Avaya Solution & Interoperability Test Lab

Application Notes for IPC UnigyV2 with Avaya Aura® Communication Manager 6.01 and Avaya Aura® Session Manager 6.2 using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for IPC UnigyV2 to interoperate with Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.2 using SIP trunks.

IPC UnigyV2 is a trading communication solution. In the compliance testing, IPC UnigyV2 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 UnigyV2 to interoperate with Avaya Aura® Communication Manager via Avaya Aura® Session Manager.

The Unigy Platform is a unified trading communications system designed specifically to make the entire trading ecosystem more productive, intelligent and efficient. Based on an SIP-enabled, open and distributed architecture, Unigy utilizes the latest, standards-based technology to create a groundbreaking, innovative Unified Trading Communications (UTC) solution.

Unigy offers a portfolio of devices and applications that serve the entire trading workflow, across the front, middle and back offices.

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 UnigyV2.

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 basic calls, display, G.711MU/ G.729, 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 UnigyV2 to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to IPC UnigyV2.

2.2. Test Results

All test cases were executed and verified. The following were the observations on IPC UnigyV2 from the compliance testing.

- Even when IPC UnigyV2 is configured with UDP, TCP protocol must be configured on Avaya Session Manager as UnigyV2 switches over to use TCP for diversions.
- During the compliance test, Network Call Redirection (REFER) was disabled on the SIP trunk between Communication Manager6.0.1 and Session Manager 6.2.

2.3. Support

Technical support on IPC UnigyV2 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 UnigyV2 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. For compliance testing, a single server hosted the Media Manager and Converged Communication Manager.

SIP trunks are used from IPC UnigyV2 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 for dialing between Central and Remote sites. Unique extension ranges were associated with Avaya Aura® Communication Manager users at the Central site (4200x and 4202x), and IPC turret users at the Remote site (7205x).

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

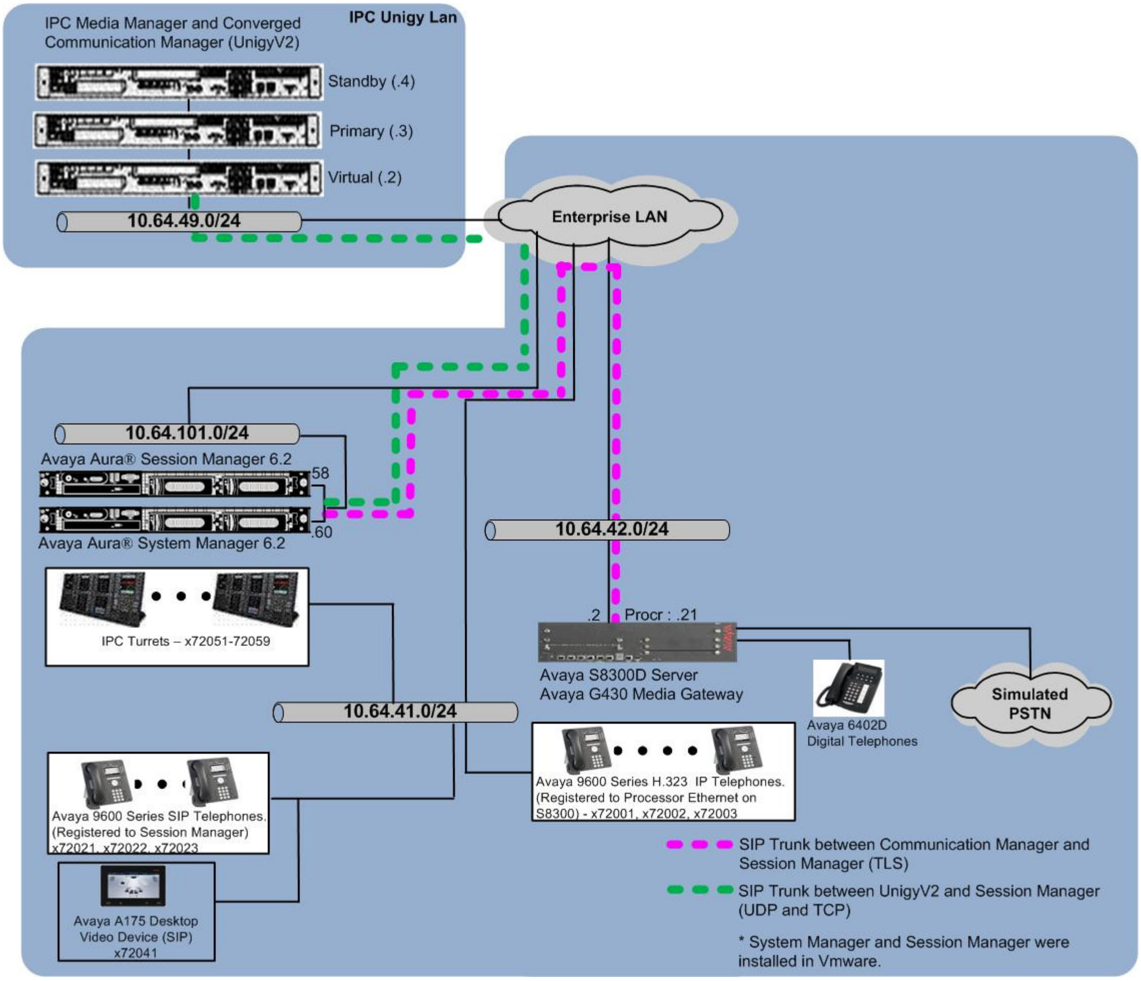

IPC Media Manager and Converged Communication Manager (UnigyV2)
IPC Unigy Lan
Standby (.4)
Primary (.3)
Virtual (.2)
10.64.49.0/24
Enterprise LAN
10.64.101.0/24
Avaya Aura® Session Manager 6.2
.58
Avaya Aura® System Manager 6.2
.60
10.64.42.0/24
IPC Turrets – x72051-72059
.2
Procr : .21
Avaya S8300D Server
Avaya G430 Media Gateway
Avaya 6402D Digital Telephones
Simulated PSTN
10.64.41.0/24
Avaya 9600 Series SIP Telephones. (Registered to Session Manager) x72021, x72022, x72023
Avaya 9600 Series H.323 IP Telephones. (Registered to Processor Ethernet on S8300) - x72001, x72002, x72003
Avaya A175 Desktop Video Device (SIP) x72041
SIP Trunk between Communication Manager and Session Manager (TLS)
SIP Trunk between UnigyV2 and Session Manager (UDP and TCP)
* System Manager and Session Manager were installed in Vmware.

Figure 1: Test Configuration of IPC UnigyV2

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 S8300D Server	R016x.00.1.510.1-19940
Avaya G430 Media Gateway	31.26
Avaya Aura® Session Manager	6.2.3.0.623006
Avaya Aura® System Manager	6.2.12.9
Avaya 96xx IP Telephone (H.323)	3.1
Avaya 96xx IP Telephone (SIP)	2.6.4
Avaya A175 Desktop Video Device (SIP)	1.0.2
IPC UnigyV2 • Media Manager • Converged Communication Manager	 02.00.00.00.1888 02.00.00.00.1888

5. Configure Avaya Aura® Communication Manager

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

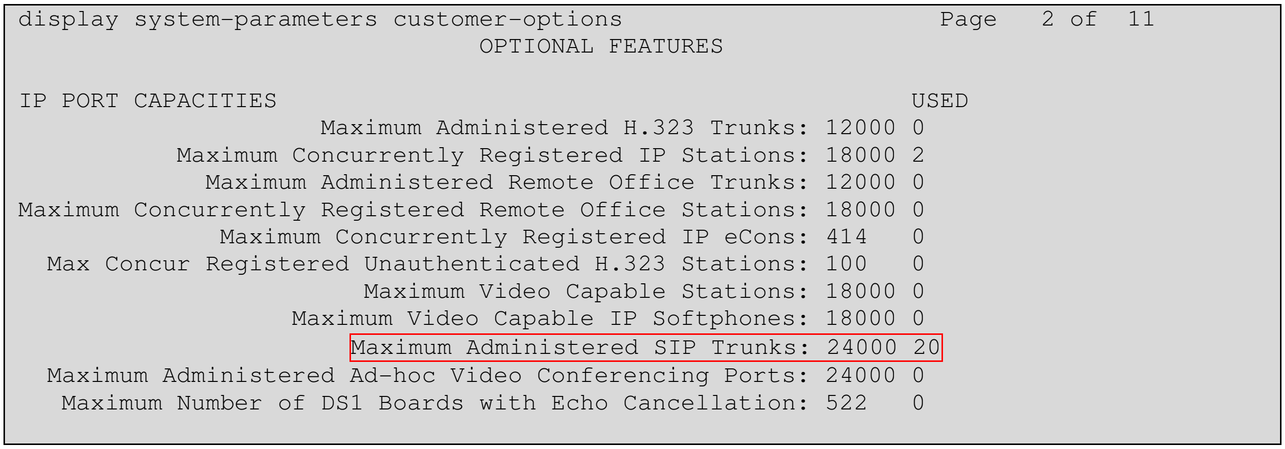
- Verify Communication Manager license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR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

In the compliance testing, a separate configuration 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.

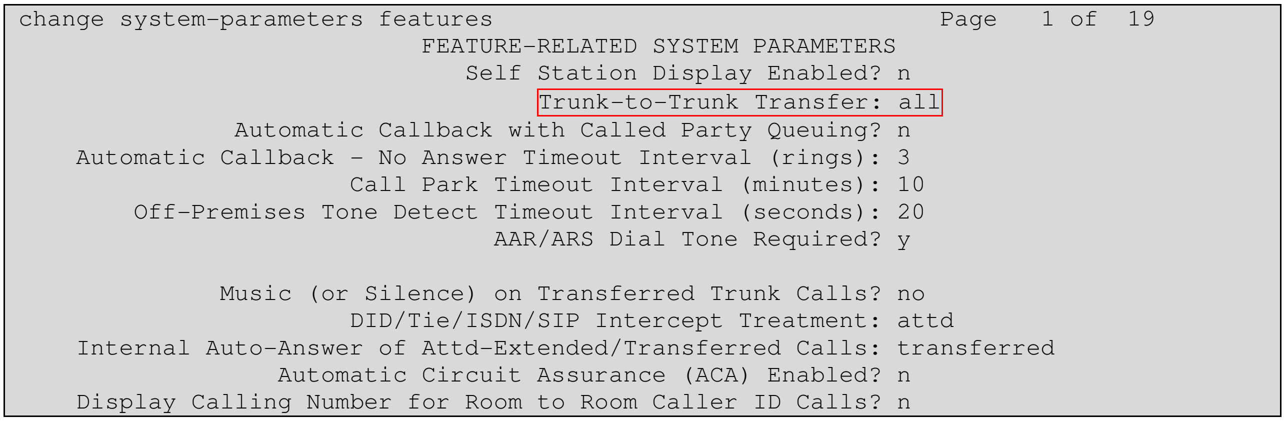


```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 2
             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 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 20
  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.

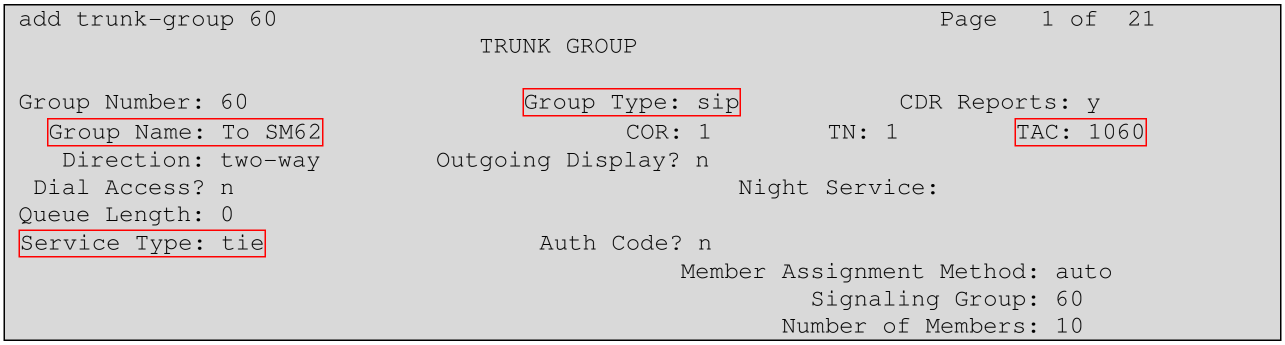


```
change system-parameters features                               Page   1 of   19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback 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: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
    Display Calling Number for Room to Room Caller ID Calls? n
```

5.3. Administer SIP Trunk Group

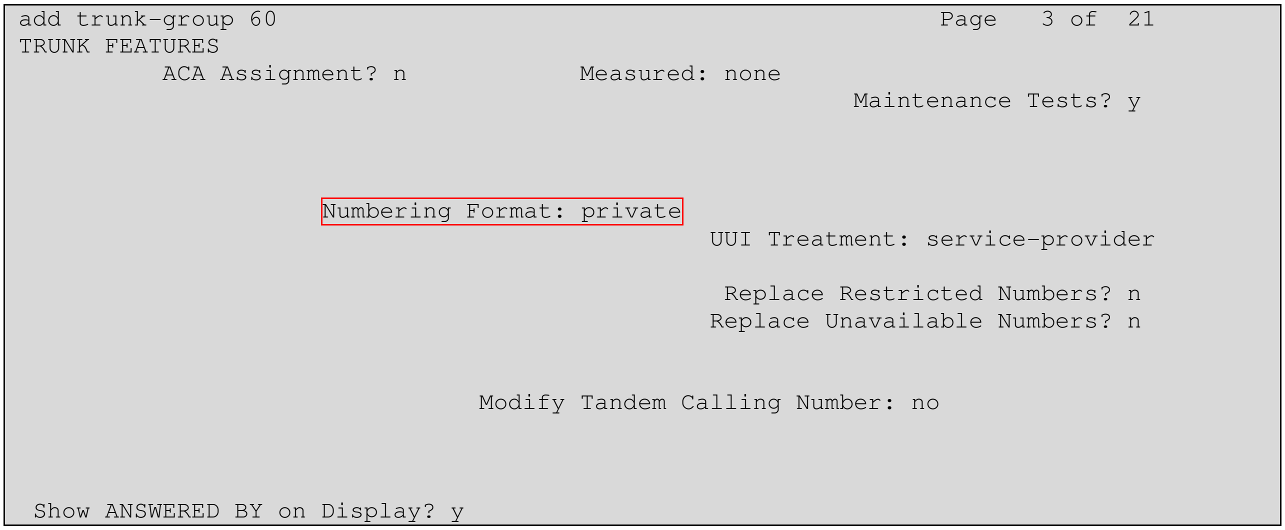
Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “60”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



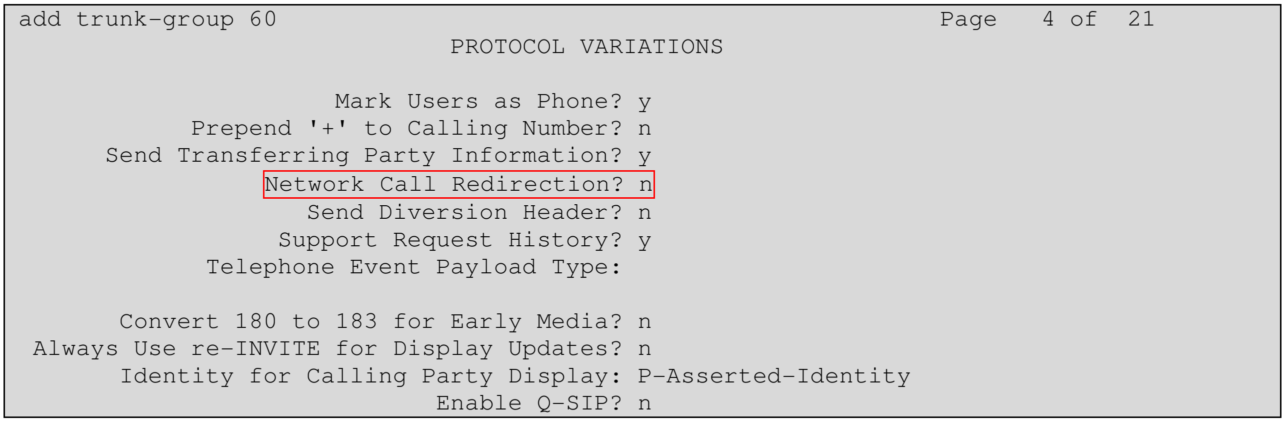
```
add trunk-group 60                                              Page   1 of   21
                                TRUNK GROUP

Group Number: 60                   Group Type: sip           CDR Reports: y
  Group Name: To SM62                     COR: 1        TN: 1        TAC: 1060
   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: 60
                                                     Number of Members: 10
```

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



```
add trunk-group 60                                              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
```

Navigate to **Page 4**, and set **Network Call Redirection** “n”, since REFER doesn’t work with IPC.

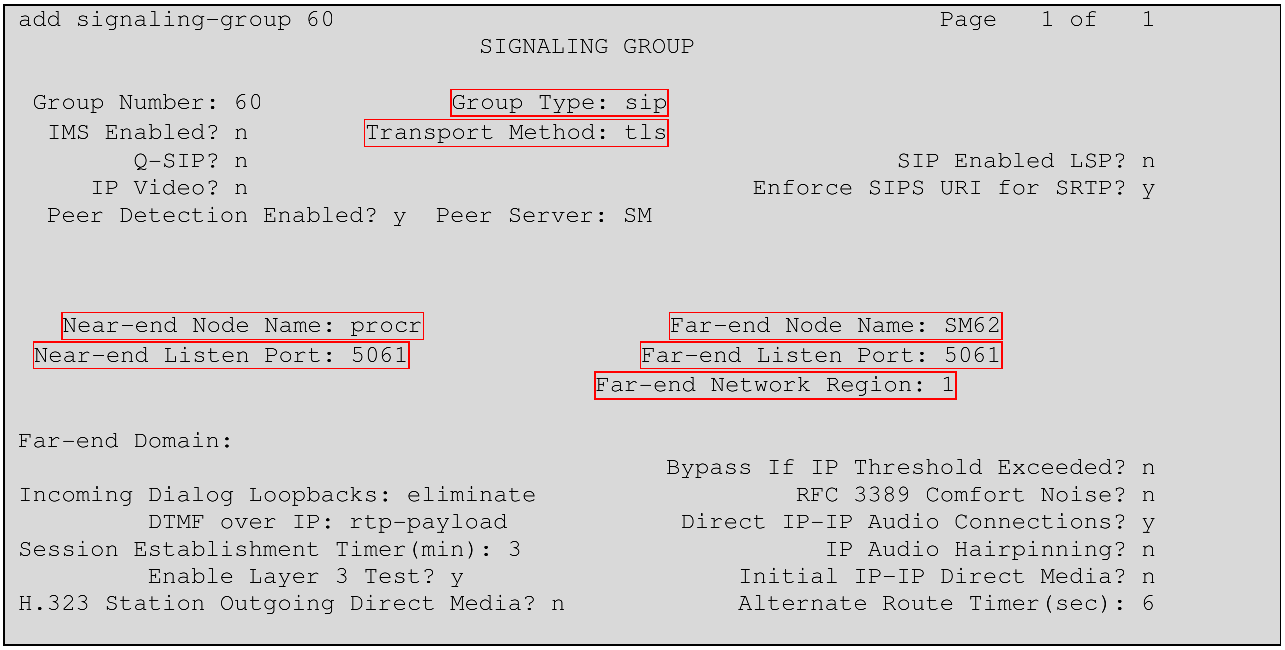


```
add trunk-group 60                                              Page   4 of  21
                              PROTOCOL VARIATIONS

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

       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                             Enable Q-SIP? n
```

5.4. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "60". 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.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration on Communication Manager.
- **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 UnigyV2.

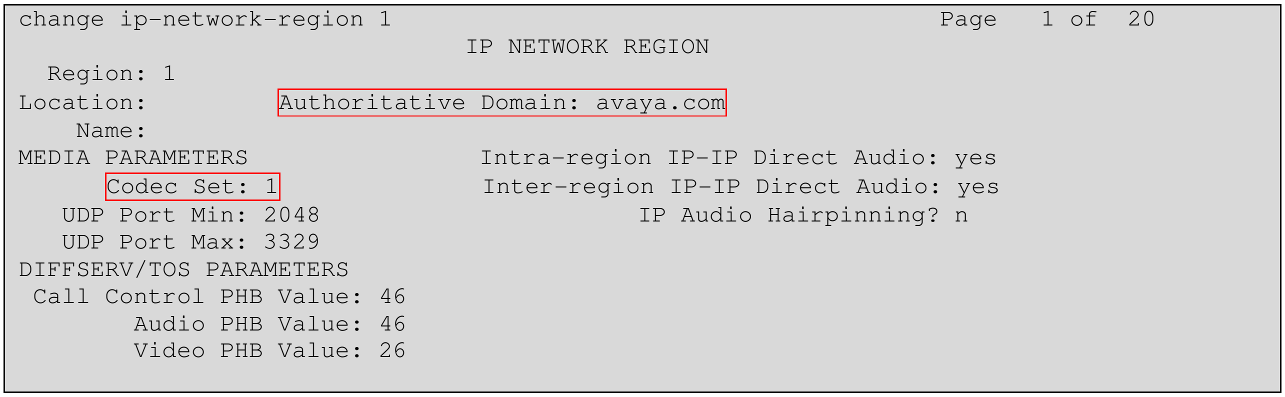


```
add signaling-group 60                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: SM62
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            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 0**.

For **Authoritative Domain**, enter “avaya.com”. Enter a descriptive **Name**. For **Codec Set**, enter an available codec set number for integration with IPC UnigyV2.

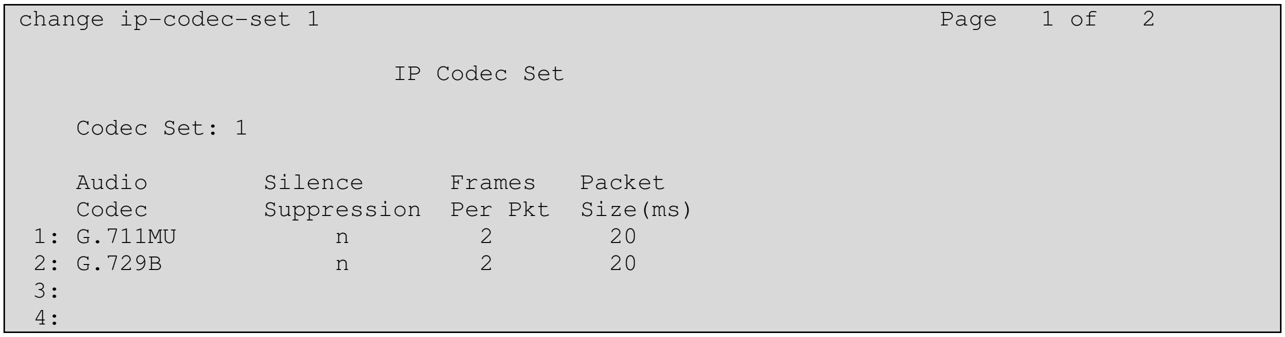


```
change ip-network-region 1                                    Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

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 UnigyV2 supports the G.711 and G.729 codec variants, and requires the codec order on Avaya to match the codec order specified on IPC UnigyV2.

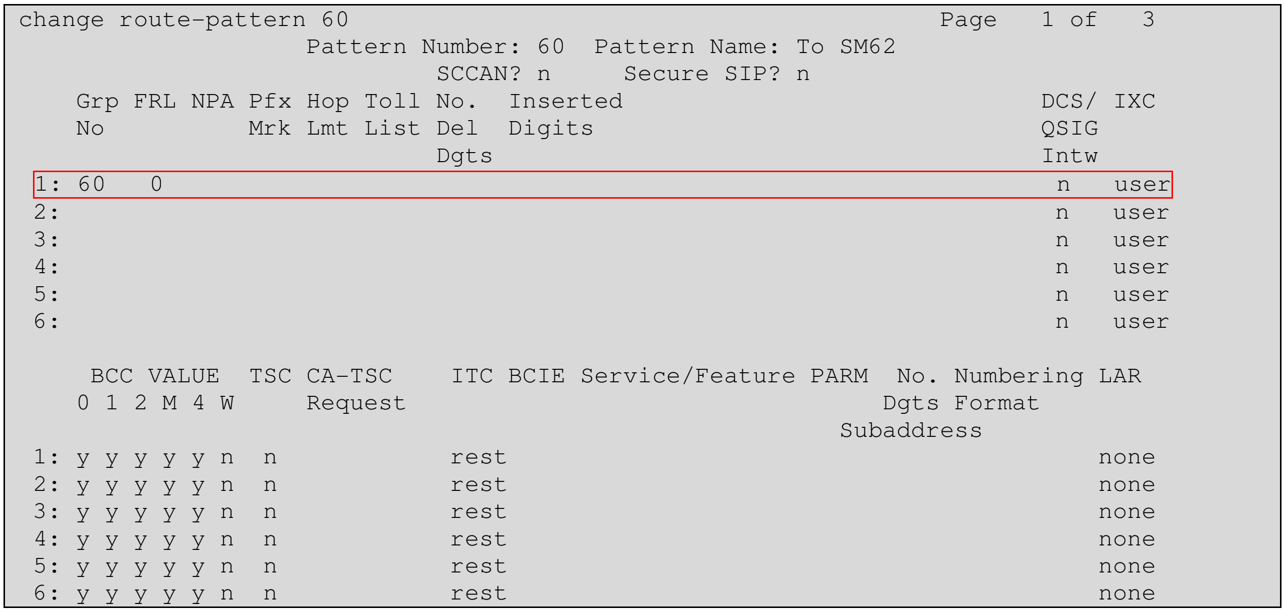


```
change ip-codec-set 1                                         Page   1 of   2

                          IP Codec Set

    Codec Set: 1

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

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 “60”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

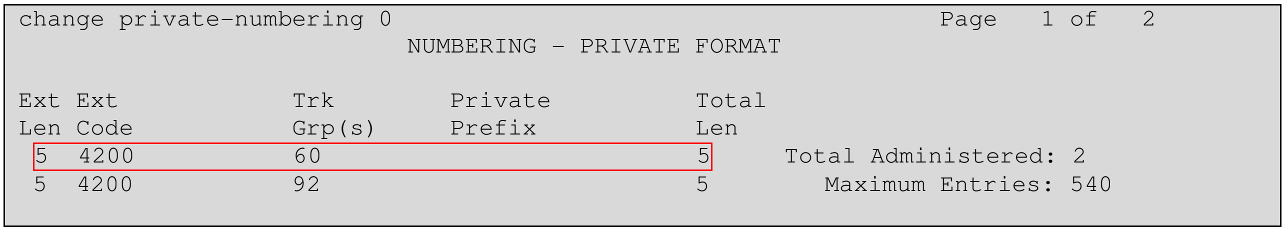
- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 0**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.



```
change route-pattern 60                                         Page    1 of    3
                    Pattern Number: 60  Pattern Name: To SM62
                             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: 60   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
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.8. 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 0**. In the example shown below, all calls originating from a 5-digit extension beginning with 4200 and routed to trunk group 92 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  4200           60                          5     Total Administered: 2
 5  4200           92                          5        Maximum Entries: 540
```

5.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 7205x 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 7205x, 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
 1303               11   0                      ars   n
 7200               5    0                      aar   n
 7202               5    0                      aar   n
 7204               5    0                      aar   n
 7205               5    0                      aar   n
                                                      n
```

5.10. Administer AAR Analysis

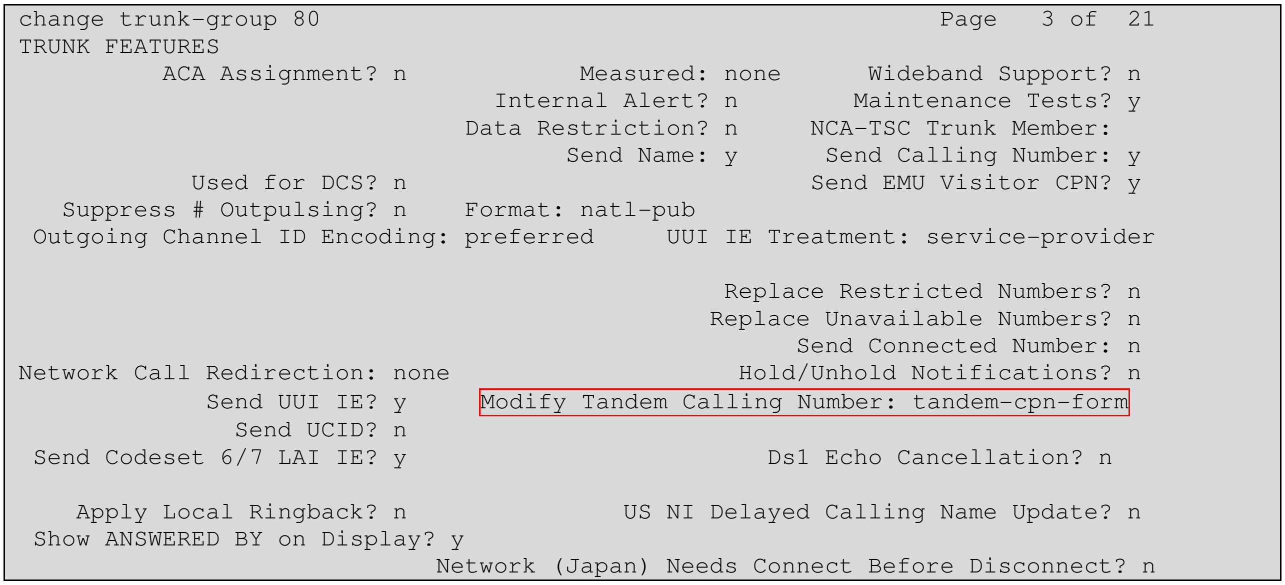
Use the “change aar analysis 0” command, and add an entry to specify how to route calls to 7205x. In the highlighted example shown below, calls with digits 7205x will be routed using route pattern “60” from **Section 0**.

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

          Dialed           Total     Route     Call    Node    ANI
          String         Min  Max   Pattern    Type    Num     Reqd
    2                    7     7      999      aar             n
    3                    7     7      999      aar             n
    4                    7     7      999      aar             n
    5                    7     7      999      aar             n
    6                    7     7      999      aar             n
    7200                 5     5      92       aar             n
    7202                 5     5      60       aar             n
    7204                 5     5      92       aar             n
    7205                 5     5      60       aar             n
                                                               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 “80”.

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.

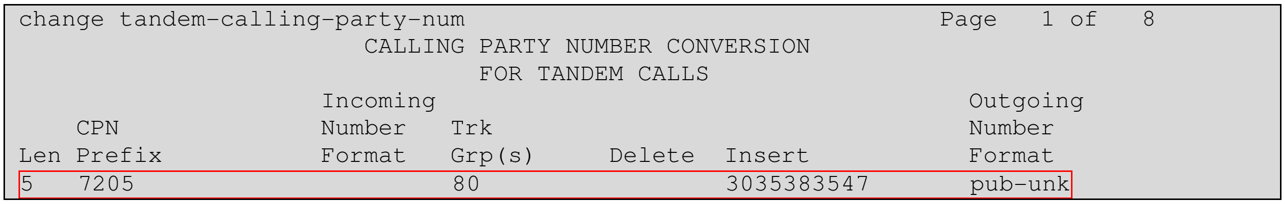


```
change trunk-group 80                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n           Measured: none      Wideband Support? n
                                Internal Alert? n        Maintenance Tests? y
                              Data Restriction? n     NCA-TSC Trunk Member:
                                     Send Name: y      Send Calling Number: y
            Used for DCS? n                           Send EMU Visitor CPN? y
   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: 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
```

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 7205 and routed to trunk group 80 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
                     Incoming                                     Outgoing
    CPN              Number   Trk                                 Number
Len Prefix           Format   Grp(s)     Delete  Insert           Format
5   7205                      80                 3035383547       pub-unk
```

6. Configure Avaya Aura® Session Manager

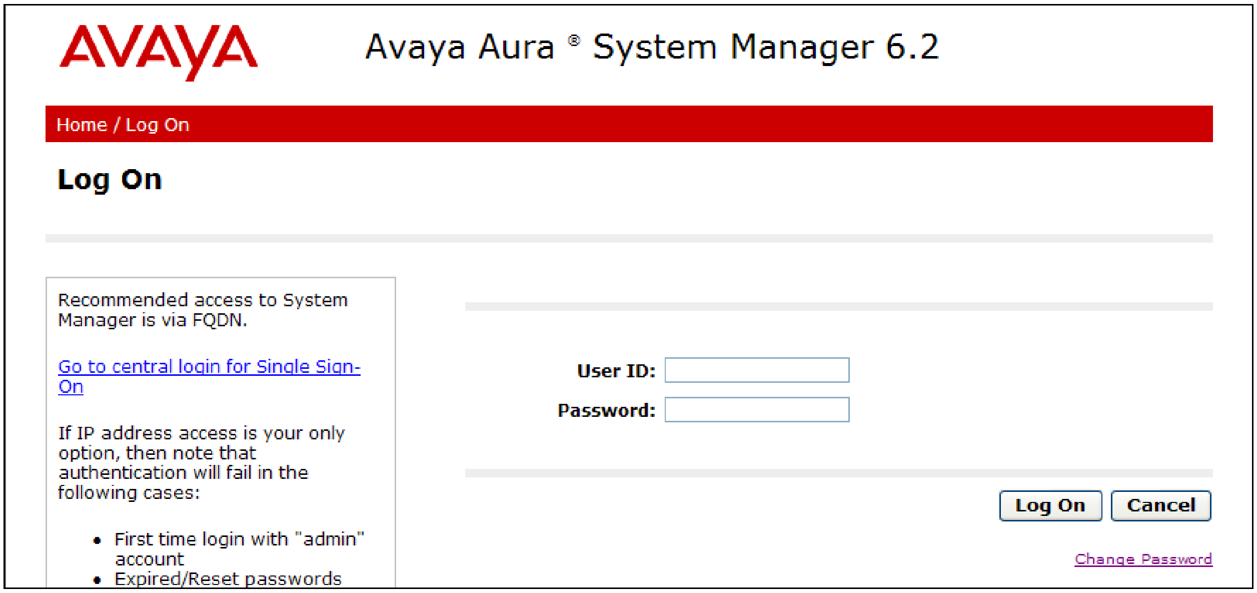
For the compliance test, Session Manager and System manager were installed and configured using VMware. The installation steps of the VMware, System manager, and Session Manager are not part of Application Notes and will not be discussed. An assumption is made that basic configurations in System Manager are already completed.

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.



Recommended access to System Manager is via FQDN.

Go to central login for Single Sign-On

If IP address access is your only option, then note that authentication will fail in the following cases:

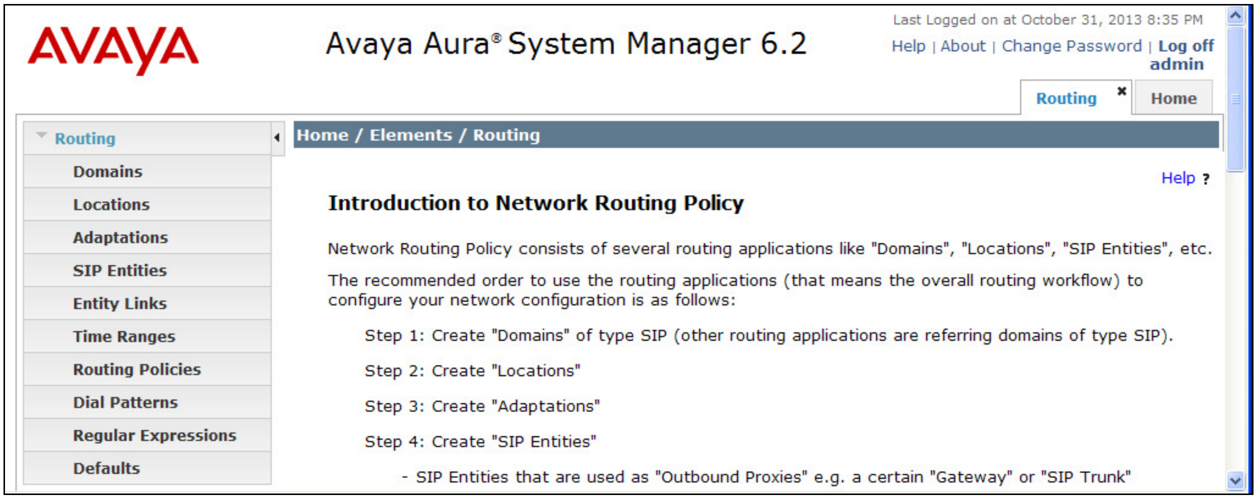
- First time login with "admin" account
- Expired/Reset passwords

User ID:

Password:

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.



Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

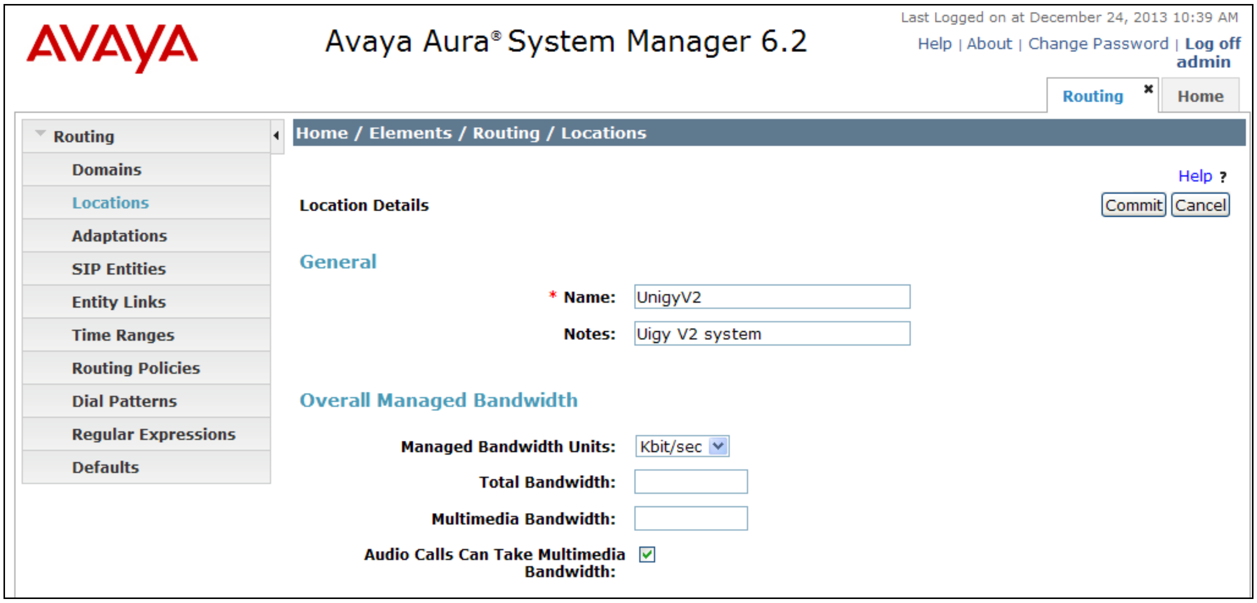
Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"

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. The following shows the location for Communication Manager 6.01.
After the changes, click the **Commit** button.

Repeat these steps to add other locations.



Location Details

General

* Name:	UnigyV2
Notes:	Uigy V2 system

Overall Managed Bandwidth

Managed Bandwidth Units:	Kbit/sec
Total Bandwidth:	
Multimedia Bandwidth:	
Audio Calls Can Take Multimedia Bandwidth:	☑

6.3. Administer Adaptations

For this compliance test, no adaptation was utilized.

6.4. Administer SIP Entities

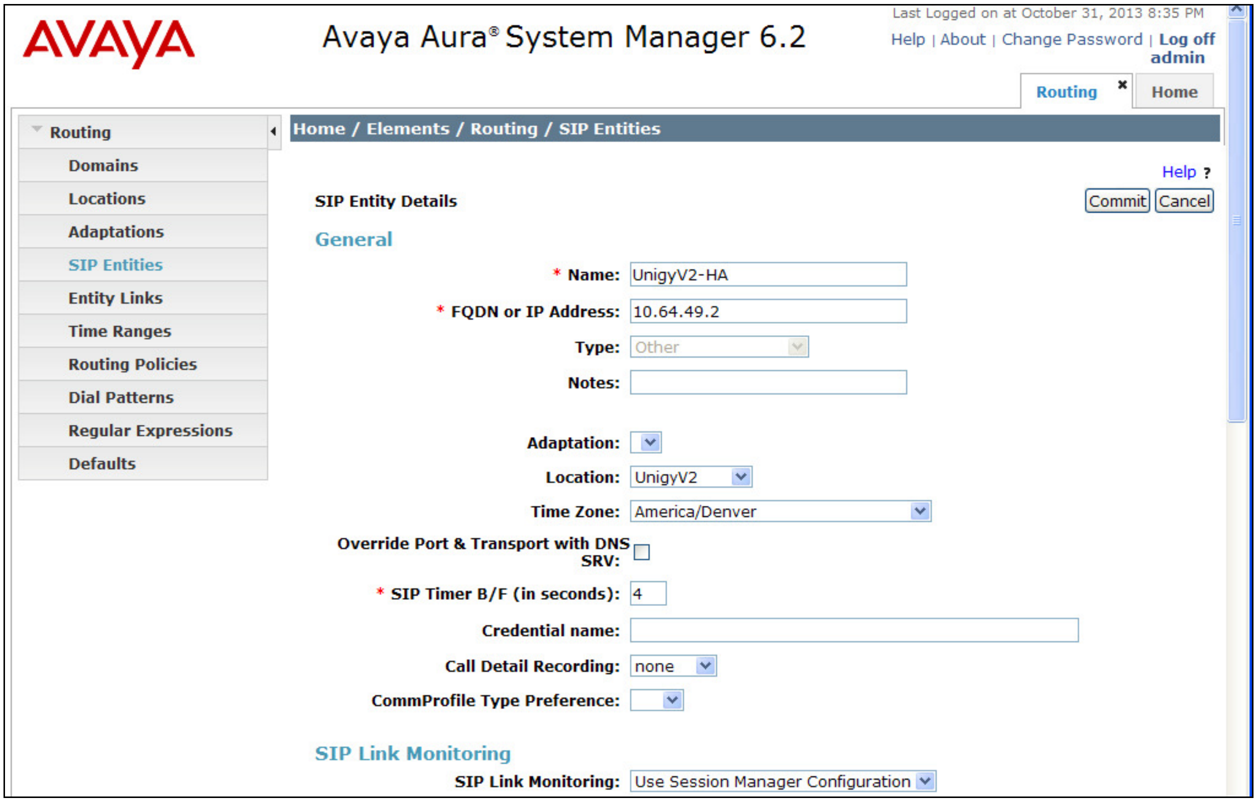
Add two new SIP entities, one for IPC, and another for the new SIP trunks for Communication Manager.

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"
- **Location:** Select the IPC location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



SIP Entity Details

General

* Name: UnigyV2-HA

* FQDN or IP Address: 10.64.49.2

Type: Other

Notes:

Adaptation:

Location: UnigyV2

Time Zone: America/Denver

Override Port & Transport with DNS SRV: □

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

CommProfile Type Preference:

SIP Link Monitoring

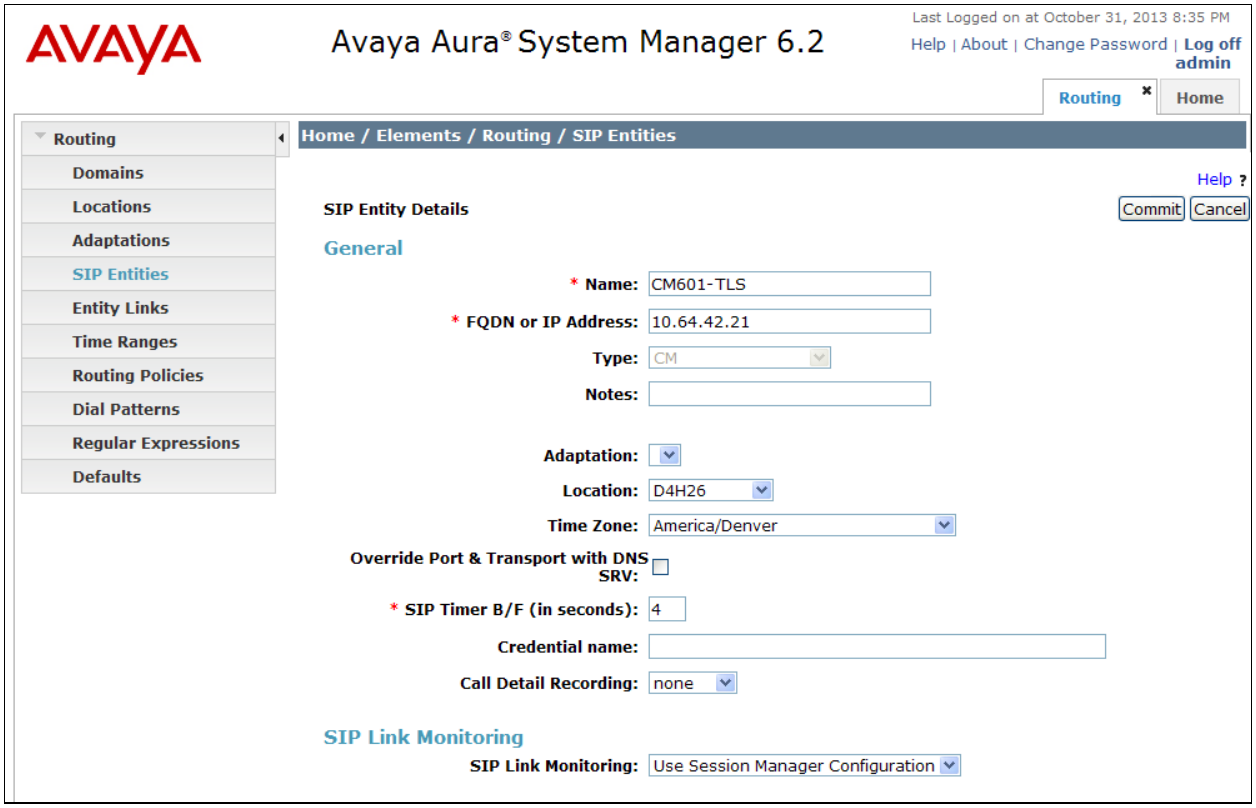
SIP Link Monitoring: Use Session Manager Configuration

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 or procr.
- **Type:** "CM"
- **Notes:** Any descriptive notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



SIP Entity Details

General

* Name: CM601-TLS

* FQDN or IP Address: 10.64.42.21

Type: CM

Notes:

Adaptation:

Location: D4H26

Time Zone: America/Denver

Override Port & Transport with DNS SRV: □

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

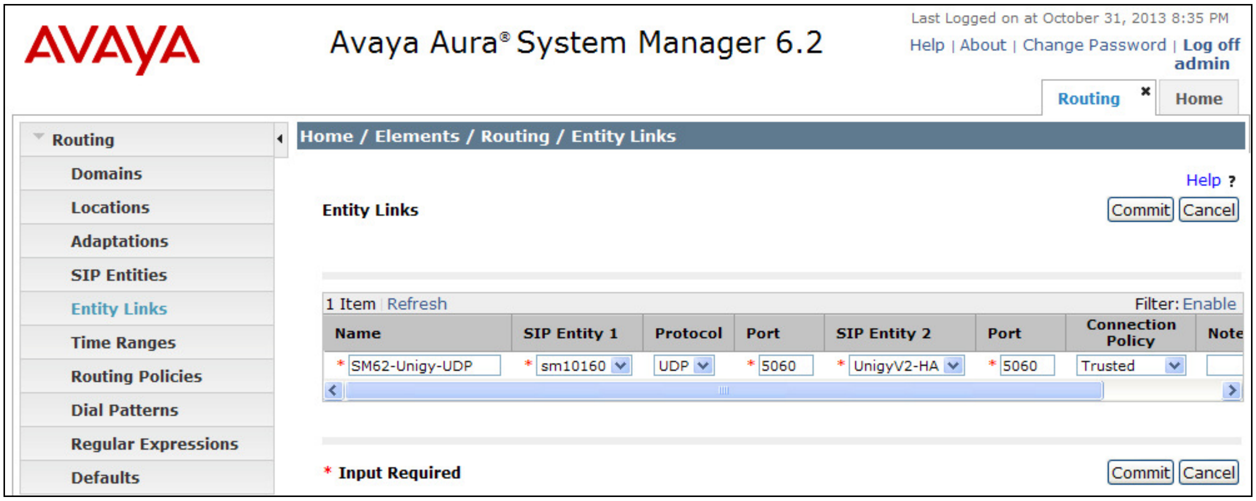
6.5. Administer Entity Links

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 "sm-10160".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The IPC entity name from **Section 6.4.1**.
- **Port:** "5060"
- **Connection Policy** Retain the default setting (trusted).

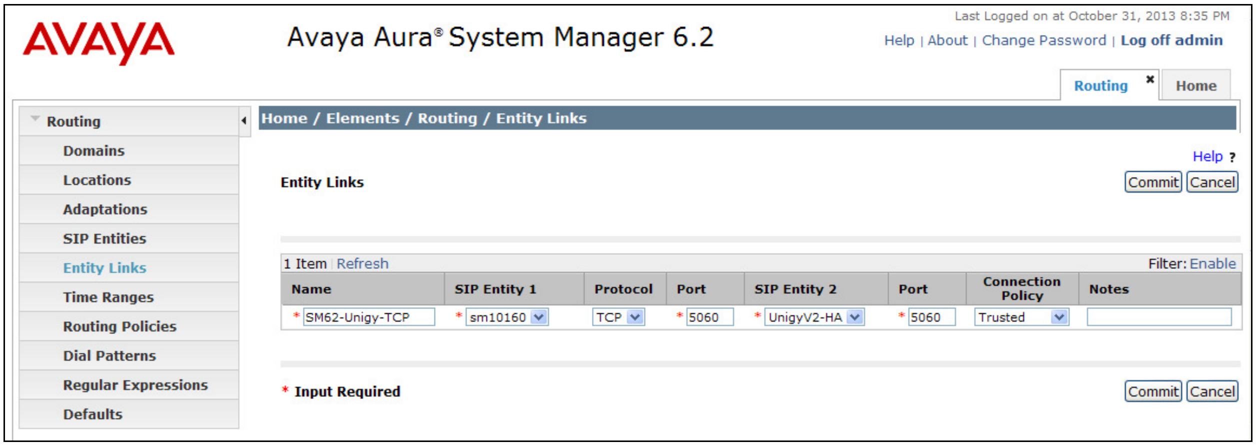


Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Note
* SM62-Unigy-UDP	* sm10160	UDP	* 5060	* UnigyV2-HA	* 5060	Trusted	

* Input Required

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



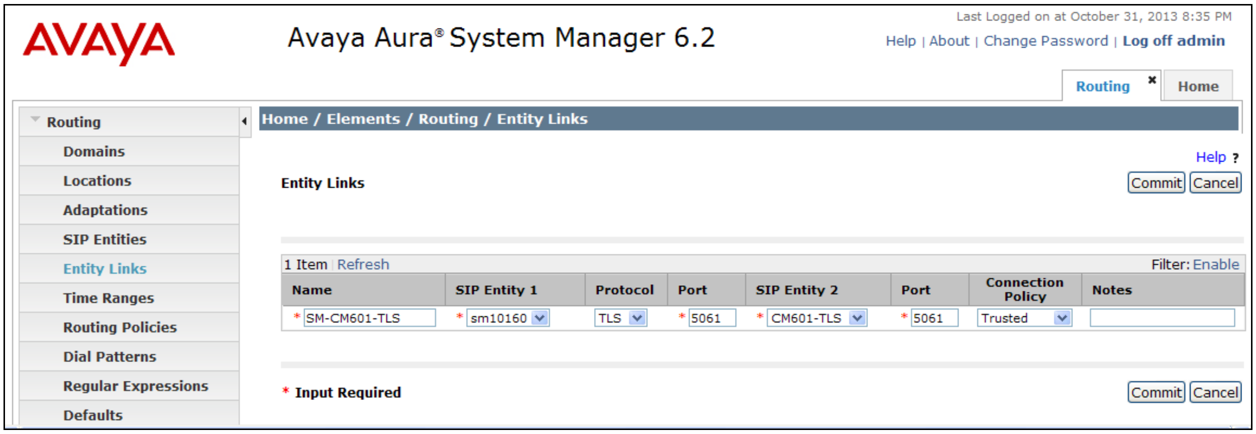
Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM62-Unigy-TCP	* sm10160	TCP	* 5060	* UnigyV2-HA	* 5060	Trusted	

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 "sm10160".
• **Protocol:**	The signaling group transport method from **Section 0**.
• **Port:**	The signaling group listen port number from **Section 0**.
• **SIP Entity 2:**	The Communication Manager entity name from **Section 6.4.2**.
• **Port:**	The signaling group listen port number from **Section 0**.
• **Connection Policy**	Retain the default setting (trusted).



Entity Links

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM-CM601-TLS	* sm10160	TLS	* 5061	* CM601-TLS	* 5061	Trusted	

6.6. Administer Routing Policies

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

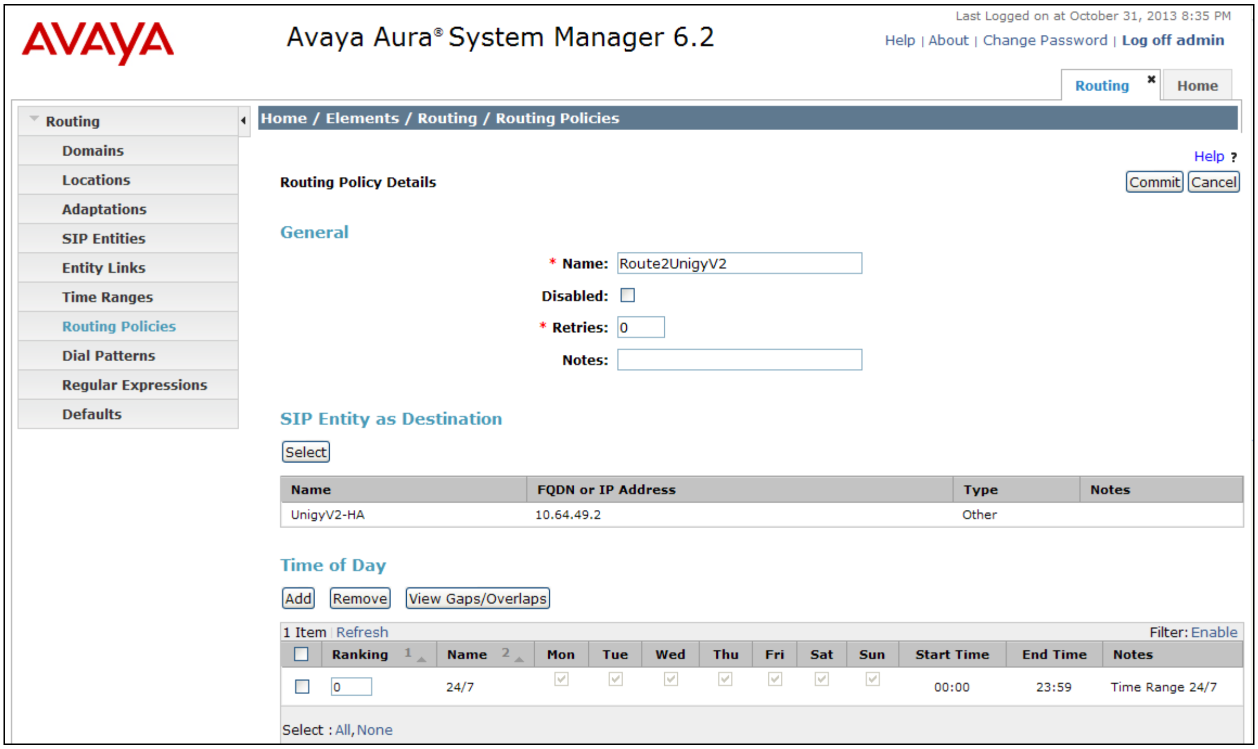
6.6.1. IPC Routing Policy

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.



Routing Policy Details

General

* Name:	Route2UnigyV2
Disabled:	□
* Retries:	0
Notes:	

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
UnigyV2-HA	10.64.49.2	Other	

Time of Day

□	Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

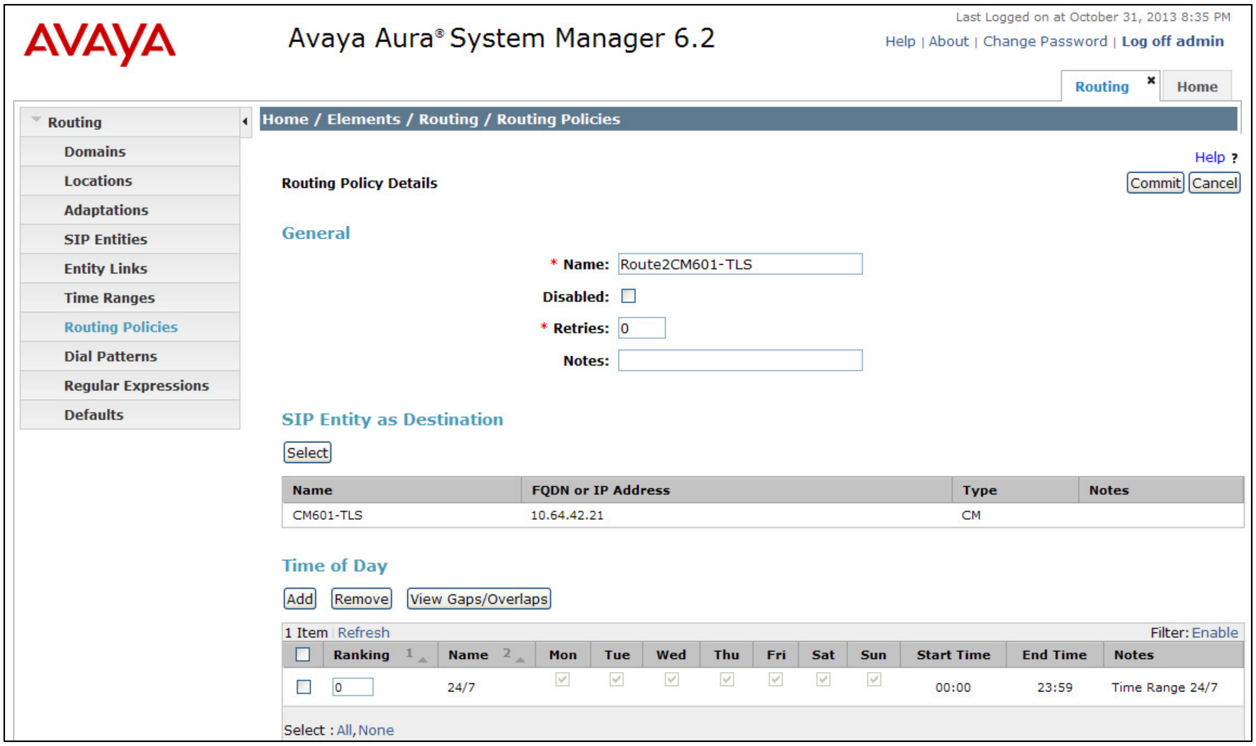
6.6.2. Communication Manager Routing Policy

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.



Routing Policy Details

General

* Name: Route2CM601-TLS

Disabled: □

* Retries: 0

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
CM601-TLS	10.64.42.21	CM	

Time of Day

□	Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
□	0	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	Time Range 24/7

6.7. Administer Dial Patterns

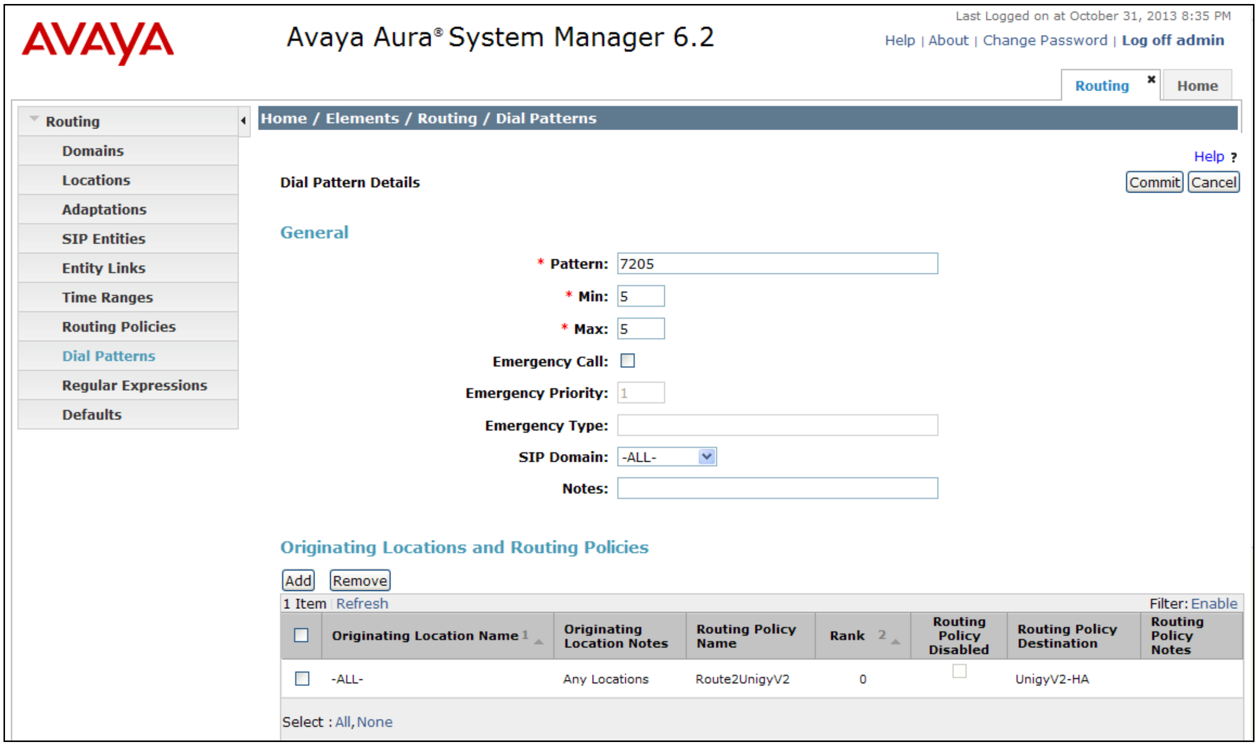
Add a new dial pattern for IPC, and update the existing dial pattern for Communication Manager.

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 (5).
- **Max:** The maximum number of digits to be matched (5).
-
- **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 selected as shown below.



Dial Pattern Details

General

* Pattern:	7205
* Min:	5
* Max:	5
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

Originating Locations and Routing Policies

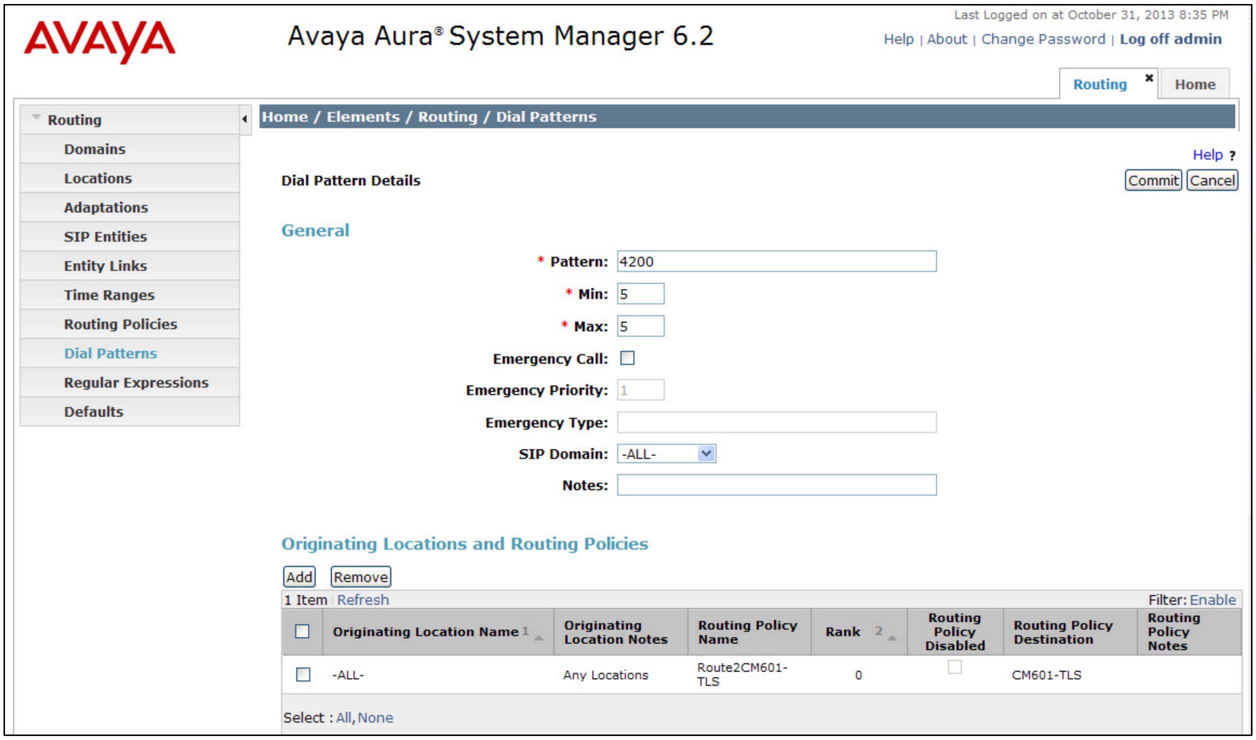
1 Item Refresh

□	Originating Location Name 1	Originating Location Notes	Routing Policy Name	Rank 2	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	-ALL-	Any Locations	Route2UnigyV2	0	□	UnigyV2-HA	

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 “4200 (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.



Dial Pattern Details

General

* Pattern:	4200
* Min:	5
* Max:	5
Emergency Call:	□
Emergency Priority:	1
Emergency Type:	
SIP Domain:	-ALL-
Notes:	

Originating Locations and Routing Policies

□	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
□	-ALL-	Any Locations	Route2CM601-TLS	0	□	CM601-TLS	

7. Configure IPC Converged Communication Manager

This section provides the procedures for configuring IPC Converged Communication 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

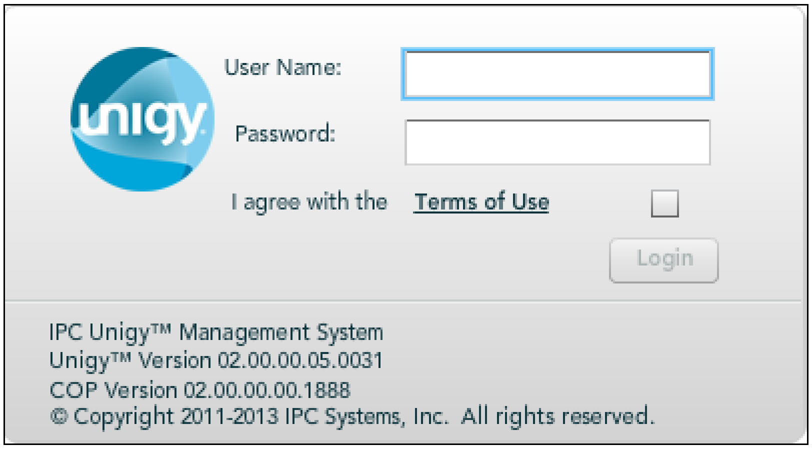
The configuration of Media Manager and/or Converged Communication Manager is typically performed by IPC installation technicians. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Launch Unigy Management System

Access the UnigyV2 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**.

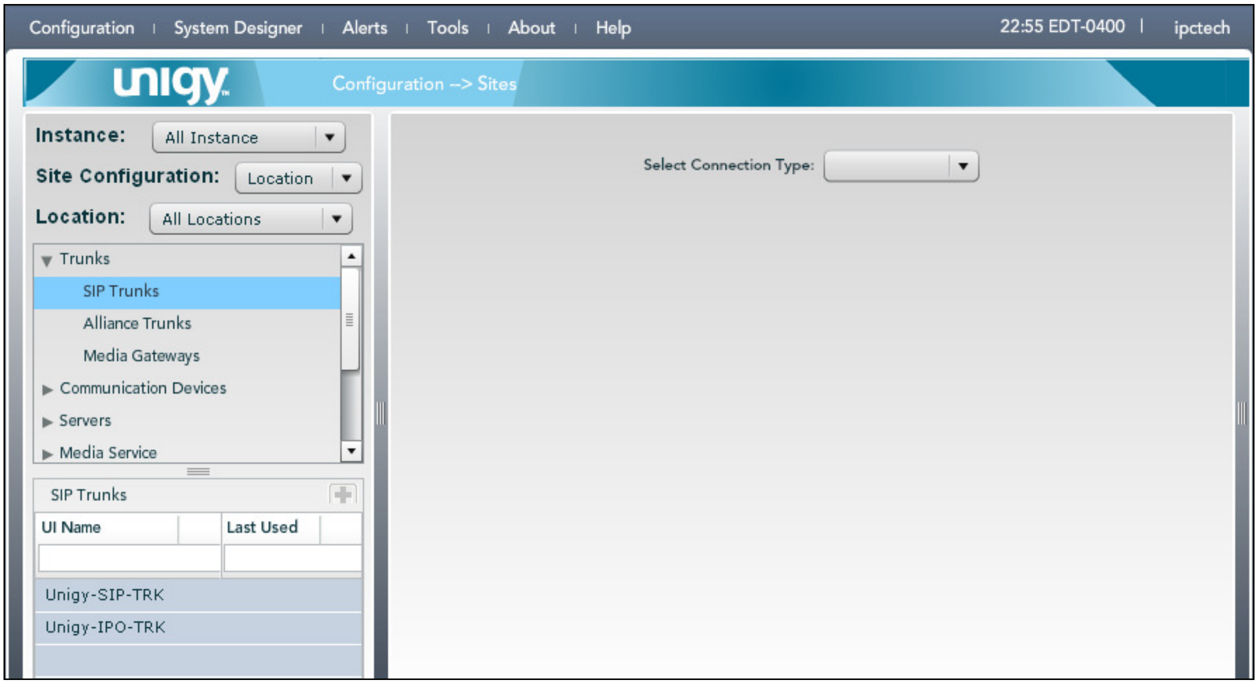


IPC Unigy™ Management System
Unigy™ Version 02.00.00.05.0031
COP Version 02.00.00.00.1888
© Copyright 2011-2013 IPC Systems, Inc. All rights reserved.

7.2. Administer SIP Trunks

Navigate to **Configuration → Site** (not shown) from the main menu at the top.

Select **Trunks → SIP Trunks** in the left pane, and click the **Add** icon (+) in the lower left pane to add a new SIP trunk. Select “Dial Tone” from the **Select Connection Type** drop-down list.



Instance: All Instance

Site Configuration: Location

Location: All Locations

- Trunks
  - SIP Trunks
  - Alliance Trunks
  - Media Gateways
- Communication Devices
- Servers
- Media Service

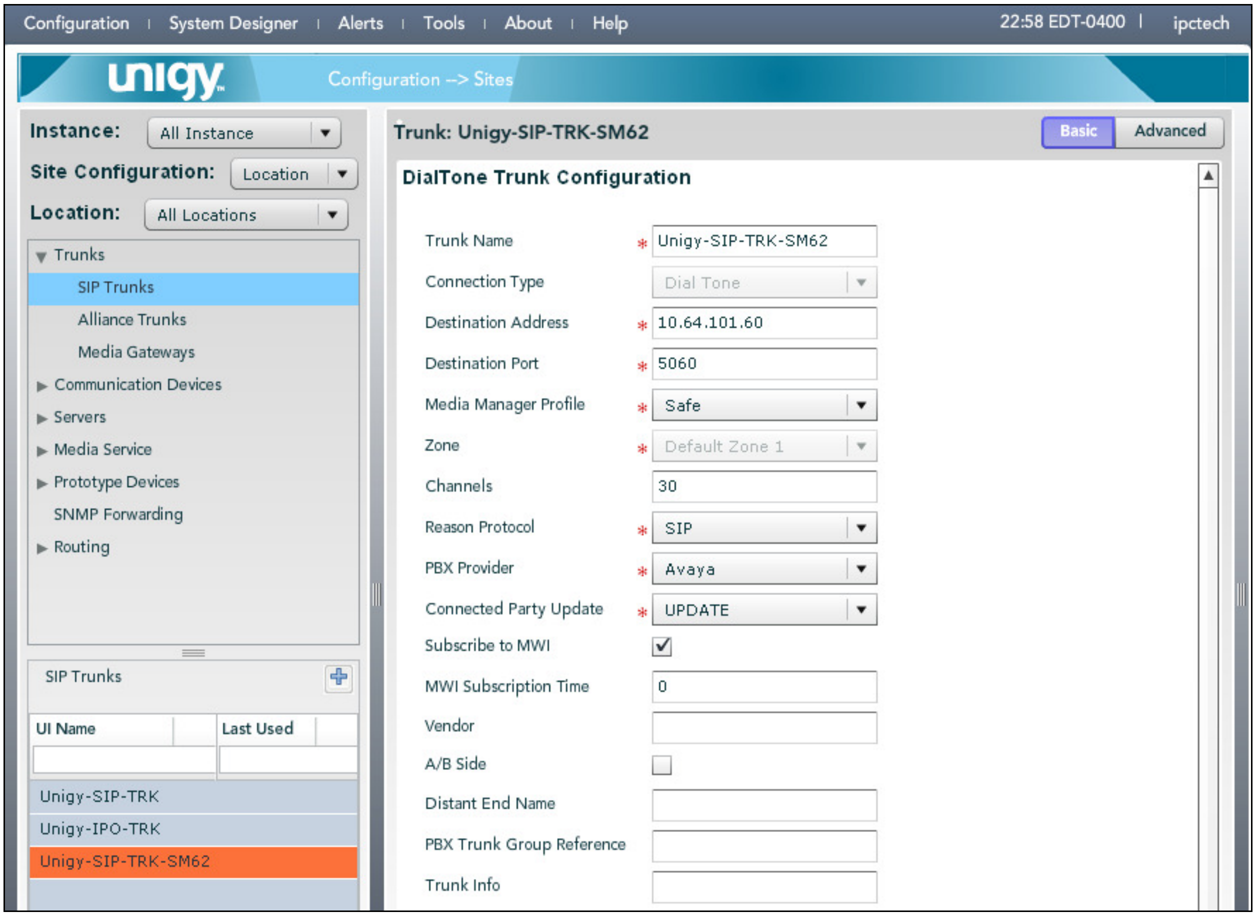
SIP Trunks

UI Name	Last Used
Unigy-SIP-TRK	
Unigy-IPO-TRK	

Select Connection Type:

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 0**.
- **Reason Protocol** “SIP”
- **PBX Provider:** “Avaya”
- **Connected Party Update:** “UPDATE”
- **Subscribe to MWI** Check the check box



Trunk: Unigy-SIP-TRK-SM62

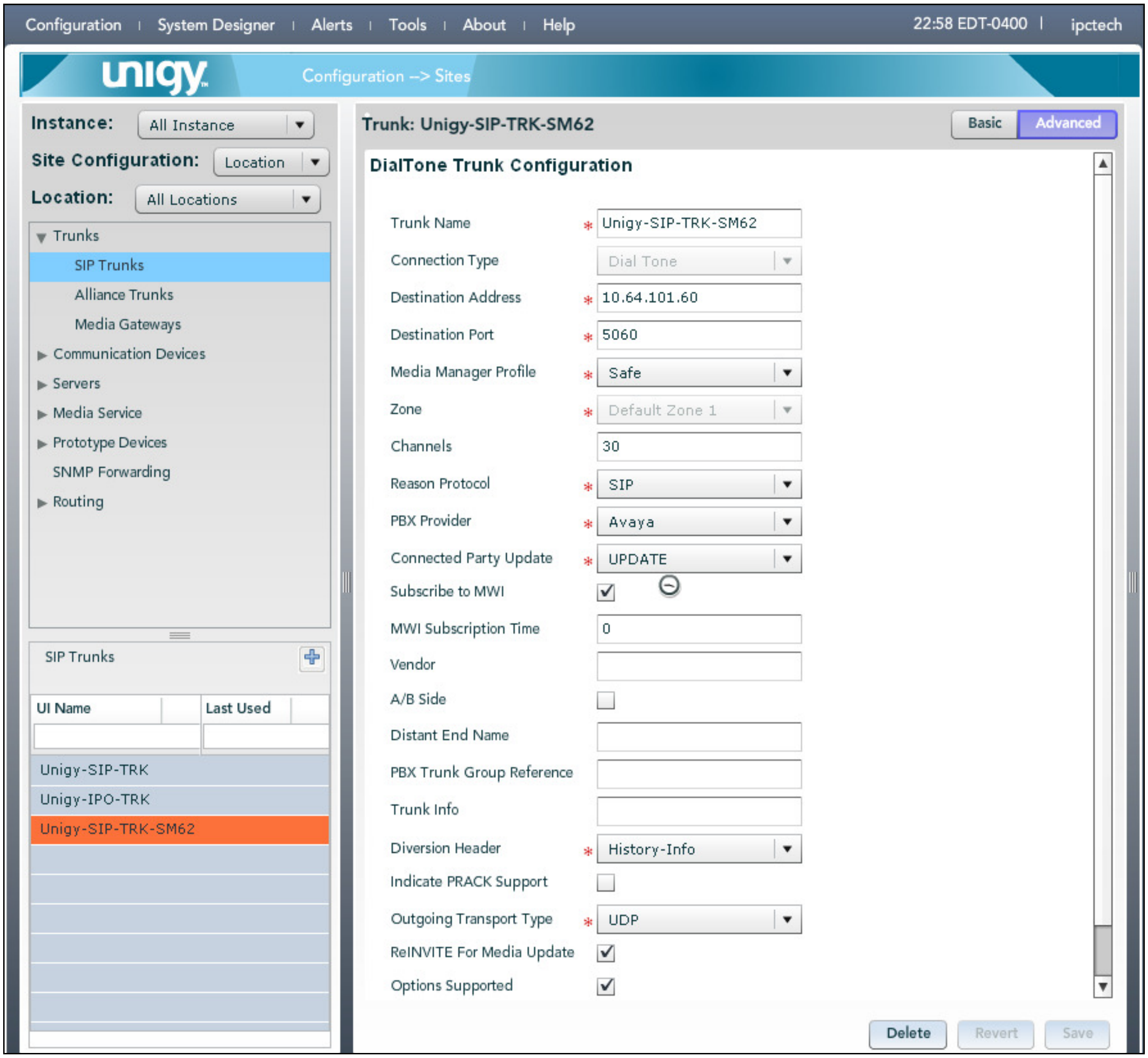
DialTone Trunk Configuration

Trunk Name	* Unigy-SIP-TRK-SM62
Connection Type	Dial Tone
Destination Address	* 10.64.101.60
Destination Port	* 5060
Media Manager Profile	* Safe
Zone	* Default Zone 1
Channels	30
Reason Protocol	* SIP
PBX Provider	* Avaya
Connected Party Update	* UPDATE
Subscribe to MWI	☑
MWI Subscription Time	0
Vendor	
A/B Side	□
Distant End Name	
PBX Trunk Group Reference	
Trunk Info	

Select the **Advanced** tab on the upper right, and enter the following values for the specified fields:

- **Diversion Header:** "History Info"
- **Outgoing Transport Type:** "UDP"

Click **Save**.



Trunk: Unigy-SIP-TRK-SM62

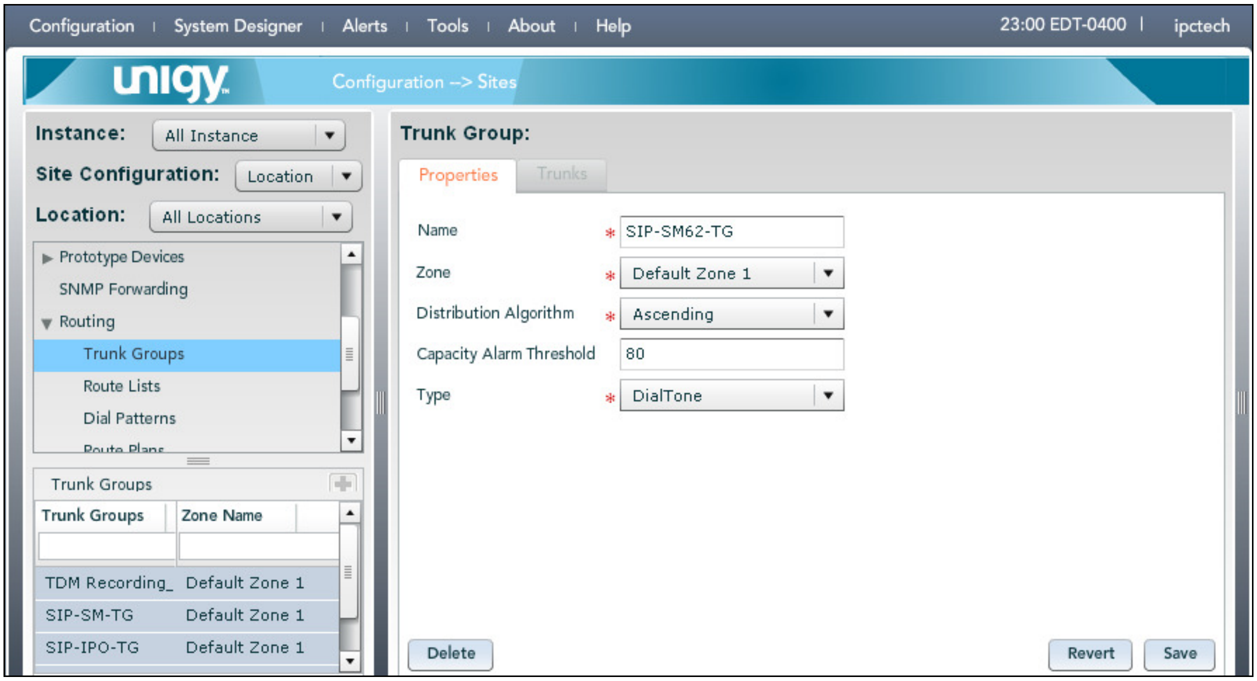
DialTone Trunk Configuration

Trunk Name *	Unigy-SIP-TRK-SM62
Connection Type	Dial Tone
Destination Address *	10.64.101.60
Destination Port *	5060
Media Manager Profile *	Safe
Zone *	Default Zone 1
Channels	30
Reason Protocol *	SIP
PBX Provider *	Avaya
Connected Party Update *	UPDATE
Subscribe to MWI	☑
MWI Subscription Time	0
Vendor	
A/B Side	☐
Distant End Name	
PBX Trunk Group Reference	
Trunk Info	
Diversion Header *	History-Info
Indicate PRACK Support	☐
Outgoing Transport Type *	UDP
ReINVITE For Media Update	☑
Options Supported	☑

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**, select “Default Zone 1” for the **Zone** field, select “Ascending” for the **Distribution Algorithm** field and click **Save**. Select the **Trunks** tab in the right pane.



Instance: All Instance

Site Configuration: Location

Location: All Locations

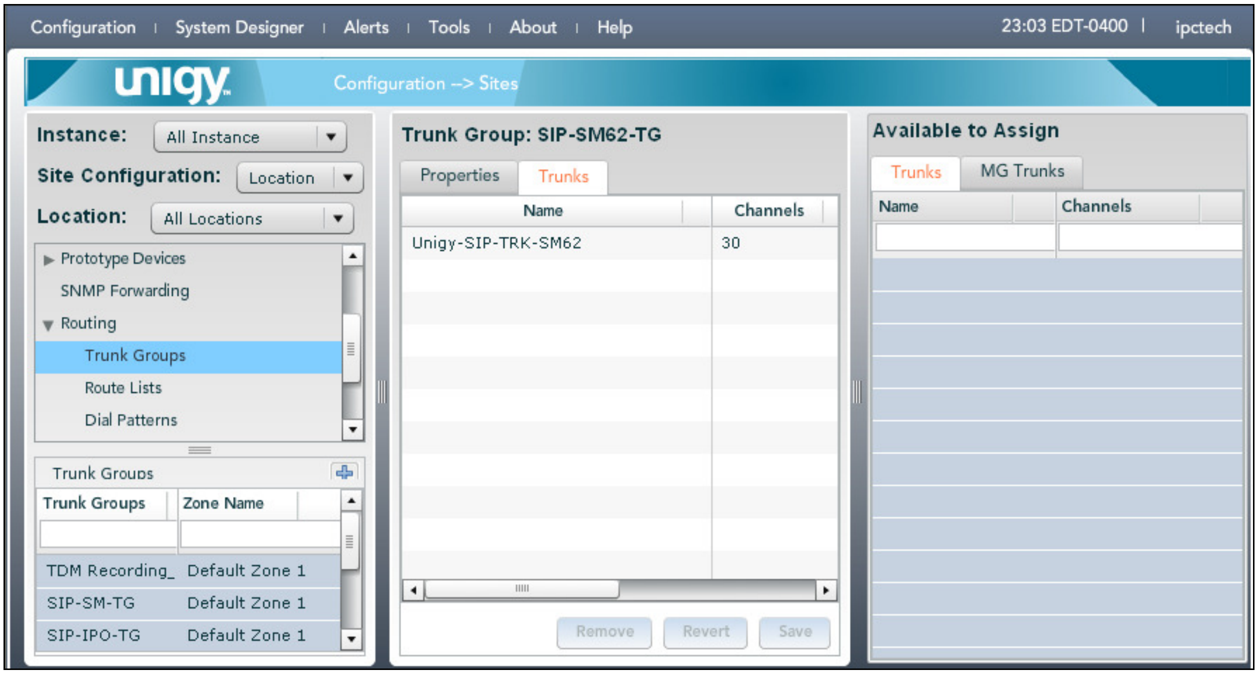
Trunk Groups

Trunk Groups	Zone Name
TDM Recording_	Default Zone 1
SIP-SM-TG	Default Zone 1
SIP-IPO-TG	Default Zone 1

Trunk Group:

Name *	SIP-SM62-TG
Zone *	Default Zone 1
Distribution Algorithm *	Ascending
Capacity Alarm Threshold	80
Type *	DialTone

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**.



Instance: All Instance

Site Configuration: Location

Location: All Locations

Trunk Groups

Trunk Groups	Zone Name
TDM Recording_	Default Zone 1
SIP-SM-TG	Default Zone 1
SIP-IPO-TG	Default Zone 1

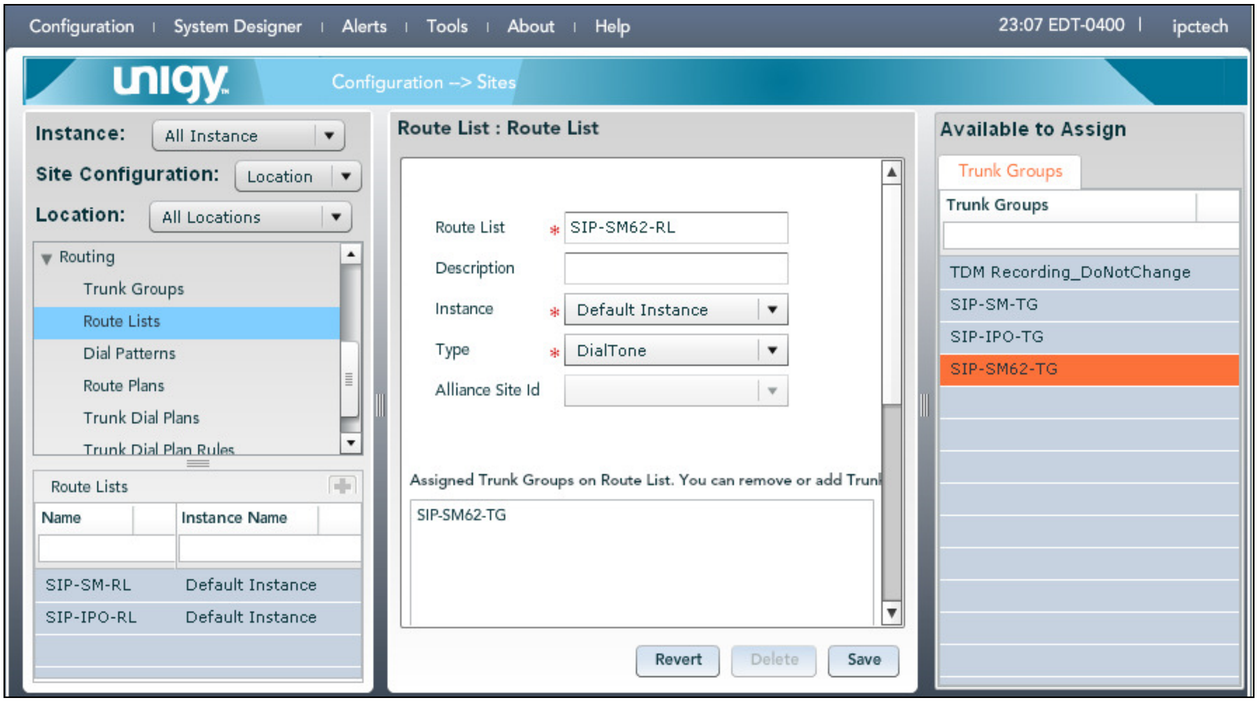
Trunk Group: SIP-SM62-TG

Name	Channels
Unigy-SIP-TRK-SM62	30

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**.



Instance: All Instance

Site Configuration: Location

Location: All Locations

Route Lists

Name	Instance Name
SIP-SM-RL	Default Instance
SIP-IPO-RL	Default Instance

Route List * SIP-SM62-RL

Description

Instance * Default Instance

Type * DialTone

Alliance Site Id

Assigned Trunk Groups on Route List. You can remove or add Trunl

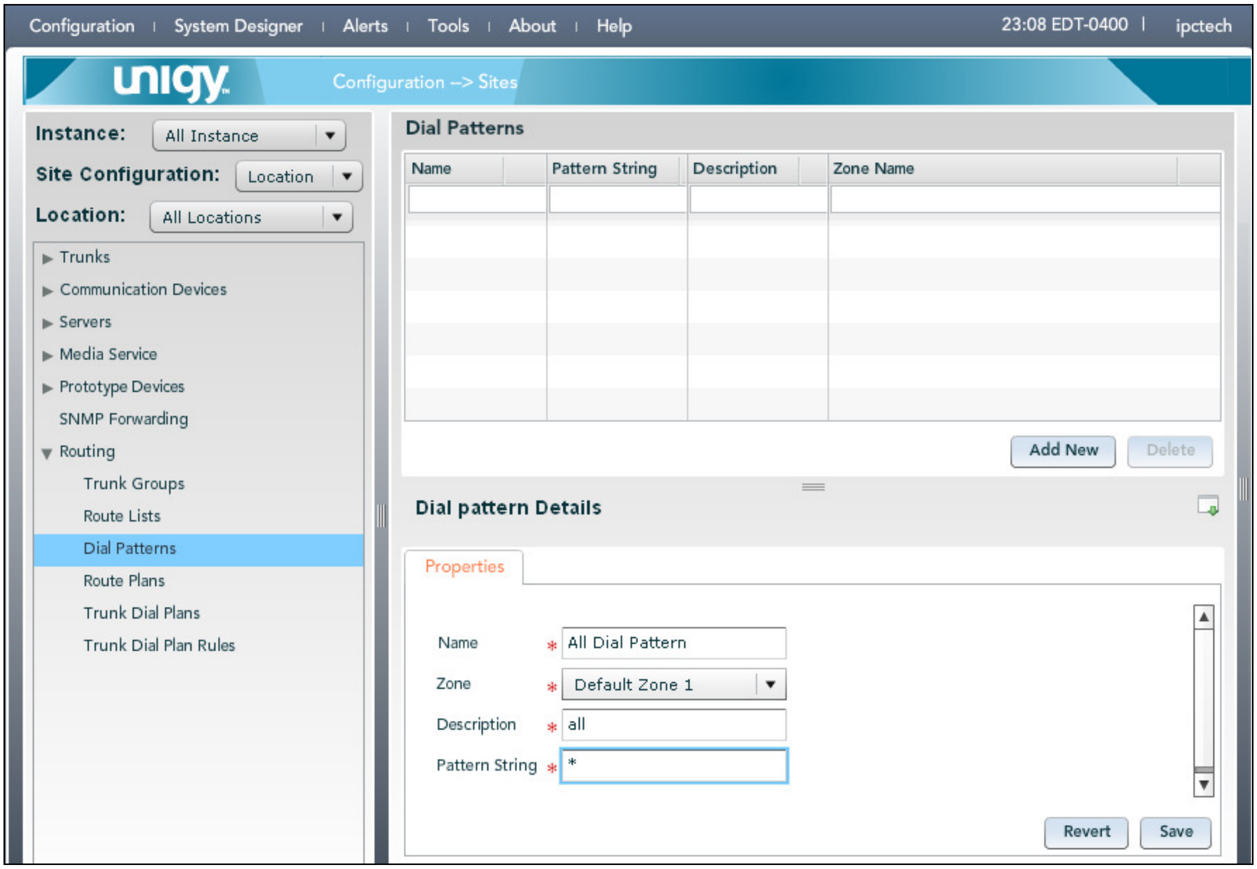
SIP-SM62-TG

Trunk Groups
TDM Recording_DoNotChange
SIP-SM-TG
SIP-IPO-TG
SIP-SM62-TG

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 "*" meaning any digits will be sent to Session Manager. Click **Save**.



Instance: All Instance

Site Configuration: Location

Location: All Locations

Name * All Dial Pattern

Zone * Default Zone 1

Description * all

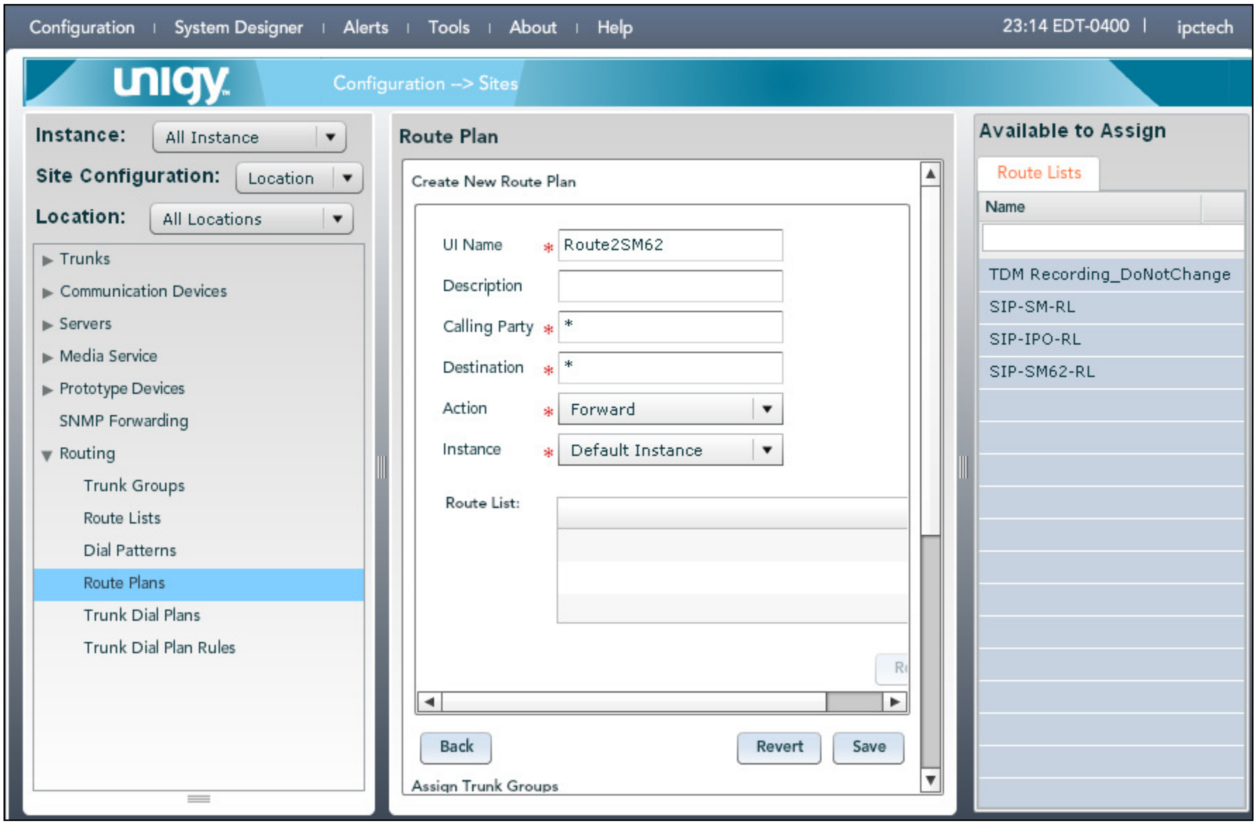
Pattern String * *

Repeat this section to add another dial pattern to reach the PSTN, and include any required prefix by Avaya Aura® Communication Manager.

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 Create New **Route Plan** middle pane, enter a descriptive **UI Name** and optional **Description**. For **Calling Party**, enter "*" to denote any calling party from UnigyV2. For **Destination** select the dial pattern for Avaya endpoints from **Section 7.5**. Select "Forward" for **Action** and click **Save**.

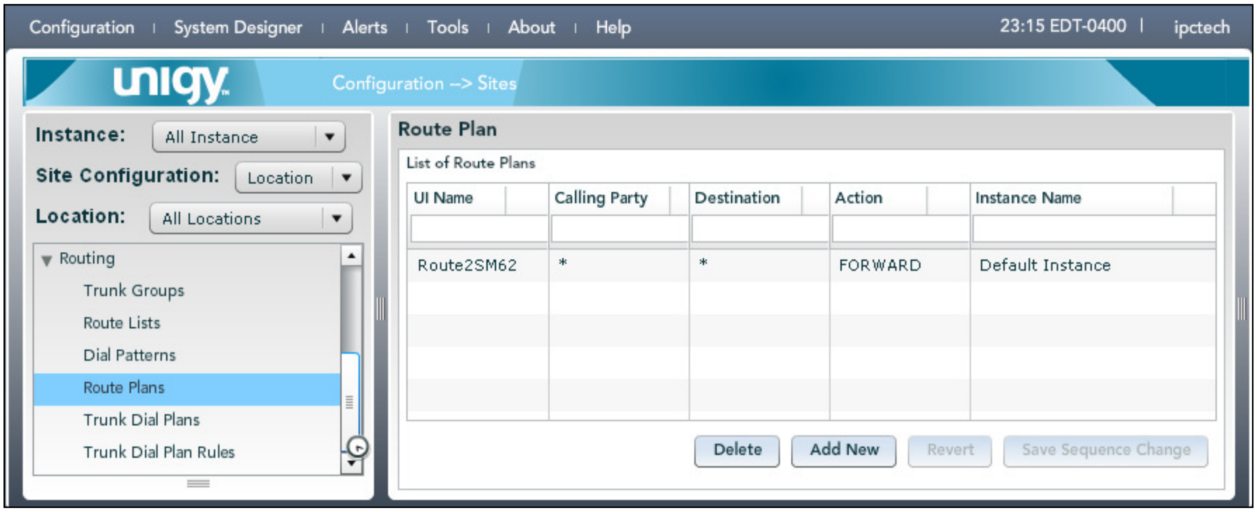


UI Name * Route2SM62
Description
Calling Party * *
Destination * *
Action * Forward
Instance * Default Instance
Route List:

Available to Assign — Route Lists:

TDM Recording_DoNotChange
SIP-SM-RL
SIP-IPO-RL
SIP-SM62-RL

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).



Instance: All Instance

Site Configuration: Location

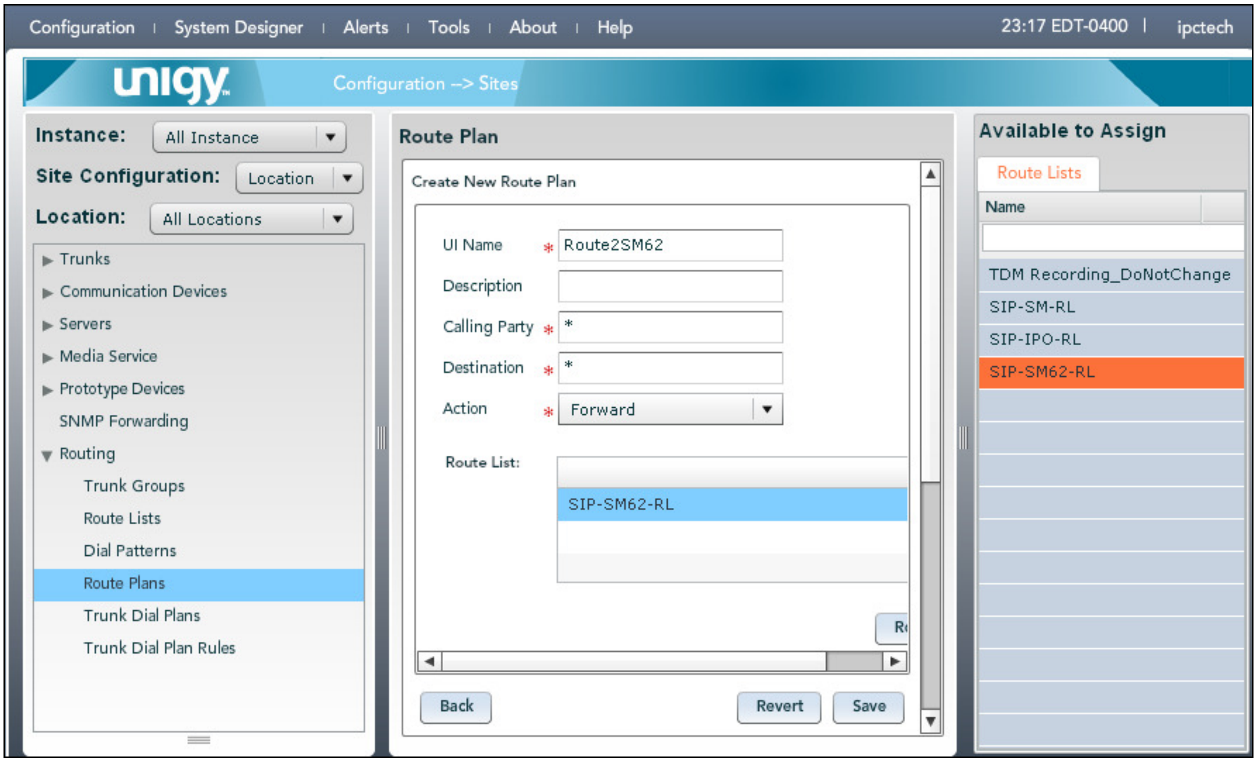
Location: All Locations

Route Plan

List of Route Plans

UI Name	Calling Party	Destination	Action	Instance Name
Route2SM62	*	*	FORWARD	Default Instance

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**.



UI Name * Route2SM62
Description
Calling Party * *
Destination * *
Action * Forward
Route List: SIP-SM62-RL

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 UnigyV2.

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 0**. Verify that all trunks are in the "in-service/idle" state as shown below.

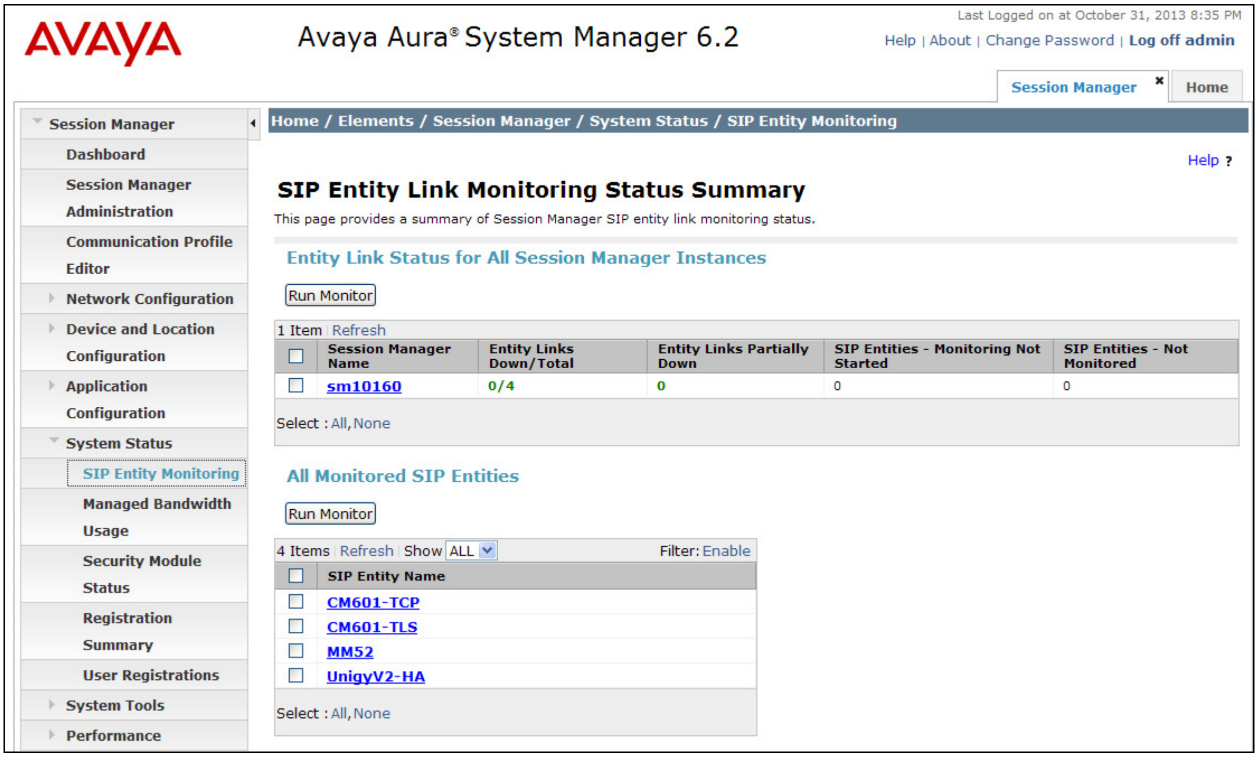
```
status trunk 60

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0060/001 T00064   in-service/idle    no
0060/002 T00065   in-service/idle    no
0060/003 T00066   in-service/idle    no
0060/004 T00067   in-service/idle    no
0060/005 T00068   in-service/idle    no
0060/006 T00069   in-service/idle    no
0060/007 T00070   in-service/idle    no
0060/008 T00071   in-service/idle    no
0060/009 T00072   in-service/idle    no
0060/010 T00073   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 0**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

```
status signaling-group 60
                        STATUS SIGNALING GROUP

        Group ID: 60
      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**.



Home / Elements / Session Manager / System Status / SIP Entity Monitoring

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

Entity Link Status for All Session Manager Instances

1 Item | Refresh

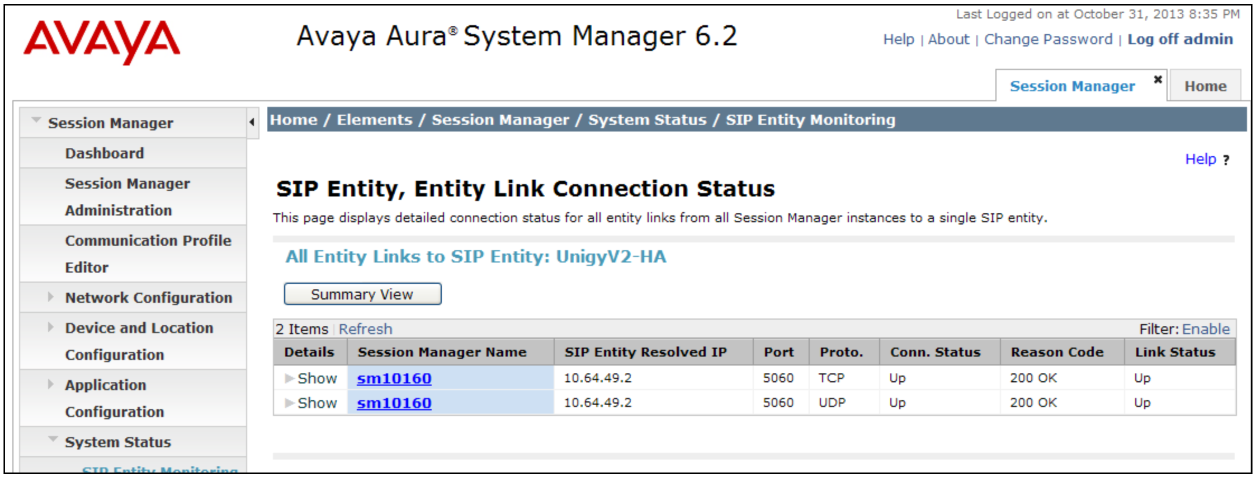
□	Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
□	sm10160	0/4	0	0	0

All Monitored SIP Entities

4 Items | Refresh | Show ALL | Filter: Enable

□	SIP Entity Name
□	CM601-TCP
□	CM601-TLS
□	MM52
□	UnigyV2-HA

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


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at October 31, 2013 8:35 PM
Help | About | Change Password | Log off admin
Session Manager
Home
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
SIP Entity Monitoring
Home / Elements / Session Manager / System Status / SIP Entity Monitoring
Help ?
Summary View
2 Items | Refresh
Filter: Enable

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: UnigyV2-HA

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▸Show	sm10160	10.64.49.2	5060	TCP	Up	200 OK	Up
▸Show	sm10160	10.64.49.2	5060	UDP	Up	200 OK	Up

8.3. Verify IPC UnigyV2

Make a call from/to an IPC turret user to/from 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 UnigyV2 to successfully interoperate with Avaya Aura® Communication Manager 6.01 using Avaya Aura® Session Manager 6.2. 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, Release 6.2, July2012, available at http://support.avaya.com.

3. *UnigyV2 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.