



DevConnect Program

Application Notes for Mitel InAttend using Mitel Attendant Connectivity Server V2.7 to interoperate with Avaya Aura® Communication Manager R10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Mitel InAttend v2.7 using Mitel Attendant Connectivity Server from Mitel Sweden AB to interoperate with Avaya Aura® Communication Manager R10.1. The Mitel solution makes use of two separate connections to Avaya Aura® Session Manager and to Avaya Aura® Application Enablement Services.

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.

1. Introduction

These Application Notes describe the configuration steps required for Mitel InAttend using Mitel Attendant Connectivity Server V2.7 from Mitel Sweden AB to interoperate with Avaya Aura® Communication Manager R10.1 utilizing a SIP trunk connection to Avaya Aura® Session Manager R10.1 and a TSAPI connection to Avaya Aura® Application Enablement Services R10.1 (AES).

Mitel InAttend is the core application in the Mitel attendant offering and an essential part in the Mitel Collaboration Management (CMG). It is a multi-featured attendant solution that is built on open standards and offers advanced collaboration features. The InAttend attendant console provides all necessary information for efficient call handling yet is fully integrated with the Mitel CMG for a complete Unified Communications experience. The InAttend SIP-based platform opens a way for integration with Avaya Aura® Communication Manager utilising a SIP connection to Avaya Aura® Session Manager using the Mitel Attendant Connectivity Server (ACS).

The ACS is responsible for the SIP connection to Session Manager and is part of the Attendant Platform which provides Private Branch Exchanges (PBX) with extended functionality. The Attendant client communicates with the private branch exchange through ACS. Using an attendant client, attendants can initiate, answer, transfer and disconnect calls. The call queuing functionality with configurable call queues also supports camp on services. Other features include automatic call distribution, which distributes the call to the attendant with the longest idle time, and direct drop to voicemail, which lets the attendant transfer calls directly to subscriber's voicemail. ACS also provides a speech attendant that enables a caller to request a user by name, and if busy, enables the caller to be transferred to an attendant, to the user's voicemail, or added to a conference. ACS also incorporates its own voicemail system.

The Mitel InAttend solution makes use of two TSAPI connections to Avaya Aura® Application Enablement Services.

- TSAPI connection from the CMG – Used to set Call Forwarding and Message Waiting.
- TSAPI connection from the InAttend server – Used to monitor devices to provide Presence information.

Mitel InAttend is made up of the following all installed on the same sever.

- Mitel Attendant Connectivity Server.
 - NeTS 5.13.25.0
 - MediaServer 1.9.145.0
 - QueueManager 2.19.25.0
- Mitel InAttend Server.
 - Collaboration Management CMG 8.6
 - Virtual Reception 8.6
 - Microsoft SQL 2019
 - Mitel InAttend Server 2.7

During compliance testing various applications such as Virtual Reception which consists of Speech Attendant and Speech Office, and these were tested alongside the InAttend console. These applications utilize the ACS to connect to Session Manager and the Mitel InAttend Server to connect to TSAPI. Each of these applications add to the overall solution and this solution will be referenced as “InAttend” throughout the remainder of this document unless there is a specific reason to refer to a specific application.

Mitel supplies, installs and configures their solution for the end customer through qualified partners. In line with Mitel’s request the configuration of InAttend is not necessarily required to be part of these Application Notes, however **Section 8** does include screen shots of the setup that was used during compliance testing.

2. General Test Approach and Test Results

The general test approach was to configure InAttend to communicate with Communication Manager as implemented on a customer’s premises using a SIP connection to Session Manager and two TSAPI connections to AES. Testing focused on verifying that ACS registered with Session Manager as a SIP Entity and both TSAPI connections showing that all features behaved as expected. Various call scenarios were performed to simulate real call types as would be observed on a customer premises. See **Figure 1** for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests.

The ACS is configured as a SIP Entity on Session Manager acting as a third-party PBX connecting to the Avaya solution over a SIP trunk. The connection was setup using TCP transport and port 5060. Calls were then made from Communication Manager to the Mitel Attendant using a Dialling Plan on Communication Manager. Calls can be made between the Mitel solution and Communication Manager extensions by a connection between the ACS and Session Manager.

The TSAPI client is installed on the InAttend server which also runs the CMG database. This client then connected to the AES using a user/password created on AES allowing the TSAPI events be passed to the InAttend server and be processed by the applications there.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member’s solution.

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and Mitel InAttend did not include use of any specific encryption features as requested by Mitel.

2.1. Interoperability Compliance Testing

The testing included:

- Verification of connectivity between Communication Manager and InAttend via Session Manager and AES
- InAttend and Speech Attendant transfers calls
- Supervised and unsupervised transfer with answer
- Directing callers to conference calls via Speech Attendant
- Call queuing and retrieval
- Detection for busy and unanswered extensions
- End to end signalling
- Call re-queuing
- Direct drop to voice mail
- Setting Call Forward and Message Waiting
- Observing Presence Information
- Serviceability tests simulating a LAN failure

2.2. Test Results

Tests were performed to ensure full interoperability of the Mitel solution with Communication Manager using the connection between the ACS and Session Manager and a TSAPI connection between the InAttend server and AES. The tests were all functional in nature and performance testing was not included. All test cases passed successfully with the following observations noted.

1. When Call Forwarding Enhanced is used to forward an Avaya extension to InAttend, be that Call Forward No Answer or Call Forward Busy, InAttend is not aware of the reason that the call is being forwarded for Call Forward No Answer, it is aware for Call Forward Busy. Mitel are aware of this and are investigating the issue.
2. When Coverage Path is used to forward an Avaya extension to InAttend, be that Call Forward No Answer or Call Forward Busy, InAttend is not aware of the reason that the call is being forwarded for Call Forward No Answer, it is aware for Call Forward Busy. Mitel are aware of this and are investigating the issue.
3. Mitel requires that a person's phone is forwarded to the conference application for a conference to take place. A Communication Manager user/extension will get a busy tone when attempting to call itself when the extension is forwarded. When the administrator of a conference needs to dial in to that conference, they will call their extension from another known source i.e., their mobile phone. This mobile number would be associated with this user/extension on the Mitel database and so this call would be recognised as the

conference administrator dialling in. A Coverage Path can also be used instead of Call Forward and this will allow the user call in from the phone itself.

4. The Transport Protocol for the SIP trunk connection between the Mitel Attendant Connectivity Server and Session Manager can be either TCP or UDP but does not currently support a TLS connection.

2.3. Support

Technical support from Mitel can be obtained through the following.

Web: www.Mitel.com/service-and-support

Tel: +1 800-722-1301

Partners can log on to <https://miaccess.mitel.com/idp/index.xhtml> where access to TeamTrack is given for reporting issues.

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, Session Manager and AES. Mitel InAttend is installed on a Windows Server 2019 OS. A network telephony server and SQL were also installed on the same server. On Communication Manager, the routing was configured to route 450x calls to Session Manager which in turn were routed to the ACS. Mitel InAttend was installed and configured on a client PC. H323, SIP and Digital phones were configured on Communication Manager to generate calls to Mitel InAttend and outbound calls to a simulated PSTN. A TSAPI connection was utilized between the Mitel InAttend server and AES.

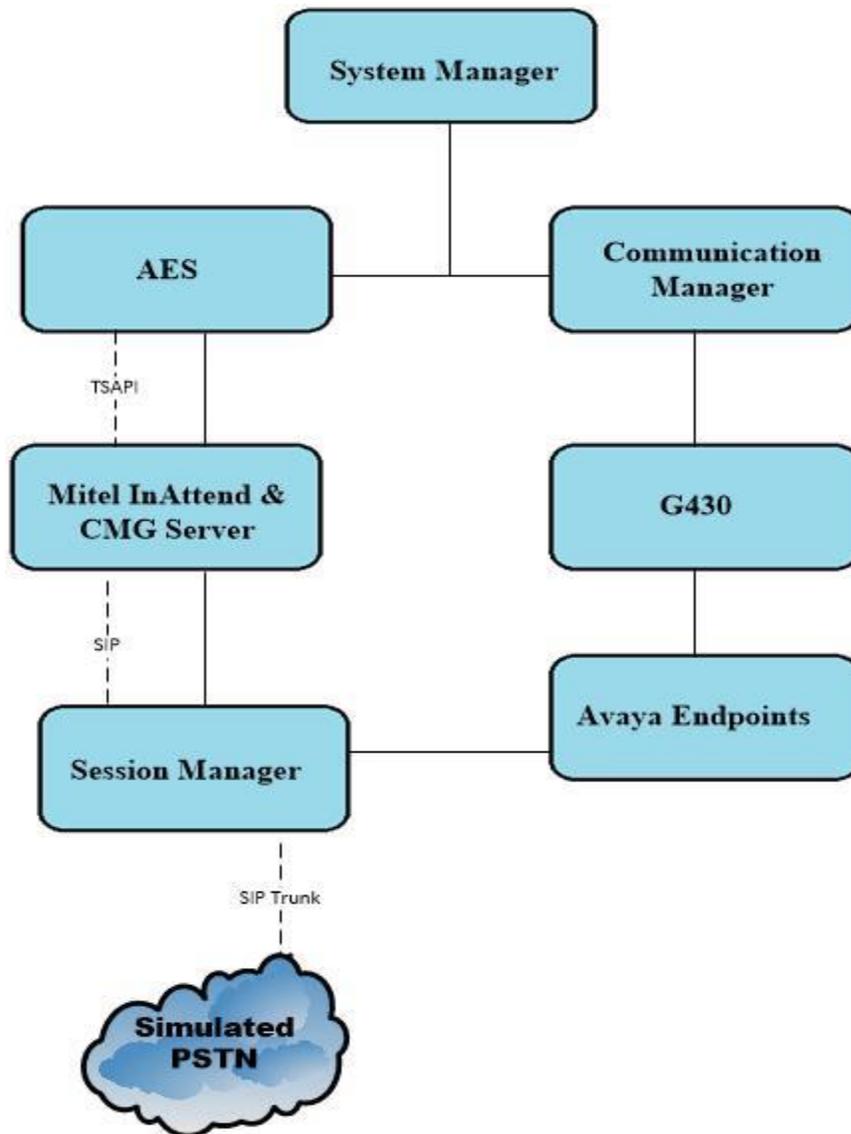


Figure 1: Avaya Aura® Communication Manager and Mitel InAttend configuration

4. Equipment and Software Validated

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

Avaya Equipment/Software	Release/ Version
Avaya Aura® System Manager	System Manager 10.1.3.0 Feature Pack 3 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.3.0.0715713
Avaya Aura® Session Manager	Session Manager R10.1 Build No. – 10.1.3.0.1013007
Avaya Aura® Communication Manager	R10.1.3.0 – FP3 R020x.01.0.974.0 Update ID 01.0.974.0-27893
Avaya Aura® Application Enablement Services	10.1.3 R10.1.3.0.0.11-0
Avaya Aura® Media Server	10.1.0.101
Avaya Media Gateway G450	42.7.0 /2
Avaya J100 Series (H323) Deskphone	6.8.5.3.2
Avaya J100 Series (SIP) Deskphone	4.0.14.0.7
Avaya 9404 Digital Deskphone	17.0
Avaya Session Border Controller (to facilitate simulated PSTN)	10.1
Mitel Equipment/Software	Release/ Version
Mitel Attendant Connectivity server running on Windows 2019	Mitel Attendant Connectivity Server includes: NeTS 5.13.25.0 MediaServer 1.9.145.0 QueueManager 2.19.25.0
Mitel InAttend server running on Windows 2019	Version 2.7 Mitel InAttend Server includes: Collaboration Management 8.6 Virtual Reception 8.6 Microsoft SQL 2019
Mitel InAttend Attendant client running on Windows 10 Enterprise	Version 2.7.28.0

Note: The Avaya Aura® platform as well as the Mitel equipment are all running on VMware.

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using 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 initial installation and configuration, please refer to the product documentation in **Section 11**.

The configuration operations described in this section can be summarized as follows:

- Verify System Parameters and Features
- Configure SIP Trunk
- Configure Call Routing for InAttend
- Configure Connection to AES
- Configure VDNs and Vectors for InAttend

Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

5.1. Verify System Parameters and Features

Each Communication Manager system will have its own setup with different System Parameters and Features configured depending on the requirement of the customer. Here is a snapshot of some of these values that were configured on the DevConnect lab for compliance testing.

5.1.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that **Maximum Administered SIP Trunks** has sufficient capacity. Each call answered by InAttend uses a minimum of one SIP trunk. Calls that are routed back to stations on Communication Manager or calls that are routed back to Communication Manager to access the PSTN will use two SIP trunks.

<code>display system-parameters customer-options</code>	Page	2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	12000	250
Maximum Concurrently Registered IP Stations:	18000	2
Maximum Administered Remote Office Trunks:	12000	0
Maximum Concurrently Registered Remote Office Stations:	18000	0
Maximum Concurrently Registered IP eCons:	414	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	18000	0
Maximum Video Capable IP Softphones:	18000	0
Maximum Administered SIP Trunks:	24000	319
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0

On **Page 4**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                               Page 4 of 12
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y                               Audible Message Waiting? y
      Access Security Gateway (ASG)? n                               Authorization Codes? 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
```

On **Page 5**, ensure that **Uniform Dialing Plan** is set to **y**.

```
display system-parameters customer-options                               Page 6 of 12
                                OPTIONAL FEATURES

      Multinational Locations? n                                     Station and Trunk MSP? y
Multiple Level Precedence & Preemption? n                               Station as Virtual Extension? y
      Multiple Locations? n
                                System Management Data Transfer? n
      Personal Station Access (PSA)? y                               Tenant Partitioning? y
      PNC Duplication? n                                           Terminal Trans. Init. (TTI)? y
      Port Network Support? y                                       Time of Day Routing? y
      Posted Messages? y                                           TN2501 VAL Maximum Capacity? y
                                Uniform Dialing Plan? y
      Private Networking? y                                         Usage Allocation Enhancements? y
```

5.1.2. Configure System Features

For the testing, **Trunk-to Trunk Transfer** was set to **all** on **Page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 11** for supporting documentation.

```
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: attd
      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
```

5.2. Configure SIP Trunk

In the **Node Names IP** form, note the IP Address of the processor interface of Communication Manager (**procr**) and the Session Manager (**sm101x**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

```
display node-names ip Page 1 of 2
      IP NODE NAMES
      Name IP Address
      sm101x 10.10.40.12
      aespri101x 10.10.40.16
      aessecc101x 10.10.40.46
      g450 10.10.40.15
      procr 10.10.40.13
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.1.1**. In this configuration, the domain name is **greanep.sil6.avaya.com**. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session manager as **ip-network region 1** is specified in the SIP signaling group.

```

display ip-network-region 1                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 1
Location: 1           Authoritative Domain: greanep.sil6.avaya.com
Name: Default region
MEDIA PARAMETERS
  Codec Set: 1
  Intra-region IP-IP Direct Audio: yes
  Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048
  UDP Port Max: 3329
  IP Audio Hairpinning? n
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5
H.323 IP ENDPOINTS
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
                                                                AUDIO RESOURCE RESERVATION PARAMETERS
                                                                RSVP Enabled? n

```

In the **IP Codec Set** form, select the audio codecs supported for calls routed over the SIP trunk to InAttend. The form is accessed via the **change ip-codec-set n** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; **G.722**, **G.711** and **G.729** were all tested. Note the **Media Encryption** includes a setting of **none** to allow for unencrypted media.

```

change ip-codec-set 1                                         Page 1 of 2
                                                                IP MEDIA PARAMETERS
Codec Set: 1
Audio          Silence      Frames   Packet
Codec          Suppression  Per Pkt  Size(ms)
1: OPUS-SWB24K
2: G.722-64K
3: G.711A      n                2        20
4: G.711MU    n                2        20
5: G.729A     n                2        20
Media Encryption
1: 1-srtp-aescm128-hmac80
2: none
                                                                Encrypted SRTCP: enforce-unenc-srtcp

```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form shown below as follows:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the appropriate setting, in this case it was set to **tls**.
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These values are taken from the **IP Node Names** form shown above.
- Set the **Near-end Node Name** to **procr**. This value is taken from the **IP Node Names** form shown above.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **sm101x**).
- Ensure that the recommended TLS port value of **5062** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured above. This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- **Far-end Domain** was set to the domain used during compliance testing.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** is set to **n**.
- The default values for the other fields may be used.

```

change signaling-group 1                                     Page 1 of 2
                                SIGNALING GROUP

Group Number: 1                      Group Type: sip
IMS Enabled? n                      Transport Method: tls
  Q-SIP? n
  IP Video? n                        Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                Far-end Node Name: sm101x
Near-end Listen Port: 5062              Far-end Listen Port: 5062
                                         Far-end Network Region: 1

Far-end Domain: greaney.sil6.avaya.com

Incoming Dialog Loopbacks: eliminate      Bypass If IP Threshold Exceeded? n
                                         RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload                Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3        IP Audio Hairpinning? n
Enable Layer 3 Test? Y                    Initial IP-IP Direct Media? n
                                         Alternate Route Timer(sec): 6

```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to and from InAttend. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager dial plan. Set the **Service Type** field to **tie**. Specify the signaling group associated with this trunk group in the **Signaling Group** field and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```

change trunk-group 1                                     Page 1 of 4
                                     TRUNK GROUP

Group Number: 1                                     Group Type: sip                                     CDR Reports: y
  Group Name: SIP TRK                               COR: 1                                     TN: 1       TAC: *801
  Direction: two-way                               Outgoing Display? y
  Dial Access? n                                     Night Service:
  Queue Length: 0
  Service Type: tie                                 Auth Code? n
                                               Member Assignment Method: auto
                                               Signaling Group: 1
                                               Number of Members: 10
  
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Mitel to prevent unnecessary SIP messages during call setup. Session refresh is used throughout the duration of the call, to check the other side has not gone away, for the compliance test a value of **600** was used.

```

change trunk-group 1                                     Page 2 of 4
  Group Type: sip

TRUNK PARAMETERS

  Unicode Name: auto

                                               Redirect On OPTIM Failure: 5000

  SCCAN? n                                         Digital Loss Group: 18
  Preferred Minimum Session Refresh Interval(sec): 600

  Disconnect Supervision - In?y Out? y

  XOIP Treatment: auto                               Delay Call Setup When Accessed Via IGAR? n
  
```

Settings on **Page 3** can be left as default. However, the **Numbering Format** in the example below is set to **private**.

```
change trunk-group 1                                     Page 3 of 4
                                     TRUNK FEATURES
      ACA Assignment? n                               Measured: none
                                                    Maintenance Tests? y

      Suppress # Outpulsing? n  Numbering Format: private
                                                    UII Treatment: service-provider
                                                    Replace Restricted Numbers? n
                                                    Replace Unavailable Numbers? n

                                                    Modify Tandem Calling Number: no

      Show ANSWERED BY on Display? y
```

Settings on **Page 4** are as follows; ensure that the **Telephone Event Payload Type** is set to **101**. Ensure that **Support Request History** is set to **y**.

```
change trunk-group 1                                     Page 4 of 21
                                     PROTOCOL VARIATIONS

                                                    Mark Users as Phone? n
      Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                                                    Send Transferring Party Information? y
                                                    Network Call Redirection? y
      Build Refer-To URI of REFER From Contact For NCR? n
                                                    Send Diversion Header? n
                                                    Support Request History? y
                                                    Telephone Event Payload Type: 101

                                                    Convert 180 to 183 for Early Media? n
      Always Use re-INVITE for Display Updates? n
      Identity for Calling Party Display: P-Asserted-Identity
      Block Sending Calling Party Location in INVITE? n
      Accept Redirect to Blank User Destination? n
                                                    Enable Q-SIP? n

      Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
      Request URI Contents: may-have-extra-digits
```

5.3. Configure Call Routing for InAttend

For compliance testing, all calls beginning with 450 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager and on to InAttend. To achieve this, automatic alternate routing (aar) would be used to route the calls.

5.3.1. Administer Dial Plan

It was decided for compliance testing that all calls beginning with 4 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager. Type **change dialplan analysis**, to make changes to the dial plan. Ensure that **4** is added with a **Total Length** of **4** and a **Call Type** of **udp**.

```

change dialplan analysis                                     Page 1 of 12
                                DIAL PLAN ANALYSIS TABLE
                                Location: all                Percent Full: 2
Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
String   Length  Type   String   Length  Type   String   Length  Type
1         4     ext   1         4     ext   1         4     ext
2         4     ext   2         4     ext   2         4     ext
3         4     udp   3         4     udp   3         4     udp
4         4     udp
8         1     fac   8         1     fac   8         1     fac
9         1     fac   9         1     fac   9         1     fac
*         3     fac   *         3     fac
  
```

5.3.2. Administer Route Selection for InAttend Calls

As digits **4xxx** were defined in the dial plan as **udp** (**Section 5.3.1**), use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to numbers beginning with **450** that are **4** digits in length will be matched. No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

```

change uniform-dialplan 4                                 Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                Percent Full: 0
Matching   Insert   Node
Pattern    Len Del   Digits   Net Conv Num
450      4 0   aar    n
                                n
  
```

Use the **change aar analysis x** command to further configure the routing of the dialed digits. Calls to InAttend begin with **450** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 1**, which contains the outbound SIP Trunk Group.

```

change aar analysis 4                                     Page 1 of 2
                AAR DIGIT ANALYSIS TABLE
                Location: all                             Percent Full: 1

Dialed          Total      Route      Call      Node  ANI
String          Min Max   Pattern   Type     Num   Reqd
450             4   4       1        lev0    Num   n
  
```

Use the **change route-pattern n** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, **Route Pattern Number 1** is used to route calls to trunk group (**Grp No**) **1**. This is the SIP Trunk configured in **Section 5.2**.

```

change route-pattern 1                                   Page 1 of 4
                Pattern Number: 1  Pattern Name: SIPTRK
                SCCAN? n          Secure SIP? n
  Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC
  No   Mrk Lmt List Del Digits  Dgts              QSIG
                                     Intw
1: 1   0
2:
3:
4:
5:
                                     n  user
                                     n  user
                                     n  user
                                     n  user
                                     n  user

  BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM  No.  Numbering  LAR
  0 1 2 M 4 W      Request
1: y y y y y n  n          unre
2: y y y y y n  n          rest
3: y y y y y n  n          rest
4: y y y y y n  n          rest
5: y y y y y n  n          rest
6: y y y y y n  n          rest
  
```

5.4. Configure Connection to Avaya Aura® Application Enablement Services

It is assumed that a connection to AES is already in place and that the TSAPI connection and switch connection between Communication Manager and AES is fully working. The following section outlines the connection that was setup for compliance testing.

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

Display the IP addresses by using the command **display node-names ip** and noting the IP address for the **procr** and the AES.

```
display node-names ip Page 1 of 2
```

IP NODE NAMES	
Name	IP Address
sm101x	10.10.40.12
aespri101x	10.10.40.16
aessec101x	10.10.40.46
g450	10.10.40.15
procr	10.10.40.13

5.4.2. 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 5.4.1**
- **Local Port:** Retain the default value of **8765**

```
change ip-services Page 1 of 3
```

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

Go to **Page 3** of the **ip-services** form and enter the following values:

- **AE Services Server:** Name obtained from the AES server, in this case **aespri101x**.
- **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 7.2**. The **AE Services Server** must 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				Page	3 of 3
AE Services Administration					
Server ID	AE Services Server	Password	Enabled	Status	
1:	aespri101x	*****	y	idle	
2:					
3:					

5.4.3. Configure CTI Link for TSAPI Service

Add a CTI link using the **add cti-link n** command, where n is the n is the cti-link number as shown in the example below this is **1**. 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		Page	1 of 3
CTI LINK			
CTI Link: 1			
Extension: 1990			
Type: ADJ-IP			
Name: aespri101x			COR: 1

5.5. Configure VDNs and Vectors for InAttend

There are two VDNs and two Vectors that need to be added to allow InAttend set the status of a user on Communication Manager using TSAPI. VDN A calls on Vector A which collects digits into VDN B which is monitored by InAttend as per **Section 8.5**.

5.5.1. Adding VDNs

There are two VDNs that are added one to collect digits and one to monitor the collected digits. Use the command **add vdn x**, where x is the vdn to be added. Each VDN uses a Vector which are outlined in **Section 5.5.2**.

```
add vdn 3082                                     Page 1 of 3
                                         VECTOR DIRECTORY NUMBER
                                         Extension: 3082                Unicode Name? n
                                         Name*: Diversion CMG
                                         Destination: Vector Number      22
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*:
                        2nd Skill*:
                        3rd Skill*:
SIP URI:
* Follows VDN Override Rules
```

Same command is used to **add VDN 3084** and this will use Vector **21**. This VDN is then referenced in **Section 8.5**.

```
add vdn 3084                                     Page 1 of 3
                                         VECTOR DIRECTORY NUMBER
                                         Extension: 3084                Unicode Name? n
                                         Name*: Hangup
                                         Destination: Vector Number      21
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*:
                        2nd Skill*:
                        3rd Skill*:
SIP URI:
* Follows VDN Override Rules
```

5.5.2. Adding Vectors

VDN 1082 on the previous page uses this **Vector 22** to collect up to **8 digits** and then routes the call to the other VDN 1084 configured again on the previous page in **Section 5.5.1**.

```
change vector 22                                     Page 1 of 6
                                                    CALL VECTOR
Number: 22                                           Name: Diversion CMG
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
  Variables? y      3.0 Enhanced? y
01 wait-time      0      secs hearing ringback
02 collect      8      digits after announcement none      for none
03 wait-time      2      secs hearing ringback
04 route-to      number 3084                        cov n if unconditionally
05
06
07
08
09
10
```

VDN 3084 uses the following Vector which simply provides **ringback** to the user while the VDN is being monitored.

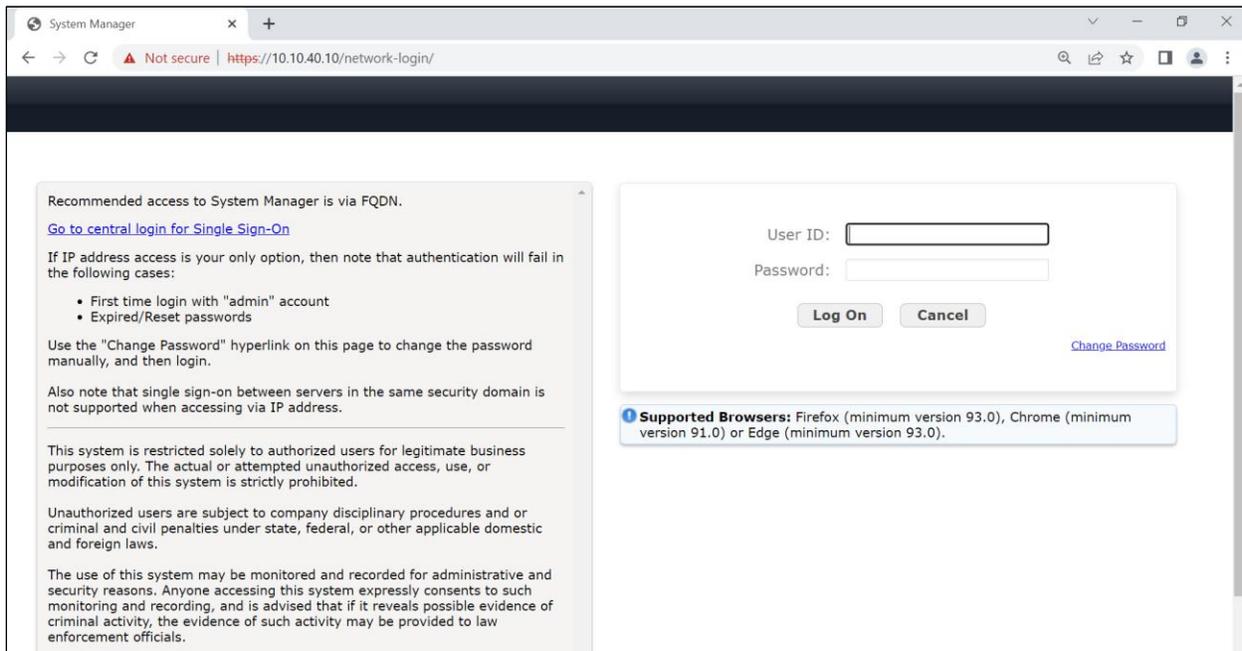
```
change vector 21                                     Page 1 of 6
                                                    CALL VECTOR
Number: 21                                           Name: Diversion 2
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
  Variables? y      3.0 Enhanced? y
01 wait-time      60      secs hearing ringback
02 stop
03
04
05
06
07
08
09
10
```

6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager to add the SIP Entity and routing to allow calls route to and from Mitel InAttend. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Configure SIP Entity
- Configure Entity Link
- Configure Routing Policy
- Configure Dial Pattern

To make changes on Session Manager a web session is established to System Manager. Log into System Manager by opening a web browser and navigating to <https://<System Manager FQDN>/SMGR>. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On**.



System Manager

Not secure | <https://10.10.40.10/network-login/>

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

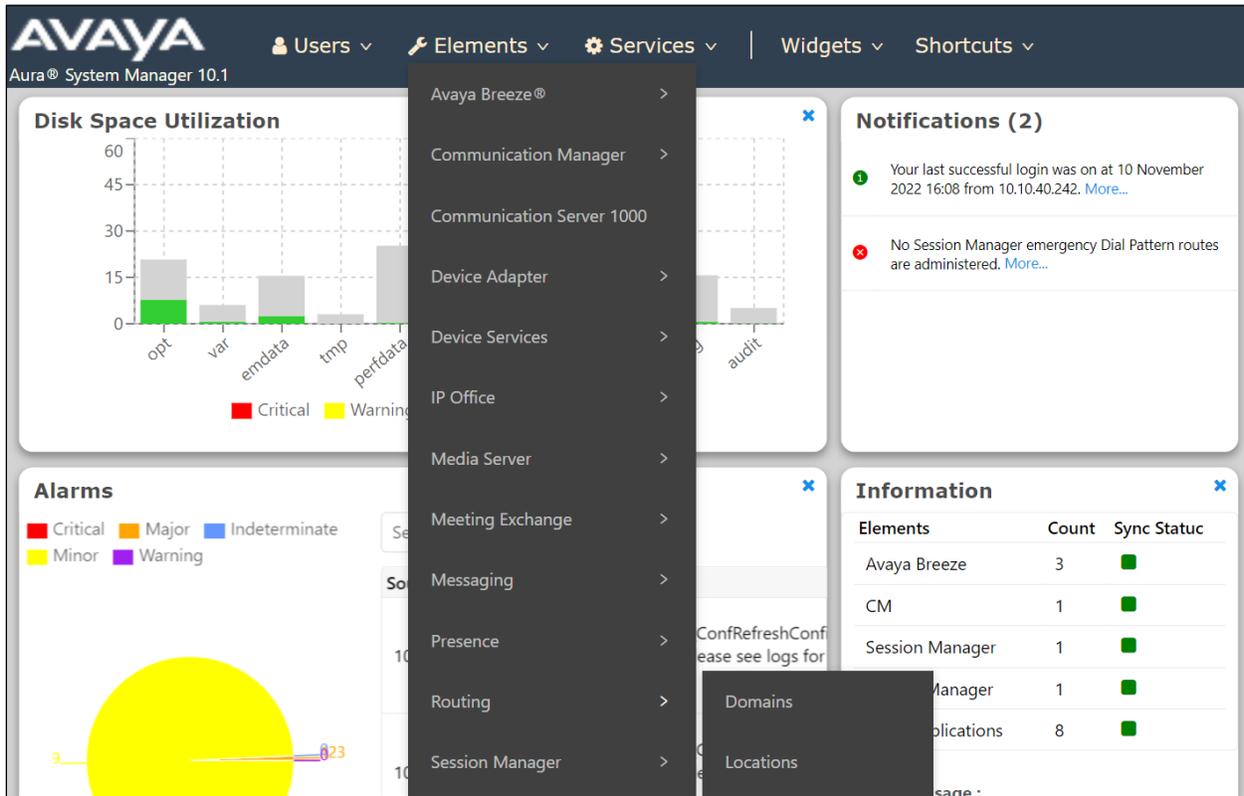
User ID:

Password:

Log On **Cancel** [Change Password](#)

Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

Once logged in navigate to **Elements** and click on **Routing**, as shown below.

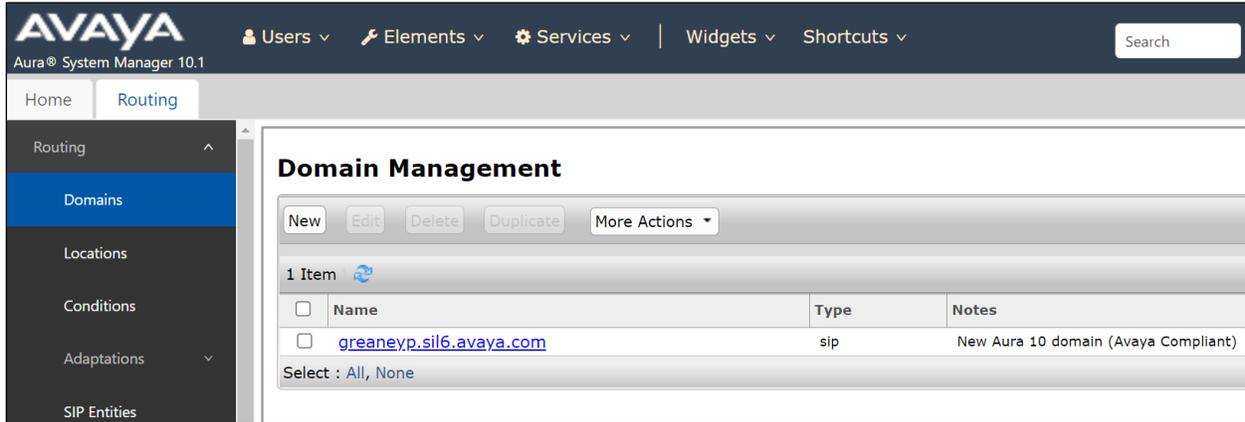


6.1. Domains and Locations

Note: It is assumed that a domain and a location have already been configured, therefore a quick overview of the domain and location that was used in compliance testing is provided here.

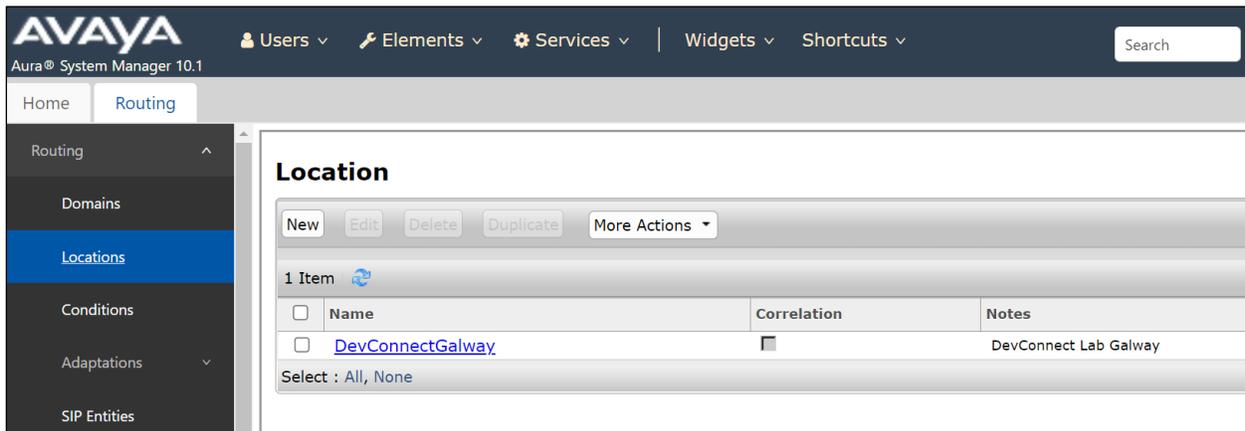
6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **greaneyp.sil6.avaya.com** as shown below. If a domain is not already in place, click on **New**. This will open a new window (not shown) where the domain can be added.



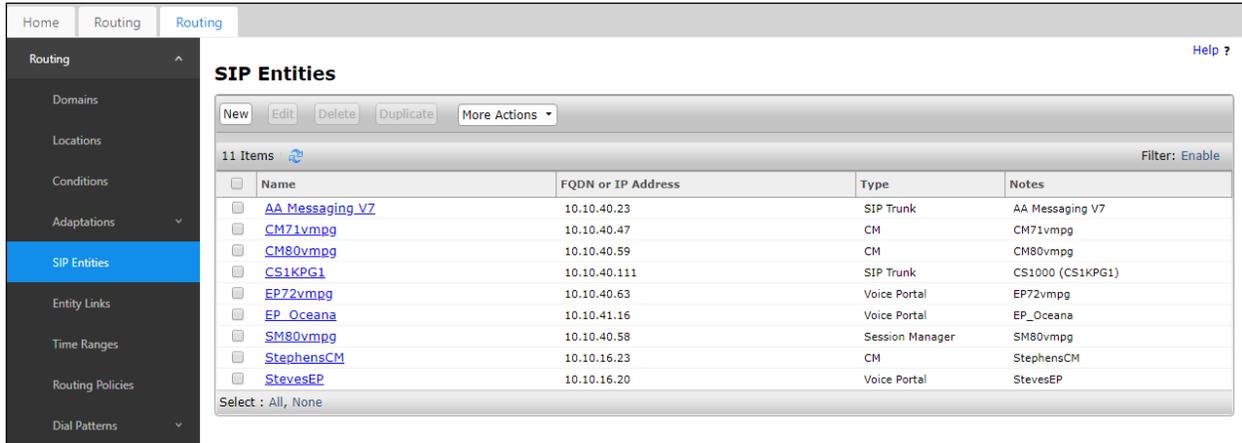
6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectGalway** which was used for compliance testing. If a location is not already in place, then one must be added to include the IP address range of the Avaya solution. Click on **New** to add a new location.



6.2. Configure Mitel InAttend SIP Entity

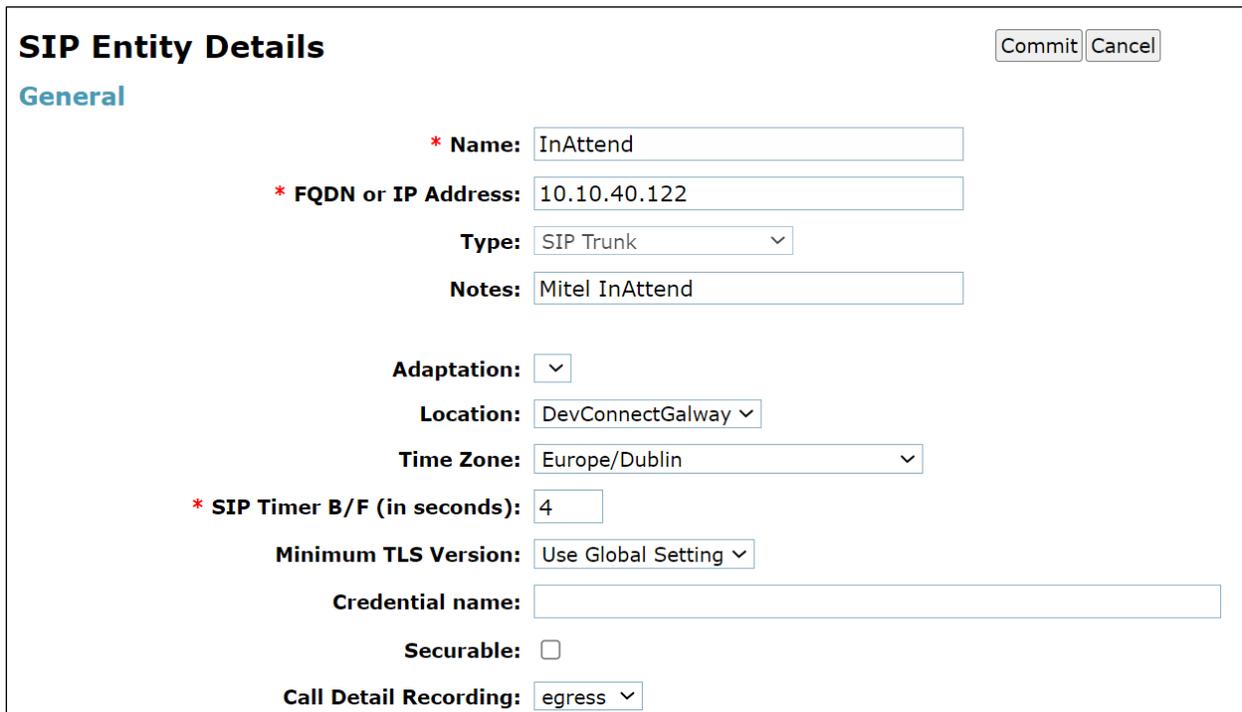
The ACS (also referred to as InAttend) is added on Session Manager as a SIP Entity with an Entity Link. Every SIP endpoint that communicated over a SIP trunk would be added as such. Click on **SIP Entities** in the left column and select **New** in the right window.



The screenshot shows the 'SIP Entities' management page. The left sidebar has 'SIP Entities' selected. The main area displays a table with 11 items. The table columns are Name, FQDN or IP Address, Type, and Notes. Below the table is a 'Select' dropdown set to 'All, None'.

Name	FQDN or IP Address	Type	Notes
AA Messaging V7	10.10.40.23	SIP Trunk	AA Messaging V7
CM71vmppg	10.10.40.47	CM	CM71vmppg
CM80vmppg	10.10.40.59	CM	CM80vmppg
CS1KPG1	10.10.40.111	SIP Trunk	CS1000 (CS1KPG1)
EP72vmppg	10.10.40.63	Voice Portal	EP72vmppg
EP_Oceana	10.10.41.16	Voice Portal	EP_Oceana
SM80vmppg	10.10.40.58	Session Manager	SM80vmppg
StephensCM	10.10.16.23	CM	StephensCM
StevesEP	10.10.16.20	Voice Portal	StevesEP

Enter a suitable **Name** for the new SIP Entity and the **IP Address** of the ACS. Enter the correct **Time Zone** and **Location** and scroll down to SIP Entity Links.



The screenshot shows the 'SIP Entity Details' configuration form. The 'General' tab is active. The form contains several fields and dropdown menus for configuring the SIP entity.

SIP Entity Details Commit Cancel

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

* **SIP Timer B/F (in seconds):**

Minimum TLS Version:

Credential name:

Securable:

Call Detail Recording:

6.3. Configure Mitel InAttend SIP Entity Link

An Entity link can be added from the SIP Entities page. Using the page from the previous page scroll down to Entity Links.

Upon scrolling down to **Entity Links** click on **Add**.

The screenshot shows the configuration interface for SIP Entity Links. At the top, there are two dropdown menus for 'Primary Session Manager Bandwidth Association' and 'Backup Session Manager Bandwidth Association'. Below these is the 'Entity Links' section with an 'Override Port & Transport with DNS SRV' checkbox. The 'Add' button is highlighted with a red box. Below the 'Add' button is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Deny New Service. Below the table is a section for 'SIP Responses to an OPTIONS Request' with 0 items and a table with columns: Response Code & Reason Phrase, Mark Entity Up/Down, and Notes.

Enter a suitable **Name** for the Entity Link and select the **Session Manager SIP Entity** for **SIP Entity 1** and the newly created InAttend SIP Entity for **SIP Entity 2**. Ensure that **TCP** is selected for the **Protocol** and that **Port 5060** is used. Click on **Commit** once finished to save the new Entity Link.

The screenshot shows the configuration interface for SIP Entity Links. At the top, there are two dropdown menus for 'Primary Session Manager Bandwidth Association' and 'Backup Session Manager Bandwidth Association'. Below these is the 'Entity Links' section with an 'Override Port & Transport with DNS SRV' checkbox. The 'Add' button is highlighted with a red box. Below the 'Add' button is a table with 1 item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Deny New Service. The configuration details for the new Entity Link are: Name: *sm101x_InAttend_5060, SIP Entity 1: sm101x, Protocol: TCP, Port: *5060, SIP Entity 2: InAttend, Port: *5060. Below the table is a section for 'SIP Responses to an OPTIONS Request' with 0 items and a table with columns: Response Code & Reason Phrase, Mark Entity Up/Down, and Notes.

6.4. Configure Routing Policy for Mitel InAttend

Click on **Routing Policies** in the left window and select **New** in the main window.

<input type="checkbox"/>	Name	Disabled	Retries	Destination	Notes
<input type="checkbox"/>	To AA Messaging VZ	<input type="checkbox"/>	0	AA Messaging V7	To AA Messaging V7
<input type="checkbox"/>	To ASCBE	<input type="checkbox"/>	0	ASBCE8vmpg	To Session Border Controller
<input type="checkbox"/>	To Capita DMS	<input type="checkbox"/>	0	Capita DMS	To Capita DMS
<input type="checkbox"/>	To Capita_DS3000	<input type="checkbox"/>	0	Capita DS3000	To Capita DS3000
<input type="checkbox"/>	To_CM71vmpg	<input type="checkbox"/>	0	CM71vmpg	To CM71vmpg
<input type="checkbox"/>	To_CM80vmpg	<input type="checkbox"/>	0	CM80vmpg	To CM80vmpg
<input type="checkbox"/>	To_CS1KPG1	<input type="checkbox"/>	0	CS1KPG1	To CS1KPG1
<input type="checkbox"/>	To_EP72vmpg	<input type="checkbox"/>	0	EP72vmpg	To EP72vmpg
<input type="checkbox"/>	To_EP_Oceana	<input type="checkbox"/>	0	EP_Oceana	To EP Oceana
<input type="checkbox"/>	To_Stephens_CM	<input type="checkbox"/>	0	StephensCM	To StephensCM
<input type="checkbox"/>	To_Steves_EP	<input type="checkbox"/>	0	StevesEP	To Steves EP

Enter a suitable **Name** for the Routing Policy and click on **Select** under **SIP Entity as Destination**, highlighted on next page.

Routing Policy Details

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type
------	--------------------	------

Select the **InAttend** SIP Entity as shown below and click on **Select**.

SIP Entities				
SIP Entities				
13 Items				Filter: Enable
	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	AACC	10.10.40.96	SIP Trunk	
<input type="radio"/>	breeze1wspaces	10.10.40.52	Avaya Breeze	Breeze 1 for wspaces
<input type="radio"/>	breeze2wspaces	10.10.40.53	Avaya Breeze	Breeze 2 for wspaces
<input type="radio"/>	breeze3wspaces	10.10.40.54	Avaya Breeze	Breeze 3 for wspaces
<input type="radio"/>	cm101x - Phones - 5061	10.10.40.13	CM	For SIP PHONES on CM
<input type="radio"/>	cm101x - SIM PSTN - 5063	10.10.40.13	CM	For Simulated SIP Trunk
<input type="radio"/>	cm101x - SIP TRUNK - 5062	10.10.40.13	CM	SIP Trunk in and out
<input type="radio"/>	Experience Portal-MPP	10.10.40.26	Voice Portal	Experience Portal
<input checked="" type="radio"/>	InAttend	10.10.40.122	SIP Trunk	Mitel InAttend
<input type="radio"/>	IP Office - SE	10.10.40.19	SIP Trunk	IP Office Server Edition
<input type="radio"/>	Messaging2019	10.10.40.75	SIP Trunk	To messaging on win 2019

The selected destination is now shown, click on **Commit** to save this.

Routing Policy Details			
General			
* Name:	<input type="text" value="To InAttend"/>		
Disabled:	<input type="checkbox"/>		
* Retries:	<input type="text" value="0"/>		
Notes:	<input type="text" value="To InAttend"/>		
SIP Entity as Destination			
<input type="button" value="Select"/>			
Name	FQDN or IP Address	Type	Notes
InAttend	10.10.40.122	SIP Trunk	Mitel InAttend
Time of Day			
<input type="button" value="Add"/>	<input type="button" value="Remove"/>	<input type="button" value="View Gaps/Overlaps"/>	
1 Item			Filter: Enable

6.5. Configure Mitel InAttend Dial Pattern

Select **Dial Patterns** in the left window and select **New** in the main window.

Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
160	4	4	<input type="checkbox"/>			greaney.p.sil6.avaya.com	ToEP810
3	4	4	<input type="checkbox"/>			greaney.p.sil6.avaya.com	3xxx route to CM101x
3201	4	4	<input type="checkbox"/>			greaney.p.sil6.avaya.com	To NovaAlert
3539173	11	11	<input type="checkbox"/>			greaney.p.sil6.avaya.com	To CM101x from SIM PSTN
3539184	11	11	<input type="checkbox"/>			greaney.p.sil6.avaya.com	To Simulated PSTN
5	4	4	<input type="checkbox"/>			-ALL-	To IP Office SE
6667	4	4	<input type="checkbox"/>			greaney.p.sil6.avaya.com	To Messaging2019 on Win 2019
68	4	4	<input type="checkbox"/>			greaney.p.sil6.avaya.com	To AACC
95	5	5	<input type="checkbox"/>			-ALL-	ToCM10

Enter the required digits for the Routing Pattern, in the example below **450** is used. This ensures that when 450x is dialed it will route to the ACS. Enter the appropriate domain for **SIP Domain** in this example the domain created in **Section 6.1.1** is added. Click on **Add** under **Originating Locations and Routing Policies** to select this Routing Policy.

Dial Pattern Details [Commit] [Cancel]

General

* Pattern:

* Min:

* Max:

Emergency Call:

SIP Domain:

Notes:

Originating Locations and Routing Policies

[Add] [Remove]

1 Item [Filter: Enable]

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes

Select the **Originating Location**, this will be the location added in **Section 6.1.2** and select the newly created Routing Policy for InAttend.

Originating Location

Apply The Selected Routing Policies to All Originating Locations

1 Item Filter: Enable

<input checked="" type="checkbox"/>	Name	Notes
<input checked="" type="checkbox"/>	DevConnectGalway	DevConnect Lab Galway

Select : All, None

Routing Policies

10 Items Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	To AACC	<input type="checkbox"/>	AACC	To AACC
<input type="checkbox"/>	To cm101x - SIM PSTN	<input type="checkbox"/>	cm101x - SIM PSTN - 5063	Calls from SIM PSTN
<input type="checkbox"/>	To cm101x - SIP Phones	<input type="checkbox"/>	cm101x - Phones - 5061	Route to CM101x - SIP Phones
<input type="checkbox"/>	To cm101x - SIP Trunk	<input type="checkbox"/>	cm101x - SIP TRUNK - 5062	Route to CM101x - SIP Trunk
<input type="checkbox"/>	ToEP810	<input type="checkbox"/>	Experience Portal-MPP	ToEP810
<input checked="" type="checkbox"/>	To InAttend	<input type="checkbox"/>	InAttend	To InAttend
<input type="checkbox"/>	To IP Office SE	<input type="checkbox"/>	IP Office - SE	To IP Office SE
<input type="checkbox"/>	To Messaging2019	<input type="checkbox"/>	Messaging2019	To Messaging on Win 2019
<input type="checkbox"/>	To NovaAlert	<input type="checkbox"/>	novaalert	To NovaAlert
<input type="checkbox"/>	To SIM PSTN	<input type="checkbox"/>	SBCE - SIM - PSTN	Simulated PSTN

With the Routing Policy selected, click on **Commit** (not shown) to finish adding the Dial Pattern.

General

* Pattern:

* Min:

* Max:

Emergency Call:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	DevConnectGalway	DevConnect Lab Galway	To InAttend	0	<input type="checkbox"/>	InAttend	To InAttend

Select : All, None

Denied Originating Locations

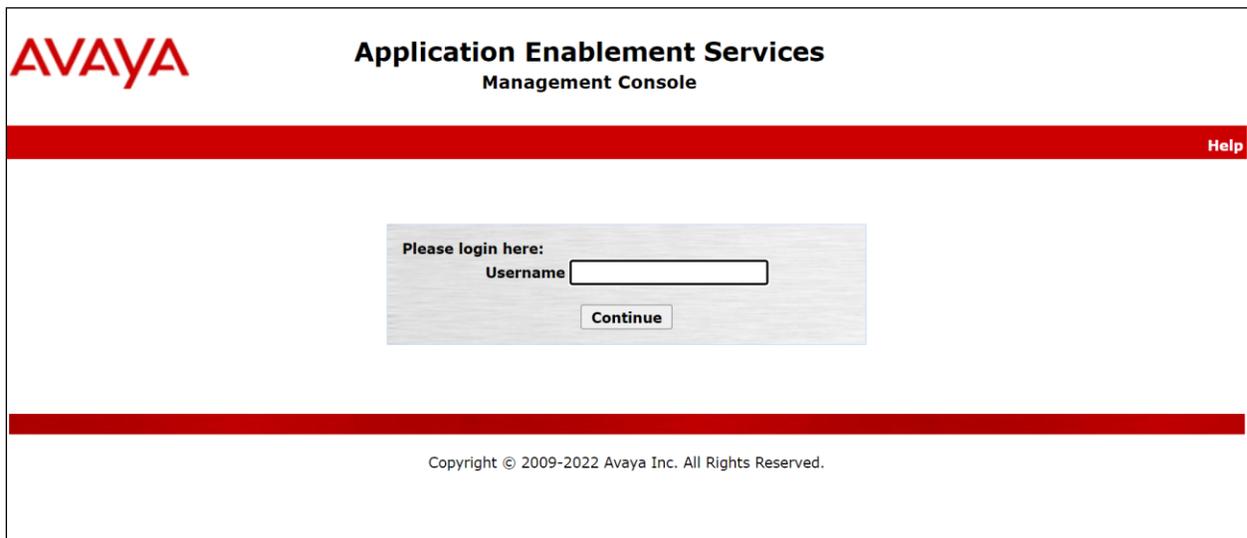
7. Configure Avaya Aura® Application Enablement Services

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

- Verify Licensing
- Switch Connection
- Administer TSAPI Link
- Identify Tlinks
- Enable TSAPI Ports
- Create CTI User
- Configure Security
- Restart AE Server

7.1. Verify Licensing

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



The screenshot shows the Avaya Application Enablement Services Management Console login page. At the top left is the Avaya logo. To its right, the text reads "Application Enablement Services Management Console". A red horizontal bar spans the width of the page, with the word "Help" in white text on the right side. In the center of the page, there is a login box with a light gray background. Inside this box, it says "Please login here:" followed by "Username" and a text input field. Below the input field is a "Continue" button. At the bottom of the page, a red horizontal bar is present, and below it, the copyright notice "Copyright © 2009-2022 Avaya Inc. All Rights Reserved." is displayed.

The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the TSAPI Service is licensed by ensuring that **TSAPI Service** is in the list of **Services** and that the **License Mode** is showing **NORMAL MODE**. If not, contact an Avaya support representative to acquire the appropriate license.

The screenshot shows the 'AE Services' management console. On the left is a navigation menu with 'AE Services' expanded. The main content area is titled 'AE Services' and contains an important notice: 'IMPORTANT: AE Services must be restarted for administrative changes to fully take effect. Changes to the Security Database do not require a restart.' Below this is a table with the following data:

Service	Status	State	License Mode	Cause*
ASAI Link Manager	N/A	Running	N/A	N/A
CVLAN Service	OFFLINE	Running	N/A	N/A
DLG Service	OFFLINE	Running	N/A	N/A
DMCC Service	ONLINE	Running	NORMAL MODE	N/A
TSAPI Service	ONLINE	Running	NORMAL MODE	N/A
Transport Layer Service	N/A	Running	N/A	N/A
AE Services HA	Not Configured	N/A	N/A	N/A

Below the table, there is a note: 'For status on actual services, please use [Status and Control](#)'. A footnote states: '* -- For more detail, please mouse over the Cause, you'll see the tooltip, or go to help page.' At the bottom, 'License Information' states: 'You are licensed to run Application Enablement (CTI) release 8.x'.

The TSAPI licenses are user licenses issued by the Web License Manager to which the Application Enablement Services server is pointed to. From the left window open **Licensing** and click on **WebLM Server Access** as shown below.

The screenshot shows the 'Licensing' management console. The left navigation menu has 'Licensing' selected. The main content area is titled 'Licensing' and contains the following instructions:

If you are setting up and maintaining the WebLM, you need to use the following:

- WebLM Server Address

If you are importing, setting up and maintaining the license, you need to use the following:

- WebLM Server Access

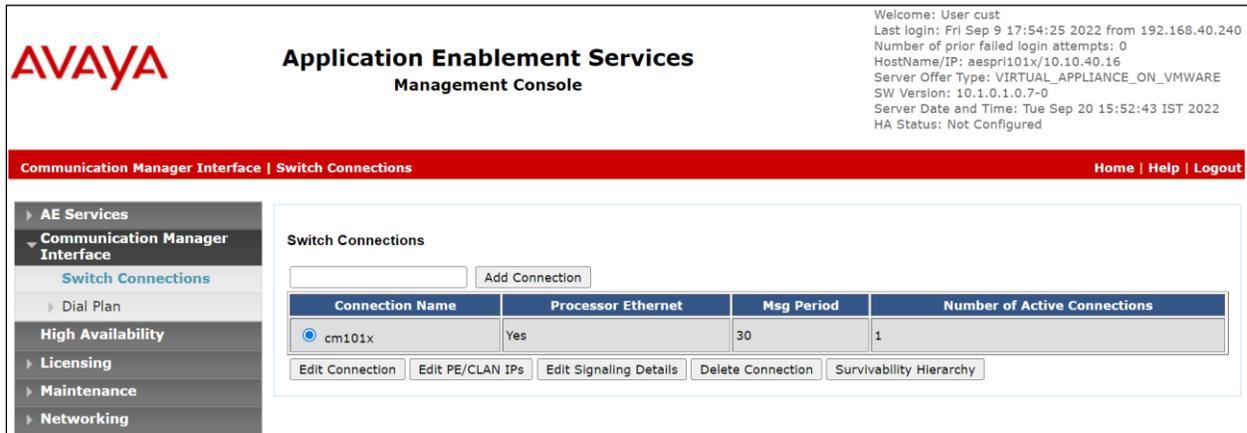
If you want to administer TSAPI Reserved Licenses or DMCC Reserved Licenses, you need to use the following:

- Reserved Licenses

NOTE: Please disable your pop-up blocker if you are having difficulty with opening this page

7.2. Create Switch Connection

Typically, the connection between the Application Enablement Services and Communication Manager is setup as part of the initial installation and would not usually be outlined in these Application Notes. From the AES Management Console navigate to **Communication Manager Interface** → **Switch Connections**, the connection to Communication Manager should be present as shown below but if one is not present one can be added by clicking on **Add Connection**.



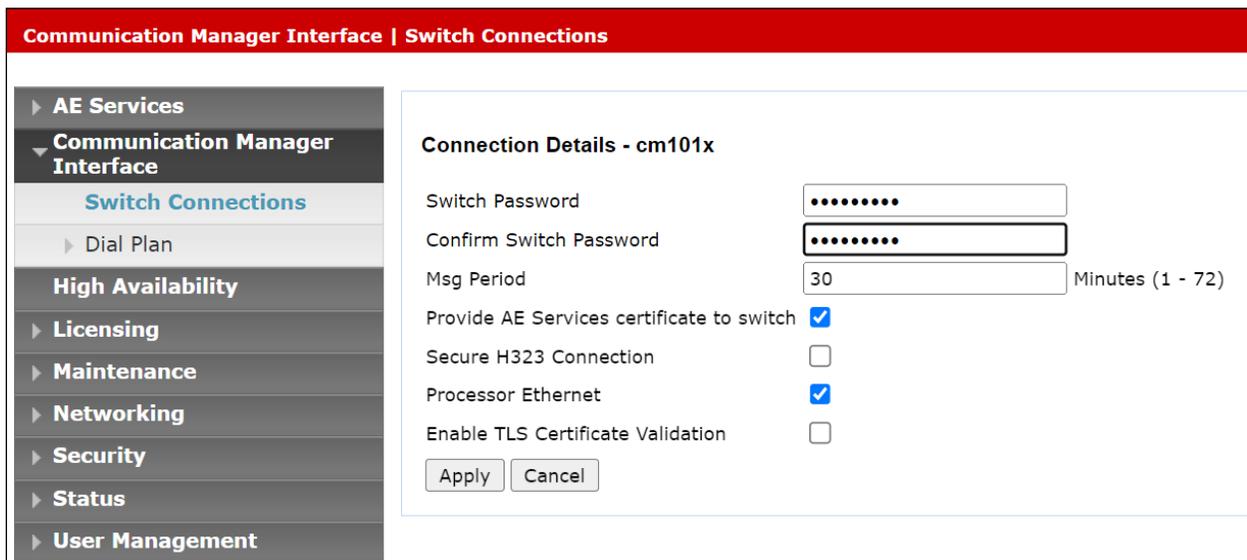
Welcome: User cust
Last login: Fri Sep 9 17:54:25 2022 from 192.168.40.240
Number of prior failed login attempts: 0
HostName/IP: aespri101x/10.10.40.16
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 10.1.0.1.0.7-0
Server Date and Time: Tue Sep 20 15:52:43 IST 2022
HA Status: Not Configured

Communication Manager Interface | Switch Connections Home | Help | Logout

Switch Connections

Connection Name	Processor Ethernet	Msg Period	Number of Active Connections
<input checked="" type="radio"/> cm101x	Yes	30	1

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 ip-services** command, described in **Section 5.4.2. Secure H323 Connection** was left unticked, as shown below. Click **Apply** to save changes.



Communication Manager Interface | Switch Connections

Connection Details - cm101x

Switch Password

Confirm Switch Password

Msg Period Minutes (1 - 72)

Provide AE Services certificate to switch

Secure H323 Connection

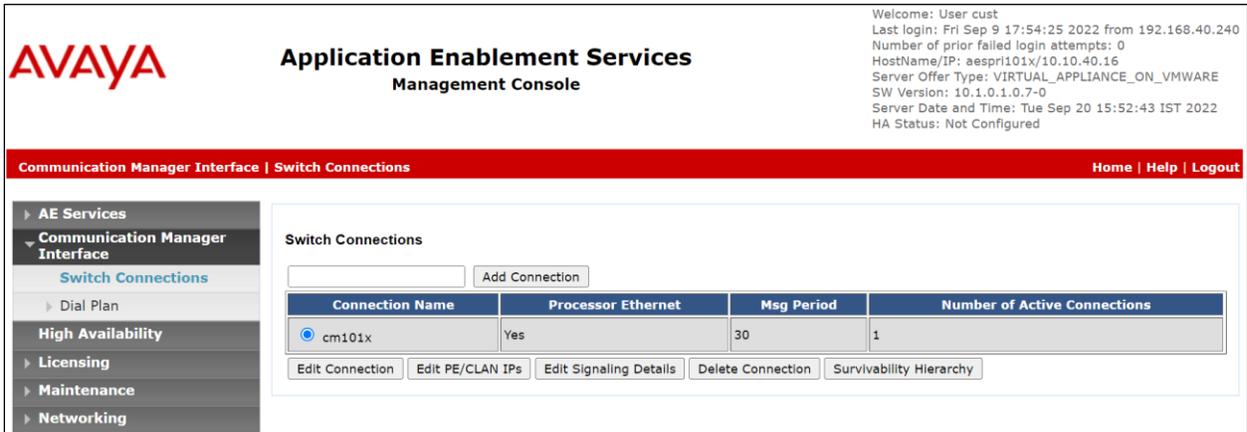
Processor Ethernet

Enable TLS Certificate Validation

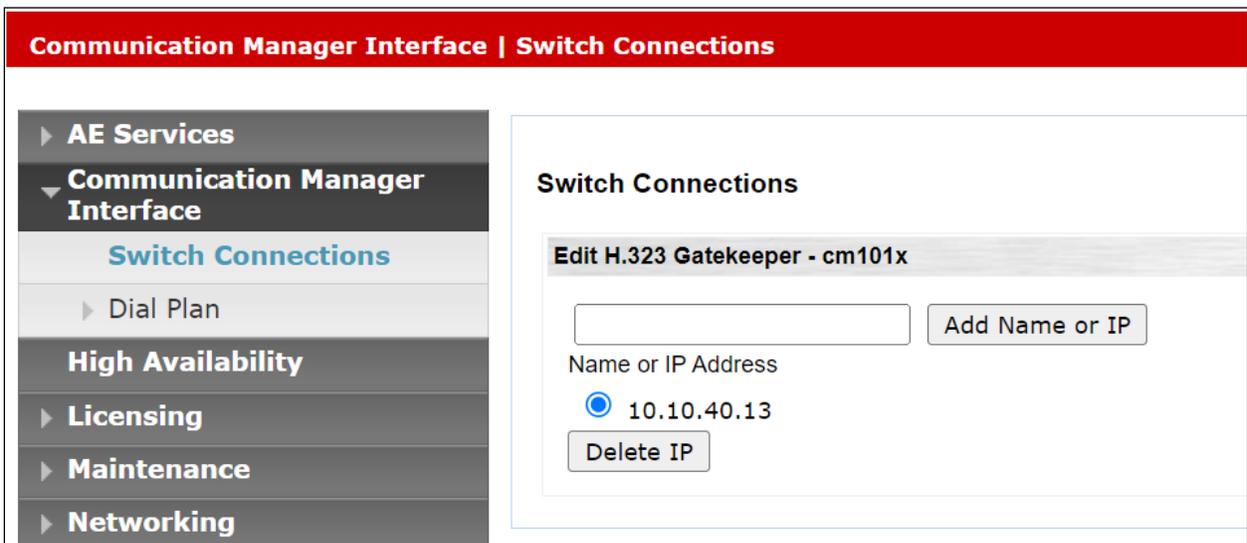
From the **Switch Connections** screen, select the radio button for the recently added switch connection and select the **Edit PE/CLAN IPs** button (not shown), see screen at the bottom of the previous page. In the resulting screen, enter the IP address of the procr, as shown in **Section 5.4.1**, that will be used for the AES connection and select the **Add/Edit Name or IP** button.



Clicking on **Edit Signaling Details** below brings up the H.323 Gatekeeper page.

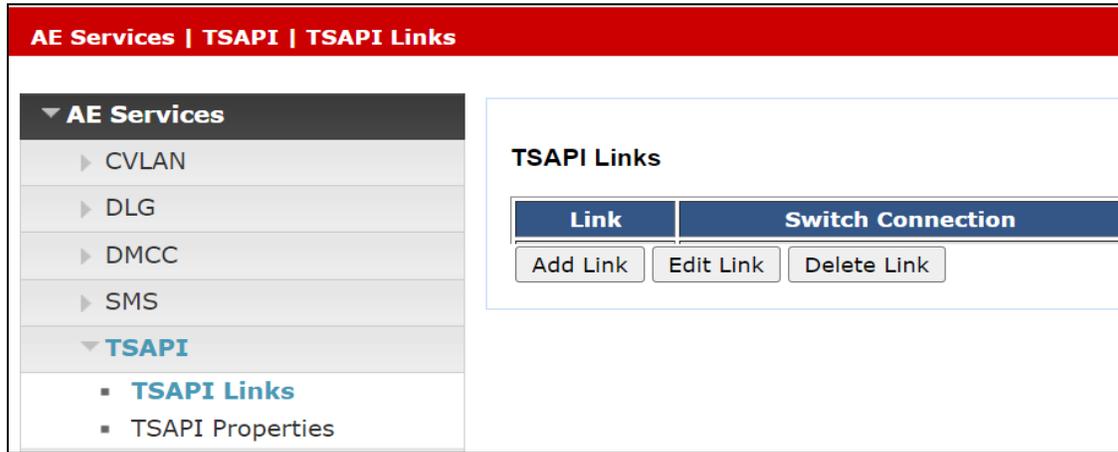


The IP address of Communication Manager is set for the **H.323 Gatekeeper**, as shown below.



7.3. Administer TSAPI link

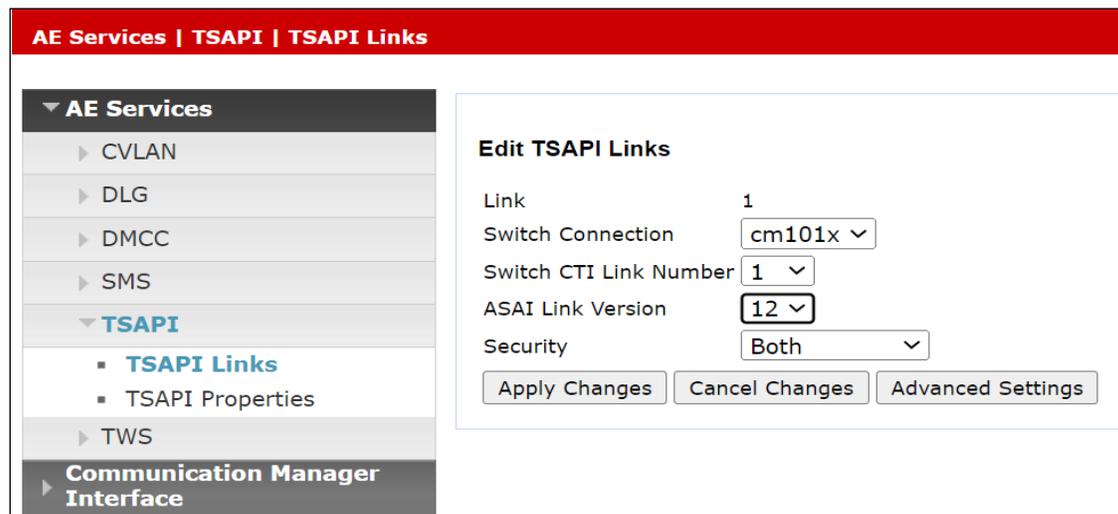
From the Application Enablement Services Management Console, select **AE Services** → **TSAPI** → **TSAPI Links**. Select **Add Link** button as shown in the screen below.



On the **Add TSAPI Links** screen (or the **Edit TSAPI Links** screen to edit a previously configured TSAPI Link as shown below), enter the following values:

- **Link:** Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm101x**, which has already been configured in **Section 7.2** from the drop-down list.
- **Switch CTI Link Number:** Corresponding CTI link number configured in **Section 5.4.3** which is **1**.
- **ASAI Link Version:** **12** was used for compliance testing but the latest version available can be chosen).
- **Security:** This can be left at the default value of **Both**.

Once completed, select **Apply Changes**.



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

Apply Changes to Link

Warning! Are you sure you want to apply the changes?
These changes can only take effect when the TSAPI server restarts.

 **Please use the Maintenance -> Service Controller page to restart the TSAPI server.**

When the TSAPI Link is completed, it should resemble the screen below.

TSAPI Links

Link	Switch Connection	Switch CTI Link #	ASAI Link Version	Security
<input checked="" type="radio"/> 1	cm101x	1	12	Both

7.4. Identify Tlinks

Navigate to **Security** → **Security Database** → **Tlinks**. Verify the value of the **Tlink Name**. This will be needed to configure Mitel Collaboration Management (CMG) module in **Section 8.5**.

Security | Security Database | Tlinks

- ▶ AE Services
- ▶ Communication Manager Interface
- High Availability
- ▶ Licensing
- ▶ Maintenance
- ▶ Networking
- ▼ **Security**
 - ▶ Account Management
 - ▶ Audit
 - ▶ Certificate Management
 - Enterprise Directory
 - ▶ Host AA
 - ▶ PAM
 - ▼ **Security Database**
 - Control
 - ⊕ CTI Users
 - Devices
 - Device Groups
 - **Tlinks**
 - Tlink Groups
 - Worktops

Tlinks

Tlink Name

AVAYA#CM101X#CSTA#AESPRI101X

AVAYA#CM101X#CSTA-S#AESPRI101X

Delete Tlink

7.5. Enable TSAPI Ports

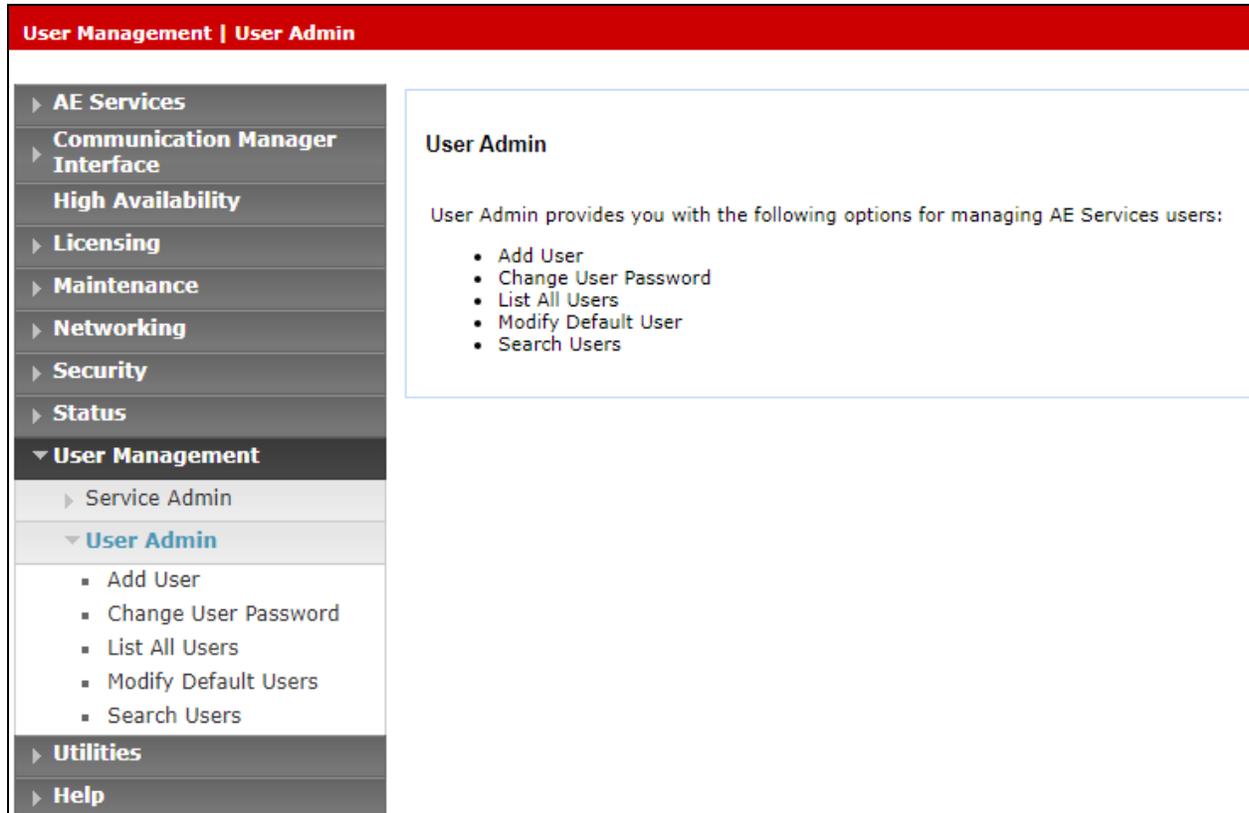
To ensure that TSAPI ports are enabled, navigate to **Networking** → **Ports**. Ensure that the TSAPI ports are set to **Enabled** as shown below.

▶ AE Services			
▶ Communication Manager Interface			
High Availability			
▶ Licensing			
▶ Maintenance			
▼ Networking			
AE Service IP (Local IP)			
Network Configure			
Ports			
TCP/TLS Settings			
▶ Security			
▶ Status			
▶ User Management			
▶ Utilities			
▶ Help			

Ports			
CVLAN Ports			
Unencrypted TCP Port	9999		Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>
Encrypted TCP Port	<input type="text" value="9998"/>		Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>
DLG Port			
TCP Port	5678		
TSAPI Ports			
TSAPI Service Port	450	Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>	
Local TLINK Ports			
TCP Port Min	1024		
TCP Port Max	1039		
Unencrypted TLINK Ports			
TCP Port Min	<input type="text" value="1050"/>		
TCP Port Max	<input type="text" value="1065"/>		
Encrypted TLINK Ports			
TCP Port Min	<input type="text" value="1066"/>		
TCP Port Max	<input type="text" value="1081"/>		
DMCC Server Ports			
Unencrypted Port	<input type="text" value="4721"/>	Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>	
Encrypted Port	<input type="text" value="4722"/>	Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>	
TR/87 Port	<input type="text" value="4723"/>	Enabled <input checked="" type="radio"/> Disabled <input type="radio"/>	

7.6. Create CTI User

A User ID and password needs to be configured for InAttend Server module to communicate with the Application Enablement Services server. Navigate to the **User Management** → **User Admin** screen then choose the **Add User** option.



The screenshot displays the 'User Management | User Admin' interface. On the left is a navigation menu with the following items: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security, Status, User Management (expanded), Service Admin, User Admin (expanded), Utilities, and Help. Under 'User Admin', the following options are listed: Add User, Change User Password, List All Users, Modify Default Users, and Search Users. The main content area on the right is titled 'User Admin' and contains the text: 'User Admin provides you with the following options for managing AE Services users:' followed by a bulleted list: Add User, Change User Password, List All Users, Modify Default User, and Search Users.

In the **Add User** screen shown below, enter the following values:

- **User Id** - This will be used by InAttend Server module in **Section 8.6**.
- **Common Name** and **Surname** - Descriptive names need to be entered.
- **User Password** and **Confirm Password** - This will be used with the InAttend Server module in **Section 8.6**.
- **CT User** - Select **Yes** from the drop-down menu.

Click on **Apply Changes** at the bottom of the screen (not shown).

High Availability	* User Id	<input type="text" value="mitel"/>
▶ Licensing	* Common Name	<input type="text" value="mitel"/>
▶ Maintenance	* Surname	<input type="text" value="mitel"/>
▶ Networking	User Password	<input type="password" value="....."/>
▶ Security	Confirm Password	<input type="password" value="....."/>
▶ Status	Admin Note	<input type="text"/>
▼ User Management	Avaya Role	<input type="text" value="None"/> ▼
▶ Service Admin	Business Category	<input type="text"/>
▼ User Admin	Car License	<input type="text"/>
▪ Add User	CM Home	<input type="text"/>
▪ Change User Password	Css Home	<input type="text"/>
▪ List All Users	CT User	<input type="text" value="Yes"/> ▼
▪ Modify Default Users	Department Number	<input type="text"/>
▪ Search Users	Display Name	<input type="text"/>
▶ Utilities	Employee Number	<input type="text"/>
▶ Help	Employee Type	<input type="text"/>
	Enterprise Handle	<input type="text"/>
	Given Name	<input type="text"/>
	Home Phone	<input type="text"/>
	Home Postal Address	<input type="text"/>

7.7. Configure Security

The CTI user permissions and the database security are set under **Security Database**.

7.7.1. Configure Database Control

The security database can be set differently depending on the requirements of the customer in question. For compliance testing, the DevConnect lab was setup as shown below, however this may be changed by opening **Control** and ticking the boxes shown.

The screenshot shows a configuration window for the Security Database. On the left is a navigation tree with the following items: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance, Networking, Security (expanded), Account Management, Audit, Certificate Management, Enterprise Directory, Host AA, PAM, Security Database (expanded), Control (selected), and CTI Users. The main content area is titled "SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services" and contains two checkboxes: "Enable SDB for DMCC Service" (unchecked) and "Enable SDB for TSAPI Service, JTAPI and Telephony Web Services" (checked). Below the checkboxes is an "Apply Changes" button.

Note: The AES Security Database (SDB) provides the ability to control a user's access privileges. The SDB stores information about Computer Telephony (CT) users and the devices they control. The DMCC service, the TSAPI service, and Telephony Web Services use this information for permission checking. Please look to **Section 11** for more information on this.

7.7.2. Associate Devices with CTI User

Navigate to **Security** → **Security Database** → **CTI Users** → **List All Users**. Select the CTI user added in **Section 7.6** and click on **Edit**.

User ID	Common Name	Worktop Name	Device ID
<input type="radio"/> asc	asc	NONE	NONE
<input checked="" type="radio"/> mitel	mitel	NONE	NONE
<input type="radio"/> nice1	nice1	NONE	NONE
<input type="radio"/> paul1	paul1	NONE	NONE
<input type="radio"/> paul2	paul2	NONE	NONE
<input type="radio"/> sytel	Sytel	NONE	NONE

In the main window ensure that **Unrestricted Access** is ticked. Once this is done click on **Apply Changes**.

Edit CTI User

User Profile: User ID: mitel
 Common Name: mitel
 Worktop Name: NONE ▾
 Unrestricted Access:

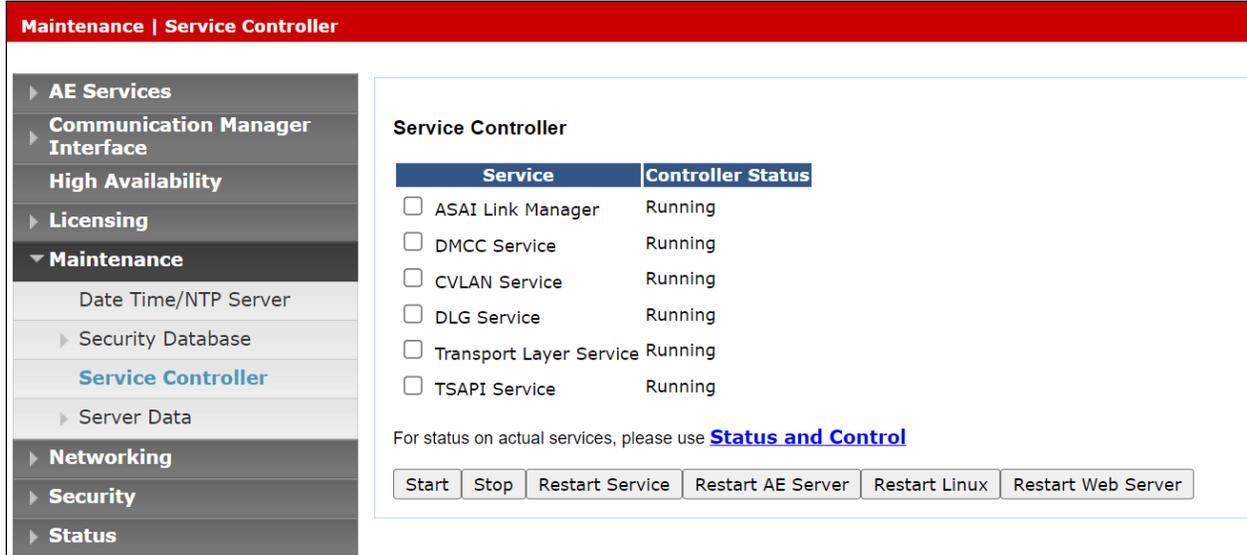
Call and Device Control: Call Origination/Termination and Device Status: None ▾

Call and Device Monitoring: Device Monitoring: None ▾
 Calls On A Device Monitoring: None ▾
 Call Monitoring:

Routing Control: Allow Routing on Listed Devices: None ▾

7.8. Restart AE Server

Once everything is configured correctly, it is best practice to restart AE Server (if possible), this will ensure that the new connections are brought up correctly. Click on the **Restart AE Server** button at the bottom of the screen.

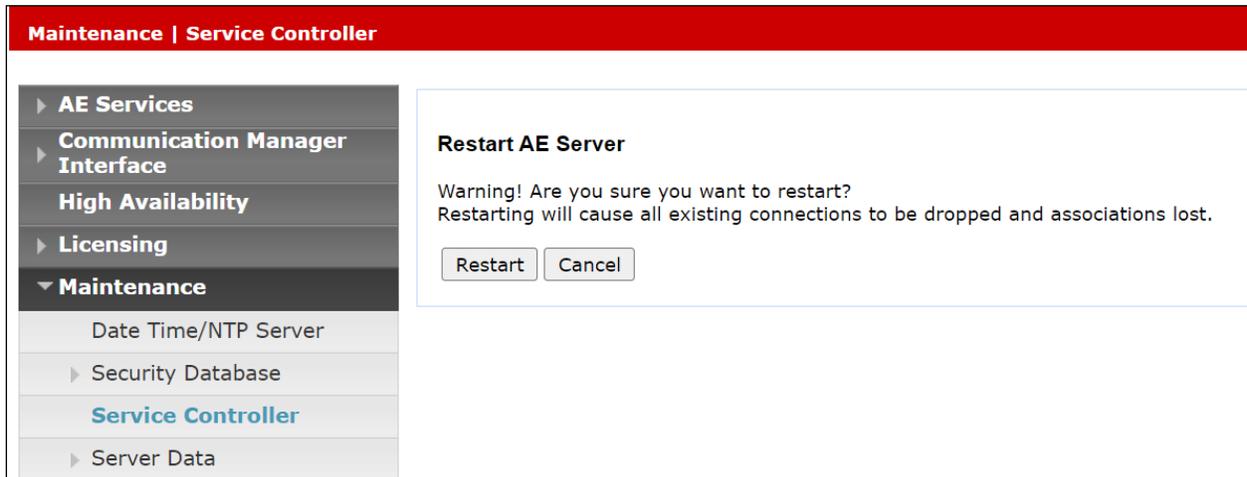


The screenshot shows the 'Maintenance | Service Controller' interface. On the left is a navigation menu with options: AE Services, Communication Manager Interface, High Availability, Licensing, Maintenance (selected), Date Time/NTP Server, Security Database, Service Controller (highlighted), Server Data, Networking, Security, and Status. The main content area is titled 'Service Controller' and contains a table with the following data:

Service	Controller Status
<input type="checkbox"/> ASAI Link Manager	Running
<input type="checkbox"/> DMCC Service	Running
<input type="checkbox"/> CVLAN Service	Running
<input type="checkbox"/> DLG Service	Running
<input type="checkbox"/> Transport Layer Service	Running
<input type="checkbox"/> TSAPI Service	Running

Below the table, there is a note: 'For status on actual services, please use [Status and Control](#)'. At the bottom, there is a row of buttons: Start, Stop, Restart Service, Restart AE Server, Restart Linux, and Restart Web Server.

A message confirming the restart will appear, click on **Restart** to proceed.



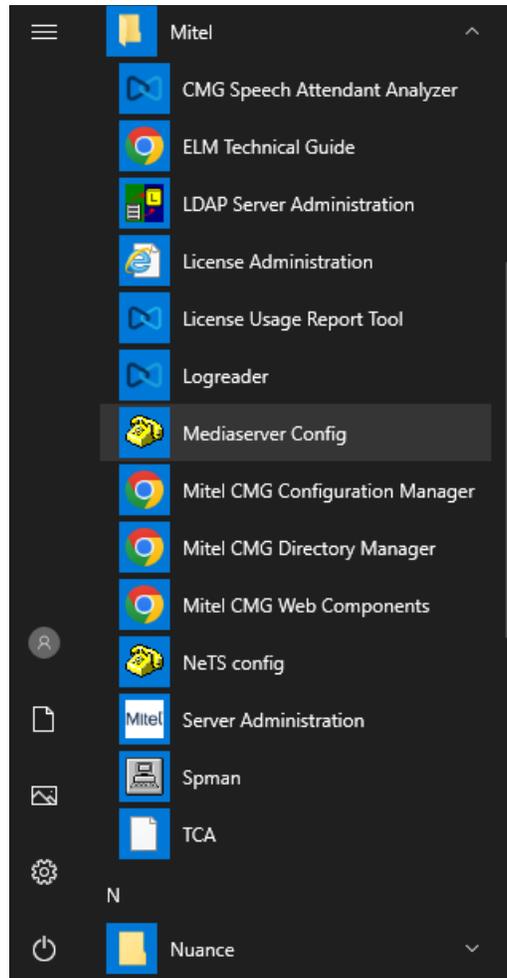
The screenshot shows the same 'Maintenance | Service Controller' interface, but with a dialog box open. The dialog is titled 'Restart AE Server' and contains the following text: 'Warning! Are you sure you want to restart? Restarting will cause all existing connections to be dropped and associations lost.' At the bottom of the dialog are two buttons: 'Restart' and 'Cancel'.

8. Configure Mitel Attendant Connectivity Server (ACS)

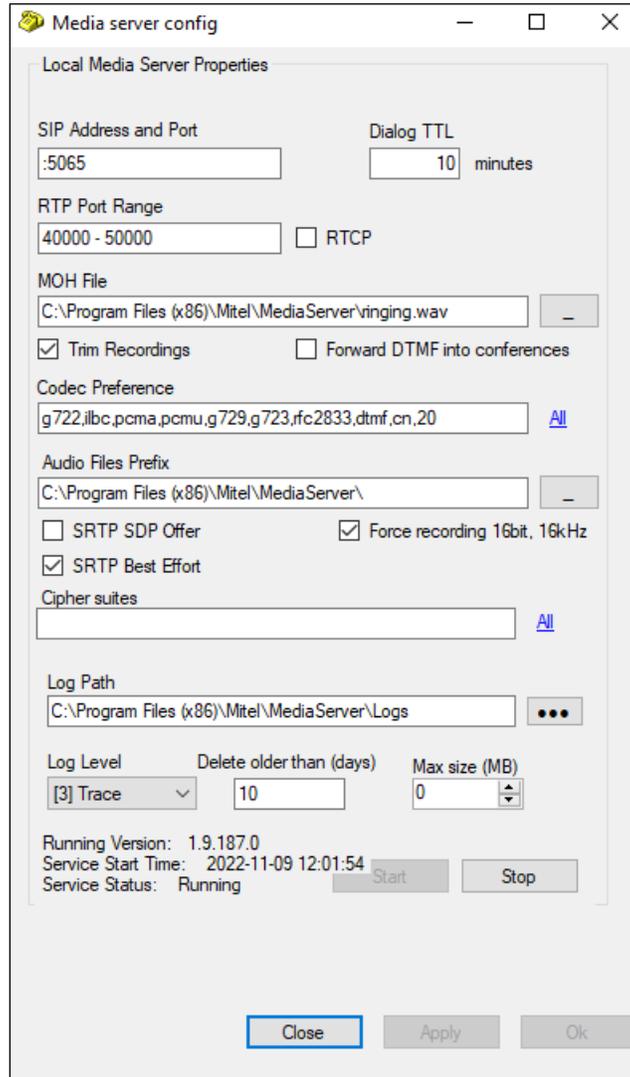
Although a Mitel engineer will setup the solution the following sections show information on the connection to Session Manager that was used for compliance testing, it may prove useful.

8.1. Mitel Media Server configuration

All Mitel applications are run from the Windows 2019 server, click on the **Mediaserver Config**, as shown below.

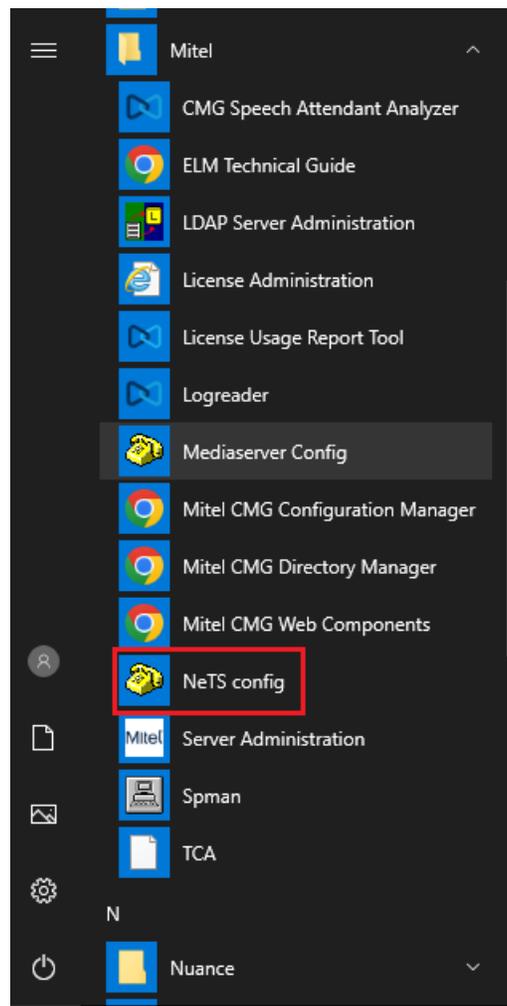


These are the settings that were used for compliance testing. Take note of the **Codec Preference** as this is where they are set. Typically, these are the default settings.

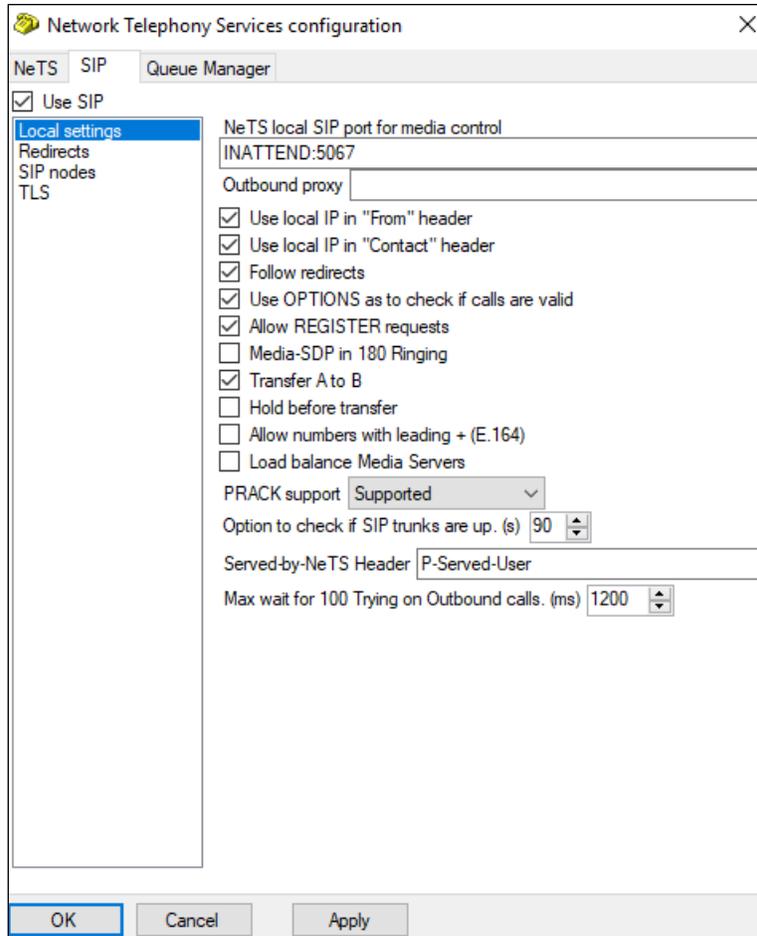


8.2. Mitel NeTS configuration

Click on the **NeTS config** as shown below.

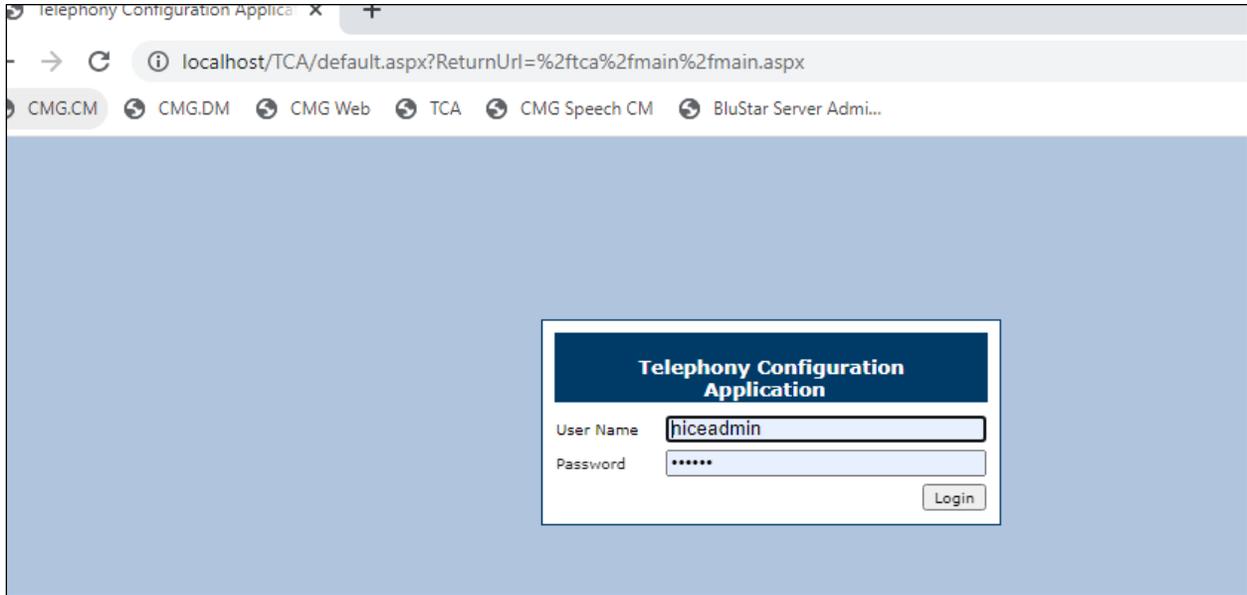


These are the settings that were used for compliance testing. The only settings that are of interest to the connection to Session Manager are found under the **SIP** tab and **Local settings**. Typically, these are the default settings.



8.3. Mitel Telephony Configuration Application (TCA) configuration

Open a web browser and browse to the ACS server name followed by TCA, for example `http://<servername>/tca`. Enter the appropriate credentials and click on **Login**.



A configuration will be setup as part of the initial installation and configuration, click on that **Configuration name**, for compliance testing this was **Inattend-Avaya**.

Telephony Configuration Application

No configuration loaded

Configurations

Configurations
Click on the configuration name to work with the configuration. To create a new configuration click on the "new" symbol to the right. The chosen configuration will be the template for the new configuration. To delete a configuration, click on the trash bin to the right of the configuration. Download a file by right clicking on the source link and choosing "Save Target As..." in the popup menu.

Configuration name	(source)	Last deployed (Server local time)		
Inattend-Avaya	(source)	2022-11-09 12:57:44		
Inattend-MX	(source)	2022-11-04 08:14:57		

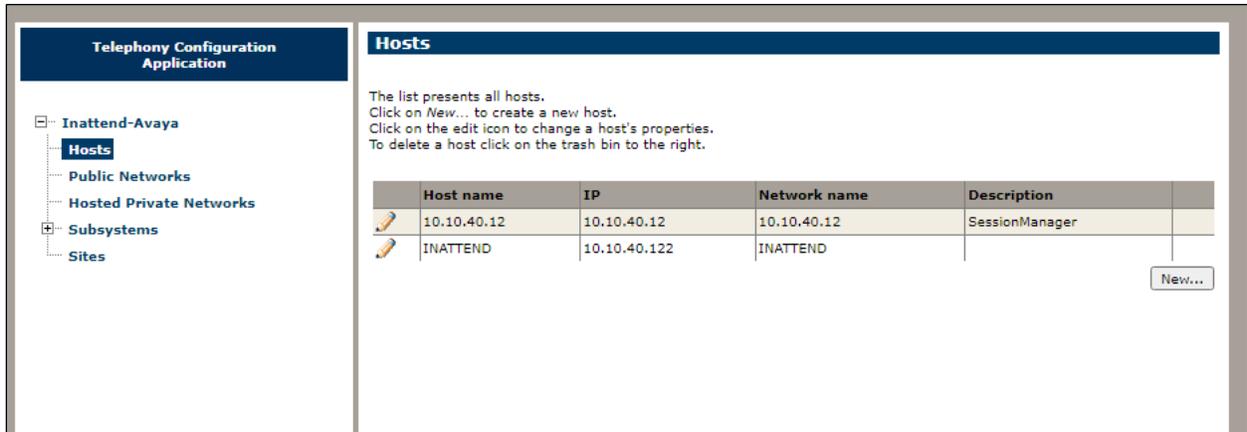
1

Templates

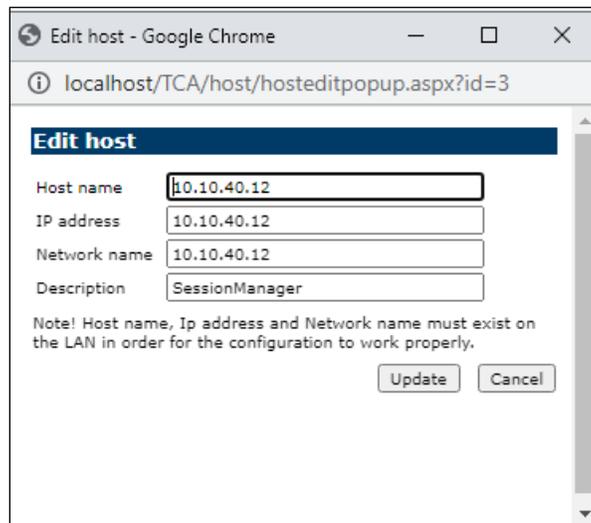
To create a new configuration from the template click on the "new" symbol to the right.

Template name	Description	
CMGVoice Without CTC - Template	CMGVoice Without CTC - Template	
CUCM SIP Template	CUCM SIP Template	
Empty Template	Empty Template	
InAttend CUCM SIP Template	InAttend CUCM SIP Template	
InAttend MX-ONE SIP Template	InAttend MX-ONE SIP Template	
Simple CUCM SIP Template	Simple CUCM SIP Template	
Single ACS Template	Single ACS Template	
Single CTC Template	Single CTC Template	
Single CTC With Queue Messages - Template	Single CTC With Queue Messages - Template	

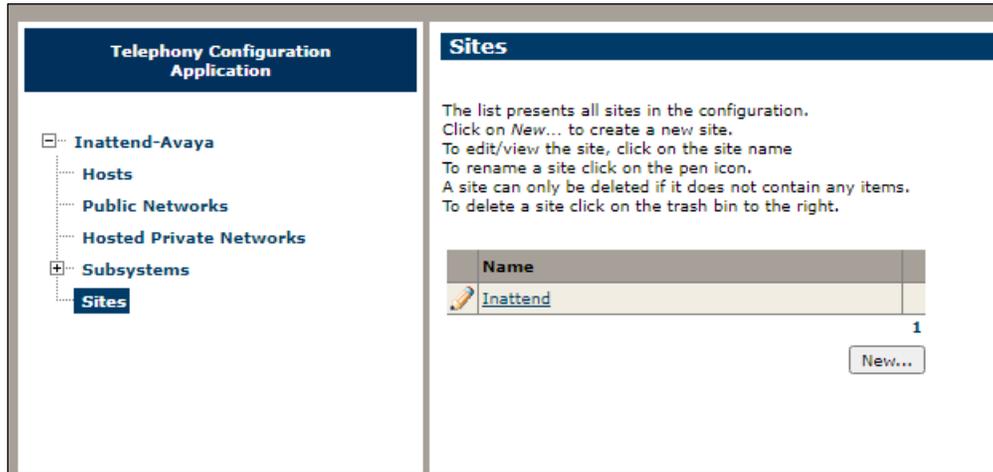
Click into **Hosts** in the left window. The hosts below were already configured by Mitel for compliance testing and clicking on the edit icon will show more information on these hosts. A new host can be added by clicking on **New** in the main window.



Enter a suitable **Host name** and **IP address**. This will be the Session Manager Security Module (SM100) IP address, as an example **10.10.40.12** was used below.

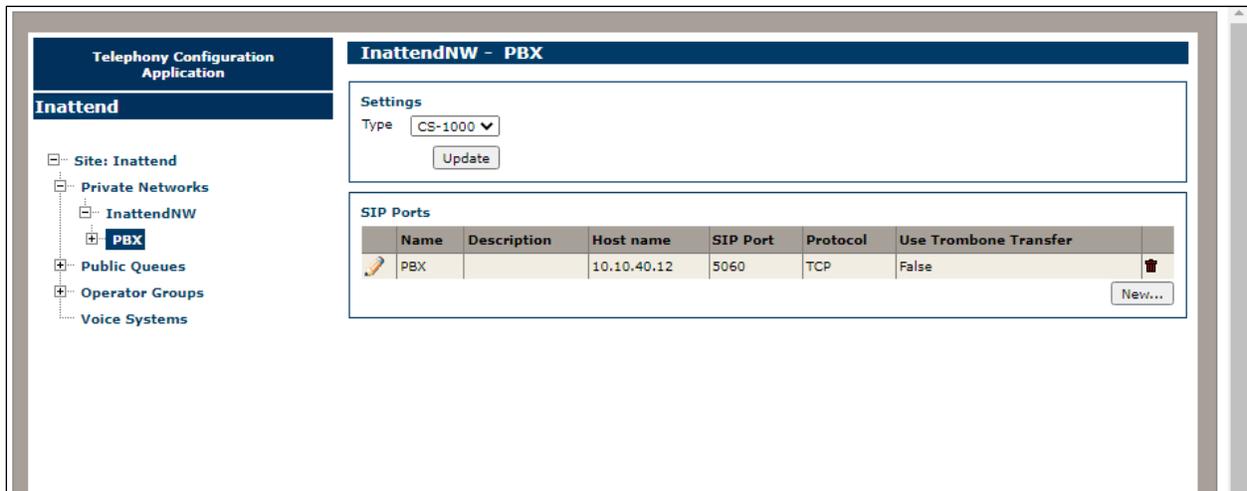


Click on **Sites** in the left window and once again a site will have been already configured during the initial setup, click on that site.



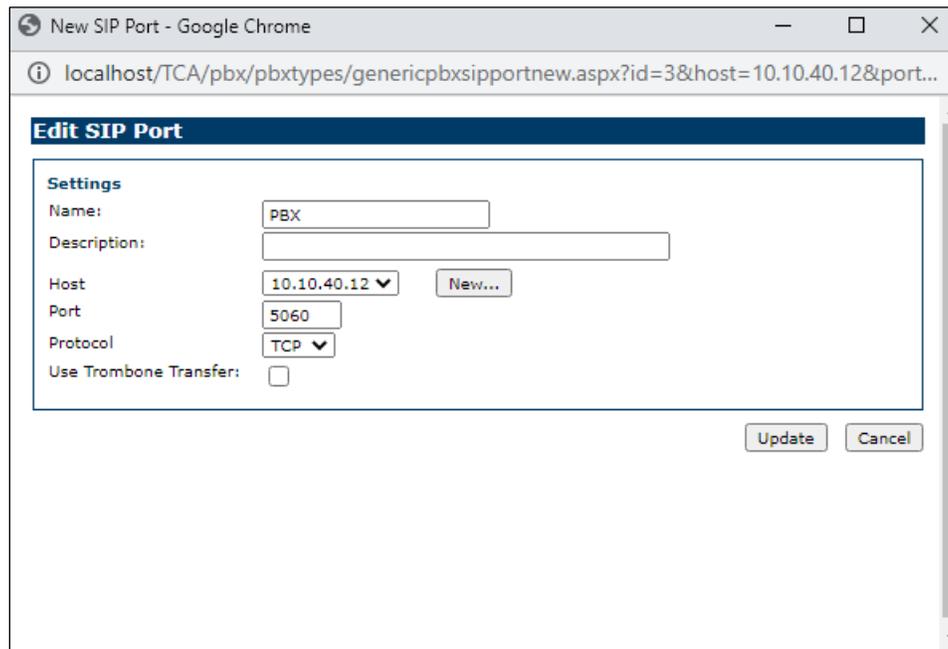
Navigate to **PBX** in the left window and click on **New** in the main window. This will create a new PBX connection. Note that the **Type** can be set to **CS-1000** before clicking on **New**.

Note: The Type being set to CS-1000 does not matter for Communication Manager, this is correct as there is no specific setting for Communication Manger and the closest is CS-1000.



Enter a suitable name and select the **Host** that was created above from the drop-down menu. The **Port** should be set to **5060** and the **Protocol** should be set to **TCP**, this will match the **Entity Link** setup in **Section 6.3**.

Note: The Protocol used can be either TCP or UDP, but it must match that setup on the Entity Link in **Section 6.3**. A connection using TLS is currently not supported.



The screenshot shows a web browser window titled "New SIP Port - Google Chrome". The address bar displays the URL: localhost/TCA/pbx/pbxtypes/genericpbxsippportnew.aspx?id=3&host=10.10.40.12&port... The main content area is titled "Edit SIP Port" and contains a "Settings" section with the following fields:

- Name: Text input field containing "PBX".
- Description: Text input field (empty).
- Host: Dropdown menu showing "10.10.40.12" and a "New..." button.
- Port: Text input field containing "5060".
- Protocol: Dropdown menu showing "TCP".
- Use Trombone Transfer: Unchecked checkbox.

At the bottom right of the form, there are two buttons: "Update" and "Cancel".

Navigate to **Domains** in the left window and note the **SIP Domain** is entered here as per **Section 6.1.1**. Devices can be entered by clicking on the **New** button at the bottom right of the screen. This will add Communication Manager extensions that can be used for other functions that are not covered in these Application Notes.

The screenshot displays the 'InattendNW - PBX - Domains - Inattend' configuration page. On the left is a navigation tree with 'Inattend' selected. The main area is divided into several sections:

- Settings:** Fields for PBX Id (1), Default internal prefix, CMG View, SIP Domain (greaney.sil6.avaya.com), SIP Domain Description (greaney.sil6.avaya.com), and Phone context. Includes an 'Update' button.
- Ports:** A table with columns: Name, Type, Host name, Port, Protocol, Description. One entry is shown: PBX, sip, 10.10.40.12, 5060, TCP. Includes an 'Add' button.
- Media servers:** A table with columns: Name, Order. One entry is shown: MS, 1. Includes an 'Add' button.
- Device ranges:** A table with columns: Description, Range, Type. Entries include Ext (4500), SA (4502-4503), Int (4501), DirectDrop (4506), Speech (4505), and Office (3000-3199). Includes a 'New...' button.

At the bottom, a status bar shows: Working with: Inattend-Avaya, Last deployed (Server local time): 2022-11-09 12:57:44, and buttons for 'Deploy' and 'Close'.

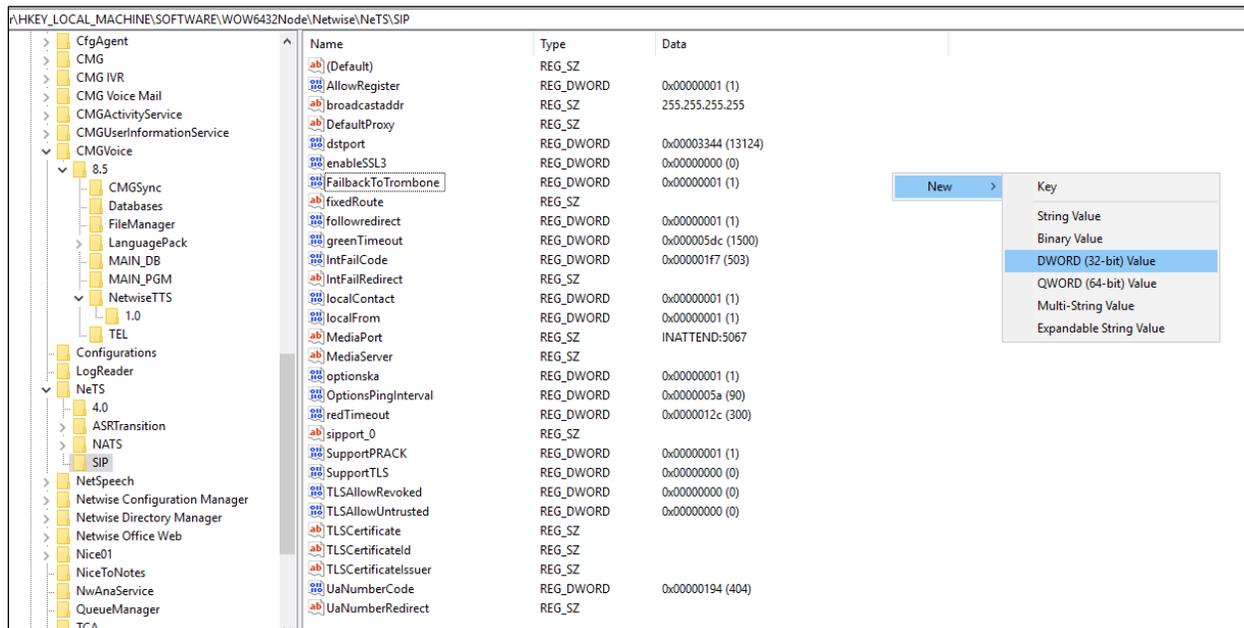
8.4. Update the Registry on the Mitel Attendant Connectivity Server

A registry setting was added to the NeTS process on the ACS server to allow a re-invite to be sent to overcome an issue found during the following scenarios:

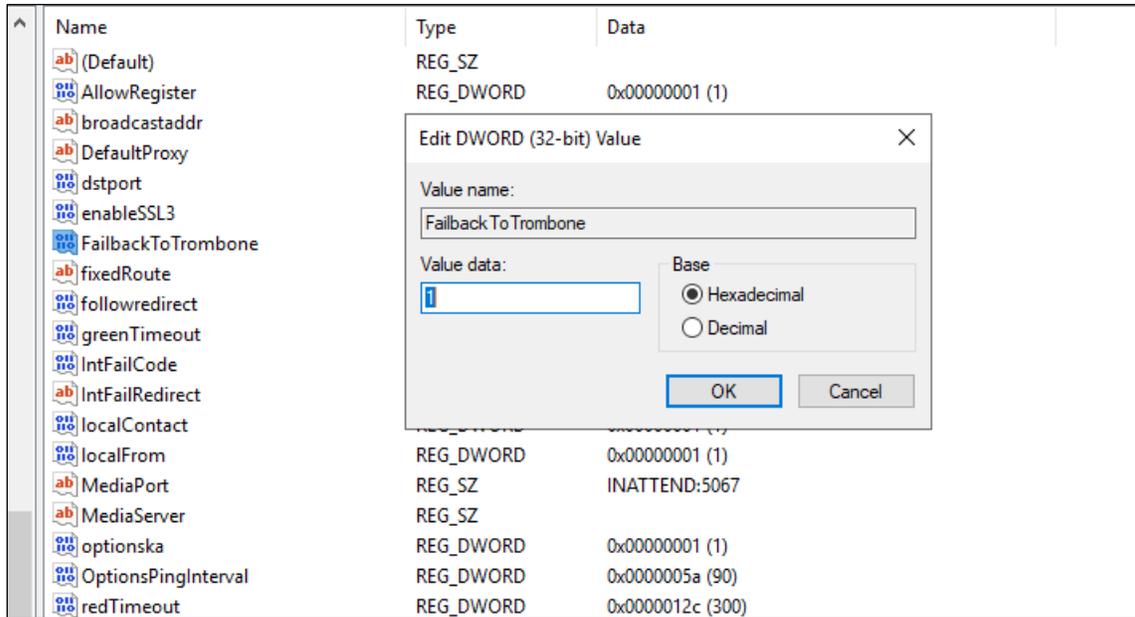
1. Caller from Communication Manager calls to the Mitel InAttend operator.
2. The operator transfers the caller to a voicemail box, 'Direct Drop' to the mailbox.

Without the update in the registry the call could not be transferred correctly. The ACS will initiate a transfer using REFER and Communication Manager sends an ACCEPT but then immediately after sends a NOTIFY message containing "481 Call Transaction does not exist". The NETS then creates a new invite with the trombone transfer, and this overcomes the issue.

The registry is updated as follows. Navigate to **Computer\HKEY_LOCAL_MACHINE\SOFTWARE\Wow6432Node\Netwise\NeTS\SIP**. In the main window, right-click anywhere on the screen and select **New → DWORD(32-bit) Value**.



Enter the name **FailbackToTrombone** as the name for the new **REG_DWORD** (not shown) and right-click on the REG_DWORD and select **Modify** as shown. Ensure that the **Value data** is set to **1** and the **Base** to **Hexadecimal**, as shown below. Click in **OK** once finished.



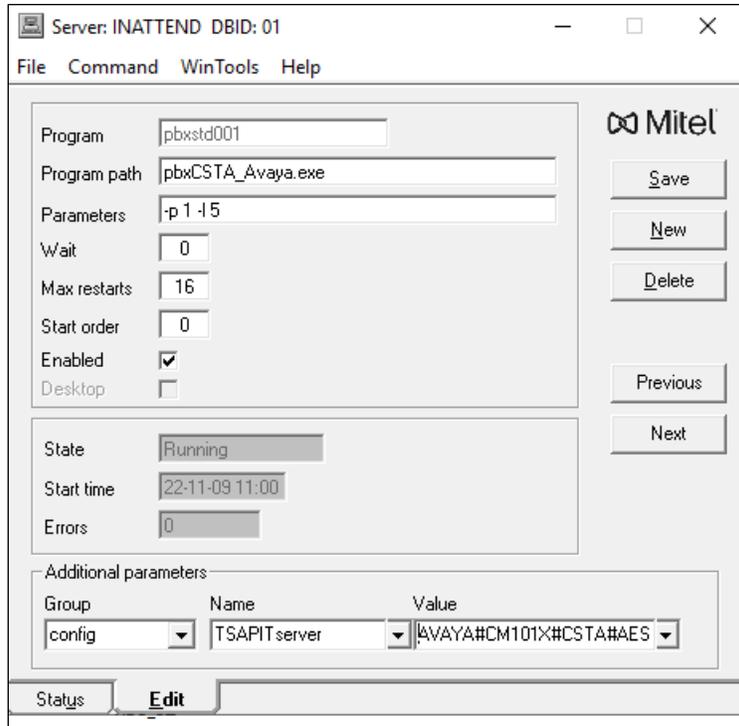
8.5. Configure TSAPI connection from Collaboration Management (CMG) module

Open SPMAN (show that screen shot that was taken of the Mitel folder), this is a Mitel application that reads the registry. The following screen is displayed. Changes are made to the **Additional parameters** at the bottom of the screen below. The **TSAPIVDN** is the VDN added in **Section 5.5.1**. This will be the second VDN added, the VDN that the collect digits Vector routes to. The **TSAPIUser** and **TSAPIPassword** is that of the CTI user added in **Section 7.6**.

Group	Name	Value
config	TSAPIVDN	1084

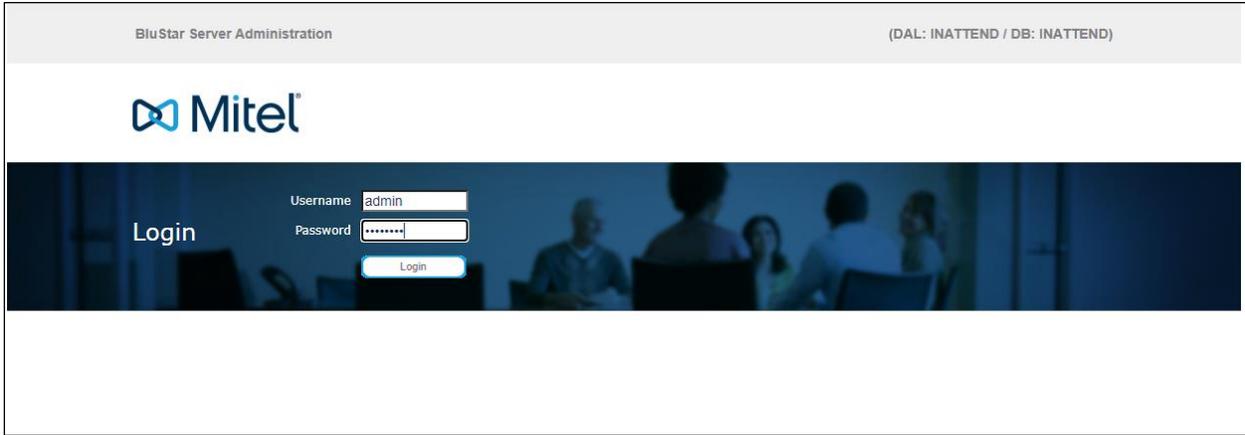
The **TSAPITserver** information is filled in from the TLINK as shown in **Section 7.4**.

Note: The unsecure link was used for the compliance testing.



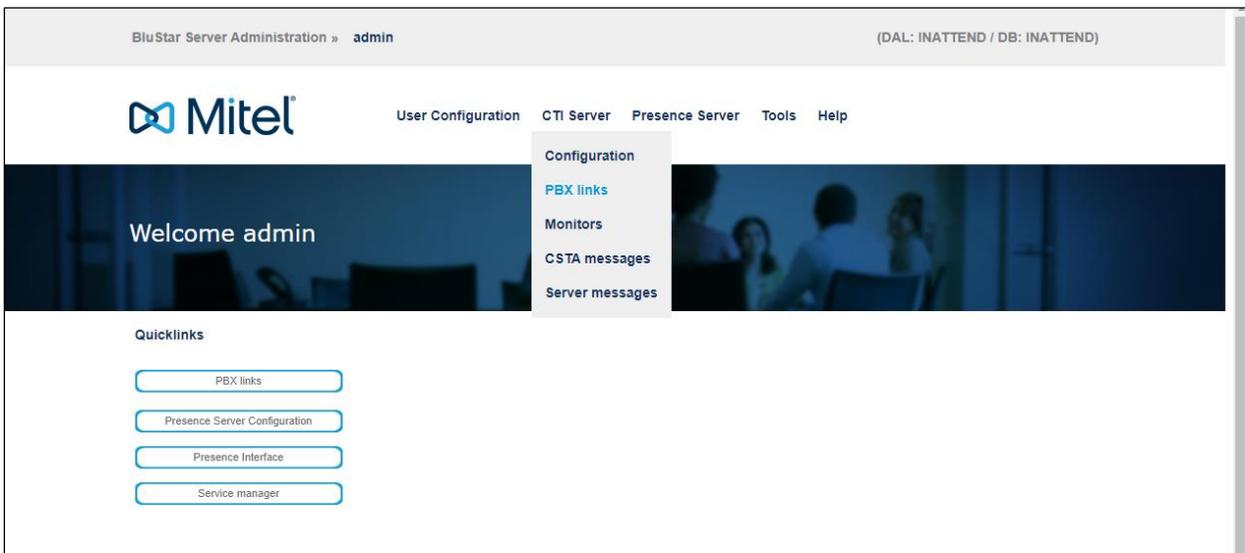
8.6. Configure TSAPI connection from the InAttend Server module

Open a web browser to the InAttend server as shown. Enter the appropriate credentials and click on **Login**.



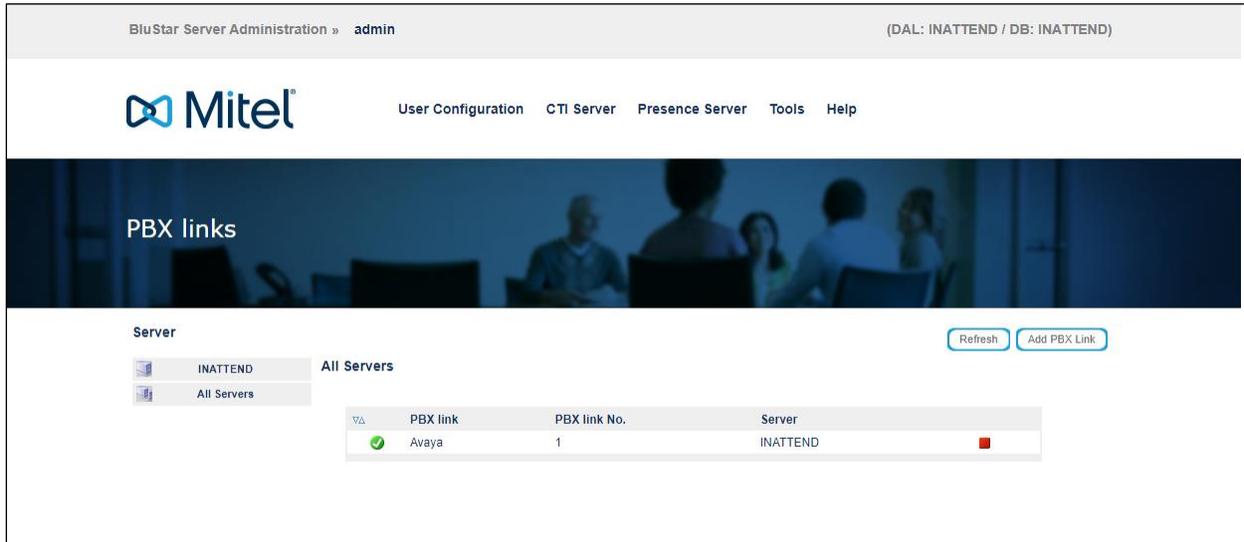
The screenshot shows the login page of the Mitel BluStar Server Administration interface. At the top, it says "BluStar Server Administration" and "(DAL: INATTEND / DB: INATTEND)". The Mitel logo is prominently displayed. Below the logo, there is a "Login" section with a background image of people in a meeting. The login form includes a "Username" field with "admin" entered, a "Password" field with masked characters, and a "Login" button.

Click on **CTI Server** → **PBX links**.

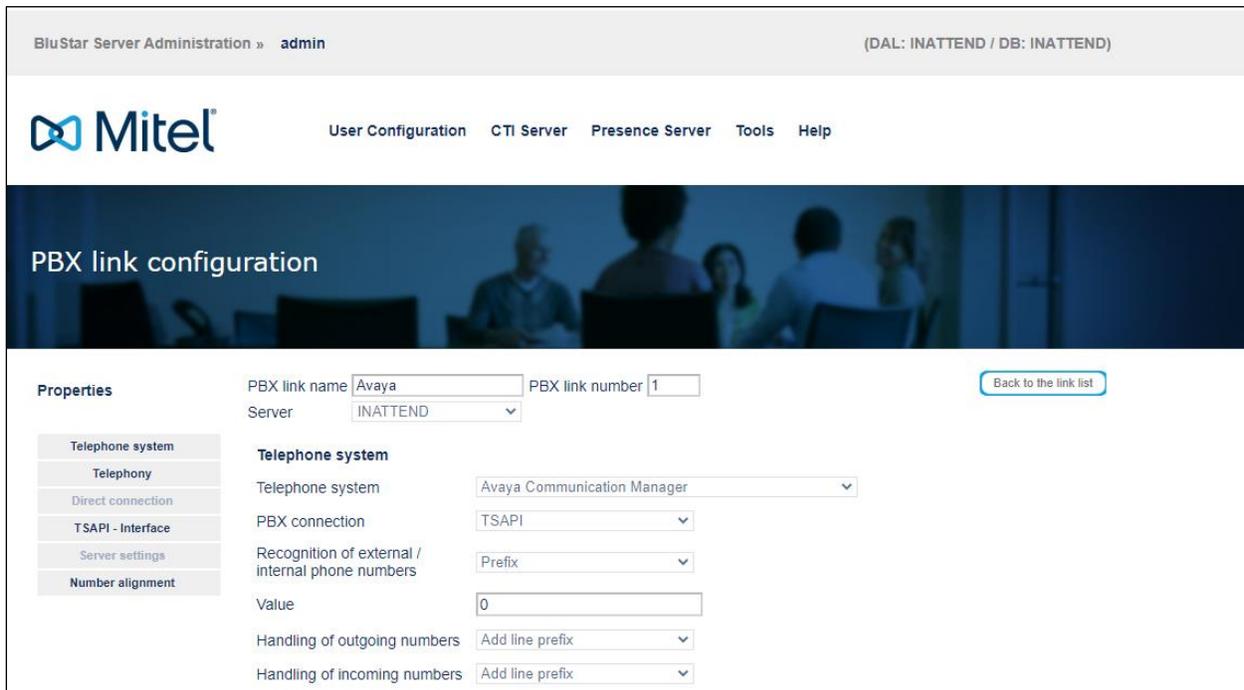


The screenshot shows the dashboard of the Mitel BluStar Server Administration interface after a successful login. The top navigation bar includes "User Configuration", "CTI Server", "Presence Server", "Tools", and "Help". The "CTI Server" menu is expanded, showing "Configuration", "PBX links", "Monitors", "CSTA messages", and "Server messages". The "PBX links" option is highlighted. Below the navigation, there is a "Welcome admin" message and a "Quicklinks" section with four buttons: "PBX links", "Presence Server Configuration", "Presence Interface", and "Service manager".

The following PBX link was already configured, but a new one can be added by clicking on **Add PBX Link**.



The following screen appears, and the **PBX link configuration** can be set. **Avaya Communication Manager** is chosen for the **Telephone system**. The **PBX connection** is set to **TSAPI** and the **Save** button can be pressed (not shown).



Pressing **Save** on the previous screen brings up the following window where the **TSAPI-Interface** details are added, which include the TLINK, TSAPI user and password. Click on **Save** again once the information is filled in.

BluStar Server Administration » admin (DAL: INATTEND / DB: INATTEND)

Mitel User Configuration CTI Server Presence Server Tools Help

PBX link configuration

Properties

PBX link name: PBX link number: [Back to the link list](#)

Server:

TSAPI - Interface

Telephony server:

Username:

Password:

Path of Csta32.dll:

The following screen is then shown containing the new connection. This connection must be started by pressing the **start icon**, highlighted below.

Mitel User Configuration CTI Server Presence Server Tools Help

PBX links

Server [Refresh](#) [Add PBX Link](#)

Server	PBX link	PBX link No.	Server
	Avaya	1	INATTEND

A successful connection will appear as green, as it is shown below.

The screenshot displays the Mitel BluStar Server Administration interface. At the top, the breadcrumb navigation shows 'BluStar Server Administration » admin' and the session information '(DAL: INATTEND / DB: INATTEND)'. The Mitel logo is on the left, and navigation links for 'User Configuration', 'CTI Server', 'Presence Server', 'Tools', and 'Help' are on the right. Below the navigation is a banner image with the text 'PBX links'. The main content area is titled 'Server' and includes a sidebar with 'INATTEND' and 'All Servers' buttons. To the right of the sidebar are 'Refresh' and 'Add PBX Link' buttons. A table lists the PBX links:

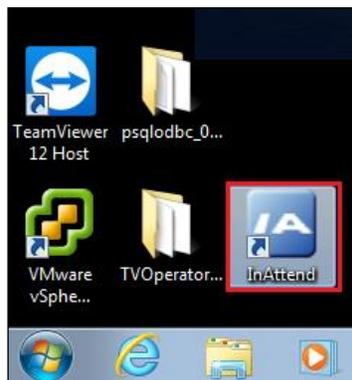
✓/✗	PBX link	PBX link No.	Server	
✓	Avaya	1	INATTEND	✗

9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Mitel solutions.

1. Make a call to the InAttend attendant and request to be transferred to a known extension. Ensure the call is connected.
2. Make a call to the InAttend attendant and request to be transferred to a known extension which is busy and request to leave a voice message. Ensure the call is transferred to voicemail and a message can be left.
3. Make a call to the Attendant queue. Ensure the attendant receives and answers the call.

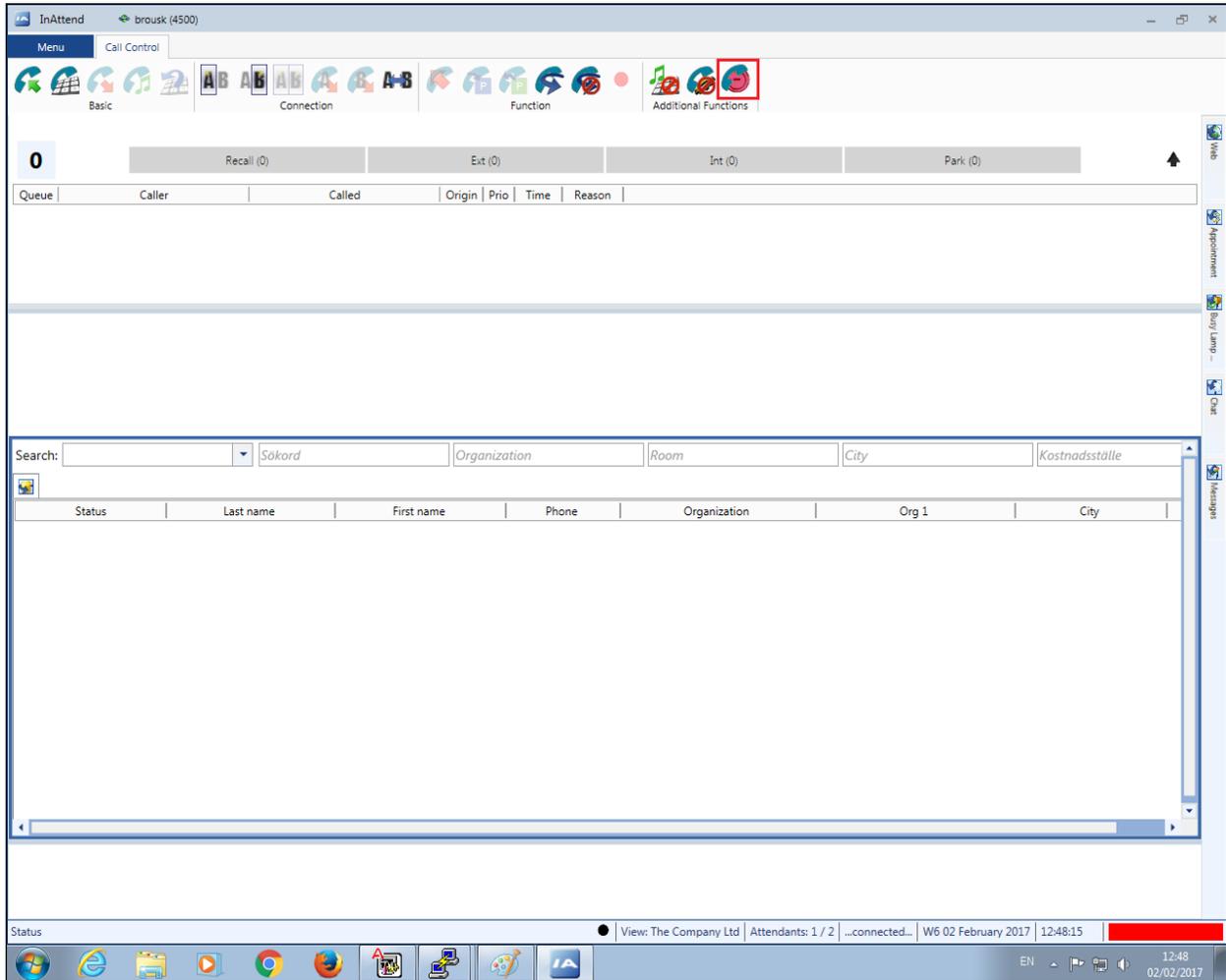
InAttend can be started from the shortcut or by navigating to the program on the client PC.



Enter the appropriate credentials and click on **Log On**.

A screenshot of the Mitel InAttend login interface. The background is dark blue. At the top left is the Mitel logo, followed by a vertical line and the text 'InAttend'. Below this, there are two input fields: 'Username' with the text 'brousk' and 'Password' with a masked password of ten dots. Below the password field is a checkbox labeled 'Remember password' which is currently unchecked. At the bottom right, there are two buttons: 'Log On' and 'Cancel'.

Once logged in the operator will be in night mode as shown below with the red bar. Click on the icon highlighted to change this to normal operation.



Once a call is presented to the attendant the caller is shown on the attendant screen and the attendant can answer the call using the mouse or keyboard. Presence information on users beginning with 100 are shown at the bottom of the screen.

The screenshot displays the InAttend software interface for user 'brousk (4500)'. The interface includes a 'Menu' bar with 'Call Control' selected, and various function icons for Basic, Connection, Function, and Additional Functions. Below the menu, there are buttons for 'Recall (0)', 'Ext (1)', 'Int (0)', and 'Park (0)'. A call log table shows a recent call from PSTN (091732000) to Avaya Night (4500) at 00:04. A large dark blue panel displays the caller information: PSTN 091732000 and INT Avaya Night 4500. Below this is a search bar with '100' entered and several filter fields. A table lists users starting with '100':

Status	Phone	Last name	First name	Organization	Org 1	City
In a call	1000	Rousk	Björn	Market Access	0008-008.1	
In a call	1001	Andersson	Mikael			

The status bar at the bottom shows 'View: The Company Ltd | Attendants: 1 / 3 | ...connected... | W50 Wednesday 11 December 2019 | 17:20:08'.

With the call answered the caller's information is displayed and this information can be augmented with information from the InAttend database.

The screenshot displays the Avaya InAttend interface. At the top, there are navigation tabs for 'Basic', 'Connection', 'Function', and 'Additional Functions'. Below these are input fields for 'Recall (0)', 'Ext (0)', 'Int (0)', and 'Park (0)'. A table with columns 'Queue', 'Caller', 'Called', 'Origin', 'Prio', 'Time', and 'Reason' is visible but empty. A large blue banner displays the caller's name 'Louise Finnigan' with extension '7001' and the call type 'INT Avaya Night' with extension '4500'. Below this is a search bar with fields for 'Sökord', 'Organization', 'Room', 'City', and 'Kostnadsställe'. The main content area shows a profile for 'Louise Finnigan' with a 'Phone no: 7001' and a 'Details' tab. The 'Details' tab contains a table with caller information:

Detail	Value
Title	Shop Manager
E-Mail	louise.finnigan@thecompany.com
	Manager
	Shop Manager
	Cashier
Sökord	Receipts
	Shop Inventory
	Salesman
	Mail
Phone	7001

At the bottom, a status bar shows 'Status', 'View: The Company Ltd', 'Attendants: 1 / 2', '...connected...', 'W6 02 February 2017', and '12:47:17'.

9.1. Verify the connection to Avaya Aura® Application Enablement Services

The following can be checked to ensure that the connections to the AES are in operation correctly.

9.1.1. Verify the link to Application Enablement Services from Communication Manager

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	12	no	aespri101x	established	865	865

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 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	aespri101x	10.10.40.16	56722	procr	683	665

9.1.2. Verify the TSAPI Link from Application Enablement Services

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**. There were two devices monitored during compliance testing and so **Associations** is showing **2** below.

TSAPI Link Details

Enable page refresh every seconds

Link	Switch Name	Switch CTI Link ID	Status	Since	State	Switch Version	Associations	Msgs to Switch	Msgs from Switch	Msgs Period
1	cm101x	1	Talking	Mon Nov 27 17:13:05 2023	Online	20	2	67	62	30

Online Offline

For service-wide information, choose one of the following:

Click in **User Status** on the screen above. A new window is displayed below showing the CTI user **Mitel** connected to receive the TSAPI events. As per **Section 1**, the Mitel InAttend solution makes use of two TSAPI connections to Application Enablement Services.

- TSAPI connection from the CMG – Used to set Call Forwarding and Message Waiting.
- TSAPI connection from the InAttend server – Used to monitor devices to provide Presence information.

CTI User Status

Enable page refresh every seconds

CTI Users

Open Streams 2
Closed Streams 0

Open Streams

Name	Time Opened	Time Closed	Tlink Name
Mitel	Wed 22 Nov 2023 03:15:16 PM GMT		AVAYA#CM101X#CSTA#AESPRI101X
Mitel	Wed 22 Nov 2023 03:15:58 PM GMT		AVAYA#CM101X#CSTA#AESPRI101X

9.2. Verify the SIP Trunk connection

The SIP trunk from Communication Manager to Session Manager can be checked using the following steps.

9.2.1. Verify Avaya Aura® Communication Manager

The following steps can be taken if there are any issues with calls being made. This should help verify the links between the products. From the SAT interface, verify the status of the SIP trunk groups by using the **status trunk n** command, where “n” is the trunk group number administered in **Section 5.2**. Verify that all trunks are in the **in-service/idle** state as shown below (just a sample of the trunks configured).

```
status trunk 1

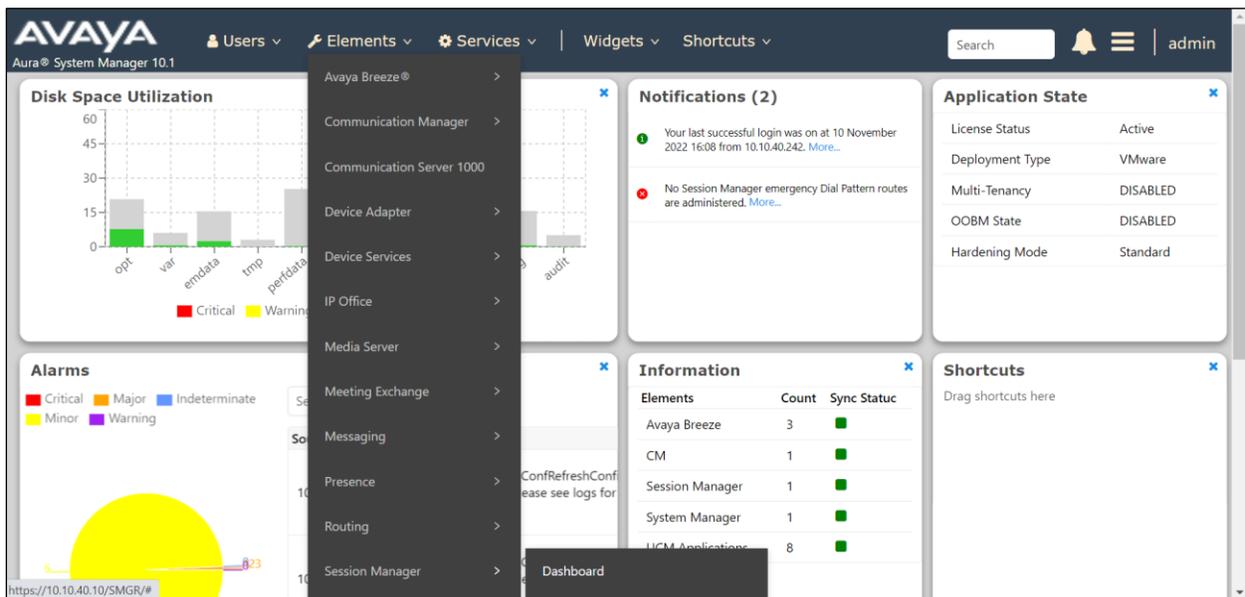
                                TRUNK GROUP STATUS

Member      Port      Service State      Mtce Connected Ports
                                Busy

0001/0001  T00001  in-service/idle    no
0001/0002  T00002  in-service/idle    no
0001/0003  T00003  in-service/idle    no
```

9.2.2. Verify InAttend SIP Entity is up

Log into System Manager as per **Section 6**. Navigate to **Elements** and click on **Session Manager**.



Selected **SIP Entity Monitoring** in the left window.

Session Manager Dashboard

This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State: [Dropdown] Shutdown System: [Dropdown] EASG: [Dropdown] Clear Logs: [Button] As of 3:05 PM

1 Item Show All Filter: Enable

Session Manager	Type	Tests Pass	Alarms	Security Module	Service State	Load Factor	Entity Monitoring	Active Call Count	Registrations	Data Replication	User Data Storage Status	License Mode	EASG
<input type="checkbox"/> sm101x	Core	✓	0/0/0	Up	Accept New Service	0/0/0	2/13	0	1/1	✓	✓	Normal	Enabled

Select : All, None

Select the **InAttend** SIP Entity.

Monitored Entities

Session Manager	Type	Monitored Entities			
		Down	Partially Up	Up	Not Monitored
<input type="checkbox"/> sm101x	Core	2	0	11	0

Select : All, None

All Monitored SIP Entities

Run Monitor

13 Items

SIP Entity Name
<input type="checkbox"/> Messaging2019
<input type="checkbox"/> AACC
<input type="checkbox"/> Experience Portal-MPP
<input checked="" type="checkbox"/> InAttend
<input type="checkbox"/> novaalert
<input type="checkbox"/> IP Office - SE
<input type="checkbox"/> cm101x - Phones - 5061

The SIP Entity should show as **UP** as it is shown below.

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

All Entity Links to SIP Entity: InAttend

Summary View

1 Item  Filter: Enable

	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	sm101x	IPv4	10.10.40.122	5060	TCP	FALSE	UP	200 OK	UP

Select : None

10. Conclusion

The interoperability of Mitel InAttend using Mitel Attendant Connectivity Server V2.7 from Mitel Sweden AB to interoperate with Avaya Aura® Communication Manager R10.1 utilizing a SIP trunk connection to Avaya Aura® Session Manager R10.1 and a TSAPI connection to Avaya Aura® Application Enablement Services was successful for this specific setup to place calls to and from InAttend. All issues and observations are outlined in **Section 2.2**.

11. Additional References

These documents form part of the Avaya official technical reference documentation suite. Further information can be obtained from <http://support.avaya.com> or from your Avaya representative.

- [1] *Administering Avaya Aura® Communication Manager* – Release 10.1
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 10.1
- [3] *Avaya Aura® Application Enablement Services Administration and Maintenance Guide* Release 10.1
- [4] *Administering Avaya Aura® Session Manager* – Release 10.1

Product Documentation for Mitel InAttend can be obtained from Mitel at:
<http://www.Mitel.com/support>

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