Avaya Solution & Interoperability Test Lab

# Application Notes for HP Networking Multi Service Router 30 Series PSTN Gateway in an Avaya Telephony Environment that includes Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya Telephones – Issue 1.0

## Abstract


These Application Notes describe the steps for configuring HP Multi Service Router 30 series PSTN Gateway in an Avaya Telephony Environment that includes Avaya Aura® Communication Manager, Avaya Aura® Session Manager and various Avaya telephones.

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 a solution for configuring a HP Networking Multi-Service Router 30 series PSTN Gateway (herein referred to as MSR30) in an Avaya telephony environment.

The MSR30 product line is a configurable set of chassis and modules that meet a wide variety of customer demands. Modules include analog and digital voice, WAN data, VPN and Power over Ethernet (PoE) Networking.

Customers who use Avaya Unified Communications products with HP hardware, such as the ProLiant servers, often require Gateways to interoperate with traditional telephone equipment.

# 2. General Test Approach and Test Results

All test cases were performed manually. The general approach was to place various types of calls to and from the simulated PSTN. Feature testing included inbound and outbound calls, transfers, conference calls, Message Waiting Indicator (MWI), FAX and voicemail. For serviceability testing, failures such as cable pulls and resets were applied. All test cases passed, except the following anomalies:

| |
|---|
| The MSR30 does not support PSTN-side hold. |

## 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing evaluated the SIP trunk between Session Manager and MSR30 with inbound, outbound, transfer, conference, MWI, FAX, and voicemail. The serviceability testing introduced failure scenarios to see if the MSR30 could resume after failure.

## 2.2. Support

Technical support on HP Networking MSR30 series can be obtained through the following:

| | |
|---|---|
| HP Contract Holders in the Americas | Use the generic HP Technical Support number 1-800-633-3600. |
| Warranty issues in the Americas | Call 1-800-334-5144. When prompted, say Networking then 3Com, or H3C or A Series. |
| All other Worldwide regions | Please use the following:<br>HP Networking telephone contact link: www.hp.com/go/telephone<br>HP Networking product support link: www.hp.com/networking/support |

## 3. Reference Configuration

**Figure 1** provides the test configuration used for the compliance testing.

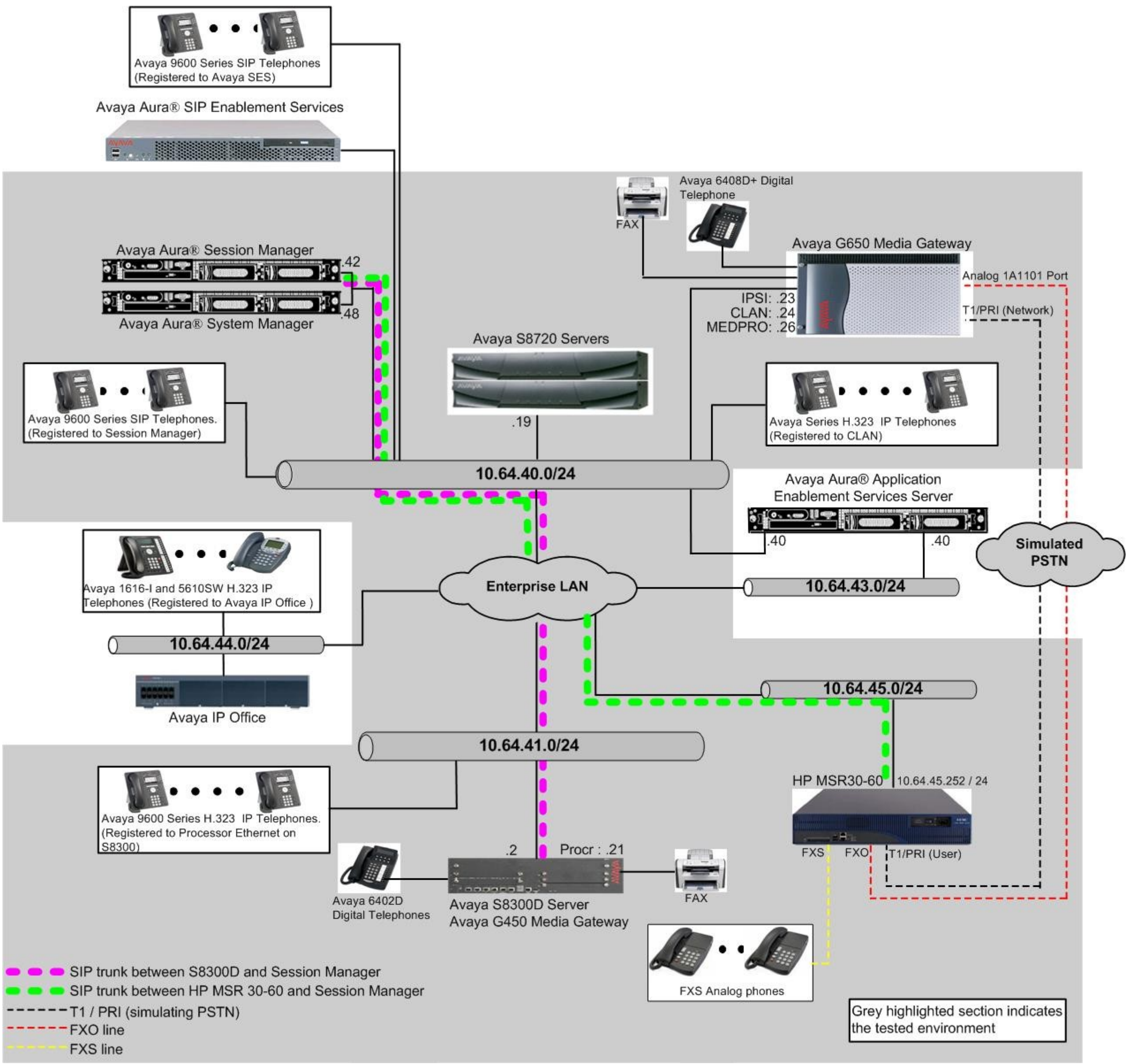



**Figure 1: The HP Networking MSR30 in the Avaya Telephony environment**

# 4. Equipment and Software Validated

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

| Equipment | | Software/Firmware |
|---|---|---|
| Avaya S8300D Server with Avaya G450 Media Gateway | | 6.0.1(R016x.00.1.510.1) w/ patch 00.1.510.1-18860 |
| Avaya Aura® System Manager | | 6.1 (R6-1-0-20-0) |
| Avaya Aura® Session Manager | | 6.1 |
| Avaya S8720 Servers with Avaya G650 Media Gateway | | Avaya Communication Manager 5.2.1 (R015x.02.1.016.4) |
| Avaya 9600 Series IP Telephones | | |
| | 9620 (H.323)<br>9630 (H.323) | 3.1<br>3.1 |
| Avaya 9600 Series SIP Telephones | | |
| | 9630 (SIP)<br>9640 (SIP)<br>9650 (SIP) | 2.6.4<br>2.6.4<br>2.6.4 |
| Avaya 6400 Series Digital Telephones | | N/A |
| Avaya C363T-PWR Converged Stackable Switch | | 4.5.14 |
| HP Networking 2910al-48G-PoE Switch | | W.14.30 |
| HP Networking Multi-Service Router 30-60 Series PSTN Gateway | | R2207P38 |
| HP 1-Port T1-Voice SIC A-MSR Module<br>HP 2-Port FXO SIC A-MSR Module<br>HP 4-Port FXO MIM A-MSR Module<br>HP 16-port FXS Voice Interface MIM Module<br>HP 16-Port 10/100 POE MIM A-MSR Module<br>HP Voice Co-Processor Module A-MSR Module<br>HP 24-Channel Voice Processor A-MSR Module | | |

# 5. Configure Avaya Testing Environment

This section describes the configuration for the Avaya telephony testing environment, shown in **Figure 1**.Telephones in the Enterprise side use a 72xxx dial plan and telephones on the PSTN side use a 2xxxx dial plan.

## 5.1. Configure Avaya Aura® Communication Manager on Enterprise side

This section describes the procedure for setting up a SIP trunk between Communication Manager and Session Manager. The steps include setting up an IP codec set, an IP network region, IP node name, a signaling group, and a trunk group. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. All SIP telephones are configured as off-PBX telephones in Communication Manager.

### 5.1.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 6400   211
                                      Maximum Stations: 2400   35
                              Maximum XMOBILE Stations: 2400   0
                    Maximum Off-PBX Telephones - EC500: 9600   0
                    Maximum Off-PBX Telephones -   OPS: 9600   18
                    Maximum Off-PBX Telephones - PBFMC: 9600   0
                    Maximum Off-PBX Telephones - PVFMC: 9600   0
                    Maximum Off-PBX Telephones - SCCAN: 0      0
                         Maximum Survivable Processors: 313    1
```

On **Page 2**of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000   30
           Maximum Concurrently Registered IP Stations: 2400   5
             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   0
                       Maximum Administered SIP Trunks: 4000   110
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000   0
   Maximum Number of DS1 Boards with Echo Cancellation: 80     0
                             Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 50     1
           Maximum TN2602 Boards with 80 VoIP Channels: 128    0
          Maximum TN2602 Boards with 320 VoIP Channels: 128    0
   Maximum Number of Expanded Meet-me Conference Ports: 300    0
```

### 5.1.2. IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and Session Manager. Enter the **change ip-codec-set <c>**command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 5.1.3** for configuring an IP network region to specify which codec sets may be used within and between network regions. The compliance test used G.711MU.

```
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:
 3:
```

On **Page 2**, set **FAX Mode** to **t.38-standard** for FAXing through the SIP trunk via the MSR30.

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

                           IP Codec Set

                               Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia:  4096:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia:  4096:Kbits

                    Mode                Redundancy
    FAX             t.38-standard        3
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

### 5.1.3. Configure IP Network Region

This section describes the steps for configuring an IP network region in Communication Manager in order to work with Session Manager. Enter the **change ip-network-region <n>**command, where **n** is a number between **1** and **250** inclusive and configure the following:

- **Authoritative Domain**–Set to the appropriate domain. During the compliance test, the authoritative domain is set to **avaya.com**. This should match the SIP Domain value on Session Manager in **Section 5.3.1**
- **Intra-region IP-IP Direct Audio**– Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in the same IP network region. The default value for this field is **yes**.
- **Codec Set**– Set the codec set number as provisioned in **Section 5.1.2**.
- **Inter-region IP-IP Direct Audio**– Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in different IP network regions. The default value for this field is **yes**.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
Codec Set: 1                    Inter-region IP-IP Direct Audio: yes
UDPPort Min: 2048                      IP Audio Hairpinning? n
UDPPort Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

### 5.1.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Session Manager. Enter the **change node-names ip** command and add a node name for Session Manager with its IP address.

```
change node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM-1                10.64.40.42
default             0.0.0.0
msgserver-ip        10.64.41.21
msgserver-sip       10.64.41.21
procr               10.64.41.21
procr6              ::
```

### 5.1.5. Configure SIP Signaling

This section describes the steps for administering a signaling group for communication between Communication Manager and Session Manager. Enter the **add signaling-group <s>**command, where **s** is an available signaling group and configure the following:

- Group Type– Set to **sip.**
- IMS Enabled – Verify the field is set to **n**. This configures the Communication Manager to function as an Evolution Server. Setting this field to **y** causes Communication Manager to function as a Feature Server.
- Near-end Node Name- Set to **procr** as shown in **Section 5.1.4**.
- Far-end Node Name - Set to **SM-1** (Session Manager) as shown in **Section 5.1.4**.
- Far-end Network Region - Set to the region configured in **Section 5.1.3**.
- Far-end Domain- Set to **avaya.com**, matching the SIP Domain in **Section 5.3.1**.

Take note of the Group Number value as it will be needed in **Section 5.1.6**.

```
add signaling-group 92                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 92              Group Type: sip
IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                                  SIP Enabled LSP? n
     IP Video? n                                        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



Near-end Node Name: procr                   Far-end Node Name: SM-1
Near-endListenPort: 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? n
         Enable Layer 3 Test? y                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

### 5.1.6. Configure SIP Trunk

This section describes the steps for administering a trunk group for communication between Communication Manager and Session Manager. Enter the **add trunk-group <t>**command, where **t** is an unallocated trunk group and configure the following:

- **Group Type**– Set to **sip**.
- **Group Name**– Enter a descriptive name.
- **TAC** (Trunk Access Code) – Set to any available trunk access code.
- **Signaling Group**– Set to the Group Number field value configured in **Section 5.1.5**.
- **Number of Members–** Allowed values are between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 92                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 92                   Group Type: sip         CDR Reports: y
Group Name: No IMS SIP trk              COR: 1        TN: 1        TAC: 1092
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                     Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 92
                                                    Number of Members: 10
```

## 5.2. Configure Avaya Aura® Communication Manager on PSTN side

This section describes the procedure for setting up a T1/PRI trunk between Communication Manager and the MSR30. The steps include setting up a DS1 card, a signaling group, and a trunk group. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface.

### 5.2.1. Configure DS1 card for T1

Enter the **add ds1 x** command, where **x** is the board number of the DS1 circuit pack. Enter a descriptive Name and set the other highlighted fields to the values indicated. Provide the following information:

- Name – A descriptive name.
- Line Coding– Select **b8zs**. This value should match the MSR30's configuration.
- Framing Mode - Select **esf**. This value should match the MSR30's configuration.
- Signaling Mode –Select **isdn-pri**.
- Connect – Select **pbx**.
- Interface – Select **network**.

*Note: Communication Manager was set to **network** on the Interface field. This means the MSR30 should be set to **user** mode.*

```
add ds1 1a09                                                      Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01A09                          Name: Temp-HP
            Bit Rate: 1.544                   Line Coding: b8zs
   Line Compensation: 1                      Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: network
   TN-C7 Long Timers? n                    Country Protocol: 1
Interworking Message: PROGress             Protocol Version: b
Interface Companding: mulaw                             CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                          T303 Timer(sec): 4


      Slip Detection? y                 Near-end CSU Type: other

                                 Block Progress Indicator? n
```

### 5.2.2. Configure Signaling Group

Enter the **add signaling-group s** command, where **s** is an unused signaling group number. Set the highlighted fields below to the values indicated.

```
add signaling-group 80                                            Page   1 of   5
                                SIGNALING GROUP

 Group Number: 80              Group Type: isdn-pri
                     Associated Signaling? y         Max number of NCA TSC: 0
Primary D-Channel: 01A0924     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC:
Trunk Group for Channel Selection: 80
      TSC Supplementary Service Protocol: a          Network Call Transfer? n
```

## 5.2.3. Configure Trunk Group

Enter the **add trunk-group t** command, where **t** is an unused trunk group number.  Set the highlighted fields below to the values indicated.

```
add trunk-group 80                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 80                   Group Type: isdn         CDR Reports: r
Group Name: Temp-HP                     COR: 1        TN: 1         TAC: 113
   Direction: two-way        Outgoing Display? y        Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie            Auth Code? n             TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

On **Page 5**, add trunk group members.

```
display trunk-group 80                                           Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  23

       Port     Code Sfx Name        Night           Sig Grp
  1: 01A0901   TN464  G                                80
  2: 01A0902   TN464  G                                80

  .  .         .      .                                .
  .  .         .      .                                .

 22: 01A0922   TN464  G                                80
 23: 01A0923   TN464  G                                80
```

## 5.2.4. Configure UDP and AAR

During the compliance test, Automatic Alternate Routing (AAR) and Uniform Dial Plan (UDP) were utilized between Communication Manager and the MSR30 via the T1/PRI trunk.  The following displays the sample UDP and AAR configuration used during the test.

Enter the **change uniform-dialplan d** command, where **d** is any digit that is valid under the provisioned dial plan.  Enter the whole or a partial extension on enterprise side for the Matching Pattern field.  Enter the length of the extension for the Len field.  Set the Del field to **0**, and the Net field is set to **aar**.

```
change uniform-dialplan 7                                        Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert              Node
  Pattern           LenDel      Digits      Net ConvNum
720                 5    0                  aar   n
```

Enter the **change aar analysis d** command, where **d** is any digit that is valid under the provisioned dial plan. Enter the whole or a partial extension on enterprise side for the Matching Pattern field. Enter the number of an unused route pattern for the Route Pattern field. The Call Type field is set to **aar**.

```
change aar analysis 7                                           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
     720                    5    5      80        aar            n
```

### 5.2.5. Configure Route Pattern

Enter the **change route-pattern r** command, where **r** is the number of the route pattern specified in previous section. Enter the number of the trunk group configured for the Grp No field. Assign a Facility Restriction Level to this routing preference for the FRL field. The FRL value **0** is the least restrictive.

```
change route-pattern 80                                         Page   1 of   3
                    Pattern Number: 80  Pattern Name: To PSTN via G3r
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 80   0                                                           n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                      none
```

## 5.3. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

This section assumes that Session Manager and System Manager have been installed, network connectivity exists between the two platforms, and the basic configuration is performed.

The following list outlines the steps for configuring Session Manager.

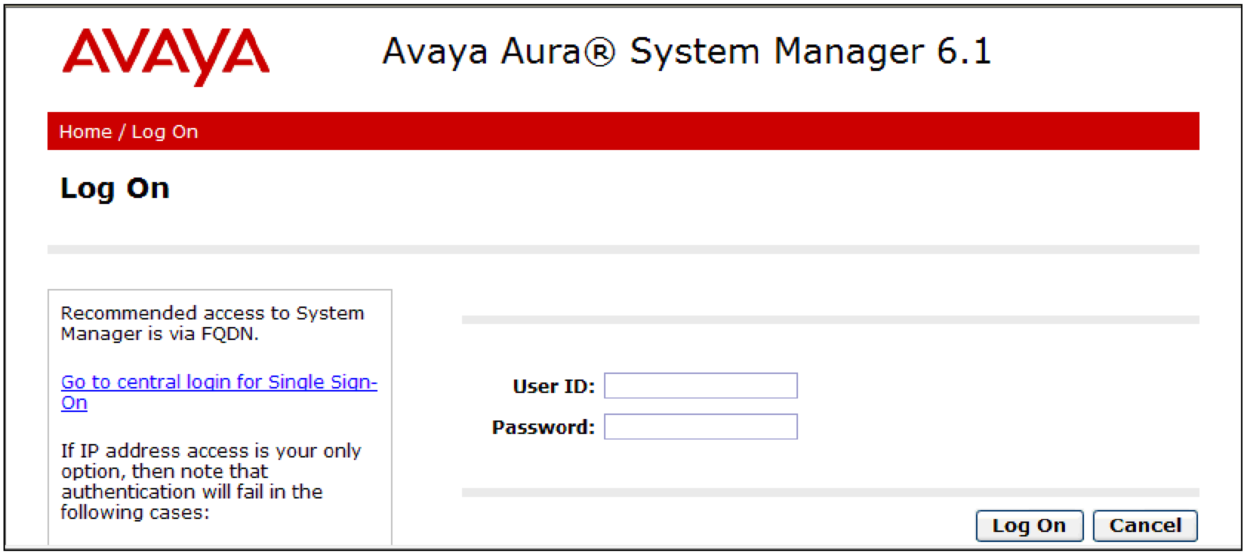
- SIP Domains
- Locations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policy
- Dial Patterns
- Manage Element
- Applications
- Application Sequence
- User Management

For a SIP trunk between Session Manager and the MSR30, the following sections need to be configured:

- **SIP Entity**
- **Entity Link**
- **Routing Policy**
- **Dial Patterns**

### 5.3.1. Configure SIP Domain

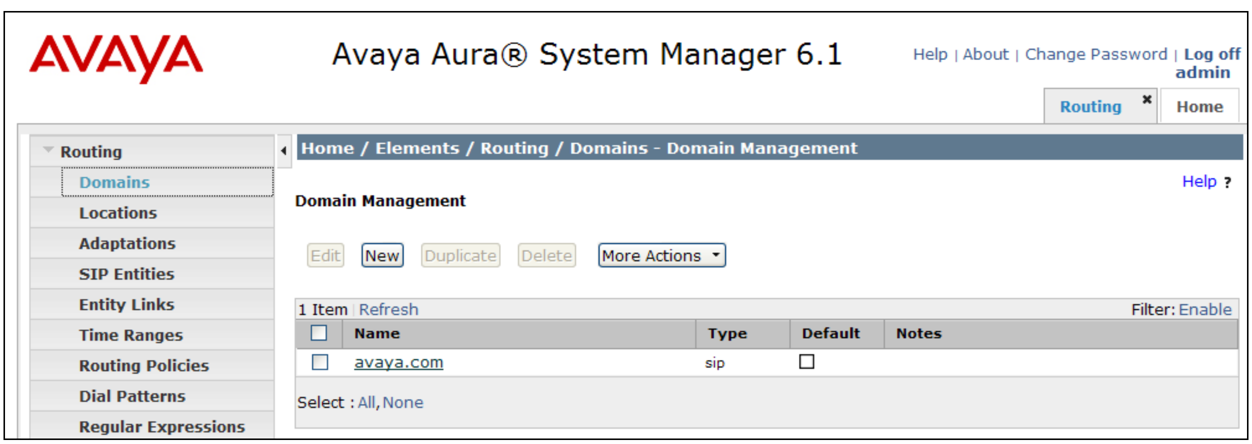
Launch a web browser, enter **https://<IP address of System Manager>/SMGR** in the URL, and log in with the appropriate credentials.




Navigate to **Elements→Routing→Domains** and click on the **New** button to create a new SIP Domain (screen not shown).Enter the following values and use defaults for the remaining fields:

- **Name** –Enter the Authoritative Domain name specified in **Section 5.1.3**, which is **avaya.com**.
- **Type** – Select **SIP**

Click **Commit** to save. The following screen shows the Domains page used during the compliance test.



### 5.3.2. Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. This is used for bandwidth management or location-based routing.

Navigate to **Routing→Locations**, and click on the **New** button to create a new location (screen not shown).

<u>General section</u>

Enter the following values and use default values for the remaining fields.

- Enter a descriptive Location in the **Name** field (e.g. **.41 Subnet**).
- Enter a description in the **Notes** field if desired.

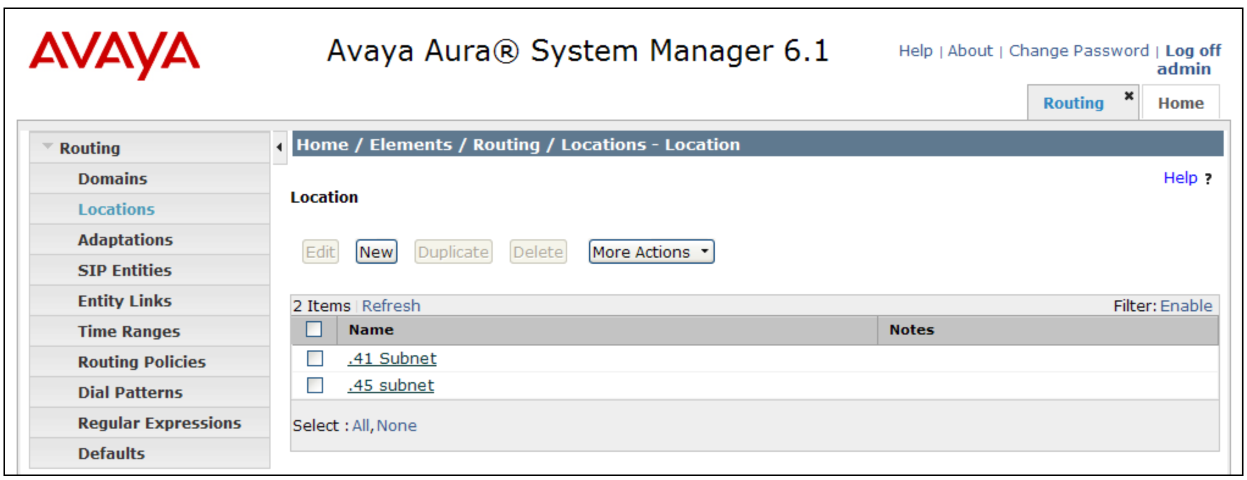
<u>Location Pattern section</u>

Click **Add** and enter the following values:

- The IP address information for the **IP address Pattern** (e.g. **10.64.41.***).
- A description in the **Notes** field if desired.

Repeat these steps in the Location Pattern section if the Location has multiple IP segments. Modify the remaining values on the form, if necessary; otherwise, use all the default values. Click on the **Commit** button.

Repeat all the steps for each new Location. The following screen shows the Location page used during the compliance test.



### 5.3.3. Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk. During the compliance test the following SIP Entities were configured:

- Session Manager itself
- Communication Manager (Avaya S8300D Server)
- Communication Manager (Avaya S8720 Servers – not shown)
- HP MSR

Navigate to **Routing →SIP Entities** and click on the **New** button to create a new SIP entity (screen not shown). Provide the following information:

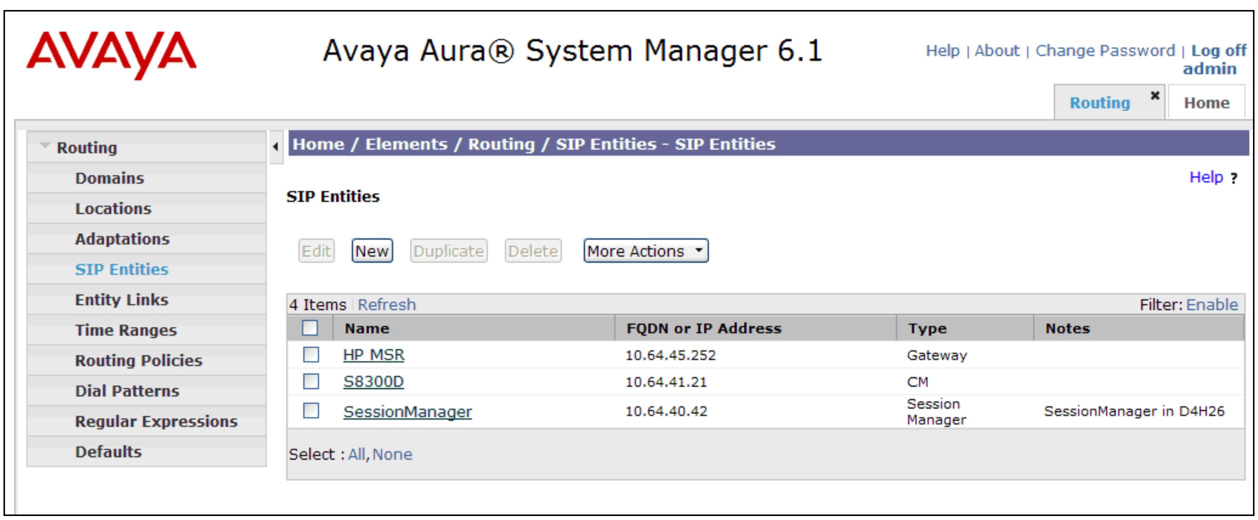
General section

Enter the following and use default values for the remaining fields:

- **Name**: Enter a descriptive name.
- **FQDN or IP Address:** Enter the IP address of the signaling interface on each:
    - Communication Manager
    - Session Manager virtual SM-100
    - HP MSR
- From the **Type** drop down menu, select a type that best matches the SIP Entity:
    - For Communication Manager, select **CM**
    - For Session Manager, select **Session Manager**
    - For the MSR, select **gateway**
- Enter a description in the **Notes** field if desired.
- Select a location from **Section 5.3.2**.
- Select the appropriate time zone.
- Accept the other default values.

Click on the **Commit** button to save each SIP entity. The following screen shows the SIP Entities page used during the compliance test.

Repeat all the steps for each new entity.



### 5.3.4. Configure Entity Links

Entity Links define the connections between the SIP Entities and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

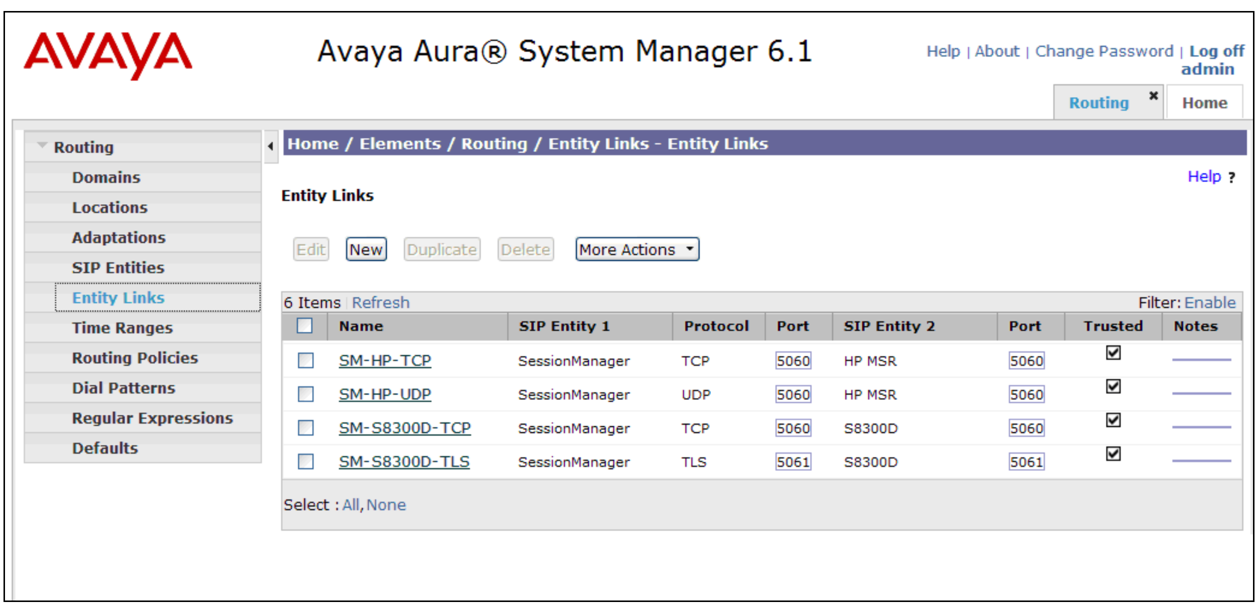
- Session Manager ⇔Communication Manager (Avaya S8300D Server)
- Session Manager ⇔MSR30

Navigate to **Routing →Entity Links** and click on the **New** button to create a new entity link (screen not shown). Provide the following information:

- **Name**: Enter a descriptive name.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity created in **Section 5.3.3** (e.g. **SessionManager**).
- In the **Protocol** drop down menu, select the protocol to be used.
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
    - TLS – 5061
    - UDP or TCP – 5060
- In the **SIP Entity 2** drop down menu, select one of the two entities in the bullet list above (which were created in **Section 5.3.3.**). In the compliance test **HP MSR** was selected.
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition. The following screen shows an Entity Links page used during the compliance test.

Repeat all the steps for each new SIP Entity Link.



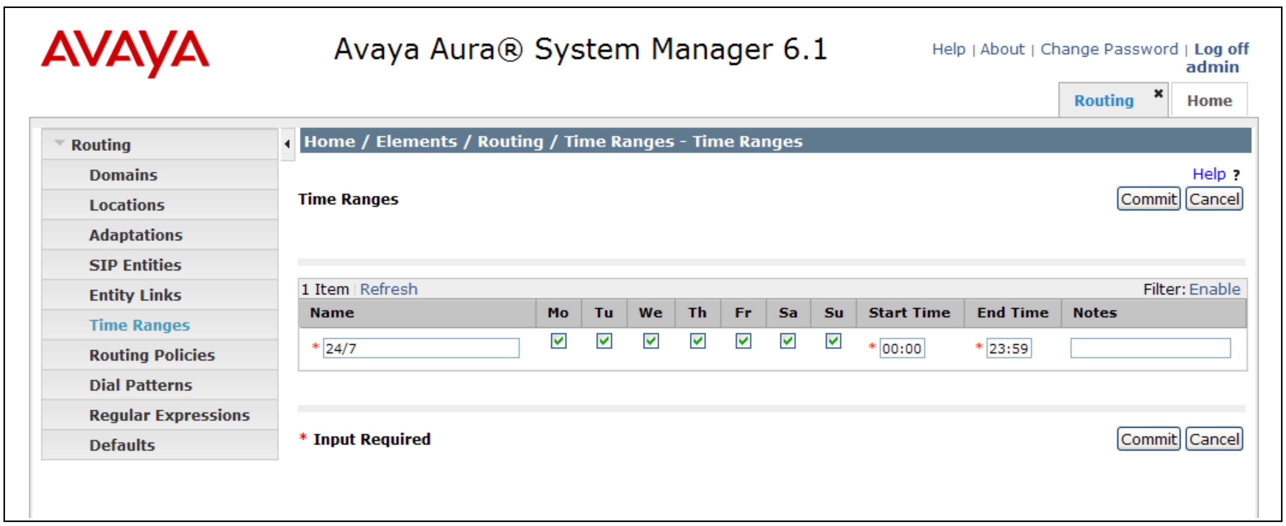
### 5.3.5. Time Ranges

Time Ranges define admission control criteria to be specified for Routing Policies (**Section 5.3.6**). In the reference configuration, no restrictions were used.

To add a Time Range, navigate to **Routing→Time Ranges**, and click on the **New** button (not shown). Provide the following information:

- Enter a descriptive name in the **Name** field (e.g. **24/7**).
- Check each day of the week.
- In the **Start Time** field, enter **00:00**.
- In the **End Time** field, enter **23:59**.
- Enter a description in the **Notes** field if desired.

Click the **Commit** button. The following screen shows the Time Range page used during the compliance test.



### 5.3.6. Configure Routing Policy

Routing Policies associate destination SIP Entities (**Section 5.3.3**) with Time of Day admission control parameters (**Section 5.3.5**) and Dial Patterns (**Section 5.3.7**). In the reference configuration, Routing Policies are defined for:

- Inbound calls to Communication Manager.
- Inbound calls to MSR30.

To add a Routing Policy, navigate to **Routing →Routing Policies** and click on the **New** button on the right (screen not shown). Provide the following information:

General section

- Enter a descriptive name in the **Name** field.
- Enter a description in the **Notes** field if desired.

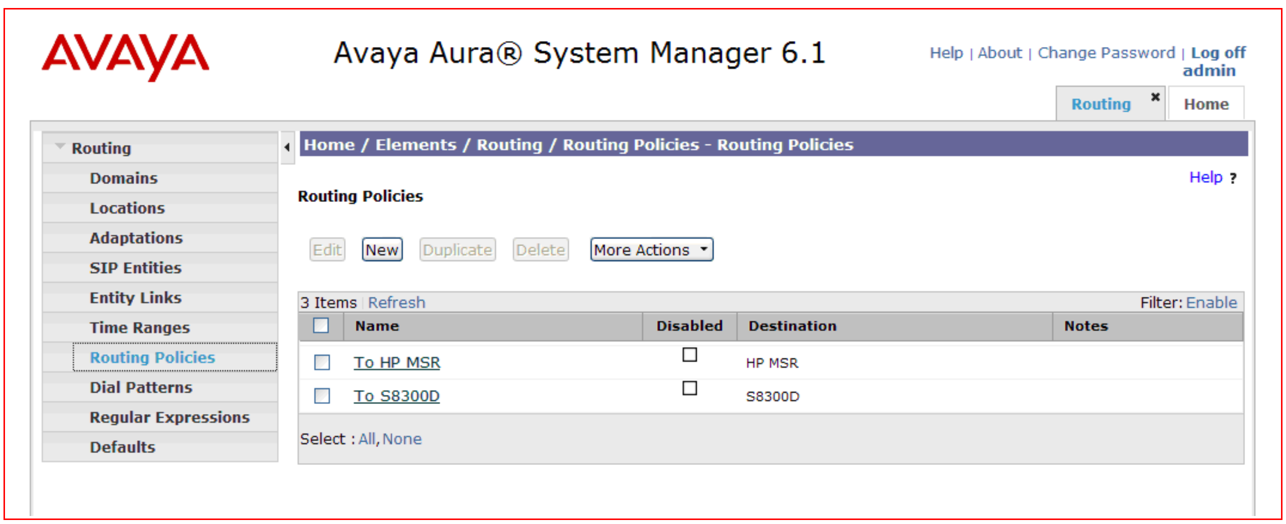
SIP Entity as Destination section

- Click the **Select** button.
- Select a SIP Entity that will be the destination for this call.
- Click the **Select** button and return to the Routing Policy Details form.

Time of Day section

- Leave default values.

Click **Commit** to save Routing Policy definition. Repeat the steps for each new Routing Policy. The following screen shows the Routing Policy used for Communication Manager during the compliance test.



### 5.3.7. Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In the compliance test, the following dial patterns are defined from Session Manager.

- 720 –Endpoints in the Avaya S8300D Server
- 2 – 2xxxx extensions in the simulated PSTN side

To add a Dial Pattern, select **Routing →Dial Patterns** and click on the **New** button (not shown) on the right pane. Provide the following information:

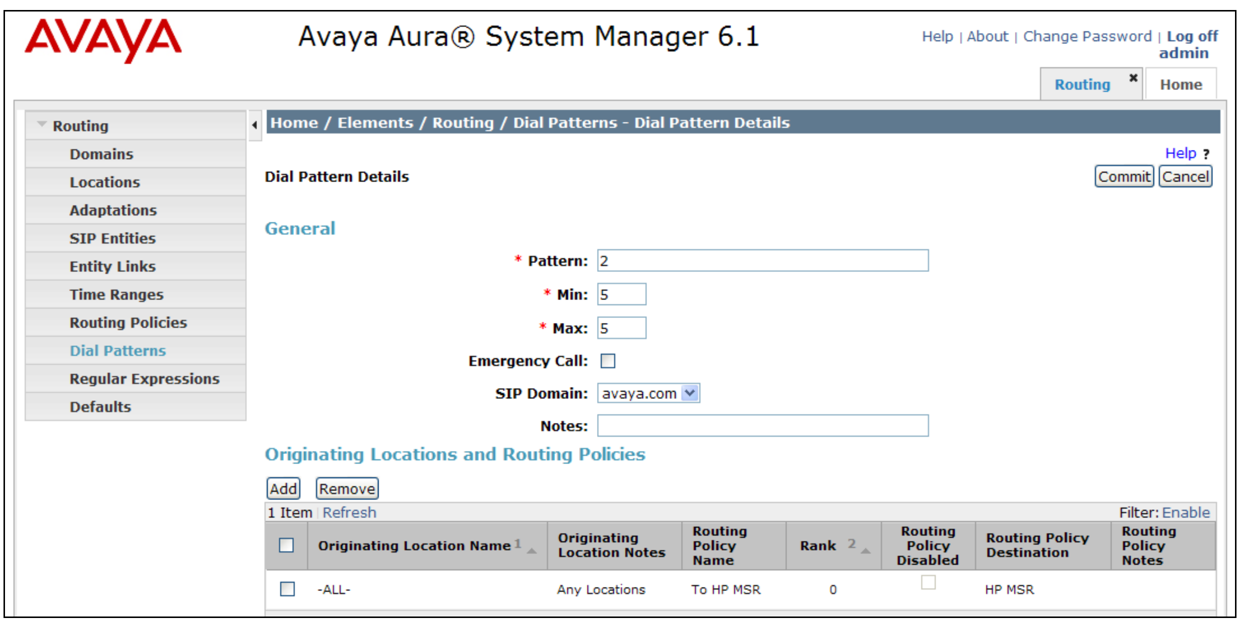
General section

- Enter a unique pattern in the **Pattern** field (e.g. **2**).
- In the **Min** field enter the minimum number of digits (e.g. **5**).
- In the **Max** field enter the maximum number of digits (e.g. **5**).
- In the **SIP Domain** drop down menu select the domain that will be contained in the Request URI *received* by Session Manager from Communication Manager.

Originating Locations and Routing Policies section

- Click on the **Add** button and a window will open (not shown).
- Click on the box for the appropriate Originating Locations, and Routing Policies (see **Section 5.3.6**) that pertain to this Dial Pattern.
  - Select the Originating Location to apply the selected routing policies to **All**.
  - Select Routing Policies to **HP MSR**.
  - Click on the **Select** button and return to the Dial Pattern window.

Click the **Commit** button to save the new definition. The following screen shows the dial pattern used for **2xxxx** during the compliance test. Repeat steps for the remaining Dial Patterns.



### 5.3.8. Configure Managed Elements

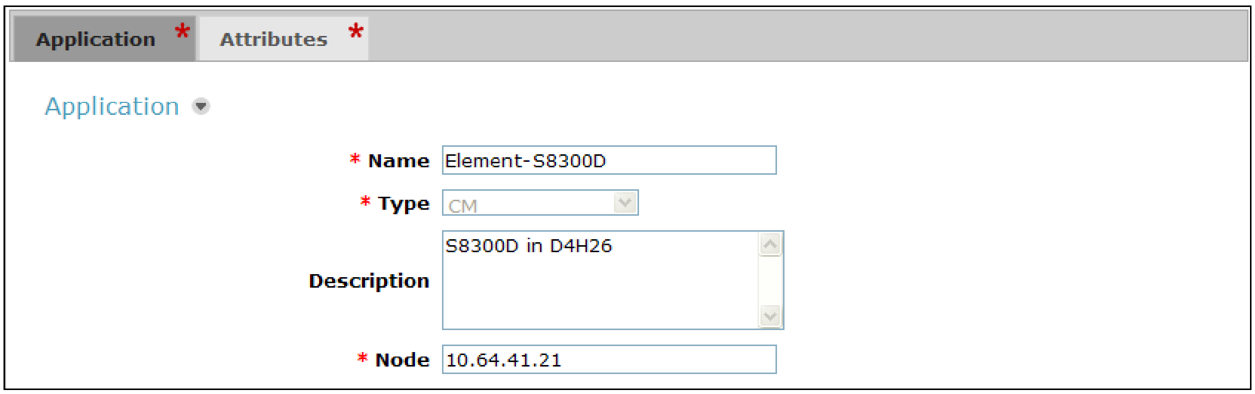
To define a new Managed Element, navigate to **Elements →Inventory→Manage Elements**. Click on the **New** button (not shown) to open the **New Entities Instance** page.

In the **New Entities Instance** Page

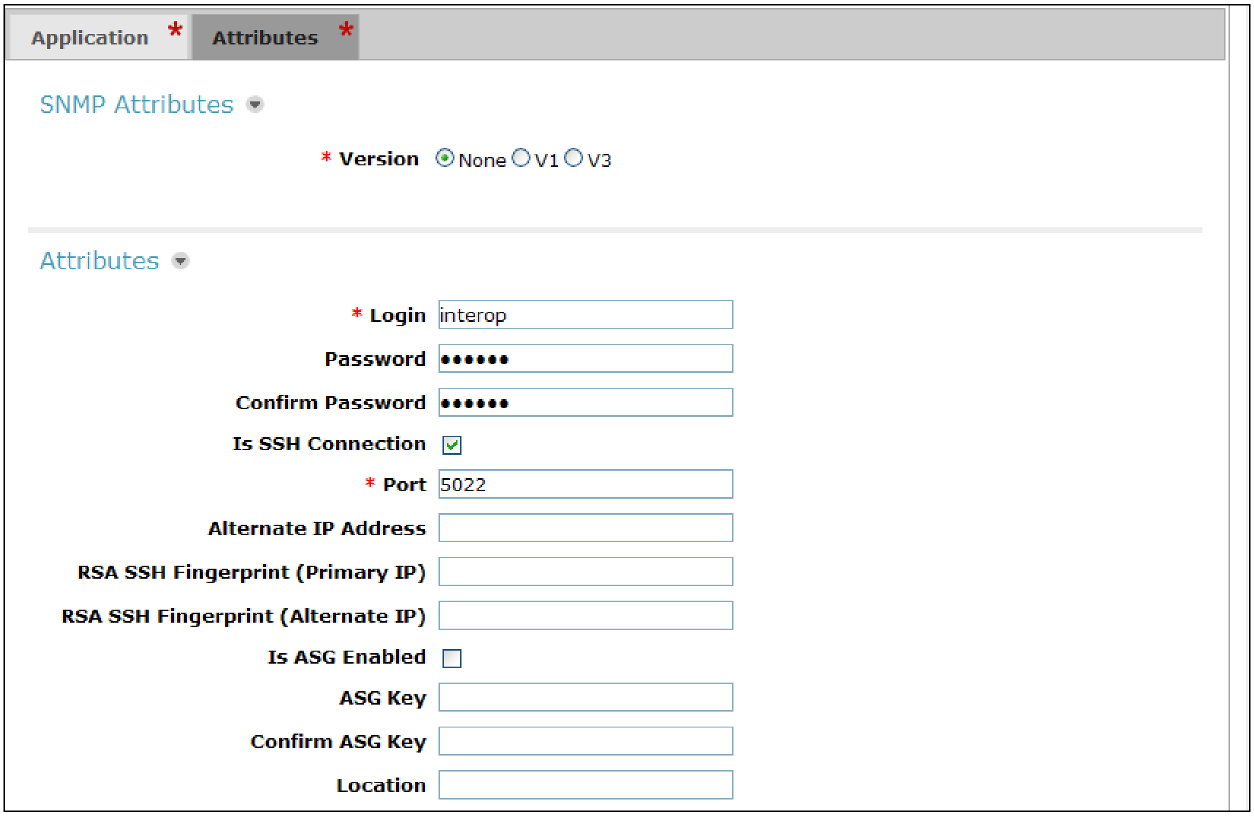
- In the **Type** field, select **CM** using the drop-down menu and the **New CM Instance** page opens (not shown).

In the **New CM Instance Page**, provide the following information:

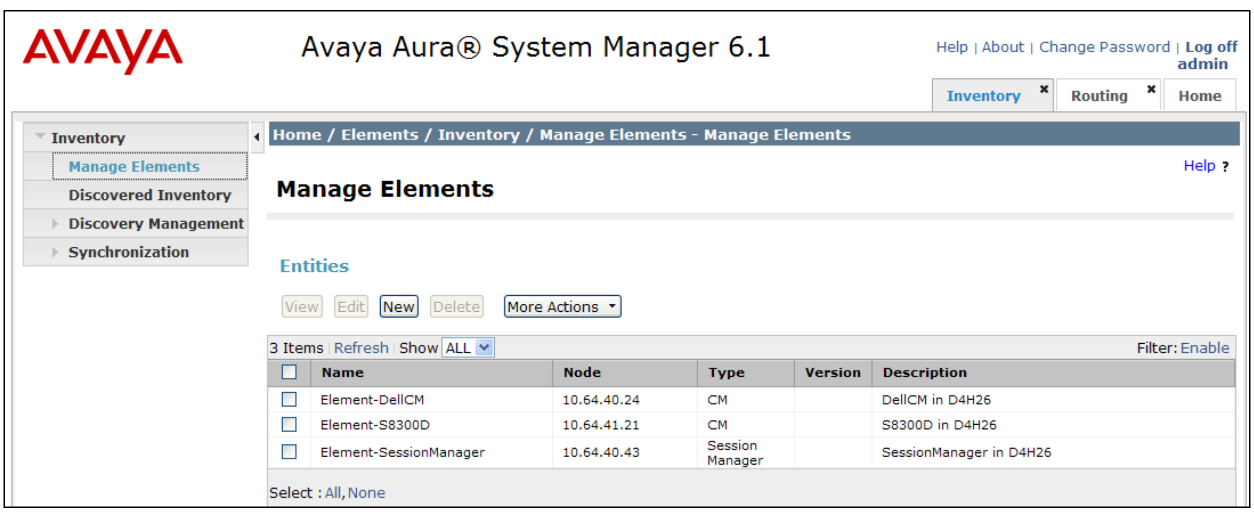
- <u>Application section</u>
  - **Name** – Enter name for Communication Manager (Evolution Server).
  - **Description -** Enter description if desired.
  - **Node** – Enter IP address of the administration interface. During the compliance test, the **procr** IP address (10.64.41.21) was utilized.




- <u>Attributes section</u>
  System Manager uses the information entered in this section to log into Communication Manager using its administration interface. Enter the following values and use default values for remaining fields.
  - **Login** – Enter login used for administration access
  - **Password** – Enter password used for administration access
  - **Confirm Password** – Repeat value entered in above field
  - **Is SSH Connection** – Check the box
  - **Port** – Verify **5022** is set



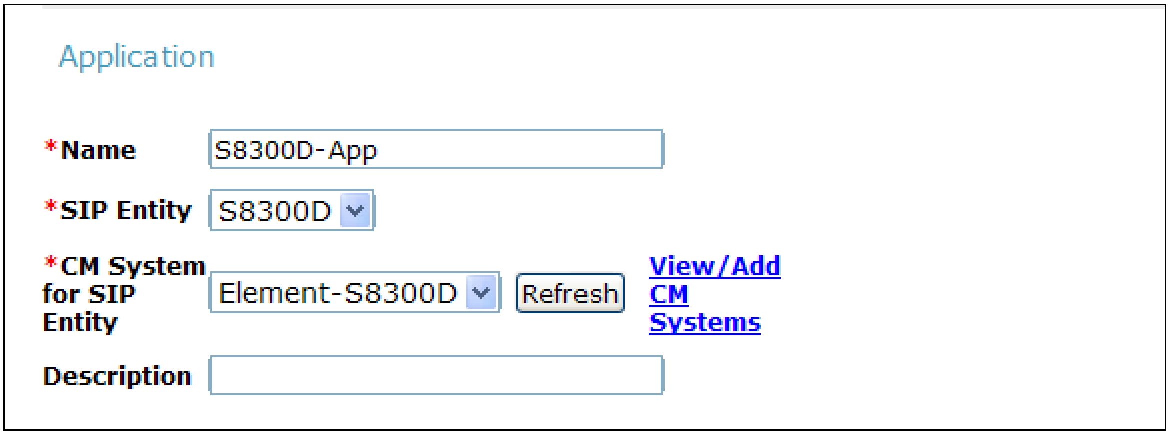

Click **Commit** (not shown) to save the element. The following screen shows the element created, **Element-S8300D**, during the compliance test.




## 5.3.9. Configure Applications

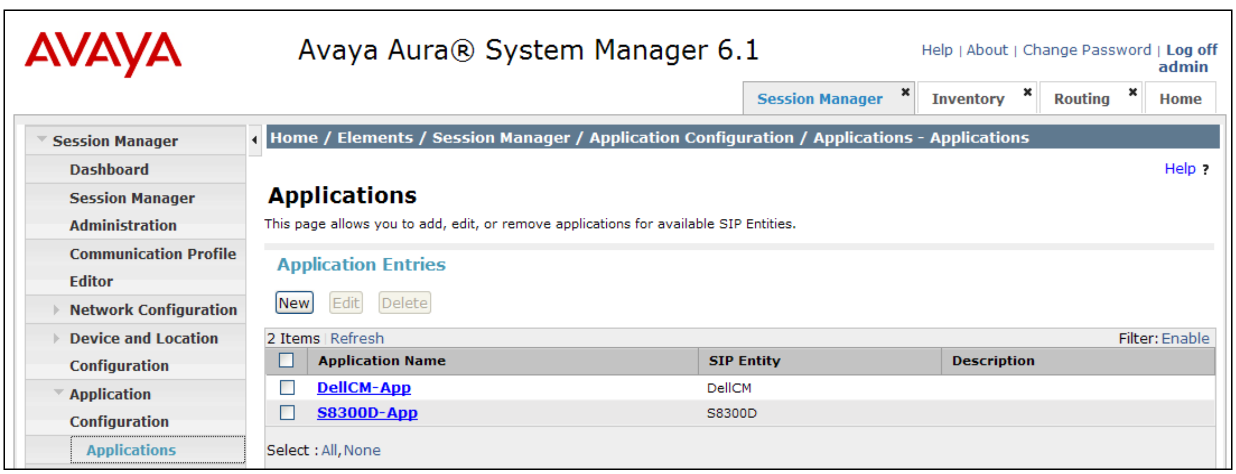
To define a new Application, navigate to **Elements → Session Manager → Application Configuration → Applications**. Click **New** (not shown) to open the Applications Editor page:

- Application Editor section
  - **Name –** Enter name for the application.
  - **SIP Entity**–Select the SIP Entity for Communication Manager defined in **Section 5.3.3.**
  - **CM System for SIP Entity** –Select the name of the Managed Element defined for Communication Manager in **Section 5.3.8.**
  - **Description–** Enter description if desired.




- Leave the fields in the Application Attributes (optional) section blank.

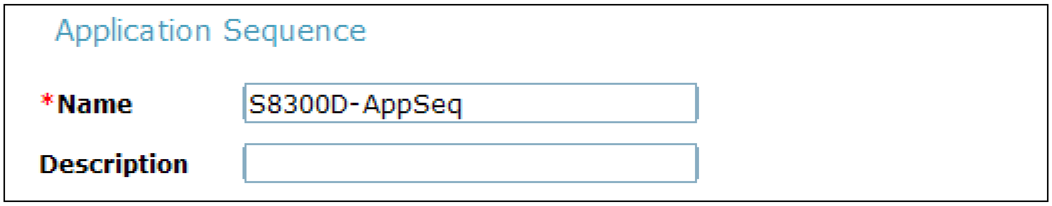
Click the **Commit** button (not shown) to save the Application. The screen below shows the Application, **S8300D-App**, defined for Communication Manager.




## 5.3.10. Define Application Sequence

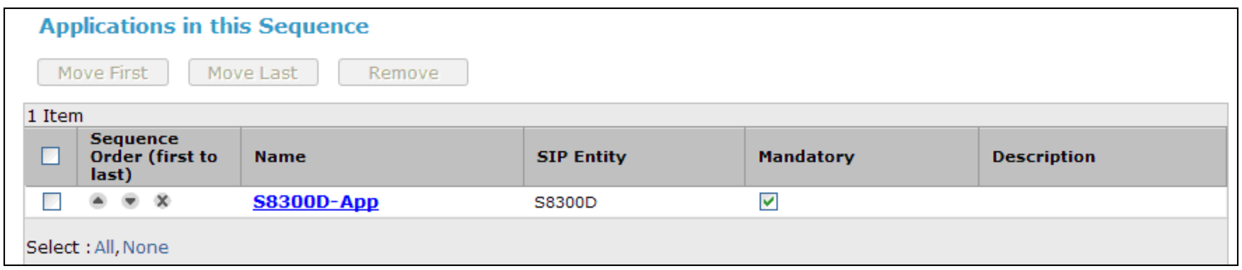
Navigate to **Elements → Session Manager → Application Configuration → Application Sequences**. Click **New** (not shown) and provide the following information:

- Sequence Name section
  - **Name** – The name for the application.
  - **Description** – Enter description, if desired.

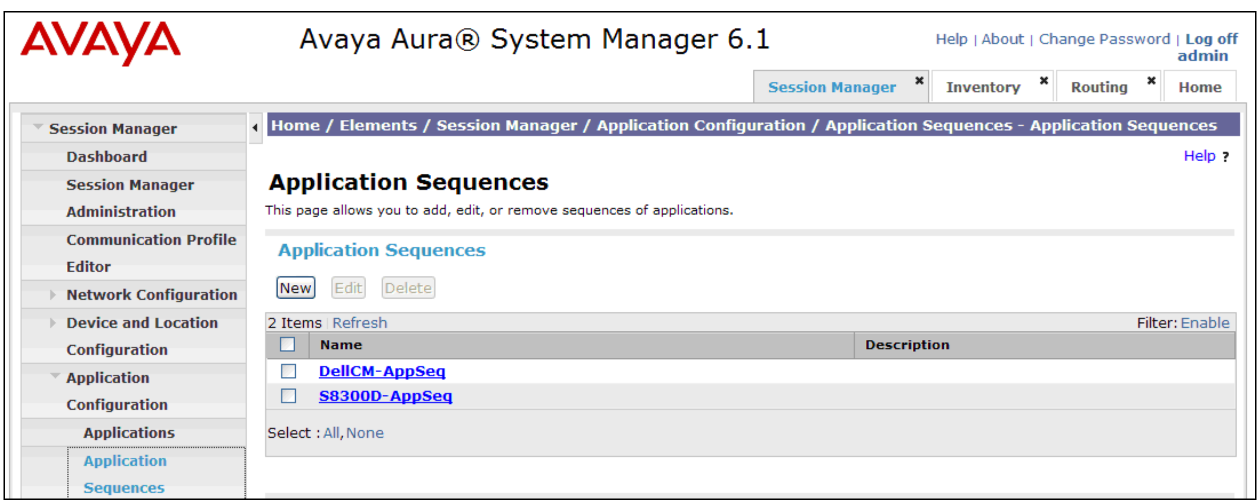



- Available Applications section
  - Click icon associated with the Application for Communication Manager defined in **Section 5.3.9** to select this application.
  - Verify a new entry is added to the Applications in this Sequence table as shown below.

Click the **Commit** button (not shown) to save the new Application Sequence.




The screen below shows the Application Sequence, **S8300D-AppSeq**, defined during the compliance test.



# 6. Configure HP Networking MSR PSTN Gateway

This section describes the configuration for the MSR30 in **Figure 1**. It is assumed that basic configuration has been performed to allow for IP and WebUI connectivity into MSR30. All steps in this section are performed using the WebUI.

Using a web browser, go to **http://<IP address of MSR30>** and log in with the appropriate credentials.


## 6.1. Introduction to the MSR Product Line

This section describes the high level features of the MSR product line. Further details can be found on the HP website.

The HP Multiple Services Router (MSR) Series is a family of modular devices with a full range of models for requirements from small offices to large data centers. The MSR product line delivers high performance, secure, integrated services on a single platform.

The MSR product line enhances network functionality, reduces complexity and simplifies management. The product line includes a variety of chassis that run Comware – the management software. Comware supports the centralized management suite IMC (Intelligent Management Center) and a comprehensive integrated security service. Additional benefits of the MSR Series include:

- Convergence of routing, switching, security and voice
- Modular, multi-bus architecture with high reliability and high performance
- Embedded encryption, quality of service, firewall, security features
- Redundant power supply and hot swapping available on select models
- Unified management platform
- Common modules across many platforms
- Open application architecture enabled
- No extra license cost for features

The HP MSR product line includes the following chassis:

| | |
|---|---|
| MSR50 Series | High performance, reliable, scalable, available in PoE and DC models |
| MSR30 Series | Medium branch office routers available in PoE and DC models. Supports external Redundant Power Supply and 1GB WAN interfaces |
| MSR20 Series | Small branch office routers |
| MSR20-1x Series | Fixed WAN interface connectivity with WLAN models |
| MSR900 Series | Ethernet 10/100 WAN connectivity with WLAN models and 4 or 8 Ethernet ports |

Power supply options for the MSR product line include:

- All MSR routers (except the MSR900) use 100～240V 50/60Hz
- MSR50 Series supports optional built-in dual Power Supply Unit and hot-swap
- MSR30 Series (starting with MSR30-16-POE) have a connector to a redundant power supply
- MSR900 Series routers use 12V input

MSR30-16, -20, -40, -60 Series and MSR50 Series support PoE:

- Standard 802.3af PoE Ethernet Switching interfaces
- MSR50 Series requires a PoE module
- MSR30 Series requires separately ordered PoE chassis

The MSR Series has a modular N-Bus architecture that provides high performance and flexibility that support a variety of interchangeable modules:

- WAN-data including MPLS, DSL, Cellular-3G, ATM, SONET, OC-3 POS, OC-3/T3/E3 and others
- Voice (T1, E1, J1, BRI)
- VPN
- Wi-Fi Access Point (b/g)
- Additional Ethernet ports
- Analog modems

Finally, the MSR product line includes features such as:

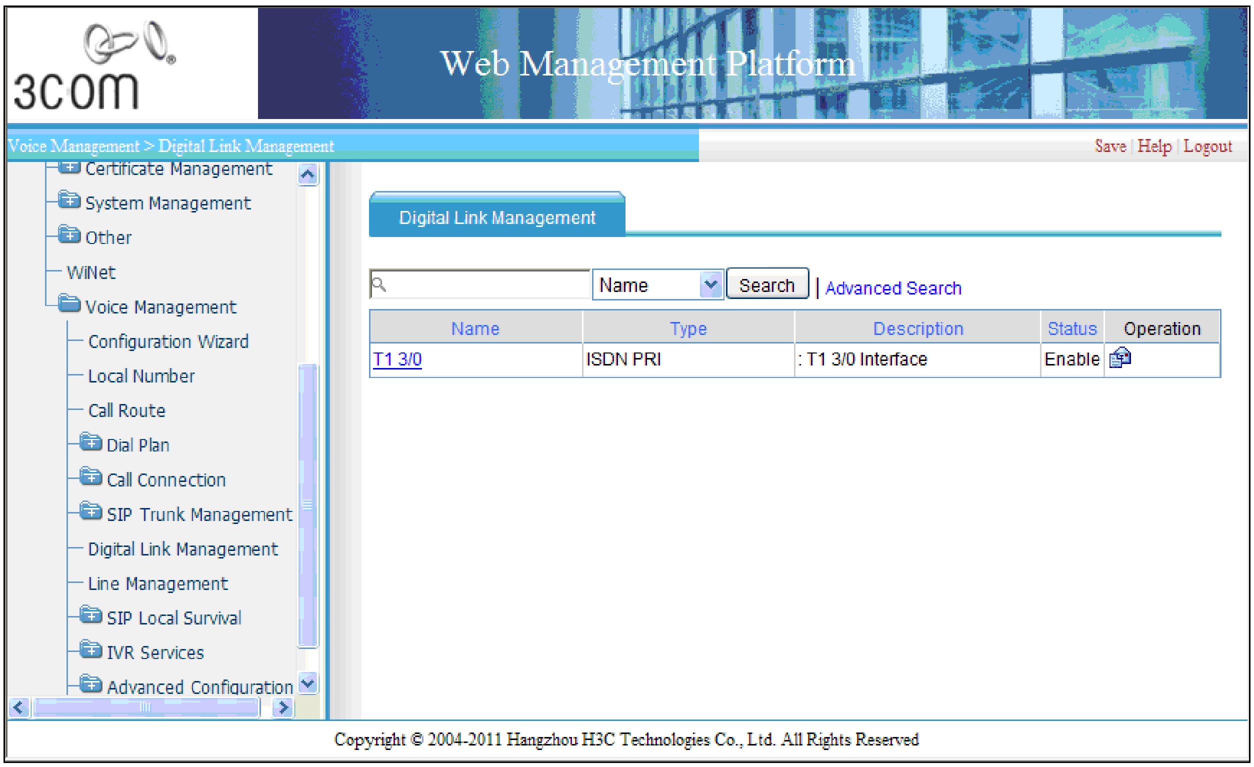
- Standards based routing, switching and wireless access
- Support for IPv4/IPv6, RIP, MPLS, IS-IS, BGP, OSPF, L2TP, GRE, PPP
- Support for IPSec, SSL, VPN, status based ASPF Firewall
- Power over Ethernet
- Support for QoS, security and VLANs

### 6.1.1. Introduction to the MSR30

The MSR30 Series are targeted for enterprise branch office or small to medium business core router applications. MSR30s have higher performance than the MSR20s and support both SIC and MIM interface modules.  The MSR 30 series has 6 different models for various density and scalability requirements and has options for DC power supply and Power over Ethernet on some models.

## 6.2. Configure T1/PRI

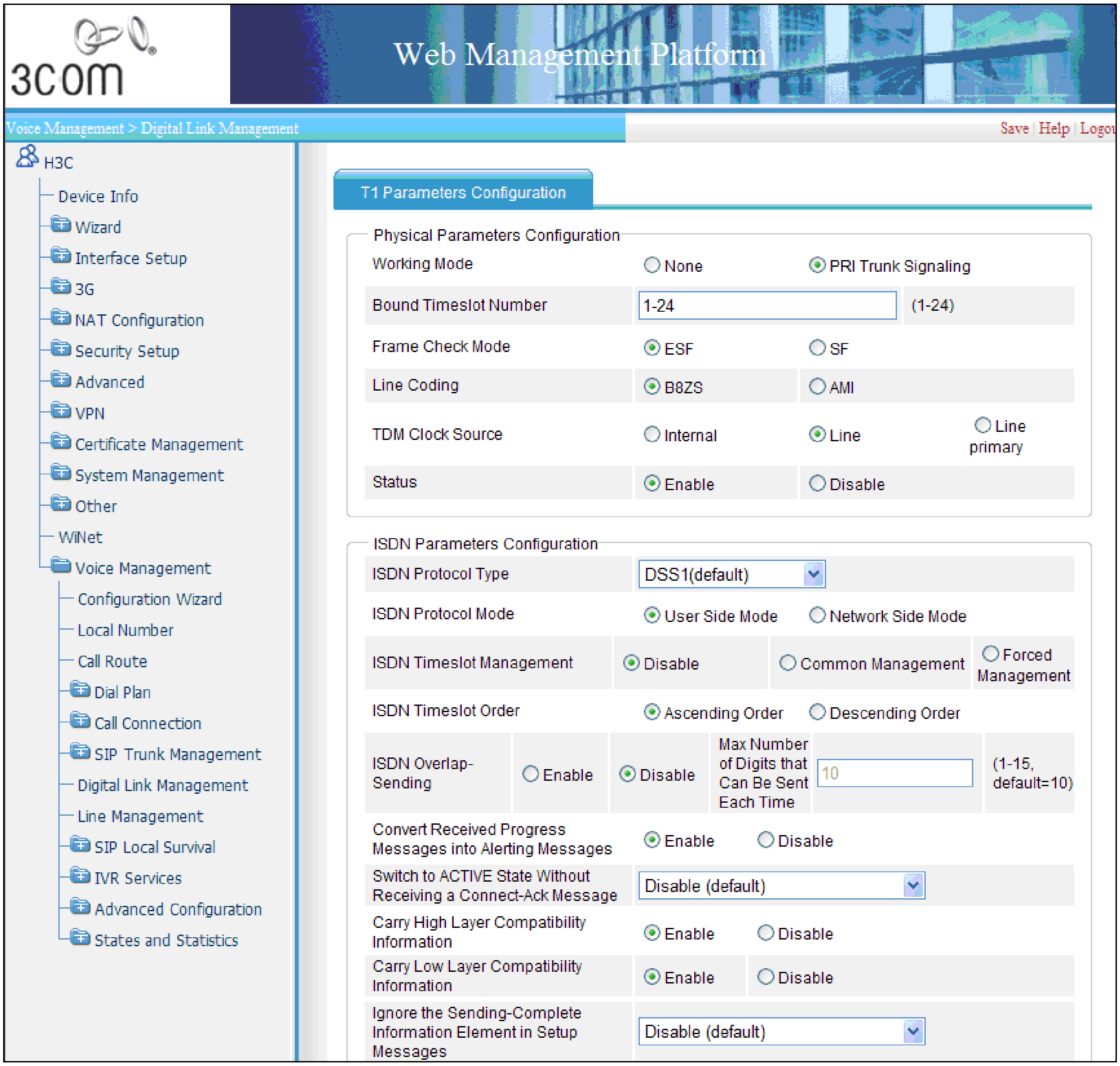
Navigate to **Voice Management→Digital Link Management** and click the icon under Operation.




In the T1 Parameters Configuration page, provide the following information:

- **Working Mode** – Select **PRI Trunk Signaling**
- **Bound Timeslot Number** – Enter **1-24**
- **Frame Check Node** – Select **ESF**
- **Line Coding** – Select **B8ZS**
- **ISDN Protocol Mode** – Select **User Side Mode**

Click the **Apply** button at the bottom (Not shown)


Web Management Platform
Voice Management > Digital Link Management
Save | Help | Logou
H3C
Device Info
Wizard
Interface Setup
3G
NAT Configuration
Security Setup
Advanced
VPN
Certificate Management
System Management
Other
WiNet
Voice Management
Configuration Wizard
Local Number
Call Route
Dial Plan
Call Connection
SIP Trunk Management
Digital Link Management
Line Management
SIP Local Survival
IVR Services
Advanced Configuration
States and Statistics
T1 Parameters Configuration
Physical Parameters Configuration
Working Mode
None
PRI Trunk Signaling
Bound Timeslot Number
1-24
(1-24)
Frame Check Mode
ESF
SF
Line Coding
B8ZS
AMI
TDM Clock Source
Internal
Line
Line primary
Status
Enable
Disable
ISDN Parameters Configuration
ISDN Protocol Type
DSS1(default)
ISDN Protocol Mode
User Side Mode
Network Side Mode
ISDN Timeslot Management
Disable
Common Management
Forced Management
ISDN Timeslot Order
Ascending Order
Descending Order
ISDN Overlap-Sending
Enable
Disable
Max Number of Digits that Can Be Sent Each Time
10
(1-15, default=10)
Convert Received Progress Messages into Alerting Messages
Enable
Disable
Switch to ACTIVE State Without Receiving a Connect-Ack Message
Disable (default)
Carry High Layer Compatibility Information
Enable
Disable
Carry Low Layer Compatibility Information
Enable
Disable
Ignore the Sending-Complete Information Element in Setup Messages
Disable (default)

## 6.3. Configure Call Route

Two routes are used during the compliance test:

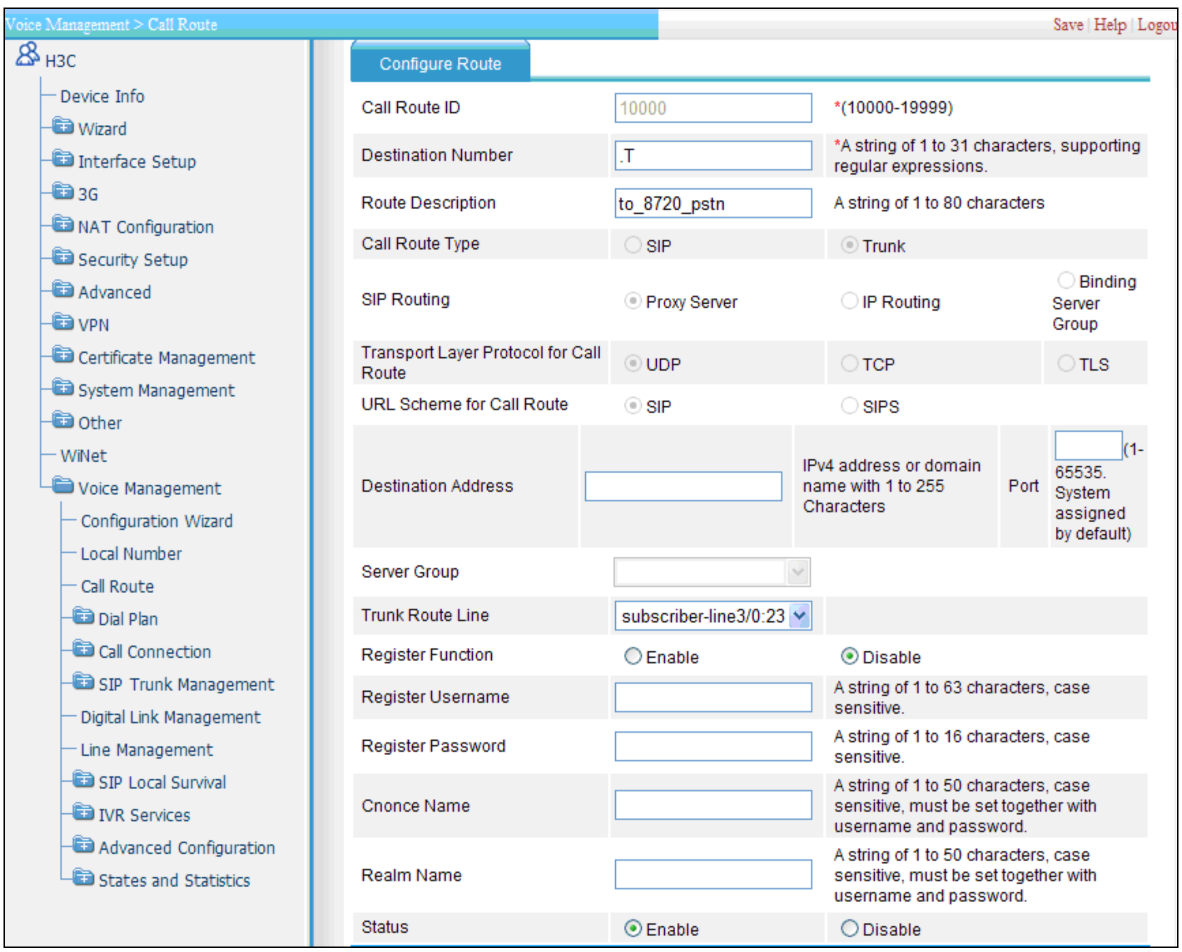
- Route to PSTN
- Route to Session Manager using a SIP trunk

### 6.3.1. Configure route to PSTN

Navigate to **Voice Management→Call Route**. To add a route, click the **Add** button (not shown) and provide the following information:

- **Call Route ID** – Enter a value within the ID range.
- **Destination Number** – Enter **.T**. Any extension other than specified in the call route will use this call route.
- **Route Description** – Enter a descriptive name.
- **Call Route Type** – Select **Trunk**.
- **Trunk Route Line** – Using the drop down list, select **subscriber-line3/0-23**.
- **Register Function** – Select **Disable**.

Click the **Apply** button.

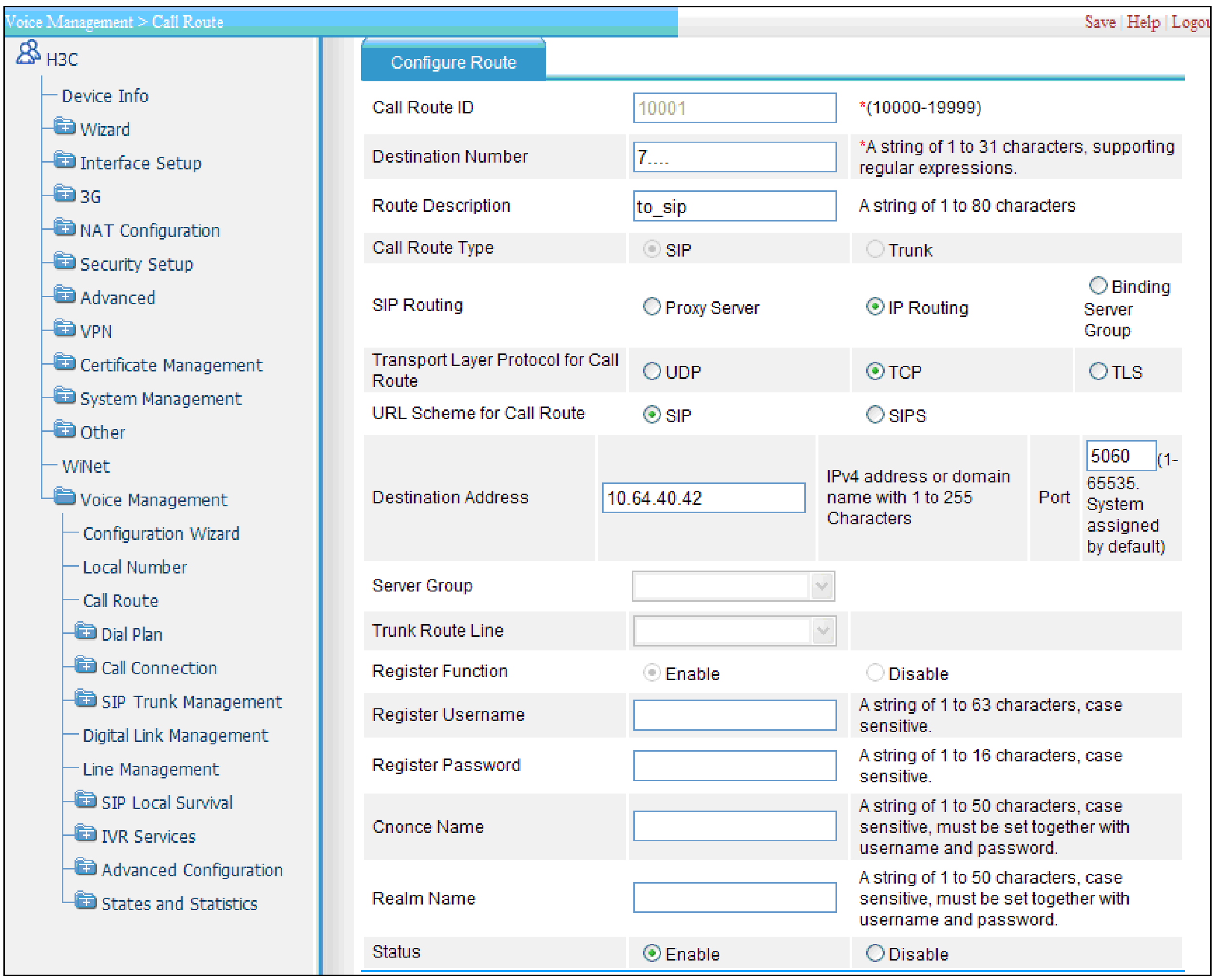


### 6.3.2. Configure route to Session Manager

Navigate to **Voice Management→Call Route**. To add a route, click the **Add** button (not shown) and provide the following information:

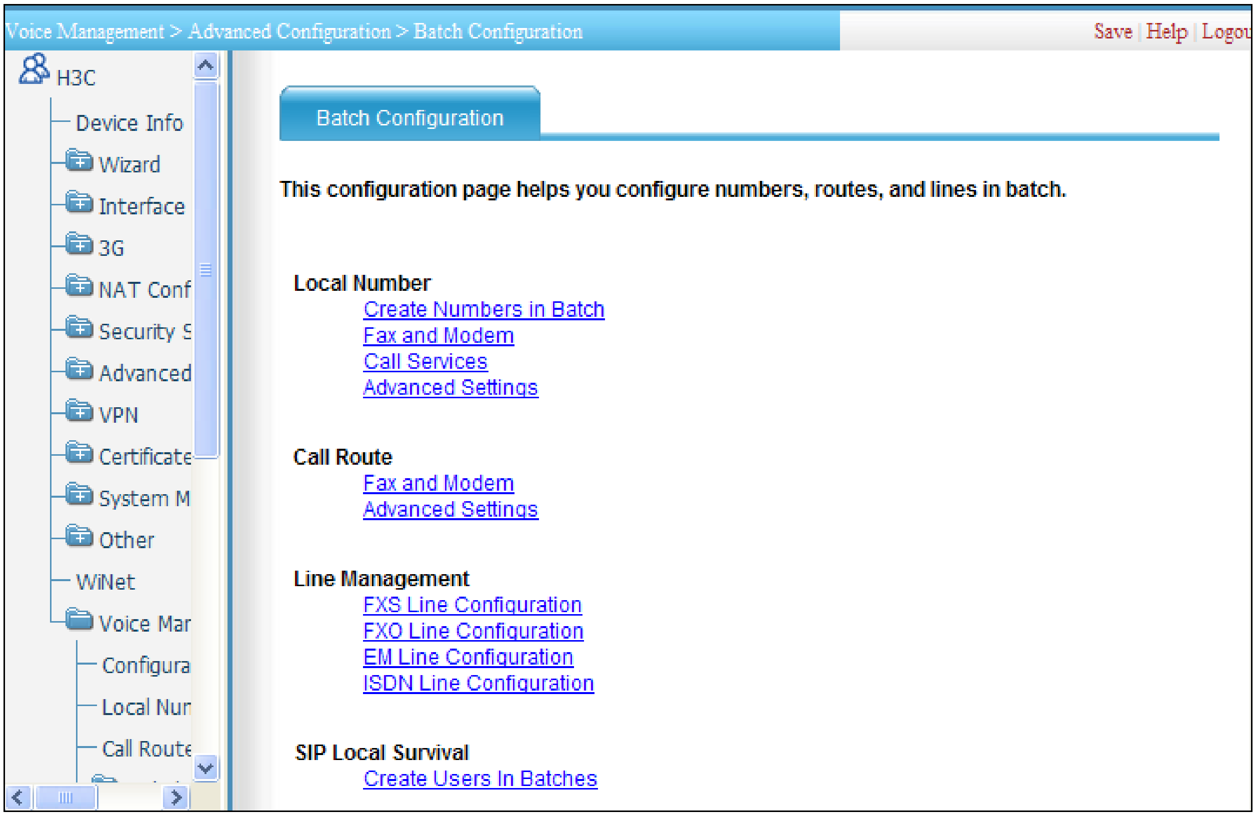
- **Call Route ID** – Enter a value within the ID range.
- **Destination Number** – Enter **7….** Any 5 digit extension starting with 7 will use this route.
- **Route Description** – Enter a descriptive name.
- **Call Route Type** – Select **SIP**.
- **SIP Routing** – Select **IP Routing**.
- **Transport Layer Protocol for Call Route** – Select **TCP**.
- **Destination Address** – Enter the Session Manager Security Module IP address.

Click the **Apply** button.



## 6.4. Configure FXS Line

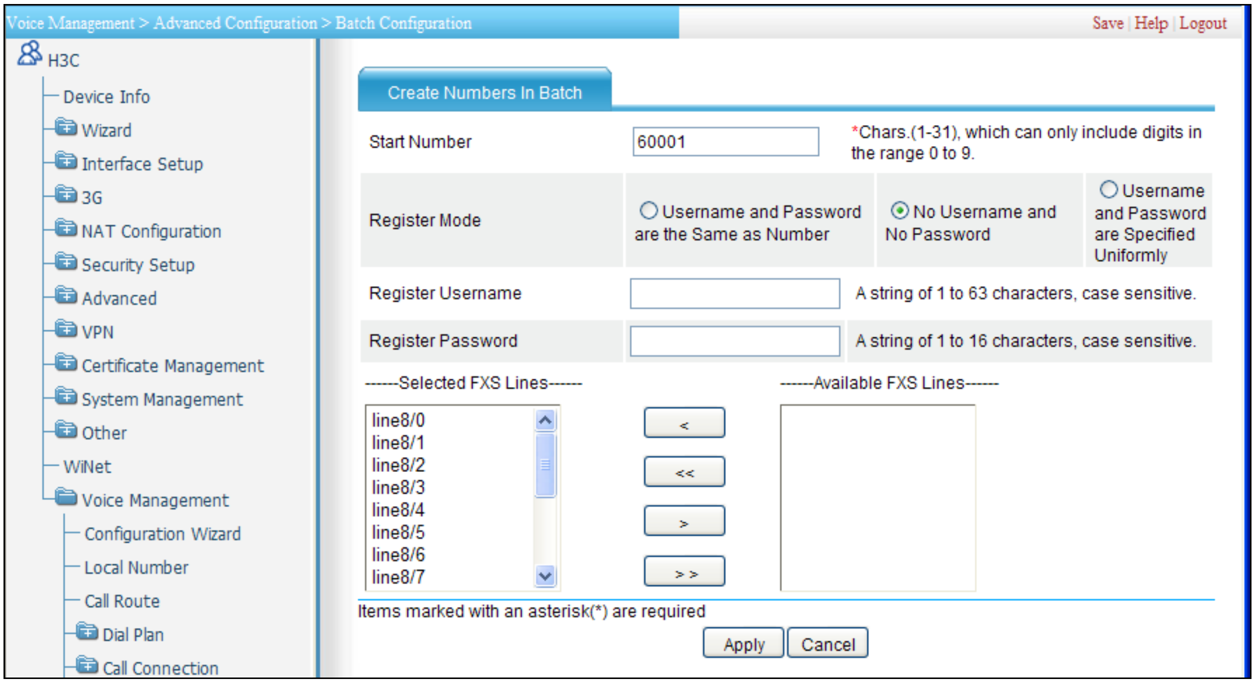
Navigate to **Voice Management →Advanced Configuration→ Batch Configuration**. Select **Create Numbers in Batch** under the Local Number section.




Provide the following information:

- **Start Number** – Enter a unique extension.
- **Register Mode** – Select **No Username and No Password**.
- Move FXS lines from the **Available FXS Lines** folder to the **Selected FXS Lines folder**.

Click the **Apply** button.




## 6.5. Configure FXO Line

In the MSR30, the FXO Subscriber Line 1/0[1] was connected to an FXS Analog circuit pack port (1A1101) in the Avaya G650 Media Gateway. The port **1A1101** is configured as type**2500** and used extension **22006**. All other values are default.

```
change station 22006                                           Page   1 of   4
                                     STATION

Extension: 22006                         Lock Messages? n               BCC: 0
Type: 2500                          Security Code:                   TN: 1
Port: 01A1101                     Coverage Path 1:                  COR: 1
     Name: FXS-1                         Coverage Path 2:                  COS: 1
                                       Hunt-to Station:                 Tests? y

STATION OPTIONS
      XOIP Endpoint type: auto                 Time of Day Lock Table:
              Loss Group: 1              Message Waiting Indicator: none
    Off Premises Station? n



          Survivable COR: internal
   Survivable Trunk Dest? y
```

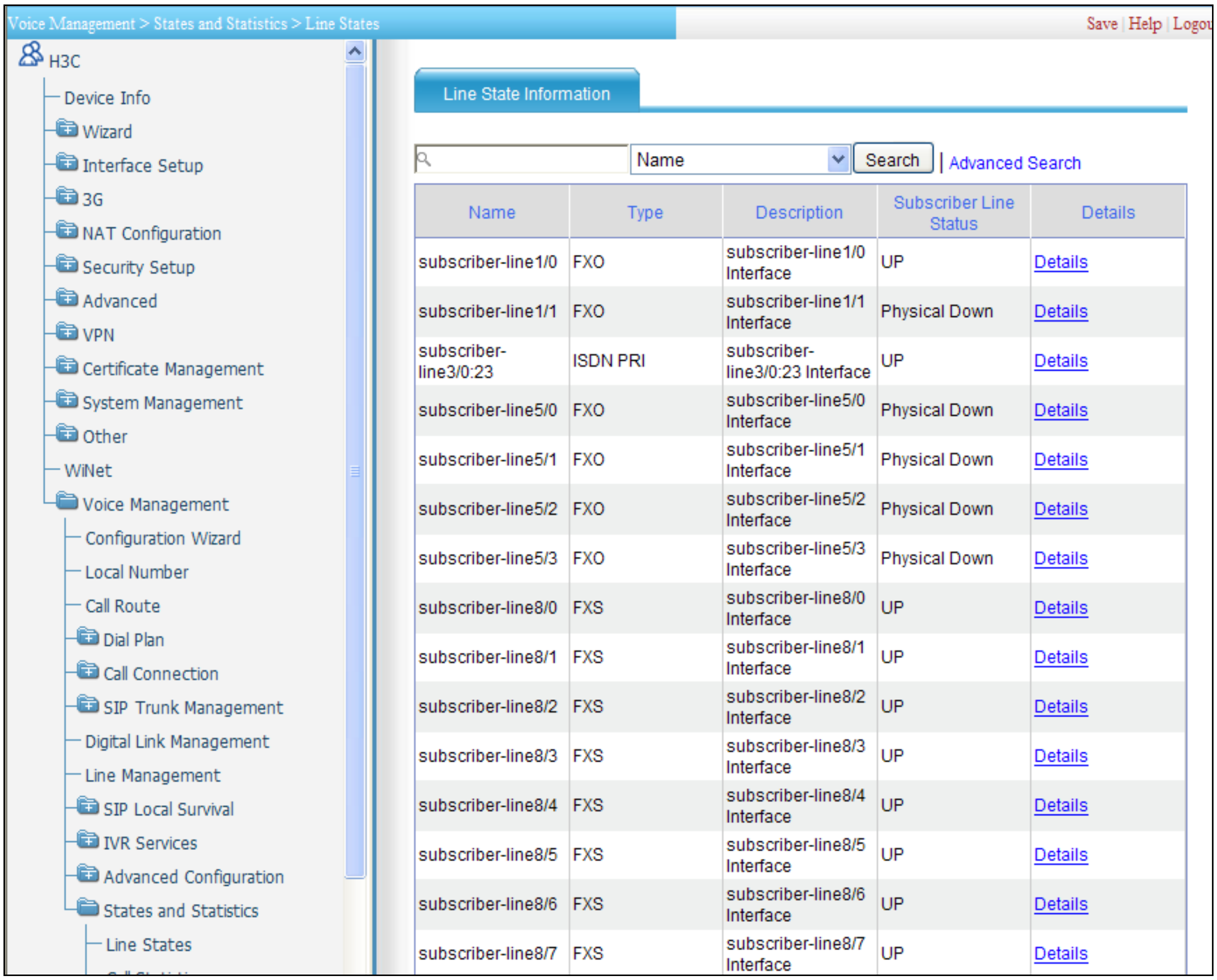
The following is the test scenario:

> When a call comes into extension 22006, the analog (FXS) port sends battery current and ring voltage to the MSR30's FXO port. The FXO port then goes

[1] The MSR uses the following convention to identify ports: <Module Identifier>/<Port Identifier>. So Subscriber Line 1/0 refers to the module in slot 1 of the chassis and port 0 on the chassis. Users can refer to the HP chassis and module documentation (see References) in order to locate the physical port.

off-hook (loop closure). When the FXO port goes off-hook, it starts to listen for DTMF digits. After a pause, the FXS port starts sending the dialed extension as DTMF digits. The MSR30 utilizes call route table to connect to the destination's extension.

To check the status of all lines, navigate to **Voice Management →States and Statistics →Line States**.



# 7. Verification Steps

## 7.1. Capturing the MSR30's configuration and call debug logs

Use the following commands to document the current MSR30 configuration and develop data for troubleshooting:

- Access the MSR's command line
- Using your terminal application, log all information to a file. (Or, be prepared to copy a lot of data in the screen buffer)
- Type "**screen-length disable**" to disable paging
- Type "**display current-configuration**" and "**display voice entity all**" to capture the current configuration
- Type "**debug voice sip all**" to capture log information
- Enable sending debug messages to console, type: "**terminal debugging**" and "**terminal monitor**"
- Confirm that debug is on by typing: "**display debug**"… and you'll see that several SIP switches are on
- Place a call
- Capture all log data printed to the console window
- To return to normal operating mode, type "**undo debug voice sip all**"
- Re-Enable paging: "**undo screen-length disable**"

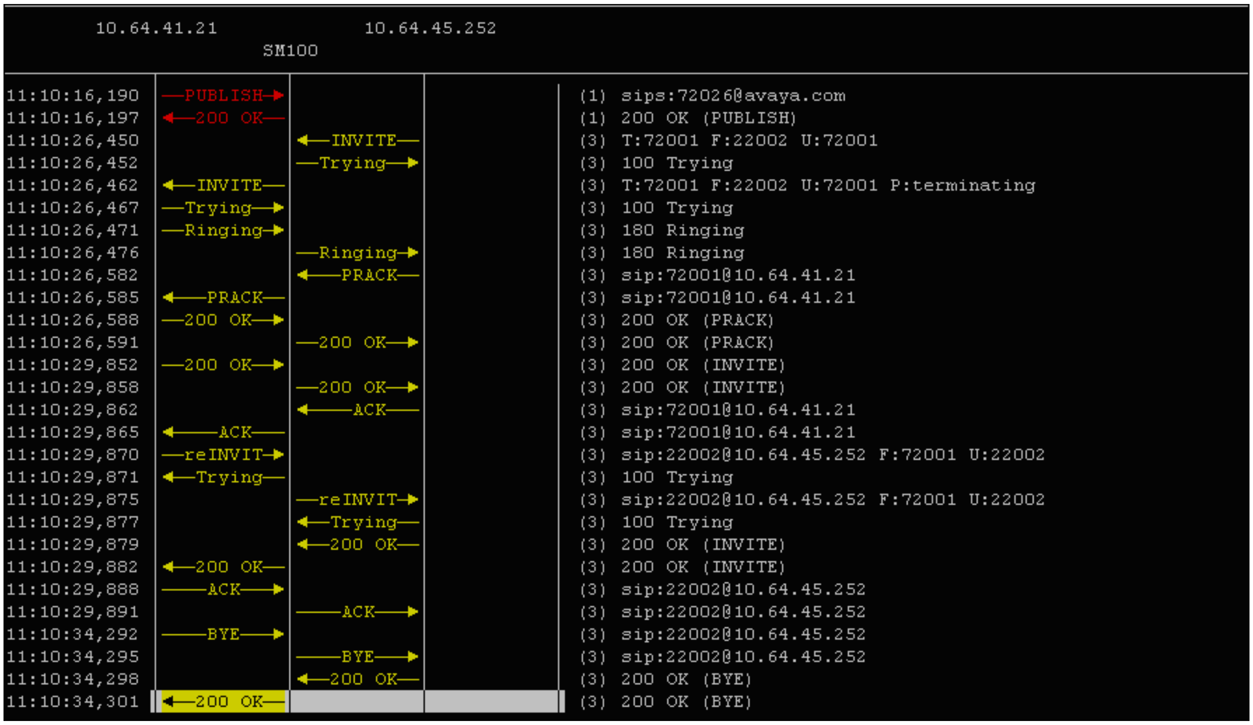
## 7.2. Verification from Avaya Aura® Communication Manager on PSTN side

The following commands can be used to check the status of the T1/PRI trunk:

- "**status trunk xx**", where xx is a trunk group number. Ensure the trunk is **up**.
- "**list trace tac yy**", where yy is a unique trunk access code.

## 7.3. Verification from Avaya Aura® Session Manager on Enterprise side

During the compliance test, the **traceSM** tool in Session Manager was utilized to capture the SIP signaling between Communication Manager (10.64.41.21), Session Manager (SM100) and the MSR30 (10.64.45.252). The following is a sample of the trace tool:




# 8. Conclusion

These Application Notes describe the procedures required to configure an HP Networking Multi-Service Router 30 Series PSTN Gateway in an Avaya Telephony environment. The HP Networking Multi-Service Router 30 Series PSTN Gateway successfully passed compliance testing.

| Anomalies |
| --- |
| The MSR30 does not support PSTN-side hold. |

# 9. Additional References

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

[1] *Administering Avaya Aura™ Communication Manager*, Release 6.0, June 2010, Issue 6.0,Document Number 03-300509
[2] *Administering Avaya Aura® Session Manager*, Release 6.1, November 2010, Issue 1.1, Document Number03-603324
[3] *Administering Avaya Aura® System Manager*, Release 6.1, November 2010

Product documentation for HP products may be found at http://www.hp.com/networking
[4] *MSR Series Routers Web-Based Configuration Guide-Release 2207(V1.05)*http://h3c.com/portal/Technical_Support___Documents/Technical_Documents/Routers/H3C_MSR_50_Series_Routers/Configuration/User_Manual/H3C_MSR_WCG-Release_2207(V1.05)
[5] *MSR Series Routers Interface Module Manual(V1.07)*http://h3c.com/portal/Technical_Support___Documents/Technical_Documents/Routers/H3C_MSR_50_Series_Routers/Installation/Installation_Manual/H3C_MSR_IMM(V1.07)
[6] *MSR 30 Routers Installation Guide(V1.05)*http://h3c.com/portal/Technical_Support___Documents/Technical_Documents/Routers/H3C_MSR_30_Series_Routers/Installation/Installation_Manual/H3C_MSR_30_Routers_IG(V1.05)

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.