Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.2 as an Evolution Server, Avaya Aura® Session Manager R6.2 and Acme Packet Net-Net 3820 SBC to support Cable and Wireless SIP IP Trunking Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Cable and Wireless SIP IP Trunking service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Acme Packet Net-Net 3820, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. Cable and Wireless is a member of the DevConnect Service Provider program.

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.

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1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Cable and Wireless SIP IP Trunking service and an Avaya SIP-enabled Enterprise Solution. The Avaya solution consists of Acme Packet Net-Net 3820, Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with the Cable and Wireless SIP IP Trunking service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and Acme Packet Net-Net 3820 SBC. The enterprise site was configured to use the SIP IP Trunking service provided by Cable and Wireless.

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 general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and Acme Packet Net-Net 3820. The enterprise site was configured to use the SIP IP Trunking service provided by Cable and Wireless. The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DDI numbers assigned by Cable and Wireless. The calls were made to H.323, SIP and analogue telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via Cable and Wireless to PSTN destinations. The calls were made from H.323, SIP and analogue telephones.
- Calls using G.711A and G.729A codec's.
- DTMF transmission using RFC 2833 with successful Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media (also known as "shuffling") with SIP and H.323 telephones, and the Avaya Desktop Video Device (Avaya DVD) running Flare Experience.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by Cable and Wireless requiring Avaya response and sent by Avaya requiring Cable and Wireless response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Cable and Wireless SIP IP Trunking service with the following observations:

- No inbound toll free numbers were tested as none were available from the Service Provider.
- No Emergency Services numbers tested as test calls to these numbers should be pre-arranged with the Operator.
- When Calling Line Identity (CLI) is restricted in the network and delivered to the enterprise, the Privacy header is not included in the INVITE.
- No ring-back was heard by the caller on calls from the PSTN forwarded unconditionally to another PSTN number. A workaround is required in the form of a HMR on the Acme Packet SBC to provide ring-back to the caller. This workaround will be required until a permanent resolution is implemented in the network.
- The conferencing of an inbound PSTN call to internal extensions is limited in that the incoming call is dropped when a fifth extension is added to the conference.
- The conferencing of an outbound PSTN call to internal extensions is limited in that the outgoing call is dropped when a fifth extension is added to the conference. With the workaround in place to provide ring-back for calls forwarded to the PSTN, this reduced such that the outgoing call is dropped when a fourth extension is added to the conference.
- The conferencing of an outbound PSTN call to additional PSTN destinations is limited in that the outgoing call is dropped when the second additional PSTN destination is added to the conference. This occurs only with the workaround in place to provide ringback for calls forwarded to the PSTN.
- T.38 Fax is not supported.
- Network Call Redirect using SIP 302 Moved Temporarily is acknowledged but the call is not routed meaning this feature can't be considered to be supported.
- When Network Call Redirect was invoked using SIP REFER, Communication Manager did not react to NOTIFY message from the network indicating that the destination was busy as it was not configured to do so. This is a Communication Manager configuration issue.
- When the number of members assigned to the SIP Trunk Group in the Communication Manager is exceeded, a SIP 500 "Service Unavailable" message is received in the network. The network re-attempts the call a number of times so that there is a delay before the caller gets an indication of failure.
- When the signalling link between the Communication Manager and the Session Manager is unavailable, a SIP 500 "Server Link Monitor Status Down" message is received in the network. The network re-attempts the call a number of times so that there is a delay before the caller gets an indication of failure.

2.3. Support

For technical support on Cable and Wireless products please use the following web link. http://www.cw.com/contact-us/.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Cable and Wireless SIP IP Trunking service. Located at the Enterprise site is an Acme Packet Net-Net 3820, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), Avaya 16xx series IP telephones (with H.323 firmware) Avaya A175 Desktop Video Device running Flare Experience, Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone running on a laptop PC configured for H.323.

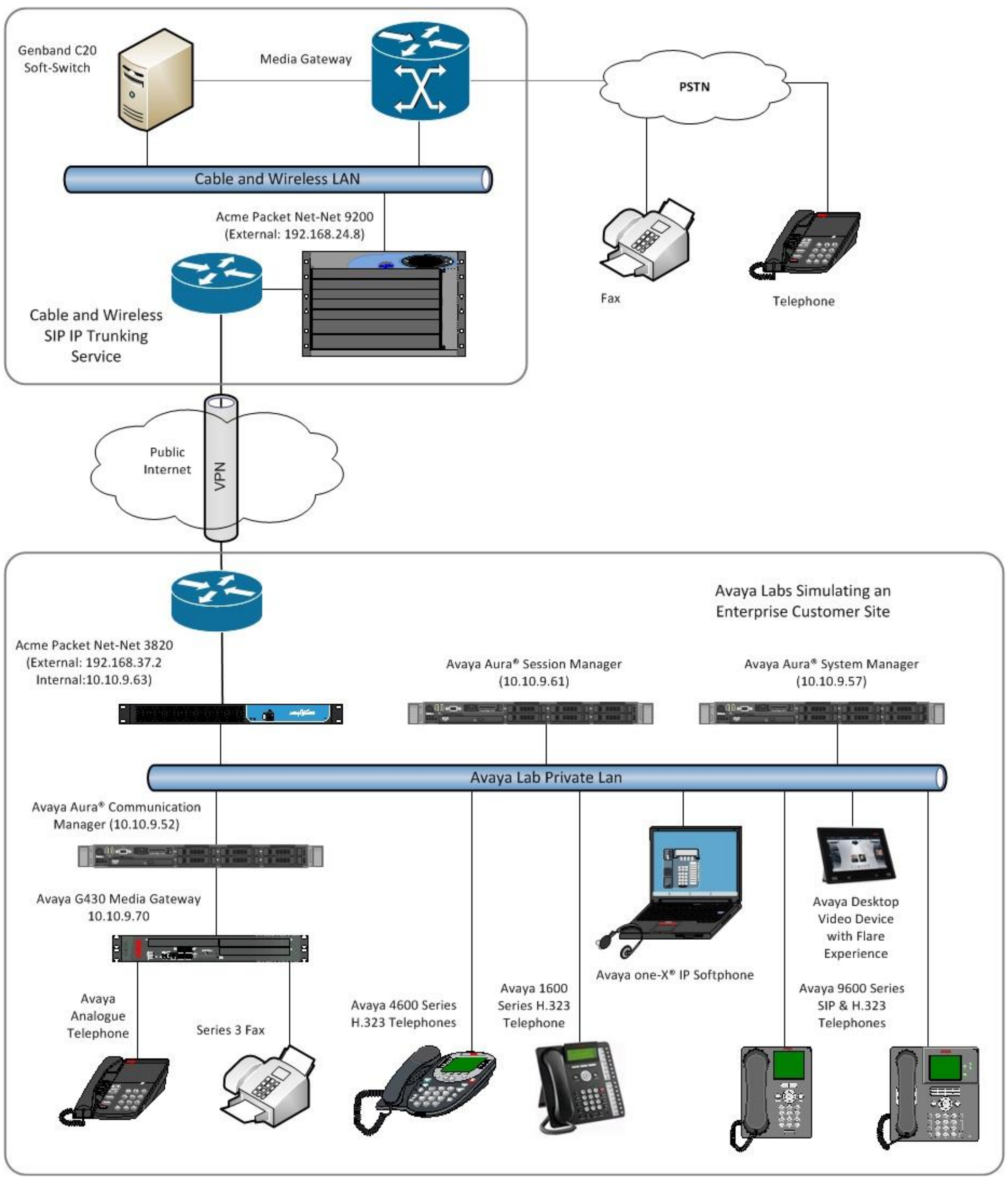


Figure 1: Test Set-up Cable and Wireless SIP IP Trunking to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Communication Manager running on Avaya S8800 Server	R6.2 Build R016x.02.0.823.0
Avaya G430 Media Gateway	FW 30.12.1
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.2 Build 6.2.0.0.620110
Avaya Aura® System Manager running on Avaya S8800 Server	R6.2 (System Platform 6.2.0.0.27, Template 6.2.12.0)
Acme Packet Net-Net 3820 SBC	SCX6.2.0 MR-11 Patch 4 (Build 1109) Build Date 08/11/12
Avaya 1616 Phone (H.323)	1.301
Avaya 4621 Phone (H.323)	2.902
Avaya 9630 Phone (H.323)	3.103
Avaya A175 Desktop Video Device (SIP)	Flare Experience Release 1.1
Avaya 9630 Phone (SIP)	R2.6 SP6
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.1.3.08-SP3-Patch2-35791
Analogue Phone	N/A
Cable and Wireless	
ACME Packet Net-Net 9200 SBC	nnSD700m11
Genband C20 Soft-Switch	CVM13 (12.0.12)

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signalling associated with the Cable and Wireless SIP IP Trunking service. For incoming calls, the Session Manager receives SIP messages from the Acme Packet Net-Net 3820 SBC and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Acme Packet Net-Net 3820 at the enterprise site that then sends the SIP messages to the Cable and Wireless network. Communication Manager Configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. The general installation of the Avaya S8800 Servers and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Cable and Wireless network, and any other SIP trunks used.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 20
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   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
```

On **Page 4**, verify that **IP Trunks** field is set to **y.**

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                        ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? y              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y              Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **SM100** and **10.10.9.61** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
SM100               10.10.9.61
Sipera-SBC          10.10.9.71
default             0.0.0.0
procr               10.10.9.52
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra**- and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled, the media stream is established directly between the enterprise end-point and the internal media interface of the Acme Packet SBC.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.

```
change ip-network-region 1                                          Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: default
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
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
```

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the IP Network Region form in **Section 5.3.** Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec's supported by Cable and Wireless were configured, namely **G.729A**, and **G.711A**.

```
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.729A            n          2        20
 2: G.711A            n          2        20
 3:
```

The Cable and Wireless SIP IP Trunking service does not currently support T.38 for transmission of fax. Although not supported as a standard configuration by Avaya, G.711 pass-through was tested. To set G.711 pass-through, navigate to **Page 2** and configure by setting the **Fax Mode** to **off** as shown below.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



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

Note: For fax to work, G.711A must be set as the preferred codec. In this configuration, the Communication Manager takes no action when fax is detected.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Cable and Wireless SIP IP Trunking service. During test, this was configured to use **TCP** and port **5060** to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of **5061** for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- Set **Peer Detection Enabled** to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Far-end Node Name** to the Session Manager (node name **SM100** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Near-end Listen Port** and **Far-end Listen Port** to **5060** (Commonly used TCP port value)
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3**. (logically establishes the far-end for calls using this signalling group as network region **1**)
- Leave **Far-end Domain** blank (allows the CM to accept calls from any SIP domain on the associated trunk)
- Set **Direct IP-IP Audio Connections** to **y**
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from the Communication Manager)

The default values for the other fields may be used.

```
change signaling-group 1                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? 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: SM100
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain:
                                              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
```

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

- Set the **Group Type** field to **sip**
- Choose a descriptive **Group Name** in test **Group 1** was used
- Specify a trunk access code (**TAC**) consistent with the dial plan, in test **101** was used
- The **Direction** is set to **two-way** to allow incoming and outgoing calls
- Set the **Service Type** field to **public-netwrk** as required setting when using the Diversion header
- Specify the signalling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**
- Specify the **Number of Members** supported by this SIP trunk group, in test this was set to **10**

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: Group 1                     COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? y
 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
```

On **Page 2** of the trunk-group form, the Preferred **Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Cable and Wireless to prevent unnecessary SIP messages during call setup.

```
Add trunk-group 1                                                Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1800

 Disconnect Supervision - In? y  Out? y
```

On **Page 3**, set the **Numbering Format** field to **public**. This allows delivery of CLI in E.164 format with a leading "+".

```
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
```

On **Page 4** of this form:

- Set **Send Diversion Header** to **y** to include the header in forwarded and transferred calls. This is not currently used by Cable and Wireless but is included as it was set for test.
- Set **Support Request History** to **n** as Cable and Wireless does not use History Info making it an unnecessary extension to the SIP INVITE
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Cable and Wireless
- Set **Always Use re-INVITE for Display Updates** to **y** as SIP UPDATE messages are not supported by Cable and Wireless for call forwarding

```
add trunk-group 1                                                Page   4 of  21
                              PROTOCOL VARIATIONS

                           Mark Users as Phone? n
                 Prepend '+' to Calling Number? n
           Send Transferring Party Information? n
                      Network Call Redirection? y
                         Send Diversion Header? y
                       Support Request History? n
                  Telephone Event Payload Type: 101


            Convert 180 to 183 for Early Media? n
      Always Use re-INVITE for Display Updates? y
            Identity for Calling Party Display: P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? N
```

5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number. In the test configuration, individual stations were mapped to send numbers allocated from the Cable and Wireless DDI range supplied. This calling party number is sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones. Note that the digits identifying the DDI range are not shown.

```
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: 8
 4  2000           1         44149nnnnnn0    12      Maximum Entries: 9999
 4  2296           1         44149nnnnnn3    12
 4  2316           1         44149nnnnnn5    12    Note: If an entry applies to
 4  2346           1         44149nnnnnn2    12    a SIP connection to Avaya
 4  2396           1         44149nnnnnn1    12    Aura(R) Session Manager,
 4  2400           1         44149nnnnnn6    12    the resulting number must
 4  2601           1         44149nnnnnn4    12    be a complete E.164 number.
```

5.8. Administer Route Selection for Outbound Calls

In the test environment, the **Automatic Route Selection (ARS)** feature was used to route outbound calls via the SIP trunk to the Cable and Wireless SIP IP Trunking service. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

```
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: *69
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 5
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

Use the **change ars analysis** command to configure the routing of dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning 0 or 00. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 1**.

```
change ars analysis 0                                          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
    0                     8    14     1       pubu         n
    00                    13   17     1       pubu         n
    00353                 10   14     1       pubu         n
    0044                  12   14     1       pubu         n
    0800                  11   11     1       pubu         n
    118                   5    6      1       pubu         n
```

Use the **change route-pattern x** command, where **x** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group **1**. Numbering Plan Indictor (NPI) of the Calling Party Number is set to E.164 and Type of Numbering (TON) is set to international by using **Numbering Format** of **intl-pub**.

```
change route-pattern 1                                         Page   1 of   3
                    Pattern Number: 1   Pattern Name: all calls
                             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: 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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                              intl-pub  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
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Cable and Wireless can be manipulated as necessary to route calls to the desired extension. In the example, the incoming DDI numbers provided by Cable and Wireless for testing are assigned to the internal extensions of the test equipment configured within the Communication Manager. The **change inc-call-handling-trmt trunk-group x** command is used to translate numbers **149nnnnnn0** to **149nnnnnn8** to the 4 digit extension by deleting all (**10**) of the incoming digits and inserting the extension number. Note that the significant digits beyond the city code have been obscured.

```
change inc-call-handling-trmt trunk-group 1                       Page   1 of  30
                     INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    10 149nnnnnn0       10  2000
 public-ntwrk    10 149nnnnnn1       10  2396
 public-ntwrk    10 149nnnnnn2       10  2346
 public-ntwrk    10 149nnnnnn3       10  2296
 public-ntwrk    10 149nnnnnn4       10  2601
 public-ntwrk    10 149nnnnnn5       10  2316
 public-ntwrk    10 149nnnnnn6       10  2400
 public-ntwrk    10 149nnnnnn7       10  6103
 public-ntwrk    10 149nnnnnn8       10  2501
```

5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 2396. Use the command **change off-pbx-telephone station mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** enter the phone that will also be called (e.g. **0035386nnnnnnn**)
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing
- Set the **Config Set** to **1**

```
change off-pbx-telephone station-mapping 2396                     Page   1 of   3
                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2396             EC500         -     0035386nnnnnnn   1          1
                                -
```

Save Communication Manager changes by entering **save translation** to make them permanent.

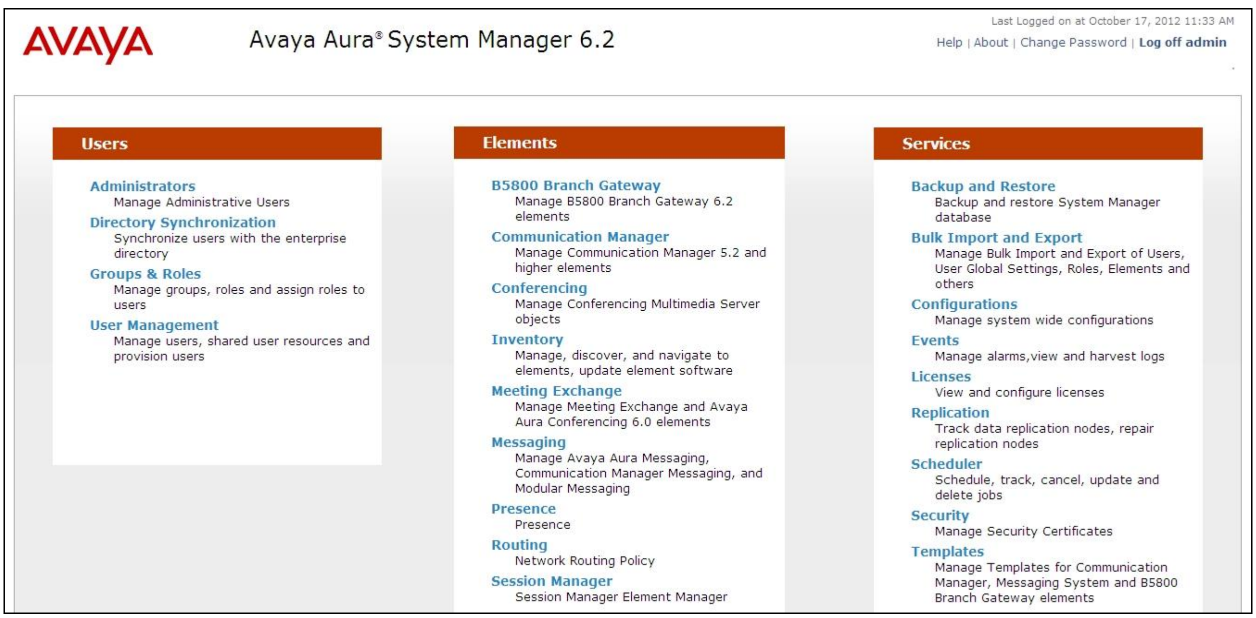
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

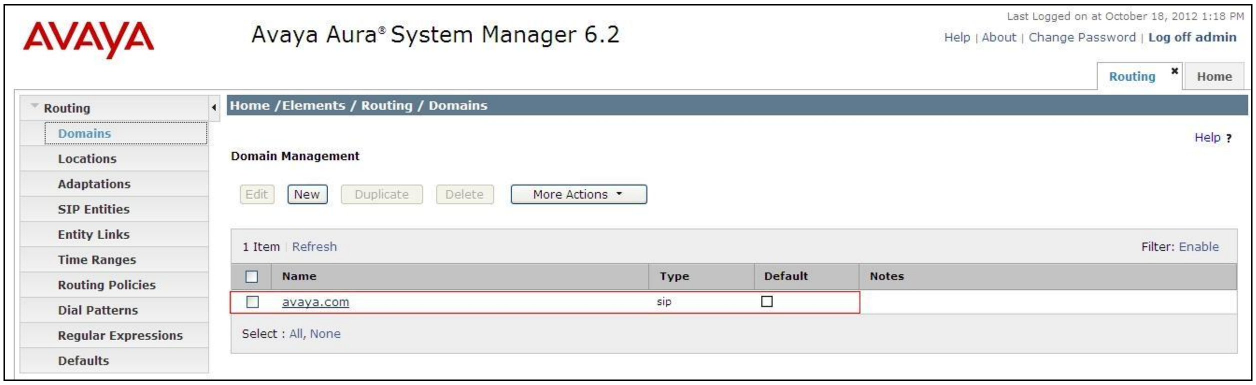
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN >/SMGR**, where <**FQDN**> is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Home tab will be presented with menu options shown below.



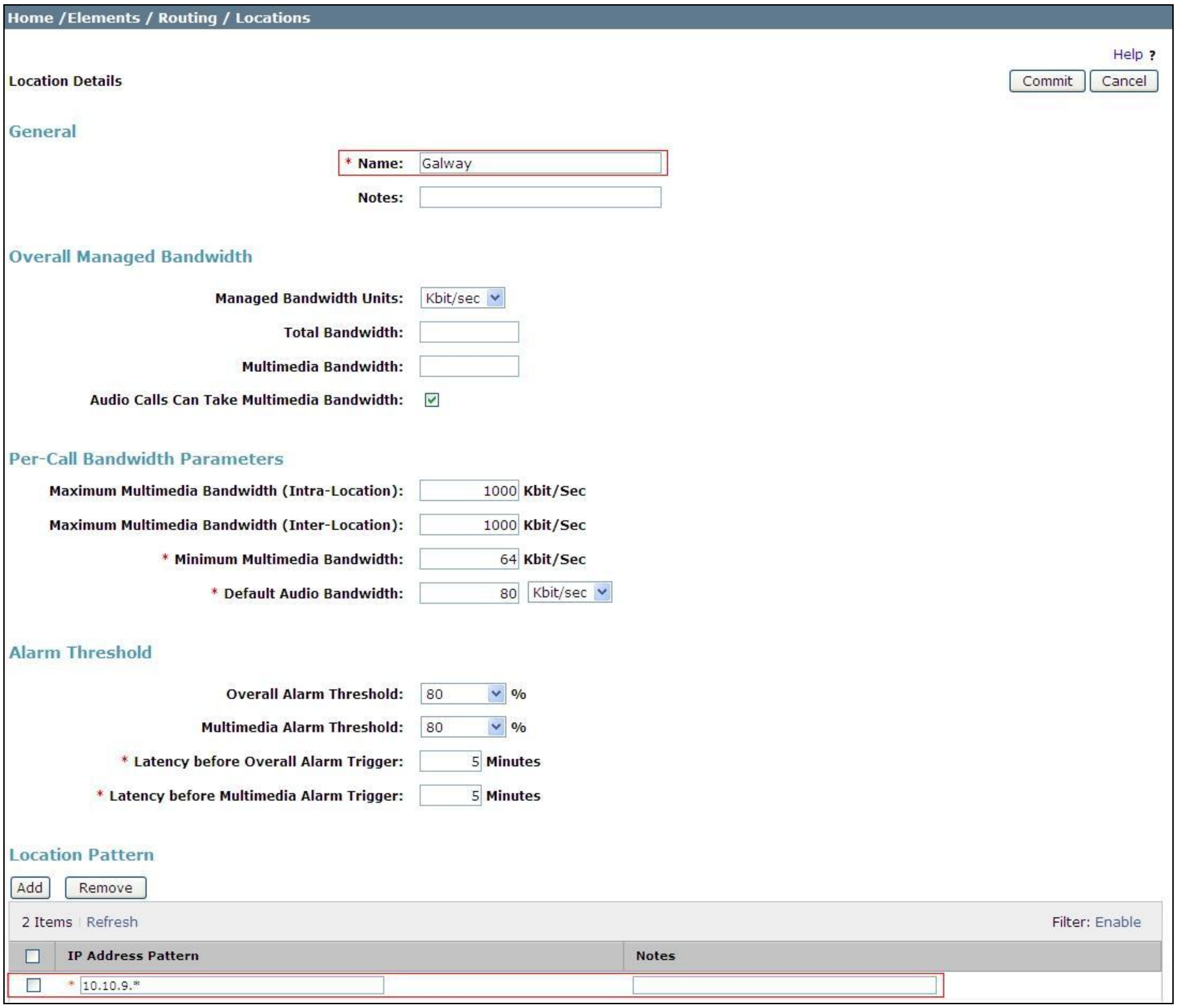
6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **avaya.com**) and optionally a description for the domain in the Notes field. Click **Commit** to save changes.



6.3. Administer Locations

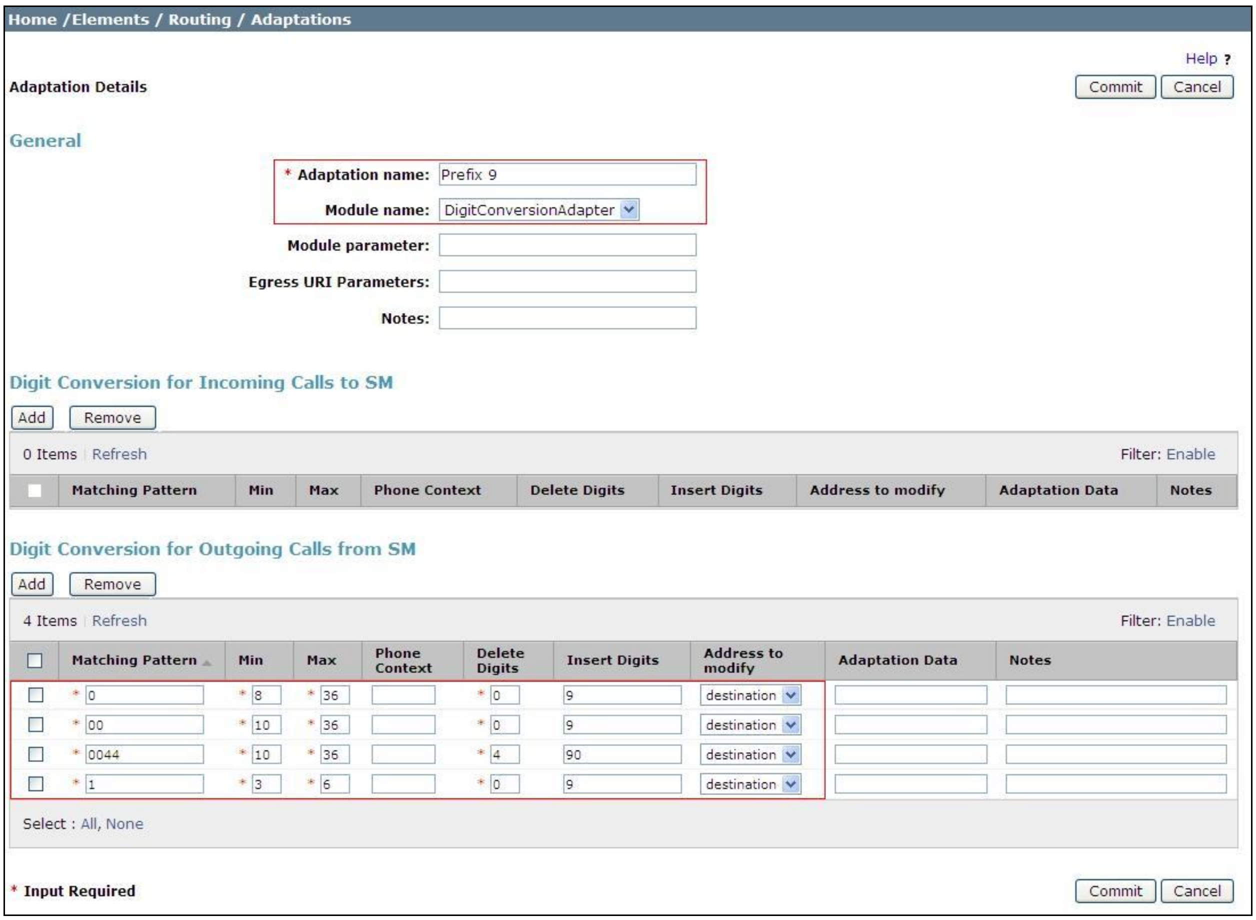
Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field, enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the test enterprise.



6.4. Administer Adaptations

Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **Digit Conversion** in the Adaptation. The example shown was used in test to prefix the called party number with a **9** which is a requirement of Cable and Wireless. The module **DigitConversionAdaptor** is used and terminating numbers starting with a **0** for national and international calls and **1** for Operator and Directory Enquiries are analysed and the **9** inserted. Additionally, UK numbers are converted from international to national format.

These rules are applied to the destination addresses.



6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity. Under **General**:

- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **SIP Trunk** for the Session Border Controller SIP entity (Note that **Gateway** was used in test, there is not currently a significant difference in functionality between the two settings)
- In the Adaptation field, enter the adaptation defined in **section 6.4** where appropriate. In test this was applied to the Acme Packet SBC
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

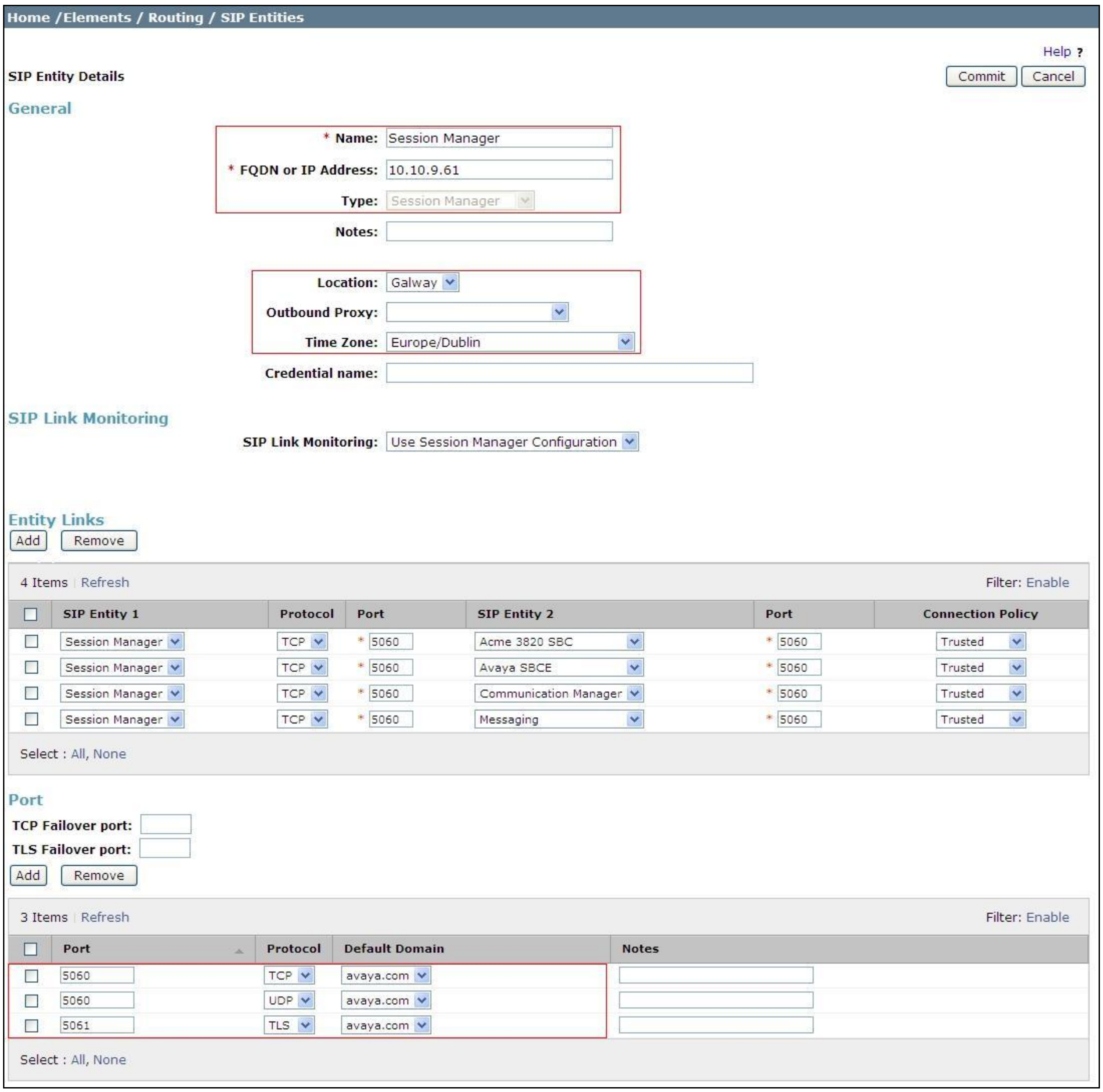
In this configuration there are three SIP Entities:

- Avaya Aura® Session Manager SIP Entity
- Avaya Aura® Communication Manager SIP Entity
- Acme Packet Net-Net 3820 SBC SIP Entity

6.5.1. Avaya Aura® Session Manager SIP Entity

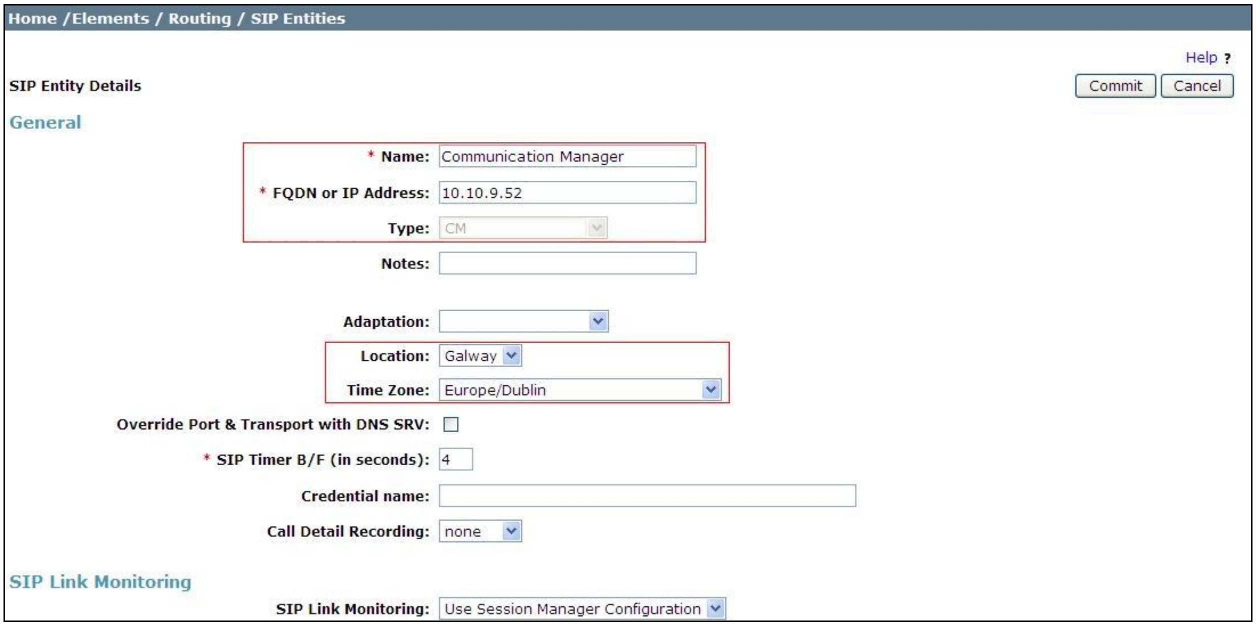
The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface. The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests
- In the **Protocol** field enter the transport protocol to be used for SIP requests
- In the **Default Domain** field, from the drop down menu select **avaya.com** as the default domain



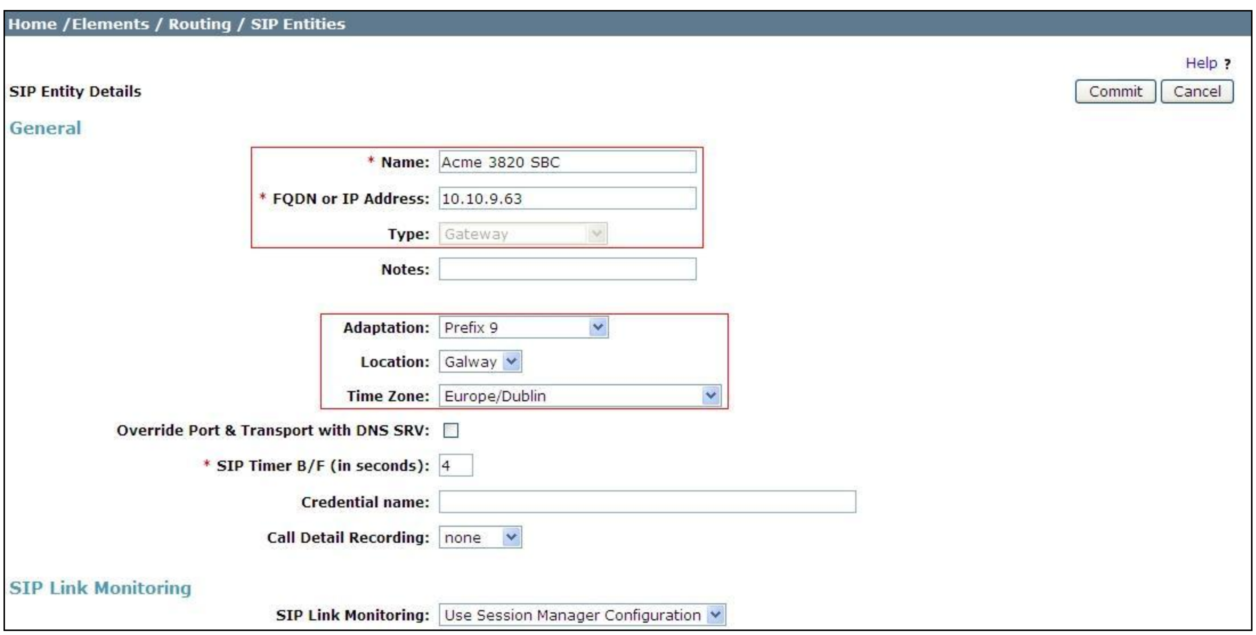
6.5.2. Avaya Aura® Communication Manager SIP Entity

The following screen shows the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling.



6.5.3. Acme Packet Net-Net 3820 SIP Entity

The following screen shows the SIP Entity for the Session Border Controller. The **FQDN or IP Address** field is set to the IP address of the Acme Packet SBC enterprise network interface.

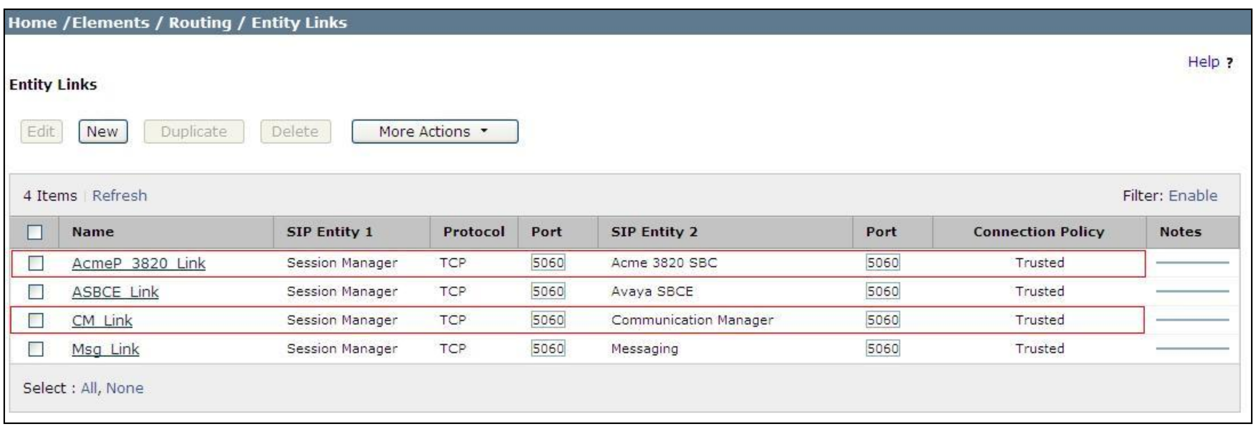


6.6. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select the name given to the Session Manager Entity, in this case **Session Manager**
- In the **Port** field enter the port number to which the other system sends its SIP requests
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests
- Select the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Click **Commit** to save changes. The following screen shows the Entity Links used in this configuration.



	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
☐	AcmeP_3820_Link	Session Manager	TCP	5060	Acme 3820 SBC	5060	Trusted	
☐	ASBCE_Link	Session Manager	TCP	5060	Avaya SBCE	5060	Trusted	
☐	CM_Link	Session Manager	TCP	5060	Communication Manager	5060	Trusted	
☐	Msg_Link	Session Manager	TCP	5060	Messaging	5060	Trusted	

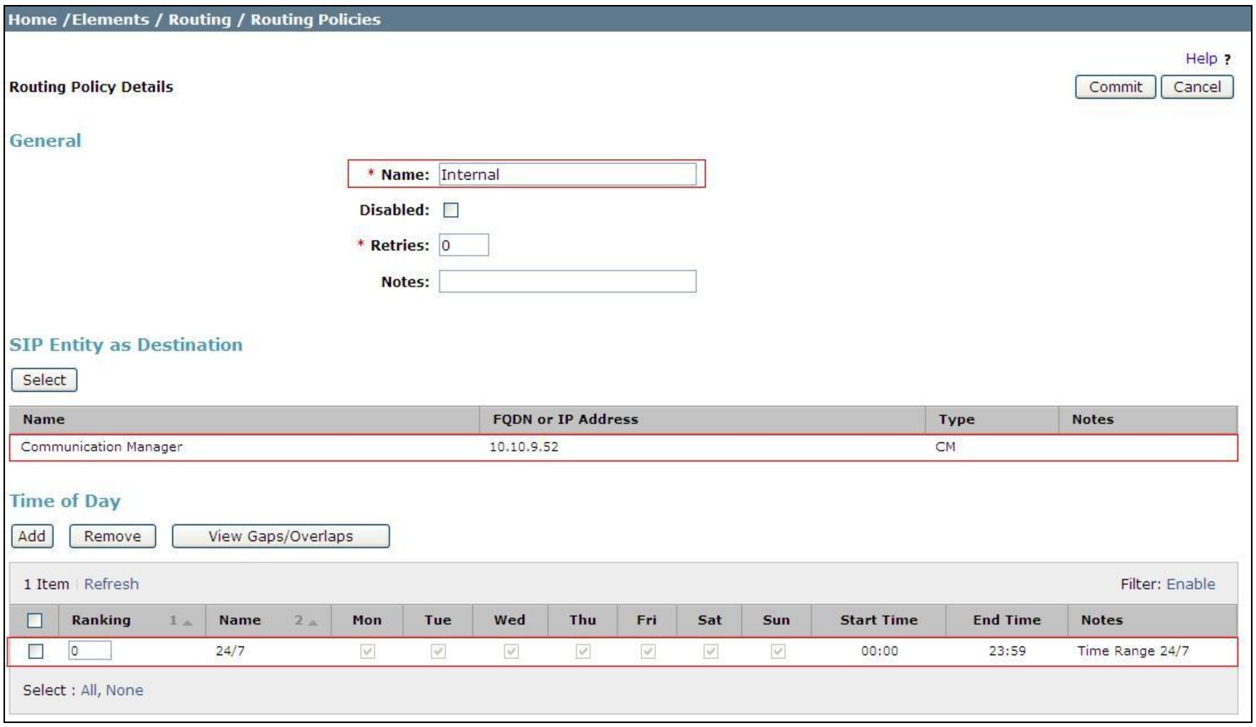
6.7. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

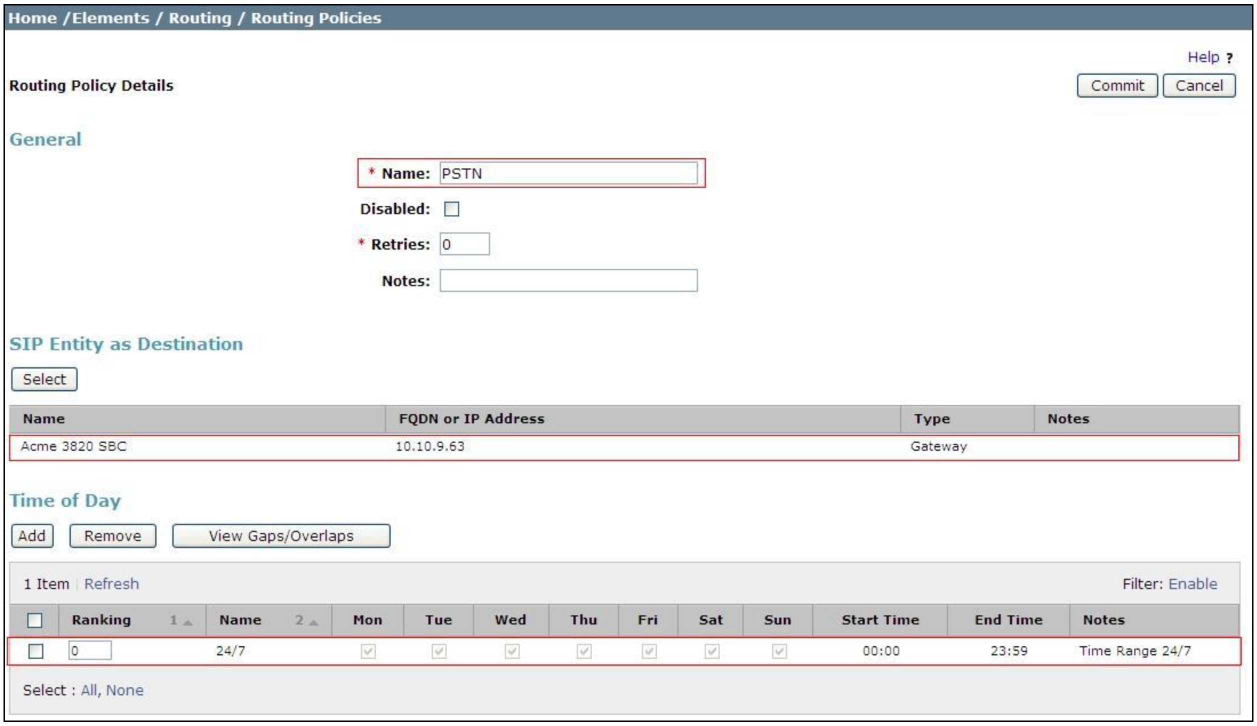
Under **General**:

- Enter an informative name in the **Name** field
- 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 then select the time range

The following screen shows the routing policy for Communication Manager.



The following screen shows the routing policy for the Acme Packet SBC.



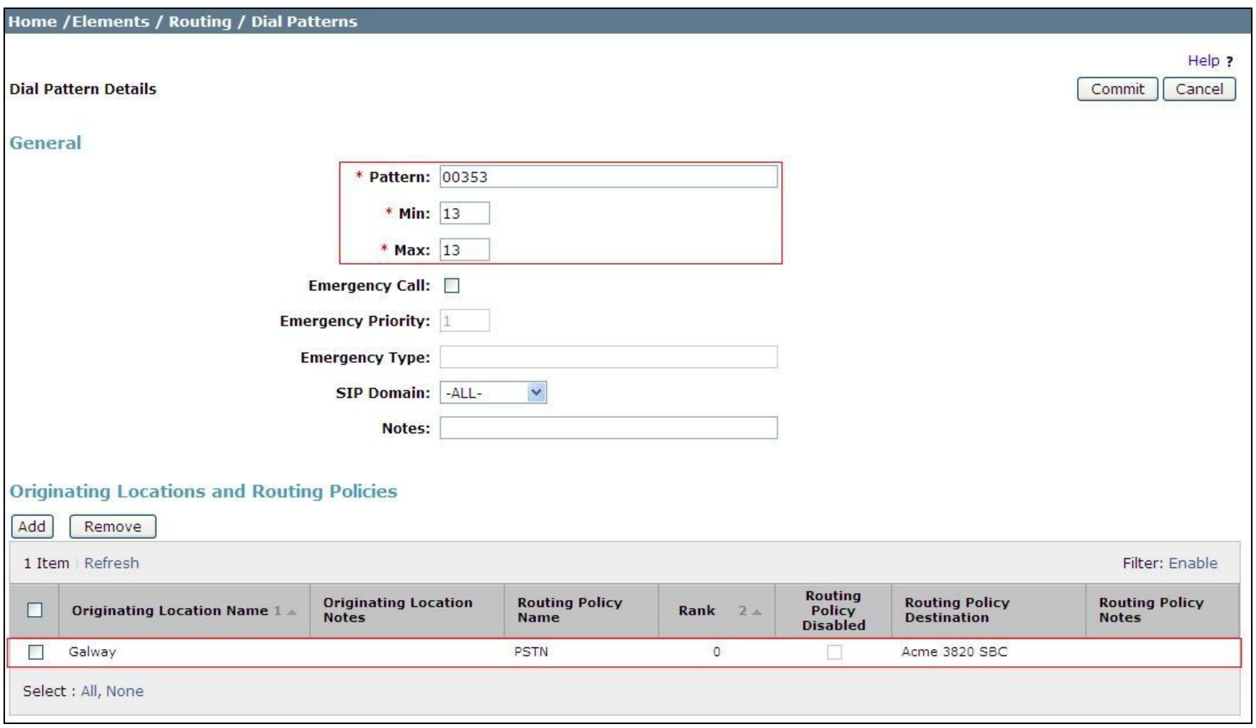
6.8. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

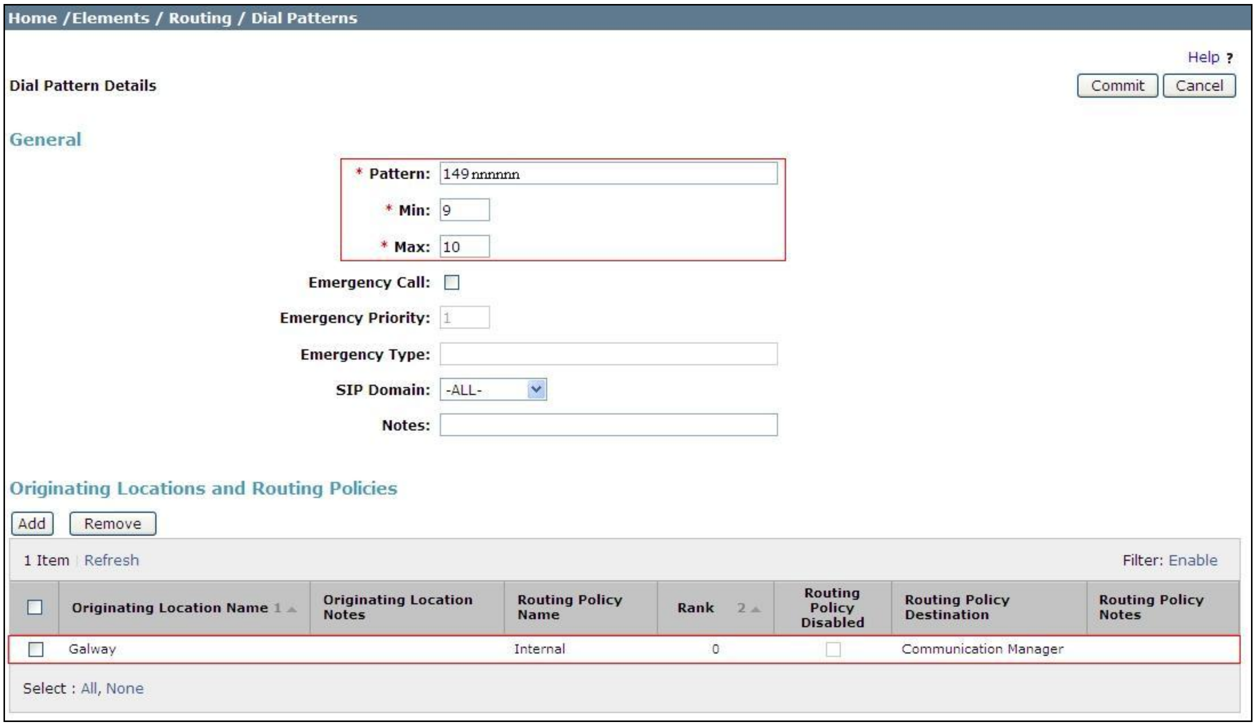
Under **General**:

- In the **Pattern** field enter a dialled number or prefix to be matched
- In the **Min** field enter the minimum length of the dialled number
- In the **Max** field enter the maximum length of the dialled number
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown), under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.6**, click **Select** button to save. The following screen shows an example dial pattern configured for the Acme Packet SBC which will route the calls out to the Cable and Wireless SIP IP Trunking service.



The following screen shows the test dial pattern configured for Communication Manager. Note that the number format received from Cable and Wireless was national with no leading 0.



6.9. Administer Application for Avaya Aura® Communication Manager

From the home tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New**.

- In the **Name** field enter a name for the application
- In the **SIP Entity** field select the SIP entity for the Communication Manager
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager

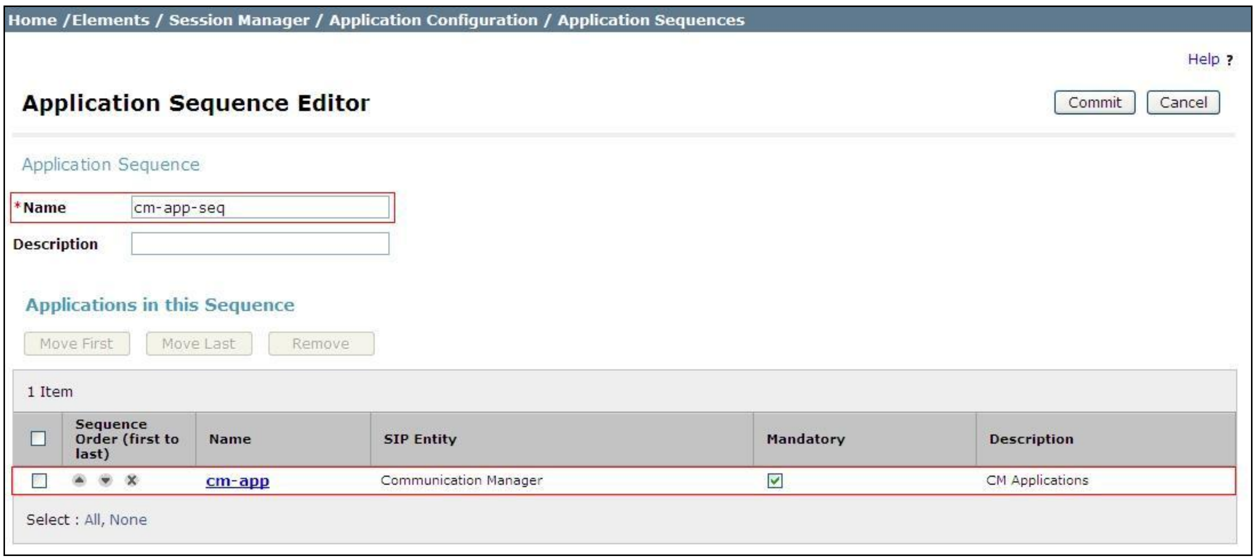
Select **Commit** to save the configuration.

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New**.

- In the **Name** field enter a descriptive name
- Under **Available Applications** (not shown), click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading

Select **Commit**.

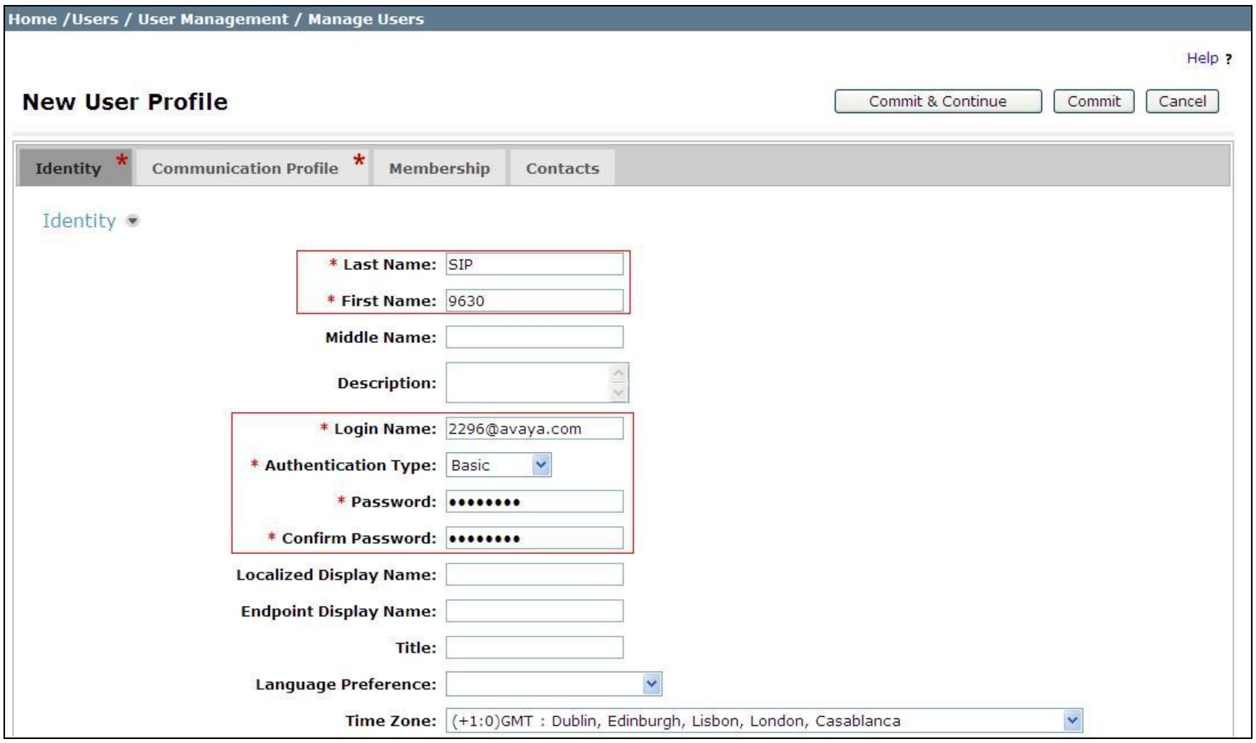


6.11. Administer SIP Extensions

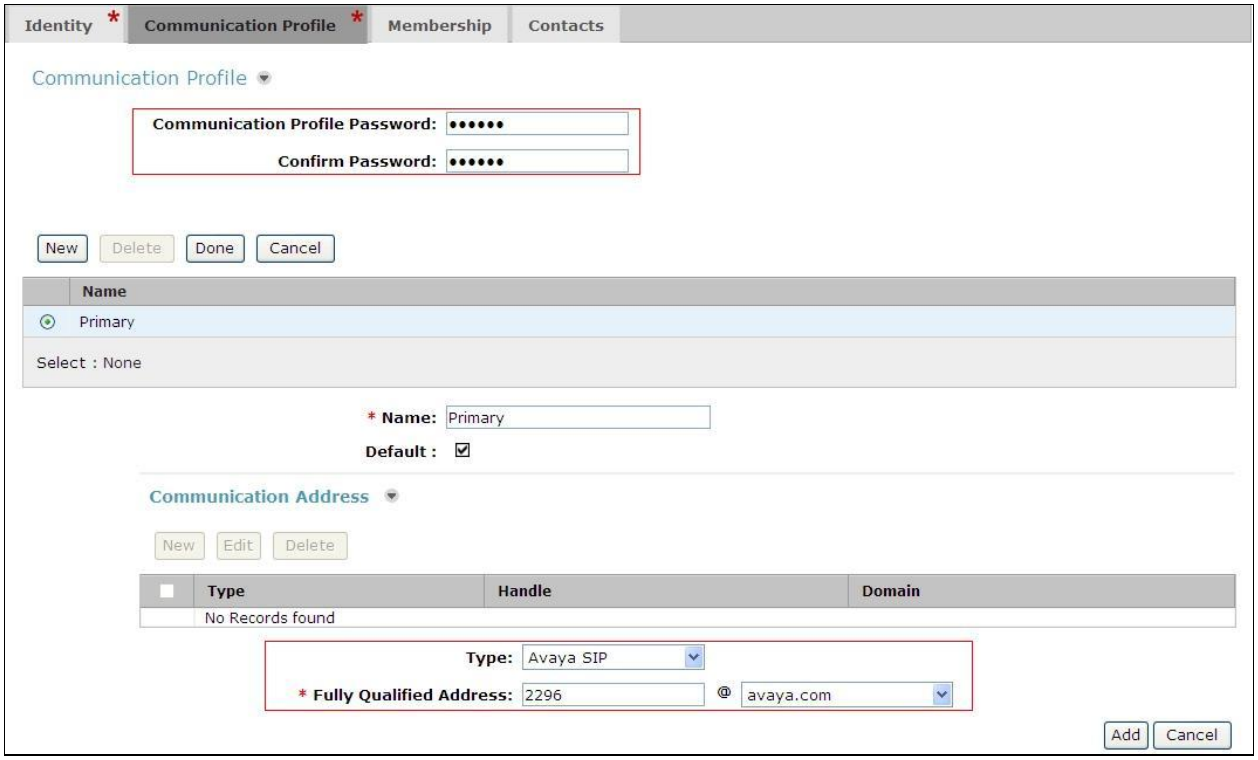
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the Home tab, select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields
- In the **Login Name** field enter a unique system login name in the form of **user@domain** (e.g. **2296@avaya.com**) which is used to create the user's primary handle
- The **Authentication Type** should be **Basic**
- In the **Password/Confirm Password** fields enter an alphanumeric password

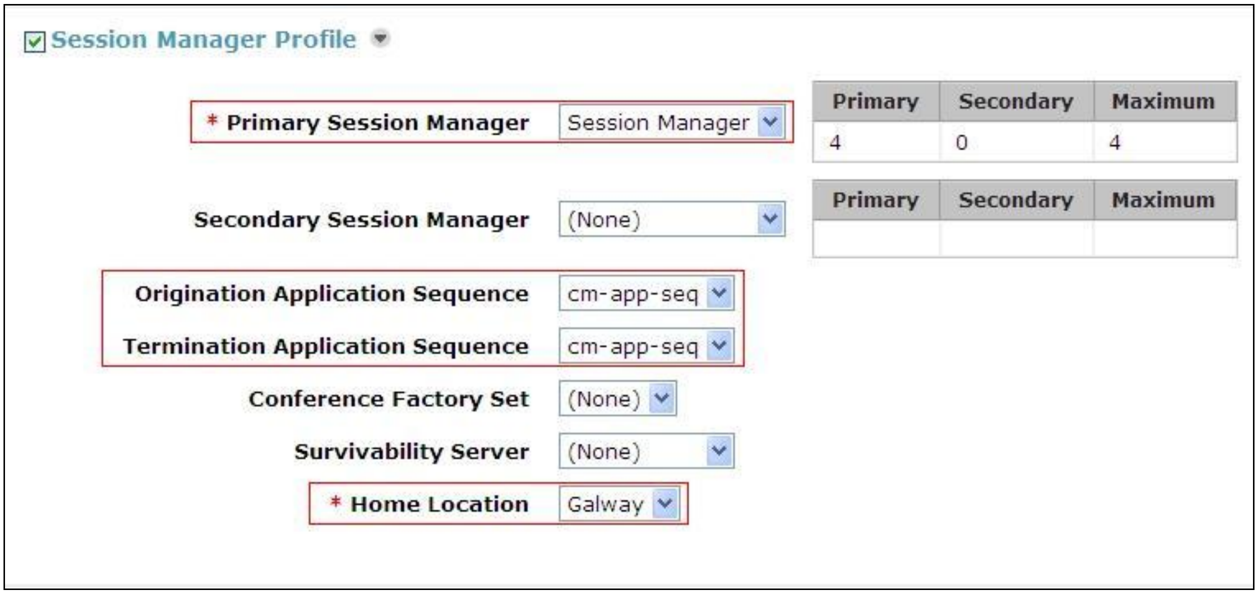


On the **Communication Profile** tab, enter a numeric **Communication Profile Password** and confirm it, then expand the **Communication Address** section and click **New**. For the **Type** field, select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



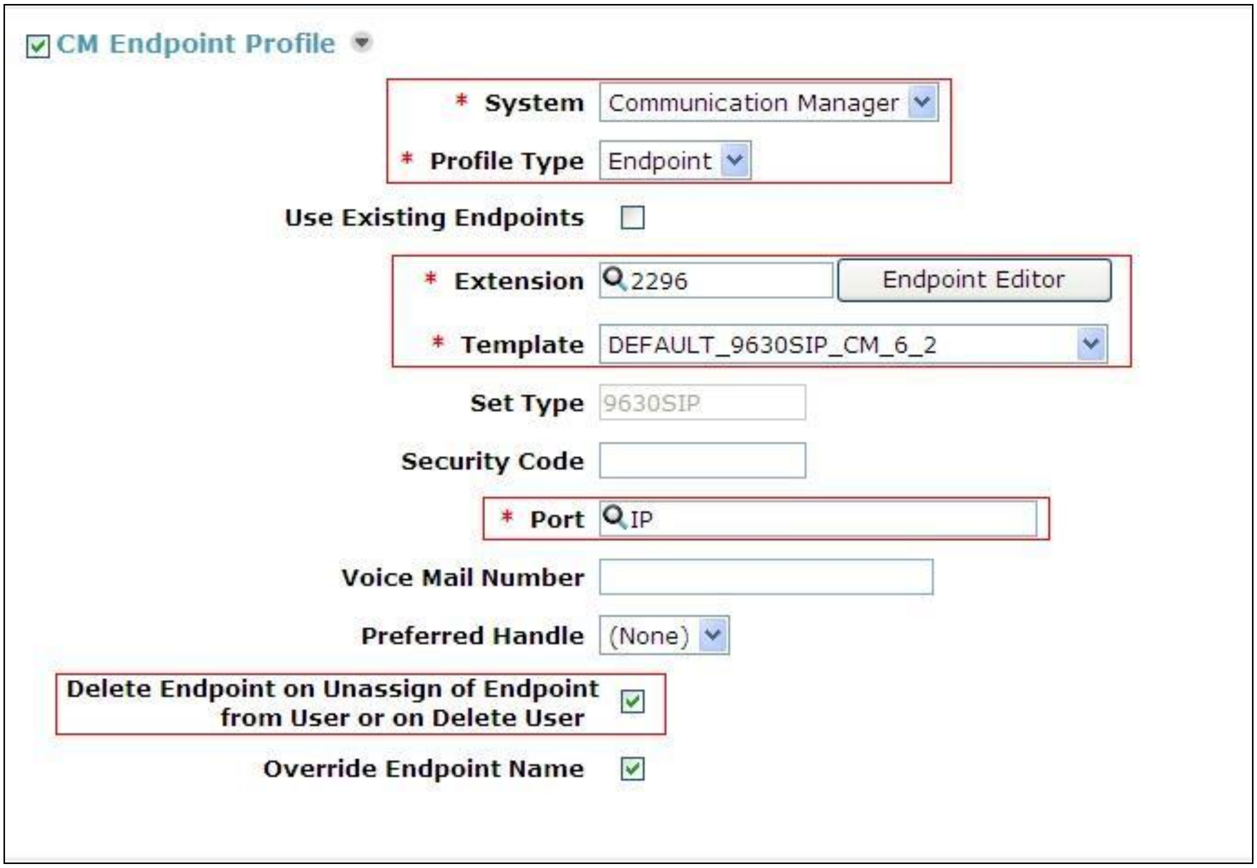
Expand the **Session Manager Profile** section.

- Make sure the **Session Manager** check box is checked
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field
- Select the appropriate application sequence from the drop-down menu in the **Origination Application Sequence** field configured in **Section 6.9**
- Select the appropriate application sequence from the drop-down menu in the **Termination Application Sequence** field configured in **Section 6.9**
- Select the appropriate location from the drop-down menu in the **Home Location** field



Expand the **Endpoint Profile** section.

- Select the Communication Manager SIP Entity from the **System** drop-down menu
- Select **Endpoint** from the drop-down menu for **Profile Type**
- Enter the extension in the **Extension** field
- Select the desired template from the **Template** drop-down menu
- For the **Port** field select **IP**
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box
- Select **Commit** to save changes and the System Manager will add the Communication Manager user configuration automatically



7. Configure Acme Packet Net-Net 3820 SBC

Refer to the printout of the test configuration provided in **Appendix A** for guidance on how to configure the Acme Packet Net-Net 3820 SBC. Note that the configuration is identical to that required for an Acme packet Net-Net 4500 SBC as they use the same firmware. There are two points worth mentioning:

- A built in Header Manipulation Rule (HMR) is used for hiding the enterprise network topology. This HMR is **ACME_NAT_TO_FROM_IP**and is applied as an **out-manipulationid** in the outside session agent. It doesn't appear in the **sip-manipulation** as it is built in, and can be viewed by typing **show built-in-sip-manipulation**
- The HMR **SIPNAT** is not used, it has been replaced by the built in HMR described above
- A HMR was developed as a workaround to the fault described in **Section 2.2** where no ring-back was heard on calls forwarded to the PSTN

7.1. Header Manipulation Rule

The HMR applied as a workaround to the ring-back fault described in Section 2.2 simply replaces the "Supported" header in 18x responses from the enterprise. This has the effect of removing the "timer" parameter in the header. This parameter invokes behavior in the CS2K that results in ringback not being played. The fault has been reproduced in the Cable and Wireless Lab and a CSR raised to Genband. The workaround described here is required until a permanent solution is implemented in the network.

The HMR is as follows:

```
sip-manipulation
       name                            Add100Rel
       description
       split-headers
       join-headers
       header-rule
              name                            Insert100rel
              header-name                     Supported
              action                          manipulate
              comparison-type                 case-sensitive
              msg-type                        reply
              methods
              match-value
              new-value                       100rel
       last-modified-by                admin@10.10.2.27
       last-modified-date              2012-10-18 07:12:32
```

It is applied in the session agent as follows:

```
session-agent
       in-manipulationid               Add100Rel
```

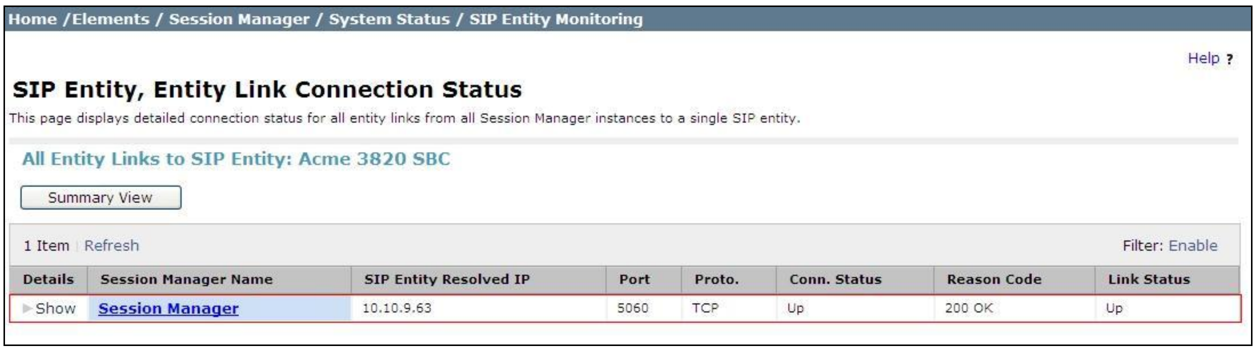
8. Configure Cable and Wireless SIP IP Trunking

The configuration of the Cable and Wireless equipment used to support the SIP IP Trunking service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Cable and Wireless equipment and system configuration please contact an authorised Cable and Wireless representative.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager Home Tab click on Session Manager and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.



Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▸ Show	Session Manager	10.10.9.63	5060	TCP	Up	200 OK	Up

Note: This is also an indication that the SIP trunk between the Acme packet SBC and the Cable and Wireless network is working effectively as OPTIONS are passed by the SBC from the Session Manager to the network

2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 1

                             TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   in-service/idle    no
```

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Acme Packet Net-Net 3820 SBC to Cable and Wireless SIP IP Trunking service. The service was successfully tested with a number of observations listed in **Section 2.2.**

11. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Installing and Configuring Avaya Aura® System Platform Release 6.2*, March 2012.
[2] *Administering Avaya Aura® System Platform Release 6.2*, February 2012.
[3] *Administering Avaya Aura® Communication Manager*, Release 6.2, February 2012.
[4] *Avaya Aura® Communication Manager Feature Description and Implementation*, February 2012, Document Number 555-245-205.
[5] *Implementing Avaya Aura® System Manager Release 6.2*, March 2012.
[6] *Implementing Avaya Aura® Session Manager*, February 2012, Document Number 03-603473
[7] *Administering Avaya Aura® Session Manager*, February 2012, Document Number 03-603324.
[8] *Net-Net 4000 S-CX6.3.0 Maintenance and Troubleshooting Guide.pdf*, https://support.acmepacket.com/
[9] *Net-Net Session Director C[xz]6.3.9 Final User Guide.pdf*, https://support.acmepacket.com/
[10] *Acme Packet HMR Developers Guide.pdf*, https://support.acmepacket.com/
[11] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

Appendix A

The configuration details provided here are the Acme Packet 3820 Net-Net SBC settings used during compliance testing. Publicly routable IP addresses have been changed to private IP addresses for security reasons.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to these application notes. Other parameter values may or may not match based on local configurations and are shown for illustrative purposes.

ANNOTATION: The following host route is required to send/receive packets to/from the SIP trunk providers' network.

```
host-routes
       dest-network                   192.168.24.8 [IP address of SIP trunk service]
       netmask                        255.255.255.0
       gateway                        192.168.37.1 [gateway for SIP traffic]
       description                    route-to-cw
       last-modified-by               admin@console
       last-modified-date             2012-09-11 07:38:12
```

ANNOTATION: The local policy below controls the routing of SIP messages from the SIP trunk service provider to the session manager.

```
local-policy
       from-address
                                      *
       to-address
                                      *
       source-realm
                                      OUTSIDE [SIP trunk service provider]
       description                    Far-side-realm
       activate-time                  N/A
       deactivate-time                N/A
       state                          enabled
       policy-priority                none
       last-modified-by               admin@console
       last-modified-date             2012-09-11 07:46:52
       policy-attribute
              next-hop                       10.10.9.61 [session manager IP address]
              realm                          INSIDE [the Enterprise realm]
              action                         none
              terminate-recursion            disabled
              carrier
              start-time                     0000
              end-time                       2400
              days-of-week                   U-S
              cost                           0
              app-protocol
              state                          enabled
              methods
              media-profiles
              lookup                         single
              next-key
              eloc-str-lkup                  disabled
              eloc-str-match
```

ANNOTATION: The local policy below controls the routing of SIP messages from session manager to the C&W SIP trunk service.

```
local-policy
       from-address
                                      *
       to-address
                                      *
       source-realm
                                      INSIDE [Enterprise SIP domain]
       description
       activate-time                  N/A
       deactivate-time                N/A
       state                          enabled
       policy-priority                none
       last-modified-by               admin@console
       last-modified-date             2012-09-11 07:48:49
       policy-attribute
              next-hop                       192.168.24.8 [SIP trunk provider address]
              realm                          OUTSIDE [SIP trunk provider realm]
              action                         none
              terminate-recursion            disabled
              carrier
              start-time                     0000
              end-time                       2400
              days-of-week                   U-S
              cost                           0
              app-protocol
              state                          enabled
              methods
              media-profiles
              lookup                         single
              next-key
              eloc-str-lkup                  disabled
              eloc-str-match
media-manager
       state                          enabled [enabled to manage voice media]
       latching                       enabled
       flow-time-limit                86400
       initial-guard-timer            300
       subsq-guard-timer              300
       tcp-flow-time-limit            86400
       tcp-initial-guard-timer        300
       tcp-subsq-guard-timer          300
       tcp-number-of-ports-per-flow   2
       hnt-rtcp                       disabled
       algd-log-level                 NOTICE
       mbcd-log-level                 NOTICE
       options                        unique-sdp-id
       red-flow-port                  1985
       red-mgcp-port                  1986
       red-max-trans                  10000
       red-sync-start-time            5000
       red-sync-comp-time             1000
       media-policing                 enabled
       max-signaling-bandwidth        10000000
       max-untrusted-signaling        100
       min-untrusted-signaling        30
       app-signaling-bandwidth        0
       tolerance-window               30
```

```
      rtcp-rate-limit                0
      trap-on-demote-to-deny         enabled
      syslog-on-demote-to-deny       disabled
      syslog-on-demote-to-untrusted  disabled
      min-media-allocation           2000
      min-trusted-allocation         4000
      deny-allocation                64000
      anonymous-sdp                  disabled
      arp-msg-bandwidth              32000
      fragment-msg-bandwidth         0
      rfc2833-timestamp              disabled
      default-2833-duration          100
      rfc2833-end-pkts-only-for-non-sig enabled
      translate-non-rfc2833-event    disabled
      media-supervision-traps        disabled
      dnsalg-server-failover         disabled
      last-modified-by               admin@10.10.2.110
      last-modified-date             2011-06-17 07:45:01
```

ANNOTATION: The following network interfaces define the IP address used on the enterprise (INSIDE) network and on the SIP trunk provider (OUTSIDE) network and the associated physical ports to which these addresses are mapped.

```
network-interface
      name                           S0P1
      sub-port-id                    0
      description                    INSIDE [the realm using this IP address]
      hostname
      ip-address                     10.10.9.63 [Acme Packet private IP address]
      pri-utility-addr
      sec-utility-addr
      netmask                        255.255.255.0
      gateway                        10.10.9.1 [private side gateway]
      sec-gateway
      gw-heartbeat
            state                          enabled
            heartbeat                      10
            retry-count                    3
            retry-timeout                  1
            health-score                   30
      dns-ip-primary
      dns-ip-backup1
      dns-ip-backup2
      dns-domain
      dns-timeout                    11
      hip-ip-list                    10.10.9.63 [allow hip to this address]
      ftp-address                    10.10.9.63 [allow ftp to this address]
      icmp-address                   10.10.9.63 [allow icmp to this address]
      snmp-address                   10.10.9.63 [allow snmp to this address]
      telnet-address                 10.10.9.63 [allow telnet to this address]
      ssh-address
      signaling-mtu                  0
      last-modified-by               admin@console
      last-modified-date             2012-09-11 07:58:11
network-interface
      name                           S0P0
      sub-port-id                    0
      description                    OUTSIDE [SIP trunk provider realm]
      hostname
      ip-address                     192.168.37.2 [Acme Packet public IP address]
      pri-utility-addr
```

```
        sec-utility-addr
        netmask                       255.255.255.0
        gateway                       192.168.37.1 [public side gateway (VPN router)]
        sec-gateway
        gw-heartbeat
                state                         enabled
                heartbeat                     10
                retry-count                   3
                retry-timeout                 3
                health-score                  30
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                   11
        hip-ip-list                   192.168.37.2 [allow hip to this address]
        ftp-address
        icmp-address                  192.168.37.2 [allow icmp to this address]
        snmp-address
        telnet-address
        ssh-address
        signaling-mtu                 0
        last-modified-by              admin@console
        last-modified-date            2012-09-11 08:27:18
phy-interface
        name                          S0P0
        operation-type                Media
        port                          0
        slot                          0
        virtual-mac
        admin-state                   enabled
        auto-negotiation              enabled
        duplex-mode                   FULL
        speed                         100
        overload-protection           disabled
        last-modified-by              admin@console
        last-modified-date            2011-03-22 05:22:58
phy-interface
        name                          S0P1
        operation-type                Media
        port                          1
        slot                          0
        virtual-mac
        admin-state                   enabled
        auto-negotiation              enabled
        duplex-mode                   FULL
        speed                         100
        overload-protection           disabled
        last-modified-by              admin@135.64.186.34
        last-modified-date            2011-03-22 07:50:27
```

ANNOTATION: The realm configuration “OUTSIDE” represents the external network on which the SIP trunk service resides.

```
realm-config
        identifier                    OUTSIDE [SIP trunk provider realm]
        description                   SIP_LAB_OUTSIDE [descriptive name]
        addr-prefix                   0.0.0.0
        network-interfaces
                                      S0P0:0
        mm-in-realm                   enabled
```

```
        mm-in-network                  enabled
        mm-same-ip                     enabled
        mm-in-system                   enabled
        bw-cac-non-mm                  disabled
        msm-release                    disabled
        qos-enable                     disabled
        generate-UDP-checksum          disabled
        max-bandwidth                  0
        fallback-bandwidth             0
        max-priority-bandwidth         0
        max-latency                    0
        max-jitter                     0
        max-packet-loss                0
        observ-window-size             0
        parent-realm
        dns-realm
        media-policy
        media-sec-policy
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        class-profile
        average-rate-limit             0
        access-control-trust-level     none
        invalid-signal-threshold       0
        maximum-signal-threshold       0
        untrusted-signal-threshold     0
        nat-trust-threshold            0
        deny-period                    30
        cac-failure-threshold          0
        untrust-cac-failure-threshold  0
        ext-policy-svr
        diam-e2-address-realm
        symmetric-latching             disabled
        pai-strip                      disabled
        trunk-context
        early-media-allow
        enforcement-profile
        additional-prefixes
        restricted-latching            none
        restriction-mask               32
        accounting-enable              enabled
        user-cac-mode                  none
        user-cac-bandwidth             0
        user-cac-sessions              0
        icmp-detect-multiplier         0
        icmp-advertisement-interval    0
        icmp-target-ip
        monthly-minutes                0
        net-management-control         disabled
        delay-media-update             disabled
        refer-call-transfer            disabled
        refer-notify-provisional       none
        dyn-refer-term                 disabled
        codec-policy
        codec-manip-in-realm           disabled
        constraint-name
        call-recording-server-id
        xnq-state                      xnq-unknown
```

```
        hairpin-id                     0
        stun-enable                    disabled
        stun-server-ip                 0.0.0.0
        stun-server-port               3478
        stun-changed-ip                0.0.0.0
        stun-changed-port              3479
        match-media-profiles
        qos-constraint
        sip-profile
        sip-isup-profile
        block-rtcp                     disabled
        hide-egress-media-update       disabled
        last-modified-by               admin@10.10.2.27
        last-modified-date             2012-10-16 08:17:41
```

ANNOTATION: The realm configuration "INSIDE" represents the enterprise network where the communication manager and session manager are located.

```
realm-config
        identifier                     INSIDE [Enterprise realm]
        description                    SIP_LAB_INSIDE [descriptive name]
        addr-prefix                    0.0.0.0
        network-interfaces
                                       S0P1:0
        mm-in-realm                    enabled
        mm-in-network                  enabled
        mm-same-ip                     enabled
        mm-in-system                   enabled
        bw-cac-non-mm                  disabled
        msm-release                    disabled
        qos-enable                     disabled
        generate-UDP-checksum          disabled
        max-bandwidth                  0
        fallback-bandwidth             0
        max-priority-bandwidth         0
        max-latency                    0
        max-jitter                     0
        max-packet-loss                0
        observ-window-size             0
        parent-realm
        dns-realm
        media-policy
        media-sec-policy
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        class-profile
        average-rate-limit             0
        access-control-trust-level     none
        invalid-signal-threshold       0
        maximum-signal-threshold       0
        untrusted-signal-threshold     0
        nat-trust-threshold            0
        deny-period                    30
        cac-failure-threshold          0
        untrust-cac-failure-threshold  0
        ext-policy-svr
        diam-e2-address-realm
```

```
        symmetric-latching             disabled
        pai-strip                      disabled
        trunk-context
        early-media-allow
        enforcement-profile
        additional-prefixes
        restricted-latching            none
        restriction-mask               32
        accounting-enable              enabled
        user-cac-mode                  none
        user-cac-bandwidth             0
        user-cac-sessions              0
        icmp-detect-multiplier         0
        icmp-advertisement-interval    0
        icmp-target-ip
        monthly-minutes                0
        net-management-control         disabled
        delay-media-update             disabled
        refer-call-transfer            disabled
        refer-notify-provisional       none
        dyn-refer-term                 disabled
        codec-policy
        codec-manip-in-realm           disabled
        constraint-name
        call-recording-server-id
        xnq-state                      xnq-unknown
        hairpin-id                     0
        stun-enable                    disabled
        stun-server-ip                 0.0.0.0
        stun-server-port               3478
        stun-changed-ip                0.0.0.0
        stun-changed-port              3479
        match-media-profiles
        qos-constraint
        sip-profile
        sip-isup-profile
        block-rtcp                     disabled
        hide-egress-media-update       disabled
        last-modified-by               admin@10.10.2.27
        last-modified-date             2012-10-16 08:18:37
```

ANNOTATION: The session agent below represents the Cable and Wireless SIP trunk service network border element. The Acme will attempt to send calls to the border element based on successful responses to the OPTIONS "ping-method". Cable and Wireless SIP trunk service border element is also specified in the session-group section below.

```
session-agent
        hostname                       192.168.24.8
        ip-address                     192.168.24.8
        port                           5060
        state                          enabled
        app-protocol                   SIP
        app-type
        transport-method               UDP
        realm-id                       OUTSIDE
        egress-realm-id
        description                    candw
        carriers
        allow-next-hop-lp              enabled
        constraints                    disabled
```

```
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume                  0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action                 Proxy
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS;hops=66
        ping-interval                   120
        ping-send-mode                  keep-alive
        ping-all-addresses              disabled
        ping-in-service-response-codes
        out-service-response-codes
        media-profiles
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid              ACME_NAT_TO_FROM_IP
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 0
        codec-policy
        enforcement-profile
        refer-call-transfer             disabled
        refer-notify-provisional        none
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        sip-profile
        sip-isup-profile
        last-modified-by                admin@10.10.2.27
        last-modified-date              2012-10-18 04:41:06
```

ANNOTATION: The session agent below represents the Communication Manager Processor Ethernet interface used in the reference configuration.

```
session-agent
       hostname                         10.10.9.61
       ip-address                       10.10.9.61
       port                             5060
       state                            enabled
       app-protocol                     SIP
       app-type
       transport-method                 UDP+TCP
       realm-id                         INSIDE
       egress-realm-id
       description                      session-manager
       carriers
       allow-next-hop-lp                enabled
       constraints                      disabled
       max-sessions                     0
       max-inbound-sessions             0
       max-outbound-sessions            0
       max-burst-rate                   0
       max-inbound-burst-rate           0
       max-outbound-burst-rate          0
       max-sustain-rate                 0
       max-inbound-sustain-rate         0
       max-outbound-sustain-rate        0
       min-seizures                     5
       min-asr                          0
       time-to-resume                   0
       ttr-no-response                  0
       in-service-period                0
       burst-rate-window                0
       sustain-rate-window              0
       req-uri-carrier-mode             None
       proxy-mode
       redirect-action                  Proxy
       loose-routing                    enabled
       send-media-session               enabled
       response-map
       ping-method                      OPTIONS;hops=66
       ping-interval                    120
       ping-send-mode                   keep-alive
       ping-all-addresses               disabled
       ping-in-service-response-codes
       out-service-response-codes
       media-profiles
       in-translationid
       out-translationid
       trust-me                         disabled
       request-uri-headers
       stop-recurse
       local-response-map
       ping-to-user-part
       ping-from-user-part
       li-trust-me                      disabled
       in-manipulationid                Add100Rel
       out-manipulationid
       manipulation-string
       manipulation-pattern
       p-asserted-id
```

```
       trunk-group
       max-register-sustain-rate      0
       early-media-allow
       invalidate-registrations       disabled
       rfc2833-mode                   none
       rfc2833-payload                0
       codec-policy
       enforcement-profile
       refer-call-transfer            disabled
       refer-notify-provisional       none
       reuse-connections              NONE
       tcp-keepalive                  none
       tcp-reconn-interval            0
       max-register-burst-rate        0
       register-burst-window          0
       sip-profile
       sip-isup-profile
       last-modified-by               admin@10.10.2.27
       last-modified-date             2012-10-18 04:45:18
```

ANNOTATION: The sip-config defines global sip-parameters, including SIP timers, SIP options, which realm to send requests to if not specified elsewhere, and enabling the SD to collect statistics on requests other than REGISTERs and INVITEs.

```
sip-config
       state                          enabled
       operation-mode                 dialog
       dialog-transparency            enabled
       home-realm-id                  INSIDE
       egress-realm-id
       nat-mode                       None
       registrar-domain               *
       registrar-host                 *
       registrar-port                 5060
       register-service-route         always
       init-timer                     500
       max-timer                      4000
       trans-expire                   32
       invite-expire                  180
       inactive-dynamic-conn          32
       enforcement-profile
       pac-method
       pac-interval                   10
       pac-strategy                   PropDist
       pac-load-weight                1
       pac-session-weight             1
       pac-route-weight               1
       pac-callid-lifetime            600
       pac-user-lifetime              3600
       red-sip-port                   1988
       red-max-trans                  10000
       red-sync-start-time            5000
       red-sync-comp-time             1000
       add-reason-header              disabled
       sip-message-len                4096
       enum-sag-match                 disabled
       extra-method-stats             disabled
       registration-cache-limit       0
       register-use-to-for-lp         disabled
       refer-src-routing              disabled
       add-ucid-header                disabled
```

```
      proxy-sub-events
      allow-pani-for-trusted-only    disabled
      pass-gruu-contact              disabled
      sag-lookup-on-redirect         disabled
      set-disconnect-time-on-bye     disabled
      last-modified-by               admin@console
      last-modified-date             2011-03-22 05:44:50
```

ANNOTATION: The SIP interface below is used to communicate with the Cable and Wireless SIP trunk service, UDP transport.

```
sip-interface
      state                          enabled
      realm-id                       OUTSIDE
      description                    candw-sip-trunk
      sip-port
            address                        192.168.37.2
            port                           5060
            transport-protocol             UDP
            tls-profile
            allow-anonymous                all
            ims-aka-profile
      carriers
      trans-expire                   0
      invite-expire                  0
      max-redirect-contacts          0
      proxy-mode
      redirect-action
      contact-mode                   none
      nat-traversal                  none
      nat-interval                   30
      tcp-nat-interval               90
      registration-caching           disabled
      min-reg-expire                 300
      registration-interval          3600
      route-to-registrar             disabled
      secured-network                disabled
      teluri-scheme                  disabled
      uri-fqdn-domain
      options                        max-udp-length=0
      trust-mode                     all
      max-nat-interval               3600
      nat-int-increment              10
      nat-test-increment             30
      sip-dynamic-hnt                disabled
      stop-recurse                   401,407
      port-map-start                 0
      port-map-end                   0
      in-manipulationid
      out-manipulationid
      manipulation-string
      manipulation-pattern
      sip-ims-feature                disabled
      operator-identifier
      anonymous-priority             none
      max-incoming-conns             0
      per-src-ip-max-incoming-conns  0
      inactive-conn-timeout          0
      untrusted-conn-timeout         0
      network-id
      ext-policy-server
```

```
        default-location-string
        charging-vector-mode            pass
        charging-function-address-mode pass
        ccf-address
        ecf-address
        term-tgrp-mode                  none
        implicit-service-route          disabled
        rfc2833-payload                 101
        rfc2833-mode                    transparent
        constraint-name
        response-map
        local-response-map
        ims-aka-feature                 disabled
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                   none
        add-sdp-invite                  disabled
        add-sdp-profiles
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                  0
        last-modified-by                admin@console
        last-modified-date              2012-10-09 07:26:56
```

ANNOTATION: The SIP interface below is used to communicate with the Session Manager, TCP transport

```
sip-interface
        state                           enabled
        realm-id                        INSIDE
        description                     Avaya-SBC
        sip-port
                address                         10.10.9.63
                port                            5060
                transport-protocol              TCP
                tls-profile
                allow-anonymous                 all
                ims-aka-profile
        carriers
        trans-expire                    0
        invite-expire                   0
        max-redirect-contacts           0
        proxy-mode
        redirect-action
        contact-mode                    none
        nat-traversal                   none
        nat-interval                    30
        tcp-nat-interval                90
        registration-caching            disabled
        min-reg-expire                  300
        registration-interval           3600
        route-to-registrar              disabled
        secured-network                 disabled
        teluri-scheme                   disabled
        uri-fqdn-domain
        trust-mode                      all
        max-nat-interval                3600
        nat-int-increment               10
        nat-test-increment              30
        sip-dynamic-hnt                 disabled
        stop-recurse                    401,407
```

```
      port-map-start                  0
      port-map-end                    0
      in-manipulationid
      out-manipulationid
      manipulation-string
      manipulation-pattern
      sip-ims-feature                 disabled
      operator-identifier
      anonymous-priority              none
      max-incoming-conns              0
      per-src-ip-max-incoming-conns   0
      inactive-conn-timeout           0
      untrusted-conn-timeout          0
      network-id
      ext-policy-server
      default-location-string
      charging-vector-mode            pass
      charging-function-address-mode pass
      ccf-address
      ecf-address
      term-tgrp-mode                  none
      implicit-service-route          disabled
      rfc2833-payload                 101
      rfc2833-mode                    transparent
      constraint-name
      response-map
      local-response-map
      ims-aka-feature                 disabled
      enforcement-profile
      route-unauthorized-calls
      tcp-keepalive                   none
      add-sdp-invite                  disabled
      add-sdp-profiles                disabled
      sip-profile
      sip-isup-profile
      tcp-conn-dereg                  0
      last-modified-by                admin@console
      last-modified-date              2012-09-11 09:04:51
```

ANNOTATION: The ChangeSupported sip-manipulation below applies the Add100Rel header rule to SIP messages between the Cable and wireless SIP trunk service and the session manager. This replaced all the parameters in the supported header from all replies (18x messages) with 100Rel effectively removing the "timer" parameter, as this was found to cause no ringback during call transfers.

```
sip-manipulation
      name                            Add100Rel
      description
      split-headers
      join-headers
      header-rule
            name                            Insert100rel
            header-name                     Supported
            action                          manipulate
            comparison-type                 case-sensitive
            msg-type                        reply
            methods
            match-value
            new-value                       100rel
      last-modified-by                admin@10.10.2.27
```

```
        last-modified-date            2012-10-18 07:12:32
```

ANNOTATION: The steering pools below define the IP Addresses and RTP port ranges on the respective realms. The “OUTSIDE” realm IP Address will be used as the media IP Address to communicate with Cable and Wireless. Likewise, the IP Address and RTP port range defined for the “INSIDE” realm steering pool will be used to communicate with the communication manager and endpoints.

```
steering-pool
        ip-address                    10.10.9.63
        start-port                    2048
        end-port                      3329
        realm-id                      INSIDE
        network-interface
        last-modified-by              admin@10.10.2.27
        last-modified-date            2012-10-10 04:23:02
steering-pool
        ip-address                    192.168.37.2
        start-port                    10000
        end-port                      20000
        realm-id                      OUTSIDE
        network-interface
        last-modified-by              admin@10.10.2.27
        last-modified-date            2012-10-10 04:26:42
system-config
        hostname
        description
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled                  enabled
        enable-snmp-auth-traps        disabled
        enable-snmp-syslog-notify     disabled
        enable-snmp-monitor-traps     disabled
        enable-env-monitor-traps      disabled
        snmp-syslog-his-table-length  1
        snmp-syslog-level             WARNING
        system-log-level              WARNING
        process-log-level             NOTICE
        process-log-ip-address        0.0.0.0
        process-log-port              0
        collect
                sample-interval               5
                push-interval                 15
                boot-state                    disabled
                start-time                    now
                end-time                      never
                red-collect-state             disabled
                red-max-trans                 1000
                red-sync-start-time           5000
                red-sync-comp-time            1000
                push-success-trap-state       disabled
        call-trace                    disabled
        internal-trace                disabled
        log-filter                    all
        default-gateway               10.10.9.1
        restart                       enabled
        exceptions
        telnet-timeout                0
```

```
       console-timeout                0
       remote-control                 enabled
       cli-audit-trail                enabled
       link-redundancy-state          disabled
       source-routing                 enabled
       cli-more                       disabled
       terminal-height                24
       debug-timeout                  0
       trap-event-lifetime            0
       default-v6-gateway             ::
       ipv6-support                   disabled
       ipv6-signaling-mtu             1500
       ipv4-signaling-mtu             1500
       cleanup-time-of-day            00:00
       last-modified-by               admin@console
       last-modified-date             2012-09-11 08:48:00
task done
acmesystem#
```

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.