Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Khomp Kmedia 6400 for E1 access - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the Khomp Kmedia 6400 Gateway to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunking along with E1 access to a simulated 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.

Table of Contents

1. Introduction

These Application Notes describe the procedure for configuring the Khomp Kmedia 6400 (Kmedia) to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunking for E1 access to a simulated PSTN.

These Application Notes present a sample configuration for an enterprise network consisting of Avaya Aura® Session Manager and Avaya Aura® Communication Manager, integrated with an Kmedia Gateway using SIP trunk and providing E1 access to a simulated PSTN.

Kmedia is a media gateway carrier grade, for converging applications in digital communication platforms (E1/T1, STM-1 or SIP), replacing several signaling and connectivity devices by a single item of equipment.

With hardware designed to work in heavy traffic environments, the Kmedia has the main protocols for NGN 's (Next Generation Networks) and universal codecs for all the channels, besides high performance and processing capacity of calls per second.

The Kmedia-6400 is expandable up to 64 E1/T1 trunks in only 2 Us, without the use of separate servers for signaling management and processing. Each trunk can be managed for maximum use of its capacity by means of the traffic distribution system, which can comply with criteria pre-established by the user, as prioritizing routes of lower cost and re-route (configuration of the waiting time in the response of the operator ahead), etc. Furthermore, the Kmedia allows the partitioning of calls in all the routes determined by the user, simultaneously.

Offering the highest density of ports and processing of the sector and the lowest operating cost for a media gateway, the Kmedia presents an average energy consumption two thirds lower than other products of similar capacity, besides occupying less space in the Data Center, aiding to reduce rental costs and contributing to reducing the environmental impact.

The Kmedia is revolutionary in the gateways market, bringing a new reality in availability, reliability, flexibility of growth and management, and also reduced physical size.

2. General Test Approach and Test Results

The general test approach was to make calls, verify codecs, and exercise common PBX features, between endpoints located in the enterprise and the simulated PSTN.

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

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability. The feature testing focused on verifying the following:

- Simulated PSTN calls from and to Avaya endpoints
- Calling with various Avaya Deskphone models
- Support for G.711A, G.711MU, G.729, G.729AB, G722 and G.726 codecs
- SIP transport using TCP
- Codec negotiation
- Telephony supplementary features, such as Hold, Call Transfer, Conference Calling and Call Forwarding
- DTMF Tone Support
- Voicemail Coverage and Retrieval
- Direct IP-to-IP Media (also known as “Shuffling”) over SIP Trunk. Direct IP-to-IP media allows compatible phones to reconfigure the RTP path after call establishment directly between the Avaya phones and the Kmedia Gateway and release media processing resources on the Avaya Media Gateway

2.2. Test Results

The Kmedia passed compliance testing.

2.3. Support

For technical support, contact Kmedia via www.khomp.com.

3. Reference Configuration

As shown in **Figure 1**, the Avaya enterprise network uses SIP trunking for call signaling internally, and with the Kmedia Gateway in order to access the simulated PSTN. The Kmedia is managed by using the web interface. Session Manager, with its SM-100 (Security Module) network interface, routes calls between the different entities using SIP Trunks. All inter-system calls are carried over these SIP trunks. Session Manager supports flexible inter-system call routing based on the dialed number, the calling number and the system location; it can also provide protocol adaptation to allow multi-vendor systems to interoperate. Session Manager is managed by Avaya Aura® System Manager via the management network interface.

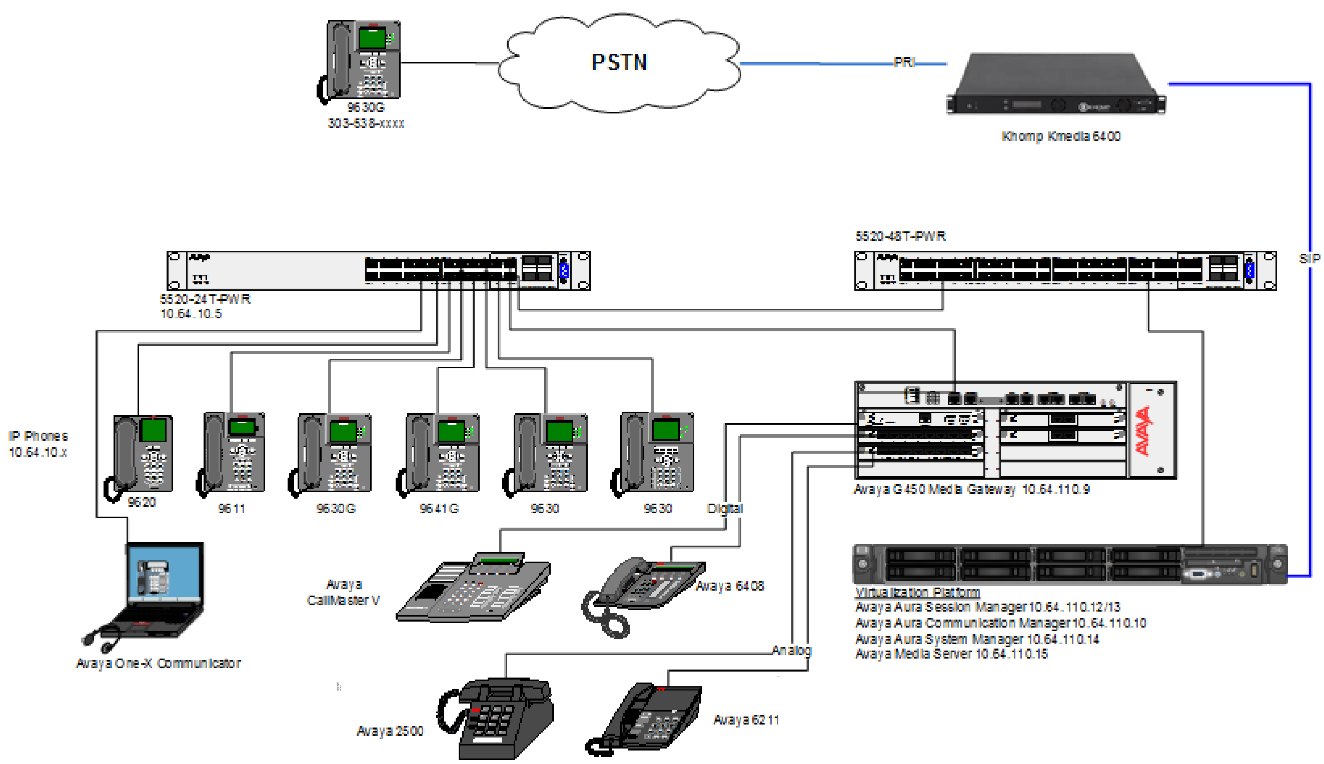


Figure 1: Compliance Test Reference Configuration

For the sample configuration shown in **Figure 1**, Session Manager, System Manager, Communication Manager, and Media Server all run in a virtual environment . These Application Notes focus on the configuration of the SIP trunks and call routing.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in a Virtual Environment	7.0 SP3
Avaya Aura® Session Manager in a Virtual Environment	7.0 SP2
Avaya Aura® System Manager in a Virtual Environment	7.0 SP2
Avaya Aura® Media Server in a Virtual Environment	7.7.0.226
Avaya 96x1 Deskphone	SIP 7.0, H.323 6.6
Avaya 6211 and 6221 Analog Phone	-
Khomp Kmedia	2.9.47

5. Configure Avaya Aura® Communication Manager

This section shows the configuration in Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assumed that the basic configuration has already been administered. For further information on Communication Manager, please consult with **Reference [1]**. The procedures include the following areas:

- Verify Communication Manager License
- Administer System Parameters Features
- Administer IP Node Names
- Administer IP Network Region and Codec set
- Administer SIP Signaling Group and Trunk Group
- Administer Route Pattern
- Administer Private Numbering
- Administer Dial Plan and AAR analysis
- Administer ARS analysis
- Administer Feature Access Codes
- Save Changes

5.1. Verify Avaya Aura® Communication Manager License

Use the **display system-parameter customer options** command to verify whether the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

Note: The license file installed on the system controls the maximum features permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  20
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  15
                       Maximum Administered SIP Trunks: 4000  10
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

5.2. Administer System Parameters Features

Use the **change system-parameters features** command to allow for trunk-to-trunk transfers. This feature is needed to allow for transferring an incoming/outgoing call from/to a remote switch back out to the same or different switch. For simplicity, the **Trunk-to-Trunk Transfer** field was set to **all** to enable all trunk-to-trunk transfers on a system wide basis.

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
                 Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                         Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? all
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n




             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? y
```

5.3. Administer IP Node Names

Use the **change node-names ip** command to add entries for Communication Manager and Session Manager that will be used for connectivity. In the sample network, the processor Ethernet interface **procr** and **10.64.110.10** are entered as **Name** and **IP Address** for the signaling in Communication Manager running in a virtual environment. In addition, **asm** and **10.64.110.13** are entered for Session Manager.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
acms                 10.64.110.18
aes                  10.64.110.15
ams                  10.64.110.16
asm                  10.64.110.13
procr                10.64.110.10
procr6               ::
```

5.4. Administer IP Network Region and Codec Set

Use the **change ip-network-region n** command, where **n** is the network region number, to configure the network region being used. In the sample network ip-network-region **1** is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **1** to use ip-codec-set 1.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Main                    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: 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
```

Use the **change ip-codec-set n** command where **n** is codec set used in the configuration. The codecs used in the compliance test are shown here. Configure the IP Codec Set as shown in the screen below.

Retain the default values for the remaining fields.

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

                          IP CODEC SET
    Codec Set: 1

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.711MU            n         2        20
 2: G.711A             n         2        20
 3: G.729AB            n         2        20
 4:
 5:
 6:
 7:

     Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: none
 2:
 3:
 4:
 5:
```

5.5. Administer SIP Trunks with Avaya Aura® Session Manager

In the test configuration, a SIP trunk was configured between Communication Manager and Session Manager for enterprise calling between Communication Manager and Session Manager registered endpoints. Additionally a SIP trunk was configured between Session Manager and the Kmedia in order to communicate between the enterprise and the simulated PSTN. To administer a SIP Trunk on Communication Manager, two steps are required: the creation of a signaling group and a trunk group.

5.5.1. Add SIP Signaling Group

Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tcp**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 5.3** i.e., **asm**
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Network Region:** **1**
- **DTMF over IP:** **rtp-payload**
- **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? n
  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? y
   Near-end Node Name: procr                  Far-end Node Name: asm
 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? 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
```

5.5.2. Add Trunk Group

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e.**, asm**)
- **TAC:** An available trunk access code (i.e., **101**)
- **Service Type:** **public-ntwrk**
- **Signaling Group:** The number of the signaling group associated (i.e., **1**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager** (must be within the limits of the total trunks available from license verified in **Section 5.1**)

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: asm                         COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

Navigate to **Page 3** and change **Numbering Format** to **public**. Use default values for all other fields.

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? Y
                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.6. Configure Route Patterns

Configure route patterns to correspond to the newly added SIP trunk group. Use the **change route pattern n** command, where **n** is an available route pattern.

The route pattern, as shown below, was configured to route calls to Session Manager and simulated PSTN endpoints.

5.6.1. Route Pattern for reaching Session Manager and Simulated PSTN Endpoints

When changing the route pattern, enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name (i.e., **asm**)
- **Grp No:** The trunk group number from **Section 5.5.2**
- **FRL:** Enter a level that allows access to this trunk, with **0** being least restrictive
- **No. Del Dgts:** **0** was entered to delete zero digits

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

5.7. Administer Public Numbering

Use the **change public-numbering** command to define the calling party number to be sent out through the SIP trunk. In the sample network configuration below, all calls originating from a **5**-digit extension (**Ext Len**) beginning with **1** (**Ext Code**) and routed through any trunk will result in a **5**-digit calling number (**Total Len**). The calling party number will be in the SIP "From" header.

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN           CPN
Len Code            Grp(s)    Prefix        Len
                                                  Total Administered: 1
 5  1                                       5        Maximum Entries: 240

                                                  Note: If an entry applies to
                                                  a SIP connection to Avaya
                                                  Aura(R) Session Manager,
                                                  the resulting number must
                                                  be a complete E.164 number.

                                                  Communication Manager
                                                  automatically inserts
                                                  a '+' digit in this case.
```

5.8. Administer Dial Plan and AAR analysis

Configure the dial plan for dialing 5-digit extensions beginning with **111** to stations registered with Session Manager.

Use the **change aar analysis n** command, where **n** is the dial string pattern to configure an **aar** entry for **Dialed String 111** (Extensions on Session Manager) to use **Route Pattern 1** (defined in **Section 5.6**). The **Call Type** was set to **lev0**.

```
change aar analysis 111                                         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
    111                   5    5     1        lev0          n
```

5.9. Administer ARS Analysis

This section provides sample Auto Route Selection (ARS) used for routing calls with dialed digits beginning with **1** which correspond to numbers accessible via the Kmedia. Use the **change ars analysis 1** command and add an entry to specify how to route calls. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Dialed String:** Dialed digits to match on
- **Total Min:** Minimum number of digits, in this case **11**
- **Total Max:** Maximum number of digits, in this case **11**
- **Route Pattern:** The route pattern number from **Section 5.6**, i.e., **1**
- **Call Type:** hnpa

Note: The additional entries may be added for different number destinations.

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

          Dialed           Total     Route    Call   Node   ANI
          String          Min  Max  Pattern   Type   Num    Reqd
    1                     11   11    1        natl          n
```

5.10. Administer Feature Access Code

Configure a feature access code to use for AAR and ARS routing. Use the **change feature access code** command to define **Access Code** for **Auto Alternate Routing (AAR)** and for **Auto Route Selection (ARS)**. In the test configuration, **8** and **9** were used respectively.

```
change feature-access-codes                                    Page   1 of  10
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
```

5.11. Save Changes

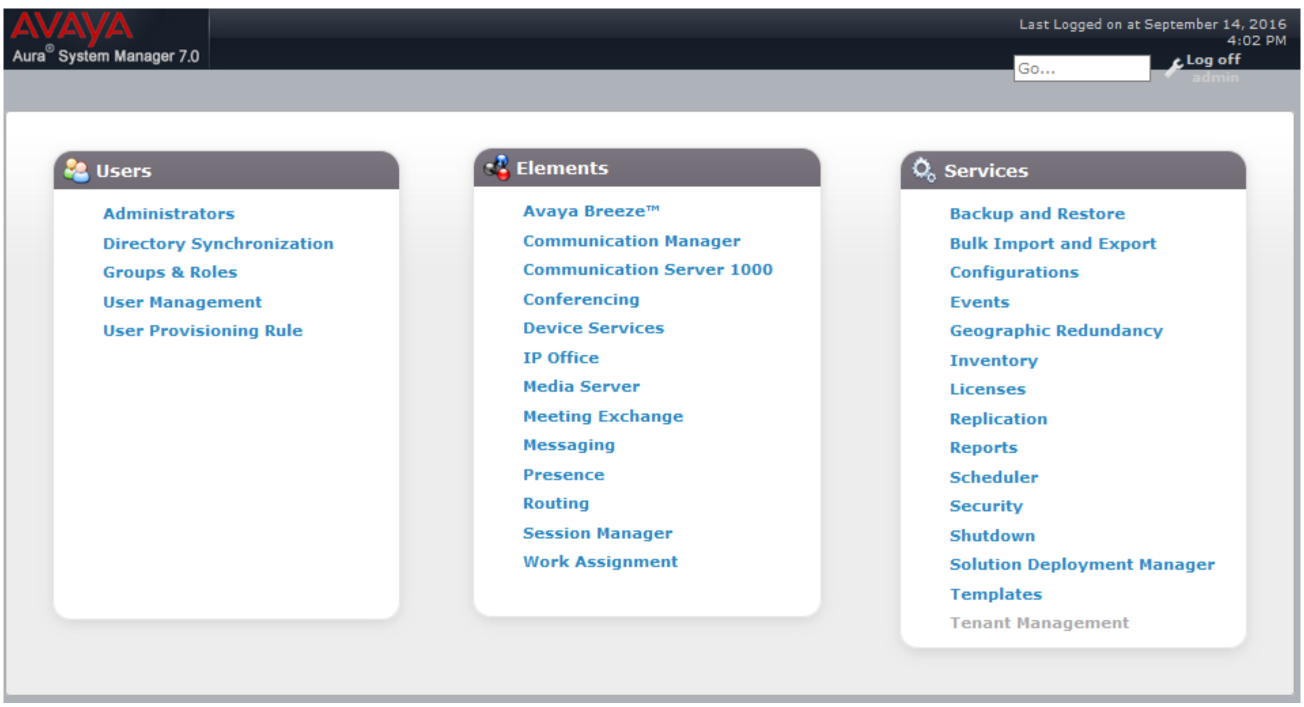
Use the **save translation** command to save all changes.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in **Reference [2]**. The procedures include adding the following items:

- Specify SIP Domain
- Add Locations
- Add Adaptations
- Add SIP Entities and Entity Links
- Add Routing Policies
- Add Dial Patterns
- Add Users for SIP Phones

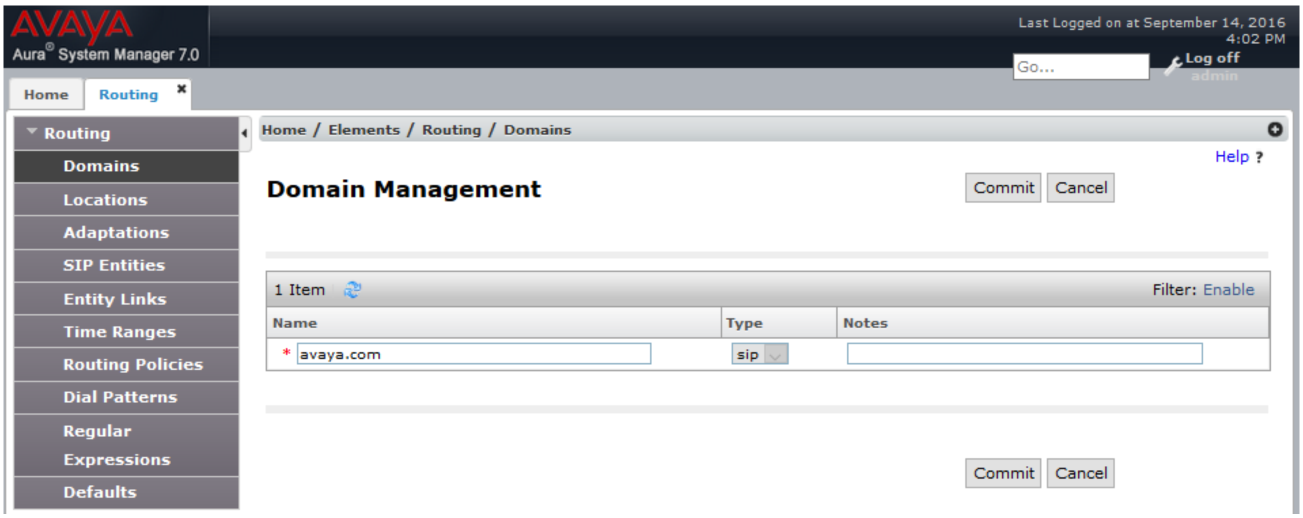
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials. The home screen as shown below is displayed. Expand the **Routing** Link under **Elements**.



6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right (not shown). The following screen will then be shown. Fill in the following fields and click **Commit**.

- **Name:** The authoritative domain name (e.g., **avaya.com**)
- **Type** Select **sip**
- **Notes:** Descriptive text (optional)



6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. A single location is added to the configuration for Communication Manager and the Kmedia Gateway. To add a location, select **Locations** on the left and click on the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

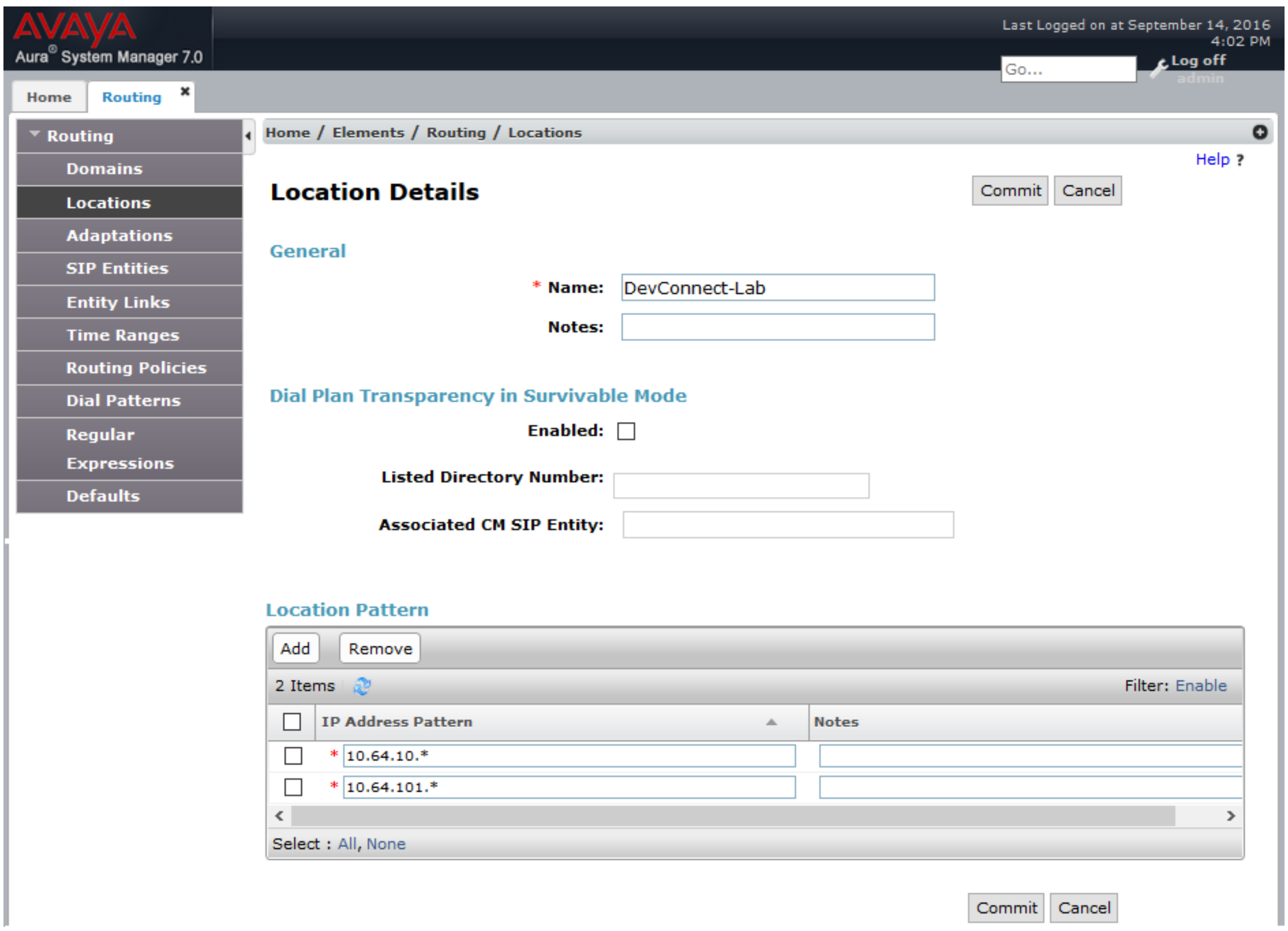
Under **General**:

- **Name:** A descriptive name

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location (optional). In these Application Notes, no pattern was defined.

Defaults can be used for the remaining fields. The screen below shows addition of the **Lab** location, which includes all the components of the compliance test environment. Click **Commit** to save.



6.3. Add Adaptations

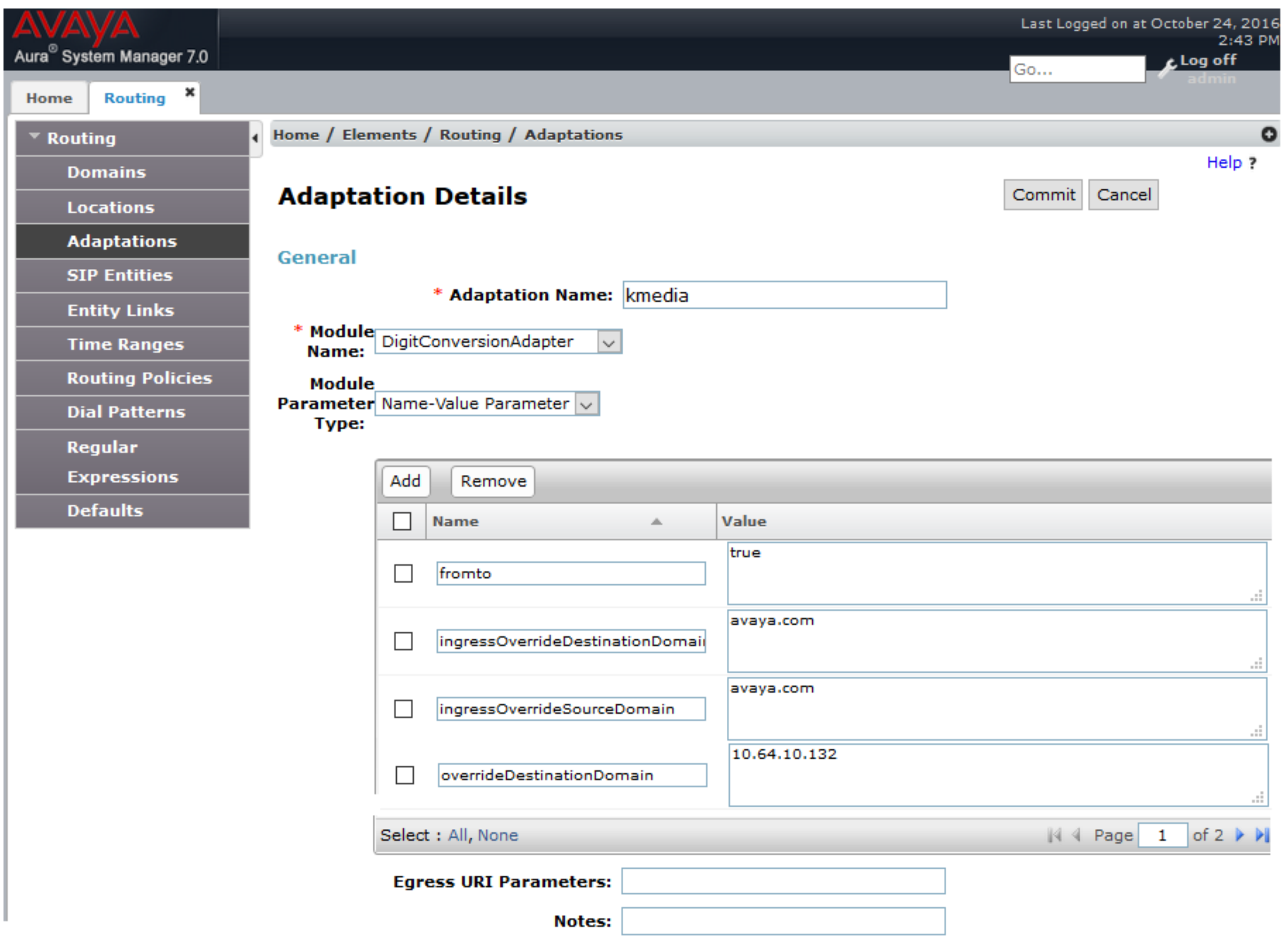
In order to maintain digit manipulation centrally on Session Manager, an adaptation module can be configured with a numbering plan offered from the PSTN Service Provider. To add an adaptation, select **Adaptations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

Under **General**:

- **Adaptation Name:** A descriptive name i.e., **kmedia**
- **Module Name:** From the dropdown list select **DigitConversionAdapter**
- **Module Parameter Type:** Configured as shown in the screen capture below.

The module parameters configured in the Screen capture overrides the domain avaya.com with the IP Address of Kmedia in SIP messages.

The screen below is the Adaptation detail page. Click **Commit** to save the changes.



6.4. Add SIP Entities and SIP Entity Links

A SIP Entity is required for each SIP-based telephony system wishing to communicate with Session Manager for call routing. In the sample configuration, a SIP Entity and SIP Entity Link is added for Communication Manager, and the Kmedia.

6.4.1. Adding Avaya Aura® Communication Manager SIP Entity and SIP Entity Link

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right (not shown).

Under **General**:

- **Name:** A descriptive name, i.e., **acm**
- **FQDN or IP Address:** IP address of the Communication Manager i.e., **10.64.110.10**
- **Type:** Select **CM**
- **Adaptation:** Select CM Adapter
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this entity

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name:** Will be populated automatically
- **SIP Entity 2:** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1:** Select Session Manager from the pull down box
- **Protocol:** Select the desired Protocol from the pull down box
- **Port:** Enter the desired port number for the Entity Link
- **Policy:** Select the appropriate Connection Policy from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for Communication Manager.

SIP Entity Details

Commit | Cancel

General

* Name:	acm
* FQDN or IP Address:	10.64.110.10
Type:	CM
Notes:	
Adaptation:	
Location:	DevConnect-Lab
Time Zone:	America/Denver
* SIP Timer B/F (in seconds):	4
Credential name:	
Securable:	☐
Call Detail Recording:	none

Loop Detection

Loop Detection Mode:	On
Loop Count Threshold:	5
Loop Detection Interval (in msec):	200

SIP Link Monitoring

SIP Link Monitoring:	Use Session Manager Configuration
Supports Call Admission Control:	☐
Shared Bandwidth Manager:	☐
Primary Session Manager Bandwidth Association:	
Backup Session Manager Bandwidth Association:	

Entity Links

Override Port & Transport with DNS SRV: ☐

Add | Remove

1 Item Filter: Enable

☐	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
☐	* asm_acm_5061_TLS	asm	TLS	* 5061	acm	* 5061	trusted	☐

Select : All, None

6.4.2. Adding Kmedia Gateway SIP Entity

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right (not shown).

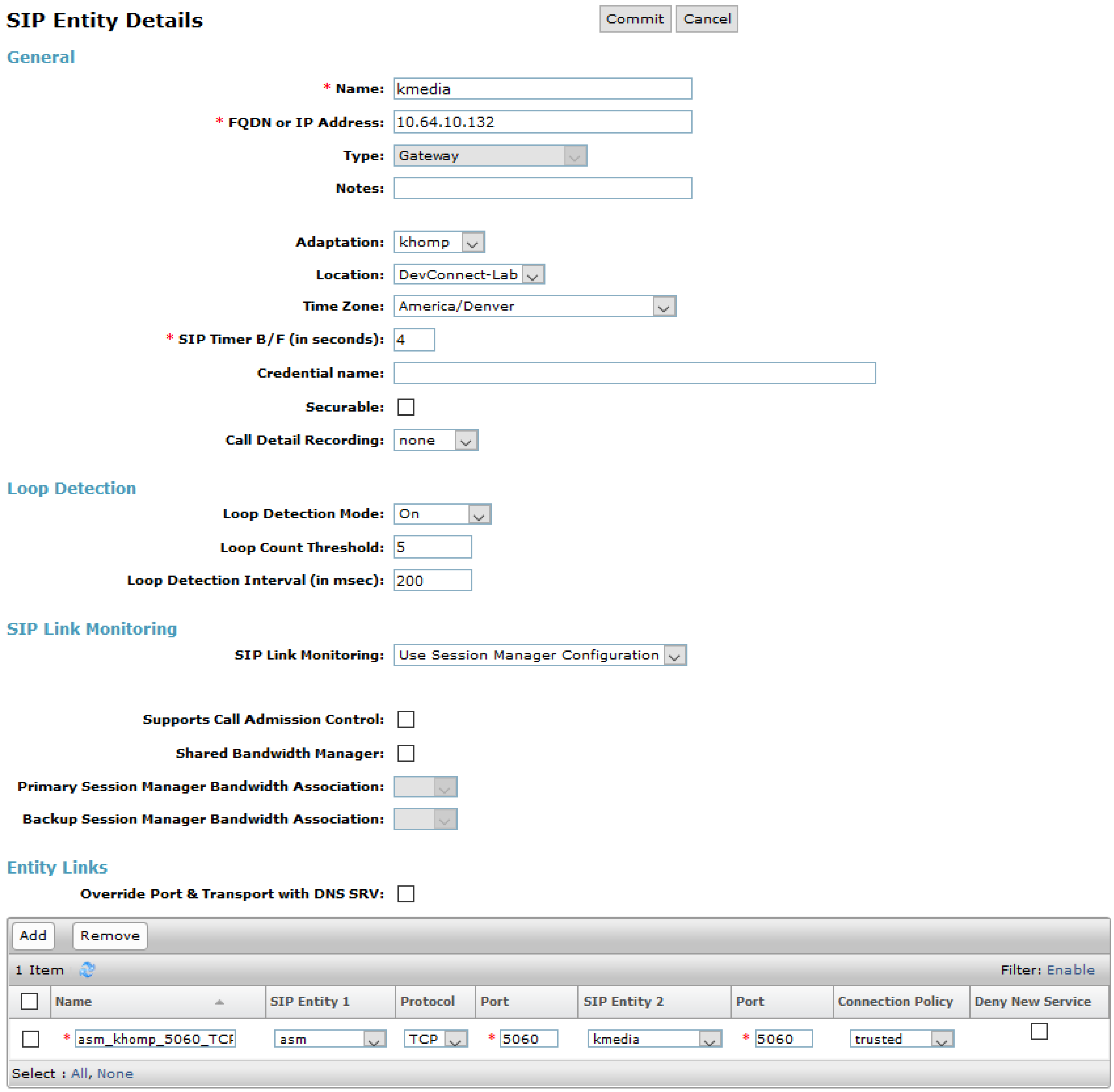
Under **General**:

- **Name:** A descriptive name, i.e., **kmedia**
- **FQDN or IP Address:** IP address of the Kmedia i.e., **10.64.10.132**
- **Type:** Select **Gateway**
- **Adaptation:** Select **kmedia**. This was configured in **Section 6.3**
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this entity
- **SIP Link Monitoring:** **Use Session Manager Configuration**

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name:** Will be populated automatically
- **SIP Entity 2:** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1:** Select Session Manager from the pull down box
- **Protocol:** Select the desired Protocol from the pull down box
- **Port:** Enter the desired port number for the Entity Link
- **Policy:** Select the appropriate Connection Policy from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for Kmedia.


SIP Entity Details
Commit
Cancel
General
* Name: kmedia
* FQDN or IP Address: 10.64.10.132
Type: Gateway
Notes:
Adaptation: khomp
Location: DevConnect-Lab
Time Zone: America/Denver
* SIP Timer B/F (in seconds): 4
Credential name:
Securable:
Call Detail Recording: none
Loop Detection
Loop Detection Mode: On
Loop Count Threshold: 5
Loop Detection Interval (in msec): 200
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:
Entity Links
Override Port & Transport with DNS SRV:
Add
Remove
1 Item
Filter: Enable
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
* asm_khomp_5060_TCP
asm
TCP
* 5060
kmedia
* 5060
trusted
Select : All, None

6.5. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 6.4**. A routing policy must be added for Communication Manager and the Kmedia Gateway. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under **General**

- Enter a descriptive **Name**

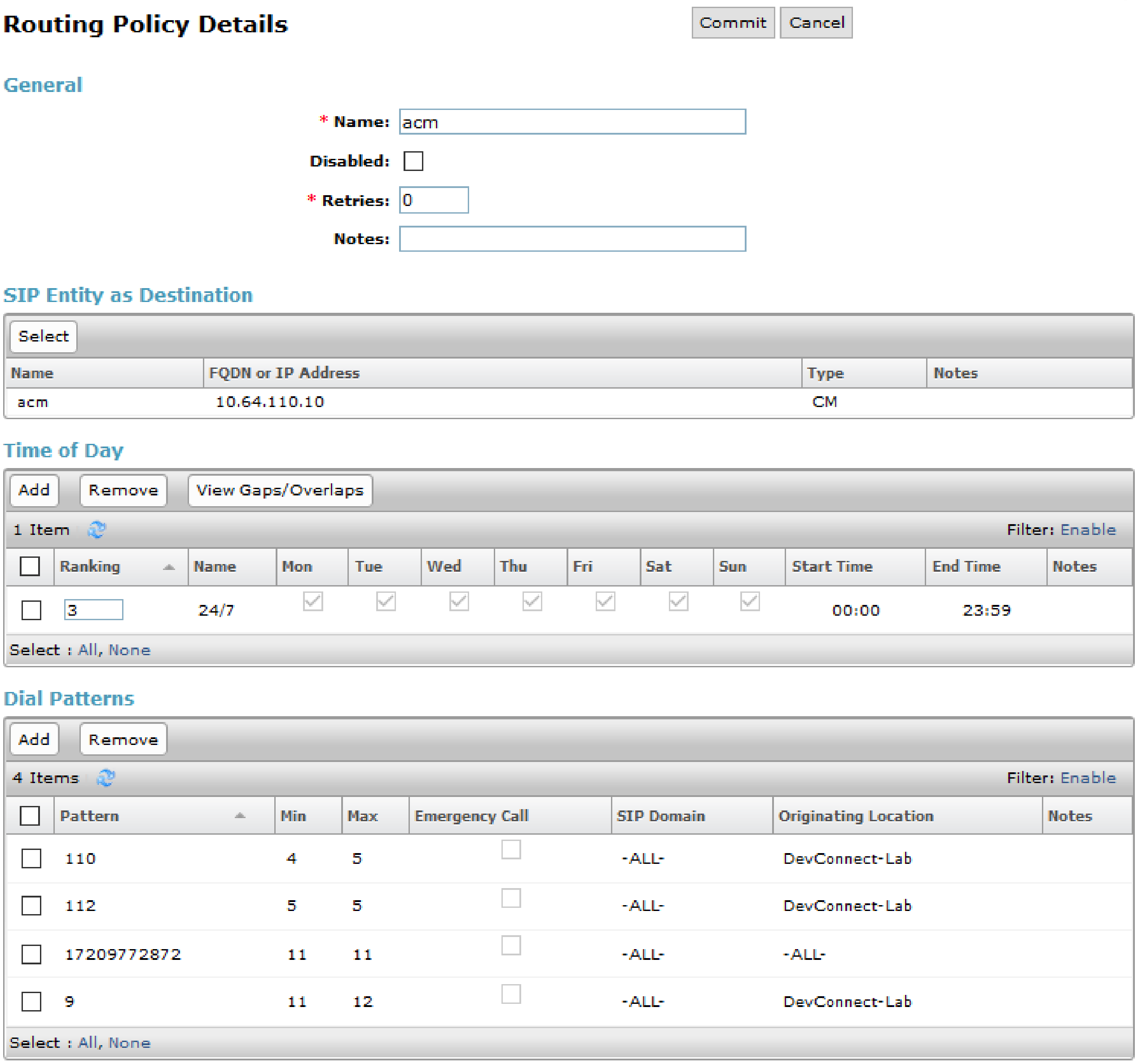
Under **SIP Entity as Destination**

- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

Under **Time of Day:**

- Click **Add**, and select the time range configured. In these Application Notes, the predefined **24/7** Time Range is used

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screens show the Routing Policies for Communication Manager and the Kmedia. Note that **Dial Patterns** (to be configured in **Section 6.6**), when configured, will be automatically displayed in the **Routing Policy Details** page.



Routing Policy Details

Commit | Cancel

General

* Name: acm

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
acm	10.64.110.10	CM	

Time of Day

Add | Remove | View Gaps/Overlaps

1 Item | Filter: Enable

☐	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	3	24/7	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	

Select : All, None

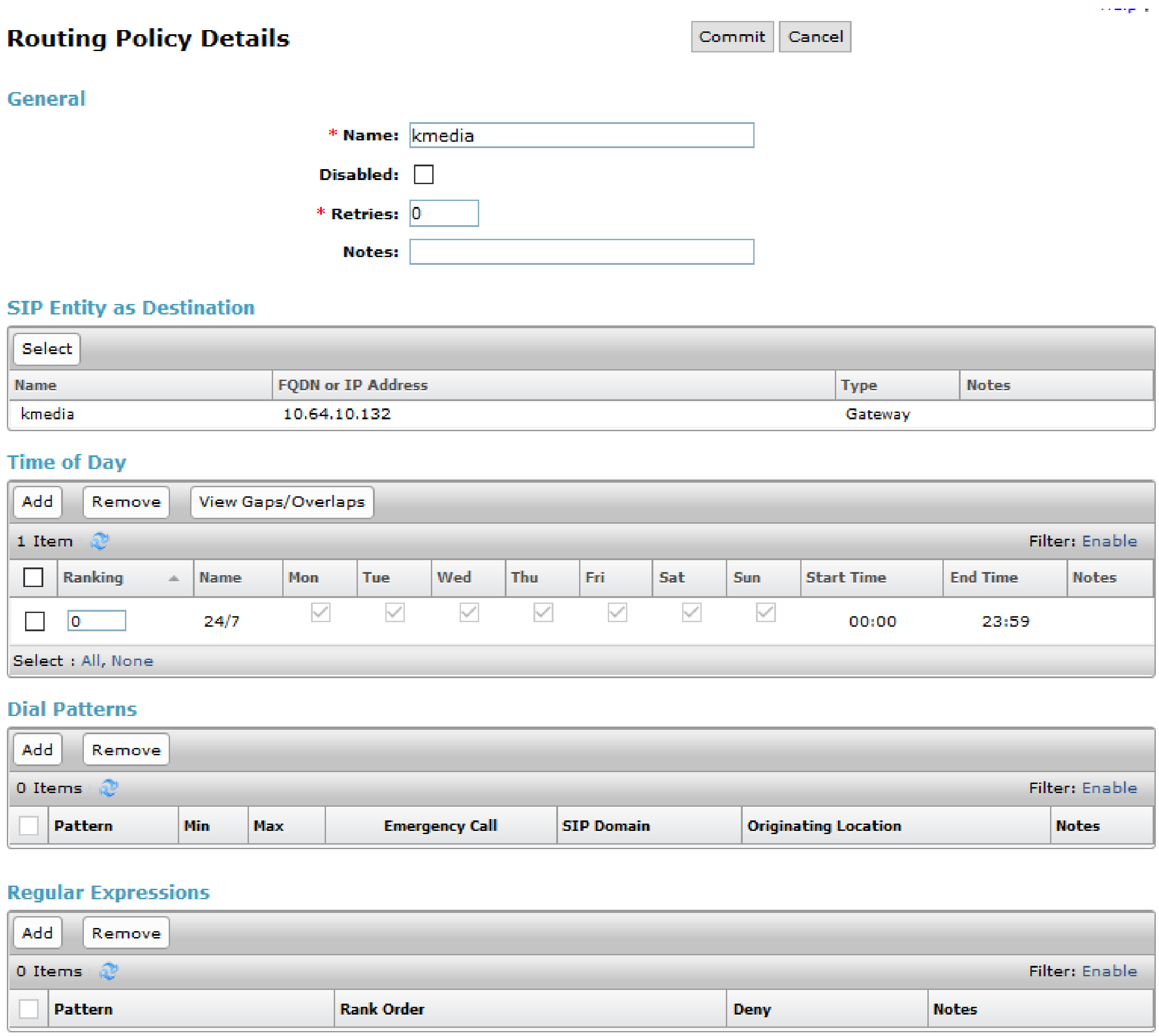
Dial Patterns

Add | Remove

4 Items | Filter: Enable

☐	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
☐	110	4	5	☐	-ALL-	DevConnect-Lab	
☐	112	5	5	☐	-ALL-	DevConnect-Lab	
☐	17209772872	11	11	☐	-ALL-	-ALL-	
☐	9	11	12	☐	-ALL-	DevConnect-Lab	

Select : All, None


Routing Policy Details
Commit
Cancel
General
* Name: kmedia
Disabled:
* Retries: 0
Notes:
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
kmedia
10.64.10.132
Gateway
Time of Day
Add
Remove
View Gaps/Overlaps
1 Item
Filter: Enable
Ranking
Name
Mon
Tue
Wed
Thu
Fri
Sat
Sun
Start Time
End Time
Notes
0
24/7
00:00
23:59
Select : All, None
Dial Patterns
Add
Remove
0 Items
Filter: Enable
Pattern
Min
Max
Emergency Call
SIP Domain
Originating Location
Notes
Regular Expressions
Add
Remove
0 Items
Filter: Enable
Pattern
Rank Order
Deny
Notes

6.6. Add Dial Patterns

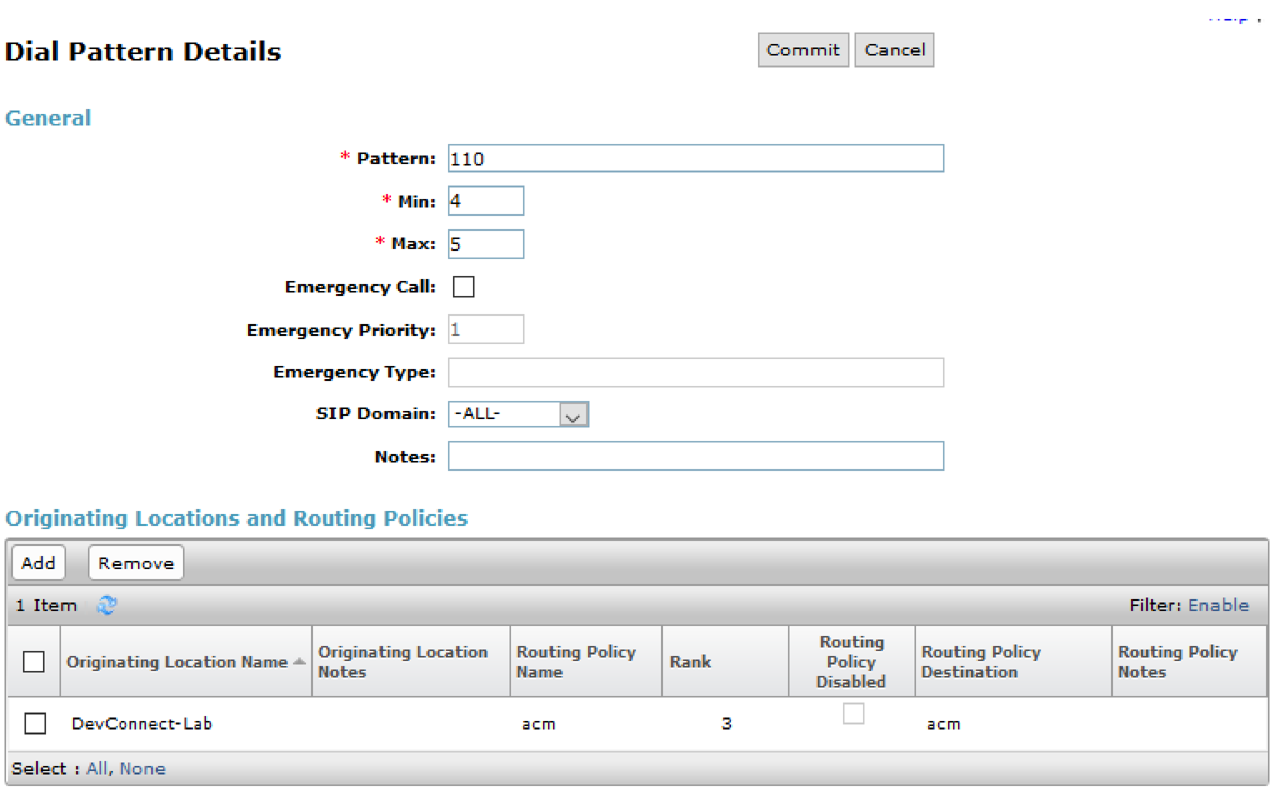
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration numbers beginning with **5** with 5-digit length reside in the Enterprise network. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not shown). Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Communication Manager.

Under **General**:

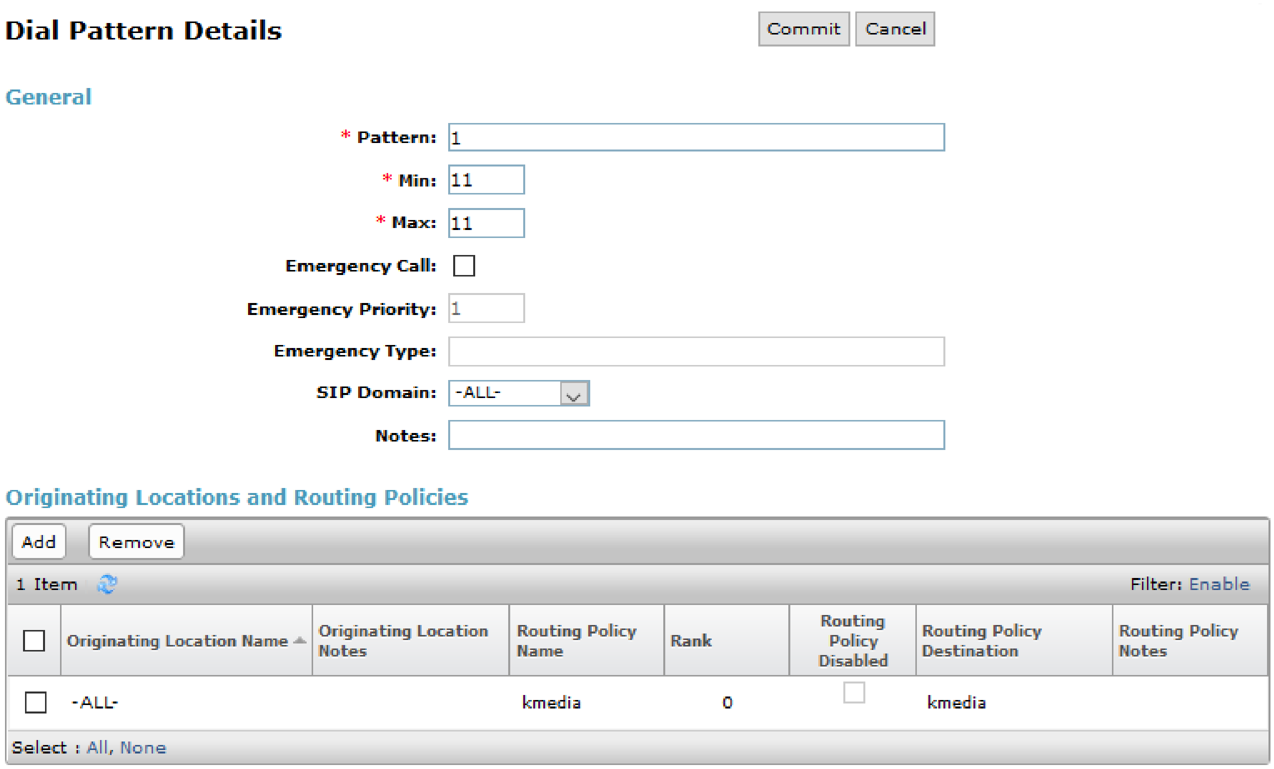
- **Pattern:** Dialed number or prefix i.e., **110**
- **Min:** Minimum length of dialed number i.e., **4**
- **Max:** Maximum length of dialed number i.e., **5**
- **SIP Domain:** Select **ALL**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for calls within the Enterprise.



The following screen shows the dial pattern definition for calls destined for the Kmedia.

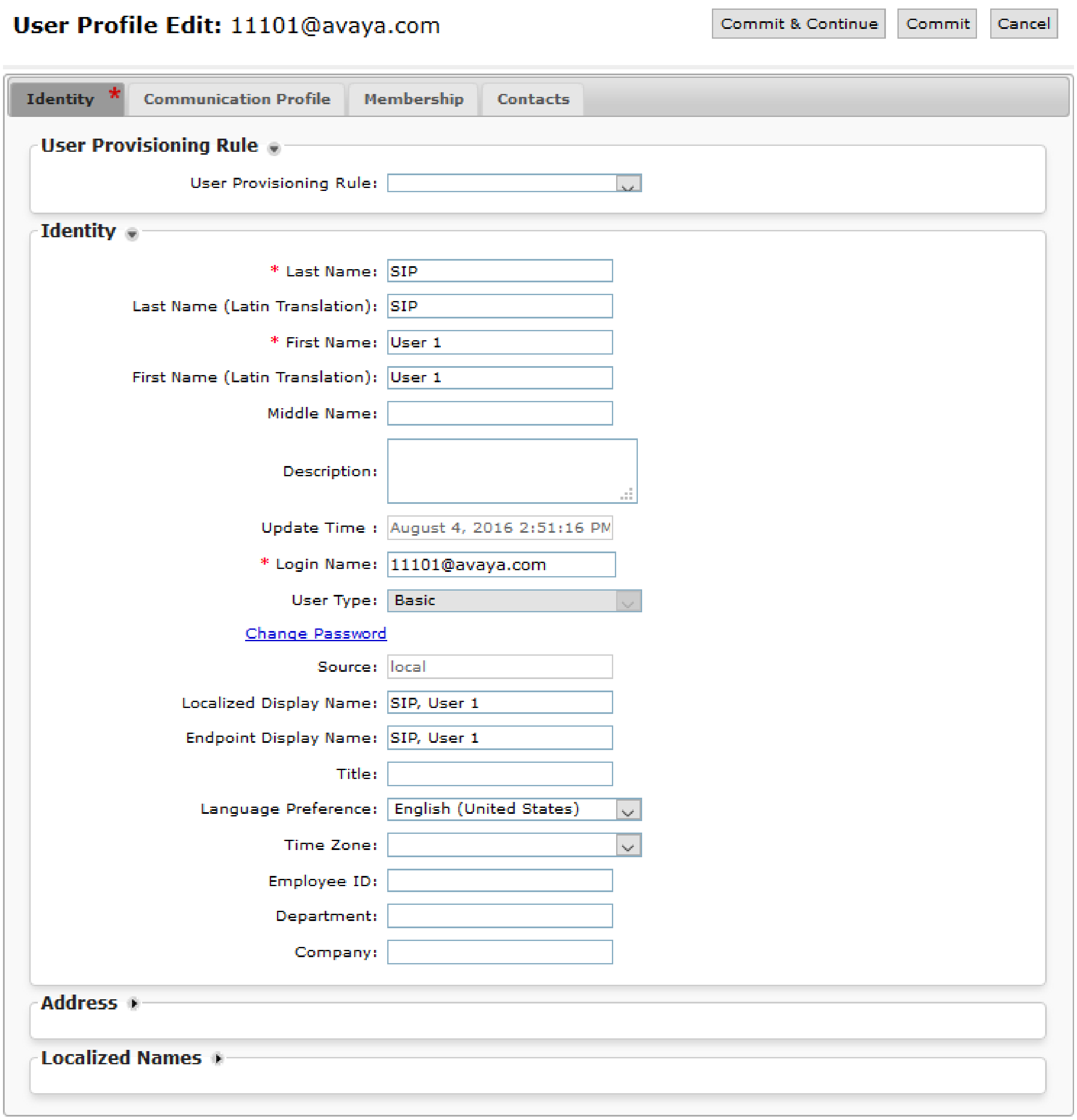


6.7. Add Users for SIP Phones

From the home screen select **Users → User Management → Manage Users** to display the **User Management** screen (not shown). Click **New** to add a user.

6.7.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "n@z", where "n" is the user extension and "z" is the domain name, in this case "**avaya.com**" used for compliance testing. Retain the default values in the remaining fields.



6.7.2. Communication Profile

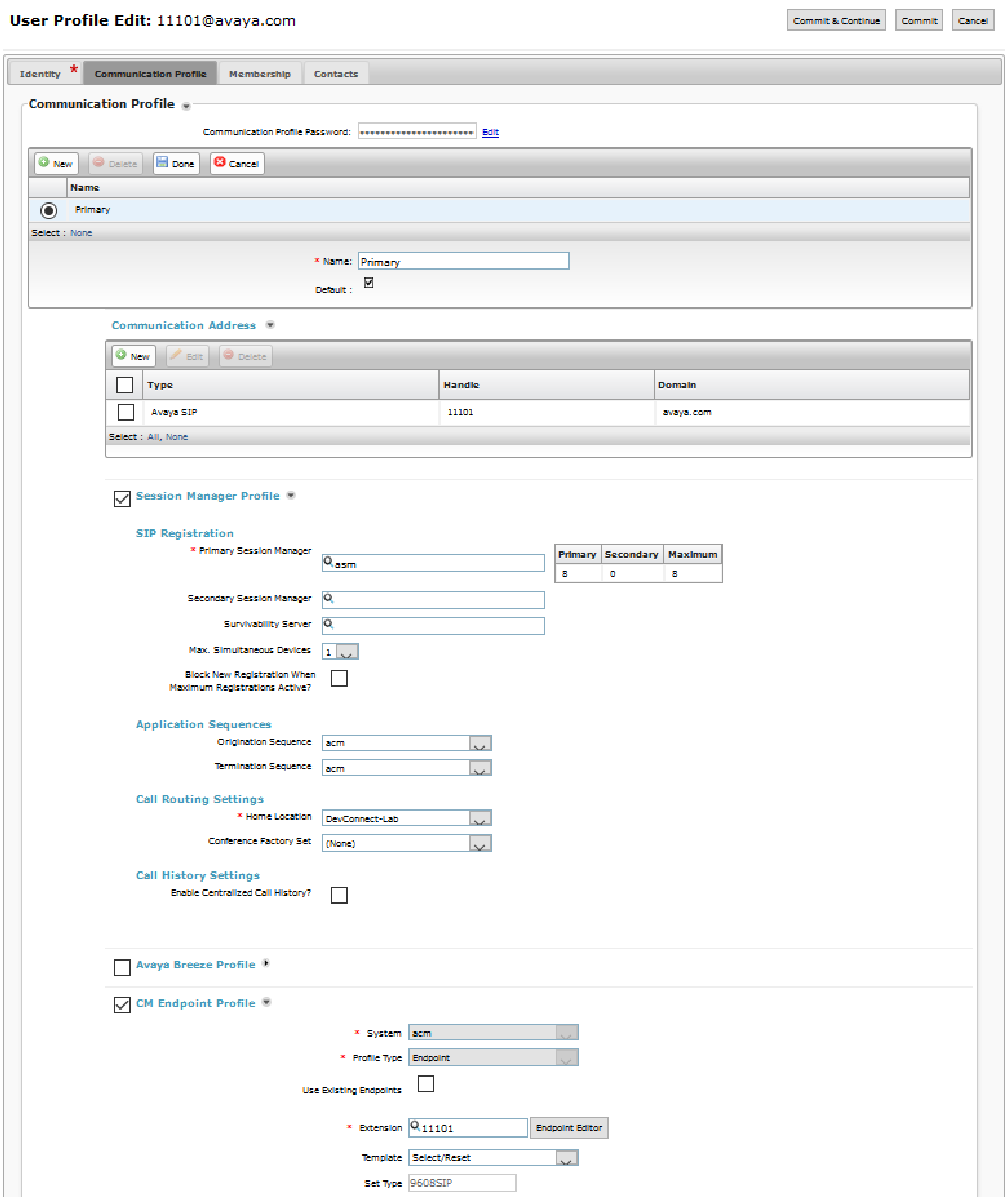
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain configured in **Section 6.7.1**. Click **Add**.

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter or select the SIP user extension configured in **Section 6.7.1**. For **Template**, select corresponding Telephone type. Retain the default values in the remaining fields.

Click **Commit** to complete the creation of the new user.


User Profile Edit: 11101@avaya.com
Commit & Continue
Commit
Cancel
Identity
Communication Profile
Membership
Contacts
Communication Profile
Communication Profile Password:
Edit
New
Delete
Done
Cancel
Name
Primary
Select : None
* Name: Primary
Default :
Communication Address
New
Edit
Delete
Type
Handle
Domain
Avaya SIP
11101
avaya.com
Select : All, None
Session Manager Profile
SIP Registration
* Primary Session Manager
asm
Primary
Secondary
Maximum
8
0
8
Secondary Session Manager
Survivability Server
Max. Simultaneous Devices
1
Block New Registration When Maximum Registrations Active?
Application Sequences
Origination Sequence
acm
Termination Sequence
acm
Call Routing Settings
* Home Location
DevConnect-Lab
Conference Factory Set
(None)
Call History Settings
Enable Centralized Call History?
Avaya Breeze Profile
CM Endpoint Profile
* System
acm
* Profile Type
Endpoint
Use Existing Endpoints
* Extension
11101
Endpoint Editor
Template
Select/Reset
Set Type
9608SIP

7. Kmedia Configuration

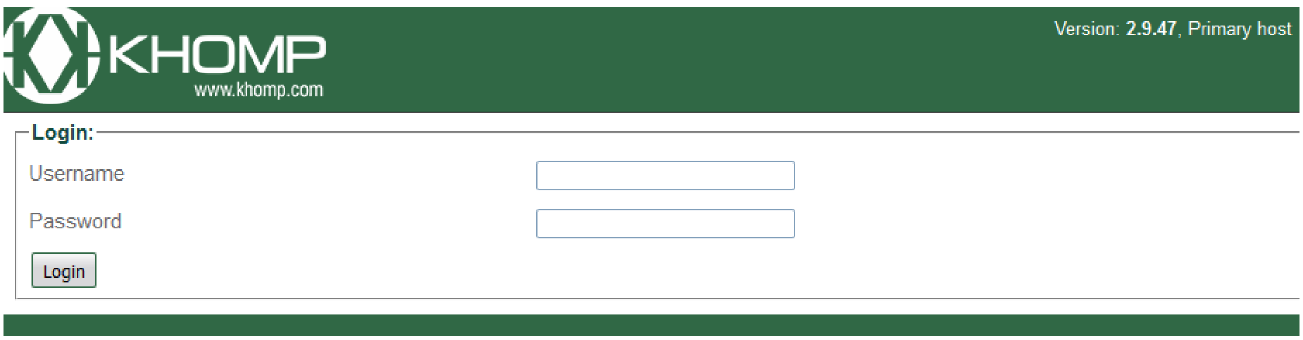
This section describes the configuration for enabling the Kmedia to interoperate with Session Manager.

7.1. Log Into Kmedia

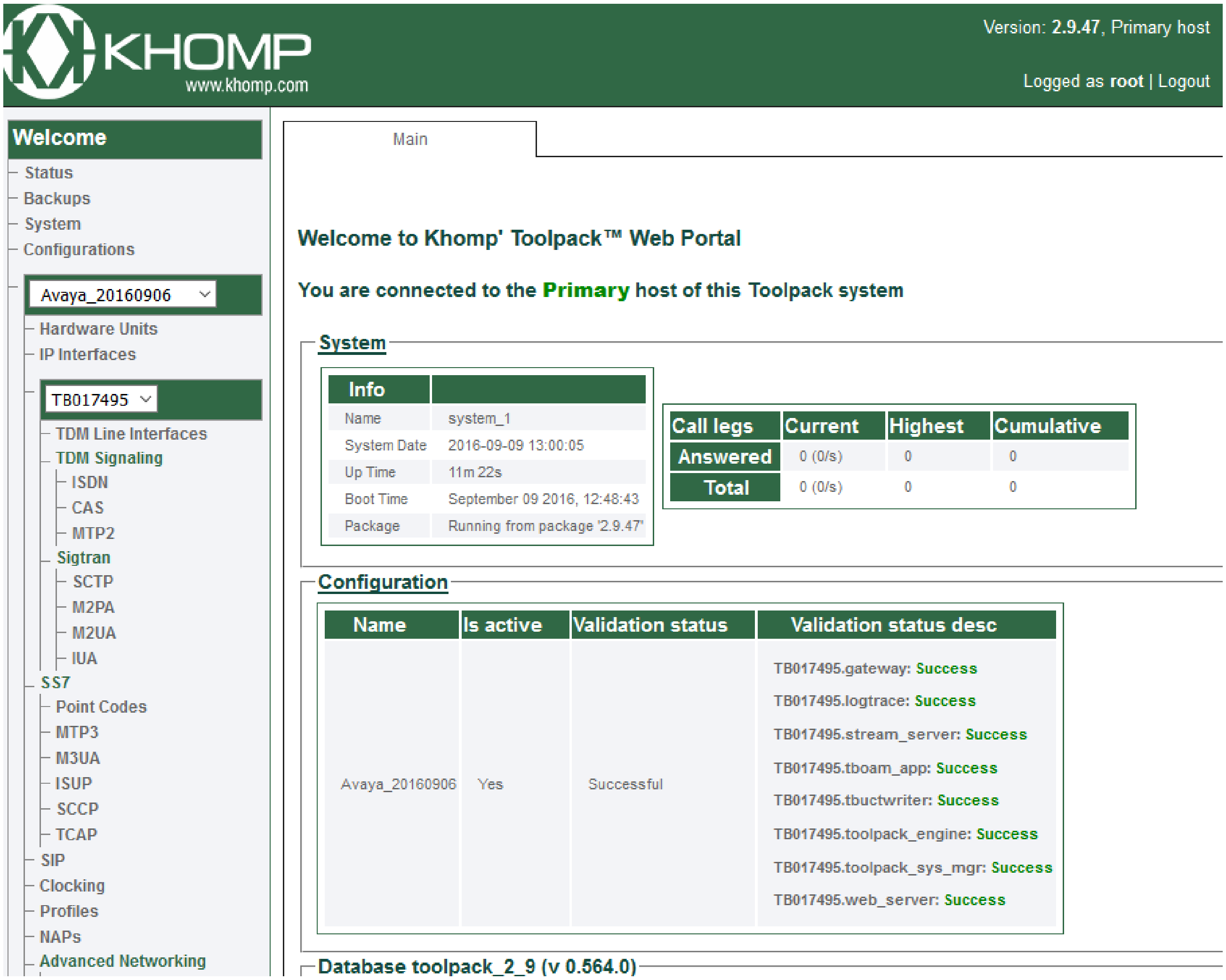
The configuration of the Kmedia Gateway is done via a Web browser. To access the device, enter the **IP address** of the Kmedia in the **Address** field of the web browser. The IP address was provisioned during initial installation.

Login credentials

The following window will appear. Log in with the proper credentials.



Once logged in the following home page is displayed.



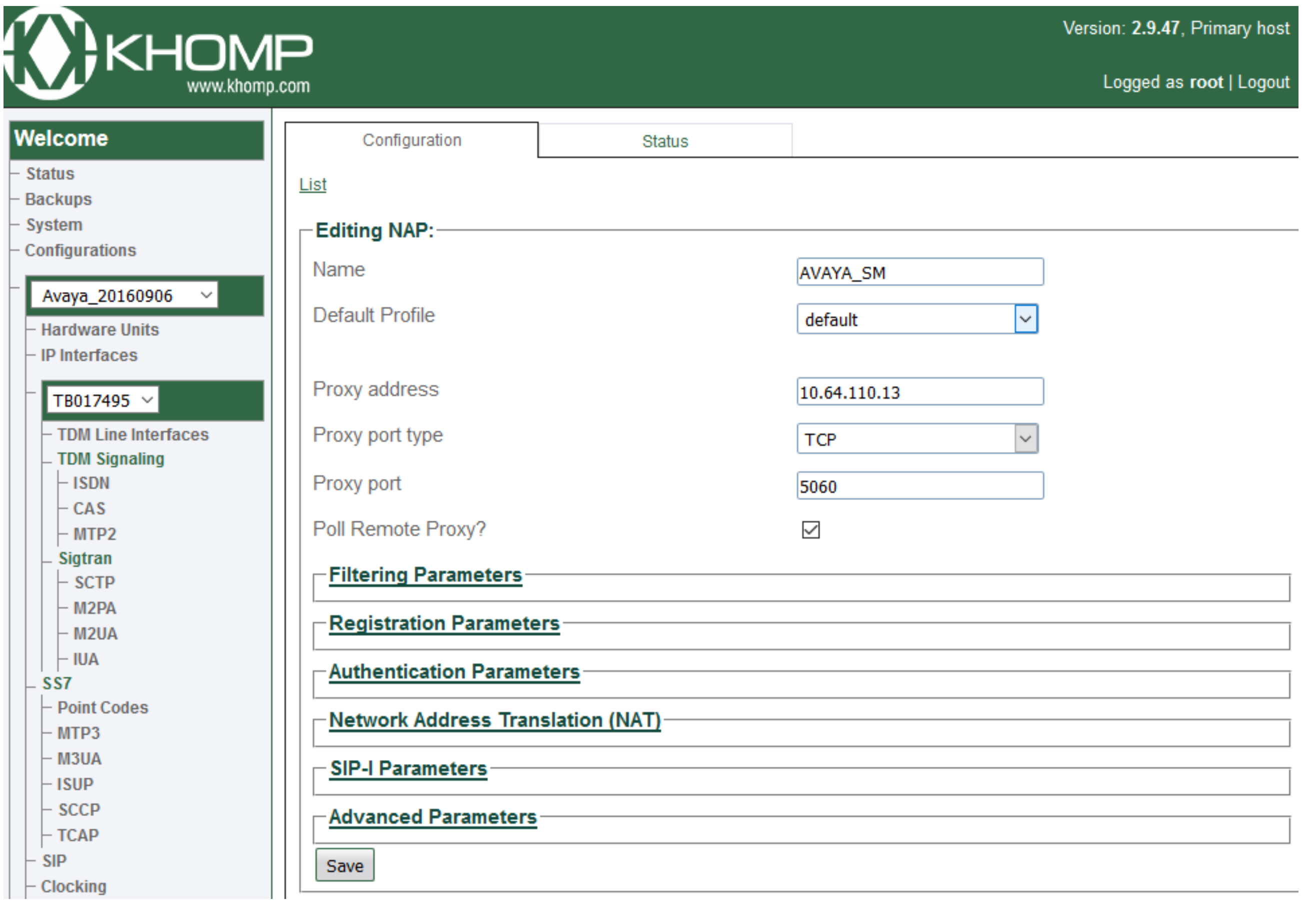
7.2. Configure NAP

To configure a NAP for Session Manager, navigate to **Configuration → NAPs → Create New NAP.**

- Type in a desired name in **Name**
- Type in the Session Managers' SM100 IP Address and Port in **Proxy Address** and **Proxy Port**.
- Set **Proxy Port** to **TCP**

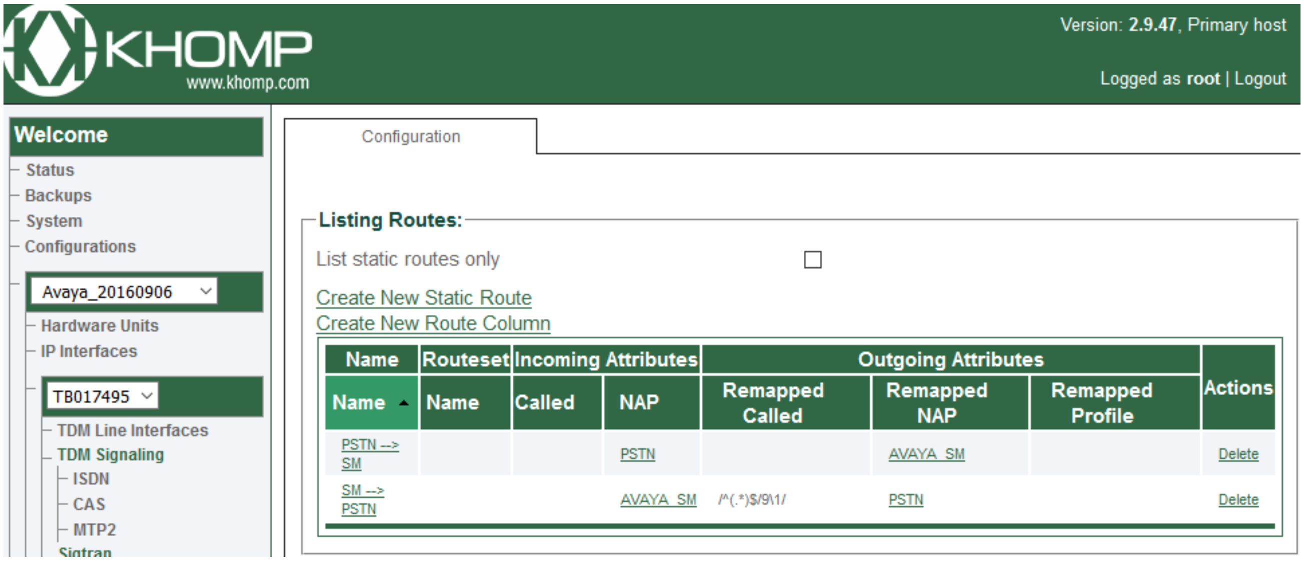
Once done select **Save** to save changes.

Similarly, based on the E1 carrier add a NAP for PSTN (not shown).

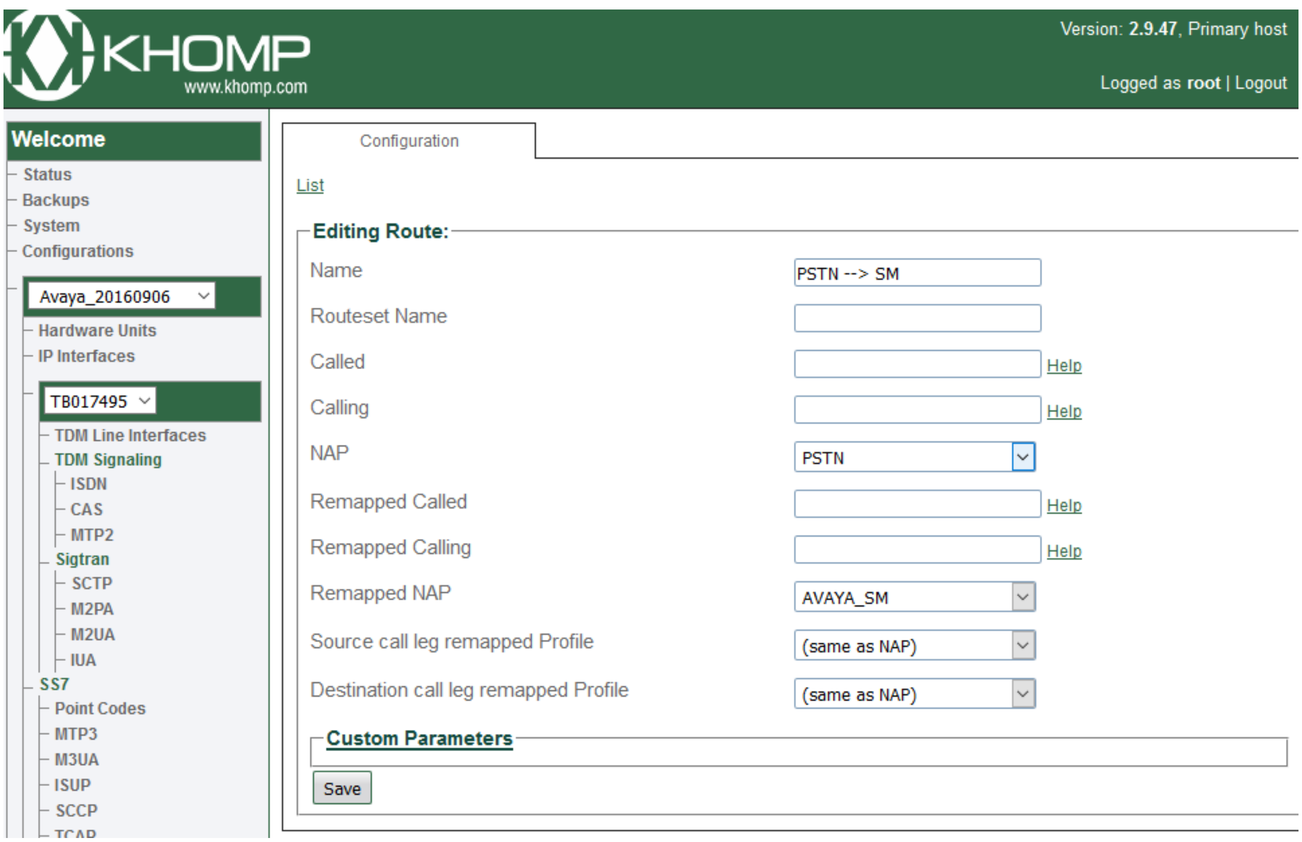


7.3. Configure Route

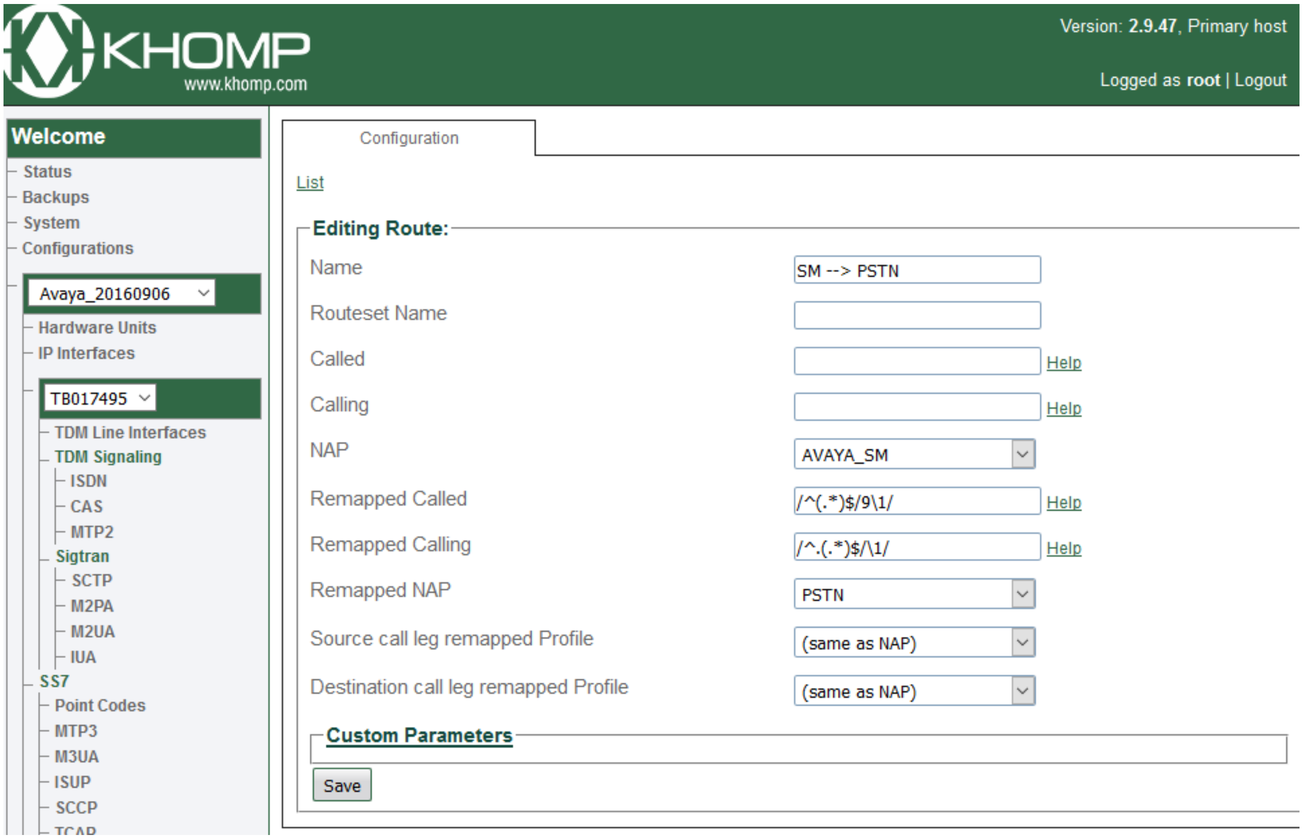
To add a route, navigate to **Configuration → Gateway → Routes.** Select **Create New Static Route** to add a new route.



The following screen capture shows the route that was added to route call from PSTN to Session Manager.

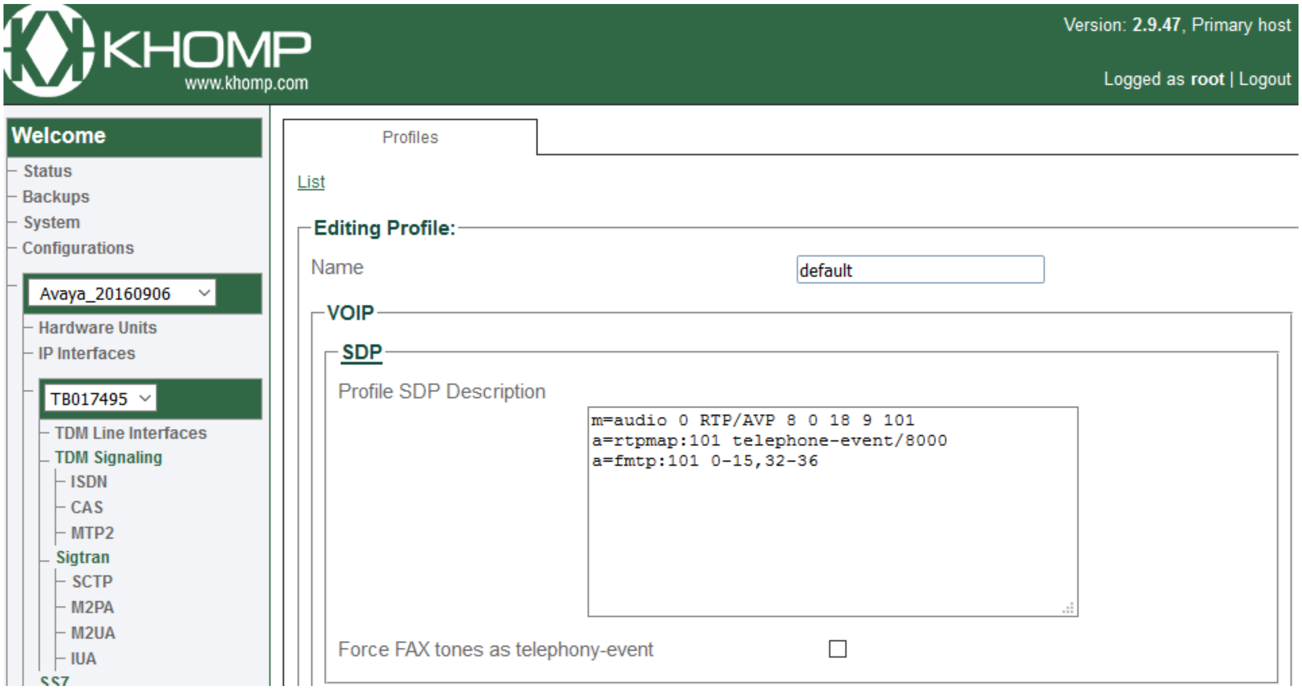


The following screen capture shows the route that was added to route calls from Session Manager to PSTN.



7.4. Configure Profile

To configure a profile, navigate to **Configuration → Profiles.** During the compliance test, **default** profile was used. Profile is used to configure audio parameters. Select **Edit** to configure audio parameters.



8. Verification Steps

This section provides the verification steps that may be performed to verify the configuration.

8.1. Verify Avaya Aura® Communication Manager Trunk Status

On Communication Manager, ensure that all the signalling groups are in service by issuing the command status **signalling-group n** where **n** is the signalling group number.

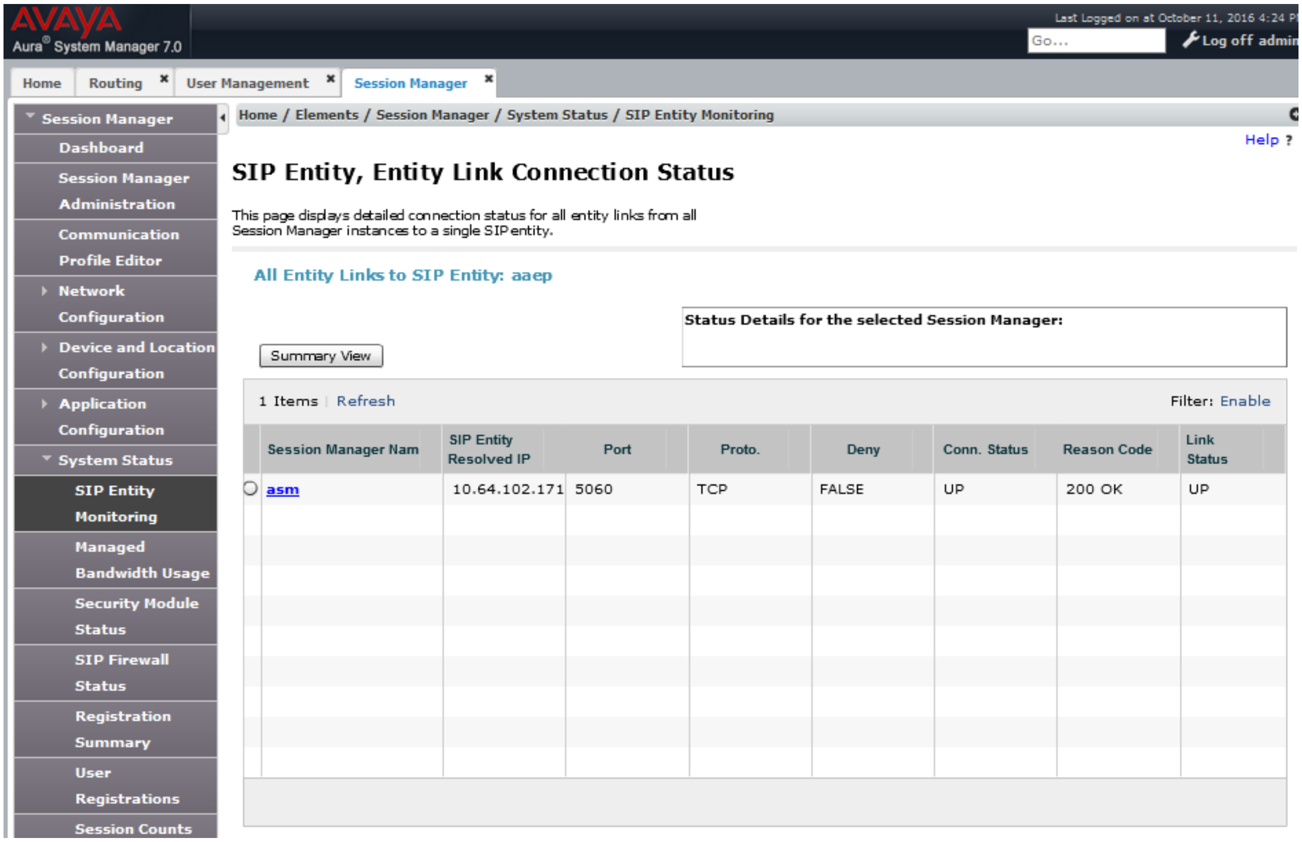
```
status signaling-group 1
                         STATUS
   SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

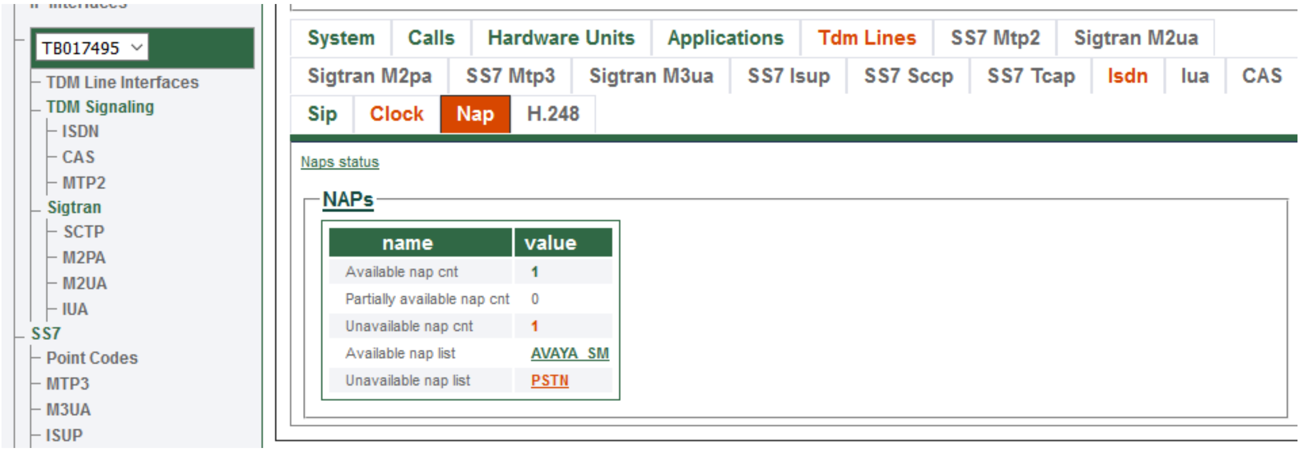
8.2. SIP Monitoring on Avaya Aura® Session Manager

From System Manager's Home screen, navigate to **Elements →Session Manager→System Status→SIP Entity Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing. The screen below shows the link status between Session Manager and the Kmedia.

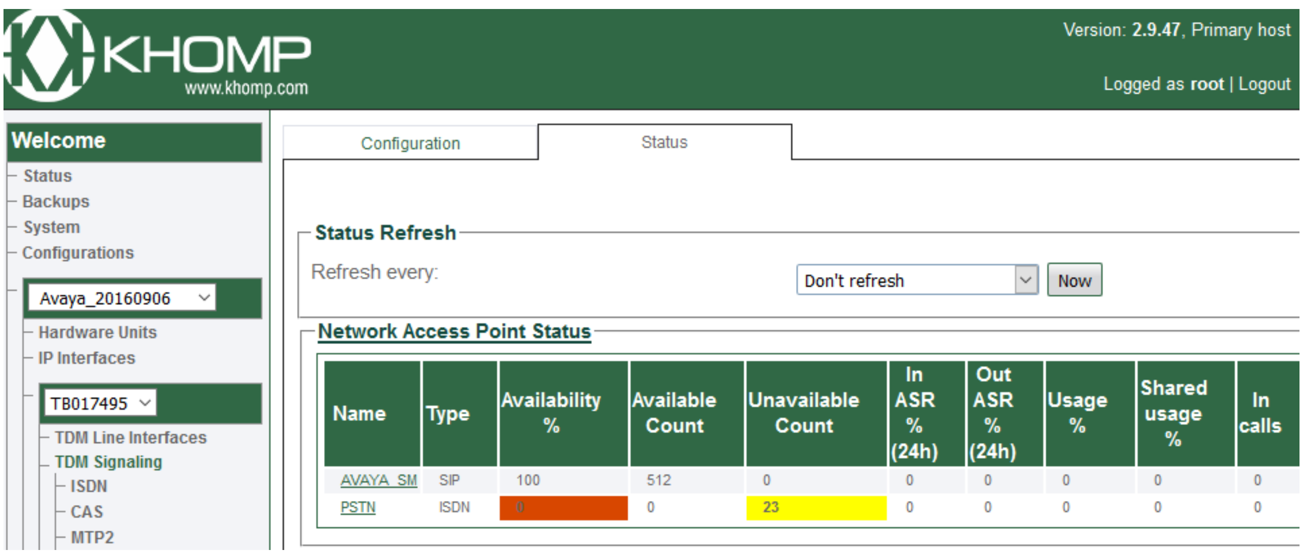


8.3. Kmedia Web Interface to Observe Status

To view the status of the SIP trunk between Kmedia and Session Manager, navigate to **Status → Nap**. Select the NAP that was added for Session Manager from the **value** column.



The NAP added for Session Manager shows 100% Availability.



9. Conclusion

These Application Notes describe the procedures required to configure the Kmedia Gateway to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Kmedia Gateway successfully passed compliance testing.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509
[2] *Administering Avaya Aura® Session Manager*, Document 03-603324
[3] Kmedia Manual

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

Product documentation for Khomp products may be found at http://www.khomp.com.

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.