



Avaya Solution & Interoperability Test Lab

Application Notes for VHT Callback using Native TSAPI with Avaya Aura® Application Enablement Services, Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the steps required to integrate VHT Callback using Native TSAPI with Avaya Aura® Application Enablement Services, Avaya Aura® Communication Manager, and Avaya Aura® Session Manager. VHT Callback is a contact center solution that calculates expected wait time and maintains caller position in a virtual queue. The integration used the Avaya Telephony Services Application Programming Interface from Avaya Aura® Application Enablement Services and the SIP trunk 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 steps required to integrate VHT Callback using Native TSAPI with Avaya Aura® Application Enablement Services (Application Enablement Services), Avaya Aura® Communication Manager (Communication Manager), and Avaya Aura® Session Manager (Session Manager). VHT Callback is a contact center solution that calculates expected wait time and maintains caller position in a virtual queue. The integration used the Avaya Telephony Services Application Programming Interface (TSAPI) from Avaya Aura® Application Enablement Services and the SIP trunk interface from 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 exceeds a pre-defined threshold, then VHT Callback routes the call over Session Manager SIP trunk to the Interactive Voice Gateway (IVG) component of VHT Callback. 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.

Note: 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.

2. General Test Approach

The feature test cases were performed both automatically and manually. Upon startup 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 User-to-User Information (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. In addition, it was verified that Communication Manager routed calls to an available agent or queued the call when the Callback or IVG servers were unavailable.

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

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

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

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

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 below and over the threshold, transfer of inbound calls with received UII data, initiation and transfer of outbound callback calls with priority and saved UII data, and unsuccessful callback attempts.
- Queue statistics using TSAPI real-time adapter in Callback.
- SIP trunk between IVG server and Session Manager using UDP transport.
- IVG response to SIP OPTIONS messages from Session Manager.

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 passed. When the wait time of incoming ACD calls exceeded a pre-defined threshold value, VHT Callback answered the call and gave the caller the option to be called back, schedule a callback, or continue waiting in queue. In addition, a queue statistics report was generated using the TSAPI real-time adapter.

2.3. Support

For technical support on VHT Callback, contact VHT Technical Support through one of the following:

- **Phone:** + 1 (866) 670-2223 (USA)
+44 (0)20 3633 4644 (EMEA)
- **Website:** <https://www.vhtcx.com/contact/contact-center-technical-support/>
- **Email:** support@vhtcx.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 IVG that connected via SIP trunks to Session Manager. 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 “avaya.com”. A 5-digit Uniform Dial Plan was used to facilitate routing of calls with Callback. In the compliance testing, calls to 78701 were routed to the IVG component of Callback.

Device Type	Extension
Skill Group Number	77
Skill Group Extension	77200
Agent Stations	77301, 78030
Agent Login IDs	76301, 76302

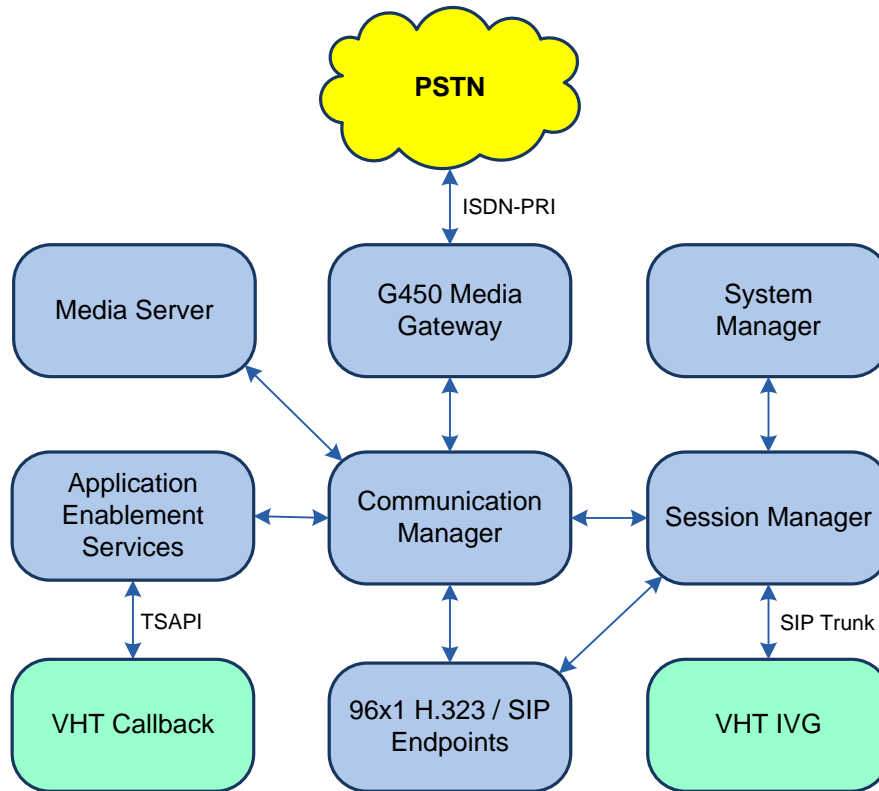


Figure 1: Compliance Testing Configuration

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	8.1.0.1.0-SP1
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.1.121
Avaya Aura® Application Enablement Services	8.1.0.0.9-1
Avaya Aura® System Manager	8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.079814
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya 96x1 IP Deskphones	6.8003 (H.323) 7.1.5.0.11(SIP)
VHT Callback using Native TSAPI on Microsoft Windows Server 2012 R2 Standard with <ul style="list-style-type: none">Avaya AES TSAPI Client	8.11.3 8.0
VHT Interactive Voice Gateway (IVG) on CentOS <ul style="list-style-type: none">Holly Voice Platform (HVP)VXML Interactive Server (VIS)Call Control Interaction Server (CCIS)	3.9.0.31 HVP-6.3.25-2816-43513 6.7.0 3.8.0

5. Configure Avaya Aura® Communication Manager

This section provides the steps for configuring Communication Manager. Administration of Communication Manager was performed using the System Access Terminal (SAT). The procedures include the following areas:

- Verify License
- Administer CTI Link
- Administer System Parameters Features
- Administer Vectors and VDNs
- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Administer AAR Call Routing

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	12
OPTIONAL FEATURES				
IP PORT CAPACITIES		USED		
Maximum Administered H.323 Trunks:		12000	0	
Maximum Concurrently Registered IP Stations:		2400	2	
Maximum Administered Remote Office Trunks:		12000	0	
Maximum Concurrently Registered Remote Office Stations:		2400	0	
Maximum Concurrently Registered IP eCons:		128	0	
Max Concur Registered Unauthenticated H.323 Stations:		100	0	
Maximum Video Capable Stations:		36000	0	
Maximum Video Capable IP Softphones:		2400	1	
Maximum Administered SIP Trunks:		12000	10	
Maximum Administered Ad-hoc Video Conferencing Ports:		12000	0	
Maximum Number of DS1 Boards with Echo Cancellation:		688	0	
(NOTE: You must logoff & login to effect the permission changes.)				

Navigate to **Page 4** and verify that the **Computer Telephony Adjunct Links** customer option is set to “y”.

```
display system-parameters customer-options                                Page  4 of 12
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
    Access Security Gateway (ASG)? 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? y              DCS Call Coverage? y
    ASAI Link Plus Capabilities? y             DCS with Rerouting? y
    Async. Transfer Mode (ATM) PNC? n
    Async. Transfer Mode (ATM) Trunking? n      Digital Loss Plan Modification? y
    ATM WAN Spare Processor? n                 DS1 MSP? y
    ATMS? y                                     DS1 Echo Cancellation? y
    Attendant Vectoring? y

(NOTE: You must logoff & login to effect the permission changes.)
```

Navigate to **Page 7** and verify that the **Vectoring (Basic)** customer option is set to “y”.

```
display system-parameters customer-options                                Page  7 of 12
                                CALL CENTER OPTIONAL FEATURES

                                Call Center Release: 8.0

                                ACD? y
                                BCMS (Basic)? y          Reason Codes? y
                                BCMS/VuStats Service Level? y      Service Level Maximizer? n
    BSR Local Treatment for IP & ISDN? y      Service Observing (Basic)? y
    Business Advocate? n                    Service Observing (Remote/By FAC)? y
    Call Work Codes? y                    Service Observing (VDNs)? y
    DTMF Feedback Signals For VRU? y      Timed ACW? y
    Dynamic Advocate? n                   Vectoring (Basic)? y
    Expert Agent Selection (EAS)? y        Vectoring (Prompting)? y
    EAS-PHD? y                          Vectoring (G3V4 Enhanced)? y
    Forced ACD Calls? n                  Vectoring (3.0 Enhanced)? y
    Least Occupied Agent? y              Vectoring (ANI/II-Digits Routing)? y
    Lookahead Interflow (LAI)? y          Vectoring (G3V4 Advanced Routing)? y
    Multiple Call Handling (On Request)? y  Vectoring (CINFO)? y
    Multiple Call Handling (Forced)? y      Vectoring (Best Service Routing)? y
    PASTE (Display PBX Data on Phone)? y    Vectoring (Holidays)? y
    Vectoring (Variables)? y
(NOTE: You must logoff & login to effect the permission changes.)
```


5.2. Administer CTI Link

Add a CTI link using the **add cti-link** command. 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: 77700	
Type: ADJ-IP	
Name: AES TSAPI Link	COR: 1
Unicode Name? n	

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 19
                           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 19
                           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 UI During Conference/Transfer? n
  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
  Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call? n
```

5.4. Administer Vectors and VDNs

Administer four 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
77201	201	Entry vector & VDN for adjunct route and failure coverage
77202	202	Hold vector & VDN for queuing inbound calls to skill at medium priority
77203	203	Callback vector & VDN for queuing outbound calls to skill at high priority
77204	204	Route vector & VDN for routing calls to IVG and failure coverage

5.4.1. Entry Vector and VDN

Modify an available vector using the **change vector** command. 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 failure from the adjunct routing step. In the compliance test, the covering point is the Hold VDN, which is administered in **Section 5.4.2**.

change vector 201	Page 1 of 6
CALL VECTOR	
Number: 201	Name: VHT Entry
Multimedia? n	Attendant Vectoring? n
Basic? y	EAS? y
Prompting? y	LAI? y
Variables? y	3.0 Enhanced? y
01 adjunct	routing link 1
02 wait-time	10 secs hearing music
03 route-to	number 77202
04	with cov n if unconditionally

Add a VDN using the **add vdn** command. Enter a descriptive **Name** and the vector number specified above for **Vector Number**. Retain the default values for all remaining fields.

add vdn 77201	Page 1 of 3
VECTOR DIRECTORY NUMBER	
Extension: 77201	
Name*: VHT Entry	
Destination: Vector Number 201	
Attendant Vectoring? n	
Meet-me Conferencing? n	
Allow VDN Override? n	
COR: 1	
TN*: 1	
Measured: none	
Report Adjunct Calls as ACD*? n	

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 77 is the existing skill group number mentioned in **Section 3**.

```
change vector 202                                     Page 1 of 6
                                     CALL VECTOR

      Number: 202           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? y      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 77      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 specified above for **Vector Number**.

```
add vdn 77202                                     Page 1 of 3
                                     VECTOR DIRECTORY NUMBER

                                     Extension: 77202
                                     Name*: VHT Hold
                                     Destination: Vector Number      202
      Attendant Vectoring? n
      Meet-me Conferencing? n
      Allow VDN Override? n
      COR: 1
      TN*: 1
      Measured: none      Report Adjunct Calls as ACD*? n
```

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 77 is the existing skill group number mentioned in **Section 3**.

change vector 203	CALL VECTOR	Page 1 of 6
Number: 203 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? y G3V4 Adv Route? y	CINFO? y BSR? y Holidays? y
Variables? y	3.0 Enhanced? y	
01 queue-to	skill 77	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 specified above for **Vector Number**.

add vdn 77203	VECTOR DIRECTORY NUMBER	Page 1 of 3
Extension: 77203		
Name*: VHT Callback		
Destination: Vector Number		203
Attendant Vectoring? n		
Meet-me Conferencing? n		
Allow VDN Override? n		
COR: 1		
TN*: 1		
Measured: none		Report Adjunct Calls as ACD*? n

5.4.4. Route Vector and VDN

Modify an available vector for Callback server to route calls to IVG using extension 78701. If the call to IVG fails for any reason, the incoming ACD call will be routed to the ACD skill where the call will either be queued or answered by an available agent. This ensures that the call is properly routed by Communication Manager even if the call attempt to IVG fails.

change vector 204	Page 1 of 6
CALL VECTOR	
Number: 204 Name: VHT Route	
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 silence
02 route-to	number 78701 with cov n if unconditionally
03 wait-time	2 secs hearing ringback
04 route-to	number 77202 with cov n if unconditionally
05 disconnect	after announcement none
06 stop	
07	

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

add vdn 77204	Page 1 of 3
VECTOR DIRECTORY NUMBER	
Extension: 77204	
Name*: VHT Route	
Destination: Vector Number 204	
Attendant Vectoring? n	
Meet-me Conferencing? n	
Allow VDN Override? n	
COR: 1	
TN*: 1	
Measured: none	Report Adjunct Calls as ACD*? n

5.5. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

change node-names ip	Page 1 of 2
IP NODE NAMES	
Name	IP Address
default	0.0.0.0
devcon-ams	10.64.102.118
devcon-sm	10.64.102.117
procr	10.64.102.115
procr6	::

5.6. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to IVG. The form is accessed via the **change ip-codec-set** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, *G.711MU* was used.

change ip-codec-set 1				Page	1 of	2
IP CODEC SET						
Codec Set: 1						
Audio	Silence	Frames	Packet			
Codec	Suppression	Per Pkt	Size (ms)			
1: G.711MU	n	2	20			
2:						
3:						
4:						
5:						
6:						
7:						

5.7. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IVG and IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Media Server. Note that calls to the PSTN are not shuffled. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

change ip-network-region 1		Page	1 of	20
IP NETWORK REGION				
Region: 1				
Location: 1		Authoritative Domain: avaya.com		
Name:		Stub Network Region: n		
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: yes		
Codec Set: 1		Inter-region IP-IP Direct Audio: yes		
UDP Port Min: 2048		IP Audio Hairpinning? n		
UDP Port Max: 3329				
DIFFSERV/TOS PARAMETERS				
Call Control PHB Value: 46				
Audio PHB Value: 46				
Video PHB Value: 26				
802.1P/Q PARAMETERS				
Call Control 802.1p Priority: 6				
Audio 802.1p Priority: 6				
Video 802.1p Priority: 5				
AUDIO RESOURCE RESERVATION PARAMETERS				
H.323 IP ENDPOINTS				
H.323 Link Bounce Recovery? y				
Idle Traffic Interval (sec): 20				
Keep-Alive Interval (sec): 5				
Keep-Alive Count: 5				
RSVP Enabled? n				

5.8. Administer SIP Signaling Group

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

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify Communication Manager (*procr*) and the Session Manager (*devcon-sm*) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Enable **Direct IP-IP Audio Connections** to allow the call to be shuffled.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 10		Page 1 of 2	
SIGNALING GROUP			
Group Number: 10	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: SM	Clustered? n	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y			
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n			
Alert Incoming SIP Crisis Calls? n			
Near-end Node Name: procr		Far-end Node Name: devcon-sm	
Near-end Listen Port: 5061		Far-end Listen Port: 5061	
		Far-end Network Region: 1	
Far-end Domain: avaya.com			
Incoming Dialog Loopbacks: eliminate		Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload		RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3		Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y		IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n		Initial IP-IP Direct Media? n	
		Alternate Route Timer(sec): 6	

5.9. Administer SIP Trunk Group

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to IVG and SIP stations. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

add trunk-group 10		Page 1 of 22	
TRUNK GROUP			
Group Number: 10	Group Type: sip	CDR Reports: y	
Group Name: To devcon-sm	COR: 1	TN: 1	TAC: 1010
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: 10	
		Number of Members: 10	

On **Page 3** of the trunk group form, set the **UI Treatment** field to *shared* and enable the **Send UCID** option.

add trunk-group 10		Page 3 of 22	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Suppress # Outpulsing? n	Numbering Format: private		
		UI Treatment: shared	
		Maximum Size of UI Contents: 128	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
		Hold/Unhold Notifications? y	
		Modify Tandem Calling Number: no	
Send UCID? y			
Show ANSWERED BY on Display? y			

5.10. Administer AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with '78'. This would cover call routing to IVG (i.e., 78701).

change uniform-dialplan 7					Page 1 of 2	
UNIFORM DIAL PLAN TABLE					Percent Full: 0	
Matching Pattern	Len	Del	Insert Digits	Net Conv Num	Node	
78	5	0		aar n		

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with "78" to route pattern 10 as shown below. Note that the **Call Type** was set to *lev0*. This entry routes calls to IVG and SIP stations.

change aar analysis 7						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
7		7	7	254	aar		n
78		5	5	10	lev0		n
8		7	7	254	aar		n
9		7	7	254	aar		n

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

change route-pattern 10										Page 1 of 3	
Pattern Number: 10										Pattern Name: To devcon-sm	
SCCAN? n		Secure SIP? n		Used for SIP stations? n							
Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Dgits	DCS/ QSIG Intw	IXC		
1: 10	0							n	user		
2:								n	user		
3:								n	user		
4:								n	user		
5:								n	user		
6:								n	user		
BCC VALUE		TSC	CA-TSC		ITC BCIE		Service/Feature	PARM Sub	Numbering	LAR	
0 1 2 M 4 W			Request					Dgts	Format		
1:	y y y y y n	n			rest				unk-unk	none	
2:	y y y y y n	n			rest					none	
3:	y y y y y n	n			rest					none	
4:	y y y y y n	n			rest					none	
5:	y y y y y n	n			rest					none	
6:	y y y y y n	n			rest					none	

6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer SIP Entities
- Administer Routing Policies
- Administer Dial Patterns

6.1. Launch System Manager

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL “https://<ip-address>”, where <ip-address> is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.

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

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

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

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

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

User ID:

Password:

[Change Password](#)

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

6.2. Administer SIP Entities

In the sample configuration, two SIP entities were added for Communication Manager and IVG.

6.2.1. SIP Entity for Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., procr) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously (not shown).
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile 'admin' are also present. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form contains the following fields: 'Name' (devcon-cm), 'FQDN or IP Address' (10.64.102.115), 'Type' (CM), 'Notes' (empty), 'Adaptation' (dropdown), 'Location' (Thornton), 'Time Zone' (America/New_York), 'SIP Timer B/F (in seconds)' (4), 'Minimum TLS Version' (Use Global Setting), 'Credential name' (empty), 'Securable' (checkbox), 'Call Detail Recording' (none), 'Loop Detection Mode' (On), 'Loop Count Threshold' (5), 'Loop Detection Interval (in msec)' (200), 'SIP Link Monitoring' (Use Session Manager Configuration), and 'CRLF Keep Alive Monitoring' (Use Session Manager Configuration). 'Commit' and 'Cancel' buttons are located at the top right of the form area.

Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. The SIP trunk from Session Manager to Communication Manager is described by an Entity link. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon-cm link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol (e.g., *TLS*).
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *Trusted*.

Click **Commit** to save the Entity Link definition.

Entity Links

Override Port & Transport with DNS SRV: ☐

Add Remove		Filter: Enable						
1 Item								
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* devcon-cm Link	devcon-sm	TLS	* 5061	devcon-cm	* 5061	trusted	<input type="checkbox"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove		Filter: Enable	
0 Items			
<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes

Commit Cancel

6.2.2. SIP Entity for IVG

A SIP Entity must be added for IVG. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of IVG.
- **Type:** Select *SIP Trunk*.
- **Location:** Select one of the locations defined previously (not shown).
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and various menu items: Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon labeled 'admin' are also present. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** VHT-IVG
- FQDN or IP Address:** 10.64.102.107
- Type:** SIP Trunk (dropdown)
- Notes:** (empty text area)
- Adaptation:** (empty dropdown)
- Location:** Thornton (dropdown)
- Time Zone:** America/New_York (dropdown)
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting (dropdown)
- Credential name:** (empty text area)
- Securable:** (checkbox, unchecked)
- Call Detail Recording:** egress (dropdown)
- Loop Detection Mode:** On (dropdown)
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)
- CRLF Keep Alive Monitoring:** Use Session Manager Configuration (dropdown)

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area. A 'Help ?' link is also visible.

Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. The SIP trunk from Session Manager to IVG is described by an Entity link. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *VHT-IVG Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol (e.g., *UDP*).
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of IVG.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *Trusted*.

Click **Commit** to save the Entity Link definition.

Entity Links

Override Port & Transport with DNS SRV: ☐

Add		Remove							
1 Item									Filter: Enable
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	
<input type="checkbox"/>	* VHT-IVG Link	devcon-sm	UDP	* 5060	VHT-IVG	* 5060	trusted	<input type="checkbox"/>	

Select : All, None

SIP Responses to an OPTIONS Request

Add		Remove				
0 Items						Filter: Enable
<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes			

Commit Cancel

6.3. Administer Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.2**. Routing policies were added for Communication Manager and IVG.

6.3.1. Routing Policy for Communication Manager

To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, version 8.1, and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile (admin) are also present. The left sidebar shows a tree view with categories like Domains, Locations, Conditions, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (selected), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Routing Policy Details' and contains three sections: General, SIP Entity as Destination, and Time of Day. The General section has fields for Name (devcon-cm Policy), Disabled (checkbox), Retries (0), and Notes. The SIP Entity as Destination section has a 'Select' button and a table with columns Name, FQDN or IP Address, Type, and Notes. The Time of Day section has buttons for Add, Remove, and View Gaps/Overlaps, and a table with columns for Ranking, Name, days of the week, Start Time, End Time, and Notes.

Routing Policy Details Commit Cancel Help ?

General

* **Name:** devcon-cm Policy

Disabled: ☐

* **Retries:** 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
devcon-cm	10.64.102.115	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

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

Select : All, None

6.3.2. Routing Policy for IVG

To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, "Aura® System Manager 8.1", and several menu items: Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon labeled "admin" are also present. The left sidebar contains a list of navigation options: Home, Routing, Domains, Locations, Conditions, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "Routing Policy Details" and includes "Commit" and "Cancel" buttons. The "General" section contains fields for "Name" (VHT-IVG Policy), "Disabled" (checkbox), "Retries" (0), and "Notes". The "SIP Entity as Destination" section features a "Select" button and a table with columns: Name, FQDN or IP Address, Type, and Notes. The table contains one entry: VHT-IVG, 10.64.102.107, SIP Trunk. The "Time of Day" section includes "Add", "Remove", and "View Gaps/Overlaps" buttons, a "Filter: Enable" option, and a table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, and Notes. The table shows one item with Ranking 0, Name 24/7, and Start/End times of 00:00 and 23:59. A "Select : All, None" option is at the bottom.

Routing Policy Details Commit Cancel Help ?

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
VHT-IVG	10.64.102.107	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

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

Select : All, None

6.4. Administer Dial Patterns

Dial patterns must be defined to direct calls to the appropriate SIP Entity. Dial patterns were added for Communication Manager and IVG.

6.4.1. Dial Patterns for Communication Manager

In the sample configuration, 5-digit extensions starting with '7' and 10-digit numbers prepended with the ARS access code '9' and prefix code '1' were routed to local stations and PSTN, respectively, via Communication Manager. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for routing calls to local stations on Communication Manager.

AVAYA
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾ Search 🔍 🔔 ≡ admin

Home Routing

Routing

Domains

Locations

Conditions

Adaptations ▾

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Origination Dial...

Dial Pattern Details Commit Cancel Help ?

General

* Pattern: 7

* Min: 5

* Max: 5

Emergency Call: ☐

SIP Domain: -ALL- ▾

Notes: CM Stations

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name ▾	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Thornton		devcon-cm Policy	0	<input type="checkbox"/>	devcon-cm	

Select : All, None

AVAYA

Aura® System Manager 8.1

Users

Elements

Services

Widgets

Shortcuts

Search

admin

Home

Routing

Routing

Domains

Locations

Conditions

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Origination Dial...

Dial Pattern Details

CommitCancel

Help

General

* Pattern: 91

* Min: 12

* Max: 12

Emergency Call: ☐

SIP Domain: -ALL-

Notes: PSTN

Originating Locations and Routing Policies

AddRemove

1 Item

Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Thornton		devcon-cm Policy	0	<input type="checkbox"/>	devcon-cm	

Select : All, None

6.4.2. Dial Pattern for IVG

In the sample configuration, 78701 was routed to IVG. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definition for routing calls to IVG.

AVAYA Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾ Search 🔍 🔔 ≡ admin

Home Routing

Routing ↑

Domains

Locations

Conditions

Adaptations ▾

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Origination Dial...

Dial Pattern Details

Commit Cancel Help ?

General

* Pattern: 78701

* Min: 5

* Max: 5

Emergency Call: ☐

SIP Domain: -ALL- ▾

Notes: VHT IVG

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Thornton		VHT-IVG Policy	0	<input type="checkbox"/>	VHT-IVG	

Select : All, None

7. Configure Avaya Aura® Application Enablement Services

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

- Launch OAM Interface
- Verify License
- Administer TSAPI Link
- Restart Service
- Obtain Tlink Name
- Administer Callback User
- Verify Security Database

7.1. 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 login screen is displayed. Log in using the appropriate credentials.



Application Enablement Services Management Console


[Help](#)

Please login here:

Username

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

**Application Enablement Services**
Management Console

Welcome: User cust
Last login: Mon Aug 5 14:02:37 2019 from 192.168.100.251
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.9-1
Server Date and Time: Thu Aug 08 11:04:41 EDT 2019
HA Status: Not Configured

Home | [Help](#) | [Logout](#)

- ▶ AE Services
- ▶ Communication Manager Interface
- ▶ High Availability
- ▶ Licensing
- ▶ Maintenance
- ▶ Networking
- ▶ Security
- ▶ Status
- ▶ User Management
- ▶ Utilities
- ▶ Help

Welcome to OAM

The AE Services Operations, Administration, and Management (OAM) Web provides you with tools for managing the AE Server. OAM spans the following administrative domains:


- AE Services - Use AE Services to manage all AE Services that you are licensed to use on the AE Server.
- Communication Manager Interface - Use Communication Manager Interface to manage switch connection and dialplan.
- High Availability - Use High Availability to manage AE Services HA.
- Licensing - Use Licensing to manage the license server.
- Maintenance - Use Maintenance to manage the routine maintenance tasks.
- Networking - Use Networking to manage the network interfaces and ports.
- Security - Use Security to manage Linux user accounts, certificate, host authentication and authorization, configure Linux-PAM (Pluggable Authentication Modules for Linux) and so on.
- Status - Use Status to obtain server status informations.
- User Management - Use User Management to manage AE Services users and AE Services user-related resources.
- Utilities - Use Utilities to carry out basic connectivity tests.
- Help - Use Help to obtain a few tips for using the OAM Help system

Depending on your business requirements, these administrative domains can be served by one administrator for all domains, or a separate administrator for each domain.

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7.2. Verify License

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

**Application Enablement Services**
Management Console

Welcome: User cust
Last login: Mon Aug 5 14:02:37 2019 from 192.168.100.251
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.9-1
Server Date and Time: Thu Aug 08 11:14:12 EDT 2019
HA Status: Not Configured

Licensing | [Home](#) | [Help](#) | [Logout](#)

- ▶ AE Services
- ▶ Communication Manager Interface
- ▶ High Availability
- ▼ Licensing
 - WebLM Server Address
 - WebLM Server Access**
 - Reserved Licenses
- ▶ Maintenance
- ▶ Networking

Licensing

If you are setting up and maintaining the WebLM, you need to use the following:

- WebLM Server Address

If you are importing, setting up and maintaining the license, you need to use the following:

- WebLM Server Access

If you want to administer TSAPI Reserved Licenses or DMCC Reserved Licenses, you need to use the following:

- Reserved Licenses

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 MEDIUM SWITCH** for the virtual server.

WebLM Home	Application Enablement (CTI) - Release: 8 - SID: 10503000	Standard License file																																				
Install license	You are here: Licensed Products > Application_Enablement > View License Capacity																																					
Licensed products	License installed on: June 28, 2019 12:26:36 PM -04:00																																					
APPL_ENAB																																						
▼ Application_Enablement																																						
View license capacity	<div>License File Host IDs: V7-94-F5-41-87-5E-01</div>																																					
View peak usage																																						
ASBCE	<div>Licensed Features</div>																																					
►Session_Border_Controller_E_AE																																						
COMMUNICATION_MANAGER	<div>13 Items Show All ▼</div> <table border="1"> <thead> <tr> <th>Feature (License Keyword)</th> <th>Expiration date</th> <th>Licensed capacity</th> </tr> </thead> <tbody> <tr> <td>Device Media and Call Control VALUE_AES_DMCC_DMC</td> <td>permanent</td> <td>10000</td> </tr> <tr> <td>AES ADVANCED LARGE SWITCH VALUE_AES_AEC_LARGE_ADVANCED</td> <td>permanent</td> <td>16</td> </tr> <tr> <td>AES HA LARGE VALUE_AES_HA_LARGE</td> <td>permanent</td> <td>1</td> </tr> <tr> <td>AES ADVANCED MEDIUM SWITCH VALUE_AES_AEC_MEDIUM_ADVANCED</td> <td>permanent</td> <td>16</td> </tr> <tr> <td>Unified CC API Desktop Edition VALUE_AES_AEC_UNIFIED_CC_DESKTOP</td> <td>permanent</td> <td>10000</td> </tr> <tr> <td>CVLAN ASAI VALUE_AES_CVLAN_ASAI</td> <td>permanent</td> <td>16</td> </tr> <tr> <td>AES HA MEDIUM VALUE_AES_HA_MEDIUM</td> <td>permanent</td> <td>1</td> </tr> <tr> <td>AES ADVANCED SMALL SWITCH VALUE_AES_AEC_SMALL_ADVANCED</td> <td>permanent</td> <td>16</td> </tr> <tr> <td>DLG VALUE_AES_DLG</td> <td>permanent</td> <td>16</td> </tr> <tr> <td>TSAPI Simultaneous Users VALUE_AES_TSAPI_USERS</td> <td>permanent</td> <td>10000</td> </tr> <tr> <td>CVLAN Proprietary Links VALUE_AES_PROPRIETARY_LINKS</td> <td>permanent</td> <td>16</td> </tr> </tbody> </table>		Feature (License Keyword)	Expiration date	Licensed capacity	Device Media and Call Control VALUE_AES_DMCC_DMC	permanent	10000	AES ADVANCED LARGE SWITCH VALUE_AES_AEC_LARGE_ADVANCED	permanent	16	AES HA LARGE VALUE_AES_HA_LARGE	permanent	1	AES ADVANCED MEDIUM SWITCH VALUE_AES_AEC_MEDIUM_ADVANCED	permanent	16	Unified CC API Desktop Edition VALUE_AES_AEC_UNIFIED_CC_DESKTOP	permanent	10000	CVLAN ASAI VALUE_AES_CVLAN_ASAI	permanent	16	AES HA MEDIUM VALUE_AES_HA_MEDIUM	permanent	1	AES ADVANCED SMALL SWITCH VALUE_AES_AEC_SMALL_ADVANCED	permanent	16	DLG VALUE_AES_DLG	permanent	16	TSAPI Simultaneous Users VALUE_AES_TSAPI_USERS	permanent	10000	CVLAN Proprietary Links VALUE_AES_PROPRIETARY_LINKS	permanent	16
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AES HA MEDIUM VALUE_AES_HA_MEDIUM	permanent	1																																				
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CVLAN Proprietary Links VALUE_AES_PROPRIETARY_LINKS	permanent	16																																				
►Call_Center																																						
►Communication_Manager																																						
MESSAGING																																						
►Messaging																																						
MSR																																						
►Media_Server																																						
SYSTEM_MANAGER																																						
►System_Manager																																						
SessionManager																																						
►SessionManager																																						
VSS																																						
►Voice_Portal																																						
Uninstall license																																						
Server properties																																						
Shortcuts																																						
Help for Licensed products																																						

7.3. 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**.

AVAYA Application Enablement Services
Management Console

Welcome: User cust
Last login: Mon Aug 5 14:02:37 2019 from 192.168.100.251
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.0.9-1
Server Date and Time: Thu Aug 08 11:15:41 EDT 2019
HA Status: Not Configured

AE Services | TSAPI | TSAPI LinksHome | Help | Logout

▼ AE Services

- ▶ CVLAN
- ▶ DLG
- ▶ DMCC
- ▶ SMS
- ▼ TSAPI
 - TSAPI Links
 - TSAPI Properties

TSAPI Links

Link	Switch Connection	Switch CTI Link #	ASAI Link Version	Security
Add Link	Edit Link	Delete 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 *devcon* is selected. For **Switch CTI Link Number**, select the CTI link number from **Section 5.2**. Retain the default values in the remaining fields.

AVAYA Application Enablement Services
Management Console

Welcome: User cust
Last login: Mon Aug 5 14:02:37 2019 from 192.168.100.251
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.0.9-1
Server Date and Time: Thu Aug 08 11:17:11 EDT 2019
HA Status: Not Configured

AE Services | TSAPI | TSAPI LinksHome | Help | Logout

▼ AE Services

- ▶ CVLAN
- ▶ DLG
- ▶ DMCC
- ▶ SMS
- ▼ TSAPI
 - TSAPI Links
 - TSAPI Properties
- ▶ TWS

Add TSAPI Links

Link1 ▼
Switch Connectiondevcon ▼
Switch CTI Link Number1 ▼
ASAI Link Version7 ▼
SecurityBoth ▼
[Apply Changes](#) [Cancel Changes](#)

7.4. 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**.

- ▶ AE Services
- ▶ Communication Manager Interface
- High Availability
- ▶ Licensing
- ▼ Maintenance
 - Date Time/NTP Server
 - ▶ Security Database
 - Service Controller**
 - ▶ Server Data
- ▶ Networking
- ▶ Security
- ▶ Status
- ▶ User Management
- ▶ Utilities
- ▶ Help

Service Controller


Service	Controller Status
<input type="checkbox"/> ASAI Link Manager	Running
<input type="checkbox"/> DMCC Service	Running
<input type="checkbox"/> CVLAN Service	Running
<input type="checkbox"/> DLG Service	Running
<input type="checkbox"/> Transport Layer Service	Running
<input checked="" type="checkbox"/> TSAPI Service	Running

For status on actual services, please use [Status and Control](#)

7.5. Obtain Tlink Name

Select **Security** → **Security Database** → **Tlinks** from the left pane. The **Tlinks** screen shows a listing of 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#DEVCON#CSTA#DEVCON-AES”. Note the use of the switch connection “DEVCON” from **Section 7.3** as part of the Tlink name.

 **Application Enablement Services**
Management Console

Welcome: User cust
Last login: Mon Aug 5 14:02:37 2019 from 192.168.100.251
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.9-1
Server Date and Time: Thu Aug 08 11:20:23 EDT 2019
HA Status: Not Configured

Security | Security Database | Tlinks**Home | Help | Logout**

▶ AE Services

▶ Communication Manager Interface

▶ High Availability

▶ Licensing

▶ Maintenance

▶ Networking

▼ **Security**

▶ Account Management

▶ Audit

▶ Certificate Management

▶ Enterprise Directory

▶ Host AA

▶ PAM

▼ **Security Database**

▪ Control

▣ CTI Users

▪ Devices

▪ Device Groups

▪ **Tlinks**

Tlinks

Tlink Name

☒ AVAYA#DEVCON#CSTA#DEVCON-AES

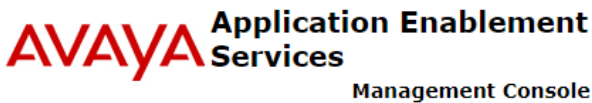
☐ AVAYA#DEVCON#CSTA-S#DEVCON-AES

Delete Tlink

7.6. 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.



Welcome: User cust
Last login: Thu Aug 8 11:52:26 2019 from 192.168.100.250
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.9-1
Server Date and Time: Thu Aug 08 12:10:18 EDT 2019
HA Status: Not Configured

[User Management](#) | [User Admin](#) | [Add User](#)[Home](#) | [Help](#) | [Logout](#)

▶ AE Services

▶ Communication Manager Interface

▶ High Availability

▶ Licensing

▶ Maintenance

▶ Networking

▶ Security

▶ Status

▼ User Management

▶ Service Admin

▼ User Admin

▪ [Add User](#)

▪ Change User Password

▪ List All Users

▪ Modify Default Users

▪ Search Users

▶ Utilities

▶ Help

Add User

Fields marked with * can not be empty.

* User Id

* Common Name

* Surname

* User Password

* Confirm Password

Admin Note

Avaya Role

Business Category

Car License

CM Home

Css Home

CT User

Department Number

Display Name


Employee Number

Employee Type

7.7. 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.

Verify that **Enable SDB for TSAPI Service, JTAPI and Telephony Web Services** is unchecked. In the event that security database is used by the customer with this parameter already enabled, then follow [3] to configure access privileges for the Callback user from **Section 7.6**.

**Application Enablement Services**
Management Console

Welcome: User cust
Last login: Thu Aug 8 11:52:26 2019 from 192.168.100.250
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.0.9-1
Server Date and Time: Thu Aug 08 12:11:39 EDT 2019
HA Status: Not Configured

Security | Security Database | Control

Home | Help | Logout

▶ AE Services

▶ Communication Manager Interface

▶ High Availability

▶ Licensing

▶ Maintenance

▶ Networking

▼ Security

▶ Account Management

▶ Audit

▶ Certificate Management

▶ Enterprise Directory

▶ Host AA

▶ PAM

▼ Security Database

▪ Control

⊞ CTI Users

SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services

☐ Enable SDB for DMCC Service

☐ Enable SDB for TSAPI Service, JTAPI and Telephony Web Services

Apply Changes

8. Configure VHT Interactive Voice Gateway (IVG)

Configuration is accomplished by accessing the browser-based IVG management system using the URL “http://<ip-address>:2020”, where <ip-address> is the IP address of the IVG server. Log in with the appropriate credentials (not shown).

From the IVG management system, navigate to **Administration** → **Service Providers** to display the **Service Provider Editor** shown below. In the **Service Provider** field, select the appropriate site name (e.g., *VHT*) and enter the desired **Domain Name** and **Domain Properties**. Scroll down to the **License Port Allocation** section and set the **Max Available Ports**.

Note: Alternatively, the VHT IVG application provisioning can be configured automatically during the install using the IVG installer. Refer to [6] for details.

The screenshot shows the VHT Service Provider Editor web interface. At the top, there is a header with the VHT logo (a red circle with 'vht' inside) and the text 'Powered by the Holly Voice Platform'. To the right of the logo, there is a version number 'HVP-6.3.25-2816-43513'. Below the logo, there are navigation tabs: 'Administration', 'Reports', 'Configuration', and 'Dashboard'. To the right of these tabs, there are dropdown menus for '<all service providers>', '<all affiliates>', and '<all applications>'. Below these, there is a user login area showing 'user: administrator' and a 'Logout' link. The main content area is titled 'Service Provider Editor' in orange. It contains three main sections: 1. 'Select Service Provider' with a dropdown menu for 'Service Provider' (set to 'VHT'), and text input fields for 'Domain Name' and 'Domain Description' (both set to 'VHT'). 2. 'Service Provider Contact Details' with text input fields for 'Name', 'Email', 'Phone', and 'Address'. 3. 'Licence Port Allocation' with text input fields for 'Max Available Ports' (set to '999') and 'Warn Ports' (set to '900'). There is an 'Edit Affiliates...' button between the first and second sections.

HVP-6.3.25-2816-43513

vht Powered by the Holly Voice Platform

<all service providers> <all affiliates> <all applications>

Administration Reports Configuration Dashboard user: administrator Logout

Service Provider Editor

Select Service Provider

Service Provider: VHT

Domain Name: VHT

Domain Description: VHT

Edit Affiliates...

Service Provider Contact Details

Name:

Email:

Phone:

Address:

Licence Port Allocation

Max Available Ports: 999 Warn Ports: 900

Scroll down to the **Application Parameters** section and click **Save Service Provider**. In the **Numbers Available** section, add the **DNIS Numbers**. The DNIS numbers were set to *78701*, which is used to route calls to IVG, and *outbound* as shown below.

The screenshot displays two configuration windows. The top window, titled "Application Parameters", features a "Key:" and "Value:" input area. Below these, a text box contains the value "type = application/voicexml+xml". To the right of this text box are three buttons: "Set", "Replace", and "Delete". Below the text box is a "Preset Parameters:" section with a dropdown menu currently showing "Set Application Type To CCXML" and a "Set" button. At the bottom of this window are three buttons: "Delete the Service Provider", "Revert", and "Save Service Provider".

Below the first window is a section titled "Service Provider Numbers" in orange text. Underneath this is a second window titled "Numbers Available". This window has a "DNIS Numbers:" label followed by two input fields separated by a hyphen. Below the input fields is a list box containing two entries: "78701 - 78701" and "outbound - outbound". To the right of the list box are three buttons: "Add", "Replace", and "Delete".

Navigate to **Administration** → **Affiliates** to display the **Affiliate Editor** shown below. In the **Service Provider** field, select the appropriate site name (e.g., *VHT*) and enter the desired **Domain Name** and **Domain Properties**. During the initial configuration of the affiliate, the **Affiliate** field should be set to *<new affiliate>* from the drop-down menu.

vht Powered by the Holly Voice Platform HVP-6.3.25-2616-43513

<all service providers> <all affiliates> <all applications>

Administration Reports Configuration Dashboard user: administrator Logout

Affiliate Editor

Select Affiliate

Service Provider: VHT

Affiliate: VHT

Domain Name: VHT-aff

Domain Description: VHT

Edit Service Provider... Edit Applications...

Affiliate Contact Details

Name:

Email:

Phone:

Address:

Licence Port Allocation

Max Available Ports: 0 Warn Ports: 0 (Available 999)

Scroll down to the **Application Parameters** section and click **Save Affiliate**. In the **Numbers Available** section, add the **DNIS Numbers**. The DNIS numbers were set to *78701*, which is used to route calls to IVG, and *outbound* as shown below.

The screenshot displays two sections of a web application interface. The top section, titled "Application Parameters", features a "Key:" label followed by an empty text input field, and a "Value:" label followed by another empty text input field. Below these fields is a large, empty rectangular area. To the right of this area are three stacked buttons: "Set", "Replace", and "Delete". Below the large area is a "Preset Parameters:" label followed by a dropdown menu currently showing "Set Application Type To CCXML". To the right of the dropdown is a "Set" button. At the bottom of this section are three buttons: "Delete the Affiliate", "Revert", and "Save Affiliate". The bottom section, titled "Numbers Available", has a "DNIS Numbers:" label followed by two empty text input fields separated by a hyphen. Below these fields is a list box containing two entries: "78701 - 78701" and "outbound - outbound". To the right of the list box are three stacked buttons: "Add", "Replace", and "Delete".

Navigate to **Administration** → **Applications** to display the **Application Editor** shown below. This section will cover the **Inbound** application. In the **Service Provider** field, select the appropriate site name (e.g., *VHT*) and affiliate added in the previous step. During the initial configuration of the application, the **Application** field should be set to *<new application>* from the drop-down menu. Next, enter the desired **Name** and **Description**.

Scroll down to the URLs section and insert the appropriate **URL** (e.g., *http://localhost:8080/VIS/PlatformSupport_HVP/Begin?Tenant=VHT&MODE=HVP*Avaya).

VHT Powered by the Holly Voice Platform

Administration Reports Configuration Dashboard

user: administrator Logout

Application Editor

Select Application

Service Provider: VHT

Affiliate: VHT

Application: VHT_Inbound

Name: VHT_Inbound

Description: VHT_Inbound

Licence Exception URL:

Edit Affiliate...

URLs

URL:

Fetch Time Out: sec

URLs: http://localhost:8080/VIS/PlatformSupport_HVP/Begin?Tenant=VHT&MODE=HVP

Add

Replace

Delete

Move Up

Move Down

In the **Application Parameters** section, add the following **Keys**:

- **ap.connhdrstodlg** = *1*
- **type** = *application/voicexml+xml*

Click **Save Application**. In the **Numbers Available** section, add the **DNIS Number**. The DNIS number that was added was *78701* as shown below.

The screenshot displays two configuration windows. The top window, titled "Application Parameters", features a "Key:" label and an empty text input field, followed by a "Value:" label and another empty text input field. Below these is a list box containing two entries: "ap.connhdrstodlg = 1" and "type = application/voicexml+xml". To the right of the list box are three buttons: "Set", "Replace", and "Delete". Below the list box is a "Preset Parameters:" label and a dropdown menu currently showing "Set Application Type To CCXML", with a "Set" button to its right. At the bottom of this window are three buttons: "Delete the Application", "Revert", and "Save Application".

Below the first window is a section header "Application Numbers" in orange text. Underneath is a second window titled "Numbers Available". It has a "DNIS Numbers:" label followed by two empty text input fields separated by a hyphen. Below this is a list box containing the entry "78701 - 78701". To the right of the list box are three buttons: "Add", "Replace", and "Delete".

Repeat the above steps for the **Outbound** application. In the **Service Provider** field, select the appropriate site name (e.g., *VHT*) and affiliate added in the previous step. During the initial configuration of the application, the **Application** field should be set to *<new application>* from the drop-down menu. Next, enter the desired **Name** and **Description**.

Scroll down to the URLs section and insert the appropriate **URL** (e.g., *http://localhost:8080/VIS/PlatformSupport_HVP/Outbound?MODE=HVP Avaya*).

The screenshot displays the VHT Application Editor interface. At the top, the VHT logo is visible with the text "Powered by the Holly Voice Platform". The navigation bar includes links for Administration, Reports, Configuration, and Dashboard. The user is logged in as "administrator".

The main section is titled "Application Editor". It contains a "Select Application" panel with the following fields:

- Service Provider: VHT
- Affiliate: VHT
- Application: VHT_outbound
- Name: VHT_outbound
- Description: VHT_outbound
- Licence Exception URL: (empty)

Below the "Select Application" panel is an "Edit Affiliate..." button. The "URLs" section contains a list of URLs with the following fields:

- URL: (empty)
- Fetch Time Out: 5 sec
- URLs: http://localhost:8080/VIS/PlatformSupport_HVP/Outbound?MODE=HVP Avaya

Buttons for "Add", "Replace", "Delete", "Move Up", and "Move Down" are located to the right of the URL list.

In the **Application Parameters** section, add the following **Key**:

- **type** = *application/voicexml+xml*

Click **Save Application**. The DNIS number that was added was *outbound* as shown below

The image shows two screenshots of a configuration interface. The top screenshot is titled "Application Parameters" and contains a "Key:" field, a "Value:" field, and a list box showing "type = application/voicexml+xml". To the right of the list box are buttons for "Set", "Replace", and "Delete". Below the list box is a "Preset Parameters:" section with a dropdown menu showing "Set Application Type To CCXML" and a "Set" button. At the bottom of the window are buttons for "Delete the Application", "Revert", and "Save Application". The bottom screenshot is titled "Application Numbers" and shows a "DNIS Numbers:" field with a range input. Below it is a list box showing "outbound - outbound". To the right of the list box are buttons for "Add", "Replace", and "Delete".

Lastly, open the `/etc/VirtualHold/toolkit.properties` file and set the **com.virtualhold.toolkit.baseurl** parameter to <http://10.64.102.108/VHTPlatformWS-V5/>, which specifies the IP address of the Callback server as shown below. This allows IVG to communicate with the Callback system.

```
# Sample configuration file for SIP Avaya - Interactive Voice Gateway integrations

# URL for the Platform Toolkit web services
# Change the [PTK_server_address] and [PTK_port] to the address and port of the server
# where the Platform Toolkit software resides
# For example, http://10.10.0.158:7000/VHTPlatformWS-v5/
# Ensure the path and VHTPlatformWS version is correct by opening it in a web browser
com.virtualhold.toolkit.baseurl=http://10.64.102.108/VHTPlatformWS-v5/
```

9. Configure VHT Callback

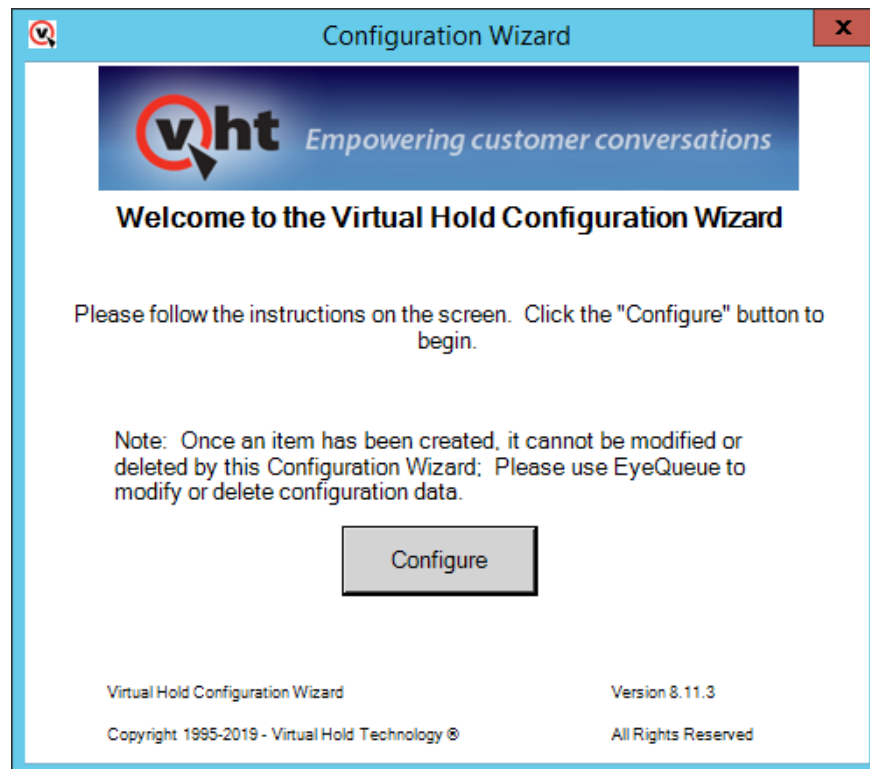
This section provides the procedures for configuring Callback. The procedures include the following areas:

- Launch VHT Configuration Wizard
- Administer Switch Connection
- Administer IVR Servers
- Administer Queues
- Administer Callback and Holding Queues
- Administer Incoming Extensions
- Administer Phone Number Configurations
- Administer Segment Variables
- Modify `site.config` File
- Configure TSAPI Real-Time Adapter

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

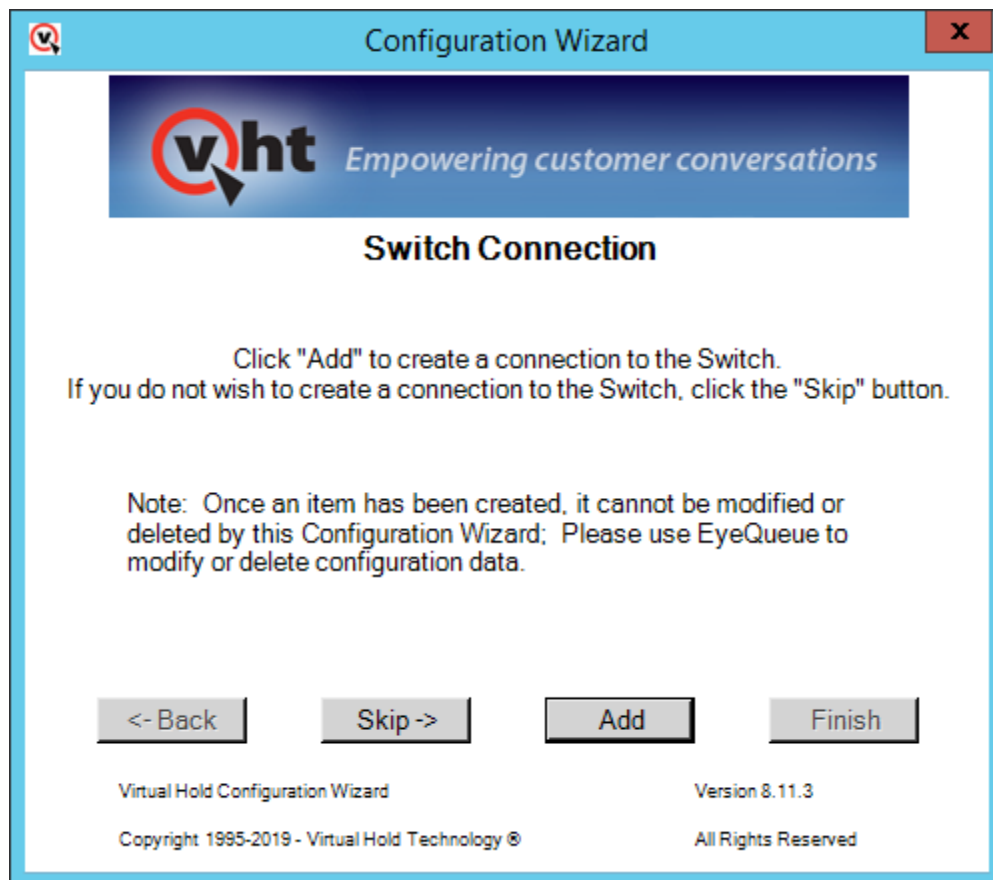
9.1. Launch Configuration Wizard

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.

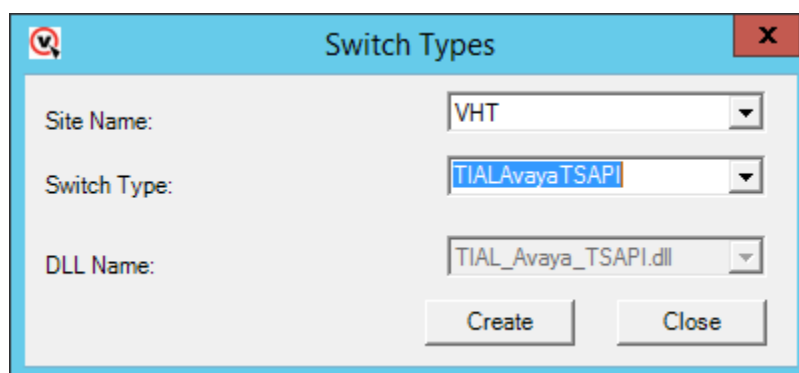


9.2. Administer Switch Connection

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 7.5**.
- **Login ID:** The Callback user credentials from **Section 7.6**.
- **Password:** The Callback user credentials from **Section 7.6**.
- **Send Extra Buffers:** The desired extra buffers.
- **Receive Queue Size:** The desired queue size.
- **Use Private Data:** Set to *TRUE*.
- **Private Data Version:** Set to '8'.

The screenshot shows the 'AES Avaya CTI' configuration window. The fields are as follows:

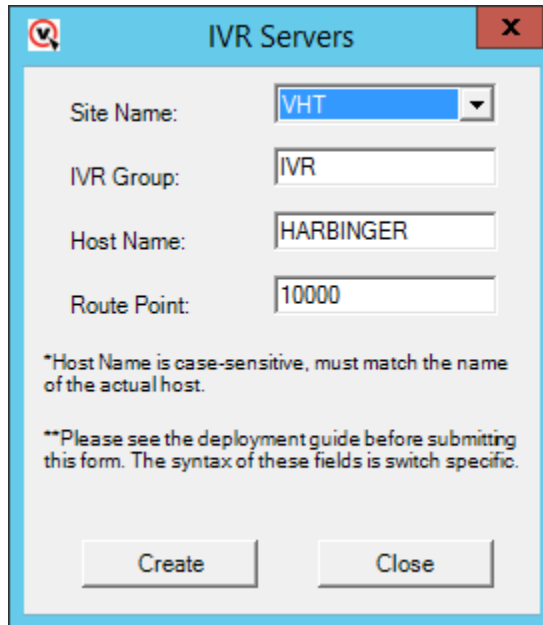
Field	Value
Site Name	VHT
VH Server ID	VHT_Test_01
Server ID	AVAYA#DEVCON#CSTA#DEV
Invoke ID Type	LIB_GEN_ID
Login ID	vht
Password	Interop123!
Application Name	virtualhold
API Version	TS2
Send Queue Size	0
Send Extra Buffers	0
Receive Queue Size	0
Receive Extra Buffers	0
Use Private Data	TRUE
Private Data Version	8

A 'Create' button is located at the bottom right of the form.

9.3. Administer IVR Servers

Continue with the wizard until the **IVR Servers** screen is displayed (not shown). Click **Add** to create IVR server.

The screen below is displayed next. Set **Host Name** to the host name of the Callback server. Even though IVG is the IVR server, the Callback server initiates the callback. The **Route Point** is just a place holder at this point.

A screenshot of a software dialog box titled "IVR Servers". The dialog has a light blue border and a title bar with a red close button. Inside, there are four labeled text fields: "Site Name" with a dropdown menu showing "VHT", "IVR Group" with the text "IVR", "Host Name" with the text "HARBINGER", and "Route Point" with the text "10000". Below these fields, there is a note: "*Host Name is case-sensitive, must match the name of the actual host." followed by another note: "**Please see the deployment guide before submitting this form. The syntax of these fields is switch specific." At the bottom of the dialog are two buttons: "Create" and "Close".

Site Name:	VHT
IVR Group:	IVR
Host Name:	HARBINGER
Route Point:	10000

*Host Name is case-sensitive, must match the name of the actual host.

**Please see the deployment guide before submitting this form. The syntax of these fields is switch specific.

Create Close

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 [4] for desired configuration of these parameters. The screenshot below shows the values used in the compliance testing.

The screenshot shows the 'Queues Setup' dialog box with the following configuration:

- Site Name:** VHT (dropdown)
- Queue ID:** VHT_Test (text field)
- Buttons:** Use Production Defaults, Use Test Defaults
- QueueSettings:**
 - Op Mode:** Normal (dropdown)
 - Turn On Threshold (sec):** 0 (spin box)
 - Call Handle Time (secs):** 45 (spin box)
 - No Ans Period (sec):** 60 (spin box)
 - Name:** VHT_Test (text field)
 - Script Number:** 1 (spin box)
 - Busy Attempts:** 3 (spin box)
 - Try Again Attempts:** 3 (spin box)
 - Mode:** Predictive (dropdown)
 - Agents Staffed Override:** TRUE (dropdown)
 - Busy Period (secs):** 60 (spin box)
 - Try Again Period (secs):** 60 (spin box)
 - Group:** (empty text field)
 - Callback Threshold (secs):** 45 (spin box)
 - No Ans Attempts:** 3 (spin box)
 - Max Attempts:** 5 (spin box)
 - Default Number of Agents:** 1 (spin box)
- Business Hours:**
 - Day Of Week:** Sun ☒ Mon ☒ Tue ☒ Wed ☒ Thu ☒ Fri ☒ Sat ☒
 - Time Begin:** 00:00 00:00 00:00 00:00 00:00 00:00 00:00
 - Time End:** 23:59 23:59 23:59 23:59 23:59 23:59 23:59
- Callbacks Offered:**
 - Day Of Week:** Sun ☒ Mon ☒ Tue ☒ Wed ☒ Thu ☒ Fri ☒ Sat ☒
 - Time Begin:** 00:00 00:00 00:00 00:00 00:00 00:00 00:00
 - Time End:** 23:59 23:59 23:59 23:59 23:59 23:59 23:59
- Callbacks Allowed:**
 - Day Of Week:** Sun ☒ Mon ☒ Tue ☒ Wed ☒ Thu ☒ Fri ☒ Sat ☒
 - Sched callbacks allowed/15 min:** 15 15 15 15 15 15 15 (spin boxes)

Queue 'VHT_Test' created.

Buttons: Create, Close

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 callback and holding 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 (e.g., *sip:77203@10.64.102.117*).

In the **Holding Queues** sub-section, enter the Hold VDN extension from **Section 5.4.2** for **Holding Queue ID** and **Route Device**. For **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 (e.g., *sip:77202@10.64.102.117*).

Retain the default values for the remaining fields.

Callback and Holding Queues

Site Name: VHT

VH Server Switch Name: VHT_Test_01

Callback Queues

☒ Use VH Server Switch Name prefix

Callback Queue ID*: 77203

Transfer Device: sip:77203@10.64.10

Callback Queue VHT_Test_01:77203 created. **Create**

Holding Queues

☒ Use VH Server Switch Name prefix

Holding Queue ID*: 77202

Route Device: 77202

Transfer Device: sip:77202@10.64.10

Holding Queue VHT_Test_01:77202 created. **Create**

*Please see the deployment guide before submitting this form. The syntax of these fields is switch specific.

*Verify VH Server Switch Name **Close**

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.

Incoming Extensions

Site Name: VHT

Queue ID: VHT_Test

VH Server Switch Name: VHT_Test_01

Incoming Extensions

Extension*: 77201

Label: Extension

Country ID: 1

Treatment Type: 11

ScriptNumber: *Please see the deployment guide before entering a script number here.

IVR Group: IVR

Holding Queue ID: VHT_Test_01:77202

Callback Queue ID: VHT_Test_01:77203

UnderThreshold Queue ID: VHT_Test_01:77202

IB IVR Extension Group: NONE

OB IVR Extension Group: NONE

Incoming Extension: 77201 created. **Create**

* Verify VH Server Switch Name **Close**

Repeat the same procedures to create an incoming extension for IVG. For **Extension**, enter the extension assigned to IVG, in this case *78701*. For **Treatment Type**, select *20*. Retain the default values in the remaining fields, including blank for **VH Server Switch Name**.

Incoming Extensions

Site Name: VHT

Queue ID: VHT_Test

VH Server Switch Name:

Incoming Extensions

Extension*: 78701

Label: Extension

Country ID: 1

Treatment Type: 20

ScriptNumber:

*Please see the deployment guide before entering a script number here.

IVR Group: IVR

Holding Queue ID: VHT_Test_01:77202

Callback Queue ID: VHT_Test_01:77203

UnderThreshold Queue ID: VHT_Test_01:77202

IB IVR Extension Group: NONE

OB IVR Extension Group: NONE

Incoming Extension: 78701 created. **Create**

* Verify VH Server Switch Name **Close**

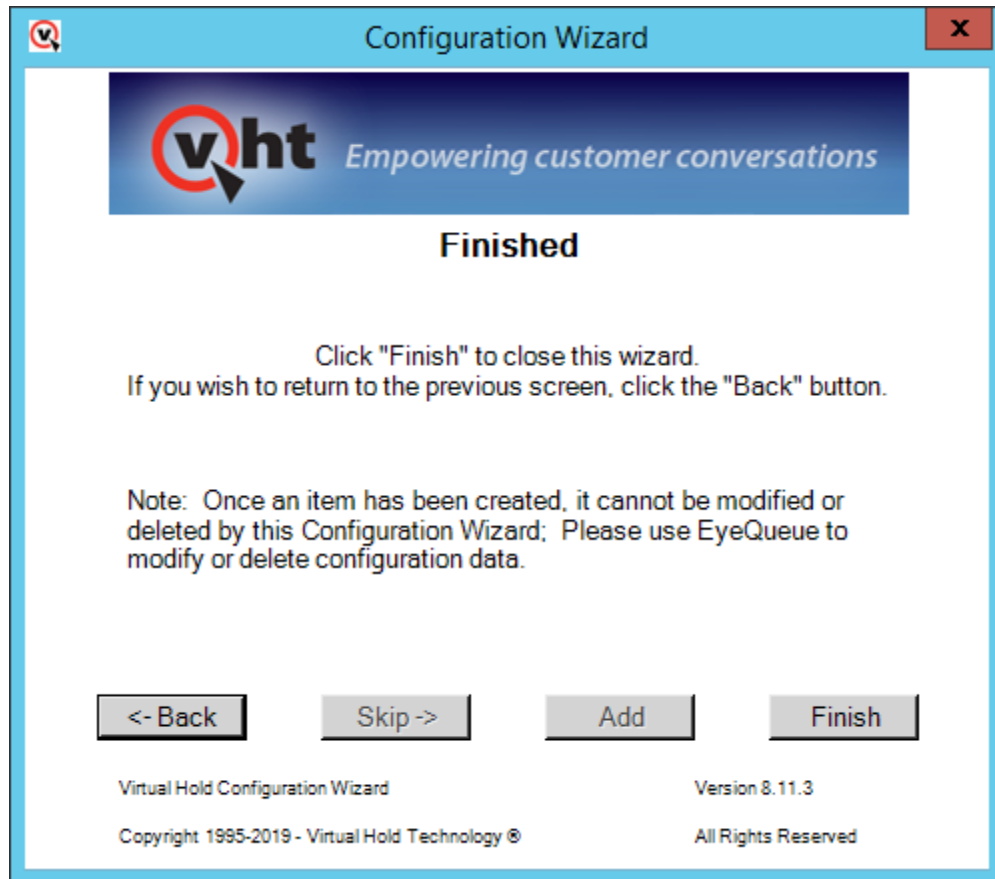
9.7. Administer Phone Number Configurations

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 and select the applicable country as shown below. For the compliance test, the Min Length field was set to '5' to allow callbacks to 5-digit extensions corresponding to local IP stations and the Max Length field was set to '12' to allow callbacks to 10-digit PSTN number prepended with a '9' (ARS access code) + '1' prefix code. Retain the default values in the remaining fields.

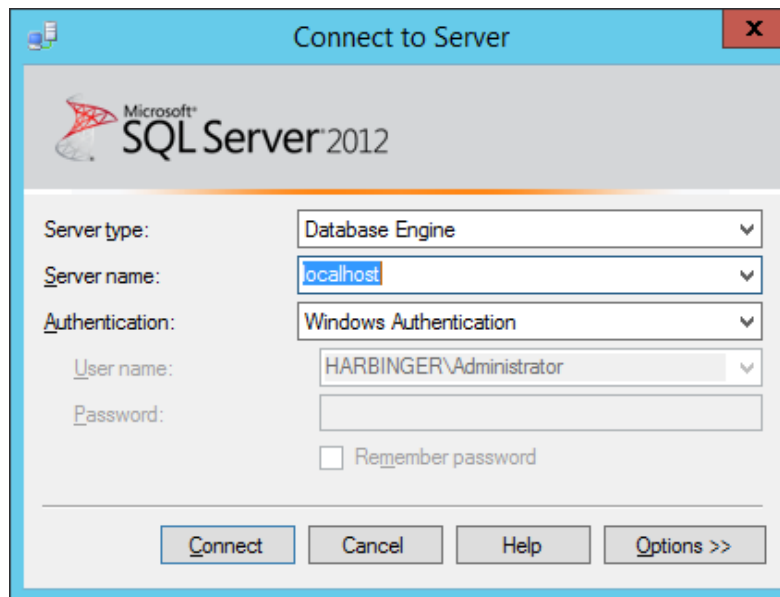
The screenshot shows a dialog box titled "PhoneNumberValidation" with a close button (X) in the top right corner. The dialog is divided into two main sections. The left section, titled "Update Country Id Dial Prefix and Suffix", contains a "Site Name" dropdown menu set to "VHT", a "Country Search" dropdown menu set to "1 - North America" with a list box below it showing "1 - North America" selected, and two empty text boxes for "Dial Prefix" and "Dial Suffix". The right section, titled "Update Phone Number Validation Min/Max Length", contains a "Site Name" dropdown menu set to "VHT", a "Country Id" dropdown menu set to "1 - North America", a "Min Length" text box set to "5", and a "Max Length" text box set to "12". Below these fields, the text "Update Successful" is displayed, followed by "Min Length: 5" and "Max Length: 12". Both sections have an "Update" button at the bottom. A "Close" button is located at the bottom right of the dialog.

When done, click **Finish** to exit the configuration wizard.



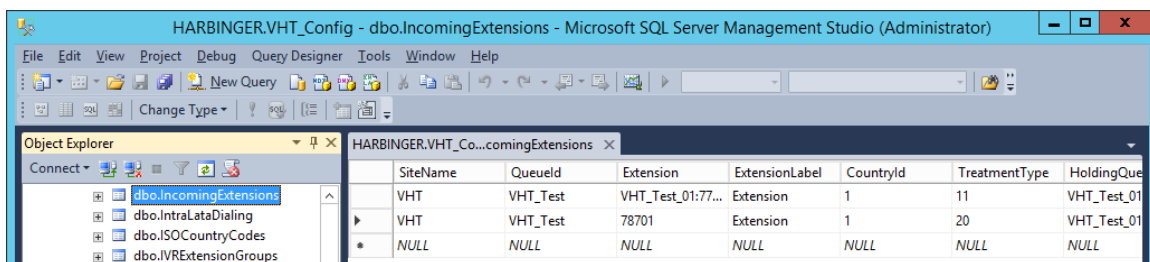
9.8. Administer Segment Variables

From the Callback server, navigate to **Start → Apps → Microsoft SQL Server 2012 → SQL Server Management Studio** to launch and connect to the SQL server.

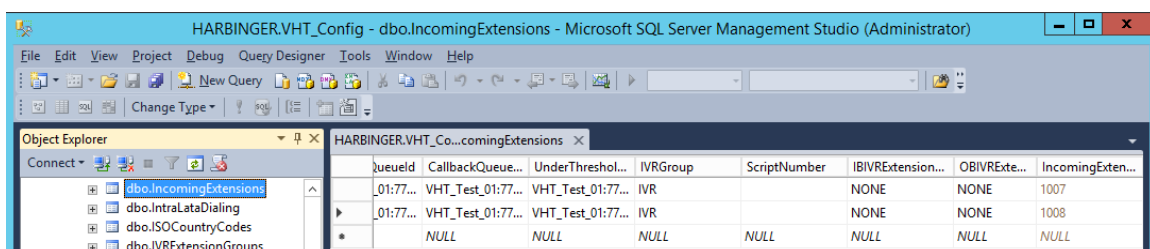


Navigate to **Databases → VHT_Config → Tables → dbo.IncomingExtensions** in the left pane, right-click the entry and select **Edit Top 200 Rows**.

Locate the entry associated with Callback with “11” as **Treatment Type**.



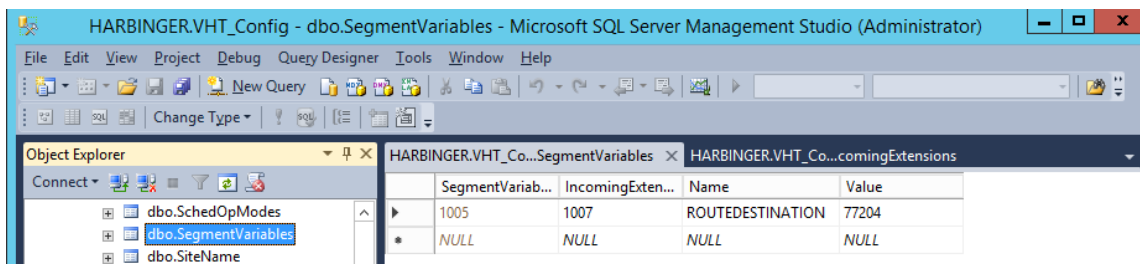
Scroll to the right to make a note of the associated **IncomingExtensionsId** value, in this case ‘1007’, as shown below.



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:** Set to *ROUTEDESTINATION*.
- **Value:** Set to the route VDN extension 77204.

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



9.9. Modify site.config File

Open the `site.config` file located in the `C:\Program Files (x86)\Virtual Hold Technology\Peripheral Monitor\` directory of the Callback server and modify the entries in bold to include the Callback server IP address (10.64.102.108), the IVG IP address (10.64.102.107), or the Session Manager IP address (10.64.102.117). The **ani** should include `<ani>@<Session Manager IP Address>`, where `<ani>` is the Automatic Number Identifier of the Callback server (e.g., 8005555555@10.64.102.117). The other entries may be left with their default values.


```

{vht_outbound_contact_client,
[
  {voice_platform, ivg_plugin},
  {ivg_environment, avaya},
  {queue_manager_connection_ping_in_seconds, 15},
  {ivr_group_name, "IVR"},
  {ivr_server_name, "harbinger"},
  {ivr_port_send_interval_ms, 2000},
  {disposition_url, "http://10.64.102.108:4153/vht/occ"},
  {disposition_timeout, 55000},
  {exclude_connections_on_failure, true},
  {time_to_exclude_on_failure_ms, 150000},
  {default_connection_attributes,
  [
    {outdial_http_options,
    [
      {timeout, 5000},
      {connect_timeout, 5000}
    ]
    },
    {request_header,
    [
      {"Accept", "application/x-www-form-urlencoded"},
      {"Content-Type", "application/x-www-form-urlencoded"}
    ]
    },
    {enable_amd, true},
    {ring_no_answer_timeout, 50000},
    {ccxml_fetch_timeout, 5000},
    {tenant, "VHT"}
  ]
  },
  {load_balanced_connections,
  [
    [
      {outdial_url, "http://10.64.102.107:8040/createsession"},
      {sip_endpoint, "10.64.102.117"},
      {failure_destination, ""},
      {dnis, "outbound"},
      {vht_ccis_uri, "http://10.64.102.107:8080/CCIS/vht_hvp.ccxml"},
      {ani, "8005555555@10.64.102.117"},
      {node_id, 7},
      {agent_priority_dnis, "agntpriority"},
      {outreach_dnis, "outreach"}
    ]
  ]
  }
]
}

```

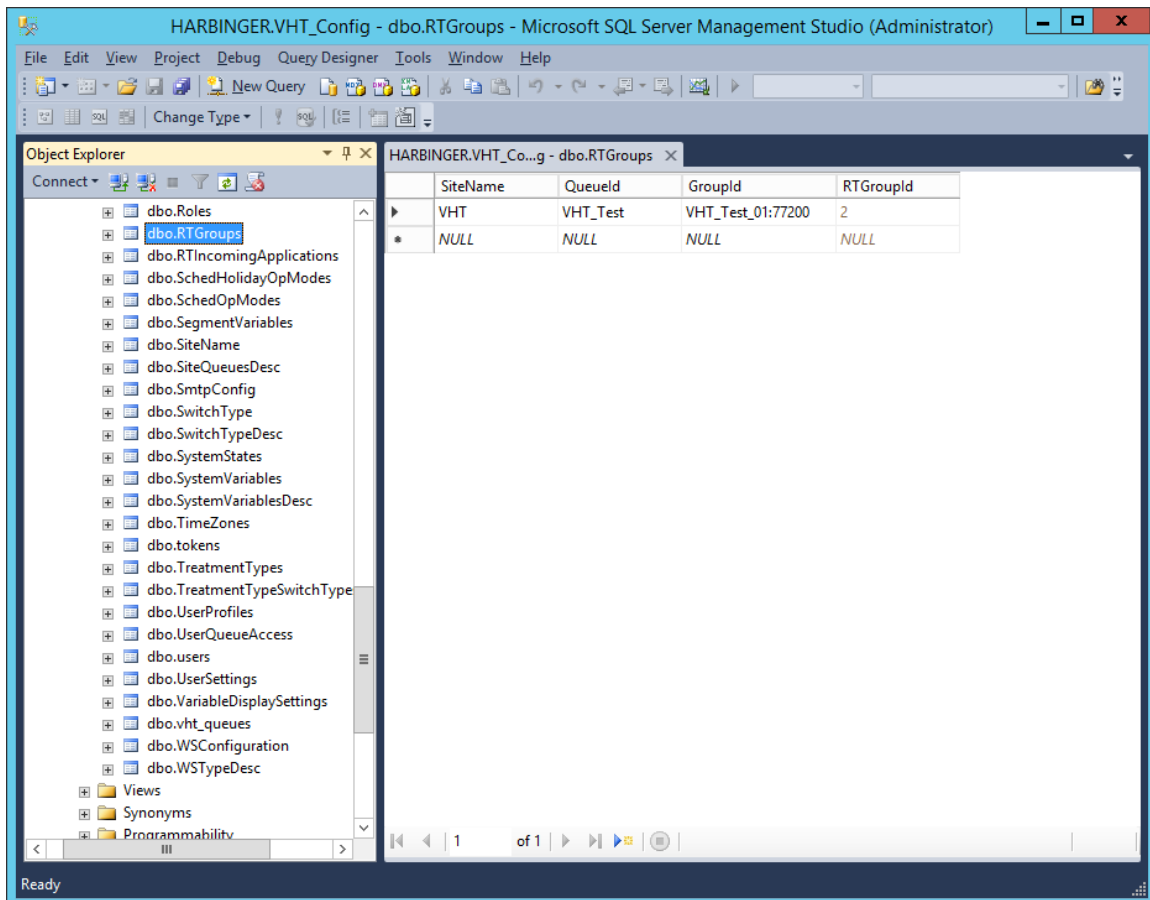
9.10. Configure TSAPI Real-Time Adapter

The Callback TSAPI Real-Time Adapter captures queue statistics, such as agent status of a monitored skill/split and can be displayed as shown in **Section 10.4**.

Open the `VHT_TsapiRealTimeAdapter_Console.exe.config` file located in the `C:\Program Files (x86)\Virtual Hold Technology\RealTimeAdapter\` directory of the Callback server and modify the entries in bold to include the Callback server IP address (10.64.102.108) for the **bolded** entries as shown below. In addition, the **SiteName** should be set to the appropriate value.

```
<?xml version="1.0" encoding="utf-8"?>
<configuration>
  <configSections>
    <sectionGroup name="VHTConfiguration">
      <section name="vhtLogging"
type="VHT.Common.Library.Configuration.Logging.VHTLoggingSection, VHT.Common.Library"
allowLocation="true" allowDefinition="Everywhere"/>
      <section name="vhtCommunication"
type="VHT.Common.Library.Configuration.Communication.VHTCommunicationSection,
VHT.Common.Library" allowLocation="true" allowDefinition="Everywhere"/></section>
    </sectionGroup>
  </configSections>
  <VHTConfiguration>
    <vhtLogging>
      <application level="10" name="TsapiRealTimeAdapter"
logFilePath="C:\Program Files (x86)\Virtual Hold Technology\VHLogs"/>
    </vhtLogging>
    <vhtCommunication>
      <QMCL reconnectIntervalSeconds="3">
        <Connections>
          <Connection connectionType="Primary">
            <Server ipAddress="10.64.102.108" port="6999"/>
            <Client ipAddress="10.64.102.108" port="0"/>
          </Connection>
        </Connections>
      </QMCL>
    </vhtCommunication>
  </VHTConfiguration>
  <appSettings>
    <add key="VhqmwmsUrl" value="http://10.64.102.108/VHQMWS/VHQMWS.asmx"/>
    <add key="SiteName" value="VHT"/>
    <add key="FrequencyMS" value="3000"/>
    <add key="UseDefaultsOnConnectionLost" value="false"/>
  </appSettings>
  <startup>
    <supportedRuntime version="v4.0" sku=".NETFramework,Version=v4.6.1"/>
  </startup>
</configuration>
```

Next, launch **SQL Server Management Studio** to launch and connect to the SQL server. Navigate to **Databases → VHT_Config → Tables → dbo.RTGroups** in the left pane, right-click the entry and select **Edit Top 200 Rows**. Ensure that an entry exists with the appropriate **SiteName**, **QueueId**, and **GroupID**, which includes the VH server ID and hunt group extension (e.g., *VHT_Test_01:77200*) as shown below.



Lastly, navigate to `HKEY_LOCAL_MACHINE\SOFTWARE\Wow6432Node\Virtual Hold` in the Windows Registry and add **ExternalTrackingId** parameter as a string value and set it to *UCID*.

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 Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd
1	7	no	devcon-aes	established	134	134

Verify the status of the SIP trunk groups by using the **status trunk** command for the trunk group number administered in **Section 5.9**. Verify that all trunks are in the *service/idle* state as shown below.

```
status trunk 10
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0010/001	T00001	in-service/idle	no
0010/002	T00002	in-service/idle	no
0010/003	T00003	in-service/idle	no
0010/004	T00004	in-service/idle	no
0010/005	T00005	in-service/idle	no
0010/006	T00006	in-service/idle	no
0010/007	T00007	in-service/idle	no
0010/008	T00008	in-service/idle	no
0010/009	T00009	in-service/idle	no
0010/010	T00010	in-service/idle	no

Verify the status of the SIP signaling groups by using the **status signaling-group** command for the signaling group number administered in **Section 5.8**. Verify that the **Group State** is *in-service* as shown below.

```
status signaling-group 10
```

```
STATUS SIGNALING GROUP
```

```
Group ID: 10
```

```
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 7.3**.

AVAYA Application Enablement Services Management Console

Welcome: User cust
Last login: Thu Aug 8 11:52:26 2019 from 192.168.100.250
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.1.0.0.9-1
Server Date and Time: Thu Aug 08 12:16:04 EDT 2019
HA Status: Not Configured

Status | Status and Control | TSAPI Service Summary

Home | Help | Logout

- ▶ AE Services
- ▶ Communication Manager Interface
- ▶ High Availability
- ▶ Licensing
- ▶ Maintenance
- ▶ Networking
- ▶ Security
- ▼ **Status**
 - Alarm Viewer
 - ▶ Logs
 - ▶ Log Manager
 - ▼ **Status and Control**
 - CVLAN Service Summary
 - DLG Services Summary
 - DMCC Service Summary
 - Switch Conn Summary
 - **TSAPI Service Summary**

TSAPI Link Details

☐ Enable page refresh every seconds

	Link	Switch Name	Switch CTI Link ID	Status	Since	State	Switch Version	Associations	Msgs to Switch	Msgs from Switch	Msgs Period
<input checked="" type="radio"/>	1	devcon	1	Talking	Mon Jul 29 12:15:53 2019	Online	18	5	31	46	30

For service-wide information, choose one of the following:

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 (not shown). Click the IVG entity name from **Section 6.2.2**.

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

AVAYA Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾ Search 🔍 admin

Home Session Manager

Session Manager ▾

- Dashboard
- Session Manager Ad...
- Global Settings
- Communication Prof...
- Network Configur... ▾
- Device and Locati... ▾
- Application Confi... ▾
- System Status ▾
- SIP Entity Monit...

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

All Entity Links to SIP Entity: VHT-IVG

Summary View

1 Item Filter: Enable

Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
devcon-sm	IPv4	10.64.102.107	5060	UDP	FALSE	UP	200 OK	UP

Select : None

10.4. Verify VHT Callback and IVG

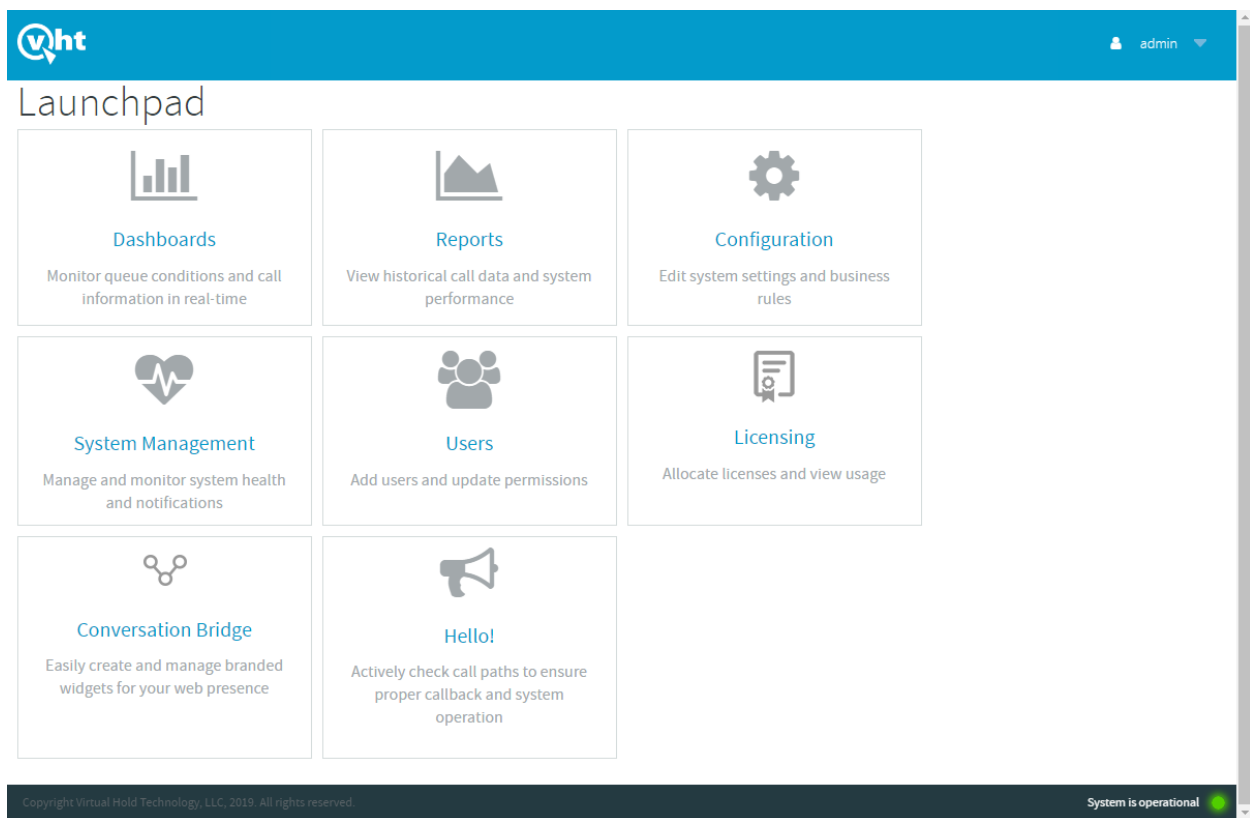
Access the Callback web-based EyeQueue application by using the URL “http://<ip-address>/EyeQueue” in an Internet browser window, where <ip-address> is the IP address of the Callback server. Log in using the appropriate credentials.

User name

Password

vht

The **Launchpad** screen below is displayed. Select **System Management**.



In **System Status**, verify that the components are in-service and that the system is operational as shown below.

The screenshot displays the VHT System Management web interface. The top navigation bar is blue with the VHT logo on the left, a heart icon and 'System Management' in the center, and a user profile icon labeled 'admin' on the right. Below this is a light gray bar with a 'Status' dropdown menu. The main content area has a large 'System Status' heading on the left and a 'Manage Components' button on the right. The central part of the interface is a light green box containing two columns of component status cards. The left column, titled 'Core (HARBINGER)', shows the mode as 'STANDALONE' with a gear icon, followed by a list of components: CTI Connector, Queue Manager, License Server, Opmode Server, Provider, Real-Time Adapter, and Report Writer. The right column, titled 'Management (HARBINGER)', lists: Launchpad, Platform Toolkit, Web Monitor, Message Bus, Outbound Contact Client, and QWatch Client. Each component name is preceded by a green circular icon with a white upward-pointing arrow. At the bottom of the interface is a dark gray footer bar containing the copyright text 'Copyright Virtual Hold Technology, LLC, 2019. All rights reserved.' on the left and 'System is operational' with a green status indicator on the right.

System Status

Manage Components

Core (HARBINGER)

Mode: STANDALONE

- CTI Connector
- Queue Manager
- License Server
- Opmode Server
- Provider
- Real-Time Adapter
- Report Writer

Management (HARBINGER)

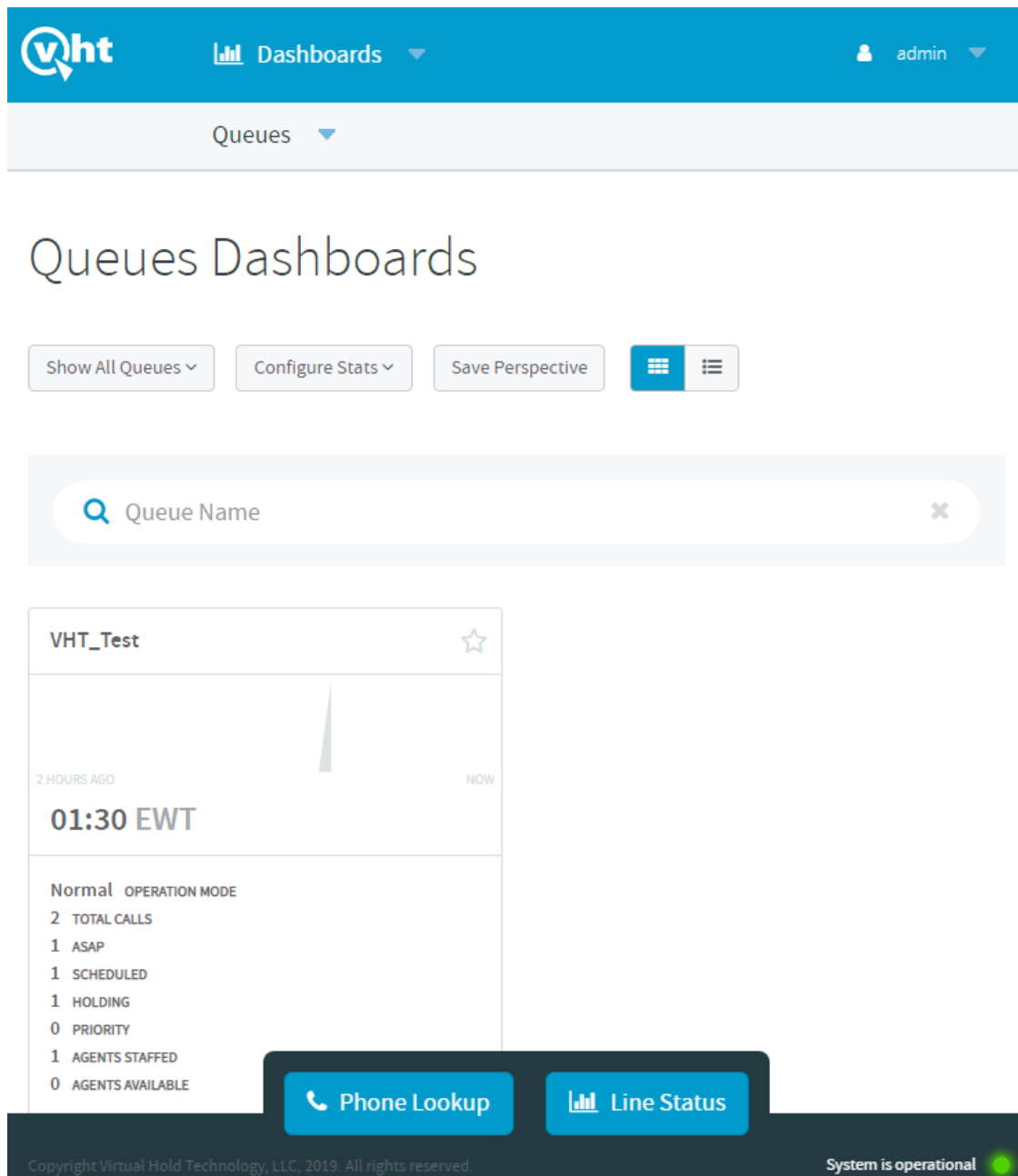
- Launchpad
- Platform Toolkit
- Web Monitor
- Message Bus
- Outbound Contact Client
- QWatch Client

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System is operational

From the **Launchpad** or from the drop-down menu at the top of the webpage, select **Dashboards**.

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



11. Conclusion

These Application Notes describe the steps required to integrate VHT Callback using Native TSAPI with Avaya Aura® Communication Manager, Avaya Aura® Application Enablement Services, and Avaya Aura® Session Manager. VHT Callback successfully handled callback requests from callers, provided estimated wait time, and reported real-time queue statistics.

12. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 2, July 2019, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019, available at <http://support.avaya.com>.
3. *Administering and Maintaining Avaya Aura® Application Enablement Services*, Release 8.1.x, Issue 2, August 2019, available at <http://support.avaya.com>.
4. *VHT Callback Configuration Guide Version 8.11 or Later*, available at <https://insight.vhtcx.com>.
5. *VHT Callback Installation Guide Version 8.10.1 or Later*, available at <https://insight.vhtcx.com>.
6. *VHT Interactive Voice Gateway (IVG) Avaya Version 3.9*, available at <https://insight.vhtcx.com>.

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