



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Enghouse Interactive Communications Portal 10.4 and CTI Media Gateway 8.5 with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2 using TCP/RTP - Issue 1.0

## Abstract

These Application Notes describe the configuration steps for Enghouse Interactive Communications Portal 10.4 and CTI Media Gateway 8.5 to successfully interoperate with Avaya Aura® Session Manager 8.1.3.2 and Avaya Aura® Communication Manager 8.1.3.2, using TCP/RTP. 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 Enghouse 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 dialed using SIP signaling. CP 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.

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. *Configuring Enghouse Interactive Communications Portal 10.4 using CTIC Media Gateway for SIP 8.2 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 calls to Enghouse Interactive Communications Portal.
- G.711A, G.711U codecs support and negotiation, with and without media 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 (support rtp-payload from 101 to 127), 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

The testing was successful. All test cases passed.

## 2.3. Support

Support for Enghouse products can be obtained as follows:

Technical support can be obtained for Enghouse Interactive as follows:

USA

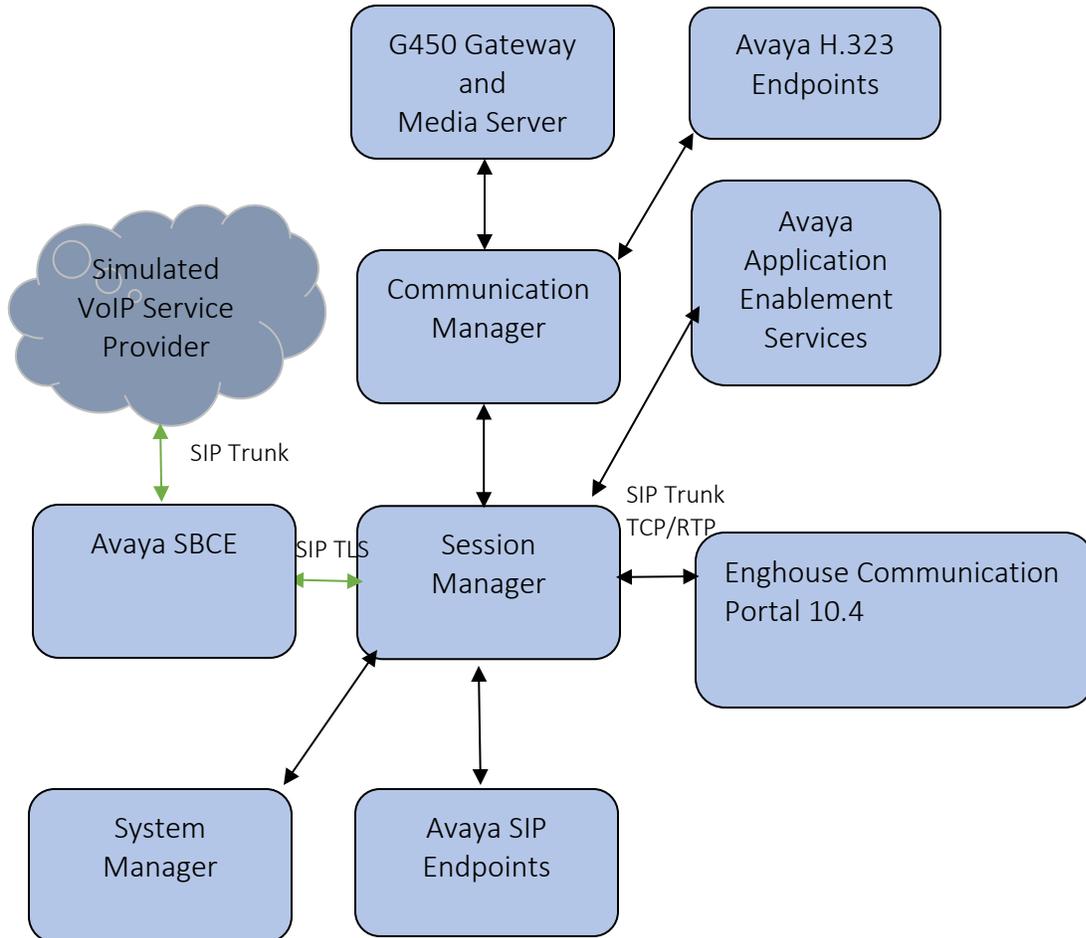
- Email: [scpsupport@enghouse.com](mailto:scpsupport@enghouse.com)
- Website: <http://enghouseinteractive.com/support.php>
- Phone: +1 800.788.9730 Self-Service
- Phone: +1 800.872.2272 Live-Service

EMEA

- Email: [envoxsupport@enghouse.com](mailto:envoxsupport@enghouse.com) / [supportenvox@syntellect.com](mailto: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 the Enghouse Communication Portal 10.4.



**Figure 1:** Test Configuration for Enghouse Communication Portal and the Avaya Platform.

## 4. Equipment and Software Validated

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

<b>Equipment/Software</b>	<b>Release/Version</b>
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 IX Workplace for Windows Softphone	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 Communication Portal CTI Media Gateway	10.4.19.9632 8.5

## 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
Private Networking? y                                              Uniform Dialing Plan? y
Processor and System MSP? y                                        Usage Allocation Enhancements? y
Processor Ethernet? y                                             Wideband Switching? y
Remote Office? y                                                  Wireless? n
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 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? 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: 2      Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 2048      IP Audio Hairpinning? y
      UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
      Call Control PHB Value: 46
      Audio PHB Value: 46
      Video PHB Value: 26
802.1P/Q PARAMETERS
      Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
      H.323 Link Bounce Recovery? y
      Idle Traffic Interval (sec): 20
      Keep-Alive Interval (sec): 5
      Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk to 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          n           2          20
3: G.729            n           2          20
4: OPUS-WB20K          n           1          20
5: G.711A           n           2          20
6:
7:

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

### 5.4. Configure Signaling Group and Trunk Group.

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager This signaling group and trunk group is used for internal calls between Avaya Endpoints and used for calls to and from 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 **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node**

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

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

```

change signaling-group 1                                     Page 1 of 3
                                SIGNALING GROUP

Group Number: 1                      Group Type: sip
  IMS Enabled? n                    Transport Method: tls
    Q-SIP? n
    IP Video? y                      Priority Video? y          Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n Peer Server: SM                      Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y
Alert Incoming SIP Crisis Calls? n
  Near-end Node Name: procr                      Far-end Node Name: smsip92
Near-end Listen Port: 5061                      Far-end Listen Port: 5061
                                                Far-end Network Region: 1

Far-end Domain: devconnect.com

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

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

```
change trunk-group 1                                     Page 1 of 4
                                     TRUNK GROUP
Group Number: 1                                     Group Type: sip                                     CDR Reports: y
Group Name: InternalCalls                             COR: 1                                     TN: 1                                     TAC: #01
Direction: two-way                                   Outgoing Display? n
Dial Access? n                                       Night Service:
Queue Length: 0
Service Type: tie                                     Auth Code? n
                                                Member Assignment Method: auto
                                                Signaling Group: 1
                                                Number of Members: 50
```

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
                                                    UII 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 follows.

```
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”, CM 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
```

			DIAL PLAN ANALYSIS TABLE			Page 1 of 12		
			Location: all			Percent Full: 2		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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 **3xxxx** 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
```

			UNIFORM DIAL PLAN TABLE			Page 1 of 2		
						Percent Full: 0		
Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node 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
```

AAR DIGIT ANALYSIS TABLE							Page 1 of 2
Location: all							Percent Full: 2
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI 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
```

Pattern Number: 1											Page 1 of 4
Pattern Name: DevC-Int											
SCCAN? n	Secure SIP? n	Used for SIP stations? n									
Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ QSIG	IXC	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 Dgts	Numbering Format	LAR	
0	1 2 M 4 W		Request						lev0-pvt	none	
1:	y y y y y	n		rest						none	
2:	y y y y y	n		rest						none	
3:	y y y y y	n		rest						none	
4:	y y y y y	n		rest						none	
5:	y y y y y	n		rest						none	
6:	y y y y y	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 Pattern for Enghouse Communication Portal.
- Configure Session Manager Application

### 6.1. Configure SIP Entities

#### 6.1.1. Configure SIP Entity for Enghouse Communication Portal

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

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

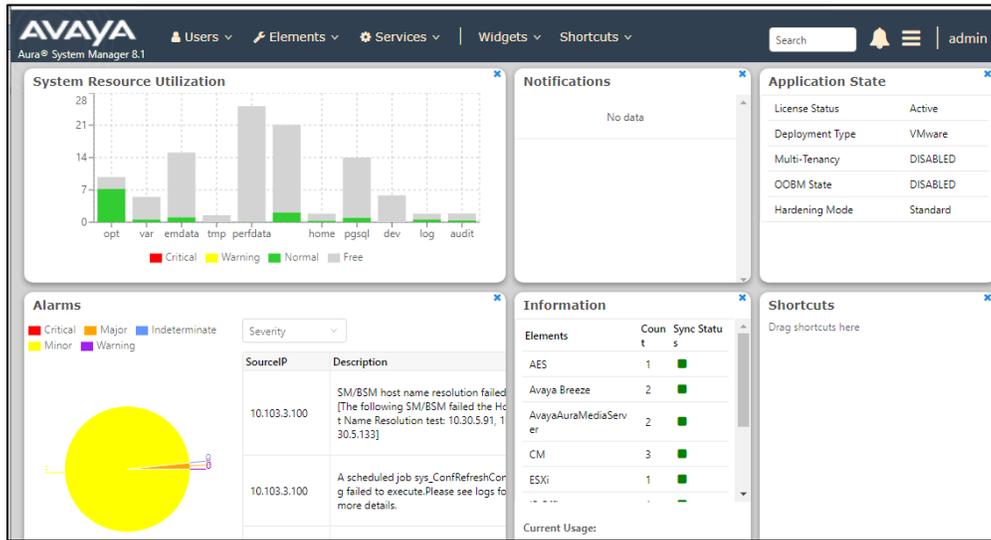
User ID:

Password:

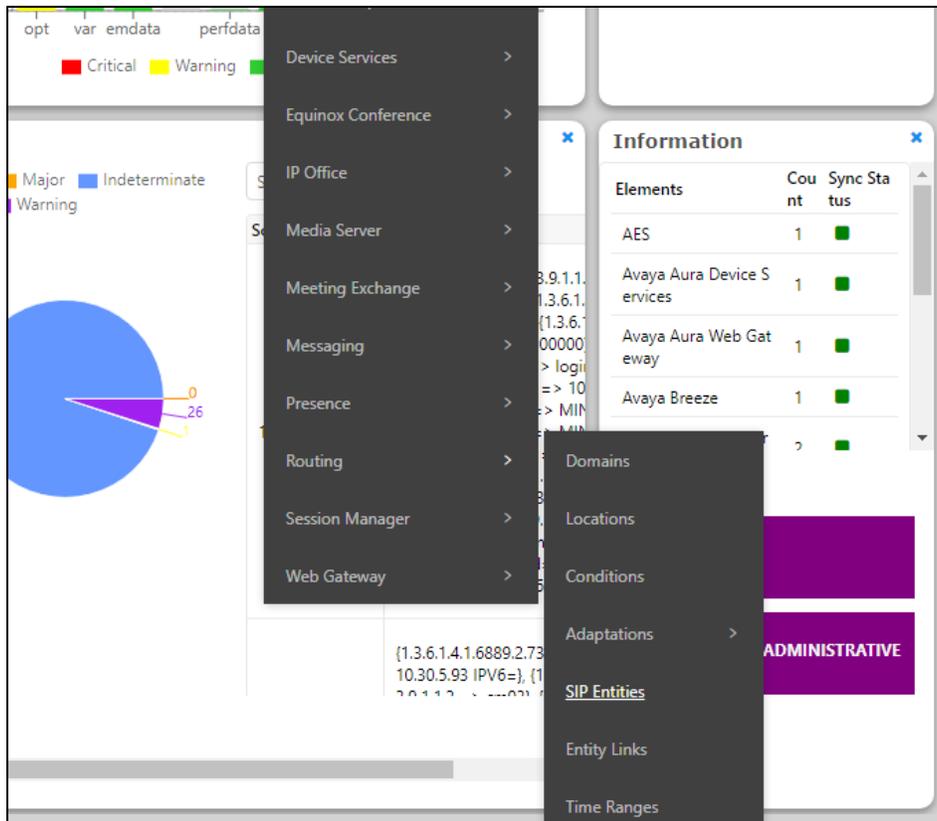
[Change Password](#)

**Supported Browsers:** Internet Explorer 11.x or Firefox 65.0, 66.0 and 67.0.

Once logged in, the following screen is displayed.



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



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

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top header includes the Avaya logo, user information, and navigation menus for Users, Elements, Services, Widgets, and Shortcuts. A search bar is located on the right. The left sidebar shows a navigation menu with options like Domains, Locations, Conditions, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, and Regular Expressions. The main content area is titled "SIP Entities" and features a toolbar with buttons for New, Edit, Delete, Duplicate, and More Actions. Below the toolbar, it indicates "17 Items" and displays a table with columns for Name, FQDN or IP Address, Type, and Notes.

<input type="checkbox"/>	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	<a href="#">BTCluster</a>	btcluster.avaya.com	Endpoint Concentrator	
<input type="checkbox"/>	<a href="#">DevConnect-AACC148</a>	10.30.5.148	SIP Trunk	
<input type="checkbox"/>	<a href="#">DevConnect-AAWG138</a>	10.30.5.138	SIP Trunk	
<input type="checkbox"/>	<a href="#">DevConnect-BSM134</a>	10.30.5.134	Session Manager	
<input type="checkbox"/>	<a href="#">DevConnect-CM93</a>	10.30.5.93	CM	
<input type="checkbox"/>	<a href="#">DevConnect-CM93PSTN</a>	10.30.5.93	SIP Trunk	
<input type="checkbox"/>	<a href="#">DevConnect-CM96</a>	cm96.hcm.com	CM	
<input type="checkbox"/>	<a href="#">DevConnect-IP Office</a>	10.128.226.178	SIP Trunk	
<input type="checkbox"/>	<a href="#">DevConnect-MPP144</a>	10.30.5.144	Voice Portal	
<input type="checkbox"/>	<a href="#">DevConnect-Officelinx145</a>	10.30.5.145	Other	
<input type="checkbox"/>	<a href="#">DevConnect-Presence</a>	10.30.5.135	Avaya Breeze	
<input type="checkbox"/>	<a href="#">DevConnect-PresenceService</a>	10.30.5.135	Presence Services	
<input type="checkbox"/>	<a href="#">DevConnect-SBC140</a>	10.30.5.140	SIP Trunk	

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 “Enghouse CP”
- **FQDN or IP Address:** The internal SIP IP address of Enghouse CP.
- **Type:** “SIP Trunk”
- **Notes:** Any desired notes.
- **Location:** Select the applicable location.
- **Time Zone:** Select the applicable time zone

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.

The screenshot shows the Avaya System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar is expanded to 'Routing', and 'SIP Entities' is selected. The main content area is titled 'SIP Entity Details' and contains the following configuration fields:

- Name:** Enghouse CP
- FQDN or IP Address:** 10.103.3.220
- Type:** SIP Trunk
- Notes:** (empty text area)
- Adaptation:** (empty dropdown)
- Location:** SaiGon
- Time Zone:** Asia/Ho\_Chi\_Minh
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty text area)
- Securable:**
- Call Detail Recording:** egress

The 'Commit' button is located in the top right corner of the form area.

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 Communication Portal entity name from this section, in this case “**Enghouse CP**”
- **Port:** “5060”
- **Connection Policy:** “trusted”

The screenshot shows the 'Entity Links' configuration page. At the top, there is a checkbox for 'Override Port & Transport with DNS SRV:'. Below this are 'Add' and 'Remove' buttons. A table below shows '1 Item' with the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Connection Policy. The table contains one row with the following values: Name: \*DevConnect-SMSIP\_Engh, SIP Entity 1: DevConnect-SMSIP, Protocol: TCP, Port: \*5060, SIP Entity 2: Enghouse CP, Port: \*5060, and Connection Policy: trusted. At the bottom, there is a 'Select' dropdown menu with options 'All' and 'None'.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	*DevConnect-SMSIP_Engh	DevConnect-SMSIP	TCP	*5060	Enghouse CP	*5060	trusted

## 6.1.2. Configure SIP Entity for Communication Manager

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

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

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

---

### SIP Entity Details

Commit Cancel

#### General

\* **Name:**

\* **FQDN or IP Address:**

**Type:**

**Notes:**

**Adaptation:**

**Location:**

**Time Zone:**

\* **SIP Timer B/F (in seconds):**

**Minimum TLS Version:**

**Credential name:**

**Securable:**

**Call Detail Recording:**

#### Loop Detection

**Loop Detection Mode:**

**Loop Count Threshold:**

**Loop Detection Interval (in msec):**

---

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

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

**Entity Links**  
Override Port & Transport with DNS SRV:

Add Remove

1 Item Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* DevConnect-SMSIP_DevC	DevConnect-SMSIP	TLS	* 5061	DevConnect-CM93	* 5061	trusted	<input type="checkbox"/>

Select : All, None

## 6.2. Configure Routing Policy for Enghouse Communication Portal

This section to add a new routing policy for routing calls to Communication 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 Communication Portal SIP entity name from **Section 6.1**

**Routing Policy Details** Commit Cancel

**General**

\* Name: To\_CP

Disabled:

\* Retries: 0

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
Enghouse CP	10.103.3.220	SIP Trunk	

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

### 6.3. Configure Dial Pattern for Enghouse Communication 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 screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar is expanded to 'Routing', and 'Dial Patterns' is selected. The main content area is titled 'Dial Patterns' and features a toolbar with 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions' buttons. Below the toolbar, it indicates '12 Items' and shows a table of dial patterns.

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	Emergency
<input type="checkbox"/>	<a href="#">0</a>	10	10	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">+1</a>	11	12	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">113</a>	3	3	<input checked="" type="checkbox"/>	Police
<input type="checkbox"/>	<a href="#">114</a>	3	3	<input checked="" type="checkbox"/>	Fire Truck
<input type="checkbox"/>	<a href="#">115</a>	3	3	<input checked="" type="checkbox"/>	Ambulance
<input type="checkbox"/>	<a href="#">3</a>	5	5	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">5</a>	4	4	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">6</a>	5	5	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">7</a>	5	5	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">8</a>	5	5	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">89999</a>	5	5	<input type="checkbox"/>	
<input type="checkbox"/>	<a href="#">9</a>	11	14	<input type="checkbox"/>	

Select : All, None

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**.

### Dial Pattern Details Commit Cancel

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations and Routing Policies**

Add Remove

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 Communication Portal. Click on **Commit** as shown below to save configuration.

### Dial Pattern Details Commit Cancel

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations and Routing Policies**

Add Remove

1 Item

	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		To_CP	0	<input type="checkbox"/>	Enghouse CP	

Select : All, None

## 7. Configuration of Enghouse Interactive Communications Portal

This section describes the steps required to configure Enghouse Interactive Communications Portal 10.4 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.

### 7.1. 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.

### 7.2. 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>

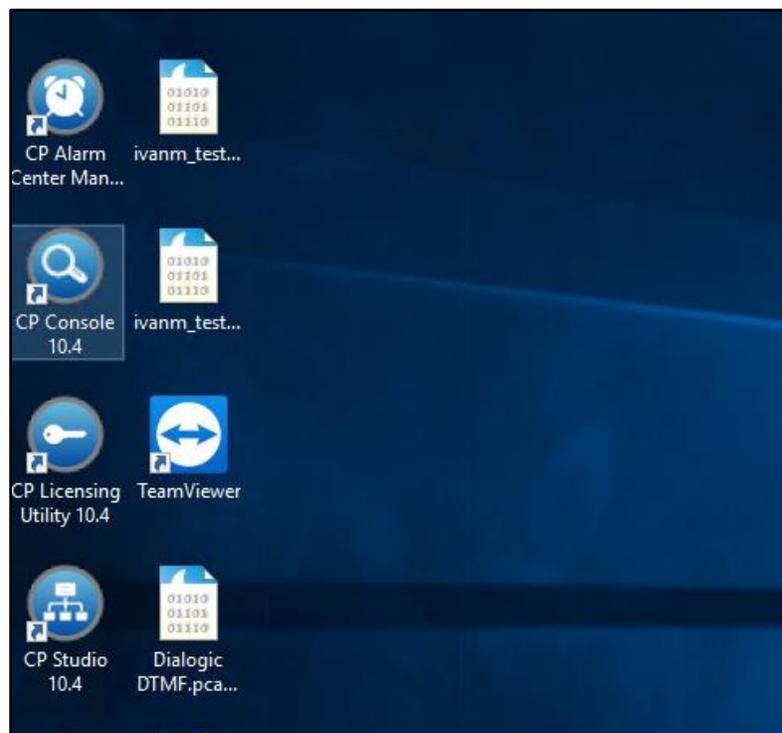
### 7.3. Set the SIP transfer type parameter

By default, the SIP transfer type is set to Refer. It should be changed to re-Invite with following steps.

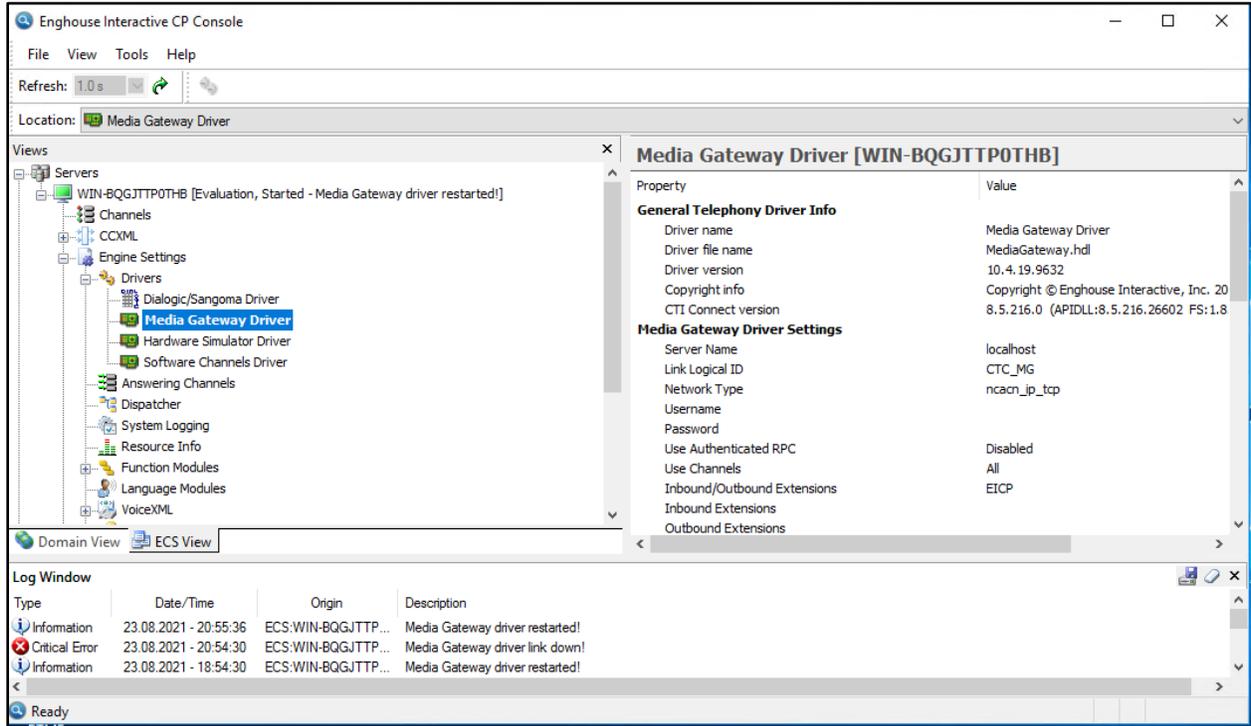
- In the <Media Gateway install folder>\conf\autoload\_configs folder, edit the csinterface.conf.xml file.
- Change the parameter `<param name="sip_transfer_type" value="refer"/>` to `<param name="sip_transfer_type" value="reinvite"/>`.
- Save the changes.

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 10.4** by double clicking on the shortcut as shown below.



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



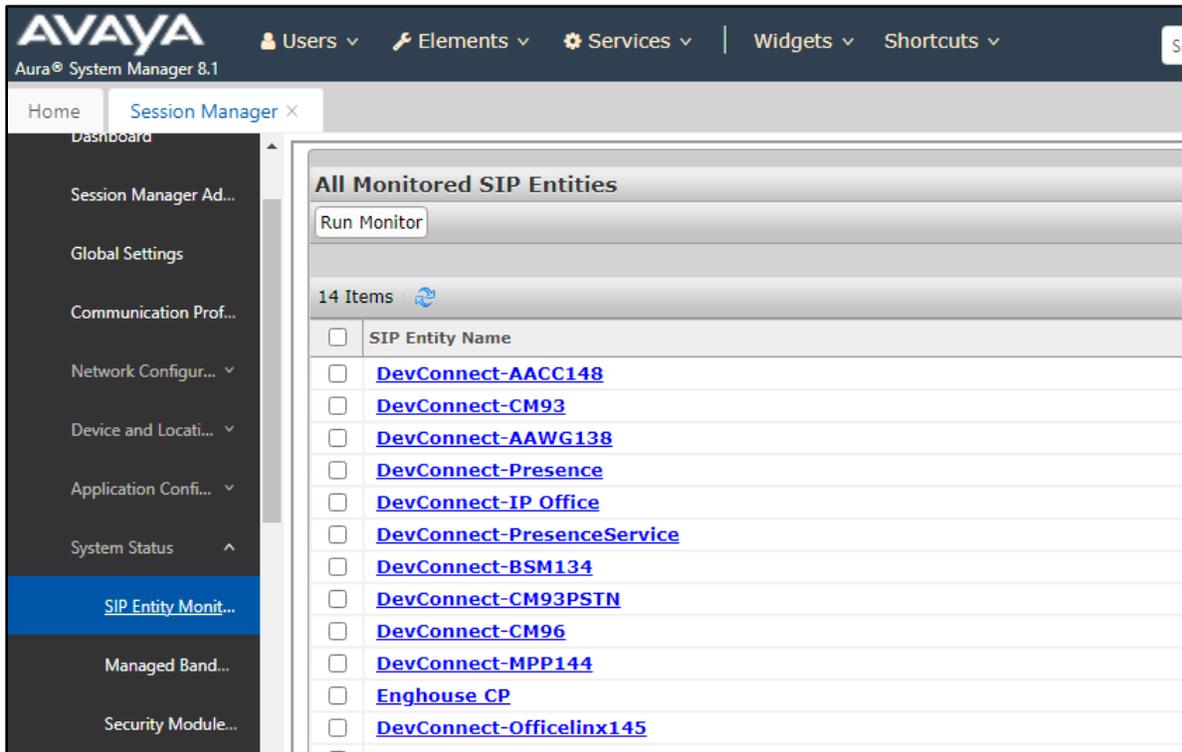
Please note that configuration of Communications Portal with regards to the setup of the IVR is outside the scope of these Application Notes, for more information on this setup please refer to **Section 10** of these Application Notes.

## 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 Links

To verify SIP connectivity, 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**.

All Entity Links to SIP Entity: Enghouse CP									
Summary View									
1 Item <span>Filter: Enable</span>									
	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	<a href="#">DevConnect-SMSIP</a>	IPv4	10.103.3.220	5060	TCP	FALSE	UP	200 OK	UP

Select : None

Select the **DevConnect-CM93** Entity and verify **Conn. Status** is **UP**.

All Entity Links to SIP Entity: DevConnect-CM93									
Summary View									
1 Item <span>Filter: Enable</span>									
Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status	
<input type="radio"/> <a href="#">DevConnect-SMSIP</a>	IPv4	10.30.5.93	5061	TLS	FALSE	UP	200 OK	UP	
Select : None									

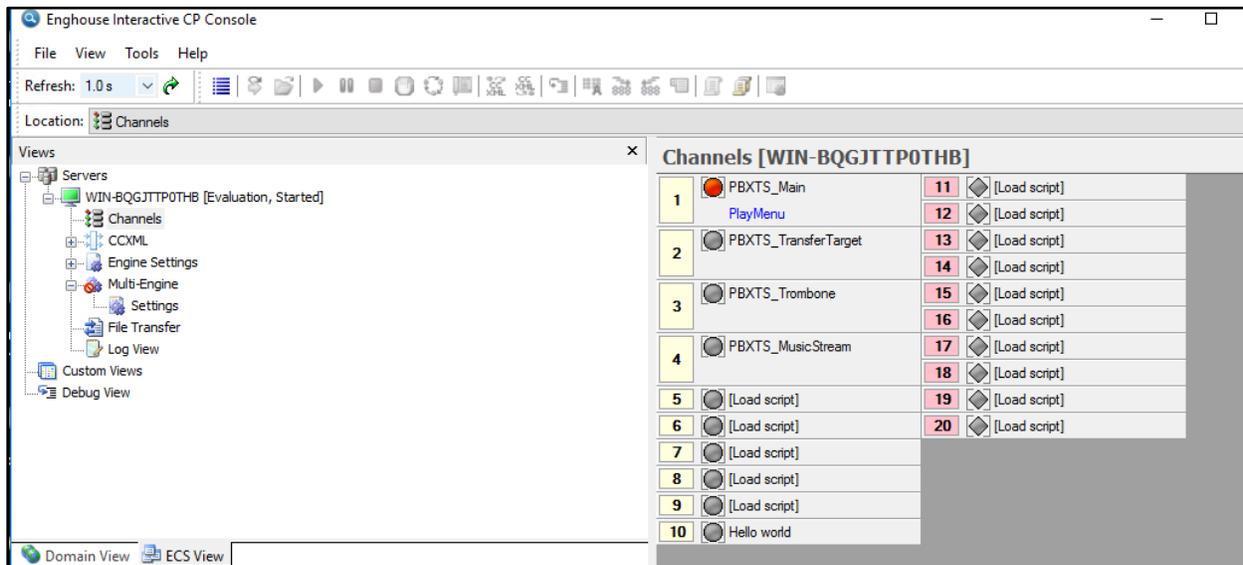
## 8.2. Verify Enghouse Interactive Communications Portal IVR scripts

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

The screenshot shows the Enghouse Interactive CP Console interface. The main window displays a list of channels for the server WIN-BQGJTTP0THB. Channel 1 is highlighted in yellow and has a green status icon, indicating it is UP. The script PBXTS\_Main is associated with it. Other channels are shown with grey status icons and 'Load script' buttons.

Channel ID	Script Name	Status	Action
1	PBXTS_Main	UP (Green)	[Load script]
2	PBXTS_TransferTarget	Down (Grey)	[Load script]
3	PBXTS_Trombone	Down (Grey)	[Load script]
4	PBXTS_MusicStream	Down (Grey)	[Load script]
5	[Load script]	Down (Grey)	[Load script]
6	[Load script]	Down (Grey)	[Load script]
7	[Load script]	Down (Grey)	[Load script]
8	[Load script]	Down (Grey)	[Load script]
9	[Load script]	Down (Grey)	[Load script]
10	Hello world	Down (Grey)	[Load script]

Place a call from the Avaya Endpoints/PSTN to Enghouse Communication 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.



## 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](http://www.enghouseinteractive.com)

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