Avaya Solution & Interoperability Test Lab

Application Notes for Enghouse Interactive Communications Portal 10.4 using Dialogic Host Media Processing (HMP) 3.0 Service Update 525 with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2 - Issue 1.0

Abstract

These Application Notes describe the configuration steps for Enghouse Interactive Communications Portal 10.4 using Dialogic Host Media Processing (HMP) 3.0 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2. 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 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 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2 using Transport Control Protocol (TCP) and Real-time Transport Protocol (RTP). Enghouse Interactive 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.

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) 10.4 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 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. Dialogic HMP 3.0 driver to perform all telephony functions on the server. This Dialogic Host Media Processing (HMP) software 3.0 facilitates the Communications Portal connectivity to Session Manager.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Enghouse Communications Portal did not include use of any specific encryption features as requested by Enghouse.

Note: Enghouse Communications Portal also support SRTP and TLS. The *Configuring Enghouse Interactive Communications Portal 10.4 using Dialogic HMP 3.0 SU525 driver with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2 using TLS and SRTP* are mentioned in others Application Notes.

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 Enghouse Interactive Communications Portal
- G.711A, G.711U codecs support and negotiation (without shuffling)
- **Call Forward** from Avaya Endpoints 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

2.2. Test Results

All functionality and serviceability test cases were completed successfully with the following was observed:

- There is no transport protocol mentioned in 200 OK messages from CP side. Avaya Session Manager need open both TCP and UDP ports

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

Figure 1 illustrates a sample configuration that consists of Avaya products and Enghouse Communications Portal with Dialogic Host Media Processing (HMP).

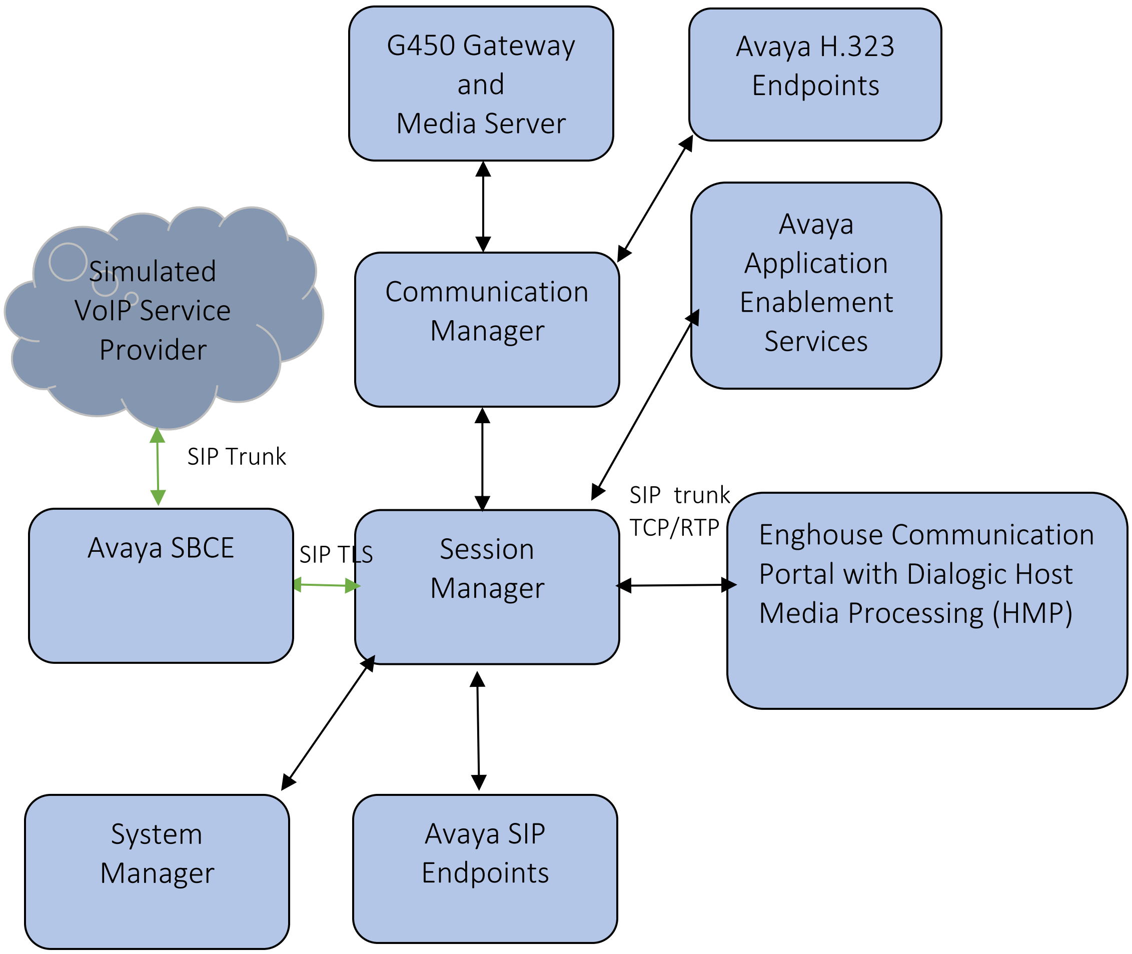


Figure 1: Test Configuration for Enghouse Communications Portal using Dialogic Host Media Processing (HMP) and the Avaya Platform.

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 .2
Avaya Aura® Session Manager in Virtual Environment	8.1.3.2
Avaya Aura® Communication Manager in Virtual Environment	8.1.3.2
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	8.1.2.0
Avaya 9608G & 9641G IP Deskphone (H.323)	6.8
Avaya Workplace Client	3.19.0
Avaya 9641 & 9621 IP Deskphone (SIP)	7.1.9
Avaya J159, J179 & J189 SIP Deskphone	4.0.9
Avaya K175 & Avaya K155	3.1.0.0
Enghouse Communications Portal with Dialogic Host Media Processing (HMP)	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 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  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 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
```

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

```
change ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: devconnect.com
    Name: SaiGon                 Stub Network Region: n
MEDIA PARAMETERS                 Intra-region IP-IP Direct Audio: yes
      Codec Set: 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 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 **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 as far as TCP or TLS (with certificates deployed) is concerned, but also if TLS protocol is set, non-secure SRTP calls will also be sent.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.722-64K                   2        20
 3: G.729            n          2        20
 4: OPUS-WB20K                  1        20
 5: G.711A           n          2        20
 6:
 7:

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

5.4. Configure 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 Communications Portal. 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 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 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 **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® System Manager

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

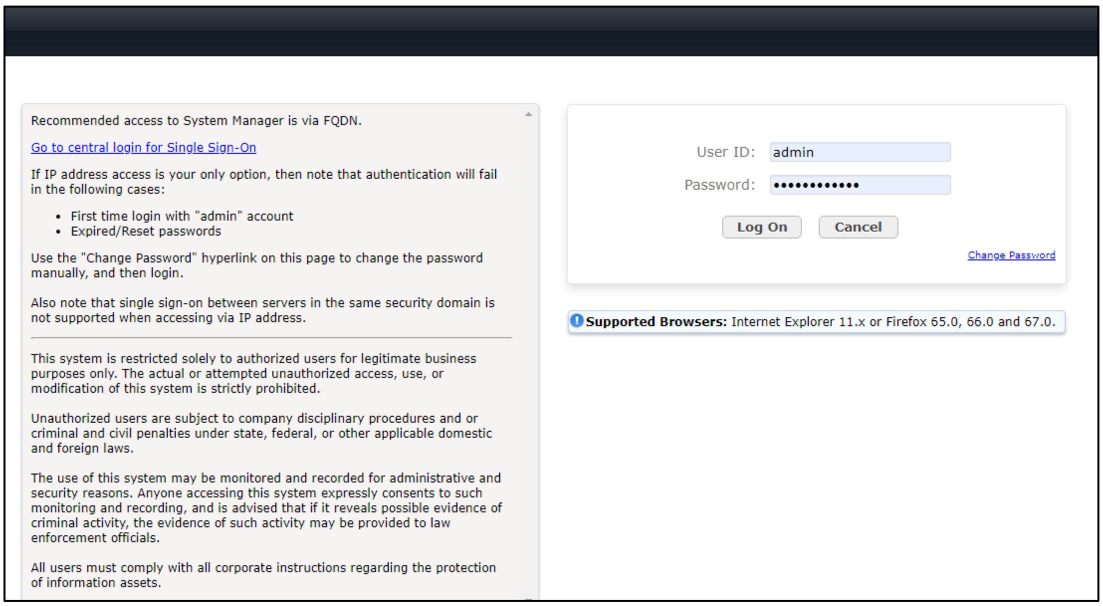
- Configure SIP Entities
- Configure Routing Policies

- Configure Dial Patterns

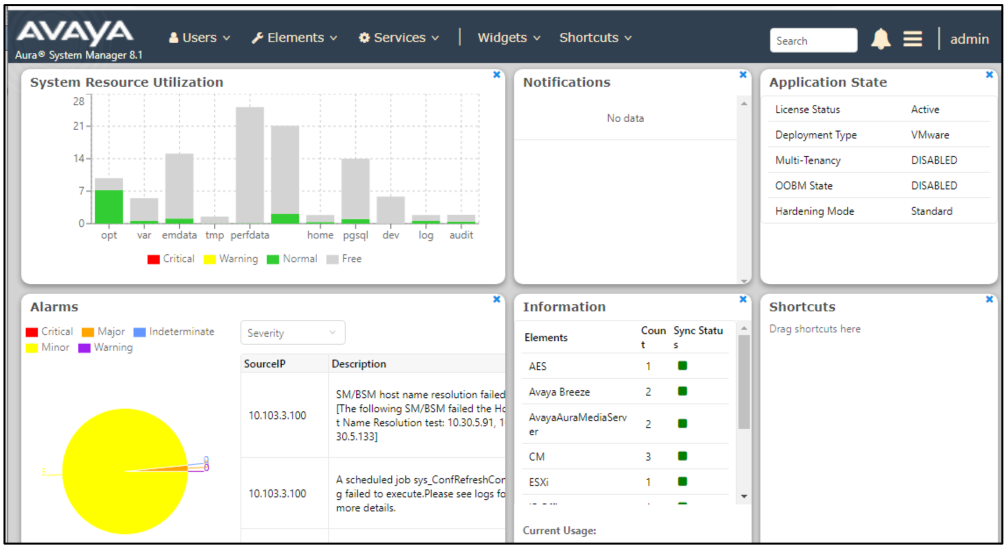
6.1. Configure SIP Entities

6.1.1. Configure SIP Entity for Enghouse Communications Portal

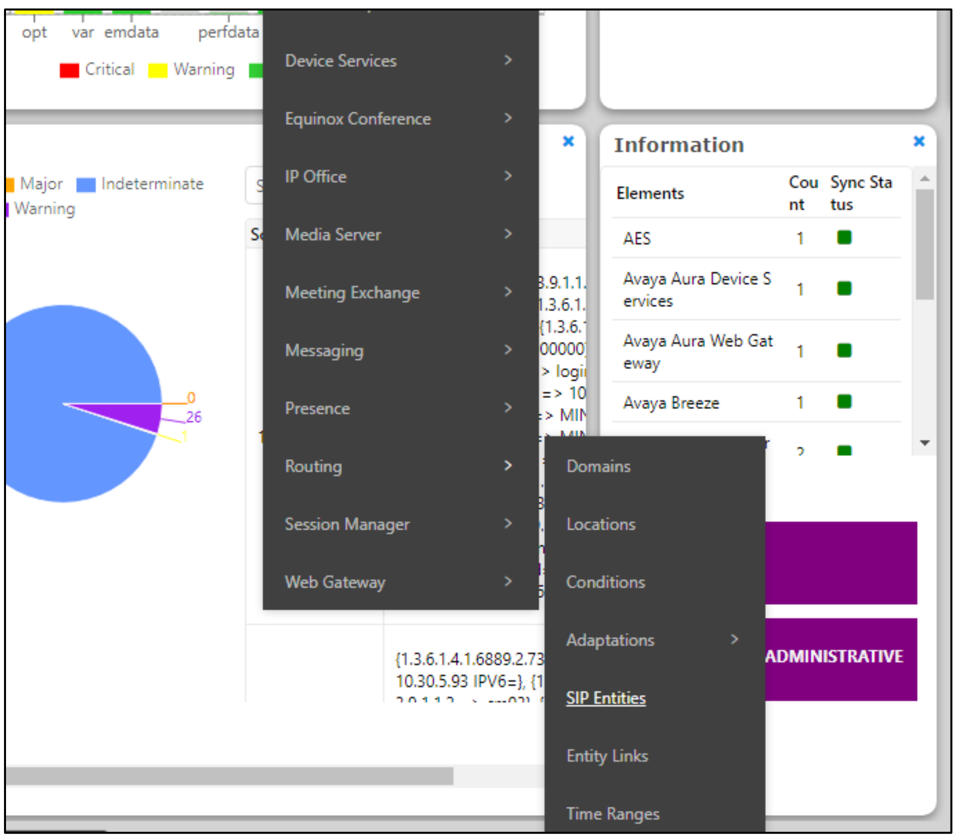
Configuration of SIP Entities is performed via System Manager. Access the System Manager Administration web interface by entering the System Manager (SMGR) URL in a web browser. Log in using appropriate credentials.



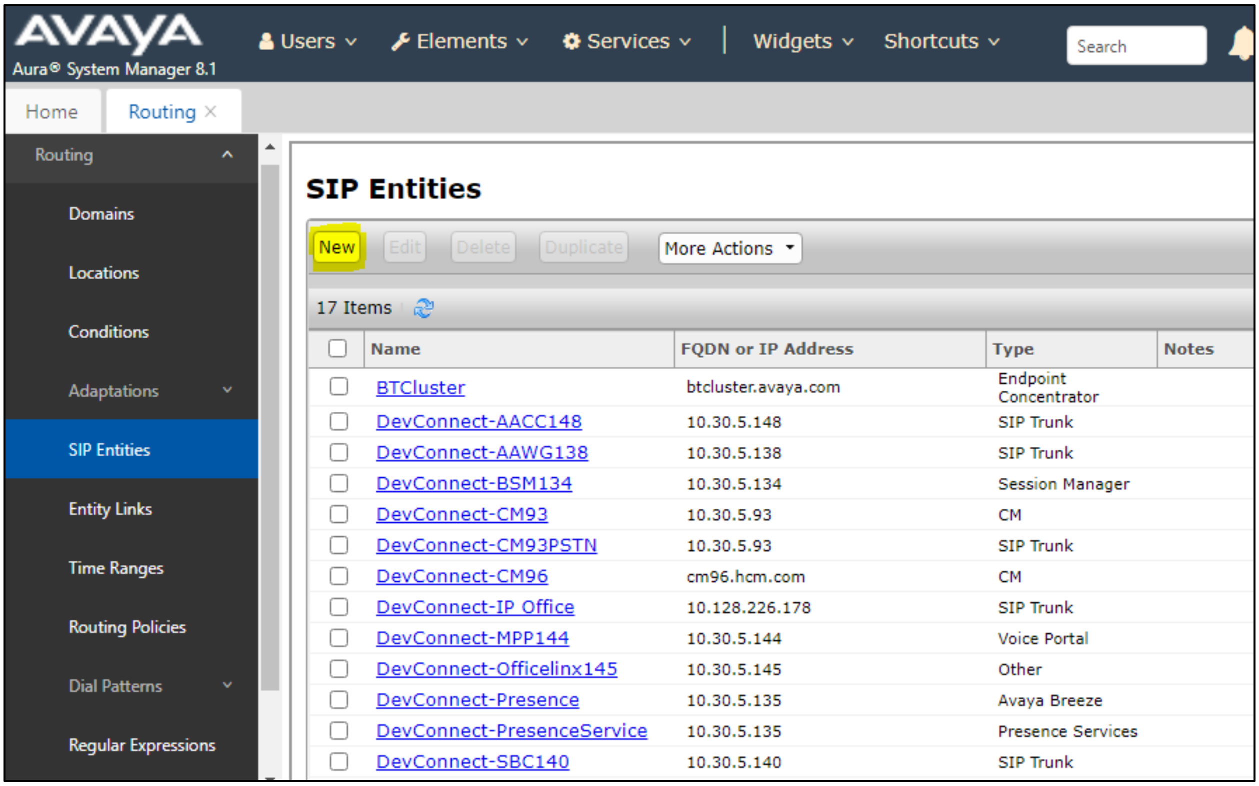
Once logged in, the following screen is displayed.



Select **Elements → Routing → SIP Entities**

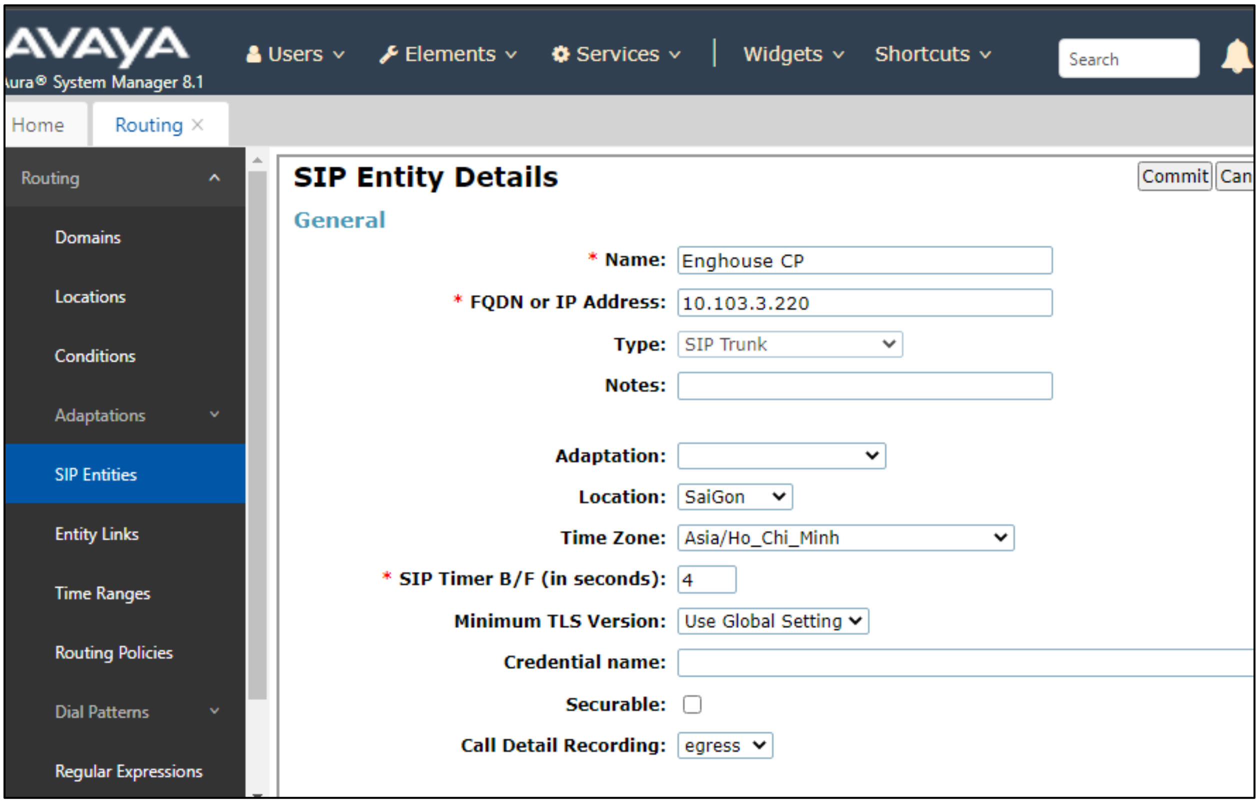


On **SIP Entities** page, press **New** to create new **SIP Entity**



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

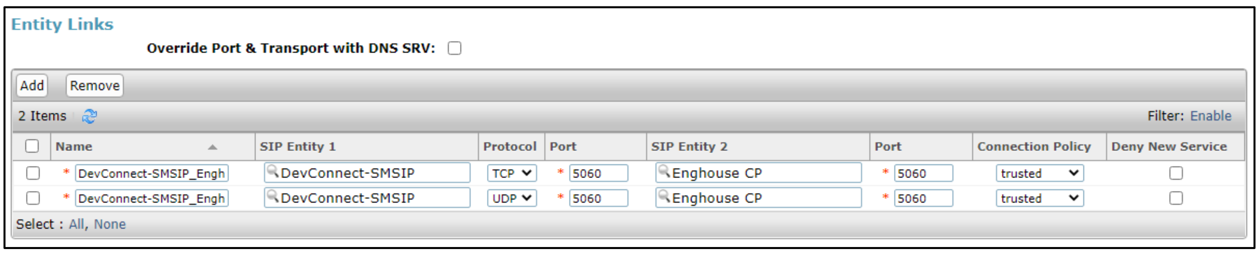
Note: The setup of a Location is specific to each site, this can be added by clicking on **Locations** on the left panel on the screen shot below, the setup of the location for this site has not been documented as part of this setup as it would be already setup as part of the site installation.



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

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "**DevConnect-SMSIP** ".
- **Protocol:** "TCP"
- **Port:** "5060"
- **SIP Entity 2:** The Communications Portal entity name from this section, in this case **"Enghouse CP"**
- **Port:** "5060"
- **Connection Policy:** "trusted"

Click Add to add one more entity link for UDP Protocol.

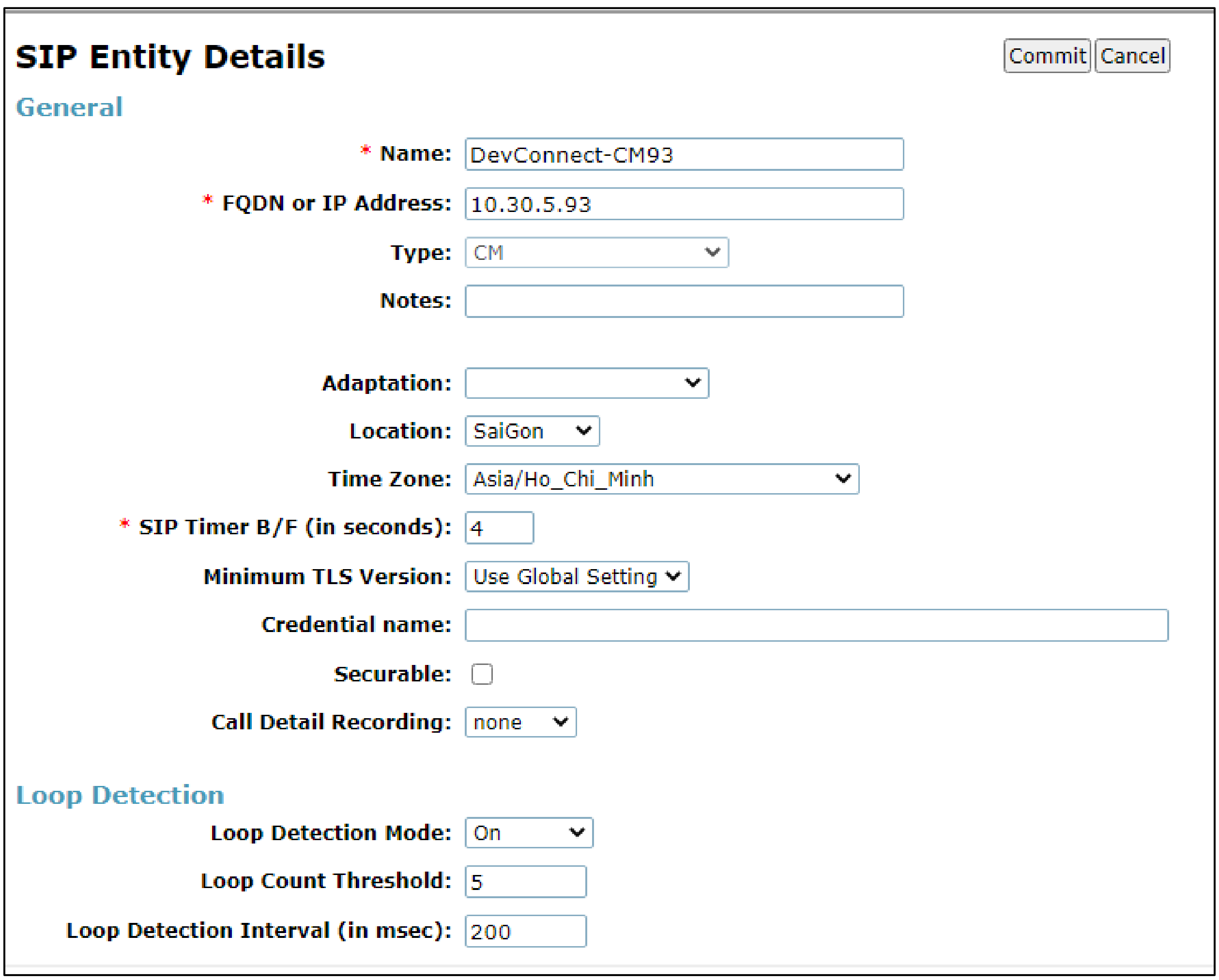


6.1.2. Configure SIP Entity for Communication Manager

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.

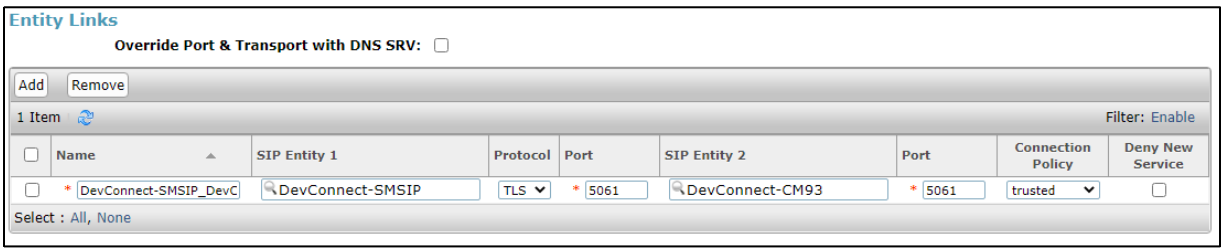
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



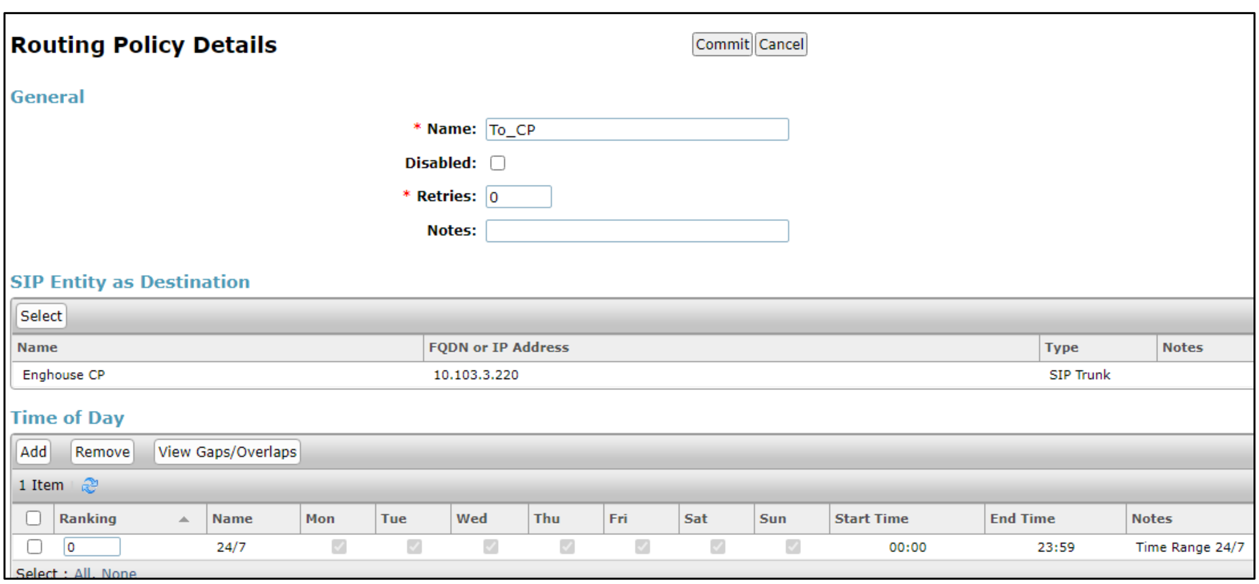
Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



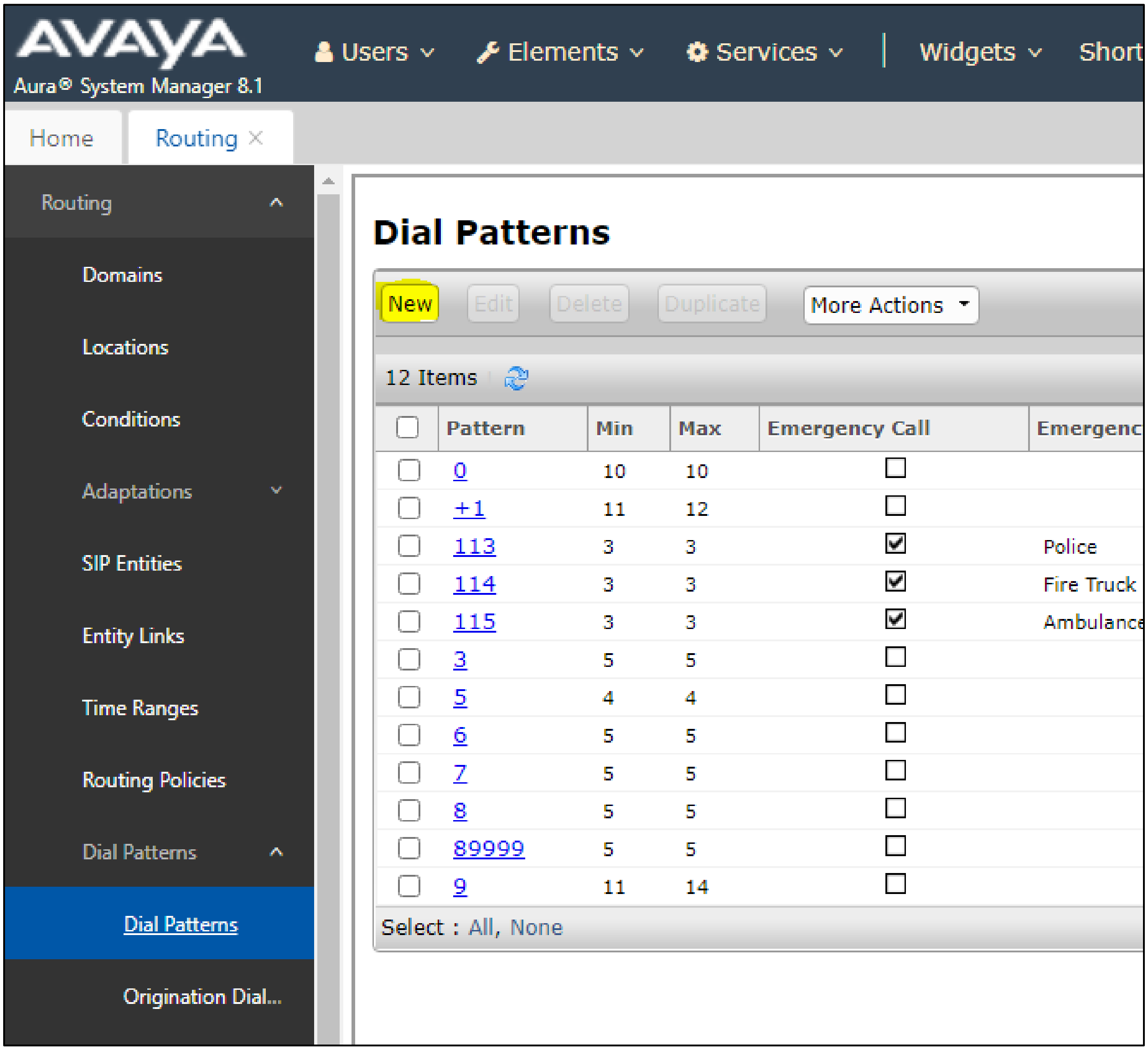
6.2. Configure Routing Policy for Enghouse Communications Portal

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**

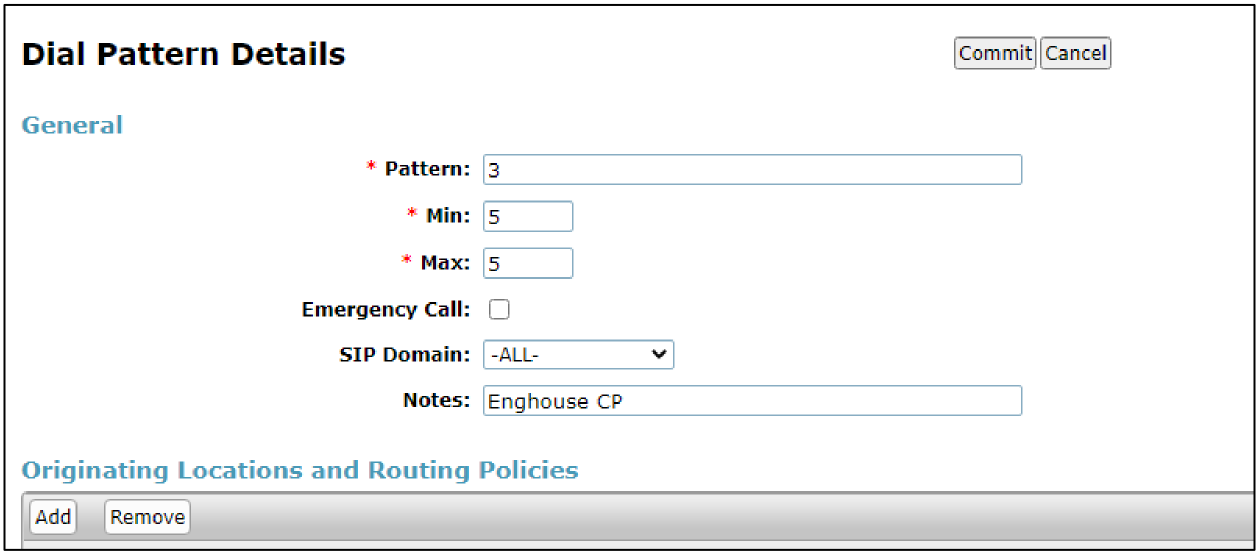


6.3. Configure Dial Pattern for Enghouse Communications Portal

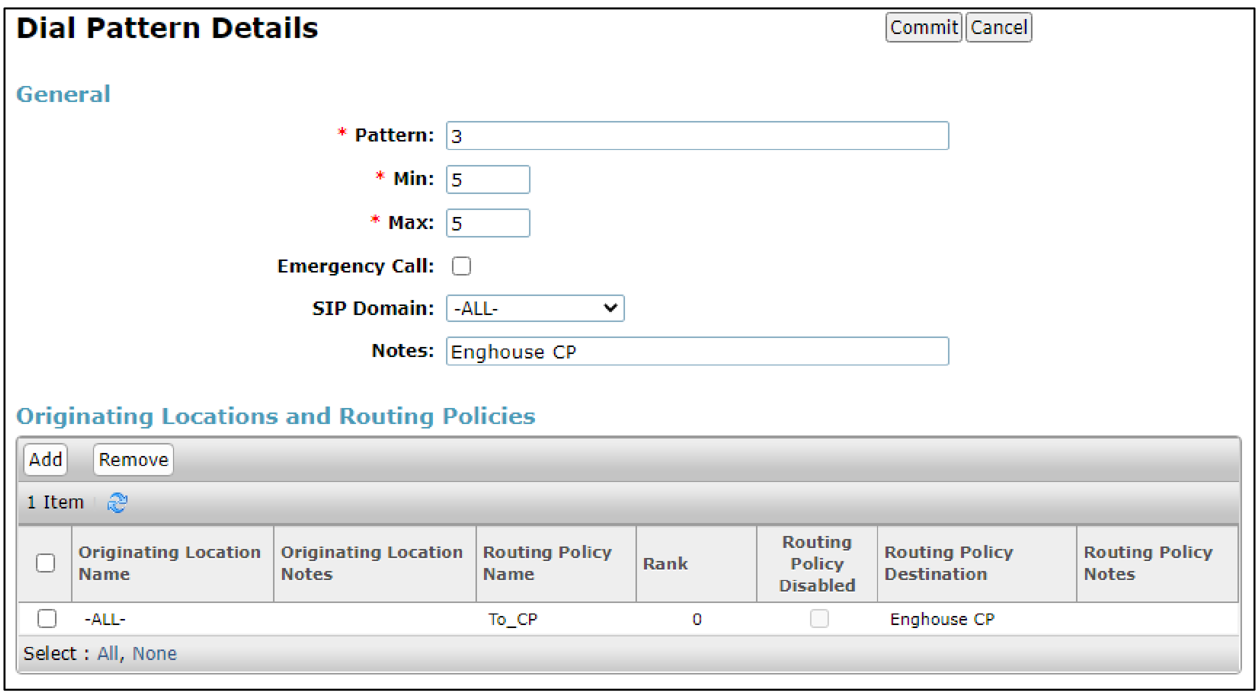
In order to route calls to the Communications Portal a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.



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

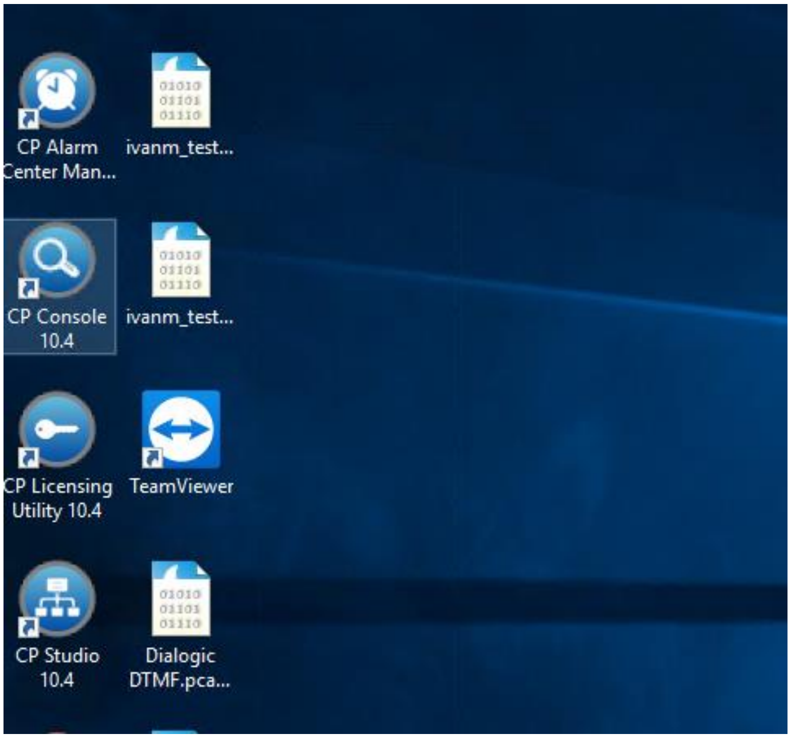


Select a preconfigured **Originating Location** and select the **Routing Polices** created in previous **Section 6.2** (not shown). The configuration below shows calls to **3xxxx** were routed to Communications Portal. Click on **Commit** as shown below to save configuration.

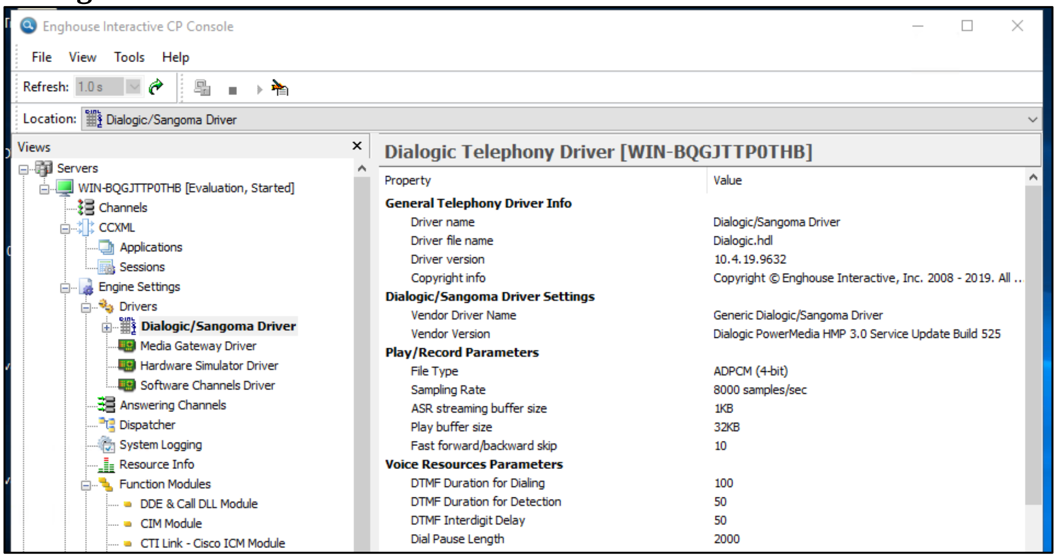


7. Configuration of Enghouse Interactive Communications Portal 10.4

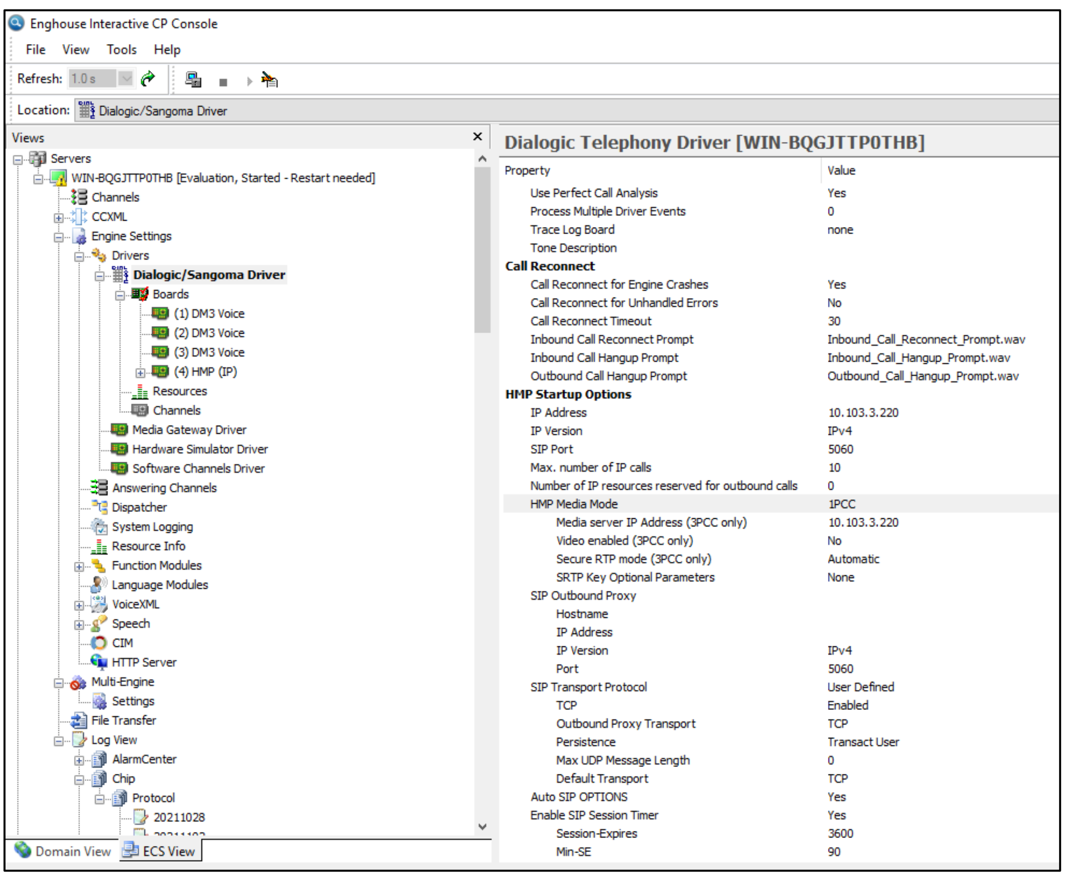
The Telephony module of Communications Portal which provides the connection to Session Manager is provided by a Dialogic Boards Driver. This driver completely caters for the telephony module of this solution. To configure the Dialogic Boards Driver, open the **CP Console 10.4** by double clicking on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



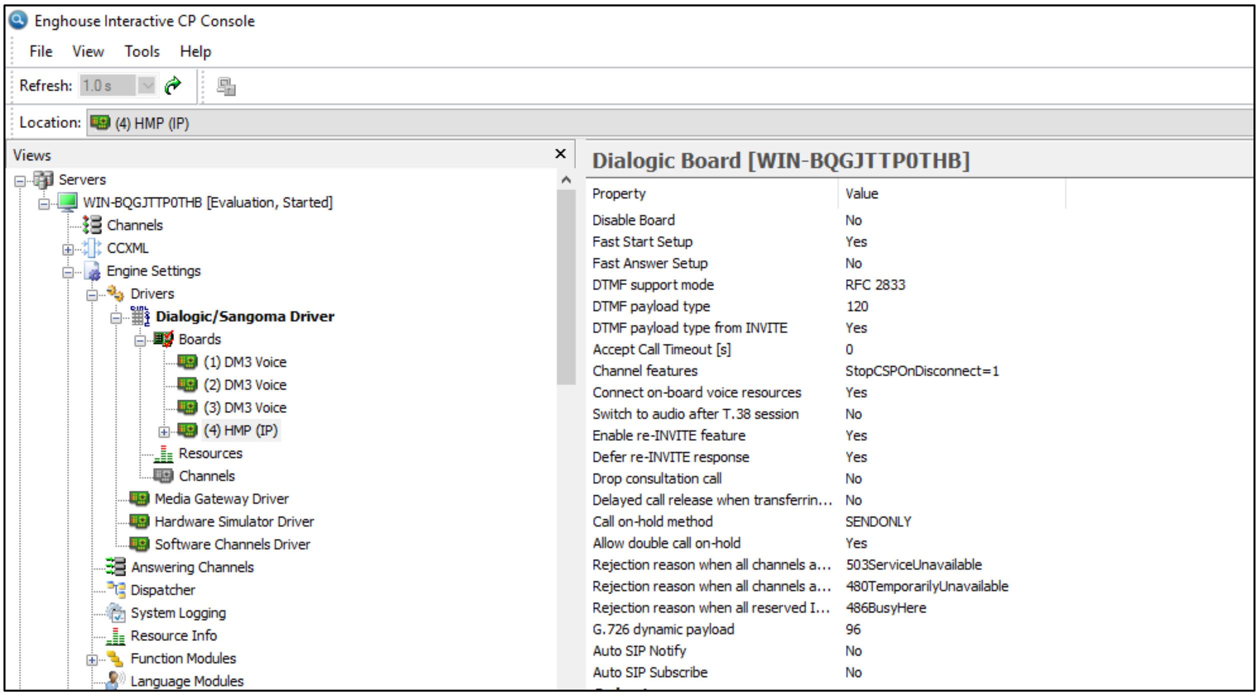
In the left window, navigate to **Servers→[Server Name]→Engine Settings → Drivers → Dialogic/Sangoma Driver**.



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 **TCP** is **enabled**, also note the **Default Transport** is set to **TCP**. And in **HMP Media Mode** select **1PCC.**



In the left window, navigate to **Servers→[Server Name]→Engine Settings → Drivers → Dialogic/Sangoma Driver → Boards → HMP (IP).** Edit **DTMF Payload type** with **120.**

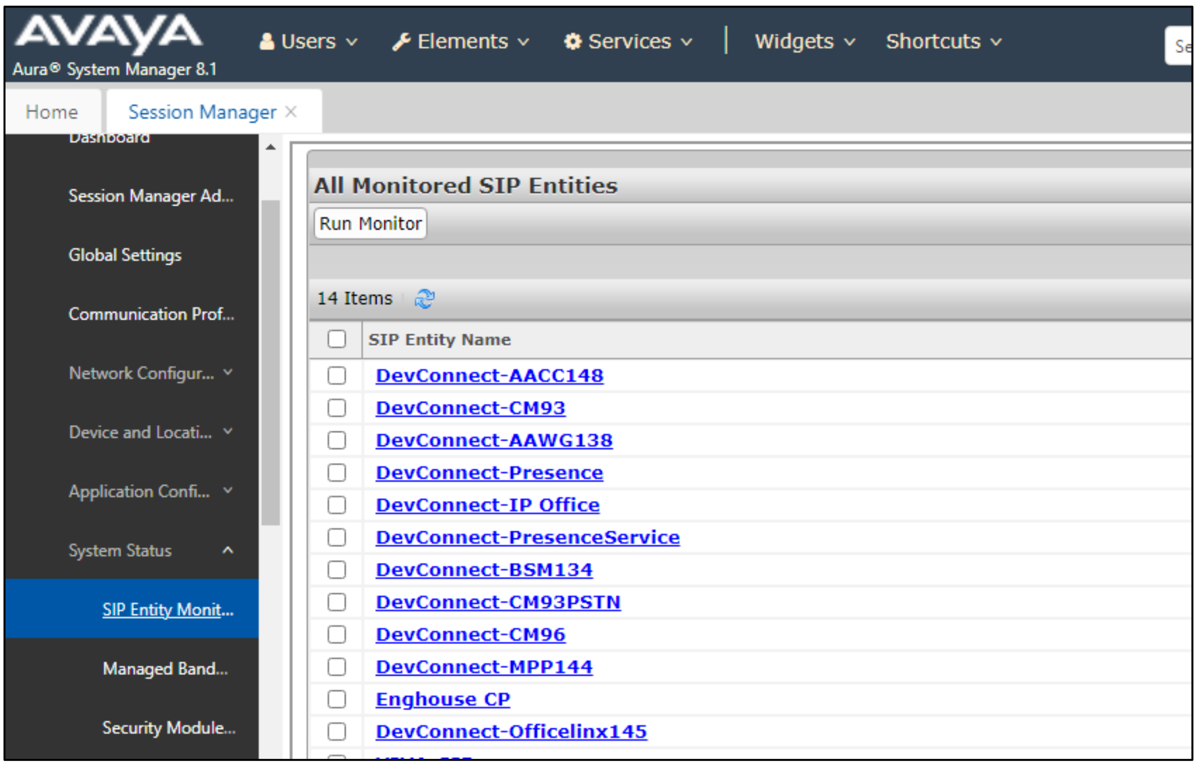


8. Verification Steps

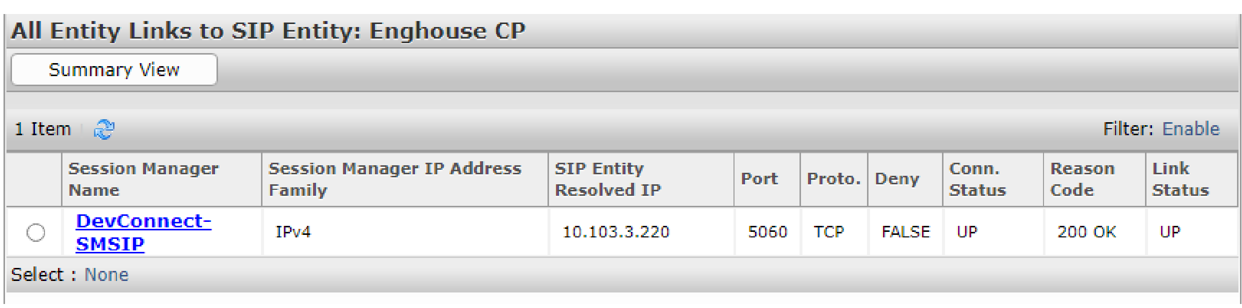
To verify a successful configuration of Enghouse Interactive Communications Portal and Session Manager/Communication Manager, a call is placed from a Communication Manager telephone to the Communications Portal with the caller getting answered successfully hearing clear and audible speech.

8.1. Verify Entity Link between Session Manager and Communications Portal

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



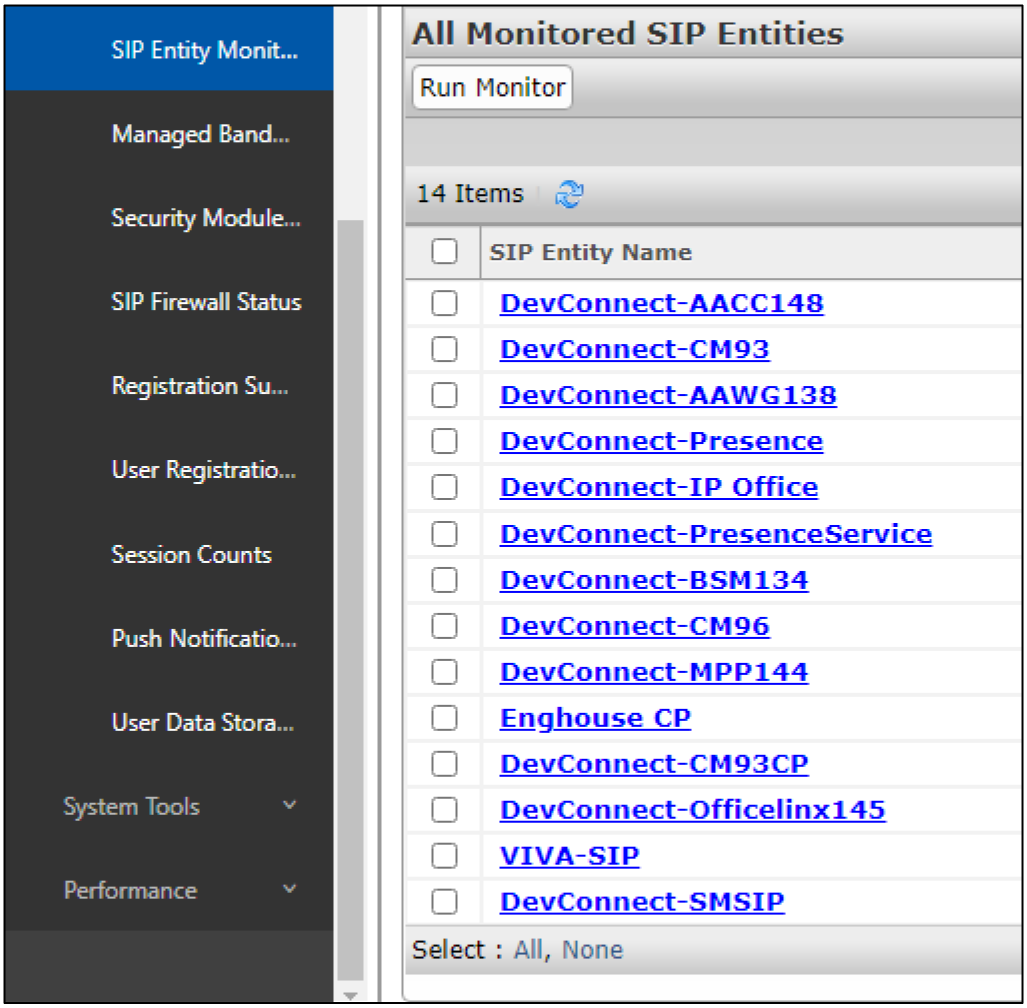
Verify **Conn. Status** is **UP.**



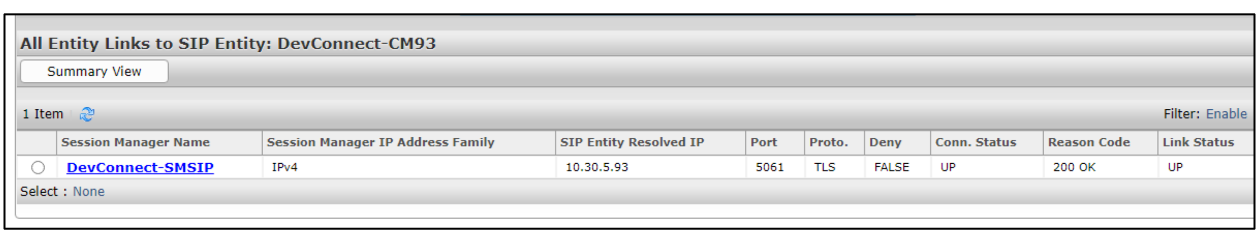
	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	DevConnect-SMSIP	IPv4	10.103.3.220	5060	TCP	FALSE	UP	200 OK	UP

8.2. Verify Entity Links between Session Manager and Communication Manager

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

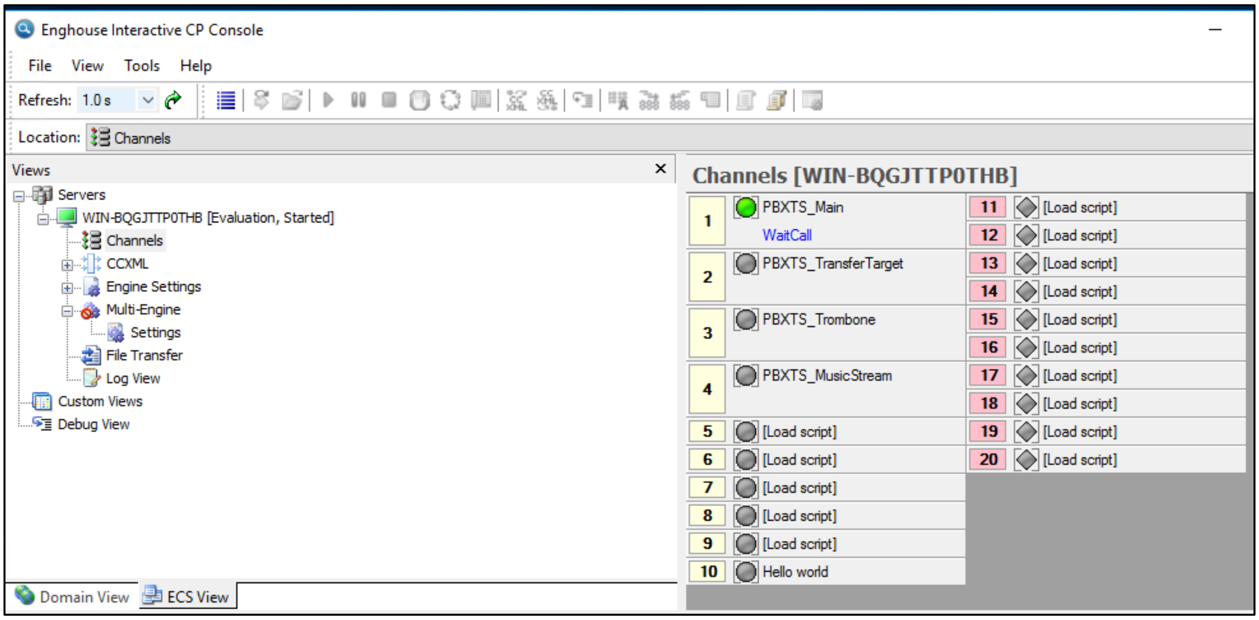


Verify **Conn. Status** is **UP.**

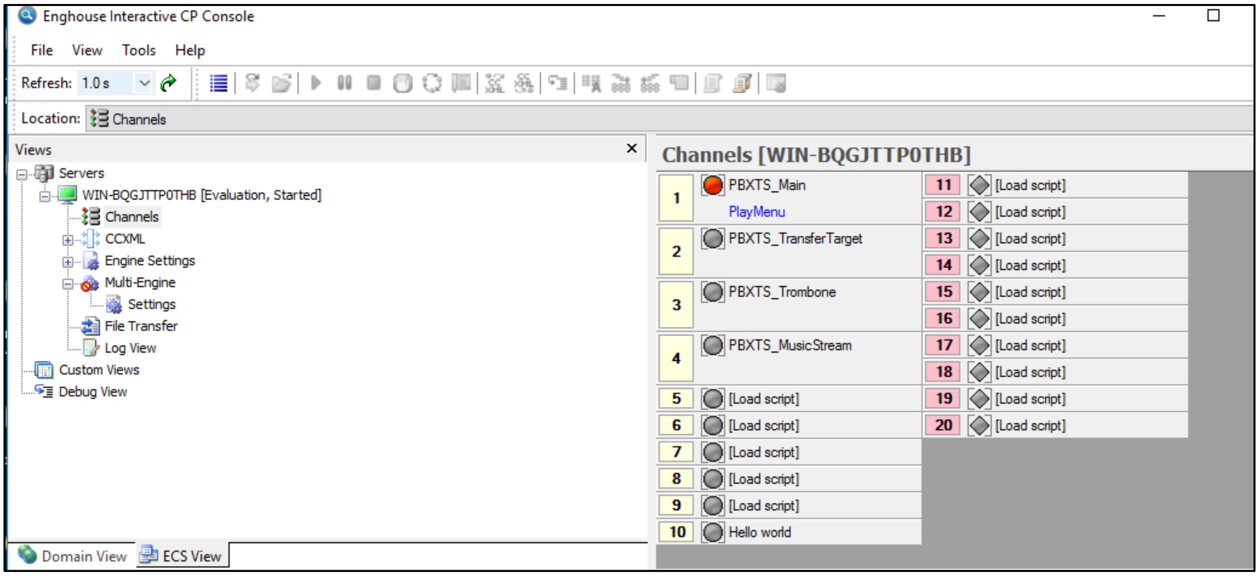


8.3. Verify Enghouse Interactive Communications Portal IVR scripts

On **CP Console 10.4,** Monitor the Channel 1 below has the script **PBXTS_Main** associated with it**,** this should also show as green.



Place a call from the Avaya Endpoints/PSTN to Enghouse Communications Portal with call number 3xxxx, ensure the call can be answered by CP. Monitor the Channel 1 below has the script **PBXTS_Main** associated with it**,** this should also show as Red.


Enghouse Interactive CP Console
File View Tools Help
Refresh: 1.0 s
Location: Channels
Views
Servers
WIN-BQGJTTP0THB [Evaluation, Started]
Channels
CCXML
Engine Settings
Multi-Engine
Settings
File Transfer
Log View
Custom Views
Debug View
Domain View
ECS View
Channels [WIN-BQGJTTP0THB]
1 PBXTS_Main
PlayMenu
2 PBXTS_TransferTarget
3 PBXTS_Trombone
4 PBXTS_MusicStream
5 [Load script]
6 [Load script]
7 [Load script]
8 [Load script]
9 [Load script]
10 Hello world
11 [Load script]
12 [Load script]
13 [Load script]
14 [Load script]
15 [Load script]
16 [Load script]
17 [Load script]
18 [Load script]
19 [Load script]
20 [Load script]

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 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2.**

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 8, Nov 2020

[2] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 8, Feb 2021

[3] *Administering the Avaya Aura® Web Gateway*, Release 3.8 Issue 2, July 2020

[4] *Administering Avaya Aura® Application Enablement Services,* Release 8.1.x, Issue 8, Feb 2021

[5] *Administering Avaya Aura® Device Services,* Release 8.0.2, Issue 4, June 2020

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

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