Avaya Solution & Interoperability Test Lab

Application Notes for PatientSafe Solutions' PatientTouch Communications with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for PatientSafe Solutions' PatientTouch Communications to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

PatientTouch Clinical Communications solution facilitates hospital care team collaboration by combining real-time clinical context with intuitive technology with minimal IT requirements. In the compliance testing, PatientTouch Communications used SIP trunks to Avaya Aura® Session Manager, for PatientTouch users 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 the 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 PatientSafe Solutions' PatientTouch Communications to interoperate with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager) using SIP trunks.

PatientTouch Communications facilitates hospital care team collaboration by combining real-time clinical context with intuitive technology with minimal IT requirements. By delivering secured texting, voice, alerts, and critical context, PatientTouch Communications provides clinicians with the right information, at the right time, about the right patient, in the right way—to deliver better, safer, Connected Patient Care. In the compliance test, PatientTouch Communications used SIP trunks to Session Manager allowing PatientTouch users to reach users on Communication Manager and on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established among PatientTouch users with Avaya SIP, H.323, and/or PSTN users.

The serviceability test cases were performed manually by disconnecting and reconnecting the LAN connection to PatientTouch Communications.

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 this DevConnect Application Note 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 this Application Note, the interface between Avaya systems and PatientSafe Solutions did not include use of any specific encryption features as requested by PatientSafe Solutions.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included; basic call, display, G.711, hold/reconnect, call forwarding and call transfer.

The serviceability testing focused on verifying the ability of PatientTouch Communications to recover from adverse conditions, such as disconnecting/reconnecting the LAN connection.

2.2. Test Results

All test cases passed with following observations:

- During the compliance test, only G711MU was utilized.
- This version of PatientTouch Communications does not support conference feature and hence it was not tested.
- PatientTouch users are not members of voice messaging system and therefore Message Waiting Indicator (MWI) feature testing is not relevant for this compliance testing.
- Forwarding feature for PatientTouch users works in following scenarios, forward on manual decline, forward no answer, forward on busy, and forward on unavailable (i.e. user being logged out).

2.3. Support

Technical support on PatientTouch Communications can be obtained through the following:

- **Phone:** +1 (858) 746-3100
- **Email:** support@patientsafesolutions.com

3. Reference Configuration

Figure 1 illustrates a sample configuration of PatientTouch Communications that consists of PatientTouch Server and clients. SIP trunks are used from PatientTouch Communications to Session Manager, to reach users on Communication Manager and on the PSTN.

A five digit dialing plan was used to facilitate dialing between the Avaya and PatientTouch sites. Unique extension ranges were associated with Communication Manager users (56xxx), and PatientTouch users (71xxx).

The configuration of Session Manager is performed via the web interface of Avaya Aura® System Manager (System Manager). The configuration of Communication Manager is performed via the SAT interface. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.

The configuration of PatientTouch Communications and on iOS devices was performed by a PatientSafe Solutions engineer prior to the solution testing. During compliance testing the PatientTouch Server was installed as a virtual machine on a VMware host server.

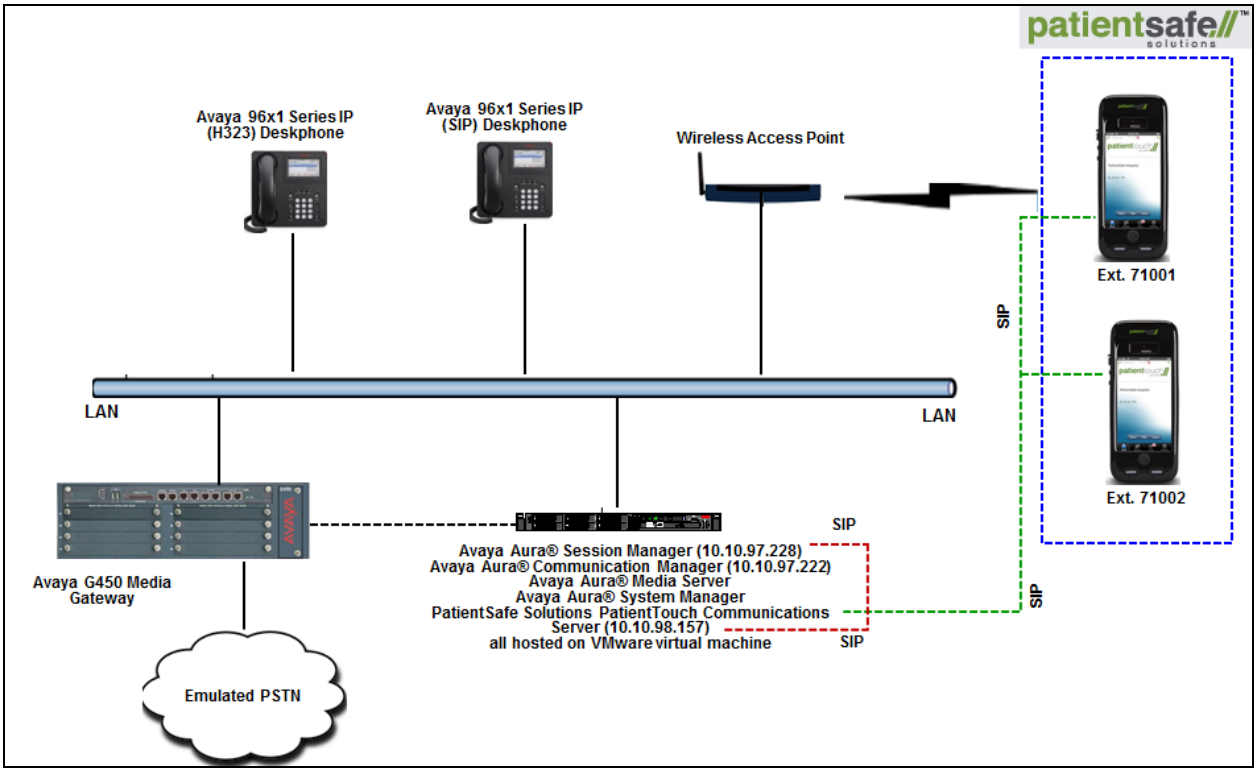


Figure 1: Avaya SIP Network with PatientSafe Solutions PatientTouch Communications Server

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 running on virtualized environment	7.1.0.0.532
Avaya Aura® Session Manager running on virtualized environment	7.1.0.0.710028
Avaya Aura® System Manager	7.1.0.0.1125193
Avaya Aura® Media Server	7.7.0.359
Avaya G450 Media Gateway	38.18.0 /1
Avaya 96x1 Series IP Telephone • 9611 (H.323) • 9641GS (SIP)	 6.6401 7.0.1.2.9
PatientSafe Solutions • PatientTouch® Communications (Server/App) • iPhone 5s (ME296LL/A)	 3.3.1.3129 /2.9.5.26099 10.3.2

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on the Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of the Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

It is implied a working system is already in place. The configuration operations described in this section can be summarized as follows: (Note: During Compliance Testing all inputs not highlighted in Bold were left as Default)

- Verify License
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer dial plan
- Administer uniform dial plan
- Administer AAR analysis

5.1. Verify License

Log in to the System Access Terminal 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.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 12000 10
          Maximum Concurrently Registered IP Stations: 18000 5
            Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
             Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 1
                      Maximum Administered SIP Trunks: 24000 54
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. 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*

```
add trunk-group 1                                          Page   1 of  22
                              TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Trunk to SM on VM          COR: 1        TN: 1        TAC: #001
   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: 24
```

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

```
add trunk-group 1                                          Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: internal
                                                      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

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


 Show ANSWERED BY on Display? y

 DSN Term? n
```

5.3. 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 node name for Session Manager
- **Near-end Listen Port:** An available port for integration with Session Manager
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**
- **Far-end Network Region:** An existing network region to use with Session Manager
- **Far-end Domain:** The applicable domain name for the network
- **Direct IP-IP Audio Connections:** *y*

```
add signaling-group 1                                        Page   1 of   3
                              SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
       Q-SIP? n
    IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 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: SM-VM
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                      Far-end Network Region: 1

Far-end Domain: bvwdev.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? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? y
         Enable Layer 3 Test? y                Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n         Alternate Route Timer(sec): 6
```

5.4. Administer SIP Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number from **Section 5.2**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.3**.
- **Number of Members:** The desired number of members, in this case *24*.

```
change trunk-group 1                                          Page   1 of  22
                              TRUNK GROUP

Group Number: 1                     Group Type: sip           CDR Reports: y
  Group Name: Trunk to SM on VM          COR: 1        TN: 1        TAC: #001
   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: 24
```

5.5. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section 5.3**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter *yes* 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 PatientTouch Communications.

```
change ip-network-region 1                                    Page   1 of  20
                            IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: bvwdev.com
    Name: Region1                Stub Network Region: n
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
```

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

```
change ip-network-region 1                                    Page   4 of  20

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

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. As per the observation noted in **Section 2.2** only configure either G.711MU or G.711A. The codec shown below was used in the compliance testing.

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

                         IP MEDIA PARAMETERS
    Codec Set: 1

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


     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
 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 PatientTouch Communications, 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 5.2**.
- **FRL:** A level that allows access to this trunk, with *0* being least restrictive.

```
change route-pattern 1                                          Page   1 of   3
                  Pattern Number: 1      Pattern Name: To SM on VM
    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                   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 Sub  Numbering LAR
    0 1 2 M 4 W     Request                           Dgts Format
 1: y y y y y n  n            rest                          lev0-pvt  none
```

5.8. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to PatientTouch Communications. Add an entry for the trunk group defined in **Section 5.2**. In the example shown below, all calls originating from a 5-digit extension beginning with 56 and routed to trunk group 1 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  56             1                           5     Total Administered: 4
                                                     Maximum Entries: 540
```

5.9. Administer Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 71xxx to PatientTouch Communications. Use the "change dialplan analysis 0" command, and add an entry to specify the use of digits pattern 71, as shown below

```
change dialplan analysis                                      Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                  Location: all            Percent Full: 2

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
    71          5   udp
```

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 71xxx to PatientTouch Communications. Note that other routing methods may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing of digits 71xxx, 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
  71             5   0                   aar  n
```

5.11. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 71xxx. In the example shown below, calls with digits 71xxx will be routed as an AAR call using route pattern *1* from **Section 5.7**.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
     71                   5    5    1         aar          n
```

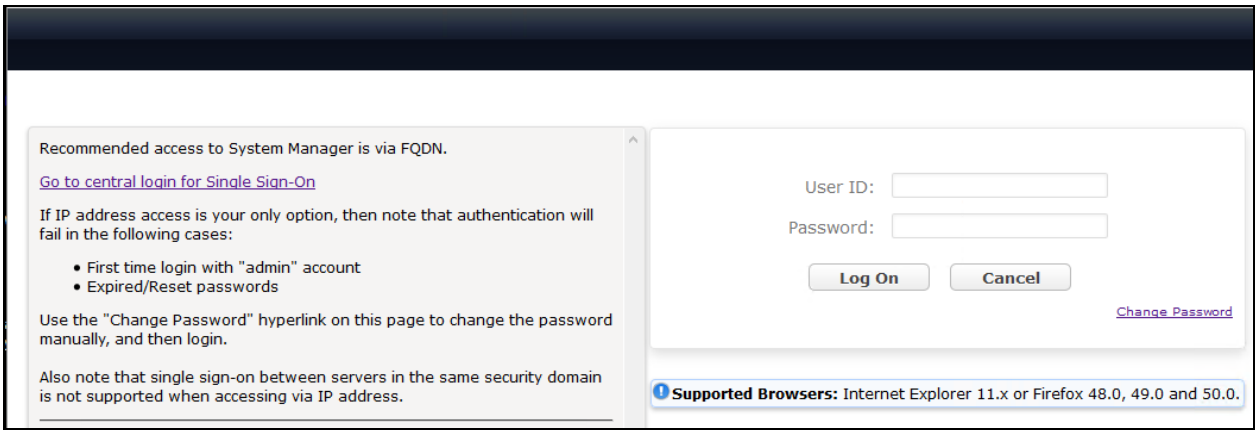
6. Configure Avaya Aura® Session Manager

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

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

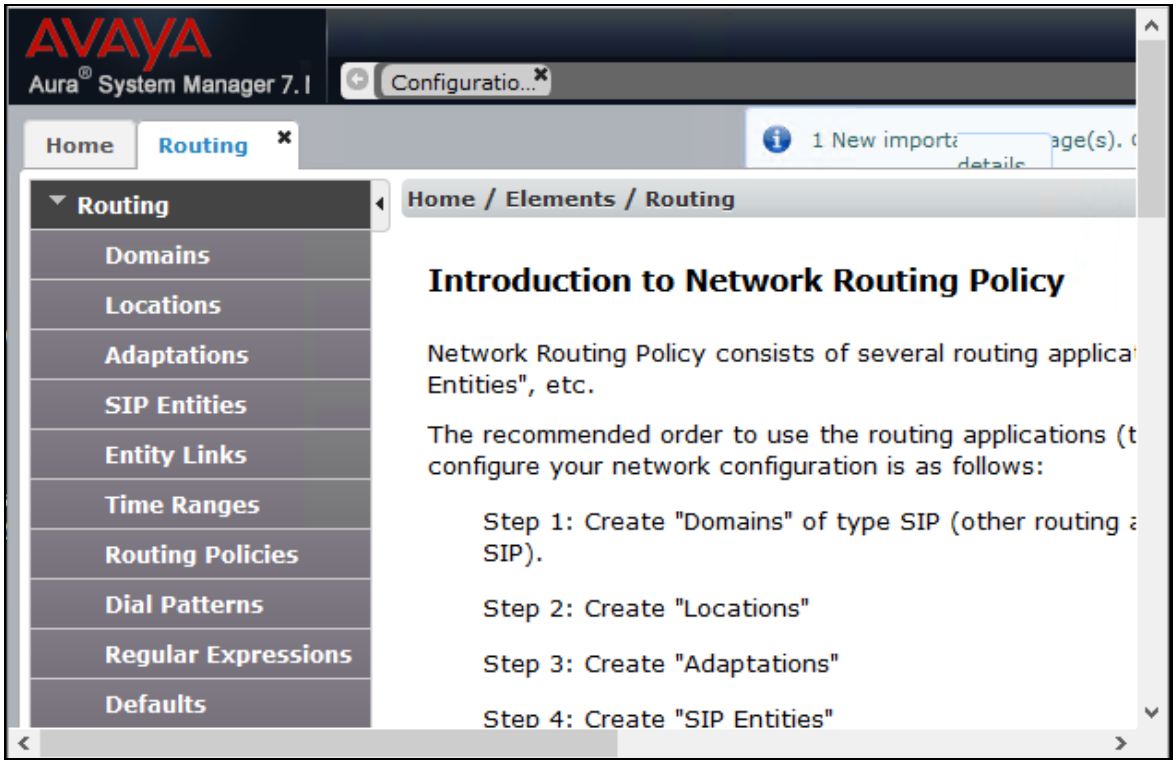
6.1. Launch System Manager

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

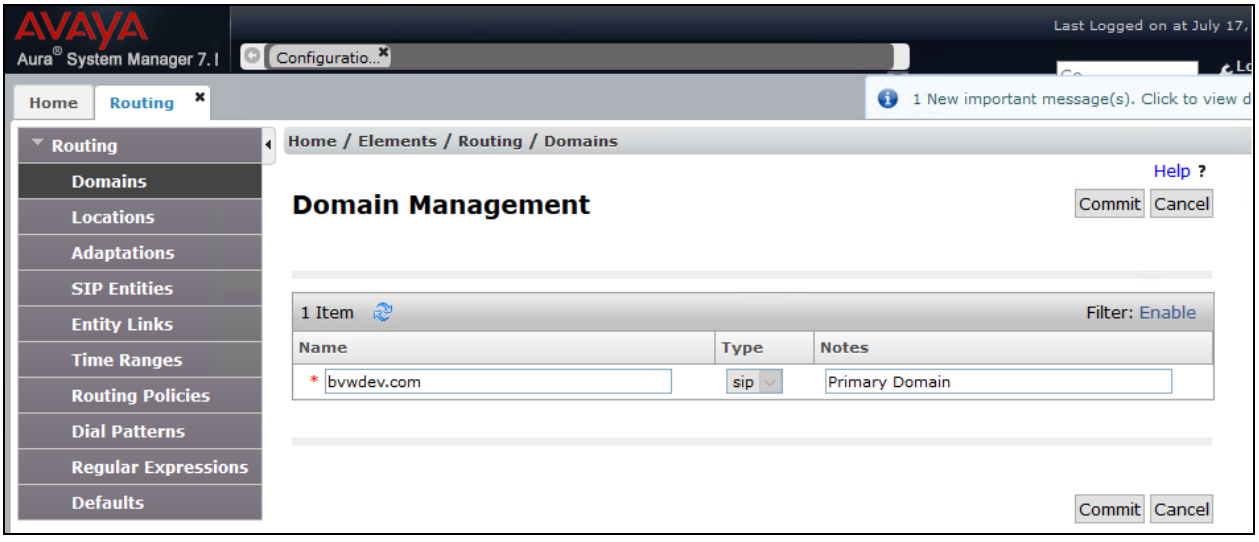


6.2. Administer Domain

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



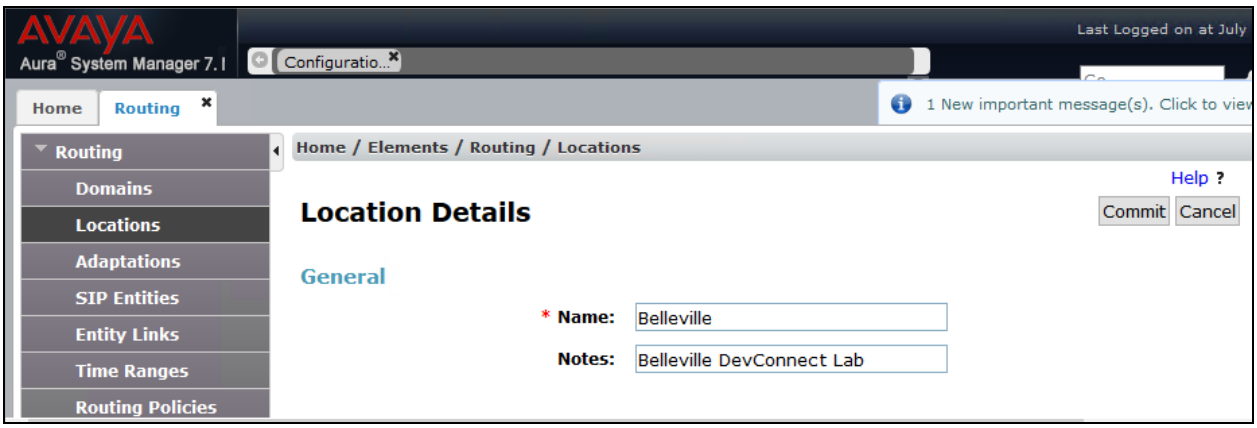
The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.



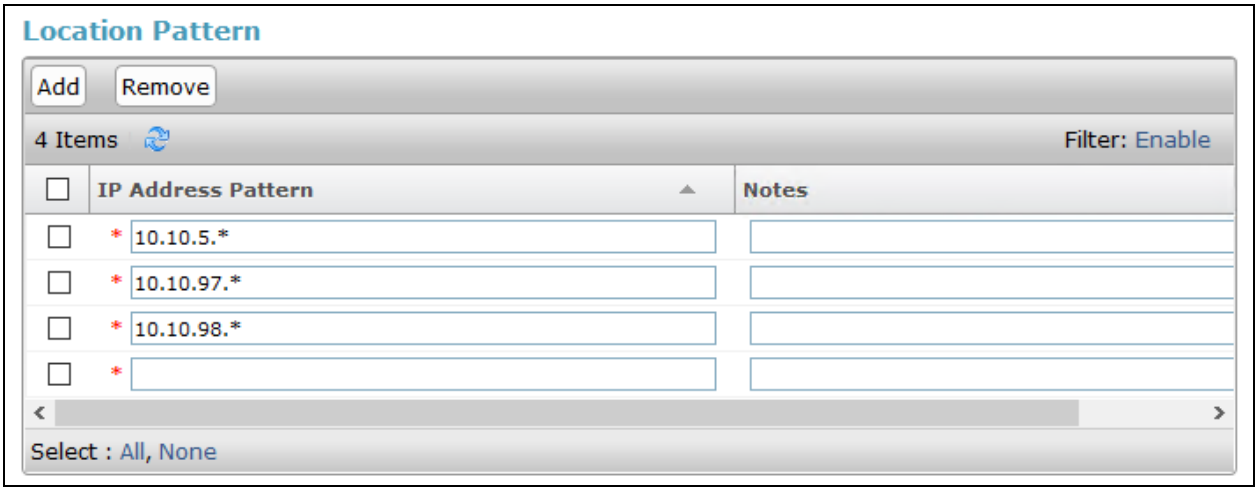
6.3. Administer Locations

Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for PatientTouch Communications.

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



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



6.4. Administer Adaptation

During compliance test, in order to make the call from and to Communication Manager via Session Manager, Adaptation to translate IP address into domain name is used for PatientTouch Communications SIP entity. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown).

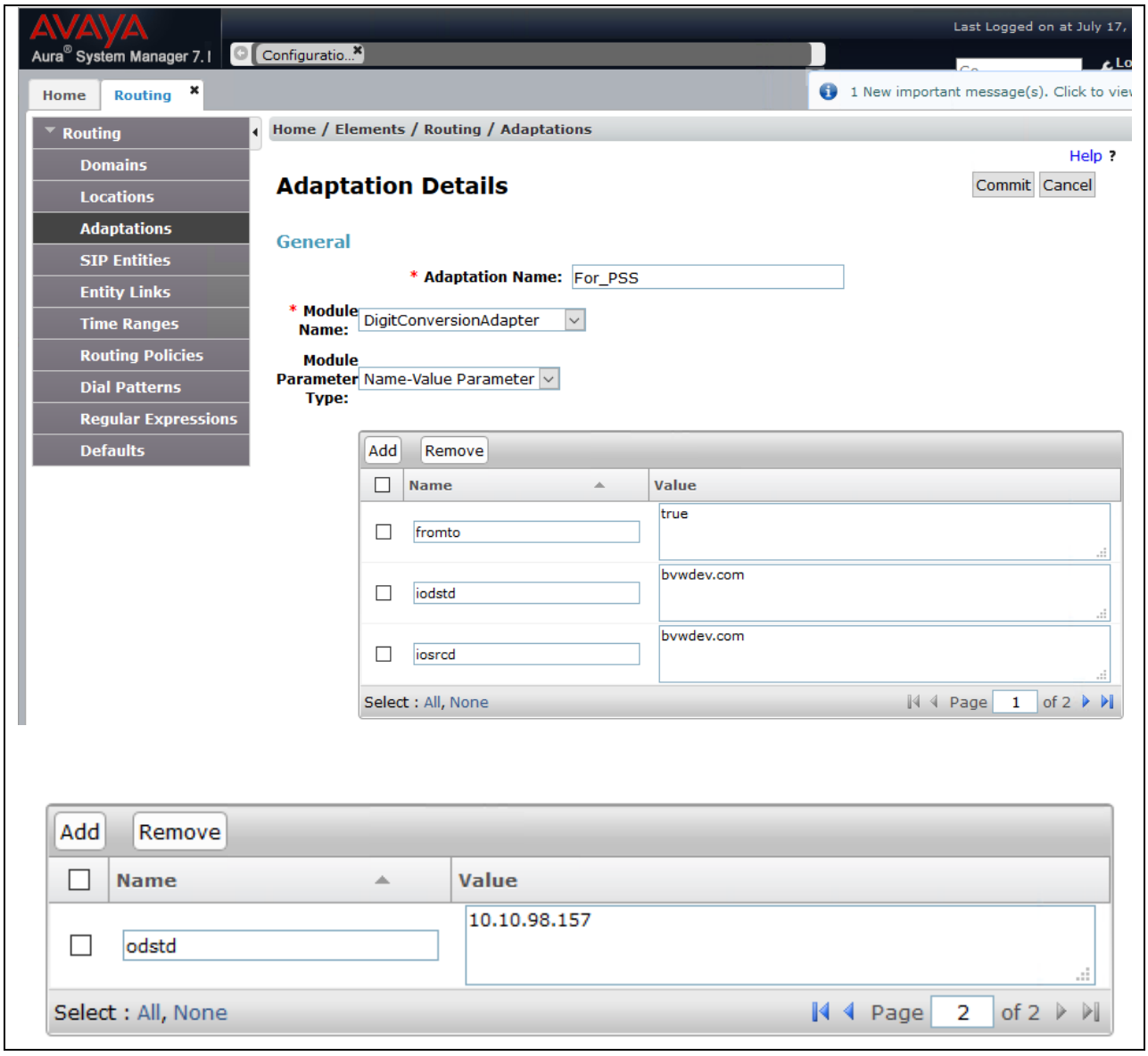
Enter the following for the PatientTouch Communications Adaptation.

- **Adaptation Name** An informative name (e.g., *For_PSS*)
- **Module Name** Select *DigitConversionAdapter*
- **Module Parameter Type** Select *Name-Value Parameter*

Click **Add** to add a new row for the following values as shown below table:

Name	**Value**
fromto	true
iodstd	Enter the domain name of system, example *bvwdev.com*
iosrcd	Enter the domain name of system, example *bvwdev.com*
odstd	Enter IP address of PatientTouch Communications, *10.10.98.157*

Once the correct information is entered click the **Commit** button. Screen below shows the Adaptation created for PatientTouch Communications.



6.5. Administer SIP Entities

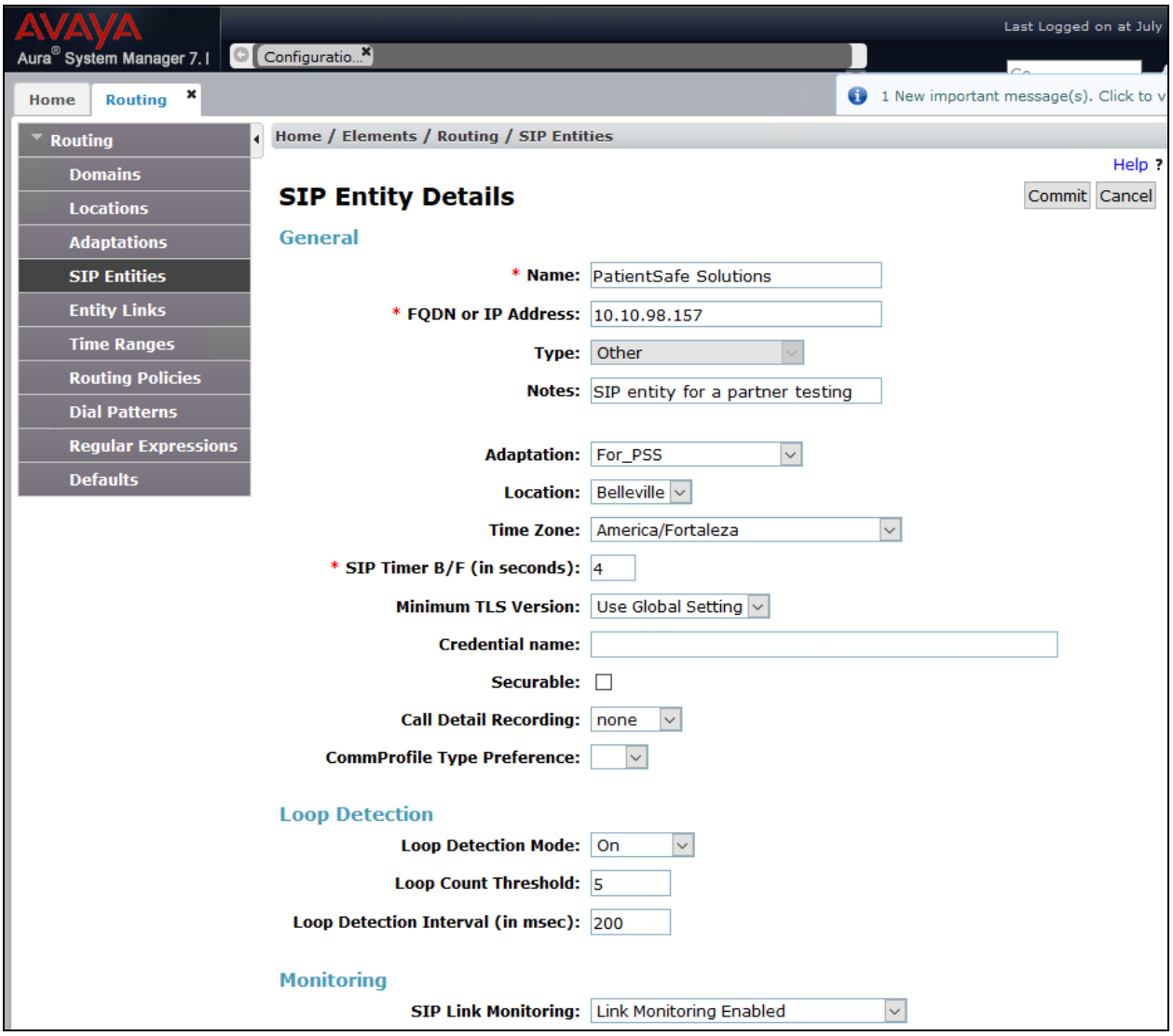
Add two new SIP entities, one for PatientTouch Communications and one for the new SIP trunks with Communication Manager.

6.5.1. SIP Entity for PatientTouch Communications

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

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

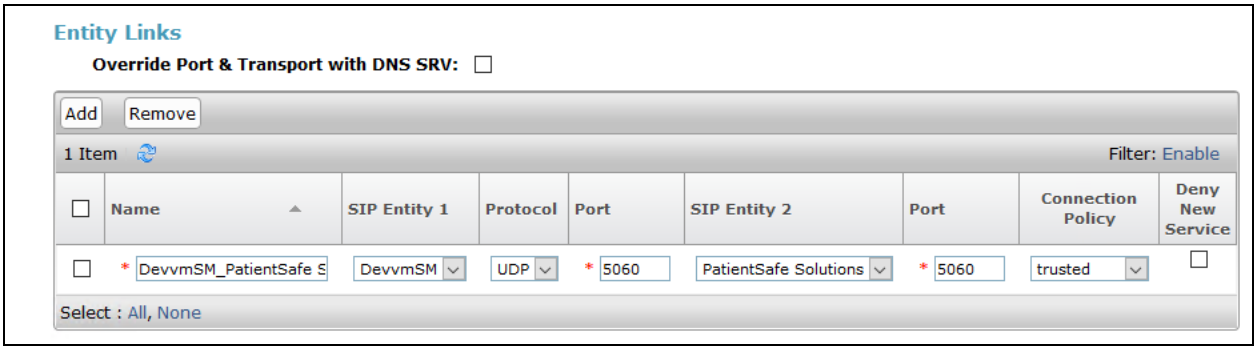
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of PatientTouch Communications server.
- **Type:** *Other*
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 6.4**.
- **Location:** Select the PatientTouch Communications location name from **Section 6.3**.
- **Time Zone:** Select the applicable time zone.



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

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

Note that only UDP protocol was tested.

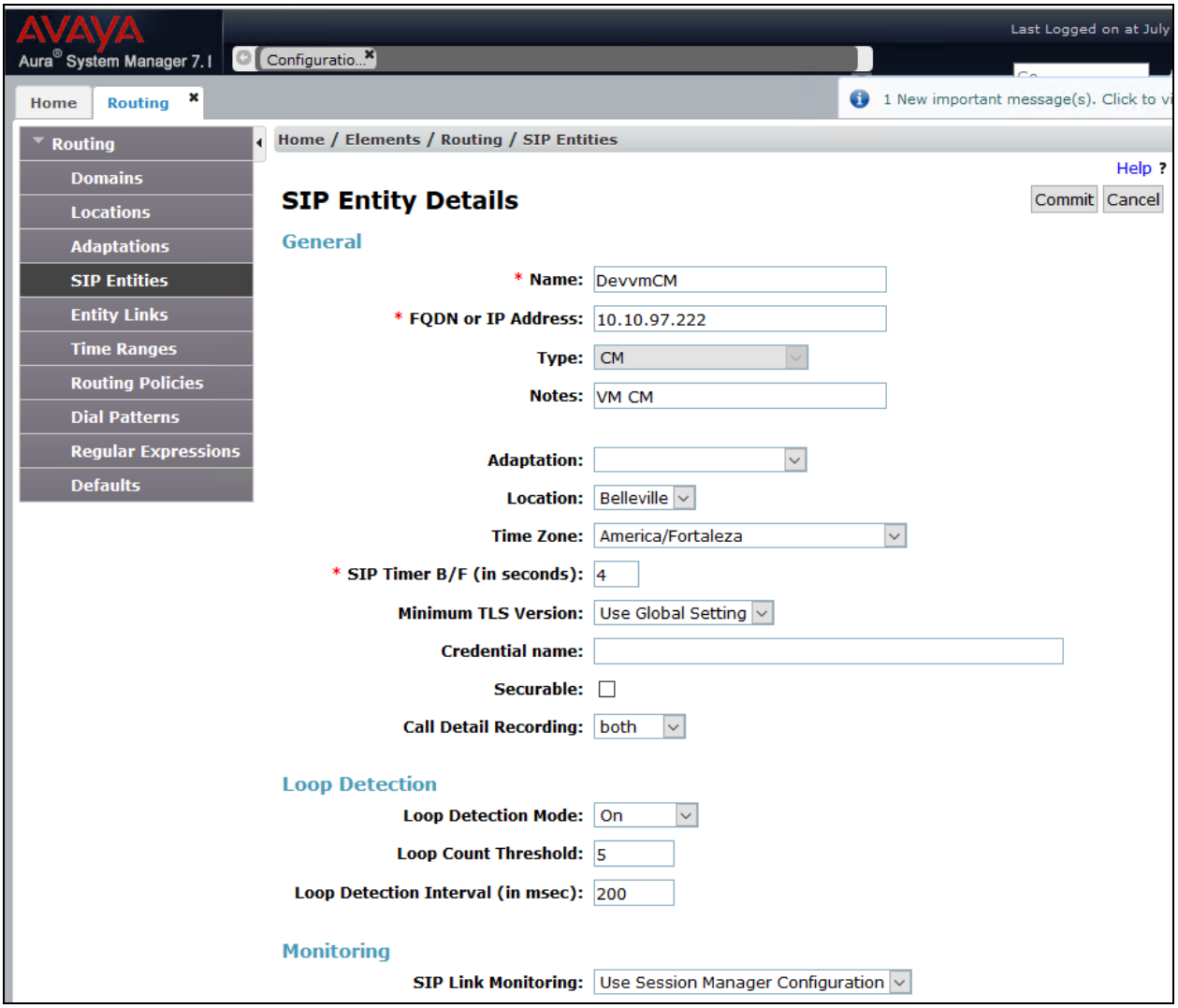


6.5.2. SIP Entity for Communication Manager

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

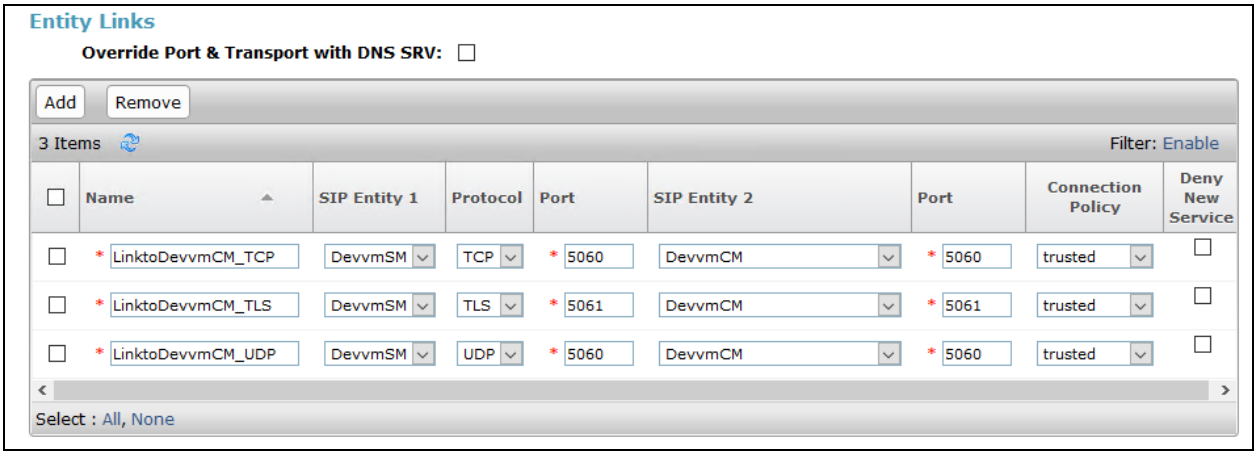
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



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

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case *DevvmSM*.
- **Protocol:** The signaling group transport (*TLS*) method from **Section 5.3**.
- **Port:** The signaling group listen port (*5061*) number from **Section 5.3**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port (*5061*) number from **Section 5.3**.
- **Connection Policy:** *trusted*



6.6. Administer Routing Policies

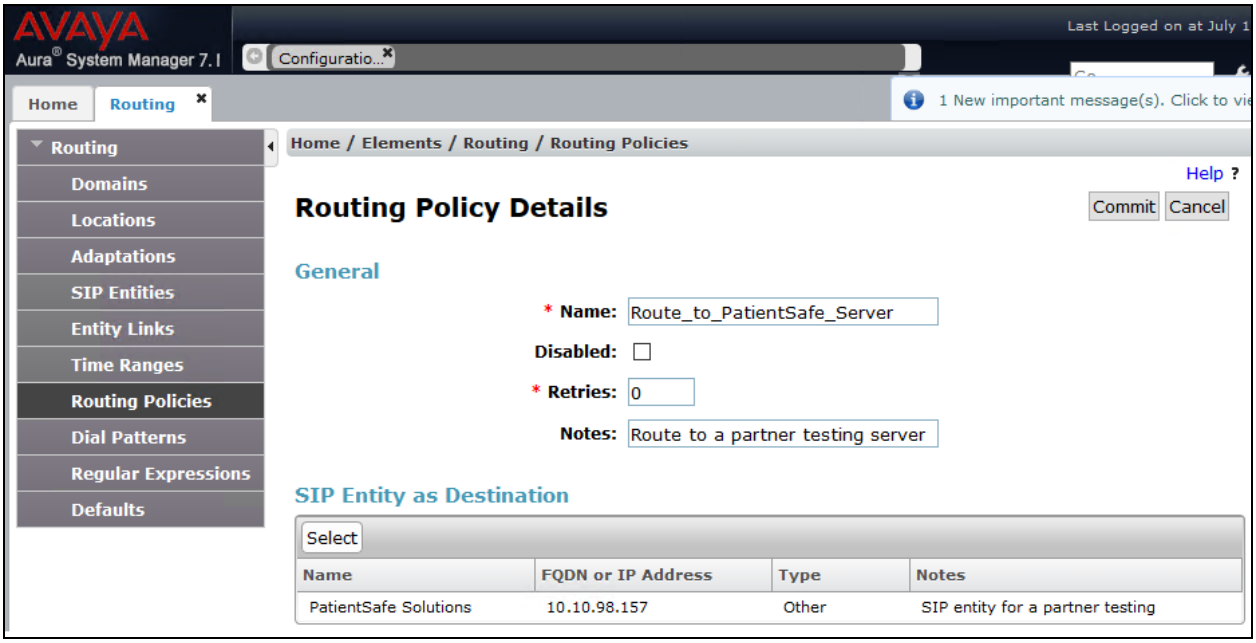
Add two new routing policies, one for PatientTouch Communications and one for the new SIP trunks with Communication Manager.

6.6.1. Routing Policy for PatientTouch Communications

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

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the PatientTouch Communications entity name from **Section 6.5.1**. The screen below shows the result of the selection.

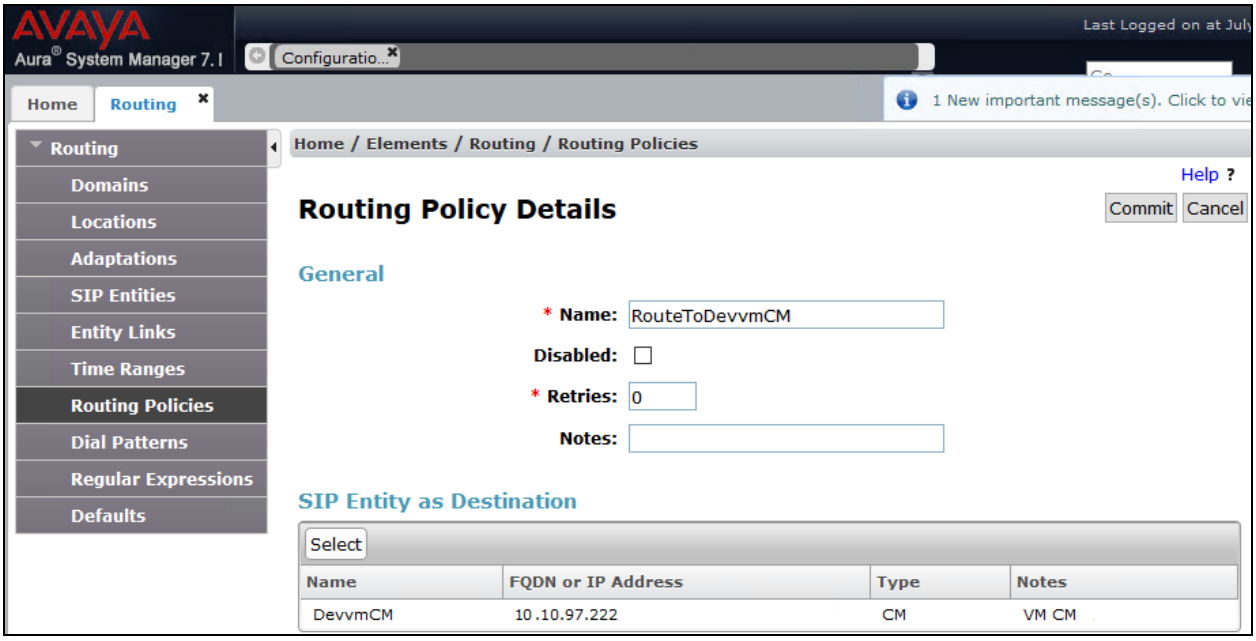


6.6.2. Routing Policy for Communication Manager

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

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.5.2**. The screen below shows the result of the selection.



6.7. Administer Dial Patterns

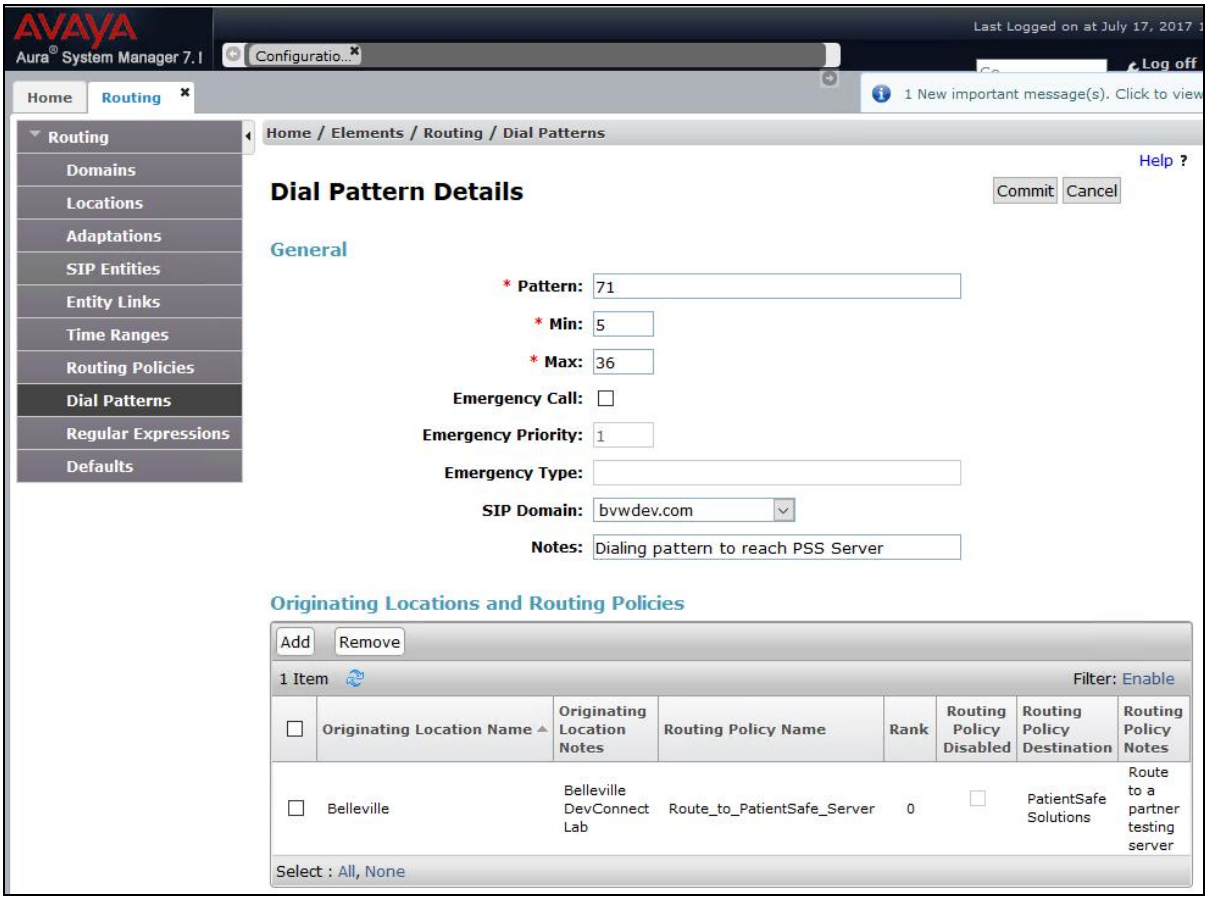
Add a new dial pattern for PatientTouch Communications and Communication Manager.

6.7.1. Dial Pattern for PatientTouch Communications

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 PatientTouch Communications. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case *71*.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 6.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching PatientTouch Communications. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations *Belleville*. The PatientTouch Communications routing policy from **Section 6.6.1**was selected as shown below.



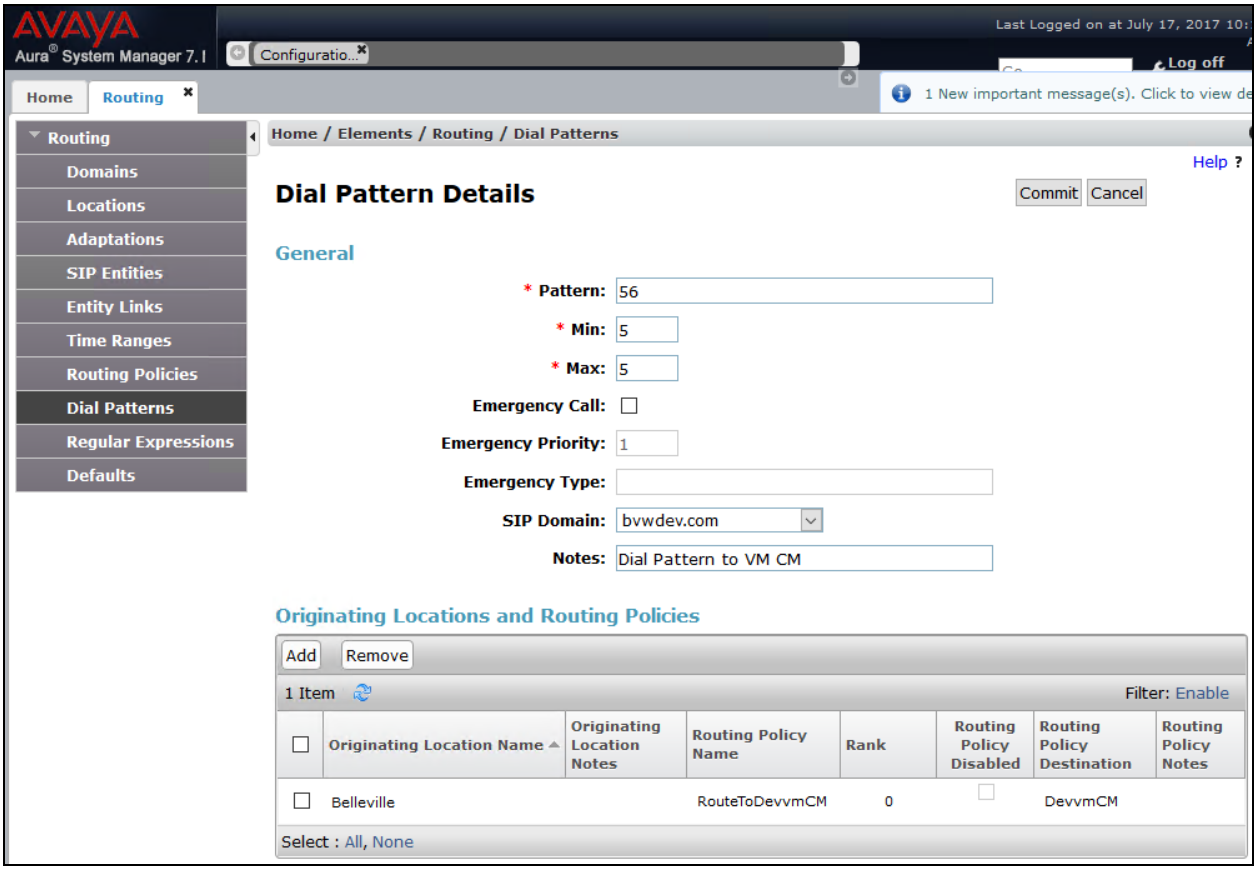
6.7.2. Dial Pattern for Communication Manager

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

- **Pattern:** A dial pattern to match, in this case *56*.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 6.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for call originations from all PatientTouch Communications endpoints in locations *Belleville*. The Communication Manager routing policy from **Section 6.6.2**was selected as shown below.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN (not shown).



7. Configure PatientSafe's PatientTouch Communications

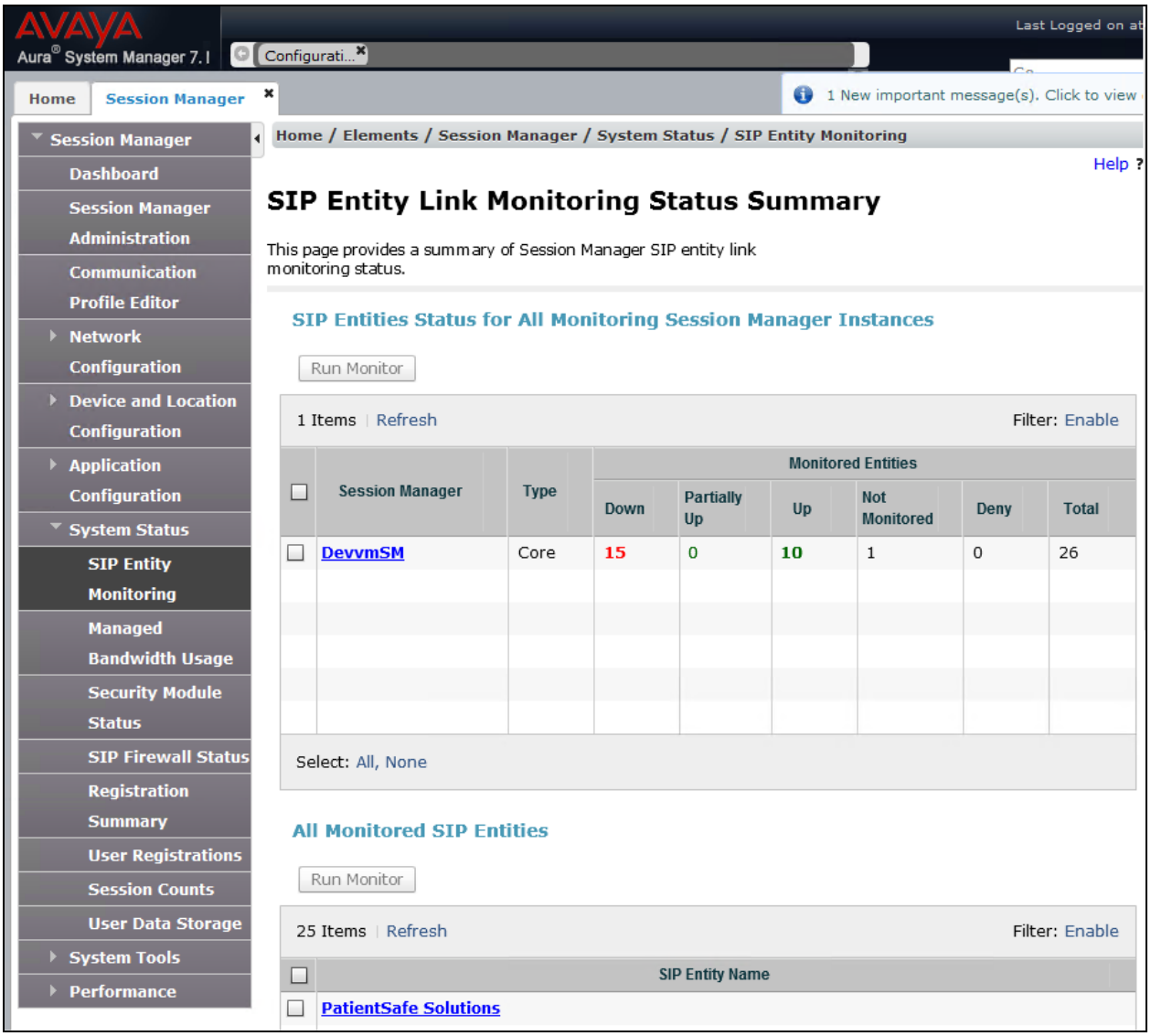
PatientSafe engineer installs, configures, and customizes the PatientTouch applications for their end customers. By PatientSafe Solutions request, installation/configuration steps were not included in these Application Notes. To acquire above information, please contact PatientSafe Solutions.

8. Verification Steps

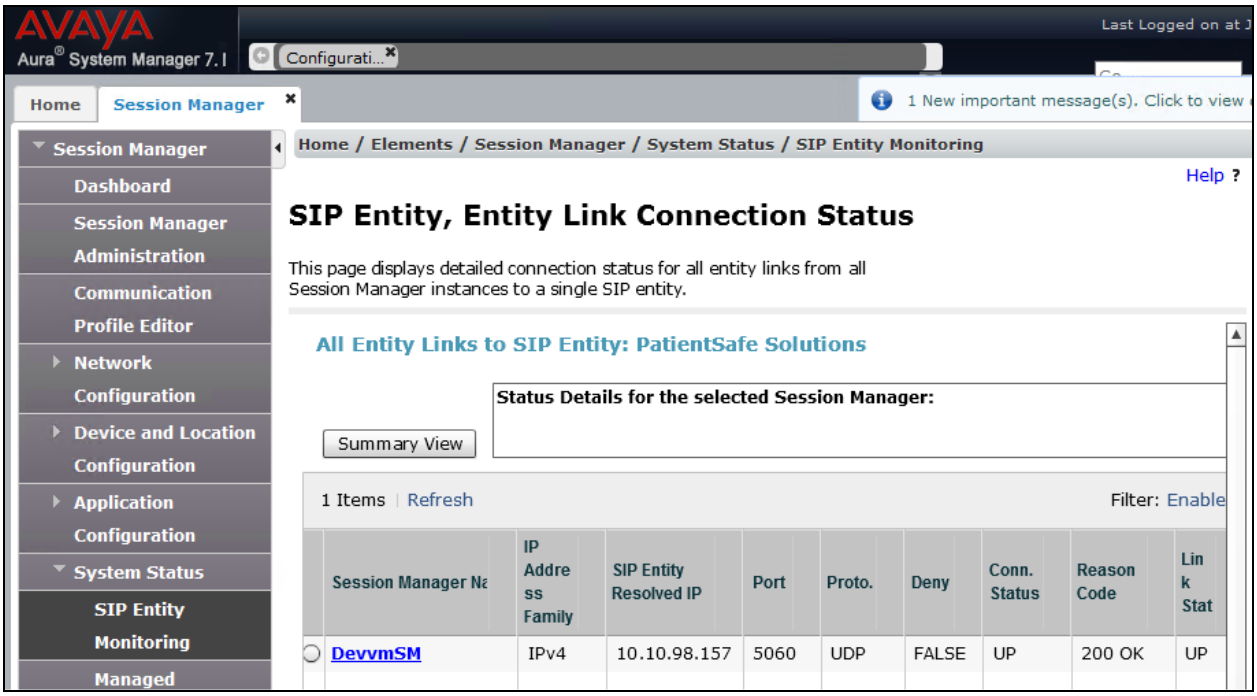
This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager.

8.1. Verify Avaya Aura® Session Manager

From the System Manager home page, 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 PatientTouch entity name from **Section 6.5.1**.



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



8.2. Verify PatientSafe Solutions' PatientTouch Communications

Please contact PatientSafe Solutions for information.

9. Conclusion

These Application Notes describe the configuration steps required for PatientSafe's PatientTouch Communications to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. 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.

Product documentation for Avaya products may be found at http://support.avaya.com.

1. *Administering Avaya Aura® Session Manager*, Release 7.1, Issue 1 May 2017
2. *Deploying Avaya Aura® System Manager*, Release 7.1, Issue 1 May 2017
3. *Administering Avaya Aura® System Manager for Release 7.1*, Release 7.1, Issue 2 May 2017
4. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1, Issue 1 May 2017

For PatientTouch Communications product documents, please contact PatientSafe Solutions.

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.