Avaya Solution & Interoperability Test Lab

Application Notes for Prism-IPX Systems PriMega Messaging Gateway with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for PriMega Messaging Gateway to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

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 outline the steps necessary to configure PriMega Messaging Gateway from Prism-IPX Systems to interoperate with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager). PriMega Messaging Gateway (hereafter referred to as PriMega) is a server-based application running on Linux platform. PriMega SIP User Agent Interfaces for Linux platforms. Interfaces are used for allowing callers to enter numeric digits for sending to pagers on PriMega platform.

PriMega connects to Communication Manager using a SIP trunk via the Session Manager. PriMega is supplied with all prerequisite software. Communication Manager also connects to PriMega when it is being used as a Paging adjunct for Meet-Me Paging features by initiating a TCP/IP connection. In this case PriMega sends TAP (Telocator Alphanumeric Protocol) messages which is an outbound protocol to pagers on the PSTN.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using Communication Manager. PriMega communicates with Communication Manager using a SIP trunk through the Session Manager. See **Figure 1** for a network diagram. A dial plan was configured on the Communication Manager to route calls to PriMega. Calls are placed to PriMega and the digits required to be sent to the pager by PriMega are entered by the caller after which PriMega automatically disconnects the call.

For the Meet-Me Paging feature, SA8312 is enabled in Communication Manager which allows a Page Line (station that is administered with the Paging station type) to interface (using TCP/IP) with paging equipment, in this case PriMega. When a Page Line is called, MultiVantage sends a TAP formatted paging message (described in document mentioned in **Section 10**) to the PriMega Paging Adjunct, which in turn, alerts the pager associated with the called page line.

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 PriMega did not include use of any specific encryption features as requested by Prism-IPX Systems.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if PriMega could resume after a link failure with Communication Manager/Session Manager. The testing included:

- Call PriMega using both internal and external users.
- Detection and confirmation by PriMega of digits entered by users.
- Termination of calls by PriMega after receiving digits from users.
- Termination of calls by PriMega if there is no activity by user within the configured time limit.
- Multiple simultaneous calls to PriMega.
- Internal and external users call the paging station and PriMega detects and acknowledges the same using the Meet-Me Paging feature.

2.2. Test Results

Tests were performed to insure full interoperability between PriMega and Communication Manager via Session Manager. The tests were all functional in nature and performance testing was not included. All test cases passed successfully.

2.3. Support

For technical support for Prism-IPX Systems products, please use the following web link. https://prism-ipx.com/.

Prism-IPX Systems can also be contacted as follows:
Phone: +1 678 242 5266
Fax: +1 678 242 5201
Web: https://prism-ipx.com/contact-us/
Email: prism-harktech_support@prism-ipx.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, which has a SIP Trunk connection to PriMega Messaging Gateway via the Session Manager. SIP and H.323 stations were configured on the Communication Manager to generate outbound calls to PriMega. For PSTN users calling PriMega, both PRI and SIP trunks were configured. For the Meet-Me Paging feature SA8312, connection between Communication Manager and PriMega is accomplished via TCP/IP.

PriMega was installed on a Linux OS platform.

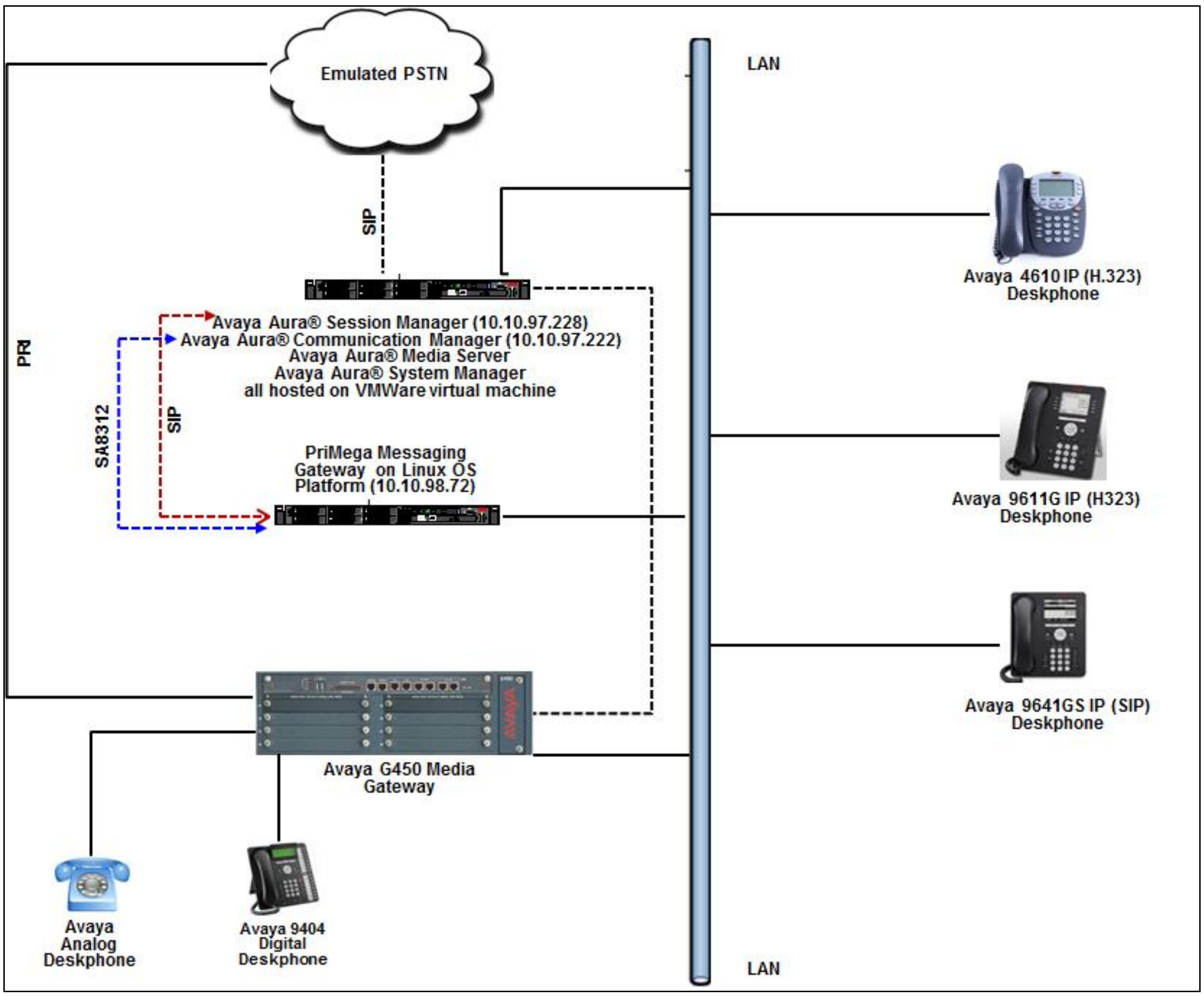


Figure 1: Avaya and PriMega Reference Configuration

4. Equipment and Software Validated

The following equipment and version were used in the reference configuration described above:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on virtualized environment	7.1.3.0.0-FP3
Avaya Aura® Session Manager running on virtualized environment	7.1.3.0.713014
Avaya Aura® System Manager running on virtualized environment	7.1.3.0
Avaya Aura® Media Server	7.8.0.384
Avaya G450 Media Gateway	39 .12 .0 /1
Avaya IP Deskphones: • 9611G (H.323) • 4610 (H.323) • 9641GS (SIP)	 6.6401 2.800 7.0.1.2.9
Avaya 9404 Digital Deskphone	18.0
Avaya Analog Deskphone	N/A
PriMega Messaging Gateway running on Linux OS	V9
PriMega Linux OS	Centos 7.4

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 dial plan
- Administer uniform dial plan
- Administer AAR analysis
- Enable special application 8312
- Configure node names
- Configure IP services
- Configure paging station type

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 34
```

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:** Enter "sip".
- **Group Name:** Enter a descriptive name.
- **TAC:** Enter an available trunk access code.
- **Service Type:** Enter "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**.

```
display 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:** Enter "sip".
- **Transport Method:** Enter "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?:** Enter "y".
- **Initial IP-IP Direct Media?:** Enter "y".

```
add signaling-group 1                                           Page   1 of   2
                                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? y
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 5.22**. 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 5.33**.
- **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 5.33**.

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 PriMega.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       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? y
   UDP Port Max: 8001
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 5.55**. Update the audio codec types in the **Audio Codec** fields as necessary. The codec shown below was used in the compliance testing.

```
display 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: G.729            n           2        20
 4:
 5:
 6:
 7:

     Media Encryption                     Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
```

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 PriMega, 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 Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 71xxx to PriMega. 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
   1            4    ext
   71           5    udp
```

5.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 71xxx to PriMega. 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
```

5.10. 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 5.77**.

```
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
```

5.11. Enable Special Applications 8312

For the Meet-Me Paging feature to be activated the Special Applications 8312 (SA8312) needs to be enabled. To enable this feature, use the "change system-parameters special-applications" command and in **Page 3**, enable the **(SA8312) - Meet-Me Paging?** by entering "y" as shown below.

```
change system-parameters special-applications                      Page   3 of  10
                             SPECIAL APPLICATIONS

                   (SA8141) - LDN Attendant Queue Priority? n
       (SA8143) - Omit Designated Extensions From Displays? n
            (SA8146) - Display Update for Redirected Calls? n
              (SA8156) - Attendant Priority Queuing by COR? n
               (SA8157) - Toll Free Vectoring until Answer? n
  (SA8201) - Start Time and 4-Digit Year CDR Custom Fields? n
                        (SA8202) - Intra-switch CDR by COS? n
                    (SA8211) - Prime Appearance Preference? n
                      (SA8240) - Station User Admin of FBI? n
                                 (SA8312) - Meet-Me Paging? y
                   (SA8323) - Idle Call Preference Display? n
```

5.12. Configure Node Names

Prior to administering PAGEx Service type, the user would have to administer all the relevant node names and their respective IP addresses by using the "change node-names ip" command. During compliance testing since PAGE1 and PAGE2 were configured, two node-names pointing to the same hardware, in this case PriMega was configured as shown below. Also, the node name for the Communication Manager is shown below. All these node names configured will be used in the next **Section 5.13**. Run the command "change node-names ip" and enter the following:

- **Name:** Enter descriptive names. For PriMega "PrismIPX1" and "PrismIPX2" were configured. For Communication Manager "procr" was configured.
- **IP Address:** Enter the IP Address of PriMega and Communication Manager.

```
change node-names ip                                                 Page   1 of   2
                                   IP NODE NAMES
    Name              IP Address
PrismIPX1           10.10.98.72
PrismIPX2           10.10.98.72
procr               10.10.97.222
```

5.13. Configure IP Services

The "change ip-services" command is to administer typical asynchronous adjuncts, e.g., PMS, CDR etc., that MultiVantage supports. The paging adjunct is in the same category of such adjuncts.

The PAGE1 and PAGE2 are considered as two separate page links when communicating with the paging adjunct. There is no priority indication when the links are assigned as PAGE1 or PAGE2. It is not required that both PAGE1 and PAGE2 are administered in the system however during compliance testing both were used. Run the "change ip-services" command and configure the following:

- **Service Type:** Enter "PAGE1" and "PAGE2".
- **Enabled:** Enter "y".
- **Local Node:** Enter "procr" which is the node name for Communication Manager as explained in **Section 5.12**.
- **Remote Node:** Enter "PrismIPX1" and "PrismIPX2" which is the node name for PriMega as explained in **Section 5.12**.
- **Remote Port:** During compliance testing "10004" and "20004" were used.

```
change ip-services                                              Page   1 of   4

                                    IP SERVICES
 Service      Enabled      Local         Local        Remote       Remote
  Type                     Node          Port         Node         Port
PAGE1          y       procr             0        PrismIPX1        10004
PAGE2          y       procr             0        PrismIPX2        20004
```

5.14. Configure Paging Station Type

SA8312 will support the page line through a station type - “PAGING”, in the station administration. When the station is administered as “PAGING” type, the port field will become a display only field with the value of ‘X’ displayed, i.e., administered without hardware (AWOH). The new station type keyword “PAGING” is displayed regardless whether the system “MeetMe Paging” option is enabled or not.

The PAGING station type is administrable only when the “Meet-Me Page” system option is enabled. However, SA8312 will not require the stations with “PAGING” type to be removed before the Meet-Me Paging (MMP) option can be disabled. To add this station type, run the “add station x” command, where “x” is an available extension and configure the following:

- **Type:** Enter “PAGING”.
- **Name:** Enter a descriptive name.
- **Send MMP Message:** Ensure that this is set to “y”.

```
add station 56503                                                    Page   1 of   4
                                     STATION

Extension: 56503                        Lock Messages? n                BCC: 0
     Type: PAGING                       Security Code:                   TN: 1
     Port: X                          Coverage Path 1:                  COR: 1
     Name: 56503, PrismIPX            Coverage Path 2:                  COS: 1
                                      Hunt-to Station:                Tests? n
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 1            Message Waiting Indicator: none
    Off Premises Station? n



          Survivable COR: internal
   Survivable Trunk Dest? y
Send MMP Message: y
```

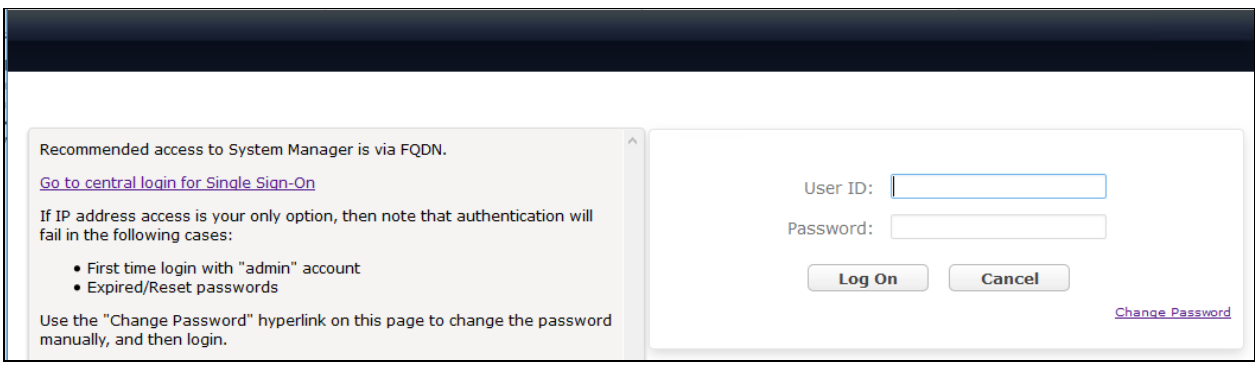
6. Configure Avaya Aura® Session Manager

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

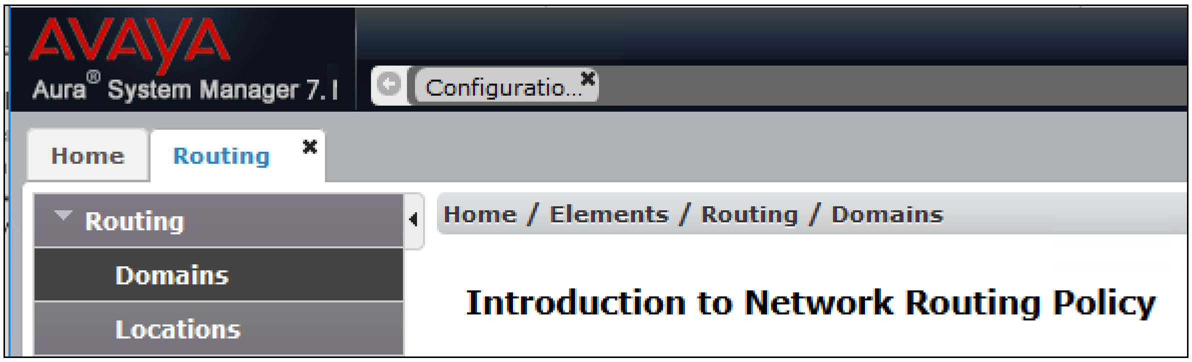
6.1. Launch System Manager

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.

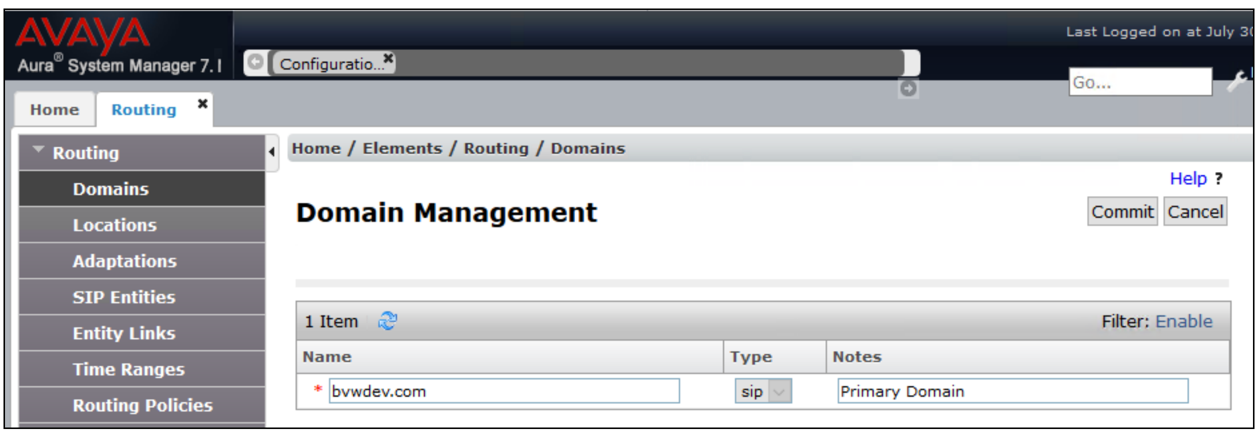


6.2. Administer Domain

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



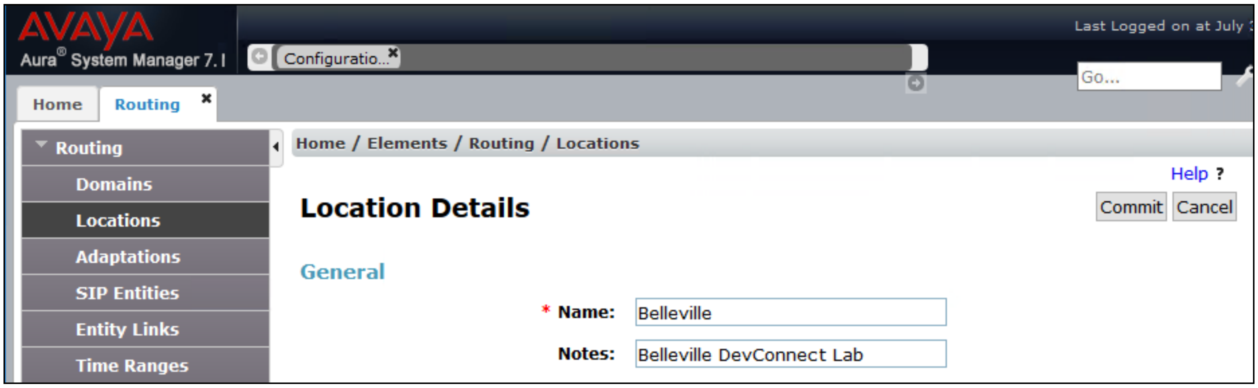
The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select "sip" from the **Type** drop down menu and provide any optional **Notes**.



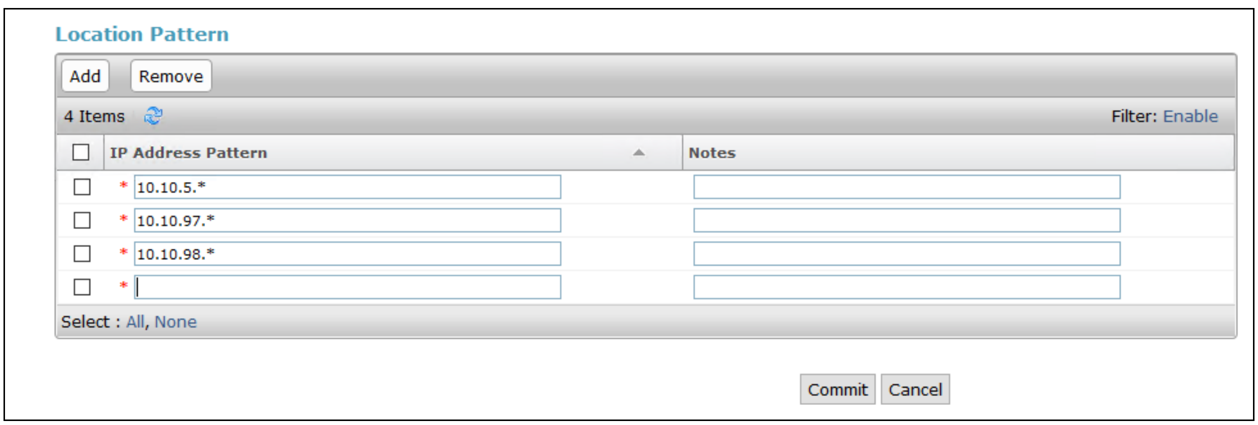
6.3. Administer Locations

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

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, to make the call from Communication Manager via Session Manager to PriMega, adaptation to translate IP address into domain name is used for PriMega SIP entity. Here is step on how to create Adaptation. 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 PriMega Adaptation.

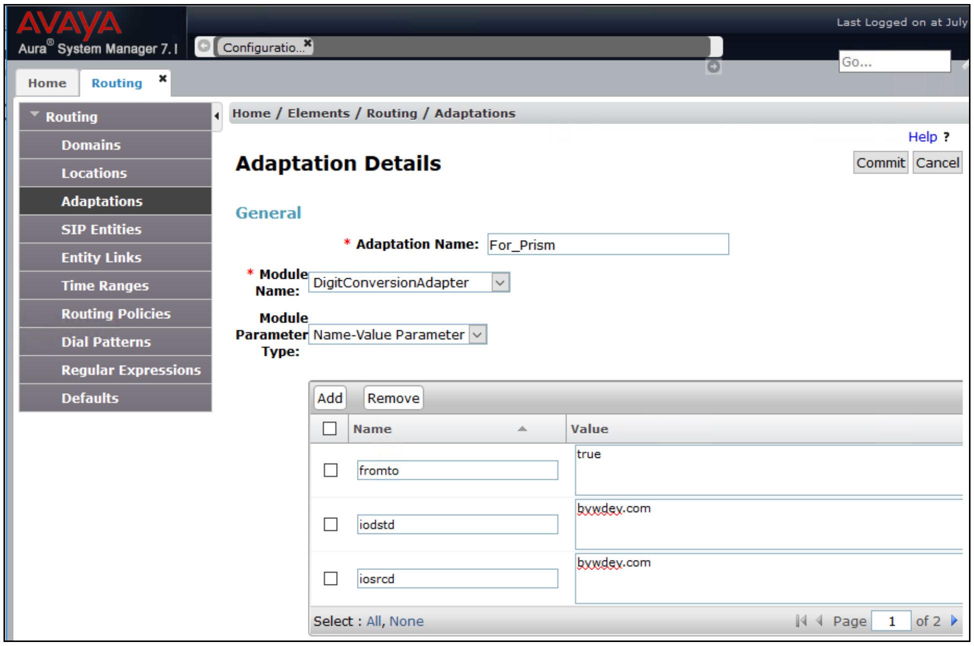
- **Adaptation Name** An informative name (e.g.," For_Prism").
- **Module Name** Select "DigitConversionAdapter".
- **Module Parameter Type** Select "Name-Value Parameter".

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

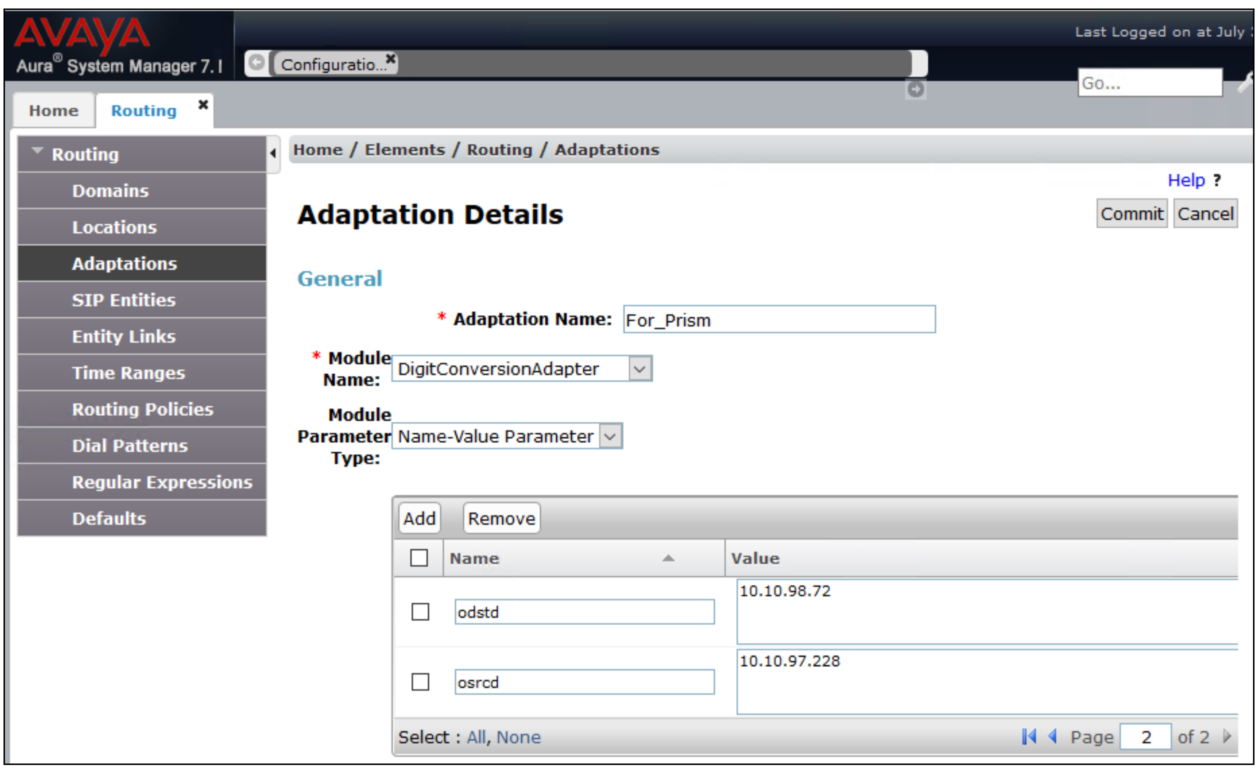
Name	**Value**
fromto	true
iodstd	Enter the domain name of system, ex: "bvwdev.com"
iosrcd	Enter the domain name of system, ex: "bvwdev.com"
odstd	Enter IP address of PriMega, ex: "10.10.98.72"
osrcd	Enter IP Address of Session Manager, ex: "10.10.97.228"

Once the correct information is entered click the **Commit** button.

Here is the screenshot show Adaptation created for PriMega.



(Continue) the screenshot show Adaptation created for PriMega:

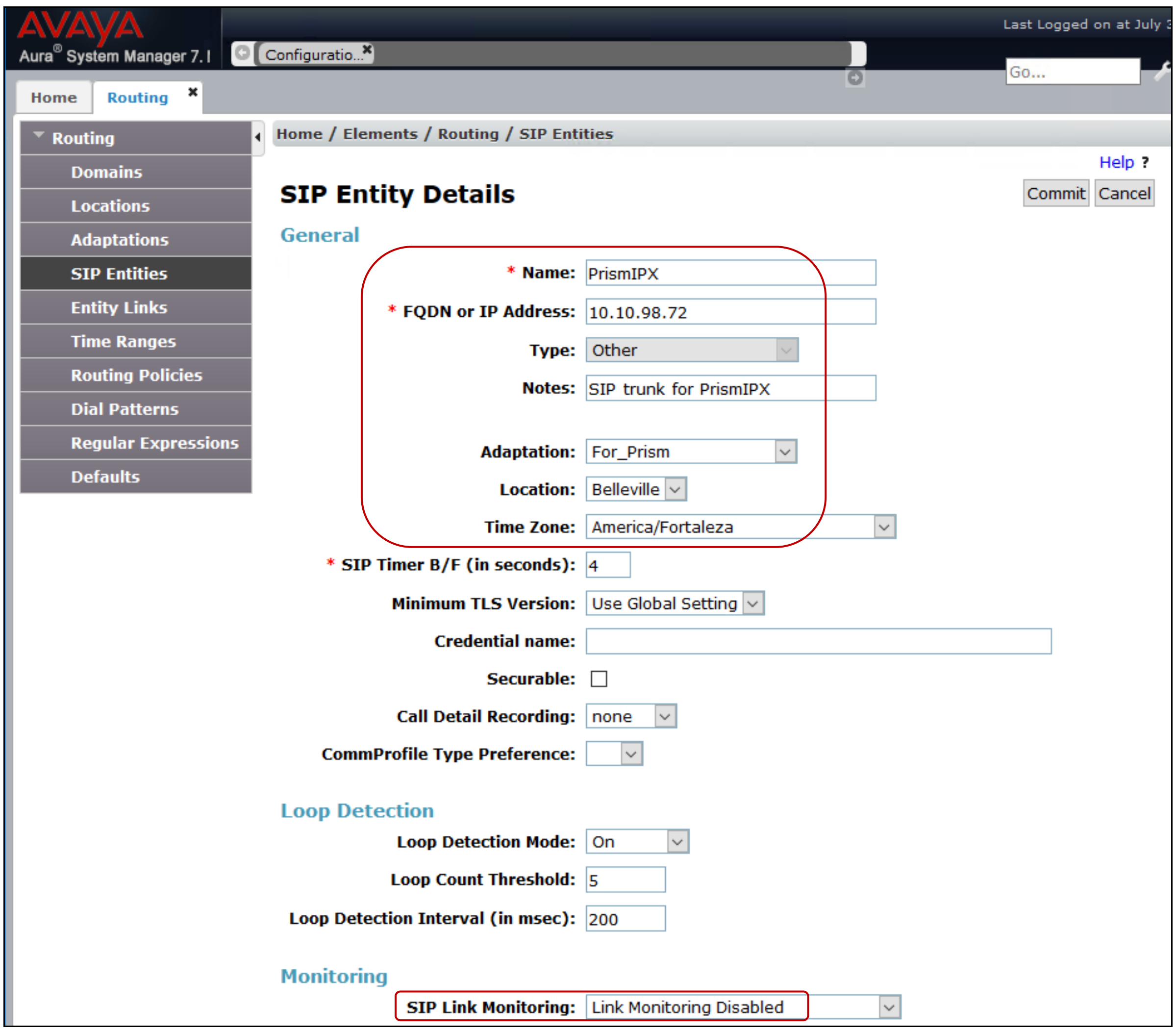


6.5. Administer SIP Entity for PriMega

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

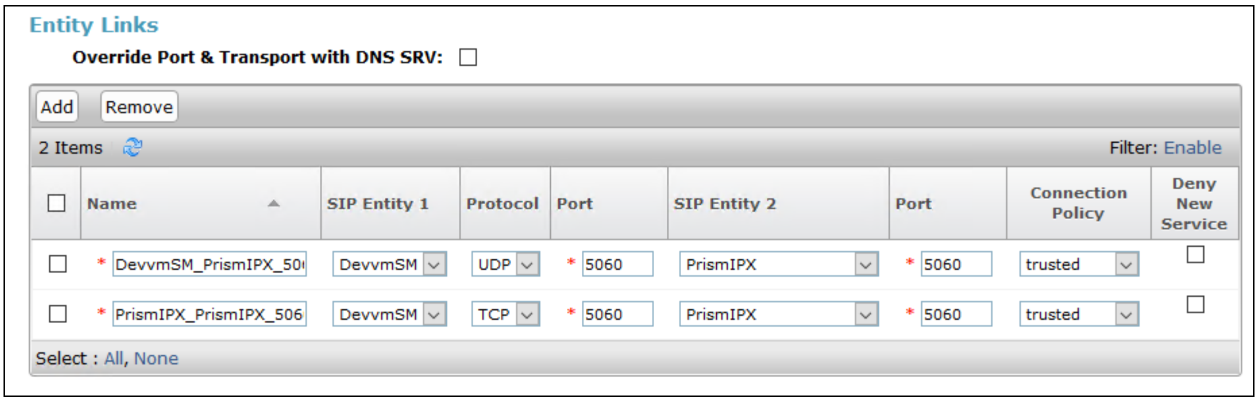
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the PriMega server.
- **Type:** Select "Other" from the drop-down menu.
- **Notes:** Any desired notes.
- **Adaptation:** Select he adaptation configured in **Section 6.4**
- **Location:** Select the PriMega location name from **Section 6.3**.
- **Time Zone:** Select the applicable time zone.
- **SIP Link Monitoring:** Select "Link Monitoring Disabled" from the drop-down menu.



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:** Select "UDP" and "TCP".
- **Port:** Enter "5060".
- **SIP Entity 2:** The PriMega entity name configured in the beginning of this section.
- **Port:** Enter "5060".
- **Connection Policy:** Select "trusted".



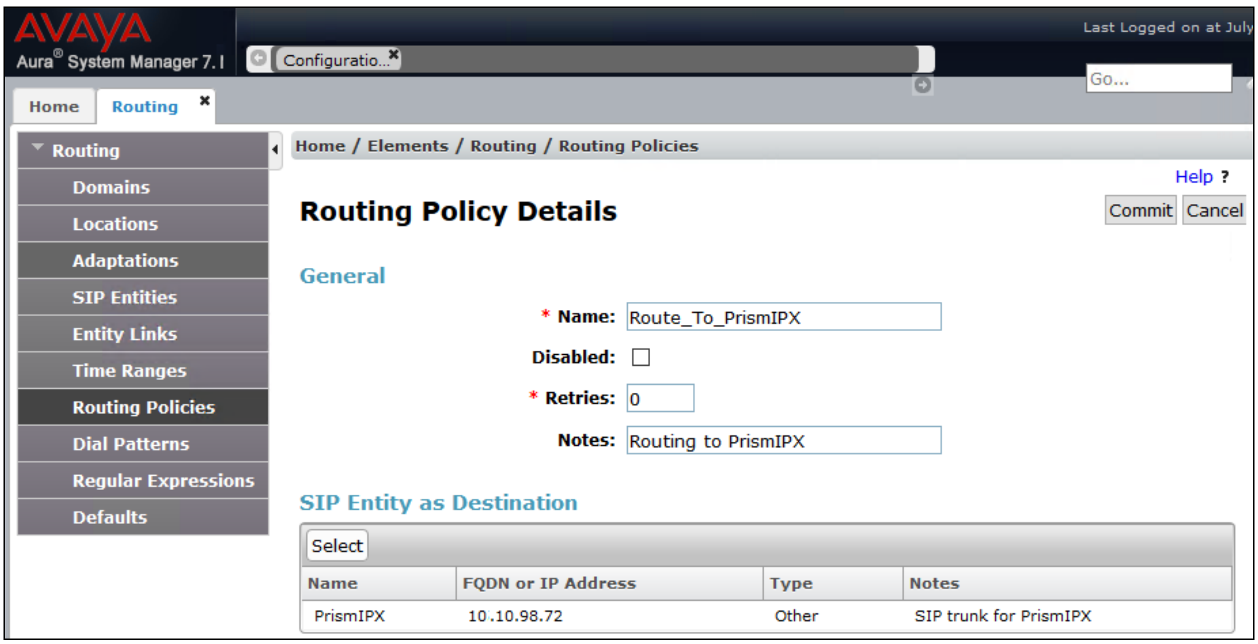
6.6. Administer Routing Policies

A new routing policy is to be added for calls to reach PriMega from the 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 PriMega.

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the PriMega entity name from **Section 6.5**. The screen below shows the result of the selection.

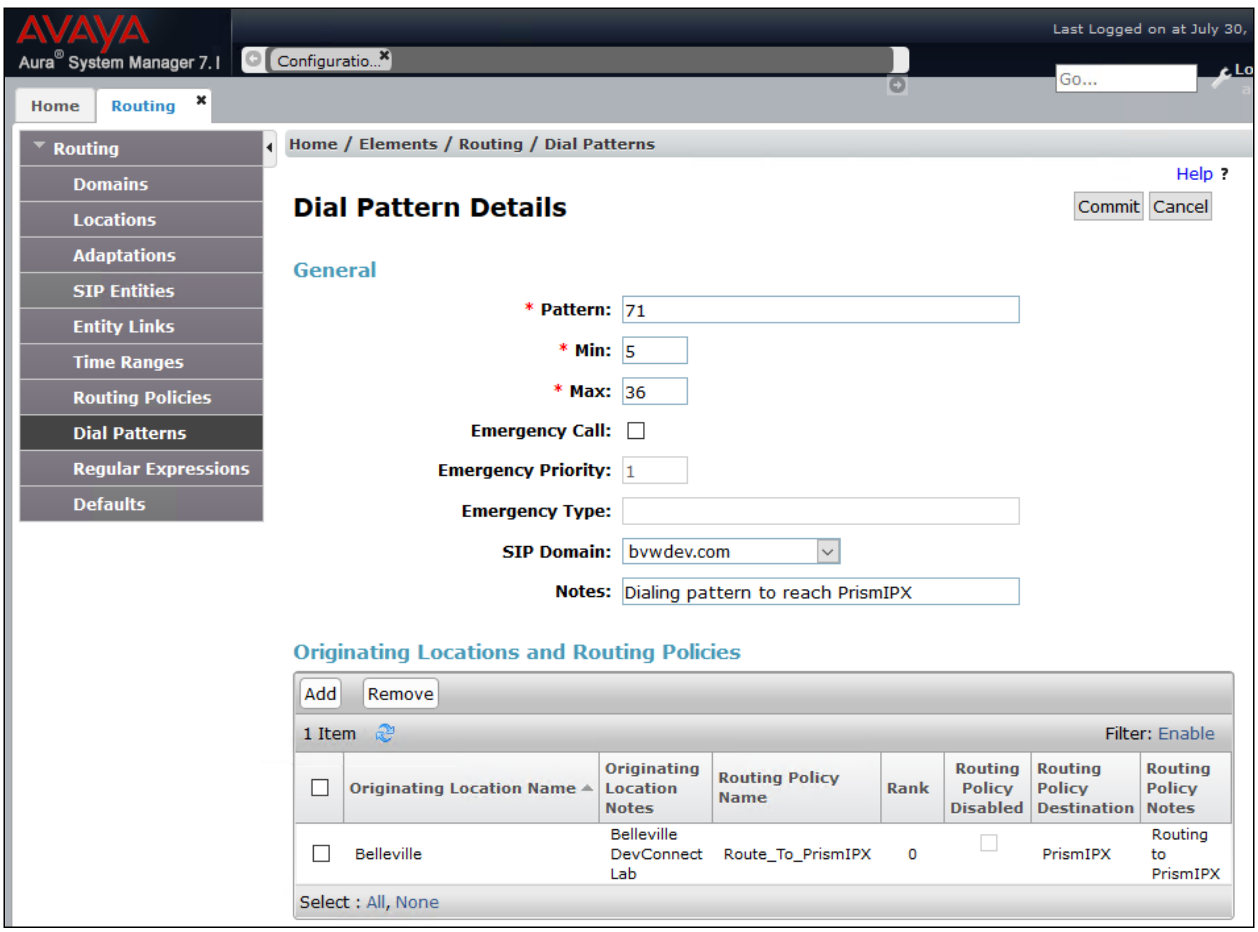


6.7. Administer Dial Patterns

Add a new dial pattern for PriMega by navigating to **Routing → Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach PriMega. 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 signaling group domain name from **Section 5.33**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching PriMega. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations "Belleville" is selected under **Originating Location Name**. The PriMega routing policy from **Section 6.6** was selected under **Routing Policy Name** as shown below.



7. Configure PriMega Messaging Gateway

PriMega Messaging Gateway is typically configured for customers by Prism-IPX Systems. For details on how to configure PriMega Messaging Gateway, contact Prism-IPX Systems by referring to **Section 2.3**. This section provides a "snapshot" of PriMega Messaging Gateway configuration used during this compliance testing. The screen shots and partial configuration shown below, supplied by Prism-IPX Systems, are provided only for reference. It does not show how to configure pagers or pager output. The PriMega software must be installed with the pika add on.

7.1. Confirm Pika configuration

Edit **/etc/pika/pikagp.cfg** and ensure IP address matches the IP that is going to be used on the server.

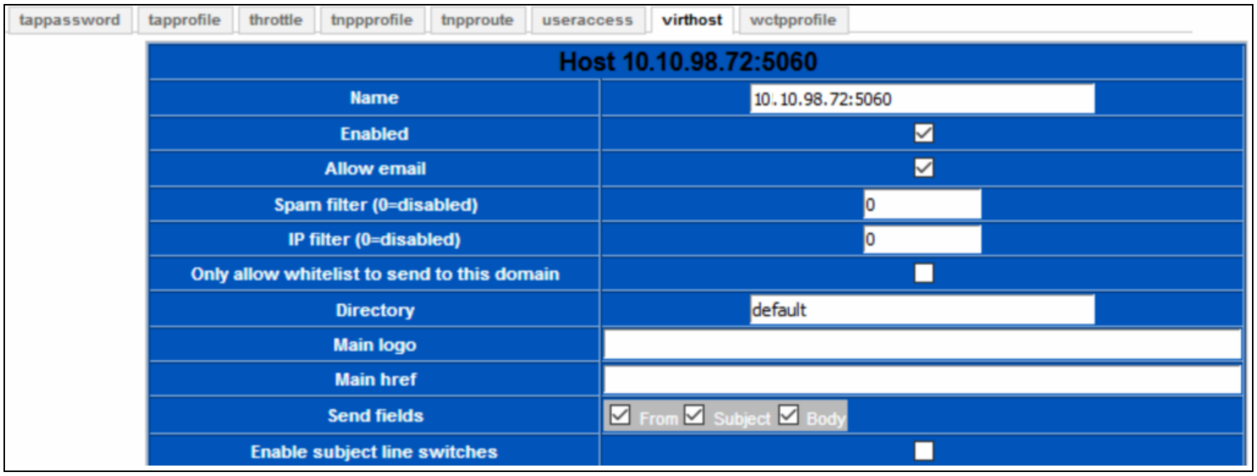
```
......

[SIP_0]
interface=10.10.98.72
ua_port=5060
server_port=0
transport=udp
domain=10.10.98.72
channels=4
audio.portrange=24000-24100
srtp_mode=disabled
.........
[USER_0]
user_name=GP_SIP_USER
domain=10.10.98.72
display_name=GP_SIP_USER
.........
```

and confirm codecs:

```
……..
codecs=g711u|g711a
……….
```

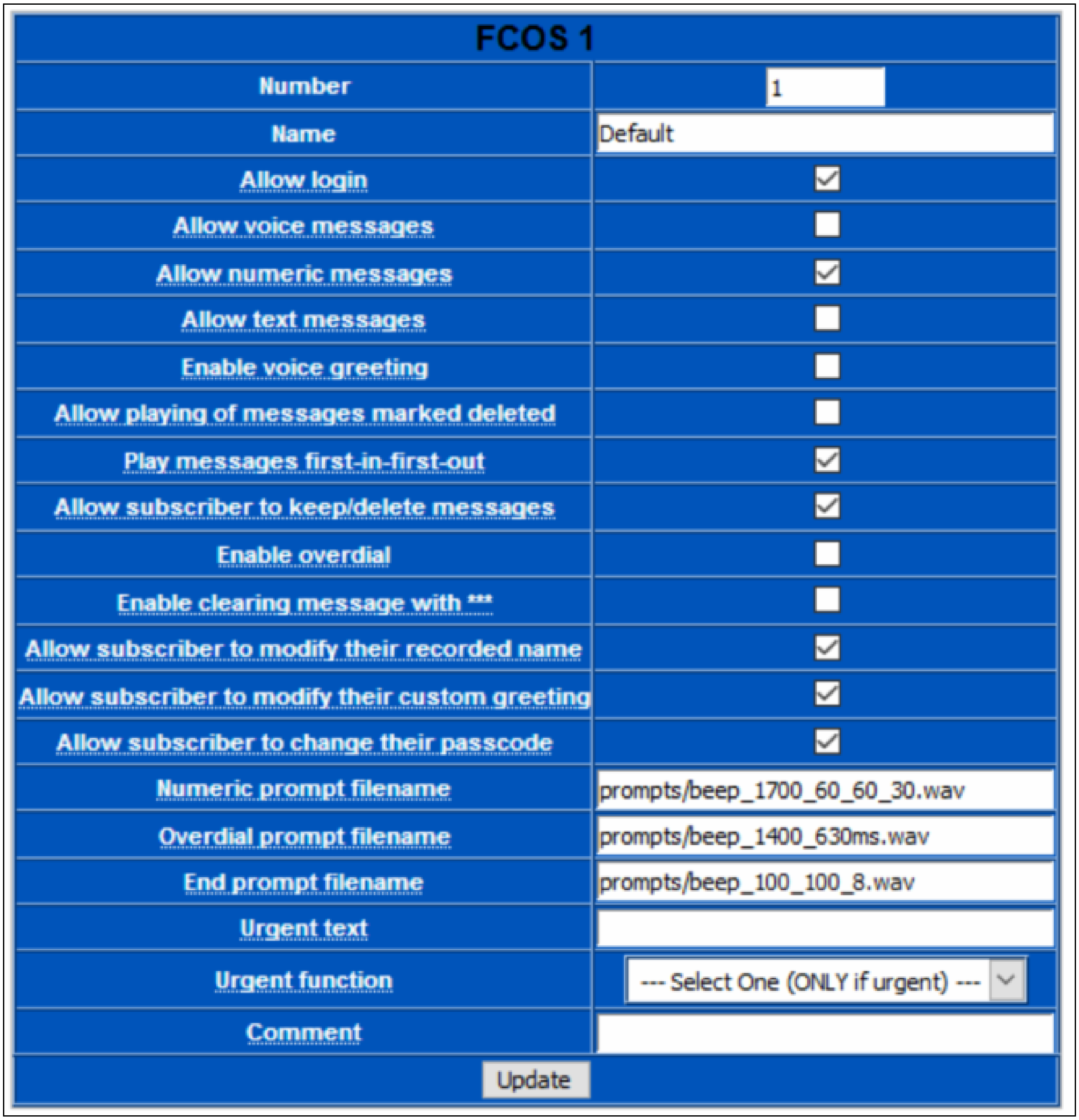
7.2. Set up virthost entry in Primega

Set up a **virthost** entry for **5060** on the IP to which the Communication Manager is pointing (all fields default except Name and Host).



7.3. Change default FCOS setting

This section shows the **FCOS1** setting used during the compliance testing.

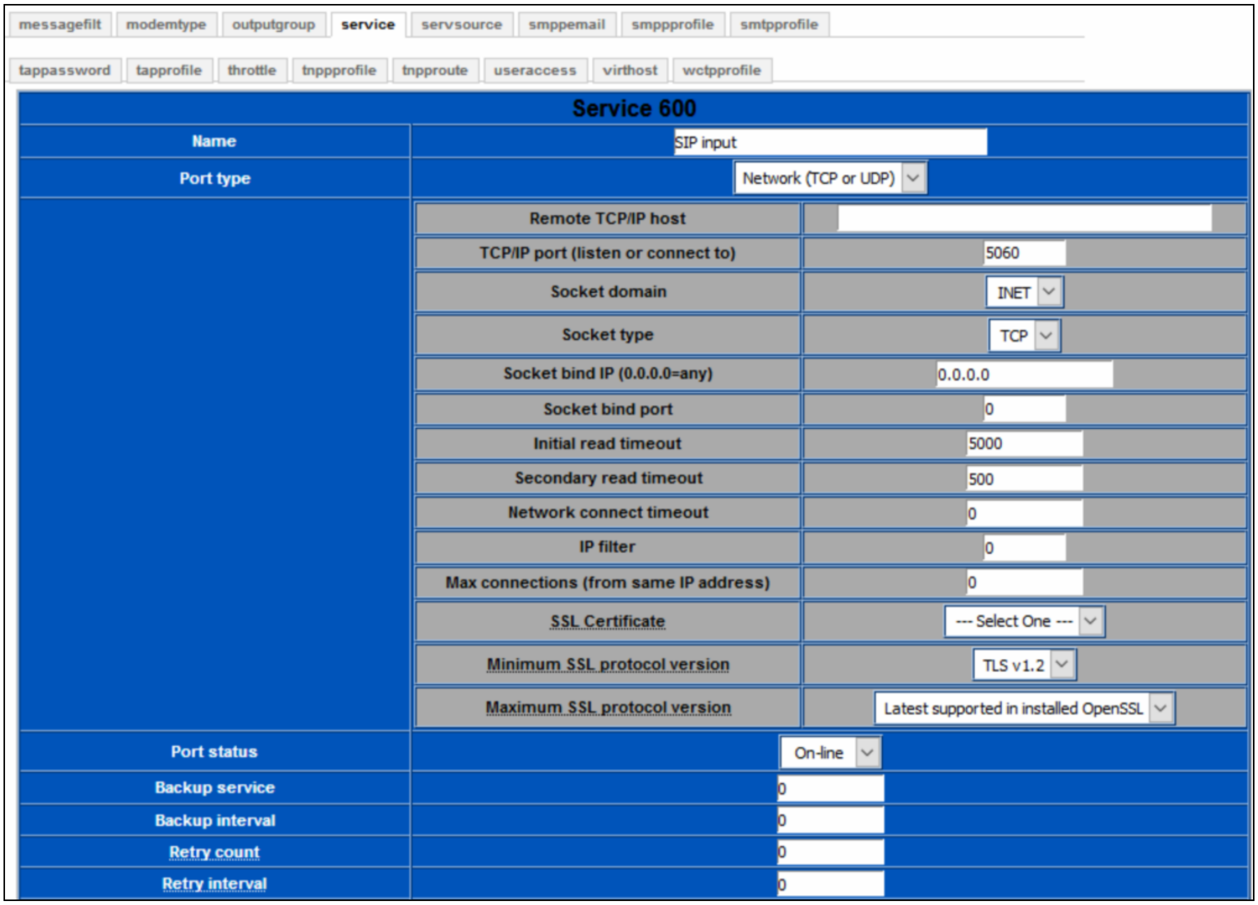


FCOS 1	
Number	1
Name	Default
Allow login	✓
Allow voice messages	
Allow numeric messages	✓
Allow text messages	
Enable voice greeting	
Allow playing of messages marked deleted	
Play messages first-in-first-out	✓
Allow subscriber to keep/delete messages	✓
Enable overdial	
Enable clearing message with ***	
Allow subscriber to modify their recorded name	✓
Allow subscriber to modify their custom greeting	✓
Allow subscriber to change their passcode	✓
Numeric prompt filename	prompts/beep_1700_60_60_30.wav
Overdial prompt filename	prompts/beep_1400_630ms.wav
End prompt filename	prompts/beep_100_100_8.wav
Urgent text	
Urgent function	--- Select One (ONLY if urgent) ---
Comment	
Update	

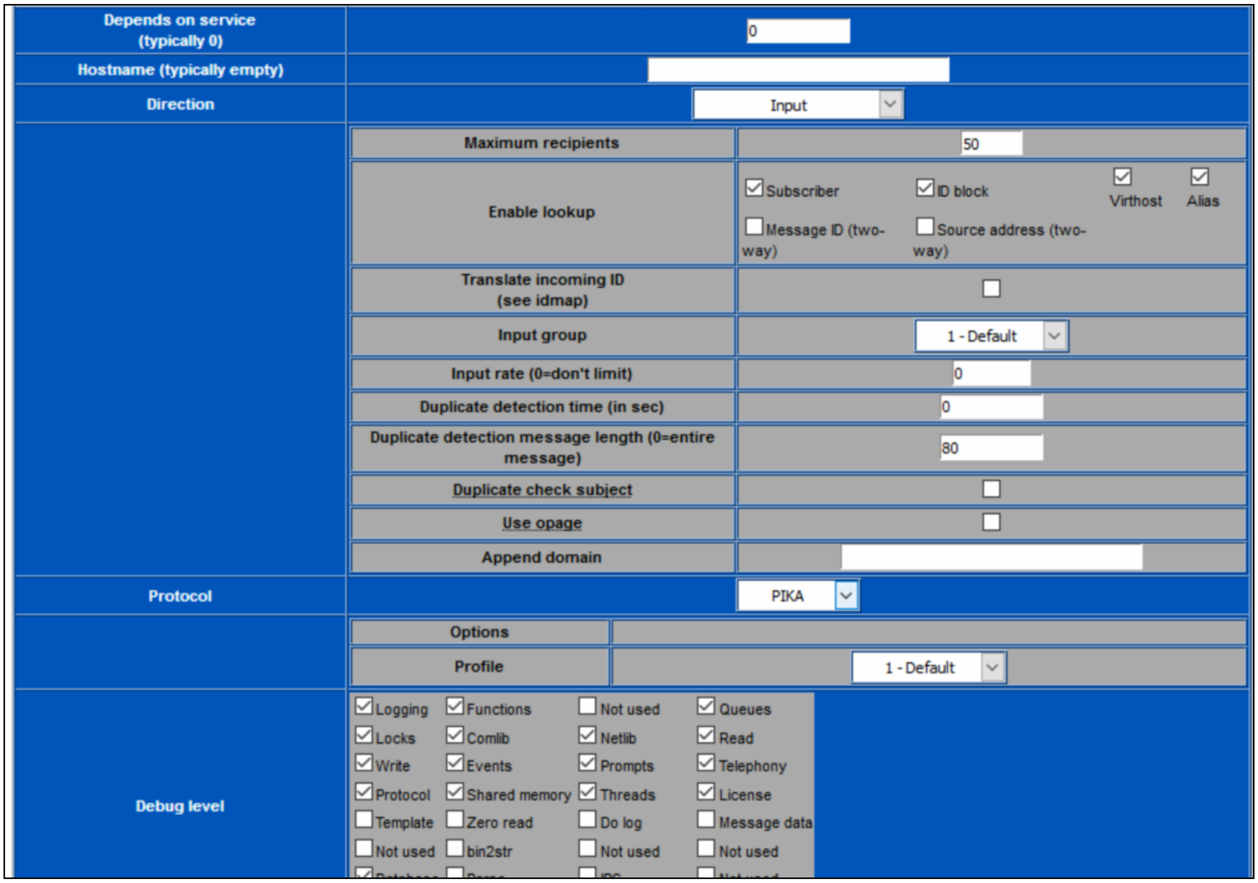
7.4. Create a SIP Input Service

This section shows the three screen shots of the SIP Input Service configured during compliance testing.

Screen below shows the configuration of the input port of 5060.



Screen below shows the configuration of the input section and protocol details.

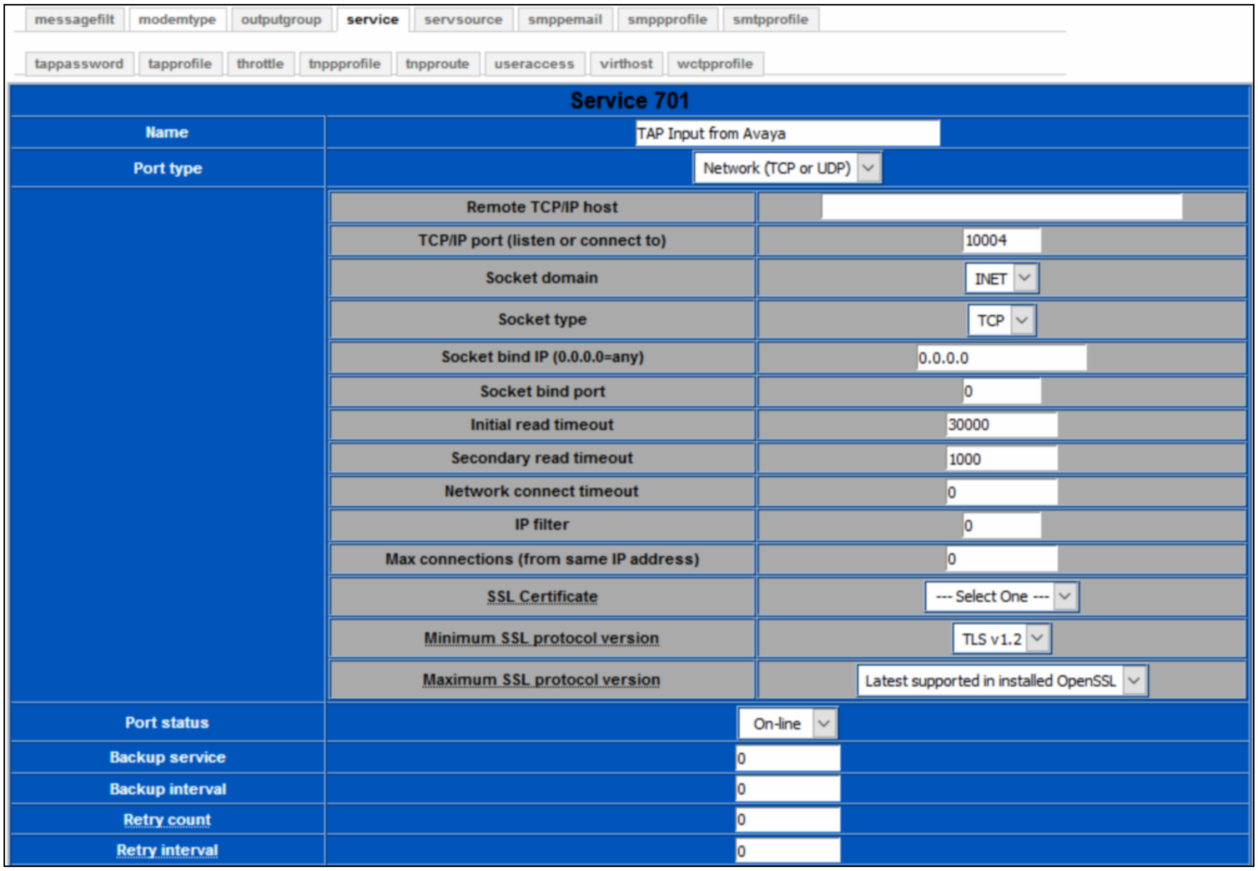


Screen below shows the configuration of the debug level, send fields and logging.

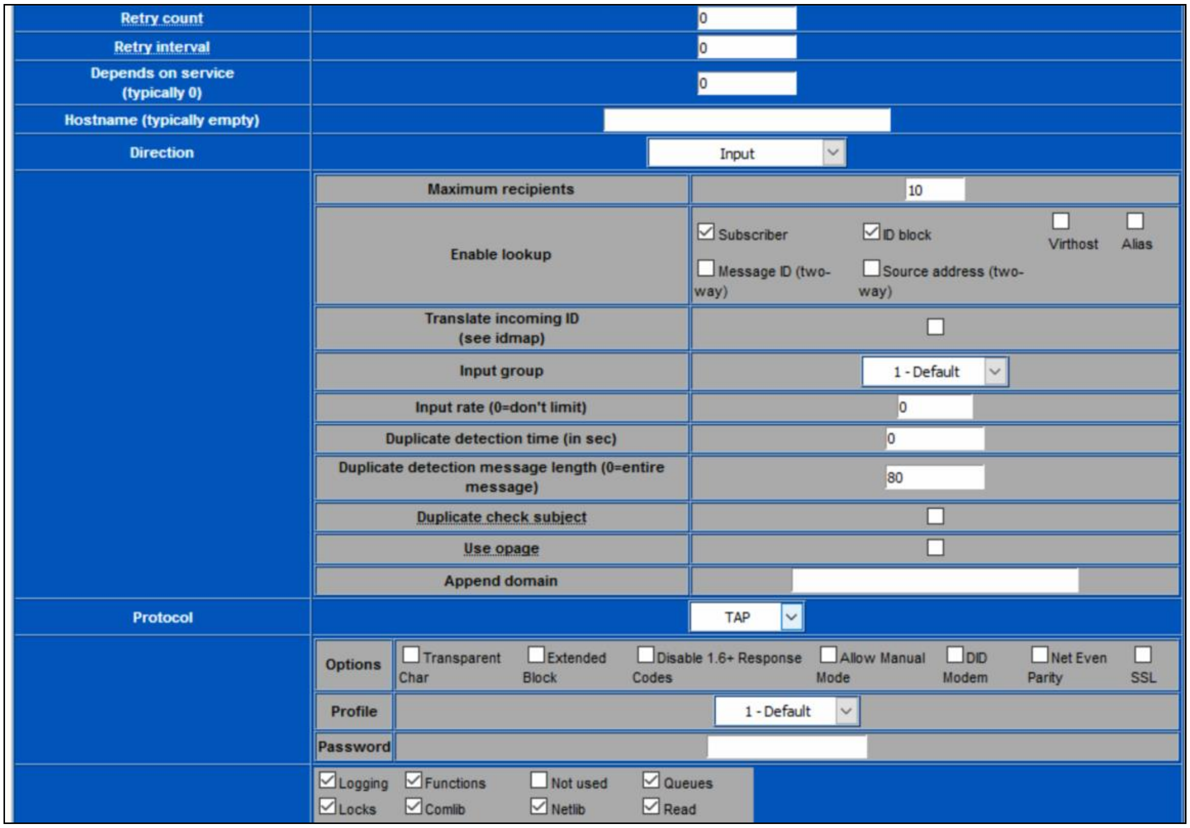
7.5. Create TAP Input Services (10004 and 20004)

This section explains the two TAP input services ports 10004 and 20004 created during compliance testing. Example below only shows configuration for 10004. Similarly, port 20004 can be created.

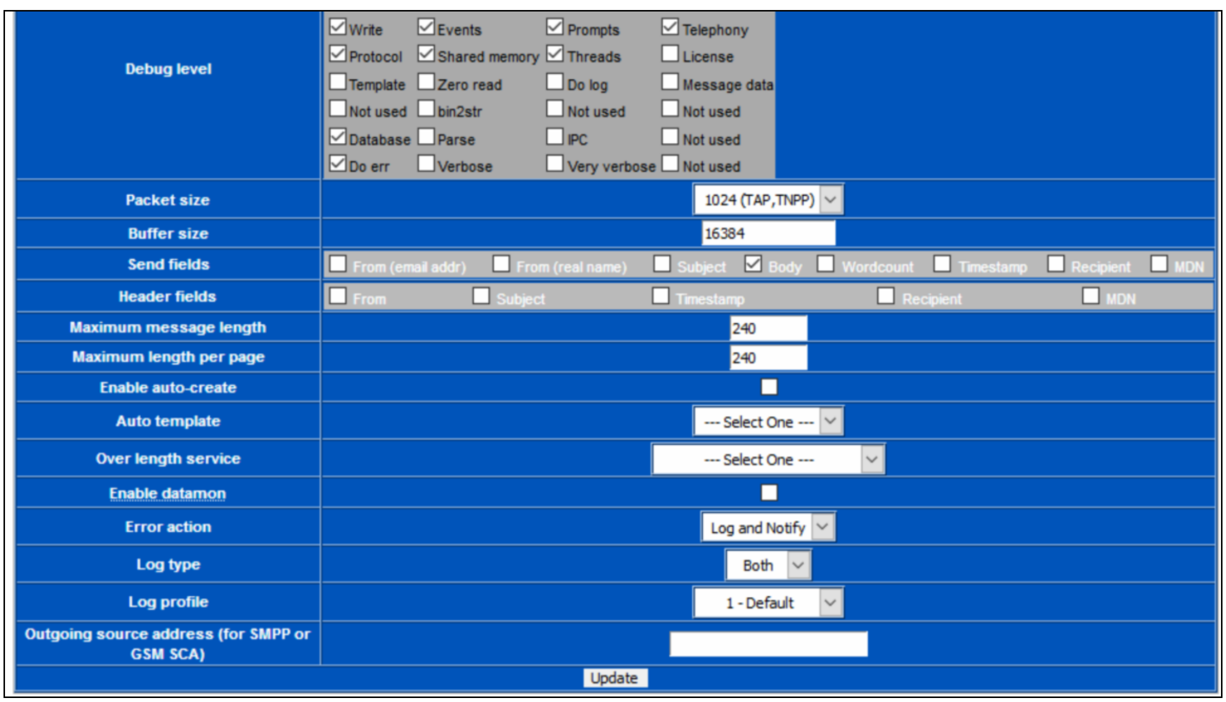
Screen below shows the configuration of the input port.



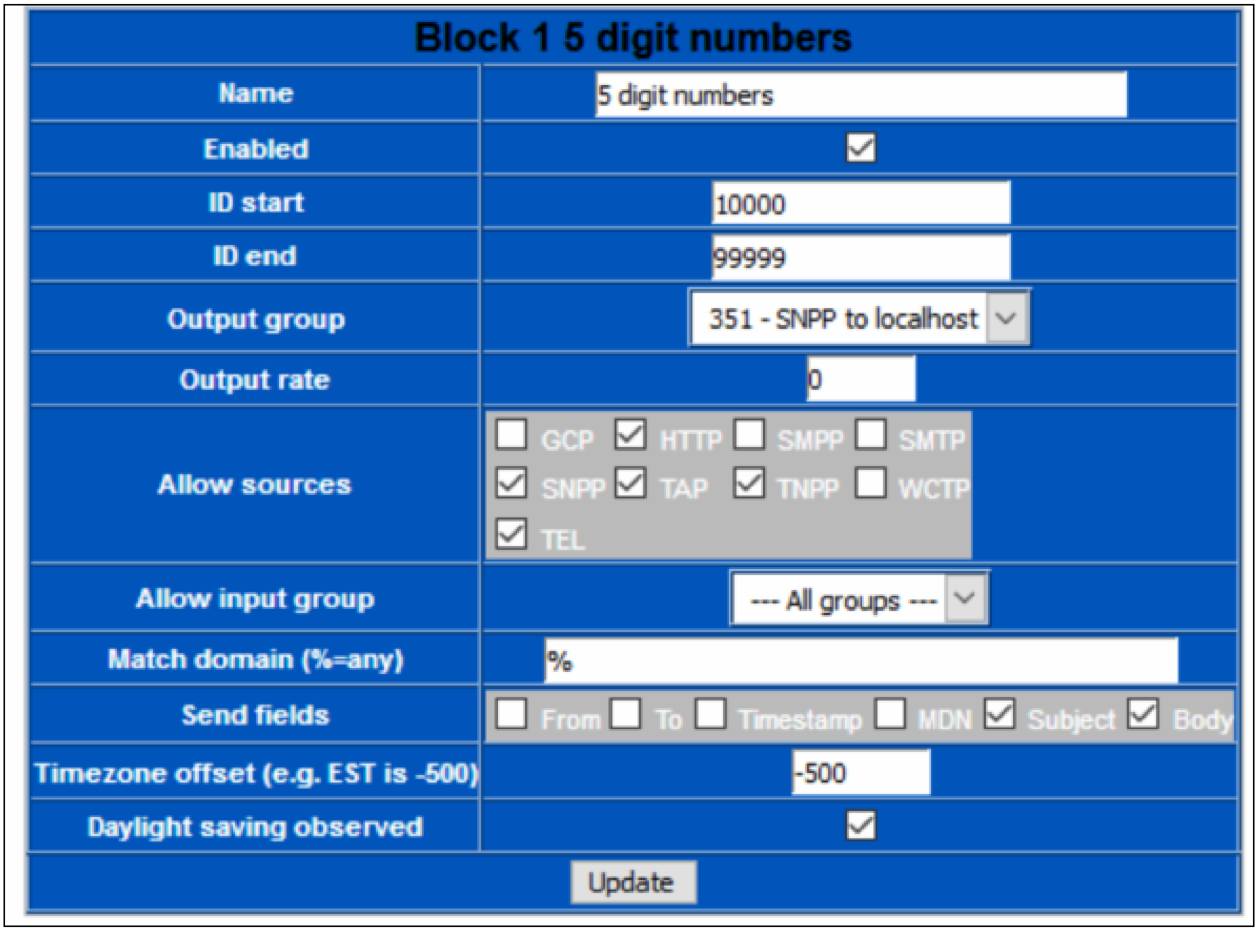
Screen below shows the configuration of the input section and protocol details.



Screen below shows the configuration of the debug level, send fields and logging.



Screen below shows the configuration of valid subscribers (that accept SIP and TAP input) with pagers or use an ID Block.



Note: *Subscribers and ID Blocks will need valid destinations and services to which to send the messages, with the connection(s) to the destination(s) working. Failure to do so will result in failure status being returned to Communication Manager.*

8. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and PriMega solution.

8.1. Verify Avaya Aura® Communication Manager Page-Link State

The following step can ensure that the communication between Communication Manager paging adjunct and PriMega is functioning correctly. Using SAT, connect to Communication Manager and check the page link status with PriMega by using the command "status page-link". Verify that the **Status** of the CTI link is **up** as shown below.

```
status page-link
                                PAGE LINK STATUS


        PAGE1 Link Status: up

        PAGE2 Link Status: up
```

8.2. Verify Page-Link Data

The following step can ensure that the data is being sent and received over the page links. To accomplish this, initiate a call to the paging station mentioned in **Section 5.14** and run the command "list trace page-links". Screen below shows the page being sent and received. It also shows the sent and receive of the heartbeat message.

```
list trace page-links                                                  Page   1

                                    LIST TRACE

time            data

09:07:25 TRACE STARTED 08/01/2018 CM Release String cold-01.0.532.0-24515
09:07:36 PAGE2 Sent: <STX>56503<CR><CR><ETX>122<CR>
09:07:37 PAGE2 Received: <ACK><CR>
09:07:43 PAGE1 Sent: Heartbeat Message
09:07:43 PAGE1 Received: Heartbeat Message
09:07:43 PAGE1 Received: PAGE EXCHANGE DISCONNECT<CR><EOT><CR>
09:07:45 PAGE1 Sent: <CR>
09:07:47 PAGE1 Sent: <CR>
09:07:47 PAGE1 Received: ID=<CR><LF>
09:07:47 PAGE1 Sent: <ESC>PG1<CR>
09:07:47 PAGE1 Received: <ACK><CR>
09:07:47 PAGE1 Received:  <ESC>[p<CR>
09:08:29 PAGE1 Sent: <STX>56503<CR><CR><ETX>122<CR>
09:08:30 PAGE1 Received: <ACK><CR>
09:08:43 PAGE2 Sent: Heartbeat Message
```

8.3. Verify PriMega Logs

The screen below shows excerpts of logs from PriMega when a paging call is made and when the paging station is called.

```
07/19 09:51:18 Received connection from 127.0.0.1
rcvd 'CALL 56204@10.10.97.228'
sent '250 OK'
rcvd 'PAGE 71072'
sent '250 Pager ID accepted'
rcvd 'MESS 1234567890'
sent '250 Message OK'
rcvd 'SEND'
sent '250 Message sent OK'
rcvd 'QUIT'
07/19 11:43:51 Received connection from 127.0.0.1
rcvd 'CALL 15149626014@10.10.97.228'
sent '250 OK'
rcvd 'PAGE 71071'
sent '250 Pager ID accepted'
rcvd 'MESS 1234567890'
sent '250 Message OK'
rcvd 'SEND'
sent '250 Message sent OK'
rcvd 'QUIT'

12:17:27.380 recv returned 13
12:17:27.380 NetRead(fd=11,rsin=): read 9 '[02]56503[0D][0D][03]'
12:17:27.380 in
NetRead(fd=11,max=4,init=99999999,sec=500,btr=4,rt=1,option=0x00000000,rsin= terms=)
12:17:27.380 NetRead(fd=11,rsin=): read 4 '122[0D]'
12:17:27.380 in process_tap(id=56503,data=)
12:17:27.380 in parse_emailaddr(buff=56503,maxname=0,maxaddr=128,maxdomain=80)
12:17:27.380 parse_emailaddr returning name='(null)' addr='56503' domain=''
12:17:27.380 in
check_id(id=56503,domain=,callpass=*****),tnppdest=,source=0x00000020,lookup=0x0000000
3,flags=0x00000001)
12:17:27.380 in sql_run_buff_vp(sqlbuff=select * from subscriber where subscriberid =
'56503')
12:17:27.382 checking idblock 1 name='5 digit numbers' domain='%' allowingroup=0
allowsource=0x00000272 (inputsource=0x00000020)
12:17:27.382 have matching idblock 1 name='5 digit numbers' domain='%'
allowsource=0x00000272
12:17:27.382 in sql_run_buff_vp(sqlbuff=select * from outputgroup where groupnum =
351)
12:17:27.383 leaving check_id(found_flags=0x00000000) retval=0
12:17:27.383 in sql_vp_copy(srcvp=           (nil))
12:17:27.383 in sql_vp_copy(srcvp=0x007f481c02ef00)
12:17:27.383 in
send_egroup(subvp=(nil),source=0x00000020,max_recip=10,smpacket=0x007f481c03de70,reqin
dex=1)
12:17:27.383 in
send_notification(smpacket=0x007f481c03de70,useraccess=(nil),service=0,reqindex=0,prot
ocol=tap,from=,to=56503,subj=,cid=,bin=N,vce=N,msg=)
12:17:27.383 in insert_message_record(protocol=TAP,messageid=,capcode=)
12:17:27.383 in NetGetConnInfo(ti=,si=,TCP,server=Y)
12:17:27.383 in NetGetHostDomain(hostmax=81,domainmax=65)
12:17:27.383 NetGetHostDomain: (host='PIKA_Test' domain='')
```

9. Conclusion

A full and comprehensive set of feature and functional test cases were performed during Compliance Testing. PriMega Messaging Gateway is considered compliant with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All test cases have passed and met all the objectives.

10. Additional References

These documents form part of the Avaya official technical reference documentation suite. Further information may be had from *http://support.avaya.com* or from the local Avaya representative.

1. *Deploying Avaya Aura® Session Manager*, Release 7.1.3. Issue 5. May 2018.
2. *Administering Avaya Aura® Session Manager*, Release 7.1.3. Issue 5. May 2018.
3. *Deploying Avaya Aura® System Manager*, 7.1.3. Issue 8. July 2018.
4. *Administering Avaya Aura® System Manager for Release 7.1.3*, Release 7.1.3. Issue 15. July 2018.
5. *Deploying Avaya Aura® Communication Manager*, Release 7.1.3. Issue 5. May 2018.
6. *Administering Avaya Aura® Communication Manager*, Release 7.1.3. Issue 7. May 2018.
7. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1.3. Issue 6. May 2018.
8. *MultiVantage® Requirements/Feature Spec: SA8312*. COMPAS ID: 92212 Issue: 1.1. Date: June 19, 2013

Product Documentation for PriMega Messaging Gateway can be obtained in the installed software or at https://wiki.harktech.com:8443/wiki/index.php (this requires registration).

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.