Avaya Solution & Interoperability Test Lab

Front-Ending Avaya Communication Server 1000 R4.5 with an Avaya G450 Media Gateway Controlled by Avaya Aura™ Communication Manager 5.2.1 to Support SIP Trunks to Avaya Aura™ Session Manager 5.2 and Avaya Modular Messaging 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration that uses an Avaya G450 Media Gateway as a PRI-QSIG/SIP gateway to connect Avaya Communication Server 1000 R4.5 (formerly known as Nortel Communication Server 1000) with Avaya Aura™ Session Manager 5.2, which in turn provides call routing support to other Avaya SIP products such as Avaya Modular Messaging 5.2.

For the sample configuration, Session Manager runs on an Avaya S8510 Server, Communication Manager runs on Avaya S8720 servers, and Avaya Communication Server 1000 runs on Avaya Communication Server 1000S. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Communication Manager.

1 Introduction

Previous Avaya Application Notes [9] describe how Release 4.5 Avaya Communication Server 1000 (formerly known as Nortel Communication Server 1000 and hereafter referred to as the CS1000) can be directly integrated with Avaya Aura™ Session Manager using SIP trunks. While effective in terms of supporting basic and supplementary call features, this configuration does have some limitations in areas such as DTMF support and call coverage[1]. There are also many installations of the CS1000 which are not SIP or IP capable. In these cases, an effective solution is to front-end the CS1000 with a PRI-QSIG/SIP gateway, which then signals on SIP trunks to Session Manager. This configuration supports basic and supplementary call features as well as RFC 2833 DTMF and message-waiting signaling for applications such as voice messaging. See [10] for one example of this technique using an AudioCodes Mediant 1000 Modular Media Gateway.

The sample configuration shown in **Figure 1** illustrates another example of front-ending using an Avaya G450 Media Gateway as the PRI-QSIG/SIP gateway. The G450 Media Gateway is controlled by Avaya Aura™ Communication Manager, which supports SIP trunks to the SM-100 (Security Module) network interface of Session Manager, which in turn performs call routing to Avaya Modular Messaging. Session Manager can support flexible inter-system call routing based on dialed number, calling number and system location, and can also provide protocol adaptation to allow multi-vendor systems to interoperate. It is managed by a separate Avaya Aura™ System Manager, which can manage multiple Session Managers by communicating with their management network interfaces. Modular Messaging expands the capabilities and features of messaging services. Centralized messaging enables the local Modular Messaging system to provide voicemail service to subscribers at both sites in a multi-site configuration.

For the sample configuration, Session Manager runs on an Avaya S8510 Server, Communication Manager runs on Avaya S8720 servers, and the CS1000 runs on Avaya Communication Server 1000S. These Application Notes should apply to other Avaya servers and Media Gateways running Communication Manager.

As shown in **Figure 1**, Communication Manager controls the G450 Media Gateway, Avaya 9630 IP Telephone (H.323), and 6408D+ Digital Telephone. The CS1000 controls the Avaya i2004 IP Telephone and 3904 Digital Telephone (formerly sold under the Nortel label). A five digit Uniform Dial Plan (UDP) is used for dialing between systems. Unique extension ranges are associated with Communication Manager (3xxxx) and Avaya Communication Server 1000 (53xxx). Session Manager routes calls based on this five digit plan, using an adaptation module to convert to the normalized eleven digit plan used in Modular Messaging.

These Application Notes will focus on configuration of the QSIG trunks, SIP trunks, dial plan support, call routing, and call coverage for voice messaging. Detailed administration of the endpoint telephones will not be described (see the appropriate documentation listed in **Section 9**).

[1] These limitations are resolved in later releases of Avaya Communication Server 1000.

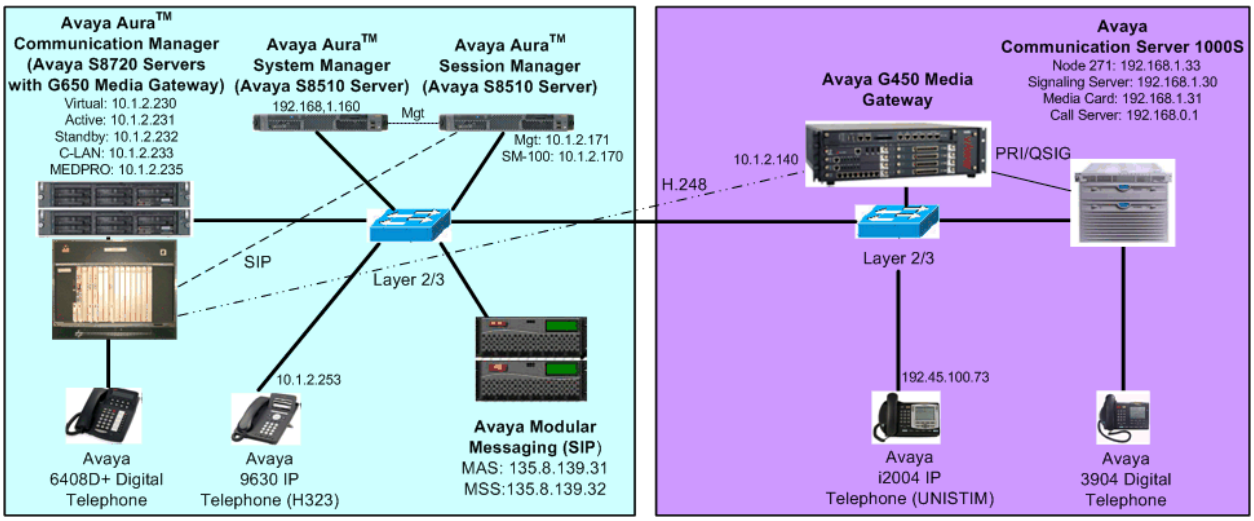


Figure 1 – Sample Configuration

2 Equipment and Software Validated

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

Hardware Component	Software Version
Avaya S8720 Servers with G450 and G650 Media Gateways	Avaya Aura™ Communication Manager 5.2.1, Load 16.4, Update 17774
Avaya S8510 Server	Avaya Aura™ Session Manager 5.2 SP 0, Load 5.2.0.1.520017
	Avaya Aura™ System Manager 5.2 Load 5.2.0.7.11 VSP patch 1.1.0.4.8
Avaya 9630 IP Telephone (H.323)	3.1
Avaya 6408D+ Digital Telephone	-
Avaya Modular Messaging Storage Server	5.2, Build 5.2-11.0
Avaya Modular Messaging Application Server	5.2, Build 5.2.150.13 (Patch 520008)
Avaya Communication Server 1000S • Call Server • Signaling Server • NTRB21 DTI/PRI TMDI Card	Avaya Communication Server 1000 Release 450w, Version 2121 sse-4.50.88 NA
Avaya (formerly Nortel) 3904 Digital Telephone	NA
Avaya (formerly Nortel) I2004 IP Telephone (UNISTIM)	C502B41

3 Configure Avaya Aura™ Communication Manager

This section describes configuring Communication Manager in the following areas. Some administration screens have been abbreviated for clarity.

- Avaya Communication Manager license
- System parameters features
- IP node names
- IP interface
- IP codec set and network region
- G450 Media Gateway
- DS1 Interface
- PRI QSIG signaling group and trunk group
- SIP signaling group and trunk group
- Route pattern
- Location and public/private numbering
- Uniform dial plan and AAR analysis
- Voice messaging hunt group
- Voice messaging coverage path
- Sample station form specifying voice messaging coverage path

3.1 Verify Avaya Aura™ Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

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

```
display system-parameters customer-options                    Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 800   200
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   47
```

3.2 Configure System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers. Submit the change.

This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch For simplicity, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 9** for more details.

```
change system-parameters features                                Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
```

3.3 Configure IP Node Names

Use the "change node-names ip" command to add entries for the C-LAN that will be used for signaling, its default gateway, and Session Manager. In this case, "clan1" and "10.1.2.233" are entered as **Name** and **IP Address** for the C-LAN, "sm1" and "10.1.2.170" are entered for the Session Manager Security Module (SM-100) interface, and "Gateway001" and "10.1.2.1" are entered for the default gateway. Note that "Gateway001" will be used to configure the IP interface for the C-LAN (see **Section 3.4**). The actual node names and IP addresses may vary. Submit these changes.

```
change node-names ip                                             Page   1 of   2
                                   IP NODE NAMES
    Name              IP Address
clan1               10.1.2.233
Gateway001          10.1.2.1
sm1                 10.1.2.170
```

3.4 Configure IP Interface for C-LAN

Add the C-LAN to the system configuration using the "add ip-interface 1a02" command. The actual slot number may vary. In this case, "1a02" is used as the slot number. Enter the C-LAN node name assigned from **Section 3.3** into the **Node Name** field.

Enter proper values for the **Subnet Mask** and **Gateway Node Name** fields. In this case, "24" and "Gateway001" are used to correspond to the network configuration in these Application Notes. Set the **Enable Interface** and **Allow H.323 Endpoints** fields to "y". Default values may be used in the remaining fields. Submit these changes.

```
add ip-interface 1a02                                          Page   1 of   3
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A02        Target socket load and Warning level: 400
           Code/Suffix: TN799  D           Receive Buffer TCP Window Size: 8320
      Enable Interface? y                           Allow H.323 Endpoints? y
                  VLAN: n                            Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5



                                 IPV4 PARAMETERS
             Node Name: clan1
           Subnet Mask: /24
     Gateway Node Name: Gateway001

         Ethernet Link: 2
         Network uses 1's for Broadcast Addresses? y
```

3.5 Configure IP Codec Set and Network Region

Configure the IP codec set to use for calls to the Avaya Communication Server 1000 via Session Manager. Use the "change ip-codec-set n" command, where "n" is an existing codec set number to be used for interoperability. Enter the desired audio codec type in the **Audio Codec** field. Retain the default values for the remaining fields and submit these changes.

```
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:
```

In the test configuration, network region "1" was used for calls to Session Manager. Use the "change ip-network-region 1" command to configure this network region. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise network (See **Section 4.1)**. This value is used to populate the SIP domain in the From header of SIP INVITE messages for outbound calls. It also must match the SIP domain in the request URI of incoming INVITEs from other systems. Enter a descriptive **Name**. For the **Codec Set** field, enter the corresponding audio codec set configured above in this section. Enable the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio**. These settings will enable direct media between Avaya IP telephones. Retain the default values for the remaining fields, and submit these changes.

```
change ip-network-region 1                                      Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: ASM
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: 10001
DIFFSERV/TOS PARAMETERS                 RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
```

3.6 Add G450 Media Gateway

The Avaya G450 Media Gateway is used to support the PRI QSIG trunk connection to the CS1000. Install and configure the G450 Media Gateway as described [6], noting its serial number, and specifying the IP address of the C-LAN configured in **Section 3.3** in its controller list. The following screen shows the G450 Media Gateway Command Line Interface commands to obtain the serial number (**show system**), and to set and verify the controller list (**set mgc list**, **show mgc list**):

```
G450-FE-???(super)# show system
System Name             :
System Location         :
System Contact          :
Uptime (d,h:m:s)        : 46,21:32:25
MV Time                 : 09:51:53 11 MAR 2010
Serial No               : 08IS38199678
Model                   : G450
.
.
.
G450-FE-???(super)# set mgc list 10.1.2.233
Done!
G450-FE-???(super)# show mgc list

CONFIGURED MGC HOST
---------------------
10.1.2.233
-- Not Available --
-- Not Available --
-- Not Available --
```

On Communication Manager, use the “add media-gateway n” command, where “n” is an unused media gateway number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Type:** “g450”
- **Name:** A descriptive name.
- **Serial No:** Serial number obtained from the G450 media gateway above

```
add media-gateway 1                                              Page   1 of   1
                              MEDIA GATEWAY
         Number: 1                              Registered?  n
           Type: g450                 FW Version/HW Vintage:
           Name: Avaya CS1000                MGP IP Address:
      Serial No: 08IS38199678         Controller IP Address:
   Encrypt Link? y                              MAC Address:
 Network Region: 1    Location: 1
                                                Site Data:
  Recovery Rule: none

Slot   Module Type            Name                     DSP Type  FW/HW version
 V1:
 V2:
 V3:
```

Make sure that the DS1 interface card (MM710) is installed in the desired slot in the gateway. When the media gateway is registered with Communication Manager, the DS1 interface should be displayed in that slot, as shown below for the sample configuration.

```
display media-gateway 1
                              MEDIA GATEWAY
         Number: 1                              Registered?  y
           Type: g450                 FW Version/HW Vintage: 30 .10 .4  /1
           Name: Avaya CS1000                MGP IP Address: 10 .1  .2  .140
      Serial No: 08IS38199678         Controller IP Address: 10 .1  .2  .233
   Encrypt Link? y                              MAC Address: 00:1b:4f:03:52:18
 Network Region: 1    Location: 1
                                                Site Data:
  Recovery Rule: none

Slot   Module Type            Name                     DSP Type  FW/HW version
 V1:   MM710                  DS1 MM                   MP80      29   3
 V2:
 V3:
```

3.7 Add DS1 Interface

The DS1 circuit pack is used for connectivity to the CS1000. Use the "add ds1 1v1" command. Note that the actual slot number may vary. In this case "1v1" is used as the slot number (see **Section 3.6**). Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Name:** A descriptive name.
- **Line Coding:** "b8zs"
- **Framing Mode:** "esf"
- **Signaling Mode:** "isdn-pri"
- **Connect:** "pbx"
- **Interface:** "peer-slave"
- **Peer Protocol:** "Q-SIG"

The **Interface** field must be complementary on both switches. For the sample configuration, Communication Manager is administered as the *peer-slave*, and the CS1000 is administered as the *peer-master* (note that **Reference** [11] shows the opposite relationship).

```
add ds1 01v1                                                    Page   1 of   2
                                 DS1 CIRCUIT PACK

            Location: 001V1                            Name: Avaya CS1000
            Bit Rate: 1.544                     Line Coding: b8zs
   Line Compensation: 1                        Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                          Interface: peer-slave
   TN-C7 Long Timers? n                       Peer Protocol: Q-SIG
Interworking Message: PROGress                         Side: b
Interface Companding: mulaw                             CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz
```

3.8 Add PRI QSIG Signaling Group and Trunk Group

Configure an ISDN trunk group to interface with the CS1000. Use the "add trunk-group n" command, where "n" is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "isdn"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Direction:** "two-way"
- **Carrier Medium:** "PRI/BRI"
- **Service Type:** "tie"

```
add trunk-group 100                                                Page   1 of  21
                                 TRUNK GROUP

Group Number: 100                  Group Type: isdn          CDR Reports: y
  Group Name: Avaya CS1000                COR: 1        TN: 1        TAC: 100
   Direction: two-way        Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? n              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n            TestCall ITC: rest
                          Far End Test Line No:
```

Navigate to **Page 2**. For the **Supplementary Service Protocol** field, enter "b" for QSIG. Retain the default values for the remaining fields.

```
add trunk-group 100                                                Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                  Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
```

Navigate to **Page 3**. Enable the **Send Name**, **Send Calling Number**, and **Send Connected Number** fields. For the **Format** field, enter "unk-pvt" to construct the calling and connected numbers using the "private numbering" table, but encode the numbering plan format as "unknown" in the ISDN messages toward the CS1000. Setting the **Internal Alert** field to "y" allows calls arriving fromCS1000 users to be treated as internal calls. For example, if a CS1000 telephone dials a Communication Manager telephone, the Communication Manager telephone will ring with the ring pattern for an internal station-station call, internal coverage criteria will apply, and the CS1000 caller will hear tones such as coverage tone, similar to a call between Communication Manager telephones.

```
add trunk-group 100                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? y        Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n              Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unk-pvt
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? n
                                                        Send Connected Number: y
                                                    Hold/Unhold Notifications? y
             Send UUI IE? y                       Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                           Ds1 Echo Cancellation? n

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                               Network (Japan) Needs Connect Before Disconnect? n
```

3.8.1 Signaling Group

Configure an ISDN signaling group for the new trunk group. Use the "add signaling-group n" command, where "n" is an available signaling group number. For the **Primary D-Channel** field, enter the slot number for the DS1 module from **Section 3.7** and port "24".

For the **Trunk Group for NCA TSC** and **Trunk Group for Channel Selection** fields, enter the ISDN trunk group number from **Section 3.8**. For the **Supplementary Service Protocol** field, enter "b" for QSIG. Maintain the default values for the remaining fields, and submit these changes.

```
add signaling-group 100                                      Page   1 of   1
                                 SIGNALING GROUP

 Group Number: 100            Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 10
                       Primary D-Channel: 001V124     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: 100
       Trunk Group for Channel Selection: 100
      TSC Supplementary Service Protocol: b          Network Call Transfer? n
```

3.8.2 Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number added in **Section 3.8**. Navigate to **Page 4.** Shown below are default values that were used during testing. If the Communication Manager Auto Callback feature will be used with CS1000 users, then the **TSC Method for Auto Callback** field must be set to "always-retain", as shown in bold below.

```
change trunk-group 100                                          Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS




   TSC Method for Auto Callback: always-retain
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                            SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
               QSIG Value-Added? n
```

Navigate to **Pages 5** and **6**. Enter all 23 ports of the DS1 module into the **Port** fields, and the corresponding **Code** and **Sfx** fields will be populated automatically. Enter the ISDN signaling group number from **Section 3.8.1** into the **Sig Grp** fields as shown below. Submit these changes.

```
change trunk-group 100                                          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: 001V101  MM710                                  100
  2: 001V102  MM710                                  100
  3: 001V103  MM710                                  100
  4: 001V104  MM710                                  100
  5: 001V105  MM710                                  100
  6: 001V106  MM710                                  100
  7: 001V107  MM710                                  100
  8: 001V108  MM710                                  100
  9: 001V109  MM710                                  100
 10: 001V110  MM710                                  100
 11: 001V111  MM710                                  100
 12: 001V112  MM710                                  100
 13: 001V113  MM710                                  100
 14: 001V114  MM710                                  100
 15: 001V115  MM710                                  100
```

```
change trunk-group 100                                          Page   6 of  21
                                  TRUNK GROUP
                                       Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:  23

       Port    Code Sfx Name        Night           Sig Grp
 16: 001V116  MM710                                  100
 17: 001V117  MM710                                  100
 18: 001V118  MM710                                  100
 19: 001V119  MM710                                  100
 20: 001V120  MM710                                  100
 21: 001V121  MM710                                  100
 22: 001V122  MM710                                  100
 23: 001V123  MM710                                  100
```

3.9 Configure SIP Signaling Group and Trunk Group

3.9.1 SIP Signaling Group

In the test configuration, trunk group "32" and signaling group "32" were used to reach Session Manager. Use the "add signaling-group n" command, where "n" is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** "sip"
- **Transport Method:** "tls"
- **Near-end Node Name:** C-LAN node name from **Section 3.3**.
- **Far-end Node Name:** Session Manager node name from **Section 3.3**.
- **Near-end Listen Port:** "5061"
- **Far-end Listen Port:** "5061"
- **Far-end Network Region:** Network region number "1" from **Section 3.5**.
- **Far-end Domain:** SIP domain name from **Section 4.1**.
- **DTMF over IP:** "rtp-payload"

```
add signaling-group 32                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 32             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n


   Near-end Node Name: clan1                 Far-end Node Name: sm1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain: avaya.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n                    Direct IP-IP Early Media? n
 Session Establishment Timer(min): 3             Alternate Route Timer(sec): 6
```

3.9.2 SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Number of Members:** The number of SIP trunks allocated for calls routed to Session Manager (must be within the limits of the total trunks configured in **Section 3.1**).

```
add trunk-group 32                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 32                   Group Type: sip           CDR Reports: y
  Group Name: To SM1                      COR: 1        TN: 1        TAC: 132
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 32
                                                   Number of Members: 4
```

Navigate to **Page 3**, and enter "public" for the **Numbering Format** field as shown below. Use default values for all other fields. Submit these changes.

```
add trunk-group 32                                                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
```

3.10 Configure Route Patterns

Create a route pattern to use for routing calls to the CS1000 using the PRI QSIG trunk. Use the "change route-pattern n" command, where "n" is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The trunk group number from **Section 3.8**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.
- **TSC:** "y" (NCA-TSCs will be used)
- **CA-TSC Request:** "none" (since CA-TSC are used for DCS but not for QSIG)
- **Numbering Format:** "unk-unk" (The numbering format and type of number for the Called Party Number will be encoded as "unknown" toward the CS1000).

```
change route-pattern 100                                        Page   1 of   3
                    Pattern Number: 100 Pattern Name: Avaya CS1000
                              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: 100  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  y  none      rest                                unk-unk   none
 2: y y y y y n  n            rest                                          none
```

Configure a route pattern for routing calls to Session Manager using the SIP trunk group. Use the "change route-pattern n" command, where "n" is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 3.9.2**.
- **FRL:** Enter a level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 32                                          Page   1 of   3
                    Pattern Number: 32  Pattern Name: To ASM
                             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: 32   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                                      none
```

3.11 Configure Location and Public/Private Numbering

Use the "change locations" command to specify the SIP route pattern to be used as a "default SIP route" for the location corresponding to the Main site. In this way, calls to non-numeric users or unknown domains will still be routed to Session Manager. Add an entry for the Main site if one does not exist already, enter the following values for the specified fields, and retain default values for the remaining fields. Submit these changes.

- **Name:** A descriptive name to denote the Main site.
- **Timezone:** An appropriate timezone offset.
- **Rule:** An appropriate daylight savings rule.
- **Proxy Sel. Rte. Pat.:** The SIP route pattern number from the previous section

```
change locations                                                 Page   1 of   1
                                    LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc  Name           Timezone Rule  NPA                                Proxy Sel
 No                   Offset                                             Rte Pat
 1:  Main            + 00:00   0                                          32
```

Use the "change public-unknown-numbering 0" command, to define the calling party number to be sent to Session Manager. Add an entry for the trunk group defined in **Section 3.9.2**. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed to trunk group 32 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header. Submit these changes.

```
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: 3
 5  3             32                         5           Maximum Entries: 9999
```

Use the "change private-numbering" command to define the calling party number to be sent to the CS1000. Add an entry for the trunk group defined in **Section 3.8**. As shown below, all calls originating from a 5-digit extension beginning with 3 and routed to trunk group 100 will result in the 5-digit calling number to be sent. Submit these changes.

```
change private-numbering 0                                       Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  3              100                         5     Total Administered: 2
                                                        Maximum Entries: 540
```

3.12 Configure Dial Plan and AAR Analysis

Configure dial plan and Automatic Alternate Routing (AAR) used for routing calls with dialed digits 53xxx to the CS1000 via the G450 Media Gateway and for calls covering to Modular Messaging via hunt group extension 33000. Use the "change uniform-dialplan 0" command, and add an entry to specify use of AAR for routing of digits 53xxx. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Matching Pattern:** Dialed prefix digits to match on, in this case "53".
- **Len:** Length of the full dialed number.
- **Del:** Number of digits to delete.
- **Net:** "aar"

Add an entry to cover calls that will cover to the voice messaging hunt group extension (33000). Session Manager will route these calls to Modular Messaging. Submit these changes.

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

   Matching                    Insert              Node
   Pattern        Len Del      Digits     Net Conv Num
  53               5   0                  aar  n
  3                5   0                  aar  n
```

Use the "change aar analysis 0" command, and add corresponding entries to specify use of the SIP trunk for non-extension numbers beginning with 3 (e.g. voice messaging hunt group 33000) and the PRI QSIG trunk for the calls to the CS1000 (53xxx). Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case "53" and "3".
- **Total Min:** Minimum number of digits.
- **Total Max:** Maximum number of digits.
- **Route Pattern:** The route pattern number from **Section 3.10**.
- **Call Type:** "aar" for voice messaging, "lev0" for private numbering (PRI/QSIG)

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    3                     5    5     32       aar          n
    53                    5    5     100      lev0         n
```

Use the "change dialplan analysis" command to define "8" as a feature access code. This will be used for AAR dialing in **Section 3.13**. Note also that the 3xxxx and 5xxxx extension ranges are defined in this form as well.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     2            5    ext
     3            5    ext
     5            5    ext
     6            5    ext
     7            5    ext
     8            1    fac
     9            1    fac
```

Use the "change feature-access-codes" command to assign the feature access code "8" to AAR.

```
change feature-access-codes                                    Page   1 of   8
                                FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *01
         Abbreviated Dialing List2 Access Code: *02
         Abbreviated Dialing List3 Access Code: *03
Abbreviated Dial - Prgm Group List Access Code: *04
                      Announcement Access Code: *05
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

3.13 Configure Voice Messaging Hunt Group

Use the "add hunt group n" command to add a hunt group to be used by the voice messaging coverage path to be defined in the next section. "n" is an unused hunt group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Number:** An unassigned hunt group number.
- **Group Name:** A meaningful name (Modular Messaging).
- **Group Extension:** An unassigned extension number.
- **Group Type:** "ucd-mia"
- **ISDN/SIP Caller Display:** "mbr-name"

```
add hunt-group 32                                                Page   1 of  60
                                   HUNT GROUP

            Group Number: 32                                 ACD? n
              Group Name: MM                               Queue? n
         Group Extension: 33000                           Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1           Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

On page 2, assign the following field values:

- **Message Center:** "sip-adjunct"
- **Voice Mail Number:** The **Group Extension** from Page 1.
- **Voice Mail Handle:** The **Group Extension** from Page 1.
- **Routing Digits:** The AAR feature access code from the previous section.

Submit these changes.

```
add hunt-group 32                                                Page   2 of  60
                                   HUNT GROUP





                       Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     33000                    33000                     8
```

3.14 Configure Voice Messaging Coverage Path

Use the "add coverage path n" command to specify a coverage path to be used for telephone users. This will specify use of the voice messaging hunt group. "n" is an unused coverage path number. Enter the hunt group number defined in the previous section in **Point 1**. Default values can be used for the remaining fields. It may be desirable to adjust the **Number of Rings** before a no-answer call goes to coverage.

```
add coverage path 32                                            Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 32
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
       Don't Answer?            y              y         Number of Rings: 2
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h32            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

3.15 Configure Coverage Path for Telephone Users

The following sample station form illustrates how to configure voice mail coverage for a given station user. Set **Coverage Path 1** to the value of the coverage path defined in the previous section.

```
change station 30001                                            Page   1 of   5
                                      STATION

Extension: 30001                         Lock Messages? n               BCC: 0
     Type: 9630                          Security Code: 123456           TN: 1
     Port: S00504                      Coverage Path 1: 32              COR: 1
     Name: AvayaH323                   Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
```

3.16 Save Translations

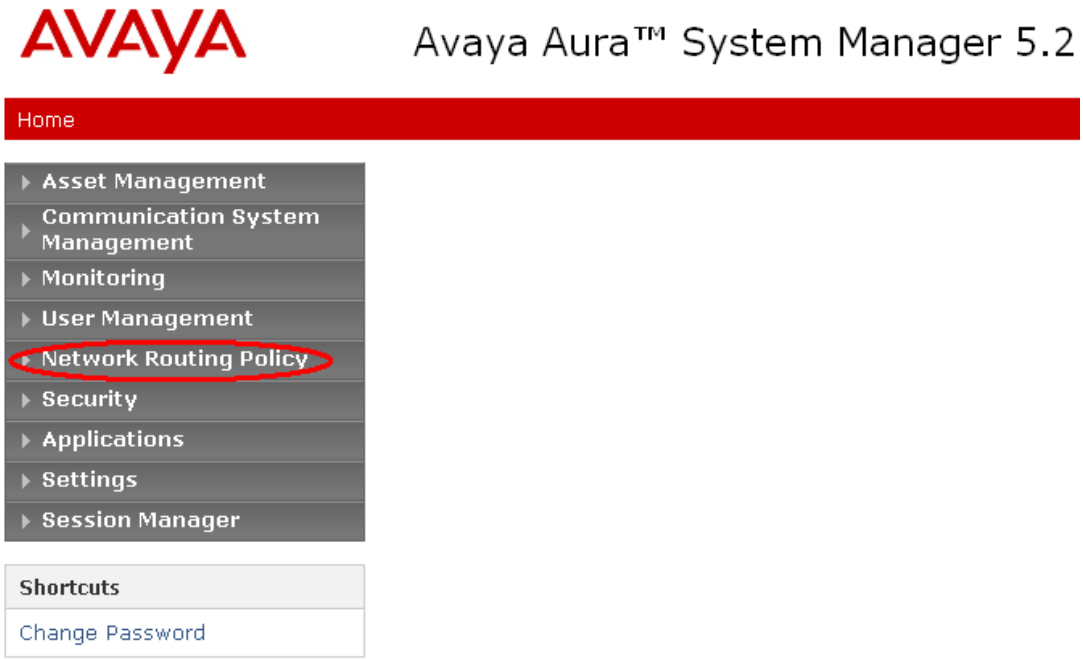
Configuration of Communication Manager is complete. Use the "save translations" command to save these changes.

4 Configure Avaya Aura™ Session Manager

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

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation for Modular Messaging
- SIP Entities corresponding to Communication Manager, CS 1000, Modular Messaging, and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager.
- Local host name resolution entries corresponding to fully qualified domain names (FQDN's) referenced in the previous steps.

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. Log in with the appropriate credentials and click on **OK** in the subsequent confirmation screen. The menu shown below is then displayed. Expand the **Network Routing Policy** Link on the left side as shown. The sub-menus displayed in the left column below will be used to configure all but the last two of the above items (**Sections 4.1** through **4.9**).

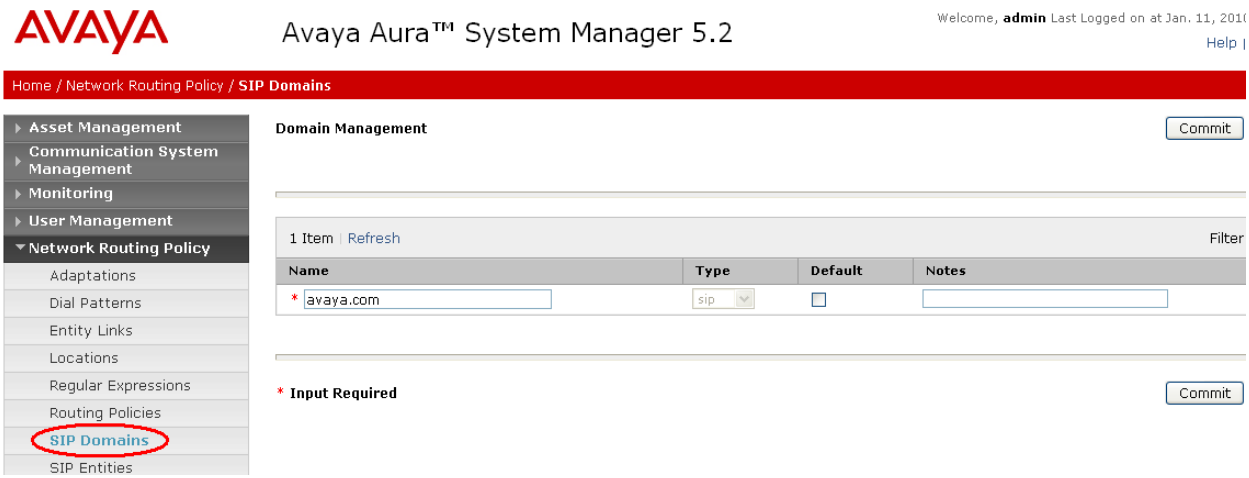


4.1 Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Select **SIP Domains** on the left and click the **New** button (not shown) on the right. Fill in the following:

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

Click **Commit**.



4.2 Add Location

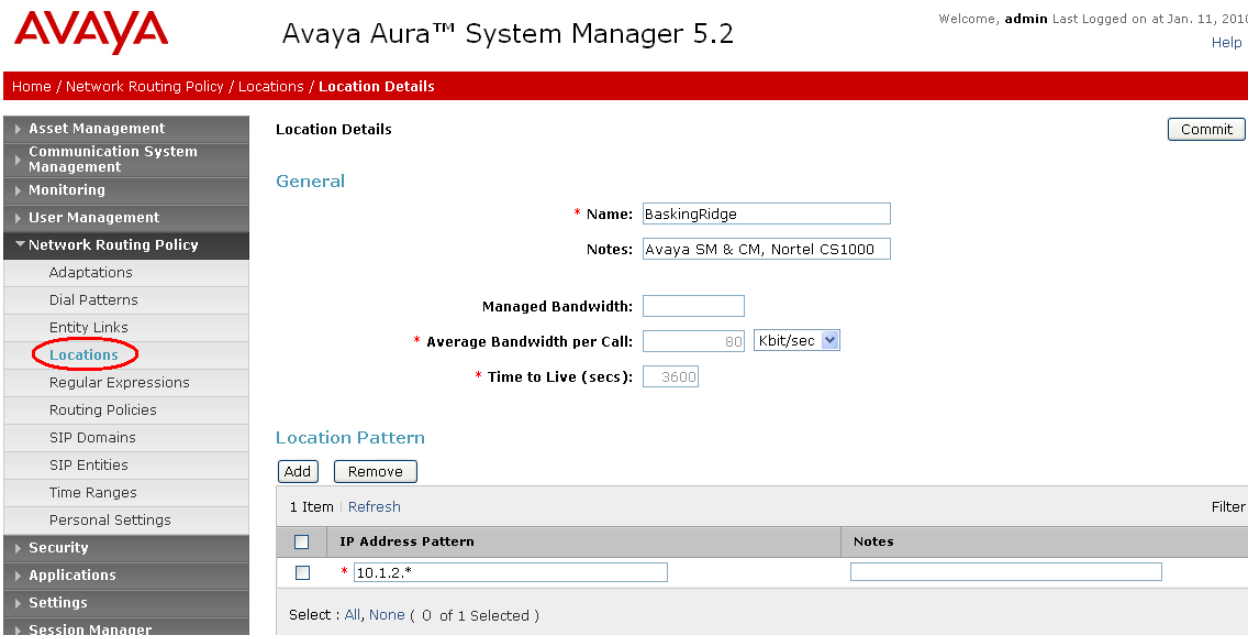
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. Under *General*, enter:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the Basking Ridge location, which includes Communication Manager, Session Manager, and Modular Messaging, and the CS1000 in the 10.1.2 subnet. Click **Commit** to save the Location definition.



The fields under *General* can be filled in to specify bandwidth management parameters between Avaya Session Manager and this location. These were not used in the sample configuration, and reflect default values. Note also that although not implemented in the sample configuration, routing policies can be defined based on location.

4.3 Add Adaptation Module

Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products. In the sample configuration, multi-site Modular Messaging represents its subscribers using 11 digit telephone numbers. The 5 digit extensions used by Communication Manager and the CS1000 are preceded by the 6 digits "120122". **DigitConversionAdapter** is used in Session Manager to convert between the 5 and 11 digit formats when routing between Modular Messaging and those systems.

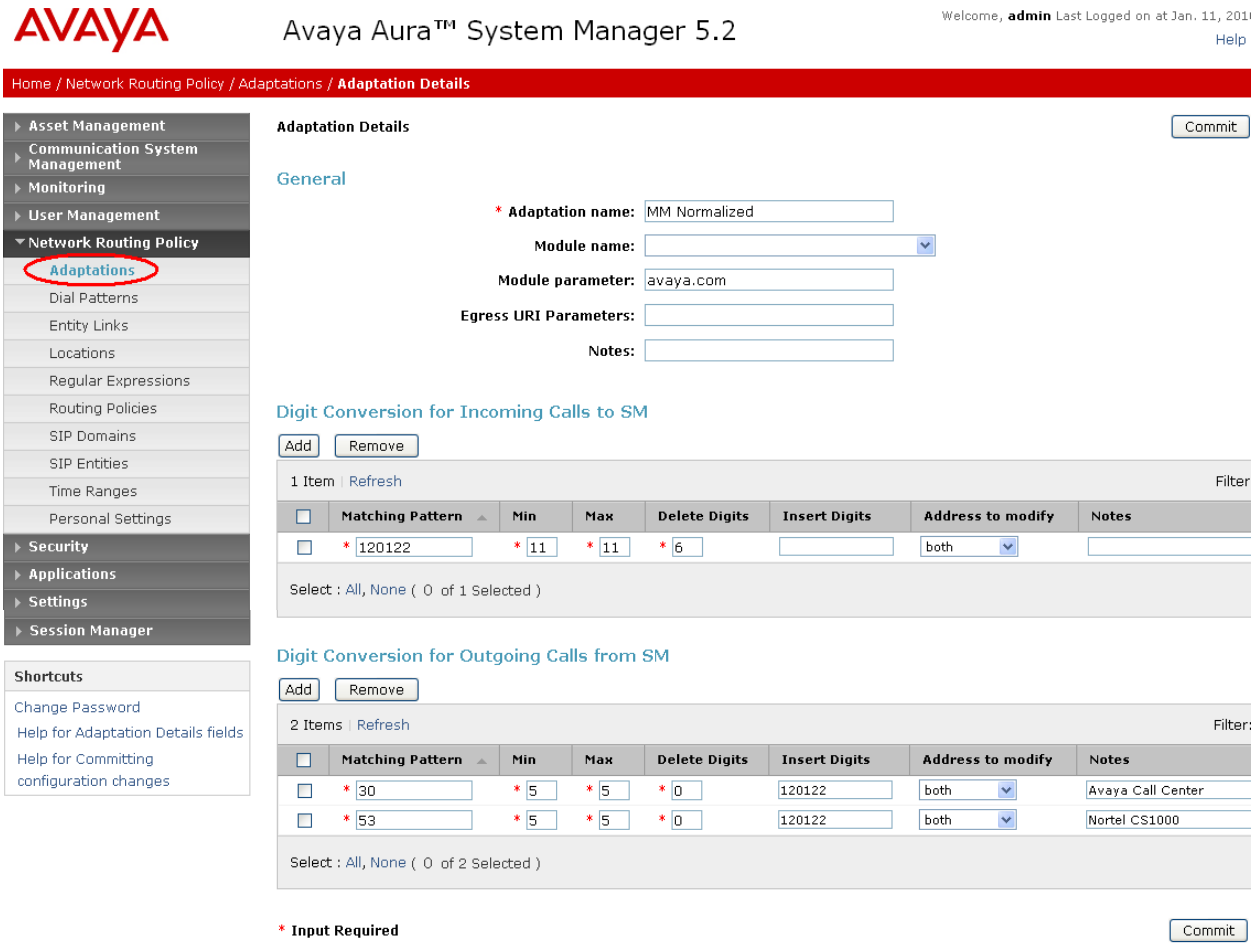
To add the generic adaptation module, select **Adaptations** on the left and click on the **New** button (not shown) on the right. Under *General*, fill in:

- **Name:** A descriptive name.
- **Adaptation Module:** Adaptation Module name and parameters (case sensitive)

The remaining fields can be left blank. Under *Digit Conversion for Incoming Calls to SM* and *Digit Conversion for Outgoing Calls from SM*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Matching Pattern:** A Reg-X expression or partial digit string used to match the incoming dialed number
- **Min:** Minimum dialed number length
- **Max:** Maximum dialed number length
- **Delete Digits:** Number of digits to delete from the beginning
- **Insert Digits:** Number of digits to insert at the beginning
- **Address to Modify:** Choose between "origination," "destination," or "both"

Click **Commit** to save the Adaptation Module definition. The screen below specifies **DigitConversionAdapter** and the SIP domain parameter "avaya.com" to be used when modifying the SIP messages. Incoming calls (SIP INVITE messages) from Modular Messaging that use 11 digit numbers will be converted to the 5 digit form by deleting the first 6 digits. Session Manager will route the call based on the resulting 5 digit extension. Calls routed to Modular Messaging will have their Request-URI, P-Asserted-Identity, and History-Info headers converted to 11 digit format by insertion of "120122" before being routed to Modular Messaging.



4.4 Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system supported by it using SIP trunks: the C-LAN board in the Avaya G650 Media Gateway and Modular Messaging. Select **SIP Entities** on the left and click on the **New** button (not shown) on the right. Under *General*, fill in:

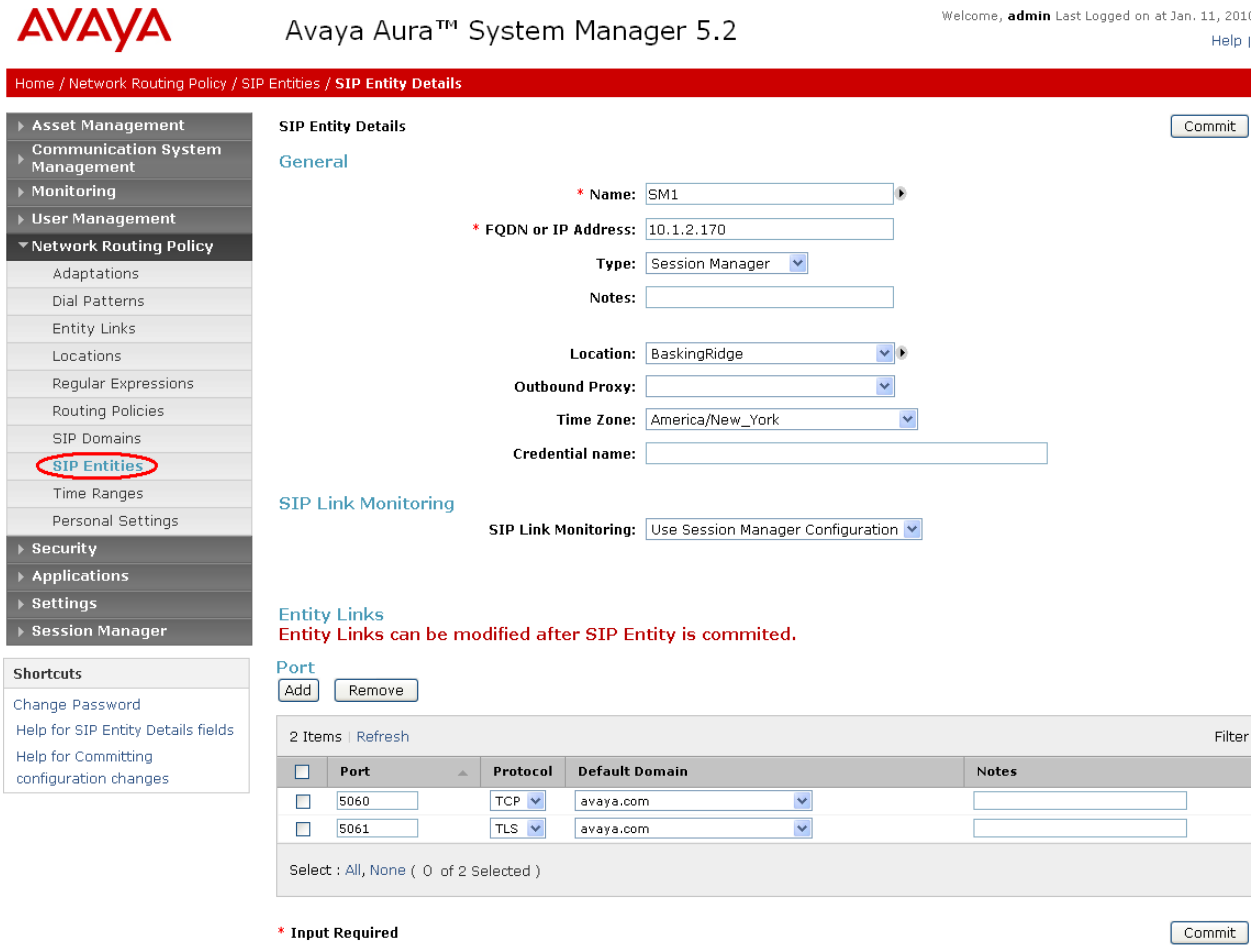
- **Name:** A descriptive name.
- **FQDN or IP Address:** FQDN or IP address of the Session Manager or the signaling interface on the telephony system.
- **Type:** "Session Manager" for Session Manager or "CM" for Communication Manager.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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

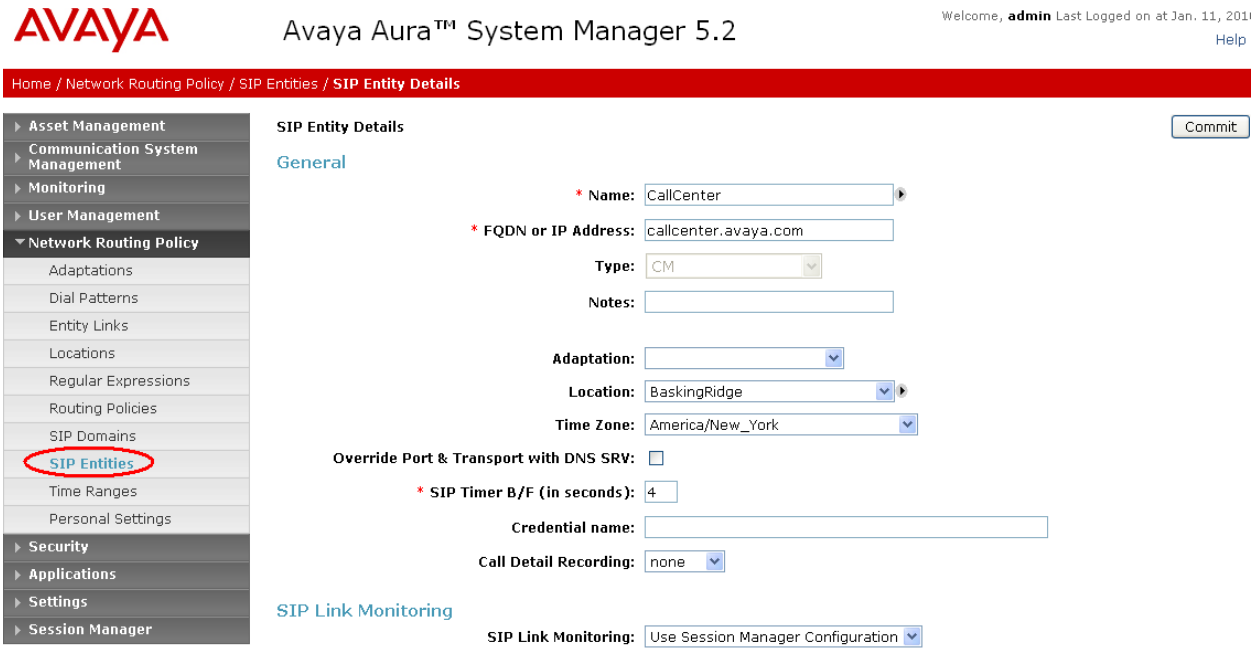
- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise (e.g., "avaya.com").

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

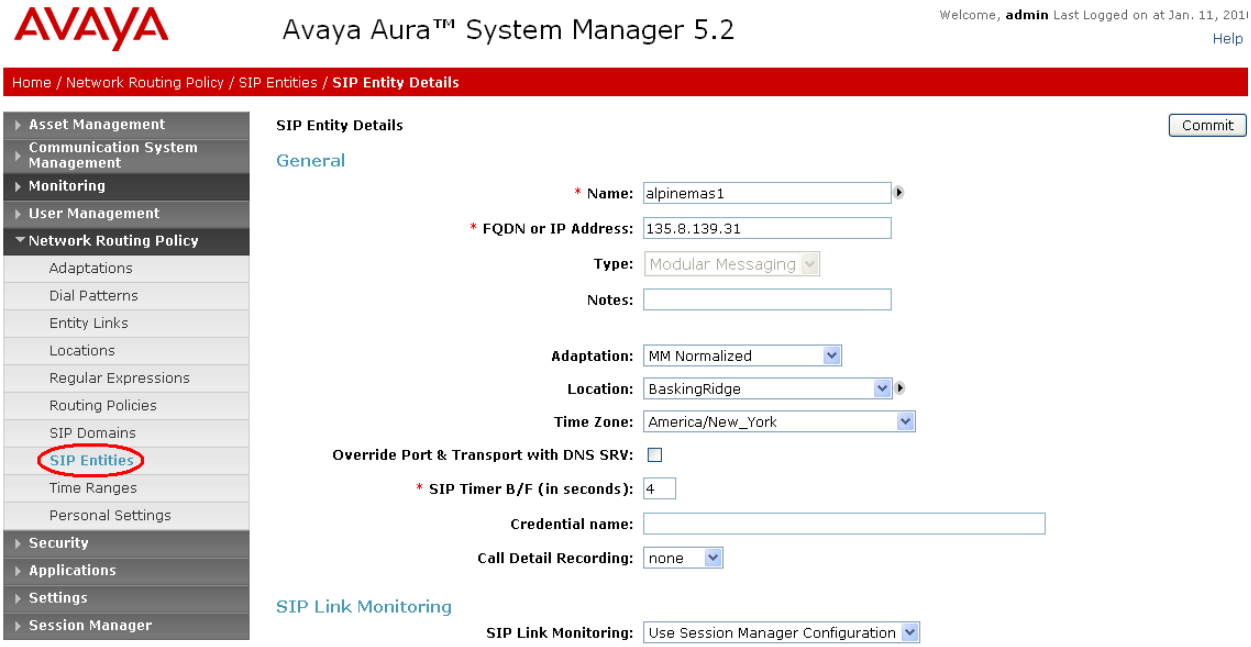
The following screen shows addition of Session Manager. The IP address of the SM-100 Security Module is entered for **FQDN or IP Address**. Two *Port* entries are added. TCP port 5060 is used for communicating with Modular Messaging and TLS port 5061 is used for communication with other Session Managers and Communication Manager.



The following screen shows addition of Communication Manager. In this case, **FQDN or IP Address** is the Fully Qualified Domain Name (FQDN) of the C-LAN board in the Avaya G650 Media Gateway. Note that although not shown in the sample configuration, definition of multiple IP addresses (e.g., C-LANs) for the same FQDN (see **Section 4.9)** will cause Session Manager to load balance call traffic among those addresses.



The following screen shows addition of the Modular Messaging Application Server (MAS) to which calls will be forwarded for busy/no-answer coverage of telephone users. **FQDN or IP Address** is the IP address of its network interface (see **Figure 1**). For **Adaptation**, select the adaptation module previously defined for dial plan digit manipulation in **Section 4.3**.

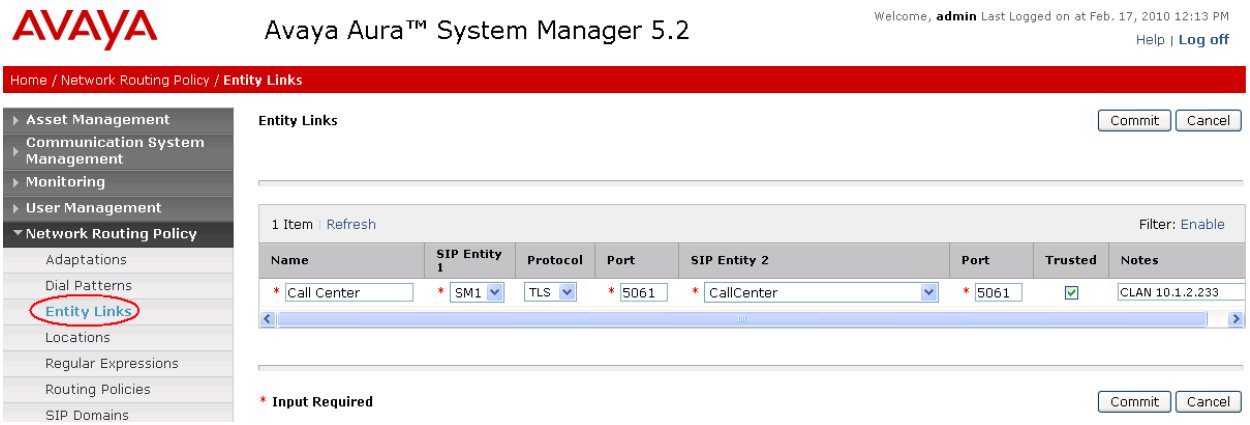


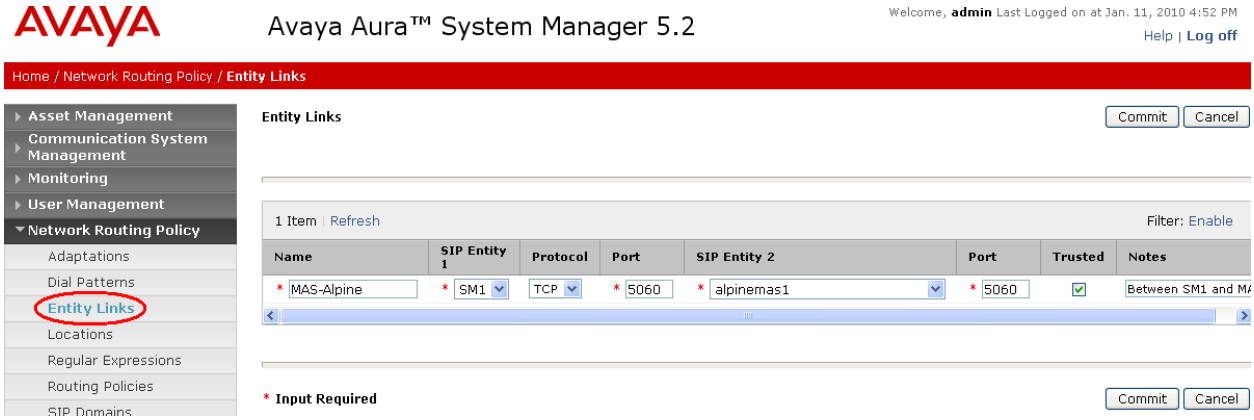
4.5 Add Entity Links

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

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in* ***Section 4.4*** *will be denied.*

Click **Commit** to save each Entity Link definition. The following screens illustrate adding the Entity Links for Communication Manager and Modular Messaging. TLS (well-known port 5061) is used for Avaya Communication Manager. TCP (well-known port 5060) was used for Modular Messaging.





4.6 Add Routing Policies

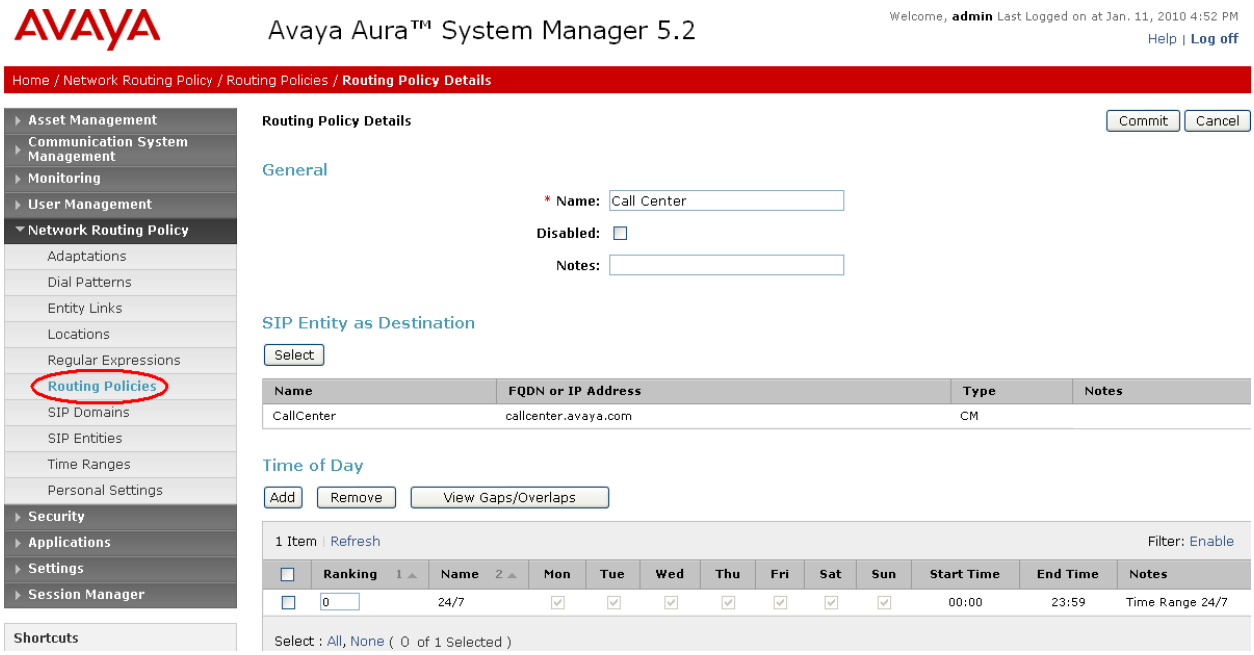
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 4.4**. Two routing policies must be added for Communication Manager and Modular Messaging. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

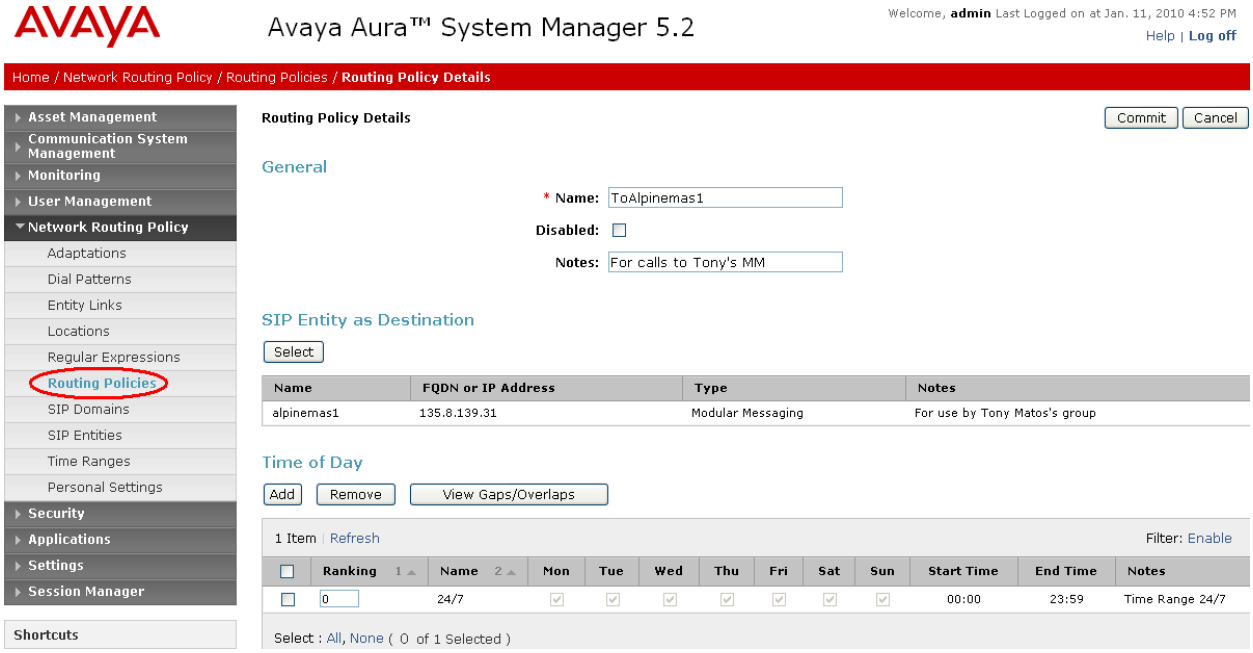
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Under *Time of Day*:
Select the default time range shown.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screens show the Routing Policies for Communication Manager and Modular Messaging.





4.7 Add Dial Patterns

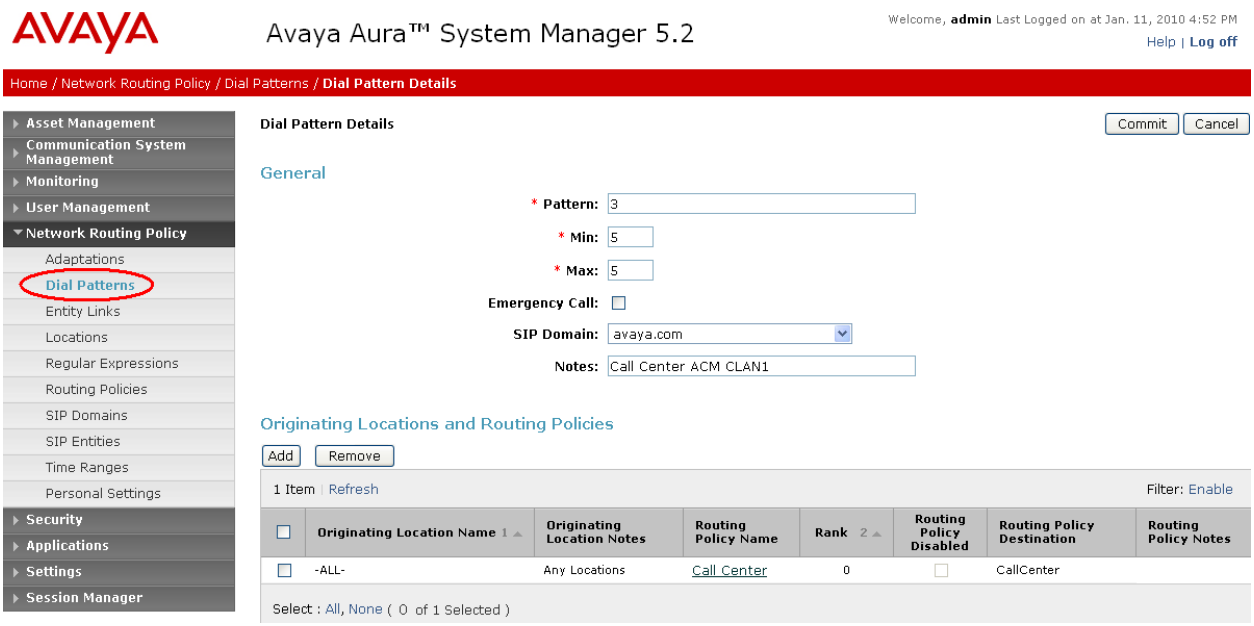
Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 5-digit extensions beginning with "3" or "53"should be routed to Communication Manager. The common access number for voice messaging for both systems is 33000, and calls to that number should be routed to Modular Messaging. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

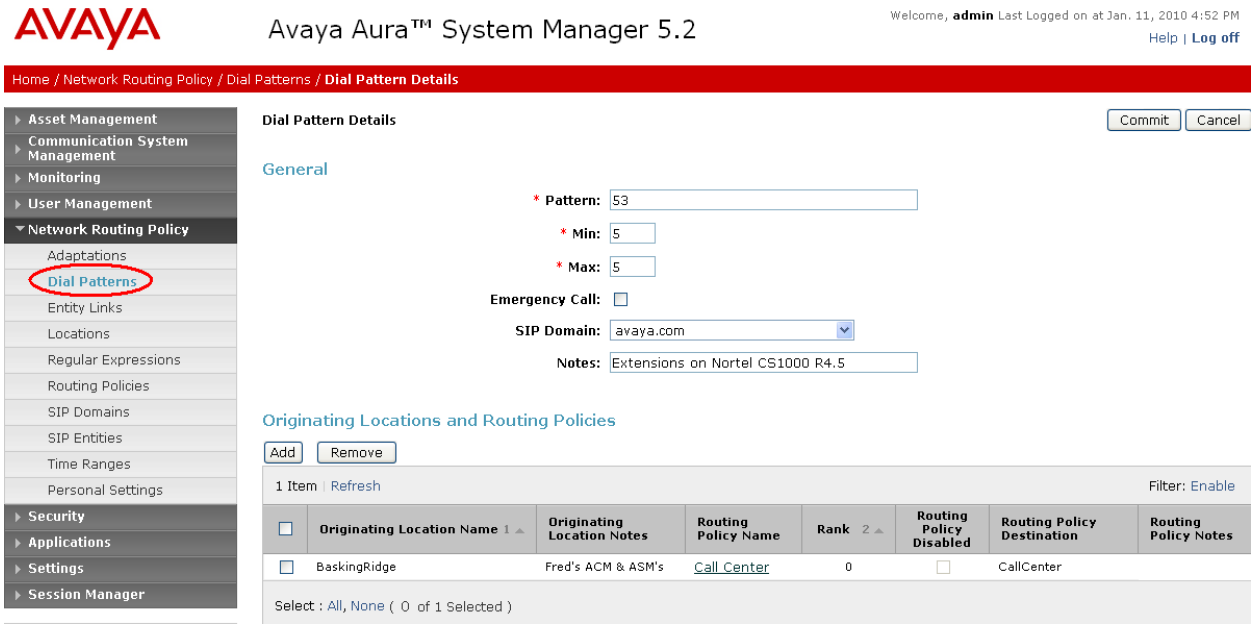
Under *General*:

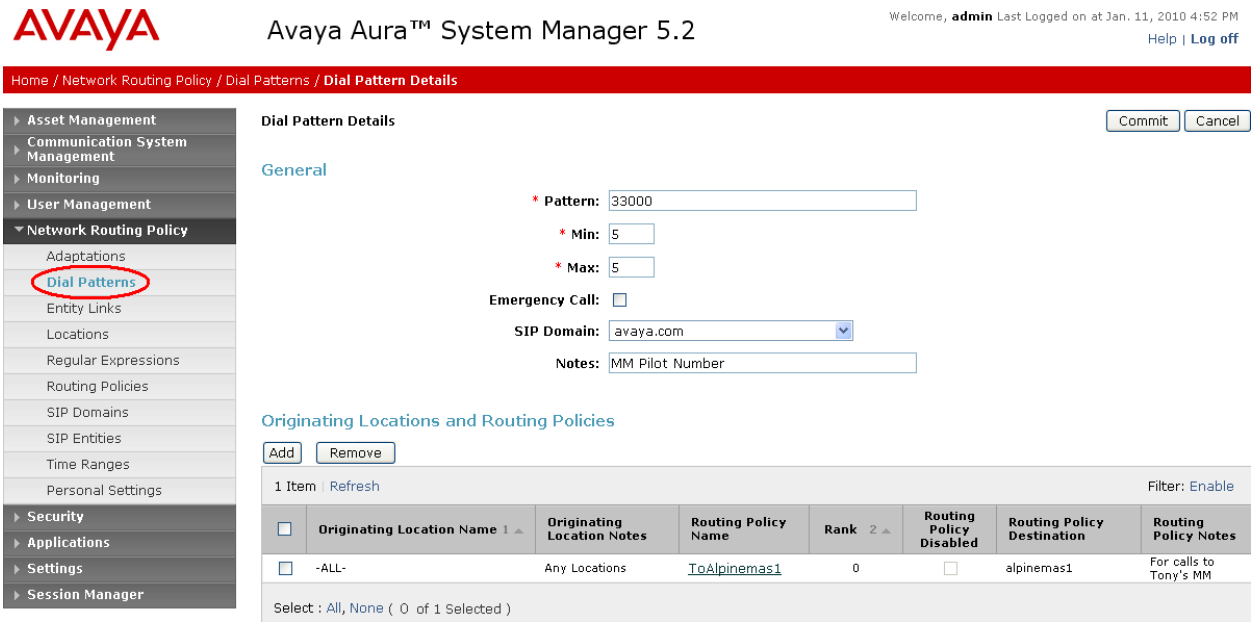
- **Pattern:** Dialed number or prefix.
- **Min:** Minimum length of dialed number.
- **Max:** Maximum length of dialed number.
- **SIP Domain:** SIP domain specified in **Section 4.1**
- **Notes:** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save each dial pattern. The following screens show the resulting three dial pattern definitions. Note that similar to Communication Manager, the dial pattern selected will correspond to the longest match of a **Pattern** with the dialed number.







4.8 Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *General*:

- **SIP Entity Name:** Select the SIP Entity added for Avaya Session Manager
- **Description**: Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager

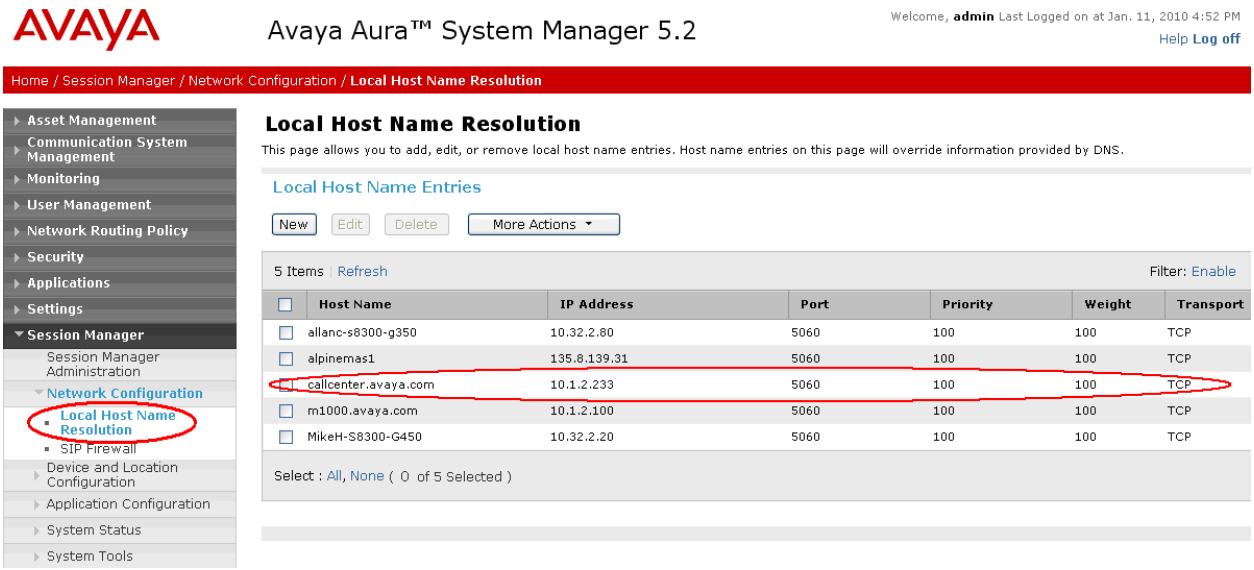
Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the resulting Session Manager definition.

4.9 Define Local Host Names

The host names (FQDN's) referenced in the definitions of the previous sections must be defined. To do so, Select **Session Manager -> Network Configuration -> Local Host Name Resolution** under the menu on the left. For each host name, click **New** and enter the following:

- **Host Name:** The FQDN used for the host
- **IP Address**: IP address of the host's network interface
- **Port:** Port number to which SIP requests are sent
- **Transport:** Transport to be used for SIP requests

Defaults can be used for the remaining fields. The **Priority** and **Weight** fields are used when multiple IP addresses are defined for the same host. The circled entry in the following screen shows the host name used in the sample configuration (see Entity Link configuration for Communication Manager in **Section 4.4**).



5 Configure Avaya Modular Messaging

In sample configuration, the Communication Manager and the CS1000 telephone systems were added as sites to an existing multi-site Modular Messaging system that was modified to support their subscribers and communication with Session Manager. The associated MAS server was named *alpinemas1*. As shown in the previous sections, Session Manager was configured to route incoming calls to *alpinemas1* (135.8.139.31*)*. *alpinemas1* was also configured to send message waiting notifications (SIP NOTIFY messages) to Session Manager. This section focuses on the following configuration steps:

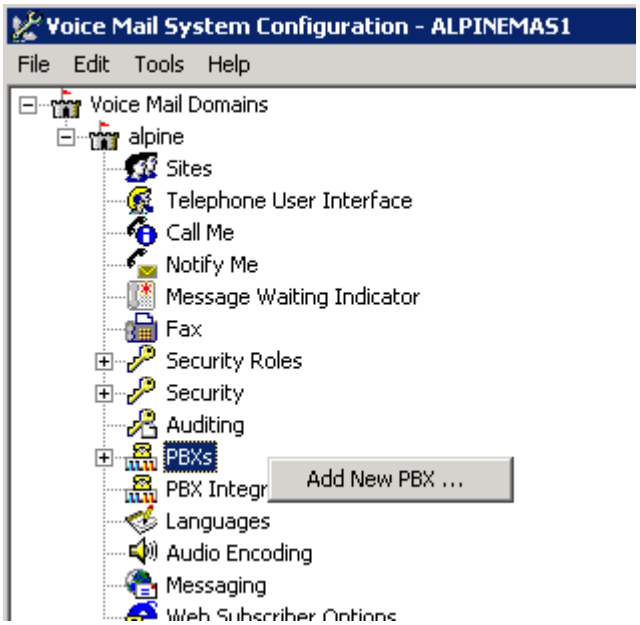
- Adding a PBX
- Configuring communication with Session Manager
- Defining dial plan translation rules

- Including the Communication Manager and CS1000 systems as sites
- Subscriber definition

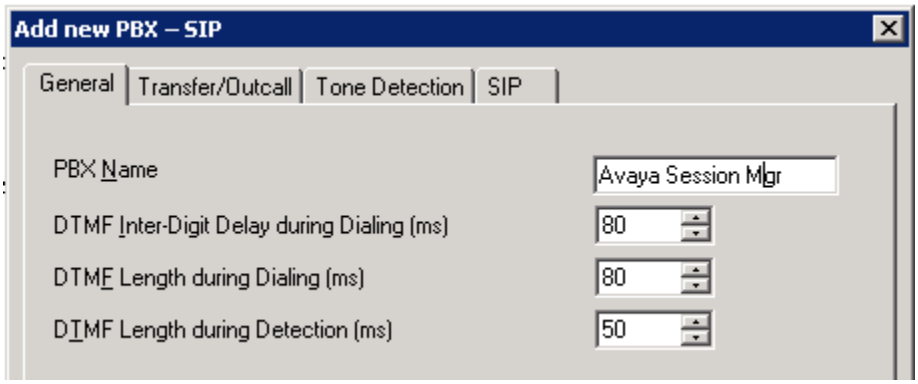
See references [7-8] in **Section 9** for standard installation and configuration information. General knowledge of the configuration tools and interfaces is assumed.

5.1 Add PBX

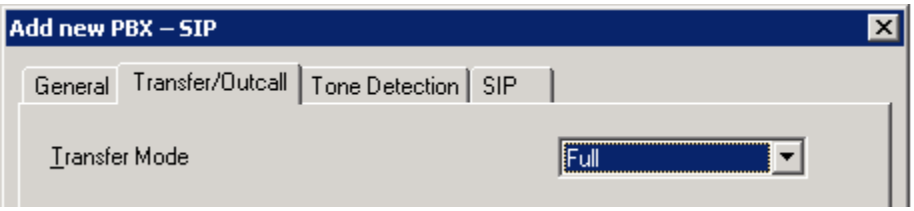
The aggregate Session Manager, Communication Manager, and CS1000 systems are defined to Modular Messaging as a "PBX." Bring up the *Voice Mail System Configuration* tool, select **PBXs**, and use button two on the mouse to select **Add New PBX**, as shown below.



On the *General* tab of the resulting displayed window, enter an appropriate **PBX Name**. Defaults can be used for the remaining fields.

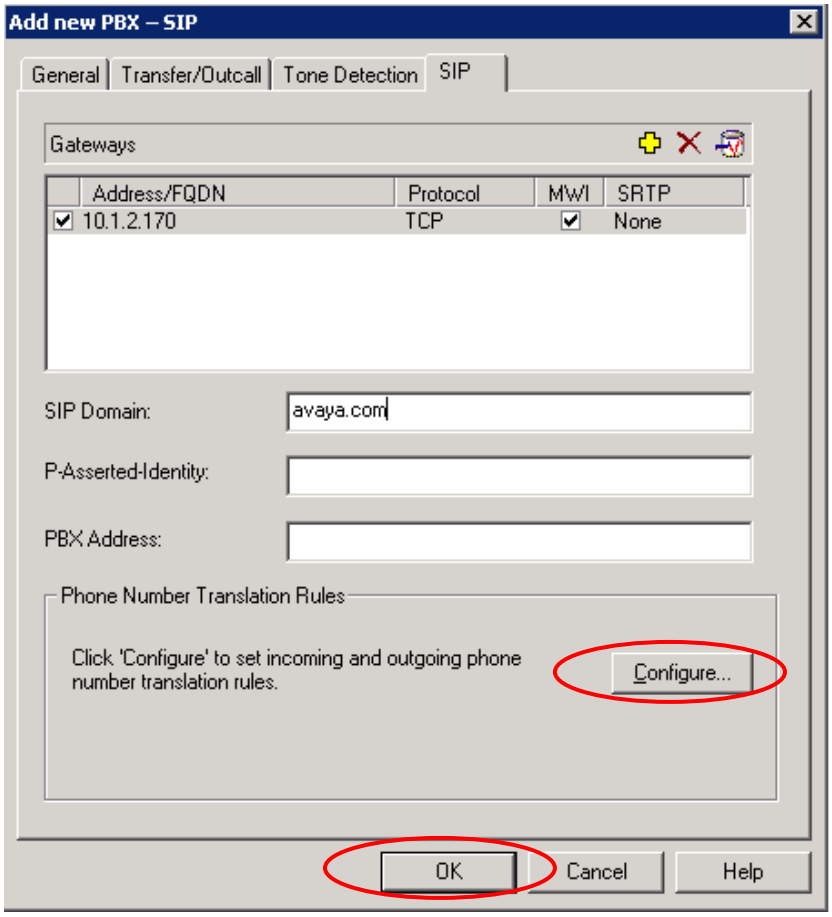


On the *Transfer/Outcall* tab, select "Full" for **Transfer Mode**.

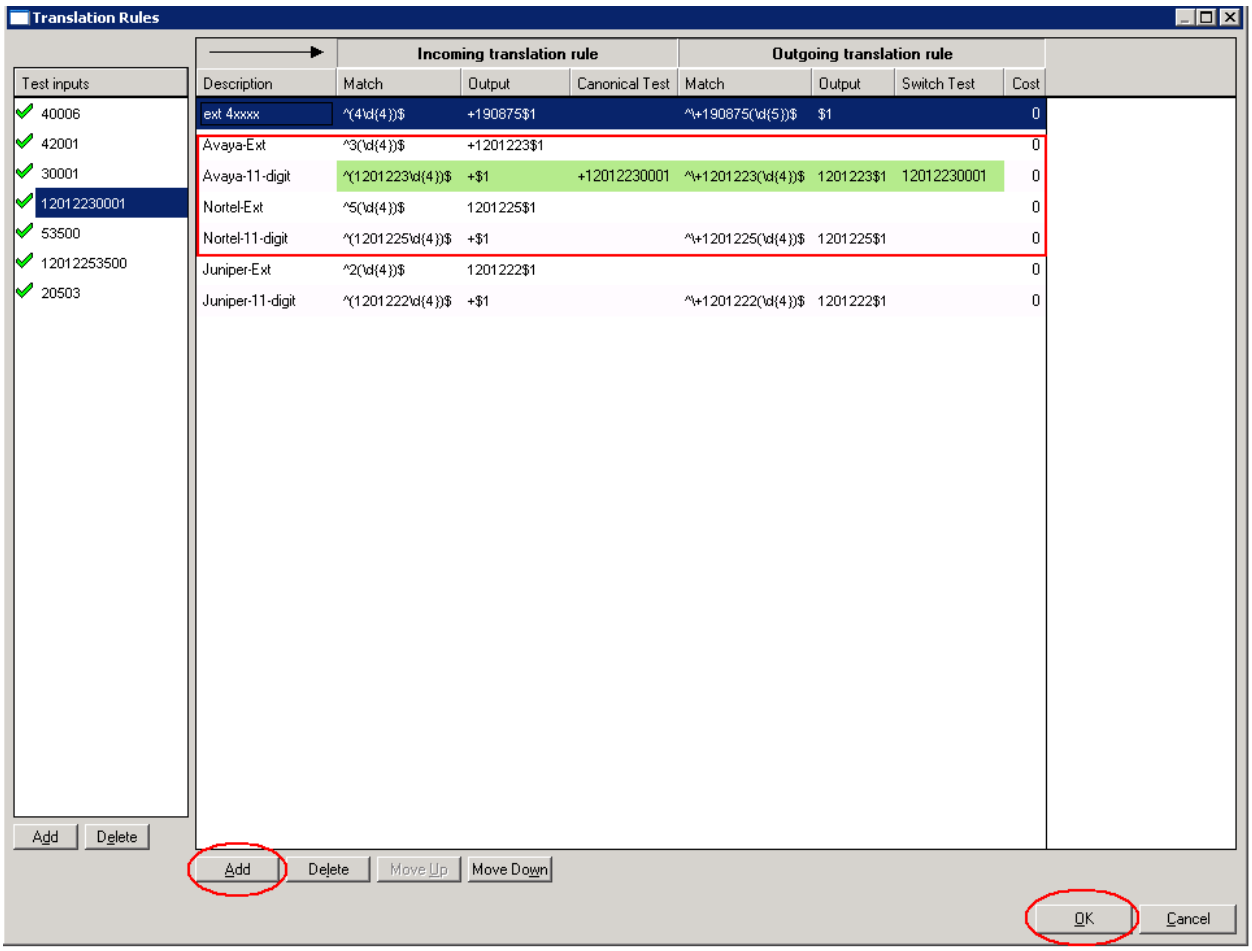


Default values can be used for the *Tone Detection* tab. On the *SIP* tab under the *Gateways* section, click on the "+" icon and add Session Manager's SM-100 IP address under **Address/FQDN**, "TCP" for **Protocol**, and check the **MWI** box so message waiting notifications will be sent. Fill in **SIP Domain** with the domain from **Section 4.1**.

Click on **Configure** to specify number translation rules for translating between the local dial plans of the Communication Manager and CS1000 telephone systems and the normalized 11 digit form used by Modular Messaging.

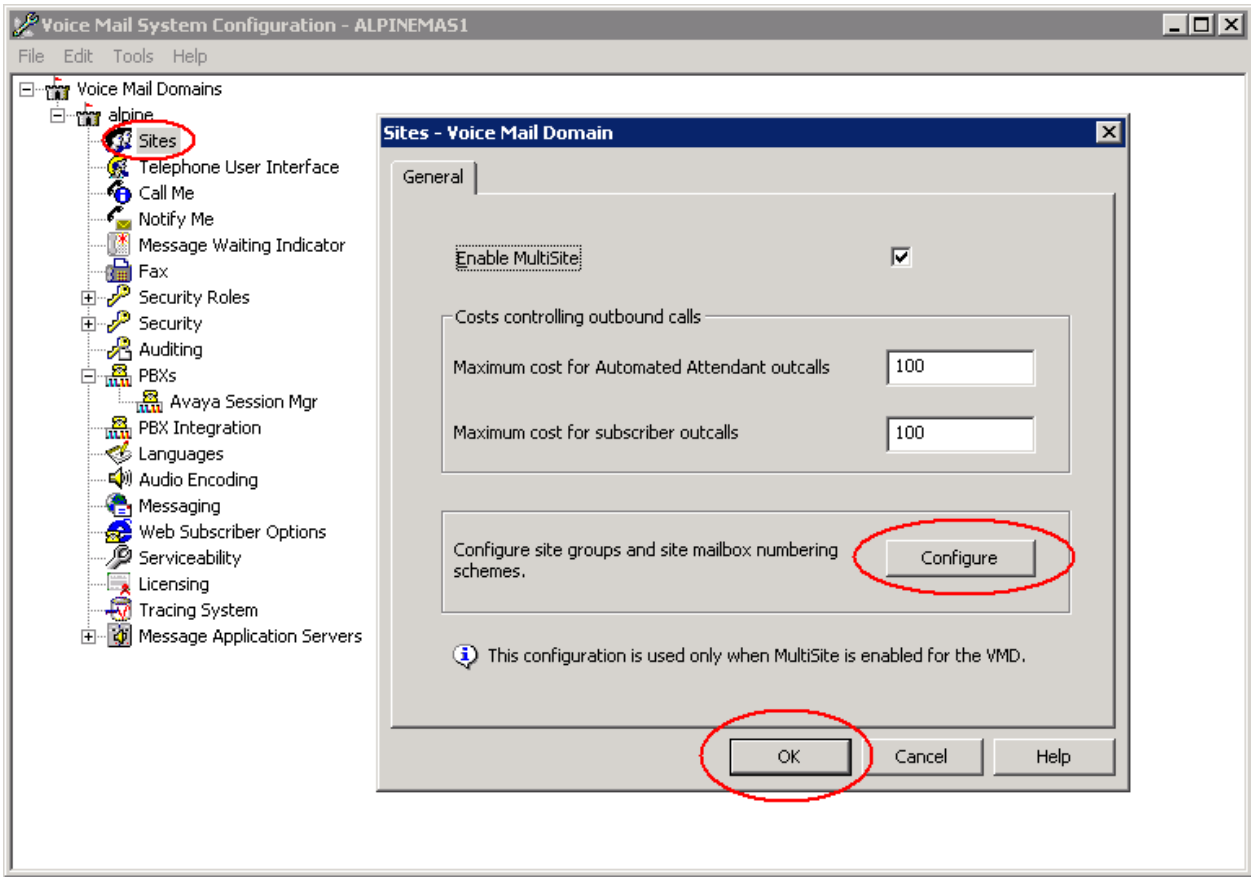


The following display appears. In the right pane, create the appropriate rules to translate between extension dialed and normalized 11 digit numbers. For the sample configuration, the last first rules were used, and were added by selecting **Add**. As described in **Section 4.3**, Session manager will translate between 11-digit numbers used by Modular Messaging and 5-digit numbering used by the telephone systems. The "Avaya-11-digit" and "Nortel-11-digit" **Incoming** and **Outgoing translation rule**s specify that Modular Messaging will not change the numbering from its 11-digit format. The "Avaya-Ext" and "Nortel-Ext" rules support features such as extension dialing by subscribers while accessing Modular Messaging, and translate the 5-digit extension format into normalized 11-digit format. In the screen below, "Avaya" corresponds to Communication Manager subscribers, and "Nortel" corresponds to CS1000 subscribers. Proper operation of the rules can be verified by adding *Test inputs* in the left pane and viewing the resulting output in the corresponding rule in the right pane. Click on **OK** when finished, then again on **OK** in the original *Add new PBX* window (see previous screen).



5.2 Add Sites

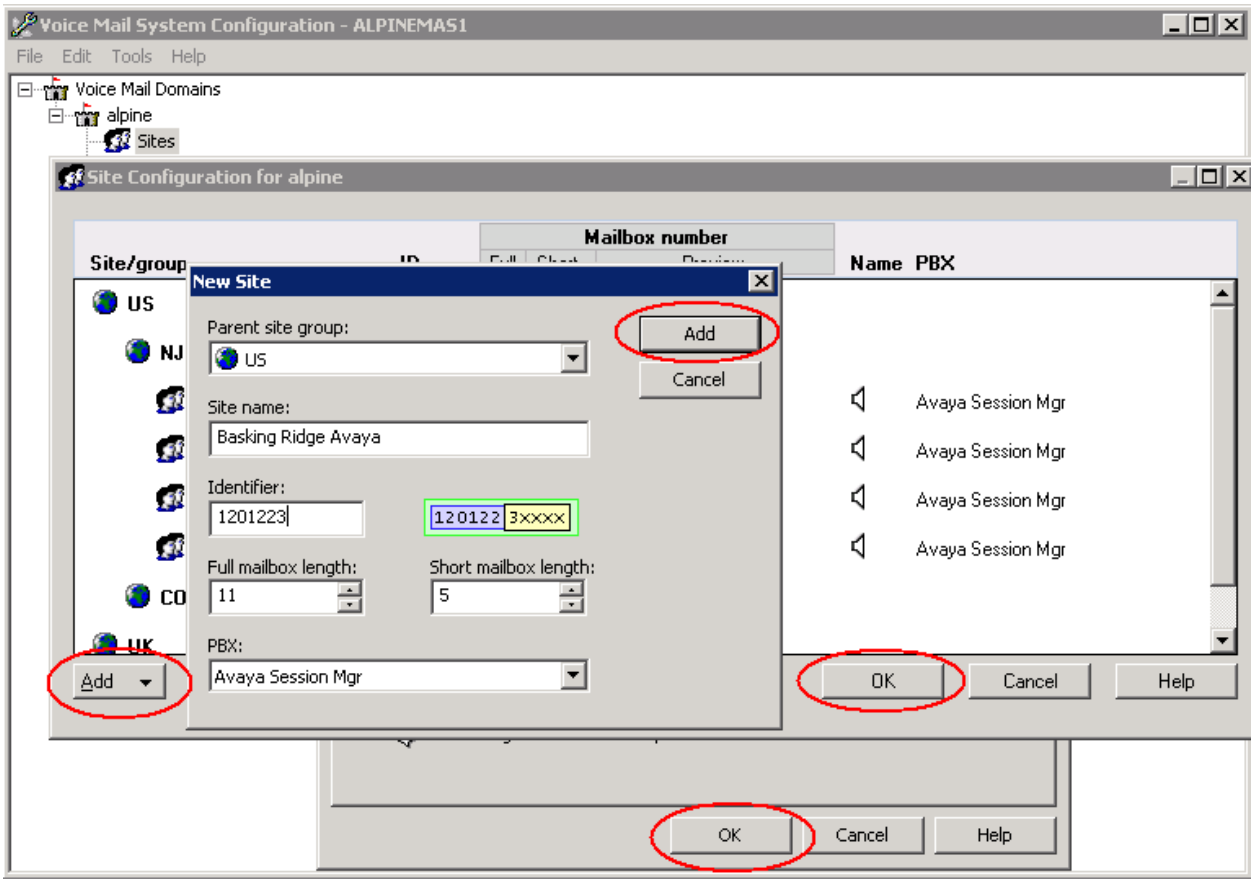
The telephone systems with different dial plans must be added as sites in Modular Messaging. This is done by double-clicking **Sites** in the *Voice Mail System Configuration* tool, as shown below. In the *Sites* window that is displayed, click on **Configure**.



The *Site Configuration* window is displayed. Click on **Add** to add the "Basking Ridge Avaya" site, and enter the following in the *New Site* window:

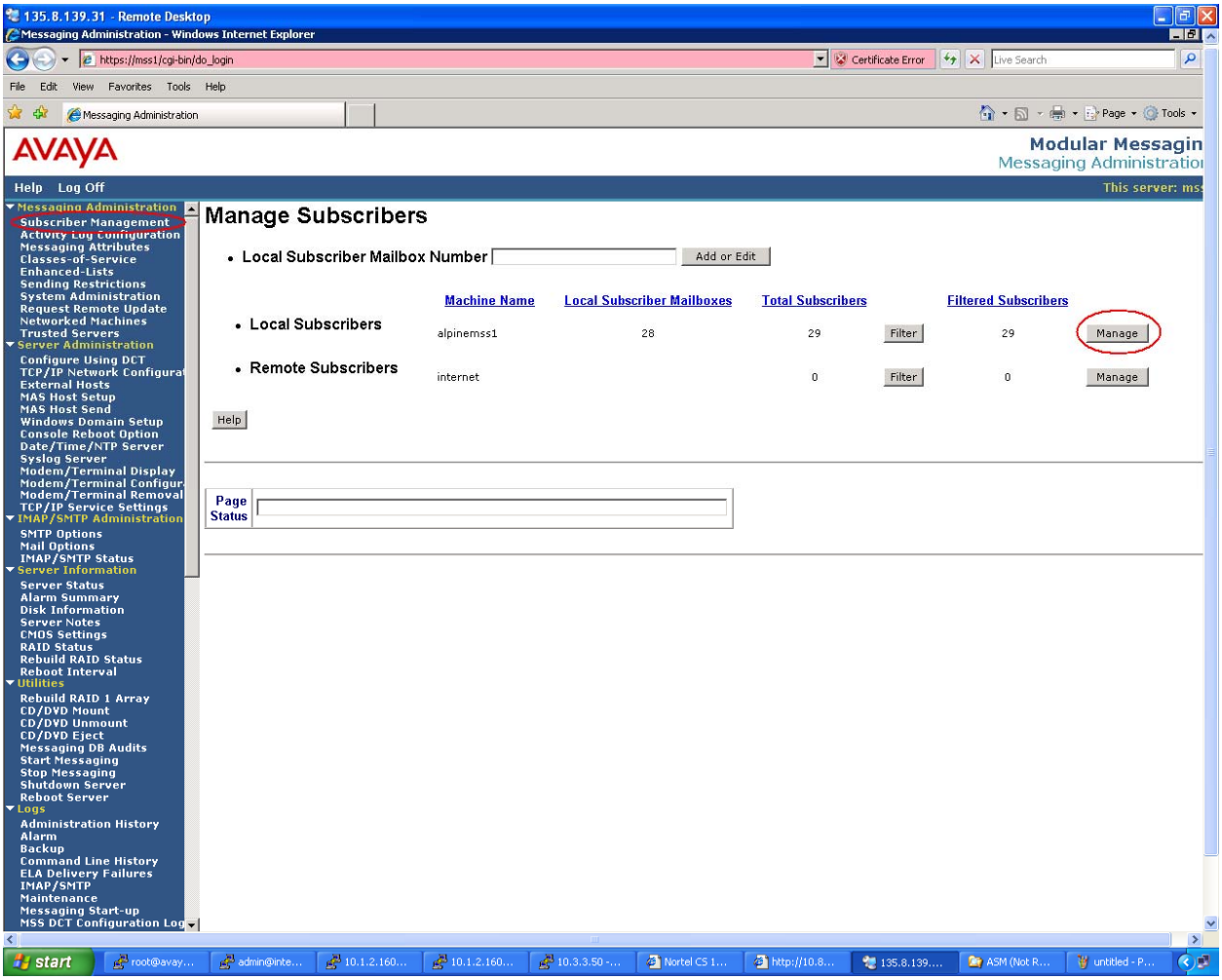
- **Parent site group:** Parent site name (e.g., "US")
- **Site name:** Site name
- **Identifier**: The unique initial n digits of the 11-digit mailbox number, identifying the site
- **Full mailbox length:** Enter "11" for the full mailbox number length
- **Short mailbox length:** Enter "5" for the extension length
- **PBX:** Enter name of the PBX added in the previous section

Click on **Add** when finished. Repeat for the site corresponding to the CS1000. The following screen shows the result of adding the Communication Manager at the Basking Ridge site. When all sites are added, click **OK** in the *Site Configuration* window, and then click on **OK** in the original *Sites* window (the first screen in this section).

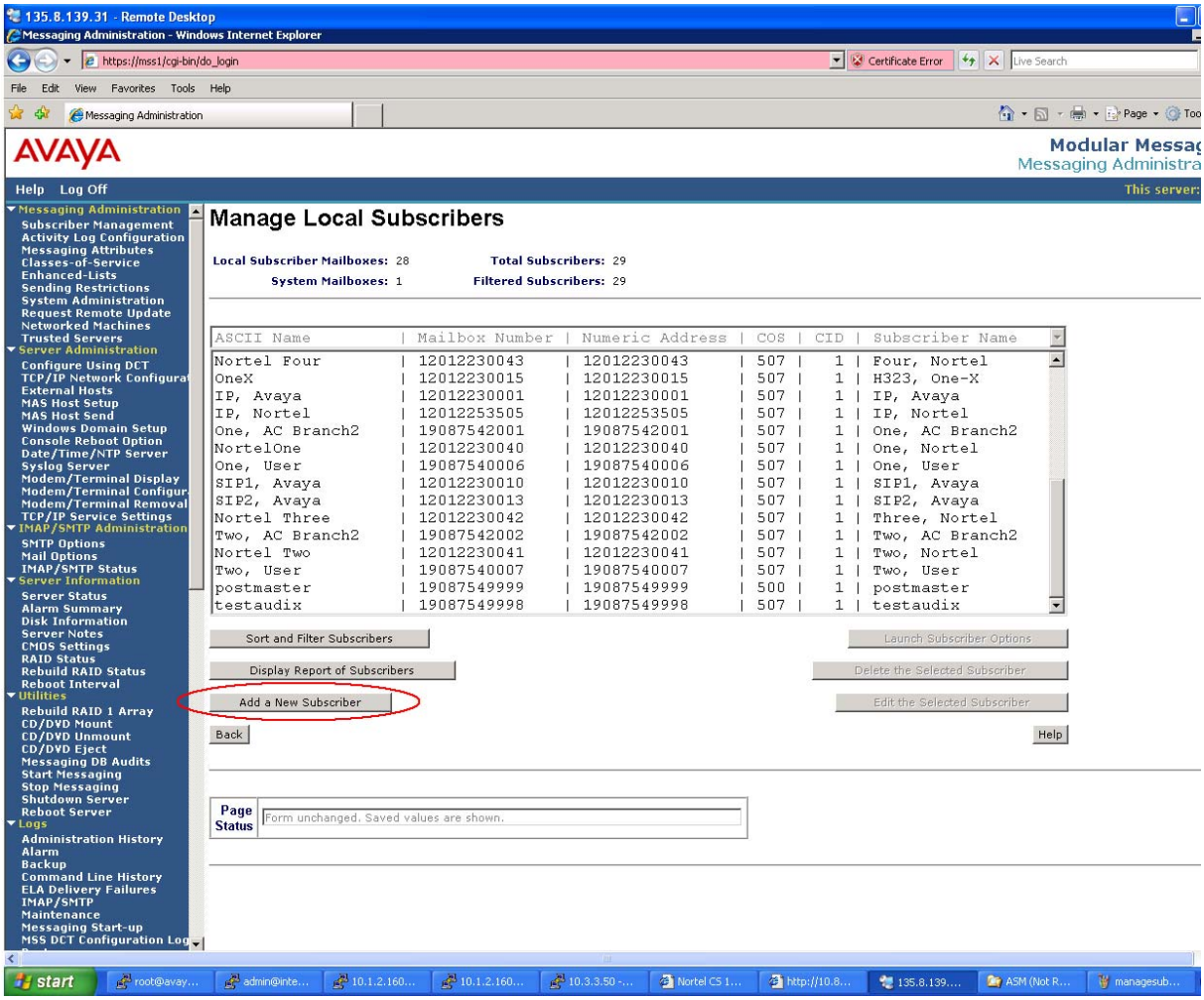


5.3 Add Subscribers

Log in to the web interface of the Modular Messaging MSS to add subscribers for each voice messaging user on the telephone systems. Select **Subscriber Management** on the left. Then select **Manage** on the right for *Local Subscribers*.



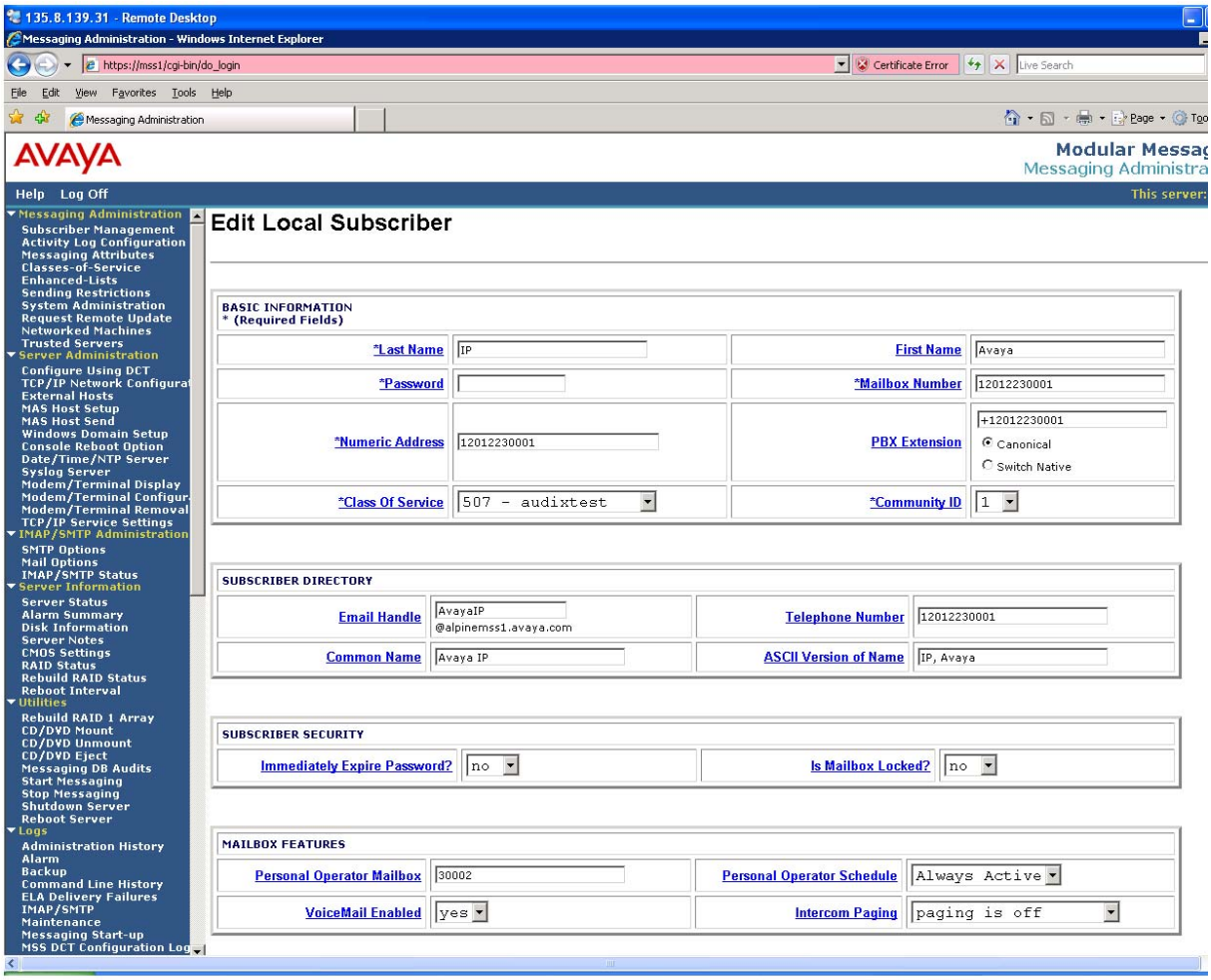
The *Manage Local Subscribers* screen is displayed. Click on **Add a New Subscriber**.

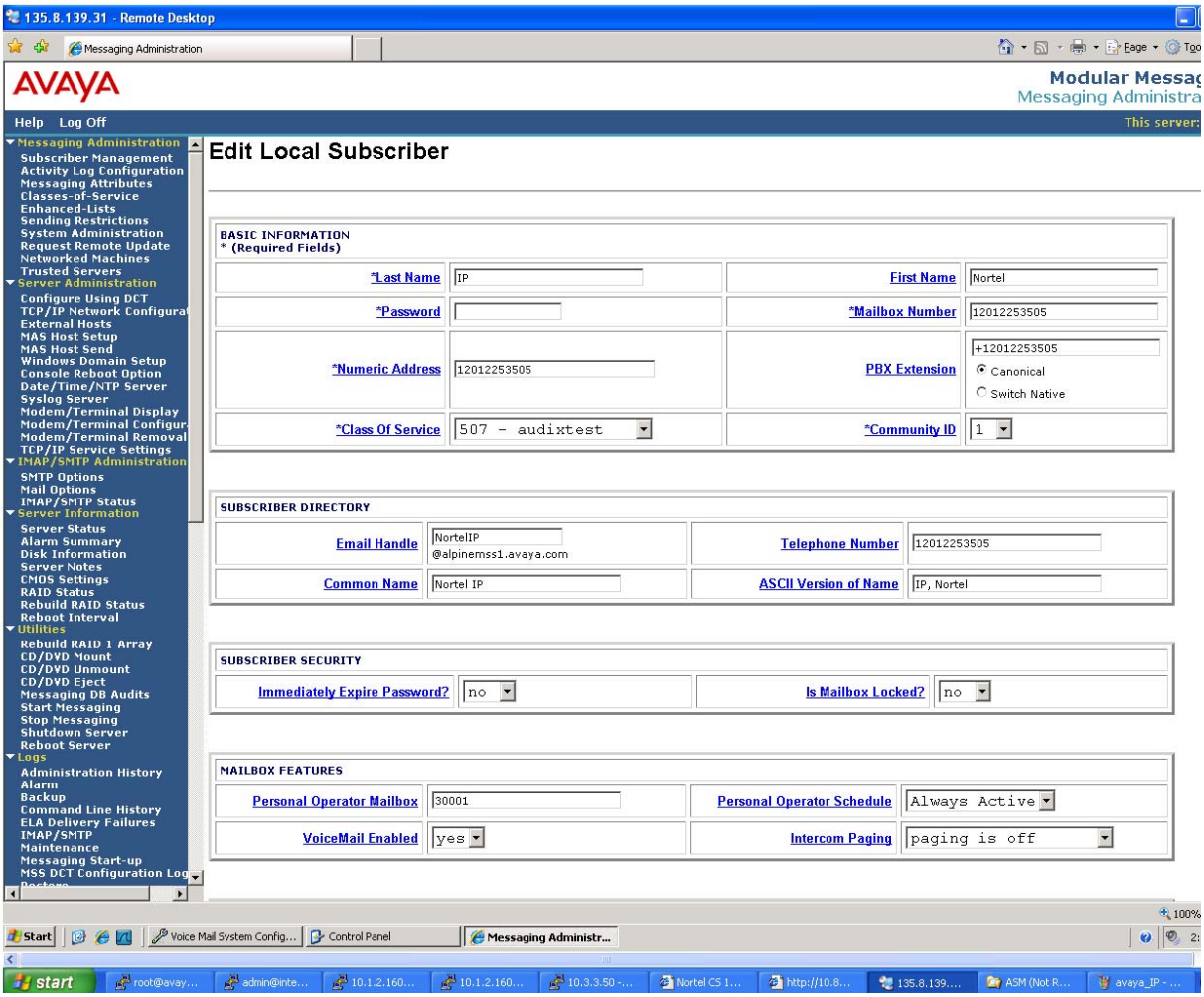


Enter the following for the new subscriber:

- **Last Name:** Subscriber last name
- **First Name**: Subscriber first name
- **Password:** Subscriber access password
- **Mailbox number:** Full 11-digit mailbox number
- **PBX Extension:** 11-digit mailbox preceded by "+" and select **Canonical**
- **Class of Service:** Select one from those configured (controls access to voice messaging features)
- **Community ID:** Select one from those configured

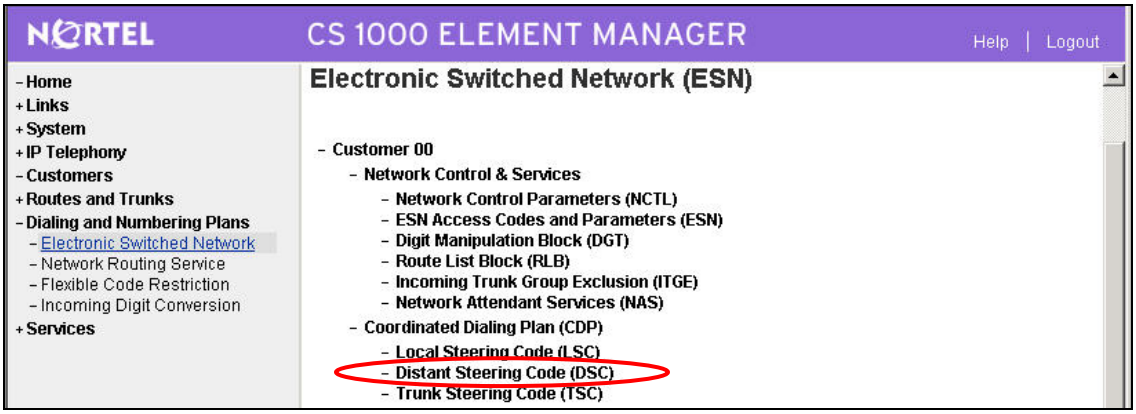
All other fields can retain default values. Select **Save** when done (not shown). The screens below show a sample subscriber definition for each system.





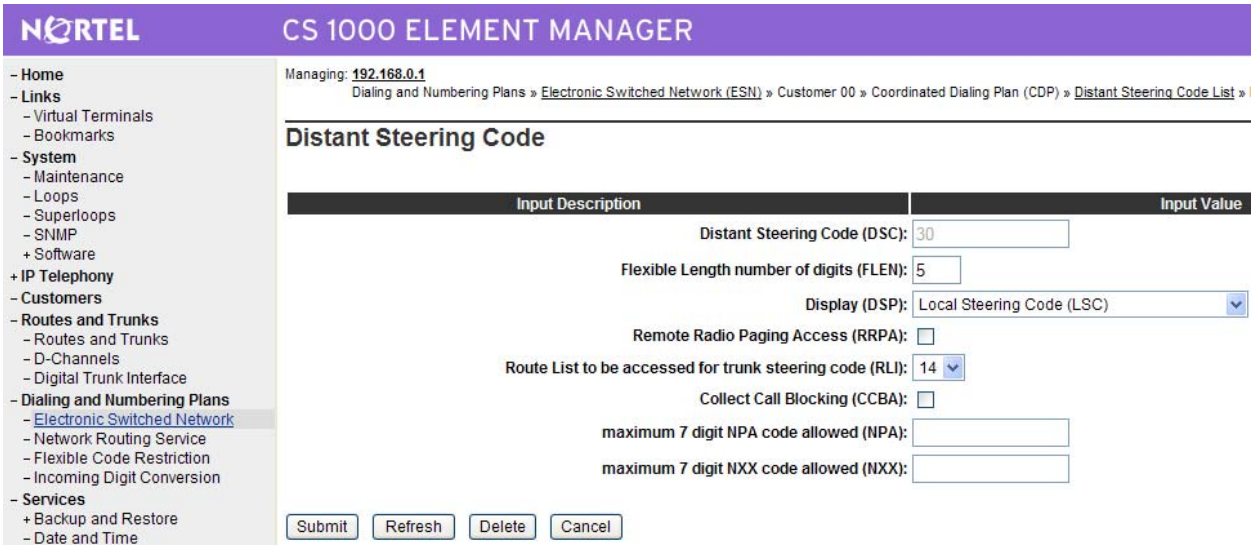
6 Configure Avaya Communication Server 1000

Configuration of the CS1000 for call routing and coverage to Modular Messaging (extension 33000) using a T1 PRI QSIG interface to the Avaya G450 Media Gateway is identical to that described in other Application Notes (see Reference [11]). In addition to those configuration steps, a route must be added for calls to Communication Manager, which has extension numbers of the form 30xxx. The CS1000 Element Manager is used to configure this routing. On the main web page, under *Dialing and Numbering Plans*, select **Electronic Switched Network**, and then click on **Distant Steering Code**.



The **Distant Steering Code List** screen is displayed. In the **Please enter a distant steering code** field, enter the dialed prefix digits to match on (in this case "30"). Click **to Add**.

The **Distant Steering Code** screen is displayed. For the **Route List to be accessed for trunk steering code (RLI)** field, select the route list index from the drop-down list that corresponds to that configured for the PRI QSIG trunk (as per Reference [11]). Retain the default values in all remaining fields, and scroll down to the bottom of the screen to click **Submit**. Repeat these two steps to add another Distant Steering Code for dialing the Modular Messaging pilot number (33000).



7 Verification Steps

7.1 Verify Avaya Aura™ Communication Manager

Verify the status of the ISDN trunk group to the CS1000 using the "status trunk" command. An example screen is shown below. Idle trunk members should show "in-service/idle".

```
status trunk 100                                                    Page   1

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0100/001 001V101  in-service/idle    no
0100/002 001V102  in-service/idle    no
0100/003 001V103  in-service/idle    no
0100/004 001V104  in-service/idle    no
0100/005 001V105  in-service/idle    no
0100/006 001V106  in-service/idle    no
0100/007 001V107  in-service/idle    no
0100/008 001V108  in-service/idle    no
0100/009 001V109  in-service/idle    no
0100/010 001V110  in-service/idle    no
0100/011 001V111  in-service/idle    no
0100/012 001V112  in-service/idle    no
0100/013 001V113  in-service/idle    no
0100/014 001V114  in-service/idle    no
```

If the trunk members are not in-service, check the signaling group status, as shown below, using the “status signaling-group” command. Verify the signaling group is “in-service” as indicated in the **Group State** and **Level 3 State** fields shown below.

```
status signaling-group 100
                          STATUS SIGNALING GROUP

        Group ID: 100                             Active NCA-TSC Count: 0
      Group Type: isdn-pri                         Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service


                            Primary D-Channel


            Port: 001V124        Level 3 State: in-service
```

If the signaling group **Level 3 State** is not in service, the health of the physical level can be checked by testing the DS1 board. Abridged output is shown below. While maintenance documentation is beyond the scope of these Application Notes, failure of the initial tests of the DS1 board likely indicate a problem with the physical layer connectivity to the CS1000 (e.g., improper cabling, framing, etc.). If test 144 fails, check that the G450 Media Gateway is deriving clock synchronization properly.

```
test board 1v1                                                          Page   1

                                  TEST RESULTS

Port        Mtce Name   Alt. Name        Test No.  Result           Error Code

001V1       MG-DS1                       138       PASS
001V1       MG-DS1                       139       PASS
001V1       MG-DS1                       140       PASS
001V1       MG-DS1                       141       PASS
001V1       MG-DS1                       142       PASS
001V1       MG-DS1                       143       PASS
001V1       MG-DS1                       144       PASS
001V1       MG-DS1                       145       PASS
001V1       MG-DS1                       146       PASS
```

Verify the status of the SIP trunk group by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 3.9.** Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 32

                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0032/001 T00226   in-service/idle    no
0032/002 T00227   in-service/idle    no
0032/003 T00228   in-service/idle    no
0032/004 T00229   in-service/idle    no
0032/005 T00230   in-service/idle    no
0032/006 T00231   in-service/idle    no
0032/007 T00232   in-service/idle    no
0032/008 T00233   in-service/idle    no
0032/009 T00234   in-service/idle    no
0032/010 T00235   in-service/idle    no
```

Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 3.9.** Verify the signaling group is "in-service" as indicated in the **Group State** field shown below.

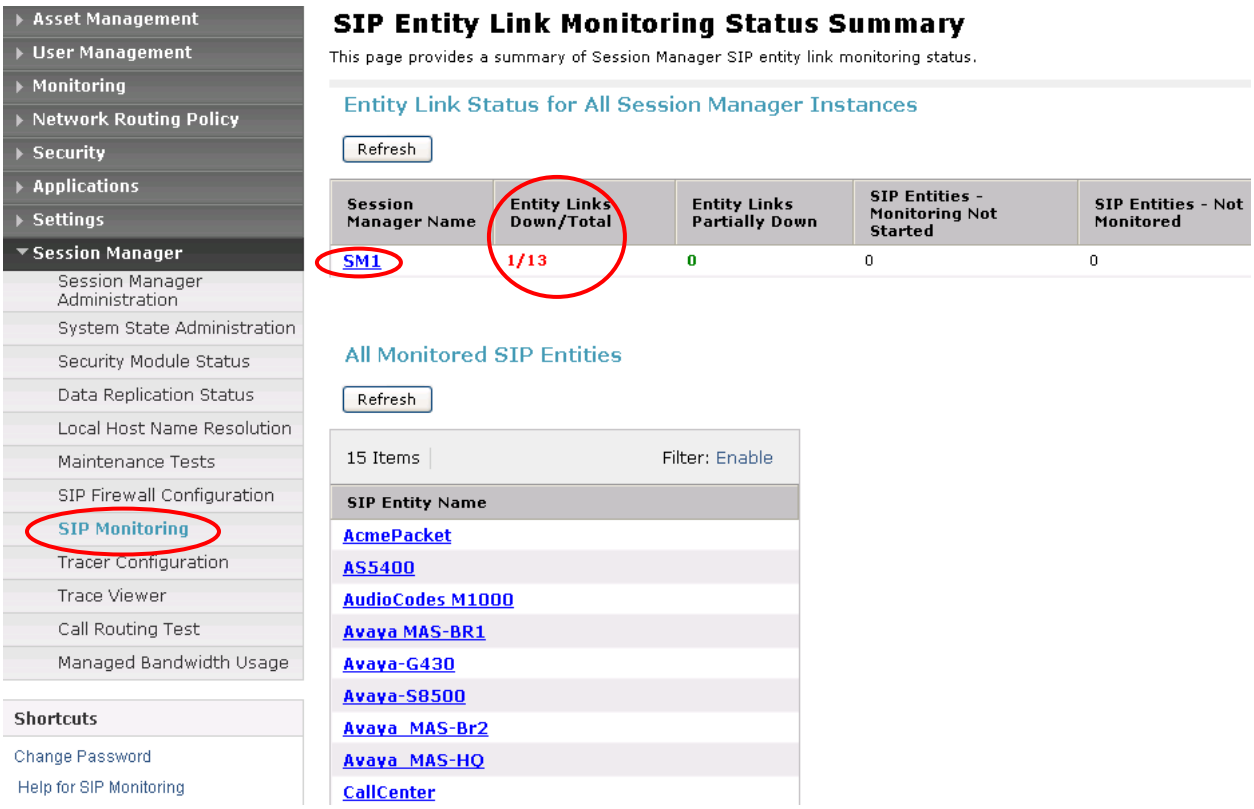
```
status signaling-group 32
                         STATUS SIGNALING GROUP

        Group ID: 32                              Active NCA-TSC Count: 0
      Group Type: sip                              Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

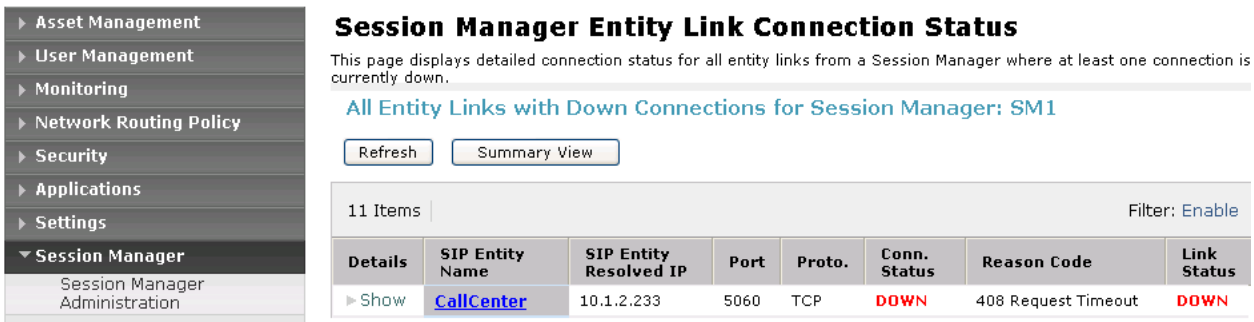
Finally, make a call between the Avaya 9600 Series IP Telephone and the Avaya i2004 IP Telephone and verify two-way audio. Verify the status of connected trunks by using the "status trunk" command for the PRI QSIG trunk group (100) to the CS1000. More information can be obtained by using "status trunk 100/x" where x is the trunk member for the in-service/active trunk member for the call.

7.2 Verify Avaya Aura™ Session Manager

Expand the **Session Manager** menu on the left and click **SIP Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing. In the sample screen below, the SIP trunk to SM1 has been busied out on Communication Manager, so one of the links is shown as down.

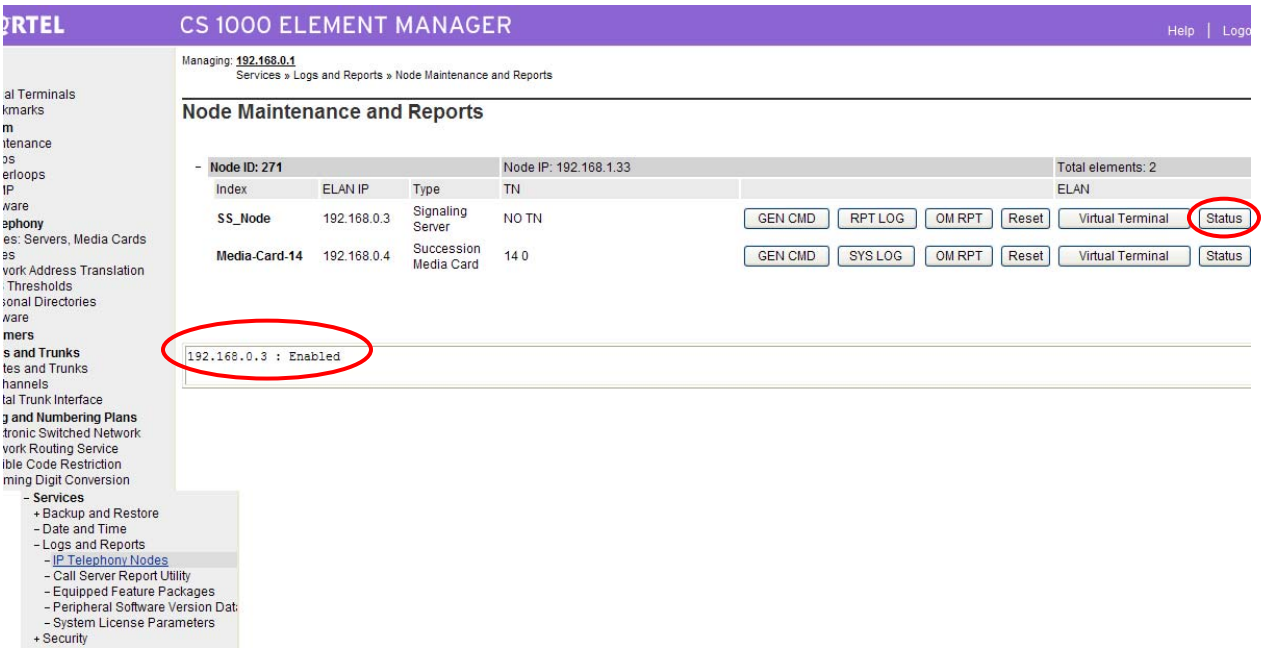


Select the corresponding Session Manager **(SM1** in this example) to view the Entity Link that is down and the Reason Code. The Reason Code reflects the result of Session Manager sending a SIP OPTIONS message to that SIP Entity.



7.3 Verify Nortel Communication Server 1000

Select **Services->Logs and Reports->IP Telephony Nodes** on the left. Click **Status** for the "SS_Node" to verify that the signaling server is enabled and operational.



See Reference [11] for verification of successful PRI QSIG trunk configuration.

7.4 Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Basic calls between various telephones on the Communication Manager 5.2 and Avaya Communication Server 1000 can be made in both directions using G.711MU. Proper display of the calling and called party name and number information was verified for all telephones with the basic call scenario.

- Supplementary calling features were verified. The feature scenarios involved additional endpoints on the respective systems, such as performing an unattended transfer to a local endpoint on the same system, and then repeating the scenario to transfer the call to a remote endpoint on the other system. The supplementary calling features verified are shown below. Note that calling/called party name and number display may not be consistent in certain scenarios for features shown in italics.

    - Unattended transfer
    - Attended transfer
    - *Hold/Unhold*
    - Consultation hold
    - Call forwarding
    - Conference

- Voice mail and voice mail calling features supported by Modular Messaging were verified, including message waiting indicator support for telephones on Communication Manager and Avaya Communication Serve 1000. Voice mail calling features included the following. Note that calling/called party name and number display may not be consistent in certain scenarios for features shown in italics.

    - Busy/no answer greetings
    - Message Waiting Indicator (MWI)
    - Send all calls
    - Coverage on call forward
    - *Personal operator*
    - *Auto-attendant*
    - *Find me*
    - *Call me*
    - *Call sender*
    - *Transfer*

8 Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 front-ended by an Avaya G450 Media Gateway can be integrated with Avaya SIP products, including Session Manager and Modular Messaging.

9 Additional References

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

Avaya Aura™ Session Manager:

[1] Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.

[2] Installing and Administering Avaya Aura™ Session Manager, Doc ID 03-603324, available at http://support.avaya.com.

[3] Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

Avaya Aura™ Communication Manager 5.2:

[4] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Doc ID 555-245-206, May, 2009, available at http://support.avaya.com.

[5] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009, available at http://support.avaya.com.

[6] *Upgrading, Migrating, and Converting Avaya Servers and Gateways, Release 5.0*, Doc ID 03-300412, January 2008, available at http://support.avaya.com.

Avaya Modular Messaging:

[7] *Release 5.2 with Avaya MSS – Messaging Application Server (MAS) Administration Guide*, November, 2009, available at http://support.avaya.com.

[8] *Avaya Modular Messaging for the Avaya Message Storage Server (MSS) Configuration – Release 5.2 Installation and Upgrades*, November, 2009, available at http://support.avaya.com.

Avaya Application Notes:

[9] *Configuring SIP Trunks among Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager 5.2, and Nortel Communication Server 1000 – Issue 1.1*, available at http://www.avaya.com.

[10] *Front-Ending Nortel Communication Server 1000 with an AudioCodes Mediant 1000 Modular Media Gateway to Support SIP Trunks to Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager 5.2 as an Access Element – Issue 1.1*, available at http://www.avaya.com.

[11] *Configure an Avaya Centralized Messaging Solution with Avaya Communication Manager and Nortel Communication Server 1000 – Issue 1.0*, available at http://www.avaya.com.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com