Avaya Solution & Interoperability Test Lab

Application Notes for Amtelco Genesis Intelligent Series with Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Amtelco Genesis Intelligent Series to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Amtelco Genesis Intelligent Series is a SIP-based solution that provides operator users with phone and call controls.

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

These Application Notes describe the configuration steps required for Amtelco Genesis Intelligent Series (Genesis) to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Genesis is a SIP-based solution that provides operator users with phone and call controls.

The Genesis solution consists of the Genesis Telephony Server, Intelligent Series Server, Intelligent Series Supervisor, and Intelligent Series Soft Agent. Operators have desktops running the Intelligent Series Soft Agent application, with dedicated audio connections via SIP with the Genesis Telephony Server.

In the compliance testing, calls from internal and external callers were routed over SIP trunks via Session Manager to Genesis for operator functions. Genesis tracked the operator states and routed calls to available operators, and populated answering operator desktops with pertinent call information such as calling and called numbers. All call controls were performed from the operator desktops.

The unsupervised transfer feature was accomplished by Genesis via use of SIP REFER, and the supervised transfer and supervised conference features were accomplished by Genesis via merge/unmerge of respective audio connections.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually with necessary operator actions such as hold and transfer performed from the operator desktops.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to the Genesis servers and/or clients.

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 feature and serviceability testing.

The feature testing included inbound, outbound, internal, external, G.711, outbound DTMF, hold/resume, drop, display, transfer, supervised conference, multiple calls, and multiple operators.

The serviceability testing focused on verifying the ability of Genesis to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to the Genesis servers and/or clients.

2.2. Test Results

All test cases were executed and verified. The following were observations on Genesis from the compliance testing.

- When the desktop running the Soft Agent application experiences a network disruption while on an active call, the Soft Agent application may disappear from the screen. A user can recover the application when the network connection is restored by using Windows task manager to manually end the Soft Agent process and then re-launch the application. For additional help reach out to Amtelco Support.

- Genesis returned 404 Not Found for OPTIONS messages from Session Manager, and this was displayed on the SIP Entity connection status screen on Session Manager. This did not appear to have any other negative impact.

2.3. Support

Technical support on Genesis can be obtained through the following:

- **Phone:** (800) 553-7679
- **Email:** service@amtelco.com
- **Web:** www.amtelco.com/Welcome.htm

3. Reference Configuration

As shown in **Figure 1**, operators have desktops running the Intelligent Series Soft Agent application, and dedicated SIP connections with the Genesis Telephony Server as part of log in. The Intelligent Series Supervisor was running on the supervisor desktop.

SIP trunks were used between the Genesis Telephony Server and Session Manager. A five digit Uniform Dial Plan was used to facilitate dialing with Genesis. Calls to extensions 52xxx were routed over the SIP trunks to Genesis. In particular, internal users on Communication Manager will dial 52222 to reach Genesis, and calls from external users will be routed with digits 52000 to Genesis.

The detailed administration of connectivity between Communication Manager and Session Manager are not the focus of these Application Notes and will not be described.

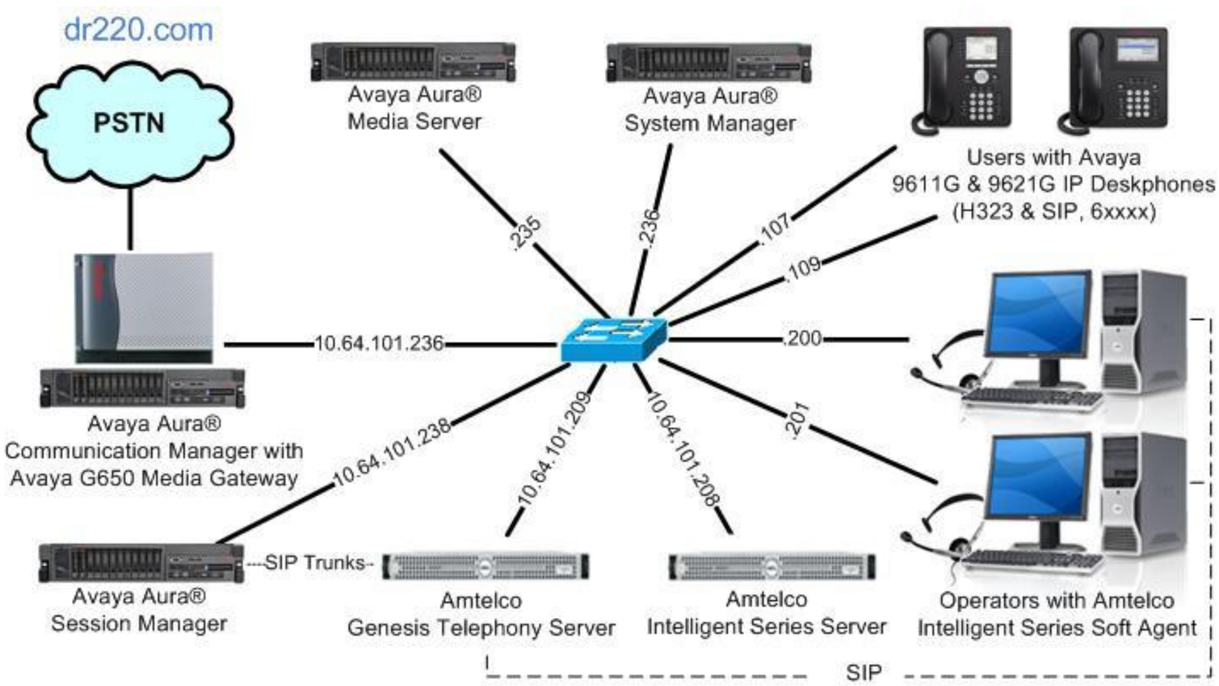


Figure 1: Compliance Testing Configuration

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 in Virtual Environment	7.0.1.1 (7.0.1.1.0.441.23169)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	7.7.0.334
Avaya Aura® Session Manager in Virtual Environment	7.0.1.1 (7.0.1.1.70114)
Avaya Aura® System Manager in Virtual Environment	7.0.1.1 (7.0.1.1.065378)
Avaya 9611GIP Deskphones (H.323)	6.6302
Avaya 9621G IP Deskphone (SIP)	7.0.1.2.9
Amtelco Genesis Telephony Server on Debian • Asterisk	3.1.2 Build 1703241247 8.6 14.2.1
Amtelco Intelligent Series Server on Microsoft Windows Server 2008 R2 Enterprise • Microsoft SQL Server 2014	5.0.6291.16498 SP1 12.0.2000.8
Amtelco Intelligent Series Supervisor on Microsoft Windows 10 Pro	5.0.6263.7
Amtelco Intelligent Series Soft Agent on Microsoft Windows 10 Pro	5.0.6263.09

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer ISDN trunk group
- Administer tandem calling party number

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with Genesis.

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

5.2. Administer System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class of Restriction or Class of Service levels. Refer to [1] for more details.

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: music Type: ext   40001
              Music (or Silence) on Transferred Trunk Calls? Call-wait
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

5.3. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "52". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"

```
add trunk-group 52                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip          CDR Reports: y
  Group Name: SIP Trunks to Genesis      COR: 1        TN: 1        TAC: 1052
   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:
                                                    Number of Members: 0
```

Navigate to **Page 3**, and enter "private" for **Numbering Format**.

```
add trunk-group 52                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

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




 Show ANSWERED BY on Display? y
```

5.4. Administer SIP Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** An existing C-LAN node name or “procr” in this case.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration with Genesis.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Genesis.
- **Far-end Domain:** The applicable domain name for the network.

For **Direct IP-IP Audio Connections**, enter “n” since Genesis does not support media shuffling.

```
add signaling-group 52                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 52             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

   Near-end Node Name: procr                 Far-end Node Name: sm7-sig
 Near-end Listen Port: 5052                Far-end Listen Port: 5052
                                        Far-end Network Region: 5
                                   Far-end Secondary Node Name:
Far-end Domain: dr220.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? n
Session Establishment Timer(min): 3                 IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                              Alternate Route Timer(sec): 6
```

5.5. Administer SIP Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case "10".

```
change trunk-group 52                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip           CDR Reports: y
  Group Name: SIP Trunks to Genesis       COR: 1        TN: 1        TAC: 1052
   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: 52
                                                     Number of Members: 10
```

5.6. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “no” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Genesis.

```
change ip-network-region 5                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 5
Location:          Authoritative Domain: dr220.com
    Name: Genesis                 Stub Network Region: n
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: no
      Codec Set: 5                Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                        IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with the network region used by the Avaya endpoints and with the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and trunk to the PSTN.

```
change ip-network-region 5                                        Page   4 of  20

 Source Region: 5     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    5      y    NoLimit                                            n       t
 2
 3
 4
 5    5                                                               all
 6
```

5.7. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Genesis supports the G.711 and G.729 codec variants, with G.729 requiring special license on Genesis. The compliance testing only covered the G.711 codec.

```
change ip-codec-set 5                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 5

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
 4:
 5:
```

5.8. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an available route pattern number to be used to reach Genesis, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 52                                         Page   1 of   3
                    Pattern Number: 52     Pattern Name: Genesis
                             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: 52   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                                        none
```

5.9. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Genesis. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 52 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

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

Ext Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  6                52                           5     Total Administered: 1
                                                           Maximum Entries: 540
```

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 52xxx to Genesis. Note that other routing methods may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing of digits 52xxx, as shown below.

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

  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num

 52              5   0                  aar  n
```

5.11. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 52xxx. In the example shown below, calls with digits 52xxx will be routed as an AAR call using route pattern "52" from **Section 5.8**.

```
change aar analysis 0                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all         Percent Full:    2

          Dialed          Total     Route     Call   Node  ANI
          String         Min  Max  Pattern    Type   Num   Reqd
    52                   5    5     52        unku         n
```

5.12. Administer ISDN Trunk Group

Use the "change trunk-group n" command, where "n" is the existing ISDN trunk group number used to reach the PSTN if applicable, in this case "13".

Navigate to **Page 3**. For **Modify Tandem Calling Number**, enter "tandem-cpn-form" to allow calling party numbers from Genesis to be modified.

```
change trunk-group 13                                                 Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                                         Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n            Format: natl-pub
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
                                                     Send Connected Number: y
Network Call Redirection: none                   Hold/Unhold Notifications? n
             Send UUI IE? y     Modify Tandem Calling Number: tandem-cpn-form
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                            Ds1 Echo Cancellation? n

    Apply Local Ringback? n               US NI Delayed Calling Name Update? n
 Show ANSWERED BY on Display? Y      Invoke ID for USNI Calling Name: variable
                             Network (Japan) Needs Connect Before Disconnect? n
```

5.13. Administer Tandem Calling Party Number

Use the "change tandem-calling-party-num" command, to define the calling party number to send to the PSTN for tandem calls from Genesis.

In the example shown below, all calls originating from a 5-digit extension beginning with 52 and routed to trunk group 13 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case "pub-unk".

```
change tandem-calling-party-num                                       Page   1 of   8
                  CALLING PARTY NUMBER CONVERSION
                          FOR TANDEM CALLS
     CPN              Trk                                Number
 Len Prefix           Grp(s)     Delete  Insert          Format

 5   6                13                 30353           pub-unk
 5   52               13                 30353           pub-unk
```

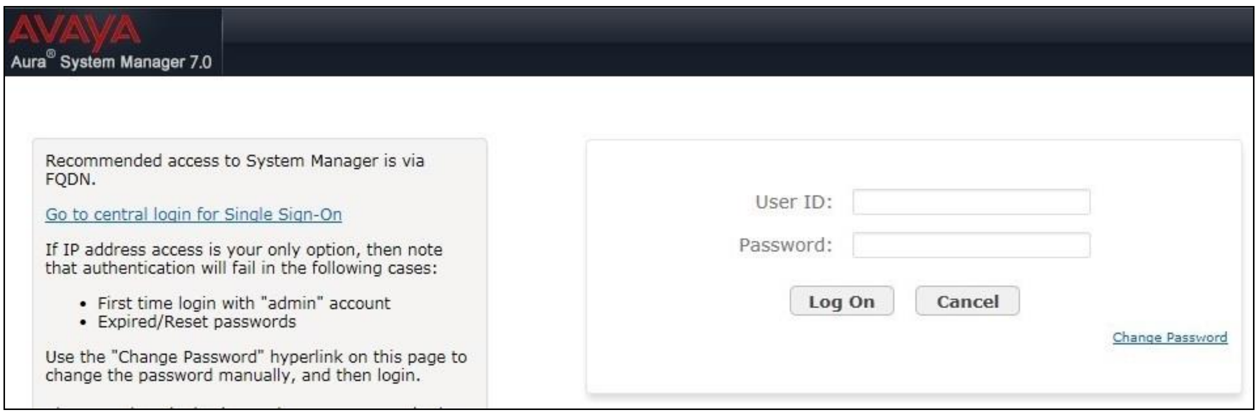
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

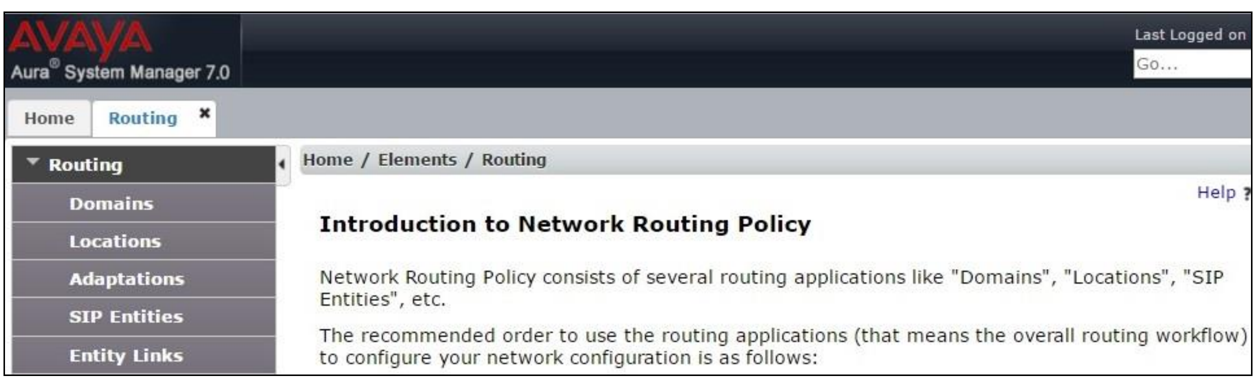
6.1. Launch System Manager

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 System Manager. Log in using the appropriate credentials.

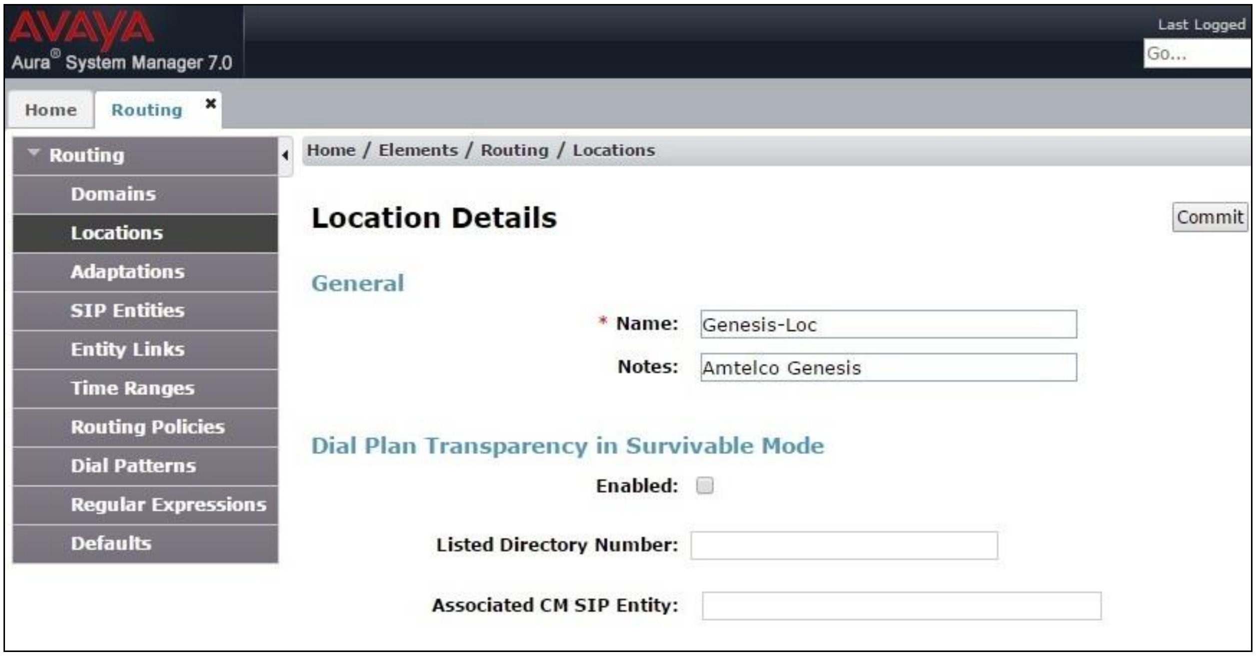


6.2. Administer Locations

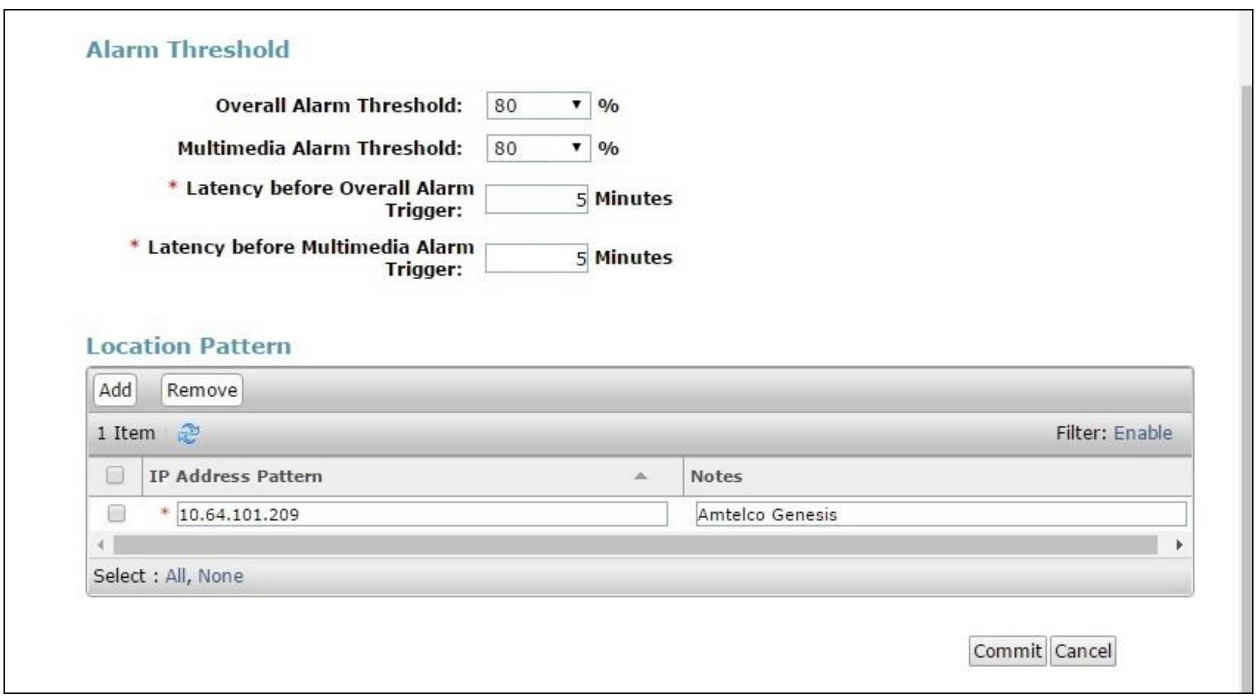
In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Genesis.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of the Genesis Telephony Server in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



6.3. Administer SIP Entities

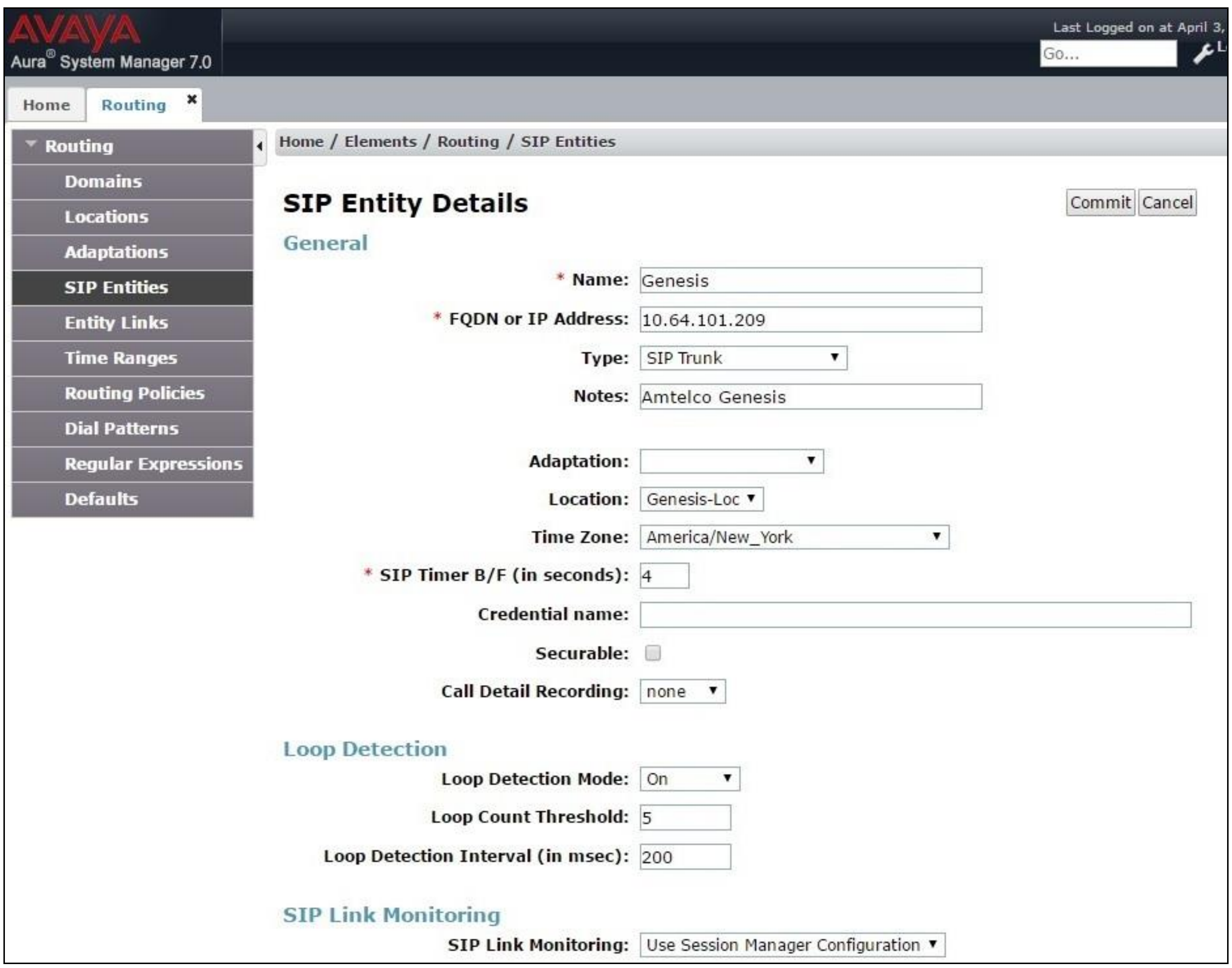
Add two new SIP entities, one for Genesis and one for the new SIP trunks with Communication Manager.

6.3.1. SIP Entity for Genesis

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Genesis.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

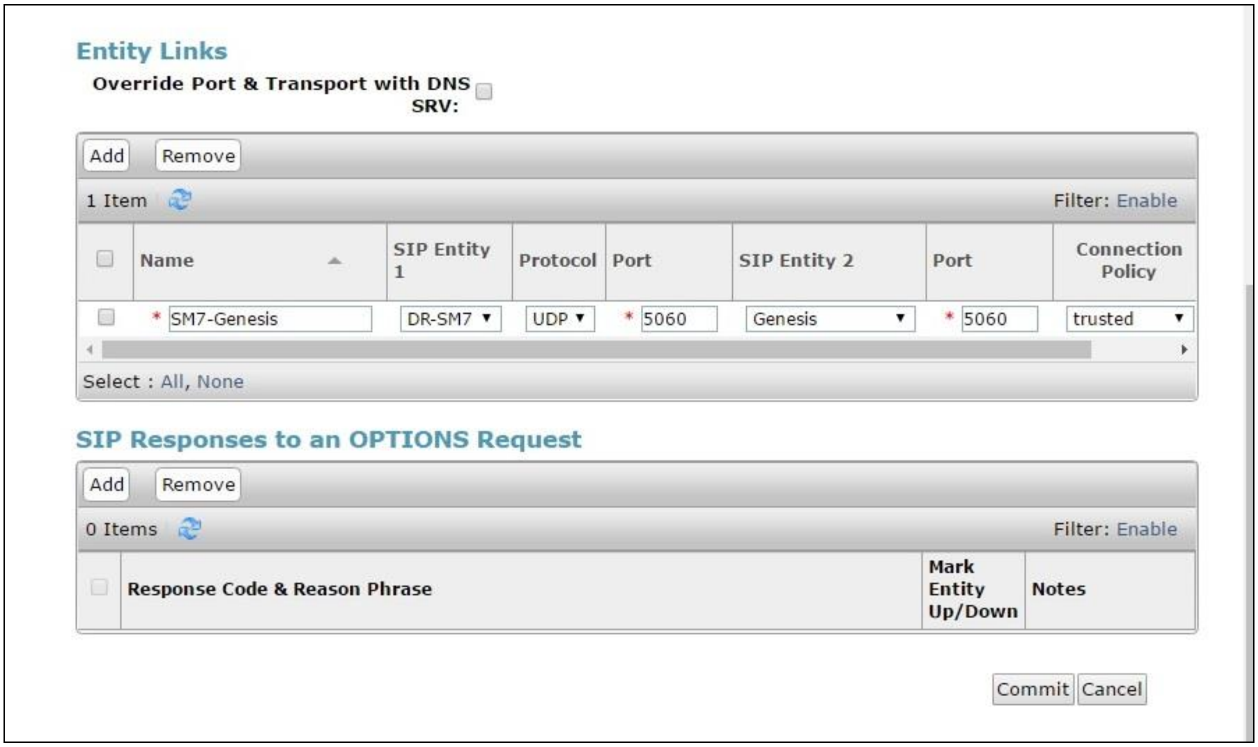
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Genesis Telephony Server.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the Genesis location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DR-SM7".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Genesis entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that Genesis can support UDP and TCP, and the compliance testing used the UDP protocol.

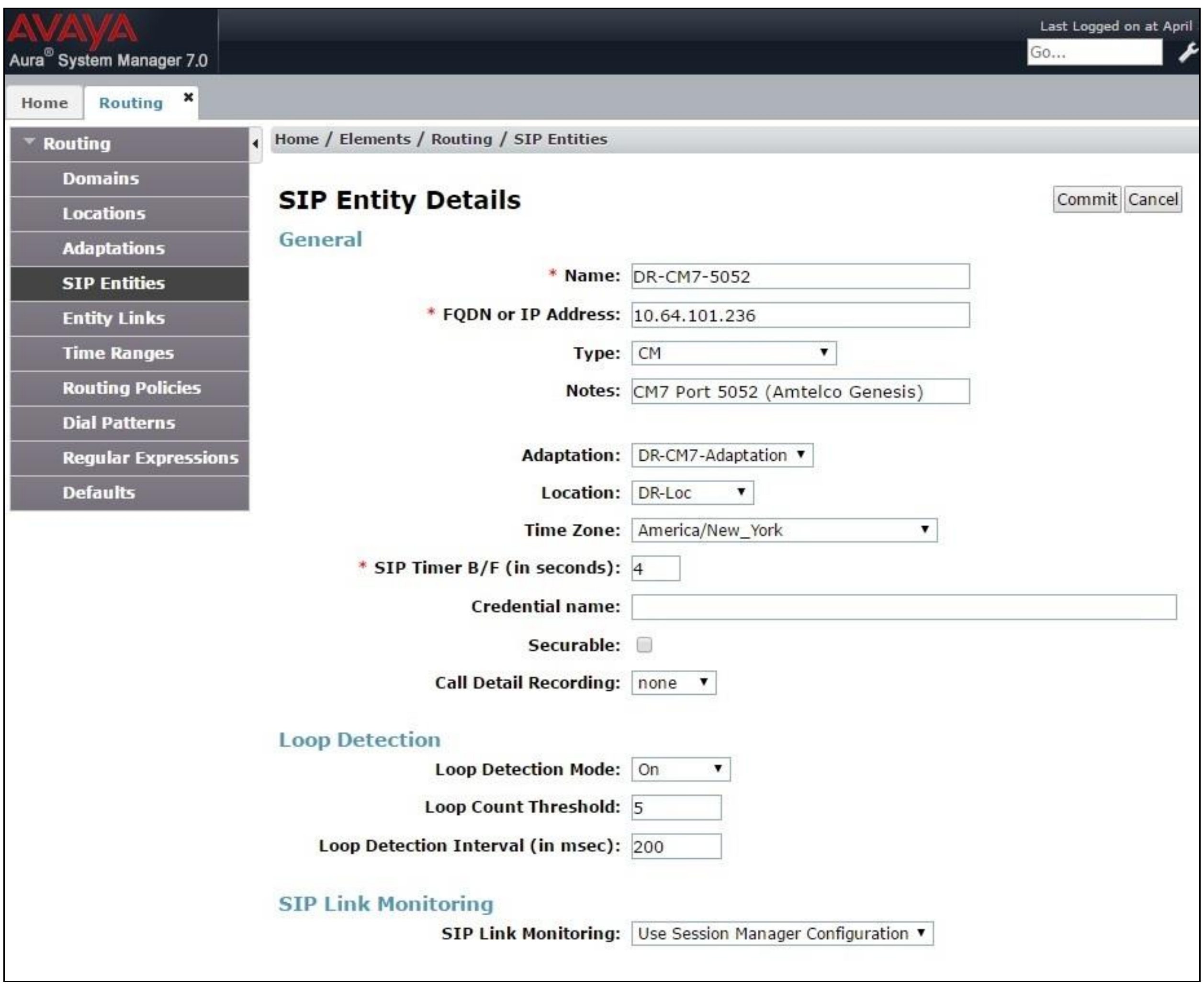


6.3.2. SIP Entity for Communication Manager

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with Genesis.

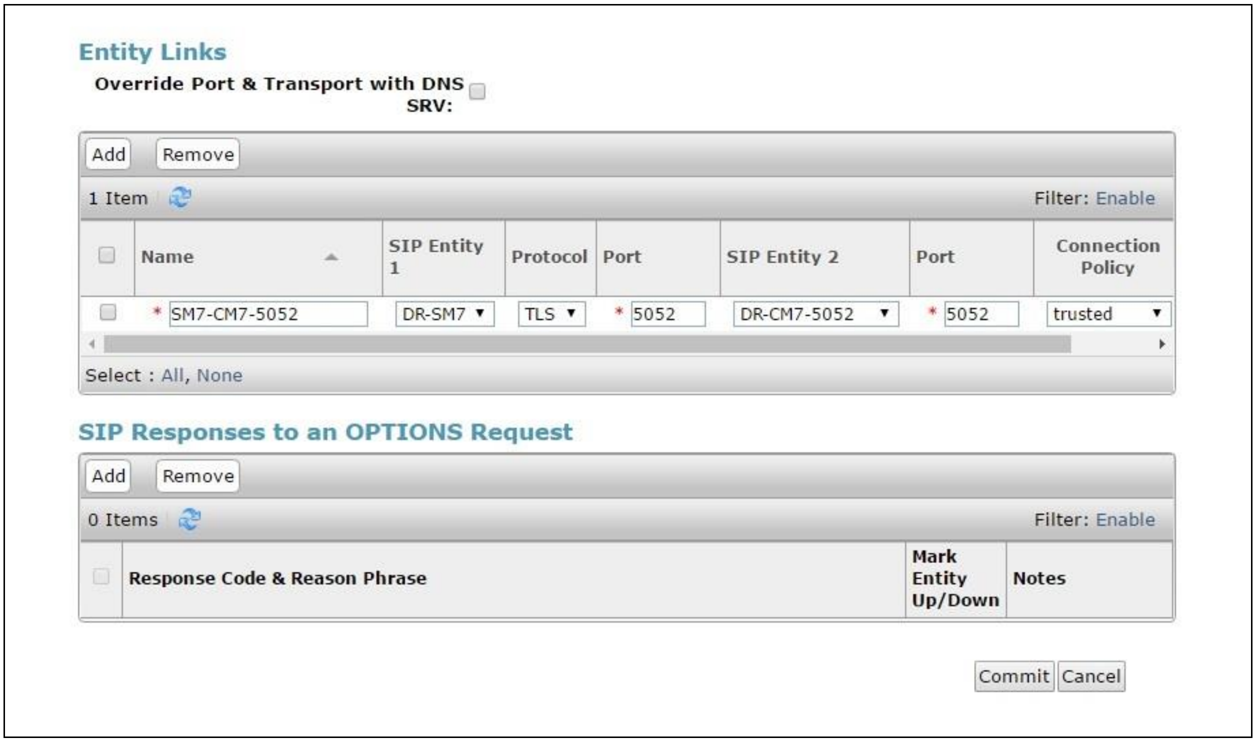
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Notes:** Any desired notes.
- **Adaptation:** Select the applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM7”.
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group far-end listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group near-end listen port number from **Section 5.4**.
- **Connection Policy:** “trusted”



6.4. Administer Routing Policies

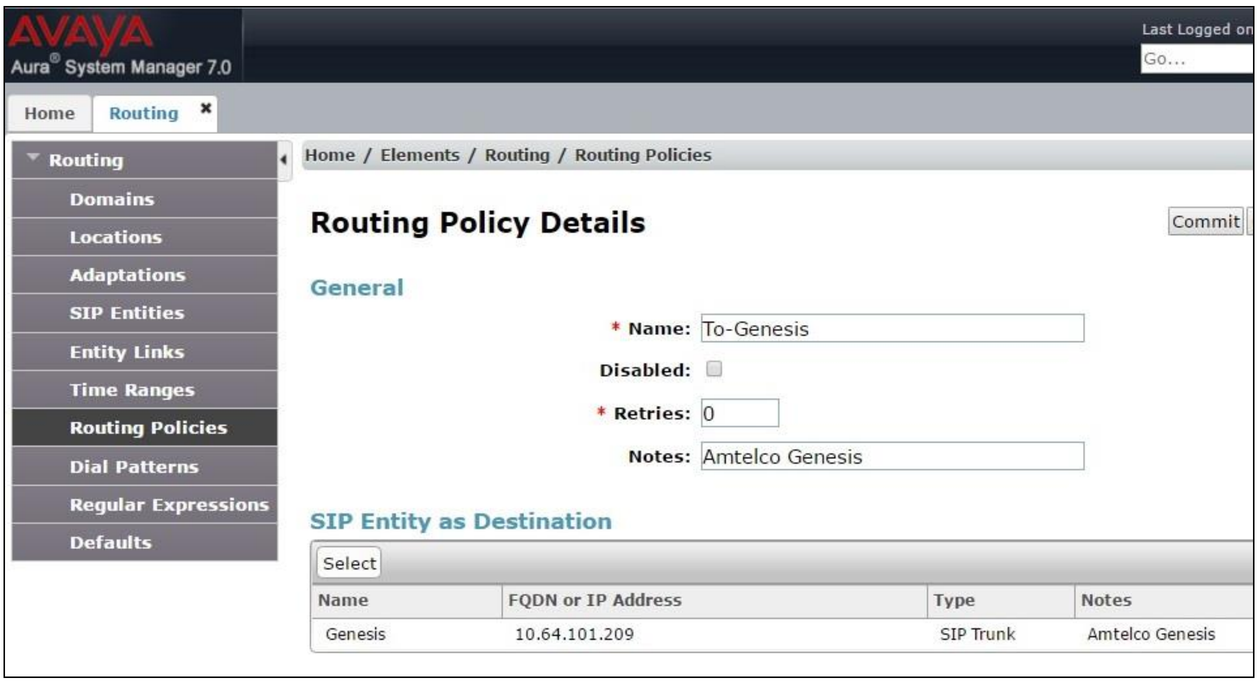
Add two new routing policies, one for Genesis and one for the new SIP trunks with Communication Manager.

6.4.1. Routing Policy for Genesis

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Genesis.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**. Enter optional **Notes**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Genesis entity name from **Section 6.3.1**. The screen below shows the result of the selection.

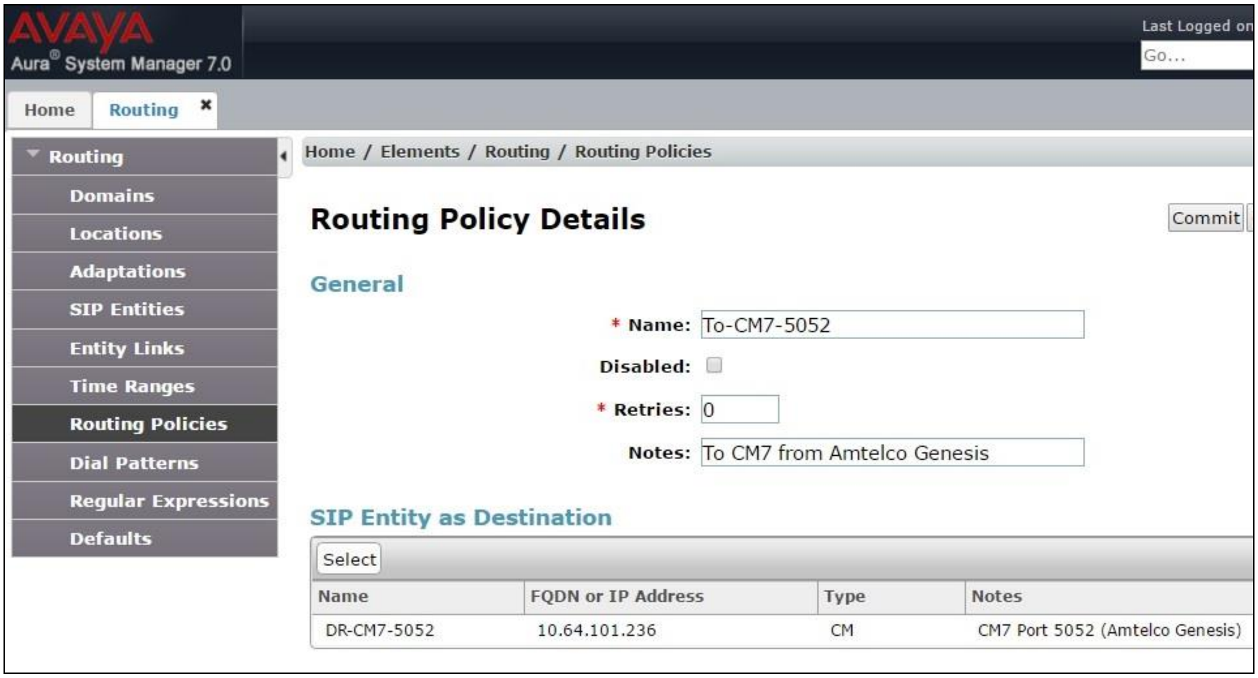


6.4.2. Routing Policy for Communication Manager

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**. Enter optional **Notes**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.3.2**. The screen below shows the result of the selection.



6.5. Administer Dial Patterns

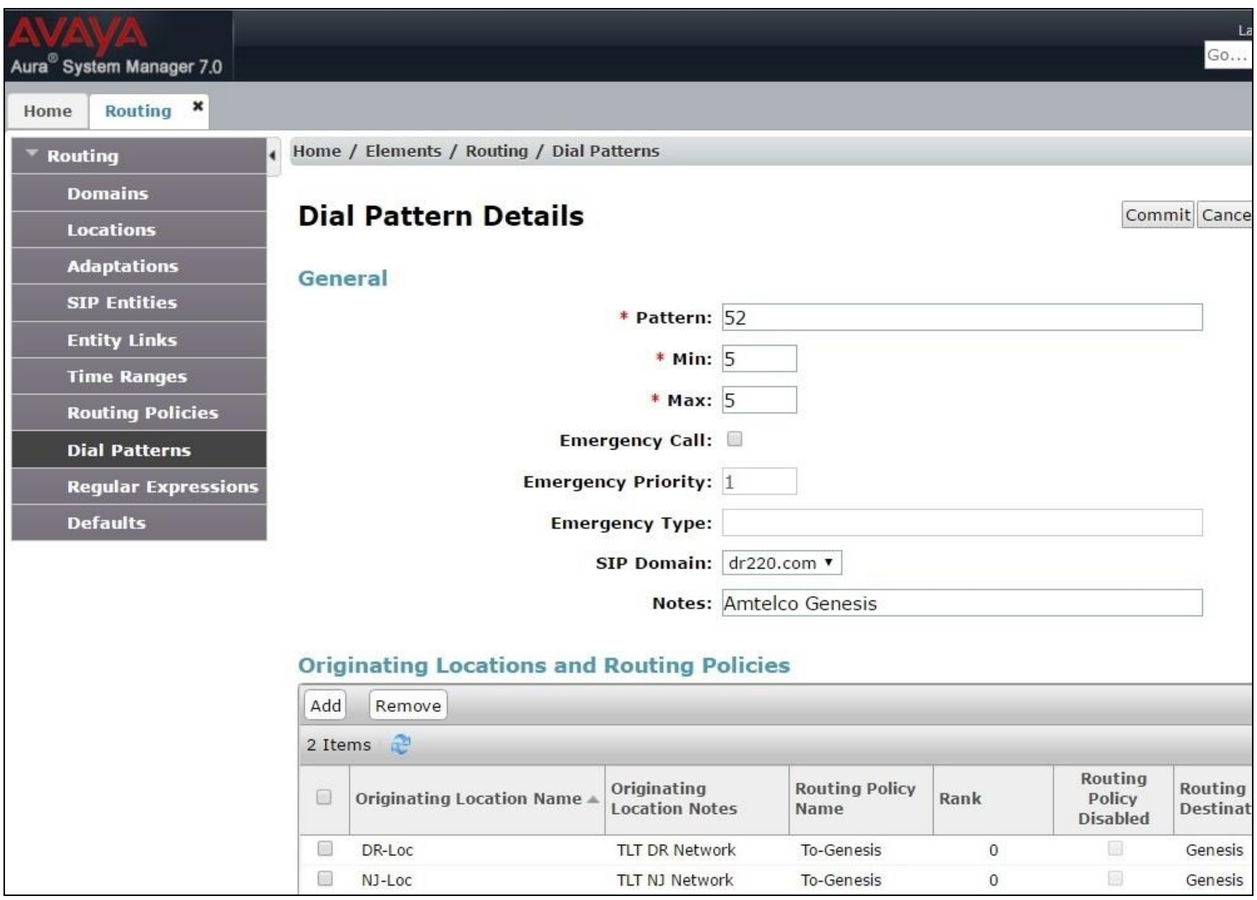
Add a new dial pattern for Genesis, and update existing dial patterns for Communication Manager.

6.5.1. Dial Pattern for Genesis

Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Genesis. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "52".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select the applicable domain, in this case "dr220.com".

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Genesis. In the compliance testing, the entry allowed for call originations from Communication Manager endpoints in locations "DR-Loc" and "NJ-Loc". The Genesis routing policy from **Section 6.4.1** was selected as shown below.

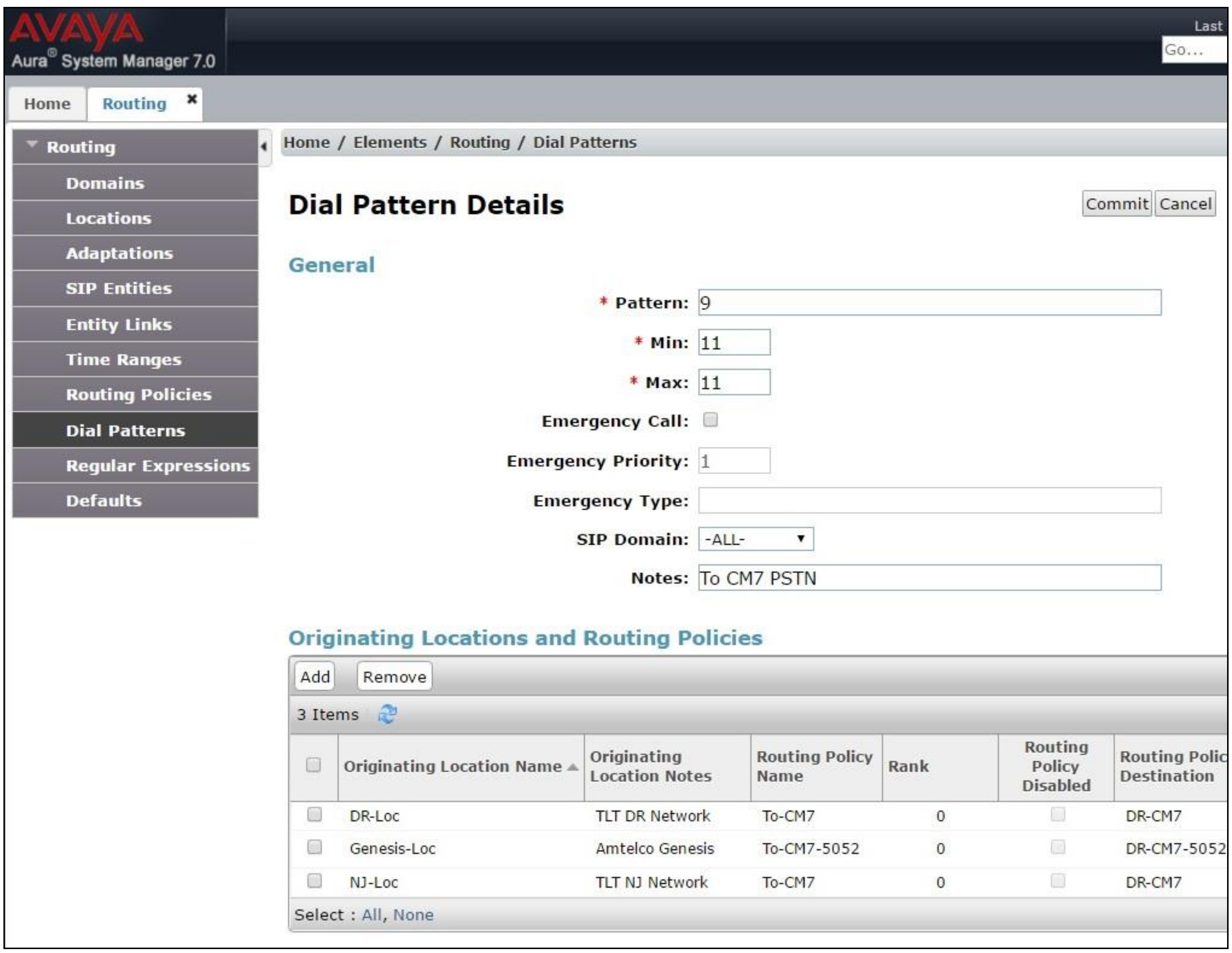


6.5.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "6" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from Genesis. In the compliance testing, the new policy allowed for call origination from the Genesis location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.4.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, operators on Genesis manually added the prefix "9" for outbound calls to the PSTN, and therefore the existing dial pattern for "9" was also changed (not shown below).



7. Configure Amtelco Genesis Intelligent Series

This section provides the procedures for configuring Genesis. The procedures include the following areas:

- Launch web interface
- Obtain application name
- Administer trunks
- Administer routes
- Administer agents
- Administer access control lists
- Launch Intelligent Series Supervisor
- Administer IS system
- Administer IS client
- Administer IS agent
- Restart IS service
- Launch Intelligent Series Soft Agent
- Administer setup

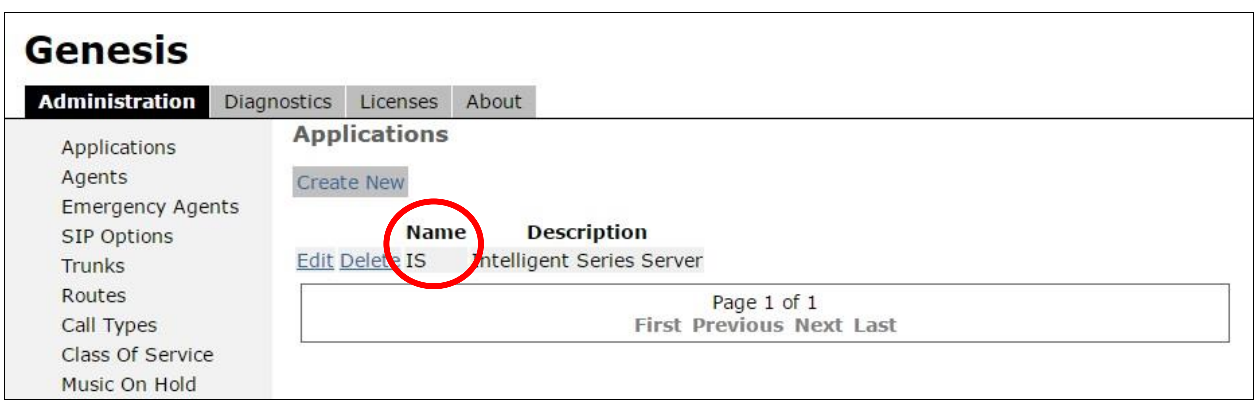
The configuration of Genesis is typically performed by Amtelco technicians. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Launch Web Interface

From a PC, launch an Internet browser window and access the Genesis web-based interface by using the URL “http://<ip-address:5080>/Admin/Application/Index”, where “ip-address” is the IP address of the Genesis Telephony Server.

7.2. Obtain Application Name

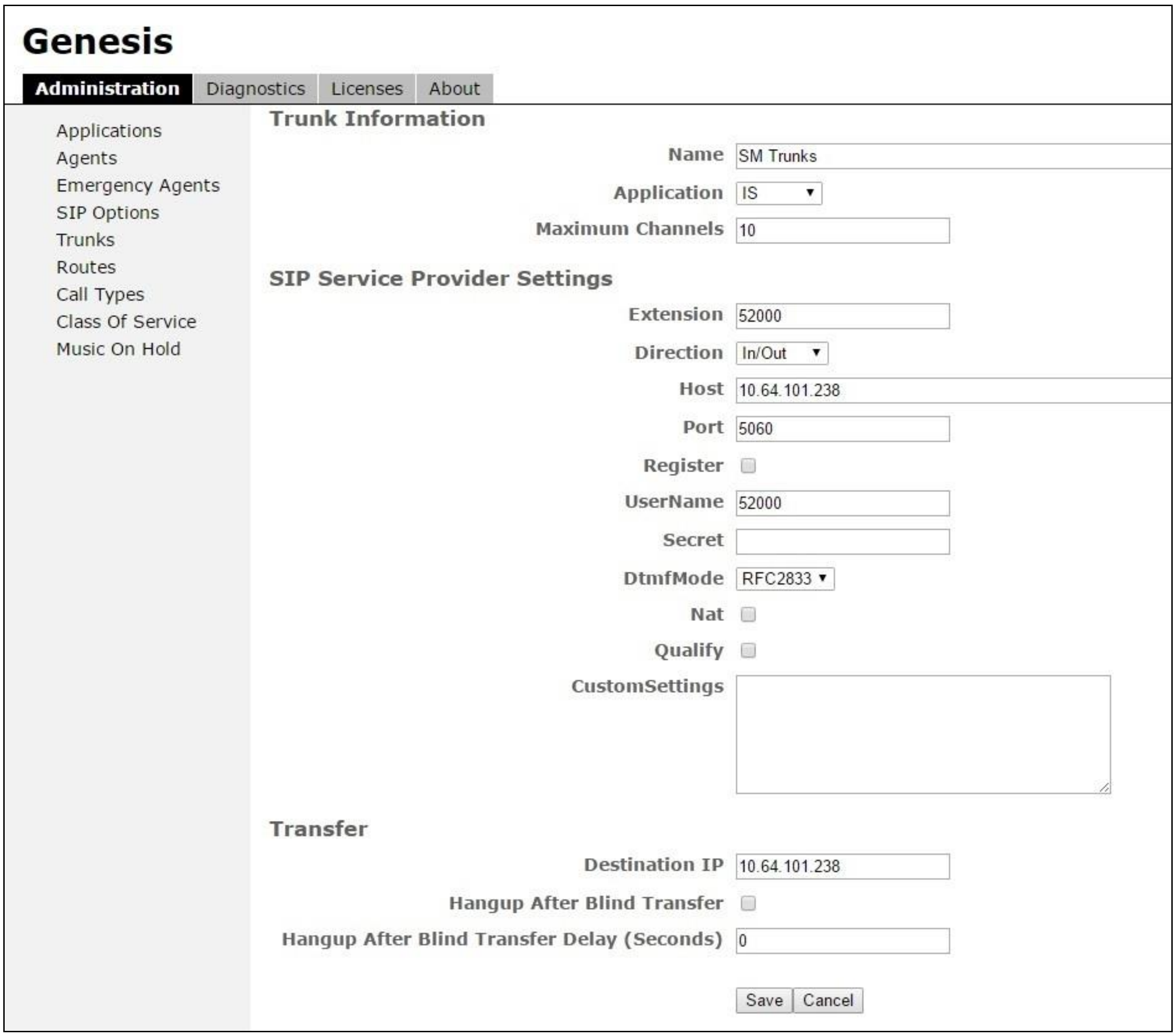
The **Applications** screen below is displayed in the right pane. Make a note of the application **Name**, in this case “IS”, which is created as part of installation. The name will be used in later sections.



7.3. Administer Trunks

Select **Trunks** in the left pane, followed by **Create New SIP Trunk** (not shown) in the updated right pane, to display the **Trunk Information** screen below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **Application:** Select the application name from **Section 7.2**.
- **Maximum Channels:** Enter the number of trunk members from **Section 5.5**.
- **Extension:** The routing extension digits from **Section 3** for calls from PSTN.
- **Host:** IP address of the Session Manager signaling interface.
- **Port:** The Genesis SIP entity port number from **Section 6.3.1**.
- **UserName:** The routing extension digits from **Section 3** for calls from PSTN.
- **Destination IP:** IP address of the Session Manager signaling interface.

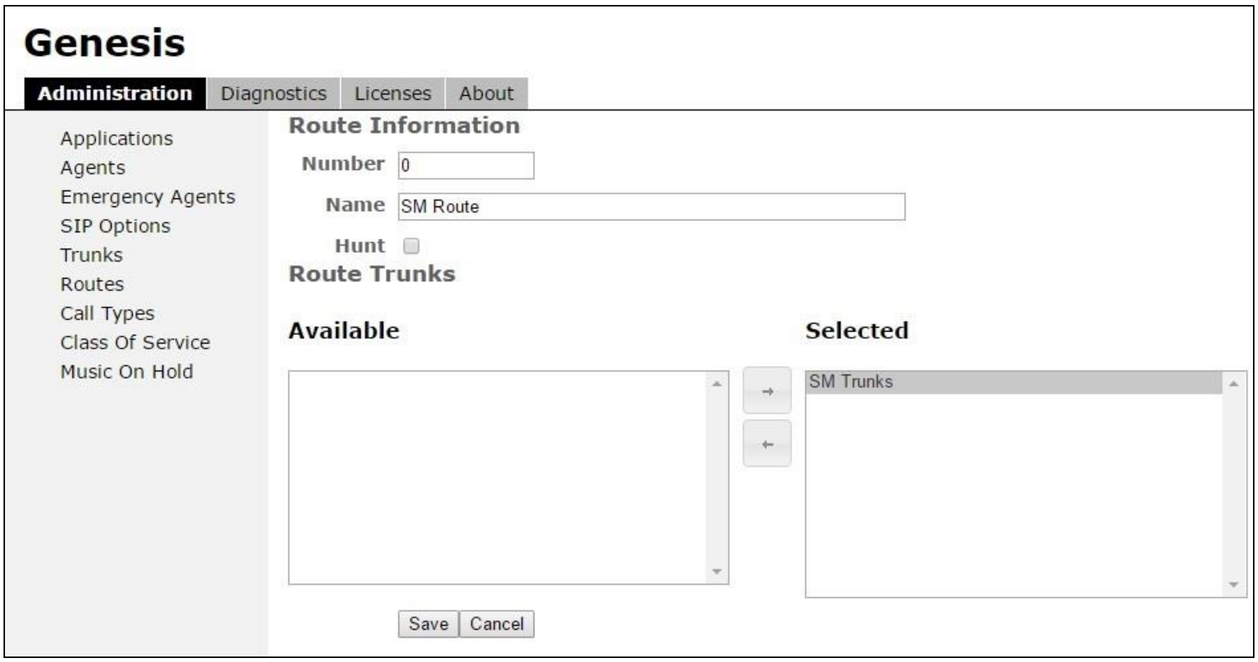


7.4. Administer Routes

Select **Routes** in the left pane, followed by **Create New Route** (not shown) in the updated right pane, to display the **Route Information** screen below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Number:** An available route number.
- **Name:** A descriptive name.

In the **Route Trunks** sub-section, select the trunk from **Section 7.3** under **Available** and move to **Selected**, as shown below.

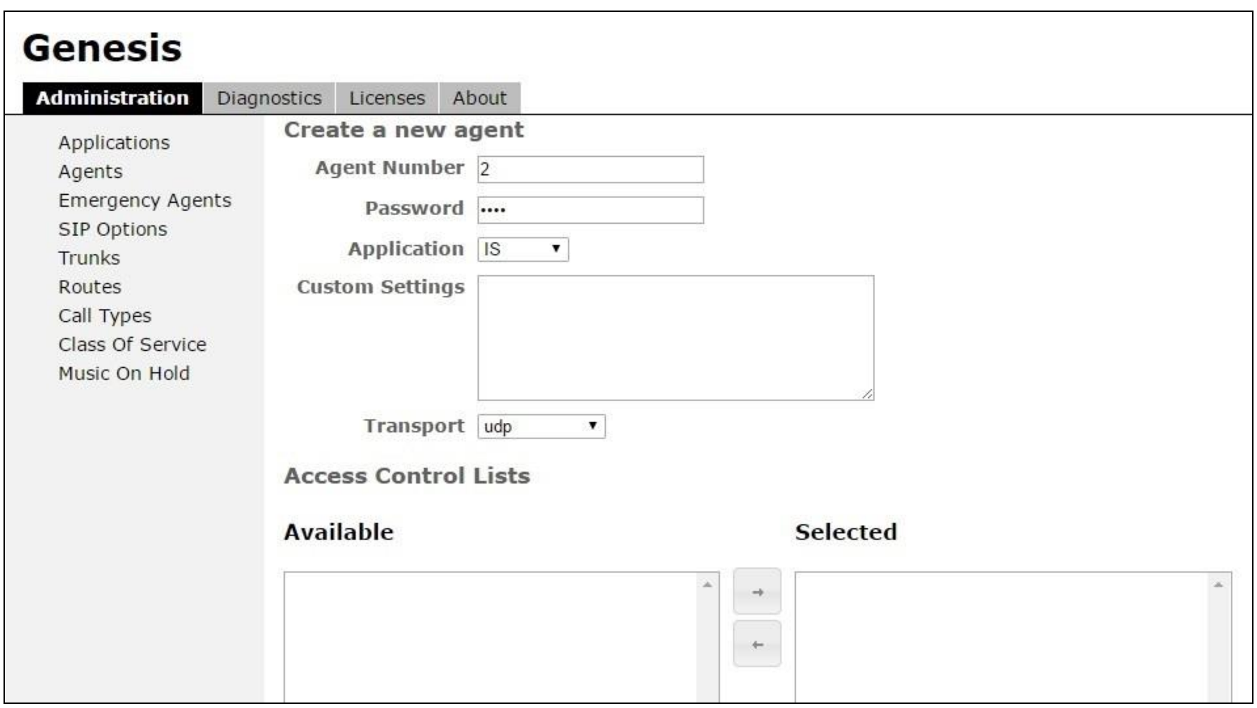


7.5. Administer Agents

Select **Agents** in the left pane, to display the **Agents** screen. One agent is needed for each operator user, and by default the first agent is automatically created, as shown below. To create additional agents, select **Create New**.

The **Create a new agent** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

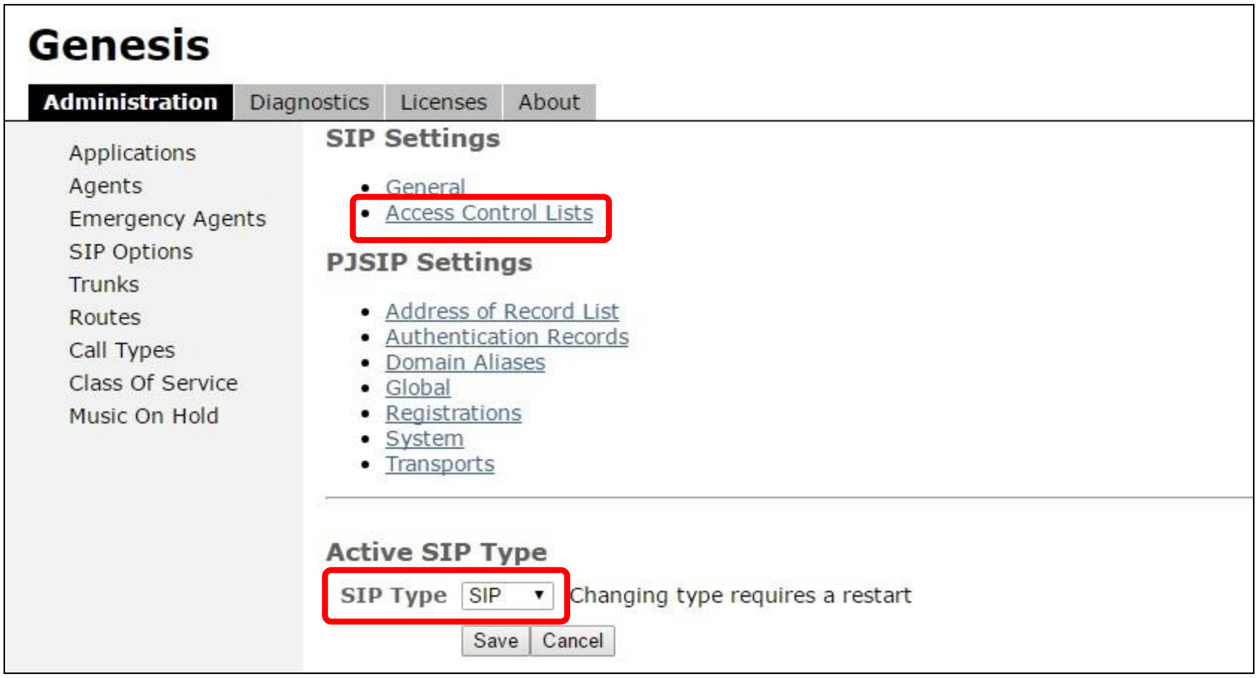
- **Agent Number:** An available agent number.
- **Password:** A desired password.
- **Application:** Select the application name from **Section 7.2**.
- **Transport:** “udp”



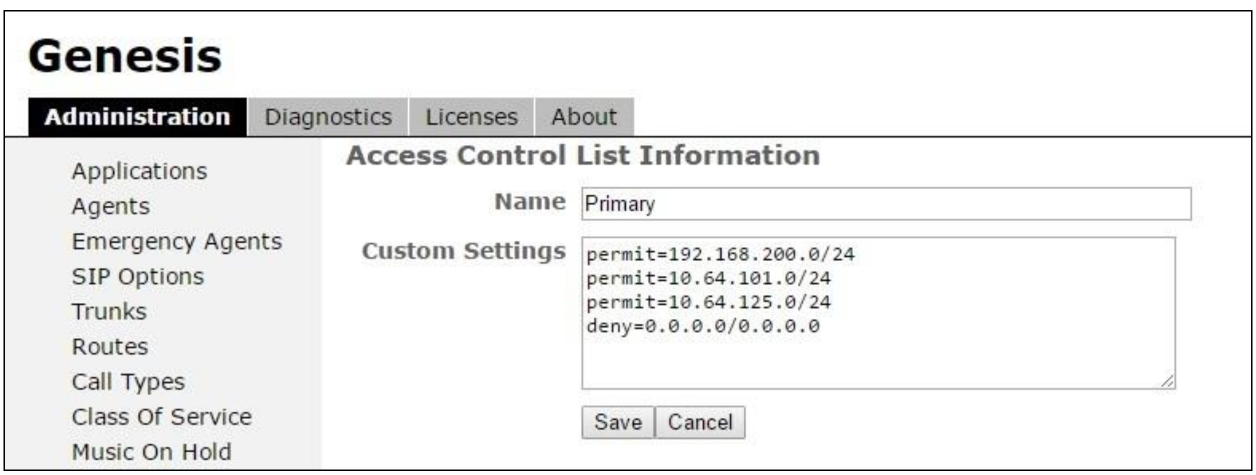
7.6. Administer Access Control Lists

Select **SIP Options** in the left pane, followed by **Access Control Lists** in the updated right pane, to display the screen below. Make certain **SIP Type** is set to "SIP", as shown below.

Select **Access Control Lists**.



The **Access Control List Information** screen is displayed. Enter a desired **Name**, and create a **permit** entry for each network subnet from **Section 3**, and create a generic **deny** entry as shown below.



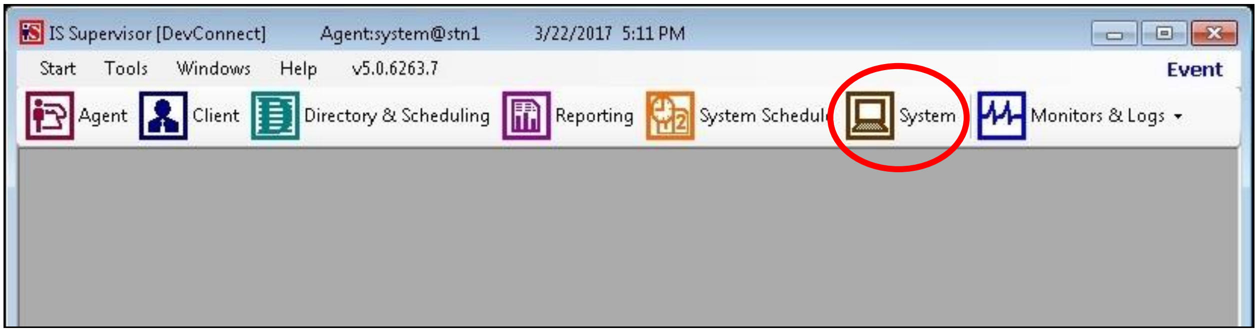
7.7. Launch Intelligent Series Supervisor

From the supervisor PC, double-click on the Intelligent Series Supervisor shortcut icon shown below, which was created as part of Intelligent Series Supervisor installation.

The **Supervisor Login** screen is displayed. Log in using the appropriate credentials.

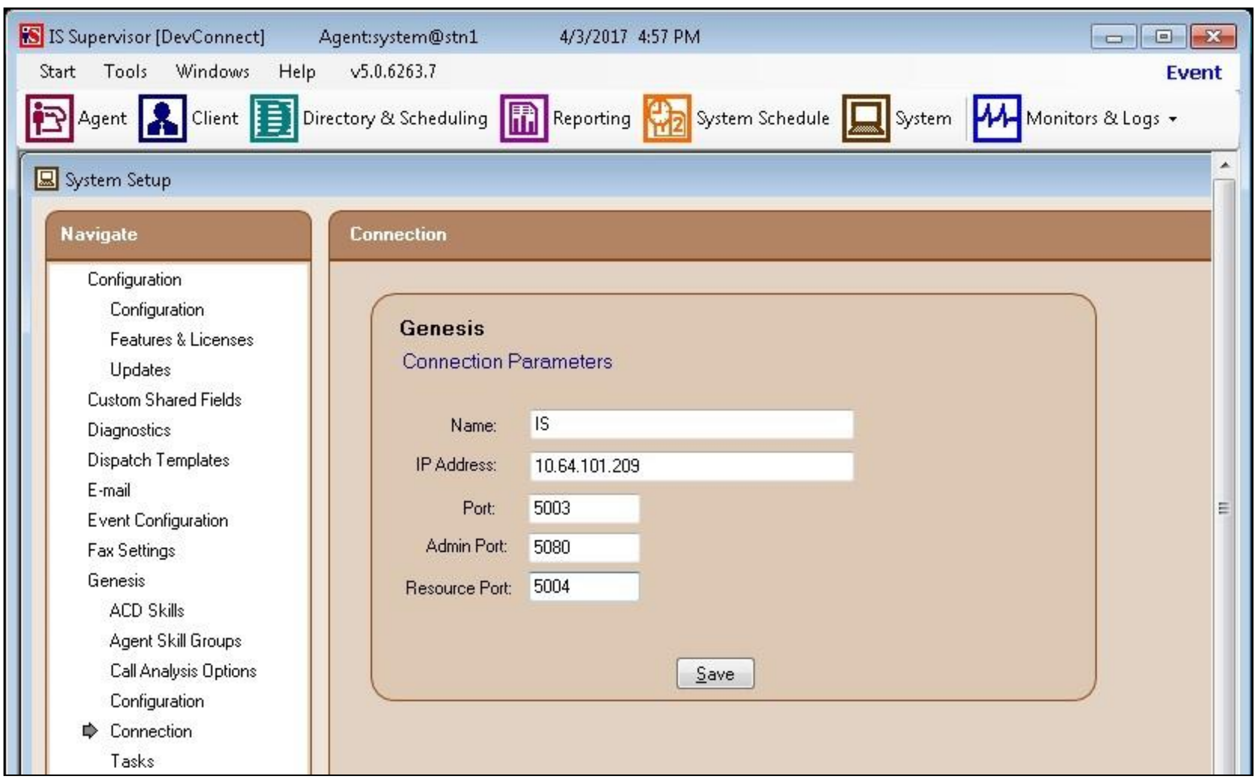
7.8. Administer IS System

The **IS Supervisor** screen is displayed. Select **System** from the top of the screen.



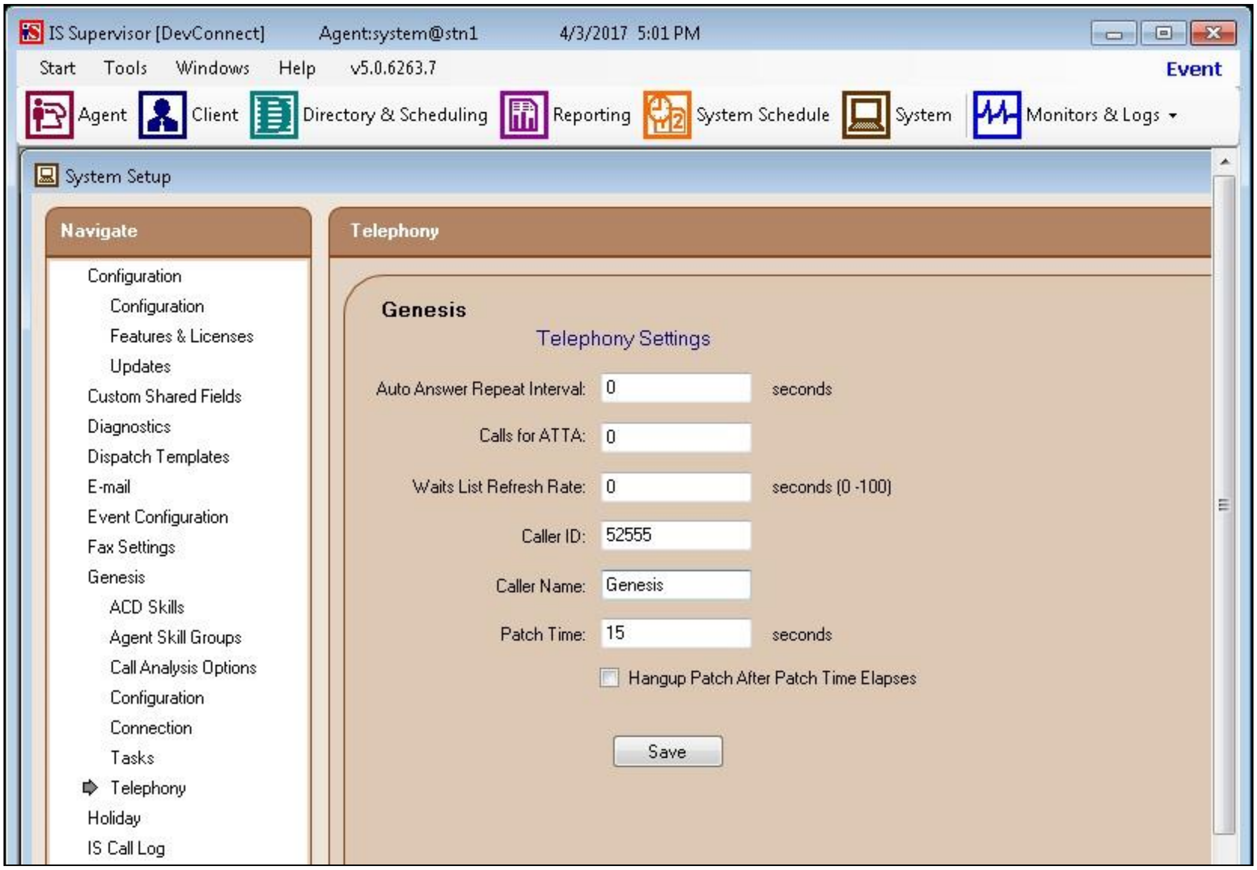
The screen is updated with **System Setup** displayed in the lower pane. Select **Genesis → Connection** from the left pane, to display the **Connection** screen in the right pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** Enter the application name from **Section 7.2**.
- **IP Address:** IP address of the Genesis Telephony Server.
- **Port:** "5003"
- **Admin Port:** "5080"
- **Resource Port:** "5004"



Select **Genesis → Telephony** from the left pane, to display the **Telephony** screen in the right pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

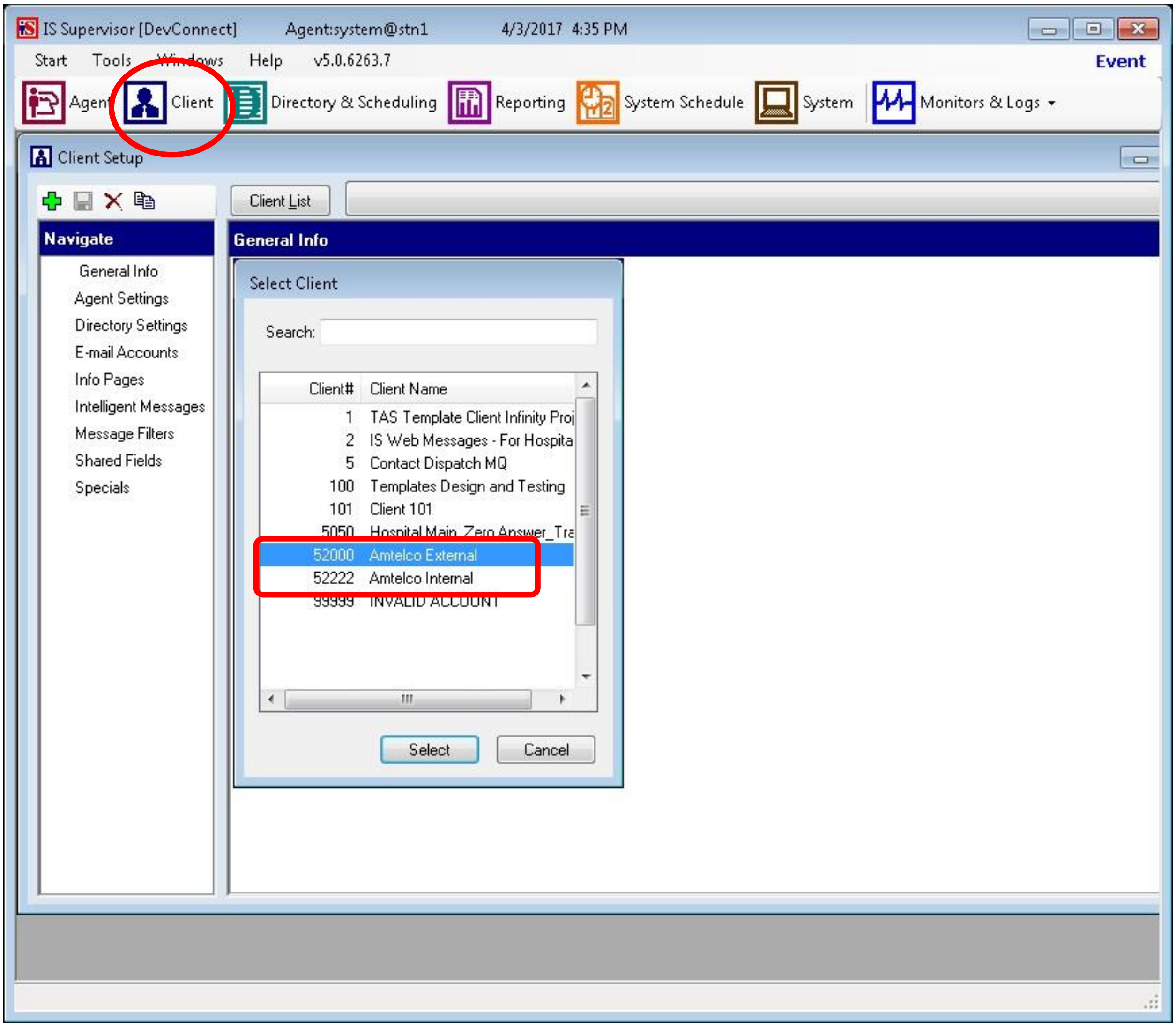
- **Caller ID:** The desired calling party extension to use for outbound calls.
- **Caller Name:** The desired calling party name to use for outbound calls.



7.9. Administer IS Client

Select **Client** from the top of the screen. The screen is updated with **Client Setup** displayed in the lower pane.

Follow reference [3] to create desired client entries to associate with called numbers for the customer network. In the compliance testing, calls from the PSTN will be routed with digits 52000 to Genesis, and calls from internal users on Communication Manager will be routed with digits 52222 to Genesis. Therefore two clients were created, as shown below.

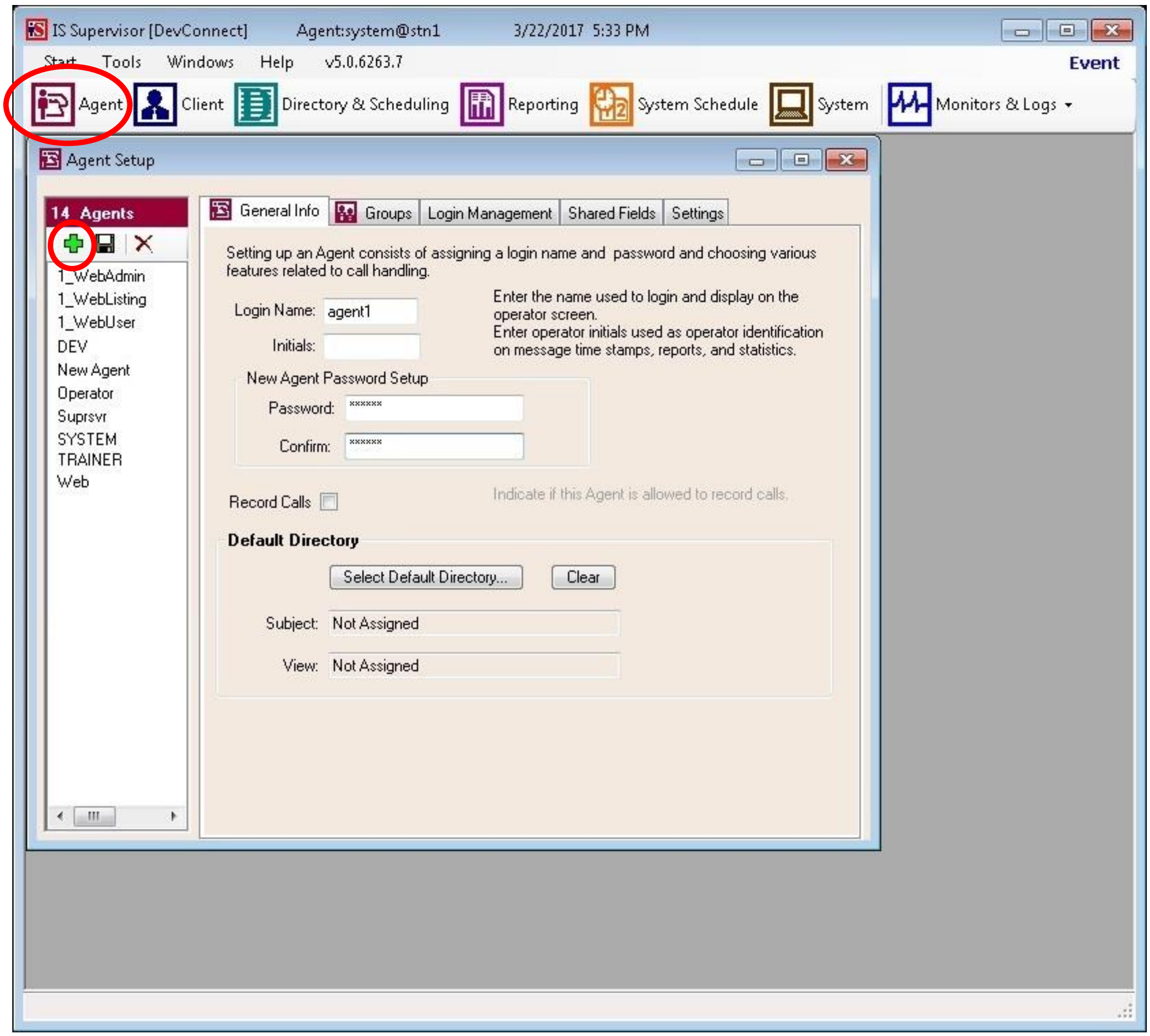


7.10. Administer IS Agent

Select **Agent** from the top of the screen. The screen is updated with **Agent Setup** displayed in the lower pane. Click on the **New Agent** icon in the left pane to create a new agent entry.

The **General Info** tab is displayed. For **Login Name**, **Password**, and **Confirm**, enter desired values. Retain the default values in the remaining fields.

One agent is needed for each operator user, and two agents were created in the compliance testing.



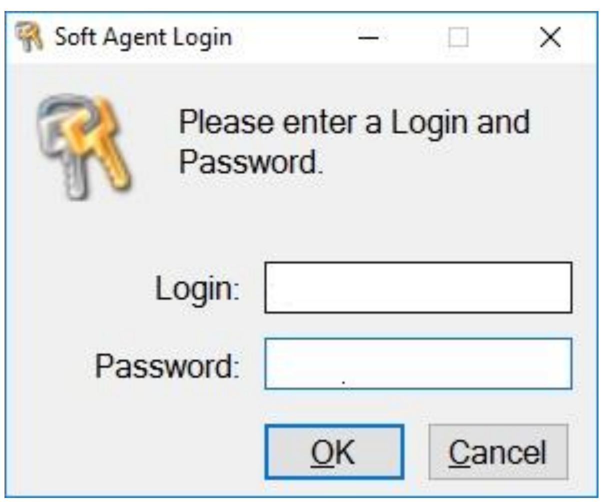
7.11. Restart IS Service

From the Intelligent Series Server, select **Start → Control Panel → Administrative Tools → Services** to display the **Services** screen. Locate and restart the **Amtelco Intelligent Series** service, as shown below.

7.12. Launch Intelligent Series Soft Agent

From an operator PC, double-click on the Soft Agent shortcut icon shown below, which was created as part of the Intelligent Series Soft Agent installation.

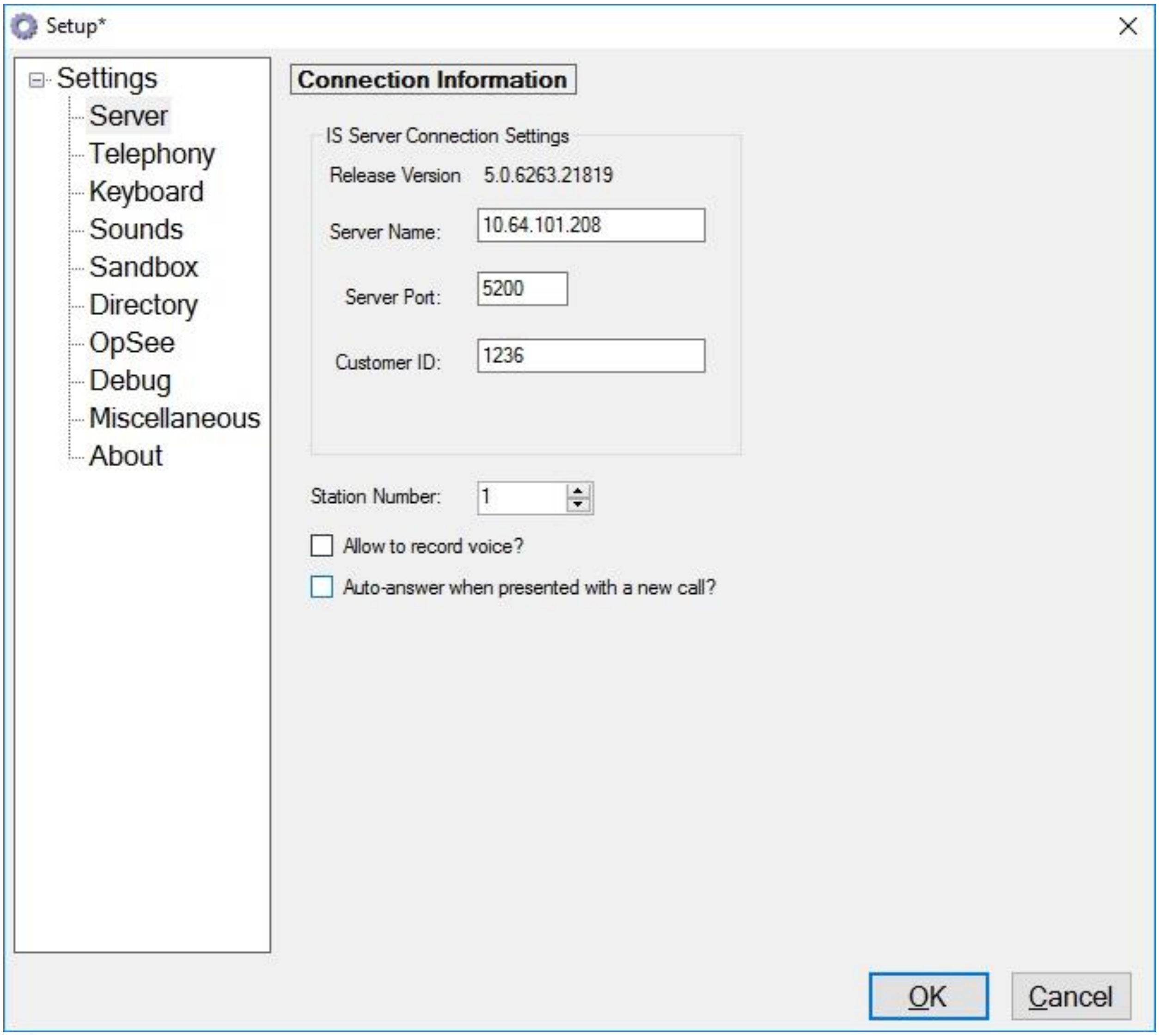
The **Soft Agent Login** screen is displayed. Press the **Ctrl** and **F12** keys together to enter setup.



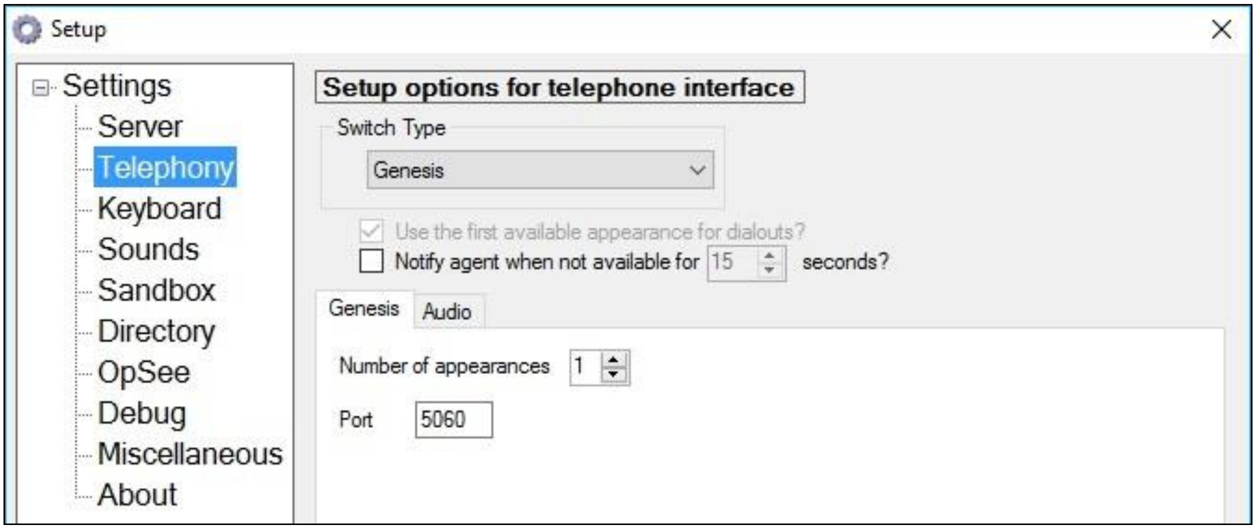
7.13. Administer Setup

The **Setup** screen below is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

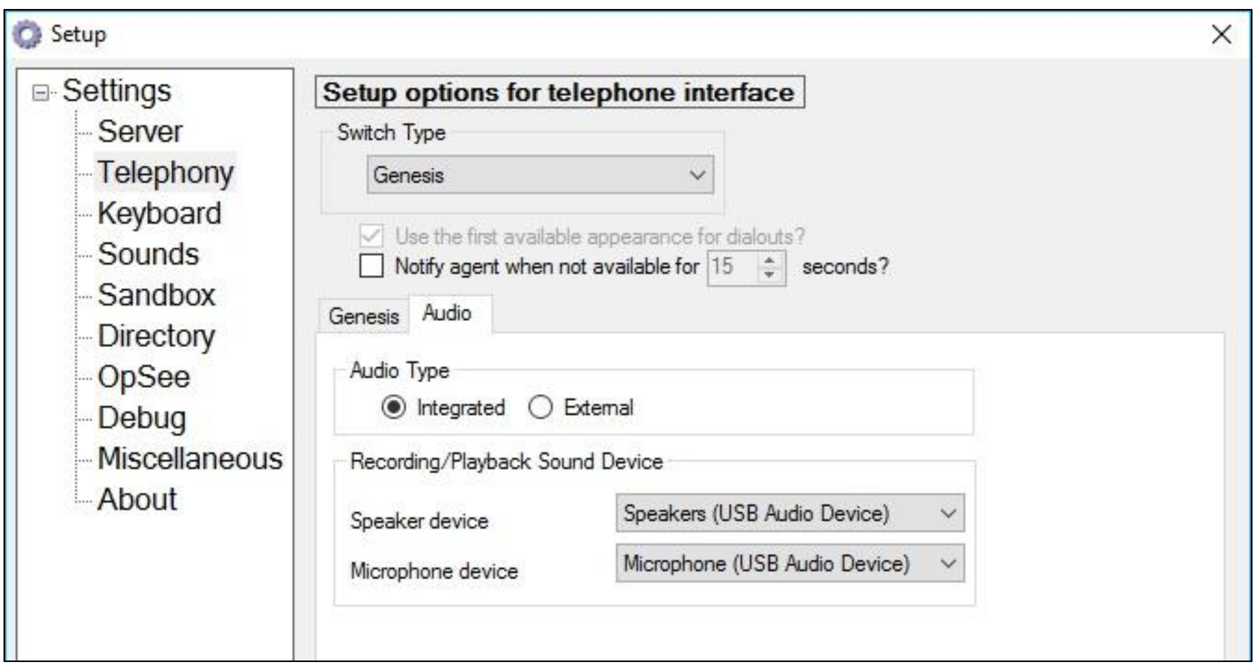
- **Server Name:** IP address of the Intelligent Series Server.
- **Server Port:** "5200"
- **Customer ID:** The unique customer ID assigned by Amtelco, in this case "1236".
- **Station Number:** An available station number, in this case "1".



Select **Settings → Telephony** from the left pane, to display the screen below. For **Switch Type**, select "Genesis". Select the desired **Number of appearances**, and enter "5060" for **Port**.

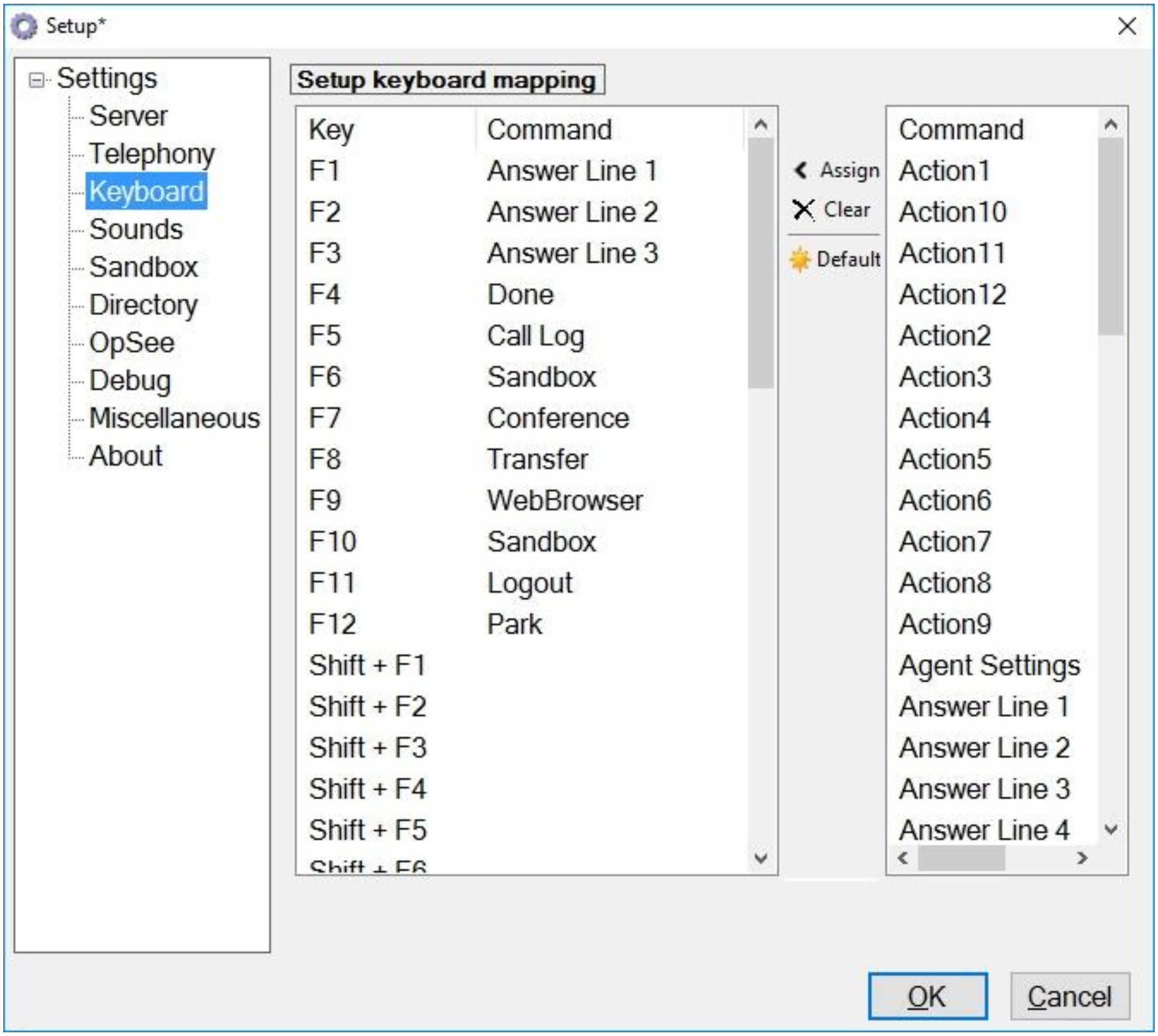


Select the **Audio** tab in the right pane, to display the screen below. For **Audio Type**, select **Integrated**. For **Speaker device** and **Microphone device**, select the applicable devices, as shown below.



Select **Settings → Keyboard** from the left pane, to display the screen below. Follow reference [3] to set the desired keyboard mapping for the agent. The setting used in the compliance testing is shown below.

Repeat **Section 7.12** and **Section 7.13** for each operator in **Section 3**. In the compliance testing, two operators were configured.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Genesis.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 52

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0052/001 T00146   in-service/idle    no
0052/002 T00147   in-service/idle    no
0052/003 T00148   in-service/idle    no
0052/004 T00149   in-service/idle    no
0052/005 T00150   in-service/idle    no
0052/006 T00151   in-service/idle    no
0052/007 T00152   in-service/idle    no
0052/008 T00153   in-service/idle    no
0052/009 T00154   in-service/idle    no
0052/010 T00155   in-service/idle    no
```

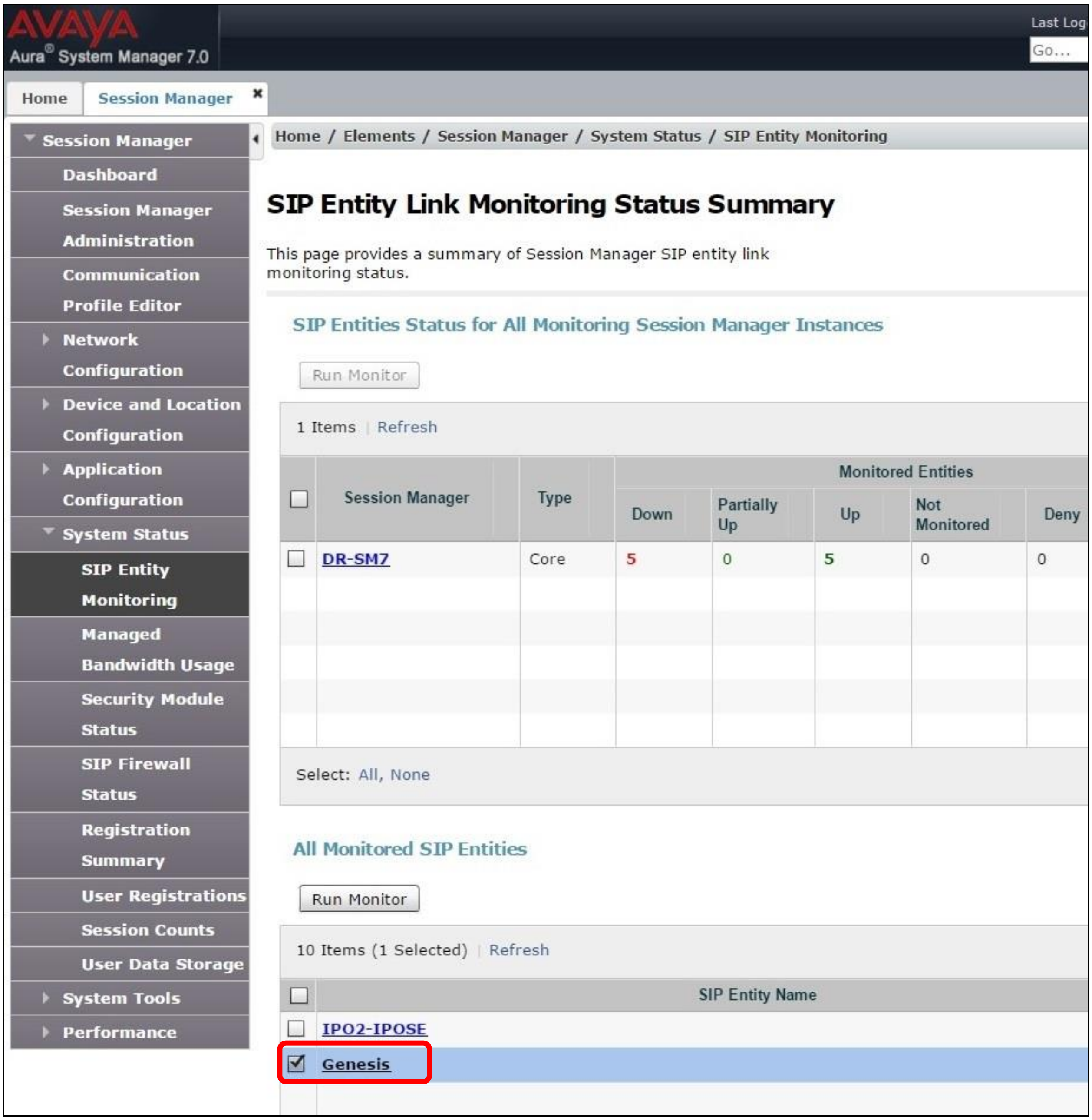
Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is "in-service", as shown below.

```
status signaling-group 52
                        STATUS SIGNALING GROUP

        Group ID: 52
      Group Type: sip

     Group State: in-service
```

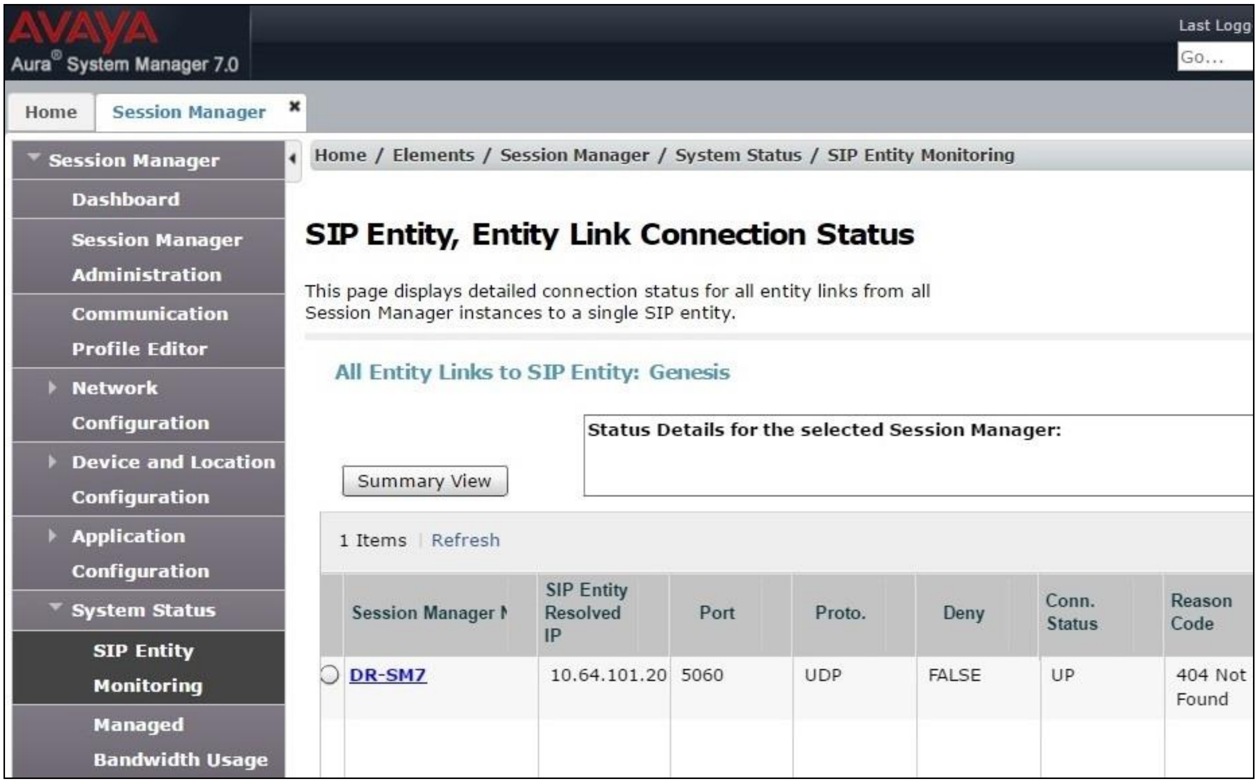
8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements** → **Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the Genesis entity name from **Section 6.3.1**.



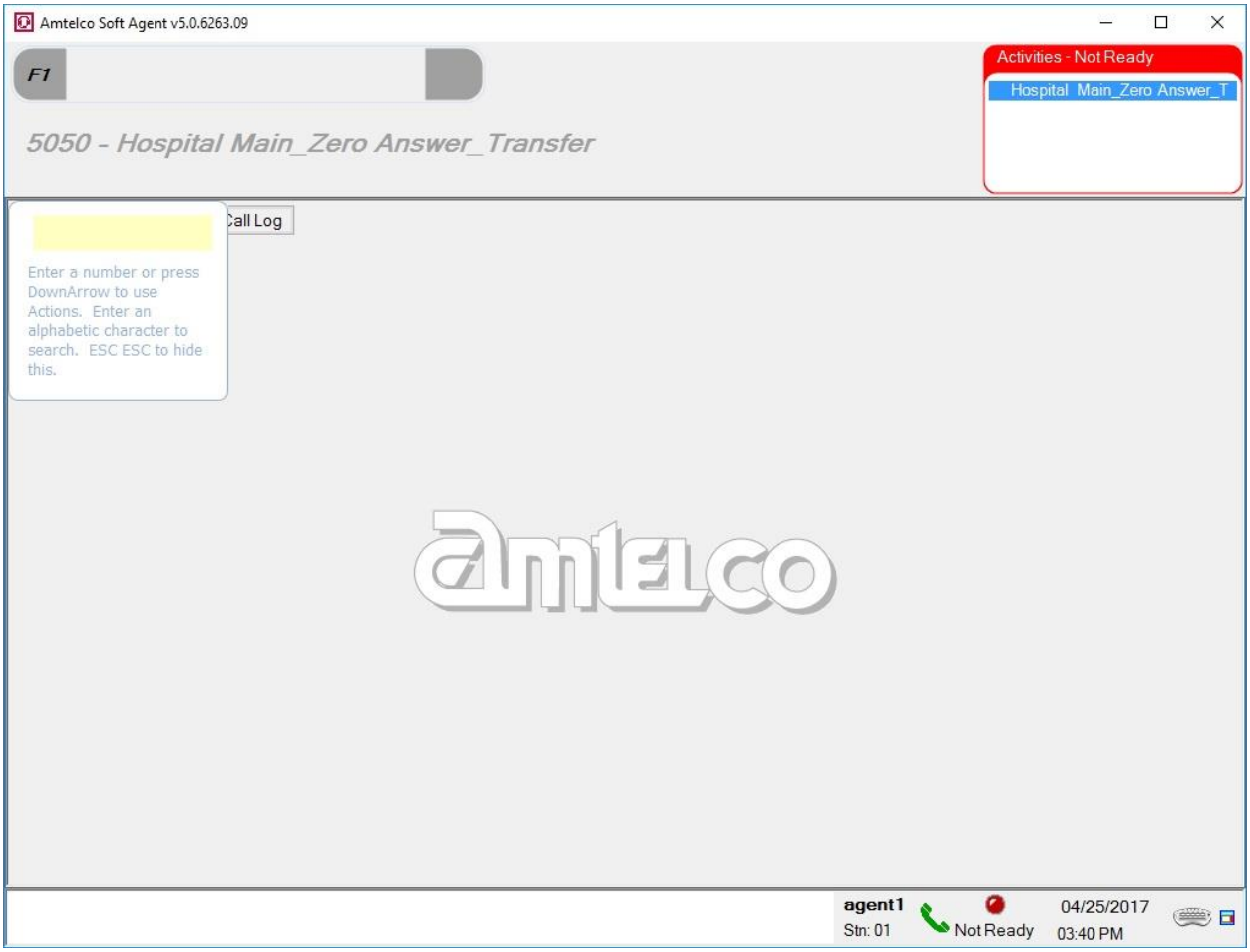
The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "UP", as shown below.



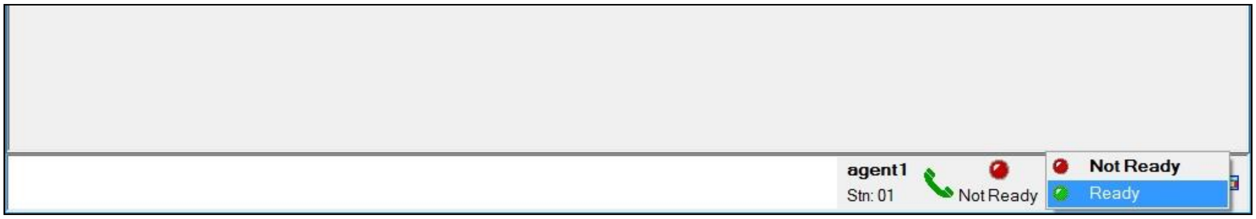
8.3. Verify Amtelco Genesis Intelligent Series

From the operator PC, follow the procedure in **Section 7.12** to launch the Intelligent Series Soft Agent and log in with the appropriate credentials from **Section 7.10**.

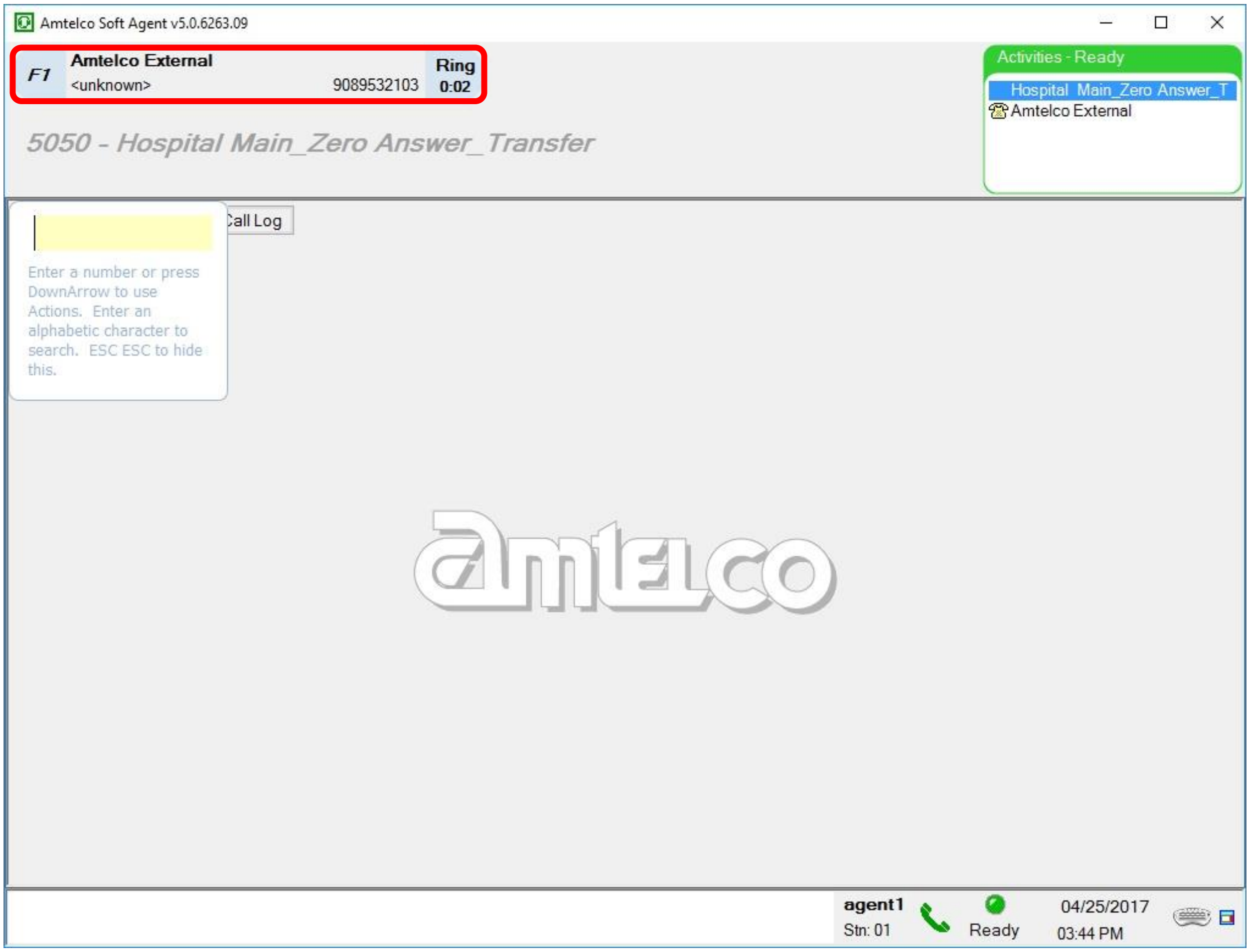
The **Amtelco Soft Agent** screen below is displayed.



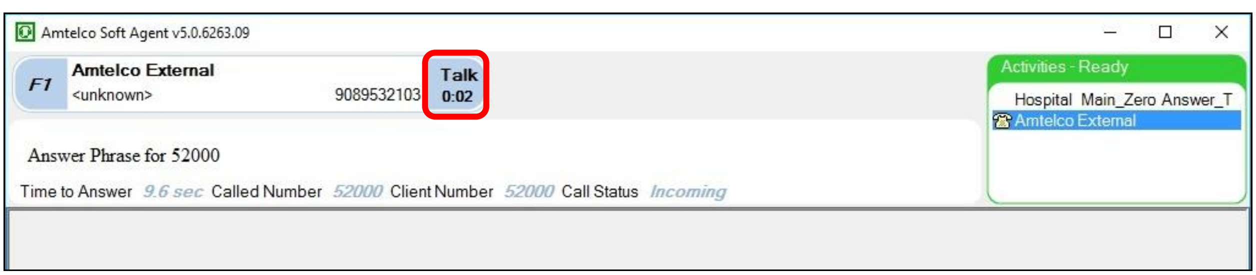
In the lower right portion of the screen, right click on **Not Ready** and select **Ready**.



Make an incoming call from PSTN to reach Genesis. Verify that the call is ringing at the available operator, and that the operator screen is updated to reflect a ringing call along with the calling party number and the called client name, as shown below.   In this case, the calling party number is **9089532103**, and the called client name is **Amtelco External**. Press the **F1** key or click in the applicable call line area highlighted below to answer the call.



Verify that the operator is connected to the PSTN with two-way talk paths. Also verify that the operator screen is updated to reflect the **Talk** state, as shown below.



9. Conclusion

These Application Notes describe the configuration steps required for Amtelco Genesis Intelligent Series to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2.1, August 2016, available at http://support.avaya.com.

2. *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016, available at http://support.avaya.com.

3. *Administering Avaya Aura® System Manager for Release 7.0.1*, Release 7.0.1, Issue 4, April 2017, available at http://support.avaya.com.

4. *System Setup Supervisor Reference Guide*, March 2017, available at https://service.amtelco.com/doclib/library.htm.

5. *Soft Agent User Reference Guide,* May 2016, available at https://service.amtelco.com/doclib/library.htm.

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.