Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager 6.3 and Avaya Aura® Session Manager 6.3 to Interoperate with Netcall Liberty 3.0 – Issue 1.0

Abstract

The Application Notes describe the configuration steps for the Netcall Liberty solution to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Netcall Liberty system can host multiple applications such as IVR and Voicemail via connection to Avaya Aura® Session Manager over a SIP Trunk.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any 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

The purpose of this document is to describe the compliance tested configuration between Avaya Aura® Session Manager, Avaya Aura® Communication Manager and the Netcall Liberty platform. This document includes a description of the configuration of both the Avaya and Netcall solutions, a description of the tests that were performed and a summary of the results.

Netcall Liberty is a Speech Enabled Multi Application Platform which integrates with the Avaya solution using a SIP trunk. Netcall Liberty enables the hosting of multiple solutions concurrently on the same platform.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of Netcall Liberty to carry out call handling and routing in a variety of scenarios using the ContactPortal (ASR) and Messaging+ applications. For the purposes of the compliance test the speech recognition and messaging applications were used to test the call handling functions of the Netcall Liberty platform. These applications were used to give a broad overview of the call handing abilities of the Netcall Liberty platform and are indicative of the way calls to other Netcall Liberty applications are handled.

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 placing calls in different scenarios, to ensure successful call routing based on the application used. Call handling and routing functions are common across Netcall Liberty, and these tests are indicative of the way Netcall Liberty processes calls. However for the purpose of the compliance testing for Netcall Liberty, the ContactPortal® ASR and Messenger+ voicemail modular applications were used, and the tests carried out included:

- Successful Speech recognition
- Barge-in
- Correct call routing
- Transfer to busy/engaged/dnd/restricted user
- SIP REFER usage
- Hold/Retrieve
- Call Transfer
- Intra switch call
- Inbound trunk call

- Outbound trunk call
- Out of band Signaling (DTMF)
- Logging

In the case of the Messaging+ application:

- Coverage to correct mailbox
- Message retrieval
- User recognition from CPN
- MWI
- Greeting recording
- Intra switch call
- Inbound trunk call

The serviceability testing focused on verifying the ability of Netcall Liberty to recover from disconnection and reconnection to the Avaya solution and power failure.

2.2. Test Results

All functionality and serviceability test cases were completed successfully with the following observation:

- In the case where an extension is called which has an unconditional divert set to another extension, and the extension being forwarded to has a divert set to the Netcall Liberty Messaging+ application, the voicemail will answer on the mailbox of the person originally called e.g. 4001 calls 4002, 4002 is on unconditional divert to 4003, 4003 has divert set to the Netcall Liberty Messaging+ application, call from 4001 covers to the voicemail box of 4002. This is by design.

2.3. Support

Technical support can be obtained for the Netcall Liberty solution as follows:

- Email: technical.support@netcall.com
- Website: http://www.netcall.com
- Phone: + 44 330 333 6100

3. **Reference Configuration**

Figure 1 illustrates the network topology used during compliance testing. System Manager and Session Manager run on individual HP Proliant GL360 Servers and Communication Manager runs on an Avaya S8300D Server circuit board installed in an Avaya G450 Media Gateway. These Application Notes focus on the configuration of Communication Manager and Session Manager for integration with Netcall Liberty. A variety of Avaya H.323, SIP, Digital and Analog Deskphones were used in the testing. Netcall Liberty was hosted on a Microsoft Windows 2008 R2 Standard server provided by Netcall.

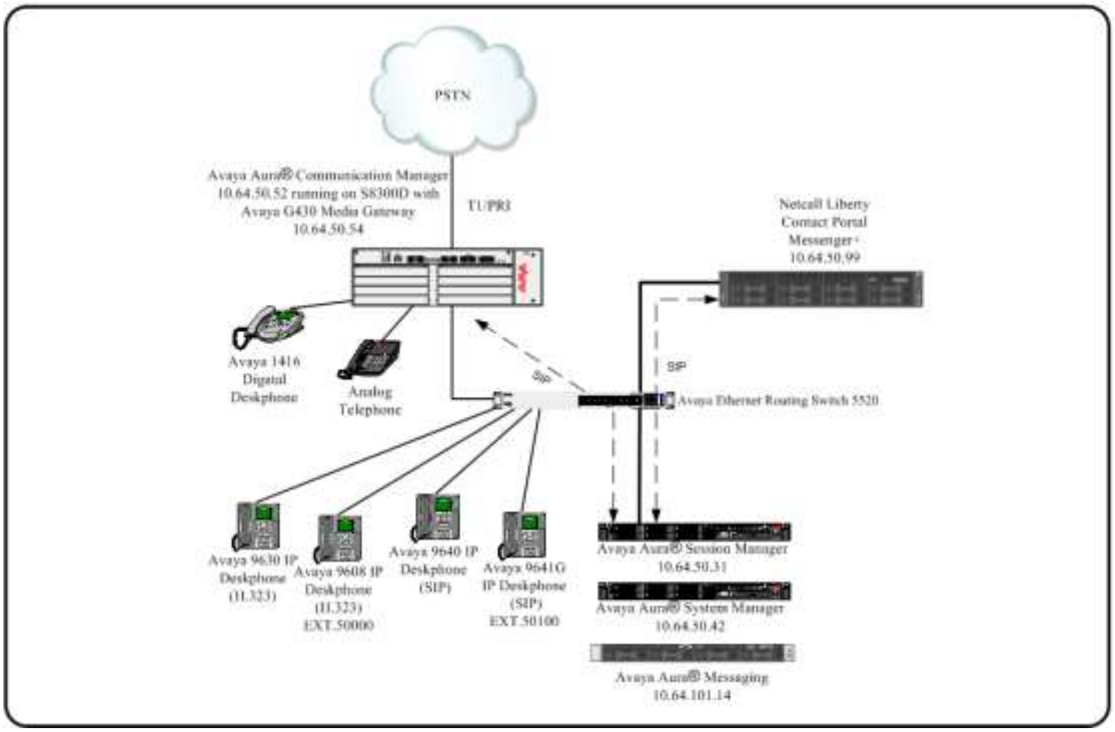


Figure 1: Avaya Aura® Communication Manager with Avaya Aura® Session Manager and Avaya Aura® System Manager and Netcall Liberty Solution.

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 running on Avaya S8300D Server	6.3.10.0-SP10
Avaya Aura® Session Manager running on HP Proliant GL360 Server	6.3.12.0.631208
Avaya Aura® System Manager	6.3.12.9.3022
Avaya Aura® Messaging	6.3.2 SP 2
Avaya 96x0 Deskphone	SIP R2_6_13-141010, H.323 R3_2_4-121214
Avaya 96x1 Deskphone	SIP R6_5_0-121114, H.323 R6_4_0_14-040314
Avaya 6211 and 6221 analog telephone	-
Avaya 1416 Digital Deskphone	Rel. 39.0
Netcall Liberty (including Contact Portal and Messenger+)	3.0

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 as referenced in **Section 10**. The configuration operations described in this section can be summarized as follows:

- Configure node-name
- Configure SIP Signaling Group
- Configure SIP Trunk
- Configure Route Pattern
- Configure AAR
- Configure Coverage Paths
- Configure Station

5.1. Configure Node Names

These application notes assume that the Session Manager SIP Signaling Interface has been configured on Session Manager. In order to create the SIP Trunk between Communication Manager and Session Manager a node-name needs to be specified for the Session Manager SIP Signaling Interface, this will be used in **Section 5.2**. Enter the command **change node-names ip** and enter an identifying **Name** for the Session Manager SIP Signaling Interface and its **IP address**, in this case **sm5031** and **10.64.50.31** respectively.

```
change node-names ip                                          Page 1 of   2
                                   IP NODE NAMES
    Name              IP Address
default             0.0.0.0
procr               10.64.50.52
procr6              ::
sm5031              10.64.50.31
```

5.2. Configure SIP Signaling Group

A signaling group must be used to specify the signaling type and node-names to be used for the SIP Trunk configured in **Section 5.3**. Enter the command **add signaling-group next**, take a note of the **Group Number**, set the **Group Type** to **sip**, **Transport Method** to **tls**, set the **Near-end Node Name** to the Communication Manager processor interface name, in this case **procr**, and the **Far-end Node Name** to that configured in **Section 5.1**, in this case **sm5031**. Leave all other settings as default.

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

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: sm5031
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region:

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

5.3. Configure SIP Trunk

To route calls between Communication Manager and Session Manager the SIP trunk must use the signaling group setup in **Section 5.2**. Enter the command **add trunk-group next**, on **Page 1** take a note of the **Group Number**, this will be used when configuring route patterns. Set the **Group Type** to **sip** and assign the trunk an identifying **Group Name.** Set the **TAC** according to the dialplan. Set the **Service Type** as **public-ntwrk** and assign the **Signaling Group** as added in **Section 5.2**, set **Number of Members** according to requirements.

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

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk TO SM5031         COR: 1        TN: 1        TAC: *002
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 30
```

On **Page 3** set the **Numbering Format** as **Private**.

```
add trunk-group next                                              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
```

On **Page 4** set the **Send Diversion Header** to **y**, **Support Request History** to **n**, and the **Identity for Calling Party Display** to **From**.

```
add trunk-group next                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? y
                                   Support Request History? n
                              Telephone Event Payload Type:


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: From
            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
```

5.4. Configure Route Pattern

In order to place calls from via Communication Manager to Session Manager and on to Netcall Liberty a route pattern must be created from Communication Manager to Session Manager. The route pattern will use the trunk created in **Section 5.3**. Enter the command **change route-pattern x** where **x** is an unused route pattern, configure an identifying **Pattern Name**, the **Grp No** setup in the previous Section, and an **FRL** accordingly.

```
change route-pattern 2                                        Page   1 of   3
                    Pattern Number: 3      Pattern Name: To sm5031
                             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: 2    0                    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                lev0-pvt  none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.5. Configure Dialplan

Enter the command **change dialplan analysis** to configure the digit string to be routed to Session Manager and onto Netcall Liberty. In the example below, **Dialed String** beginning with **64** and a **Total Length** of **5** is set to the **call type udp**, this will route the dialed string to the uniform dialplan for further classification.

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 3

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   1           11   ext
   2           5    ars
   3           5    ext
   4           5    ext
   5           5    aar
   6           5    ext
   64          5    udp
   67          5    aar
   7           5    ext
   8           1    fac
   9           1    fac
   *           3    fac
   *           4    dac
   #           4    fac
```

5.6. Configure Uniform Dialplan

Enter the command **change uniform-dialplan 85**, set the **Matching Pattern** to **64**, the **Len** to **5** and the **Net** to **aar**, this will route the 5 digit number beginning with **64** to the **aar** table where a route pattern is defined.

```
change uniform-dialplan 64                                     Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                           Percent Full: 0

  Matching                    Insert              Node
  Pattern        Len Del      Digits      Net Conv Num
 64               5   0                   aar  n
```

5.7. Configure AAR

The AAR table must be configured in order that calls to Netcall Liberty are routed using the pattern configured in **Section 5.4**. Enter the command **change aar analysis 0**, in this example when a **5** digit number beginning with digits **64** is dialed, it will be placed using route pattern **2**. Configure the **Dialed String 64**, **Min 5** and **Max 5**, **Route Pattern 2** and **Call Type aar** as shown below.

```
change aar analysis 64                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 3

          Dialed            Total      Route     Call    Node   ANI
          String           Min  Max   Pattern    Type    Num    Reqd
    64                     5    5      2         aar            n
```

5.8. Configure Coverage Paths

Cover paths must be administered in order to route calls to the Netcall Liberty Messaging+ application. For the purposes of the compliance test, both a cover path and a remote cover path were added. Enter the command **change coverage remote 1**, enter the extension assigned in **Section 7.3** for voicemail deposit, in this case **64999** in an available field, in this example **01**. This is referred to as r1.

```
change coverage remote 1                                        Page   1 of  23

                       REMOTE CALL COVERAGE TABLE
                       ENTRIES FROM 1    TO 1000

01: 64999                 16:                        31:
02:                       17:                        32:
03:                       18:                        33:
04:                       19:                        34:
05:                       20:                        35:
06:                       21:                        36:
07:                       22:                        37:
08:                       23:                        38:
09:                       24:                        39:
10:                       25:                        40:
11:                       26:                        41:
12:                       27:                        42:
13:                       28:                        43:
14:                       29:                        44:
15:                       30:                        45:
```

Enter the command **add coverage path next**, take a note of the **Coverage Path Number**, in this case **1**, and enter **r1** in **Point 1**.

```
add coverage path next                                        Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?            n               n
               Busy?            y               y
       Don't Answer?            y               y        Number of Rings: 2
                All?            n               n
 DND/SAC/Goto Cover?            y               y
   Holiday Coverage?            n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: r1             Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

5.9. Configure Station

A station must be administered with the coverage path configured in the previous section for use with the Netcall Liberty Messaging+ application. Enter the command **add station x** in this case station **60004** is used. Configure according to requirements and set the **Coverage Path** to **1**.

```
add station next                                              Page   1 of   5
                                     STATION

Extension: 60004                         Lock Messages? n             BCC: 0
     Type: 9630                          Security Code:                TN: 1
     Port: IP                          Coverage Path 1: 1             COR: 1
     Name: Tom Watson                  Coverage Path 2:               COS: 1
                                       Hunt-to Station:             Tests? y
STATION OPTIONS
                Location:                    Time of Day Lock Table:
              Loss Group: 19           Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 60004
            Speakerphone: 2-way                 Mute Button Enabled? y
        Display Language: english                    Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                            IP SoftPhone? n

                                                           IP Video? n
                              Short/Prefixed Registration Allowed: default

                                                 Customizable Labels? y
```

Additionally the voice-mail number must be configured on Page 4 for proper use of the Message Button on the H.323 Deskphone.

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

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:

    voice-mail 64999
```

5.10. Save Translation

Enter the command **Save Translation** to save the Communication Manager changes made in the previous sections.

6. Configure Avaya Aura® Session Manager

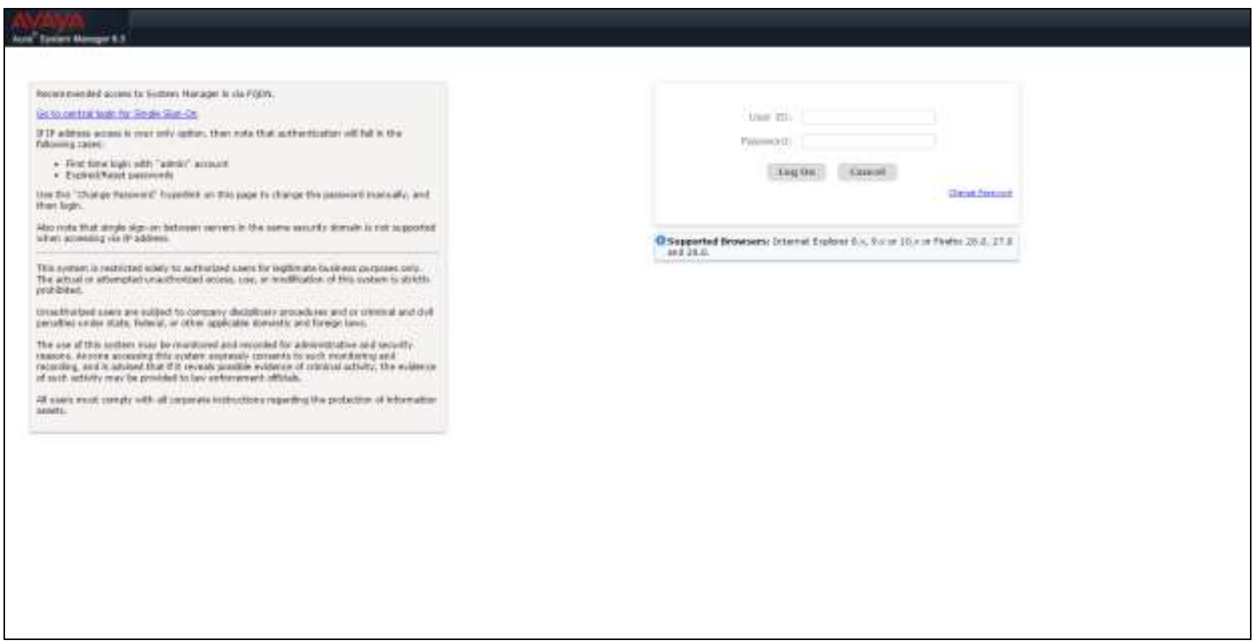
System Manager is used to configure Session Manager SIP Entities and manages the connections between related SIP trunks and endpoints. It is also used to configure dial patterns and route calls according to defined rules. The configuration and verification operations illustrated in this section were all performed using the System Manager Web Interface.

This section provides the procedure for configuring Session Manager. For further reference documents, refer to **Section 10** of this document. The procedures include the following areas:

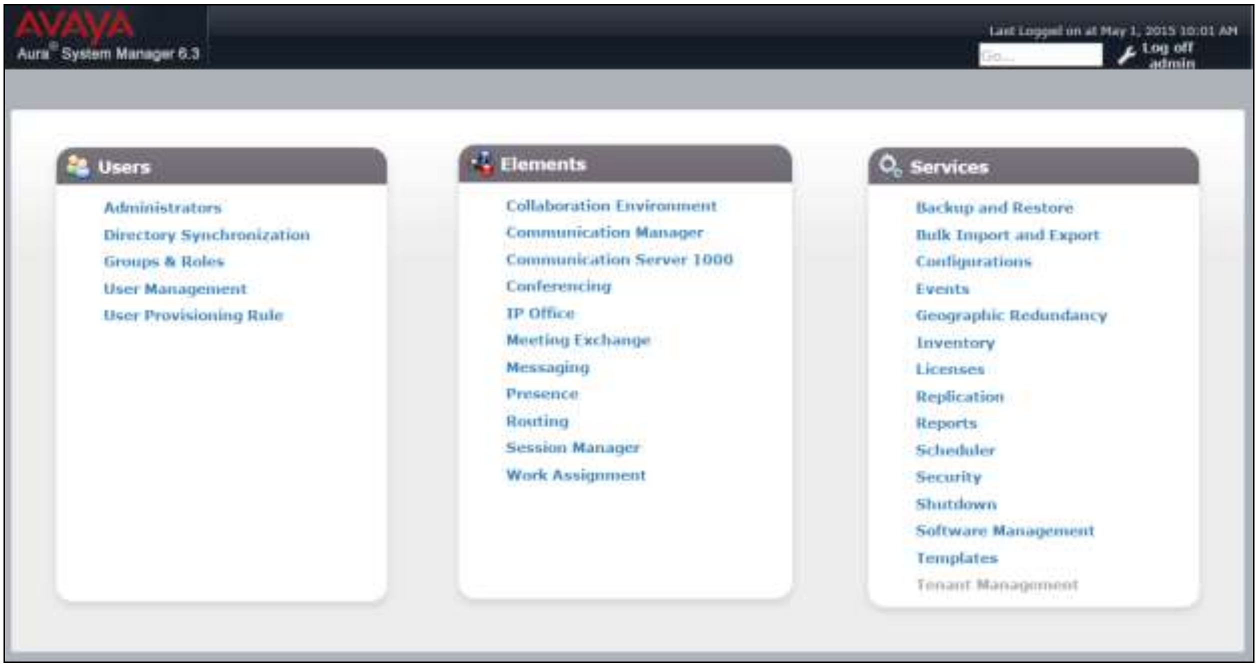
- Log in to System Manager
- Administer Adaptation
- Administer SIP Entity
- Administer Entity Link
- Administer Routing Policies
- Administer Dial Patterns

6.1. Log in to Avaya Aura® System Manager

Log into the System Manager web interface using the System Manager IP address, in this case https://10.64.50.42/network-login/ enter the appropriate credentials and click **Log On**.

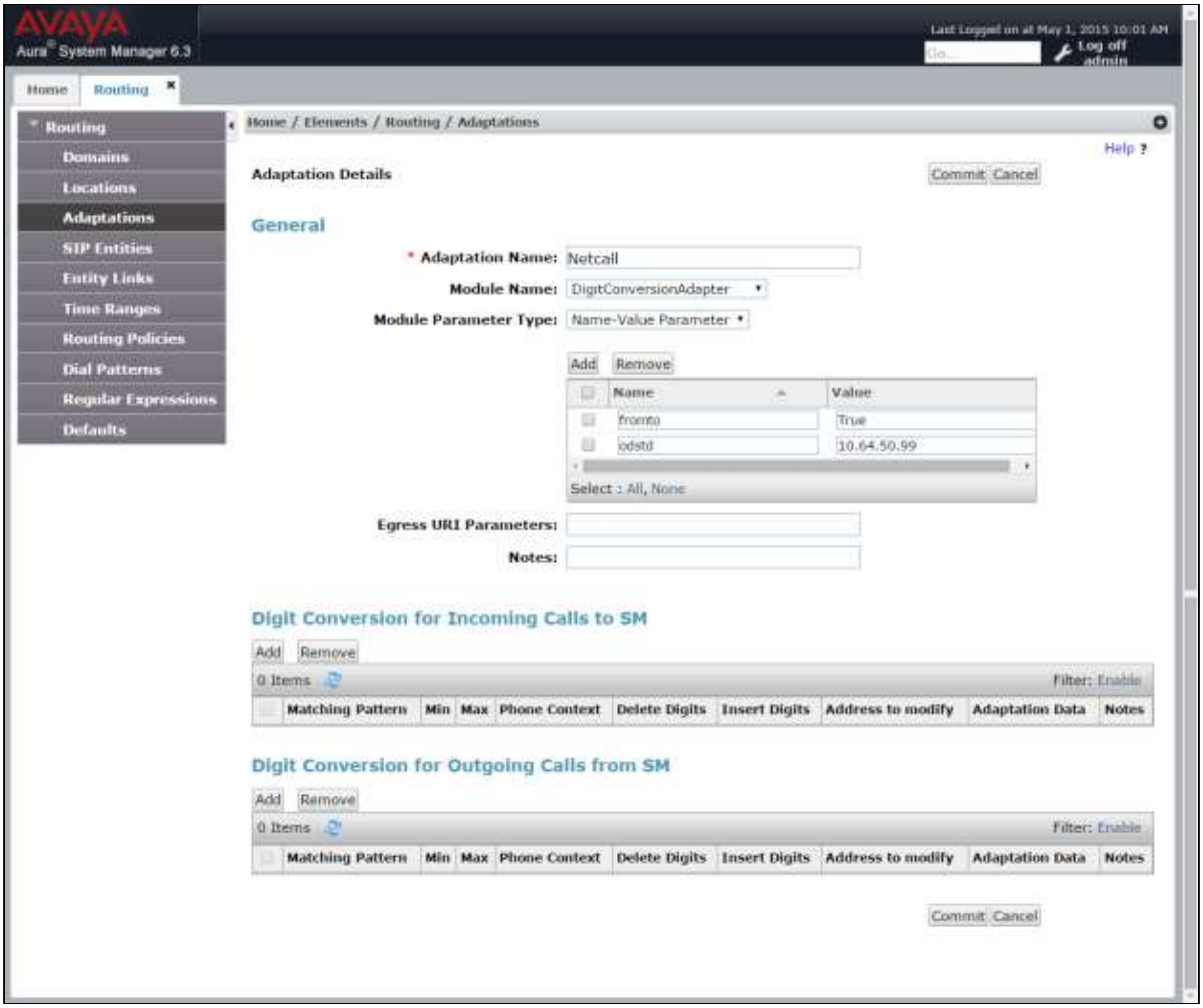


The home screen is divided into three sections with hyperlinked categories below.



6.2. **Administer Adaptation**

In order for successful interoperation of Netcall Liberty with Session Manager over a SIP trunk, an Adaptation must be configured to replace the outbound domain destination with the IP address of the Netcall Liberty server. Click **Home → Elements → Routing →Adaptations→New** assign and identifying **Adaptation name**, select **DigitConversionAdapter** from the drop down box next to **Module Name,** then select **Name-Value Parameter** from the drop down box next to **Module Parameter Type** this will cause the Add, Remove buttons, and Name, Value fields to appear. Click the **Add** button and enter **fromto** in the **Name** field and **True** in the **Value** field and click on **Commit**.. Repeat this step for adding overrideDestinationDomain (**odstd**) with the IP address of the Netcall Liberty Server (**10.64.50.99**).

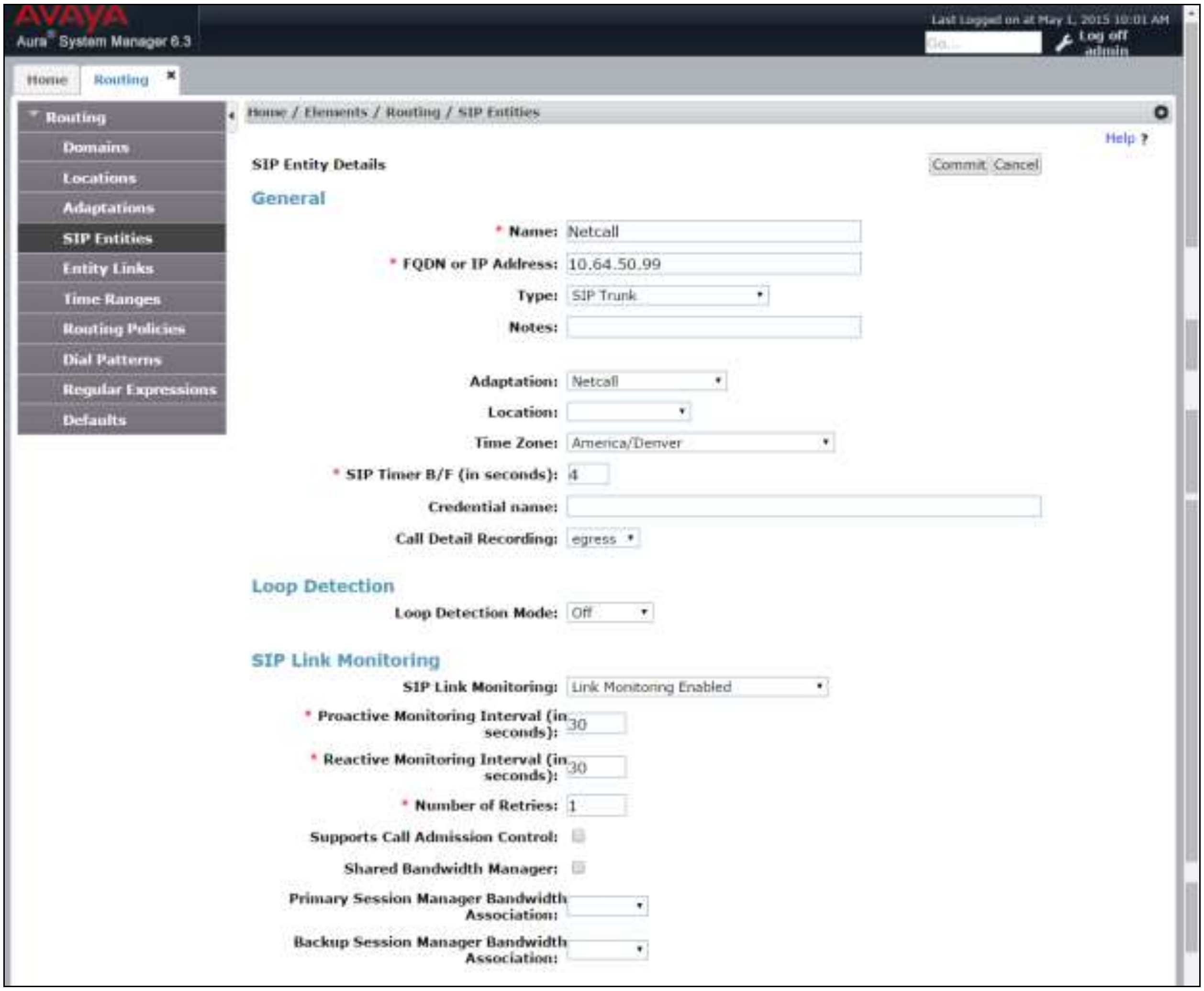

AVAYA
Aura® System Manager 6.3
Last Logged on at May 1, 2015 10:01 AM
Log off admin
Home
Routing
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Adaptations
Help ?
Adaptation Details
Commit Cancel
General
* Adaptation Name: Netcall
Module Name: DigitConversionAdapter
Module Parameter Type: Name-Value Parameter
Add Remove
Name
Value
fromto
True
odstd
10.64.50.99
Select : All, None
Egress URI Parameters:
Notes:
Digit Conversion for Incoming Calls to SM
Add Remove
0 Items
Filter: Enable
Matching Pattern
Min
Max
Phone Context
Delete Digits
Insert Digits
Address to modify
Adaptation Data
Notes
Digit Conversion for Outgoing Calls from SM
Add Remove
0 Items
Filter: Enable
Matching Pattern
Min
Max
Phone Context
Delete Digits
Insert Digits
Address to modify
Adaptation Data
Notes
Commit Cancel

6.3. Administer SIP Entity

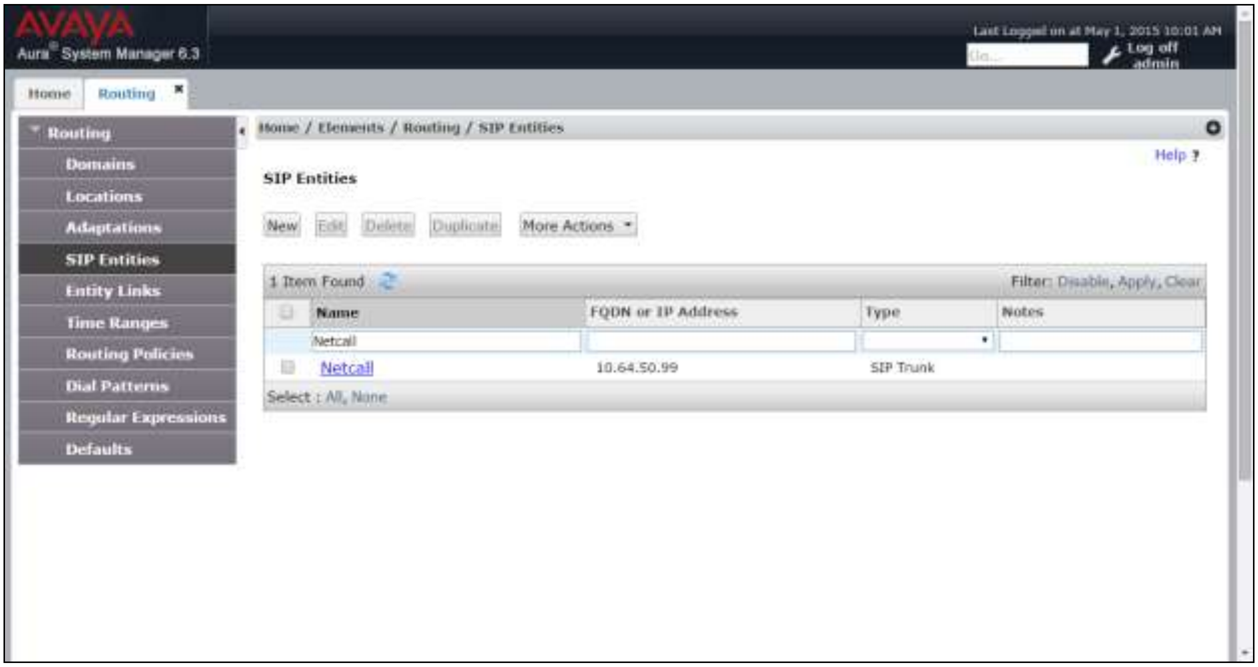
This section details the steps to create SIP Entity for the Netcall Liberty Solution.

6.3.1. Configure Netcall Liberty Entity

Click **Home → Elements → Routing → SIP Entities → New** assign an identifying **Name**, the **FQDN or IP Address** for the Netcall Liberty server, set the **Type** to **SIP Trunk**, choose the **Adaptation** configured in **Section 6.4** from the drop down box, leave all other settings default and click **Commit**.



The screen below will now be displayed confirming the entry.

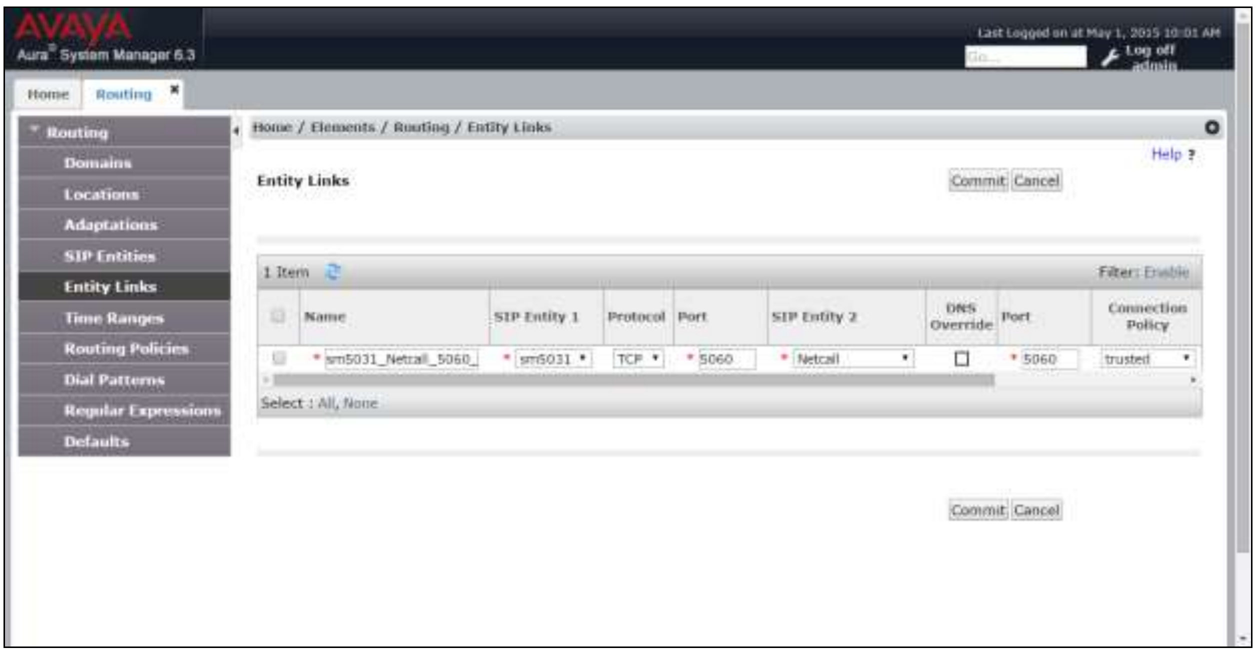


6.4. Administer SIP Entity Link

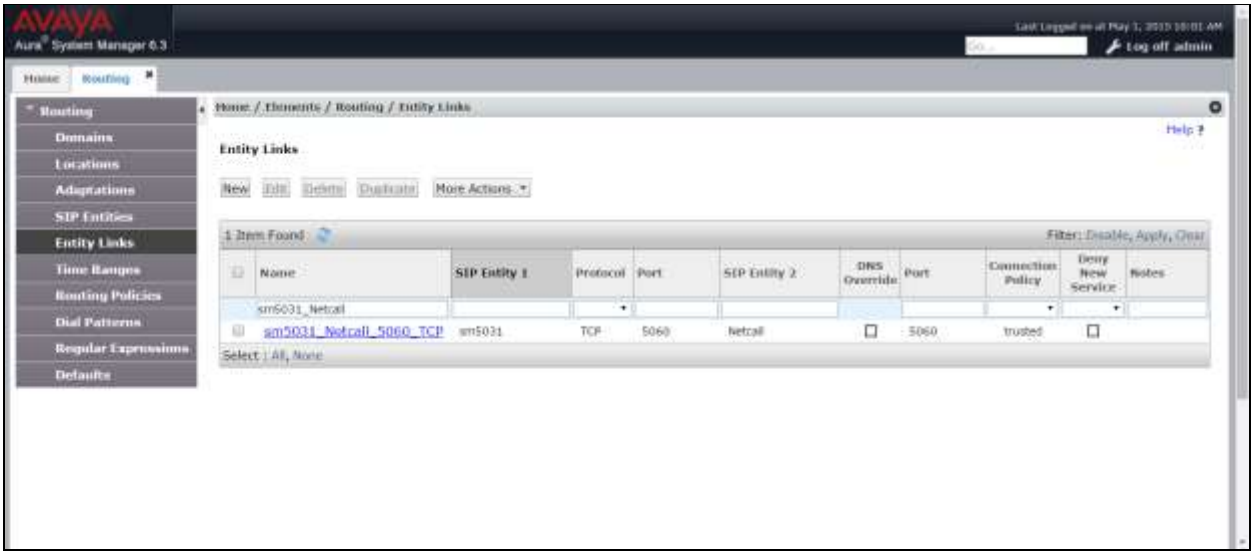
A SIP Trunk between a Session Manager and a telephony system is described by an Entity Link. An Entity Link needs to be created between Session Manager and Netcall Liberty.

6.4.1. Administer SIP Entity Link from Avaya Aura® Session Manager to Netcall Liberty

Click on **Home → Elements → Routing → Entity Links → New** assign an identifying **Name** choose the entity assigned to the Session Manager SIP Signaling Interface as **SIP Entity 1**, set the **Protocol** as **TCP**, enter **5060** for the **Port**, choose the Netcall Liberty entity as **SIP Entity 2** and set the **Port** to **5060**, select **Trusted** from the **Connection Policy** drop down box. Click **Commit** when done. This establishes the Session Manager end of the SIP Trunk to Netcall Liberty.



The screen below will be shown confirming the entry.



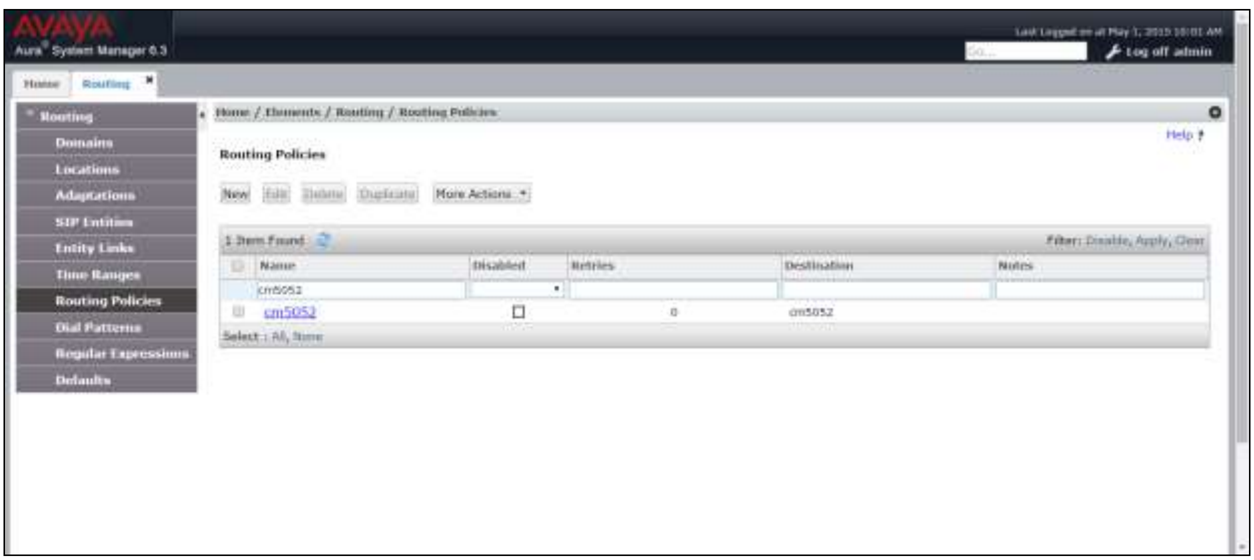
6.5. Administer Routing Policies

To complete the routing configuration, a Routing Policy is created. Routing policies direct how calls will be routed to an attached system. Two routing policies must be created, one for the Communications Manager and the second for Netcall Liberty. These will be associated with the Dial Patterns created in **Section 6.10**.

6.5.1. Create Routing Policy to Avaya Aura® Communication Manager

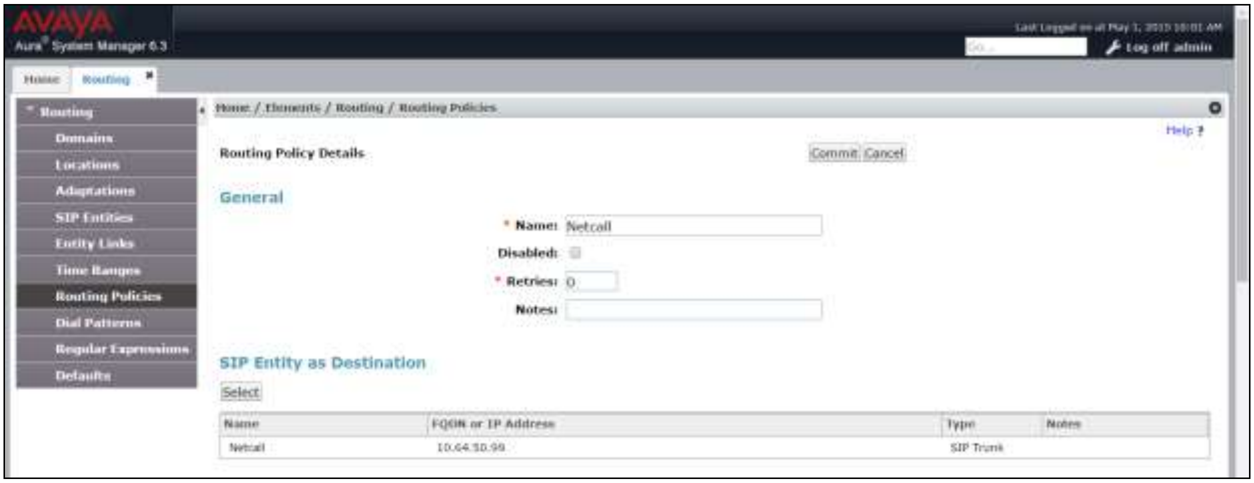
Click **Home → Elements → Routing → Routing Polices →New** assign an indentifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select** and choose the Communication Manager SIP Entity and click **Select**. The Communication Manager SIP Entity is assumed to have been previously created and the configuration of that entity is not shown in this document. Click **Commit** when done.

The screen below will be shown confirming the entry.

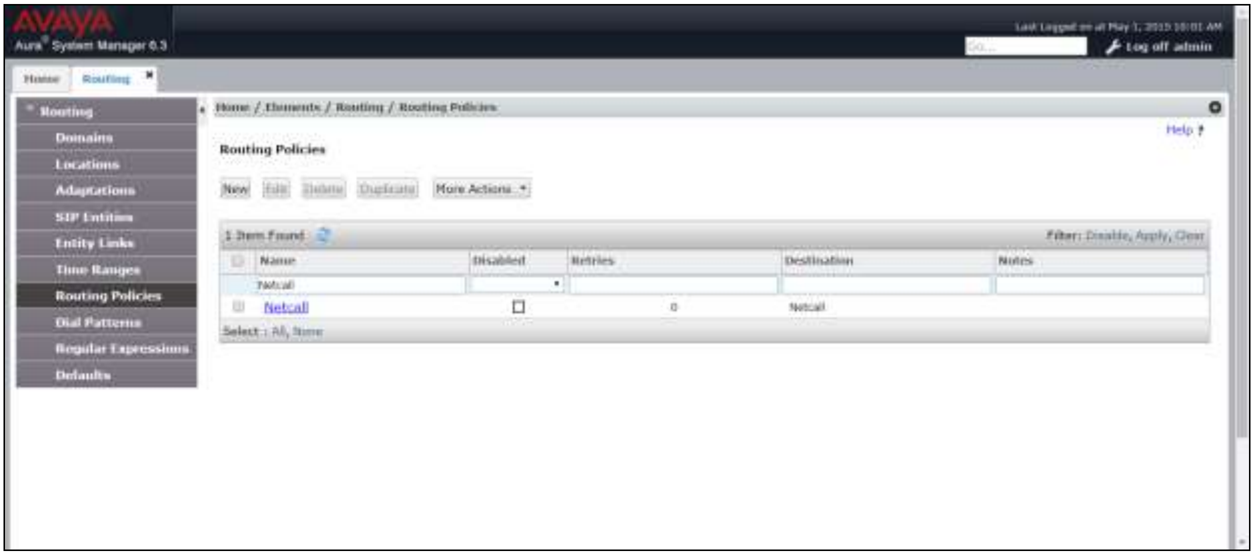


6.5.2. Create Routing Policy to Netcall Liberty

Click **Home → Elements → Routing → Routing Polices →New** assign an identifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select** and choose the **Netcall Liberty** SIP Entity and click **Select**. Click **Commit** when done.



The screen below will be shown confirming the entry.

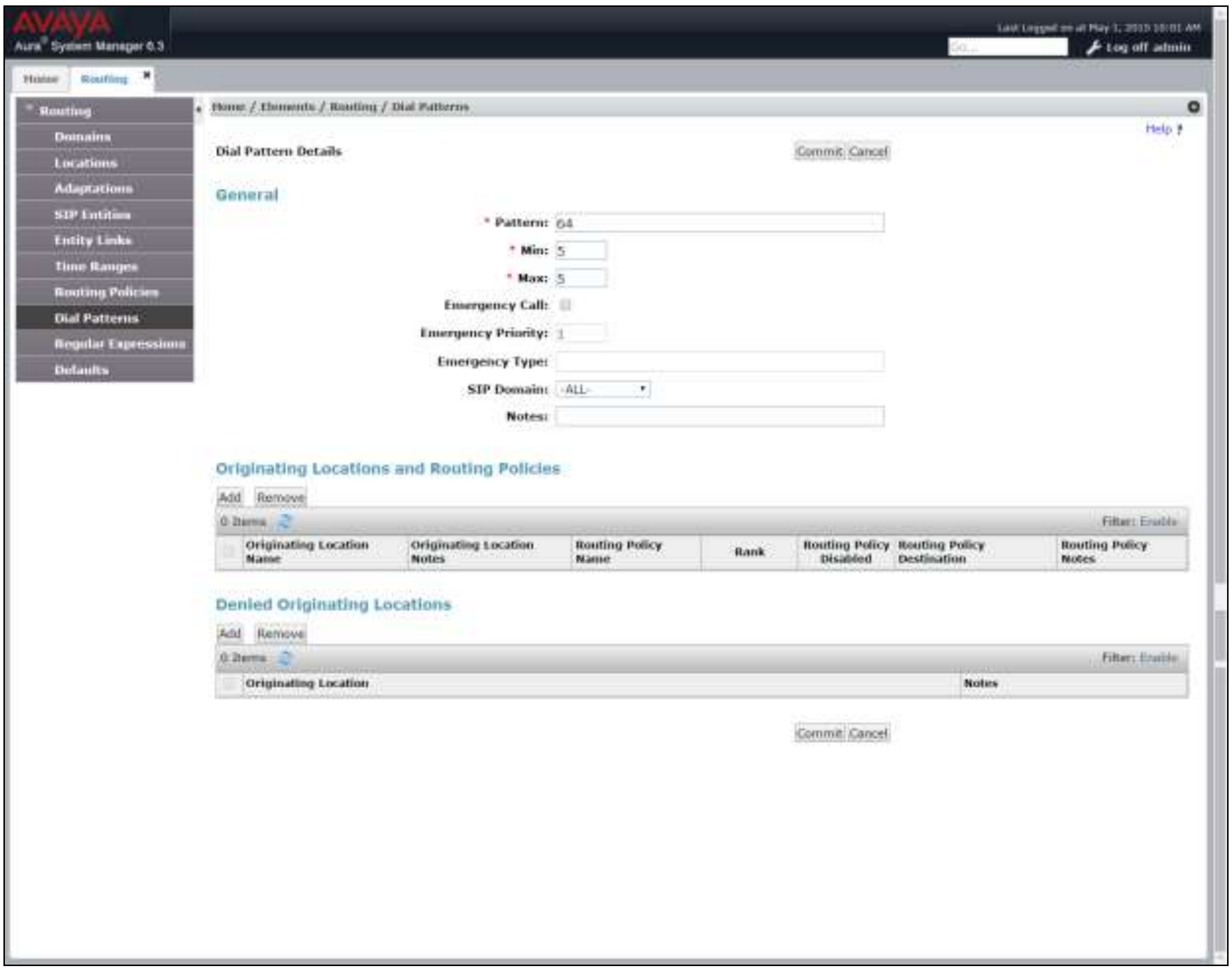


6.6. Administer Dial Patterns

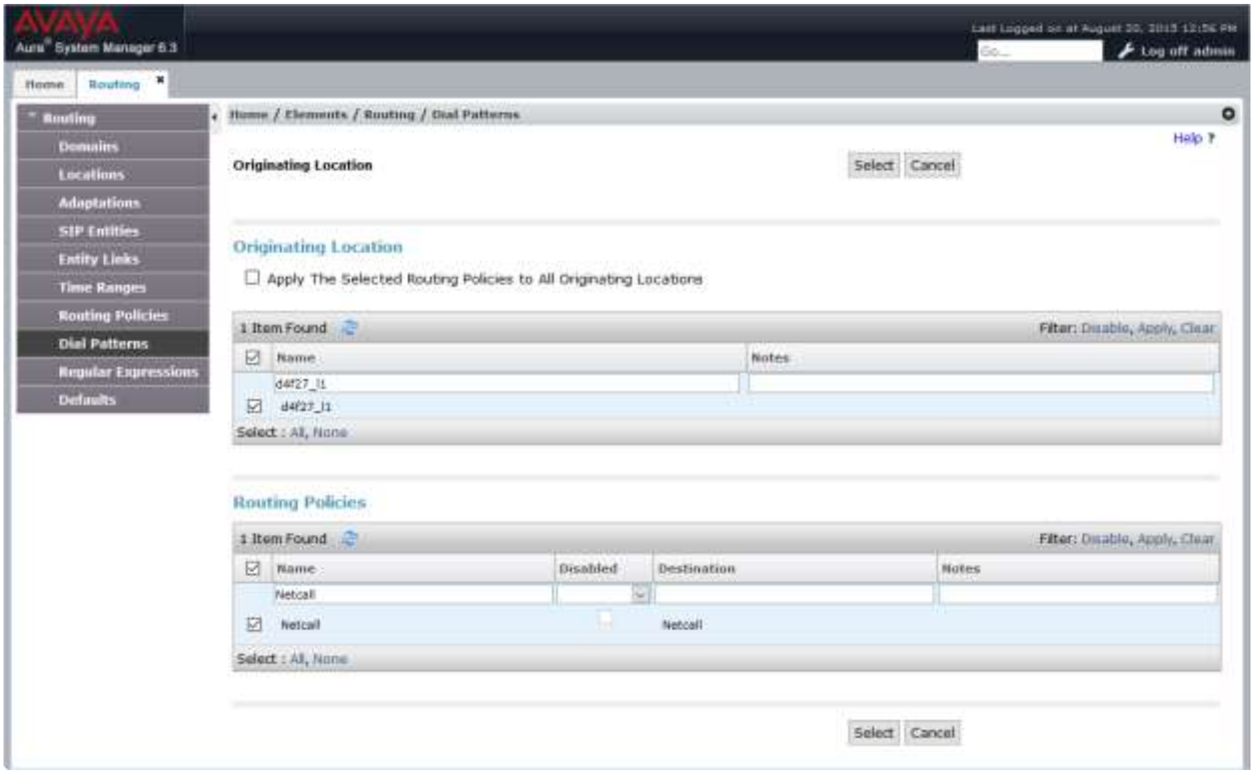
As one of its main functions, Session Manager routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing, which will direct calls based on the number dialed to the appropriate system.

6.6.1. Create Dial Pattern for calls to Netcall Liberty

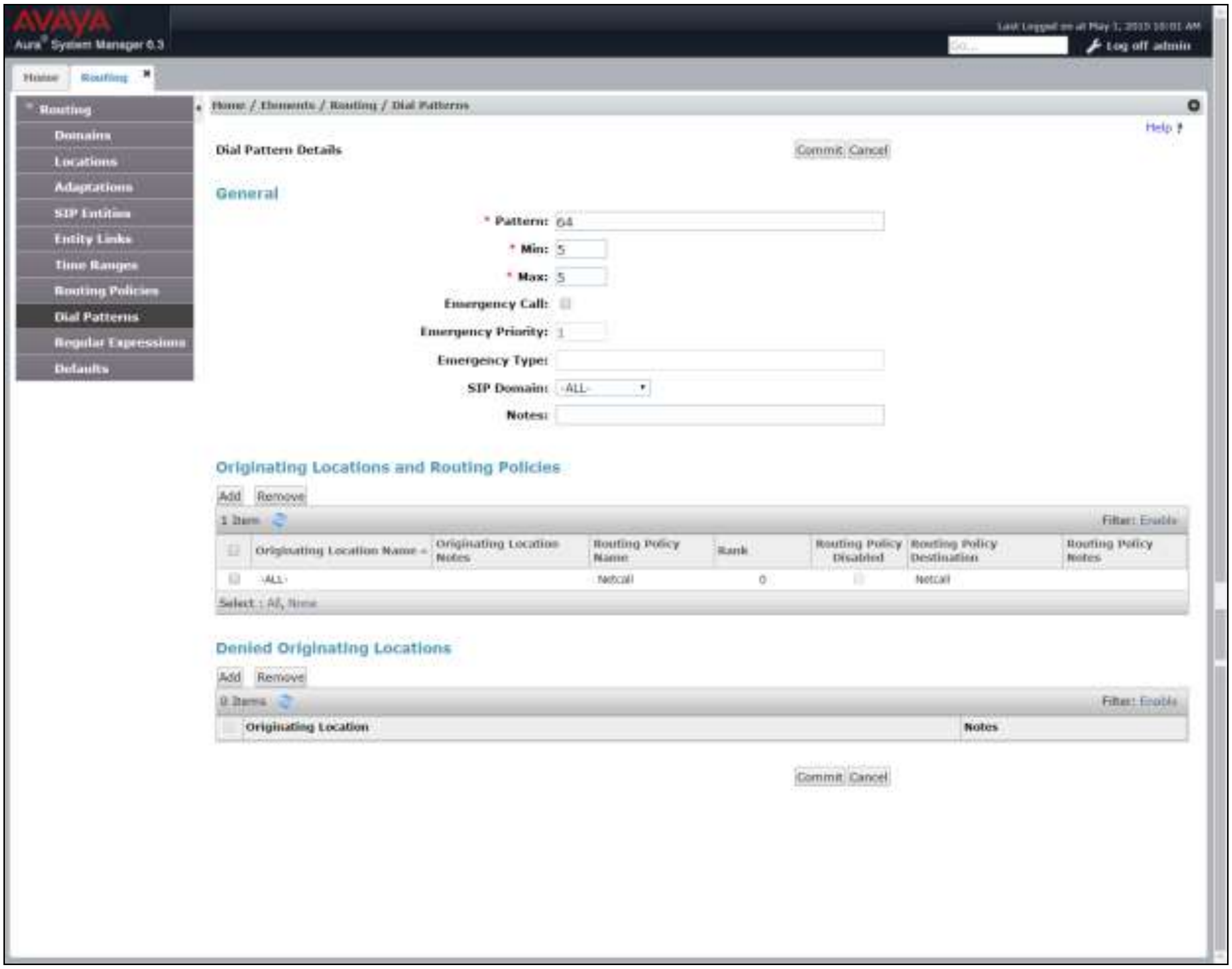
In **Section 5.7** Communication Manager is configured to route **5** digit strings beginning with **64** to Session Manager. To create a Dial Pattern to route these digits from Session Manager to Netcall Liberty click **Home → Elements → Routing → Dial Patterns →New.** Under **Pattern** enter the numbers presented to Session Manager by Communication Manager in the **Patterns** box. Set **Min** and **Max** digit string length, and set **SIP Domain** to **ALL**.



In the **Originating Locations and Routing Policies** section of the web page, click **Add.** In the **Origination Location** section place a tick in the box next to **Apply The Selected Routing Policies to All Originating Locations**, in the **Routing Policies** section click the routing policy created for Netcall Liberty. Click **Select** when done.



Click **Commit** when complete.



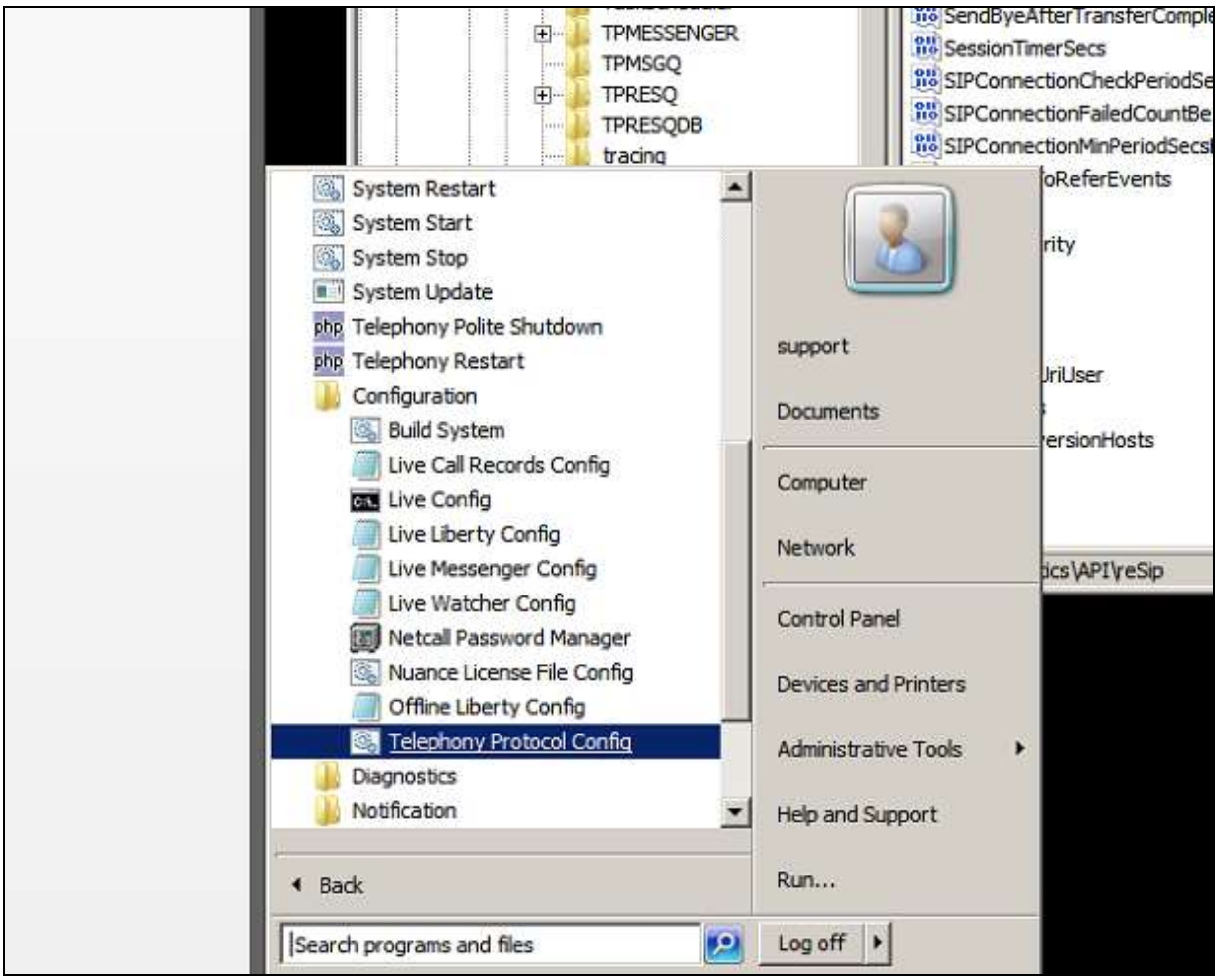
7. Configure Netcall Liberty

These Application Notes assume that a SIP build of Netcall Liberty v18.16 with a G5 Netcall Liberty processor or above is provided and installed by Netcall. Administration of the interface between Netcall Liberty and the Avaya solution is summarized as follows:

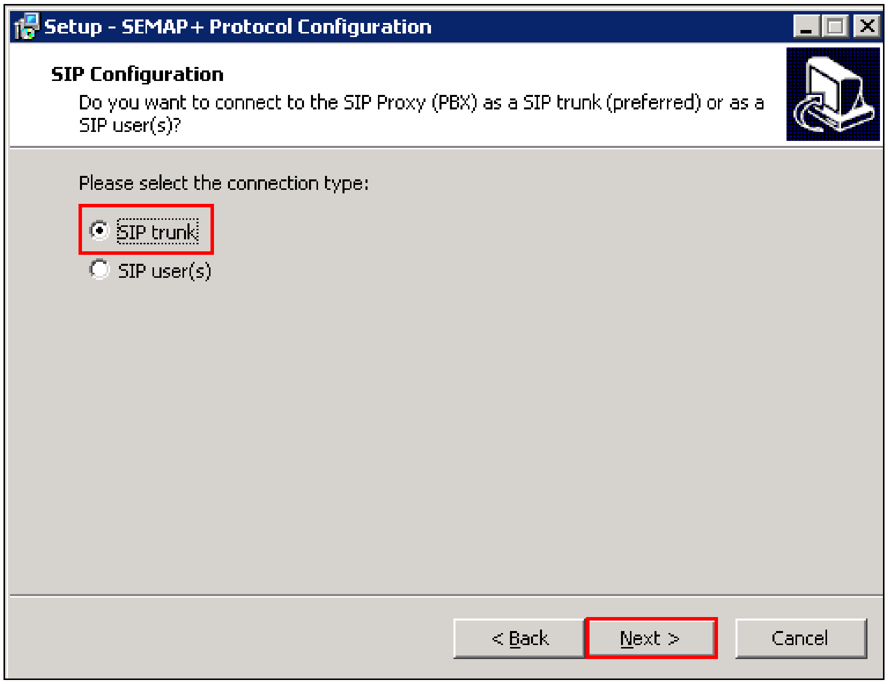
- Netcall Liberty Protocol Configuration Wizard Setup
- Configure Netcall Liberty Avaya MWI Special Settings
- Configure Netcall Liberty Routing Rules
- Configure Netcall Liberty Number Range Details
- Configure Netcall Liberty Test User and Mailbox

7.1. Netcall Liberty Protocol Configuration Wizard Setup

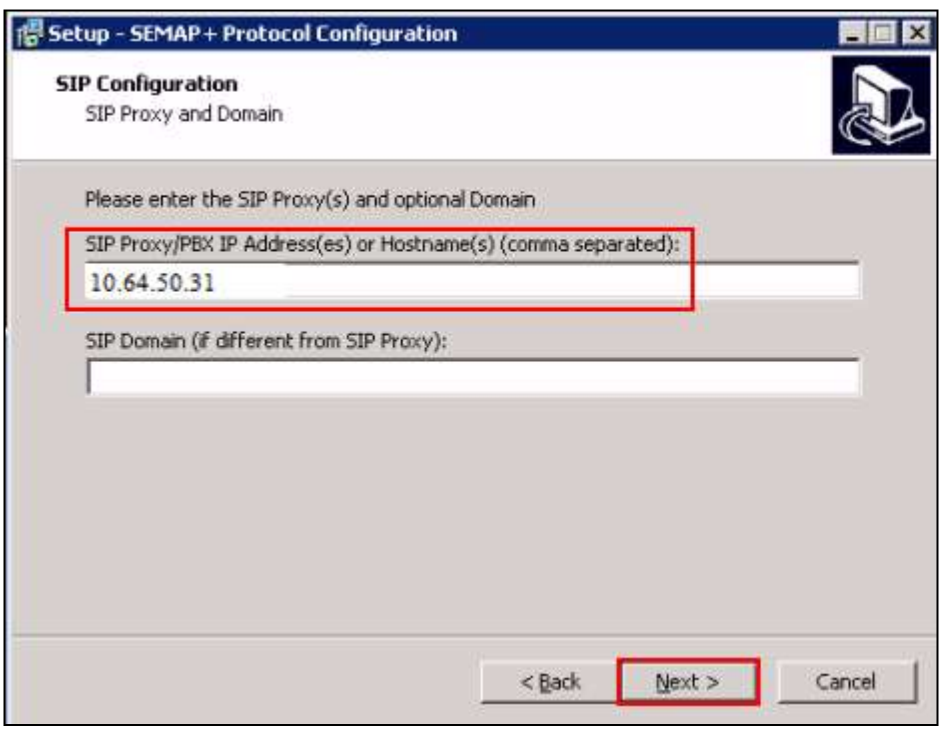
On the Netcall Liberty server log in to the Windows OS and click **Start → All Programs → Netcall Liberty -> Configuration → Telephony Protocol Config**.



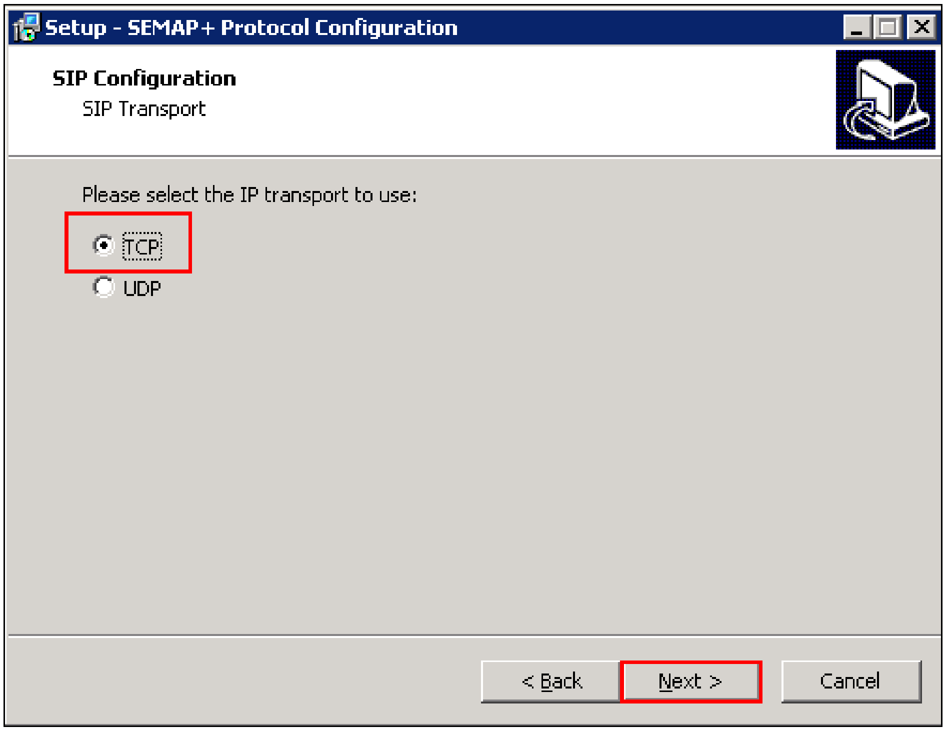
A familiar setup Microsoft Software Installer will start, choose the relevant Netcall Liberty program option (not shown) and click next. At the **SIP Configuration** page click the radio button next to **SIP trunk** and click **Next**.



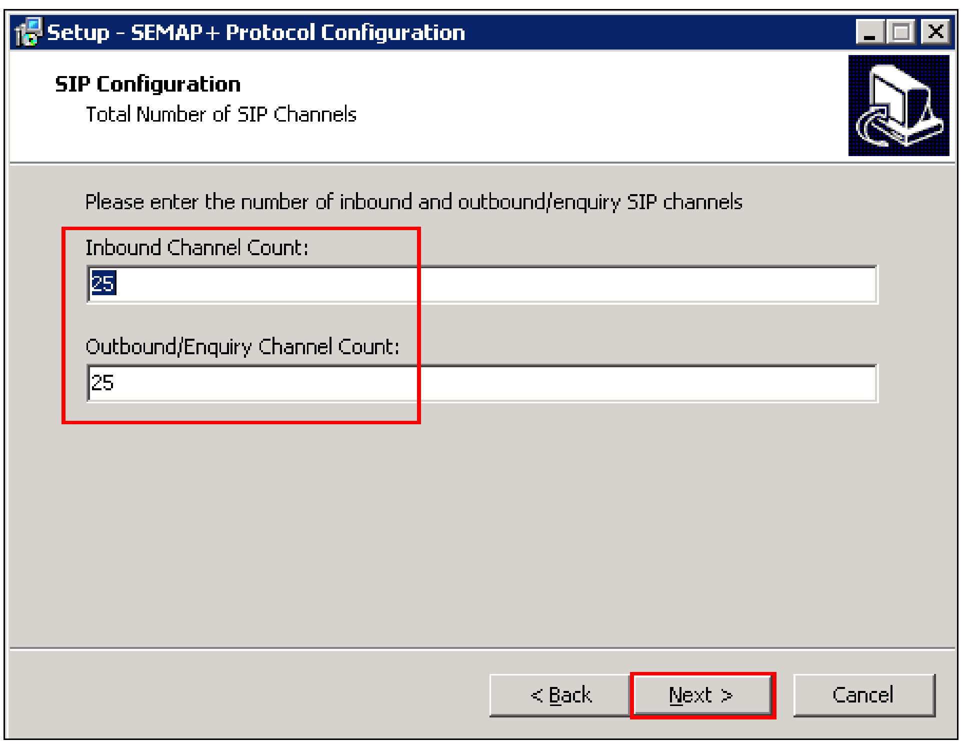
Enter the IP address of the Session Manager SIP Signalling interface in the **SIP Proxy/PBX IP Address(es) or Hostname(s) (comma separated):** box as shown below and click **Next**.



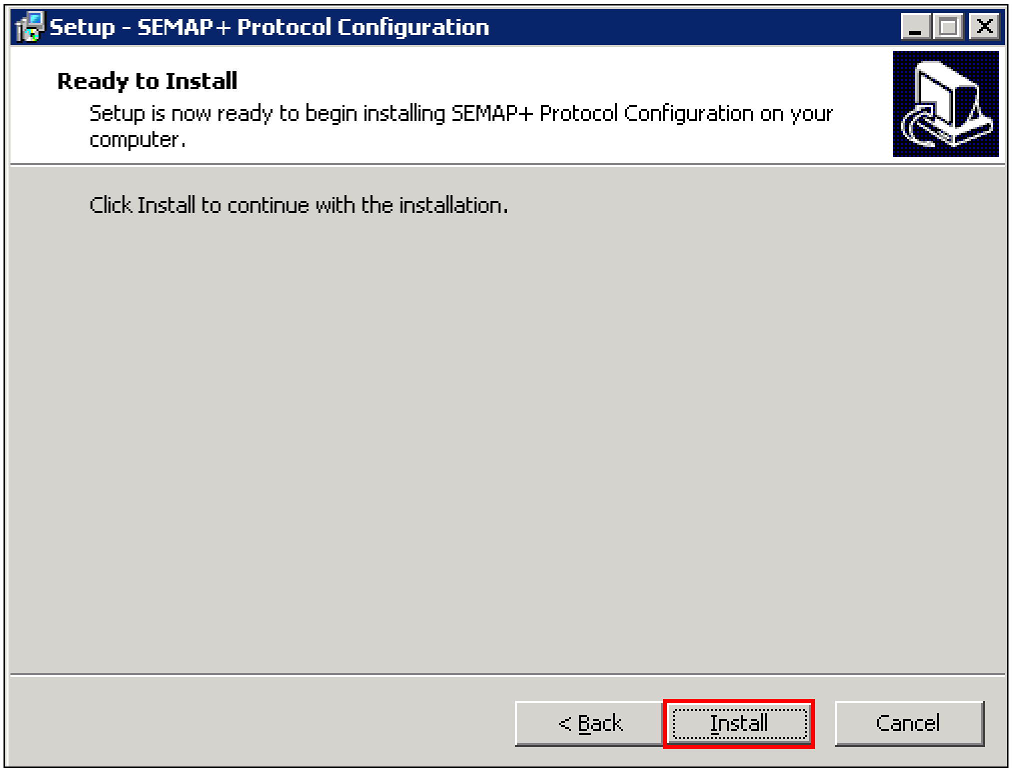
In the screen that appears, select **TCP** as the IP transport to use, and click **Next**.



Enter the **Inbound Channel Count** and **Outbound/Enquiry Channel Count**, according to requirements, as shown below and click **Next**.

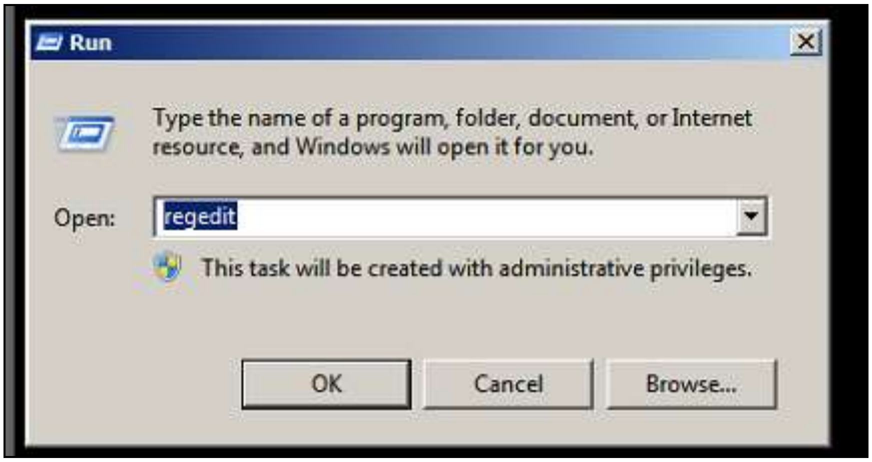


Click **Install** to complete the installation with the configuration specified.



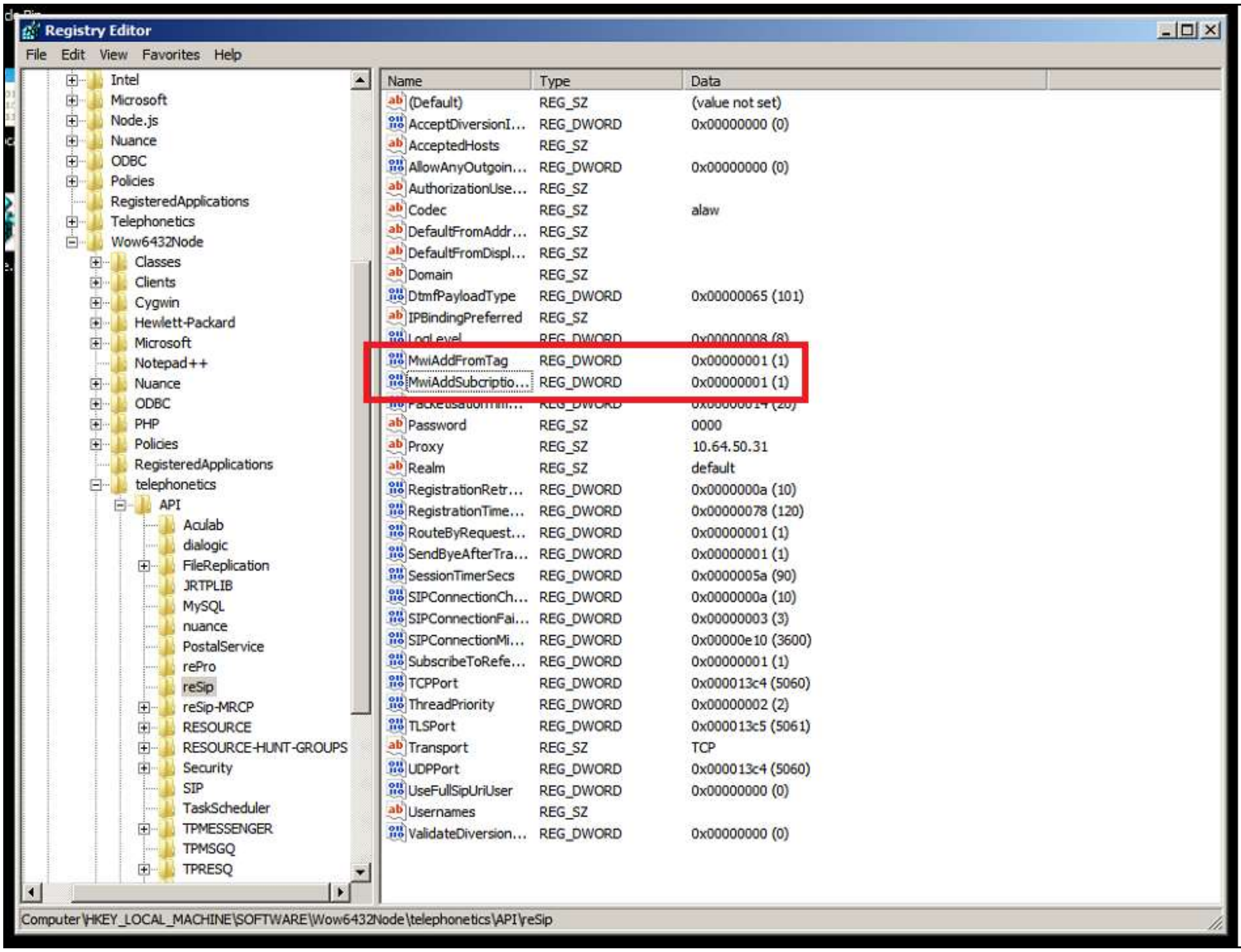
7.2. Configure Netcall Liberty Avaya MWI Special Settings

On the SEMAP system two registry keys need to be added to ensure the MWI is sent to Session Manager in the correct format. From the Netcall Liberty OS, click **Start → Run** type **regedit** in the Open box, and click **OK**.



Navigate to **[HKEY_LOCAL_MACHINE\SOFTWARE\wow6432node\Telephonetics\API\reSip]** and add the following as shown below:

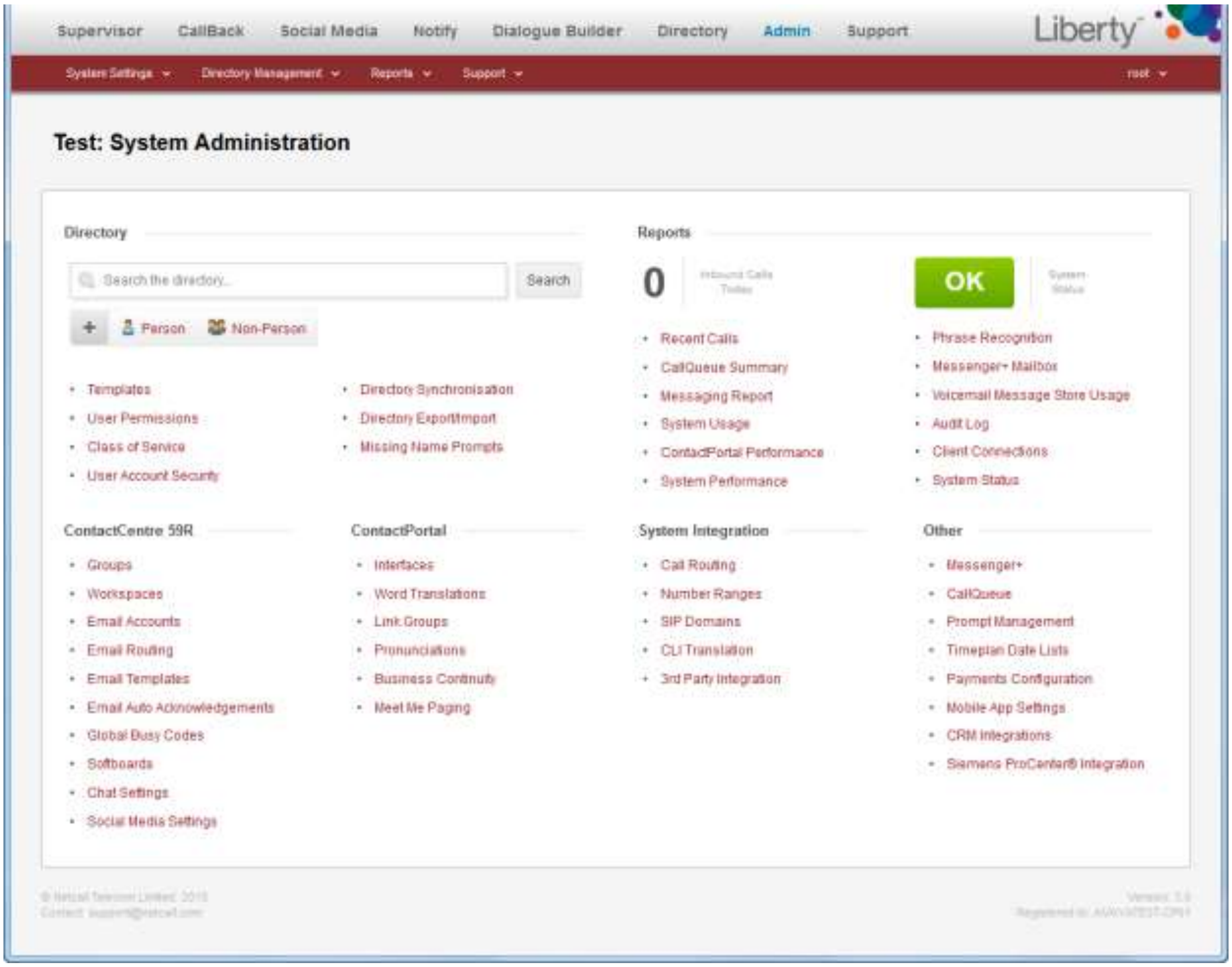
- **MwiAddSubcriptionState=dword:00000001**
- **MwiAddFromTag=dword:00000001**



Registry Editor
File Edit View Favorites Help
Intel
Microsoft
Node.js
Nuance
ODBC
Policies
RegisteredApplications
Telephonetics
Wow6432Node
Classes
Clients
Cygwin
Hewlett-Packard
Microsoft
Notepad++
Nuance
ODBC
PHP
Policies
RegisteredApplications
telephonetics
API
Aculab
dialogic
FileReplication
JRTPLIB
MySQL
nuance
PostalService
rePro
reSip
reSip-MRCP
RESOURCE
RESOURCE-HUNT-GROUPS
Security
SIP
TaskScheduler
TPMESSENGER
TPMSGQ
TPRESQ
Name Type Data
(Default) REG_SZ (value not set)
AcceptDiversionI… REG_DWORD 0x00000000 (0)
AcceptedHosts REG_SZ
AllowAnyOutgoin… REG_DWORD 0x00000000 (0)
AuthorizationUse… REG_SZ
Codec REG_SZ alaw
DefaultFromAddr… REG_SZ
DefaultFromDispl… REG_SZ
Domain REG_SZ
DtmfPayloadType REG_DWORD 0x00000065 (101)
IPBindingPreferred REG_SZ
MwiAddFromTag REG_DWORD 0x00000001 (1)
MwiAddSubcriptio… REG_DWORD 0x00000001 (1)
Password REG_SZ 0000
Proxy REG_SZ 10.64.50.31
Realm REG_SZ default
RegistrationRetr… REG_DWORD 0x0000000a (10)
RegistrationTime… REG_DWORD 0x00000078 (120)
RouteByRequest… REG_DWORD 0x00000001 (1)
SendByeAfterTra… REG_DWORD 0x00000001 (1)
SessionTimerSecs REG_DWORD 0x0000005a (90)
SIPConnectionCh… REG_DWORD 0x0000000a (10)
SIPConnectionFai… REG_DWORD 0x00000003 (3)
SIPConnectionMi… REG_DWORD 0x00000e10 (3600)
SubscribeToRefe… REG_DWORD 0x00000001 (1)
TCPPort REG_DWORD 0x000013c4 (5060)
ThreadPriority REG_DWORD 0x00000002 (2)
TLSPort REG_DWORD 0x000013c5 (5061)
Transport REG_SZ TCP
UDPPort REG_DWORD 0x000013c4 (5060)
UseFullSipUriUser REG_DWORD 0x00000000 (0)
Usernames REG_SZ
ValidateDiversion… REG_DWORD 0x00000000 (0)
Computer\HKEY_LOCAL_MACHINE\SOFTWARE\Wow6432Node\telephonetics\API\reSip

7.3. Configure Netcall Liberty Routing Rules

The Netcall Liberty routing rules must be configured with the access number assigned in **Section 5.6**. Access the web interface of the Netcall Liberty server using http://IP_OF_Netcall Liberty_SERVER and log in using the credentials provided by Netcall. The following home page will appear.



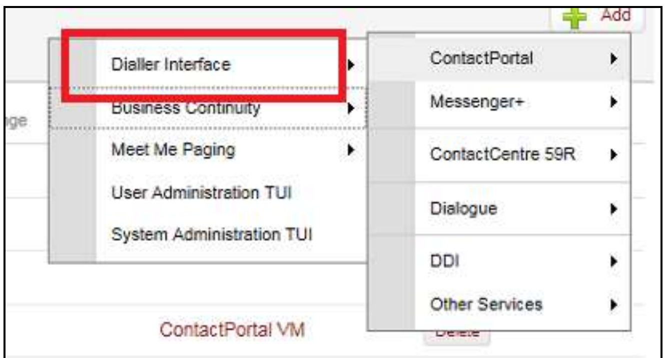
Click **Admin → Call Routing**.

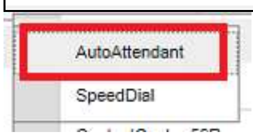
Click on the **Add** button.

For the purposes of the compliance test, three call routing rules are created:

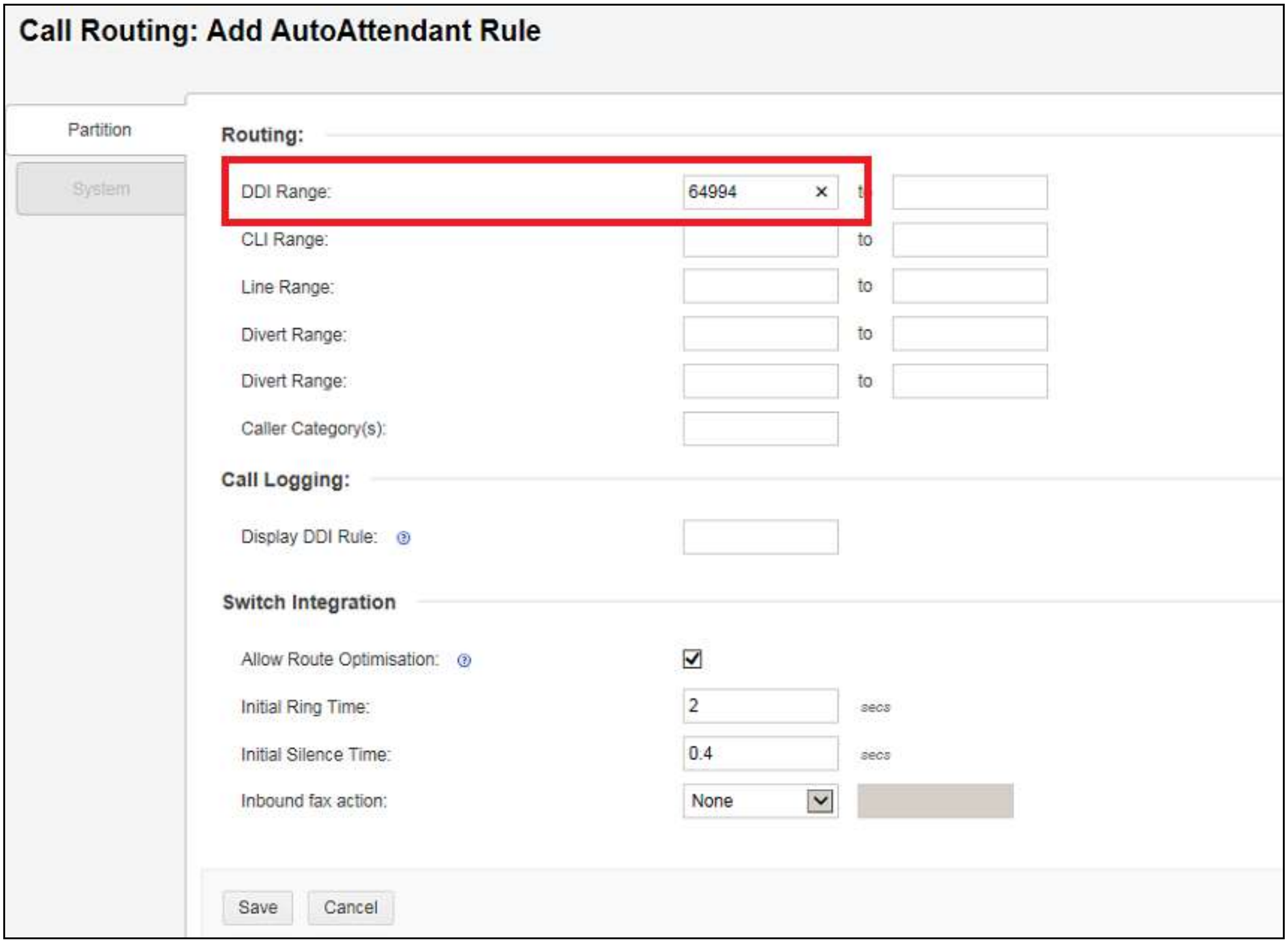
- 64994 – ContactPortal AutoAttendant
- 69999 – Voicemail Deposit
- 69999 - Voicemail Retrieval

Select **AutoAttendant** under the **ContactPortal/Dialer Interface** section, from the **Add a new routing rule** drop down box.



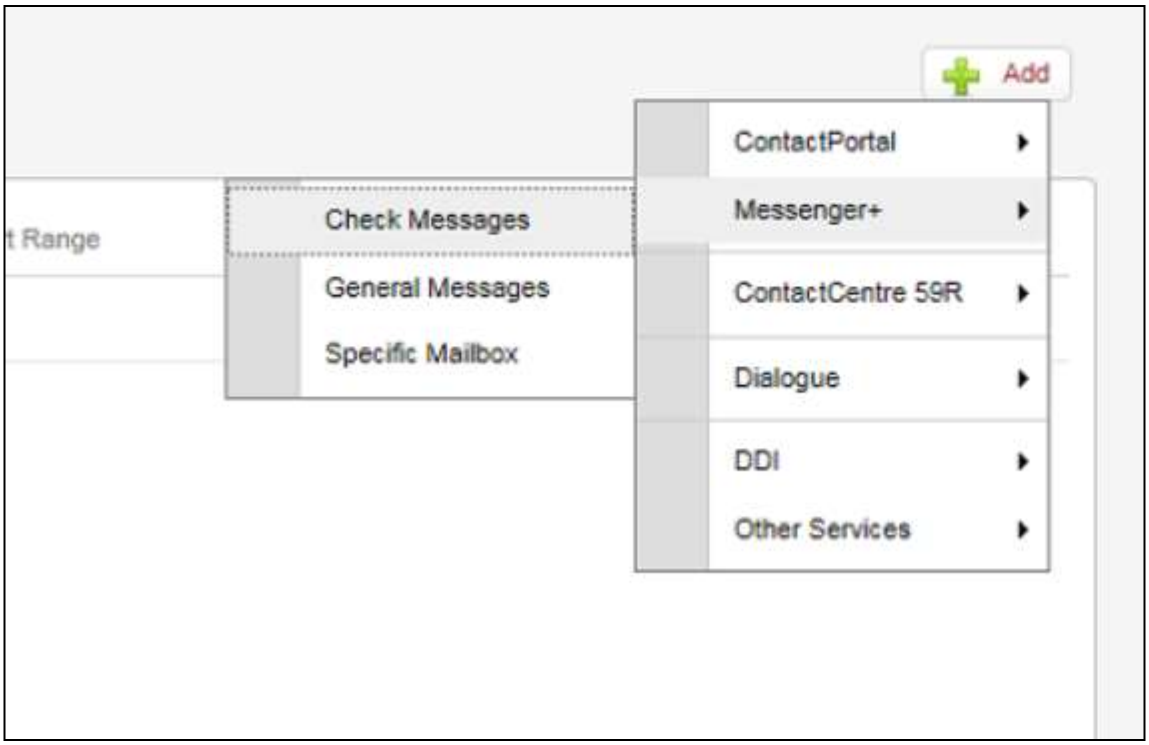


Enter **64994** in the **DDI Range** box, leave all other settings as default and click **Save**.

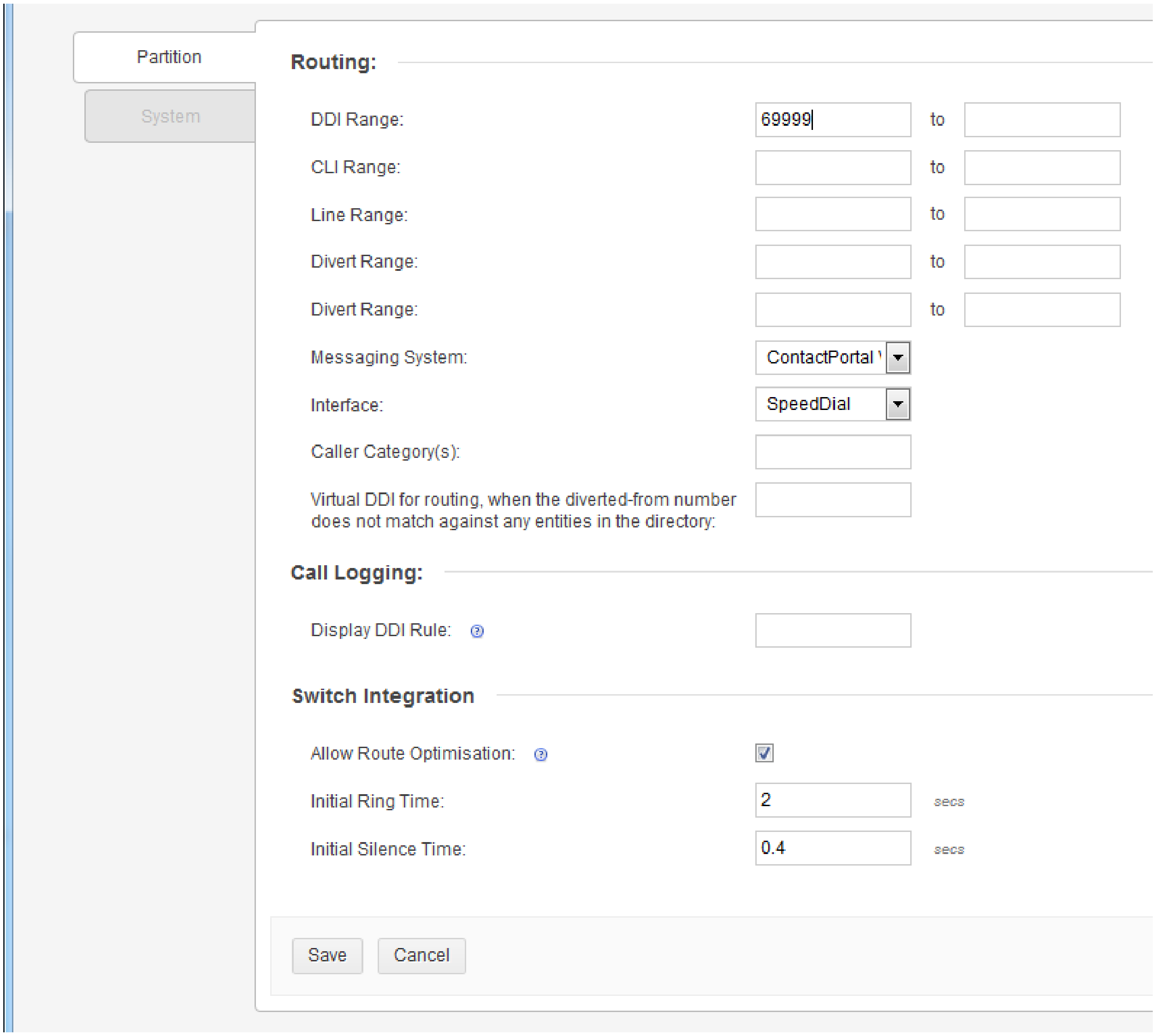


The screen below will appear showing the added **AutoAttendant** Rule.

Select **Check Messages** under the **Messenger+** section, from the **Add a new routing rule** drop down box.

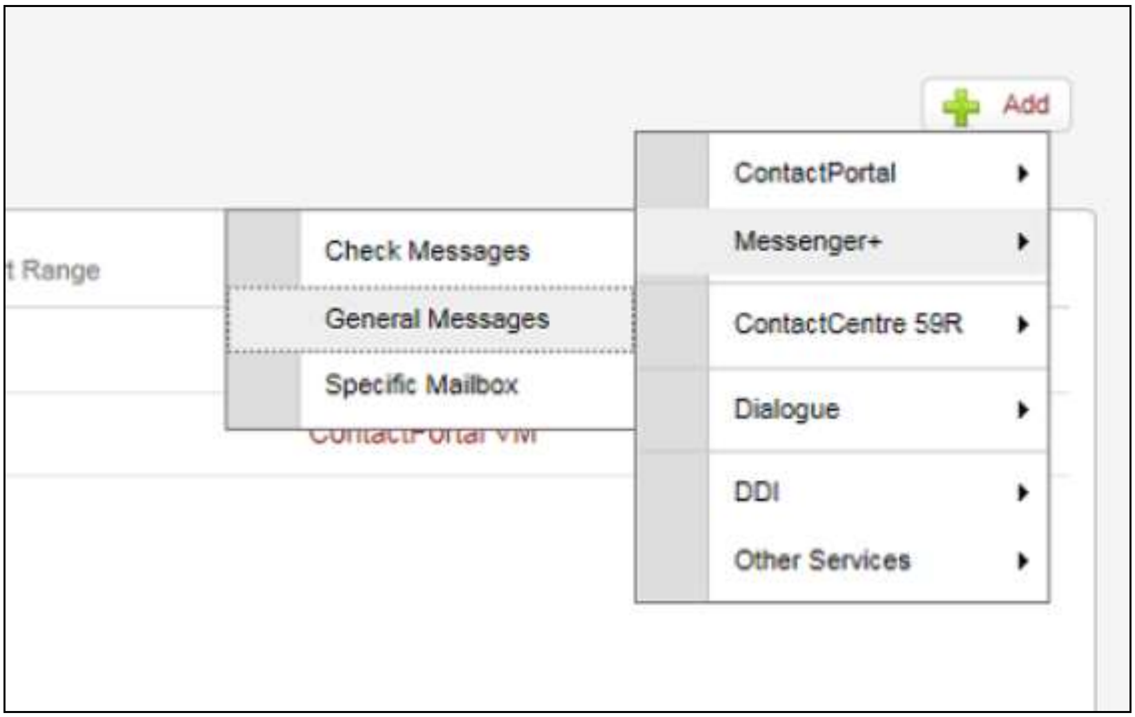


Enter **69999** in the **DDI** field, leave all others settings as default and click **Add Rule**.

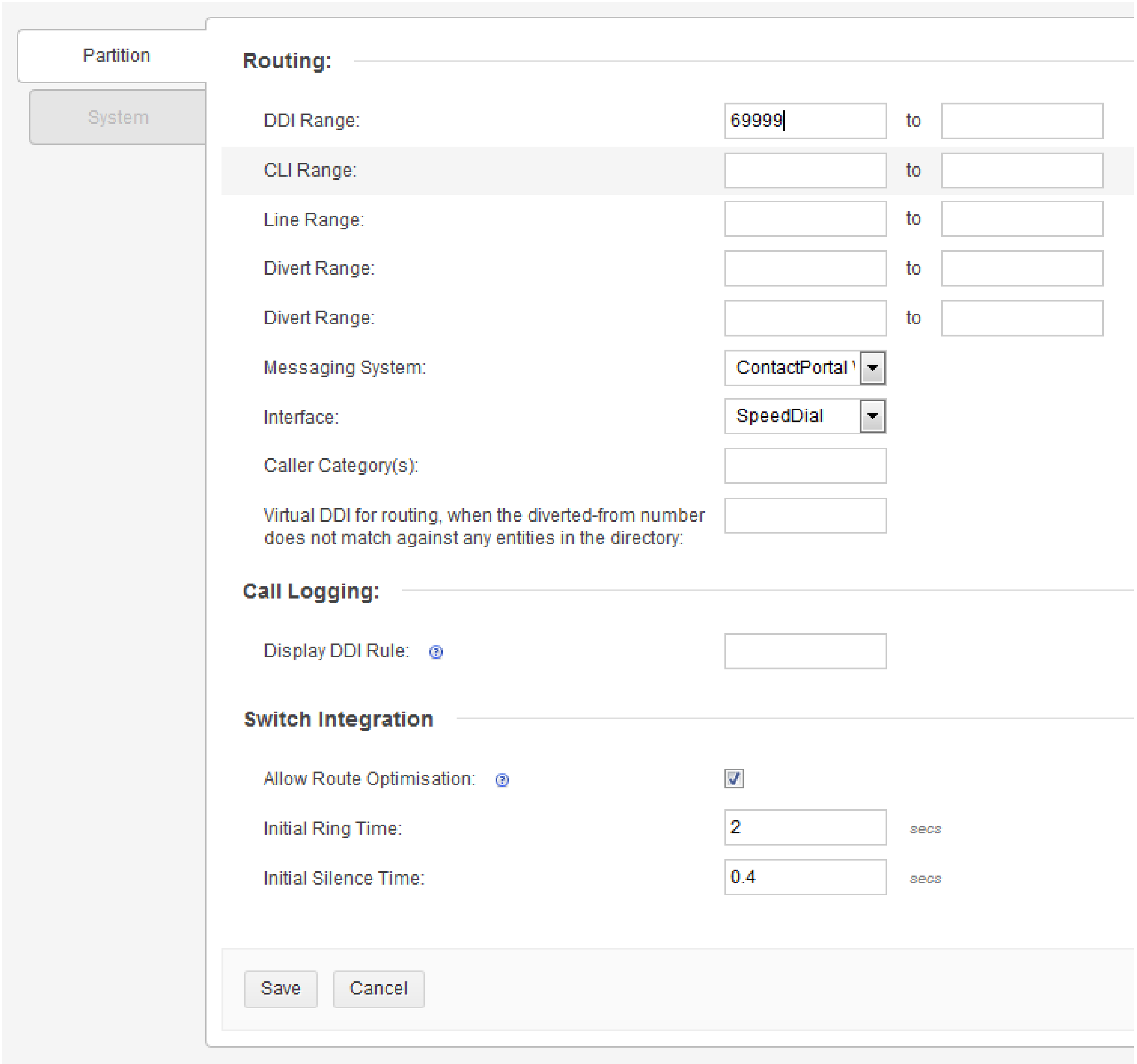


The screen below will appear showing the added **Check Messages** Rule.

Select **General Messages** under the **Messenger+** section, from the **Add a new routing rule** drop down box.



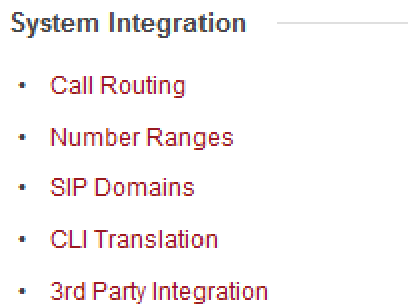
Enter **69999** in the **DDI** field, leave all others settings as default and click **Add Rule**.



The screen below will appear showing the added **General Messages** Rule.

7.4. Configure Netcall Liberty Number Range Details

Routing rules must be configured to define internal, external and barred numbers. From the home page, click **Admin → Number Ranges**.

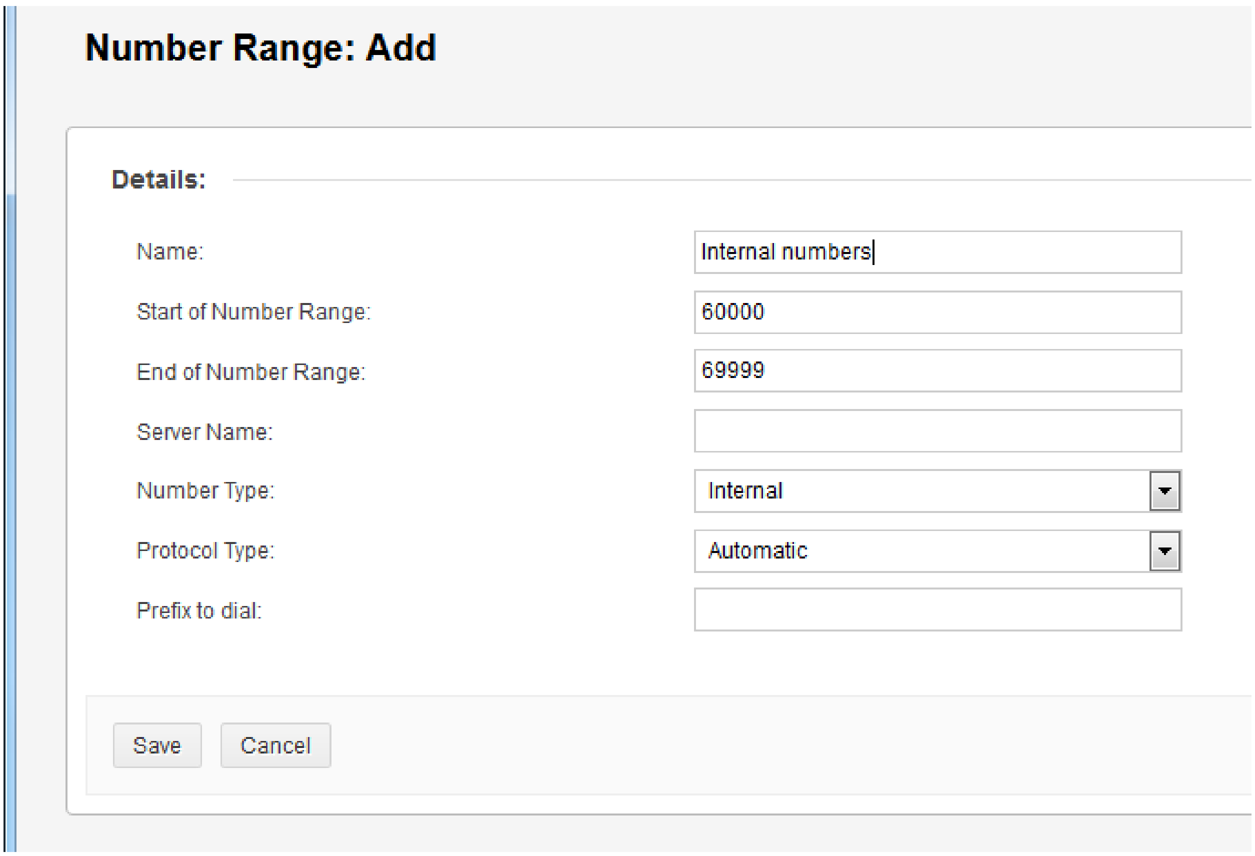


Click the **Add** button.

For the purposed of the compliance test, three new routing rules are added:

- Internal Numbers
- External Number
- Barred Number

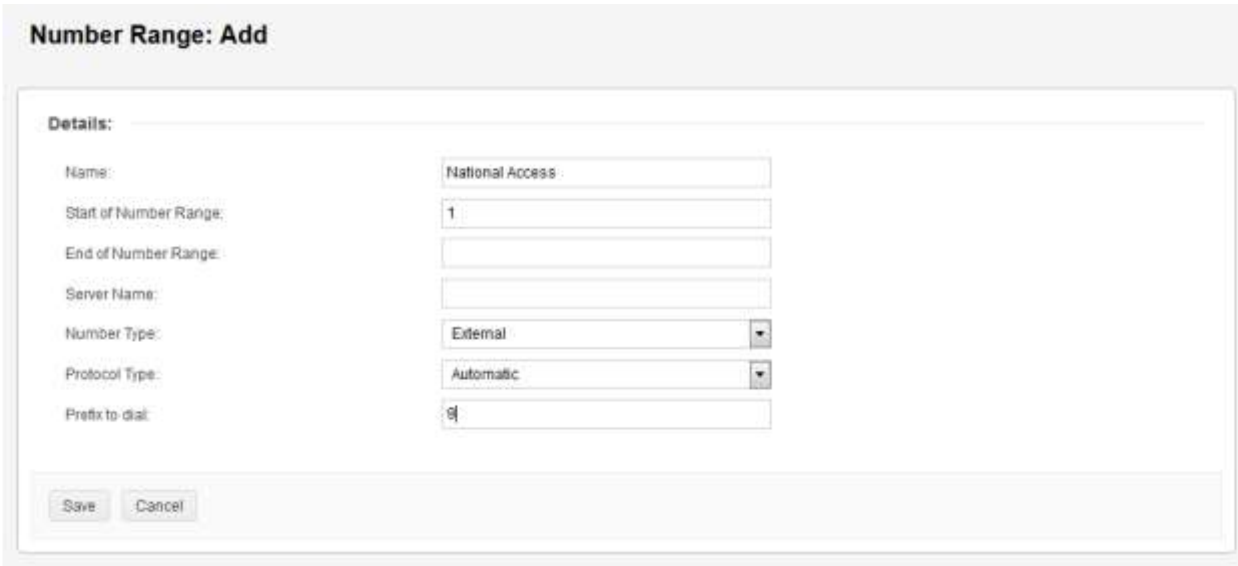
In the **Name** field enter an identifying name for the rule, in this case **Avaya Internal Test Extensions**, enter the **Start of Number Range** and **End of Number Range** accordingly, in this case **60000** and **69999** respectively, For compliance testing, extensions 60000 through 69999 represented internal extensions. Leave all other fields as default and click **Add**.



The following screen will appear displaying the new rule.

Note that **Optimise** is disabled, this rule can be edited to enable Optimise. This feature of Netcall Liberty uses SIP REFER to free up Netcall Liberty resources.

Click the **Add button again,** in the **Name** field enter an identifying name for the rule, in this case **Outside Line**, set the **Start of Number Range** as **1**, this it the first digit for an external call, select **External** from the **Number Type** drop down box, and set **Prefix to dial** as **9** leave all other fields as default and click **Save**.

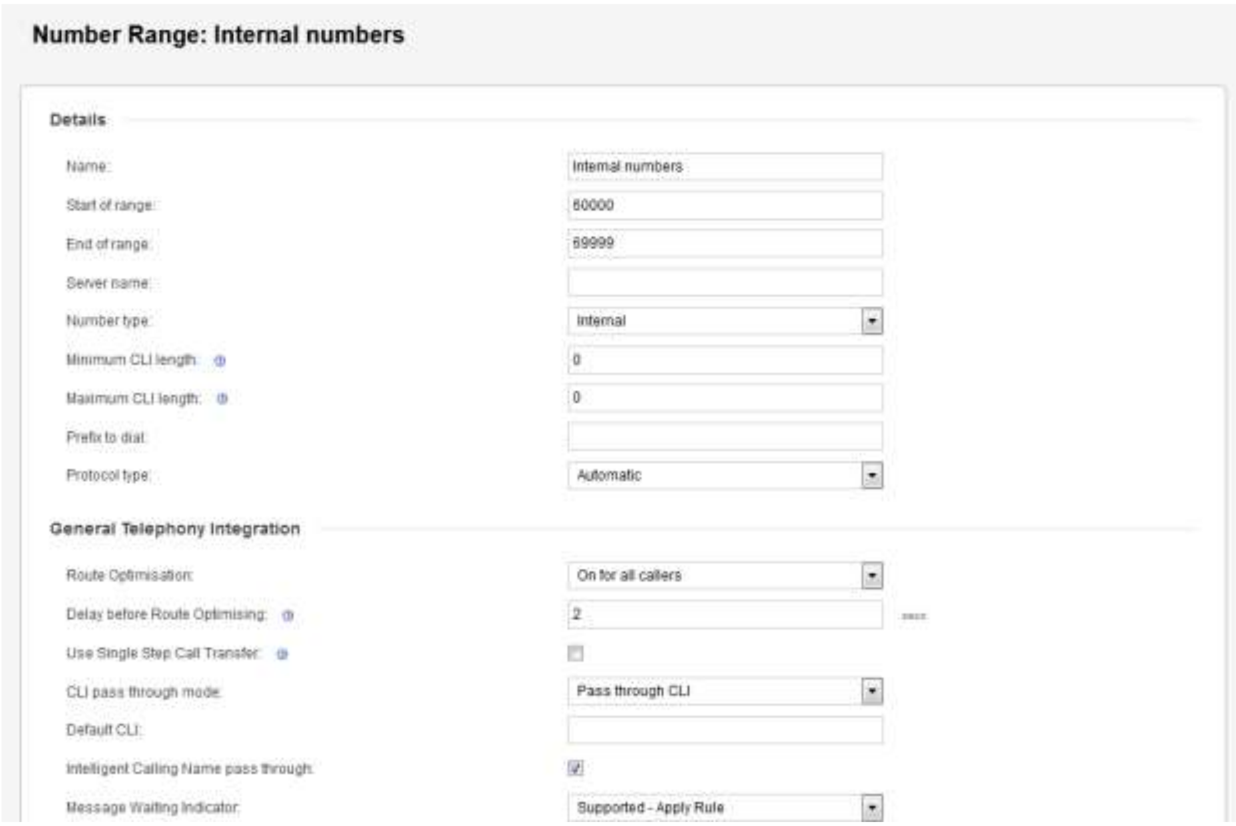


The following screen will appear displaying the new rule.

Click **Add a new rule,** in the **Name** field enter an identifying name for the rule, in this case **Barred Number**, set the **Start of Number Range** as **60002**, leave all other fields as default and click **Add**.

The following screen will appear displaying the new rule.

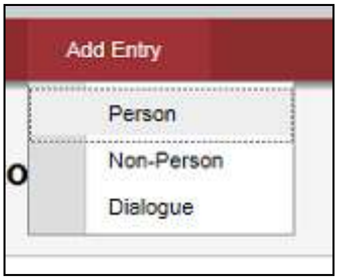
Once administered, the routing rules can be edited, click on the routing rule to edit, by default **Route Optimisation** is **Off** and **Delay** is set to **0,** change **Route Optimisation** to **On for all callers** and **Delay Before Route Optimising (secs)** to **2**. This will enable SIP REFER to take place 2 seconds after the established call.



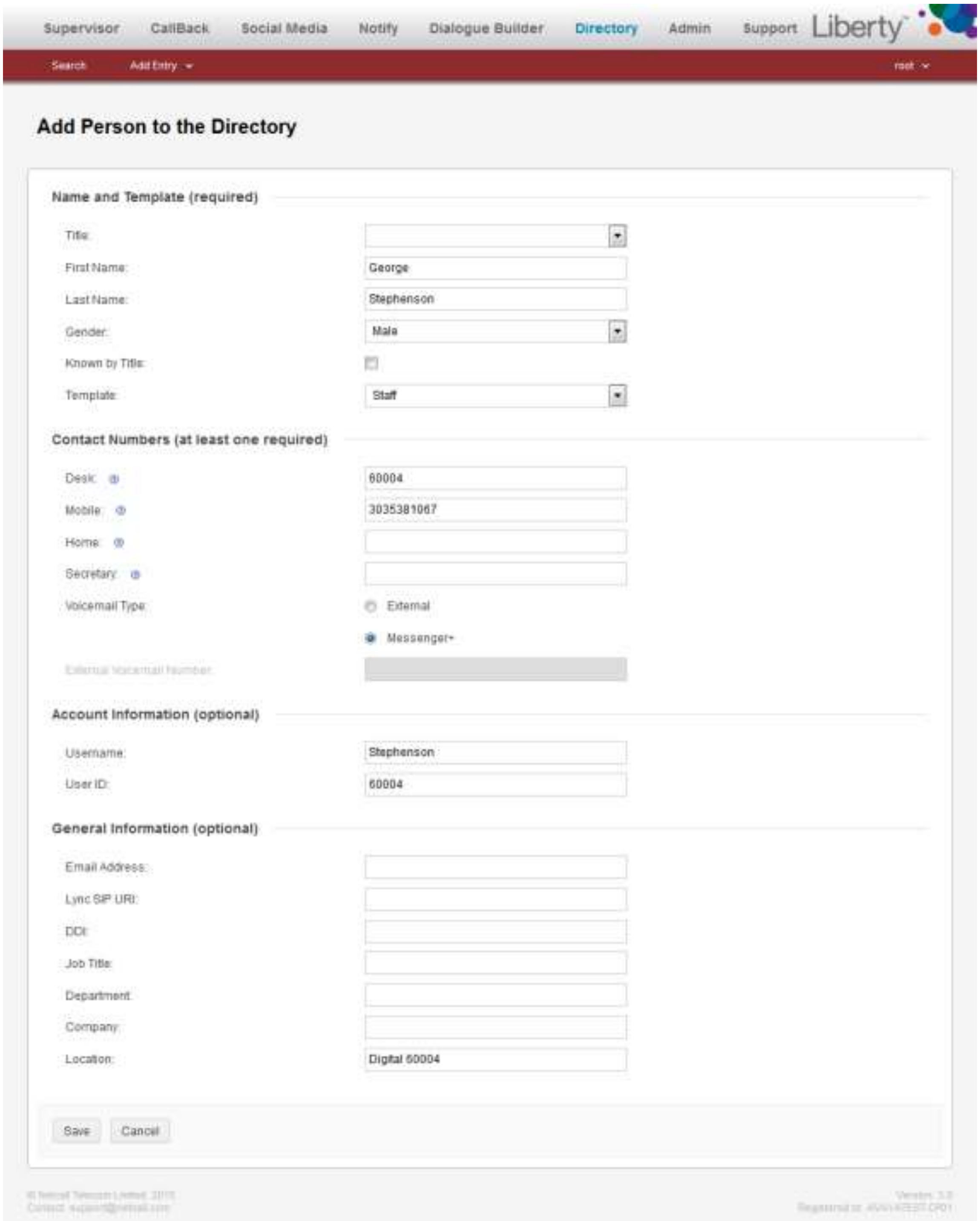
Scroll down the page and select **Always reINVITE** from the **SIP reINVITE Policy** drop down box, this will enable shuffling, it is not recommended that this is used where call recording is used in the solution. Click **Update** when done.

7.5. Configure Netcall Liberty Test User and Mailbox

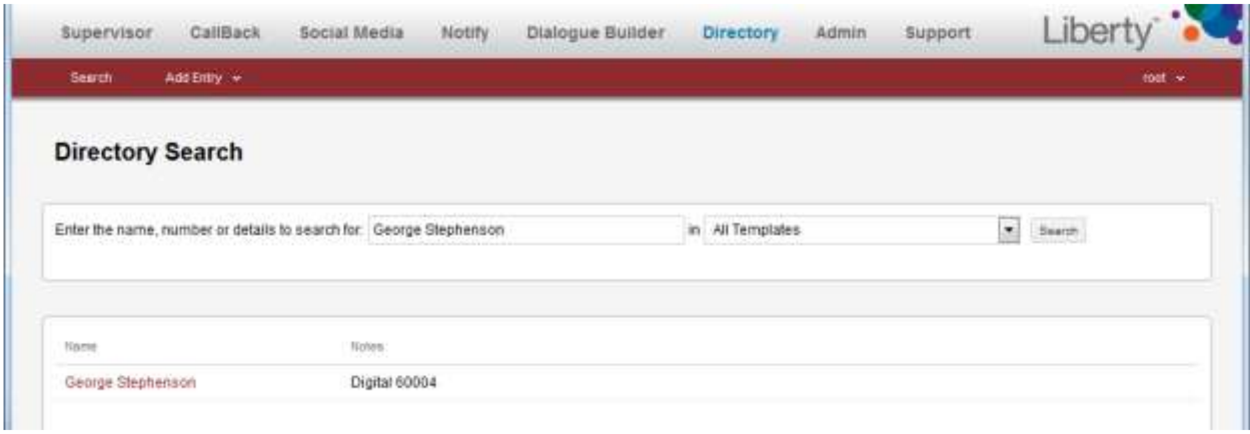
For the purposes of the compliance test, users are added manually. From the home page, click **Directory** and the screen below will appear, click on **Add Entry → Person**.



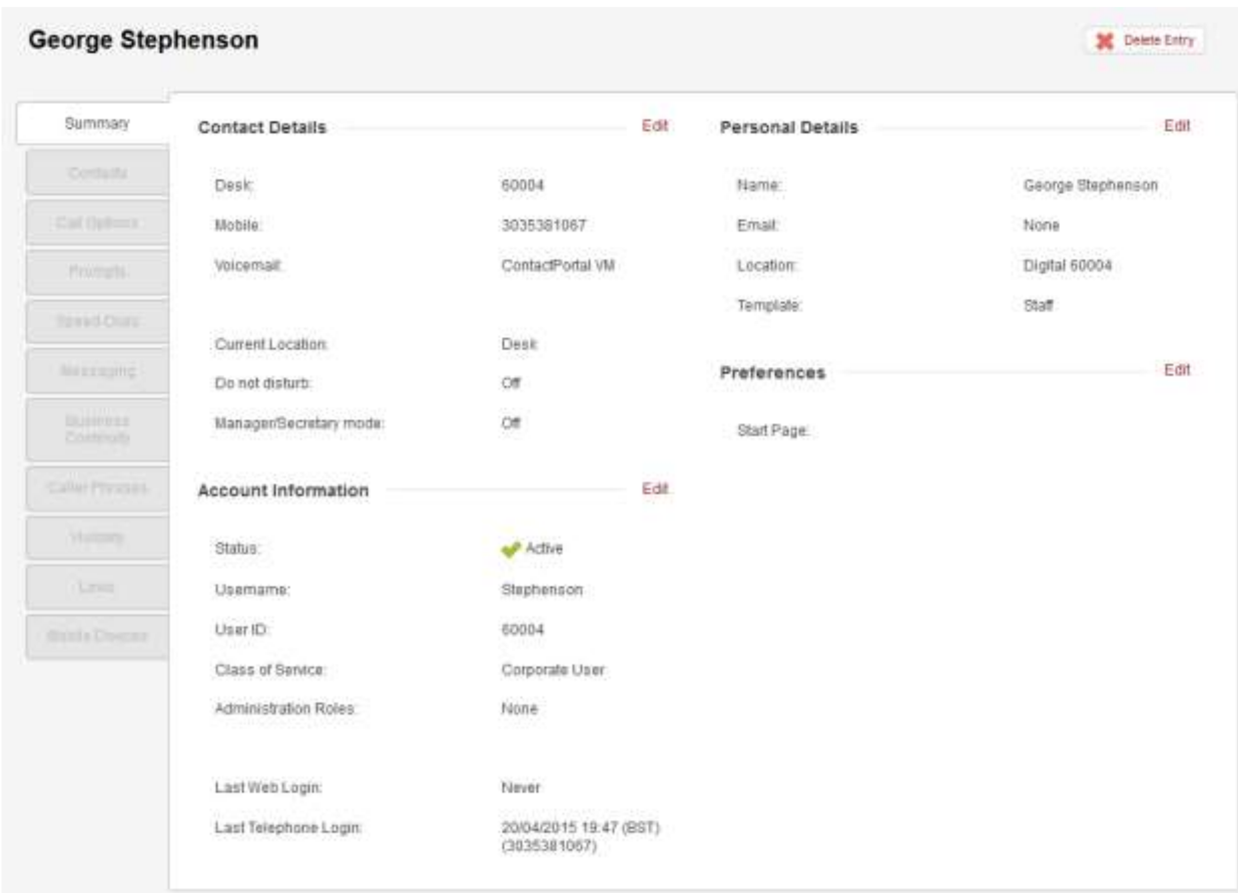
Enter the details and extension number as required, in this case the **Title, First Name** and **Last Name** are defined. Under the **Contact Numbers** section the **Desk** number is set to **4001** and a tick is placed next to **Messenger+** to define that this user has a voicemail box. In the **General Information** section the **User ID** is set to **60004** , click **Add Entry** (not shown) when done.



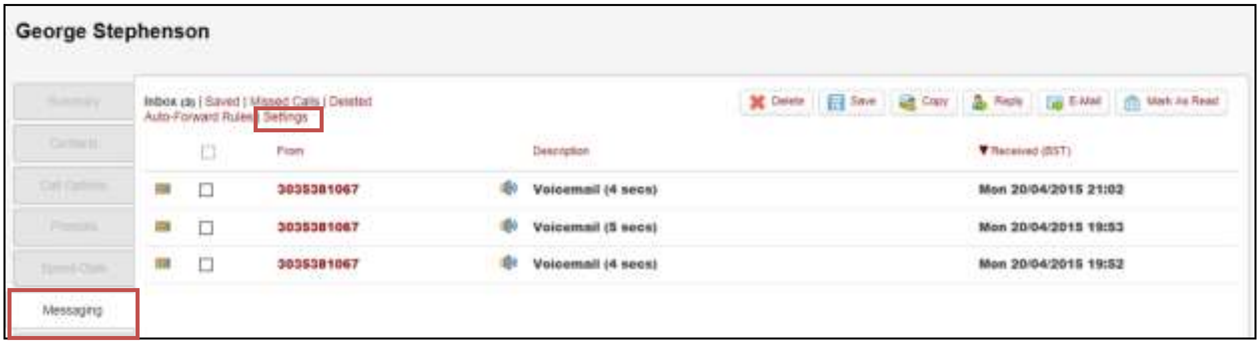
From the home page click **Directory → Search.** Enter the name of the added user and click the **Search button**.



Click the **Name** (e.g. Georg Stephenson) from the search result above.



Click **Messaging** and then **Settings**.



Ensure a tick is placed next to **MWI** and click **Update Notification Settings** (not shown).

8. Verification Steps

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

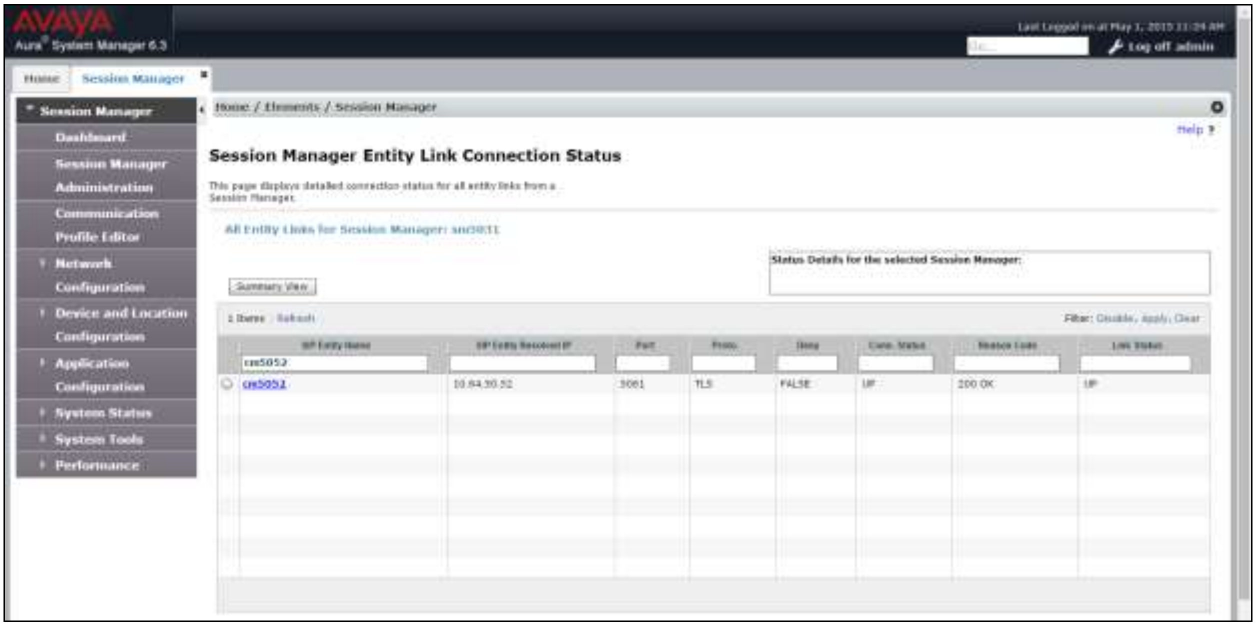
8.1. Verify Avaya Aura® Communication Manager SIP Trunk

Enter the command **status trunk x** where x is the trunk configured in **Section 5.3**. Confirm all channels are **in-service/idle**.

```
status trunk 2                                                      Page   1

                              TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0002/001 T00009   in-service/idle    no
0002/002 T00010   in-service/idle    no
0002/003 T00011   in-service/idle    no
0002/004 T00012   in-service/idle    no
0002/005 T00013   in-service/idle    no
0002/006 T00014   in-service/idle    no
0002/007 T00015   in-service/idle    no
0002/008 T00016   in-service/idle    no
0002/009 T00017   in-service/idle    no
0002/010 T00018   in-service/idle    no
0002/011 T00019   in-service/idle    no
0002/012 T00020   in-service/idle    no
0002/013 T00021   in-service/idle    no
0002/014 T00022   in-service/idle    no
```

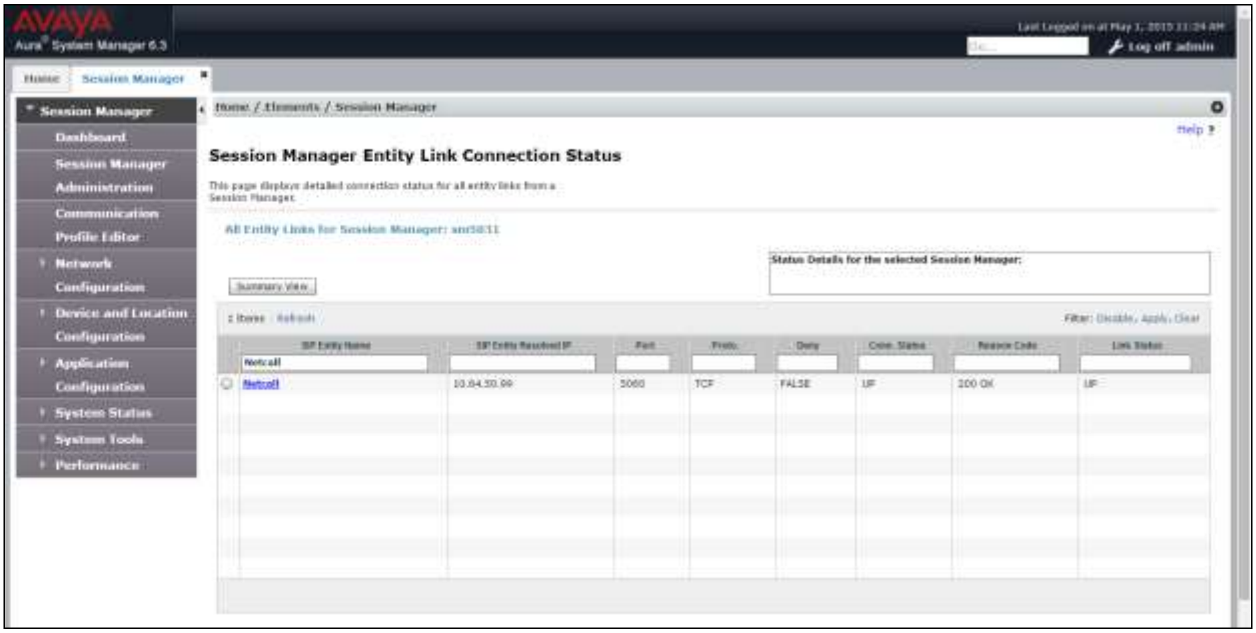
8.2. Verify Avaya Aura® Communication Manager Entity Link Status

From the System Manager web interface click **Home → Elements → Session Manager → System Status → SIP Entity Monitoring →Commgr → Show** verify **Conn. Status** and **Link Status** is **UP**.



8.3. Verify Netcall Liberty Entity Link Status

From the System Manager web interface click **Home → Elements → Session Manager → System Status → SIP Entity Monitoring →Netcall Liberty → Show** verify **Conn. Status** and **Link Status** is **UP**.



AVAYA
Aura® System Manager 6.3
Log off admin
Home
Session Manager
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
System Tools
Performance
Home / Elements / Session Manager
Help ?
Session Manager Entity Link Connection Status
All Entity Links for Session Manager: sm5831
Status Details for the selected Session Manager:
Summary View
Refresh
SIP Entity Name
SIP Entity Resolved IP
Port
Proto.
Deny
Conn. Status
Reason Code
Link Status
Netcall
10.64.50.99
5060
TCP
FALSE
UP
200 OK
UP

8.4. **Verify Netcall Liberty Subsystem Status**

From the Netcall Liberty web interface, click **System Staus**, verify all systems are **UP**.

8.5. **Verify Netcall Liberty Access**

Manually place a call to the Netcall Liberty and verify successful interpretation of ASR and appropriate response of the Messaging+ application.

8.6. Verify Netcall Liberty Logging and Netcall Liberty Version

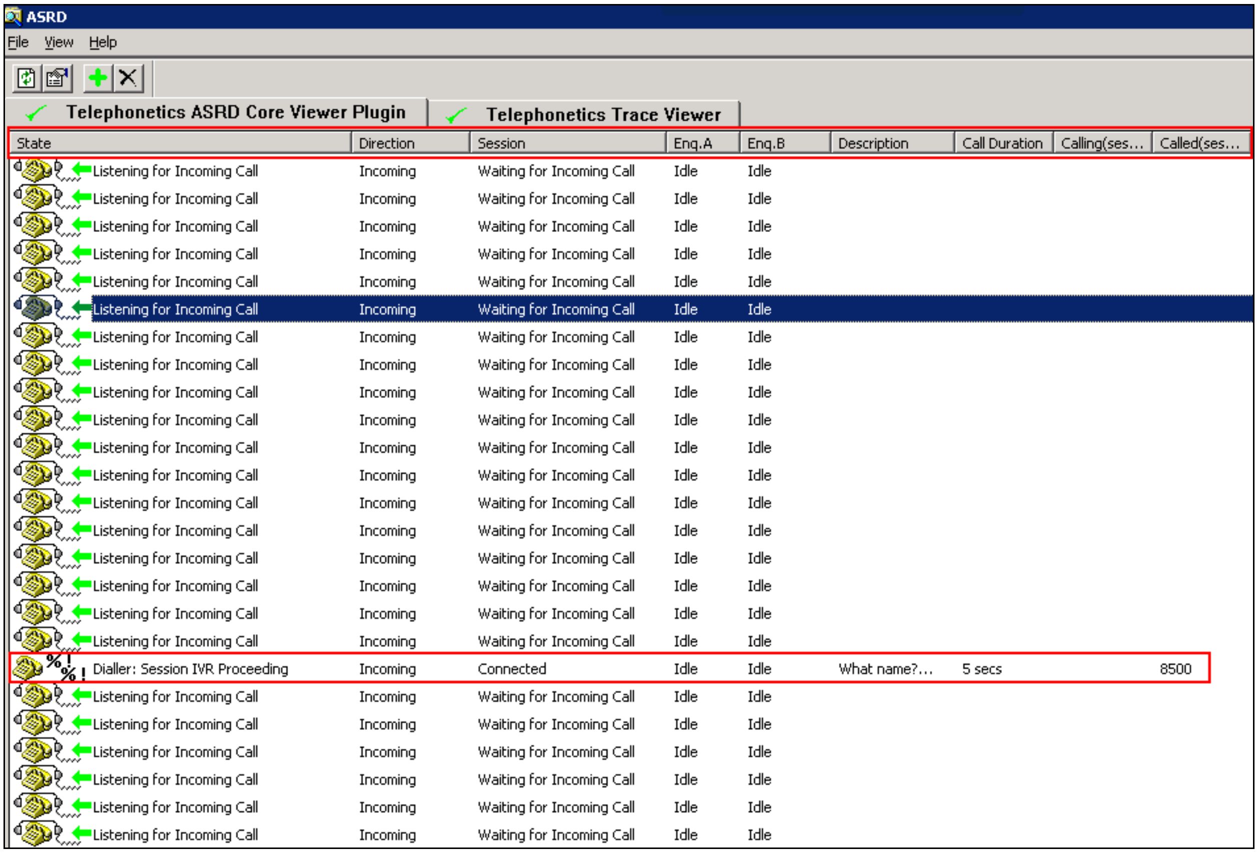
On the Netcall Liberty server, browse to **E:\tptrace\<date>.log** and verify the file contains application logging information including SIP messaging. Confirm that the **User-Agent** is shown as **ContactPortal/18.16**.

```
OPTIONS sip:10.10.16.201:5060;transport=tcp SIP/2.0
Via: SIP/2.0/ ;branch=z9hG4bK-d8754z-9e3c2854a804f834-1---d8754z-;rport
Max-Forwards: 70
Contact: <sip:contactportal>
To: <sip:10.10.16.201:5060;transport=tcp>
From: <sip:contactportal@10.10.16.201>;tag=415d4964
Call-ID: YTYzOTA3OGM4MDQ2ZTVmZmFjNjQzMWUxMjA1MWUzY2E.
CSeq: 1 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO, NOTIFY
Supported: timer
Content-Length: 0
```

```
Contact: <sip:8500@10.10.17.2>
To: <sip:8500@10.10.17.2>;tag=bc77c625
From: "PSTN,4000"<sip:anonymous@anonymous.invalid>;tag=0d84ae9625de117f4f475d9000
Call-ID: 0d84ae9625de118f4f475d9000
CSeq: 1 INVITE
Session-Expires: 1200;refresher=uac
Min-SE: 1200
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO, NOTIFY
Content-Type: application/sdp
Supported: timer
User-Agent: ContactPortal/18.16
Content-Length: 207
```

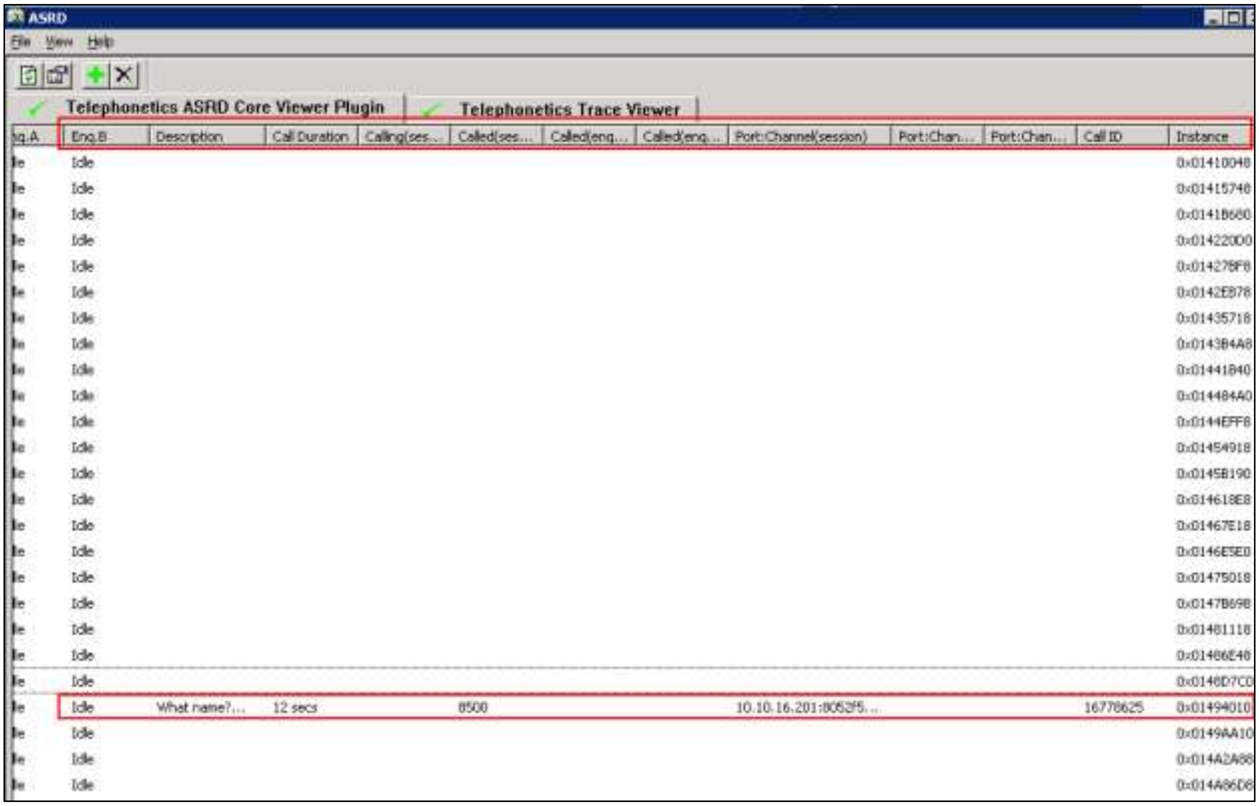
8.7. Verify Netcall Liberty Real Time Line Viewer

Click **Start→ ContactPortal (System) → Real Time Line Viewer**, place call to Netcall Liberty and verify the **State, Direction, Session, Description, Call Duration** and **Called** are accurately presented.



State	Direction	Session	Eng.A	Eng.B	Description	Call Duration	Calling(ses…	Called(ses…
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Dialler: Session IVR Proceeding	Incoming	Connected	Idle	Idle	What name?…	5 secs		8500
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				
Listening for Incoming Call	Incoming	Waiting for Incoming Call	Idle	Idle				

On the same screen, use the bar at the bottom of the screen to scroll to the right and verify the **Port:Channel(session)** column displays a connection to the IP address of the Session Manager SIP Signaling Interface.



9. Conclusion

These Application Notes describe the configuration steps required for Netcall Liberty to successfully interoperate with Avaya Aura® Communication Manager, Avaya Aura® System Manager and Avaya Aura® Session Manager. All functionality and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager – Release 6.0, Issue 6.0, June 2010*

[2] *Administering Avaya Aura® Session Manager – Release 6.1, Issue 1, November 2010*

Netcall Liberty Documentation can be found at http://www.netcall.com

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.