Avaya Solution & Interoperability Test Lab

Application Notes for British Telecom (Financial Technology Services) Session Manager with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate British Telecom (Financial Technology Services) Session Manager with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. British Telecom Session Manager is a SIP proxy that interoperates with Avaya Aura® Session Manager via a SIP Trunk. It is used to route calls to the British Telecom trading turrets.

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

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

1. Introduction

These Application Notes describe the configuration steps required to successfully integrate British Telecom (BT) Session Manager with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The British Telecom Session Manager is a SIP Proxy that uses Session Manager to route calls between Communication manager and BT trading turrets via a SIP Trunk.

2. General Test Approach and Test Results

The general test approach was to configure the BT Session Manager to communicate with the Session Manager via a SIP Trunk.

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.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on carrying out different call scenarios with good quality audio. The tests included:

- SIP trunk is connected and in-service.
- BT Turret can make and receive calls via BT Session Manager.
- BT Turret can transfer and conference via BT Session Manager.
- BT Turret can recover from loss of service via BT Session Manager.

2.2. Test Results

All test cases were passed with the following observations.

- Avaya SIP session timers must be set to 30 minutes to allow all session refreshes to be done by the BT Session Manager every 23 minutes. This setting is configured in **Section 5**.

2.3. Support

For technical support for BT Session Manager and BT Netrix turrets the BT Unified Trading Interoperability Team can be contact at:
Email: Unified.trading.interop.team@bt.com

3. Reference Configuration

The configuration shown in **Figure 1** was used during the compliance test of BT Session Manager with Session Manager and Communication Manager. BT Session Manager utilizes a SIP trunk to communicate with BT Netrix and BT Hi Touch Turrets and Avaya handsets.

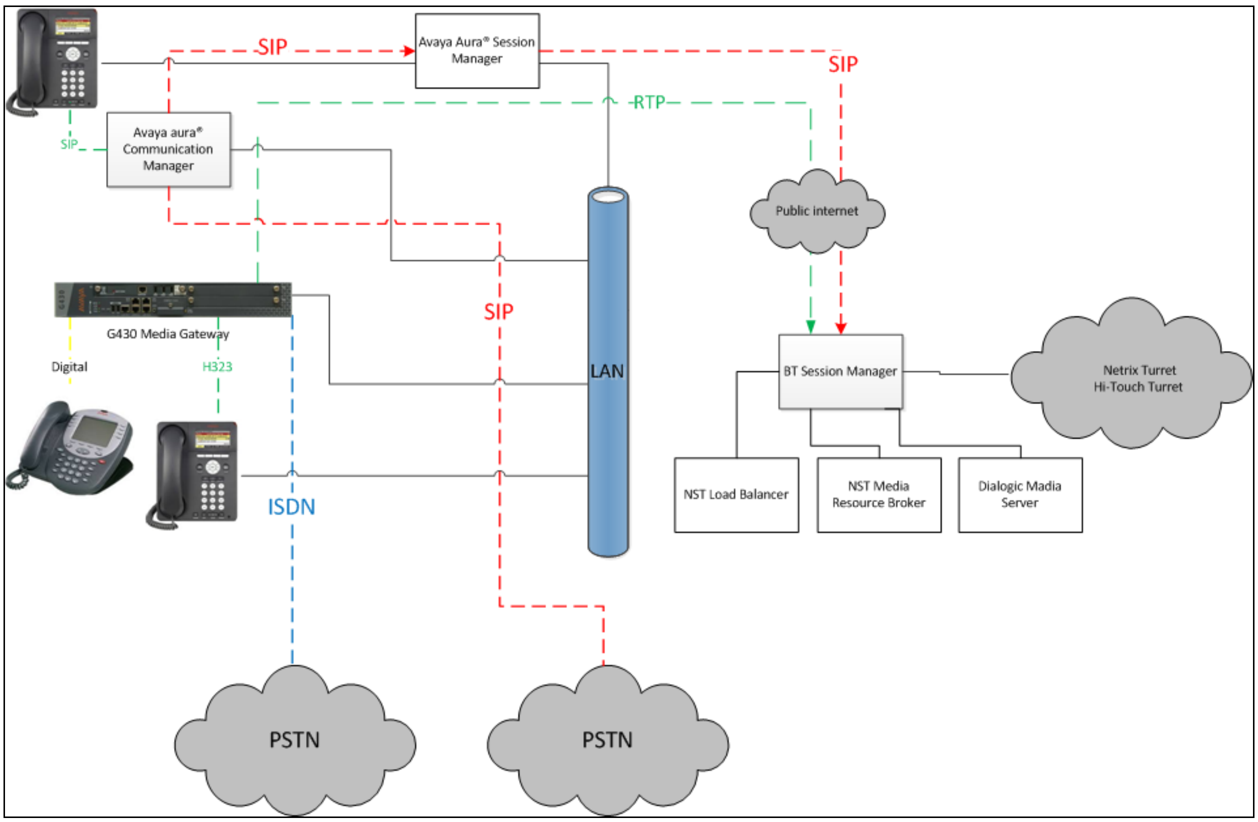


Figure 1: Connection of British Telecom Session Manager with Avaya Aura® Session Manager and Avaya Aura® Communication Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	R7.0 Build R017x.00.0.441.0 Version 7.0.1.1.0.441.23169 Updates: 00.0.441.0-23169 PLAT-rhel6.5-0010
Avaya Aura® Session Manager	R7.0.1 Build 7.0.1.1.70114
Avaya Aura® System Manager	R7.0.1.2 Build 7.0.0.0.16266 Update 7.0.1.2.075662 Service Pack 2
Avaya 96x1 Series IP Deskphones H.323	6.6029
Avaya 96x1 Series IP Deskphones SIP	7.0.0-080615
Avaya 2420 Series Digital Deskphones	Rel 6 FWV 6
British Telecom Session Manager (Co-Hosted).	4.2.1.3
British Telecom NST Load Balancer	1.3.23
British Telecom NST Media Resource Broker (MRB)	1.2.23
British Telecom Dialogic Media Server.	2.4.12290
British Telecom SIP Netrix Turret	4.2.0.13
British Telecom Hi Touch Turret	1.3.5.1

5. Configure Avaya Aura® Communication Manager

This section describes the steps required to allow Communication Manager to communicate with Session Manager. Is it assumed that Communication Manager is installed and configured before implementing the configuration steps. For all other provisioning information, such as initial installation and configuration, please refer to the product documentation in **Section 11**.

The configuration illustrated in this section was performed using Communication Manager System Administration Terminal (SAT).

Configuration steps include:

- Check SIP Trunk Licensing.
- Add entries in the Dial Plan for use with SIP Trunk and routing via BT Session Manager.
- Administer SIP Trunk (to Session Manager).
- Add Route Pattern.

Using the ***display system-parameters customer-options*** command go to **page 2** and check that the system is sufficiently licensed for SIP Trunks.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             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: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 10
```

Use the ***change node-names ip*** command to add the Session Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM1677              10.10.16.77
default             0.0.0.0
procr               10.10.16.27
procr6              ::-
```

Use ***change dialplan analysis*** to add a **3** digit dial access code(**dac**) for use in the SIP Trunk, a unform dial plan(**udp**) entry for calling out over the SIP Trunk and check that there is an entry for feature access codes(**fac**).

```
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
   2           7   udp
   7           3   dac
   8           5   udp
   8           7   udp
   827         7   ext
   9           1   fac
   *           3   fac
   #           3   fac
```

Use ***add-signaling-group x*,** where x is the number of the group required. Set **Transport Method** to **tcp, Near-end Node Name** to **procr** and **Far-end Node Name** to the entry added in **node-names**. Set the **Far-end Network Region** to **1** and **Direct IP-IP Audio Connections?** to **n.**

```
add signaling-group 76                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 76             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  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: SM1677
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

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

Use ***add trunk-group ,x*** where x is the number administered for the signaling group. On **Page 1,** set the **Group Type** to **sip**. Set the **TAC** to suitable entry based on the dial plan **dac** administered above. Set the **Service Type** to **tie**, **Signaling group** to the one administered above and **Number of Members** to a number satisfactory for call routing required (**255** shown is the max for this type of trunk group).

```
add trunk-group 76                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 76                   Group Type: sip          CDR Reports: y
  Group Name: ToSM7                       COR: 1        TN: 1        TAC: 776
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                          Member Assignment Method: auto
                                                   Signaling Group: 76
                                                 Number of Members: 255
```

On **Page 2,** set the **Preferred Minimum Session refresh Interval(sec): to 1800** as this is a time greater than the BT Session Manager refresh interval.

```
change trunk-group 76                                           Page   2 of  22
      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): 1800

 Disconnect Supervision - In? y  Out? y


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




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

On P**age 3,** set the **Numbering Format**. For this test, the **private** numbering table was used to set the calling party number format and send it over the SIP trunk.

```
   add trunk-group 76                                                  Page   3 of  21
   TRUNK FEATURES
             ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y




                        Numbering Format: private
                                                         UUI Treatment: service-provider

                                                          Replace Restricted Numbers? n
                                                         Replace Unavailable Numbers? n

                                                           Hold/Unhold Notifications? y
                                   Modify Tandem Calling Number: no




    Show ANSWERED BY on Display? y
```

Next, a route pattern needs to be added so that call can be routed out of Communication Manager to Session Manager. Use ***change route-pattern x,*** where x is the number of the SIP trunk created. Enter the Trunk group created above beside the first **Grp No,** an **FRL** of **0**.

```
change route-pattern 76                                               Page   1 of   3
                    Pattern Number: 76     Pattern Name: ToSM7
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 76   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  none
```

An Alternate Route Selection(ARS) entry must be made for dialing the external numbers that are to be routed over the SIP trunk to BT Session Manager. Use ***change aar analysis x,*** where x is the first number in the dialed string. Set **Dialed String** to **x**, **Total Min/Max** to the length of the number to be dialed, **Route Pattern** to the one administered above and **Call Type** to **aar.**

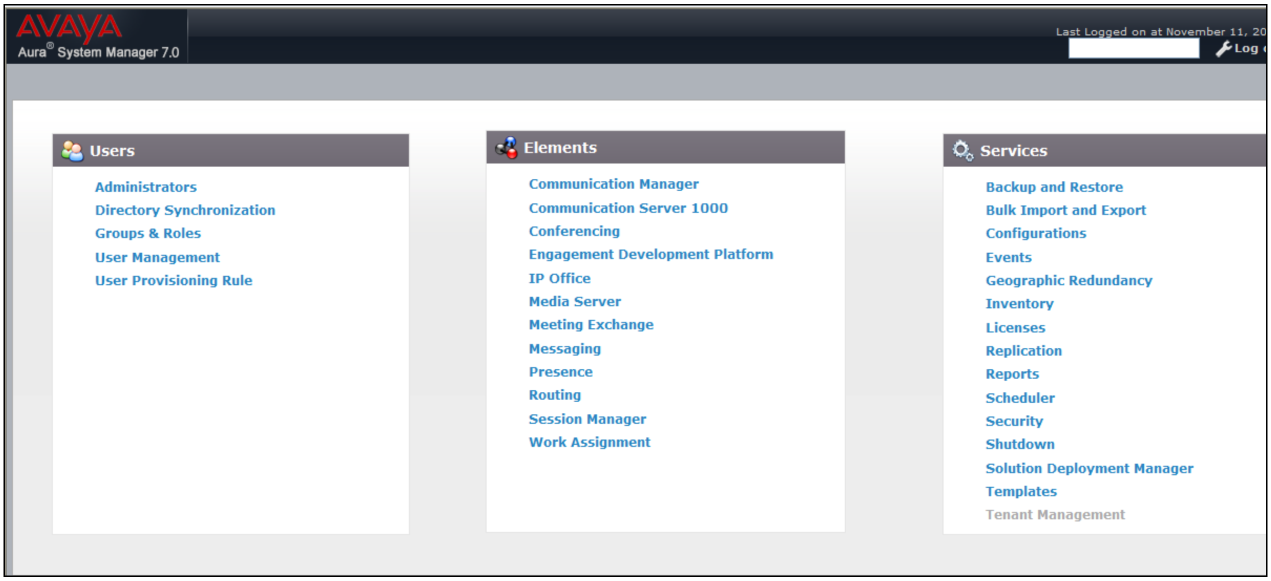
```
change aar analysis 3                                                 Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 2

          Dialed          Total     Route    Call   Node  ANI
          String         Min  Max  Pattern   Type   Num   Reqd
    3                    4    4     76       aar          n
```

6. Configure Avaya Aura® Session Manager

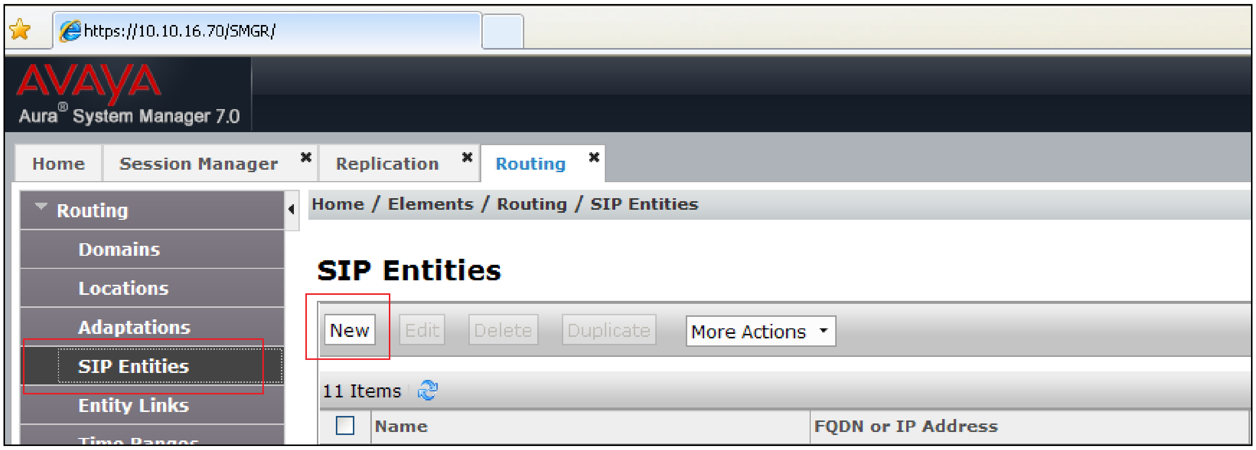
In this section, the configuration steps required to connect BT Session Manager to Session Manager as a SIP entity are described. It is assumed that an existing Session manager instance has already been installed and configured as this is out with the scope of this document. All Configuration steps were carried out using Avaya Aura® System Manager. Configuration steps will include:

- Adding a BT Session Manager SIP Entity.
- Adding an Entity Link.
- Adding a Routing Policy.
- Adding a Dial Pattern.

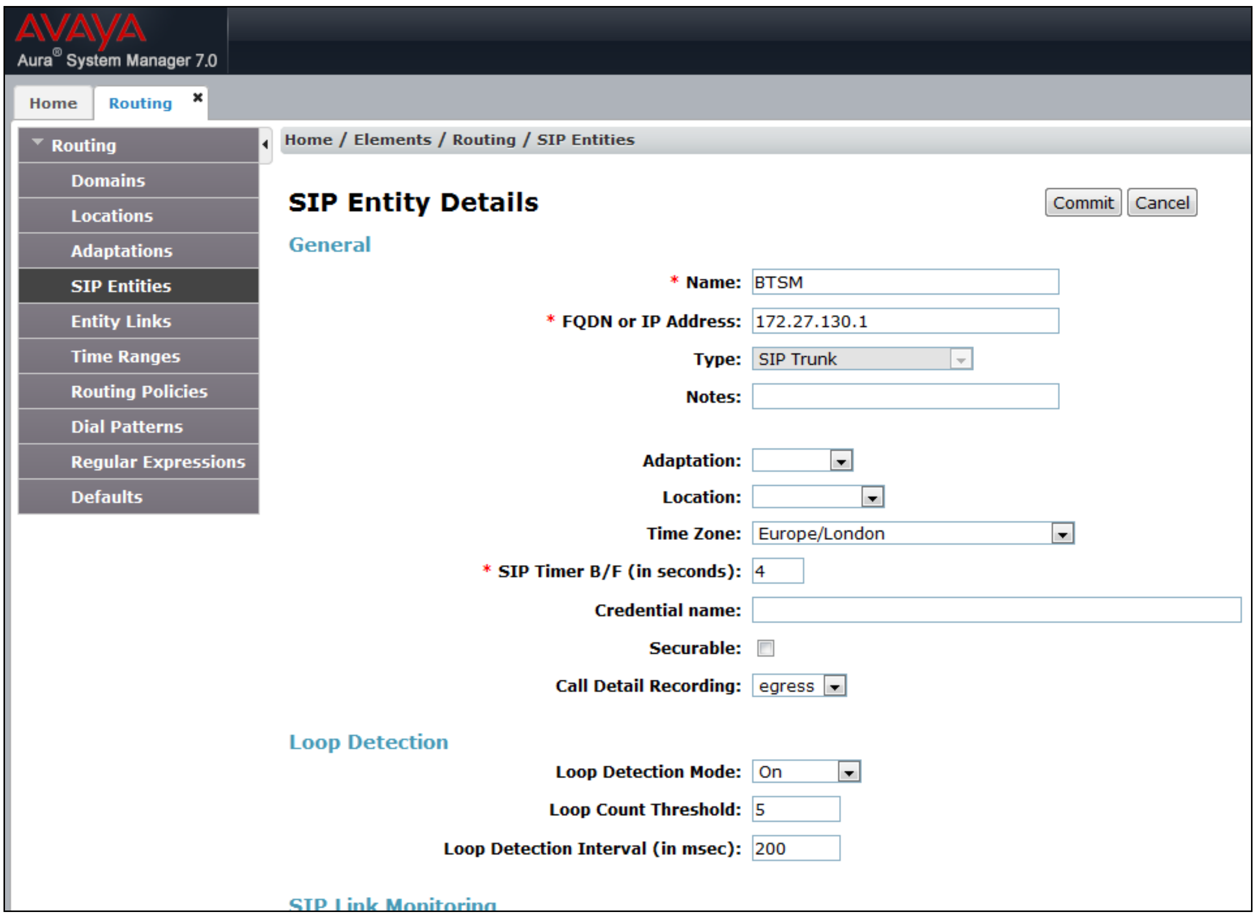
From the System Manager home screen select **Elements→Routing**



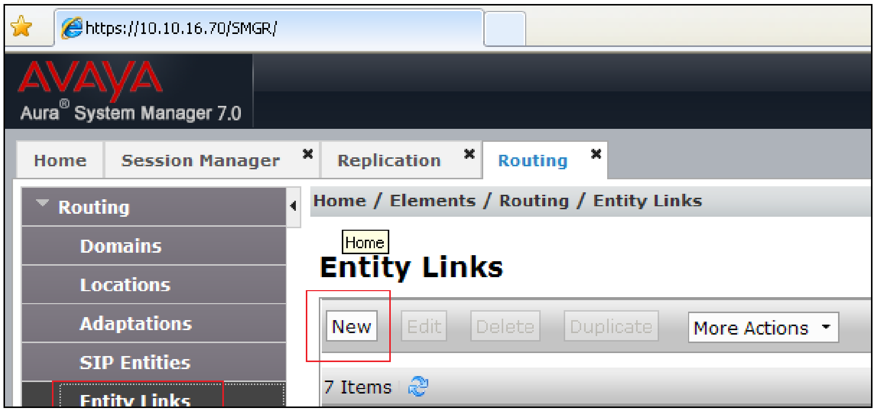
Select **SIP Entities** from the left hand menu and click on **New** to add the BT Session Manager entity.



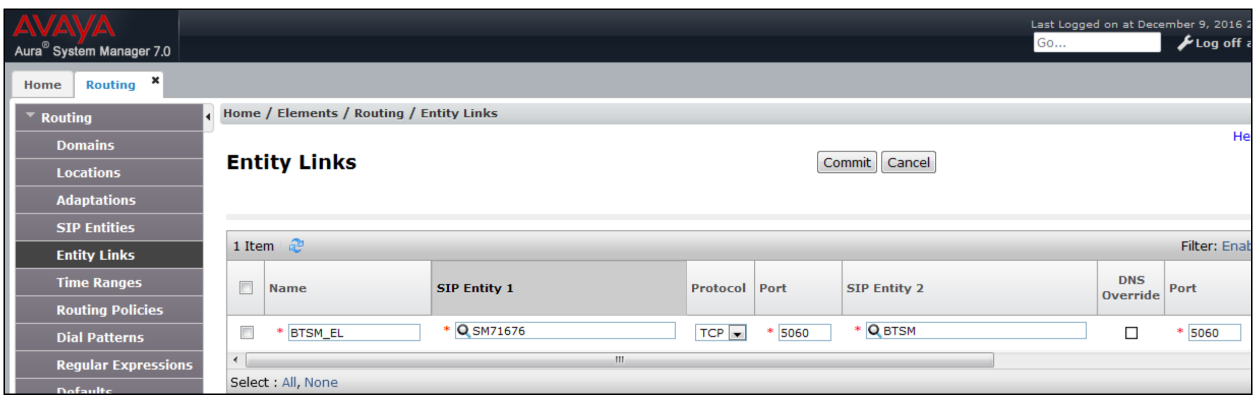
Enter a descriptive **Name** and the **FQDN or IP Address** of the BT Session Manager. Set **Type** as **SIP Trunk** and choose a **Time Zone** from the drop down menus. Click on **Commit** to save the changes.



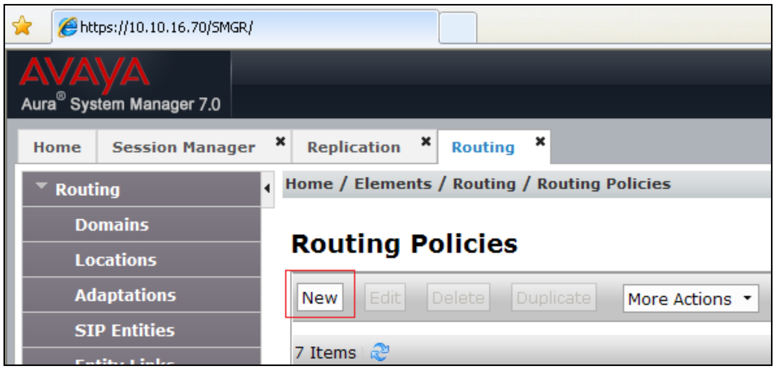
Next, add an Entity link between the BT Session Manager and Session Manager entities. Select **Entity Links** from the left hand menu and click on **New.**



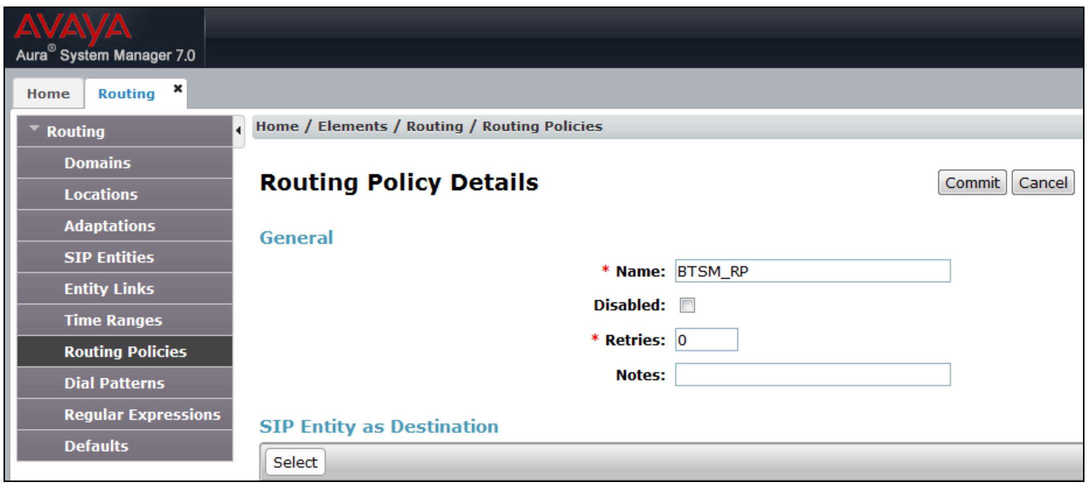
Enter a descriptive **Name** and then select the Session Manager as **SIP Entity 1** from the drop down. Select the **BTSM** entity as **SIP Entity 2**. Select the **Protocol** administered on the BT Session Manager server. **TCP** was used during testing. The ports will automatically change to the default **5060**. Click on **Commit** to save changes.



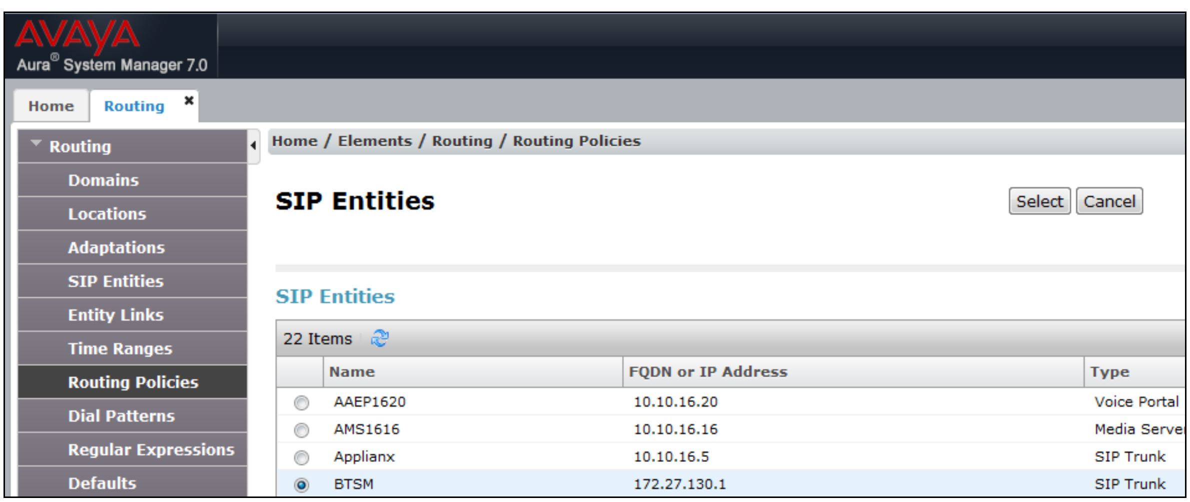
From the left hand menu, select Routing Policies (not shown) and click on **New**.



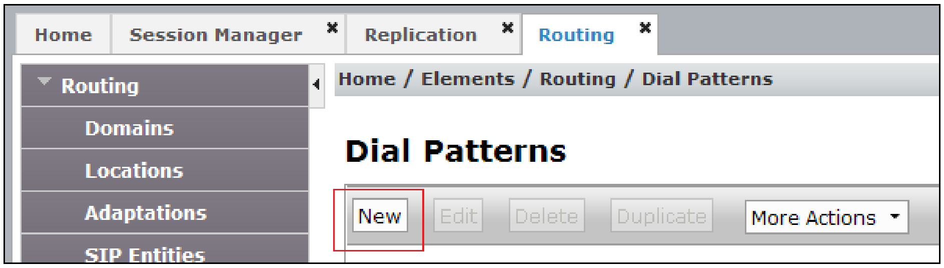
Enter a descriptive **Name** and under **SIP Entity as Destination** click on **Select**.



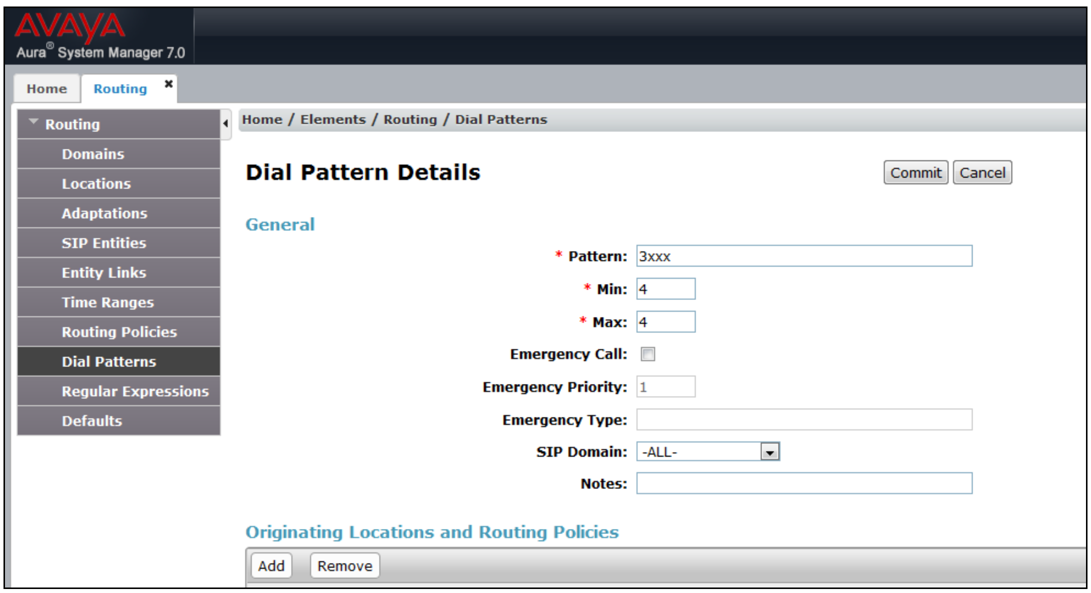
From the list of **SIP Entities,** select the **BTSM** entity and click on **Select** to save changes.



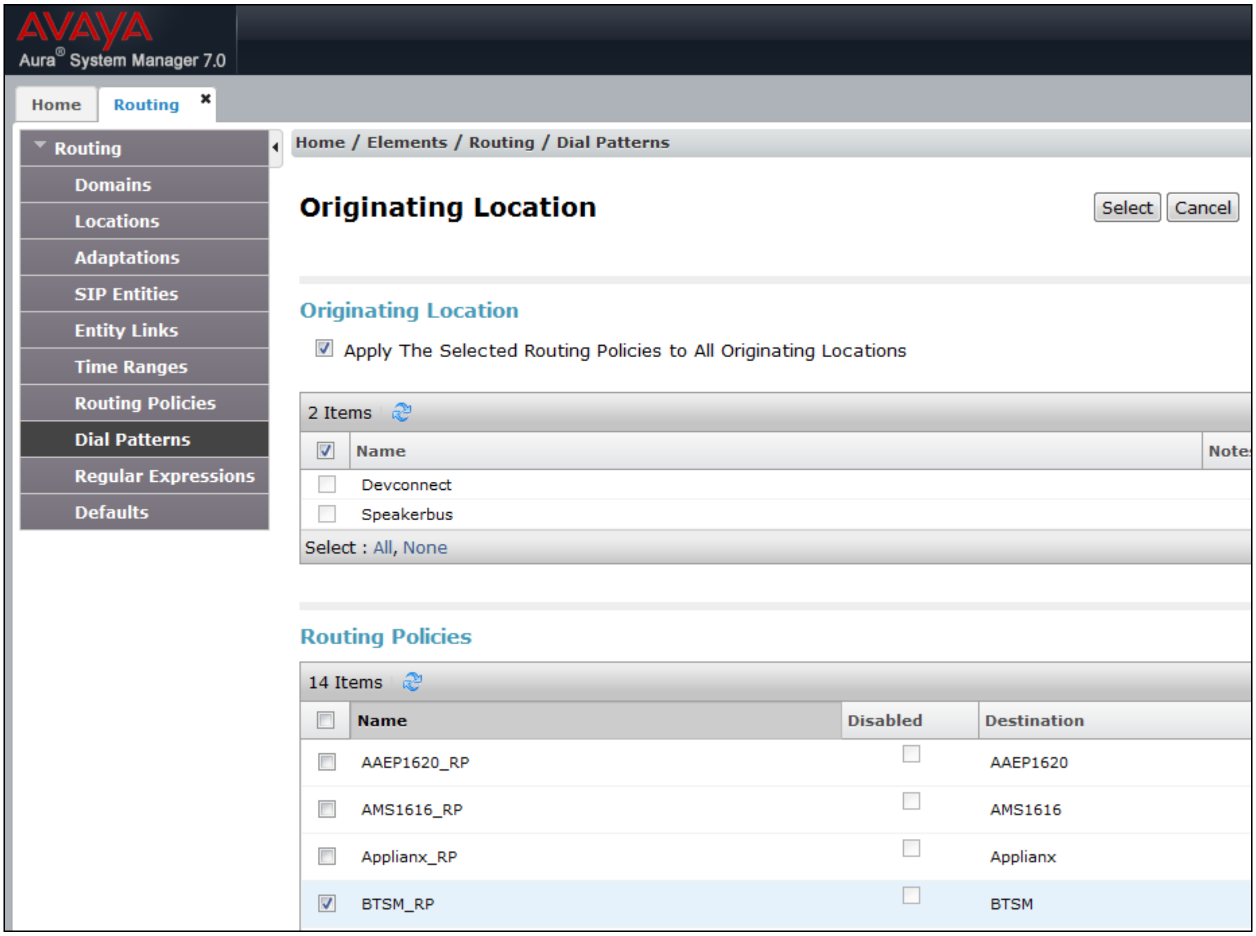
From the left hand menu, select **Dial Patterns** (not shown) and click on **New**.



Enter the **Pattern** that will route calls to the BT Session Manager server and set the **Min** and **Max** to the length of the number to be dialed. Under **Originating Location and Routing Policies** click on **Add**.

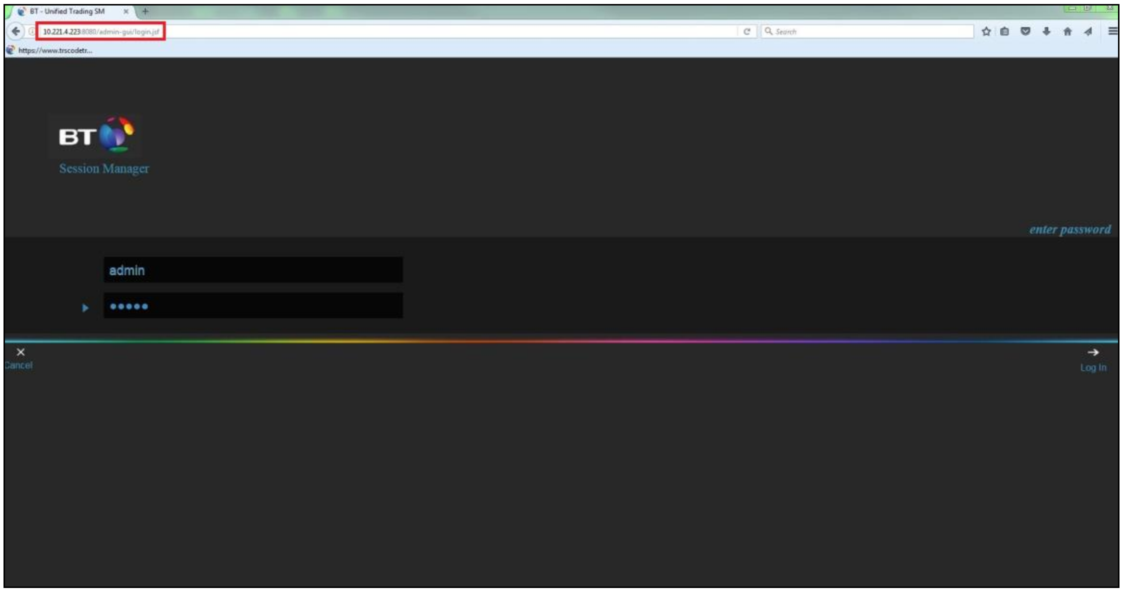


Select **Apply the Selected Routing Policy to All Originating Locations** and under **Routing Policies** select the BT Session Manager Routing Policy added above.

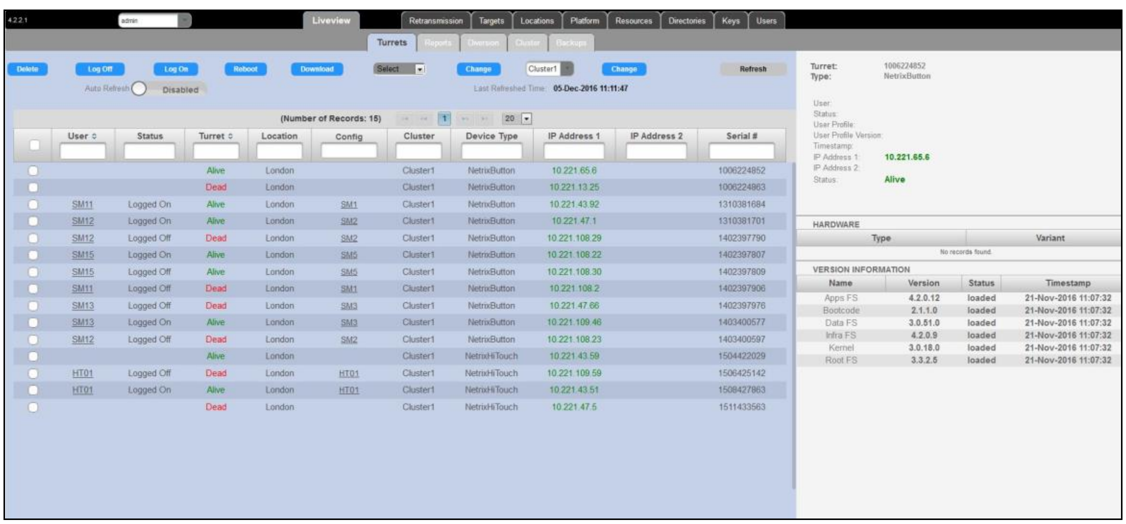


7. British Telecom Session Manager Configuration

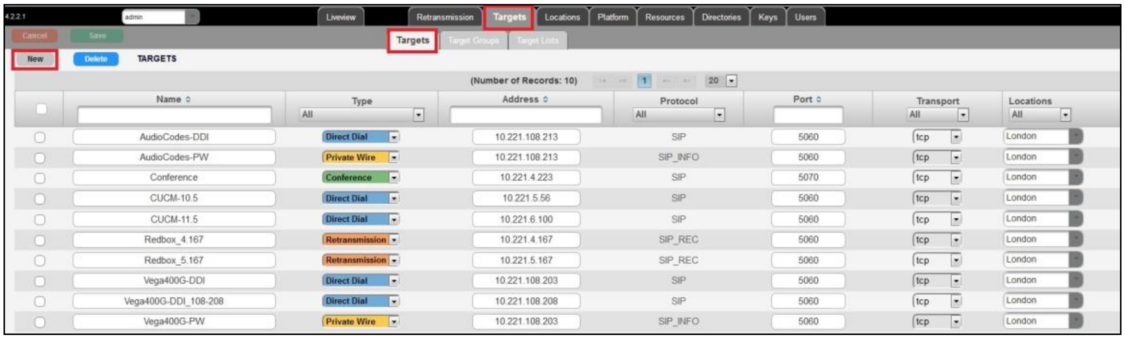
In the first instance, please browse to the BT Session Manager VIP Address followed by port 8080, for example http://10.221.4.223:8080



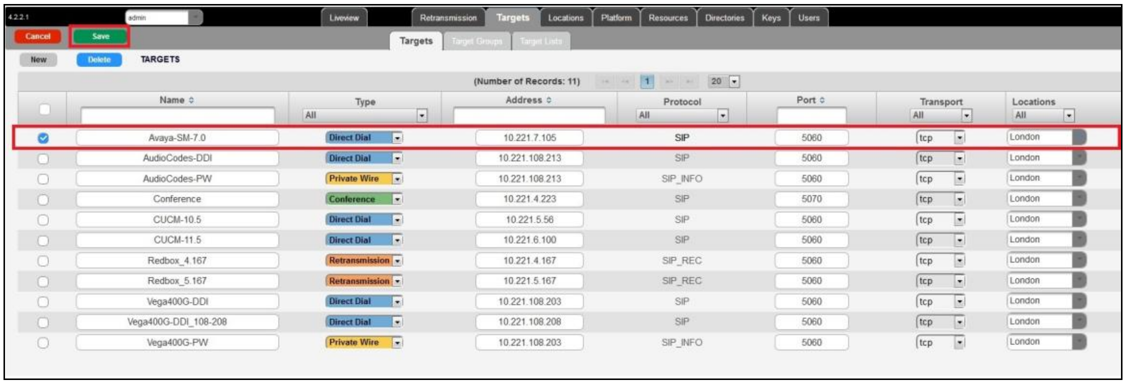
The LiveView screen is shown, which is the default webpage after logging in.



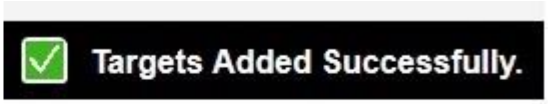
Using the tabs highlighted below, enter the Target configuration area and select **New.**



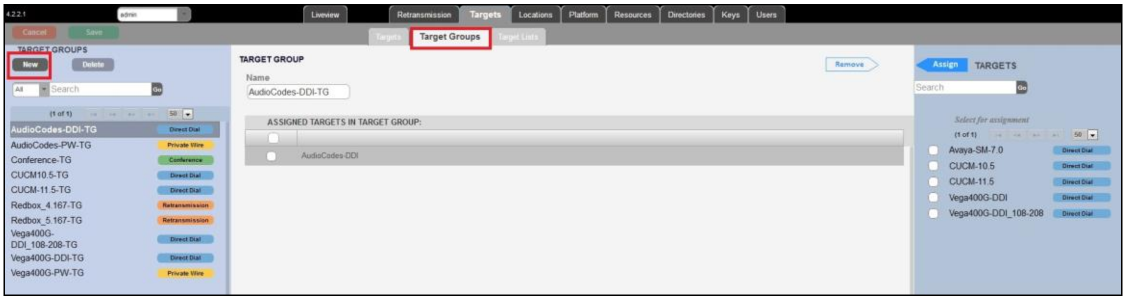
Enter the Target name, the IP Address of the Avaya Session Manager and the port and select **Save.**



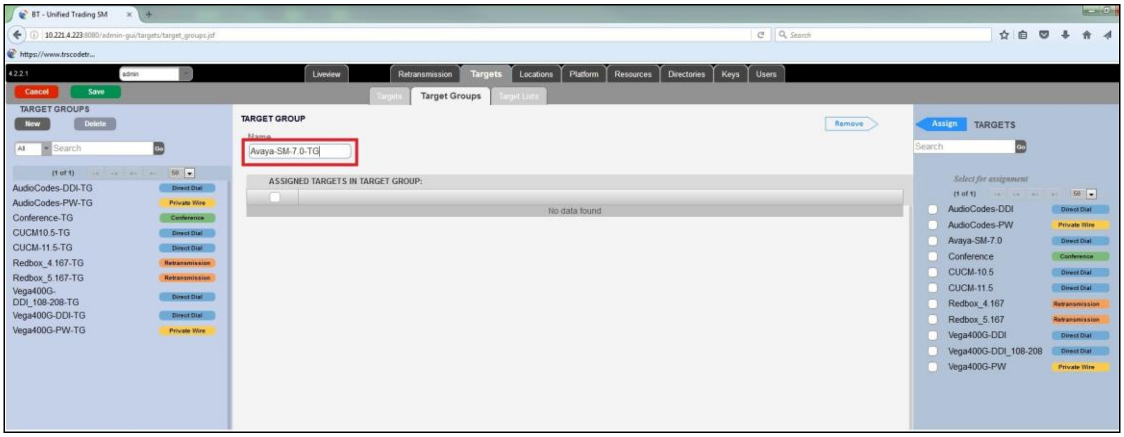
At the bottom the UI, check that the Target has saved to the Database successfully.



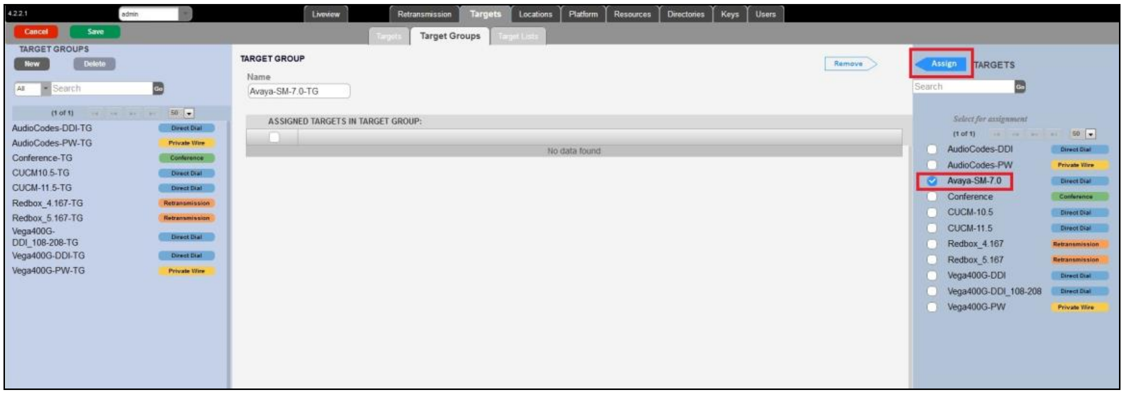
Now move to the Target Group tab as highlighted in the picture below and select **New.**



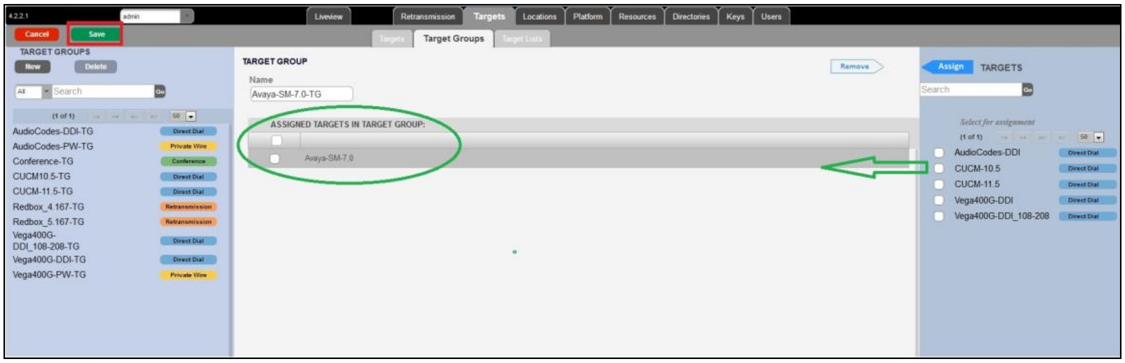
A text box will appear, enter your chosen Target Group name.



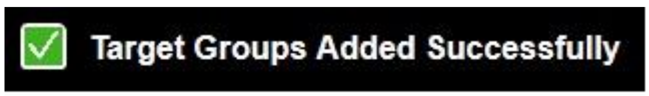
Select the Target created earlier by selecting the checkbox next to the Target Name and then select **Assign.**



The Target will now move into the centre pane as shown in the picture below, now select **Save.**

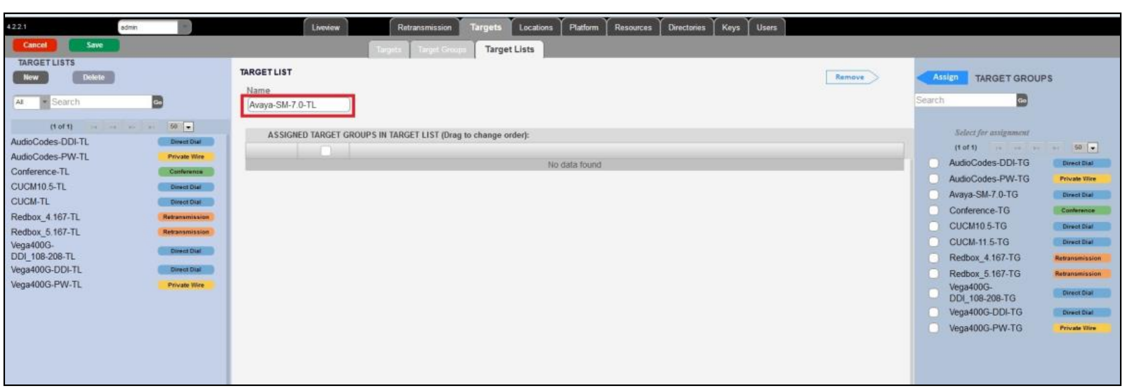


Ensure that the Target Group saves to the Database correctly by checking for the below message at the bottom of the UI.



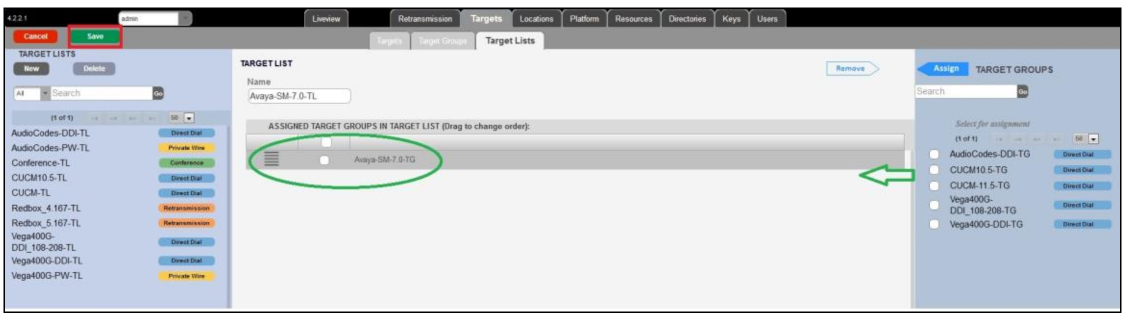
Now move to the Target List tab as highlighted in the picture below and select **New**

A text box will appear, enter the chosen Target List name.



Select the Target Group that was created earlier by selecting the checkbox next to the Target Name and then select **Assign.**

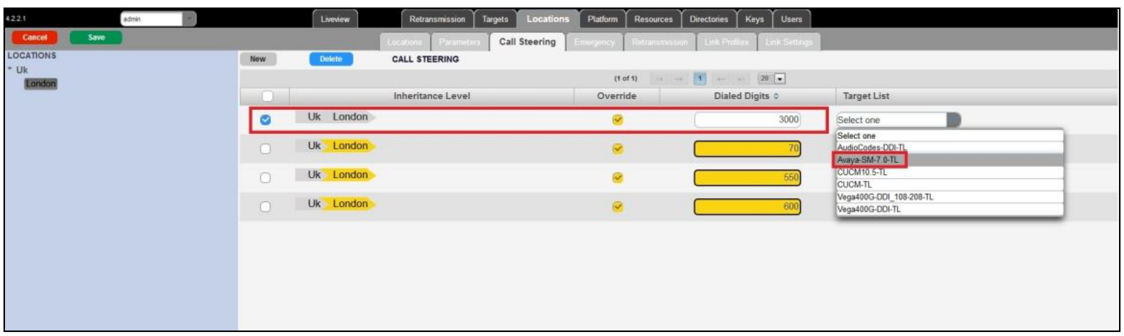
The Target Group will now move into the centre pane as shown in the picture below, now select **Save.**



Ensure that the Target List saves to the Database correctly by checking for the below message at the bottom of the UI.

The next step is to route the Avaya digits to the newly created SIP Trunk to the Avaya Aura® platform. This is defined in Call Steering, use the picture below to navigate to this area of the UI and select **New.**

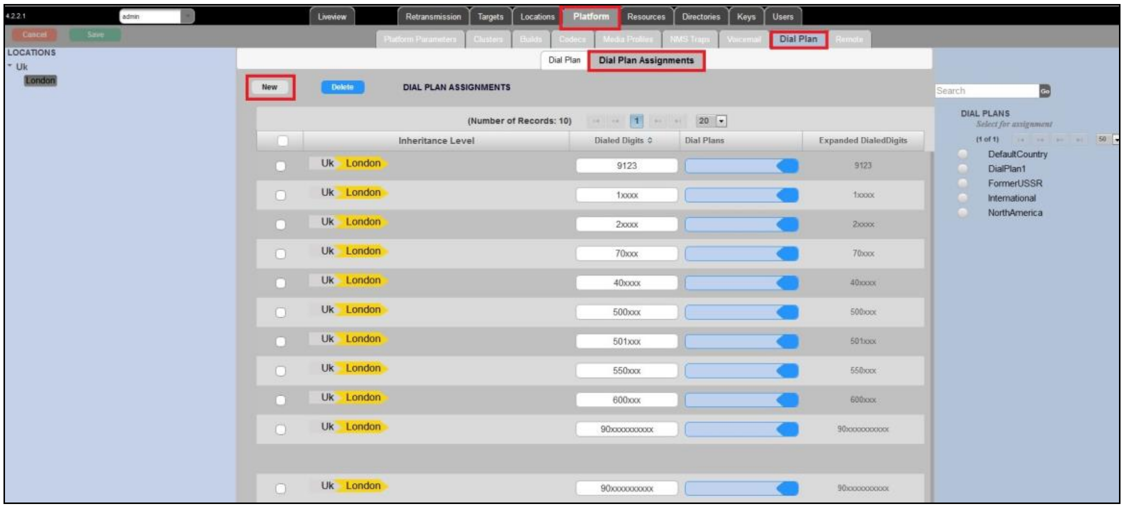
Enter the Digit information in the box that appears as shown below, here we are defining a range of 600000 - 600099. Use the drop down box to assign the Avaya Target List to these digits.



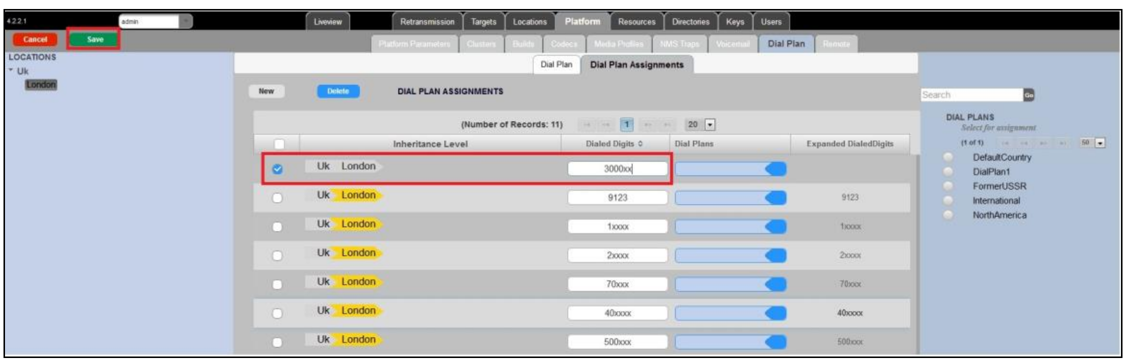
Select **Save**.

To confirm that the digits will then be saved to the Call Steering table, please ensure that you see the below message at the bottom of the UI.

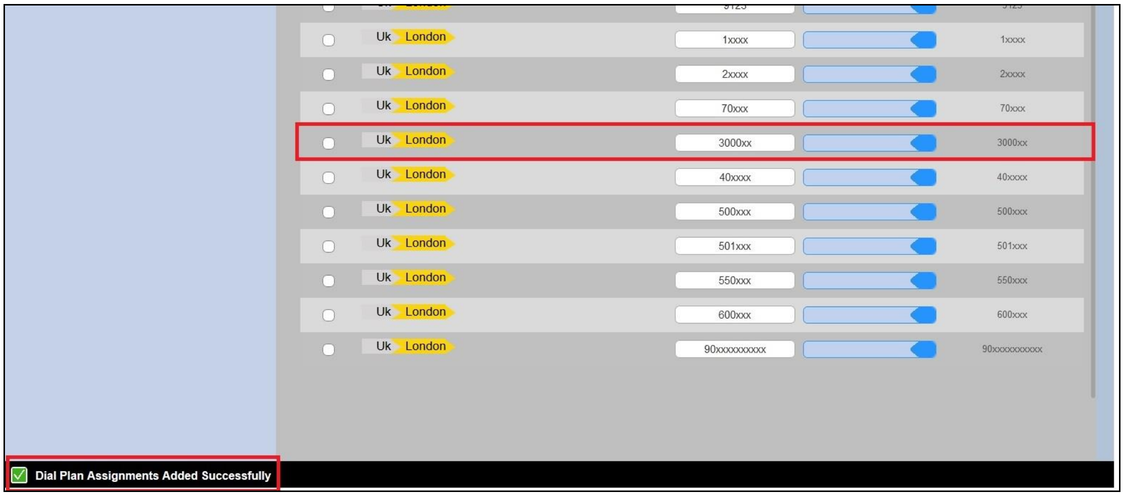
The next step is to define the digit string length into the Dial Plan to ensure that there is no unnecessary post dial delay. Navigate to the Dial Plan Assignment area as shown below and select **New.**



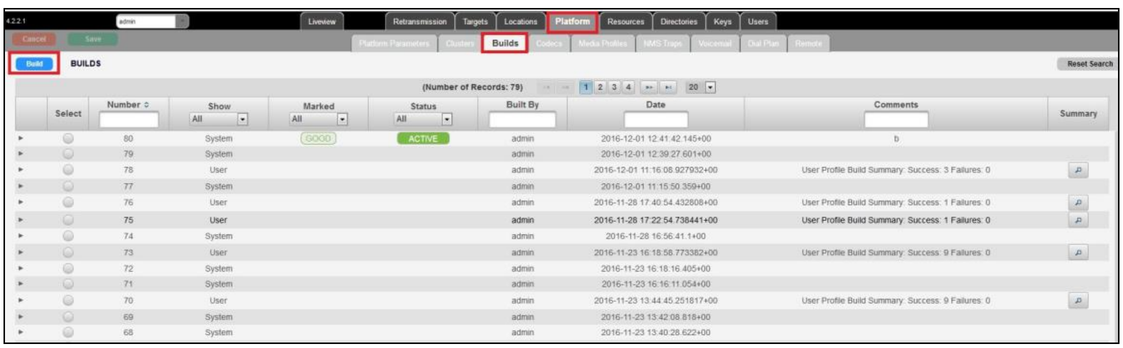
Complete the digit length by using the “X” character to denote any one single digit and select **Save.**



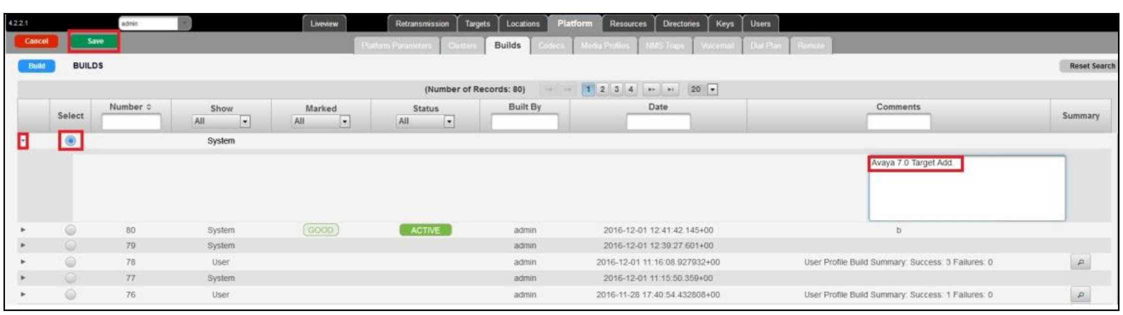
Ensure that the dial plan is saved by ensuring that the digits are entered correctly into the table after selecting Save and that the **Dial Plan Assignments Added Successfully** message can be seen at the bottom of the UI.



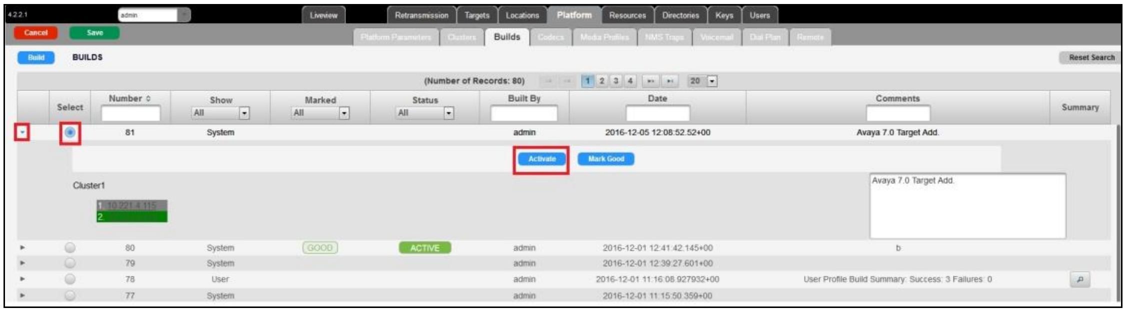
The last step is to save and activate the changes. Navigate to the Build page within the UI as shown below and select **Build.**



A new build window will appear as shown below. Enter the reason for the build and select **Save.**



Back at the Build page, find the build created and select the small arrow next to the Radio button as shown in the picture below and select **Activate.**



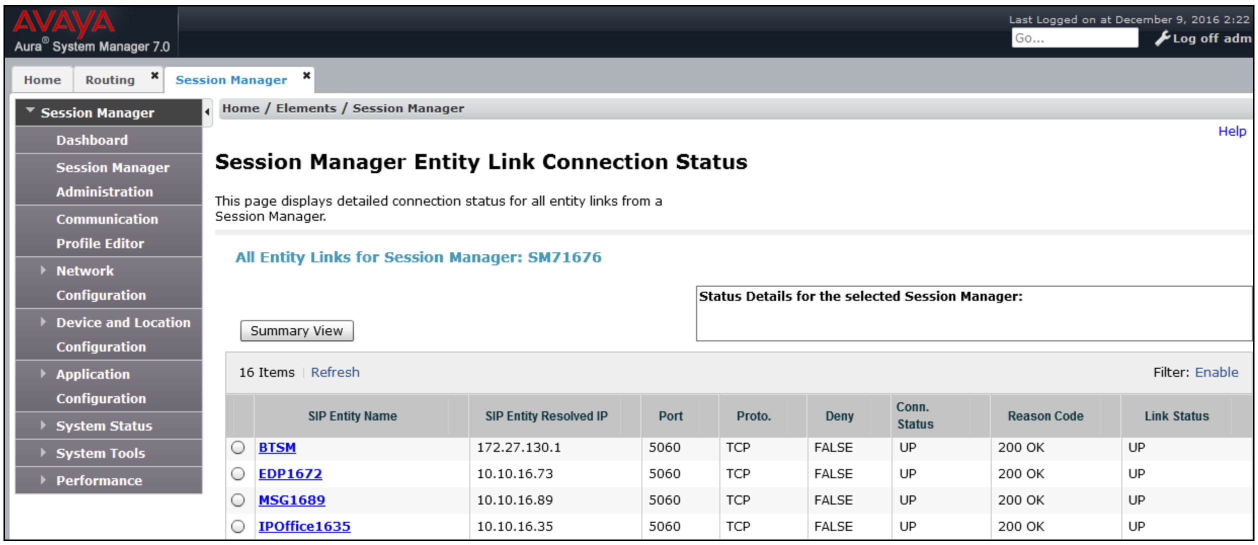
On Build page, the build will show up as **Active**.

8. Verification Steps

This section describes the checks that can be carried out to verify the connection between Netrix Turrets and , Session Manager and Communication Manager.

8.1. Avaya Aura® Session Manager Verification

From the main System Manager webpage, select Session Manager from the Elements section (not shown). Select Dashboard from the left hand menu and click on the entry under Entity Monitoring(not shown). The BT Session manager entity is listed and will show with **Conn. Status** and **Link Status** as **UP** and **Reason Code** as **200 OK**.

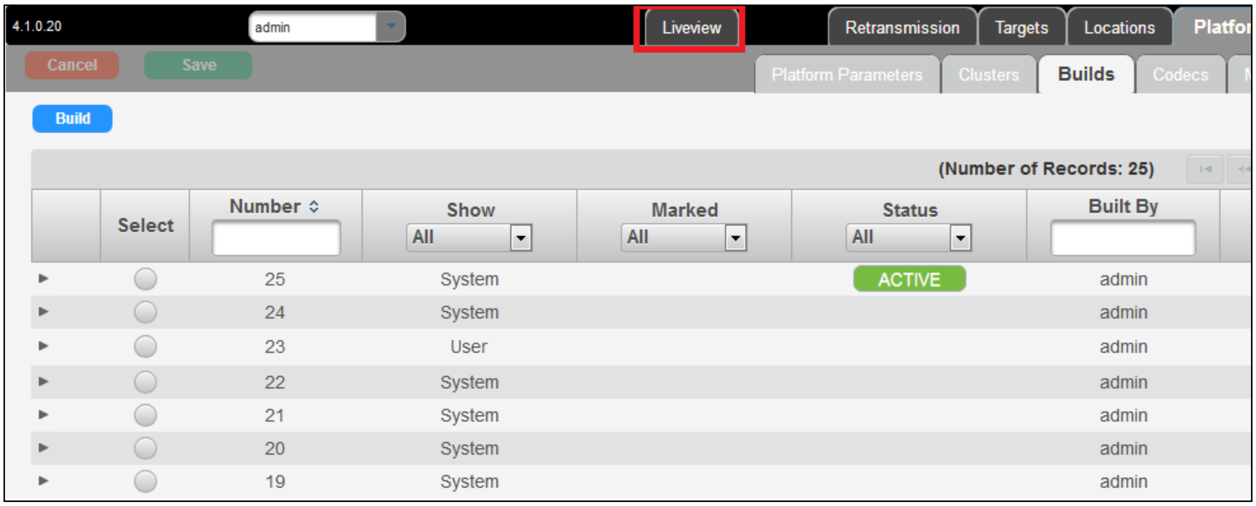


8.2. Netrix Verification

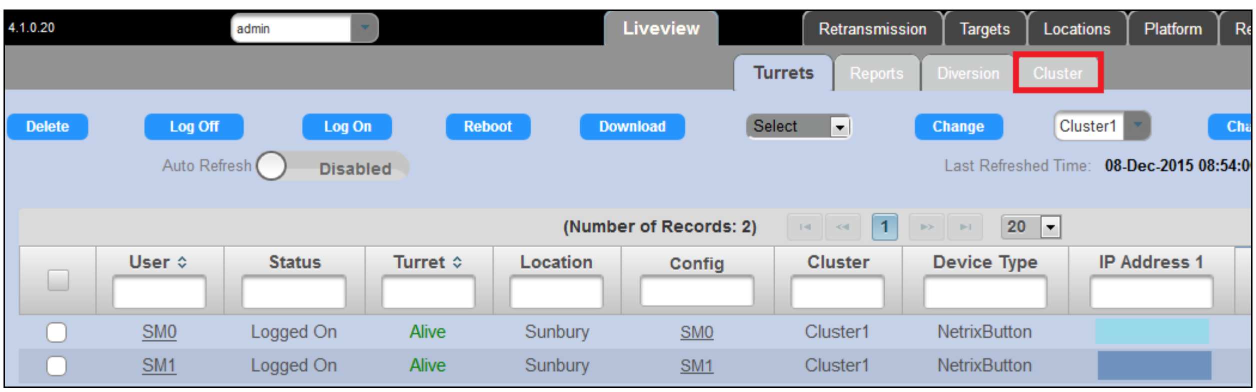
Log into BT Session Manager by browsing to the IP address of the BT Session Manager Virtual IP Address followed by the port used for the UI, this is normally 8080.

For this example, http://<Session Manager IP>:8080 is used.

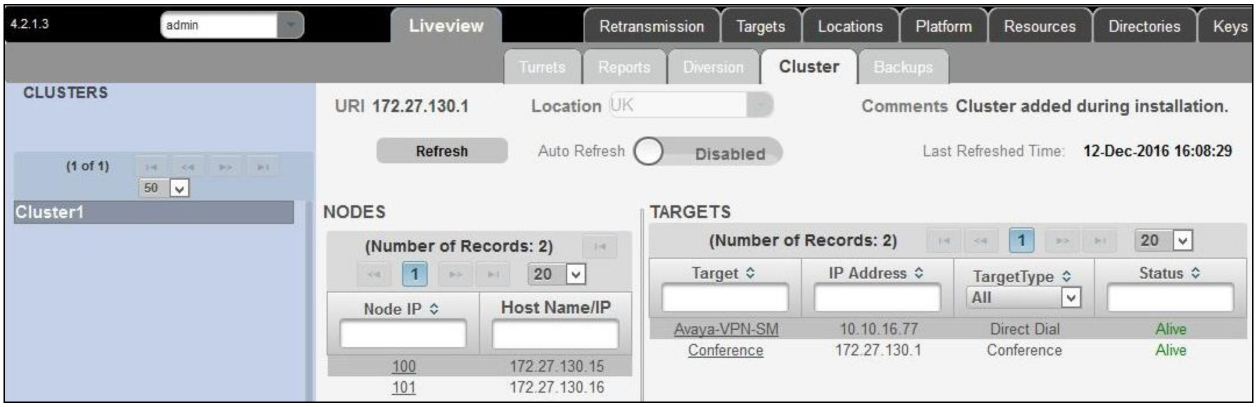
Select **Live View**.



Select **Cluster**.



Ensure that the Target created is showing as Alive in Liveview.



9. Conclusion

These Application Notes describe the configuration steps required for British Telecom Session Manager to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2**.

10. Additional References

This section references the Avaya and BT product documentation that are relevant to these Application Notes.
Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
[3] *Administering Avaya Aura® Session Manager*, Release 7.0, 03-603324
[4] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*, August 2015

Information regarding Product documentation for BT Netrix Trading Turret can be obtained by contacting the BT support email in **Section 2.3**

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.