Avaya Solution & Interoperability Test Lab

Application Notes for IPC Unigy with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for IPC Unigy to interoperate with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager using SIP trunks.

IPC Unigy is a trading communication solution. In the compliance testing, IPC Unigy used SIP trunks to Avaya Aura® Session Manager, for turret users on IPC to reach users on Avaya Aura® Communication Manager and on the PSTN.

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 IPC Unigy to interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager.

IPC Unigy is a trading communication solution. In the compliance testing, IPC Unigy used SIP trunks to Avaya Aura® Session Manager, for turret users on IPC to reach users on Avaya Aura® Communication Manager and on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established among IPC turret users with Avaya SIP, Avaya H.323, and/or PSTN users. Call controls were performed from various users to verify the call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet cable to IPC Unigy.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included basic call, display, G.711MU, G.729AB, codec negotiation, hold/reconnect, DTMF, call forwarding unconditional/ring-no-answer/busy, blind/attended transfer, and attended conference.

The serviceability testing focused on verifying the ability of IPC Unigy to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to IPC Unigy.

2.2. Test Results

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

- IPC does not support domain name, therefore the domain name on the Avaya SIP trunk group and network region must be left blank to accommodate this.
- IPC does not support media shuffling, therefore corresponding parameters must be disabled on the Avaya signaling group and network region. Furthermore, IPC does not support asymmetric codec, so the supported codec order must be in sync between IPC and Avaya.
- IPC does not support interpretation of DMTF digits from Avaya endpoints, so the DTMF tests only covered the Avaya interpretation of DMTF digits from the IPC turrets.
- In an outgoing call from IPC turret to the PSTN, the IPC turret display will show "null" as the connected number. Note that the name of the PSTN endpoint can still be shown on the display, and that incoming calls from the PSTN to the IPC turrets have proper displays.
- In transfer scenarios involving IPC turrets transferring calls to Avaya SIP endpoints, the Avaya SIP endpoints will see "wlssuser" in the display upon completion of transfer, as sent from IPC.
- The dial pattern string specified on IPC must contain the exact number of digits.
- For call forwarding scenarios involving Avaya SIP endpoints calling IPC turrets that are forwarded back to Avaya endpoints, the Avaya SIP endpoint will show two active call appearances after the call diverts.
- Multiple divert buttons on the turret can lead to turret performance degradation.
- For blind transfer scenarios involving IPC turrets as the called party and Avaya SIP or PSTN as the calling and forward-to parties, there is no talk path for the resultant call with IPC returning 500 Internal Error. The workaround is to use attended transfer instead.
- Even when IPC Unigy is configured with UDP, the TCP protocol much be configured to be allowed on Avaya Session Manager as Unigy switches over to use TCP for diversions.

2.3. Support

Technical support on IPC Unigy can be obtained through the following:

- **Phone:** (800) NEEDIPC, (203) 339-7800
- **Email:** systems.support@ipc.com

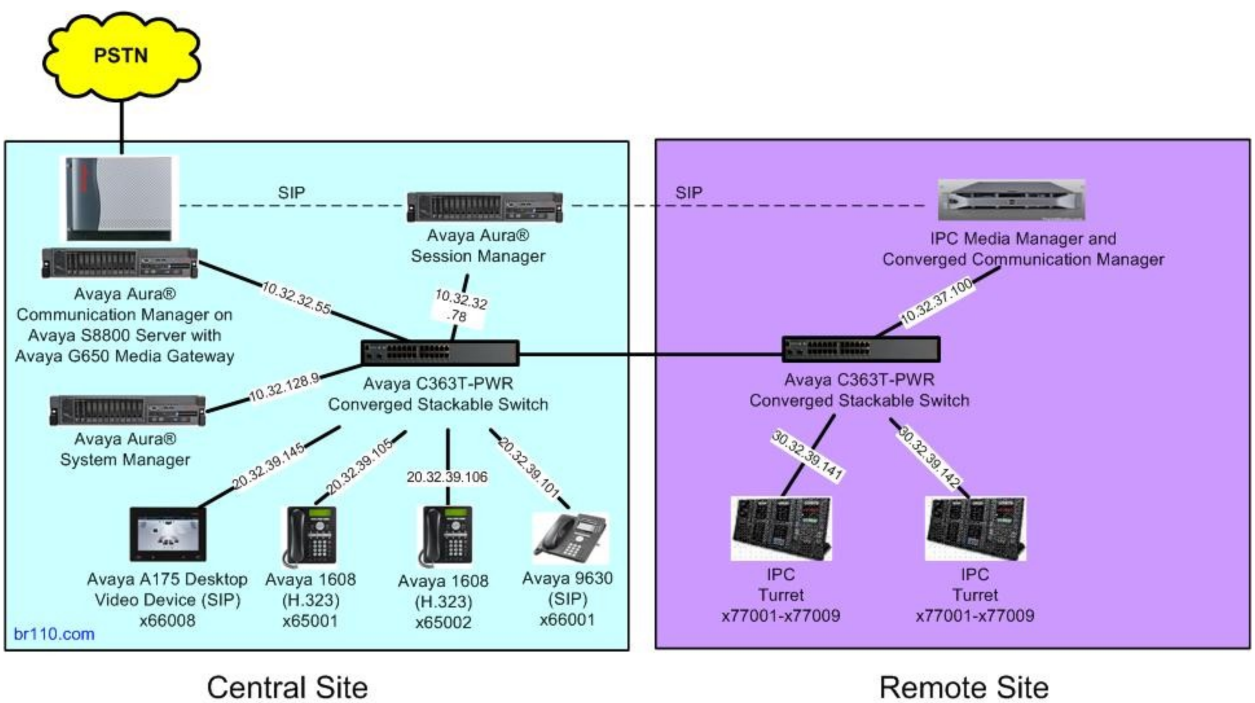
3. Reference Configuration

As shown in the test configuration below, IPC Unigy at the Remote Site consists of the Media Manager, Converged Communication Manager, and Turrets. The Media Manager and Converged Communication Manager are typically deployed on separate servers. In the compliance testing, the same server hosted the Media Manager and Converged Communication Manager.

SIP trunks are used from IPC Unigy to Avaya Aura® Session Manager, to reach users on Avaya Aura® Communication Manager and on the PSTN.

A five digit Uniform Dial Plan (UDP) was used to facilitate dialing between the Central and Remote sites. Unique extension ranges were associated with Avaya Aura® Communication Manager users at the Central site (6xxxx), and IPC turret users at the Remote site (77xxx).

The detailed administration of basic connectivity between Avaya Aura® Communication Manager and Avaya Aura® Session Manager is not the focus of these Application Notes and will not be described.



4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager on Avaya S8800 Server	6.0.1 SP2 with special patch 18993 (R016x.00.1.510.1-18993)
Avaya G650 Media Gateway • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor • TN464HP DS1 Interface	 HW01 FW038 HW20 FW122 HW02 FW024
Avaya Aura® Session Manager	6.1 SP2
Avaya Aura® System Manager	6.1 SP2
Avaya 1608 IP Telephone (H.323)	1.3
Avaya 9630 IP Telephone (SIP)	2.6.4
Avaya A175 Desktop Video Device (SIP)	1.0.2
IPC Unigy • Media Manager • Converged Communication Manage • Turrets	 01.00.00.01.0003 01.00.00.01.0003 01.00.00.01.0003

5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- 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 the IPC turret users.

5.1. Verify Communication Manager 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.

```
change system-parameters customer-options                       Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 6
           Maximum Concurrently Registered IP Stations: 18000 0
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer System Parameters Features

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

This feature is needed to be able to transfer an incoming call from IPC back out to IPC (incoming trunk to outgoing trunk), and to transfer an outgoing call to IPC to another outgoing call to IPC (outgoing trunk to outgoing trunk). 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? y
                                    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: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: none
                  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 "77". 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 77                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 77                   Group Type: sip          CDR Reports: y
  Group Name: IPC Unigy                   COR: 1        TN: 1        TAC: 1077
   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 77                                              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
```

Navigate to **Page 4**, and enter "101" for **Telephone Event Payload Type**.

```
change trunk-group 77                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                     Send Diversion Header? n
                   Support Request History? y
              Telephone Event Payload Type: 101
```

5.4. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "77". 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.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration with IPC Unigy.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region for integration with IPC Unigy.

For **Far-end Domain**, leave the field blank since IPC Unigy does not support domain name. For **Direct IP-IP Audio Connections**, enter "n" since IPC Unigy does not support shuffling.

```
add signaling-group 77                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 77             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: Clan-1                  Far-end Node Name: S8800-SM-SIG
 Near-end Listen Port: 5077                  Far-end Listen Port: 5077
                                          Far-end Network Region: 7
                                     Far-end Secondary Node Name:
Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                  Alternate Route Timer(sec): 6
```

5.5. 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**, leave the field blank. 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 IPC Unigy.

```
change ip-network-region 7                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 7
Location: 1       Authoritative Domain:
    Name: IPC Unigy
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. In the compliance testing, all Avaya endpoints are on network region "1".

```
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
 4
 5
 6
 7    7                                                                 all
 8
```

5.6. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.5**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that IPC Unigy supports the G.711 and G.729 codec variants, and requires the codec order on Avaya to match the codec order specified on IPC Unigy. The codec order shown below matched the default order on IPC Unigy.

```
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:
```

5.7. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach IPC, in this case "77". 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 77                                         Page   1 of   3
                    Pattern Number: 77  Pattern Name: IPC Unigy
                             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: 77   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
```

5.8. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to IPC. 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 77 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                77                            5     Total Administered: 2
                                                            Maximum Entries: 540
```

5.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 77xxx to IPC. 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 77xxx, 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

 77              5   0                   aar  n
```

5.10. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 77xxx. In the example shown below, calls with digits 77xxx will be routed as an AAR call using route pattern "77" from **Section 5.7**.

```
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
   77                     5    5     77         unku          n
```

5.11. 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, in this case "10".

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

```
change trunk-group 10                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? 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: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
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
                             Network (Japan) Needs Connect Before Disconnect? n
 DSN Term? n
```

5.12. 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 IPC turret users.

In the example shown below, all calls originating from a 5-digit extension beginning with 7 and routed to trunk group 10 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                10                 90884            pub-unk
 5   7                10                 90884            pub-unk
```

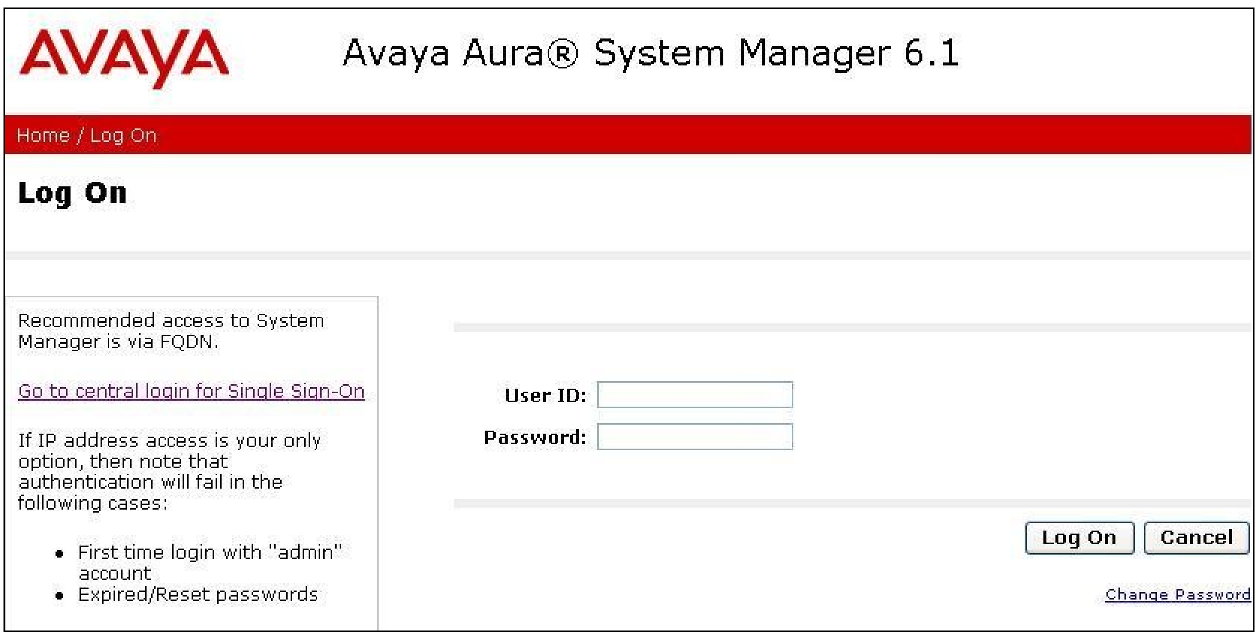
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® 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

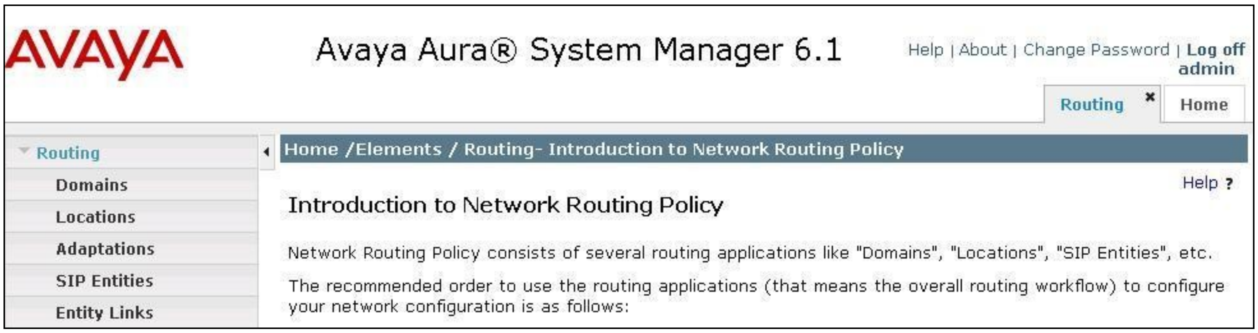
6.1. Launch System Manager

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.

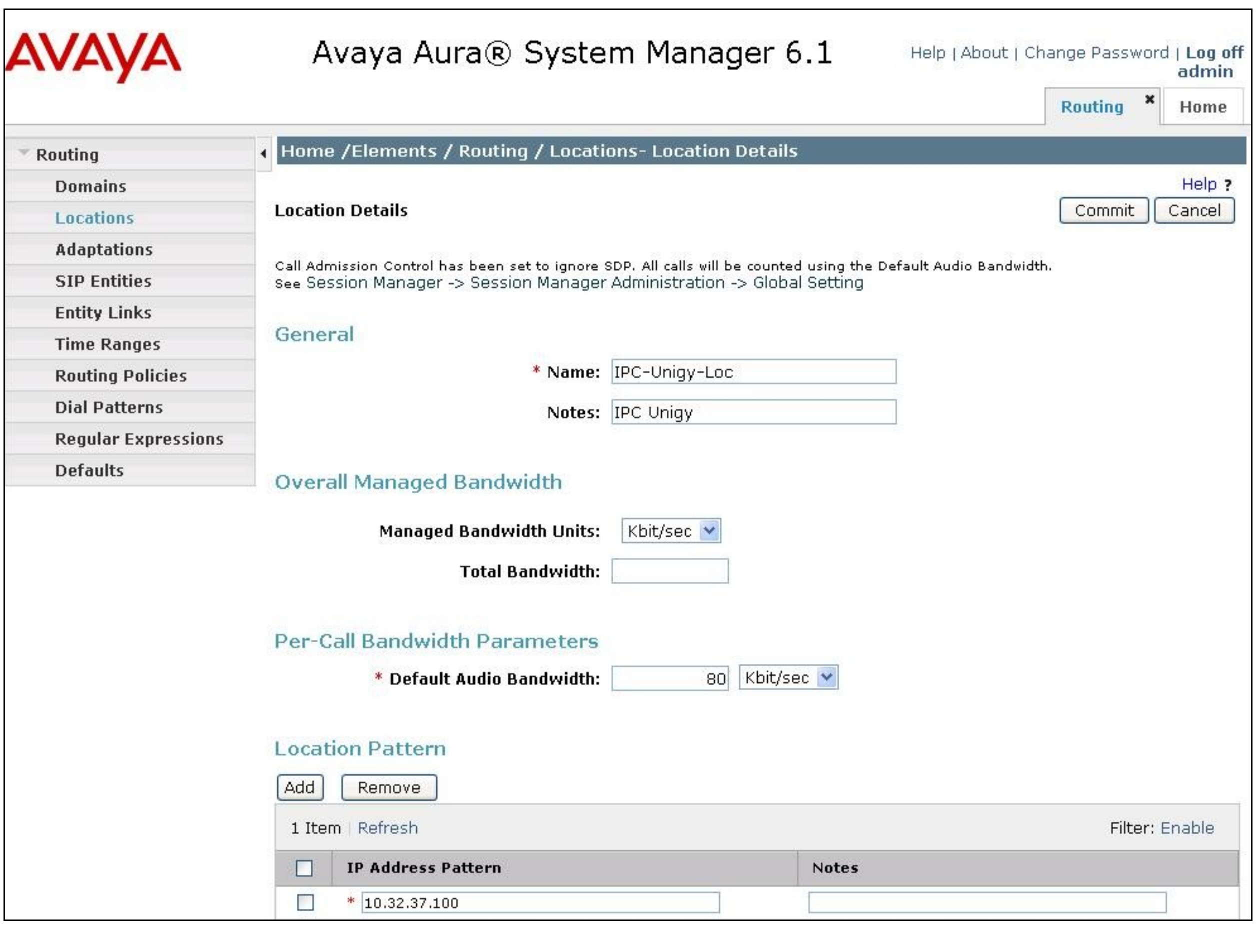


6.2. Administer Locations

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 IPC.



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**, as shown below. Retain the default values in the remaining fields.

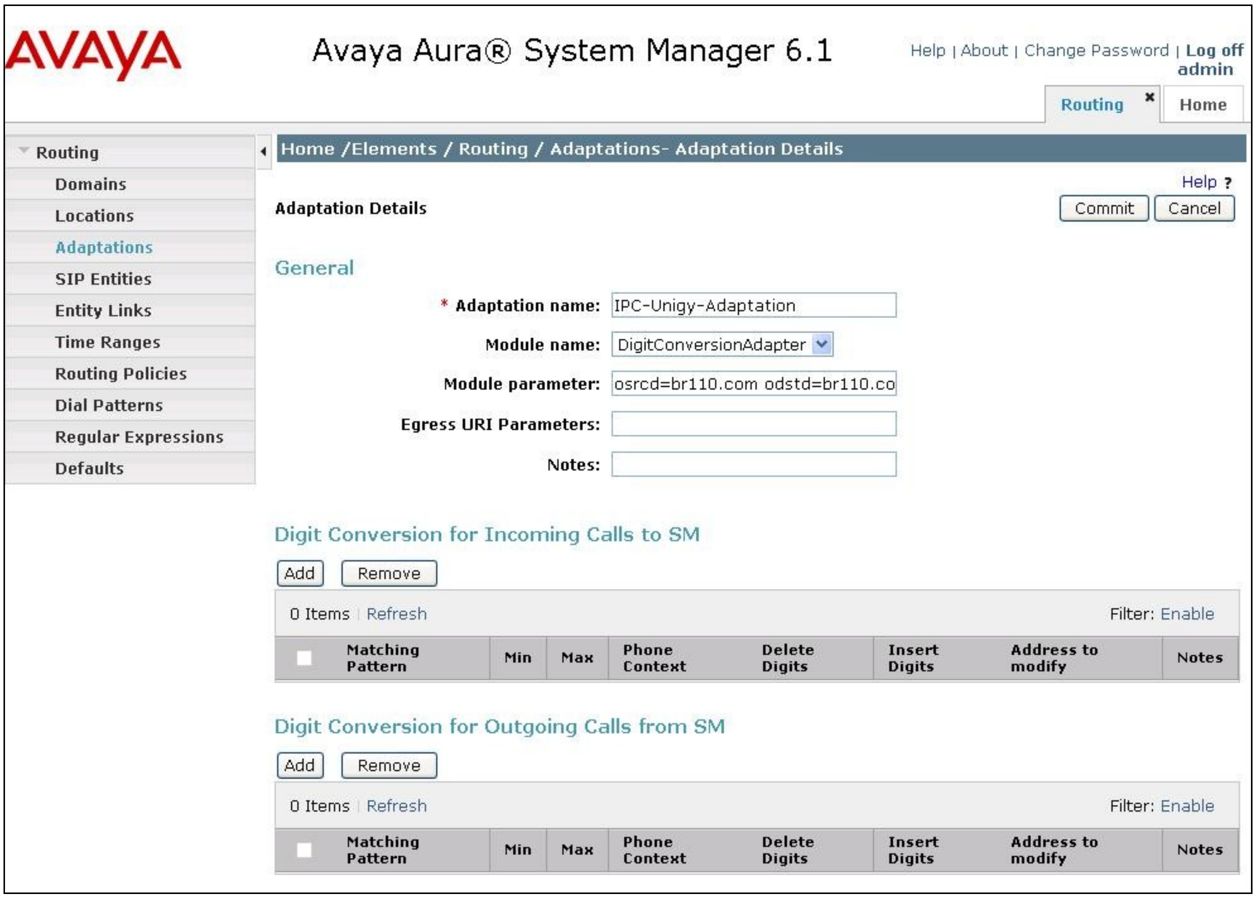


6.3. Administer Adaptations

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

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 iosrcd=br110.com iodstd=br110.com", where "br110.com" is the applicable domain. This will set the source and destination domains for all incoming and outgoing calls for IPC.



6.4. Administer SIP Entities

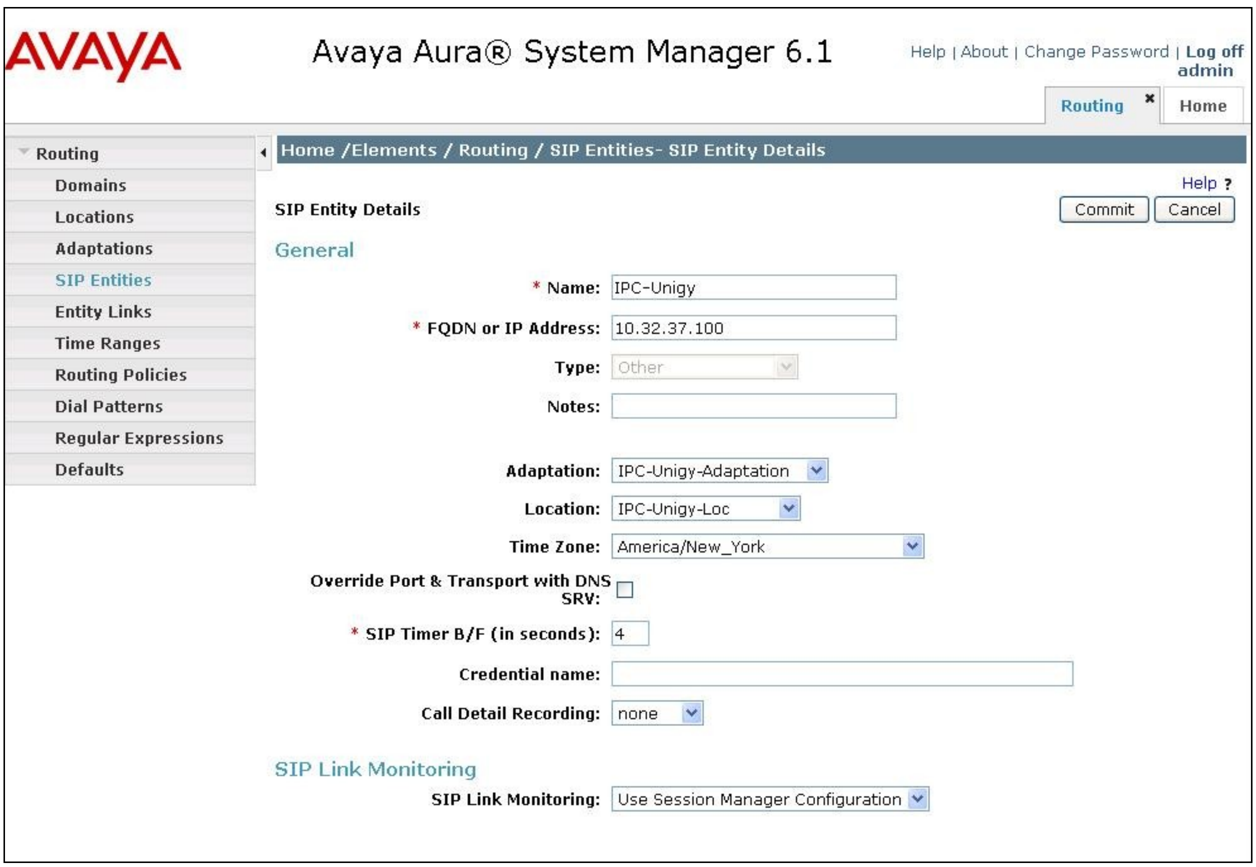
Add two new SIP entities, one for IPC, and another for the new SIP trunks for Communication Manager.

6.4.1. IPC SIP Entity

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

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 IPC Media Manager server.
- **Type:** "Other"
- **Adaptation:** Select the IPC adaptation name from **Section 6.3**.
- **Location:** Select the IPC location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

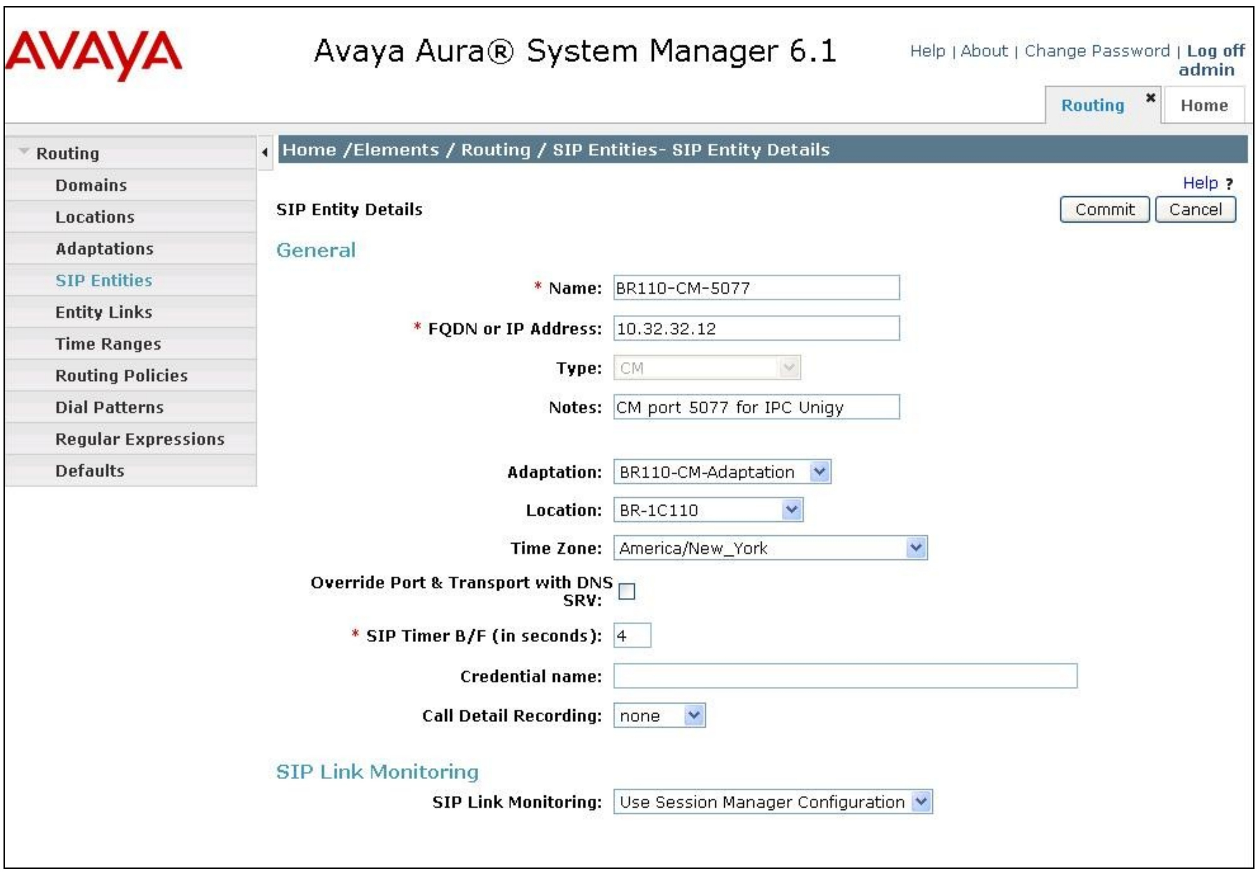


6.4.2. Communication Manager SIP Entity

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 IPC.

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.
- **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.



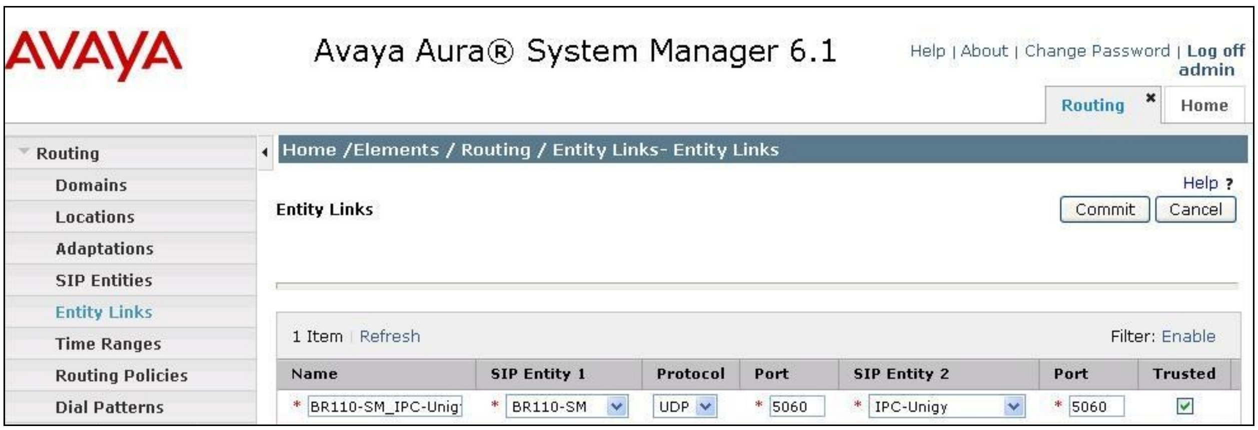
6.5. Administer Entity Links

Add three new entity links, two for IPC, and another for Communication Manager.

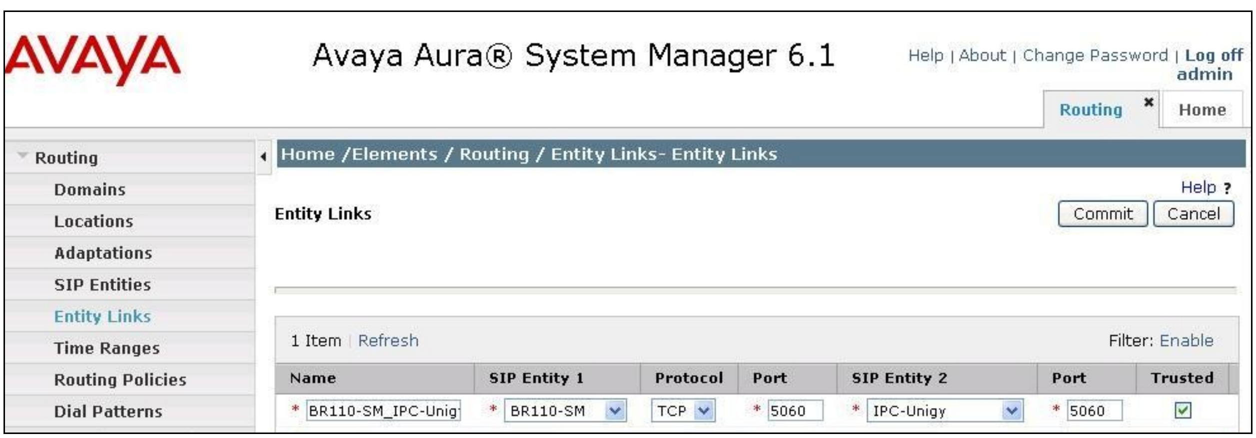
6.5.1. IPC Entity Links

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-SM".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The IPC entity name from **Section 6.4.1**.
- **Port:** "5060"
- **Trusted:** Retain the check.



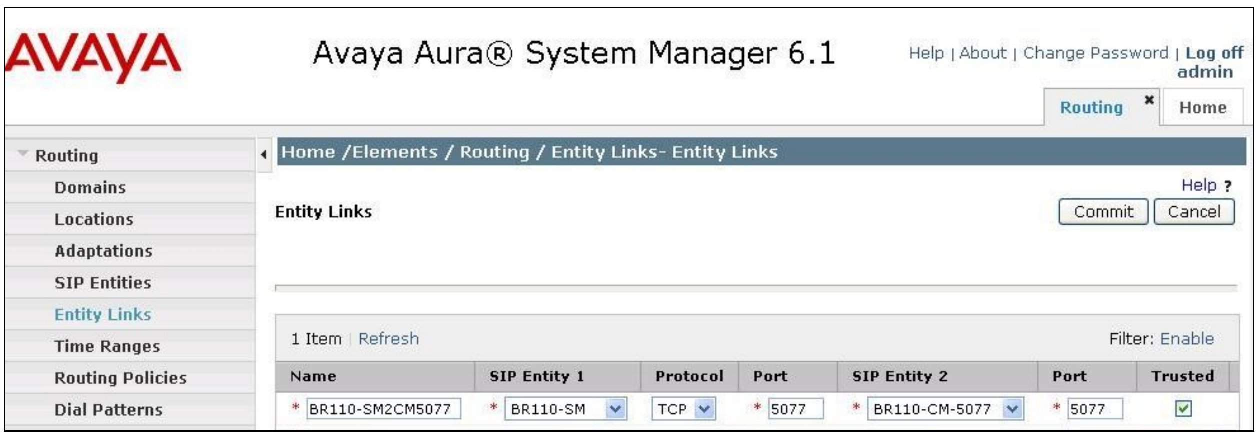
Repeat and add another entity link for IPC with "TCP" as Protocol, as shown below.



6.5.2. Communication Manager Entity Links

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-SM".
- **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**.
- **Trusted:** Retain the check.



6.6. Administer Routing Policies

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

6.6.1. IPC Routing Policy

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

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 IPC entity name from **Section 6.4.1** in the listing (not shown).

Retain the default values in the remaining fields.

6.6.2. Communication Manager Routing Policy

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

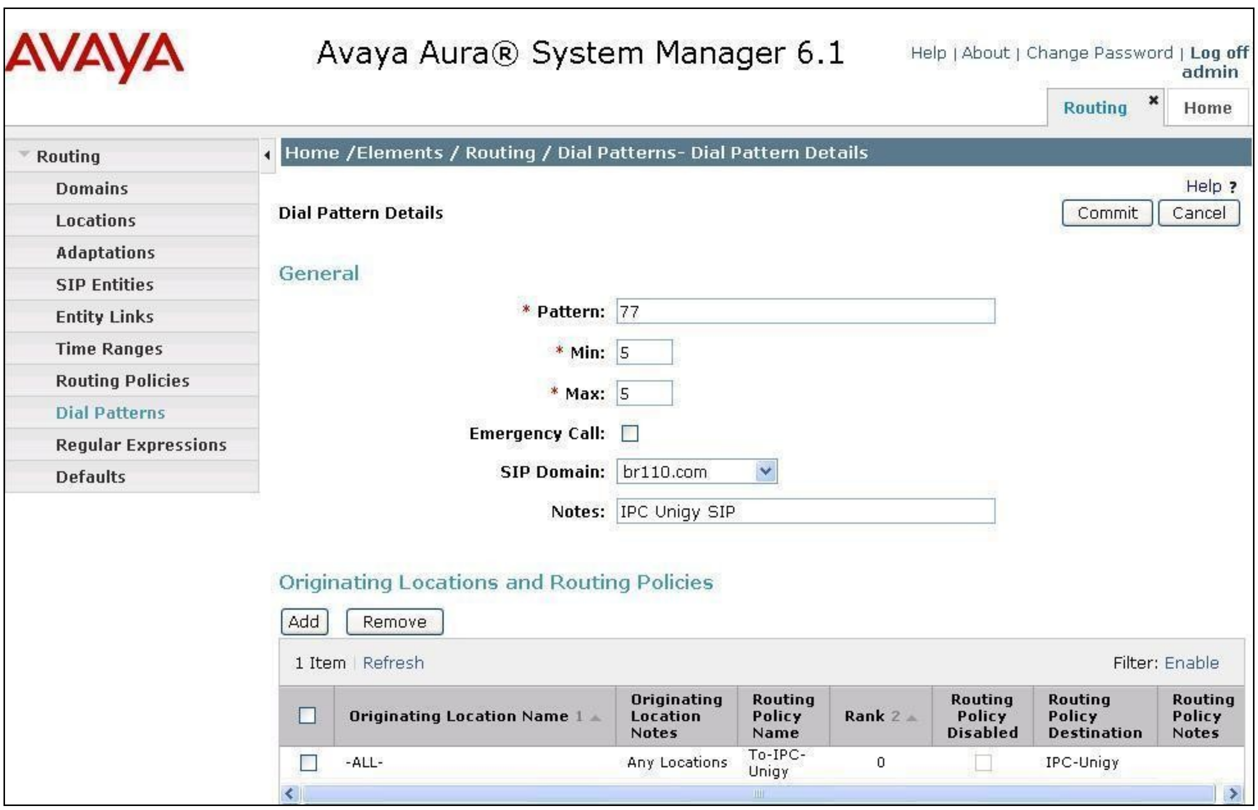
Add a new dial pattern for IPC, and update the existing dial pattern for Communication Manager.

6.7.1. IPC Dial Pattern

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 IPC turret users. 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 Communication Manager domain name from **Section 3**.
- **Notes:** Any desired description.

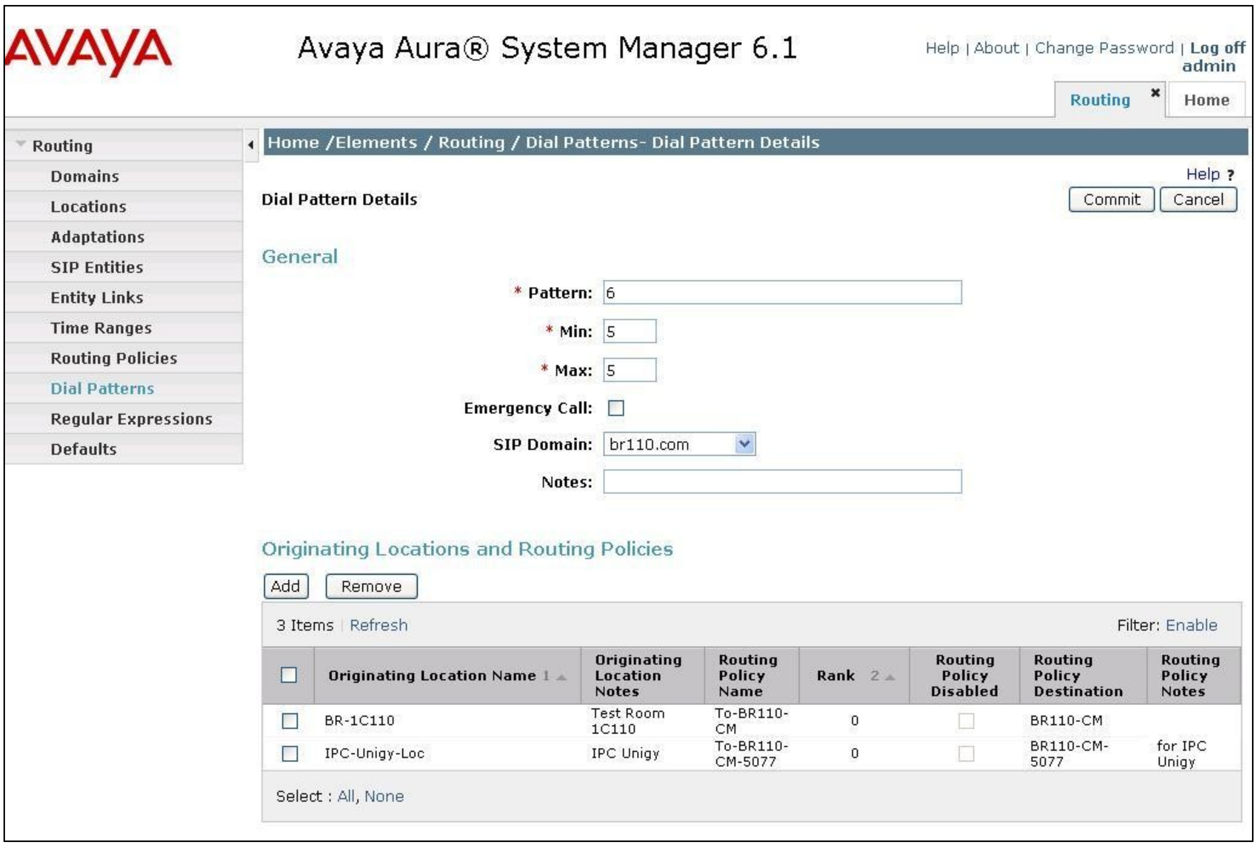
In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching IPC turret users. In the compliance testing, the policy allowed for call origination from all locations, and the IPC routing policy from **Section 6.6.1** was select as shown below.



6.7.2. Communication Manager Dial Pattern

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 "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 IPC turret users. In the compliance testing, the policy allowed for call origination from the IPC 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.



7. Configure IPC Media Manager

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

- Launch Unigy Management System
- Administer SIP trunks
- Administer trunk groups
- Administer route lists
- Administer dial patterns
- Administer route plans

The configuration of Media Manager is typically performed by IPC installation technicians. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Launch Unigy Management System

Access the Unigy Management System web interface by using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Media Manager. Log in using the appropriate credentials.

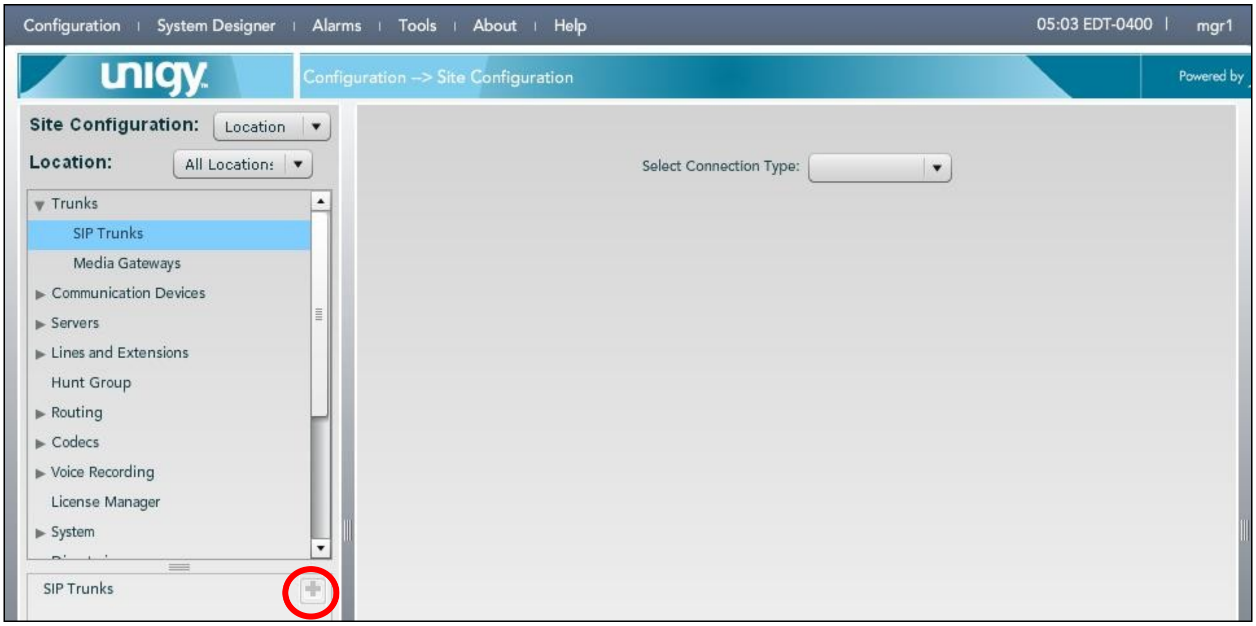
The screen below is displayed. Enter the appropriate credentials. Check **I agree with the Terms of Use**, and click **Login**.

In the subsequent screen (not shown), click **Continue**.

7.2. Administer SIP Trunks

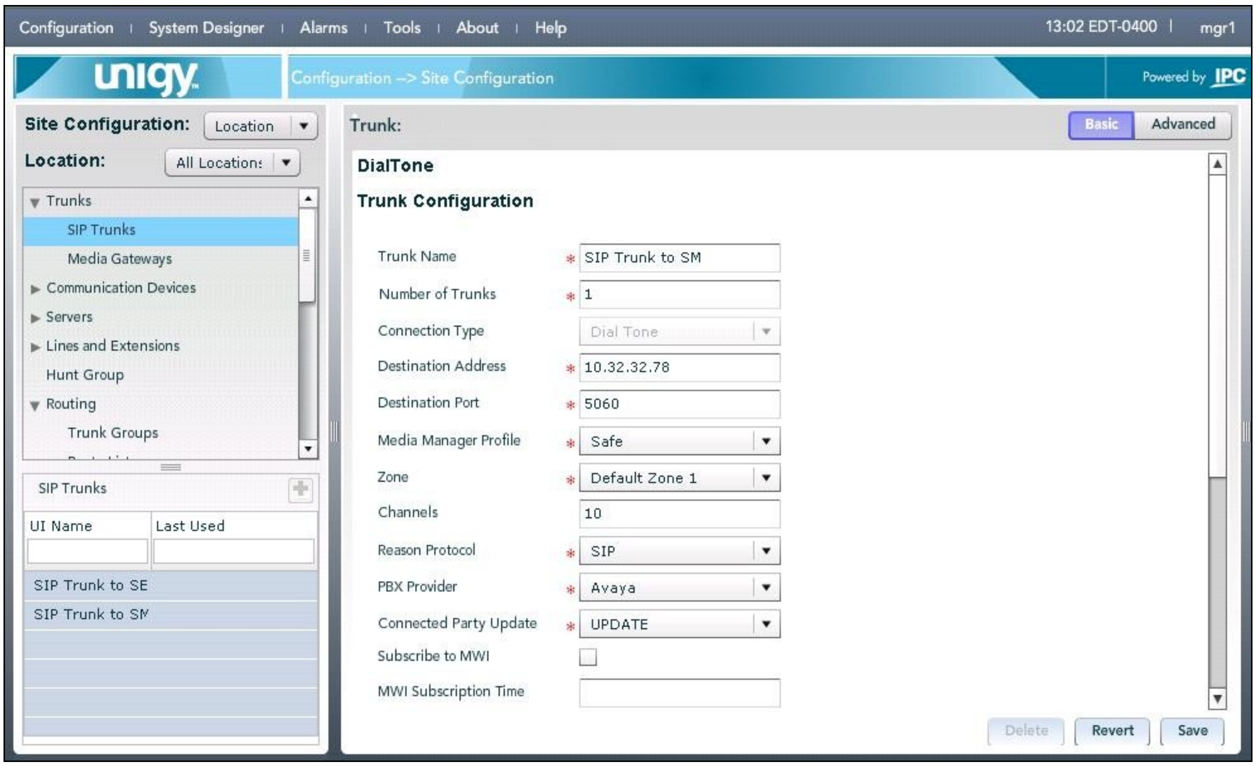
Select **Trunks > SIP Trunks** in the left pane, and click the **Add** icon in the lower left pane to add a new SIP trunk.

The screen below is displayed. Select "Dial Tone" from the **Select Connection Type** drop-down list.



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

- **Trunk Name:** A descriptive name.
- **Destination Address:** IP address of the Session Manager signaling interface.
- **Destination Port:** The port number from **Section 6.5.1**.
- **Zone:** An available zone, in this case "Default Zone 1".
- **Channels:** The number of SIP trunk group members from **Section 5.3**.
- **PBX Provider:** "Avaya"
- **Connected Party Update:** "UPDATE"

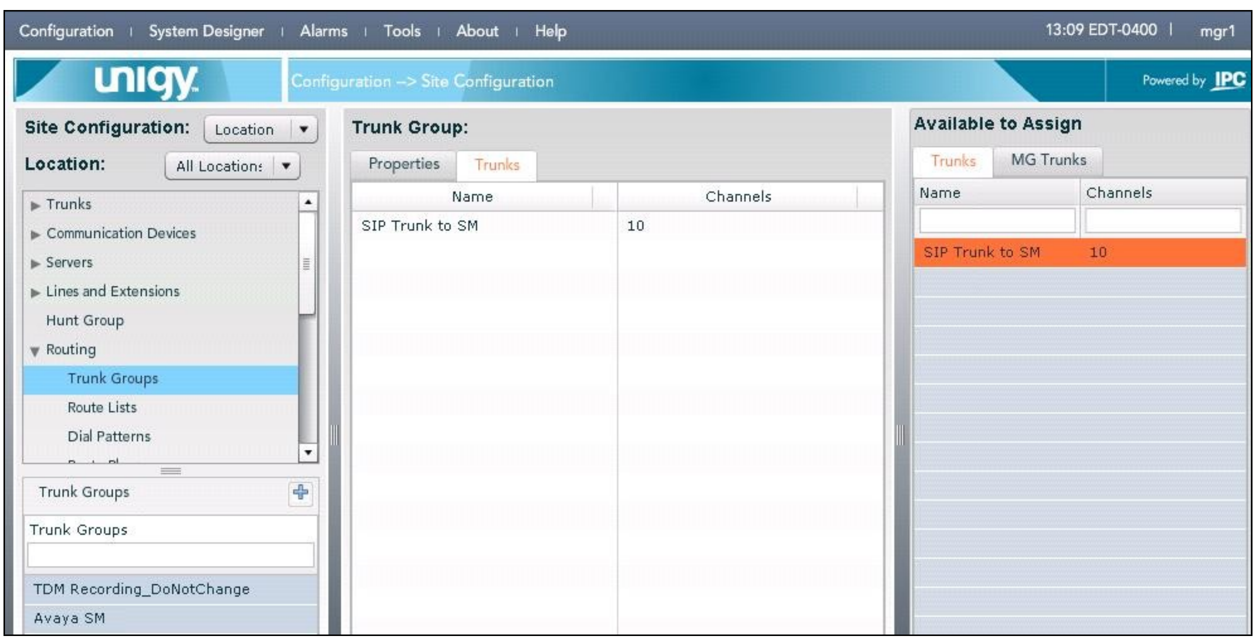


7.3. Administer Trunk Groups

Select **Routing > Trunk Groups** in the left pane, and click the **Add** icon in the lower left pane to add a new trunk group.

The **Trunk Group** screen is displayed in the right pane. In the **Properties** tab, enter a descriptive **Name**, and click **Save** (not shown). Select the **Trunks** tab in the right pane.

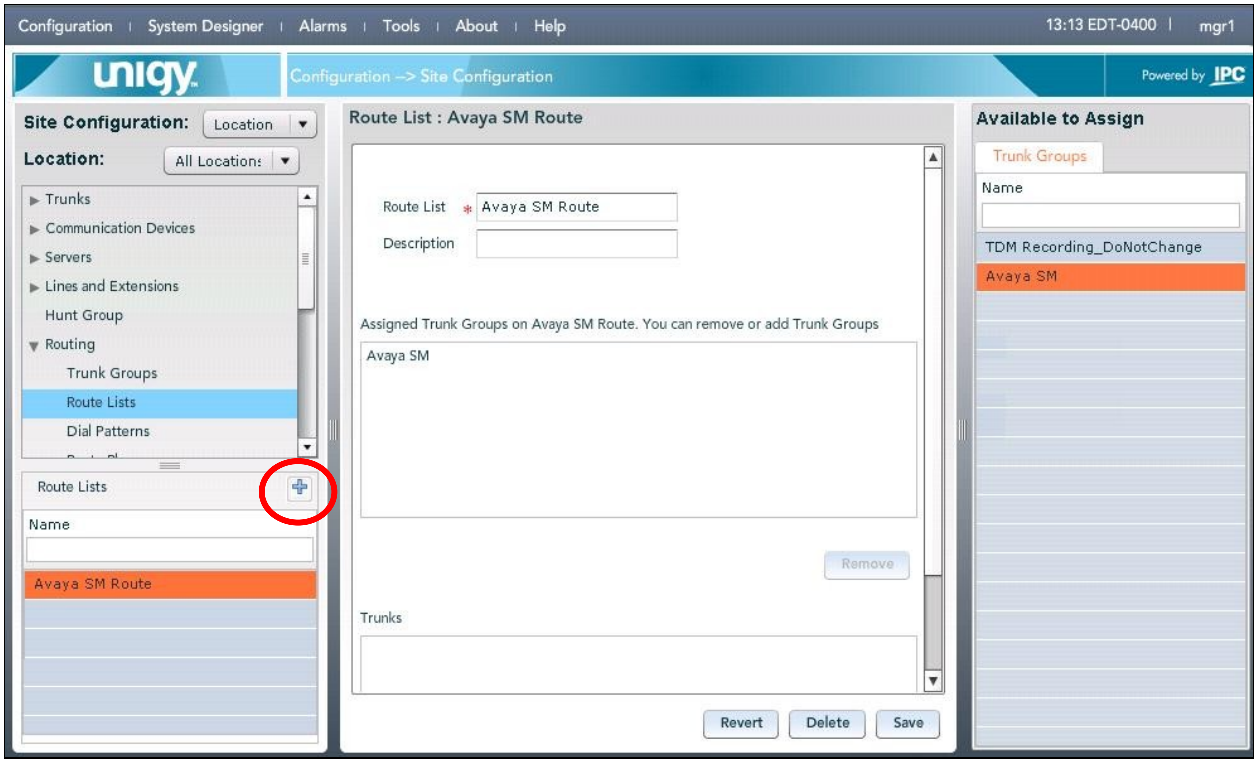
The screen is updated with three panes. In the rightmost pane, select the **Trunks** tab to display a list of trunks. Select the SIP trunk from **Section 7.2** in the rightmost pane and drag to the middle pane as shown below. Click **Save** (not shown).



7.4. Administer Route Lists

Select **Routing > Route Lists** in the left pane, and click the **Add** icon in the lower left pane to add a new route list.

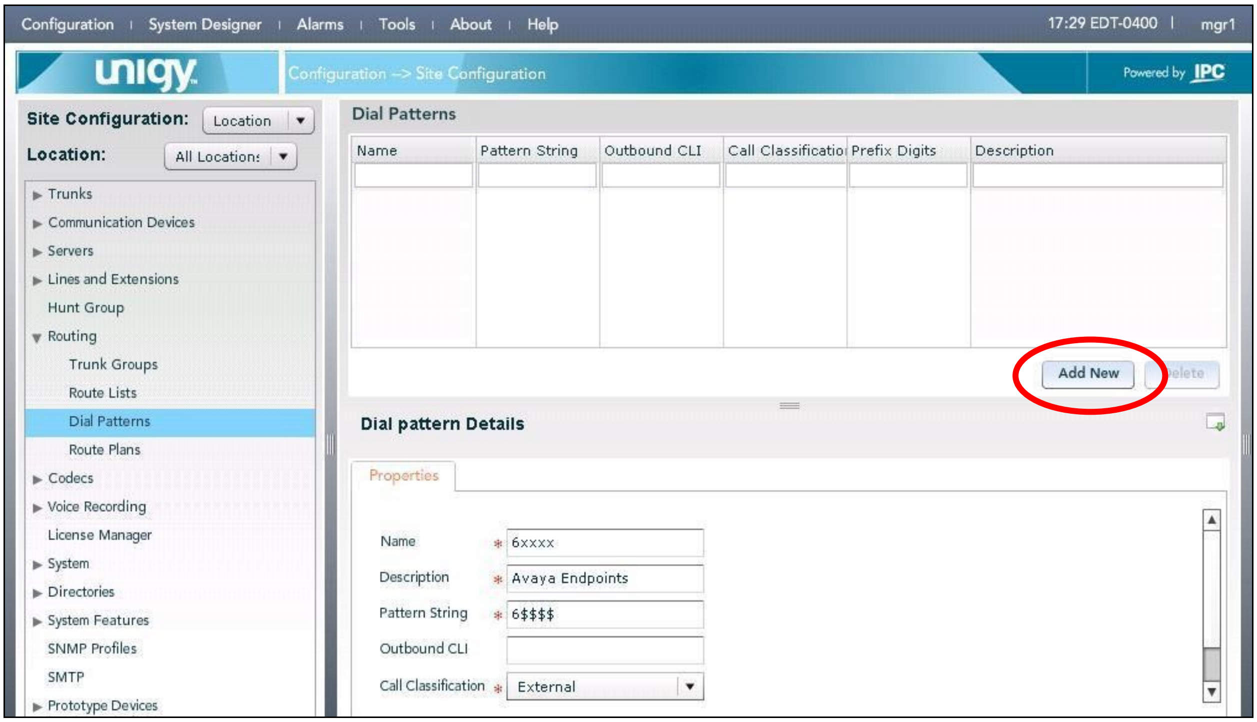
The **Route List** screen is displayed in the middle pane. For **Route List**, enter a descriptive name. In the right pane, select the trunk group from **Section 7.3** and drag into the **Assigned Trunk Groups on Route List** sub-section in the middle pane, as shown below. Click **Save**.



7.5. Administer Dial Patterns

Select **Routing > Dial Patterns** in the left pane, to display the **Dial Patterns** screen in the right pane. Click **Add New** in the upper right pane.

In the **Dial pattern Details** sub-section in the lower right pane, enter the desired **Name** and **Description**. For **Pattern String**, enter the dial pattern to match for Avaya endpoints, in this case "6$$$$" with "$" matching to any digit. For **Call Classification**, select "External". Click **Save** (not shown).

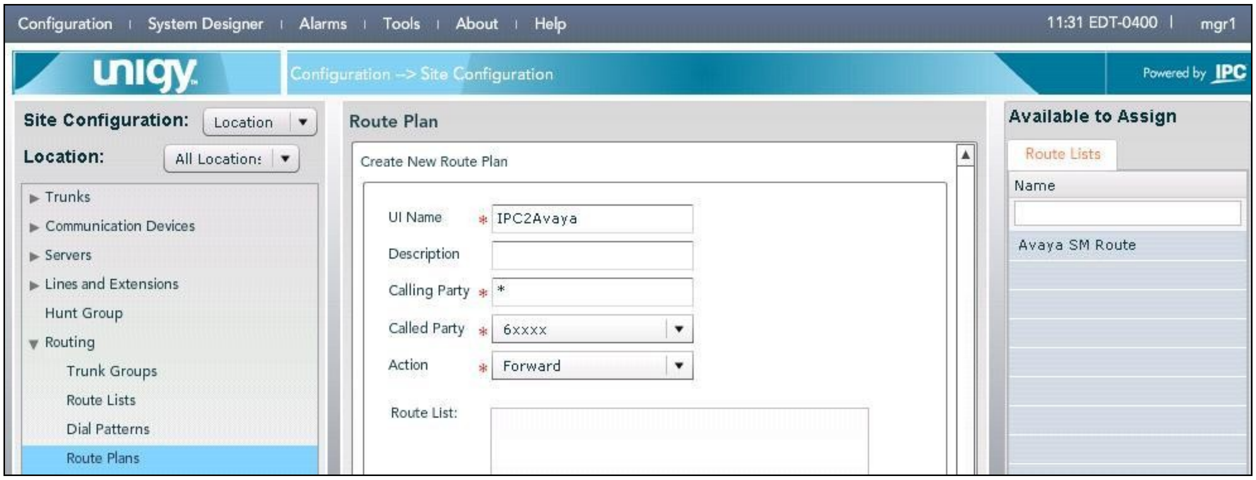


Repeat this section to add another dial pattern to reach the PSTN, and include any required prefix by Avaya Aura® Communication Manager. In the compliance testing, two dial patterns were created as shown below.

7.6. Administer Route Plans

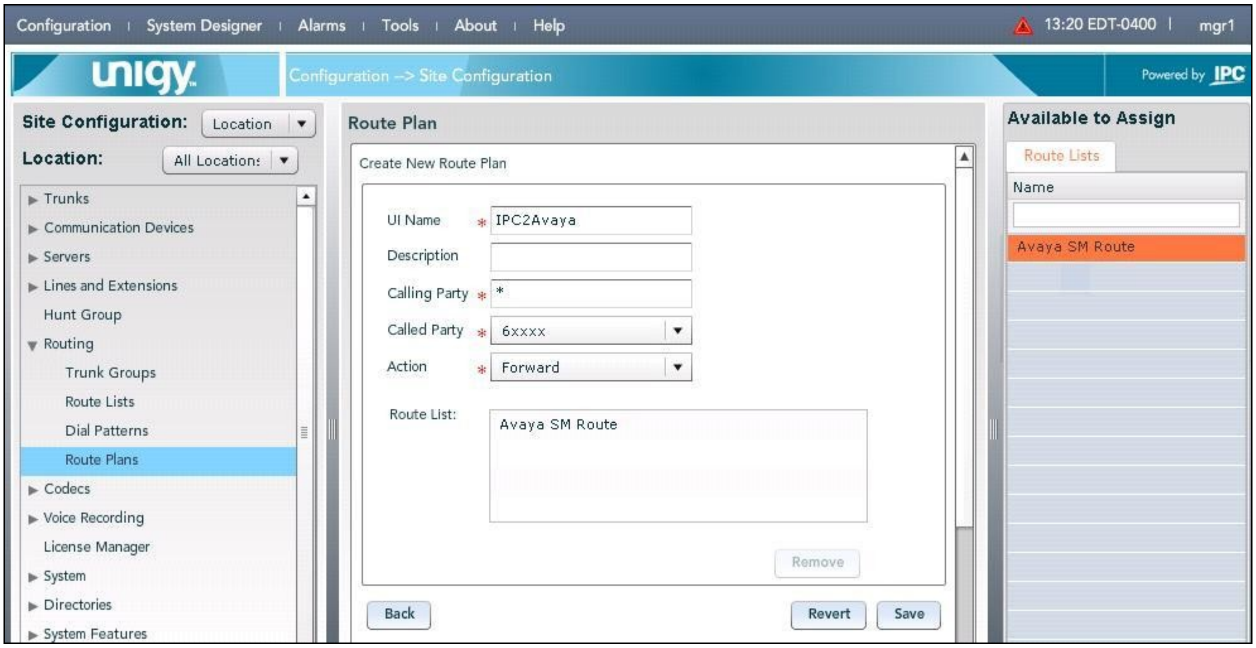
Select **Routing > Route Plans** in the left pane, and click **Add New** (not shown) in the right pane to create a new route plan.

The screen is updated with three panes, as shown below. In the **Route Plan** middle pane, enter a descriptive **UI Name** and optional **Description**. For **Calling Party**, enter "*" to denote any calling party from Unigy. For **Called Party**, select the dial pattern for Avaya endpoints from **Section 7.5**. Select "Forward" for **Action**, and click **Save** (not shown).



The screen is updated with the newly created route plan. Select the route plan, and click **Edit** toward the bottom of the screen (not shown).

The screen is updated with three panes again, as shown below. In the right pane, select the route list from **Section 7.4** and drag into the **Route List** sub-section in the middle pane, as shown below. Click **Save**.



Repeat this section to add another route plan for the PSTN. In the compliance testing, two route plans were created as shown below.

8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and IPC Unigy.

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.

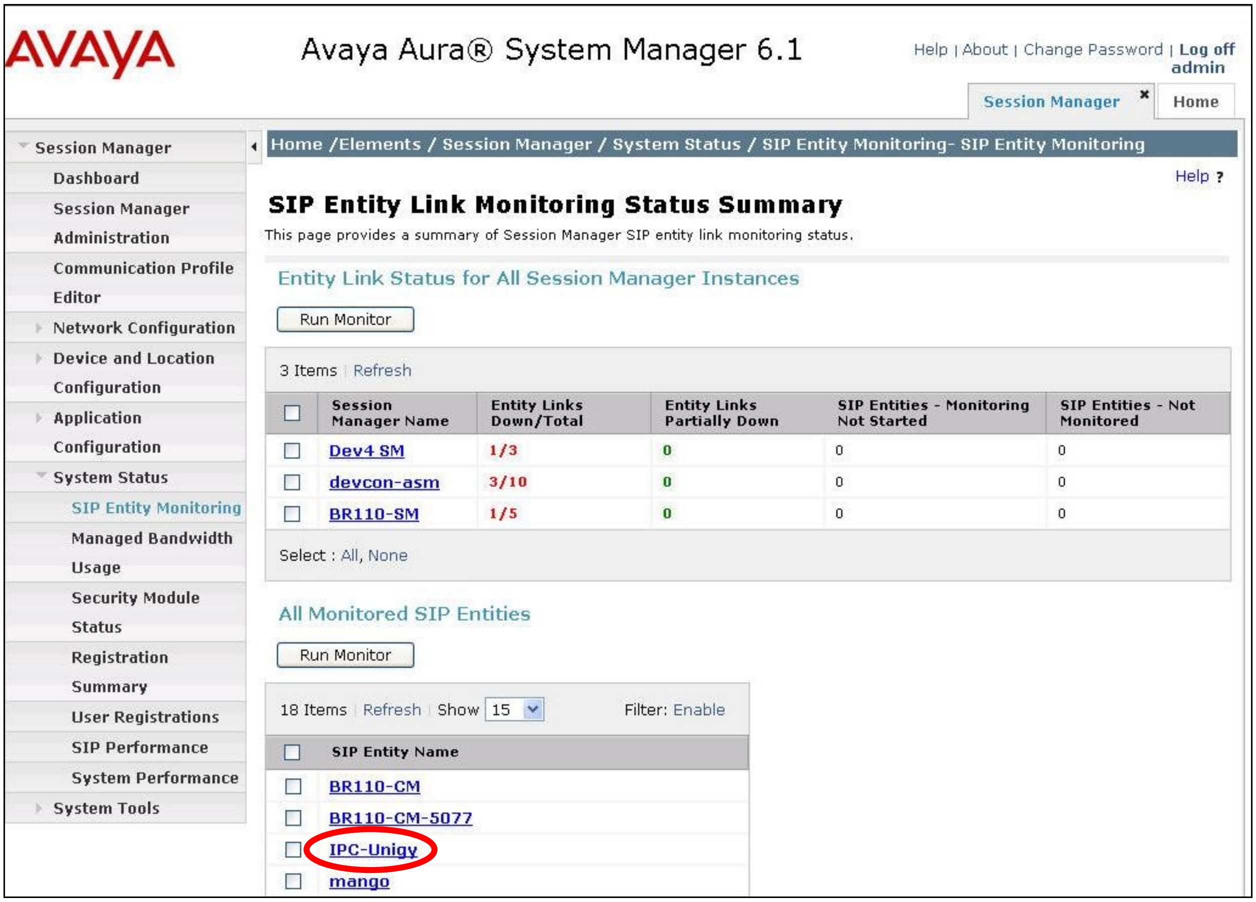
```
status trunk 77

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0077/001 T00135   in-service/idle    no
0077/002 T00136   in-service/idle    no
0077/003 T00137   in-service/idle    no
0077/004 T00138   in-service/idle    no
0077/005 T00139   in-service/idle    no
0077/006 T00140   in-service/idle    no
0077/007 T00141   in-service/idle    no
0077/008 T00142   in-service/idle    no
0077/009 T00143   in-service/idle    no
0077/010 T00144   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.

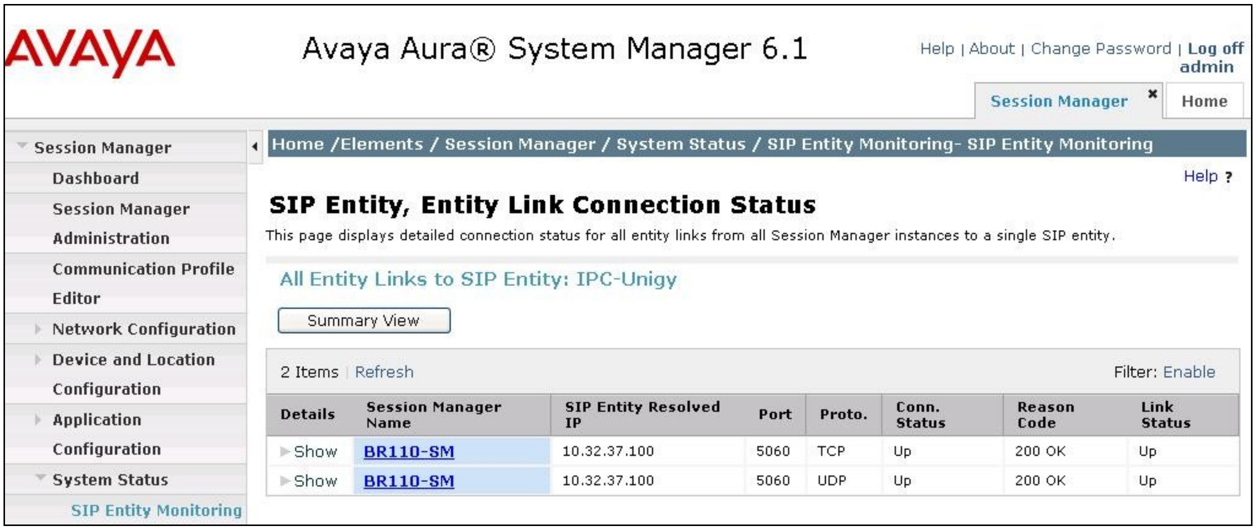
```
status signaling-group 77
                         STATUS SIGNALING GROUP

        Group ID: 77
      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 on the IPC 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 IPC Unigy

Make a call from an IPC turret user to an Avaya endpoint. Verify that the call can be connected with two-way talk paths.

9. Conclusion

These Application Notes describe the configuration steps required for IPC Unigy to successfully interoperate with Avaya Aura® Communication Manager 6.0.1 using 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 6.0, Release 6.0, June 2010, available at http://support.avaya.com.

2. *Administering Avaya Aura™ Session Manager*, Document Number 03-603324, Issue 3, Release 6.0, August 2010, available at http://support.avaya.com.

3. *Unigy 1.1 System Configuration*, Part Number B02200187, Release 00, upon request to IPC Support.

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.