Avaya Solution & Interoperability Test Lab

Application Notes for Configuring VHT Virtual Hold using Native TSAPI Interface with Avaya Aura® Experience Portal, Avaya Aura® Application Enablement Services, Avaya Aura® Session Manager, and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate VHT Virtual Hold with Avaya Aura® Experience Portal, Avaya Aura® Application Enablement Services, Avaya Aura® Session Manager, and Avaya Aura® Communication Manager.

VHT Virtual Hold is a contact center solution that calculates the estimated wait time for an incoming call and maintains the caller's position in a virtual queue. VHT Virtual Hold can call the user back and connect to an agent when the caller's turn comes up. The integration with Avaya Aura® Experience Portal is achieved through an inbound and an outbound VXML application. The integration with Avaya Aura® Communication Manager is achieved through Native TSAPI Interface and the Avaya Aura® Application Enablement Service TSAPI service for event monitoring and adjunct routing support. Calls to Virtual Hold VXML applications are routed using H.323 connections from Avaya Aura® Communication Manager or using SIP connections from Avaya Aura® Communication Manager via Avaya Aura® Session Manager.

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.

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1. Introduction

These Application Notes describe the configuration steps required to integrate VHT Virtual Hold with Avaya Aura® Experience Portal, Avaya Aura® Application Enablement Services (AES), Avaya Aura® Session Manager, and Avaya Aura® Communication Manager.

VHT Virtual Hold is a contact center intelligent queue management solution that calculates the Estimated Wait Time (EWT) for an incoming call and maintains the caller's position in a virtual queue. VHT Virtual Hold can call the user back and connect to an agent when the caller's turn comes up. VHT Virtual Hold consists of Virtual Hold Queue Manager and Virtual Hold VXML Interaction Server (VIS). Virtual Hold Queue Manager is responsible for making routing decisions and maintaining the virtual queue. Virtual Hold VXML Interaction Server allows for Avaya Aura® Experience Portal supported VXML applications, developed by VHT for inbound and outbound calls, and is responsible for interactions with Avaya Aura® Experience Portal. The integration with Avaya Aura® Communication Manager is achieved through Native TSAPI Interface and the AES TSAPI service for event monitoring and adjunct routing support.

As calls come into the contact center, VHT Virtual Hold monitors the EWT and determines how calls are treated. If the EWT is less than the turn-on threshold, the calls are routed to a queue, as normal, to be answered by an agent. If the EWT is more than the turn-on threshold, the callers are offered several options. One option is to save the caller's places in line and call back when it is their turn. Another option is to stay in the queue to wait being answered by an agent. The third option is to receive a callback at a later time chosen by the caller. If the first option is chosen, the caller provides phone number and name and then hangs up. When it is nearly the caller's turn in queue, VHT Virtual Hold calls the caller back, verifies that the caller is on the line, and transfers the call to the agent queue at high priority, which makes the call the next one to be answered by an agent.

VHT Virtual Hold uses a Native TSAPI Interface element to interact with the Avaya Aura® Application Enablement Services' TSAPI service to query and monitor the agent's state and service speed, and uses the provided CTI event reports to calculate the EWT. Incoming calls are routed to the inbound VXML application via Avaya Aura® Experience Portal, where VHT Virtual Hold can play the EWT to the caller and provide the caller with options. Virtual Hold VXML Interaction Server uses the Application Interface Web Service provided by Avaya Aura® Experience Portal to launch the outbound VXML application and send callback requests.

Calls to Virtual Hold VXML applications are routed using H.323 connections from Avaya Aura® Communication Manager or using SIP connections from Avaya Aura® Communication Manager via Avaya Aura® Session Manager.

2. General Test Approach and Test Results

This section describes the compliance test approach, test coverage, test results, and the support information.

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 purpose of this compliance testing was to verify interoperability between Virtual Hold and Avaya products including Experience Portal, Application Enablement Services, Session Manager, and Communication Manager.

The testing was performed on two configurations.

- Experience Portal and Communication Manager connected via H.323 connections
- Experience Portal, Session Manager, and Communication Manager connected via SIP connections

Different sets of VDNs/vectors in Communication Manager were used to support the two configurations. The UUI (User to User Information) feature was available and tested only in the SIP configuration.

The interoperability compliance test included events, feature and, serviceability testing.

- The event testing used internal logs to verify receiving and proper handling of CTI events by Virtual Hold.
- The feature testing entailed placing calls manually from a PSTN phone to Experience Portal and verifying the following:
    - Adjunct route by Virtual Hold
    - Virtual Hold VXML applications launch
    - Experience Portal using SIP and H.323 as VoIP Connections.
    - Experience Portal Call Detail Report and Alarm/Warning generation.
    - Virtual Hold playing Estimated Wait Time
    - Virtual Hold handling of caller options including callback, scheduled callback, and staying in queue.
    - Virtual Hold storing and passing UUI in callback calls (SIP Configuration only).
- The serviceability testing focused on verifying the ability of Experience Portal and Virtual Hold to recover after a network outage or server reboot.

2.2. Test Results

All test cases were executed. The following observations were made:

- Virtual Hold did not support the Retrieved event when a held call was resumed.
- With the SIP configuration, a UUI that was part of an original call was not passed in the callback call. This functionality will be fixed in VIS 4.3.1.
- With the SIP configuration, a call to the Entry VDN (see **Section 4**, **Step 8** for definition) received a reorder tone if no MPP is available on Experience Portal.

2.3. Support

To obtain technical support for Virtual Hold:

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

3. Reference Configuration

The diagram below illustrates the test configurations.

For this test effort, two different configurations were tested:

- Experience Portal and Communication Manager connected via H.323 connections
- Experience Portal and Communication Manager connected via SIP connections using Session Manager

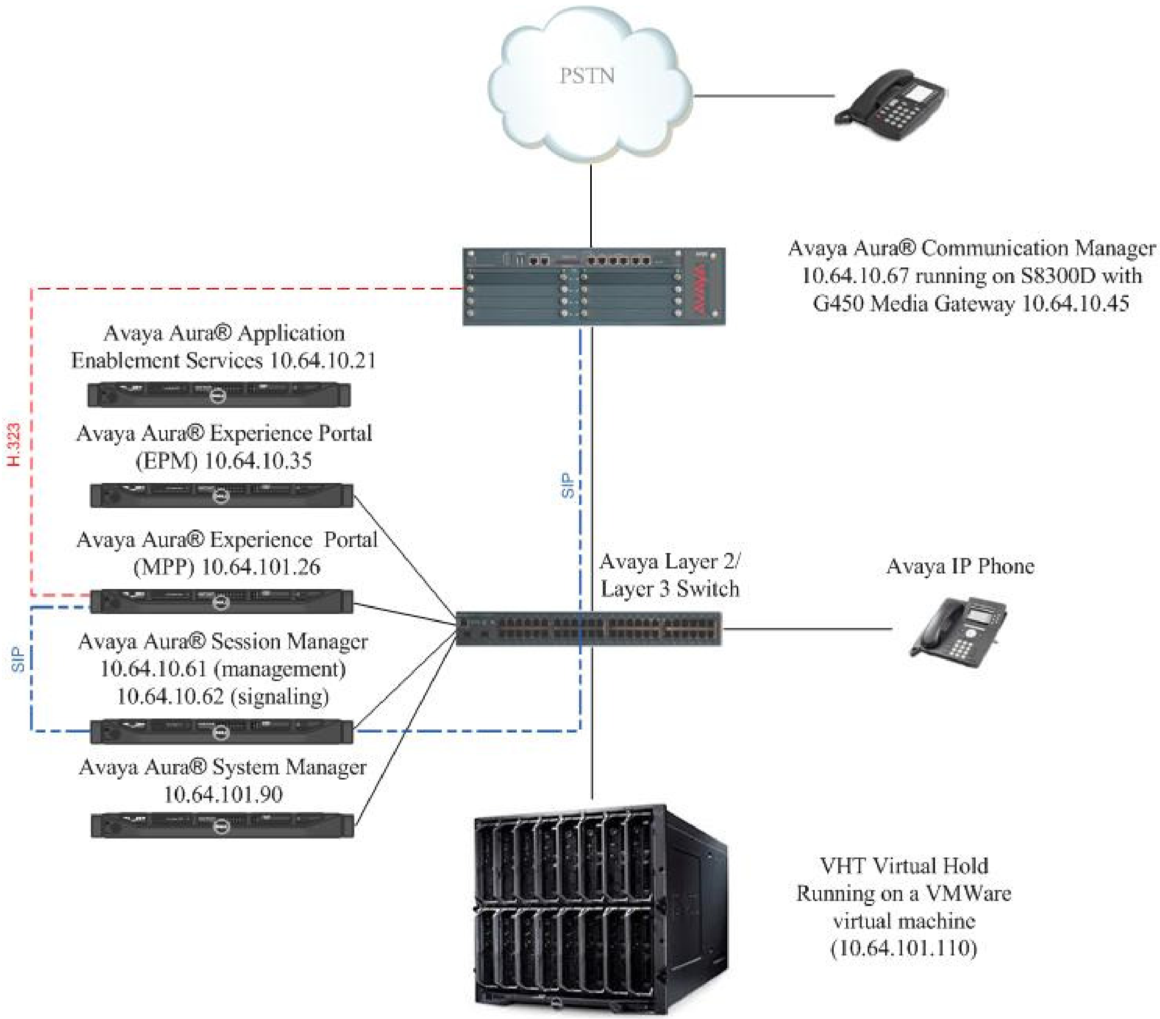


Figure 1: Test Configuration

3.1. Equipment and Software Validated

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

Equipment/Software	Version
Avaya Aura® Experience Portal running on HP Proliant DL360 G7 Server	6.0 SP2
Avaya Aura® Communication Manager running on Avaya S8300 Server	6.3 patch 20553
Avaya G450 Media Gateway MGP MM710 T1 Module	 HW 1 FW 31.20.0 HW 04 FW 015
Avaya Aura® Session Manager running on HP Proliant DL360 G7 Server	6.3.2.0.632023
Avaya Aura® System Manager running on a VMWare virtual machine	6.3.0 FP2
Avaya Aura® Application Enablement Services running on Dell PowerEdge R610 server	6.3
Virtual Hold Server running on a VMware host with Windows 2008 Server R2 SP1 64-bit Operating System Queue Manager VXML Interaction Server Native TSAPI Interface	 7.6.6 R6 4.2.2 R2 6.20.100.1362

4. Configure Avaya Aura® Communication Manager

This section describes the Communication Manager configuration for supporting the VHT Virtual Hold solution. Certain subsections apply only to the H.323 configuration or the SIP configuration and will be noted accordingly.

It is assumed that the following administration is already in place and will not be described in this section.

- SIP trunk group to Session Manager
- Route Pattern that maps to the SIP trunk group

The configuration of Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent.

The configuration procedures fall into the following areas:

- Verify Communication Manager Licenses
- Configure System Parameters Features
- Configure Cti-link
- Configure H.323 Connections to Experience Portal
- Configure Hunt Group for Contact Center Agents
- Configure VDNs and Vectors for H.323 Configuration
- Configure Automatic Alternate Routing (AAR)
- Configure VDNs and Vectors for SIP Configuration
- Configure Converse Data Return Feature Access Code
- Configure UUI Treatment for Trunk Group

Step	Description
1.	**Communication Manager Licenses** Verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the **display system-parameters customer-options** command to verify that the **Computer Telephony Adjunct Links** option is set to **y** on **Page 3**. If this option is not set to **y**, then contact the Avaya sales team or business partner for a proper license file.

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

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

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

```
change 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? y
         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
                   Forced ACD Calls? n    Vectoring (ANI/II-Digits Routing)? y
               Least Occupied Agent? y    Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? y                    Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y     Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? y                 Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? y                Vectoring (Variables)? y
```

Step	Description
2.	**System-Parameters Features** Enter the **change system-parameters features** command and navigate to **Page 5**. Set the **Create Universal Call ID (UCID)** field to **y.**

```
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: 1
```

On **Page 13,** set the **Send UCID to ASAI** field to **y.**

```
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
          Service Level Algorithm for SLM: actual
    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? n
                                  Send UCID to ASAI? y
          For ASAI Send DTMF Tone to Call Originator? y
  Send Connect Event to ASAI For Announcement Answer? n
```

Step	Description

3.	**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 1                                        Page   1 of   3`
	`                                    CTI LINK`
	` CTI Link: 1`
	`Extension: 6201`
	`     Type: ADJ-IP`
	`                                                              COR: 1`
	`     Name: TSAPI`

Step	Description
4.	**H.323 Connections to Experience Portal (H.323 Configuration only)** To create H.323 connections to Experience Portal requires the following steps: • Create a Auto-Available Skill (AAS) • Create a number of H.323 IP stations with station type 7434ND • Create a number of agents that are tied in with the above H.323 IP stations and registered to the Auto-Available Skill **Create Auto-Available Skill:** Use the **add hunt-group n** command to create an AAS skill, where **n** is an available hunt group number (e.g. **55**). On **Page 1**, enter a descriptive name in the **Group Name** field and an available extension in the **Group Extension** field. Set **ACD**, **Queue,** and **Vector** fields to **y**.

```
add hunt-group 55                                              Page   1 of   4
                                 HUNT GROUP

            Group Number: 55                              ACD? y
              Group Name: VH H323                       Queue? y
         Group Extension: 62155                        Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                  MM Early Answer? n
           Security Code:             Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

On **Page 2**, set the **Skill** field to **y**.

```
add hunt-group 55                                              Page   2 of   4
                                 HUNT GROUP

                   Skill? y      Expected Call Handling Time (sec): 180
                     AAS? y        Service Level Target (% in sec): 80 in 20
                Measured: both
     Supervisor Extension:


      Controlling Adjunct: none


        VuStats Objective:

   Multiple Call Handling: none
```

Continue on next page.

Step	Description
	Create H.323 IP Stations: Use the **add station n** command, where **n** is a valid unused station number (e.g. **25520**). On **Page 1**, • Set the **Type** field to **7434ND** • Set the **Port** field to **IP** • Enter a descriptive name in the **Name** field • Enter a **Security Code**, which will later be used by Experience Portal. • Set the **Display Module** field to **y** • Set the **IP Softphone** field to **y**

```
add station 25520                                              Page   1 of   6
                                     STATION

Extension: 25520                         Lock Messages? n               BCC: 0
     Type: 7434ND                        Security Code:                  TN: 1
     Port: IP                          Coverage Path 1:                 COR: 1
     Name: AEP Station                 Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                     Message Lamp Ext: 25520
          Display Module? y
        Display Language: english                Coverage Module? n

          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y
                                               Remote Office Phone? n
                                                IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default
```

On **Page 2**, set the **Multimedia Mode** field to **enhanced**.

```
add station 25520                                              Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                      EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
     MWI Served User Type:                      Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 25520         Always Use? n IP Audio Hairpinning? y
```

Continue on the next page.

Step	Description
	On **Page 6**, set the **#1 Display Button** field to **normal**. Please note that the administration of the **Display Module** field on **Page 1** and the **#1 Display Button** field on **Page 6** are required in order for the Routing VDN (See **Step 6** for definition) number to be passed to Experience Portal. `add station 25520` Page 6 of 6 STATION DISPLAY BUTTON ASSIGNMENTS **1: normal** 2: Repeat this step to add station 25521 with the same Security Code. **Create AAS Agents:** For each H.323 IP station created, add an Auto Answer agent using **add agent-loginID n**, where **n** is an available agent ID (e.g. **5520**), and set the **Port Extension** field to the H.323 IP station extension. On **Page 1**, • Set the **AAS** field to **y** • Enter a **Security Code** • Set the **Port Extension** field to the corresponding H.323 IP station extension created in this step. **add agent-loginID 5520** Page 1 of 2 AGENT LOGINID **Login ID: 5520** **AAS? y** Name: VH AUDIX? n TN: 1 LWC Reception: spe COR: 1 LWC Log External Calls? n Coverage Path: AUDIX Name for Messaging: **Security Code:** **Port Extension: 25520** LoginID for ISDN/SIP Display? n Auto Answer: station MIA Across Skills: system ACW Agent Considered Idle: system Aux Work Reason Code Type: system On **Page 2**, **line 1**, set the **SN field** to the hunt group created earlier in this step. Set the **SL** field to **1**. **add agent-loginID 5520** Page 2 of 2 AGENT LOGINID Direct Agent Skill: Service Objective? n Call Handling Preference: skill-level Local Call Preference? n SN RL SL SN RL SL **1: 55 1** 16: Repeat this step to add agent 5521.

Step	Description
5.	**Create Hunt-Group for Contact Center Agents** Administer a hunt group for Call Center Agents by using the **add hunt-group n** command, where **n** is an available hunt group number. On **Page 1**, enter a descriptive name in the **Group Name** field and an available extension in the **Group Extension** field. Set **ACD**, **Queue,** and **Vector** fields to **y**.

```
add hunt-group 51                                                Page   1 of   4
                                  HUNT GROUP

            Group Number: 51                                ACD? y
              Group Name: Skill 51                        Queue? y
         Group Extension: 62151                          Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                    MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

On **Page 2**, set the **Skill** field to **y**.

```
add hunt-group 51                                                Page   2 of   4
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n        Service Level Target (% in sec): 80 in 20
                 Measured: both
     Supervisor Extension:


      Controlling Adjunct: none


        VuStats Objective:

   Multiple Call Handling: none


 Timed ACW Interval (sec):          After Xfer or Held Call Drops? n
```

For the compliance testing, two agents with extensions 25004 and 25005 and agent Login ids 2504 and 2505 were configured as available agents for the above hunt group.

```
list agent-loginID 2504 count 2

                              AGENT LOGINID
Login ID     Name           Extension     Dir Agt  AAS/AUD        COR Ag Pr SO
               Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv Skil/Lv

2504         IP Agent 4     25004                                  1   lvl
                 51/01     /       /       /       /       /       /       /
2505         IP Agent 5     25005                                  1   lvl
                 51/01     /       /
```

Step	Description
6.	**VDNs and Vectors for H.323 Configuration (H.323 Configuration only)**

Administer a set of Vector Directory Numbers (VDNs) and vectors as follows:

- Entry VDN/vector: To perform adjunct route with the Virtual Hold Queue Manager.
- Routing VDN/vector: To perform converse-on function with the Virtual Hold Queue Manager.
- Holding VDN/vector: To queue incoming calls to the agent skill at medium priority.
- Callback VDN/vector: To queue callback calls to the agent skill at high priority.

Entry VDN and Vector

Modify an available vector using the **change vector n** command, where **n** is an existing vector number.

Following configuration was used during compliance testing.

```
change vector 100                                              Page   1 of   6
                                  CALL VECTOR

    Number: 100              Name: VH H323 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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 adjunct      routing link 1
03 wait-time    5   secs hearing ringback
04 route-to     number 62102            with cov n if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 the **Destination** field. Retain the default values for all remaining fields.

```
add vdn 62100                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 62100
                                 Name*: VH H323 Entry
                           Destination: Vector Number        100
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? y
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Continue on the next page.

Step	Description
	Routing VDN and Vector Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 101                                               Page   1 of   6
                                  CALL VECTOR

    Number: 101              Name: VH H323 Routing
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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 converse-on  skill 55   pri m passing none   and none
03 collect      1    digits after announcement none     for none
04 goto step    6             if digits           =      1
05 route-to     number 62102            with cov n if unconditionally
06 disconnect   after announcement none
07 stop
08
```

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 the **Destination** field. Retain the default values for all remaining fields.

```
add vdn 62101                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 62101
                                 Name*: VH H323 Routing
                           Destination: Vector Number        101
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? y
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Continue on the next page.

Step	Description
	Holding VDN and Vector Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 102                                               Page   1 of   6
                                    CALL VECTOR

    Number: 102              Name: VH H323 Holding
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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 queue-to     skill 51   pri m
03 wait-time    30  secs hearing ringback
04 goto step    3             if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 the **Destination** field. Retain the default values for all remaining fields.

```
add vdn 62102                                                   Page   1 of   3
                              VECTOR DIRECTORY NUMBER

                               Extension: 62102
                                   Name*: VH H323 Holding
                             Destination: Vector Number        102
                     Attendant Vectoring? n
                    Meet-me Conferencing? n
                      Allow VDN Override? n
                                     COR: 1
                                     TN*: 1
                                Measured: none
```

Continue on the next page.

Step	Description
	Callback VDN and Vector Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 103                                               Page   1 of   6
                                  CALL VECTOR

    Number: 103              Name: VH H323 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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 queue-to     skill 51   pri h
03 wait-time    30  secs hearing ringback
04 goto step    3             if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 **the Destination** field. Retain the default values for all remaining fields.

```
add vdn 62103                                                   Page   1 of   3
                              VECTOR DIRECTORY NUMBER

                             Extension: 62103
                                 Name*: Vh H323 Callback
                           Destination: Vector Number        103
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Step	Description
7.	**Automatic Alternate Routing (AAR) (SIP Configuration only)** For the compliance test, AAR was used to route calls to Experience Portal via a SIP trunk to Session Manager. A Route Pattern 10 was pre-configured to use Trunk Group 10, which is a SIP trunk connected to Session Manager. For the compliance test, use the **change aar analysis** command to add an entry to AAR table as follows:. • Enter **257** in the **Dialed String** field. • Enter **5** and **5** to the **Total Min** and **Total Max** fields. • Enter **10** to the **Route Pattern** field. • Enter **aar** in the **Call Type** field. With the above entry, all calls with dialed digits of 257xx will be routed over Trunk Group 10 to Session Manager. In the compliance test, extension 25798 is associated with the Experience Portal inbound application.

```
change aar analysis 257                                      Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    257                   5    5      10        aar          n
```

Step	Description
8.	**VDNs and Vectors for SIP Configuration (SIP Configuration only)** Administer a set of Vector Directory Numbers (VDNs) and vectors as follows: • Entry VDN/vector: To perform adjunct route with the Virtual Hold Queue Manager • Holding VDN/vector: To queue incoming calls to the agent skill at medium priority. • Callback VDN/vector: To queue callback calls to the agent skill at high priority. **Entry VDN and Vector** Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 110                                               Page   1 of   6
                                  CALL VECTOR

    Number: 110              Name: VH SIP 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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 adjunct      routing link 1
03 wait-time    10  secs hearing ringback
04 route-to     number 62111            with cov n if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 62110                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 62110
                                 Name*: VH SIP Entry
                           Destination: Vector Number        110
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Continue on the next page.

Step	Description
	Holding VDN and Vector Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 111                                               Page   1 of   6
                                  CALL VECTOR

    Number: 111              Name: VH SIP Holding
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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 queue-to     skill 51   pri m
03 wait-time    30  secs hearing ringback
04 goto step    3             if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 the **Destination** field. Retain the default values for all remaining fields.

```
add vdn 62111                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 62111
                                 Name*: VH SIP Holding
                           Destination: Vector Number        111
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Continue on the next page.

Step	Description
	Callback VDN and Vector Modify an available vector using the **change vector n** command, where **n** is an existing vector number. Following configuration was used during compliance testing.

```
change vector 112                                               Page   1 of   6
                                  CALL VECTOR

    Number: 112             Name: VH SIP 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? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 queue-to     skill 51   pri h
03 wait-time    30  secs hearing ringback
04 goto step    3             if unconditionally
05 disconnect   after announcement none
06 stop
07
```

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 the **Destination** field. Retain the default values for all remaining fields.

```
add vdn 62112                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 62112
                                 Name*: VH SIP Callback
                           Destination: Vector Number        112
                  Attendant Vectoring? n
                 Meet-me Conferencing? n
                   Allow VDN Override? n
                                  COR: 1
                                  TN*: 1
                             Measured: none
```

Step	Description

9.	**Feature Access Code (H.323 Configuration only)**
	Enter the **change feature-access-codes** command. On **Page 7**, set the **Converse Data Return Code** field to **#12**.
	
	<pre>change feature-access-codes Page 7 of 10
	                               FEATURE ACCESS CODE (FAC)
	
	                         Call Vectoring/Prompting Features
	
	   Converse Data Return Code: #12
	
	Vector Variable 1 (VV1) Code:
	Vector Variable 2 (VV2) Code:
	Vector Variable 3 (VV3) Code:
	Vector Variable 4 (VV4) Code:
	Vector Variable 5 (VV5) Code:
	Vector Variable 6 (VV6) Code:
	Vector Variable 7 (VV7) Code:
	Vector Variable 8 (VV8) Code:
	Vector Variable 9 (VV9) Code:</pre>

10.	**UUI Treatment for SIP Trunk Group (SIP Configuration only)**
	Enter the **change trunk-group n** command where **n** is the trunk group number of the SIP trunk to Session Manager. Set the **UUI Treatment** field to **shared** and **Send UCID** field to **yes**.
	
	<pre>change trunk-group 10 Page 3 of 22
	TRUNK FEATURES
	          ACA Assignment? n Measured: none
	                                                          Maintenance Tests? y
	
	                     Numbering Format: private
	                                                UUI Treatment: shared
	                                              Maximum Size of UUI Contents: 128
	                                                 Replace Restricted Numbers? n
	                                                Replace Unavailable Numbers? n
	
	                                Modify Tandem Calling Number: no
	               Send UCID? y
	
	 Show ANSWERED BY on Display? y</pre>

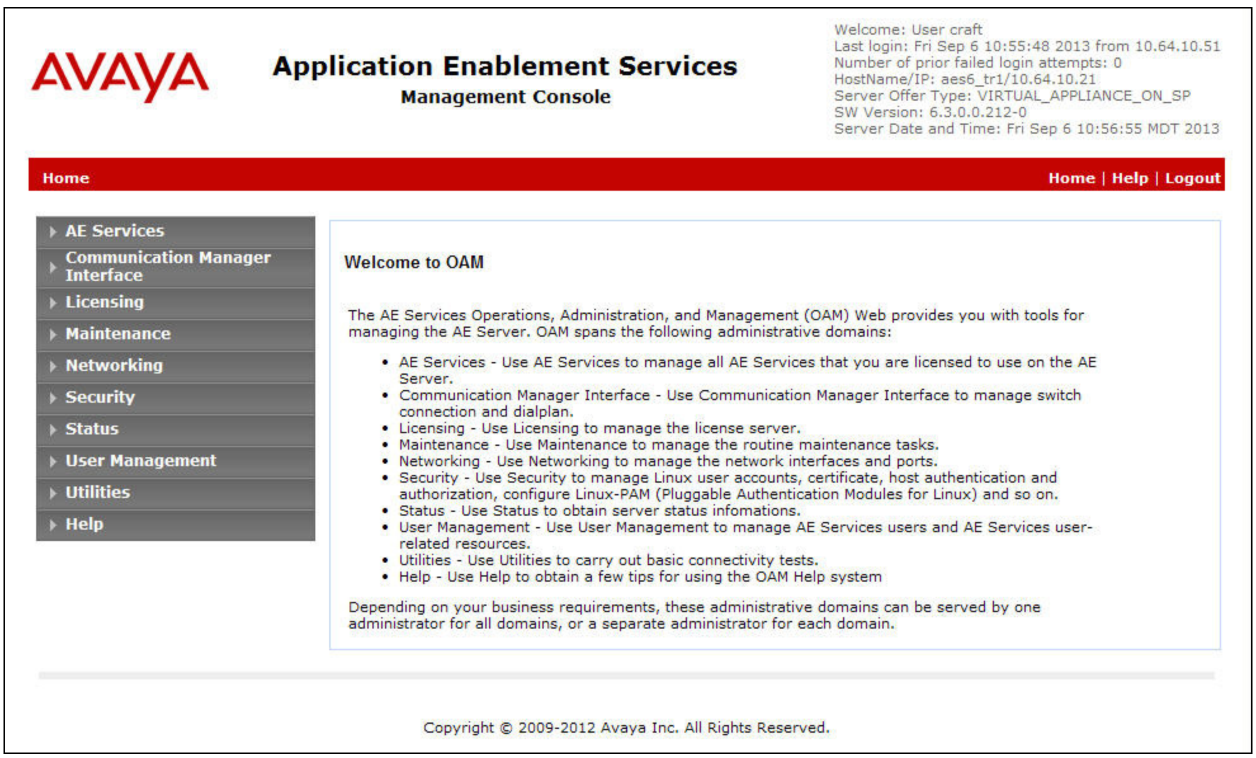
5. Configure Avaya Aura® Application Enablement Services

The configuration of Application Enablement Services is performed via a web browser. Enter **https://<ip-addr>** in the URL field of a web browser where **<ip-addr>** is the IP address of the Application Enablement Services server. After a login step, the **Welcome to OAM** page is displayed.

The configuration procedures fall into the following areas:

- Confirm TSAPI Licenses
- Add TSAPI Links
- Note the Tlink Information
- Restart TSAPI Service
- Configure Virtual Hold User
- Enable Unrestricted Access for Virtual Hold User

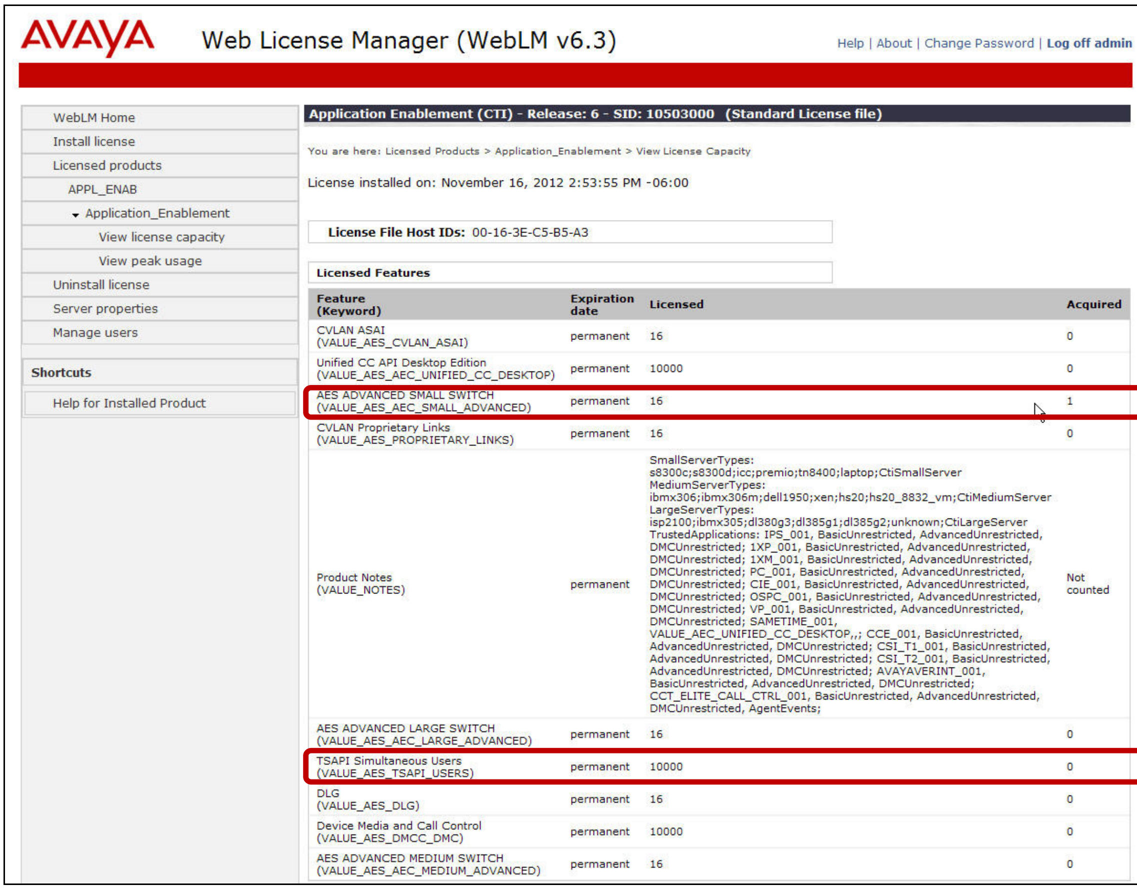
It is assumed that the configuration of a switch connection to Communication Manager is already in place and therefore will not be described here.



Step	Description
1.	**Confirm TSAPI Licenses**

Virtual Hold uses a TSAPI Advanced (**VALUE_AES_AEC_xxxxx_ADVANCED)** license for adjunct routing and a TSAPI Basic (**VALUE_AES_TSAPI_USERS**) license for each VDN being monitored. If the licensed quantities are not sufficient for the implementation, contact the Avaya sales team or business partner for a proper license file.

From the left pane of the Application Enablement Services Management Console, click **Licensing → WebLM Server Access**. A **Web License Manager** login window is displayed. Enter proper credentials to log in. Click **Licensed products → APPL_ENAB → Application_Enablement** from the left pane. The Application Enablement Services license is displayed in the right pane. Ensure that there are enough **VALUE_AES_AEC_xxxxx_ADVANCED** and **VALUE_AES_TSAPI_USERS** licenses available.


AVAYA Web License Manager (WebLM v6.3)

Help | About | Change Password | Log off admin

- WebLM Home
- Install license
- Licensed products
- APPL_ENAB
- Application_Enablement
- View license capacity
- View peak usage
- Uninstall license
- Server properties
- Manage users

Shortcuts

- Help for Installed Product

Application Enablement (CTI) - Release: 6 - SID: 10503000 (Standard License file)

You are here: Licensed Products > Application_Enablement > View License Capacity

License installed on: November 16, 2012 2:53:55 PM -06:00

License File Host IDs: 00-16-3E-C5-B5-A3

Licensed Features

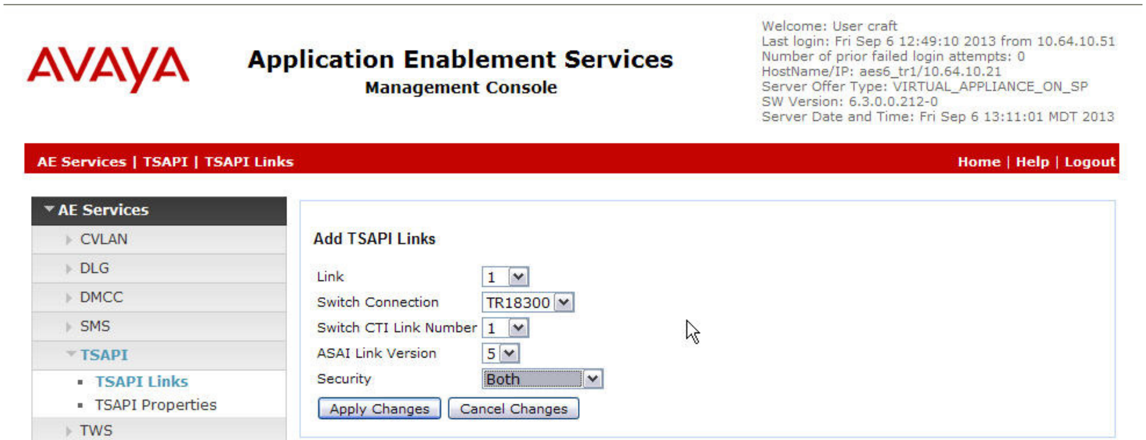
Feature (Keyword)	Expiration date	Licensed	Acquired
CVLAN ASAI (VALUE_AES_CVLAN_ASAI)	permanent	16	0
Unified CC API Desktop Edition (VALUE_AES_AEC_UNIFIED_CC_DESKTOP)	permanent	10000	0
AES ADVANCED SMALL SWITCH (VALUE_AES_AEC_SMALL_ADVANCED)	permanent	16	1
CVLAN Proprietary Links (VALUE_AES_PROPRIETARY_LINKS)	permanent	16	0
Product Notes (VALUE_NOTES)	permanent	SmallServerTypes: s8300c;s8300d;icc;premio;tn8400;laptop;CtiSmallServer MediumServerTypes: ibmx306;ibmx306m;dell1950;xen;hs20;hs20_8832_vm;CtiMediumServer LargeServerTypes: isp2100;ibmx305;dl380g3;dl385g1;dl385g2;unknown;CtiLargeServer TrustedApplications: IPS_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; 1XP_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; 1XM_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; PC_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; CIE_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; OSPC_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; VP_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; SAMETIME_001, VALUE_AEC_UNIFIED_CC_DESKTOP,,; CCE_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; CSI_T1_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; CSI_T2_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; AVAYAVERINT_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted; CCT_ELITE_CALL_CTRL_001, BasicUnrestricted, AdvancedUnrestricted, DMCUnrestricted, AgentEvents;	Not counted
AES ADVANCED LARGE SWITCH (VALUE_AES_AEC_LARGE_ADVANCED)	permanent	16	0
TSAPI Simultaneous Users (VALUE_AES_TSAPI_USERS)	permanent	10000	0
DLG (VALUE_AES_DLG)	permanent	16	0
Device Media and Call Control (VALUE_AES_DMCC_DMC)	permanent	10000	0
AES ADVANCED MEDIUM SWITCH (VALUE_AES_AEC_MEDIUM_ADVANCED)	permanent	16	0

Step	Description
2.	**Add TSAPI Links**

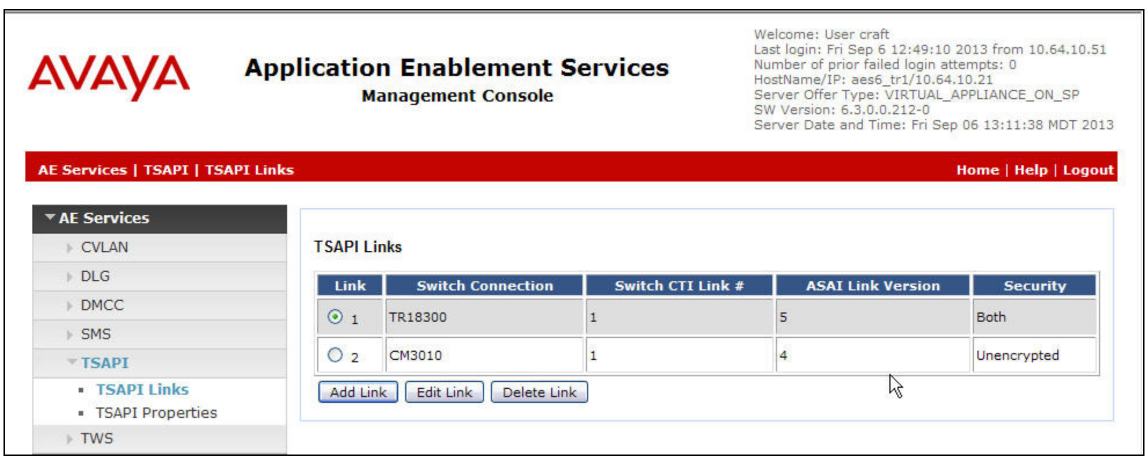
Navigate to the **AE Services → TSAPI → TSAPI Links** page to add a TSAPI Link. Click **Add Link** (not shown).

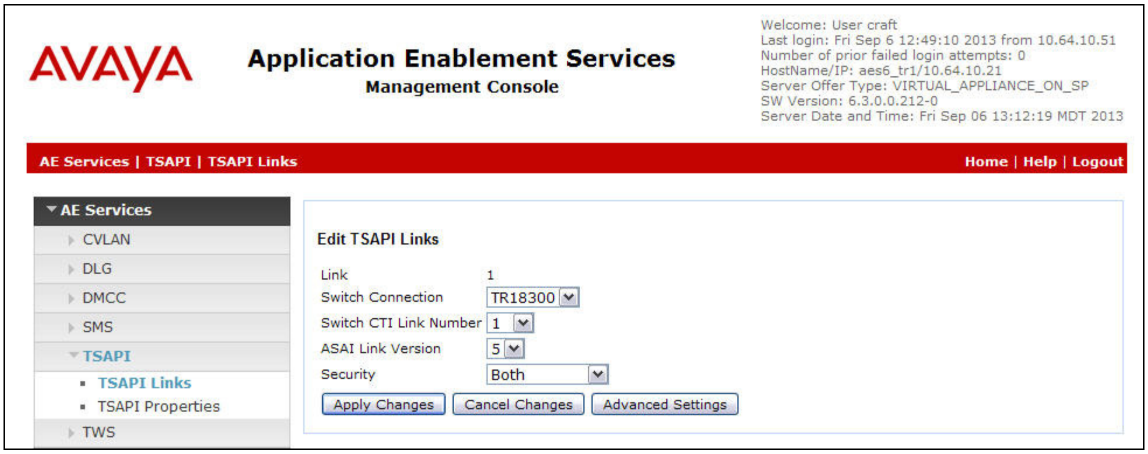
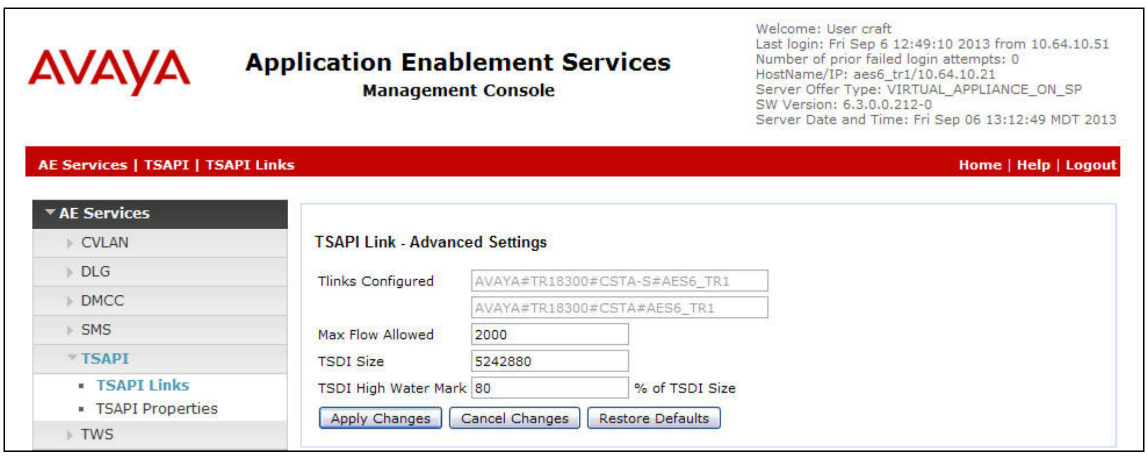
Select a **Switch Connection** and the **Switch CTI Link Number** using the drop down menus. The **Switch CTI Link Number** must match the number configured in the **cti-link** form in **Section 4**, **Step 3**. Select **5** in the **ASAI Link Version** field and **Both** in the **Security** field.

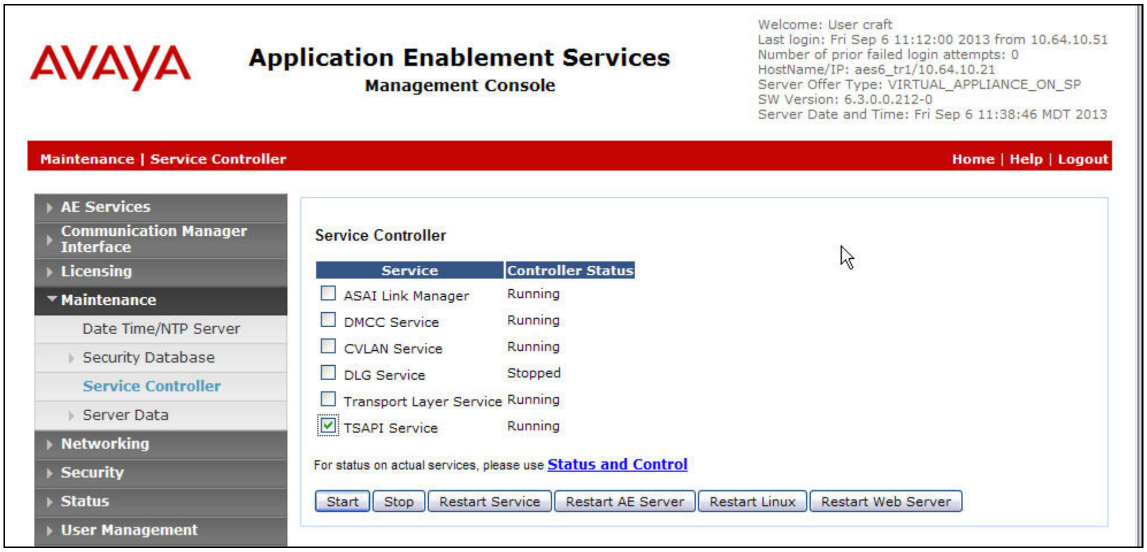
Click **Apply Changes**.



It returns to the **TSAPI Links** screen which shows that the **TR18300** link has been added.



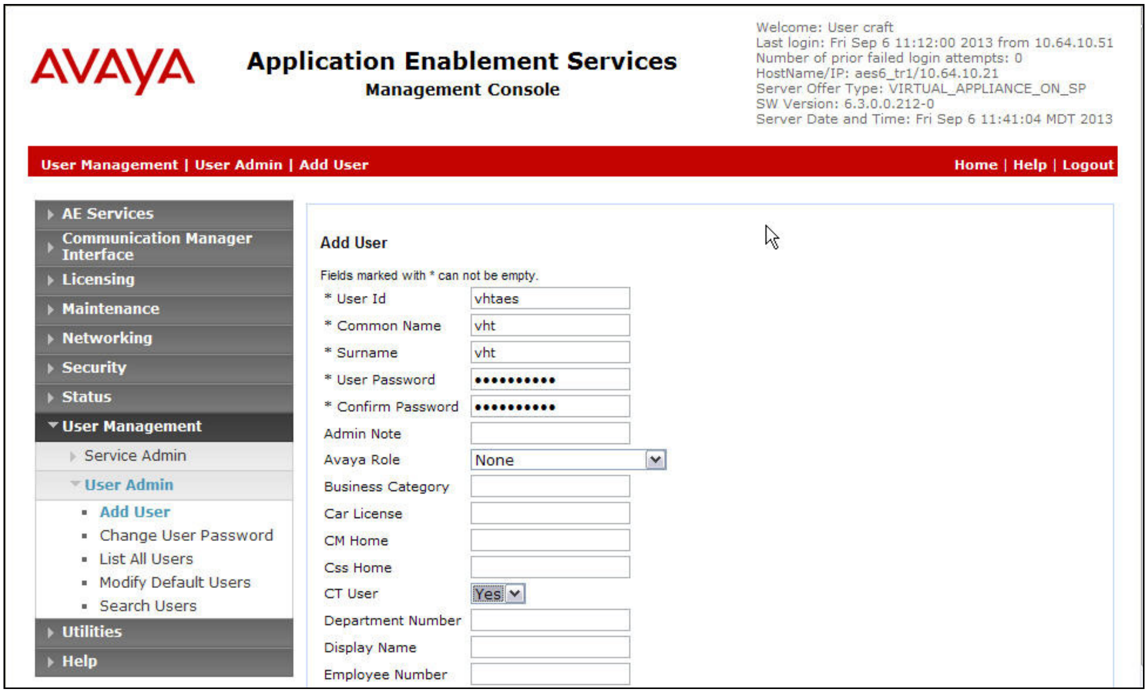
Step	Description
3.	**Note the Tlink Information** Select the **TR18300** TSAPI Link and click **Edit Link**. The **Edit TSAPI Links** screen is displayed.  Click the **Advanced Settings** button. The **TSAPI Link – Advanced Settings** screen is displayed. Note the value in the **Tlinks Configured** field which will be used for Virtual Hold configuration in **Section 8**. 

Step	Description
4.	**Restart TSAPI Service** Select **Maintenance → Service Controller** from the left pane, to display the **Service Controller** screen in the right pane. Check the **TSAPI Service**, and click **Restart Service**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>

Configure Virtual Hold user
In the left pane, select **User Management → User Admin → Add User**. The **Add User** panel will be displayed. Enter an appropriate **User Id, Common Name, Surname,** and **User Password**. Select **Yes** from the **CT User** dropdown list.

Click **Apply** at the bottom of the page (not shown) to save the entry.

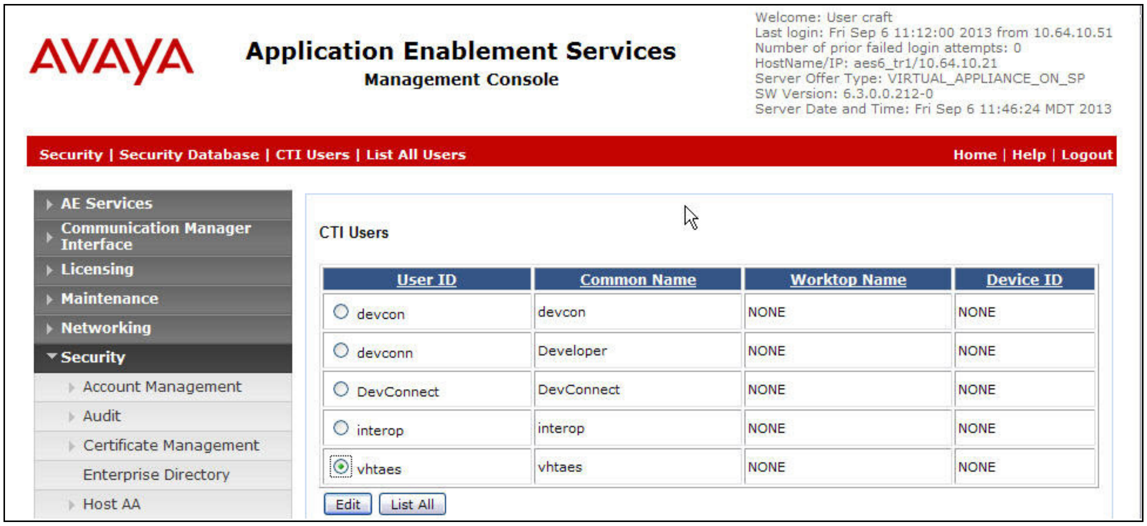


</td></tr>
</table>

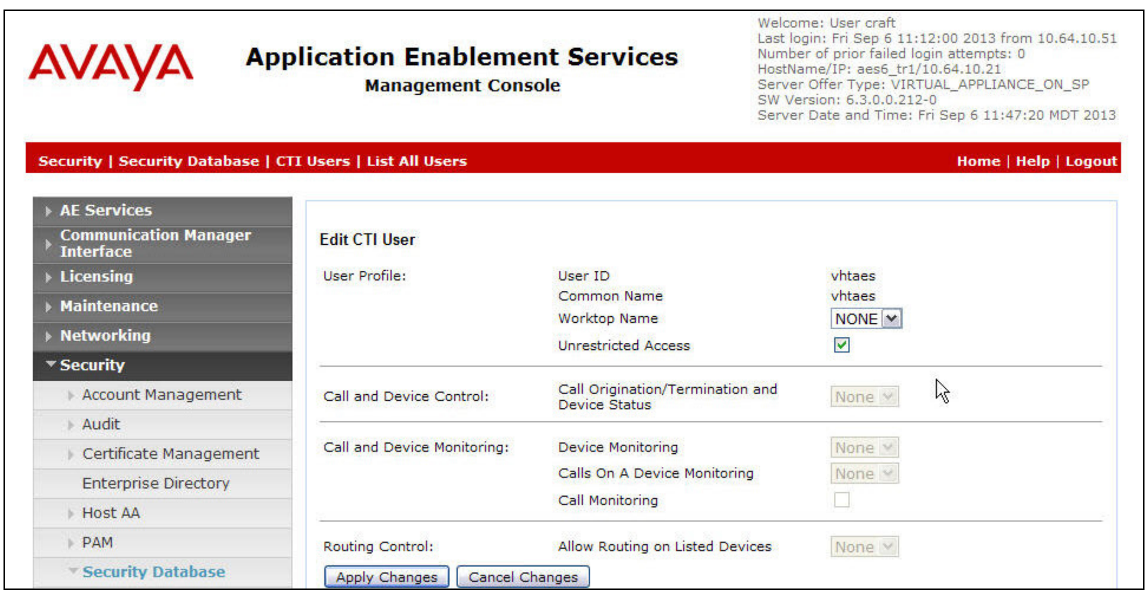
Step	Description
6.	**Enable Unrestricted Access for Virtual Hold User**

If the Security Database (SDB) is enabled on Application Enablement Services, set the Virtual Hold user account to Unrestricted Access to enable access to any device. This step avoids the need to duplicate administration.

Navigate to **Security → Security Database → CTI Users → List All Users**. The **CTI Users** screen is displayed. Select the **vhtaes** user and click **Edit**.



On the **Edit CTI User** page, check the **Unrestricted Access** box and click the **Apply Changes** button. Click **Apply** when asked to confirm the change on the **Apply Changes to CTI User Properties** dialog (not shown).



6. Configure Avaya Aura® Session Manager

This section provides the steps for configuring Session Manager to route calls to Experience Portal. It is assumed that basic administration for Session Manager such as Domain, Locations, and Time Range, as well as the configuration for an entity link to Communication Manager are already in place and therefore will not be described here.

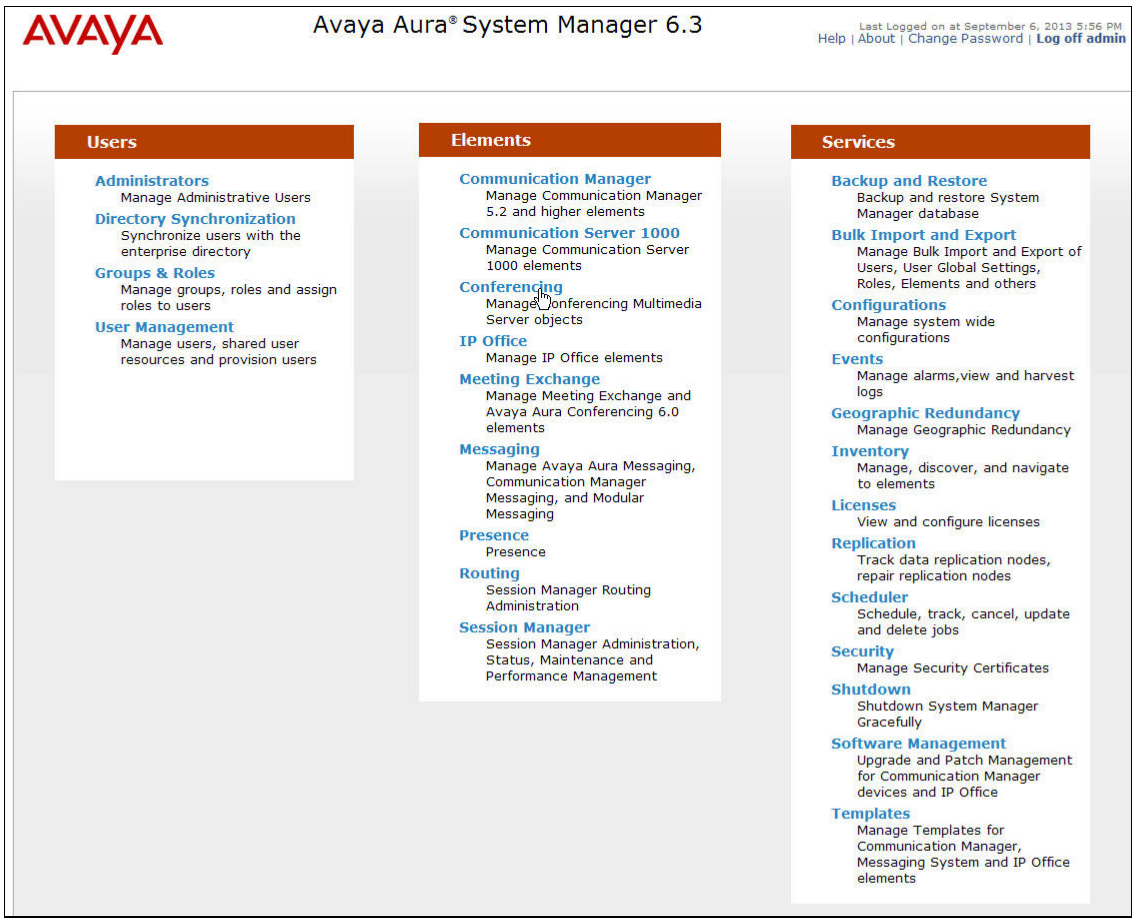
The configuration procedures fall into the following areas:

- Launch System Manager
- Configure SIP Entity for Experience Portal
- Configure Entity Link for Experience Portal
- Configure Routing Policy for Experience Portal
- Configure Dial Pattern for Experience Portal

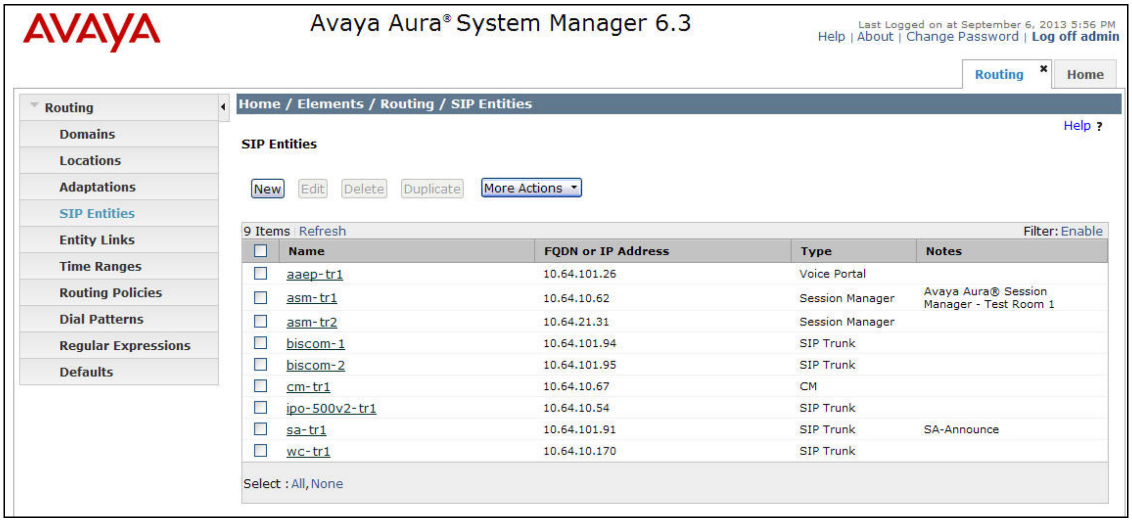
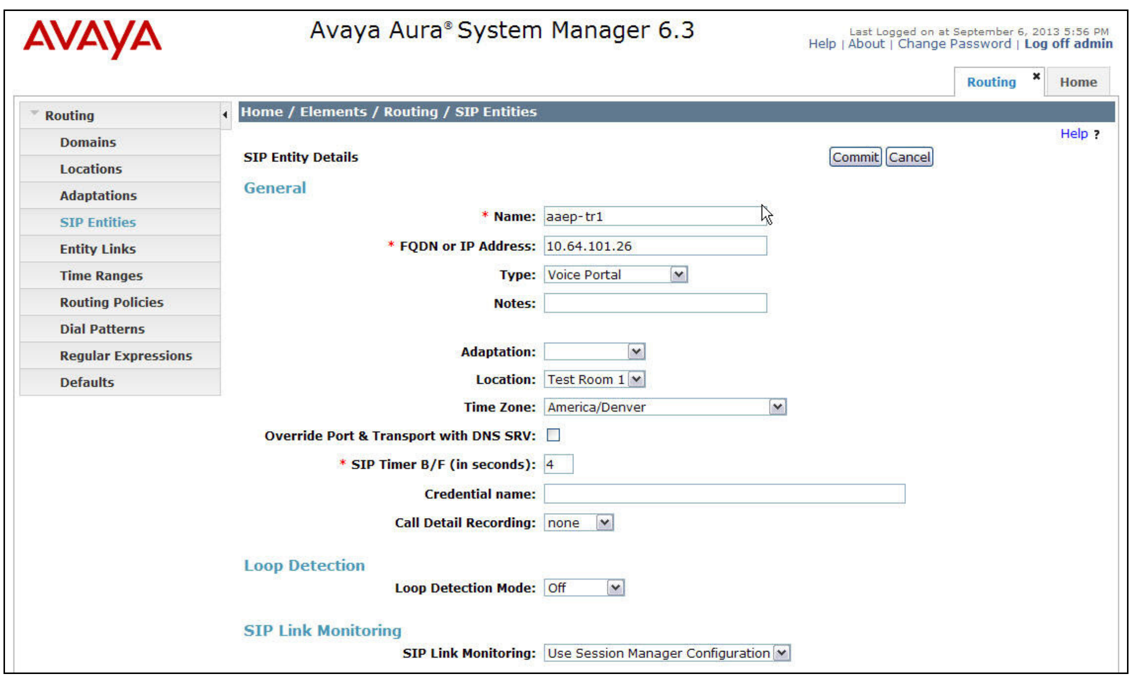
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>

Launch System Manager
Session Manager is configured using browser access to System Manager. Enter **https://<*ip-addr*>** into the URL field of a web browser, where **<*ip-addr*>** is the IP address or qualified domain name of the System Manager. Login using appropriate credentials.

The home page is a navigation screen as shown below. Each of these links will open a new tab from which to navigate to the details of the managed environment. Click **Routing**.

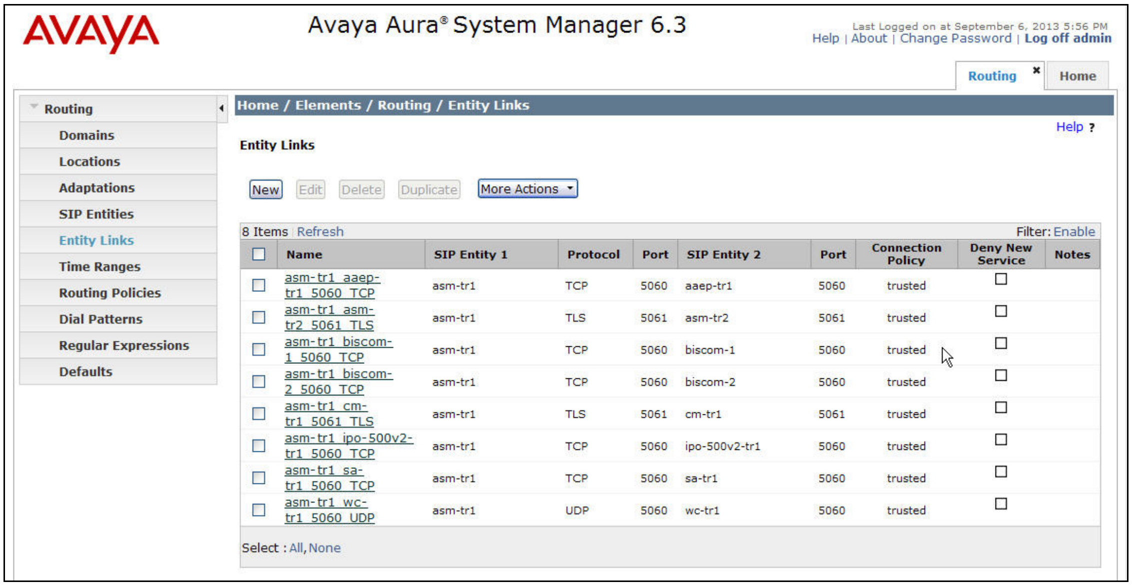


</td></tr>
</table>

Step	Description
2.	**Configure SIP Entity for Experience Portal** On the left pane, click **SIP Entities**. The **SIP Entities** screen is displayed.  Click **New**. The **SIP Entity Details** screen is displayed. Enter a descriptive name to the **Name** field and the IP Address or Fully Qualified Domain Name of the Experience Portal to the **FQDN or IP Address** field. Select **Voice Portal** from the dropdown menu of the **Type** field. Set the **Location** and **Time Zone** fields to proper values. Click **Commit**. 

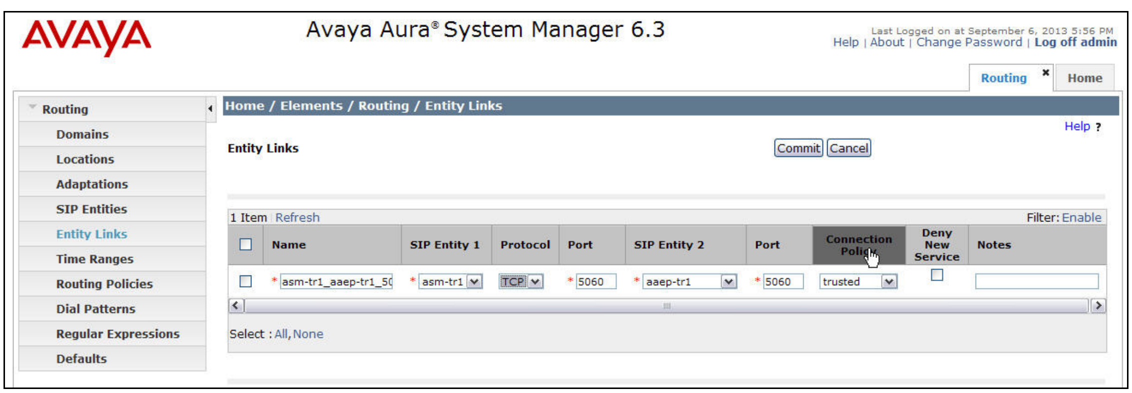
Step	Description
3.	**Configure Entity Link for Experience Portal**

On the left pane, click **Entity Links**. The Entity Links screen is displayed.

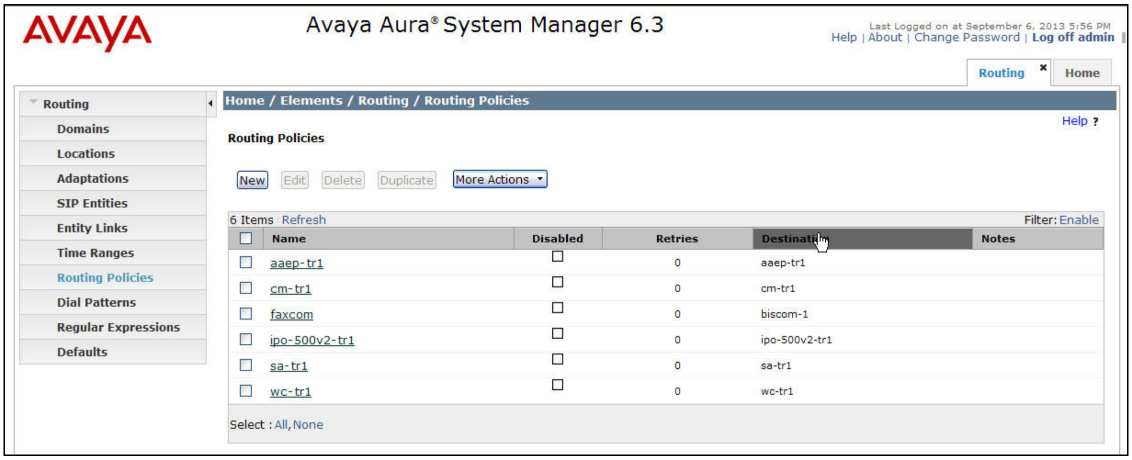


Click **New**. Enter the following values and click **Commit**.

- Add a **Name**
- Set the **SIP Entity 1** field to **asm-tr1 which is the SIP entity for Session Manager**.
- Set the **Protocol** field to **TCP**.
- Set the **SIP Entity 2** to the SIP Entity defined in **Step 2**.
- **Set** both **Port** fields to **5060**.
- Set the **Connection Policy** field to **trusted**.

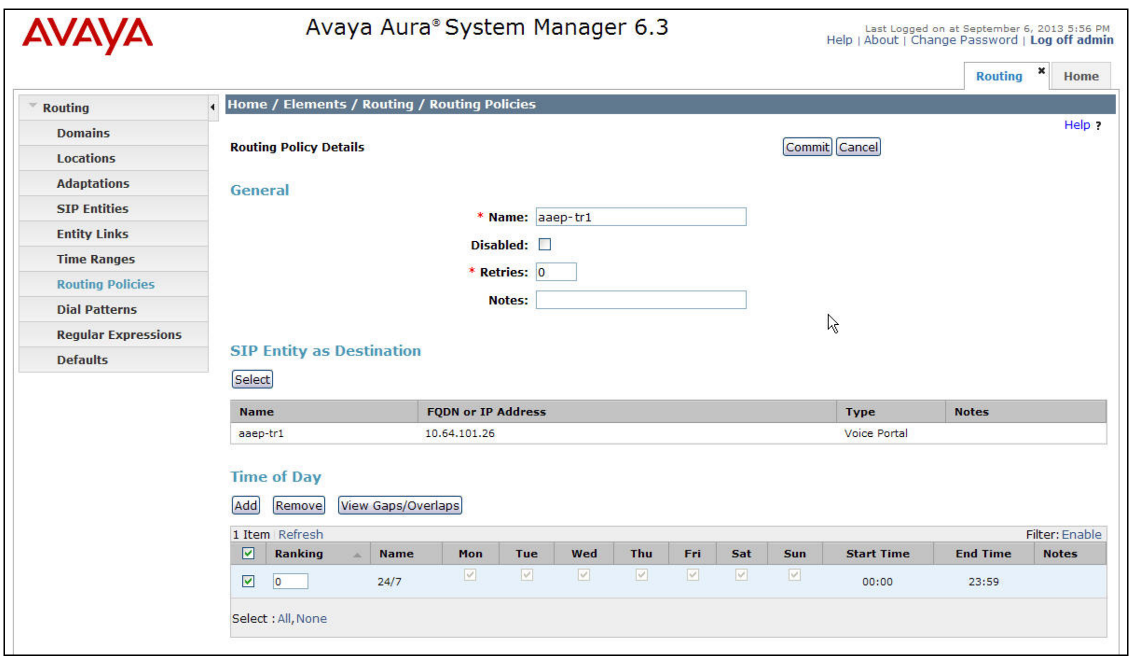


Step	Description
4.	**Configure Routing Policy** On the left pane, click **Routing Policies**. The **Routing Policies** screen is displayed.



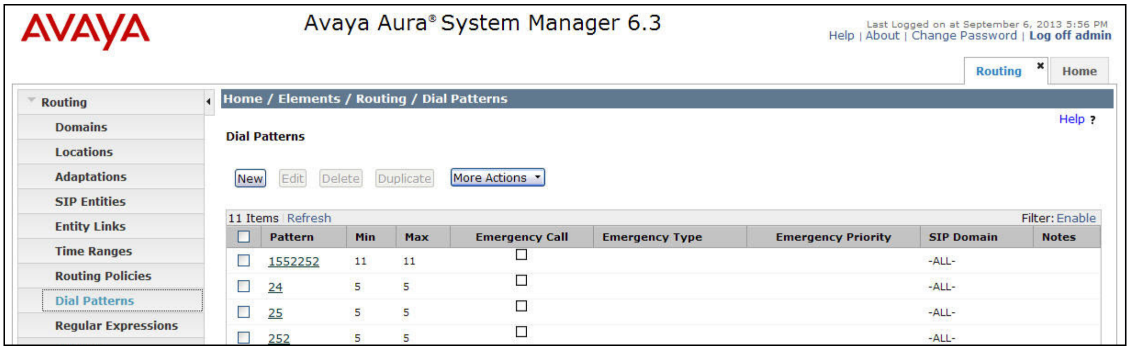
Click **New**. The **Routing Policy Details** screen is displayed. Configure the following and click **Commit**.

- Enter a descriptive name to the **Name** field.
- Under **SIP Entity as Destination** section, click **Select.** A new window is displayed (not shown). Enter the SIP Entity configured in **Step 2**. Click **Select**.



Step	Description
5.	**Configure Dial Pattern for Experience Portal**

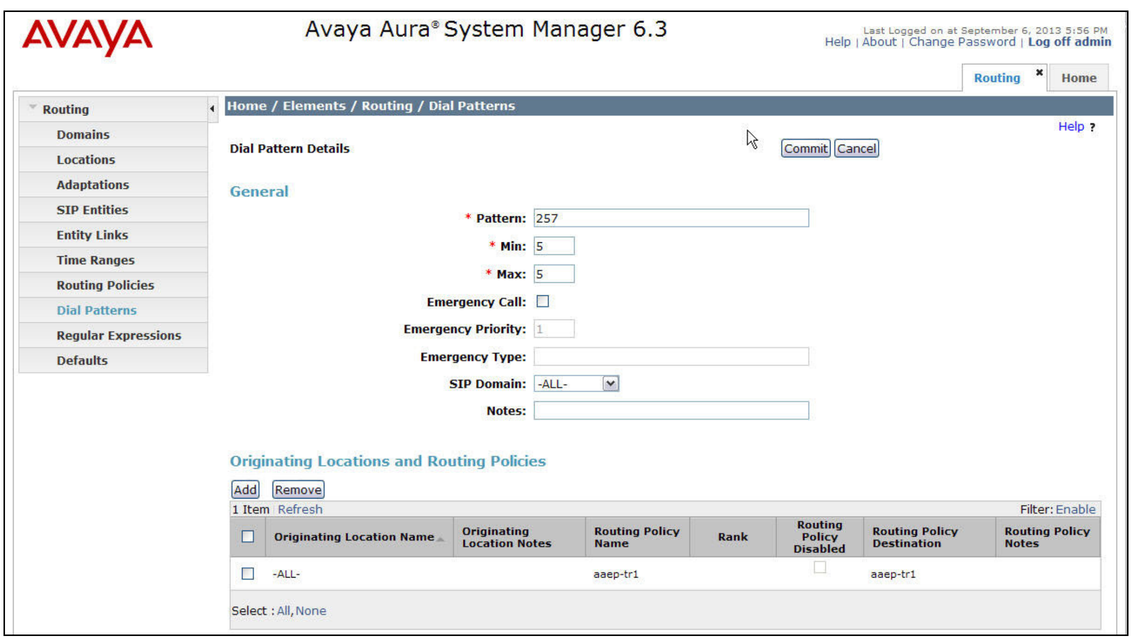
On the left pane, click **Dial Patterns**. The **Dial Pattern**s screen is displayed.



Click **New**. The **Dial Pattern Details** screen is displayed. Configure the following and click **Commit**.

- Set the **Pattern** field to **257**.
- Set the **Min** and **Max** fields to **5** and **5**.
- Set the **SIP Domain** field to **-ALL-**.
- Under the **Originating Locations and Routing Policies** section, click **Add.** A new window is displayed (not shown). Select the **Routing Policy** configured in **Step 4** and then check the **Apply The Selected Routing Policies to All Originating Locations** checkbox. Click **Select.**
- Click **Commit**.

The Dial Pattern configuration directs all calls with 257xx destinations to Experience Portal.



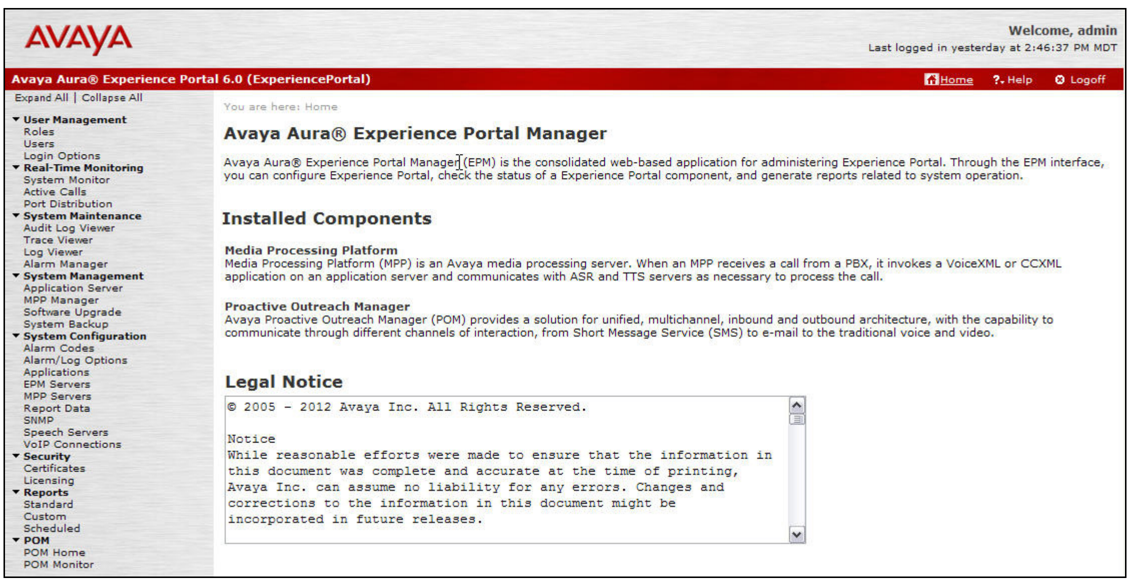
7. Configure Avaya Aura® Experience Portal

This section provides the steps to configure Experience Portal using the Experience Portal Manager (EPM) web interface to support the Virtual Hold solution.

The configuration procedures fall into the following areas:

- Launch Experience Portal Manager
- Configure VoIP Connections for H.323 Configuration
- Configure VoIP Connections for SIP Configuration
- Configure Web Services Authentication Parameters
- Configure Applications for H.323 Configuration
- Configure Applications for SIP Configuration

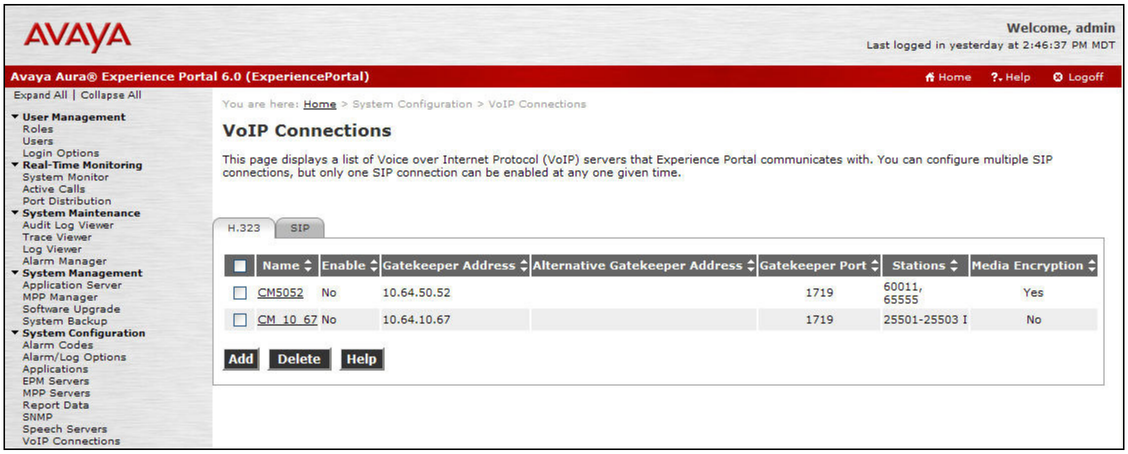
Step	Description
1.	**Launch Experience Portal Manager** Type in **http://*<ip-addr>*/** as the URL in a web browser, where ***<ip-addr>*** is the IP address of Experience Portal Manager. Log in with proper credentials.



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>

Configure VoIP Connections

On the left pane, click **VoIP Connections**.



</td></tr>
</table>

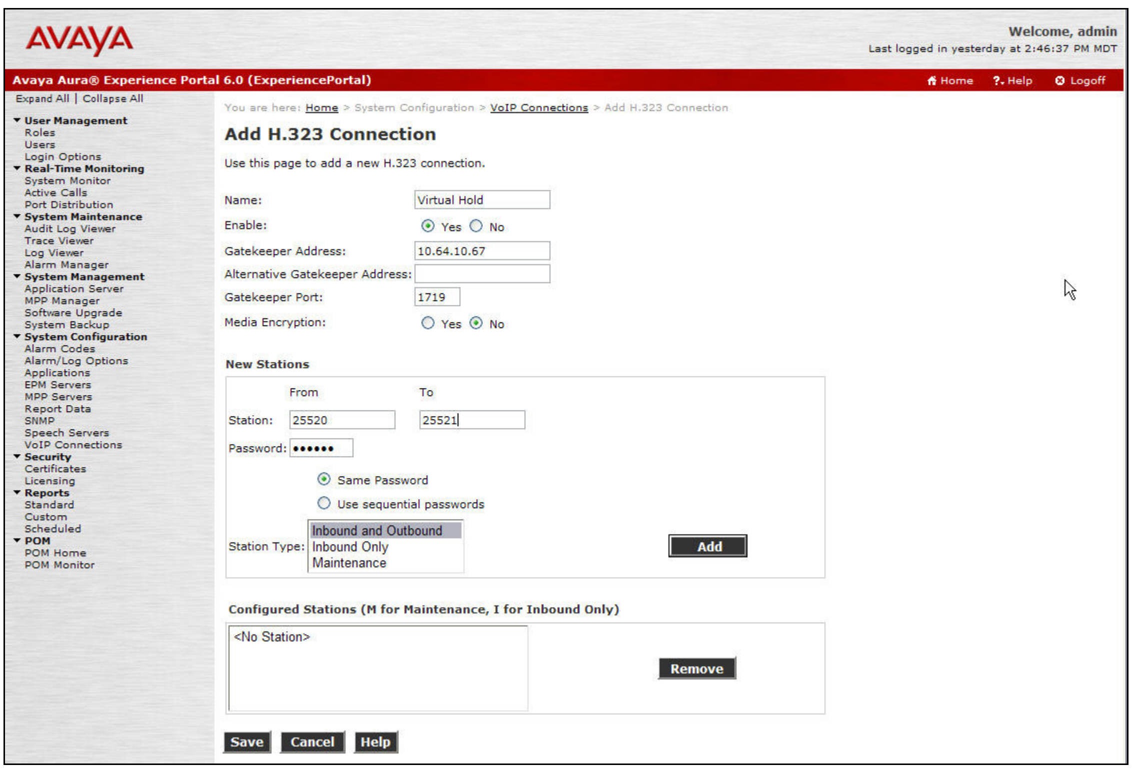
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

Configure H.323 Connections (H.323 Configuration only)

To add a H.323 Connection, click the **H.323** tab followed by **Add**.

- Type in **Name**
- Fill in **Gatekeeper Address.** Gatekeeper address is the IP address of the Communication Manager
- Set the **Media Encryption** field to **No**.
- Enter the stations configured in **Section 4**, **Step 4** in the **Station From** and **To** fields. Enter the stations’ Security Code from **Section 4**, **Step 4** in the **Password** field and select the **Same Password** radio button. Select **Inbound and Outbound** as the **Station Type**. Click **Add**.
- The rest of the values are left at **Default**.
- Click **Save.**

Note: If both H.323 Connections and SIP Connections are configured, only one type connections can be enabled at a given time. Another type must be disabled by setting the **Enable** field to **No**.



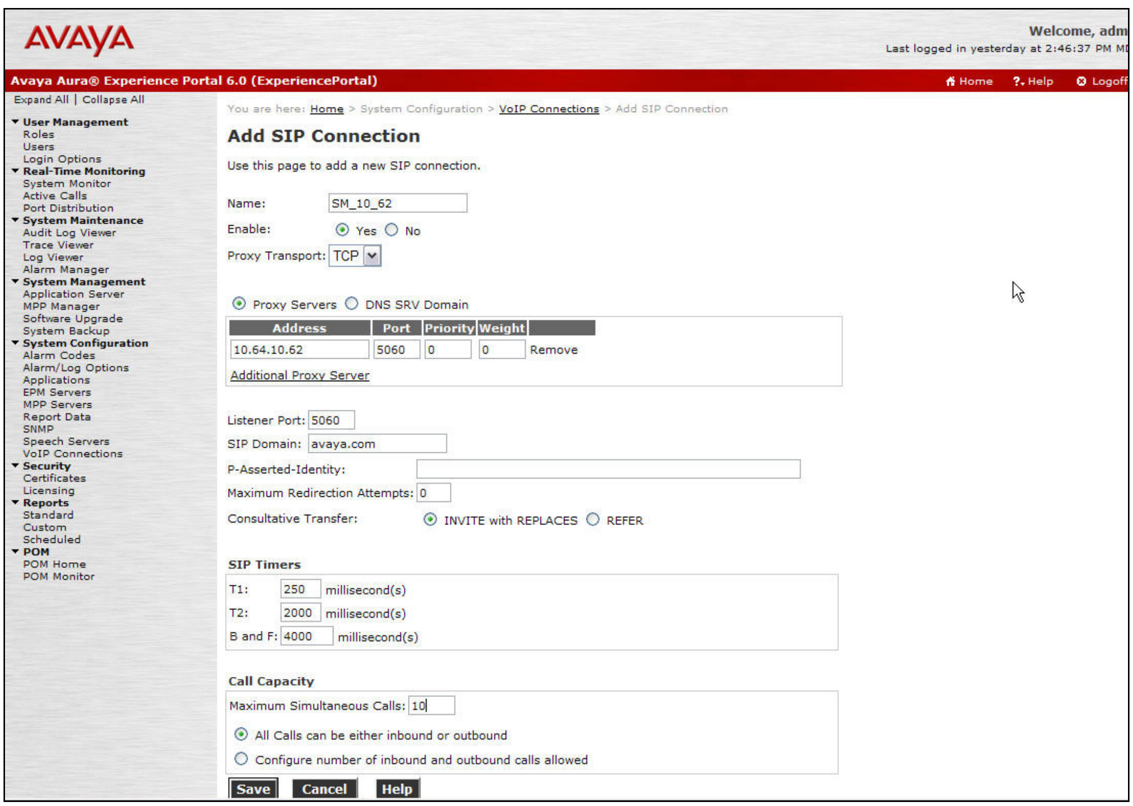
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>

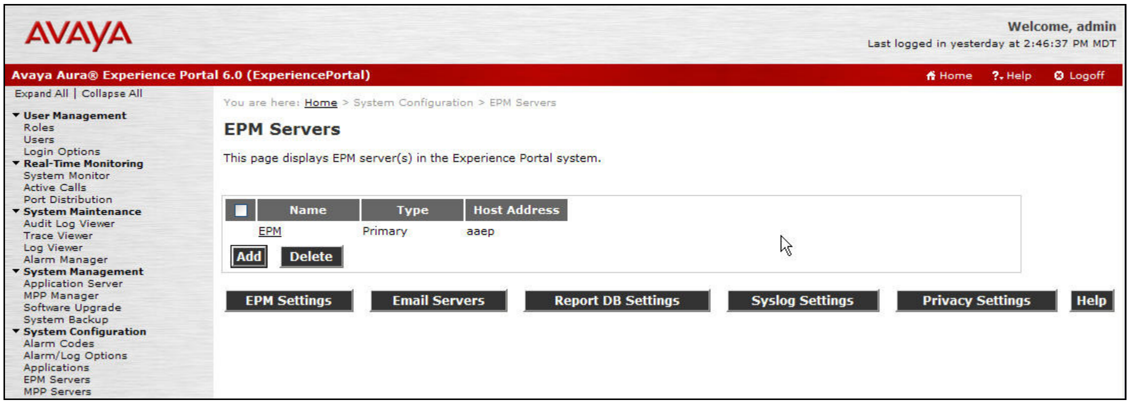
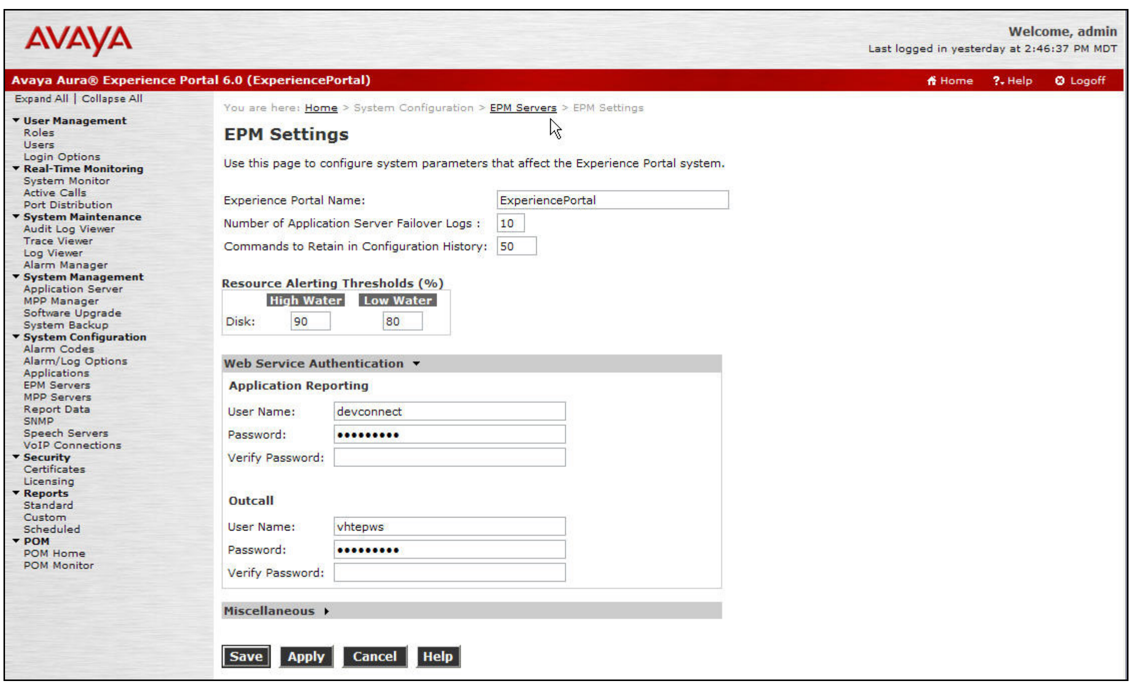
Configure SIP Connections (SIP Configuration only)
To add a **SIP Connection**, click **SIP** tab on the **VoIP Connections** page and then click **Add**.

- Fill in **Name**.
- In the **Address** and **Port** fields, fill the the IP address and Port of Session Manager.
- In the **SIP Domain** field, type in the domain pre-configured in Session Manager.
- Set the **Maximum Simultaneous Calls** field to **10**.
- The rest of the values are left at **default values**.
- Click **Save.**

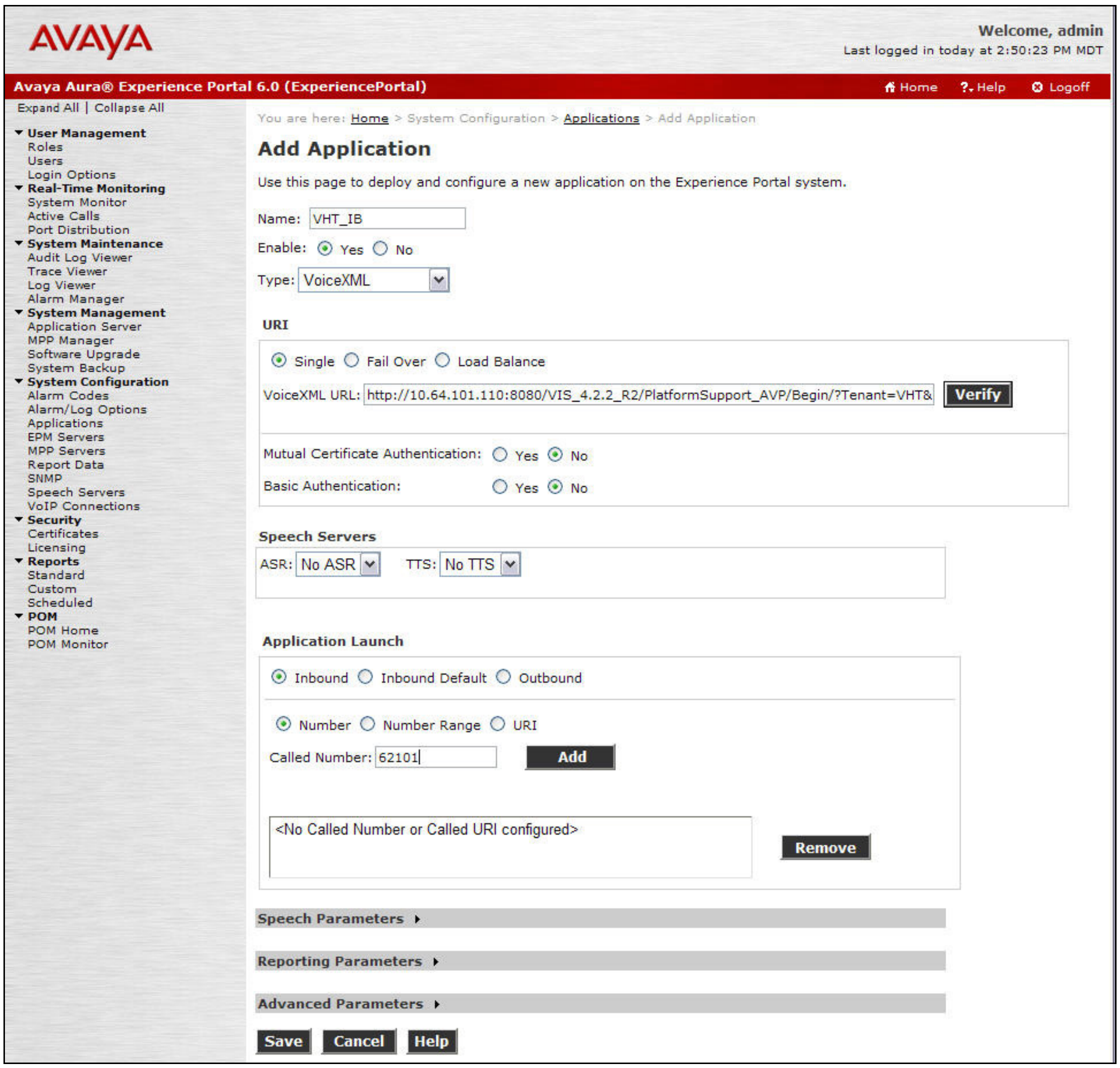
Note: If both H.323 Connections and SIP Connections are configured, only one type can be enabled at a given time. Another type must be disabled by setting the **Enable** field to **No**.



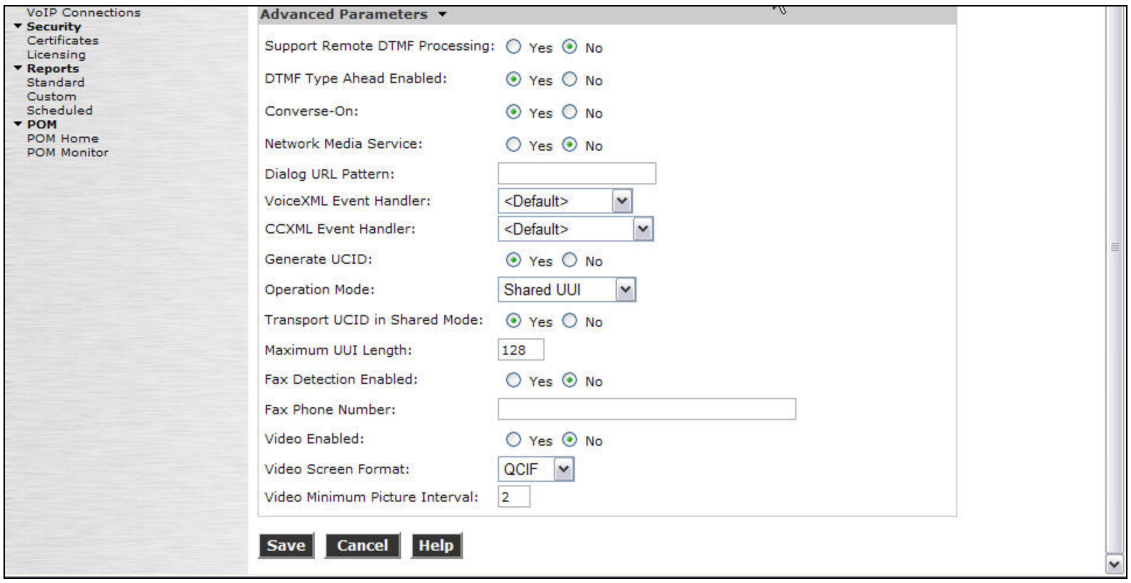
</td></tr>
</table>

Step	Description
5.	**Configure Web Service Authentication Parameters** On the left pane, click **System Configuration → EPM Servers**. The **EPM Servers** screen is displayed.  Click **EPM Settings**. The **EPM Settings** screen is displayed. Under the **Web Service Authentication** section, **Outcall** sub-section, type in **Username, Password** and **Verify Password**. This information will be used by Virtual Hold to initiate an outbound call. Click **Save**. 

Step	Description
6.	**Configure Inbound Application for H.323 Configuration** On the left pane, navigate to **System Configuration → Applications**. The **Application screen** is displayed (not shown). Click **Add**. The **Add Application** screen is displayed. • Fill in **Name**. • For **Type**, select **VoiceXML** from the drop down menu. • Fill in **VoiceXML URL**: http://10.64.101.110:8080/VIS_4.2.2_R2/PlatformSupport_AVP/Begin/?Tenant=VHT&MODE=AVP, where **10.64.101.110** and **8080** are the IP Address and Tomcat Port of the Virtual Hold Server • Set the **Called Number** field to **62101** which is the extension of the H.323 Routing VDN configured in **Section 4**, **Step 6** and click **Add**.



Continue on the next page.

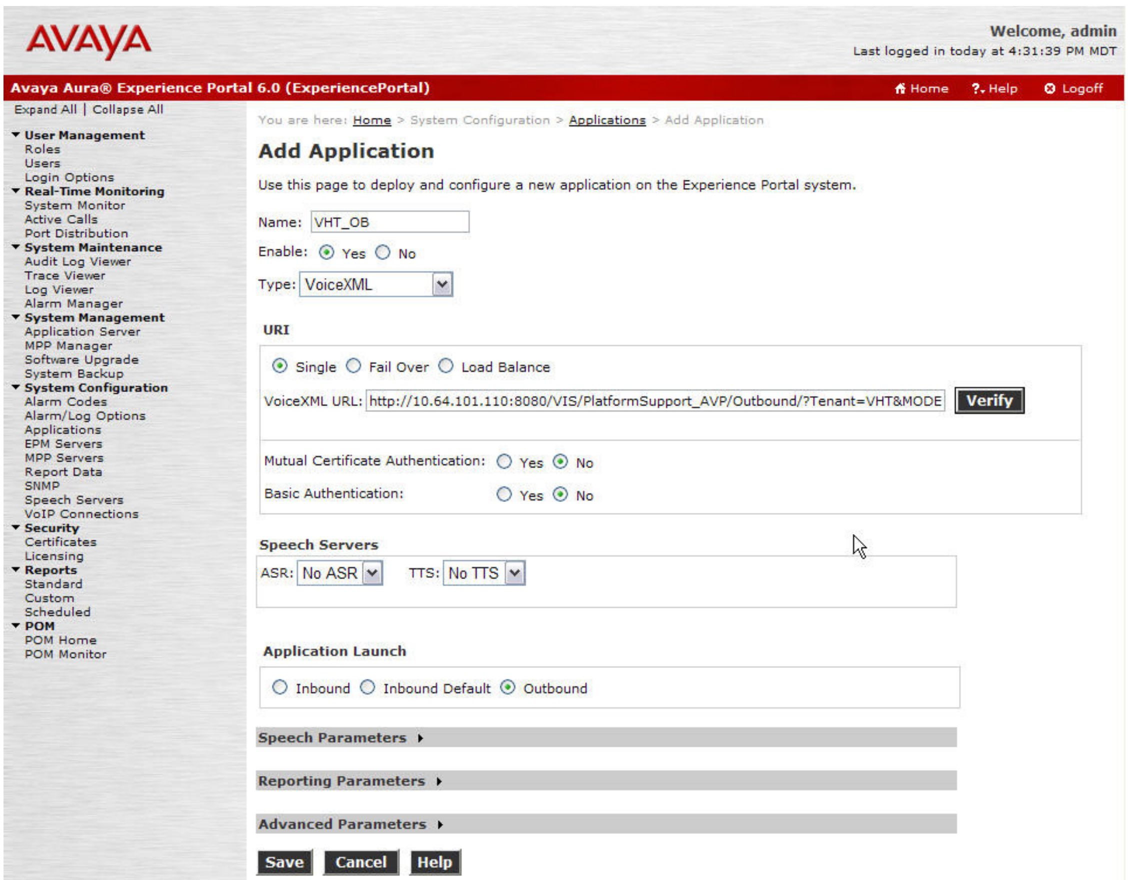
Step	Description
	Click **Advanced Parameters** to expand. • Set the **Converse-On** field to **Yes**. • Set the **Generate UCID** field to **Yes**. • Set the **Operation Mode** field to **Shared UUI**. • Set the **Transport UCID in Shared Mode** field to **Yes**. • Click **Save**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>

Configure Outbound Application for H.323 Configuration

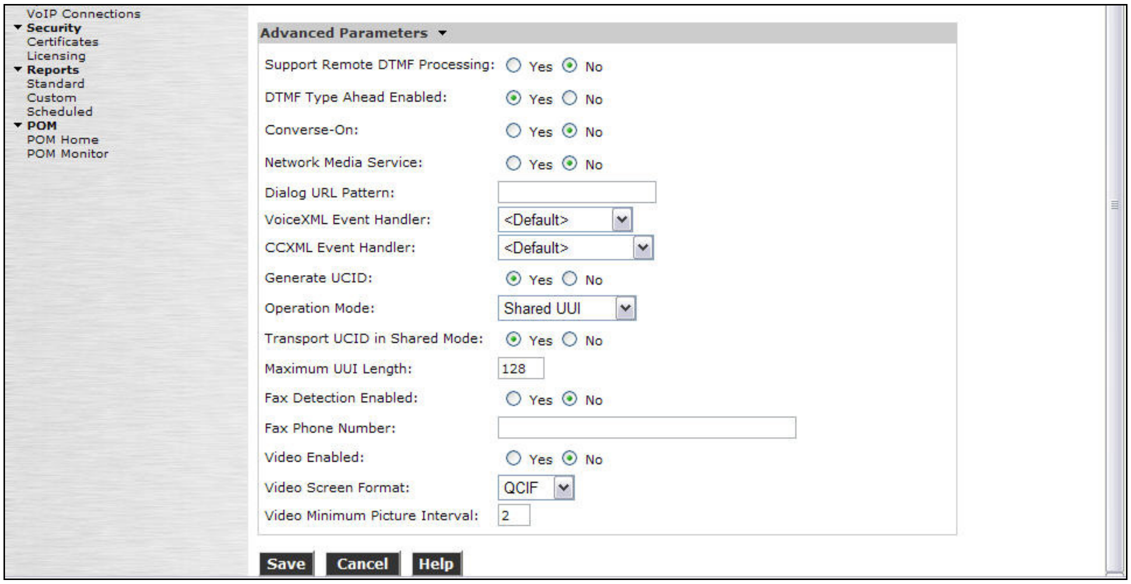
On the Application screen (not shown), click **Add**. The **Add Application** screen is displayed.

- Fill in **Name**.
- For **Type**, select **VoiceXML** from the drop down menu.
- Fill in **VoiceXML URL**: http://10.64.101.110:8080/VIS_4.2.2_R2/PlatformSupport_AVP/Outbound/?Tenant=VHT&MODE=AVP, where **10.64.101.110** and **8080** are the IP Address and Tomcat Port of the Virtual Hold Server
- Set the **Application Launch** section to **Outbound**.



Continue on the next page.

</td></tr>
</table>

Step	Description
	Click **Advanced Parameters** to expand. • Set the **Generate UCID** field to **Yes**. • Set the **Operation Mode** field to **Shared UUI**. • Set the **Transport UCID in Shared Mode** field to **Yes**. • Click **Save**. 
8.	**Configure Inbound Application for SIP Configuration** The procedure is the same as **Step 6** except the following: • Set the **VoiceXML URL** field to http://10.64.101.110:8080/VIS_4.2.2_R2/PlatformSupport_AVP/Begin/?Tenant=VHT&MODE=AVPSIP • Set the **Called Number** field to **25798**. • Set the **Converse-On** field to **No**.
9.	**Configure Outbound Application for SIP Configuration** The procedure is the same as **Step 7** except the following: • Set the **VoiceXML URL** field to http://10.64.101.110:8080/VIS_4.2.2_R2/PlatformSupport_AVP/Outbound/?Tenant=VHT&MODE=AVPSIP

8. Configure VHT Virtual Hold

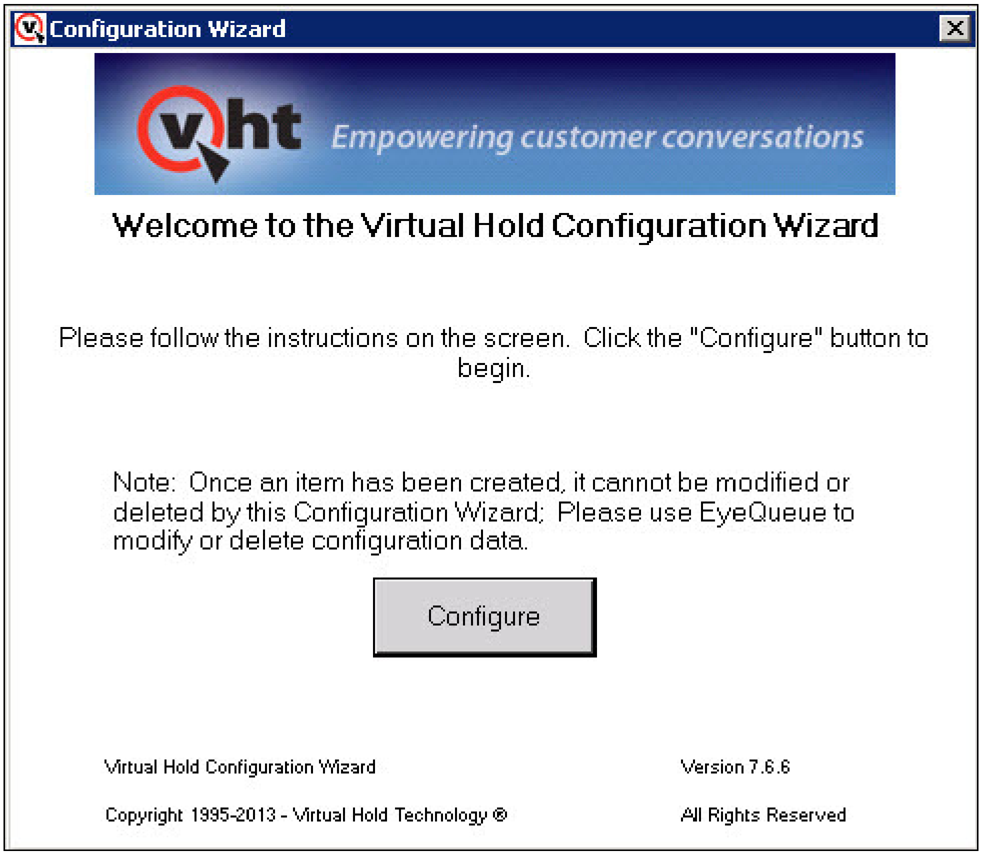
The Virtual Hold software runs under Windows 2008 Server R2 SP2 64bit operating system. Configuration of Virtual Hold is done through the following elements:

- VHT Configuration Wizard
- SQL Server Management Studio
- Text based configuration files

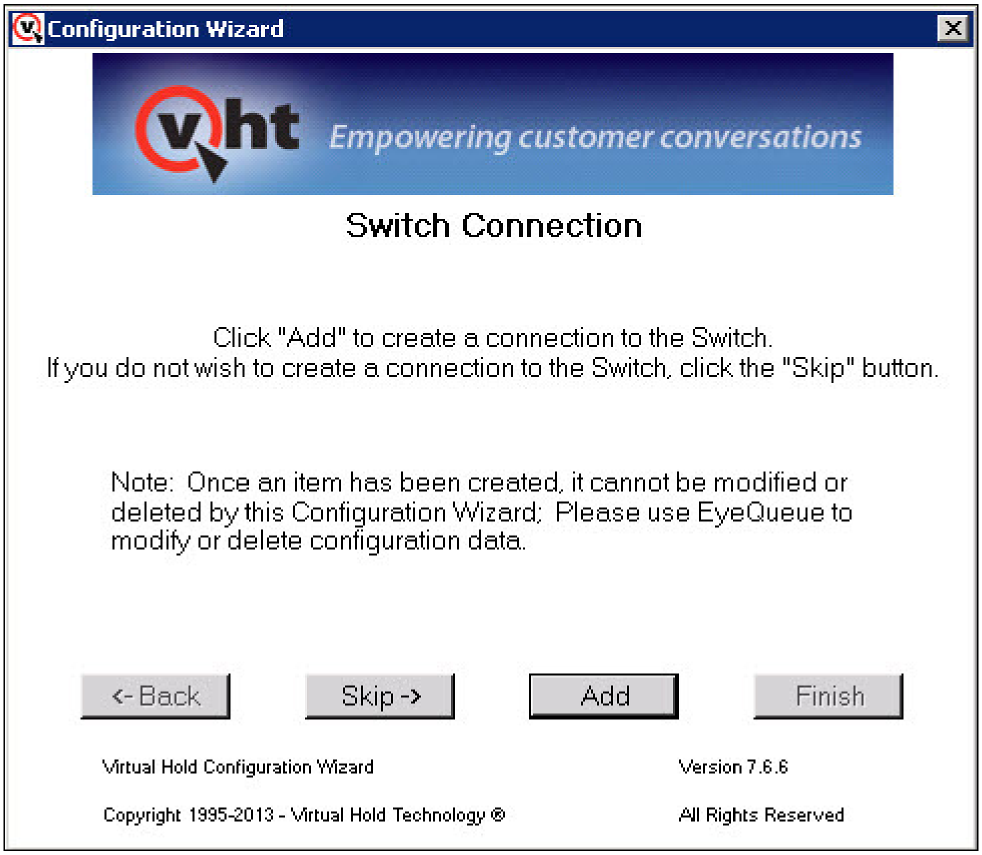
The configuration procedures fall into the following areas:

- Using VHT Configuration Wizard
    - Launch VHT Configuration Wizard
    - Configuration Switch Connection
    - Configure AES Avaya CRI
    - Configure IVR Servers (H.323 Configuration only)
    - Configure Queues
    - Configure Callback and Holding Queues
    - Configure Incoming Extensions
    - Configure Phone Number Configuration
- Using SQL Server Management Studio
    - Configure Segment Variables
- Using text files
    - OutboundIVR_AVP.xml
    - toolkit.properties

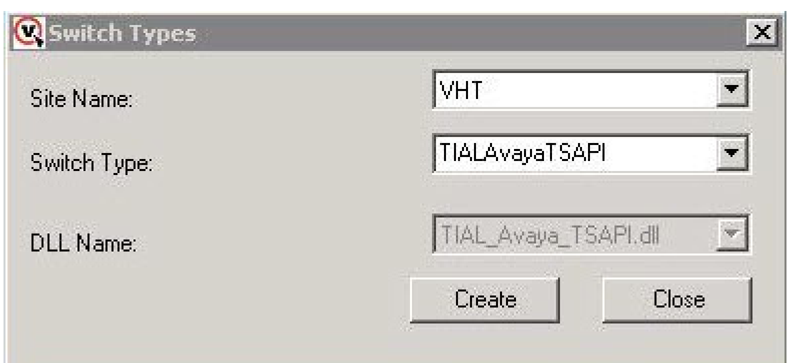
Step	Description
1.	Log in the Virtual Hold server with proper credentials. Open **VHT Configuration Wizard** by navigating to **Start →All Programs → Virtual Hold Technology → Configuration → VHT Configuration Wizard**.

Step	Description
2.	On the **Welcome to the Virtual Hold Configuration Wizard** page, click **Configure.** 

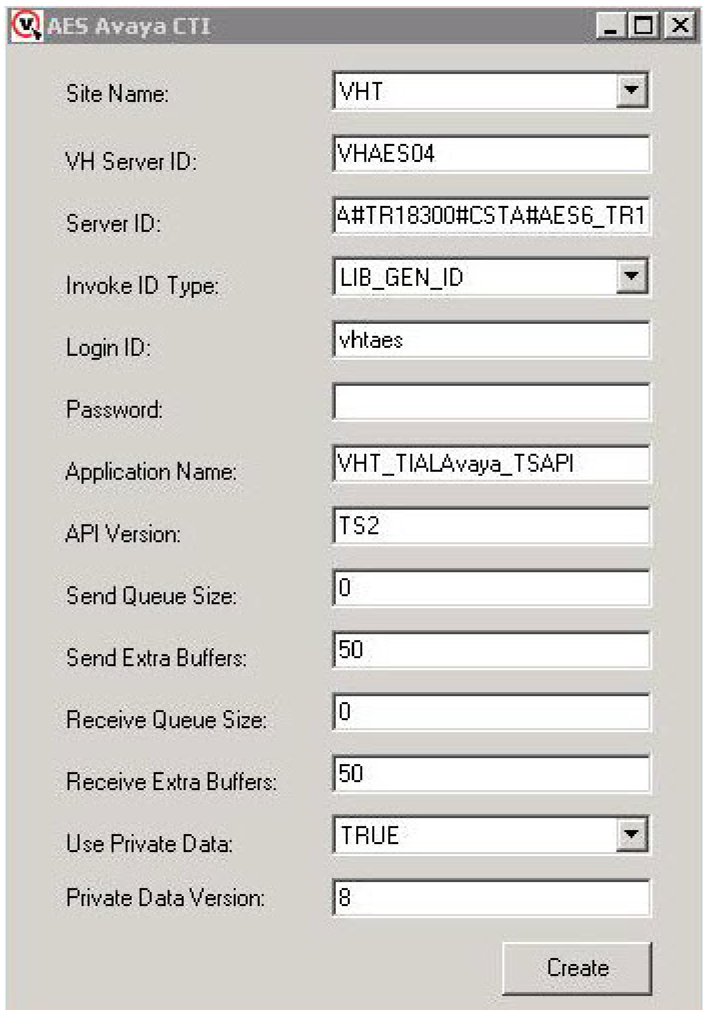
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>On the Switch Connection page, click Add.

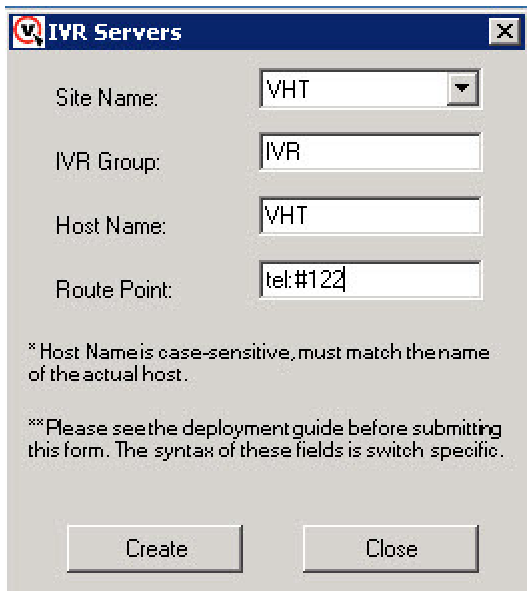


</td></tr>
<tr><td>4.</td><td>The Switch Types window is displayed with the Site Name already populated. Select TIALAvayaTSAPI from the drop-down menu of the Switch Type field. Click Create.

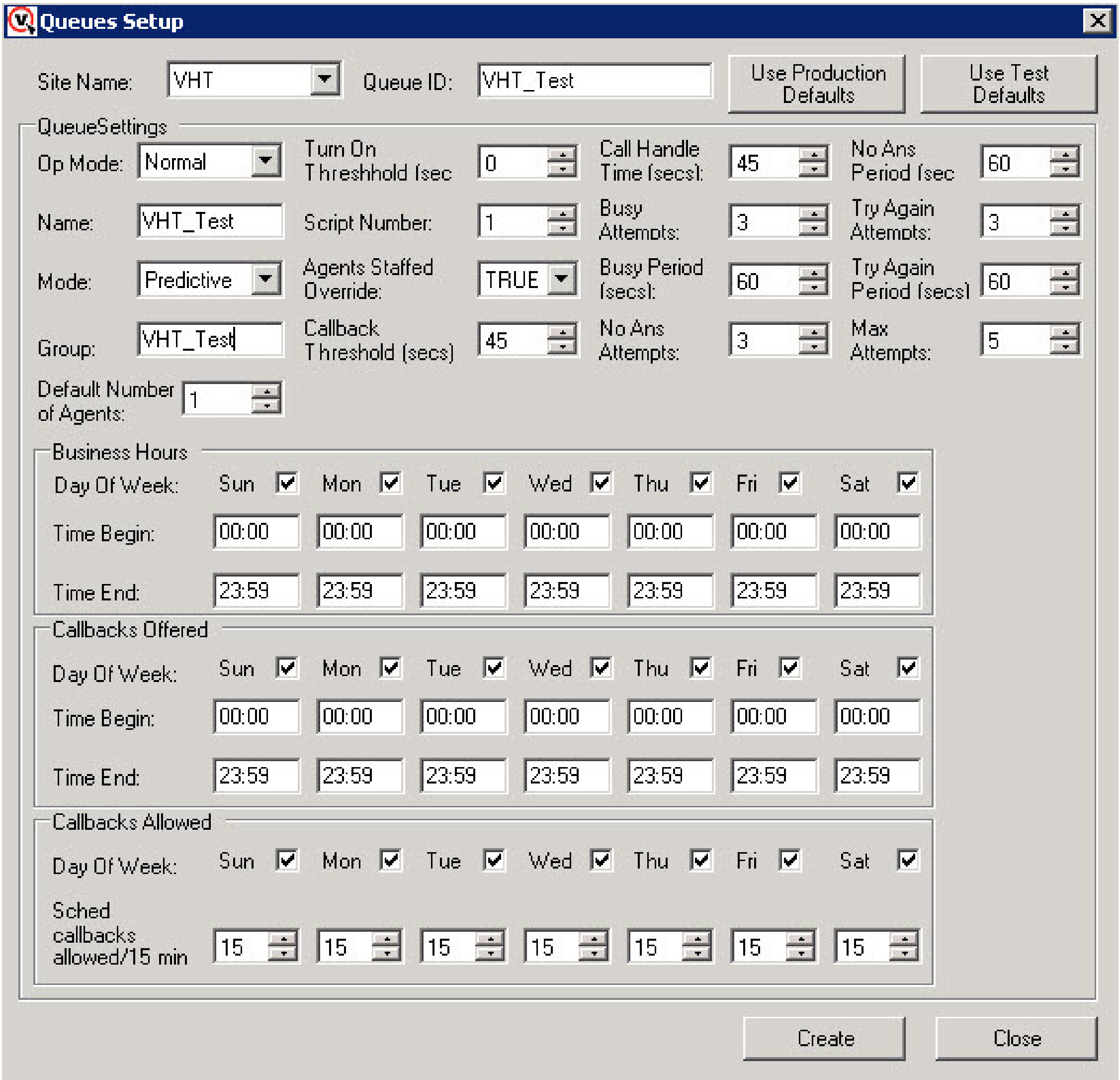


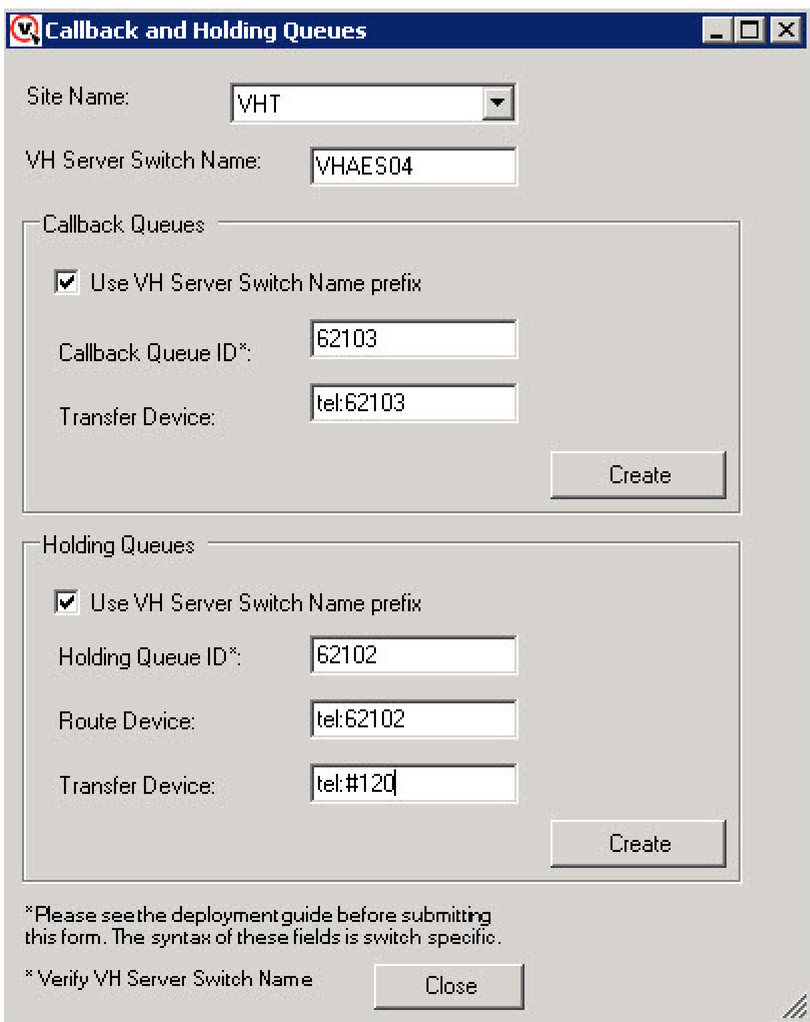
</td></tr>
</table>

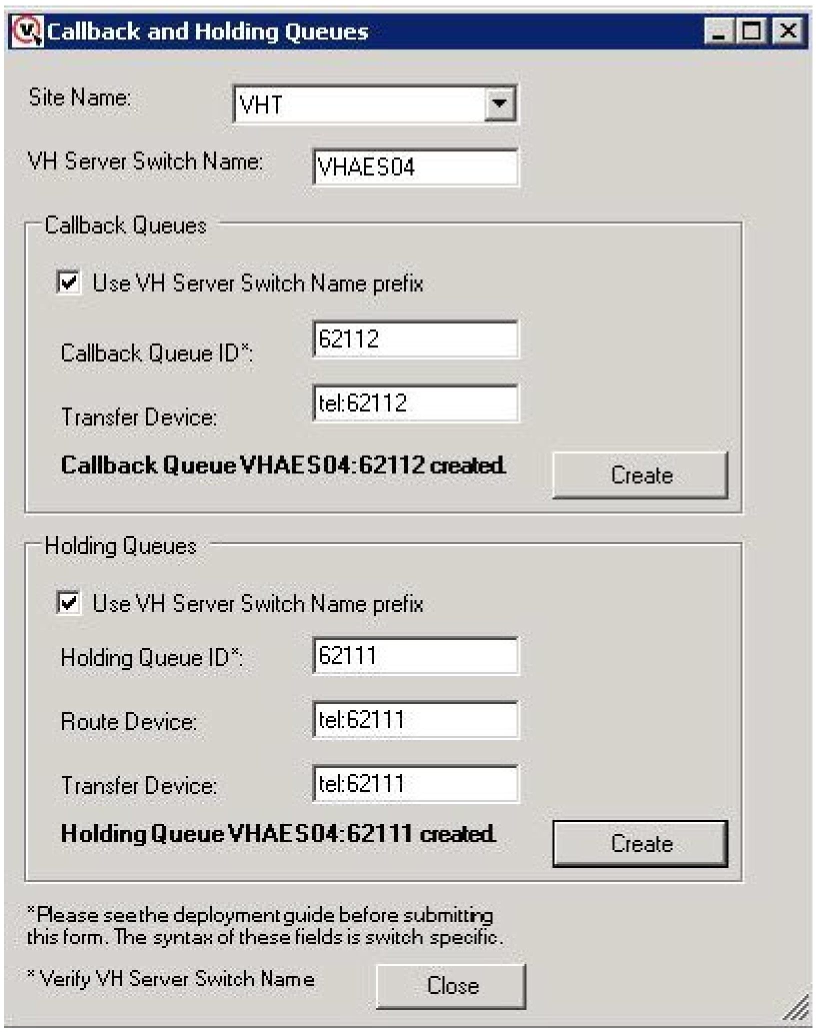
Step	Description
5.	The **AES Avaya CTI** window is displayed. • Set the **VH Server ID** field to a descriptive name • Set the **Server ID** field to one of the tlinks noted in **Section 5**, **Step 3** • Set the **Login ID** and **Password** field to the **User Id** and **User Password** values configured in **Section 5**, **Step 5** • Set the **Application Name** field to **VHT_TIALAvaya_TSAPI** • Set the **Send Extra Buffers** and **Receive Extra Buffers** field to **50** • Set the **Use Private Data** field to **TRUE** • Set the **Private Data Version** field to **8** Click **Create** followed by **Close**. 

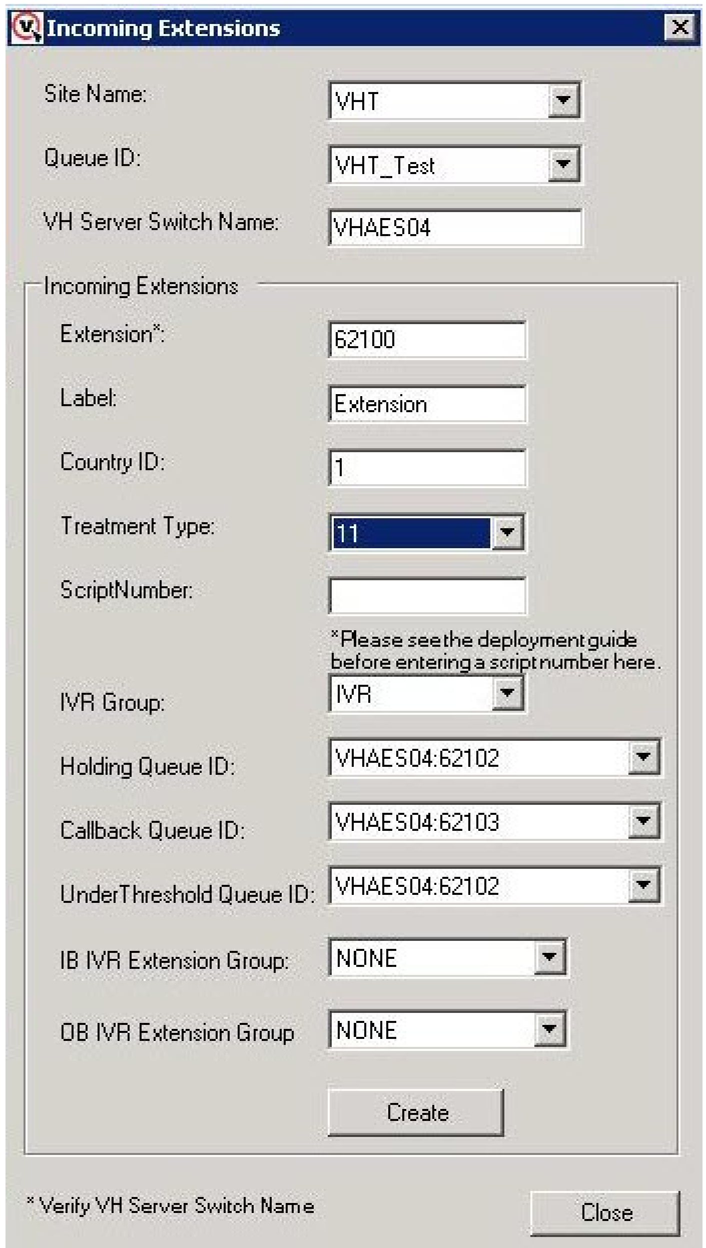
Step	Description
6.	Skip the **Agent Groups** and **Agents** page (not shown). The following procedure is for the H.323 Configuration: On the **IVR Servers** page, click **Add** (not shown). The **IVR Servers** window is displayed. Enter **tel:#122** in the **Route Point** field where **#12** is the **Converse Data Return Code** configured in **Section 4, Step 9**. Keep the values in other fields and click **Create** followed by **Close**. For the SIP Configuration, skip the **IVR Servers** page. 

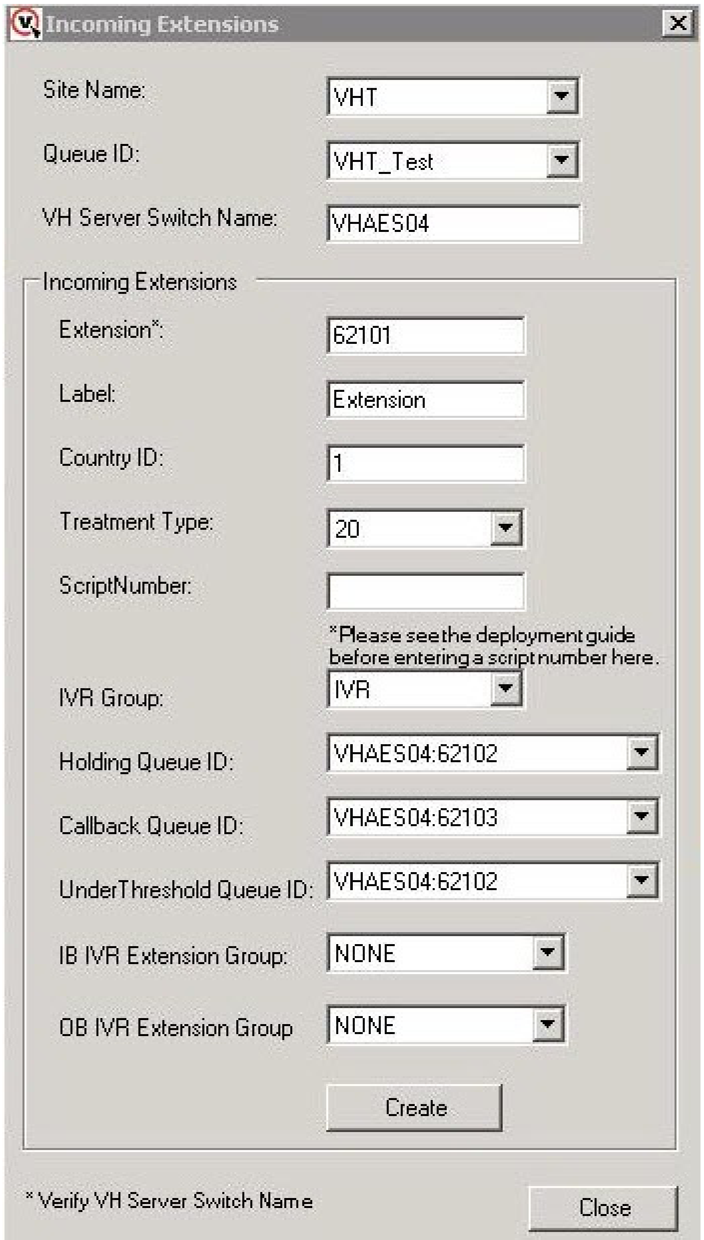
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Skip the IVR Extensions page and click Add on the Queues page (not shown). The Queues Setup window is displayed. Enter a group name in the Group field (for reporting purpose) and accept the defaults value for all other fields. Click Create followed by Close.

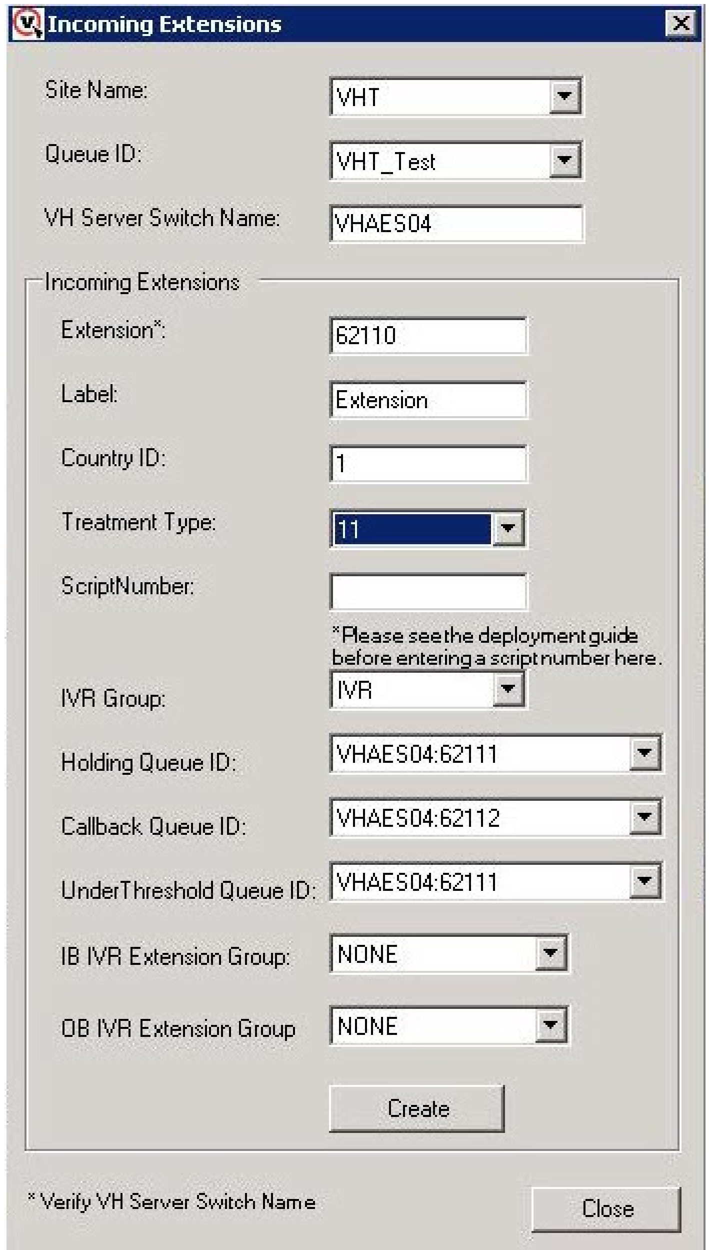

</td></tr>
</table>

Step	Description
8.	On the **Callback** and **Holding Queues** page, click **Add** (not shown). The **Callback and Holding Queues** window is displayed. The following procedure is for the H.323 Configuration: In the **Callback Queues** section, enter the Callback VDN configured in **Section 4**, **Step 6** (**62103**) in the **Callback Queue ID** field. Enter the same value with the string **tel:** appended to it in the **Transfer Device** field. Click **Create**. In the **Holding Queues** section, enter the Holding VDN configured in **Section 4**, **Step 6** (**62102**) in the **Holding Queue ID** field. Enter the same value with the string **tel:** appended to it in the **Route Device** field. Enter **tel:#120** in the **Transfer Device** field where **#12** is the **Converse Data Return Code** configured in **Section 4, Step 9**. Click **Create** followed by **Close**.  Continue on the next page.

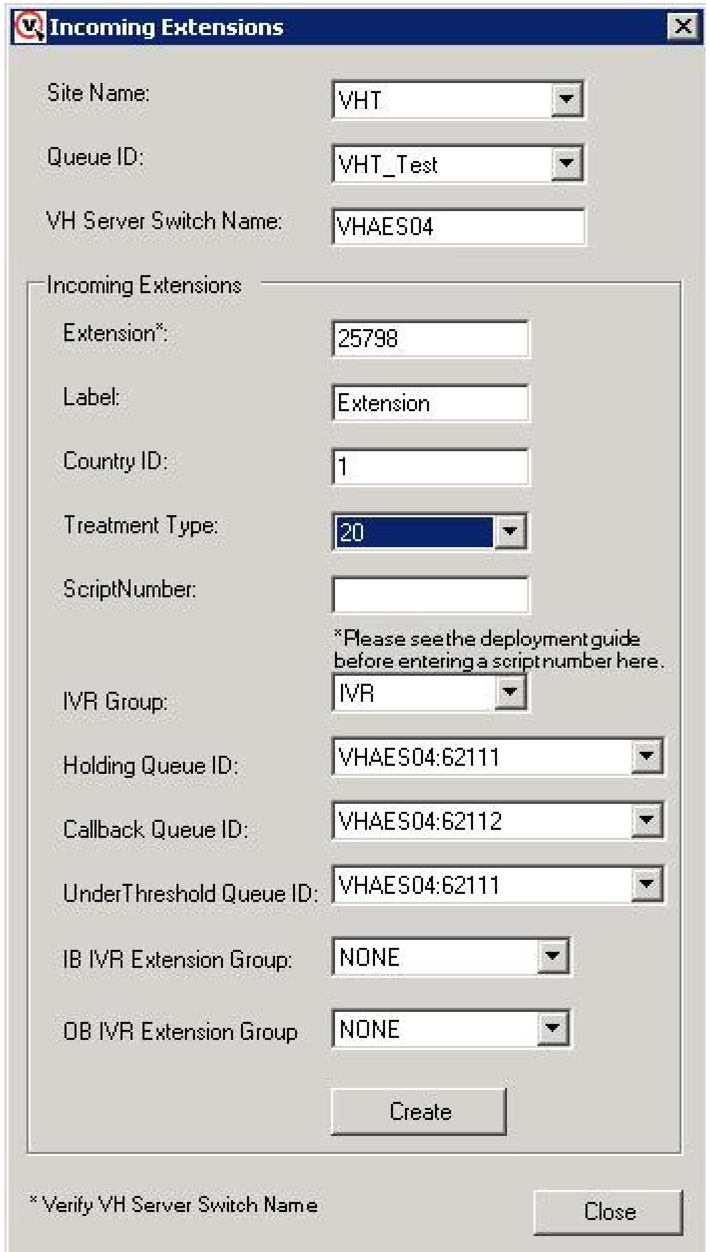
Step	Description
	The following procedure is for the SIP Configuration: In the **Callback Queues** section, enter the Callback VDN configured in **Section 4**, **Step 8** (**62112**) in the **Callback Queue ID** field. Enter the same value with the string **tel:** appended to it in the **Transfer Device** field. Click **Create**. In the **Holding Queues** section, enter the Holding VDN configured in **Section 4**, **Step 8** (**62111**) in the **Holding Queue ID** field. Enter the same value with the string **tel:** appended to it in the **Route Device** field and the **Transfer Device** field. Click **Create** followed by **Close**. 

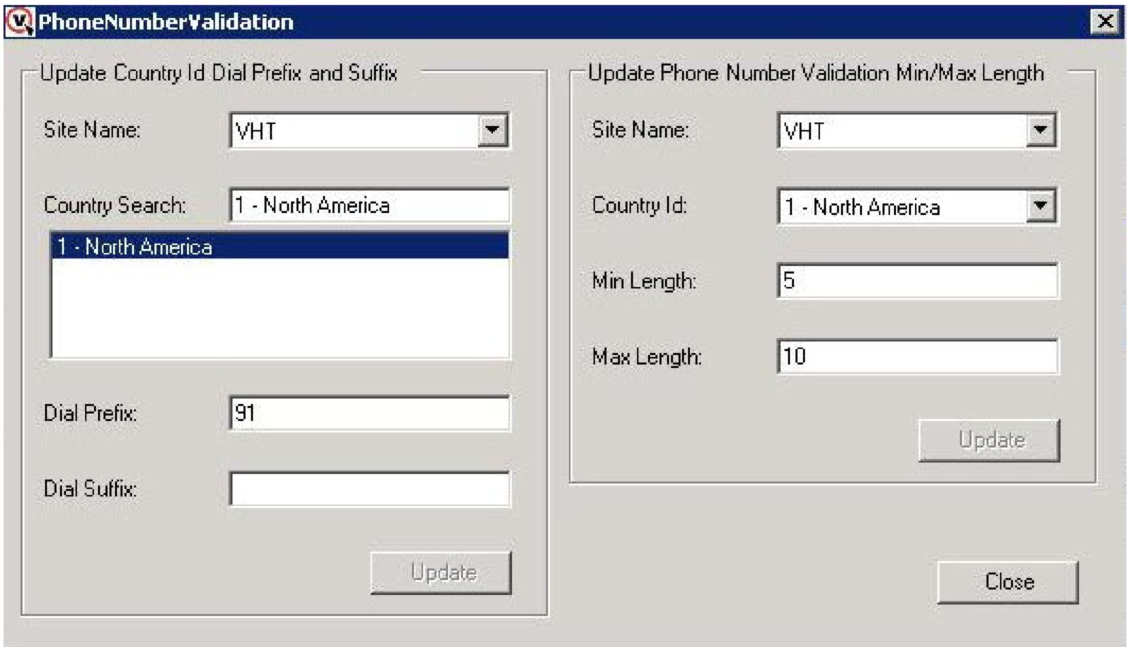
Step	Description
9.	On the **Incoming Extensions** page, click **Add** (not shown). The **Incoming Extensions** window is displayed. The following procedure is for the H.323 Configuration: Enter the Entry VDN configured in **Section 4**, **Step 6** (**62100**) in the **Extension** field and **11** in the **Treatment Type** field. Click **Create**.  Continue on the next page.

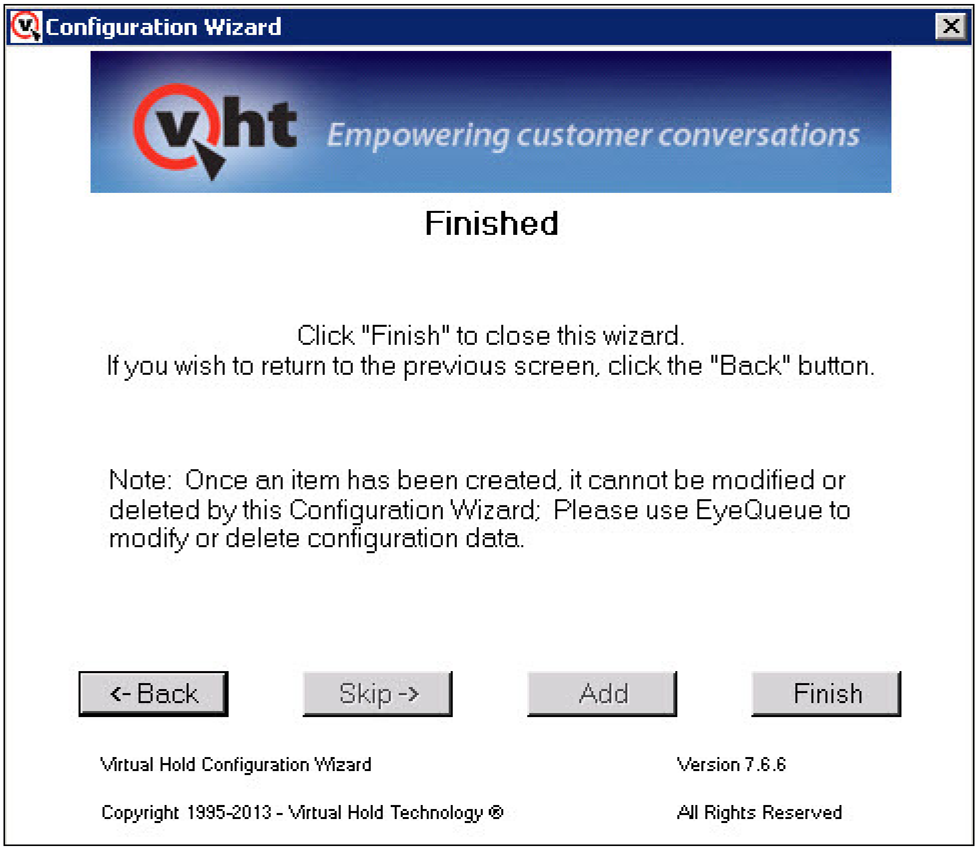
Step	Description
	Repeat the step for the Routing VDN. Enter the Routing VDN configured in **Section 4**, **Step 6** (**62101**) in the **Extension** field and **20** in the **Treatment Type** field. Click **Create** followed by **Close**.  Continue on the next page.

Step	Description
	The following procedure is for the SIP Configuration: Enter the Entry VDN configured in **Section 4**, **Step 8** (**62110**) in the **Extension** field and **11** in the **Treatment Type** field. Click **Create**.  Continue on the next page.

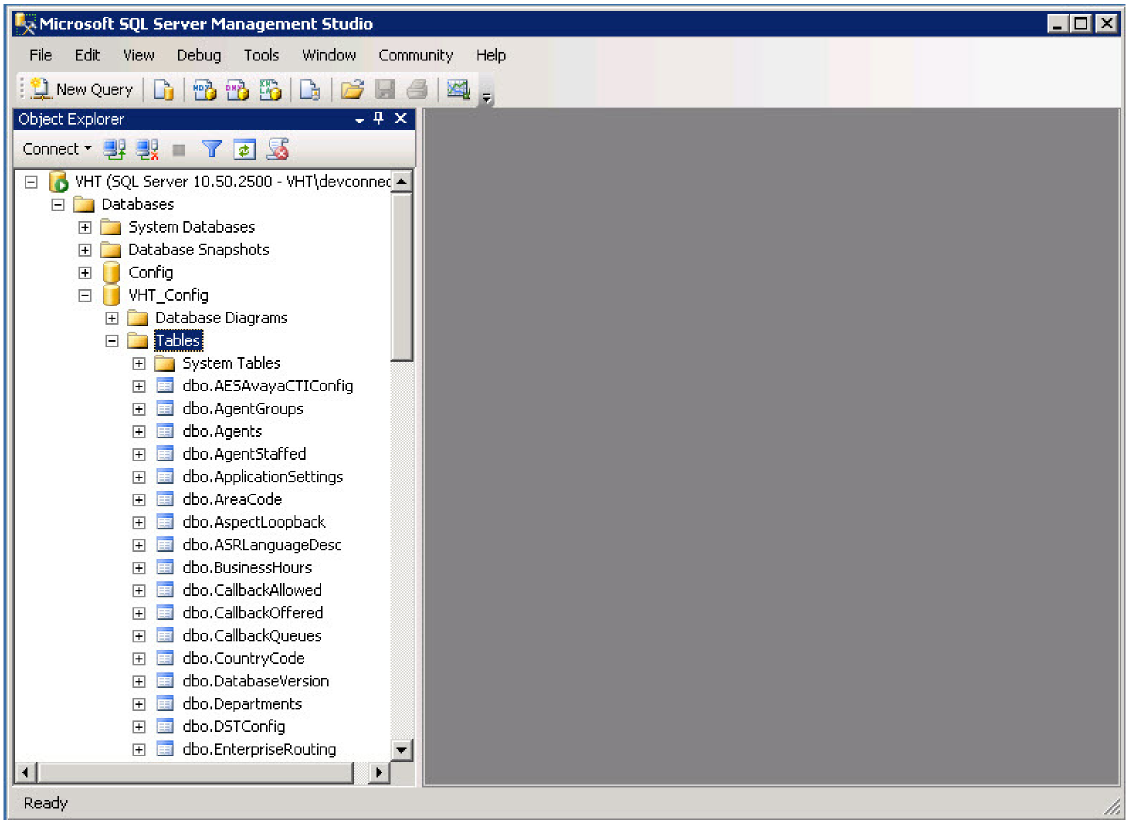
Step	Description
	Repeat the step and enter the extension associated with the Experience Portal inbound application, configured in **Section 7**, **Step 8** (**25798**), in the **Extension** field and **20** in the **Treatment Type** field. Click **Create** followed by **Close**.

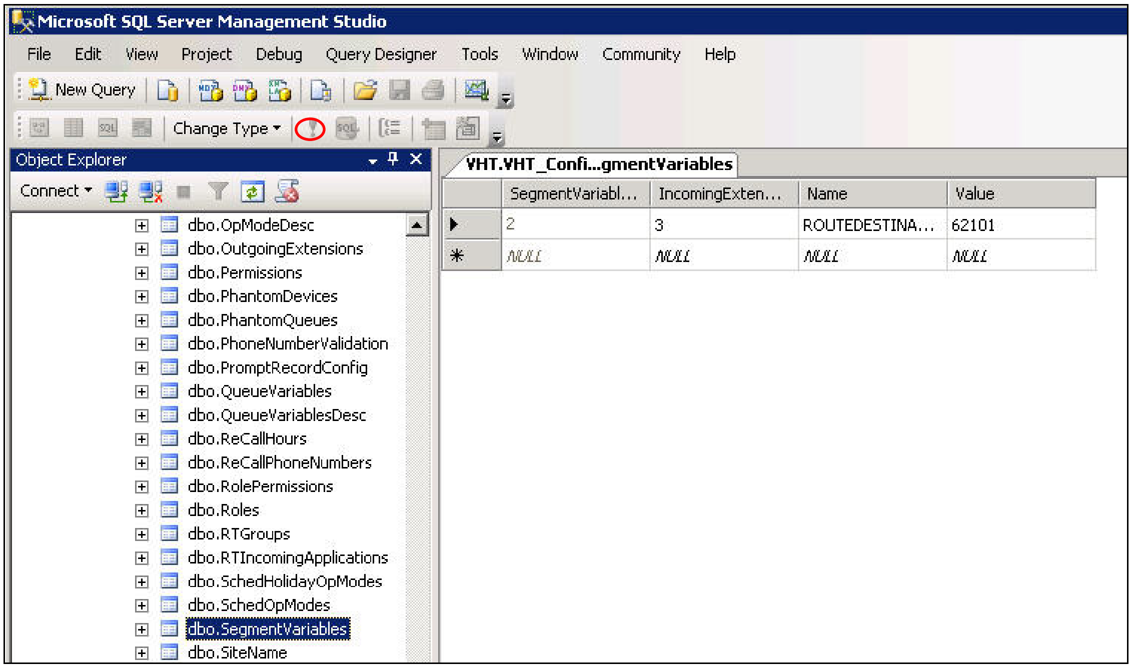


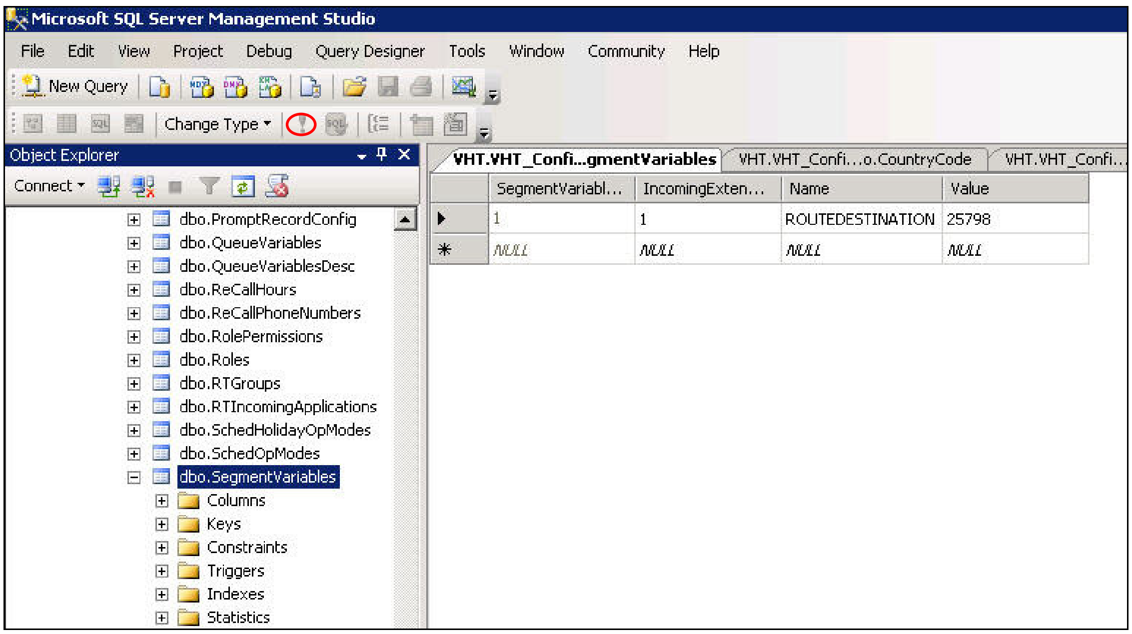
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10.</td><td>Skip the Incoming Application page. On the Phone Number Configuration page, click add (not shown). The PhoneNumberValidation window is displayed. From the drop-down menu of the Country Search field select 1 – North America. Enter 91 in the Dial Prefix field which allows out bound calls to use the ARS (Automatic Route Selection) capability in Communication Manager. Click Update in the left pane. Enter 5 in the Min Length field and 10 in the Max Length field. Click Update in the right pane followed by Close.

</td></tr>
</table>

Step	Description
11.	The **Finished** page is displayed. Click **Finish**. 

Step	Description
12.	On the Virtual Hold server, open **SQL Server Management Studio** by navigating to **Start →All Programs → Microsoft SQL Server 2008 R2 → SQL Server Management Studio**. The **Connect to Server** window is displayed. Click **Connect**. 

Step	Description
13.	From the left pane under **Object Explorer**, navigate to **<Server Hostname>→ Databases → VHT_Config → Tables** where the **<Server Hostname>** is the hostname of the database server. 

Step	Description
14.	An entry has to be created in the **SegmentVariables** table to map an Entry VDN as an incoming extension (See **Section 8**, **Step 9**) to a route destination. Right click **dbo.SegmentVariables** in the left pane and click **Edit Top 200 Rows** (not shown). The following procedure is for the H.323 Configuration: Enter the **Incoming Extension Id** assigned to the Entry VDN (**3**) in the **IncomingExtensionId** field. The **Incoming Extension Id** can be found in the **IncomingExtension** table (not shown). Enter **ROUTEDESTINATION** in the **Name** field and the Routing VDN configured in **Section 4**, **Step 6** (**62101**) in the **Value** field. Click the **Execute SQL** (not shown) button in the **Tools** bar.  Continue on the next page.

Step	Description
	The following procedure is for the SIP Configuration: Enter the **Incoming Extension Id** assigned to the Entry VDN (**1**) in the **IncomingExtensionId** field. The **Incoming Extension Id** can be found in the **IncomingExtension** table (not shown). Enter **ROUTEDESTINATION** in the **Name** field and the extension associated with the Experience Portal inbound application, configured in **Section 7**, **Step 8** (**25798**), in the **Value** field. Click the **Execute SQL** (not shown) button in the **Tools** bar. From the Virtual Hold server, navigate to **Start → Control Panel → Administrative Tools → Services**, restart the **VHT_QueueManager** service. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>15.</td><td>
Navigate to C:\Program Files (x86)\Virtual Hold Technology folder and open OutboundIVR_AVP.xml using notepad.

Replace the IP Address in the URI field with the IP Address of Experience Portal Manager.
Set ApplicationName to the name of the outbound application configured in Section 7, Step 7.
Set AppInterfaceUsername and AppInterfacePassword to the Username and Password configured in Section 7, Step 5.


```
<?xml version="1.0" encoding="utf-8"?>
<LoadBalancerManager>
        <DefaultID>NONE</DefaultID>
        <NumberOfConnectionSets>1</NumberOfConnectionSets>
        <ConnectionSet1>
                <Count>1</Count>
                <Identifier>VHT_Test</Identifier>
                <FirstConnection>Connection1</FirstConnection>
                <LastConnection>Connection1</LastConnection>
                <Connection1>
                        <URI>http://10.64.10.35:8080/axis/services/AppIntfWS</URI>
                        <OutboundANI>8005555555</OutboundANI>
             <!-- AVP provisioned Virtual Hold outbound application -->
                        <ApplicationName>VHT_OB</ApplicationName>
             <AppInterfaceUsername>vhtepws</AppInterfaceUsername>
             <AppInterfacePassword>xxxxxxxxxx</AppInterfacePassword>
             <ConnectTimeout>30</ConnectTimeout>

        <MaxConcurrentOutboundDialRequests>2</MaxConcurrentOutboundDialRequests>

        <WebServiceClientTimeoutInMilliSeconds>180000</WebServiceClientTimeoutInMilliSec
onds>

<SessionParameters>enable_call_classification=true;detect_greeting_end=true</SessionPar
ameters>
             <URLParameters></URLParameters>
                        <TimeToExcludeOnFailure>150000</TimeToExcludeOnFailure>
                        <NextConnectionOnSuccess>Connection1</NextConnectionOnSuccess>
                        <NextConnectionOnFailure>Connection1</NextConnectionOnFailure>

        <NextConnectionOnNoResourcesAvailable>Connection1</NextConnectionOnNoResourcesAv
ailable>
                </Connection1>
        </ConnectionSet1>
</LoadBalancerManager>
```


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>16.</td><td>
Navigate to C:\VirtualHold folder and open toolkit.properties using notepad.
The following procedure is for the H.323 Configuration:

Replace the IP address in the baseurl and webaudiopath parameters with the IP address of the Virtual Hold server.
Set the defaultdestination parameter to the Holding VDN configured in Section 4, Step 6 with a prefix of tel: (tel:62102)
Set the useDnisAsSegment parameter to true
Set the useexternalrouting parameter to true
Set the avp.normaltransferdtmf parameter to tel:#122
Set the disconnectontransfer parameter to tel:#121
Set the avp.disconnectdtmf parameter to tel:#121

In the last three bullet items, the string #12 in the values is the Converse Data Return Code configured in Section 4, Step 9.

```
#sample configuration file for VHT
com.virtualhold.toolkit.loopback=false
com.virtualhold.toolkit.debug=true

#URL for the PTK webservices
com.virtualhold.toolkit.baseurl=http://10.64.101.110/VHTPlatformWS-v4/

#Name file configuration
com.virtualhold.toolkit.audiopath=C:/Program Files/Apache Software Foundation/Tomcat
7.0/webapps/ROOT
com.virtualhold.toolkit.webaudiopath=http://10.64.101.110:8080/

#Default transfer destination if destination cannot be retrieved from PTK
com.virtualhold.toolkit.defaultdestination=tel:62102

#Set this to true if you want to use the call's DNIS as the incoming PTK segment.
com.virtualhold.toolkit.useDnisAsSegment=true

# Default transfer mode (use disconnectontransfer = true if your routing engine retains
call control after <disconnect /> )
com.virtualhold.toolkit.inbound.useexternalrouting=true
com.virtualhold.toolkit.outbound.useexternalrouting=false

# Default transfer mode (use disconnectontransfer = true if your routing engine retains
call control after <disconnect /> )
# Also, this property can be overriden with the URL query string parameter
DisconnectOnTransfer
com.virtualhold.toolkit.avp.normaltransferdtmf=tel:#122
com.virtualhold.toolkit.disconnectontransfer=tel:#121
com.virtualhold.toolkit.avp.disconnectdtmf=tel:#121

#Time group ranges - used in day/time selection
com.virtualhold.toolkit.earlymorning=(12:00 am|6:00 am)
com.virtualhold.toolkit.morning=(6:00 am|12:00 pm)
com.virtualhold.toolkit.afternoon=(12:00 pm|5:00 pm)
com.virtualhold.toolkit.evening=(5:00 pm|9:00 pm)
com.virtualhold.toolkit.night=(9:00 pm|11:59 pm)

#
com.virtualhold.toolkit.avp.uuistoredinascii = false
```

Continue on the next page.
</td></tr>
</table>

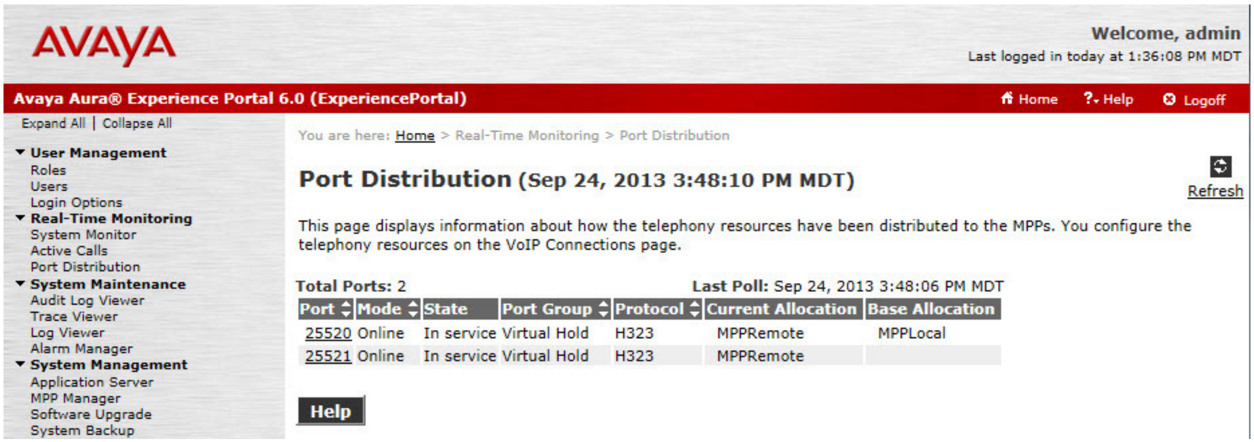
Step	Description
	The following procedure is for the SIP Configuration: • Replace the IP address in the **baseurl** and **webaudiopath** paramters with the IP address of the Virtual Hold server. • Set the **defaultdestination** parameter to the Holding VDN configured in **Section 4**, **Step 8** with a prefix of **tel:** (**tel:62111)** • Set the **useDnisAsSegment** parameter to **true** • Set the **useexternalrouting** parameter to **false** • Set the **disconnectontransfer** parameter to **false** • Set the **avp.uuistoredinascii** parameter to **false**

```
#sample configuration file for VHT
com.virtualhold.toolkit.loopback=false
com.virtualhold.toolkit.debug=true

#URL for the PTK webservices
com.virtualhold.toolkit.baseurl=http://10.64.101.110/VHTPlatformWS-v4/

#Name file configuration
com.virtualhold.toolkit.audiopath=C:/Program Files/Apache Software Foundation/Tomcat
7.0/webapps/ROOT
com.virtualhold.toolkit.webaudiopath=http://10.64.101.110:8080/

#Default transfer destination if destination cannot be retrieved from PTK
com.virtualhold.toolkit.defaultdestination=tel:62111

#Set this to true if you want to use the call's DNIS as the incoming PTK segment.
com.virtualhold.toolkit.useDnisAsSegment=true


# Default transfer mode (use disconnectontransfer = true if your routing engine retains
call control after <disconnect /> )
com.virtualhold.toolkit.inbound.useexternalrouting=false
com.virtualhold.toolkit.outbound.useexternalrouting=false

# Default transfer mode (use disconnectontransfer = true if your routing engine retains
call control after <disconnect /> )
# Also, this property can be overriden with the URL query string parameter
DisconnectOnTransfer
com.virtualhold.toolkit.disconnectontransfer=false

#Time group ranges - used in day/time selection
com.virtualhold.toolkit.earlymorning=(12:00 am|6:00 am)
com.virtualhold.toolkit.morning=(6:00 am|12:00 pm)
com.virtualhold.toolkit.afternoon=(12:00 pm|5:00 pm)
com.virtualhold.toolkit.evening=(5:00 pm|9:00 pm)
com.virtualhold.toolkit.night=(9:00 pm|11:59 pm)

#
com.virtualhold.toolkit.avp.disconnectdtmf=tel:#121

# Used for AVP integrations only
# If UUI data is in hexidecimal format then uuistoredinascii should be false.
# If UUI data is in ASCII format then uuistoredinascii should be true.
# The default value is false, so if the below uuistoredinascii line is not defined it
behaves as if it were set to false.
com.virtualhold.toolkit.avp.uuistoredinascii=false
```

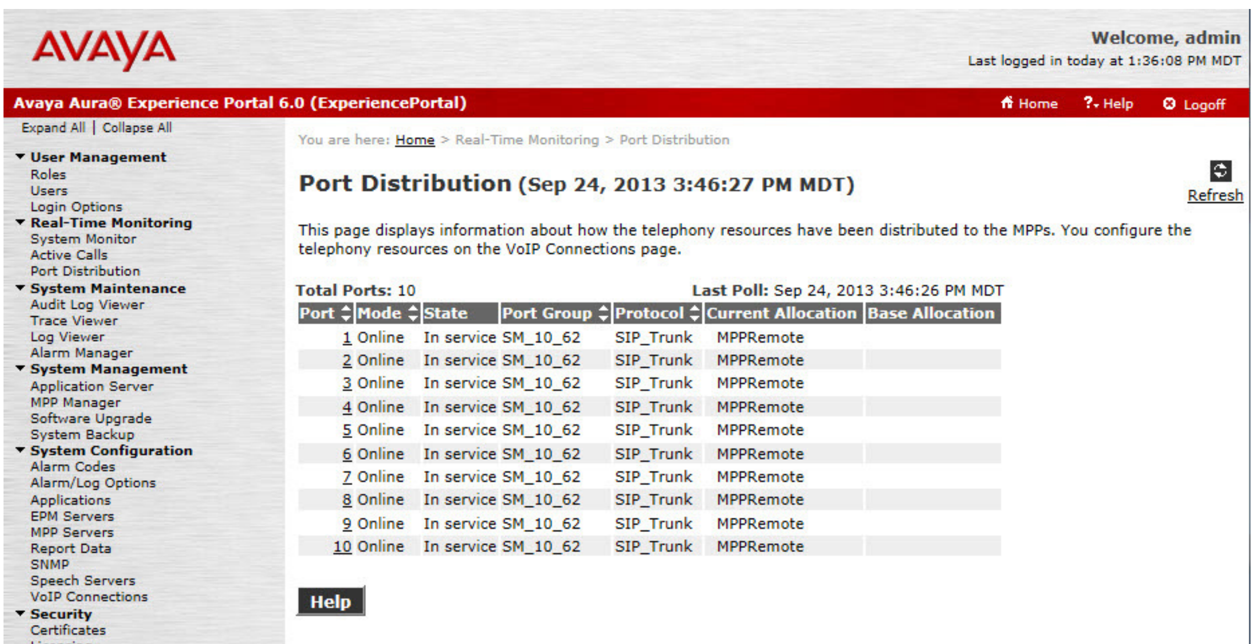
9. Verification Steps

9.1. Avaya Aura® Experience Portal

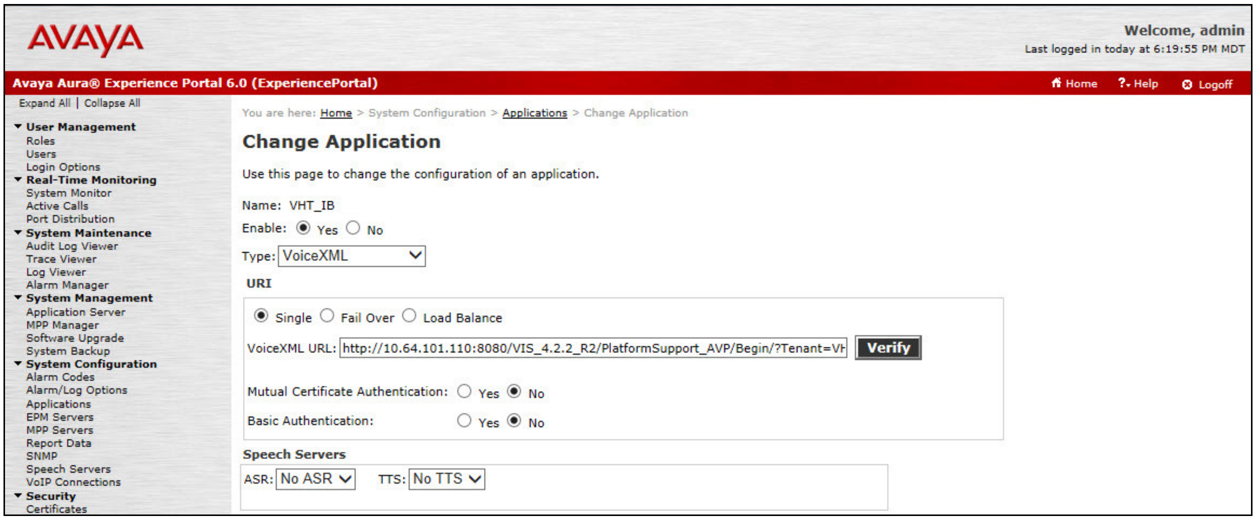
To verify **VoIP Connections** in Experience Portal, click **Real Time Monitoring → Port Distribution** in the left pane. The **State** for the configured ports should be **In service**. The following screenshot is for the H.323 configuration.



The following screenshot is for the SIP configuration.



Click **System Configuration → Applications** in the left pane to display the **Applications** page (not shown). Click the **VHT_IB** application link on the page. The **Change Application** page is displayed. Click the **Verify** button next to the **VoiceXML URL** field.



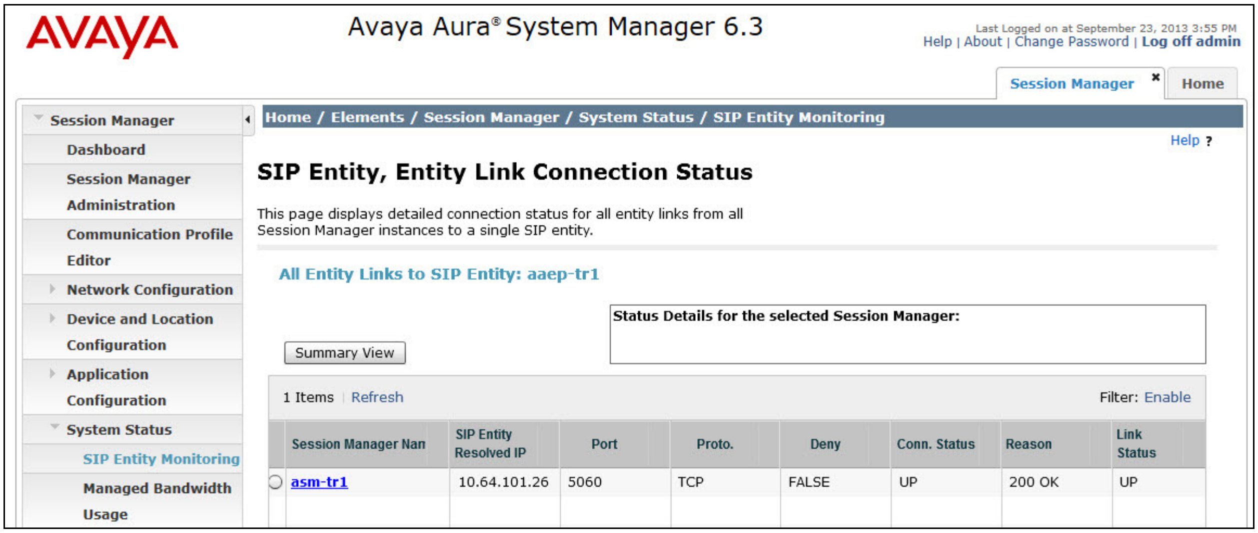
Verify that that the following page is displayed as an indication that the application is accessible.


```
  <?xml version="1.0" encoding="UTF-8"?>
- <vxml version="2.1" xmlns="http://www.w3.org/2001/vxml">
    <property value="0" name="documentmaxage"/>
    <property value="0" name="documentmaxstale"/>
  - <form scope="document" id="InitialForm">
      <var name="PLATFORM_ANI" expr="''"/>
      <var name="PLATFORM_DNIS" expr="''"/>
      <var name="avpUCID" expr="''"/>
      <var name="avpAAI" expr="''"/>
    - <block name="InitialBlock">
        <assign name="PLATFORM_ANI" expr="session.connection.remote.uri"/>
        <assign name="PLATFORM_DNIS" expr="session.connection.local.uri"/>
        <assign name="avpUCID" expr="session.avaya.ucid"/>
        <assign name="avpAAI" expr="session.connection.aai"/>
        <submit namelist="PLATFORM_ANI PLATFORM_DNIS avpUCID avpAAI" method="post" next="/VIS/-/next?
          Action_07381e87a39f48a5b7add4802eb951f7=success.filled&MODE=AVPSIP"/>
      </block>
    - <catch event="connection.disconnect.hangup">
        <goto next="/VIS/-/next?
          Action_07381e87a39f48a5b7add4802eb951f7=error.disconnect.hangup&MODE=AVPSIP"/>
      </catch>
    - <catch event="externalmessage.cpa.machine">
        <goto next="/VIS/-/next?
          Action_07381e87a39f48a5b7add4802eb951f7=externalmessage.cpa.machine&MODE=AVPSIP"/>
      </catch>
    - <catch event="externalmessage.cpa.beep">
        <goto next="/VIS/-/next?
          Action_07381e87a39f48a5b7add4802eb951f7=externalmessage.cpa.beep&MODE=AVPSIP"/>
      </catch>
    </form>
  - <catch event="connection.disconnect.hangup">
      <goto next="/VIS/-/abort?MODE=AVPSIP"/>
    </catch>
  </vxml>
```


Repeat the procedure for the **VHT_OB** application.

9.2. Avaya Aura® Session Manager

To verify connectivity to Experience Portal, click on **Session Manager** on the Home page of System Manager web interface. Navigate to **Session Manager → System Status → SIP Entity Monitoring**. Locate the SIP Entity for Experience Portal under **All Monitored SIP Entities** and click on it. The **Conn. Status** and **Link Status** fields should display **Up**.



10. Conclusion

These Application Notes describe the configuration steps required to integrate VHT Virtual Hold with Avaya Aura® Experience Portal, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and Avaya Aura® Application Enablement Services. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

11. Additional References

This section references the Avaya and Virtual Hold documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] Avaya Aura® Application Enablement Services Administration and Maintenance Guide, Release 6.3, Issue 1, May 2013
[2] Administering Aura® Experience Portal, April 2012
[3] Virtual Hold ACD Configuration Guide for Avaya, May 17, 2013
[4] Virtual Hold AVP/AEP Integration Guide, May 15, 2013
[5] Virtual Hold Version 7.6 EyeQueue User Guide, July 18, 2013
[6] Virtual Hold Version 7.6 System Requirements, July 25, 2013
[7] Virtual Hold Version 7.6 Release Notes, July 18, 2013

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