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.

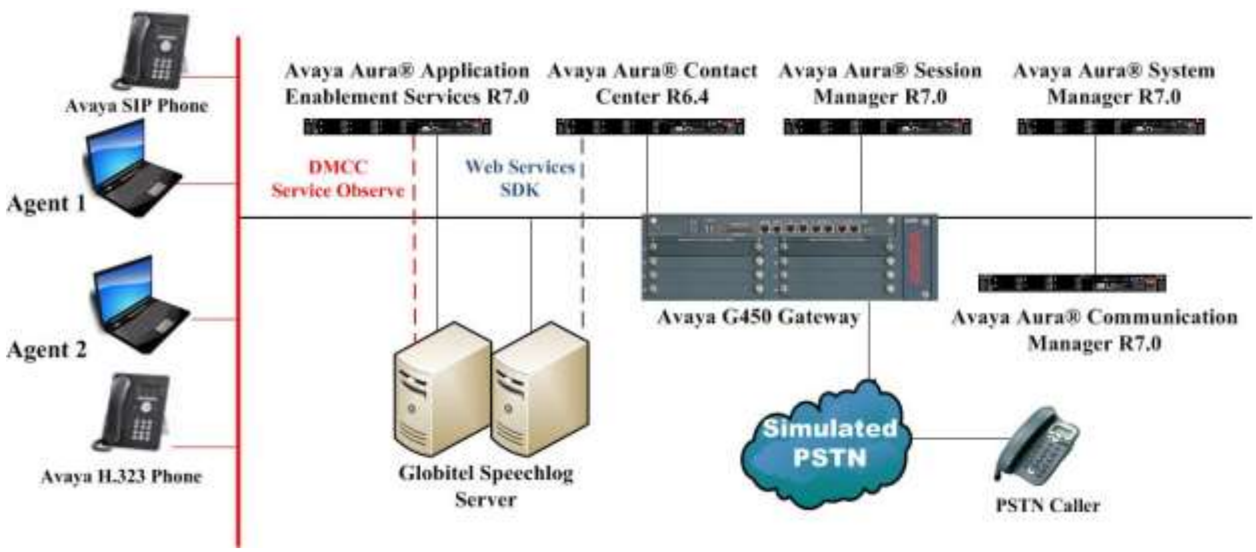


Figure 1: Connection of Globitel SpeechLog Voice Recorder with Avaya Aura® Contact Center connected to Avaya Aura® Communication Manager R7.0 and Avaya Aura® Application Enablement Services R7.0.

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
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                        DS1 MSP? y
                                 ATMS? y            DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

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       Password          Enabled    Status
                   Server
      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:
```

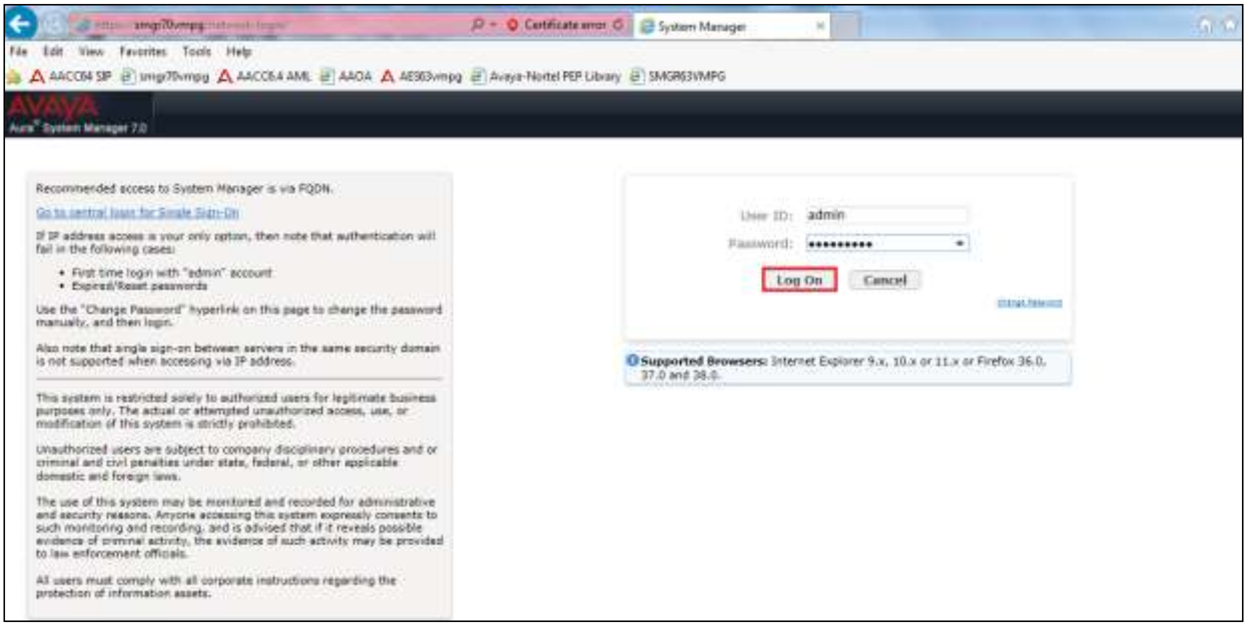
5.6. Configure H323 Stations for Service Observation

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**.

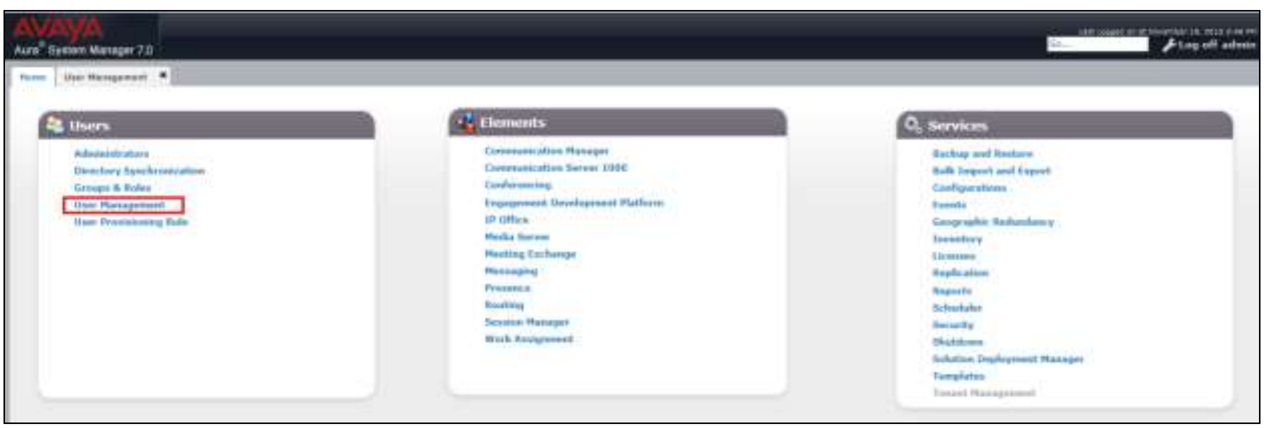
```
change station x                                              Page   1 of   5
                                     STATION

Extension: 7000                          Lock Messages? n               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
```

5.7. Configure SIP Stations for Service Observation

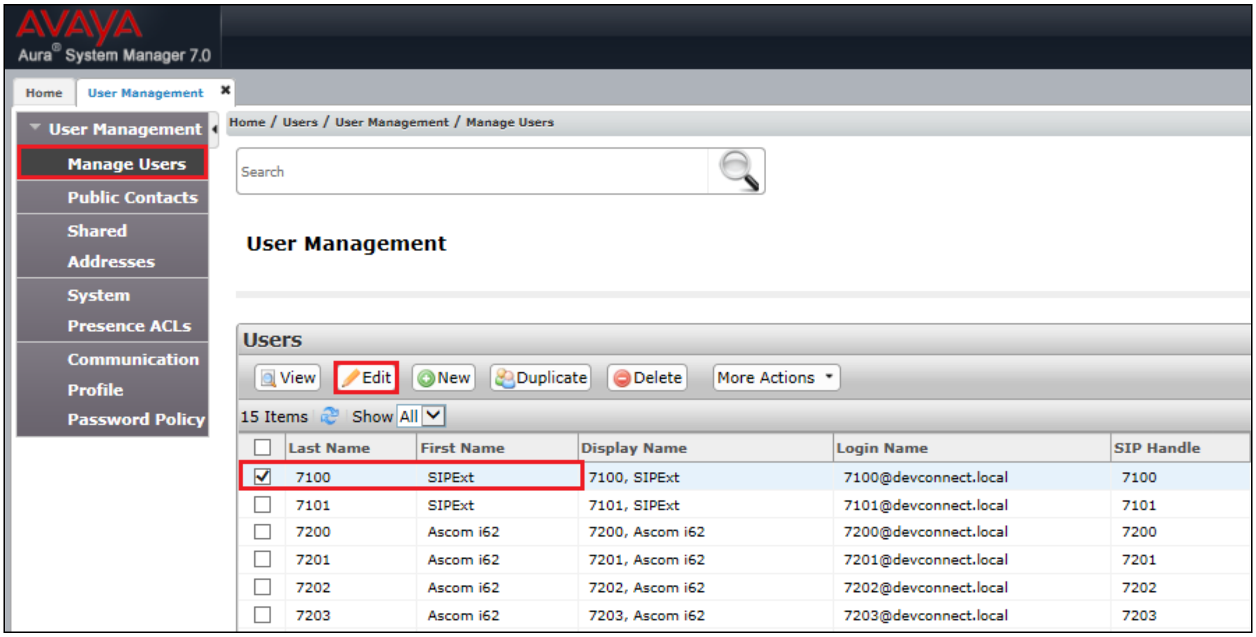
The configuration of SIP phones on Communication Manager must be carried out from System Manager. Access the System Manager using a Web Browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager or **http://<IP Address >/SMGR**. Log in using appropriate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



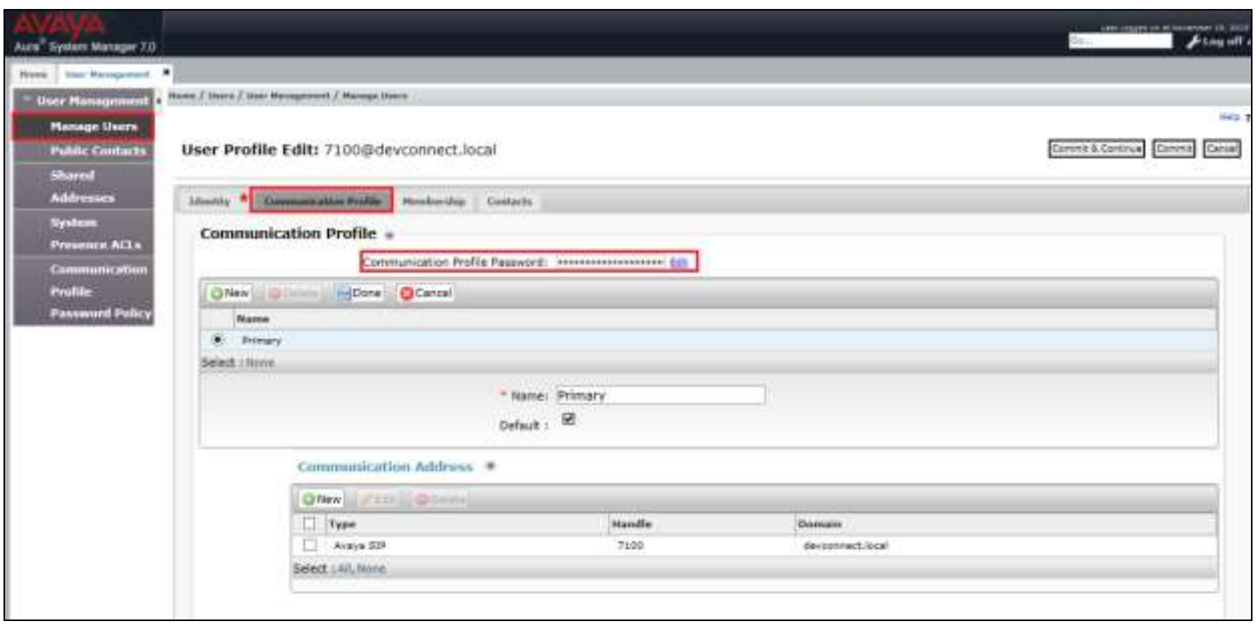
From the home page click on **User Management** highlighted below.



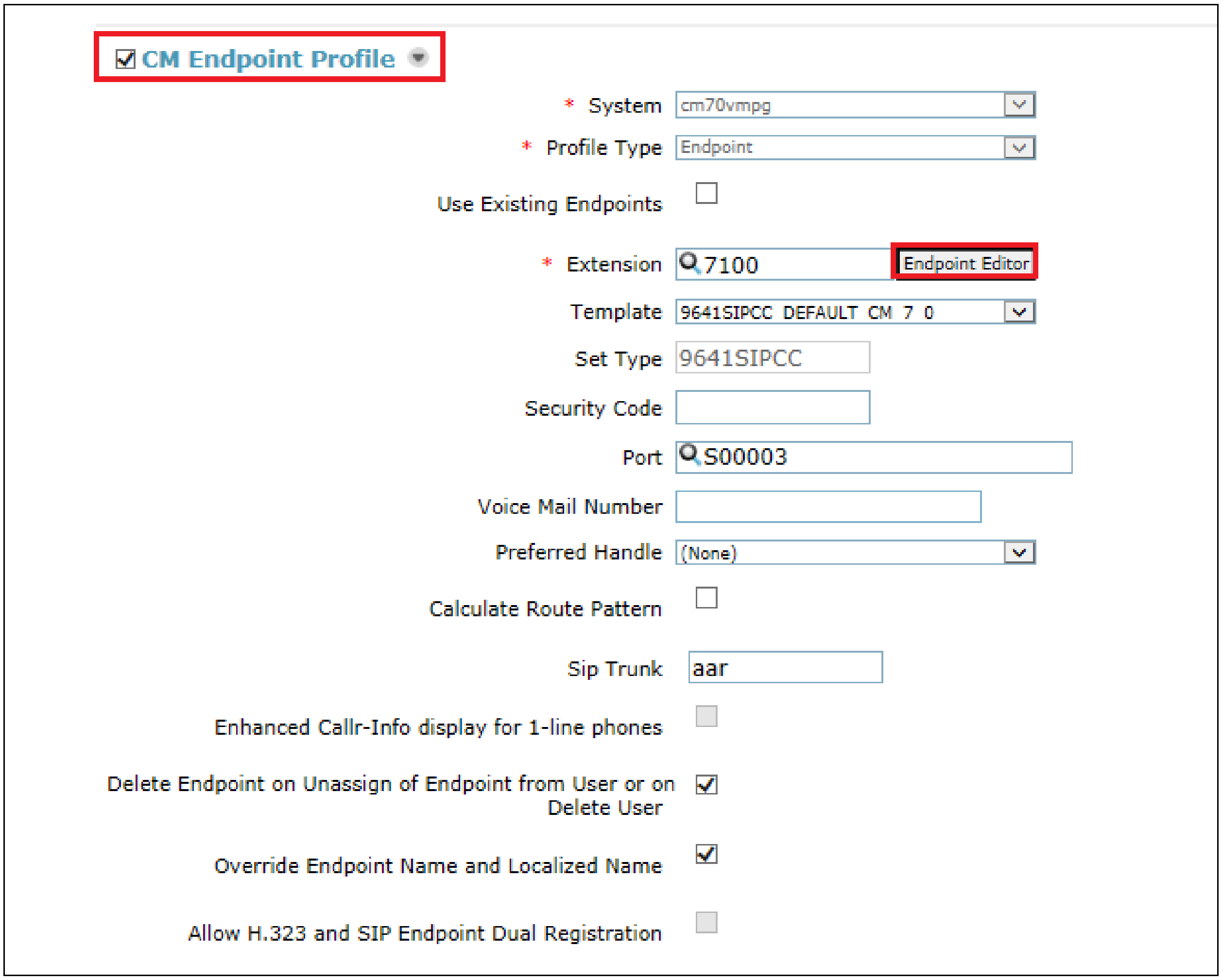
Click on **Manager Users** in the left window. Select the station to be edited and click on **Edit**.



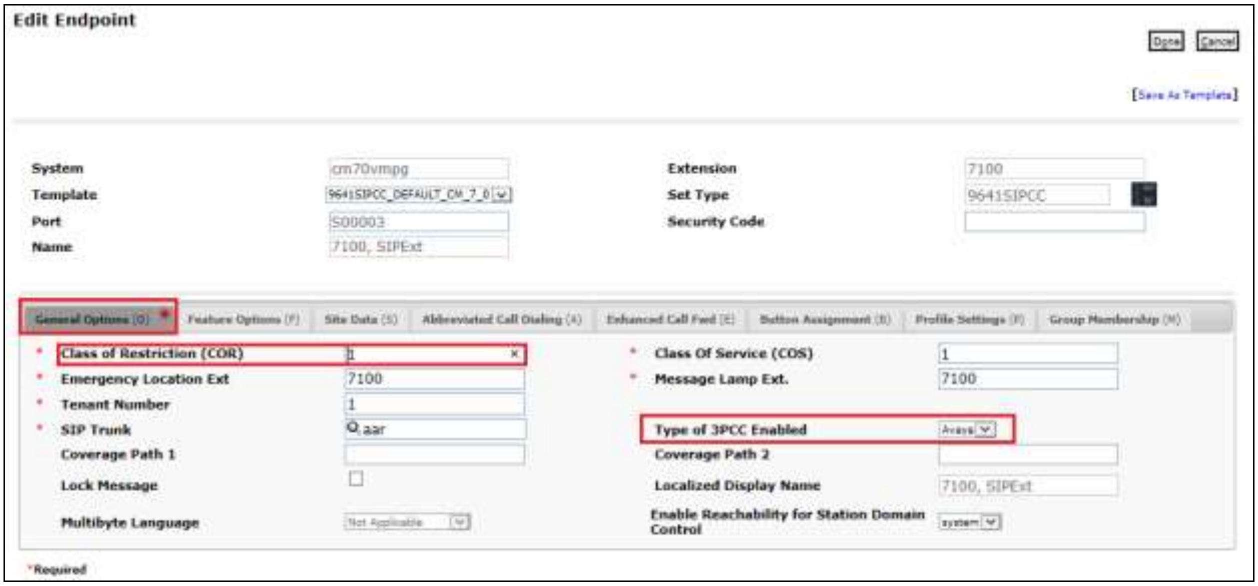
Click on the **Communication Profile** tab. Ensure that the **Communication Profile Password** is known and if not click on edit to change it.



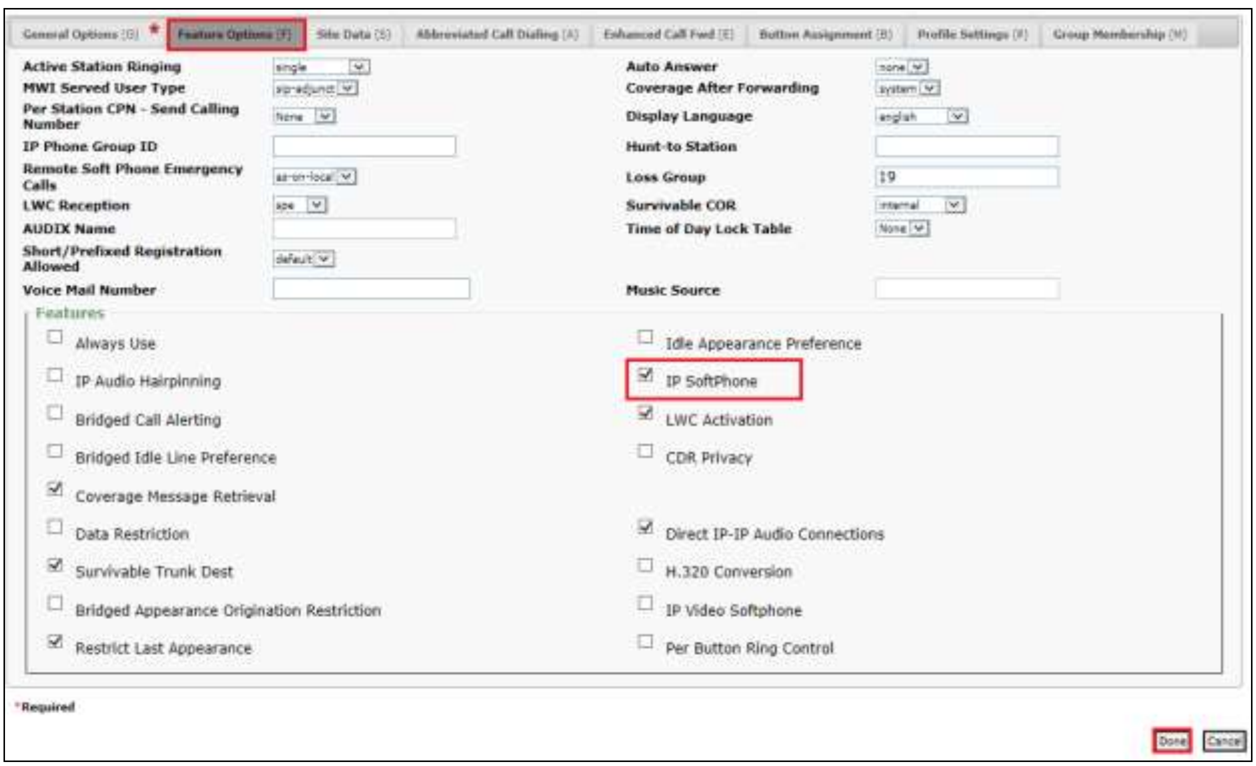
From the same page scroll down to **CM Endpoint Profile** and enter a suitable **Extension** number and select the correct **Template** then click on **Endpoint Editor** to make further changes.



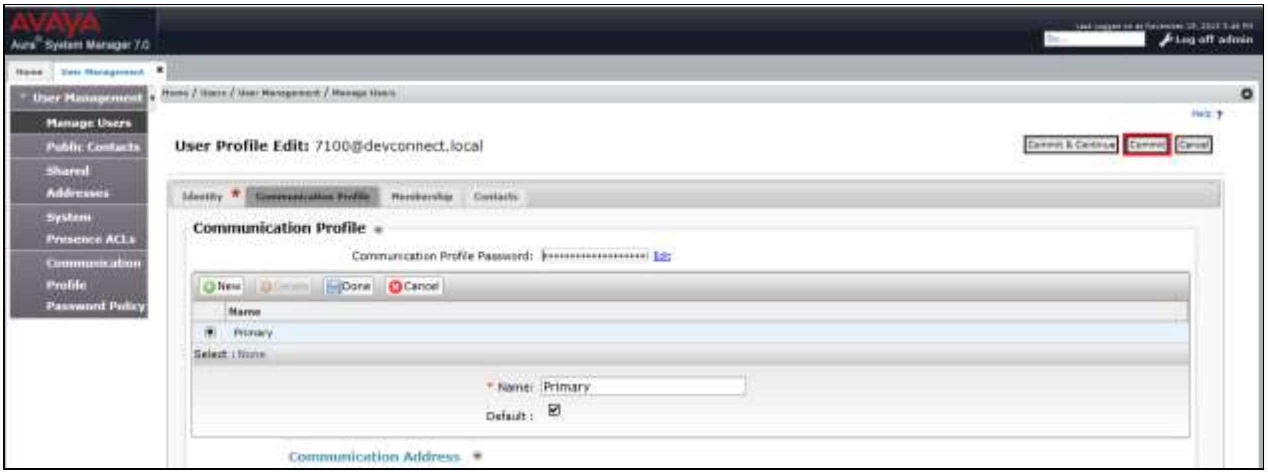
In the **General Options** tab ensure that **Type of 3PCC Enabled** is set to **Avaya** as is shown below. Also that Class of Restriction is set to that configured in **Section 5.5**.



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.



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
                                           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
```

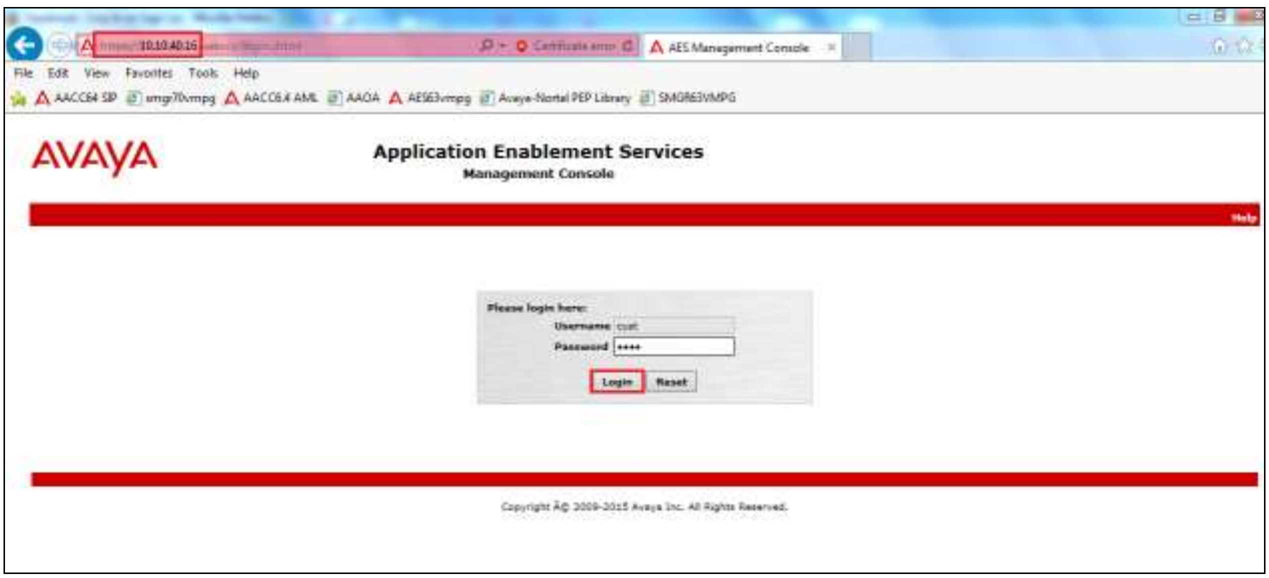
6. Configure Avaya Aura® Application Enablement Services

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

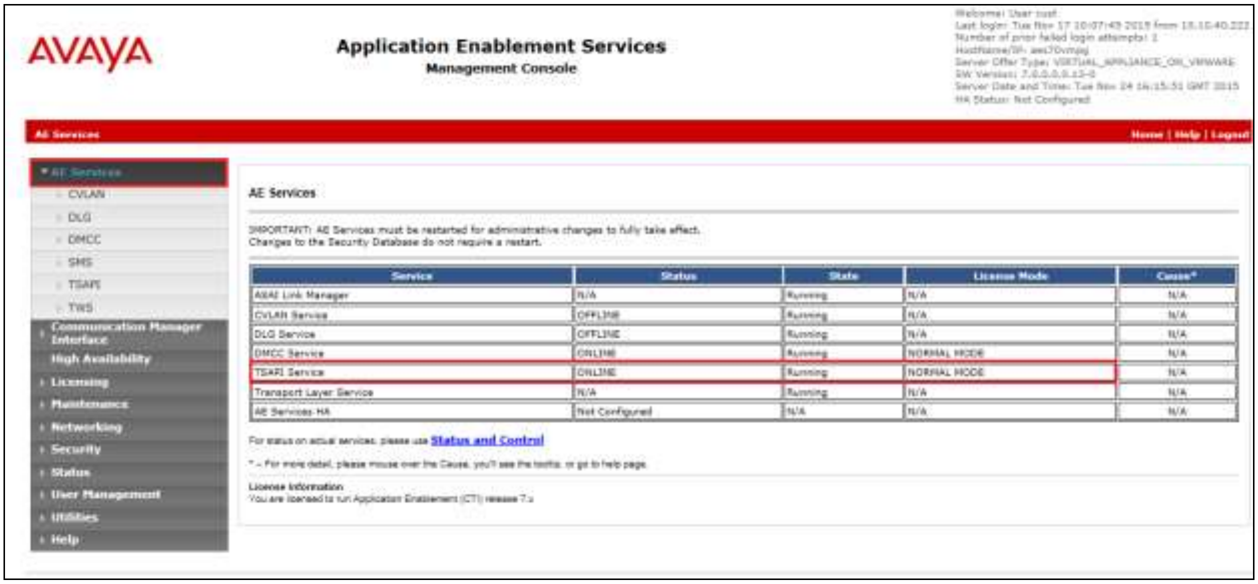
- Verify Licensing
- Create Switch Connection
- Administer TSAPI link
- Identify Tlinks
- Enable TSAPI Ports
- Create CTI User
- Associate Devices with CTI User

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.



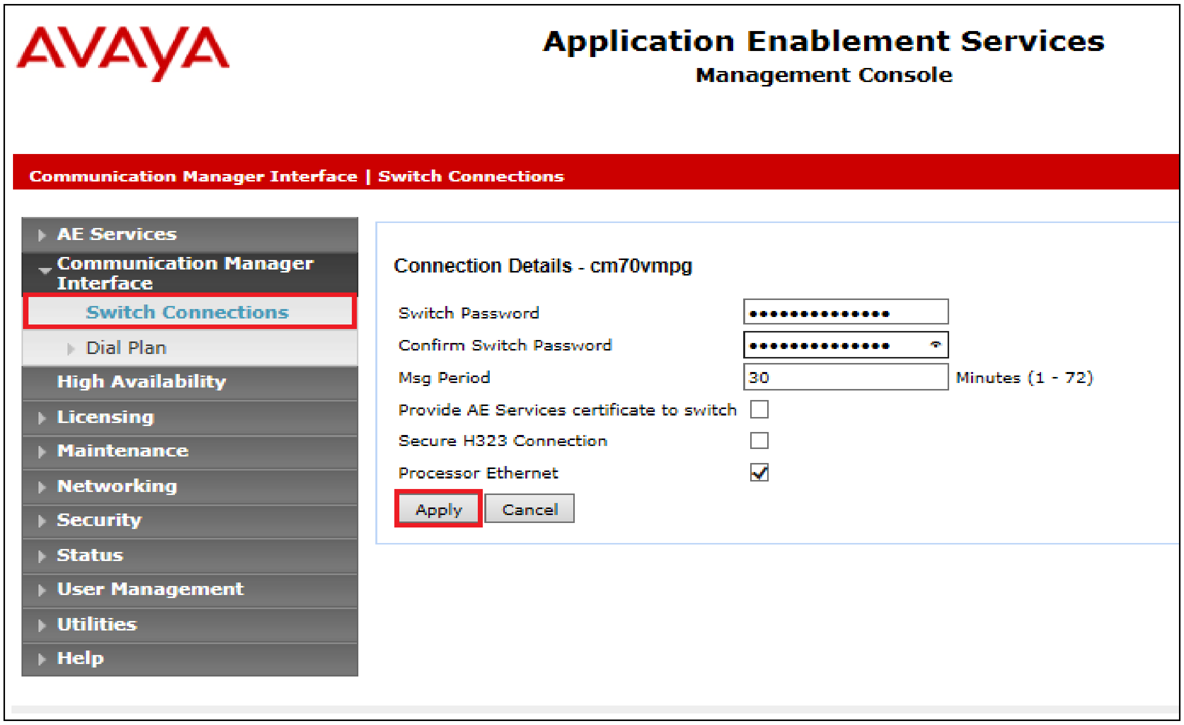
The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the TSAPI Service is licensed by ensuring that **TSAPI Service** is in the list of **Services** and that the **License Mode** is showing **NORMAL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



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.

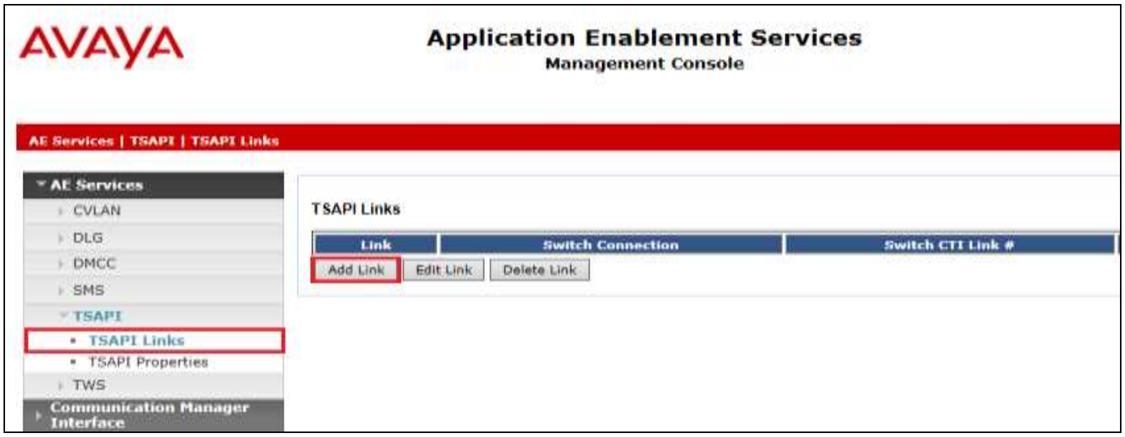
In the resulting screen enter the **Switch Password**. The Switch Password must be the same as that entered into Communication Manager AE Services Administration screen via the **change ip-services** command described in **Section 5.3**. Default values may be accepted for the remaining fields. Click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



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.

6.3. Administer TSAPI link

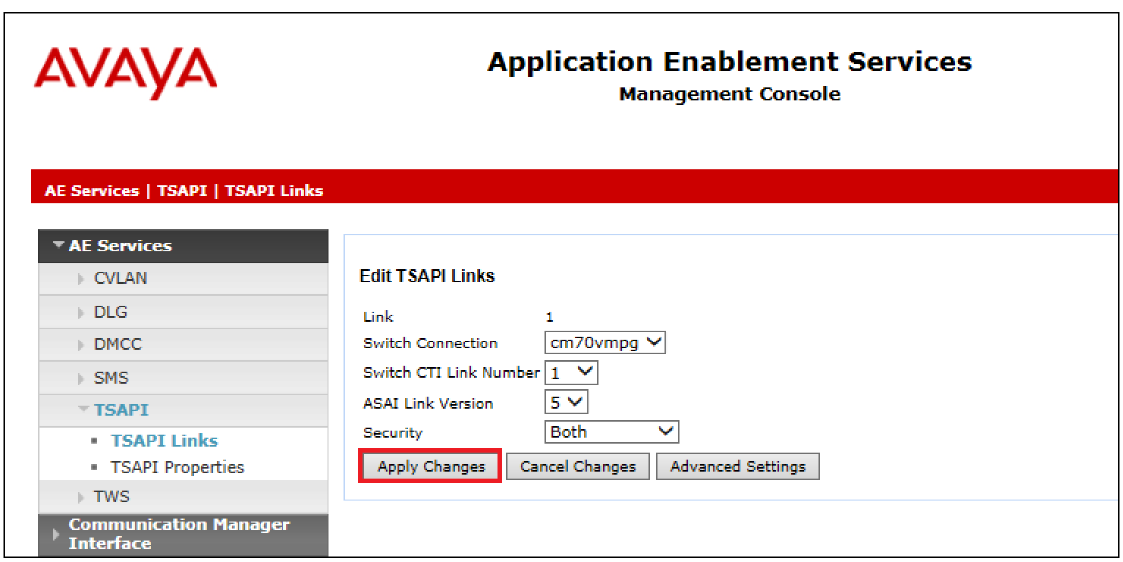
From the Application Enablement Services Management Console, select **AE Services** → **TSAPI** → **TSAPI Links**. Select **Add Link** button as shown in the screen below.



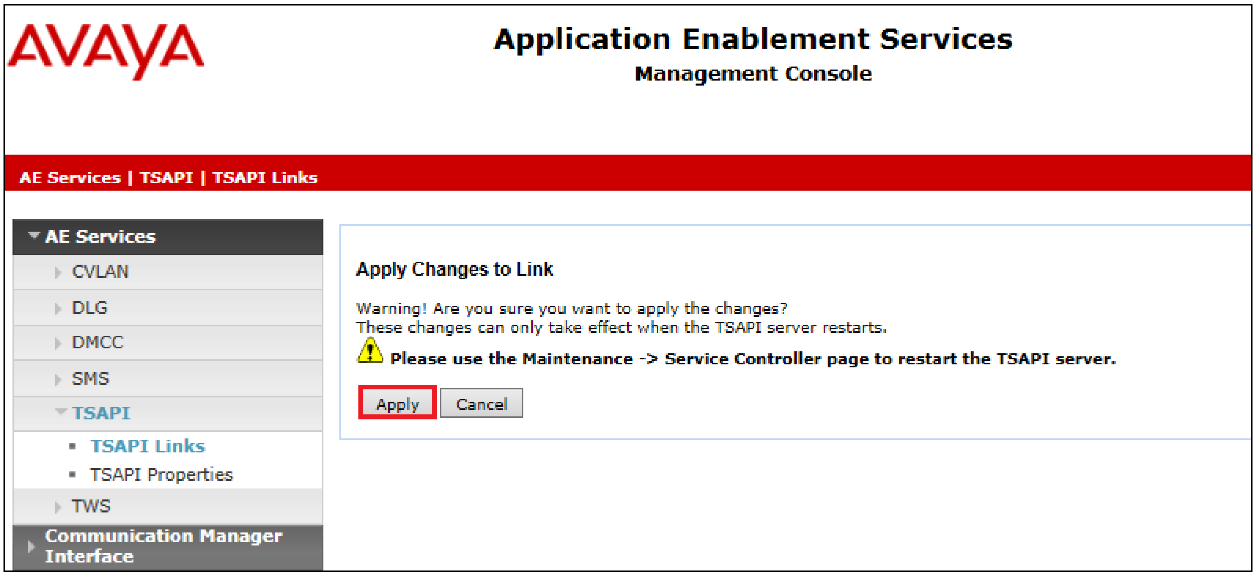
On the **Add TSAPI Links** screen (or the **Edit TSAPI Links** screen to edit a previously configured TSAPI Link as shown below), enter the following values:

- **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**.

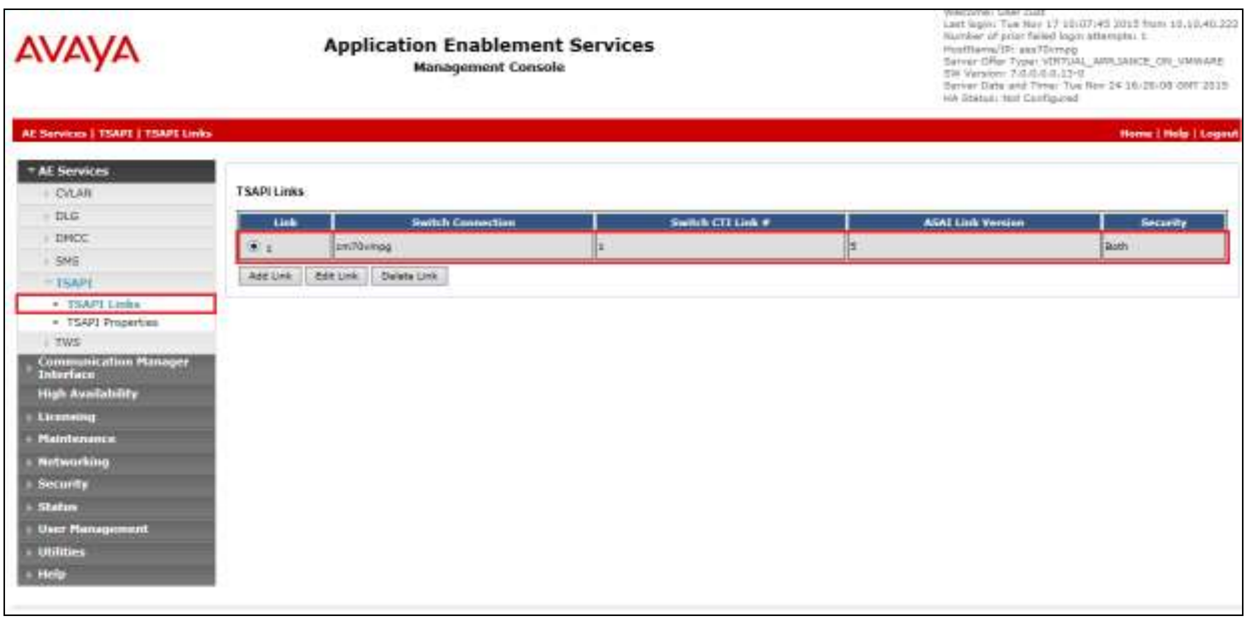
Once completed, select **Apply Changes**.



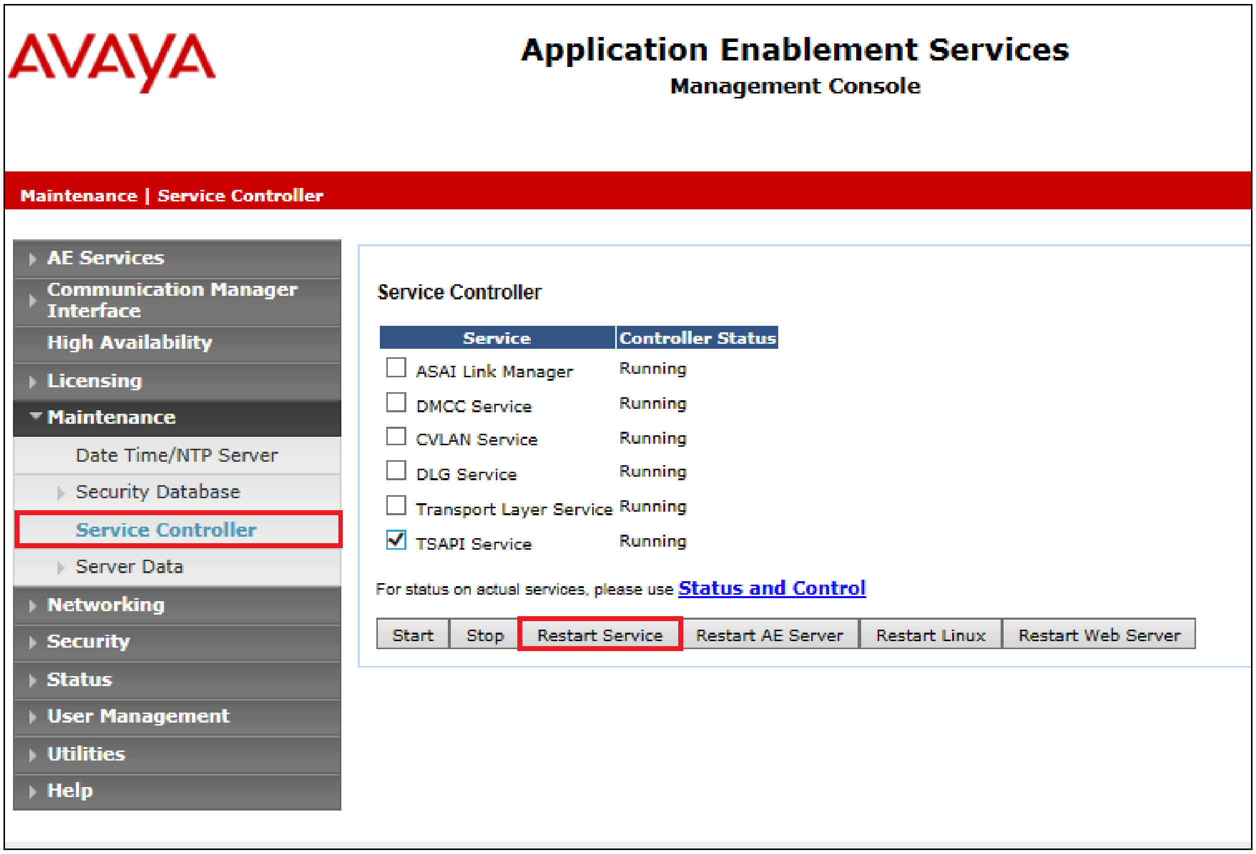
Another screen appears for confirmation of the changes made. Choose **Apply**.



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

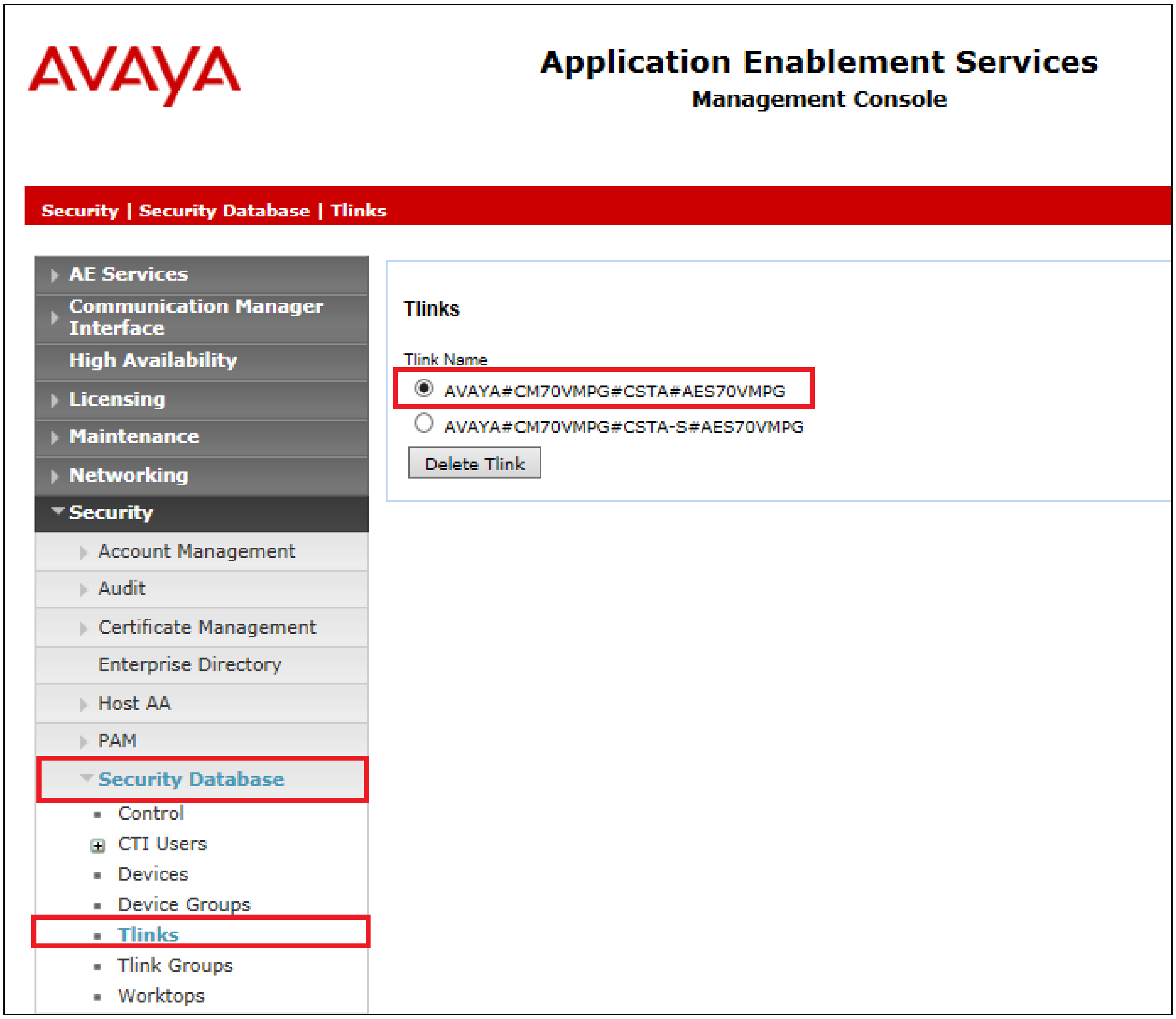


The TSAPI Service must be restarted to effect the changes made in this section. From the Management Console menu, navigate to **Maintenance → Service Controller**. On the Service Controller screen, tick the **TSAPI Service** and select **Restart Service**.



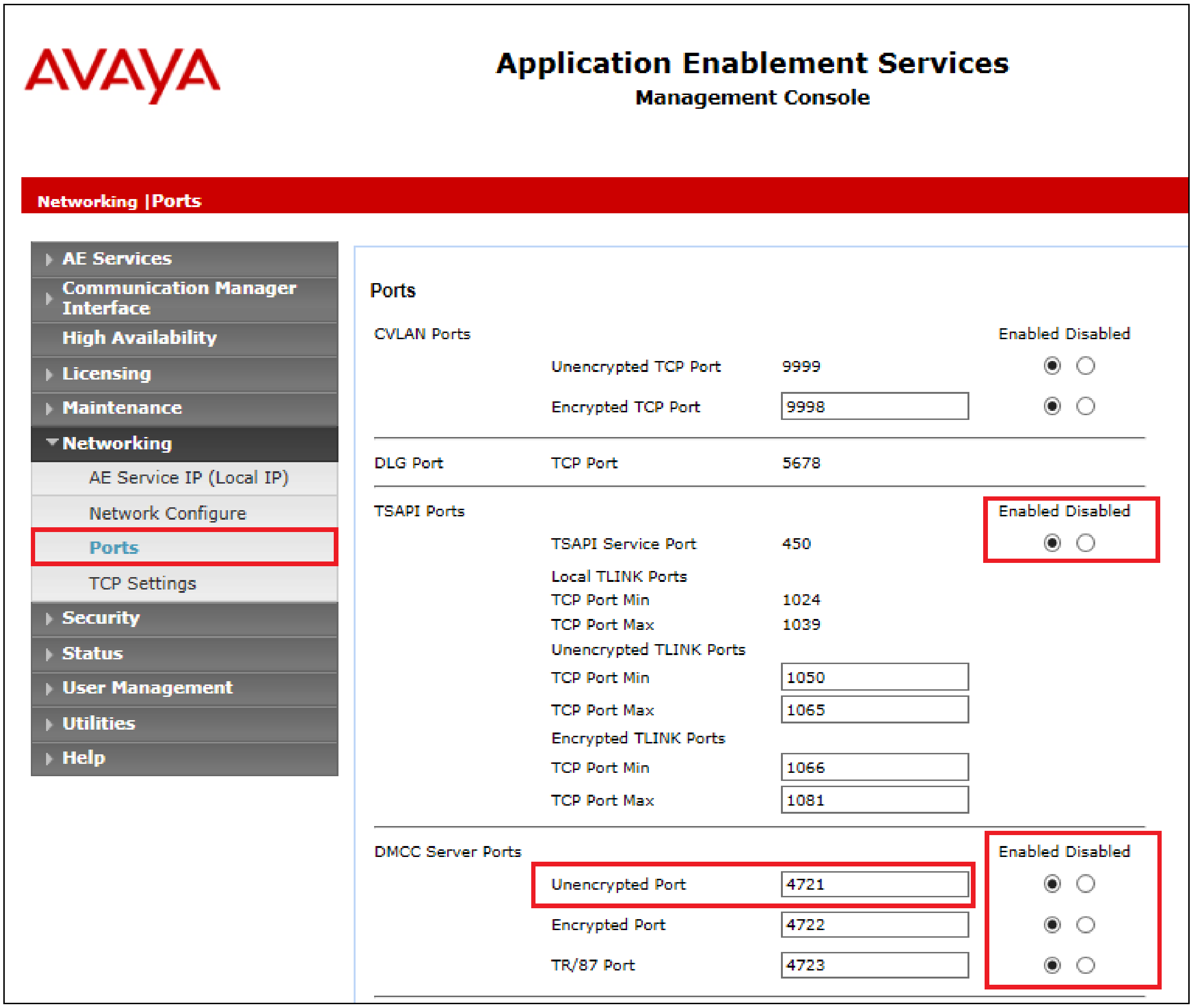
6.4. Identify Tlinks

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**.



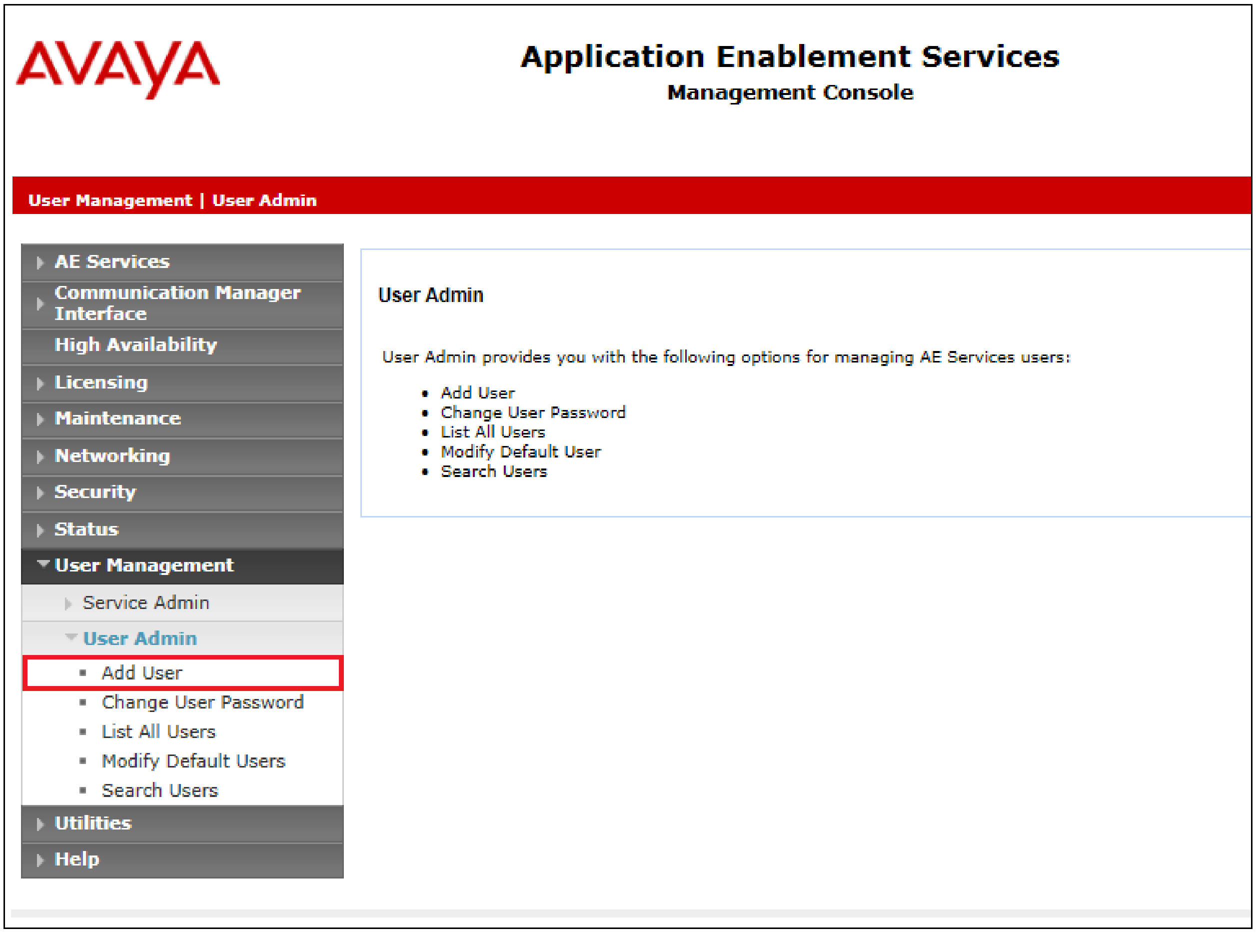
6.5. Enable TSAPI Ports

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**.



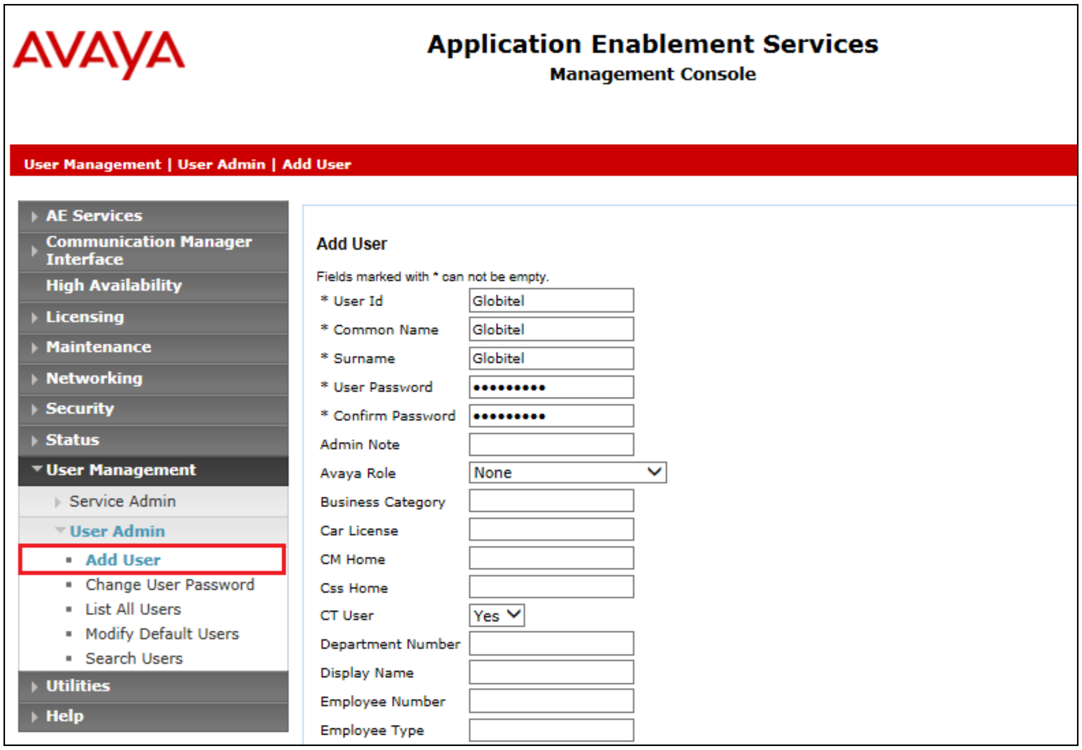
6.6. Create CTI User

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.

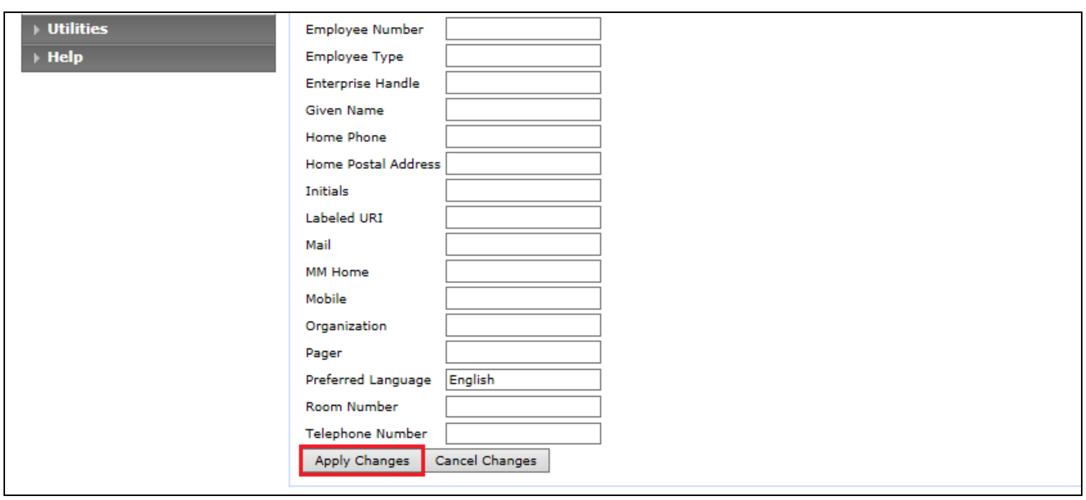


In the **Add User** screen shown below, enter the following values:

- **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.

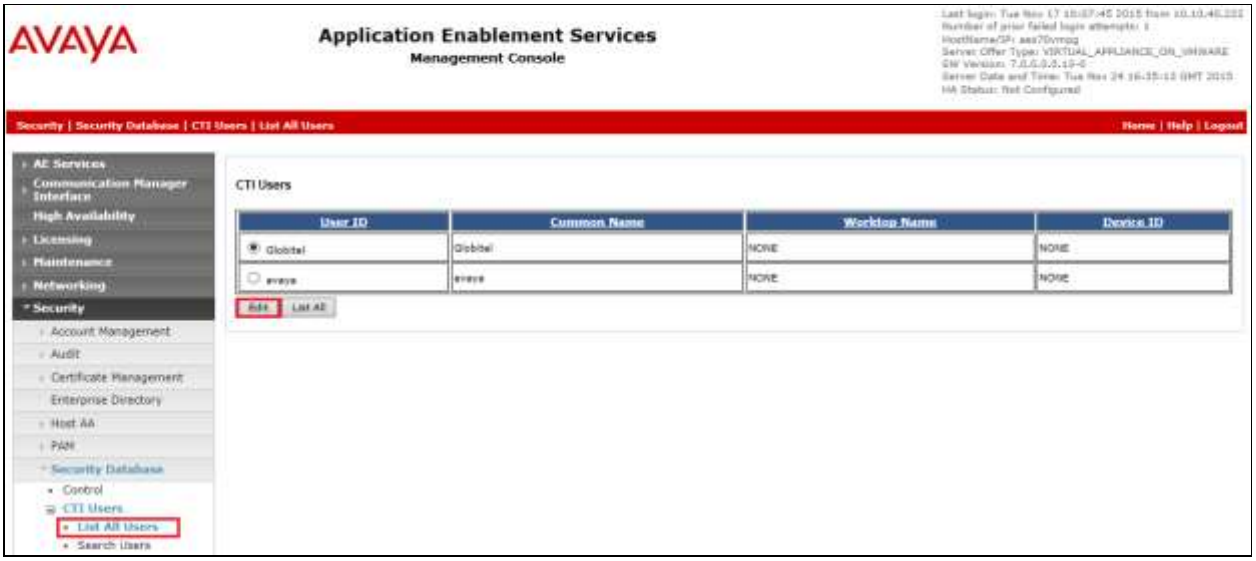


Scroll down and click on **Apply Changes**.

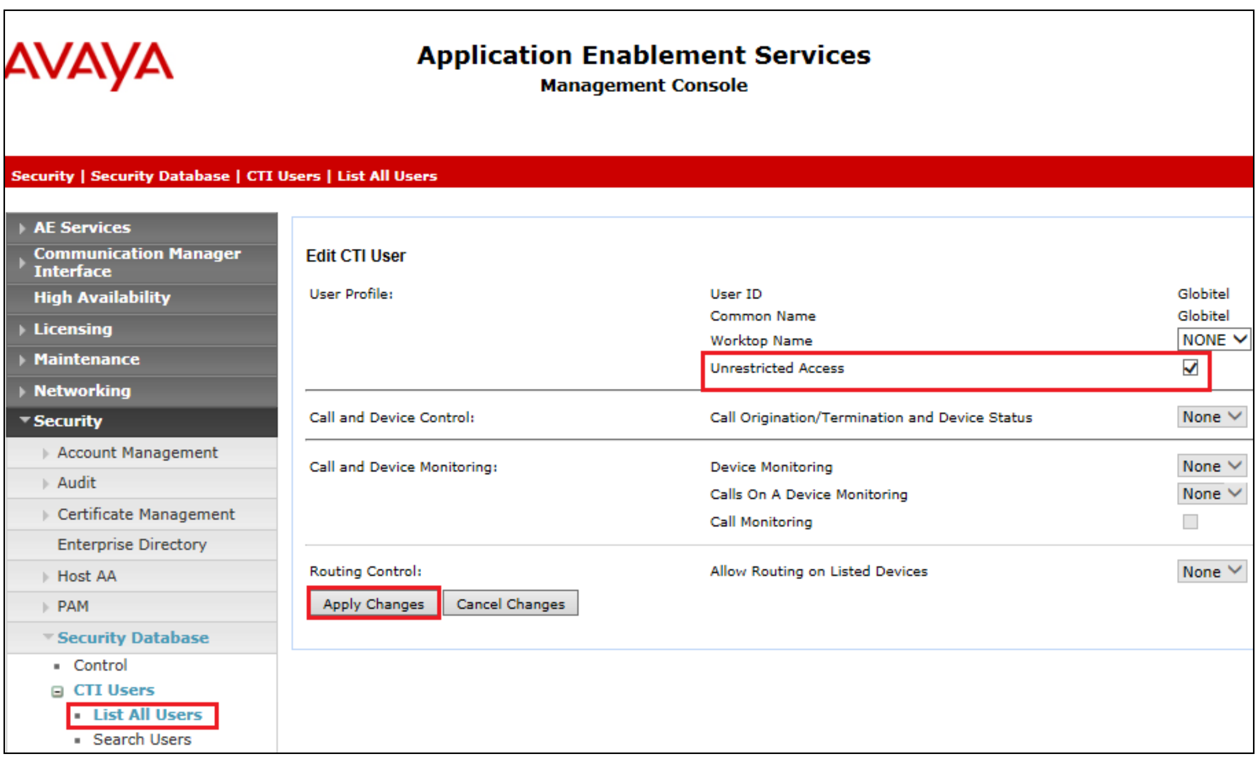


6.7. Associate Devices with CTI User

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



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

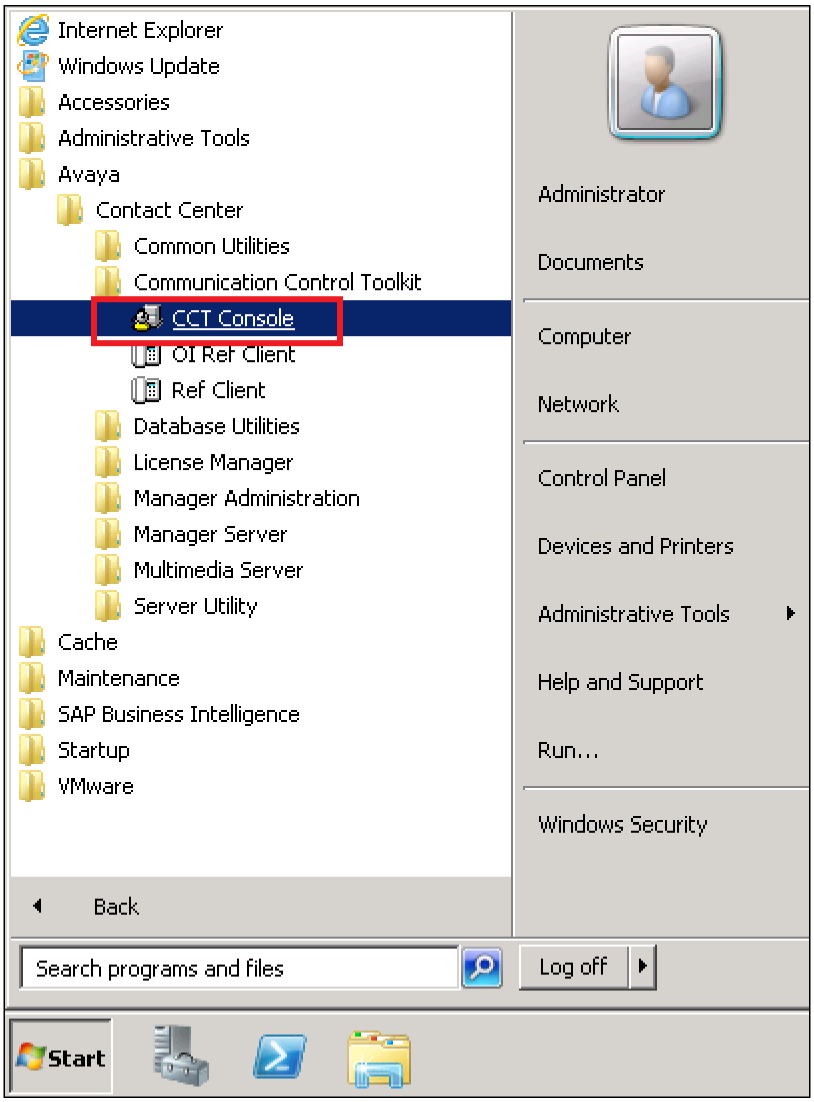


7. Configure Avaya Aura® Contact Center

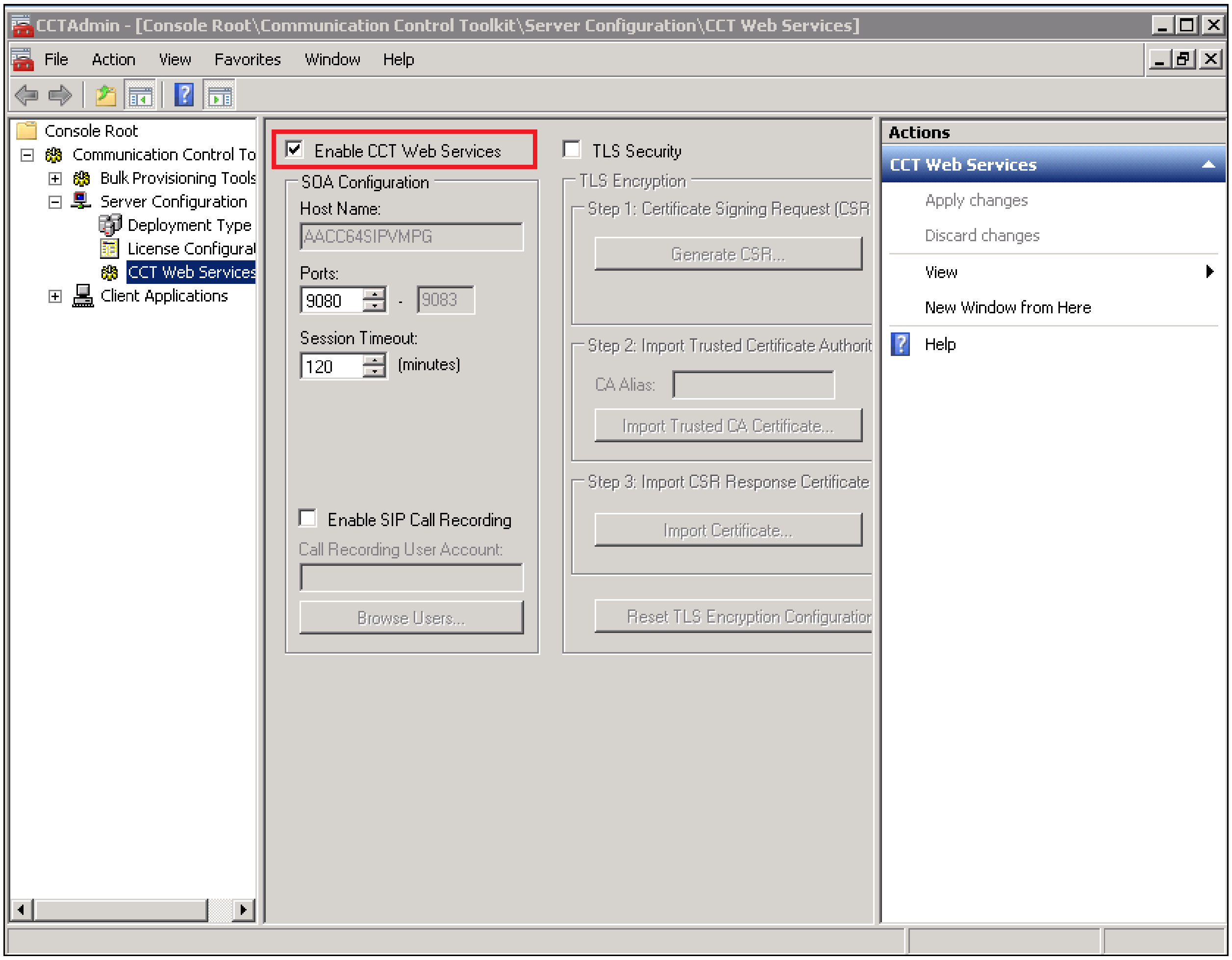
Web Services need to be running on the Contact Center server and this along with a CCT user is what is required for SpeechLog Voice Recorder to connect to Contact Center in order to monitor the agent's calls. This user is created in CCT and the agents that are to be monitored are associated with that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

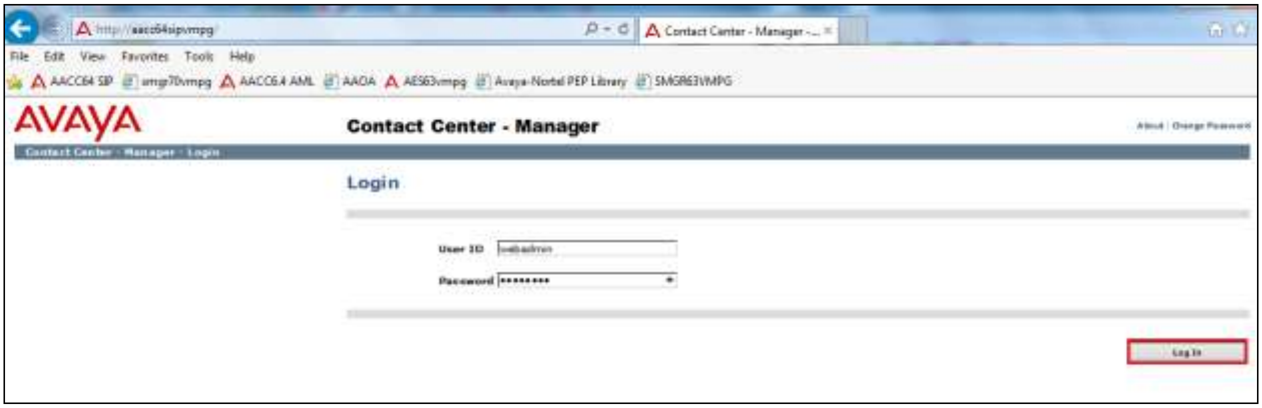


Ensure that **Enable CCT Web Services** is ticked, if not tick this and close the window. Upon closing the option to save this will be presented (not shown below).



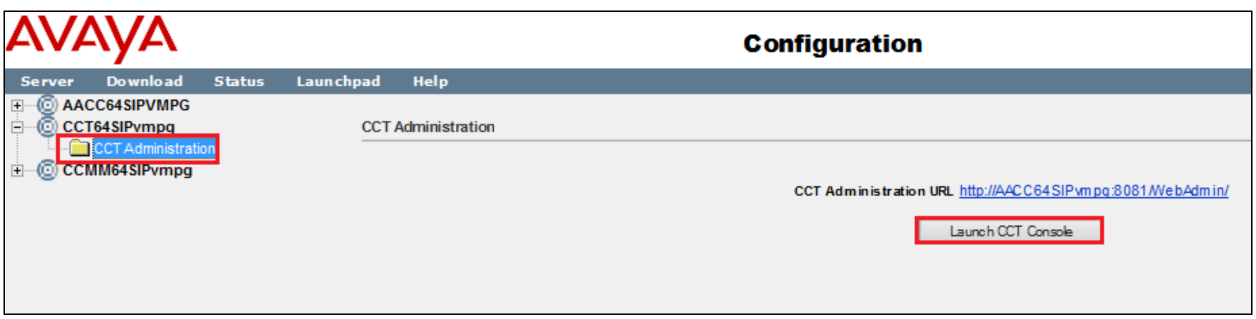
7.2. Create a CCT user

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

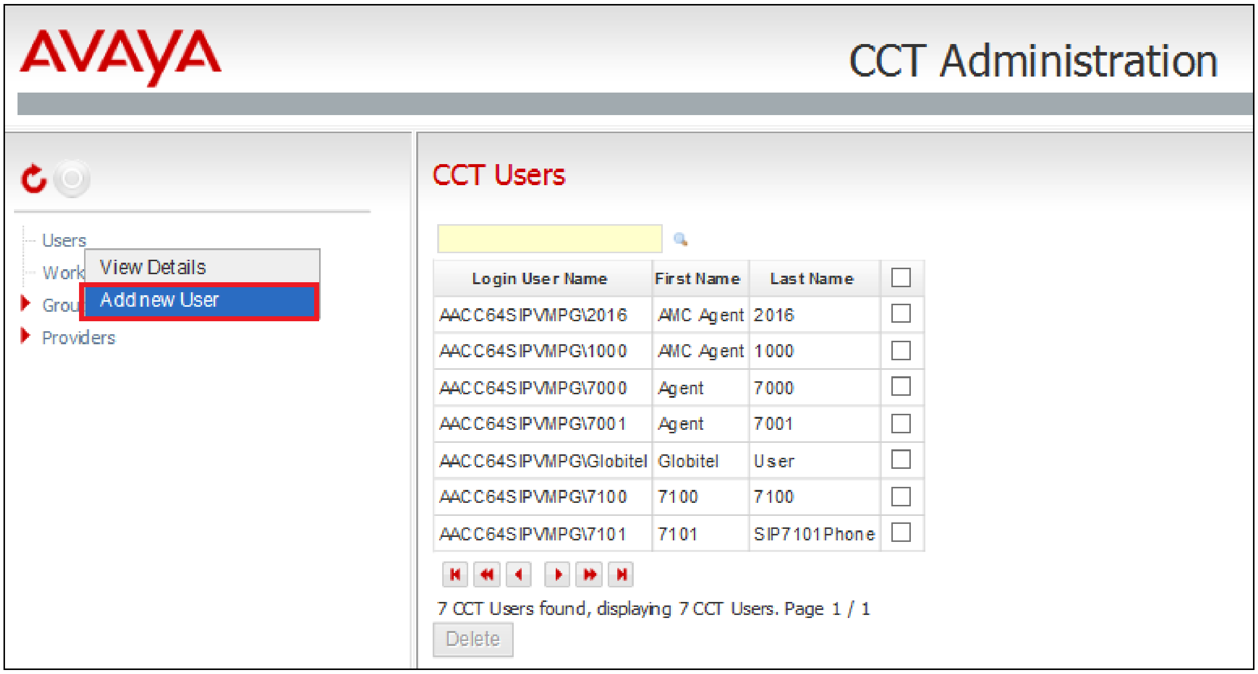


Once logged in select Configuration.

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



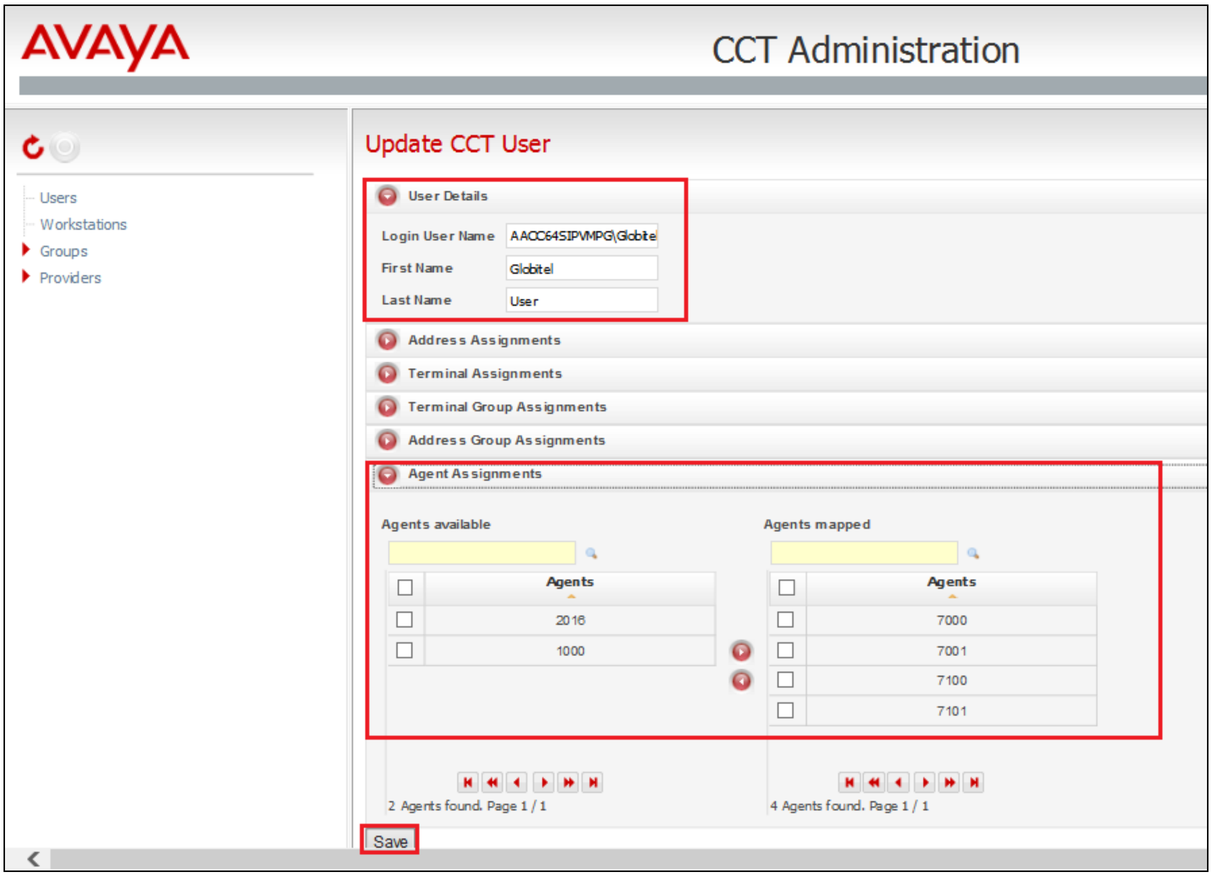
Right click on **Users** in the left window and click on **Add new User**.



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.

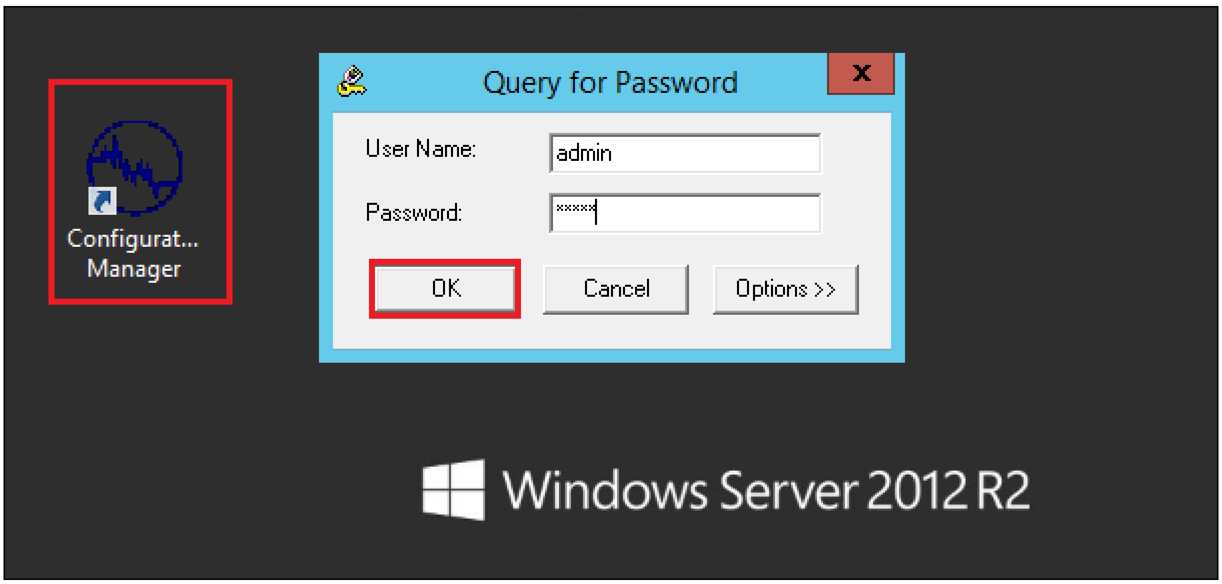
Under **Agent Assignments** select the agents that are to be monitored. In the example below agents **7000**, **7001**, **7100** and **7101** are all being monitored.



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**.

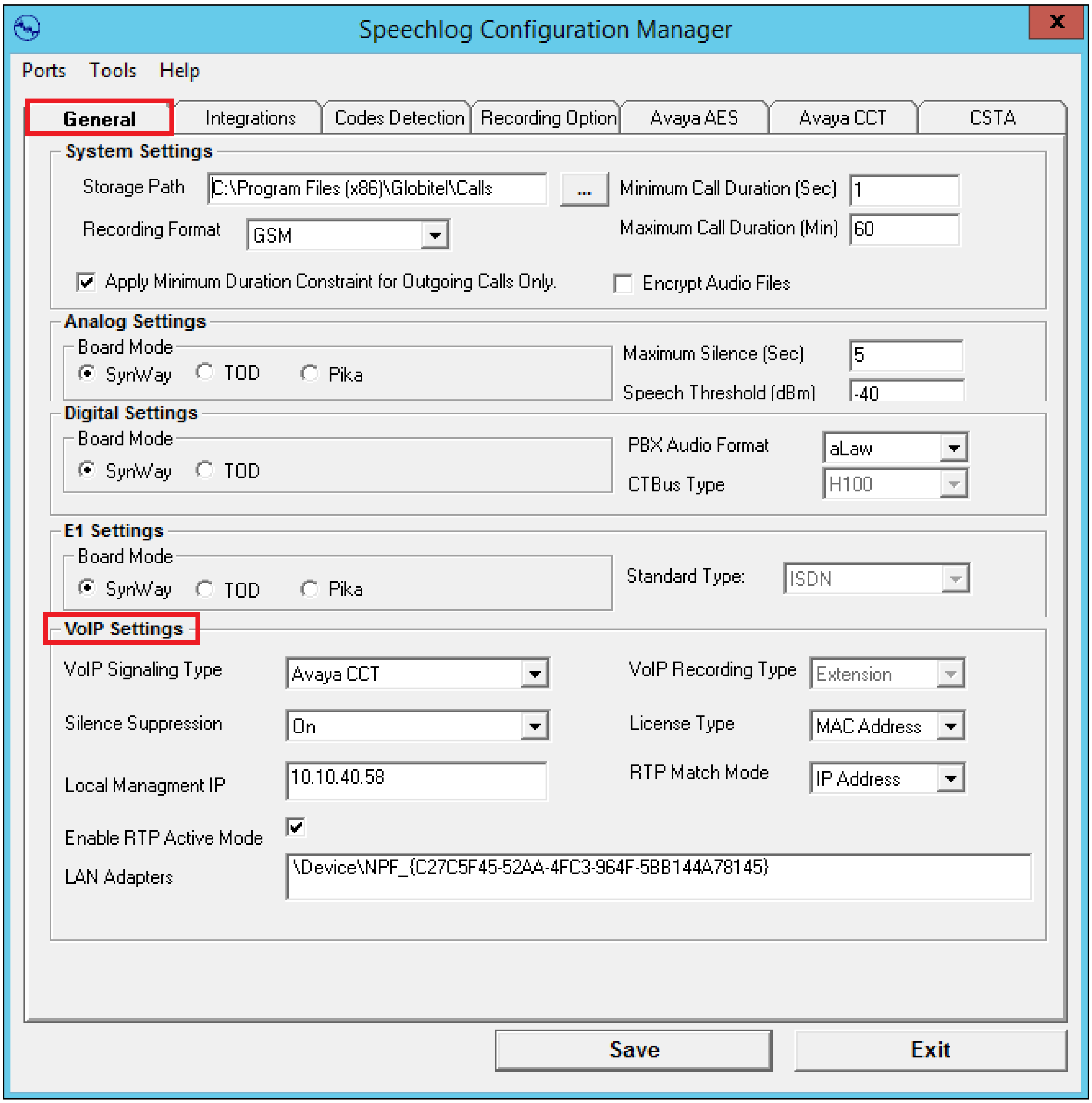


8.1. Configure connection to Avaya Aura® Application Enablement Services and Avaya Aura® Contact Center

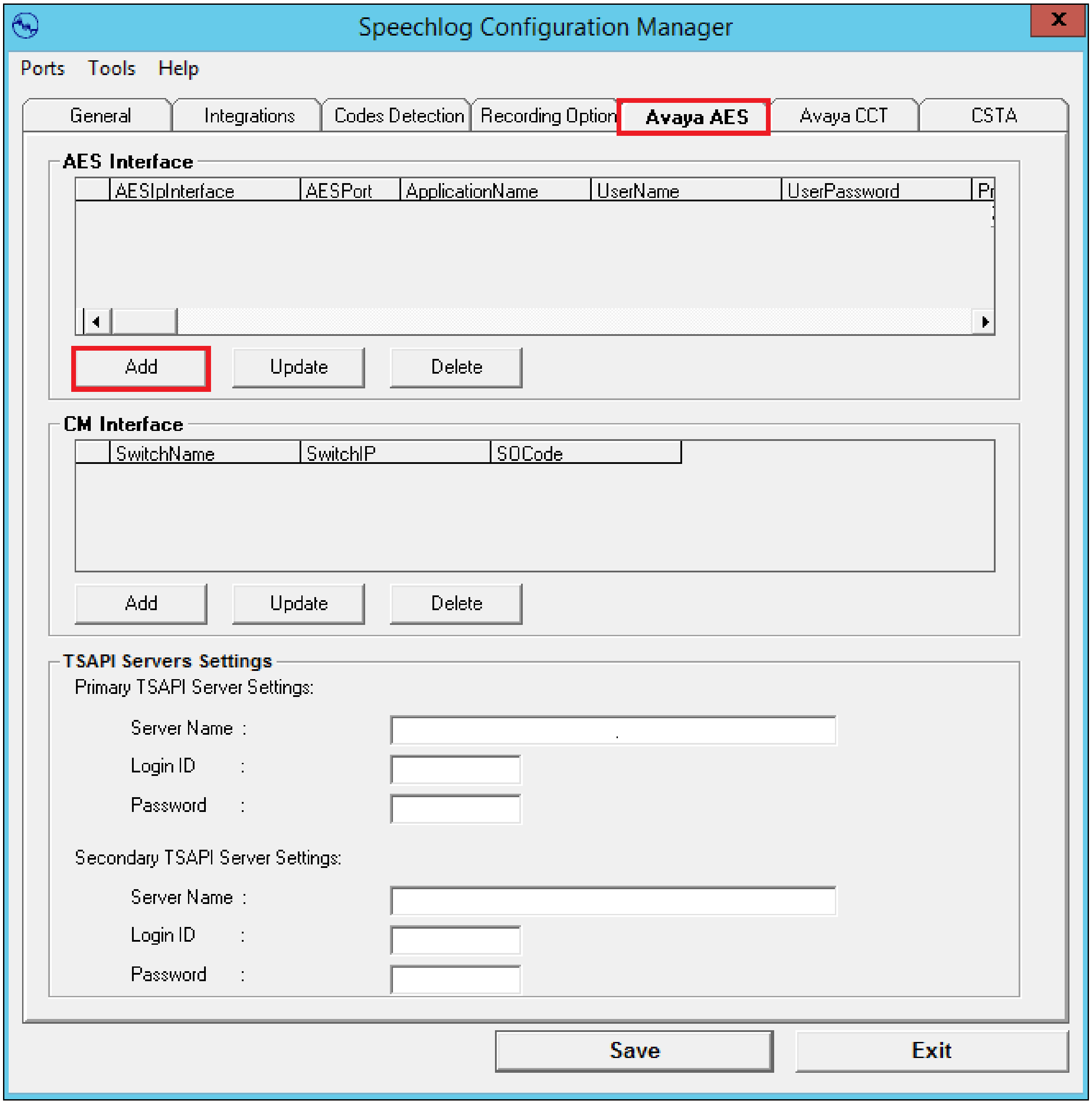
Click on the **General** tab, the **VoIP Settings** should be set as follows:

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

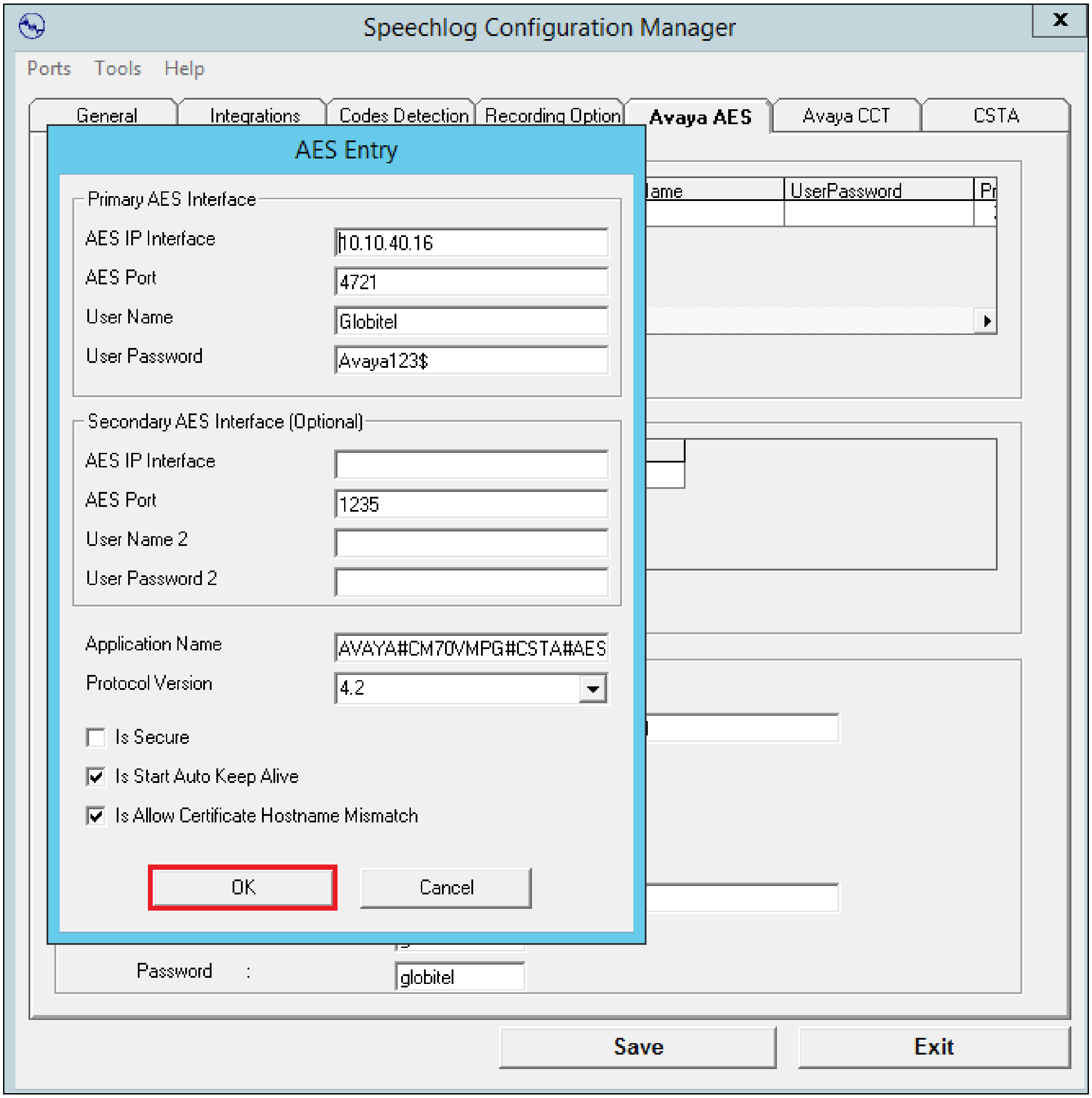
The other settings can be left as default.



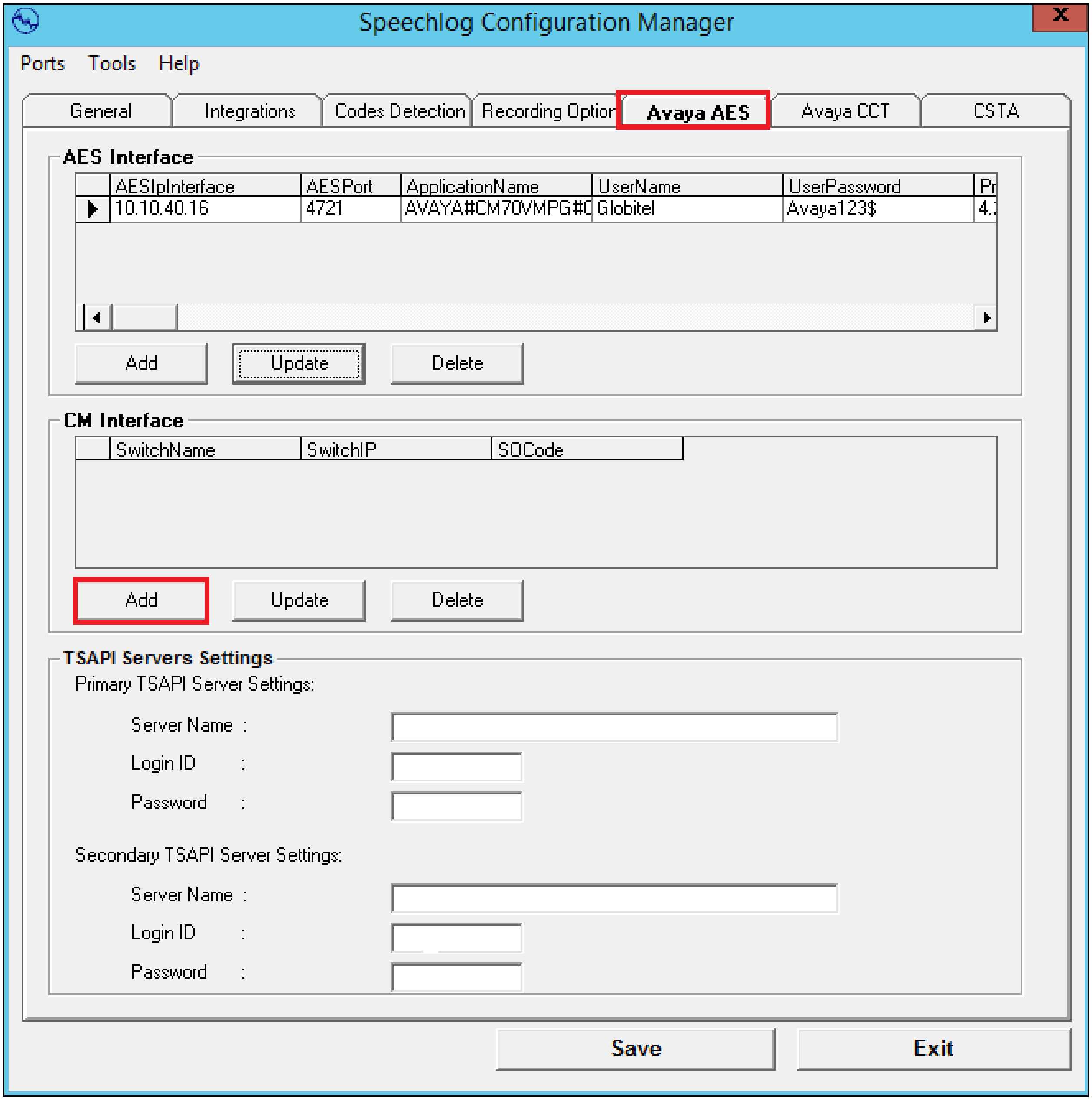
Click on the **Avaya AES** tab and **Add** a new **AES Interface**.



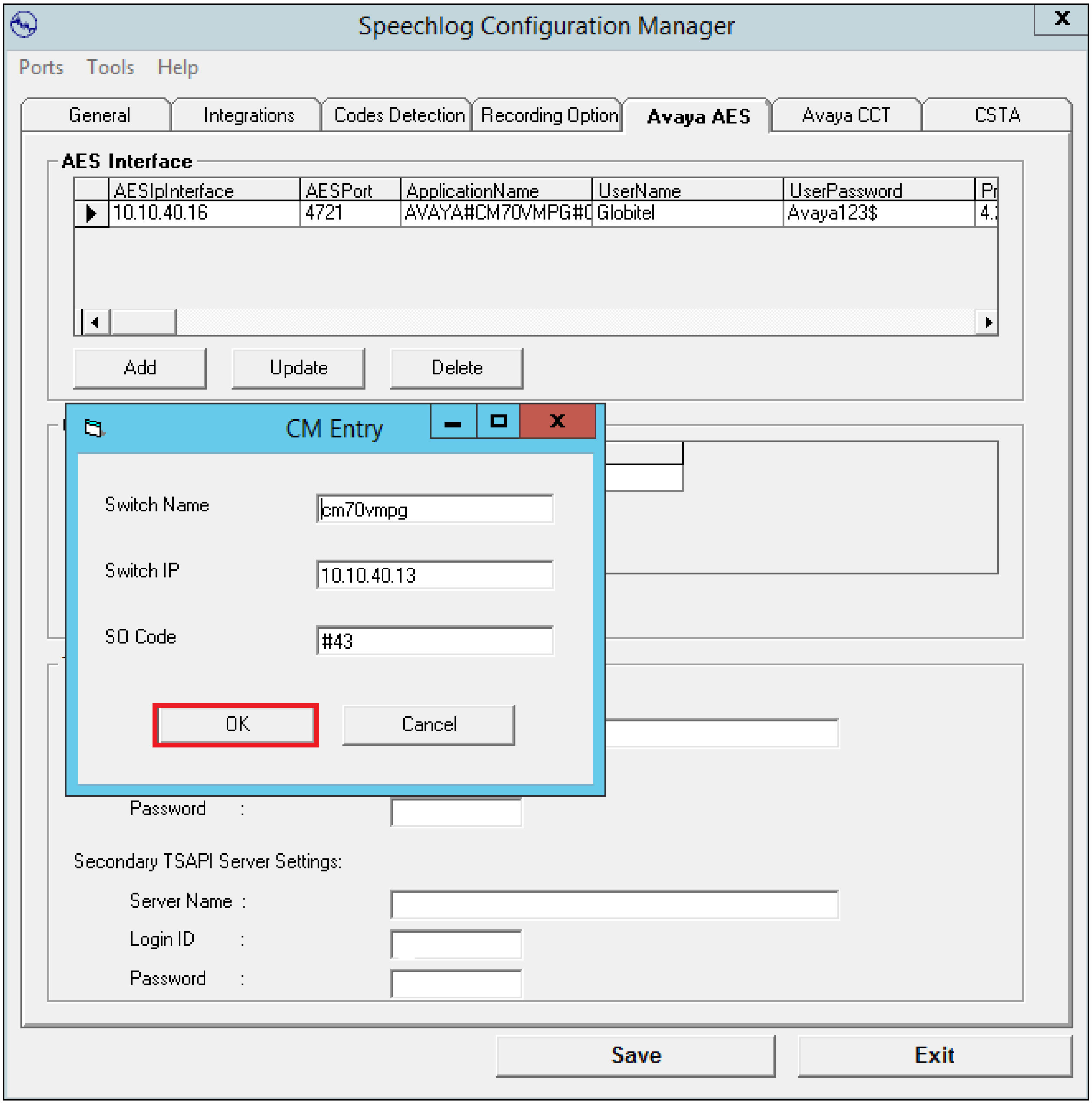
Enter the AES IP address for the **AES IP Interface** and the **AES Port** which should be the DMCC unsecured port number as shown in **Section 6.5**. The **User Name** and **User Password** is that which was configured in **Section 6.6**. The **Application Name** is the Tlink also secured from **Section 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.



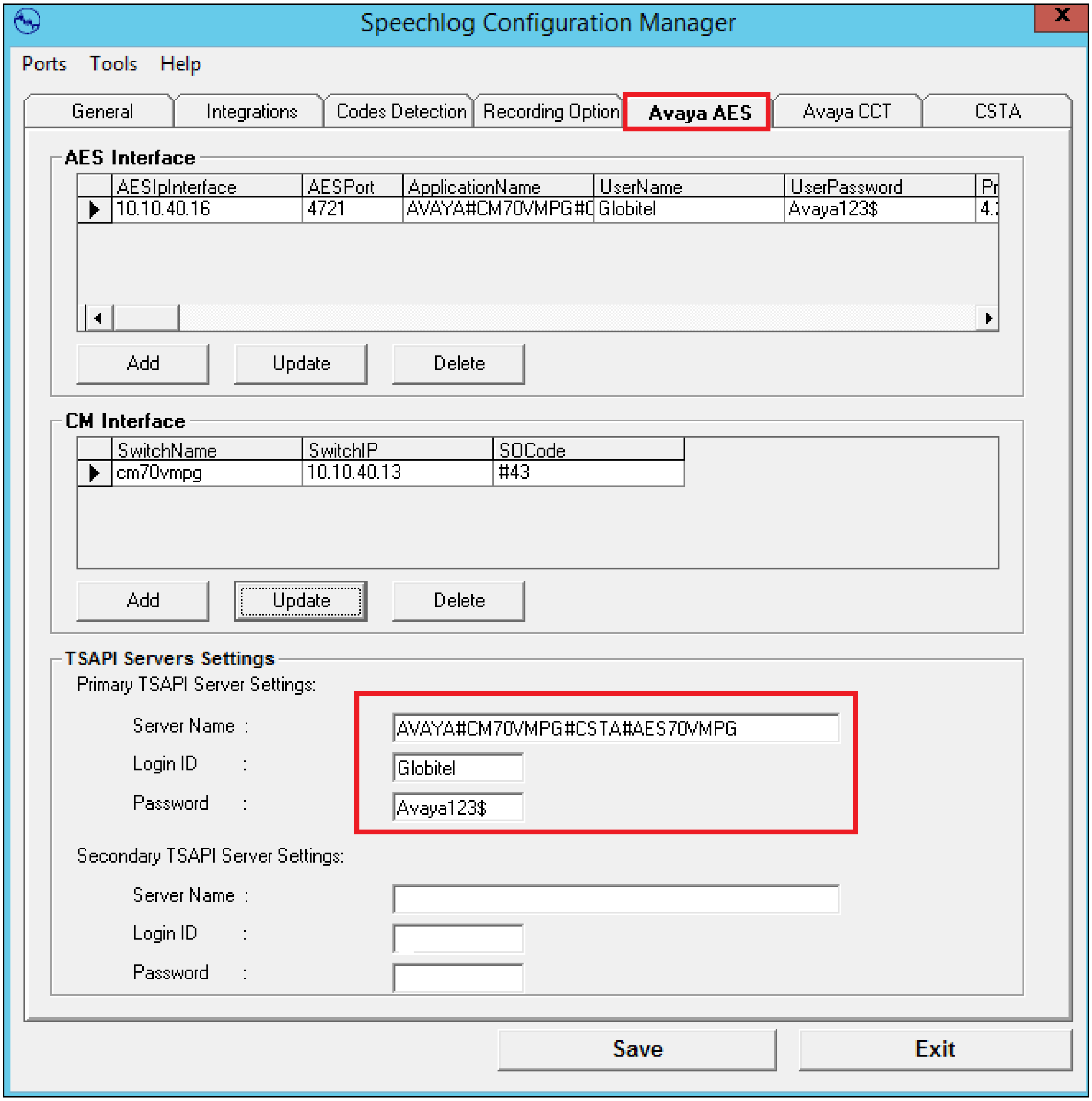
A new Communication Manager interface also needs to be added. Click on **Add** under the **CM Interface** section.



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.



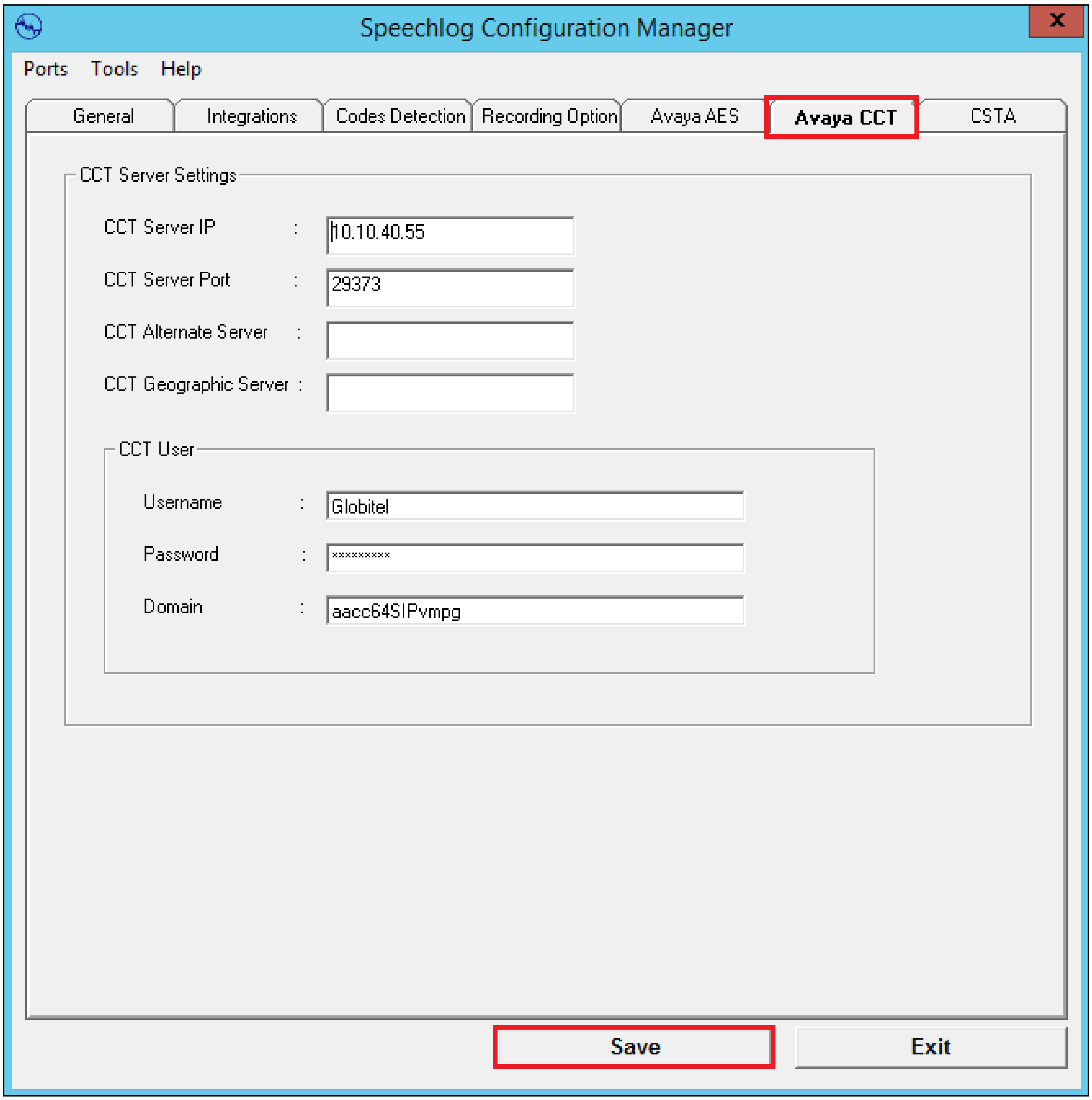
The **TSAPI Servers Settings** must now be completed, enter the Tlink information obtained from **Section 6.4** as the **Server Name**, the **Login ID** and **Password** are that of the AES user created in **Section 6.6**. Click on **Save** to save all the information just entered in this section.



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

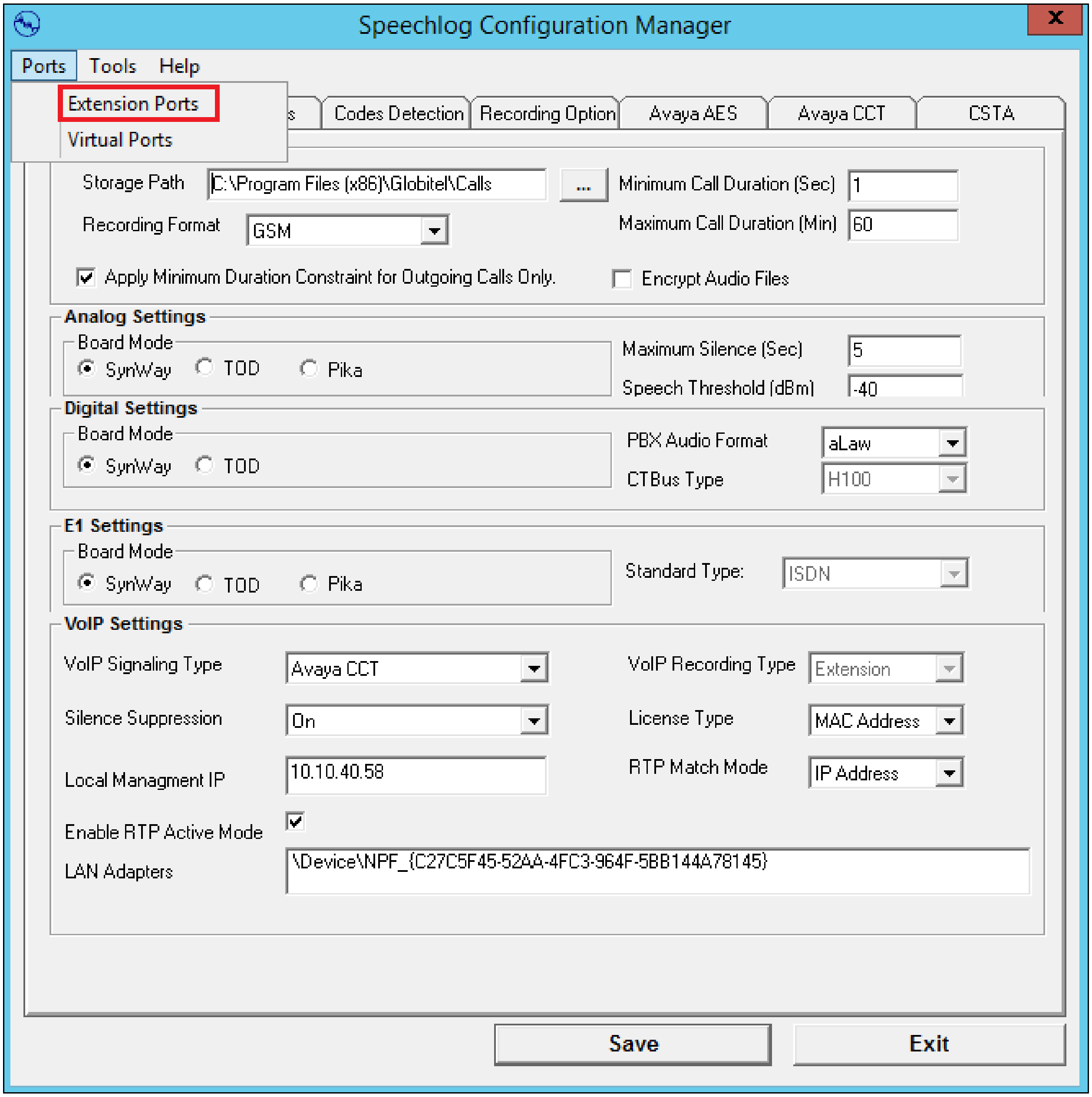
- **CCT Server IP** Enter the IP Address of the Avaya Contact Center Server which is hosting the CCT module.
- **CCT Server Port** This is hardcoded as **29373**.
- **Username** This is the username of the CCT user created in **Section 7.0**.
- **Password** This is the password for the user above.
- **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.

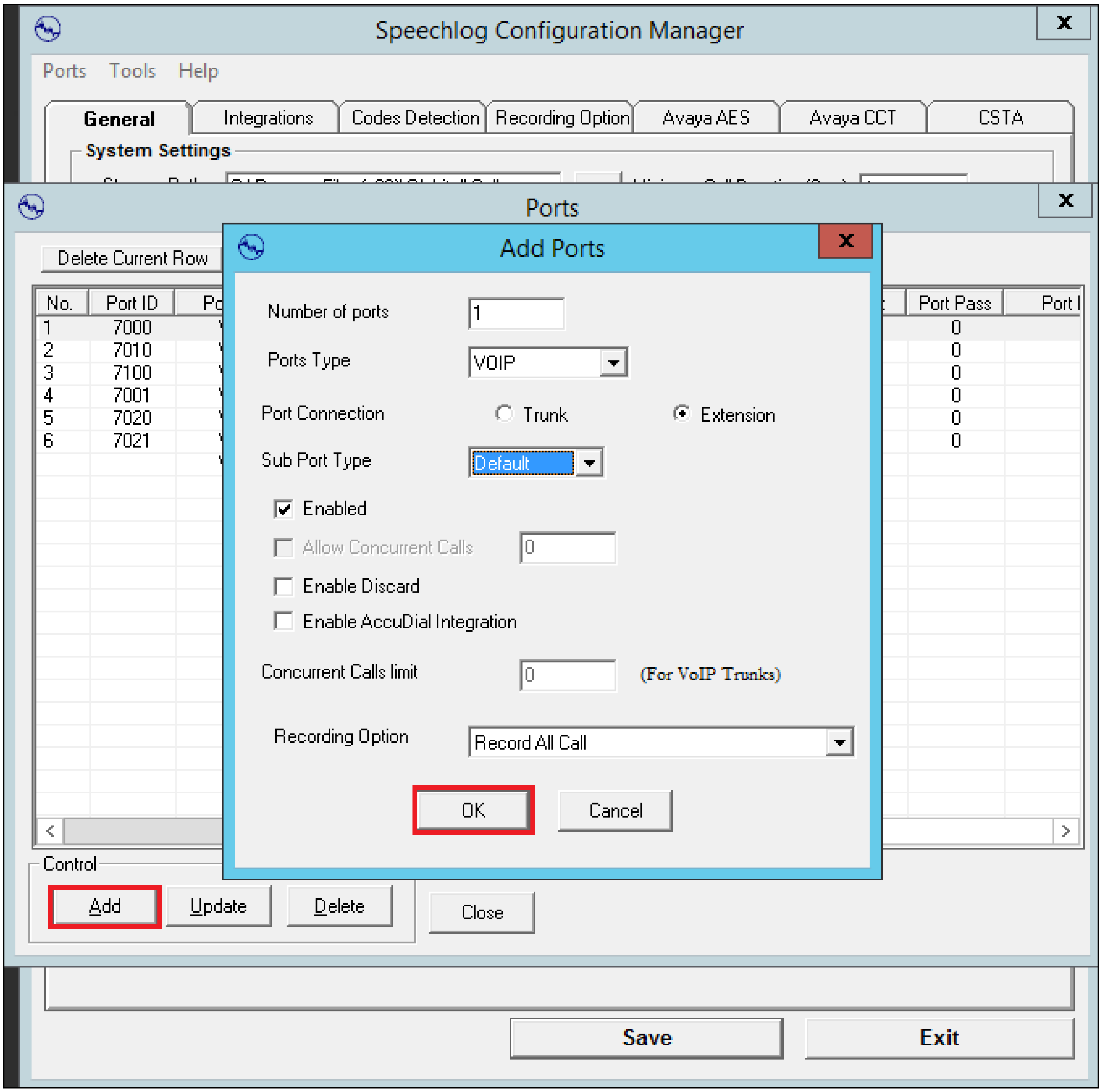


8.2. Configure Extensions to be recorded

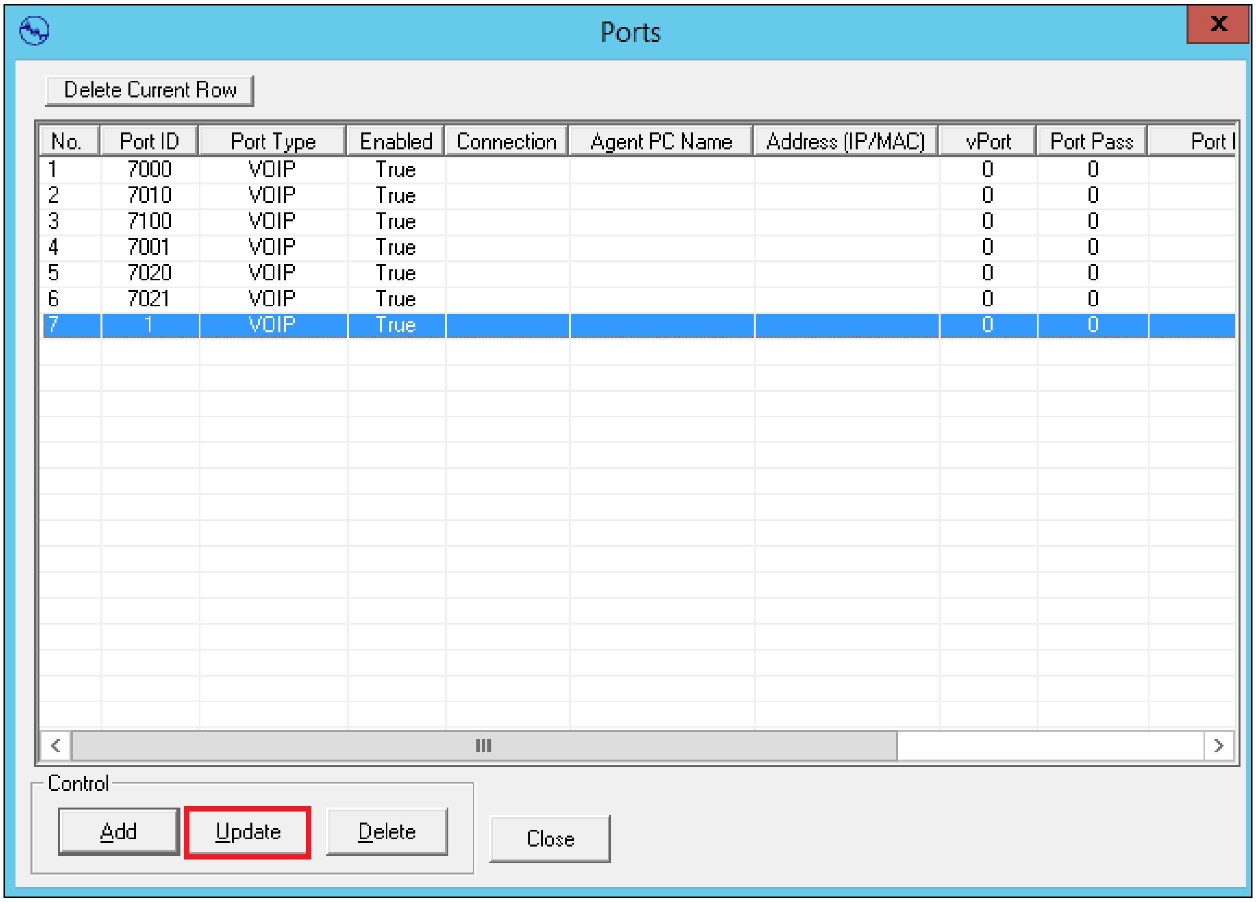
From the top left of the page click on **Ports → Extension Ports**.



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.



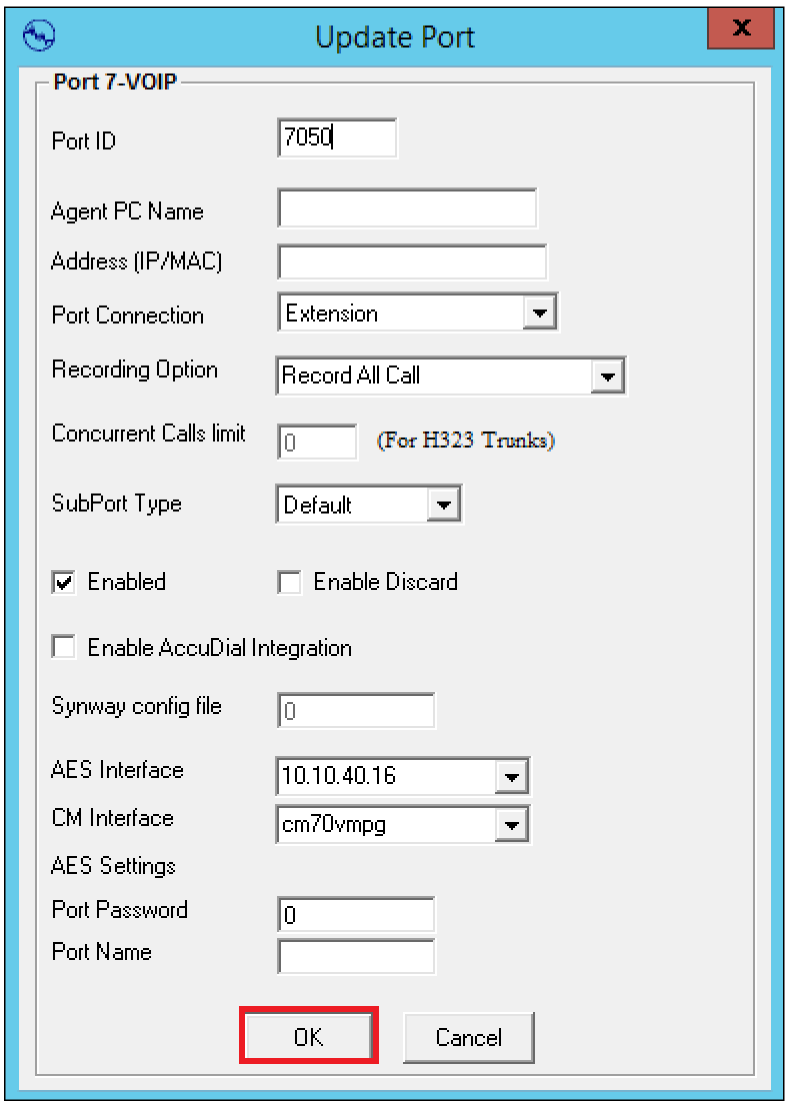
Highlight the new extension and click on **Update**.



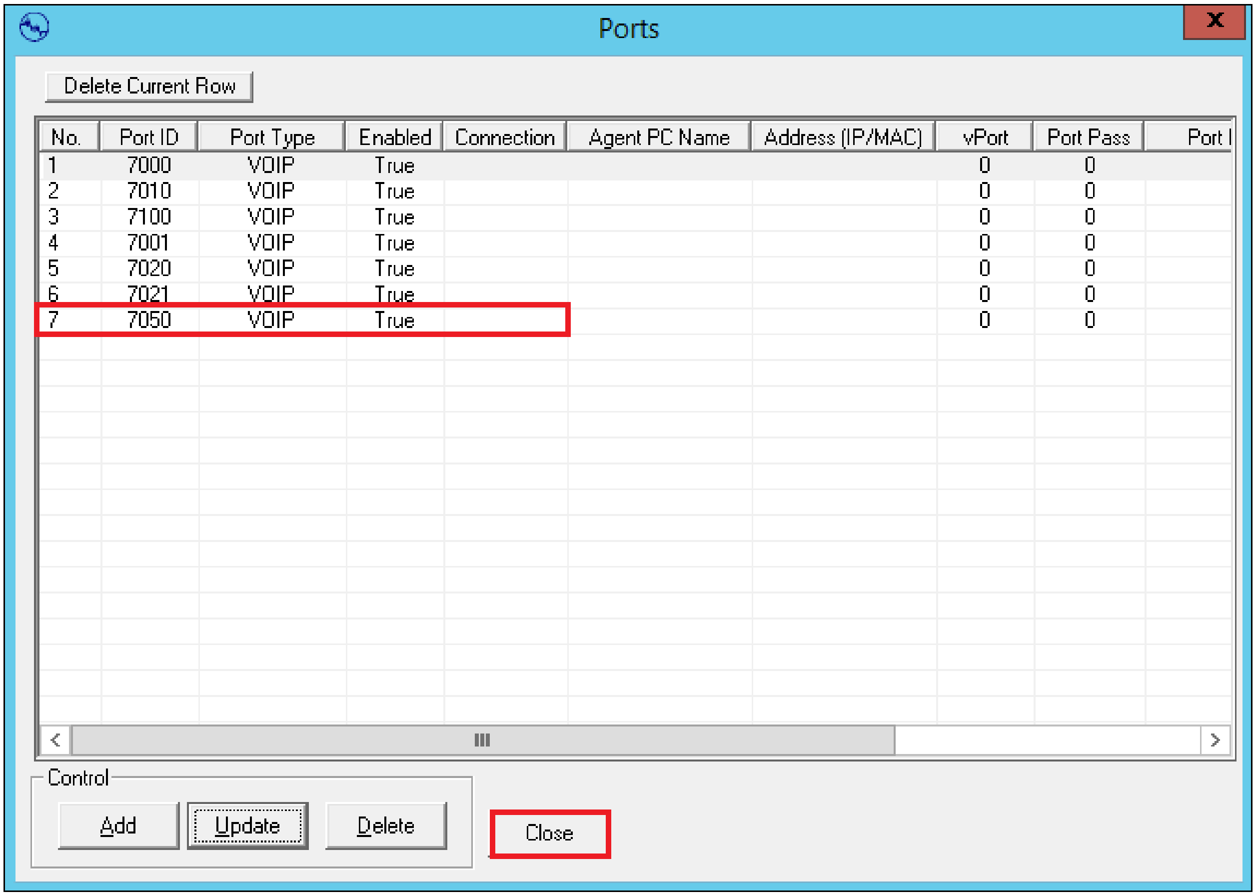
Enter the following information:

- **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.

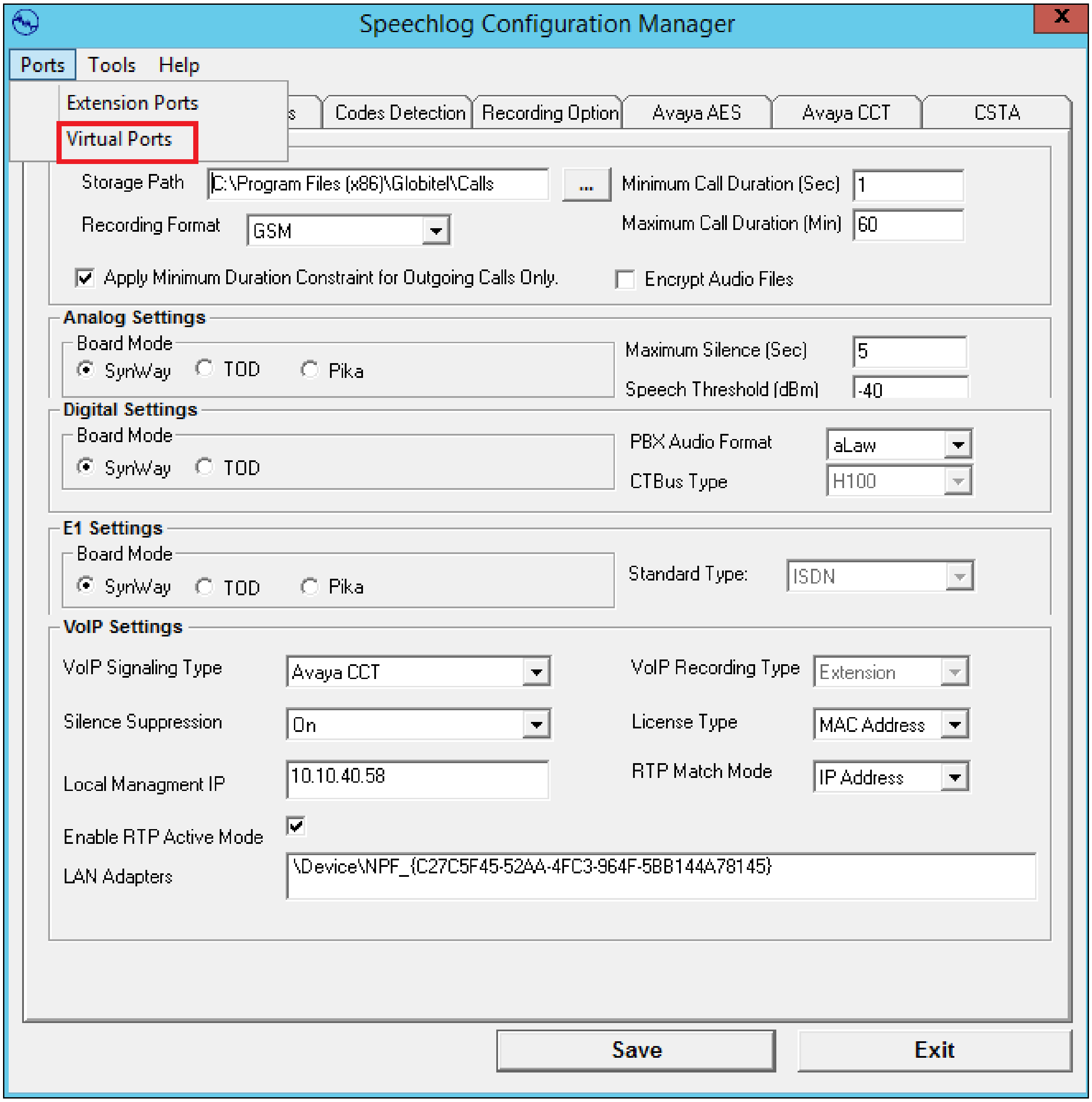


With this new extension updated, click on **Close** to continue.

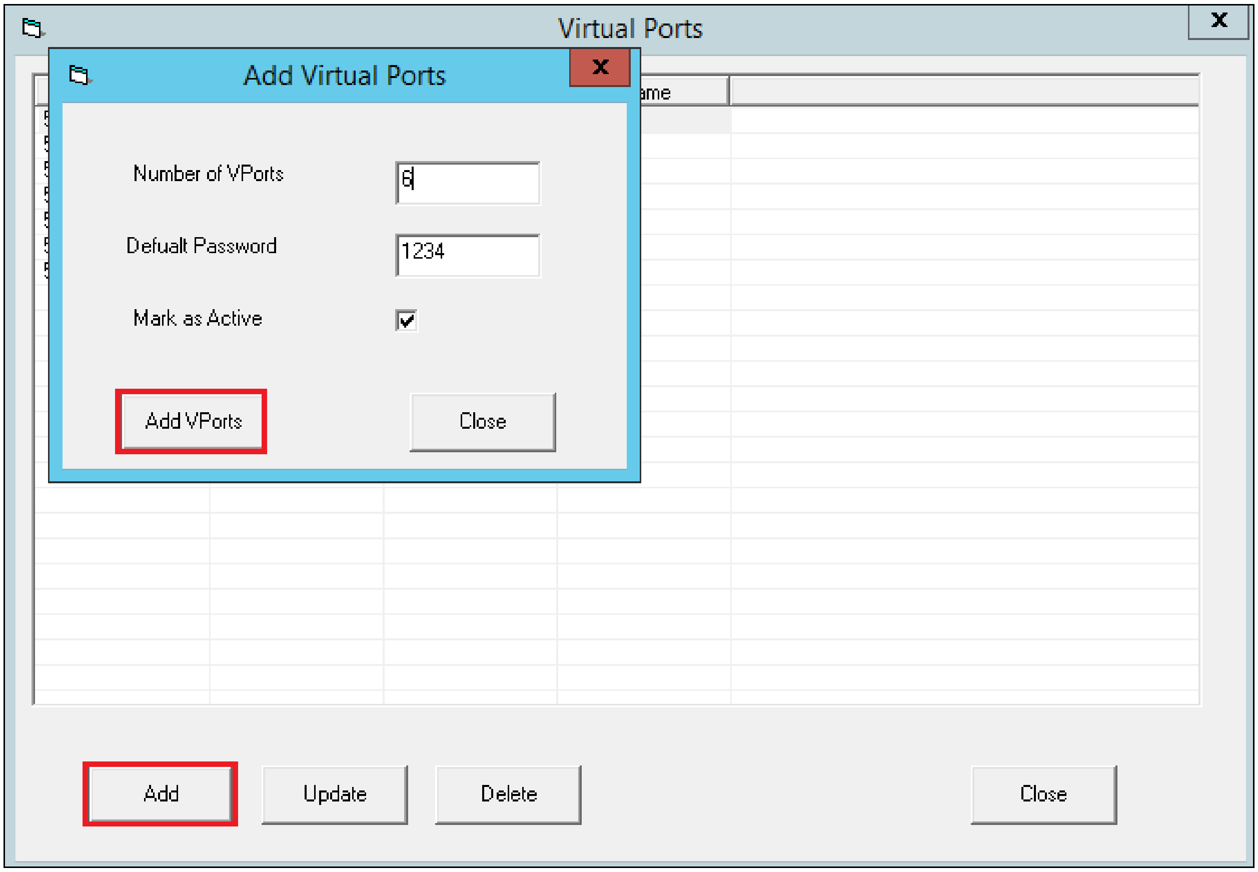


8.3. Configure Virtual Ports

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

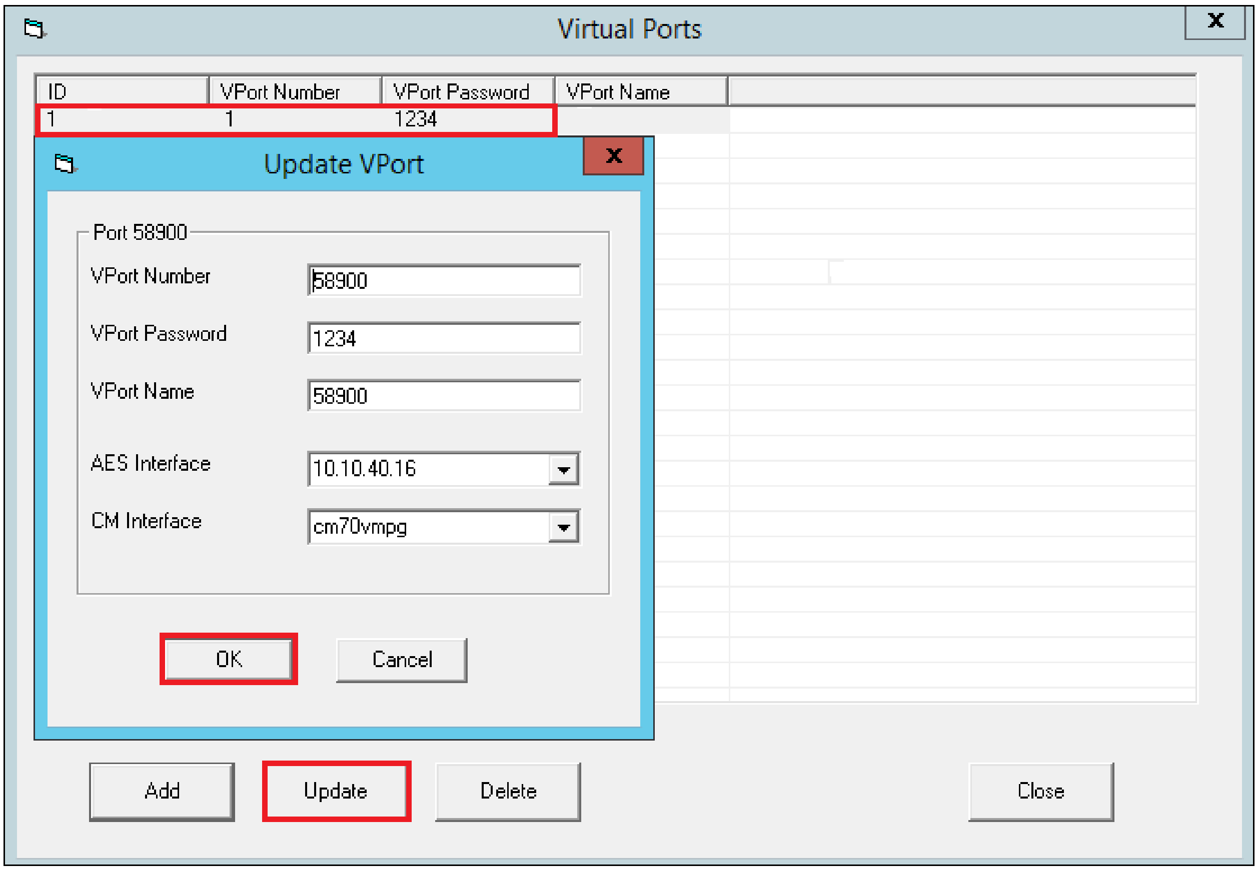


From the **Virtual Ports** window, click on **Add** (not shown). Enter the **Number of VPorts**, in the example below **6** virtual ports were added with a **Default Password** of **1234**, this being the password of the virtual stations created in **Section 5.8**. Click on **Add VPorts**.

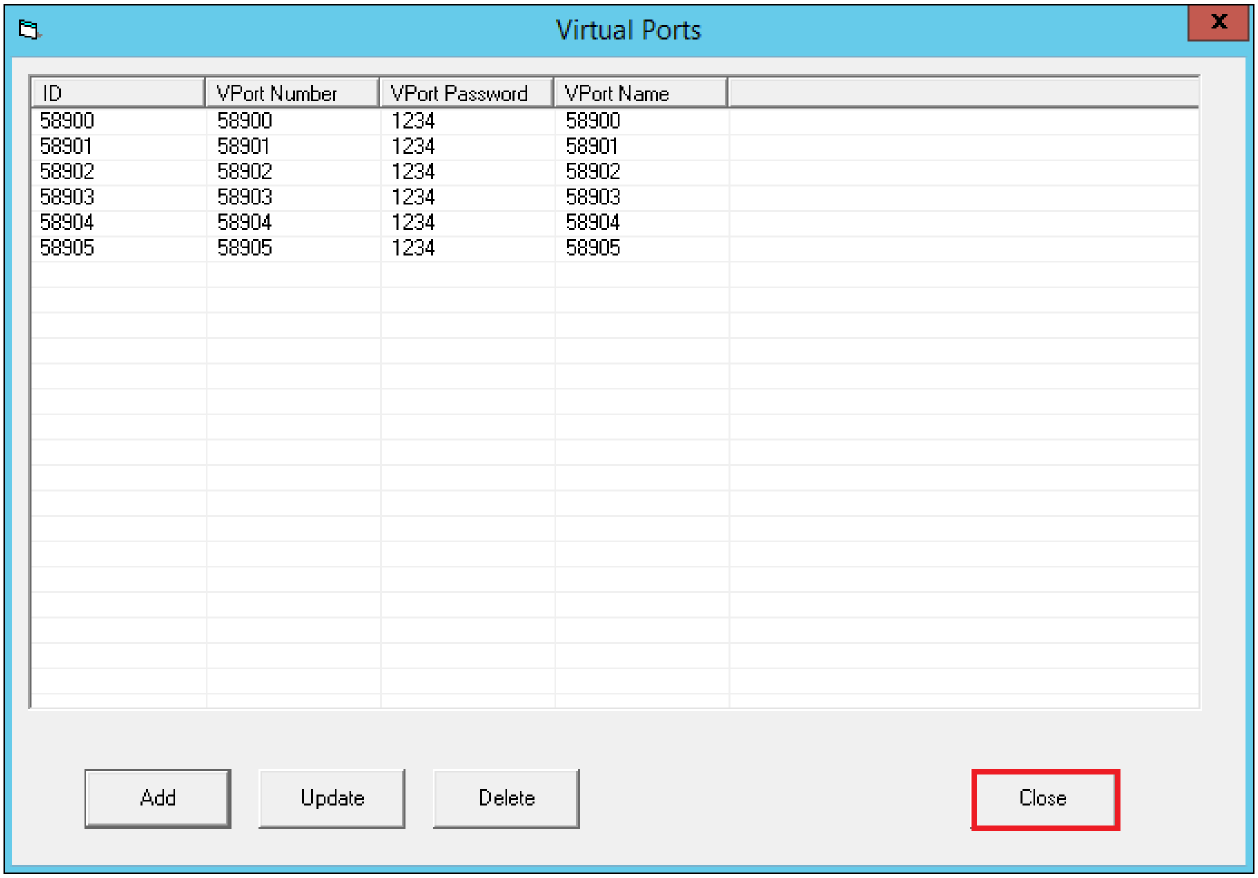


Highlight the new port and click on **Update**, this opens the **Update VPort** window where the following needs to be configured.

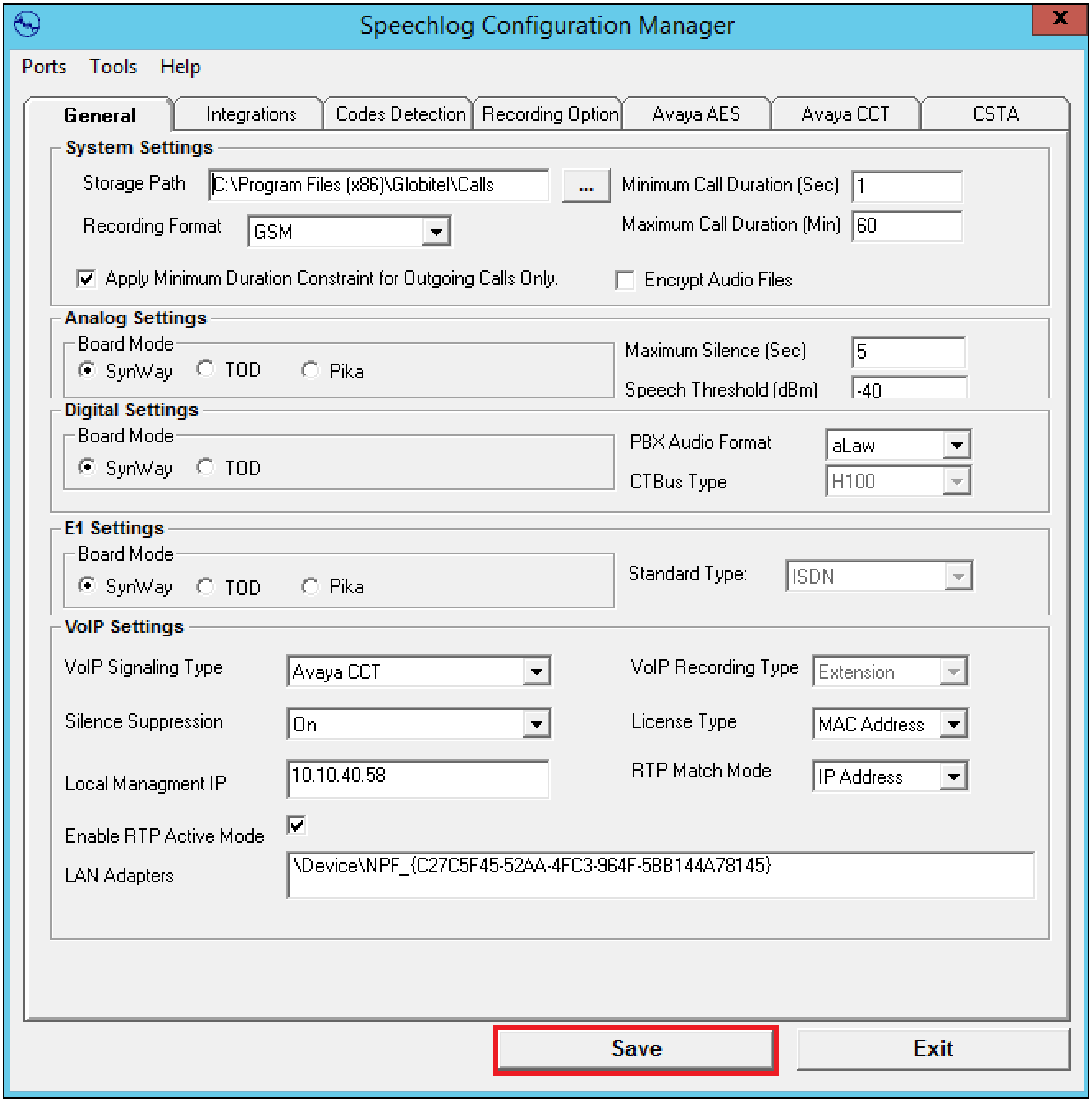
- **VPort Number** Virtual extension number from **Section 5.8**.
- **VPort Password** The password of the virtual extension created in **section 5.8**.
- **VPort Name** Any Suitable name will do, it's recommended to keep this the same as the extension number for convenience.
- **AES Interface** The 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**



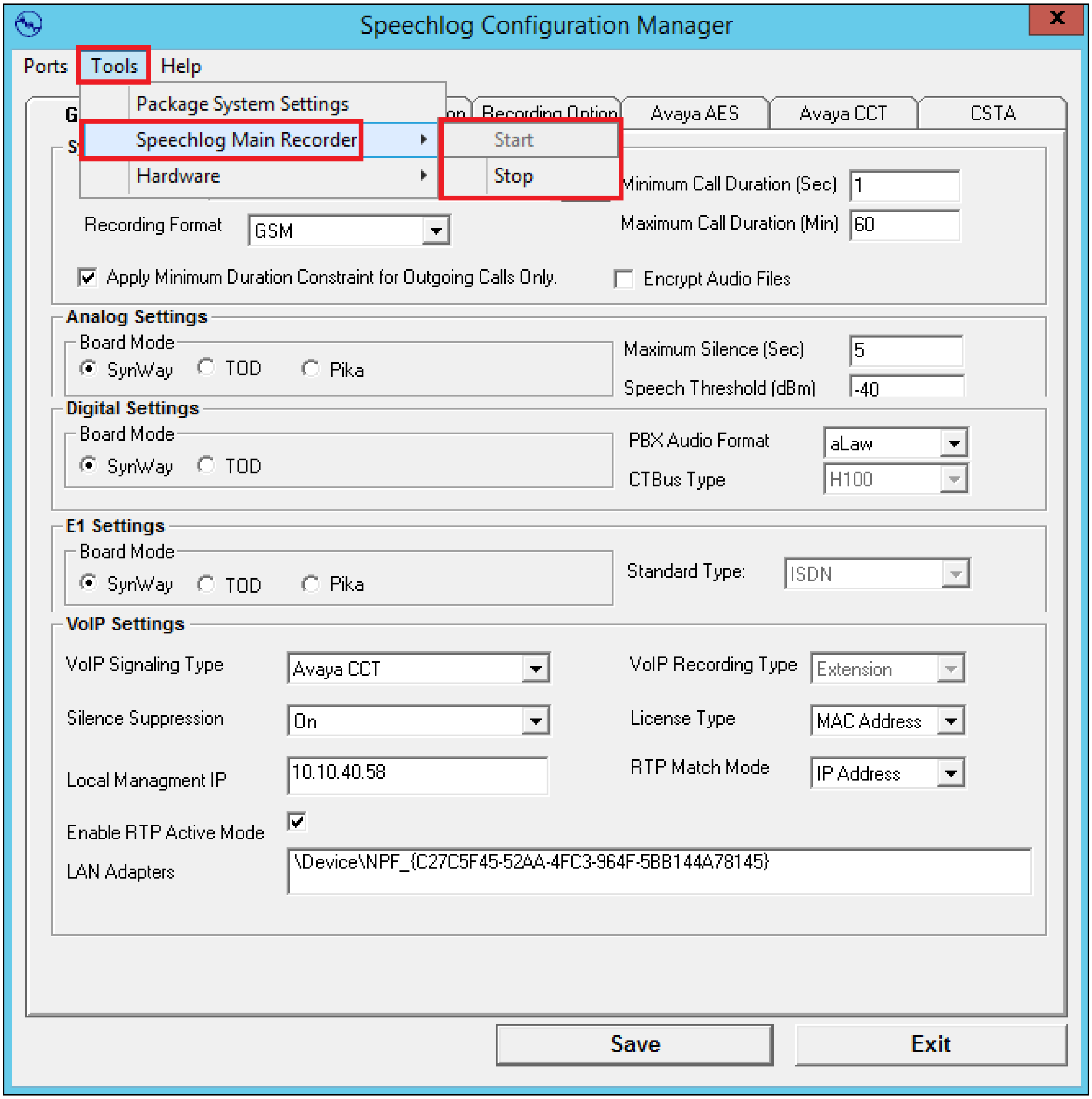
When all the virtual ports are configured click on **Close** to continue.



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



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.

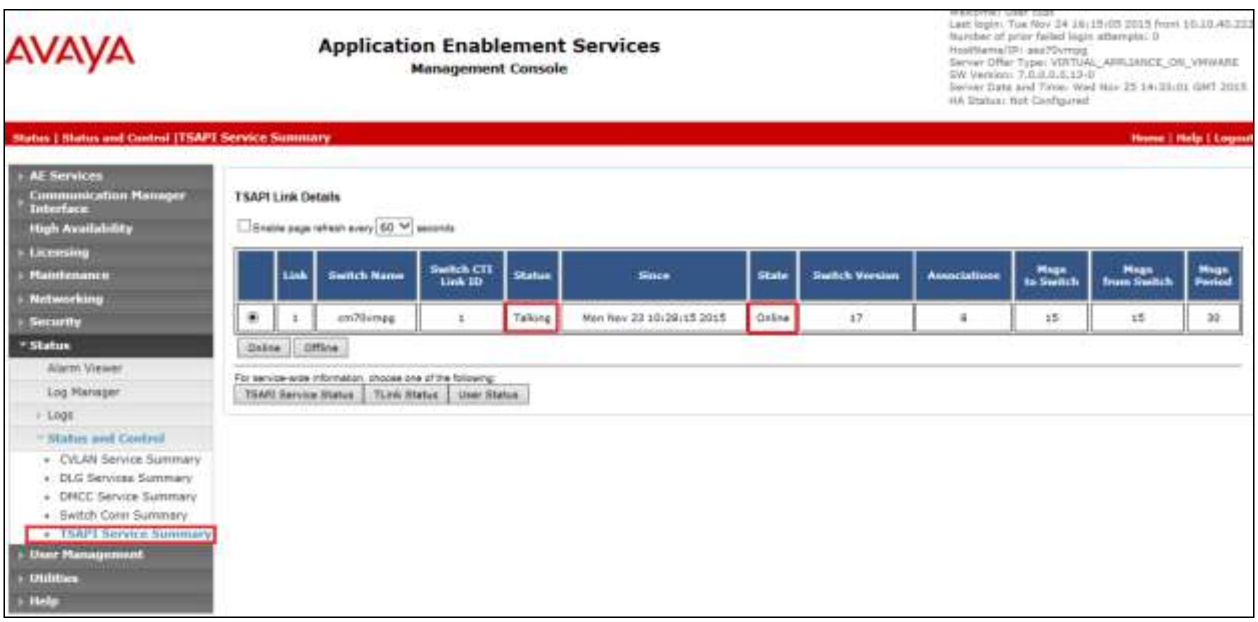
9.1. Verify Avaya Aura® Communication Manager CTI Service State

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   Version     Mnt   AE Services      Service       Msgs     Msgs
Link              Busy    Server          State        Sent     Rcvd

1         4       no     aes70vmpg       established   18       18
```

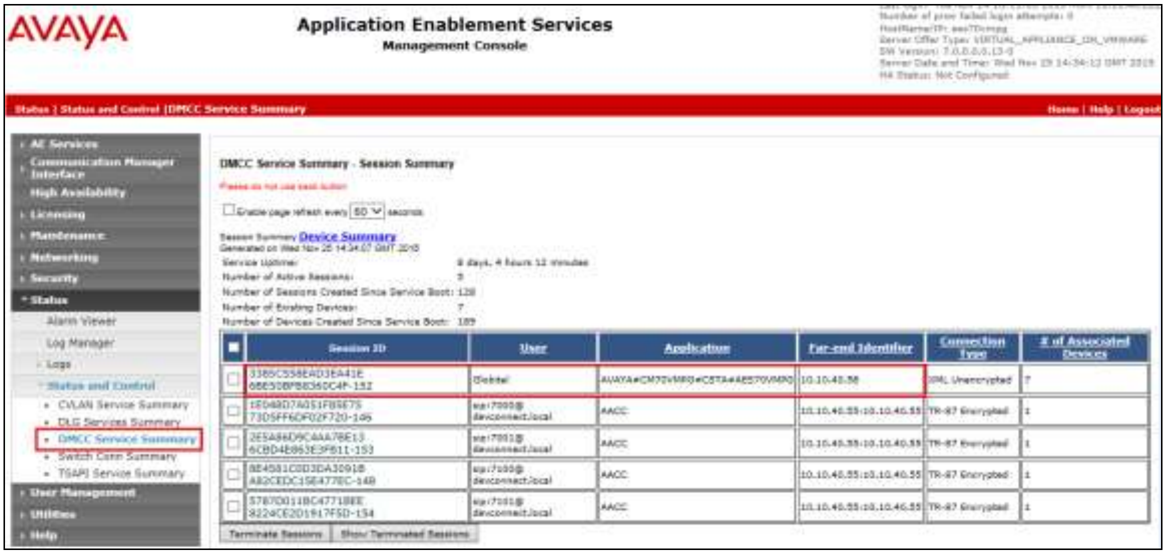
9.2. Verify TSAPI link on AES

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**.



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 **Globite**l is connected from the IP address **10.10.40.58**, which is the SpeechLog Voice Recorder server.

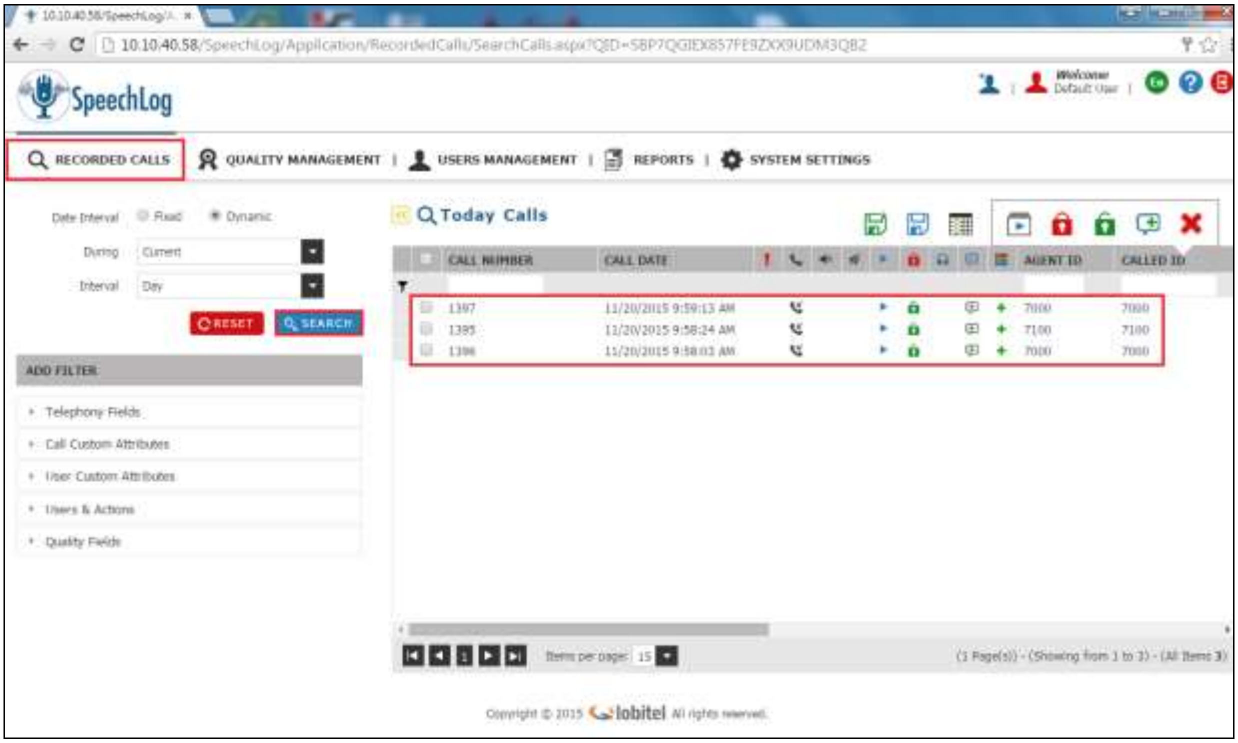


9.4. Verify calls are being recorded

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:

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

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
```

```
display station 58900                                           Page   3 of   6
                                     STATION

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





Enhanced Callr-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
```

```
display station 58900                                           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
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.