



Application Notes for VHT Callback using Genesys T-Server 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 Genesys T-Server 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 Genesys T-Server 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.

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 an Avaya Aura® 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 UUI data, initiation and transfer of outbound callback calls with priority and saved UUI data, and unsuccessful callback attempts.
- Queue statistics using Genesys real-time adapter in Callback.

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 Genesys 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.virtualhold.com/contact/contact-center-technical-support/>
- **Email:** support@virtualhold.com

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. The Callback configuration consisted of the Callback server and 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 787xx were routed to the IVG component of Callback.

Device Type	Extension
Skill Group Number	77
Skill Group Extension	77200
Agent Stations	77301, 77302

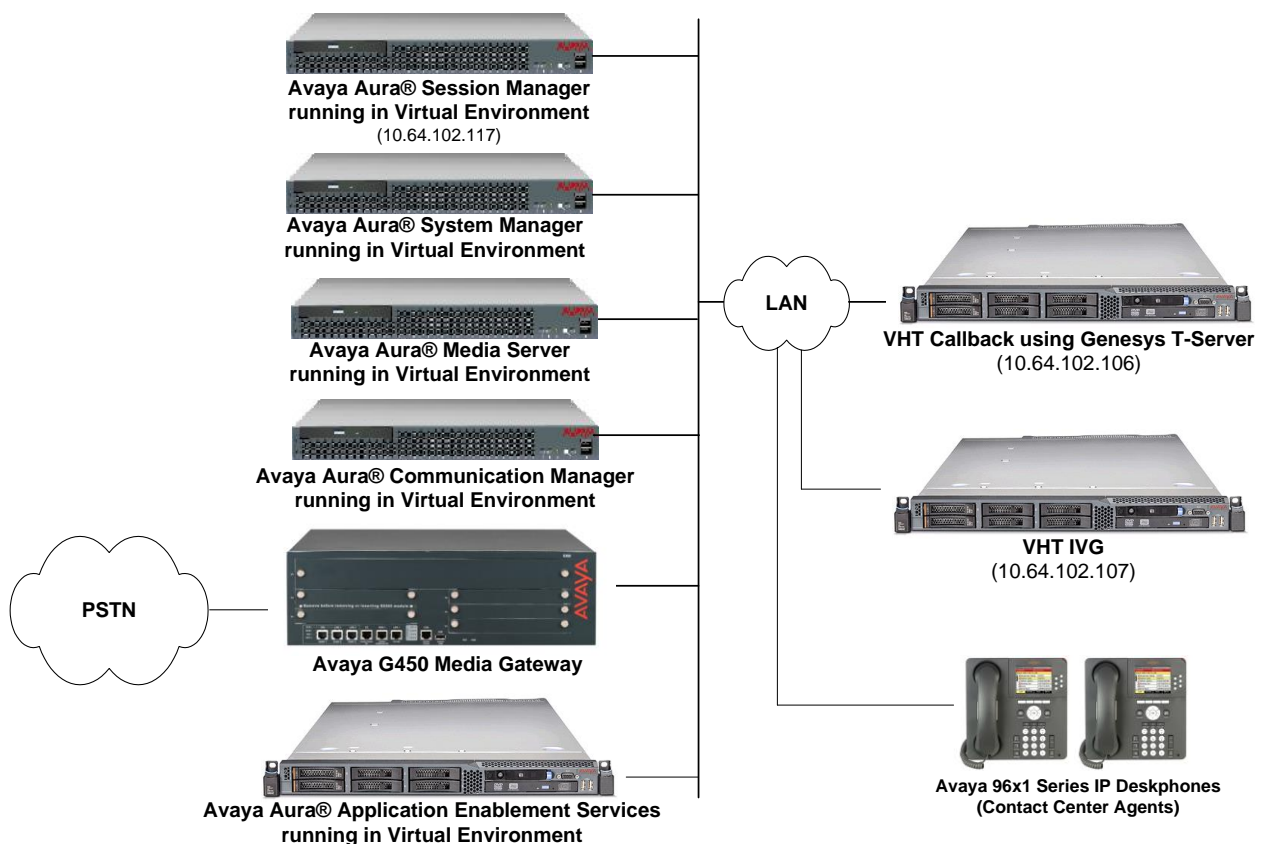


Figure 1: Compliance Testing Configuration

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 running in a Virtual Environment with an Avaya G450 Media Gateway	7.1 (R017x.01.0.532.0)
Avaya Aura® Media Server running in a Virtual Environment	7.7.0.375
Avaya Aura® Application Enablement Services running in a Virtual Environment	7.1 (7.1.0.0.0.17-0)
Avaya Aura® Session Manager running in a Virtual Environment	7.1 (7.1.0.0.710028)
Avaya Aura® System Manager running in a Virtual Environment	7.1.0.0 (Build No. 7.1.0.0.1125193 Software Update Revision No: 7.1.0.0.116654)
Avaya 96x1 IP Deskphones	6.6401U (H.323) 7.0.1.4.6 (SIP)
VHT Callback using Native TSAPI on Microsoft Windows Server 2012 R2 Standard with <ul style="list-style-type: none">Genesys T-Server for Avaya TSAPIAvaya AES TSAPI Client	8.8.3.612 8.1.010.09 7.1.0.67
VHT Interactive Voice Gateway (IVG) on CentOS <ul style="list-style-type: none">VXML Interactive Server (VIS)	3.2.0.114 6.2.0

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 Network Region
- Administer IP Codec Set
- 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:		18000	4	
Maximum Administered Remote Office Trunks:		12000	0	
Maximum Concurrently Registered Remote Office Stations:		18000	0	
Maximum Concurrently Registered IP eCons:		128	0	
Max Concur Registered Unauthenticated H.323 Stations:		100	0	
Maximum Video Capable Stations:		36000	2	
Maximum Video Capable IP Softphones:		18000	0	
Maximum Administered SIP Trunks:		12000	10	
Maximum Administered Ad-hoc Video Conferencing Ports:		12000	0	
Maximum Number of DS1 Boards with Echo Cancellation:		522	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: 7.0

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

(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	
	COR: 1
Name: AES TSAPI Link	

5.3. Administer System Parameters Features

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

```
change system-parameters features                               Page 5 of 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 UII 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 Codec	Silence Suppression	Frames Per Pkt	Packet 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			
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 Link Bounce Recovery? y		RSVP Enabled? n	
Idle Traffic Interval (sec): 20			
Keep-Alive Interval (sec): 5			
Keep-Alive Count: 5			

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	
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	
Send UCID? y		Modify Tandem Calling Number: no	
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	Node Num
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 routes calls to IVG and SIP stations.

change aar analysis 7				Page 1 of 2	
AAR DIGIT ANALYSIS TABLE				Percent Full: 2	
Location: all					
Dialed String	Total Min	Max	Route Pattern	Call Type	Node Num ANI Req'd
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	Hop	Toll	No.	Inserted			DCS/	IXC	
			Mrk	Lmt	List	Del	Digits			QSIG		
							Dgts			Intw		
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 48.0, 49.0 and 50.0.

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 7.1 web interface. The left sidebar shows a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and contains a form with the following fields and values:

- Name:** devcon-cm
- FQDN or IP Address:** 10.64.102.115
- Type:** CM
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** Thornton
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- Securable:** (unchecked)
- 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

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

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. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. 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

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"/>	* 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

0 Items
Filter: Enable

<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 7.1 web interface. The left sidebar shows a navigation menu with 'Routing' selected, and 'SIP Entities' highlighted under the 'Routing' section. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. The 'General' tab is active, showing the following fields: 'Name' (VHT-IVG), 'FQDN or IP Address' (10.64.102.107), 'Type' (SIP Trunk), 'Notes' (empty), 'Adaptation' (empty), 'Location' (Thornton), 'Time Zone' (America/New_York), 'SIP Timer B/F (in seconds)' (4), 'Minimum TLS Version' (Use Global Setting), 'Credential name' (empty), 'Securable' (unchecked), 'Call Detail Recording' (egress), 'Loop Detection Mode' (On), 'Loop Count Threshold' (5), 'Loop Detection Interval (in msec)' (200), and 'SIP Link Monitoring' (Use Session Manager Configuration). The 'Loop Detection' and 'Monitoring' sections are 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. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. 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 7.1 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.1', and a user session summary: 'Last Logged on at July 31, 2017 2:19 PM' with a 'Go...' button and a 'Log off admin' link. A notification banner states '1 New important message(s). Click to view details'. The left sidebar contains a tree view with 'Routing' expanded, showing sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (selected), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Home / Elements / Routing / Routing Policies' and displays the 'Routing Policy Details' form. The form has 'Commit' and 'Cancel' buttons. The 'General' section includes fields for 'Name' (containing 'devcon-cm Policy'), 'Disabled' (checkbox), 'Retries' (set to 0), and 'Notes'. The 'SIP Entity as Destination' section features a 'Select' button and a table with the following data:

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

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 7.1 interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 7.1", and a "Last Logged on at July 31, 2017 2:19 PM" timestamp. A "GO..." search bar and a "Log off admin" link are also present. The main navigation menu on the left lists various routing-related options: Home, Routing, Domains, Locations, 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 contains a "Commit" button and a "Cancel" button. The "General" section includes 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 the following data:

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

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.

The screenshot shows the Avaya Aura System Manager 7.1 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Dial Pattern Details' and includes a 'General' tab. The form fields are as follows:

- Pattern:** 7
- Min:** 5
- Max:** 5
- Emergency Call:** ☐
- Emergency Priority:** 1
- Emergency Type:**
- SIP Domain:** -ALL-
- Notes:** CM Stations

Below the form is a section titled 'Originating Locations and Routing Policies' with an 'Add' button and a 'Remove' button. It contains a table with one item:

	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	

At the bottom of the table, there is a 'Select' dropdown menu with options: All, None.

The following screen shows the dial pattern definition for routing calls to PSTN via Communication Manager.

AVAYA
Aura® System Manager 7.1

Last Logged on at July 31, 2017 2:19 PM
GO... Log off admin

Home Routing * 1 New important message(s). Click to view details

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel Help ?

General

* Pattern: 91

* Min: 12

* Max: 12

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes: PSTN

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

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 7.1

Last Logged on at July 31, 2017 2:19 PM
GO... Log off admin

Home Routing * 1 New important message(s). Click to view details

Home / Elements / Routing / Dial Patterns

Dial Pattern Details Commit Cancel Help ?

General

* Pattern: 787

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

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

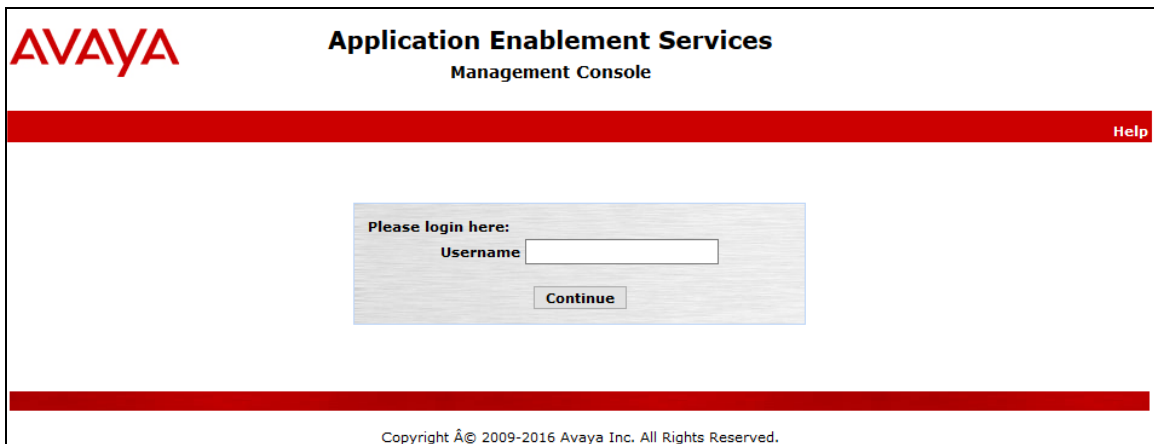
7. Configure Avaya Aura® Application Enablement Services

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

- Launch OAM Interface
- Verify License
- Administer TSAPI Link
- Administer TCP Settings
- Restart Service
- Obtain Tlink Name
- Administer 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.



The screenshot shows the Avaya Application Enablement Services Management Console login interface. At the top left is the Avaya logo. To its right, the text "Application Enablement Services" and "Management Console" is displayed. A red horizontal bar spans the width of the page, with a "Help" link on the right. In the center, there is a login box with the text "Please login here:" followed by a "Username" label and a text input field. Below the input field is a "Continue" button. At the bottom of the page, a red horizontal bar is present, and below it, the copyright notice "Copyright © 2009-2016 Avaya Inc. All Rights Reserved." is displayed.

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

The screenshot displays the Avaya Application Enablement Services Management Console. The top header includes the Avaya logo, the title "Application Enablement Services Management Console", and a welcome message for user "cust" with login details. A red navigation bar contains "Home", "Help", and "Logout" links. On the left, a sidebar lists various services: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security, Status, User Management, Utilities, and Help. The main content area, titled "Welcome to OAM", explains that the OAM web provides tools for managing the AE Server across several administrative domains. It lists these domains with brief descriptions: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security, Status, User Management, Utilities, and Help. A note at the bottom states that these domains can be managed by one administrator or separate administrators.

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:07:49 EDT 2017
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.

The screenshot shows the Avaya Application Enablement Services Management Console with the "Licensing" section selected in the left sidebar. The main content area, titled "Licensing", provides instructions on how to set up and maintain the WebLM. It lists the required information for setting up and maintaining the WebLM, and for administering TSAPI Reserved Licenses or DMCC Reserved Licenses. The sidebar also shows sub-items under "Licensing": WebLM Server Address, WebLM Server Access, and Reserved Licenses.

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:09:25 EDT 2017
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

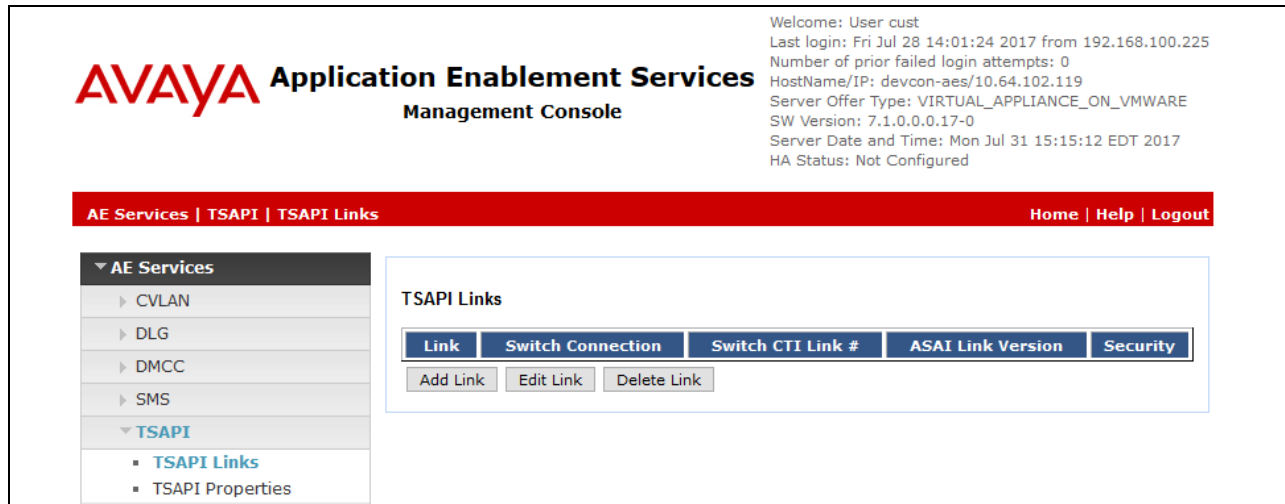
The **Web License Manager** screen below is displayed. Select **Licensed Products** → **APPL_ENAB** → **Application_Enablement** in the left pane to display the **Application Enablement (CTI)** screen in the right pane.

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.

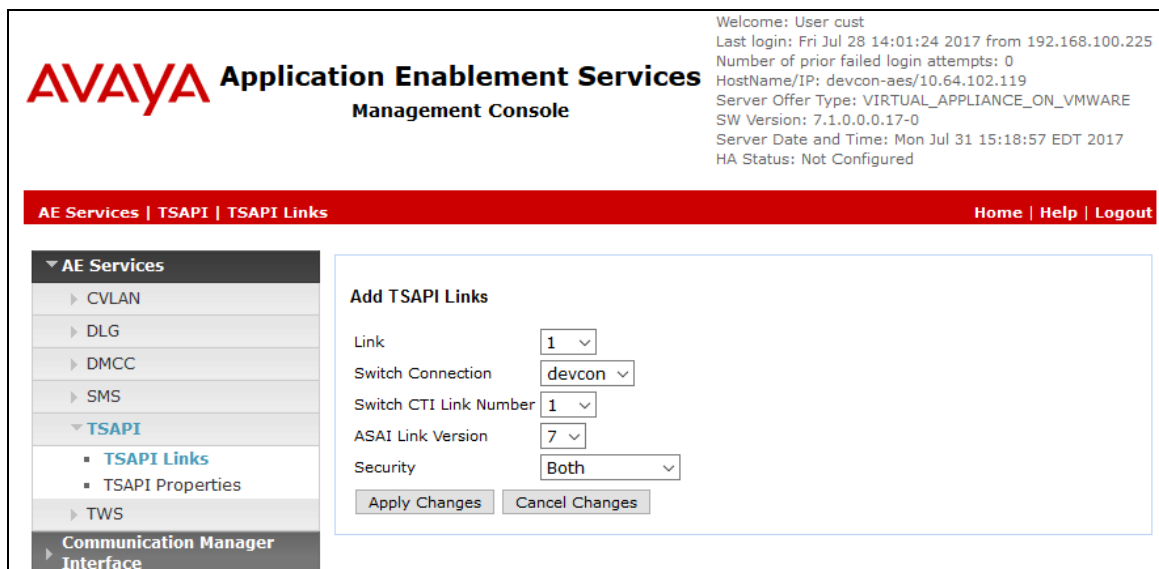
Feature (License Keyword)	Expiration date	Licensed capacity
Unified CC API Desktop Edition VALUE_AES_AEC_UNIFIED_CC_DESKTOP	permanent	1000
CVLAN ASAI VALUE_AES_CVLAN_ASAI	permanent	16
Device Media and Call Control VALUE_AES_DMCC_DMC	permanent	1000
AES ADVANCED SMALL SWITCH VALUE_AES_AEC_SMALL_ADVANCED	permanent	3
DLG VALUE_AES_DLG	permanent	16
TSAPI Simultaneous Users VALUE_AES_TSAPI_USERS	permanent	1000
AES ADVANCED LARGE SWITCH VALUE_AES_AEC_LARGE_ADVANCED	permanent	3
CVLAN Proprietary Links VALUE_AES_PROPRIETARY_LINKS	permanent	16
Product Notes VALUE_NOTES	permanent	SmallServerTypes: s8300c;s8300d;icc;premio;tn8400;laptop;CtiS MediumServerTypes: ibmx306;ibmx306m;dell1950;xen;hs20;hs20_ LargeServerTypes: isp2100;ibmx305;dl380g3;dl385g1;dl385g2;ur TrustedApplications: IPS_001, BasicUnrestricted, DMCUnrestricted; IXP_001, BasicUnrestricted, DMCUnrestricted; IXM_001, BasicUnrestricted, DMCUnrestricted; PC_001, BasicUnrestricted, DMCUnrestricted; CIE_001, BasicUnrestricted, DMCUnrestricted; OSPC_001, BasicUnrestricted, DMCUnrestricted; VP_001, BasicUnrestricted, DMCUnrestricted; SAMETIME_001, VALUE_AES_UNIFIED_CC_DESKTOP;; CCE_0 AdvancedUnrestricted, DMCUnrestricted; CS1, AdvancedUnrestricted, DMCUnrestricted; CS1, AdvancedUnrestricted, DMCUnrestricted; AVA BasicUnrestricted, AdvancedUnrestricted, DMC CCT_ELITE_CALL_CTRL_001, BasicUnrestricted, DMCUnrestricted, AgentEvents; ANAV_001, B AdvancedUnrestricted, DMCUnrestricted, Age UNIFIED_DESKTOP_001, BasicUnrestricted, A DMCUnrestricted, AgentEvents; AACCC_001, B AdvancedUnrestricted, DMCUnrestricted; CE_ BasicUnrestricted, AdvancedUnrestricted, DMC TP_CLIENT_001, BasicUnrestricted, , , AgentE AgentEvents; EXT_CLIENT_002, , , AgentE AgentEvents; EXT_CLIENT_004, , , AgentE AgentEvents; AAWFO_SELECT_001, BasicUn AdvancedUnrestricted, DMCUnrestricted;
AES ADVANCED MEDIUM SWITCH VALUE_AES_AEC_MEDIUM_ADVANCED	permanent	3

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**.



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.



7.4. Administer TCP Settings

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

AVAYA **Application Enablement Services**
Management Console

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:20:23 EDT 2017
HA Status: Not Configured

Networking | TCP / TLS SettingsHome | Help | Logout

▶ AE Services

▶ Communication Manager Interface

▶ High Availability

▶ Licensing

▶ Maintenance

▼ Networking

AE Service IP (Local IP)

Network Configure

Ports

TCP/TLS Settings

▶ Security

▶ Status

▶ User Management

▶ Utilities

▶ Help

TCP / TLS Settings

TLSv1 Protocol Configuration

☐ Support TLSv1.0 Protocol

☐ Support TLSv1.1 Protocol

☒ Support TLSv1.2 Protocol

TCP Retransmission Count

☐ Standard Configuration (15)

☒ TSAPI Routing Application Configuration (6)

Apply Changes

Restore Defaults

Cancel Changes

Note: A smaller TCP Retransmission Count reduces the amount of time that the AE Services server waits for a TCP acknowledgement before closing the socket. Select the Standard Configuration setting unless this AE Services server is used by TSAPI routing applications.

Warning: This setting applies to all TCP and TLS sockets on the AE Services Server and so it should be used with caution.

7.5. Restart Service

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

AVAYA **Application Enablement Services**
Management Console

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:21:40 EDT 2017
HA Status: Not Configured

Maintenance | Service ControllerHome | Help | Logout

▶ 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](#)

StartStopRestart ServiceRestart AE ServerRestart LinuxRestart Web Server

7.6. 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.

The screenshot displays the Avaya Application Enablement Services Management Console. The top header includes the Avaya logo and the title "Application Enablement Services Management Console". On the right, a welcome message for user "cust" is shown, along with login details and system status. A red navigation bar contains links for "Security", "Security Database", "Tlinks", "Home", "Help", and "Logout". The left sidebar lists various services, with "Security" expanded to show "Security Database" and "Tlinks" selected. The main content area, titled "Tlinks", shows a "Tlink Name" field with two radio button options: "AVAYA#DEVCON#CSTA#DEVCON-AES" (selected) and "AVAYA#DEVCON#CSTA-S#DEVCON-AES". A "Delete Tlink" button is also present.

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.0.17-0
Server Date and Time: Mon Jul 31 15:22:34 EDT 2017
HA Status: Not Configured

AVAYA Application Enablement Services
Management Console

Security | Security Database | Tlinks Home | Help | Logout

Tlinks

Tlink Name

☒ AVAYA#DEVCON#CSTA#DEVCON-AES
☐ AVAYA#DEVCON#CSTA-S#DEVCON-AES

Delete Tlink

7.7. Administer Callback User

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

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

AVAYA **Application Enablement Services**
Management Console

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:24:21 EDT 2017
HA Status: Not Configured

User Management | User Admin | Add UserHome | 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 Idvht

* Common Namevht

* Surnamevht

* User Password••••••••

* Confirm Password••••••••

Admin Note

Avaya RoleNone

Business Category

Car License

CM Home

Css Home

CT UserYes

Department Number

Display Name

Employee Number

7.8. Verify Security Database

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

Verify that **Enable SDB for TSAPI Service, JTAPI and Telephony Web Services** retained the default value of unchecked. In the event that security database is used by the customer with this parameter already enabled, then follow [2] to configure access privileges for the Callback user from **Section 7.7**.

The screenshot displays the Avaya Application Enablement Services Management Console. The top header includes the Avaya logo and the title "Application Enablement Services Management Console". A welcome message and system information are shown in the top right corner. The main navigation pane on the left lists various services, with "Security" expanded to show "Security Database" and "Control" selected. The main content area displays the "SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services" configuration page, which includes two unchecked checkboxes and an "Apply Changes" button.

Welcome: User cust
Last login: Fri Jul 28 14:01:24 2017 from 192.168.100.225
Number of prior failed login attempts: 0
HostName/IP: devcon-aes/10.64.102.119
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.1.0.0.17-0
Server Date and Time: Mon Jul 31 15:25:21 EDT 2017
HA Status: Not Configured

AVAYA Application Enablement Services Management Console

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 “https://<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**.

The screenshot shows the VHT Service Provider Editor interface. At the top, there is a header with the VHT logo and the text "Powered by the Holly Voice Platform". Below the header, there is a navigation bar with links for "Administration", "Reports", "Configuration", and "Dashboard". On the right side of the header, there are dropdown menus for "<all service providers>", "<all affiliates>", and "<all applications>". The user is logged in as "administrator" and can click "Logout".

The main content area is titled "Service Provider Editor". It contains three sections:

- Select Service Provider:** This section has three input fields: "Service Provider:" (a dropdown menu with "VHT" selected), "Domain Name:" (a text box with "VHT" entered), and "Domain Description:" (a text box with "VHT" entered). There is an "Edit Affiliates..." button to the right of these fields.
- Service Provider Contact Details:** This section has four input fields: "Name:", "Email:", "Phone:", and "Address:", each with a corresponding text box.
- Licence Port Allocation:** This section has two input fields: "Max Available Ports:" (a text box with "999" entered) and "Warn Ports:" (a text box with "0" entered).

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 shows two configuration windows. The top window, titled "Application Parameters", has fields for "Key:" and "Value:". Below these is a list box for "Preset Parameters:" with the option "Set Application Type To CCXML" selected. To the right are buttons for "Set", "Replace", and "Delete". At the bottom of this window are buttons for "Delete the Service Provider", "Revert", and "Save Service Provider". The bottom window, titled "Numbers Available", has a "DNIS Numbers:" label and a list box containing "78701 - 78701" and "outbound - outbound". To the right of this list box are buttons for "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.

The screenshot shows the "VHT" logo and "Powered by the Holly Voice Platform" text at the top left. The top right shows the version "HVP-6.3.7-2391-40364" and navigation links for "Administration", "Reports", "Configuration", and "Dashboard". The user is logged in as "administrator". The main section is titled "Affiliate Editor". It contains a "Select Affiliate" section with dropdown menus for "Service Provider:" (set to "VHT"), "Affiliate:" (set to "VHT-Aff"), and text input fields for "Domain Name:" (set to "VHT-Aff") and "Domain Description:" (set to "VHT-Aff"). Below this are buttons for "Edit Service Provider..." and "Edit Applications...". The "Affiliate Contact Details" section has input fields for "Name:", "Email:", "Phone:", and "Address:". The "Licence Port Allocation" section has input fields for "Max Available Ports:" (set to "0") and "Warn Ports:" (set to "0"), with a note "(Available 999)".

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 panels. The top panel, titled "Application Parameters", includes fields for "Key:" and "Value:", a large text area, and a "Preset Parameters:" dropdown menu currently showing "Set Application Type To CCXML". To the right of these fields are buttons for "Set", "Replace", and "Delete", along with a "Set" button for the preset parameters. Below this panel are three buttons: "Delete the Affiliate", "Revert", and "Save Affiliate". The bottom panel, titled "Numbers Available", features a "DNIS Numbers:" label, two input fields separated by a hyphen, and a list box containing "78701 - 78701" and "outbound - outbound". To the right of the list box are buttons for "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).

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 image shows two configuration windows from a software interface.

Application Parameters

Key: Value:

Preset Parameters:

Buttons: Set, Replace, Delete, Set

Buttons: Delete the Application, Revert, Save Application

Application Numbers

Numbers Available

DNIS Numbers: -

Buttons: Add, Replace, 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=HVPavaya*).

The image shows the VHT Application Editor interface. The header includes the VHT logo, "Powered by the Holly Voice Platform", and navigation links: Administration, Reports, Configuration, Dashboard. The user is logged in as "administrator".

Application Editor

Select Application

Service Provider:
 Affiliate:
 Application:
 Name:
 Description:
 Licence Exception URL:

Buttons: Edit Affiliate...

URLs

URL:
 Fetch Time Out: sec
 URLs:
 Buttons: Replace, Delete, Move Up, Move Down

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

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

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

The screenshot displays two sections of a software interface. The top section, titled "Application Parameters", contains a "Key:" field with the value "type" and a "Value:" field with the value "application/voicexml+xml". Below these fields is a list box containing the entry "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 this section are buttons for "Delete the Application", "Revert", and "Save Application". The bottom section, titled "Application Numbers", contains a "Numbers Available" header. Below this is a "DNIS Numbers:" section with two input fields, both containing the text "outbound", separated by a hyphen. Below the input fields is a list box containing the entry "outbound - outbound". To the right of the list box are buttons for "Add", "Replace", and "Delete".

9. Configure VHT Callback

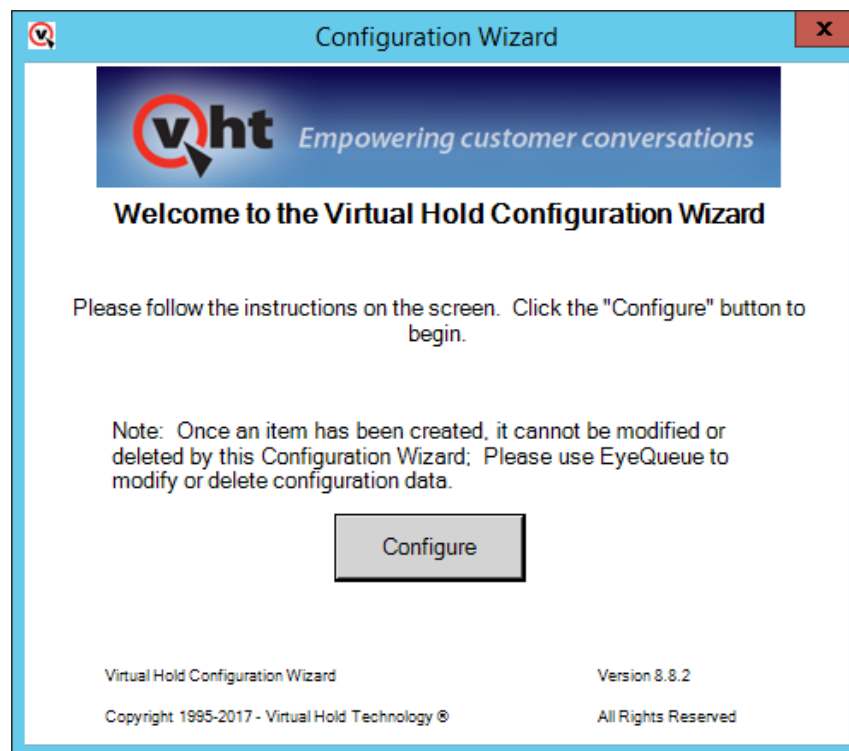
This section provides the procedures for configuring Callback. The procedures include the following areas:

- Launch VHT Configuration Wizard
- Administer Switch Connection
- Administer Genesys CTI T-Server Connections
- 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.

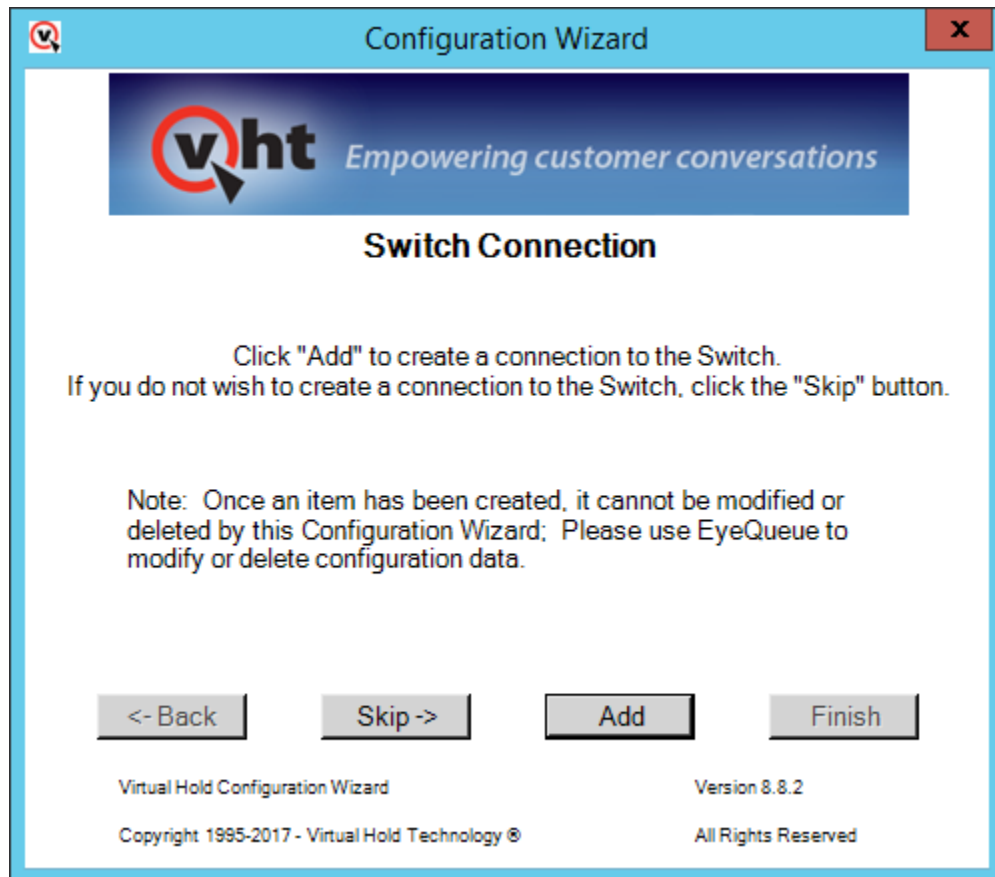
9.1. Launch Configuration Wizard

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.

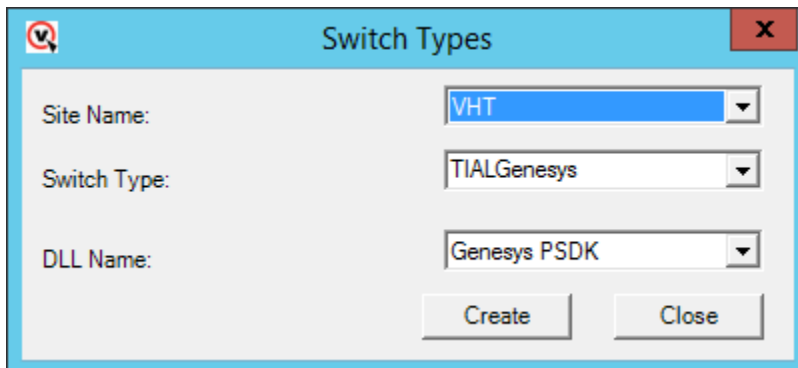


9.2. Administer Switch Connection

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 *TIALGenesys* from the drop-down list. For **DLL Name**, select *Genesys PSDK* 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.



9.3. Administer Genesys CTI T-Server Connections

Continue with the wizard until the **Genesys CTI T-Server Connections** screen is displayed (not shown). Click **Add** to create connection.

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

- **T-server Switch Name:** A descriptive name (e.g., *Avaya_Switch*).
- **Host IP Address A:** Set to the Callback IP address.
- **Host Port A:** Set to *4000*.
- **Host IP Address B:** Set to the Callback IP address.

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

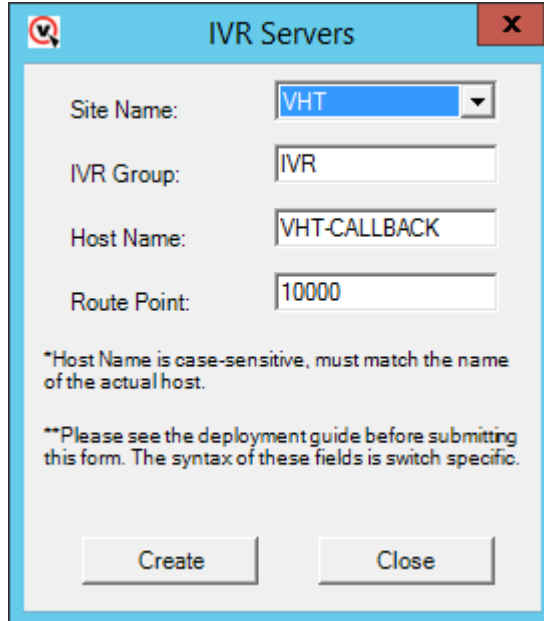
Field	Value
Site Name	VHT
T-Server Switch Name	Avaya_Switch
Host IP Address A	10.64.102.106
Host Port A	4000
Host IP Address B	10.64.102.106
Host Port B	
Redundancy Mode	None
Reconnect Interval	2000
Register All Devices	FALSE
Accept Only These Events	
Protocol	

At the bottom, the status message reads: 'GenesysCTI: Avaya_Switch created.' The 'Create' button is highlighted with a dashed border.

9.4. 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.



The image shows a software window titled "IVR Servers" with a standard Windows-style title bar (minimize, maximize, close buttons). The window contains a form with the following fields:

- Site Name:** A dropdown menu with "VHT" selected.
- IVR Group:** A text input field containing "IVR".
- Host Name:** A text input field containing "VHT-CALLBACK".
- Route Point:** A text input field containing "10000".

Below the input fields, there are two lines of instructional text:

- *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.

At the bottom of the window, there are two buttons: "Create" and "Close".

9.5. Administer Queues

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

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

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:** VHT_Test (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 (all checked)
 - Time Begin:** 00:00 (all days)
 - Time End:** 23:59 (all days)
- Callbacks Offered:**
 - Day Of Week:** Sun, Mon, Tue, Wed, Thu, Fri, Sat (all checked)
 - Time Begin:** 00:00 (all days)
 - Time End:** 23:59 (all days)
- Callbacks Allowed:**
 - Day Of Week:** Sun, Mon, Tue, Wed, Thu, Fri, Sat (all checked)
 - Sched callbacks allowed/15 min:** 15 (all days)

Queue 'VHT_Test' created.

Buttons: Create, Close

9.6. 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**. For **Route Device** and **Transfer Device**, enter “sip:x@y”, where “x” is the Hold VDN extension, and “y” is the IP address of the Session Manager signaling interface (e.g., *sip:77202@10.64.102.117*).

Retain the default values for the remaining fields.

Callback and Holding Queues

Site Name: VHT

T-Server Switch Name: Avaya_Switch

Callback Queues

☒ Use T-Server Switch Name prefix

Callback Queue ID*: 77203

Transfer Device: sip:77203@10.64.10

Callback Queue Avaya_Switch:77203 created.

Create

Holding Queues

☒ Use T-Server Switch Name prefix

Holding Queue ID*: 77202

Route Device: sip:77202@10.64.10

Transfer Device: sip:77202@10.64.10

Holding Queue Avaya_Switch:77202 created.

Create

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

* Verify T-Server Switch Name

Close

9.7. 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

T-Server Switch Name: Avaya_Switch

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: Avaya_Switch:77202

Callback Queue ID: Avaya_Switch:77203

UnderThreshold Queue ID: Avaya_Switch:77202

IB IVR Extension Group: NONE

OB IVR Extension Group: NONE

Incoming Extension: 77201 created. **Create**

* Verify T-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

T-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: Avaya_Switch:77202

Callback Queue ID: Avaya_Switch:77203

UnderThreshold Queue ID: Avaya_Switch:77202

IB IVR Extension Group: NONE

OB IVR Extension Group: NONE

Incoming Extension: 78701 created. **Create**

* Verify T-Server Switch Name **Close**

9.8. 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: "Update Country Id Dial Prefix and Suffix" on the left and "Update Phone Number Validation Min/Max Length" on the right.

Update Country Id Dial Prefix and Suffix:

- Site Name: VHT (dropdown menu)
- Country Search: 1 - North America (dropdown menu with a list box below it showing "1 - North America" selected)
- Dial Prefix: (empty text field)
- Dial Suffix: (empty text field)
- Update Successful Dial Prefix: (text label)
- Dial Suffix: (text label)
- Update: (button)

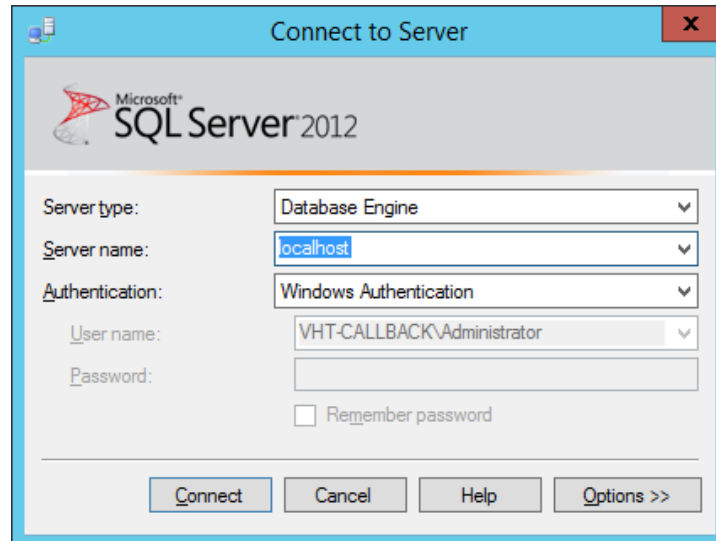
Update Phone Number Validation Min/Max Length:

- Site Name: VHT (dropdown menu)
- Country Id: 1 - North America (dropdown menu)
- Min Length: 5 (text field)
- Max Length: 12 (text field)
- Update Successful Min Length: 5 Max Length: 12 (text label)
- Update: (button)

At the bottom right of the dialog is a "Close" button.

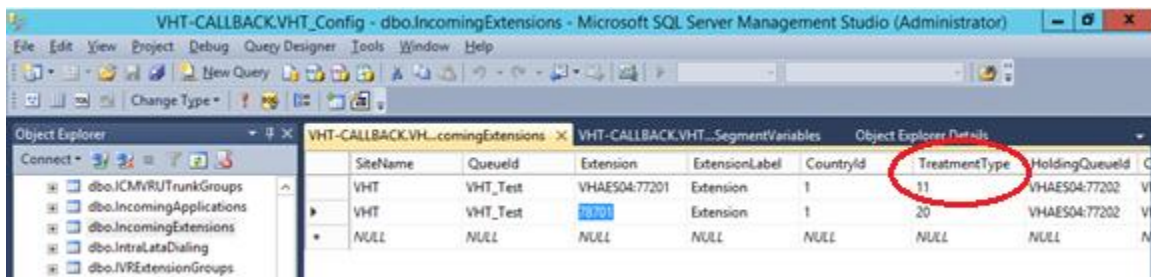
9.9. Administer Segment Variables

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



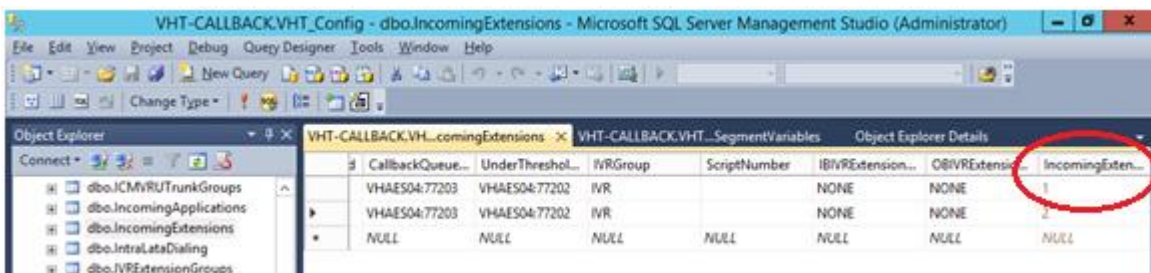
Navigate to **Databases → VHT_Config → Tables → dbo.IncomingExtensions** in the left pane, right-click the entry and select **Edit Top 200 Rows**.

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



SiteName	QueueId	Extension	ExtensionLabel	CountryId	TreatmentType	HoldingQueueId
VHT	VHT_Test	VHAES04:77201	Extension	1	11	VHAES04:77202
VHT	VHT_Test	70701	Extension	1	20	VHAES04:77202
NULL	NULL	NULL	NULL	NULL	NULL	NULL

Scroll to the right to make a note of the associated **IncomingExtensionsId** value, in this case ‘1’, as shown below.

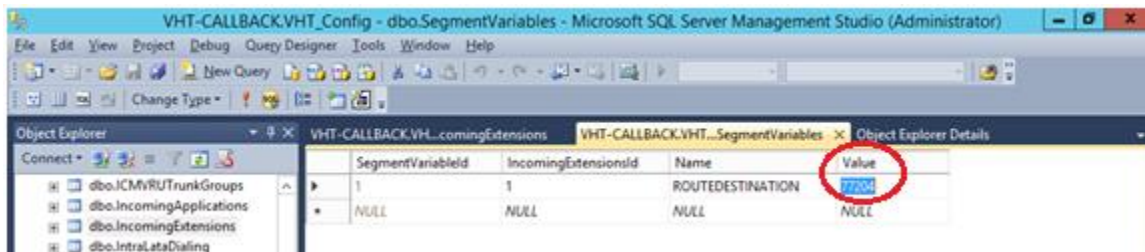


CallbackQueue...	UnderThreshol...	IVRGroup	ScriptNumber	IBIVRExtension...	OBIVRExtension...	IncomingExten...
VHAES04:77203	VHAES04:77202	IVR		NONE	NONE	1
VHAES04:77203	VHAES04:77202	IVR		NONE	NONE	2
NULL	NULL	NULL	NULL	NULL	NULL	NULL

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).



9.10. 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 (10.64.102.106), 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., 3306702285@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, "vht-callback"},
  {ivr_port_send_interval_ms, 2000},
  {disposition_url, "http://10.64.102.106:4153/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, false},
      {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/vht-ivg/vht_hvp.ccxml"},
        {ani, "3306702285@10.64.102.117"},
        {node_id, 5},
        {agent_priority_dnis, "agntpriority"},
        {agent_result_url, "http://10.64.102.107:8080/vht-ivg/agent_priority"},
        {outreach_dnis, "outreach"}
      ]
    ]
  }
]
}
}

```

9.11. Configure Genesys Real-Time Adapter

The Callback Genesys 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_GenesysRealTimeAdapter_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 (10.64.102.106) for the **bolded** entries as shown below. In addition, the **SiteName** should be set to the appropriate value.

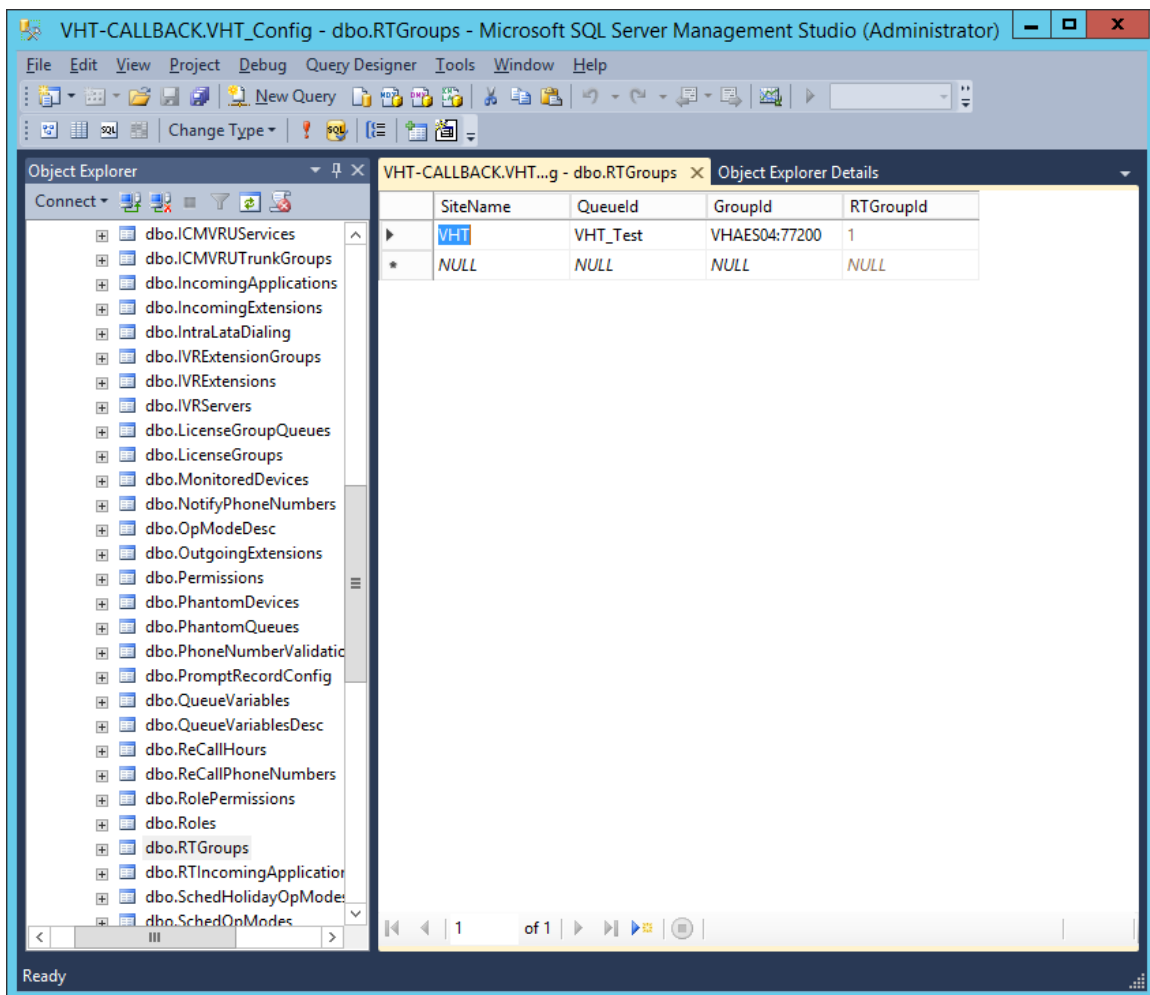
```
<?xml version="1.0" encoding="utf-8"?>
<configuration>
  <configSections>
    <sectionGroup name="VHTConfiguration">
      <section name="vhtLogging"
type="VHT.Common.Library.Configuration.Logging.VHTLoggingSection, VHT.Common.Library"
allowLocation="true" allowDefinition="Everywhere"/>
      <section name="vhtCommunication"
type="VHT.Common.Library.Configuration.Communication.VHTCommunicationSection,
VHT.Common.Library" allowLocation="true" allowDefinition="Everywhere"/>
      <section name="statServer"
type="RealTimeAdapters.Configuration.Sections.StatServerSection, RealTimeAdapters"
allowLocation="true" allowDefinition="Everywhere"/>
    </sectionGroup>
  </configSections>
  <VHTConfiguration>
    <vhtLogging>
      <application level="10" name="GenesysRealTimeAdapter"
logFilePath="C:\Program Files (x86)\Virtual Hold Technology\VHLogs"/>
    </vhtLogging>
    <vhtCommunication>
      <QMCL reconnectIntervalSeconds="3">
        <Connections>
          <Connection connectionType="Primary">
            <Server ipAddress="10.64.102.106" port="6999"/>
            <Client ipAddress="10.64.102.106" port="0"/>
          </Connection>
        </Connections>
      </QMCL>
    </vhtCommunication>
    <statServer tenant="Resources" password="" clientName="VHTGenRTAdapter"
intervalFrequencySecs="15"> <!-- ipVersion -->
      <servers>
        <add name="primary" host="10.64.102.106" port="5100"/>
      </servers>
      <!-- <callsInAcd statType="" /> -->
      <agentsAvailable statType="VHT_CurrNumberWaitStatuses"
category="CurrentNumber" subject="AgentStatus" mainMask="WaitForNextCall"/>
      <agentsStaffed statType="VHT_CurrAgentsLoggedIn" category="CurrentNumber"
subject="AgentStatus" mainMask="*,~NotMonitored,~LoggedOut"/>
    </statServer>
  </VHTConfiguration>
  <appSettings>
    <add key="VhqmwsUrl" value="http://10.64.102.106/VHQMWS/VHQMWS.asmx"/>
    <add key="SiteName" value="VHT"/>
    <add key="UseTialAdapter" value="TRUE"/>
  </appSettings>
</configuration>
```

```

    <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., *VHAES04:77200*) 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**.


Application Enablement Services
Management Console

Welcome: User cust
Last login: Mon Jul 31 15:07:14 2017 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: 7.1.0.0.17-0
Server Date and Time: Tue Aug 01 13:03:23 EDT 2017
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
▶ Log Manager
▶ Logs
▼ 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	Fri Jul 21 12:18:31 2017	Online	17	7	134	134	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 7.1

Last Logged on at August 1, 2017 1:07 PM
Go... Log off admin

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

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.

All Entity Links to SIP Entity: VHT-IVG

Status Details for the selected Session Manager:

Summary View

1 Items Refresh Filter: Enable

Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/> devcon-sm	IPv4	10.64.102.107	5060	UDP	FALSE	UP	200 OK	UP

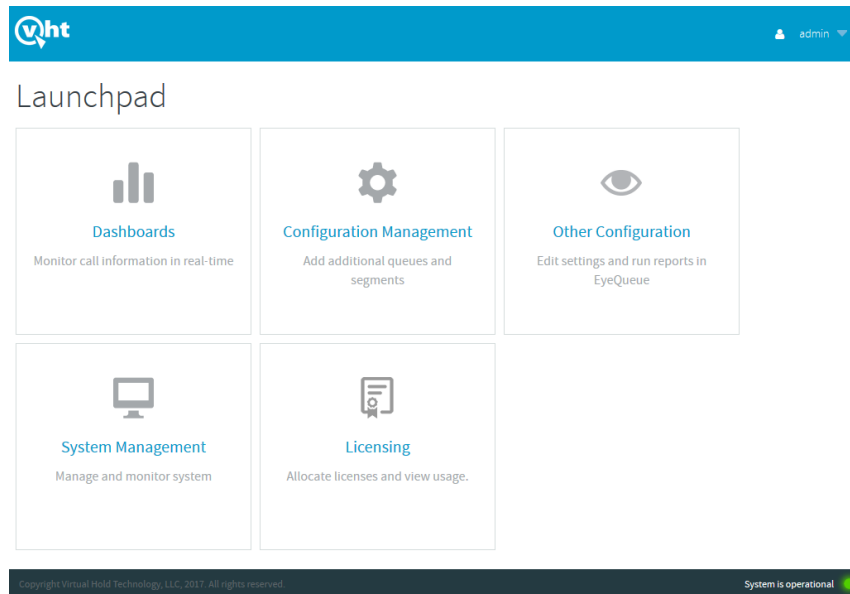
10.4. Verify VHT Callback and IVG

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.

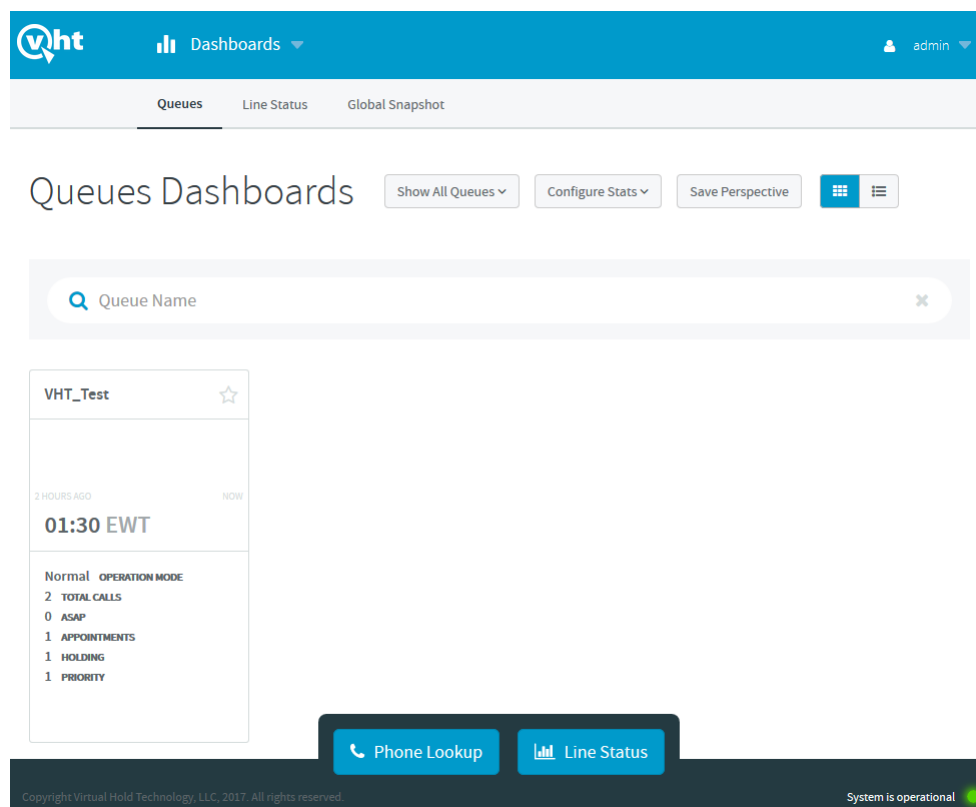
User name

Password

The **Launchpad** screen below is displayed. 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 screen is updated reflecting proper active calls and expected wait time (EWT) as shown below.



To verify the **Queue Statistics** using the Genesys real-time adapter, select **Other Configuration** from the **Launchpad** screen. The screen below is displayed. Select **Queue Watch**.

QueueWatch

QueueWatch is a dashboard that allows real-time viewing of ASAP callbacks and scheduled callbacks in the Virtual Hold system. Also, this area lets you cancel specific callbacks.

This area of EyeQueue allows you to see calls that are currently being treated by Virtual Hold, pending callbacks (ASAP callbacks) and scheduled callbacks (Appointments). This area also lets you cancel specific callbacks.

[Back](#)

From the **QueueWatch** screen (not shown), select **Queue Statistics** to display real-time data for the ACD split. The following data shows an ACD split with two agents that are unavailable and one caller waiting for a callback.

Queue Statistics - Mozilla Firefox

vht-callback/VHQueueWatch/AjaxQueueStats.aspx?SiteName=VHT&Columns=QueueName|OpMode|ModeStatus|EWT|AgentsAvailable|AgentsStaffe

Queue Name	Op Mode	Mode Status	EWT	Agents Available	Agents Staffed	ACD Queue	Holding Queue	ASAP Callbacks	Priority Queue	Calls in IVR	Total Calls in VH	Appts	Retries
VHT_Test	Normal		00:00:23	0	2	0	0	1	0	0	1	0	0

11. Conclusion

These Application Notes describe the steps required to integrate VHT Callback using Genesys T-Server 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 7.1, Issue 1, May 2017, available at <http://support.avaya.com>.
2. *Administering and Maintaining Avaya Aura® Application Enablement Services*, Release 7.1, Issue 1, May 2017, available at <http://support.avaya.com>.
3. *Virtual Hold Configuration Guide Version 8.8 or Later*, July 31, 2017, available upon request to Virtual Hold Support.
4. *Virtual Hold Installation Guide Version 8.8 or Later*, July 12, 2017, available upon request to Virtual Hold Support.

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