Avaya Solution Interoperability Lab

Configuring Avaya Aura™ Session Manager 5.2 with Avaya Aura™ Communication Manager Access Element, Avaya Voice Portal and Avaya Aura™ Communication Manager Feature Server – Issue 1.0

Abstract

These Application Notes describe the configuration of Avaya Aura™ Session Manager R5.2, Avaya Aura™ Communication Manager Access Element R5.2.1 with an Avaya G650 Media Gateway, Avaya Aura™ Communication Manager operating as a Feature Server, and Avaya Voice Portal R5 to support a Voice Portal First (VP 1st) solution.

- Avaya Aura™ Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies and registrations for SIP endpoints.
- Avaya Aura™ Communication Manager operates as a Feature Server to support SIP endpoints which communicate with Avaya Aura™ Session Manager over SIP trunks.
- Avaya G650 Media Gateway consolidates PSTN facilities by concentrating and routing the calls to Avaya Aura™ Communication Manager Access Element which communicates with Avaya Aura™ Session Manager and Avaya Voice Portal over SIP trunks.
- Avaya Voice Portal is a Web services based, speech enabled interactive voice response system that can accept traditional DTMF touch tone inputs and prerecorded audio files for output, as well as VoiceXML2.0 compliant speech applications to guide callers through call flows.

These Application Notes provide information for the setup, configuration, and verification of the call flows tested on the Voice Portal First (VP 1st) solution.

Table of Contents:

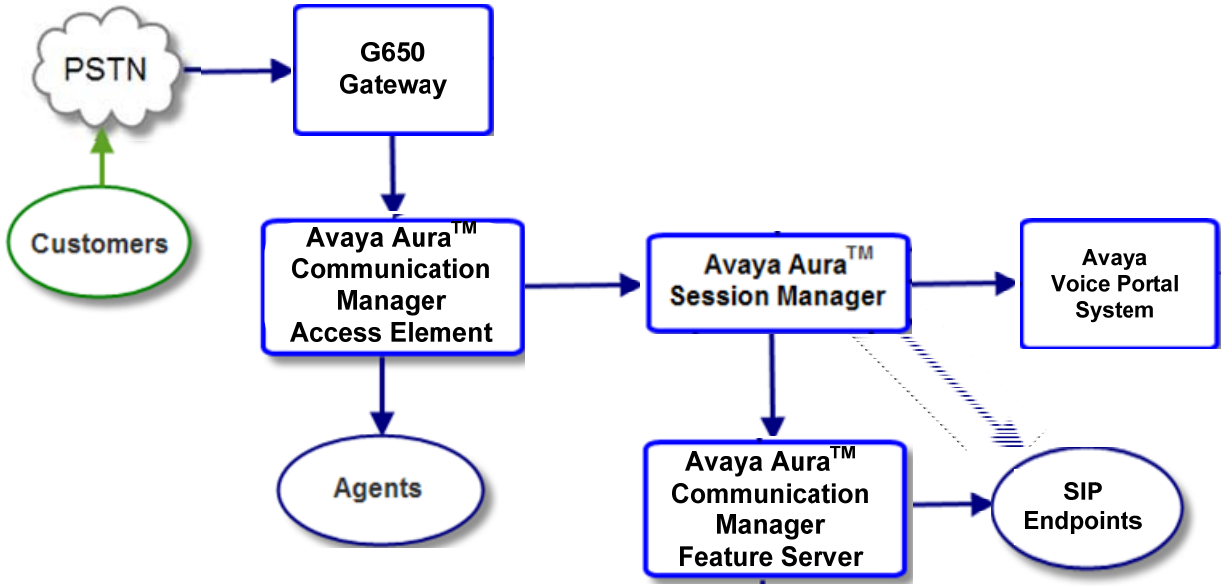
1. Introduction

The Voice Portal First solution in the sample configuration is comprised of five Avaya products – Avaya Aura™ Communication Manager Access Element with an Avaya G650 Media Gateway, Avaya Aura™ Communication Manager Feature Server, Avaya Aura™ Session Manager, and Avaya Voice Portal.

In a typical scenario, a customer call enters an Avaya G650 Media Gateway through the Public Switched Telephone Network (PSTN) over a DS1 trunk. The Avaya G650 Media Gateway delivers the call to the Avaya Aura™ Communication Manager Access Element. Communication Manager routes the calls to the Session Manager over a network connection using a SIP trunk.

The Session Manager routes the call to the Voice Portal system where the customer interacts with a self service application. After completing the self service application, the customer may opt to speak to a contact center agent. If so, the Voice Portal system delivers the call back to Session Manager, which then routes the call back to the Communication Manager Access Element. After arriving at a Communication Manager, the caller is connected to an agent. Alternatively, the call can originate from a SIP endpoint registered to the Session Manager. In this scenario, the Communication Manager Feature Server routes the call to the Session Manager over a SIP trunk.

The figure provides an overview of a typical Voice Portal First solution.



Voice Portal First Solution Overview

These Application Notes describe the administrative steps required for configuring the Avaya products that comprise the Voice Portal First solution.

1.1. Voice Portal First Solution Overview

The following section describes the components of the Voice Portal First Solution.

Core Site

1.1.1. Avaya Aura™ Communication Manager Access Element

Avaya Aura™ Communication Manager provides Call Center Software functionality when a customer elects to talk with an agent. Calls from VP application are delivered to Communication Manager Access Element via direct SIP trunks through Session Manager.

1.1.2. Avaya G650 Media Gateway (G650)

The Avaya G650 Media Gateway provides consolidation of PSTN facilities into SIP.

Data Center

1.1.3. Avaya Aura™ Session Manager (SM)

Avaya Aura™ Session Manager is a SIP proxy/routing engine that is capable of routing SIP requests throughout a network. The Avaya Aura™ System Manager (SMGR) provides administration. Session Manager provides the following functionality:

User Relation Element (URE)
The User Relation Element provides a mapping for all devices associated with a user, acting as a registrar and location server. It also provides origination and termination routing for a user based on their configured and registered features by routing sessions in progress to appropriate feature servers or delivery platforms for featured services. There may be multiple User Relation Elements – all share the same configuration data and real time data.

SIP Routing Element (SRE)
The SIP Routing Element provides site to site routing services including number/name resolution, richly manages network ingress and egress including carrier selection for least cost, time of day, load balancing, and media preferences. There may be multiple SIP Routing Elements – all share the same configuration data and some real time data.

Session Manager does the following:

- Routes SIP sessions across the network with centralized routing policies

- Centralizes SIP registrations and location services
- Scales to support up to 25,000 locations and up to 250,000 users[1]
- Enables applications to be decomposed and distributed across the enterprise network
- Introduces application sequencing preparing for applications to run alongside Communication Manager.
- Provides the gateway for the enterprise for external SIP adjuncts.
- Is available with geographically dispersed redundancy, that is, the Session Manager instances can be spread across distance (WAN) but in the event of loss of one Session Manager instance, service still continues to operate normally.
- All the Session Manager instances in an enterprise function as a whole, providing continuous service to all users in the event of Session Manager one instance failures.
- Each Session Manager instance operating in the "active" mode, processes INVITE, REGISTER, SUBSCRIBE and other SIP messages
- Each user in the enterprise is assigned two Session Manager instances to support that user's proxy, registrar, application sequencer, and event-handler.

1.1.4. Avaya Aura™ Communication Manager Feature Server

Avaya Aura™ Communication Manager operating as a Feature Server supports IP Multimedia Subsystem (IMS)-SIP users that are registered to Avaya Session Manager. The Communication Manager server is connected to Session Manager via an IMS-enabled SIP signaling group.

1.1.5. Avaya Voice Portal (VP)

VP is a Web based speech enabled interactive voice response system that can accept traditional DTMF touch tone inputs and prerecorded audio files for output. It uses VoiceXML2.0 compliant speech applications to guide callers through self service call flows.

Avaya Voice Portal is comprised of a Voice Portal Management System (VPMS) server, a Media Processing Platform (MPP) server, a Web Application Server, and a Speech Processing server. Avaya Dialog Designer (DD) application is deployed on a web application server to perform a custom self service workflow.

Voice Portal Management System (VPMS) manages the MPPs and provides a web interface for administering VP.

Media Processing Platform (MPP) provides the main processing for self service applications. Details are described below:

- Uses H.323, SIP, and RTP protocols to communicate with external services, such as Session Manager.

[1] If there are "N" Session Manager instances in an enterprise, then the total capacity of users that can be supported is = (N-1) X 50,000.

- Runs the Avaya VoiceXML browser to interpret VoiceXML2.0 compliant speech applications.
- Provides proxy interfaces to communicate with the TTS (Text To Speech) servers and ASR (Automatic Speech Recognition) servers. The MPP uses Media Resource Control Protocol (MRCP) to control ASR and TTS servers.

Web Application Server – The web application server utilizes a workflow defined in Dialog Designer to provide the self service application. The MPP calls the application server and coordinates media resources available for processing the call.

Speech Server – The speech server provides Automatic Speech Recognition (ASR) and Text To Speech (TTS) capabilities.

1.2. Network Topology

As shown in **Figure 1,** the Avaya 9600-Series IP Telephone (H.323) and 2420 Digital Telephone are supported by Avaya Aura™ Communication Manager Access Element. The Communication Manager Access Element is connected over a SIP trunk to the Avaya Aura™ Session Manager, using its SM-100 (Security Module) network interface. All inter-system calls are carried over these SIP trunks.

Avaya Aura™ Session Manager is managed by a separate Avaya Aura™ System Manager. Avaya 9620 IP Telephones configured as SIP users utilize the Avaya Aura™ Session Manager User Registration feature and require a Communication Manager operating as a Feature Server.

For the sample configuration, Avaya Aura™ Session Manager runs on an Avaya S8510 Server, and Avaya Aura™ Communication Manager 5.2.1 runs on an Avaya S8730 Server with Avaya G650 Media Gateway. Two Avaya Aura™ Session Managers are deployed as a pair of active-active redundant servers. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura™ Communication Manager 5.2.1.

These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of Communication Manager Feature Server and the endpoint telephones will not be described (see the appropriate documentation listed in **Section 9**).

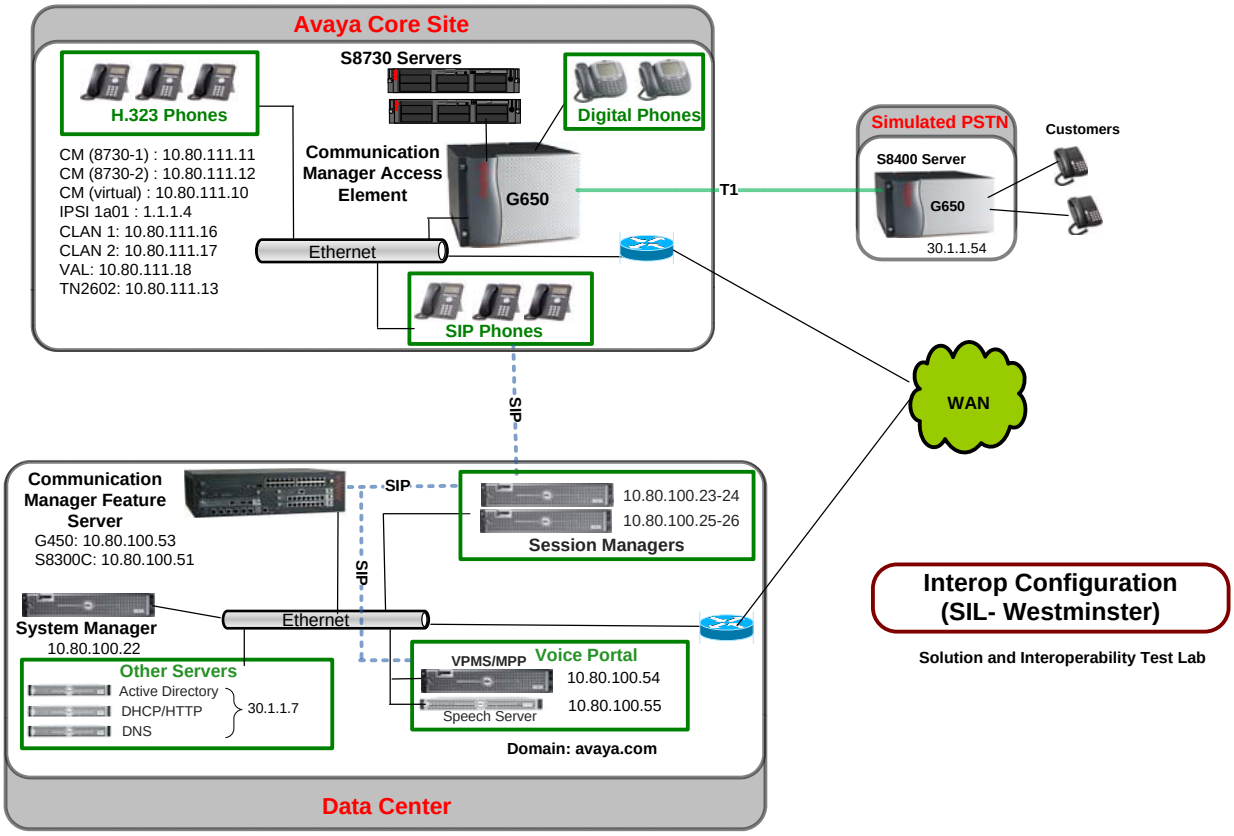


Figure 1: Voice Portal First - Sample Configuration

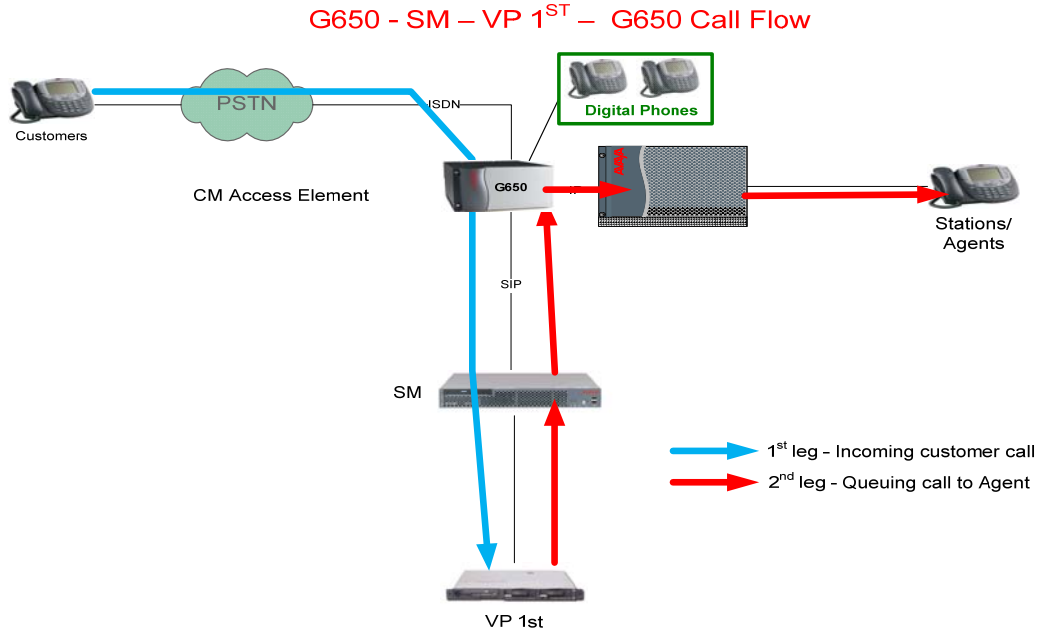
1.3. Equipment and Software Validated

The following equipment and software were used for the sample configuration.

Component	Software Version
Avaya Aura™ Session Manager on Avaya S8510 server	Release 5.2.0.1.520017-11-18-2009
Avaya Voice Portal • Voice Portal Management System • Media Processing Platform • Nuance Speech Server • Tomcat Application Server	 5.0.0.2.0104 5.0.0.0.4106 NSS05.0.2 Tomcat 5.5
Avaya Aura™ Communication Manager Access Element • Avaya S8730 Server	5.2.1 R015x.02.1.016.4
Avaya Aura™ Communication Manager Feature Server • Avaya S8300 Server	5.2.1 R015x.02.1.016.4
Avaya G650 Media Gateway • IPSI (TN2312BP) • C-LAN (TN799DP)	 TN2312BP HW14 FW040 TN799DP HW01 FW034

Component	Software Version
• IP Media Resource 320 (TN2602AP) • DS1 Interface (TN464F)	TN2602AP HW02 FW051 TN464F 000020
Avaya IP Telephones: • 9650 • 9630 • 9620	 FW: 2.0 FW:1.50 FW:1.5
Avaya SIP Phones • 9650	FW: 2.5.5.16
Avaya Digital Telephones (2420D)	-

1.4. Call Flow — SIP Trunk to Avaya Aura™ Communication Manager



Scenario: Customer calls VP for self service and selects option to talk to an Agent.

1. Customer calls VP application using either PSTN phone or internal Avaya telephone and is handled by Avaya Aura™ Communication Manager Access Element.
2. Avaya Aura™ Communication Manager routes the call to the Session Manger via SIP trunk.
3. Avaya Aura™ Session Manager routes the call to an MPP on Voice Portal System.
4. The MPP maps the DNIS (Dialed Number identification Service) number to a speech application and begins the self service workflow.
5. Customer decides to talk to an agent and selects the appropriate option from the workflow.

6. The MPP places an outbound call to Avaya Aura™ Session Manager as part of a SIP REFER method.
7. Session Manager forwards the REFER destination to the Avaya Aura™ Communication Manager.
8. The Avaya Aura™ Communication Manager accepts the REFER message and then delivers the call to an agent[2].

2. Configuring Avaya Aura™ Communication Manager Access Element

This section describes configuring Avaya Aura™ Communication Manager Access Element using Avaya Site Administrator. These instructions assume a CLAN and Media Processor board are already installed and configured on the Communication Manager Access Element. Some administration screens have been abbreviated for clarity.

- Verify System Capabilities and Communication Manager Licensing
- Administer IP codec set and network region
- Administer IP node names
- Administer IP interface
- Administer SIP trunk group and signaling group
- Administer route patterns
- Administer numbering plan
- Administer VDN for calls transferred from VP application

After completing these steps, the "**save translations**" command should be performed.

2.1. Verify System Capabilities and Licensing

This section describes the procedures to verify the correct system capabilities and licensing have been configured. If there is insufficient capacity or a required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

2.1.1. SIP Trunk Capacity Check

Issue the **display system-parameters customer-options** command to verify that an adequate number of SIP trunk members are administered for the system as shown below:

[2] Note that resulting communication path is from customer, through the G650, to Communication Manager, thereby freeing resources on Voice Portal.

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

IP PORT CAPACITIES                                               USED
                     Maximum Administered H.323 Trunks: 500   0
           Maximum Concurrently Registered IP Stations: 18000 4
             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: 100   0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 50    20
```

2.1.2. AAR/ARS Routing Check

Verify that **ARS** is enabled (on page 3 of system-parameters customer options).

```
display system-parameters customer-options                    Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n           Audible Message Waiting? n
        Access Security Gateway (ASG)? n               Authorization Codes? n
        Analog Trunk Incoming Call ID? n                         CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                           CAS Main? n
Answer Supervision by Call Classifier? n                  Change COR by FAC? n
                                   ARS? y  Computer Telephony Adjunct Links? y
                  ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                       DCS (Basic)? y
          ASAI Link Core Capabilities? y                  DCS Call Coverage? y
: 18000          0
```

2.1.3. Configure Trunk-to-Trunk Transfers

Use the "**change system-parameters features**" command to enable trunk-to-trunk transfers.

This feature is needed 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.

```
change system-parameters features                               Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
```

2.1.4. Configure Codec Type

Issue the **change ip-codec-set n** command where **n** is the next available number. Enter the following values:

- Enter "**G.711MU**" for type of Audio Codec

The value administered here will be used in Voice Portal configuration.

- Silence Suppression: Retain the default value "**n**".
- Frames Per Pkt: Enter "**2**".
- Packet Size (ms): Enter "**20**".
- Media Encryption: Enter the value based on the system requirement. For the sample configuration, "none" was used.

```
change ip-codec-set 1                                    Page   1 of   2
                           IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2: G.729             n          2        20
 3:

     Media Encryption
 1: none
```

2.2. Set IP Network Region

Using the **change ip-network-region 1** command, set the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio** fields to "**yes**". For the **Codec Set** enter the corresponding audio codec set configured in **Section 2.1**. Set the **Authoritative Domain** to the correct SIP domain for the configuration.

```
change ip-network-region 1                               Page   1 of  19
                                IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 16585
```

2.3. Add Node Names and IP Addresses

Using the **change node-names ip** command, add the node-name and IP for the CLANs and the Session Manager, if not already previously added. Note the node names of the CLANs which will later be used to configure the SIP trunks between the Avaya G650 and the Session Manager.

Node names for other SIP entities in the solution such as Voice Portal do not have to be administered on the Communication Manager Access Element since the Access Element does not directly connect to these entities.

*Note that these may have been already created and do not need to be re-created if the names are already present in the node-names list.

```
change node-names ip                                        Page   1 of   2
                                   IP NODE NAMES
    Name                IP Address
8730-1              10.80.111.11
8730-2              10.80.111.12
ASM1                10.80.100.24
ASM2                10.80.100.26
CLAN-1              10.80.111.16
CLAN-2              10.80.111.17
```

2.4. Configure SIP Signaling Group and Trunk Group

2.4.1. Create a Signaling Group for SIP Trunk to Avaya Aura™ Session Manager

In the sample configuration, trunk group "20" and signaling group "20" were used to connect to Avaya Aura™ Session Manager. Issue the **add signaling-group n** command, where "n" is an available signaling group number to create a SIP trunk to the Session Manager. Fill in the indicated fields as shown below. Default values can be used for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tcp[3]"
- **Near-end Node Name:** C-LAN node name from **Section 2.3**.
- **Far-end Node Name:** Session Manager node name from **Section 2.3**.
- **Near-end Listen Port:** "5060"
- **Far-end Listen Port:** "5060"
- **Far-end Domain:** this field should be left blank[4]
- **DTMF over IP:** "rtp-payload"
- **Session Establishment Timer:** "3" [5]

```
display signaling-group 20                                      Page   1 of   1
                                SIGNALING GROUP

Group Number: 20             Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n
     IP Video? n

    Near-end Node Name: CLAN-2                Far-end Node Name: ASM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region:
Far-end Domain:
                                            Bypass If IP Threshold Exceeded? n

       DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

[3] TCP was used for the sample configuration. However, TLS would typically be used in production environments.
[4] To support the ability for the Voice Portal application to transfer calls back to agents logged into Communication Manager, the name of the far-end domain should be left blank.
[5] If agents are not expected to answer the transferred call from the VP application within 3 minutes, this value may need to be increased.

2.4.2. Add a SIP Trunk Group to Connect to Avaya Aura™ Session Manager

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

*Note that the number of members determines how many simulataneous calls can be processed by the trunk through Session Manager.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Signaling Group:** The number of the signaling group added in **Section 2.4.1**
- **Number of Members:** The number of members in the SIP trunk to be allocated to calls routed to Session Manager (must be within the limits of the total number of trunks configured in **Section 2.1.1**).

One the add command is completed, trunk members will be automatically generated based on the value in the **Number of Members** field.

```
add trunk-group 20                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 20                   Group Type: sip           CDR Reports: y
  Group Name: SIP to ASM1 for VP app      COR: 1        TN: 1        TAC: #20
Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 20
                                                    Number of Members: 10
```

On page 2, set the **Preferred Minimum Session Refresh Interval** to 1200. Note: to avoid extra SIP messages, all SIP trunks connected to Session Manager should be configured with a minimum value of 1200.

```
add trunk-group 20                                          Page   2 of  21
                                  Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                       Preferred Minimum Session Refresh Interval(sec): 1200
```

On page 3, set **Numbering Format** to be *public*. Use default values for all other fields.

```
add trunk-group 20                                          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



 Show ANSWERED BY on Display? y
```

2.5. Configure Route Pattern

Use the "**change route-pattern n**" command, when n is an available number to define a route pattern for routing calls to the VP application over the SIP trunk group defined in **Section 2.4.2.** In the sample configuration, route pattern 20 was created as shown below:

```
change route-pattern 20                                    Page   1 of  3

                    Pattern Number: 20  Pattern Name: to VP via 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: 20   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
```

2.6. Administer Numbering Plan

2.6.1. Administer dial plan

Use the "**change dialplan analysis**" command, to define any extension numbers associated with the VP application.

In the sample configuration, VDN "522-1000" is used for agents supporting the VP application.

```
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
     0         1    attd
     1         2    dac
     2         6    aar
     400       7    ext
     500       5    ext
     522       7    ext
     666       7    ext
```

2.6.2. Administer ARS analysis

This section provides the configuration of the Automatic Route Selection (ARS) pattern used in the sample configuration for routing “522-2000” calls to the VP application.

Note that other methods of routing may be used.

Use the “**change ars analysis n**” command where **n** is a valid number defined in the dialplan to add an entry for routing the dialed number of “522-2000” to Voice Portal application over the route pattern defined in **Section 2.5.**

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    222                   7    7     1        hnpa         n
    255                   7    7     1        hnpa         n
    303                   10   10    10       hnpa         n
    333                   7    7     999      hnpa         n
    4                     7    7     1        hnpa         n
    411                   3    3     deny     svcl         n
    5                     7    7     999      hnpa         n
    511                   7    7     999      hnpa         n
    522                   7    7     20       hnpa         n
    555                   7    7     deny     hnpa         n
    611                   3    3     1        svcl         n
    666                   7    7     10       hnpa         n
    7                     7    7     2        hnpa         n
    8                     7    7     999      hnpa         n
    811                   3    3     1        svcl         n
```

2.7. Administer Hunt-Group, VDN and Vector for calls transferred from VP application

2.7.1. Define Hunt-Group

Use “**add hunt-group x**” command where **x** is an available number to define a hunt-group (skill) for the agents who will receive calls from the VP application.

For the sample configuration, skill 522 was created. Since the Voice Portal application can support multiple, simultaneous calls, set the Queue and Queue Limit fields on page 1 to support queuing.

```
add hunt-group 522                                          Page   1 of  3
                                  HUNT GROUP

            Group Number: 522                               ACD? y
              Group Name: VP Agents                       Queue? y
          Group Extension: 522-2222                      Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold: 100  Port:
  Time Warning Threshold: 100  Port:
```

On page 2, optionally set **Measured** field to “internal” to enable BCMS to monitor agent and queue status. If another reporting product such as Avaya Call Management System is available, this field should be set to either “external” or “both”.

```
add hunt-group 522                                          Page   2 of  3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n        Service Level Target (% in sec): 80 in 20
                 Measured: internal
     Supervisor Extension:
```

Use “**add vdn xxx**” command where **xxx** is a valid extension number to define a VDN as the number the VP application will use to transfer calls to an agent.

For the sample configuration, VDN “522-1000” was created. Optionally, set Measured field to “internal” to enable BCMS to monitor agent and queue status.

```
add vdn 5221000                                             Page   1 of  2
                                                      VECTOR DIRECTORY NUMBER

                              Extension: 522-1000
                                  Name*: Sample VP application
                            Destination: Vector Number        1000

                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: internal
        Acceptable Service Level (sec): 20
```

2.7.2. Administer Vector

Use "**change vector xxx"** command where **xxx** is an available vector to define a vector for processing calls from the VP application.
For the sample configuration, vector 1000 was created. Since the Voice Portal application can support multiple, simultaneous calls, define the first step in the vector to queue to the skill (hunt-group) defined in **Section 2.7.1.**

```
change vector 1000                                          Page   1 of  6
                                  CALL VECTOR

    Number: 1000             Name: Sample VP app
Multimedia? n                                Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? n  ASAI Routing? y
 Prompting? n   LAI? y  G3V4 Adv Route? y   CINFO? n   BSR? y   Holidays? n
 Variables? n   3.0 Enhanced? N

01 queue-to     skill 522  pri h
```

3. Configure the Avaya Aura™ Communication Manager Feature Server

Assuming the Avaya Aura™ Communication Manager Feature Server was already installed and configured, this section provides the procedures for any additional configuration needed when an Avaya Aura™ Communication Manager Feature Server is part of the Voice Portal First solution.

3.1. Enable Private Numbering

SIP Users registered to Session Manager need to be added to either the private or public numbering table on the Communication Manager Feature Server. For the sample configuration, private numbering was used.

Use the "**change system-parameters customer-options**" command to verify that Private Networking is enabled as shown below:

```
display system-parameters customer-options              Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? y            Station and Trunk MSP? y
Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                     Multiple Locations? y
                                            System Management Data Transfer? n
         Personal Station Access (PSA)? y              Tenant Partitioning? n
                         PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                   Port Network Support? n              Time of Day Routing? n
                        Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                       Uniform Dialing Plan? y
                     Private Networking? y    Usage Allocation Enhancements? y
               Processor and System MSP? y
                     Processor Ethernet? y              Wideband Switching? n
```

3.2. Configure Private Numbering Plan

To enable SIP endpoints to dial extensions defined in the Communication Manager Access Element, use the "**change private-numbering x"** command, where x is the number used to identify the private number plan to create entries for any extension numbers assigned in the Communication Manager Access Element. For the sample configuration, extension numbers starting with 522-XXXX will be used to access the Voice Portal application.

- **Ext Len:** Enter the extension length allowed by the dial plan
- **Ext Code:** Enter leading digit (s) from extension number
- **Trunk Grp:** Enter the SIP Trunk Group number for the SIP trunk between the Feature Server and Session Manager
- **Private Prefix:** Leave blank unless an enterprise canonical numbering scheme is defined in Session Manager. If so, enter the appropriate prefix.

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

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 7  5              10                          7     Total Administered: 2
 7  6              10                          7        Maximum Entries: 540
```

Note: After a change on Communication Manager Feature Server which alters the dial plan, synchronization between Communication Manager Feature Server and Session Manager needs to be completed and SIP phones must be re-registered. To request an on demand synchronization, log into the System Manager console and use the **Synchronize CM Data** feature under the Communication System Management menu.

4. Configure Avaya Aura™ Session Manager

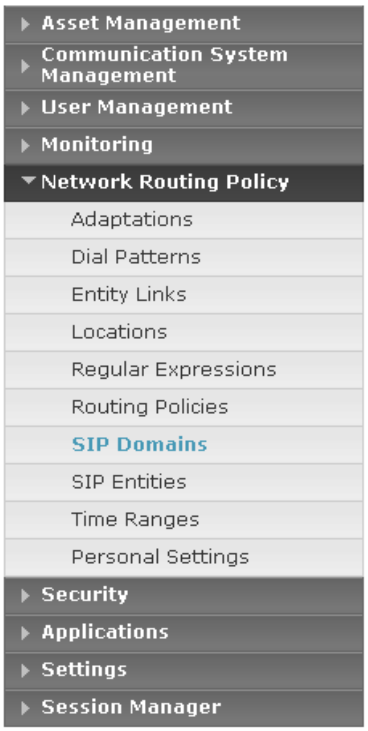
This section provides the procedures for configuring Avaya Aura™ Session Manager. The procedures include adding the following items:

- Administer SIP domain
- Define Logical/physical Locations that can be occupied by SIP Entities
- For each SIP entity in the sample configuration:
    - Define SIP Entity
    - Define Entity Links, which define the SIP trunk parameters used by Avaya Aura ™ Session Manager when routing calls to/from SIP Entities
    - Define Routing Policies, which control call routing between the SIP Entities
    - Define Dial Patterns, which govern to which SIP Entity a call is routed

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura™ System Manager, using the URL “http://<ip-address>/SMGR”, where “<ip-address>” is the IP address of Avaya Aura™ System Manager.

Log in with the appropriate credentials and accept the Copyright Notice.

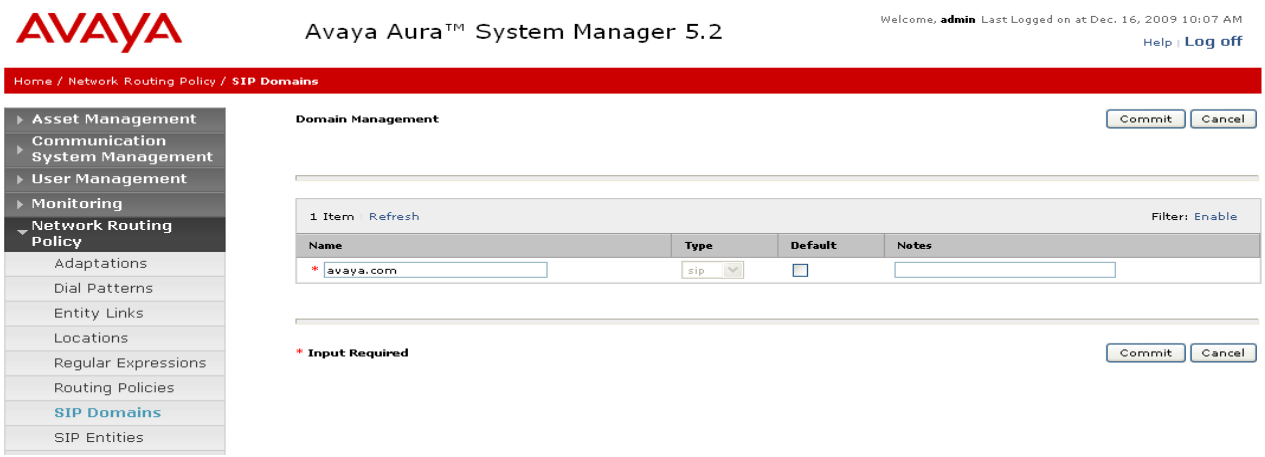
Expand the **Network Routing Policy** Link on the left side of Navigation Menu. Select a specific item such as SIP Domains. When the specific item is selected, the color of the item will change to blue as shown below:



4.1. Administer SIP Domains

- Expand Network Routing Policy and select **SIP Domains**.
    - Click **New**
    - In the *General* Section, under *Name* add a descriptive name. Under *Notes* add a brief description.
    - Click **Commit** to save.

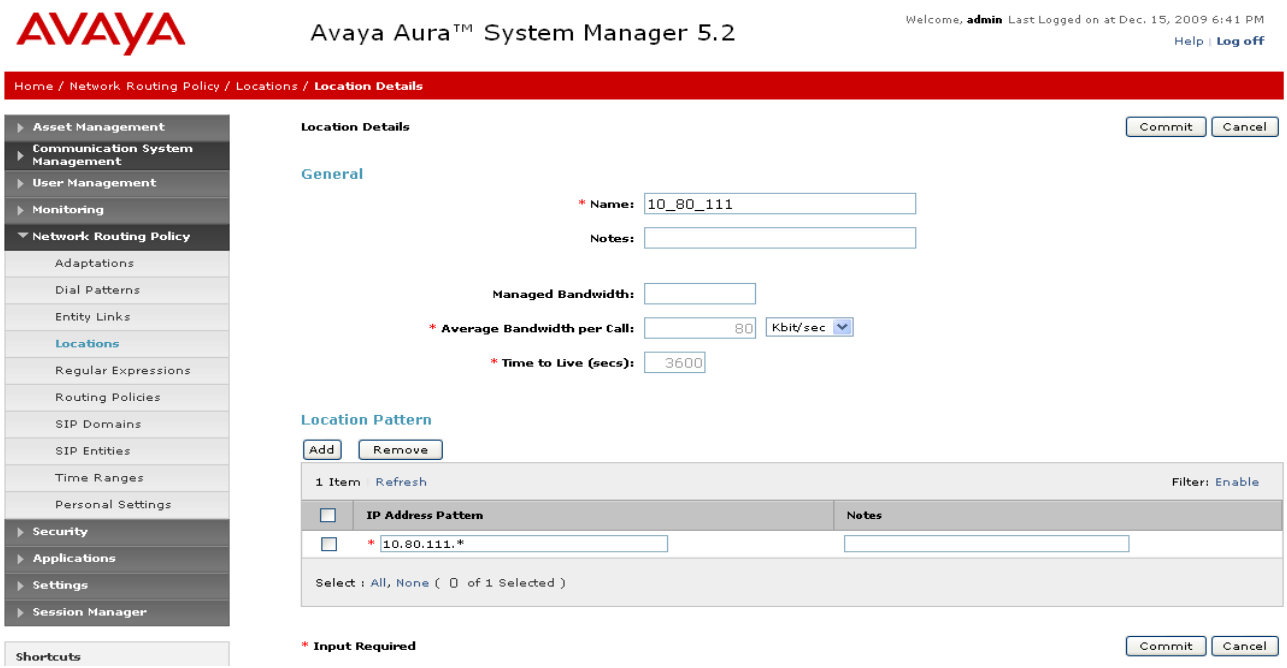
The screen below shows the information for sample configuration.



4.2. Define Locations

- Expand Network Routing Policy and select **Locations.** Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.
    - Click **New**
    - In the *General* Section, under *Name* add a descriptive name.
    - Under *Notes* add a brief description.
    - In the *Location Pattern* Section, under IP Address Pattern enter pattern used to logically identify the location. Under *Notes* add a brief description.
    - Click **Commit** to save.

The screen below shows the information for Communication Manager Access Element in the sample configuration.

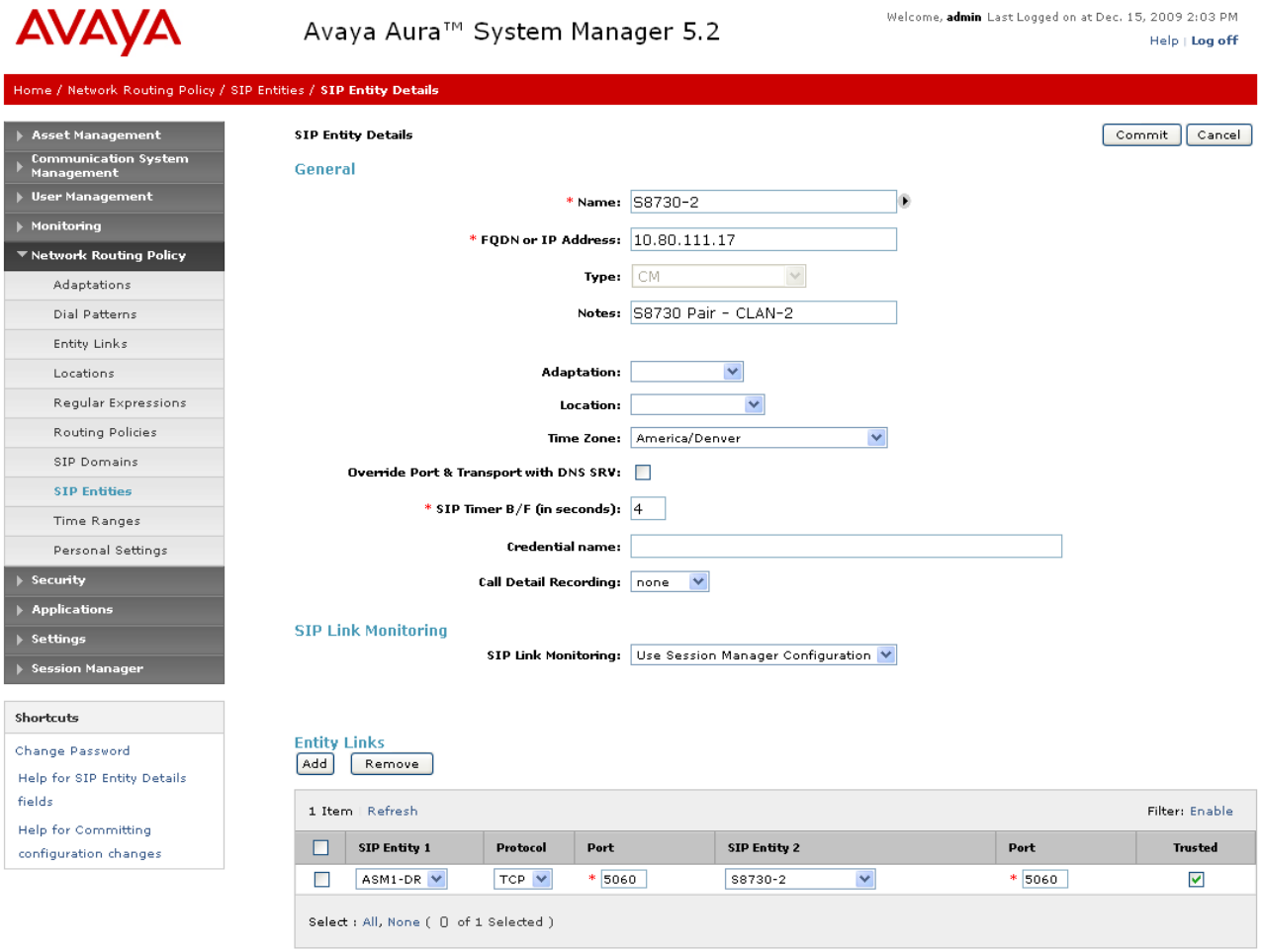


4.3. Add Avaya Aura™ Communication Manager Access Element

4.3.1. Define SIP Entity for the Communication Manager Access Element

- Expand Network Routing Policy
  - Select SIP Entities
    - Click **New**
    - In the *General* Section, under *Name* add an identifier for the Communication Manager. Under *FQDN or IP Address* enter the IP Address of the Communication Manager. Under *Type* select CM. Under *Notes* add a brief description.
    - *Location:* From the drop-down select the Location added in **Section 4.2**. Note: since location-based routing was not used in the sample configuration, selecting a value for location field is optional.
    - Click **Commit** to save.

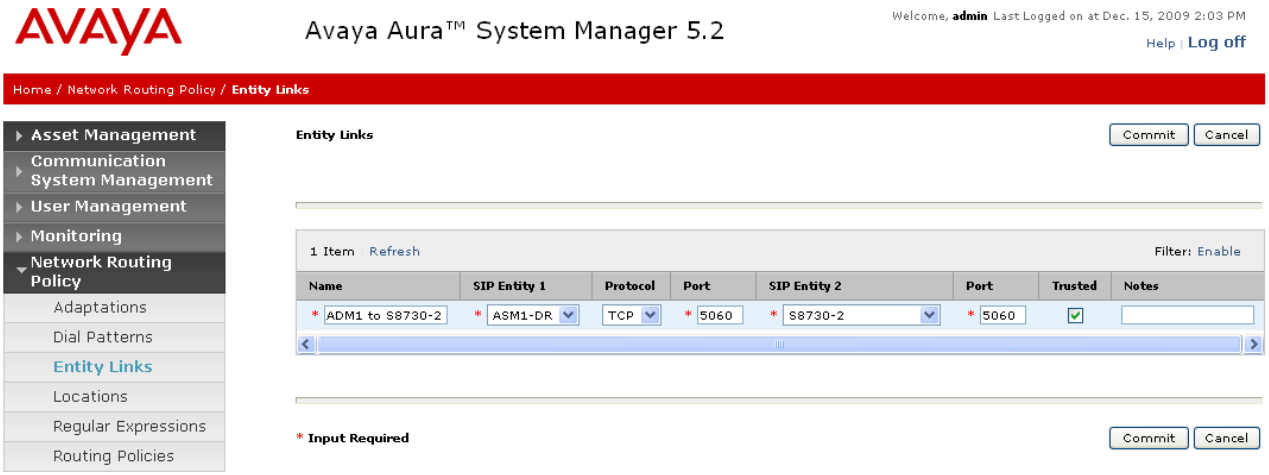
In the sample configuration, a SIP entity was defined for each of the CLAN boards in the Avaya G650 Media Gateway. The following screen shows addition of one of the SIP entities for the Communication Manager Access Element. The IP address used is that of the C-LAN board in the Avaya G650 Media Gateway.



4.3.2. Define an Entity Link for Communication Manager Access Element

- Network Routing Policy
  - o Entity Links
    - Click **New**
    - Under *Name,* enter an identifier for the Communication Manager Access Element.
    - Under *SIP Entity 1* drop-down select the appropriate Session Manager. Under *Port* dropdown select the correct port for the Session Manager.
    - Under *SIP Entity 2* drop-down select the SIP Entity added in **Section 4.3.1** for the Communication Manager Access Element. Under *Port* dropdown select the correct port for the Communication Manager. Select it as a *Trusted* host. Under *Protocol* dropdown select the required protocol.
    - Under *Notes* add a brief description.
    - Click **Commit** to save.

The following screen shows the entity link defined for the Communication Manager Access Element.



4.3.3. Define Routing Policy for Communication Manager Access Element

- Network Routing Policy
  - o Routing Policies
    - Click **New**
    - In the 'General' section, under Name add an identifier to define the routing policy for the Communication Manager. Under *Notes* add a brief description.
    - In the 'SIP Entity as Destination' section, click on **Select**.
    - The SIP Entity List page opens.
      Select one of the SIP entries for the Communication Manager added in **Section 4.3.1** and click on **Select**
    - The selected SIP Entity displays on the Routing Policy Details page.
    - Click on **Commit** to save.

Shown below is the updated screen for the sample configuration.

Home / Network Routing Policy / Routing Policies / **Routing Policy Details**

- Asset Management
- Communication System Management
- User Management
- Monitoring
- 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

Shortcuts

- Change Password
- Help for Routing Policy Details fields
- Help for SIP Entity List
- Help for Time Range List
- Help for Pattern List
- Help for Regular Expressions List
- Help for Committing configuration changes

Routing Policy Details Commit Cancel

General

* **Name:** to S8730 CM

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
S8730-1	10.80.111.16	CM	S8730 Pair CLAN-1

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking 1	Name 2	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	✓	✓	✓	✓	✓	✓	✓	00:00	23:59	Time Range 24/7

Select : All, None (0 of 1 Selected)

Dial Patterns

Add Remove

4 Items Refresh Filter: Enable

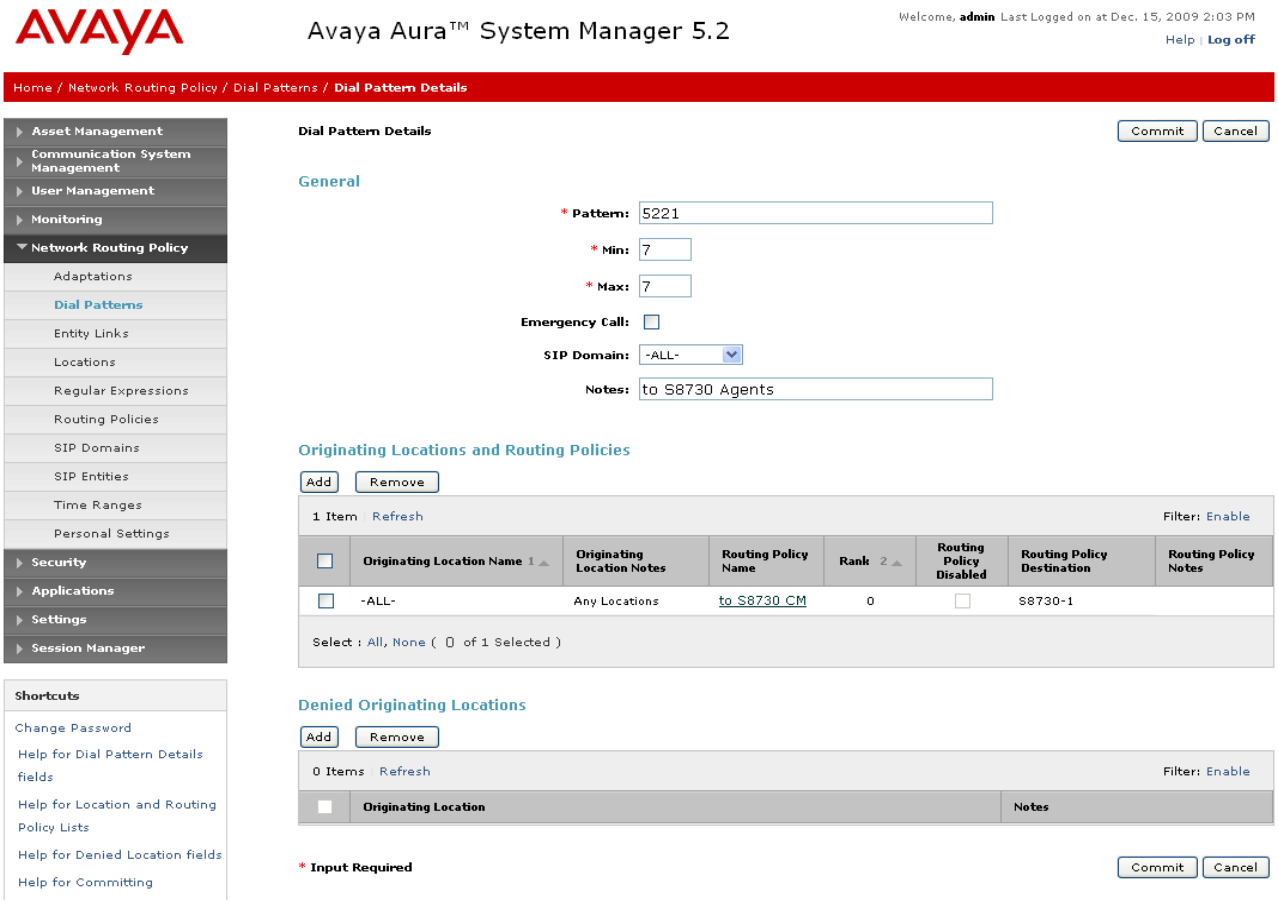
Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
400	7	7	☐	-ALL-	-ALL-	
5221	7	7	☐	-ALL-	-ALL-	to S8730 Agents
5223	7	7	☐	-ALL-	-ALL-	direct call to VP VDN on S8730
6664	7	7	☐	-ALL-	-ALL-	to S8730 CM

Select : All, None (0 of 4 Selected)

4.3.4. Define Dial Plan for calls to Communication Manager Access Element

- Expand Network Routing Policy
  - Dial Patterns
    - Click **New**
    - In the 'General' section, under *Pattern* add the number that the Voice Portal will dial-out to reach an agent on Communication Manager. Under *Min* enter the minimum number digits that must be dialed. Under *Max* enter the maximum number digits that may be dialed.
    - Under SIP Domain drop-down, select the SIP Domain added in **Section 4.1** or select "All" if the system can accept incoming call from all SIP domains
    - Under *Notes* add a brief description.
    - In the 'Originating Locations and Routing Policies' section click on **Add**
      - The 'Locations and Routing Policy List' page opens.
      - Under Locations, select the desired location.
    - Under Routing Policies, select the one defined for Avaya Communication Manager in **Section 4.3.3** and click on **Select**.

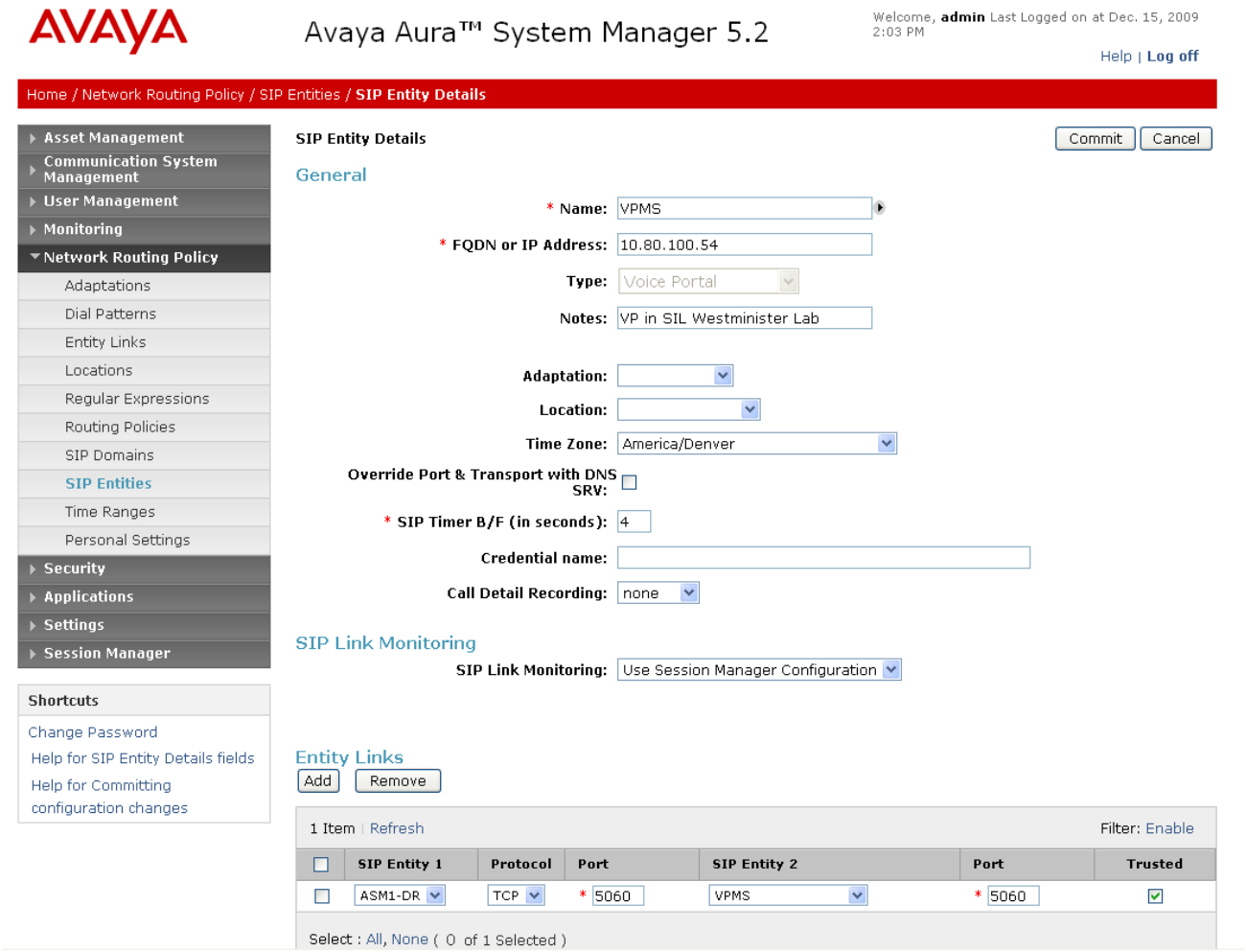
Shown below is the updated screen for the sample configuration.



4.4. Add Voice Portal System

4.4.1. Define SIP Entity for Voice Portal

- Expand Network Routing Policy
  - o Select SIP Entities
    - Click **New**
    - In the *General* Section, under *Name* add an identifier for the Voice Portal. Under *FQDN or IP Address* enter the Host Name or IP address of the Voice Portal server[6]. Under *Type* select VP. Under *Notes* add a brief description.
    - Click **Commit** to save.

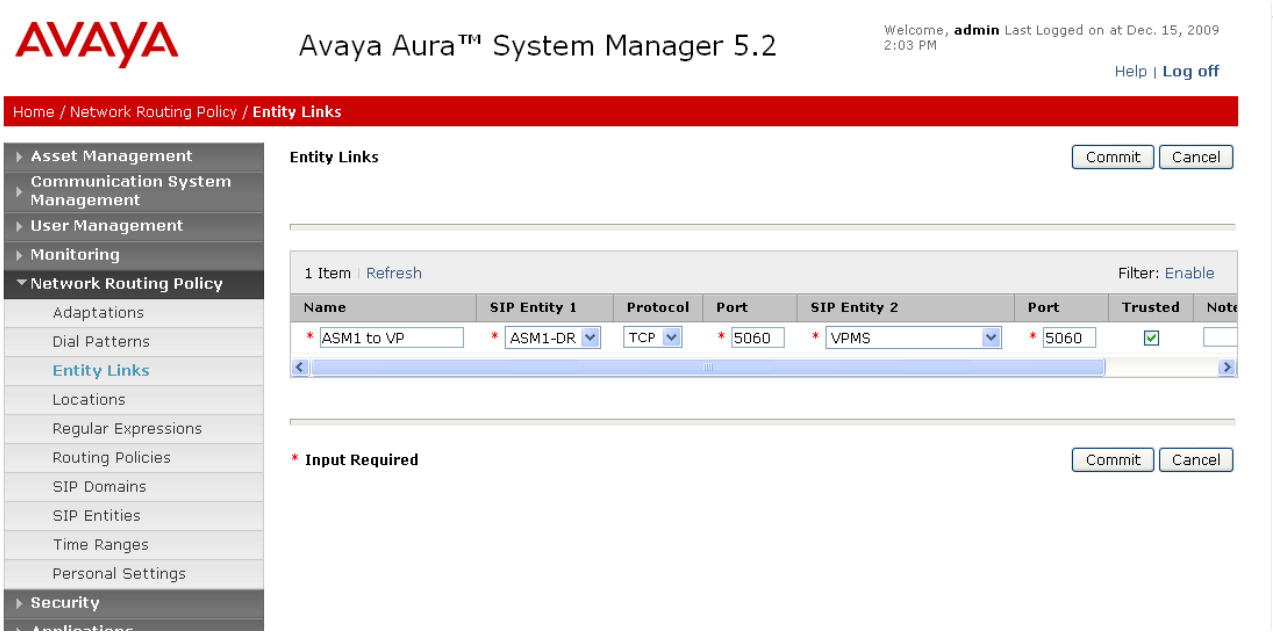


[6] Note: in the sample configuration, the MPP is running on the same server as the VPMS. In configurations with multiple MPPs, a SIP entity would need to be defined for each MPP.

4.4.2. Define the Entity Links for Voice Portal

- Expand Network Routing Policy
  - o Select Entity Links
    - Click **New**
    - Under *Name* enter an identifier for the Avaya Voice Portal.
    - Under *SIP Entity 1* drop-down select the appropriate Session Manager. Under *Port* dropdown select the correct port for the Session Manager.
    - Under *SIP Entity 2* drop-down select the SIP Entity added in **Section 4.4.1** for the Avaya Voice Portal. Under *Port* drop-down select the correct port for the Avaya Voice Portal. Select it as a *Trusted* host. Under *Protocol* dropdown select the required protocol. Under *Notes* add a brief description.
    - Click **Commit** to save.

Shown below is the updated screen for the sample configuration.

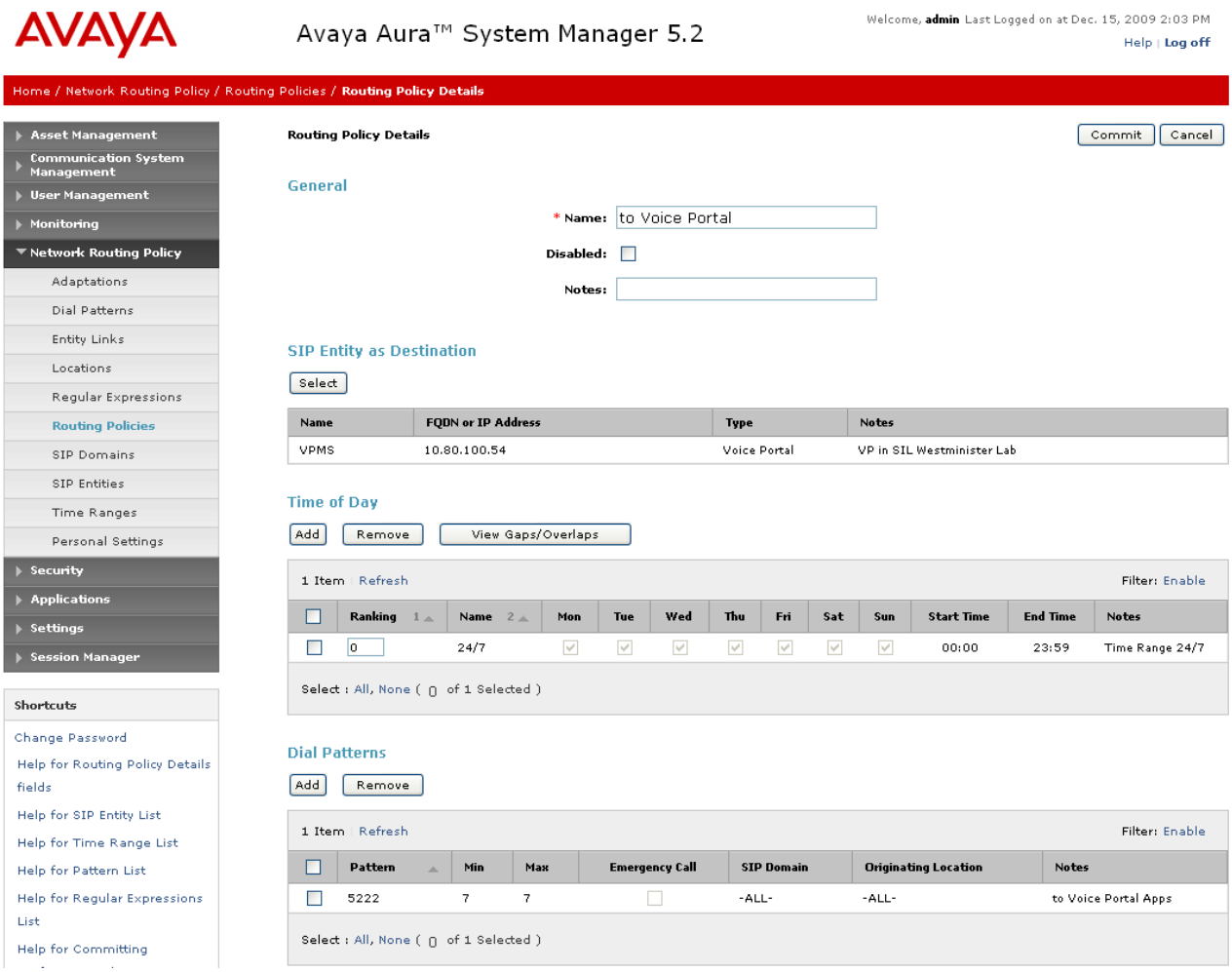


4.4.3. Define Routing Policies for Voice Portal

- Network Routing Policy
  - o Routing Policies
    - Click **New**

- In the 'General' section, under Name add an identifier to define the routing policy for the Avaya Voice Portal. Under *Notes* add a brief description.
- In the 'SIP Entity as Destination' section, click on **Select**
- The SIP Entity List page opens.
  Select the entry of the Avaya Voice Portal added in the above steps, and click **Select.**
- The selected SIP Entity displays on the Routing Policy Details page.
- Click on **Commit** to save.

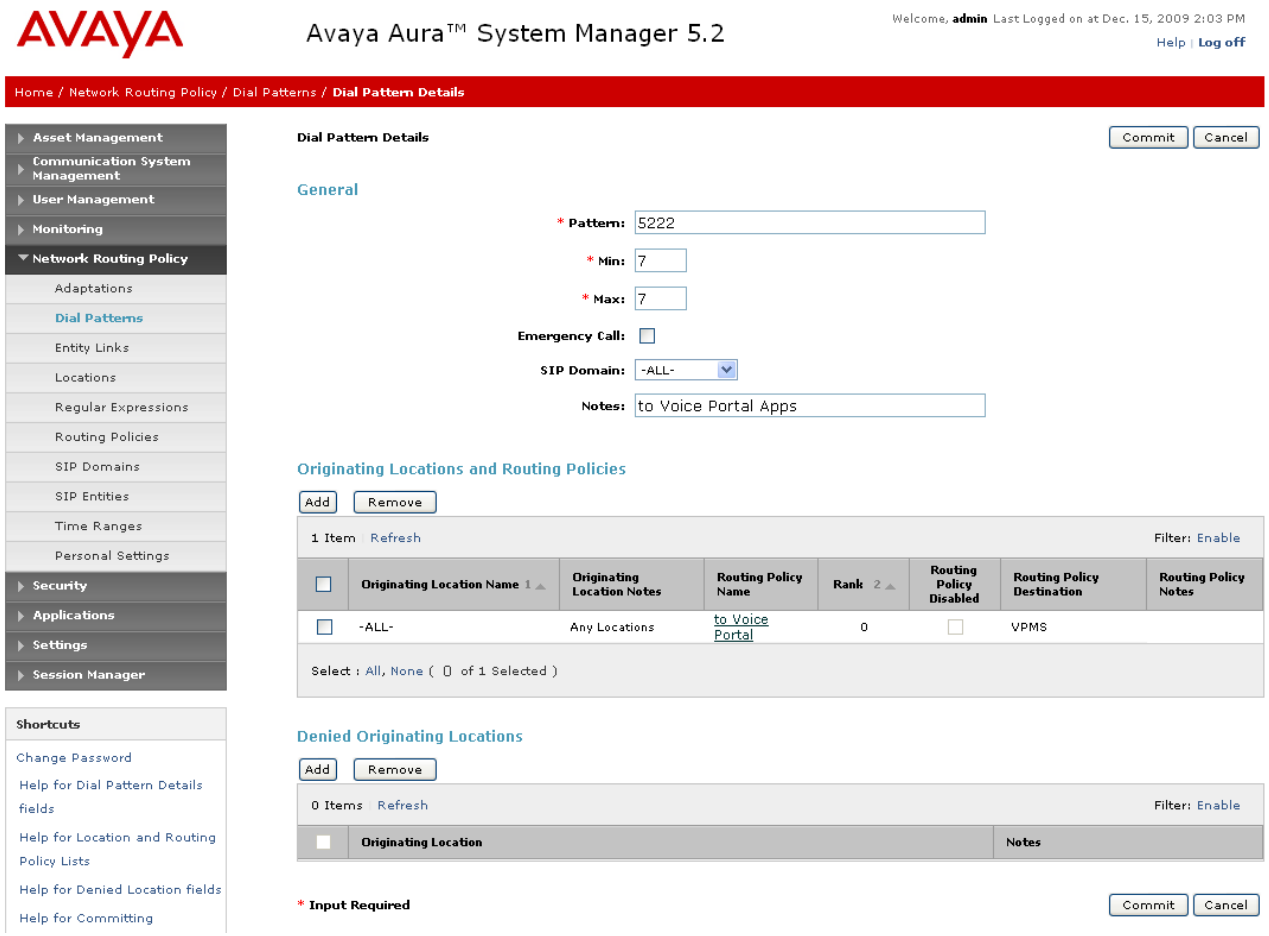
Shown below is the updated screen for the sample configuration.



4.4.4. Define Dial Plan to Route Calls to Voice Portal

- Expand Network Routing Policy
  - Select Dial Patterns
    - Click **New**
    - In the ‘General’ section under Pattern, add the dialed string of the number associated with the Voice Portal application. Under Min, enter the minimum number digits that must be dialed. Under Max, enter the maximum number digits that may be dialed.
    - Under SIP Domain drop-down, select the appropriate SIP Domain.
    - Under *Notes* add a brief description.
    - In the ‘Originating Locations and Routing Policies’ section click on **Add**
      - The ‘Locations and Routing Policy List’ page opens.
      - Under Locations, select the desired location or select the ALL options
      - Under Routing Policies select the one defined for the Voice Portal in **Section 4.4.3** and click on **Select**.

Shown below is the updated screen for the sample configuration.

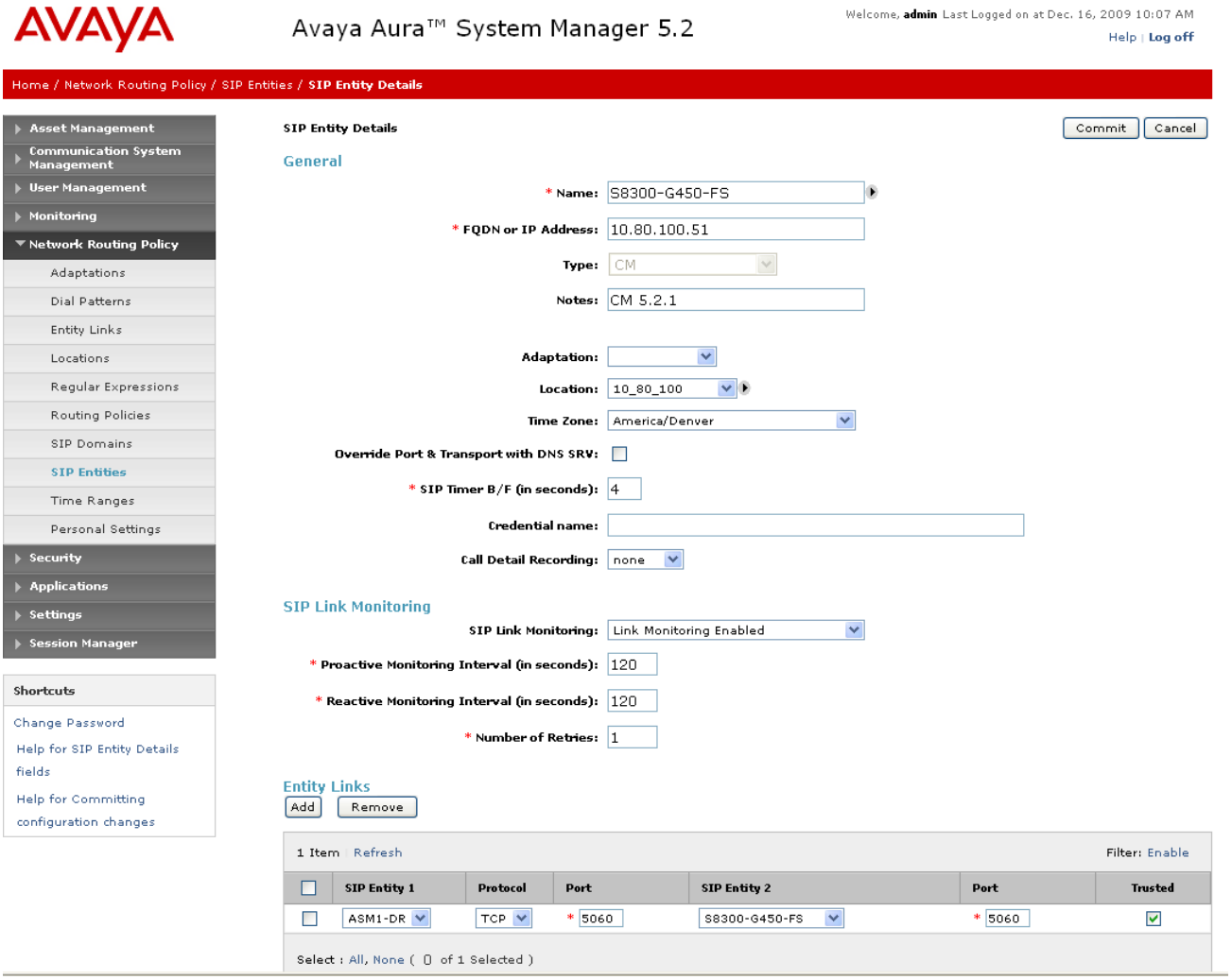


4.5. Administration of Avaya Aura™ Communication Manager Feature Server

Detailed administration of Communication Manager Feature Server and the endpoint telephones will not be described (see the appropriate documentation listed in **Section 9**). The following section captures relevant screens for Communication Manager Feature Server applicable for these Application Notes.

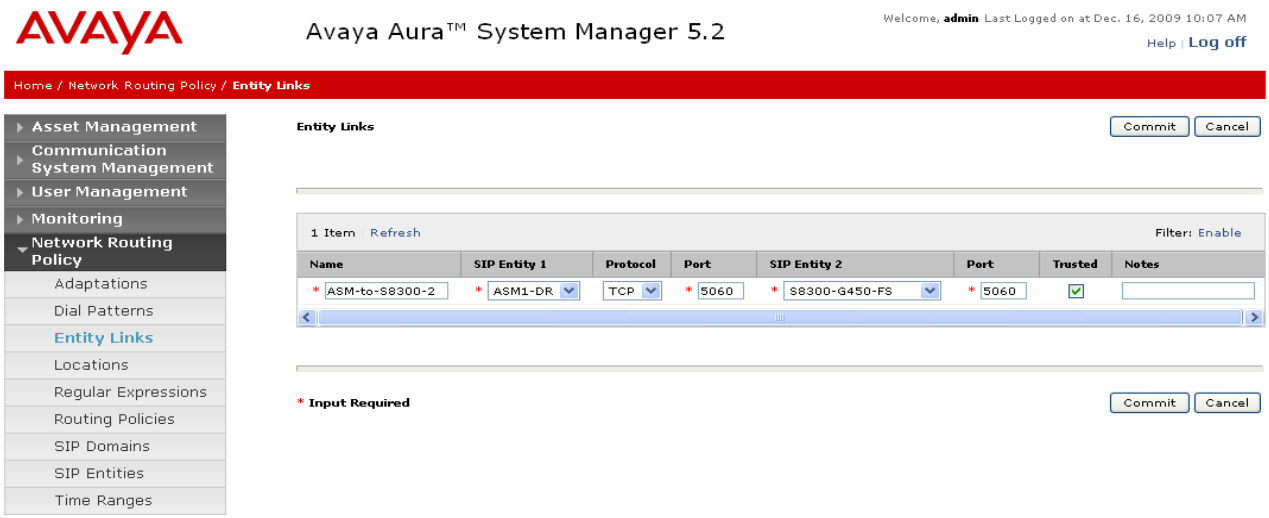
4.5.1. Define SIP Entity

The following screen shows addition of Communication Manager Feature Server. The IP address used is that of the S8300C server.



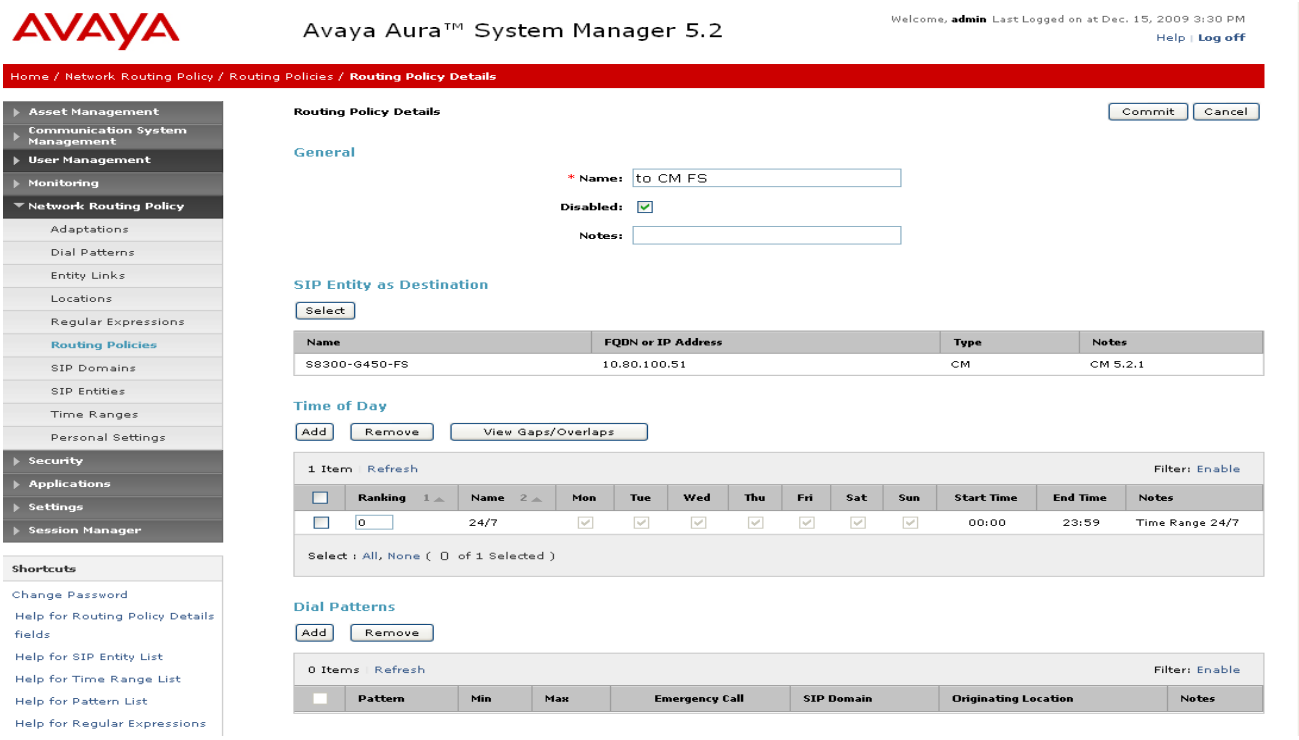
4.5.2. Define the Entity Link

The following screen shows the entity link defined for the Communication Manager Feature Server.



4.5.3. Define the Routing Policy

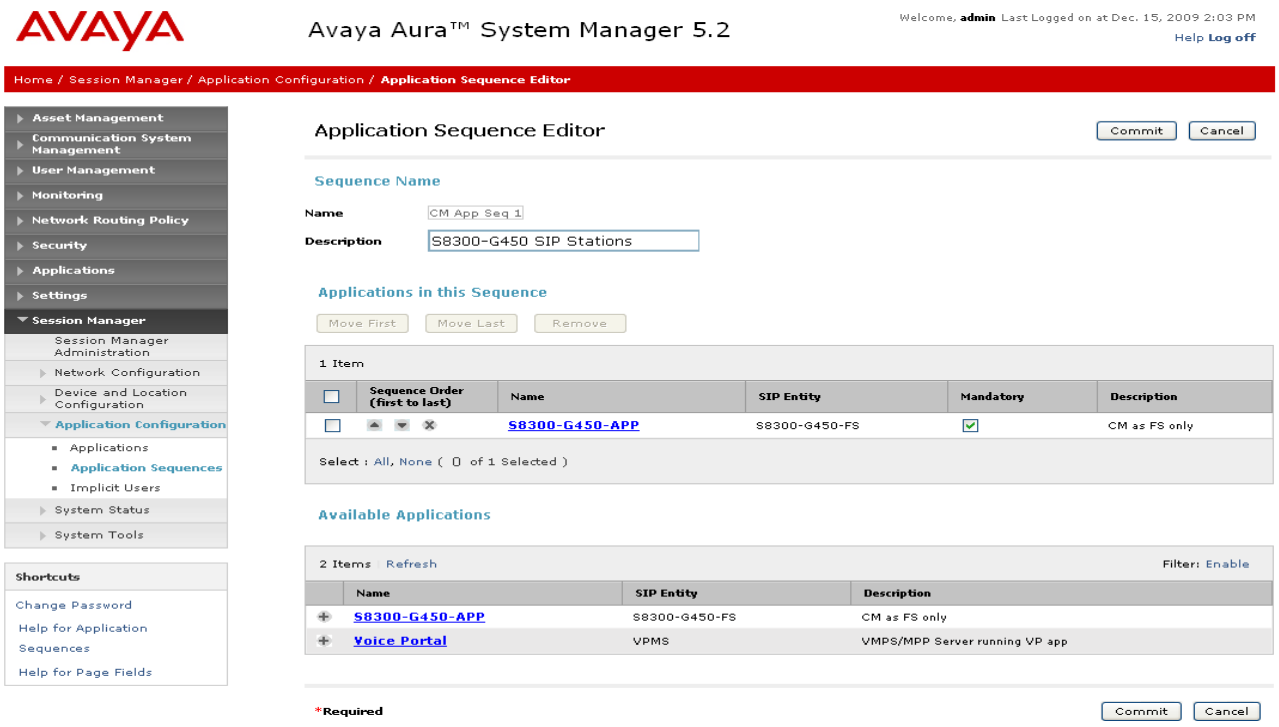
Since the SIP users are registered on Session Manager, the routing policy defined for the Communication Manager Feature Server does not need to include any dial patterns as shown below:



4.5.4. Define Application Sequence

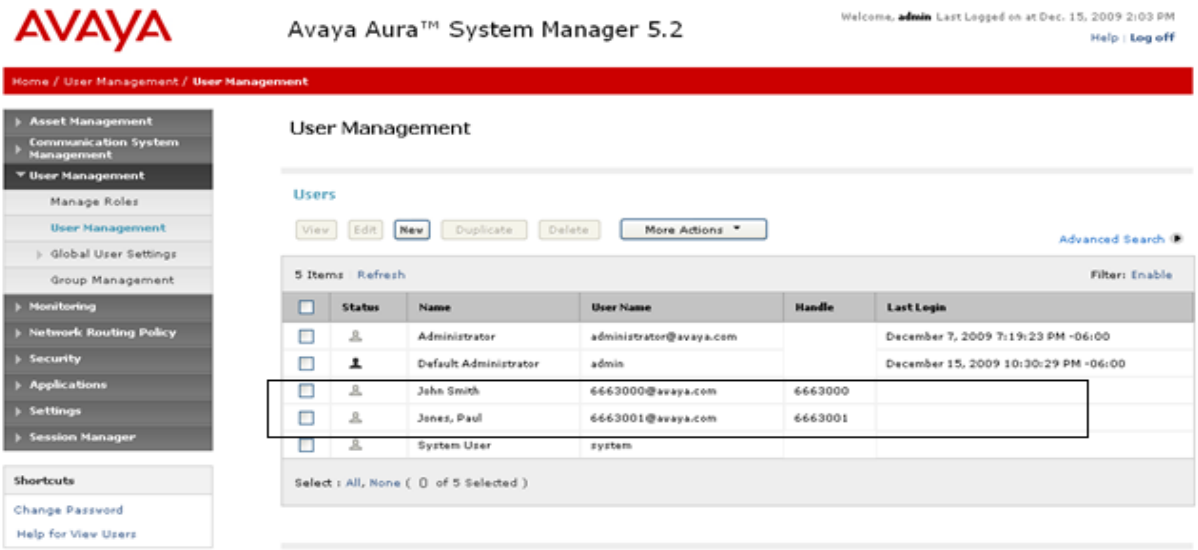
First, verify an application has been defined for the Communication Manager Feature Server as shown below:

Second, define an application sequence for the Communication Manager Feature Server as shown below:

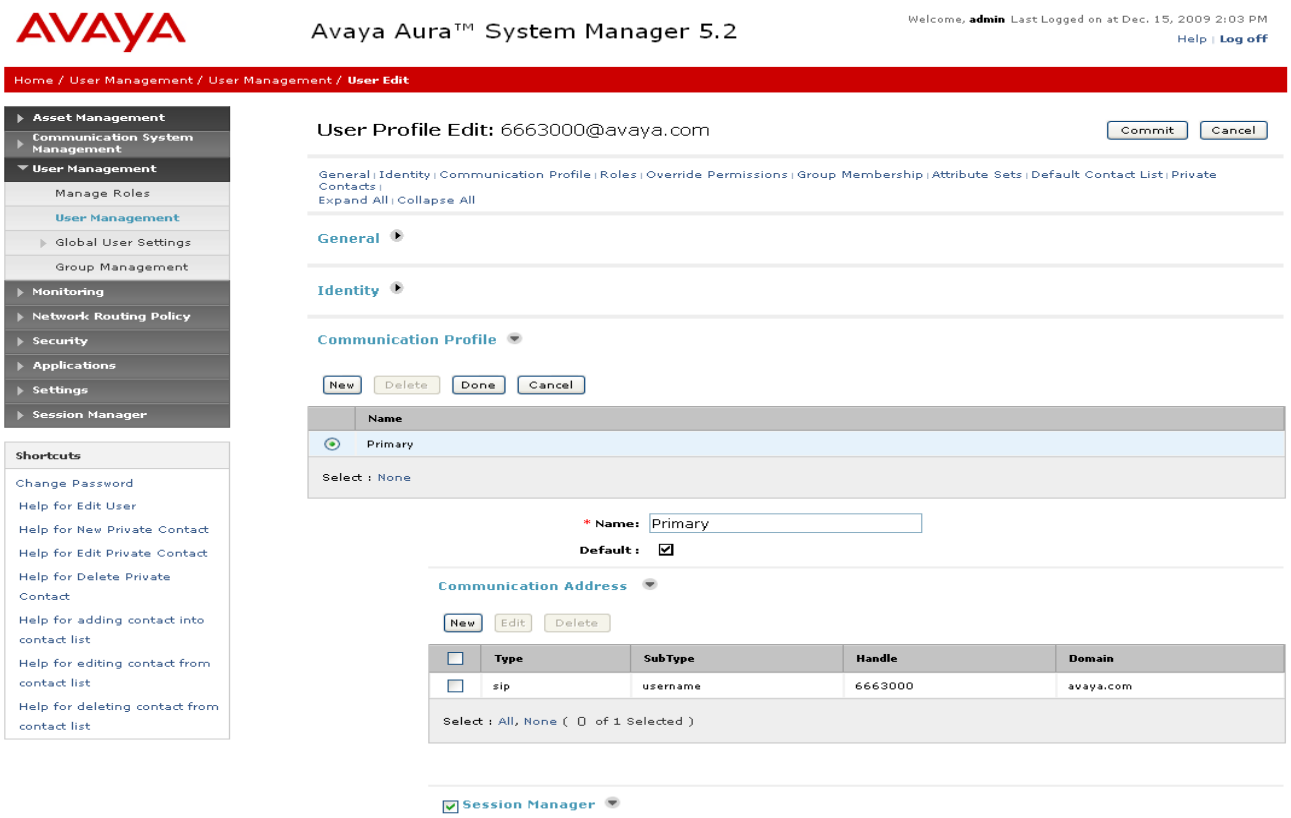


4.5.5. Verify Registrations of SIP Endpoints

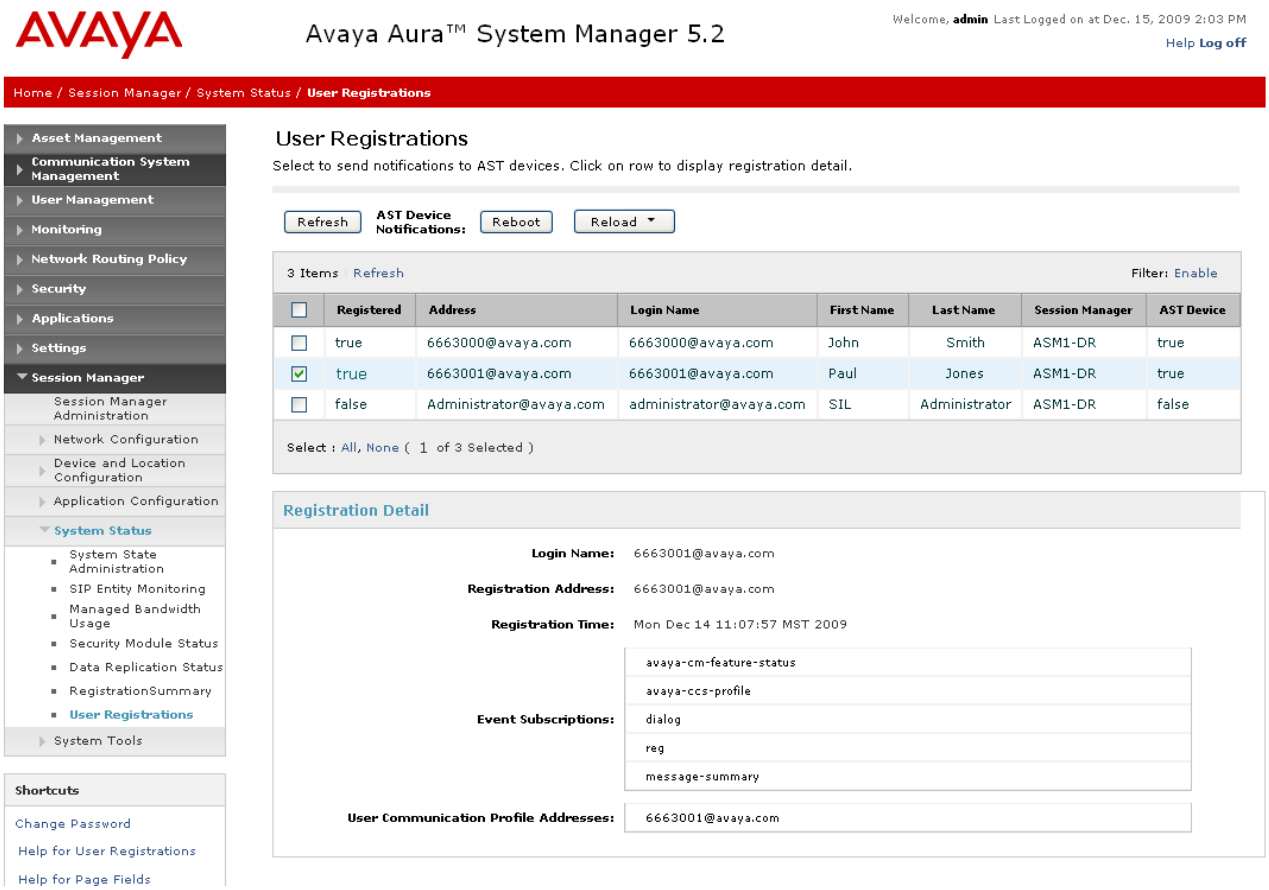
Verify SIP users have been created in the Session Manager. In the sample configuration, two SIP users were created as shown in the highlighted area below:



Verify the application sequence defined in **Section 4.5.4** is assigned to the SIP users by assigning the appropriate SIP communication profile as shown below:



Verify the SIP endpoints have successfully registered with the Session Manager as shown below:



5. Configure the Voice Portal

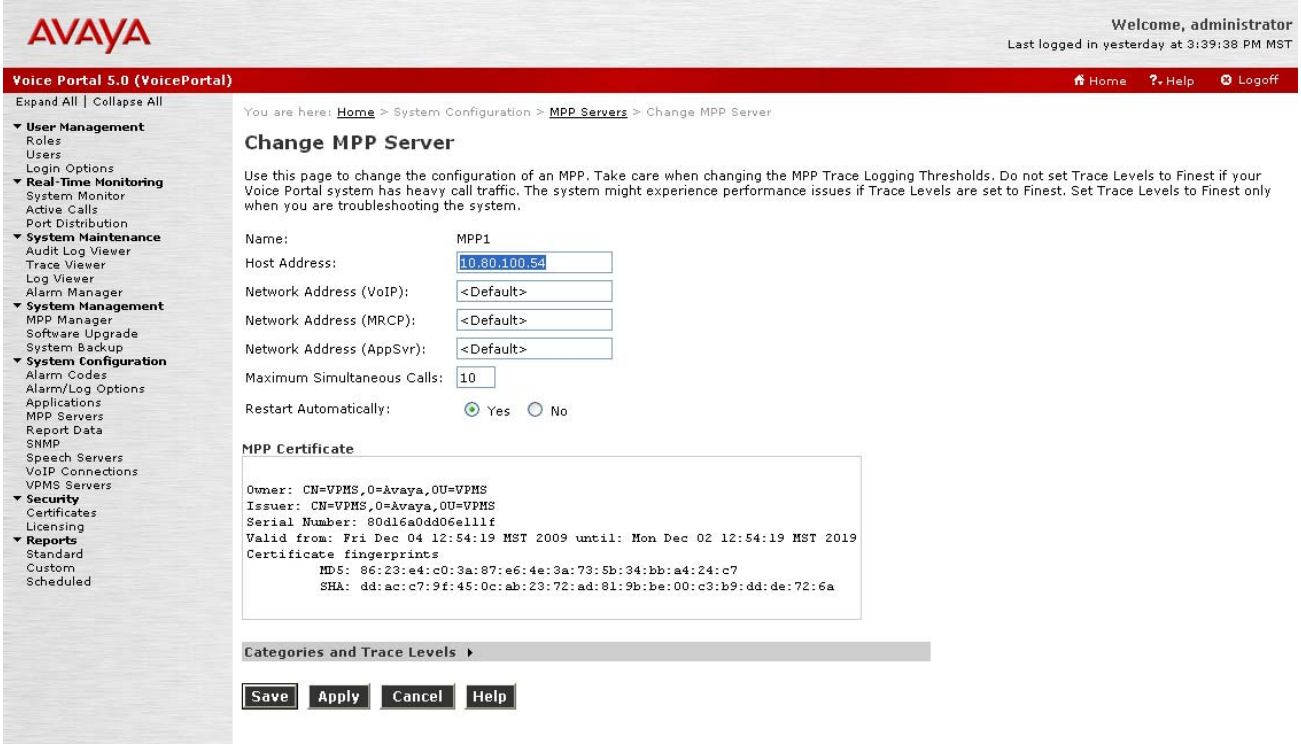
Log in to the Web Administration page of Voice Portal with Administrative rights.

5.1. System Configuration

5.1.1. MPP Servers

Add the installed MPP with details of the appropriate IP addresses, maximum simultaneous calls, etc. For the sample configuration, the MPP server is located at IP address 10.80.100.54.

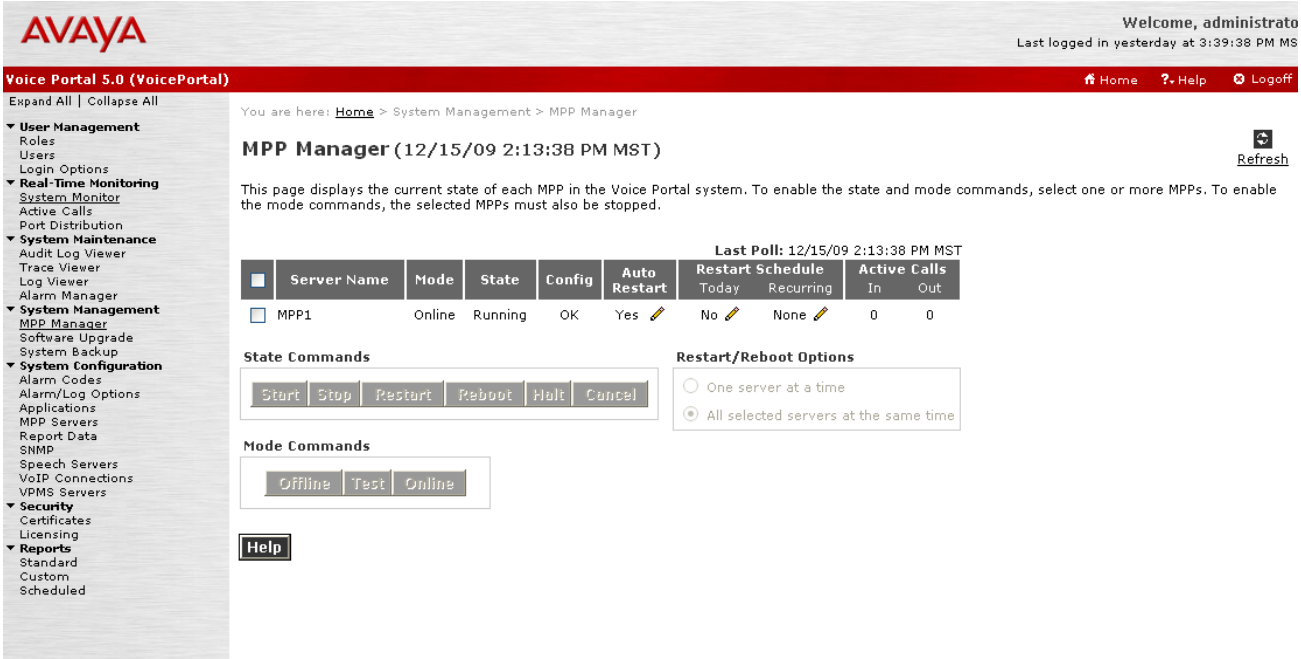
Refer to Voice Portal documentation for additional details in **Section 9**.



5.1.1. Verify MPP is in Service

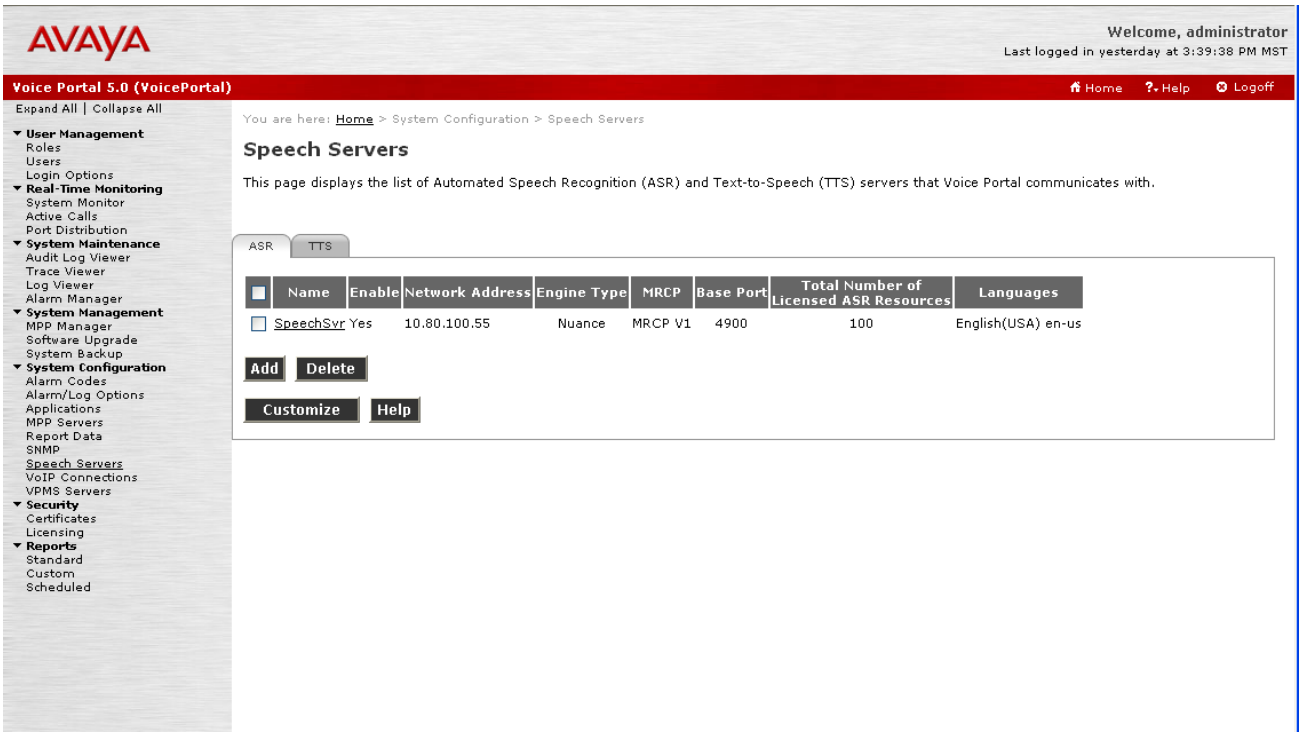
Under System Management select MPP Manager, on the left hand side. This will list all the administered MPPs. To place a MPP in service, check the box to the left of the server name and click on Start.

The screen shot below shows the MPP is Online and in the Running State.



5.2. Configure the Speech Server

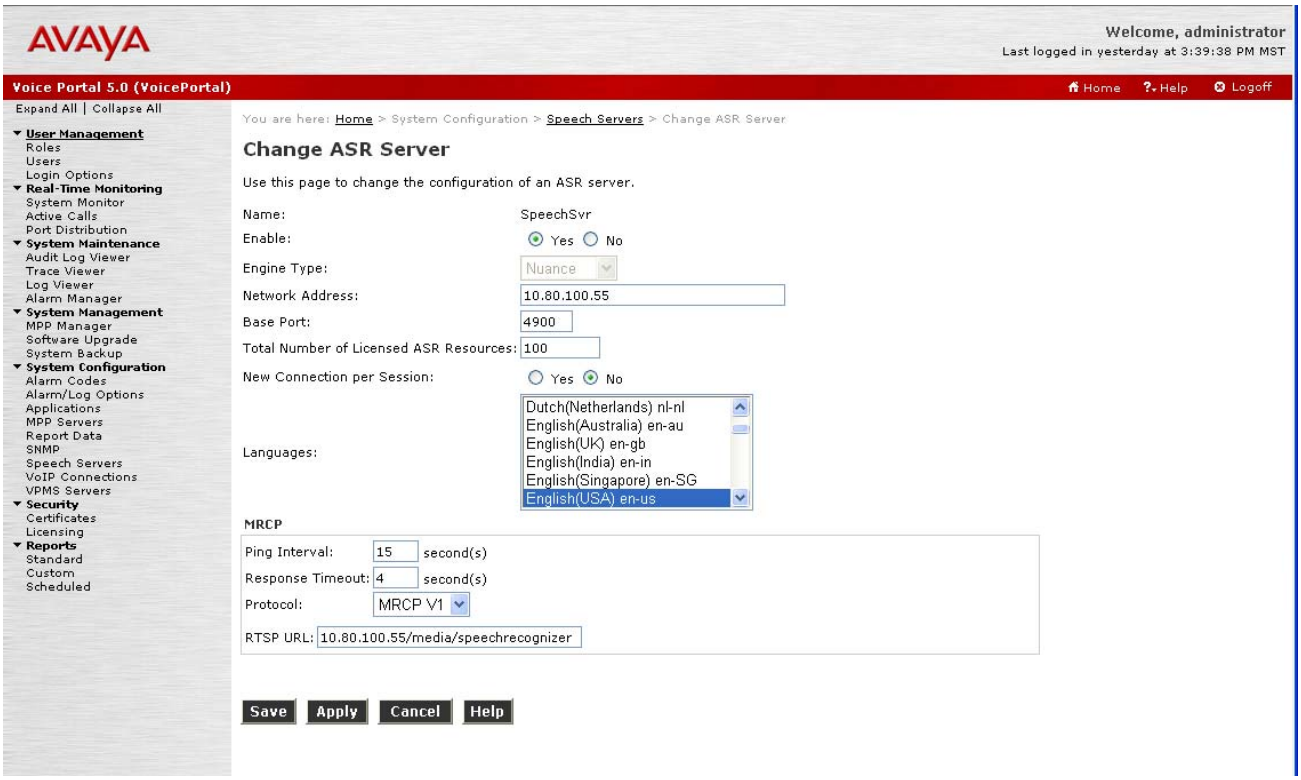
Under System Configuration, Select Speech Servers. Verify the Speech Server is correctly configured with the IP address of the host machine running the application server and speech server. The configuration of the Speech Server for the sample configuration is shown below:



5.2.1. Add the Automated Speech Recognition Server (ASR)

On the Speech Servers page when the ASR tab is highlighted, select the Add button at the bottom of the page and enter the required details such as the IP address of the speech server host machine, Engine Type such as Nuance, etc.

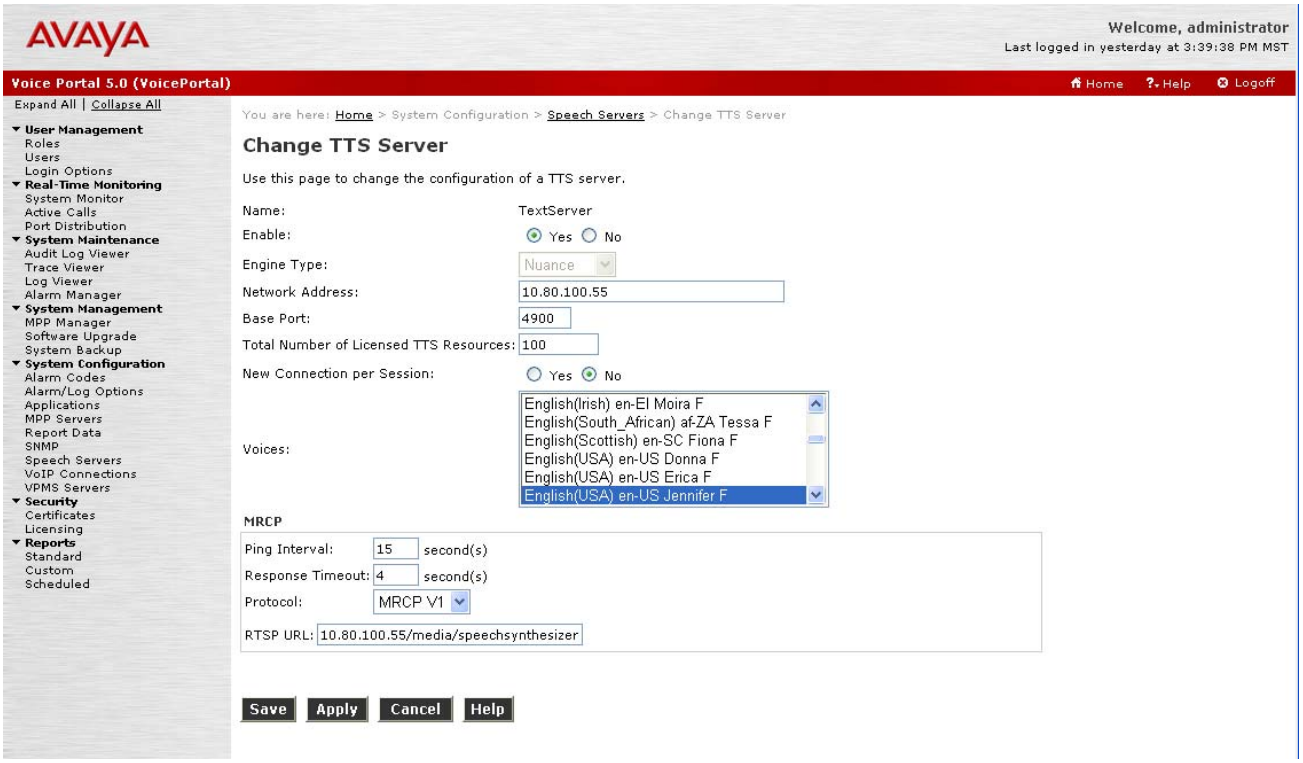
Below is a screen after ASR is added for the sample configuration.



5.3. Add the Text-To-Speech Server (TTS)

On the Speech Servers page, select the TTS tab and enter the required details such as the IP address of the TTS host machine, Engine Type such as Nuance, etc.

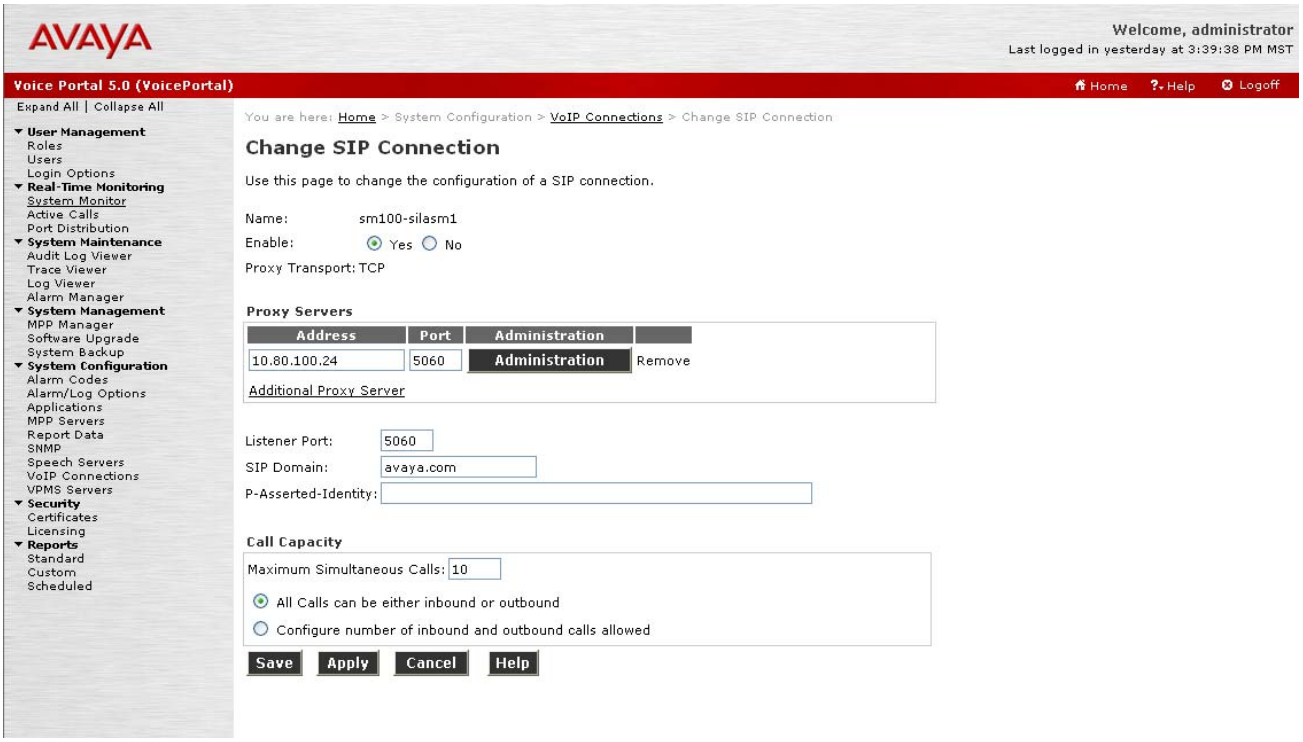
Below is the screen after TTS is added for the sample configuration.



5.4. Add a SIP Connection for Session Manager

Under System Configuration, select VoIP Connections. Select the SIP tab and Click on New. Add the SIP connection details which would contain details of the SIP interface to Session Manager. Enter IP address of the Session Manager Security Module (SM 100) under Proxy Servers. Verify the port number matches the port number defined in **Section 4.4.1** for the Voice Portal SIP entity. Finally, verify the SIP domain matches the SIP domains defined in Session Manager.

Below is a screen after the SIP Connection is added for the sample configuration.

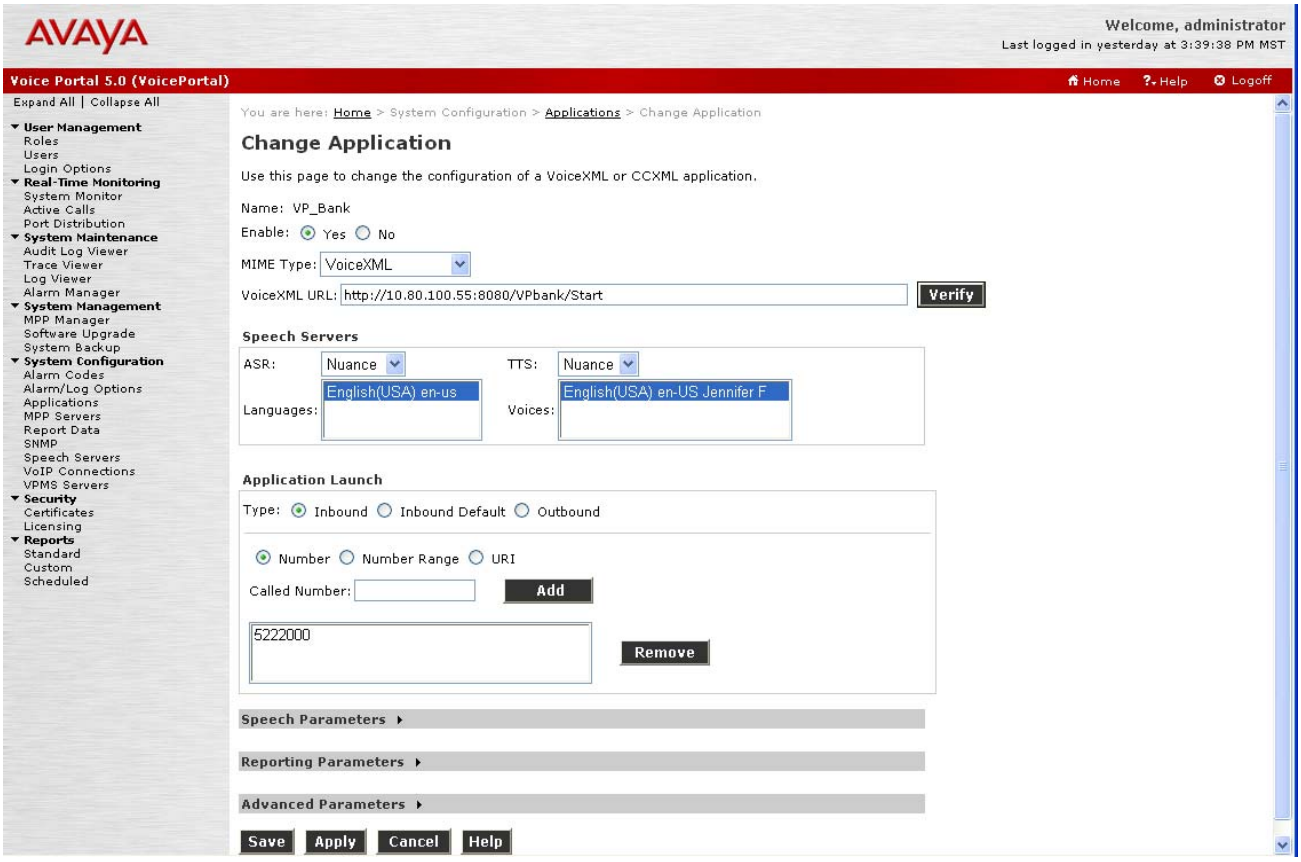


5.5. Setup Self Service Applications

Under System Configuration → Applications, add the Application URL, the Speech Servers configured, and the DNIS number to access the application that will be used in the sample configuration.

As shown below for the sample configuration, the following values were entered:

- url to the VP Bank application:
  http://10.80.100.55:8080/VPBank/Start
- DNIS of 5222000

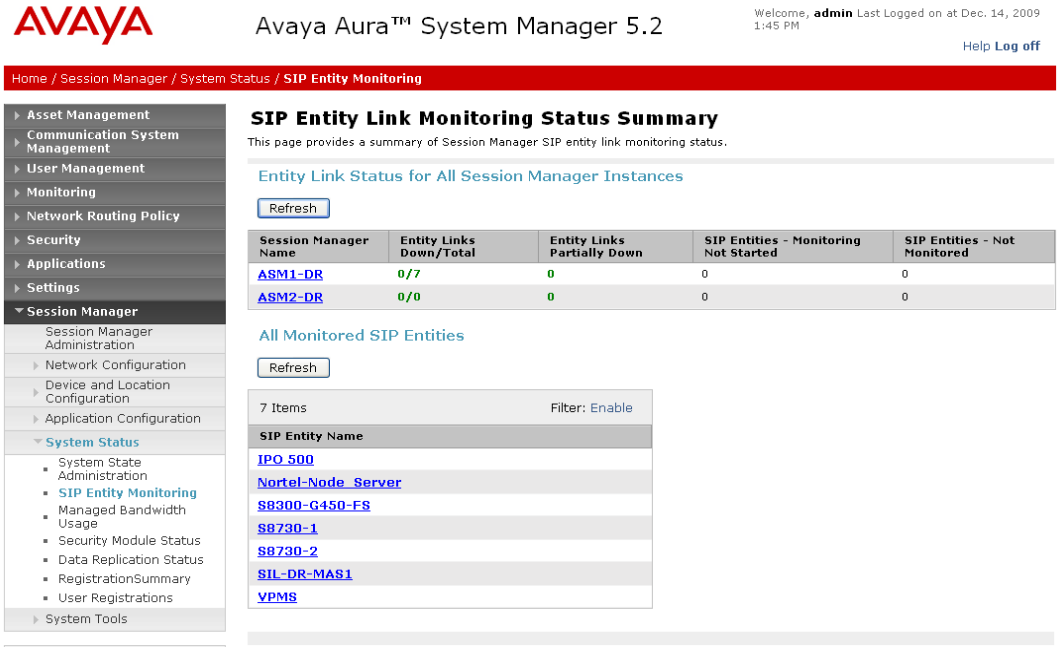


6. Verification Steps

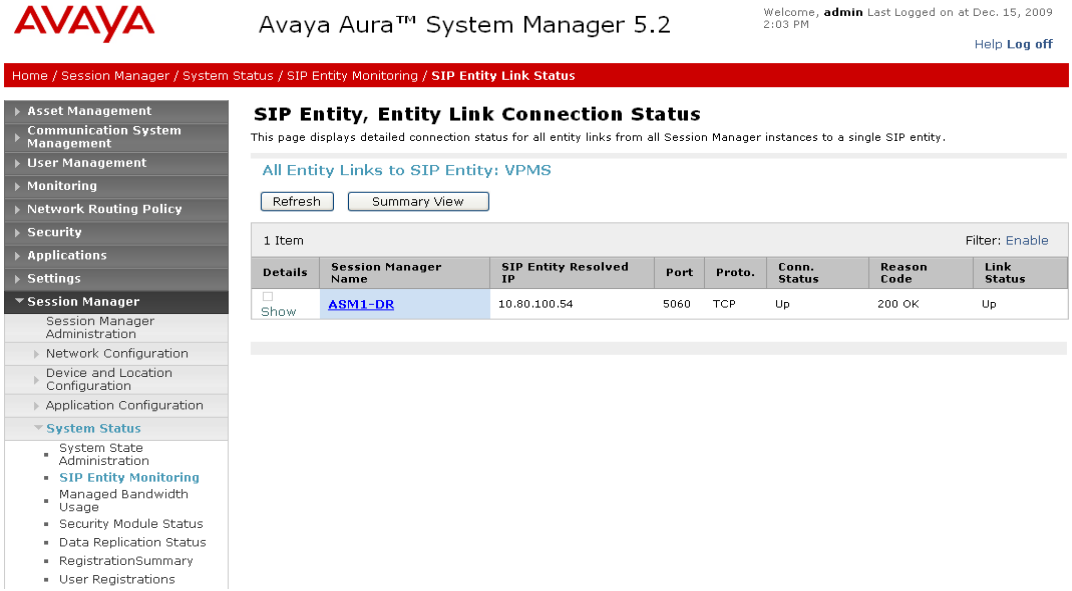
This section provides the tests that can be performed on Voice Portal, Communication Manager and Session Manager to verify proper configuration of these systems.

6.1. Verify Avaya Aura™ Session Manager Configuration

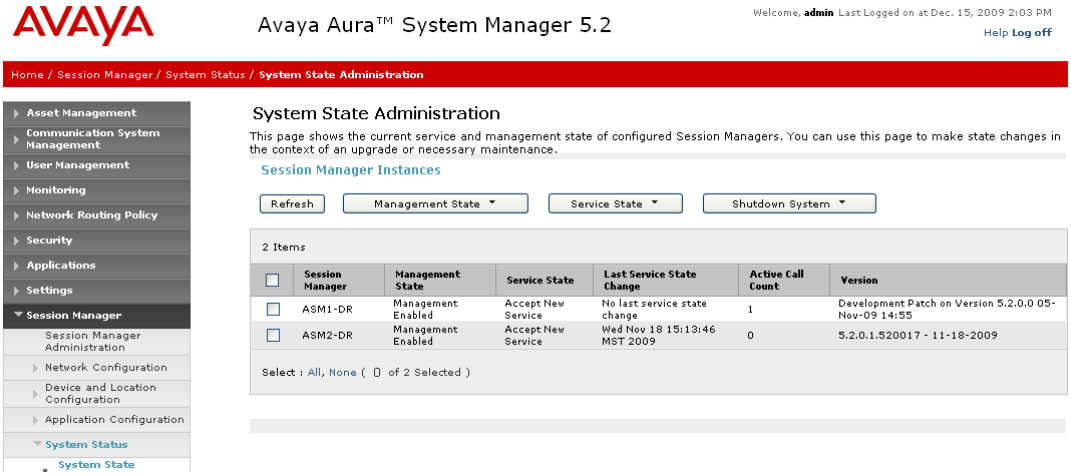
Expand the Session Manager menu on the left and click SIP Entity Monitoring. Verify all SIP Entity Links are operational as shown below:



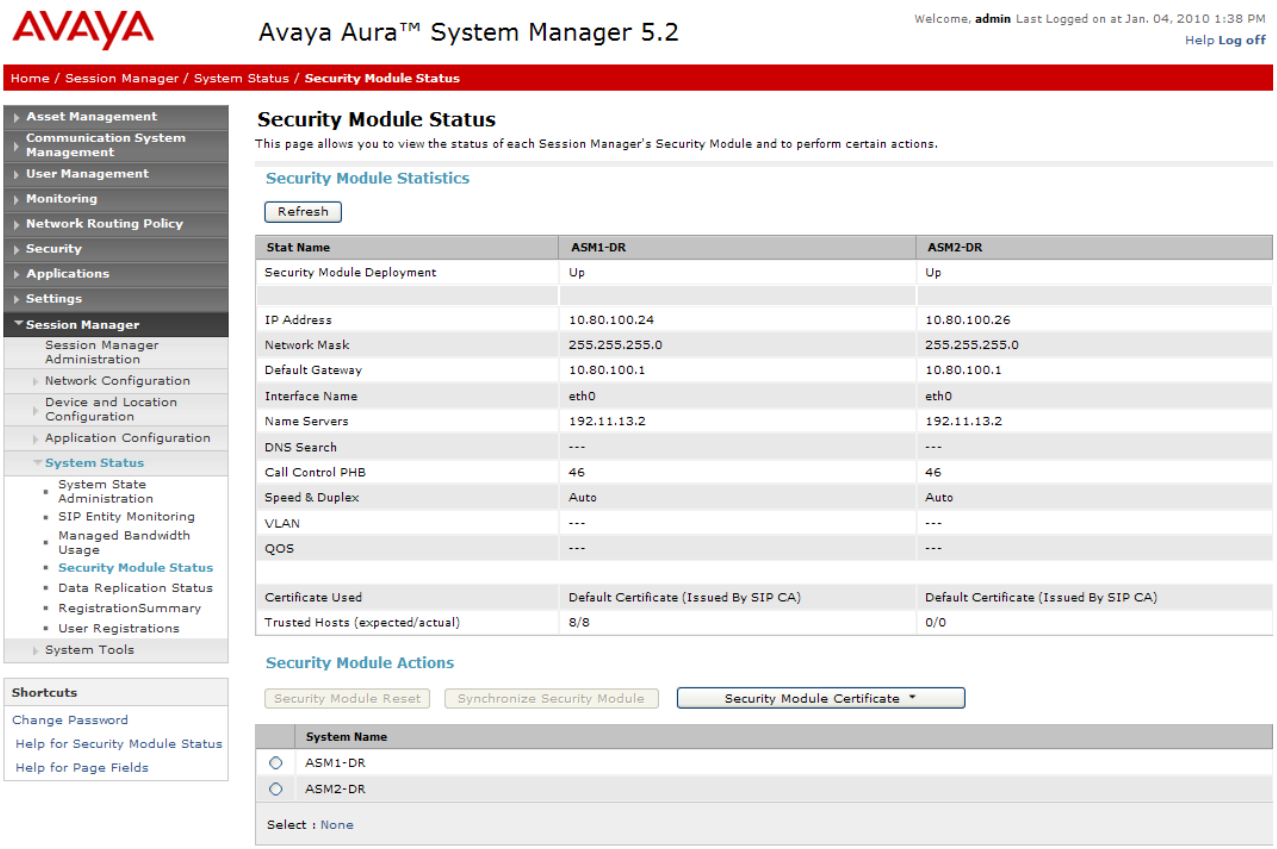
Select the corresponding SIP Entity for the VP system and verify the link is up as shown below:



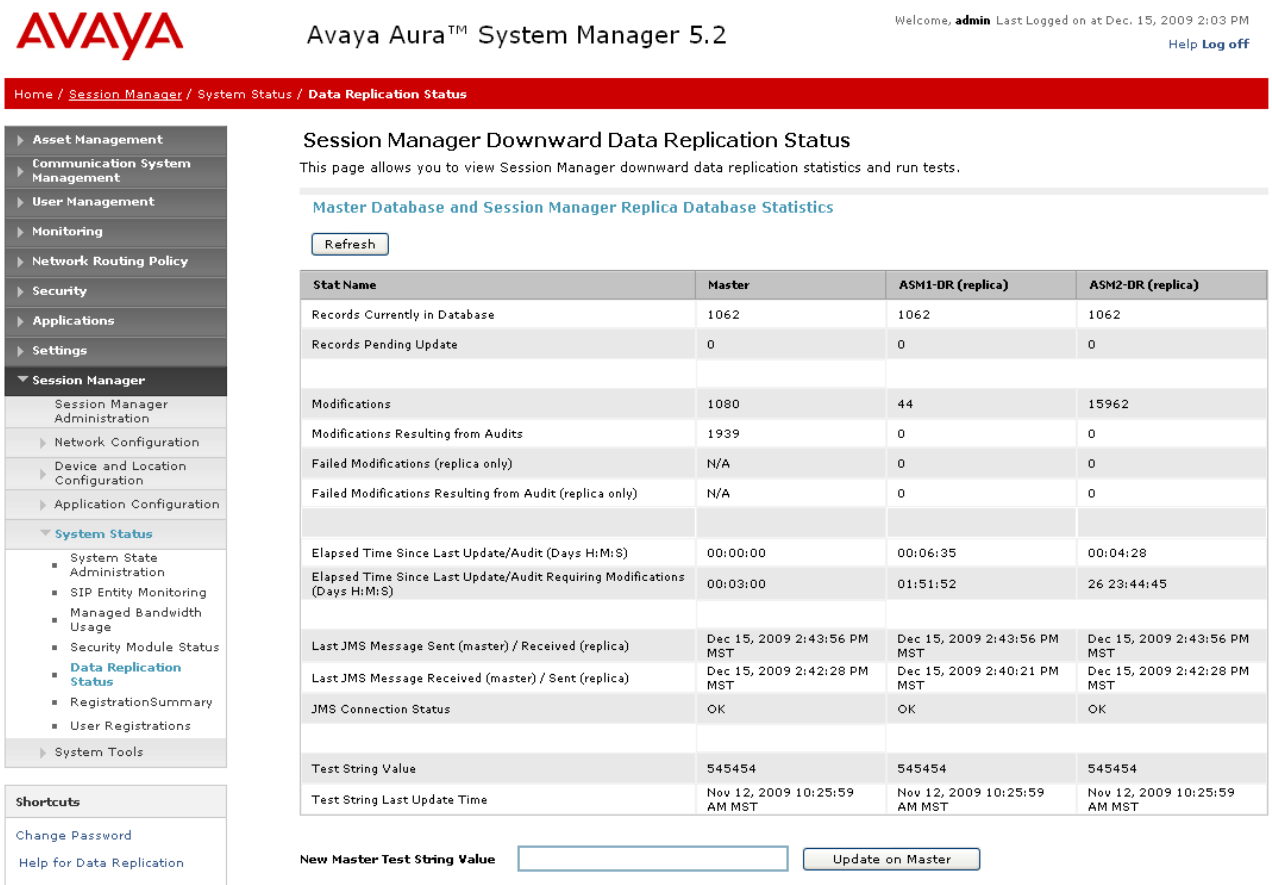
Verify the overall system status for the specific Session Manager as shown below:



Verify the status of the Security Module (SM 100 card) for the specific Session Manager as shown below:



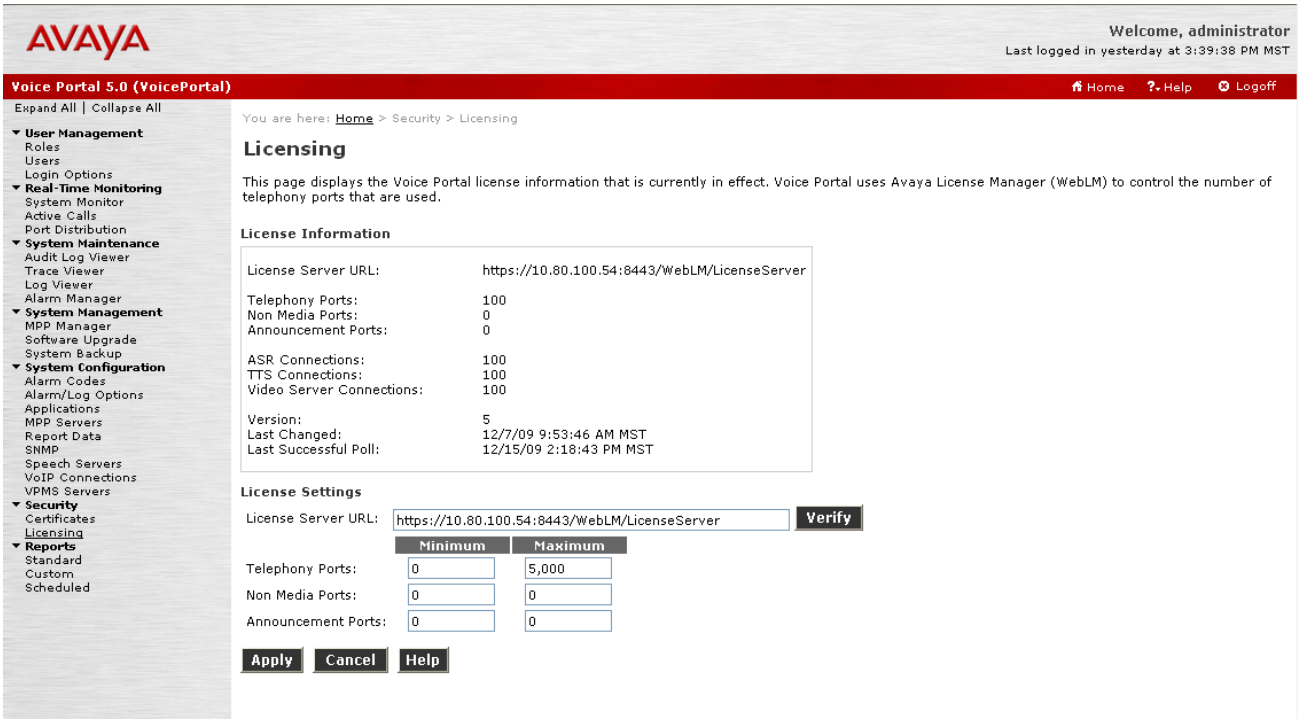
Finally, verify the data replication status as shown below:



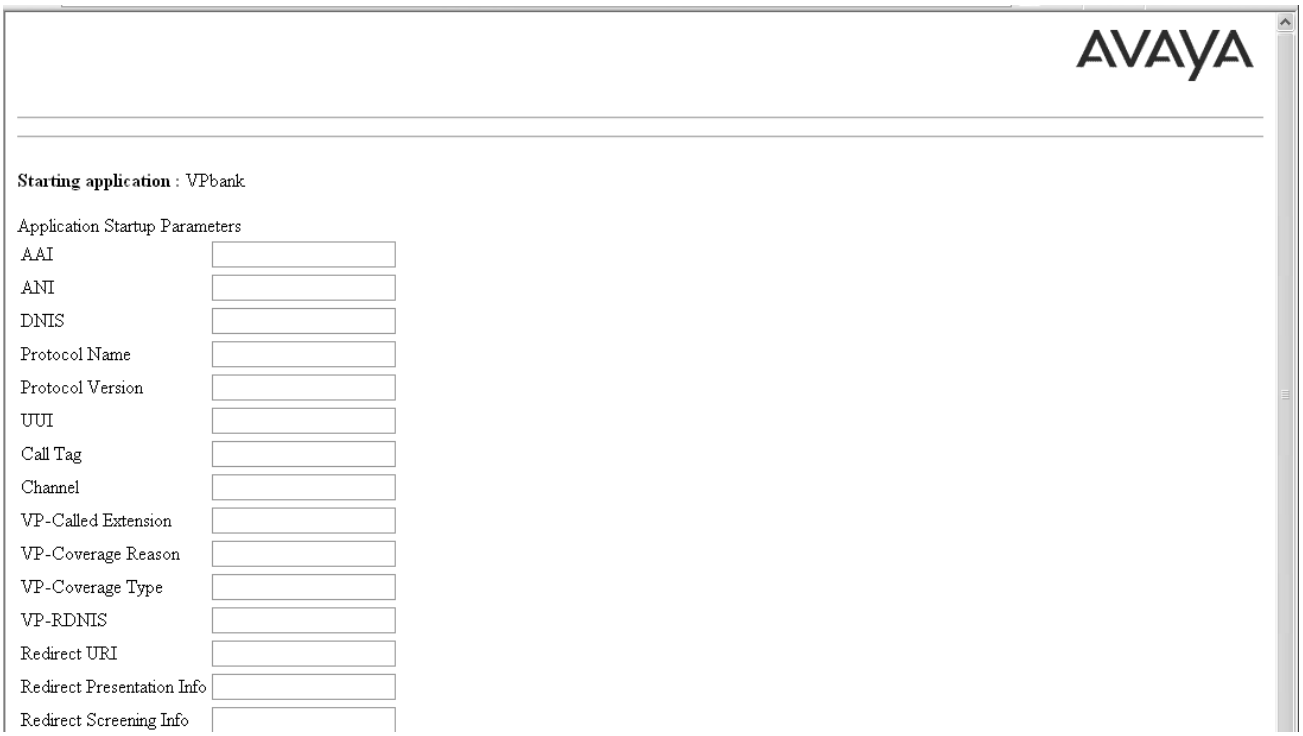
Stat Name	Master	ASM1-DR (replica)	ASM2-DR (replica)
Records Currently in Database	1062	1062	1062
Records Pending Update	0	0	0
Modifications	1080	44	15962
Modifications Resulting from Audits	1939	0	0
Failed Modifications (replica only)	N/A	0	0
Failed Modifications Resulting from Audit (replica only)	N/A	0	0
Elapsed Time Since Last Update/Audit (Days H:M:S)	00:00:00	00:06:35	00:04:28
Elapsed Time Since Last Update/Audit Requiring Modifications (Days H:M:S)	00:03:00	01:51:52	26 23:44:45
Last JMS Message Sent (master) / Received (replica)	Dec 15, 2009 2:43:56 PM MST	Dec 15, 2009 2:43:56 PM MST	Dec 15, 2009 2:43:56 PM MST
Last JMS Message Received (master) / Sent (replica)	Dec 15, 2009 2:42:28 PM MST	Dec 15, 2009 2:40:21 PM MST	Dec 15, 2009 2:42:28 PM MST
JMS Connection Status	OK	OK	OK
Test String Value	545454	545454	545454
Test String Last Update Time	Nov 12, 2009 10:25:59 AM MST	Nov 12, 2009 10:25:59 AM MST	Nov 12, 2009 10:25:59 AM MST

6.2. Verify Voice Portal Configuration

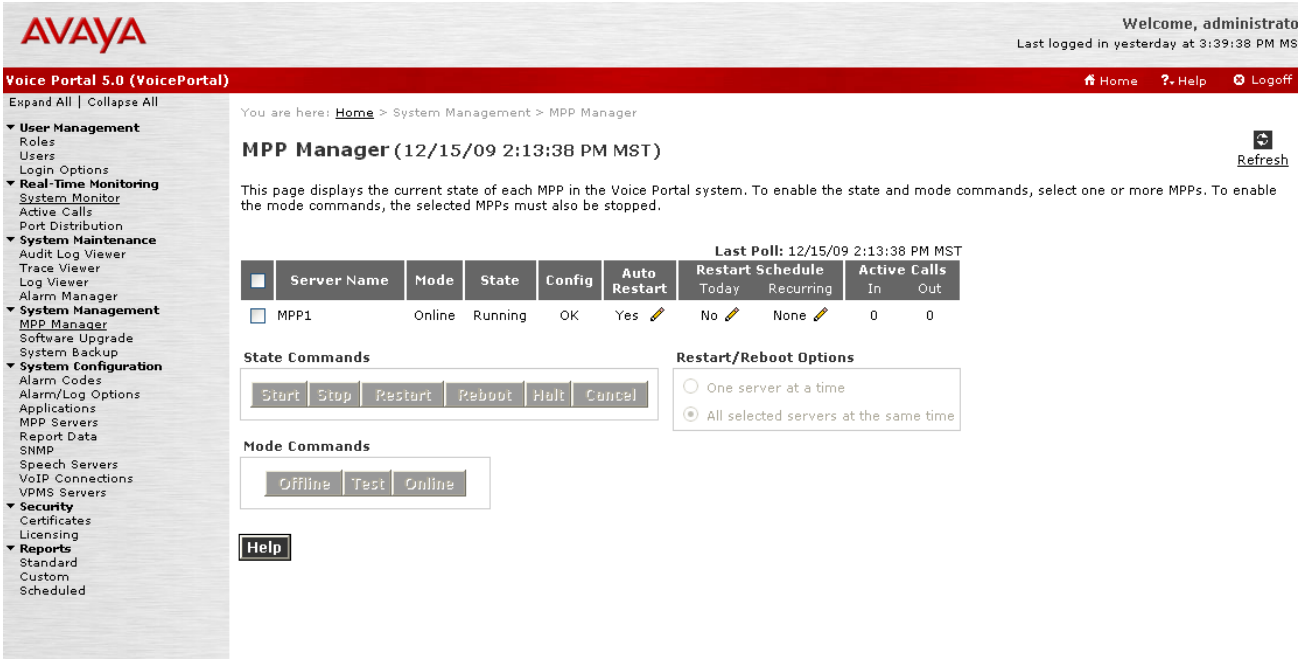
Verify the correct licenses for the VP system by accessing the Licensing link as shown below:



Verify the application has been correctly configured by selecting the sample VP bank application and selecting the “Verify” option. If the system has been correctly configured, a separate window should be displayed as shown below:



Use the **Real-Time Monitoring → System Monitor** or **System Management → MPP Manager** to ensure MPPs are online, running and receiving calls as shown below:



6.3. Verify Avaya Aura™ Communication Manager Access Element Configuration

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 2.4.2**. Verify that all trunks are in the "in-service/idle" state as shown below:

```
status trunk 20
                              TRUNK GROUP STATUS
Member   Port     Service State      Mtce Connected Ports
                                     Busy0020/001 T00057   in-service/idle
no
0020/002 T00058   in-service/idle    no
0020/003 T00059   in-service/idle    no
0020/004 T00060   in-service/idle    no
0020/005 T00061   in-service/idle    no
0020/006 T00062   in-service/idle    no
0020/007 T00063   in-service/idle    no
0020/008 T00064   in-service/idle    no
0020/009 T00065   in-service/idle    no
0020/010 T00066   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 2.4.1.**

Verify the signaling group is “in-service” as indicated in the **Group State** field shown below:

```
status signaling-group 20
                          STATUS SIGNALING GROUP

        Group ID: 20                                Active NCA-TSC Count: 0
      Group Type: sip                                Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

6.4. Verification Scenarios

For all scenarios, perform the following steps:

Step 1: Log in an agent on the Communication Manager Access Element.
Step 2: Verify caller is able to hear the appropriate prompts in the VP application.
Step 3: Select the option to talk to an agent and verify call is delivered to the agent.

6.4.1. Call Scenarios Verified

- Basic Call flow from external PSTN phone
  - Verify Communication Manager is able to route external call to VP application through the Session Manager.
  - Verify call is successfully transferred to the agent
- Basic Call flow from internal Avaya phones (includes both digital and IP stations)
  - Verify Communication Manager is able to route internal call to VP application through the Session Manager.
  - Verify call is successfully transferred to the agent
- Basic Call flow from SIP endpoints registered to Session Manager
  - Verify Communication Manager is able to route call from SIP endpoint to VP application through the Session Manager.
  - Verify call is successfully transferred to the agent
- Multiple Calls
  - Verify VP application is able to support multiple, simultaneous calls.
  - Verify first call can still be successfully transferred to the agent
  - Verify other calls are queued
- Queuing
  - Place multiple, simultaneous calls to VP application
  - Verify first call is successfully transferred to the agent
  - Verify calls queued for > 5 minutes are delivered to agent once agent becomes available

6.4.2. Verify status on Communication Manager Access Element

Use the Communication Manager SAT command, '**list trace tac #'**, where **tac #** is the trunk access code defined in **Section 2.4.2** to trace trunk group activity for the SIP trunk between the Session Manager and Communication Manager as shown below:

```
list trace tac #20                                          Page   1

                                LIST TRACE

time            data

17:35:44     dial 5222000 route:UDP|HNPA|ARS
17:35:44     term trunk-group 20    cid 0x200
17:35:44     dial 5222000 route:UDP|HNPA|ARS
17:35:44     route-pattern  20 preference 1  cid 0x200
17:35:44     seize trunk-group 20 member 10  cid 0x200
17:35:44     Setup digits 5222000
17:35:44     Calling Number & Name NO-CPNumber H.323 4621 IP
17:35:44     Proceed trunk-group 20 member 10  cid 0x200
17:35:44     G711MU ss:off ps:20
             rgn:1 [10.80.100.54]:23374
             rgn:1 [10.80.111.13]:9660
17:35:44     xoip options: fax:Relay modem:off tty:US  uid:0x50042
             xoip ip: [10.80.111.13]:9660
17:35:44     active trunk-group 20 member 10  cid 0x200
17:35:44     G711MU ss:off ps:20
```

Use the Communication Manager SAT command, '**list trace vdn**" where **vdn** is the VDN number defined in **Section 2.7.1** to trace calls from the VP Application are correctly processed on the Communication Manager Access Element as shown below:

```
list trace vdn 5221000

                                LIST TRACE VDN

          vec
time      prt st data

17:36:57    0  0 ENTERING TRACE cid 515
17:36:57 1000  1 vdn e5221000 bsr appl   0 strategy 1st-found override n
17:36:57 1000  1 queue-to
17:36:57 1000  1      queueing to skill 522 pri h
17:36:57 1000  1 Local Agent Preference=n
17:36:57 1000  1 Agent Login ID: 50001  Logged in at station: 4001000
17:36:57 1000  1 LEAVING VECTOR PROCESSING cid 515
17:36:57 1000  1 TRACE COMPLETE cid 515
```

Use the Communication Manager SAT command, "**mon bcms system**" to verify calls are in queue or delivered to agents as shown by screen below.

```
monitor bcms system                                                    Page   1 of   1
                                BCMS SYSTEM STATUS
                                                   Date:   13:33 THU DEC 10 2009

                                AVG                   AVG          AVG   AVG    % IN
                   CALLS OLDEST SPEED AVAIL ABAND ABAND ACD        TALK AFTER SERV
SKILL NAME         WAIT  CALL   ANS   AGENT CALLS TIME  CALLS      TIME CALL  LEVL

VP Agents             2   0:09  0:00     0     0  0:00     3      0:26  0:00   67
```

6.4.3. Verify status on Voice Portal

- Verify the prompts from the application can be heard.
- Go to the VPMS webpage "Real-Time Monitoring → System Monitor → Active Calls" to ensure MPPs are online, running and receiving calls as shown below:

Active Calls (12/10/09 11:09:33 AM MST)

Refresh

This page displays the status of all the active calls being handled by the Voice Portal system.

Total Active Calls: 3 **Last Poll:** 12/10/09 11:09:33 AM MST

Port	Port Group	Protocol	Call Type	MPP Server	Start Time	Calling Number/URI	Called Number/URI	Application	ASR Server	TTS Server
1	sm100-silasm1	SIP_Trunk	Inbound	MPP1	12/10/09 11:08:52 AM MST	anonymous	tel:5222000	VP_Bank	SpeechSvr	TextServer
2	sm100-silasm1	SIP_Trunk	Inbound	MPP1	12/10/09 11:09:09 AM MST	tel:6663000	tel:5222000	VP_Bank	SpeechSvr	TextServer
3	sm100-silasm1	SIP_Trunk	Inbound	MPP1	12/10/09 11:09:26 AM MST	anonymous	tel:5222000	VP_Bank	SpeechSvr	TextServer

6.4.4. Verify status on Session Manager

The SIP Tracing Viewer on System Manager can be used to display SIP message traces between Session Manager and SIP entities, based on configurable filters. For more information on how to configure SIP tracing, see Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325.

7. Acronyms

ARS	Automatic Route Selection (Routing on Communication Manager)
ASR	Automatic Speech Recognition
BCMS	Basic Call Management System (used for monitoring ACD calls)
CLAN	Control LAN (Control Card in Communication Manager)
DCP	Digital Communications Protocol
DD	Avaya Voice Portal Dialog Designer
DNIS	Dialed Number identification Service
DTMF	Dual Tone Multi Frequency
FQDN	Fully Qualified Domain Name (hostname for Domain Naming Resolution)
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPSI	IP-services interface (Control Card in Communication Manager)
IVR	Interactive Voice Response
LAN	Local Area Network
MPP	Media Processing Platform (Voice Portal Server)
MRCP	Media Resource Control Protocol
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SAT	System Access Terminal
SIL	Solution Interoperability Lab
SIP	Session Initiation Protocol
SM	Avaya Aura™ Session Manager
SMGR	Avaya Aura™ System Manager
SNMP	Simple Network Management Protocol
SRE	SIP Routing Element
SSH	Secure Shell
SSL	Secure Socket Layer
TAC	Trunk Access Code (Communication Manager Trunk Access)
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TLS	Transport Layer Security
TTS	Text To Speech
URE	User Relation Element
URL	Uniform Resource Locator
VDN	Vector Directory Number
VP	Voice Portal
VP 1st	Voice Portal First
VPMS	Voice Portal Management Server
WAN	Wide Area Network
XML	eXtensible Markup Language

8. Conclusion

These Application Notes describe how to configure the Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element, Avaya Aura™ Communication Manager operating as a Feature Server, Avaya G650 Media Gateway, and Avaya Voice Portal (VP) to support a Voice Portal First (VP 1st) solution.

Interoperability testing included verification of basic calls to the VP application from several types of endpoints including ability to transfer call to an agent on Communication Manager.

9. Additional References

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

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.

Communication Manager

4) Hardware Description and Reference for Avaya Aura™ Communication Manager (COMCODE 555-245-207) http://support.avaya.com/elmodocs2/comm_mgr/r4_0/avayadoc/03_300151_6/245207_6/245207_6.pdf
5) 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.
6) Administering Avaya Aura™ Communication Manager, Doc ID 03-300509, May 2009, available at http://support.avaya.com.
7) Administering Avaya Aura™ Communication Manager as a Feature Server, Doc ID 03-603479, November 2009, available at http://support.avaya.com

Voice Portal

8) Administering Voice Portal, available at http://support.avaya.com
9) Configuring Avaya Voice Portal with Avaya Communication Manager and Designing a Sample Speech Application using Dialog Designer – Issue 1.0 : http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/voiceportal.pdf

Avaya Application Notes

1) Voice Portal First Solution: Configuring Avaya Aura™ Session Manager with Avaya G860 High Density Trunk Gateway, Avaya Aura™ Communication Manager and Avaya Voice Portal – Issue 1.0, available at http://www.avaya.com

2) Configuring 9600-Series SIP Phones on Avaya Aura™ Session Manager Release 5.2, 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