Avaya Solution & Interoperability Test Lab

Application Notes for Datapulse Intuition Enterprise Server with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the Datapulse Intuition Enterprise Server to act as an attendant gateway using Avaya Aura® Session Manager to connect to Avaya Aura® Communication Manager. Calls are queued in the Datapulse Intuition Enterprise Server and then sent to the Datapulse Intuition Enterprise Attendant application where calls can be distributed as required. Datapulse Intuition Enterprise Server is a windows based application that can operate many instances of the Datapulse Intuition Attendant.

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

1. Introduction

The purpose of this document is to describe the steps required for Datapulse Intuition Enterprise Server to interoperate successfully with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Datapulse Intuition Enterprise Server is configured to receive calls sent by Avaya Aura® Communication Manager via Avaya Aura® Session Manager. Calls are queued by the Datapulse Intuition Enterprise Server and distributed to Datapulse Intuition Enterprise Attendant application. Calls can be distributed to agents, stations and external numbers as required using either a blind or announced method of transfer. The Datapulse Enterprise Server can also route unanswered or busy extension calls to the Intuition Enterprise Attendant application. These calls are presented to the Intuition Enterprise Attendant application as busy or unanswered calls giving the operator information with regard to the intended recipient of the call and the reason the call has been redirected.

2. General Test Approach and Test Results

To test the interoperability of Datapulse Intuition Enterprise Server with Session Manager and Communication Manager, calls were presented and distributed to the Intuition Enterprise Attendant application using a number of scenarios designed to test the features of Datapulse Intuition Enterprise Server.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The feature testing verified Intuition Enterprise Server's ability to receive and process inbound calls. The serviceability testing introduced failure scenarios to see if Intuition Enterprise Server could recover from failures in connectivity to Session Manager and Communication Manager. Interoperability compliance testing covered the following features and functionality

- Ability for an inbound call from different sources to be answered.
- Ability for an inbound call to be forwarded to the required number.
- Ability for an inbound call to be retrieved when intended recipient is not available.
- Ability for the Intuition Enterprise Attendant application to make calls receive and transfer calls.

2.2. Test Results

During testing Intuition Enterprise Server carried out all tasks with the expected outcome in each scenario.

2.3. Support

For technical support of Datapulse products contact the Datapulse Service Desk:
Web: http://www.datapulse.com
Email: support@datapulse.com
Telephone: +1 800 657 1530 in North America
+44 (0) 118 972 8400 in Europe
+61 433 986 344 in Asia
+ 971 4 501 5600 in Middle East & Africa

3. Reference Configuration

The configuration used as an example in these Application Notes is shown in **Error! Reference source not found.**. The diagram illustrates an Enterprise SIP environment. The configuration consists of an S8800 server running Communication Manager connected to a G650 Media Gateway, Session Manager and System Manager that allow LAN connectivity to H.323 Avaya IP handsets. The configuration also includes a Windows 2003 server running Intuition Enterprise Server and a Windows PC running Intuition Enterprise Attendant.

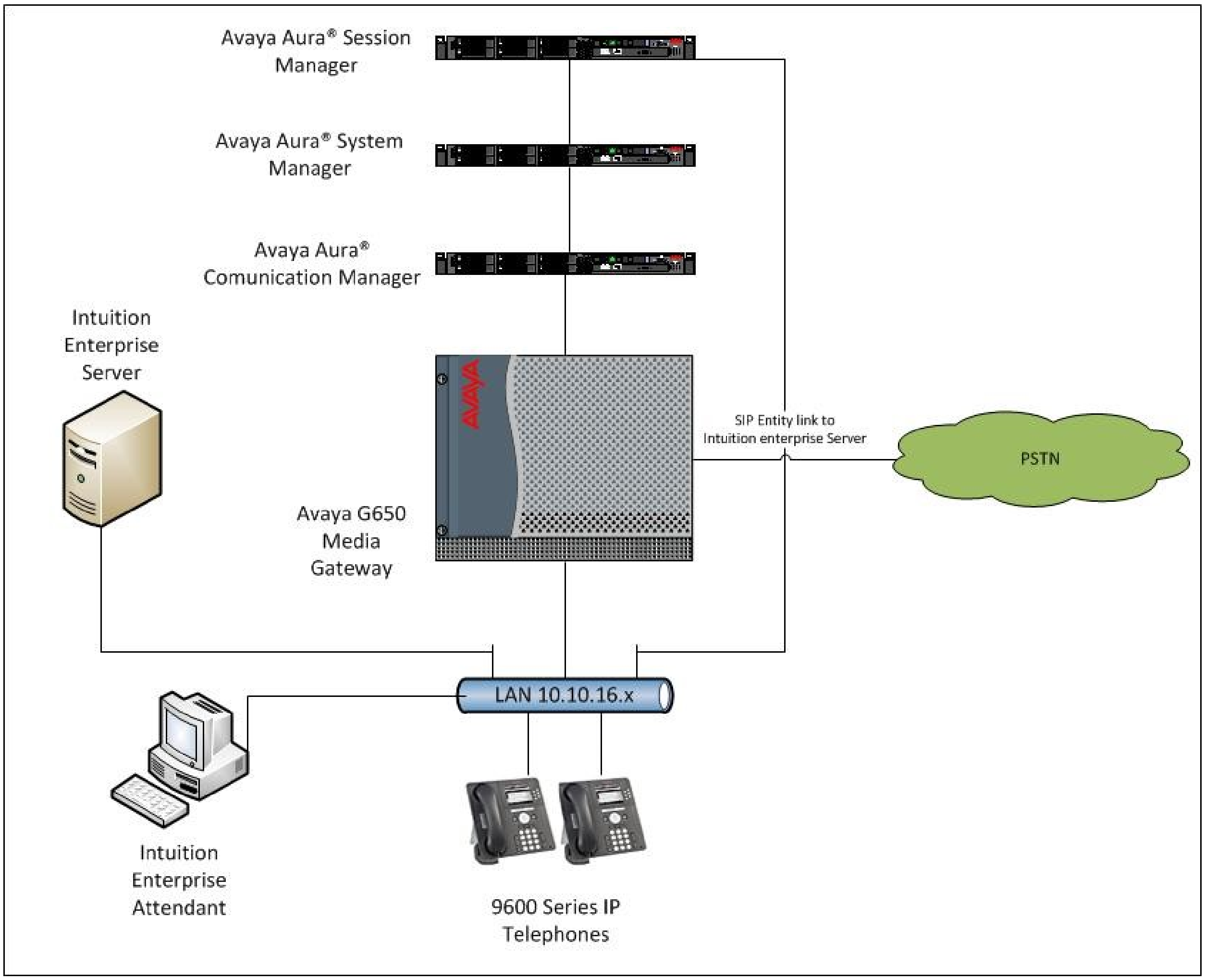


Figure 1: Intuition Enterprise Server with SIP Enterprise Configuration.

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server with G650 Media Gateway	Avaya Aura® Communication Manager 6.0.1 (R016x.00.0.345.0) with Service Pack 1 (Patch 18444)
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (Build 6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager 6.1 (Build 6.1.0.4.5072-6.1.4.113)
Avaya 9600 Series Handsets	3.1 (H.323)
Datapulse Intuition Enterprise Server	3.1
Datapulse Intuition Enterprise Attendant	3.1

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the Communication Manager to communicate with Intuition Enterprise Server via a SIP Trunk. Log in to Communication Manager with the appropriate credentials. The following steps are performed using the SAT (System Access Terminal) on Communication Manager:

- Verify system capacity
- Define the Dial Plan
- Define Node Name
- Define SIP Trunk to Avaya Aura® Session Manager
- Define Call Routing

5.1. Verify System Capacity

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 the **Maximum Administered SIP Trunks** allowed in the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  11
                                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 5
                       Maximum Administered SIP Trunks: 24000 349
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Define Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. The number used to access Intuition enterprise server must be administered as an **aar** type. A **4** digit entry using **4** as the first number is used in this example.

```
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
  309        4    ext
  350        4    ext
  4          4    aar
```

5.3. Define Node Name

Use the **change node-names ip** command to add the Session Manager security module **Name** and **IP Address.** The **CLAN** IP address should also be noted for use in the **Near-end Node Name** field in **Section 5.4.**

```
change node-names ip
                                  IP NODE NAMES
   Name                 IP Address
AES522               10.10.16.25
CLAN                 10.10.16.31
CM521                10.10.16.23
Gateway              10.10.16.1
IPbuffer             10.10.16.184
Intuition            10.10.16.51
SM61                 10.10.16.201
```

5.4. Define SIP Trunk

Use the **add signaling-group** command to define the **Group Type, Transport Method, Near-end and Far-End Node Names** and **Ports** to be used by the SIP Trunk.

```
add signaling-group 6
                                SIGNALING GROUP

 Group Number: 6              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                          SIP Enabled LSP? n
     IP Video? n                                Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: CLAN                   Far-end Node Name: SM61
 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? y
Session Establishment Timer(min): 3                 IP Audio Hairpinning? n
         Enable Layer 3 Test? n                Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Use **add trunk-group** command to define the SIP Trunk group. On **Page 1** the **Group Type** is set to **sip** and the **Service Type** is set to **tie**. The **Signaling Group** administered above is added as well as the **Number of Members** required.

```
add trunk-group 6                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 6                    Group Type: sip           CDR Reports: y
  Group Name: Intuition Server            COR: 1        TN: 1       TAC: 707
   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: 6
                                                     Number of Members: 10
```

On **Page 3** set the **Numbering Format** to **private**

```
add trunk-group 6                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y



                     Numbering Format: private
                                             UUI Treatment: service-provider

                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n



                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.5. Define Call Routing

Use the **change aar analysis** command to configure the **Dialed String, Route Pattern**, **Call Type** and **Total Min/Max** digits to be dialed.

```
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
           444             4    4     6        unku          n
```

Use the **change route-pattern** command to configure the trunk **Grp No** and **FRL**.

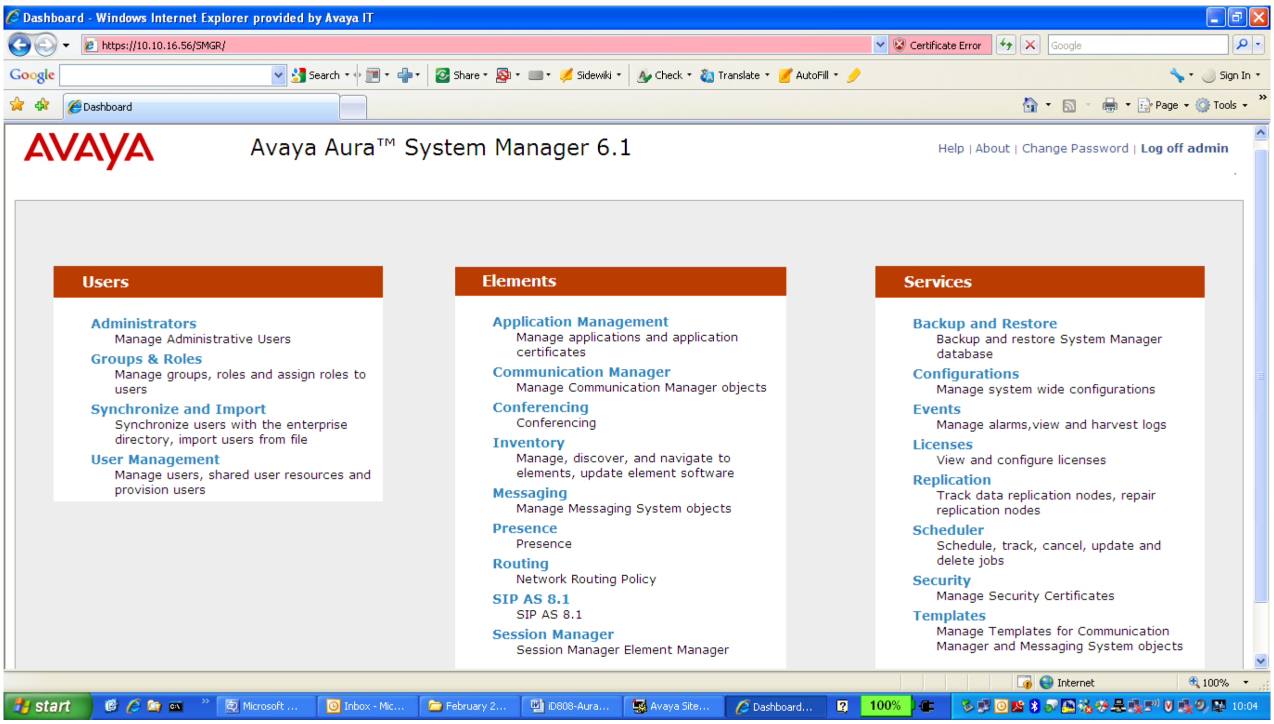
```
change route-pattern 6                                       Page   1 of   3
                    Pattern Number: 7   Pattern Name: Intuition
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 6    0                                                           n  user
```

6. Configure Avaya Aura® Session Manager

These steps are designed to set up Session Manager to communicate with Intuition Enterprise Server as a SIP Entity and allow calls to be routed.

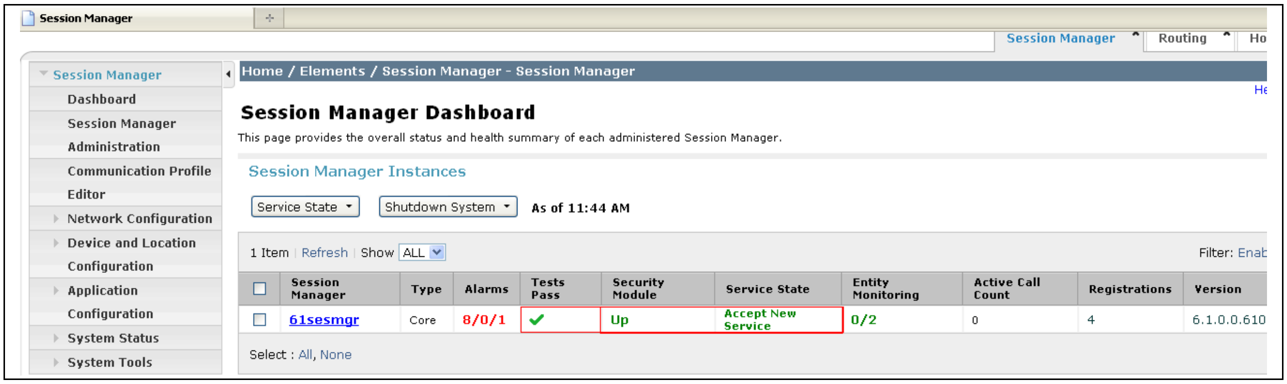
6.1. Logging in to Avaya Aura® System Manager

To access the administration web interface, enter **http://<ip-addr>/SMGR** as the URL in an Internet browser. Where <ip-addr> is the IP address of System Manager on System Platform. Log in with the appropriate credentials. The Dashboard screen is displayed, as shown below.

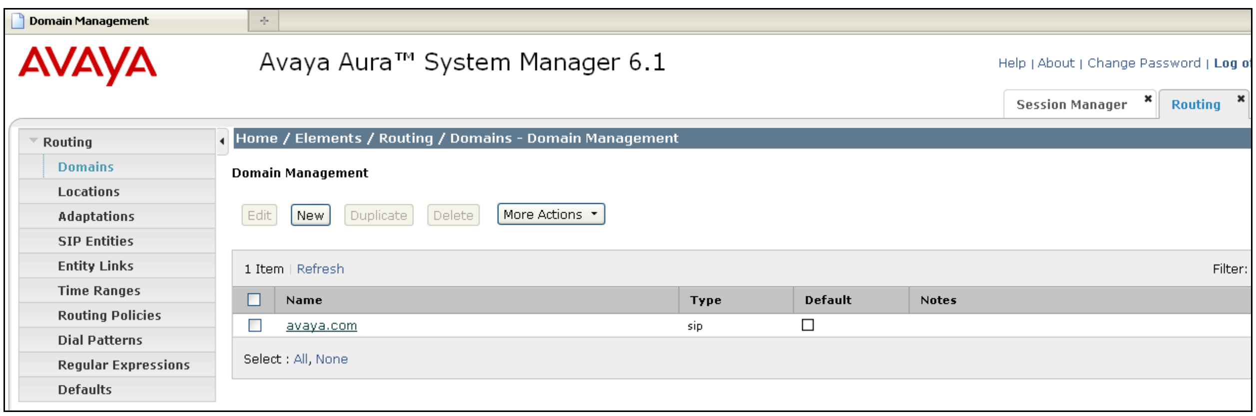


6.2. Verify System Properties

From the Dashboard of the web interface, choose Session Manager from the Elements section. Verify that a green tick shows under **Tests Passed**, **Security Module** is **Up** and **Service State** is set to **Accept New Service**.

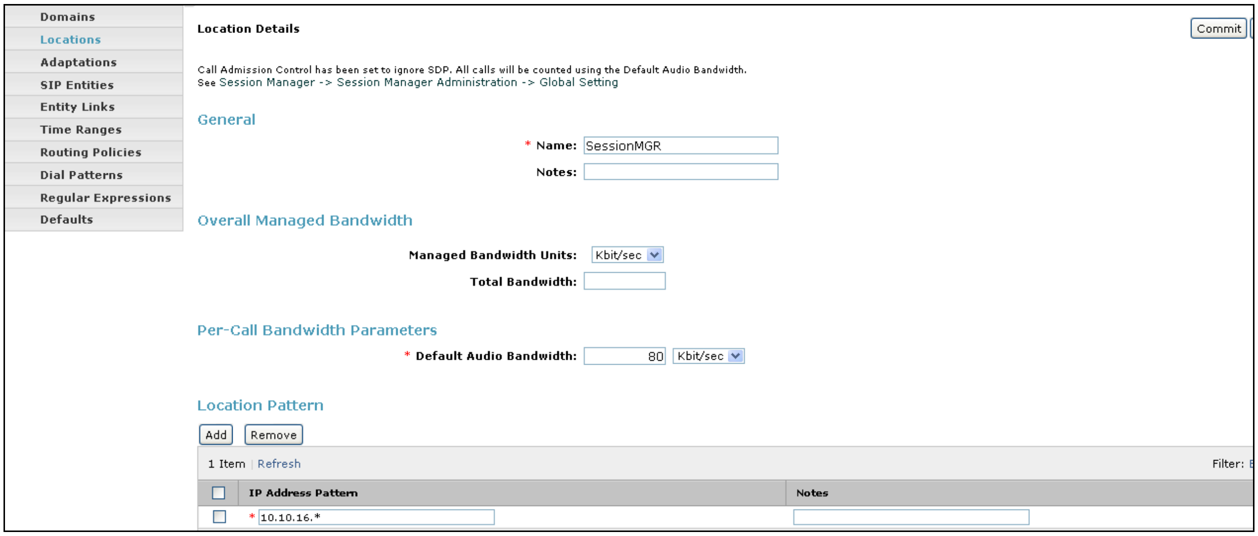


Next go to Routing from the Elements section of the Dashboard and select Domains and check the administered domain.



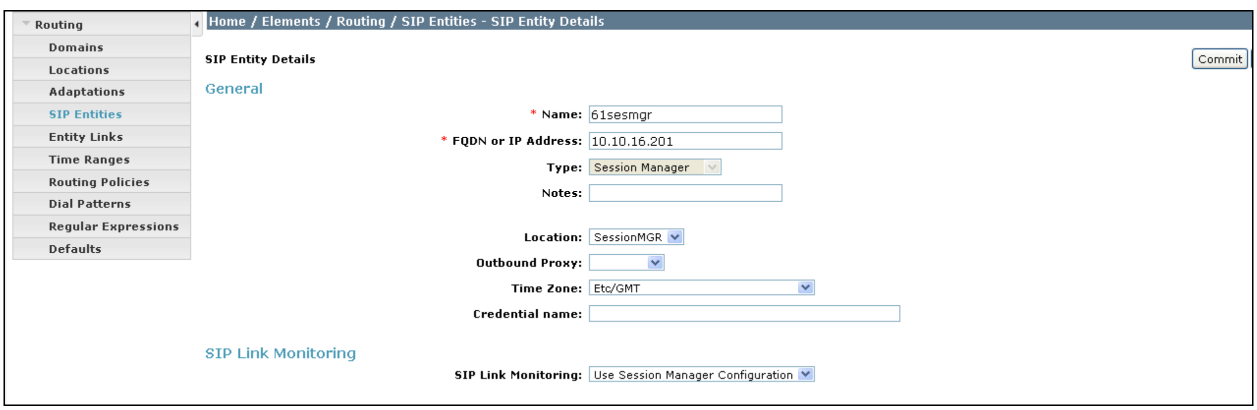
6.3. Add Location

Select Routing form the Elements section of the Dashboard and chose **Locations.** Click on the new button (not shown) and add a **Name** and the IP Address range in the format shown under **Location Pattern → IP Address Pattern**. Click on the **Commit** button to save.

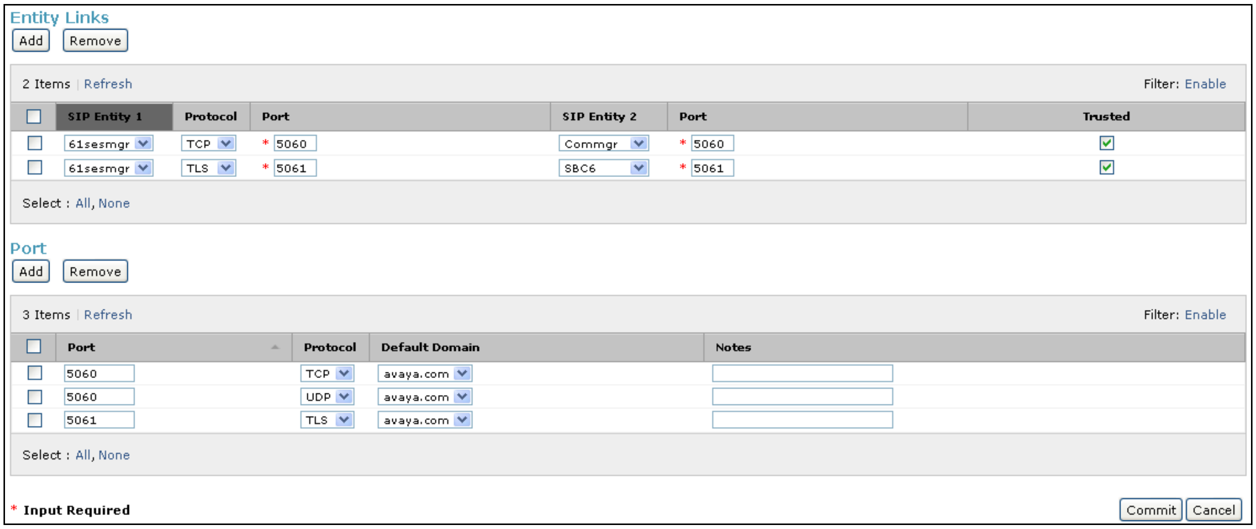


6.4. Create a SIP entity

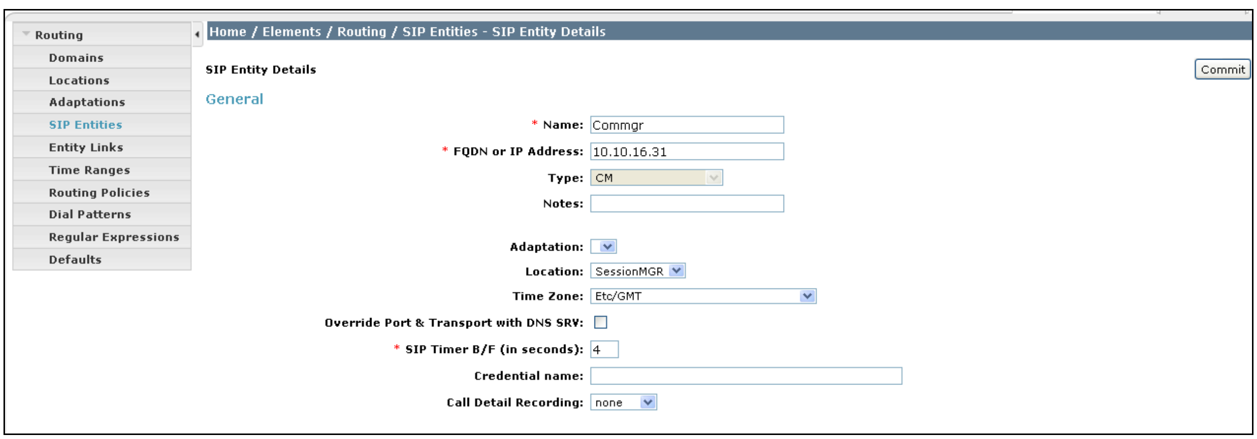
From the Elements section of the Dashboard choose **Routing**. From the left hand side menu choose **SIP Entities**. Click on the **New** button (not shown) and enter the required information, **Location** of the Session Manager Manager created in **Section 6.3** and set the **Type** to Session Manager.



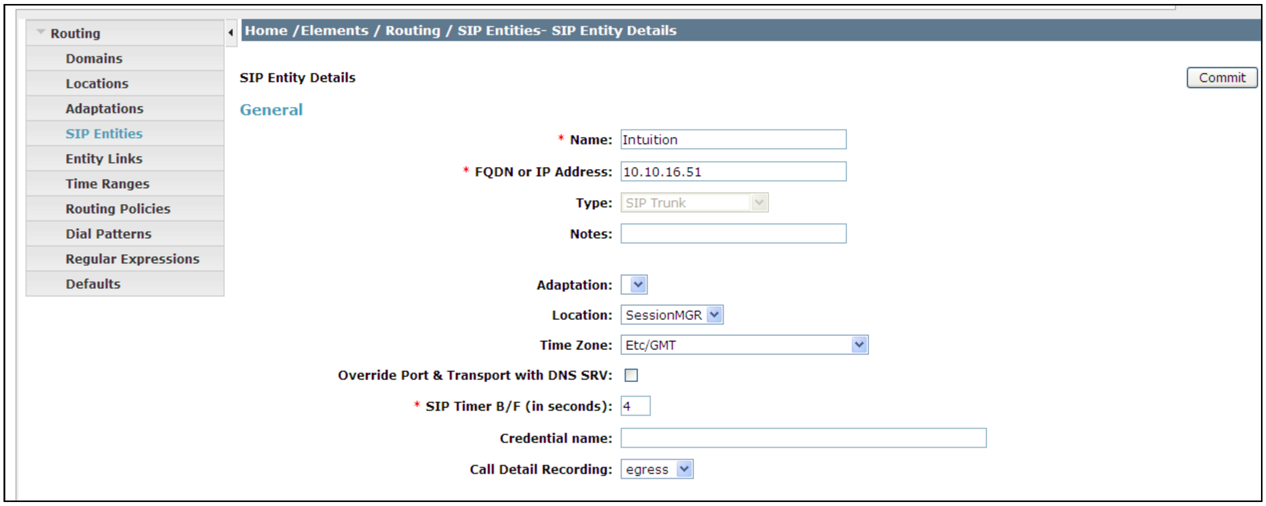
Add the protocol and port information in the **Port** section shown below. The **Entity Links** section will automatically populate after the link is added in **Section 6.5** Click **Commit** to save the changes.



Repeat above steps to add Communication Manager entity by setting the **Type** field to **CM**. Click on the **Commit** button to save.

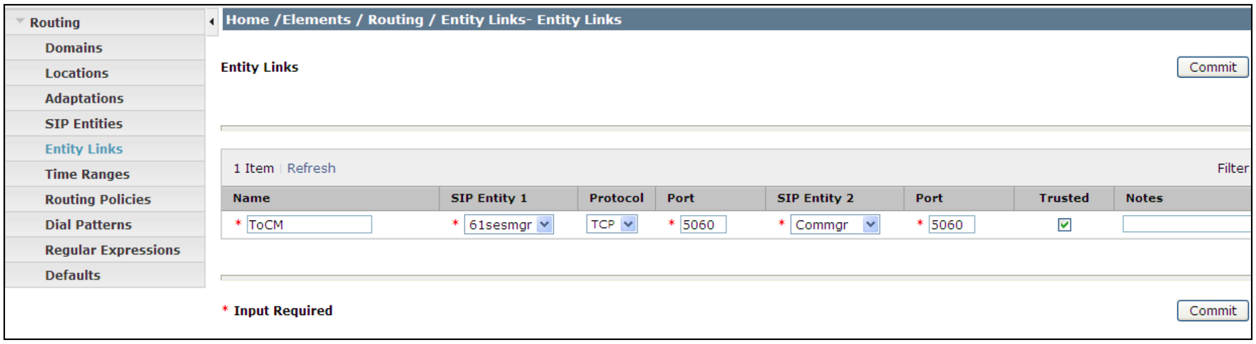


Repeat above steps to add the Intuition Enterprise Server by setting the **Type** field to **SIP Trunk**. Click on the **Commit** button to save.



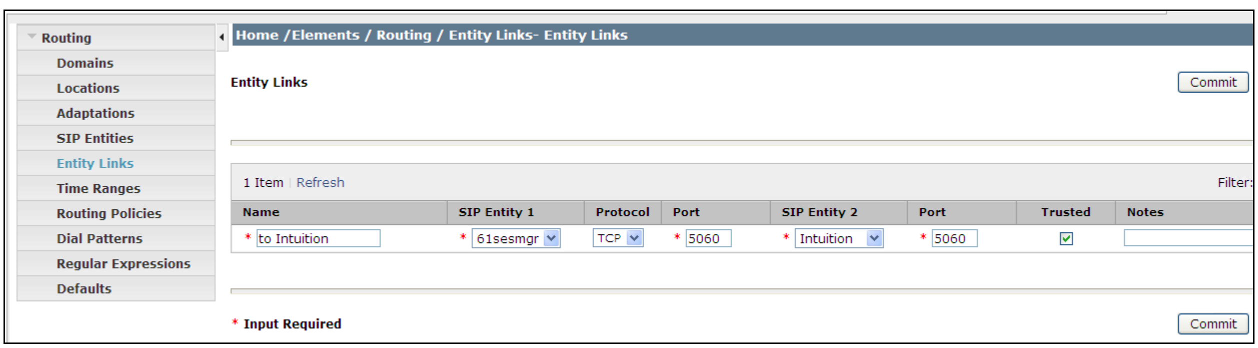
6.5. Add a Communication Manager Entity Link

From the Routing menu choose **Entity Links**. Click on the New button (not Shown) give the link an appropriate **Name** and then choose the entities added in **Section 6.4,** the **Protocol** to use (TCP used in this example) and the **Port** used to communicate on. Click on the **Commit** button to save.



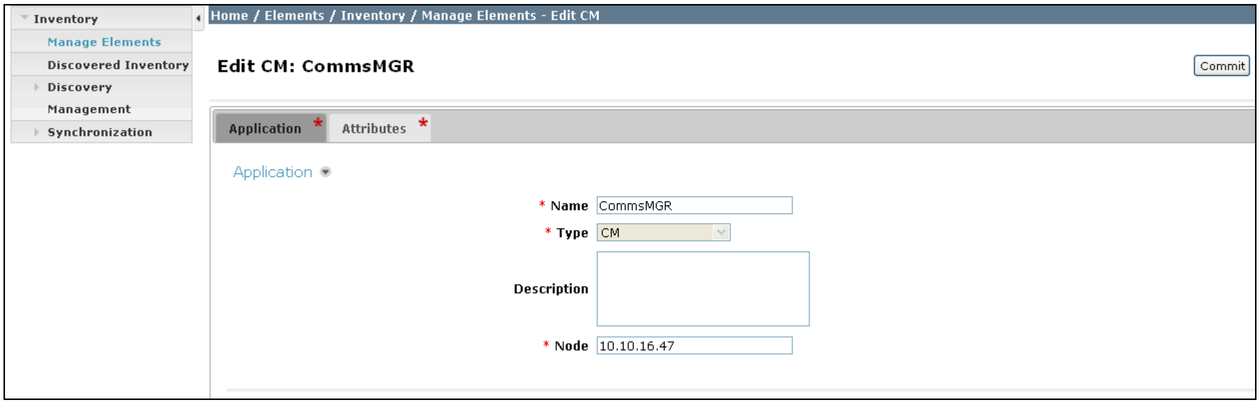
6.6. Add an Intuition Entity Link

From the **Routing** menu choose **Entity Links**. Click on the New button (not Shown) give the link an appropriate name and then choose the entities added in **Section 6.4,** the **Protocol** to use (TCP used in this example) and the **Port** used to communicate on. Click on the **Commit** button to save.



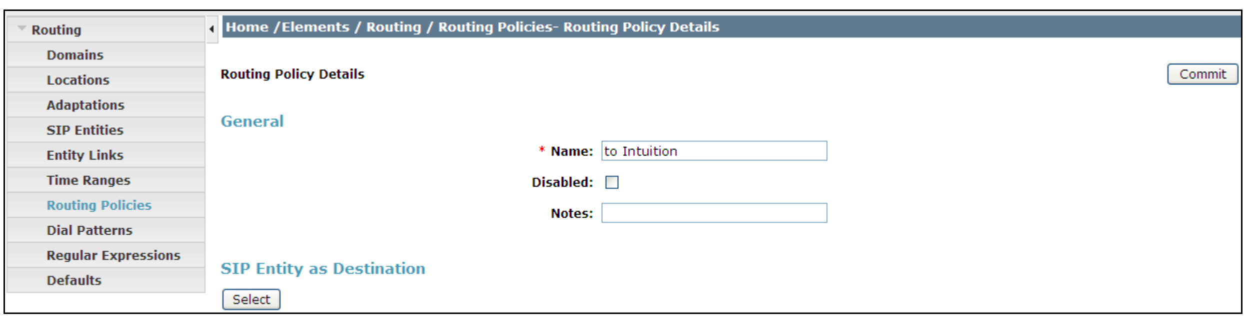
6.7. Add Communication Manager Managed Element

From the **Elements** section of the Dashboard choose **Inventory** and then **Manage Elements**. Click the New button and enter a valid **Name**, **Type** as **CM** and the **Node** as the Communication Manager Server IP address. Click on **Commit** to save.

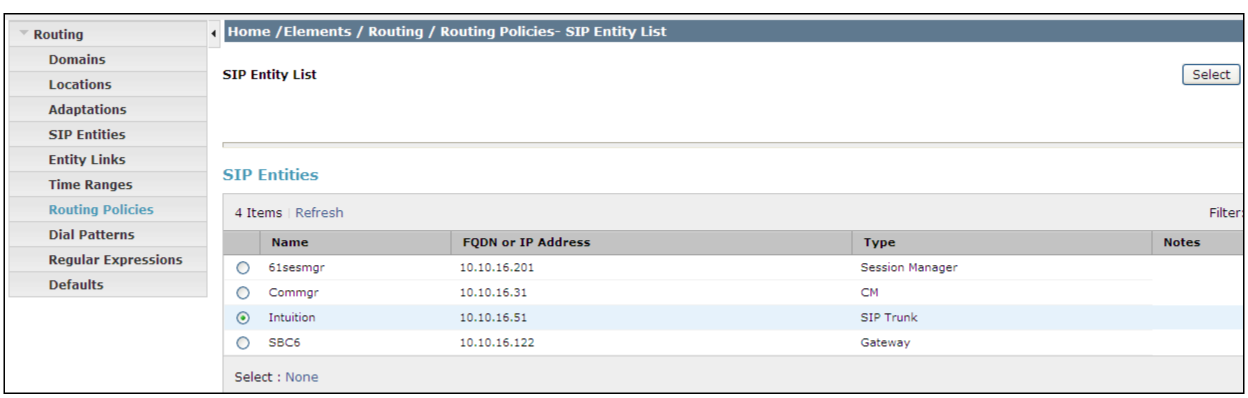


6.8. Add Routing Policy

From the **Elements** section of the dashboard choose **Routing** and then **Routing Polices**. Click on the **New** button (not shown) and add a **Name** for the policy. Click on the **Select** button under **SIP Entity as Destination**.

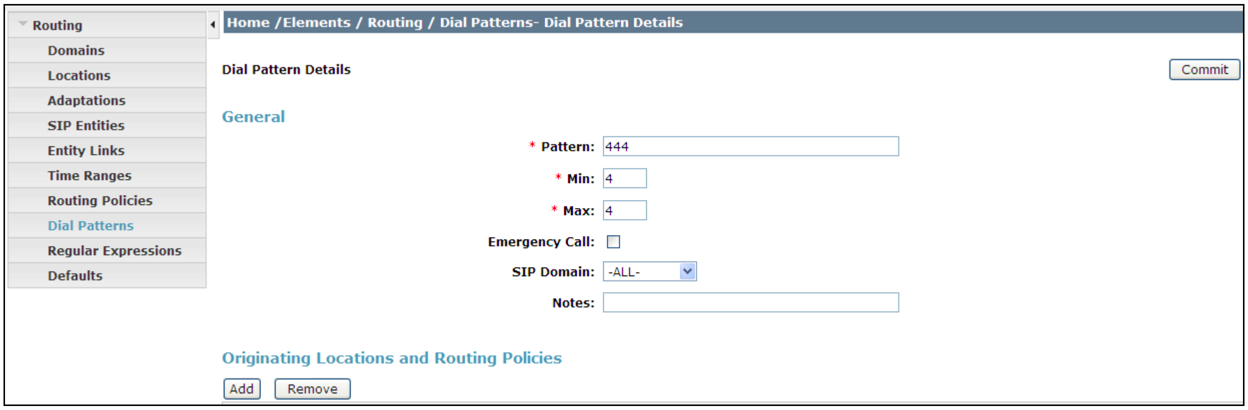


Select the Intuition Enterprise Server entity as a Destination and click on the **Select** button. When routed back to the previous screen click on the **Commit** button (not show) to save.

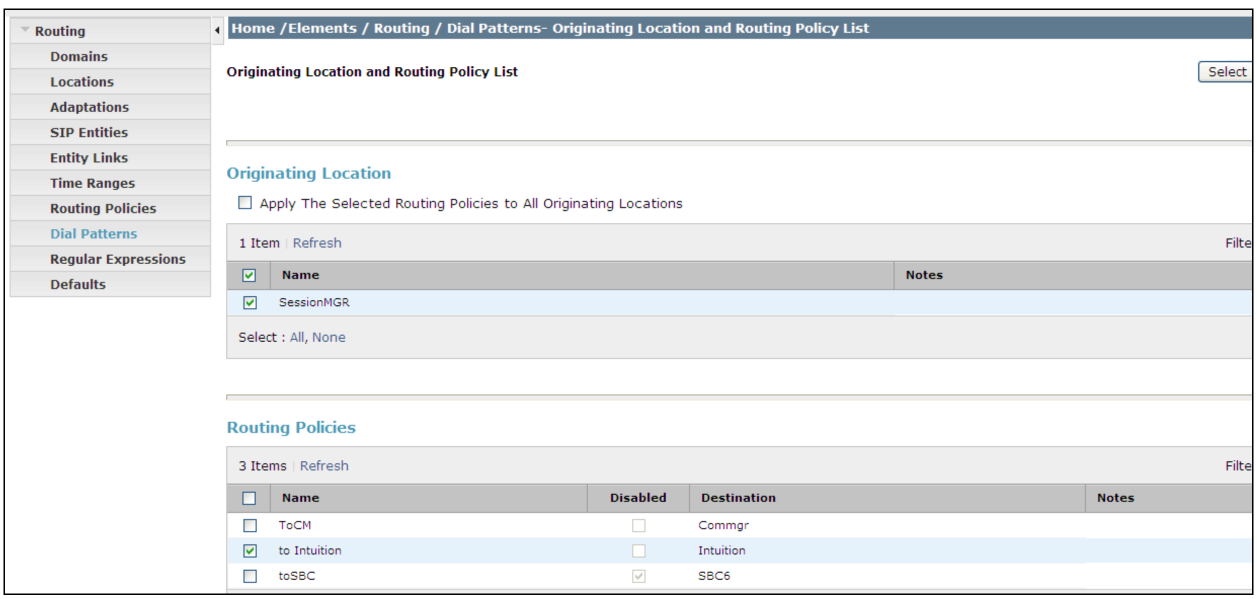


6.9. Add Dial Pattern

Select **Dial Patterns** from the **Routing** menu and click on the **New** button (not shown). Enter the **Pattern** used for dialing the Intuition Enterprise Server and the **Min** and **Max** digits. Click on the **Add** button under **Originating Locations and Routing Policies**.



Check the **Originating Location** and **Routing policies** required and click on the **Select** button. Click on the **Commit** button when routed back to the previous page to save.



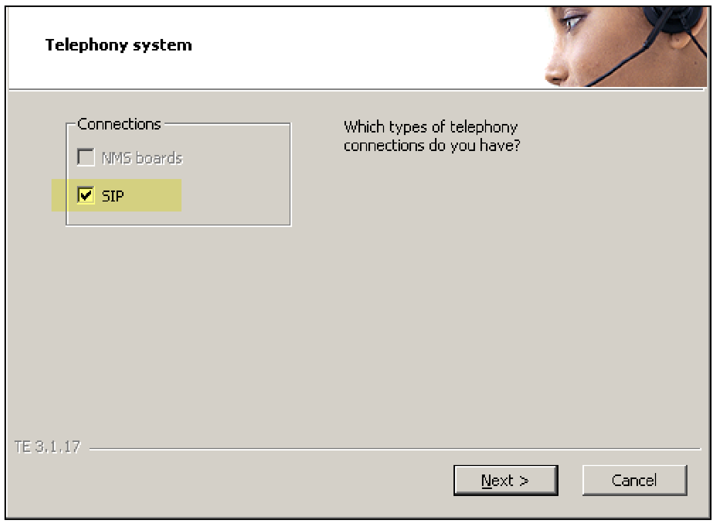
Note: Dial Pattern will automatically be added to the routing policies selected.

7. Configure Datapulse Intuition Enterprise Server

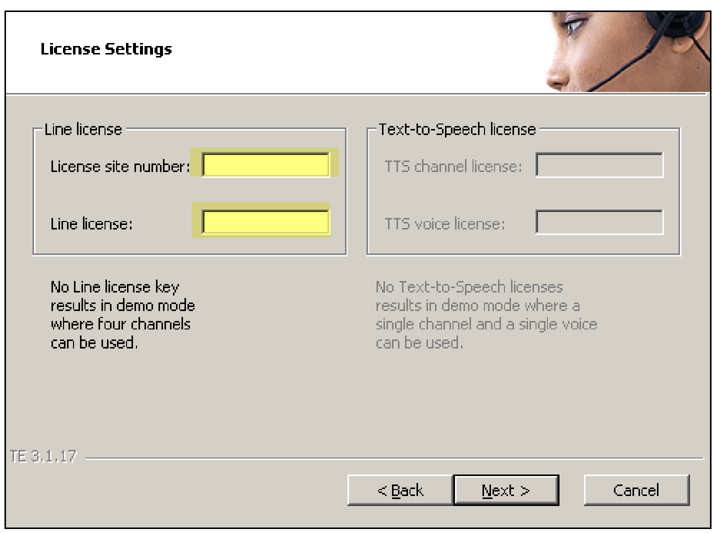
These steps are designed to set up Intuition Enterprise server to interoperate with CM and Session Manager using SIP.

7.1. General setup

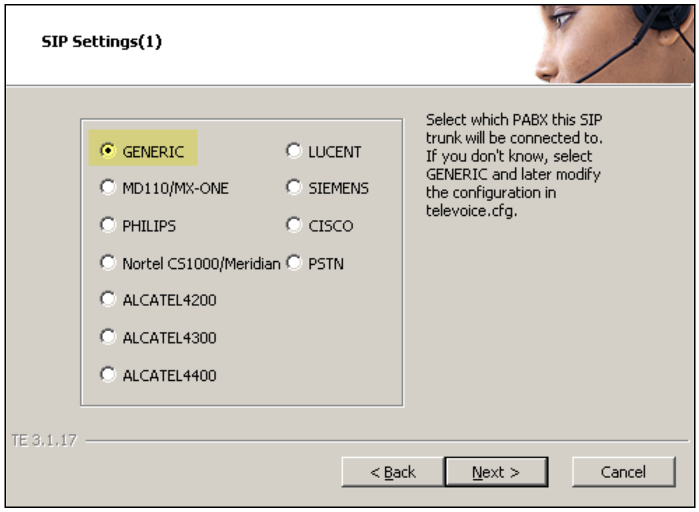
Start the Line interface configuration program from **Programs → Intuition Enterprise** on the start menu. Make sure **SIP** is selected and click **Next**.



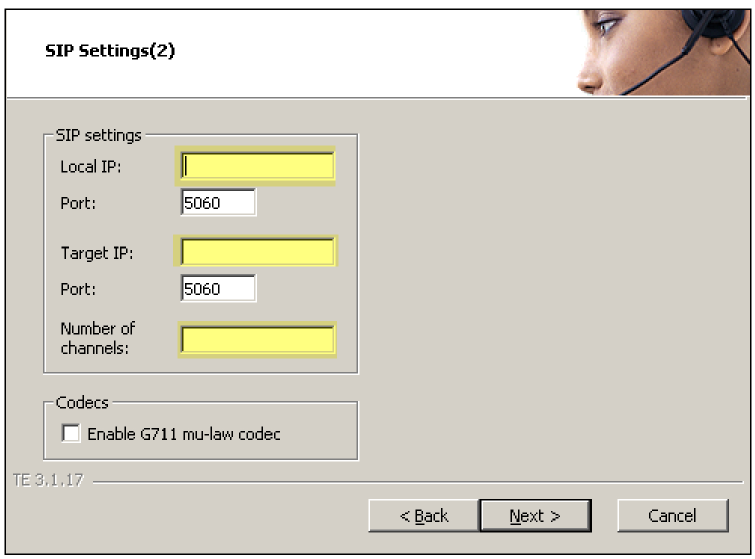
Enter your **License site number** and **Line License** codes and click **Next**.



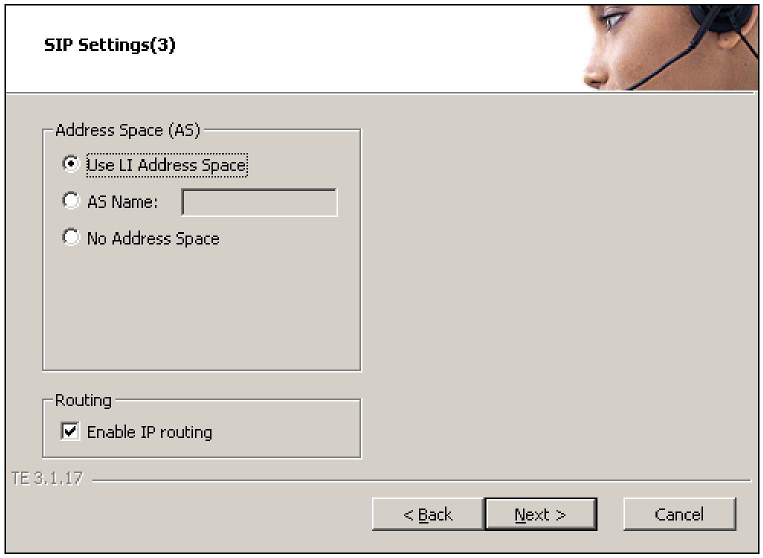
Make sure **GENERIC** is selected and click **Next**.



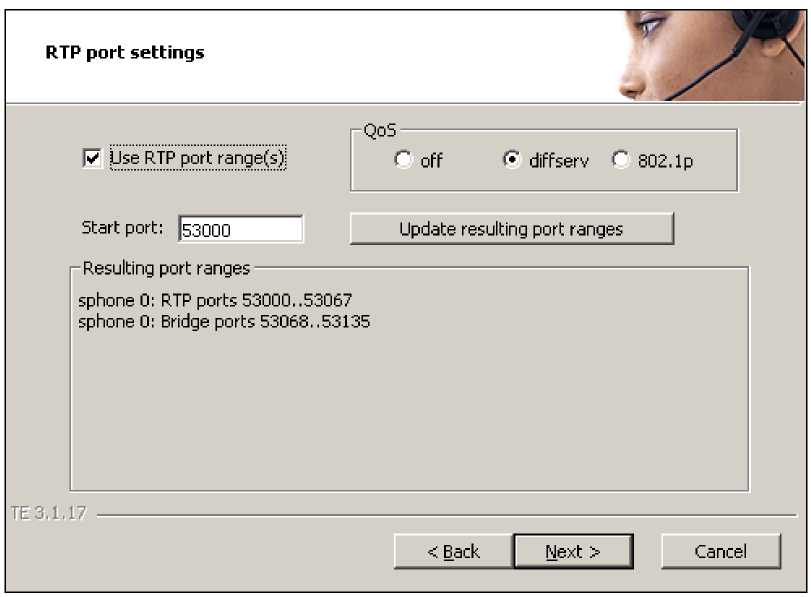
Enter your **Local IP**, the **Target IP** that relates to the Communication Manager interface from **Section 5.3** and **Number of channels** that should be activated. When done, click **Next**.



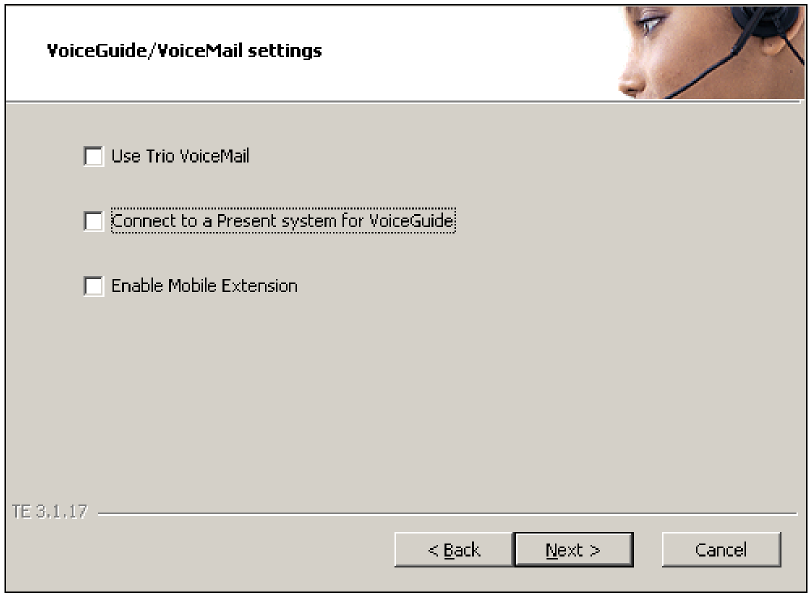
Click **Next** to use the default settings.



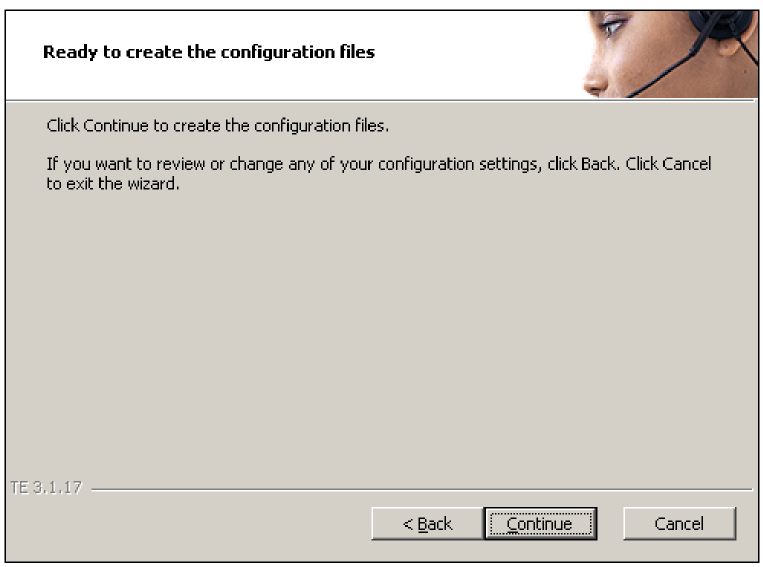
Click **Next** to use the default setting.



Click **Next** to use the default setting.



Click **Continue.**



7.2. Special configuration for Avaya Aura® Session Manager

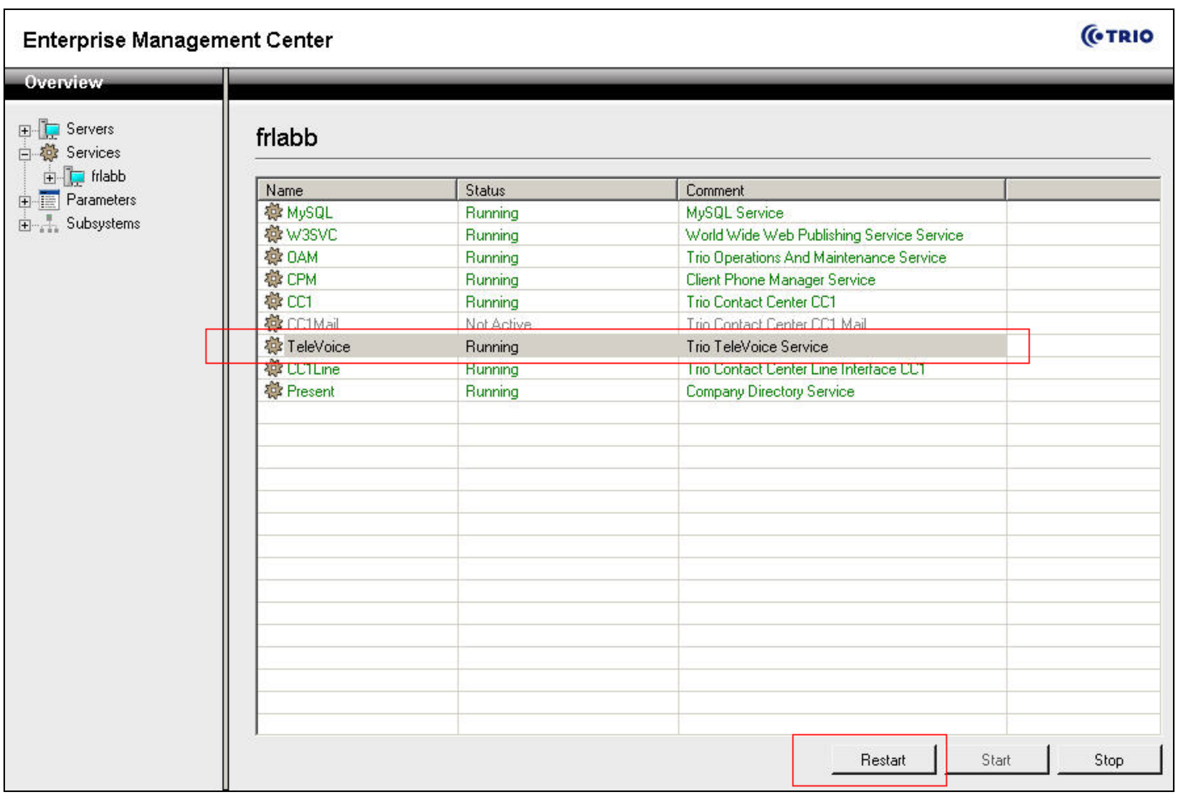
Open C:\TE\ProgramData\LI\cfg\televoice.cfg. Find the [SIP_1] Section.

```
[sip_1]
        signallingprotocol=sip
        localHost=10.10.16.51
        targetHost=10.10.16.201
        uriScheme=1
        transferPoint=afterAnswer
        usetcp = 1
```

Make sure 'uriScheme' is set to '1'
Add the row 'ucetecp =1'

7.3. Restart the TeleVoice service

From the start menu on the Intuition Enterprise Server open **Programs → Intuition Enterprise → Enterprise Management Center** select **TeleVoice** service and click on the **Restart** button to activate changes.



8. Verification Steps

The following steps can be used to verify and troubleshoot the installation of the configuration tested in this document.

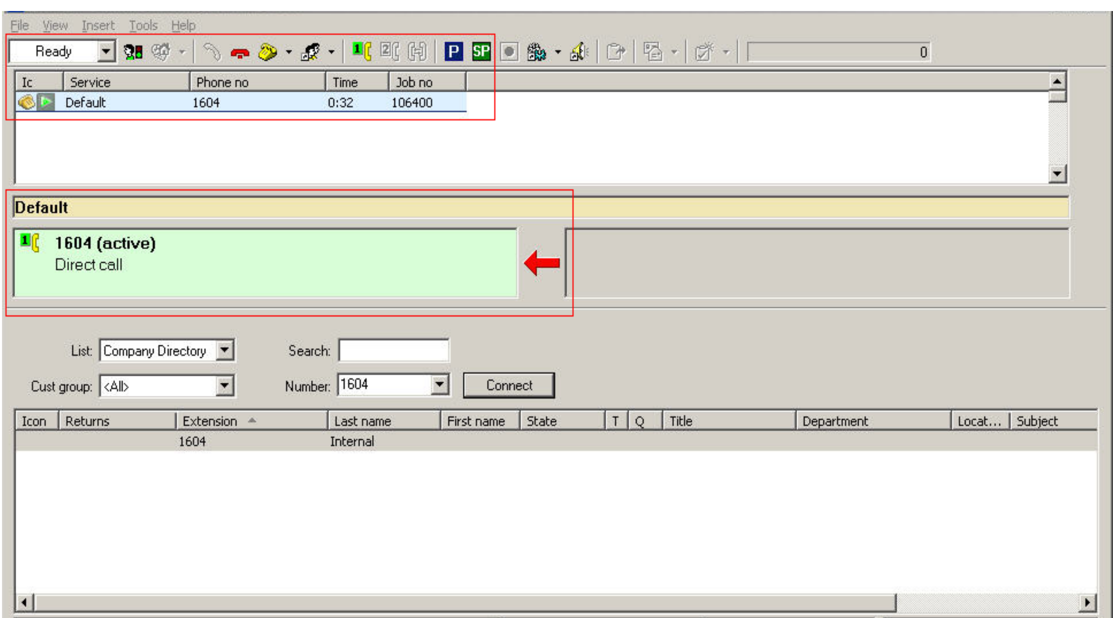
8.1. SIP Trunk Status

Make a call to the Intuition Enterprise server and answer the call using Intuition Enterprise Agent. Use the **status trunk** command to view **in-service active** trunk carrying the call.

```
status trunk 7

                              TRUNK GROUP STATUS

Member   Port     Service State       Mtce Connected Ports
                                      Busy

0007/001 T00629   in-service/idle     no
0007/002 T00630   in-service/idle     no
0007/003 T00631   in-service/idle     no
0007/004 T00632   in-service/active   no    S00019
0007/005 T00633   in-service/idle     no
0007/006 T00634   in-service/idle     no
0007/007 T00635   in-service/idle     no
0007/008 T00636   in-service/idle     no
0007/009 T00637   in-service/idle     no
0007/010 T00638   in-service/idle     no
```

8.2. Active Call Information

Make a call the Intuition Enterprise Server and answer the call using Intuition Enterprise Attendant.



9. Conclusion

These Application Notes have described the administration steps required to use Datapulse Intuition Enterprise Server with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All features were tested successfully with the configuration detailed above.

10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, 9th August 2010, Document Number 03-300509.

[2] *SIP Support in Avaya Aura® Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.

[3] *Installing and configuring Avaya Aura® Session Manager*, 5th January 2011, Document Number 03-603473.

[4] *Session Initiation Protocol Service Examples draft-ietf-sipping-service-examples-15*, Internet-Draft, 11th July 2008, available at http://tools.ietf.org/html/draft-ietf-sipping-service-examples-15

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.