Avaya Solution & Interoperability Test Lab

Application Notes for configuring Enghouse Interactive Communications Portal 10.1 using CTIC Media Gateway for SIP 8.2 with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 using TLS and SRTP - Issue 1.0

Abstract

These Application Notes describe the configuration steps for Enghouse Interactive Communications Portal 10.1 using CTIC Media Gateway for SIP 8.2 to successfully interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 using TLS/SRTP. Communications Portal 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 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 for Enghouse Interactive Communications Portal 10.1 to successfully interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 using Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP). Enghouse Interactive Communications Portal (formerly Syntellect Communications Portal) is an open, standards-based platform with integrated application development and management components.

- Voice self-service solutions, such as interactive voice response (IVR), interactive voice and video response (IVVR), outbound dialing, and speech-enabled self-service systems.
- SMS, email, standards-based voice mail.
- Contact center solutions, including outbound dialing, intelligent routing applications and screen pop applications.
- Unified communications solutions, including standards-based voice-mail systems and applications that combine traditional voice, IP telephony, video messaging, SMS, email, and fax communication.

The primary goal of the Transport Layer Security (TLS) protocol is to provide privacy and data integrity between two communicating computer applications. When secured by TLS, connections between a client (e.g., Enghouse Interactive Communications Portal 10.1) and a server (e.g., Avaya Aura® Session Manager) have one or more of the following properties:

- The connection is private because symmetric cryptography is used to encrypt the data transmitted. The keys for this symmetric encryption are generated uniquely for each connection and are based on a shared secret negotiated at the start of the session. The server and client negotiate the details of which encryption algorithm and cryptographic keys to use before the first byte of data is transmitted. The negotiation of a shared secret is both secure and reliable.
- The identity of the communicating parties can be authenticated using public-key cryptography. This authentication can be made optional, but is generally required for at least one of the parties (typically the server).
- The connection is reliable because each message transmitted includes a message integrity check using a message authentication code to prevent undetected loss or alteration of the data during transmission.

The Secure Real-time Transport Protocol (or SRTP) defines a profile of RTP (Real-time Transport Protocol), intended to provide encryption, message authentication and integrity, and replay protection to the RTP data in both unicast and multicast applications. Since RTP is closely related to RTCP (Real Time Control Protocol) which can be used to control the RTP session, SRTP also has a sister protocol, called Secure RTCP (or SRTCP); SRTCP provides the same security-related features to RTCP, as the ones provided by SRTP to RTP. Utilization of SRTP or SRTCP is optional to the utilization of RTP or RTCP; but even if SRTP/SRTCP are used, all provided features (such as encryption and authentication) are optional and can be separately enabled or disabled. The only exception is the message authentication feature which is indispensably required when using SRTCP.

2. General Test Approach and Test Results

The IVR application telephony functionality of Communications Portal 10.1 (CP) was the only module tested. This IVR application (CP script) connects to Session Manager as a SIP Trunk entity and can be integrated with Communication Manager by passing SIP calls to and from the PBX. Session Manager directs the call over SIP trunks to CP scripts which in turn handles the call depending on the digits dialled using SIP signaling. Communications Portal utilizes CTI Media Gateway driver to perform all telephony functions on the server. This CTI Media Gateway facilitates the Communications Portal connectivity to Session Manager.

Note: It is assumed that all TLS connections are already established between System Manager, Session Manager and Communication Manager along with the H323 and SIP deskphones that were used as part of the compliance testing. The description of any such setup is deemed as being outside the scope of these Application Notes.

Note: All test cases were performed with Direct IP –IP Audio Connection set to N. Note that any other setting does not work when using SRTP.

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.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on placing various calls to the Communications Portal IVR:

- **Basic Inbound/Outbound** – Tests inbound calls to Enghouse Interactive Communications Portal.
- **Call Hold** – Tests held calls to/from Enghouse Interactive Communications Portal.
- **Call Transfer** – Tests transferred calls to/from Enghouse Interactive Communications Portal.
- **IVR Functionality** – Tests of various IVR features like is ANI/DNIS detection, leaving voice message/voice mail (Recording), DTMF collection, Barge-in and Trombone Referral on the Enghouse Interactive Communications Portal.
- **Failover/Service** – Tests the behaviour of Enghouse Interactive Communications Portal when there are certain failed conditions and verifying the ability of Communications Portal to recover from disconnection and reconnection to the Avaya solution.

2.2. Test Results

All functionality and serviceability test cases were completed successfully, however the following issues were observed.

- All test cases were performed with Direct IP –IP Audio Connection set to N. Note that any other setting does not work when using SRTP. See **Section 5.6**.
- Setting Encrypted SRTCP to any other option but best-effort would not establish a new call. See **Section 5.5**.
- CLID on phone called by CP is not updated with the correct CLID after transfer is completed. To resolve this, the script was updated to send the FROM information. This is configurable on a per customer basis and is not hardcoded as part of the SIP firmware.
- Compliance testing was also carried out using TCP/RTP and all test cases passed with Direct IP –IP Audio Connection set to both Y and N.

2.3. Support

Technical support 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

The configuration in **Figure 1** was used to compliance test Enghouse Interactive Communications Portal 10.1 with Session Manager and Communication Manager using SIP signalling over SIP trunks to route calls from Communication Manager to Communications Portal 10.1. The TLS protocol is to provide privacy and data integrity between two communicating computer applications and SRTP used to provide encryption, message authentication and integrity, and replay protection to the RTP data.

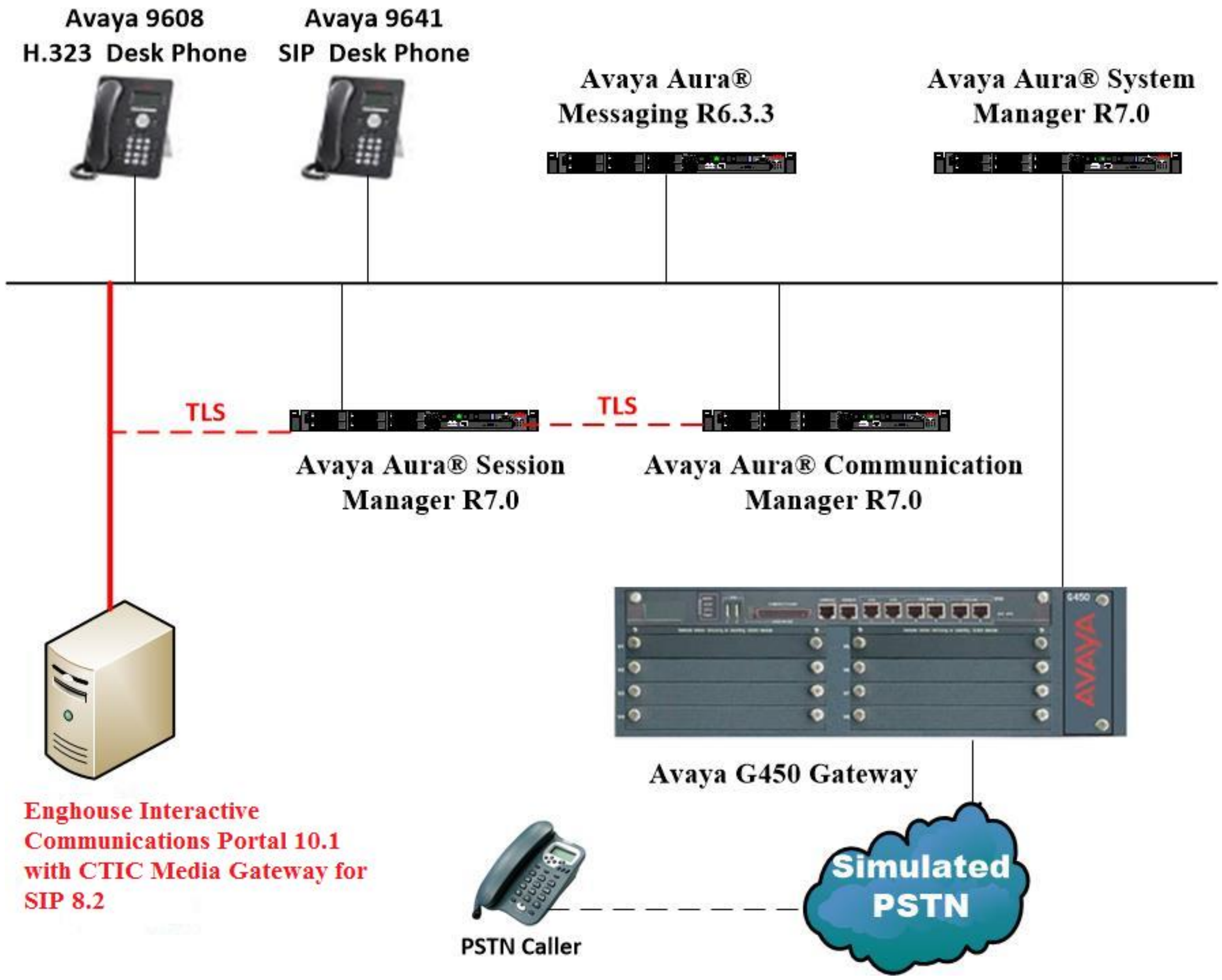


Figure 1: Connection of Enghouse Interactive Communications Portal 10.1 with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0

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 running on a virtual server	System Manager 7.0.1.0 Build No. - 7.0.0.0.16266 Software Update Revision No: 7.0.1.0.064859 Feature Pack 1
Avaya Aura® Session Manager running on a virtual server	Session Manager R7.0 SP1 Build No. – 7.0.1.0.701007
Avaya Aura® Communication Manager running on a virtual server	R7.0 R017x.00.0.441.0 00.0.441.0-23012
Avaya Aura® Messaging running on a virtual server	R6.3.3
Avaya G450 Gateway	37.19.0 /1
Avaya 9608 H323 Deskphone	96x1 H323 Release 6.6.028
Avaya 9608 SIP Deskphone	96x1 SIP Release 7.0.0.39
Enghouse Interactive Communications Portal running on Windows 2012 R2	Communications Portal 10.1 with CTIC Media Gateway for SIP 8.2 SP1F

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 11**.

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

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

Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

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 Communications Portal 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 2 SIP trunks.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 12000 250
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 319
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

On **Page 3**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              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
```

On **Page 5**, ensure that **Uniform Dialing Plan** is set to **y**.

```
display system-parameters customer-options                      Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? y
                     Multiple Locations? n
                                             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
```

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 11** 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? no
                      DID/Tie/ISDN/SIP Intercept Treatment: attd
    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 **8** is used for AAR and **9** for ARS routing.

```
display feature-access-codes                                        Page   1 of  10
                               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: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *25    Deactivation: #25
```

5.3. Administer Dial Plan

It was decided for compliance testing that all calls beginning with 62 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager and therefore to Communications Portal. 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 **6** is added with a **Total Length** of **4** and a **Call Type** of **udp**.

```
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
   2           4   ext
   3           4   ext
   4           4   udp
   5           4   ext
   6           4   udp
   7           3   dac
   8           1   fac
   9           1   fac
   *           3   fac
   #           3   fac
```

5.4. Administer Route Selection for Communications Portal Calls

As digits **6**xxx were defined in the dial plan as udp (**Section 5.3**) use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to numbers beginning with **62** that are **4** digits in length will be matched. No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

```
change uniform-dialplan 6                                              Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                     Percent Full: 0

  Matching                   Insert              Node
  Pattern       Len Del      Digits      Net Conv Num
  62             4   0                   aar  n
                                              n
```

Use the **change aar analysis** x command to further configure the routing of the dialed digits. Calls to Communications Portal begin with **62** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 1**, which contains the outbound SIP Trunk Group.

```
change aar analysis 62                                                Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed                 Total     Route    Call   Node  ANI
    String                Min  Max  Pattern   Type   Num   Reqd
    62                    4    4     1        unku         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 configured in **Section 5.6**.

```
change route-pattern 1                                                 Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIPTRK
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                  DCS/ IXC
    No          Mrk Lmt List Del  Digits                                    QSIG
                             Dgts                                           Intw
 1: 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  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                         Subaddress
 1: y y y y y n  n            unre                                             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: y y y y y n  n            rest                                             none
```

5.5. Configure Network Region and IP Codec

In the Node Names IP form, note the IP Address of the **procr** and the Session Manager (**sm70vmpg**). 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.

```
display node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AMS77vmpg           10.10.40.17
CMS18vmpg           10.10.40.36
IPO500V2            10.10.40.20
IPOSE               10.10.40.25
PGDECT              10.10.40.50
aes70vmpg           10.10.40.26
default             0.0.0.0
procr               10.10.40.13
procr6              ::
sm70vmpg            10.10.40.12
```

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.local**. 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.

```
display ip-network-region 1                                       Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devconnect.local
    Name: Default region
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
```

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk to Communications Portal. 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 example below includes **G.711A** (a-law), which is supported by Communications Portal. Note the **Media Encryption** has been set to **1-srtp-aescm128-hmac80**, this is the encryption that is support by Communications Portal and must be set correctly on each side to allow secure RTP (SRTP). In order for SRTP to work properly, **Encrypted SRTCP** needed to be set to **best-effort** as shown below.

Note: Setting **Encrypted SRTCP** to any other option would not establish a new call.

```
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.711A           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption                  Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2:
 3:
 4:
 5:
```

5.6. Configure SIP Trunk

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form shown below as follows:

- 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 **sm70vmpg**), 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 **n**.
- The default values for the other fields may be used.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? 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: sm70vmpg
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                        Far-end Network Region: 1

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

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to and from Communications Portal. 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  21
                                TRUNK GROUP

Group Number: 1                  Group Type: sip           CDR Reports: y
  Group Name: SIP TRK                  COR: 1        TN: 1        TAC: *11
   Direction: two-way        Outgoing Display? y
 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
```

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 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 **600** was used.

```
change trunk-group 1                                              Page   2 of  21
      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): 600

 Disconnect Supervision - In? y  Out? y


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

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  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



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

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n

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




 Show ANSWERED BY on Display? y
```

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

```
change trunk-group 1                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        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
```

6. Configure Avaya Aura® Session Manager

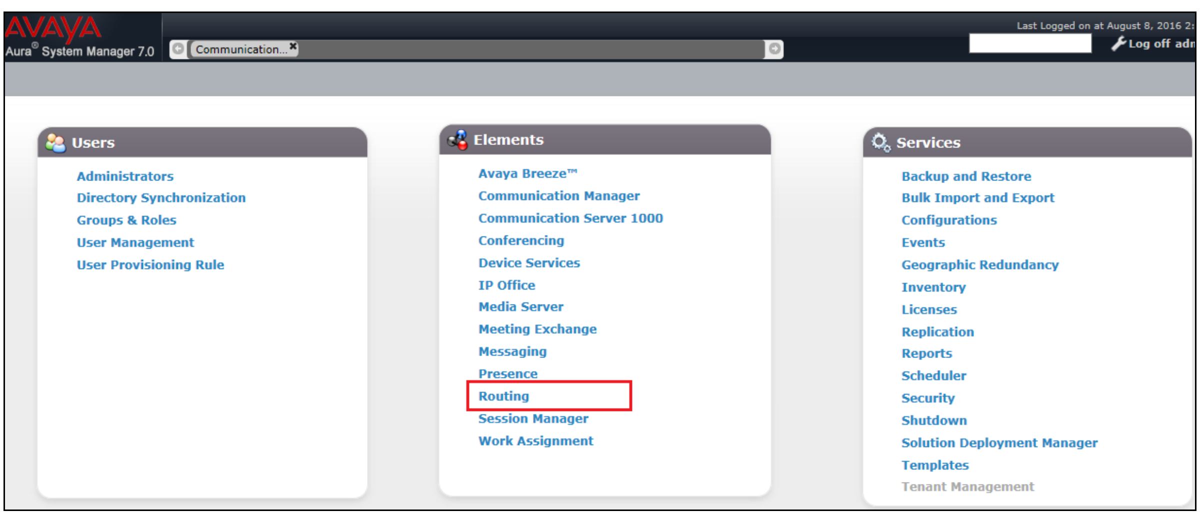
This section provides the procedures for configuring Session Manager. Session Manager is configured using a web browser connecting to System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager.
- Display configured SIP Domain.
- Configure SIP Entities.
- Configure Routing Policies.
- Configure Dial Patterns.

6.1. Log in to Avaya Aura® System Manager

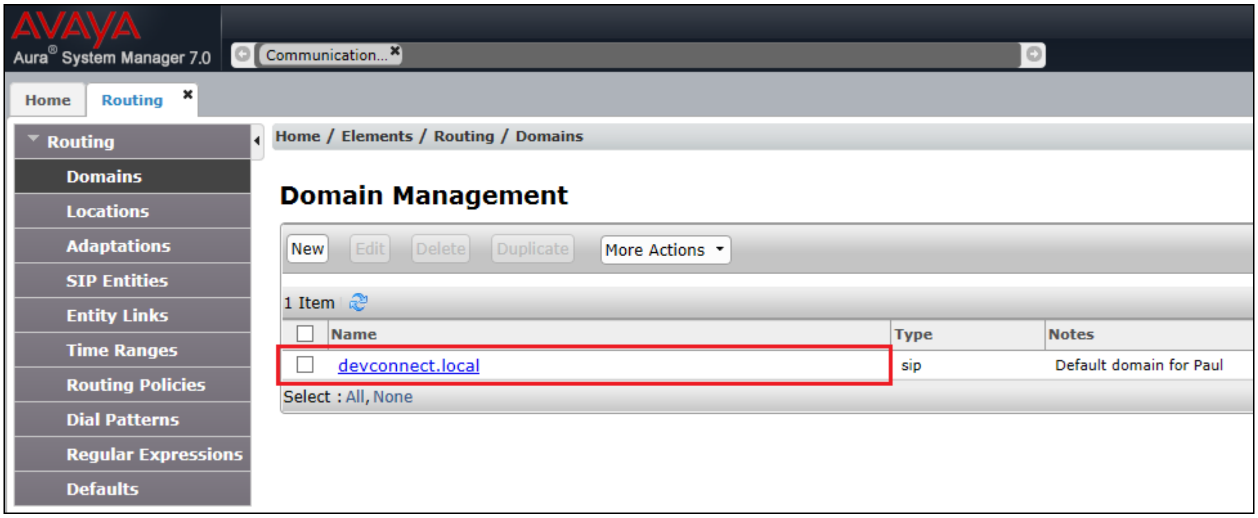
Access the System Manager using a web browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager or **http://<IP Adddress >/SMGR**. Log in using appropriate credentials.

Once logged ion click on **Routing** as highlighted.

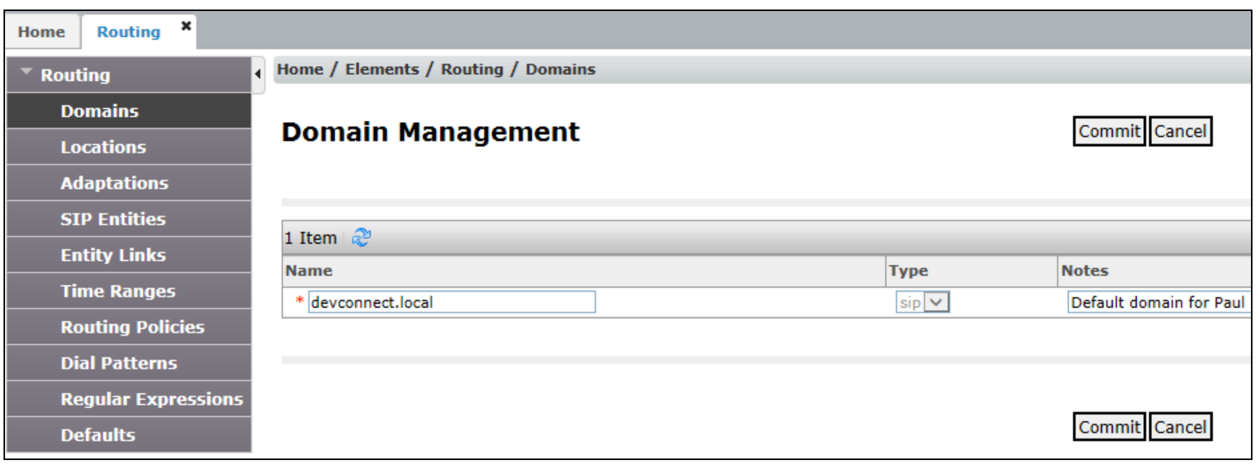


6.2. Display configured SIP Domain

Click on **Domains** in the left window. For compliance testing a domain had already been previously added called **devconnect.local**, this is displayed below and if there is not a domain already configured click on **New**.

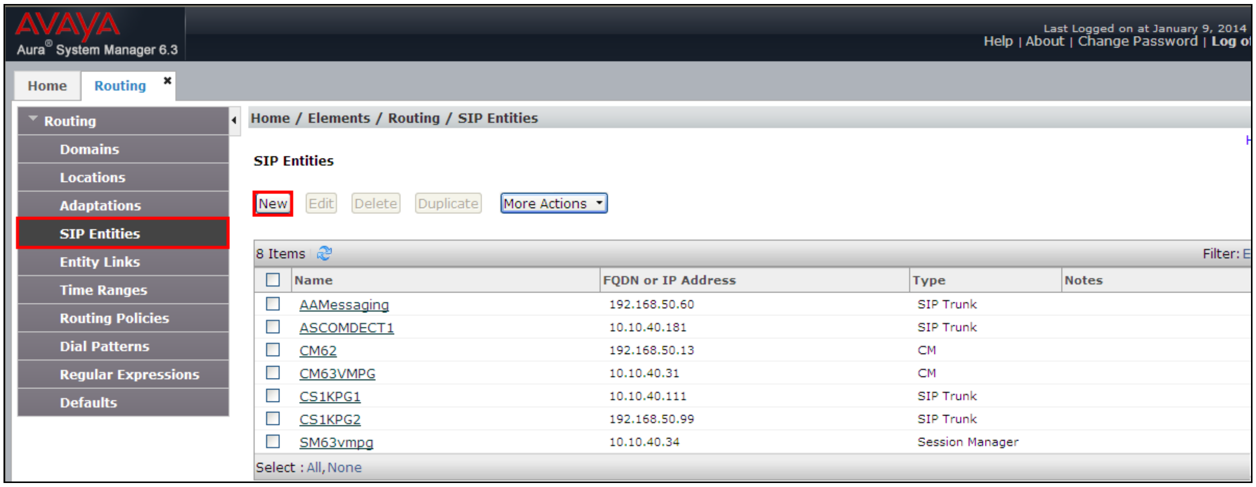


If a new domain is to be added this should be entered as shown below. Click on **Commit** once done.



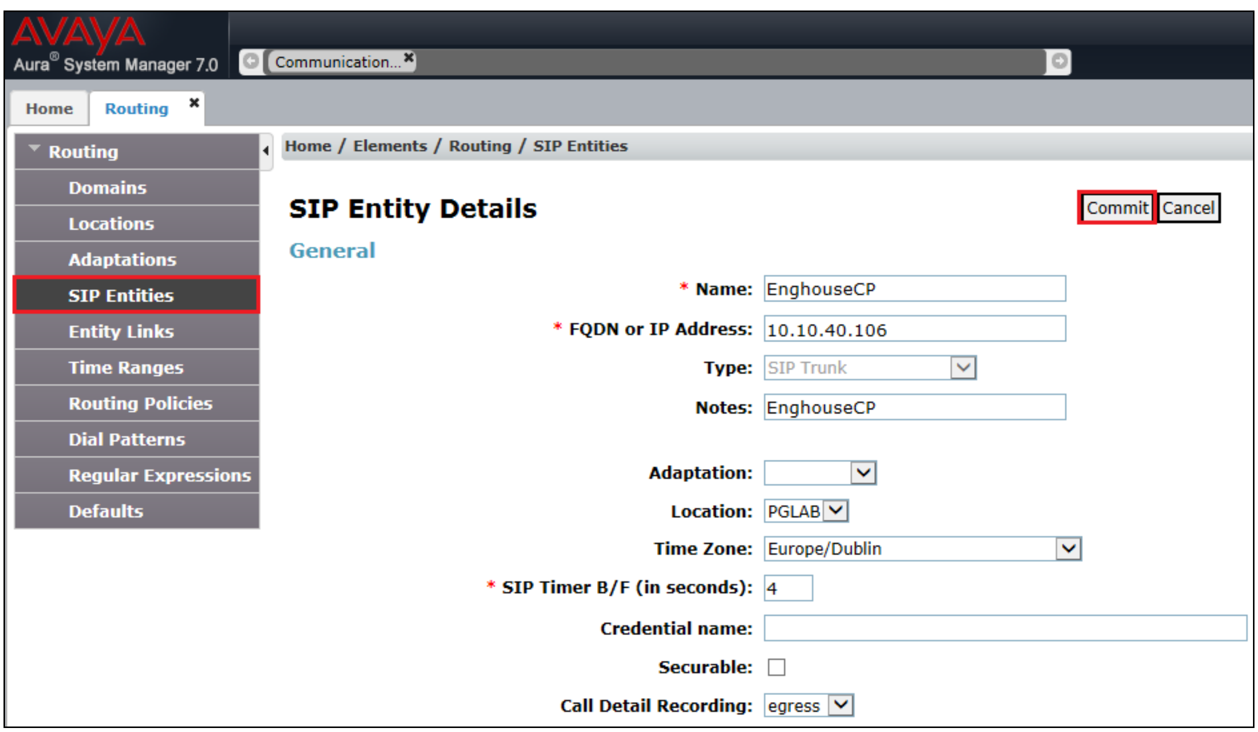
6.3. Configure SIP Entity for Enghouse Interactive Communications Portal

Select **SIP Entities** from the left window and click on **New** in the main window.



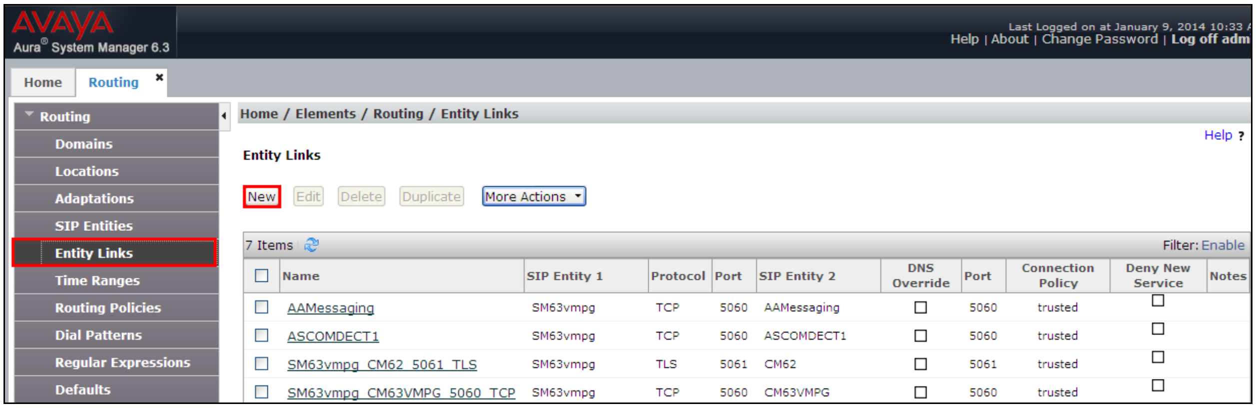
Enter a suitable **Name** and ensure that the correct **Location** and **Time Zone** are entered correctly, click on **Commit** to save the new entity.

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.

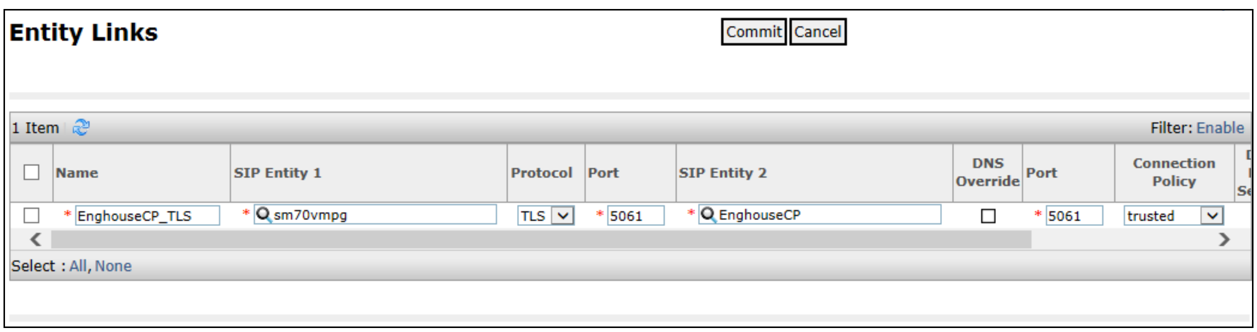


6.4. Configure Entity Link for Enghouse Interactive Communications Portal

Select **Entity Link** from the left window and click on **New** in the main window.

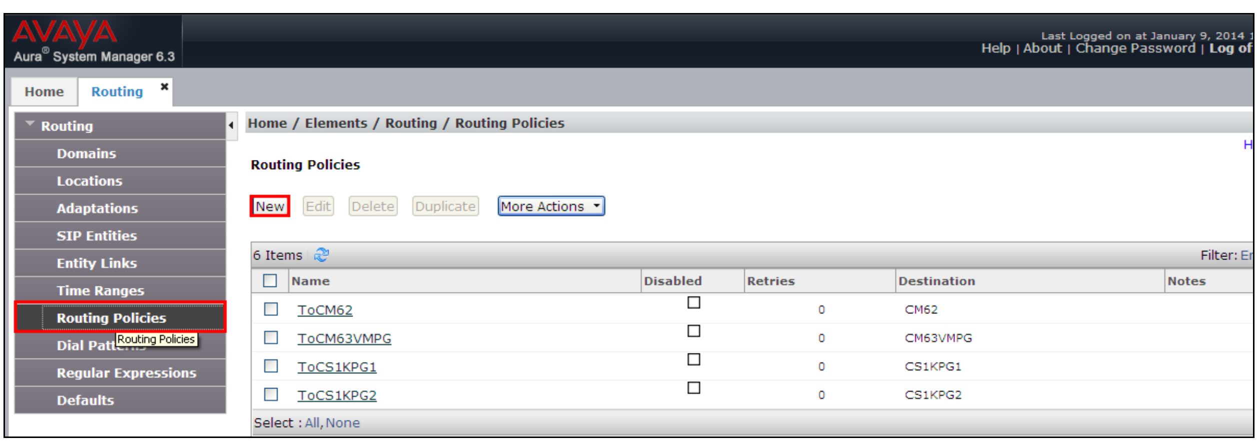


Select the correct **SIP Entity** that was created in **Section 6.3** and ensure that **TLS** is used as the **Protocol**. Note the **Port** is **5061**. Click on **Commit** once the information is entered correctly.

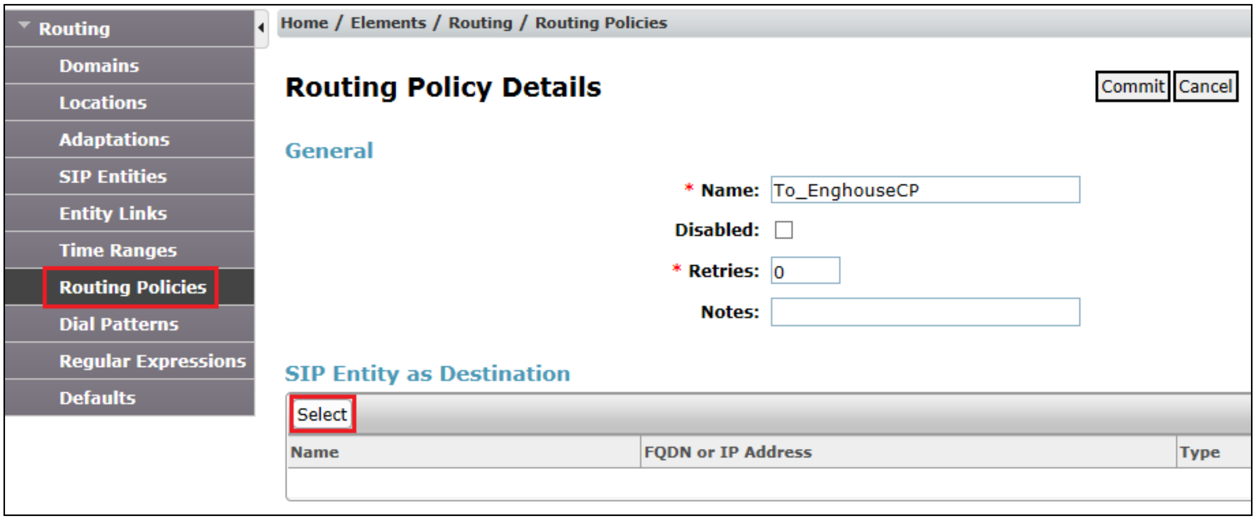


6.5. Configure Routing Policy for Enghouse Interactive Communications Portal

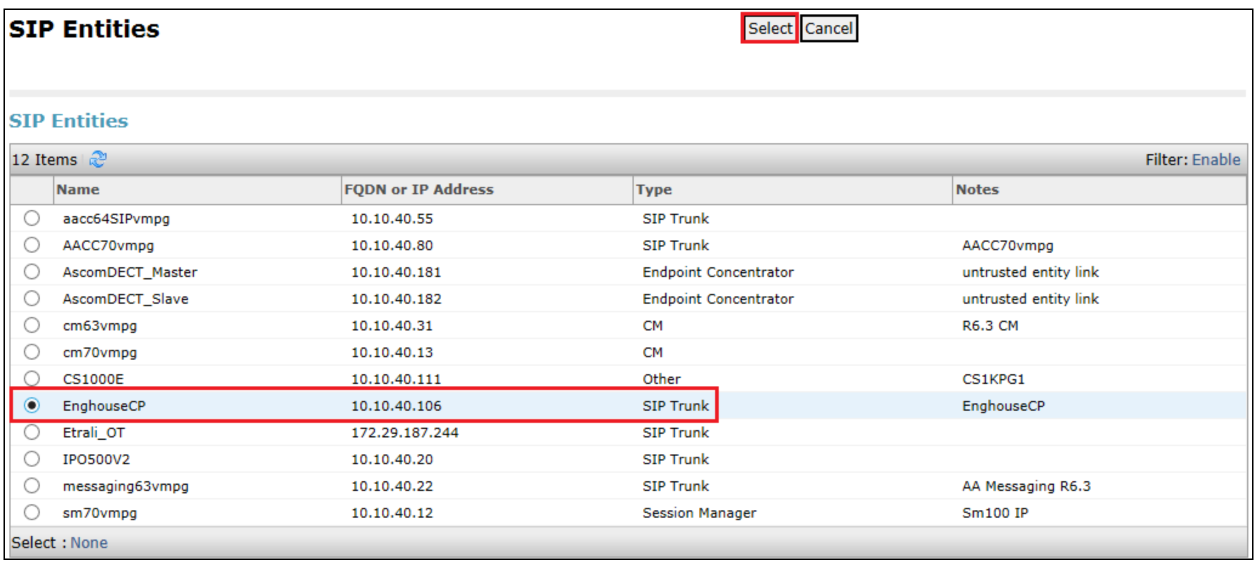
Select **Routing Policies** from the left window and click on **New** in the main window.



Enter a suitable **Name** and click on **Select** highlighted in order to associate this routing policy with a SIP Entity.

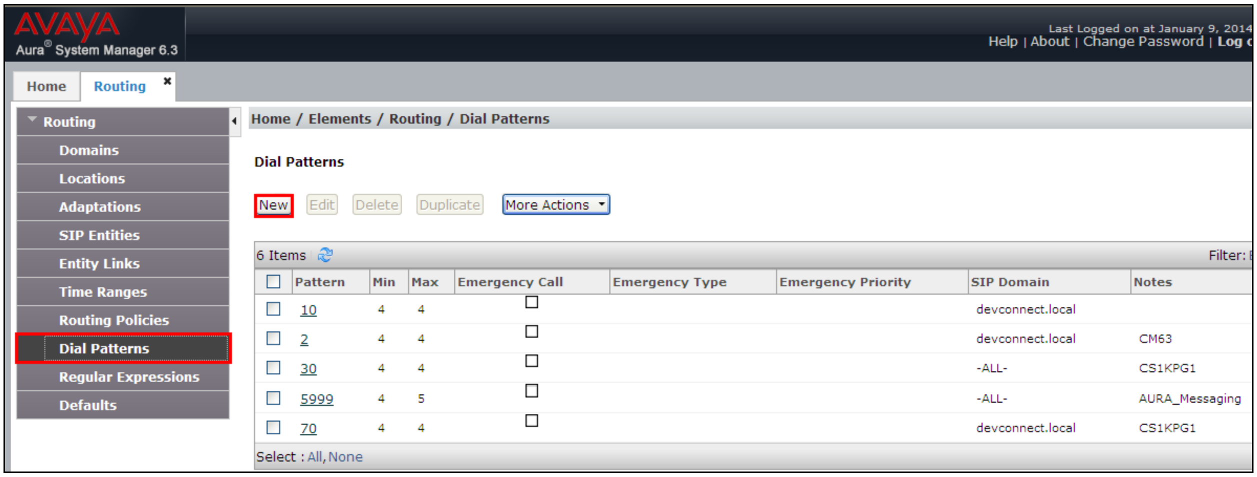


Select the **EnghouseCP** SIP Entity created in **Section 6.3** and click on **Commit** when done (not shown).

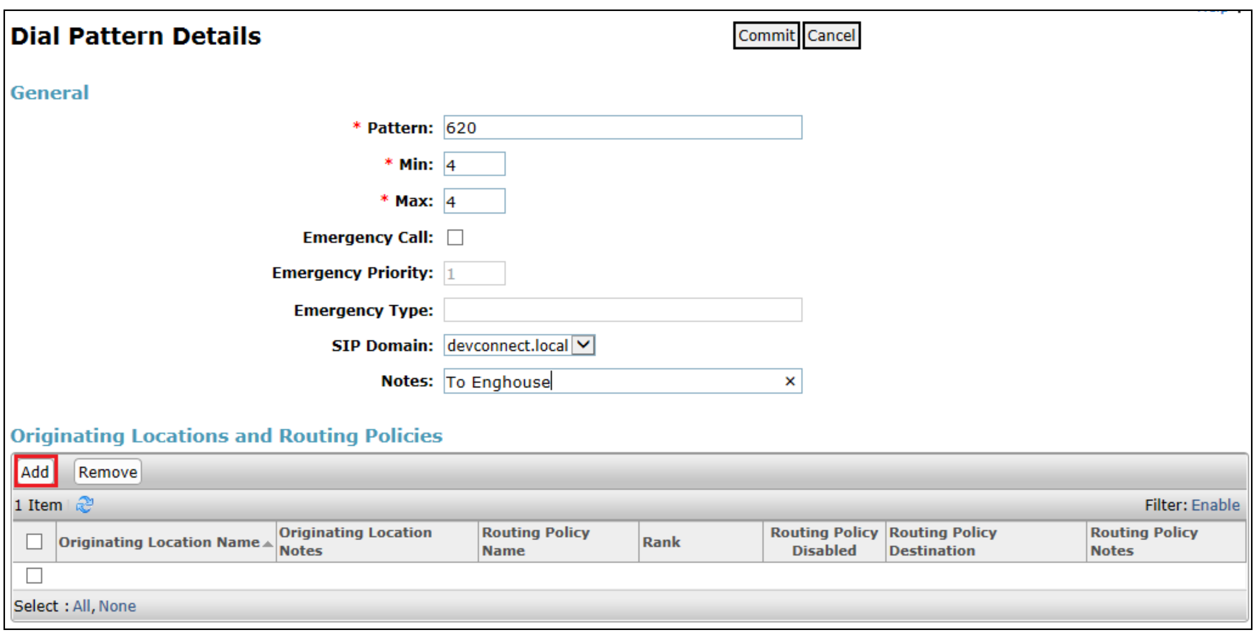


6.6. Configure Dial Pattern for Enghouse Interactive Communications Portal

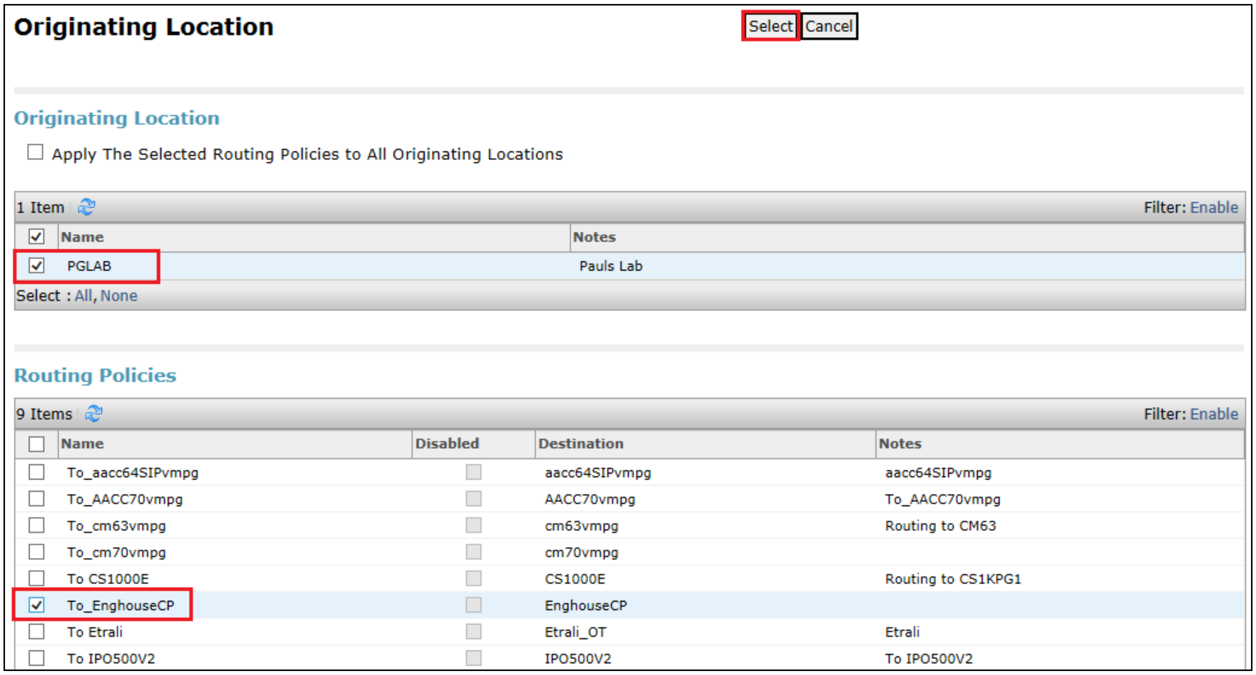
In order to route calls to the Communications Portal a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.



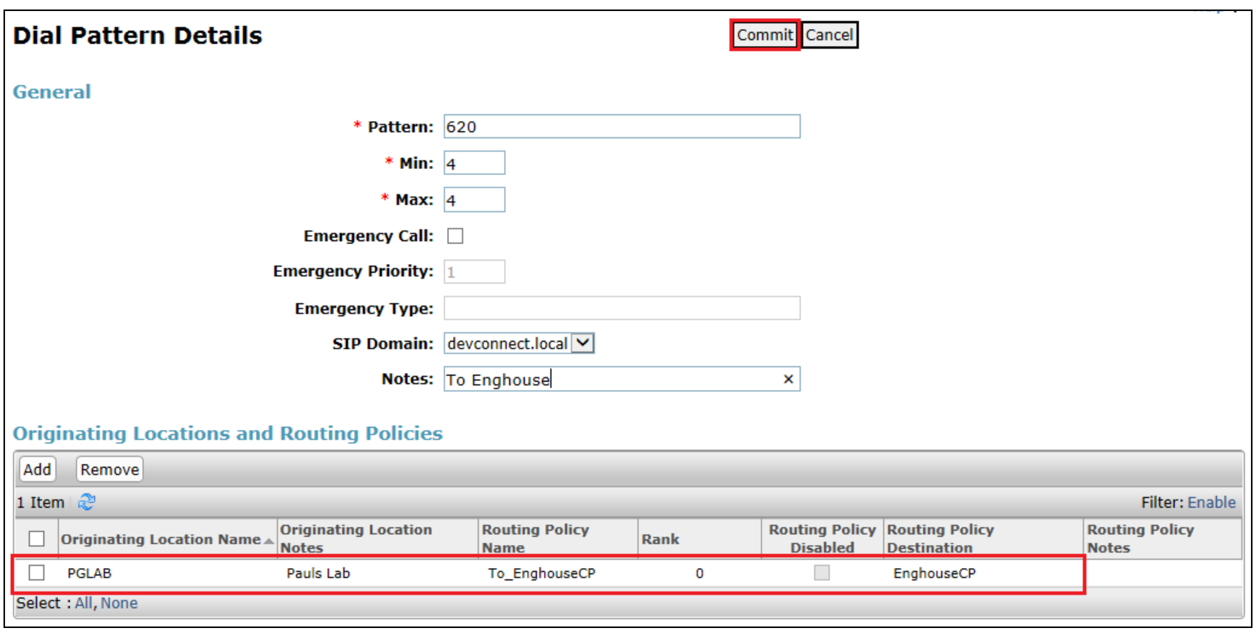
Enter the number to be routed noting this will be the same number outlined in **Section 5.4**. Note the **SIP Domain** is that configured in **Section 6.2**. Click on **Add** to select the SIP Entity.



Tick on the **Originating Location** as shown below and select the **Enghouse** Routing Policy. Click on **Select** once complete.

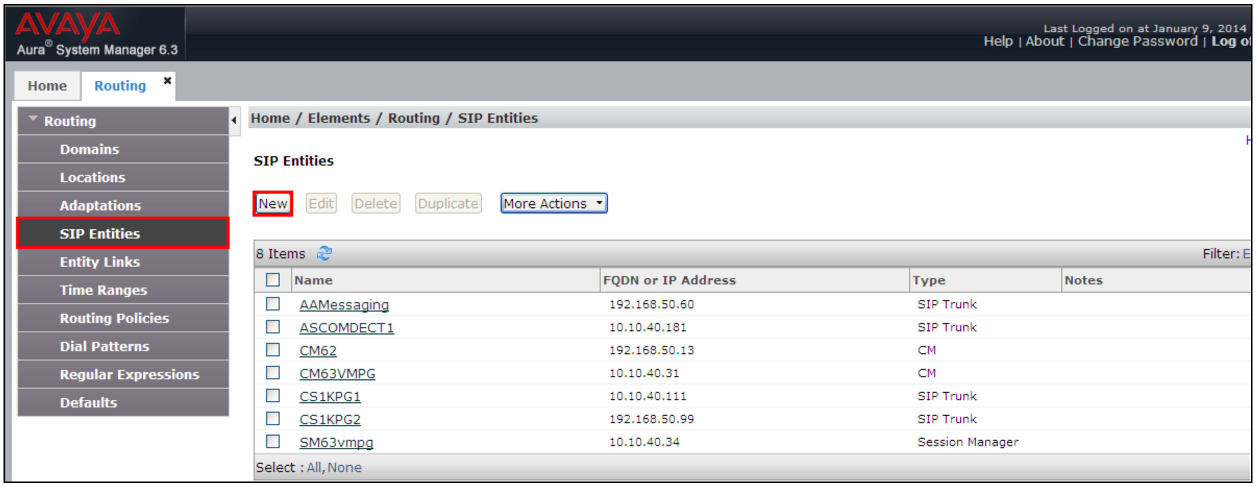


With the new Routing Policy in place, click on **Commit** as shown below.

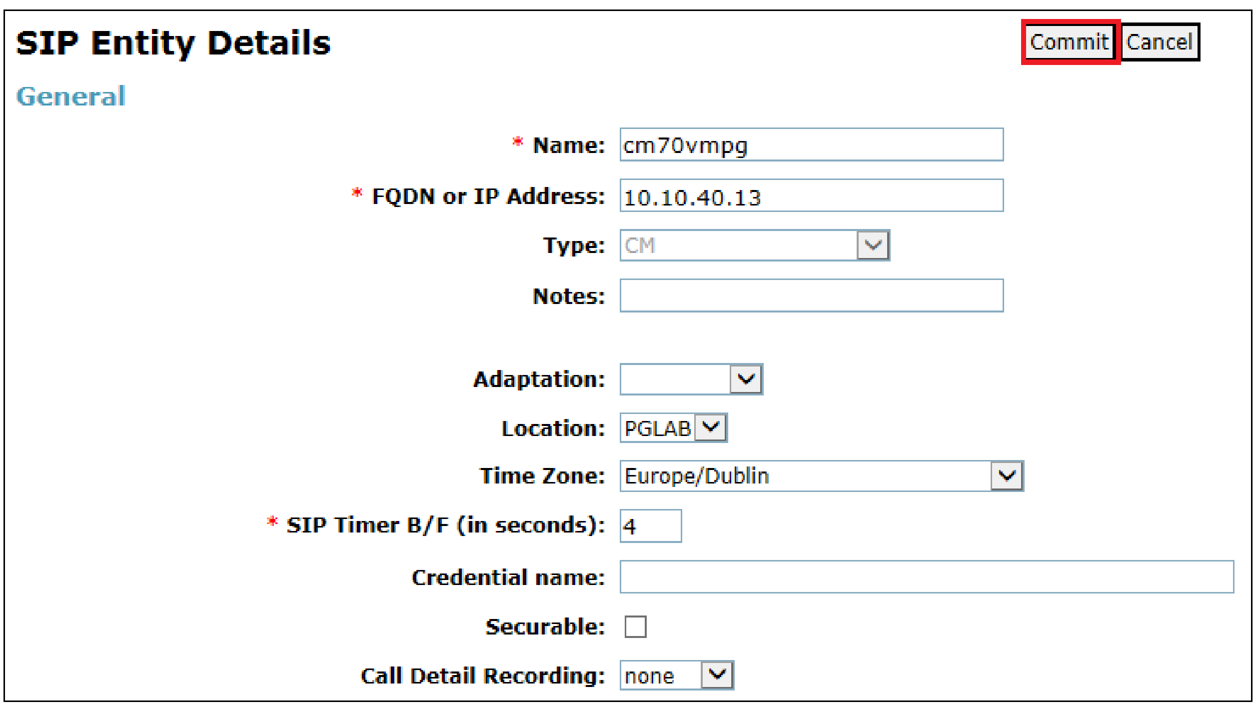


6.7. Configure Avaya Aura® Communication Manager SIP Entity

The following SIP Entity, SIP Entity Link, Routing Policy and Dial Pattern were already in place prior to compliance testing. The following sections are included to show an example of how to add these in the event they are not already present. Select **SIP Entities** from the left window and click on **New** in the main window.

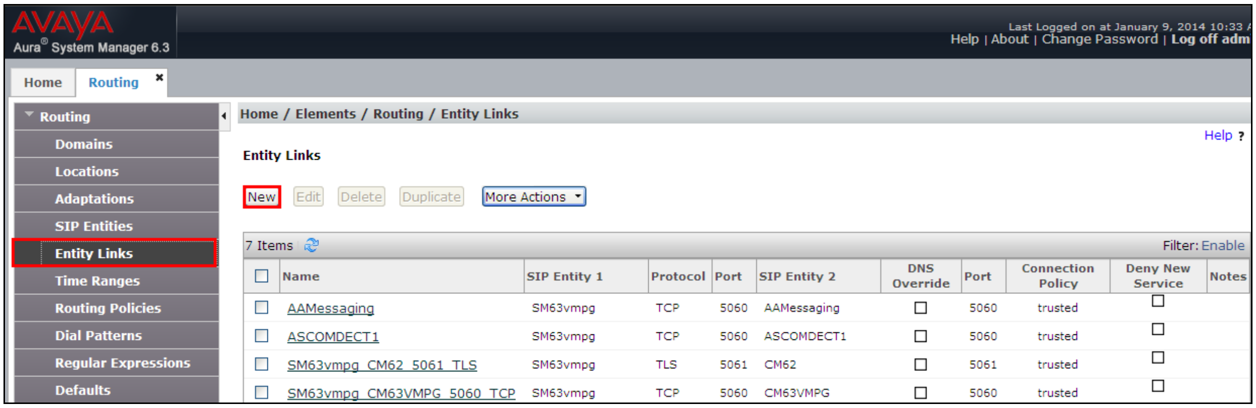


Enter a suitable **Name** and ensure the **Location** and the correct **Time Zone** is entered. Click on **Commit** once all is entered correctly.

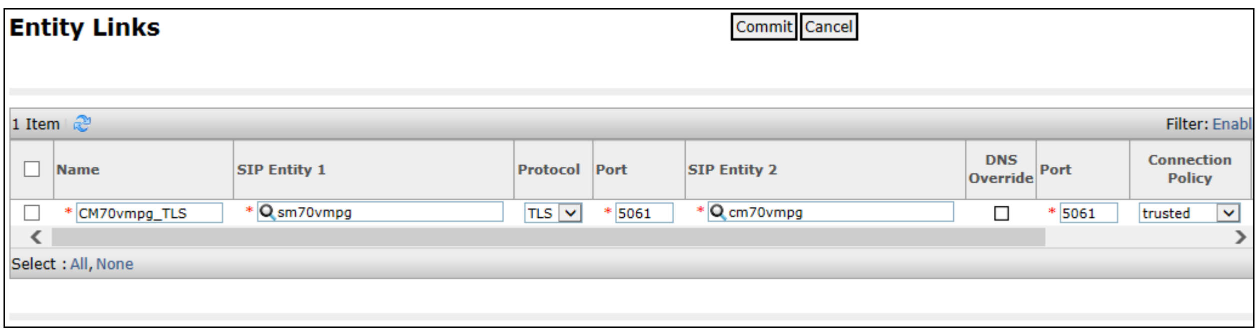


6.8. Configure Avaya Aura® Communication Manager Entity Link

Select **Entity Link** from the left window and click on **New** in the main window.

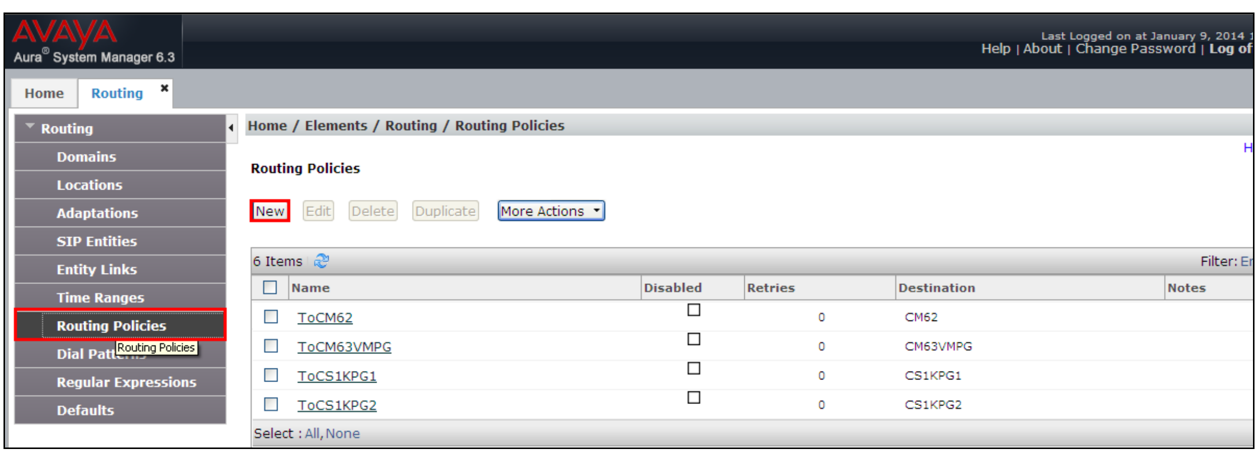


Select the correct **SIP Entity** that was created in **Section 6.7** and ensure that **TLS** is used as the **Protocol**. Note the **Port** is **5061**. Click on **Commit** once entered correctly.

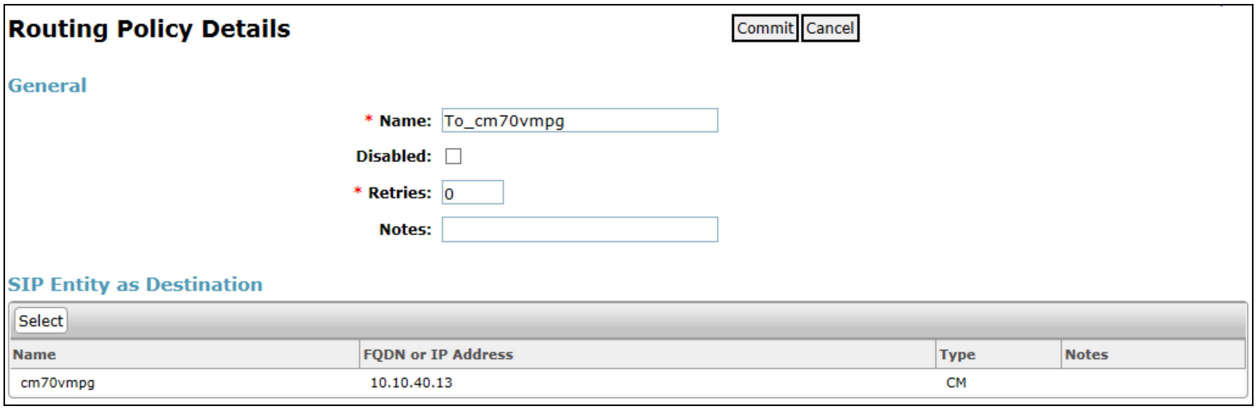


6.9. Configure Avaya Aura® Communication Manager Routing Policy

Select **Routing Policies** from the left window and click on **New** in the main window.



Enter a suitable **Name** and click on **Select** highlighted in order to associate this routing policy with a SIP Entity. Select the **Communication Manager** SIP Entity created in **Section 6.7** (not shown) and click on **Commit** when done.


Routing Policy Details

Commit Cancel

General

* Name: To_cm70vmpg

Disabled:

* Retries: 0

Notes:

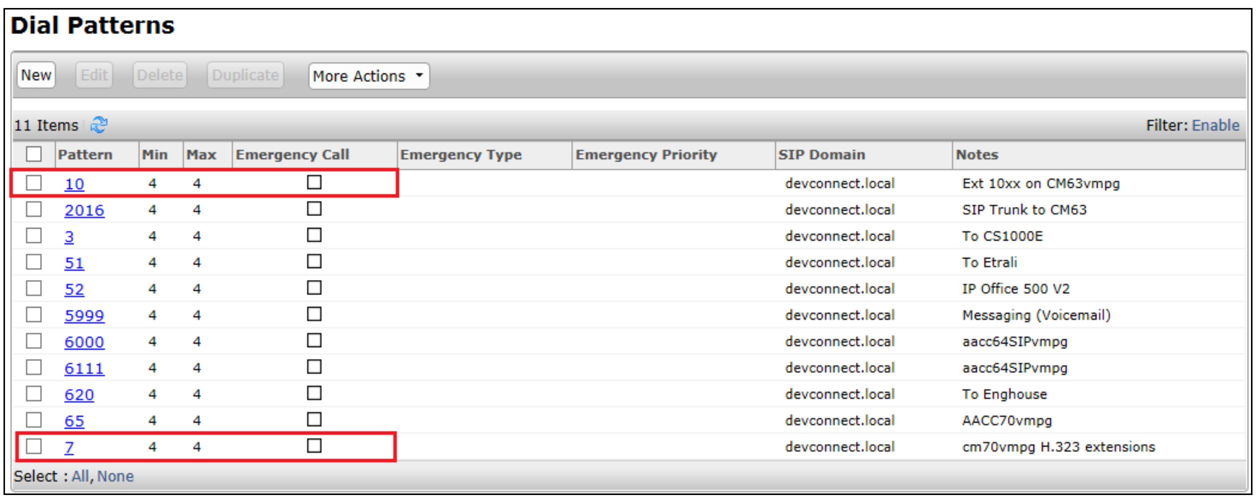
SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
cm70vmpg	10.10.40.13	CM	

6.10. Configure Avaya Aura® Communication Manager Dial Pattern

In order to route calls to Communication Manager 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 two dial patterns highlighted below were added in the same manner as outlined in **Section 6.6**.


Dial Patterns

New Edit Delete Duplicate More Actions

11 Items Filter: Enable

Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
10	4	4	☐			devconnect.local	Ext 10xx on CM63vmpg
2016	4	4	☐			devconnect.local	SIP Trunk to CM63
3	4	4	☐			devconnect.local	To CS1000E
51	4	4	☐			devconnect.local	To Etrali
52	4	4	☐			devconnect.local	IP Office 500 V2
5999	4	4	☐			devconnect.local	Messaging (Voicemail)
6000	4	4	☐			devconnect.local	aacc64SIPvmpg
6111	4	4	☐			devconnect.local	aacc64SIPvmpg
620	4	4	☐			devconnect.local	To Enghouse
65	4	4	☐			devconnect.local	AACC70vmpg
7	4	4	☐			devconnect.local	cm70vmpg H.323 extensions

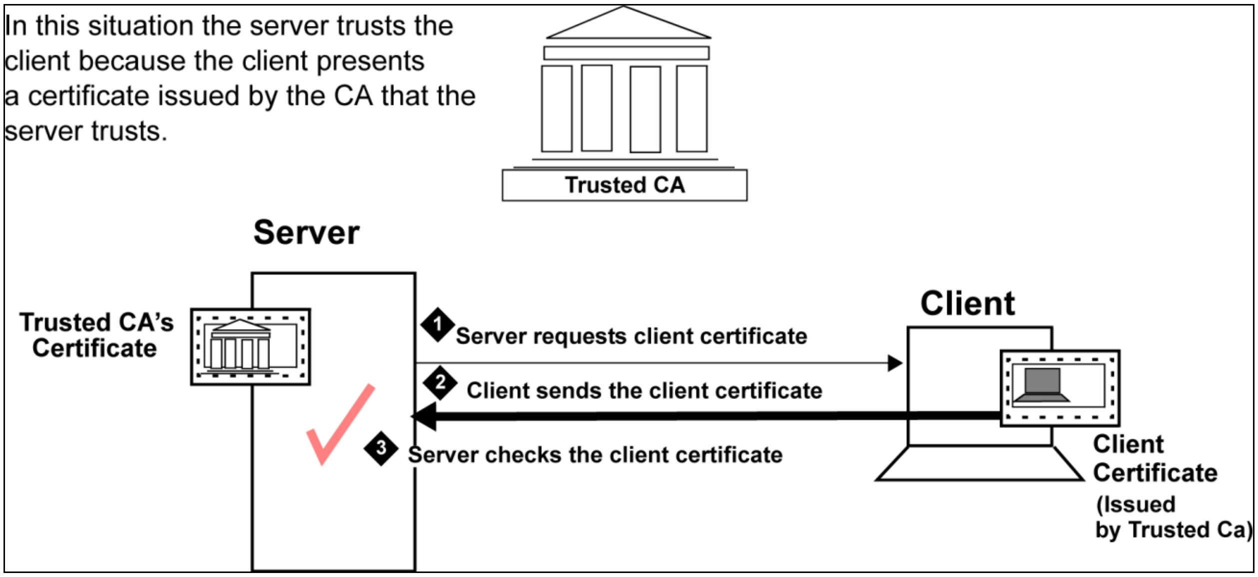
Select : All, None

7. TLS Management

Transport Layer Security (TLS) establishes unique cryptographically secure sessions for each connection request. When an entity (client) establishes a TLS session to a remote entity (server), TLS provides authentication, privacy and reliability. The following section describes TLS certificate installation so that clients authenticate servers or, optionally, servers authenticate clients.

Client authentication is similar to server authentication, except that the roles are reversed. The client authentication comes into play for encrypted connections with **mutual authentication**, and applies in addition to server authentication.

As shown below, the server also sends a request to the client for the client identity certificate, the client then sends its identity certificate to the server, and the server checks the received client identity certificate against the CA certificates that are saved in its trust store, to verify that the client identify certificate is indeed signed by a CA that the server trusts.



In the mutual authentication case, in addition to everything that was required for server authentication, the client must also have a client identity certificate created and signed by a trusted CA, and the server must have a copy of the CA certificate to be used in authenticating the client identity certificate.

In summary, the required additional manual procedures include:

- Avaya Aura® System Manager provides the third-party CA trusted Root Certificate. Install third-party CA trusted Root Certificate on Enghouse Communication Portal Server.
- Generate Certificate Signing Request and Private Key on Enghouse Communication Portal Server. Enghouse supplies a Certificate Signing Request (CSR) which is then signed by the Avaya Aura® System Manager Certificate Authority (CA).
- The resulting Identity Certificate is then installed on the Enghouse Communication Portal Server. Install the Third-Party Signed Identity Certificate into Enghouse Communication Portal Server.

Note: It is assumed the generating of the third-party CA trusted Root Certificate has been previously completed and is not discussed here. See the **Appendix** for the procedure for adding the Communication Portal Entity on the Certificate Authority and for generating the signed identity certificate.

8. Configuration of Enghouse Interactive Communications Portal 9.0

This section describes the steps required to configure Enghouse Interactive Communications Portal 10.1 to interoperate with Session Manager and Communication Manager. These steps include:

- Media Gateway Driver Configuration.
- Configuration file creation.
- Change Outbound Dial plan.
- Set the SIP transfer type parameter.
- Enable TLS and SRTP in the CTIC Media Gateway for SIP.

8.1. Media Gateway driver configuration

When using Media Gateway perform the following steps to modify the configuration parameters in the Media Gateway configuration files.

- Create the avaya.xml gateway configuration file.
- Change the outbound dial plan.
- Set the SIP transfer type parameter.

8.2. Create the avaya.xml gateway configuration file

To configure CP for this integration, prepare a gateway configuration file by performing the following steps.

- In the <Media Gateway install folder>\conf\sip_profiles\external folder, create a new text (.txt) file named *avaya.xml* with the following content. By default, Media Gateway is installed to C:\Program Files\Enghouse Interactive\Media Gateway.
- <include>
- <gateway name="AVAYA">
- Enter the IP address for Session Manager in the **realm** parameter value.
- <param name="realm" value="xxx.xxx.xxx.xxx"/>
- <param name="username" value="not-used"/>
- <param name="password" value="not-used"/>
- <param name="register" value="false"/>
- <param name="caller-id-in-from" value="false"/>
- <param name="register-transport" value="tcp"/>
- </gateway>
- </include>

8.3. Change the outbound dial plan

To configure CP for this integration, you must change the outbound dial plan configuration file by performing the following steps.

- In the <Media Gateway install folder>\conf\autoload_configs folder, edit the csdialplan.conf.xml file.
- Comment the following line: **<!-- <param pattern="^(.+@.+)$" value="sofia/external/$1"**/> -->
- Add the following line immediately below the line you commented: **<param pattern="^(.+@.+)$" value="sofia/gateway/AVAYA/$1"/>**
- Save the changes.

8.4. Set the SIP transfer type parameter

By default, the SIP transfer type is set to Refer. You must change transfer type to re-Invite with following steps.

- In the <Media Gateway install folder>\conf\autoload_configs folder, edit the csinterface.conf.xml file.
- Change the parameter **<param name="sip_transfer_type" value="refer"/>** to **<param name="sip_transfer_type" value="reinvite"/>**.
- Save the changes.

8.5. Enable TLS and SRTP in the CTIC Media Gateway for SIP

To enable TLS you need to generate a Cert Signing Request (CSR) and private key on the CP server system first.

- To do that open and update "openssl.cnf" file located in <Communications Portal install folder>\ Tools\OpenSSL:

[CA_default]
default_md = sha256 # use public key default MD

[req]
default_md = sha256

[req_distinguished_name]
countryName = Country Name (2 letter code)
countryName_default = IE
countryName_min = 2
countryName_max = 2
stateOrProvinceName = State or Province Name (full name)
stateOrProvinceName_default = Connacht
localityName = Locality Name (eg, city)
localityName_default = Galway
0.organizationName = Organization Name (eg, company)
0.organizationName_default = Avaya

organizationalUnitName = Organizational Unit Name (eg, section)
organizationalUnitName_default = DEVCONNECT
commonName = Common Name (e.g. server FQDN or YOUR name)
commonName_default = ENGHOUSE2012CP

// Extensions to add to a certificate request
basicConstraints = CA:FALSE
keyUsage = nonRepudiation, digitalSignature, keyEncipherment, dataEncipherment, keyAgreement
extendedKeyUsage = serverAuth, clientAuth
subjectAltName = @alt_names

[alt_names]
DNS.1 = smgr1bgvm.avaya.com

When it is asking use default values in [ ] by pressing ENTER.

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

openssl_scp.exe req -out ENGHOUSE2012CP.csr -new -newkey rsa:2048 -nodes -sha256 -keyout ENGHOUSE2012CP.key -config openssl.cnf
Cert Signing Request (CSR) .csr file and private key file are generated. CSR file then can be sent to Avaya which can make the Identiy Certificate (.pem file).

- Manually concatenate your private key file, Identity Certificate file and Root Certificate Authority (provided by Avaya, too) file into "**tls.pem**" file. Copy that file into <Media Gateway install folder>\conf\ssl folder.

- In the <Media Gateway install folder>\conf folder, edit the vars.xml file. In the <!-- External SIP Profile --> section change the parameter

<**X-PRE-PROCESS cmd="set" data="external_ssl_enable=false"**/> to <**X-PRE-PROCESS cmd="set" data="external_ssl_enable=true"**/>

- In the <Media Gateway install folder>\conf\sip_profiles\external folder, edit avaya.xml file. Edit the parameter

<**param name="register-transport" value="tcp"**/> to <**param name="register-transport" value="tls"**/>

This completes the steps to set TLS with Media Gateway. To complete the Media Gateway configuration, SRTP has to be enabled too.

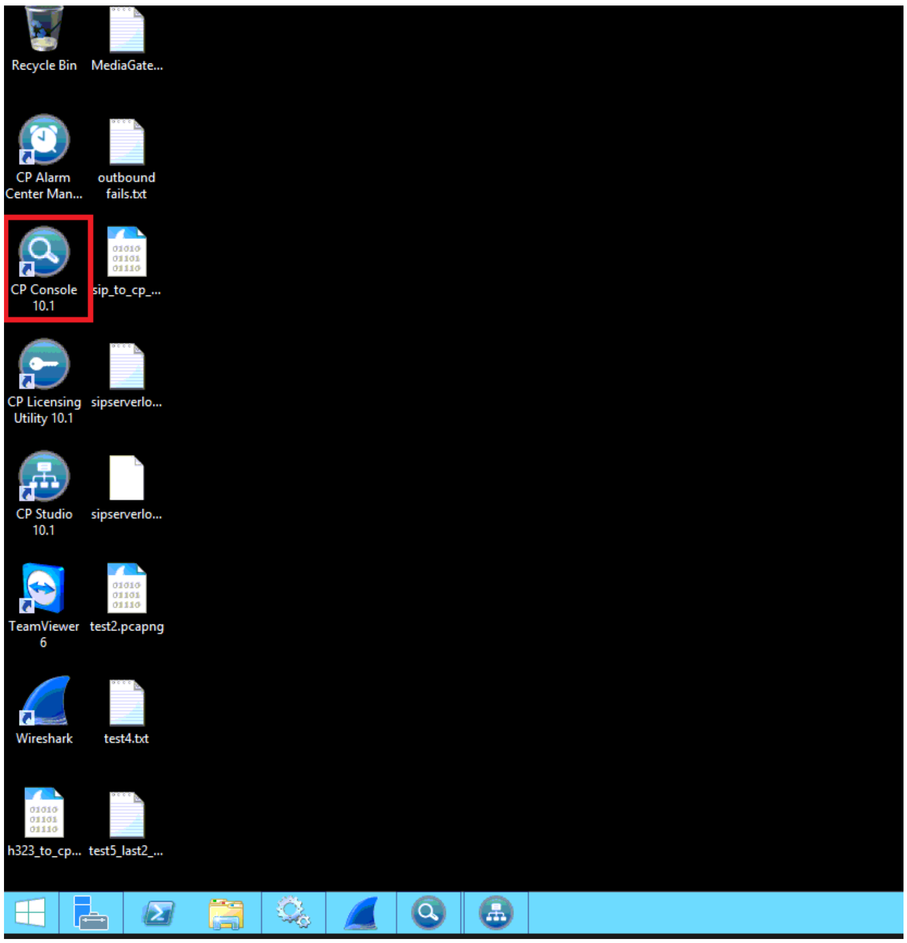
- In the <Media Gateway install folder>\conf\autoload_configs folder, edit the csdialplan.conf.xml file. Change the parameter:

<**param pattern="^(.+@.+)$" value="sofia/gateway/AVAYA/$1"**/> to <**param pattern="^(.@.)$" value="**

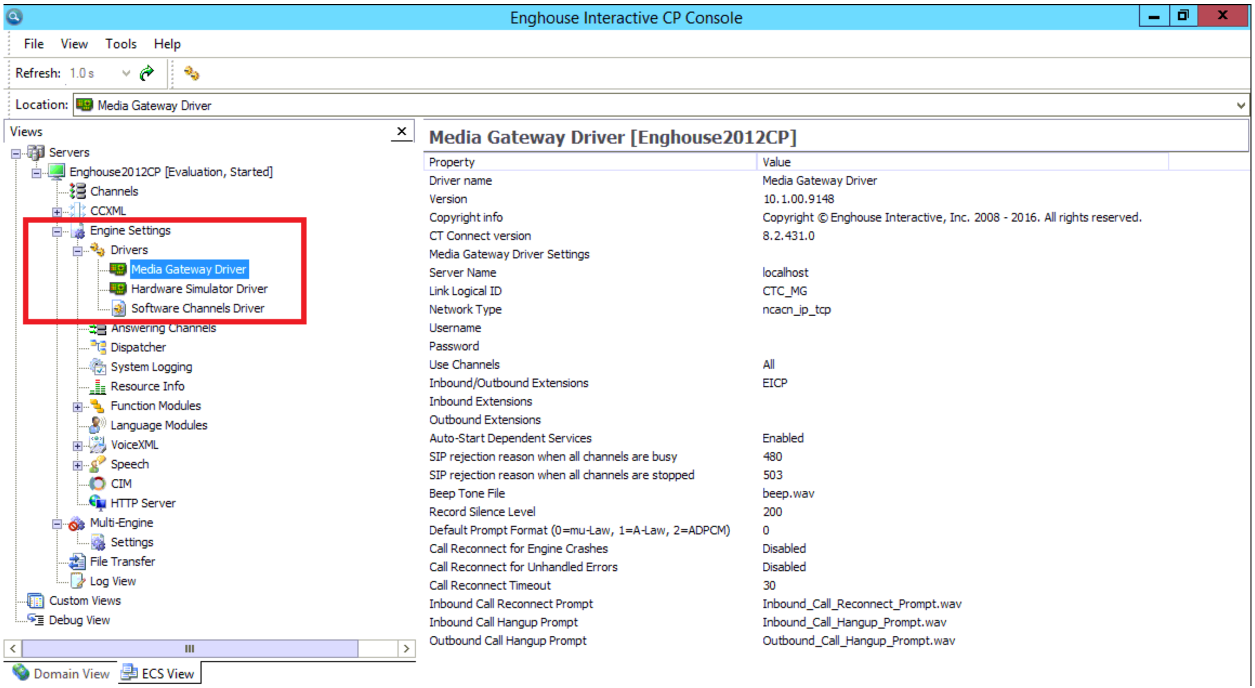
{sip_secure_media=true,rtp_secure_media=true:AES_CM_128_HMAC_SHA1_80,sdp_secure_savp_only=true} sofia/gateway/AVAYA/$1"/>

To complete the CP configuration, you must stop the CP engine, stop the Media Gateway service (if it is already started) and restart the CP Engine.

To configure the Media Gateway Driver open the **CP Console 9.0** by double clicking on the shortcut as shown below.



In the left window, navigate to **Servers →[Server Name]→Engine Settings→Drivers→Media Gateway Driver**.



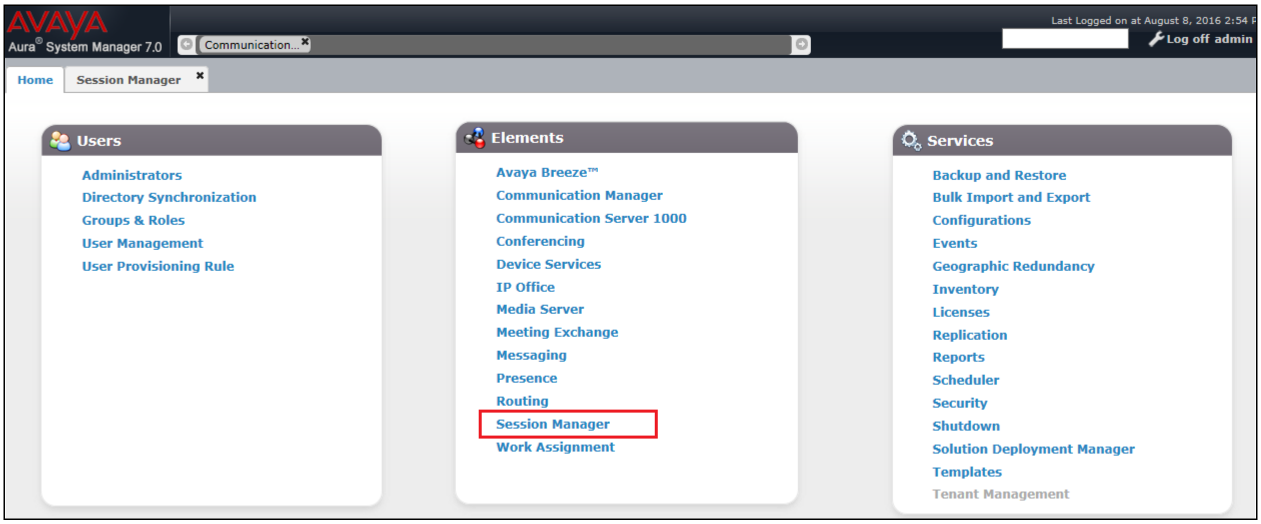
Please note that configuration of Communications Portal with regards to the setup of the IVR is outside the scope of these Application Notes, for more information on this setup please refer to **Section 11** of these Application Notes.

9. Verification Steps

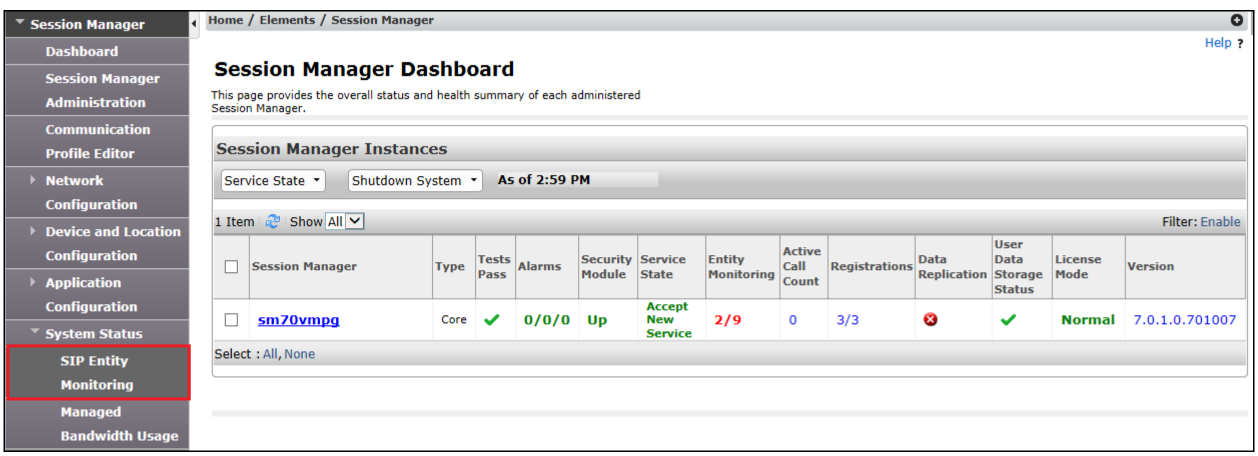
To verify a successful configuration of Enghouse Interactive Communications Portal and Session Manager/Communication Manager a call is placed from a Communication Manager telephone to the Communications Portal with the caller getting answered successfully hearing clear and audible speech.

9.1. Verify Enghouse Interactive Communications Portal SIP Entity is up

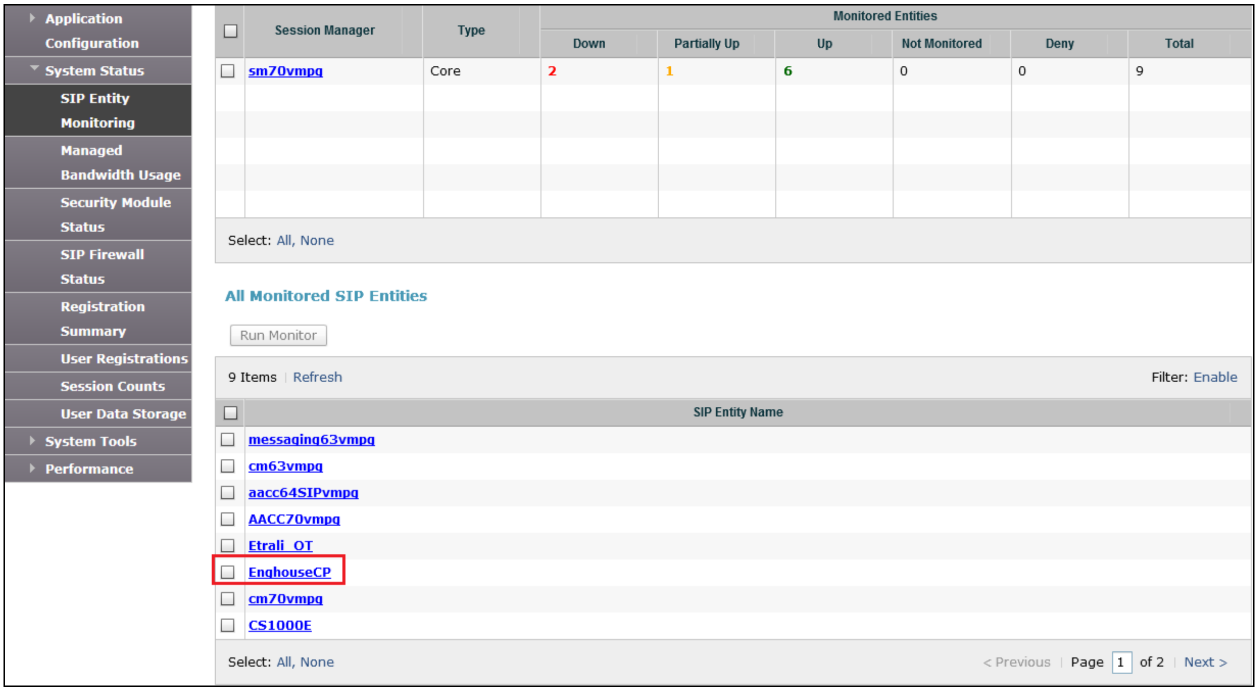
Log in to System Manager as per **Section 6.1**. From the main menu select Session Manager as shown below.



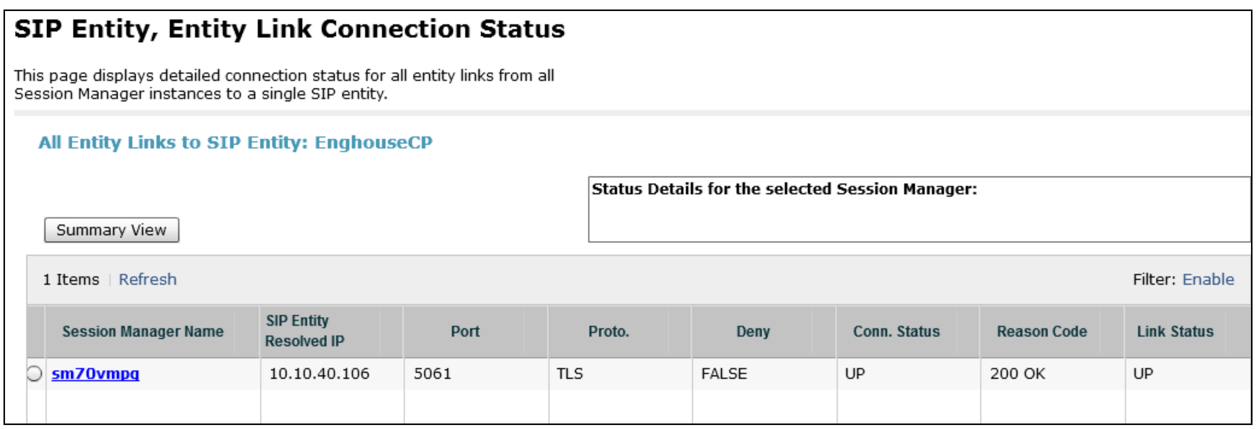
Navigate to **System Status → SIP Entity Monitoring**.



Select the **EnghouseCP** SIP Entity.

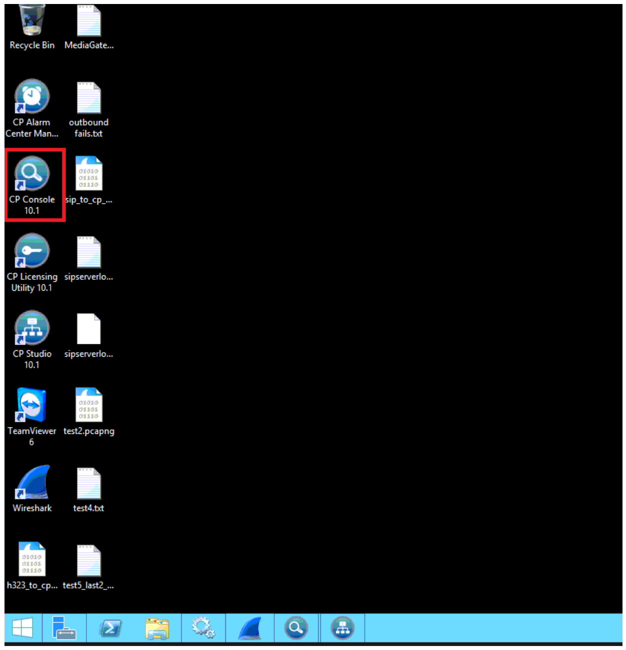


Note that both the **Conn. Status** and **Link Status** show **UP**.

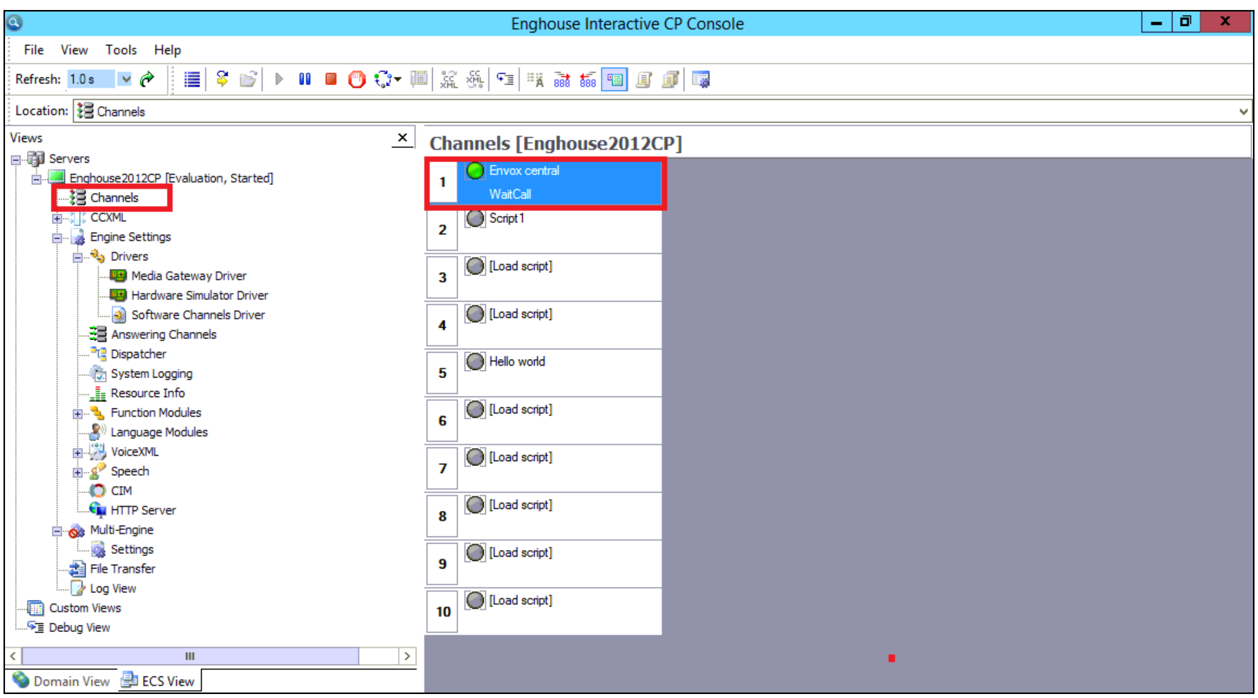


9.2. Verify Enghouse Interactive Communications Portal IVR script

Open the **CP Console 9.0** by double clicking on the shortcut as shown below.



Channel 1 below has the script **Envox Central** associated with it**,** this should also show as green.



10.Conclusion

These Application Notes describe the configuration steps required for Enghouse Interactive Communications Portal 10.1 to successfully interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2.**

11.Additional References

This section references the Avaya and Enghouse product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at *http://support.avaya.com*.

[1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509

[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* Document ID 555-245-205

[3] *Administering Avaya Aura® Session Manager,* Release 7.0, 03-603324

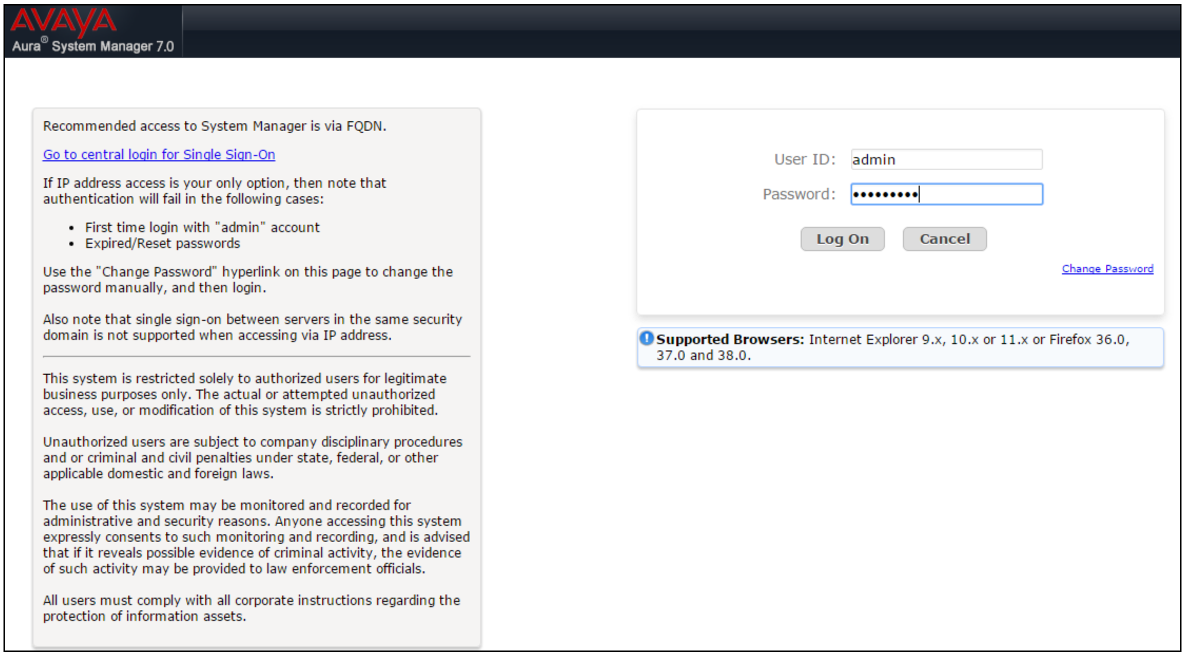
Product documentation for Enghouse Interactive Communications Portal can be obtained by visiting the following website, www.enghouseinteractive.com

Appendix

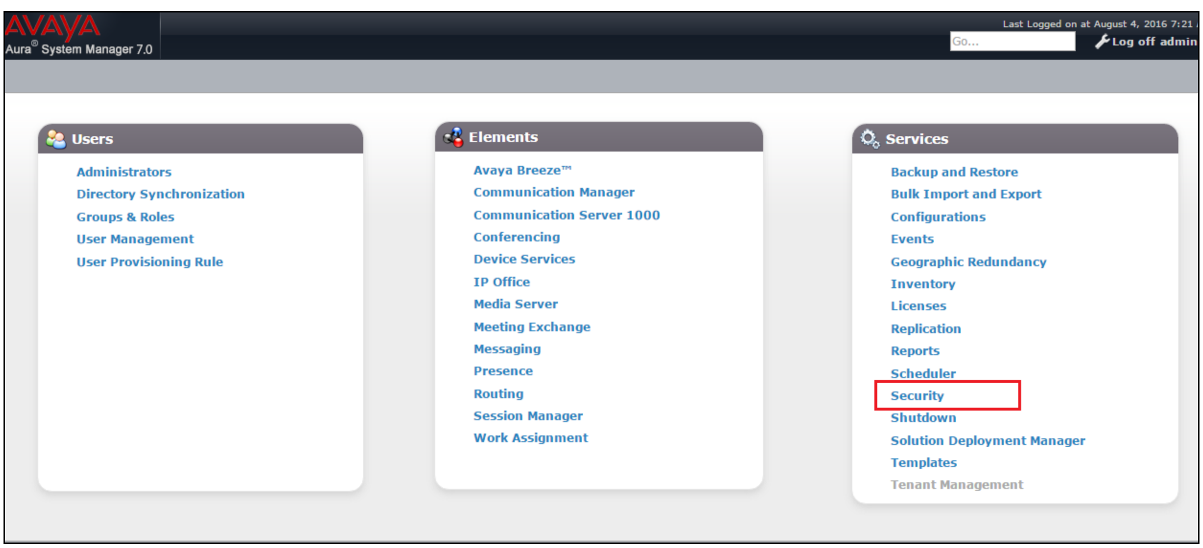
The following section shows the creation of the Enghouse Communications Portal End Entity on the 3rd party CA in order to sign the CSR generated by Communications Portal.

Add End Entity

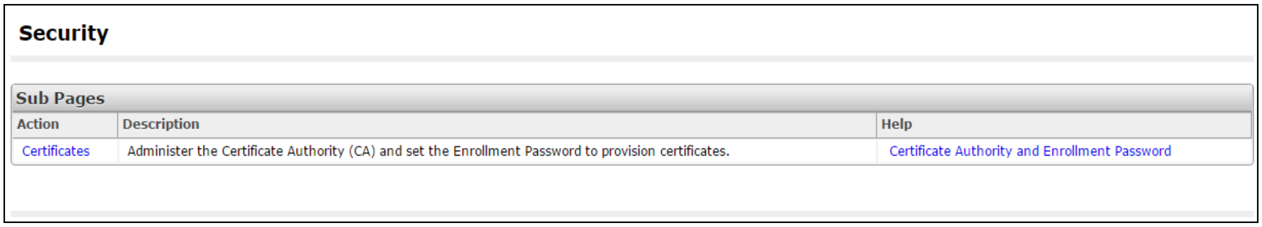
The 3rd party endpoint (Communications Portal) is added to the CA as an end entity. Log in to the Certificate Authority, in this case a System Manager.



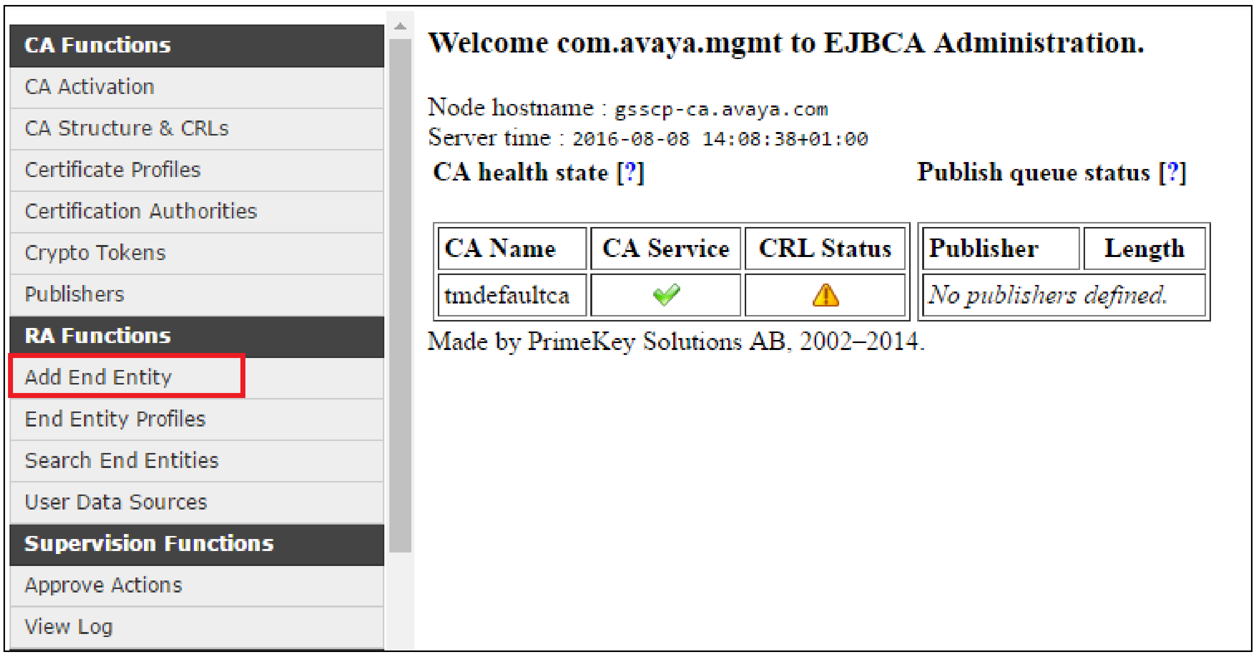
Click on **Security** from the main menu.



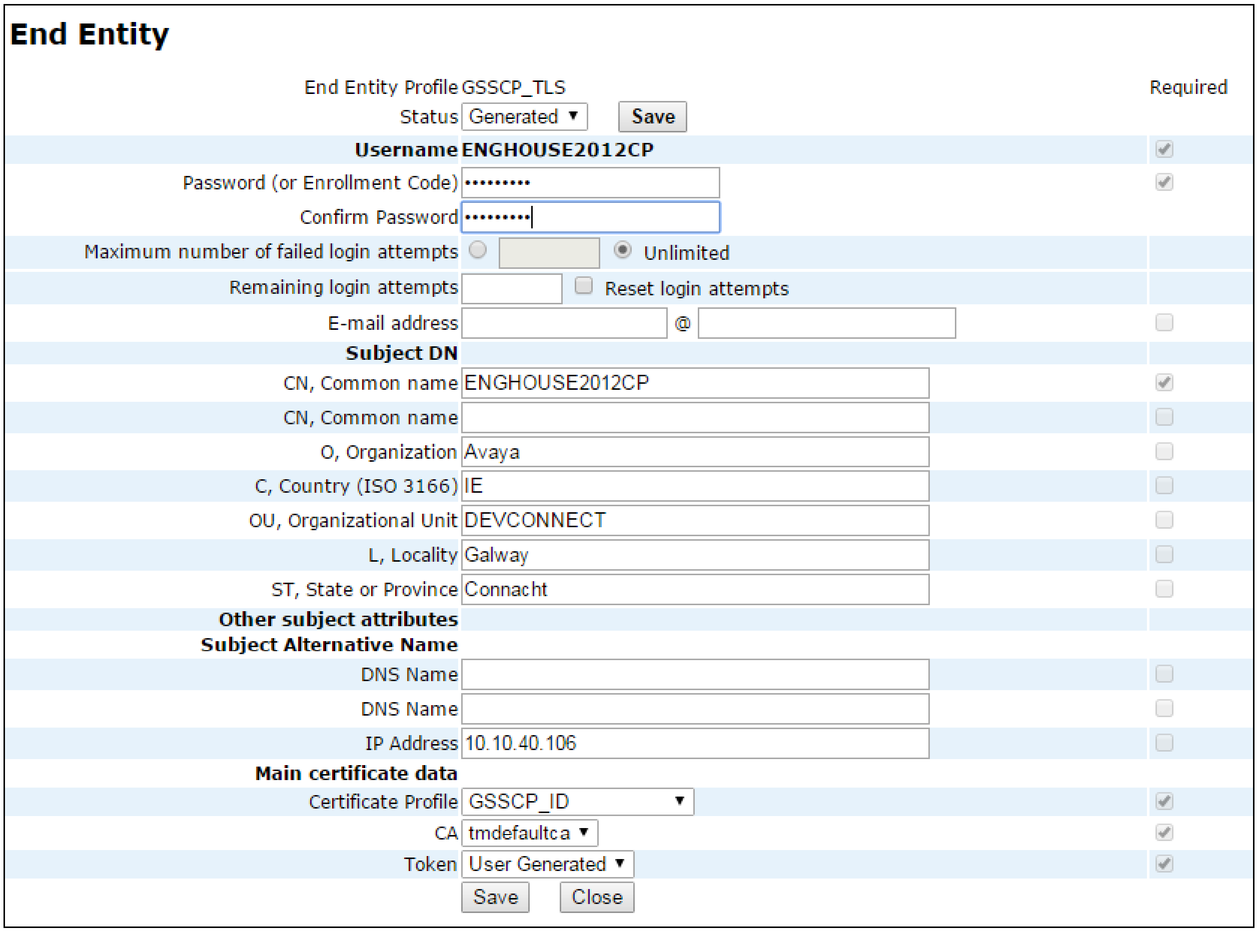
Click on **Certificates**.



Click on **Add End Entity**.

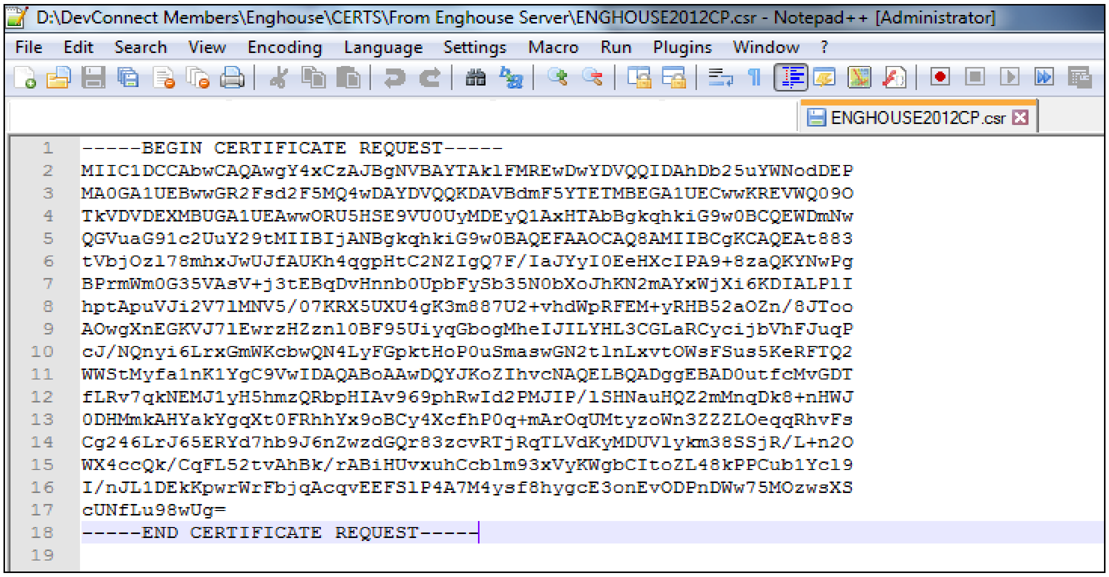


The following is an example of the **End Entity** that was added for compliance testing. Take note of the **Password (orEnrollment Code)**, this will be required later, the **IP address** will be that of the Enghouse Communications Portal and the **Common name** and **Username** should be hostname associated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



Generate the certificate signing request

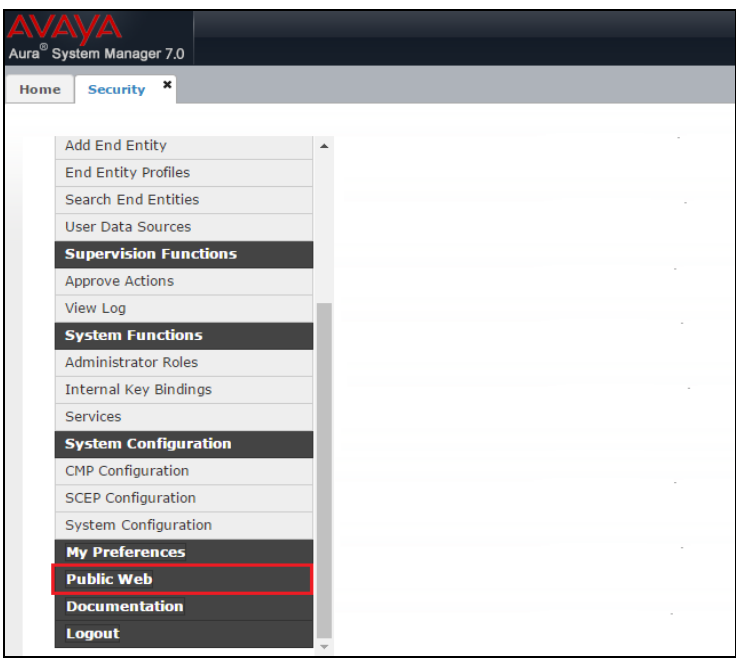
In public key infrastructure (PKI) systems, a certificate signing request (also CSR or certification request) is a message sent from an applicant to a certificate authority in order to apply for a digital identity certificate. The most common format for CSRs is the PKCS #10 specification and another is the Signed Public Key and Challenge Spkac format generated by some Web browsers. The 3rd party application will generate the CSR, the following is the CSR generated by Enghouse.



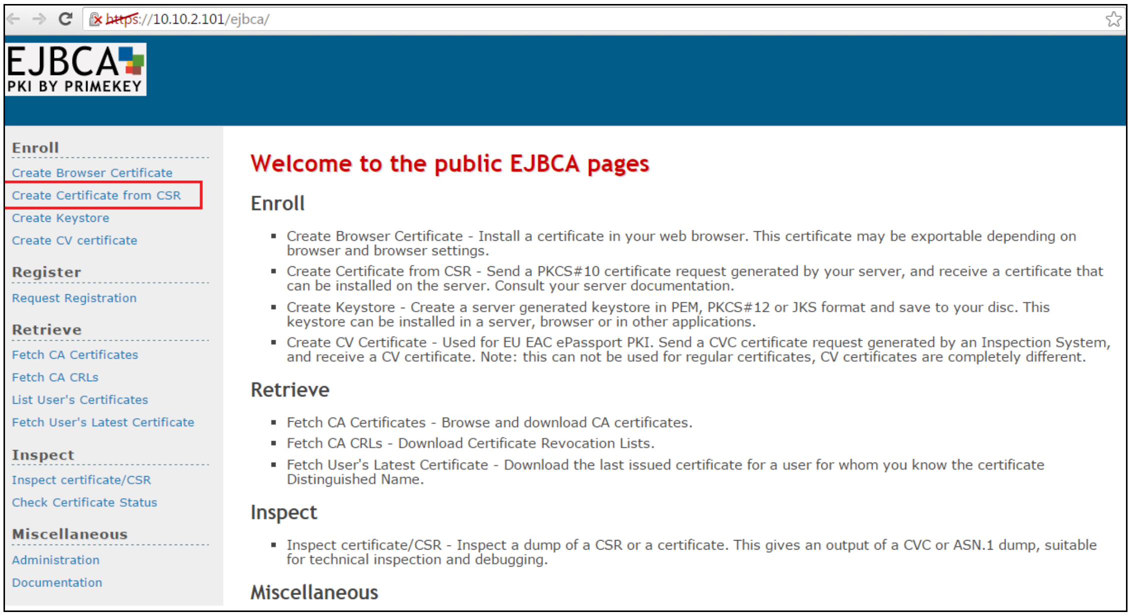

```
-----BEGIN CERTIFICATE REQUEST-----
MIIC1DCCAbwCAQAwgY4xCzAJBgNVBAYTAklFMREwDwYDVQQIDAhDb25uYWNodDEP
MA0GA1UEBwwGR2Fsd2F5MQ4wDAYDVQQKDAVBdmF5YTETMBEGA1UECwwKREVWQ09O
TkVDVDEXMBUGA1UEAwwORU5HSE9VU0UyMDEyQ1AxHTAbBgkqhkiG9w0BCQEWDmNw
QGVuaG91c2UuY29tMIIBIjANBgkqhkiG9w0BAQEFAAOCAQ8AMIIBCgKCAQEAt883
tVbjOzl78mhxJwUJfAUKh4qgpHtC2NZIgQ7F/IaJYyI0EeHXcIPA9+8zaQKYNwPg
BPrmWm0G35VAsV+j3tEBqDvHnnb0UpbFySb35N0bXoJhKN2mAYxWjXi6KDIALPlI
hptApuVJi2V7lMNV5/07KRX5UXU4gK3m887U2+vhdWpRFEM+yRHB52aOZn/8JToo
AOwgXnEGKVJ7lEwrzHZznl0BF95UiyqGbogMheIJILYHL3CGLaRCycijbVhFJuqP
cJ/NQnyi6LrxGmWKcbwQN4LyFGpktHoP0uSmaswGN2tlnLxvtOWsFSus5KeRFTQ2
WWStMyfa1nK1YgC9VwIDAQABoAAwDQYJKoZIhvcNAQELBQADggEBAD0utfcMvGDT
fLRv7qkNEMJ1yH5hmzQRbpHIAv969phRwId2PMJIP/1SHNauHQZ2mMnqDk8+nHWJ
0DHMmkAHYakYgqXt0FRhhYx9oBCy4XcfhP0q+mArOqUMtyzoWn3ZZZLOeqqRhvFs
Cg246LrJ65ERYd7hb9J6nZwzdGQr83zcvRTjRqTLVdKyMDUVlykm38SSjR/L+n2O
WX4ccQk/CqFL52tvAhBk/rABiHUvxuhCcblm93xVyKWgbCItoZL48kPPCub1Ycl9
I/nJL1DEkKpwrWrFbjqAcqvEEFSlP4A7M4ysf8hygcE3onEvODPnDWw75MOzwsXS
cUNfLu98wUg=
-----END CERTIFICATE REQUEST-----
```


Generate the Identity Certificate

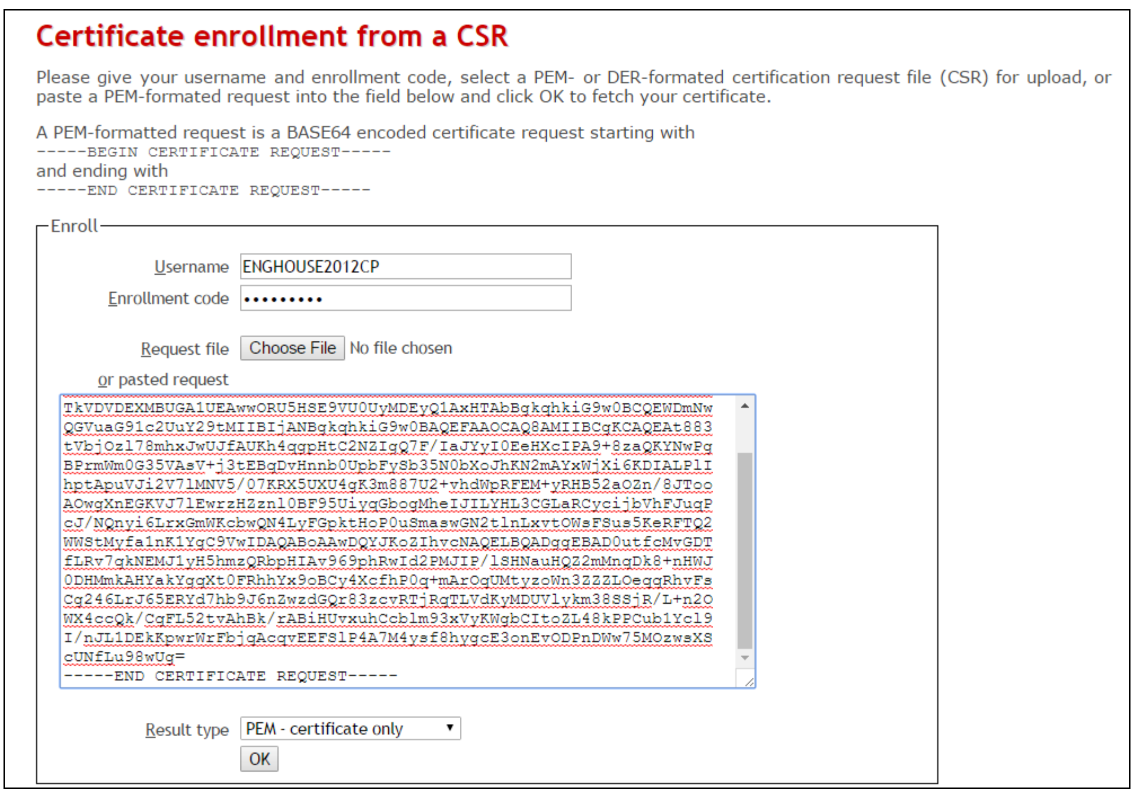
From the CA, click on the **Public Web** down the left side of the page.



The following web page is opened where a **Certificate from CSR** can be generated.

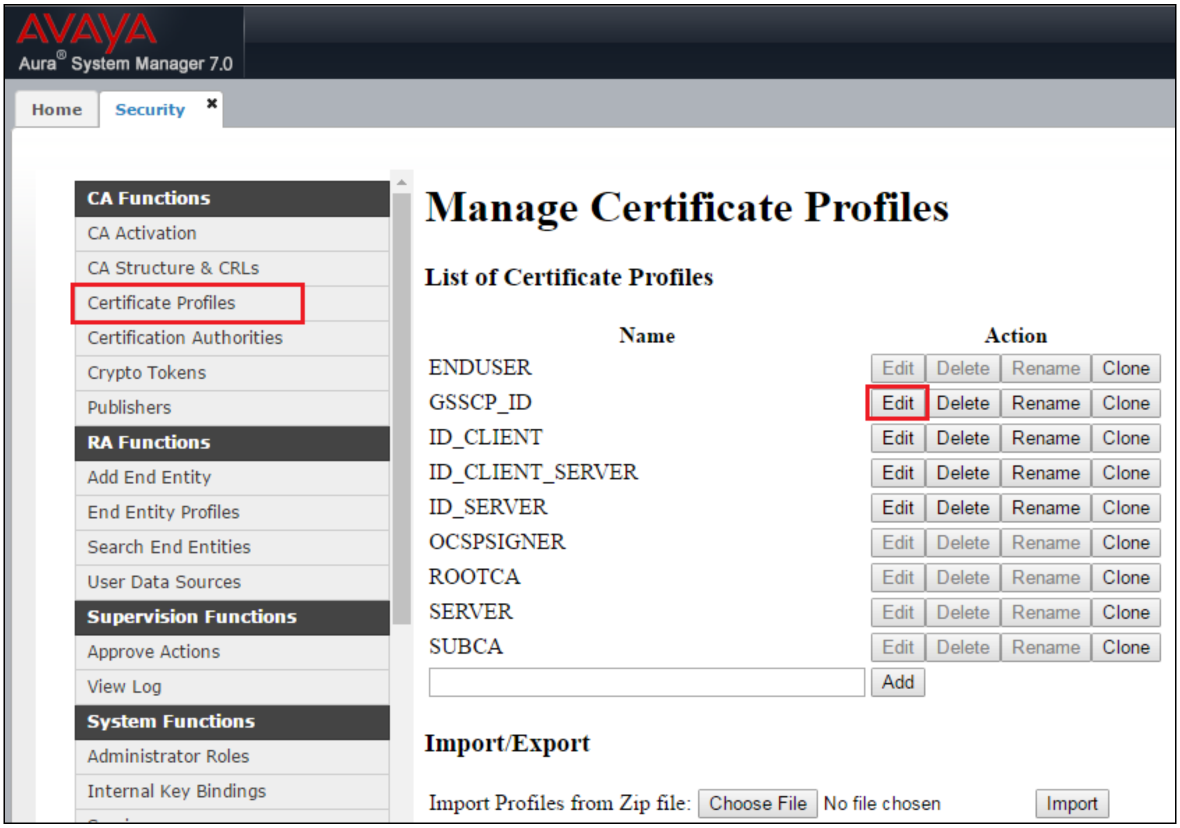


Copy and paste in the CSR request, this is taken from the CSR generated by Enghouse as shown on the previous page.

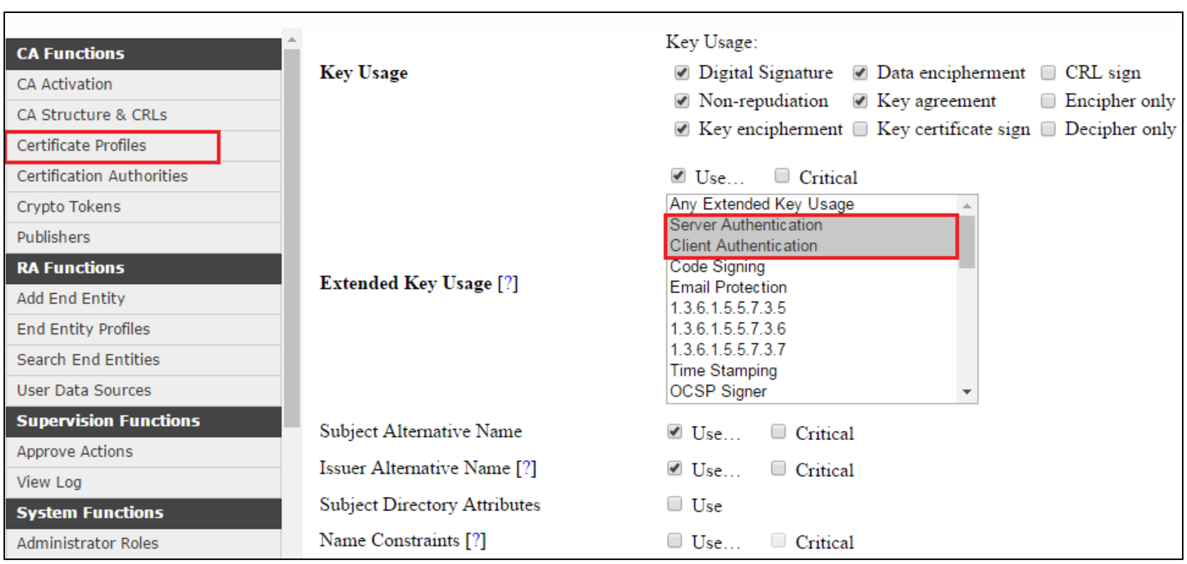


Mutual Authentication

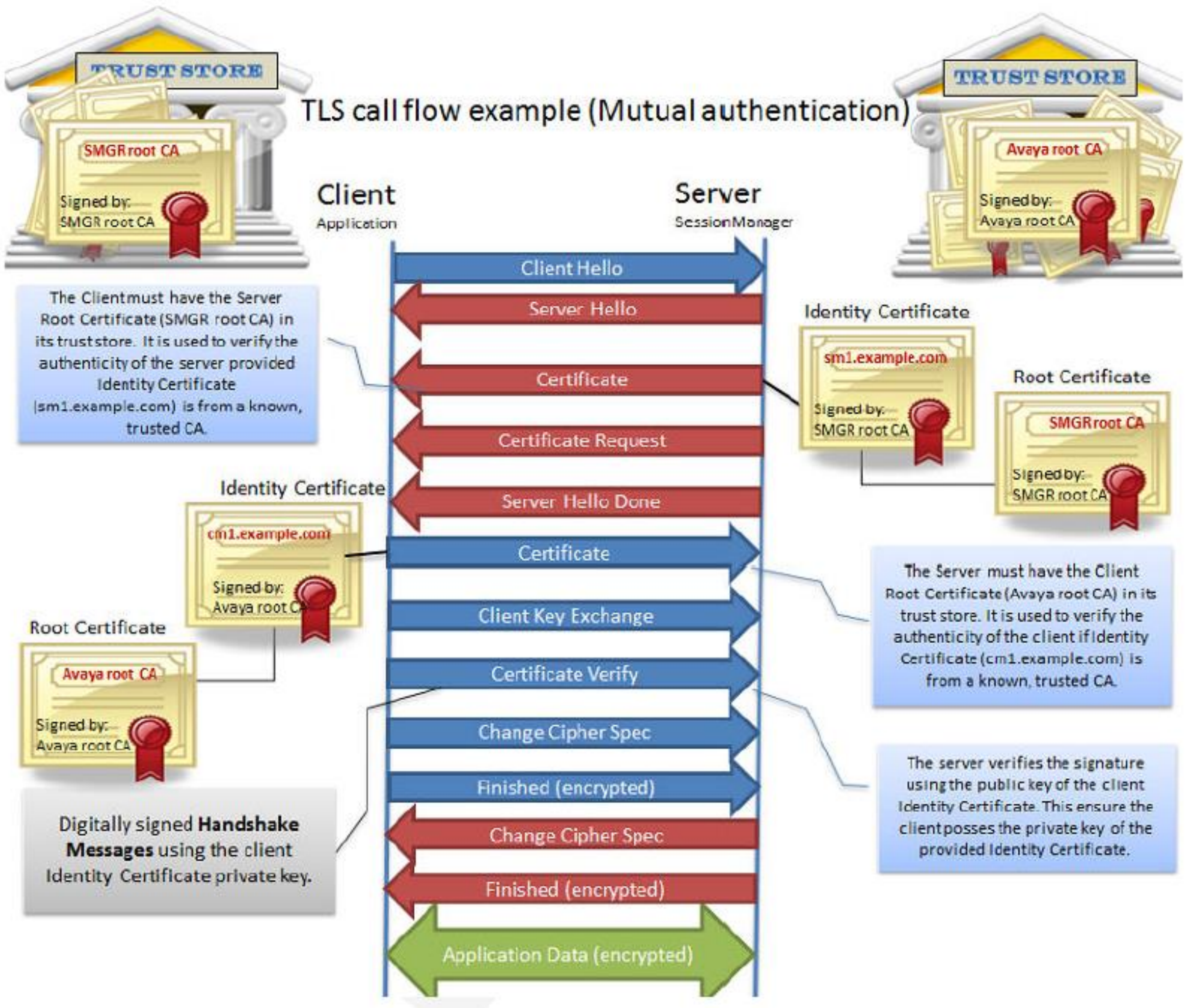
Mutual Authentication is determined in the Certificate Profile. Click on **Certificate Profiles** in the left window, this will display a list of profiles and edit the appropriate profile as shown below.



The profile below shows that both Server and Client Authentication are selected.



The flow below show what happens in the Mutual Authentication scenario.



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.