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 using a PRI NI-1 trunk to Support SIP Trunks to Avaya Aura™ Session Manager 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration that uses an Avaya G450 Media Gateway as a PRI NI-1/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 can provide call routing support to other Avaya SIP products.

For the sample configuration, Session Manager runs on an Avaya S8510 Server, Communication Manager runs on Avaya S8720 servers, and 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 [8] 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. Since there are installations of the CS1000 which are not SIP or IP capable, an effective solution is to front-end the CS1000 with a PRI-QSIG gateway, which then signals on SIP trunks to Session Manager. Application notes are also available documenting the configuration steps for this arrangement [9, 10]. However, there are also many installations of the CS1000 which are not PRI-QSIG capable. In these cases, front-ending using PRI-NI-1 instead of QSIG can be employed. This configuration supports basic and supplementary call features. These Application Notes document the configuration steps and parameter settings required to support front-ending the CS1000 with an Avaya G450 Media Gateway controlled by Avaya Aura™ Communication Manager using PRI NI-1, such that the CS1000 can be integrated with Avaya Aura™ Session Manager via SIP trunks.

The sample configuration is shown in **Figure 1**. The G450 Media Gateway is controlled by Communication Manager, which supports SIP trunks to the SM-100 (Security Module) network interface of Session Manager. 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.

For the sample configuration, Session Manager and System Manager run on Avaya S8510 Servers, 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.

These Application Notes will focus on configuration of the PRI NI-1 trunk, SIP trunk, dial plan support, and call routing. Detailed administration of the endpoint telephones will not be described.

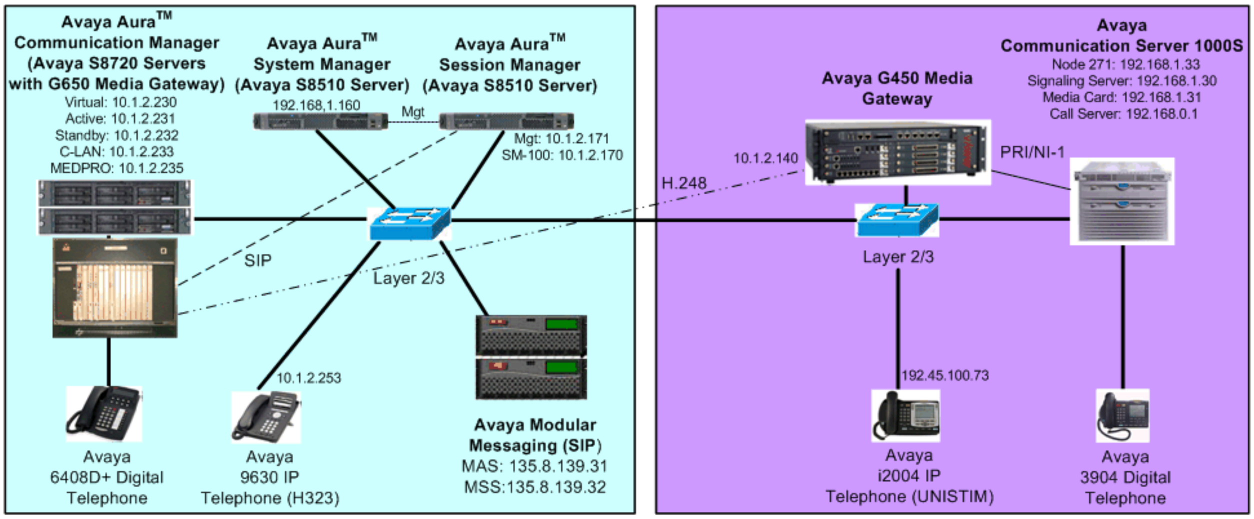


Figure 1 – Sample Configuration

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, Service Pack 2 (18111)
Avaya S8510 Server	Avaya Aura™ Session Manager 5.2 SP 2 (522007)
	Avaya Aura™ System Manager 5.2 SP 2 (522007)
Avaya 9630 IP Telephone (H.323)	3.1
Avaya 6408D+ Digital Telephone	-
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

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 should 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

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers and 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 8** 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
```

Use the "change system-parameters special-applications" command to enable Special Application **SA7161**, which enables compatible signaling on the PRI NI-1 interface to the CS1000.

```
change system-parameters special-applications                   Page   1 of   9
                             SPECIAL APPLICATIONS

WARNING: Special App features are intended to serve specific needs and are not
         recommended for general use. Activating one or more of these features
         may result in unpredictable system behavior. Please review information
         at http://support.avaya.com before feature activation.

Number of Features Activated: 3      Number of Restricted Features Activated: 0


           (SA7161) - NORTEL SL1 PRI and DMS Names Display? y
                         (SA7291) - TAAS Pickup During Day? n
```

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 other SIP products 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. In the sample configuration, the basic G.711 mu-law codec is used for the Avaya 9600 series IP Telephones.

```
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 NI-1 trunk connection to the CS1000. Install and configure the G450 Media Gateway as described in [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:** "network"
- **Peer Protocol:** "sl1"

The **Interface** field must be complementary on both switches. For the sample configuration, Communication Manager must be administered as the *network*, and the CS1000 must be administered as the *user*. Note that the CS1000 can only be administered as *user* for an NI-1 interface.

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

            Location: 001V1                           Name: Avaya CS1K
            Bit Rate: 1.544                    Line Coding: b8zs
   Line Compensation: 1                       Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                        Interface: network
   TN-C7 Long Timers? n                   Country Protocol: sl1
Interworking Message: PROGress
Interface Companding: mulaw                           CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                             T303 Timer(sec): 4


      Slip Detection? n                 Near-end CSU Type: other
```

3.8 Add PRI NI-1 Signaling Group and Trunk Group

3.8.1 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 CS1K                 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:
TestCall BCC: 4
```

Navigate to **Page 2**. For the **Supplementary Service Protocol** field, enter "a", and for **Codeset to Send Display**, enter "0". Retain default values for the remaining fields.

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

TRUNK PARAMETERS
         Codeset to Send Display: 0     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a     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 calls 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                            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
Network Call Redirection: none                    Hold/Unhold Notifications? n
             Send UUI IE? y                     Modify Tandem Calling Number? n
               Send UCID? y
 Send Codeset 6/7 LAI IE? y                           Ds1 Echo Cancellation? n

    Apply Local Ringback? n             US NI Delayed Calling Name Update? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

3.8.2 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 **Group Type**, enter "isdn-pri". For the **Trunk Group for NCA TSC** and **Trunk Group for Channel Selection** fields, enter the ISDN trunk group number from **Section 3.8.1**. For the **TSC Supplementary Service Protocol** field, enter "a". Maintain the default values for the remaining fields, and submit these changes.

```
add signaling-group 100                                         Page   1 of   5
                                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: a         Network Call Transfer? n
```

3.8.3 Trunk Group Members

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 3.8.2** 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 NI-1 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 3.8.2**.
- **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 NI-1)
- **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 SM1
                             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 time zone 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
```

Since Communication Manager is configured as an "Access Element" with respect to Session Manager, the SIP trunk is not IMS-enabled (see **Section 3.9.1**). Outbound calls from Communication Manager may go to a public gateway. Therefore, calling parties for calls that use this trunk should have public numbering treatment. 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
```

Since the PRI trunk to the CS1000 is used for intra-enterprise calls, the calling parties should have private numbering treatment. 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 3.8.2**. 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. 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. This allows callers to use extension dialing without being required to dial an AAR access code.

- **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”

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
```

Use the “change aar analysis 0” command, and add corresponding entries to specify use of the PRI NI-1 trunk for 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”.
- **Total Min:** Minimum number of digits.
- **Total Max:** Maximum number of digits.
- **Route Pattern:** The route pattern number from **Section 3.10**.
- **Call Type:** “lev0” for private numbering (PRI NI-1)

```
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
    53                    5    5     100       lev0          n
```

Use the “change dialplan analysis” command to define the 3xxxx and 5xxxx extension ranges.

```
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
```

3.13 Save Translations

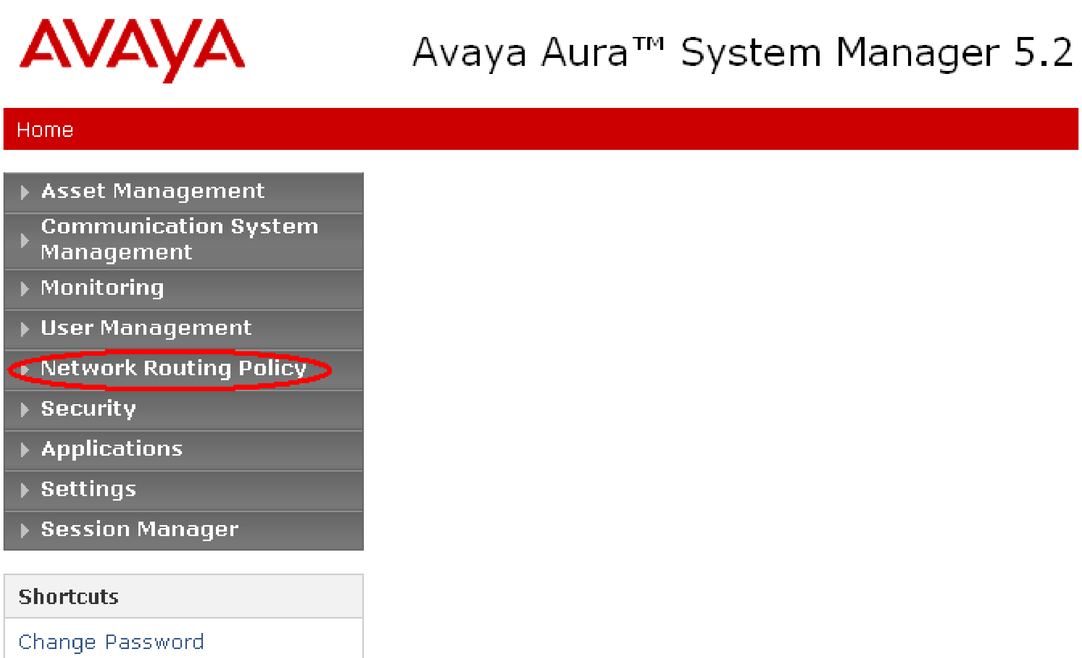
Configuration of Communication Manager is complete. Use the “save translations” command to save these changes.

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
- SIP Entities corresponding to Communication Manager, CS 1000, 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**).

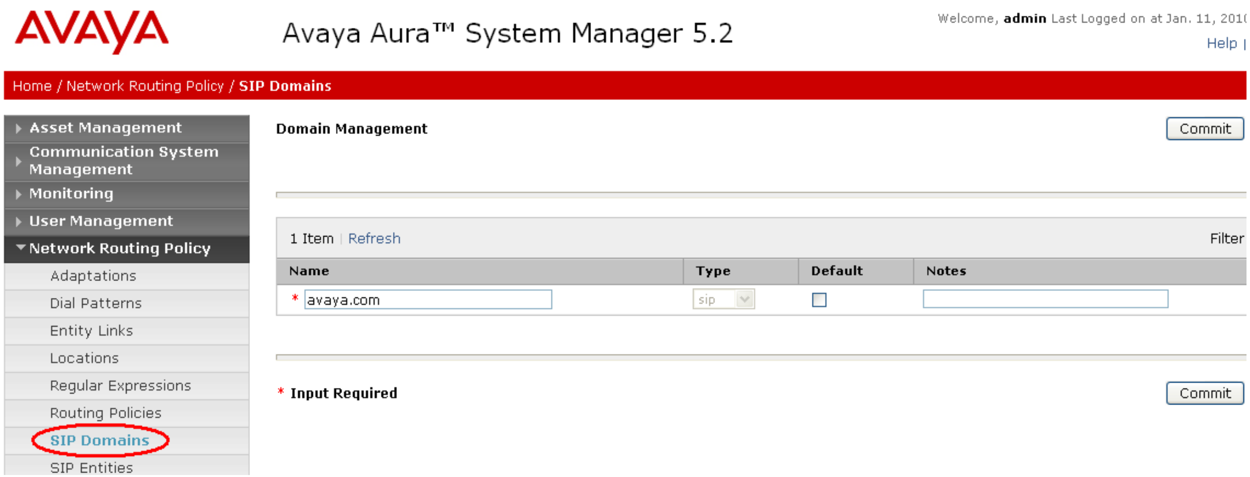


4.1 Specify SIP Domain

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

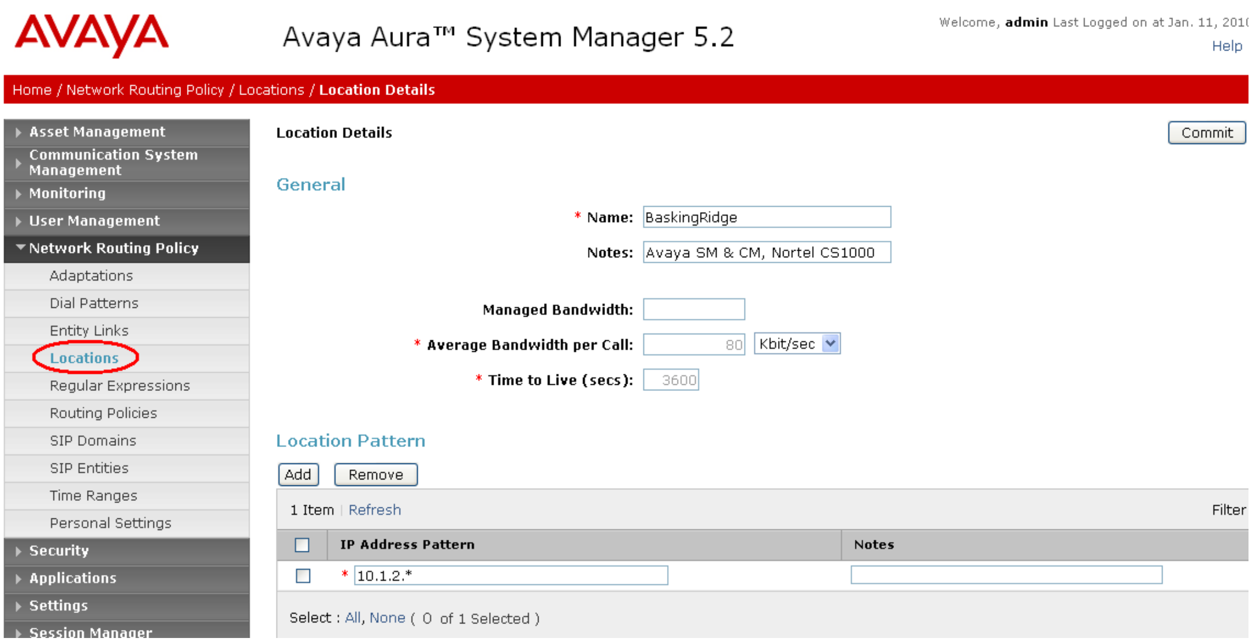
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. Under **General**, enter:

- **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 the CS1000[1] in the 10.1.2.0/24 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.

[1] Note that even though the CS1000 is in a different subnet than 10.1.2.0/24, since the only access to it is via the PRI NI-1 interface on the Communication Manager controlled G450 Media Gateway, from the perspective of Session Manager, routing to the CS1000 if via Communication Manager, which resides in the 10.1.2.0/24 subnet.

4.3 Add SIP Entities

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

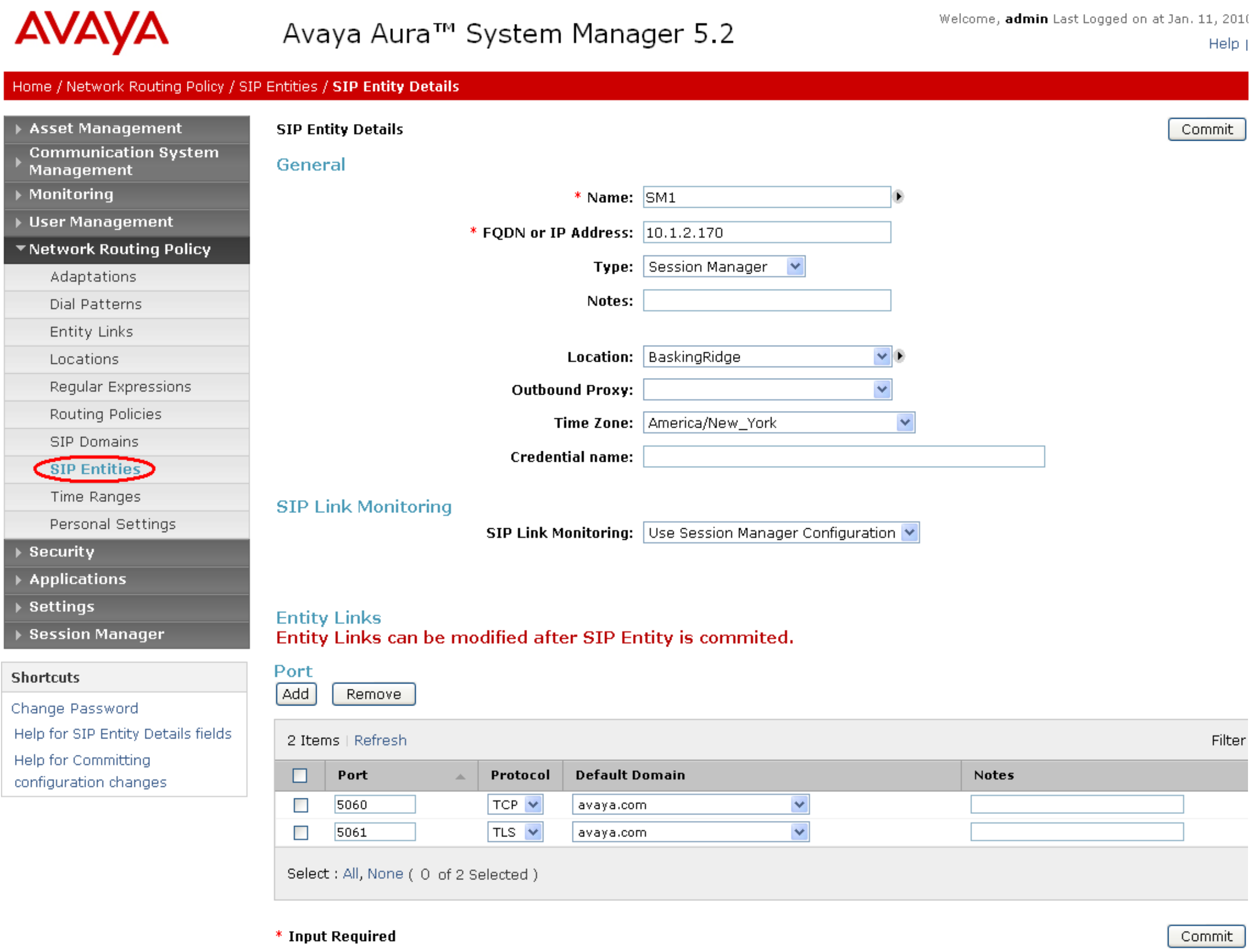
- **Name:** A descriptive name.
- **FQDN or IP Address:** FQDN or IP address of the Session Manager or the signaling interface on the telephony system.
- **Type:** "Session Manager" for Session Manager or "CM" for Communication Manager.
- **Location:** Select one of the locations defined previously.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

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

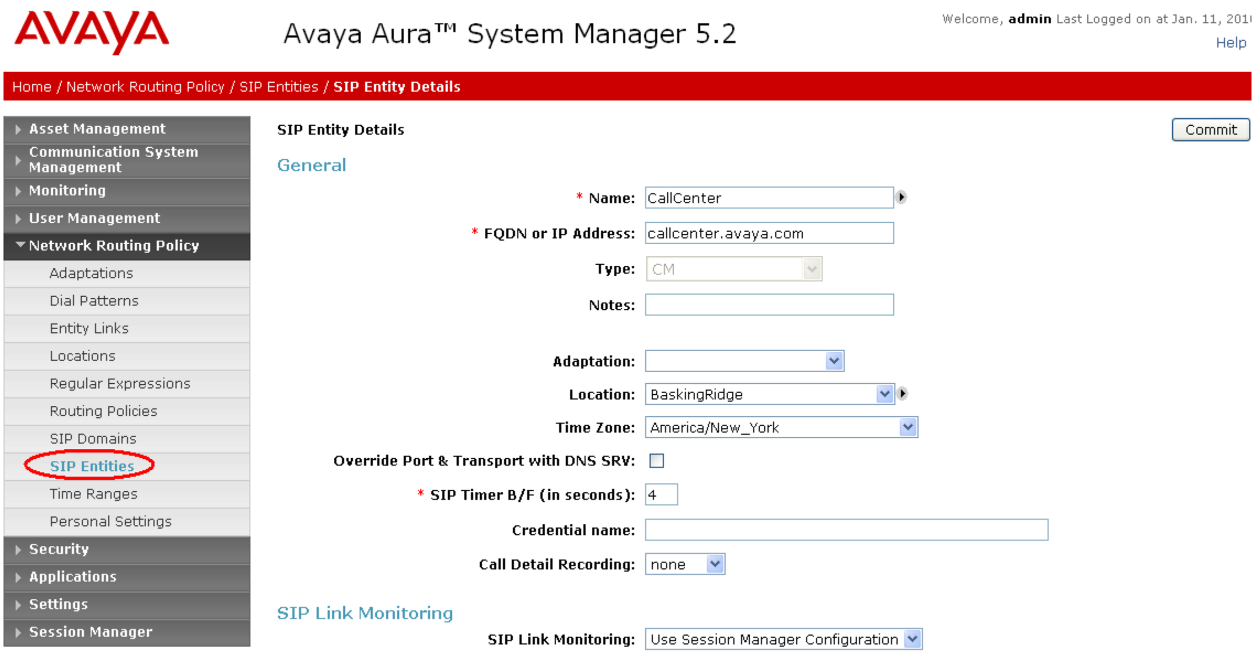
- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise (e.g., "avaya.com").

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 shown. The TLS port 5061 is used for communication with other Session Managers and Communication Manager. TCP port 5060 is used for communicating with other SIP entities not addressed in this sample configuration.



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 4.8)** will cause Session Manager to load balance call traffic among those addresses.

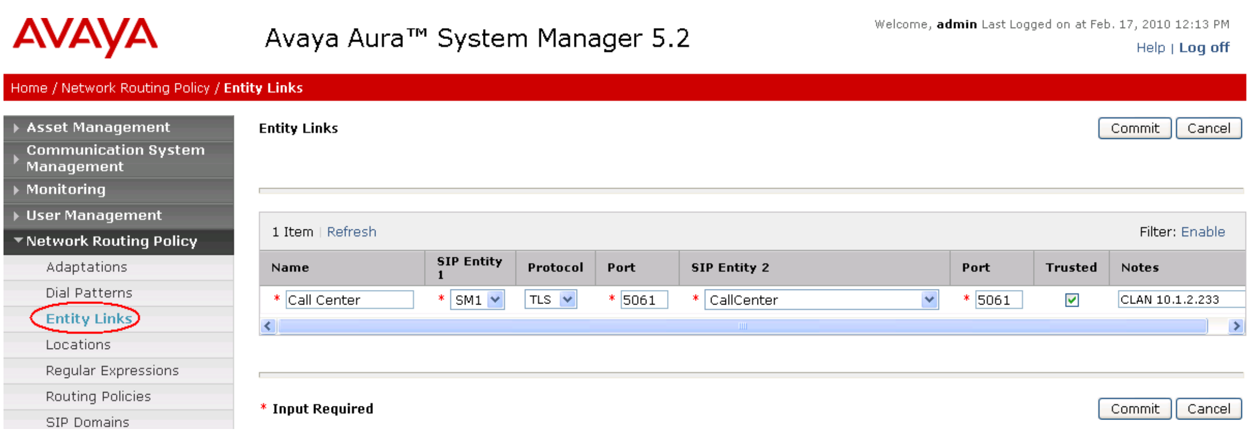


4.4 Add Entity Links

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 SIP Entity dedfined in **Section 4.3**..
- **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 4.3** will be denied.*

Click **Commit** to save the Entity Link definition. The following screen illustrates adding the Entity Link for Communication Manager.



4.5 Add Routing Policies

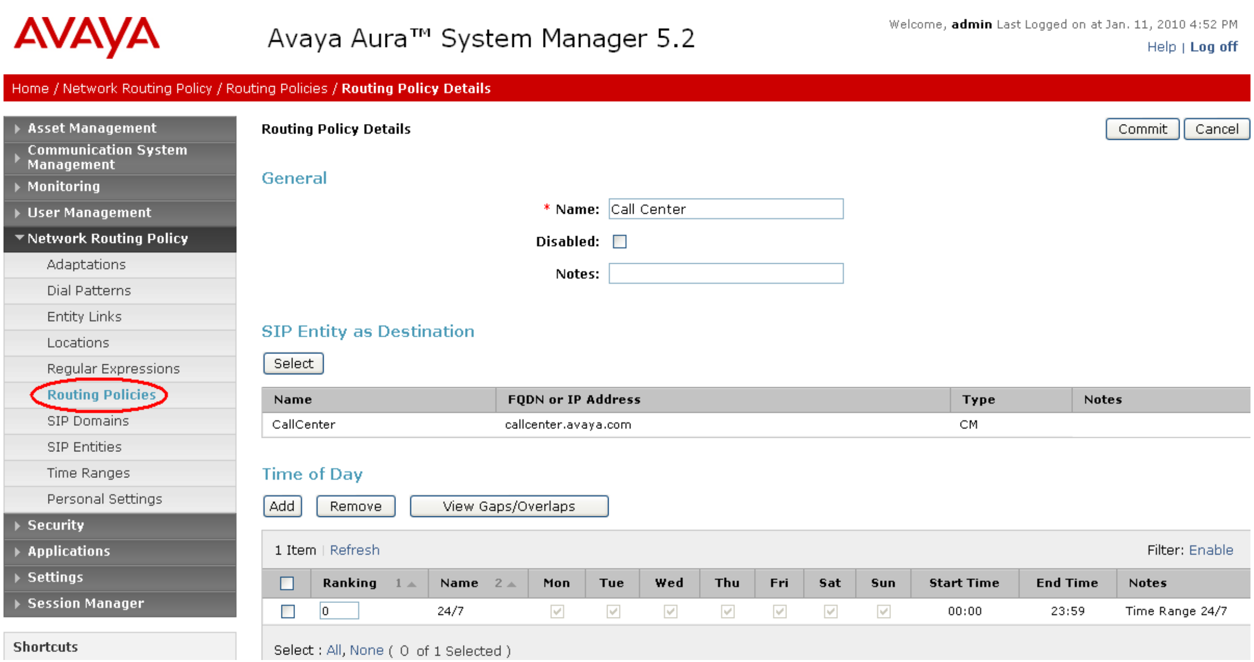
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 4.3**. A routing policy must be added for Communication Manager. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. 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 screen shows the Routing Policy for users on Communication Manager and the CS1000.



4.6 Add Dial Patterns

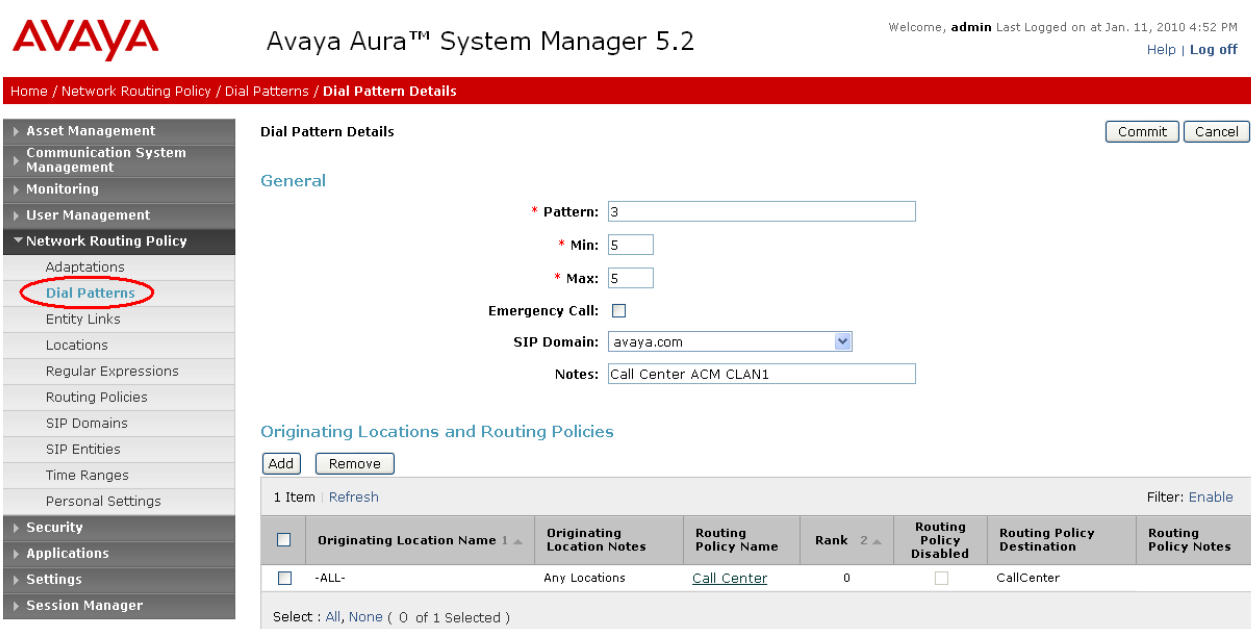
Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 5-digit extensions beginning with "3" (Communication Manager) or "53" (CS1000) should be routed to Communication Manager. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

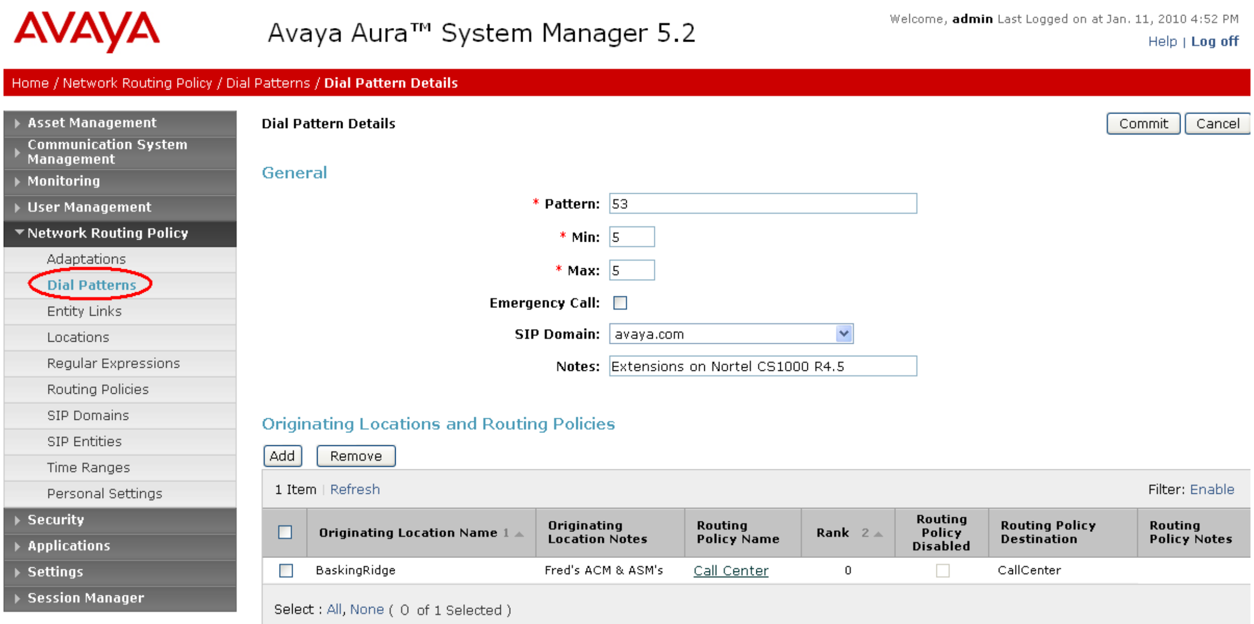
Under **General**:

- **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 two 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.




AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Jan. 11, 2010 4:52 PM
Help | Log off
Home / Network Routing Policy / Dial Patterns / Dial Pattern Details
Asset Management
Communication System Management
Monitoring
User Management
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
Settings
Session Manager
Dial Pattern Details
Commit
Cancel
General
* Pattern: 53
* Min: 5
* Max: 5
Emergency Call:
SIP Domain: avaya.com
Notes: Extensions on Nortel CS1000 R4.5
Originating Locations and Routing Policies
Add
Remove
1 Item | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
BaskingRidge
Fred's ACM & ASM's
Call Center
0
CallCenter
Select : All, None (0 of 1 Selected)

4.7 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

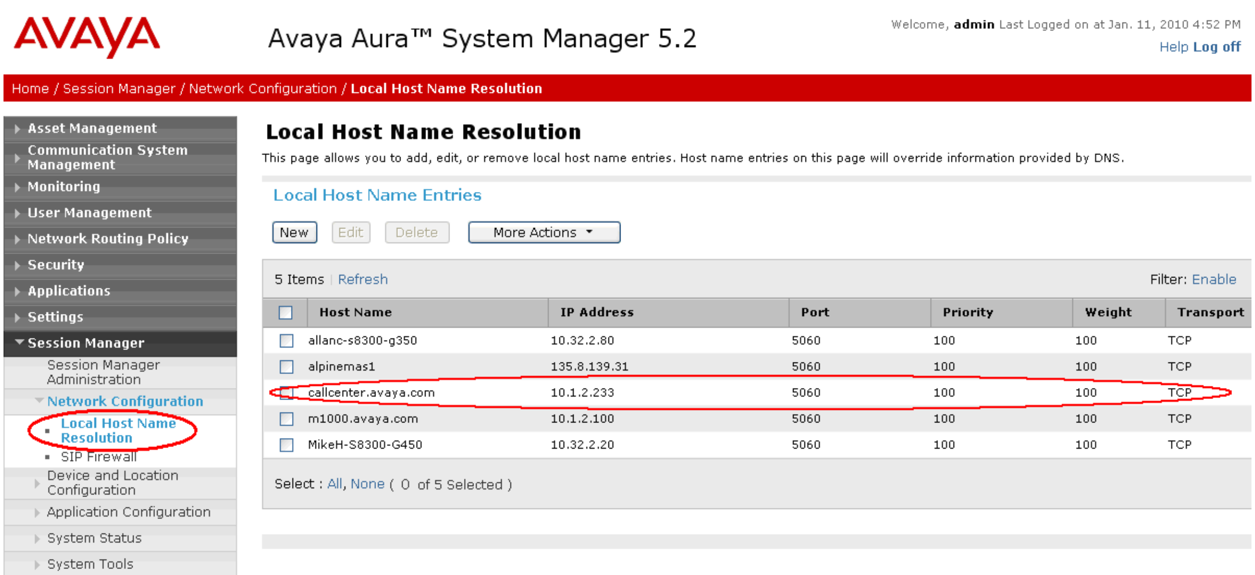
Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the resulting Session Manager definition.

4.8 Define Local Host Names

Any host names (FQDN's) referenced in SIP Entity definitions 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 defines the host name referenced in the SIP Entity configuration for Communication Manager in **Section 4.3**).



5 Configure Avaya Communication Server 1000

This section describes configuration of the CS1000 for call routing using a T1 PRI NI-1 interface to the Avaya G450 Media Gateway. These Application Notes assume that ISDN PRI is not being configured for the first time, so error detection thresholds and clock synchronization control are assumed to be in place. If not, refer to the ISDN Primary Rate Interface document in [7] for detailed descriptions. Furthermore, these Application Notes use the Coordinated Dial Plan (CDP) feature to route calls from the Avaya Communication Server 1000, over the PRI NI-1 trunks to Communication Manager. The CDP feature is assumed to be already enabled on Avaya Communication Server 1000, and therefore will not be described in detail.

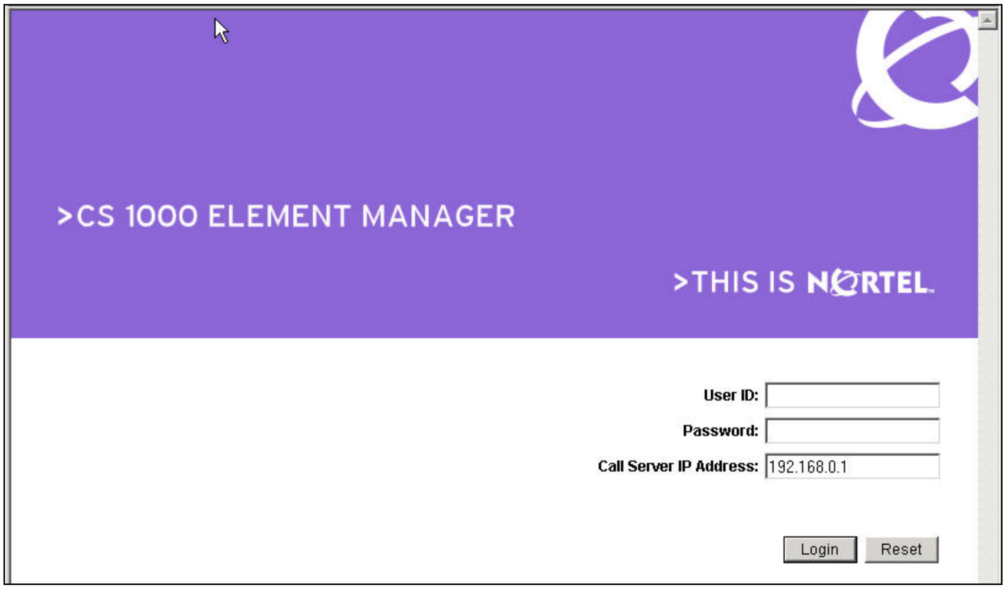
The procedures below describe the details of configuring the CS1000:

- Launch Element Manager
- Verify equipped feature packages
- Administer TMDI card
- Administer D-Channel
- Administer routes and trunks
- Administer route list block
- Administer distant steering code
- Enable TMDI card
- Enable D-Channel automatic establishment
- Enable D-Channel service messages

5.1 Launch Element Manager

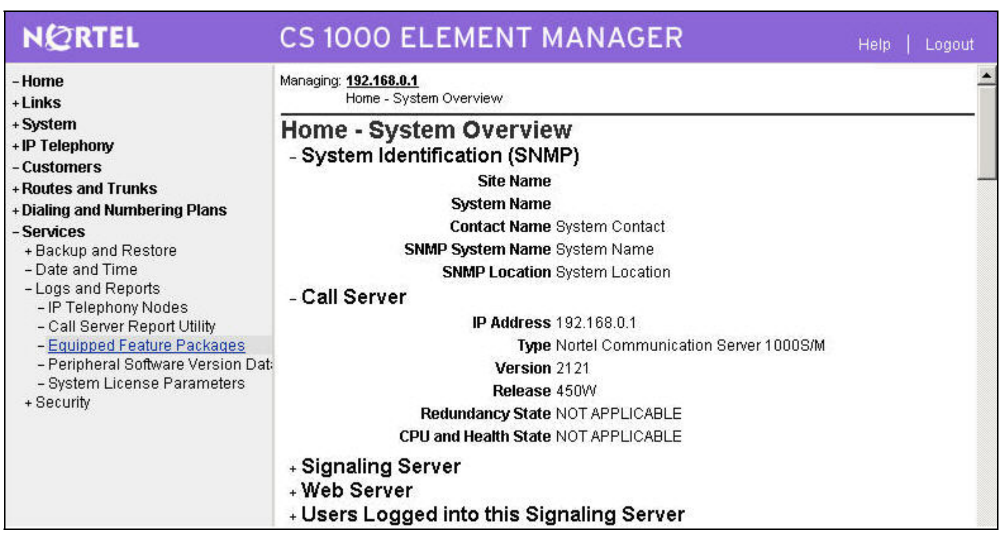
Access the CS1000 web based interface Element Manager by using the URL "http://<ip-address>" in an Internet browser window, where "<ip-address>" is the IP address of the Signaling Server. Note that the IP address for the Signaling Server may vary, and in this case "192.168.1.30" is used.

The **CS 1000 ELEMENT MANAGER** screen is displayed. Enter the appropriate credentials, retain the automatically populated value in the **Call Server IP Address** field, and click **Login**.



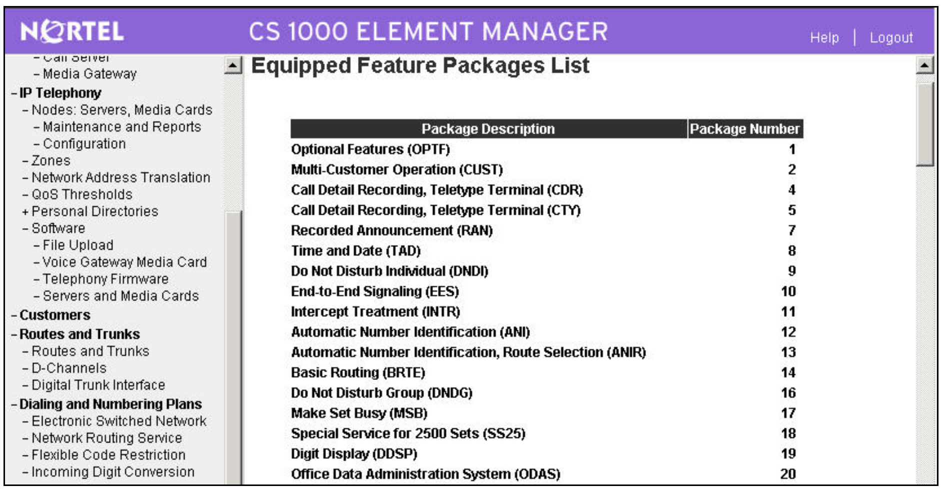
5.2 Verify Equipped Feature Packages

The **Home – System Overview** screen is displayed. Select **Services > Logs and Reports > Equipped Feature Packages** in the left pane.



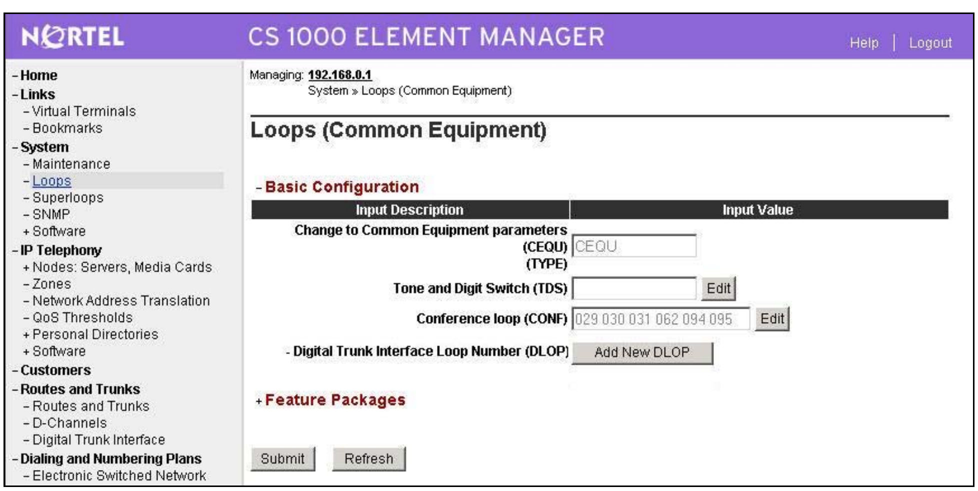
The **Equipped Feature Packages List** screen is displayed next, and shows a listing of the licensed feature packages in sequential order by package number. Scroll down the right pane as necessary to verify that the following feature packages are equipped:

- 19 Digit Display (DDSP)
- 59 Coordinated Dialing Plan (CDP)
- 95 Calling Party Name Display (CPND)
- 145 Integrated Services Digital Network (ISDN)
- 146 Primary Rate Access (CO) (PRA)
- 154 2.0 Mb/s Primary Rate Interface (PRI2)
- 184 Overlap Signaling (M1 to M1 and M1 to 1TR6 CO) (OVLP)
- 202 International Primary Rate Access (CO) (IPRA)



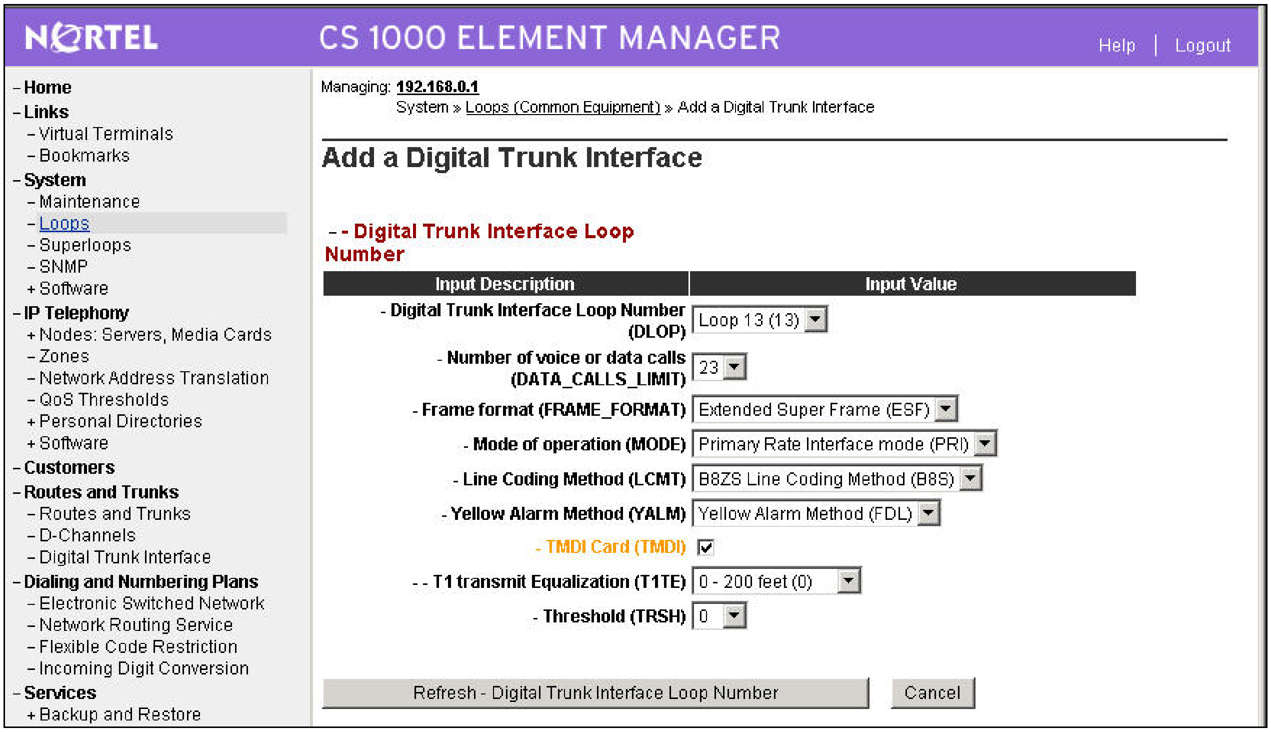
5.3 Administer TMDI Card

Select **System > Loops** from the left pane to display the **Loops (Common Equipment)** screen. In the **Digital Trunk Interface Loop Number (DLOP)** field, click **Add New DLOP** to add a digital trunk interface to the TMDI card.

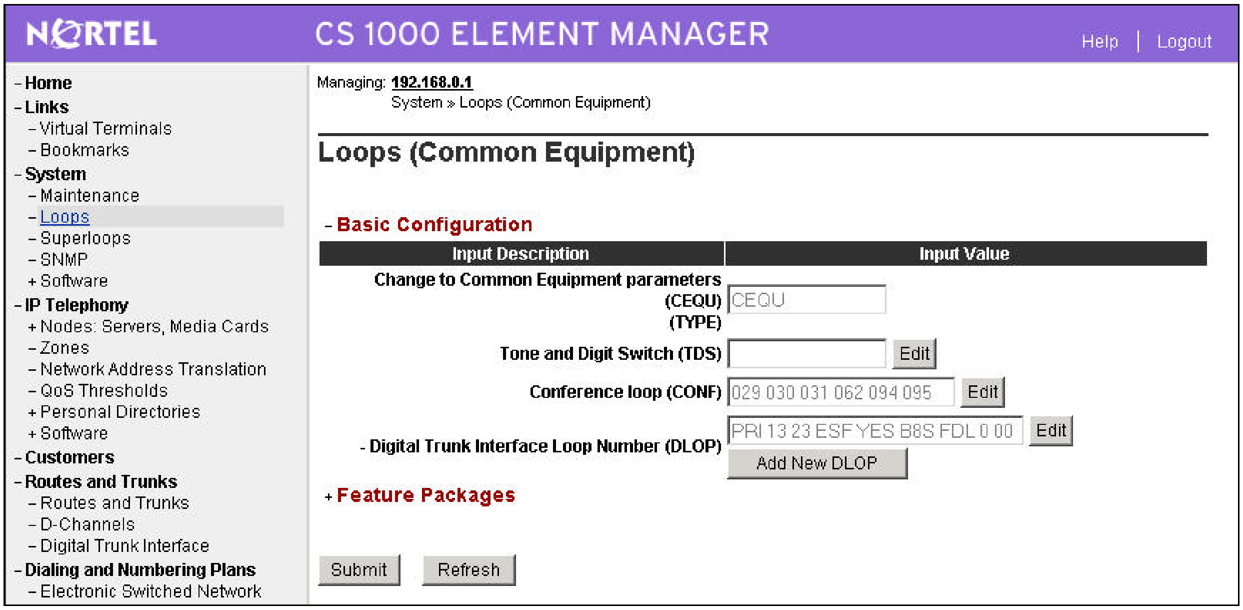


The **Add a Digital Trunk Interface** screen is displayed next. For the **Digital Trunk Interface Loop Number (DLOP)** field, select the loop number corresponding to the physical slot location of the TMDI card. In this case, "Loop 13 (13)" is selected from the drop-down list.

For the **Number of voice or data calls (DATA_CALLS_LIMIT)** field, select "23" from the drop-down list, to match the number of trunk members configured in Communication Manager in **Section 3.8.3**. For the **Mode of operation (MODE)** field, select "Primary Rate Interface mode (PRI)" from the drop-down list. For the **Threshold (TRSH)** field, select "0" from the drop-down list. Retain the default values for all remaining fields, and click **Refresh – Digital Trunk Interface Loop Number** at the bottom of the screen.

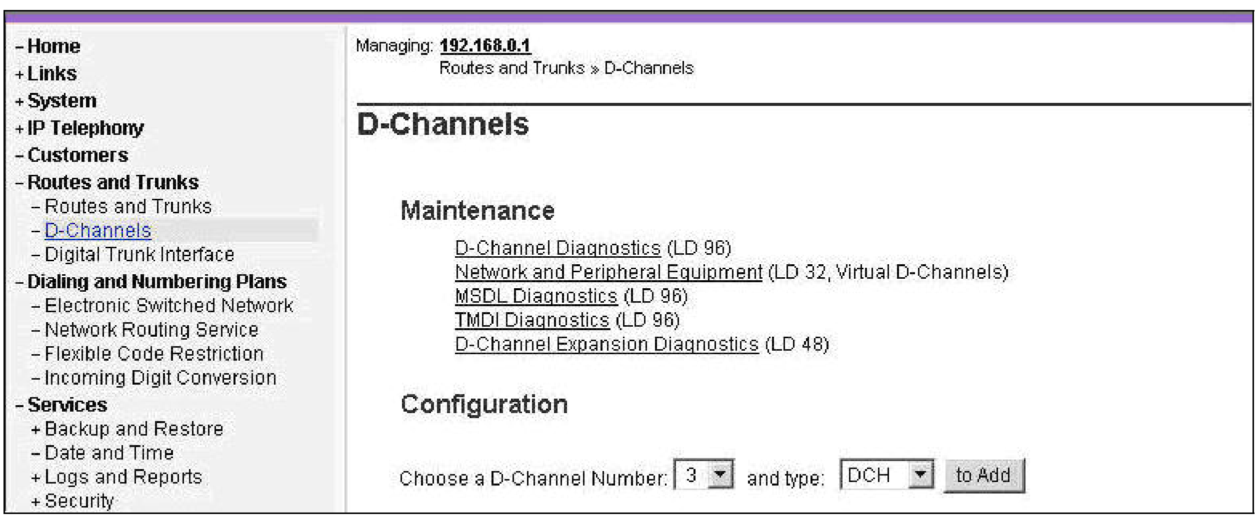


The **Loops (Common Equipment)** screen is displayed again, and updated with values in the **Digital Trunk Interface Loop Number (DLOP)** field. Click **Submit**.



5.4 Administer D-Channel

Select **Routes and Trunks > D-channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-Channel from the drop-down list (in this case "3"). Click **to Add**.

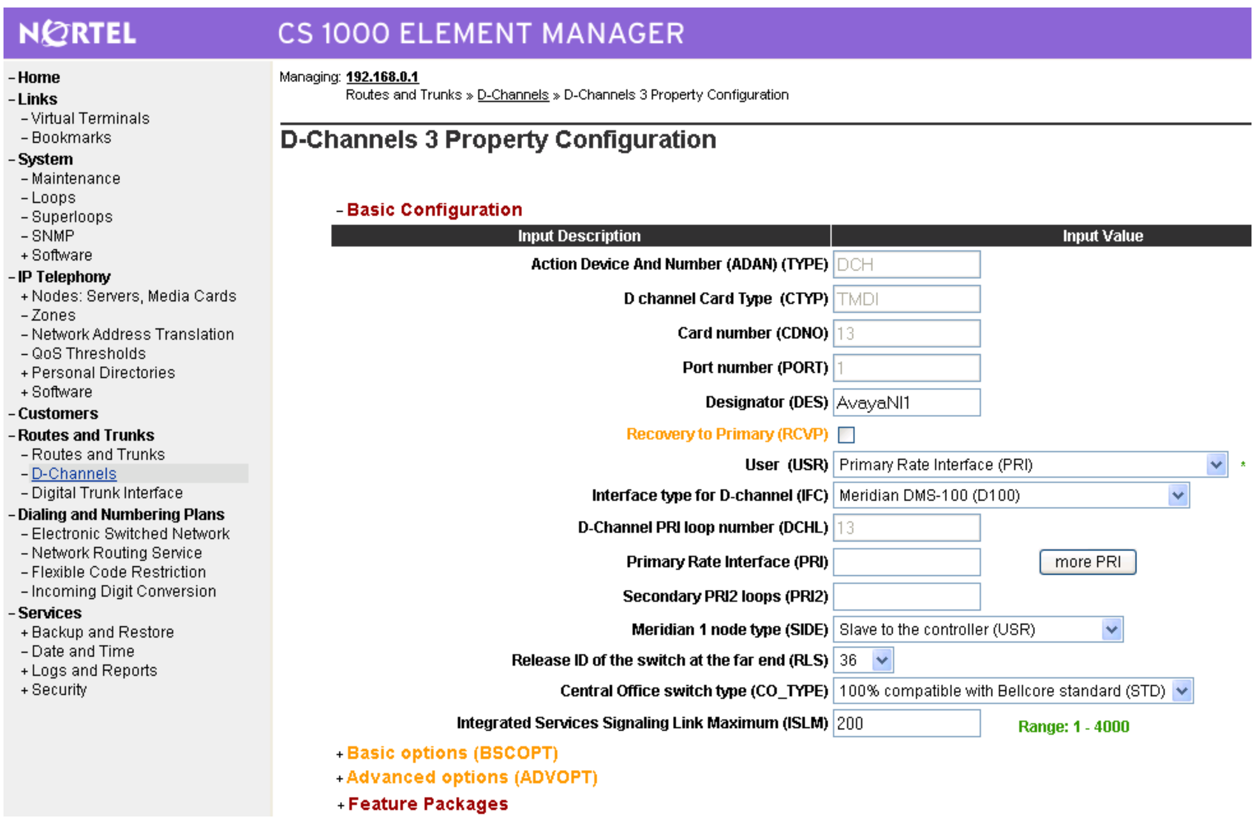


The **D-Channels 3 Property Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **D channel Card Type (CTYP):** "D-Channel on TMDI card (TMDI)"
- **Card number (CDNO):** Select the physical TMDI card location, in this case "13".
- **Port number (PORT):** "1"
- **Designator (DES):** A descriptive text.
- **User (USR):** "Primary Rate Interface (PRI)"
- **Interface type for D-channel:** "Meridian DMS-100 (D100)"
- **D-Channel PRI loop number**: The digital trunk interface loop number from **Section 5.3**.

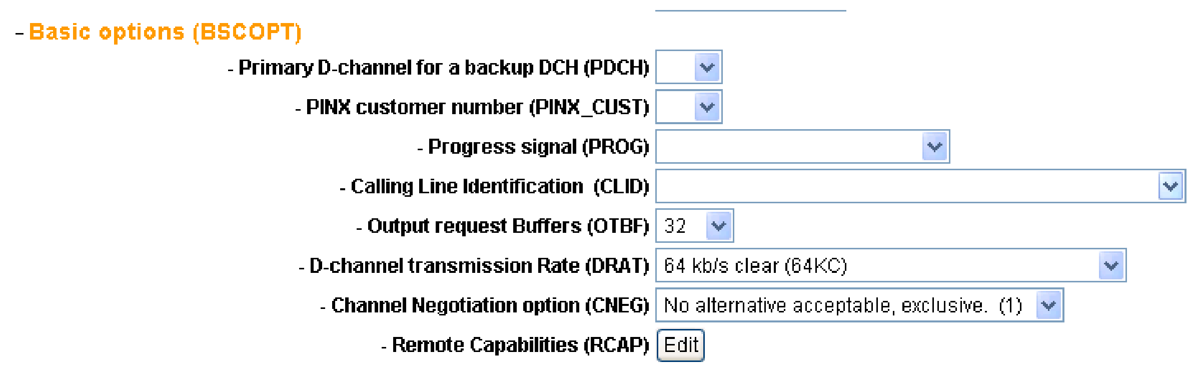
Note that the **Meridian 1 node type (SIDE)** field defaults to "Slave to the controller (USR)", which complements the DS1 interface setting on Avaya Communication Manager in **Section 3.7**.

For the **Release ID of the switch at the far end (RLS)** field, select "36" from the drop-down list. The following screen shows the configured D-Channel. Select **Basic options (BSCOPT)** toward the bottom of the screen to expand it.



For the **D-channel transmission Rate (DRAT)** field, select “64 kb/s clear (64KC)” from the drop-down list.

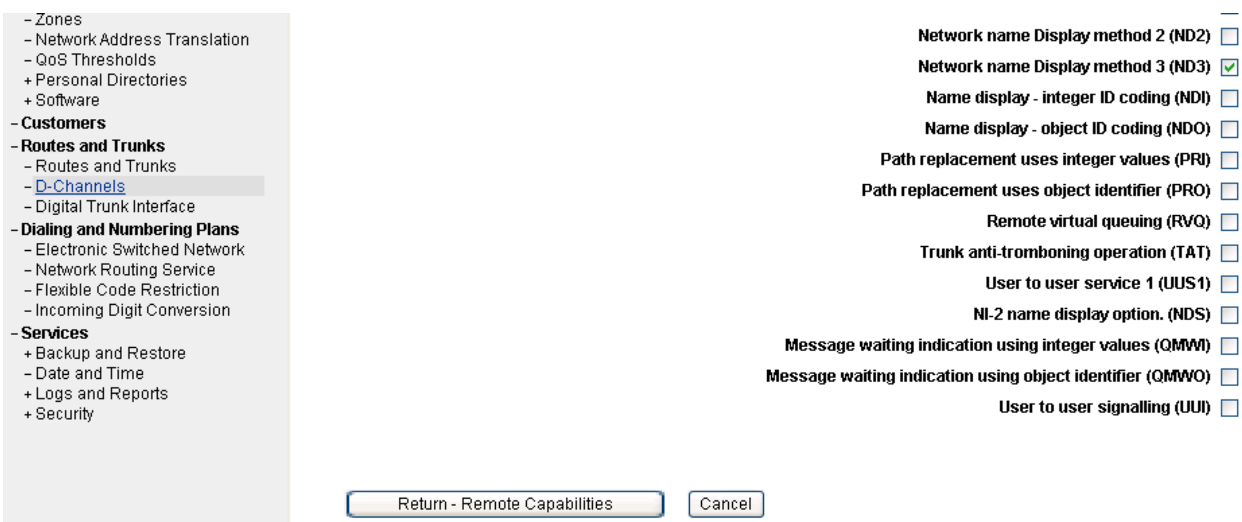
Retain the default values in the remaining fields, and click **Edit** next to the **Remote Capabilities (RCAP)** field.



The **Remote Capabilities Configuration** screen is displayed next. Scroll down the screen as necessary to check the following capability:

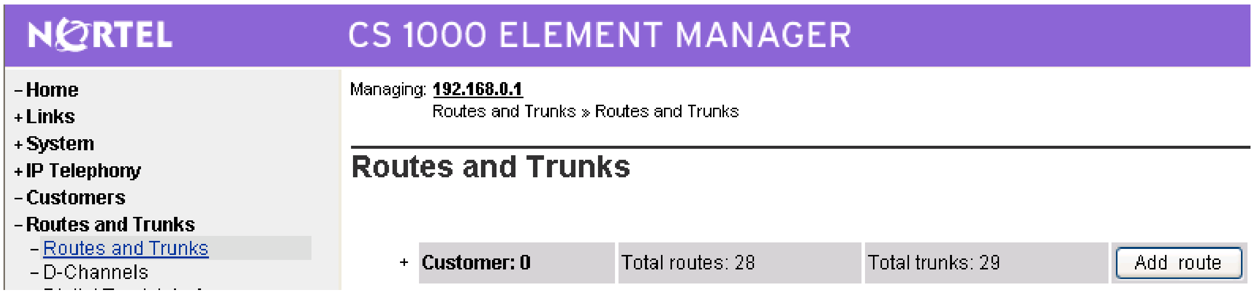
- **Network name Display Method 3 (ND3)**

Click **Return – Remote Capabilities** at the bottom of the screen. The **D-Channels 3 Property Configuration** screen is displayed again (not shown below). Click **Submit**.



5.5 Administer Routes and Trunks

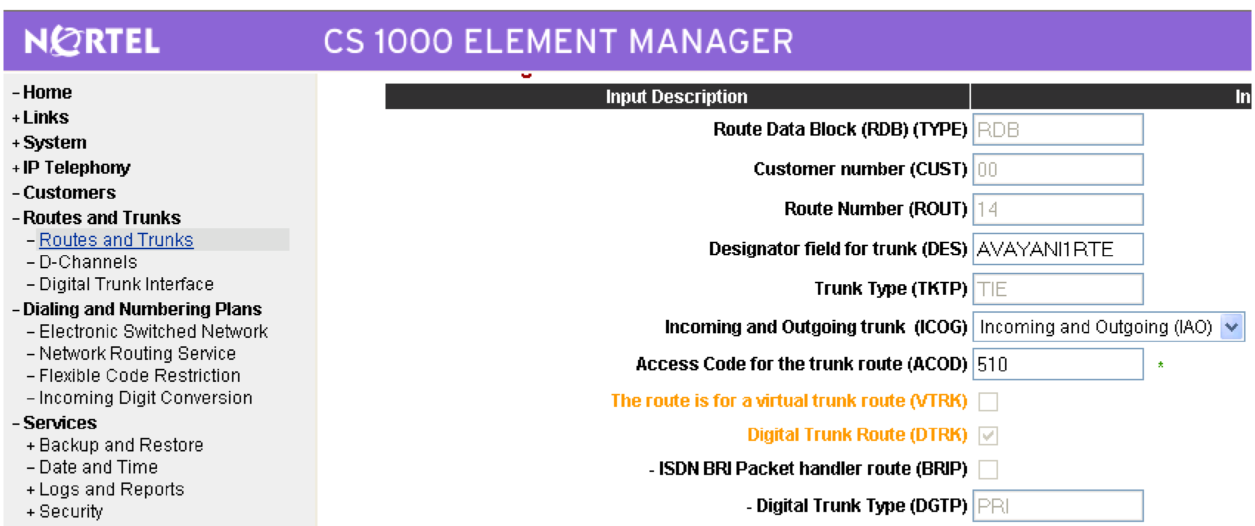
Select **Routes and Trunks** > **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. Next to the applicable **Customer** row, click **Add route**.



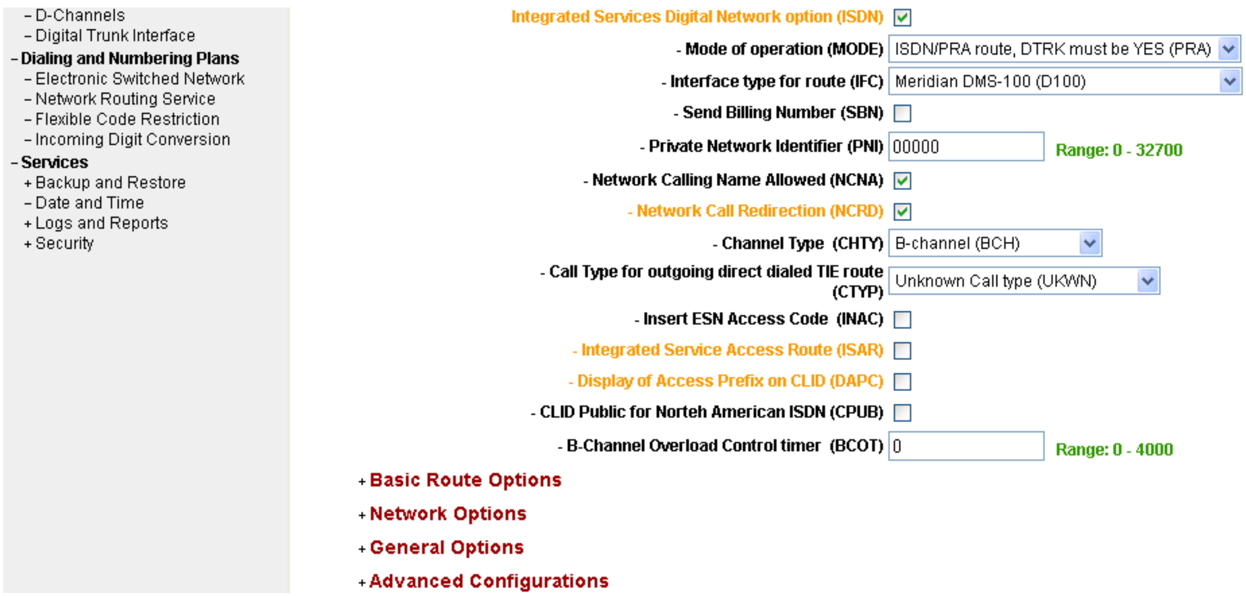
The **Customer 0, New Route Configuration** screen is displayed next. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Route Number (ROUT):** Select an available route number.
- **Designator field for trunk (DES):** A descriptive text.
- **Trunk Type (TKTP):** "TIE trunk data block (TIE)"
- **Incoming and Outgoing trunk (ICOG):** "Incoming and Outgoing (IAO)"
- **Access Code for the trunk route (ACOD):** An available access code.

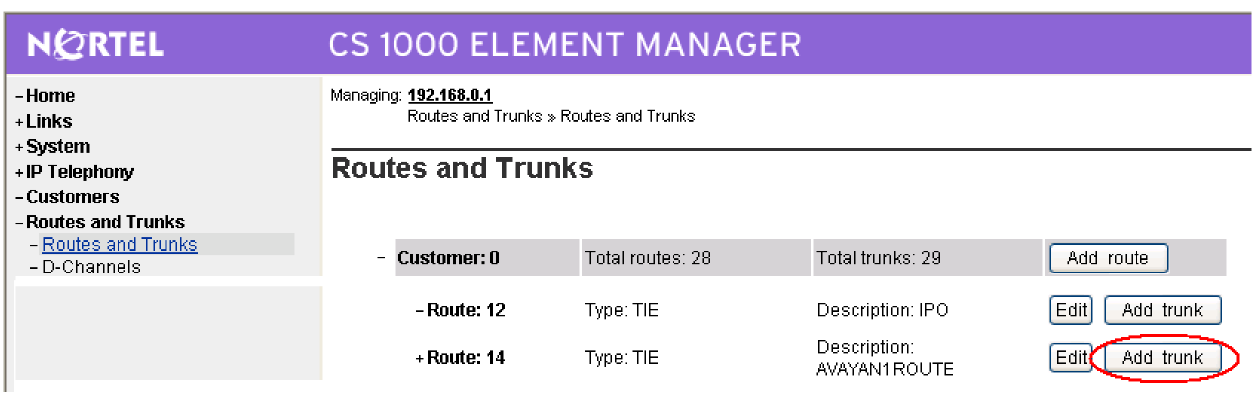
Verify that **Digital Trunk Route (DTRK)** is checked and the **Digital Trunk Type (DGTP)** field is set to "PRI". The following screen shows the resulting configuration for the route.



Scroll down the screen, check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. For the **Mode of operation (MODE)** field, select "ISDN/PRA route, DTRK must be YES (PRA)" from the drop-down list. For the **Interface type for route (IFC)** field, select "Meridian DMS-100 (D100)" from the drop-down list. For the **Call Type for outgoing direct dialed TIE route (CTYP)** field, select "Unknown Call type (UKWN)" from the drop-down list. Check the check boxes **Network Calling Name Allowed (NCNA)** and **Network Call Redirection (NCRD)**. Scroll down to the bottom of the screen, and click **Submit**.

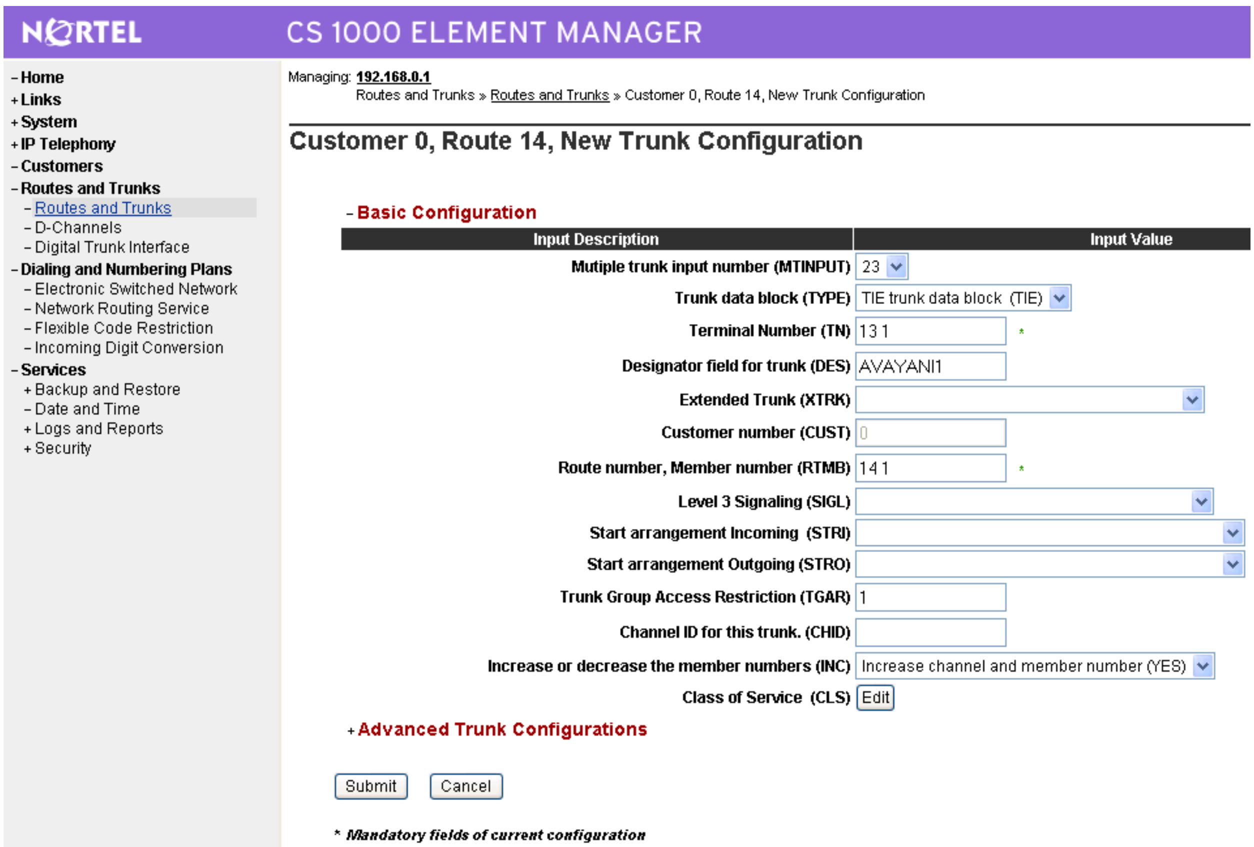


The **Routes and Trunks** screen is displayed again, and updated with the newly added route. Click the **Add trunk** button next to the newly added route.



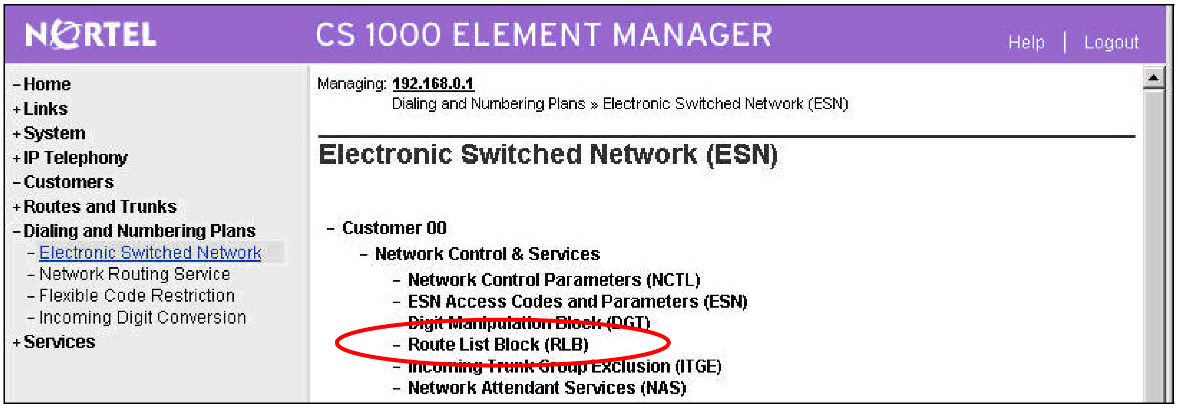
The **Customer 0, Route 14, New Trunk Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit**. The **Multiple trunk input number (MTINPUT)** field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk.

- **Multiple trunk input number (MTINPUT):** "23" (must match values in **Sections 5.3** and **3.8.3**
- **Terminal Number (TN):** The TMDI slot number and port "1".
- **Designator field for trunk (DES):** A descriptive text.
- **Route number, Member number (RTMB):** Current route number and starting member.
- **Trunk Group Access Restriction (TGAR):** Desired trunk group access restriction level.

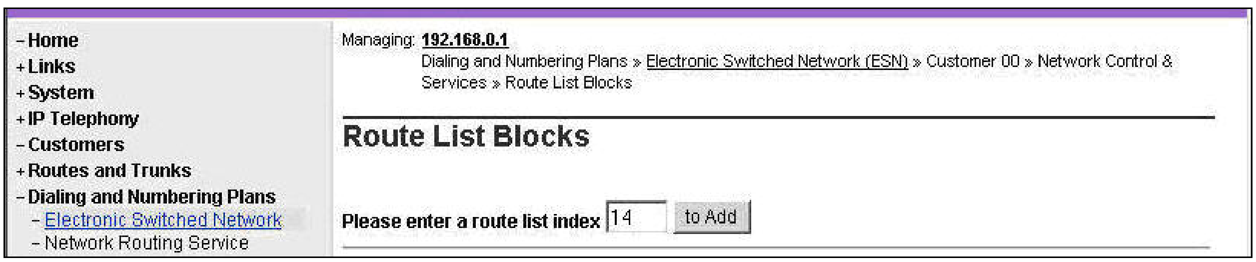


5.6 Administer Route List Block

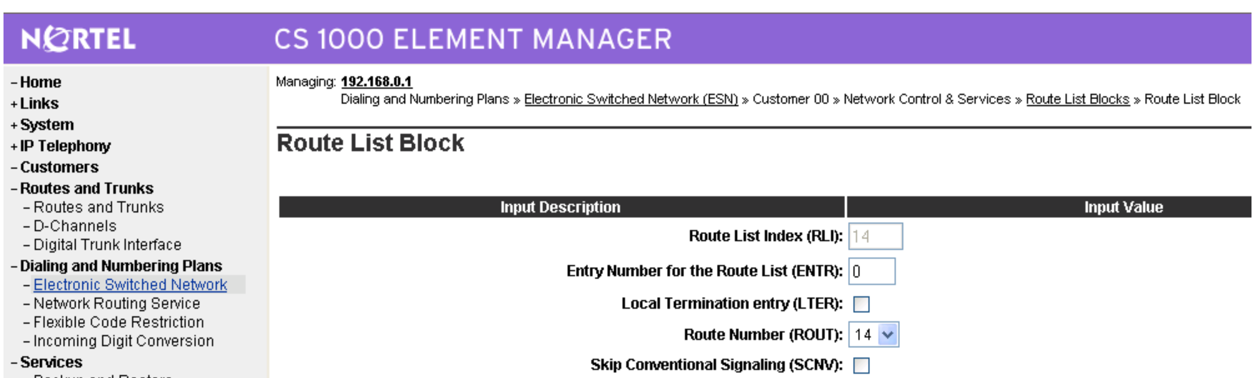
Select **Dialing and Numbering Plans > Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Route List Block (RLB)**.



The **Route List Blocks** screen is displayed. In the **Please enter a route list index** field, enter an available route list block number (in this case "14"). Click **to Add**.

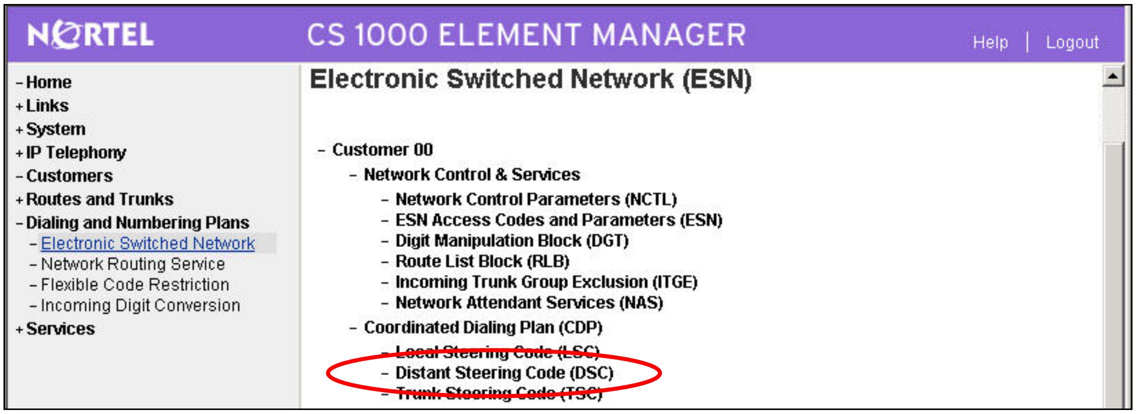


The **Route List Block** screen is updated with a listing of parameters. For the **Route Number (ROUT)** field, select the route number from **Section 5.5**. Retain the default values for the remaining fields, and scroll down to the bottom of the screen and click **Submit** (not shown).

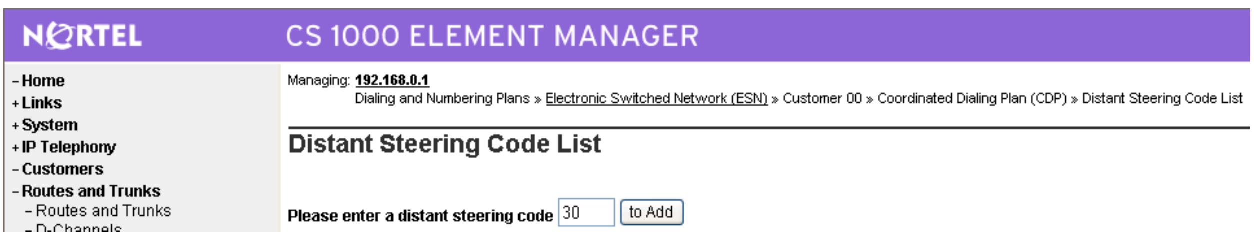


5.7 Administer Distant Steering Code

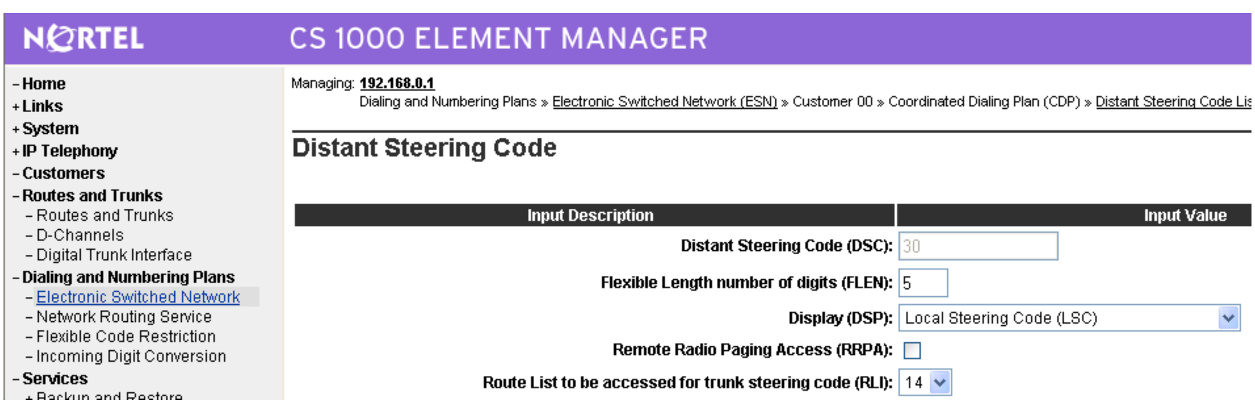
The **Electronic Switched Network (ESN)** screen is displayed again. Select **Distant Steering Code (DSC)** to add an entry to route 30xxx calls to Avaya Communication Manager.



The **Distant Steering Code List** screen is displayed next. In the **Please enter a distant steering code** field, enter the dialed prefix digits to match on (in this case "30"). This specification will match dialed extensions of the form 30xxx. 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 in **Section 5.6** from the drop-down list. Retain the default values in all remaining fields, and scroll down to the bottom of the screen to click **Submit** (not shown).



5.8 Enable TMDI Card

The remaining steps required to bring the PRI NI-1 trunk into service are best accomplished using the Command Line Interface (CLI) of the CS1000. Access the CLI via a hyper terminal application running on a PC, which has a serial cable connected to the CS1000 serial port, or via *telnet* to the IP address of the CS1000 Signaling Server (192.168.1.30 in the sample configuration).

The CLI is a character-based serial interface to the operating system and overlay programs on each system component. The program issues a prompt for input, and the system administrator enters the appropriate response through the keyboard followed by the **Enter** key. The output from the CS1000 command line interface has been trimmed down in the subsequent sections in order to focus on the key settings for the configuration. Values highlighted in bold represent values entered by the system administrator.

Command	Comment
> **login** USERID? **xxxxx** PASS? **yyyyy**	Issue the login command. Enter a valid user ID. Enter a valid user password.
TTY #00 LOGGED IN xxxxx 16:52 24/5/2010	A sample response indicating successful log in.
> **ld 96** . **enl tmdi 13 all**	Use load 96 to enable the TMDI card. Enable the TMDI card with the physical slot number of the TMDI card and the option "all".

5.9 Enable D-Channel Automatic Establishment

Use the CLI to enable automatic establishment for the administered D-Channel.

Command	Comment
> **ld 96** . **enl auto 3**	Use load 96 to enable automatic establishment for the D-Channel. Enable the D-Channel automatic establishment with the D-Channel number, in this case "3".

5.10 Enable D-Channel Service Messages

Service messages are used to maintain control and obtain status of the PRI trunk. By default, service messages are disabled for PRI interface type "Meridian DMS-100 (D100)". They must be enabled in this configuration for the individual B channels to come into service. The CLI can be used to enable service messages on the D-Channel. Note that the D-Channel may have to be disabled and enabled before and after enabling service messages.

Command	Comment
> **ld 96** . **dis dch 3** . **enl serv 3** . **enl dch 3**	Use load 96 to enable service messages for the D-Channel. Disable the D-Channel, in this case "3". Enable D-Channel service messages. Enable the D-Channel again.

6 Verification Steps

6.1 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 show OOS/PINS (out of service, pending in service), verify that service messages have been enabled on the D-Channel in the CS1000 (See **Section 5.10**). 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. One should also verify that the trunk parameters specified for Communication Manager (**Sections 3.7-3.8**) and the CS1000 (**Sections 5.4-5.5**) have been configured correctly.

```
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 3.9.2**. 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 3.9.1**. Verify the signaling group is "in-service" as indicated in the **Group State** field shown below.

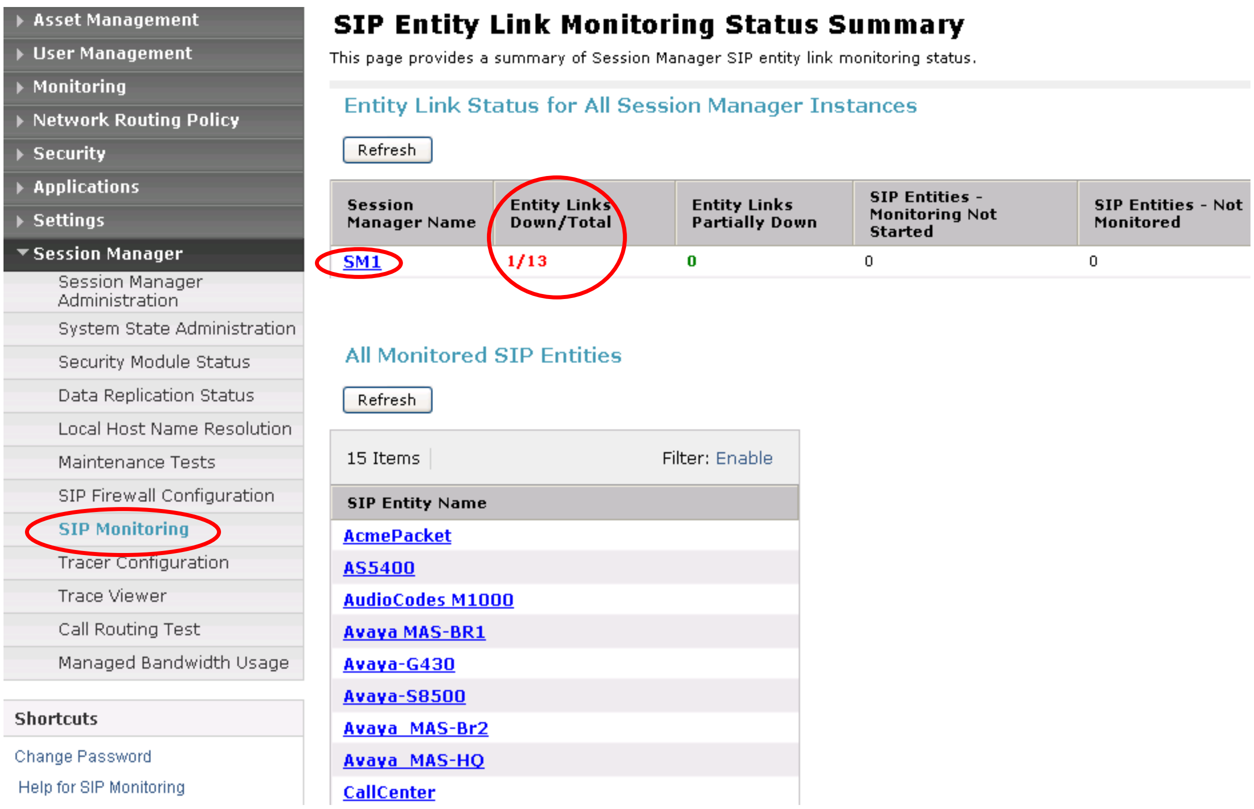
```
status signaling-group 32
                         STATUS SIGNALING GROUP

        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
```

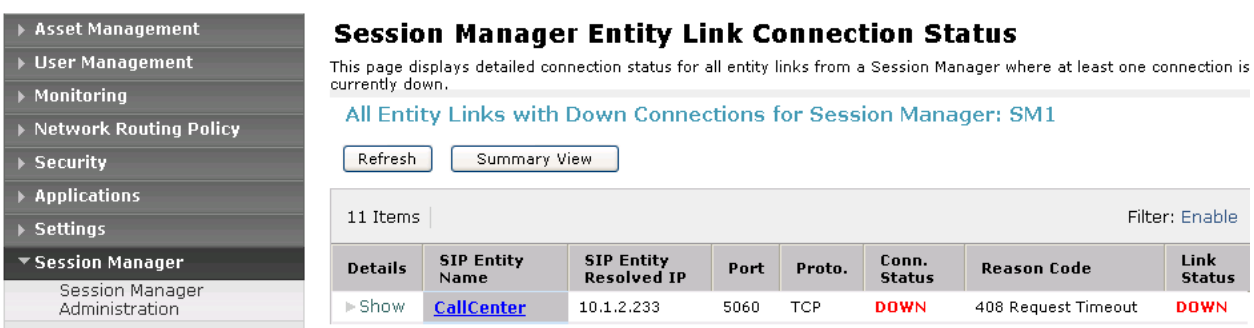
Finally, make a call between the Avaya 9600 Series IP Telephone and the Avaya i2004 IP Telephone and verify two-way audio. Verify the status of connected trunks by using the "status trunk" command for the PRI QSIG trunk group (100) to the CS1000. More information can be obtained by using "status trunk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

6.2 Verify Avaya Aura™ Session Manager

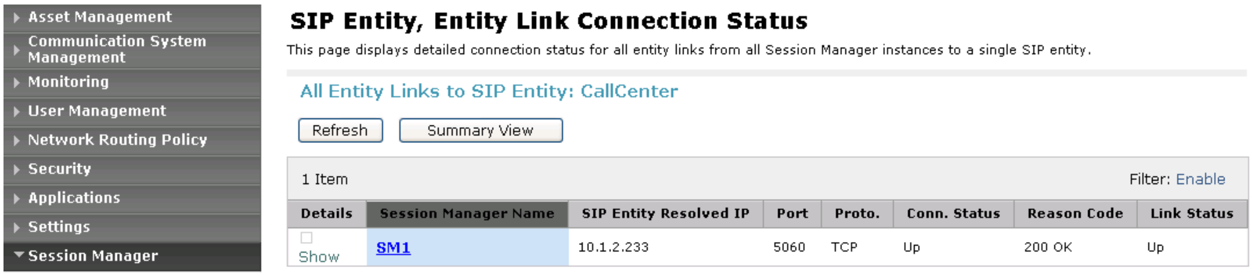
Expand the **Session Manager** menu on the left and click **SIP Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing. In the sample screen below, the SIP trunk to SM1 has been busied out on Communication Manager, so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

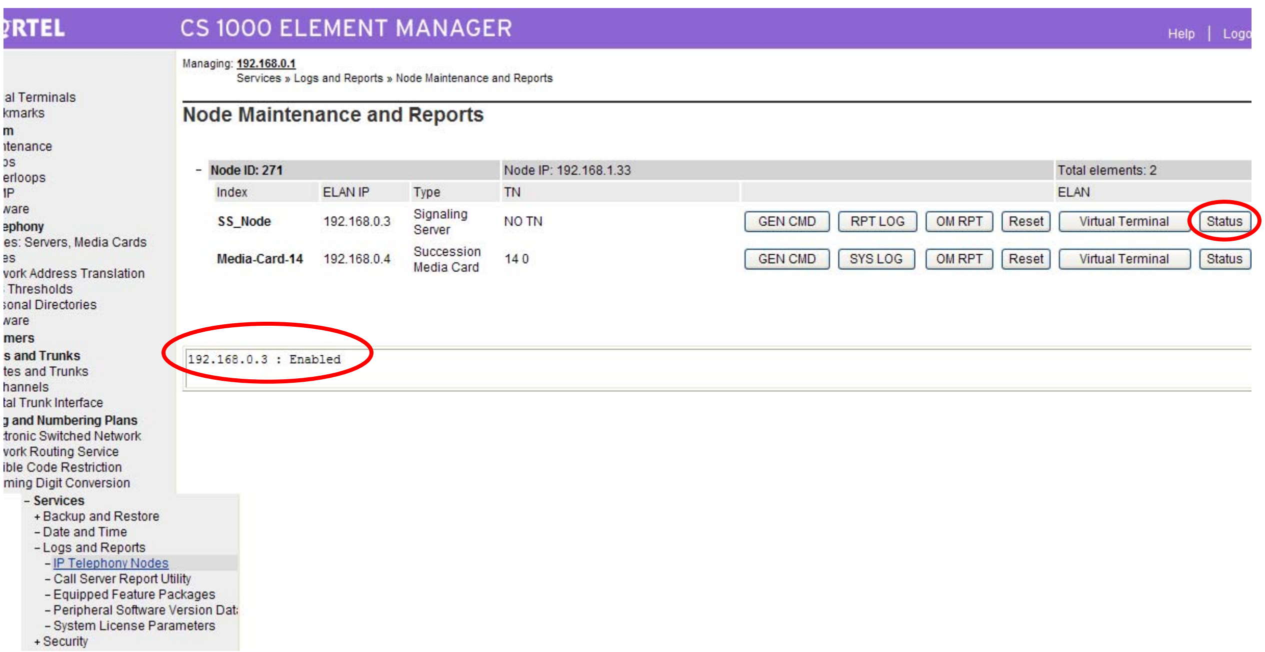


Once the source of the problem has been located, verify that the link status has changed as shown below. Note that the time period for the status to change is dependent on the SIP Link Monitoring parameter settings defined for the SIP Entity corresponding to Communication Manager. See [2,3] for more information.

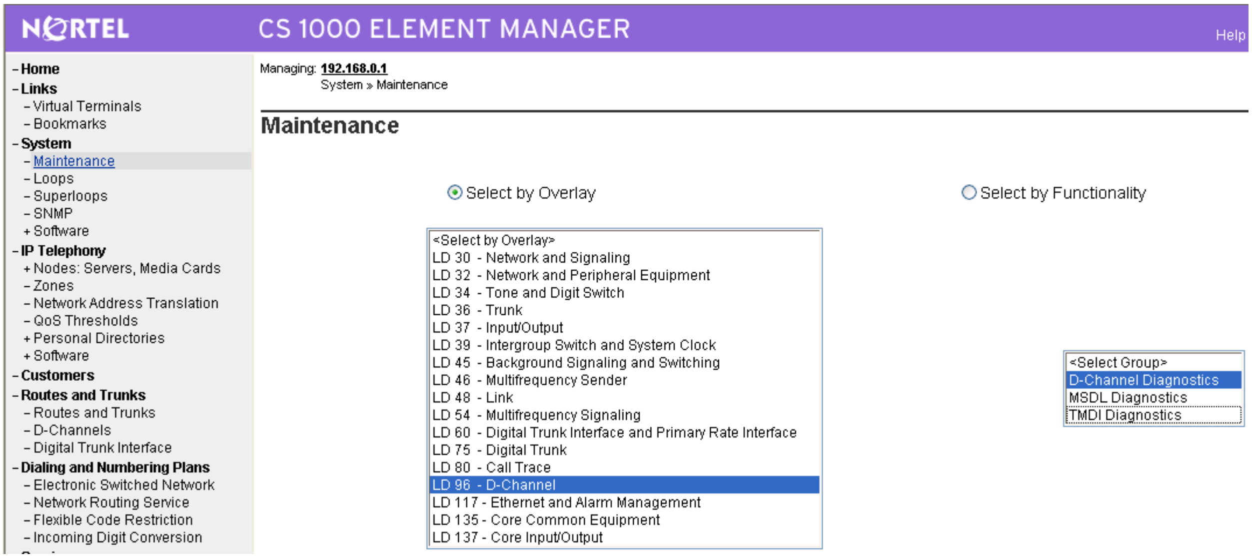


6.3 Verify Nortel Communication Server 1000

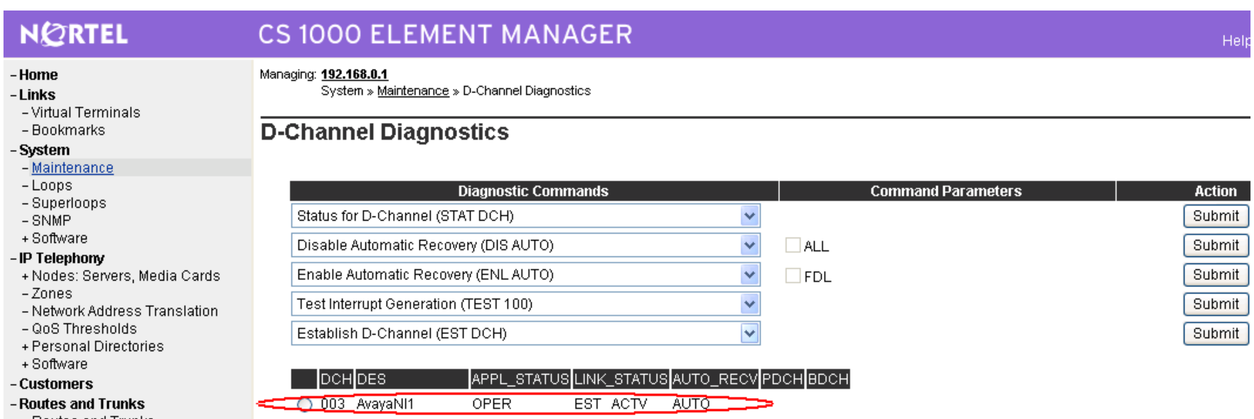
Select **Services->Logs and Reports->IP Telephony Nodes** on the left. Click **Status** for the "SS_Node" to verify that the signaling server is enabled and operational.



Select **System->Maintenance** on the left, and then select **Select by Overlay** and then **LD 96 – D-Channel** in the center window. Select **D-Channel Diagnostics** in the window that appears on the right.



The screen that results, shown below, will indicate the operational state of the D-Channel, in this case “EST ACTV” under the LINK_STATUS column. Other diagnostic tests can be run from this page by selecting the D-Channel, one of the **Diagnostic Commands**, and then clicking on **Submit**.



6.4 Verification Scenarios

Verification scenarios for the configuration described in these Application Notes are listed below. Note that there are some telephone display limitations as described in **Section 6.5**.

- Basic calls between various telephones on the Communication Manager 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.

    - Call hold/unhold
    - Unattended transfer
    - Attended transfer
    - Call forwarding
    - Conference
    - Calling number block

6.5 Telephone Display Limitations

The following are limitations in name and number displays for some of the scenarios verified in the previous section. Unless otherwise specified below, telephone displays will show the correct name and number. Abbreviations are used for brevity (CM = Communication Manager, CS1K = Communication Server 1000).

6.5.1 Basic Calls

When calling from CM to CS1K, the CM telephone shows called number during the ringing phase, and connected name when the call is answered.

When calling from CS1K to CM, the CS1K telephone displays the called number.

6.5.2 Call Hold/Unhold

For a call between CM and CS1K that is held by the CS1K user, when the CS1K user takes the call off hold, the CS1K telephone display changes to the local CS1K PRI trunk designation.

6.5.3 Call Transfer

For a call between CM and CS1K that is transferred to a party on the same PBX as the transferring party, the transferred party's display is not updated (i.e., displays the name of the transferring party).

For a call between CM and CS1K that is transferred to a party that is not on the same PBX as the transferring party, the displays of both the transferred and transfer-to party are not updated (i.e., display the name of the transferring party). Also note that the trunks on the transfer-to party PBX are not released.

6.5.4 Call Forward

For a call from CS1K to CM that is forwarded to another telephone, the CS1K telephone display is not updated (i.e., displays the originally called party).

7 Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 front-ended by an Avaya G450 Media Gateway via a PRI NI-1 trunk can be integrated with Session Manager and Communication Manager.

8 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] *Administering Avaya Aura™ Session Manager*, Release 5.2, Issue 2.0, November 2009, Document Number 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 5.2.1:

[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™ Session Manager*, Release 5.2, Issue 2.0, November 2009, Document Number 03-603324, 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 Communication Server 1000 4.5:

[7] *ISDN Primary Rate Interface Installation and Configuration*, Nortel Communication Server 1000 Release 4.5, Document Number 553-3001-201, available on the Nortel Communication Server Electronic Reference Library CD.

Avaya Application Notes:

[8] *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.

[9] *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.

[10] *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*, 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