Avaya Solution & Interoperability Test Lab

Application Notes for Enghouse Interactive Communications Portal 10.4 using Dialogic Host Media Processing 3.0 with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3 with TLS/SRTP- Issue 1.0

Abstract

These Application Notes describe the configuration steps for Enghouse Interactive Communications Portal 10.4 using Dialogic Host Media Processing 3.0 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3. Communications Portal is an IVR application that connects to Session Manager via a SIP Trunk.

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 Enghouse Interactive Communications Portal 10.4 using Dialogic Host Media Processing (DHP) 3.0 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3 using Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP) via a SIP Trunk. Enghouse Interactive Communications Portal is an open, standards-based platform with integrated application development and management components. Enghouse Interactive Communications Portal provide voice self service solutions, such as interactive voice response, outbound dialing, voice recording and speech enabled self service systems.

2. General Test Approach and Test Results

Interoperability testing contained functional tests mentioned in **Section 2.1**. All test cases were performed manually. The general test approach was to validate successful handling of inbound/outbound calls to and from the Communications Portal (CP) to verify IVR application telephony functionality. The IVR application telephony functionality of CP was the only module tested. This IVR application (CP script) connects to Session Manager via a SIP Trunk. Session Manager directs inbound calls over SIP trunk to CP script which in turn handles the call based on the digits dialed using SIP signaling. CP uses Dialogic HMP to perform all telephony functions on the server.

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 this Application Note, the interface between Avaya systems and Enghouse Interactive Communications Portal solution utilized enabled securities capabilities with TLS/SRTP.

2.1. Interoperability Compliance Testing

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

- **Basic Inbound/Outbound** – Tests inbound/outbound calls to and from CP.
- G.711A, G.711U codecs support and negotiation (without shuffling).
- **Call Forward** from Avaya Endpoints to CP.
- **Call Hold** – Tests held calls to/from CP.
- **Call Transfer** – Tests transferred calls to/from CP.
- **IVR Functionality** – Tests of various IVR features such as ANI/DNIS detection, voice recording, DTMF collection on the CP.
- **Failover/Service** – Tests the behaviour of CP when there are certain failed conditions such as Media shuffling, session refresh, options, long hold, multiple calls, isolated network from CP server, power off/on CP server

2.2. Test Results

The testing was successful. All test cases passed.

2.3. Support

Technical support on Communications Portal can be obtained for Enghouse Interactive as follows:
USA

- Email: scpsupport@enghouse.com
- Website: http://enghouseinteractive.com/support.php
- Phone: +1 800.788.9730 Self-Service
- Phone: +1 800.872.2272 Live-Service

EMEA

- Email: envoxsupport@enghouse.com / supportenvox@syntellect.com
- Website: http://www.enghouseinteractive.com/services/support/
- Phone: +44 870.220.2205

3. Reference Configuration

Figure 1 illustrates a sample configuration that consists of Avaya products and CP with Dialogic HMP.

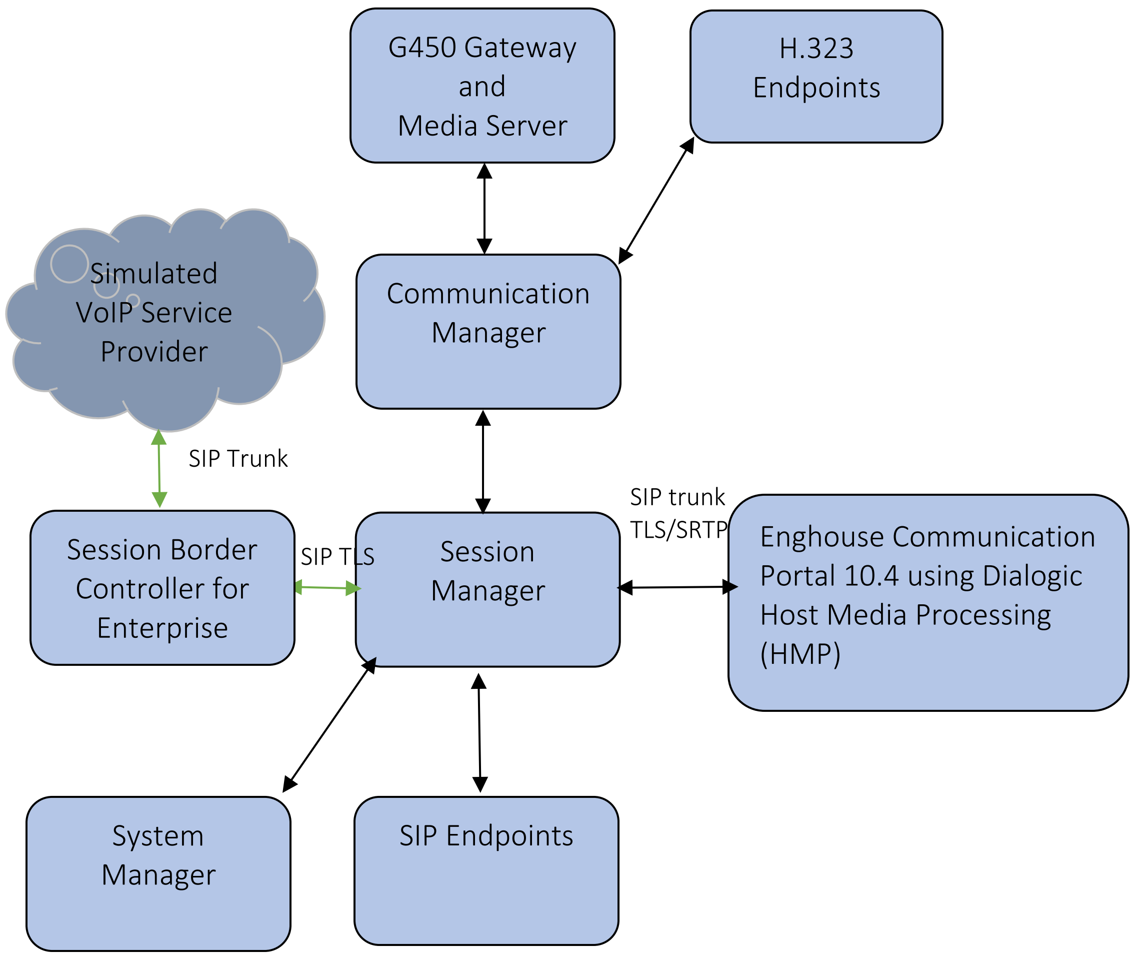


Figure 1: Test Configuration for Enghouse Interactive Communications Portal using Dialogic Host Media Processing (HMP) and the Avaya Platform.

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 J159 H323 Deskphone	6.8
Avaya Workplace Client for Windows	3.22.0
Avaya J179 & J189 SIP Deskphone	4.0.9
Avaya Vantage™ K175 & K155	3.1.0.0
Enghouse Communications Portal - Dialogic Host Media Processing	10.4.19.9632 3.0 Service Update 525

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 Signaling Group and Trunk Group
- Administer Dial Plan
- Administer Route Selection for Communications Portal 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 uses 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 two 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




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

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 **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 **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: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
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 codecs supported for calls routed over the SIP trunk to CP. 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 Communications Portal. 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 with non-secure RTP 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:
```

In the **node-names ip** form, note the IP address of the **procr** and the 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
```

5.4. Configure SIP 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 CP. 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 **n** and **Peer Server** set to **SM**.
- 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
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Enghouse Interactive to prevent unnecessary SIP messages during call setup. Session refresh is used throughout the duration of the call, to check the other side has not gone away, for the compliance test a value of **900** was used.

```
change trunk-group 1                                            Page   2 of   4
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

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
```

Settings on **Page 4** are as follow.

```
change trunk-group 1                                            Page   4 of   4
                              PROTOCOL VARIATIONS

                                         Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? y
                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
     Resend Display UPDATE Once on Receipt of 481 Response? y
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
          Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

Note: With the field **Resend Display UPDATE once on Receipt of 481 Response** is set to **y**, Communication Manager will send a SIP UPDATE message for 481 response received from far end to avoid display incorrectly in some race condition cases.

5.5. Administer Dial Plan

It was decided for the compliance testing that all calls beginning with 3 and a total length of 5 digits were to be sent across the SIP trunk to Session Manager and therefore to CP. 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 **3** 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             10 udp
 7             5  ext
 8             5  ext
 9             1  fac
 *             3  fac
 #             3  dac
```

5.6. Administer Route Selection for Communications Portal Calls

As digits **3**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 **3** 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 3                                    Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                           Percent Full: 0

  Matching                      Insert                Node
  Pattern            Len Del    Digits       Net Conv Num
 3                   5   0                   aar n
 4                   10  0                   ars n
                                                 n
```

Use the **change aar analysis** x command to further configure the routing of the dialed digits. Calls to Communications Portal begin with **3** 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
    3                     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 via System Manager. The procedures include the following areas:

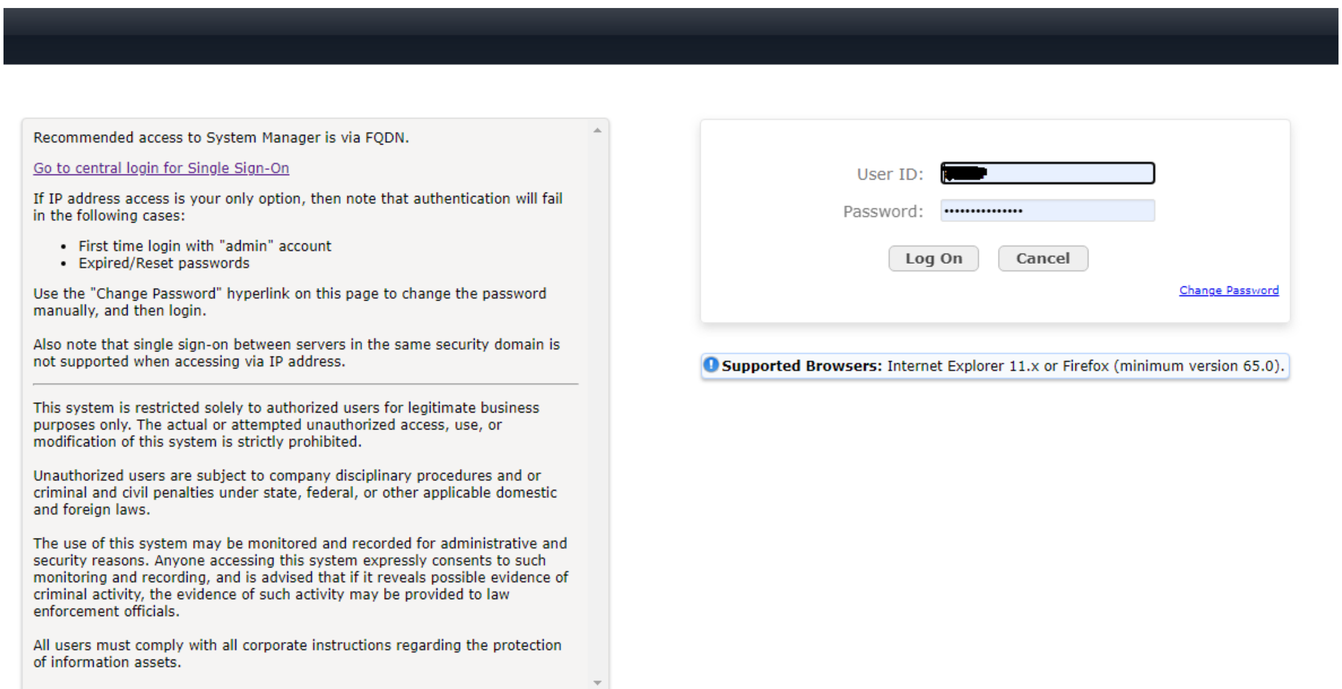
- Configure SIP Entities
- Configure Routing Policies
- Configure Dial Patterns

6.1. Configure SIP Entities

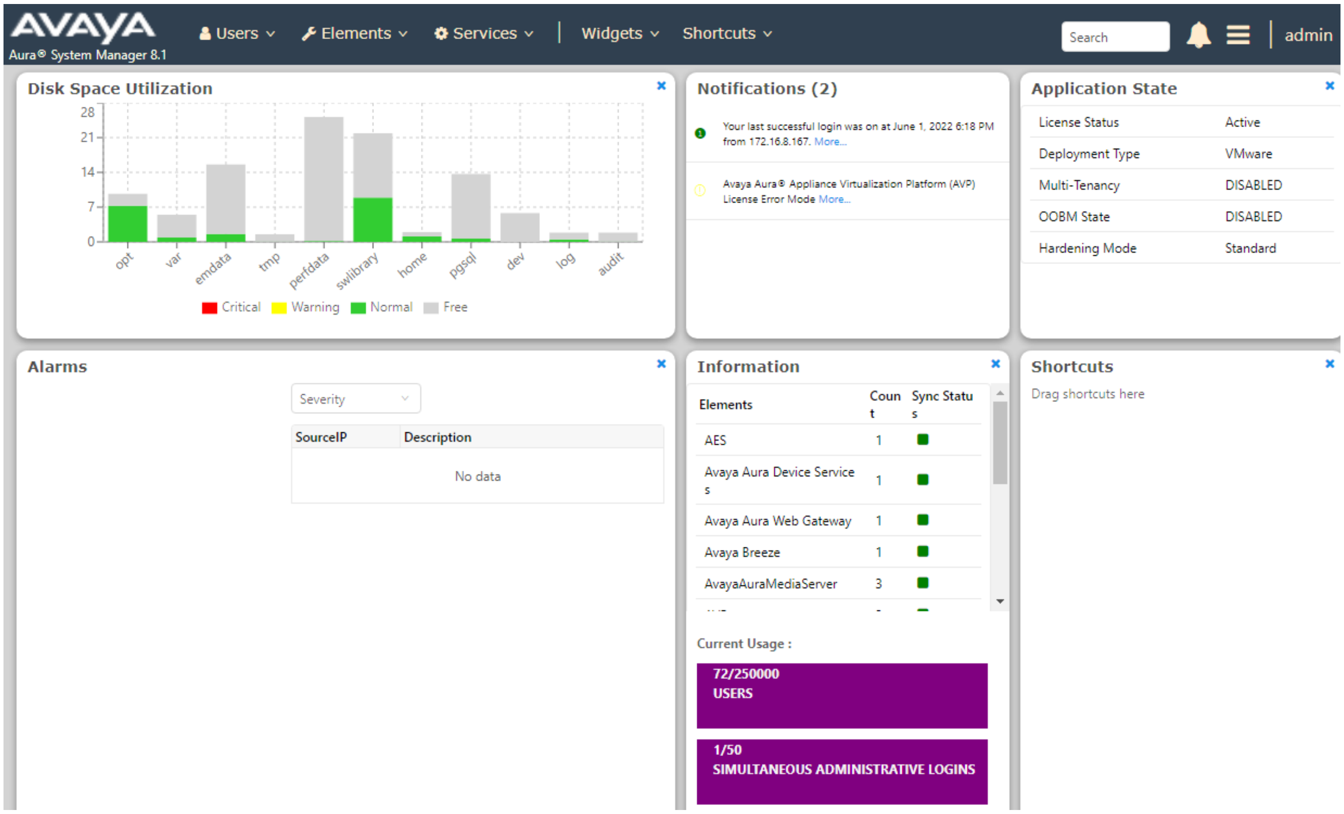
This section provides the procedures for configuring SIP Trunk from Session Manager to Communication Manager and CP

6.1.1. Configure SIP Entity for Communications Portal

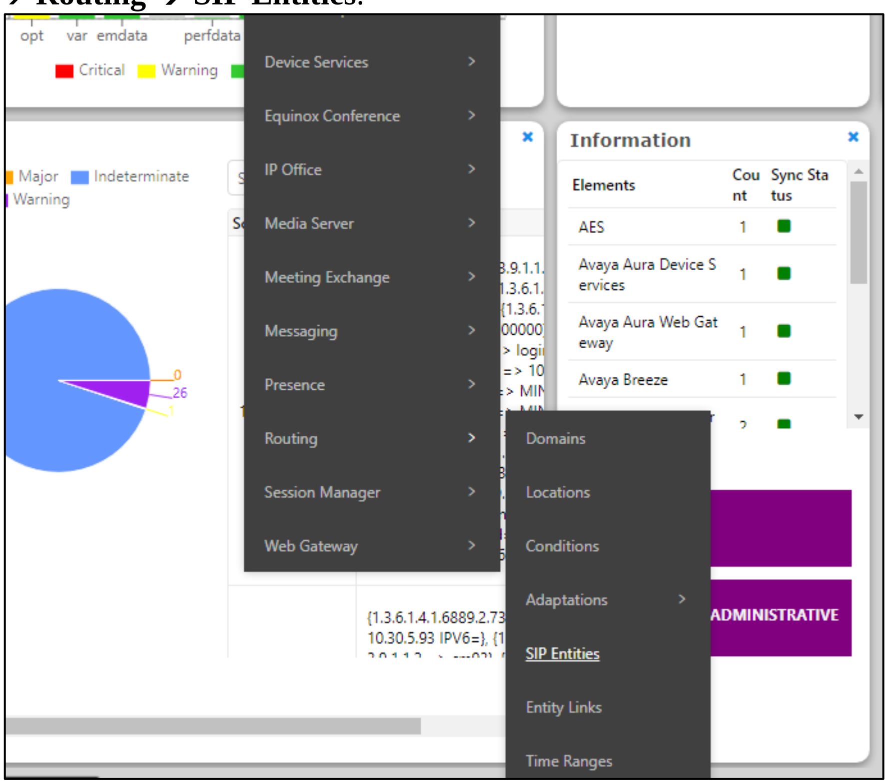
Configuration of SIP Entities is performed via System Manager. Access the System Manager Administration web interface by entering the System Manager (SMGR) URL in a web browser. Log in using appropriate credentials.



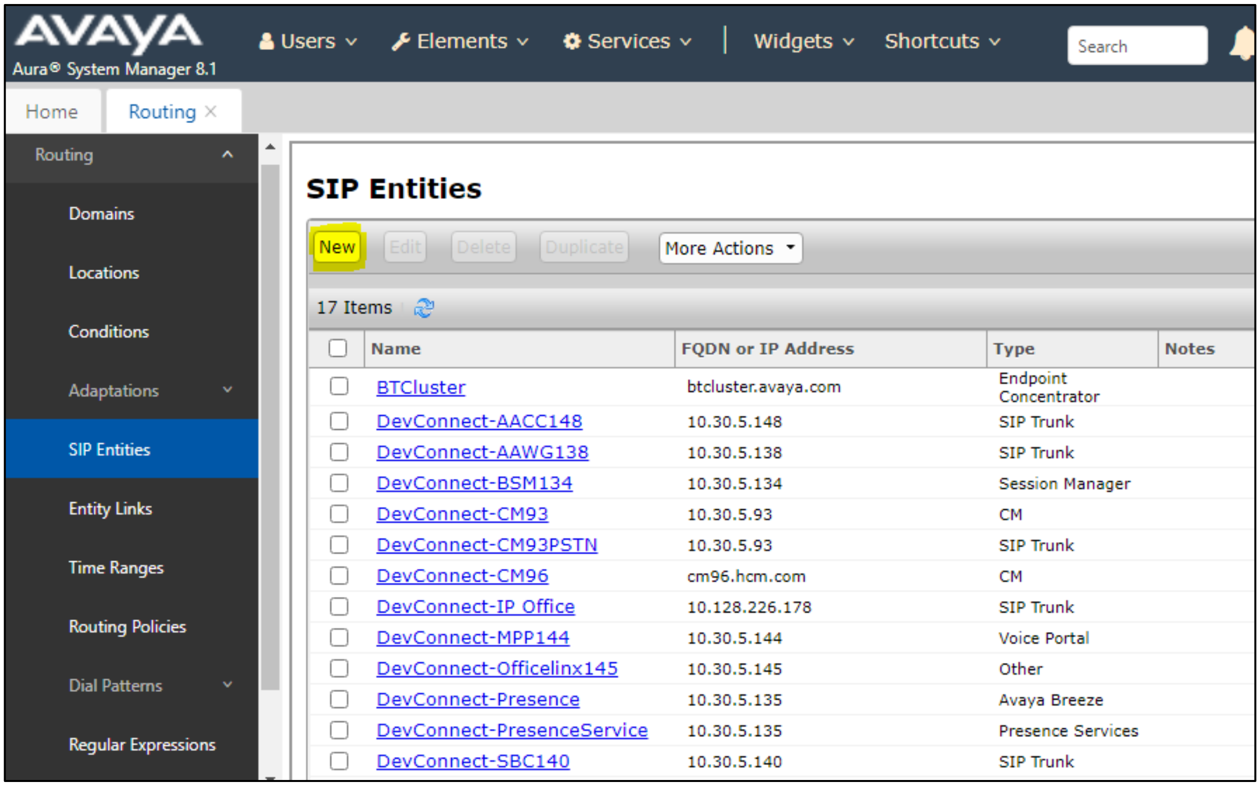
Once logged in, the following screen is displayed.



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



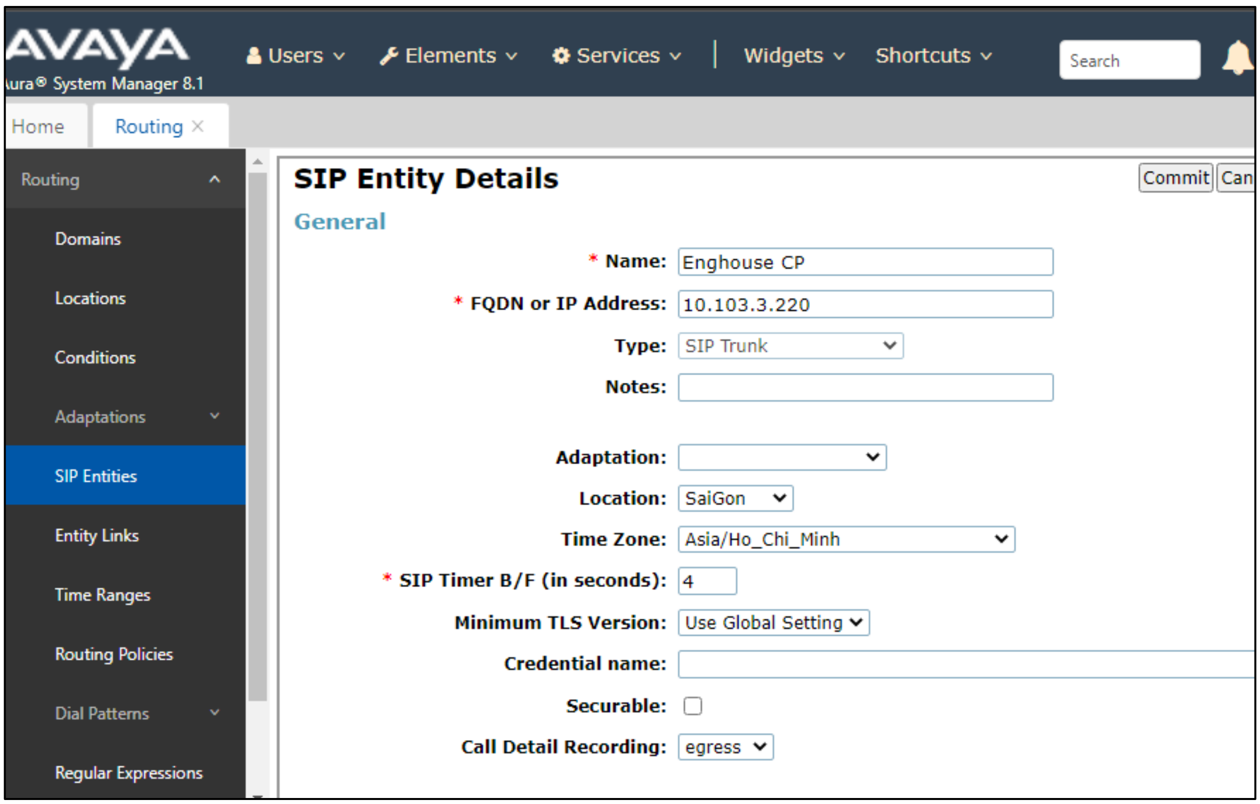
On **SIP Entities** page, press **New** to create new **SIP Entity**.



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

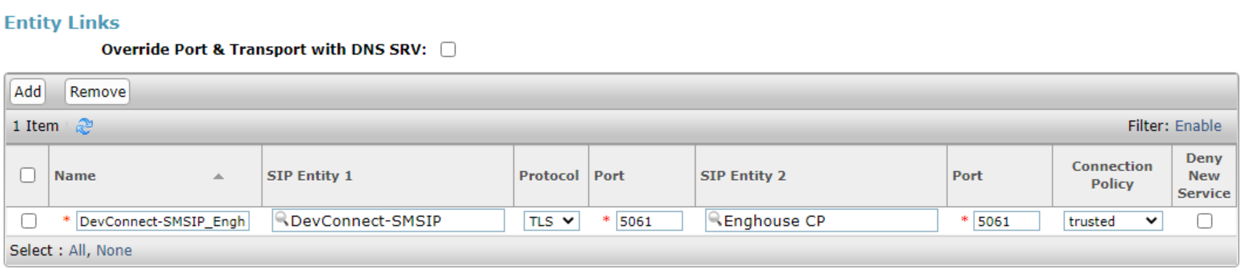
- **Name:** A descriptive name, example **Enghouse CP**.
- **FQDN or IP Address:** The proc IP address of CP.
- **Type:** **SIP Trunk**
- **Notes:** Any desired notes.
- **Location:** Select the applicable location.
- **Time Zone:** Select the applicable time zone.

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:** **TLS**
- **Port:** **5061**
- **SIP Entity 2:** The Communications Portal entity name from this section, in this case "Enghouse CP".
- **Port:** **5061**
- **Connection Policy:** **trusted**

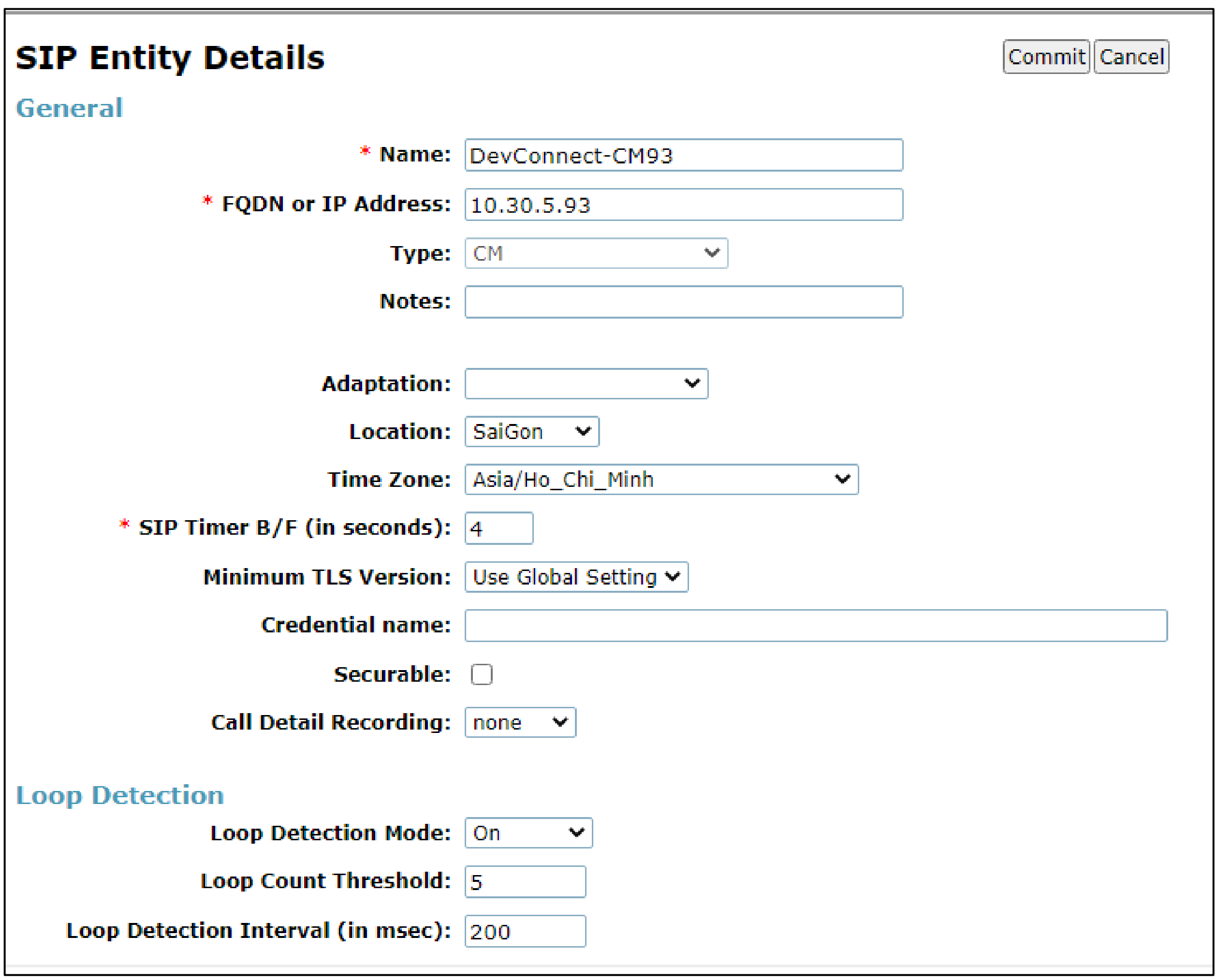


6.1.2. Configure SIP Entity for Communication Manager

Add new 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.

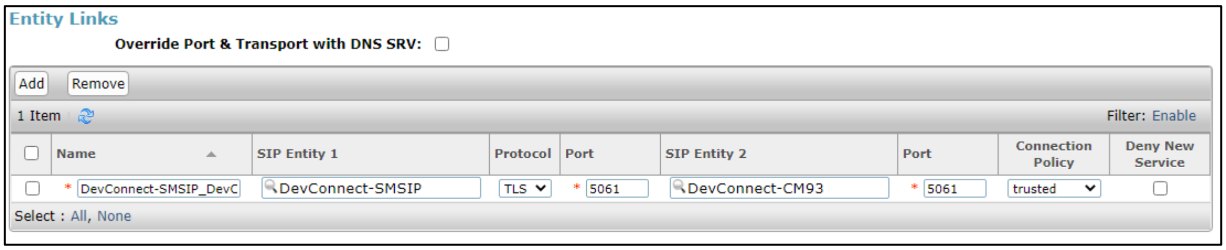
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name, example **DevConnect-CM93**.
- **FQDN or IP Address:** The proc IP address of Communication Manager.
- **Type:** **CM**
- **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 Communication Manager entity name from this section, in this case **DevConnect-CM93**
- **Port:** **5061**
- **Connection Policy:** **trusted**

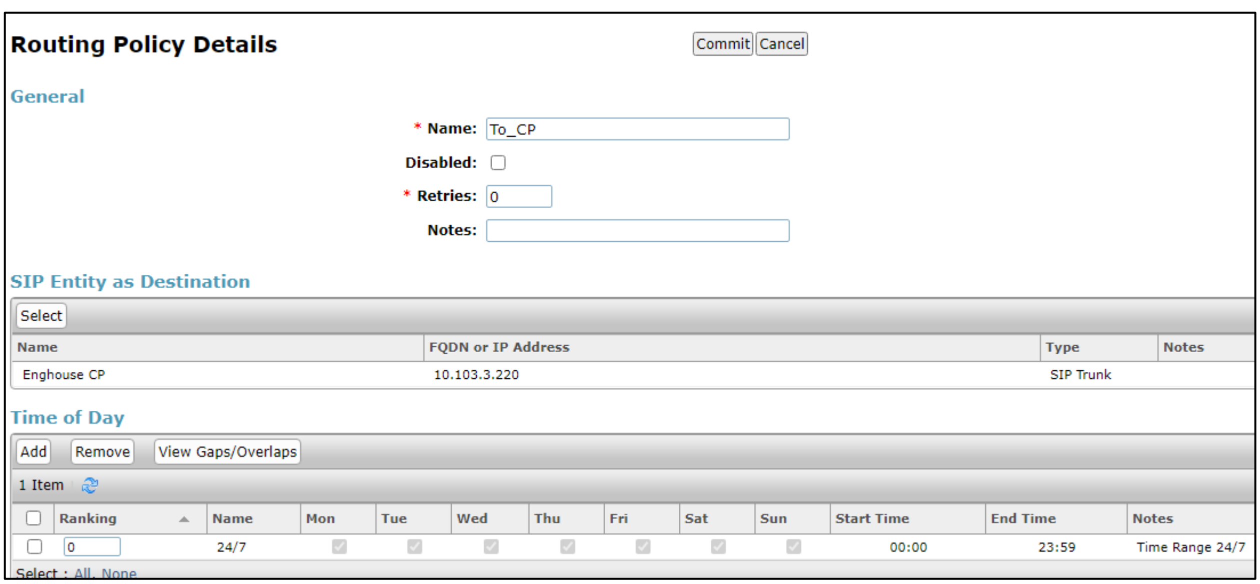


6.2. Configure Routing Policy for Communications Portal

This section to add a new routing policy for routing calls to Communications Portal. 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.

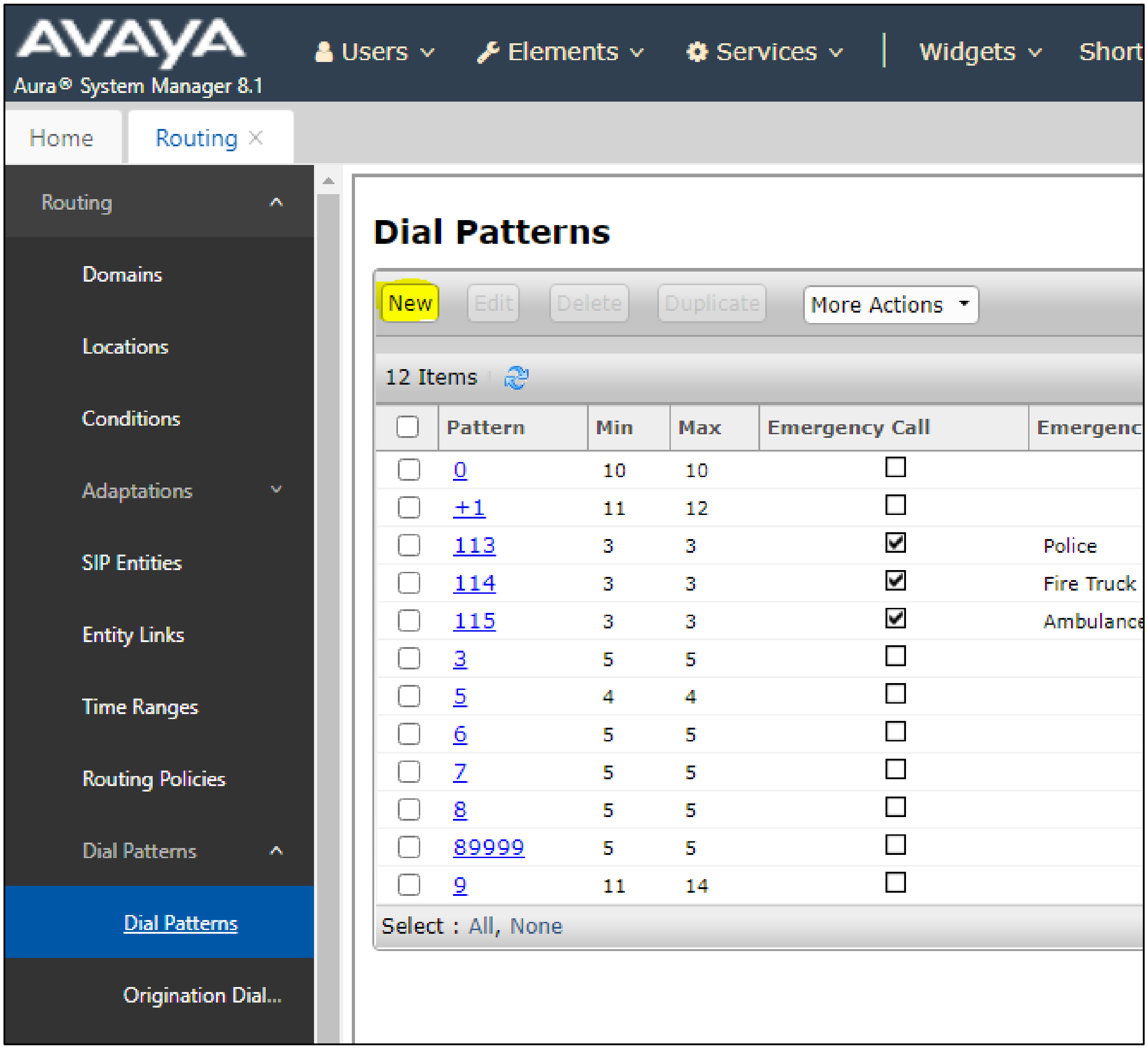
The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**. Enter optional **Notes**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communications Portal entity name from **Section 6.1.1**.

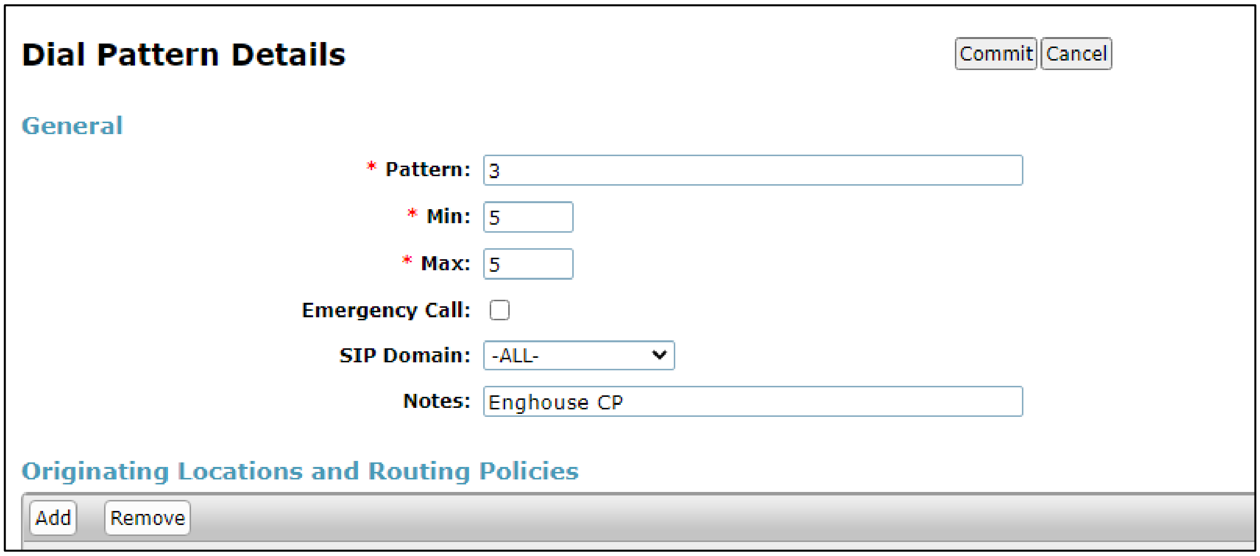


6.3. Configure Dial Pattern for Communications Portal

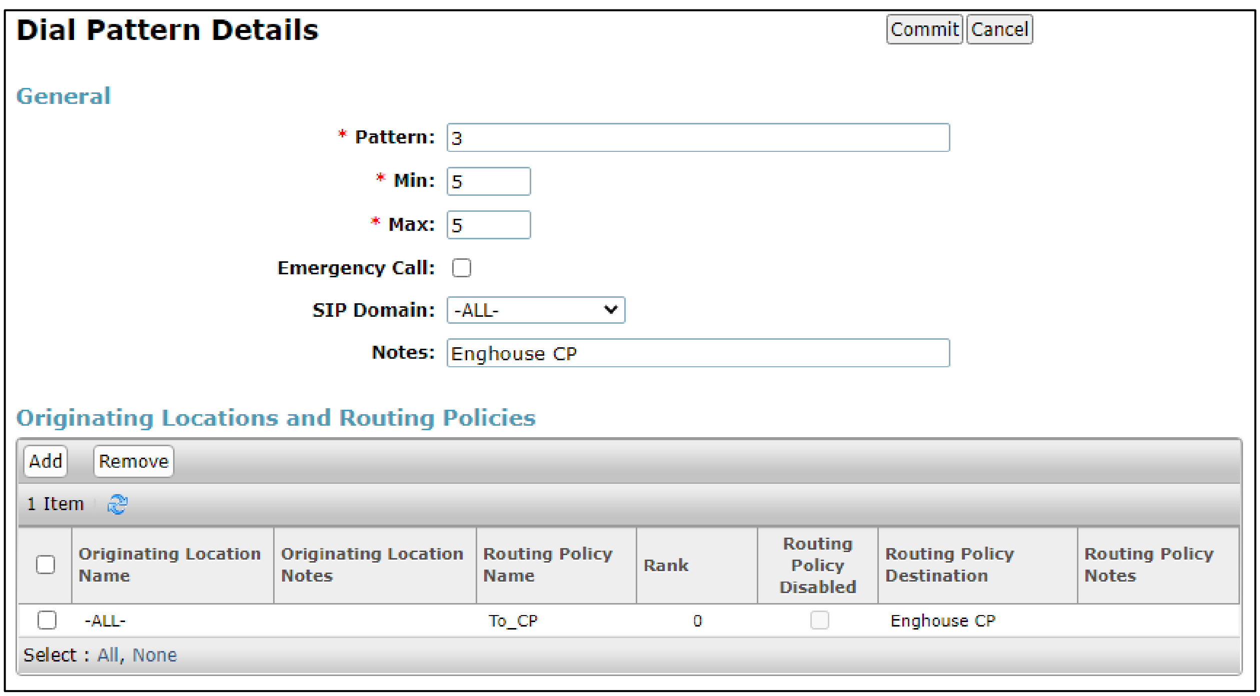
In order to route calls to the Communications Portal a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.



The **Dial Pattern Details** screen is displayed. Enter the number to be routed noting this will be the same number outlined in **Section 5.5**. 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 **3xxxx** were routed to Communications Portal. Click on **Commit** as shown below to save configuration.



7. Configure Enghouse Interactive Communications Portal

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

- Generate CSR and Private key for TLS SIP Trunk
- Configuration of Communications Portal

7.1. Generate CSR and Private key for TLS SIP Trunk

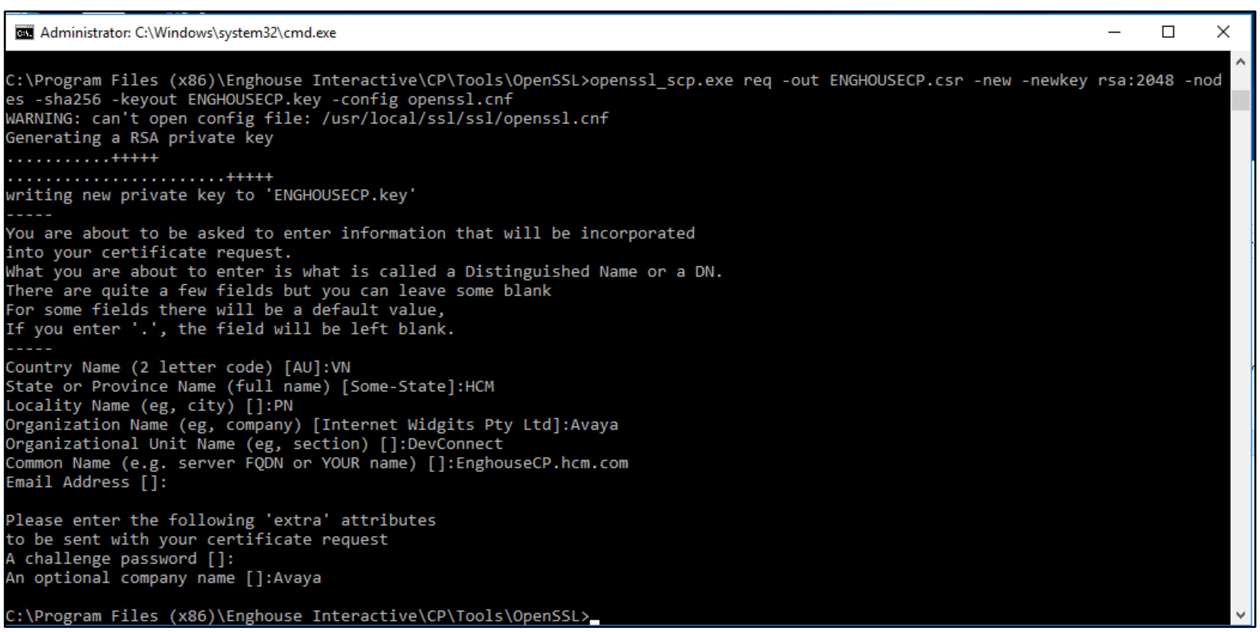
To enable TLS, generating a Cert Signing Request (CSR) and private key on the CP server system first.

- Use command prompt and open folder <Communications Portal install folder>\ Tools\OpenSSL and enter following command:

openssl_scp.exe req -out ENGHOUSECP.csr -new -newkey rsa:2048 -nodes -sha256 - keyout ENGHOUSECP.key -config openssl.cnf

Enter following information as below:
Country Name (2 letter code) [AU]:**VN**
State or Province Name (full name) [Some-State]:**HCM**
Locality Name (eg, city) []:**PN**
Organization Name (eg, company) [Internet Widgits Pty Ltd]:**Avaya**
Organizational Unit Name (eg, section) []:**DevConnect**
Common Name (e.g. server FQDN or YOUR name) []:**EnghouseCP.hcm.com**
Email Address []:

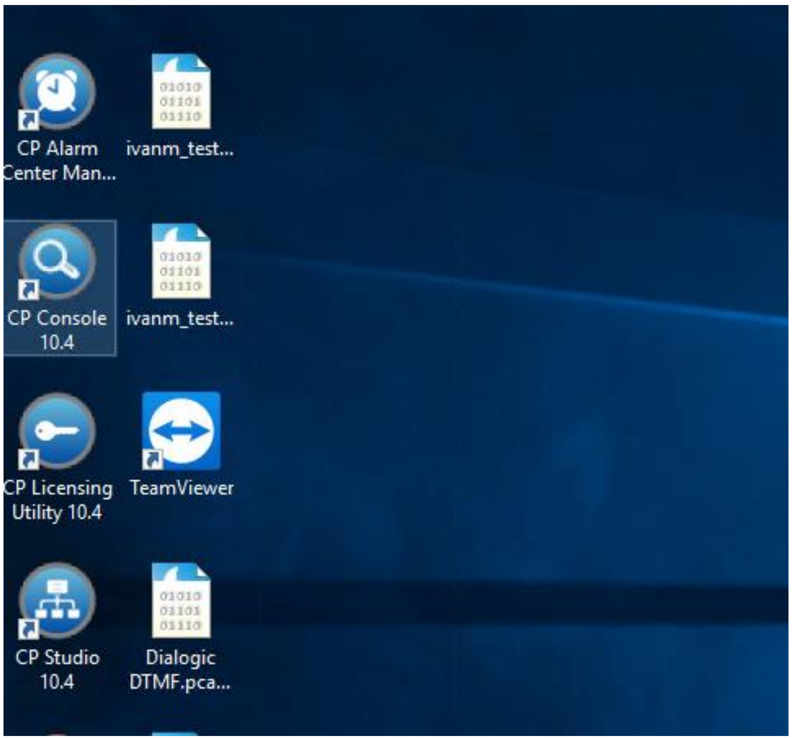
Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:
An optional company name []:**Avaya**



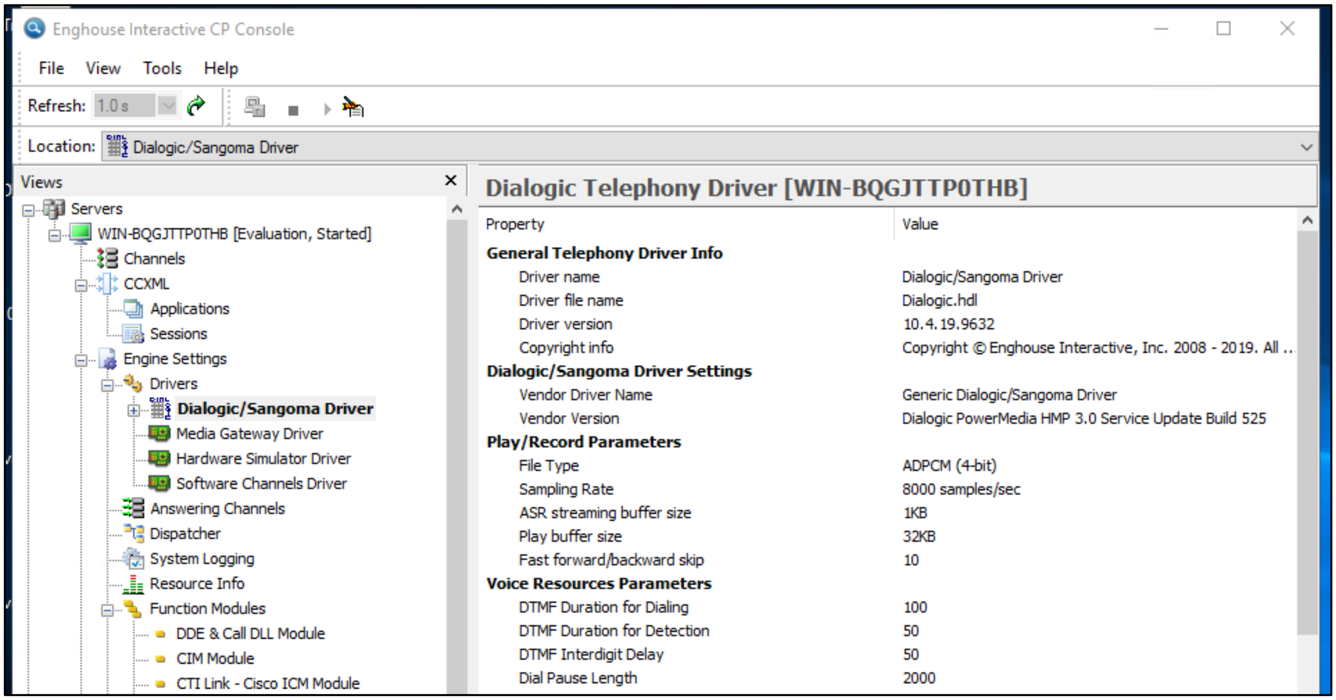
Cert Signing Request (CSR) file **ENGHOUSECP.csr** and private key file **ENGHOUSECP.key** are generated. CSR file then can be sent to Avaya which can make the Identity Certificate (.pem file).

7.2. Configuration of Communications Portal

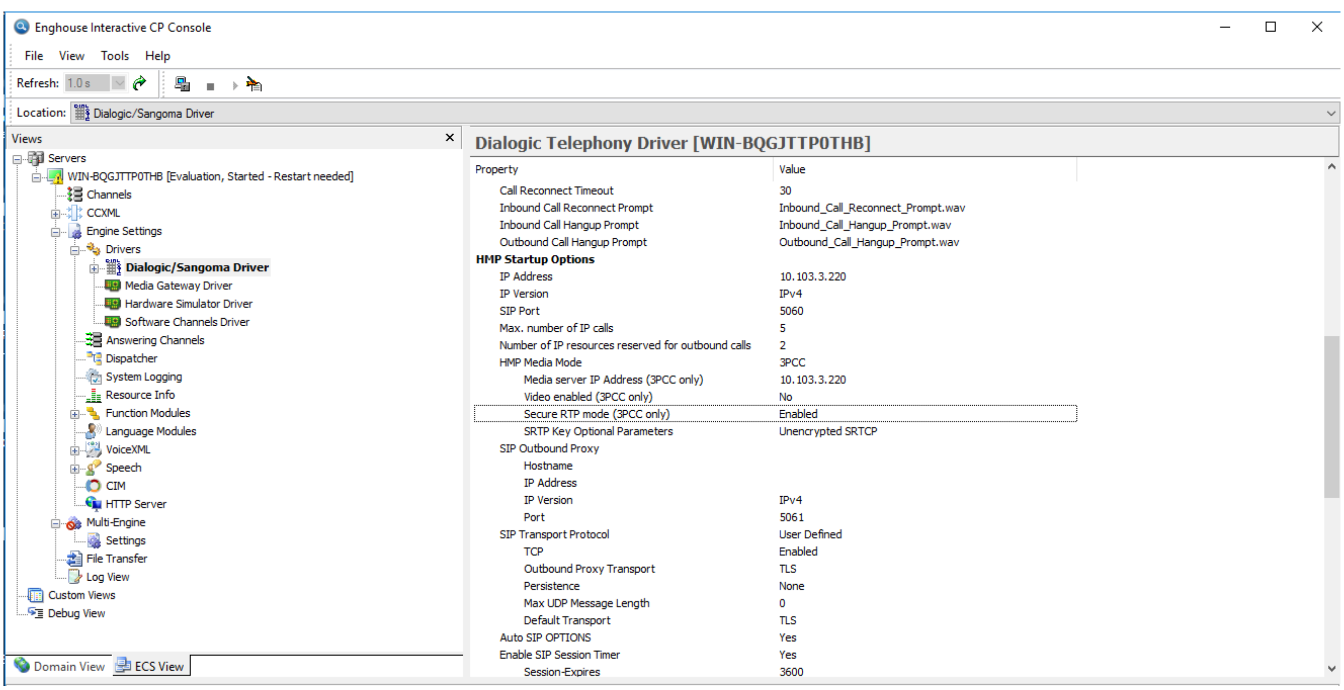
The Telephony module of Communications Portal which provides the connection to Session Manager is provided by a Dialogic Boards Driver. This driver completely caters for the telephony module of this solution. To configure the Dialogic Boards Driver, open the **CP Console 10.4** by double clicking on the shortcut as shown below.



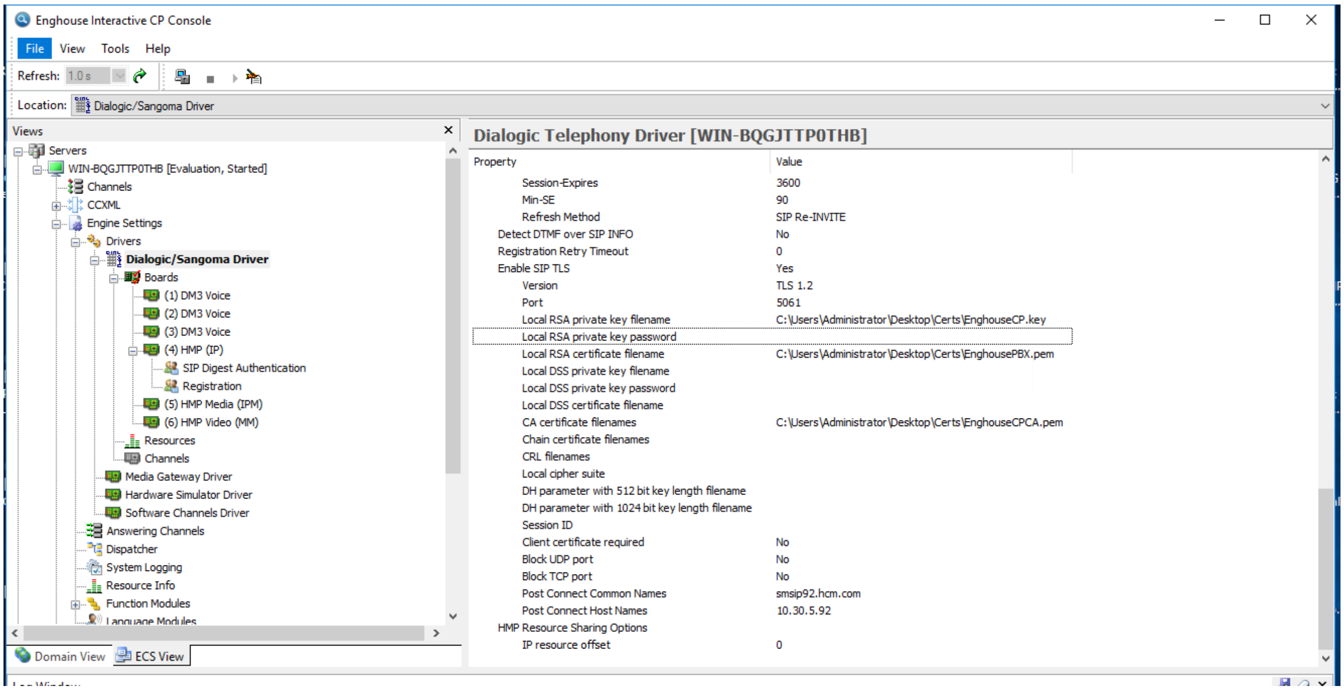
In the left window, navigate to **Servers→**[**Server Name**]**→Engine Settings → Drivers → Dialogic/Sangoma Driver**.



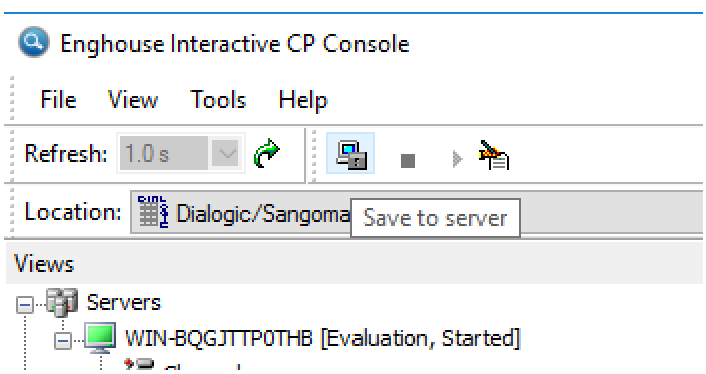
In the main window scroll down to **HMP Startup Options**, ensure that **Auto SIP OPTIONS** is set to **Yes**. **SIP Transport Protocol** is set to **User Defined** and **TLS** is **enabled**, also note the **Default Transport** is set to **TLS**. And in **HMP Media Mode** select **3PCC**.



Scroll down, Select **Enable SIP TLS** as **Yes** with **Version TLS 1.2** and **Port 5061**. In **Local RSA private key filename** field select private key with full location path already created on **Section 7.1**. Select **Local RSA certificate filename** with identity certificate and **CA certificate filenames** with all trusted CA certificate. Enter **Post Connect Common Names** with Avaya Session Manager SIP FQDN and **Post Connect Host Names** with SM SIP IP Address.



Click Save to server to save all configuration and restart CP.

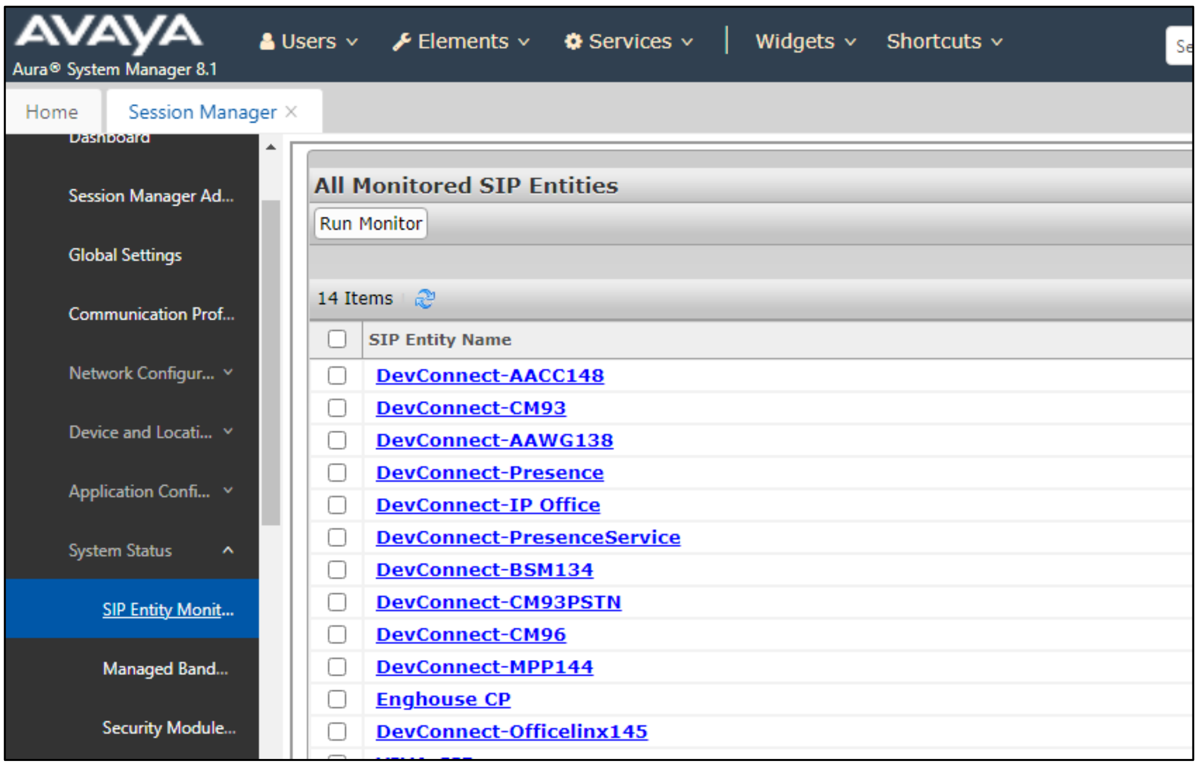


8. Verification Steps

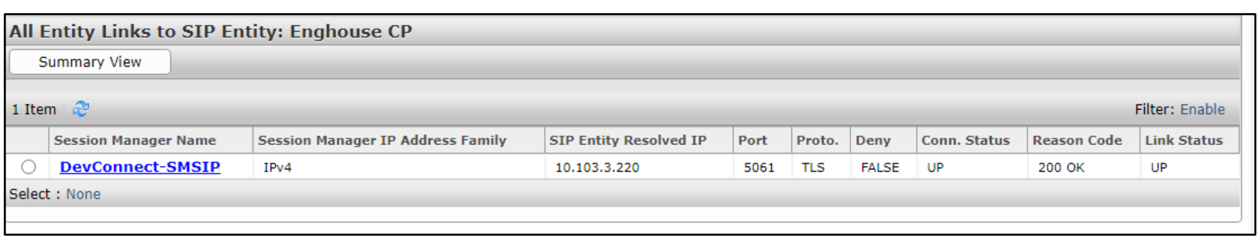
To verify a successful configuration of Enghouse Interactive Communications Portal and Session Manager/Communication Manager, a call is placed from a PSTN telephone to the Communications Portal with the caller getting answered successfully hearing clear and audible speech.

8.1. Verify Avaya Aura® Session Manager

To verify SIP connectivity to Communications Portal, on System Manager, navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring**. Under the **All Monitored SIP Entities**, select the **Enghouse CP** Entity.



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



8.2. Verify SRTP is being used via Avaya Aura® Communication Manager

When calls are active on CP, verify that SRTP is being used by checking the status of the SIP trunk between Communication Manager and Session Manager that's being used for the call. Use the status trunk command and navigate to Page 3 to verify that SRTP is being used for the call as shown below. Use the **status trunk 1** command to determine which trunk member is active, example trunk member **11** is active as shown below

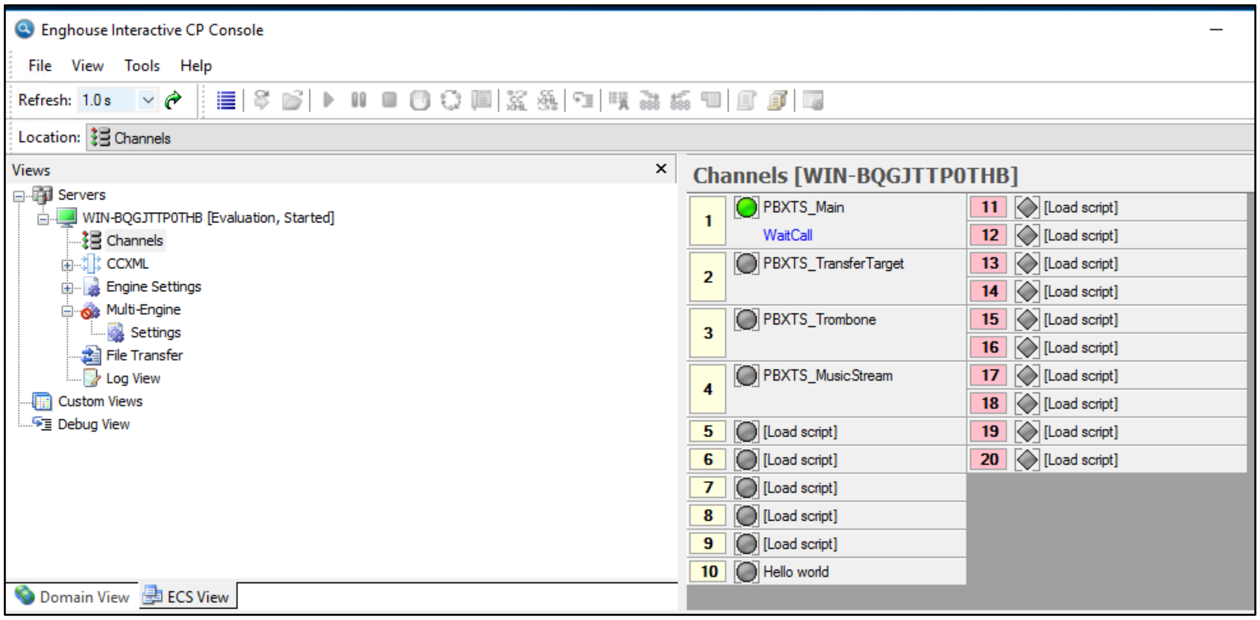
```
status trunk 1                                                        Page   1

                             TRUNK GROUP STATUS

Member    Port    Service State      Mtce Connected Ports
                                     Busy

0001/0001 T000001 in-service/idle    no
0001/0002 T000002 in-service/idle    no
0001/0003 T000003 in-service/idle    no
0001/0004 T000004 in-service/idle    no
0001/0005 T000005 in-service/idle    no
0001/0006 T000006 in-service/idle    no
0001/0007 T000007 in-service/idle    no
0001/0008 T000008 in-service/idle    no
0001/0009 T000009 in-service/idle    no
0001/0010 T000010 in-service/idle    no
0001/0011 T000011 in-service/active  no   T000249
0001/0012 T000012 in-service/idle    no
0001/0013 T000013 in-service/idle    no
0001/0014 T000014 in-service/idle    no
```

Use **command status trunk 01/011**, **on page 3** to check RTP status of the call, example for media shuffling enable (Direct IP-IP Audio Connections set to y) as shown below

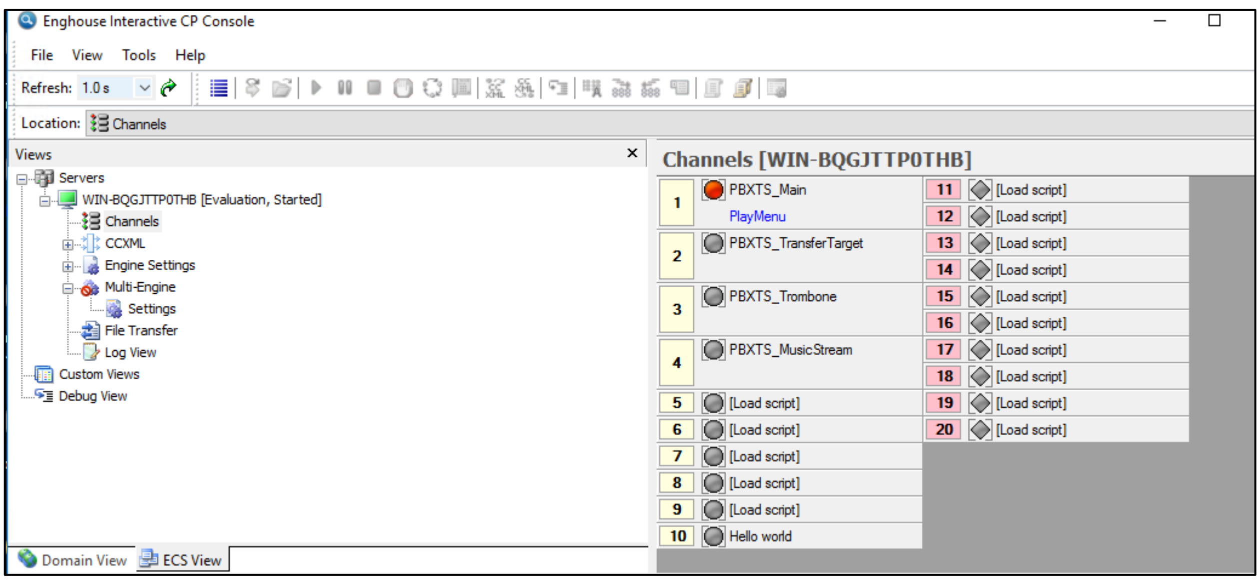
```
status trunk 01/11                                             Page   3 of   3
                      SRC PORT TO DEST PORT TALKPATH
src port: T000011
T000011:TX:10.103.3.220:29016/g711u/20ms/1-srtp-aescm128-hmac80
T000249:RX:10.30.5.99:37392/g711u/20ms/1-srtp-aescm128-hmac80








dst port: T000249
```

8.3. Verify Enghouse Interactive Communications Portal

On **CP Console 10.4**, monitor the Channel 1 below show as green that mean CP is available to receive the call.



Place a call from the PSTN to CP ensure the call can be answered by CP. Monitor the Channel 1 below should also show as Red.



9. Conclusion

These Application Notes describe the configuration steps required for Enghouse Interactive Communications Portal 10.4 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.3 and Avaya Aura® Communication Manager 8.1.3.3. 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 12, July 2021
2. *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 10, Sept 2021
3. *Administering Avaya Aura® System Manager*, Release 8.1.x, Issue 17, Nov 2021
4.

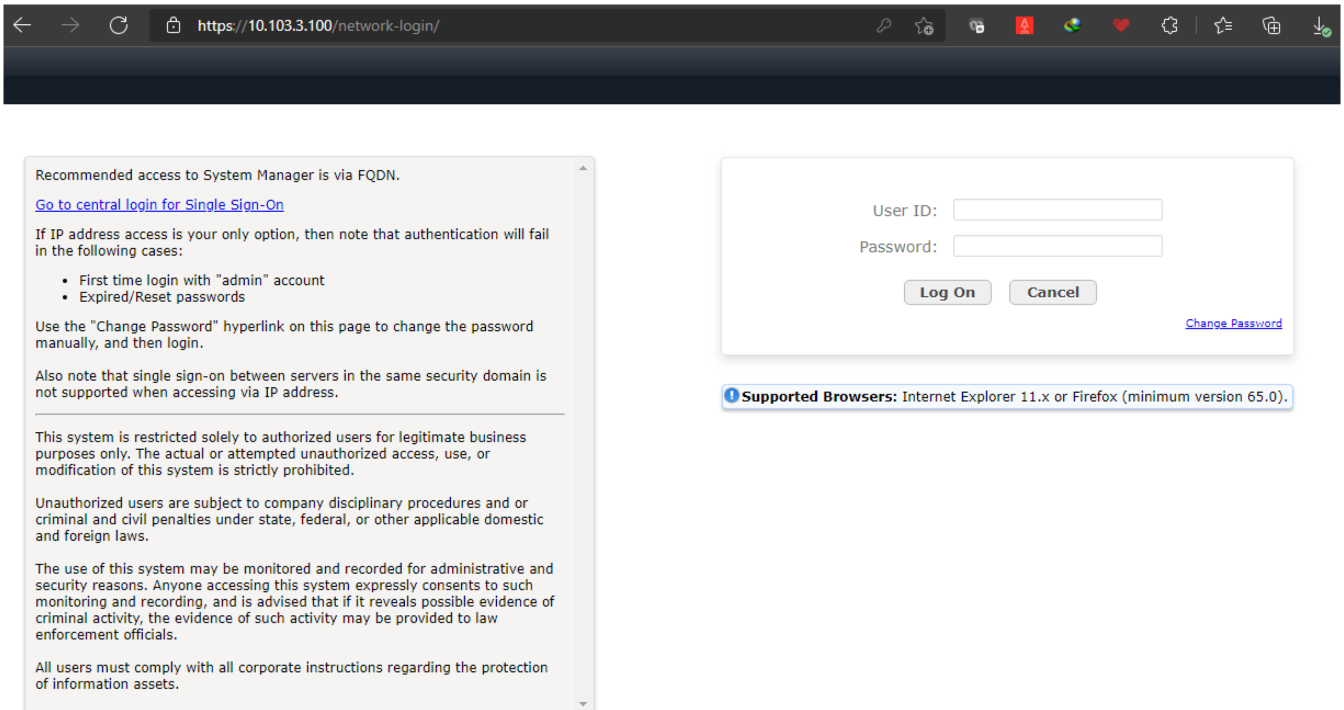
Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.

Appendix

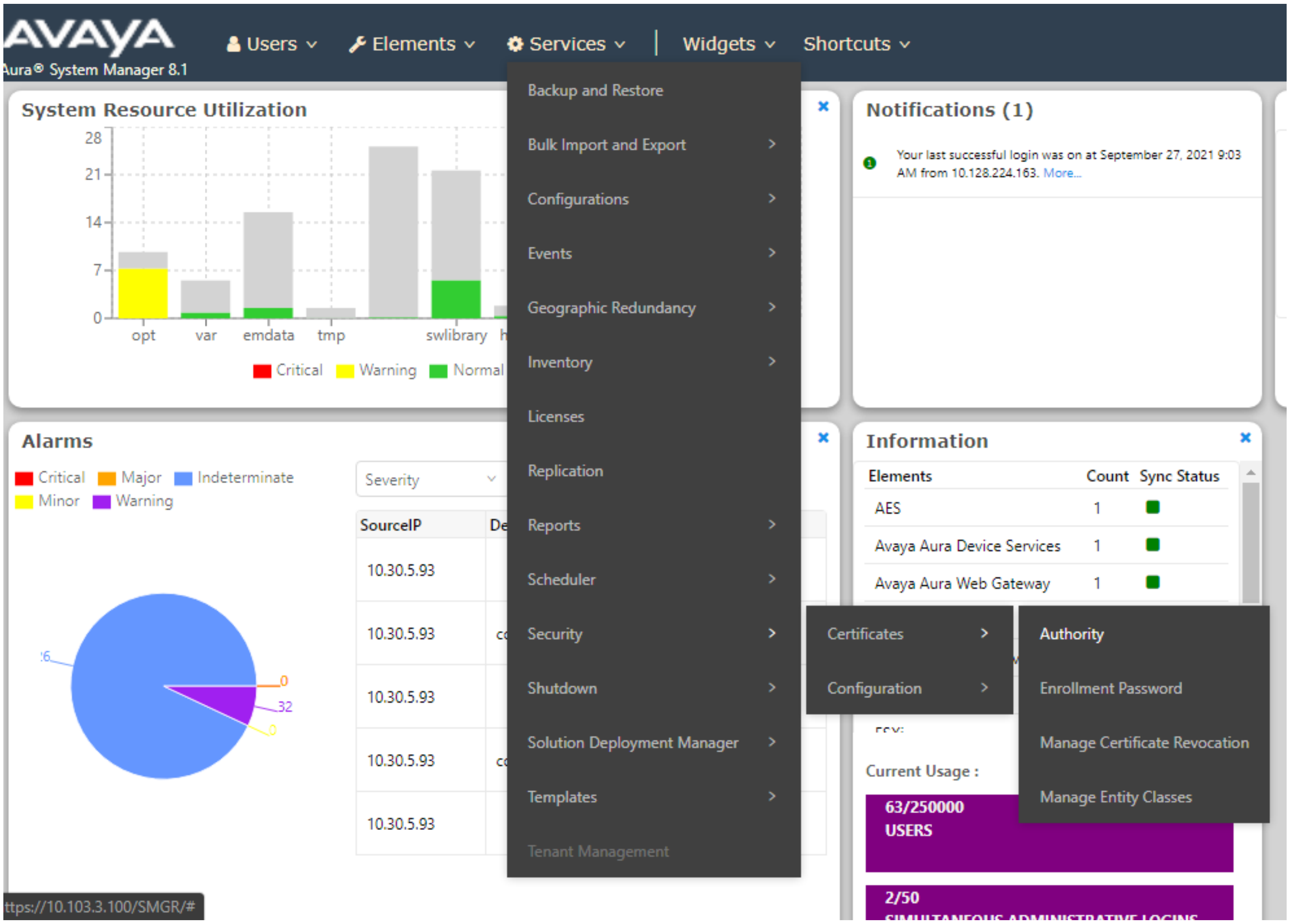
The following section shows the creation of the Enghouse Communications Portal End Entity on the SMGR CA in order to sign the CSR generated by Communications Portal.

Add End Entity

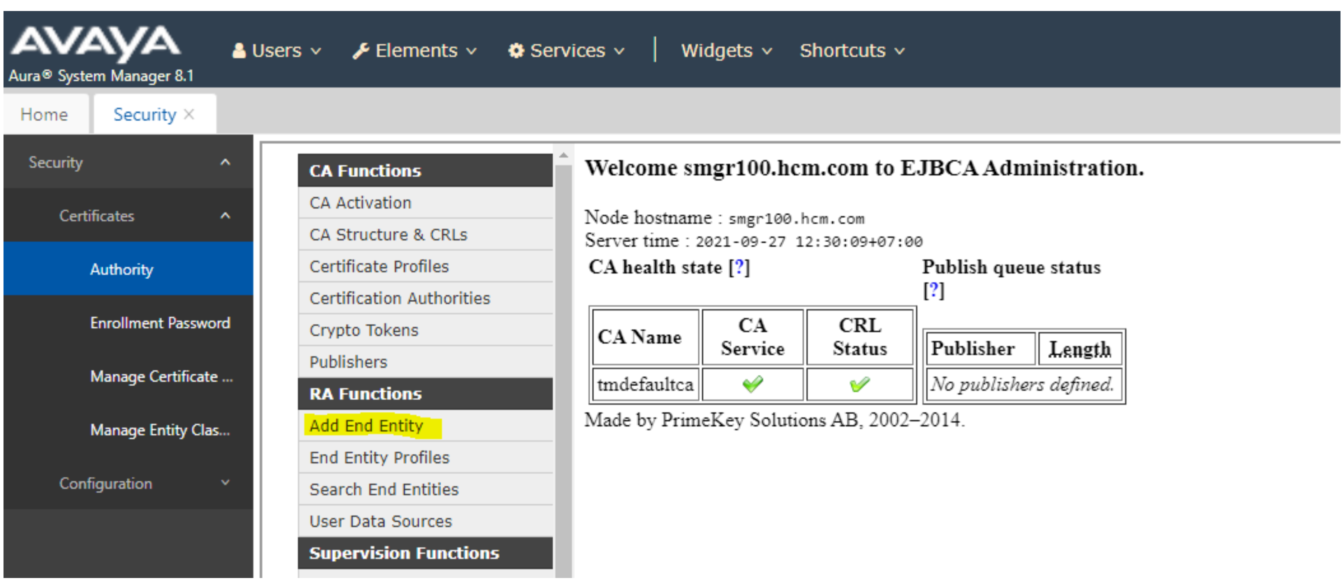
The 3rd party endpoint (Communications Portal) is added to the CA as an end entity. Log in to the Certificate Authority, in this case a System Manager.



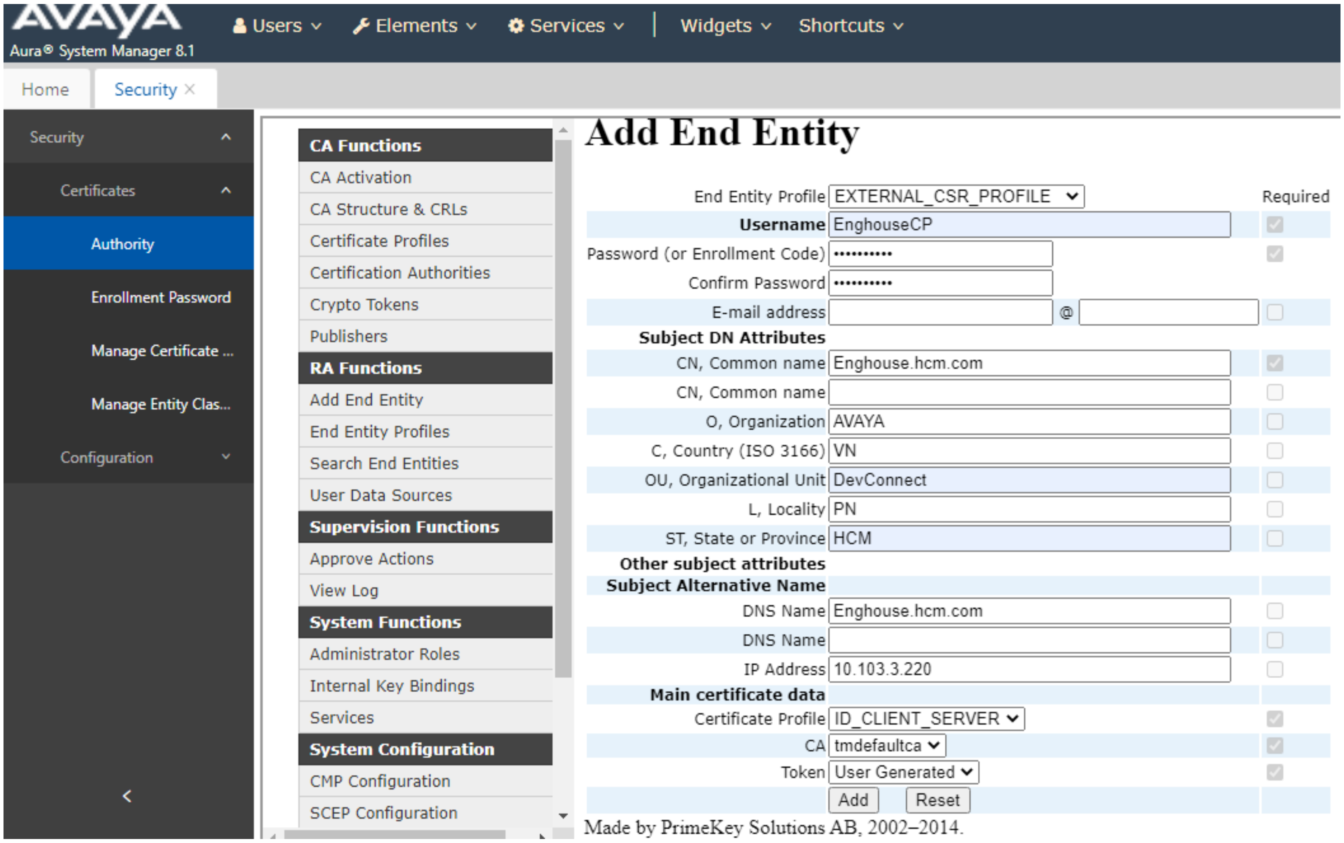
Click on **Services → Security → Certificates → Authority** from the main menu.



Click on **Add End Entity**

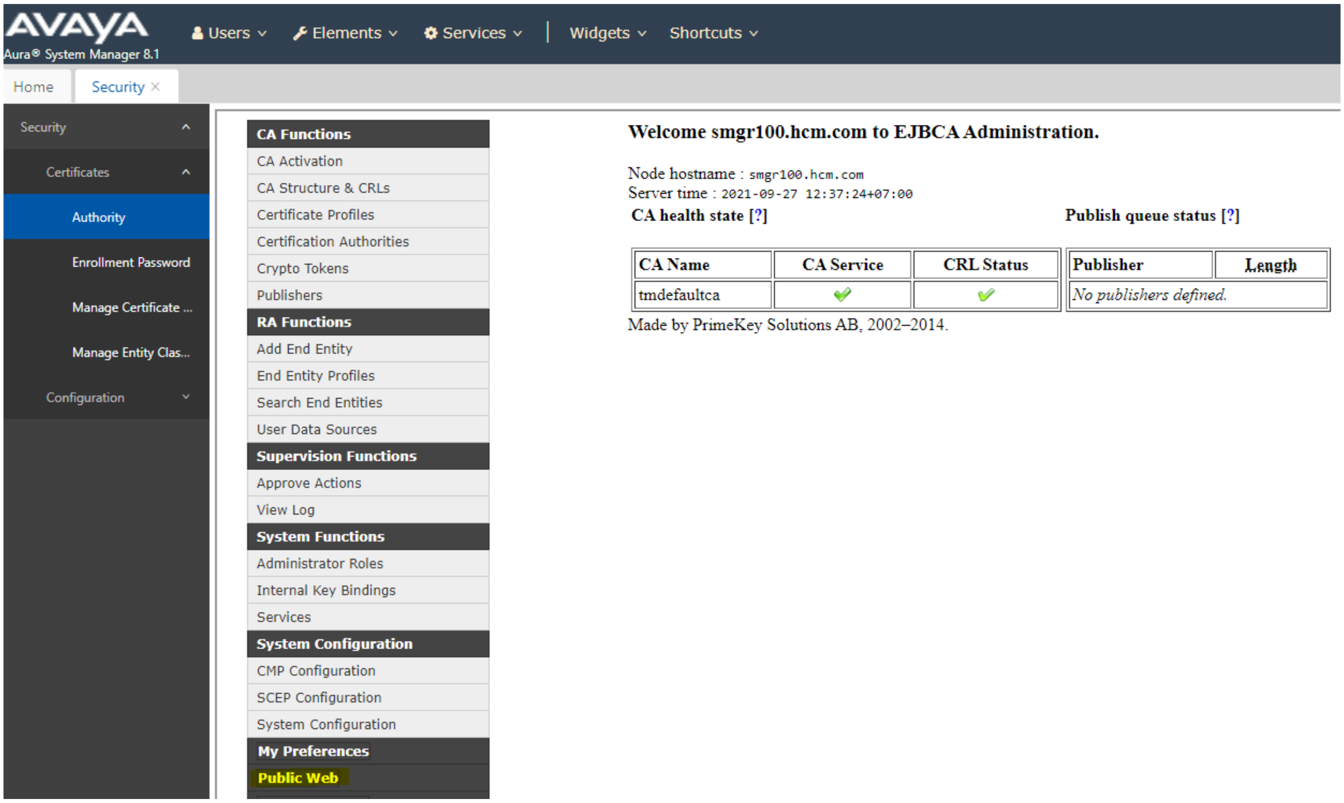


The following is an example of the **End Entity** that was added for compliance testing. Take note of the **Password (or Enrollment Code)**, this will be required later, the **IP address** will be that of the Communications Portal and the **Common name** and **Username** should be hostname associated with the Communications Portal. Click on **Save** once the information has been filled in correctly.

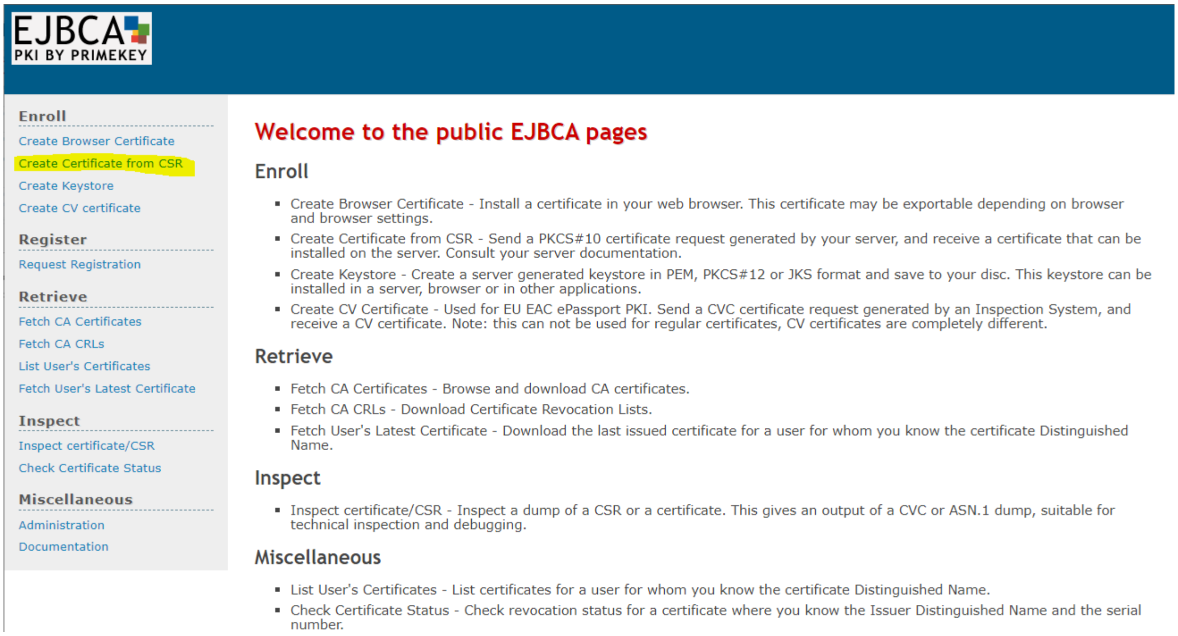


Generate the Identity Certificate

From the CA, click on the **Public Web** down the left side of the page.



The following web page is opened, click on **Create Certificate from CSR**



Choose CSR file **EnghouseCP.csr**, this is taken from the CSR generated by Enghouse as shown on the previous page. Select Result type with **PEM – full certificate chain** and click **OK** to download **Identity Certificate.**