

Avaya Solution & Interoperability Test Lab

Application Notes for configuring Enghouse Presence Suite R11.1 to interoperate with Avaya Aura® Communication Manager R8.0.1 and Avaya Aura® Application Enablement Services R8.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for Enghouse Presence Suite to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services. Presence Suite is a multi-channel contact management suite which handles voice, text chat, email and web contact and call recording mechanisms. Presence Suite integrates with the Avaya solution by using the Telephony Services Application Programmer Interface (TSAPI) provided by Avaya Aura® Application Enablement Services to monitor and control agent stations, and handle routing of external calls and uses single step conferencing implemented via DMCC over TSAPI to record calls.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

Introduction

These Application Notes describe the compliance tested configuration of Enghouse Presence Suite R11.1 to interoperate with Avaya Aura® Communication Manager R8.0.1 using Avaya Aura® Application Enablement Services R8.0. Presence Suite is a multi-channel contact management suite able to handle voice, e-mail, call recording and web chat contact mechanisms. The Telephony Services Application Programmer Interface (TSAPI) provided by Avaya Aura® Application Enablement Services is used to monitor and control agent stations, generate phantom calls for non-voice contacts and handle routing of external calls. The call recording module uses Avaya Aura® Communication Manager's Single Step Conferencing (SSC) feature via the Device, Media, and Call Control (DMCC) service provided by the Avaya Aura® Application Enablement Services (AES) to capture the audio and call details for recording agent calls.

Device Media Call Control (DMCC) works by allowing software vendors to create soft phones, in memory on a recording server, and use them to monitor and record other phones. This is purely a software solution and does not require telephony boards or any wiring beyond a typical network infrastructure.

Presence Suite consists of a number of modules. The following modules were tested.

- Presence Voice Outbound
- Presence Voice Inbound
- Presence Mail Interactions
- Presence Web Interactions
- Presence Recording

Upon starting the Presence Server application, the application automatically queries Avaya Aura® Application Enablement Services for device status and requests monitoring. The Presence Server specifies where to route each call and hence how to handle the calls, based on agent status information that the application tracks from CTI device query results and event reports received from Avaya Aura® Application Enablement Services.

General Test Approach and Test Results

Testing included validating the correct operation of typical contact center functions including, inbound and outbound service calls. Functionality testing included basic telephony operations such as answer, hold/retrieve, transfer, and conference. This was carried out for the inbound and outbound service calls. Email, web call back, web chat and call recording were also tested. Additional features such as call capturing, direct agent transfer and malicious calls were tested. The serviceability test cases were performed manually by busying out and releasing the CTI link and by disconnecting and reconnecting LAN cables.

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 this DevConnect Application Note 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 Presence Suite did not include use of any specific encryption features as requested by Enghouse.

2.1 Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on verifying Presence Suite handling of TSAPI messages in the areas of routing, call control and event notification. The serviceability testing focused on verifying the Presence Suite ability to recover from adverse conditions, such as stopping the TSAPI Service, taking the CTI link offline and disconnecting the Ethernet cable from all the devices in the solution.

The following modules were tested.

- Presence Voice Outbound
- Presence Voice Inbound
- Presence Mail Interactions
- Presence Web Interactions
- Presence Recording

Calls were places to a VDN to test inbound calls, outbound calls were initiated by the Presence Suite, both email and web chat were tested using phantom calls to route calls to the agent. Call recording was tested by repeating the inbound and outbound calls specifically for call recording.

2.2 Test Results

All test cases passed successfully.

2.3 Support

Technical support can be obtained for Enghouse Presence Suite as follows:

Email: Presence.Support@enghouse.comWebsite: https://www.enghouseinteractive.es/en

• Phone: +34 93 10 10 300

Reference Configuration

Figure 1 shows the network topology during interoperability testing. A Communication Manager with an Avaya G450 Media Gateway and Media Server was used as the hosting PBX. Presence Suite, including Presence Agent PC's, are connected to the LAN and controls the Avaya H323 and SIP IP telephones via Application Enablement Services using TSAPI and Presence Recording using DMCC Single Step Conference.

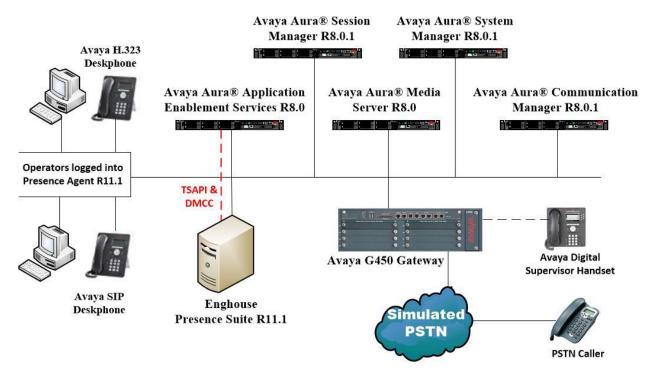


Figure 1: Avaya Aura® Communication Manager R8.0.1 and Aura® Application Enablement Services R8.0 with Enghouse Presence Suite R11.1 configuration

Equipment and Software Validated

All the hardware and associated software used in the compliance testing is listed below.

Avaya Equipment	Software / Firmware Version
Avaya Aura® System Manager running on a virtual server	System Manager 8.0.1.1 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.11.039340 Service Pack 1
Avaya Aura® Session Manager running on a virtual server	Session Manager R8.0.1 Build No. – 8.0.1.1.801103
Avaya Aura® Communication Manager running on a virtual server	R8.0.1.1.0 – FP1SP1 R018x.00.0.822.0 Update ID 00.0.822.0-25183
Avaya Aura® Application Enablement Services running on Virtual Server	R8.0 Build No – 8.0.0.0.6-0
Avaya Media Gateway G450	41.10.1 /1
Avaya Aura® Media Server	Appliance Version R8.0.0.6 Media Server 8.0.0.150 Element Manager 8.0.0.150
Avaya 96x1 SIP Deskphone	7.1.2.0.14
Avaya J179 H323 Deskphone	6.7.002U
Avaya 9408 Digital Deskphone	V2.0
Enghouse Equipment	Software / Firmware Version
Enghouse Presence Suite running on Windows Server 2016 Server	R11.1
Enghouse Presence Client running on Windows 10	R11.1

Configure Avaya Aura® Communication Manager

The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration and verification operations illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). The configuration operations described in this section can be summarized as follows:

- Verify System Features
- Administer SIT Treatment for Call Classification
- Administer Hunt Groups, Vectors and VDN's
- Administer Class of Restriction
- Administer Agent Logins
- Administer Agent Stations
- Administer Phantom Stations
- Note procr IP Address for AES Connectivity
- Configure Transport link for AES Connectivity
- Configure CTI Link for TSAPI Service
- Configure H323 Stations for Single Step Conference
- Configure SIP Stations for Single Step Conference
- Configure Virtual Stations

5.1 Verify System Features

Use the **display system-parameters customer-options** command to verify that Communication Manager has permissions for features illustrated in these Application Notes. On **Page 4**, ensure that **Computer Telephony Adjunct Links?** is set to **y** and **Answer Supervision by Call Classifier?** is set to **y** as shown below.

```
display system-parameters customer-options
                                                                     4 of 12
                                                              Page
                              OPTIONAL FEATURES
   Abbreviated Dialing Enhanced List? y
                                                Audible Message Waiting? y
                                                Authorization Codes? y
       Access Security Gateway (ASG)? y
       Analog Trunk Incoming Call ID? y
                                                             CAS Branch? n
A/D Grp/Sys List Dialing Start at 01? y
                                                               CAS Main? n
Answer Supervision by Call Classifier? y
                                                      Change COR by FAC? n
                                ARS? y Computer Telephony Adjunct Links? y
                ARS/AAR Partitioning? y Cvg Of Calls Redirected Off-net? y
         ARS/AAR Dialing without FAC? y
                                                            DCS (Basic)? y
                                                      DCS Call Coverage? y
         ASAI Link Core Capabilities? y
         ASAI Link Plus Capabilities? y
                                                      DCS with Rerouting? y
     Async. Transfer Mode (ATM) PNC? n
 Async. Transfer Mode (ATM) Trunking? n Digital Loss Plan Modification? y
             ATM WAN Spare Processor? n
                                                                DS1 MSP? v
                              ATMS? y
                                                  DS1 Echo Cancellation? y
                 Attendant Vectoring? y
```

On **Page 7**, verify the following customer options are set to **y** as shown below.

- ACD? to v
- Vectoring (Basic)? to y
- Expert Agent Selection (EAS)? to y

```
display system-parameters customer-options
                                                                     7 of 12
                                                              Page
                        CALL CENTER OPTIONAL FEATURES
                         Call Center Release: 8.0
                               ACD? v
                                                              Reason Codes? v
                      BCMS (Basic)? y
                                                   Service Level Maximizer? n
                                        Service Observing (Basic)? y
         BCMS/VuStats Service Level? y
                                        Service Observing (Remote/By FAC)? y
  BSR Local Treatment for IP & ISDN? y
                                                  Service Observing (VDNs)? y
                 Business Advocate? n
                   Call Work Codes? y
                                                                 Timed ACW? y
      DTMF Feedback Signals For VRU? y
                                                         Vectoring (Basic)? y
                                                     Vectoring (Prompting)? y
                  Dynamic Advocate? n
      Expert Agent Selection (EAS)? y
                                                 Vectoring (G3V4 Enhanced)? y
                           EAS-PHD? y
                                                  Vectoring (3.0 Enhanced)? y
                  Forced ACD Calls? n Vectoring (ANI/II-Digits Routing)? y
                                         Vectoring (G3V4 Advanced Routing)? y
              Least Occupied Agent? y
         Lookahead Interflow (LAI)? y
                                                         Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? y
                                                      Vectoring (Holidays)? y
  PASTE (Display PBX Data on Phone)? y
                                                     Vectoring (Variables)? y
```

Use the command display system-parameters features and on Page 1, verify that the Trunkto-Trunk Tranfer option is set to all as shown below.

```
display system-parameters features
                                                                Page
                                                                       1 of 19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
   Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
       Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
              Music (or Silence) on Transferred Trunk Calls? no
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
            Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? N
```

On Page 10 ensure that Station Tone Forward Disconnect is set to silence as shown below.

```
display system-parameters features
                                                               Page 10 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS
                                           Update Transferred Ring Pattern? n
                Pull Transfer: n
         Outpulse Without Tone? y
                                             Wait Answer Supervision Timer? n
        Misoperation Alerting? n
                                               Repetitive Call Waiting Tone? n
   Allow Conference via Flash? y
                                  Network Feedback During Tone Detection? y
 Vector Disconnect Timer (min):
   Hear Zip Tone Following VOA? y System Updates Time On Station Displays? n
                    Station Tone Forward Disconnect: silence
                           Level Of Tone Detection: precise
          Charge Display Update Frequency (seconds): 30
                           Date Format on Terminals: dd/mm/yy
                        Onhook Dialing on Terminals? n
               Edit Dialing on 96xx H.323 Terminals? n
                  Allow Crisis Alert Across Tenants? n
                   Send DTMF Over Telecommuter Link? y
ITALIAN DCS PROTOCOL
  Italian Protocol Enabled? n
```

On **Page 11**, verify that the **Expert Agent Selection (EAS) Enabled?** option is set to **y** as shown below.

```
display system-parameters features
                                                              Page 11 of 19
                       FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
 EAS
        Expert Agent Selection (EAS) Enabled? y
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension:
                                                                Delay:
   Message Waiting Lamp Indicates Status For: station
                          Work Mode On Login: aux
 VECTORING
                   Converse First Data Delay: 0
                                                     Second Data Delay: 2
              Converse Signaling Tone (msec): 100
                                                        Pause (msec): 70
                   Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? n
                           BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
 SERVICE OBSERVING
             Service Observing: Warning Tone? y
                                                   or Conference Tone? n
   Allowed with Exclusion: Service Observing? n
                                                    SSC? n
            Allow Two Observers in Same Call? y
```

On Page 12 ensure that ACW Agents Considered Idle is set to y.

```
display system-parameters features
                                                                Page 12 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS
 AGENT AND CALL SELECTION
                         MIA Across Splits or Skills? n
                          ACW Agents Considered Idle? y
                    AUX Agents Considered Idle (MIA)? n
                      AUX Agent Remains in LOA Queue? n
                          Call Selection Measurement: current-wait-time
    Service Level Supervisor Call Selection Override? n
                                 Auto Reserve Agents: none
       Block Hang-up by Logged-in Auto-Answer Agents? n
 CALL MANAGEMENT SYSTEM
    REPORTING ADJUNCT RELEASE (determines protocol used by appl link)
                                      CMS (appl mis):
                                  AAPC/IQ (appl ccr):
                               BCMS/VuStats LoginIDs? y
                   BCMS/VuStats Measurement Interval: hour
           BCMS/VuStats Abandon Call Timer (seconds):
                     Validate BCMS/VuStats Login IDs? n
                            Clear VuStats Shift Data: on-login
                 Remove Inactive BCMS/VuStats Agents? n
```

On **Page 13**, verify that **Call Classification After Answer Supervision** option is set to **y** as shown below.

```
display system-parameters features
                                                                Page 13 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER MISCELLANEOUS
          Callr-info Display Timer (sec): 10
                         Clear Callr-info: next-call
        Allow Ringer-off with Auto-Answer? n
    Reporting for PC Non-Predictive Calls? n
            Agent/Caller Disconnect Tones? n
          Interruptible Aux Notification Timer (sec): 3
             Zip Tone Burst for Callmaster Endpoints: double
 ASAI
                   Copy ASAI UUI During Conference/Transfer? n
               Call Classification After Answer Supervision? y
                                         Send UCID to ASAI? y
                 For ASAI Send DTMF Tone to Call Originator? y
         Send Connect Event to ASAI For Announcement Answer? n
 Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call? n
```

5.2 Administer Special Information Tones Treatment for Call Classification

This form is used to specify the treatment of Special Information Tones (SIT) used for outbound call management type calls with USA tone characteristics. Enter the **change sit-treatment** command. Set the **Pause Duration** to **0.8** and **Talk Duration** to **3.0**. Please note this may vary depending on the country where the PBX is installed.

```
change sit-treatment

SIT TREATMENT FOR CALL CLASSIFICATION

SIT Ineffective Other: dropped

SIT Intercept: dropped

SIT No Circuit: dropped

SIT Reorder: dropped

SIT Vacant Code: dropped

SIT Unknown: dropped

AMD Treatment: dropped

Pause Duration (seconds): 0.8

Talk Duration (seconds): 3.0
```

5.3 Administer Hunt Groups, Call Vectors and Vector Directory Numbers

In order for calls to be routed to agents, Hunt Groups (skills), Vectors, and Vector Directory Numbers (VDN) must be configured. Typically a unique VDN and Hunt Group would be setup for each of the following modules which were tested. However for these Application Notes the configuration shows the setup of only the Presence Inbound VDN, Vector and Hunt Group as an example.

- Presence Voice Outbound
- Presence Voice Inbound
- Presence Mail Interactions
- Presence Web Interactions
- Presence Recording

5.3.1 Hunt Groups

Enter the **add hunt-group n** command where **n** in the example below is **98**. On **Page 1** of the **hunt-group** form, assign a **Group Name** and **Group Extension** valid under the provisioned dial plan. Set the following options to **y** as shown below.

- ACD? to y
- Queue? to y
- Vector? to y

```
add hunt-group 98
                                                          Page
                                                                1 of
                                                                       4
                                 HUNT GROUP
           Group Number: 98
                                                         ACD? y
            Group Name: PresenceInbound
                                                       Queue? y
        Group Extension: 4808
                                                      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:
```

On **Page 2**, set the **Skill** field to **y** as shown below.

```
add hunt-group 98

Skill? y

AAS? n

Measured: none
Supervisor Extension:

Controlling Adjunct: none

Multiple Call Handling: none

Timed ACW Interval (sec): After Xfer or Held Call Drops? n
```

Repeat the above steps to create a hunt groups for the outbound service, web chat and email.

5.3.2 Vectors

Enter the **change vector n** command, where **n** is the vector number. The adjunct routing link enables Presence Suite to specify the destination of a call. The **adjunct routing link** number is defined by the position of the AESVCS link on page three of the ip-services (not shown), in this case Server ID **1.**

The call is then queued to the skill set out on the VDN in the 1st Skill field on the next page.

```
1 of
change vector 44
                                                                        Page
                                      CALL VECTOR
    Number: 44
                                Name: DevConnect Vector
Multimedia? y 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 adjunct routing link 1
02 wait-time 2 secs hearing
03 queue-to skill 1st pri m
04 wait-time 10 secs hearing
05 goto step 3 if
                  2 secs hearing ringback
                 10 secs hearing music
                      if unconditionally
06 stop
07
08
09
10
11
12
```

5.3.3 Vector Directory Numbers (VDN)

Enter the **add vdn n** command, where **n** is an available extension number. On **Page 1** assign a **Name** for the VDN and set the **Vector Number** to the relevant vector.

```
add vdn 4908
                                                                   1 of
                                                            Page
                           VECTOR DIRECTORY NUMBER
                            Extension: 4908
                                Name*: PresenceInbound
                          Destination: Vector Number
                                                            44
                  Attendant Vectoring? n
                 Meet-me Conferencing? n
                   Allow VDN Override? n
                                  COR: 1
                                  TN*: 1
                             Measured: none
                                               Report Adjunct Calls as ACD*? n
       VDN of Origin Annc. Extension*:
                           1st Skill*: 98
                           2nd Skill*:
                           3rd Skill*:
* Follows VDN Override Rules
```

5.4 Administer Class of Restriction

Enter the **change cor x** command where **x** corresponds to the Class of Restriction to be used for the agent login IDs in **Section 5.5**. On **Page 1**, set the **Direct Agent Calling** to **y**. This will allow agents to be called directly once they are logged in.

```
CLASS OF RESTRICTION

COR Number: 1
COR Description: DefaultCOR_PG

FRL: 7
Can Be Service Observed? y
Calling Party Restriction: none
Can Be A Service Observer? y
Called Party Restriction: none
Time of Day Chart: 1
Priority Queuing? n
Priority Queuing? n
Restriction Override: none
Restricted Call List? n

Access to MCT? y
Group II Category For MFC: 7
Send ANI for MFE? n
MF ANI Prefix:
Hear System Music on Hold? y
Can Use Directed Call Pickup? y
Can Use Directed Call Pickup? y
Group Controlled Restriction: inactive
```

5.5 Administer Agent Logins

Enter the **add agent-loginID n** command; where **n** is an available extension number. Enter a descriptive name for the agent in the **Name** field. Ensure the **COR** field is set to **1** which relates to the COR configured in **Section 5.4**. The **Auto Answer** field is set to **station** except for those logins that will be used for outbound services. In that case, the field will be set to **all**. Configure a password as required.

```
add agent-loginID 4405
                                                           Page 1 of
                                AGENT LOGINID
               Login ID: 4405
                                                               AAS? n
                                                             AUDIX? n
                   Name: PresenceAgent1
                     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: station
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**, assign a skill to the agent by entering the relevant hunt group number created in **Section 5.3.1** for **SN** and entering a skill level of **1** for **SL**. In this case, an agent is able to handle both inbound and outbound calls is created. Set the **Direct Agent Skill** to the Inbound hunt group **98**.

```
Change agent-loginID 4405

AGENT LOGINID

Direct Agent Skill: 98

Call Handling Preference: skill-level

SN RL SL

1: 98

1 16:
2: 96

1 17:
3: 97

1 18:
4:
19:
5: 20:
6:
7:
```

Repeat this task accordingly for any additional inbound or outbound agents required.

5.6 Administer Agent Stations

For each station that agents will log in to, enter the command **change station n**, where **n** is the station extension. On **Page 1** ensure that **IP SoftPhone** is set to **y** as shown below.

```
change station 4000
                                                                    Page 1 of 5
                                       STATION
                                       Lock Messages? n
Security Code: *
Coverage Path 1: 2
Coverage Path 2:
Extension: 4000
                                                                           BCC: 0
     Type: 9608
                                                                             TN: 1
     Port: S00000
                                                                            COR: 1
     Name: 4000, H323User
                                                                            COS: 1
                                         Hunt-to Station:
                                                                          Tests? n
STATION OPTIONS
                                             Time of Day Lock Table:
              Loss Group: 19 Personalized Ringing Pattern: 1
       Speakerphone: 2-way
Display Language: english
able GK Node Name:
                                                Message Lamp Ext: 4000
                                              Mute Button Enabled? y
                                                     Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal
                                                 Media Complex Ext:
   Survivable Trunk Dest? y
                                                        IP SoftPhone? y
                                                  IP Video Softphone? n
                                Short/Prefixed Registration Allowed: default
                                                 Customizable Labels? y
```

On **Page 4**, the following buttons must be assigned as shown below:

- aux-work Agent is logged in to the ACD but is not available to take a call.
- **manual-in** Agent is available to accept ACD calls.
- **after-call** Agent state after the ACD call is completed. The agent is not available.
- **release** State when the call is dropped.

```
change station 4000
                                                           Page
                                                                  4 of
                                    STATION
 SITE DATA
                                                      Headset? n
      Room:
      Jack:
                                                      Speaker? n
     Cable:
                                                     Mounting: d
     Floor:
                                                  Cord Length: 0
  Building:
                                                    Set Color:
ABBREVIATED DIALING
                             List2:
                                                       List3:
    List1:
BUTTON ASSIGNMENTS
                                       5: manual-in
1: call-appr
                                                             Grp:
2: call-appr
                                        6: after-call
                                                             Grp:
                                       7: release
 3: call-appr
             RC: Grp:
 4: aux-work
                                       8::
```

Note: The same changes should be made on a SIP station. Changes to SIP stations are made using System Manager as shown in **Section 5.12**.

5.7 Administer Phantom Stations

Presence Suite uses stations via AES to initiate calls on Communication Manager. These stations will be used to place calls to customers for outbound services as well as to place calls to agents in order to reserve an agent to handle the outbound call. Use the command **add station n**, enter a descriptive name for **Name**, the **Type** should be set to **6408D+** and enter **X** for the **Port**.

```
add station 28800
                                                             Page 1 of
                                    STATION
                                        Lock Messages? n
Extension: 28800
                                                                     BCC: 0
                                        Security Code:
    Type: 6408D+
                                                                       TN: 1
                                      Coverage Path 1:
                                                                      COR: 1
    Port: X
    Name: PresencePhantom
                                      Coverage Path 2:
                                                                      cos: 1
                                     Hunt-to Station:
STATION OPTIONS
            Time of Day Lock Table:
Loss Group: 2 Personalized Ringing Pattern: 1
Data Module? n Message Lamp But
           Speakerphone: 2-way
                                               Message Lamp Ext: 4850
                                           Mute Button Enabled? y
       Display Language: english
         Survivable COR: internal
                                             Media Complex Ext:
   Survivable Trunk Dest? y
                                               IP SoftPhone? n
                                             Remote Office Phone? n
                                                       IP Video? n
```

5.8 Note procr IP Address for Avaya Aura® Application Enablement Services Connectivity

Display the procr IP Address by using the command **display node-names ip** and noting the IP address for the **procr** and AES (aes80vmpg).

display node-name	mes ip		Page	1 of	2
		IP NODE NAMES			
Name	IP Address				
SM100	10.10.40.58				
aes80vmpg	10.10.40.56				
default	0.0.0.0				
g450	10.10.40.15				
procr	10.10.40.59				

5.9 Configure Transport Link for Avaya Aura® Application Enablement Services Connectivity

To administer the transport link to AES use the **change ip-services** command. On **Page 1** add an entry with the following values:

- **Service Type:** should be set to **AESVCS**.
- Enabled: set to y.
- Local Node: set to the node name assigned for the procr in Section 5.8.
- Local Port Retain the default value of 8765.

change ip-services				Page	1 of	4		
Service Type AESVCS	Enabled Y	Local Node procr	IP SERVICES Local Port 8765	Remote Node	Remote Port			

Go to **Page 4** of the **ip-services** form and enter the following values:

- AE Services Server: Name obtained from the AES server, in this case aes80vmpg.
- **Password:** Enter a password to be administered on the AES server.
- Enabled: Set to y.

Note: The password entered for **Password** field must match the password on the AES server in **Section 6.2**. The **AE Services Server** should match the administered name for the AES server, this is created as part of the AES installation, and can be obtained from the AES server by typing **uname –n** at the Linux command prompt.

change ip-services AE Services Administration					4 of	4
Server ID	AE Services Server	Password	Enabled	Status		
1: 2:	aes80vmpg	*****	У	idle		
3:						

5.10 Configure CTI Link for TSAPI Service

Add a CTI link using the **add cti-link n** command. Enter an available extension number in the **Extension** field. Enter **ADJ-IP** in the **Type** field, and a descriptive name in the **Name** field. Default values may be used in the remaining fields.

```
add cti-link 1

CTI Link: 1

Extension: 2002

Type: ADJ-IP

COR: 1

Name: aes80vmpg
```

5.11 Configure H323 Stations for Single Step Conference

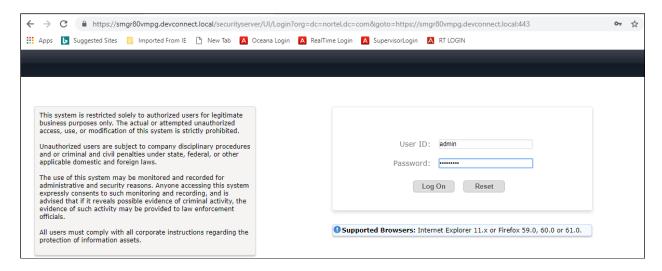
No extra specific changes were made during compliance testing for and H.323 stations in order for recording and single step conference to work. The screen below shows an example of a H.323 phone that was tested.

```
diaplay station 2000
                                                                    Page
                                                                           1 of
                                    STATION
                                      Lock Messages? n
Security Code: 1234
Extension: 2000
                                                                          BCC: 0
    Type: 9608
                                                                       TN: 1
    Port: S00101
                                      Coverage Path 1:
                                                                       COR: 1
    Name: H323 2000
                                      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: 2000
           Speakerphone: 2-way
                                             Mute Button Enabled? y
       Display Language: english
Survivable GK Node Name:
         Survivable COR: internal
                                               Media Complex Ext:
                                                    IP SoftPhone? y
  Survivable Trunk Dest? y
                                              IP Video Softphone? n
                             Short/Prefixed Registration Allowed: default
```

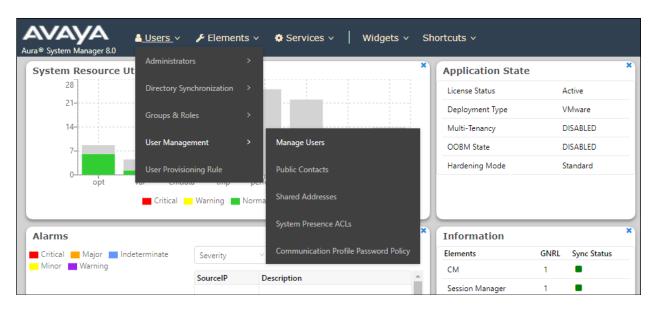
5.12 Configure SIP Stations for Single Step Conference

There should be no extra changes required for SIP stations to work using Single Step Conference. Changes of SIP phones on Communication Manager must be carried out from System Manager. Access the System Manager using a web browser by entering <a href="http://<FQDN">http://<FQDN >/SMGR, where <a href="http://<IP Address">FQDN is the fully qualified domain name of System Manager or <a href="http://<IP Address">http://<IP Address >/SMGR. Log in using appropriate credentials.

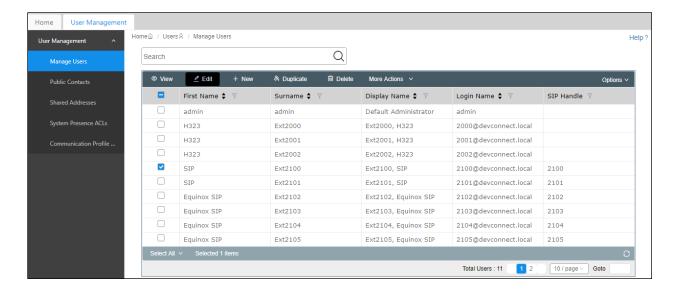
Note: The following shows changes a SIP extension and assumes that the SIP extension has been programmed correctly and is fully functioning.



From the home page click on Users \rightarrow User Management \rightarrow Manage Users as highlighted below.

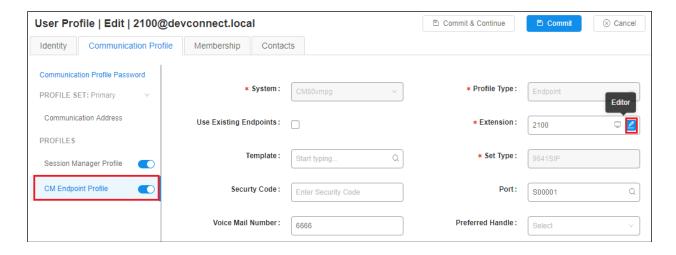


Select the station to be edited and click on **Edit**. The example below shows that SIP extension **2100** is selected.

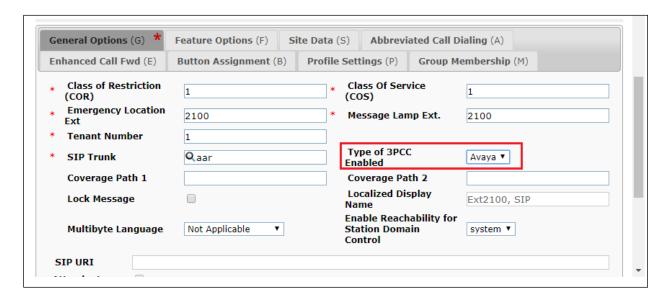


To set the password for the SIP extension click on **Communication Profile Password** in the left window and set the password in the main window (not shown here).

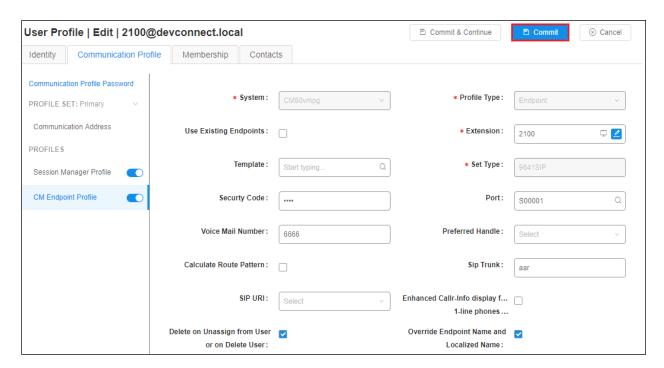
Click on the **CM Endpoint Profile** in the left window. Click on the **Editor** icon in the main window.



Ensure that **Type of 3PCC Enabled** is set to **Avaya**. Click on the **Feature Options** tab after that. Ensure that both the **Class of Restriction (COR)** and the **Class of Service (COS)** are set correctly. Scroll up or down and click on **Done** (not shown).



Click on Commit, as shown.



5.13 Configure Virtual Stations

Add virtual stations to allow Presence Suite record calls using Single Step Conference. Type **add station x** where x is the extension number of the station to be configured also note this extension number for configuration required in **Section 7.2.1**. Note the **Security Code** and ensure that **IP SoftPhone** is set to **y**.

add station 28903 Page 1 of 6 STATION Extension: 28903 Lock Messages? n BCC: 0 Lock Messages? n
Security Code: 1234 Type: 4624 TN: 1 Coverage Path 1: Port: S00101 COR: 1 cos: 1 Name: Recorder Coverage Path 2: Hunt-to Station: STATION OPTIONS Time of Day Lock Table: Loss Group: 19 Personalized Ringing Pattern: 1 Message Lamp Ext: 28903 Speakerphone: 2-way
Display Language: english Mute Button Enabled? y 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

Configure Avaya Aura® Application Enablement Services Server

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

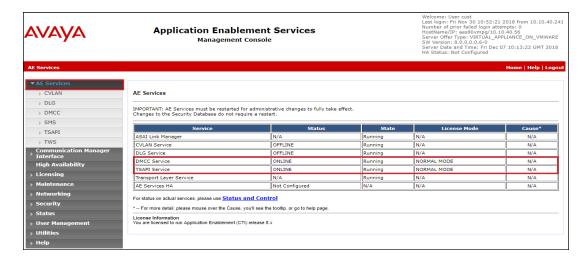
- Verify Licensing
- Create Switch Connection
- Administer TSAPI link
- Create CTI User
- Enable Unrestricted Access for CTI User
- Identify TLinks
- Configure Networking Ports

6.1 Verify Licensing

To access the maintenance console, enter **https://<ip-addr>** as the URL in an Internet browser, where <ip-addr> is the active IP address of AES. The login screen is displayed, log in with the appropriate credentials and then select the **Login** button.

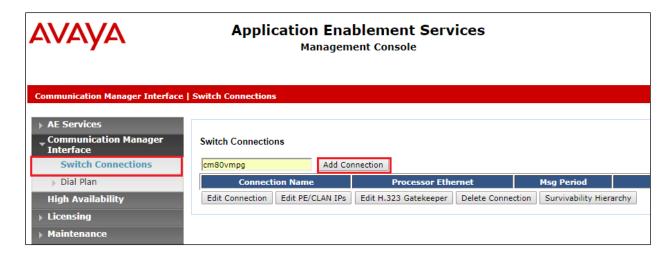


The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the **TSAPI Service** and **DMCC Service** are licensed by ensuring that the **License Mode** is showing **NORMAL MODE**.

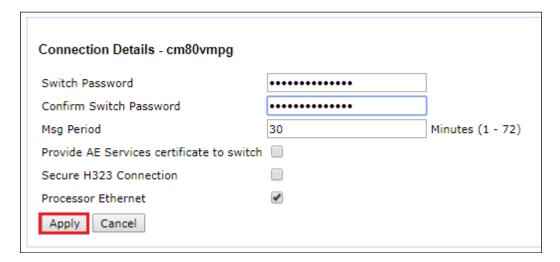


6.2 Create Switch Connection

From the AES Management Console navigate to **Communication Manager Interface > Switch Connections** to set up a switch connection. Enter in a name for the Switch Connection to be added and click the **Add Connection** button.



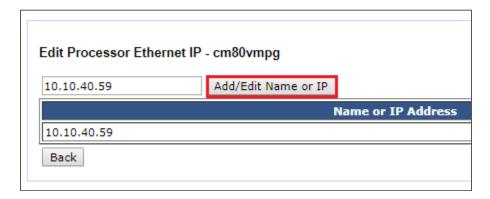
In the resulting screen enter the **Switch Password**, the Switch Password must be the same as that entered into Communication Manager AE Services Administration screen via the **change ipservices** command, described in **Section 5.9**. Default values may be accepted for the remaining fields. Click **Apply** to save changes.



From the **Switch Connections** screen, select the radio button for the recently added switch connection and select the **Edit CLAN IPs** button.

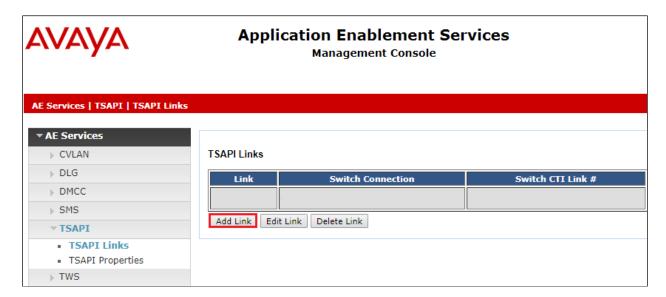


In the resulting screen, enter the IP address of the **procr** as shown in **Section 5.8** that will be used for the AES connection and select the **Add Name or IP** button.



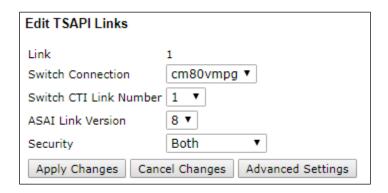
6.3 Administer TSAPI link

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

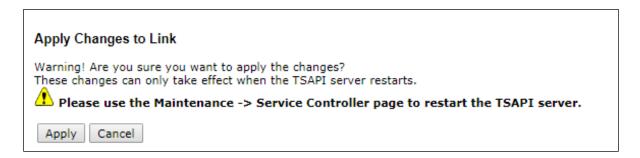


On the **Add TSAPI Links** screen, enter the following values:

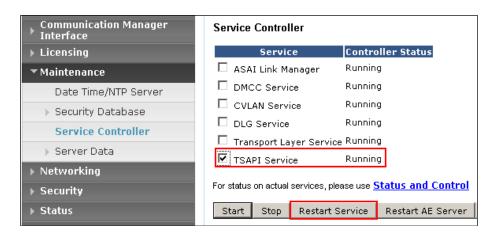
- **Link:** Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm80vmpg**, which has already been configured in **Section 6.2**, from the drop-down list.
- Switch CTI Link Number: Corresponding CTI link number configured in Section 5.10.
- **ASAI Link Version:** This can be left at the default value of **8**.
- **Security:** This can be left at the default value. The value **both** was used in this test.
- Once completed, select **Apply Changes**.



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



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

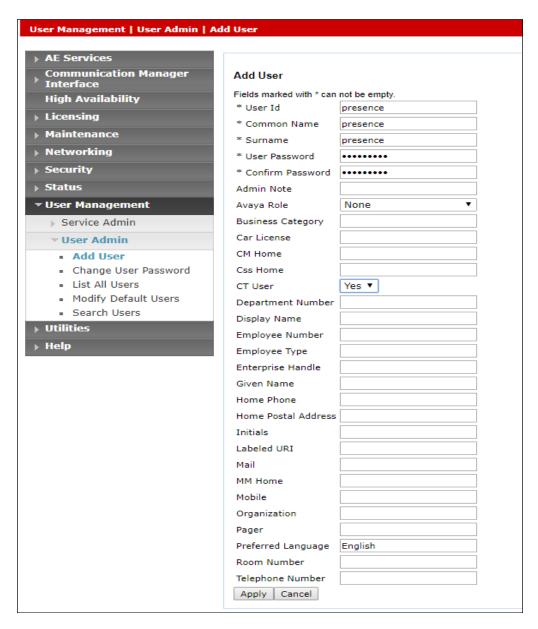


6.4 Create CTI User

A User ID and password needs to be configured for the Presence Suite server to communicate as a TSAPI client with the Application Enablement Services. Navigate to the **User Management** → **User Admin** and choose the **Add User** option (not shown). In the **Add User** screen shown below, enter the following values:

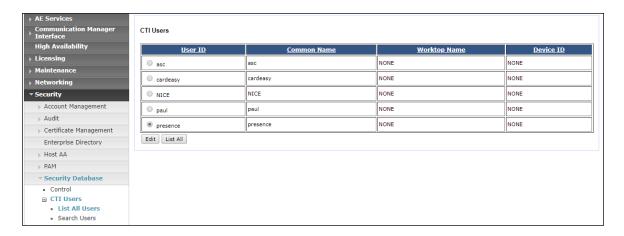
- User Id This will be used by Presence Suite in Section 7.1.
- Common Name and Surname Descriptive names need to be entered.
- User Password and Confirm Password This will be used with the User Id in Section 7.1.
- **CT User -** Select **Yes** from the drop-down menu.

Complete the process by choosing **Apply** at the bottom of the screen.



6.5 Enable Unrestricted Access for CTI User

Navigate to the **CTI Users** screen by selecting **Security** → **Security Database** → **CTI Users** → **List All Users**. Select the user that was created in **Section 6.4** and select the **Edit** button.



The **Edit CTI User** screen appears. Check the **Unrestricted Access** box and **Apply Changes** at the bottom of the screen.



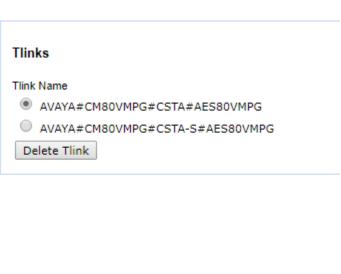
A screen appears to confirm applied changes to CTI User, choose **Apply**. This CTI user should now be enabled.



6.6 Identify Tlinks

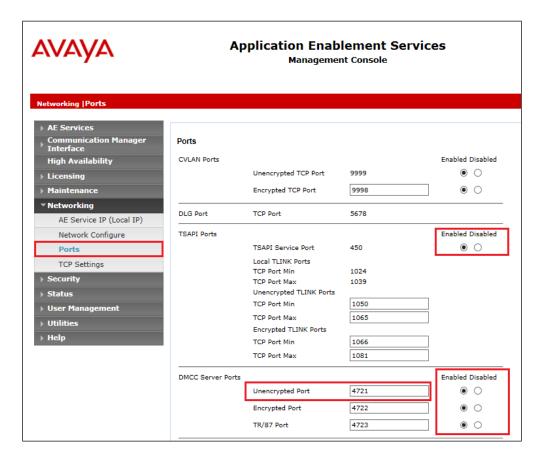
Navigate to **Security** → **Security Database** → **Tlinks**. Verify the value of the **Tlink Name**. This will be needed to configure Presence Suite in **Section 7.1**.



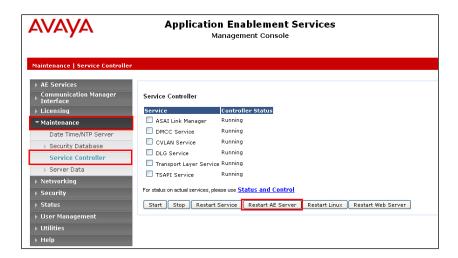


6.7 Configure Networking Ports

To ensure that TSAPI and DMCC ports are enabled, navigate to **Networking** → **Ports**. Ensure that the TSAPI ports are set to **Enabled** as shown below. Ensure that the **DMCC Server Ports** are also **Enabled** and take note of the **Unencrypted Port 4721** which will be used later in **Section 7.4.1**.



Once all the necessary changes are made it is a good idea to restart of the AE Server. Navigate to **Maintenance** \rightarrow **Service Controller**. In the main screen select **Restart AE Server** highlighted.

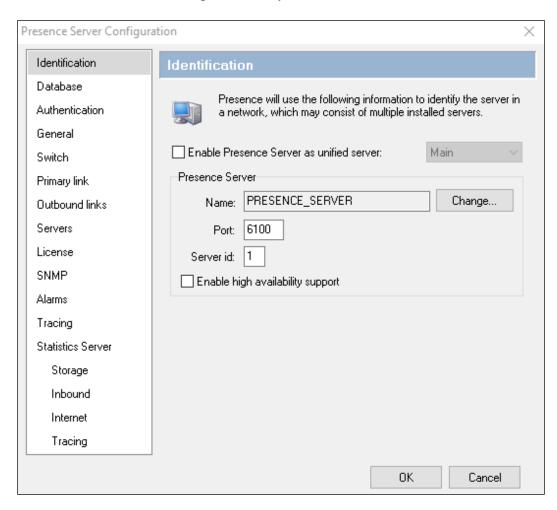


Configure Enghouse Presence Suite

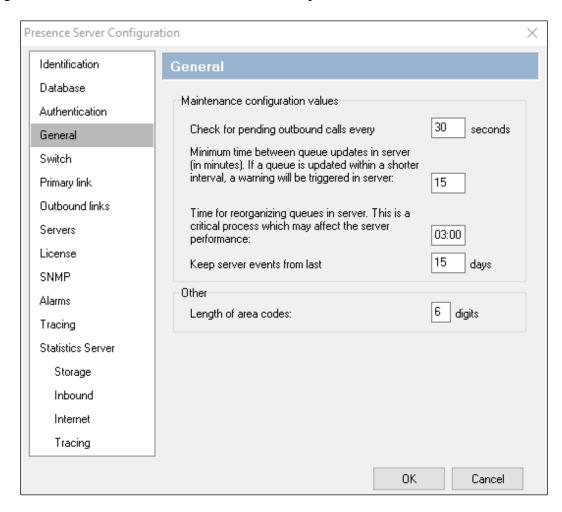
The Presence Suite includes the Presence Server, Presence Mail Interactions Server, Presence Web Interactions Server, Presence Recording Server, Presence Administrator, Presence Web Supervisor, and Presence Agent. The Presence Server and the Oracle database were pre-installed on the same machine for convenience during the compliance testing. The Presence server was configured and provided by Enghouse Interactive. An outline of the configuration relevant to the Avaya solution integration is detailed below.

7.1 Presence Server Configuration

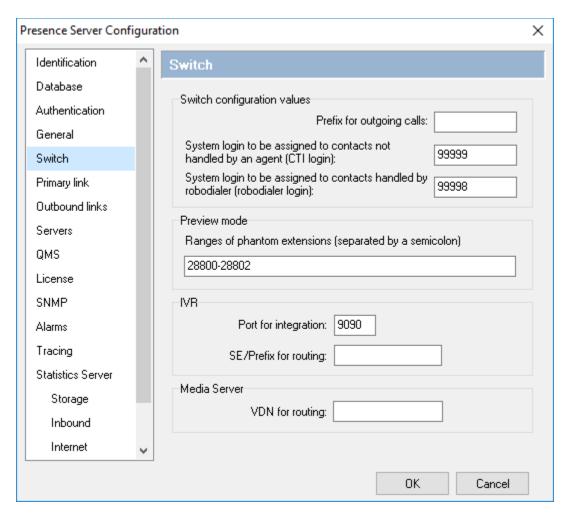
Launch the Presence Server configuration application by double clicking the **pcoservercfg.exe** located in the pre-installed Presence folder on the Presence Server (not shown). Select the **Identification** option from the menu on the left side of the screen, enter the **Server name** as **PRESENCE SERVER** as used for the identification of the server. The **Port** can be set to **6100**. Note that the actual value for server port can vary.



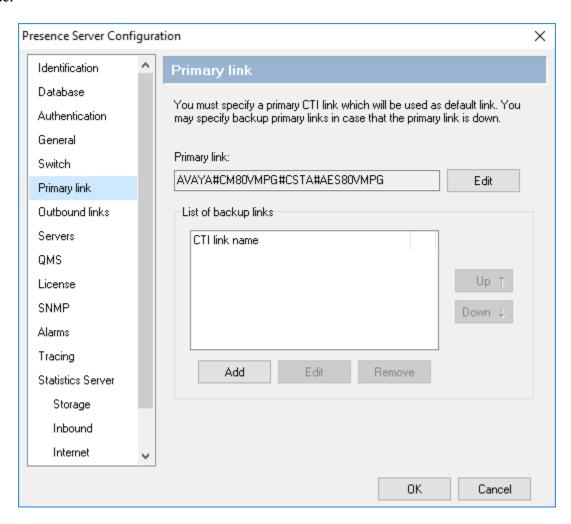
Select **General** from the menu on the left side of the screen. If desired, the Maintenance configuration values can be altered here, for the compliance test the default values were retained.



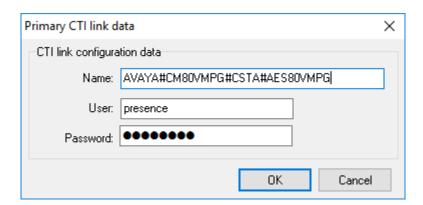
Select the **Switch** option from the menu on the left side of the screen. The **System login to be assigned to contacts not handled by an agent (CTI login)** field should be set to a value supplied by Presence, the value used for this configuration is **99999**. Enter the phantom extensions configured in **Section 5.7** in the **Ranges of phantom extensions** parameter.



Select the **Primary link** menu on the left side of the screen and choose the **Edit** button to enter a value.



In the resulting pop-up box enter the Tlink name from **Section 6.6** in the **Name** field. For the **User** and **Password** fields enter the user name and password configured on the Application Enablement Services in **Section 6.4**. Click **OK**.

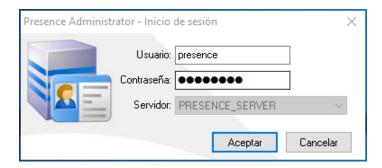


7.2 Presence Service Configuration

A number of services for inbound, outbound, email and internet were configured via the Presence Administrator. This section covers the basic configuration for each type of service. Please refer to **Section 10** for detailed documentation on configuring Presence Suite services.

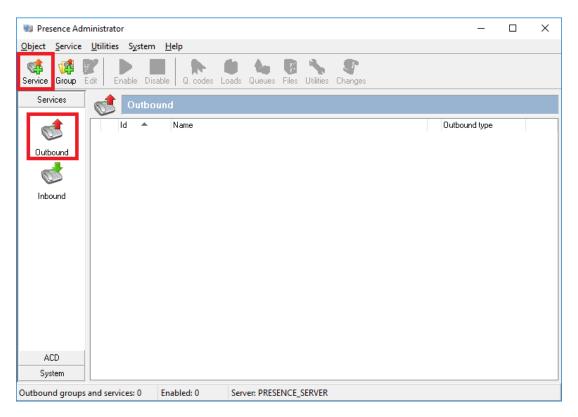
7.2.1 Logging in to Presence Administrator

Launch the **Presence Administrator** application by double clicking the **proadmin.exe** located in the Presence folder (not shown). The username and password that appear in the **User** and **Password** fields are created during the Presence Server installation.

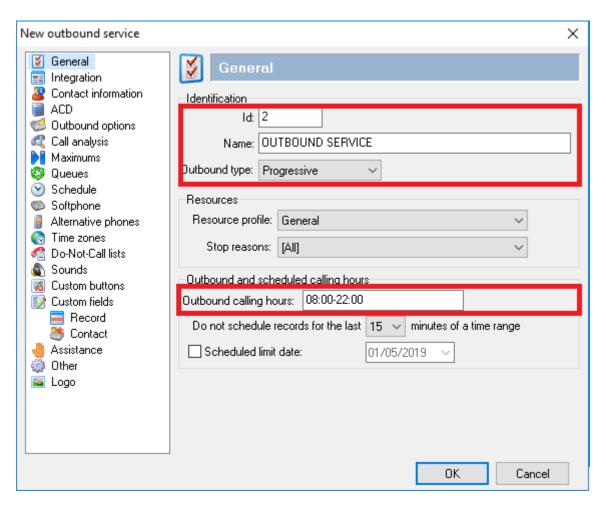


7.2.2 Outbound Service

After logging in to Presence Administrator the following screen will be displayed. Select **Services** \rightarrow **Outbound** from the Presence Administrator main menu on the left hand side. Click the **Service** button to configure an outbound service.

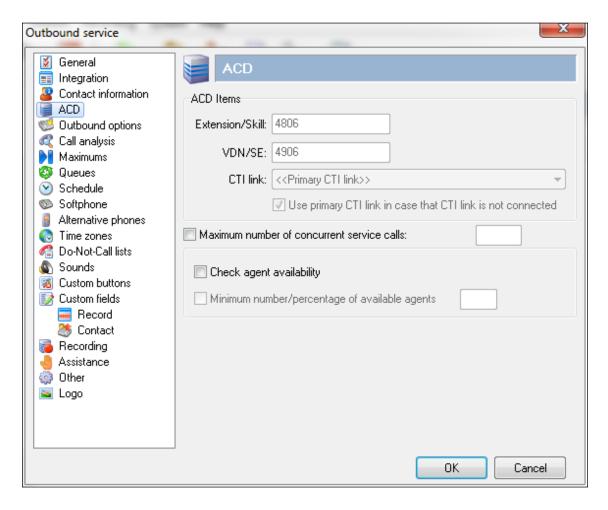


In the resulting screen, select **General** from the menu on the left hand side and enter an **Id** and a **Name** for the outbound service. In the **Outbound type** field select the type of outbound service, this specifies the mode in which the outbound service will operate, for further details of the type of outbound service available please refer to documentation in **Section 10**. In the **Outbound** calling hours field set the time range for which the outbound service will be active. All other fields are left with their default values.

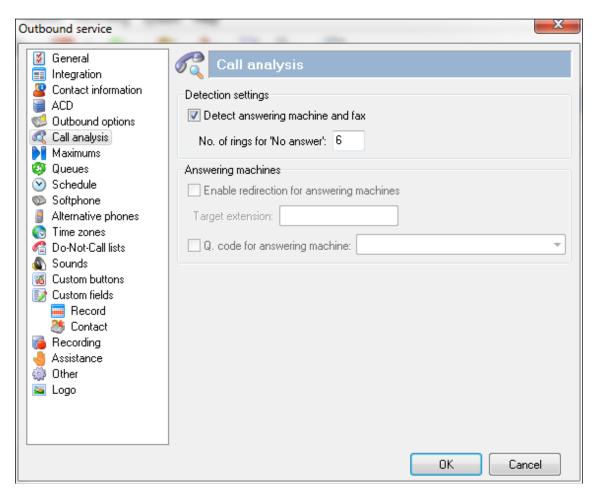


Select **ACD** from the left hand side menu and moving to the right. In the **Extension/Skill** field enter the extension number assigned to the outbound hunt group configured in **Section 5.3.1**. In the **VDN/SE** field enter the VDN number assigned to Outbound calls. In the test configuration only one CTI link was configured so the **CTI Link** field is set to **<<Pri>Primary CTI Link>>** if multiple CTI links exist on the system then the specific CTI link can be specified. All other field may be left at their default values.

Note: Typically a unique VDN/Skill would be setup for each module such as Inbound, Outbound, Webchat etc., in these screen shots taken for compliance testing the same VDN may have been used for some of these services.

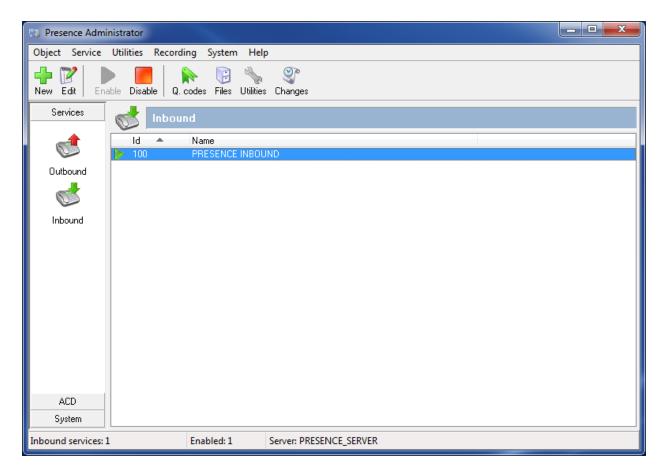


Select **Call analysis** from the left hand side menu. The fields in the right hand side define how the outbound service should behave following an unsuccessful attempt at contacting the customer. For testing, the **Detect answering machine and fax** box are checked and the **No. of rings for 'No answer'** is set to **6**, as shown in the screen below. Click **OK** to complete the outbound service configuration.

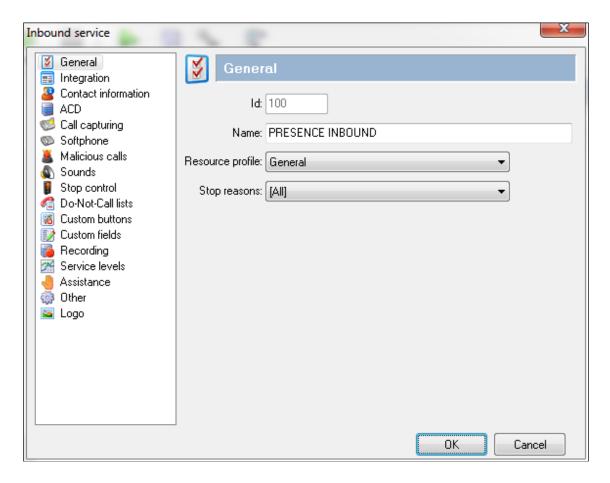


7.2.3 Inbound Service

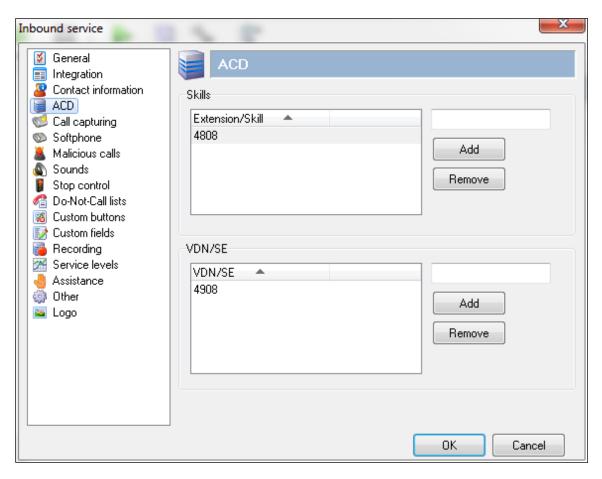
To configure an inbound service, from the left hand side select **Services →Inbound** from the Presence Administrator main menu. Click the **New** button.



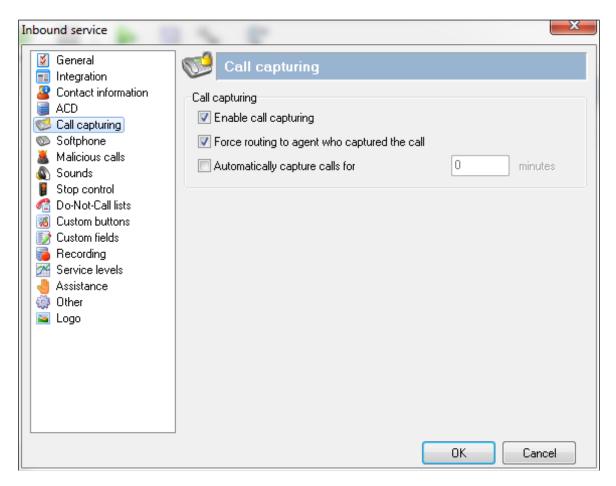
In the resulting screen, select **General** from the menu on the left hand side and enter an **Id** and a **Name** for the inbound service. All other fields are left with their default values.



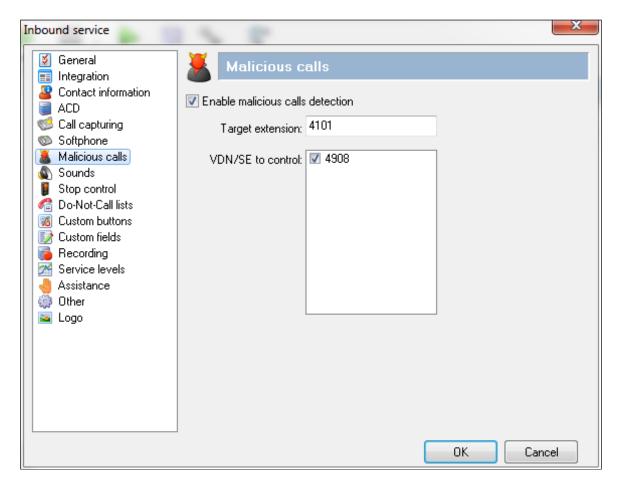
Select **ACD** from the left hand side menu and moving to the right, under the heading **Skills** enter the skill group extensions configured in **Section 5.3.1** that will handle inbound calls in the untitled box (this includes email and web chat call types) and click **Add**. The skill group extensions will then appear to the left in the **Extension/Skill** box. Under the heading **VDN/SE** enter the VDN configured in **Section 5.3.3** that will handle inbound calls in the untitled box and click **Add**. The VDN will then appear to the left in the **VDN/SE** box.



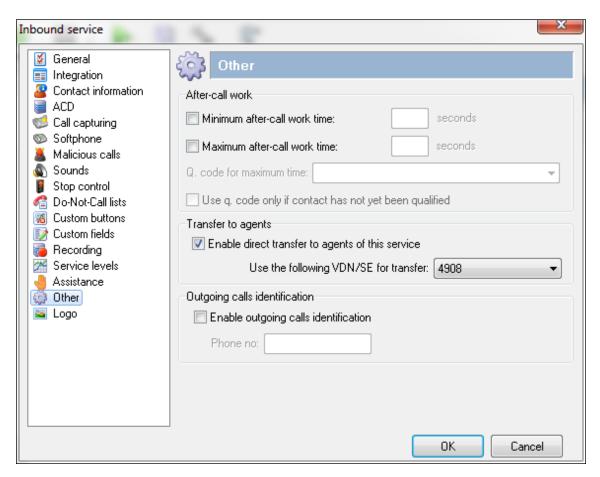
Select **Call capturing** from the left hand side menu and moving to the right, select the **Enable call capturing.** Force routing to agent who captured the call was checked for this compliance testing but is each users preference. These options allow an agent to mark an inbound call so that if the caller rings back while that agent is logged onto the system, the call will be routed again to the agent who tagged the call.



Select **Malicious calls** from the left hand side menu and moving to the right, select the **Enable malicious calls detection** check box. This option allows agents to mark calls as malicious, so that the caller can be directed to another location such as a supervisor position if they call back again. In the **Target extension** field enter the extension that any malicious calls will be redirected to. In the **VDN/SE to control** field select the VDNs this option will be available on.

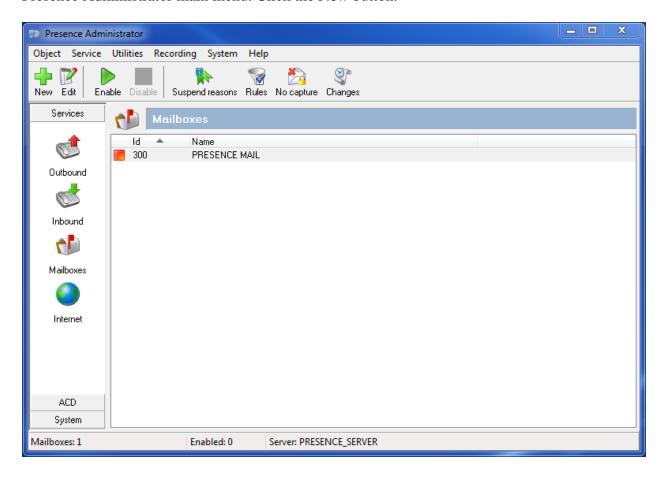


Select Other from the left hand side menu and moving to the right, select the Enable direct transfer to agents of this service check box. Enter the direct agent transfer VDN assigned in the Use the following VDN/SE for transfer field. Click OK to complete the inbound service configuration.

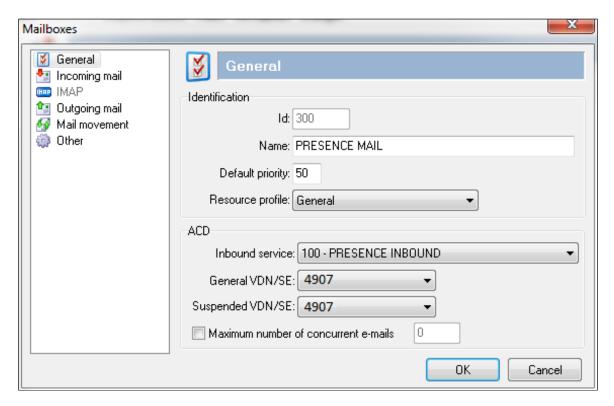


7.2.4 Email Service

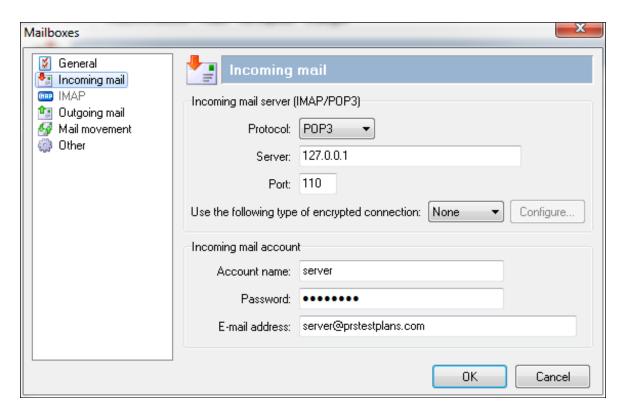
To configure an email service, from the left hand side select **Services** → **Mailboxes** from the Presence Administrator main menu. Click the **New** button.



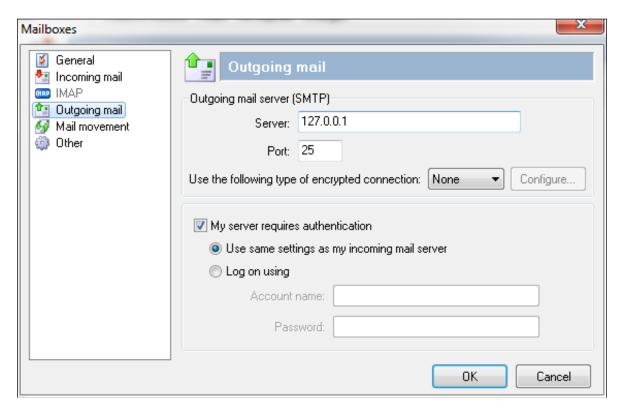
In the resulting screen, select **General** from the menu on the left hand side and enter an **Id** and a **Name** for the email service. Referring to **Section 5.3**, in the **General VDN/SE** field enter the VDN assigned for email and enter the VDN assigned for suspended emails in the **Suspended VDN/SE** field. This is to allow each incoming email to be reported on. When the email arrives the VDN is called and the agent is placed on work.



Select **Incoming mail** from the left hand side menu. This window allows administrator to specify the POP3 server and account from which to download incoming mails. In the **Server** field enter the POP3 mail server address. For the interoperability testing this was the same IP address as the Presence Server. The POP3 port of **110** is entered into the **Port** field. Under the **Incoming mail account** heading enter the **Account name**, **Password** and **E-mail address** associated with the POP3 mail account.

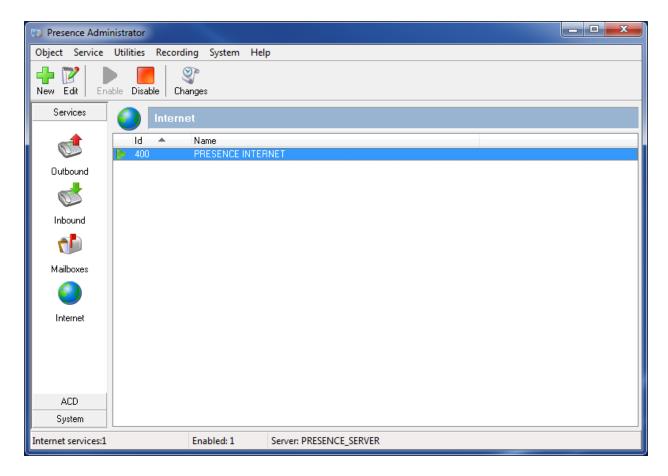


Select **Outgoing mail** from the left hand side menu and moving to the right, define the SMTP server that will be used to send response emails from Presence agents. Enter an IP address in the server field. For the interoperability testing this was the same IP address as the Presence Server. The SMTP port of **25** is entered into the **Port** field. Click **OK** to complete the email service configuration.

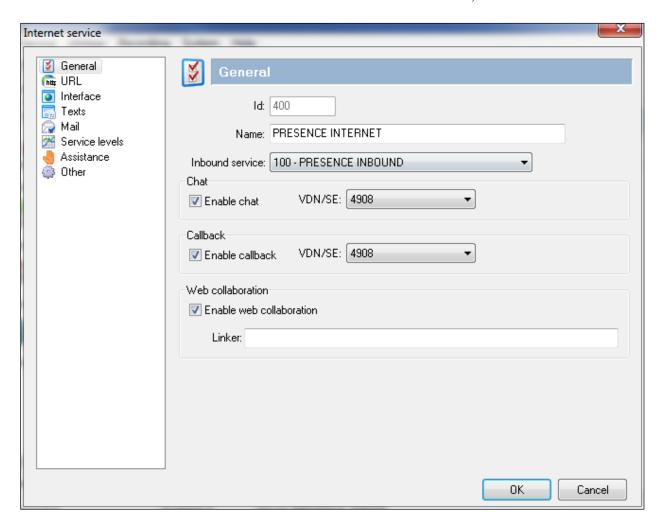


7.2.5 Web Chat / Web Call Back

To configure a web service, from the left hand side select **Services** → **Internet** from the Presence Administrator main menu. Click the **New** button.

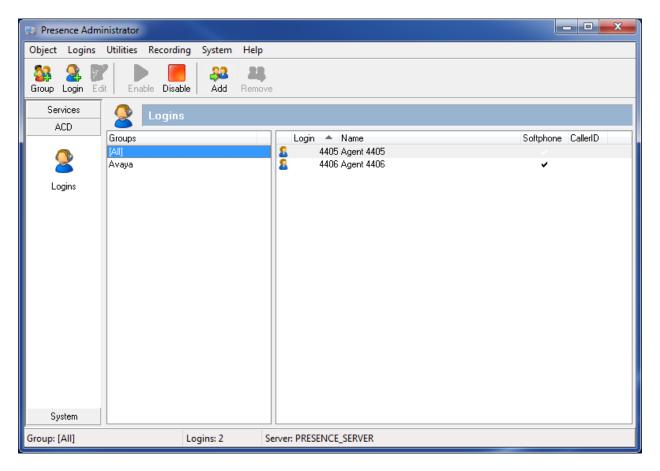


In the resulting screen, select **General** from the menu on the left hand side and enter an **Id** and a **Name** for the web service. The **Enable chat** and **Enable callback** check boxes should be selected and the relevant VDN for each entered into the **VDN/SE** field, click **OK** when done.

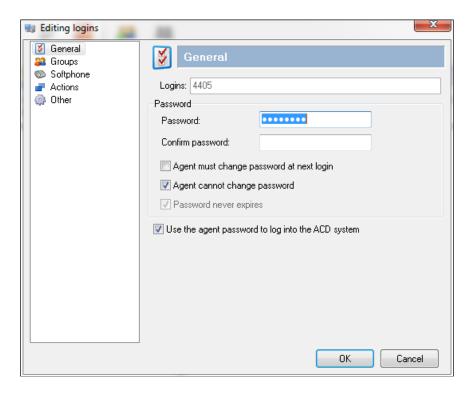


7.2.6 Add ACD Agent Logins

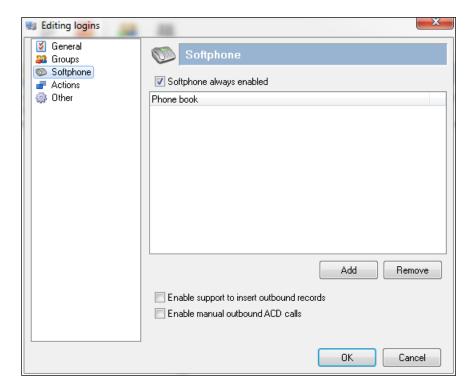
To add the agent logins administered on Communication Manager for use by Presence Suite, from the left hand pane of the Presence Administrator main menu select $\mathbf{ACD} \rightarrow \mathbf{Logins}$ and click the \mathbf{Login} button.



In the **Logins** field, enter a Communication Manager Agent Login ID and a password, as configured in **Section 5.5**. Best practice is to tick **Agent cannot change password** as shown.



Click on **Softphone** in the left pane, and place a tick in the **Softphone always enabled** field. Click **OK** when done.

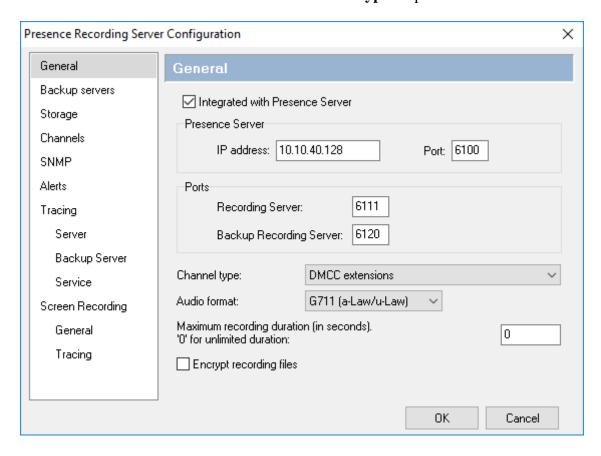


7.3 Configure Presence Recording

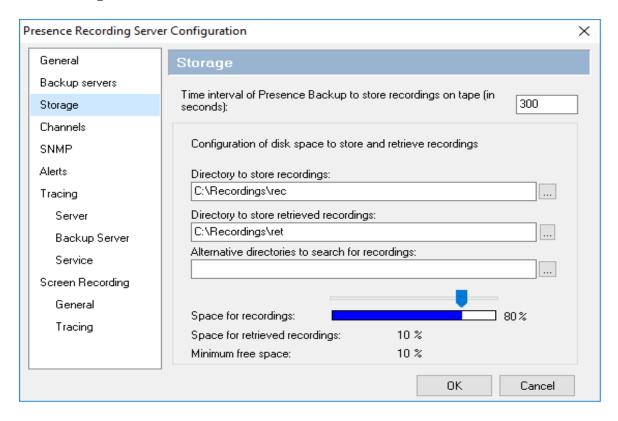
Launch the Presence Recording Server configuration application by double clicking the **precservercfg.exe** located in the pre-installed Presence folder on the Presence server (not shown).

7.3.1 Presence Recording Server Configuration

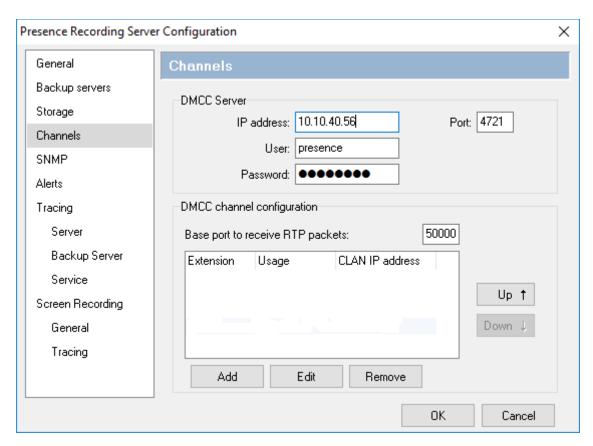
Select the **General** option from the menu on the left side of the screen, tick the **Integrated with Presence Server** box and enter the **IP address** of the Presence Server and the port used for connection. Select **DMCC extensions** from the **Channel type** drop-down box.



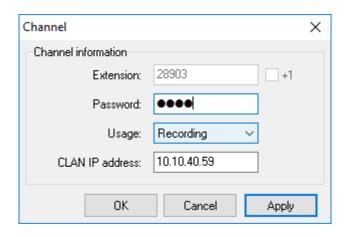
Click on **Storage** in the left-hand pane and enter an appropriate directory in the **Directory to store recordings** field.



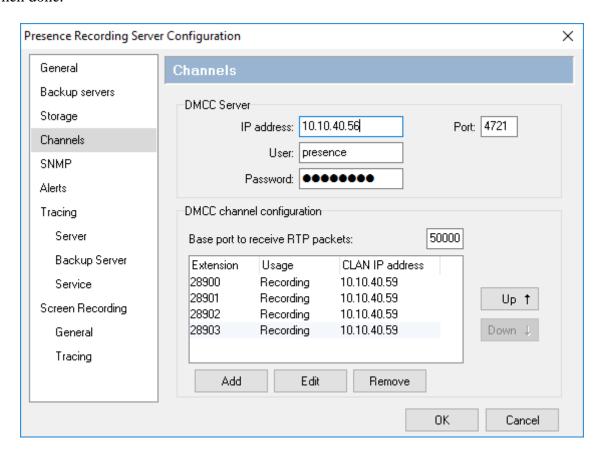
Click on **Channels** in the left-hand pane. In the **DMCC Server** section enter the IP address of the AES server and the AES user configured for the Presence Suite installation, enter the port configured for connectivity to AES (the default is **4721**). In the **DMCC channel configuration** section, click **Add**.



Enter a valid recording channel **Extension** and **Password** as configured in **Section 5.13**. Enter the **CLAN IP address** and select **Recording** from the **Usage** drop-down box. Click **OK** when done. Repeat as necessary. For playback channels, select **Playback** from the **Usage** drop-down box.



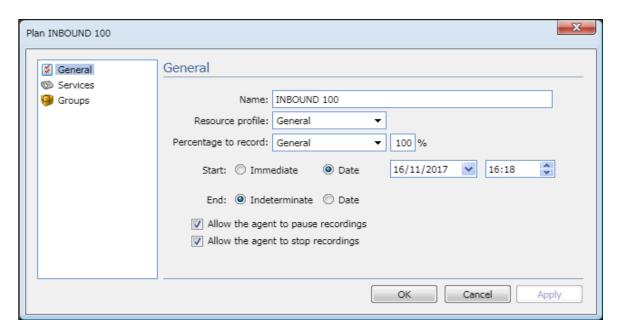
The screen shown below will appear, displaying all recording and playback channels, click \mathbf{OK} when done.



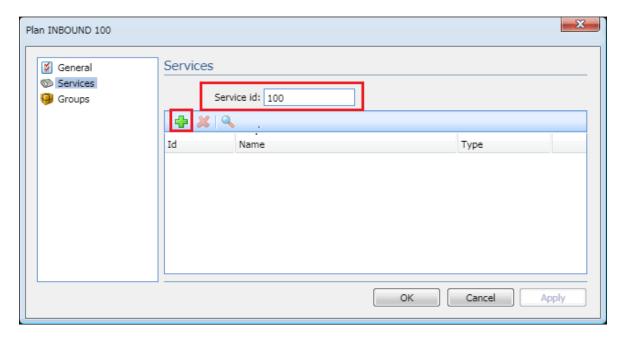
7.3.2 Configure Recording Plan for Inbound Service

Recording plans must be configured according to the call recordings required. This plan is created for the inbound calls to agents.

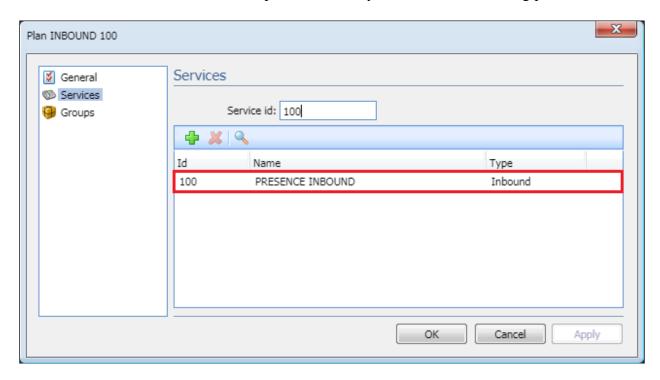
Using the Presence Web Supervisor, click on Administration \rightarrow Recording \rightarrow Plans \rightarrow New (not shown). In the displayed window, assign an identifying Name and set the **Percentage to record** as required, in this case 100%. Configure the **Start** and **End** parameters as appropriate.



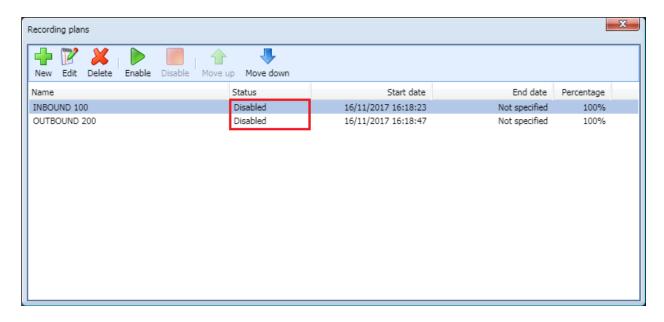
Click on **Services** in the left-hand pane, enter the inbound service identifier in the **Service ID** box and click the plus icon.



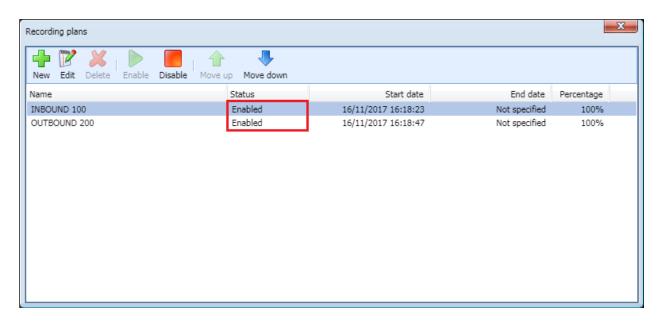
This will add the relevant configured service to the recording plan, in this case **PRESENCE INBOUND**. Click **OK** when done. Repeat as necessary for additional recording plans.



The screen below will be displayed, summarizing the added recording plans. Note that the status shows **Disabled**.



Select each one in turn and click **Enable**, the status will now appear as **Enabled**.



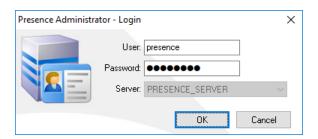
Calls that are placed via either of these services will be recorded according to the recording plan configured above.

7.3.3 Add stations to Presence Recording

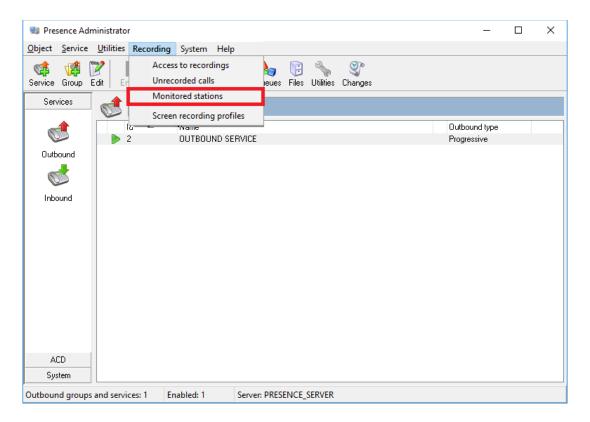
To record non-agent stations, each station that is to be recorded must be added to Presence Recording. Create a recording group adding these stations and finally create a recording plan adding this group.

Note: In the example below extensions 8270001 and 8270002 are added to be recorded by Presence Recording.

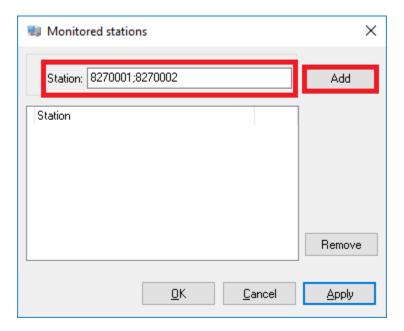
Launch the **Presence Administrator** application by double clicking the **proadmin.exe** located in the Presence folder (not shown). The username and password that appear in the **User** and **Password** fields are created during the Presence server installation.



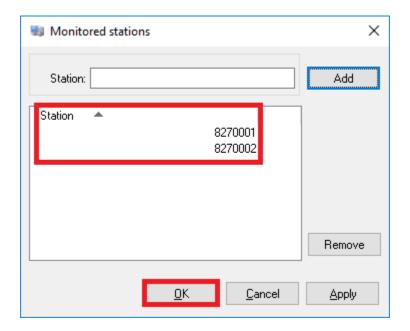
Select **Recording** → **Monitored stations** from the Presence Administrator menu. Calls that are placed via either of these stations will be recorded according to the recording plan configured in **Section 7.3.5.**



In the resulting screen, add both stations in the Station field and click Add.

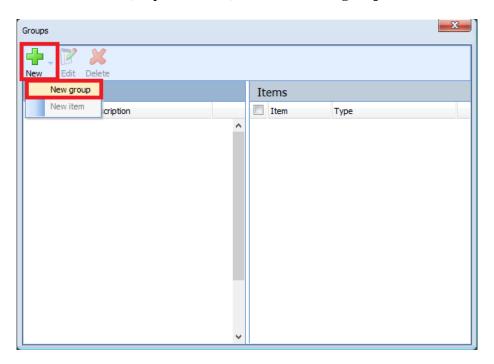


Then click OK to add the stations into Presence Recording and close the screen.

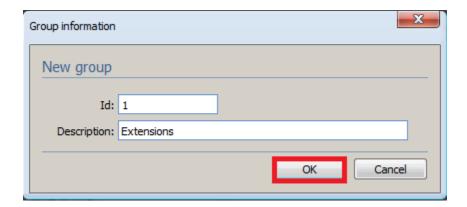


7.3.4 Create a new recording group

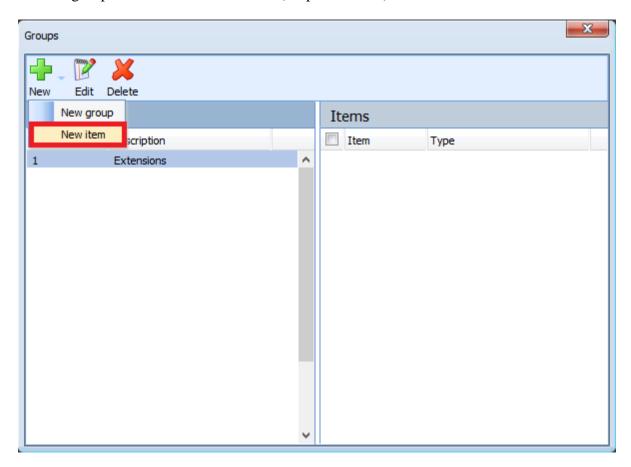
Using the Presence Web Supervisor application, double click on **Administration** → **Recording** → **Groups** and click on **New** (drop-down box) and select **New group**.



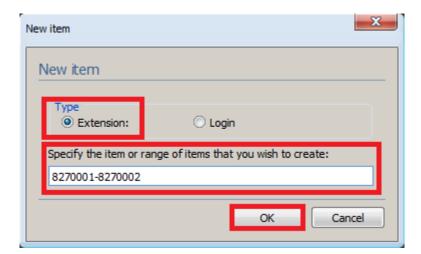
Enter the details for the new group. Note any number is used for code. Click on \mathbf{OK} when finished.



Select the group created and click on New (drop-down box) and select New item.



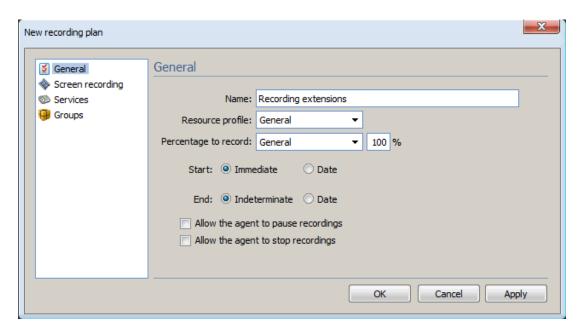
Select Extension as the Type and the extensions to be added. Click on **OK** once done.



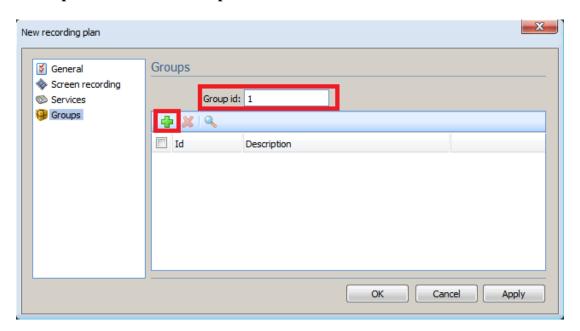
Once **OK** is clicked above the following screen shows the added stations.

7.3.5 Create a recording plan for extensions

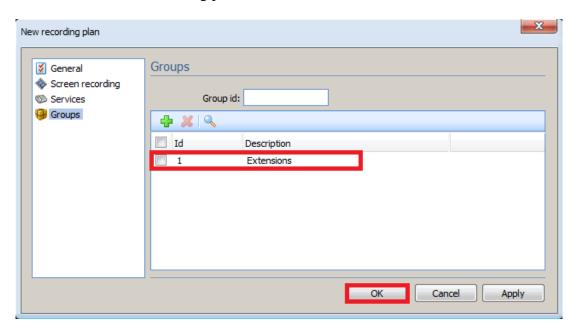
Double click on Administration → Recording → Plans → New (not shown). In the displayed New recording plan window, enter a Name, the Resource profile is pre-selected, Percentage to record is set to 100%. Start and End is set to Immediately and Indeterminate respectively. Click on OK once done.



On the **Groups** window set the **Group id** created click on the Add icon.



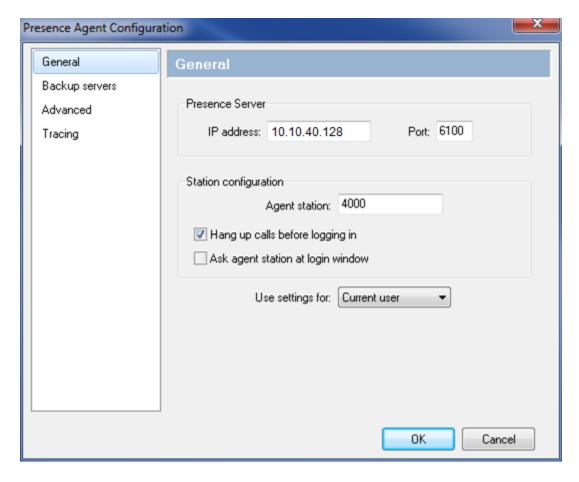
Click on \mathbf{OK} to create the recording plan.

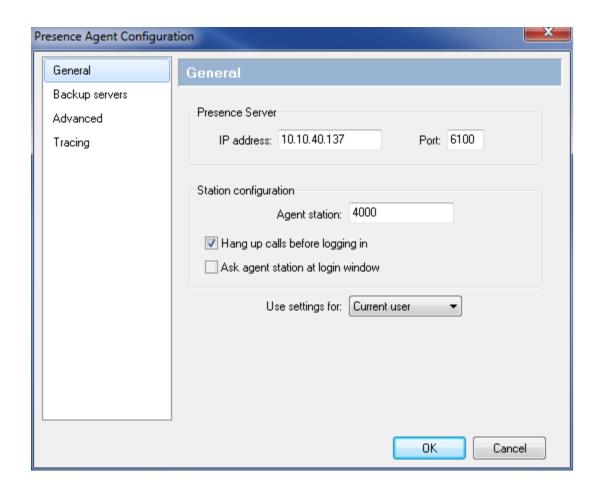


In the screen that display the added recording plans (not shown), select the recording plan created and click **Enable**, the status will now appear as **Enabled**.

7.4 Presence Agent Configuration

The following steps are carried out on the Presence Suite Agent PC. Prior to installing the Presence agent, ensure that the DBExpress driver (dbexpoda40.dll) is located in the C:\Windows\System32 directory. The DBExpress driver allows the agent application to communicate with the Oracle database. Installing this driver eliminates the need to install the Oracle client. Launch the Presence agent configuration application by double clicking the proagentcfg.exe located in the C: → Presence folder. Enter the Presence Server IP: address as 10.10.40.128. The Presence Server port can be left as the default value of 6100. Enter the extension of the agent that will be using this workstation in the Agent station field. Check both the Hang up calls before logging in check box. In the field Use settings for choose Current user from the drop down menu. Click OK. This step is needed for each agent configured; only the agent station field will vary.



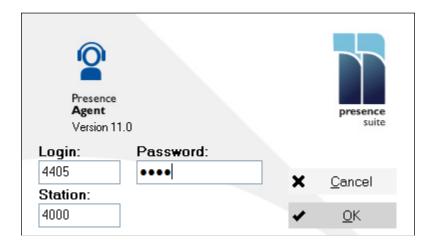


Verification Steps

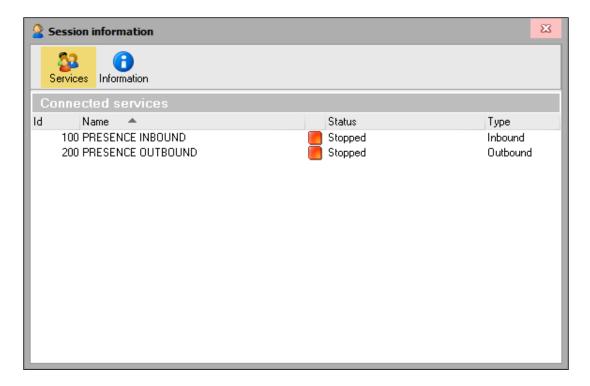
This section provides the tests that can be performed to verify correct configuration of Communication Manager, Application Enablement Services and Presence Suite.

8.1 Verify Presence Suite

Launch the Presence agent configuration application by double clicking the **pcoagent.exe** located in the Presence folder (not shown). Enter the agent **Login** and **Password** configured in **Section 5.5** and click on **OK**.



In the next screen, click on the **Services** button in the task bar. The service set up for the agent will be displayed.



A task bar is present at the top of the Agent PC. Click on the green arrow to put the agent in to an available state.



The information status on the task bar goes to available indicating the agent is ready to receive calls.

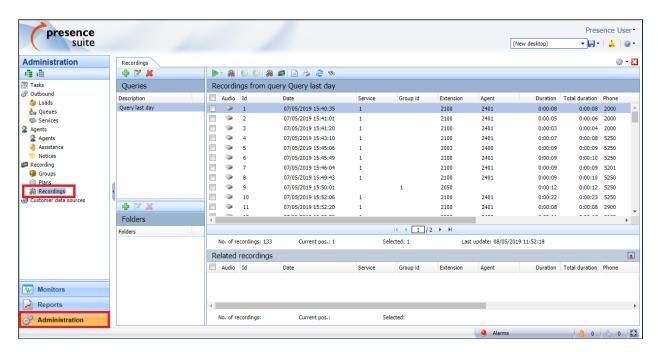


An outbound call is placed and answered.

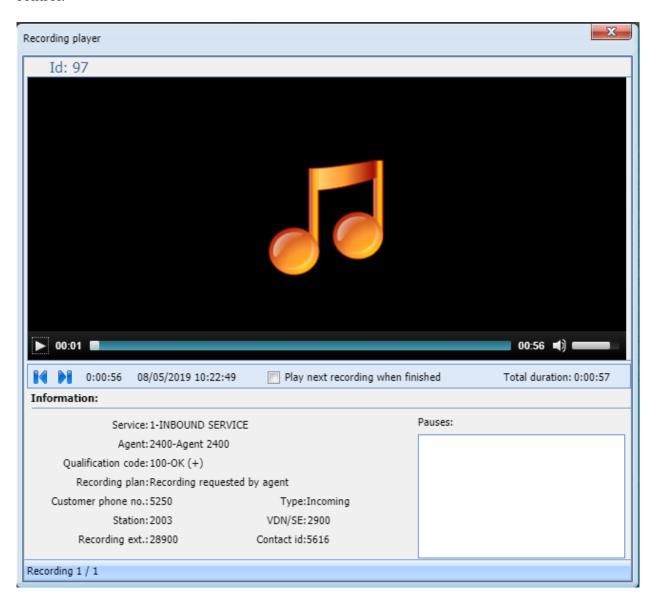


8.2 Verify Presence Recording

Using Presence Web Supervisor, click **Administration** \rightarrow **Recording** \rightarrow **Recordings**, visually verify correct recording detail as shown below. Select a recording and click on Play audio only in browser to be played, the audio will be played in the computer.



The audio will be played in the computer. The screen below will be displayed allowing playback control.



8.3 Verify Avaya Aura® Communication Manager CTI Link

The following steps can ensure that the communication between Communication Manager and the Application Enablement Services server is functioning correctly. Check the TSAPI link status with Application Enablement Services by using the command **status aesvcs cti-link**. Verify the **Service State** of the TSAPI link is **established**.

status aesvcs cti-link											
AE SERVICES CTI LINK STATUS											
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Rcvd					
1	8	no	aes80vmpg	established	87	61					

Use the command **status aesvcs interface** to verify that the status **Local Node** of Application Enablement Services interface is connected and **listening**.

status aesvcs interface											
AE SERVICES INTERFACE STATUS											
Local Node	Enabled?	Number of Connections	Status								
procr	yes	1	listening								

Verify that the there is a link with the Application Enablement Services and that messages are being sent and received by using the command **status aesvcs link**.

status	aesvcs link									
	AE SERVICES LINK STATUS									
Srvr/ Link	AE Services Server	Remote IP	Remote Port	Local Node	Msgs Sent	Rcvd				
01/01	aes80vmpg	10.10.40.56	57650	procr	683	665				

8.4 Verify TSAPI Link and DMCC

This section will verify both the TAPI and DMCC links.

8.4.1 Verify TSAPI Link

On the AES Management Console verify the status of the TSAPI link by selecting **Status Status** and **Control TSAPI Service Summary** to display the **TSAPI Link Details** screen. Verify the status of the TSAPI link by checking that the **Status** is **Talking** and the **State** is **Online**.



8.4.2 Verify Avaya Aura® Application Enablement Services DMCC Service

The following steps are carried out on AES to validate that the communication link between AES and the Presence Recording server is functioning correctly. Verify the status of the DMCC service by selecting Status → Status and Control → DMCC Service Summary. The DMCC Service Summary − Session Summary screen is displayed as shown below. It shows a connection to the Presence server, IP address 10.10.40.128. The Application is shown as cmapiApplication, and the Far-end Identifier is given as the IP address 10.10.40.128 as expected. The User is shown as the user created for the CTI user for Presence Recording.

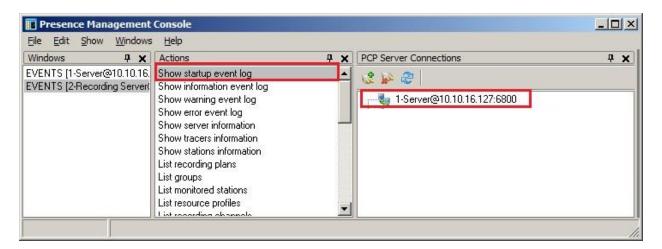




8.5 Verify Presence Suite CTI Connection

One of the available methods to confirm correct startup is a startup log which can be accessed from Presence Management Console. Navigate to C: → Presence → pmconsole.exe (not shown). A startup log commences when the Presence Server is trying to load and connect to the Application Enablement Services server. Click on the item named Server@10.10.16.127:6800 in the PCP Server Connections pane of the Management Console. To open the startup event log, double click Show startup event log in the Actions pane.

Note: The example below shows a connection to another AES server not the server mentioned in this document.



Verify successful CTI connection and service startup.

Conclusion

These Application Notes describe the configuration steps required for Presence Suite R11.1 to successfully interoperate with Avaya Aura® Communication Manager R8.0.1 using Avaya Aura® Application Enablement Services R8.0. All feature functionality and serviceability test cases were completed successfully with observations noted in **Section 2.2.**

Additional References

This section references the Avaya and Presence Suite product documentation that are relevant to these Application Notes.

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

- [1] Administering Avaya Aura® Communication Manager, Document ID 03-300509
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Document ID 555-245-205
- [3] Avaya Aura® Application Enablement Services Administration and Maintenance Guide Release 8.0

The following documentation is available on request from Enghouse at www.enghouseinteractive.es/en

- [4] ACD Sys Presence Administrator Manual Presence Suite, V11.1
- [5] Presence Installation Guides Presence Software, V11.1
- [6] PBX/ACD Requirements Presence Software, V11.1

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