Avaya Solution & Interoperability Test Lab

Application Notes for NMS Communications Vision Media Gateway VG2000 and Vision Signaling Server VS5000 with Avaya Voice Portal and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to enable calls between Avaya Voice Portal and the PSTN through a SIP infrastructure consisting of Avaya SIP Enablement Services and the NMS Communications Vision Media Gateway VG2000. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls to Avaya SIP telephones or the PSTN. The Vision Media Gateway or VG2000 combines signaling and media gateway functions providing PSTN access to SIP-based telephony networks.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to enable calls between Avaya Voice Portal and the PSTN through a SIP infrastructure consisting of Avaya SIP Enablement Services and the NMS Communications Vision Media Gateway VG2000. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls to Avaya SIP telephones or the PSTN. The Vision Media Gateway or VG2000 combines signaling and media gateway functions providing PSTN access to SIP-based telephony networks. In this configuration, the VG2000 connects to Avaya Voice Portal through a SIP interface on Avaya SIP Enablement Services and to the PSTN via an SS7 interface. The VG2000 has separate network connections for IP-based call signaling (SIP) and for IP-based media (RTP). The Vision Signaling Server or VS5000 provides an interface to the SS7 network using the ISUP signaling protocol and terminates the SS7 signaling link. Refer to **Figure 1** for an illustration of the test configuration.

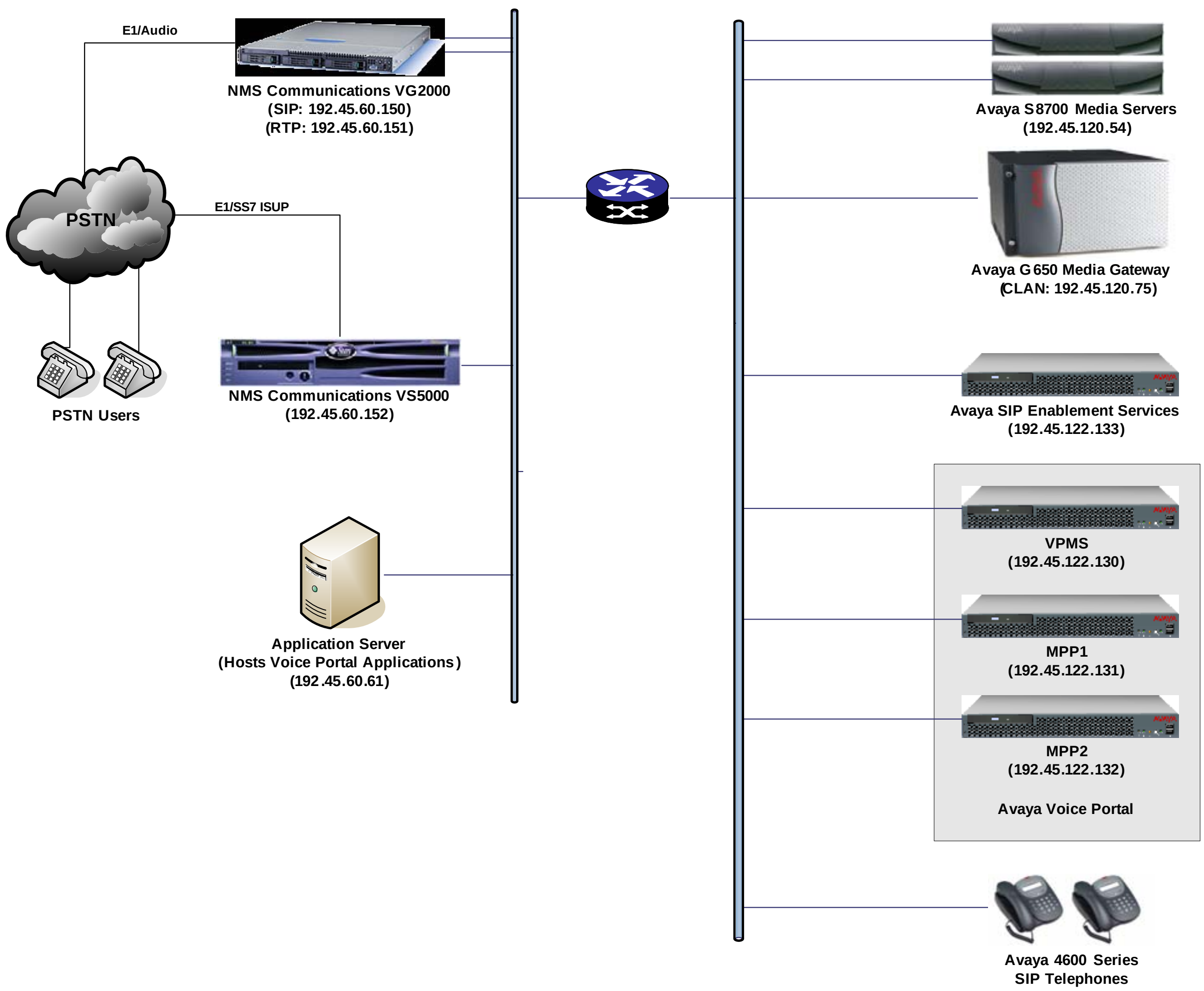


Figure 1: SIP-to-PSTN Interworking with the NMS Communications VG2000 & VS5000

1.1. Equipment and Software Validated

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

Equipment	Software
Avaya Voice Portal	4.0.0.1 Build 0003
Avaya SIP Enablement Services	4.0.0.0-033.6
Avaya S8700 Servers with a G650 Media Gateway	Avaya Communication Manager 4.0 (R014x.00.0.730.5)
Avaya 4600 Series SIP Telephones	2.2.2
NMS Communications Vision Media Gateway Model VG2000 with CG6565 PSTN Interface	Version 1.0 (Build 5997)
NMS Communications Vision Signaling Server Model VS5000 with TX4000 SS7 Signaling Board	Version 1.0
Application Server – HTTP Server running on Windows 2003 Server	Service Pack 2

2. Configure Avaya Communication Manager

Avaya Communication Manager was used in this configuration to support the Avaya SIP telephones. Avaya Communication Manager is in the call flow for all calls originating and terminating on Avaya SIP telephones, and extends telephony features to the SIP telephones. This required configuring a SIP trunk, off-PBX stations (OPS), and call routing on Avaya Communication Manager. A SIP trunk was established between Avaya Communication Manager and Avaya SIP Enablement Services. Off-PBX stations (OPS) were administered for each Avaya SIP telephone registered with Avaya SIP Enablement Services. Since the main focus of these Application Notes is on establishing calls between Avaya Voice Portal and the PSTN through a SIP infrastructure consisting of the Avaya SIP Enablement Services and the NMS Communications Vision Media Gateway, the configuration steps for Avaya Communication Manager are not covered here. Refer to [3] for details on configuring SIP trunks, OPS stations, and call routing.

3. Configure Avaya Voice Portal

This section covers the administration of Avaya Voice Portal. In this configuration, Voice Portal connected to the IP network via a SIP interface. Voice Portal configuration required:

- Importing certificates for TLS authentication
- Configuring a SIP connection
- Adding MPP servers
- Configuring the VoIP audio format (mu-law or a-law)
- Adding a speech server
- Adding applications
- Starting the MPP servers

Avaya Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter http://<ip-addr>/VoicePortal as the URL in an Internet browser, where *<ip-addr>* is the IP address of the VPMS. Log in using the Administrator user role. The screen shown in **Figure 2** is displayed.

Note: All of the screens in this section are shown after the Voice Portal had been configured. Don't forget to save the screen parameters after configuring Avaya Voice Portal.

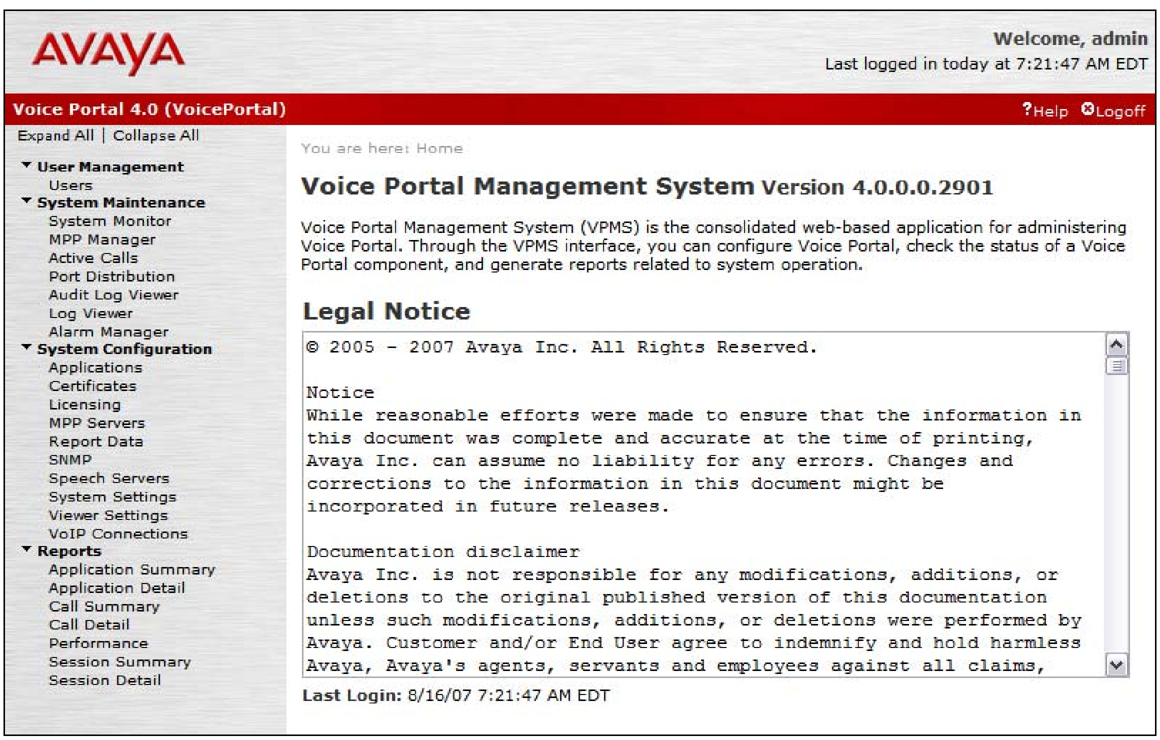


Figure 2: VPMS Main Screen

Install Certificate. In this configuration, Avaya Voice Portal used TLS authentication over the SIP interface to Avaya SIP Enablement Services. To install the certificate for TLS authentication, navigate to the **Certificates** page and select the **Root Certificate** tab. Specify the directory path where the certificate is located and the password, and then click **Install**. **Figure 3** shows a certificate that has already been installed.

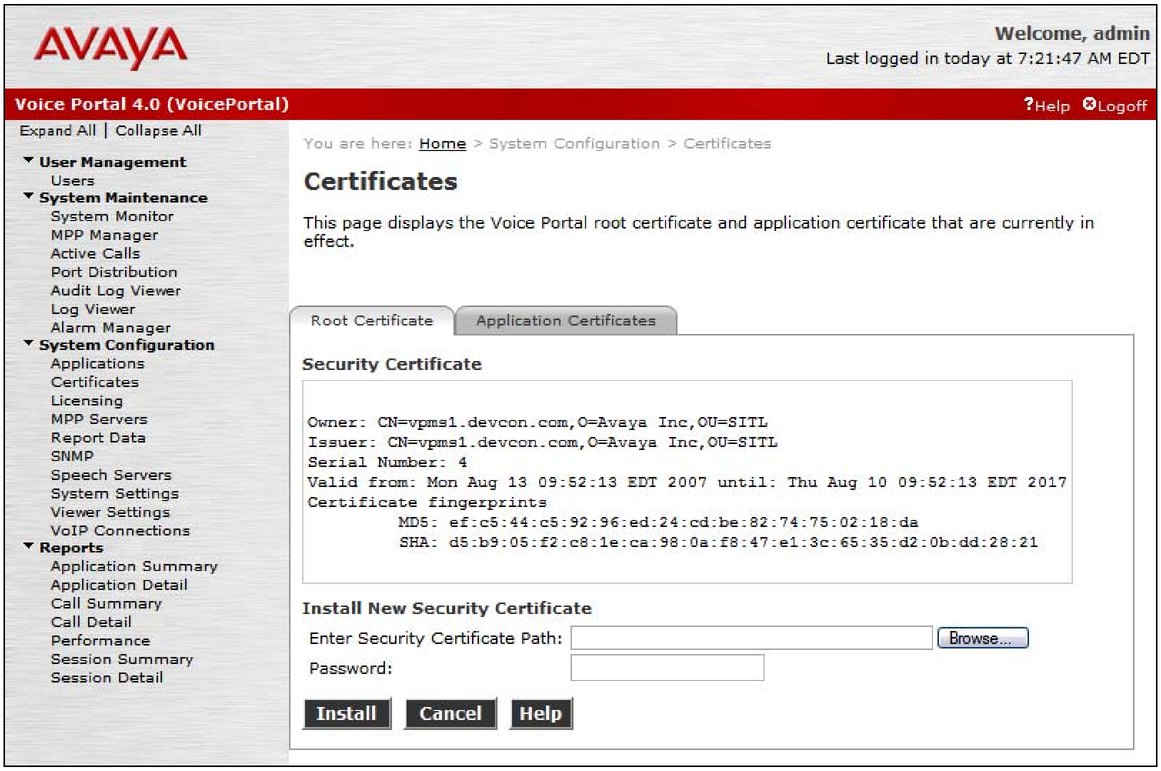


Figure 3: Certificates

Configure the SIP Connection. To configure a SIP connection, navigate to the **VoIP Connections** page and then click on the **SIP** tab. In the SIP tab shown in **Figure 4**, the following parameters must be configured:

- Select TLS as the **Proxy Transport**
- Specify the IP address of Avaya SIP Enablement Services in the **Proxy Server Address** field
- Set the **Proxy Server Port** and **Listener Port** fields to '5061' for TLS
- Set the **SIP Domain** (e.g., avaya.com)
- Set the **Maximum Simultaneous Calls** and **Number of Outbound Calls Allowed**. In this example, a maximum of 40 calls is supported between two MPP servers.
- Under the SIP Proxy Certificate section, click on the **Trust this Certificate** checkbox (not shown in the screen below)
- Accept the default values for the other fields

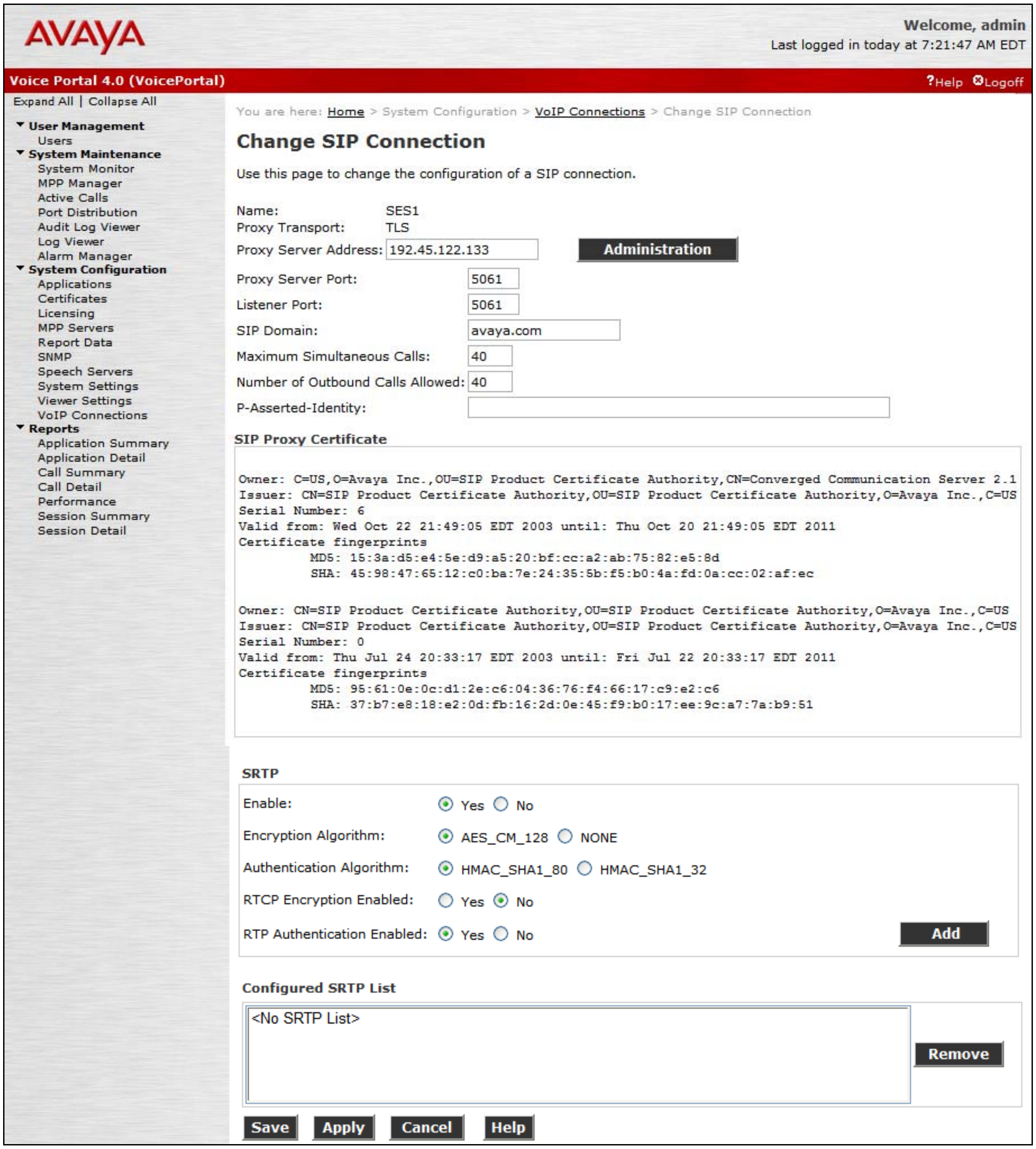


Figure 4: SIP Connection

Add the MPP Servers. Add the two MPP servers[1] by navigating to the **MPP Servers** screen. In the MPP Server configuration page, specify a descriptive name and the **Host Address** of each MPP server. Also, specify the **Maximum Simultaneous Calls** supported by each MPP server. **Figure 5** shows the configuration for the first MPP server. Repeat these steps for the second MPP server.

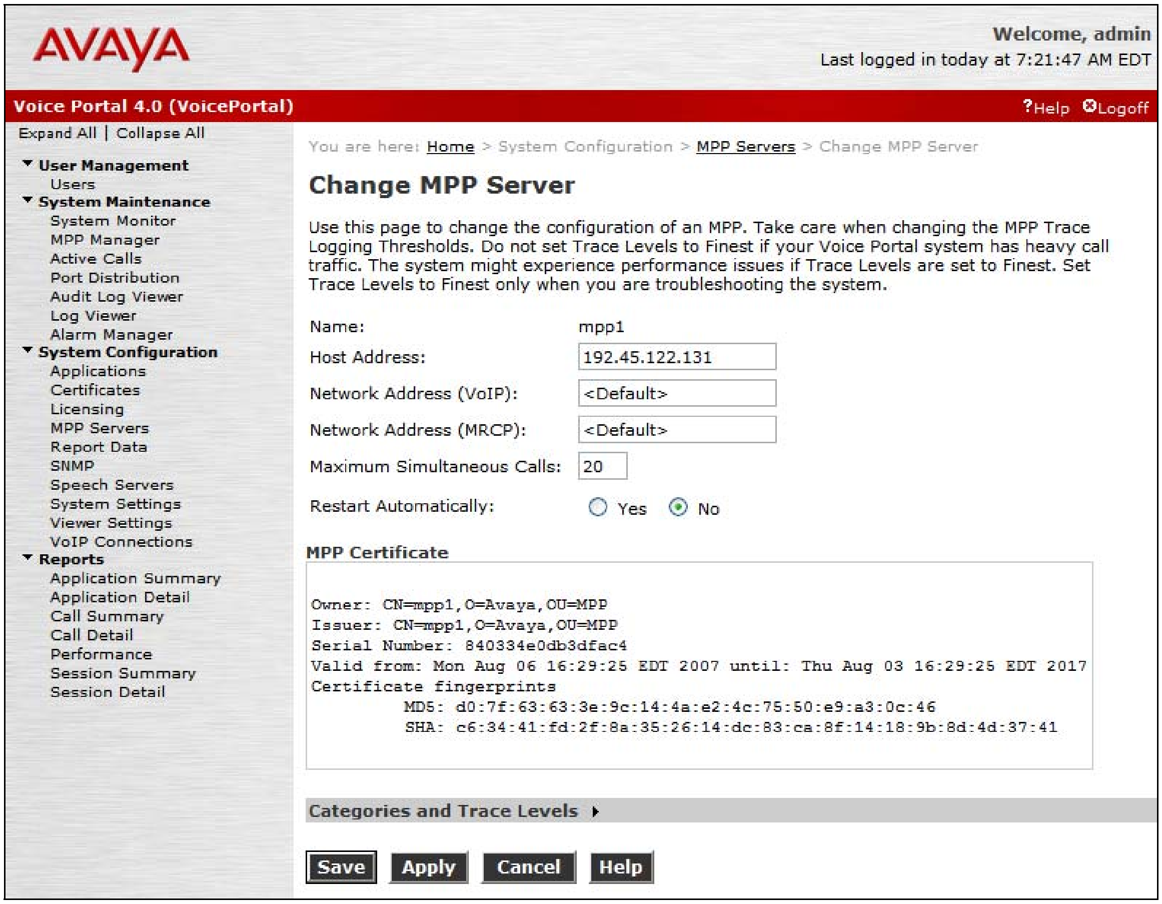


Figure 5: MPP Server

[1] Although two MPP servers were used in this configuration, only one MPP is required when deploying a Voice Portal system.

Configure the VoIP Audio Format. The **VoIP Audio Format** for the MPP servers is configured in the **VoIP Settings** screen. The VG2000 supports both mu-law and a-law. The **MPP Native Format** field in **Figure 6** is set to *audio/basic* for mu-law.

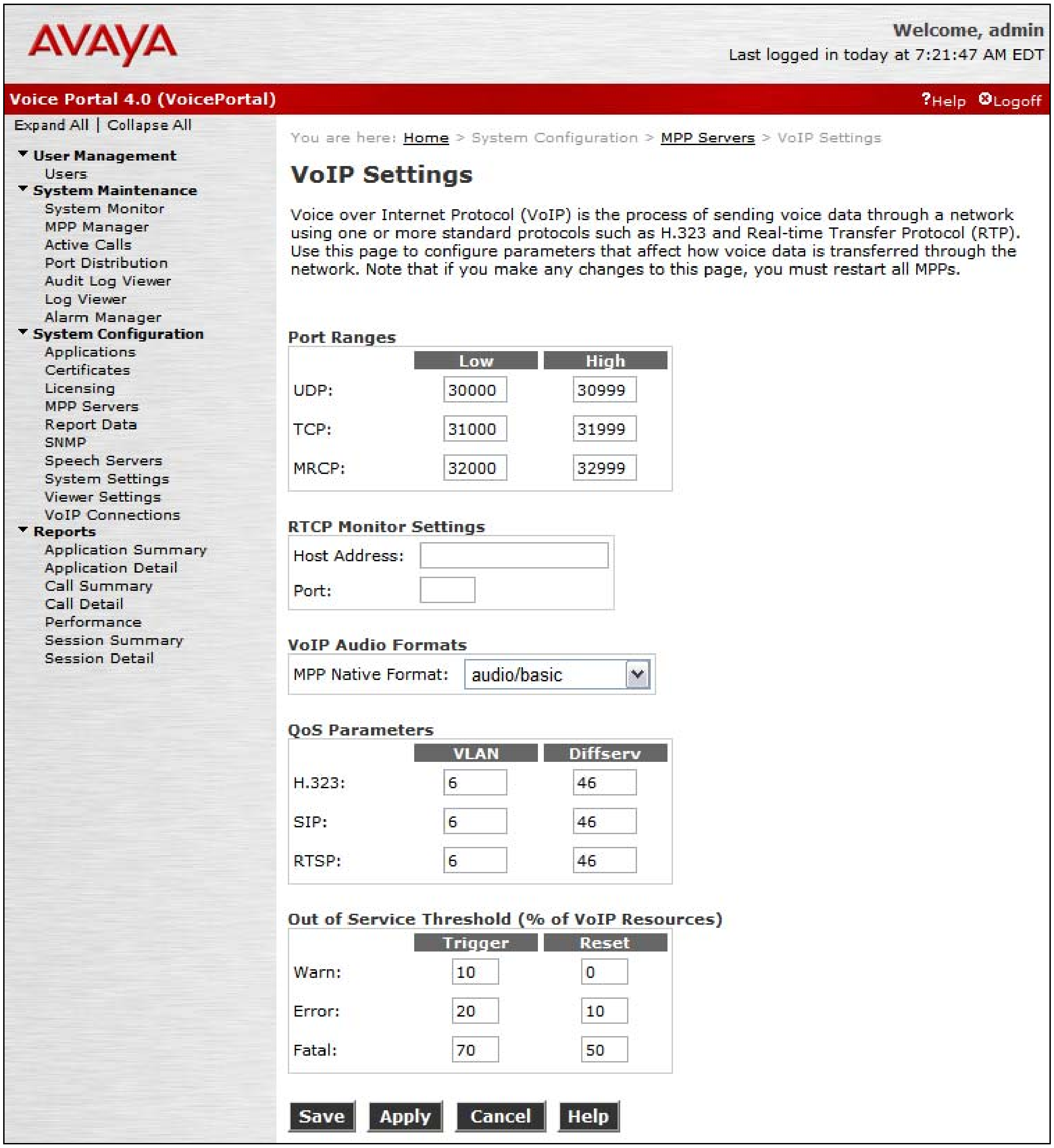


Figure 6: VoIP Settings

Add an ASR Server. To configure the ASR server, click on **Speech Servers** in the left pane, select the **ASR** tab, and then click **Add**. **Figure 7** shows the screen after the ASR server has already been configured. Set the **Engine Type** to the appropriate value. In this configuration, a Nuance ASR server was used so the engine type was set to *Nuance*. Set the **Network Address** field to the IP address of the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

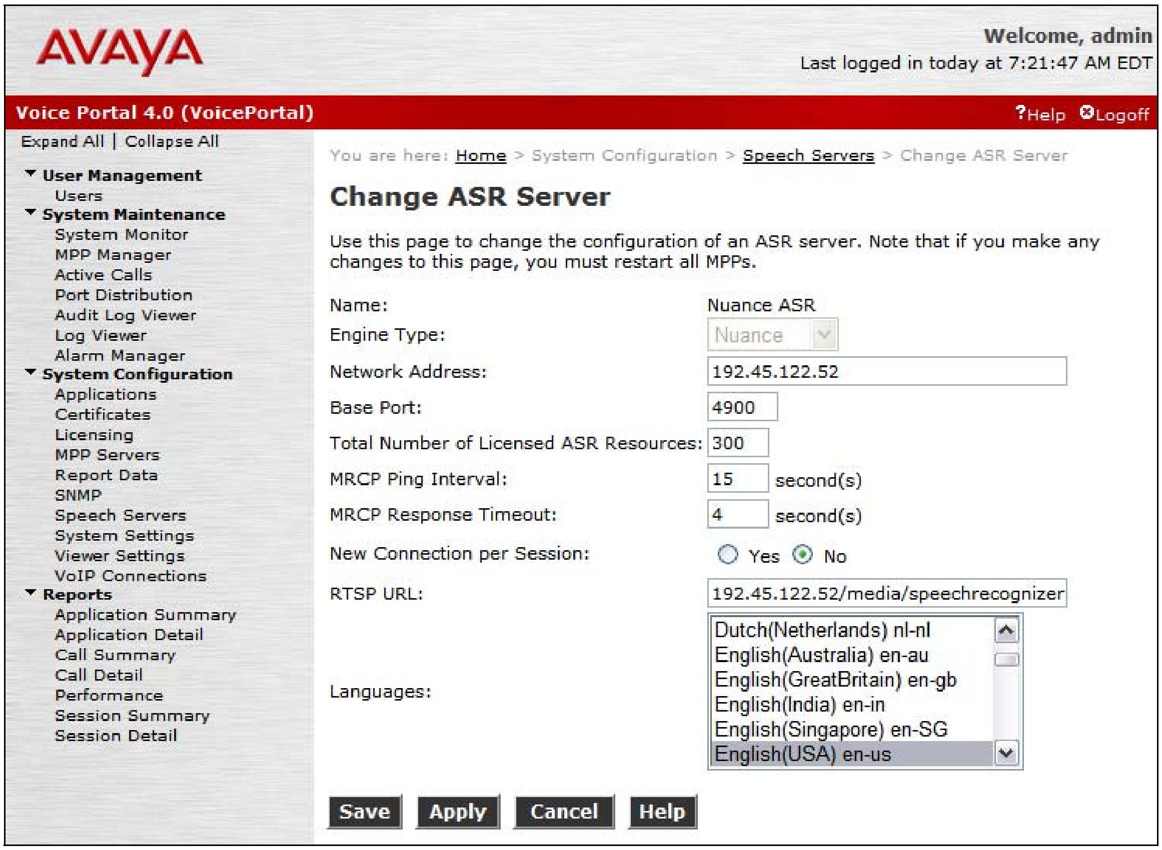


Figure 7: ASR Speech Servers

Add a TTS Server. To configure the TTS server, click on **Speech Servers** in the left pane, select the **TTS** tab, and then click **Add**. **Figure 8** shows the screen after the TTS server has already been configured. Set the **Engine Type** to the appropriate value. In this configuration, a Nuance TTS server was used so the engine type was set to *Nuance*. Set the **Network Address** field to the IP address of the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

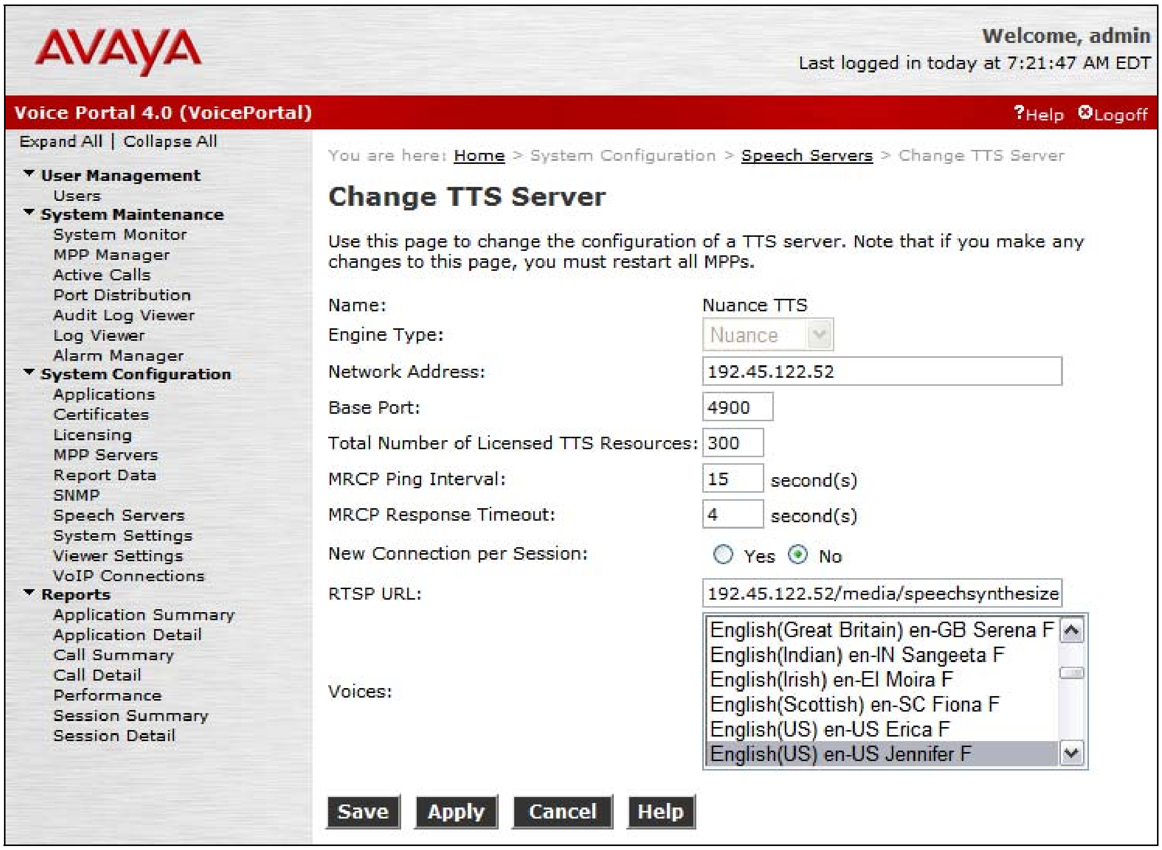


Figure 8: TTS Server

Add an Application. On the **Applications** page, add a Voice Portal application. Specify a **Name** for the application, set the **MIME Type** field to the appropriate value (e.g., VoiceXML), and set the **VoiceXML URL** field to point to a VoiceXML application on the application server. Next, specify the type of ASR and TTS servers to be used by the application and the called number that invokes the application. The called number is entered into the **Called Number** field and then the **Add** button is clicked. The **Applications** screen is shown in **Figure 9**.

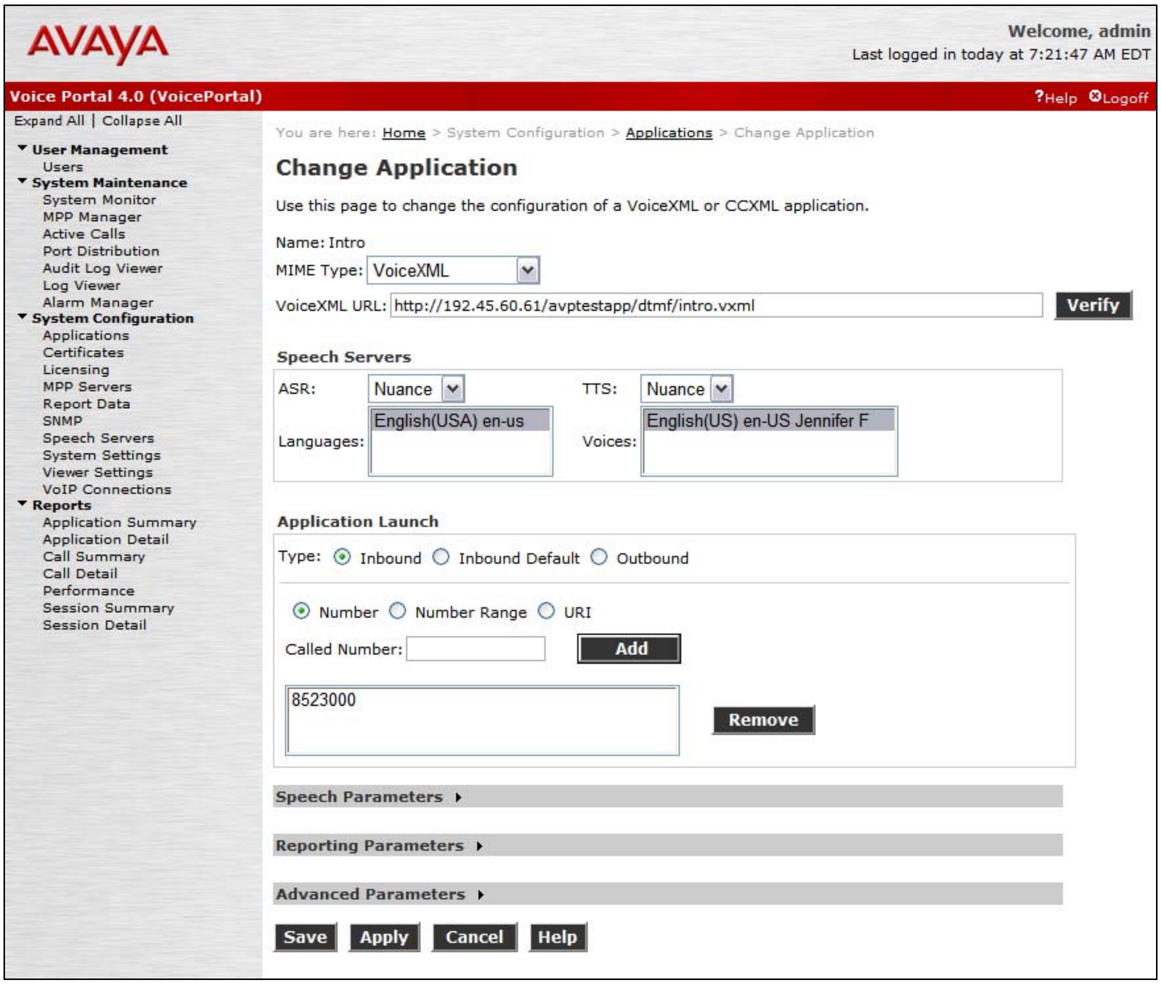


Figure 9: Applications

Start the MPP Servers. Start the MPP servers from the **MPP Manager** page shown in **Figure 10**. Select each MPP and then click the **Start** button. The **Mode** of each MPP should be *Online* and the **State** should be *Running*.

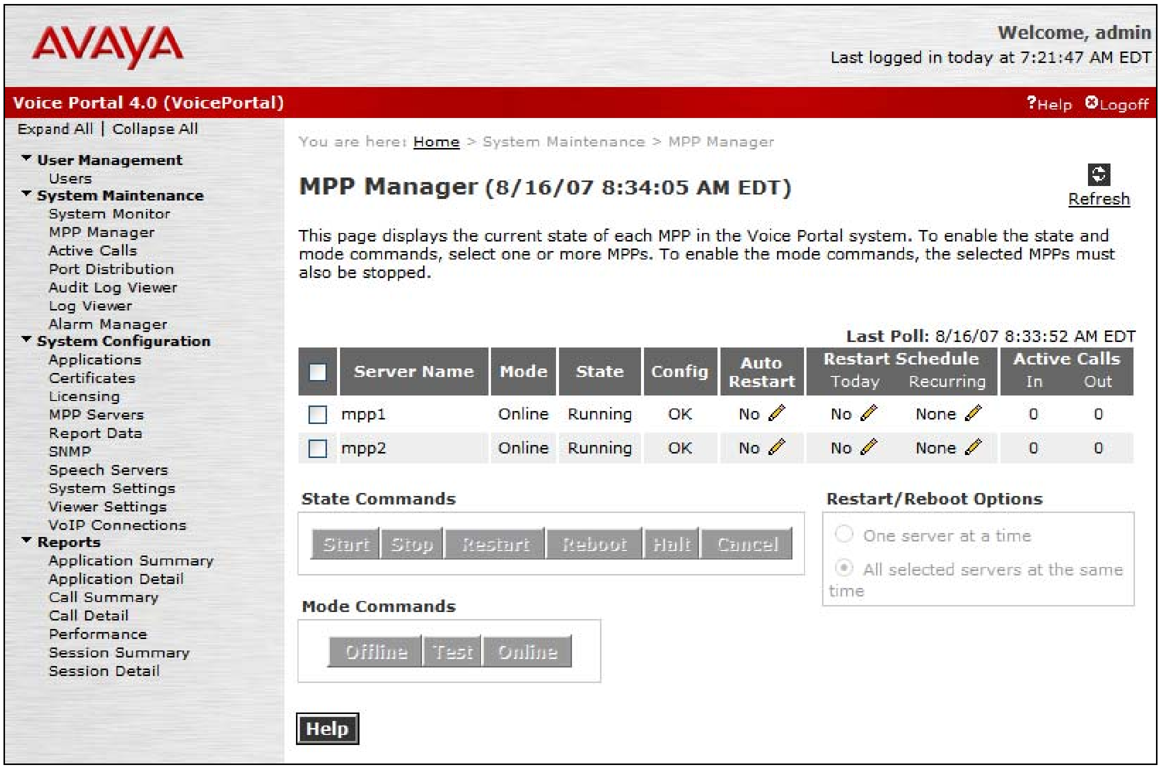


Figure 10: MPP Manager

4. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya SIP Enablement Services (SES) with a focus on configuring:

- **Trusted Certificates** for TLS authentication to Voice Portal
- a **Host Address Map** for routing calls to the PSTN
- a **Trusted Host** for the VG2000
- **Adjuncts** for Voice Portal
- **Application IDs** for the Voice Portal application

Basic configuration of Avaya SIP Enablement Services, including Users and Media Server Interfaces, are not included in these Application Notes. Refer to [3] for more information.

Avaya SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter http://<ip-addr>/admin as the URL in an Internet browser, where *<ip-addr>* is the IP address of Avaya SES. Log in with the appropriate credentials and select the Launch Administration Web Interface link. To install the certificates for TLS authentication, navigate to the **Trusted Certificates** page and click the **Import** button and specify the path to the certificate. The first two certificates shown in **Figure 11** are used for Voice Portal.

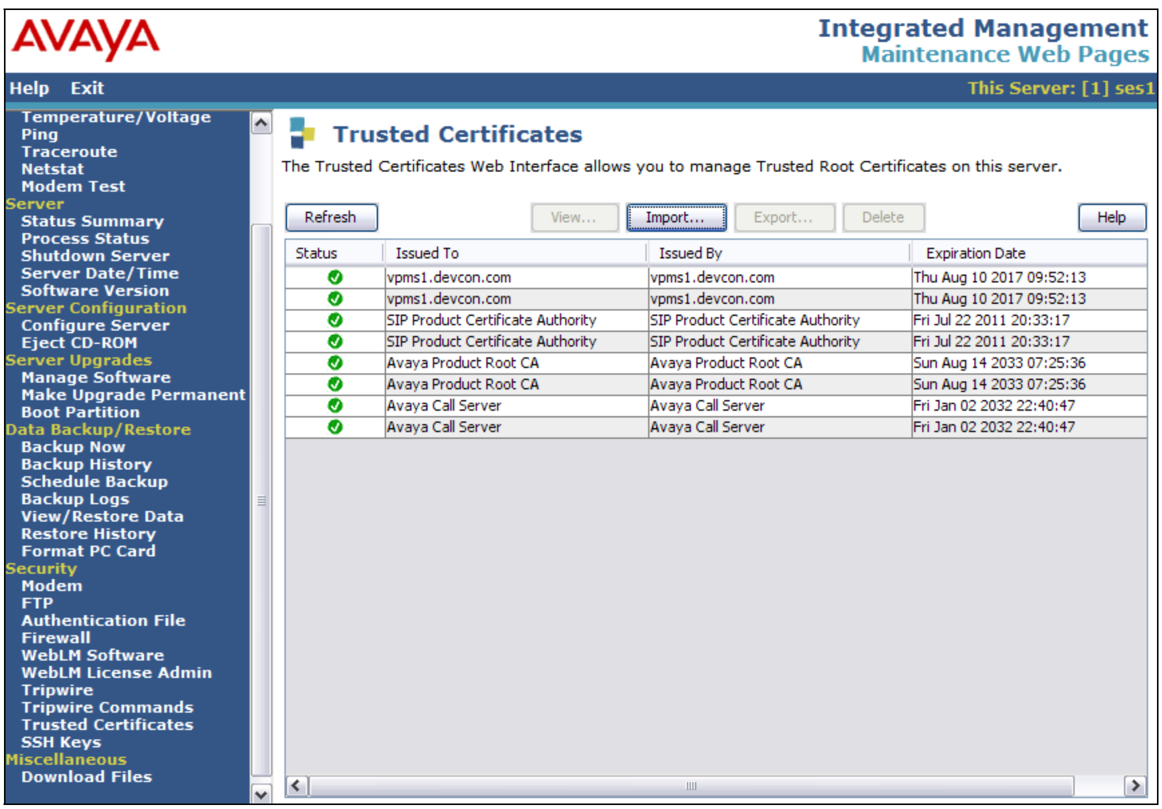


Figure 11: Trusted Certificates

Note: All the configuration screens in this section were captured after they were already configured. Configuration data is displayed in *edit* screens instead of *add* screens.

Add a Host Address Map. To route calls from the Avaya SIP-based network to the PSTN, 7-digit numbers beginning with '538' were dialed. In order for the SIP calls to be routed to the PSTN through the VG2000, a **Host Address Map** is required on Avaya SES. In this case, the Host Map Entry screen should appear as shown in **Figure 12**. The **Pattern** field in the Host Map Entry was set to `^sip:538[0-9]{4}`. The **Host Contact** specified the IP address of the VG2000 and the transport protocol and port.

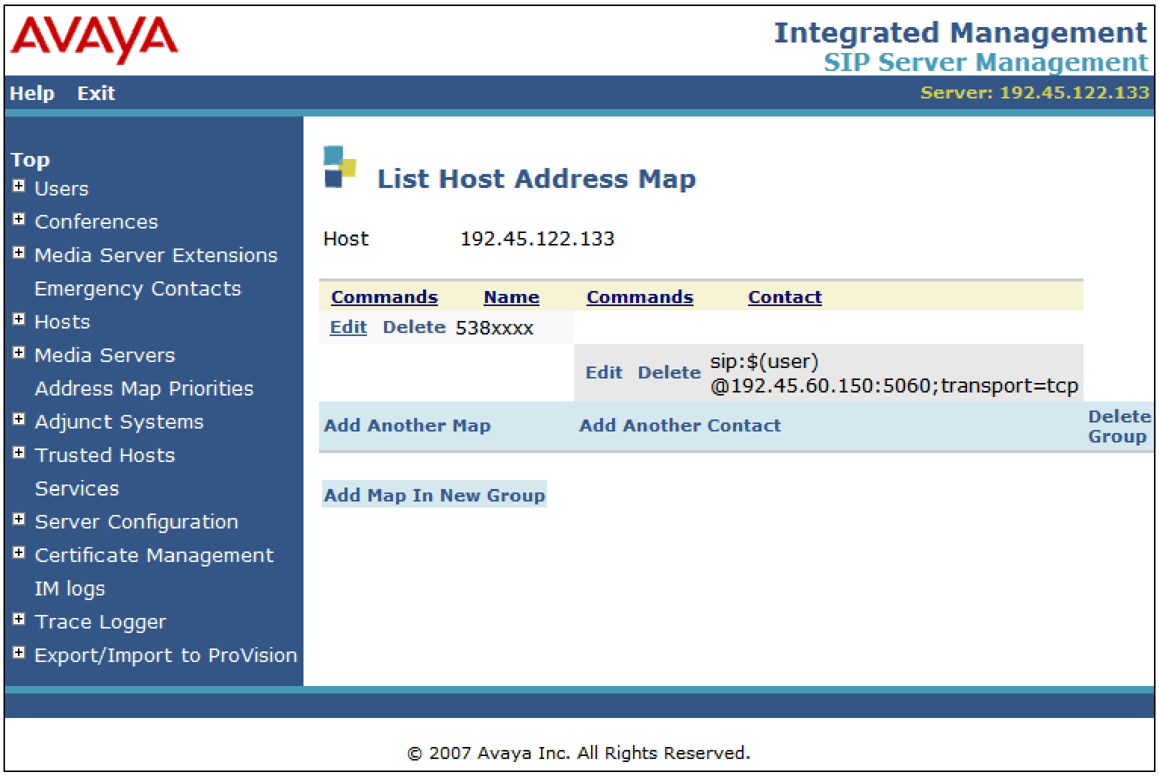


Figure 12: Host Address Map

To route calls from the PSTN to Avaya Voice Portal, 7-digit numbers beginning with '852' were received by Avaya SES. These calls were routed directly from Avaya SES to Avaya Voice Portal because **Application IDs** were configured in Avaya SES that matched the incoming number. No address maps were required. Application IDs are shown in **Figure 9**.

Add a Trusted Host. Configure the Vision Media Gateway as a trusted host so that Avaya SES does not challenge its incoming SIP requests. Navigate to the **Trusted Host** page and configure the IP address of the VG2000 and provide a descriptive comment. The **Trusted Host** page is shown in **Figure 13**.

Figure 13: Trusted Host

Add Adjunct System. Provision an adjunct system for the VPMS. Provide a System Name and select the Avaya SES with which this adjunct system will be integrated. **Figure 14** displays the **Adjunct System** page configured for the VPMS.

Figure 14: Adjunct System

Add Adjunct Servers. Add an adjunct server, associated with the adjunct system configured in **Figure 14**, for each MPP server. Specify the **Server Name**, **Server ID**, **Link Type**, and the **Server IP Address**. The **Server ID** is the server's handle or extension used to directly call this adjunct server. In this configuration, TLS is enabled between the Voice Portal and the SES. **Figure 15** shows the configuration for one MPP server. Repeat this configuration for the other MPP server.

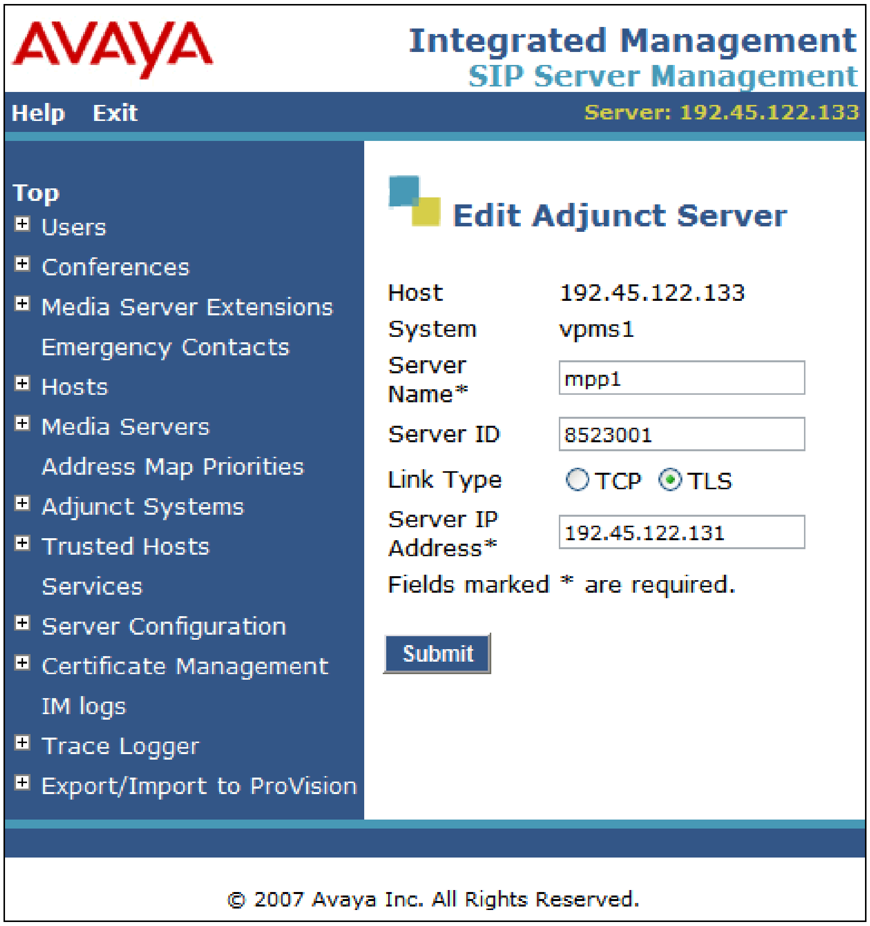


Figure 15: Adjunct Server

Add Applications IDs. Add **Application IDs** for the applications configured on the Voice Portal in **Figure 9**. This Application ID is associated with the adjunct system configured in **Figure 14**. When the called number of an incoming call matches the Application ID field in this form, the SES will route the call to the adjunct system. Figure 16 displays the Application ID form.

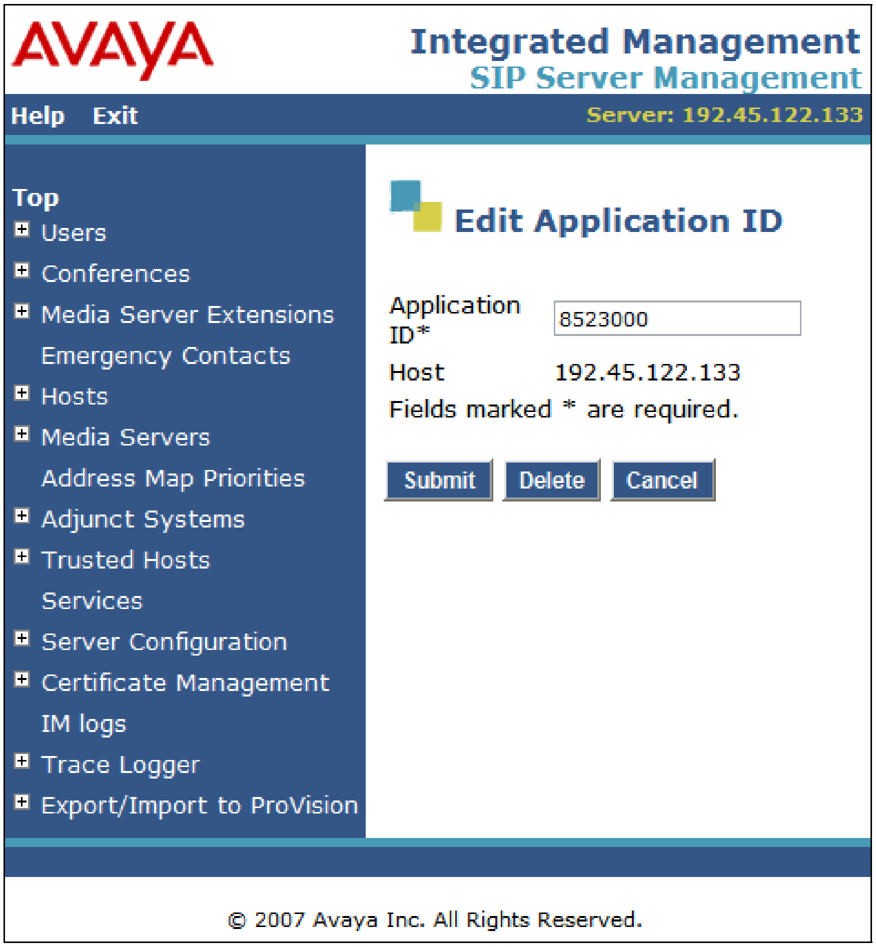


Figure 16: Application ID

5. Configure NMS Communications Vision Media Gateway

This section provides the procedures for configuring the Vision Media Gateway (VG2000). The procedures require two distinct operations:

- Configuring the Vision Media Gateway host IP network parameters
- Configuring the Vision Media Gateway telephony network interfaces

5.1. Configure the IP Network Parameters

The IP network parameters of the Vision Media Gateway are configured using standard Linux administration tools. Refer to [5] for instructions.

5.2. Configure Telephony Network Interfaces

To configure the telephony network interfaces of the Vision Media Gateway, access the Configuration Manager by launching Mozilla Firefox browser version 2.0 or later. Browse to the IP address of the Vision Media Gateway using port 8080 by specifying "http://<ip-address>:8080" in the URL. Log in using administrator credentials. The Vision Media Gateway welcome screen shown in **Figure 17** is displayed.

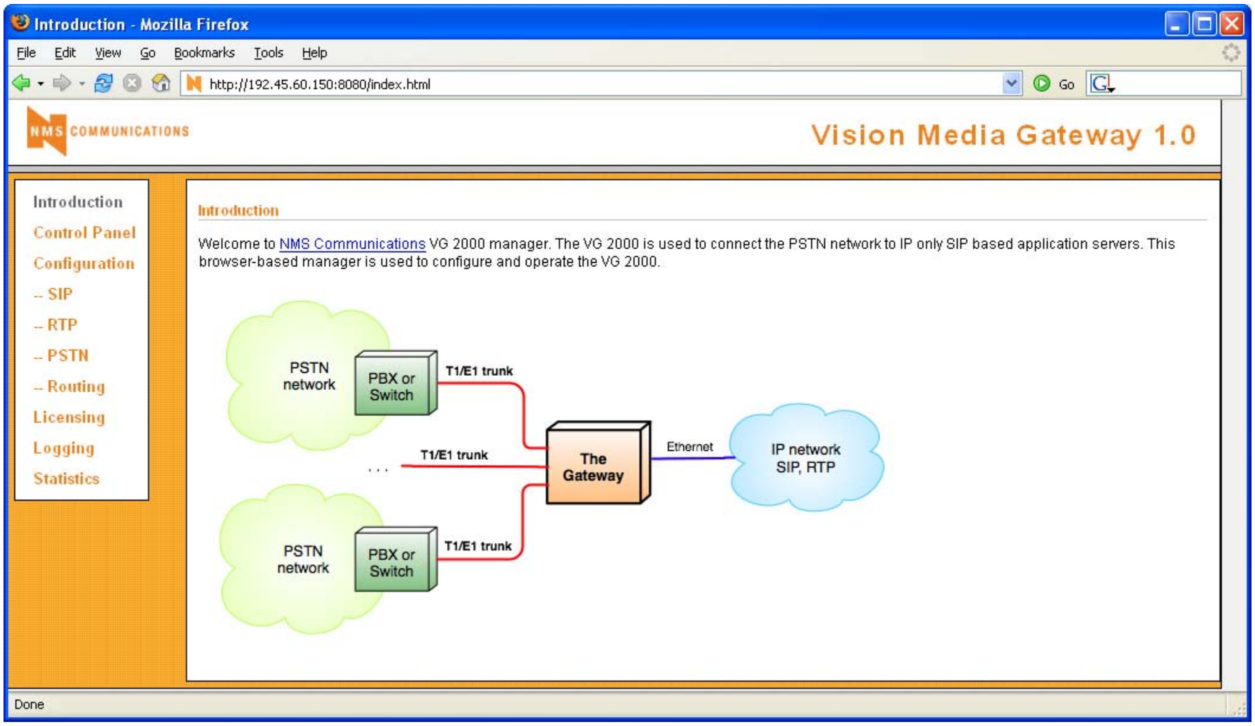


Figure 17: VG2000 Welcome Screen

Configure the SIP Interface. SIP messages are exchanged between the VG2000 and Avaya SES using the TCP transport protocol and Port 5060. Navigate to the **SIP Configuration** screen and configure the parameters as shown in **Figure 18**. Under the **General** section, specify the IP address of the VG2000 in the IP Address field, set the **Port** field to *5060*, and enable TCP support. Under the **Outbound Proxy** section, set the **Transport** field to *TCP*, configure the **Host or IP Address** field to the IP address of Avaya SES, and set the **Port** field to *5060*. Click **Save All**.

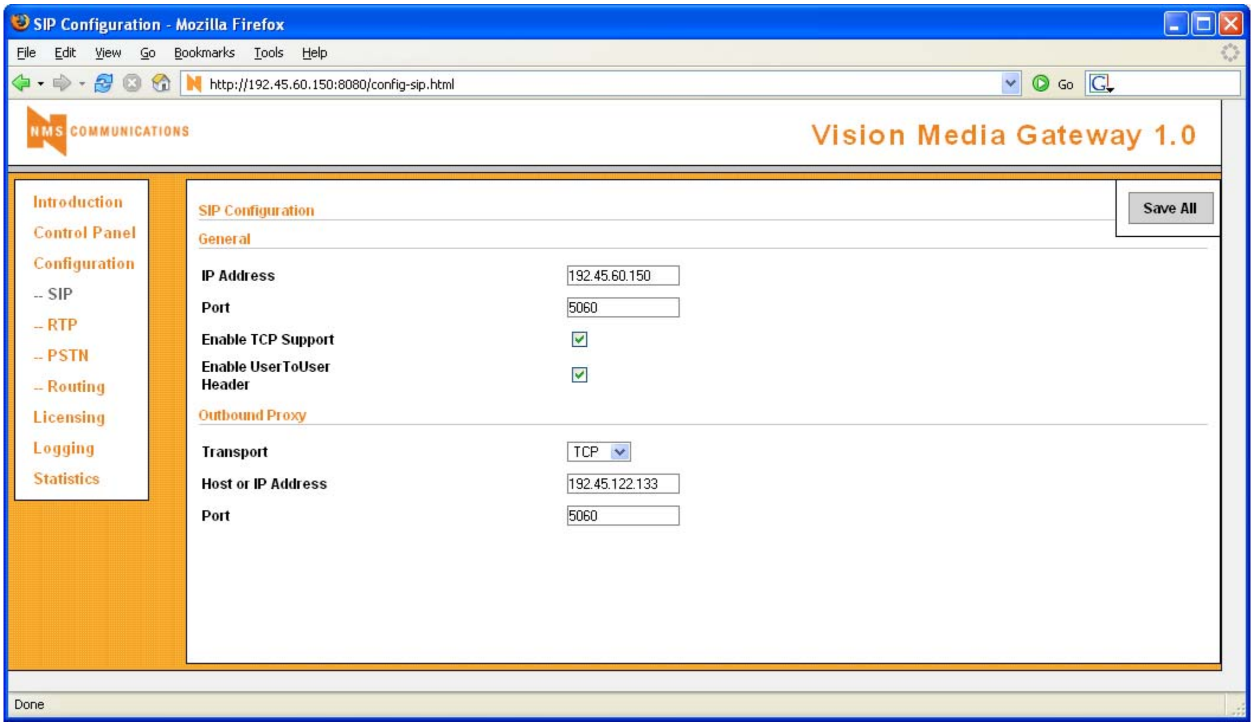


Figure 18: SIP Configuration

Configure the RTP Interface. Navigate to the **RTP Configuration** screen. Provide the IP network configuration associated with the RTP interface of the Vision Media Gateway. Accept the default **Base Port** of *8000* and enable **RFC 2833 support**. Click **Save All**.

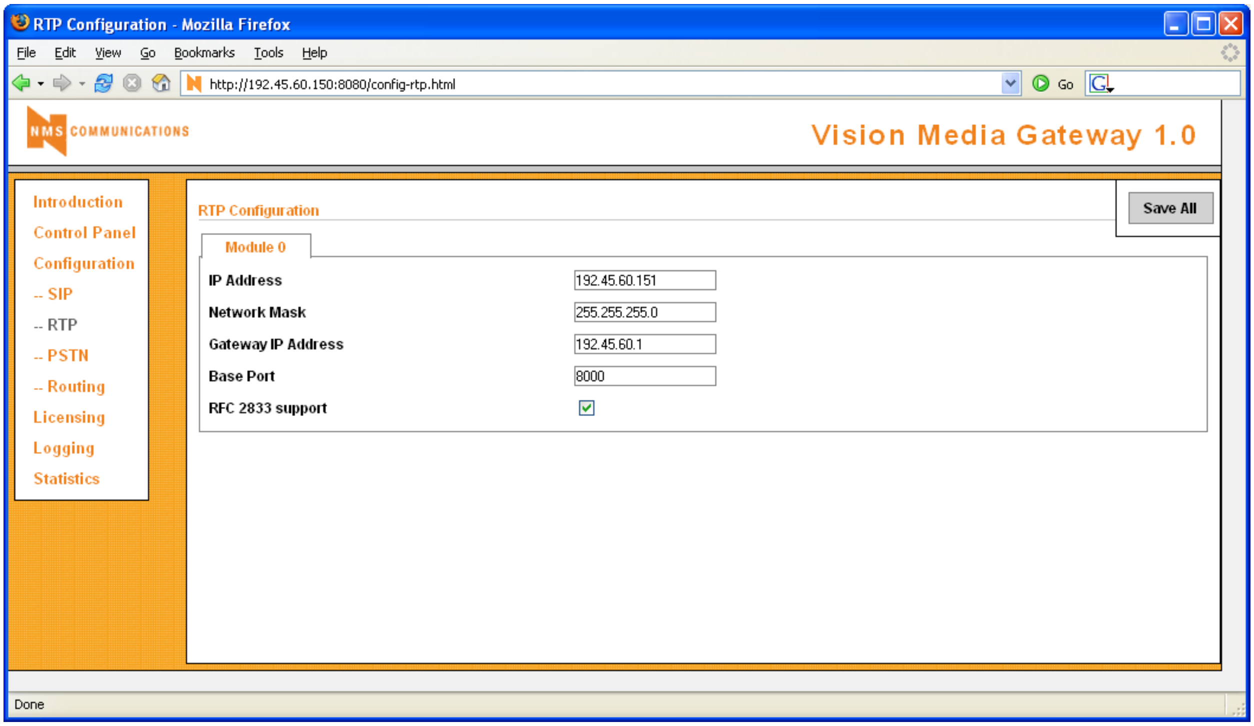


Figure 19: RTP Configuration

Configure the SS7 Interface. Once the SIP and RTP interfaces are configured, configure the SS7 interface to the PSTN. In the **PSTN Configuration** screen, configure the SS7 interface as shown in **Figure 20**. In this configuration, the trunk parameters were configured for *E1* with *HDB3* line coding and the *CEPT* frame type. The SS7 interface used the *ISUP* signaling protocol. The **Circuit Mask** specifies which voice circuits of the interface to use. For example, *0x7FFFFFF* enables all 31 voice circuits on the E1 interface. Click **Save All**.

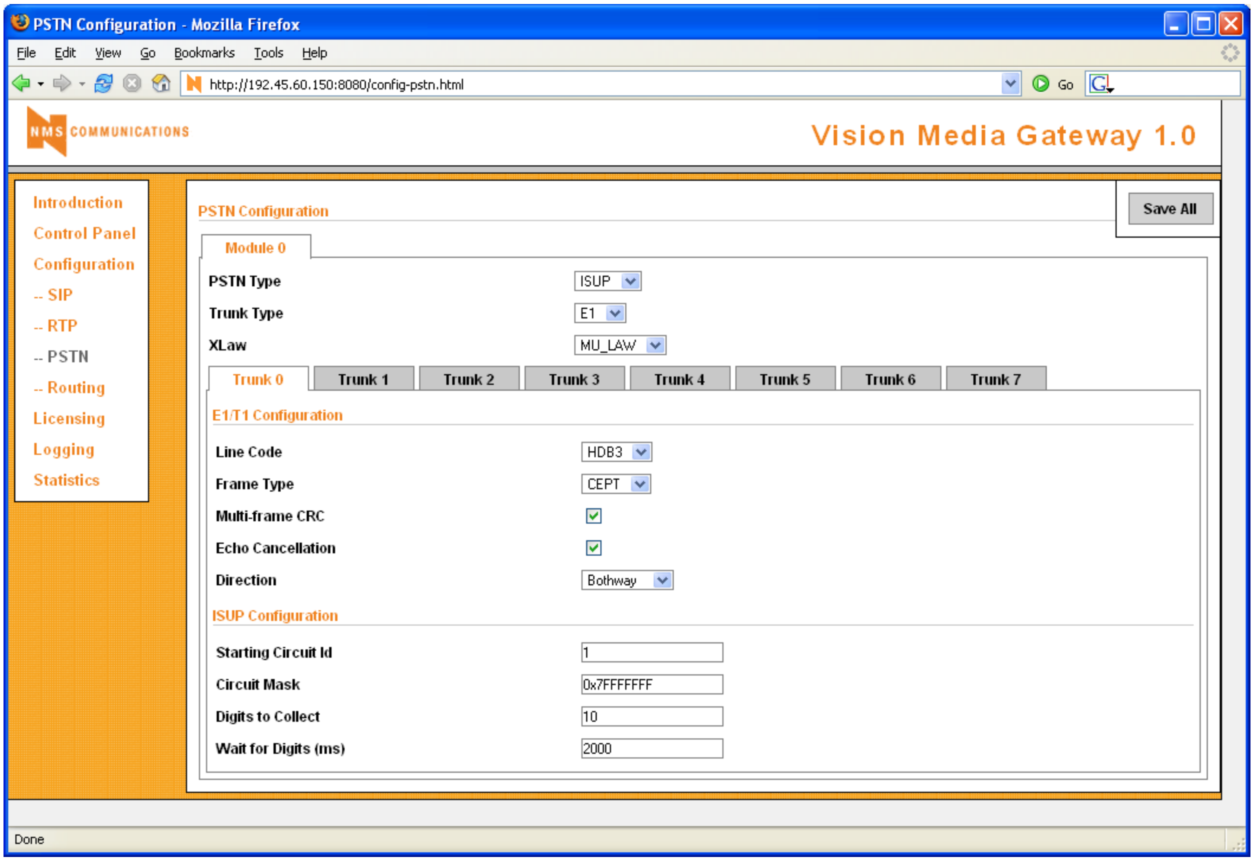


Figure 20: PSTN Configuration

Configure Call Routing. Use the Vision Media Gateway **Routing Configuration** screen to create a set of routing rules for the gateway. Routing table entries include from and to (or calling and called) patterns that are matched against the calling and called addresses received for an incoming call. For each matched pattern set, the routing table specifies the outgoing call mode (for example, SIP to PSTN or PSTN to SIP) and the substitution pattern for constructing the outgoing from and to addresses from the incoming addresses for that call. Separate routing rules are also required for transferred calls. Patterns are specified as JavaScript regular expressions.

In this configuration, two routes were used to route calls between the SIP and PSTN networks and two routes were used to route transferred calls. Below is a brief description of each call route. When multiple routes exist with the same Mode value, the order of the routes is important. The routes with the lower IDs take precedence over the ones with a higher ID.

- **Routing ID 1**: Routes calls from the PSTN to the SIP-based network (i.e., **Mode** is set to *pstn→sip*). In this configuration, the VG2000 received 10-digit dial strings, stripped off the area code, and sent the resulting dial string to Avaya SES in the SIP signaling.
- **Routing ID 2**: Routes calls from the SIP-based network to the PSTN (i.e., **Mode** is set to *sip→pstn*). In this configuration, 7-digit numbers beginning with '538' were passed to the VG2000. The VG2000 prepended the 732 area code to the dial string and sent the 10-digit number to the PSTN over the SS7 interface.
- **Routing IDs 3 and 4**: These routes are used for routing transferred calls to the PSTN and SIP networks.

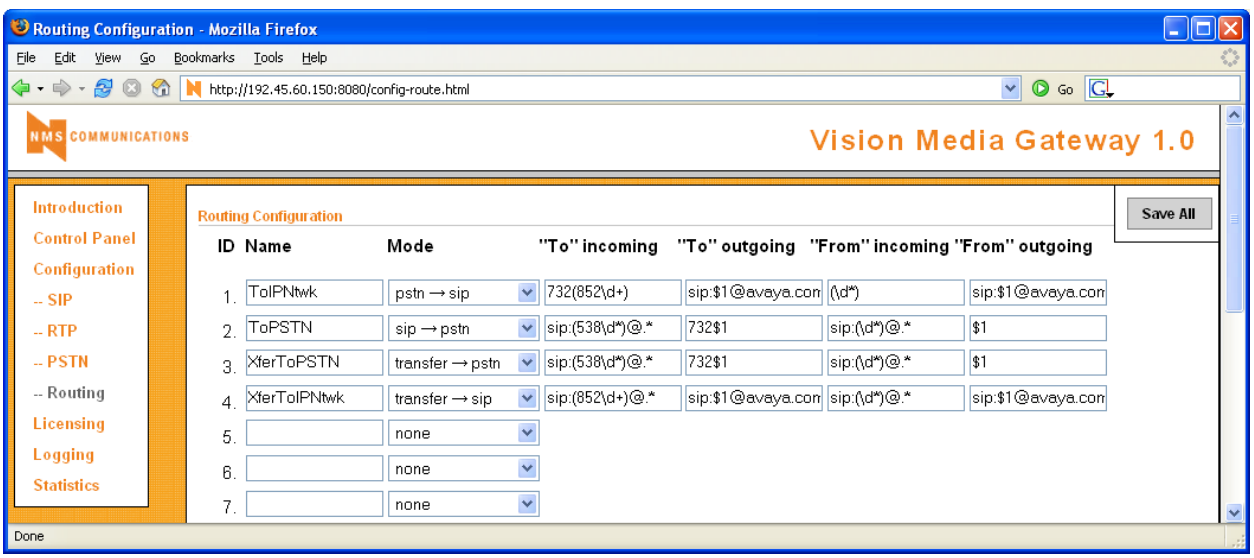


Figure 21: Routing Configuration

6. Configure NMS Communications Vision Signaling Server

This section provides the procedures for configuring the Vision Signaling Server (VS5000). The procedures require two distinct operations:

- Configuring the Vision Signaling Server IP network parameters
- Configuring the Vision Signaling Server SS7 network interface

6.1. Configure the IP Network Parameters

The Visions Signaling Server IP network configuration is managed by scripts and the procedures is described in [7].

6.2. Configure SS7 Interface

The Vision Signaling Server network configuration is based on two configuration files:

- `txcfg1.txt`: TDM configuration file, which defines the physical characteristics of the T1/E1 trunks.
- `ss7_config_default.xml`: SS7 configuration, which defines the SS7 network configuration including signaling links, link sets, routes, and circuit groups.

The configuration files used during testing are contained in the appendix of this document. Refer to [7] for information on how to configure these files.

7. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify calls between Avaya Voice Portal and the PSTN through the NMS Communications Vision Media Gateway, which served as a SIP-to-PSTN gateway. This section covers the general test approach and the test results.

7.1. General Test Approach

The interoperability compliance test included feature, serviceability, and performance load testing. The feature testing focused on verifying the following:

- Placing calls from the Avaya SIP-based network to the PSTN
- Placing calls from the PSTN to Avaya Voice Portal
- Call transfers from Voice Portal to users on the PSTN and the Avaya SIP-based network
- Performing Blind, Supervised, and Bridged call transfers from Voice Portal to users on the PSTN and Avaya SIP-based network
- Sending UUI during call transfers from Voice Portal to the PSTN
- Receiving UUI from the PSTN to Voice Portal

The serviceability testing focused on verifying the ability of the VG2000 and VS5000 to recover from adverse conditions, such as power failures and disconnecting cables to the IP network.

The performance load testing was accomplished by repeatedly establishing 240 calls through the Vision Media Gateway to Voice Portal for an extended period of time.

7.2. Test Results

All test cases passed. Avaya Voice Portal was successful in establishing and transferring calls to users on the PSTN through the NMS Communications Vision Media Gateway. However, it was noted during testing that the VS5000 needs to be powered up manually after a power outage.

8. Verification Steps

This section provides the verification steps that may be performed to verify that the Voice Portal can establish calls to the PSTN through the Vision Media Gateway.

1. From the VPMS web interface, verify that the MPP servers are online and running in the **System Monitor** page shown in **Figure 22**.

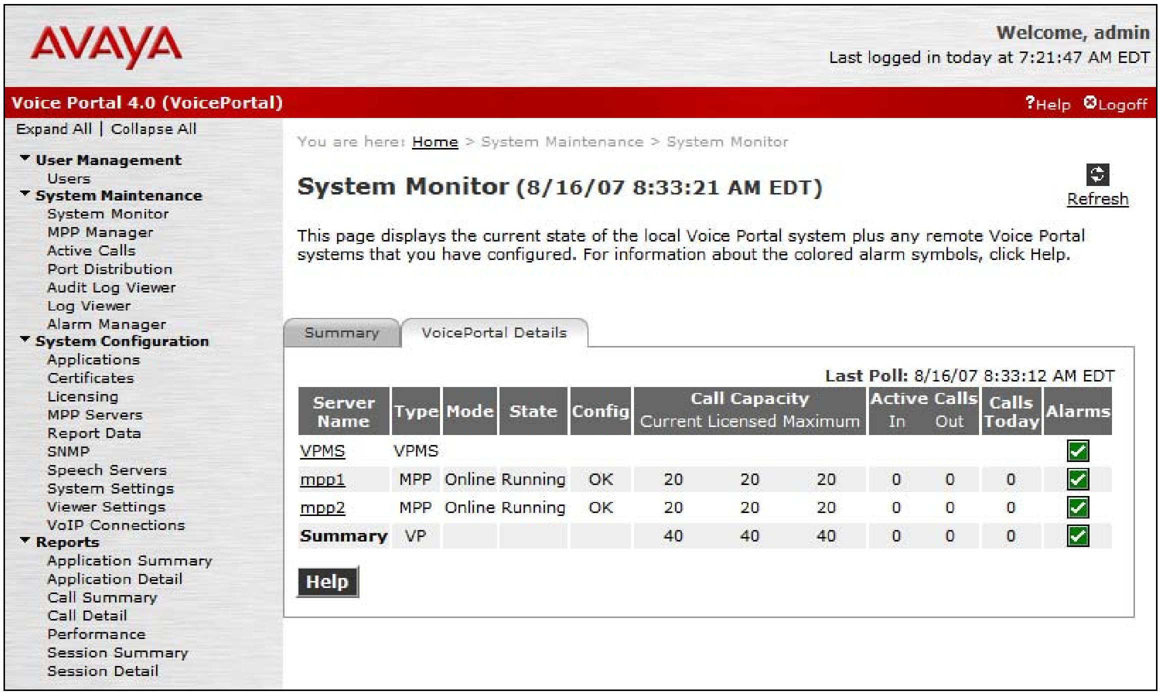
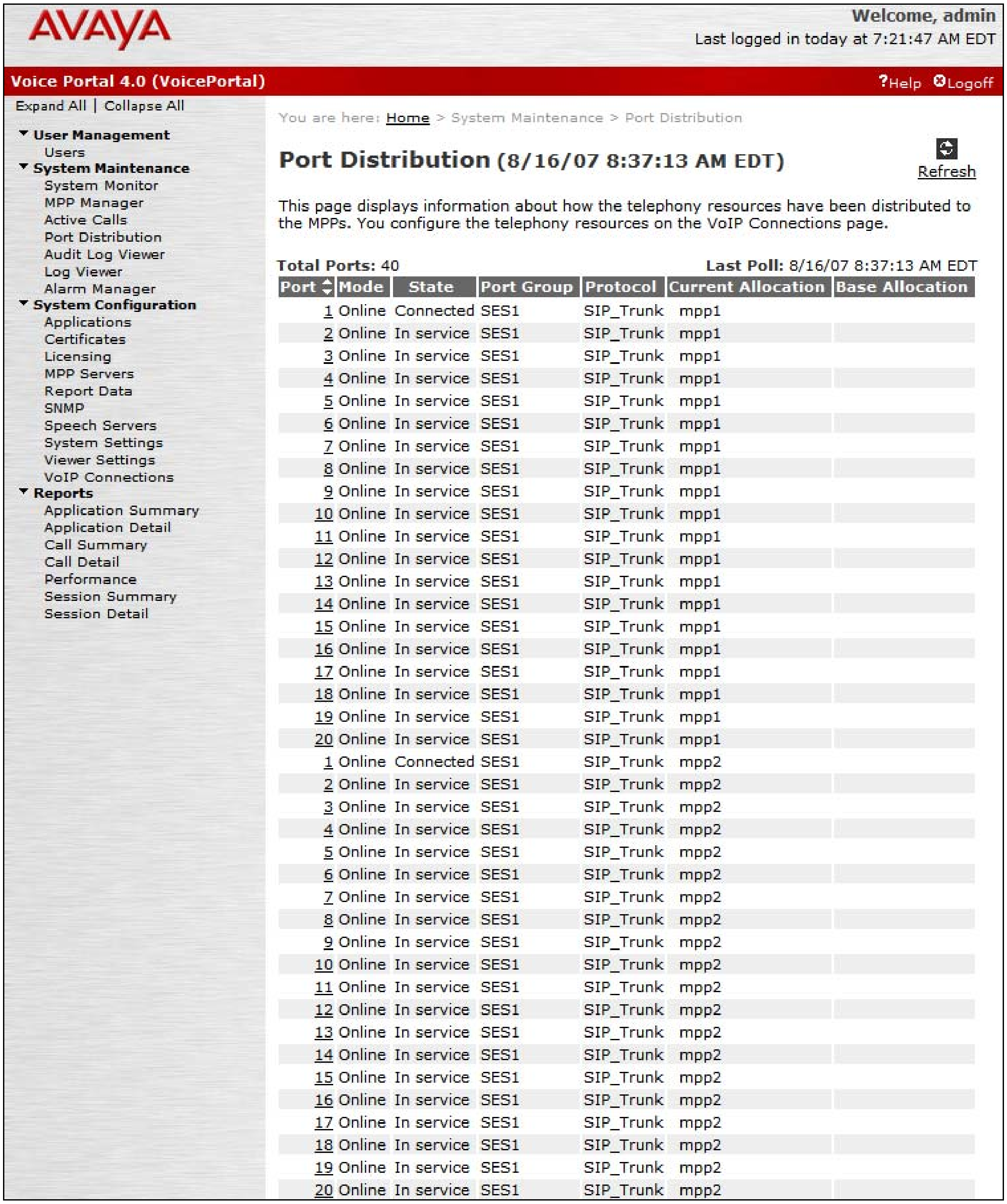


Figure 22: System Monitor

2. From the VPMS web interface, verify that the ports on the MPP servers are in-service in the **Port Distribution** page shown in Figure 23.



Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
1	Online	Connected	SES1	SIP_Trunk	mpp1	
2	Online	In service	SES1	SIP_Trunk	mpp1	
3	Online	In service	SES1	SIP_Trunk	mpp1	
4	Online	In service	SES1	SIP_Trunk	mpp1	
5	Online	In service	SES1	SIP_Trunk	mpp1	
6	Online	In service	SES1	SIP_Trunk	mpp1	
7	Online	In service	SES1	SIP_Trunk	mpp1	
8	Online	In service	SES1	SIP_Trunk	mpp1	
9	Online	In service	SES1	SIP_Trunk	mpp1	
10	Online	In service	SES1	SIP_Trunk	mpp1	
11	Online	In service	SES1	SIP_Trunk	mpp1	
12	Online	In service	SES1	SIP_Trunk	mpp1	
13	Online	In service	SES1	SIP_Trunk	mpp1	
14	Online	In service	SES1	SIP_Trunk	mpp1	
15	Online	In service	SES1	SIP_Trunk	mpp1	
16	Online	In service	SES1	SIP_Trunk	mpp1	
17	Online	In service	SES1	SIP_Trunk	mpp1	
18	Online	In service	SES1	SIP_Trunk	mpp1	
19	Online	In service	SES1	SIP_Trunk	mpp1	
20	Online	In service	SES1	SIP_Trunk	mpp1	
1	Online	Connected	SES1	SIP_Trunk	mpp2	
2	Online	In service	SES1	SIP_Trunk	mpp2	
3	Online	In service	SES1	SIP_Trunk	mpp2	
4	Online	In service	SES1	SIP_Trunk	mpp2	
5	Online	In service	SES1	SIP_Trunk	mpp2	
6	Online	In service	SES1	SIP_Trunk	mpp2	
7	Online	In service	SES1	SIP_Trunk	mpp2	
8	Online	In service	SES1	SIP_Trunk	mpp2	
9	Online	In service	SES1	SIP_Trunk	mpp2	
10	Online	In service	SES1	SIP_Trunk	mpp2	
11	Online	In service	SES1	SIP_Trunk	mpp2	
12	Online	In service	SES1	SIP_Trunk	mpp2	
13	Online	In service	SES1	SIP_Trunk	mpp2	
14	Online	In service	SES1	SIP_Trunk	mpp2	
15	Online	In service	SES1	SIP_Trunk	mpp2	
16	Online	In service	SES1	SIP_Trunk	mpp2	
17	Online	In service	SES1	SIP_Trunk	mpp2	
18	Online	In service	SES1	SIP_Trunk	mpp2	
19	Online	In service	SES1	SIP_Trunk	mpp2	
20	Online	In service	SES1	SIP_Trunk	mpp2	

Figure 23: Port Distribution

3. From a Linux terminal on the Vision Signaling Server, verify that the physical connectivity is established using the commands specified in **Figure 24**.

```
root@mgw:~
File  Edit  View  Terminal  Tabs  Help
[13:17:51] vs5000:/> cpcon
Console program V3.2 : [Hit Return to Activate, 'quit' to Exit]
                       [For multi-screen reply, 'more' to scroll]
Copyright 1996-2005: NMS Communications

(1) [con] >frstatus

Trunk    State            RX Alarms       TX Alarms          Loop     B C  Testing
=====    =============    =============   ================   ===========   =========
   1     SYNC             NO ALARMS       NO ALARMS                        NONE
   2     SYNC             NO ALARMS       NO ALARMS                        NONE
   3     NO CONNECT           R           NO ALARMS                        NONE
   4     NO CONNECT           R           NO ALARMS                        NONE

(1) [con] >
```

Figure 24: Frame Status

4. From a Linux terminal on the Vision Signaling Server, verify that the SS7 interface is in-service using the commands specified in **Figure 25**.

```
root@mgw:~
File  Edit  View  Terminal  Tabs  Help

[11:30:22] vs5000:/opt/hs-data/raid/nms_hearsay/cfg/oam> ss7cli

ss7cli V1.0 - Copyright 2006: NMS Communications


[ss7cli] > status link *

Link  L3 State    L2 State                Inh  Blk  Congested  L2FlowCtlLvl
----  ----------  ----------------------  ---  ---  ---------  ------------
 0    Active      In Service                        false      0
 1    Active      In Service                        false      0

[ss7cli] > status route *
                                                   Num of    Num of
Rte  DPC          OPC          RteState     Cong   Cong LS   Act LS   Adj SP Rst
---  -----------  -----------  -----------  -----  --------  ------   ----------
  2  2            1            Available    false     0         1        true

[ss7cli] >
```

Figure 25: Link Status

5. From the Configuration Manager of the Vision Media Gateway, navigate to the **Control Panel** and verify that SS7 interface and channels are in-service as shown in **Figure 26**. In this example, four channels are in-service. The checkboxes are enabled so that the system comes up automatically after rebooting or cycling power.

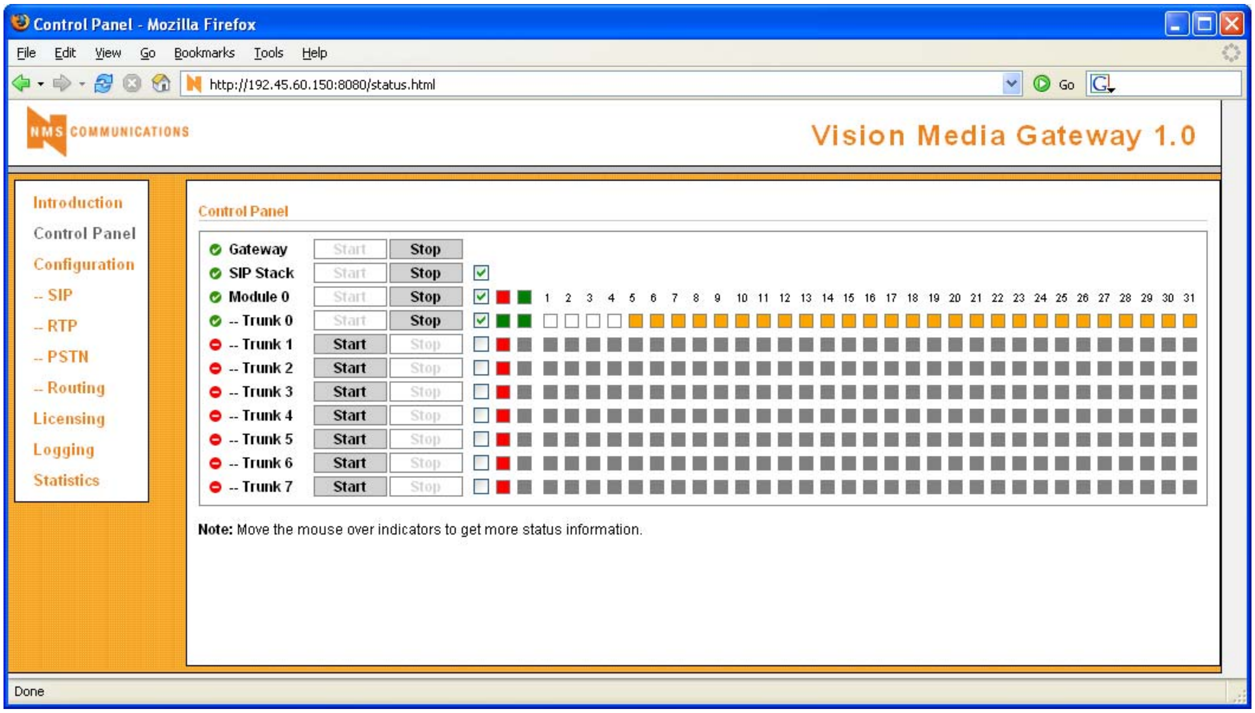


Figure 26: Control Panel

6. From a phone on the PSTN, place a call to an application on Voice Portal. Verify that the call is established successfully and that the proper greeting is provided. Transfer the call to another user.

9. Support

NMS Communications Technical support is available by phone, web, and Email.

- **Phone:** (508) 271-1333
- **Web:** http://www.nmscommunications.com/Support
- **Email:** tech_support@nmss.com

10. Conclusion

These Application Notes describe the configuration steps required so that the NMS Communications Vision Media Gateway can allow calls between Avaya Voice Portal and the PSTN through a SIP infrastructure. All feature, serviceability, and performance test cases were completed successfully.

11. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Document 03-300509, Issue 3.1, February 2007, available at http://support.avaya.com.

[2] *Feature Description and Implementation for Avaya Communication Manager*, Document 555-245-205, Issue 5, February 2007, available at http://support.avaya.com.

[3] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series and S8700 series Media Server*, Document 555-245-206, Issue 2007, May 2007, available at http://support.avaya.com.

[4] *Installing the VG 2000 Gateway*, Version 1.0, July 2007, Document ID 9000-62494-10, available from NMS Communications.

[5] *Vision Media Gateway Configuration and Administration Manual*, Document ID 9000-62701-10, Version 1.0, July 2007, available from NMS Communications.

[6] *Installing the Vision VS 5000 Signaling Server*, Document ID 9000-62672-11, Version 1.1, June 2007, available from NMS Communications.

[7] *Vision Signaling Server Configuration and Administration Manual*, Document ID 9000-62666-11, Version 1.1, July 2007, available from NMS Communications.

APPENDIX: Vision Signaling Server Configuration Files

A sample TDM configuration file, `txcfg1.txt,` is displayed below.

```
################################################################################
#  NMS Communications TX Series E1/T1/J1 and H.100 / H.110 Configuration File
#
#  ==============================================================================
#  General Information:
#  ==============================================================================
#  This file contains sample configuration information to help you set up
#  communication between E1/T1/J1/H100/LOCAL channels on the TX Board.
#  A separate instance of this file is usually generated
#  for each TX Series board that is installed in your system.
#
#  Both H.100 and H.110 are treated identically in this file.
#
#  This configuration file is provided as an input file to the TXCONFIG
#  utility.  The TXCONFIG commands that relate to initial TX board
#  configuration are described below.  Additional commands are also provided
#  by TXCONFIG.  These commands are used to perform other [non-configuration]
#  related functions.  Enter '?' to the TXCONFIG utility for further details.
################################################################################

################################################################################
#  HearSay/MyCaller CONFIGURATION:
#  ==============================================================================
#  Configure as clock master with trunk 1 as primary clock source and no
#  fallback clock.
################################################################################

#-------------------------------------------------------------------------------
clock net=1 a

#-------------------------------------------------------------------------------
#     Trunk     Framing   Encoding   Loop Master
e1cfg 1         ccs       hdb3       false
e1cfg 2         ccs       hdb3       false
e1cfg 3         ccs       hdb3       false
e1cfg 4         ccs       hdb3       false

#-------------------------------------------------------------------------------
#    PortNum    L|H|E|T|J   Trunk   Channel     Speed
port 1          e1          1       16
port 2          e1          2       16
port 3          e1          3       16
port 4          e1          4       16
```

A sample SS7 configuration file, `ss7_config_default.xml`, is displayed below.


```xml
<Properties xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
    xsi:noNamespaceSchemaLocation="ss7_config.xsd">
  <Version>1.0</Version>
  <AutoStart>Yes</AutoStart>
  <State>OutOfService</State>
  <Config>
   <MtpConfig>
     <GenConfig>
        <OPC>1</OPC>
        <DefaultVariant>ITU</DefaultVariant>
        <DefaultDPC>2</DefaultDPC>
        <NodeType>SP</NodeType>
        <MaxLinks>2</MaxLinks>
        <MaxUsers>2</MaxUsers>
        <MaxLinksets>2</MaxLinksets>
        <MaxRoutes>32</MaxRoutes>
        <RestartRequired>false</RestartRequired>
     </GenConfig>
     <NsapConfig Index="1">
         <!--  Nsap #1 used by ISUP   -->
     </NsapConfig>
     <NsapConfig Index="2">
         <!--  Nsap #2 reserved for future use by SCCP or other MTP user -->
     </NsapConfig>
     <LinkConfig Index="1">
        <!--  Link #1: first link to DPC 2, terminates on SS701 server  -->
        <Server>SS701</Server>
        <PortNumber>1</PortNumber>
        <LinkSLC>0</LinkSLC>
        <Linkset>1</Linkset>
        <Ssf>National</Ssf>
     </LinkConfig>
     <LinkConfig Index="2">
        <!--  Link #1: second link to DPC 2, terminates on SS702 server  -->
        <Server>SS701</Server>
        <PortNumber>2</PortNumber>
        <LinkSLC>1</LinkSLC>
        <Linkset>1</Linkset>
        <Ssf>National</Ssf>
     </LinkConfig>
     <LinksetConfig Index="1">
        <!--  Linkset #1: direct link set to DPC 2 (DefaultDpc)  -->
        <TargetNmbActLinks>2</TargetNmbActLinks>
        <Route Index="1">
          <RouteNumber>2</RouteNumber>
          <Priority>0</Priority>
        </Route>
```

```xml
  &lt;/LinksetConfig&gt;
  &lt;RouteConfig Index="1"&gt;
      <!-- NOTE: This first route is the UP (inbound) route, therefore its DPC is the local point
 code -->
      &lt;DPC&gt;1&lt;/DPC&gt;
      &lt;Ssf&gt;National&lt;/Ssf&gt;
      &lt;Direction&gt;Up&lt;/Direction&gt;
  &lt;/RouteConfig&gt;
  &lt;RouteConfig Index="2"&gt;
      &lt;!-- Direct route to attached DPC 2, via link set #1 --&gt;
      &lt;DPC&gt;2&lt;/DPC&gt;
      &lt;Ssf&gt;National&lt;/Ssf&gt;
      &lt;Direction&gt;Down&lt;/Direction&gt;
      &lt;AdjRoute&gt;true&lt;/AdjRoute&gt;
  &lt;/RouteConfig&gt;
&lt;/MtpConfig&gt;
&lt;IsupConfig&gt;
  &lt;GenConfig&gt;
      &lt;DefaultVariant&gt;ITUWHITE&lt;/DefaultVariant&gt;
      &lt;MaxCircuits&gt;256&lt;/MaxCircuits&gt;
      &lt;MaxGroups&gt;32&lt;/MaxGroups&gt;
      &lt;MaxCallRefs&gt;256&lt;/MaxCallRefs&gt;
      &lt;ExtElmts&gt;true&lt;/ExtElmts&gt;
  &lt;/GenConfig&gt;
  &lt;CircConfig Index="1"&gt;
     &lt;!-- 1st voice E1 to attached DPC 2 --&gt;
     &lt;Circuit&gt;1&lt;/Circuit&gt;
     &lt;CIC&gt;1&lt;/CIC&gt;
     &lt;NumCircuits&gt;31&lt;/NumCircuits&gt;
     &lt;Direction&gt;Bothway&lt;/Direction&gt;
     &lt;UnusedCircuits&gt;None&lt;/UnusedCircuits&gt;
     &lt;Ssf&gt;National&lt;/Ssf&gt;
  &lt;/CircConfig&gt;
  &lt;CircConfig Index="2"&gt;
     &lt;!-- 2nd voice E1 to attached DPC 2 --&gt;
     &lt;Circuit&gt;33&lt;/Circuit&gt;
     &lt;CIC&gt;33&lt;/CIC&gt;
     &lt;NumCircuits&gt;31&lt;/NumCircuits&gt;
     &lt;Direction&gt;Bothway&lt;/Direction&gt;
     &lt;UnusedCircuits&gt;None&lt;/UnusedCircuits&gt;
     &lt;Ssf&gt;National&lt;/Ssf&gt;
  &lt;/CircConfig&gt;
  &lt;CircConfig Index="3"&gt;
     &lt;!-- 3rd voice E1 to attached DPC 2 --&gt;
     &lt;Circuit&gt;65&lt;/Circuit&gt;
     &lt;CIC&gt;65&lt;/CIC&gt;
     &lt;NumCircuits&gt;31&lt;/NumCircuits&gt;
     &lt;Direction&gt;Bothway&lt;/Direction&gt;
```

```
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
<CircConfig Index="4">
    <!-- 4th voice E1 to attached DPC 2 -->
    <Circuit>97</Circuit>
    <CIC>97</CIC>
    <NumCircuits>31</NumCircuits>
    <Direction>Bothway</Direction>
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
<CircConfig Index="5">
    <!-- 5th voice E1 to attached DPC 2 -->
    <Circuit>129</Circuit>
    <CIC>129</CIC>
    <NumCircuits>31</NumCircuits>
    <Direction>Bothway</Direction>
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
<CircConfig Index="6">
    <!-- 6th voice E1 to attached DPC 2 -->
    <Circuit>161</Circuit>
    <CIC>161</CIC>
    <NumCircuits>31</NumCircuits>
    <Direction>Bothway</Direction>
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
<CircConfig Index="7">
    <!-- 7th voice E1 to attached DPC 2 -->
    <Circuit>193</Circuit>
    <CIC>193</CIC>
    <NumCircuits>31</NumCircuits>
    <Direction>Bothway</Direction>
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
<CircConfig Index="8">
    <!-- 8th voice E1 to attached DPC 2 -->
    <Circuit>225</Circuit>
    <CIC>225</CIC>
    <NumCircuits>31</NumCircuits>
    <Direction>Bothway</Direction>
    <UnusedCircuits>None</UnusedCircuits>
    <Ssf>National</Ssf>
</CircConfig>
```

```
    <UsapConfig Index="1">
        <!-- Only 1 ISUP user, the ssp_server -->
    </UsapConfig>
    <NsapConfig Index="1">
        <!-- Matches the MTP Nsap reserved for use by ISUP -->
        <Ssf>National</Ssf>
    </NsapConfig>
  </IsupConfig>
  <SspConfig>
    <GenConfig>
       <Server1>SS701</Server1>
       <Server2>SS702</Server2>
    </GenConfig>
  </SspConfig>
  </Config>
</Properties>
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.