



Avaya Solution & Interoperability Test Lab

Application Notes for Sabio CallBack with Avaya Voice Portal, Avaya Aura™ Application Enablement Services and Avaya Aura™ Communication Manager – Issue 1.0

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.

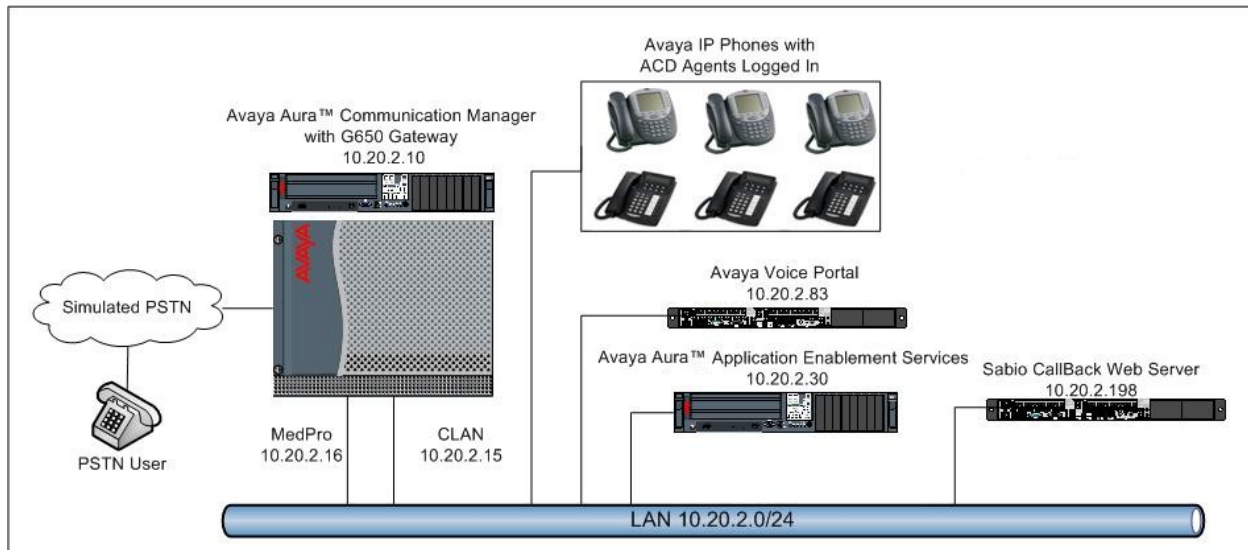


Figure 1: Network Topology

3 Equipment and Software Validated

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

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

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**.

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

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

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

On **Page 9** confirm **Adjunct Routing**, **CTI Stations**, **Phantom Calls** and **Agent States** are set to **y**.

```
display system-parameters customer-options                               Page 9 of 11
      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                                     Page 11 of 18
      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		Page	1 of 8
FEATURE ACCESS CODE (FAC)			
Abbreviated Dialing List1 Access Code:			
Abbreviated Dialing List2 Access Code:			
Abbreviated Dialing List3 Access Code:			
Abbreviated Dial - Prgm Group List Access Code:			
Announcement Access Code: *56			
Answer Back Access Code: *59			
Attendant Access Code:			
Auto Alternate Routing (AAR) Access Code:			
Auto Route Selection (ARS) - Access Code 1: 9		Access Code 2:	
Automatic Callback Activation:		Deactivation:	
Call Forwarding Activation Busy/DA: All:		Deactivation:	
Call Forwarding Enhanced Status: Act:		Deactivation:	

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		Page	5 of 8
FEATURE ACCESS CODE (FAC)			
Automatic Call Distribution Features			
After Call Work Access Code:			
Assist Access Code:			
Auto-In Access Code: *27			
Aux Work Access Code: *28			
Login Access Code: *25			
Logout Access Code: *26			
Manual-in Access Code: *29			

On **Page 6** define a **Converse Data Return Code**. This is required to allow Sabio CallBack to return data to the Communication Manager.

```
change feature-access-codes                                     Page 6 of 8
                                FEATURE ACCESS CODE (FAC)

                                Call Vectoring/Prompting Features

    Converse Data Return Code: *12

Vector Variable 1 (VV1) Code:
Vector Variable 2 (VV2) Code:
Vector Variable 3 (VV3) Code:
Vector Variable 4 (VV4) Code:
Vector Variable 5 (VV5) Code:
```

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.

```
change node-names ip                                         Page 1 of 2
                                IP NODE NAMES

    Name                      IP Address
CLAN                        10.20.2.15
Gateway                    10.20.2.1
MEDPRO                       10.20.2.16
PC4.1                        10.20.2.60
RDTT                         10.20.2.41
SiteB                        10.10.15.13
VPCLAN                       10.20.2.18
aesserver                  10.20.2.30
announce                     10.20.2.17
```

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		Page 1 of 3	
IP INTERFACES			
Type:	C-LAN		
Slot:	01A02	Target socket load and Warning level: 400	
Code/Suffix:	TN799 D	Receive Buffer TCP Window Size: 8320	
Enable Interface?	y	Allow H.323 Endpoints? y	
VLAN:	n	Allow H.248 Gateways? y	
Network Region:	1	Gatekeeper Priority: 5	
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		Page 1 of 4			
IP SERVICES					
Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port
AESVCS	y	CLAN	8765		

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.

change ip-services		AE Services Administration			Page 4 of 4
Server ID	AE Services Server	Password	Enabled	Status	
1:	aesserver	aeserverpw123	y	in use	
2:					

4.7 Configure CTI Link for TSAPI Service

Add a CTI link using the **add cti-link n** 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.

add cti-link 11		Page 1 of 3	
CTI Link: 11		CTI LINK	
Extension: 720			
Type: ADJ-IP			
		COR: 1	
Name: aes server			

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		Page 1 of 6
STATION		
Extension: 31020	Lock Messages? n	BCC: 0
Type: 7434ND	Security Code: 1234	TN: 1
Port: IP	Coverage Path 1:	COR: 1
Name: VoicePortal	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 2	Personalized Ringing Pattern: 1	
Data Module? n	Message Lamp Ext: 31020	
Display Module? y		
Display Language: english	Coverage Module? n	
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? y	
	IP Video Softphone? n	

On **Page 2** set **MultiMedia Mode** to **enhanced**

add station 31020		Page 2 of 6
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance? n	
LWC Activation? y	Coverage Msg Retrieval? y	
LWC Log External Calls? n	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
Redirect Notification? y	Idle Appearance Preference? n	
Per Button Ring Control? n	Bridged Idle Line Preference? n	
Bridged Call Alerting? n	Restrict Last Appearance? y	
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State: enabled	
Multimedia Mode: enhanced		
MWI Served User Type:	Display Client Redirection? n	
AUDIX Name:	Select Last Used Appearance? n	
	Coverage After Forwarding? s	
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 31020	Always Use? n IP Audio Hairpinning? y	

On **Page 6** add a **normal** button to the station

add station 31020	STATION	Page 6 of 6
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	HUNT GROUP	Page 1 of 3
Group Number: 90		
Group Name: VoicePortalHG	ACD? y	
Group Extension: 490	Queue? n	
Group Type: ucd-mia	Vector? y	
TN: 1		
COR: 1	MM 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	HUNT GROUP	Page 2 of 3
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                                     Page 1 of 2
                                AGENT LOGINID
      Login ID: 34020                                AAS? y
      Name: VoicePortal                                AUDIX? n
      TN: 1                                           LWC Reception: none
      COR: 1                                           LWC Log External Calls? n
      Coverage Path:                                AUDIX Name for Messaging:
      Security Code:
      Port Extension: 31020                        LoginID for ISDN/SIP Display? n
```

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                                     Page 2 of 2
                                AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
      Call Handling Preference: skill-level              Local Call Preference? n

      SN  RL  SL          SN  RL  SL          SN  RL  SL          SN  RL  SL
      1:  90   1          16:                31:                46:
      2:                17:                32:                47:
```

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                                           Page 1 of 5
                                STATION
      Extension: 31030                                Lock Messages? n          BCC: 0
      Type: CTI                                       Security Code: 1234        TN: 1
      Port: X                                         Coverage Path 1:          COR: 1
      Name: SabioCallBackSTN1                       Coverage Path 2:          COS: 1
                                                    Hunt-to Station:

      STATION OPTIONS
                                Time of Day Lock Table:
      Loss Group: 1                                Personalized Ringing Pattern: 1
      Data Module? n                                Message Lamp Ext: 31030
      Display Module? n
```

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**.

change hunt-group 53		Page 1 of 3
HUNT GROUP		
Group Number: 53	ACD? y	
Group Name: SabioInboundSkill	Queue? y	
Group Extension: 453	Vector? y	
Group Type: ucd-mia		
TN: 1		
COR: 1	MM Early Answer? n	
Security Code:	Local Agent Preference? n	
ISDN/SIP Caller Display:		

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

change hunt-group 53		Page 2 of 3
HUNT GROUP		
Skill? y	Expected Call Handling Time (sec): 180	
AAS? n		
Measured: none		
Supervisor Extension:		

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		Page 1 of 3
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	MM 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		Page 2 of 3
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				Page 1 of 6	
CALL VECTOR					
Number: 53		Name: SabioPhase1			
Basic? y		EAS? y		Meet-me Conf? n	Lock? n
Prompting? y		LAI? n		ANI/II-Digits? n	ASAI Routing? y
Variables? y		3.0 Enhanced? y		BSR? n	Holidays? n
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	Page 1 of 3
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	Extension: 71053 of 3	
VECTOR DIRECTORY NUMBER		
VDN VARIABLES*		
Var	Description	Assignment
V1	VDN Number	71053
V2		

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

```
change vector 54                                     Page 1 of 6
CALL VECTOR
Number: 54      Name: SabioPhase2
Basic? y      EAS? y      G3V4 Enhanced? y      Meet-me Conf? n      Lock? n
Prompting? y  LAI? n      G3V4 Adv Route? y      ANI/II-Digits? n      ASAI Routing? y
Variables? y  3.0 Enhanced? y      CINFO? n      BSR? n      Holidays? n
01 wait-time  1      secs hearing silence
02 converse-on  skill 90      pri m passing V1      and A
03 collect    1      digits after announcement none      for none
04 goto step  9      if digits      =      1
05 queue-to   skill 53      pri m
06 wait-time  10      secs hearing ringback
07 goto step  6      if unconditionally
08 stop
09 disconnect after announcement none
10 stop
```

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**.

```
add vdn 71054                                     Page 1 of 3
VECTOR DIRECTORY NUMBER
Extension: 71054
Name*: Sabio Phase 2
Destination: Vector Number      54
Meet-me Conferencing? n
Allow VDN Override? n
COR: 1
TN*: 1
Measured: internal
```

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**.

change vdn 71054			Page 3 of 3		
VECTOR DIRECTORY NUMBER					
VDN VARIABLES*					
Var	Description	Assignment			
V1	VDN Number	71054			
V2					

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.

change variables						Page 1 of 39
VARIABLES FOR VECTORS						
Var	Description	Type	Scope	Length	Start Assignment	VAC
A	CallBackPhase2Value	collect	G	6	1 999999	
B						
C						

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.

change vector 55				Page 1 of 6	
CALL VECTOR					
Number: 55		Name: Sabio CB Out			
		Meet-me Conf? n		Lock? n	
Basic? y		EAS? y		ANI/II-Digits? n	
Prompting? y		G3V4 Enhanced? y		ASAI Routing? y	
Variables? y		LAI? n		CINFO? n	
		G3V4 Adv Route? y		BSR? n	
		3.0 Enhanced? y		Holidays? n	
01 wait-time		5 secs hearing silence			
02 queue-to		skill 54 pri 1			
03 wait-time		60 secs hearing ringback			
04 goto step		3 if unconditionally			
05 stop					

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                                     Page 1 of 3
                                         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                                     Page 1 of 6
                                         CALL VECTOR

      Number: 56      Name: AVP access

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

4.14.9 Configure Avaya Voice Portal Access VDN

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**.

```
add vdn 71056                                     Page 1 of 3
                                         VECTOR DIRECTORY NUMBER

      Extension: 71056
      Name*: Sabio VoicePortal Access
      Destination: Vector Number      56

Meet-me Conferencing? n
Allow VDN Override? n
      COR: 1
      TN*: 1
      Measured: none

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

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.

The image shows the Avaya Voice Portal 4.1 login interface. At the top, the 'AVAYA' logo is displayed in red. Below it, a red horizontal bar contains the text 'Voice Portal 4.1'. The main area is white and contains a 'User Name:' label followed by a text input field. Below the input field is a 'Submit' button. At the bottom left, there is a link labeled 'Change Password'. At the bottom center, the copyright notice '© 2009 Avaya Inc. All Rights Reserved.' is visible.

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.

AVAYA Welcome
Last logged in 7/20/10 at 1:14

Voice Portal 4.1 (VoicePortal) Home Help

Expand All | Collapse All

- ▼ **User Management**
 - Users
- ▼ **System Maintenance**
 - System Monitor
 - MPP Manager
 - Active Calls
 - Port Distribution
 - Audit Log Viewer
 - Log Viewer
 - Alarm Manager
- ▼ **System Configuration**
 - Applications
 - Certificates
 - Licensing
 - MPP Servers
 - Report Data
 - SNMP
 - Speech Servers
 - Viewer Settings
 - VoIP Connections**
 - VPMS Servers
- ▼ **Reports**
 - Application Summary
 - Application Detail
 - Call Summary
 - Call Detail
 - Performance
 - Session Summary
 - Session Detail

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > Add H.323 Connection

Add H.323 Connection

Use this page to add a new H.323 connection.

Name:

Enable: ☒ Yes ☐ No

Gatekeeper Address:

Alternative Gatekeeper Address:

Gatekeeper Port:

Media Encryption: ☐ Yes ☒ No

New Stations

From	To
Station: <input type="text" value="31020"/>	<input type="text" value="31029"/>
Password: <input type="password" value="****"/>	
<input checked="" type="radio"/> Same Password	
<input type="radio"/> Use sequential passwords	
Station Type: <input type="text" value="Inbound and Outbound"/>	
<input type="button" value="Add"/>	

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.

AVAYA Welc
Last logged in 7/20/10 at 1:4

Voice Portal 4.1 (VoicePortal) Home Help

Expand All | Collapse All

▼ User Management
Users

▼ System Maintenance
System Monitor
MPP Manager
Active Calls
Port Distribution
Audit Log Viewer
Log Viewer
Alarm Manager

▼ System Configuration
Applications
Certificates
Licensing
MPP Servers
Report Data
SNMP
Speech Servers
Viewer Settings
VoIP Connections
VPMS Servers

▼ Reports
Application Summary
Application Detail
Call Summary
Call Detail
Performance
Session Summary
Session Detail

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > Change Application

Change Application

Use this page to change the configuration of a VoiceXML or CCXML application.

Name:

Enable: ☒ Yes ☐ No

MIME Type:

VoiceXML URL:

Speech Servers

ASR: TTS:

Application Launch

Type: ☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

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.

AVAYA Last logged in 7/20

Voice Portal 4.1 (VoicePortal) Home

Expand All | Collapse All

User Management
Users

System Maintenance
System Monitor
MPP Manager
Active Calls
Port Distribution
Audit Log Viewer
Log Viewer
Alarm Manager

System Configuration
Applications
Certificates
Licensing
MPP Servers
Report Data
SNMP
Speech Servers
Viewer Settings
VoIP Connections
VPMS Servers

Reports
Application Summary
Application Detail
Call Summary
Call Detail
Performance
Session Summary
Session Detail

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > [Change Application](#)

Change Application

Use this page to change the configuration of a VoiceXML or CCXML application.

Name: SabioCB_AgentOffer

Enable: ☒ Yes ☐ No

MIME Type: VoiceXML

VoiceXML URL: <http://10.20.2.198/SIC/callback/CallbackAgentOfferFront.jsp> **Verify**

Speech Servers

ASR: No ASR TTS: No TTS

Application Launch

Type: ☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number: **Add**

Remove

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.

AVAYA Application Enablement Operations Administration and Maintenance

You are here: > Administration > Switch Connections

Switch Connections

DevConnCM Add Connection

Connection Name	Number of Active Connections
<input checked="" type="radio"/> ACMG650	1
<input type="radio"/> acm	0

Edit Connection Edit CLAN IPs Edit H.323 Gatekeeper Delete Connection

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.

AVAYA Application Enablement Operations Administration and Maintenance

You are here: > Administration > Switch Connections

Set Password - DevConnCM

Please note the following:
* Changing the password affects only new connections, not open connections.

Switch Password

Confirm Switch Password

SSL ☒

Apply Cancel

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.

AVAYA Application Enablement Operations Administration and Maintenance

You are here: > Administration > Switch Connections

Edit CLAN IPs - DevConnCM

10.20.2.15 Add Name or IP

Name or IP Address	Status
--------------------	--------

Delete IP

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 link 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.

The screenshot shows the 'Add / Edit TSAPI Links' form. The left sidebar contains a navigation tree with 'Administration' expanded, showing 'CTI Link Admin' and 'TSAPI Links'. The main content area has a breadcrumb trail: 'You are here: > Administration > CTI Link Admin > TSAPI Links'. The form fields are: 'Link:' with a dropdown set to '3', 'Switch Connection:' with a dropdown set to 'DevConnCM', 'Switch CTI Link Number:' with a dropdown set to '11', 'ASAI Link Version' with a dropdown set to '4', and 'Security' with a dropdown set to 'Both'. Below the fields are two buttons: 'Apply Changes' and 'Cancel Changes'.

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**.

The screenshot shows the 'Tlinks' page. The left sidebar contains a navigation tree with 'Administration' expanded, showing 'Security Database' and 'Tlinks'. The main content area has a breadcrumb trail: 'You are here: > Administration > Security Database > Tlinks'. The page title is 'Tlinks'. Below the title is a list of 'Tlink Name' entries. The first entry is 'AVAYA#DEVCONNCM#CSTA#AESESERVER' and the second entry is 'AVAYA#DEVCONNCM#CSTA-S#AESESERVER'. The second entry is selected with a radio button. Below the list are two buttons: 'Edit Tlink' and 'Delete Tlink'.

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).

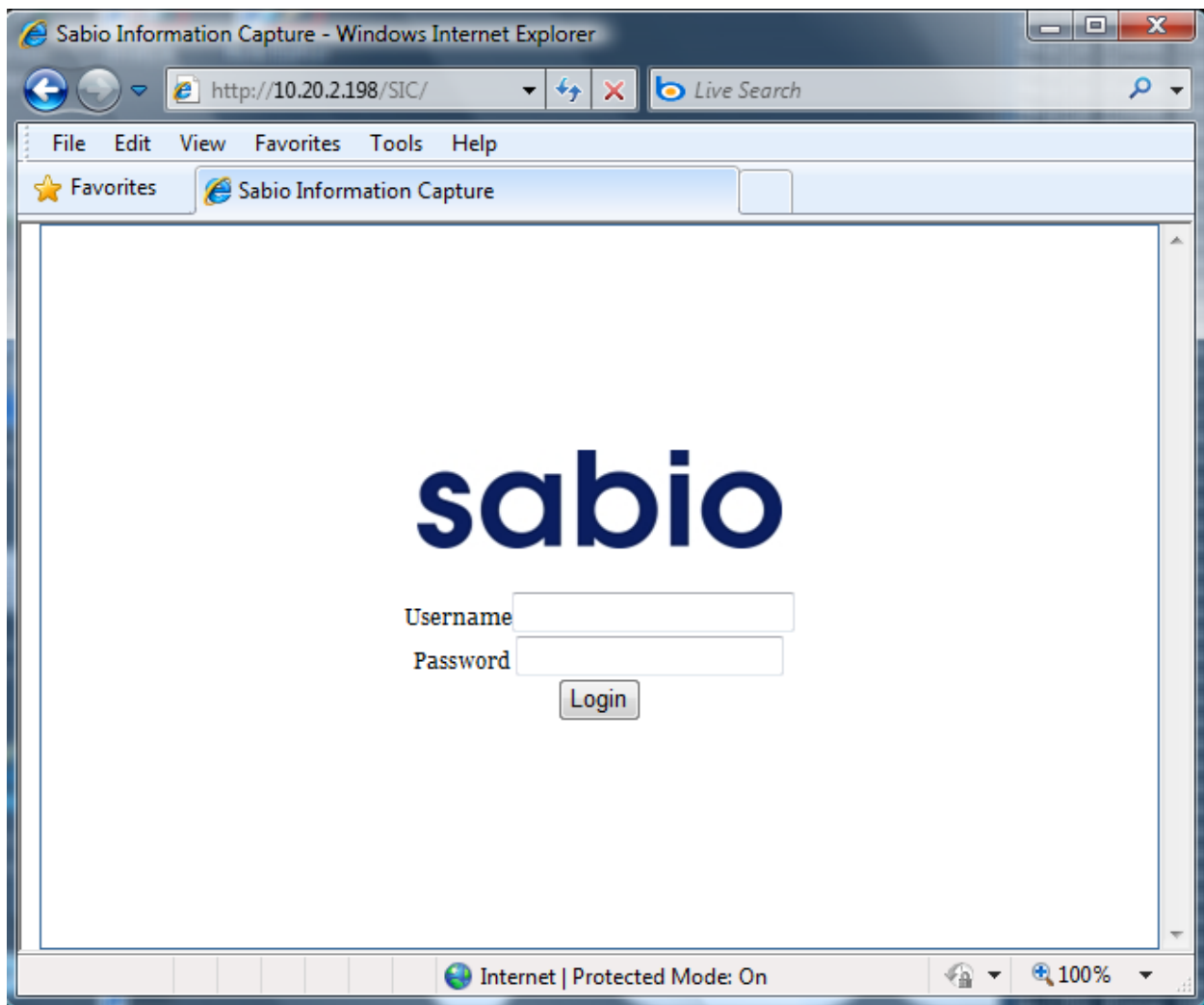
The screenshot displays the Avaya Application Enablement Services (AES) Administration and Monitoring (AAM) web interface. The top header shows the Avaya logo and the title 'Application Enablement Services Operations Administration and Monitoring'. Below the header, a breadcrumb trail indicates the current location: 'You are here: > User Management > Add User'. The left sidebar contains a navigation menu with the following items: 'User Management Home', 'User Management' (expanded), 'List All Users', 'Add User' (selected), 'Search Users', 'Modify Default User', 'Change User Password', 'Service Management', and 'Help'. The main content area is titled 'Add User' and includes a note: 'Fields marked with * can not be empty.' The form contains the following fields: '* User Id' (text input, value: 'sabio'), '* Common Name' (text input, value: 'sabio'), '* Surname' (text input, value: 'sabio'), '* User Password' (password input, masked with dots), '* Confirm Password' (password input, masked with dots), 'Admin Note' (text area), 'Avaya Role' (dropdown menu, value: 'None'), 'Business Category' (text input), 'Car License' (text input), 'CM Home' (text input), 'Css Home' (text input), 'CT User' (dropdown menu, value: 'Yes'), and 'Department Number' (text input). Red rectangular boxes highlight the compulsory fields: '* User Id', '* Common Name', '* Surname', '* User Password', '* Confirm Password', and 'CT User'.

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.

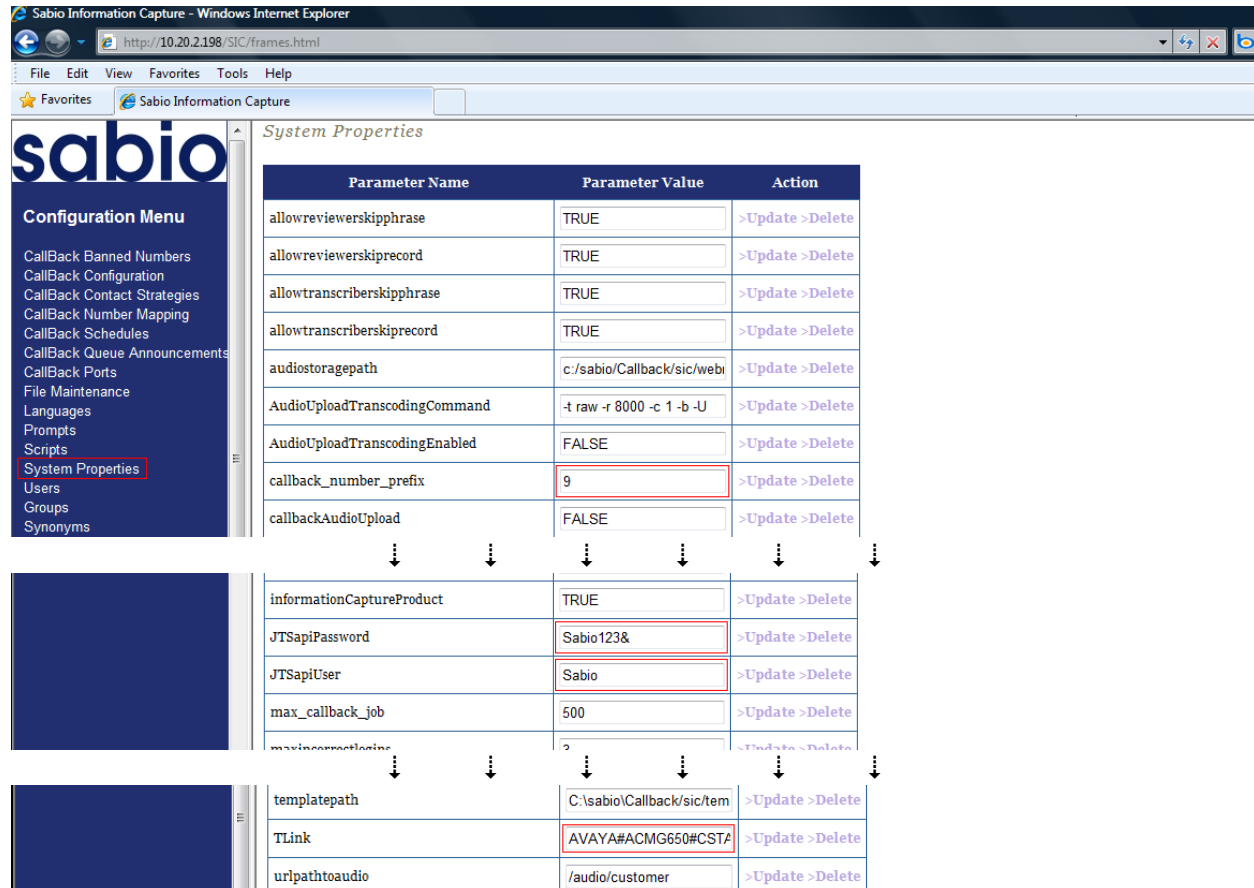


7.2 Configure System Properties

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**.



Parameter Name	Parameter Value	Action
allowreviewerskipphrase	TRUE	>Update >Delete
allowreviewerskiprecord	TRUE	>Update >Delete
allowtranscriberskipphrase	TRUE	>Update >Delete
allowtranscriberskiprecord	TRUE	>Update >Delete
audiostoragepath	c:/sabio/Callback/sic/web	>Update >Delete
AudioUploadTranscodingCommand	-t raw -r 8000 -c 1 -b -U	>Update >Delete
AudioUploadTranscodingEnabled	FALSE	>Update >Delete
callback_number_prefix	9	>Update >Delete
callbackAudioUpload	FALSE	>Update >Delete
informationCaptureProduct	TRUE	>Update >Delete
JTSapiPassword	Sabio123&	>Update >Delete
JTSapiUser	Sabio	>Update >Delete
max_callback_job	500	>Update >Delete
maxrecordingtime	?	>Update >Delete
templatepath	C:/sabio/Callback/sic/tem	>Update >Delete
TLink	AVAYA#ACMG650#CSTA	>Update >Delete
uripathtoaudio	/audio/customer	>Update >Delete

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.

The screenshot shows the Sabio Information Capture web interface in a Windows Internet Explorer browser. The address bar displays `http://10.20.2.198/SIC/frames.html`. The sidebar menu on the left includes the following items: Configuration Menu, CallBack Banned Numbers, CallBack Configuration, CallBack Contact Strategies, CallBack Number Mapping, CallBack Schedules, CallBack Queue Announcements, **CallBack Ports** (highlighted with a red box), File Maintenance, Languages, Prompts, Scripts, System Properties, Users, Groups, and Synonyms. The main content area is titled *CallBack Ports* and contains a table of existing ports and a form to enter new port details.

ID	IVRPort	XPort	Active	Action
75	71056	31030	true	>Delete >Update
76	71056	31031	true	>Delete >Update
77	71056	31032	true	>Delete >Update

Below the table is a form titled "Enter new CallBack Port Details" with the following fields:

- IVRPort:** 71056
- XPort:** 31033
- Active:** true (dropdown menu)

A **Submit** button is located below the form.

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.

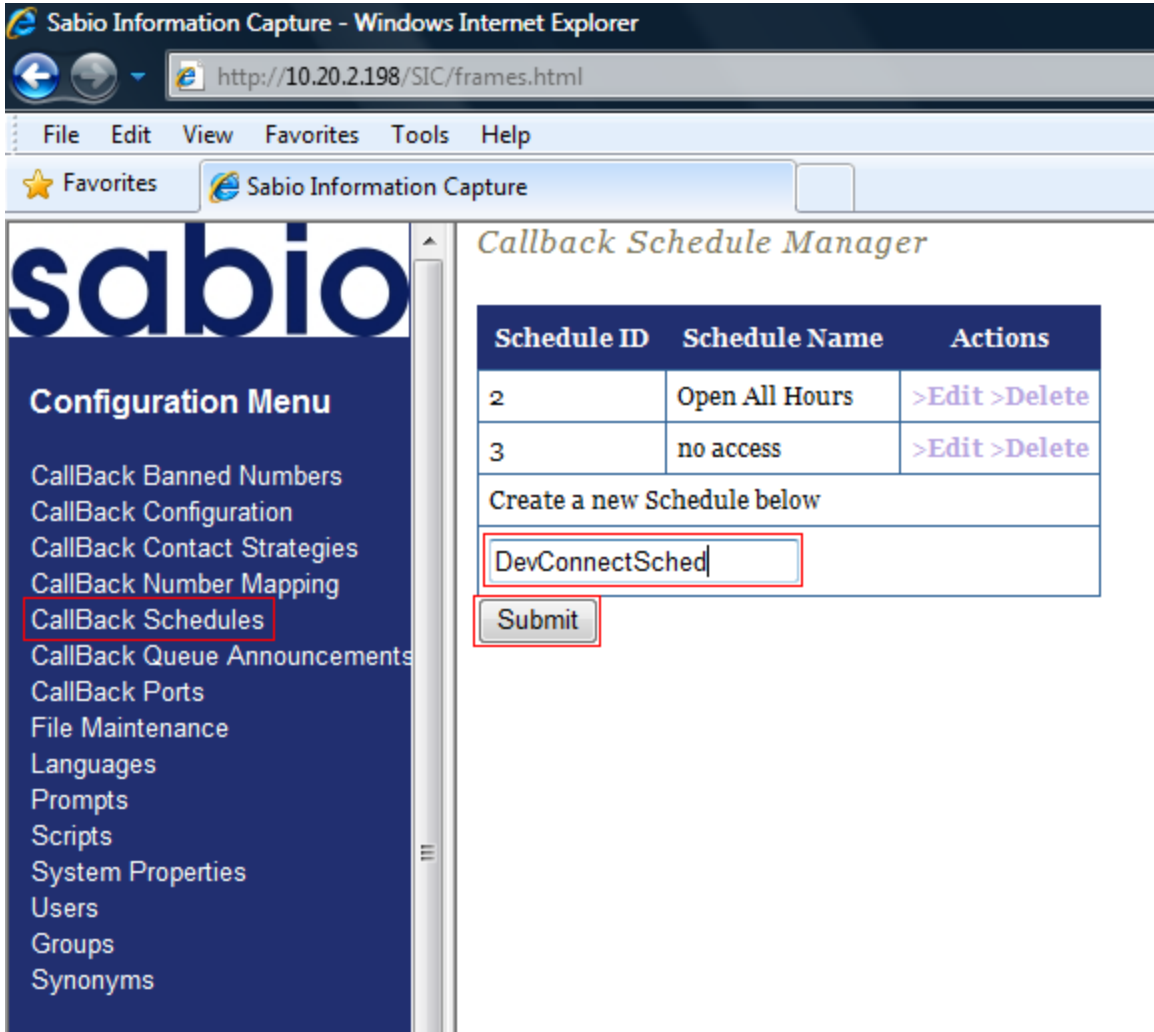
The screenshot shows a web browser window titled "Sabio Information Capture - Windows Internet Explorer" with the address bar showing "http://10.20.2.198/SIC/frames.html". The browser's menu bar includes File, Edit, View, Favorites, Tools, and Help. The Favorites bar shows "Sabio Information Capture". The main content area is divided into two panes. The left pane, titled "sabio", contains a "Configuration Menu" with the following items: CallBack Banned Numbers (highlighted with a red box), CallBack Configuration, CallBack Contact Strategies, CallBack Number Mapping, CallBack Schedules, CallBack Queue Announcements, CallBack Ports, File Maintenance, Languages, Prompts, Scripts, System Properties, Users, Groups, and Synonyms. The right pane, titled "Banned Numbers", contains a table with the following data:

ID	Banned Number/Regex	Action
3	999	>Delete
11	02081231001	>Delete
12	0208123101[0-9]{1}	>Delete

Below the table is a text input field with the placeholder text "Enter new Banned Number/Regex below if required". Below the input field is a "Submit" button (highlighted with a red box).

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 screenshot shows the Sabio Information Capture web interface in a Windows Internet Explorer browser. The address bar shows the URL `http://10.20.2.198/SIC/frames.html`. The left pane displays the **sabio** logo and a **Configuration Menu** with the following items: CallBack Banned Numbers, CallBack Configuration, CallBack Contact Strategies, CallBack Number Mapping, **CallBack Schedules** (highlighted with a red box), CallBack Queue Announcements, CallBack Ports, File Maintenance, Languages, Prompts, Scripts, System Properties, Users, Groups, and Synonyms. The main content area is titled **CallBack Schedule Manager** and contains a table with the following data:

Schedule ID	Schedule Name	Actions
2	Open All Hours	>Edit >Delete
3	no access	>Edit >Delete

Below the table, there is a heading **Create a new Schedule below** and a text input field containing `DevConnectSched` (highlighted with a red box). Below the input field is a **Submit** button (highlighted with a red box).

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.

Script Details

ID	45
Name	DevConnectSched
<input type="button" value="Update"/>	

Create new Schedule entry

Type	<input type="text" value="Dialing"/>
Day Of Week	<input type="text" value="Tuesday"/>
Start Time	<input type="text" value="09:00"/>
End Time	<input type="text" value="17:00"/>
<input type="button" value="Create"/>	

Entries

Entry ID	Type	Day Of Week	Start Time	End Time	Action
485	<input type="text" value="Dialing"/>	<input type="text" value="Monday"/>	<input type="text" value="09:00"/>	<input type="text" value="17:00"/>	>Update >Delete
490	<input type="text" value="Offer"/>	<input type="text" value="Monday"/>	<input type="text" value="11:00"/>	<input type="text" value="14:00"/>	>Update >Delete

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 screenshot shows a web browser window titled "Sabio Information Capture - Windows Internet Explorer" with the address bar displaying "http://10.20.2.198/SIC/frames.html". The browser's menu bar includes File, Edit, View, Favorites, Tools, and Help. The Favorites bar shows "Sabio Information Capture".

The main content area is divided into two panes. The left pane, titled "sabio", contains a "Configuration Menu" with the following items: CallBack Banned Numbers, CallBack Configuration, CallBack Contact Strategies, CallBack Number Mapping, CallBack Schedules, CallBack Queue Announcements, CallBack Ports, File Maintenance, Languages, Prompts, Scripts, System Properties, Users, Groups, and Synonyms. The "CallBack Contact Strategies" item is highlighted.

The right pane, titled "Callback Contact Strategy Manager", contains a table with the following data:

Contact Strategy ID	Contact Strategy Name	Actions
3	test	>Edit >Delete

Below the table, there is a heading "Create a new Contact Strategy below" followed by a text input field containing "4 Attempts x 5>15 mins". A "Submit" button is located below the input field.

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.

Script Details

ID	52
Name	4 Attempts x 5>15 mins
<input type="button" value="Update"/>	

Create new contact strategy entry

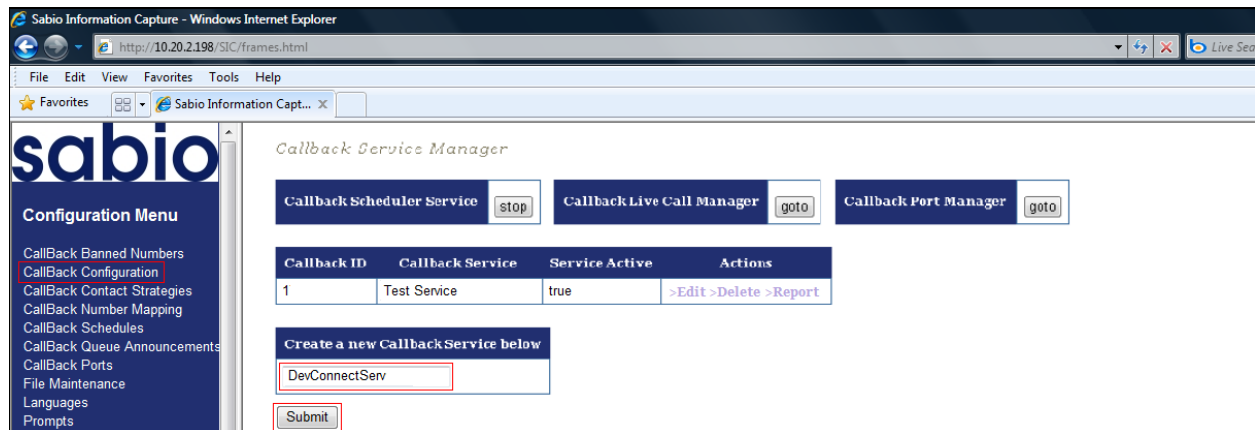
Delay

Entries

Entry ID	Sequence	Delay	Action
42051	1	5	>Update
42052	2	10	>Update
42053	2	10	>Update
42054	2	15	>Update >Delete

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 be 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 with not 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 screen shot, as such, the screen shot below shows only some example values for illustration purposes.

Modify Existing Callback Service Language Configuration

Lang ID	Description	Def Queue	Initial Delay	Min Number Length	Max Number Length	Number Prefix
o	English UK	71055	10	0	0	9

Test Menu

- Callback Enter Callback
- Test Banned Number

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.

Configuration Menu

- CallBack Banned Numbers
- CallBack Configuration
- CallBack Contact Strategies
- CallBack Number Mapping**
- CallBack Schedules
- CallBack Queue Announcements
- CallBack Ports
- File Maintenance
- Languages
- Prompts
- Scripts
- System Properties
- Users
- Groups
- Synonyms

Reporting Menu

- CallBack Activity

Create new Number Mapping entry

CallBack Definition: Test Service(1)

CallBack Configuration: English UK(48)

Handback Type: Disconnect

Inbound Number:

Outbound Number:

Create

Entries

Entry ID	Callback Service	Callback Configuration	Type	Inbound Number	Outbound Number	Action
3	DevConnectSen(44)	English UK(48)	Converse Multi Stage Stage 1	71053	71055	>Update >Delete
4	DevConnectSen(44)	English UK(48)	Converse Multi Stage Stage 2	71054	71055	>Update >Delete

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						
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd
11	4	no	aesserver	established	15	15

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						Page 4 of 4
Grp/Mod	Grp/Mod	Grp/Mod	Grp/Mod	Grp/Mod		
90/AI	/	/	/	/	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**

AVAYA Voice Portal 4.1 (VoicePortal) Home Help

You are here: [Home](#) > System Maintenance > MPP Manager

MPP Manager (8/17/10 10:54:27 AM GMT)

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, selected MPPs must also be stopped.

Last Poll: 8/17/10 10:54:22 AM GMT

Server Name	Mode	State	Config	Auto Restart	Restart Schedule Today	Restart Schedule Recurring	Active Calls In	Active Calls Out
MPP1	Online	Running	OK	Yes	No	None	0	0

State Commands
Start Stop Restart Reboot Halt Cancel

Restart/Reboot Options
☐ One server at a time
☒ All selected servers at the same time

Mode Commands
Offline Test Online

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.

Sabio Information Capture - Windows Internet Explorer

http://10.20.2.198/SIC/frames.html

File Edit View Favorites Tools Help

Sabio Information Capt... x Avaya Voice Portal Manag...

sabio

Configuration Menu

- CallBack Banned Numbers
- CallBack Configuration
- CallBack Contact Strategies
- CallBack Number Mapping
- CallBack Schedules
- CallBack Queue Announcements
- Prompts
- Scripts
- Synonyms

Reporting Menu

- CallBack Activity
- CallBack Audit
- CallBack Available Capacity
- CallBack Live Manager
- CallBack Queue

Callback Live Call Manager

Show All Ports [Go](#)

ID	IVRPort	XPort	Active	Pooled Status	Call Active	Job Active	C/B Thread Timeout	Job ID	Destination	Job Timeout	Callback Def ID	Callback Conf ID	Attended
75	71056	31030	true	Not Initialized	false	false							
76	71056	31031	true	OUT	true	true	qvdnThread:30961secs	callback-group.job-callback-665	restricted	31965secs	665	48	2
77	71056	31032	true	IN	false	false							

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*
3. *Application Enablement Services Administration and Maintenance Guide Document No. 02-300357, May 2008*
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>

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