Avaya Solution & Interoperability Test Lab

Application Notes for the Amcom PC/PSAP with Avaya Aura™ Communication Manager and Avaya Aura™ Application Enablement Services - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Application Enablement Services, Avaya IP and Digital Telephones, and Amcom PC/PSAP desktop applications.

Amcom PC/PSAP is a powerful Windows-based intelligent E911 workstation solution for a campus or municipality. Using the existing PBX telephone system as an "Automatic Number Identification (ANI)/Automatic Location Information (ALI) controller", Amcom PC/PSAP eliminates the need for external proprietary switching solutions and is able to perform all necessary telephony functions from the call taker's PC keyboard. Amcom PC/PSAP integrates with Amcom CTI Layer, which is a middleware between Amcom PC/PSAP and Avaya Aura™ Application Enablement Services, to control and monitor phone states.

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 a compliance-tested configuration comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Application Enablement Services, Avaya IP and Digital Telephones, and Amcom PC/PSAP applications.

Amcom Communications PC/PSAP is a PC and LAN based system, which allows Communication Manager to be used in a PSAP (Public Safety Answering Position – a physical location where 911 emergency telephone calls are received and then routed to the proper emergency services by the security agent or "911 operator" at the PSAP). Campuses or municipalities can set up a public or private PSAP using Amcom PC/PSAP, which has the capabilities to extract ANI (Automatic Number Identification – phone number of the caller) from Emergency 911 trunks and retrieve corresponding ALI (Automatic Location Information – information about the call based on the ANI such as name, phone number, address, nearest cross street, etc.). Amcom PC/PSAP integrates with Amcom CTI Layer, which is a middleware between Amcom PC/PSAP and Application Enablement Services, to control and monitor phone states.

It is the Amcom CTI Layer service that actually uses the Application Enablement Services Device and Media Control Application Programming Interface (API) to share control of and monitor a physical telephone and receive the same terminal and first party call information received by the physical telephone. Amcom PC/PSAP in turn uses the Amcom CTI Layer service to control and monitor a physical telephone. The PC/PSAP applications regularly provide the Database server with call and lamp state information concerning the controlled telephones.

1.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of the compliance testing was primarily on verifying the interoperability between Amcom PC/PSAP, Application Enablement Services, and Communication Manager.

Amcom PC/PSAP allows a user to operate a physical telephone and view call and telephone display information through a graphical user interface (GUI). During compliance testing, calls were successfully placed to and from Avaya IP and Digital Telephones that were controlled and monitored by the Amcom PC/PSAP application. Basic telephone and call operations on Avaya IP and Digital telephones were tested using the Amcom desktop application.

1.2. Support

Technical support for the Amcom PC/PSAP solution can be obtained by contacting Amcom:

- URL – https://secure5.inet7.com/amcomsoftware-com/Support/online.aspx
- Phone – (888) 797-7487

2. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with an Application Enablement Services server and Avaya S8720 Media Servers with G650 Media Gateway. The PC/PSAP was located in a different VLAN. Endpoints include Avaya 9600 Series H.323 IP Telephones, Avaya 4625 H.323 IP Telephone, and an Avaya 6408D Digital Telephone. An Avaya S8300 Server with an Avaya G450 Media Gateway was included in the test to provide an inter-switch scenario.

Note: Basic administration of Application Enablement Services server is assumed. For details, see [2].

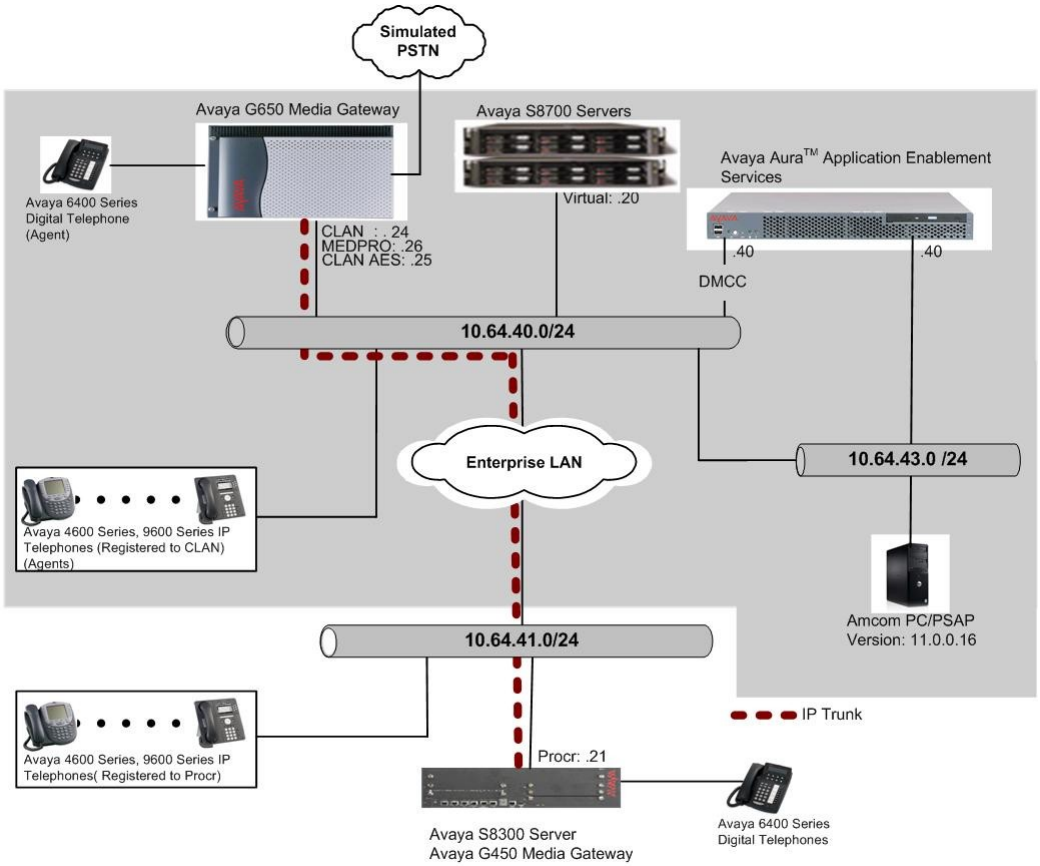


Figure 1: Sample Configuration.

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8720 Media Servers	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface TN799DP C-LAN Interface TN2302AP IP Media Processor	FW40 FW34 FW51
Avaya S8300C Media Server with Avaya G450 Media Gateway	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4)
Avaya Aura™ Application Enablement Services Server	5.2 (r5-2-0-98-0)
Avaya 4625SW IP Telephone	2.5
Avaya 9600 Series IP Telephones	
9620 (H.323) 9630 (H.323) 9650 (H.323)	3.1 3.1 3.1
Avaya 6424D+ Digital Telephone	-
Amcom PC/PSAP	11.0.0.16

4. Configure Avaya Communication Manager

This section describes the procedure for administering the transport link to the AES server, setting up a Feature Access Codes. Abbreviated dialing, and controlled telephones.

Log into the System Access Terminal (SAT) to verify that Avaya Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Verify that the **ASAI Link Core Capabilities**, and **Computer Telephony Adjunct Links** customer option is set to "y" on (Page 3) and there are sufficient **IP_API_A** licenses (Page 10). If not, then contact the Avaya sales team or business partner for a proper license file.

4.1. Configure IP Services for AES

Notes: *The section 4.1 was performed at the Avaya S8720 Media Server with an Avaya G650 Media Gateway side.*

Enter the **change node-names ip** command. In the compliance-tested configuration, the CLAN IP address was used for registering H.323 endpoints, and the CLAN-AES IP address was used for connectivity to Application Enablement Services.

```
change node-names ip                                          Page   1 of   1
                                  IP NODE NAMES
    Name                 IP Address          Name              IP Address
CDR_buffer             192.45 .80 .250                          .   .   .
CLAN                   10.64.40.24                              .   .   .
CLAN-AES               10.64.40.25                              .   .   .
G350                   10.64.42.21                              .   .   .
MEDPRO                 10.64.40.26                              .   .   .
S8300                  10.64.41.21                              .   .   .
default                0   .0   .0   .0                        .   .   .
```

Enter the **change ip-services** command. On **Page 1**, configure the Service Type field to **AESVCS** and the Enabled field to **y**. The Local Node field should be pointed to the **CLAN-AES** board that was configured previously in the IP NODE NAMES form in this section. During the compliance test, the default port was used for the Local Port field.

```
change ip-services                                            Page   1 of   4

                                  IP SERVICES
 Service      Enabled      Local        Local       Remote       Remote
  Type                     Node         Port        Node         Port
AESVCS         y       CLAN-AES          8765
```

On **Page 4**, enter the hostname of the Application Enablement Services server for the AE Services Server field. The server name may be obtained by logging in to the Application Enablement Services server using ssh, and running the command **uname –a**. Enter an

alphanumeric password for the Password field. Set the Enabled field to **y**. The same password will be configured on the Application Enablement Services server in **Section 5.2**.

```
change ip-services                                              Page   4 of   4
                           AE Services Administration

   Server ID    AE Services       Password          Enabled     Status
                   Server
      1:       server1           xxxxxxxxxxxxxxx       y         idle
      2:
      3:
      4:
      5:
```

4.2. Configure feature-access-codes

Notes: The sections, 4.2, 4.3, 4.5, and 4.6, were performed at the Avaya S8300 Media Server with an Avaya G450 Media Gateway side.

Enter the **change feature-access-codes** command. On **Page 1** of the FEATURE ACCESS CODE (FAC) form, verify the Auto Route Selection (ARS) – Access Code 1 field is set to **9**.

```
change feature-access-codes                                     Page   1 of   7
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:

      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                  Automatic Callback Activation:      Deactivation:
Call Forwarding Activation Busy/DA:        All:       Deactivation:
```

4.3. Configure Dialplan

Enter the **change dialplan analysis** command. Create a single digit dial string with 9 and associate it with **Feature Access Code (fac)**.

```
change dialplan analysis                                        Page   1 of  12
                           DIAL PLAN ANALYSIS TABLE
                                                          Percent Full:    1

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    attd        4       5    ext
        10      4    dac         5       5    ext
        11      3    dac         6       5    ext
        12      3    fac         7       5    ext
        126     6    aar         8       1    fac
        13      3    fac         9       1    fac
        14      3    fac         *       3    fac
        15      3    fac         #       3    fac
```

4.4. Configure Hunt Group

Notes: *The section 4.4 was performed at the Avaya S8720 Media Server with an Avaya G650 Media Gateway side.*

Enter the **add hunt-group n** command, where **n** is an unused hunt group number. On **Page 1** of the HUNT GROUP form, assign a descriptive Group Name and Group Extension valid in the provisioned dial plan at the S8720 Media Server with a G650 Media Gateway side.

```
add hunt-group 51                                          Page   1 of  60
                                HUNT GROUP

            Group Number: 51                              ACD? n
              Group Name: 911-hunt                       Queue? n
         Group Extension: 22999                         Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1           Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:
```

On **Page 3**, add the 911 member extensions, which will be forwarded when a caller dials 911.

```
add hunt-group 51                                          Page   3 of  60
                                HUNT GROUP
          Group Number: 51      Group Extension: 22999     Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /3
                                     Total Administered Members: 3
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)           Ext      Name (24 characters)
   1: 22001                                 14:
   2: 22002                                 15:
   3: 22003                                 16:
   4:                                       17:
   5:                                       18:
```

4.5. Configure Auto Route Selection (ARS)

Enter the **change ars analysis 11** command. When a caller dials 911, the first digit (9) indicates that it is an ARS call. Therefore, the next two digits (11) are utilized in the ARS table.

```
change ars analysis 11                                       Page   1 of   2
                           ARS DIGIT ANALYSIS TABLE
                                  Location:  all            Percent Full:    1

          Dialed           Total     Route    Call    Node  ANI
          String          Min  Max  Pattern   Type    Num   Reqd
    11                    2    2    51        emer          n
    120                   11   11   deny      fnpa          n
    1200                  11   11   deny      fnpa          n
    121                   11   11   deny      fnpa          n
```

4.6. Configure Route Pattern

Enter the **change route-pattern r** command, where r is a route-pattern number. In the following route-pattern, two digits (11) are removed and replace it with 22999 (911 hunt group extension). The extension, 22999, will be sent to the trunk group 51.

```
display route-pattern 51                                        Page   1 of   3
                   Pattern Number: 51  Pattern Name: temp-911
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 51   0                    2   22999                               n   user
 2:                                                                  n   user
 3:                                                                  n   user
     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                  Dgts Format
                                                          Subaddress
 1: y y y y y n  n            rest                                     none
 2: y y y y y n  n            rest                                     none
 3: y y y y y n  n            rest                                     none
```

4.7. Configure Feature Access Codes (FAC)

Notes: *The section 4.7 to 4.9 was performed at the Avaya S8720 Media Server with an Avaya G650 Media Gateway side.*

Enter the **display feature-access-codes** command. On **Page 5** of the FEATURE ACCESS CODE (FAC) form, configure and enable the following access codes:

- After Call Work Access Code
- Auto-In Access Code
- Aux Work Access Code
- Login Access Code
- Logout Access Code

```
display feature-access-codes                                    Page   5 of   9
                            FEATURE ACCESS CODE (FAC)

                        Automatic Call Distribution Features

                  After Call Work Access Code: 120
                         Assist Access Code: 121
                       Auto-In Access Code: 122
                      Aux Work Access Code: 123
                         Login Access Code: 124
                       Logout Access Code: 125
                      Manual-in Access Code: 126
     Service Observing Listen Only Access Code: 127
     Service Observing Listen/Talk Access Code: 128
         Service Observing No Talk Access Code:
                   Add Agent Skill Access Code: 130
                Remove Agent Skill Access Code: 131
             Remote Logout of Agent Access Code: 132
```

4.8. Configure Abbreviated Dialing

Enter the **add abbreviated-dialing group g** command, where **g** is the number of an available abbreviated dialing group. In the **DIAL CODE** list, enter the Feature Access Codes for ACD Login and Logout from **Section 4.7**.

```
add abbreviated-dialing group 1                                  Page   1 of   1
                        ABBREVIATED DIALING LIST

              Group List: 1        Group Name: Call Center
    Size (multiple of 5): 5       Program Ext:              Privileged? n
DIAL CODE
     11: 124
     12: 125
     13:
```

4.9. Configure Controlled Telephones

Enter the **change station r** command, where **r** is the extension of a registered, physical Avaya IP or Digital telephone. On **Page 1** of the STATION form, enter a phone Type, descriptive name, Security Code and set IP SoftPhone field to **y** to allow the physical station to be controlled by a softphone such as the CTI Layer application.

```
add station 22001                                                Page   1 of   5
                                    STATION

Extension: 22001                       Lock Messages? n                BCC: 0
     Type: 4625                        Security Code: *                 TN: 1
     Port: S00416                    Coverage Path 1:                  COR: 1
     Name: DMCC-1                    Coverage Path 2:                  COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                              Message Lamp Ext: 22001
           Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english             Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y

                                            IP Video Softphone? n
```

On **Page 4** of the station form, for **ABBREVIATED DIALING List 1**, enter the abbreviated dialing group configured in **Section 4.8**. On **Pages 4** and **5** of the station forms, configure the following BUTTON ASSIGNMENTS in addition to the call-appr (call appearance) buttons:

- aux-work
- abrv-dial – configure two of these buttons, one for Login and one for Logout, along with the Dial Codes from Abbreviated Dialing **List1** for ACD Login and Logout, respectively.
- after-call
- auto-in (On Page 5)
- release (On Page 5)

```
add station 22001                                          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: personal 1         List2: group     1        List3:

BUTTON ASSIGNMENTS
 1: call-appr                              5: aux-work     RC:     Grp:
 2: call-appr                              6: abrv-dial   List: 2 DC: 11
 3: brdg-appr  B:1  E:22101                7: abrv-dial   List: 2 DC: 12
 4: brdg-appr  B:2  E:22101                8: after-call           Grp:
```

```
add station 22001                                          Page   5 of   5
                                    STATION

FEATURE BUTTON ASSIGNMENTS

 9: auto-in              Grp:
10: release
```

Repeat the instructions provided in this section for each physical station that is to be controlled / monitored by an Amcom CTI Layer.

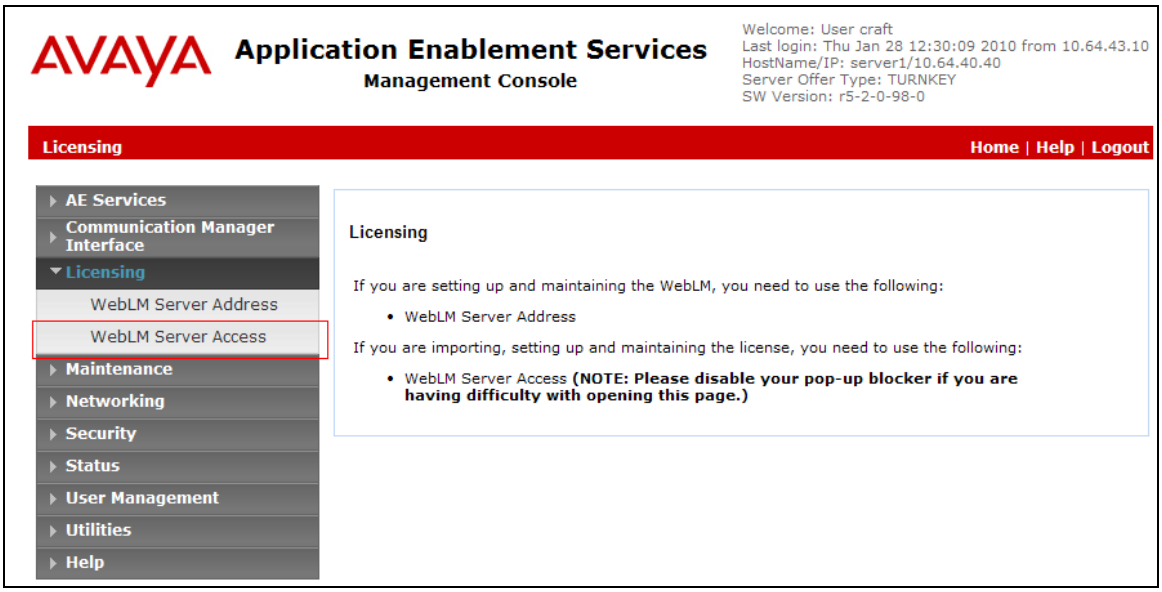
5. Configure Avaya Application Enablement Services

The Avaya Application Enablement Services server enables Computer Telephony Interface (CTI) applications to control and monitor telephony resources on Communication Manager.

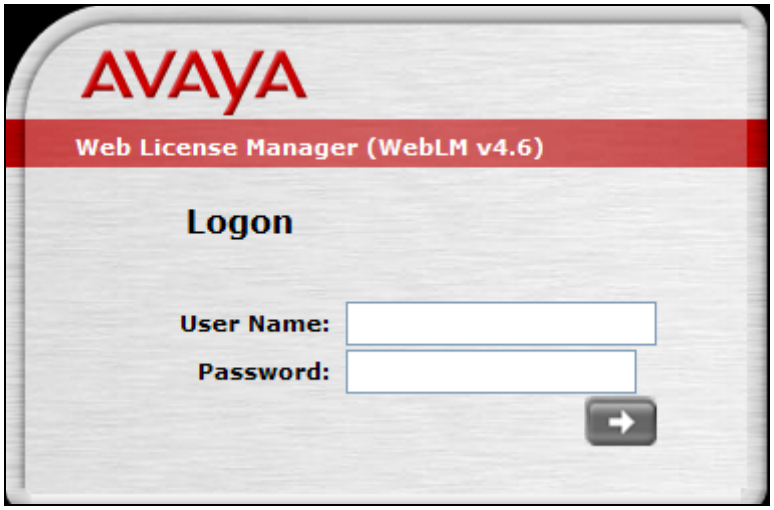
This section assumes that installation and basic administration of the Application Enablement Services server has been performed. The steps in this section describe the configuration of a Switch Connection, a CTI user, a CMAPI port.

5.1. Device and Media Control API Station Licenses

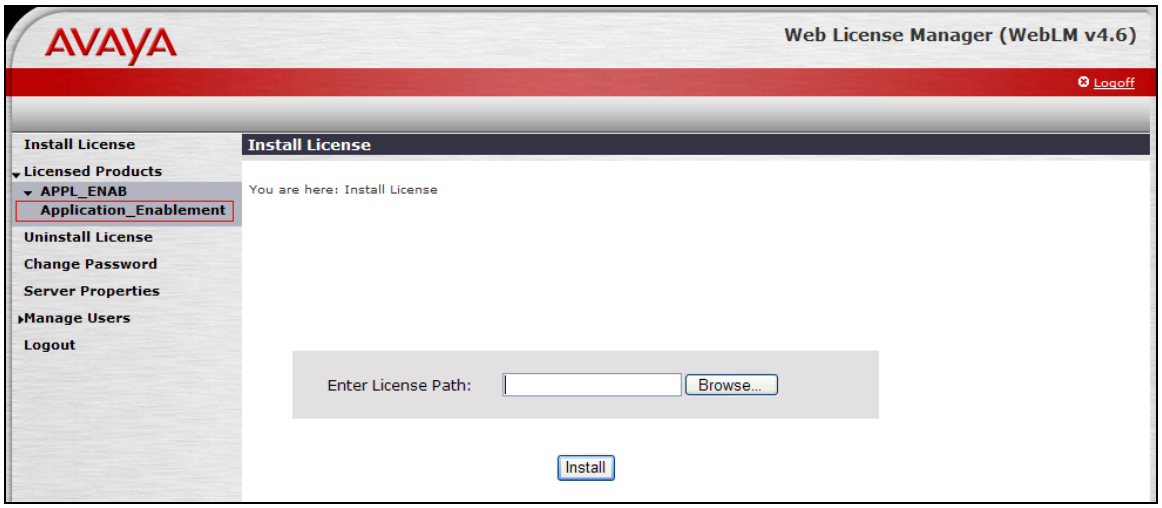
The Amcom CTI Service instances appear as "virtual" stations/softphones to Communication Manager. Each of these virtual stations, hereafter called Device and Media Control API station, requires a license. Note that this is separate and independent of Avaya IP Softphone licenses, which are required for Avaya IP Softphones but not required for Device and Media Control API stations. To check and verify that there are sufficient DMCC licenses, log in to https://<IP address of the Application Enablement Services server>/index.jsp, and enter appropriate login credentials to access the Application Enablement Services Management Console page. Select the **Licensing → WebLM Server Access** link from the left pane of the window.



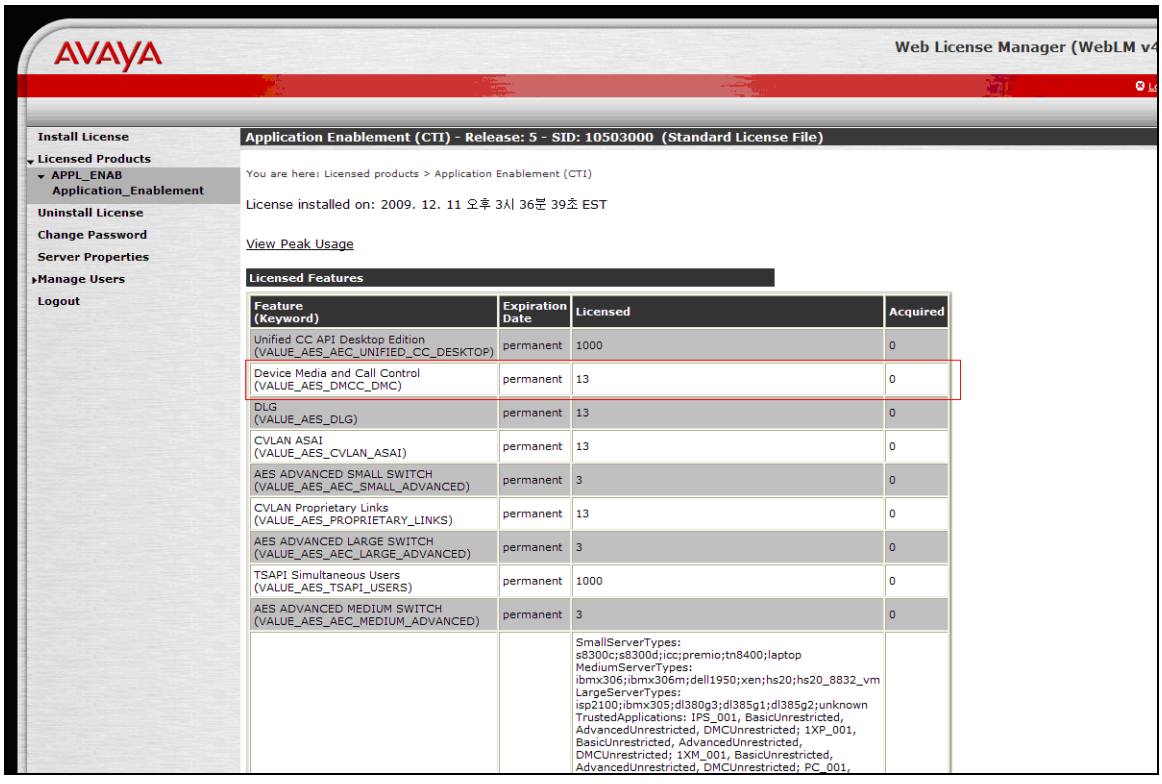
Provide appropriate login credentials to access the Web License Manager page.



On the Install License page, select **License Products → Application_Enablement** link from the left pane of the window.

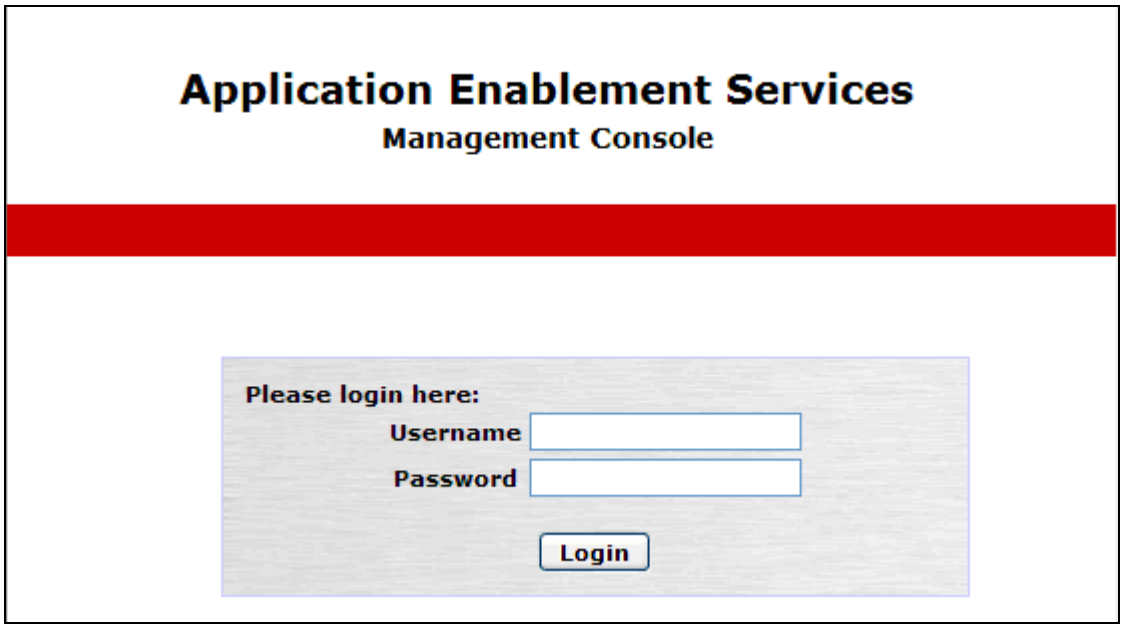


On the Licensed Features page, verify that there are sufficient DMCC licenses.

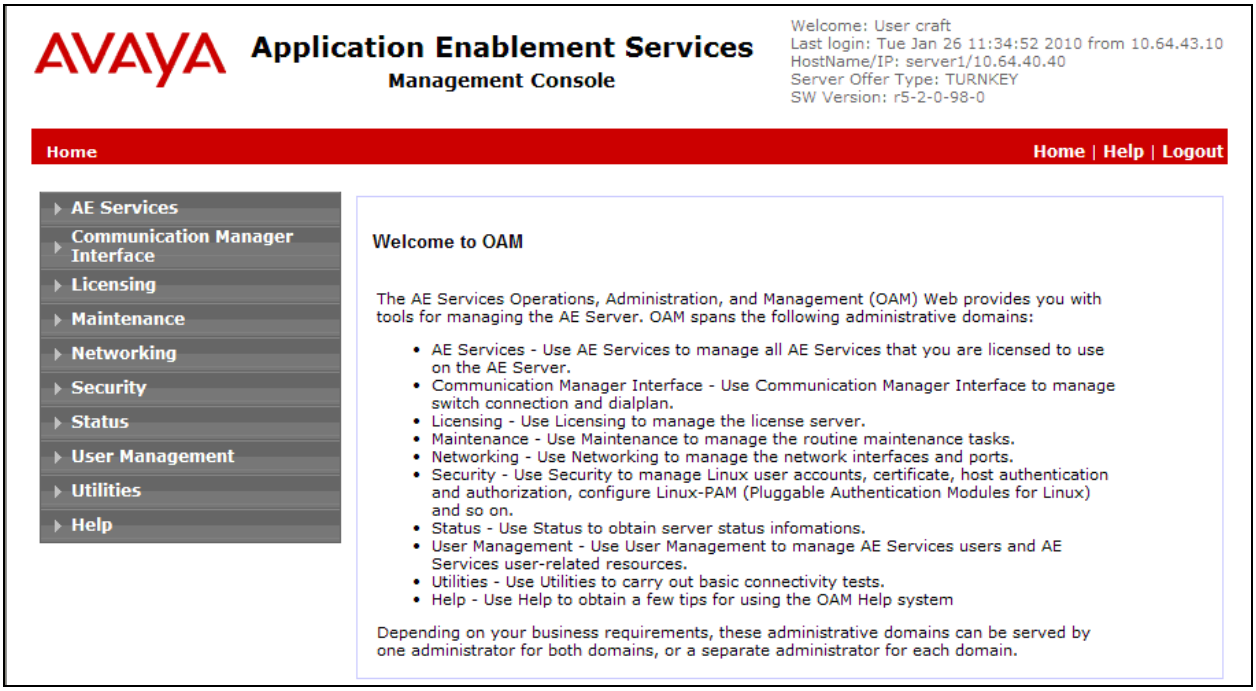


5.2. Configure Switch Connection

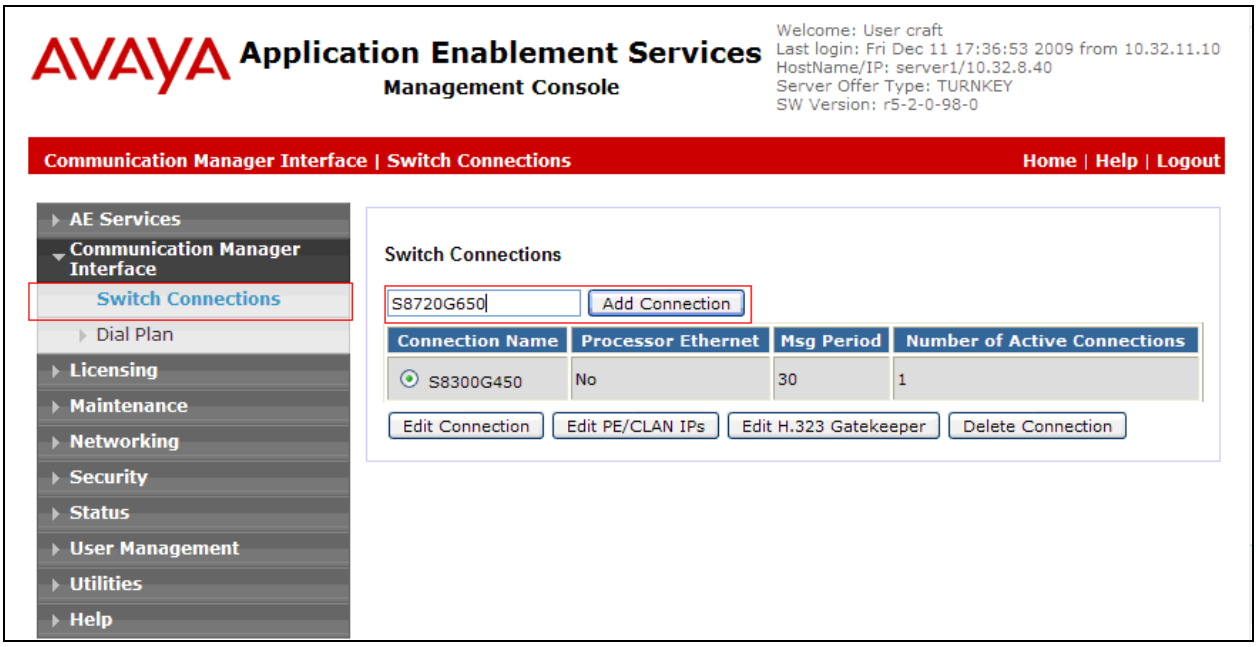
Launch a web browser, enter https://<IP address of the Application Enablement Services server> in the address field, and log in with the appropriate credentials for accessing the Application Enablement Services Management Console pages.



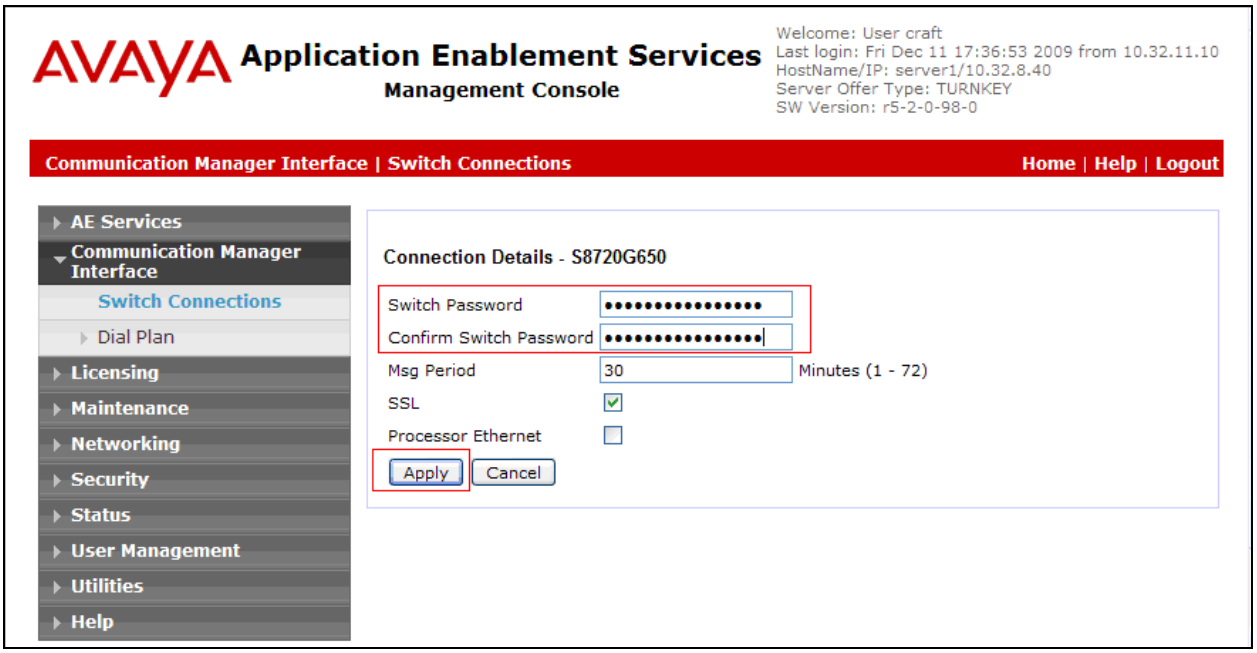
Click on **Communication Manager Interface → Switch Connection** in the left pane to invoke the Switch Connections page.



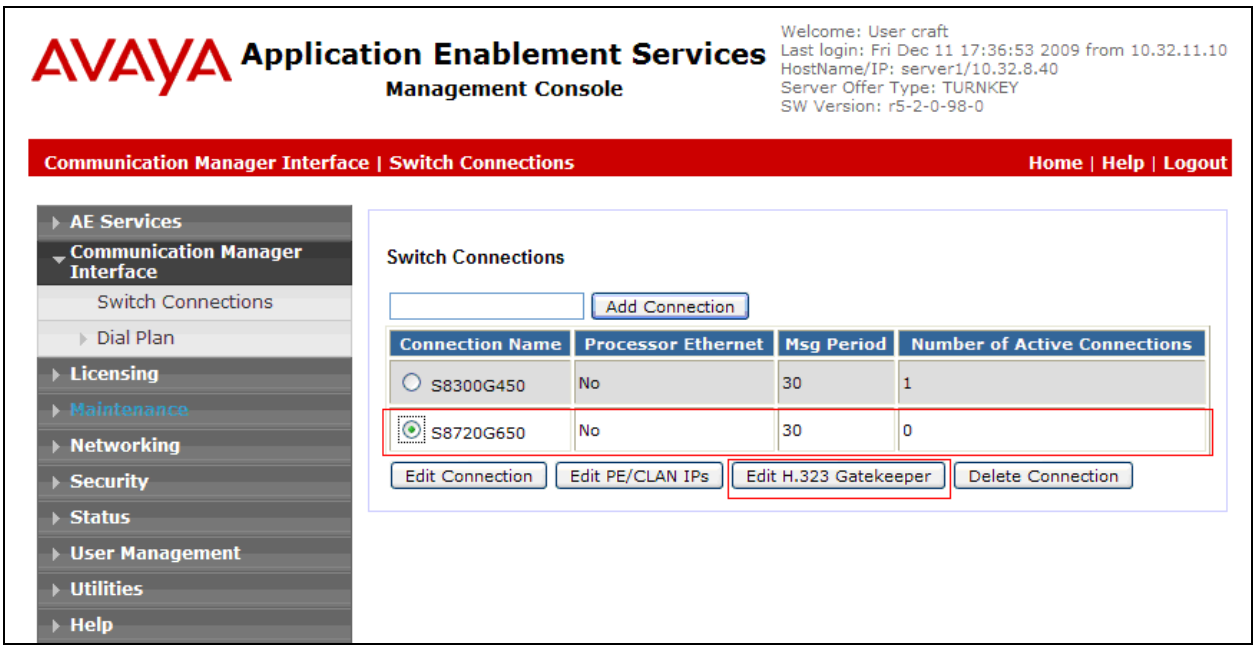
A Switch Connection defines a connection between the Application Enablement Services server and Communication Manager. Enter a descriptive name for the switch connection and click on **Add Connection**.



The next window that appears prompts for the Switch Connection password. Enter the same password that was administered in Avaya Communication Manager in **Section 4.1**. Click on **Apply**.

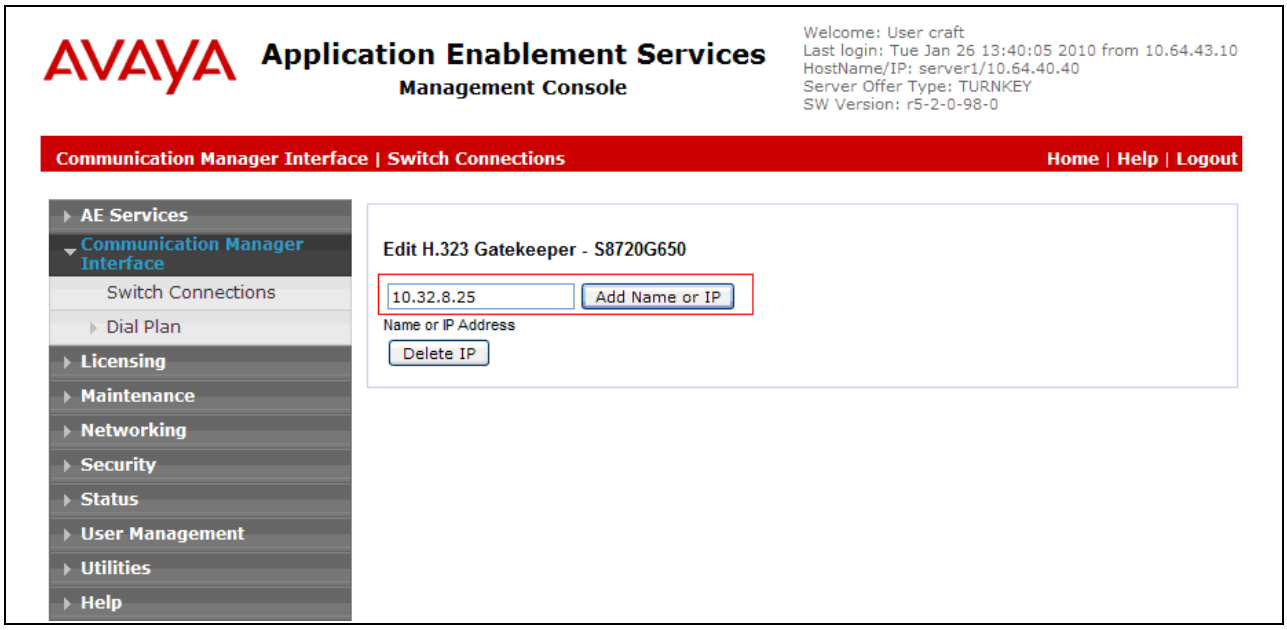


After returning to the Switch Connections page, select the radio button corresponding to the switch connection added previously, and click on the **Edit H.323 Gatekeeper** button for DMCC call control and monitor.



On the **Edit H.323 Gatekeeper – S8720G650** page, enter the C-LAN IP address which will be used for the DMCC service. Click on **Add Name or IP**. Repeat this step as necessary to add other C-LAN boards enabled with Application Enablement Services.

Note: *Avaya recommends using a CLAN board for phone registration, and another CLAN board for H.323 Gatekeeper.*



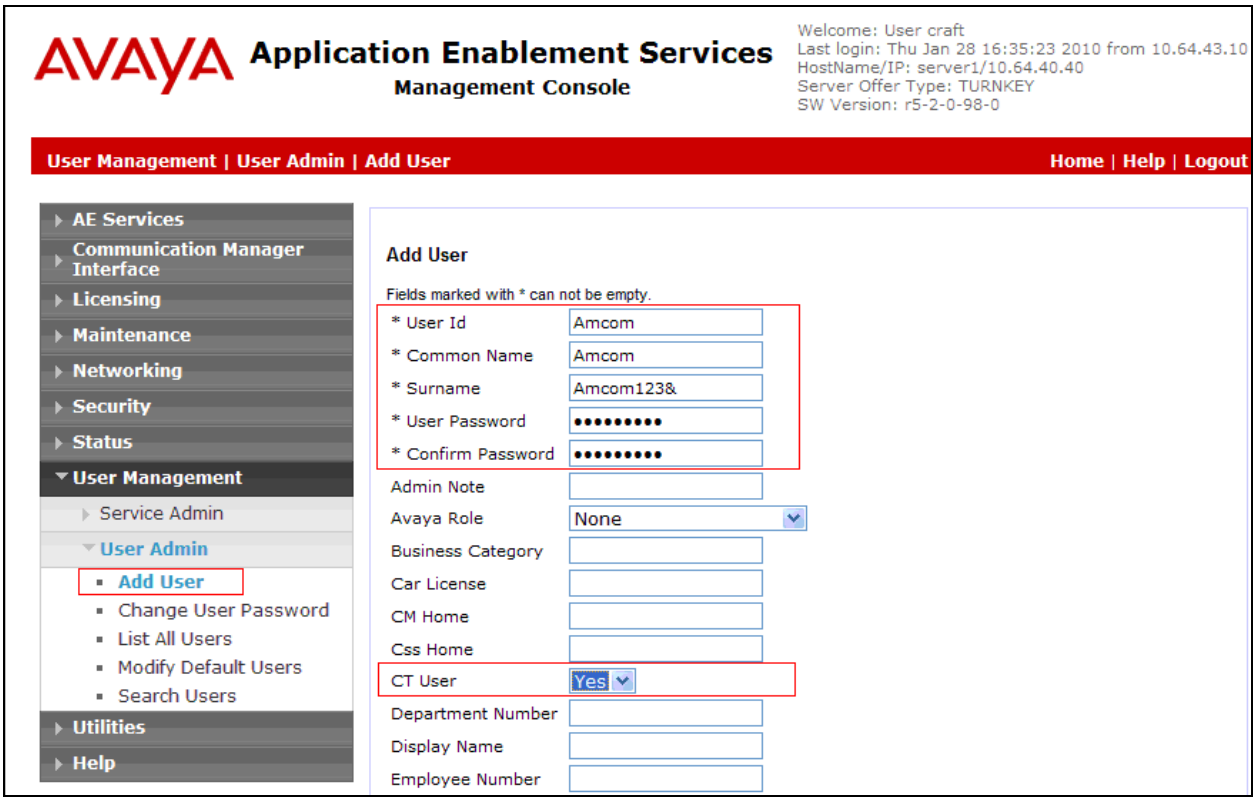
5.3. Configure the CTI Users

Navigate to **User Management → User Admin → Add User** link from the left pane of the window. On the Add User page, provide the following information:

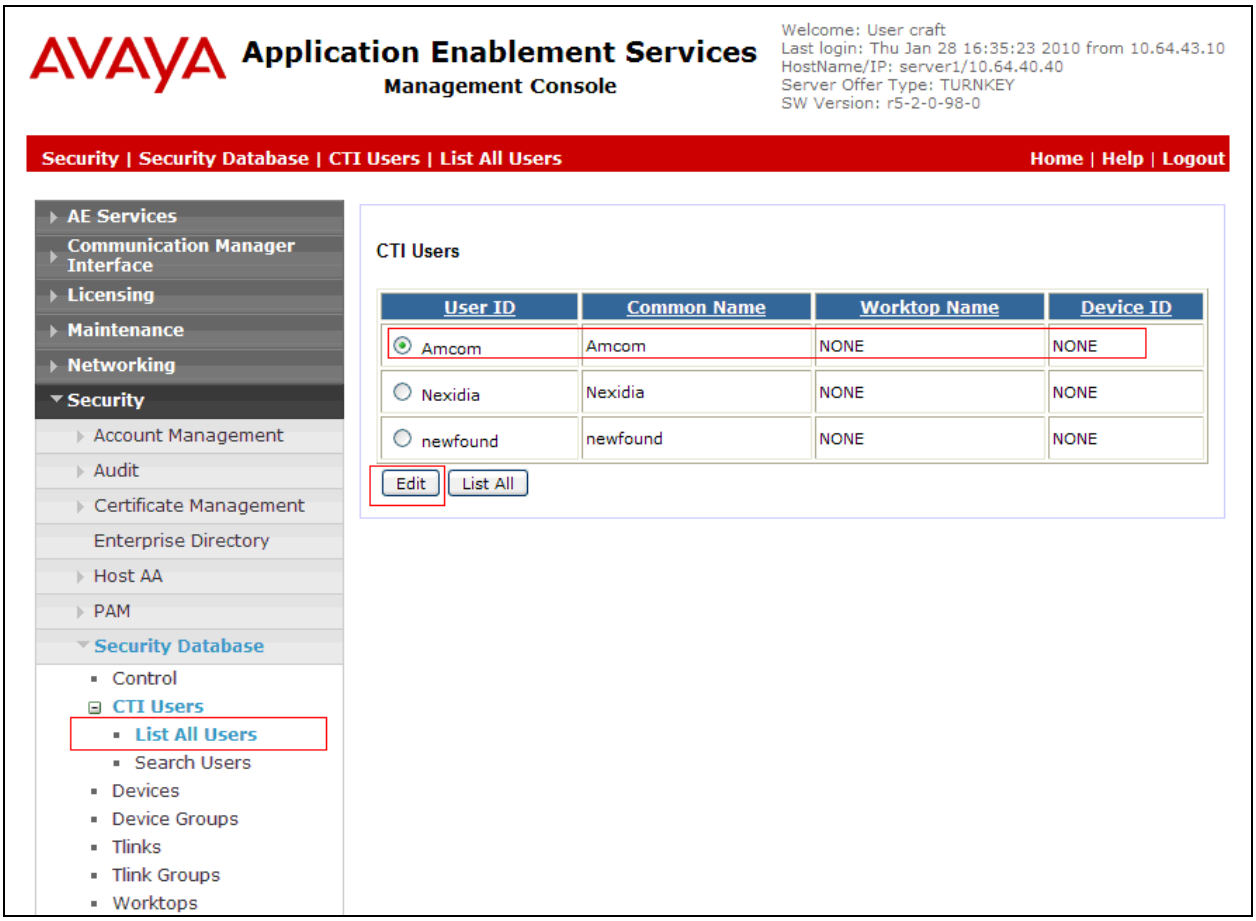
- User Id
- Common Name
- Surname
- User Password
- Confirm Password

The above information (User ID and User Password) must match with the information configured in the CTI Layer Configuration page in **Section 6**.

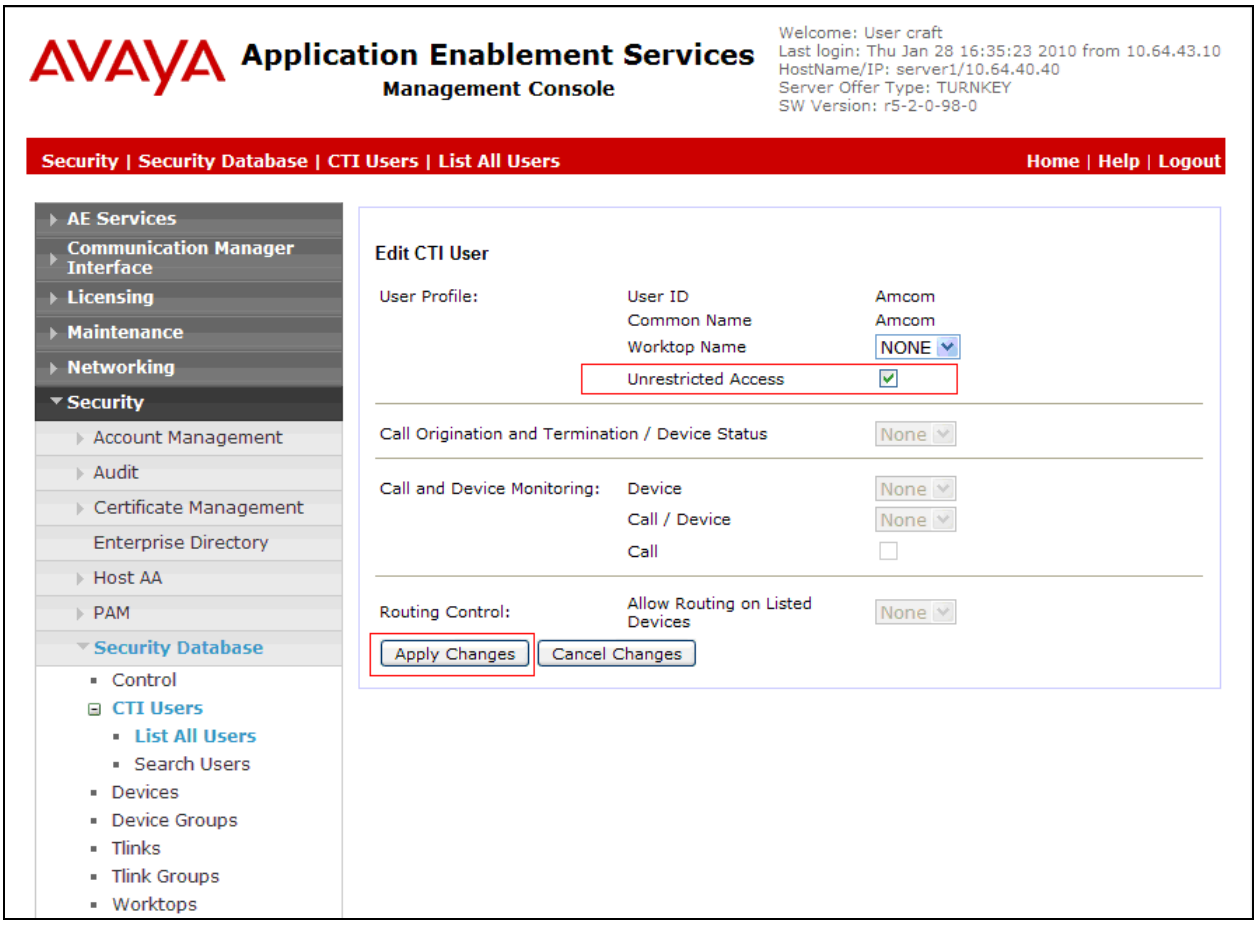
Select **Yes** using the drop down menu on the CT User field. This enables the user as a CTI user. Default values may be used in the remaining fields. Click the **Apply** button (not shown) at the bottom of the screen to complete the process.



Once the user is created, navigate to the **Security → Security Database → CTI Users → List All Users** link from the left pane of the window. Select the User ID created previously, and click the **Edit** button to set the permission of the user.

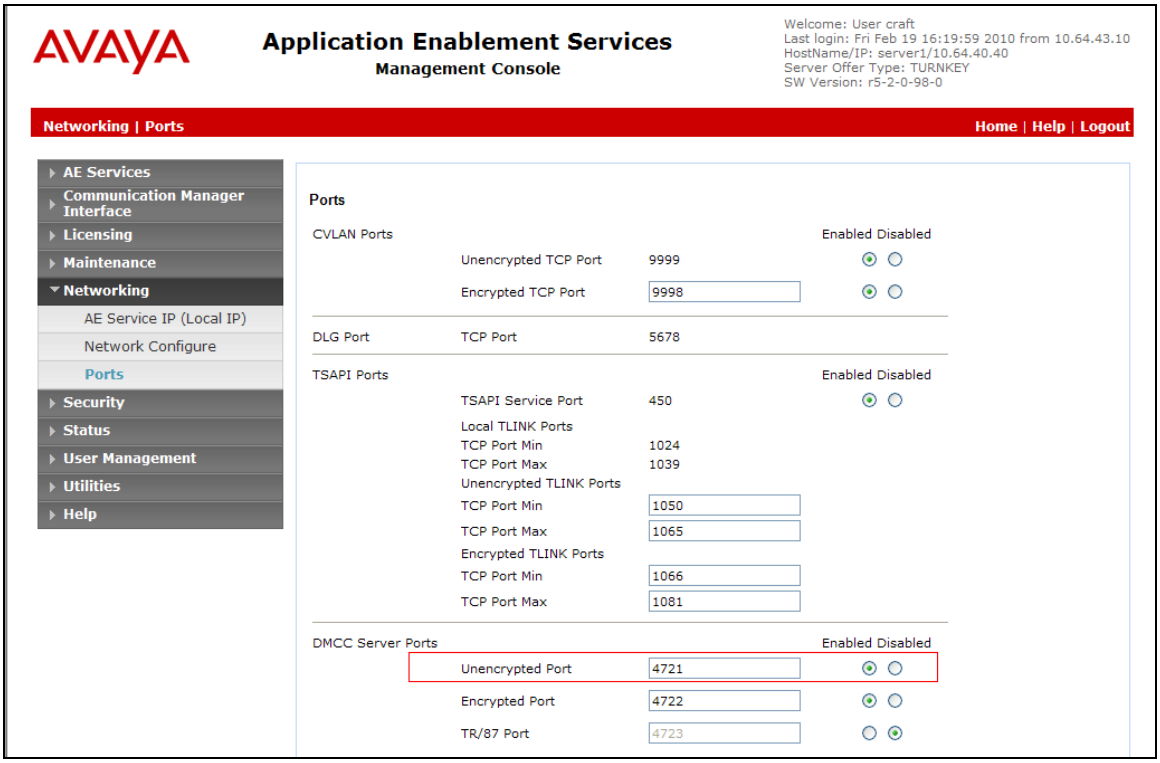


Provide the user with unrestricted access privileges by checking the **Unrestricted Access** button. Click on the **Apply Changes** button.



5.4. Configure the CTI Port

Navigate to the **Networking → Ports** link, from the left pane of the window, to set the DMCC server port. During the compliance test, the default port values were utilized. The following screen displays the default port values. Since the unencrypted port was utilized during the compliance test, set the Unencrypted Port field to **Enabled**. Default values may be used in the remaining fields. Click the **Apply Changes** button (not shown) at the bottom of the screen to complete the process.



6. Configure Amcom PC/PSAP

Amcom installs, configures, and customizes the PC/PSAP applications for their end customers.

7. General Test Approach and Test Results

The general approach was to exercise basic telephone and call operations on Avaya IP and Digital telephones using the aforementioned Amcom desktop application. The main objectives were to verify that:

- The user may successfully use PC/PSAP to perform off-hook, on-hook, dial, answer, hold, retrieve, transfer, conference, and release operations on the physical telephone.
- The agent user may successfully use PC/PSAP to log into and out of an ACD, and move between agent work modes.
- Manual operations performed on the physical telephone are correctly reflected in the PC/PSAP GUI.

- PC/PSAP and manual telephone operations may be used interchangeably; for example, go off-hook using PC/PSAP and manually dial digits.
- Display and call information on the physical telephone is accurately reflected in the PC/PSAP GUI.
- Call states are consistent between PC/PSAP and the physical telephone.

The objectives of **Section 7** were verified. For serviceability testing, the Amcom CTI Service was able to regain control of the physical telephone after restarts of the Amcom CTI Service, the computer on which it runs, and the Application Enablement Services server. In addition, after Amcom CTI Service lost network connectivity to the Application Enablement Services server, it was able to recover the existing session to the Application Enablement Services server when network connectivity was restored before the session expired, and establish a new session when network connectivity was restored after the previous session expired.

8. Verification Steps

The following steps may be used to verify the configuration:

- From the Amcom client computers, ping IP interfaces, in particular the Application Enablement Services server, and verify connectivity.
- For the physical IP telephones, verify that the physical telephones are registered by using the **list registered-ip-stations** command on the SAT. For the physical Digital telephones, verify that the telephones are attached to the correct ports.
- Go off-hook and on-hook on the controlled telephones manually and using PC/PSAP, and verify consistency.
- Place and answer calls from the controlled telephones manually and using PC/PSAP, and verify consistency.

9. Conclusion

These Application Notes described a compliance-tested configuration comprised of Avaya Aura™ Communication Manager Release 5.2.1, Avaya Aura™ Application Enablement Services Release 5.2, Avaya IP and Digital Telephones, and the Amcom PC/PSAP application. Amcom PC/PSAP allows a user to operate a physical telephone and view call and telephone display information through a graphical user interface (GUI). During compliance testing, calls were successfully placed to and from Avaya IP and Digital Telephones that were controlled and monitored by the Amcom PC/PSAP application.

10. Additional References

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

[1] *Administering Avaya Aura™ Communication Manager*, Issue 5.0, May 2009, Document Number 03-300509

[2] *Avaya Aura™ Application Enablement Services Administration and Maintenance Guide*, Issue 11, November 2009, Document Number 02-300357

Product information for Amcom products may be found at http://www.amcomsoft.com/products.cfm.

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.