Avaya Solution & Interoperability Test Lab

Configuring a Survivable SIP Gateway Solution using the Avaya Secure Router SR2330 10.2.1, Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0, and Avaya Modular Messaging 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration of a Survivable SIP Gateway Solution using the Avaya Secure Router SR2330, Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0, and Avaya Modular Messaging 5.2 in both Central and Distributed Trunking configurations.

The Survivable SIP Gateway Solution addresses the risk of service disruption for SIP endpoints deployed at remote branch locations if connectivity to the centralized Avaya SIP call control platform located at the main site is lost. Connectivity loss can be caused by WAN access problems being experienced at the branch or by network problems at the centralized site blocking access to the Avaya SIP call control platform, or by Avaya Aura™ Session Manager going out of service. The solution monitors connectivity health from the remote branch to the centralized Avaya SIP call control platform. When connectivity loss is detected, Avaya one-X® Deskphone SIP 9600 Series IP Telephones as well as the Avaya Secure Router dynamically switch to survivability mode, restoring telephony services at the branch for intra-branch and PSTN calls.

The Avaya Secure Routers 2330 and 4134 support SIP gateway capability and SIP survivability and are intended for use as survivable SIP gateways and integrated branch routers. The results shown in this document were obtained using the SR2330 platform. The SR2330 and SR4134 share common software, interface modules and software licenses and the same results should be expected from the SR4134 platform.

Testing was conducted at the Avaya Solution and Interoperability Test Lab at the request of Avaya Unified Branch Product Management.

1. Introduction

These Application Notes present a sample configuration of a Survivable SIP Gateway Solution using the Avaya Secure Router SR2330, Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0 and Avaya Modular Messaging 5.2 in both Central and Distributed Trunking configurations.

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The Avaya Secure Routers 2330 and 4134 support SIP gateway capability and SIP survivability, and are intended for use as survivable SIP gateways and integrated branch routers. The results shown in this document were obtained using the SR2330 platform. The SR2330 and SR4134 share common software, interface modules, and software licenses, and the same results should be expected from the SR4134 platform.

2. Overview

2.1. Avaya Secure Router 2330

The Avaya Secure Router 2330, referred to as the SR2330 throughout the remainder of this document, takes on various roles based on call flows and network conditions. The following lists these roles:

- Branch router
- SIP Media Gateway (FXO interfaces to PSTN, FXS interfaces to analog endpoints)
- SIP Survivability Module (Registrar and Proxy, dynamically activated on detection of lost connectivity to Session Manager)

When the SR2330 is serving the Registrar/Proxy role, it is said to be in *Survivable Mode*.

2.2. Avaya one-X® Deskphone SIP 9600 Series IP Telephones and Avaya one-X® Communicator

The one-X Deskphone SIP 9600 Series IP Telephone, referred to as Avaya 9600 SIP Phone in these Application Notes, is also a key component of the Survivable SIP Gateway Solution. The 6.0 firmware release of the Avaya 9600 SIP Phone includes feature capabilities specific to SIP survivability, enabling the phone to monitor connectivity to Session Manager and dynamically fail over to the local SR2330 as an alternate or survivable SIP server. See **Reference [7]** for additional information on the Avaya 9600 SIP Phone.

The one-X Communicator is a SIP soft phone running on Windows PC. Like the Avaya 9600 SIP Phones, the one-X Communicator Release 6.0 supports SIP survivability.

2.3. Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager

Session Manger is a routing hub for SIP calls among connected SIP telephony systems. The System Manager provides management functions for the Session Manager. In the sample configuration, SIP trunks link the Session Manager to the Communication Manager and Avaya Modular Messaging at the Headquarters and the SR2330 at the Branch site. The Session Manager also functions as the SIP registrar and proxy in the Aura™ network and as such provides SIP call control for all SIP phones registered with it.

In the sample configuration, the Communication Manager at the main Headquarters/Datacenter location operates as an Evolution Server (hereafter abbreviated ES). As such, it supports calling features for the SIP phones in the main location as well as in the branch locations (in normal mode); it also supports non-SIP telephones (H.323, digital and analog) and interworking between SIP and non-SIP telephones. Unless otherwise specified in these Application Notes, the term "Communication Manager" refers to the *Evolution Server*.

2.4. PSTN Trunking Configurations

The Survivable SIP Gateway Solution can interface with the PSTN in either a Centralized Trunking or a Distributed Trunking configuration. These trunking options determine how branch calls to and from the PSTN will be routed by Session Manager and Communication Manager over the corporate network.

Assuming an enterprise consisting of a main Headquarters/Datacenter location and multiple distributed branch locations all inter-connected over a corporate WAN, the following defines Centralized Trunking and Distributed Trunking as related to the Survivable SIP Gateway Solution:

Centralized Trunking: All PSTN calls, inbound to the enterprise and outbound from the enterprise, are routed to/from PSTN media gateways centrally located at the Headquarters/Datacenter location.

Distributed Trunking: PSTN call routing can be determined based on the originating source location using Location Based Routing on Session Manager. Local calls from branch locations can be routed back to the same branch location and terminate on the FXO interface of the local SR2330 branch gateway. This has the potential benefits of saving bandwidth on the branch access network, off loading the WAN and centralized media gateway resources, avoiding toll charges, and reducing latency. The Distributed Trunking call flows presented in **Section 2.6** provide additional details of how calls are routed based on the location of the caller and the number being called.

The Centralized Trunking and Distributed Trunking configurations share mostly the same configuration procedures on Communication Manager and Sessions Manager. The configuration procedures in the ensuing text implement the Distributed Trunking configuration. Differences specific to Centralized Trunking will be pointed out where appropriate.

2.5. Network Modes

PSTN call routing is further determined within each of the trunking configurations based on the network status of each branch.

Normal Mode: A Branch has WAN connectivity to the main Headquarters/Datacenter location and the centralized Avaya SIP call control platform is being used for all branch calls.

Survivable Mode: A Branch has lost network connectivity to the Headquarters/Datacenter location and the local branch SR2330 SIP call control is being used for all calls at that branch. Note: if the Session Manager loses connectivity to the WAN or has gone out of service, all branches will go into Survivable Mode simultaneously.

2.6. Call Flows

This section presents the primary call flows for the Survivable SIP Gateway Solution in a Distributed Trunking configuration for both Normal Mode and Survivable Mode. The components included in these call flows are based on the components used in the sample configuration presented in these Application Notes.

2.6.1. Distributed Trunking – Normal Mode

Overview:

- **SIP Call Control**: All SIP call control and call routing is provided by the centralized Session Manager and Communication Manager.
- **Branch PSTN Outbound Local**: Session Manager Location Based Routing is used to route these calls to the local branch SR2330 FXO interface. In the case of Centralized Trunking configuration, the Session Manager Location Based Routing will route these local calls from the branches to a centralized Avaya G450 Media Gateway for onward routing to the PSTN.
- **Branch PSTN Outbound Non-Local**: PSTN outbound calls from the branch to non-local numbers are routed to a centralized Avaya G450 Media Gateway controlled by the Communication Manager ES.
- **Branch PSTN Inbound**: Calls from the PSTN to a branch Listed Directory Number (LDN) enter the enterprise network at the local branch SR2330 FXO interface, then route to the Session Manager/Communication Manager for call treatment.
- **Headquarters PSTN Inbound**: Calls to Headquarters endpoints enter the enterprise network at the Headquarters Avaya G450 Media Gateway controlled by the Communication Manager ES.

Call Flows:

1. **Avaya 9600 SIP Phone at branch to H.323 IP phone at Headquarters**
   Avaya 9600 SIP → Session Manager → Communication Manager ES → H.323 IP phone

2. **Avaya 9600 SIP Phone at branch to Digital/Analog phone at Headquarters**
   Avaya 9600 SIP → Session Manager → Communication Manager ES → Avaya G450 Media Gateway → Digital/Analog phone

3. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Local Number**
   Avaya 9600 SIP→ Session Manager → Communication Manager ES → Session Manager → SR2330 FXO → PSTN phone

4. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Long Distance Number**
   Avaya 9600 SIP → Session Manager → Communication Manager ES → Avaya G450 Media Gateway → PSTN phone

5. **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch**
   Avaya 9600 SIP → Session Manager → Communication Manager ES → Session Manager → Avaya 9600 SIP

6. **PSTN phone to Branch LDN assigned to Avaya 9600 SIP phone**
   PSTN phone → SR2330 FXO → Session Manager → Communication Manager ES → Session Manager → Avaya 9600 SIP

Figure 1 presents a high level network view of the Distributed Trunking Normal Mode call flows.

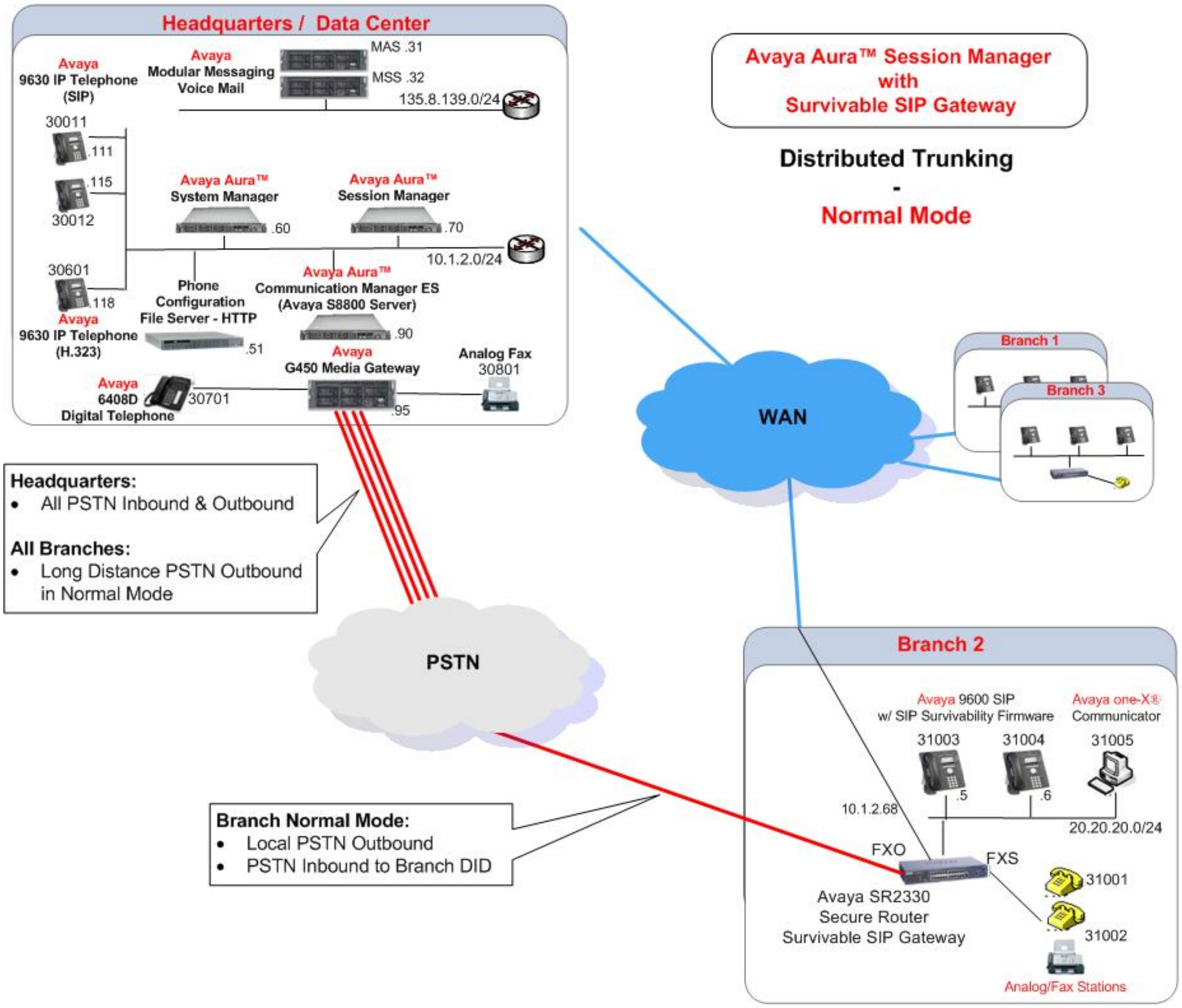


Figure 1: Distributed Trunking – Normal Mode

2.6.2. Distributed Trunking – Survivability Mode

Overview:

- **SIP Call Control**: All SIP call control and call routing is provided by the local branch SR2330.
- **SIP Registration**: All branch Avaya 9600 SIP Phones are transitioned and registered to the SR2330.
- **All Branch PSTN Outbound**: Local and Non-Local: Routed to the SR2330 FXO interface.
- **Branch PSTN Inbound**: Calls from the PSTN to a branch Listed Directory Number (LDN) or Direct Inward Dialing (DID) number enter the network at the local branch SR2330 FXO interface. The SR2330 routes the call to a phone assigned to the FXO interface.

Call Flows:

1. **Avaya 9600 SIP Phone at branch to PSTN endpoint – Local & Non-Local**
   Avaya 9600 SIP → SR2330 FXO → PSTN phone

2. **PSTN phone to Branch LDN or DID assigned to Avaya 9600 SIP phone.**
   PSTN phone → SR2330 FXO → Avaya 9600 SIP

3. **Avaya 9600 SIP Phone at branch to Avaya 9600 SIP phone at same branch.**
   Avaya 9600 SIP → SR2330 → Avaya 9600 SIP

Figure 2 presents a high level view of the Distributed Trunking Survivable Mode call flows.

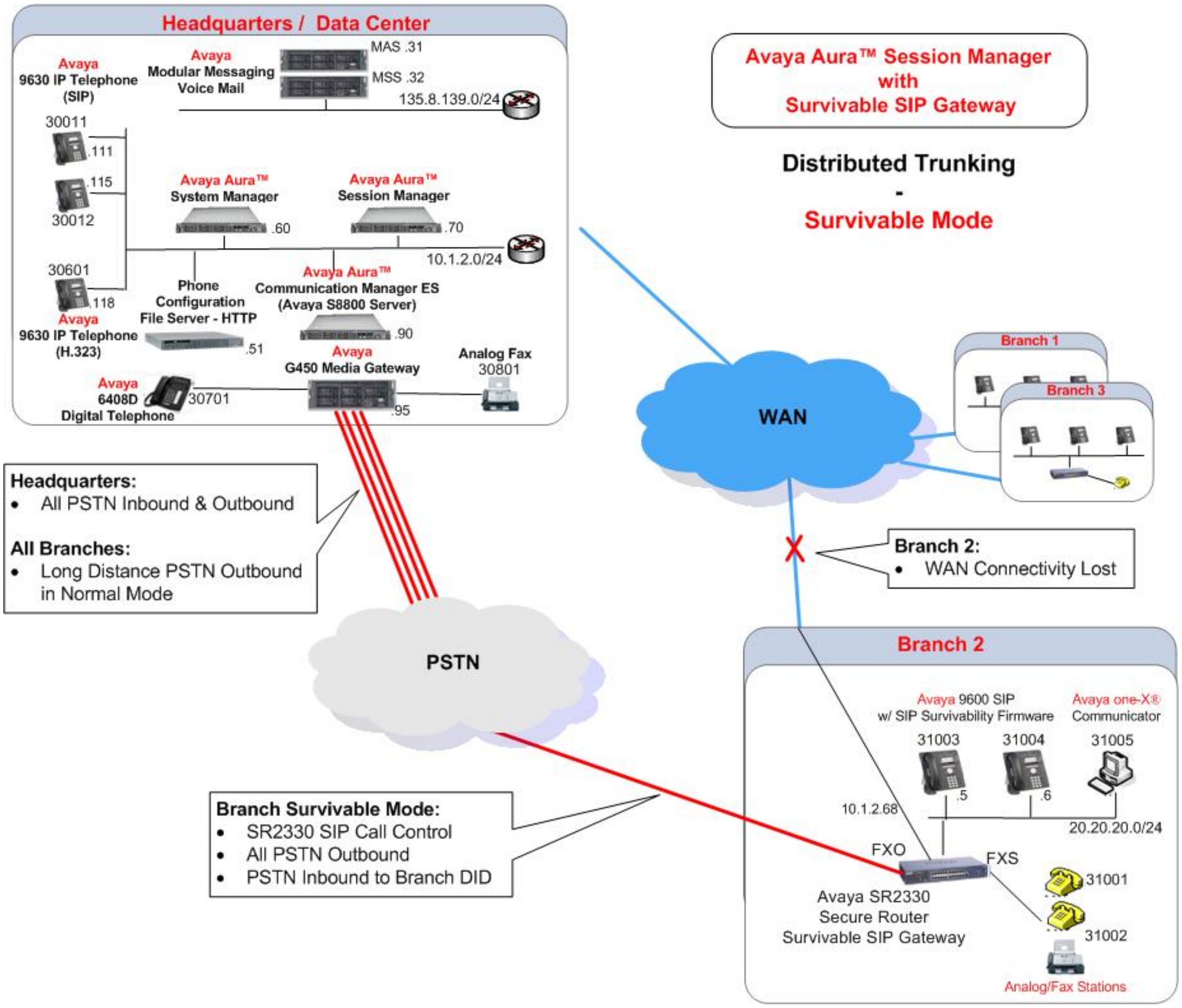


Figure 2: Distributed Trunking – Survivable Mode

2.6.3. Detailed Call Flow: Branch PSTN Outbound Local – Normal Mode

Many of the Session Manager and Communication Manager configuration steps presented in **Section 4** and **Section 5** are to support the location based routing requirements of the Branch PSTN Outbound Local – Normal Mode call flow. The details of this call flow, specific to the sample configuration, are included here as a reference for better understanding of the linkage of the various configuration steps. As mentioned earlier, the term “Communication Manager” refers to Communication Manager ES.

Branch PSTN Outbound Local – Normal Mode:
Branch 2 Avaya 9600 SIP Phone user dials the following local PSTN number:

9 1-908-848-5703.

1. Branch 2 Avaya 9600 SIP Phone sends SIP INVITE to Session Manager with dialed digit string of 919088485703.
2. Session Manager receives the SIP INVITE and identifies the Avaya 9600 SIP Phone user has an assigned Communication Manager Extension. Session Manager forwards the SIP INVITE to Communication Manager.
3. Communication Manager receives the SIP INVITE from Session Manager on SIP Trunk Group Number 60.
4. Communication Manager identifies the IP address of the Avaya 9600 SIP Phone in the Contact field of the SIP INVITE message as an IP address mapped to IP Network Region 10. This will be used to determine the proper codec set to use.
5. The leading 9 in the dialed digit string is identified by Communication Manager as the ARS Access Code. The 9 is removed from the dialed digit string.
6. The ARS Digit Analysis Table is queried for a match on the remaining digits 19088485703.
7. A match on 1908848 is found and Route Pattern 60 is chosen.
8. Route Pattern 60 routes the call to SIP Trunk Group Number 60.
9. Communication Manager sends a new SIP INVITE to Session Manager over SIP Trunk Group Number 60 with the dialed digits of 19087665703.
10. Session Manager matches on the digits 1908848 of the dialed number and identifies the calling phone as part of Location “SR2330 Branch 2” and identifies the next hop as the Branch 2 SR2330 with IP address 20.20.20.1 using TCP port 5080.
11. Session Manager forwards the SIP INVITE with dialed digits string 19088485703 to the Branch 2 SR2330.
12. The Branch 2 SR2330 internally routes the call to an FXO interface for termination on the PSTN.

2.7. Network Topology

The network implemented for the sample configuration shown in **Figure 1** and **Figure 2** is modeled on an enterprise consisting of a main Headquarters/Datacenter location and multiple distributed branch locations all inter-connected over a corporate WAN. While three branch locations are shown, the Branch 2 configuration was implemented and described in this document.

The Headquarters location hosts Session Manager and the Communication Manager ES, running on an Avaya S8800 server, controlling an Avaya G450 Media Gateway with PSTN trunks. The Headquarters network is mapped to IP Network Region 1 within Communication Manager ES. The Distributed Trunking capabilities of the solution utilize the location based call routing features of Session Manager, and IP codec set selection features of Communication Manager ES, and requires the information presented in **Table 1** below.

IP Network	IP Network Region	Location	Area Code & Exchange
10.1.2.0/24	1	Basking Ridge	732-829
20.20.20.0/24	10	Branch 2	908-848

Table 1 – Network Information

The Headquarters location also hosts the following centralized components: an Avaya Modular Messaging voice mail platform and an Avaya IP Phone Configuration File Server. The configuration details of these components are considered out of scope of these Application Notes and are therefore not included.

The Avaya IP Phone Configuration File Server contains the 46xxsettings.txt file used by Avaya IP phones to set the values of phone configuration parameters. **Section 7** includes the parameters of the 46xxsettings.txt file used by the Avaya 9600 SIP Phone for survivability. The Avaya Modular Messaging voice mail platform can be reached by dialing the internal extension configured as the voice mail access number or pilot number, or by dialing a PSTN number that also terminates to Modular Messaging. The internal or private extension is configured in the 46xxsettings.txt file as the default voice mail access number to dial when the Message button of the Avaya 9600 SIP Phone is pressed while the phone is in Normal Mode. The external PSTN number is configured in the 46xxsettings.txt file as an alternate voice mail access number to dial when the Message button of the Avaya 9600 SIP Phone is pressed while the phone is in Survivable Mode. This enables branch users to continue to access the centralized voice mail platform while in Survivable Mode via the PSTN using the Message button. Traditional Message Waiting Indication via the telephone is not available while the phone is in Survivable Mode. The messaging system, such as Avaya Modular Messaging, may enable other methods of notification that a message has been delivered.

In addition to the Avaya 9600 SIP Phones, the following types of endpoints are also set up at the Headquarters location:

- An Avaya 9600 H.323 Phone connected to the Headquarters LAN
- An Avaya 6408D Digital Phone connected to the Avaya G450 Media Gateway
- An analog phone connected to the Avaya G450 Media Gateway

- An analog fax machine connected to the Avaya G450 Media Gateway

All the above endpoints are natively configured on the Communications Manager ES and interwork with the SIP endpoints at the main site and the branch sites.

The branch location consists of several Avaya 9600 SIP Phones (including an one-X Communicator SIP soft phone) and an SR2330 Secure Router with two PSTN Analog trunks on FXO interfaces and two analog phones / fax machines on FXS interfaces. A flat network has been implemented at the branch. In the sample configuration (see **Figure 3**), the SR2330 uses its LAN side IP address (20.20.20.1) for SIP signaling with the local branch SIP phones. Its SIP Media Gateway Module listens for SIP requests on port 5080. Requests can come from either the Session Manager in the Headquarters location or the SIP Survivability Module within the SR2330. In survivable mode, the SIP Survivability Module listens on port 5060 for SIP requests from the branch one-X Deskphone SIP 9600 Series IP Telephones, and proxies those requests to the Media Gateway module as necessary (e.g., for calls to the FXS or FXO interfaces). In Normal Mode, the IP telephones signal through the SR2330 directly to Session Manager.

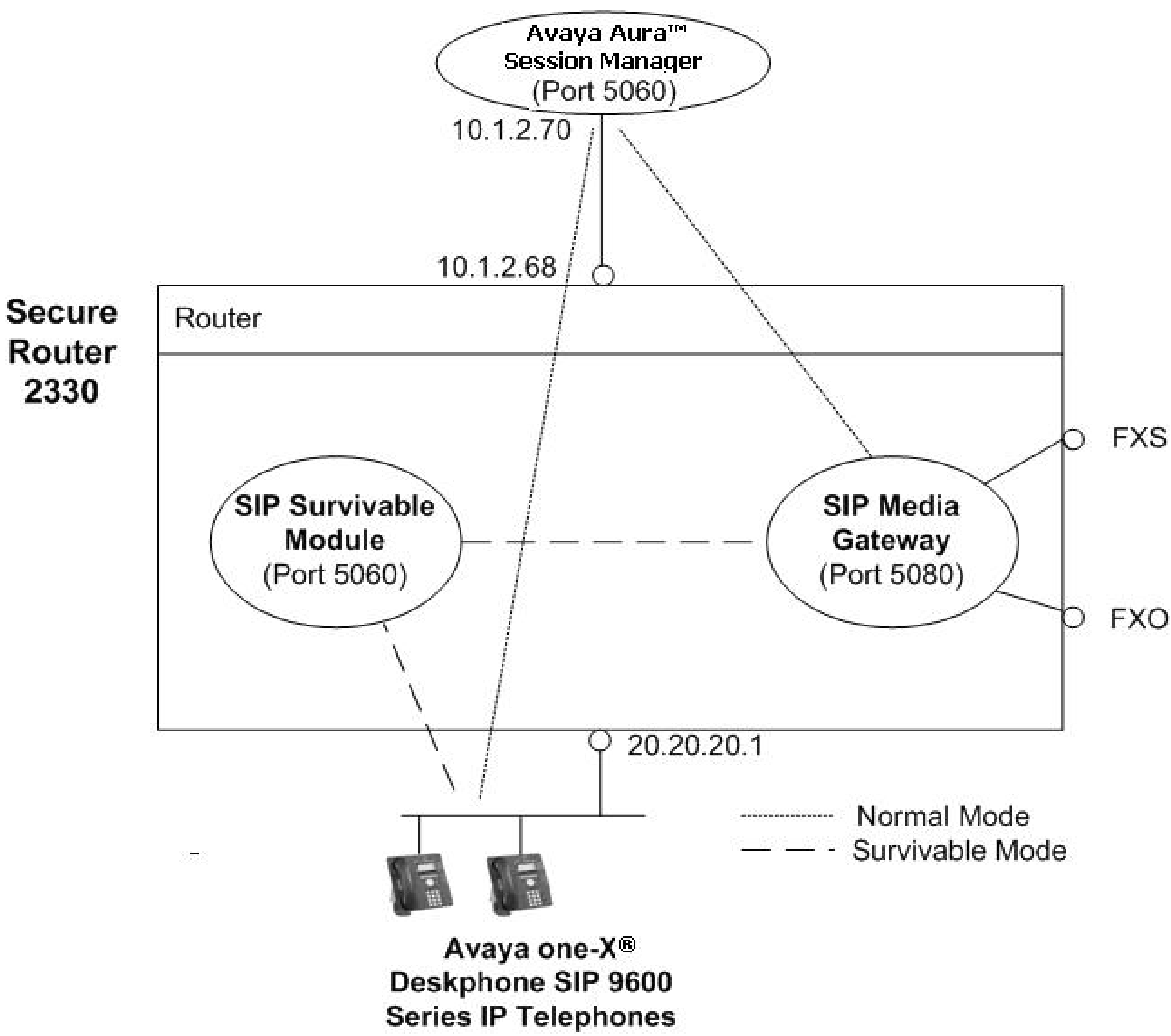


Figure 3: Avaya SR2330 in Sample Configuration

3. Equipment and Software Versions

The information in these Application Notes is based on the software and hardware versions listed in **Table 2**.

Device Description	Versions Tested
Avaya S8800 Server with G450 Media Gateway	Avaya Aura™ Communication Manager 6.0 (pre-GA) R016x.00.0.344.0 with Patch 1003
Avaya S8800 Server	Avaya Aura™ Session Manager 6.0.0.0 (pre-GA), Build 34013 (Sprint 34)
Avaya S8800 Server	Avaya Aura™ System Manager 6.0.0.0 (pre-GA), Build 34012
Avaya S8800 Server	Avaya Modular Messaging 5.2, Build 9.2.150.13 (Patch 520008)
Avaya S8800 Server	Avaya Modular Messaging 5.2, Build 5.2-11.0
Avaya one-X® Deskphone 9600 Series IP Telephones (SIP)	2.6 pre-GA release (SIP96xx_2_6_2_9.bin)
Avaya one-X® Communicator (SIP)	6.0.0.17 (pre-GA)
Avaya one-X® Deskphone 9600 Series IP Telephones (H323)	3.1
Avaya 6210 Analog Telephone	-
Brother Intellifax analog fax machines	-
Avaya SR2330 Secure Router	10.2.1.101

Table 2 – Software/Hardware Version Information

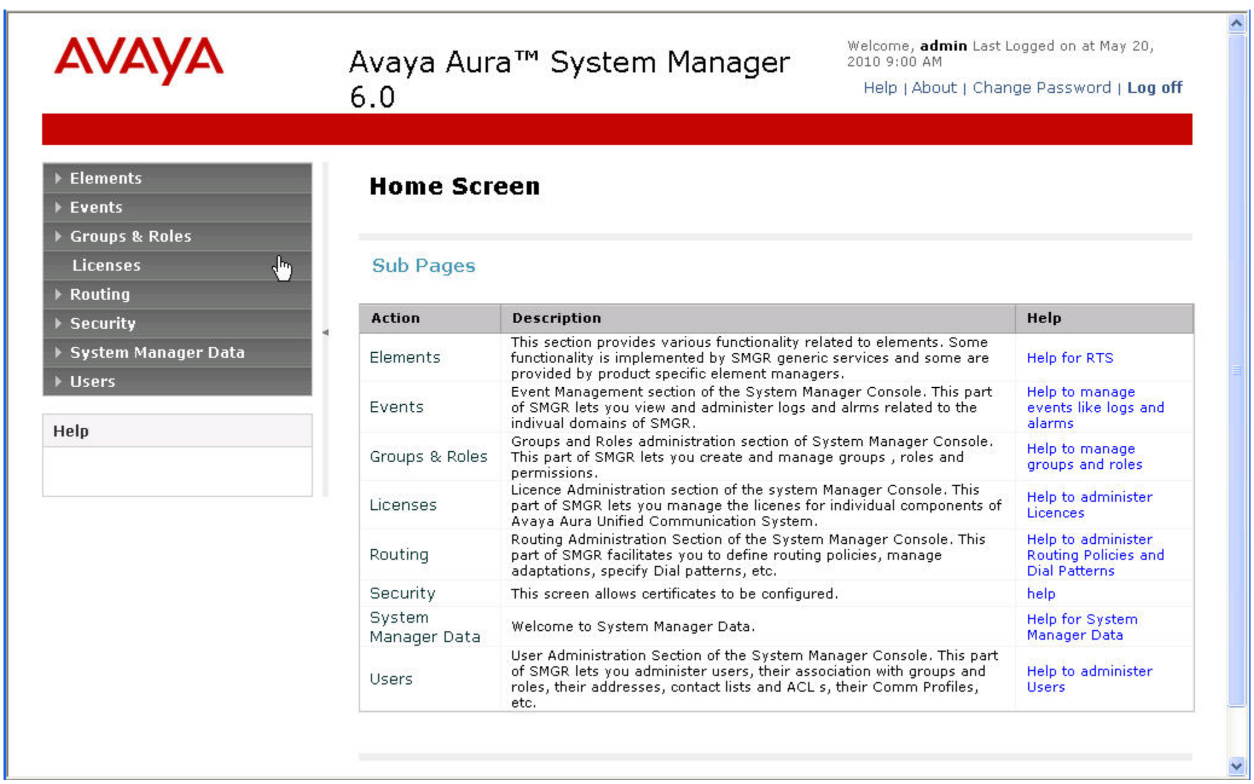
4. Configure Avaya Aura™ Session Manager

This section describes the administration steps for Session Manager that implements the Survivable SIP Gateway Solution. The following areas are covered:

- SIP domain
- Location configuration for Branch 2
- SIP Element configuration for the SR2330 Secure Router
- Element Link, which defines the SIP trunk parameters used by Session Manager when routing calls to/from the SR2330 Secure Router
- Location based routing policy corresponding to the dial plan definitions
- Dial pattern configurations for routing calls to long distance and branch-local PSTN destinations
- User configuration for branch SIP and analog telephones

It is assumed that the basic configuration of the Session Manager, including Session Manager Element administration and configuration of a SIP Element Link between Session Manager and the Communication Manager ES, has already been completed.

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **OK** in the subsequent confirmation screen. The menu shown below is then displayed. Expand the **Routing** Link on the left side. The sub-menus displayed in the left column under **Routing** will be used to configure all of the above parameters except the SIP users (**Sections 4.1** through **4.6**).

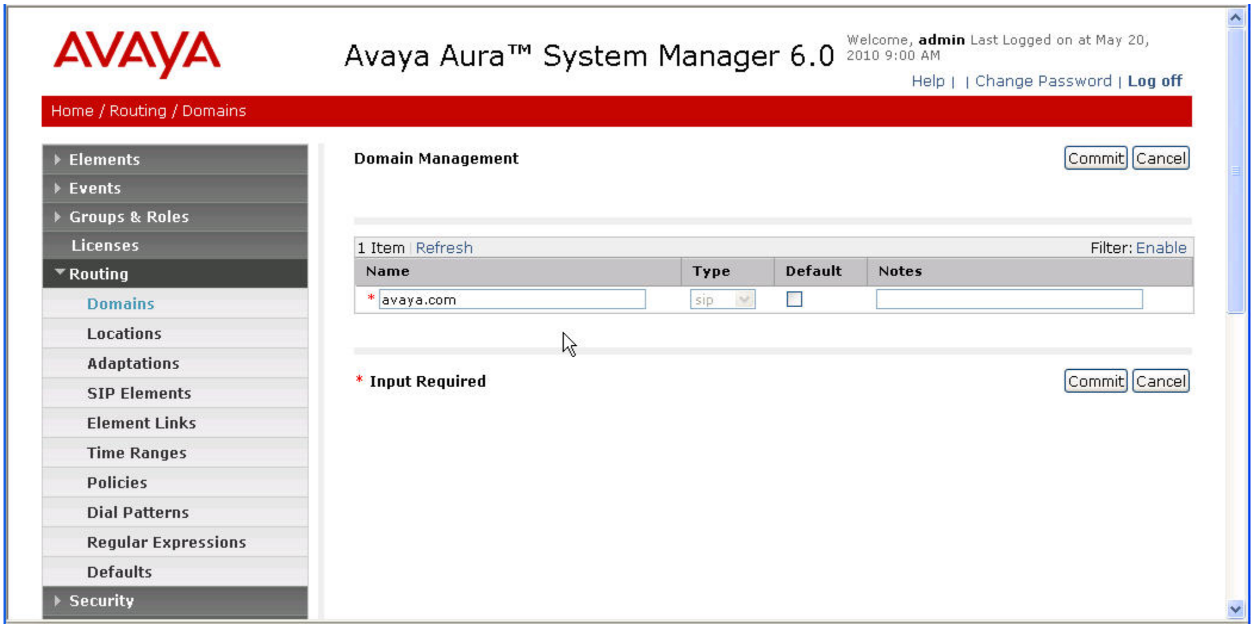


4.1. Specify SIP Domain

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

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

Click **Commit**.



4.2. Add Location

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

Under *General*, enter:

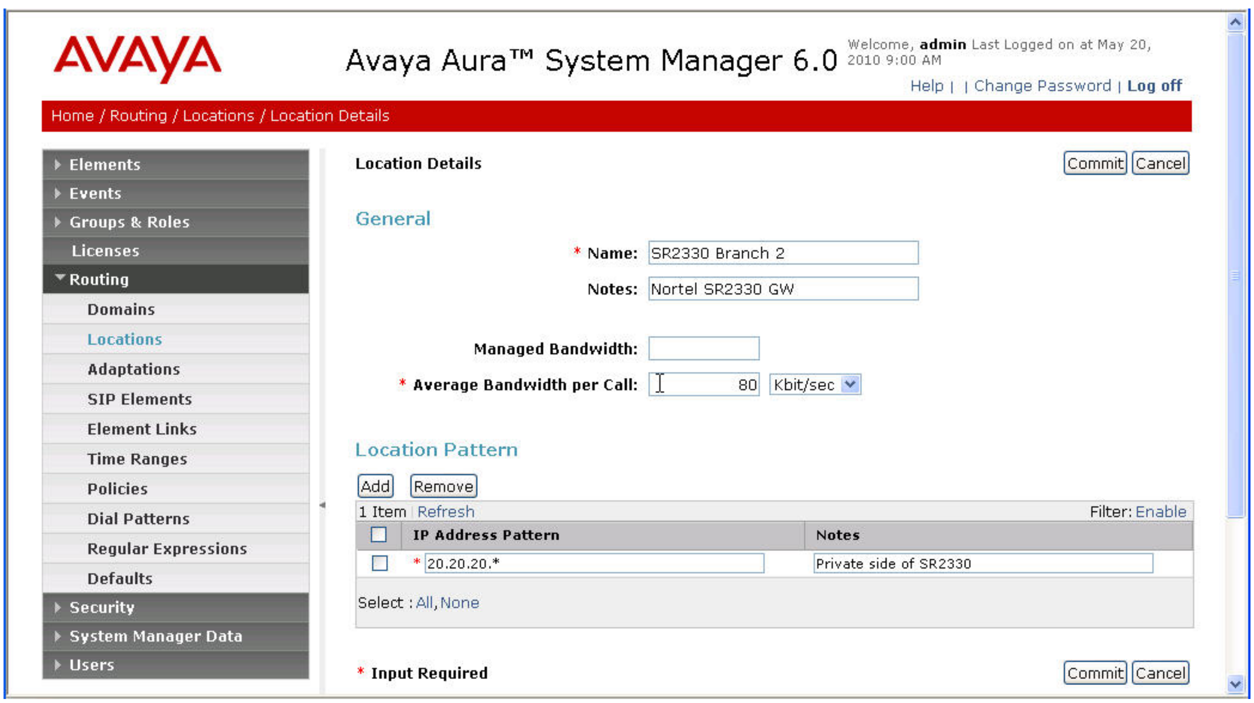
- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

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

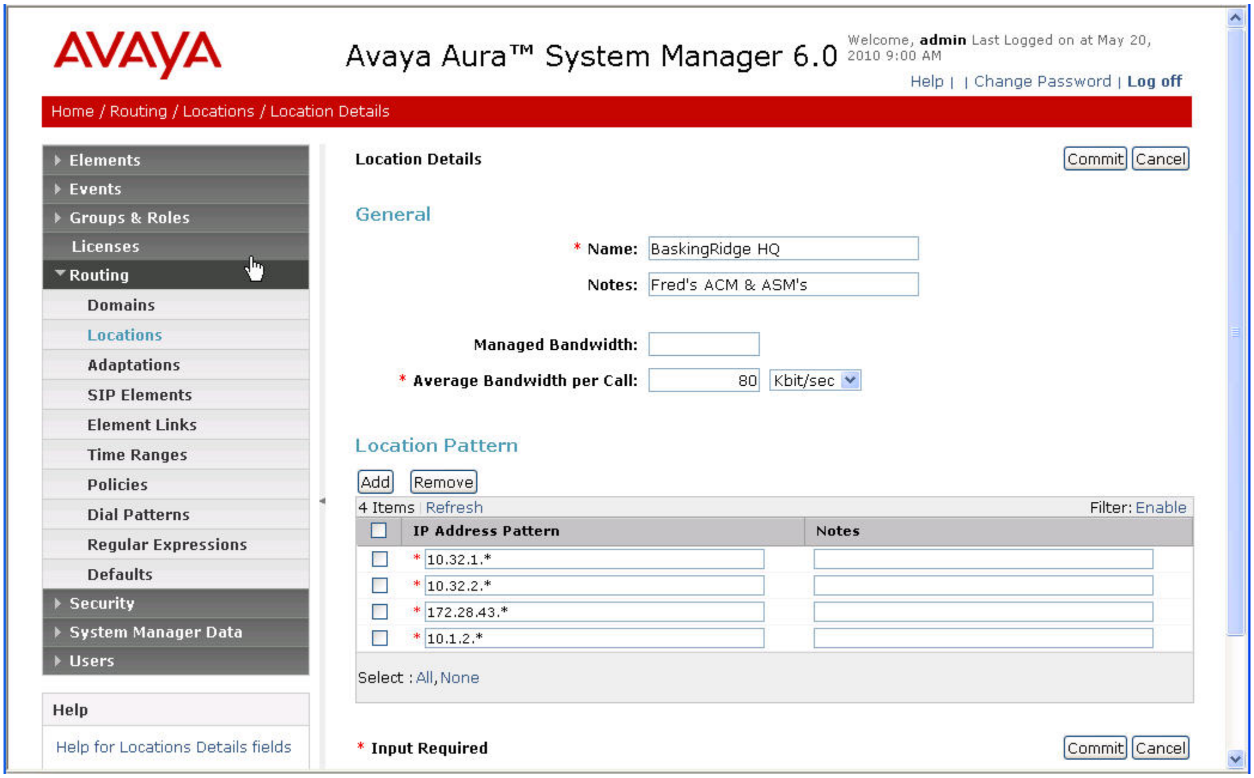
The fields under *General* can be filled in to specify bandwidth management parameters between Session Manager and this location. These were not used in the sample configuration, and reflect default values.

The screen below shows addition of the Branch 2 location, which includes the SR2330 Secure Router and the Avaya 9630 IP Telephones (SIP) in the 20.20.20.0/24 subnet.[1]. This location will be used in the configuration of routing policies for Branch 2. Click **Commit** to save the Location definition.



[1] Note that even though the *WAN* interface on the SR2330 is 10.1.2.68, the *SIP signaling* interface on the SR2330 is bound to the LAN side 20.20.20.1 address. See **Section 6**.

The screen below shows addition of the “BaskingRidge HQ” Location for the Headquarters site, which includes Session Manager (10.1.2 subnet), Communication Manager (10.1.2 subnet), and all SIP telephones located at this location (10.1.2 subnet). Other IP addresses in the screen were not used by the sample configuration. Click **Commit** to save the **Location Details** definition.



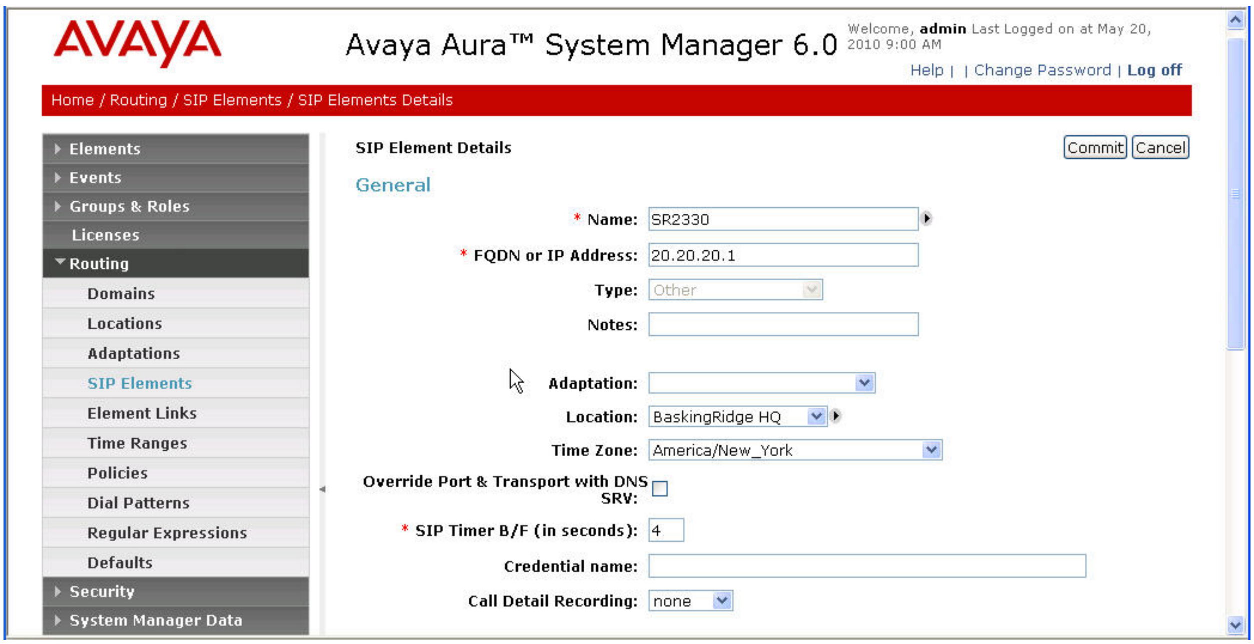
4.3. Add SIP Element

This section describes the configuration of the SIP Element corresponding to the SR2330 in Branch 2. Select **Routing** → **SIP Elements** on the left and click on the **New** button (not shown) on the right.

Under *General*, fill in:

- **Name:** A descriptive name.
- **FQDN or IP Address:** FQDN or IP address of the SIP signaling interface on the SR2330 (See **Section 6**).
- **Type:** "Other".
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save the SIP Entity definition. The screen below shows the resulting SIP Entity configured for the SR2330.

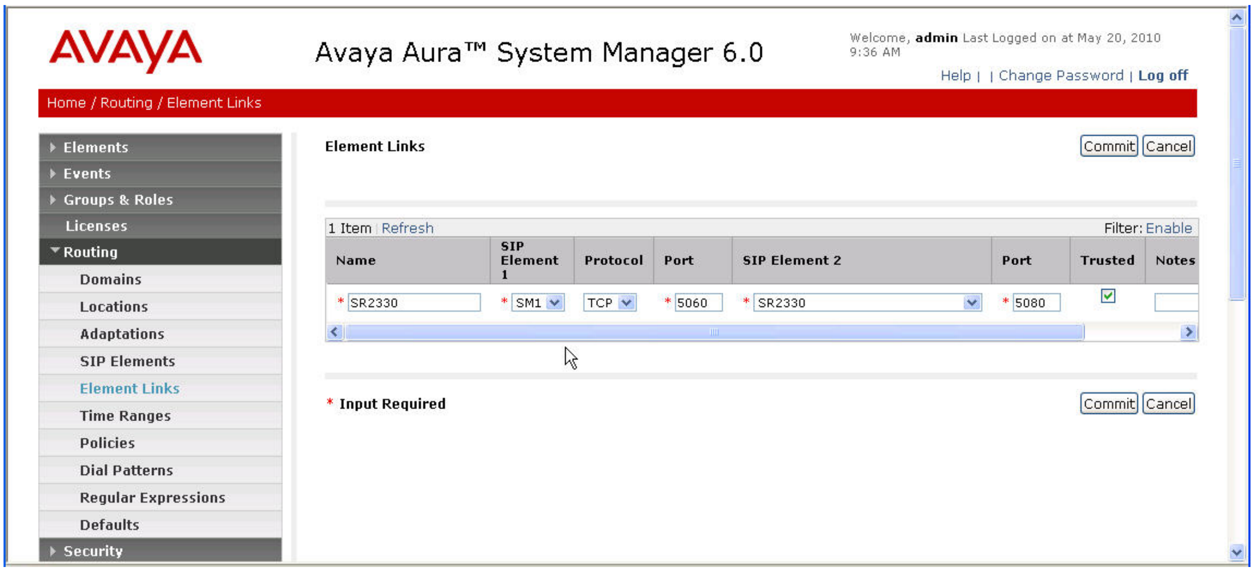


4.4. Add Element Link

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

- **Name:** A descriptive name.
- **SIP Element 1:** Select the desired Session Manager.
- **Protocol** Select TCP.
- **Port:** Port number to which the SR2330 sends SIP requests.
- **SIP Element 2:** Select the name of the other system.
- **Port:** Port number on which the SR2330 receives SIP requests (See **Figure 3** in **Sections 2.7** and **Section 6**).
- **Trusted:** Check this box.
  Note: If this box is not checked, calls from the SIP Element 2 will be denied by Session Manager.

Click **Commit** to save the Entity Link definition. The following screen shows adding the Element Link connecting Session Manager and the branch SR2330.



4.5. Add Routing Policies

Routing Policies describe the conditions under which calls will be routed to SIP Entities. The Routing Policies can be thought of as routing destinations with routing conditions.

The inter-branch and intra-branch calls between phones using extension numbers do not need Routing Policies since all the phones, both at the Headquarters and in the branches, are administered on the Communication Manager and register to the Session Manager. However, calls to the PSTN need Routing Policies to determine where they are going to be routed for eventual termination to the PSTN. These calls could go out to the PSTN through the PSTN trunks on the Avaya G450 Media Gateway at the Headquarters, or they could go out through the analog trunks (a.k.a Service Provider CO lines) connected to the FXO ports on the branch SR2330.

Separate Routing Policies need to be created for sending PSTN-bound calls. In the case of Centralized Trunking arrangement, all PSTN-bound calls, regardless of the call originations (either from the Headquarters or from the branches), should be sent to the Headquarters for onward routing to the PSTN. In the case of Distributed Trunking, all PSTN-bound calls from the Headquarters plus the Long Distance toll calls from the branch locations should be routed to the Headquarters for PSTN termination, but local calls from the branch should be routed to the local branch SR2330 for termination to the PSTN through the FXO interfaces on the branch SR2330.

For the sample configuration, two routing policies were added for routing calls to PSTN: one with the Headquarters Communication Manager as the routing destination; one with the Branch 2 SR2330 as the routing destination.

To add a routing policy, select **Routing → Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

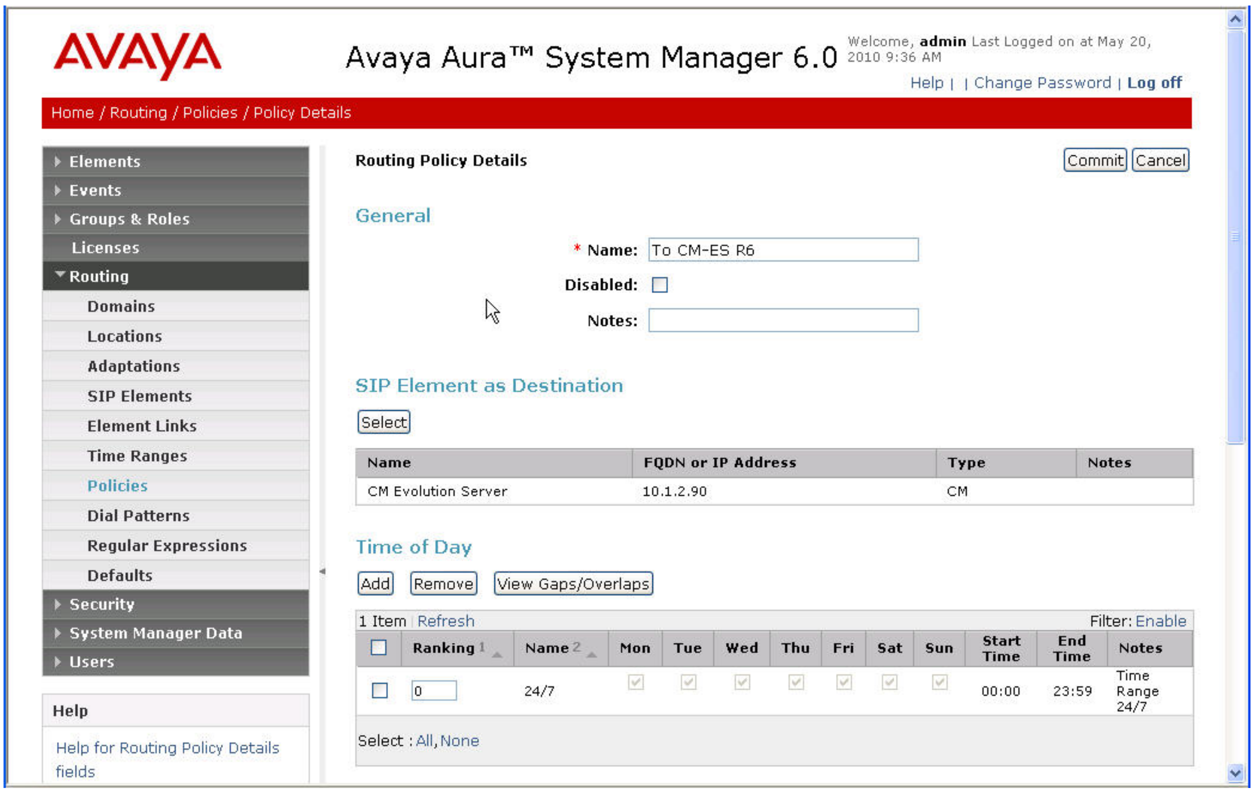
Under *General*:
Enter a descriptive name in **Name**.

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

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

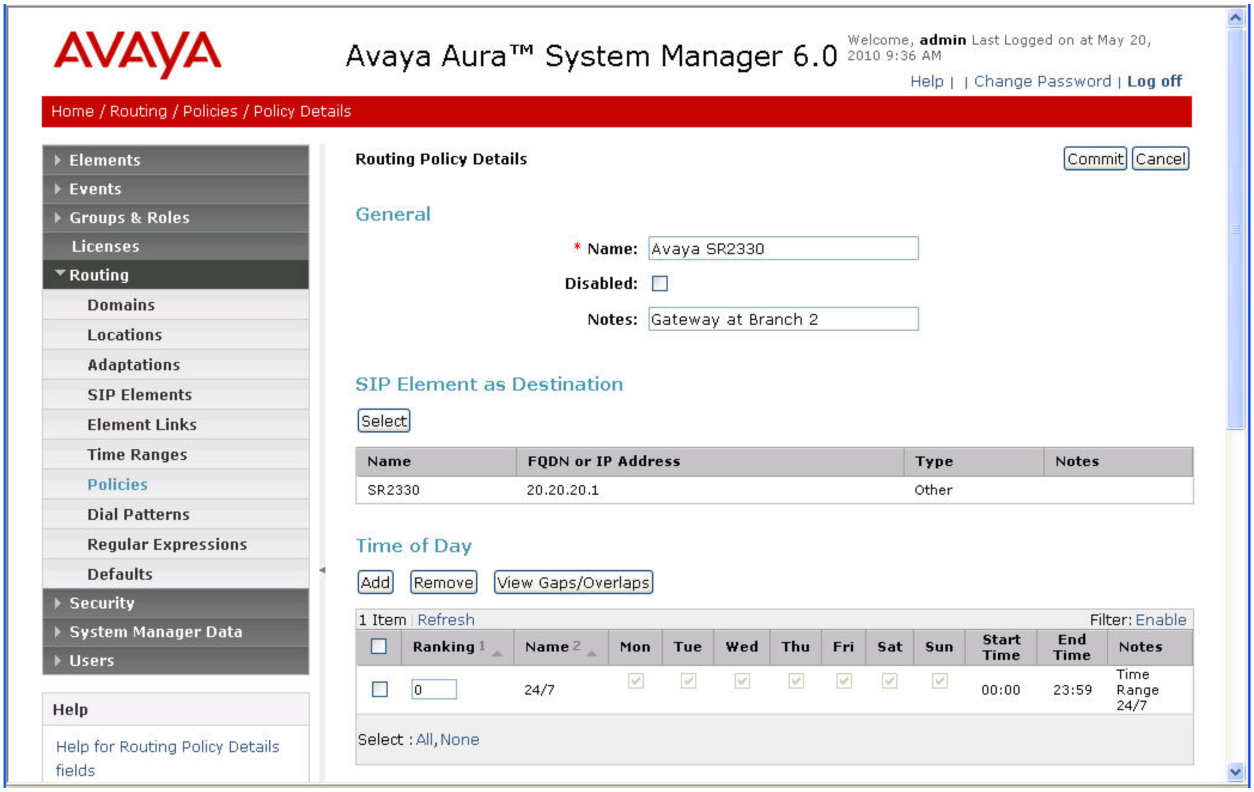
Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition. The following screen shows the Routing Policy for the Branch 2 SR2330.

The following screen shows the Routing Policy configured in the sample configuration for routing PSTN calls to the Headquarters where the Communication Manager would send these calls through the Avaya G450 Media Gateway to the PSTN[2].



[2] The configurations on Communication Manager and the Avaya Media Gateway for routing calls to the PSTN (Route Pattern, PSTN Trunk/Signaling Groups, T1/E1 interfaces, etc.) are standard configurations out of scope of these Application Notes, and are therefore not included.

The following screen shows the Routing Policy configured in the sample configuration for routing PSTN calls to the local branch SR2330.



4.6. Add Dial Patterns

Define a Dial Pattern for matching calls based on dialed digits. A Dial Pattern is then associated with a Routing Policy to direct calls with the matched dialed digit strings to the destinations (SIP Elements as specified in Routing Policies). **Note:** Calls to Branch users do not require a dial pattern, since they are directly registered to Session Manager.

To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

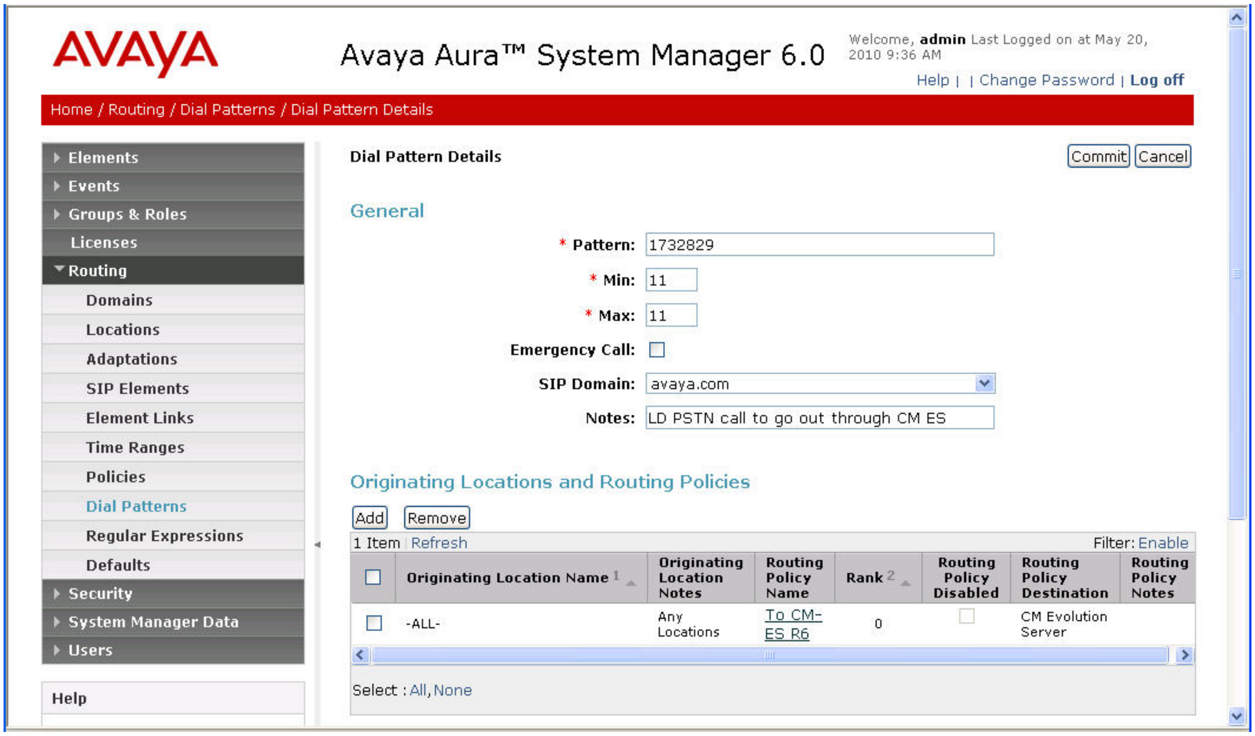
Under *General*:

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

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the origination Location and the Routing Policy from the lists.

Default values can be used for the remaining fields. Click **Commit** to save each dial pattern.

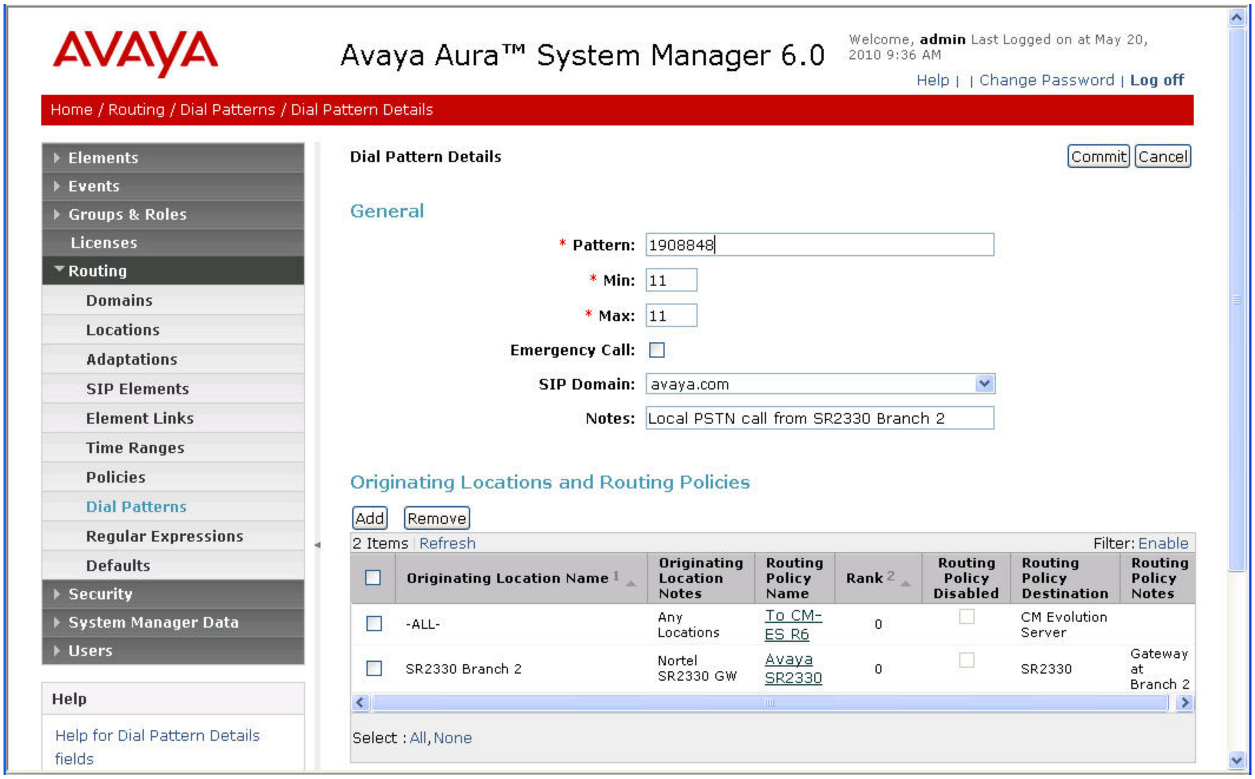
The following screen shows the Dial Pattern defined for routing calls to the PSTN with the dialed number 1-732-829-XXXX. Since the calls to the “732” area code are Long-Distance toll calls for the sample branch in the sample configuration, “-ALL-“ is selected for **Originating Location Name** and “To CM-ES R6” (as configured in **Section 4.5**) was selected for **Routing Policy Name.** With these settings, calls from both the Headquarters and the branch would be routed to the Headquarters Communication Manger for termination to the PSTN through the Avaya Media Gateway.



The following screen shows the Dial Pattern defined for routing calls to the dialed number 1-908-848-XXXX.

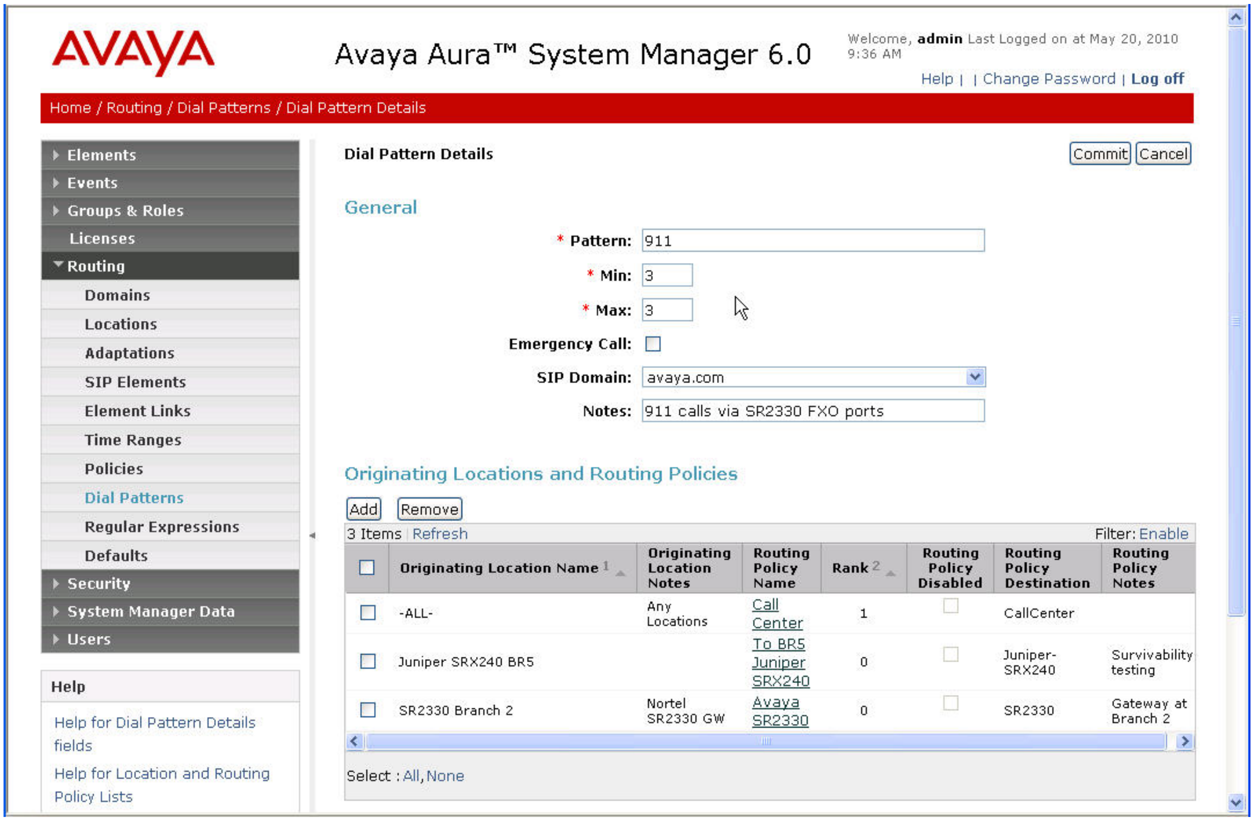
Since the calls to the “908” area code are local calls for the sample branch in the sample configuration, the Location “SR2330 Branch 2” (as defined in **Section 4.2**) was selected for **Originating Location Name** so that calls to the “908” area code from the sample branch would be routed to the SIP Element “SR2330” as specified in the Routing Policy “Avaya SR2330”. The branch SR2330 would route these calls out to the PSTN through its FXO interface ports.

A second entry was specified for the “1908848” Dial Pattern that would route calls to the “908” area code from all Locations, except the sample branch, to the Headquarters Communication Manger for termination to the PSTN through the Avaya Media Gateway. **Note**: In the case of Centralized Trunking arrangement, the entry for routing the local calls to the sample branch is not needed since all PSTN calls, regardless of their origination locations, should be routed to the central location for termination to the PSTN.



Note: In the sample configuration, the branch phones are restricted to make calls only to the 1-908-848-XXXX (local) and 1-732-829-XXXX (long distance) numbers. In real deployment, the Dial Patterns should be modified as appropriate in accordance with business policies.

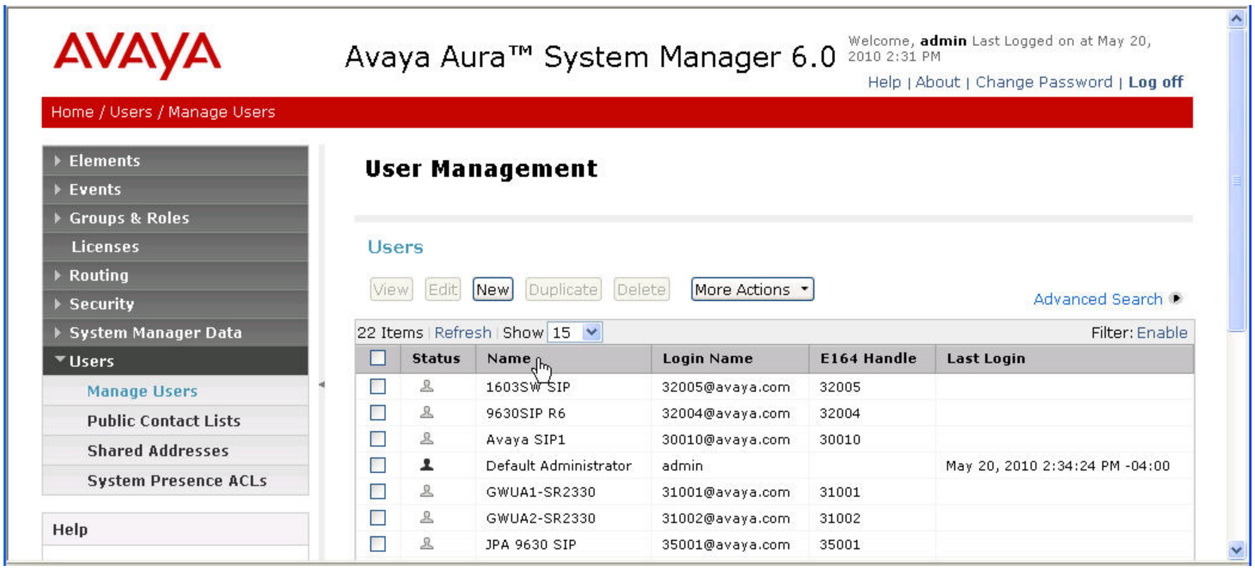
The following screen shows the Dial Pattern defined for routing 911 emergency calls. As shown by the 3rd entry under *Originating Locations and Routing Policies*, 911 calls from the branch location in the Normal Mode will be routed back to the branch SR2330 to go out, through the FXO interfaces on the branch SR2330, to the local Emergency Response Center. In the Survivable Mode, the routing policy on the branch SR2330 will route 911 calls from the branch to the PSTN through its FXO interfaces too.



4.7. SIP Users

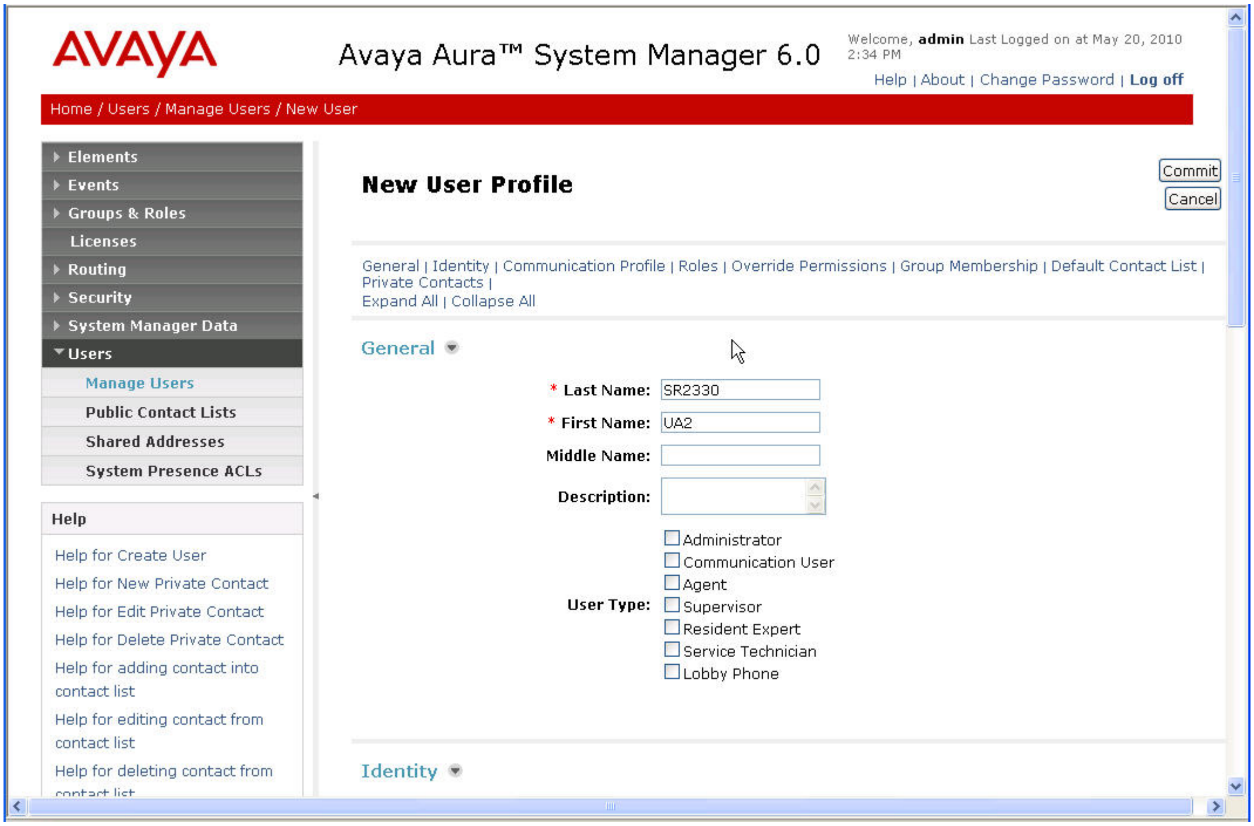
This section describes the administration of SIP telephones in Session Manager, and applies to the 9600 series SIP telephones, the one-X Communication soft phone as well as the analog telephones connected to the FXS ports of the SR2330, which registers with Session Manager on their behalf. It is assumed that the SIP trunk between Communication Manager and Session Manager has already been provisioned. **References [3]** and **[4]** contain information on configuring SIP trunks between Communication Manager and Session Manager. The following screens show a sample configuration for an Avaya 9630 SIP phone whose extension is 31003. The same procedure should be followed for all branch IP and analog telephones.

On the main configuration page, select **Users → Manage Users** from the left configuration tree, and click **New** to administer a new telephone user.



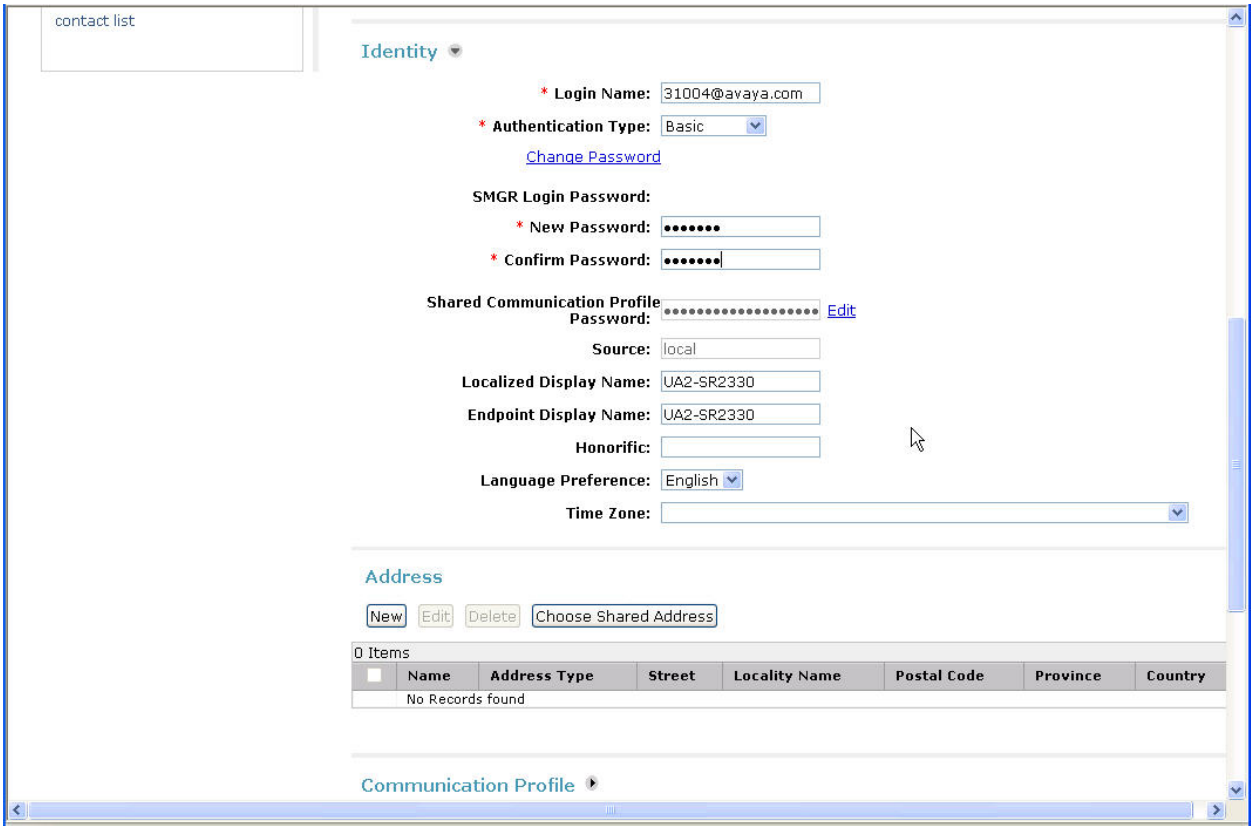
This will create a new User Profile.

In the *General* section of New User Profile, enter a **Last Name** and **First Name**. Note that fields marked with * are required to be filled in.



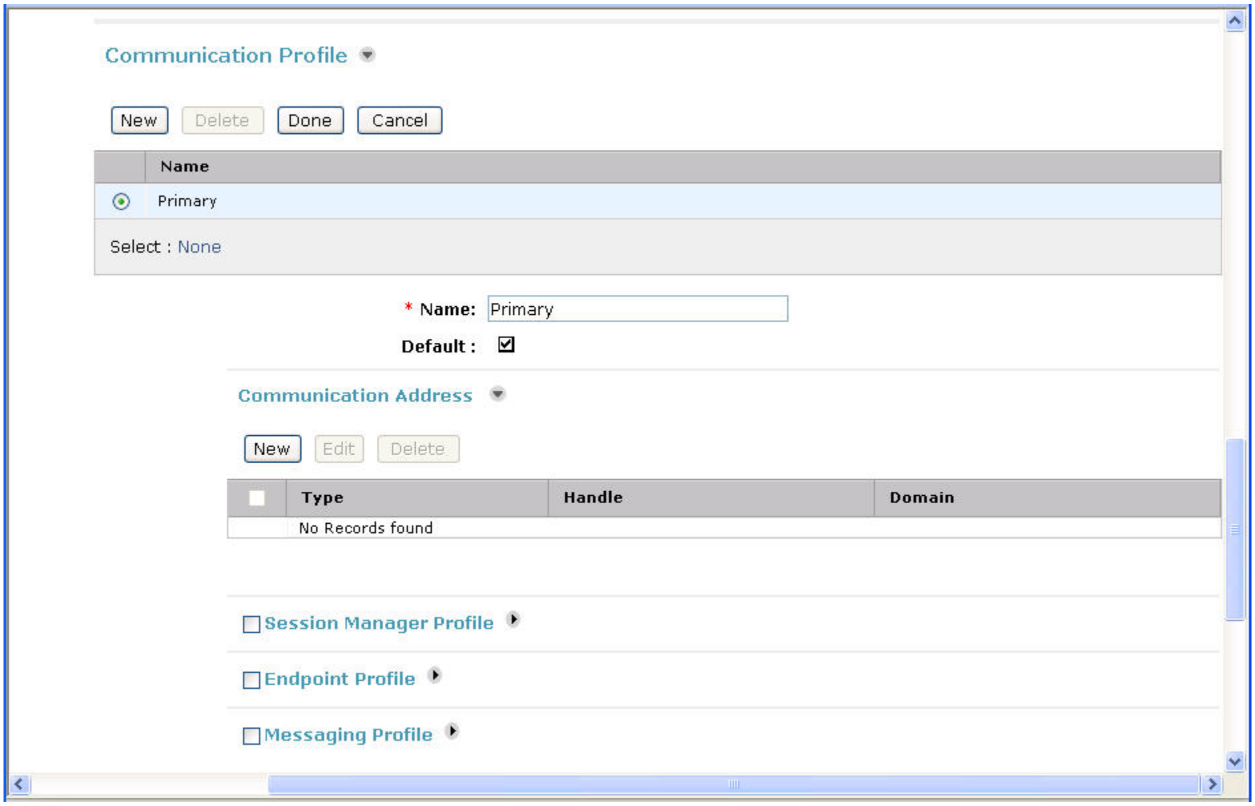
The information above is what was entered for extension 31004, a 9600 SIP Telephone in the branch.

In the *Identity* section, enter a Login Name, for example “31004@avaya.com”, and the required passwords, as shown below. Note that the **Shared Communication Profile Password** is the one the telephone is required to use when registering to Session Manager. It is also recommended to enter both display names with the same data. **SMGR Login Password**, while required, was not used in this sample configuration, and can be any value.



The information above is what was entered for extension 31004. Note that the passwords are not displayed when viewing an endpoint’s configuration.

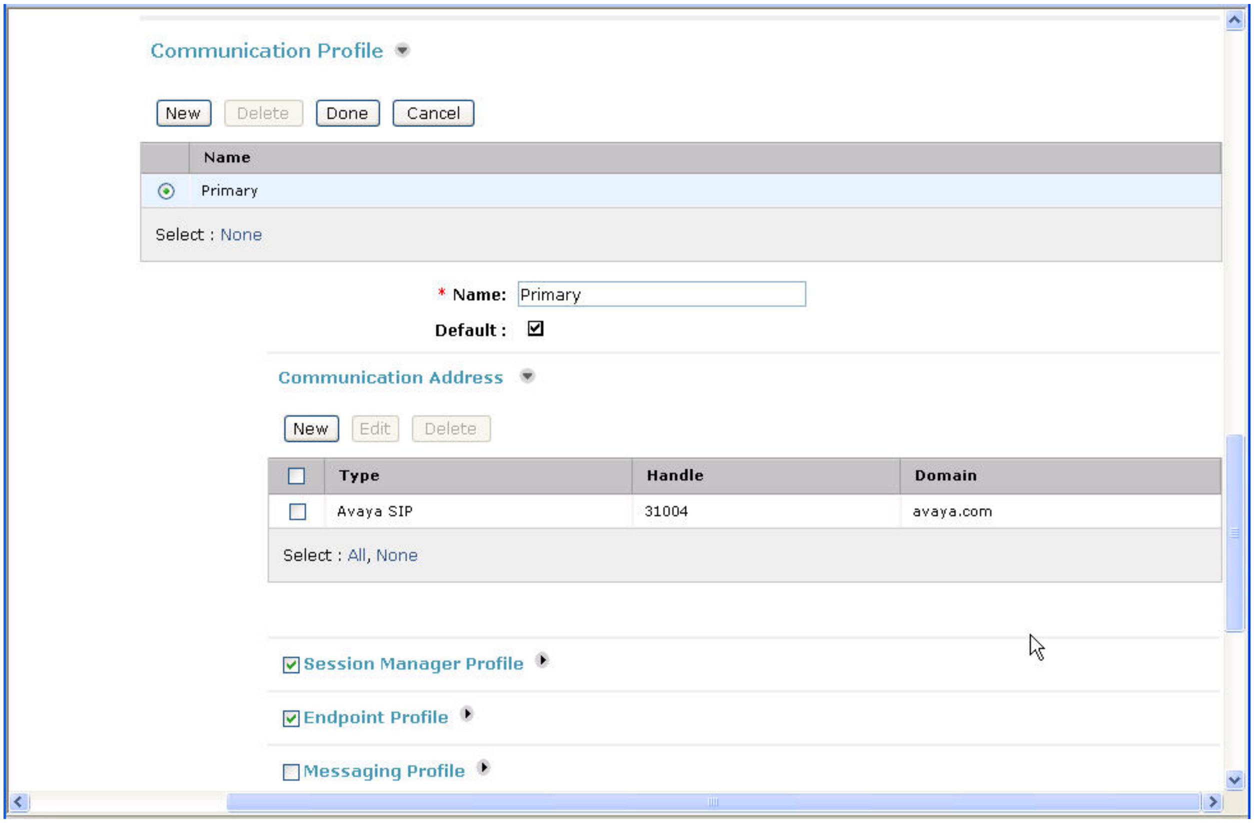
In the *Communication Profile* section, there are three sub-sections that need to be filled in: Communication Address, Session Manager Profile, and Endpoint Profile.



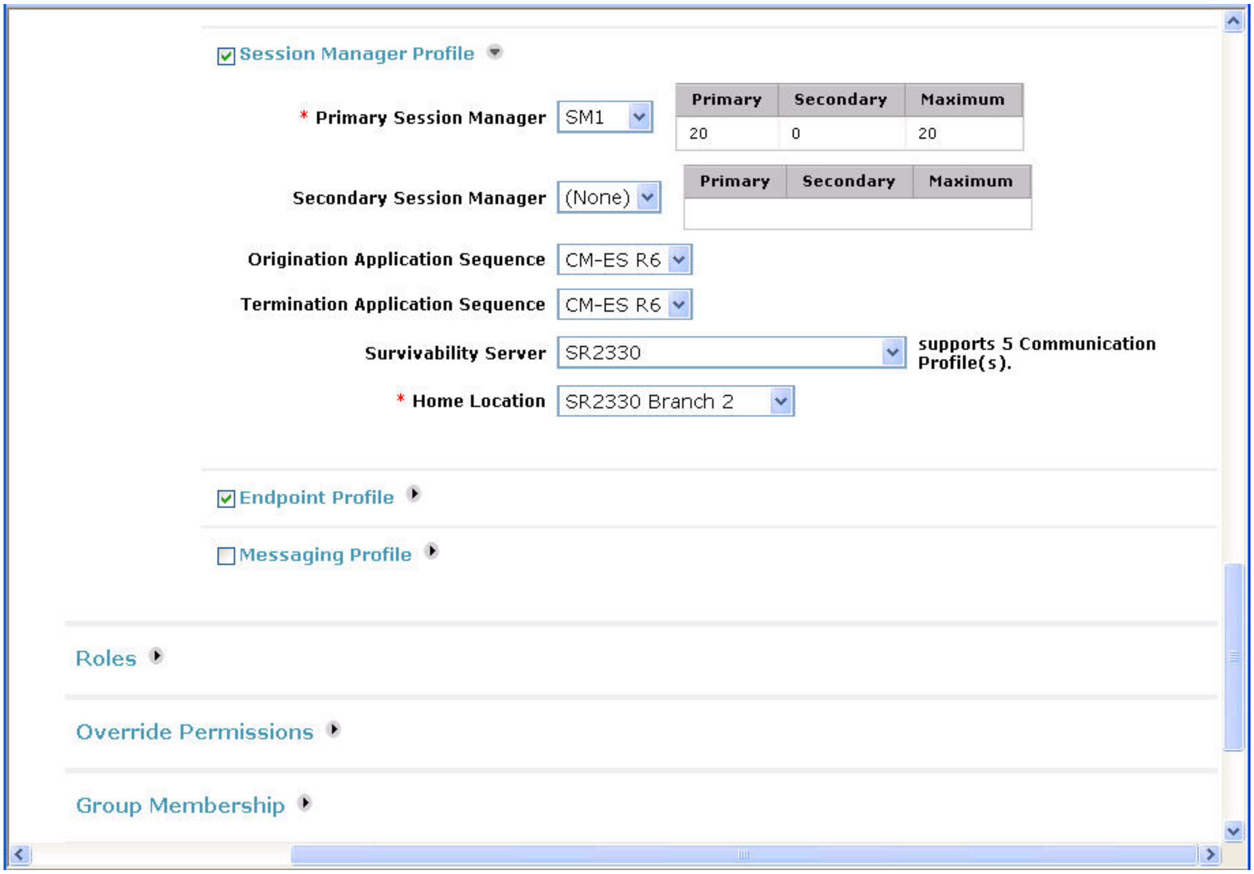
Click **New** under *Communication Address*. Enter appropriate values in the following fields and use defaults for the remaining fields:

- **Type**: Select "Avaya SIP"
- **Fully Qualified Address**: Enter the extension and select the domain as specified in **Section 4.1**

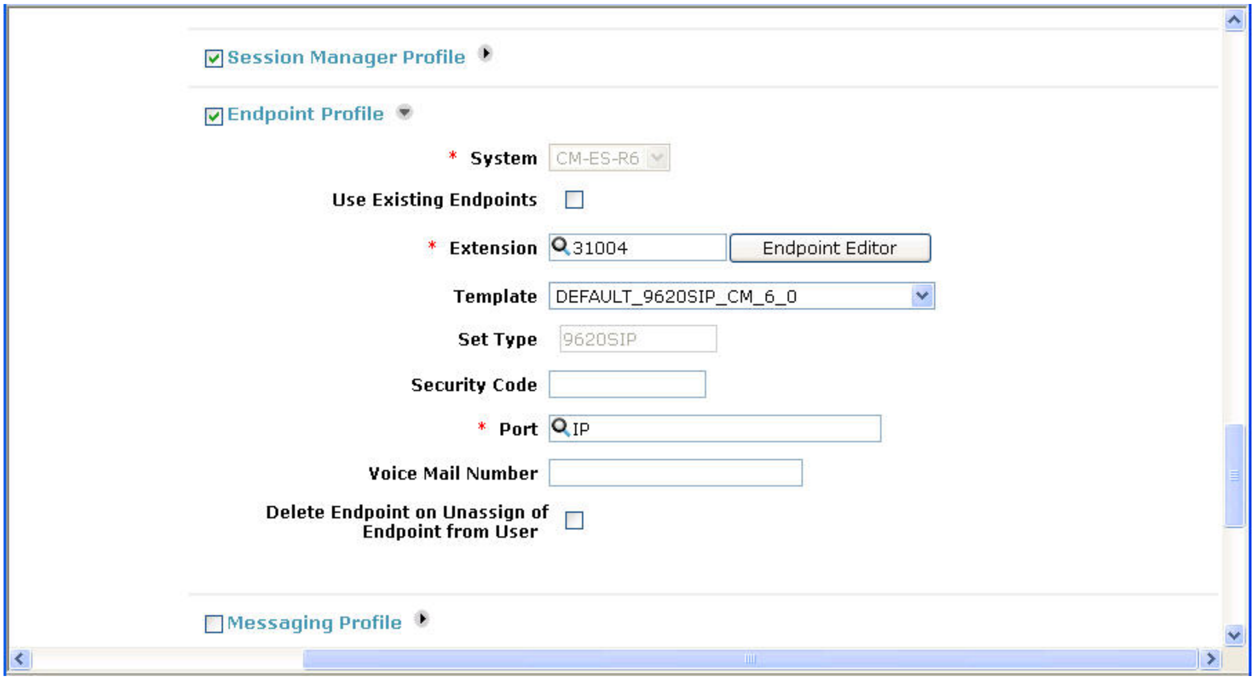
Click on **Add** to add the record with the above information. The table entry for the added record is shown in the screen below.



Click on the tick box next to **Session Manager Profile** and expand this section. Select the appropriate **Session Manager Instance** from the list. Select the appropriate **Origination** and **Termination Application Sequence** (these items should have already been configured as part of the Session Manager standard setup; see **Reference [4]** for details). Select the branch SR2330 SIP Element for **Survivability Server** and the branch Location for **Home Location**. The screen below shows what was used for extension 31004.



Click on the box next to **Endpoint Profile** and expand the section. Enter the appropriate **System**, which is the SIP Element for Communication Manager ES. Leave **Use Existing Stations** unchecked, causing Session Manager to automatically generate the station and Off-PBX station-mapping forms in Communication Manager[3]. Enter an **Extension**, and select "DEFAULT_9620SIP_CM_6_0" for the **Template**[4]. Leave the **Security Code** blank. Select "IP" for the **Port** field. The screen below shows what was used for extension 31004.



When done click 

at the bottom of the web page. Repeat the above steps for each telephone to be configured.

[3] System Manager uses the **Localized Display Name** field to populate the **Name** field in the station form in Communication Manager. Additional fields can be populated in Communication Manager later, if needed. See **Section 5.6**.

[4] This value for the **Template** can also be used for the analog telephone users supported by the FXS interfaces on the SR2330.

5. Configure Avaya Aura™ Communication Manager

This section shows the necessary steps to configure Communication Manager ES to support the Survivable SIP Gateway Solution. It is assumed that the basic configuration on Communication Manager and the required licensing, as well as the configuration required for accessing Avaya Modular Messaging have already been administered. See **Reference [1]** for additional information. The configurations on Communication Manager and the Avaya Media Gateway for routing calls to the PSTN (Route Pattern, PSTN Trunk/Signaling Groups, T1/E1 interfaces, etc.) are standard configurations out of scope of these Application Notes, and are therefore not included. All commands discussed in this section are executed on Communication Manager using the System Access Terminal (SAT).

The administration procedures in this section include the following areas. Some administration screens have been abbreviated for clarity.

- Configure IP Node Names
- Configure IP Codec Set
- Configure IP Network Regions
- Configure IP Network Map
- Configure SIP Signaling Group and Trunk Group
- Configure Private Numbering
- Configure Automatic Route Selection (ARS)
- Configure Route Pattern
- Update stations

5.1. Configure IP Node Names

Use the **change node-names ip** command to add an entry for the Session Manager that the Communication Manager will connect to. The **Name** "SM1" and **IP Address** "10.1.2.70" are entered for the Session Manager. The configured node-name "SM1" will be used later on in the SIP Signaling Group administration (**Section 5.5.1**).

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AS5400              10.3.3.40
Edge                10.3.3.60
Home1               10.3.3.50
Home2               10.3.3.41
SES                 10.3.3.50
SM1                 10.1.2.70
SurvCM              10.32.2.80
default             0.0.0.0
msgserver           10.3.3.14
procr               10.1.2.90
```

5.2. Configure IP Codec Set

The voice codec to be used is defined in the IP Codec Set form. For the sample configuration, a single codec set is used with a single codec defined. The **change ip-codec-set** command is shown below to define Codec Set 1 where the "G.711MU" codec is entered. On **Page 2** of the IP Codec Set form, set **Fax Mode** to "t.38-standard".

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2:
 3:
 4:
```

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         pass-through         0
    Clear-channel   n                    0:
```

5.3. Configure IP Network Regions

IP Network Regions are defined for each branch location as well as the Headquarters location as defined in **Table 1** using the **change ip-network-region** command. The values used in the sample configuration for Headquarters IP Network Region 1 are shown below. The **Name**, **Codec Set** and **Authoritative Domain** field values shown are specific to the sample configuration. All remaining fields have been left at default values. The **Authoritative Domain** is the SIP domain name defined on Session Manager (see **Section 4.1**) and used throughout the enterprise for SIP communications.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name: HQ CM and SIP Phones
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? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

The values used in the sample configuration for Branch 2 IP Network Region 10 are shown below (abbreviated). The **Name**, **Codec Set** and **Authoritative Domain** field values shown are specific to the sample configuration. All remaining fields have been left at default values. Follow the same steps to create the IP Network Regions for other branch locations.

```
change ip-network-region 10                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 10
Location:         Authoritative Domain: avaya.com
    Name: Branch SR2330
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? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
```

The following screen illustrates a portion of **Page 4** for IP Network Region 10. The connectivity between network regions is specified under the **Inter Network Region Connection Management** heading. For example, codec set "1" is specified for connections between network region 10 and network region 1. If bandwidth usage is a concern, a different codec set (e.g., G.729) could be defined that uses compression between the Headquarters and Branch 2, and would be specified here.

```
change ip-network-region 10                                   Page   4 of  20

 Source Region: 10    Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1     y    NoLimit                                             n       t
 2
```

5.4. Configure IP Network Map

IP addresses are used to associate a device with a specific IP Network Region. The **change ip-network-map** command is used to perform the IP address to IP Network Region mapping. The IP Address Mapping used in the sample configuration is shown below based on the information from **Table 1** (other entries in the table below were not used in the sample configuration). In this case, the full subnets for the Headquarters (10.1.2.0/24) and Branch 2 (20.20.20.0/24) were entered with the corresponding IP Network Region numbers.

```
change ip-network-map                                         Page   1 of  63
                              IP ADDRESS MAPPING

                                              Subnet Network     Emergency
 IP Address                                   Bits   Region VLAN Location Ext
 --------------------------------------------- ------ ------ ---- -------------
 FROM: 1.0.0.2                                /32    50     n
   TO: 1.0.0.2
 FROM: 10.1.2.0                               /24    1      n
   TO: 10.1.2.255
 FROM: 10.3.3.0                               /      1      n
   TO: 10.3.3.255
 FROM: 10.32.2.0                              /24    1      n
   TO: 10.32.2.255
 FROM: 20.20.20.0                             /24    10     n
   TO: 20.20.20.255
 FROM: 65.206.67.0                            /24    4      n
   TO: 65.206.67.255
```

5.5. Configure SIP Signaling Group and Trunk Group

A SIP signaling group and an associated trunk group were configured between Communication Manager ES and Session Manager in the sample configuration. The signaling and the trunk group were used for call signaling and media transport to/from SIP phones registered to Session Manager including phones in the sample branch (when in Normal Mode).

5.5.1. SIP Signaling Groups

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

- **Group Type**: "sip"
- **Transport Method**: "tcp"
- **Near-end Node Name**: "procr" node name for Communication Manager
- **Far-end Node Name**: "SM1" Session Manager node name
- **Near-end Listen Port**: "5060"
- **Far-end Listen Port**: "5060"
- **Far-end Network Region**: IP Network Region number "1" from **Section 5.3**
- **Far-end Domain**: SIP domain name from **Section 4.1** and **Section 5.3**
- **DTMF over IP**: "rtp-payload"
- **Alternate route Timer (sec)**: "10"

```
add signaling-group 60                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 60              Group Type: sip
  IMS Enabled? n         Transport Method: tcp
        Q-SIP? n                                               SIP Enabled LSP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 10
```

5.5.2. SIP Trunk Group

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

- **Group Type**: "sip"
- **Group Name**: Descriptive text
- **TAC**: An available trunk access code as per dialplan
- **Service Type**: "tie"
- **Signaling Group**: The signaling group number as configured in **Section 5.5.1**
- **Number of Members**: Equal to the maximum number of concurrent calls supported

```
add trunk-group 60                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 60                Group Type: sip          CDR Reports: y
  Group Name: SM1                      COR: 1        TN: 1        TAC: 160
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 60
                                                     Number of Members: 100
```

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

```
add trunk-group 60                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.6. Configure Private Numbering

Use the **change private-numbering 0** command to define the calling party number to be sent. In the example shown below, all calls originating from a 5-digit extension beginning with “3” (Headquarters and branch extensions) and routed across trunk group 60 (configured in **Section 5.5.2**) will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

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

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  2                                            5     Total Administered: 5
 5  3                60                          5        Maximum Entries: 540
 5  4                                            5
 5  5                                            5
 5  7                                            5
```

5.7. Configure Automatic Route Selection (ARS)

The ARS entries highlighted in this section focus on the local and long distance dialing from branch locations.

5.7.1. ARS Access Code

The sample configuration designates “9” as the ARS Access Code as shown below on the **change feature-access-codes** form. Calls with a leading 9 will be directed to the ARS routing table.

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: 621
         Abbreviated Dialing List2 Access Code: 622
         Abbreviated Dialing List3 Access Code: 623
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: 626
                       Answer Back Access Code: 625
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *5     Deactivation: #5
Call Forwarding Activation Busy/DA: *2     All: 612    Deactivation: #2
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: 624
                       Call Pickup Access Code: *6
CAS Remote Hold/Answer Hold-Unhold Access Code: #6
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

5.7.2. ARS Digit Analysis

The **change ars analysis x** command is used to make global routing entries where "x" is the dialed digit string to configure in the ARS digit Analysis table. The global ARS table used in the sample configuration is shown below. Calls to PSTN with **Dialed String** of 1 + 10 digits, and emergency 911 calls will select **Route Pattern** "60", which will select the SIP trunk to Session Manager. Note that all PSTN numbers are configured to route to Session Manager, which can use location based routing to determine the next SIP Element in the network to send the calls to. Also note that the sample configuration restricts PSTN calls to the 732 and 908 area codes with specific Exchange numbers. In real deployment, the dialed numbers in the ARS table should be modified as appropriate in accordance with business policies.

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

          Dialed            Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    1732829               11   11     60       hnpa          n
    1908848               11   11     60       hnpa          n
    911                   3    3      60       emer          n
                                                             n
```

5.8. Route Pattern

Use the **change route-pattern** command to modify the route pattern for calls routed to Session Manager. The changes made to Route Pattern 60 in the sample configuration are highlighted below. Route Pattern 60 uses SIP Trunk Group "60", which was configured to connect to Session Manager in **Section 5.5.2**. In the case of the sample configuration, this Route Pattern causes all digits to be sent to Session Manager for onward routing based on Dial Pattern and source Location. The Facility Restriction Level (**FRL**) specification allows access to this trunk, "0" being the least restrictive.

```
change route-pattern 60                                             Page   1 of   3
                    Pattern Number: 60  Pattern Name: SM FS
                             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: 60   0                    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: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
 4: y y y y y n  n            rest                                        none
```

5.9. Update Stations

A station must exist on Communication Manager for each SIP user account created in Session Manager. The extension assigned to the Communication Manager station must match the **Extension** under the **Station Profile** for the corresponding user on Session Manager. As described in **Section 4.7**, Session Manager will automatically create the station and Off-PBX station-mapping forms when the Session Manager user is created. The forms generated for the user added in **Section 4.7** are shown below, and are sufficient for basic calling features. Additional fields can be entered or existing fields edited by using the **change station** command. Typical fields modified are **Coverage Path 1** on **Page 1**, **MWI Served User Type** on **Page 2**, and various feature **BUTTON ASSIGNMENTS** on **Page 4**. Note that feature buttons are only supported on the Avaya 9600 SIP phones and not on the analog telephones connected to the FXS ports of the SR2330.

```
change station 31004                                          Page   1 of   6
                                     STATION

Extension: 31004                         Lock Messages? n              BCC: 0
     Type: 9620SIP                       Security Code:                 TN: 1
     Port: S00093                      Coverage Path 1: 60             COR: 1
     Name: UA2-SR2330                  Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 31004

        Display Language: english

          Survivable COR: internal
   Survivable Trunk Dest? y                            IP SoftPhone? n

                                                           IP Video? n
```

```
change station 31004                                          Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                        Coverage Msg Retrieval? y
                                                              Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
                                                Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
                                                       EC500 State: enabled

    MWI Served User Type: sip-adjunct

                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 31004        Always Use? n IP Audio Hairpinning? n
```

```
change station 31004                                          Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                              Headset? n
       Jack:                                              Speaker? n
      Cable:                                             Mounting: d
      Floor:                                          Cord Length: 0
   Building:                                            Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                           4: call-fwd   Ext:
 2: call-appr                           5: cpn-blk
 3: call-appr                           6:
```

The following screen shows **Page 1** of the updated STATION form for extension 31005 which is assigned to the one-X Communicator soft phone. Note the setting for **IP SoftPhone?** is set to **y**.

```
change station 31005                                          Page   1 of   6
                                     STATION

Extension: 31005                        Lock Messages? n               BCC: 0
     Type: 9600SIP                      Security Code:                  TN: 1
     Port: S00102                     Coverage Path 1: 60              COR: 1
     Name: UA3-SR2330                 Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
             Loss Group: 19        Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 31005
           Speakerphone: 2-way               Mute Button Enabled? y
       Display Language: english                Expansion Module? n
 Survivable GK Node Name:
         Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? y

                                             IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default
```

The following are the **off-pbx-telephone station-mapping** screens automatically generated by Session Manager. No changes to this form are typically required**.**

```
change off-pbx-telephone station-mapping 31004                Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 31004            OPS          -      31004            aar        1
```

Page 2 is of the form is displayed below.

```
change off-pbx-telephone station-mapping 31004                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 31004         OPS      3        both         all         none
```

Repeat any desired modifications to stations added by Session Manager. The following list command output summarizes the configuration relevant to the branch phones in the sample configuration. Use the command **list off-pbx-telephone station-mapping 31*** to note the station extensions created. Each Avaya SIP Telephone at the branch (e.g., 31003 and 31004), each analog device connected to an FXS port on the SR2330 (e.g., 31001 and 31002), and each one-X Communicator soft phone (e.g., 31005) can be observed. The corresponding registration of these users to Session Manager is shown in **Section 8.3**.

```
list off-pbx-telephone station-mapping 31*

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station        Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                  Set    Select  Mode         Allowed

31001          OPS        31001            1  /   aar     both         all
31002          OPS        31002            1  /   aar     both         all
31003          OPS        31003            1  /   aar     both         all
31004          OPS        31004            1  /   aar     both         all
31005          OPS        31005            1  /   aar     both         all
```

6. Configure Avaya SR2330 Secure Router

Presented below is an annotated version of the SR2330 Secure Router configuration used in Branch. There are two main SIP components in the SR2330 as used in the survivable SIP Gateway solution: the SIP Media Gateway and the SIP Survivability Module (SSM). The SIP Media Gateway implements a Back-to-Back User Agent (B2BUA) and provides call processing support for locally connected FXS and FXO ports, as well as SIP to PSTN gateway capabilities. It registers on behalf of its configured FXS ports to Session Manager in the Normal Mode. The branch 9600 SIP telephones and the one-X Communicator SIP soft phone register to Session Manager as their primary SIP registrar, and simultaneously to the SSM as the secondary registrar. See **Section 2.7** and **Figure 3** for more details on the SIP signaling configuration implemented below.

The entries in the SR2330 configuration contained in this section were the actual commands used to have the default settings on SR2330 changed for the sample configuration. Default settings on SR2330 not relevant to the sample configuration are not listed.

```
# SR2330 system configuration file (.CFG).

system logging
  console
    priority crit
```

```
    no enable
    exit console
  syslog
    host_ipaddr 10.1.2.49
    module alarms local0 none
    module dos local0 none
    module forwarding local0 none
    module voip-ssm-cdr local0 none
    module voip-cdr local0 none
    enable
    exit syslog
  exit logging
hostname SR

# WAN side Ethernet interface
interface ethernet 0/1
  ip address 10.1.2.68 255.255.255.0
  exit ethernet

# LAN side Ethernet interface
interface ethernet 0/2
  ip address 20.20.20.1 255.255.255.0
  exit ethernet

ftp_server

telnet_server
telnet_timeout 0

# Required to support domain lookup in survivable mode.
# Avaya 9600 IP Telephones use the domain name (avaya.com) in the
# Refer-To header of REFER messages during transfer scenarios.
# This entry allows the SR2330 to resolve the domain name to its
# private side IP address.
ip host_add avaya.com 20.20.20.1

# Default route is toward the WAN
ip route 0.0.0.0/0 10.1.2.1

# Dialed digit translations used in survivable mode to strip the first 9
# before sending out on the local FXO trunk
voice translation-rule 1
  rule 1 /9.........../ //
  rule 2 /9911/ /911/
  exit translation-rule
voice translation-profile strip9
  translate called 1
  exit translation-profile

voice service voip
  # sip signaling options
  # Configure the SIP Media Gateway to listen on port 5080
  sip
    bind all ipv4:20.20.20.1:5080
    rel1xx disable
    exit sip
  # enable t.38 fax
  fax protocol t38
```

```
  # One single codec must be configured for calls
  # to be supported from FXS analog phones to 9600 SIP phones
  # in survivable mode.
  codec 1 g711ulaw 160
  # Support for RFC 2833 DTMF. Configuration of rtp payload-type must match
  # the DTMF_PAYLOAD_TYPE setting in 46xxsettings.txt file for Avaya 96xx
  # phones
  rtp payload-type nte 101
  dtmf-relay rtp-nte

  # SIP Survivable Module
  ssm
  # To configure the SIP Survivability Module, bind the IP interface
  # for SIP traffic using default port 5060
    bind ip ipv4:20.20.20.1
  # Enable the SSM
    enable
    sip-server
      # Specify the SSM domain name to be used with Session Manager
      domain dns:avaya.com
      exit sip-server
  # Point the SSM to the SIP Media Gateway IP interface as the
  # default gateway (specifying the non-default port)
    default-gateway ipv4:20.20.20.1:5080 transport tcp
    exit ssm
  exit voip

# SIP user registration parameters
sip-ua
  # Primary Registrar is Session Manager, secondary is SSM
  sip-server ipv4:10.1.2.70
  sip-server ipv4:20.20.20.1:5060 secondary
  transport tcp
  registrar ipv4:10.1.2.170 expires 3600
  # Enable SIP OPTIONS messages for keepalives
  keepalive target sip-server
  keepalive target sip-server secondary
  keepalive timer 40
  exit sip-ua

# FXO trunk group name (will include 3/1 and 3/2)
trunk group pstn

# FXO ports

voice-port 3/1
  signal loop-start
  # battery-reversal configuration is specific to the analog line from
  # specific service provider.  The default setting “no battery-reversal
  # answer” was used in the validation test with Verizon-provisioned analog
  # lines.  The configuration “battery-reversal answer” might need to be used
  # for analog trunk lines from other service providers.
  ring-number 2
  caller-id enable 1
  station number 9087661495
  no shutdown
  # Calls into this FXO port are routed to this extension (via Session
```

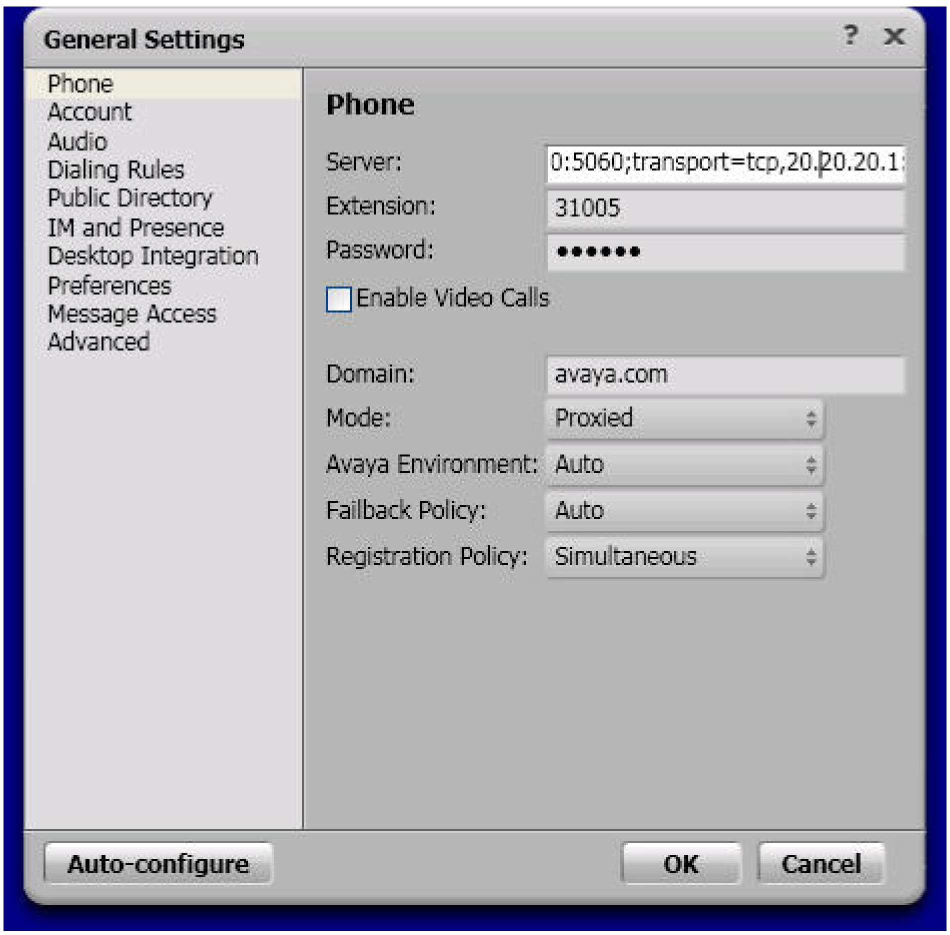
```
  # Manager or SMM depending on normal or survivable mode, respectively
  connection plar 31003
  trunk-group pstn
  exit voice-port

voice-port 3/2
  signal loop-start
  ring-number 2
  caller-id enable 1
  station number 9087661527
  no shutdown
  # Calls into this FXO port are routed to this extension (via Session
  # Manager or SMM depending on normal or survivable mode, respectively
  connection plar 31002
  trunk-group pstn
  exit voice-port

# FXS ports for the two analog phones
voice-port 1/1
  signal loop-start
  # User name
  station name fxs1
  # Extension number
  station number 31001
  no shutdown
  # Interdigit timeout governs how fast call is launched when user stops
  # dialing
  timeouts interdigit 3
  exit voice-port

voice-port 1/2
  signal loop-start
  station name fxs2
  station number 31002
  no shutdown
  timeouts interdigit 3
  exit voice-port

# Routing for 1-AAA-NNN-CCCC calls dialed by branch phones that are delivered
# by Session Manager
dial-peer voice pots 6
  destination-pattern 1.%
  # Send out local FXO ports
  trunkgroup pstn
  # Send all digits
  forward-digits all
  no shutdown
  exit pots

# Routing for calls to the local analog phones on FXS ports
dial-peer voice pots 1
  # Extension as known by ACM & ASM
  destination-pattern 31001
  port 1/1
  forward-digits all
  no shutdown
  # Userid and password used to register to Session Manager on behal
  # of the phone
```

```
  authentication 31001 31001
  register e164
  exit pots

dial-peer voice pots 2
  destination-pattern 31002
  port 1/2
  forward-digits all
  no shutdown
  # Userid and password used to register to Session Manager on behal
  # of the phone
  authentication 31002 31002
  register e164
  exit pots

# Same routing as for dial peer pots #6, except it handles dialing the
# initial “9”
dial-peer voice pots 5
  destination-pattern 91.%
  trunkgroup pstn
  forward-digits all
  no shutdown
  # Strip the “9”
  translation-profile outgoing strip9
  exit pots
```

7. Configure Avaya SIP Phones

The Session Manager 6.0 Release supports configuration of a Survivability Server for SIP Users (see screenshot for *Session Manager Profile* in **Section 4.7**). However the specific pre-GA version of the Session Manager software used in the validation test does not support this capability yet. Consequently, the specification of the Survivability Server for the Avaya 9600 Series SIP Telephones is handled the same way as with previous versions of the Session Manager software using the phone configuration file 46xxsettings.txt.

The specification of the Survivability Server for one-X Communicator SIP soft phone is set by manually typing in the same setting for **SIP_CONTROLLER_LIST** in 46xxsettings txt in the **Server** field of the **Phone** configuration screen:



The following 46xxsettings.txt file parameters/values were used for the testing:

46xxsettings.txt Parameter Name	Value Used in Sample Configuration	Description
SIP_CONTROLLER_LIST	10.1.2.70:5060;transport=tcp, 20.20.20.1:5060;transport=tcp	A priority list of SIP Servers for the phone to use for SIP services. The port and transport use the default values of 5061 and TLS when not specified. The setting used in the sample configuration shows the values used for this parameter for a phone in Branch 2. The Session Manager is the first priority SIP Server listed using port and transport of 5060 and TCP. Separated by a comma, the Branch 2 Avaya SR2330 Secure Router is the next priority SIP Server using port 5060 and TCP transport. The SIP Server list for each branch would require different values for the SIP_CONTROLLER_LIST, e.g. the list for Branch 1 phones will include the Session Manager and the Branch 1 Avaya SR2330 Secure Router while the list for Branch 2 phones will include the Session Manager and the Branch 2 Avaya SR2330 Secure Router . To accomplish this, the GROUP system value mechanism can be implemented as described in [8].
FAILBACK_POLICY	Auto	While in Survivable Mode, determines the mechanism to use to fail back to the centralized SIP Server. **Auto** = the phone periodically checks the availability of the primary controller and dynamically fails back.
FAST_RESPONSE_TIMEOUT	2	The timer terminates SIP INVITE transactions if no SIP response is received within the specified number of seconds after sending the request. Useful when a phone goes off-hook after connectivity to the centralized SIP Server is lost, but before the phone has detected the connectivity loss. The default value is 4 seconds. After the SIP INVITE is terminated, the phone immediately transitions to Survivable Mode.
MSGNUM	33000	The number dialed when the Message button is pressed and the phone is in Normal Mode.

46xxsettings.txt Parameter Name	Value Used in Sample Configuration	Description
PSTN_VM_NUM	19088485960	The number dialed when the Message button is pressed and the phone is in Survivable Mode.
RECOVERYREGISTERWAIT	60	A Reactive Monitoring Interval. If no response to a "maintenance check" REGISTER request is received within the timeout period, the phone will retry the monitoring attempt after a randomly selected delay of 50% - 90% of this parameter.
DIALPLAN	[2-8]xxxx | 91xxxxxxxxxx | 9xxxxxxxxxx | 9911 | 911	Enables the acceleration of dialing when the WAN is down and the Avaya SR2330 Secure Router is active, by defining the dial plan used in the phone. In normal mode, the Avaya telephone does not require these settings to expedite dialing. The dialplan values used in the phone will generally match the values used by the Avaya SR2330 Secure Router in **Section 7**. See [8] for additional format details on the DIALPLAN parameter.
PHNEMERGNUM	911	Number dialed when emergency soft key is pressed.
DISCOVER_AVAYA_ENVIRONMENT	1	Automatically determines if the active SIP Server is an Avaya server or not.
SIPREGPROXYPOLICY	simultaneous	A policy to control how the phone treats a list of proxies in the SIP_CONTROLLER_LIST parameter **alternate** = remain registered with only the active controller **simultaneous** = remain registered with all available controllers
DTMF_PAYLOAD_TYPE	101	Specifies the RTP payload type to be used for RFC 2833 signaling. Set to be compatible with corresponding setting on SR2330.
SIPDOMAIN	avaya.com	The enterprise SIP domain. Must be the same for all SIP controllers in the configuration. SIPDOMAIN is set to “avaya.com” in the sample configuration.

Table 3 – Configuration Entries in 46xxsettings.txt File

8. General Test Approach and Test Results

This section describes the validation test used to verify the sample configuration for the Survivable SIP Gateway Solution using the Avaya SR2330 Secure Router in the branch. This section covers the general test approach and the test results.

8.1. General Test Approach

The general test approach was to break and restore network connectivity from the branch site to the Headquarters site to verify that:

- When network connectivity is broken, the branch SR2330 automatically assumes the SIP proxy and SIP registrar functions. In this Survivable Mode, the branch phones can still call each other and reach PSTN through the SR2330 FXO analog trunk interface.
- When network connectivity is restored, the Session Manager at the Headquarters location automatically assumes the SIP proxy and SIP registrar functions for providing centralized SIP call control. In this Normal Mode, PSTN access by phones at both the headquarters and branch sites are through the Avaya Media Gateway at the central location with the exception that local non-toll calls from the branch phones are routed to the PSTN through the branch SR2330.

8.2. Test Results

The following features and functionality were verified. Any observations related to these tests are listed at the end of this section:

- In Normal Mode, the Session Manager located at the central site serves as the SIP registrar and proxy for phones at both the central and branch sites; in Survivable Mode, the SR2330 located at the branch location serves as the SIP registrar and proxy for the branch phones.
- Branch phones register to the Session Manager and the branch SR2330 simultaneously. Switching between the Normal and the Survivable Modes is automatic and within a reasonable time span (see **Section 9.5** and **9.6**).
- In Normal Mode, calls can be placed between phones at the main site and the branch site, and among phones within the site.
- In Normal Mode
  - With Distributed Trunking configuration, local non-toll calls from the branch are routed to the PSTN through the FXO interfaces on the branch SR2330; long-distance toll calls from the branch phones are routed to the PSTN through the Avaya Media Gateway at the central location.
  - With Centralized Trunking configuration, all calls to the PSTN from the branch, including both local and long distance toll calls, are routed to the PSTN through the Avaya Media Gateway at the central location.
- In Survivable Mode, calls can be placed among phones within the branch. In addition, branch phones can still place calls to the PSTN (and to the phones at Headquarters via PSTN) using the FXO interface on the branch SR2330.
- PBX features including Hold, Transfer, Call Waiting, Call Forwarding and Conference on Avaya 9600 SIP Phones in both Normal and Survivable Modes.
- Analog phones connected to the FXS ports on the branch SR2330 register properly to the central Session Manager in Normal Mode.

- Messaging system access by branch phones (through internal access number in Normal Mode and PSTN call in Survivable Mode) and proper function of MWI (Messaging Waiting Indicator) on the Avaya 9600 IP Phones in the Normal Mode.
- Faxing between the branch and the Headquarters (in Normal Mode) and between the branch and PSTN (in both Normal Mode and Survivable Mode).
- Proper system recovery after power outage and loss/restoration of IP connection.

The following problems were observed in the validation testing:

1. **Calls from branch to PSTN via FXO interface on the branch SR2330 require longer Alternate Route Timer on Communication Manager ES**.
   The timer (on the Signaling Group) was increased to 10 seconds from the default setting of 6 seconds, otherwise call fails.

2. **Hold (switch-hook flash) is not supported for SR2330 FXS phones**.
   This may be supported in the next release after 10.2.1. During the validation test, the hold is a local phone function on the Avaya 6211 analog set.

3. **Domain name is required to be configured in SR2330 to be used in Survivable Mode**.
   If a domain name is not configured in SR2330, the transfer will fail because the Refer-To header sent by the Avaya 9600 SIP phone contains the FQDN domain name, and the SR2330 cannot resolve it.

4. **Reliability of provisional responses must be disabled in SR2330 in order for call from SIP phone to FXS port to be successful in Survivable Mode**.
   If not, the call will fail since the Avaya 9600 SIP phone will send BYE immediately after it has received a *200 OK* without SDP from SR2330. SR2330 does not include SDP in the *200 OK* since SDP negotiation has happened with previous PRACK exchanges. The command to disable reliability of provisional responses on SR2330 is “rel1xx disable” under configure/voice/service/voip/sip.

5. **Caller ID is not supported on SR2330 for SIP phones in Survivable Mode**
   SIP phone display will show calling/called numbers. However, Caller ID is supported for FXS-connected phones.

6. **“Alternate” registration mode must be configured on the SIP 9600 phone for DTMF tones to work properly**.
   With “Simultaneous” registration mode, any key press on the Avaya SIP 9600 phone generates double DTMF tones. The Avaya 9600 phone development indicated that the early versions of 96xx 6.0 firmware (including the version used in the validation test) did have problems with DTMF that would be fixed in later versions.

7. **Media Shuffle must be turned off for one-X Communicator to work with Modular Messaging in Normal Mode**.
   With Media Shuffle on, Modular Messaging does not detect any DTMF tones generated from one-X Communicator.

8. **Faxing from Headquarters (fax machine connected to a port on the Analog Module in the Avaya G450 Media Gateway) to Branch (fax machine connected to an SR2330 FXS port) fails with tested Communication Manager software**.
   The failure was due to an incorrect setting of *T.38MaxBitRate* in the SDP contained in the *200 OK* response to a SIP RE-INVITE for T.38 fax from SR2330. This defect is fixed in Communication Manager Release 6.0 Build 345 and later.

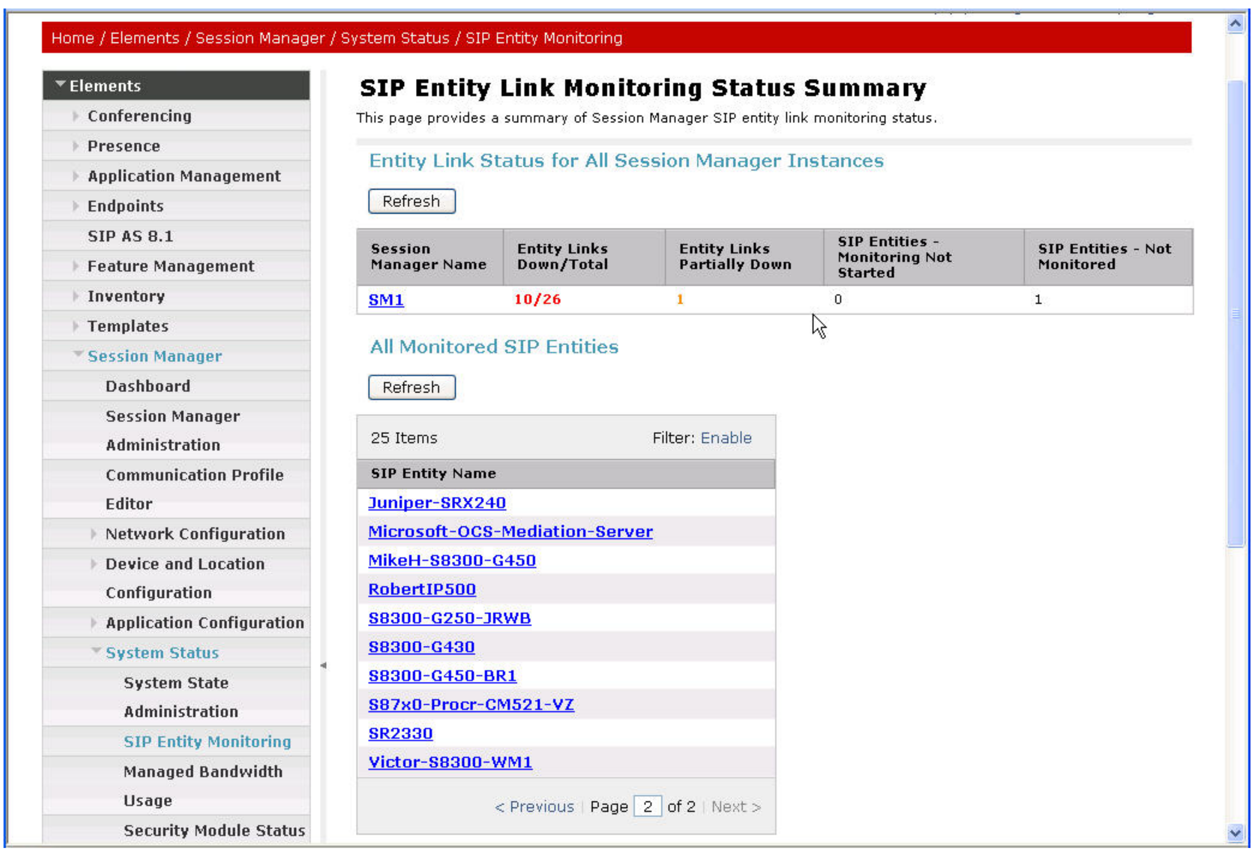
9. **SR2330 does not provide local ring-back for PSTN calls from branch phones in Survivable Mode**.
   The calling phone hears silence for about 5 seconds before hearing ring-back from PSTN.

9. Verification Steps

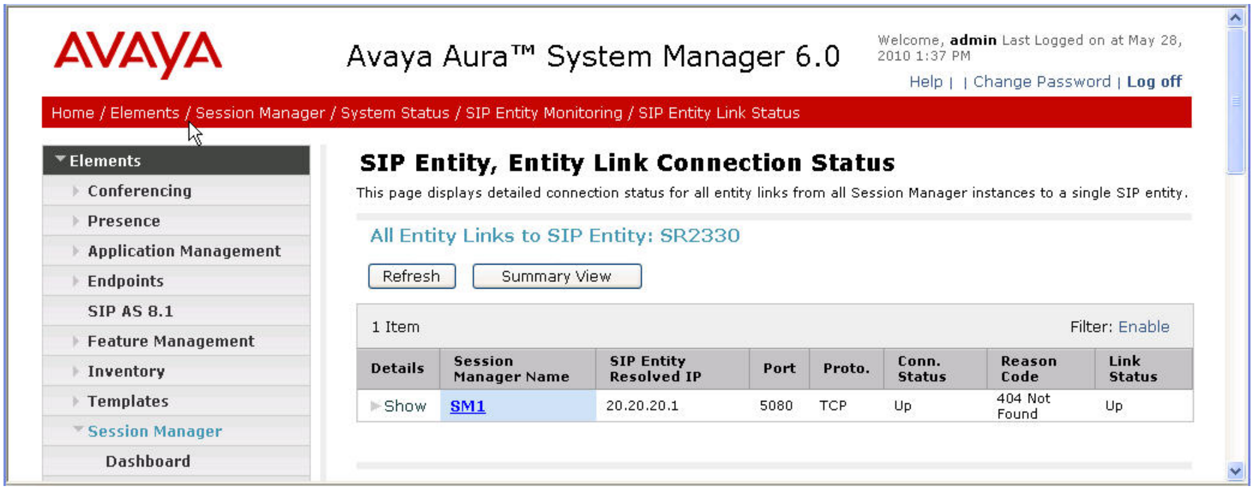
This section contains the procedures that can be used to verify the sample configuration.

9.1. Avaya Aura™ Session Manager/Avaya Secure Router 2330 Link Status

From the left navigation panel of the browser-based GUI of System Manager, select **Elements → Session Manager → System Status → SIP Entity Monitoring**. The SIP entities monitored by Session Manager are listed on the lower portion of the **SIP Entity Link Monitoring Status Summary** page as shown below. Select the entity corresponding to the SR2330, in this case "SR2330".



The next screen displayed will show the entity link status, which is determined by the SR2330 response to a SIP OPTIONS message periodically sent by Session Manager. In this case the "404 Not Found" response by the SR2330 indicates that SIP signaling on the SR2330 is functional, and so the **Conn. Status** is shown as "Up".



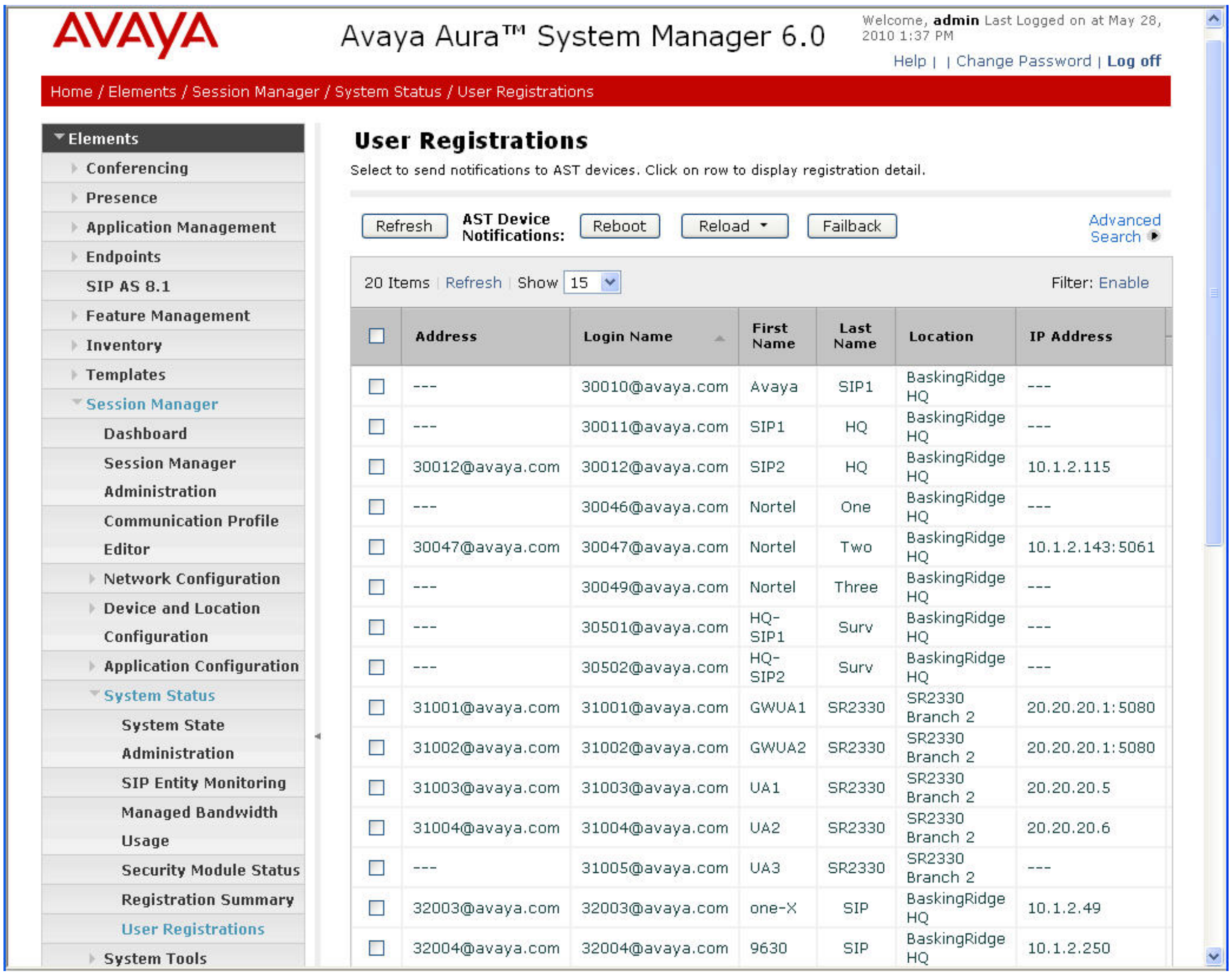
9.2. Avaya Secure Router 2330 Registered Users

The command line interface of the SR2330 can be used to determine the registration state of the Analog phones connected to the FXS interfaces of the SIP Media Gateway Module. Below are shown the command and the output, which indicates that the two FXS interfaces configured with extensions 31001 and 31002 have successfully registered with Session Manager.

```
SR# show sip-ua register status


Line      peer                               expires(sec)
registered
4         31001                              3600          yes
5         31002                              3600          yes
```

9.3. Avaya Aura™ Session Manager Registered Users in Normal Mode

To verify registration of the SR2330 supported analog stations connected to the FXS interfaces, the Avaya 9600 IP phones and the one-X Communicator soft phone, navigate to **Elements → Session Manager → System Status → User Registrations**. The screen below shows branch extensions 31001 and 31002 corresponding to the analog stations, 31003 and 31004 corresponding to the Avaya 9600 IP phones, and extension 31005 corresponding to the soft phone. Extensions 30011 and 30012 are Avaya 9600 IP phones at the Headquarters. Note that extension 30011 at the Headquarters and extension 31005 at the branch were not logged in at the time the screenshot was taken.



9.4. Verify Basic Calls

In the Normal Mode, make calls between Headquarters and the branch; verify that the calls are successful with two-way talk-path. Make calls between the PSTN and the branch through the Headquarters; verify that the calls are successful with two-way talk-path.

In the Survivable Mode, make calls between branch phones; verify that the calls are successful with two-way talk-path. Make calls between the PSTN and the branch through the FXO interfaces on the branch SR2330; verify that the calls are successful with two-way talk-path.

9.5. Timing Expectations for Fail-over to Survivable Mode

In the event that a failure occurs and a branch is unable to communicate with the central Session Manager there will be a length of time before Avaya 9600 SIP Telephones acquire service from the Secure Router 2330. This section is intended to set approximate expectations for this length of time. In practice, failover timing will depend on a variety of factors. Using the configuration described in these Application Notes, when the IP WAN is disconnected, idle Avaya 9600 SIP phones in the branch will typically display the "Acquiring Service…" screen in approximately 75-120 seconds. With multiple identical idle phones in the same branch, it would not be unusual for some phones to register to the SR2330 before others, with the earliest registering in approximately one minute and the latest registering in approximately two minutes.

Note that attempting to place a call from a branch phone during this time will trigger acquisition of service from the SR2330 immediately. Likewise, if a phone receives an intra-branch call before it has acquired service, it will do so immediately and then the call will be successfully delivered. In other words, the Avaya SIP Telephones in the branch can typically place and receive calls processed by the SR2330 approximately two minutes after the branch is isolated by a WAN failure.

9.6. Timing Expectations for Fail-back to Normal Mode

This section is intended to set approximate expectations for the length of time before Avaya 9600 SIP phones registered to the SR2330 in Survivable Mode will re-acquire service from Session Manager for normal service, once branch communication with the central Session Manager is restored. In practice, failover timing will depend on a variety of factors. Using the configuration described in these Application Notes, when the IP WAN is restored such that the branch telephones can again reach the Session Manager, idle Avaya 9600 SIP phones in the branch will typically be registered with Session Manager in one minute or less. With multiple identical idle phones in the same branch, it would not be unusual for some phones to register back with the Session Manager before others. For example, some may register within 15 seconds, others within 30 seconds, with others registering in approximately one minute.

10. Conclusion

SIP endpoints deployed at remote branch locations risk a loss of service if a break in connectivity to the centralized SIP call control platform occurs. Connectivity loss can be caused by WAN access problems being experienced at the branch or network problems at the centralized site blocking access to the Avaya SIP call control platform. The Avaya Survivable SIP Gateway Solution minimizes service disruptions to the remote branch SIP endpoints.

These Application Notes present the configuration steps to implement the Survivable SIP Gateway Solution using Avaya Secure Router 10.2.1 with Avaya Aura™ Communication Manager 6.0 (pre-GA), Avaya Aura™ Session Manager 6.0 (pre-GA) and Avaya Modular Messaging 5.2. The validation test verified the configuration for the Avaya Survivable SIP Gateway Solution .

11. References

Avaya Application Notes and additional resources can be found at the following web address http://www.avaya.com/usa/resources/. Product documentation for Avaya products may be found at http://support.avaya.com/.
The following Avaya references are relevant to these Application Notes. Note that pre-6.0 versions of the Avaya documents are listed since Release 6.0 of the Avaya Aura™ products is not GA yet:

[1] *Administering Avaya Aura™ Communication Manager*, Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

[2] *Administering Network Connectivity on Avaya Aura™ Communication Manager*, Issue 14, May 2009, Document Number 555-233-504.

[3] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206.

[4] *Administering Avaya Aura™ Session Manager*, Release 5.2, Issue 2.0, November 2009, Document Number 03-603324.

[5] *Avaya Aura™ Communication Manager Screen Reference*, Issue 1.0, May 2009, Document Number 03-602878.

[6] *Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2* — Issue 1.0, February 2010, Avaya Solution Interoperability Lab Application Notes.

[7] *Avaya one-X® Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.5*, Issue 5, November 2009, Document Number 16-601944.

[8] *Avaya one-X® Communicator Troubleshooting* — Issue 3, December 2009, Document Number 16-603218.

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.