Avaya Solution & Interoperability Test Lab

Application Notes for Sangoma IMG 2020 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Sangoma IMG 2020 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Sangoma IMG 2020 is an integrated media gateway that enables interworking between IP and PSTN networks. In the compliance test, Sangoma IMG 2020 delivered SIP signaling and media from an Avaya SIP-based enterprise network to the PSTN via ISDN-PRI. Sangoma IMG 2020 connected to Avaya Aura® Session Manager via a SIP trunk and to the PSTN via an ISDN-PRI trunk.

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 integrate the Sangoma IMG 2020 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Sangoma IMG 2020 is an integrated media gateway that enables interworking between IP and PSTN networks. In the compliance test, Sangoma IMG 2020 delivered SIP signaling and media from an Avaya SIP-based enterprise network to the PSTN via ISDN-PRI. Sangoma IMG 2020 connected to Avaya Aura® Session Manager via a SIP trunk and to the PSTN via an ISDN-PRI trunk.

Note: These Application Notes will focus on the configuration of the SIP trunk between Sangoma IMG 2020 and Avaya Aura® Session Manager and call routing. The configuration of the ISDN-PRI trunk to the PSTN is outside the scope of these Application Notes and will not be covered.

2 General Test Approach and Test Results

Interoperability compliance testing covered feature and serviceability testing. The feature testing focused on establishing calls between an Avaya SIP-based enterprise network and the PSTN, where all calls traversed Sangoma IMG 2020. Various telephony features, such as hold/resume, call transfers, conference calls, call forwarding, and calling number display/block were exercised.

The serviceability testing focused on verifying that IMG 2020 came back into service after re-establishing IP network connectivity and after a reboot.

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

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

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

For the testing associated with this Application Note, the interface between Avaya systems and Sangoma IMG 2020 did not include use of any specific encryption features as requested by Sangoma.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Establishing SIP trunk between IMG 2020 and Session Manager and verifying the exchange of SIP Options messages.
- Establishing voice calls between an Avaya SIP-based enterprise network and the PSTN with all calls traversing IMG 2020. SIP and H.323 endpoints on the Avaya enterprise network were used.
- Support of direct IP-to-IP media (also known as "Shuffling" which allows IP endpoints to send audio RTP packets directly to each other without using media resources on the Avaya Media Gateway or Avaya Aura® Media Server).
- Telephony features such as hold/resume, call transfers, conference calls, call forwarding, calling number display/block, and voicemail coverage.
- Establishing T.38 fax calls between an Avaya SIP-based enterprise network and the PSTN with fax calls traversing IMG 2020.
- Support of G.711 mu-law and G.729 codecs.
- DTMF support using RFC2833.
- Proper system recovery after reconnecting IMG2020 to the IP network and after a reboot.

2.2 Test Results

All test cases passed with the following observations:

- When Sangoma IMG 2020 receives a fax call, it's supposed to detect the fax tone in order to switch the call from G.711 to T.38 fax. During the compliance test, IMG 2020 was not properly detecting the fax tone from the computer fax software being used; and therefore, didn't classify the call as a fax call preventing it from sending the re-Invite with T.38 fax. The call was being detected as a data call and not a fax call. The workaround is to disable **Modem Behavior** as shown in **Section 7.5.2** so that IMG 2020 would assume the incoming call is a fax call and switch from G.711 to T.38 fax. If the customer is using modems in their network, then this field should be enabled and fax calls would have to use G.711 since the re-Invite with T.38 fax would not be sent to Communication Manager.

- For incoming PSTN calls, with the calling party number included, an Adaptation was required to replace the domain in the P-Asserted-Identity header on the egress side only; otherwise, the call would not complete. The domain had to match the configured domain in the SIP signaling group on Communication Manager. Leaving the domain in the SIP signaling group blank would allow all domains to match so this would be another solution. Refer to **Section 6.2**.

2.3 Support

For technical support on Sangoma IMG2020, contact Sangoma Customer Support via the web at https://www.sangoma.com/support/customer-support.

3 Reference Configuration

The network diagram below illustrates the test configuration. In this configuration, Sangoma IMG 2020 delivered SIP signaling and media from an Avaya SIP-based enterprise network to the PSTN via ISDN-PRI. Sangoma IMG 2020 connected to Avaya Aura® Session Manager via a SIP trunk and to the PSTN via an ISDN-PRI trunk. Voice calls were established with Avaya H.323 / SIP Deskphones and T.38 fax calls were established using computer fax software.

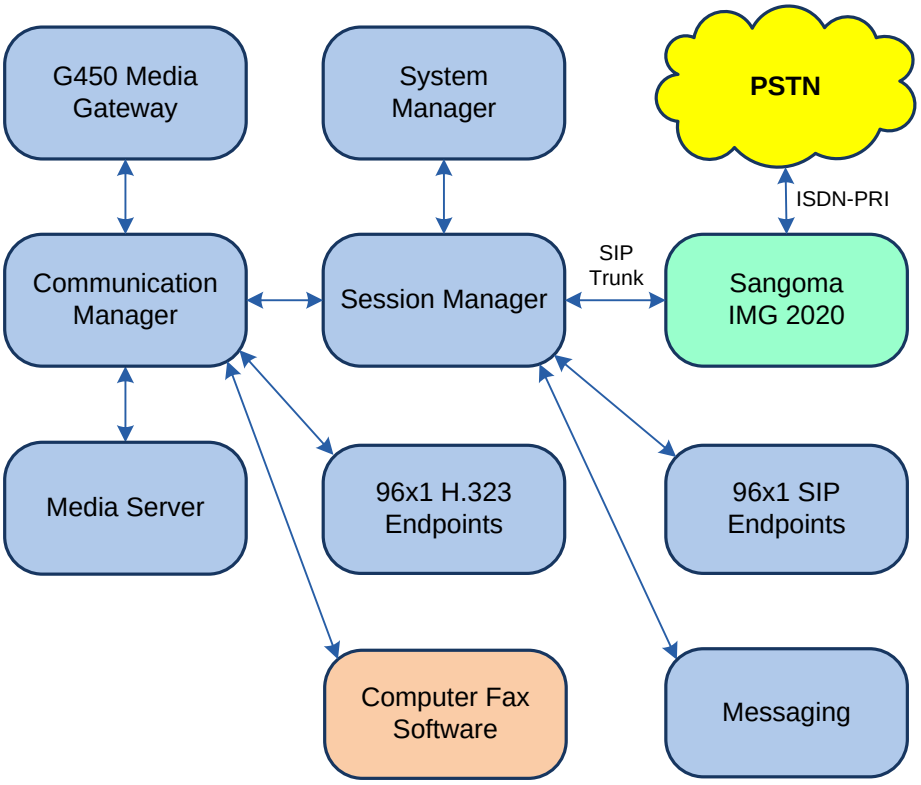


Figure 1: Avaya SIP-based Enterprise Network with Sangoma IMG 2020

4 Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager	8.0.1.1.0-FP1SP1 (R018x.00.0.822.0 with Patch 25183)
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.0.173
Avaya Aura® System Manager	8.0.1.1 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.1.1.039340 Service Pack 1
Avaya Aura® Session Manager	8.0.1.1.801103
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.8003 (H.323) 7.1.5.0.11 (SIP)
Sangoma IMG 2020	2.3.1

5 Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Trunk Group to Session Manager
- Administer Private Numbering Format
- Administer Incoming Call Handling Treatment
- Administer ARS Call Routing for PSTN Calls

  Note: AAR call routing to Avaya SIP endpoints registered to Session Manager is not covered in these Application Notes.

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

5.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                          Page   1 of   2
                                IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.2 Administer IP Codec Set

In the **IP Codec Set** form, specify the audio codec types supported for calls routed over the SIP trunk to IMG 2020. The form is accessed via the **change ip-codec-set 1** command. The default settings of the **IP Codec Set** form are shown below. IMG 2020 supports the G.711 and G.729 codecs.

```
change ip-codec-set 1                                          Page   1 of   2

                        IP Codec Set
   Codec Set: 1

   Audio         Silence      Frames   Packet
   Codec         Suppression  Per Pkt  Size(ms)
1: G.711MU            n          2        20
2: G.729              n          2        20
3:
```

5.3 Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                     Page   1 of  20
                            IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                        Stub Network Region: n
MEDIA PARAMETERS                 Intra-region IP-IP Direct Audio: yes
      Codec Set: 1               Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                        IP Audio Hairpinning? n
   UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4 Administer SIP Trunk to Session Manager

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

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify the Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- Enable **Initial IP-IP Direct Media**.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                        Page   1 of   2
                              SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
       Q-SIP? n
    IP Video? n                                  Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                      Clustered? n
 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: devcon-sm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                      Far-end Network Region: 1

Far-end Domain: avaya.com
                                           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? y                  Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Avaya SIP endpoints and for PSTN calls over the IMG 2020. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                          Page   1 of  22
                                TRUNK GROUP

Group Number: 10                    Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm                 COR: 1        TN: 1        TAC: 1010
   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: 10
                                                    Number of Members: 10
```

On **Page 2**, the **Preferred Minimum Session Refresh Interval (sec)** field was set to a value equal to or greater than the **Session Timer** set on IMG 2020 in **Section 7.5.2**. The **Session Timer** was set to 600 seconds.

```
add trunk-group 10                                          Page   2 of   5
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y


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

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

On **Page 3,** set the **Numbering Format**. For this test, the **private** numbering table was used to set the calling party number format for outgoing calls. In this case, 5-digit extensions beginning with '7' were converted to 732-777-xxxx as shown in **Section 5.5**.

```
add trunk-group 10                                          Page   3 of   5

TRUNK FEATURES

          ACA Assignment? n            Measured: none

                                                          Maintenance Tests? y

   Suppress # Outpulsing? n  Numbering Format: private

                                              UUI Treatment: shared

                                           Maximum Size of UUI Contents: 128

                                              Replace Restricted Numbers? n

                                             Replace Unavailable Numbers? n

                                               Hold/Unhold Notifications? y

                                Modify Tandem Calling Number: tandem-cpn-form

               Send UCID? y

 Show ANSWERED BY on Display? y
```

5.5 Administer Private Numbering Format

In the **Numbering – Private Format** form, add entry to prepend *73277* to calls from extensions beginning with '7' and routed over SIP trunk group 10. This will convert the calling number from a 5-digit extension to a 10-digit number in the format 732-777-xxxx. The calling number could then be displayed by the called party, if supported.

```
change private-numbering 0                                    Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext               Trk        Private          Total
Len Code              Grp(s)     Prefix           Len
 5  7                 10         73277            10 Total Administered: 1
                                                       Maximum Entries: 540
```

5.6 Administer Incoming Call Handling Treatment

The DID numbers for PSTN calls to the Avaya enterprise network were in the format of 732-777-xxxx, where the last 5 digits mapped to the extension. The calls were received by Communication Manager on SIP trunk group 10. The **Incoming Call Handling Treatment** table for trunk group 10 was modified to delete the first 5 digits, as shown below, so that the calls would terminate on the proper extension.

```
change inc-call-handling-trmt trunk-group 10                  Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             10 73277            5
```

5.7 Administer ARS Call Routing for PSTN Calls

PSTN calls are routed to Session Manager over a SIP trunk via ARS call routing. Add an entry to the ARS analysis form to route calls in the 908 or 732 area codes to route pattern 10 as shown below. The ARS feature access code of '9' steers calls to ARS routing.

```
change ars analysis 7                                        Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                  Location: all            Percent Full: 1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    732                   10   10    10       hnpa         n
    908                   10   10    10       hnpa         n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 to Session Manager as shown below.

```
change route-pattern 10                                      Page   1 of   3
                  Pattern Number: 10     Pattern Name: To devcon-sm
    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: 10   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                               unk-unk   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 Configure Avaya Aura® Session Manager

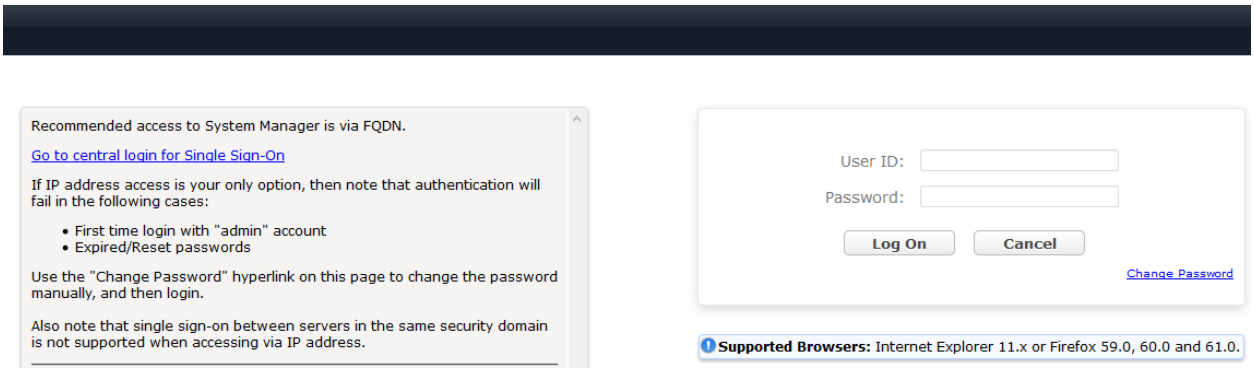
This section provides the procedure for configuring Session Manager. The procedure includes adding the following items:

- Launch System Manager
- Adaptations to modify SIP messages for egress only
- SIP Entities corresponding to Session Manager, Communication Manager, and IMG 2020
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies
- Dial Patterns
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager

Note: It is assumed that basic configuration of Session Manager has already been completed. This section will focus on the configuration of the SIP trunk to IMG 2020 and routing calls to it.

6.1 Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

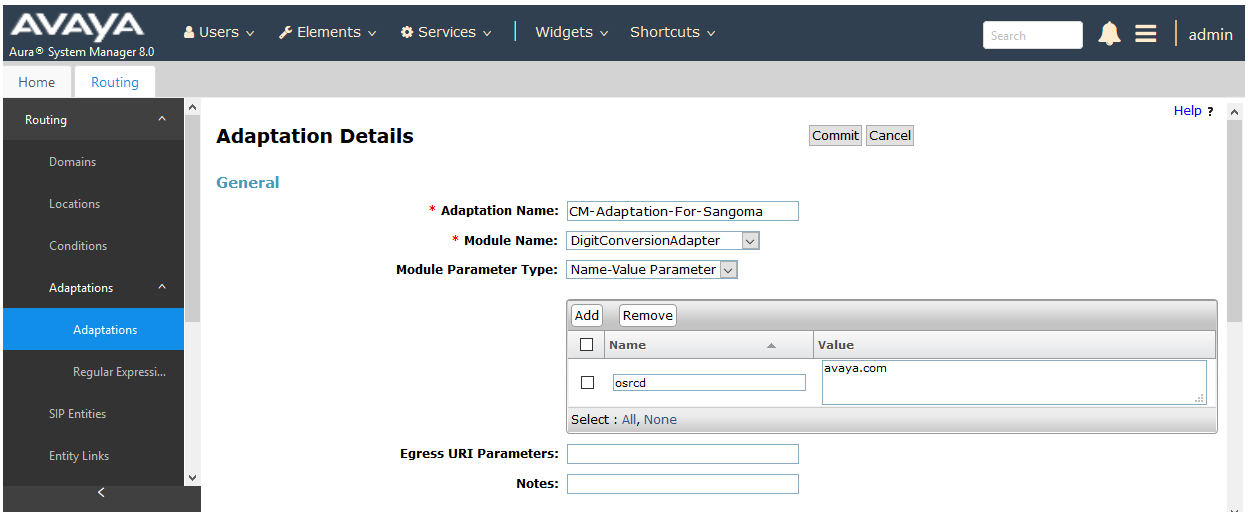


6.2 Add Adaptation

Session Manager can be configured with Adaptations that can modify SIP messages before or after routing decision have been made; for example, changing the domain in the P-Asserted-Idenity in a SIP INVITE message. To create an **Adaptation** that will be applied to the Communication Manager SIP entity in **Section 6.3.2**, navigate to **Elements → Routing → Adaptations** and click on the **New** button (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields.

- **Adaptation Name:** Enter a descriptive name for the Adaptation (e.g., *CM-Adaptation-For-Sangoma*).
- **Module Name:** Select **DigitConversionAdapter**.
- **Module Parameter Type:** Select **Name-Value Parameter**. The section will expand with an area to enter **Name** and **Value** pairs. Click **Add**. To replace the domain in the P-Asserted-Identity header on the egress side of Session Manager (i.e., towards Communication Manager) enter the keyword **osrcd** in the **Name** field and *avaya.com* in the **Value** field. This will replace the IP address used in the domain with *avaya.com*, the domain configured in the SIP signaling group in **Section 5.4**.



6.3 Add SIP Entities

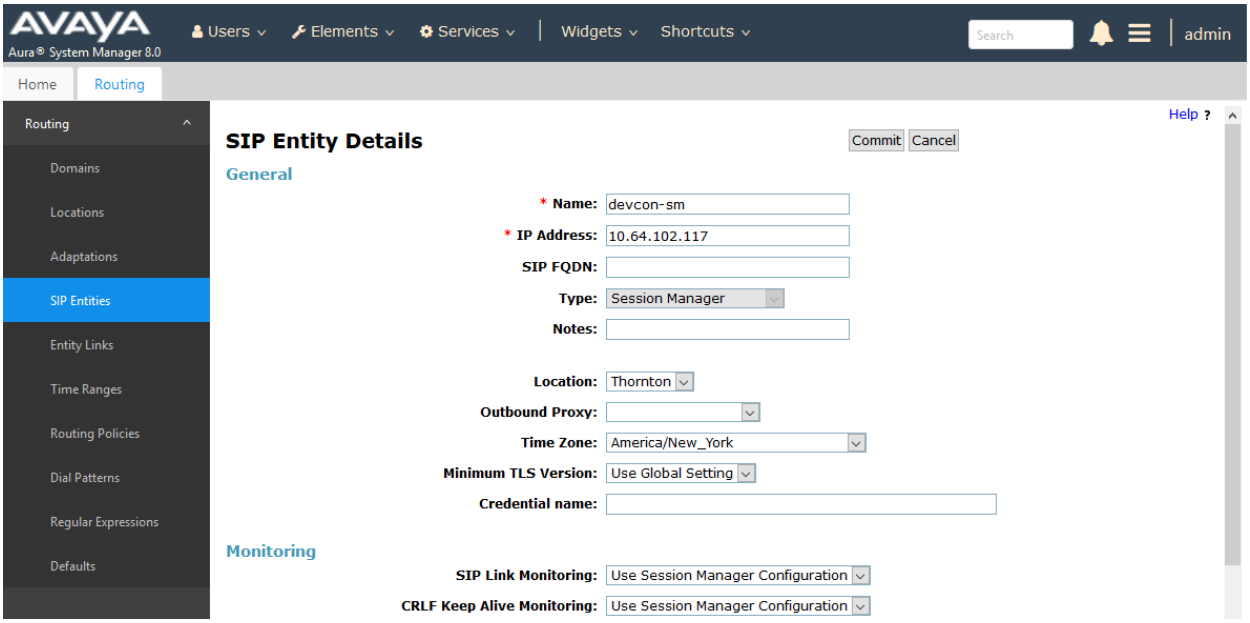
This section covers the configuration of SIP entities for Session Manager, Communication Manager, and IMG 2020.

6.3.1 Avaya Aura® Session Manager

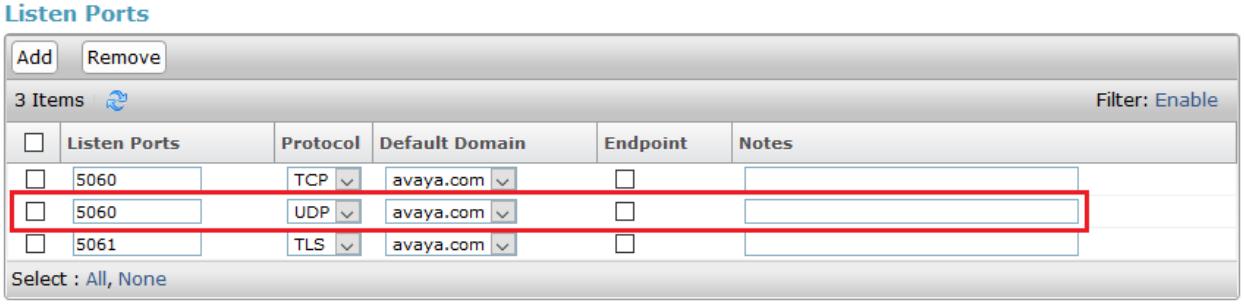
From the System Manager **Home** screen, navigate to **Elements → Routing → SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined.
- **Time Zone:** Time zone for this location.



Scroll down to the **Listen Ports** section and verify that the UDP transport network protocol used by IMG 2020 is specified as shown below.



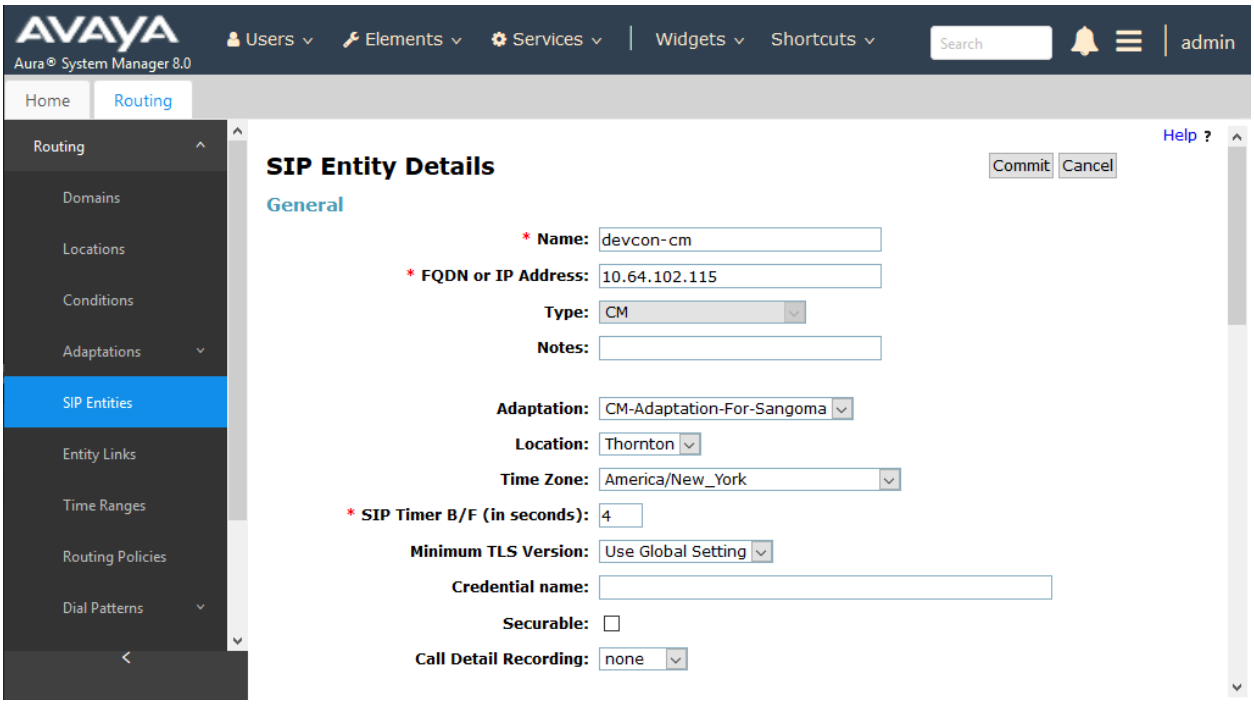
6.3.2 Avaya Aura® Communication Manager

A SIP Entity must be added for the Communication Manager. From the System Manager **Home** screen, navigate to **Elements → Routing → SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Communication Manager (*procr*).
- **Type:** Select *CM*.
- **Adaptation:** Adaptation configured in **Section 6.2**.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



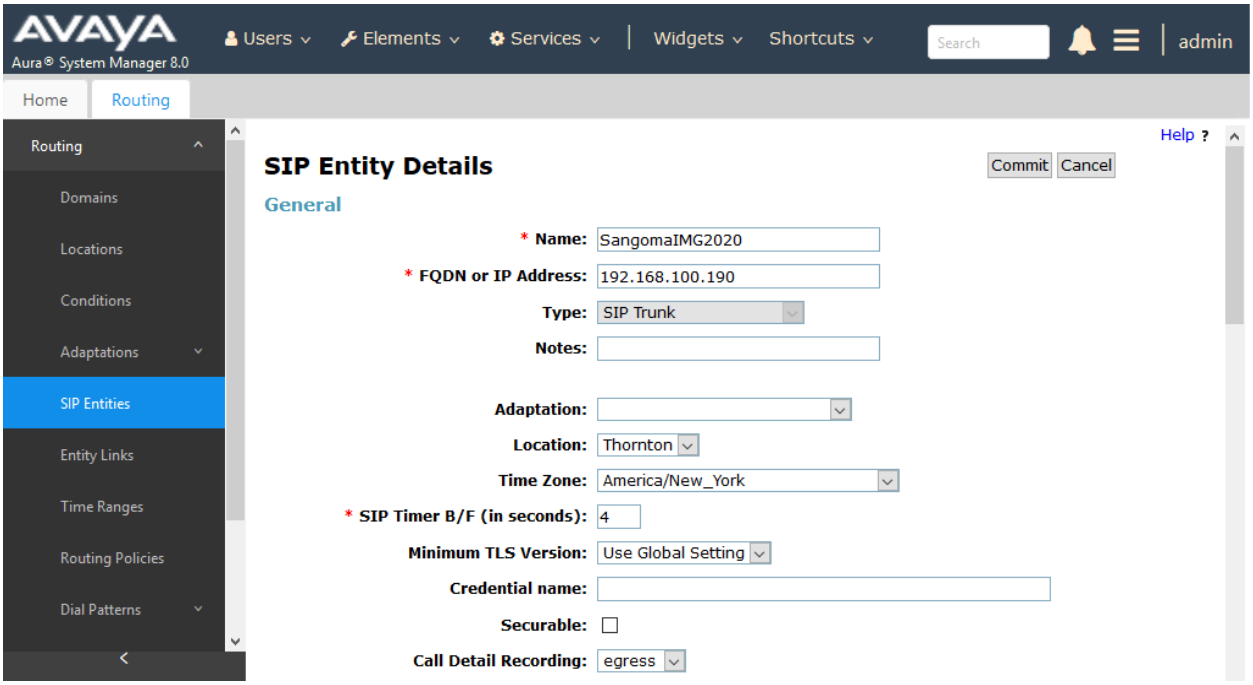
6.3.3 Sangoma IMG 2020

A SIP Entity must be added for IMG 2020. To add a SIP Entity, navigate to **Elements → Routing → SIP Entities** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IMG 2020 IP address.
- **Type:** Select *SIP Trunk*.
- **Location:** Select one of the locations previously defined.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



6.4 Add Entity Links

This section covers the configuration of Entity Links for Communication Manager and IMG 2020.

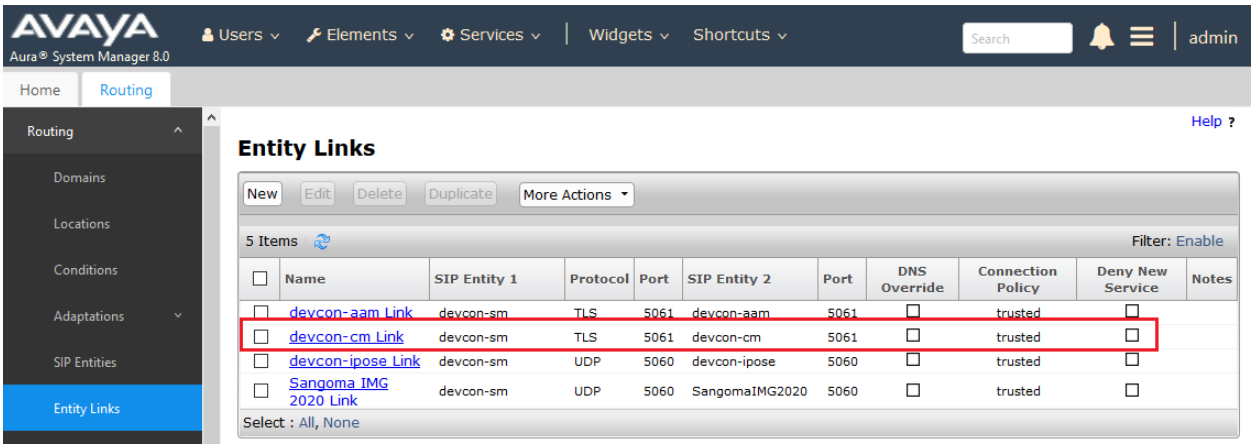
6.4.1 Communication Manager Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon-cm Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *trusted*. *Note: If trusted is not selected, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

Click **Commit** to save the Entity Link definition.

The following Entity Link is between Session Manager and Communication Manager.

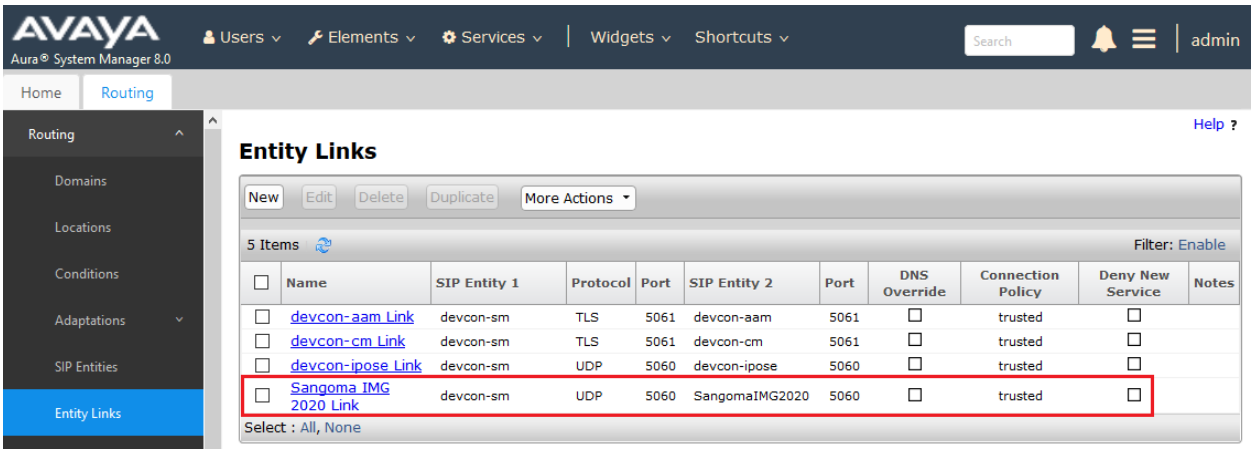


6.4.2 Sangoma IMG 2020 Entity Link

The SIP trunk between Session Manager and IMG 2020 is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Sangoma IMG 2020 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select UDP transport protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the *SangomaIMG2020* SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in* ***Section 6.3.3*** *will be denied.*

Click **Commit** to save the Entity Link definition.



6.5 Add Routing Policies

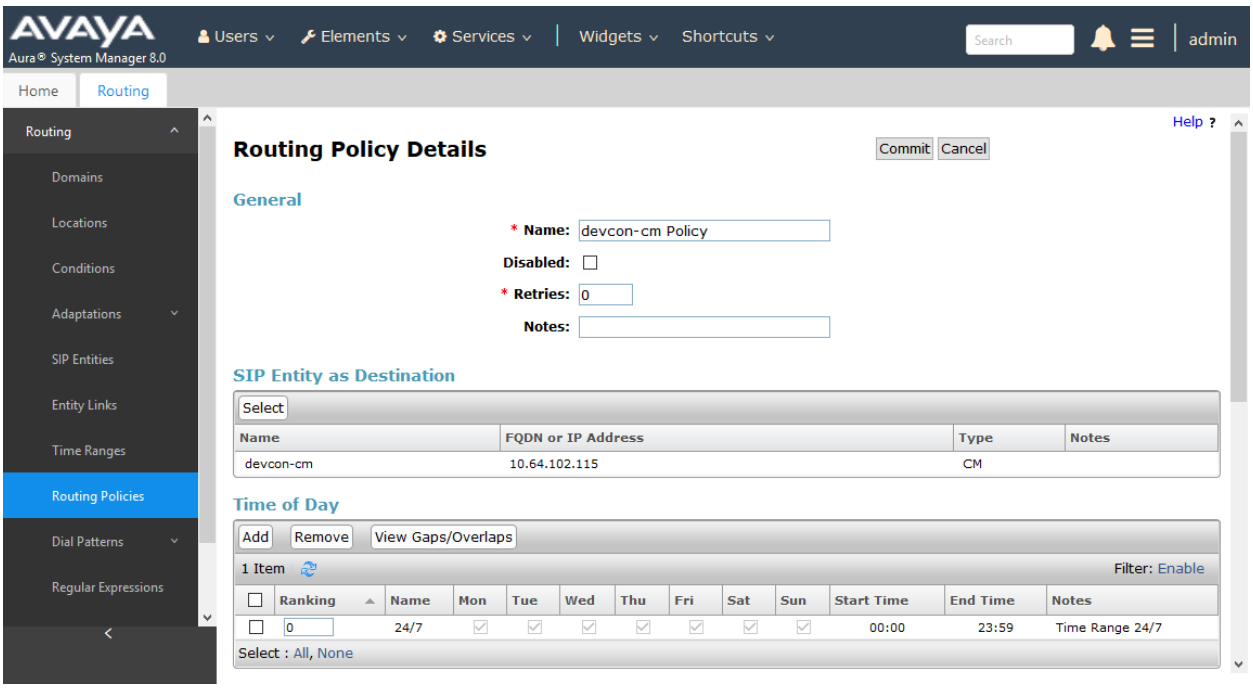
Routing policies describe the conditions under which calls are routed to Communication Manager and IMG 2020 SIP entities. To add a routing policy, navigate to **Elements → Routing → Routing Policies** and click on the **New** button (not shown). The following screen is displayed. Fill in the following:

Under *General*:
Enter a descriptive name in **Name**.

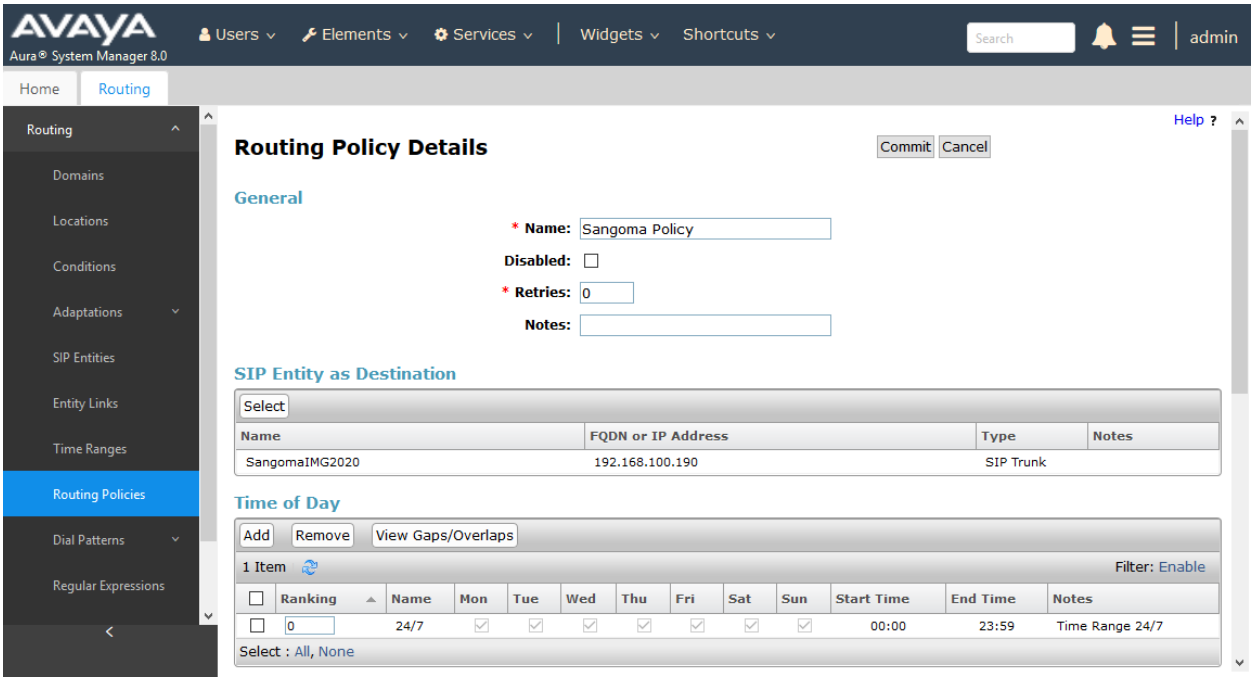
Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition.

The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for IMG 2020.



6.6 Add Dial Patterns

Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, 10-digit numbers starting with 73277 are routed to Communication Manager and calls to 732 or 908 area codes are routed to IMG 2020.

To add a dial pattern, navigate to **Elements → Routing → Dial Patterns** and click on the **New** button (not shown). Fill in the following:

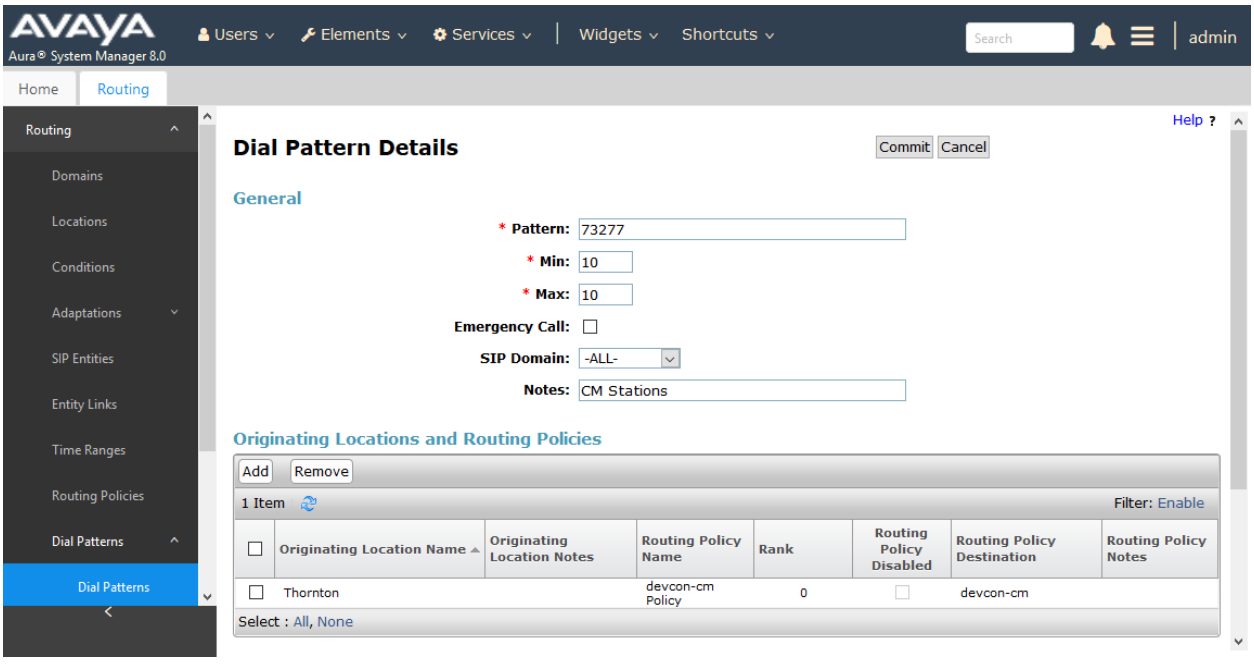
Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

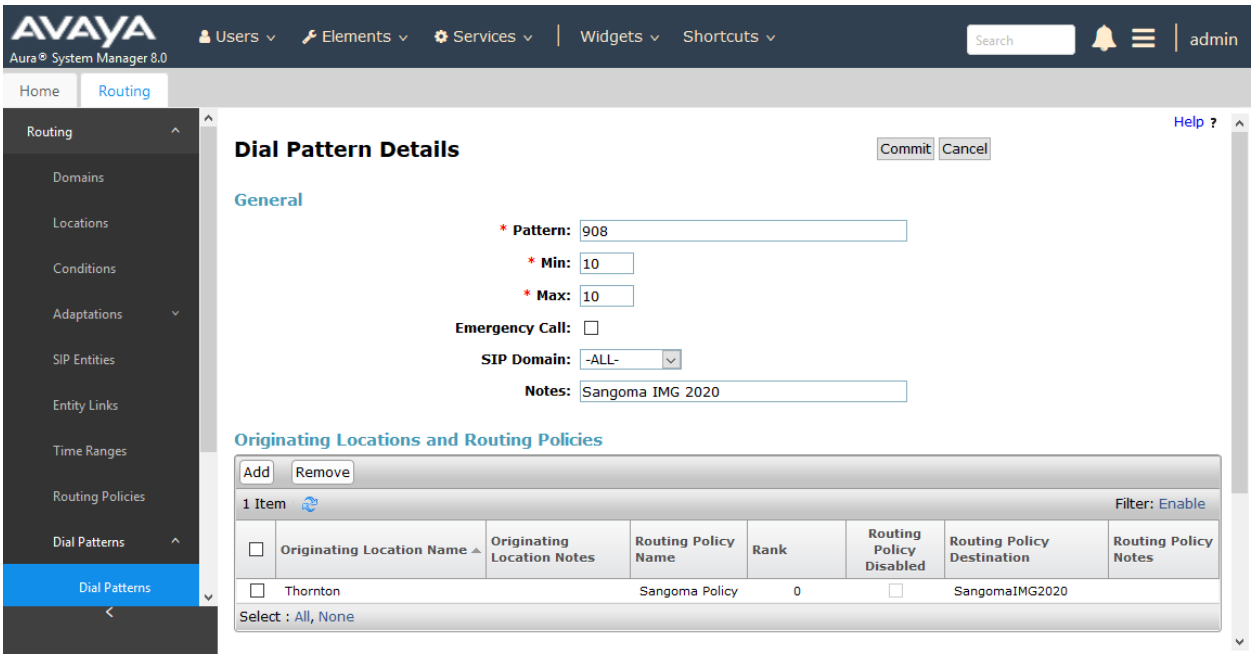
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern.

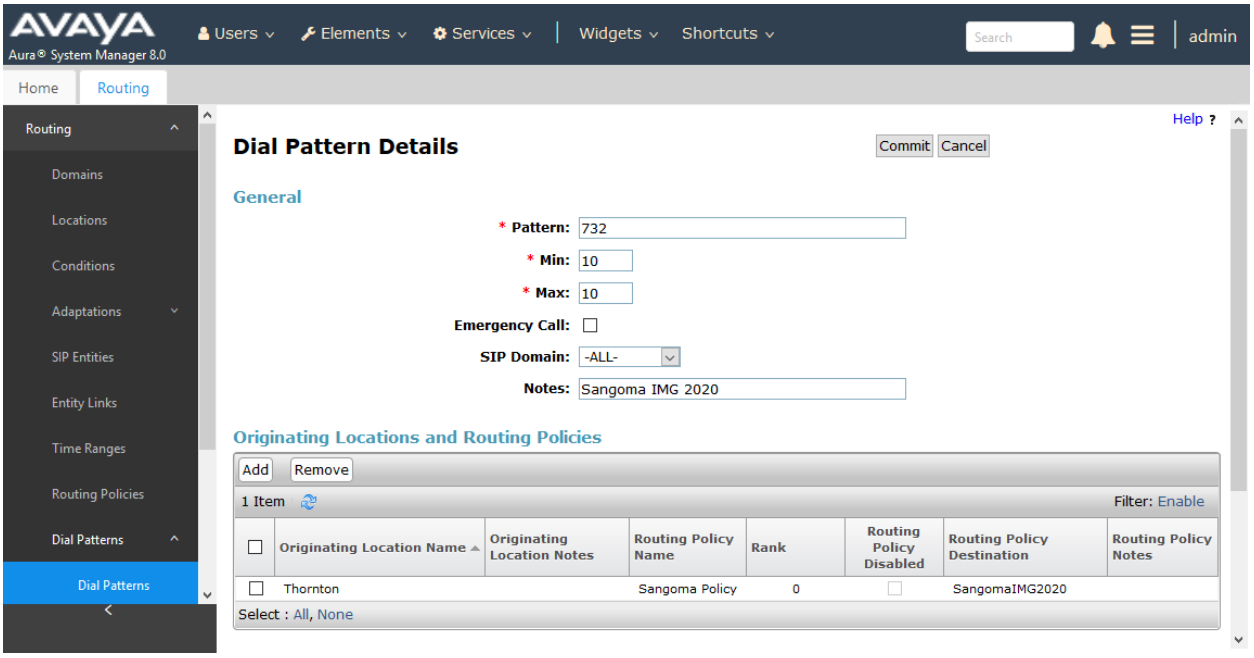
The following **Dial Pattern** shows the dial pattern definition for 73277 being routed to Communication Manager.



The following **Dial Pattern** shows the dial pattern definition for calls in the 908 area code being routed to IMG 2020.



The following **Dial Pattern** shows the dial pattern definition for calls in the 732 area code being routed to IMG 2020.



6.7 Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

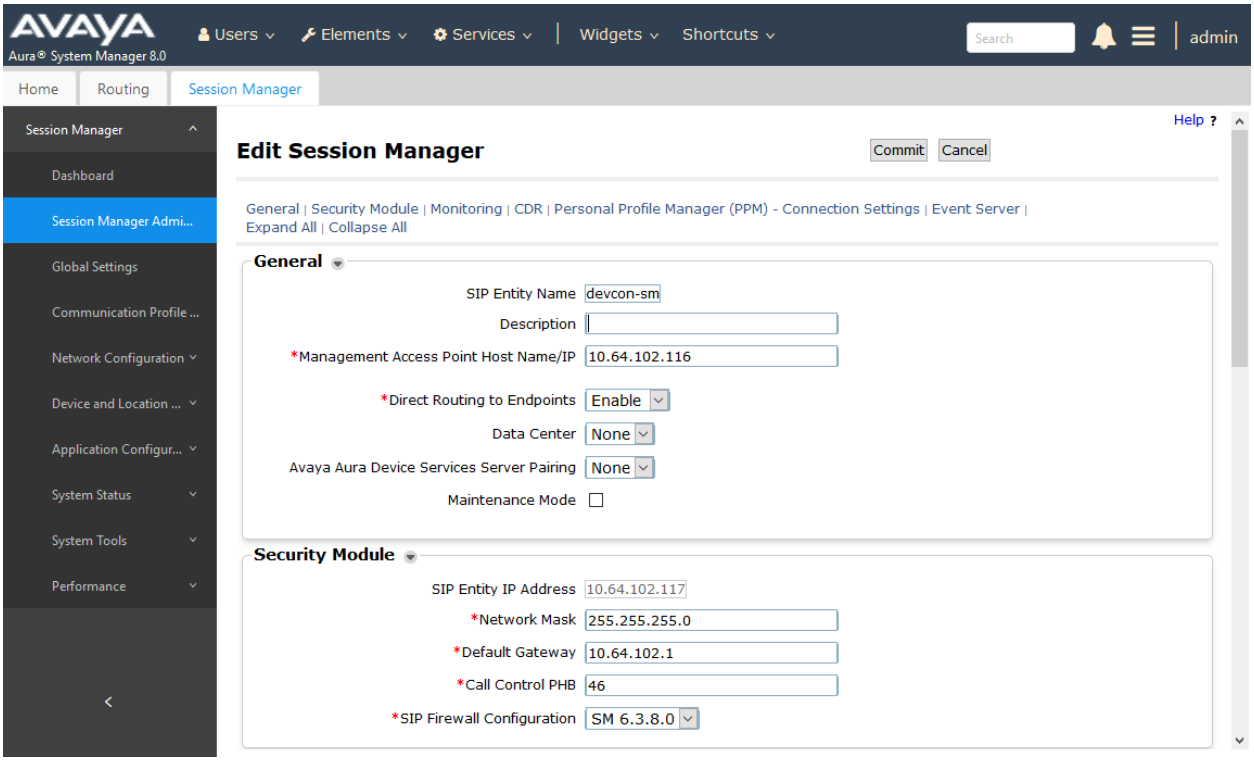
Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

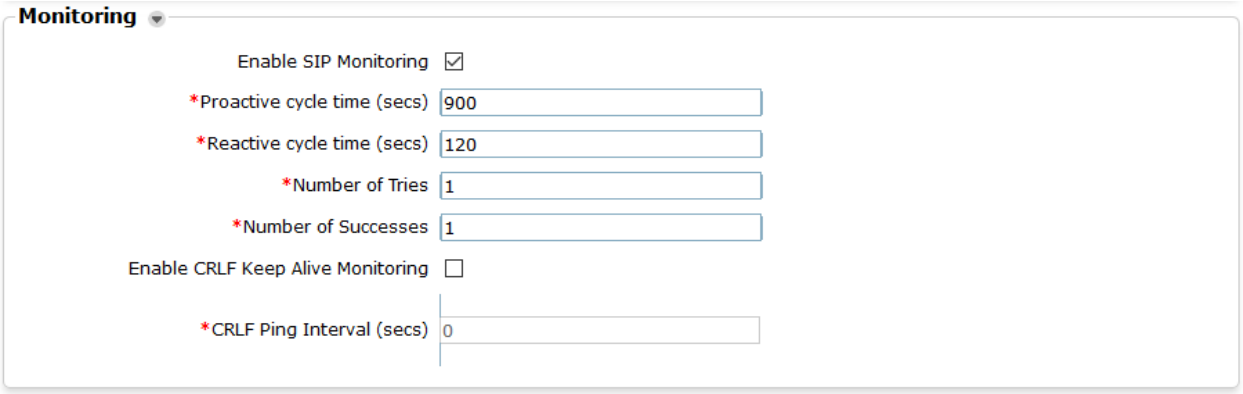
Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Commit** to add this Session Manager.



The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to IMG 2020. Use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every 900 secs. If there is no response, Session Manager will send a SIP Options message every 120 secs.



7 Configure Sangoma IMG 2020

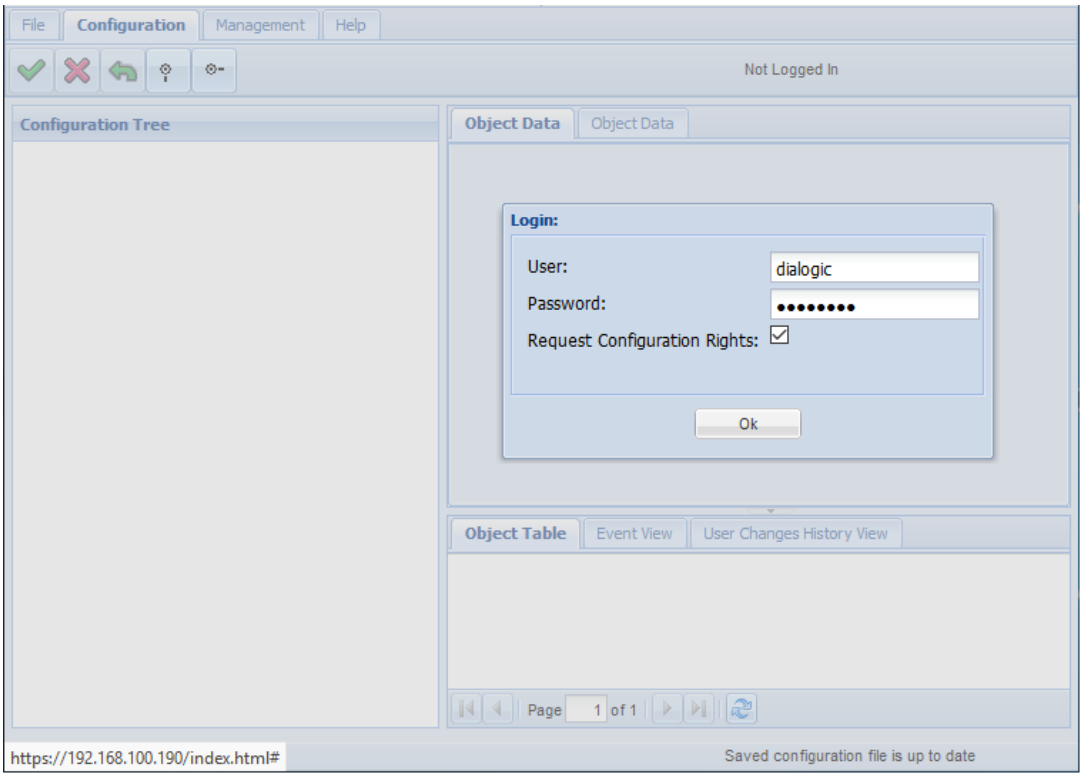
The IMG 2020 was configured with two trunk interfaces, a SIP trunk used to connect to Session Manager and an ISND-PRI trunk used to connect to the PSTN. This section will cover the configuration of the SIP trunk, media interfaces, and call routing. It is assumed that the ISDN-PRI trunk has already been completed. Refer to **[4]** for additional configuration details.

This section covers the following configuration areas:

- Launch Web GUI
- Create IMG 2020 Physical Node
- Create IP Interface for Media
- Create IP Interface for Signaling
- Configure Profiles
- Configure External Network Element
- Configure Routing
- Save Configuration

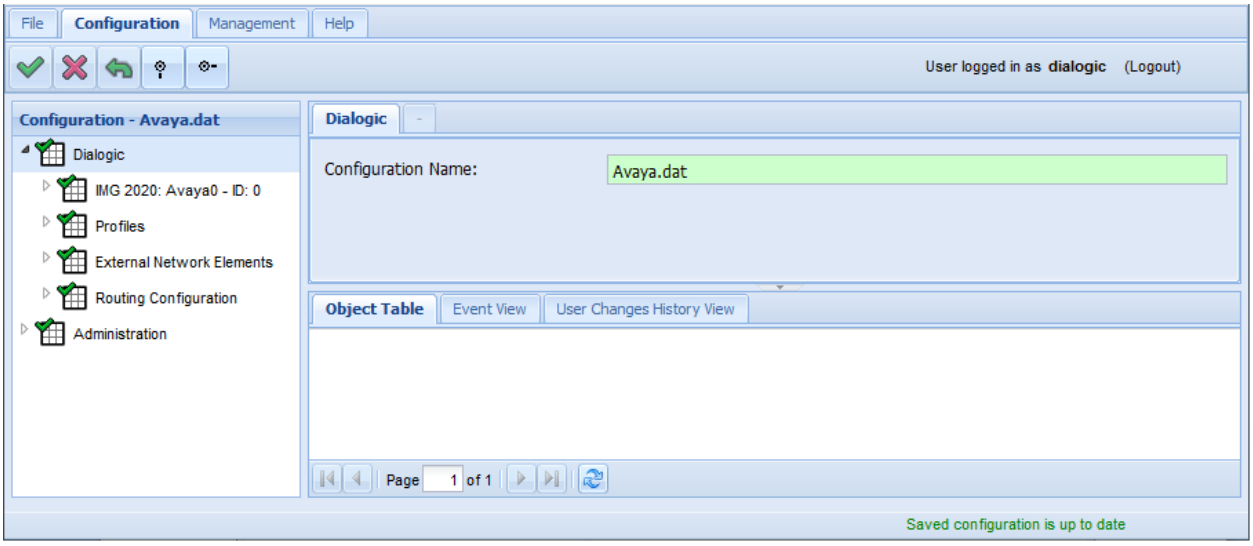
7.1 Launch Web GUI

The Web GUI is a real-time web based Graphical User Interface used to configure, monitor, and provision the IMG 2020. Access the Web GUI by using the URL "https://ip-address" in a supported web browser, where "ip-address" is the IMG 2020 IP address. Log in using the appropriate credentials when prompted as shown below.



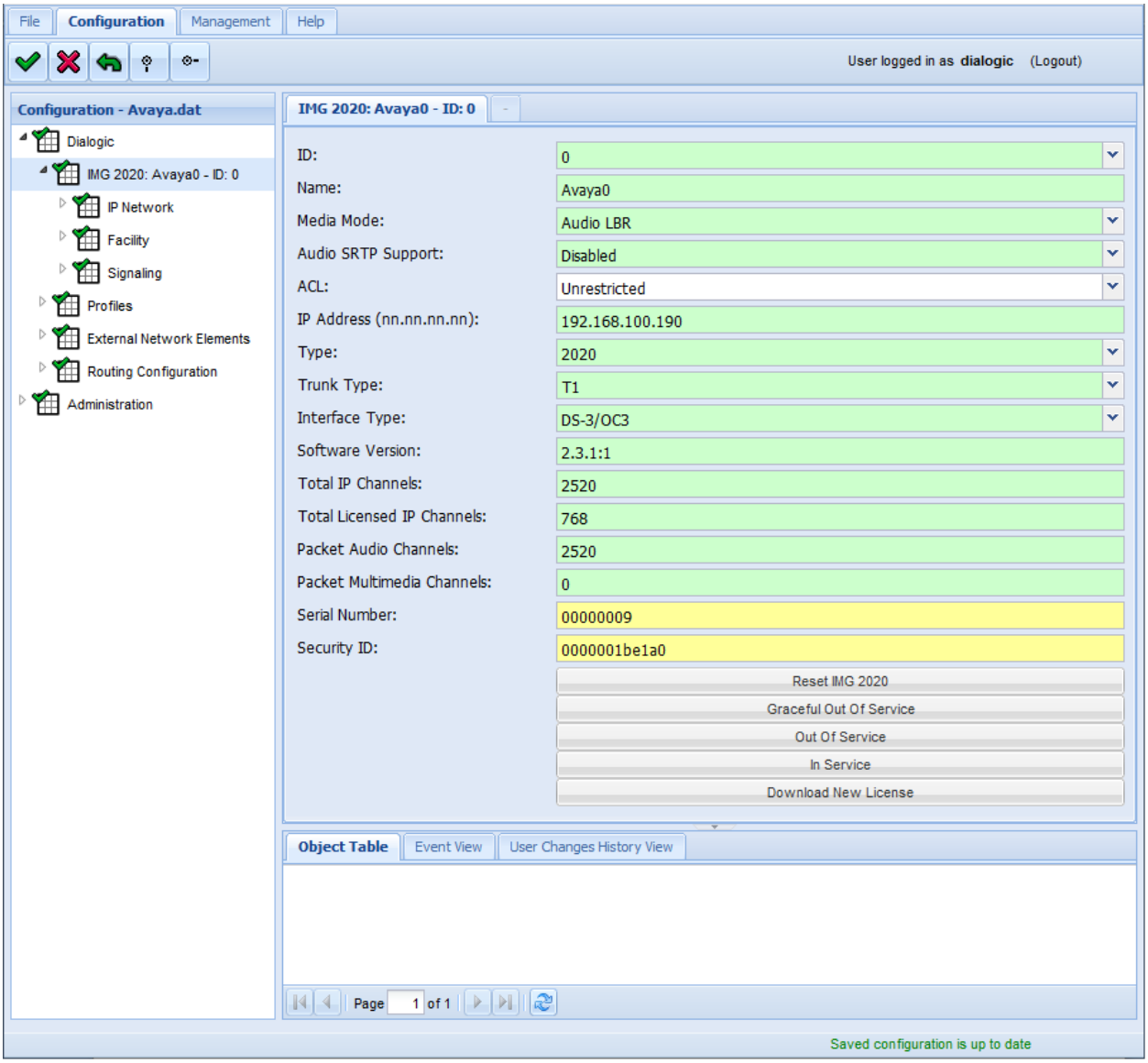
The following window is displayed from which basic functionality on the IMG 2020 is configured.

Note: After configuring an object, click on the commit check mark .



7.2 Create IMG 2020 Physical Node

Create the physical IMG 2020 node. Right-click on the **Dialogic** object in the configuration tree and select **New IMG 2020**. Specify a **Name** (e.g., *Avaya0*) to identify the node and enter the IMG 2020 IP address (e.g., *192.168.100.190*). Select the **Media Mode** from the drop-down menu (i.e., *Audio LBR*). Keep the default values for the remaining fields.

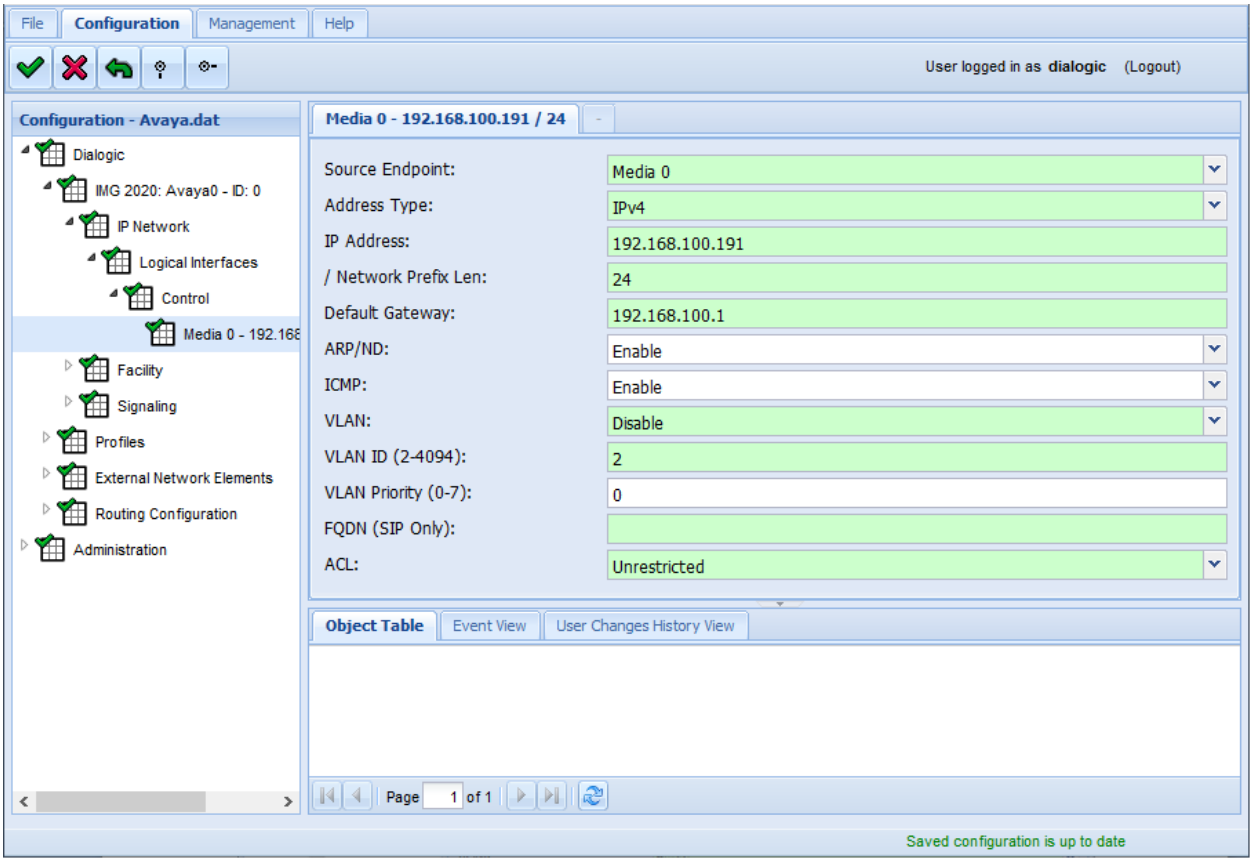


7.3 Create IP Interface for Media

To stream media (i.e., RTP data), an interface on the IMG 2020 must be configured and be given a Media 0 Endpoint IP address. Follow the steps below.

1. Right-click on the IMG 2020 Physical Node (i.e., **IMG 2020** in the configuration tree) and select **New Network**.
2. Right-click on the **IP Network** object and select **New Logical Interfaces**.
3. Right-click on the **Logical Interfaces → Control** object and select **New IP Interface**.

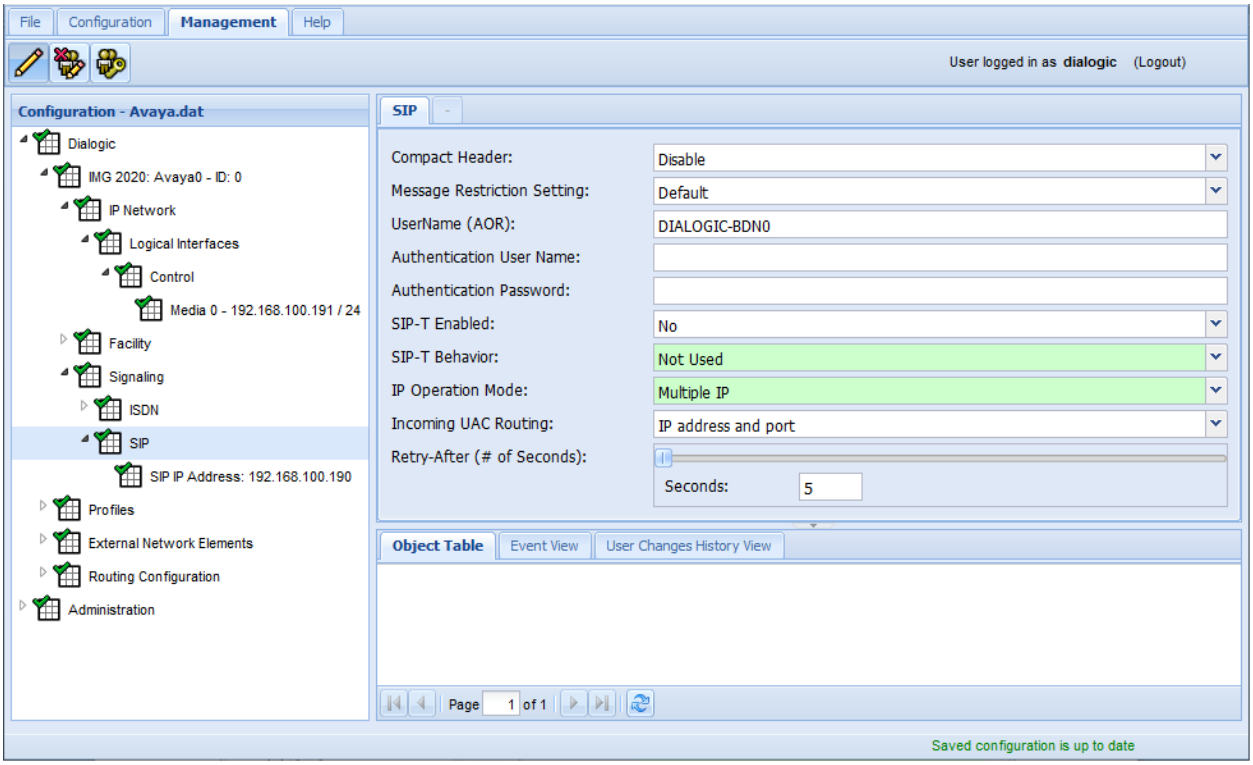
In the **IP Interface** screen, set the **Source Endpoint** to *Media 0*. In the **IP Address** field, enter an IP address that will be used for RTP data and enter the **Network Prefix Len** and **Default Gateway**.



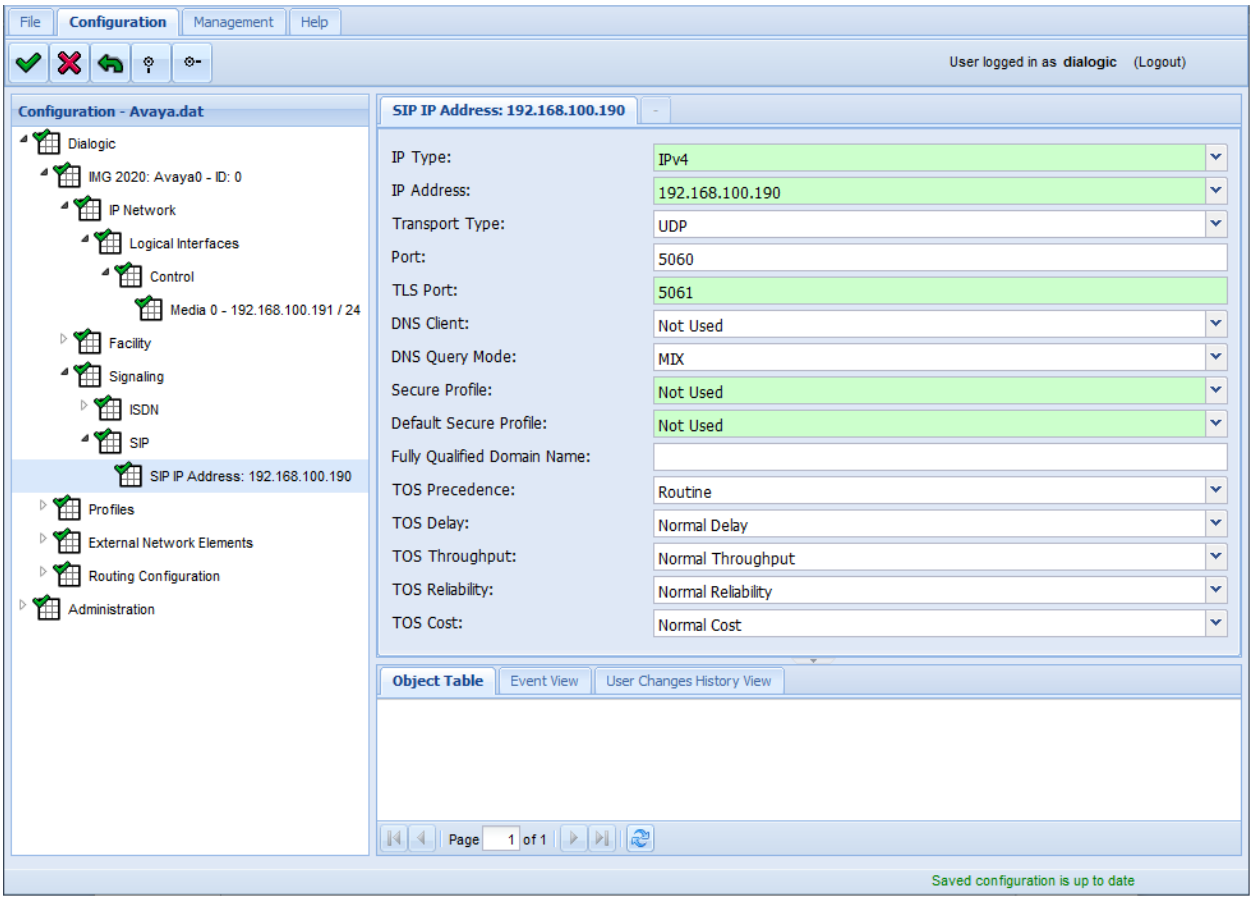
7.4 Create IP Interface for Signaling

A Signaling object needs to be configured to allow SIP signaling to be utilized on the IMG 2020. To configure the SIP signaling interface, right-click on the **IMG 2020** Physical Node and select **New Signaling**. Next, right-click on the **Signaling** object that was created and select **New SIP**. Select the **SIP** object in the configuration tree and configure the fields as shown below. Set the **IP Operation Mode** to *Multiple IP*.

Note: For the compliance test, there was only one IP address defined for SIP, but it is recommended to set the **IP Operation Mode** field to *Multiple IP* to allow another SIP address to be added in the future without having to perform a major re-configuration.



Right-click on the **SIP** object and select **New IP Address**. The **SIP IP Address** screen is displayed. Select the appropriate **IP Address** from the drop-down menu (e.g., *192.168.100.190*). Set the **Transport Type** to *UDP* and the **Port** to *5060*. Keep the default values for the remaining fields.



7.5 Configure Profiles

Some objects need to be associated with a specific profile for them to function in a network. The profiles are first created and then associated with an object. This section covers the configuration of the **IP Profiles** and **SIP Profiles**. The **IP Profile** will be associated with the **Routing** object created in **Section 7.7** and the **SIP Profile** will be associated with the **External Network Element** object created in **Section 7.6**.

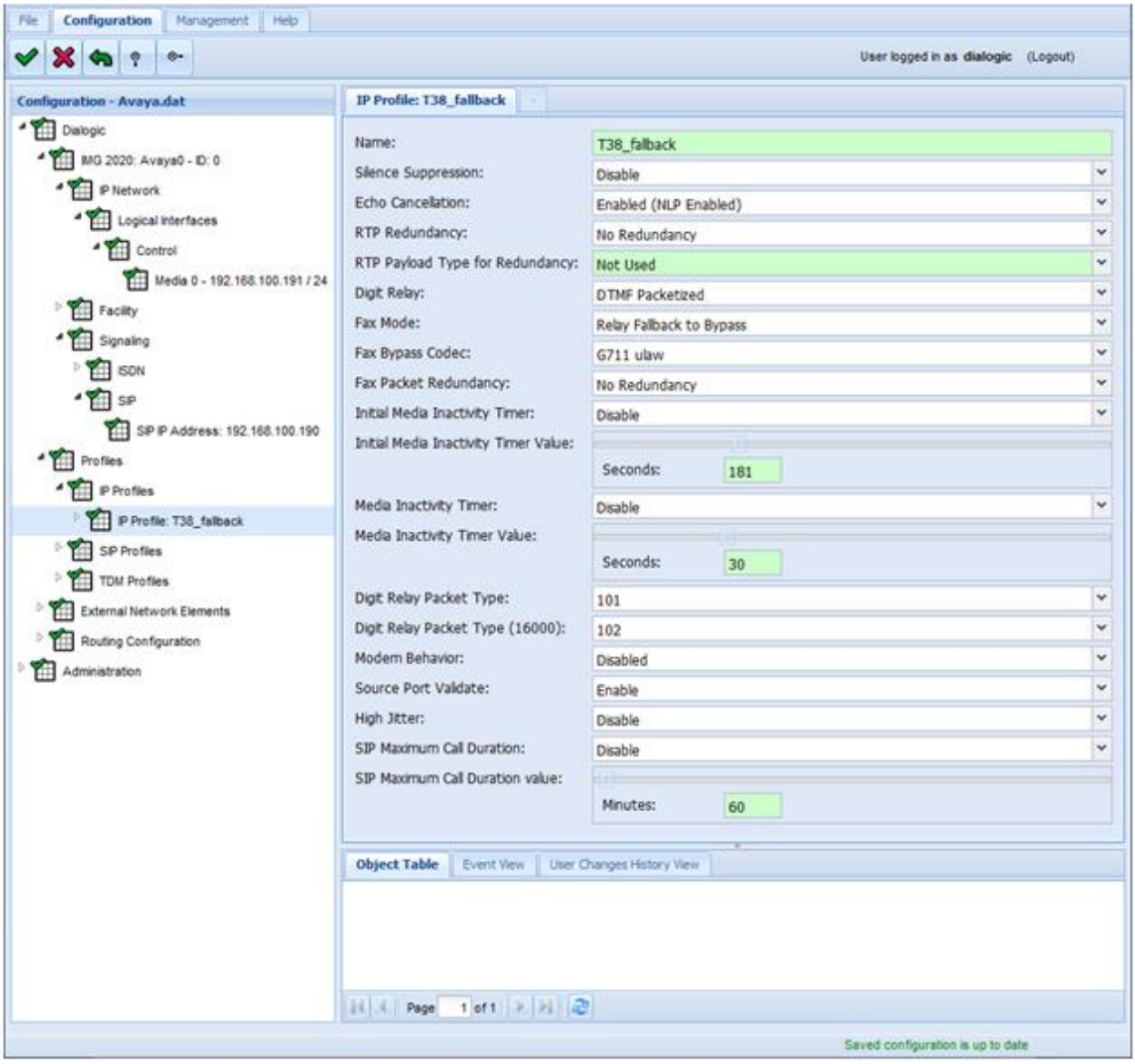
7.5.1 Configure IP Profile

The **IP Profile** will be configured for T.38 fax, DTMF using RFC2833, and allow G.711 and G.729 codecs along with the preference order. Follow the procedure below to create the **IP Profile**.

1. Right-click on the **Dialogic** object and select **New Profiles**.
2. Right-click on the **Profiles** object and select **New IP Profiles**.
3. Right-click on the **IP Profiles** object and select **New IP Profile**.

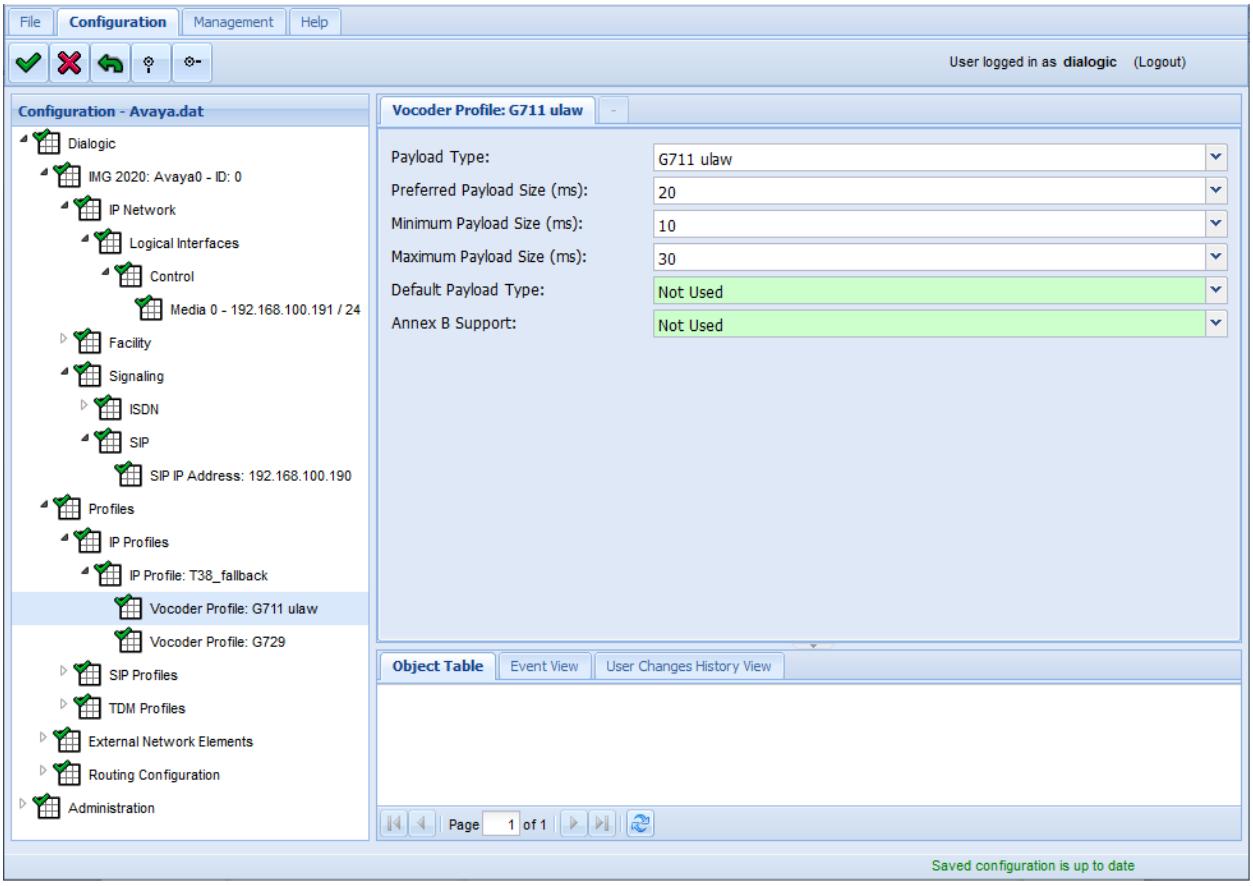
In the **IP Profile** screen, enter a descriptive in the **Name** field (e.g., *T38_fallback*). For **Digit Relay**, select *DTMF Packetized* to use RFC 2833. For **Fax Mode**, select *Relay Fallback to Bypass*. For **Fax Bypass Codec**, select *G711 ulaw*. Keep the default values for the remaining fields.

Note: For the compliance test, **Modem Behavior** was set to *Disabled* so that IMG 2020 would assume the incoming call is a fax call, as opposed to a data call, and switch from G.711 to T.38; otherwise, IMG 2020 would not properly detect the fax tone from the computer fax software used during the compliance test. If the customer is using modems in their configuration, then this field should be enabled and fax calls would use G.711 since the T.38 re-Invite from IMG 2020 would not be sent to Communication Manager.

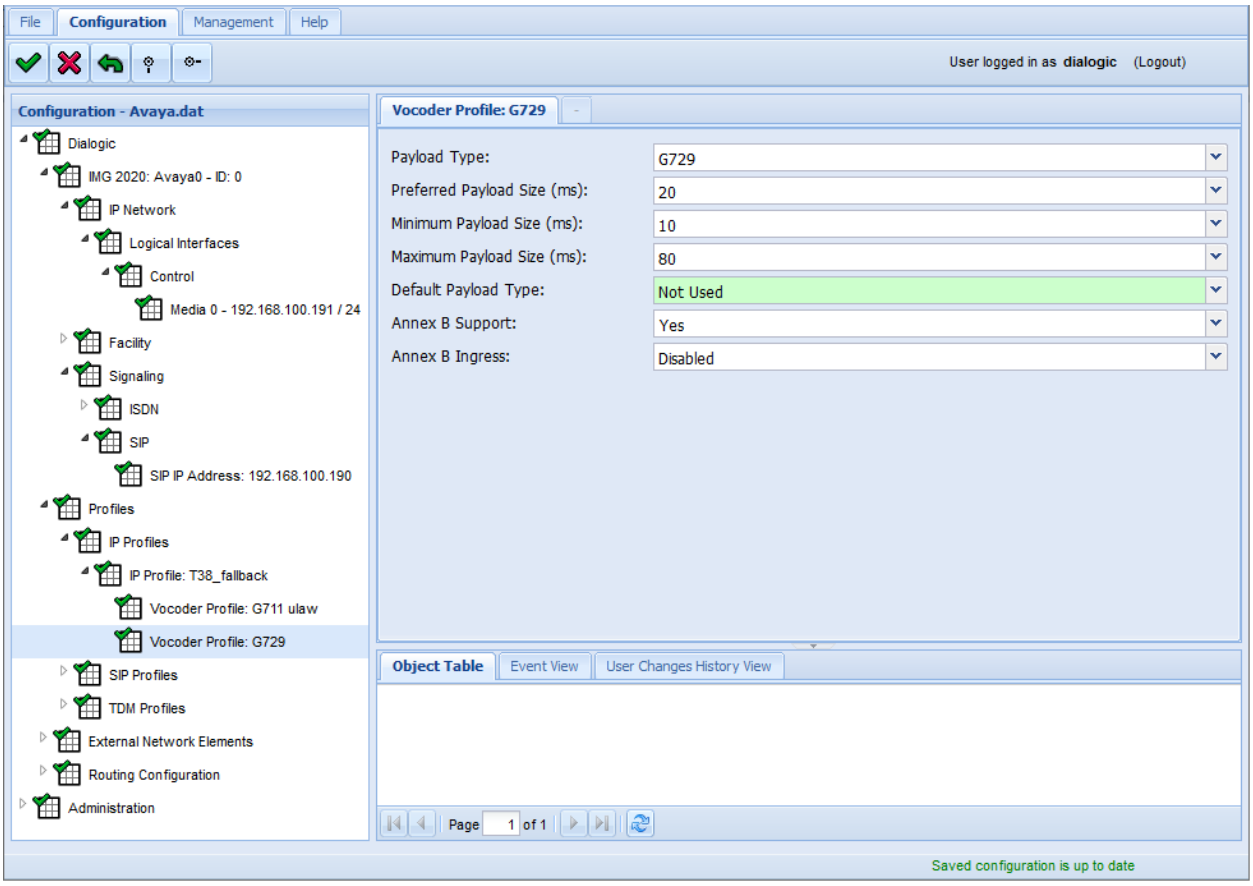


In the next step, the G.711 and G.729 codecs will be associated with the profile. Note that the order in which the codecs are listed indicate their preference order.

Right-click on the **IP Profile: T38_fallback** object (name may vary) and select **New Vocoder Profile** to associate the G.711 codec with the profile. In the **Vocoder Profile**, set the **Payload Type** to *G711 ulaw*.



Repeat the step above to associate the G.729 codec with the profile. Right-click on the **IP Profile: T38_fallback** object and select **New Vocoder Profile** to associate the G.729 codec with the profile. In the **Vocoder Profile**, set the **Payload Type** to *G729* and **Annex B Support** to *Yes*.

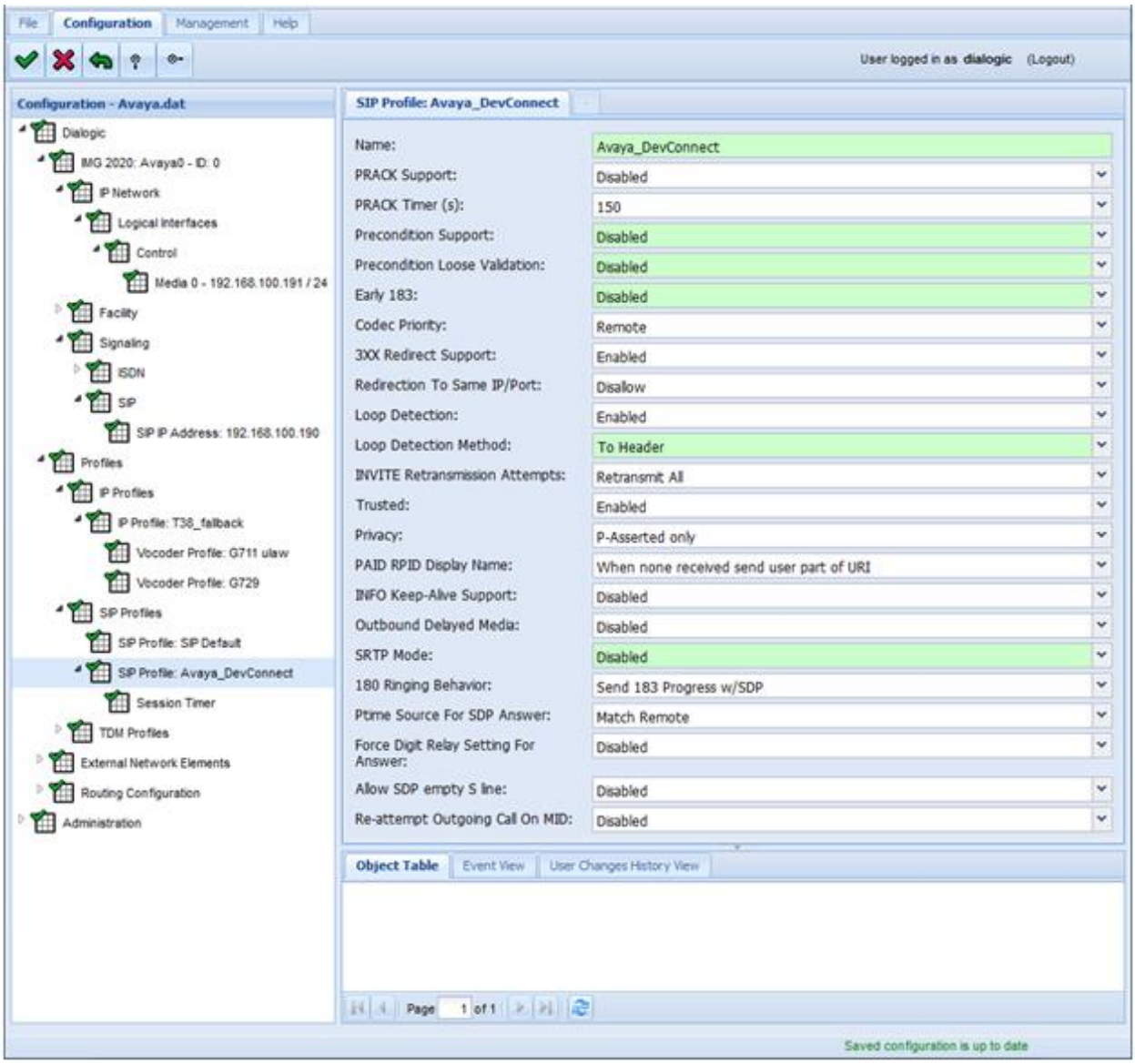


7.5.2 Configure SIP Profile

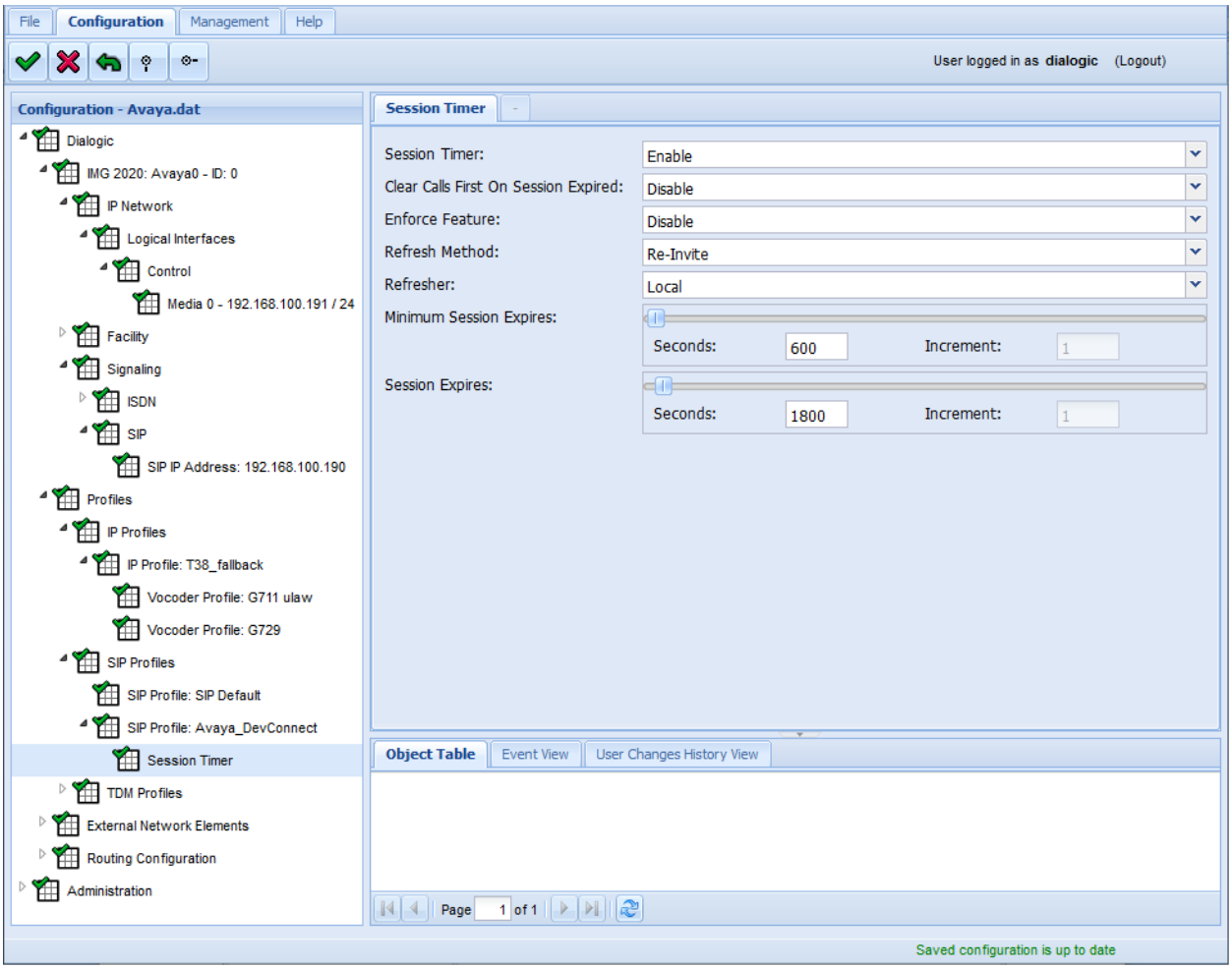
The **SIP Profile** configures the attributes and features needed to communicate with a specific external gateway (i.e., Session Manager and Communication Manager). To configure the **SIP Profile**, follow these steps.

1. Right-click on the **Profiles** object and select **New SIP Profiles**.
2. Right-click on the **SIP Profiles** object and select **New SIP Profile**.

In the **SIP Profile** screen, enter a descriptive name in the **Name** field (e.g., *Avaya_DevConnect*). For **Codec Priority**, select **Remote**. This gives the codec preference order of the remote side priority during codec negotiation. Keep the default values for the remaining fields.



Right-click on the **SIP Profile: Avaya_DevConnect** object (name may vary) and select **New SIP Session Timer**. In the **Session Timer** screen, set the **Minimum Session Expires** field to the appropriate value. For the compliance test, this timer was set to 600 seconds to match the default value in the SIP trunk group on Communication Manager.

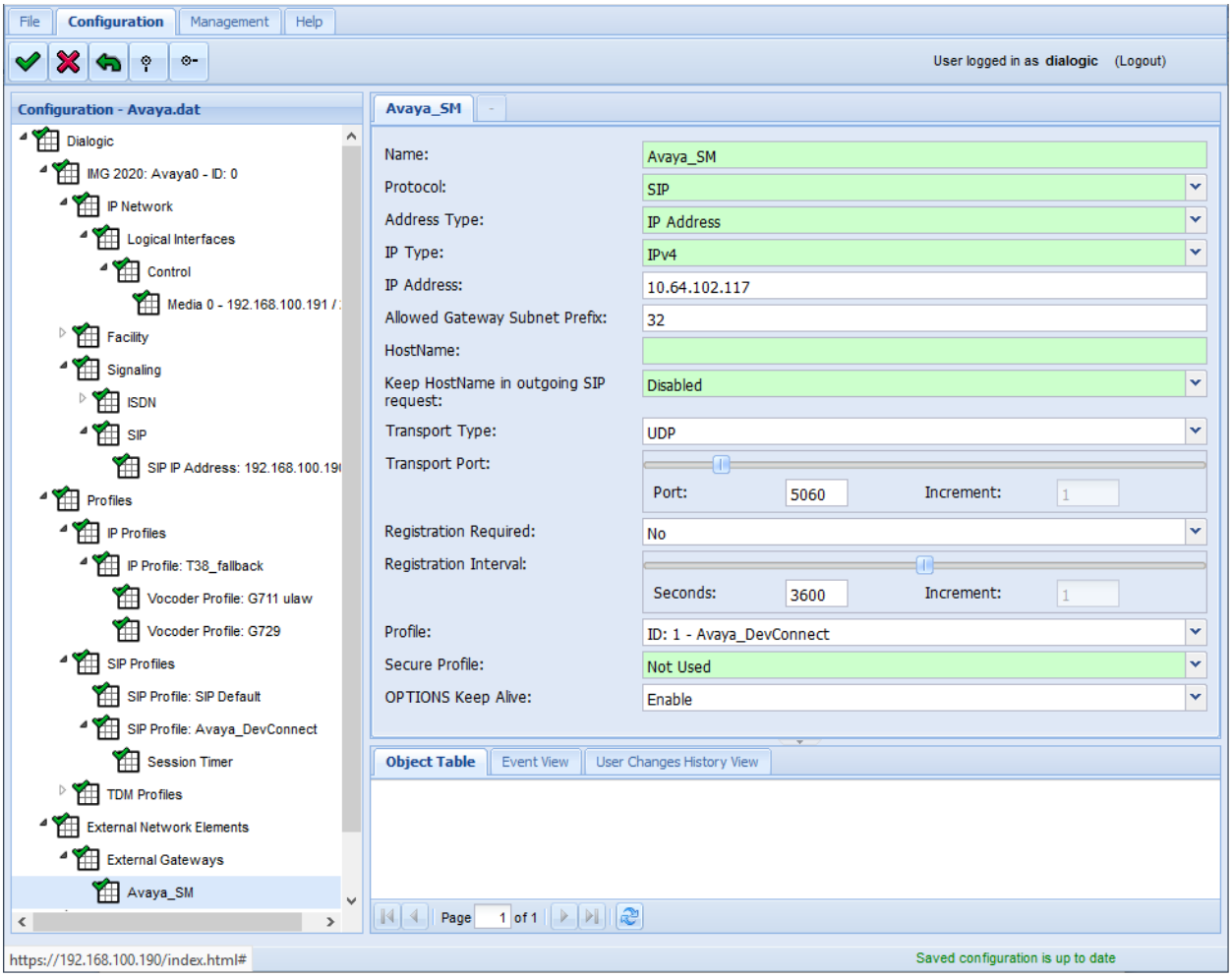


7.6 Configure External Network Element

Configure the **External Network Element**, which in this case the external SIP gateway is Session Manager. The **External Network Element** will be associated with the **Channel Group** configured in **Section 7.7.1**. To create the **External Network Element**, follow these steps.

1. Right-click on the **Dialogic** object and select **New Network Elements**.
2. Right-click on the **External Network Elements** object and select **New External Gateways**.
3. Right-click on the **External Gateways** object and select **New External Gateway**.

In the **External Gateway** screen, provide a descriptive name in the **Name** field (e.g., *Avaya_SM*). Set the **Protocol** to *SIP* and the **IP Address** to the IP address of the Session Manager signaling interface (e.g., *10.64.102.117*). Set the **Transport Type** and **Transport Port** to *UDP* and *5060*, respectively. Set the **Profile** to *Avaya_DevConnect* to associate the external network element with the **SIP Profile** created in **Section 7.5.2**. Lastly, enable **OPTIONS Keep Alive** to allow IMG 2020 to send SIP Options messages to Session Manager.



7.7 Configure Routing

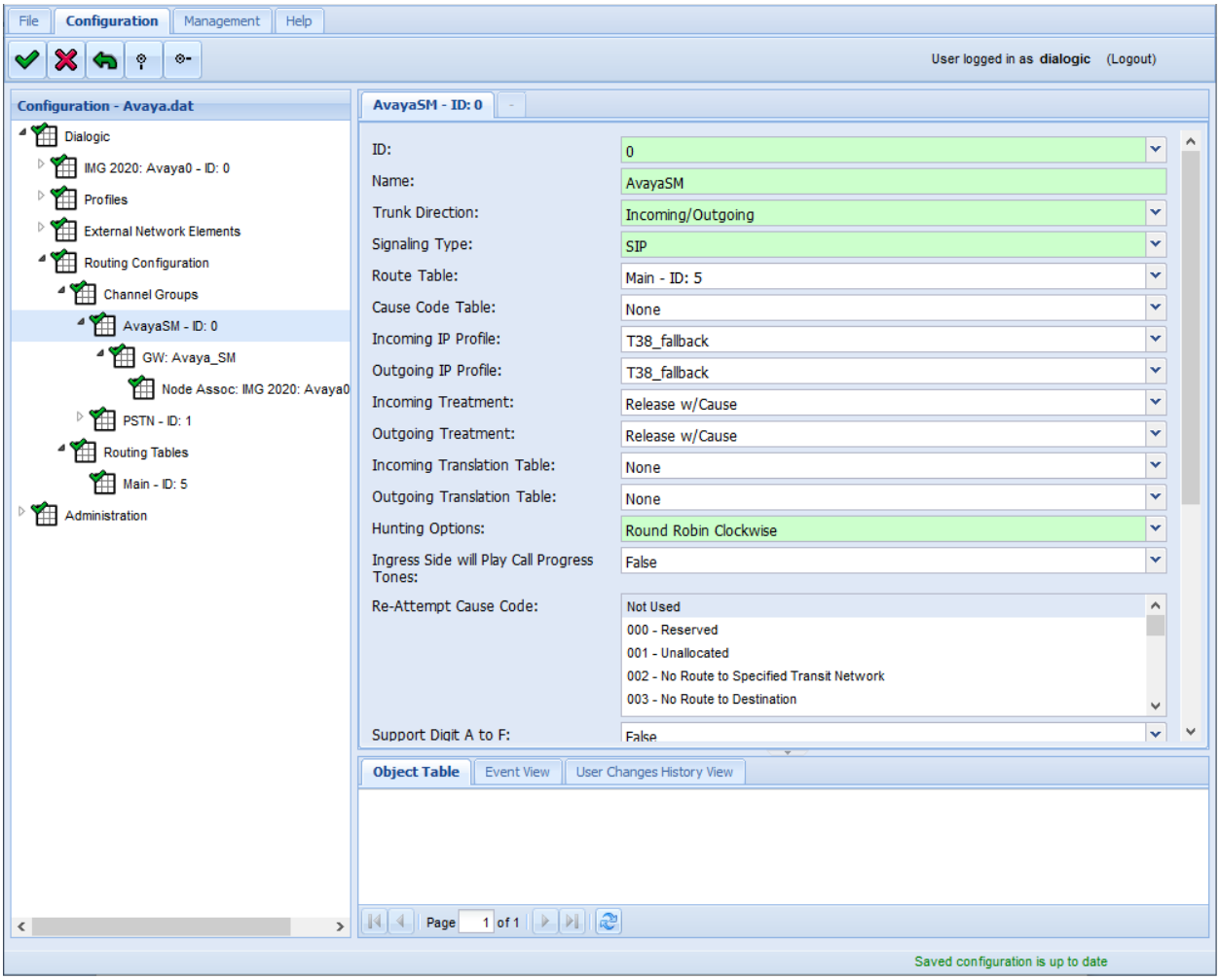
This section covers the configuration for routing calls between the Avaya enterprise network and the PSTN. For the compliance test, basic routing was set up where calls received from the SIP trunk was automatically routed out the ISDN-PRI trunk and vice versa. To configure routing, **Channel Groups** and **Routing Tables** need to be created as specified in this section.

7.7.1 Configure Channel Group

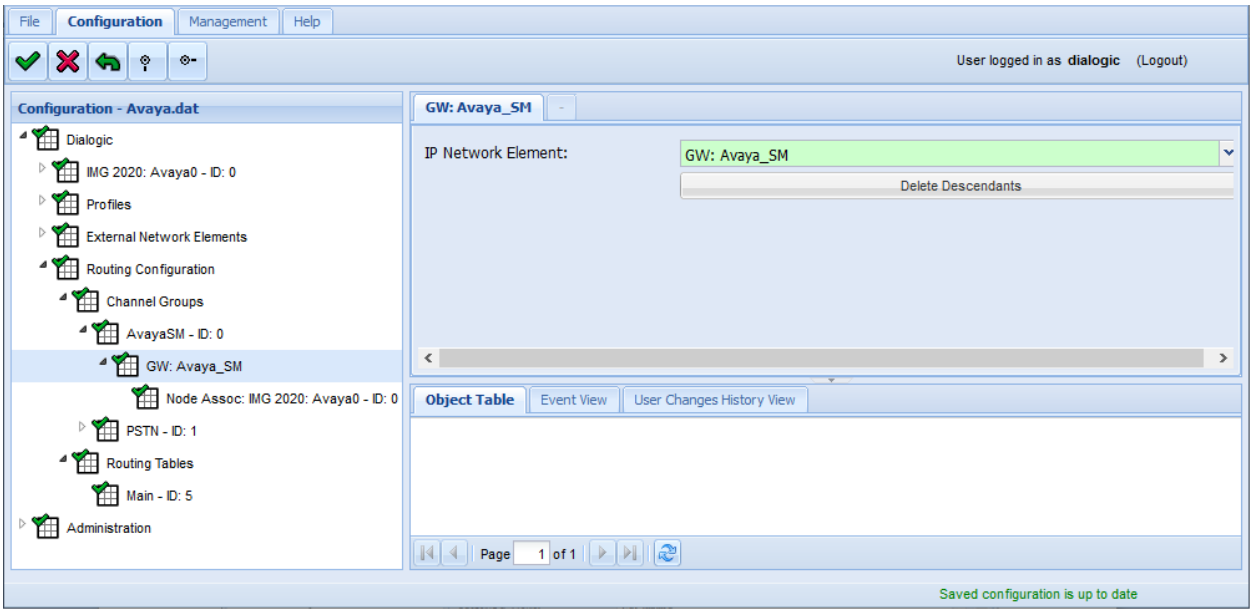
To configure the Channel Group for the SIP trunk, follow these steps.

1. Right-click on the **Dialogic** object and select **New Routing Configuration**.
2. Right-click on the **Routing Configuration** object and select **New Channel Groups**.
3. Right-click on the **Channel Groups** object and select **New Channel Group**.

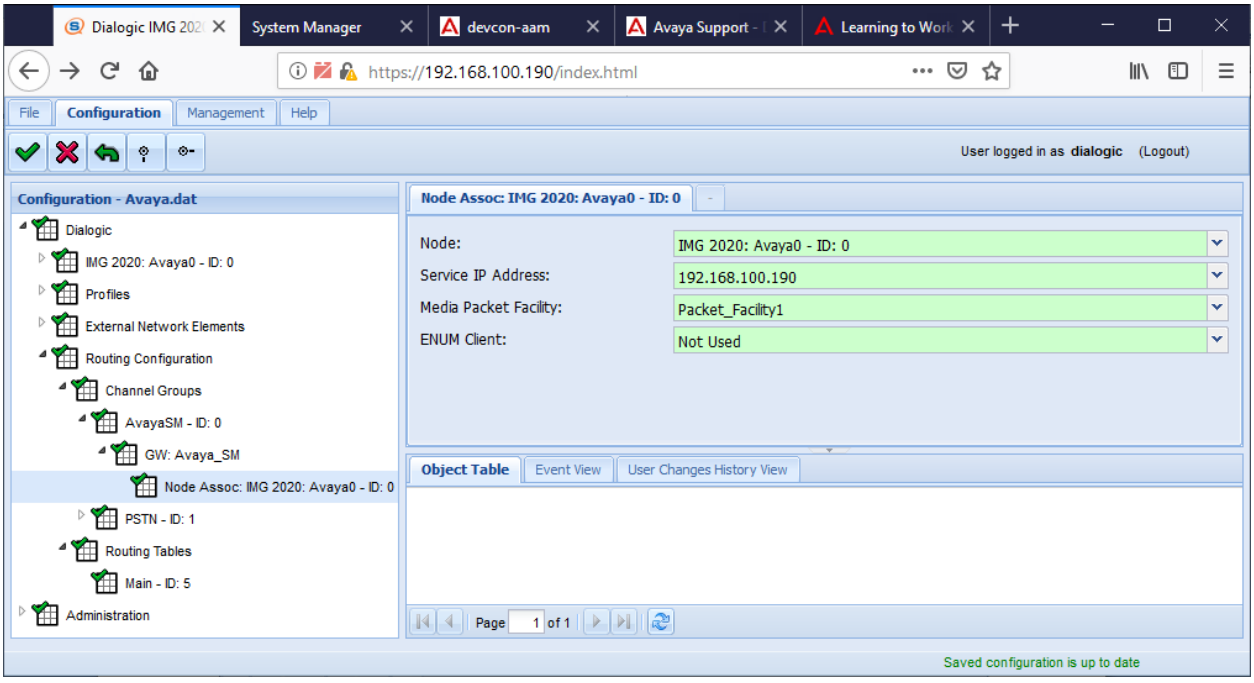
In the **Channel Group** screen, provide a descriptive name in the **Name** field (e.g., *AvayaSM*). Set the **Signaling Type** to *SIP* and set the **Incoming IP Profile** and **Outgoing IP Profile** to the *T38_fallback* IP Profile created in **Section 7.5.1**. Set the **Route Table** to the one configured in **Section 7.7.2**. Keep the default values for the remaining fields.



Right-click on the **AvayaSM** channel group object (name may vary) and select **New IP Network Element**. In the **Network Element** screen, set **IP Network Element** to the **Avaya_SM External Gateway** configured in **Section 7.6**.



Right-click on the **GW: Avaya_SM** object (name may vary) and select **New Node Association**. Set **Node** to the IMG 2020 Physical Node created in **Section 7.2**, set the **Service IP Address** to the IMG 2020 Signaling IP address (e.g., *192.168.100.190*), and set the **Media Packet Facility** to *Packet_Facility1*.

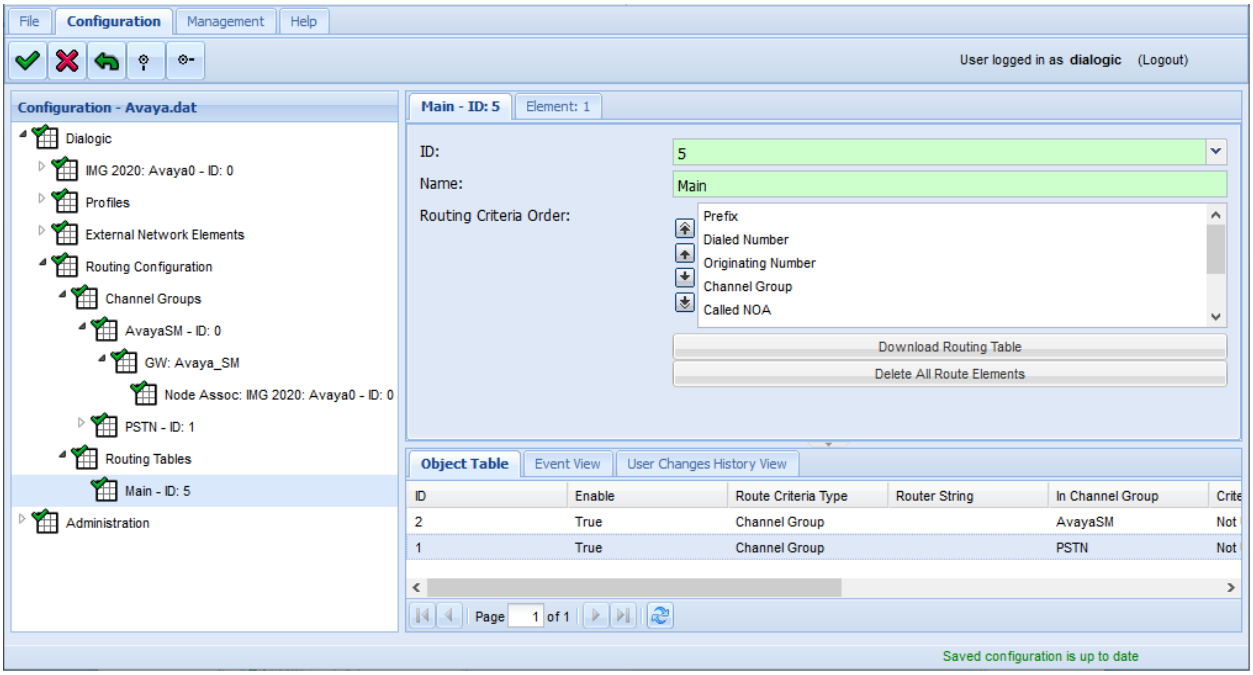


7.7.2 Configure Routing Table

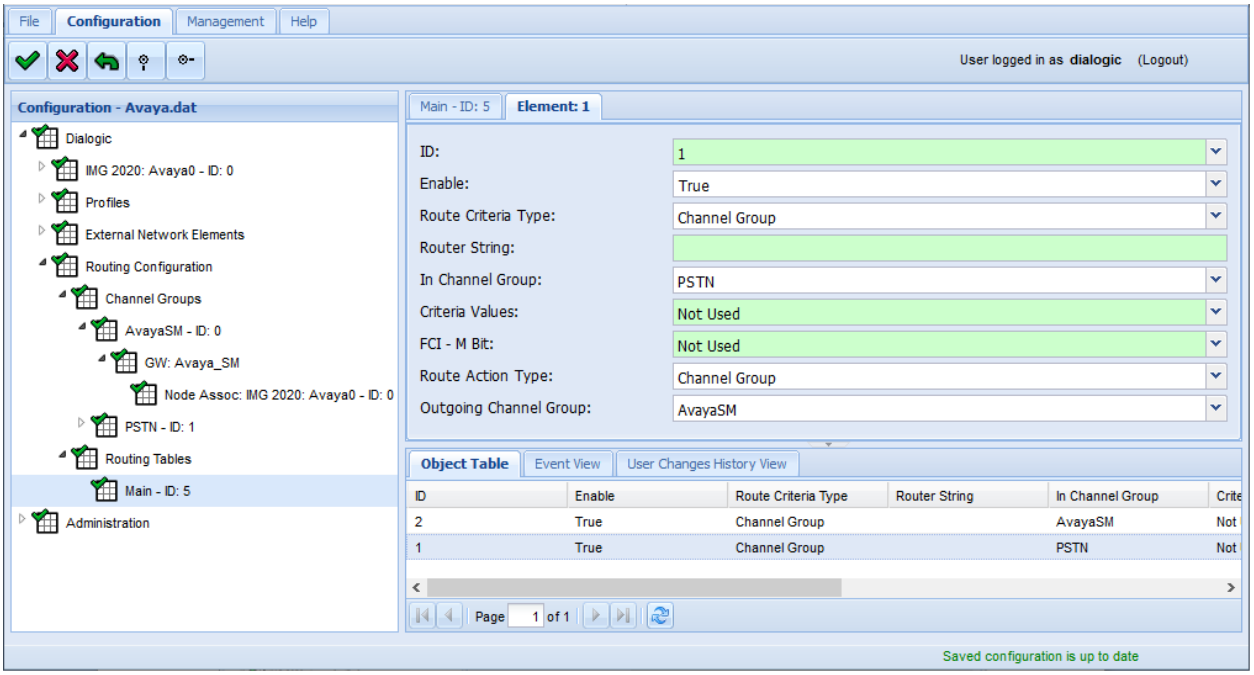
For the compliance test, basic routing was set up where calls received from the SIP trunk was automatically routed out the ISDN-PRI trunk and vice versa. To create the Routing table, follow these steps.

1. Right-click on the **Routing Configuration** object and select **New Routing Table**.
2. Right-click on the **Routing Tables** object and select **New Route Table**.

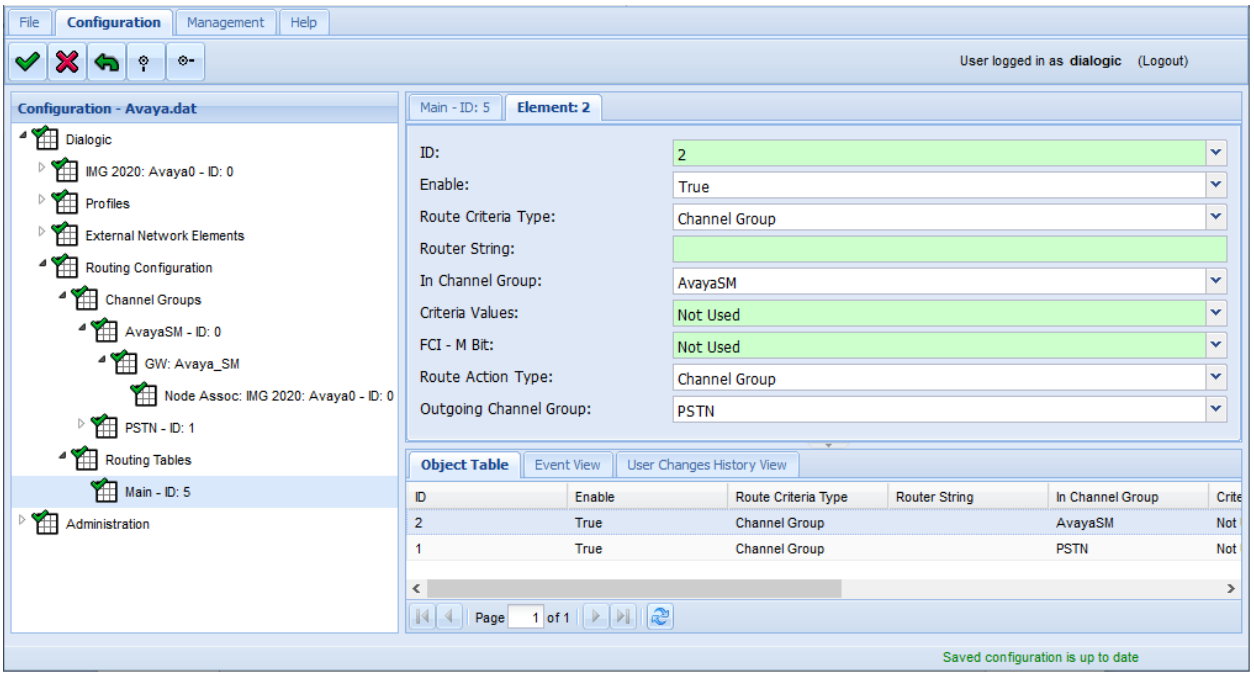
In the **Route Table** screen, enter a descriptive name in the **Name** field (e.g., *Main*).



Right-click on the **Main – ID 5** object (name may vary) and select **New Route Element**. In the **Element** tab, set **Route Criteria Type** and **Route Action Type** to *Channel Group*. Set the **In Channel Group** to the channel group for the ISDN-PRI interface (not shown in these Application Notes). Set the **Outgoing Channel Group** to the *AvayaSM* channel group configured in **Section 7.7.1**. This routing configuration specifies that incoming calls on the ISDN-PRI trunk from the PSTN will be routed out the SIP trunk to the Avaya enterprise site.

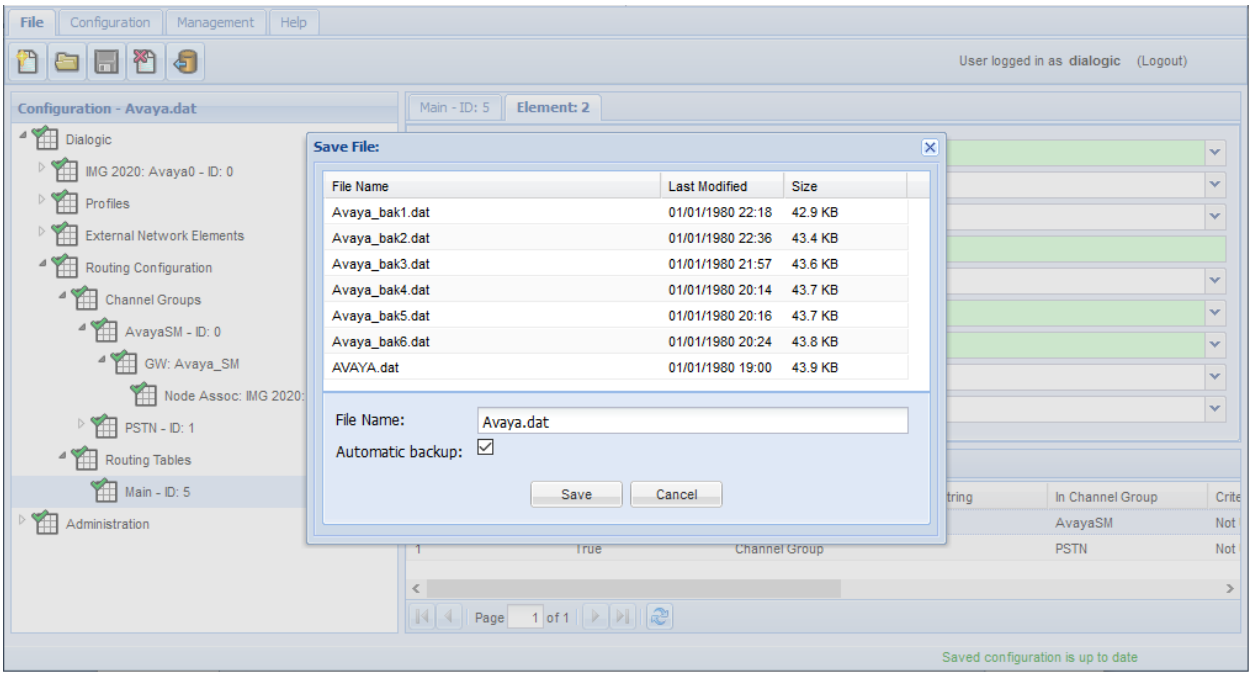


Repeat the step above. Right-click on the **Main – ID 5** object (name may vary) and select **New Route Element**. In the **Element** tab, set **Route Criteria Type** and **Route Action Type** to *Channel Group*. Set the **In Channel Group** to the *AvayaSM* channel group configured in **Section 7.7.1**. Set the **Outgoing Channel Group** to the channel group for the ISDN-PRI interface (not shown in these Application Notes). This routing configuration specifies that incoming calls on the SIP trunk from the Avaya enterprise site will be routed out the ISDN-PRI trunk to the PSTN.



7.8 Save Configuration

Once the configuration is complete, save the configuration by navigating to **File → Save** and specify the appropriate **File Name** in the dialog box as shown below. Click **Save**.



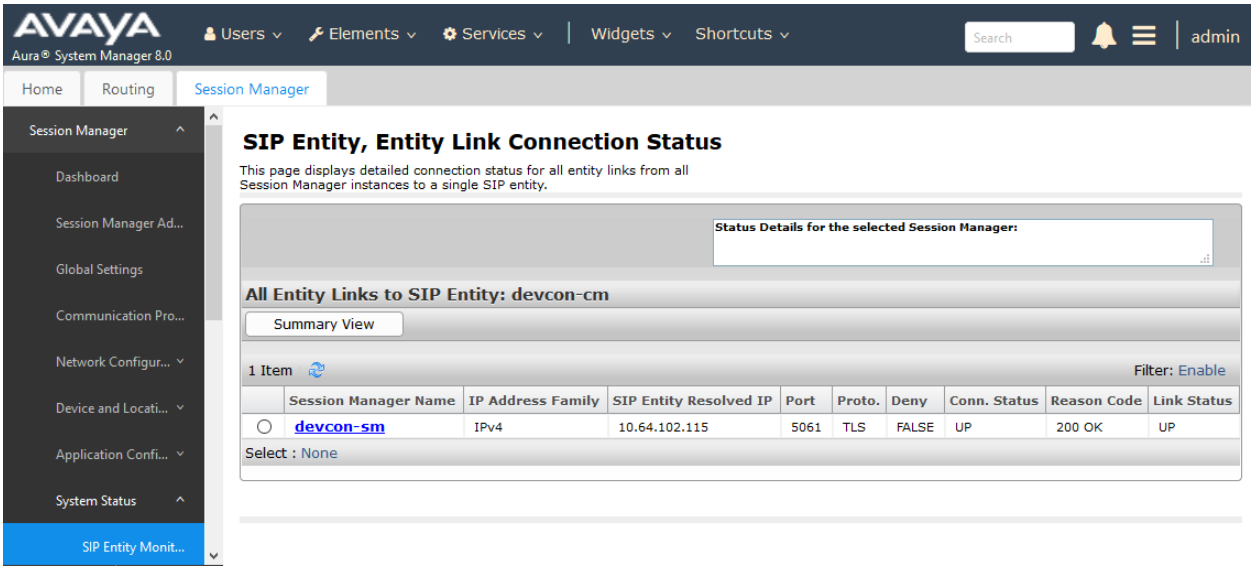
8 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, Sangoma IMG 2020.

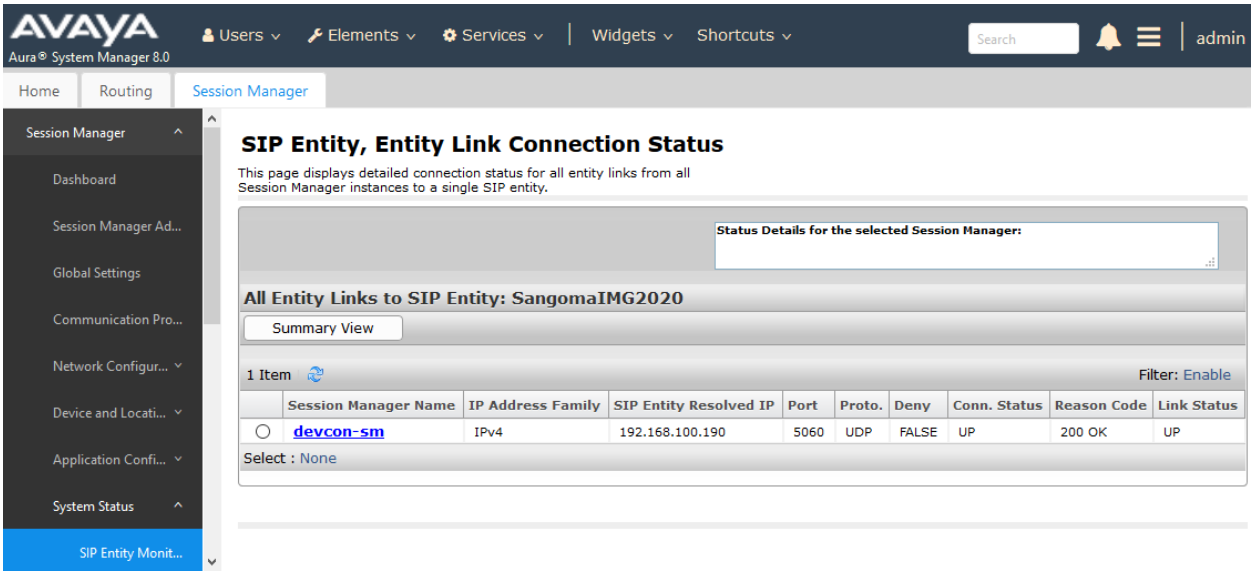
1. The **status trunk** command may be used on Communication Manager to view the **Service State** of the SIP trunk to Session Manager. The trunk members should be in the *in-service/idle* state when no calls are active as shown below.

```
status trunk 10

                            TRUNK GROUP STATUS

Member    Port    Service State       Mtce Connected Ports
                                      Busy

0010/0001 T00001  in-service/idle     no
0010/0002 T00002  in-service/idle     no
0010/0003 T00003  in-service/idle     no
0010/0004 T00004  in-service/idle     no
0010/0005 T00005  in-service/idle     no
0010/0006 T00006  in-service/idle     no
0010/0007 T00007  in-service/idle     no
0010/0008 T00008  in-service/idle     no
0010/0009 T00009  in-service/idle     no
0010/0010 T00010  in-service/idle     no
```

2. Alternatively, the connection status of the SIP trunk between Communication Manager and Session Manager may be viewed on System Manager by navigating to **Elements → Session Manager → System Status → SIP Entity Monitoring** and clicking on the appropriate SIP entities. Below is the status of the SIP trunk to Communication Manager. The **Conn. Status** should be *UP*.



3. As in the step above, the connection status of the SIP trunk between IMG 2020 and Session Manager may also be viewed on System Manager. Below is the status of the SIP trunk to IMG 2020. The **Conn. Status** should be *UP*.



4. Place an outgoing PSTN call from an Avaya H.323 or SIP Deskphone and verify that the call is established successfully with two-way audio.

5. Place an incoming PSTN call to an Avaya H.323 or SIP Deskphone and verify that the call is established successfully with two-way audio.

6. Verify that fax calls are successful between the Avaya enterprise network and the PSTN.

9 Conclusion

These Application Notes describe the configuration steps required to integrate the Sangoma IMG 2020 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voice and fax calls were successfully established between the Avaya enterprise site and the PSTN. In addition, telephony features such as hold/resume, call transfers, conference calls, call forwarding, calling number display/block, and voicemail coverage were also exercised. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10 References

This section references the Avaya and Sangoma documentation relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.0.1, Issue 3, December 2018, available at http://support.avaya.com.

[2] *Administering Avaya Aura® System Manager for Release 8.0.1*, Release 8.0.x, Issue 7, January 2019, available at http://support.avaya.com.

[3] *Administering Avaya Aura® Session Manager*, Release 8.0.1, Issue 3, December 2018, available at http://support.avaya.com.

[4] *Sangoma IMG 2020 User Manual*, available at https://wiki.sangoma.com/display/DIMG/IMG+2020+Documentation

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.