



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Globitel SpeechLog Voice Recorder to interoperate with Avaya Aura® Contact Center R6.4 and Avaya Aura® Communication Manager R7.0 using DMCC Service Observe to record calls - Issue 1.0

Abstract

These Application Notes describe the configuration steps for the Globitel SpeechLog Voice Recorder to interoperate with the Avaya solution consisting of an Avaya Aura® Contact Center, Avaya Aura® Communication Manager R7.0, Avaya Aura® Session Manager R7.0, and Avaya Aura® Application Enablement Services R7.0.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps for the Globitel SpeechLog Voice Recorder to interoperate with the Avaya solution consisting of an Avaya Aura® Contact Center R6.4, Avaya Aura® Communication Manager R7.0, Avaya Aura® Session Manager R7.0, and Avaya Aura® Application Enablement Services R7.0.

When a call is to be recorded, SpeechLog Voice Recorder performs recording using two connections to the Avaya Solution.

- Using CCT Web Services to receive events on agent and call info.
- Using TSAPI link with AES to get call events (ringing, established, released, etc.).
- Using Device Media Call Control to perform service observe between the extension to be recorded and a configured virtual softphone enabled station.

SpeechLog Voice Recorder uses Device Media Call Control Service Observe to record Avaya Aura® Contact Centre (Contact Centre) skillset calls on Avaya Aura® Communication Manager deskphones. Device Media Call Control (DMCC) works by allowing software vendors to create soft phones, in memory on a recording server, and use them to monitor and record other phones. This is purely a software solution and does not require telephony boards or any wiring beyond a typical network infrastructure.

Globitel's SpeechLog Voice Recorder is fully integrated into a LAN (Local Area Network), and includes easy-to-use Web based applications that can be used to retrieve telephone conversations from a comprehensive long-term calls database. SpeechLog Voice Recorder uses the Communication Manager feature "Service Observe" to observe a call on an extension; this way the call is recorded and can be played back at a later time.

2. General Test Approach and Test Results

The compliance testing focuses on the recording of Contact Centre skillset calls on Communication Manager deskphones. SpeechLog Voice Recorder connects to Communication Control Toolkit (CCT) Web Services in order to obtain events pertaining to specific Contact Centre skillset calls. SpeechLog Voice Recorder can then record the call based on the events it receives.

Recording of Contact Centre skillset calls can be achieved using CCT Web Services and DMCC to record calls. The recording application sends a message to the DMCC integration application to begin recording the voice stream coming to that softphone extension. SpeechLog Voice Recorder utilises a CTI through Avaya Aura® Application Enablement Services (AES) to record calls on Communication Manager deskphones using Service Observe. In this message, the recorder passes along the softphone extension to be recorded along with the location and filename of the recording. Test cases are executed to exercise a sufficiently broad segment of functionality to have a reasonable expectation of interoperability in production configurations.

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

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on placing and recording calls in different call scenarios with good quality audio recordings and accurate call records. The tests included:

- **Contact Centre Inbound/Outbound Calls** - Test call recording for inbound/outbound calls to the AACC Agents from PSTN callers.
- **Contact Centre Hold/Transferred/Conference calls** – Test call recording for calls transferred to and in conference with PSTN callers.
- **Serviceability testing** - The behaviour of SpeechLog Voice Recorder under different simulated failure conditions on the Avaya platform were also observed.

2.2. Test Results

All functionality and serviceability test cases were completed successfully.

2.3. Support

Technical support can be obtained for Globitel SpeechLog Voice Recorder at:

Globitel
Khalda, Amman, Jordan.
support@globitel.com
Hotline: +962 (7) 97315050
Phone: +962 (6) 5300 130
Fax: +962 (6) 5300 144
P.O. Box 1786 Amman 11821 Jordan

3. Reference Configuration

The configuration in **Figure 1** was used to compliance test Globitel SpeechLog Voice Recorder with Avaya Aura® Contact Center connecting to Avaya Aura® Communication Manager R7.0 and Avaya Aura® Application Enablement Services R7.0 using DMCC Service Observe.

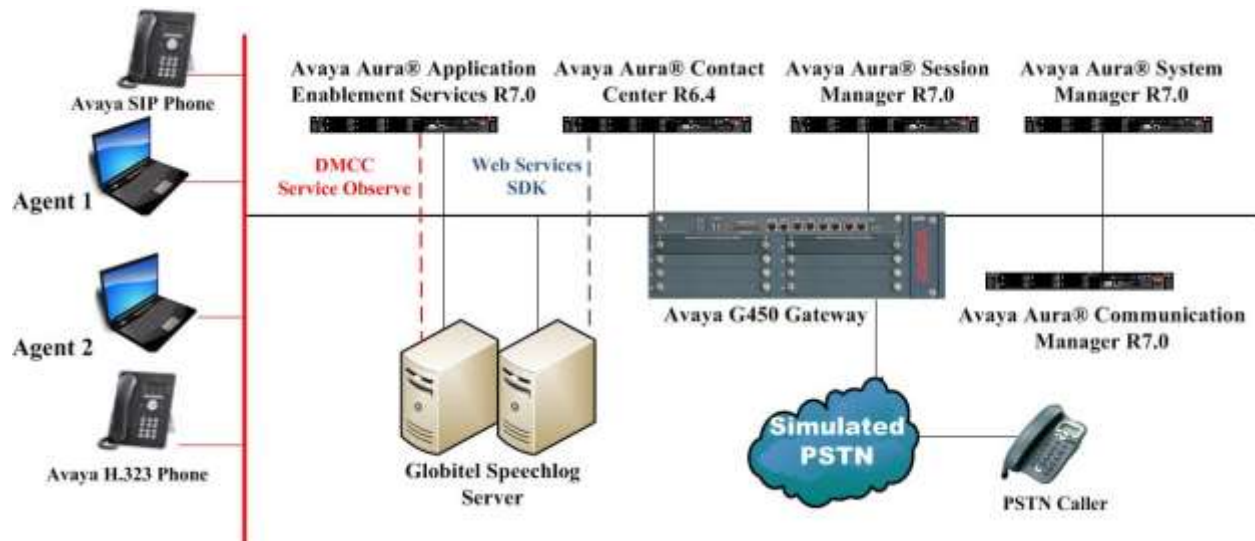


Figure 1: Connection of Globitel SpeechLog Voice Recorder with Avaya Aura® Contact Center connected to Avaya Aura® Communication Manager R7.0 and Avaya Aura® Application Enablement Services R7.0.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager running on Virtual Server	R7.0.0.0.0 Build 7.0.0.0.16266-7.0.9.9.902 SW Update Revision No. 7.0.0.0.3873
Avaya Aura® Session Manager running on Virtual Server	R7.0.0.0.700007
Avaya Aura® Communication Manager running on Virtual Server	R7.0 Build 017x.00.0.441.0
Avaya Aura® Application Enablement Services running on a Virtual Server	R7.0 Build No – 7.0.0.0.0.13-0
Avaya Aura® Contact Center running on a Virtual Server	R6.4 Service Pack 15
Avaya G450 Gateway	37.19.0 /1
Avaya 9608 H323 Deskphone	96x1 H323 Release 6.6.028
Avaya 9641 SIP Deskphone	96x1 SIP Release 6.5.0.17
Globitel SpeechLog Voice Recorder	7.0

5. Configure Avaya Aura® Communication Manager

The information provided in this section describes the configuration of Communication Manager relevant to this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

The configuration illustrated in this section was performed using Communication Manager System Administration Terminal (SAT).

5.1. Verify System Features

Use the **display system-parameters customer-options** command to verify that Communication Manager has permissions for features illustrated in these Application Notes. On **Page 3**, ensure that **Computer Telephony Adjunct Links?** is set to **y** as shown below.

display system-parameters customer-options		Page	3 of 11
OPTIONAL FEATURES			
Abbreviated Dialing Enhanced List?	y	Audible Message Waiting?	y
Access Security Gateway (ASG)?	n	Authorization Codes?	y
Analog Trunk Incoming Call ID?	y	CAS Branch?	n
A/D Grp/Sys List Dialing Start at 01?	y	CAS Main?	n
Answer Supervision by Call Classifier?	y	Change COR by FAC?	n
ARS?	y	Computer Telephony Adjunct Links?	y
ARS/AAR Partitioning?	y	Cvg Of Calls Redirected Off-net?	y
ARS/AAR Dialing without FAC?	y	DCS (Basic)?	y
ASAI Link Core Capabilities?	n	DCS Call Coverage?	y
ASAI Link Plus Capabilities?	n	DCS with Rerouting?	y
Async. Transfer Mode (ATM) PNC?	n	Digital Loss Plan Modification?	y
Async. Transfer Mode (ATM) Trunking?	n	DS1 MSP?	y
ATM WAN Spare Processor?	n	DS1 Echo Cancellation?	y
ATMS?	y		
Attendant Vectoring?	y		

5.2. Note procr IP Address for Avaya Aura® Application Enablement Services Connectivity

Display the procr IP address by using the command **display node-names ip** and note the IP address for the **procr** and AES (**aes70vmpg**).

display node-names ip		Page	1 of 2
IP NODE NAMES			
Name	IP Address		
SM100	10.10.40.12		
aes70vmpg	10.10.40.16		
default	0.0.0.0		
PGDECT	10.10.40.50		
procr	10.10.40.13		

5.3. Configure Transport Link for Avaya Aura® Application Enablement Services Connectivity

To administer the transport link to AES 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 procr in **Section 5.2**
- **Local Port:** Retain the default value of **8765**.

change ip-services					Page	1 of	3
IP SERVICES							
Service	Enabled	Local	Local	Remote	Remote		
Type		Node	Port	Node	Port		
AESVCS	y	procr	8765				

Go to **Page 3** of the **ip-services** form and enter the following values:

- **AE Services Server:** Name obtained from the AES server, in this case **aes70vmpg**.
- **Password:** Enter a password to be administered on the AES server.
- **Enabled:** Set to **y**.

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

change ip-services				Page	3 of 3
AE Services Administration					
Server ID	AE Services Server	Password	Enabled	Status	
1:	aes70vmpg	*****	y	in use	
2:					
3:					

5.4. 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 1			Page 1 of 3		
CTI LINK					
CTI Link: 1					
Extension: 7777					
Type: ADJ-IP					
			COR: 1		
Name: aes70vmpg					

5.5. Configure Communication Manager for Service Observation

Type **display cor x**, where x is the COR number in the screen above, to check the existing Class of Restriction. Ensure that **Can be Service Observed** is set to **y**, if not type **change cor x** to make a change to the Class or Restriction. This value needs to be enabled in order for Service Observe to work for call recording.

```
display cor 1                                     Page 1 of 23
                                     CLASS OF RESTRICTION
COR Number: 1
COR Description:

FRL: 0                                           APLT? y
Can Be Service Observed? y                   Calling Party Restriction: all-toll
Can Be A Service Observer? y                   Called Party Restriction: none
Time of Day Chart: 1                           Forced Entry of Account Codes? n
Priority Queuing? n                             Direct Agent Calling? y
Restriction Override: all                       Facility Access Trunk Test? n
Restricted Call List? n                         Can Change Coverage? n
Unrestricted Call List: 1
Access to MCT? y                               Fully Restricted Service? n
Group II Category For MFC: 7                   Hear VDN of Origin Annc.? n
Send ANI for MFE? n                           Add/Remove Agent Skills? n
MF ANI Prefix:                               Automatic Charge Display? n
Hear System Music on Hold? y PASTE (Display PBX Data on Phone)? n
Can Be Picked Up By Directed Call Pickup? y
Can Use Directed Call Pickup? y
Group Controlled Restriction: inactive
```

Type **change system-parameters features**, on **Page 11** ensure that **Allow Two Observes in Same Call** is set to **y**.

```
change system-parameters features                Page 11 of 19
                                     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: 0                    Second Data Delay: 2
Converse Signaling Tone (msec): 100             Pause (msec): 70
Prompting Timeout (secs): 10
Interflow-qpos EWT Threshold: 2
Reverse Star/Pound Digit For Collect Step? n
Available Agent Adjustments for BSR? n
BSR Tie Strategy: 1st-found
Store VDN Name in Station's Local Call Log? n
SERVICE OBSERVING
Service Observing: Warning Tone? y              or Conference Tone? n
Service Observing/SSC Allowed with Exclusion? n
Allow Two Observers in Same Call? y
```


Type **change feature-access-codes** to access the feature codes on Communication Manager. Scroll to **Page 5** in order to view or change the **SERVICE OBSERVING** access codes. Note the **Service Observing Listen Only Access Code** is **#43**; this will be required in **Section 7.1** during the setup of SpeechLog Voice Recorder.

change feature-access-codes	Page 5 of 10
FEATURE ACCESS CODE (FAC)	
Call Center Features	
AGENT WORK MODES	
After Call Work Access Code:	#36
Assist Access Code:	
Auto-In Access Code:	#38
Aux Work Access Code:	#39
Login Access Code:	#40
Logout Access Code:	#41
Manual-in Access Code:	#42
SERVICE OBSERVING	
Service Observing Listen Only Access Code: #43	
Service Observing Listen/Talk Access Code:	#44
Service Observing No Talk Access Code:	
Service Observing Next Call Listen Only Access Code:	
Service Observing by Location Listen Only Access Code:	
Service Observing by Location Listen/Talk Access Code:	
AACC CONFERENCE MODES	
Restrict First Consult Activation:	Deactivation:
Restrict Second Consult Activation:	Deactivation:

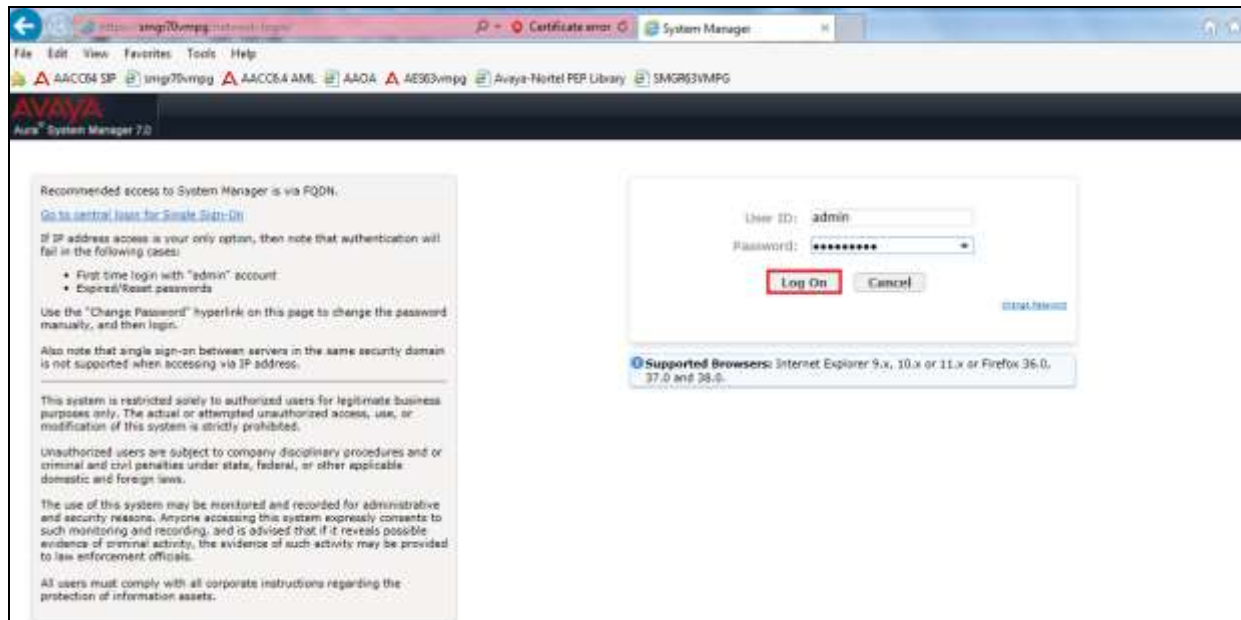
5.6. Configure H323 Stations for Service Observation

All endpoints that are to be monitored by SpeechLog Voice Recorder will need to have IP Softphone set to y. IP Softphone must be enabled in order for DMCC Service Observe to work. Type **change station x** where x is the extension number of the station to be monitored, also note this extension number for configuration required in **Section 7.2**. Note the **Security Code** and ensure that **IP SoftPhone** is set to y. Also ensure that the correct Class of Restriction (**COR**) is set to that configured in **Section 5.5**.

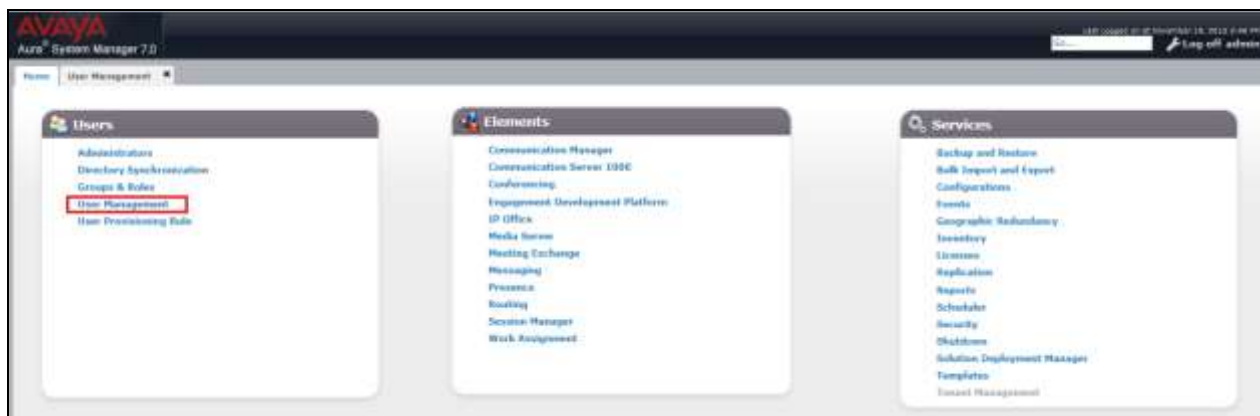
change station x	Page 1 of 5
STATION	
Extension: 7000	Lock Messages? n
Type: 9608	Security Code: *
Port: S00000	Coverage Path 1: 1
Name: Ext7000	Coverage Path 2:
	Hunt-to Station:
	BCC: 0
	TN: 1
	COR: 1
	COS: 1
	Tests? y
STATION OPTIONS	
	Time of Day Lock Table:
Loss Group: 19	Personalized Ringing Pattern: 1
	Message Lamp Ext: 7000
Speakerphone: 2-way	Mute Button Enabled? y
Display Language: english	Button Modules: 0
Survivable GK Node Name:	
Survivable COR: internal	Media Complex Ext:
Survivable Trunk Dest? y	IP SoftPhone? y
	IP Video Softphone? n
	Short/Prefixed Registration Allowed: yes
	Customizable Labels? y

5.7. Configure SIP Stations for Service Observation

The configuration of SIP phones on Communication Manager must be carried out from System Manager. Access the System Manager using a Web Browser by entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager or **http://<IP Address>/SMGR**. Log in using appropriate credentials.



From the home page click on **User Management** highlighted below.



Click on **Manager Users** in the left window. Select the station to be edited and click on **Edit**.

AVAYA
Aura® System Manager 7.0

Home / User Management

Home / Users / User Management / Manage Users

Search

User Management

Users

View Edit New Duplicate Delete More Actions

15 Items Show All

	Last Name	First Name	Display Name	Login Name	SIP Handle
<input checked="" type="checkbox"/>	7100	SIPEXt	7100, SIPEXt	7100@devconnect.local	7100
<input type="checkbox"/>	7101	SIPEXt	7101, SIPEXt	7101@devconnect.local	7101
<input type="checkbox"/>	7200	Ascom i62	7200, Ascom i62	7200@devconnect.local	7200
<input type="checkbox"/>	7201	Ascom i62	7201, Ascom i62	7201@devconnect.local	7201
<input type="checkbox"/>	7202	Ascom i62	7202, Ascom i62	7202@devconnect.local	7202
<input type="checkbox"/>	7203	Ascom i62	7203, Ascom i62	7203@devconnect.local	7203

Click on the **Communication Profile** tab. Ensure that the **Communication Profile Password** is known and if not click on edit to change it.

AVAYA
Aura® System Manager 7.0

Home / User Management

Home / Users / User Management / Manage Users

User Profile Edit: 7100@devconnect.local

Commit & Continue Commit Cancel

Identity Communication Profile Relationship Contacts

Communication Profile

Communication Profile Password: [REDACTED]

New Cancel

Name

Primary

Select: None

Name: Primary

Default: ☒

Communication Address *

New Cancel

Type	Handle	Domain
<input type="checkbox"/> Avaya SIP	7100	devconnect.local

Select: All, None

From the same page scroll down to **CM Endpoint Profile** and enter a suitable **Extension** number and select the correct **Template** then click on **Endpoint Editor** to make further changes.

☒ **CM Endpoint Profile**

* System

* Profile Type

Use Existing Endpoints ☐

* Extension **Endpoint Editor**

Template

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Calculate Route Pattern ☐

Sip Trunk

Enhanced Callr-Info display for 1-line phones ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User ☒

Override Endpoint Name and Localized Name ☒

Allow H.323 and SIP Endpoint Dual Registration ☐

In the **General Options** tab ensure that **Type of 3PCC Enabled** is set to **Avaya** as is shown below. Also that Class of Restriction is set to that configured in **Section 5.5**.

Edit Endpoint

System: cm70vmpg Extension: 7100
 Template: 9641SIPCC_DEFAULT_CM_7_8 Set Type: 9641SIPCC
 Port: 500003 Security Code:
 Name: 7100, SIPExt

General Options (G) * Feature Options (F) Site Data (S) Abbreviated Call Dialing (A) Enhanced Call Feed (E) Button Assignment (B) Profile Settings (P) Group Membership (M)

* Class of Restriction (COR) 1 * Class Of Service (COS) 1
 * Emergency Location Ext 7100 * Message Lamp Ext. 7100
 * Tenant Number 1
 * SIP Trunk Q aar
 Coverage Path 1
 Lock Message ☐
 Multibyte Language Not Applicable
 Type of 3PCC Enabled Avaya
 Coverage Path 2
 Localized Display Name 7100, SIPExt
 Enable Reachability for Station Domain Control system

* Required

Click on the **Feature Options** tab and ensure that **IP Softphone** is ticked as shown. Click on **Done**, at the bottom of the screen, once this is set.

General Options (G) * Feature Options (F) Site Data (S) Abbreviated Call Dialing (A) Enhanced Call Feed (E) Button Assignment (B) Profile Settings (P) Group Membership (M)

Active Station Ringing single
 MWI Served User Type sign-out
 Per Station CPN - Send Calling Number None
 IP Phone Group ID
 Remote Soft Phone Emergency Calls aar-local
 LWC Reception xps
 AUDIX Name
 Short/Prefixed Registration Allowed default
 Voice Mail Number

Auto Answer none
 Coverage After Forwarding system
 Display Language english
 Hunt-to Station
 Loss Group 19
 Survivable COR internal
 Time of Day Lock Table None
 Music Source

Features

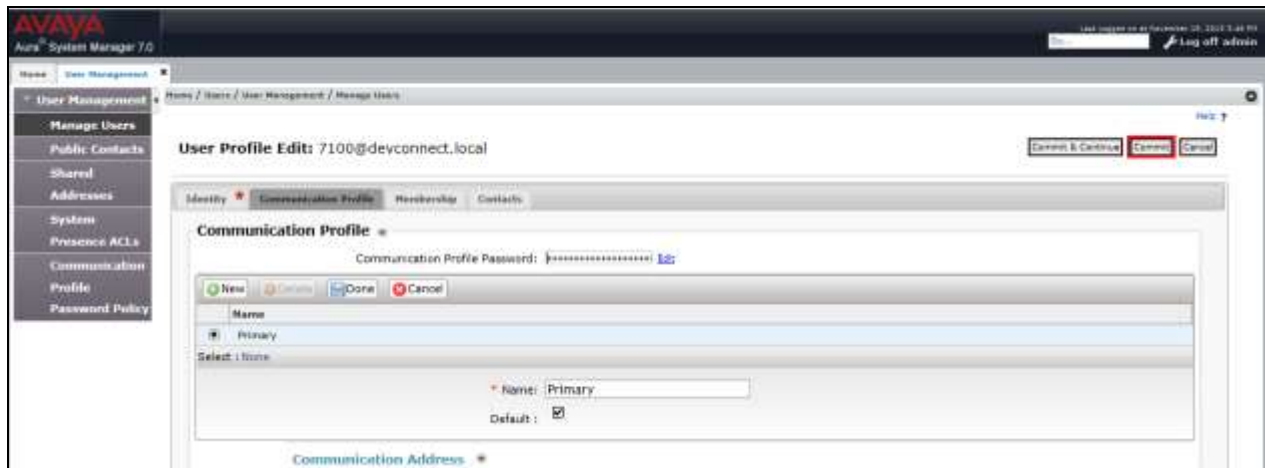
☐ Always Use
☐ IP Audio Hairpinning
☐ Bridged Call Alerting
☐ Bridged Idle Line Preference
☒ Coverage Message Retrieval
☐ Data Restriction
☒ Survivable Trunk Dest
☐ Bridged Appearance Origination Restriction
☒ Restrict Last Appearance

☐ Idle Appearance Preference
☒ IP SoftPhone
☒ LWC Activation
☐ CDR Privacy
☒ Direct IP-IP Audio Connections
☐ H.323 Conversion
☐ IP Video Softphone
☐ Per Button Ring Control

* Required

Done Cancel

Click on **Commit** once this is done to save the changes.



5.8. Configure Virtual Stations for Service Observe

Add virtual stations to allow SpeechLog Voice Recorder to record calls using Service Observe. Type **add station x** where x is the extension number of the station to be configured, also note this extension number for configuration required in **Section 7.1**. Note the **Security Code** and ensure that **IP SoftPhone** is set to **y**. Note also the **COR** for the stations, this will be set to that configured in **Section 5.5**.

add station 58900		Page 1 of 6
STATION		
Extension: 58900	Lock Messages? n	BCC: 0
Type: 4624	Security Code: *	TN: 1
Port: S00026	Coverage Path 1:	COR: 1
Name: Recorder1	Coverage Path 2:	COS: 1
	Hunt-to Station:	Tests? y
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
Speakerphone: 2-way	Personalized Ringing Pattern: 1	
Display Language: english	Message Lamp Ext: 58900	
Survivable GK Node Name:	Mute Button Enabled? y	
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? y	
	IP Video Softphone? n	
	Short/Prefixed Registration Allowed: default	

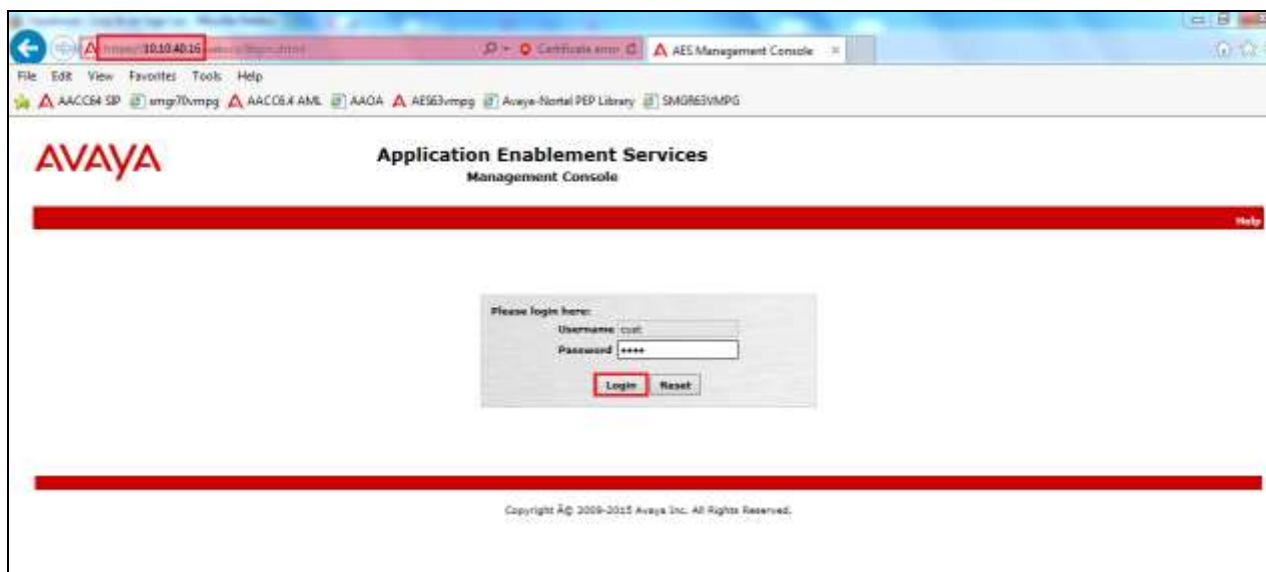
6. Configure Avaya Aura® Application Enablement Services

This section provides the procedures for configuring Application Enablement Services. The procedures fall into the following areas:

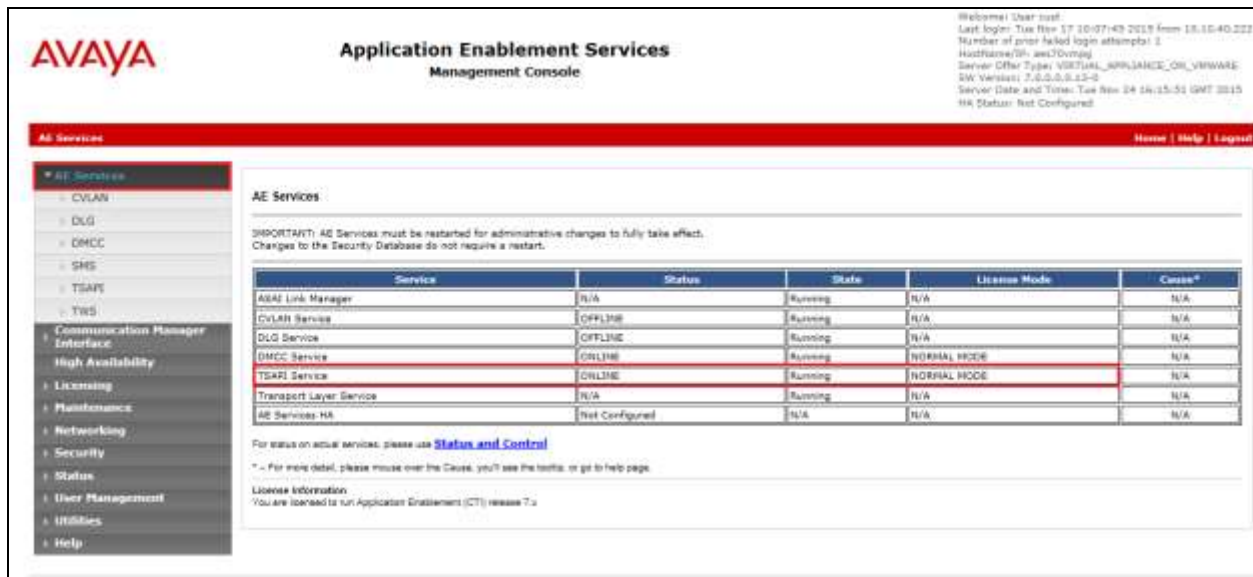
- Verify Licensing
- Create Switch Connection
- Administer TSAPI link
- Identify Tlinks
- Enable TSAPI Ports
- Create CTI User
- Associate Devices with CTI User

6.1. Verify Licensing

To access the AES Management Console, enter **https://<ip-addr>** as the URL in an Internet browser, where <ip-addr> is the IP address of AES. At the login screen displayed, log in with the appropriate credentials and then select the **Login** button.



The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the TSAPI Service is licensed by ensuring that **TSAPI Service** is in the list of **Services** and that the **License Mode** is showing **NORMAL MODE**. If not, contact an Avaya support representative to acquire the proper license for your solution.



AVAYA Application Enablement Services Management Console

Welcome! User: root
Last login: Tue Nov 17 10:07:45 2015 from 10.10.40.222
Number of prior failed login attempts: 1
HostName/IP: ams70vmg
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.0.0.0.13-0
Server Date and Time: Tue Nov 24 16:15:51 GMT 2015
HA Status: Not Configured

AE Services

IMPORTANT: AE Services must be restarted for administrative changes to fully take effect. Changes to the Security Database do not require a restart.

Service	Status	State	License Mode	Cause*
ASAE Link Manager	N/A	Running	N/A	N/A
CVLAN Service	OFFLINE	Running	N/A	N/A
DLG Service	OFFLINE	Running	N/A	N/A
DMCC Service	ONLINE	Running	NORMAL MODE	N/A
TSAPI Service	ONLINE	Running	NORMAL MODE	N/A
Transport Layer Service	N/A	Running	N/A	N/A
AE Services HA	Not Configured	N/A	N/A	N/A

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

* - For more detail, please mouse over the Cause, you'll see the tooltip, or go to help page.

License Information:
You are licensed to run Application Enablement (CT) release 7.0

6.2. Create Switch Connection

From the AES Management Console navigate to **Communication Manager Interface** → **Switch Connections** to set up a switch connection. Enter a name for the Switch Connection to be added and click the **Add Connection** button.



AVAYA Application Enablement Services Management Console

Welcome! User: root
Last login: Tue Nov 17 10:07:45 2015 from 10.10.40.222
Number of prior failed login attempts: 1
HostName/IP: ams70vmg
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 7.0.0.0.13-0
Server Date and Time: Tue Nov 24 16:16:56 GMT 2015
HA Status: Not Configured

Communication Manager Interface | Switch Connections

Switch Connections

Enter Name: **Add Connection**

Connection Name	Processor Ethernet	Plug Period	Number of Active Connections
Edit Connection Edit PE/CLAN 3ds Edit H.323 Gatekeeper Delete Connection Survivability Hierarchy			

In the resulting screen enter the **Switch Password**. The Switch Password must be the same as that entered into Communication Manager AE Services Administration screen via the **change ip-services** command described in **Section 5.3**. Default values may be accepted for the remaining fields. Click **Apply** to save changes.

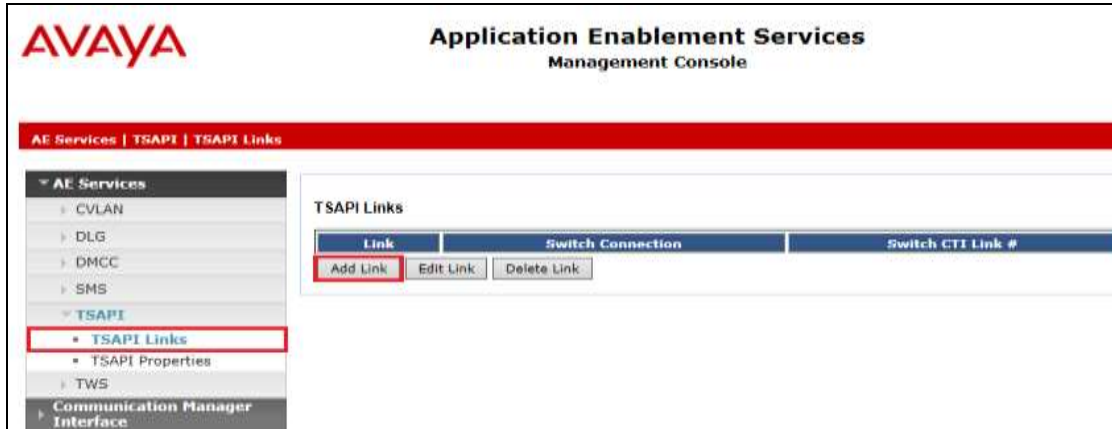
The screenshot shows the Avaya Application Enablement Services Management Console. The left sidebar contains a navigation menu with the following items: AE Services, Communication Manager Interface (selected), Switch Connections (highlighted with a red box), Dial Plan, High Availability, Licensing, Maintenance, Networking, Security, Status, User Management, Utilities, and Help. The main content area is titled 'Connection Details - cm70vmppg' and contains the following fields: Switch Password (password field), Confirm Switch Password (password field), Msg Period (30 Minutes (1 - 72)), Provide AE Services certificate to switch (checkbox), Secure H323 Connection (checkbox), and Processor Ethernet (checked checkbox). The 'Apply' button is highlighted with a red box.

From the **Switch Connections** screen, select the radio button for the recently added switch connection and select the **Edit PE/CLAN IPs** button (not shown, see screen at the bottom of the previous page). In the resulting screen, enter the IP address of the procr as shown in **Section 5.2** that will be used for the AES connection and select the **Add/Edit Name or IP** button.

The screenshot shows the Avaya Application Enablement Services Management Console. The left sidebar is the same as the previous screenshot. The main content area is titled 'Edit Processor Ethernet IP - cm70vmppg' and contains a table with the following columns: Name or IP Address. The table has one row with the value '10.10.40.13'. The 'Add/Edit Name or IP' button is highlighted with a red box. There is also a 'Back' button at the bottom left of the table.

6.3. Administer TSAPI link

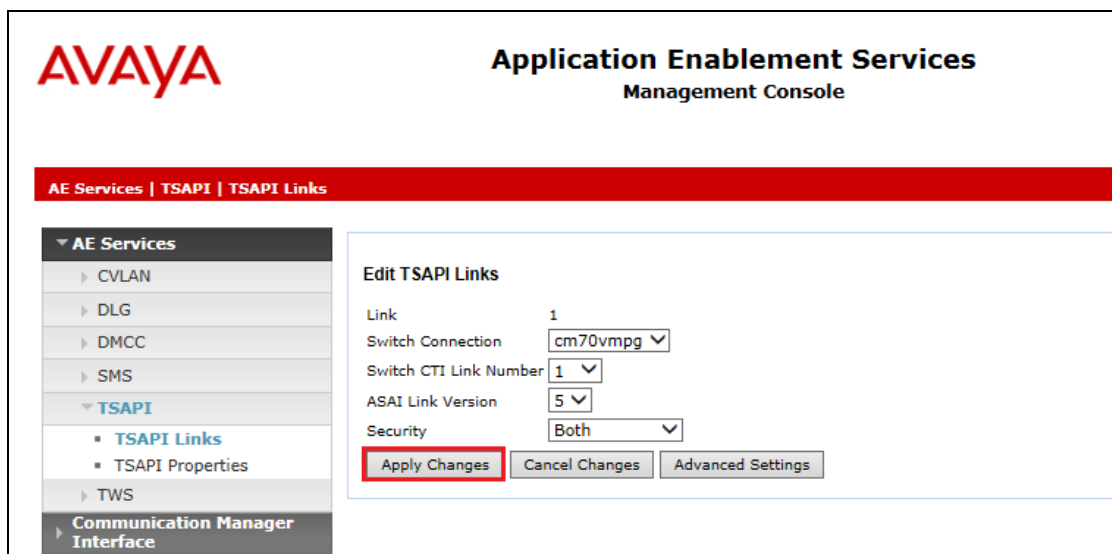
From the Application Enablement Services Management Console, select **AE Services** → **TSAPI** → **TSAPI Links**. Select **Add Link** button as shown in the screen below.



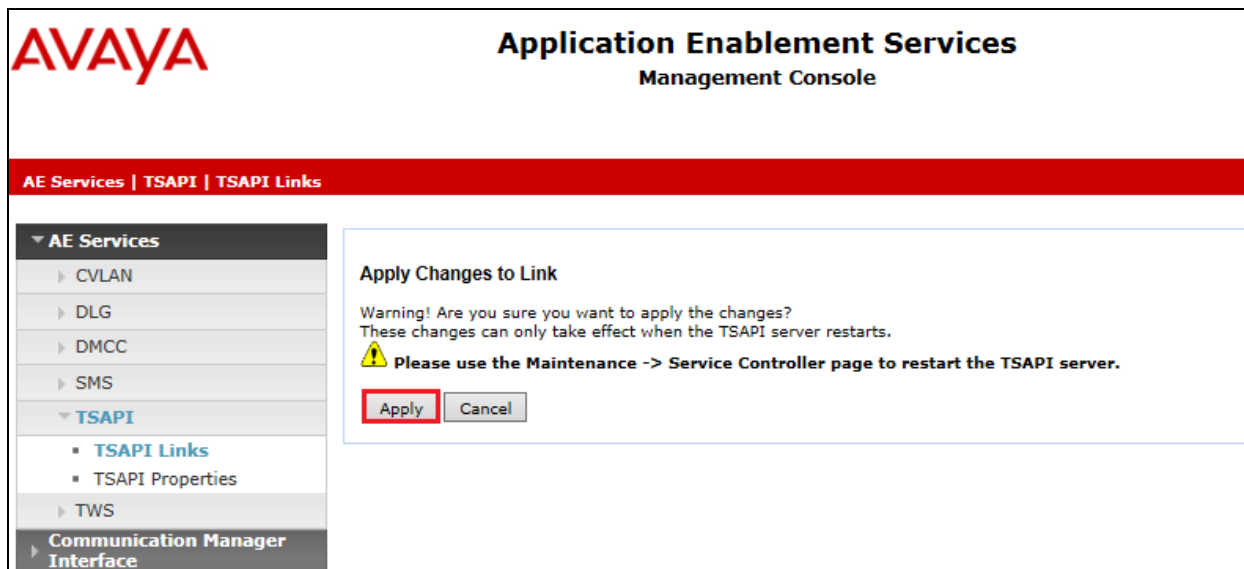
On the **Add TSAPI Links** screen (or the **Edit TSAPI Links** screen to edit a previously configured TSAPI Link as shown below), enter the following values:

- **Link:** Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm70vmpg**, which has already been configured in **Section 6.2**, from the drop-down list.
- **Switch CTI Link Number:** Corresponding CTI link number configured in **Section 5.4** which is **1**.
- **ASAI Link Version:** This can be left at the default value of **5**.
- **Security:** This can be left at the default value of **both**.

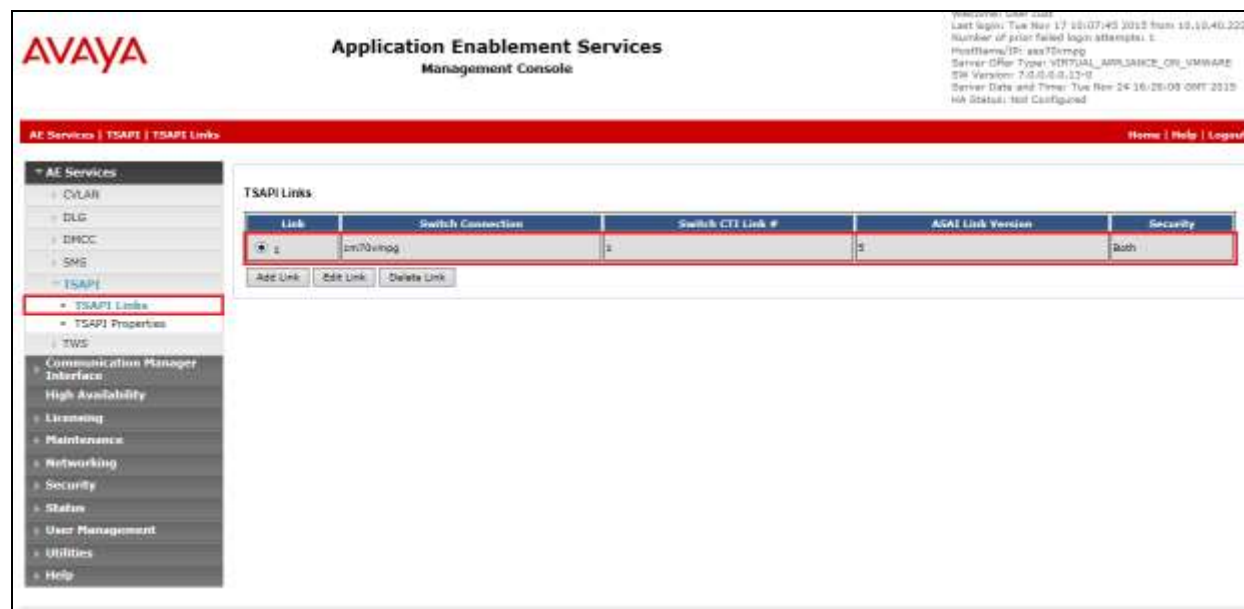
Once completed, select **Apply Changes**.



Another screen appears for confirmation of the changes made. Choose **Apply**.



When the TSAPI Link is completed, it should resemble the screen below.



The TSAPI Service must be restarted to effect the changes made in this section. From the Management Console menu, navigate to **Maintenance** → **Service Controller**. On the Service Controller screen, tick the **TSAPI Service** and select **Restart Service**.

AVAYA **Application Enablement Services**
Management Console

Maintenance | Service Controller

Service Controller

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

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

6.4. Identify Tlinks

Navigate to **Security** → **Security Database** → **Tlinks**. Verify the value of the **Tlink Name**. This will be needed to configure SpeechLog Voice Recorder in **Section 7.1**.

The screenshot displays the Avaya Application Enablement Services Management Console. The top header features the Avaya logo and the title "Application Enablement Services Management Console". A red navigation bar contains the text "Security | Security Database | Tlinks". On the left, a sidebar menu lists various services, with "Security Database" and its sub-item "Tlinks" highlighted with red boxes. The main content area, titled "Tlinks", shows a "Tlink Name" field with two radio button options: "AVAYA#CM70VMPPG#CSTA#AES70VMPPG" (selected) and "AVAYA#CM70VMPPG#CSTA-S#AES70VMPPG". A "Delete Tlink" button is located below the options.

6.5. Enable TSAPI Ports

To ensure that TSAPI ports are enabled, navigate to **Networking → Ports**. Ensure that the TSAPI ports are set to **Enabled** as shown below. Ensure that the **DMCC Server Ports** are also **Enabled** and take note of the **Unencrypted Port 4721** which will be used later in **Section 7.1**.

AVAYA Application Enablement Services Management Console

Networking | Ports

Ports

CVLAN Ports

			Enabled Disabled
Unencrypted TCP Port	9999		<input checked="" type="radio"/> <input type="radio"/>
Encrypted TCP Port	<input type="text" value="9998"/>		<input checked="" type="radio"/> <input type="radio"/>

DLG Port

TCP Port	
5678	

TSAPI Ports

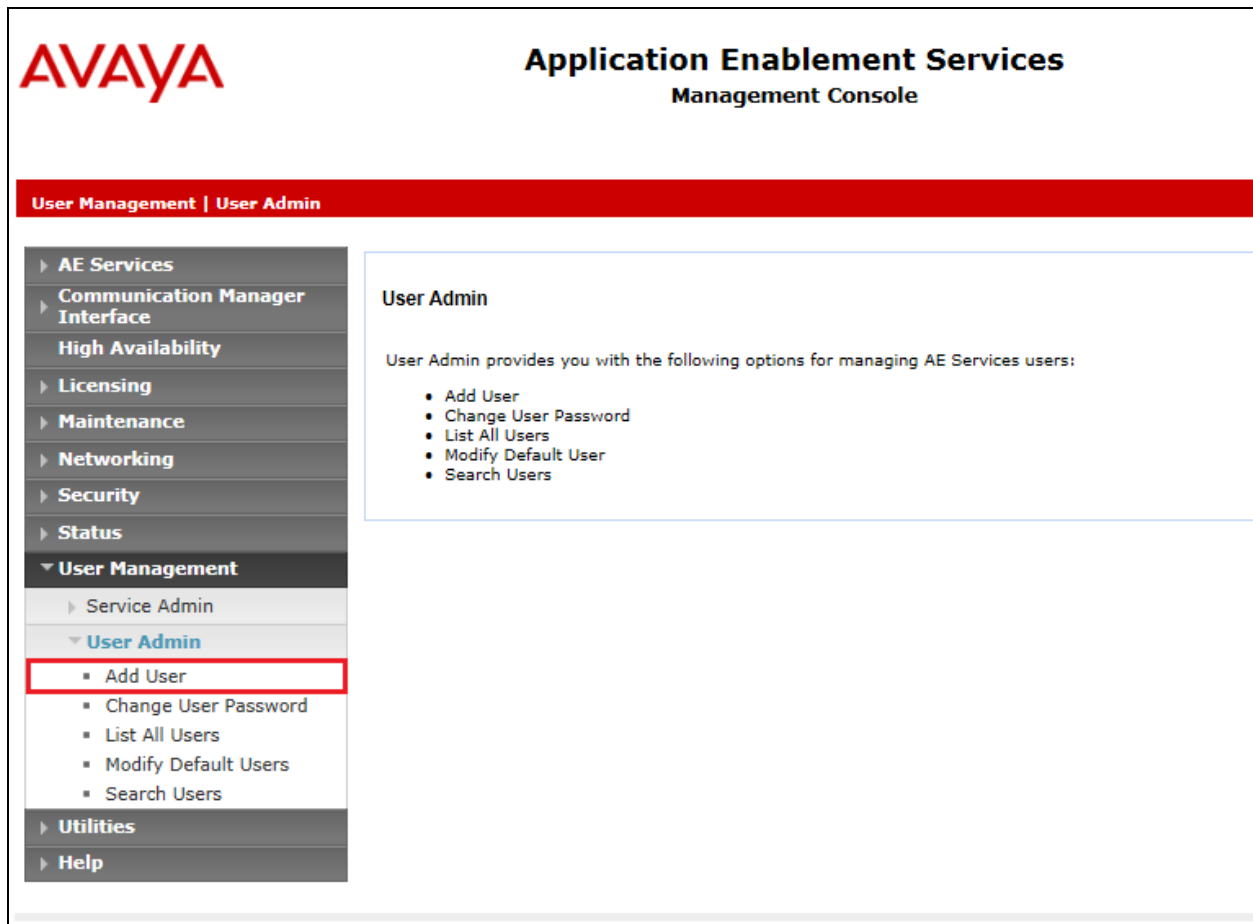
		Enabled Disabled
TSAPI Service Port	450	<input checked="" type="radio"/> <input type="radio"/>
Local TLINK Ports		
TCP Port Min	1024	
TCP Port Max	1039	
Unencrypted TLINK Ports		
TCP Port Min	<input type="text" value="1050"/>	
TCP Port Max	<input type="text" value="1065"/>	
Encrypted TLINK Ports		
TCP Port Min	<input type="text" value="1066"/>	
TCP Port Max	<input type="text" value="1081"/>	

DMCC Server Ports

		Enabled Disabled
Unencrypted Port	<input type="text" value="4721"/>	<input checked="" type="radio"/> <input type="radio"/>
Encrypted Port	<input type="text" value="4722"/>	<input checked="" type="radio"/> <input type="radio"/>
TR/87 Port	<input type="text" value="4723"/>	<input checked="" type="radio"/> <input type="radio"/>

6.6. Create CTI User

A User ID and password needs to be configured for SpeechLog Voice Recorder to communicate with the Application Enablement Services server. Navigate to the **User Management** → **User Admin** screen then choose the **Add User** option.



In the **Add User** screen shown below, enter the following values:

- **User Id** - This will be used by the SpeechLog Voice Recorder setup in **Section 7.1**.
- **Common Name** and **Surname** - Descriptive names need to be entered.
- **User Password** and **Confirm Password** - This will be used with the SpeechLog Voice Recorder setup in **Section 7.1**.
- **CT User** - Select **Yes** from the drop-down menu.

AVAYA Application Enablement Services Management Console

User Management | User Admin | Add User

Add User

Fields marked with * can not be empty.

* User Id

* Common Name

* Surname

* User Password

* Confirm Password

Admin Note

Avaya Role

Business Category

Car License

CM Home

Csm Home

CT User

Department Number

Display Name

Employee Number

Employee Type

Scroll down and click on **Apply Changes**.

Employee Number

Employee Type

Enterprise Handle

Given Name

Home Phone

Home Postal Address

Initials

Labeled URI

Mail

MM Home

Mobile

Organization

Pager

Preferred Language

Room Number

Telephone Number

Apply Changes

6.7. Associate Devices with CTI User

Navigate to **Security** → **Security Database** → **CTI Users** → **List All Users** and click on **Edit Users**.

The screenshot shows the Avaya Application Enablement Services Management Console. The left sidebar contains a navigation menu with categories: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, and Security. Under Security, the 'Security Database' is expanded, and 'CTI Users' is selected, with 'List All Users' highlighted. The main content area displays a table titled 'CTI Users' with columns: User ID, Common Name, Worktop Name, and Device ID. The table contains two rows: one for 'Global' and one for 'myapp'. Below the table are 'Edit' and 'List All' buttons. The top right corner shows system information including the last login time and server status.

User ID	Common Name	Worktop Name	Device ID
Global	Global	NONE	NONE
myapp	myapp	NONE	NONE

In the main window ensure that **Unrestricted Access** is ticked. Once this is done click on **Apply Changes**.

The screenshot shows the 'Edit CTI User' page in the Avaya Application Enablement Services Management Console. The left sidebar is the same as the previous screenshot, with 'List All Users' highlighted. The main content area is titled 'Edit CTI User' and contains a form for editing user profile and permissions. The 'User Profile' section includes fields for User ID, Common Name, and Worktop Name, all set to 'Global'. The 'Unrestricted Access' checkbox is checked. The 'Call and Device Control' section has a dropdown for 'Call Origination/Termination and Device Status' set to 'None'. The 'Call and Device Monitoring' section has dropdowns for 'Device Monitoring' and 'Calls On A Device Monitoring', both set to 'None', and an unchecked checkbox for 'Call Monitoring'. The 'Routing Control' section has a dropdown for 'Allow Routing on Listed Devices' set to 'None'. At the bottom, there are 'Apply Changes' and 'Cancel Changes' buttons.

User Profile:	
User ID	Global
Common Name	Global
Worktop Name	NONE
Unrestricted Access	<input checked="" type="checkbox"/>

Call and Device Control:	
Call Origination/Termination and Device Status	None

Call and Device Monitoring:	
Device Monitoring	None
Calls On A Device Monitoring	None
Call Monitoring	<input type="checkbox"/>

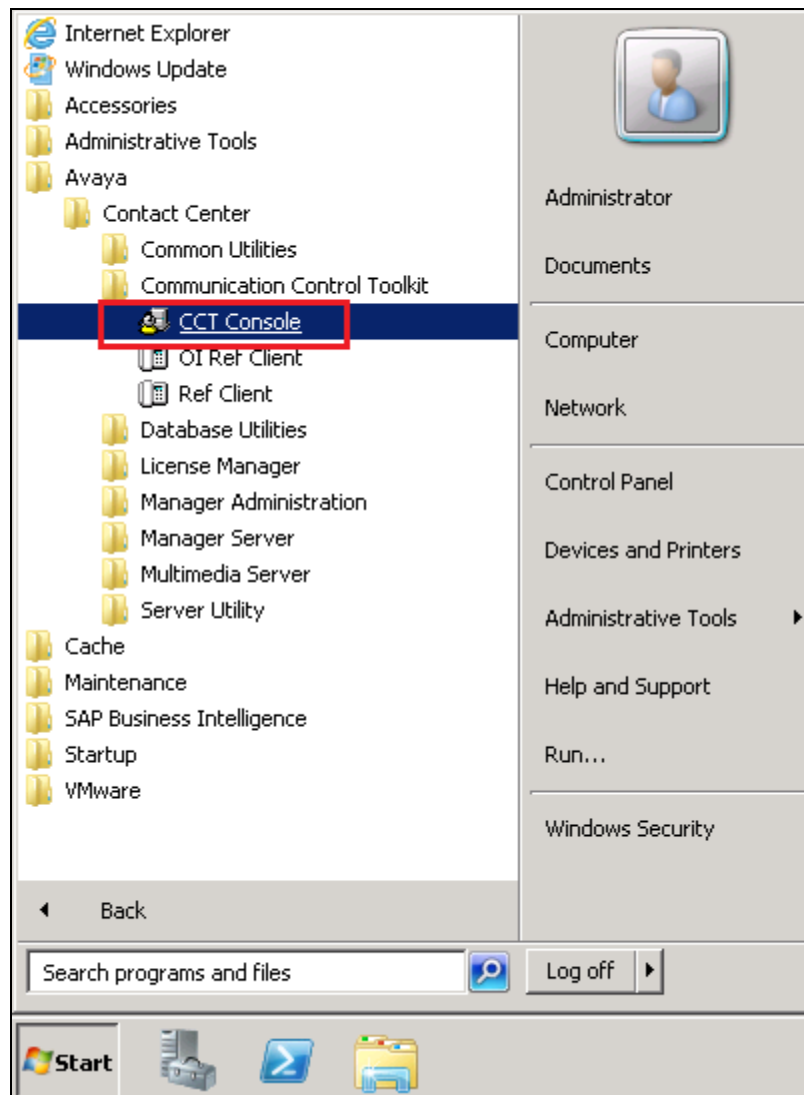
Routing Control:	
Allow Routing on Listed Devices	None

7. Configure Avaya Aura® Contact Center

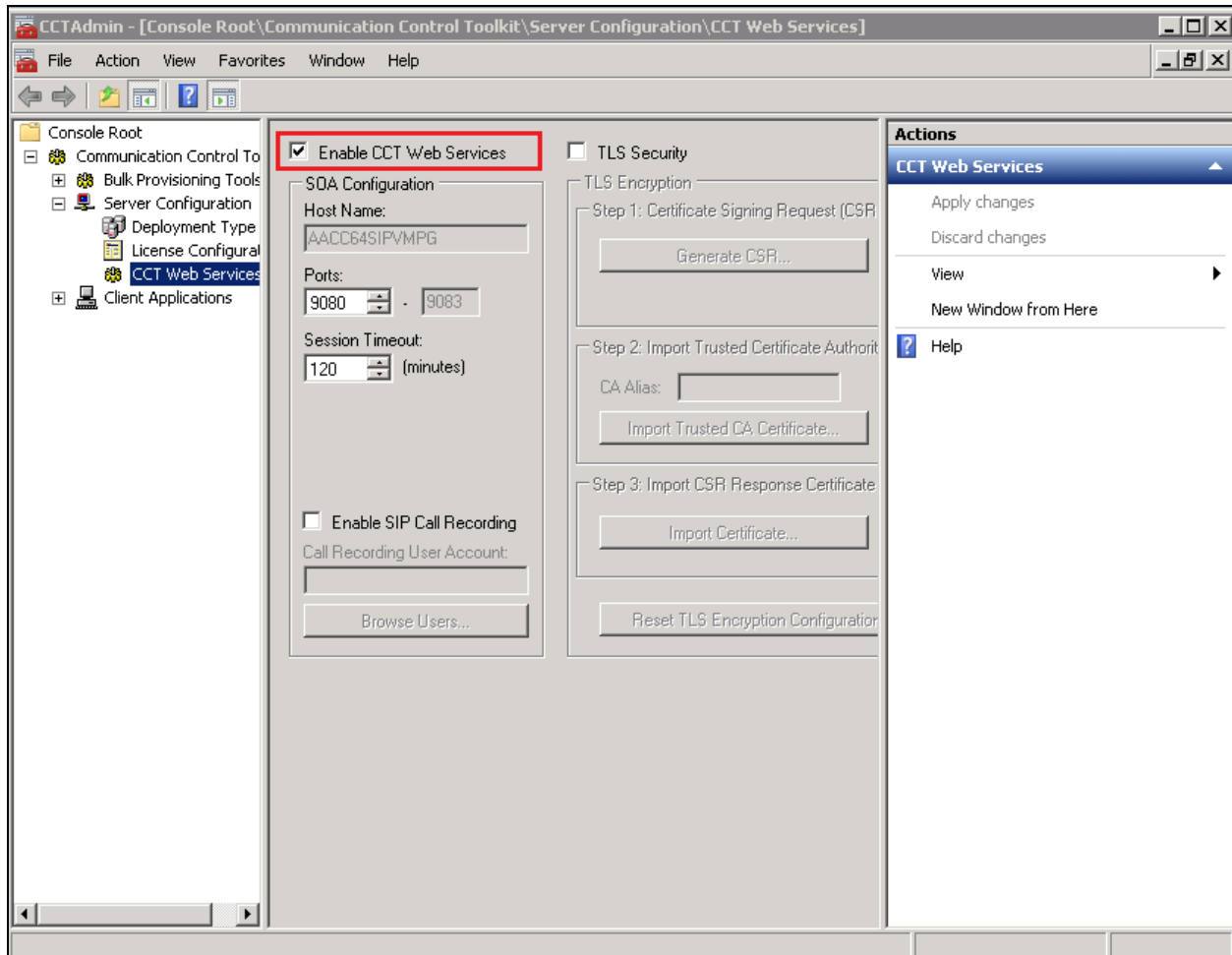
Web Services need to be running on the Contact Center server and this along with a CCT user is what is required for SpeechLog Voice Recorder to connect to Contact Center in order to monitor the agent's calls. This user is created in CCT and the agents that are to be monitored are associated with that new CCT user.

7.1. Verify Web Services are running on Avaya Aura® Contact Center

From the server open the CCT Console by navigating to Start → Programs → Avaya →

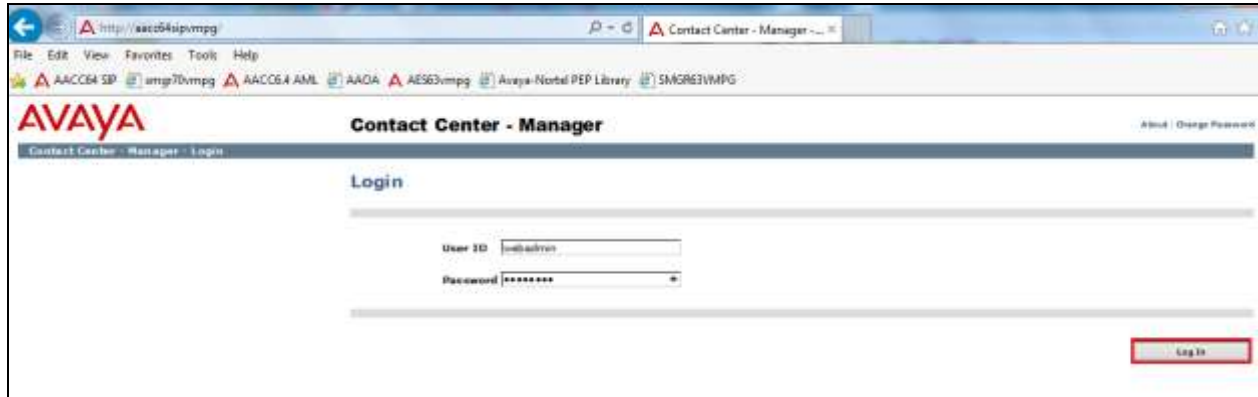


Ensure that **Enable CCT Web Services** is ticked, if not tick this and close the window. Upon closing the option to save this will be presented (not shown below).



7.2. Create a CCT user

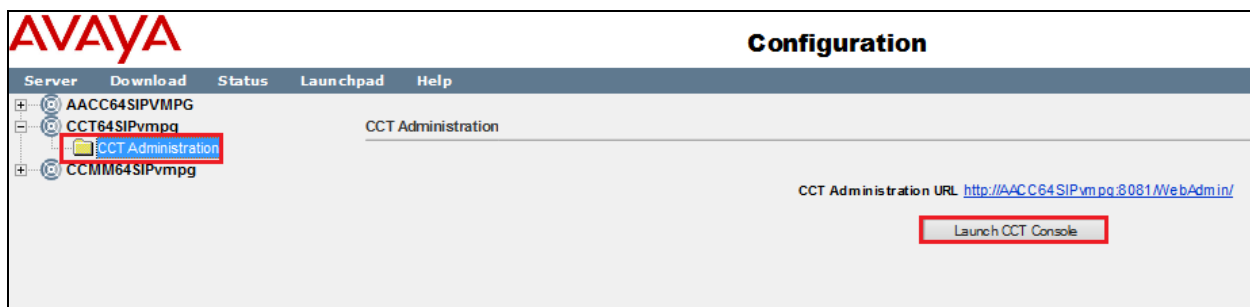
Open a browser session to the Contact Center server, enter the appropriate credentials and click on **Log In**.



Once logged in select Configuration.



Open the CCT server from the left window as shown and click on **Launch CCT Console** from the main window.



Right click on **Users** in the left window and click on **Add new User**.

AVAYA

CCT Administration

Users

Work

Groups

Providers

View Details

Add new User

CCT Users

Login User Name	First Name	Last Name	
AAC C64SIP\VMPG\2016	AMC Agent	2016	<input type="checkbox"/>
AAC C64SIP\VMPG\1000	AMC Agent	1000	<input type="checkbox"/>
AAC C64SIP\VMPG\7000	Agent	7000	<input type="checkbox"/>
AAC C64SIP\VMPG\7001	Agent	7001	<input type="checkbox"/>
AAC C64SIP\VMPG\Globitel	Globitel	User	<input type="checkbox"/>
AAC C64SIP\VMPG\7100	7100	7100	<input type="checkbox"/>
AAC C64SIP\VMPG\7101	7101	SIP7101Phone	<input type="checkbox"/>

7 CCT Users found, displaying 7 CCT Users. Page 1 / 1

Delete

Enter the following details under **User Details**:

- **Login User Name** This will be in the format Domain\Username.
- **First Name** Enter a suitable first name.
- **Last Name** Enter a suitable last name.

Under **Agent Assignments** select the agents that are to be monitored. In the example below agents **7000**, **7001**, **7100** and **7101** are all being monitored.

AVAYA CCT Administration

Update CCT User

User Details

Login User Name: AACC645IPVMPG\Globtel
First Name: Globtel
Last Name: User

Agent Assignments

Agents available

Agents
2016
1000

2 Agents found. Page 1 / 1

Agents mapped

Agents
7000
7001
7100
7101

4 Agents found. Page 1 / 1

Save

8. Configure Globitel SpeechLog Voice Recorder

The installation of Globitel SpeechLog Voice Recorder is usually carried out by an engineer from Globitel and is outside the scope of these Application Notes. For information on the installation of SpeechLog Voice Recorder contact Globitel as per the information provided in **Section 2.3**.

The following sections outline the process involved in connecting SpeechLog Voice Recorder to the Avaya solution. All configuration of SpeechLog Voice Recorder for a connection with the AES is performed using **Configuration Manager** located on the SpeechLog Voice Recorder server. From the SpeechLog Voice Recorder server open Configuration Manager, this will bring up a log in window, enter the appropriate credentials and click on **OK**.



8.1. Configure connection to Avaya Aura® Application Enablement Services and Avaya Aura® Contact Center

Click on the **General** tab, the **VoIP Settings** should be set as follows:

- **VoIP Signaling Type** Set to **Avaya CCT**.
- **Local Management IP** Set to the IP address of the SpeechLog Voice Recorder server.

The other settings can be left as default.

The screenshot shows the 'Speechlog Configuration Manager' window. The 'General' tab is selected and highlighted with a red box. Within this tab, the 'VoIP Settings' section is also highlighted with a red box. The settings are as follows:

Section	Setting	Value
System Settings	Storage Path	C:\Program Files (x86)\Globitel\Calls
	Recording Format	GSM
	Minimum Call Duration (Sec)	1
	Maximum Call Duration (Min)	60
Analog Settings	Board Mode	SynWay
	Maximum Silence (Sec)	5
Digital Settings	Board Mode	SynWay
	PBX Audio Format	aLaw
E1 Settings	Board Mode	SynWay
	Standard Type	ISDN
VoIP Settings	VoIP Signaling Type	Avaya CCT
	Silence Suppression	On
	Local Management IP	10.10.40.58
	VoIP Recording Type	Extension
	License Type	MAC Address
	RTP Match Mode	IP Address
Enable RTP Active Mode		<input checked="" type="checkbox"/>
LAN Adapters		\Device\NPF_{C27C5F45-52AA-4FC3-964F-5BB144A78145}

Buttons at the bottom: Save, Exit

Click on the **Avaya AES** tab and **Add** a new **AES Interface**.

The screenshot shows the 'Speechlog Configuration Manager' application window. The 'Avaya AES' tab is selected and highlighted with a red box. The window contains three main sections: 'AES Interface', 'CM Interface', and 'TSAPI Servers Settings'. The 'AES Interface' section has a table with columns: AESInterface, AESPort, ApplicationName, UserName, UserPassword, and Pr. Below the table are 'Add', 'Update', and 'Delete' buttons, with the 'Add' button highlighted by a red box. The 'CM Interface' section has a table with columns: SwitchName, SwitchIP, and SQCode, and 'Add', 'Update', and 'Delete' buttons. The 'TSAPI Servers Settings' section has two sub-sections: 'Primary TSAPI Server Settings' and 'Secondary TSAPI Server Settings', each with fields for 'Server Name', 'Login ID', and 'Password'. At the bottom of the window are 'Save' and 'Exit' buttons.

Speechlog Configuration Manager

Ports Tools Help

General Integrations Codes Detection Recording Option **Avaya AES** Avaya CCT CSTA

AES Interface

AESInterface	AESPort	ApplicationName	UserName	UserPassword	Pr
--------------	---------	-----------------	----------	--------------	----

◀ ▶

Add Update Delete

CM Interface

SwitchName	SwitchIP	SQCode
------------	----------	--------

Add Update Delete

TSAPI Servers Settings

Primary TSAPI Server Settings:

Server Name :

Login ID :

Password :

Secondary TSAPI Server Settings:

Server Name :

Login ID :

Password :

Save Exit

Enter the AES IP address for the **AES IP Interface** and the **AES Port** which should be the DMCC unsecured port number as shown in **Section 6.5**. The **User Name** and **User Password** is that which was configured in **Section 6.6**. The **Application Name** is the Tlink also secured from **Section 6.4**. The **Protocol Version** should be set to **4.2** and the tick boxes shown below should be ticked. Click on **OK** to continue.

Speechlog Configuration Manager

Ports Tools Help

General Integrations Codes Detection Recording Option **Avaya AES** Avaya CCT CSTA

AES Entry

Primary AES Interface

AES IP Interface 10.10.40.16

AES Port 4721

User Name Globitel

User Password Avaya123\$

Secondary AES Interface (Optional)

AES IP Interface

AES Port 1235

User Name 2

User Password 2

Application Name AVAYA#CM70VMPG#CSTA#AES

Protocol Version 4.2

☐ Is Secure

☒ Is Start Auto Keep Alive

☒ Is Allow Certificate Hostname Mismatch

OK Cancel

Password : globitel

Save Exit

A new Communication Manager interface also needs to be added. Click on **Add** under the **CM Interface** section.

The screenshot shows the 'Speechlog Configuration Manager' application window. The 'Avaya AES' tab is selected and highlighted with a red box. The window contains three main sections: 'AES Interface', 'CM Interface', and 'TSAPI Servers Settings'. The 'AES Interface' section has a table with one entry and 'Add', 'Update', and 'Delete' buttons. The 'CM Interface' section has an empty table and 'Add', 'Update', and 'Delete' buttons, with the 'Add' button highlighted by a red box. The 'TSAPI Servers Settings' section has fields for 'Primary TSAPI Server Settings' and 'Secondary TSAPI Server Settings', each with 'Server Name', 'Login ID', and 'Password' fields.

AESInterface	AESPort	ApplicationName	UserName	UserPassword	Pr
10.10.40.16	4721	AVAYA#CM70VMPG#0	Globitel	Avaya123\$	4.0

SwitchName	SwitchIP	SOCode
------------	----------	--------

TSAPI Servers Settings

Primary TSAPI Server Settings:

Server Name :

Login ID :

Password :

Secondary TSAPI Server Settings:

Server Name :

Login ID :

Password :

Save **Exit**

Enter the name of the Communication Manager for **Switch Name** and the IP Address of the Communication Manager server for **Switch IP**. The **SO Code** is the Service Observe feature access code that was configured in **Section 5.5**. Click on **OK** to continue.

The screenshot shows the 'Speechlog Configuration Manager' application window. The 'Avaya AES' tab is selected. The 'AES Interface' section contains a table with the following data:

AESInterface	AESPort	ApplicationName	UserName	UserPassword	Pr
10.10.40.16	4721	AVAYA#CM70VMPG#C	Globitel	Avaya123\$	4.3

Below the table are 'Add', 'Update', and 'Delete' buttons. A 'CM Entry' dialog box is open in the foreground, containing the following fields:

- Switch Name: cm70vmpg
- Switch IP: 10.10.40.13
- SO Code: #43

The 'OK' button in the 'CM Entry' dialog is highlighted with a red rectangle. Below the dialog, there are fields for 'Password', 'Secondary TSAPI Server Settings' (Server Name, Login ID, Password), and 'Save' and 'Exit' buttons at the bottom.

The **TSAPI Servers Settings** must now be completed, enter the Tlink information obtained from **Section 6.4** as the **Server Name**, the **Login ID** and **Password** are that of the AES user created in **Section 6.6**. Click on **Save** to save all the information just entered in this section.

The screenshot shows the 'Speechlog Configuration Manager' window with the 'Avaya AES' tab selected. The interface includes a menu bar (Ports, Tools, Help) and several configuration sections.

AES Interface

AESInterface	AESPort	ApplicationName	UserName	UserPassword	Pr
▶ 10.10.40.16	4721	AVAYA#CM70VMPG#C	Globitel	Avaya123\$	4.3

Buttons: Add, Update, Delete

CM Interface

SwitchName	SwitchIP	SOCode
▶ cm70vmpg	10.10.40.13	#43

Buttons: Add, Update, Delete

TSAPI Servers Settings

Primary TSAPI Server Settings:

Server Name :	AVAYA#CM70VMPG#CSTA#AES70VMPG
Login ID :	Globitel
Password :	Avaya123\$

Secondary TSAPI Server Settings:

Server Name :	
Login ID :	
Password :	

Buttons: Save, Exit

Click on the **Avaya CCT** tab. Enter the following credentials:

- **CCT Server IP** Enter the IP Address of the Avaya Contact Center Server which is hosting the CCT module.
- **CCT Server Port** This is hardcoded as **29373**.
- **Username** This is the username of the CCT user created in **Section 7.0**.
- **Password** This is the password for the user above.
- **Domain** This is the domain to which the Contact Center belongs to or the name of the Contact Center server if no domain is being used.

Click on **Save** to continue.

The screenshot shows the 'Speechlog Configuration Manager' application window. The 'Avaya CCT' tab is selected and highlighted with a red box. The window contains two main sections: 'CCT Server Settings' and 'CCT User'. In the 'CCT Server Settings' section, the 'CCT Server IP' is set to '10.10.40.55', 'CCT Server Port' is '29373', and 'CCT Alternate Server' and 'CCT Geographic Server' are empty. In the 'CCT User' section, the 'Username' is 'Globitel', 'Password' is masked with 'xxxxxxx', and 'Domain' is 'aacc64SIPvmg'. At the bottom of the window, the 'Save' button is highlighted with a red box, and the 'Exit' button is also visible.

Section	Field	Value
CCT Server Settings	CCT Server IP	10.10.40.55
	CCT Server Port	29373
	CCT Alternate Server	
	CCT Geographic Server	
CCT User	Username	Globitel
	Password	xxxxxxx
	Domain	aacc64SIPvmg

8.2. Configure Extensions to be recorded

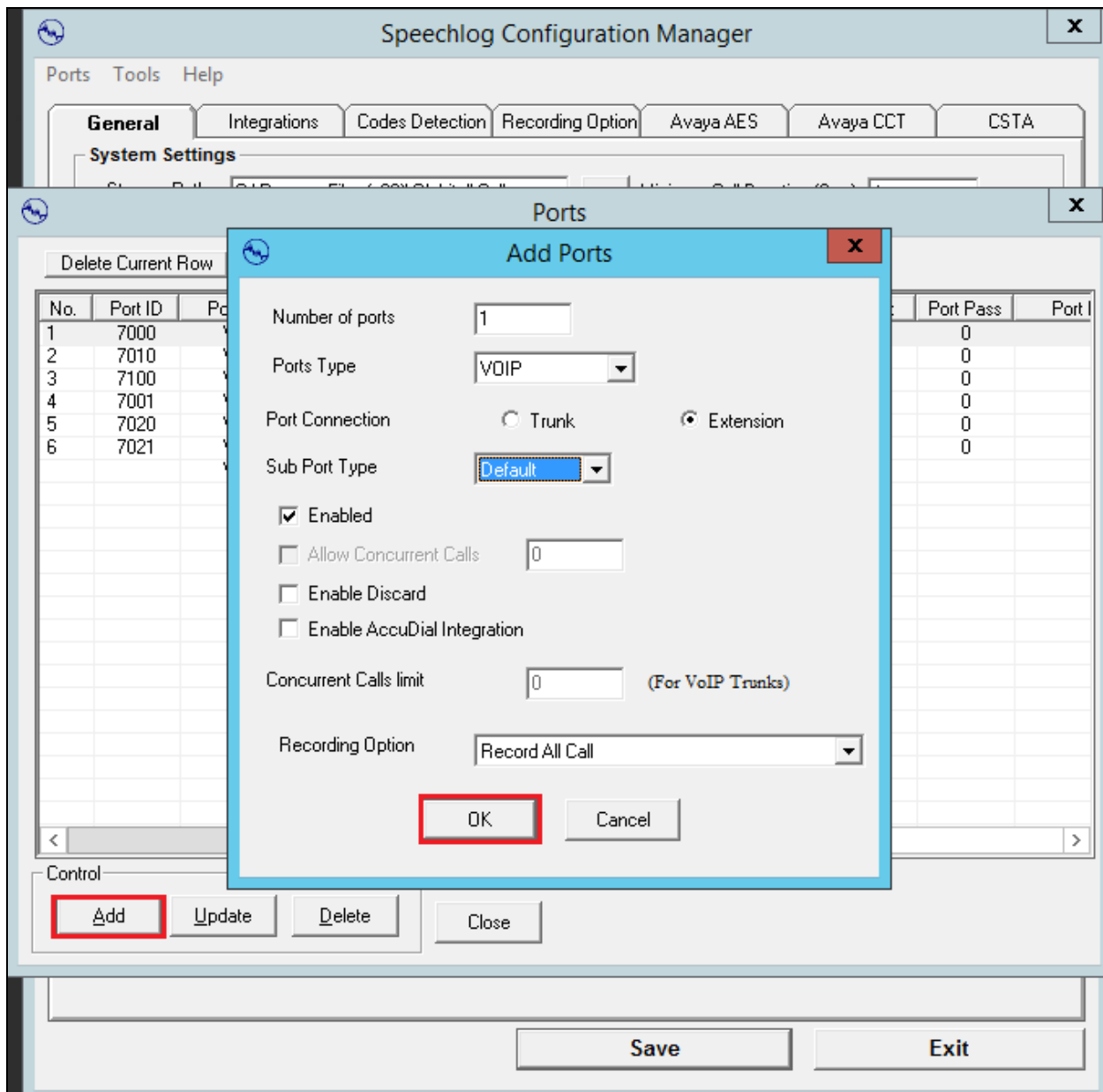
From the top left of the page click on **Ports** → **Extension Ports**.

The screenshot shows the 'Speechlog Configuration Manager' window. The 'Ports' tab is selected, and the 'Extension Ports' sub-tab is highlighted with a red box. The configuration is organized into several sections:

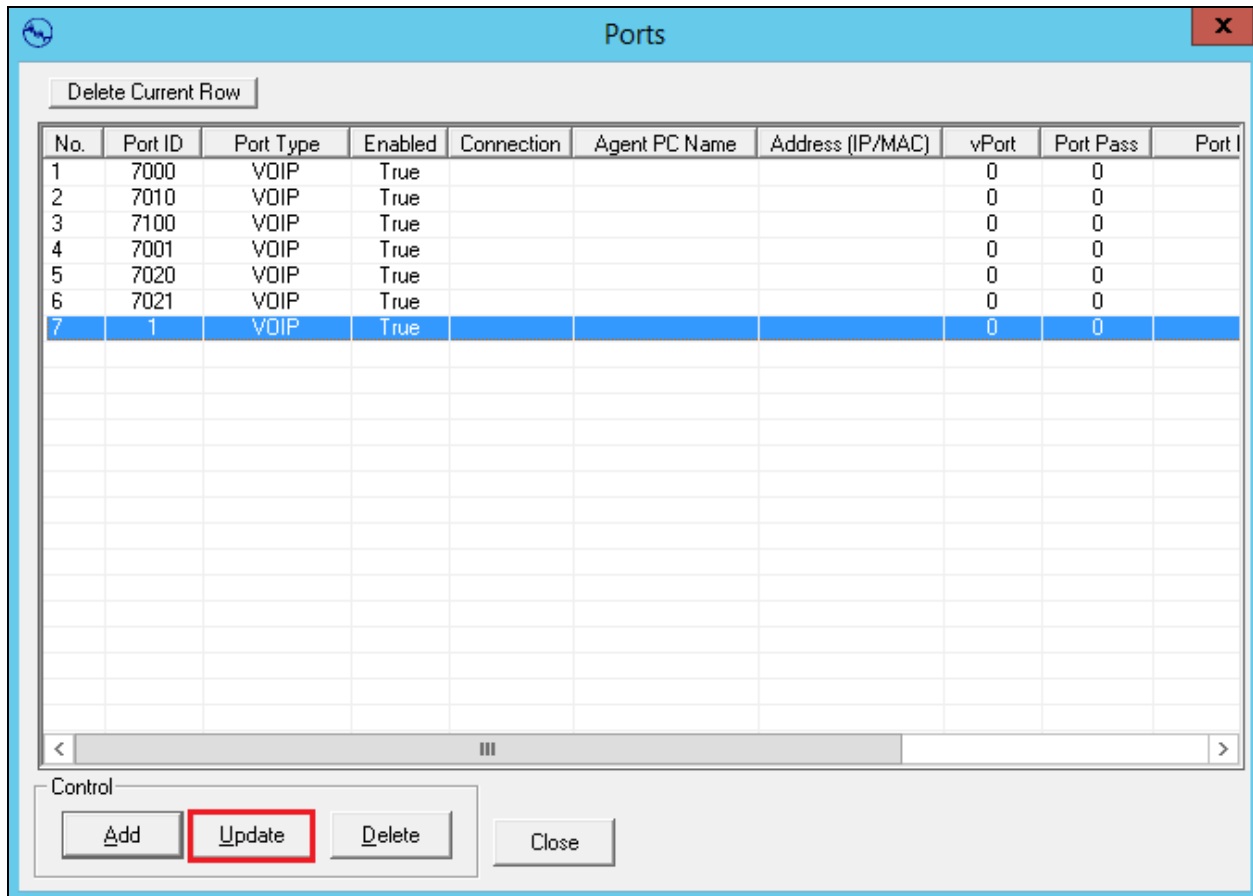
- General Settings:**
 - Storage Path: C:\Program Files (x86)\Globitel\Calls
 - Recording Format: GSM
 - Minimum Call Duration (Sec): 1
 - Maximum Call Duration (Min): 60
 - ☒ Apply Minimum Duration Constraint for Outgoing Calls Only.
 - ☐ Encrypt Audio Files
- Analog Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD, ☐ Pika
 - Maximum Silence (Sec): 5
 - Speech Threshold (dBm): -40
- Digital Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD
 - PBX Audio Format: aLaw
 - CTBus Type: H100
- E1 Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD, ☐ Pika
 - Standard Type: ISDN
- VoIP Settings:**
 - VoIP Signaling Type: Avaya CCT
 - VoIP Recording Type: Extension
 - Silence Suppression: On
 - License Type: MAC Address
 - Local Management IP: 10.10.40.58
 - RTP Match Mode: IP Address
 - ☒ Enable RTP Active Mode
 - LAN Adapters: \Device\NPF_{C27C5F45-52AA-4FC3-964F-5BB144A78145}

At the bottom of the window are 'Save' and 'Exit' buttons.

The example below shows the addition of one extension but note that a number of extensions would usually be added together. The **Ports** window is opened, click on **Add** at the bottom left of this window, this will bring up the **Add Ports** window as shown where the **Number of ports** to be added window is filled in. The **Ports Type** should be **VOIP** and the **Port Connection** is **Extension**. The **Sub Port Type** can be left as **Default** and the **Enabled** box should be ticked. For compliance testing the **Recording Option** was set to **Record All Call**. Click on **OK** to add this one port.



Highlight the new extension and click on **Update**.



The screenshot shows a window titled "Ports" with a close button (X) in the top right corner. Below the title bar is a button labeled "Delete Current Row". The main area contains a table with the following columns: No., Port ID, Port Type, Enabled, Connection, Agent PC Name, Address (IP/MAC), vPort, Port Pass, and Port I. The table has 7 rows of data, with the 7th row highlighted in blue. Below the table is a horizontal scrollbar. At the bottom of the window is a "Control" panel with four buttons: "Add", "Update", "Delete", and "Close". The "Update" button is highlighted with a red rectangular box.

No.	Port ID	Port Type	Enabled	Connection	Agent PC Name	Address (IP/MAC)	vPort	Port Pass	Port I
1	7000	VOIP	True				0	0	
2	7010	VOIP	True				0	0	
3	7100	VOIP	True				0	0	
4	7001	VOIP	True				0	0	
5	7020	VOIP	True				0	0	
6	7021	VOIP	True				0	0	
7	1	VOIP	True				0	0	

Control

Add Update Delete Close

Enter the following information:

- **Port ID** The extension number of the phoneset to be recorded.
- **Port Connection** **Extension.**
- **Recording Option** **Record All Call.**
- **Enabled Box** Ticked.
- **AES Interface** The IP address of the AES, which is the AES Interface configured in **Section 5.2**
- **CM Interface** The Communication Manager Interface that was configured in **Section 6.2**

Click on **OK** to continue.

Update Port

Port 7-VOIP

Port ID: 7050

Agent PC Name:

Address (IP/MAC):

Port Connection: Extension

Recording Option: Record All Call

Concurrent Calls limit: 0 (For H323 Trunks)

SubPort Type: Default

☒ Enabled ☐ Enable Discard

☐ Enable AccuDial Integration

Synway config file: 0

AES Interface: 10.10.40.16

CM Interface: cm70vmpg

AES Settings

Port Password: 0

Port Name:

OK Cancel

With this new extension updated, click on **Close** to continue.

The screenshot shows a window titled "Ports" with a close button (X) in the top right corner. Below the title bar is a button labeled "Delete Current Row". The main area contains a table with the following columns: No., Port ID, Port Type, Enabled, Connection, Agent PC Name, Address (IP/MAC), vPort, Port Pass, and Port I. The table has 7 rows of data, with the 7th row highlighted by a red border. Below the table is a horizontal scrollbar. At the bottom of the window is a "Control" panel with four buttons: "Add", "Update", "Delete", and "Close". The "Close" button is highlighted with a red box.

No.	Port ID	Port Type	Enabled	Connection	Agent PC Name	Address (IP/MAC)	vPort	Port Pass	Port I
1	7000	VOIP	True				0	0	
2	7010	VOIP	True				0	0	
3	7100	VOIP	True				0	0	
4	7001	VOIP	True				0	0	
5	7020	VOIP	True				0	0	
6	7021	VOIP	True				0	0	
7	7050	VOIP	True				0	0	

Control

Add Update Delete Close

8.3. Configure Virtual Ports

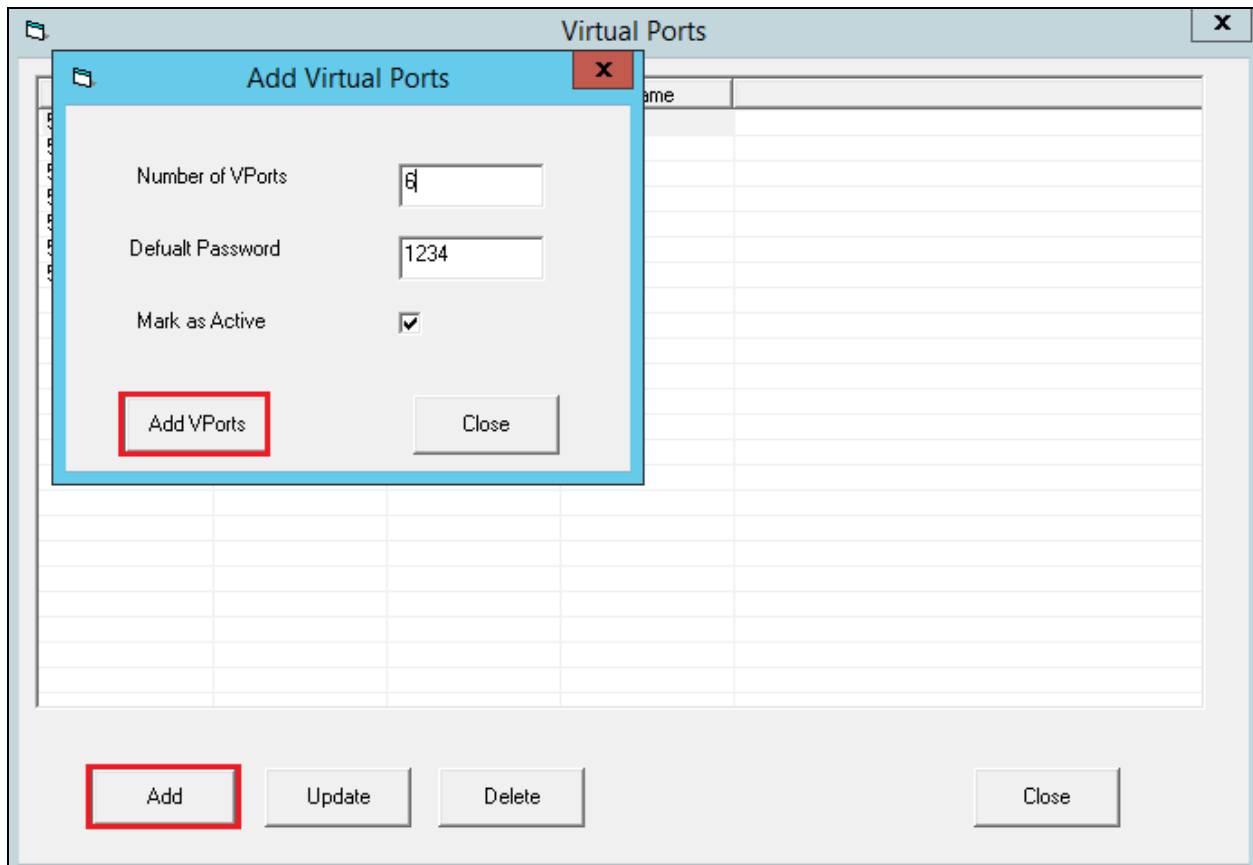
From the top left of the window click on **Ports** → **Virtual Ports**.

The screenshot shows the 'Speechlog Configuration Manager' window with the 'Ports' tab selected. The 'Virtual Ports' sub-tab is highlighted with a red box. The configuration is organized into several sections:

- Storage Path:** C:\Program Files (x86)\Globitel\Calls
- Recording Format:** GSM
- Minimum Call Duration (Sec):** 1
- Maximum Call Duration (Min):** 60
- ☒ Apply Minimum Duration Constraint for Outgoing Calls Only.
- ☐ Encrypt Audio Files
- Analog Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD, ☐ Pika
 - Maximum Silence (Sec): 5
 - Speech Threshold (dBm): -40
- Digital Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD
 - PBX Audio Format: aLaw
 - CTBus Type: H100
- E1 Settings:**
 - Board Mode: ☒ SynWay, ☐ TOD, ☐ Pika
 - Standard Type: ISDN
- VoIP Settings:**
 - VoIP Signaling Type: Avaya CCT
 - VoIP Recording Type: Extension
 - Silence Suppression: On
 - License Type: MAC Address
 - Local Management IP: 10.10.40.58
 - RTP Match Mode: IP Address
 - ☒ Enable RTP Active Mode
 - LAN Adapters: \Device\NPF_{C27C5F45-52AA-4FC3-964F-5BB144A78145}

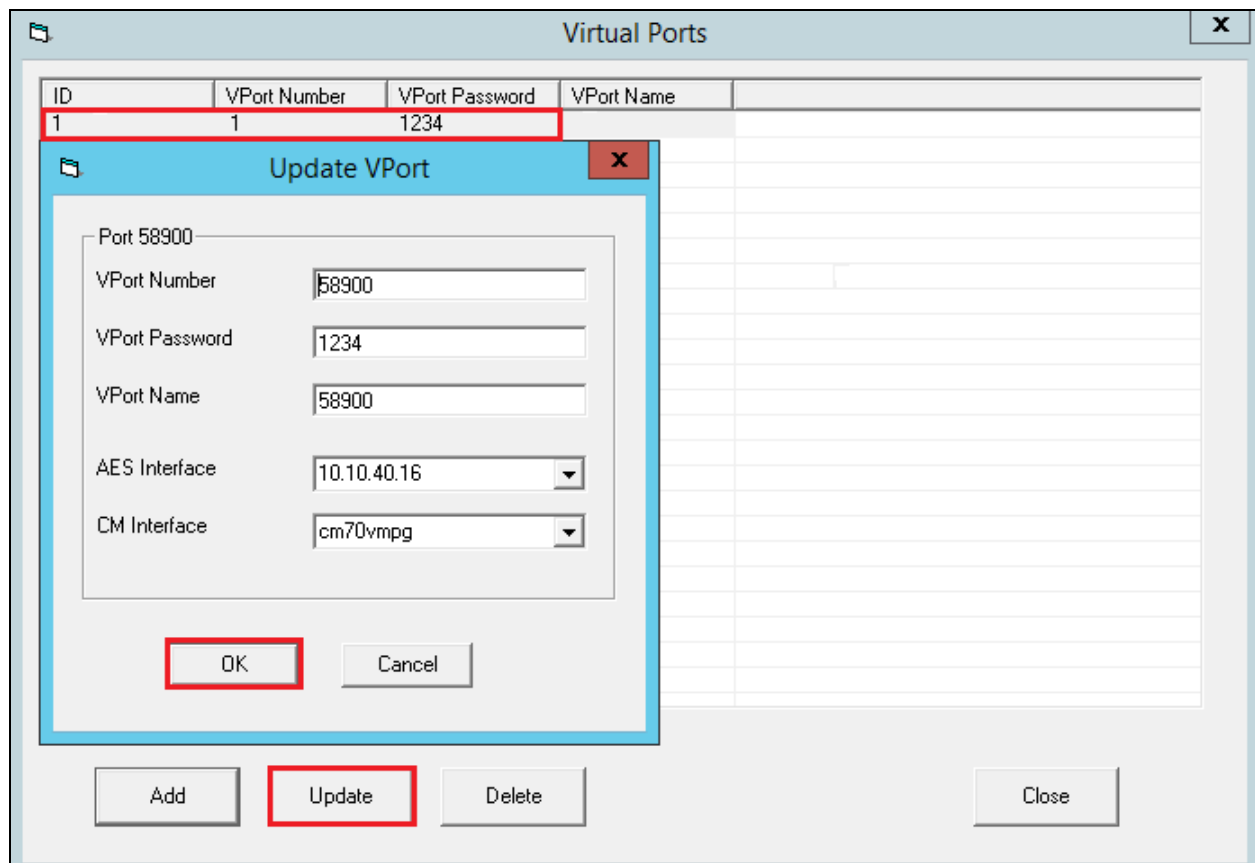
At the bottom of the window are 'Save' and 'Exit' buttons.

From the **Virtual Ports** window, click on **Add** (not shown). Enter the **Number of VPorts**, in the example below **6** virtual ports were added with a **Default Password** of **1234**, this being the password of the virtual stations created in **Section 5.8**. Click on **Add VPorts**.



Highlight the new port and click on **Update**, this opens the **Update VPort** window where the following needs to be configured.

- **VPort Number** Virtual extension number from **Section 5.8**.
- **VPort Password** The password of the virtual extension created in **section 5.8**.
- **VPort Name** Any Suitable name will do, it's recommended to keep this the same as the extension number for convenience.
- **AES Interface** The IP address of the AES, which is the AES Interface configured in **Section 5.2**
- **CM Interface** The Communication Manager Interface that was configured in **Section 6.2**



When all the virtual ports are configured click on **Close** to continue.

[illegible]

Click on **Save** at the bottom of the screen.

Speechlog Configuration Manager

Ports Tools Help

General Integrations Codes Detection Recording Option Avaya AES Avaya CCT CSTA

System Settings

Storage Path ... Minimum Call Duration (Sec)

Recording Format Maximum Call Duration (Min)

☒ Apply Minimum Duration Constraint for Outgoing Calls Only. ☐ Encrypt Audio Files

Analog Settings

Board Mode ☒ SynWay ☐ TOD ☐ Pika Maximum Silence (Sec)

Speech Threshold (dBm)

Digital Settings

Board Mode ☒ SynWay ☐ TOD PBX Audio Format

CTBus Type

E1 Settings

Board Mode ☒ SynWay ☐ TOD ☐ Pika Standard Type:

VoIP Settings

VoIP Signaling Type VoIP Recording Type

Silence Suppression License Type

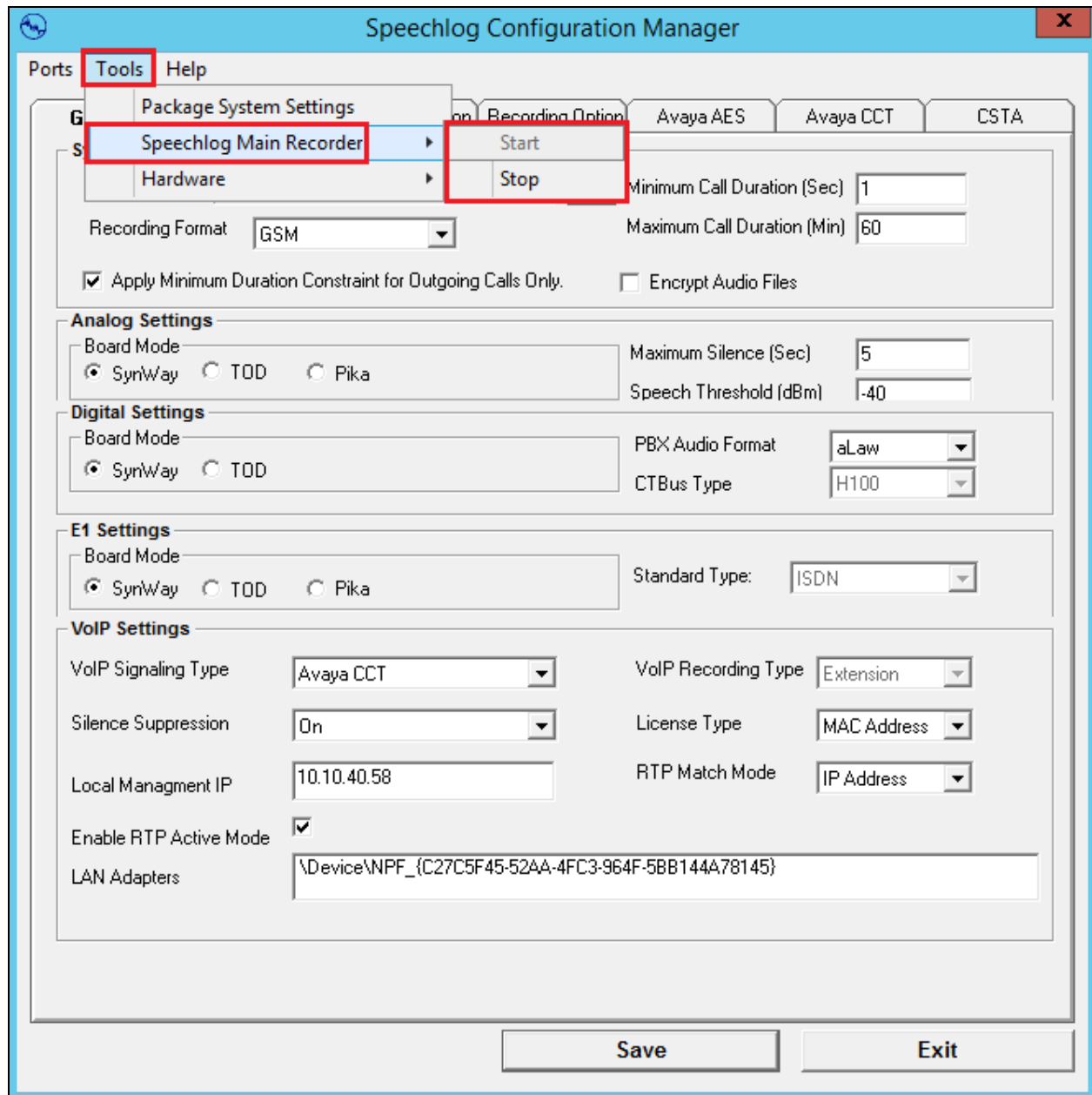
Local Management IP RTP Match Mode

Enable RTP Active Mode ☒

LAN Adapters

Save **Exit**

In order to complete the setup and place the recorder into service, the main recorder service should be stopped and started. Note that this may already be in a stopped state, if so simply start the service. If stopping and starting the service allow a minute between these actions in order to allow all services to stop correctly before starting them back up again.



This concludes the setup of the SpeechLog Voice Recorder Server for a connection to CCT and AES for DMCC Service Observe recording.

9. Verification Steps

This section provides the steps that can be taken to verify correct configuration of Globitel's SpeechLog Voice Recorder and Avaya Aura® Application Enablement Services.

9.1. Verify Avaya Aura® Communication Manager CTI Service State

Before the connection between SpeechLog Voice Recorder and the AES is checked, the connection between Communication Manager and AES can be checked to ensure it is functioning correctly. Check the AESVCS link status by using the command **status aesvcs cti-link**. Verify the **Service State** of the CTI link is **established**.

status aesvcs cti-link						
AE SERVICES CTI LINK STATUS						
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd
1	4	no	aes70vmpg	established	18	18

9.2. Verify TSAPI link on AES

On the AES Management Console verify the status of the TSAPI link by selecting **Status → Status and Control → TSAPI Service Summary** to display the **TSAPI Link Details** screen. Verify the status of the TSAPI link by checking that the **Status** is **Talking** and the **State** is **Online**.

The screenshot shows the Avaya Application Enablement Services Management Console. The left sidebar contains a navigation menu with options like AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security, and Status. The main area displays the 'TSAPI Link Details' screen. At the top, there is a header with the Avaya logo and the title 'Application Enablement Services Management Console'. Below the header, there is a status bar indicating 'Status | Status and Control | TSAPI Service Summary'. The main content area shows a table with the following columns: Link, Switch Name, Switch CTI Link ID, Status, State, Switch Version, Associations, Msgs to Switch, Msgs from Switch, and Msgs Period. The table contains one row with the following data: Link 1, Switch Name cm70vmpg, Switch CTI Link ID 1, Status Talking, State Online, Switch Version 17, Associations 4, Msgs to Switch 15, Msgs from Switch 15, and Msgs Period 30. Below the table, there are buttons for 'Online' and 'Offline'. At the bottom, there is a section for 'For service-side information, choose one of the following:' with buttons for 'TSAPI Service Status', 'Link Status', and 'User Status'.

9.3. Verify DMCC link on AES

Verify the status of the DMCC link by selecting **Status** → **Status and Control** → **DMCC Service Summary** to display the **DMCC Service Summary – Session Summary** screen. The screen below shows that the user **Globitel** is connected from the IP address **10.10.40.58**, which is the SpeechLog Voice Recorder server.

The screenshot shows the Avaya Application Enablement Services Management Console. The left sidebar contains navigation links: All Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security, Status, Alarm Viewer, Log Manager, Logs, Status and Control, CUCM Service Summary, TCC Service Summary, DMCC Service Summary (highlighted), Switch Ldap Summary, TSPS Service Summary, User Management, Utilities, and Help. The main content area displays the 'DMCC Service Summary - Session Summary' page. It includes a summary section with statistics: Session Summary, Device Summary, Generated on Wed Nov 25 14:34:07 GMT 2015, Service (10000), 8 days, 4 hours, 12 minutes, Number of Active Sessions: 3, Number of Sessions Created Since Service Boot: 128, Number of Existing Devices: 7, and Number of Devices Created Since Service Boot: 189. Below this is a table with columns: Session ID, User, Application, Far-end Identifier, Connection Type, and # of Associated Devices. The table lists several sessions, with the first one showing a user 'Globitel' connected from IP '10.10.40.58' via 'A/VYA#CM72VM90#CSTA#AES70VM90' using 'OHL Unencrypted' connection type.

Session ID	User	Application	Far-end Identifier	Connection Type	# of Associated Devices
3165C55E4018A41E 08E50B78E350C4F-152	Globitel	A/VYA#CM72VM90#CSTA#AES70VM90	10.10.40.58	OHL Unencrypted	7
1F048D7A051F85E75 73D5FF6F02F720-146	wa/7000@ @connect.local	AACC	10.10.40.25:10.10.40.55	TR-67 Encrypted	1
2F5A84D9C4AA78E13 6CB04E863E3F011-153	wa/7001@ @connect.local	AACC	10.10.40.38:10.10.40.55	TR-67 Encrypted	1
0E4581C0D3D43091B A82CEDC15E4778C-148	wa/7000@ @connect.local	AACC	10.10.40.25:10.10.40.55	TR-67 Encrypted	1
5787D011BC47718EE 8224CE201917F5D-134	wa/7001@ @connect.local	AACC	10.10.40.25:10.10.40.55	TR-67 Encrypted	1

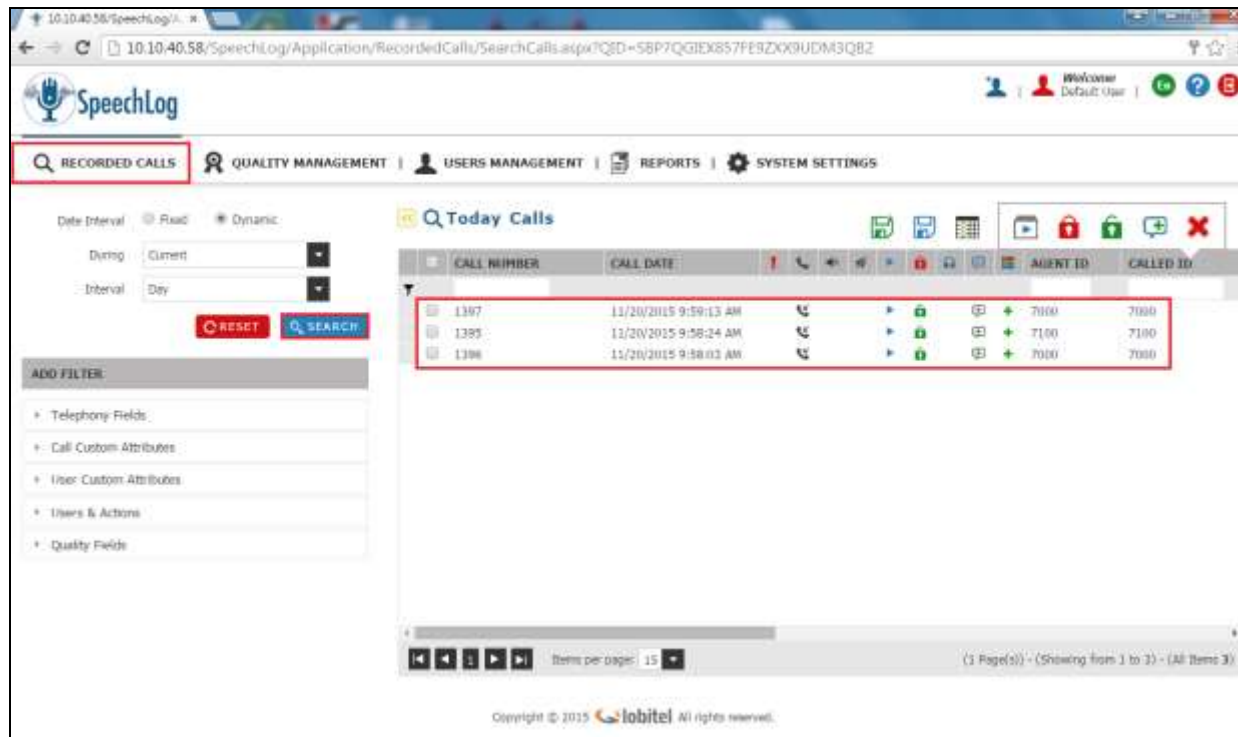
9.4. Verify calls are being recorded

From any of the monitored Avaya endpoints make a series of inbound and outbound calls. Once these calls are completed they should be available for playback through a web browser to the SpeechLog Voice Recorder server.

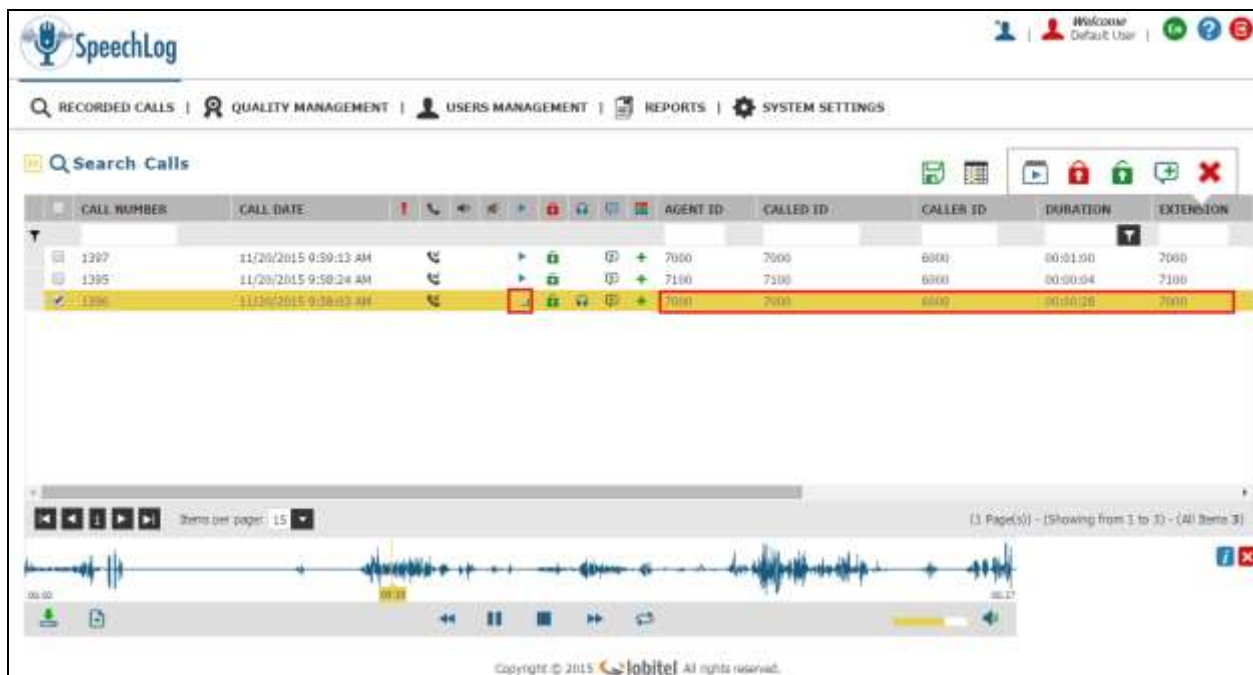
Open a browser session to the SpeechLog Voice Recorder server as is shown below. Enter the appropriate credentials and click on **Sign in**.



A list of calls should now be available for playback. If not a simple **Search** for calls on the day of recording should produce a list like the one shown below.



Click on the play icon of the call that is to be played back. This call should then be played back as shown below and should be audible through any connected speakers or headphones.



10. Conclusion

These Application Notes describe the configuration steps required for Globitel's SpeechLog Voice Recorder to successfully interoperate with Avaya Aura® Contact Center R6.4 connecting to Avaya Aura® Communication Manager R7.0 using Avaya Aura® Application Enablement Services R7.0 using DMCC Service Observe to record calls. All feature functionality and serviceability test cases were completed successfully with some issues and observations noted in **Section 2.2**.

11. Additional References

This section references the Avaya and Globitel product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] *Avaya Aura® Contact Center Configuration – Avaya Aura® Unified Communications Platform Integration* Release 6.4 44400-521 Issue 05.02
- [2] *Avaya Aura® Contact Center Server Administration* Release 6.4 44400-610 Issue 05.01
- [3] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
- [5] *Avaya Aura® Application Enablement Services Administration and Maintenance Guide* Release 7.0

Technical support can be obtained for Globitel SpeechLog Voice Recorder at:

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Khalda, Amman, Jordan.
support@globitel.com
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Phone: +962 (6) 5300 130
Fax: +962 (6) 5300 144
P.O. Box 1786 Amman 11821 Jordan

Appendix

Avaya 9608 H.323 Deskphone

This is a printout of the Avaya 9608 H.323 deskphone used during compliance testing.

display station 7000	Page 1 of 5	
STATION		
Extension: 7000	Lock Messages? n	BCC: 0
Type: 9608	Security Code: *	TN: 1
Port: S00000	Coverage Path 1: 1	COR: 1
Name: Ext7000	Coverage Path 2:	COS: 1
	Hunt-to Station:	Tests? y
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 7000	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english	Button Modules: 0	
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? y	
	IP Video Softphone? n	
	Short/Prefixed Registration Allowed: yes	
	Customizable Labels? y	

display station 7000	Page 2 of 5	
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		
	EMU Login Allowed?	n
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State:	enabled
Multimedia Mode: enhanced	Audible Message Waiting?	n
MWI Served User Type: sip-adjunct	Display Client Redirection?	n
	Select Last Used Appearance?	n
	Coverage After Forwarding?	s
	Multimedia Early Answer?	n
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections?	y
Emergency Location Ext: 7000	Always Use? n IP Audio Hairpinning?	n

```

display station 7000                                     Page 3 of 5
                                     STATION

      Conf/Trans on Primary Appearance? n
      Bridged Appearance Origination Restriction? n      Offline Call Logging? y
      Require Mutual Authentication if TLS? n

      Call Appearance Display Format: disp-param-default
      IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n

      ENHANCED CALL FORWARDING
      Forwarded Destination      Active
Unconditional For Internal Calls To: 7101                n
      External Calls To: 7101                n
      Busy For Internal Calls To:                n
      External Calls To:                n
      No Reply For Internal Calls To:                n
      External Calls To:                n

      SAC/CF Override: n

```

```

display station 7000                                     Page 4 of 5
                                     STATION

SITE DATA
  Room:                      Headset? n
  Jack:                      Speaker? n
  Cable:                    Mounting: d
  Floor:                  Cord Length: 0
  Building:                Set Color:

ABBREVIATED DIALING
  List1:                  List2:                  List3:

BUTTON ASSIGNMENTS
  1: call-appr            5: call-park
  2: call-appr            6:
  3: call-appr            7:
  4: extnd-call           8:

  voice-mail

```


Avaya Virtual softphone

This is a printout of the Avaya virtual extension used during compliance testing.

display station 58900	Page 1 of 6	
STATION		
Extension: 58900	Lock Messages? n	BCC: 0
Type: 4624	Security Code: *	TN: 1
Port: S00026	Coverage Path 1:	COR: 1
Name: Recorder1	Coverage Path 2:	COS: 1
	Hunt-to Station:	Tests? y
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19	Personalized Ringing Pattern: 1	
	Message Lamp Ext: 58900	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
Survivable GK Node Name:		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? y	
	IP Video Softphone? n	
	Short/Prefixed Registration Allowed: default	

display station 58900	Page 2 of 6	
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Auto Select Any Idle Appearance?	n
LWC Activation? y	Coverage Msg Retrieval?	y
LWC Log External Calls? n	Auto Answer:	none
CDR Privacy? n	Data Restriction?	n
Redirect Notification? y	Idle Appearance Preference?	n
Per Button Ring Control? n	Bridged Idle Line Preference?	n
Bridged Call Alerting? n	Restrict Last Appearance?	y
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
Service Link Mode: as-needed	EC500 State:	enabled
Multimedia Mode: enhanced	Audible Message Waiting?	n
MWI Served User Type:	Display Client Redirection?	n
AUDIX Name:	Select Last Used Appearance?	n
	Coverage After Forwarding?	s
	Multimedia Early Answer?	n
Remote Softphone Emergency Calls: as-on-local	Direct IP-IP Audio Connections?	y
Emergency Location Ext: 58900	Always Use? n IP Audio Hairpinning?	n

Page 3 of 6

STATION

```

Conf/Trans on Primary Appearance? n
Bridged Appearance Origination Restriction? n    Offline Call Logging? y

```

Enhanced Caller-Info Display for 1-Line Phones? n

ENHANCED CALL FORWARDING

				Forwarded Destination	Active
Unconditional	For	Internal Calls To:			n
		External Calls To:			n
Busy	For	Internal Calls To:			n
		External Calls To:			n
No Reply	For	Internal Calls To:			n
		External Calls To:			n

SAC/CF Override: n

Page 4 of 6

STATION

SITE DATA

```

Room:                               Headset? n
Jack:                               Speaker? n
Cable:                             Mounting: d
Floor:                             Cord Length: 0
Building:                           Set Color:

```

ABBREVIATED DIALING

```
List1:          List2:          List3:
```

BUTTON ASSIGNMENTS

1: call-appr	7:
2: call-appr	8:
3: call-appr	9:
4:	10:
5:	11:
6:	12:

display station 58900

Page 5 of 6

STATION

FEATURE BUTTON ASSIGNMENTS

13:	19:
14:	20:
15:	21:
16:	22:
17:	23:
18:	24:

display station 58900

Page 6 of 6

STATION

SOFTKEY BUTTON ASSIGNMENTS

- 1: directory
- 2: drop
- 3: int-aut-an
- 4: timer
- 5: priority
- 6: auto-cback
- 7: abr-prog
- 8: abr-spchar Char: ~p
- 9: lwc-store
- 10: ringer-off
- 11: btn-view
- 12: admin

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