Avaya Solution & Interoperability Test Lab

Application Notes for Amtelco Infinity Intelligent SIP Attendant Console with Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Amtelco Infinity SIP Attendant Console to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. Amtelco Infinity SIP Attendant Console is a SIP-based soft phone solution that provides phone and operator state controls during call handling.

In the compliance testing, Amtelco Infinity SIP Attendant Console used the SIP trunks interface from Avaya Aura® Session Manager to provide attendant consoles for Avaya Aura® Communication Manager.

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 Infinity SIP Attendant Console to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. Amtelco Infinity SIP Attendant Console is a SIP-based soft phone solution that provides phone and operator state controls during call handling.

In the compliance testing, Amtelco Infinity SIP Attendant Console used the SIP trunks interface from Avaya Aura® Session Manager to provide attendant consoles for Avaya Aura® Communication Manager.

The Amtelco Infinity SIP Attendant Console solution consists of an Infinity server and attendants with desktop computers running Amtelco Infinity Telephone Agent. The Infinity server controls routing of calls to/from the attendants, and with all attendant related activities such as answer/drop calls performed from Amtelco Infinity Telephone Agent.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually with necessary attendant actions such as hold and transfer performed from the attendant desktops to verify various call scenarios. The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to the Infinity server and to the attendants.

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 G.711, G.729, codec negotiation, DTMF, hold/resume, drop, display, blind transfer, attended conference, inbound, outbound, multiple calls, and multiple agents.

The serviceability testing focused on verifying the ability of Infinity to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connections to the Infinity server and to the attendants.

2.2. Test Results

All test cases were executed and passed. The following were observations on Infinity from the compliance testing.

- Infinity does not support media shuffling.
- The Infinity XDS VoIP card requires a Gigabit network connection.

2.3. Support

Technical support on Infinity 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**, attendants are running the Infinity Telephone Agent soft phone application on the desktops, and the administrator is running the Infinity Supervisor.

SIP trunks are used between Infinity SIP Attendant Console and Session Manager. A five digit Uniform Dial Plan was used to facilitate dialing with Infinity. Calls to extensions 52xxx are routed over the SIP trunks to Infinity. In particular, internal users on Communication Manager will dial 52222 to reach Infinity, and calls from external users will be routed with digits 52000 to Infinity. Infinity will route the received call to an available attendant, and populate the answering attendant with pertinent information for the call.

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

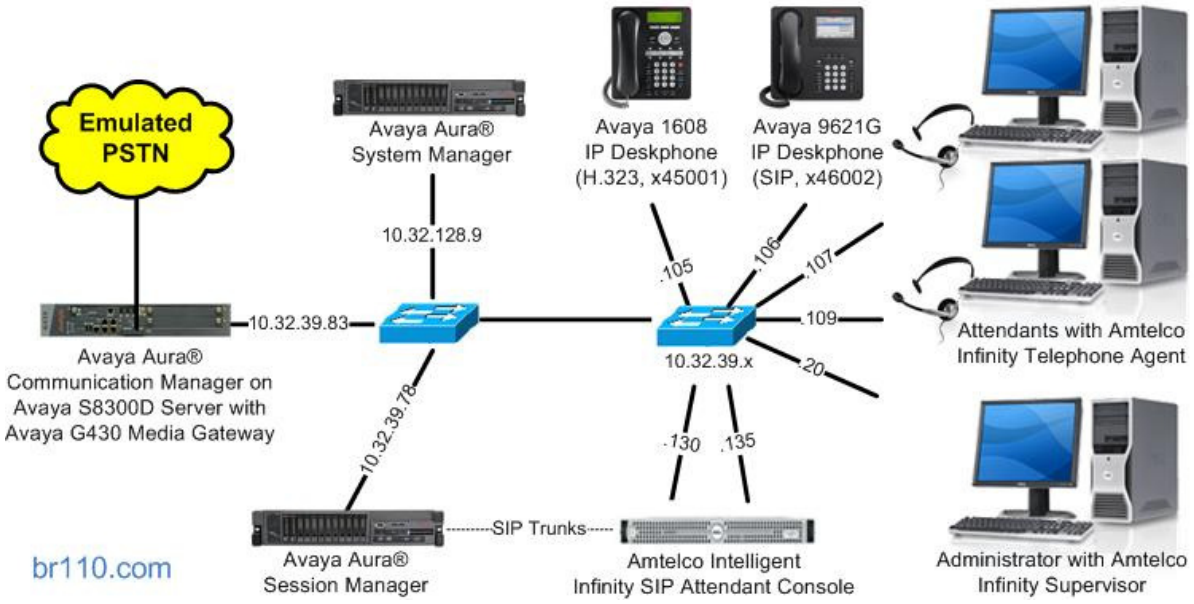


Figure 1: Compliance Testing Configuration

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 on Avaya S8300D Server with Avaya G430 Media Gateway	6.3 (R016x.03.0.124.0-20756)
Avaya Aura® Session Manager	6.3 SP2
Avaya Aura® System Manager	6.3.2
Avaya 1608 IP Deskphone (H.323)	1.3.3
Avaya 9621G IP Deskphone (SIP)	6.2.2
Amtelco Infinity Intelligent SIP Attendant Console • XDS VoIP Card	5.61.06 Patch 1 4.47
Amtelco Infinity Supervisor	5.60.0020
Amtelco Infinity Telephone Agent	5.60.4769.08

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

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

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  11
                              OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  66
           Maximum Concurrently Registered IP Stations: 2400  2
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  6
                       Maximum Administered SIP Trunks: 4000  40
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    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  20
                            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 (or Silence) on Transferred Trunk Calls? no
            DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                Automatic Circuit Assurance (ACA) Enabled? n


          Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? 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 Infinity      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


                               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:** "tcp"
- **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 Infinity.
- **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 Infinity.
- **Far-end Domain:** The applicable domain name for the network.

For **Direct IP-IP Audio Connections**, enter "n" since Infinity does not support shuffling.

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

 Group Number: 52             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
       Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: 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: NJ-SM-Sig
 Near-end Listen Port: 5052               Far-end Listen Port: 5052
                                       Far-end Network Region: 7
                                  Far-end Secondary Node Name:
Far-end Domain: br110.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? n
                                               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 Infinity      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 Infinity.

```
change ip-network-region 7                                    Page   1 of  20
                            IP NETWORK REGION
  Region: 7
Location:         Authoritative Domain: br110.com
    Name: Infinity              Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 7              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" is used by the Avaya endpoints and trunk to the PSTN.

```
change ip-network-region 7                                    Page   4 of  20

 Source Region: 7     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    7     y    NoLimit                                        n       t
 2
 3
```

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 Infinity supports the G.711 and G.729 codec variants. The codec shown below were used in the compliance testing.

```
change ip-codec-set 7                                          Page   1 of   2

                         IP Codec Set

   Codec Set: 7

   Audio        Silence      Frames   Packet
   Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.729AB          n          2        20
 3:
 4:
 5:
```

5.8. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Infinity, 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: Infinity
                                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 Infinity. 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 4 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  4              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 Infinity. Note that other methods of routing may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing 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
```

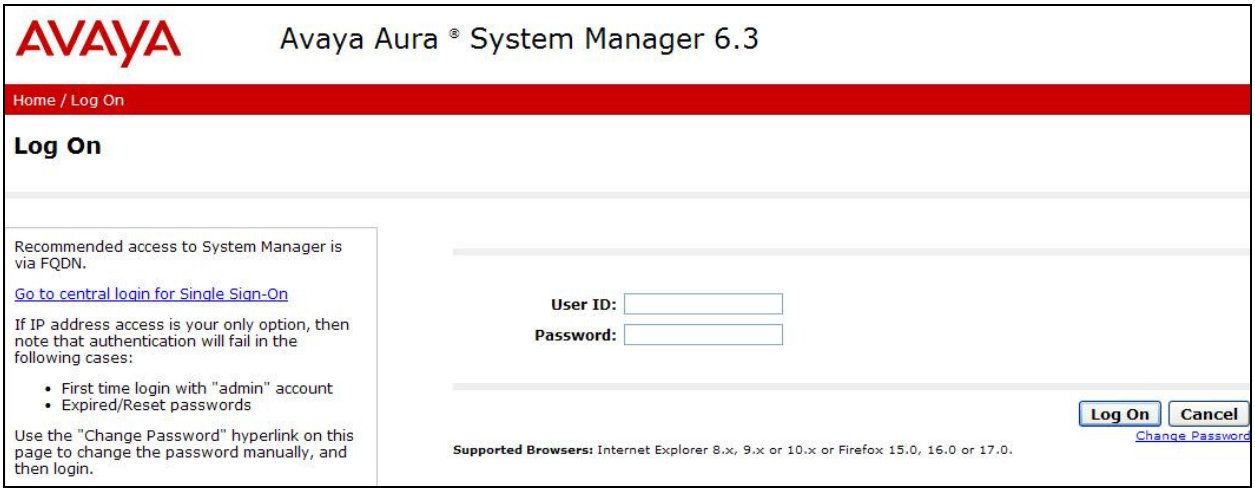
6. Configure Avaya Aura® Session Manager

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

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

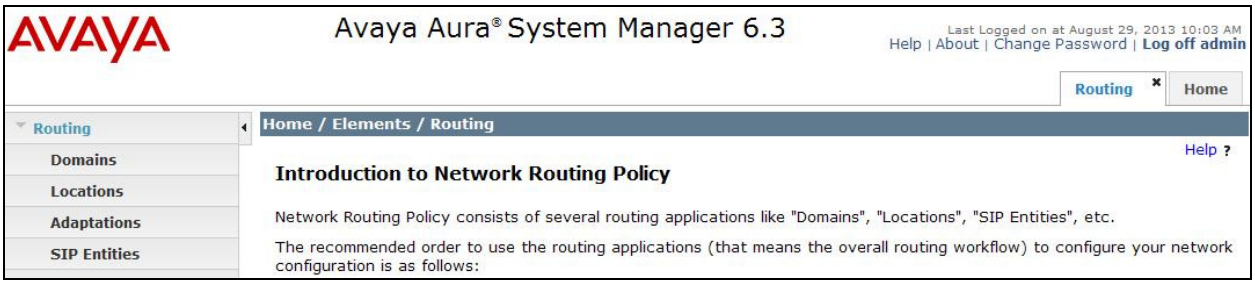
6.1. Launch System Manager

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 the System Manager server. Log in using the appropriate credentials.

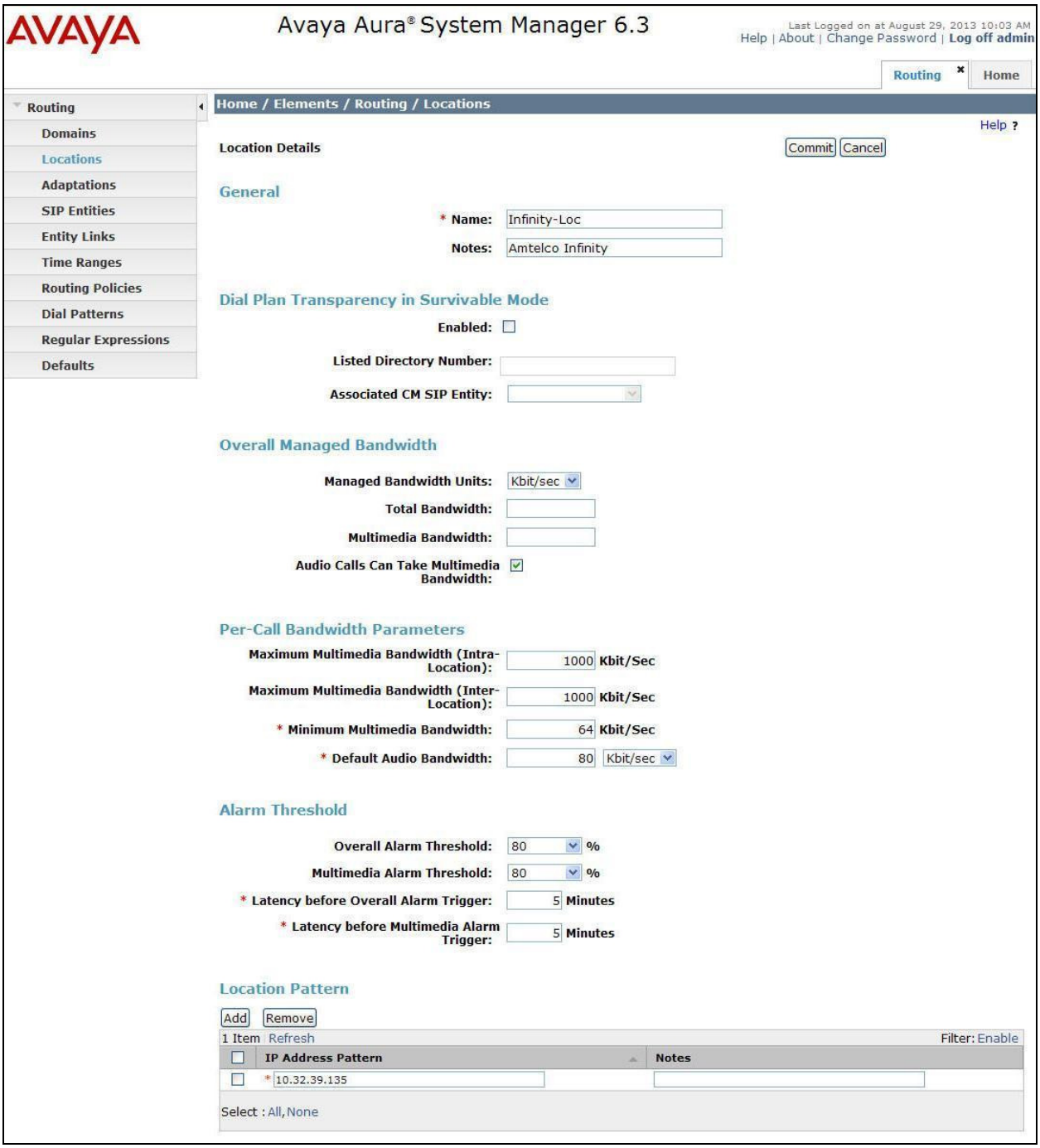


6.2. Administer Locations

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 Infinity.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. In the **Location Pattern** sub-section, click **Add** and enter the applicable **IP Address Pattern** for the Infinity IP board from **Section 7.2**, as shown below. Retain the default values in the remaining fields.

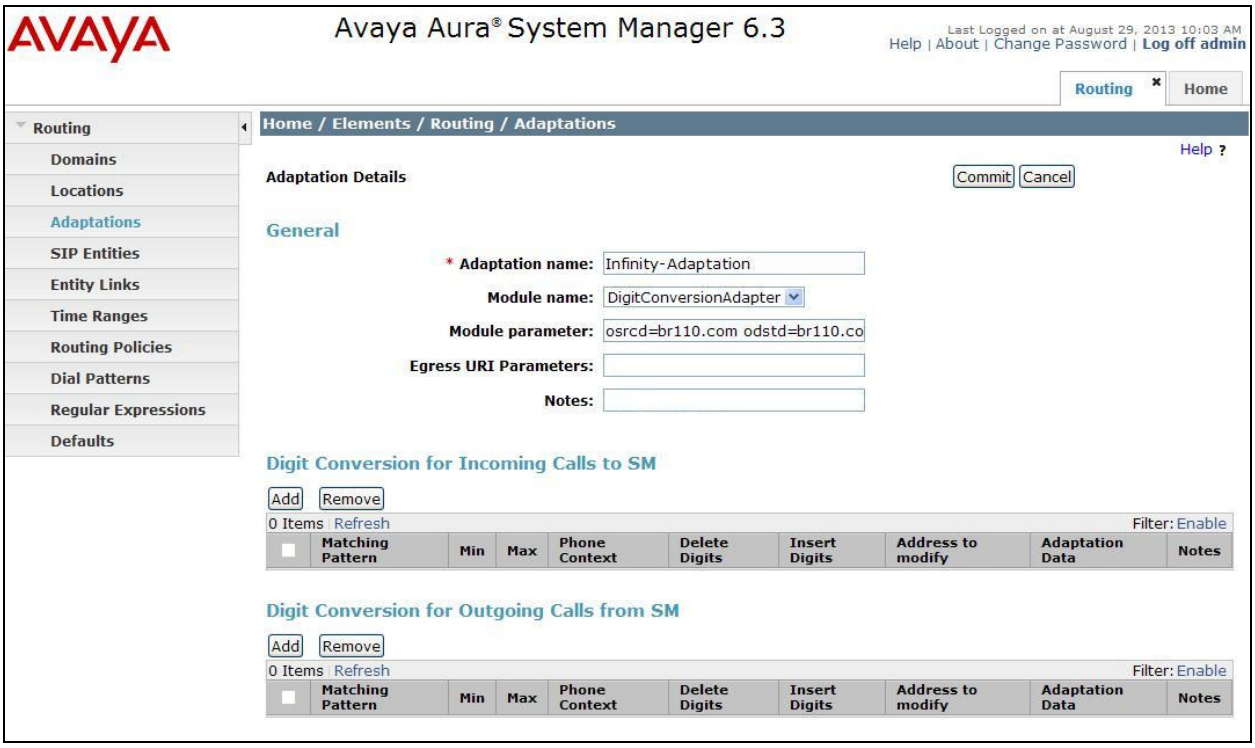


6.3. Administer Adaptations

Select **Routing → Adaptations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new adaptation for Infinity.

The **Adaptation Details** screen is displayed. In the **General** sub-section, enter a descriptive **Adaptation name**. For **Module name**, select "DigitConversionAdapter".

For **Module parameter**, enter "osrcd=br110.com odstd=br110.com, where "br110.com" is the applicable domain. This will set the source and destination domains for all incoming and outgoing calls for Infinity.



6.4. Administer SIP Entities

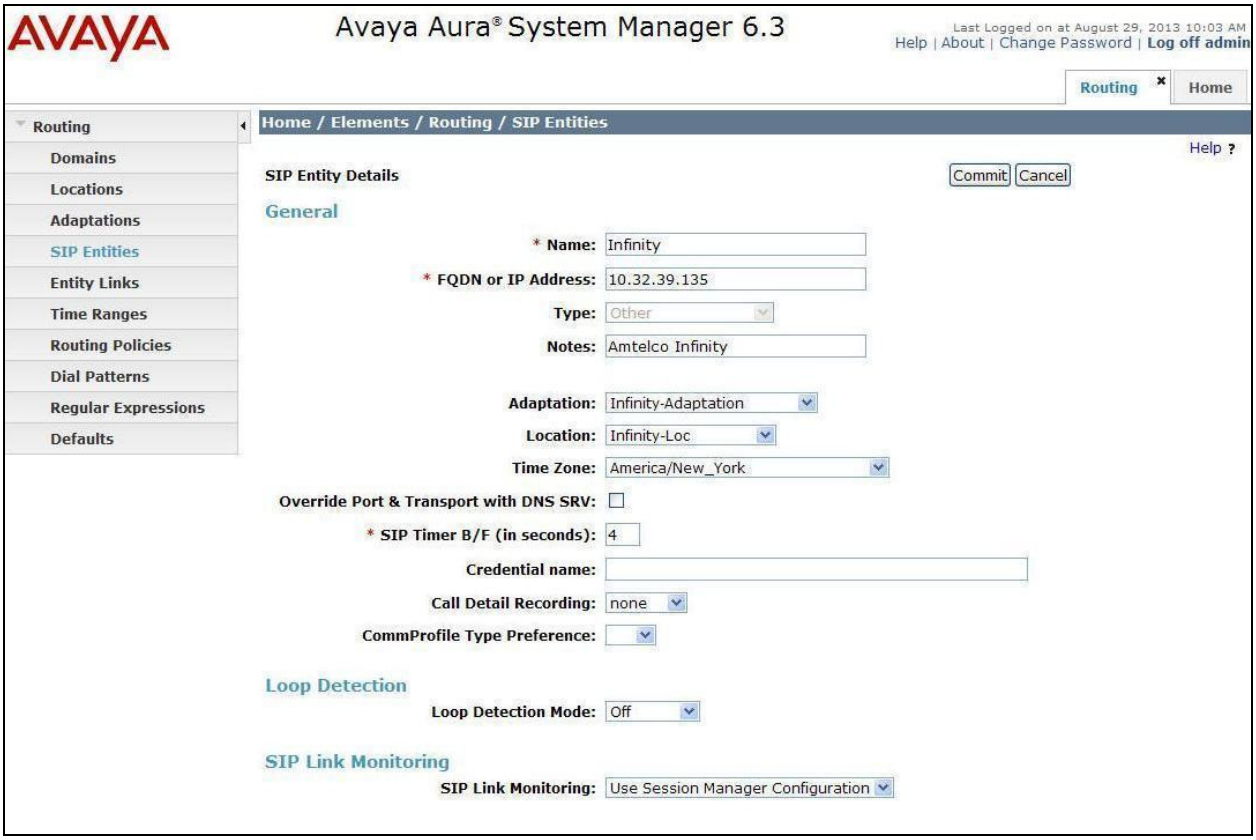
Add two new SIP entities, one for Infinity, and another for the new SIP trunks for Communication Manager.

6.4.1. SIP Entity for Infinity

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

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 Infinity IP board from **Section 7.2**.
- **Type:** "Other"
- **Adaptation:** Select the Infinity adaptation name from **Section 6.3**.
- **Location:** Select the Infinity location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

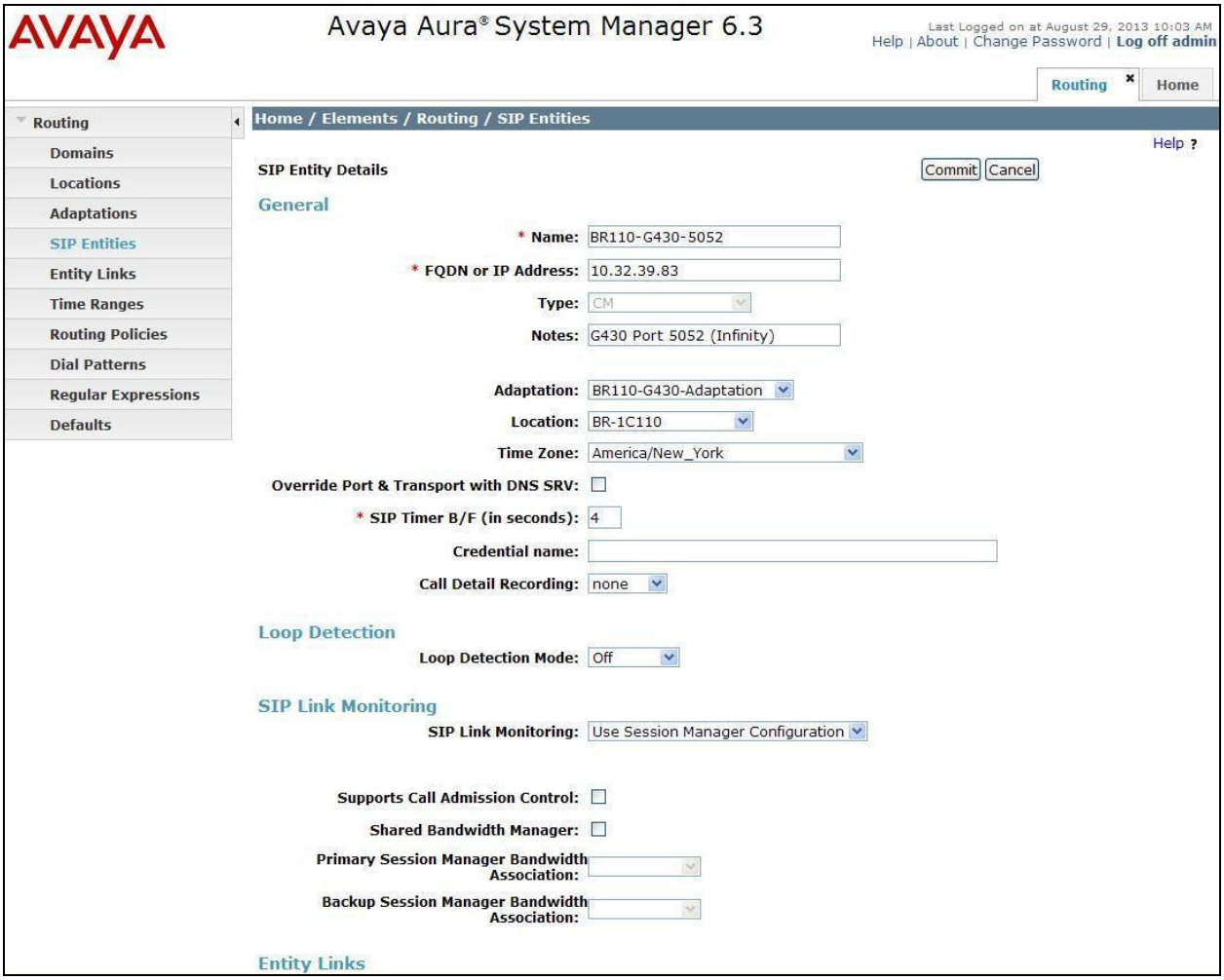


6.4.2. SIP Entity for Communication Manager

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 Infinity.

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 descriptive notes.
- **Adaptation:** Select the applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



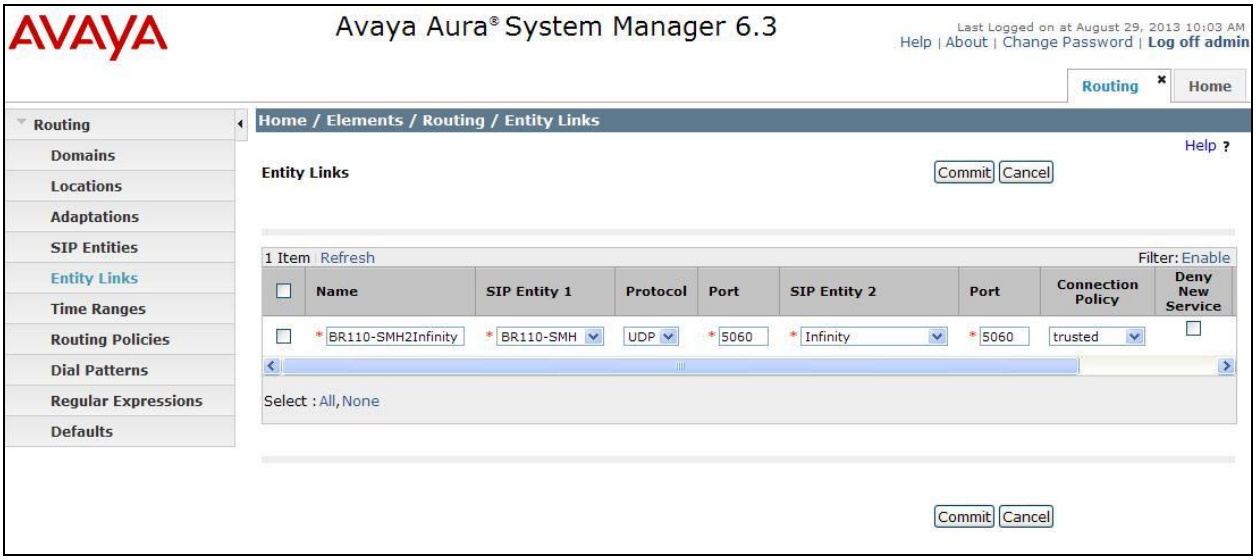
6.5. Administer Entity Links

Add two new entity links, one for Infinity and one for Communication Manager.

6.5.1. Entity Link for Infinity

Select **Routing → Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for IPC. The **Entity Links** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

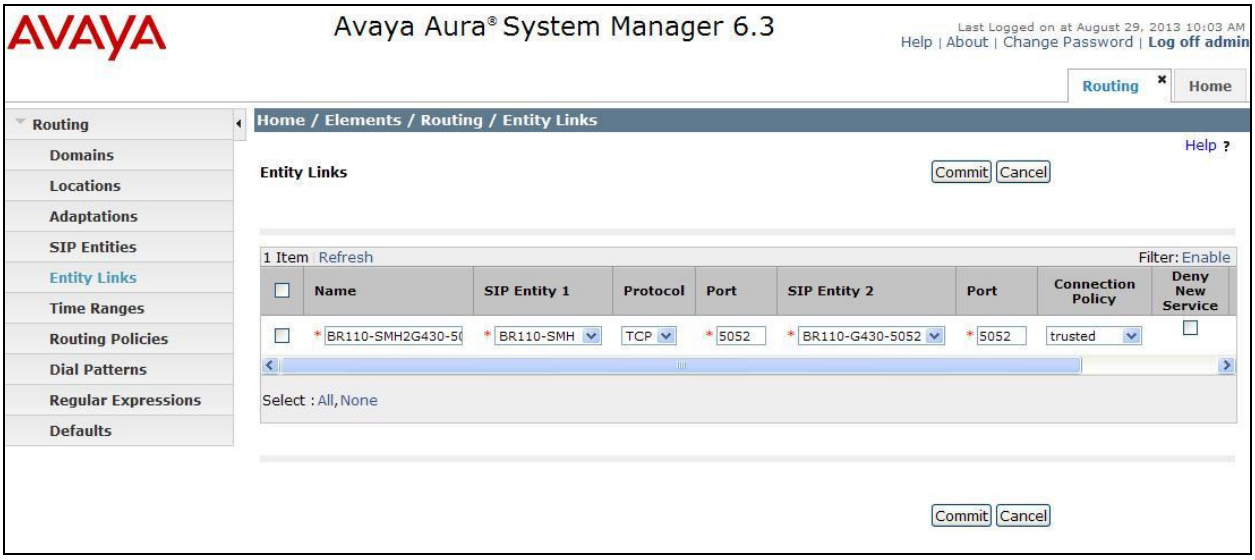
- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "BR110-SMH".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Infinity entity name from **Section 6.4.1**.
- **Port:** "5060"
- **Connection Policy:** "trusted"



6.5.2. Entity Link for Communication Manager

Select **Routing → Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for Communication Manager. The **Entity Links** screen is displayed. 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 "BR110-SMH".
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from **Section 6.4.2**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **Connection Policy:** "trusted"



6.6. Administer Routing Policies

Add two new routing policies, one for Infinity and one for Communication Manager.

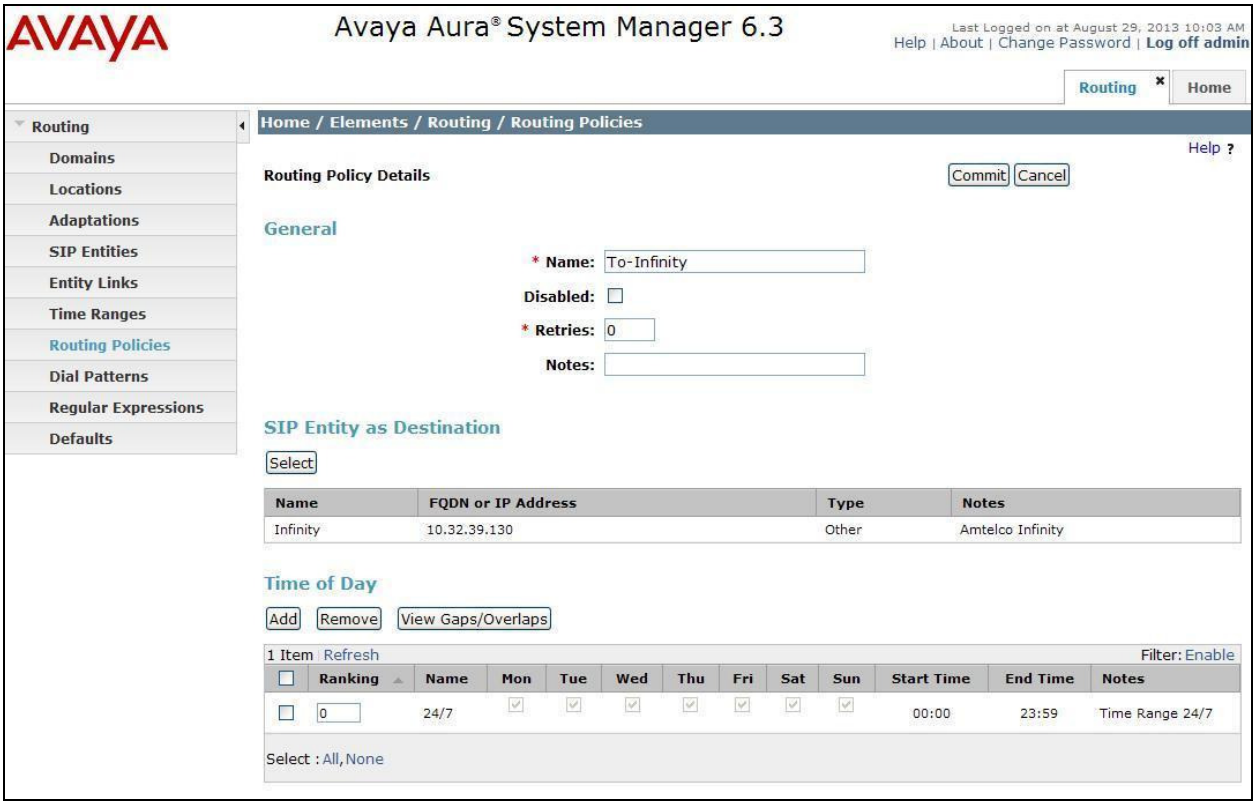
6.6.1. Routing Policy for Infinity

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

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Infinity entity name from **Section 6.4.1** in the listing (not shown).

Retain the default values in the remaining fields.



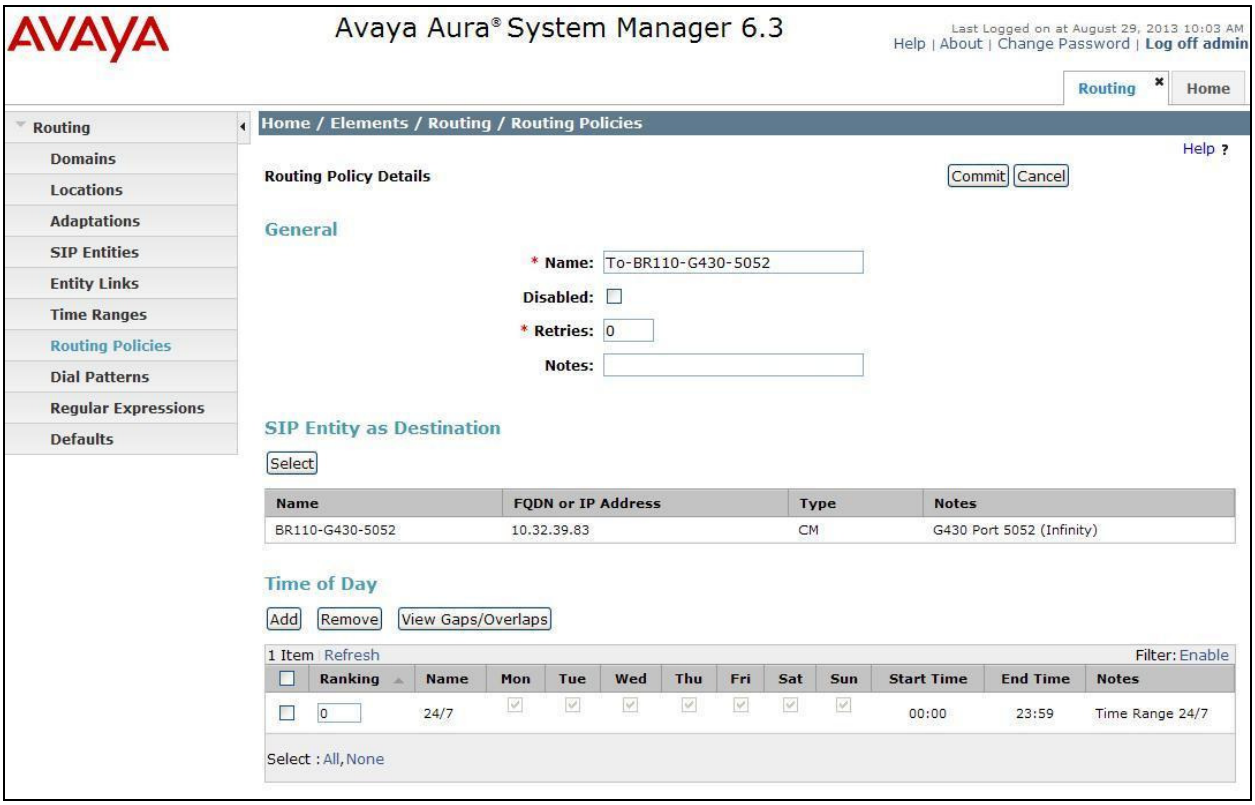
6.6.2. Routing Policy for Communication Manager

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**.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.4.2** in the listing (not shown).

Retain the default values in the remaining fields.



6.7. Administer Dial Patterns

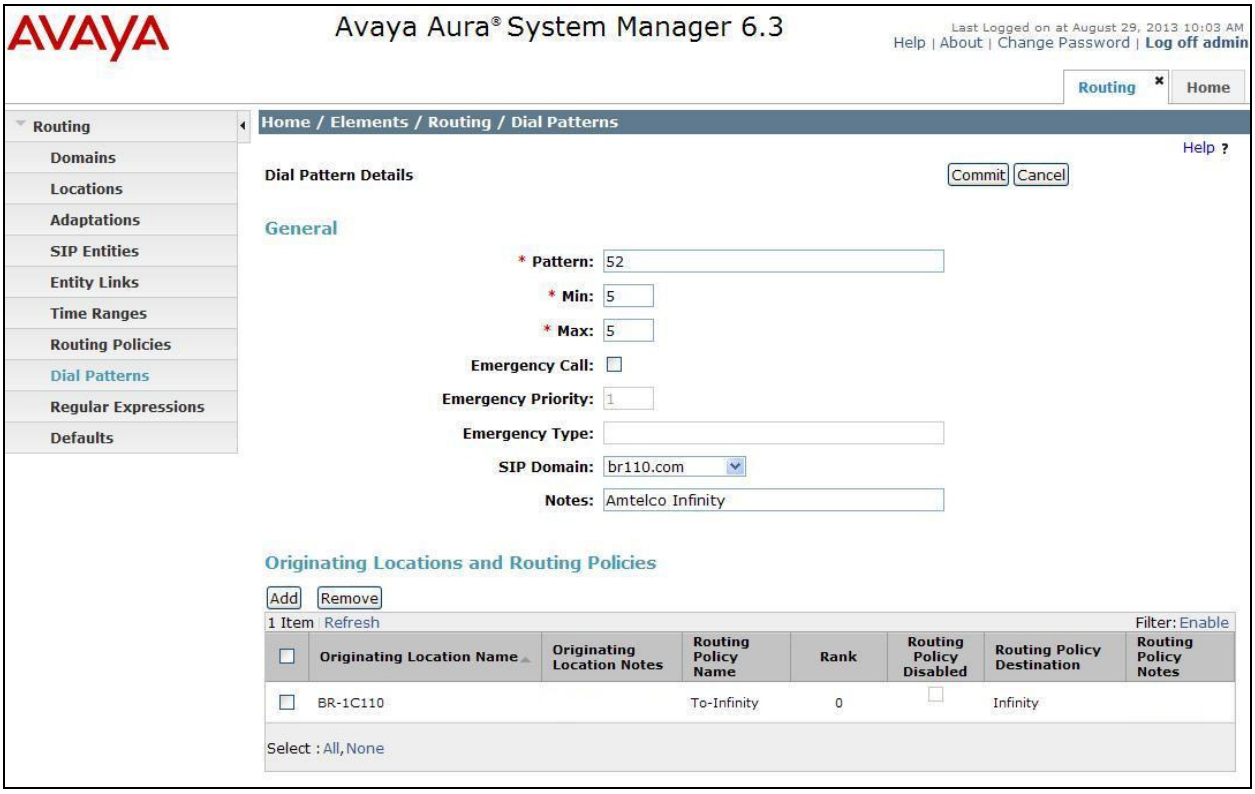
Add a new dial pattern for Infinity, and update the existing dial pattern for Communication Manager.

6.7.1. Dial Pattern for Infinity

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 Infinity. 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.
- **Min:** The minimum number of digits to be matched.
- **Max:** The maximum number of digits to be matched.
- **SIP Domain:** The signaling group domain name from **Section 5.4**.
- **Notes:** Any desired description.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching Infinity. In the compliance testing, the policy allowed for call origination from the Communication Manager location "BR-1C110", and the Infinity routing policy from **Section 6.6.1** was selected as shown below.

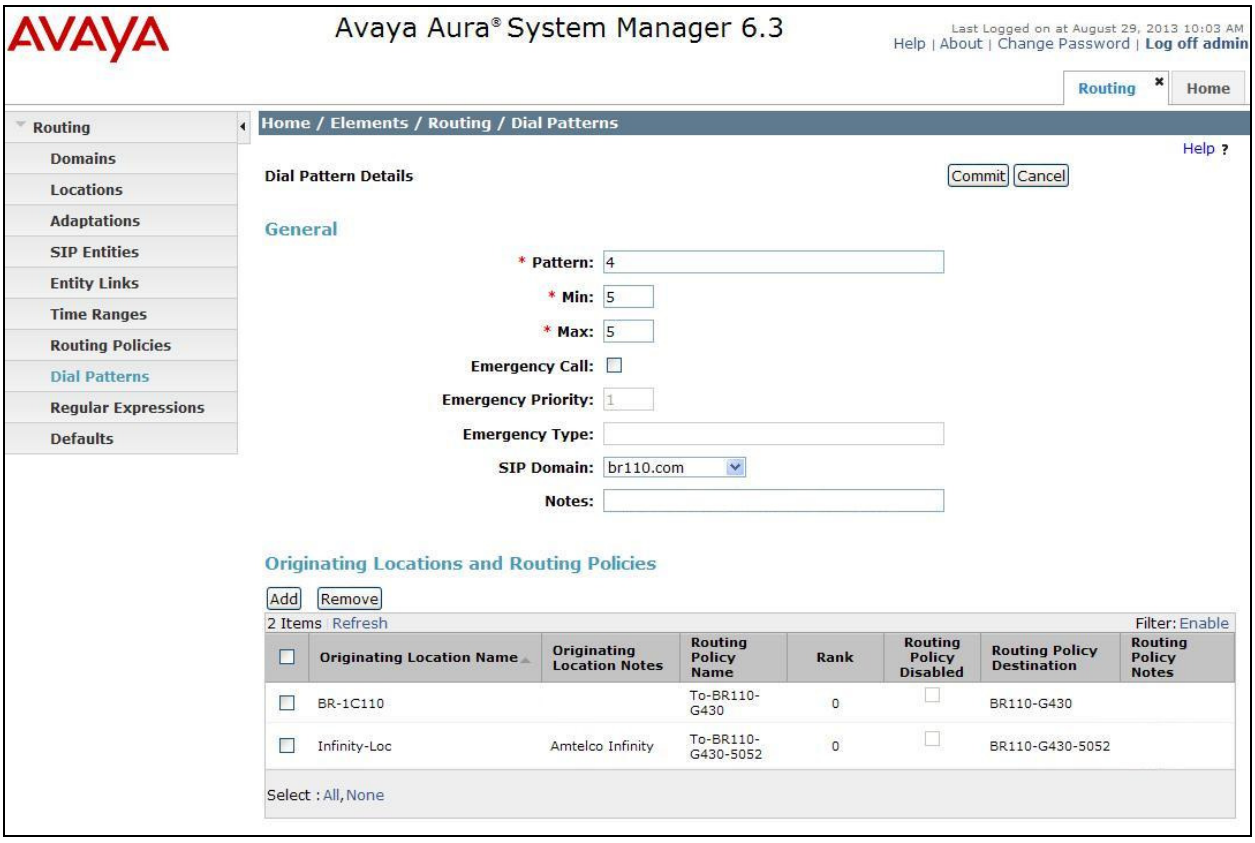


6.7.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "4" (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 Infinity. In the compliance testing, the new policy allowed for call origination from the Infinity location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.6.2** 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.



7. Configure Amtelco Infinity Intelligent SIP Attendant Console

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

- Launch Infinity Supervisor
- Administer billing number and board settings
- Administer SIP route
- Administer clients
- Administer system settings

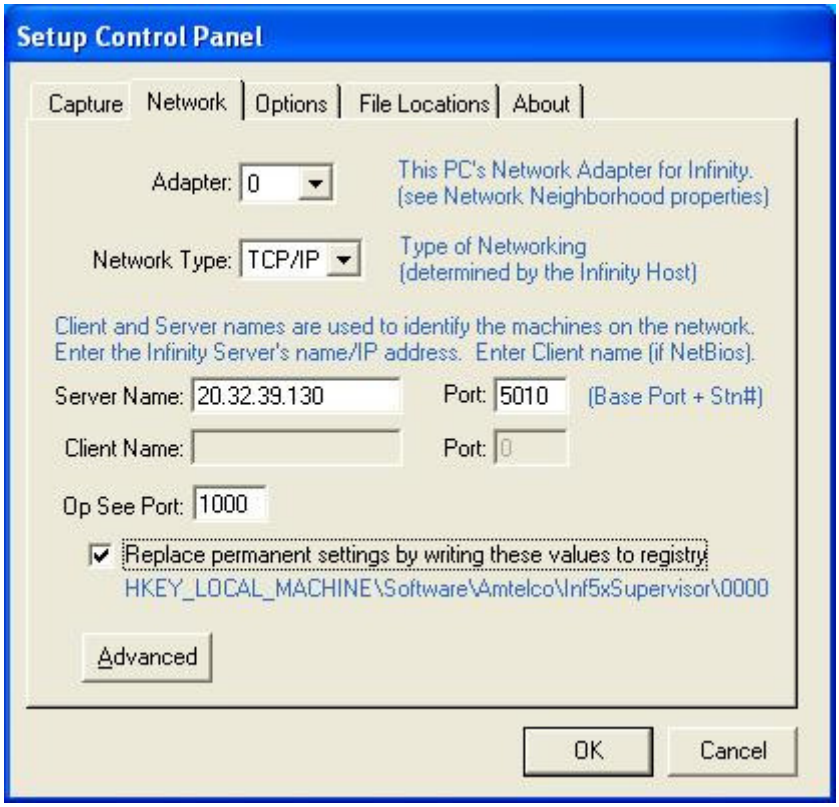
7.1. Launch Infinity Supervisor

From a PC running the Amtelco Infinity Supervisor application, select **Start → All Programs → AMTELCO → Infinity Supervisor** to display the **Infinity Supervisor Login** screen below.

Upon initial log in, prior to entering the credentials, press the **Ctrl** and **F12** key.

The **Setup Control Panel** screen is displayed. For **Server Name**, enter the IP address of the Infinity server that interfaces with attendants, in this case "20.32.39.130". Retain the default values in the remaining fields.

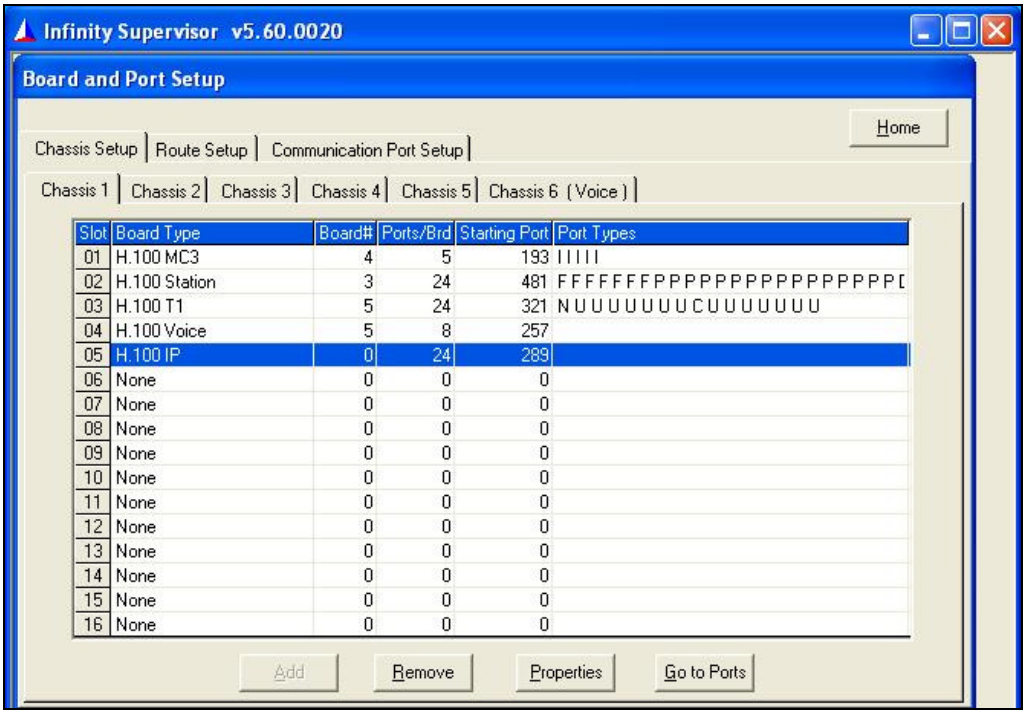
The **Infinity Supervisor Login** screen shown earlier is displayed again. Log in using the appropriate credentials.



7.2. Administer Billing Number and Board Settings

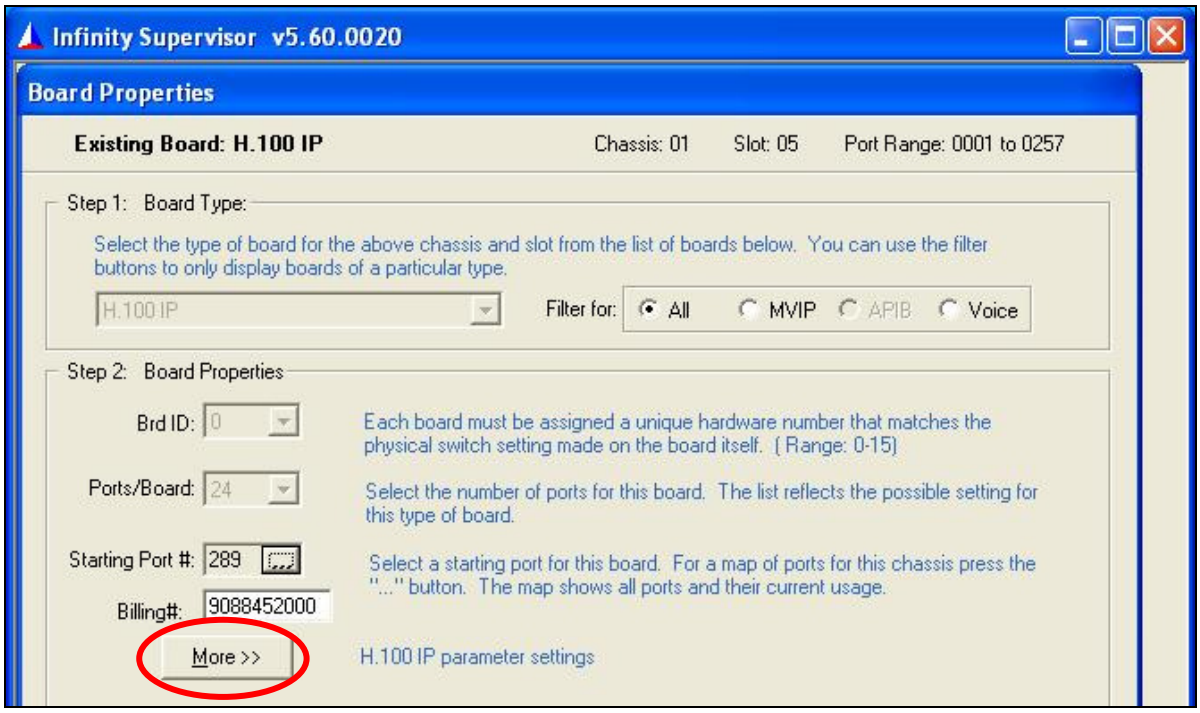
The **Infinity Supervisor** screen is displayed next. Select **BOARDS and PORTS**.

The **Board and Port Setup** screen pops up. Select the **H.100 IP** entry, and click **Properties**.

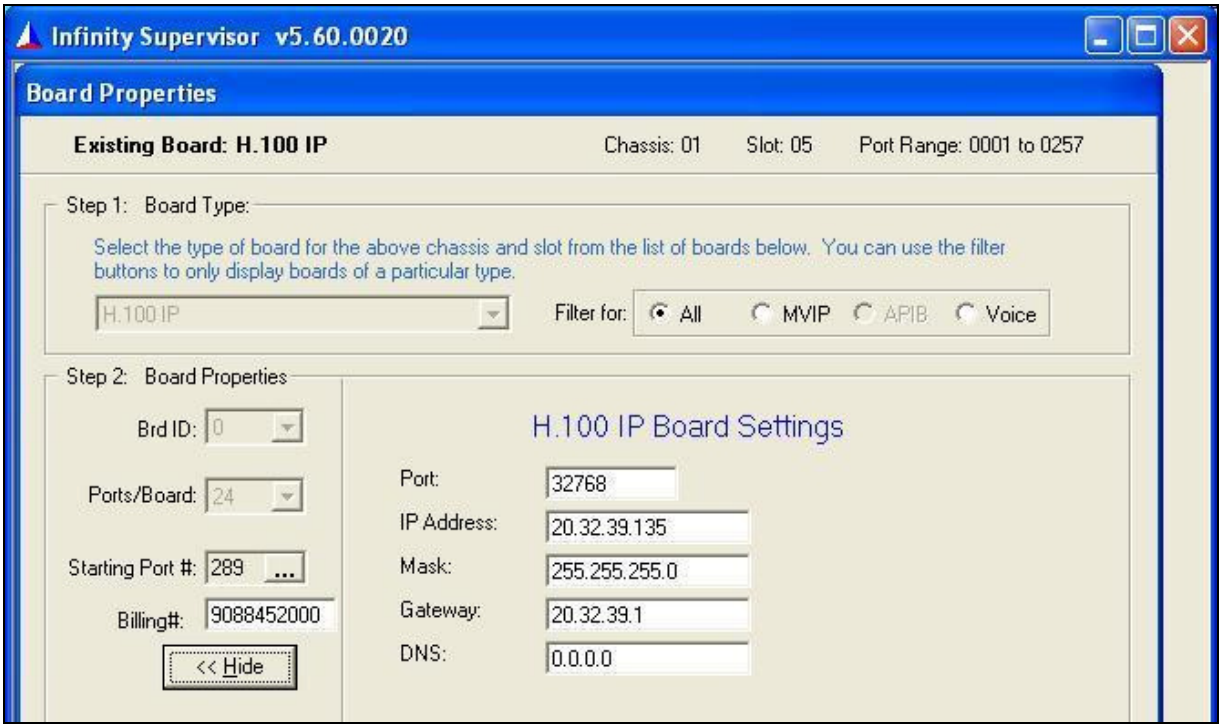


Slot	Board Type	Board#	Ports/Brd	Starting Port	Port Types
01	H.100 MC3	4	5	193	IIIII
02	H.100 Station	3	24	481	FFFFFFFFPPPPPPPPPPPPPPPP[
03	H.100 T1	5	24	321	NUUUUUUUCUUUUUUU
04	H.100 Voice	5	8	257	
05	H.100 IP	0	24	289	
06	None	0	0	0	
07	None	0	0	0	
08	None	0	0	0	
09	None	0	0	0	
10	None	0	0	0	
11	None	0	0	0	
12	None	0	0	0	
13	None	0	0	0	
14	None	0	0	0	
15	None	0	0	0	
16	None	0	0	0	

The **Board Properties** screen pops up next. For **Billing#,** enter the applicable number to use for outbound calls for billing purposes, and click **More**.



The **Board Properties** screen is updated with the **H.100 IP Board Settings** sub-section. For **Port**, enter "32768". Enter the pertinent network information for the remaining fields.

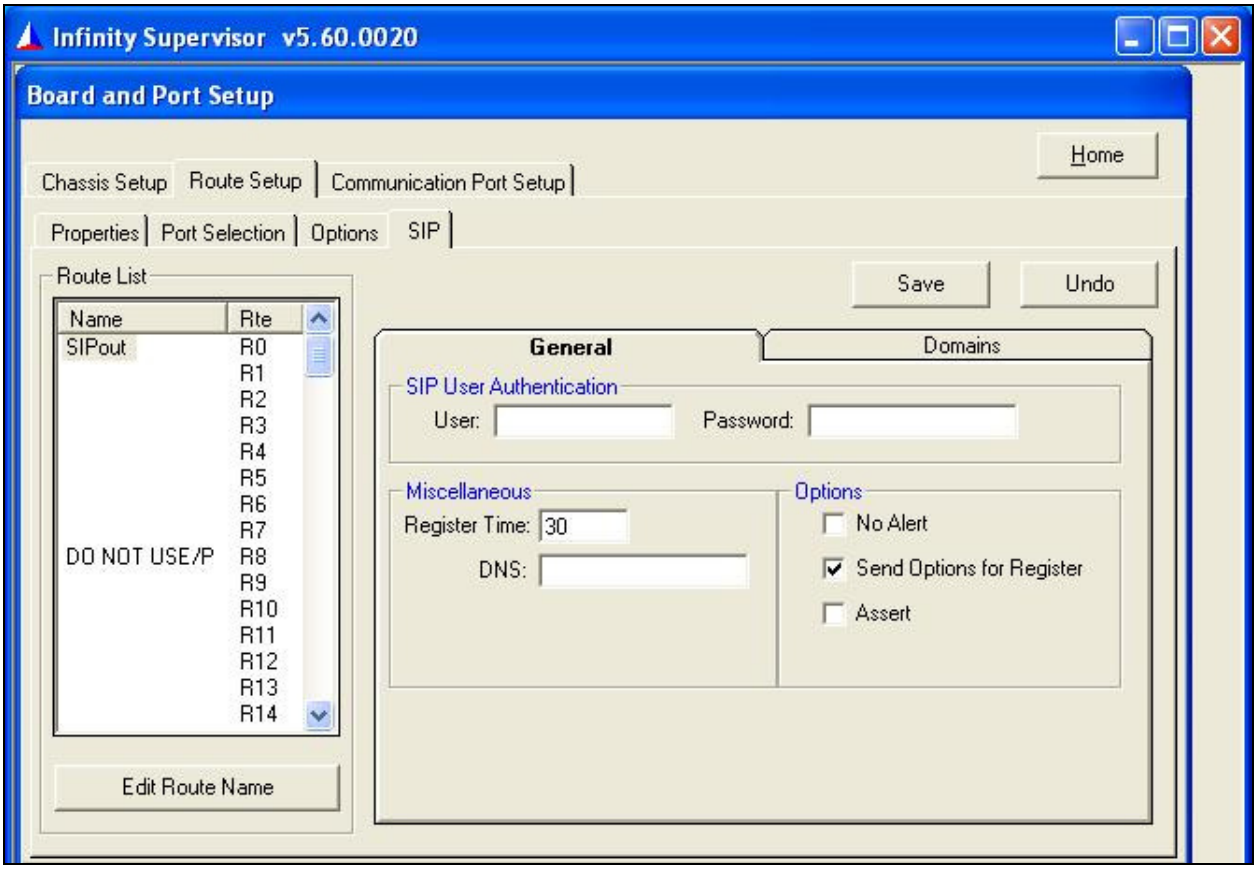


7.3. Administer SIP Route

The **Board and Port Setup** screen is displayed next. Select the **Route Setup → SIP** tab, followed by the **General** sub-tab.

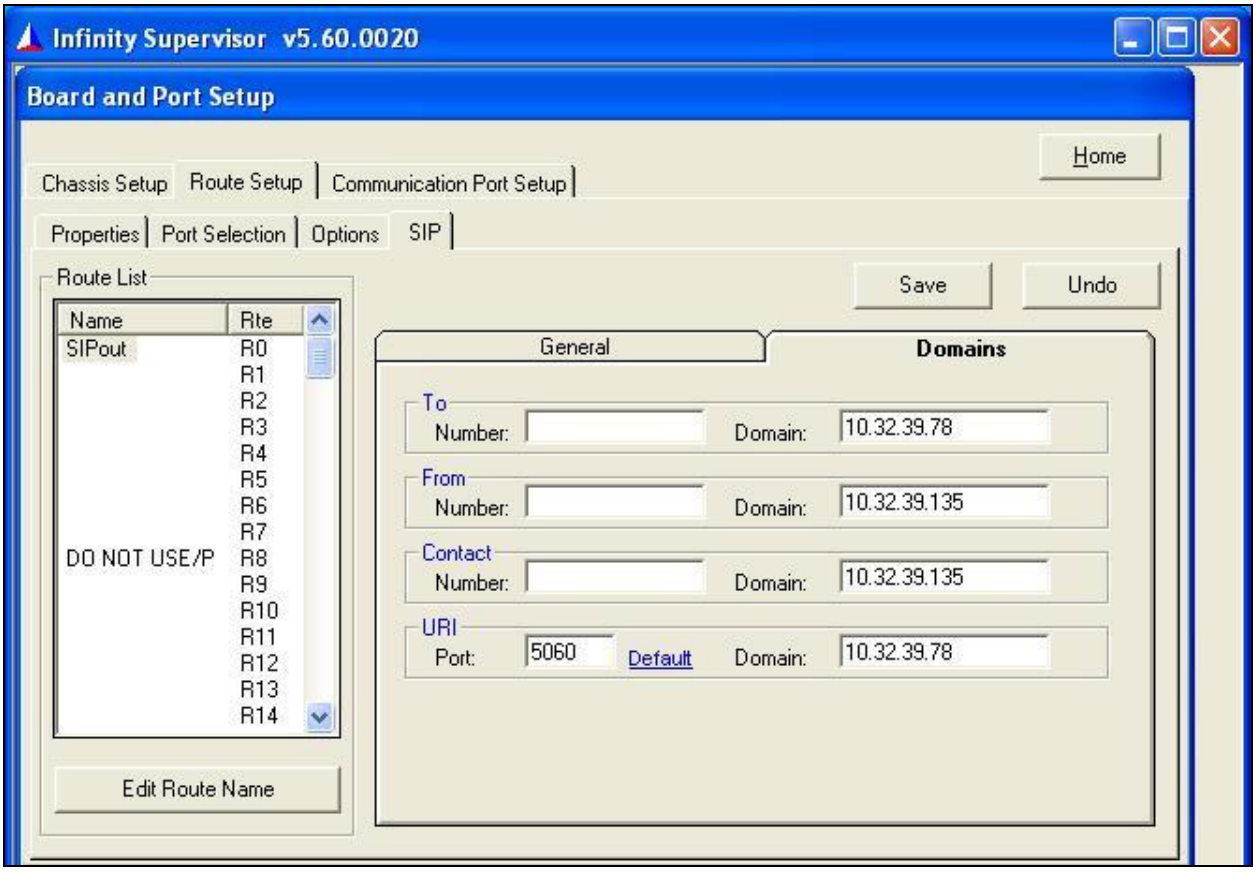
Under **Options**, check **Send Options for Register**.

For **Register Time**, enter a desired interval for the Options message.



Select the **Domains** sub-tab. Enter the following values for the specified fields, and retain the default values for the remaining fields.

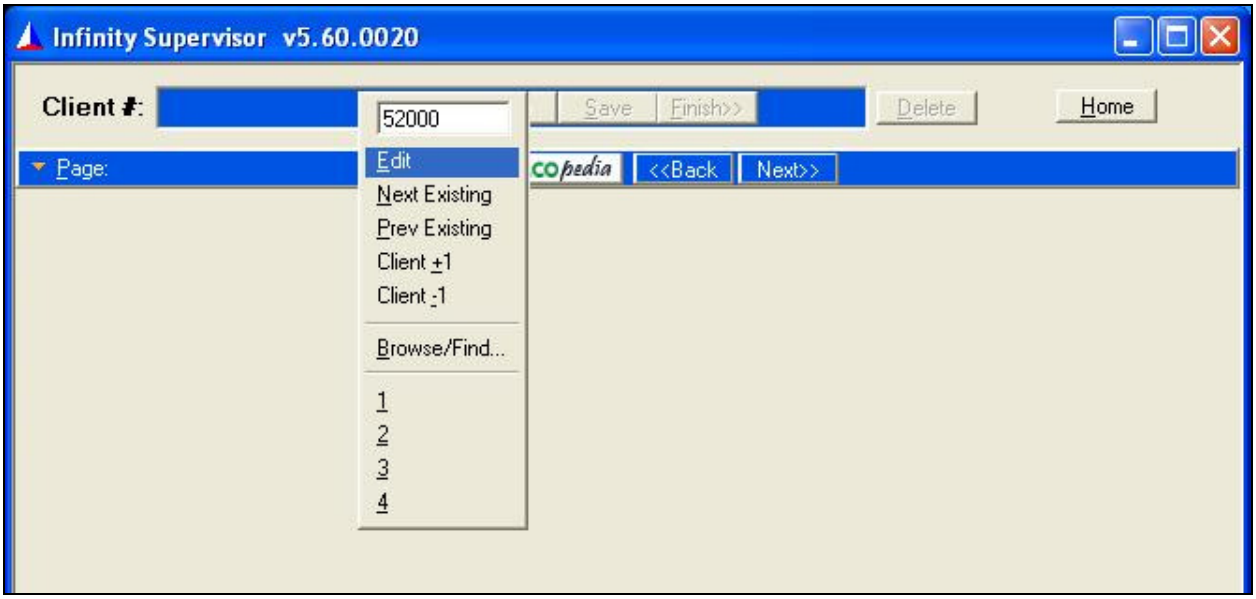
- **To Domain:** IP address of the Session Manager signaling interface.
- **From Domain:** IP address of the IP board from **Section 7.2**.
- **Contact Domain:** IP address of the IP board from **Section 7.2**.
- **URI Port:** Infinity SIP entity port number from **Section 6.5.1**.
- **URI Domain:** IP address of the Session Manager signaling interface.



7.4. Administer Clients

From the **Infinity Supervisor** screen shown below, select **CLIENT**.

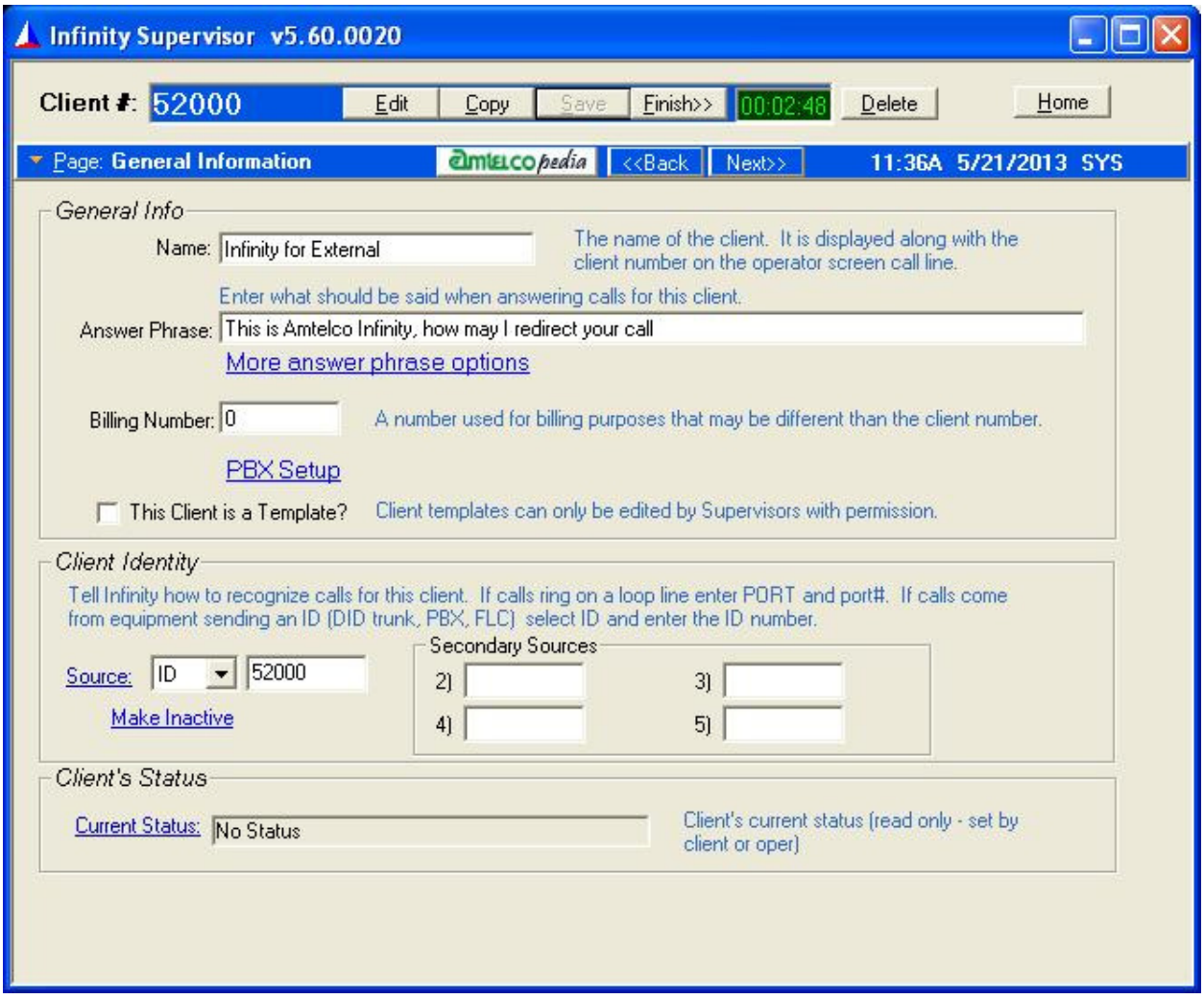
The screen below is displayed. Enter an available client number, in this case "52000". Confirm adding a new client in the next pop-up dialog box (not shown).



The screen is updated as shown below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** Name to display to the attendant when answering calls to this client.
- **Answer Phrase:** Guidance phrase for what to say when answering calls to this client.
- **Source ID:** The phone number identification for this client.

Repeat this section to administer all needed clients. In this compliance testing, calls from the PSTN will be routed with digits 52000 to Infinity, and calls from internal users on Communication Manager will be routed with digits 52222 to Infinity. Therefore two clients were created.



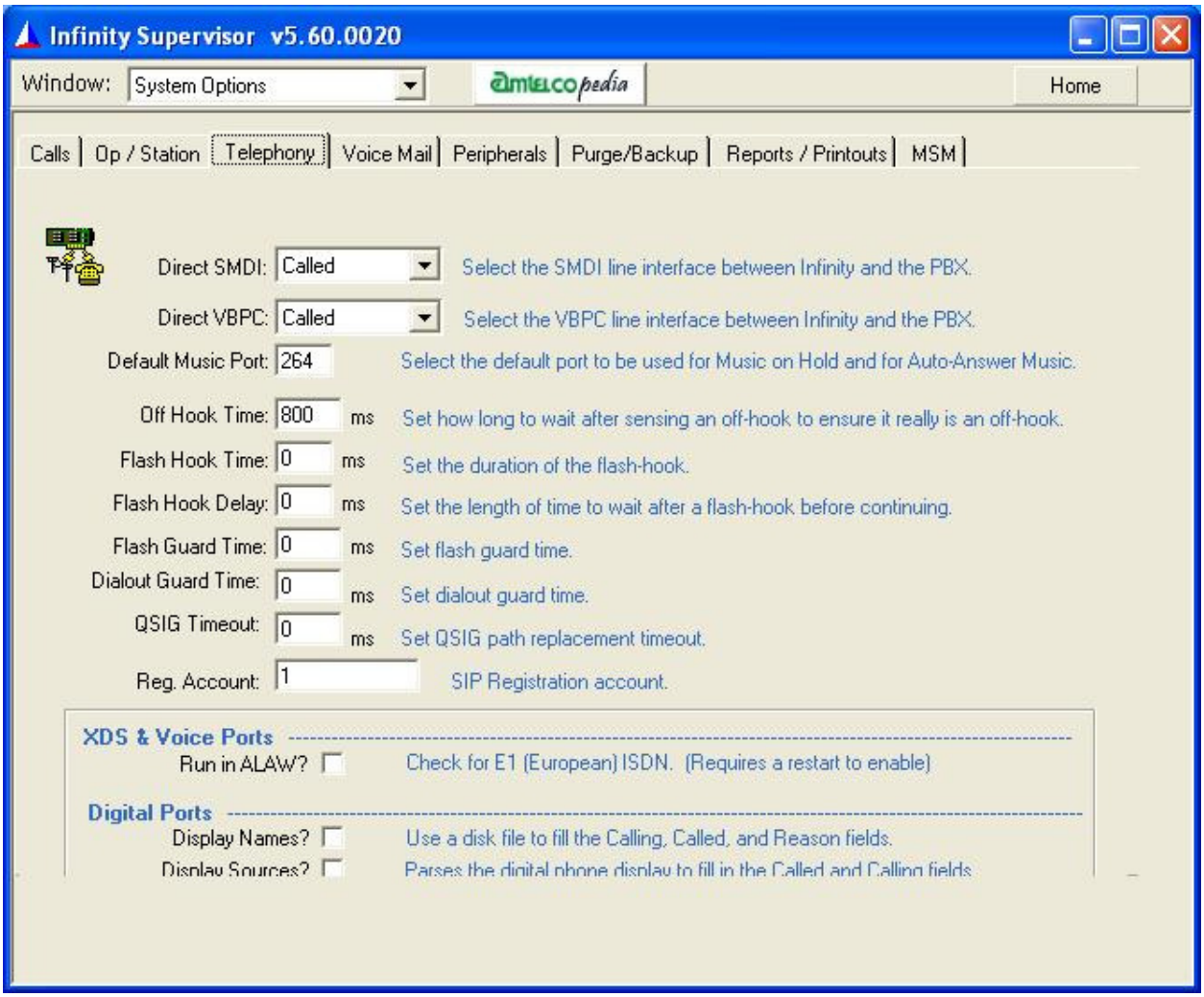
7.5. Administer System Settings

From the **Infinity Supervisor** screen shown below, select **SYSTEM SETTINGS**.

The screen below is displayed. For **Window**, select **System Options** from the drop-down list.

Select the **Telephony** tab, and enter a valid account number for **Reg. Account**.

Reboot the Infinity server.



8. Verification Steps

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

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.

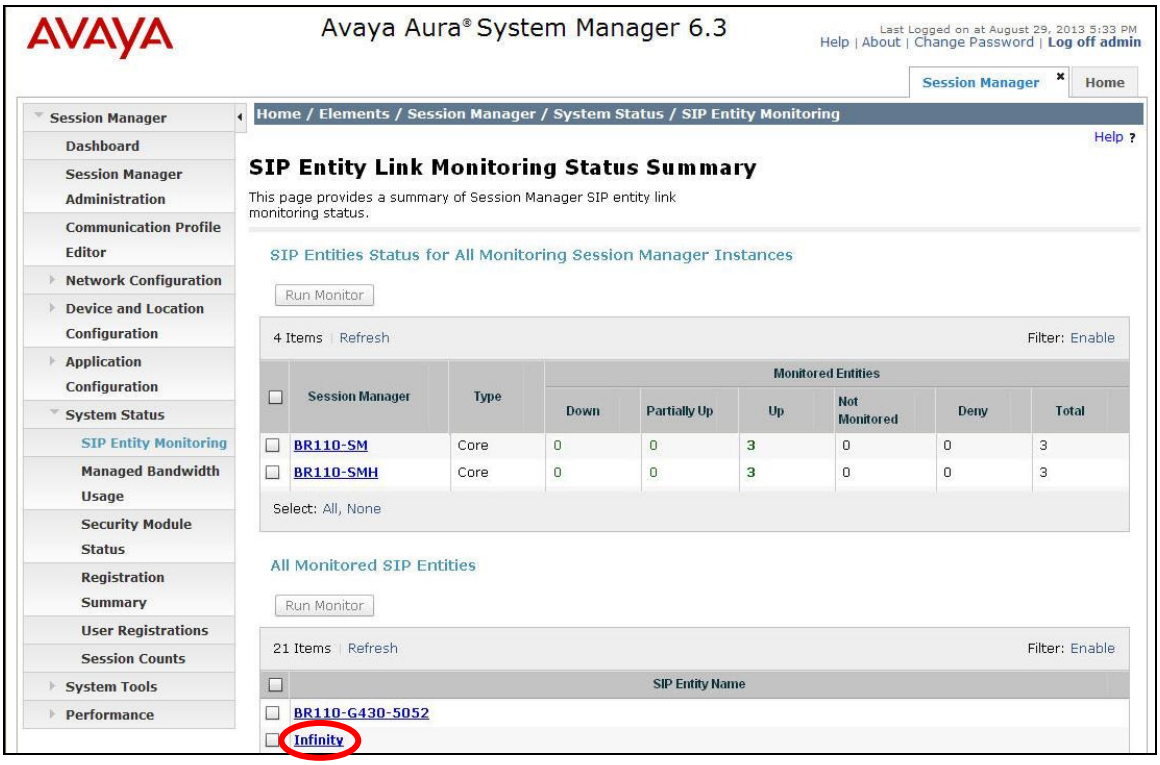
```
status trunk 52

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0052/001 T00011   in-service/idle    no
0052/002 T00012   in-service/idle    no
0052/003 T00013   in-service/idle    no
0052/004 T00014   in-service/idle    no
0052/005 T00015   in-service/idle    no
0052/006 T00016   in-service/idle    no
0052/007 T00017   in-service/idle    no
0052/008 T00018   in-service/idle    no
0052/009 T00019   in-service/idle    no
0052/010 T00020   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 signaling group is "in-service" as indicated in the **Group State** field shown below.

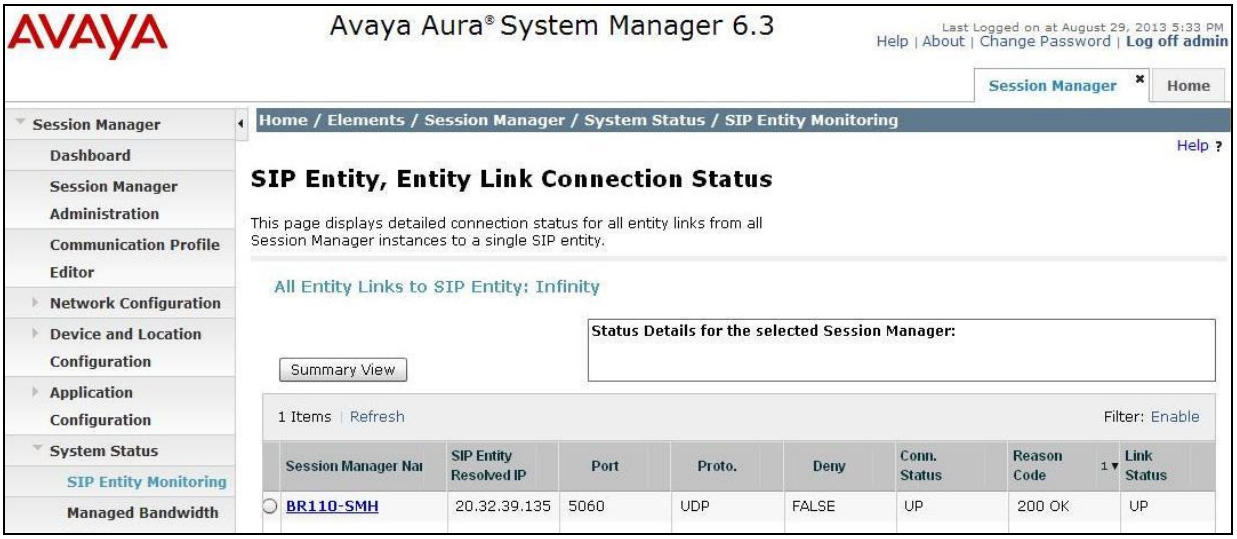
```
status signaling-group 52
                         STATUS SIGNALING GROUP

       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 Infinity entity name from **Section 6.4.1**.



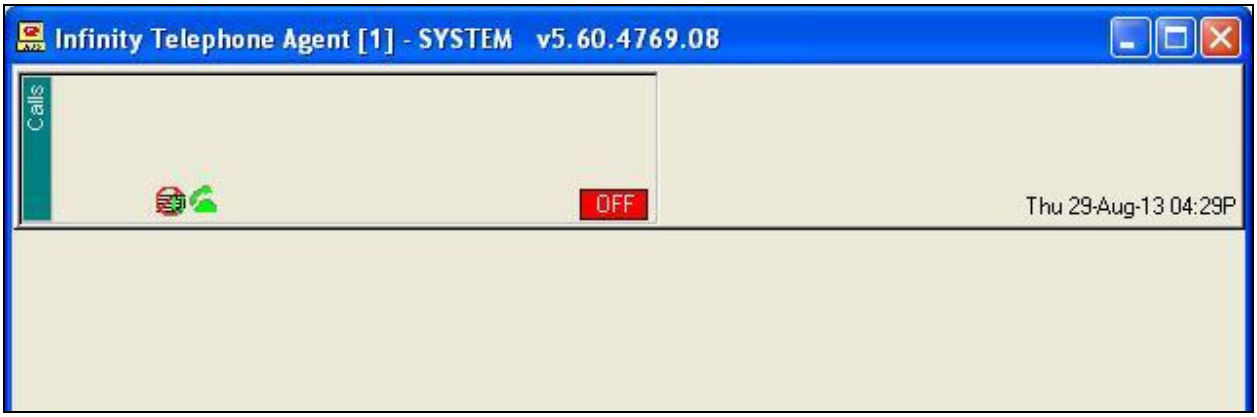
The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that **Conn Status** and **Link Status** are "Up", as shown below.



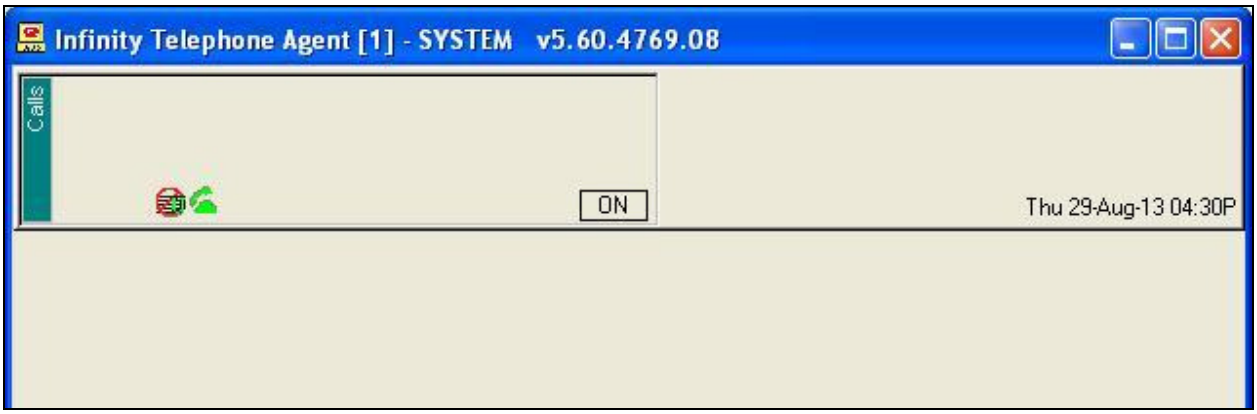
8.3. Verify Infinity Intelligent SIP Attendant Console

From an attendant PC running the Amtelco Infinity Telephone Agent application, select **Start → All Programs → AMTELCO → Infinity Telephone Agent** to display the **Infinity Telephone Agent** screen below. Log in using the appropriate credentials.

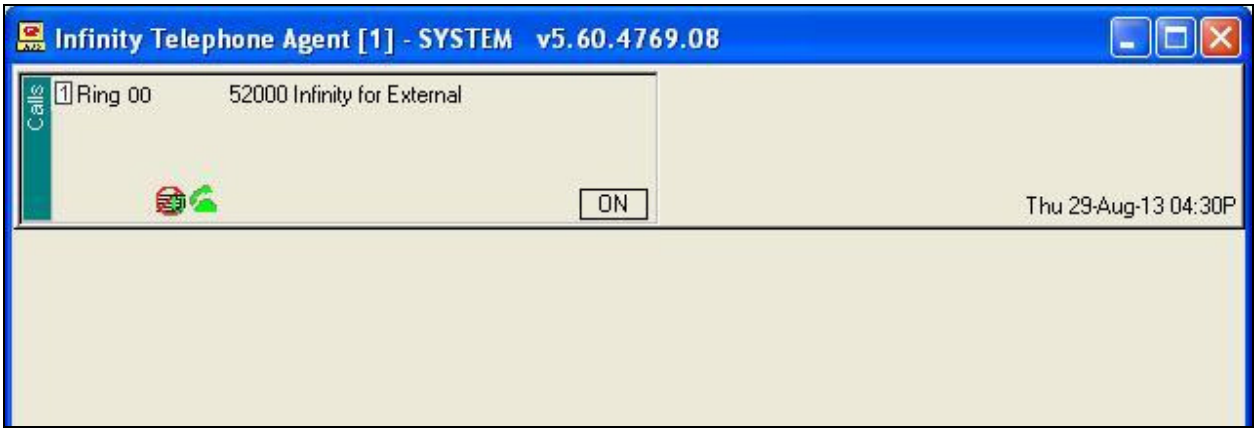
The screen below is displayed next. Click **OFF** to toggle into available.



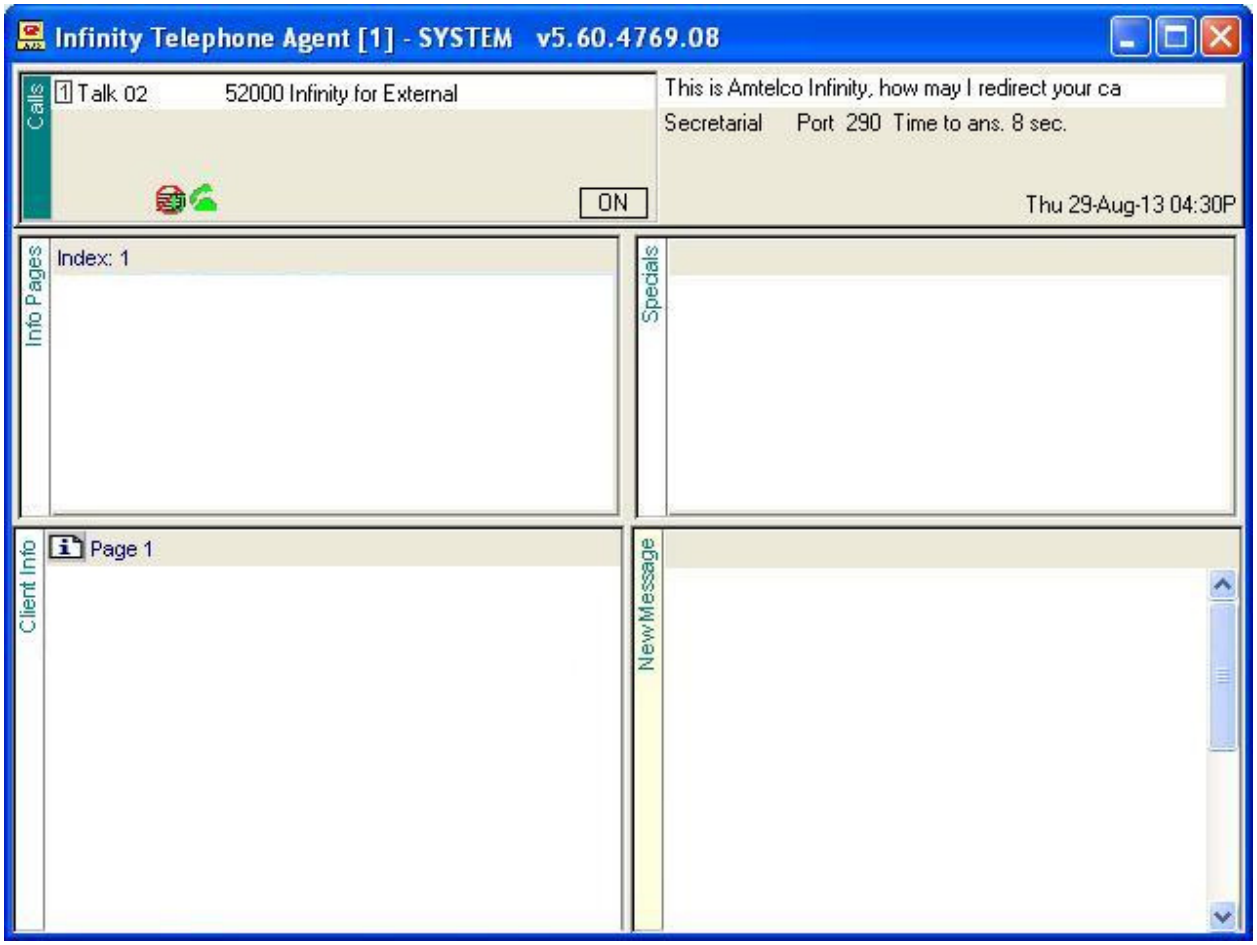
Verify the status is updated to **ON**, as shown below.



Make an incoming call from the PSTN to reach Infinity. Verify that an available attendant hears the alerting tone, and that the attendant screen is updated showing the alerting call. Also verify that the display information reflects the proper client ID and name from **Section 7.4**.



Press **F1** to answer the call. Verify that the attendant is connected to the PSTN with two-way talk paths, and that the screen is updated with the proper guidance phrase from **Section 7.4**.



9. Conclusion

These Application Notes describe the configuration steps required for Amtelco Infinity Intelligent SIP Attendant Console to successfully interoperate with Avaya Aura® Session 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*, Document 03-300509, Issue 8, Release 6.3, May 2013, available at http://support.avaya.com.

2. *Administering Avaya Aura® Session Manager*, Release 6.3, June 2013, available at http://support.avaya.com.

3. *Administering Avaya Aura® System Manager*, Release 6.3, May 2013, available at http://support.avaya.com.

4. *Infinity Supervisor Reference Guide*, Version 232M072, November 2012, available at http://service.amtelco.com.

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