Voice Monitor** icon, , from Desktop to display the **eONE Voice Monitor** screen. Verify that the **Status** for all ports is "Line is Idle", as shown below.



9. Conclusion

These Application Notes describe the configuration steps required for Computer Instruments eONE to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. All feature and serviceability test cases were completed.

10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 10, Release 7.1, August 2017, available at http://support.avaya.com.

2. *Administering Avaya Aura® Session Manager*, Release 7.1, Issue 7, September 2017, available at http://support.avaya.com.

3. *Installing eONE*, available from http://www.instruments.com.

4. *eONE Application Server*, available from http://www.instruments.com.

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.