Avaya Solution & Interoperability Test Lab

Application Notes for Configuring ESNA Office-LinX with Avaya Aura® Application Enablement Services and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring ESNA Office-LinX to interoperate with Avaya Aura® Application Enablement Services and Avaya Aura® Communication Manager.

The Office-LinX is a voice processing system that functions with an organization's existing telephone system to enhance its overall telecommunications environment.

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 procedure for configuring ESNA Office-LinX to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

ESNA Office-LinX is a voice processing system that functions with an organization's existing telephone system to enhance its overall telecommunications environment.

ESNA Office-LinX acts as a unified messaging solution offering call and voice messaging control over the phone, web, or via client applications from the user's desktop PC or mobile smart device . System Administrative functions may be performed either by using a touchtone telephone or the Windows interface from the Voice Mail server.

However, during this compliance test, the focus was on the TSAPI CTI link (3rd party Call Control) integration between ESNA Office-LinX and Avaya Aura® Application Enablement Services server.

2. General Test Approach and Test Results

The general test approach was to place calls to ESNA Office-LinX, using a coverage path and hunt group. The main objectives were to verify the following:

- Successfully establish calls to ESNA Office-LinX from SIP and H.323 telephones attached to Session Manager or Communication Manager.
- Successfully transfer from ESNA Office-LinX to SIP and H.323 telephones attached to Session Manager or Communication Manager.
- Using the Office-LinX UC Client Manager, verify the call state is consistent with the physical phone.
- Using the Office-LinX UC Client Manager, calls can be answered / transferred/ Held / Resumed and Hung up.

For serviceability testing, failures such as cable pulls and resets were applied. All test cases passed.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included features and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability between ESNA Office-LinX and Application Enablement Services.

2.2. Test Results

The test objectives were verified. For serviceability testing, ESNA Office-LinX operated properly after recovering from failures such as cable disconnects and resets of ESNA Office-LinX, Communication Manager, and Application Enablement Services.

2.3. Support

Technical support for the ESNA Office-LinX solution can be obtained by contacting ESNA:

- URL – techsupport@esna.com
- Phone – (905) 707-1234

3. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with a Session Manager and an Avaya S8300D Server with an Avaya G450 Media Gateway. Endpoints include Avaya 9600 Series SIP telephones, Avaya 9600 Series H.323 IP telephones, and an Avaya 6408D Digital telephone. Avaya S8720 Servers with Avaya G650 Media Gateway were included in the test to provide an inter-switch scenario.

ESNA Office-LinX does not register with the Session Manager as an endpoint but instead is configured as a trusted SIP entity.

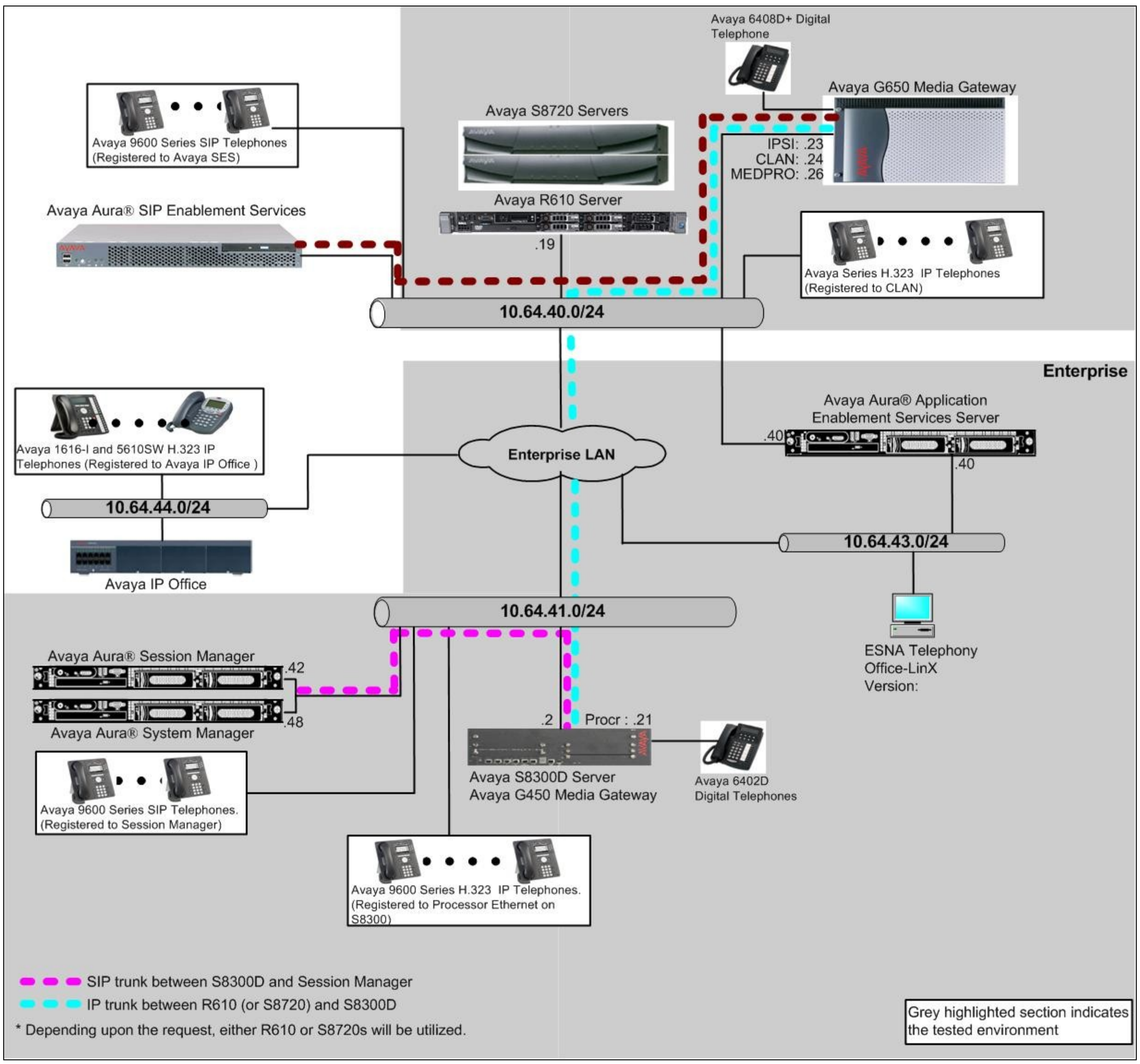


Figure 1: Test Configuration of ESNA Office-LinX

4. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya S8300D Server with Avaya G450 Media Gateway		Avaya Aura® Communication Manager 6.0.1(R016x.00.1.510.1) w/ patch 00.1.510.1-19303
Avaya Aura® System Manager		6.1.5.0
Avaya Aura® Session Manager		6.1.5.0
Avaya S8720 Servers with Avaya G650 Media Gateway (used for inter-switch test scenarios)		Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4) w/ patch 02.1.016.4 - 18365
Avaya Aura® Application Enablement Services		6.1 (R6-1-0-20-0)
Avaya 4600 and 9600 Series SIP Telephones		
	9620 (SIP) 9630 (SIP) 9650 (SIP)	2.5 2.5 2.5
Avaya 4600 and 9600 Series IP Telephones		
	4625 (H.323) 9620 (H.323) 9630 (H.323) 9650 (H.323)	2.9 3.1 3.1 3.1
Avaya 6408D+ Digital Telephone		-
ESNA Office-LinX		8.5 SP1

5. Configure Avaya Aura® Communication Manager

In the compliance test, Communication Manager was set up as an Evolution Server. This section describes the procedure for setting up a SIP trunk between Communication Manager and Session Manager. The steps include setting up an IP codec set, an IP network region, IP node names, a signaling group, a trunk group, a SIP station, and the TSAPI CTI link. Before a trunk can be configured, it is necessary to verify that there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. All SIP telephones, except ESNA Office-LinX, are configured as off-PBX telephones in Communication Manager.

5.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses.

If not, contact an authorized Avaya account representative to obtain additional licenses

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                                 Software Package: Standard
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 6400  185
                                      Maximum Stations: 500   19
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 10    0
                    Maximum Off-PBX Telephones -   OPS: 500   9
                    Maximum Off-PBX Telephones - PBFMC: 10    0
                    Maximum Off-PBX Telephones - PVFMC: 10    0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 0     0
```

On **Page 2** of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed.

If not, contact an authorized Avaya account representative to obtain additional licenses.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  20
           Maximum Concurrently Registered IP Stations: 2400  3
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 10    0
                       Maximum Administered SIP Trunks: 4000  110
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 8     0
```

5.2. IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and Session Manager. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 5.3** for configuring the IP network region to specify which codec sets may be used within and between network regions.

***Note**: ESNA Office-LinX supports G.711MU and G.711A. During the compliance test, G711MU was utilized.*

```
change ip-codec-set 1                                          Page   1 of   2

                          IP Codec Set
    Codec Set: 1

    Audio         Silence       Frames   Packet
    Codec         Suppression   Per Pkt  Size(ms)
 1: G.711MU            n           2         20
```

5.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. During the compliance test, the authoritative domain is set to **avaya.com**. This should match the SIP Domain value used in Session Manager.
- **Codec Set** – Set the codec set number as provisioned in **Section 5.2**.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Session Manager in Communication Manager. Enter the **change node-names ip** command, and add a node name for Session Manager along with its IP address.

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

    Name                  IP Address
CLAN                    10.64.40.24
SM-1                    10.64.41.42
default                 0.0.0.0
procr                   10.64.41.21
procr6                  ::
rdtt                    10.64.43.10
s8300-lsp               10.64.42.21
```

5.5. Configure SIP Signaling

Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to **sip**.
- **IMS Enabled** – Verify that the field is set to **n**. Setting this filed to **y** will cause Communication Manager to behave as a Feature Server.
- **Transport Method** – Set to **tls** (Transport Layer Security).
- **Near-end Node Name** – Set to **procr** as displayed in the IP Node Names **Section 5.4**.
- **Far-end Node Name** – Set to the Session Manager name configured in **Section 5.4**.
- **Far-end Network Region** – Set to the region configured in **Section 5.3**.
- **Far-end Domain** – Set to **avaya.com**. This should match the Authoritative Domain value in **Section 5.3**.
- **Direct IP-IP Audio Connections** – Set to **y**, as shuffling was enabled during the compliance test.

```
add signaling-group 92
                                SIGNALING GROUP

 Group Number: 92             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM


   Near-end Node Name: procr                 Far-end Node Name: SM-1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                       IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Configure Trunk Group

To configure the trunk group, enter the **add tunk-group <t>** command, where **t** is an available trunk group and configure the following:

- **Group Type** – Set the Group Type field to **sip**.
- **Group Name** – Enter a descriptive name.
- **TAC** (Trunk Access Code) – Set to any available trunk access code.
- **Service Type** – Set the Service Type field to **tie**.
- **Signaling Group** – Set to the Group Number field value for the signalling group configured in **Section 5.5**
- **Number of Members** – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 92                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 92                   Group Type: sip         CDR Reports: y
  Group Name:SM_41_42                     COR: 1        TN: 1        TAC: 1092
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 92
                                                     Number of Members: 20
```

On **Page 3**, during the compliance test, the Numbering Format field was set to **private.**

```
add trunk-group 92                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? Y
 DSN Term? N
```

5.7. Configure Hunt Group

This section describes the steps for administering a hunt group in Communication Manager. Enter the **add hunt-group <h>** command, where **h** is an available hunt group number. The following fields were configured for the compliance test.

- **Group Name** – Enter a descriptive name
- **Group Extension** – Enter an available extension that is valid in the provisioned dial plan.

```
Add hunt-group 93                                              Page   1 of  60
                                  HUNT GROUP

            Group Number: 93                                ACD? n
              Group Name: ESNA                            Queue? n
         Group Extension: 72036                          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 2**, provide the following information:

- **Message Center** – Enter **sip-adjunct**, indicating the type of messaging adjunct used for this hunt group. This value will also be used in the Station form.
- **Voice Mail Number** – Enter the Voice Mail Number, which is the extension of ESNA Office-LinX.
- **Voice Mail Handle** –Enter the Voice Mail Handle which is the extension of ESNA Office-LinX.
- **Routing Digit (e.g., AAR/ARS Access Code)** – Enter the AAR Access Code as defined in the Feature Access Code form.

```
add hunt-group 93                                              Page   2 of  60
                                  HUNT GROUP



                      Message Center: sip-adjunct

     Voice Mail Number          Voice Mail Handle          Routing Digits
                                                     (e.g., AAR/ARS Access Code)
     72036                      72036                      8
```

5.8. Configure Coverage Path

This section describes the steps for administering a coverage path in Communication Manager. Enter the **add coverage path <s>** command, where **s** is a valid coverage path number. The **Point1** value of **h93** is used to represent the hunt group number 93 created in **Section 5.7**. Default values for the other fields may be used.

```
add coverage path 93                                            Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 93
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
        Don't Answer?            y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h93           Rng:    Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

5.9. Configure SIP Endpoint

SIP endpoints and off-pbx-telephone stations will be automatically created in Communication Manager when users (SIP endpoints) are created in Session Manager.

5.10. Configure Route Pattern

For the trunk group created in **Section 5.6**, define the route pattern by entering the **change route-pattern <r>** command, where **r** is an unused route pattern number. The route pattern consists of a list of trunk groups that can be used to route a call. The following screen shows that route-pattern 93 will utilize trunk group 93 to route calls. Default values for the other fields may be used.

```
change route-pattern 93                                         Page   1 of   3
                    Pattern Number: 93  Pattern Name: 2ESNA-SM1
                             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: 93  0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 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: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.11. Configure AAR Analysis

For the AAR Analysis Table, create the dial string that will map calls to the Office-LinX via the route pattern created in **Section 5.10**. Enter the **change aar analysis <x>** command, where **x** is the starting digit. The dialed string created in the AAR Digit Analysis table should contain a map to the Office-LinX system extension, which is configured as x72036. During the configuration of the aar table, the Call Type field was set to **unku**.

```
change aar analysis 720                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 3

          Dialed            Total      Route     Call    Node   ANI
          String           Min  Max   Pattern    Type    Num    Reqd
          7202              5    5      92        unku            n
          7203              5    5      92        unku            n
```

5.12. Configure TSAPI CTI Link

This section describes the CTI link configuration between Communication Manager and Application Enablement Services. Enter the **add cti-link <m>** command, where **m** is a number between 1 and 64, inclusive. Enter a valid Extension under the provisioned dial plan. Set the **Type** field to **ADJ-IP** and assign a descriptive **Name** to the CTI link. Default values may be used in the remaining fields.

```
add cti-link 4                                              Page   1 of   3
                                    CTI LINK
 CTI Link: 4
Extension: 72000
     Type: ADJ-IP
                                                                    COR: 1
     Name: TSAPI
```

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 **procr** which was configured in **Section 5.4**. During the compliance test, the default port was utilized 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      procr             8765
CDR1                   procr             0        pcr              5852
CDR2                   procr             0        rdtt-1           9004
```

On **Page 4**, enter the hostname of the AES server for the **AE Services Server** field. The server name may be obtained by logging into the AES server using ssh and running the **uname –a** command. Enter an alphanumeric password for the **Password** field. Set the **Enabled** field to **y**. The same password will be configured on the AES server in **Section 6.1**.

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

   Server ID     AE Services        Password            Enabled     Status
                    Server
      1:         aes           [redacted]                  y           idle
      2:
      3:
      4:
      5:
      6:
```

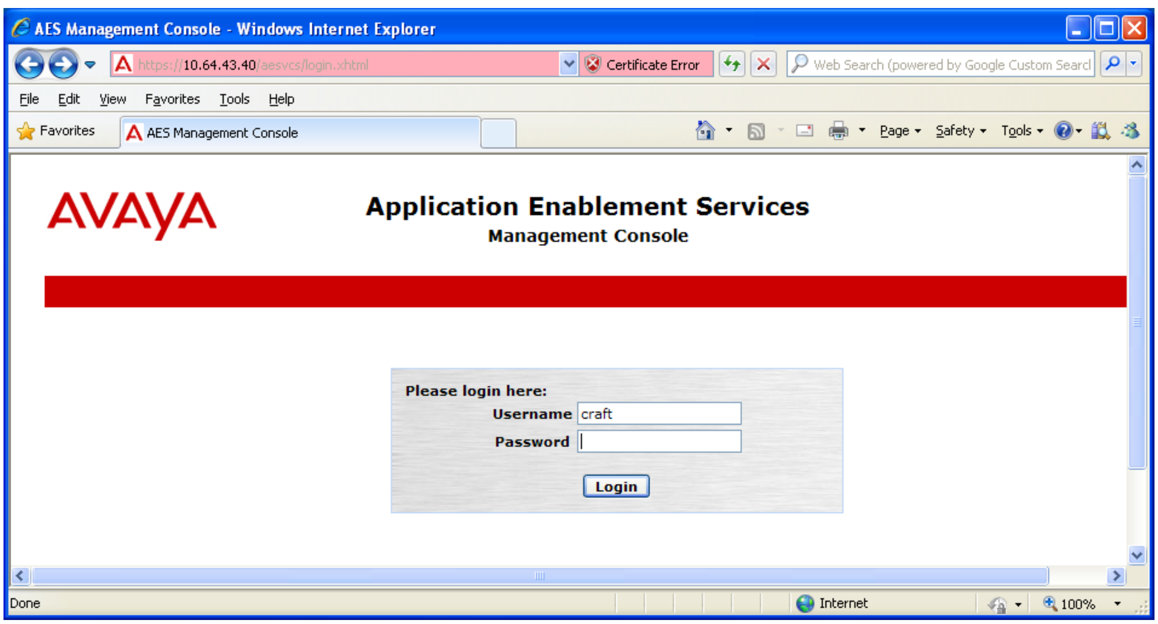
6. Configure Avaya Aura® Application Enablement Services

Application Enablement Services enable Computer Telephony Interface (CTI) applications to control and monitor telephony resources on Communication Manager. Application Enablement Services receives requests from CTI applications and forwards them to Communication Manager. Conversely, Application Enablement Services receives responses and events from Communication Manager and forwards them to the appropriate CTI applications.

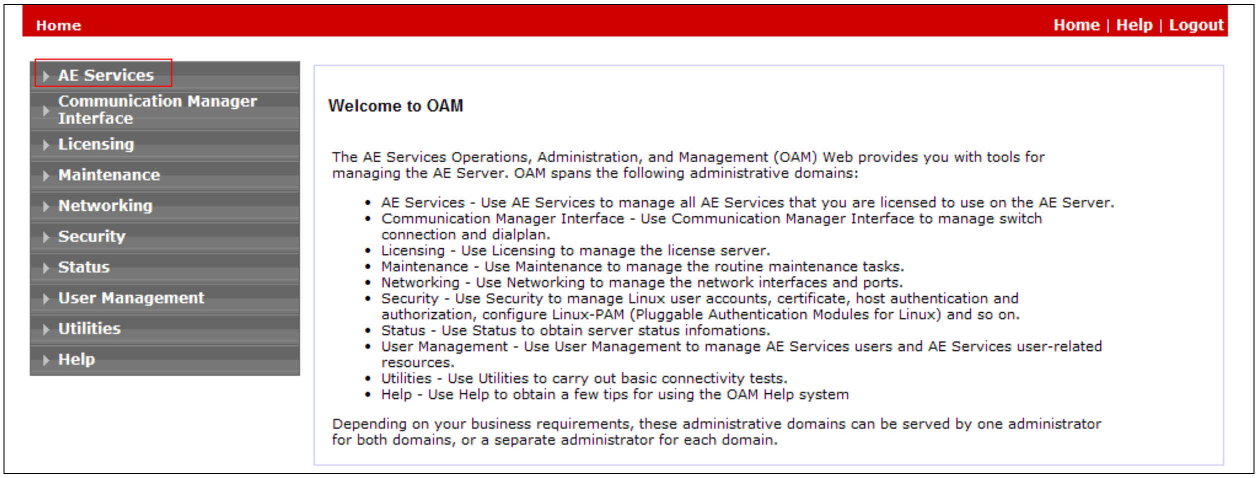
This section assumes that installation and basic administration of the Application Enablement Services server has been performed. The steps in this section describe the configuration of a Switch Connection, creating a CTI link for TSAPI, and a CTI user.

6.1. Configure Switch Connection

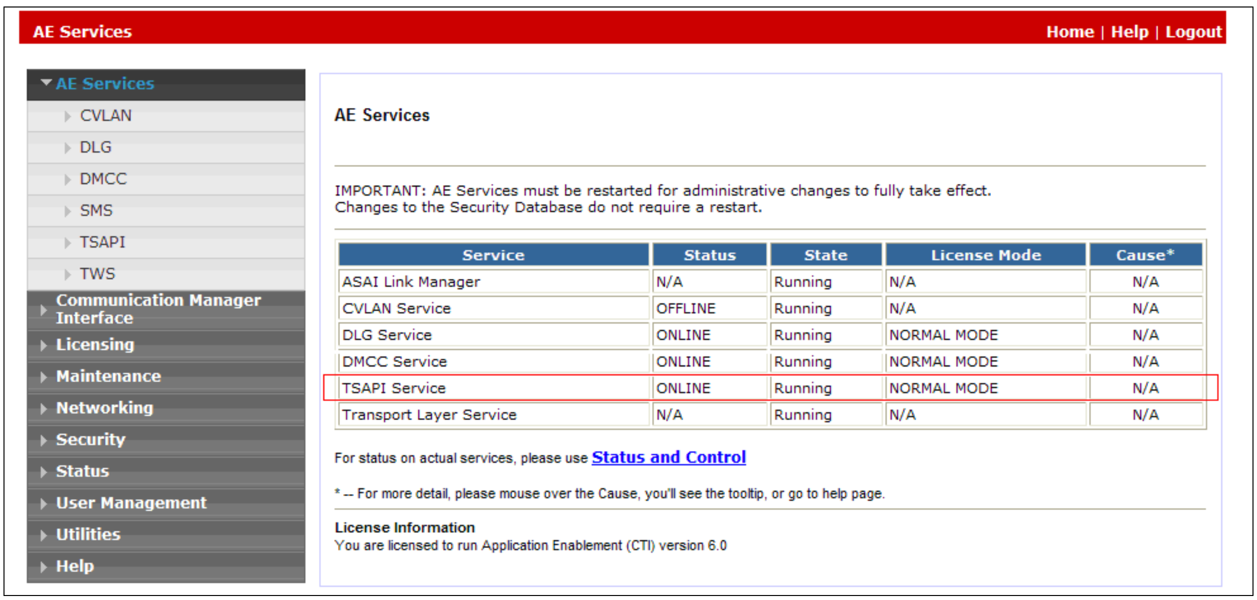
Launch a web browser, enter https://<IP address of AES server> in the URL, and log in with the appropriate credentials for accessing the Application Enablement Services Management Console page.



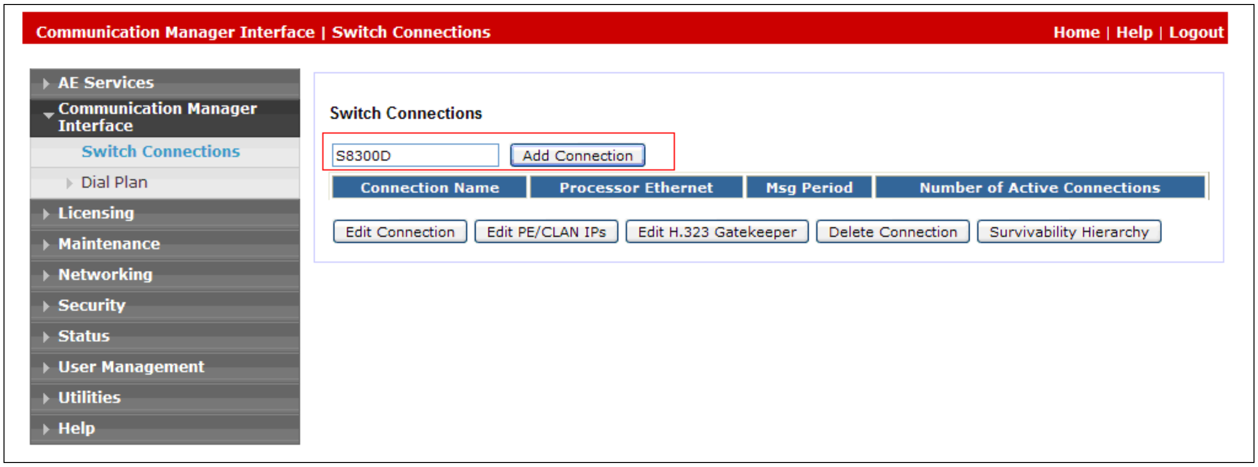
The Welcome to OAM screen is displayed next. Select **AE Services** from the left pane.



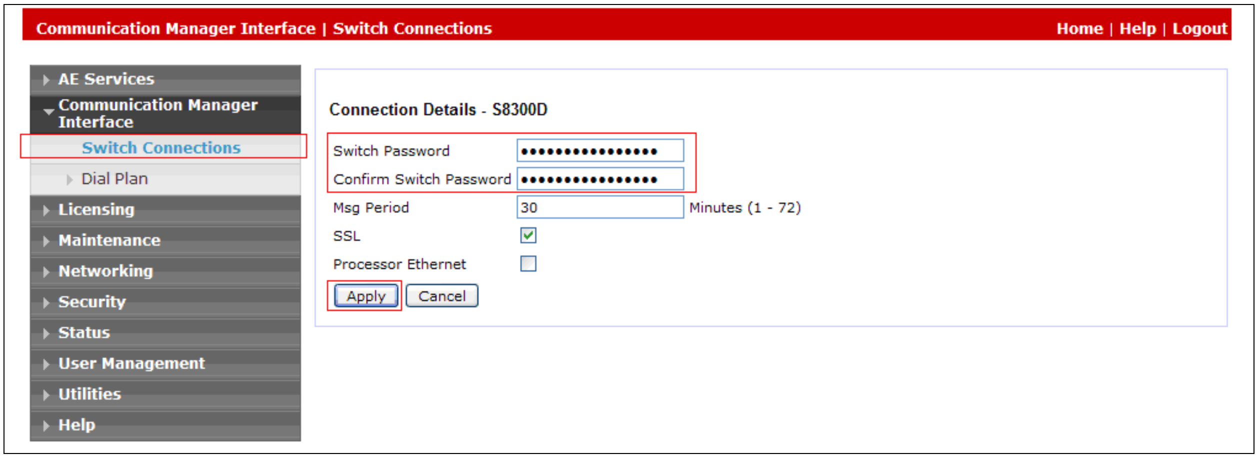
Verify that AES is licensed for the TSAPI service, as shown in the screen below.



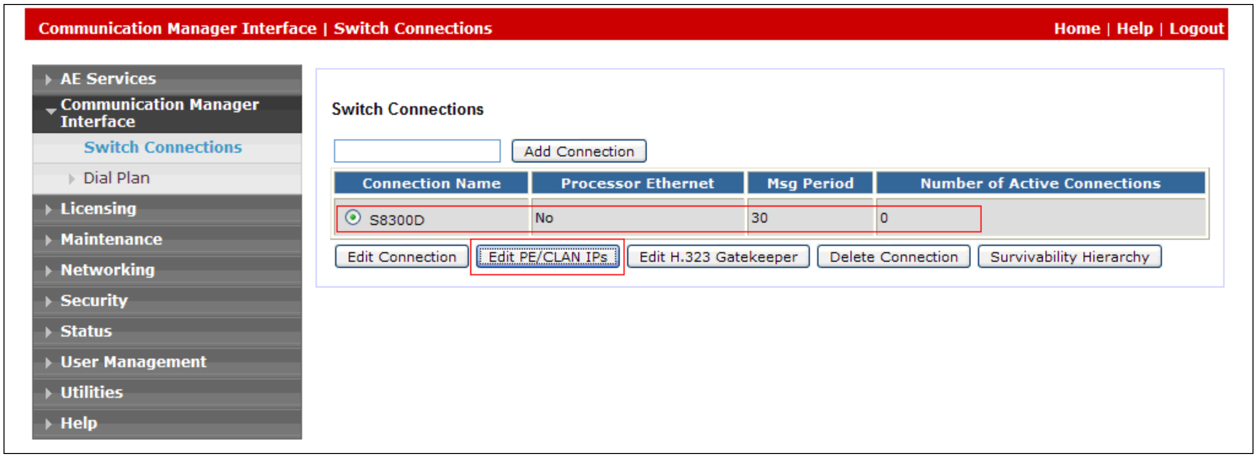
Click on **Communication Manager Interface→ Switch Connections** in the left pane to invoke the Switch Connections page. A Switch Connection defines a connection between the Application Enablement Services server and Communication Manager. Enter a descriptive name for the switch connection and click on **Add Connection**.



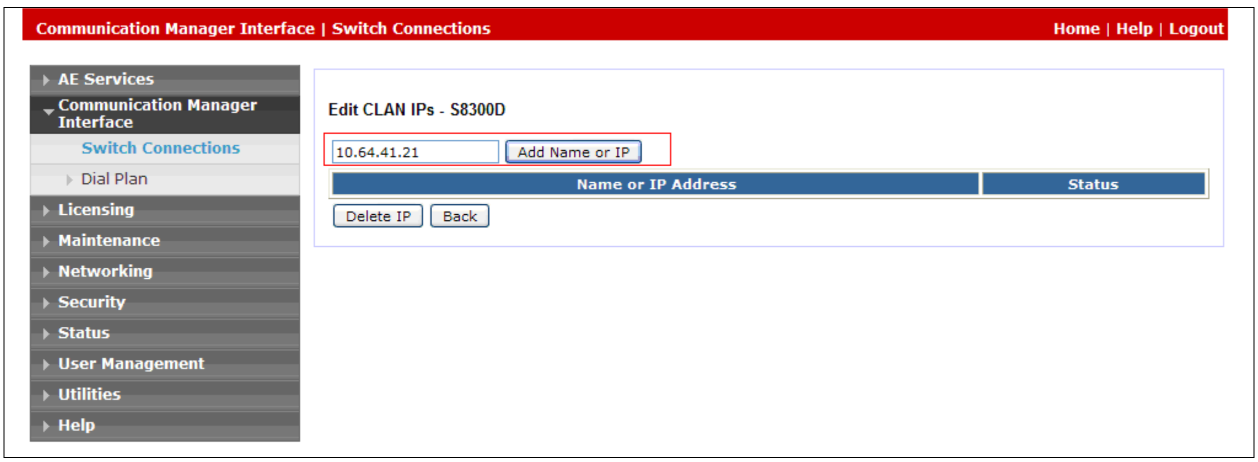
The next window that appears prompts for the Switch Password. Enter the same password that was administered on Communication Manager in **Section 5.12** Default values may be used in the remaining fields. Click on **Apply**.



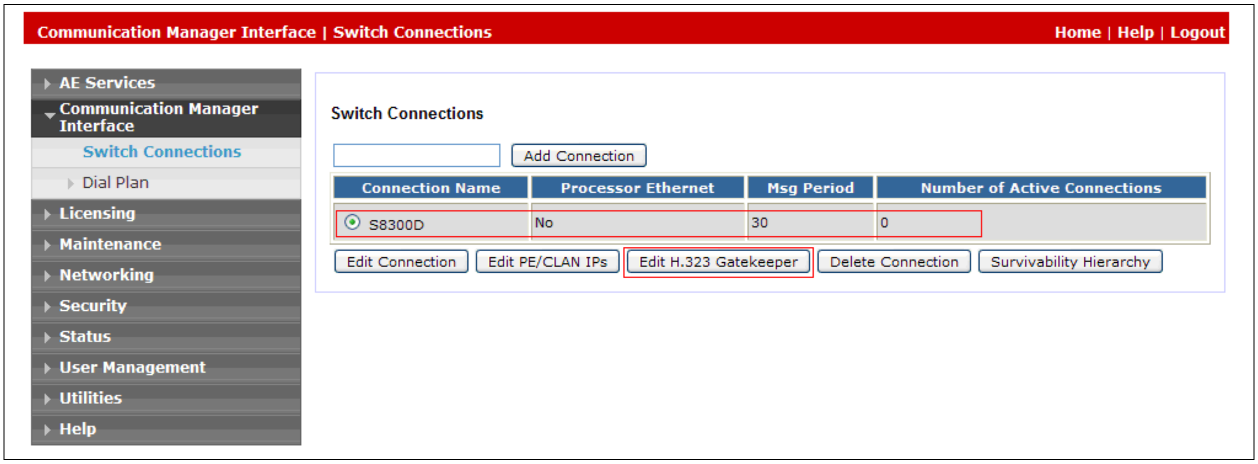
After returning to the Switch Connections page, select the radio button corresponding to the switch connection added previously, and click on **Edit PE/CLAN IPs**.



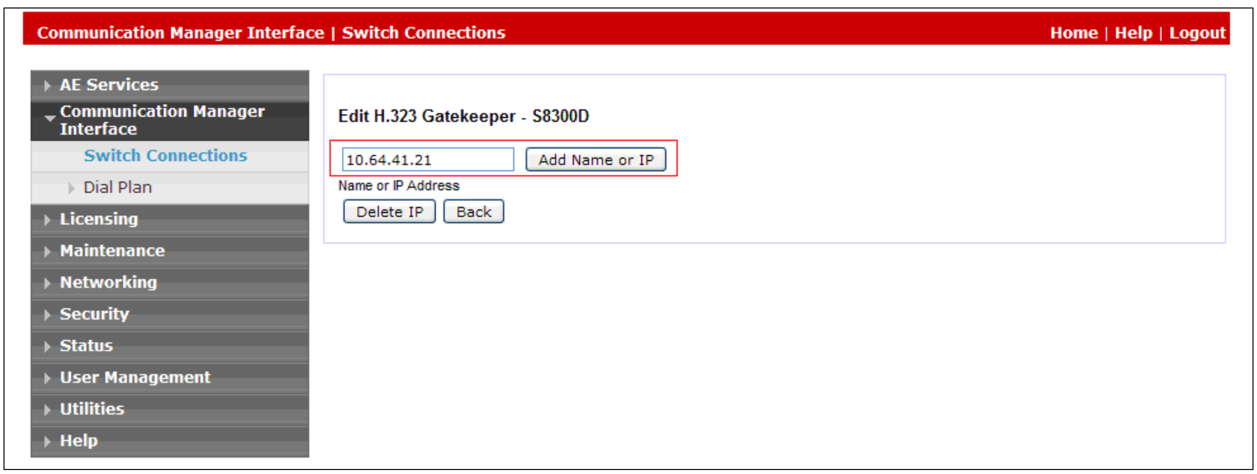
Enter the IP address of Procr used for Application Enablement Services connectivity listed in **Section 5.4**, and click on **Add Name or IP**.



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



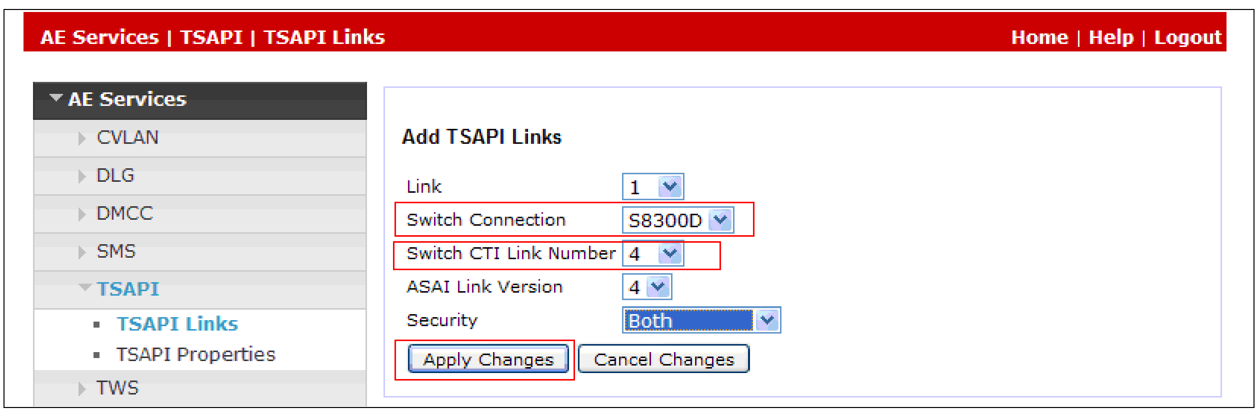
Enter the IP address of Procr used for Application Enablement Services connectivity listed in **Section 5.4**, and click on **Add Name or IP**.



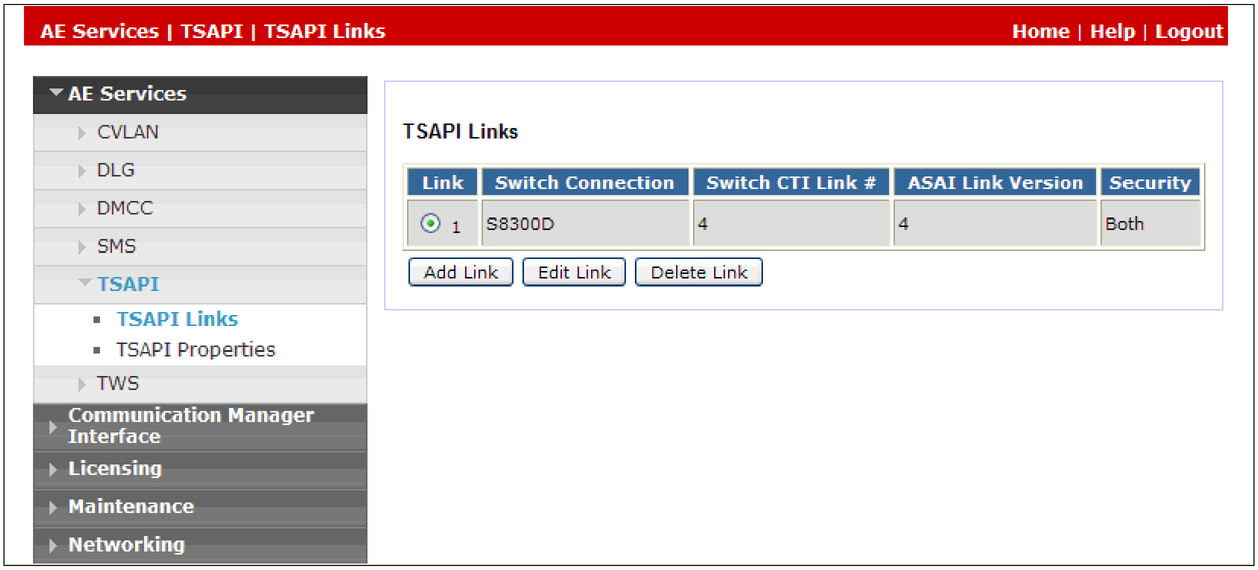
6.2. Configure TSAPI CTI Link

Navigate to **AE Services → TSAPI → TSAPI Links** to configure the TSAPI CTI link. Click the **Add Link** button to start configuring the TSAPI link.

Select the switch connection using the drop-down menu. Select the switch connection configured in **Section 6.1**. Select the **Switch CTI Link Number** using the drop-down menu. The CTI link number should match with the number configured in the cti-link form in **Section 5.12**. Click **Apply Changes**.



The following screen shows the TSAPI CTI link configuration.

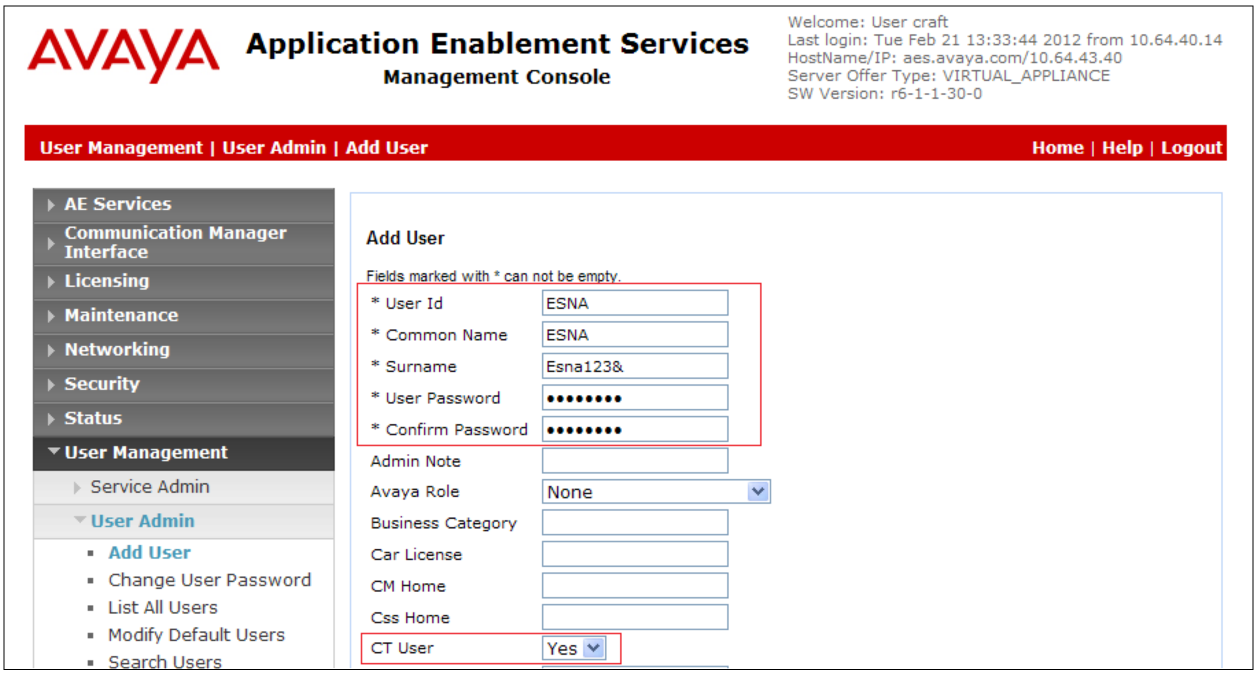


6.3. Configure CTI User

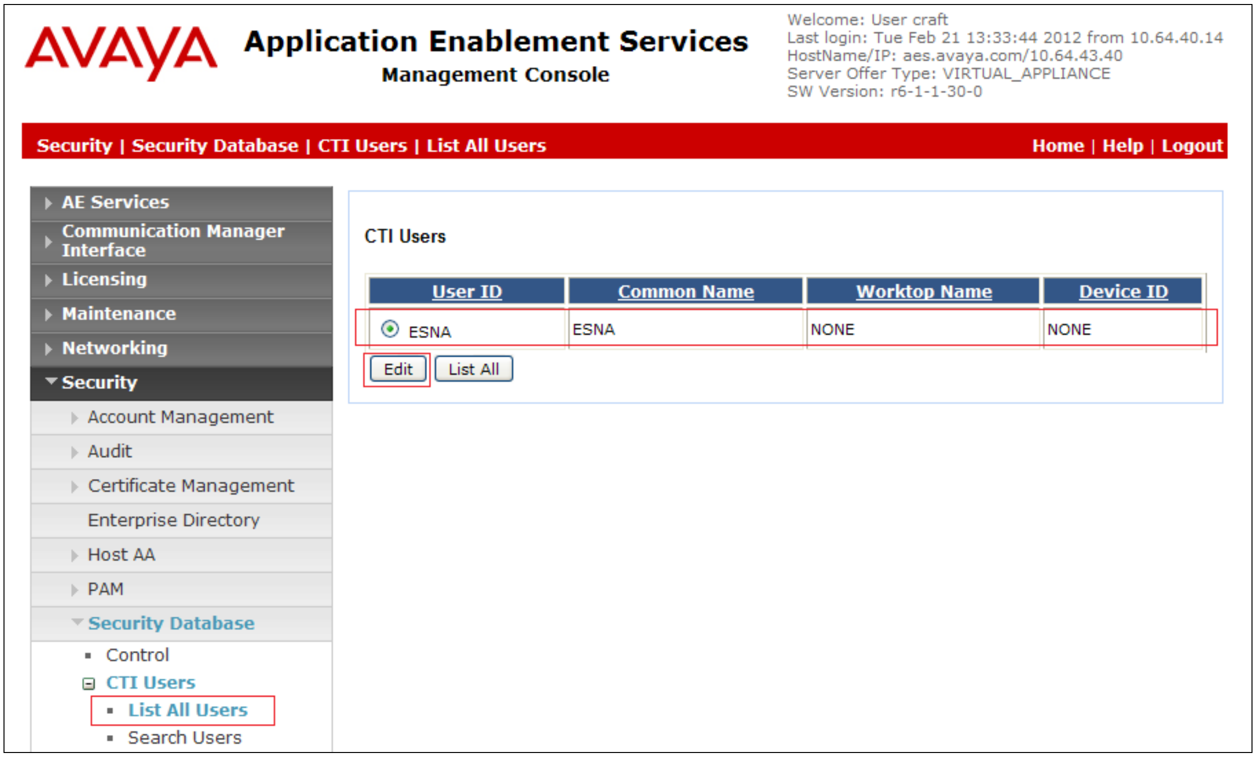
Navigate to **User Management → User Admin→Add User**. On the Add User page, provide the following information:

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

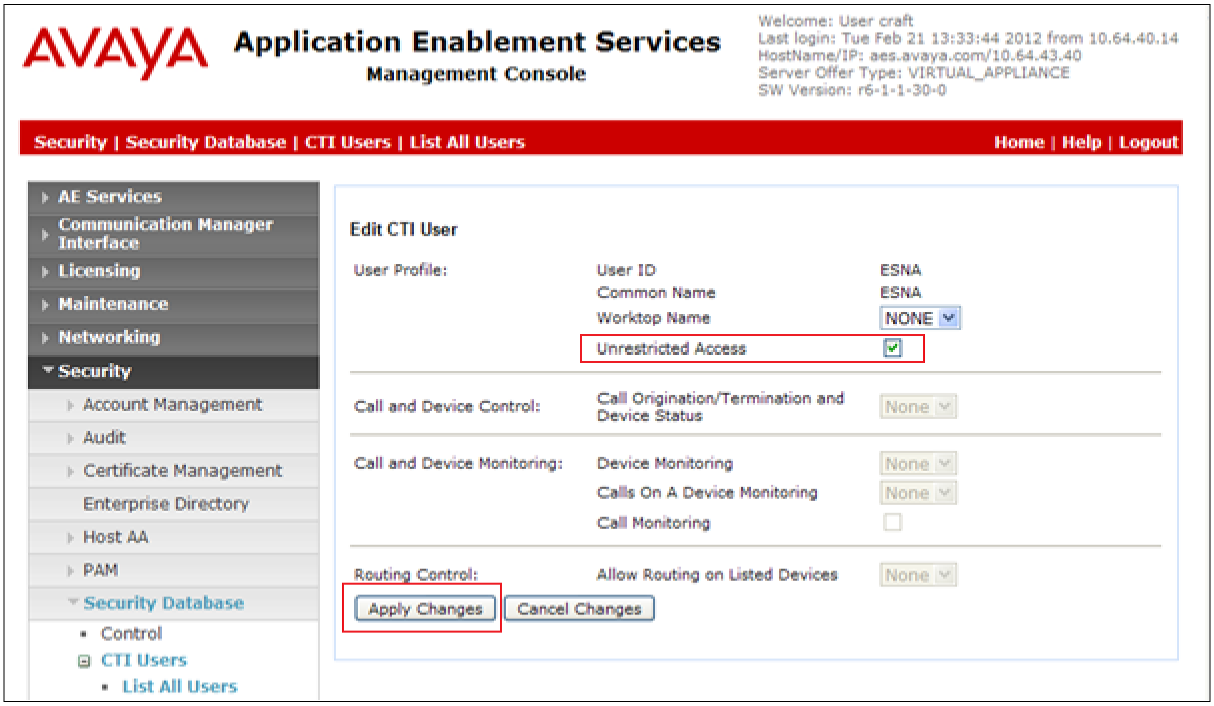
Select **Yes** using the drop-down menu on the **CT User** field. This enables the user as a CTI user. Click the **Apply** button (not shown here) at the bottom of the screen to complete the process. Default values may be used in the remaining fields.



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

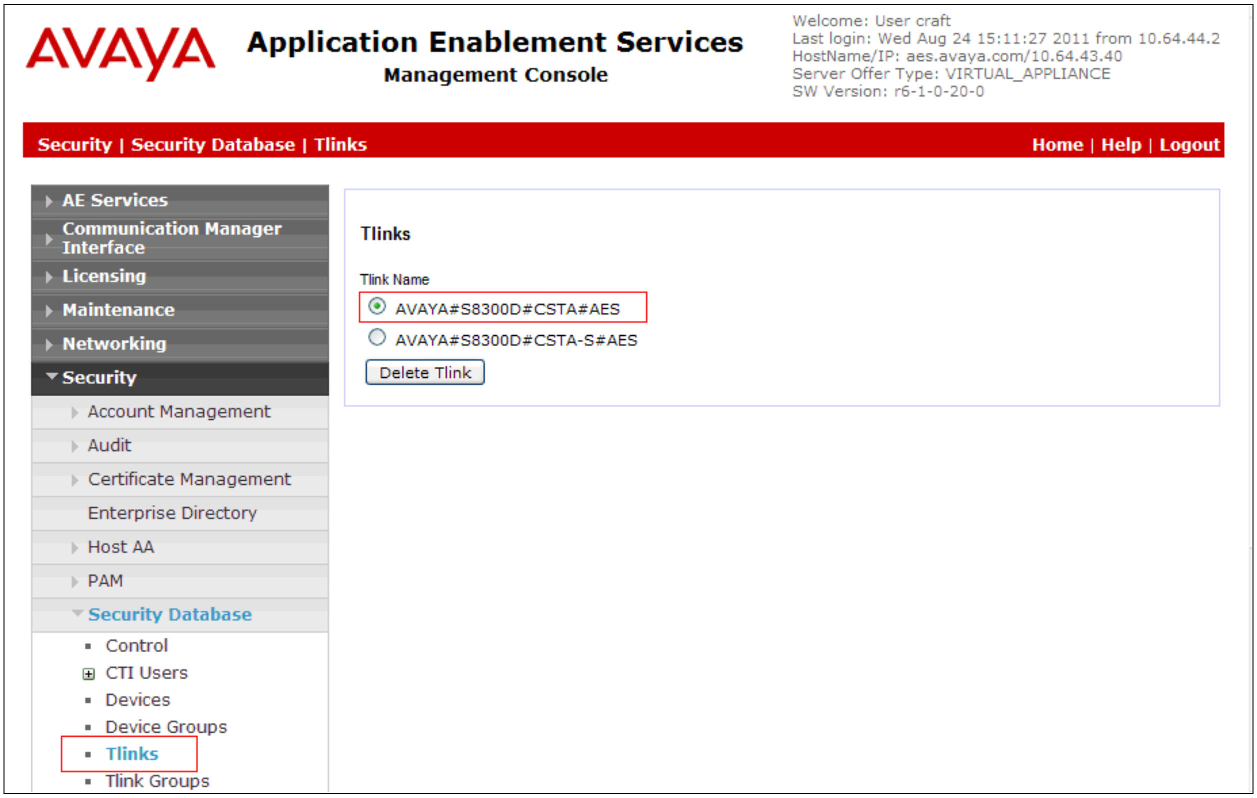


Provide the user with unrestricted access privileges by checking the **Unrestricted Access** check box. Click the **Apply Changes** button.



*Note: Since the CTI user was given **Unrestricted Access** during the compliance test, the following section becomes optional.*

Navigate to the **Security → Security Database → Tlinks** page and verify the Tlink name. The following screen shows the Tlink used during the compliance test.



7. Configure Avaya Aura® Session Manager

Since the focus of the compliance test was to verify the integration between Communication Manager and Application Enablement Services, this section will not discuss the configuration of Session Manager.

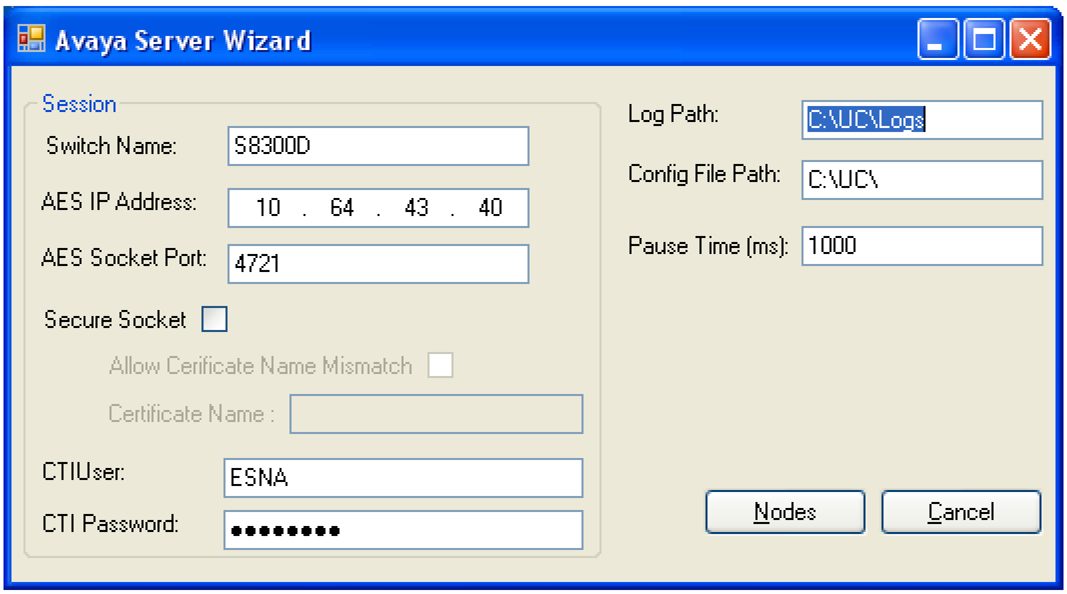
However, for configuring Session Manager, refer to [8]

8. Configure ESNA Office-LinX

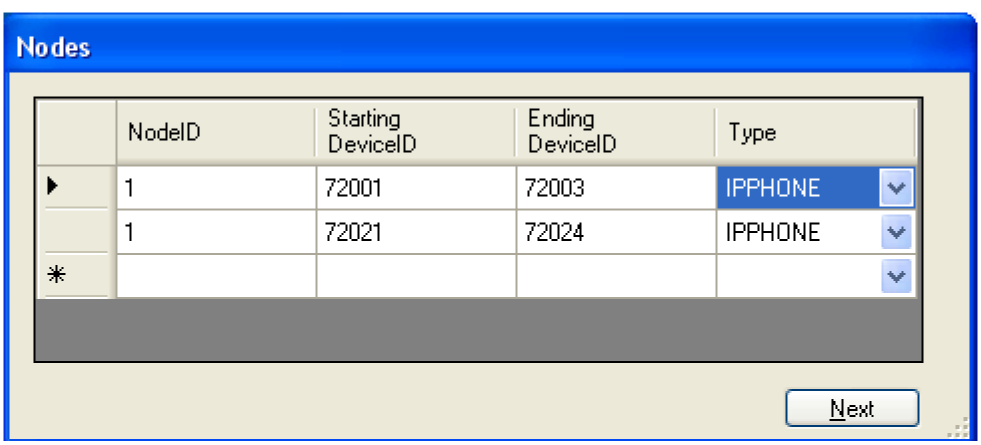
ESNA installs, configures, and customizes the Office-LinX application for their customers. Thus, this section only describes the interface configuration so that the Office-LinX can talk to Application Enablement Services and Communication Manager. To configure ESNA Office-LinX, navigate to **Start → All Programs → Office LinX → AvayaServerWizard**. From the Avaya Server Wizard page, provide the following information:

- **Switch Name** – Enter the Switch Connection name created in **Section 6.1**.
- **AES IP Address** – Enter the IP address of the Application Enablement Services server.
- **AES Socket Port** – Enter the unsecured port
- **CTI User** – Enter the CTI user created in **Section 6.3**.
- **CTI Password** – Enter the password created in **Section 6.3**.

Click the **Nodes** button to configure the monitoring extension.



The following screen shows the monitoring extension



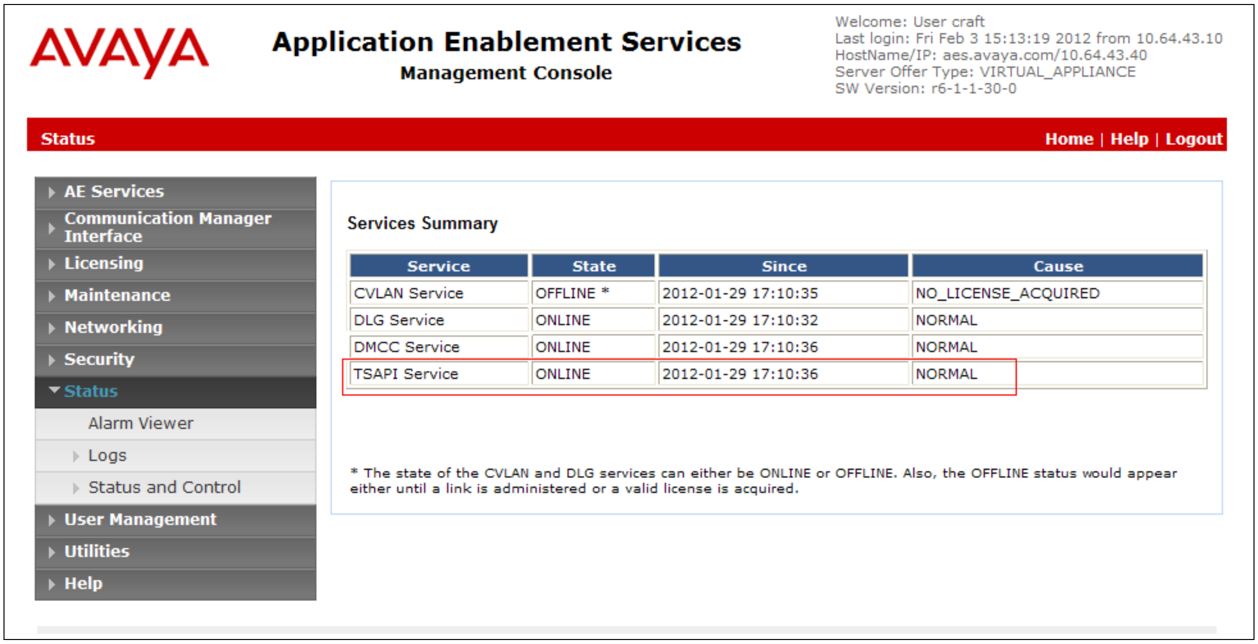
For configuring ESNA Office-LinX, refer to item [7] in **Section 10**.

9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager and Application Enablement Services with ESNA Office-LinX.

9.1. Verify Avaya Aura® Application Enablement Services

From the Application Enablement Services Management Console web pages, select **Status** from the left pane and verify the state of the TSAPI Service is set to **NORMAL**.



9.2. Verify Avaya Aura® Communication Manager

Verify the status of the administered CTI link by using the **status aesvcs cti-link** command. Verify the **Service State** is **established** for the CTI link number administered in **Section 5.12**, as shown below.

```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

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

1              no                     down          0        0
4     4        no    aes              established   75       75
```

10. Conclusion

These Application Notes describe the procedures required to configure ESNA Office-LinX to interoperate with Application Enablement Services and Communication Manager. ESNA Office-LinX successfully passed compliance testing.

11. Additional References

The following Avaya product documentation can be found at http://support.avaya.com

[1] *Administering Avaya Aura™ Communication Manager*, Release 6.0, June 2010, Issue 6.0, Document Number 03-300509

[2] *Administering Avaya Aura® Session Manager*, Release 6.1, November 2010, Issue 1.1, Document Number03-603324

[3] *Administering Avaya Aura® System Manager*, Release 6.1, November 2010

The following documentation was provided by ESNA.

[4] *Configuring Avaya AES Server Properties*, February 2012, Document Version: 8.5 (1)

[5] *Office-LinX Server Configuration Guide*, March 2012, Document Version: 8.5 (3)

[6] *Office-LinX Server Installation Guide*, March 2012, Document Version: 8.5 (4)

[7] *Office-LinX Integration with Avaya Aura CM using SES and AES,* Sep 2011, Version 8.5(1)

[8] *Application Notes for Configuring ESNA Office-LinX with Avaya Aura® Session Manager and Avaya Aura® Communication Manager* - Issue 1.0

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.