Avaya Solution & Interoperability Test Lab

Application Notes for VHT Callback 8.0 with Avaya Aura® Application Enablement Services 6.3 and Avaya Aura® Session Manager 6.3 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for VHT Callback 8.0 to interoperate with Avaya Aura® Communication Manager 6.3, Avaya Aura® Application Enablement Services 6.3, and Avaya Aura® Session Manager 6.3. VHT Callback is a contact center solution that calculates expected wait time and maintains caller position in virtual queue.

The integration used the Avaya Telephony Services Application Programming Interface from Avaya Aura® Application Enablement Services, and the SIP trunks interface from Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for VHT Callback 8.0 to interoperate with Avaya Aura® Communication Manager 6.3, Avaya Aura® Application Enablement Services 6.3, and Avaya Aura® Session Manager 6.3. VHT Callback is a contact center solution that calculates expected wait time and maintains caller position in virtual queue.

VHT Callback can call users back and connect to agents when the caller's turn comes up. The integration used the Avaya Telephony Services Application Programming Interface (TSAPI) from Avaya Aura® Application Enablement Services, and the SIP trunks interface from Avaya Aura® Session Manager.

The TSAPI interface is used by VHT Callback to monitor VDNs and to query status of ACD queues. The information obtained from the TSAPI event reports is used to calculate the expected wait time. All incoming ACD calls are routed by VHT Callback using the TSAPI adjunct routing capabilities. When the expected wait time for an ACD queue reaches a pre-defined threshold, then VHT Callback specifies for the call to route over Avaya Aura® Session Manager SIP trunks to the Interactive Voice Gateway (IVG) component of VHT Callback. The IVG will play the expected wait time announcement and provide caller with options to continue to wait in queue or to be called back.

Callers that decide to wait in queue will be transferred by VHT Callback to a Hold VDN on Communication Manager, which queues the call to the ACD skill group.

Callers that decide to be called back will be prompted for callback number and time, and VHT Callback will track the caller position in the virtual queue. When it is almost time for the caller to be serviced from the virtual queue, VHT Callback will place an outbound callback call via IVG and Avaya Aura® Session Manager SIP trunks to the PSTN destination, with call progress tones and tone detection handled by IVG. When the callback call is connected and accepted by the PSTN destination, VHT Callback then uses SIP REFER to transfer the callback call to a Callback VDN on Communication Manager, which queues the call to the ACD skill group with priority.

2. General Test Approach and Test Results

The feature test cases were performed both automatically and manually. Upon start of the Callback application, the application automatically sends TSAPI queries for ACD skill group status, route registers for the Entry VDN, and requests monitoring of VDNs. For the manual part of the testing, incoming calls were made to the monitored VDNs to enable adjunct route and event reports to be sent to Callback. Manual call controls from the customer and agent telephones were exercised to verify remaining event reports, and the proper scheduling and delivering of callback calls.

The UUI data test cases were performed by using vector variables to assign UUI data to inbound calls, and verified by reviewing the TSAPI log and the SIP REFER message associated with inbound transferred and outbound callback calls.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to the Callback server and to the IVG component.

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

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on Callback:

- Use of TSAPI query service to query status on skill group.
- Use of TSAPI event report service to monitor VDNs.
- Use of TSAPI routing service to route incoming calls.
- Use of SIP messages to answer and transfer inbound calls, and to initiate and transfer outbound callback calls.
- Proper handling of call scenarios involving G.711, DTMF, REFER, expected wait time under and over the threshold, transfer of inbound calls with received UUI data, initiation and transfer of outbound callback calls with priority and saved UUI data, and unsuccessful callback calls.

The serviceability testing focused on verifying the ability of Callback to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to the Callback server and to the IVG component.

2.2. Test Results

All test cases were executed and verified. The following were observations on Callback from the compliance testing.

- The calling party number was not available on the outbound callback calls.
- Upon receipt of the BYE message from Session Manager as part of a transferred call, IVG sent a 480 Temporarily Unavailable without any noticeable adverse impact.

2.3. Support

Technical support on Callback can be obtained through the following:

- **Phone:** (866) 670-2223
- **Email:** support@virtualhold.com

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. The Callback configuration consisted of the Callback server, and an IVG that connected via SIP trunks with Session Manager.

The configuration of Session Manager was performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, Session Manager, Application Enablement Services, and of contact center devices is not the focus of these Application Notes and will not be described.

The pre-existing contact center devices used in the compliance testing are shown in the table below. Additional vectors and VDNs need to be created, as described in **Section 5.4**. The applicable domain for the network is "dr220.com". A five digit Uniform Dial Plan was used to facilitate routing of calls with Callback. In the compliance testing, calls to 32xxx were routed to the IVG component of Callback.

Device Type	Extension
Skill Group Number	81
Skill Group Extension	65081
Agent Stations	65001, 65002

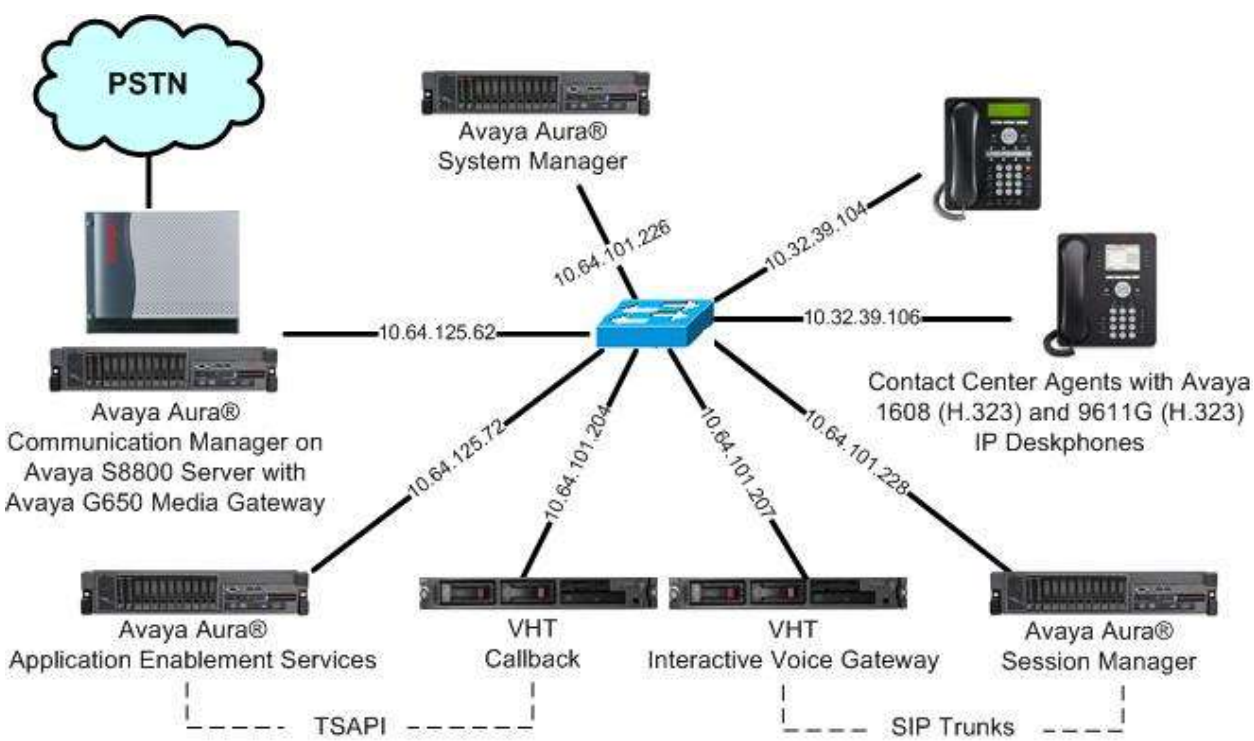


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager on Avaya S8800 Server with Avaya G650 Media Gateway	6.3.9 (R016x.03.0.124.0-21971)
Avaya Aura® Application Enablement Services	6.3.3 SP1 (6.3.3.1.10-0)
Avaya Aura® Session Manager	6.3.11.0.631103
Avaya Aura® System Manager	6.3.11.8.2933
Avaya 1616 IP Deskphone (H.323)	1.350B
Avaya 9611G IP Deskphone (H.323)	6.4.0.14
VHT Callback on Microsoft Windows Server 2008 R2 Enterprise • Microsoft SQL Server 2008 R2 • Avaya TSAPI Windows Client (csta32.dll)	8.0.6.1075 SP 1 10.50.1600.1 5.2.1.474
VHT IVG on CentOS	1.1.0-20150320145933 6.5

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer CTI link
- Administer system parameters features
- Administer vectors and VDNs
- Administer SIP signaling group
- Administer SIP trunk group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer uniform dial plan
- Administer AAR analysis

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

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for **Maximum Administered SIP Trunks**.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

Navigate to **Page 3**, and verify that the **Computer Telephony Adjunct Links** customer option is set to "y".

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

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                      DCS (Basic)? y
          ASAI Link Core Capabilities? n                DCS Call Coverage? y
          ASAI Link Plus Capabilities? n               DCS with Rerouting? y
```

Navigate to **Page 6**, and verify that the **Vectoring (Basic)** customer option is set to "y".

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 6.0

                                ACD? y                        Reason Codes? y
                       BCMS (Basic)? y           Service Level Maximizer? n
         BCMS/VuStats Service Level? y         Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? y   Service Observing (Remote/By FAC)? y
                  Business Advocate? n          Service Observing (VDNs)? y
                    Call Work Codes? y                         Timed ACW? y
      DTMF Feedback Signals For VRU? y                 Vectoring (Basic)? y
                   Dynamic Advocate? n             Vectoring (Prompting)? y
       Expert Agent Selection (EAS)? y         Vectoring (G3V4 Enhanced)? y
                            EAS-PHD? y          Vectoring (3.0 Enhanced)? y
```

5.2. Administer CTI Link

Add a CTI link using the "add cti-link n" command, where "n" is an available CTI link number. Enter an available extension number in the **Extension** field. Note that the CTI link number and extension number may vary. Enter "ADJ-IP" in the **Type** field, and a descriptive name in the **Name** field. Default values may be used in the remaining fields.

```
add cti-link 2                                                   Page   1 of   3
                                    CTI LINK
 CTI Link: 2
Extension: 60100
     Type: ADJ-IP
                                                                     COR: 1

     Name: AES CTI Link
```

5.3. Administer System Parameters Features

Use the “change system-parameters features” command to enable **Create Universal Call ID (UCID)**, which is located on **Page 5**. For **UCID Network Node ID**, enter an available node ID.

```
change system-parameters features                               Page   5 of  20
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint:               Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                     Switch Name:
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT: station
                EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
      Delay Sending RELease (seconds): 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n
              Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? y    UCID Network Node ID: 27
```

Navigate to **Page 13**, and enable **Send UCID to ASAI**. This parameter allows for the universal call ID to be sent to Callback.

```
change system-parameters features                               Page  13 of  20
                        FEATURE-RELATED SYSTEM PARAMETERS
 CALL CENTER MISCELLANEOUS
           Callr-info Display Timer (sec): 10
                         Clear Callr-info: next-call
        Allow Ringer-off with Auto-Answer? n

    Reporting for PC Non-Predictive Calls? n

            Agent/Caller Disconnect Tones? n
          Interruptible Aux Notification Timer (sec): 3
             Zip Tone Burst for Callmaster Endpoints: double

  ASAI
            Copy ASAI UUI During Conference/Transfer? y
        Call Classification After Answer Supervision? y
                                   Send UCID to ASAI? y
          For ASAI Send DTMF Tone to Call Originator? n
  Send Connect Event to ASAI For Announcement Answer? n
```

5.4. Administer Vectors and VDNs

Administer three sets of vectors and VDNs shown below for routing of calls to Callback. Note that the VDN extensions and vector numbers can vary.

VDN	Vector	Purpose
60901	901	Entry vector & VDN for adjunct route and failure coverage
60902	902	Hold vector & VDN for queuing inbound calls to skill at medium priority
60903	903	Callback vector & VDN for queuing outbound calls to skill at high priority

5.4.1. Entry Vector and VDN

Modify an available vector using the "change vector n" command, where "n" is an existing vector number. The vector will be used to provide adjunct route to the CTI link defined in **Section 5.2**.

Note that the vector **Number**, **Name**, **wait-time** and **route-to number** parameter settings may vary. The **route-to number** is used as the covering point to provide failure coverage in case of failures from the adjunct routing step. In the compliance testing, the covering point is the Hold VDN, which is administered in **Section 5.4.2**.

```
change vector 901                                              Page   1 of   6
                                  CALL VECTOR

    Number: 901             Name: VHT Entry
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 adjunct      routing link 2
02 wait-time    10  secs hearing music
03 route-to     number 60902            with cov n if unconditionally
04
```

Add a VDN using the "add vdn n" command, where "n" is an available extension number. Enter a descriptive **Name**, and the vector number from above for **Vector Number**. Retain the default values for all remaining fields.

```
add vdn 60901                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 60901
                                 Name*: VHT Entry
                           Destination: Vector Number        901
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

5.4.2. Hold Vector and VDN

Modify an available vector to queue incoming calls to the ACD skill group at medium priority. Note that the vector **Number**, **Name**, **queue-to skill** and **wait-time** parameter settings may vary, and that "81" is the existing skill group number from **Section 3**.

```
change vector 902                                              Page   1 of   6
                                  CALL VECTOR

    Number: 902              Name: VHT Hold
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing silence
02 queue-to     skill 81   pri m
03 wait-time    20  secs hearing ringback
04 goto step    3             if unconditionally
05
```

Add a VDN with an available extension as shown below. Enter a descriptive **Name**, and the vector number from above for **Vector Number**.

```
add vdn 60902                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 60902
                                 Name*: VHT Hold
                           Destination: Vector Number        902
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

5.4.3. Callback Vector and VDN

Modify an available vector to queue callback calls to the ACD skill group at high priority. Note that the vector **Number**, **Name**, **queue-to skill** and **wait-time** parameters may vary, and that "81" is the existing skill group number from **Section 3**.

```
change vector 903                                              Page   1 of   6
                                  CALL VECTOR

    Number: 903             Name: VHT Callback
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 queue-to     skill 81   pri h
02 wait-time    20  secs hearing ringback
03
```

Add a VDN with an available extension as shown below. Enter a descriptive name for **Name**, and the vector number from above for **Vector Number**.

```
add vdn 60903                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 60903
                                 Name*: VHT Callback
                           Destination: Vector Number        903
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

5.5. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "32". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tls"
- **Near-end Node Name:** An existing C-LAN node name.
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Callback.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Callback.
- **Far-end Domain:** The applicable domain name for the network.

```
add signaling-group 32                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 32             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: clan                  Far-end Node Name: DR-SMW-Sig
 Near-end Listen Port: 5032                Far-end Listen Port: 5032
                                        Far-end Network Region: 3

Far-end Domain: dr220.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? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "32". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Signaling Group:** The signaling group number from **Section 5.5**.
- **Number of Members:** The desired number of members, in this case "10".

```
add trunk-group 32                                             Page   1 of  21
                                TRUNK GROUP

Group Number: 54                   Group Type: sip           CDR Reports: y
  Group Name: VHG IVG                     COR: 1        TN: 1        TAC: 1032
   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: 32
                                                     Number of Members: 10
```

5.7. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.5**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Authoritative Domain:** The applicable domain for the network.
- **Name:** A descriptive name.
- **Intra-region IP-IP Direct Audio:** “yes”
- **Inter-region IP-IP Direct Audio:** “yes”
- **Codec Set:** An available codec set for integration with Callback.

```
change ip-network-region 3                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 3
Location:          Authoritative Domain: dr220.com
    Name: VHT IVG                 Stub Network Region: n
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: yes
      Codec Set: 3                Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                        IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with network region used by Avaya endpoints and by the trunk to the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and by the trunk to the PSTN.

```
change ip-network-region 3                                     Page   4 of  20

 Source Region: 3     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC  R  L    e
 1    3
 2
 3    3                                                                all
 4
 5
 6
 7
 8
```

5.8. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.7**. Update the audio codec types in the **Audio Codec** fields as necessary. G.711MU was the only codec covered in the compliance testing.

```
change ip-codec-set 3                                          Page   1 of   2

                          IP Codec Set

    Codec Set: 3

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

5.9. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Callback, in this case "32". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.6**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 32                                        Page   1 of   3
                    Pattern Number: 54  Pattern Name: VHT IVG
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 32   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                        none
```

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing of inbound calls to Callback at destination 32xxx, which will be returned by Callback as the adjunct route destination in the compliance testing. Note that other routing methods may be used.

Use the “change uniform-dialplan 0” command, and add an entry to specify the use of AAR for routing digits 32xxx, as shown below.

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

  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num

 32              5   0                  aar  n
```

5.11. Administer AAR Analysis

Use the “change aar analysis 0” command, and add an entry to specify how to route calls to 32xxx. In the example shown below, calls with digits 32xxx will be routed as an AAR call using route pattern “32” from **Section 5.9**.

```
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
    32                    5    5     32       aar          n
```

6. Configure Avaya Aura® Application Enablement Services

This section provides the procedures for configuring Application Enablement Services. The procedures include the following areas:

- Launch OAM interface
- Verify license
- Administer TSAPI link
- Administer TCP settings
- Restart service
- Obtain Tlink name
- Administer Callback user
- Verify security database

6.1. Launch OAM Interface

Access the OAM web-based interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Application Enablement Services server.

The **Please login here** screen is displayed. Log in using the appropriate credentials.

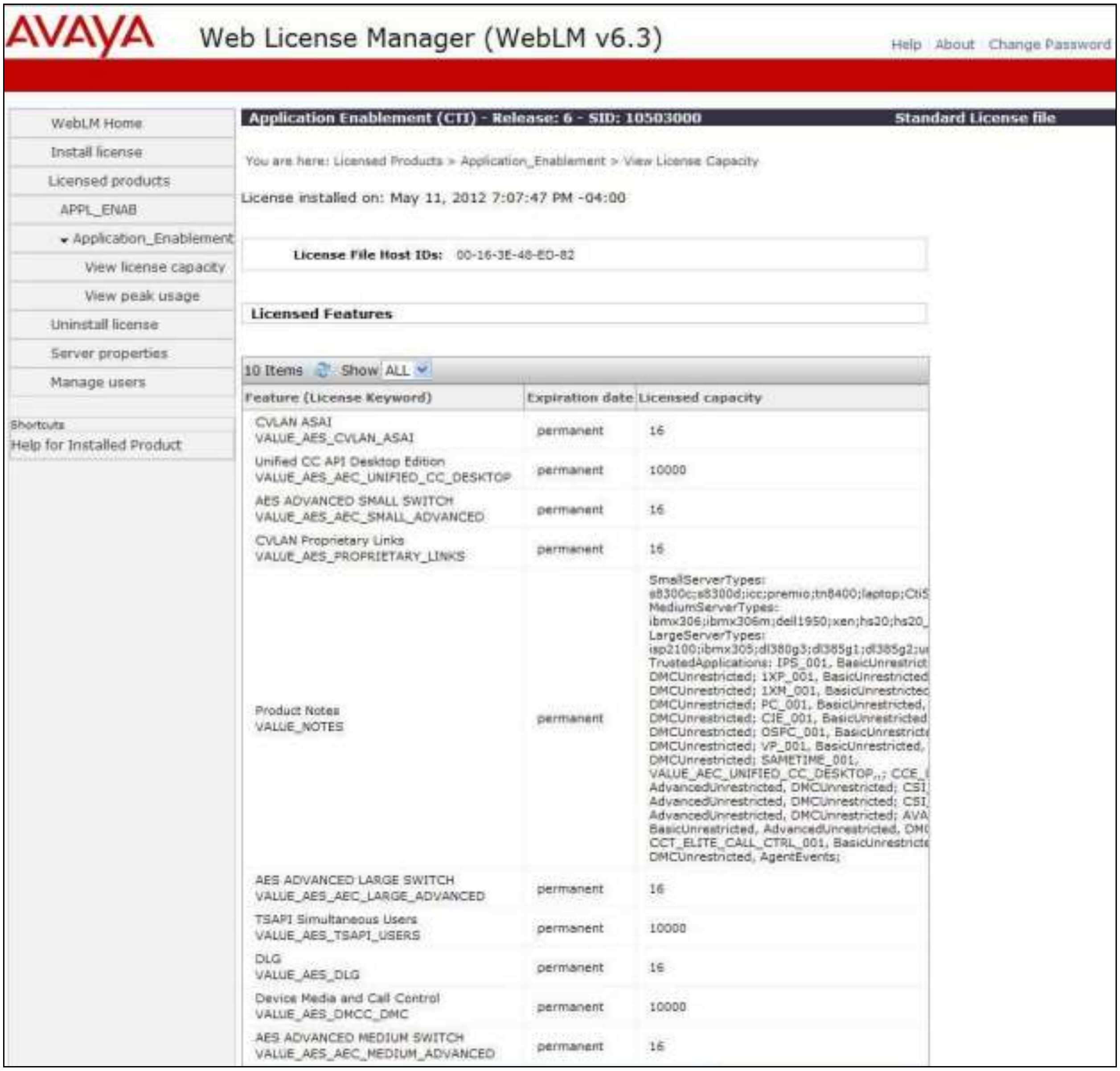
The **Welcome to OAM** screen is displayed next.

6.2. Verify License

Select **Licensing → WebLM Server Access** in the left pane, to display the **Web License Manager** pop-up screen (not shown), and log in using the appropriate credentials.

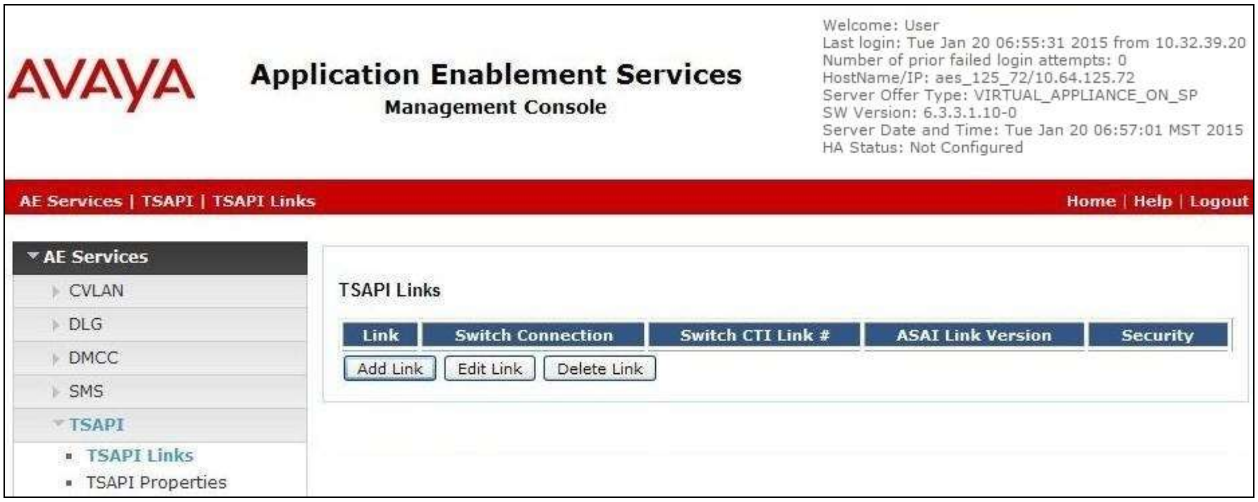
The **Web License Manager** screen below is displayed. Select **Licensed products → APPL_ENAB → Application_Enablement** in the left pane, to display the **Application Enablement (CTI)** screen in the right pane.

Verify that there are sufficient licenses for **TSAPI Simultaneous Users**, as shown below. Also verify that there is an applicable advanced switch license, in this case **AES ADVANCED LARGE SWITCH** for the Avaya S8800 Server.



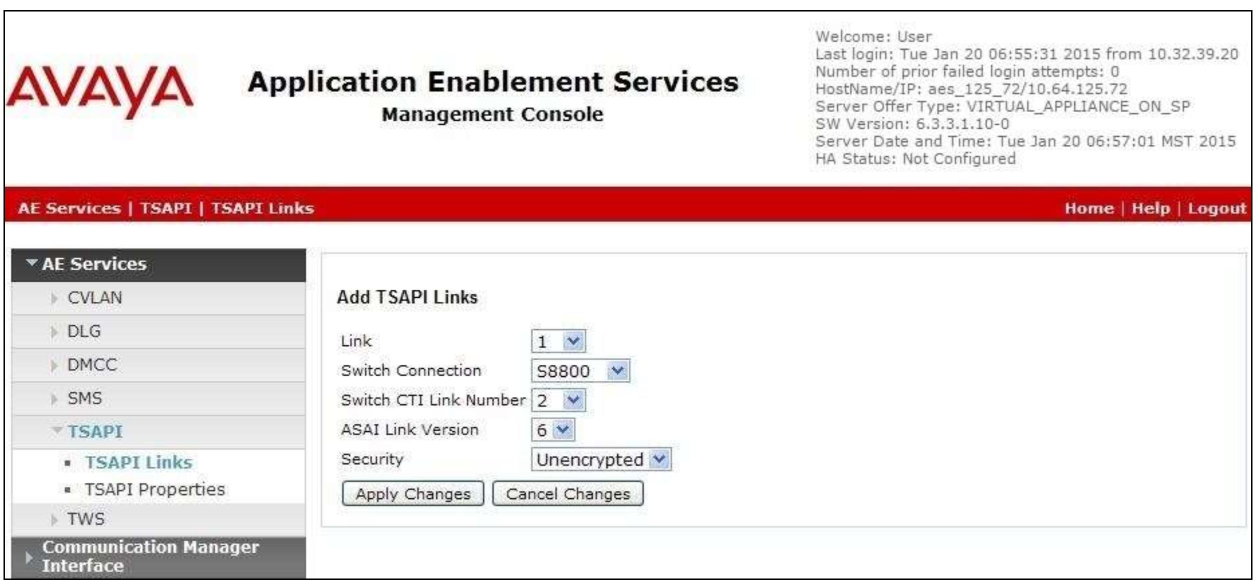
6.3. Administer TSAPI Link

Select **AE Services → TSAPI → TSAPI Links** from the left pane of the **Management Console**, to administer a TSAPI link. The **TSAPI Links** screen is displayed, as shown below. Click **Add Link**.



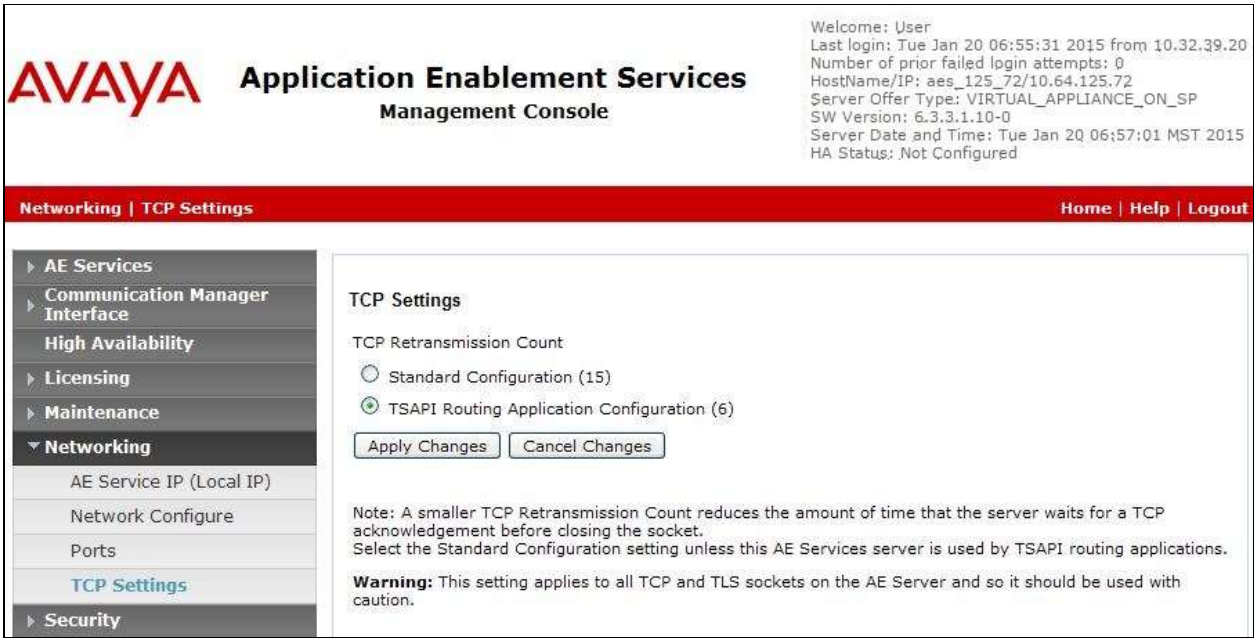
The **Add TSAPI Links** screen is displayed next.

The **Link** field is only local to the Application Enablement Services server, and may be set to any available number. For **Switch Connection**, select the relevant switch connection from the drop-down list. In this case, the existing switch connection "S8800" is selected. For **Switch CTI Link Number**, select the CTI link number from **Section 5.2**. Retain the default values in the remaining fields.



6.4. Administer TCP Settings

Select **Networking → TCP Settings** from the left pane, to display the **TCP Settings** screen in the right pane. For **TCP Retransmission Count**, select **TSAPI Routing Application Configuration**, as shown below.



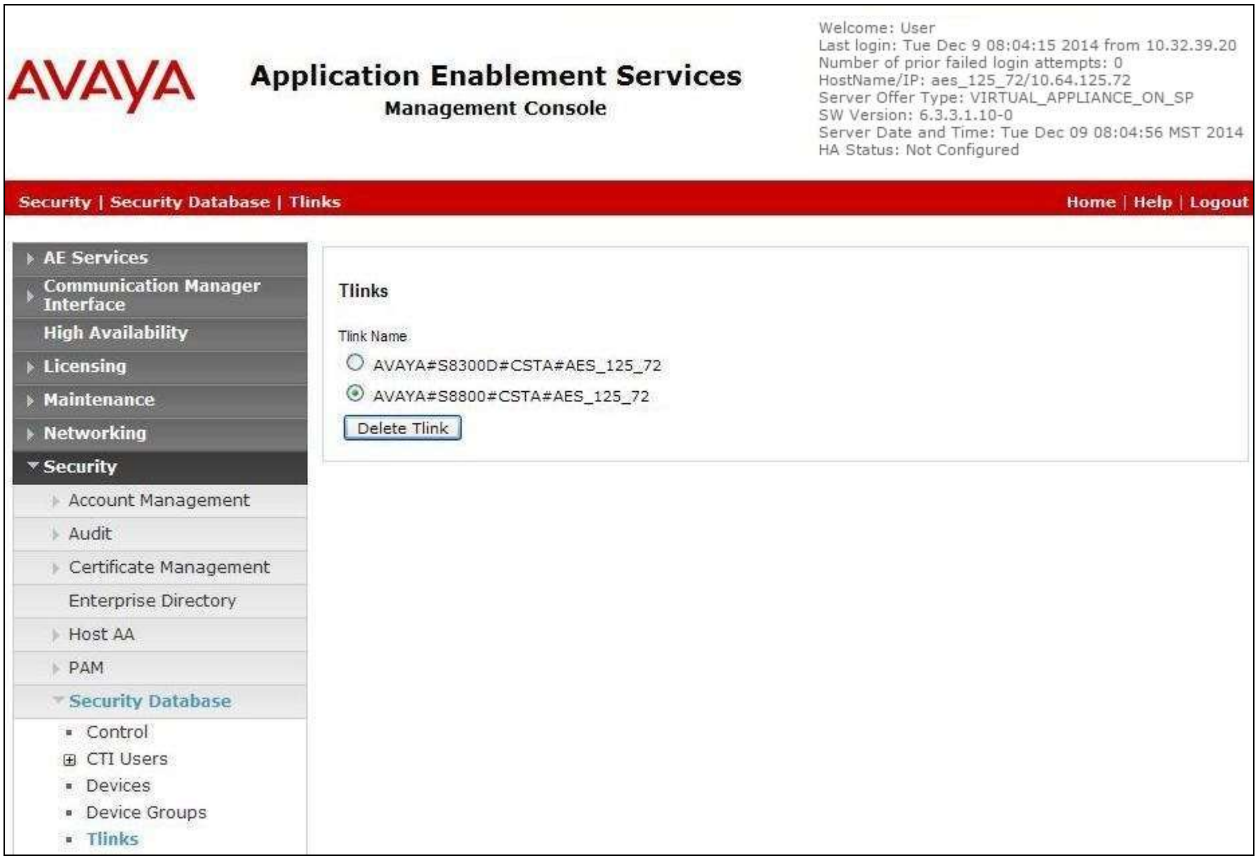
6.5. Restart Service

Select **Maintenance → Service Controller** from the left pane, to display the **Service Controller** screen in the right pane. Check **TSAPI Service** as shown below, and click **Restart Service**.

6.6. Obtain Tlink Name

Select **Security → Security Database → Tlinks** from the left pane. The **Tlinks** screen shows a listing of the Tlink names. A new Tlink name is automatically generated for the TSAPI service. Locate the Tlink name associated with the relevant switch connection, which would use the name of the switch connection as part of the Tlink name. Make a note of the associated Tlink name, to be used later for configuring Callback.

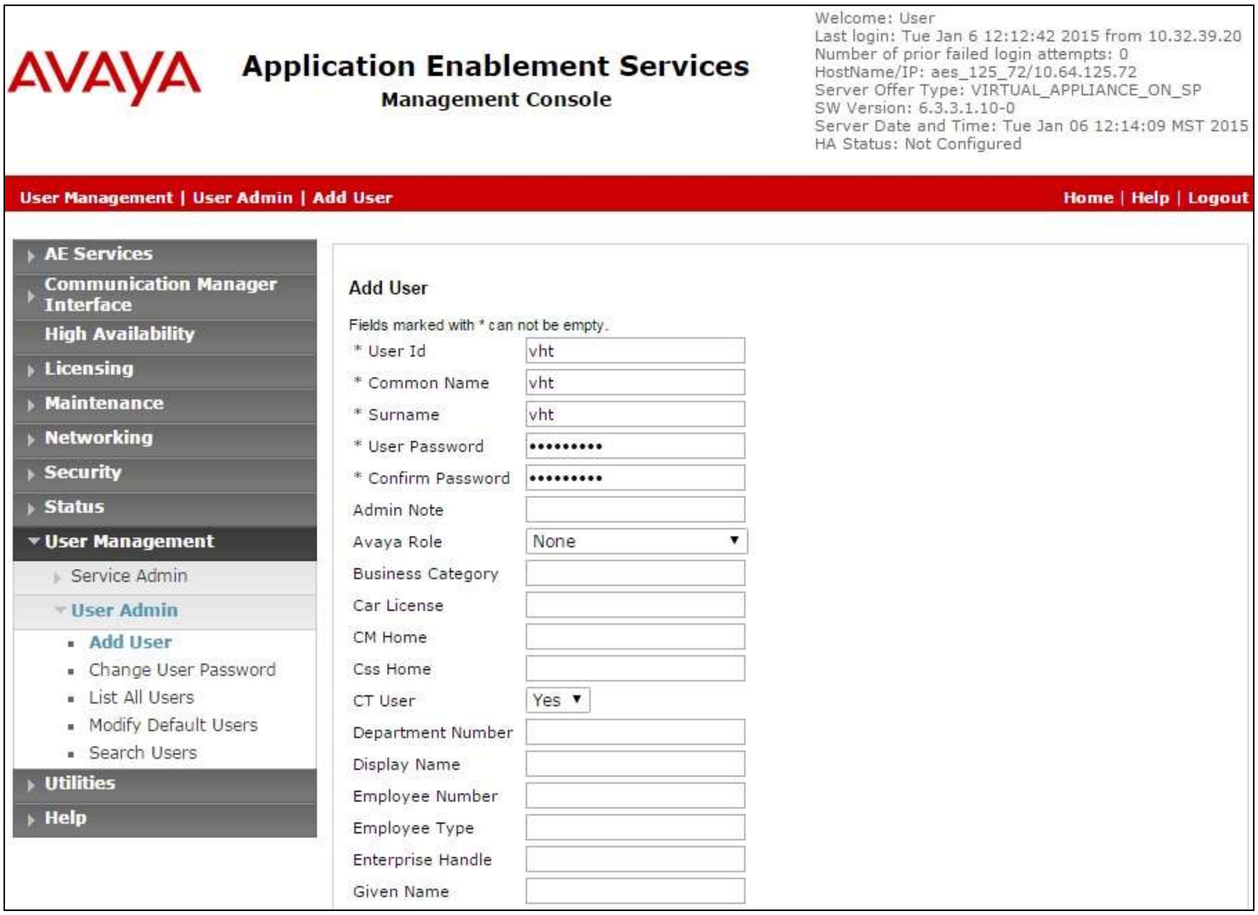
In this case, the associated Tlink name is "AVAYA#**S8800**#CSTA#AES_125_72". Note the use of the switch connection "S8800" from **Section 6.3** as part of the Tlink name.



6.7. Administer Callback User

Select **User Management → User Admin → Add User** from the left pane, to display the **Add User** screen in the right pane.

Enter desired values for **User Id**, **Common Name**, **Surname**, **User Password**, and **Confirm Password**. For **CT User**, select "Yes" from the drop-down list. Retain the default value in the remaining fields.



6.8. Verify Security Database

Select **Security → Security Database → Control** from the left pane, to display the **SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services** screen in the right pane.

Make certain that **Enable SDB for TSAPI Service, JTAPI and Telephony Web Services** retained the default value of unchecked. In the event that security database is used by the customer with this parameter already enabled, then follow [2] to configure access privileges for the Callback user from **Section 6.7**.

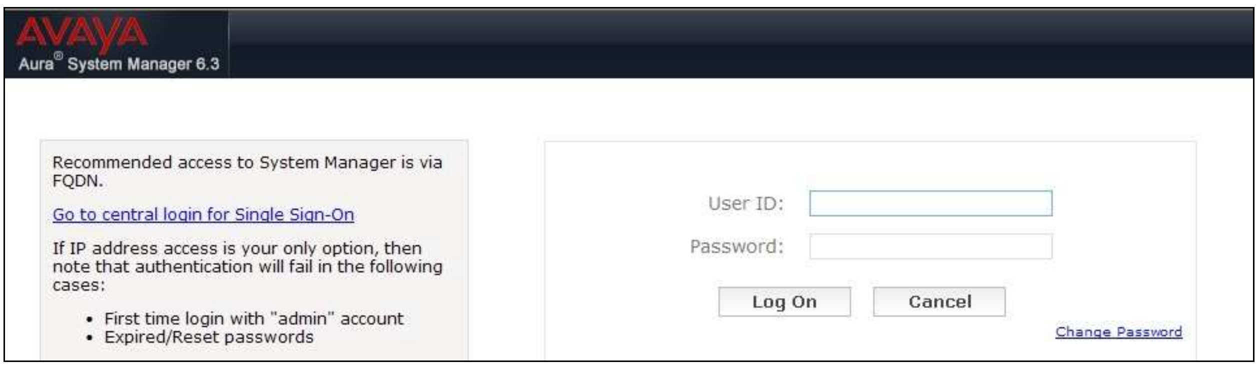
7. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

7.1. Launch System Manager

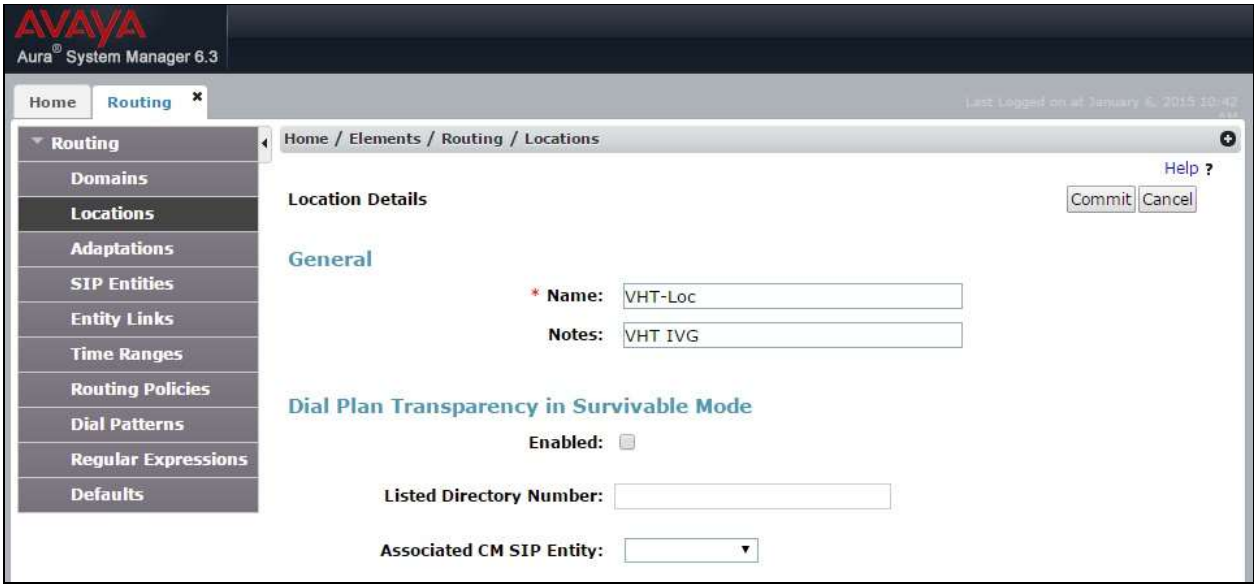
Access the System Manager web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of System Manager. Log in using the appropriate credentials.



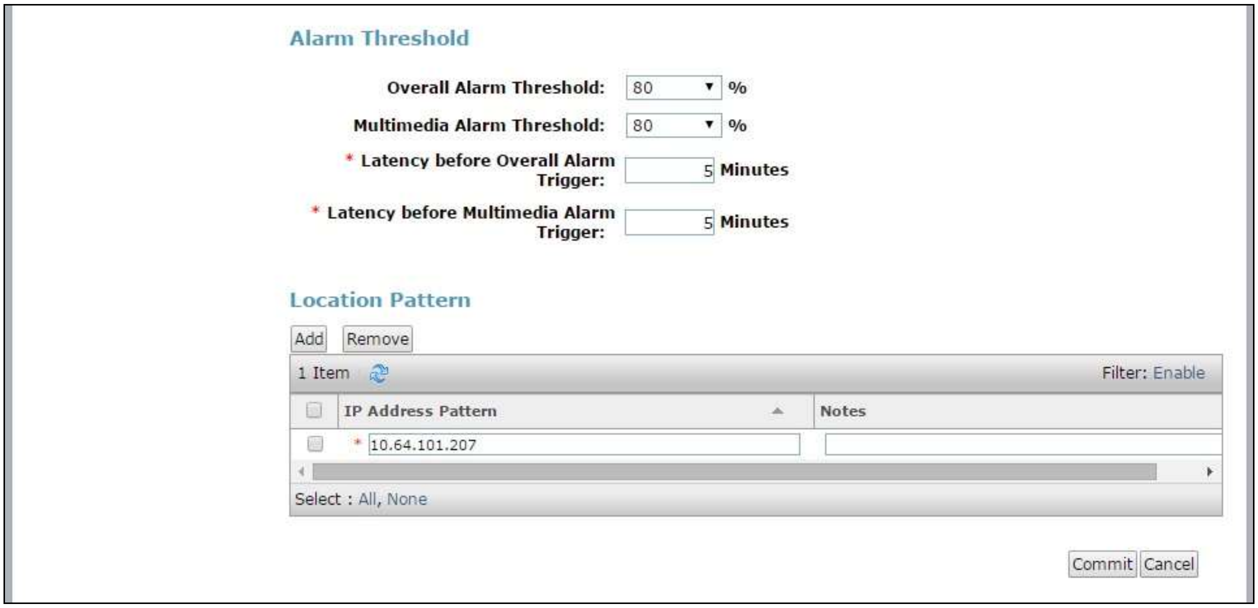
7.2. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for IVG.

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



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address pattern of IVG in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



7.3. Administer SIP Entities

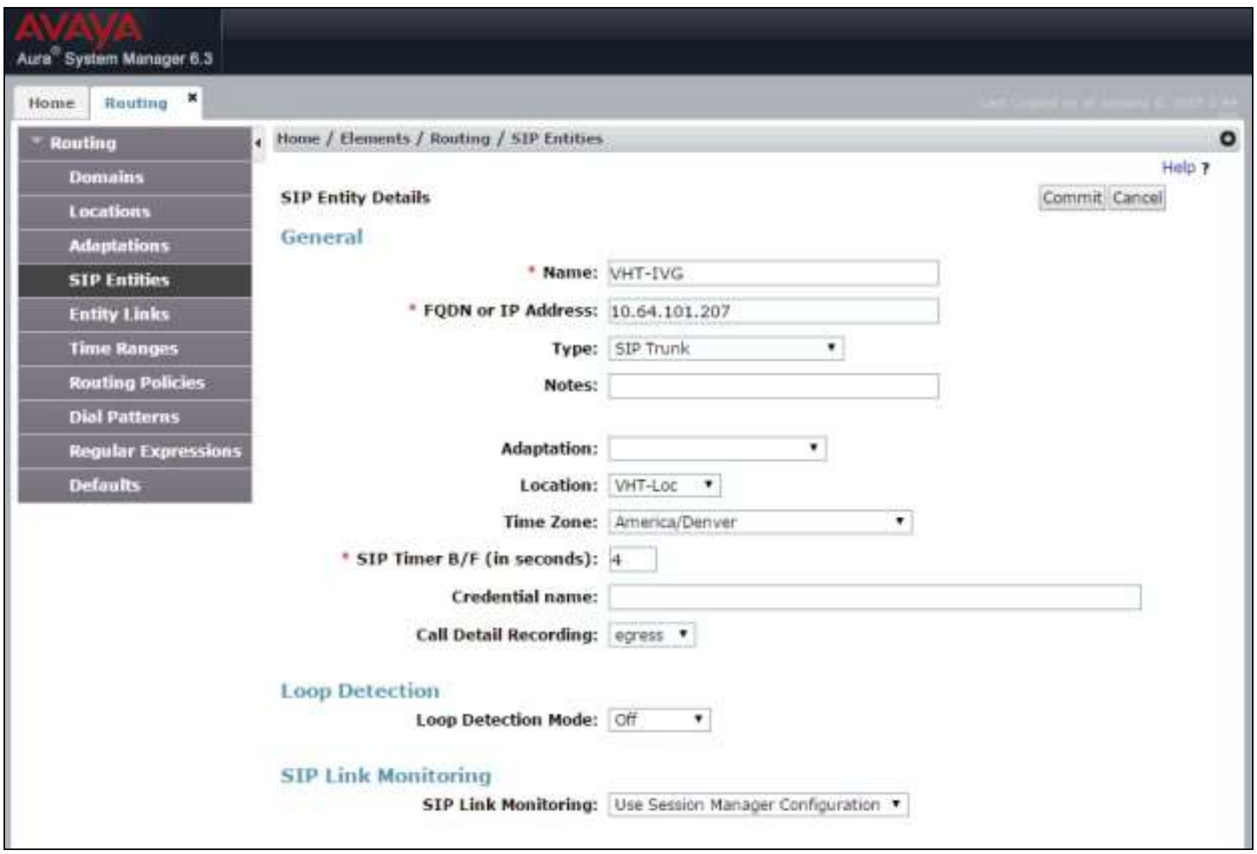
Add two new SIP entities, one for IVG and one for the new SIP trunks with Communication Manager.

7.3.1. SIP Entity for IVG

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

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

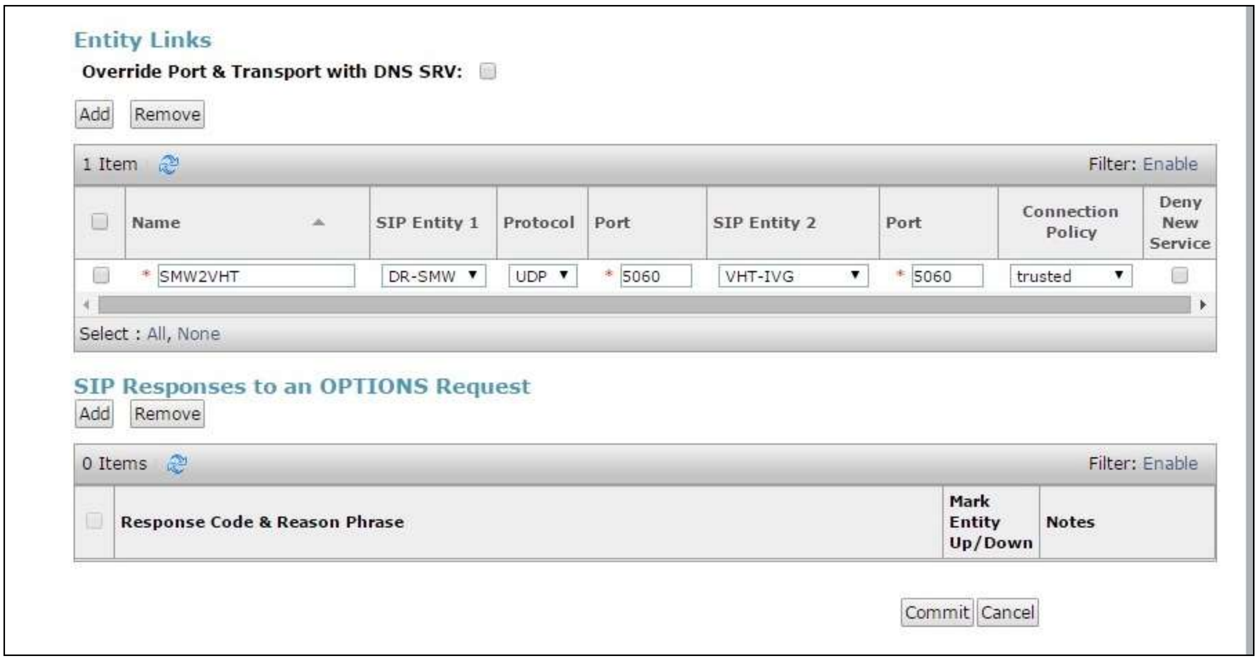
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of IVG.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the Callback location name from **Section 7.2**.
- **Time Zone:** Select the applicable time zone.



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 "DR-SMW".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The IVG entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that only UDP is supported by IVG.

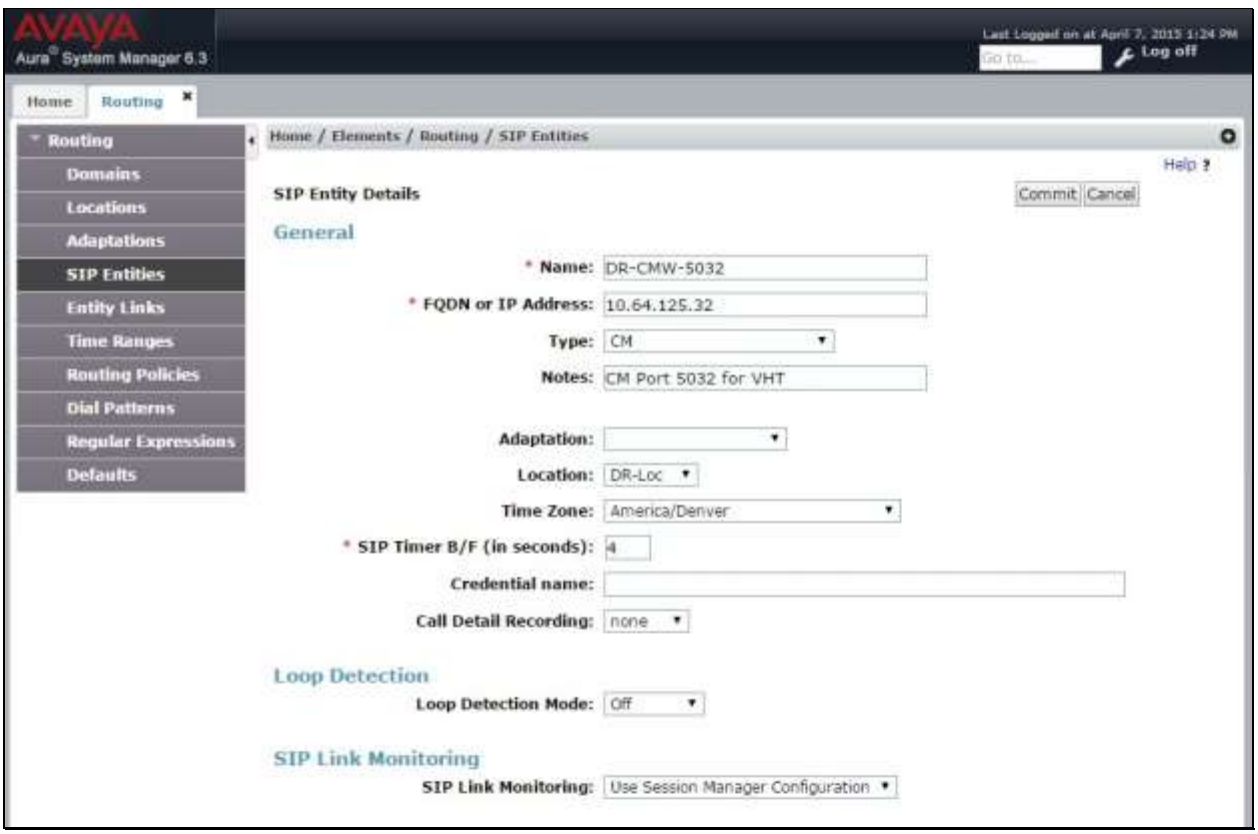


7.3.2. SIP Entity for Communication Manager

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with IVG.

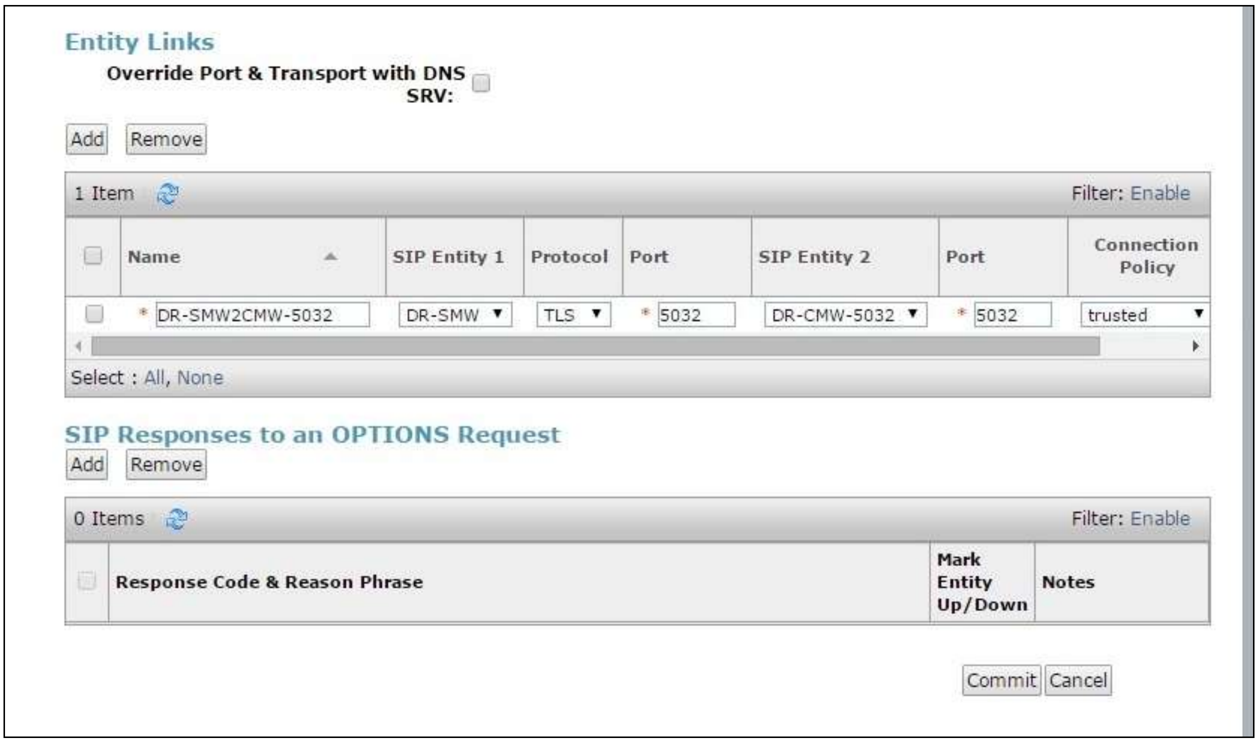
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Notes:** Any desired notes.
- **Adaptation:** Select any applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



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

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DR-SMW".
- **Protocol:** The signaling group transport method from **Section 5.5**.
- **Port:** The signaling group listen port number from **Section 5.5**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port number from **Section 5.5**.
- **Connection Policy:** "trusted"



7.4. Administer Routing Policies

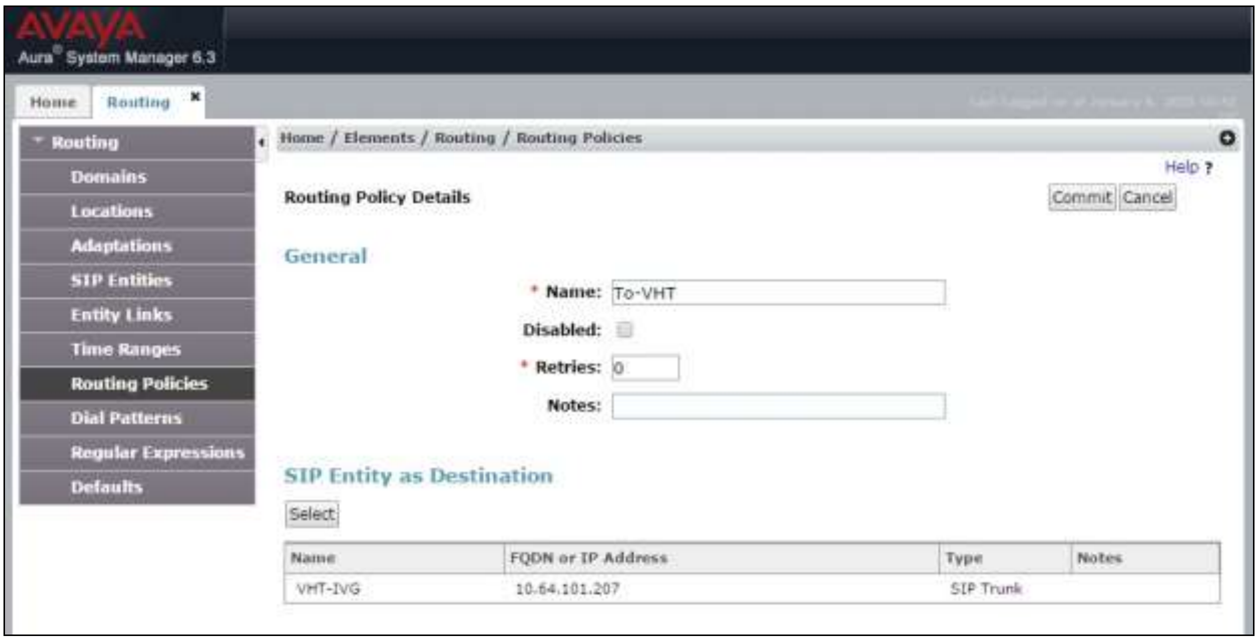
Add two new routing policies, one for IVG and one for the new SIP trunks with Communication Manager.

7.4.1. Routing Policy for IVG

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

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the IVG entity name from **Section 7.3.1**. The screen below shows the result of the selection.

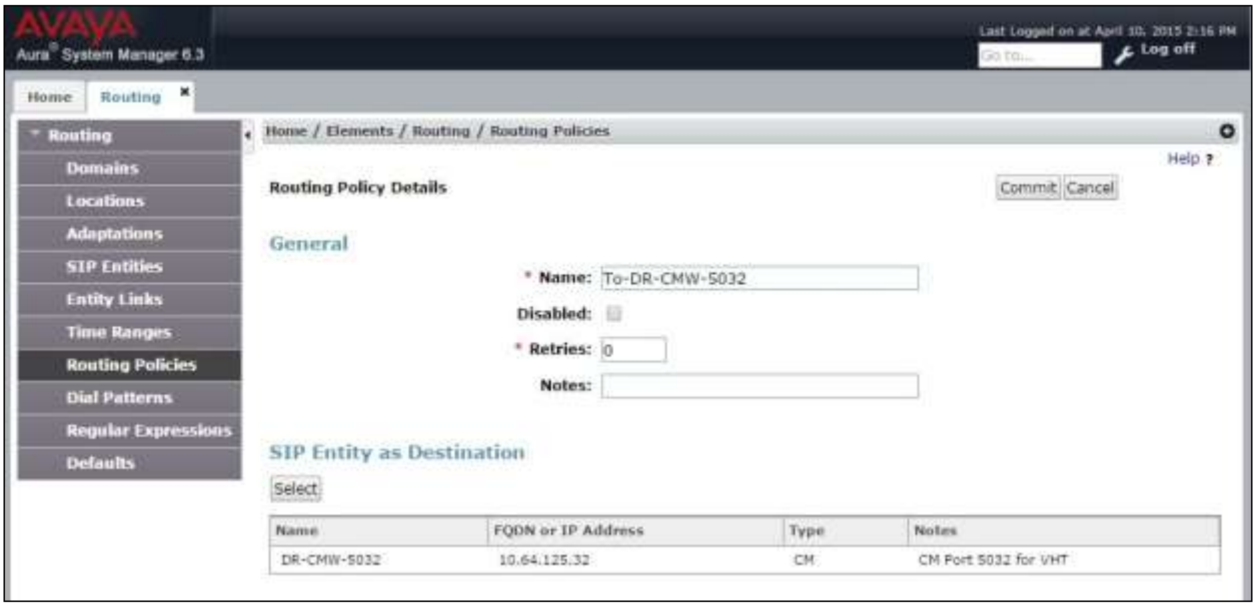


7.4.2. Routing Policy for Communication Manager

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

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 7.3.2**. The screen below shows the result of the selection.



7.5. Administer Dial Patterns

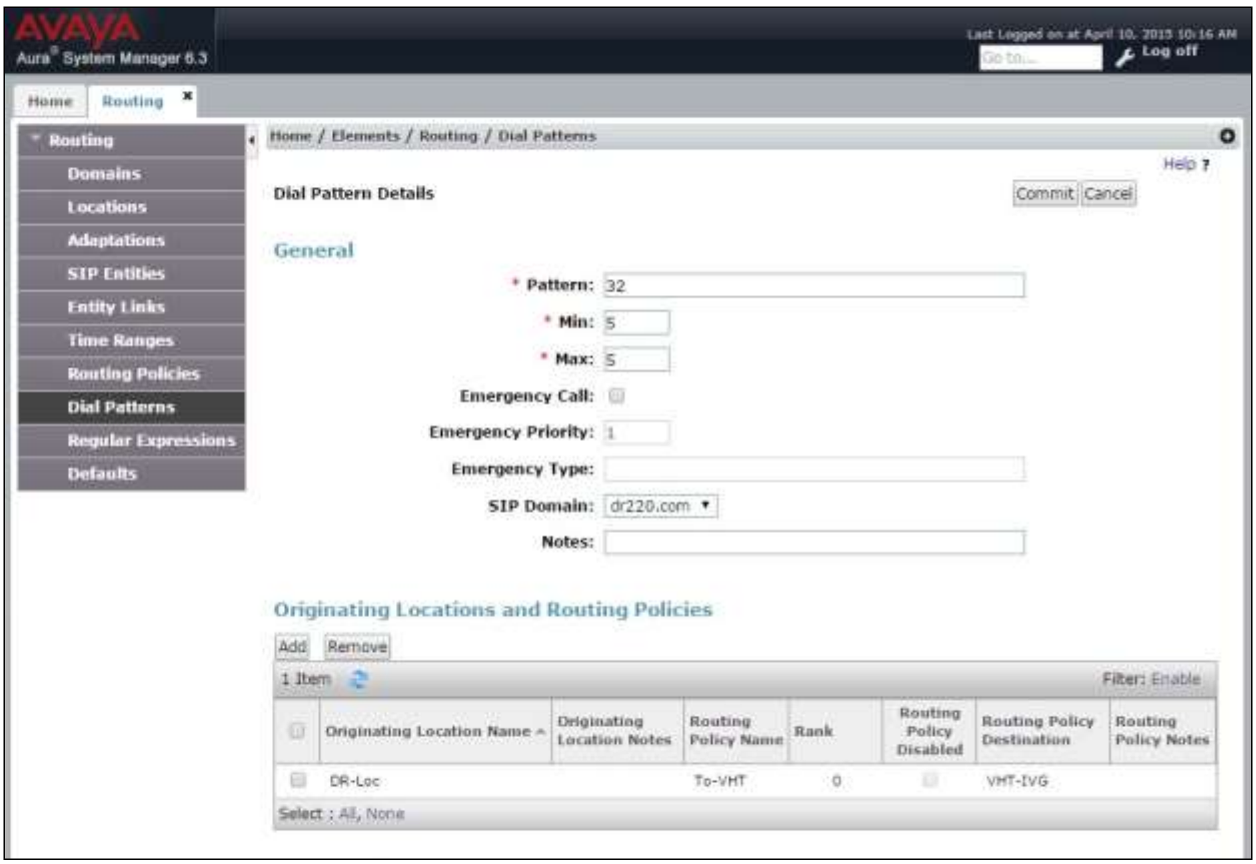
Add a new dial pattern for IVG, and update existing dial patterns for Communication Manager.

7.5.1. Dial Pattern for IVG

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

- **Pattern:** A dial pattern to match, in this case “32”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 5.5**.

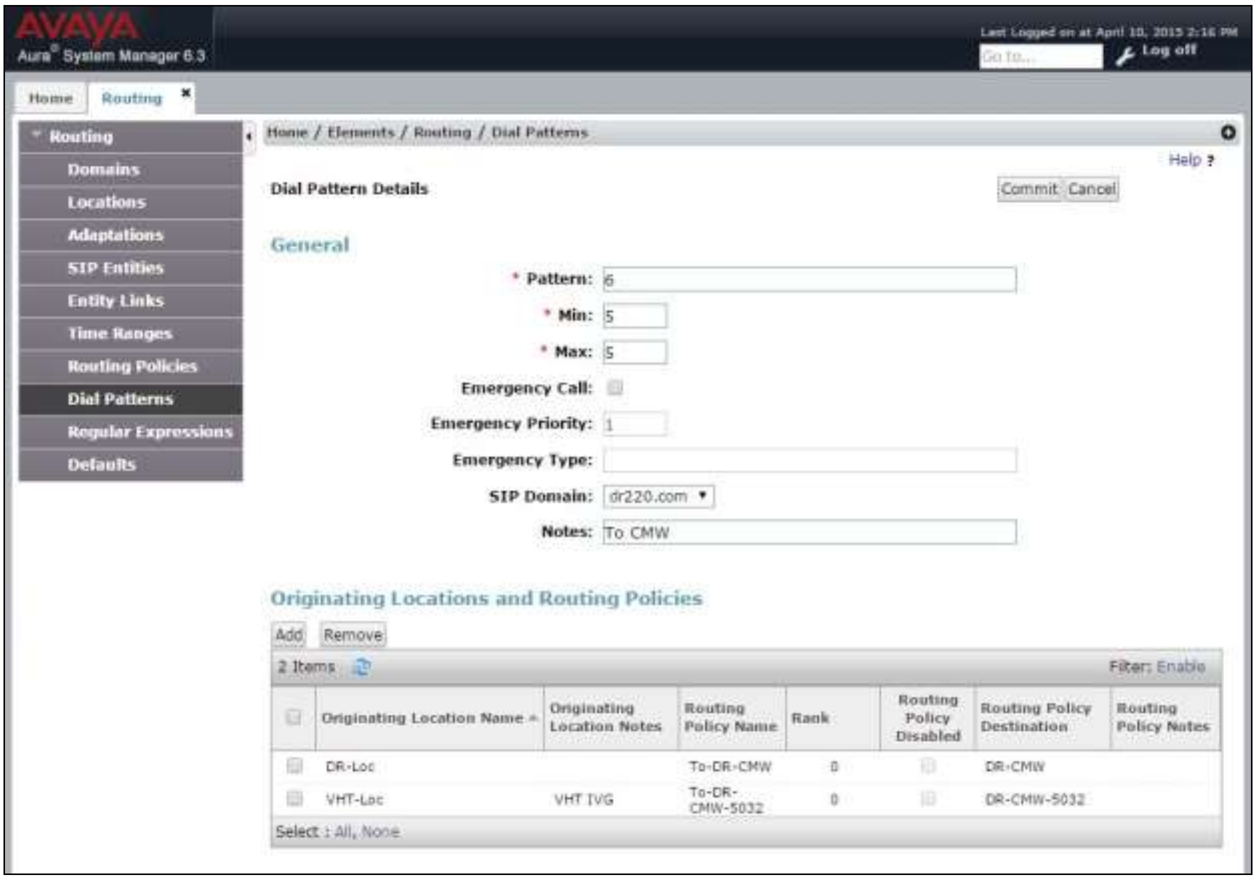
In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching IVG. In the compliance testing, the policy allowed for call origination from the Communication Manager location “DR-Loc”, and the IVG routing policy from **Section 7.4.1**was selected as shown below.



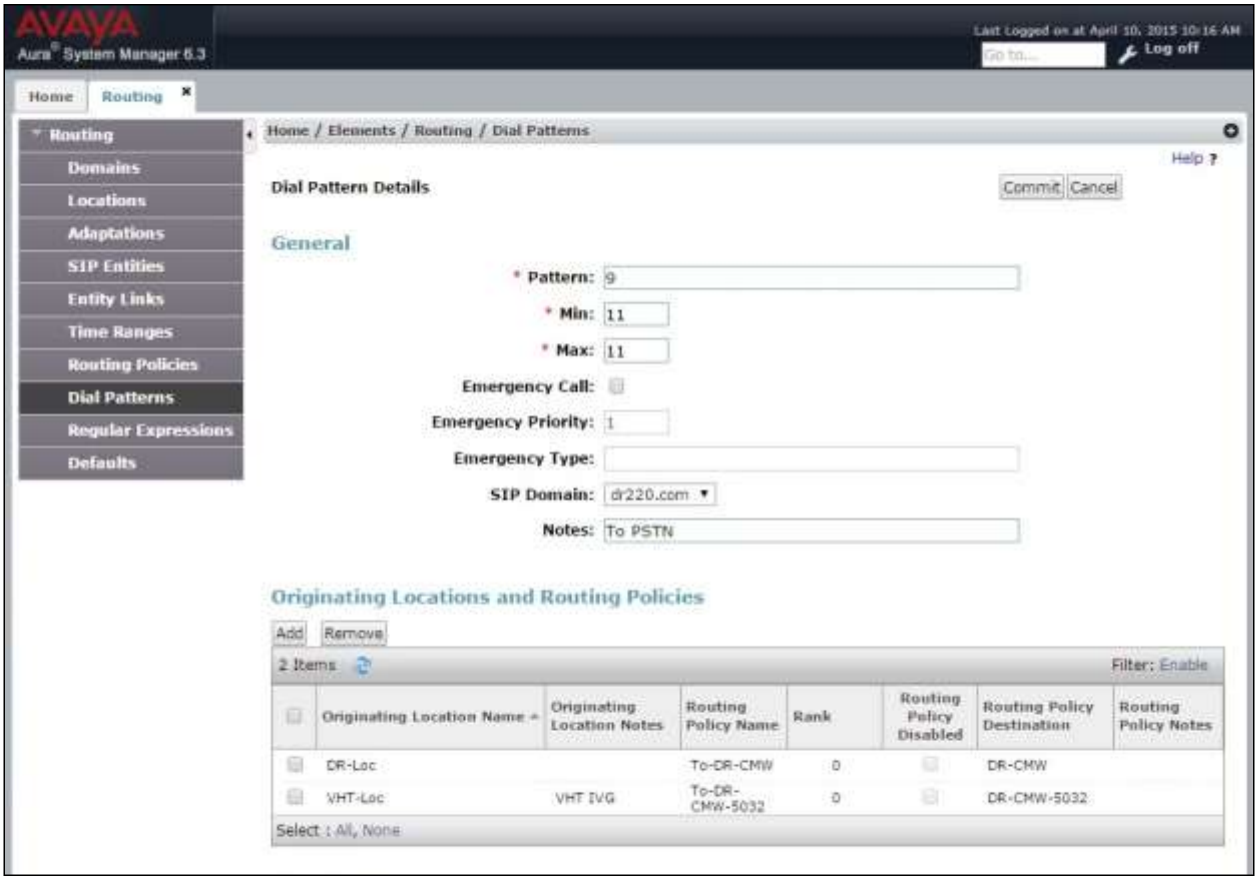
7.5.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the first existing and applicable dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "6" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new entry as necessary for calls from IVG. In the compliance testing, the new entry allowed for call origination from the IVG location from **Section 7.2**, and the Communication Manager routing policy from **Section 7.4.2** was selected as shown below. Retain the default values in the remaining fields.



Follow the procedures in this section to make similar changes to any other applicable dial patterns that IVG will be using to reach Communication Manager. In the compliance testing, one other dial pattern to reach the PSTN via Communication Manager was applicable and updated, as shown below.



8. Configure VHT IVG

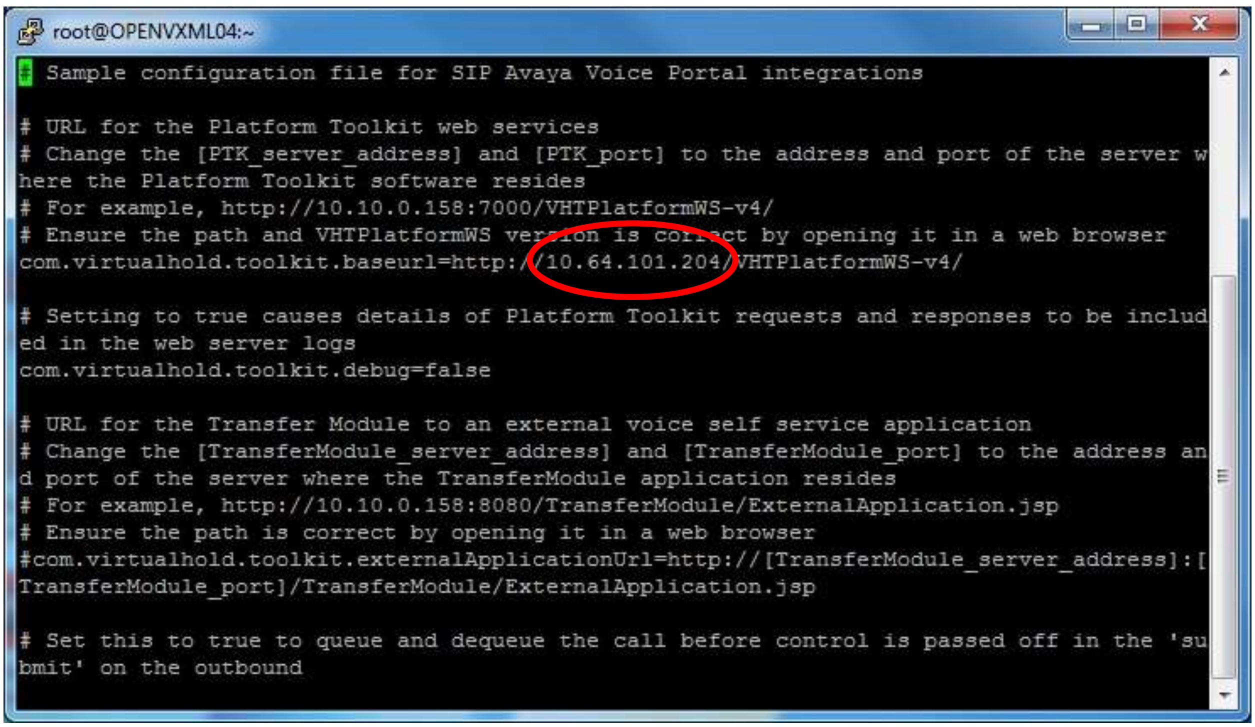
This section provides the procedures for configuring IVG. The procedures include the following areas:

- Administer toolkit properties
- Administer assigned extensions

The configuration of IVG is typically performed by VHT integration engineers. The procedural steps are presented in these Application Notes for informational purposes.

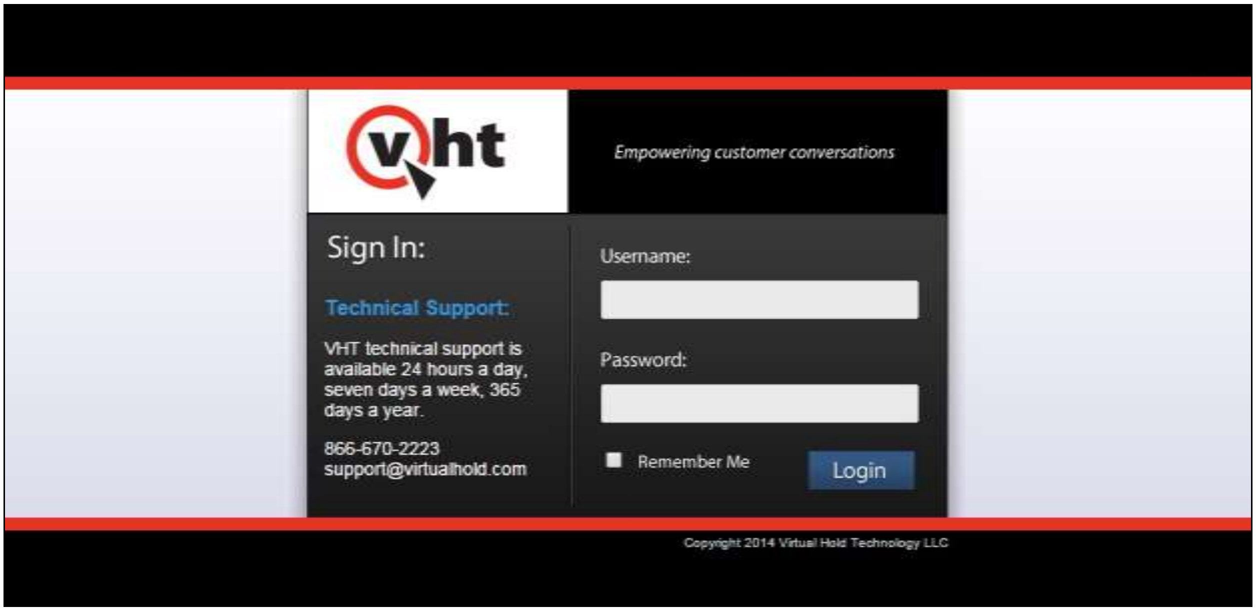
8.1. Administer Toolkit Properties

Log in to the Linux shell of IVG. Use the "vi /etc/VirtualHold/toolkit.properties" command to edit the file. Locate and replace the string "[PKT_server_address]:[PTK_port]" with the IP address of the Callback server. The screenshot below was captured after the replacement.

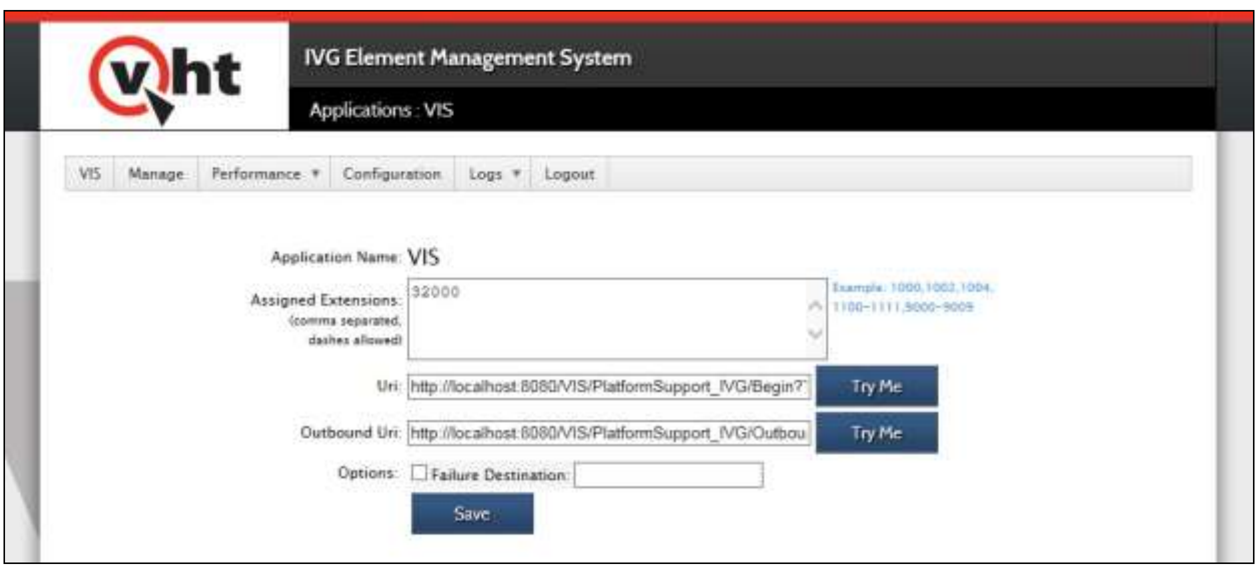


8.2. Administer Assigned Extensions

Access the IVG web interface by using the URL “http://ip-address:8080/ ivg-ems” in an Internet browser window, where “ip-address” is the IP address of IVG. Log in using the appropriate credentials.



The **IVG Element Management System** screen is displayed. For **Assigned Extensions**, enter the extension assigned to IVG, in this case “32000”. Retain the default values in the remaining fields.



9. Configure VHT Callback

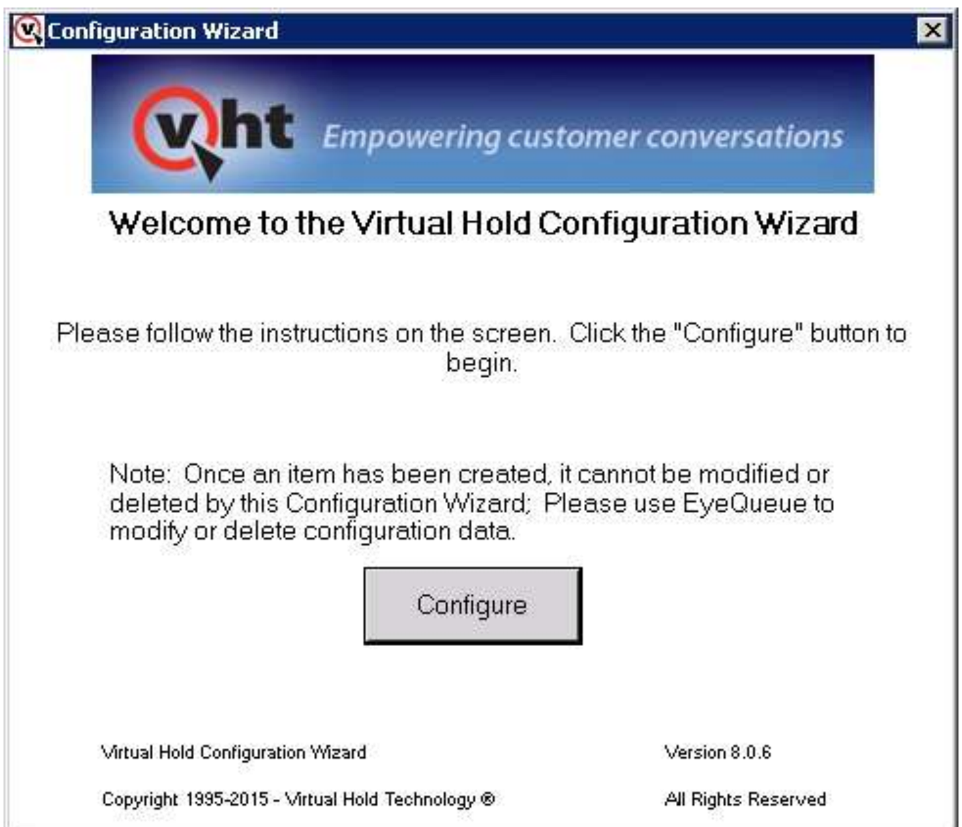
This section provides the procedures for configuring Callback. The procedures include the following areas:

- Launch configuration wizard
- Administer switch connection
- Administer agent groups
- Administer queues
- Administer callback and holding queues
- Administer incoming extensions
- Administer phone number configurations
- Administer route destinations

The configuration of Callback is typically performed by VHT integration engineers. The procedural steps are presented in these Application Notes for informational purposes.

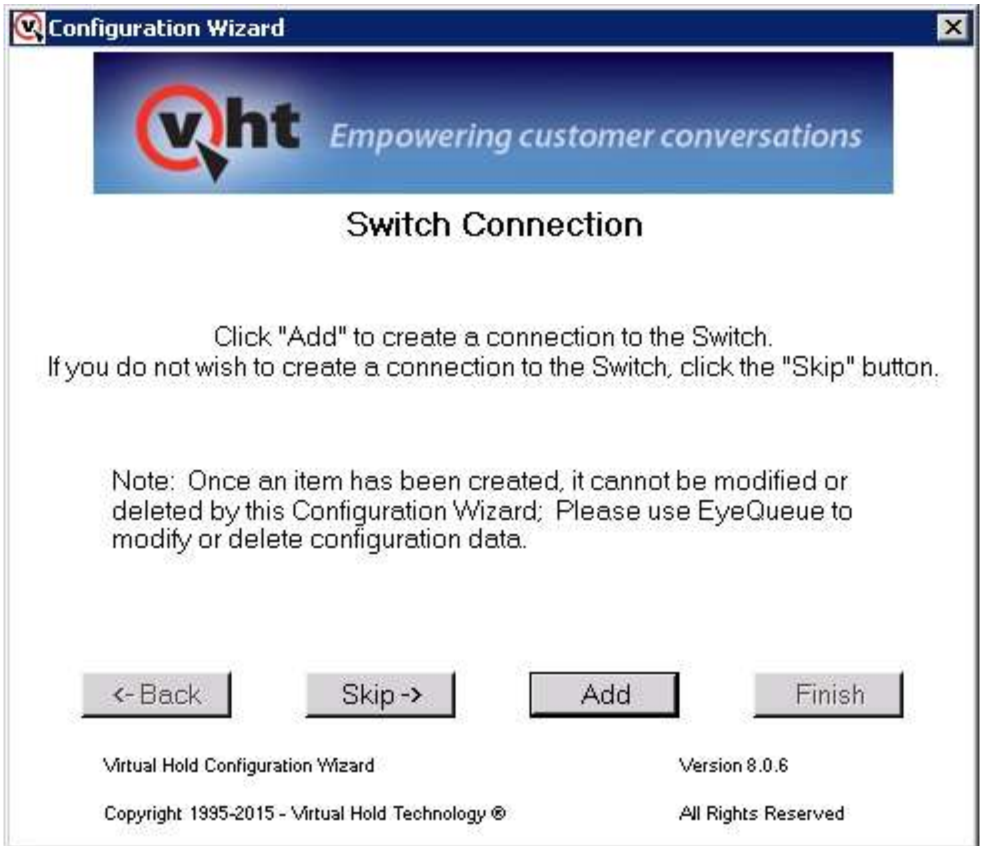
9.1. Launch Configuration Wizard

From the Callback server, navigate to **Start → All Programs → Virtual Hold Technology → Configuration → VHT Configuration Wizard** to launch the wizard. The **Welcome to the Virtual Hold Configuration Wizard** screen is displayed. Click **Configure** to proceed.



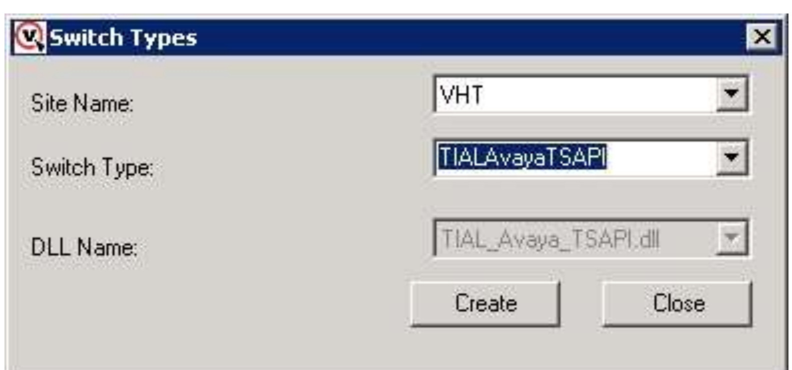
9.2. Administer Switch Connection

The **Switch Connection** screen is displayed. Click **Add** to create a connection to the switch.



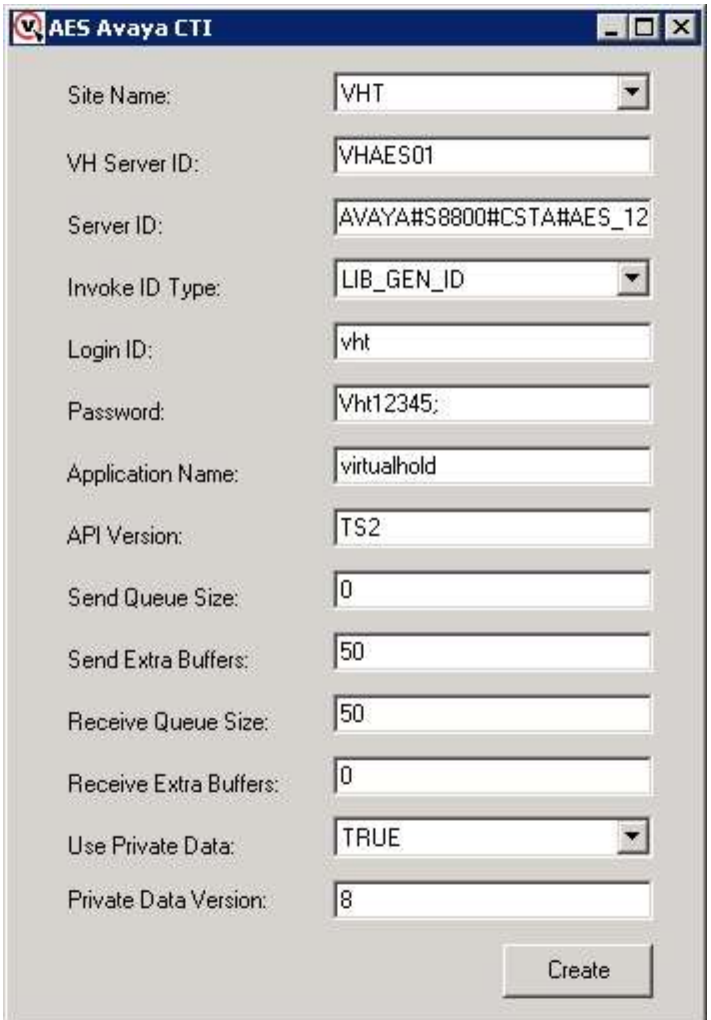
The **Switch Types** screen is displayed next. For **Switch Type**, select "TIALAvayaTSAPI" from the drop-down list. Note that the value of **Site Name** was automatically populated, and was created as part of installation.

Retain the default values in the remaining fields.



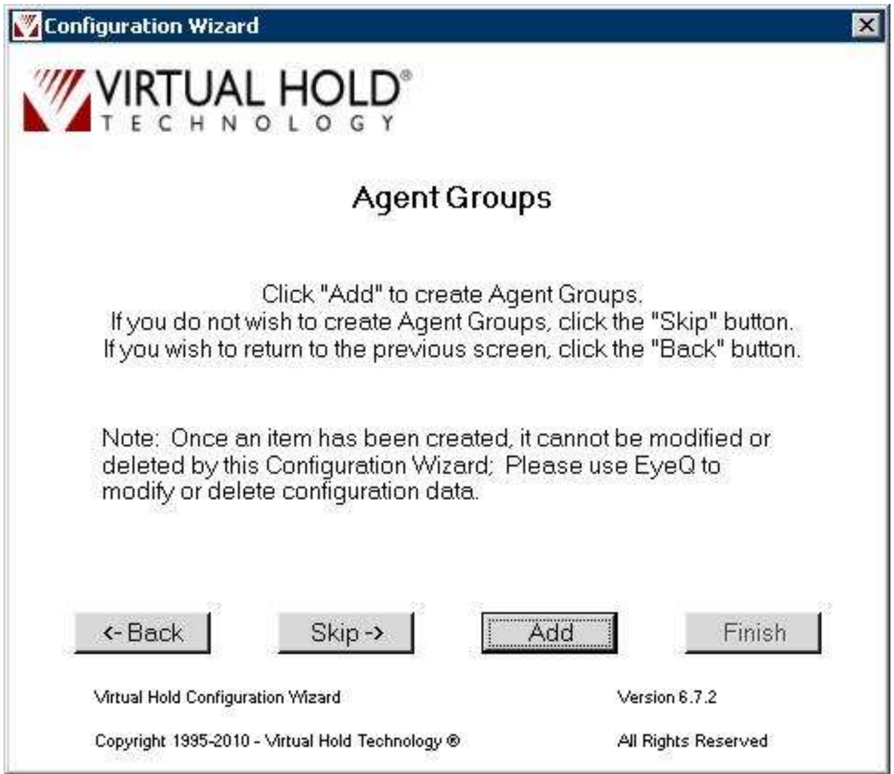
The **AES Avaya CTI** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **VH Server ID:** A descriptive name.
- **Server ID:** The Tlink name from **Section 6.6**.
- **Login ID:** The Callback user credentials from **Section 6.7**.
- **Password:** The Callback user credentials from **Section 6.7**.
- **Send Extra Buffers:** The desired extra buffers.
- **Receive Queue Size:** The desired queue size.
- **Use Private Data:** “TRUE”
- **Private Data Version:** “8”



9.3. Administer Agent Groups

The **Agent Groups** screen is displayed next. Click **Add**.



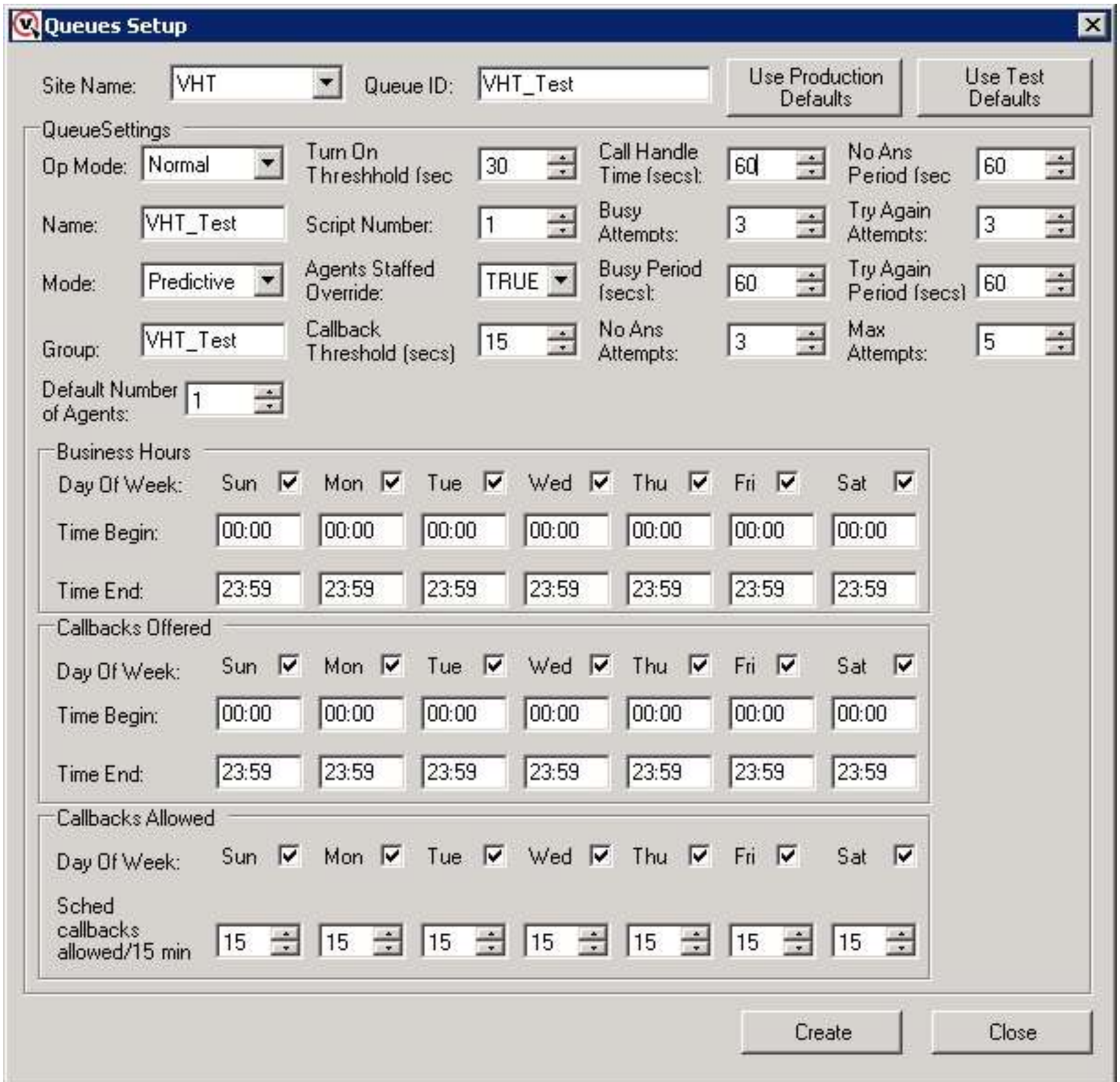
The screen below is displayed next. This screen is used to define the skill group. Retain the default value for **Site Name**. For **Starting Agent Group**, enter "x:y:z", where "x" is the desired agent group name, "y" is the VH server ID from **Section 9.2**, and "y" is the skill group extension from **Section 3**.

In the compliance testing, the value "VHT_TEST:VHAES01:65081" was used.

9.4. Administer Queues

Continue with the wizard until the **Queues** screen is displayed (not shown). Click **Add** to create queues.

The **Queues Setup** screen is displayed next. Consult reference [3] for desired configuration of these parameters. The screenshot below shows the values used in the compliance testing.



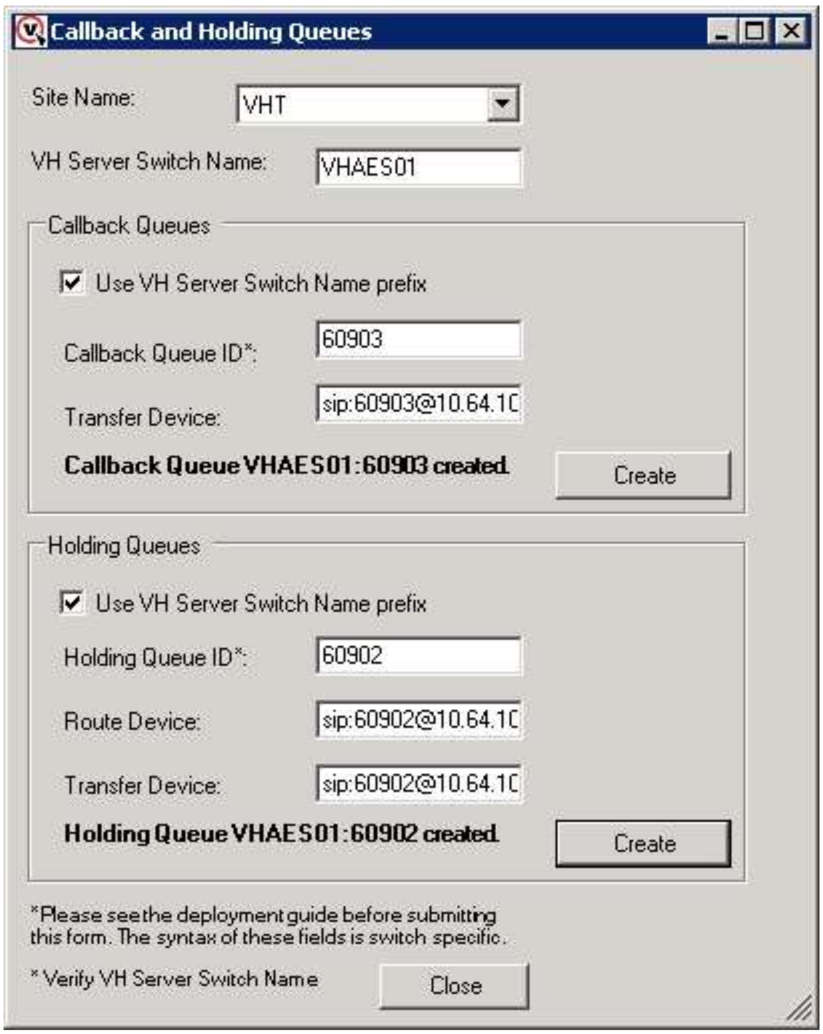
9.5. Administer Callback and Holding Queues

Continue with the wizard until the **Callback and Holding Queues** screen is displayed (not shown). Click **Add** to create queues. The screen below is displayed next.

In the **Callback Queues** sub-section, enter the Callback VDN extension from **Section 5.4.3** for **Callback Queue ID**. For **Transfer Device**, enter "sip:x@y", where "x" is the Callback VDN extension, and "y" is the IP address of the Session Manager signaling interface.

In the **Holding Queues** sub-section, enter the Hold VDN extension from **Section 5.4.2** for **Holding Queue ID**. For **Route Device** and **Transfer Device**, enter "sip:x@y", where "x" is the Hold VDN extension, and "y" is the IP address of the Session Manager signaling interface.

Retain the default values for the remaining fields.

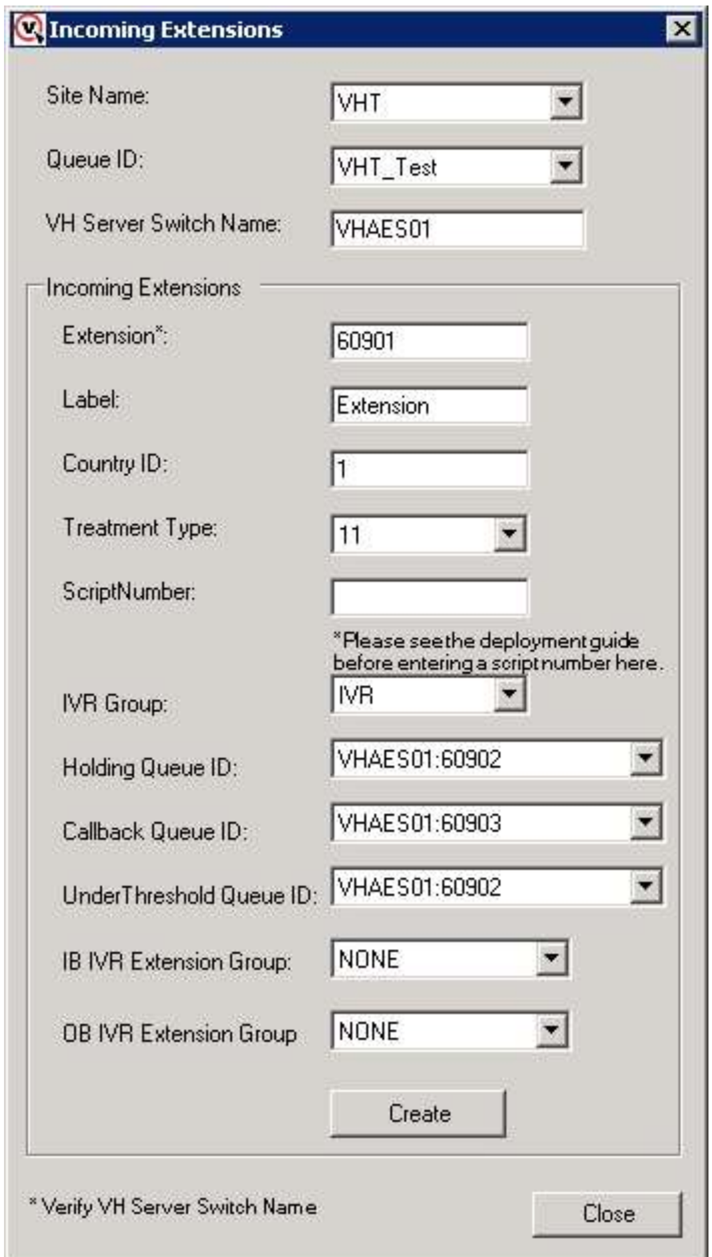


9.6. Administer Incoming Extensions

Continue with the wizard until the **Incoming Extensions** screen is displayed (not shown). Click **Add** to create an incoming extension for Callback.

The screen below is displayed next. For **Extension**, enter the Entry VDN extension from **Section 5.4.1**. For **Treatment Type**, select "11".

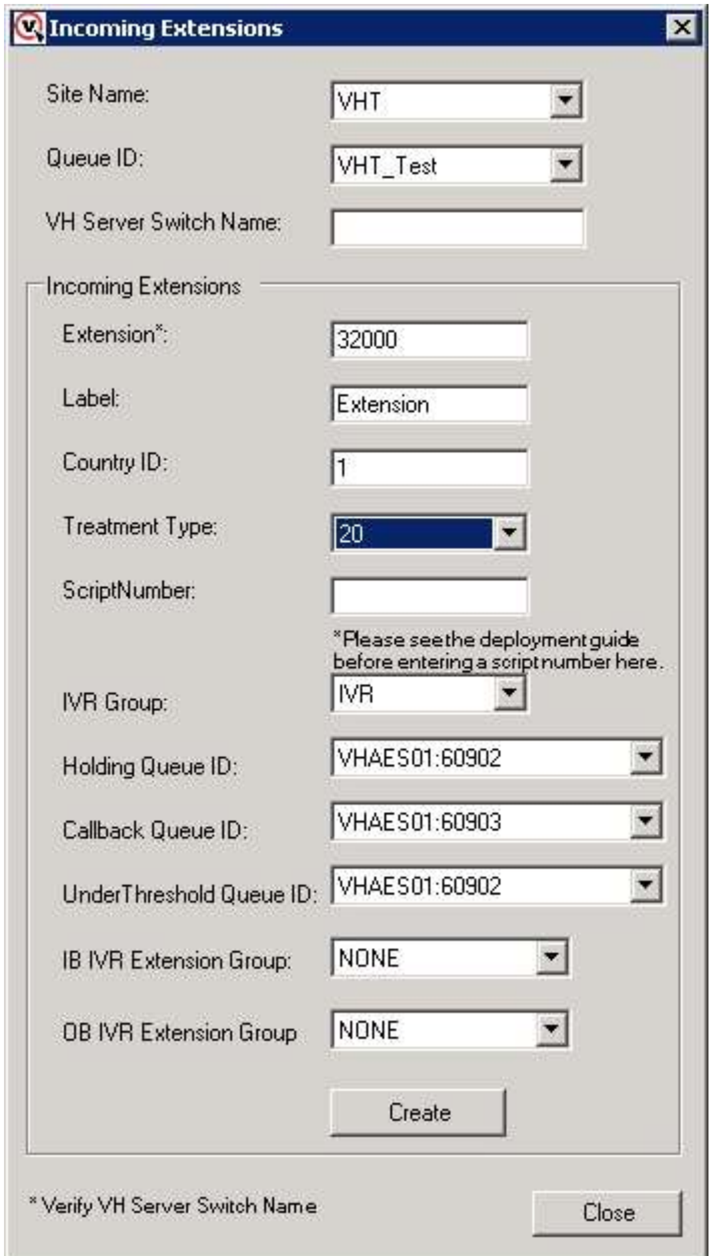
Retain the default values in the remaining fields.



Repeat the same procedures to create an incoming extension for IVG.

For **Extension**, enter the extension assigned to IVG, in this case "32000". For **Treatment Type**, select "20".

Retain the default values in the remaining fields, including blank for **VH Server Switch Name**.

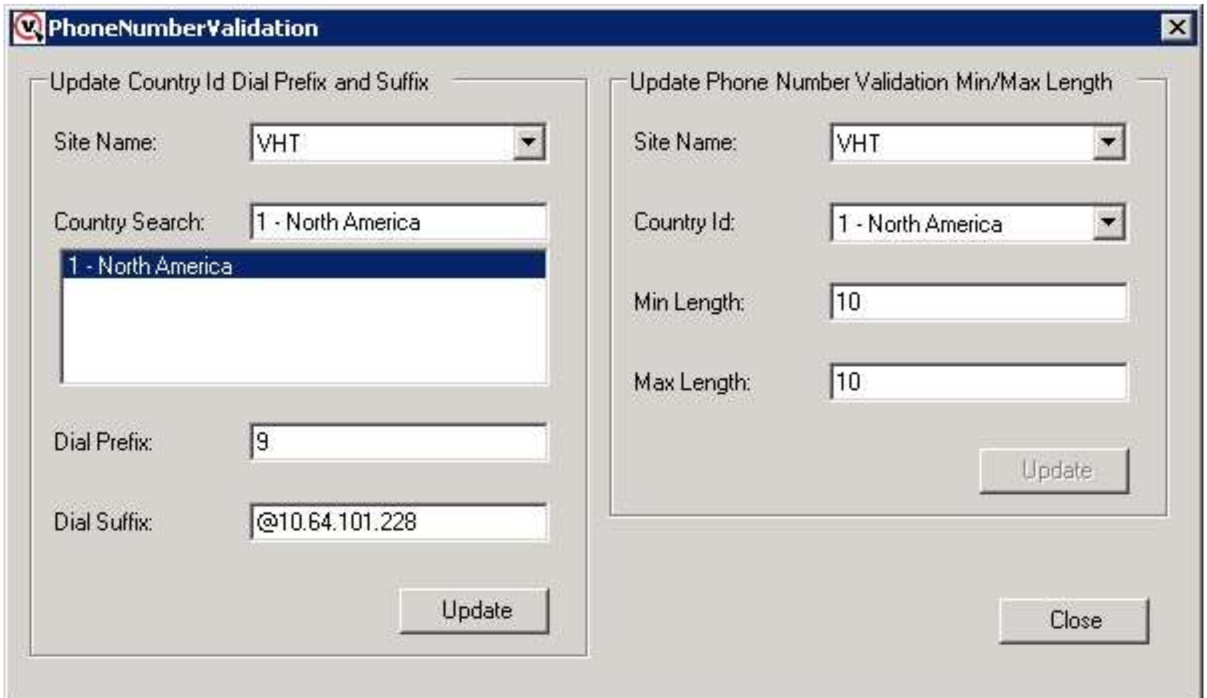


9.7. Administer Phone Number Configurations

Continue with the wizard until the **Phone Number Configurations** screen is displayed (not shown). Click **Add** to create phone number configuration, the screen below is displayed next.

For **Country Search**, locate the applicable country, which will enable the **Dial Prefix** and **Dial Suffix** fields. For **Dial Prefix**, enter any applicable dialing prefix for the network. For **Dial Suffix**, enter "@y", where "y" is the IP address of the Session Manager signaling interface.

Retain the default values in the remaining fields.

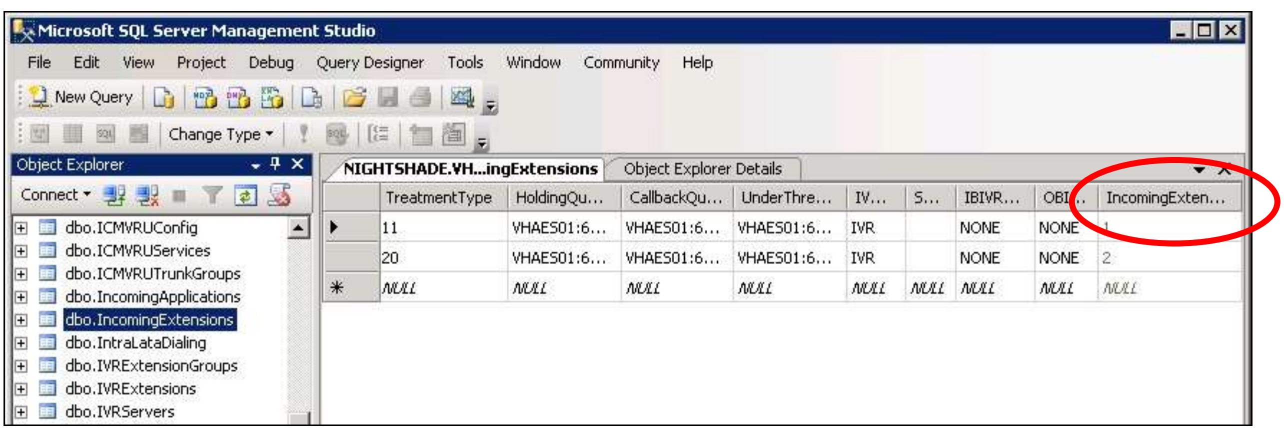


9.8. Administer Route Destination

From the Callback server, navigate to **Start → All Programs → Microsoft SQL Server 2008 R2 → SQL Server Management Studio** to launch and connect to the SQL server.

Navigate to **Databases → VHT_Config → Tables → dbo.IncomingExtensions** in the left pane, right click the entry and select **Edit Top 200 Rows**.

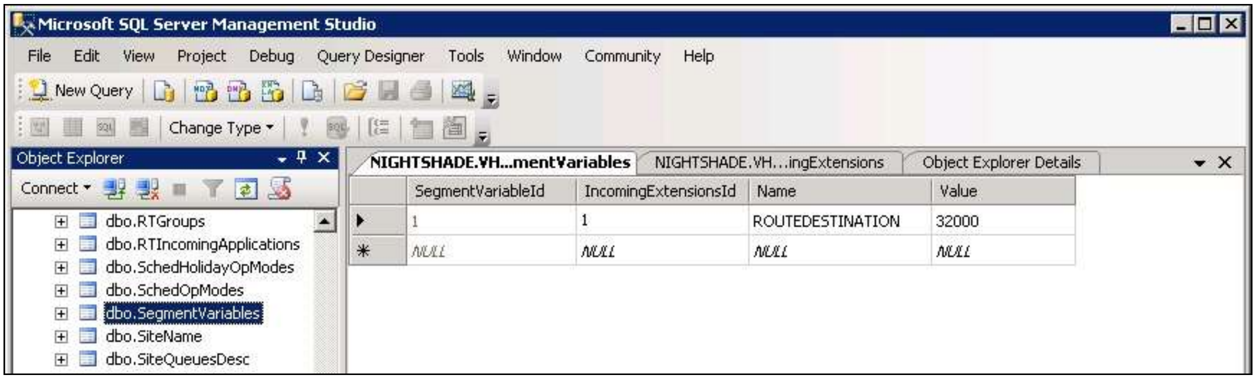
Locate the entry associated with Callback with "11" as **Treatment Type**. Make a note of the associated **IncomingExtensionsId** value, in this case "1", as shown below.



Scroll down to **dbo.SegmentVariables** in the left pane, right click the entry and select **Edit Top 200 Rows**. Add an entry and enter the following values for the specified fields, and retain the default values for the remaining fields.

- **IncomingExtensionsId:** The value from the **dbo.IncomingExtensions** table from above.
- **Name:** "ROUTEDESTINATION"
- **Value:** The assigned extension to IVG, in this case "32000".

Restart the VHT Core Monitor and VHT Peripheral Monitor services (not shown).



10. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Application Enablement Services, Session Manager, Callback and IVG.

10.1. Verify Avaya Aura® Communication Manager

On Communication Manager, verify the status of the administered CTI link by using the "status aesvcs cti-link" command. Verify that the **Service State** is "established" for the CTI link number administered in **Section 5.2**, as shown below.

```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

CTI   Version  Mnt   AE Services      Service       Msgs     Msgs
Link           Busy  Server           State         Sent     Rcvd

1              no                     down          0        0
2     6        no    aes_125_72       established   603      601
```

Verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.6**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 32

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0032/001 T00113   in-service/idle    no
0032/002 T00114   in-service/idle    no
0032/003 T00115   in-service/idle    no
0032/004 T00116   in-service/idle    no
0032/005 T00117   in-service/idle    no
0032/006 T00118   in-service/idle    no
0032/007 T00119   in-service/idle    no
0032/008 T00120   in-service/idle    no
0032/009 T00121   in-service/idle    no
0032/010 T00122   in-service/idle    no
```

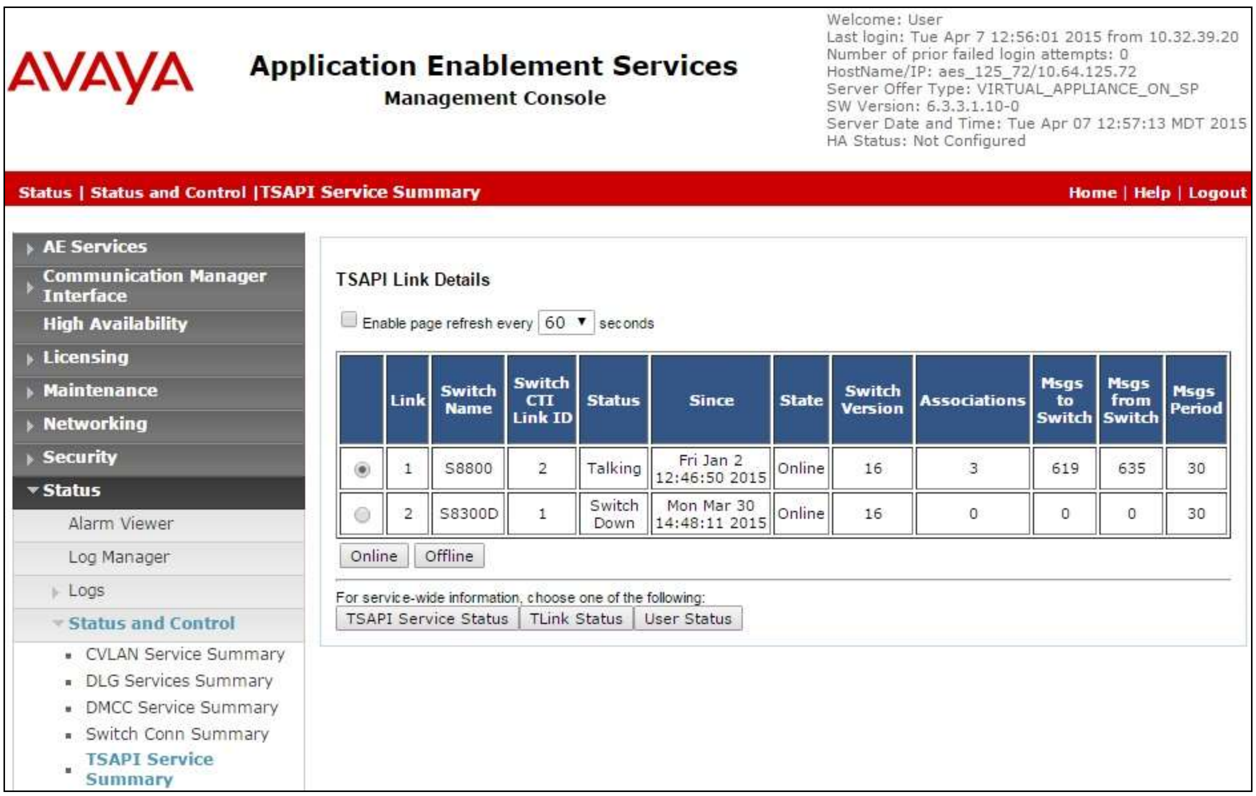
Verify the status of the SIP signaling groups by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 5.5**. Verify that the **Group State** is “in-service”, as shown below.

```
status signaling-group 32
                         STATUS SIGNALING GROUP

        Group ID: 32
      Group Type: sip

     Group State: in-service
```

10.2. Verify Avaya Aura® Application Enablement Services

On Application Enablement Services, verify the status of the TSAPI link by selecting **Status → Status and Control → TSAPI Service Summary** from the left pane. The **TSAPI Link Details** screen is displayed.

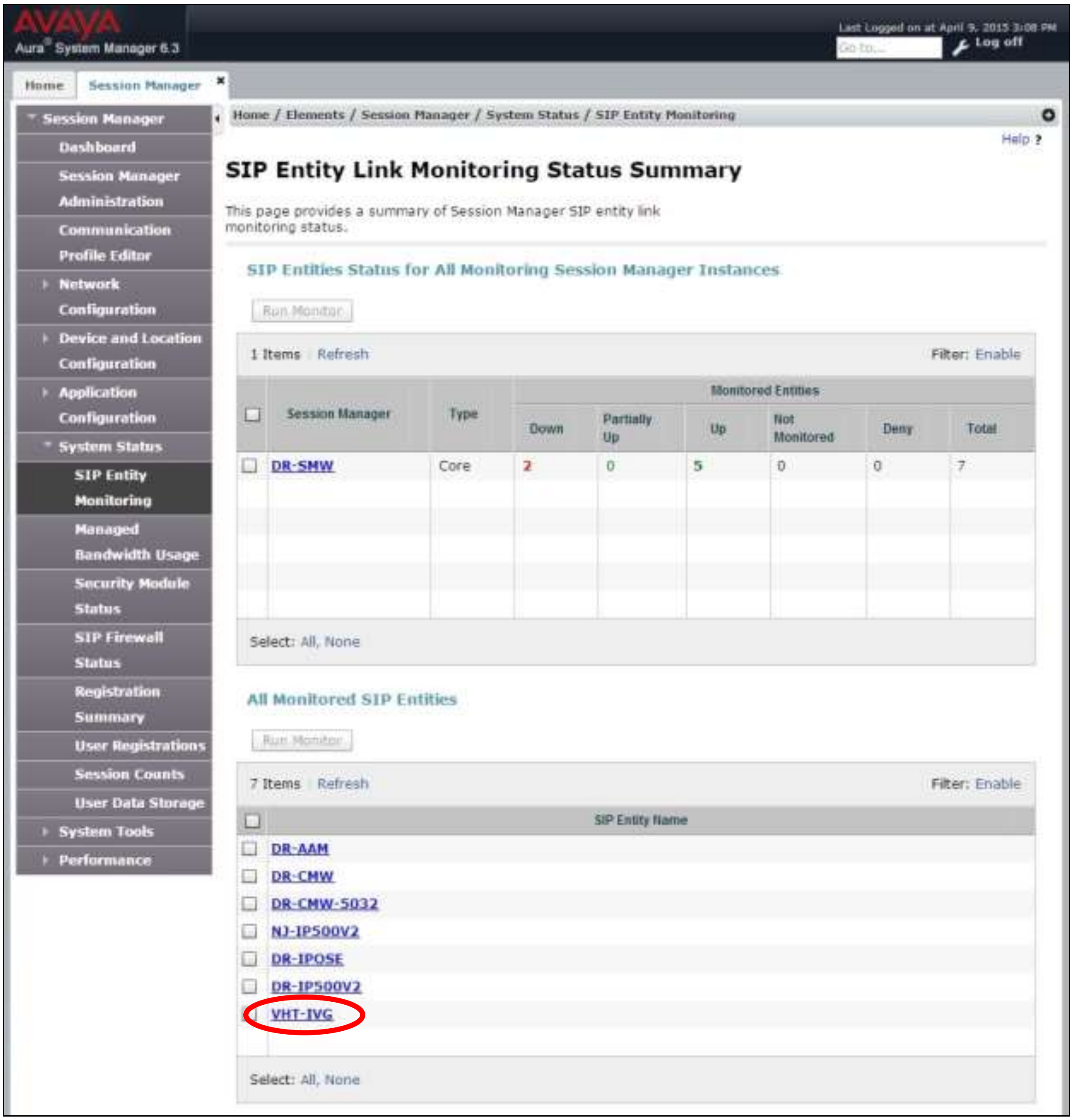
Verify the **Status** is “Talking” for the TSAPI link administered in **Section 6.3**, and that the **Associations** column reflects the number of VDNs from **Section 5.4**.



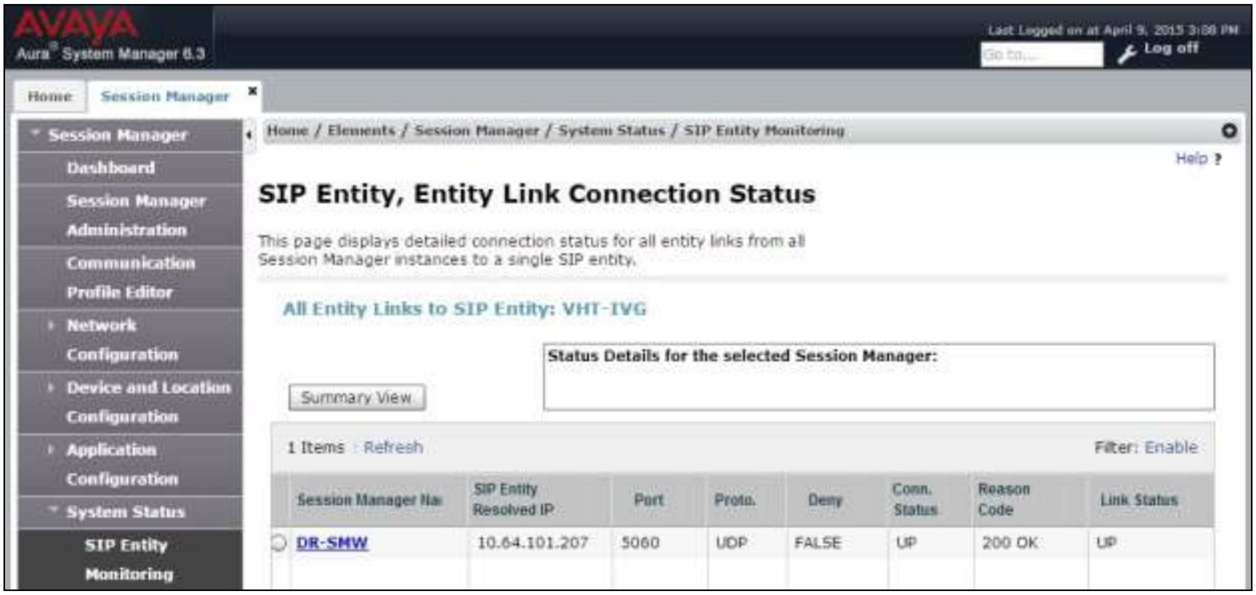
10.3. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the IVG entity name from **Section 7.3.1**.

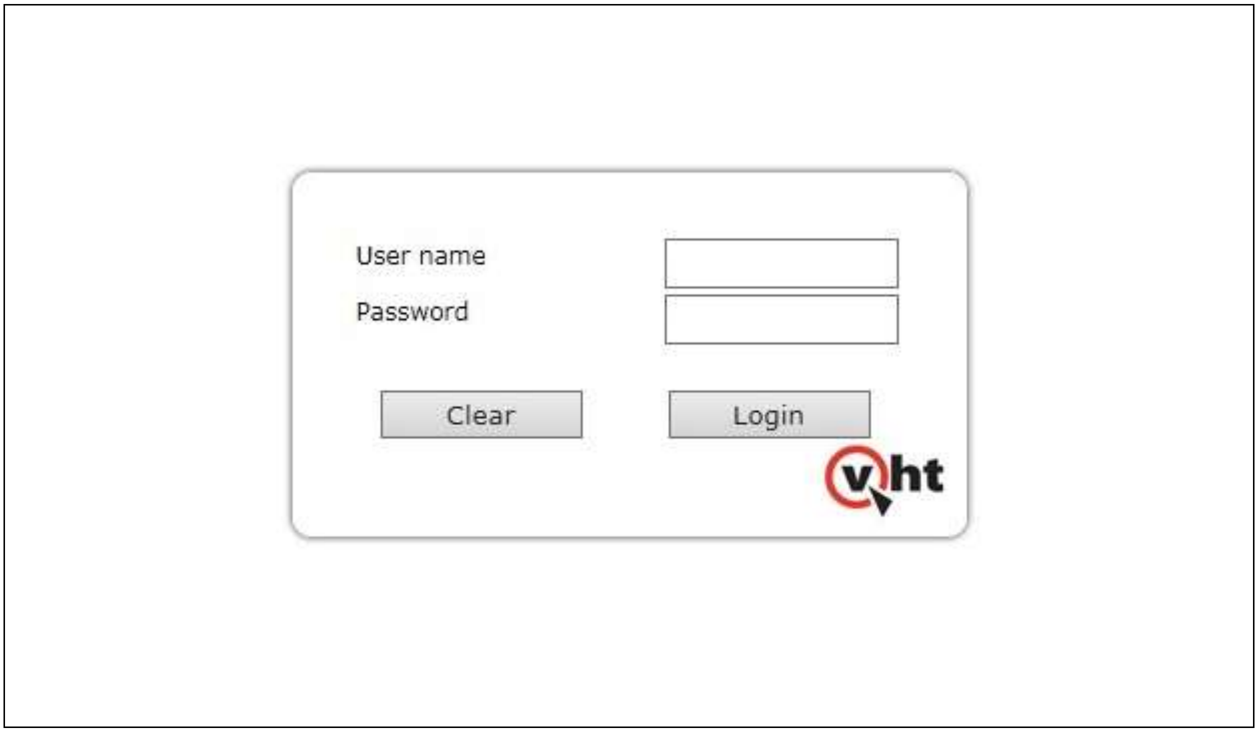


The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn. Status** and **Link Status** are "UP", as shown below.

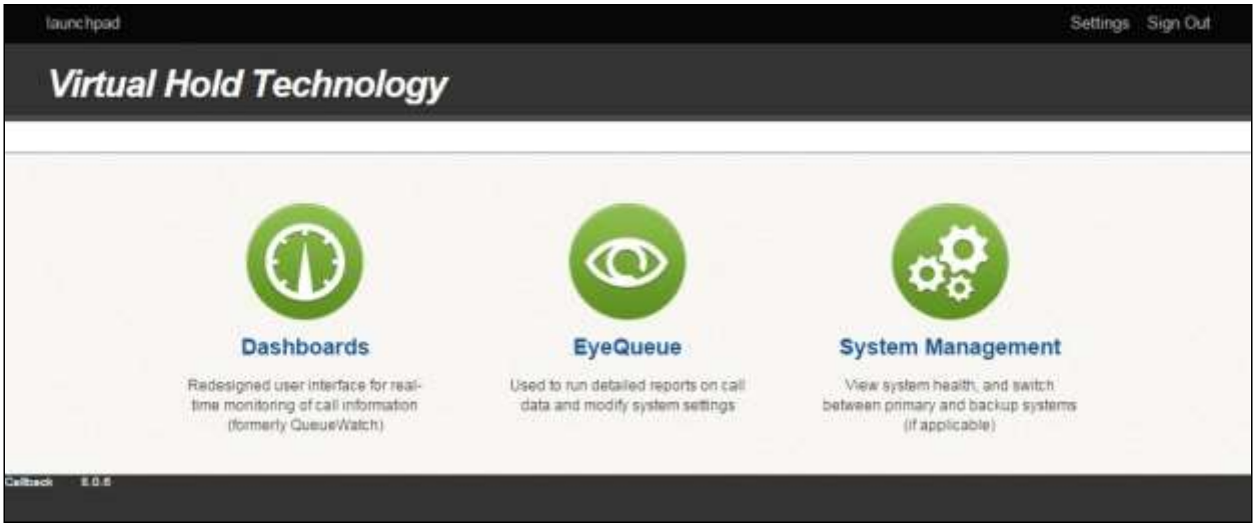


10.4. Verify VHT Callback and IVG

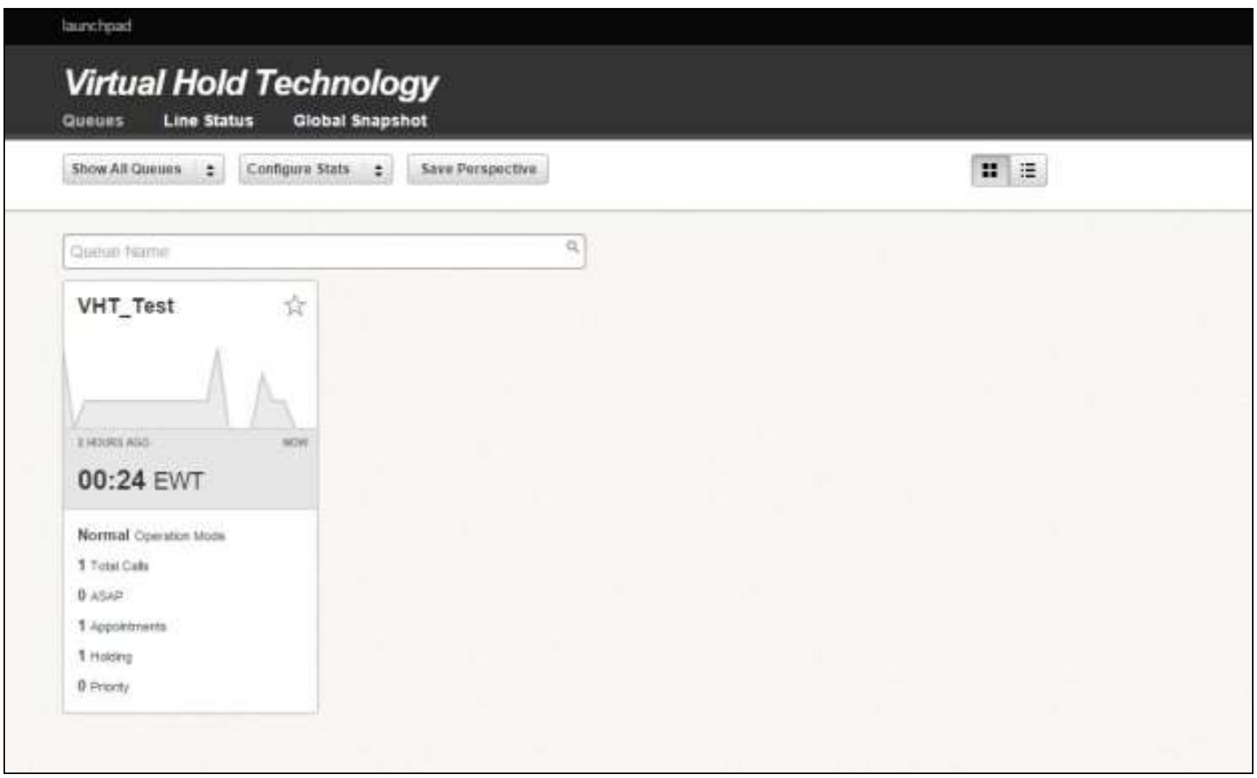
Access the Callback web-based EyeQueue application by using the URL "http://ip-address/ EyeQueue" in an Internet browser window, where "ip-address" is the IP address of the Callback server. Log in using the appropriate credentials.



The screen below is displayed. Select **Dashboards**.



Make a few incoming ACD calls, with active call at the agent, call optioned to stay in queue, and call scheduled for callback. Verify that the screen is updated reflecting proper active calls and expected wait time (EWT), as shown below.



11. Conclusion

These Application Notes describe the configuration steps required for VHT Callback 8.0 to successfully interoperate with Avaya Aura® Communication Manager 6.3, Avaya Aura® Application Enablement Services 6.3, and Avaya Aura® Session Manager 6.3. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

12. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 10, Release 6.3, June 2014, available at http://support.avaya.com.

2. *Avaya Aura® Application Enablement Services Administration and Maintenance Guide*, Release 6.3, 02-300357, June 2014, available at http://support.avaya.com.

3. *Virtual Hold Deployment Guide,* Version 8.0.6, available upon request to Virtual Hold Support.

4. *Interactive Voice Gateway (IVG) Configuration Guide,* Version 1.1.0, available upon request to Virtual Hold Support.