

Avaya Solution & Interoperability Test Lab

Application Notes for IPC Unigy v5.2 sp2 with Avaya Aura® Session Manager R10.1 and Avaya Aura® Communication Manager R10.1 using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for IPC Unigy v5.2 sp2 to interoperate with Avaya Aura® Session Manager R10.1 and Avaya Aura® Communication Manager R10.1 using SIP trunks.

IPC Unigy is a trading communication solution. In the compliance testing, IPC Unigy uses SIP trunks to Avaya Aura® Session Manager. Using the SIP trunks, IPC Unigy users with IPC MAX/TOUCH endpoints (turrets) were able to reach users on Avaya Aura® Communication Manager and on the PSTN.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1 Introduction

These Application Notes describe the configuration steps required for IPC Unigy v5.2 sp2 (Unigy) to interoperate with Avaya Aura® Session Manager R10.1 (Session Manager) and Avaya Aura® Communication Manager R10.1 (Communication Manager). Unigy integrates with Session Manager via SIP Trunks (UDP). IPC MAX/TOUCH endpoints (turrets) provide calling functionality.

The Unigy Platform is a unified trading communications system designed specifically to make the entire trading ecosystem more productive, intelligent and efficient. Based on a 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. Calls were placed from various users to verify the call scenarios.

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

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and Unigy did not include use of any specific encryption features as requested by IPC.

2.1 Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included basic call, display, G.711MU, G.729, hold/reconnect, DTMF, call forwarding unconditional/ring-no-answer/busy, blind/attended transfer, and conference. Messaging interoperability is not verified except for sending DTMF tones to the server.

The serviceability testing focused on verifying the ability of Unigy to recover from adverse conditions, simulated by disabling/reenabling the entity links to Unigy.

2.2 Test Results

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

- During the compliance test media shuffling was disabled, as shown in **Section 5.2** and **Section 5.4**. (IPC requested)
- DTMF tones sent to turrets were not heard on their handset. Tones sent from turrets were heard on Avaya handsets and messaging.
- The G.729 codec was not supported unless a license was enabled on Unigy.
- Call forwarding on turrets to Avaya endpoints, other IPC turrets, and PSTN are configured using Unigy features.

2.3 Support

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

• **Phone:** +1-(800)-NEED-IPC, +1-(203) 339-7800

• Email: systems.support@ipc.com

3 Reference Configuration

As shown in the test configuration below, Unigy consists of the Media Manager (MM), Converged Communication Manager (CCM), and turret endpoints. The Media Manager and Converged Communication Manager are typically deployed on separate servers.

SIP trunks are used from Unigy to Session Manager to reach Avaya users (SIP and H.323) and the simulated PSTN via Avaya Session Border Controller for Enterprise.

A five-digit dial plan was used to facilitate dialing between Communication Manager and Unigy. Unique extension ranges were associated with Communication Manager users (70xxx for H.323 and SIP), and IPC turret users (7205x).

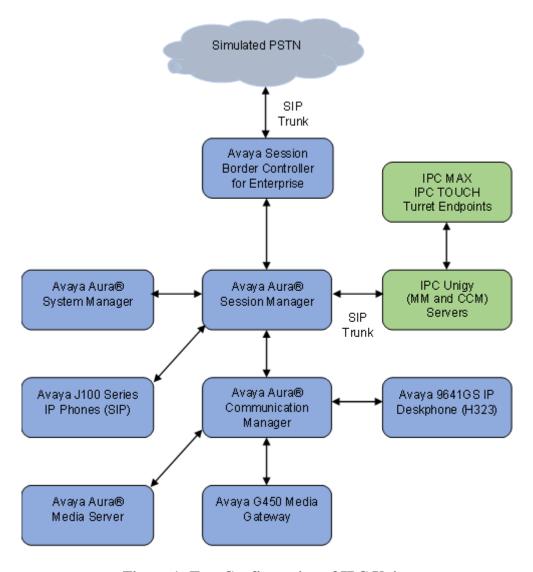


Figure 1: Test Configuration of IPC Unigy

4 Equipment and Software Validated

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

Equipment/Software	Release/Version				
Avaya Aura® Communication Manager	10.1.0.2-SP2				
running on Virtualized Environment	01.0.974.0-27607				
Avaya Aura® System Manager running on	10.1.0.2 Service Pack 2				
Virtualized Environment	10.1.0.2.0715160				
Avaya Aura® Session Manager running on	10.1.0.2 Service Pack 2				
Virtualized Environment	10.1.0.02.1010215				
Avaya Session Border Controller for	10.1.1.0-35-218720				
Enterprise running on Virtualized Environment	10.1.1.0-33-210720				
Avaya Aura® Media Server running on	10.1.0.101				
Virtualized Environment					
Avaya G450 Media Gateway	42.7.0				
Avaya 9641GS IP Deskphone (H.323)	6.8.5.3.2				
Avaya J179/J189 IP Phones (SIP)	4.0.13.0.6				
IPC Unigy					
Media Manager	05.02.02.00.0033				
Converged Communication Manager	05.02.02.00.0033 HF3: 05.02.02.00.9000303				
IPC TOUCH Turret Endpoints	05.02.02.00.0033				
IPC MAX Turret Endpoints	05.02.02.00.0033				

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 SIP signaling group
- Administer SIP trunk group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer AAR analysis
- Administer Trunk Group Calling Number
- Administer Tandem Calling Party Number

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 12
                               OPTIONAL FEATURES
IP PORT CAPACITIES
                                                         USED
               Maximum Administered H.323 Trunks: 12000
      Maximum Concurrently Registered IP Stations: 2400
        Maximum Administered Remote Office Trunks: 12000
Max Concurrently Registered Remote Office Stations: 2400
         Maximum Concurrently Registered IP eCons: 128
    Max Concur Reg Unauthenticated H.323 Stations: 100
                                                             0
                  Maximum Video Capable Stations: 36000
                                                             1
              Maximum Video Capable IP Softphones: 150
                                                             21
                  Maximum Administered SIP Trunks: 12000
  Max Administered Ad-hoc Video Conferencing Ports: 12000
  Max Number of DS1 Boards with Echo Cancellation: 688
```

5.2 Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "1". 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: Set to "1".

• **Direct IP-IP Audio Connection:** "n" to turn off Media Shuffling.

```
add signaling-group 1
                                                              Page 1 of
                               SIGNALING GROUP
Group Number: 1
                             Group Type: sip
 IMS Enabled? n
                       Transport Method: tls
      Q-SIP? n
    IP Video? y
                        Priority Video? n
                                               Enforce SIPS URI for SRTP? n
 Peer Detection Enabled? y Peer Server: SM
                                                               Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
  Near-end Node Name: procr
                                           Far-end Node Name: sm10
Near-end Listen Port: 5061
                                         Far-end Listen Port: 5061
                                      Far-end Network Region: 1
Far-end Domain: avaya.com
                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate
                                            RFC 3389 Comfort Noise? n
        DTMF over IP: rtp-payload
                                            Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 65
                                                      IP Audio Hairpinning? n
        Enable Layer 3 Test? y
                                                Alternate Route Timer(sec): 6
```

5.3 Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "1". 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"

• **Signaling Group:** Number of signaling group configured in previous section.

• **Number of Members:** As required in the environment.

```
add trunk-group 1
                                                         Page
                                                                1 of
                              TRUNK GROUP
                                Group Type: sip
Group Number: 1
                                                        CDR Reports: y
                                                   TN: 1 TAC: 101
 Group Name: SM Trunk 1
                                       COR: 1
  Direction: two-way Outgoing Display? y
Dial Access? n
                                               Night Service:
Queue Length: 0
Service Type: tie
                                 Auth Code? n
                                           Member Assignment Method: auto
                                                    Signaling Group: 1
                                                  Number of Members: 10
```

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

```
add trunk-group 1
                                                                    3 of
                                                             Page
TRUNK FEATURES
         ACA Assignment? n
                                      Measured: both
                                                          Maintenance Tests? y
  Suppress # Outpulsing? n Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                               Modify Tandem Calling Number: no
              Send UCID? y
Show ANSWERED BY on Display? Y
DSN Term? n
```

Navigate to **Page 5** and disable **Network Call Redirection** (REFER) since REFER is not supported on Unigy.

```
add trunk-group 1
                                                             Page 5 of 5
                              PROTOCOL VARIATIONS
                                      Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                      Send Transferring Party Information? n
                                 Network Call Redirection? n
                                     Send Diversion Header? n
                                  Support Request History? y
                              Telephone Event Payload Type: 120
                        Convert 180 to 183 for Early Media? n
                 Always Use re-INVITE for Display Updates? y
                        Identity for Calling Party Display: P-Asserted-Identity
           Block Sending Calling Party Location in INVITE? n
                Accept Redirect to Blank User Destination? n
          Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                               Request URI Contents: may-have-extra-digits
```

5.4 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 5.2**.

For **Authoritative Domain**, set to the pertinent domain, in this case "avaya.com". 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 Unigy.

```
change ip-network-region 1
                                                                Page
                                                                       1 of 20
                               IP NETWORK REGION
Region: 1 NR Group: 1
Location: Authoritative Domain: avaya.com
   Name: Main
                               Stub Network Region: n
MEDIA PARAMETERS
                                Intra-region IP-IP Direct Audio: no
     Codec Set: 1
                              Inter-region IP-IP Direct Audio: no
  UDP Port Min: 2048
                                          IP Audio Hairpinning? y
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
       Audio PHB Value: 46
       Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5
                                      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
                                                         RSVP Enabled? n
 H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

5.5 Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.4**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Unigy supports G.711MU and G.729. For G.729, IPC needs to install a license.

```
change ip-codec-set 1

IP MEDIA PARAMETERS
Codec Set: 1

Audio Silence Frames Packet
Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 20
2: G.729 n 2 20
```

5.6 Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Unigy, in this case "1". 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 1
                                             Page
                                                  1 of
            Pattern Number: 1
                             Pattern Name: sm81
  SCCAN? n Secure SIP? n Used for SIP stations? n
  Grp FRL NPA Pfx Hop Toll No. Inserted
                                                  DCS/ IXC
  No Mrk Lmt List Del Digits
                                                  QSIG
                   Dgts
                                                  Intw
1: 1 0
                                                  n user
2:
                                                  n user
3:
                                                  n
                                                     user
4:
                                                  n
                                                     user
5:
                                                  n
                                                     user
6:
                                                     user
   0 1 2 M 4 W Request
                                  Dgts Format
1: yyyyn n
                                             unk-unk none
                    rest.
2: yyyyyn n
                    rest
                                                    none
```

5.7 Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Unigy. In the example shown below, all calls originating from a 5-digit extension beginning with 5 or 7 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

5.8 Administer AAR Analysis

Use the "change aar analysis 720" command and add an entry to specify how to route calls to 720xx. In the highlighted example shown below, calls with digits 720xx will be routed using route pattern "1" from **Section 5.6**.

change aar analysis 720					Page 1 of	2			
AAR DIGIT ANALYSIS TABLE									
	Location: all				Percent Full: 0				
Dialed	Total	Route	Call	Node	ANI				
String	Min Max	Pattern	Type	Num	Reqd				
720	5 5	1	aar		n				

5.9 Administer Trunk Group Calling Number

Use the "change trunk-group n" command, where "n" is the SIP trunk group from **Section 5.3**, in this case "1".

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

```
change trunk-group 1
TRUNK FEATURES

ACA Assignment? n

Measured: both

Maintenance Tests? y

Suppress # Outpulsing? n Numbering Format: private

UUI Treatment: shared

Maximum Size of UUI Contents: 128

Replace Restricted Numbers? n

Replace Unavailable Numbers? n

Modify Tandem Calling Number: tandem-cpn-form

Send UCID? y

Show ANSWERED BY on Display? Y

DSN Term? n
```

5.10 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 720 and routed to trunk group 1 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	67			
CALLING PARTY NUMBER CONVERSION								
FOR TANDEM CALLS								
		Incoming	Outgoing			Ou	tgoing	
	CPN	Number	Trunk			Nu	mber	
Len	Prefix	Format	Group(s)	Delete	Insert	Fo	rmat	
5	720		1	30	Эхххуууу	pub	-unk	

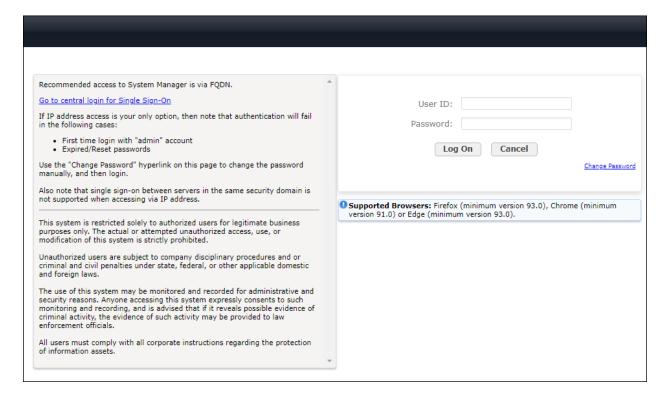
6 Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. It is assumed that the basic configuration is already in place. This section discusses the following areas:

- Launch System Manager
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

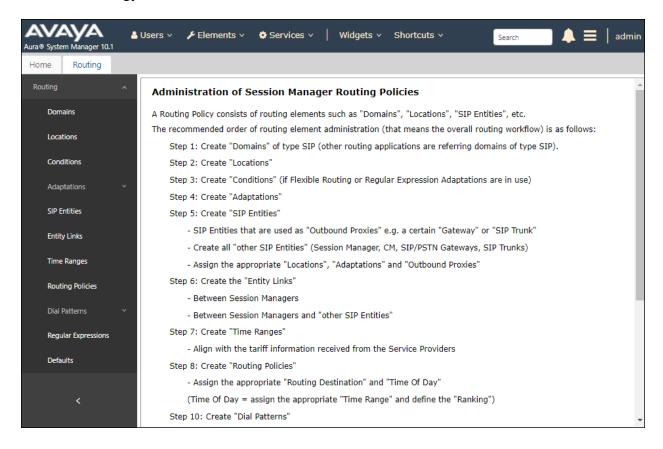
6.1 Launch System Manager

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

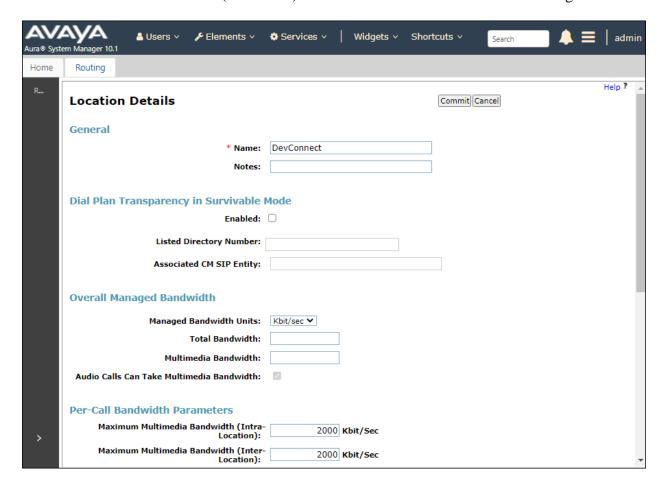


6.2 Administer Locations

In the subsequent screen (not shown), select **Elements** → **Routing** to display the **Administration of Session Manager Routing Policies** screen below. Select **Routing** → **Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a location for Unigy.



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 of "10.64.*" (not shown). Retain the default values in the remaining fields.



6.3 Administer Adaptations

Add an adaptation to translate incoming/outgoing SIP headers. Select **Adaptations** \rightarrow **Adaptations** from the left pane and click **New** (not shown) to add a new adaptation for Unigy.

The Adaptation Details screen is displayed. Enter the following values for the specified fields:

• Adaptation Name: A descriptive name.

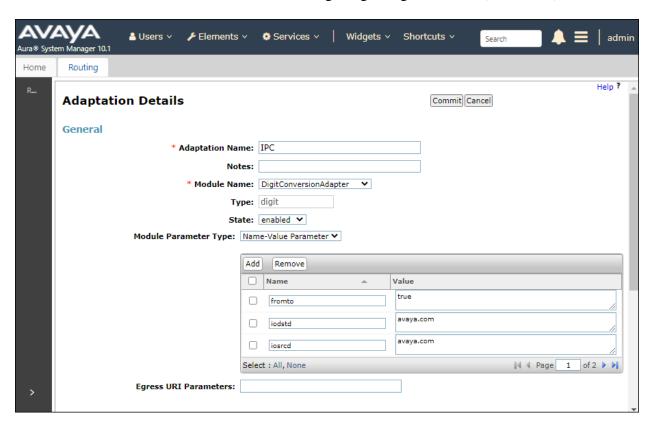
Module Name: "DigitConversionAdapter"
 Module Parameter Type: "Name-Value Parameter"

Click **Add** to add the adaptation name value pairs as specified:

• fromto: "true"

iodstd: The pertinent domain name.
iosrcd: The pertinent domain name.
odstd: "ipc.com" (not shown).

• **osrcd:** The Session Manager signaling IP address (not shown).



6.4 Administer SIP Entities

Add two new SIP entities, one for Unigy, and another for the SIP trunk for Communication Manager.

6.4.1 Unigy SIP Entity

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

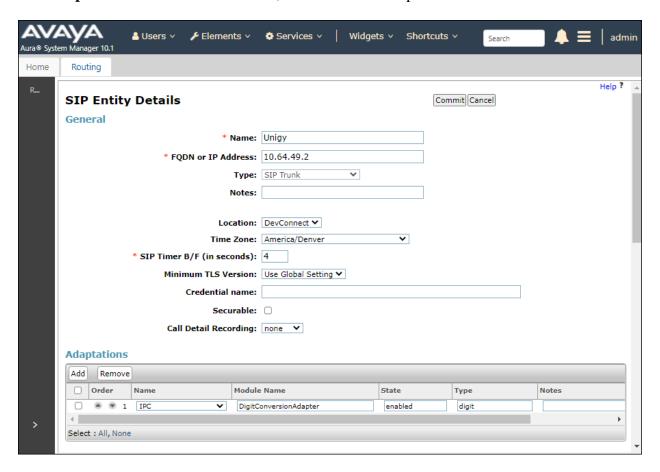
• FQDN or IP Address: The IP address of the Unigy Media Manager server.

• Type: "SIP Trunk"

• **Location:** Select the Unigy location name from **Section 6.2**.

• **Time Zone:** Select the applicable time zone.

• Adaptation: Click Add, then select the Adaptation Name from Section 6.3.



6.4.2 Communication Manager SIP Entity

Select **Routing** \rightarrow **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 Unigy.

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.

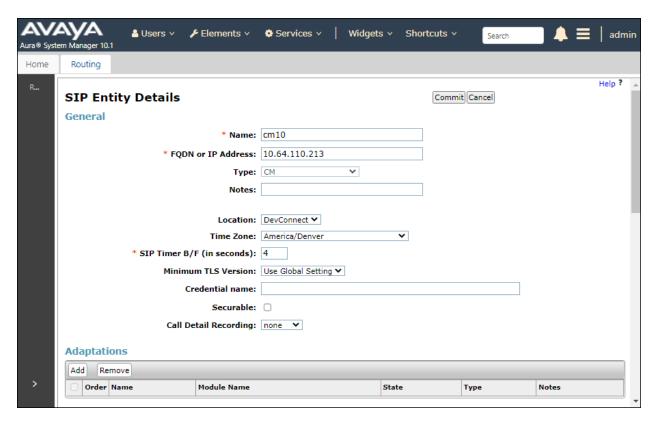
• FQDN or IP Address: The IP address of Communication Manager.

• **Type:** "CM"

• **Notes:** Any descriptive notes.

• **Location:** Select the location administered in **Section 6.2.**

• **Time Zone:** Select the applicable time zone.



6.5 Administer Entity Links

Add entity links for Unigy and for Communication Manager.

6.5.1 Unigy Entity Link

Select **Routing** \rightarrow **Entity Links** from the left pane and click **New** in the subsequent screen (not shown) to add a new entity link for Unigy.

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.

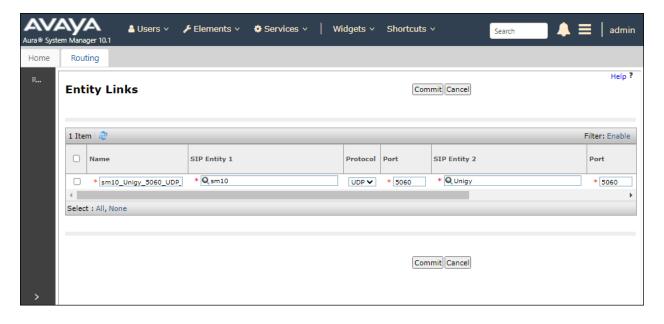
• **SIP Entity 1:** The administered Session Manager SIP entity name, e.g., "sm10".

Protocol: "UDP" Port: "5060"

• **SIP Entity 2:** The administered Unigy SIP entity name from **Section 6.4.1**.

• **Port:** "5060"

• Connection Policy: "trusted" (not shown).



6.5.2 Communication Manager Entity Link

Select **Routing** \rightarrow **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 administered Session Manager SIP entity name, e.g., "sm10".

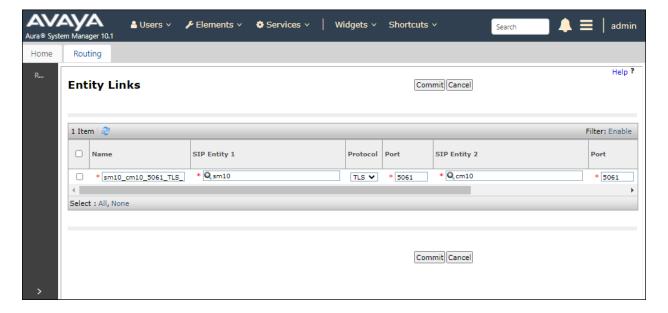
Protocol: "TLS" Port: "5061"

• **SIP Entity 2:** The administered Communication Manager SIP entity name

from **Section 6.4.2**.

• **Port:** "5061"

• Connection Policy: "trusted" (not shown)



6.6 Administer Routing Policies

Add two new routing policies, one for Unigy, and another for Communication Manager. The routing policies are linked to matching digits in dial plans defined in **Section 6.7** below. Then digits matching that dial plan entry are routed to the proper destination.

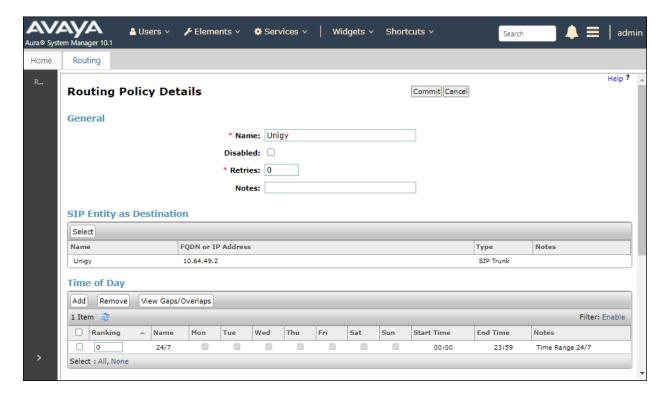
6.6.1 Unigy Routing Policy

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

The **Routing Policy 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

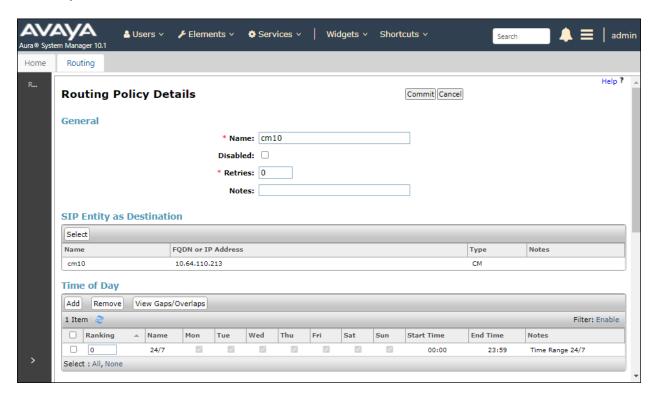
• **SIP Entity as Destination:** Click **Select** and choose the Unigy SIP entity administered in **Section 6.4.1**.



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. Enter the following values for the specified fields and retain the default values for the remaining fields.

- Name: A descriptive name
- **SIP Entity as Destination:** Click **Select** and choose the Communication Manager entity administered in **Section 6.4.2**.



6.7 Administer Dial Patterns

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

6.7.1 Unigy Dial Pattern

Select **Routing** \rightarrow **Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Unigy 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.

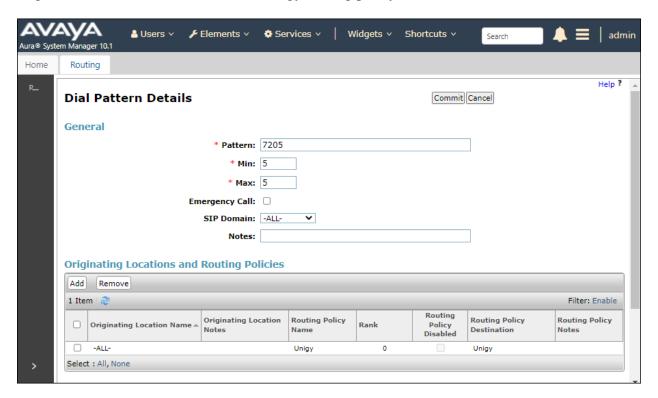
• **Pattern:** A dial pattern to match.

Min: The minimum number of digits to be matched.
Max: The maximum number of digits to be matched.

• SIP Domain: "ALL"

• **Notes:** Any desired description (optional).

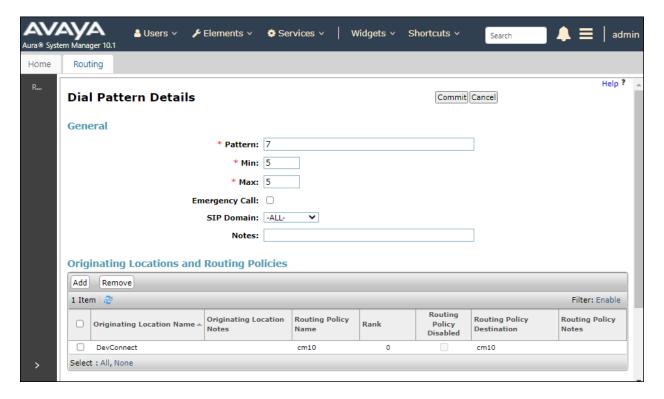
In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching Unigy turret users. In the compliance testing, the policy allowed for call origination from all locations, and the Unigy routing policy from **Section 6.6.1** was selected.



6.7.2 Communication Manager Dial Pattern

Select **Routing** \rightarrow **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 **7** (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 Unigy turret users. 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 Unigy Converged Communication Manager

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

- Launch Unigy Management System
- Administer SIP Trunks
- Administer Trunk groups
- Administer Route Lists
- Administer Zone Dial Patterns
- Administer Route Plans

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

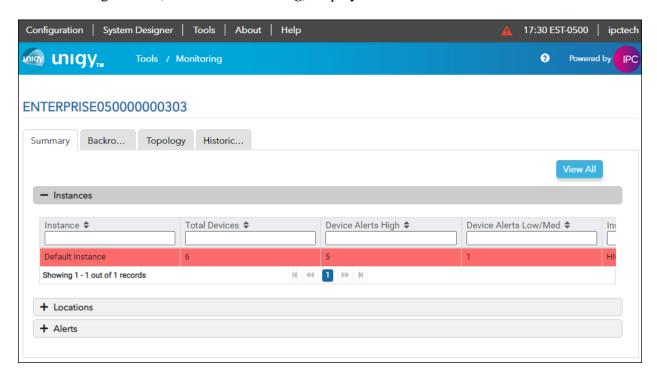
7.1 Launch Unigy Management System

Access the Unigy Management System web interface by using the URL "http://<ip-address>" in an Internet browser window, where "ip-address" is the virtual IP address (VIP) of the Zone or in a standalone environment the IP address of the CCM.

The screen below is displayed. Enter the appropriate credentials. Check **I agree with the Terms** of Use and click **Login** (not shown). In the subsequent screen (not shown), click **Continue**.

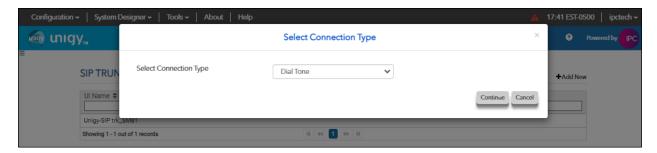


The following screen (**Tools** \rightarrow **Monitoring**) displays.



7.2 Administer SIP Trunks

Navigate to **Configuration** → **Sites** from the top Main Toolbar. Select **Trunks** → **SIP Trunks** (not shown) in the left pane and click the **Add New** icon (not shown) in the upper right pane to add a new SIP trunk. Select "Dial Tone" from the **Select Connection Type** drop-down list. Click **Continue**.



The screen below is displayed next. Select **Show Advanced** (not shown) on the top right and enter the following values for the specified fields.

• **Trunk Name:** A descriptive name.

• **Destination Address:** IP address of the Session Manager signaling interface.

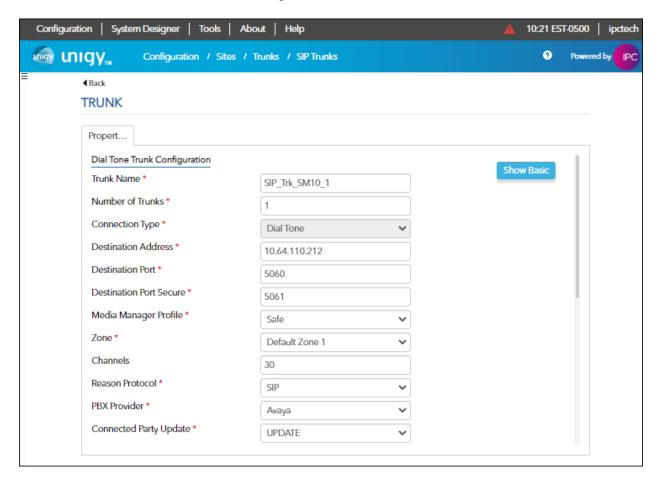
• **Destination Port:** The port number from **Section 6.5.1**.

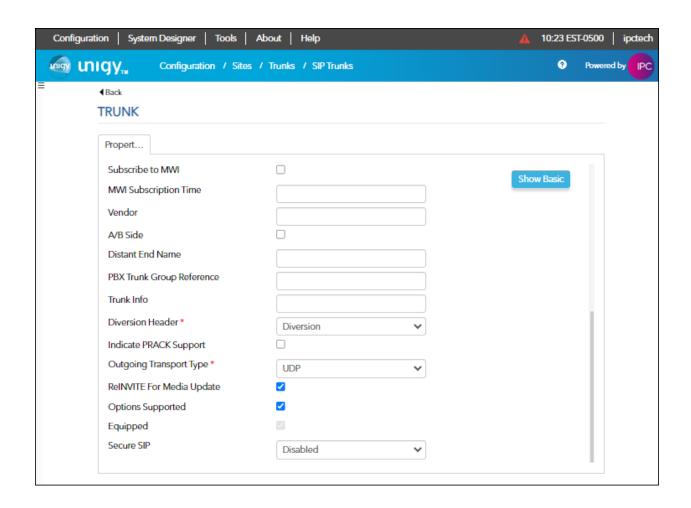
• **Zone:** An available zone, in this case "Default Zone 1".

• **Channels:** The number of SIP trunk group members.

Reason Protocol: "SIP"
PBX Provider: "Avaya"
Connected Party Update: "UPDATE"
Diversion Header "Diversion"
Outgoing Transport Type "UDP"

Retain the default values in the remaining fields.

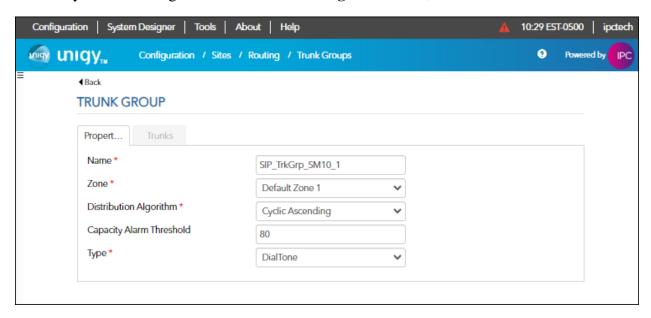




7.3 Administer Trunk Groups

Select **Routing** \rightarrow **Trunk Groups** in the left pane and click the **Add New** icon in the upper right pane to add a new trunk group.

In the **Properties** tab, enter a descriptive **Name**, select "Default Zone 1" for the **Zone** field, select "Cyclic Ascending" for the **Distribution Algorithm** field, and click **Save**.

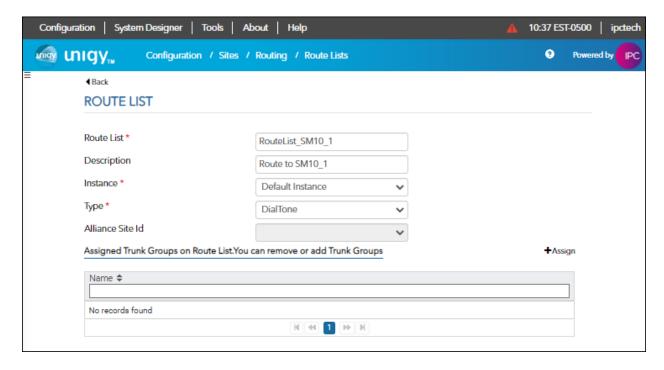


Select the **Trunks** tab. Click on the **+Assign** icon on the upper right to display available trunks. Select the SIP trunk from **Section 7.2** (not shown). Click **Save**.

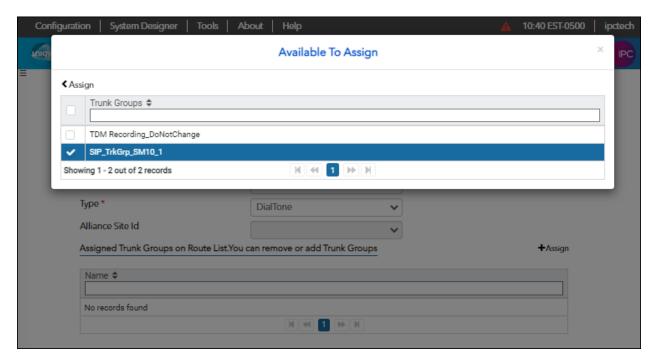
7.4 Administer Route Lists

Select **Routing** → **Route Lists** in the left pane and click the +**Add New** icon in the upper right to add a new route list.

The **ROUTE LIST** screen is displayed. For **Route List**, enter a descriptive name. Input a description in the **Description** field if desired. Select "Default Instance" for **Instance**.



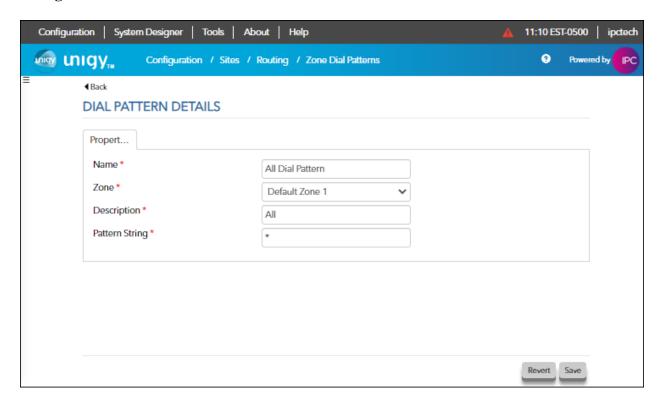
Click the +Assign icon and select the trunk group from Section 7.3.



Click the **Assign** icon to return to the route lists window. Click **Save**.

7.6 Administer Zone Dial Patterns

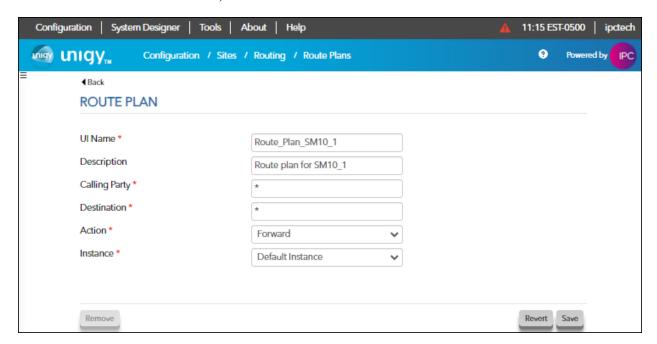
Select Routing → Zone Dial Pattern. Click Add New. Enter an appropriate value for Name. Select Default Zone 1 for Zone. Enter an appropriate value for Description. Input * for Pattern String. Click Save.



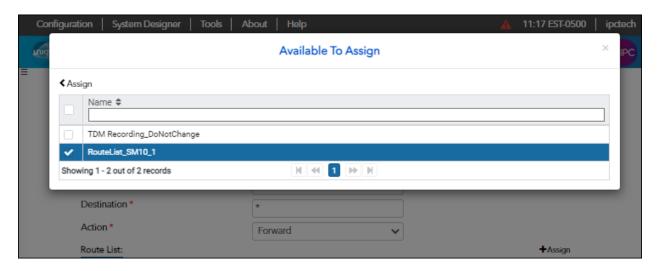
7.7 Administer Route Plans

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

In the ROUTE PLAN pane, enter a descriptive **UI Name** and optional **Description**. For **Calling Party**, enter "*" to denote any calling party from Unigy. For **Destination**, enter "*". For **Action**, select "Forward". For **Instance**, select "Default Instance". Click **Save**.



Click +Assign to open the Available To Assign window. Select the Route List from Section 7.4. Click on the <Assign icon to return to the route plan window. Click Save.



8 Verification Steps

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

8.1 Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk group 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.

```
TRUNK GROUP STATUS
Member
                Service State
                                   Mtce Connected Ports
         Port.
                                    Busy
0001/0001 T000001 in-service/idle
0001/0002 T000002 in-service/idle
0001/0003 T000003 in-service/idle
0001/0004 T000004 in-service/idle
                                    no
0001/0005 T000005 in-service/idle
0001/0006 T000006 in-service/idle
                                   no
0001/0007 T000007 in-service/idle
                                    no
0001/0008 T000008 in-service/idle
0001/0009 T000009 in-service/idle
0001/0010 T000010 in-service/idle
                                    no
```

Verify the status of the SIP signaling group by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.2**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

```
status signaling-group 1
STATUS SIGNALING GROUP

Group ID: 1
Group Type: sip

Group State: in-service
```

Verify the codec set specified is used in the calls made between Avaya sets and the turret sets. For example, with the codec set to only G.729 as below:

```
change ip-codec-set 1

IP MEDIA PARAMETERS

Codec Set: 1

Audio Silence Frames Packet

Codec Suppression Per Pkt Size(ms)

1: G.729

n 2 20
```

The trunk status on the call should show the codec used:

```
status trunk 1/1 Page 3 of 3

SRC PORT TO DEST PORT TALKPATH

src port: T000001

T000001:TX:10.64.10.201:2048/g729/20ms

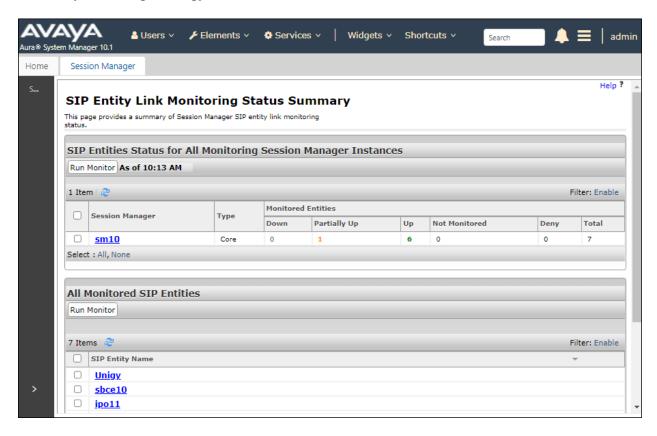
AMS1:RX:10.64.110.214:6050/g729/20ms:TX:cnfID:0

AMS1:RX:cnfID:0:TX:10.64.110.214:6052/g729/10ms

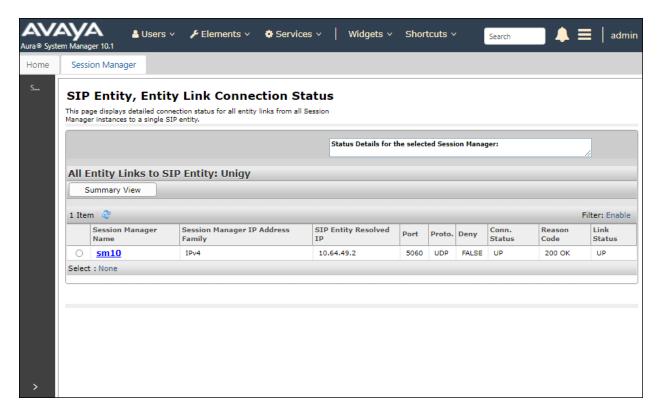
T000009:RX:10.64.10.214:16426/g729/10ms
```

8.2 Verify Avaya Aura® Session Manager

The SIP trunk status between Session Manager and Unigy is verified through System Manager. From the System Manager home page (not shown), select **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Under **All Monitored SIP Entities**, click on the Unigy SIP entity name, e.g., "Unigy", 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 calls in both directions from IPC MAX and TOUCH turret endpoints to Avaya endpoints. Verify calls connect with two-way talk paths.

9 Conclusion

These Application Notes describe the configuration steps required for IPC Unigy 5.2 sp2 to successfully interoperate with Avaya Aura[®] Session Manager R10.1 and Avaya Aura[®] Communication Manager R10.1. All feature and serviceability test cases were completed with observations as noted in **Section 2.2**.

10 Additional References

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

The following Avaya product documentation can be found at http://support.avaya.com.

- 1. Administering Avaya Aura® Communication Manager, Release 10.1.x, Issue 2, September 2022.
- 2. Avaya Aura® Communication Manager Feature Description and Implementation, Issue 6, Release 10.1, September 2022.
- 3. Administering Avaya Aura® Session Manager, Release 10.1, Issue 4, September 2022.
- 4. Administering Avaya Aura® System Manager, Issue 7, Release 10.1.x, September 2022.

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