Avaya Solution & Interoperability Test Lab

Application Notes for Voxtronic Voxlog Professional Voice Recorder with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Voxtronic Voxlog Professional voice recorder to interoperate with Avaya Aura® Communication Manager using Avaya Aura® Application Enablement Services. Voxtronic Voxlog Professional voice recorder is a call recording solution. In the compliance testing, Voxtronic Voxlog Professional voice recorder used the Telephony Services Application Programming Interface from Avaya Aura® Application Enablement Services to monitor stations on Avaya Aura® Communication Manager. This solution also used the Service Observe feature via the Avaya Aura® Application Enablement Services Device, Media, and Call Control interface to capture the media associated with the monitored stations for call recording.

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

Table of Contents

1. Introduction

These Application Notes describe the configuration used to enable the Voxtronic Voxlog Professional voice recorder to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services. The Voxlog Professional voice recorder offers various methods of voice recording. For the purpose of the tests described by these Application Notes, the Avaya Aura® Communication Manager Service Observe feature was used.

Voxlog Professional can be configured to monitor specific local endpoints and record calls made to or from those endpoints. Calls between or among local endpoints which are each monitored produce multiple voice files: one for each monitored endpoint.

Voxlog Professional does not record calls which are made from Avaya telephone bridged appearances.

1.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing:

- The following test scenarios were used to test the various Voxlog Professional features:
  - Basic call
  - Hold/retrieve
  - Transfer / Blind transfer
  - Conferencing
  - Hunt group calls
  - Calls to/from bridged appearances
- Voxlog Professional's robustness was tested by verifying its ability to recover from interruptions to its external connections including:
  - The LAN connection between and Voxlog Professional and the network
  - The connection of the PBX to the network
- Voxlog Professional's robustness was further tested by verifying its ability to recover from power interruptions to the following components:
  - The Voxlog Professional server
  - The Avaya Aura® Communication Manager Server to which the Voxlog Professional is attached.

1.2. Support

Support for Voxlog Professional is available at:

General technical support from Voxlog Professional can be obtained by sending mail to: support@voxtronic.com.

Support enquiry can also be presented through the form available at the Voxtronic web site: http://www.voxtronic.com/en/Request-Support.

Direct support contact:
Ferenc Czifra
Email: **ferenc.czifra@voxtronic.com**
Telephone: +43 1 817 4846 / 343

2. Reference Configuration

The following diagram shows the configuration used for compliance testing**.**

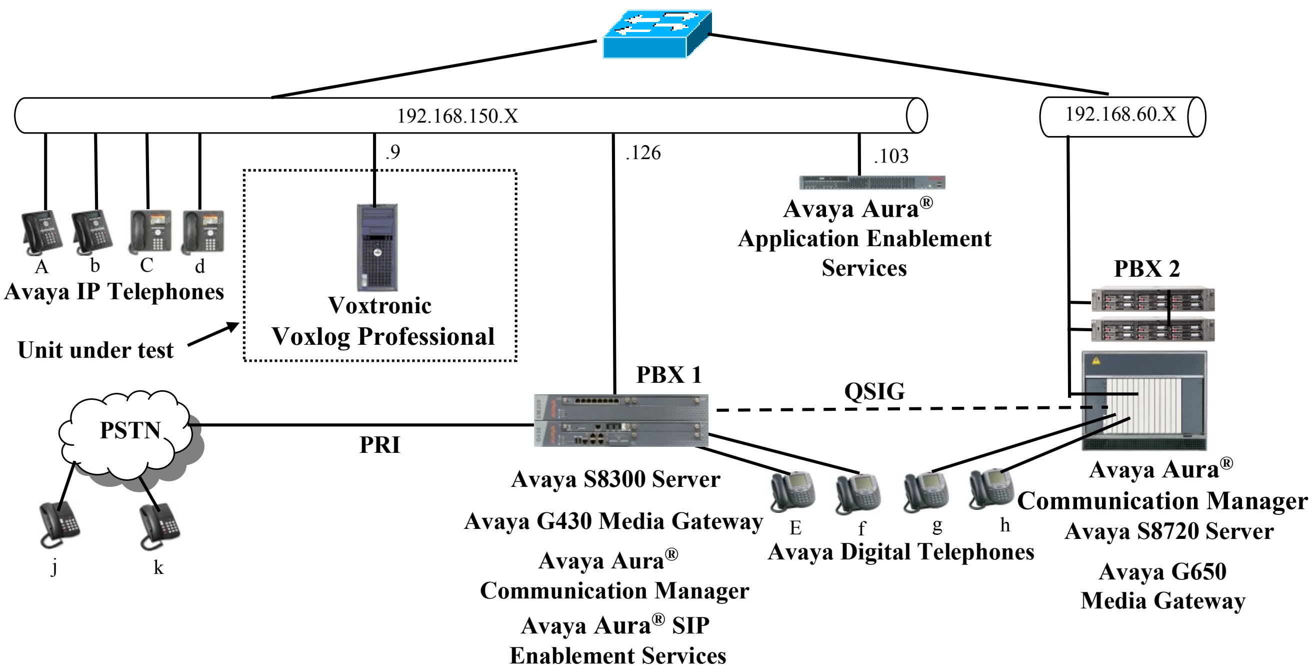

Figure 1: **Voxlog Professional Test Configuration**

In the above diagram, the Voxtronic Voxlog Professional records voice conversations from telephones attached to PBX 1. The TSAPI and DMCC services provided by Application Enablement Services are used to monitor call activity and capture voice streams associated with PBX 1. The Voxtronic Voxlog Professional voice recorder is attached to PBX 1 via the local area network. PBX 2 is included in the configuration solely to test the ability to monitor conversations which traverse a trunk to a networked PBX. The stations attached to PBX 2 are not monitored by Voxtronic Voxlog Professional.

When a call is to be recorded, the Voxlog Professional voice recorder uses the Communication Manager Service Observe feature to initiate monitoring for calls which it wishes to record. The voice stream for such calls is received via the LAN interface to PBX 1.

The PBX 2 system is attached to PBX 1 via an IP/QSIG interface, and is used as a networked PBX system. This allows remote networked telephones (g, h) to be included in the test.

The following table contains additional information about each of the telephones shown in **Figure 1**. A "*" in the "Monitored" column indicates that the telephone is monitored by the Voxlog Professional voice recorder. Note that one Virtual CTI Station is required for each endpoint which is to be monitored, as these are used by Voxlog Professional to initiate Service Observe operations.

Phone	Monitored	Model	Extension
A	*	Avaya 9640G (SIP)	10182
b		Avaya 9640G (SIP)	10184
C	*	Avaya 9630G (H.323)	10183
d		Avaya 1608 (H.323)	10065
E	*	Avaya 2410	
f		Avaya 2410	
g		Avaya 2410	60007
h		Avaya 2410	60008
j		N/A	069 7505 6174
k		N/A	015209160934
l		Hunt Group (A & C)	11304
x		CTI Station	11401
y		CTI Station	11402
z		CTI Station	11403

Table 1: Device Monitor Configuration

3. Equipment and Software Validated

Component	Version
Avaya G430 Media Gateway	30.14.0
Avaya Aura® Communication Manager Avaya Aura® SIP Enablement Services (co-resident)	R015x.02.1.016.4 Patch: 18365
Avaya Aura® Application Enablement Services	5.2.2
Avaya Aura® Application Enablement Services TSAPI Client	5.2 Build 483
Avaya 96xx H.323 Telephones	3.1.1
Avaya 96xx SIP Telephones	2.5.0
Voxtronic Voxlog Professional	3.10.0.810
Voxtronic Voxlog Professional platform OS: MS Win XP	SP3

Table 2: Hardware/Software Component Versions

4. Configure Avaya Aura® Communication Manager

The configuration information in this section covers only PBX 1 – the system to which the Voxlog Professional voice recorder is attached.

The configuration and verification operations illustrated in this section were all performed using the Communication Manager System Administration Terminal (SAT).

The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in references **Error! Reference source not found.** and **Error! Reference source not found.**.

4.1. Verify system-parameters customer-options

Use the **display system-parameters customer options** command to verify that Communication Manager is configured to meet the minimum requirements to run Voxlog Professional. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Concurrently Registered IP Stations (Page 2)	This must be sufficient to support the total number of IP stations.
IP Stations (Page 4)	This parameter must be set to "y".
Service Observing (Basic) (Page 6)	This parameter must be set to "y".
IP_Phone (Page 10)	This parameter must be set to the number of IP stations plus 1 for each station which is to be monitored.

Table 3: System-Parameters Customer-Options Parameters

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 100   40
           Maximum Concurrently Registered IP Stations: 450   3
             Maximum Administered Remote Office Trunks: 450   0
Maximum Concurrently Registered Remote Office Stations: 450   0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   30
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 1     1
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 2: System-Parameters Customer-Options Screen, Page 2

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? n                             ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? n
    Enterprise Survivable Server? n                           ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                  ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? n
     External Device Alarm Admin? n                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                 Multifrequency Signaling? y
      Global Call Classification? n         Multimedia Call Handling (Basic)? n
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n               Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

Figure 3: System-Parameters Customer-Options Screen, Page 4

```
display system-parameters customer-options                      Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                ACD? y                        Reason Codes? n
                       BCMS (Basic)? n             Service Level Maximizer? n
         BCMS/VuStats Service Level? n           Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n     Service Observing (Remote/By FAC)? n
                  Business Advocate? n            Service Observing (VDNs)? n
                    Call Work Codes? n                           Timed ACW? n
      DTMF Feedback Signals For VRU? n                  Vectoring (Basic)? y
                   Dynamic Advocate? n              Vectoring (Prompting)? n
       Expert Agent Selection (EAS)? y          Vectoring (G3V4 Enhanced)? n
                            EAS-PHD? y           Vectoring (3.0 Enhanced)? n
                  Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? n
               Least Occupied Agent? n     Vectoring (G3V4 Advanced Routing)? n
          Lookahead Interflow (LAI)? n                  Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n      Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n               Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n              Vectoring (Variables)? n
```

Figure 4: System-Parameters Customer-Options Screen, Page 6

```
display system-parameters customer-options                      Page  10 of  11
                     MAXIMUM IP REGISTRATIONS BY PRODUCT ID

Product ID  Rel. Limit          Used
IP_API_A       : 100            0
IP_API_B       : 100            0
IP_API_C       : 100            0
IP_Agent       : 100            0
IP_IR_A        : 100            0
IP_NonAgt      : 100            0
IP_Phone       : 450            2
IP_ROMax       : 450            0
IP_Soft        : 100            0
IP_Supv        : 100            0
IP_eCons       : 68             0
oneX_Comm      : 450            1
```

Figure 5: System-Parameters Customer-Options Screen Page 10

4.2. Configure system-parameters features

Use the **change system-parameters customer options** command to set the parameters as shown in the following table.

Parameter	Usage
Allow Two observers in Same Call	Set this parameter to “y”.

Table 4: System-Parameters Customer-Options Parameters

```
change system-parameters features                               Page  11 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                   Delay:
    Message Waiting Lamp Indicates Status For: station

  VECTORING
                    Converse First Data Delay: 0      Second Data Delay: 2
               Converse Signaling Tone (msec): 100         Pause (msec): 70


    Reverse Star/Pound Digit For Collect Step? n


   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
              Service Observing: Warning Tone? y     or Conference Tone? n
     Service Observing Allowed with Exclusion? n
             Allow Two Observers in Same Call? y
```

Figure 6: System-Parameters Features Screen, Page 1

4.3. Configure Feature Access Codes

Use the **change feature-access-codes** command to assign an access code to the Service Observing No Talk Access Code to allow SpeechLog to activate the Service Observe operation.

Parameter	Usage
Service Observing No Talk Access Code	Assign an otherwise unused access code which is configured by Voxlog to activate this feature.

Table 5: Parameters for the Feature Access Codes (FAC)

```
change change feature-access-codes                                     Page   5 of   8
                               FEATURE ACCESS CODE (FAC)

                         Automatic Call Distribution Features

                    After Call Work Access Code: *70
                             Assist Access Code: *71
                            Auto-In Access Code: *72
                           Aux Work Access Code: *73
                              Login Access Code: *74
                             Logout Access Code: *75
                          Manual-in Access Code: *76
      Service Observing Listen Only Access Code: *220
      Service Observing Listen/Talk Access Code: *221
          Service Observing No Talk Access Code: *222
                   Add Agent Skill Access Code: *77
                Remove Agent Skill Access Code: *78
            Remote Logout of Agent Access Code: *79
```

Figure 7: Feature Access Code (FAC) Form, Page 5

4.4. Configure Avaya Aura® Application Enablement Services Interface

Use the **change ip-services** command to configure the interface to the Application Enablement Services server, as shown in the following table.

Parameter	Usage
Service Type (Page 1)	Enter "AESVCS".
Enabled (Page 1)	Enter "y" to enable the service.
Local Node (Page 1)	Enter the IP node name for the CLAN or Processor Ethernet interface, as appropriate. For the Avaya G430 Media Gateway, enter "procr" to select the gateway's integrated Processor Ethernet interface.
AE Services Server (Page 4)	Enter the name that was assigned to the Application Enablement Services server when it was installed.
Password (Page 4)	Enter the password that was assigned to the switch connection, as shown in **Figure 30**.
Enabled (Page 4)	Enter "y" to enable the connection.

Table 6: IP Services Parameters

```
change ip-services                                              Page   1 of   4

                                   IP SERVICES
 Service     Enabled     Local        Local       Remote      Remote
  Type                   Node         Port        Node        Port
AESVCS        y      procr            8765
```

Figure 8: IP Services Form, Page 1

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

   Server ID    AE Services       Password          Enabled    Status
                   Server
      1:      AES              interop123456789      y         in use
```

Figure 9: IP Services Form, Page 4

Use the **add cti-link** command to add a CTI link by TSAPI. The link number can be any value between 1 and 64 which is not currently assigned to another link. The link number specified must be the same value that is used in the "Switch CTI Link Number" field shown in **Figure 33**. Use an unused extension as the value for the "Extension" parameter. The value chosen for the "Name" parameter is a matter of personal preference.

```
add cti-link 4                                              Page   1 of   3
                                    CTI LINK
 CTI Link: 4
Extension: 19996
     Type: ADJ-IP
                                                                       COR: 1

     Name: AES-devcon223-tsapi
```

Figure 10: CTI Link Form

4.5. Configure Network Region

Use the **change ip-network-region** command to assign an appropriate domain name to be used by Communication Manager.

```
change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: ffm.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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

Figure 11: IP Network Region Form

4.6. Configure Stations

4.6.1. Configure IP Stations

Use the **add station** command to create each of the IP stations listed in **Table 1**, using the values shown in the following table.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name, which is used for identification purposes.
Security Code	Enter an appropriate numeric string to be used as a security code.

Table 7: Configuration IP Stations

```
add station 10183                                            Page   1 of   5
                                     STATION

Extension: 10183                         Lock Messages? n              BCC: 0
     Type: 9630                          Security Code: 123456          TN: 1
     Port: S00007                      Coverage Path 1:                COR: 1
     Name: extn 10183                  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: 10183
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n



                                              Customizable Labels? yy
```

Figure 12: Station Form

4.6.2. Configure Virtual CTI Stations

Use the **add station** command to create a station for each of the Virtual CTI Stations listed in **Table 1**. A separate Virtual CTI Station is required for each station to be monitored. These stations are subsequently assigned by the Voxtronic Voxlog Professional for monitoring in **section 7.** The extension list of the provisioned Virtual CTI Stations is specified for Voxlog Professional by the **recordingDevices** parameter in **avayarps_config .xml** (see **Section 7**). Note that the station numbers must be sequential.

Parameter	Usage
Type (Page 1)	Enter "9620".
Name (Page 1)	Any alphanumeric string can be assigned as an extension name.
Security Code (Page 1)	Enter a common security code for all provisioned Virtual CTI Stations.
IP Softphone (Page 1)	Enter "y".
Button 5 (Page 4)	Create a "serv-oberv" button.

Table 8: Virtual CTI Station Parameters

```
add station 11401                                            Page   1 of   5
                                     STATION

Extension: 11401                         Lock Messages? n             BCC: 0
     Type: 9620                          Security Code: 1234           TN: 1
     Port: S00017                      Coverage Path 1:               COR: 1
     Name: CTI 11401                   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: 11401
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y

                                               IP Video Softphone? n


                                              Customizable Labels? y
```

Figure 13: Virtual CTI Station Form, Page 1

```
add station 11401                                            Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                   List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                           4:
 2: call-appr                           5: serv-obsrv
 3: call-appr                           6:


    voice-mail Number:
```

Figure 14: Virtual CTI Station Form, Page 4

4.7. Configure Hunt Group

Use the **add hunt-group** command to create a hunt group which is used to test the ability of Voxlog Professional to monitor hunt groups. Assign an unused extension to the hunt group. Add extensions for telephones “A” and “C” to the hunt group, which are assigned to IP phones which are monitored by Voxlog Professional.

Parameter	Usage
Group Name (Page 1)	Any alphanumeric string can be used as a Group Name.
Group Extension (Page 1)	Use an unused extension which is compatible with the dial plan.
GROUP MEMBER ASSIGNMENTS (Page 3)	Add the extensions which are to be assigned to this hunt group. For this test, extensions “A” and “C” are used.

Table 9: Configuration IP Stations

```
add hunt-group 3                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 3                               ACD? n
              Group Name: A + C                         Queue? n
         Group Extension: 11304                        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:
```

Figure 15: Hunt Group Form, Page 1

```
add hunt-group 3                                            Page   3 of  60
                                  HUNT GROUP
         Group Number: 3    Group Extension: 11304         Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /2
                                           Total Administered Members: 2
GROUP MEMBER ASSIGNMENTS
      Ext           Name(19 characters)       Ext           Name(19 characters)
   1: 10182         extn 10182            14:
   2: 10183         extn 10183            15:
   3:                                     16:
   4:                                     17:
   5:                                     18:
   6:                                     19:
   7:                                     20:
   8:                                     21:
   9:                                     22:
  10:                                     23:
  11:                                     24:
  12:                                     25:
  13:                                     26:

  At End of Member List
```

Figure 16: Hunt Group Form, Page 3

4.8. Configure SIP Interface to SES

Use the **add signaling-group** command to configure the Signaling Group parameters for the SIP trunk group. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Enter the Group Type as “sip”.
Co-Resident SES	Enter “y”.
Far-end Network Region	Enter the number of the network region configured in **Figure 11**.

Table 10: Signaling-Group Parameters for SIP Interface

```
add signaling-group 1                                           Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n         Co-Resident SES? y






   Near-end Node Name: procr                Far-end Node Name: procr
 Near-end Listen Port: 6001               Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain:

                                             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
                                                  Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Figure 17: Signaling Group Form

Use the **add trunk-group** command to configure the SIP interface to Avaya SES. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Specify the Group Type as “sip”.
Group Name	Select an appropriate name to identify the device.
TAC	Specify a trunk access code that can be used to provide dial access to the trunk group.
Service Type	Designate the trunk as a “tie” line to a peer system.
Signaling Group	Enter the number assigned to the SIP signaling group shown in **Figure 17**.
Number of Members	Specify sufficient number of members to support the maximum simultaneous connections required.

Table 11: Trunk-Group Parameters for the SIP Interface

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: ses                          COR: 1        TN: 1        TAC: *001
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                     Signaling Group: 1
                                                   Number of Members: 30
```

Figure 18: Trunk Group Form

After the configuration steps for Avaya Aura® Communication Manager are complete, enter the command **save translation** to make these changes permanent.

5. Configure Avaya Aura® SIP Enablement Services

Configure SIP Enablement Service by entering "<SES IP Address>/admin" in a web browser. After entering the administrator name and password, the following screen content is displayed:

Figure 19: SIP Enablement Service "Top" Configuration Screen

5.1. Server Configuration

Select “System Properties” from the “Server Configuration” menu from the left pane of the screen. Enter values in this screen as shown in the following table:

Parameter	Usage
SIP Domain	Enter the same SIP domain name assigned in **Figure 11**.
License Host	Enter the IP address of the license host, in this case the IP address of the SES server.
Management System Login / Password	Enter the authentication credentials for Communication Manager administrative login.

Table 12: Parameters for System Properties

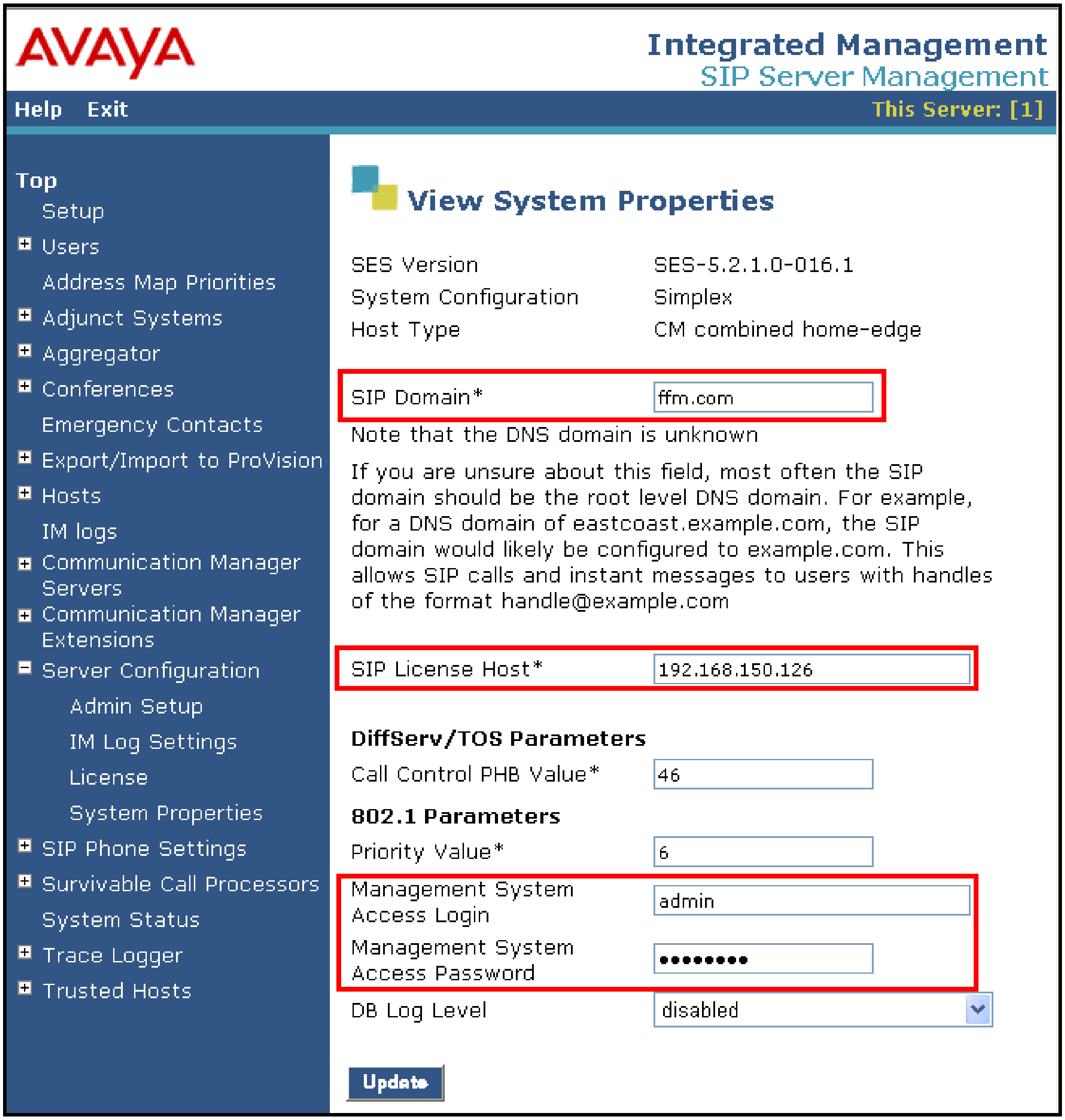

Figure 20: System Properties Screen

Navigate to “Server Configuration” → “Admin Setup”, select the “This server is the SES Master…” radio button, and click “Setup”.

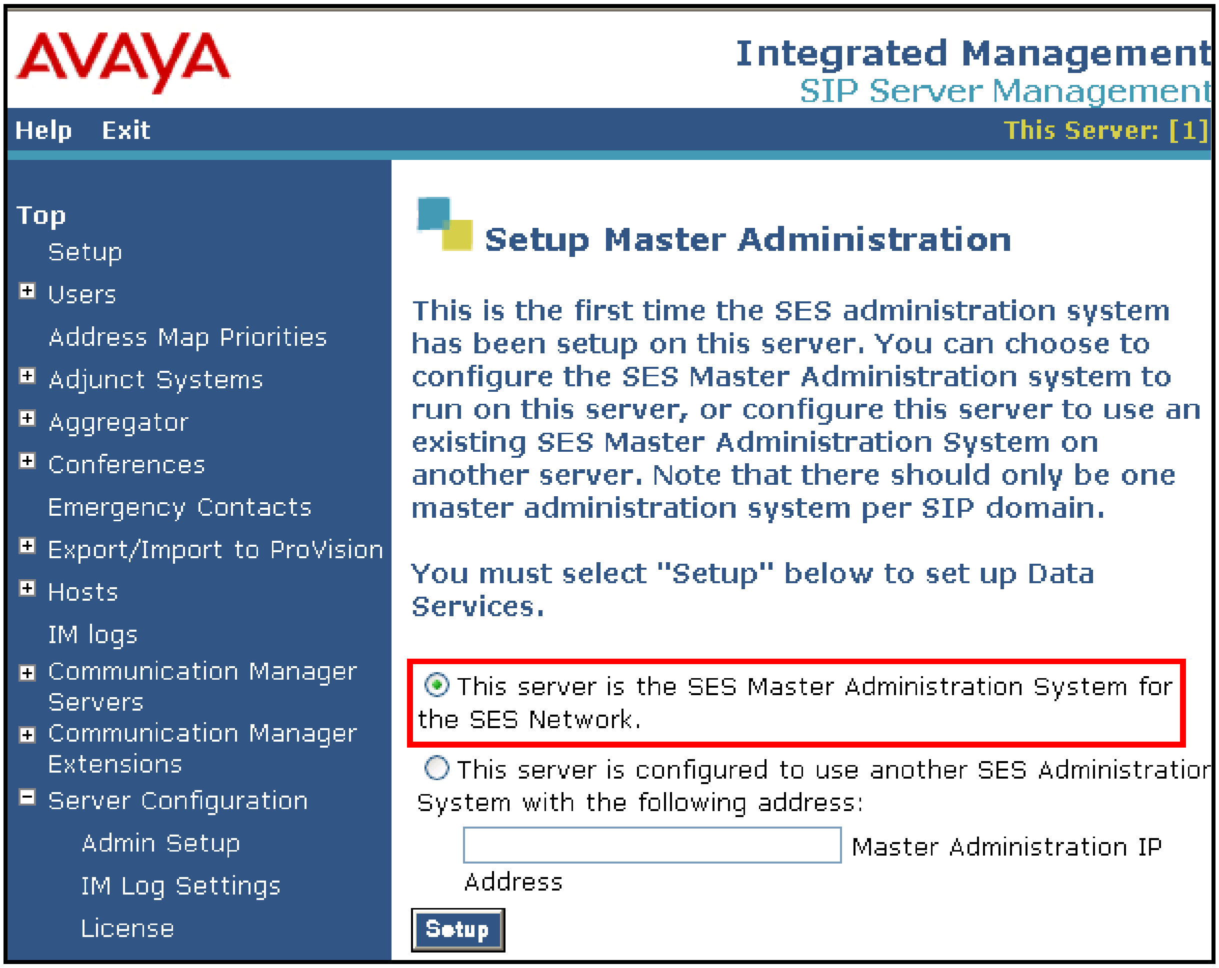

Figure 21: Master Administration Screen

5.2. Add Hosts

Navigate to "Hosts" → "Add Host" from the left pane of the top level screen shown in **Figure 19**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed.

Parameter	Usage
Profile Service Password	Enter the password which was entered from the initial setup script when SES was installed.

Table 13: "Add Host" Parameters

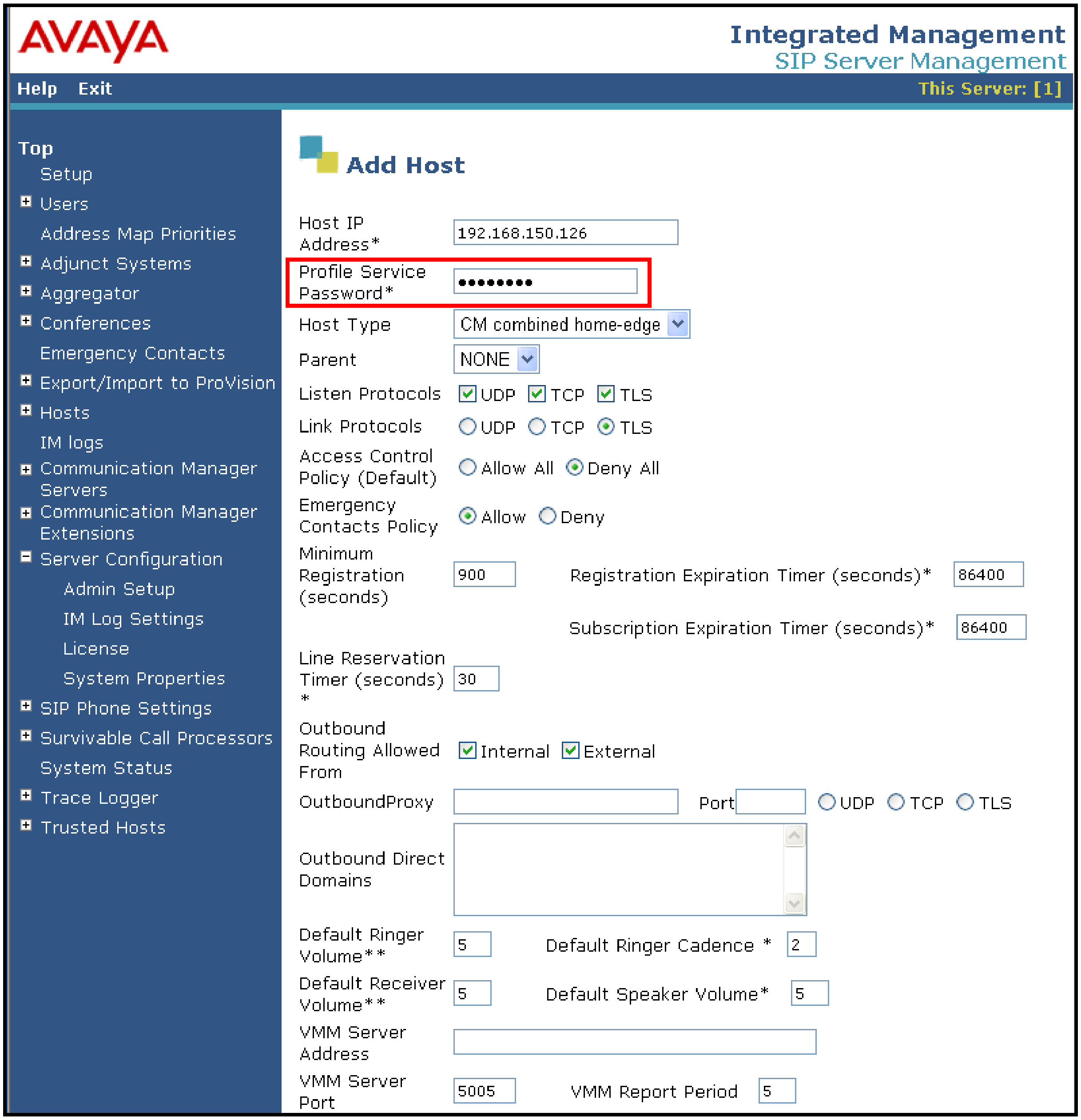

Figure 22: Add Host Screen

Navigate to "Communication Manager Servers" → "list", enter the "Near-end Listen Port" value assigned to the SIP trunk in **Figure 17** into the "SIP Trunk Port" field, and click "Update".

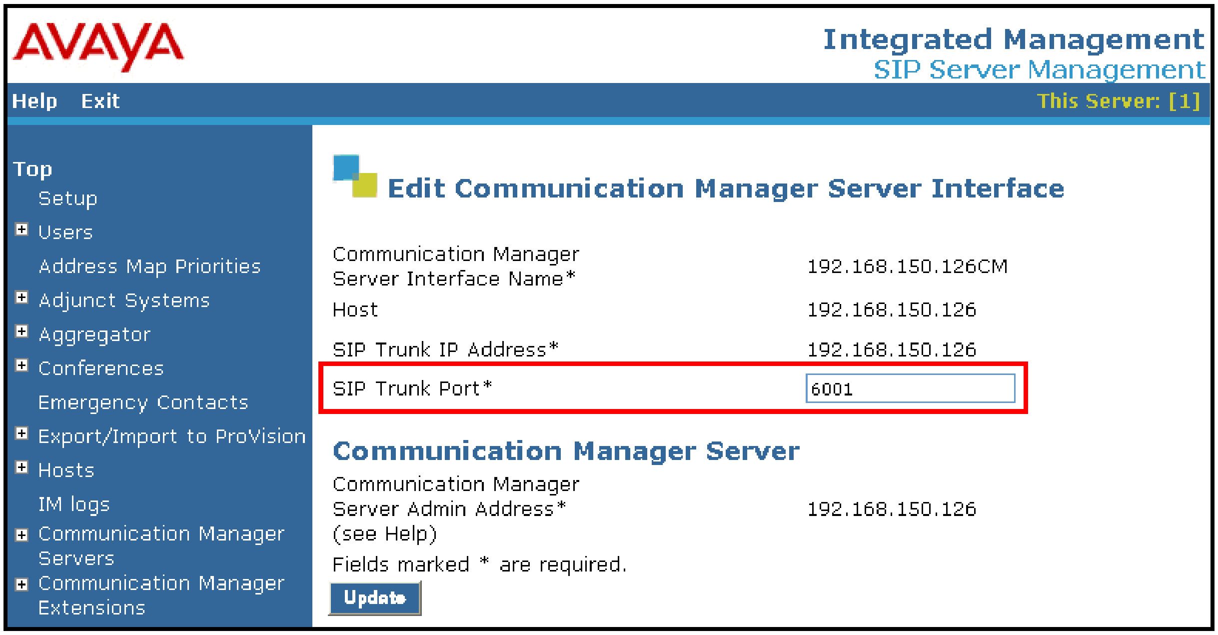

Figure 23: Edit Communication Manager Server Interface Screen

Navigate to “Users” → “Add”. For each of the SIP phones in **Table 1**, enter the parameters shown in the following table, and click “Add”.

Parameter	Usage
Primary Handel	Enter the SIP phone extension.
User ID	Enter the SIP phone extension.
Password / Confirm	Enter password with which the phone is to use for registration.
First Name / Last Name	Enter first and last names to identify the telephone user.
Add Communication Manager Extension	Check this box.

Table 14: SIP User Parameters

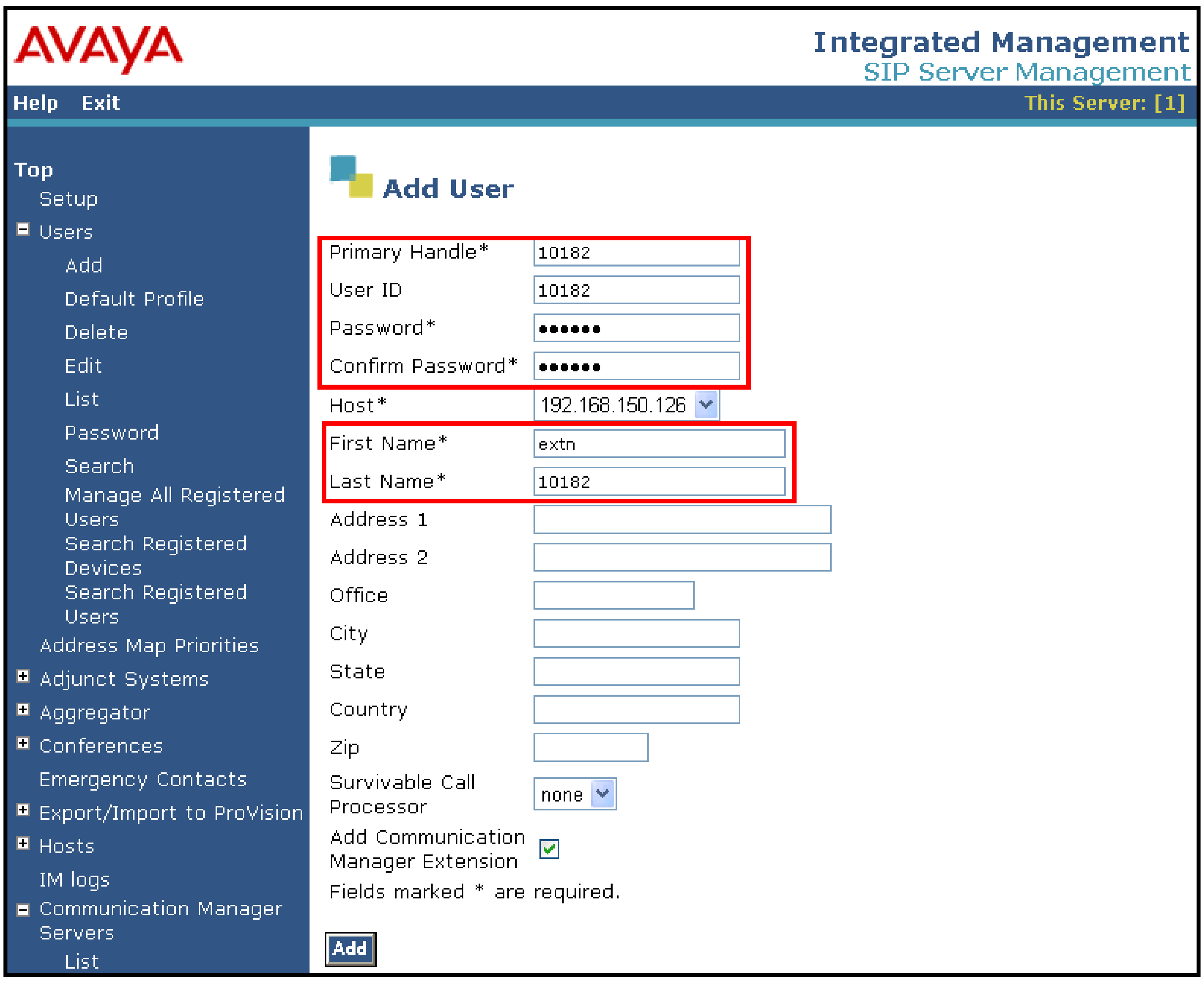

Figure 24: Add User Screen

Enter the SIP phone extension and click “Add”.

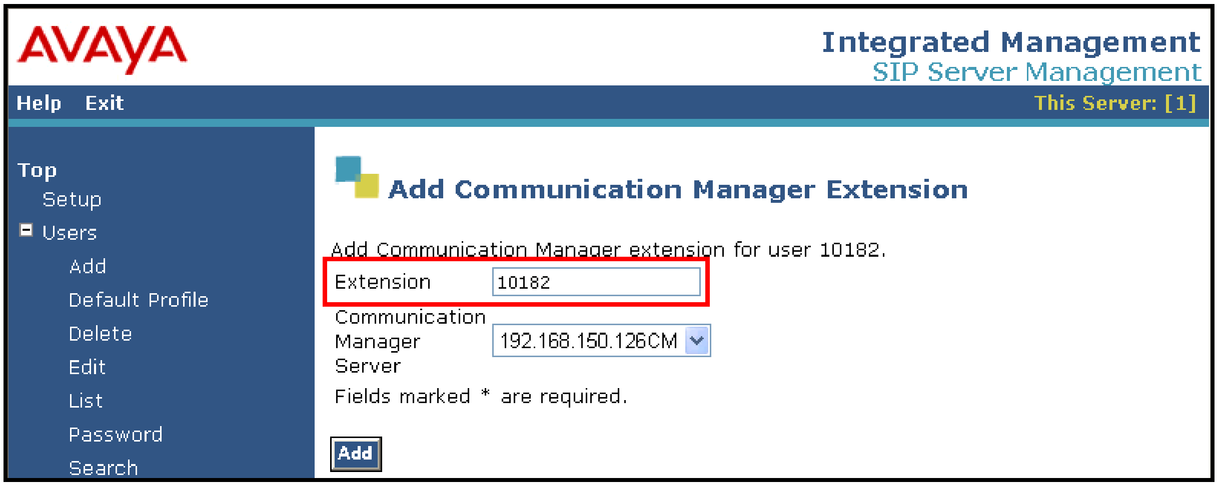

Figure 25: Add Extension Screen

6. Configure Avaya Aura® Application Enablement Services

The Application Enablement Services server is configured via a web browser by accessing the following URL:

https://<AES server address>/

Click "Continue To Login".

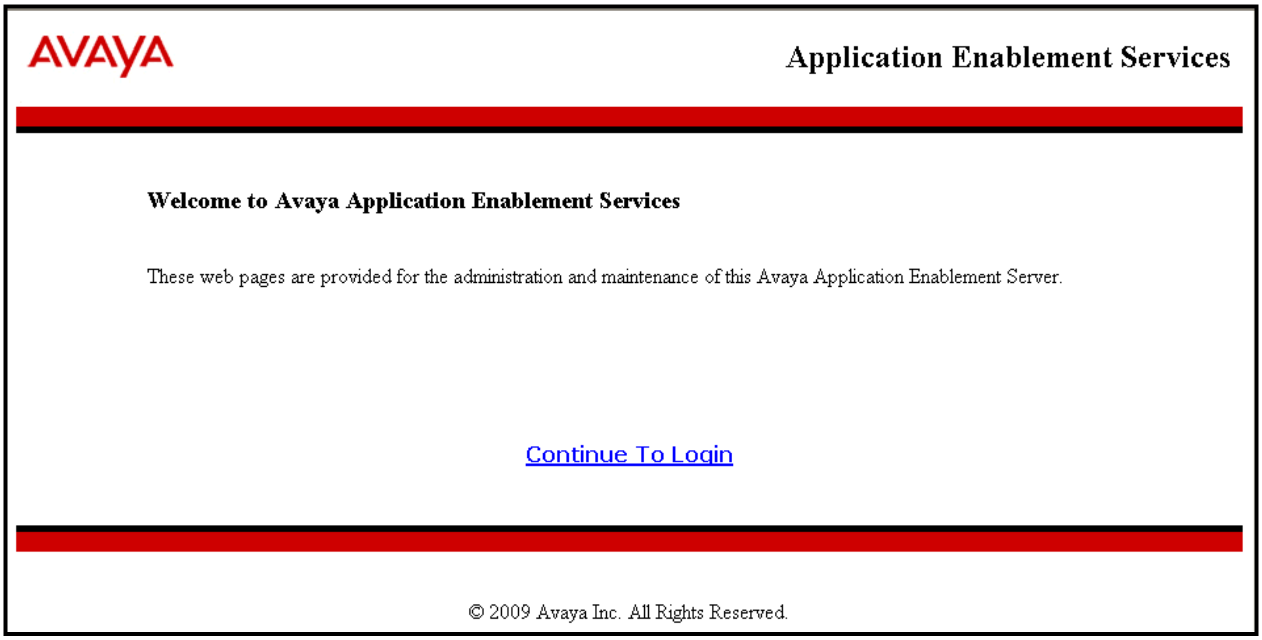

Figure 26: Avaya Application Enablement Services Welcome Screen

Once the login screen appears, enter the login credentials for performing administrative activities.

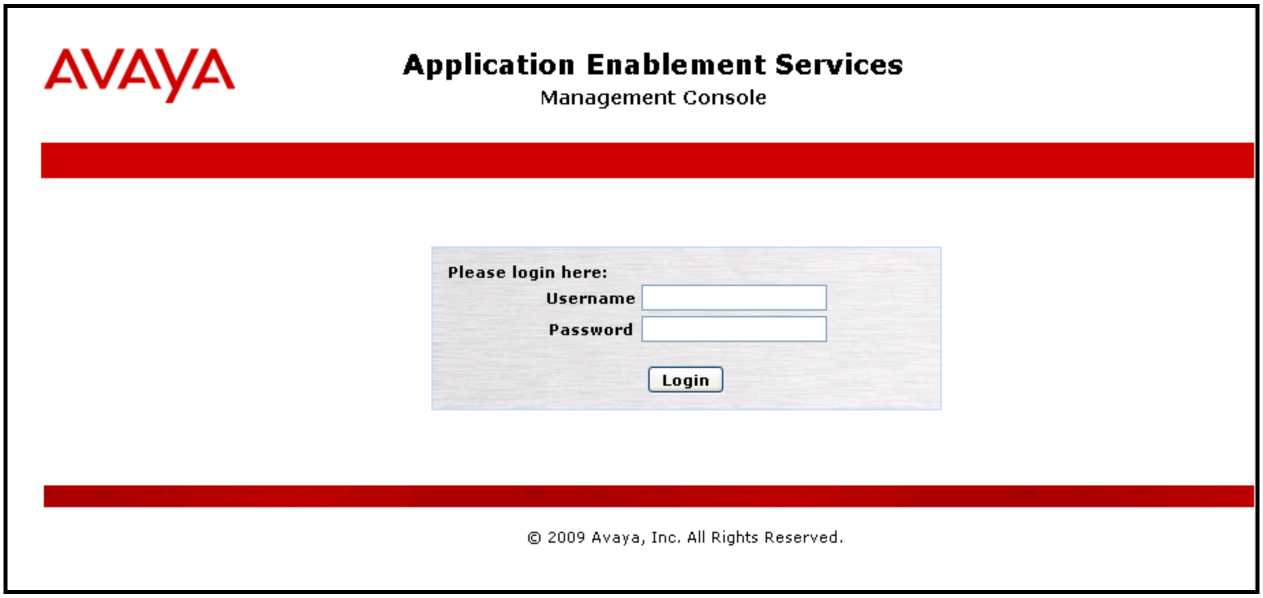

Figure 27: Application Enablement Services Login Screen

Click "AE Services" in left frame.

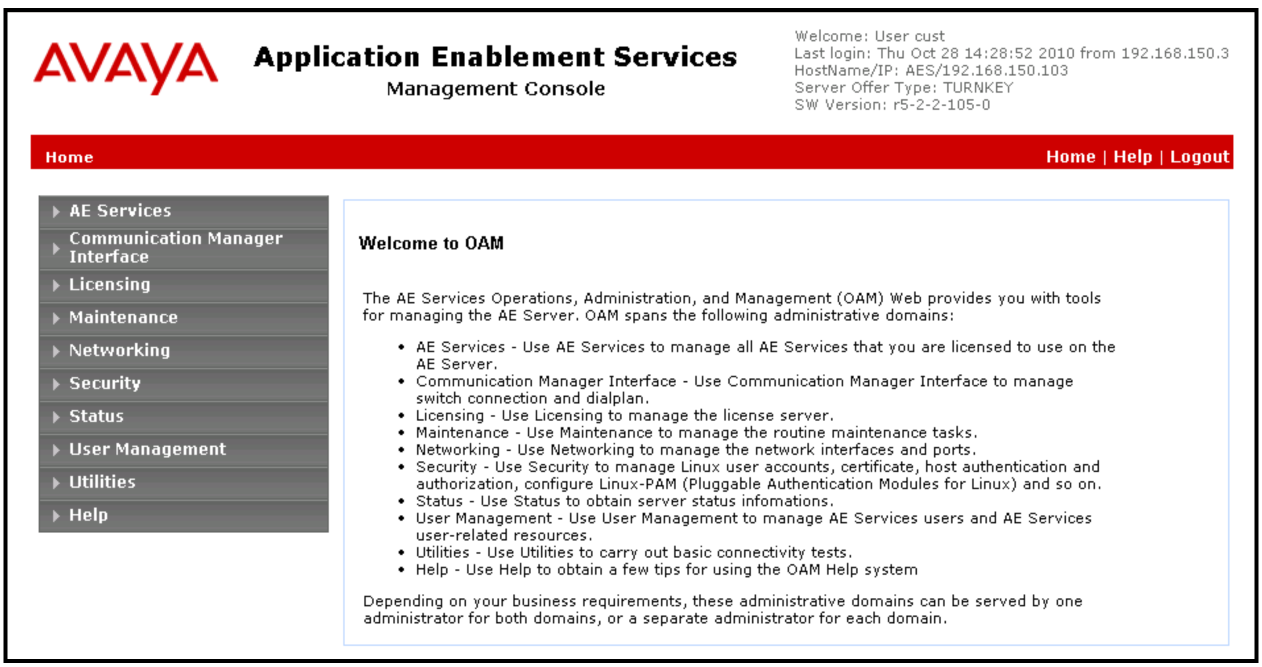

Figure 28: Application Enablement Services Main Screen

Navigate to **Communication Manager Interface→Switch Connections**. Enter the name of the Switch Connection to be added, and click on the "Add Connection" button. This name should match what will be used by the Voxtronic Voxlog Professional in **section 7.**

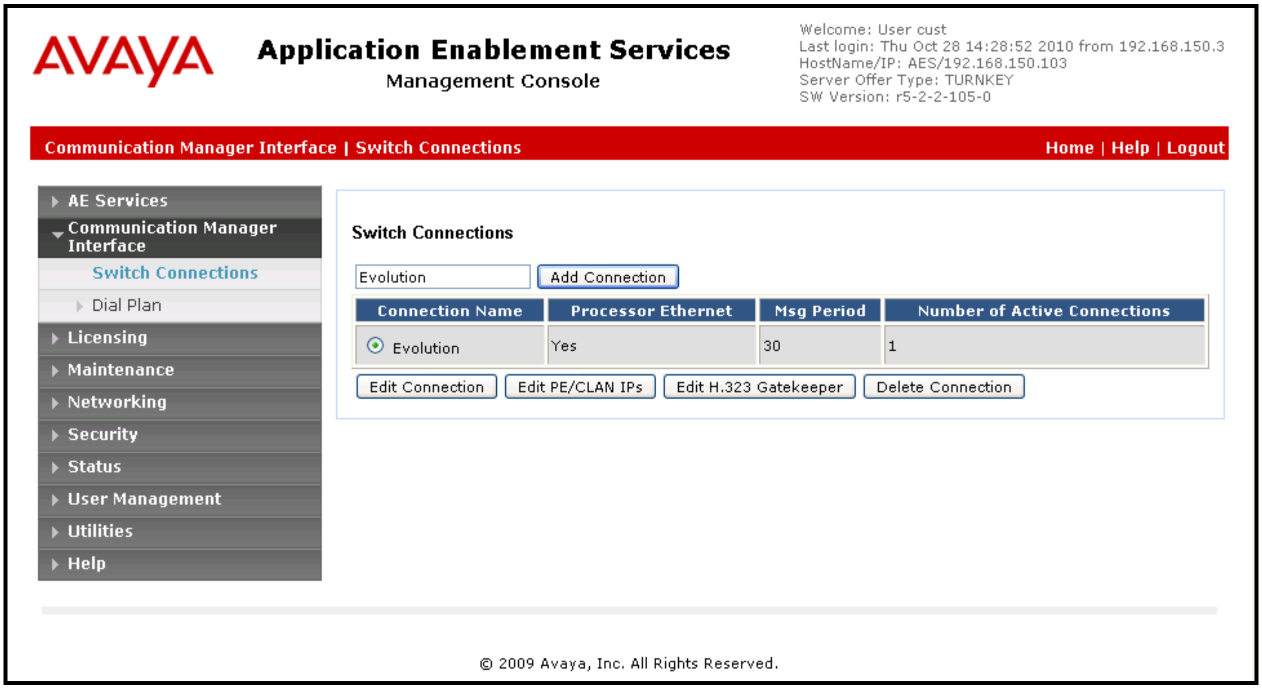

Figure 29: Switch Connection Screen

The Communication Manager Interface | Switch Connections is presented. At this point, enter the screen fields as described in the following table, and click the "Apply" button.

Parameter	Usage
Switch Password	The Switch Password must be the same as was entered into the Communication Manager AE Services Administration form via the "change ip-services" command, described in **Figure 9**. Passwords must consist of 12 to 16 alphanumeric characters
SSL	SSL (Secure Socket Layer) is enabled by default. Keep the default setting unless you are adding a Switch Connection for a DEFINITY Server CSI
Processor Ethernet	Check this box.

Table 15: Configuration of Switch Password

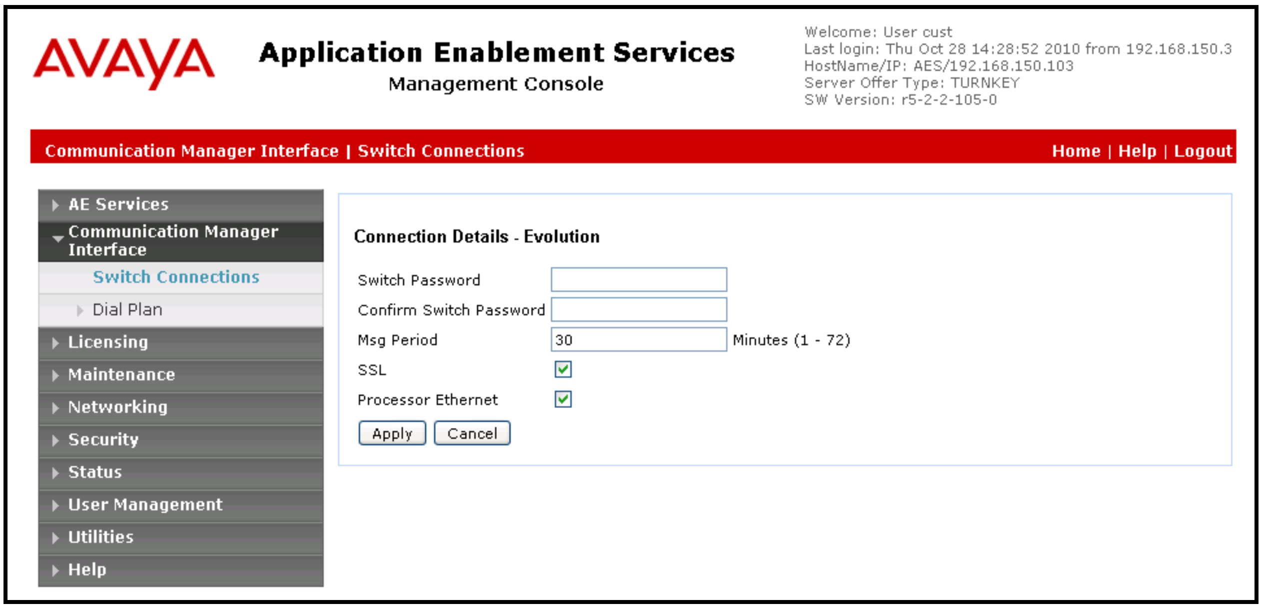

Figure 30: Set Switch Password Screen

From the **Communication Manager Interface→Switch Connections** screen, click the "Edit PE/CLAN IPs" button, (not shown), to display the screen shown below. Enter the IP address of the Processor Ethernet or CLAN that Application Enablement Services will use for communication with the switch, and click the "Add/Edit Name or IP" button.

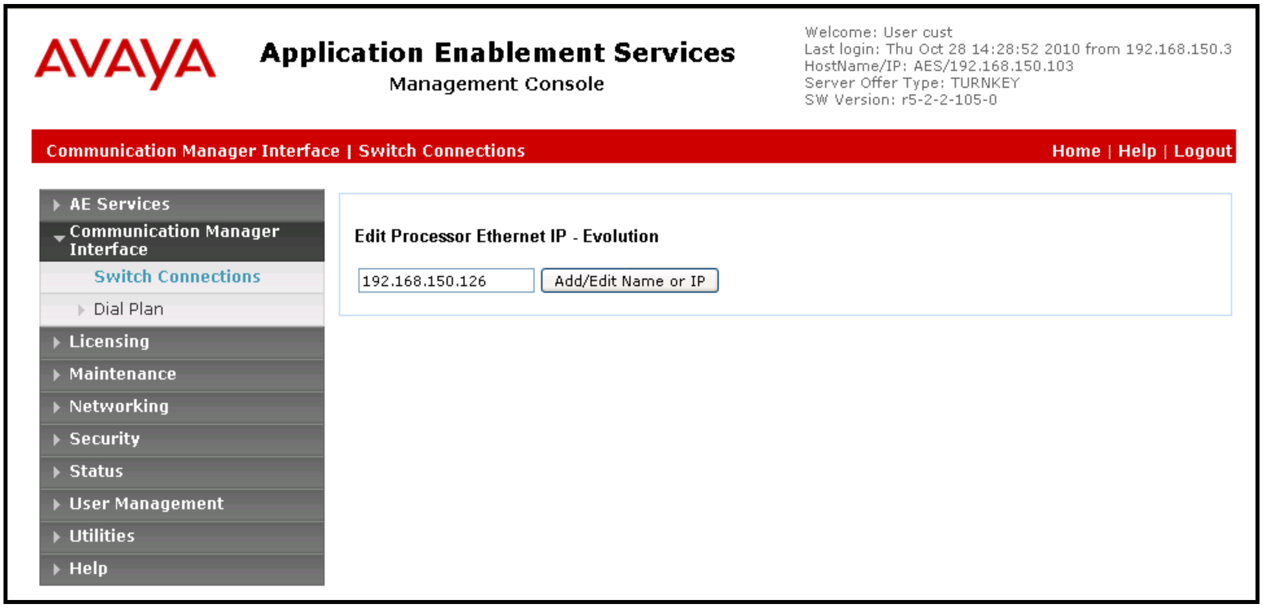

Figure 31: Edit Processor Ethernet IP Screen

Navigate to **AE Services→TSAPI→TSAPI Links.** The following screen is displayed. Click the "Add Link" button.

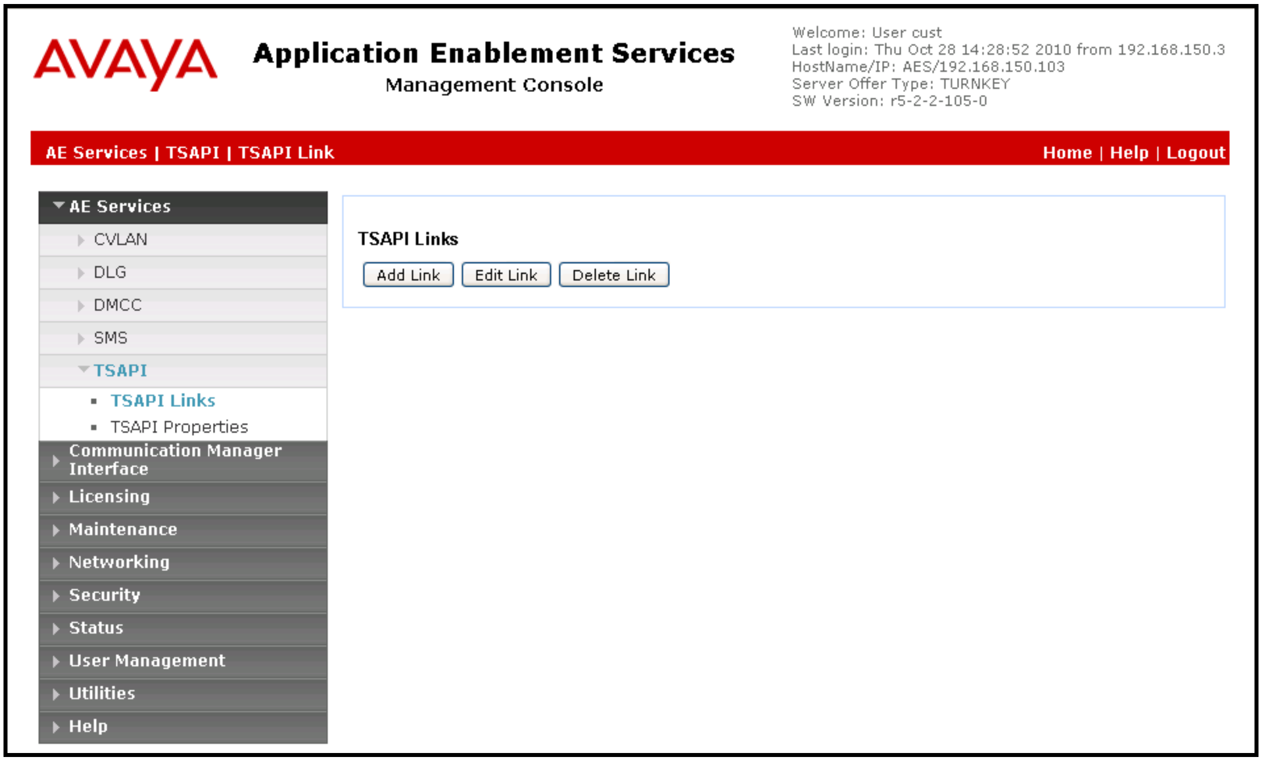

Figure 32: TSAPI Links Screen

Fill in the parameters for the link to be added. The "Link" parameter must be a unique value between 1 and 16. The "Switch Connection" parameter should be the name of the Avaya Server which is to be controlled by this link. The value for the TSAPI "Switch CTI Link Number" must be a value between 1 and 64, and must be the same as was used in the Avaya Aura® Communication Manager "add cti-link" configuration command in **Figure 10**. Click the "Apply Changes" button.

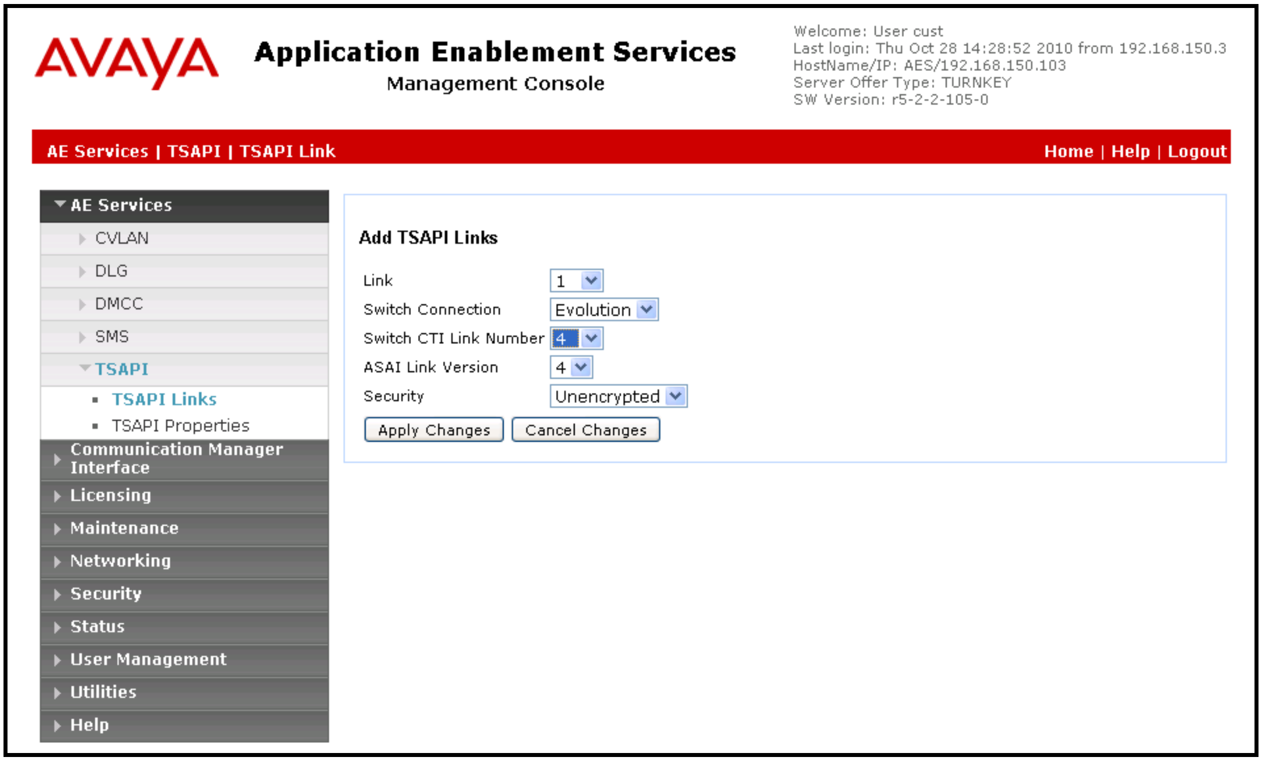

Figure 33: Add TSAPI Link Screen

Navigate to **User Management→User Admin→Add User**. The "CT User" field for this user must be set to "Yes". In this case, the Application Enablement Services user is the Voxlog Professional application, which uses Application Enablement Services to monitor stations and initiate switching operations. The "User Id" and "User Password" must be the same as what will be configured for Voxtronic Voxlog Professional in **Section 7**.

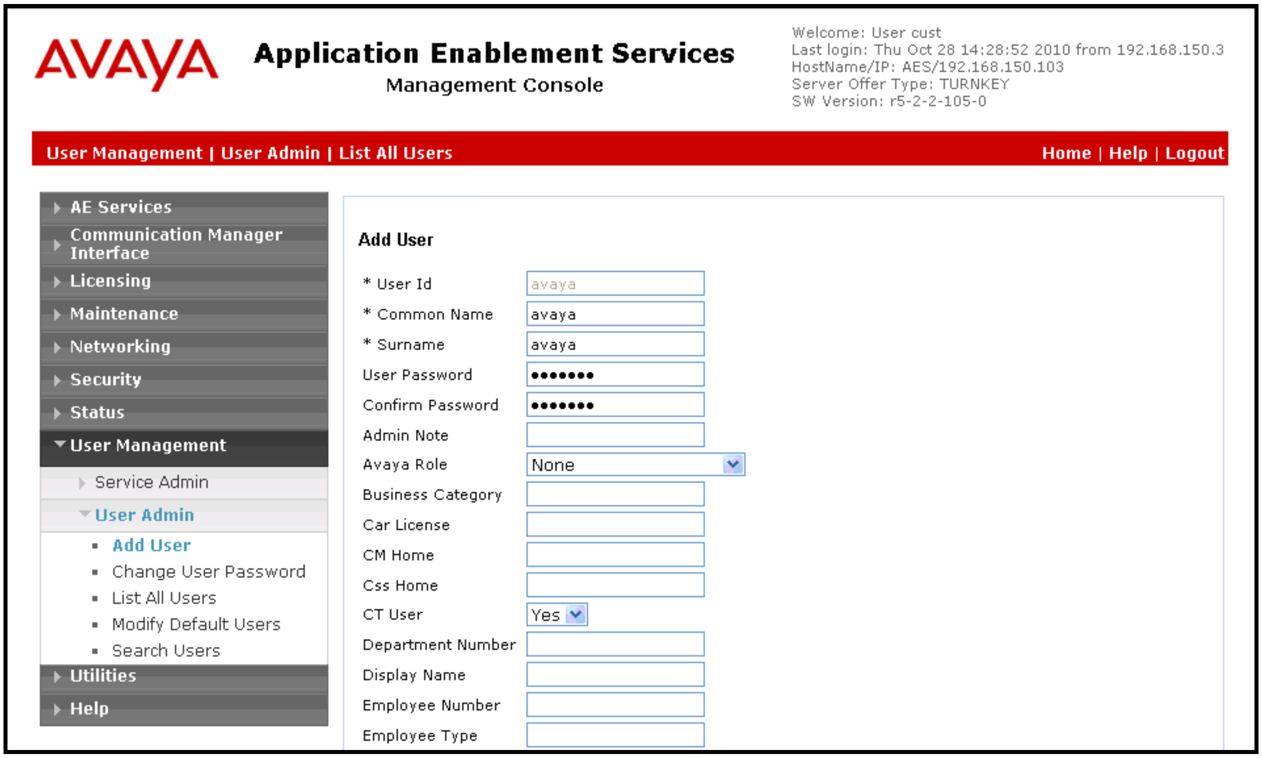

Figure 34: Add User Screen

Navigate to **Security → Security Database → CTI Users→ List All Users**, and then click "Edit User" for the newly added user "avaya", (not shown). Enable "Unrestricted Access" and click "Apply Changes".

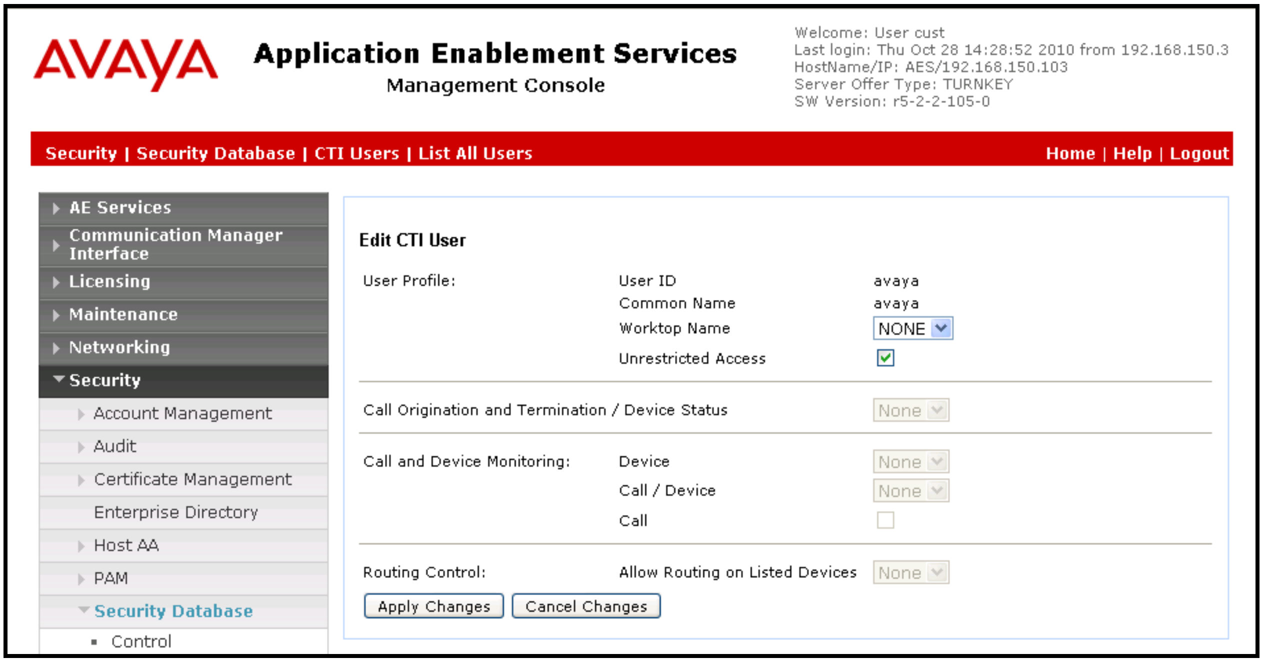

Figure 35: Edit CTI User Screen

Navigate to **Networking → Ports** and configure the DMCC Server Ports as shown in the following table.

Parameter	Usage
Unencrypted Port	Set this port to 4721.

Table 16: Avaya Aura® Application Enablement Services Port Parameters

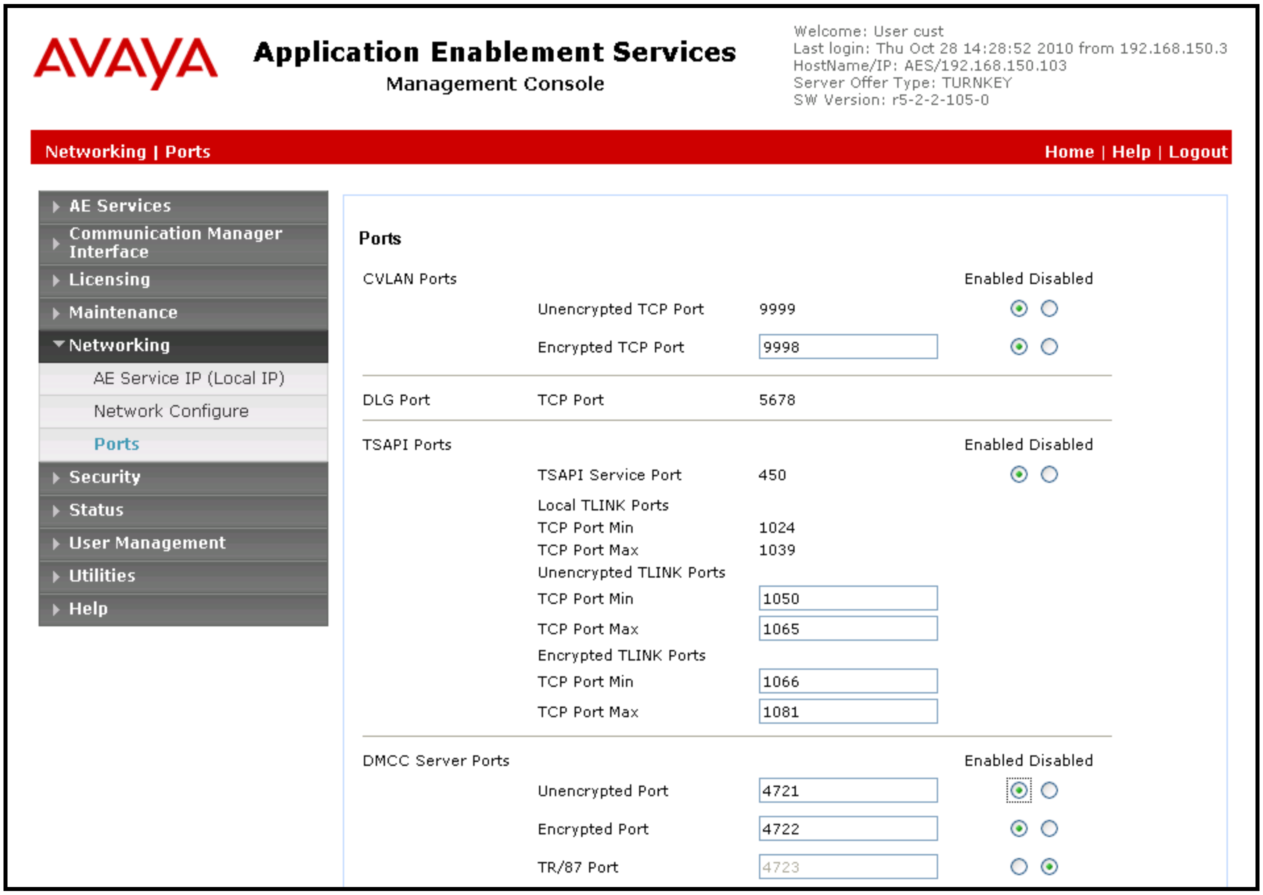

Figure 36: Application Enablement Services Port Configuration

7. Configure Voxtronic Voxlog Professional Server

The configuration values for the external interfaces used by Voxlog Professional are read from the **avayarps_config.xml** configuration file, located in the server's **C:\voxtronic\voxct_v3\bin\rps\avayarps** directory This XML property file contains a series of keys and associated values for various interface settings. This file can be edited with a text editor. The format of this file is shown in the following illustration.

```
<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<properties version="1.0">
    <entry key="key 1">value 1</entry>
    …
    <entry key="key n">value n</entry>
 </properties>
```

Figure 37: Voxtronic Voxlog Configuration File Format

The following table shows the various key names and associated values which were used to configure Voxlog to interoperate with Communication Manager.

Key	Value
recordingMethod	Selects the Client-side IP call recording method: SINGLE_STEP_CONFERENCING or SERVICE_OBSERVING. Set this value to "**SERVICE_OBSERVING**".
switchLinkName	Name of the communication link between the AE-Services and the Communication Manager**.** This must match the name used for the "Switch Connection" in **Figure 29**.
CMServerIpAddr	IP-Address of the Communication Manager Server (CLAN/PROCR/Gatekeeper). Set this to the IP address of the Communication Manager PROCR interface.
DMCCServerIpAddr	IP-Address off the AES DMCC-Server**.** Set this to the IP address of the Avaya Application Enablement Server.
DMCCServerPort	Port number of the AES DMCC-Server (service). Set this value to 4721.
DMCCServerUserName	User name of the AES DMCC-Server (service).. Enter the user name configured in **Figure 34**.
DMCCServerPassword	Password of the AES DMCC-Server (service)**..** Enter the password configured in **Figure 34**.
voxlogServerIpAddr	IP-Address of the Voxlog Server. Set this to the IP address of the Voxlog Professional Server.
voxlogServerPort	Voxlog Professional Server Port Set this value to: 6814
voxlogRtpIpAddr	IP-Address of the local Media-Stream. The RTP stream is forwarded to this local IP address by the Communication Manager. Set this to the IP address of the Voxlog Professional Server.
voxlogRtpPortRange	Port range of the local Media-Streams in Voxlog Professional. The

	RTP Media Stream ports are assigned continuously from this port number. Set this value to 6000.
recordingDevices	Comma-separated list of Virtual CTI Station (Recording-Device) extensions. The extensions are provisioned with the CM in **Figure 13**.
commonRDPassword	Common password of the Virtual CTI stations (Recording-Devices). Enter the common “Security Code” value assigned to all of the Virtual CTI stations in **Figure 13**.
recorderCodecs	Comma-separated list of codecs used by Voxlog Media Processor. Specifies the allowed Media Stream (RTP) Payload Types. Set this value to “g711A, g711U”.
recorderPacketSize	Size of the configured Media Stream (RTP) packets in milliseconds. Set this value to 20.
serviceObservingFAC	Feature Access Code (FAC) of Service Observing. Enter the value for the “Service Observing No Talk Access Code” configured in **Figure 7**.
sessionDurationTimer	Avaya AES Session duration time in seconds. Set this value to 60.
sessionCleanupDelay	Avaya AES Session cleanup delay in seconds. Set this value to 300.
applicationDescription	Avaya application description. Set this to: "Voxlog Professional - Client-side IP Call Recording"
ownExternalNumber	External Telephone Number of the location where the Voxlog Professional is running.

Table 17: Voxtronic Voxlog Professional Configuration Values

The following **avayarps_config.xml** configuration was used for the compliance test:

<?xml version="1.0" encoding="UTF-8" standalone="no" ?>
<!DOCTYPE properties *(View Source for full doctype...)*>
\- <properties version="**1.0**">
<comment>Config file for Avaya DMCC</comment>
\- <!--
```
 1.  Client-side Ip-call recording method:
             SINGLE_STEP_CONFERENCING |
             SERVICE_OBSERVING
```
-->
<entry key="**recordingMethod**">SERVICE_OBSERVING</entry>
\- <!--
```
 2.  Name of the communication link between AES and CM
             e.g.: S8500
                   Evolution
```
-->
<entry key="**switchLinkName**">Evolution</entry>

```
- <!--
 3.  CM-Server: IP-Address
              e.g.: 10.64.120.12
                    192.168.150.126
-->
<entry key="CMServerIpAddr">192.168.150.126</entry>
- <!--
 4.  AES DMCC-Server: IP-Address (See: cmapi.server_ip)
              e.g.: 10.64.120.15
                    192.168.150.103
-->
<entry key="DMCCServerIpAddr">192.168.150.103</entry>
- <!--
 5.  AES DMCC-Server: Port:      (See: cmapi.server_port)
              Unsecure connection: 4721,
              Secure   connection: 4722
-->
<entry key="DMCCServerPort">4721</entry>
- <!--
 6.  AES DMCC-Server: User name  (See: cmapi.username)
              e.g.: aessim
                    avaya
-->
<entry key="DMCCServerUserName">avaya</entry>
- <!--
 7.  AES DMCC-Server: Password   (See: cmapi.password)
              e.g.: AESsim123#
                    ou812
-->
<entry key="DMCCServerPassword">ou812</entry>
- <!--
 8.  Voxlog Professional Server: IP-Addresss
              192.168.1.4
              192.168.150.13
-->
<entry key="voxlogServerIpAddr">192.168.150.13</entry>
- <!--
 9.  Voxlog Professional Server: Port
              6814
-->
<entry key="voxlogServerPort">6814</entry>
- <!--
 10. Media-Stream Recorder: IP-Address (MediaProcessor of Voxlog
Professional)
              e.g.: 10.10.101.215
                    192.168.150.13
-->
<entry key="voxlogRtpIpAddr">192.168.150.13</entry>
- <!--
 11. Media-Stream Recorder: Port-Range
```

```
            e.g.: 6000
-->
<entry key="voxlogRtpPortRange">6000</entry>
- <!--
 12. Provisioned comma-separated list of Recording-Device (CTI) extensions.
            e.g.: 46001, 46002, 46003, 46004
                  11401, 11402, 11403, 11404
-->
<entry key="recordingDevices">11401, 11402, 11403, 11404</entry>
- <!--
 13. Common password of all Recording-Devices
            e.g.: 1234
-->
<entry key="commonRDPassword">1234</entry>
- <!--
 14. Comma-separated list of codecs in MediaProcessor. Allowed: g711A, g711U,
g723
            e.g.: g711A, g711U
-->
<entry key="recorderCodecs">g711A, g711U</entry>
- <!--
 15. Size of RTP media packets in milliseconds (For RTP PayloadType:  A-law,
u-law, G.723).
            e.g.: 20
-->
<entry key="recorderPacketSize">20</entry>
- <!--
 16. Provisioned Feature Access Code (FAC) of Service Observing:
            e.g.: 'Service Observing Listen Only' ==> *220
                  'Service Observing Listen/Talk' ==> *221
                  'Service Observing No Talk'     ==> *222
-->
<entry key="serviceObservingFAC">*222</entry>
- <!--
 17. Avaya AES Session duration timer in seconds
            Allowed: 5..7200
            60
-->
<entry key="sessionDurationTimer">60</entry>
- <!--
 18. Avaya AES Session cleanup delay in seconds
            Allowed: 0..7200
            300
-->
<entry key="sessionCleanupDelay">300</entry>
- <!--
 19. Avaya application description
            "Voxlog Professional - Client-side IP Call Recording"
-->
```

<entry key="**applicationDescription**">"Voxlog Professional - Client-side IP Call Recording"</entry>
- <!--
 20. External Telephone Number of the location running Voxlog Professional
            e.g.: 90739887
-->
<entry key="**ownExternalNumber**">"90739887"</entry>
</properties>

8. General Test Approach and Test Results

The compliance testing between Voxtronic Voxlog Professional and Communication Manager was performed manually. The tests were all functional in nature, and no performance testing was done. The test method employed can be described as follows:

- Avaya Aura® Communication Manager was configured to support various local IP telephones, as well as a networked PBX connection and a PSTN connection.
- An E1 PSTN interface was attached to Avaya Aura® Communication Manager.
- The Voxtronic Voxlog Professional was configured to monitor various telephones attached to Avaya Aura® Communication Manager.
- The major Voxtronic Voxlog Professional features and functions were verified using the above-mentioned local and external telephones, including the ability to record calls made to and from
  - Locally attached IP and digital telephones
  - Telephones attached to the PSTN via E1 trunk.
  - Telephones attached to a networked PBX via QSIG trunk.

The tests performed are shown is **Section 1.1**. All tests performed produced the expected result.

9. Verification Steps

The correct installation and configuration of Voxtronic Voxlog Professional voice recorder can be verified by performing the following steps using the SAT terminal from PBX 1.

- Use the “status aesvcs cti-link” command to verify that the TSAPI link allocated in **Figure 10** is “established”.

```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

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

1              no                     down          0        0
2              no                     down          0        0
3              no                     down          0        0
4     4        no    AES              established   15       15
```

Figure 38: Status Aesvcs Cti-link Screen

- Login to Avaya Aura® Application Enablement Services, and navigate to the **AE Services** screen. Verify that the DMCC and TSAPI Services are licensed, ONLINE, and Running.

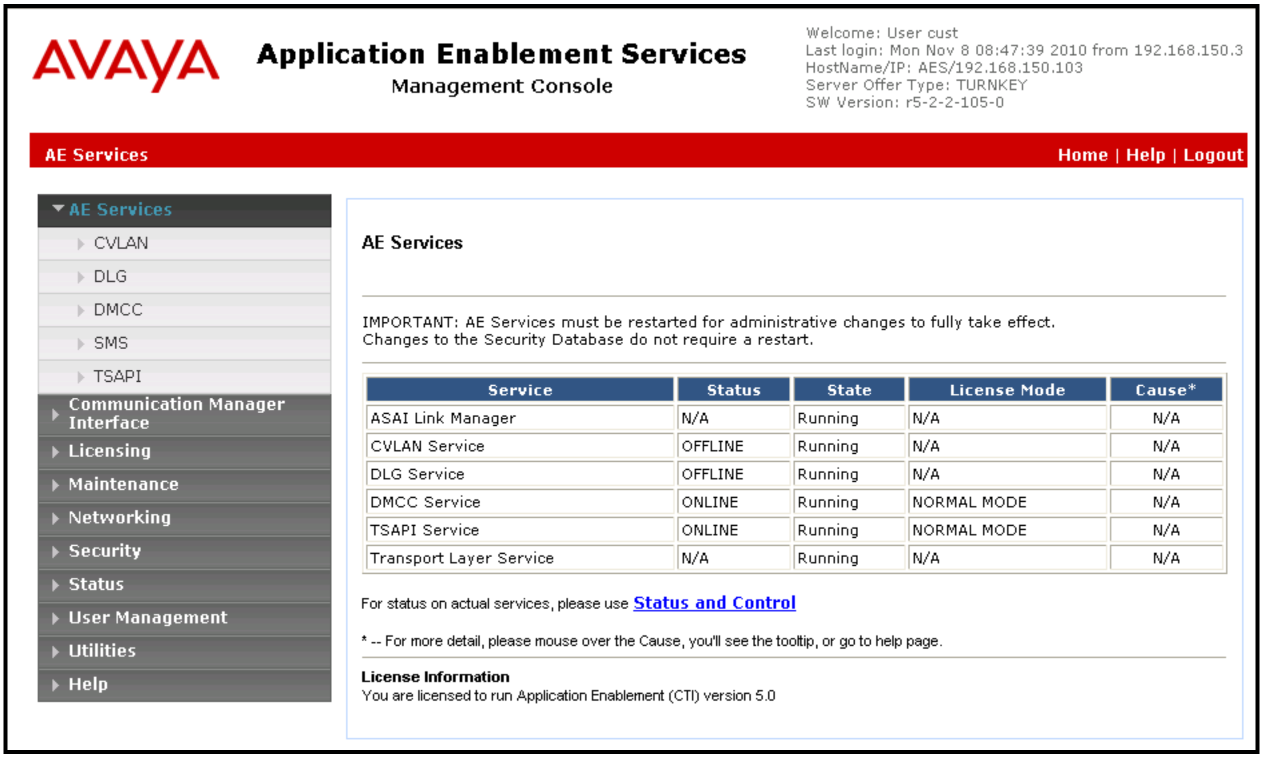

Figure 39: Application Enablement Services AE Services Screen

- Navigate to **Status → Status and Control -> Switch Conn Summary,** select the PBX 1, and click "Switch Connection Details". Verify that the connection state is "Online" and "Talking".

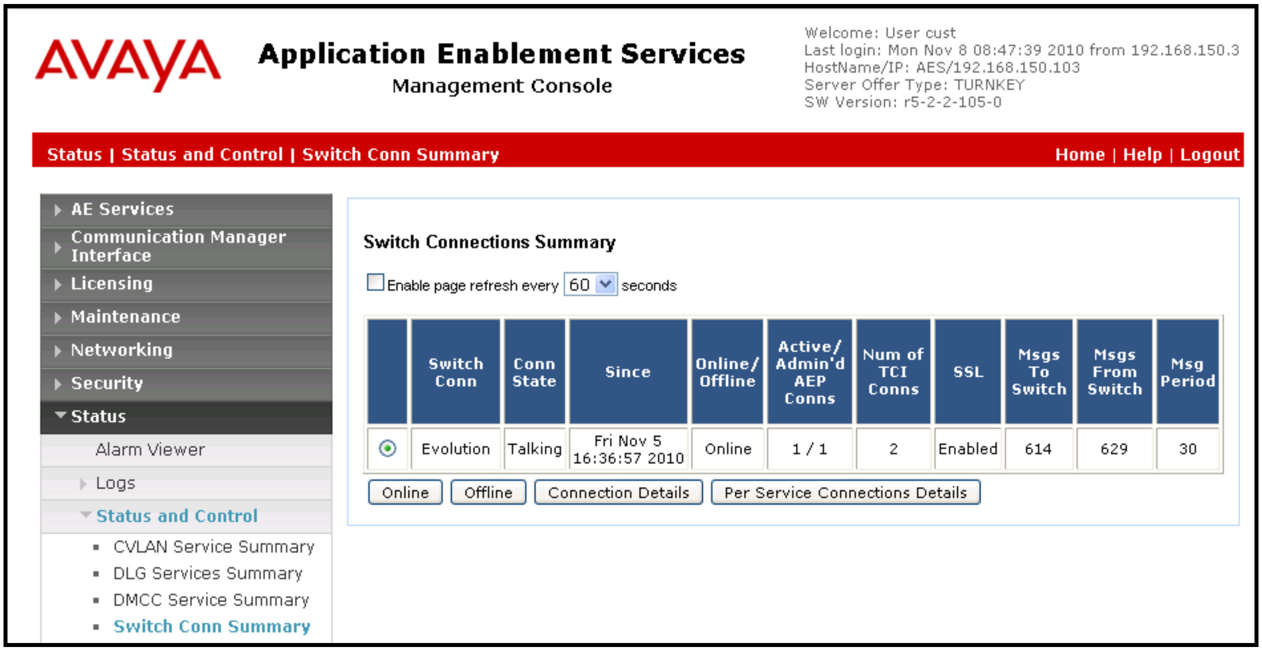

Figure 40: Application Enablement Services Switch Connection Details Screen

- Navigate to **Status → Status and Control -> TSAPI Service Summary** and click "Details" for "TSAPI Service". Verify that the TSAPI service for PBX 1 is "Online" and "Talking".

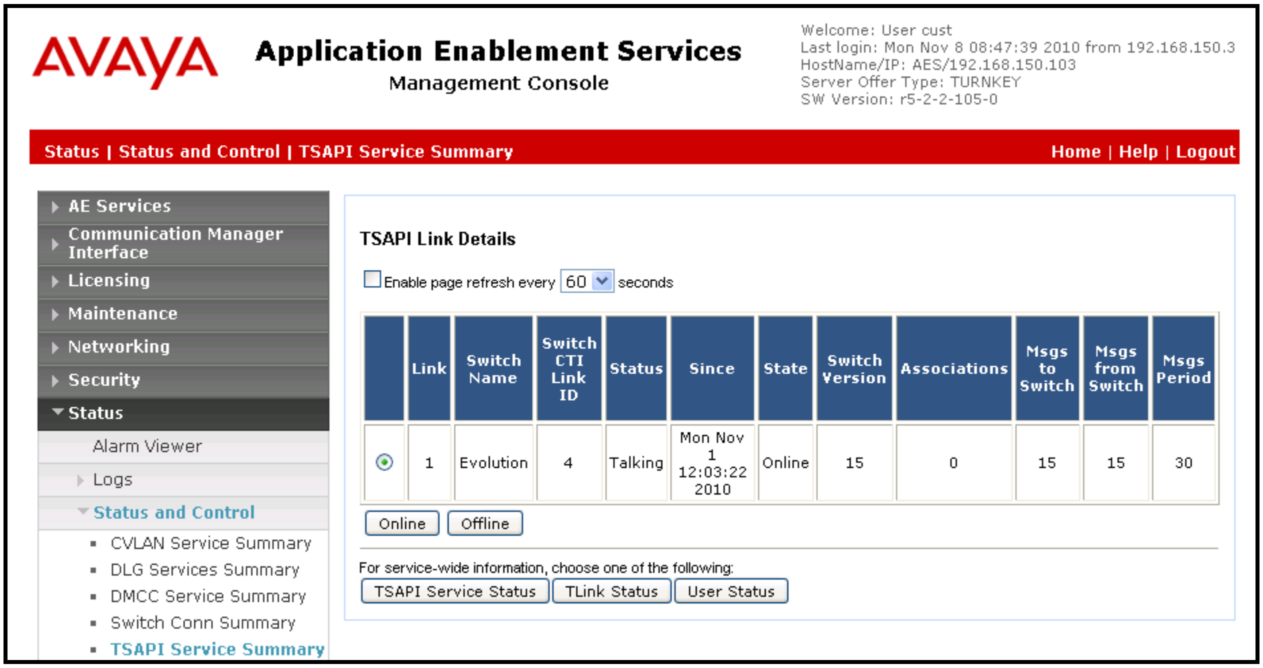

Figure 41: TSAPI Link Details Screen

- Navigate to **Status → Status and Control → DMCC Service Summary** and click "Service Summary". Verify that the Voxtronic Voxlog Professional has established a session.

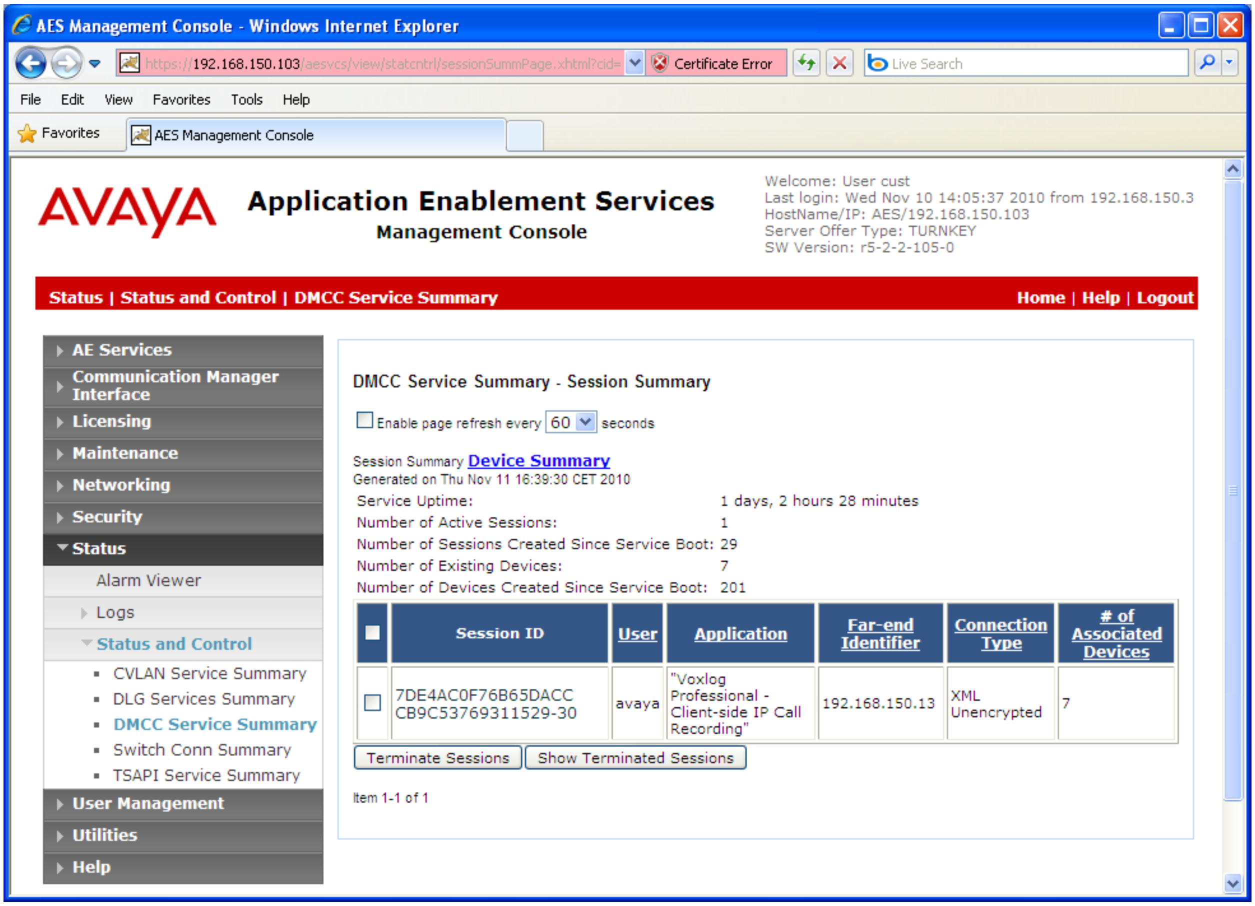

Figure 42: DMCC Service Summary Screen

- Navigate to **Status → Status and Control → DMCC Service Summary** and click "Device Summary". Verify that the Voxtronic Voxlog Professional has registered each of the CTI stations.

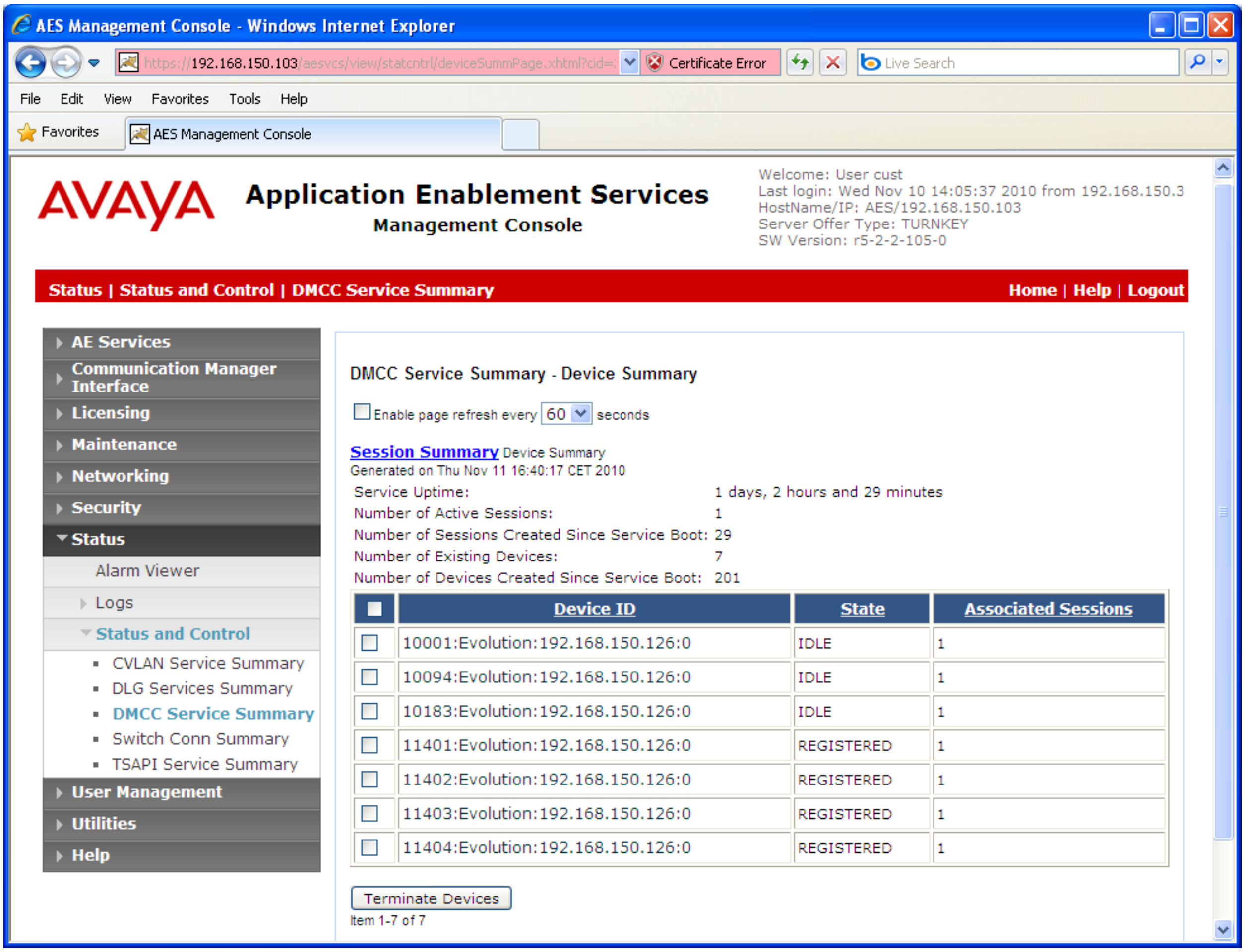

Figure 43: DMCC Device Summary Screen

10. Conclusion

These Application Notes describe the compliance testing of the Voxtronic Voxlog Professional voice recorder with Avaya Aura® Communication Manager. Silent monitoring via the Service Observe recording method offered by the Voxlog Professional was tested. A detailed description of the configuration required for both the Avaya and the Voxtronic equipment is documented within these Application Notes. The Voxlog Professional passed all of the tests performed, which included both functional and robustness tests.

11. References

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

[2] *Avaya Aura™ Communication Manager Feature Description and Implementation,* May 2009, Issue 7, Document Number 555-245-205.

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

[4] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server*, May 2009, Document Number 03-602508

[5] Voxlog User Manual Web Interface V.3.2 1. - *VLX3 - Benutzerhandbuch 3.2.2.5.En.doc*

[6] Voxlog Technical Manual - *VLX3 - Technisches Handbuch V7_EN.doc*

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.