Avaya Solution & Interoperability Test Lab

Application Notes for Spok MediCall, utilizing Spok CTI Layer, with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services - Issue 1.1

Abstract

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura® Communication Manager, Avaya Aura® Application Enablement Services, Avaya IP Telephones, and Spok MediCall desktop application.

Spok MediCall allows a user to operate a physical telephone and view call and telephone display information through a graphical user interface (GUI). Spok MediCall integrates with Spok CTI Layer, which is a middleware between Spok MediCall and Avaya Aura® Application Enablement Services, to control and monitor phone states.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 a compliance-tested configuration comprised of Avaya Aura® Communication Manager, Avaya Aura® Application Enablement Services (AES), Avaya IP (J169\J179) Telephones, and Spok MediCall applications.

Spok MediCall is a Windows-based attendant console application. Spok MediCall allows a user to operate a physical telephone and view call and telephone display information through a graphical user interface (GUI). Spok MediCall integrates with Spok CTI Layer, which is a middleware between Spok MediCall and AES, to control and monitor phone states.

The Spok CTI Layer service uses the AES Device and Media Call Control (DMCC) and Telephony Services Application Programming Interface (TSAPI) via DMCC to share control of and monitor a physical telephone and receive the same terminal and first party call information received by the physical telephone. Spok MediCall in turn uses the Spok CTI Layer service to control and monitor a physical telephone.

2. 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 Spok desktop application. Typical call scenarios including inbound, outbound, internal, external, and various conference and transfer were performed. The main objectives were to verify that:

- The user may successfully use Spok MediCall to perform off-hook, on-hook, dial, answer, hold, retrieve, transfer, conference, and release operations on the physical telephone.
- Spok MediCall and manual telephone operations may be used interchangeably; for example, go off-hook using Spok MediCall and manually dial digits.
- Display and call information on the physical telephone is accurately reflected in the Spok MediCall GUI.
- Call states are consistent between Spok MediCall and the physical telephone.

For serviceability testing, failures such as network disconnects, and resets were applied.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Spok made use of encrypted DMCC connections.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of the compliance test was primarily on verifying the interoperability between Spok MediCall, AES, and Communication Manager.

2.2. Test Results

All test cases were executed and passed. Note that the MediCall application only supports attended transfers.

2.3. Support

Technical support for the Spok MediCall solution can be obtained by contacting Spok:

- URL – http://www.spok.com
- Phone – (888) 797-7487

3. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with an AES, Communication Manager, Media Server and Avaya G430 Media Gateway. Spok MediCall is configured to be in the same network as the enterprise. Endpoints include Avaya J100 Series H.323 IP Telephones and Avaya Endpoints.

Note: Basic administration of Communication Manager and AES server is assumed. For details, see [1] and [2].

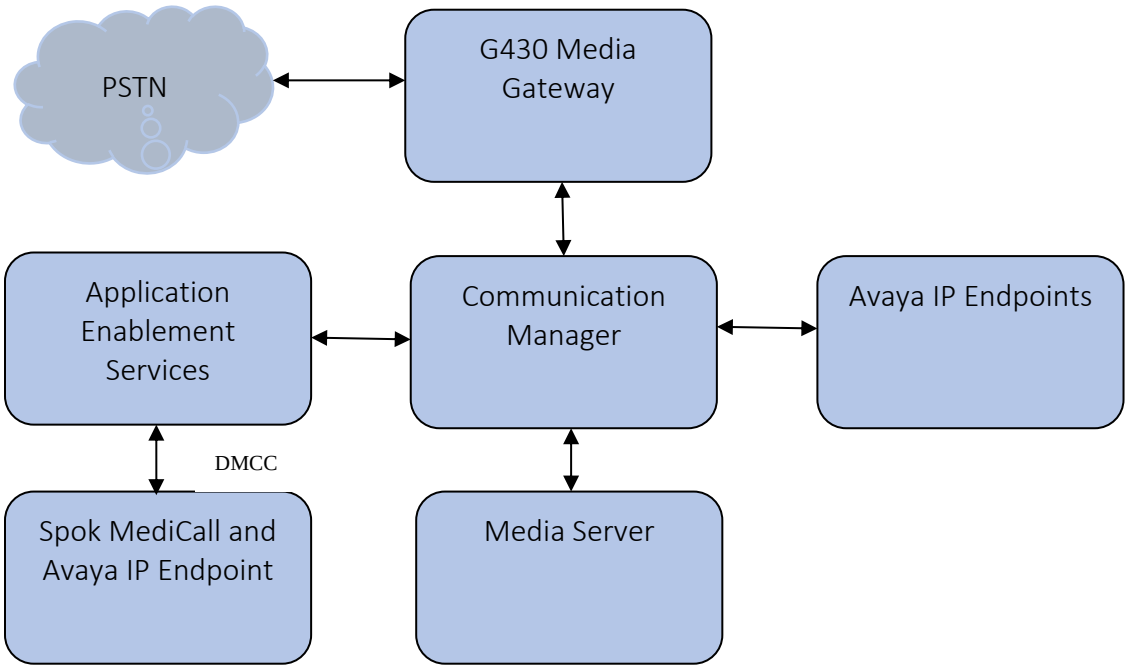


Figure 1: Spok MediCall Test Configuration

4. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided. All servers (except G430 Media Gateway) were on VM infrastructure, including the Spok components:

Equipment	Software/Firmware
Avaya Aura® Communication Manager	8.1.3.1.0-FP3SP1
Avaya Aura® Application Enablement Services	8.1.3.1.0.7-0
Avaya Aura® Media Server	8.0.2.163
Avaya G430 Media Gateway	41.34.1/1
Avaya Endpoints	
J169\J179 (H.323)	6.8502
Spok MediCall	11.x (11.11.28) 11.13
Spok CTI Layer	7.x (7.0.0.6) 7.4

5. Configure Avaya Aura® Communication Manager

This section describes the procedures for configuring Abbreviated Dialing, and controlled telephones. Standard connectivity was in place for AES and other Avaya components and are not covered in this document. A System Access Terminal session was used to perform these steps.

5.1. Configure System Parameters Features

Enter the **change system-parameters features** command. -Verify the setting of **Auto Hold** and **Transfer Upon Hang-Up** features. Please consult with Spok to confirm which combination of these features will work best in the environment, some combinations may cause rare conflicts.

~~Verify the **Auto Hold** and **Transfer Upon Hang-Up** features are enabled.~~

```
change system-parameters features                              Page   6 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
         Public Network Trunks on Conference Call: 5              Auto Start? y
    Conference Parties with Public Network Trunks: 6                Auto Hold? y
 Conference Parties without Public Network Trunks: 6           Attendant Tone? n
         Night Service Disconnect Timer (seconds): 180          Bridging Tone? n
                  Short Interdigit Timer (seconds): 3          Conference Tone? n
              Unanswered DID Call Timer (seconds):              Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30    Mode

change system-parameters features                              Page   7 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

CONFERENCE/TRANSFER
               Abort Transfer? n                 No Dial Tone Conferencing? n
        Transfer Upon Hang-Up? y        Select Line Appearance Conferencing? n
             Abort Conference? n                                      Unhold? n
   No Hold Conference Timeout: 60   Maximum Ports per Expanded Meet-me Conf: 7
                                                    12-party Conferences? n
                 External Ringing for Calls with Trunks? remote-only
```

5.2. Configure Abbreviated Dialing

Enter the **add abbreviated-dialing system** command. In the **DIAL CODE** list, enter the Feature Access Codes for ACD Login and Logout. These codes will be used by Spok MediCall extensions.

```
add abbreviated-dialing system                                  Page   1 of   1
                        ABBREVIATED DIALING LIST
                                SYSTEM LIST

Size (multiple of 5): 5      Privileged? n     Label Language:english
DIAL CODE                              LABELS (FOR STATIONS THAT DOWNLOAD LABELS)
     01: *54                                 01: Log-in
     02: *55                                 02: Log-out
     03:                                     03: **************
     04:                                     04: **************
     05:                                     05: **************
```

5.3. Configure Stations

During the compliance testing one extension was configured for Spok MediCall. Enter the **change station *n*** command, where ***n*** is the extension of a station.

Extension 30012 was used by Spok MediCall for controlling an Avaya Endpoint. On **Page 1** of the **station** form, enter a phone **Type**, descriptive **Name**, **Security Code**, **Button Modules** and set **IP SoftPhone** field to **y** to allow the physical station to be controlled by a softphone such as the Spok MediCall application. Note that J100 series phones use 9611 as station type for H.323 firmware configurations.

```
change station 30012                                            Page   1 of   7
                                    STATION

Extension: 30012                        Lock Messages? n                BCC: 0
     Type: 9611                       Security Code: *                 TN: 1
     Port: S000007                  Coverage Path 1:                  COR: 1
     Name: Spok Medicall            Coverage Path 2:                  COS: 1
Unicode Name? n                    Hunt-to Station:                 Tests? y
STATION OPTIONS
                                         Time of Day Lock Table:
             Loss Group: 19           Personalized Ringing Pattern: 1
                                        Message Lamp Ext: 30012
           Speakerphone: 2-way           Mute Button Enabled? y
        Display Language: english             Button Modules: 2
 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

                                       Customizable Labels? y
```

On **Page 2,** set **Auto Select Any Idle Appearance** to **y**.

```
change station 30012                                          Page   2 of   7
                                    STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? y
          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:                       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: 30012             Always Use? n IP Audio Hairpinning? n
```

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

```
change station 30012                                          Page   4 of   7
                                    STATION
 SITE DATA
      Room: __________                                  Headset? n
      Jack: _____                                       Speaker? n
     Cable: _____                                      Mounting: d
     Floor: __________                               Cord Length: 0
  Building: __________                                 Set Color:

ABBREVIATED DIALING
     List1: system          List2: ________           List3:



BUTTON ASSIGNMENTS
 1:call-appr                          5:call-appr
 2:call-appr                          6:q-calls     Grp: 1
 3:call-appr                          7:
 4:call-appr                          8:


change station 30012                                          Page   6 of   7
                                    STATION

                         BUTTON MODULE #1 ASSIGNMENTS

 1:brdg-appr  B:1 E:30001             13:brdg-appr  B:1 E:30002
 2:brdg-appr  B:2 E:30001             14:brdg-appr  B:2 E:30002
 3:brdg-appr  B:3 E:30001             15:
 4:brdg-appr  B:4 E:30001             16:
 5:brdg-appr  B:5 E:30001             17:
 6:brdg-appr  B:6 E:30001             18:
 7:                                   19:
 8:abrv-dial   List: 1 DC: 01    HL? n 20:
 9:auto-in             Grp:           21:
10:aux-work     RC:    Grp:           22:
11:after-call          Grp:           23:
12:                                   24:

change station 30012                                          Page   7 of   7
                                    STATION

                         BUTTON MODULE #2 ASSIGNMENTS

 1:brdg-appr  B:1 E:30003             13:brdg-appr  B:1 E:30006
 2:                                   14:brdg-appr  B:2 E:30006
 3:                                   15:brdg-appr  B:3 E:30006
 4:                                   16:brdg-appr  B:4 E:30006
 5:abrv-dial   List: 1 DC: 02    HL? n 17:brdg-appr  B:5 E:30006
 6:                                   18:brdg-appr  B:6 E:30006
 7:                                   19:

11:                                   23:togle-swap
12:                                   24:release
```

5.4. Configure Hunt Group

Enter the **add hunt-group *n*** command, where ***n*** is an unused hunt group number. On **Page 1** assign a descriptive **Group Name** and an available **Group Extension** as per the dial plan. Also, set **ACD**, **Queue** and **Vector** to **y**. The Hunt group configured here was used by MediCall agents to log onto ACD.

```
add hunt-group 21                                               Page   1 of   4
                                  HUNT GROUP

           Group Number: 21                                 ACD? y
             Group Name: Hunt Group 21                    Queue? y
        Group Extension: 31020                           Vector? y
             Group Type: ucd-mia
                     TN: 1
                    COR: 1                      MM Early Answer? n
          Security Code:                  Local Agent Preference? n
 ISDN/SIP Caller Display:

            Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

5.5. Configure VDNs

Use the **add vdn *n*** command to add a new VDN, where ***n*** is an available extension as per the dial plan.

On **Page 1,** provide a descriptive **Name** and available **Vector Number** in **Destination.**

```
add vdn 31501                                                Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                              Extension: 31501                    Unicode Name? n
                                  Name*: Spok VDN
                            Destination: Vector Number        21
                    Attendant Vectoring? n
                   Meet-me Conferencing? n
                     Allow VDN Override? n
                                    COR: 1
                                    TN*: 1
                               Measured: both     Report Adjunct Calls as ACD*? n
        Acceptable Service Level (sec): 20

        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:

SIP URI:
```

5.6. Configure Vector

To configure a vector, use the **change vector *n*** command, where ***n*** is the vector used during the adding the VDN. A simple vector is configured to queue calls to hunt group 21.

```
change vector 21                                              Page   1 of   6
                                CALL VECTOR

    Number: 21               Name: Spok Vector
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 queue-to     skill 21   pri m
03 wait-time    30  secs hearing music
04 goto step    2             if unconditionally
05
```

5.7. Configure Agent Extensions

Enter the **add agent-loginID *n*** command, where ***n*** is an available extension according to the dial plan. This extension will be used by Spok MediCall to log onto ACD. During the compliance test, two agent extensions were added, 12021 and 12022. On **Page 1**, specify a **Name** of the agent, **Password**, and set **Auto Answer** to **none**.

```
add agent-loginID 32021                                      Page   1 of   2
                                AGENT LOGINID

               Login ID: 32021                Unicode Name? n   AAS? n
                   Name: Spok Agent 1                          AUDIX? n
                     TN: 1        Check skill TNs to match agent TN? n
                    COR: 1
          Coverage Path:                              LWC Reception: spe
          Security Code:                      LWC Log External Calls? n
          Attribute:                         AUDIX Name for Messaging:

                                        LoginID for ISDN/SIP Display? n
                                                            Password:
                                                   Password (enter again):
                                                         Auto Answer: none
 AUX Agent Remains in LOA Queue: system              MIA Across Skills: system
AUX Agent Considered Idle (MIA): system    ACW Agent Considered Idle: system
             Work Mode on Login: system    Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :
    WARNING:  Agent must log in again before changes take effect
```

On **Page 2,** configure the Skill Number that was configured earlier in this document and specify a skill level.

```
add agent-loginID 32021                                      Page   2 of   2
                                AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
Call Handling Preference: skill-level                 Local Call Preference? n

    SN   RL SL          SN   RL SL
 1: 21      1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
 4:                 19:                 34:                 49:
```

6. Configure Avaya Aura® Application Enablement Services

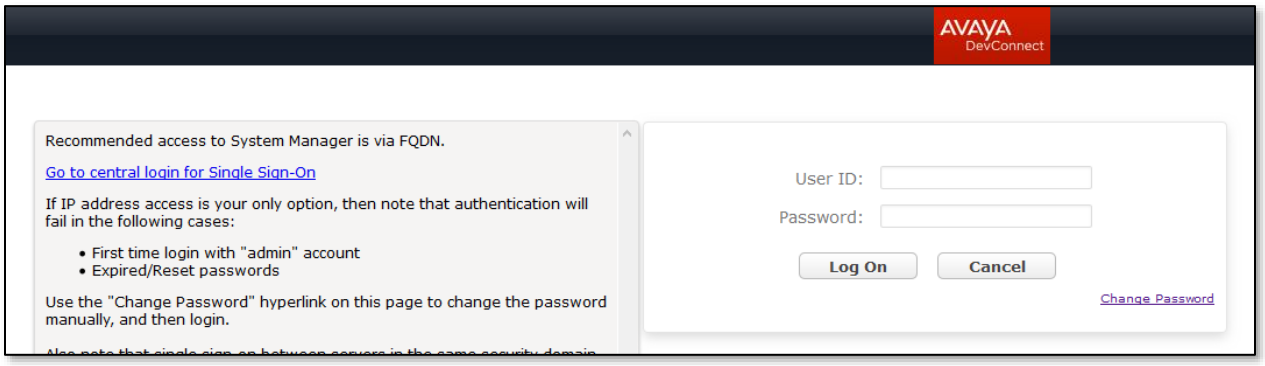
The Application Enablement Services server enables Computer Telephony Interface (CTI) applications to control and monitor telephony resources on Communication Manager.

This section assumes that installation and basic administration of the AES server has been performed. The steps in this section describe the configuration of a CTI user, a DMCC port and TLS Version, Root Certificate, and Tlink information.

6.1. Device and Media Call Control API Station Licenses

The Spok MediCall Service instances appear as "virtual" stations/softphones to Communication Manager. Each of these virtual stations, hereafter called Device and Media Call 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 Call 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 AES Management Console page.
Select the **Licensing ➔ WebLM Server Access** link from the left pane of the window (not shown). During the compliance testing, Avaya Aura® System Manager was used as a license server.

Provide appropriate login credentials and log in.



Navigate to **Services → Licenses** (not shown). On the WebLM Home page, select **License Products → Application_Enablement** (not shown) link from the left pane of the window.

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

Note on DMCC Licenses: The Spok MediCall application uses an existing station for Console Operators. Thus, the Communication Manager license requires enough station license capacity to accommodate these. The DMCC licenses can be purchased as either Basic (just the AES DMCC requirement), or Full (which bundles a Communication Manager station RTU with the AES DMCC).

Note: TSAPI licenses (one per agent station) are also required.

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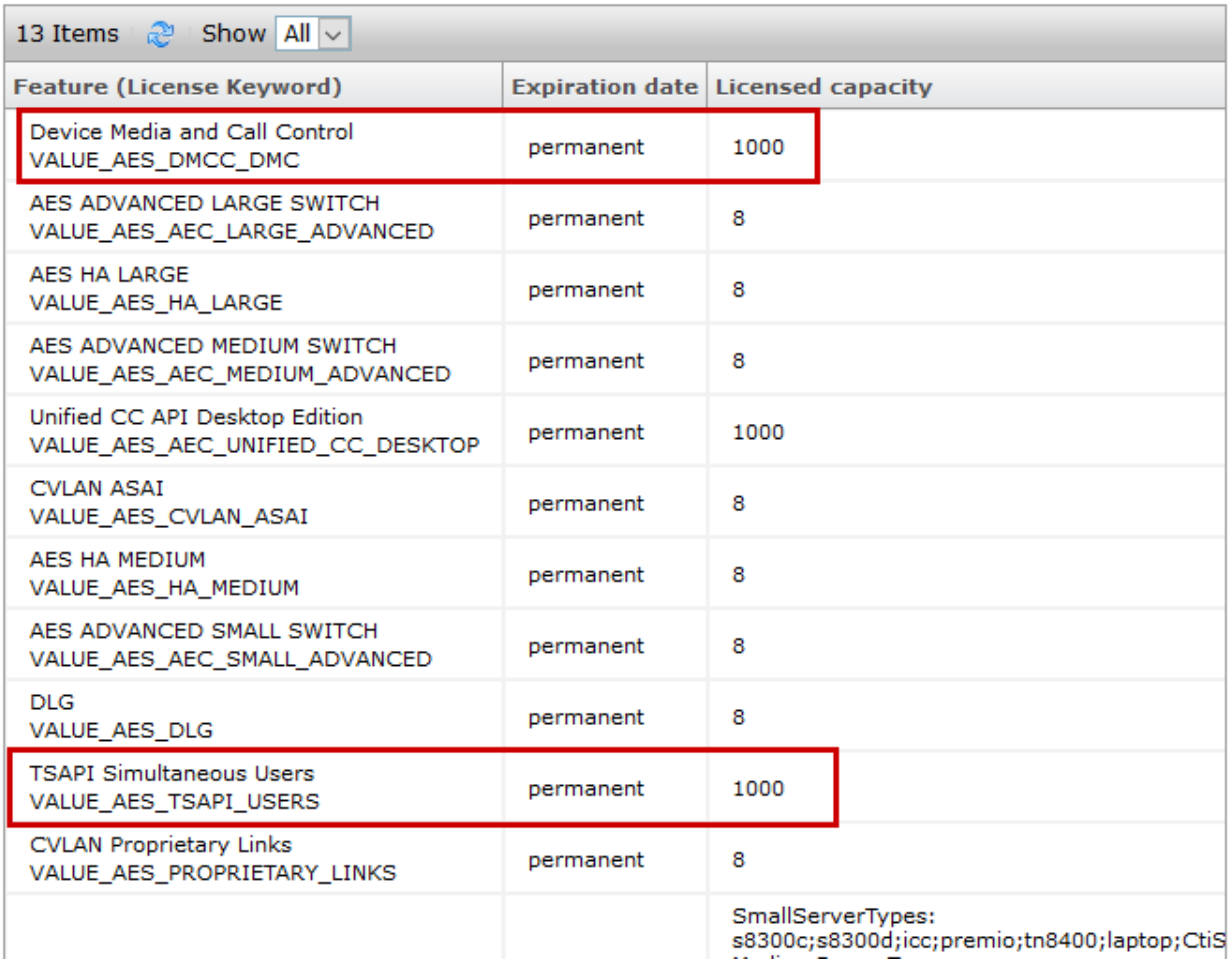


13 Items Show All

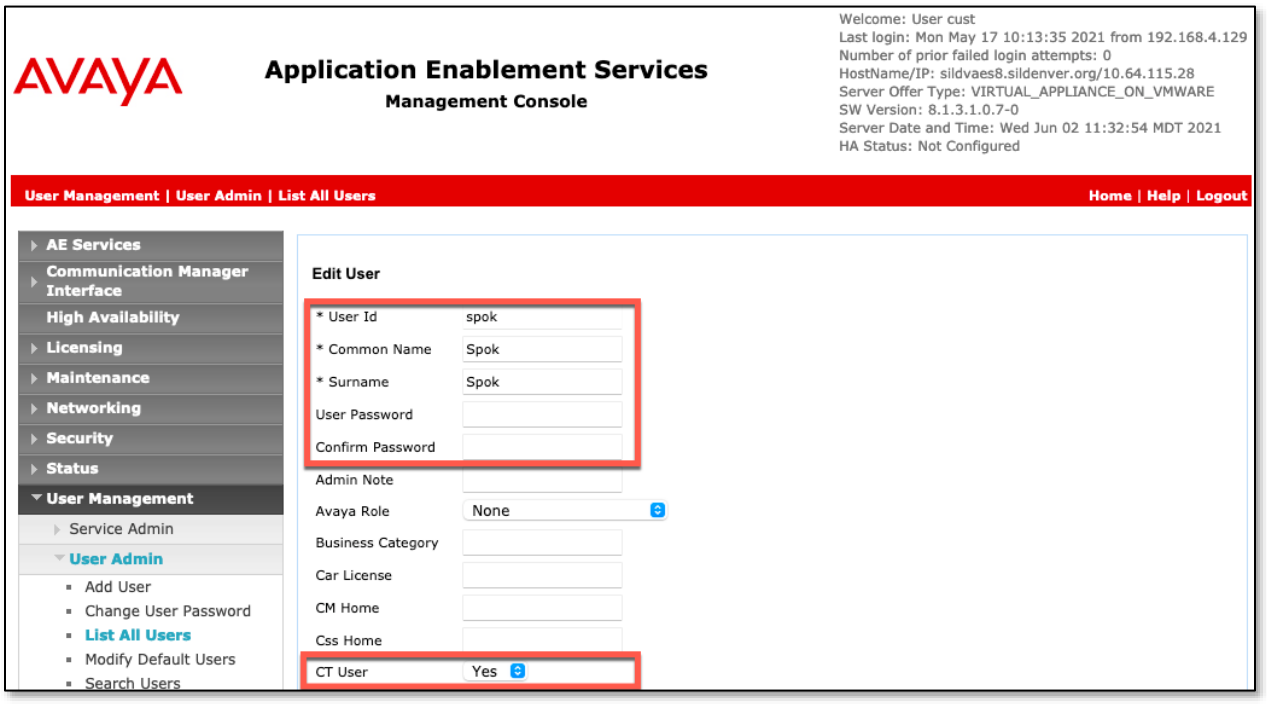
Feature (License Keyword)	Expiration date	Licensed capacity
Device Media and Call Control VALUE_AES_DMCC_DMC	permanent	1000
AES ADVANCED LARGE SWITCH VALUE_AES_AEC_LARGE_ADVANCED	permanent	8
AES HA LARGE VALUE_AES_HA_LARGE	permanent	8
AES ADVANCED MEDIUM SWITCH VALUE_AES_AEC_MEDIUM_ADVANCED	permanent	8
Unified CC API Desktop Edition VALUE_AES_AEC_UNIFIED_CC_DESKTOP	permanent	1000
CVLAN ASAI VALUE_AES_CVLAN_ASAI	permanent	8
AES HA MEDIUM VALUE_AES_HA_MEDIUM	permanent	8
AES ADVANCED SMALL SWITCH VALUE_AES_AEC_SMALL_ADVANCED	permanent	8
DLG VALUE_AES_DLG	permanent	8
TSAPI Simultaneous Users VALUE_AES_TSAPI_USERS	permanent	1000
CVLAN Proprietary Links VALUE_AES_PROPRIETARY_LINKS	permanent	8
		SmallServerTypes: s8300c;s8300d;icc;premio;tn8400;laptop;CtiS

6.2. Configure the CTI User

Navigate to **User Management → User Admin → Add User** link from the left pane of the window. On the Add User page (not shown), provide the following information:

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

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. The Edit User page below shows the configuration previously configured for this user.

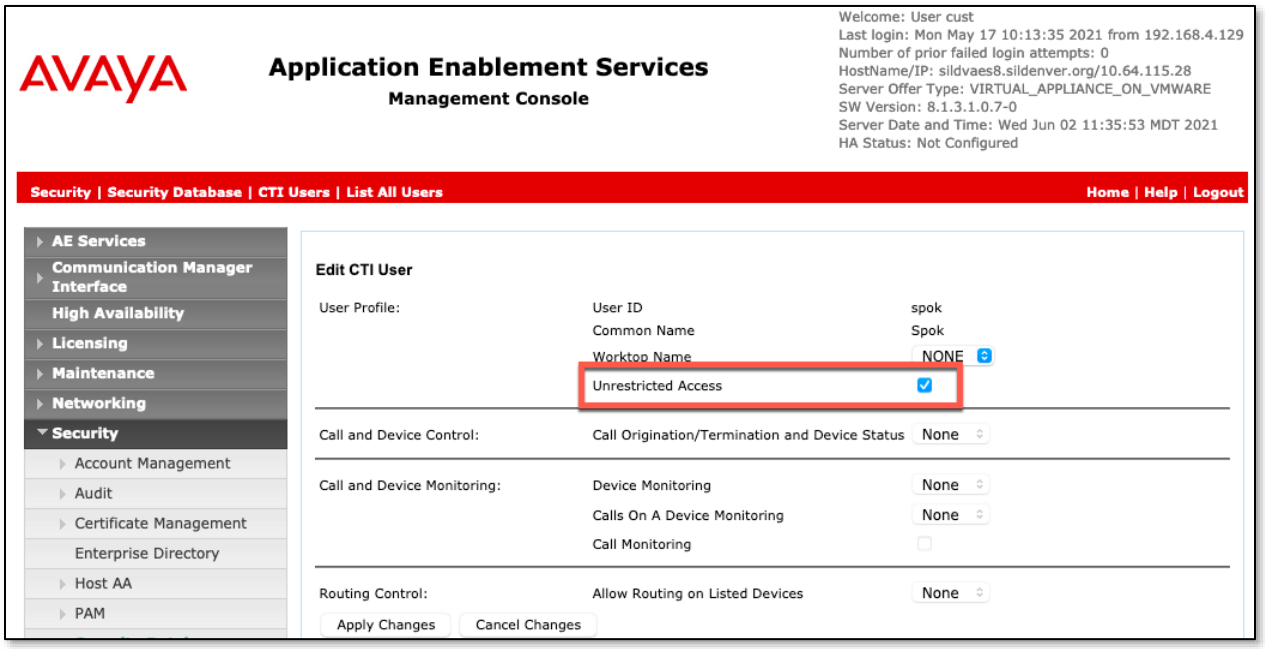


The above information (User ID and User Password) must match with the information configured in the Spok MediCall Configuration page in **Section 7**.

The Following step is only necessary if the Security Database is enabled for DMCC and TSAPI (**Security → Security Database → Control** – not shown).

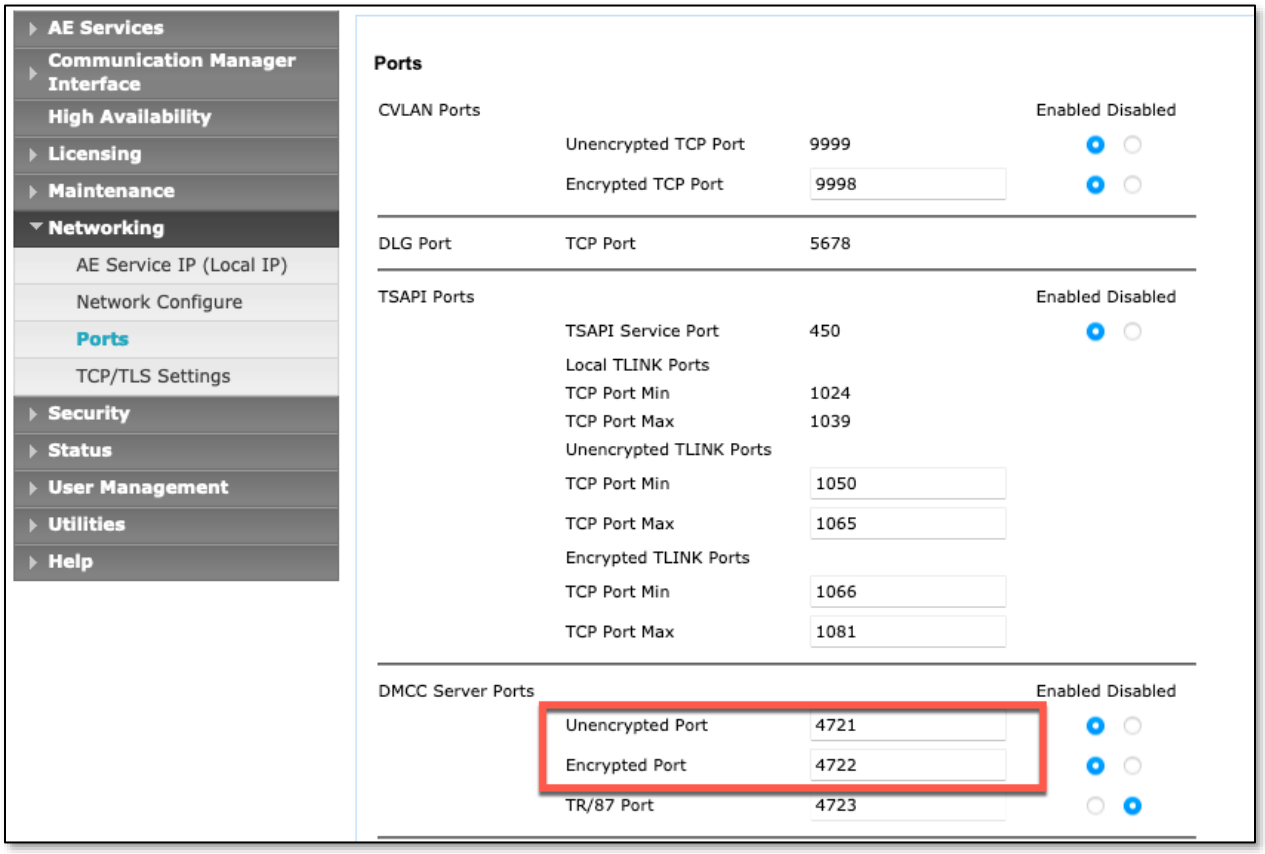
Once the user is created, navigate to the **Security → Security Database → CTI Users → List All Users** link from the left pane of the window. Select the User ID created previously and click the **Edit** button to set the permission of the user (not shown).

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



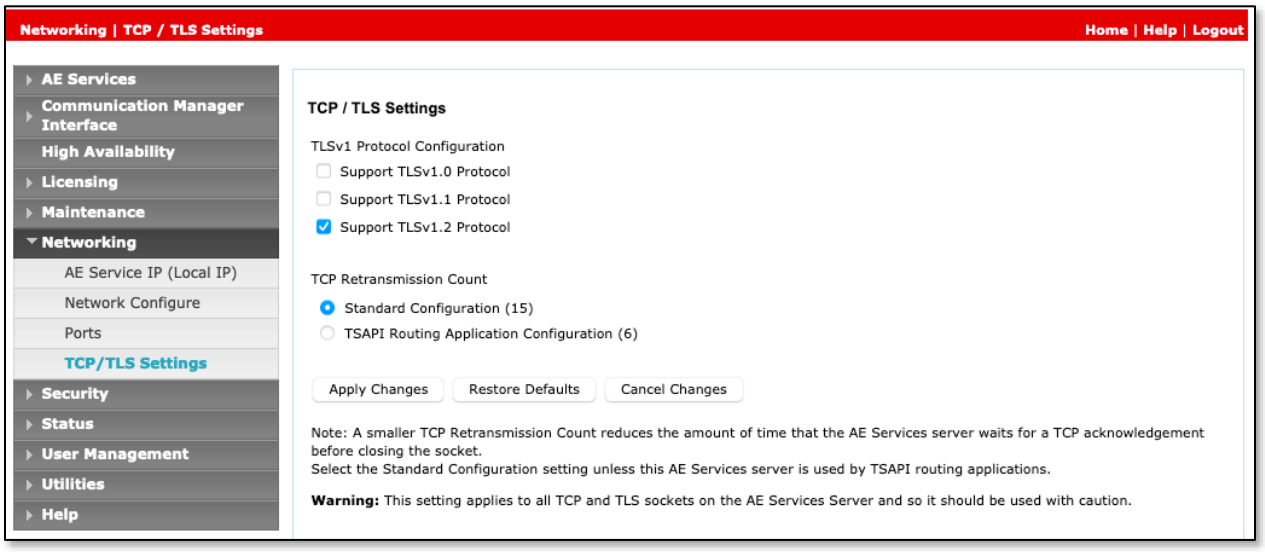
6.3. Configure the DMCC Port

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. Both **Unencrypted** and **Encrypted Port** were used during the compliance test. Click the **Apply Changes** button (not shown) at the bottom of the screen to complete the process.



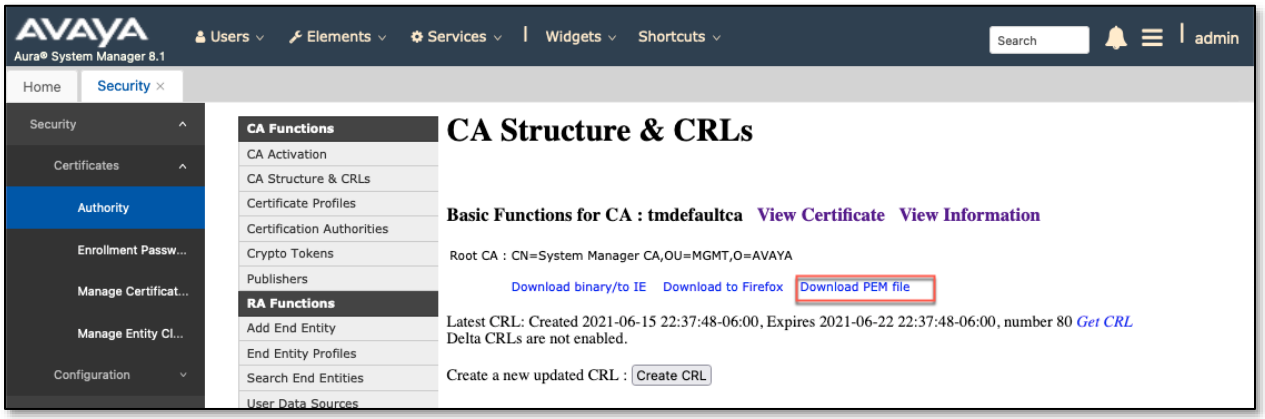
6.4. Configure TLS Version

Navigate to the **Networking → TCP/TLS Settings** page and verify that TLS Version 1.2 is checked. This will be used in **Section 7** when configuring Spok MediCall.



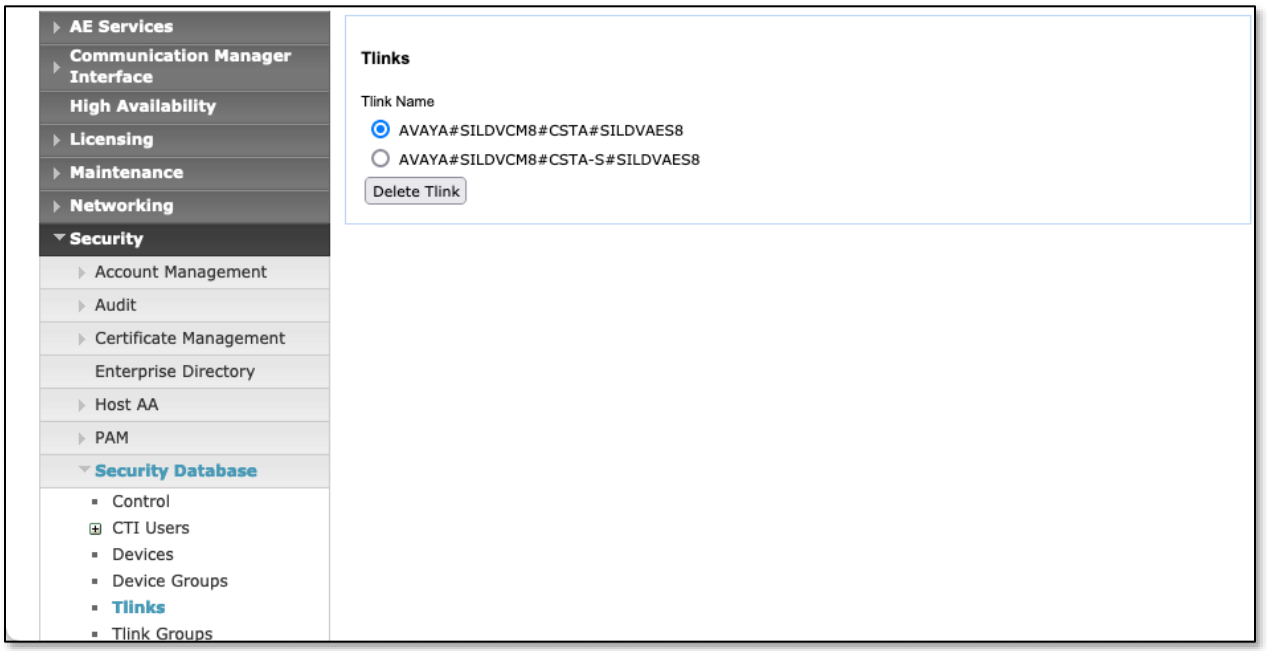
6.5. Obtain Root Certificate

In order to configure the application to use secure links, download the root certificate for the environment, in this case Avaya Aura® System Manager issued certificates to AES. The following illustrates how to download this from Avaya Aura® System Manager.



6.6. Obtain the Tlink

Navigate to the **Security → Security Database → Tlinks** and note the Tlink name for use when configuring the Spok solution in the next section.



7. Configure Spok MediCall

Spok installs, configures, and customizes the Spok MediCall application for their end customers. Spok MediCall integrates with Spok CTI Layer, which is a middleware installed on the same PC that the Spok Medicall is installed on, to control and monitor the phone states.

Note: Avaya phones as the network supplier for the agent workstations is not supported by Spok. Agent workstations should have its own network connection, separate from Avaya phones.

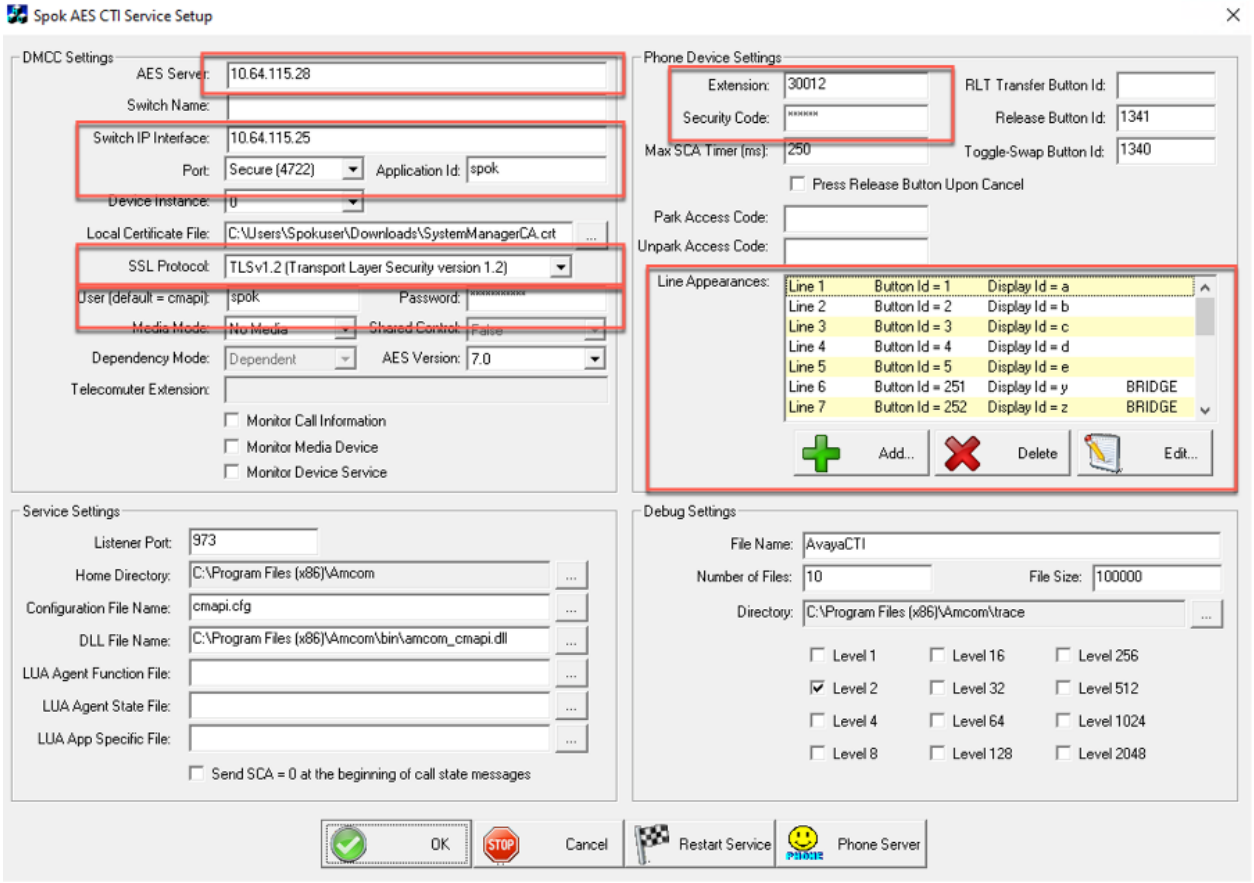
The following shows the **Spok AES CTI Services Setup** page. This is an application installed when the Spok software is installed on the PC and is accessed for the Programs list on the PC. Provide the following information:

Under DMCC Settings

- **AES Server** – Enter the IP address of AES.
- **Switch IP Address** – Enter the procr IP address of Communication Manager.
- **Port** – Enter the port utilized during the compliance test.
- **SSL Protocol** – Select Version 1.2 to match the AES settings in **Section 6**.
- **User** – Enter the user name created for Spok MediCall from **Section 6**.
- **Password** – Enter the password created for Spok MediCall from **Section 6**.

Under Phone Device Settings

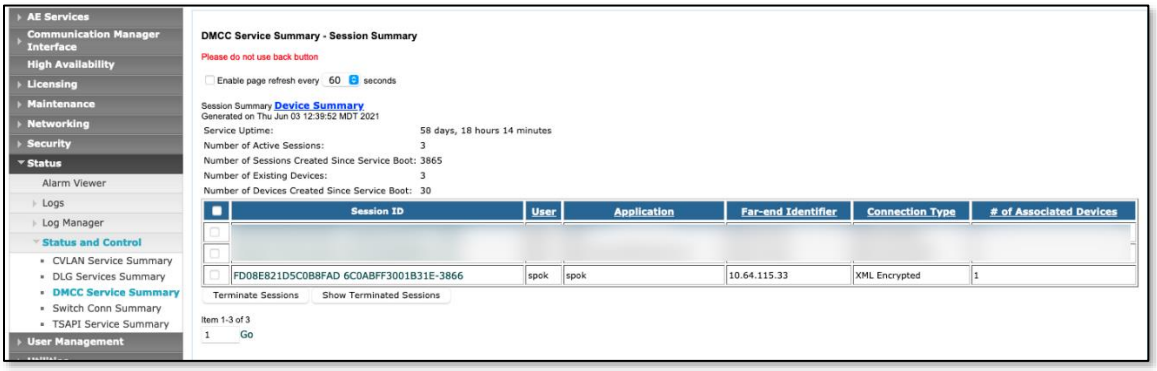
- **Extension**: Enter the extension that will be controlled by Spok MediCall from **Section 5**.
- **Security Code**: Enter the security code for the controlled station from **Section 5**.
- **Release Button** – Enter the Release button assigned for the controlled station from **Section 5.3**.
- **Line Appearances** – Configure line appearances as per **Section 5**.

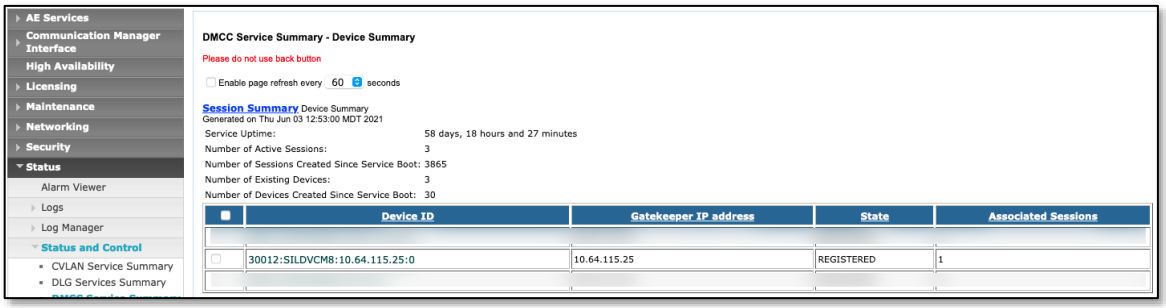


8. Verification Steps

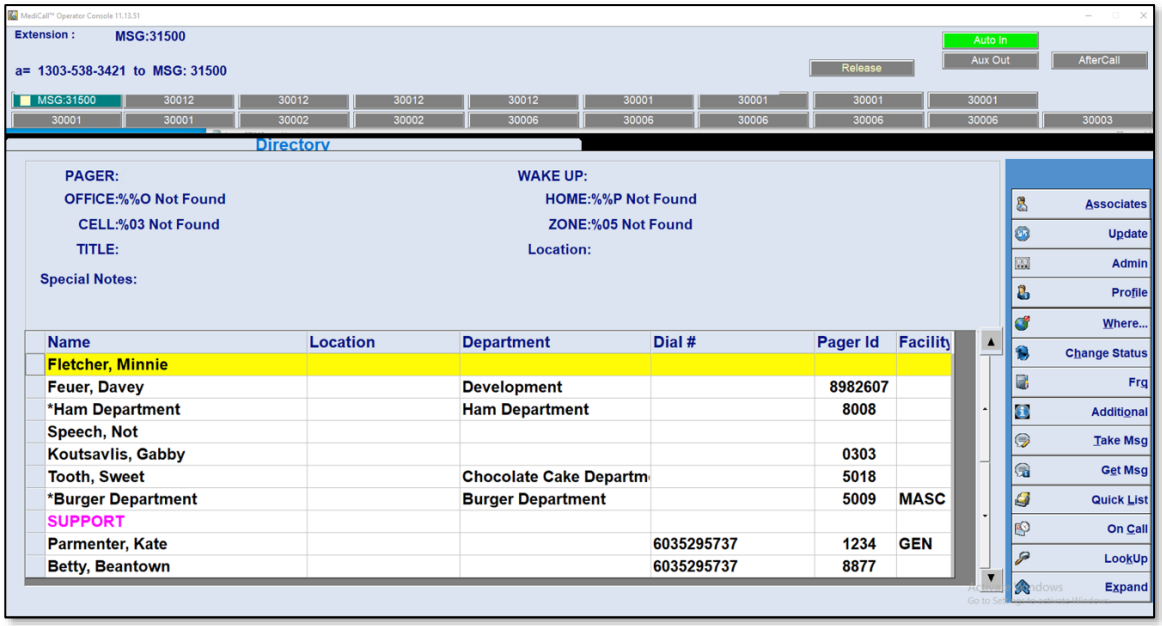
The following steps may be used to verify the configuration:

- Verify Spok MediCall is successfully connected to AES via AES Management console. Navigate to **Status → Status and Control → DMCC Service Summary**. Verify the **State** of Spok MediCall user is **REGISTERED**.





- Place and answer calls from the controlled telephones manually and use Spok MediCall and verify consistency.



9. Conclusion

These Application Notes described a compliance-tested configuration comprised of Communication Manager, AES, Avaya J169\179 IP Telephones, and the Spok MediCall application. Spok MediCall 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 Telephones that were controlled and monitored by the Spok MediCall application.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.
[1] *Administering Avaya Aura® Communication Manager, Release 8.1.x*
[2] *Administering Avaya Aura® Application Enablement Services, Release 8.1.x*

Product information for Spok products may be found at http://www.spok.com.

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.