Abstract

These Application Notes describe the configuration steps required to integrate Sabio CallBack with an Avaya Automatic Call Distribution (ACD) solution. The compliance tested solution consisted of Avaya Aura™ Communication Manager, Avaya Voice Portal, Avaya Aura™ Application Enablement Services and a web server running Sabio CallBack.

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 compliance tested configuration using Avaya Aura™ Communication Manager, Avaya Voice Portal, Avaya Aura™ Application Enablement Services and Sabio CallBack. The Sabio CallBack solution is designed to add value to the customer experience as well as increasing call centre efficiency by allowing customers, during peak times, to schedule a call back rather than waiting in queue for a call centre agent.

Avaya Voice Portal is used to access VXML applications on the Sabio CallBack web server. Voice prompts stored on the Sabio CallBack web server are used to offer the call back, and where necessary, prompt for required information. Once the call back has been accepted, a set of pre-defined rules schedule when the call back will occur. To achieve the call back Avaya Aura™ Application Enablement Services is used to place a virtual call in to a queue to reserve an agent. When the agent is reserved, Avaya Aura™ Application Enablement Services is used to place a second call to the customer and conference the two called parties together.

Using the call vectoring functionality available on Avaya Aura™ Communication Manager the Sabio CallBack solution can be incorporated into pre-existing call routing strategies. These Application Notes describe the minimum call vector requirements for Sabio CallBack, details on vector programming can be found in References [1] and [2] Section 11.

1.1 Interoperability Compliance Testing

The interoperability compliance testing focused on the ability of the Sabio solution to interoperate with the Avaya solution. The following is a summary of the feature, functionality and serviceability testing that was undertaken:

- Access to Sabio CallBack from Call vector
- CLI recognition and confirmation
- Number prompt when no CLI present
- Numbers barred from requesting a call back rejected
- Invalid number formats rejected
- Maximum number of call back attempts
- Defined schedule allows/ prevents call backs respectively
- Call queuing scenarios such as multiple calls, agents busy, agents logged off, etc.
- Call back failures including, busy, unobtainable and unanswered calls
- Additional data capture when call back is accepted
- Agent whisper replay of data capture
- Agent accept and reject of call back
- Failure scenarios including recovery from network failures and system failures

1.2 Support

For technical support of Sabio products, please check for the local support contact at the following web address: www.sabio.co.uk
2 Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The solution consisted of Communication Manager, Voice Portal, Application Enablement Services (AE Services) and Sabio CallBack web server. Voice Portal is connected to Communication Manager using H.323 VoIP connections with 10 ports configured, Voice Portal also communicates with Sabio CallBack using HTTP. Sabio CallBack communicates with Communication Manager through a TSAPI connection with Application Enablement Services server to obtain call information and perform call control activities.

3 Equipment and Software Validated

All the hardware and associated software used in the compliance testing is listed below.

<table>
<thead>
<tr>
<th>Equipment</th>
<th>Software Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya S8730 Server</td>
<td>Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4)</td>
</tr>
<tr>
<td>Avaya G650 Media Gateway</td>
<td></td>
</tr>
<tr>
<td>- CLAN TN799DP</td>
<td>HW15, FM49</td>
</tr>
<tr>
<td>- IP Media Processor TN2602AP</td>
<td>HW01, FM34</td>
</tr>
<tr>
<td>Avaya Voice Portal</td>
<td>Avaya Voice Portal 4.1 (VPMS 4.1.0.3.0111) (MPP 4.1.0.3-0002)</td>
</tr>
<tr>
<td>Avaya Aura™ Application Enablement Services</td>
<td>Avaya Aura™ Application Enablement Services 4.2.2 patch 4 (r4-2-2-31-0)</td>
</tr>
<tr>
<td>Sabio Web Server</td>
<td>Sabio CallBack v2.1.24.0</td>
</tr>
</tbody>
</table>

Table 1: Hardware and Software Version Numbers
4 Configuration of Avaya Aura™ Communication Manager

These Application Notes assume that Communication Manager is installed and operational. This section describes the steps for configuring the Communication Manager call routing and to work with Sabio CallBack as well as the integration steps for Voice Portal and Application Enablement Services. All configurations in the section are administered using the System Access Terminal (SAT). The procedures covered include the following:

4.1 Confirm Necessary Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Log into the Communication Manager SAT interface and use the `display system-parameters customer-options` command to determine these values. On Page 3 verify the fields ARS and Computer Telephony Adjunct Links are set to y.

<table>
<thead>
<tr>
<th>display system-parameters customer-options</th>
<th>Page 3 of 11</th>
</tr>
</thead>
<tbody>
<tr>
<td>OPTIONAL FEATURES</td>
<td></td>
</tr>
<tr>
<td>Abbreviated Dialing Enhanced List? n</td>
<td>Audible Message Waiting? n</td>
</tr>
<tr>
<td>Access Security Gateway (ASG)? n</td>
<td>Authorization Codes? y</td>
</tr>
<tr>
<td>Analog Trunk Incoming Call ID? n</td>
<td>CAS Branch? n</td>
</tr>
<tr>
<td>A/D Grp/Sys List Dialing Start at 01? n</td>
<td>CAS Main? n</td>
</tr>
<tr>
<td>Answer Supervision by Call Classifier? n</td>
<td>Change COR by FAC? n</td>
</tr>
<tr>
<td>ARS? y</td>
<td>Computer Telephony Adjunct Links? y</td>
</tr>
<tr>
<td>ARS/AAR Partitioning? y</td>
<td>Cvg Of Calls Redirected Off-net? n</td>
</tr>
</tbody>
</table>

On Page 6 confirm that the call centre features highlighted below are activated, i.e. set to y.

<table>
<thead>
<tr>
<th>display system-parameters customer-options</th>
<th>Page 6 of 11</th>
</tr>
</thead>
<tbody>
<tr>
<td>CALL CENTER OPTIONAL FEATURES</td>
<td></td>
</tr>
<tr>
<td>Call Center Release: 5.0</td>
<td></td>
</tr>
<tr>
<td>ACD? y</td>
<td>Reason Codes? n</td>
</tr>
<tr>
<td>BCMS (Basic)? y</td>
<td>Service Level Maximizer? n</td>
</tr>
<tr>
<td>BCMS/VuStats Service Level? n</td>
<td>Service Observing (Basic)? y</td>
</tr>
<tr>
<td>BSR Local Treatment for IP &amp; ISDN? n</td>
<td>Service Observing (Remote/By FAC)? n</td>
</tr>
<tr>
<td>Business Advocate? n</td>
<td>Service Observing (VDNs)? n</td>
</tr>
<tr>
<td>Call Work Codes? n</td>
<td>Timed ACW? n</td>
</tr>
<tr>
<td>DTMF Feedback Signals For VRU? n</td>
<td>Vectoring (Basic)? y</td>
</tr>
<tr>
<td>Dynamic Advocate? n</td>
<td>Vectoring (Prompting)? y</td>
</tr>
<tr>
<td>Expert Agent Selection (EAS)? y</td>
<td>Vectoring (G3V4 Enhanced)? y</td>
</tr>
<tr>
<td>EAS-PHD? n</td>
<td>Vectoring (3.0 Enhanced)? y</td>
</tr>
<tr>
<td>Forced ACD Calls? n</td>
<td>Vectoring (ANI/II-Digits Routing)? n</td>
</tr>
<tr>
<td>Least Occupied Agent? n</td>
<td>Vectoring (G3V4 Advanced Routing)? y</td>
</tr>
<tr>
<td>Lookahead Interflow (LAI)? n</td>
<td>Vectoring (CINFO)? n</td>
</tr>
<tr>
<td>Multiple Call Handling (On Request)? n</td>
<td>Vectoring (Best Service Routing)? n</td>
</tr>
<tr>
<td>Multiple Call Handling (Forced)? n</td>
<td>Vectoring (Holidays)? n</td>
</tr>
<tr>
<td>PASTE (Display PBX Data on Phone)? n</td>
<td>Vectoring (Variables)? y</td>
</tr>
</tbody>
</table>
On Page 9 confirm Adjunct Routing, CTI Stations, Phantom Calls and Agent States are set to y.

```
display system-parameters customer-options

ASAI ENHANCED FEATURES

  Adjunct Routing? y
  CTI Stations? y
  Increased Adjunct Route Capacity? n
  Phantom Calls? y

ASAI PROPRIETARY FEATURES

  Agent States? y

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

4.2 Define System Features

Use change system-parameters features to administer system-wide features. On Page 11, there are a number of settings that affect the behavior when a converse-on vector step is used. For the compliance test a converse-on vector step is used to access Sabio CallBack. The settings used during the compliance test are highlighted below.

```
change system-parameters features

FEATURE-RELATED SYSTEM PARAMETERS

CALL CENTER SYSTEM PARAMETERS

  EAS
    Expert Agent Selection (EAS) Enabled? y
    Minimum Agent-LoginID Password Length:
    Direct Agent Announcement Extension: Delay:
    Message Waiting Lamp Indicates Status For: station

VECTORING

  Converse First Data Delay: 1 Second Data Delay: 1
  Converse Signaling Tone (msec): 100 Pause (msec): 70
  Prompting Timeout (secs): 10

  Reverse Star/Pound Digit For Collect Step? n
```
4.3 Define Feature Access Codes (FAC)

A FAC (feature access code) should be defined for each feature that will be used. Use **change feature-access-codes** to define the required access codes. On Page 1, define a **Auto Route Selection (ARS) - Access Code 1**. This is required by Sabio CallBack when placing external calls.

---

**change feature-access-codes**

<table>
<thead>
<tr>
<th>FEATURE ACCESS CODE (FAC)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abbreviated Dialing List1 Access Code:</td>
</tr>
<tr>
<td>Abbreviated Dialing List2 Access Code:</td>
</tr>
<tr>
<td>Abbreviated Dialing List3 Access Code:</td>
</tr>
<tr>
<td>Abbreviated Dial - Prgm Group List Access Code:</td>
</tr>
<tr>
<td>Announcement Access Code: *56</td>
</tr>
<tr>
<td>Answer Back Access Code: *59</td>
</tr>
<tr>
<td>Attendant Access Code:</td>
</tr>
<tr>
<td>Auto Alternate Routing (AAR) Access Code:</td>
</tr>
<tr>
<td><strong>Auto Route Selection (ARS) - Access Code 1</strong>: 9 Access Code 2:</td>
</tr>
<tr>
<td>Automatic Callback Activation:</td>
</tr>
<tr>
<td>Deactivation:</td>
</tr>
<tr>
<td>Call Forwarding Activation Busy/DA:</td>
</tr>
<tr>
<td>All:</td>
</tr>
<tr>
<td>Deactivation:</td>
</tr>
<tr>
<td>Call Forwarding Enhanced Status:</td>
</tr>
<tr>
<td>Act:</td>
</tr>
<tr>
<td>Deactivation:</td>
</tr>
</tbody>
</table>

---

On Page 5 define a FAC for each of the following:

- **Auto-In Access Code**: When activated, this feature will set the ACD agent to a state where they are available to handle calls, upon completion of a call the agent will automatically be made available again.

- **Aux Work Access Code**: When activated, this feature will set the ACD agent to an Auxiliary work state, this is the default state for an agent upon first login.

- **Login Access Code**: This feature allows ACD agents to log in to an extension.

- **Logout Access Code**: This feature allows ACD agents to log out of an extension.

- **Manual-in Access Code**: When activated this feature will set the ACD agent to a state where they are available to handle calls, upon completion of a call the agent will be unavailable until the feature is activated again.

---

**change feature-access-codes**

<table>
<thead>
<tr>
<th>FEATURE ACCESS CODE (FAC)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automatic Call Distribution Features</td>
</tr>
<tr>
<td>After Call Work Access Code:</td>
</tr>
<tr>
<td>Assist Access Code:</td>
</tr>
<tr>
<td><strong>Auto-In Access Code</strong>: *27</td>
</tr>
<tr>
<td><strong>Aux Work Access Code</strong>: *28</td>
</tr>
<tr>
<td><strong>Login Access Code</strong>: *25</td>
</tr>
<tr>
<td><strong>Logout Access Code</strong>: *26</td>
</tr>
<tr>
<td><strong>Manual-in Access Code</strong>: *29</td>
</tr>
</tbody>
</table>
4.4 Configure Node-Names IP

Use the `change node-names ip` command. Add an entry in the node-names form for the CLAN, AE Services server and the default gateway used for the IP network the CLAN will be connected to. A Name and IP Address should be added for each. The values used during the interoperability test are highlighted below.
4.5 Configure CLAN for AE Services Connectivity

Add the CLAN to the system configuration using the `add ip-interface n` command, where `n` is the CLAN board location. Enter the CLAN node name assigned in Section 4.4 to the Node Name field. Enter values for the Subnet Mask and Gateway Address fields. In this case, /24 and 10.20.2.1 are used to correspond to the network configuration in these Application Notes. Set the Enable Interface field to y, and use a separate Network Region for the CLAN dedicated for AE Services connectivity. Default values may be used in the remaining fields.

```
add ip-interface 01a02

IP INTERFACES

Type: C-LAN
Slot: 01A02
Code/Suffix: TN799 D
Enable Interface? y
VLAN: n
Network Region: 1

IPV4 PARAMETERS

Node Name: CLAN
Subnet Mask: /24
Gateway Node Name: Gateway

Ethernet Link: 1
Network uses 1's for Broadcast Addresses? y
```

4.6 Configure Transport Link for AE Services Connectivity

To administer the transport link to AE Services, use the `change ip-services` command. On Page 1, add an entry with the following values:

- **Service Type**: should be set to AESVCS
- **Enabled**: set to y
- **Local Node**: set to the node name assigned for the CLAN in Section 4.4.
- **Local Port**: Retain the default value of 8765.

```
change ip-services

IP SERVICES

<table>
<thead>
<tr>
<th>Service Type</th>
<th>Enabled</th>
<th>Local Node</th>
<th>Local Port</th>
<th>Remote Node</th>
<th>Remote Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>AESVCS</td>
<td>y</td>
<td>CLAN</td>
<td>8765</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
Go to Page 4 of the ip-services form, and enter the following values:

- **AE Services Server**: Name obtained from the AE Services server, in this case *aesserver*
- **Password**: Enter a password to be administered on the AE Services server
- **Enabled**: Set to *y*

**Note:** The name and password entered for the **AE Services Server** and **Password** fields must match the name and password on the AE Services server in **Section 6.2**. The administered name for the AE Services server is created as part of the AE Services installation, and can be obtained from the AE Services server by typing `uname –n` at the Linux command prompt.

<table>
<thead>
<tr>
<th>change ip-services</th>
<th>AE Services Administration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Server ID</td>
<td>AE Services Server</td>
</tr>
<tr>
<td>1:</td>
<td>aesserver</td>
</tr>
<tr>
<td>2:</td>
<td></td>
</tr>
</tbody>
</table>

### 4.7 Configure CTI Link for TSAPI Service

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

<table>
<thead>
<tr>
<th>add cti-link 11</th>
<th>CTI LINK</th>
</tr>
</thead>
<tbody>
<tr>
<td>CTI Link: 11</td>
<td></td>
</tr>
<tr>
<td>Extension: 720</td>
<td></td>
</tr>
<tr>
<td>Type: ADJ-IP</td>
<td></td>
</tr>
<tr>
<td>Name: aesserver</td>
<td></td>
</tr>
<tr>
<td>COR: 1</td>
<td></td>
</tr>
</tbody>
</table>
4.8 Configure Avaya Voice Portal H.323 Stations

For these Application Notes, H.323 stations will provide the integration between Communication Manager and Voice Portal. A call to these stations will be routed to Voice Portal which will run a VXML script from the Sabio CallBack server. Use the command add station n. In the station form, set the Type to 7434ND, set Port to IP and provide a descriptive Name. Specify a Security Code, which will be used in Section 5.2 when configuring Voice Portal and set the Display Module and IP SoftPhone fields to y.

```
add station 31020

Extension: 31020
Type: 7434ND
Port: IP
Name: VoicePortal

STATION OPTIONS

Display Module? y
Display Language: english
Survivable COR: internal
Survivable Trunk Dest? y

Time of Day Lock Table:
Loss Group: 2
Data Module? n
Personalized Ringing Pattern: 1
Coverage Module? n
Media Complex Ext: 31020

On Page 2 set MultiMedia Mode to enhanced
```

```
add station 31020

STATION

FEATUR e OPTIONS

LWC Reception: spe
LWC Activation? y
Auto Select Any Idle Appearance? n
Coverage Msg Retrieval? y
Auto Answer: none
Data Restriction? n
Idle Appearance Preference? n
Bridged Idle Line Preference? n
Restrict Last Appearance? y
Active Station Ringing: single

H.320 Conversion? n
Service Link Mode: as-needed
Multimedia Mode: enhanced
EC500 State: enabled

MWI Served User Type:
AUDIX Name:
Display Client Redirection? n
Select Last Used Appearance? n
Coverage After Forwarding? s
Remote Softphone Emergency Calls: as-on-local
Emergency Location Ext: 31020
Always Use? n
IP Audio Hairpinning? y
```

On Page 2 set MultiMedia Mode to enhanced.
On Page 6 add a normal button to the station

```
add station 31020
```

DISPLAY BUTTON ASSIGNMENTS

1: normal
2: 

Repeat the above steps for each Voice Portal station. In this configuration, ten Voice Portal stations were configured with an extension range of 31020-31029.

### 4.9 Configure Avaya Voice Portal Hunt Group

To access Voice Portal from a Vector, a hunt group is used to deliver calls to agents that are logged into the stations configured in Section 4.8. To add a hunt group, use the command `add hunt-group n`. Set the ACD field to y to allow the hunt group to be assigned to agents. Enter a descriptive name for Group Name, set Group Extension to an available extension number and set Group Type to ucd-mia. Set the Vector field to y to allow the hunt group to be vector controlled.

```
add hunt-group 90
```

```
Group Number: 90
ACD? y
Group Name: VoicePortalHG
Queue? n
Group Extension: 490
Vector? y
Group Type: ucd-mia
TN: 1
COR: 1
M&M Early Answer? n
Security Code: 
Local Agent Preference? n
ISDN/SIP Caller Display: grp-name
```

On Page 2, set the Skill and AAS fields to y. AAS will allow the agents to automatically log into the stations configured in Section 4.8 with this hunt group

```
add hunt-group 90
```

```
Skill? y
Expected Call Handling Time (sec): 180
AAS? y
Measured: none
Supervisor Extension:
```
4.10 Configure Avaya Voice Portal Agents

To add an agent login ID, use the command `add agent-loginID n`. Enter a descriptive name for Name and set the AAS field to y to allow agent to automatically log in to the station defined in the Port Extension field. The Port Extension field should be set to one of the stations configured in Section 4.8, each Voice Portal agent should be assigned to a different station.

```
add agent-loginID 34020
```

On Page 2 assign a skill to the agent by entering the hunt group configured in Section 4.9 for SN and entering a skill level of 1 for SL.

```
add agent-loginID 34020
```

4.11 Configure Sabio CallBack CTI stations

Sabio CallBack uses CTI stations via the AE Services to initiate calls on Communication Manager, the CTI stations will be used to place calls to customers after a CallBack has been scheduled and to place calls to agents in order to reserve an agent to handle the customer callback. Use the command add station n. Enter a descriptive name for Name, set the Type field to CTI, enter a Security Code that Sabio CallBack will use to login as the station and enter X for the Port

```
add station 31030
```

Repeat the above steps for each Sabio CallBack CTI station. In this configuration, only three CTI stations were configured with an extension range of 31030-31032.
4.12 Configure Inbound Hunt Group

The call flows used with Sabio CallBack require two skill groups, this Skill group will be used for handling inbound calls when a Callback is either not offered or is not accepted. To configure the inbound skill group, run the command `add hunt-group n`. Set the ACD, Queue and Vector field to y. Enter a descriptive name for Group Name, set Group Extension to an available extension number and set Group Type to ucd-mia.

<table>
<thead>
<tr>
<th>Group Number: 53</th>
<th>ACD? y</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Name: SabioInboundSkill</td>
<td>Queue? y</td>
</tr>
<tr>
<td>Group Extension: 453</td>
<td>Vector? y</td>
</tr>
<tr>
<td>Group Type: ucd-mia</td>
<td></td>
</tr>
<tr>
<td>TN: 1</td>
<td></td>
</tr>
<tr>
<td>COR: 1</td>
<td></td>
</tr>
<tr>
<td>MM Early Answer? n</td>
<td></td>
</tr>
<tr>
<td>Security Code:</td>
<td></td>
</tr>
<tr>
<td>Local Agent Preference? n</td>
<td></td>
</tr>
<tr>
<td>ISDN/SIP Caller Display:</td>
<td></td>
</tr>
</tbody>
</table>

On Page 2 set the Skill field to y.

<table>
<thead>
<tr>
<th>Group Number: 53</th>
<th>Skill? y</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAS? n</td>
<td></td>
</tr>
<tr>
<td>Measured: none</td>
<td></td>
</tr>
<tr>
<td>Supervisor Extension:</td>
<td></td>
</tr>
</tbody>
</table>

Expected Call Handling Time (sec): 180
4.13 Configure CallBack Hunt Group

This Skill group will be used to reserve an agent to handle a CallBack. To configure the CallBack skill group run the command add hunt-group n. Set the ACD, Queue and Vector field to y. Enter a descriptive name for Group Name, set Group Extension to an available extension number and set Group Type to ucd-mia.

```
add hunt-group 54

HUNT GROUP

Group Number: 54  ACD? y
Group Name: SabioCallBackSkill  Queue? y
Group Extension: 454  Vector? y
Group Type: ucd-mia
TN: 1
COR: 1  MMM Early Answer? n
Security Code:  Local Agent Preference? n
ISDN/SIP Caller Display:
```

On Page 2 set the Skill field to y.

```
add hunt-group 54

HUNT GROUP

Skill? y  Expected Call Handling Time (sec): 180
AAS? n
Measured: none
Supervisor Extension:
```

4.14 Configure Vectors and VDNs

Four sets of Vectors and VDNs are required for Sabio CallBack. Following is a summary of each vector and VDN.

- **Phase 1**: The phase one vector and VDN is the first call routing point that the customer will hit, the customer is queued to the inbound skill group and depending on call centre conditions the customer will be offered a Call back. If the call back is accepted then the call is routed to Phase 2.
- **Phase 2**: The phase two vector and VDN route the customer to Voice Portal passing new variables so that the callback can be scheduled.
- **CallBack**: The CallBack Vector and VDN are used by Sabio CallBack to reserve an agent to handle the customer CallBack.
- **Voice Portal Access**: The Voice Portal Access vector and VDN are used by Sabio CallBack to access Voice Portal

The configuration for each set of vector and VDN is covered in more detail in the following sections.
4.14.1 Configure Phase 1 Vector

Use the **change vector n** command to configure the vector that will be used with the phase 1 VDN. Shown below is the Phase 1 Vector that was used for the interoperability test. To better understand what the vector is doing a brief explanation for some of the vector steps follows:

- **Line 02** is queuing the call to the inbound skill group **53** configured in **Section 4.12**
- **Line 03** is passing the variable V1 to Voice Portal by conversing on the Voice Portal skill **90** which was configured in **Section 4.9**
- **Line 04** will collect any digits entered by the customer, Sabio CallBack prompts the customer to press 1 if they wish to schedule a callback
- **Line 05** moves the call to line **13** within the vector if the customer enters **1**. If the customer has not entered **1** then the call will continue to queue for an agent in skill **53**
- **Line 13** if the call reaches this line then the customer has accepted the call back offer and is sent to the phase 2 VDN for further processing

**Note:** This is a sample vector, it is possible to provide additional call treatment within the vector such as queue announcements and time of day routing, please see reference [2] for further information.

```
change vector 53

CALL VECTOR

Number: 53
Name: SabioPhase1
Meet-me Conf? n
Lock? n

Basic? y
EAS? y
G3V4 Enhanced? y
ANI/II-Digits? n
ASAI Routing? y

Prompting? y
LAI? n
G3V4 Adv Route? y
CINFO? n
BSR? n
Holidays? n

Variables? y
3.0 Enhanced? y

01 wait-time 2 secs hearing ringback
02 queue-to skill 53 pri m
03 converse-on skill 90 pri m passing V1 and wait
04 collect 1 digits after announcement none for none
05 goto step 13 if digits = 1
06 wait-time 10 secs hearing ringback
07 converse-on skill 90 pri m passing V1 and 10
08 collect 1 digits after announcement none for none
09 goto step 13 if digits = 1
10 wait-time 10 secs hearing ringback
11 goto step 3 if unconditionally
12 stop
13 route-to number 71054 with cov n if unconditionally
14 stop
```
4.14.2 Configure Phase 1 VDN

Use the command `add vdn n`. Enter an available extension number for **Extension**, enter a descriptive name for **Name** and enter vector number **53** that was configured in the previous section as **Destination : Vector Number**. Set **Allow VDN override** to **y**, this will allow the phase 2 VDN to become the active VDN extension if the call back offer is accepted.

```
add vdn 71053

VECTOR DIRECTORY NUMBER

Extension: 71053
Name*: Sabio Phase 1
Destination: Vector Number 53

Meet-me Conferencing? n
Allow VDN Override? y
COR: 1
TN*: 1
Measured: internal

1st Skill*: 
2nd Skill*: 
3rd Skill*: 
```

On **Page 3** configure the variable that is used by line **03** of the vector configured in **Section 4.14.1**. For the **V1** variable enter a descriptive name for **Description** and enter the VDN extension number for **Assignment**.

```
add vdn 71053

VECTOR DIRECTORY NUMBER

VDN VARIABLES*

<table>
<thead>
<tr>
<th>Var</th>
<th>Description</th>
<th>Assignment</th>
</tr>
</thead>
<tbody>
<tr>
<td>V1</td>
<td>VDN Number</td>
<td>71053</td>
</tr>
<tr>
<td>V2</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
4.14.3 Configure Phase 2 Vector

Use the change vector n command to configure the vector that will be used with the phase 2 VDN. Shown below is the Phase 2 Vector that was used for the interoperability test. To better understand what the vector is doing a brief explanation for some of the vector steps follows:

- **Line 02** is passing the variables V1 and A to Voice Portal by conversing on the Voice portal skill 90 which was configured in Section 4.9. The presence of variable A indicates to Sabio Callback that the call back offer has been accepted and the script to schedule the call back should be invoked.
- **Line 03** will collect a digit returned by Sabio Callback via Voice Portal.
- **Line 04** moves the call to line 9 within the vector if Sabio Callback returns 1. If Sabio Callback does not return a 1 then the Call Back was unsuccessful and the call will continue to the next line in the vector, line 05.
- **Line 05** will queue the call for an agent in the inbound skill 53.
- **Line 09** will disconnect the call after the caller has successfully scheduled a call back.

<table>
<thead>
<tr>
<th>change vector 54</th>
<th>CALL VECTOR</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number: 54</td>
<td>Name: SabioPhase2</td>
</tr>
<tr>
<td>Meet-me Conf? n</td>
<td>Lock? n</td>
</tr>
<tr>
<td>Basic? y</td>
<td>EAS? y</td>
</tr>
<tr>
<td>G3V4 Enhanced? y</td>
<td>ANI/II-Digits? y</td>
</tr>
<tr>
<td>Prompting? y</td>
<td>LAI? n</td>
</tr>
<tr>
<td>G3V4 Adv Route? y</td>
<td>INFO? n</td>
</tr>
<tr>
<td>Variables? y</td>
<td>BSR? n</td>
</tr>
<tr>
<td>01 wait-time</td>
<td>Holidays? n</td>
</tr>
<tr>
<td>02 converse-on</td>
<td>1 secs hearing silence</td>
</tr>
<tr>
<td>03 collect</td>
<td>digits after announcement none for none</td>
</tr>
<tr>
<td>04 goto step</td>
<td>if digits = 1</td>
</tr>
<tr>
<td>05 queue-to</td>
<td>skill 53 pri m</td>
</tr>
<tr>
<td>06 wait-time</td>
<td>10 secs hearing ringback</td>
</tr>
<tr>
<td>07 goto step</td>
<td>6 if unconditionally</td>
</tr>
<tr>
<td>08 stop</td>
<td></td>
</tr>
<tr>
<td>09 disconnect</td>
<td>after announcement none</td>
</tr>
<tr>
<td>10 stop</td>
<td></td>
</tr>
</tbody>
</table>

4.14.4 Configure Phase 2 VDN

Use the command add vdn n. Enter an available extension number for Extension, enter a descriptive name for Name and enter vector number 54 that was configured in the previous section as the Destination: Vector Number.

<table>
<thead>
<tr>
<th>add vdn 71054</th>
<th>VECTOR DIRECTORY NUMBER</th>
</tr>
</thead>
<tbody>
<tr>
<td>Extension:</td>
<td>71054</td>
</tr>
<tr>
<td>Name*:</td>
<td>Sabio Phase 2</td>
</tr>
<tr>
<td>Destination:</td>
<td>Vector Number 54</td>
</tr>
<tr>
<td>Meet-me Conferencing? n</td>
<td></td>
</tr>
<tr>
<td>Allow VDN Override? n</td>
<td>1</td>
</tr>
<tr>
<td>COR:</td>
<td>1</td>
</tr>
<tr>
<td>TN*:</td>
<td>1</td>
</tr>
<tr>
<td>Measured:</td>
<td>internal</td>
</tr>
</tbody>
</table>
On Page 3, configure the first variable that is used by line 02 of the vector configured in Section 4.14.3. For the V1 variable enter a descriptive name for Description and enter the VDN extension number for Assignment.

<table>
<thead>
<tr>
<th>Var</th>
<th>Description</th>
<th>Assignment</th>
</tr>
</thead>
<tbody>
<tr>
<td>V1</td>
<td>VDN Number</td>
<td>71054</td>
</tr>
<tr>
<td>V2</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

4.14.5 Configure Phase 2 Variable

To configure the second variable, variable A, that is used by line 02 of the vector configured in Section 4.14.3 run the command change variables. Enter a descriptive name for Var A in Description and set the Type to collect. Set the Scope field to G meaning that variable A is a global variable. The Length, Start and Assignment parameters should be agreed with Sabio. The screen shot below shows the values used during interoperability testing.

4.14.6 Configure Sabio CallBack Vector

This vector is used by Sabio CallBack to reserve agents to handle a customer call back by placing a call in queue for skill 54. Use the change vector n command to configure the vector that will be used with the CallBack VDN. Shown below is the CallBack Vector that was used for the interoperability test.

<table>
<thead>
<tr>
<th>Number: 55</th>
<th>Name: Sabio CB Out</th>
</tr>
</thead>
<tbody>
<tr>
<td>Meet-me Conf?</td>
<td>n</td>
</tr>
<tr>
<td>Lock?</td>
<td>n</td>
</tr>
<tr>
<td>Basic?</td>
<td>y</td>
</tr>
<tr>
<td>EAS?</td>
<td>y</td>
</tr>
<tr>
<td>G3V4 Enhanced?</td>
<td>y</td>
</tr>
<tr>
<td>ANI/II-Digits?</td>
<td>n</td>
</tr>
<tr>
<td>ASAI Routing?</td>
<td>y</td>
</tr>
<tr>
<td>Prompting?</td>
<td>y</td>
</tr>
<tr>
<td>LAI?</td>
<td>n</td>
</tr>
<tr>
<td>G3V4 Adv Route?</td>
<td>y</td>
</tr>
<tr>
<td>CINFO?</td>
<td>n</td>
</tr>
<tr>
<td>BSR?</td>
<td>n</td>
</tr>
<tr>
<td>Holidays?</td>
<td>n</td>
</tr>
<tr>
<td>Variables?</td>
<td>y</td>
</tr>
<tr>
<td>01 wait-time</td>
<td>5 secs hearing silence</td>
</tr>
<tr>
<td>02 queue-to</td>
<td>skill 54 pri 1</td>
</tr>
<tr>
<td>03 wait-time</td>
<td>60 secs hearing ringback</td>
</tr>
<tr>
<td>04 goto step</td>
<td>3 if unconditionally</td>
</tr>
<tr>
<td>05 stop</td>
<td></td>
</tr>
</tbody>
</table>

change vdn 71054

change variables

change vector 55

The screen shot below shows the values used during interoperability testing.

change vector 55

change variables
4.14.7 Configure CallBack VDN

Use the command **add vdn n**. Enter an available extension number for **Extension**, enter a descriptive name for **Name** and enter vector number **55** that was configured in the previous Section 4.14.6 as **Destination : Vector Number**.

```
add vdn 71055

VECTOR DIRECTORY NUMBER

  Extension: 71055
  Name*: SabioCallBack OUT
  Destination: Vector Number 55

  Meet-me Conferencing? n
  Allow VDN Override? n
  COR: 1
  TN*: 1
  Measured: internal

  1st Skill*:
  2nd Skill*:
  3rd Skill*:
```

4.14.8 Configure Avaya Voice Portal Access Vector

This vector is used by Sabio CallBack to access Voice Portal by placing a call in queue for skill **90**. Use the change vector n command to configure the vector that will be used with the Voice Portal Access VDN. Shown below is the Voice Portal Access Vector that was used for the interoperability test.

```
change vector 56

CALL VECTOR

  Number: 56
  Name: AVP access

  Meet-me Conf? n
  Lock? n
  Basic? y
  EAS? y
  G3V4 Enhanced? y
  ANI/II-Digits? n
  ASAI Routing? y
  Prompting? y
  LAI? n
  G3V4 Adv Route? y
  CINFO? n
  BSR? n
  Holidays? n
  Variables? y
  3.0 Enhanced? y

01 wait-time 1 secs hearing silence
02 queue-to skill 90 pri m
03 wait-time 5 secs hearing silence
04 stop
05
06
07
08
09
10
11
12
```

Use the command `add vdn n`. Enter an available extension number for `Extension`, enter a descriptive name for `Name` and enter vector number 56 that was configured in the previous Section 4.14.8 as `Destination : Vector Number`.

<table>
<thead>
<tr>
<th>add vdn 71056</th>
</tr>
</thead>
<tbody>
<tr>
<td>VECTOR DIRECTORY NUMBER</td>
</tr>
<tr>
<td>Extension: 71056</td>
</tr>
<tr>
<td>Name*: Sabio VoicePortal Access</td>
</tr>
<tr>
<td>Destination: Vector Number 56</td>
</tr>
<tr>
<td>Meet-me Conferencing? n</td>
</tr>
<tr>
<td>Allow VDN Override? n</td>
</tr>
<tr>
<td>COR: 1</td>
</tr>
<tr>
<td>TN*: 1</td>
</tr>
<tr>
<td>Measured: none</td>
</tr>
<tr>
<td>1st Skill*:</td>
</tr>
<tr>
<td>2nd Skill*:</td>
</tr>
<tr>
<td>3rd Skill*:</td>
</tr>
</tbody>
</table>
5 Configure Avaya Voice Portal
This section covers the administration of Voice Portal. Voice Portal is configured via an Internet browser using Voice Portal Management System (VPMS) web interface. It is assumed that Voice Portal software and the license file have already been installed. In this configuration, Voice Portal is connected to the Communication Manager via H.323.

5.1 Logging in to Avaya Voice Portal
Voice Portal is configured via the VPMS web interface. To access the web interface enter <http://<ip-addr>/VoicePortal> as the URL in an Internet browser, where <ip-addr> is the IP address of the VPMS. The login screen is displayed, log in with the appropriate Administrator user credentials.
5.2 Configuring H.323 Connection for Avaya Aura™ Communication Manager

To configure the H.323 connection for Communication Manager, navigate to the VoIP Connections → Add H.323 Connection. In the Add H.323 Connection screen, specify a Name and enter the IP address of the CLAN interface in the Gatekeeper Address field. Set the Station From, To and Password fields according to the stations configured in Section 4.8. Highlight Inbound and Outbound for Station Type. Accept the default values for the other fields and click the Add button.
5.3 Add Applications

Sabio CallBack requires two applications to be added to Voice Portal. Navigate to System Configuration → Applications and for the first application. On the Change Applications page, specify a Name for the application, set the MIME Type field to VoiceXML, and set the VoiceXML URL field to a URL provided by Sabio that will point to an application on the Sabio Callback server. Next, the Phase 1 and Phase 2 VDN number configured in Section 4.14.2 and 4.14.4 are entered into the Called Number field, click the Add button to enter each number. Click on Save once completed (not shown). The screen shot below shows the application after it has been configured.
For the second application, on the Change Applications page, specify a Name for the application, set the MIME Type field to VoiceXML, and set the VoiceXML URL field to a URL provided by Sabio that will point to an application on the Sabio Callback server. Next, the Voice Portal Access VDN number configured in Section 4.14.9 is entered into the Called Number field, click the Add button to enter the number. Click on Save once completed (not shown). The screen shot below shows the application after it has been configured.
6 Configure Avaya Aura™ Application Enablement Services

This section covers the administration of AE Services (Application Enablement Services). AE Services is configured via an Internet browser using the Administration web interface. It is assumed that AE Services software and the license file have already been installed.

6.1 Logging in to Avaya Aura™ Application Enablement Services

To access the administration web interface, enter https://<ip-addr>/MVAP as the URL in an Internet browser, where <ip-addr> is the active IP address of AE Services. The login screen is displayed, log in with the appropriate credentials and then select the Login button.
6.2 Add Switch Connection

From the left pane of the Administration web interface, navigate to Administration → Switch Connections. Enter a name for the switch connection to be added and select the Add Connection button.

In the resulting screen, enter and confirm the Switch Password. This must match the password configured in Section 4.6. When finished select the Apply button.

Back in the Switch Connections screen select the radio button for the recently added switch connection and select the Edit CLAN IPs button (not shown). In the resulting screen enter the IP address of the CLAN that will be used for the AE Services connection and select the Add Name or IP button.
6.3 Add TSAPI Link

From the left pane of the Administration web interface, navigate to Administration → CTI Link Admin → TSAPI Links. For Link select the next available link number using the drop down menu. For the Switch Connection field select the switch connection defined in Section 6.2. The Switch CTI Link Number must match the CTI configured number in Section 4.7. Ensure that the ASAI Link Version field is set to 4. Set the Security field to Both to create a secure and a non-secure TSAPI link. When all the values have been set, select the Apply Changes button.

![Add / Edit TSAPI Links](image1)

Once the TSAPI link has been added navigate to Administration → Security Database → Tlinks to view the Tlink Name. The secure connection is the second link in the screen below and is denoted by the characters CSTA-S in the Tlink Name.

![Add / Edit TSAPI Links](image2)
6.4 Add TSAPI User

From the left pane of the Administration web interface, navigate to **User Management → Add User**. From the **Add User** screen enter values for all of the compulsory fields marked with *. The **User ID** and **User Password** are used in the configuration of Sabio CallBack. In addition to the compulsory fields the **CT User** field should be set to **Yes**. When complete select the **Apply** button (not shown).
7 Configure Sabio CallBack

This section covers the administration of Sabio CallBack. Sabio CallBack is configured via an Internet browser using the Administration web interface. It is assumed that Sabio CallBack software and the license file have already been installed. For additional information on installation tasks please contact Sabio using the details in Section 1.2

7.1 Logging in to Sabio CallBack

To access the administration web interface, enter http://<ip-addr>/SIC/ as the URL in an Internet browser, where <ip-addr> is the active IP address of Sabio CallBack. The login screen is displayed, log in with the appropriate credentials and then select the Login button.
7.2 Configure System Properties

From the left pane of the Administration web interface, select the System Properties option. In the System Properties screen the following fields should be configured:

- **Callback_number_prefix** set this field to the ARS Feature access code assigned in Section 4.3.
- **JTSapiPassword** set this field to the password that was configured for the TSAPI user in Section 6.4
- **JTSapiUser** set this field to the TSAPI username configured in Section 6.4
- **TLink** Set this field to the TLINK configured in Section 6.3

**Note:** For brevity, some fields have been omitted from the following screen shots. The fields configured in this section are the fields that were required for the interoperability test. For information on configuring any other fields not covered in these Application Notes please refer to the Sabio Support details in Section 1.2.
7.3 Configure CallBack Ports

From the left pane of the Administration web interface, select the CallBack Ports option. In the CallBack Ports screen under the Enter new CallBack Port Details heading, enter the VDN number assigned in Section 4.14.9 as the IVRPort. For the XPort field, enter an extension for one of the CTI stations Configured in Section 4.11. Set the Active field to true to activate the port and then select the Submit button. Ports that have been configured are shown at the top of the page.
7.4 Configure Banned Numbers

Numbers that are configured as a banned numbers are prevented from scheduling a call back. Banned numbers can be entered as individual numbers or a number pattern match using regular expression. An example is the pattern of 0208123101[0-9]{1} which will match on all calls having digits beginning with 0208123101 and one additional digit. To configure a Banned Number from the left pane of the Administration web interface, select the CallBack Banned Numbers option. In the Banned Numbers screen under the heading Enter new Banned Number/Regex below if required, enter a number or regular expression pattern, and then select the Submit button. Banned numbers that have been configured will then appear above.
7.5 Configure CallBack Schedule

From the left pane of the Administration web interface, select the CallBack Schedules option. In the CallBack Schedule Manager screen under the Create a new Schedule below heading, enter a name for the schedule to create, and then select the Submit button.
The **CallBack Schedule Details** screen is displayed. In the **CallBack Schedule Details** screen, under the heading **Create new Schedule entry** use the drop down menu to select the **Type** of schedule entry to be configured, select **Offer** to configure an entry that specifies a time period when call backs will be offered to inbound callers and select **Dialing** to configure an entry that specifies a time period when a caller requested call back can be dialed. The schedules are defined by day of the week, use the drop down menu to select the **Day Of Week** and specify a start and stop time in the **Start Time** and **End Time** fields respectively. Click the **Create** button and the schedule entry will be displayed under the **Entries** heading. Once all of the required schedule entries are configured select the **Update** button towards the top of the screen to save the changes.
7.6 Configure CallBack Contact Strategy

CallBack strategies are used to define the frequency and occurrence of re-tries following a failed attempt at contact i.e. the customer is unavailable. From the left pane of the Administration web interface, select the **CallBack Contact Strategies** option. In the **CallBack Contact Strategy Manager** screen under the **Create a new Contact Strategy below** heading, enter a name for the strategy to create, and then select the **Submit** button.
The **CallBack Contact Strategy Details** screen is displayed. In the **CallBack Contact Strategy Details** screen, under the heading **Create new contact strategy entry** enter a delay time in minutes, this will be the amount of time before a call back is retried. Click the **Create** button and the delay entry will be displayed under the **Entries** heading. The **Sequence** column displays the order that each delay will be used. Once all of the required delay entries are configured select the **Update** button towards the top of the screen to save the changes.
7.7 Add CallBack Configuration

A CallBack configuration defines how a call back will behave. From the left pane of the Administration web interface, select the CallBack Configuration option. In the resulting CallBack Service Manager screen under the heading Create a new CallBack Service below, enter a name for the CallBack Configuration to create, and then select the Submit button.

After creating the CallBack Configuration at least one language must be defined. A CallBack Configuration can have multiple languages configured within it, each language can be individually configured to provide different behavior. The following values must administered:

- **Def Queue** is set to the VDN number used to reserve an agent for a CallBack, configured in Section 4.14.7.
- **Initial Delay** is set to the amount of time in seconds the system will wait before attempting to reserve an agent for a CallBack.
- **Min Number Length** is set to the minimum length a callers number must be in order to be valid.
- **Max Number Length** is set to the maximum length a callers number must be in order to be valid.
- **Number Prefix** is a number that will prefix to outbound calls, for the interoperability test this was set to the ARS FAC defined in Section 4.3.
- **Max Concurrent Callbacks** is the maximum number of active scheduled call backs allowed at any one time, once this number is exceeded callback will no longer be offered/accepted.
- **Queued Max Concurrent Callbacks** is the number of calls that can be queued for an agent at any one time.
- **Data Capture Script** is used to define a script that can capture additional information from the caller beyond the telephone number.
- **Schedule** is set to the CallBack Schedule defined in Section 7.5.
- **Agent Acceptance Required** is used to activate the functionality that provides the ability for agents to pre-screen the call back and choose if to accept or reject it.
- **Agent AutoDial Delay** is only relevant when Agent Acceptance Required is activated. This field used to define a time limit in seconds that will force the agent to accept the callback if no response has been received from the agent. A value of 0 will disable this
timeout limit.

- **Auto CLI** will, if set to **yes**, recognize and read back the callers CLI when a call back is requested. The caller then has the option to accept the call back on the presented CLI or enter an alternative number. If set to **no**, the caller will be prompted to enter a number regardless of any presented CLI.

- **Number Confirmation** will, if set to **yes**, read back the caller entered number for confirmation before accepting the call back. If set to **no**, the caller entered number will be accepted without confirmation.

- **Contact Strategy** is set to the CallBack Strategy defined in **Section 7.6**.

- **Timezone** is set to the timezone to be used for the combination of CallBack Configuration/Language being configured.

All other fields can be left with their default values. Once all fields have been configured, select the **create** button (not shown) to save the changes. It is not possible to display all the values configured in a screenshot, as such, the screenshot below shows only some example values for illustration purposes.
7.8 Configure Number Mapping

From the left pane of the Administration web interface, select the CallBack Number Mapping option. In the CallBack Number Mapping screen under the Create a new Number Mapping entry heading, use the drop down menu to select the CallBack Definition that the Number mapping is being added for, configured in Section 7.7. Next select the CallBack Configuration language to be used. For the Inbound Number field, enter either the Phase 1 VDN configured in Section 4.14.2 or the Phase 2 VDN configured in Section 4.14.4. For the Outbound Number field enter the Voice Portal Access VDN configured Section 4.14.9. The Handback Type will depend on the number being entered in the Inbound Number field, if the Phase 1 VDN is being configured then Converse Multi Stage Stage 1 should be selected if the Phase 2 VDN is being configured then Converse Multi Stage Stage 2 should be selected. Select the Create button and the number mapping entry will be displayed under the Entries heading. The screen shot below was taken after the number mapping used for testing was configured and the values used can be seen under the Entries heading.

8 General Test Approach and Test Results

This section describes the interoperability testing used to verify Sabio CallBack Solution. The interoperability testing included feature and serviceability testing. The feature testing focused on verifying the following:

- Access to Sabio CallBack from call vector
- CLI recognition and confirmation
- Number prompt when no CLI present
- Numbers barred from requesting a call back rejected
- Invalid number formats rejected
- Maximum number of call back attempts
- Defined schedule allows/ prevents call backs respectively
- Call queuing scenarios such as multiple calls, agents busy, agents logged off, etc.
- Call back failures including, busy, unobtainable and unanswered calls
• Additional data capture when call back is accepted
• Agent whisper replay of data capture
• Agent accept and reject of call back

The serviceability testing focused on verifying the ability of Sabio Callback to recover from adverse conditions, such as power failures and disconnecting cables from the IP network.

9 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, AE Services, Voice Portal and Sabio CallBack.

9.1 Verify Avaya Aura™ Communication Manager

Verify the status of the administered CTI link by using the `status aesvcs cti-link` command. The Service State should show as `established`.

```
status aesvcs cti-link

AE SERVICES CTI LINK STATUS

<table>
<thead>
<tr>
<th>CTI Link</th>
<th>Version</th>
<th>Mnt</th>
<th>AE Services</th>
<th>Service</th>
<th>Msgs</th>
<th>Msgs</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

```

Verify the status of the Voice Portal agents by using the `status station n` command, where `n` is the agent login ID of a Voice Portal agent. Go to Page 4 and confirm that Grp/Mod displays 90/AI indicating that the agent is in an auto-in state for skill 90.

```
status station 34020

ACD STATUS

<table>
<thead>
<tr>
<th>Grp/Mod</th>
<th>Grp/Mod</th>
<th>Grp/Mod</th>
<th>Grp/Mod</th>
<th>Grp/Mod</th>
</tr>
</thead>
<tbody>
<tr>
<td>90/AI</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
</tr>
<tr>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
<td>/</td>
</tr>
</tbody>
</table>

On ACD Call? no

Occupancy: 75.0
```

9.2 Verify Avaya Aura™ Application Enablement Services

From the CTI OAM Administration menu, verify the status of the administered CTI link by selecting Status and Control ➔ Switch Conn Summary. The Conn State should show Talking.
9.3 Verify Avaya Voice Portal

From the VPMS web interface, click **System Management → MPP Manager**. On the MPP Manager page, verify that the MPP server is **Online** and **Running**.

![MPP Manager Page]

Place a call to Voice Portal by dialing the Phase 1 VDN. Verify that the Sabio CallBack application answers the call and that the application is able to recognize the callers CLI.

9.4 Verify Sabio CallBack

From the Sabio CallBack web interface verify the status of the Sabio CallBack by clicking **CallBack live Manager** option in the left pane. Confirm that all the port IDs show **true** underneath the **Active** column.
10 Conclusion
These Application Notes describe the configuration steps required to use Sabio CallBack with Avaya Aura™ Communication Manager, Avaya Aura™ Application Enablement Services and Avaya Voice Portal. All functionality and serviceability test cases were completed successfully.

11 Additional References
This section references the Avaya and Sabio product documentation that are relevant to these Application Notes.
Product documentation for Avaya products may be found at http://support.avaya.com
1. Administering Avaya Aura™ Communication Manager, Document No. 03-300509, May 2009
2. Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Document No. 07-600780, April 2009
4. CN3915 Avaya Voice Portal Date: 11/09 Rev: A Intg Type: (H.323 EAS) Software Application, Jan 2009

Documentation for Sabio products may be requested from Sabio at http://www.sabio.co.uk