Configure a Survivable SIP Gateway using the Avaya Secure Router 10.3.3 in an Avaya Aura® Infrastructure using Centralized Trunking – Issue 1.0

Abstract

These Application Notes present a sample configuration of a Survivable SIP Gateway Solution using the Avaya Secure Router 10.3.3, Avaya Aura® Session Manager 6.2 FP1, Avaya Aura® Communication Manager 6.2.3 Avaya Aura® Conferencing 6.0.1, and Avaya Aura® Messaging 6.2 in a Centralized Trunking Configuration.

In a centralized SIP server architecture, the remote branches make use of the call processing resources available at a central location, generally located at the corporate headquarters. The SIP survivability feature enhances the feature set of the Avaya Secure Router 2330 (SR2330) by providing business continuity to the branch office in the event of a SIP server failure or of a WAN connection outage to the corporate headquarters.
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1. Introduction

These Application Notes present a sample configuration of a Survivable SIP Gateway Solution using the Avaya Secure Router, Avaya Aura® Session Manager 6.2, Avaya Aura® Communication Manager 6.2.3, Avaya Aura® Conferencing 6.0.1 and Avaya Aura® Messaging 6.2 in a Centralized Trunking Configuration. With this feature, employees at the branch office can continue to use SIP phones to place and receive intra-site calls and make calls over the PSTN, including 911 calls, after a SIP server or WAN connection failure. The SIP survivability module (SSM) is a software-only subsystem on the SR2330 that provides SIP survivability capabilities. The SSM operates as a SIP Back to Back User Agent (B2BUA) that can back up a central SIP server by providing basic call services to connected endpoints at the branch if the WAN connection to the central SIP server fails.

A Remote Branch Office (RBO) makes use of the VoIP infrastructure deployed at a central corporate location. During normal operation when the WAN is up and running, routing decisions for all calls (intra RBO, inter-RBO, PSTN) are made at the central infrastructure deployed at the headquarters. When the central SIP server or WAN connection goes down, the Branch phones and SIP Gateway enter the survivable mode and start sending calls to the SSM instead of sending calls to the SM. SSM handles intra-RBO calls and PSTN calls. SSM routes the PSTN calls to SIP Gateway which in turn routes the calls over trunks to PSTN network. Inter-RBO VoIP calls cannot be handled because the central infrastructure is not reachable. However, Inter-RBO calls can still be made if they are routable using PSTN trunks.

The SIP Media Gateway provides connections to FXS phones and to PSTN trunks, and the SSM provides survivability capabilities for the SIP and analog Phones.

The SR2330 SSM operates in two modes – Normal mode and Survivable mode. In normal mode, sip phones, analogue phones register and make calls directly (SSM not involved) to the SIP Server located in the head office. In survivable mode, the SSM or the SIP survivable Module supports SIP server functionality to provide basic call features to the SIP endpoints at the branch, and also supports local registrar functionality to store registrations. Survivable mode is also called backup mode as the SSM functions as a backup server to the central SIP server in this mode. In the configuration with Avaya Aura® setup, SSM will always be in the backup mode.

1.1. 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 (PRI interfaces to PSTN, FXS interfaces to analog endpoints). SR2330 also provides FXO, BRI, E1R2 and CAS interfaces, not covered in these application notes. The test configuration has been with T1 PRI, but other trunk interfaces can be configured as per the PSTN branch link.
- SIP Survivability Module (Registrar and Proxy)

When the SR2330 is serving the Registrar/Proxy role, it is said to be in Survivable Mode.
1.2. Avaya SIP 96xx and 96x1 Series IP Telephone
The Avaya SIP 96xx and 96x1 Series IP Telephones are key components of the Survivable SIP Gateway Solution. The 6.2 firmware release of the Avaya 96x1 (9608, 9611, 9621 and 9641) and 2.6.7 firmware release of the 96xx (9620, 9630, 9640, and 9670) SIP Phone include feature capabilities specific to SIP survivability, enabling the phones to monitor connectivity to Avaya Aura® Session Manager and dynamically fail over to the local SR2330 as an alternate or survivable SIP server. Please see References [11] & [12] in Section 11 for additional information on the Avaya 96xx/96x1 SIP Phones.

1.3. Avaya Aura® Communication Manager
The Headquarter or Datacenter location in the sample configuration includes a Communication Manager that acts as the Evolution Server. Avaya Aura® Communication Manager supports non-SIP, SIP and centralized PSTN trunks.

2. Interoperability Testing
The testing covered interoperability testing of Secure Router 2330 at the branch location with Avaya Aura® Communication Manager Evolution Server Release 6.2 FP1 and Avaya Aura® Session Manager Release 6.2 FP1 at the Core Location. This includes integration with centralized applications such as Avaya Aura® Messaging and Avaya Aura® Conferencing.

2.1. 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 Avaya Aura® Session Manager and Avaya Aura® Communication Manager over the corporate network.

In a sample scenario of 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. The Centralized Trunking arrangement has the same network configuration as the Distributed Trunking except that in Normal Mode, all PSTN calls from the branch, both local and long-distance toll calls, are routed through the central T1/E1 facilities on the Avaya G650 Media Gateway located at the Headquarters location.

The call flows presented in Section 2.3 provide additional details of how calls are routed based on the location of the caller and the number being called.

Distributed Trunking: Local calls from a branch location can be routed back to the same branch location and terminate on the PRI 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 sample configuration presented in these Application Notes implements a Centralized Trunking configuration.

2.2. Call Routing in Centralized Trunking Mode

1. Branch Analog phone to Headquarter Phone:
   When a call is initiated from an analog phone or other POTS endpoint at the branch, the default SR2330 operation is to build a SIP Uniform Resource Identifier (URI) and forward a SIP Invite to the central SIP server for call routing. The SIP server IP address is specified on the SR2330 using the ‘sip-ua sipserv’ command. In a Centralized Trunk deployment the call is sent over PSTN trunk on the G650 PRI interface in the headquarters.

2. Analog to Analog Branch phones
   To establish calls between the two analog phones, you must configure a POTS dial peer on each SR2330. The POTS dial peer specifies the parameters and dialed digits for the analog phone connection.

3. Headquarter to Branch and Branch to Branch SIP phones
   In the sample configuration Branch numbers are dialed with a 20 pre-fix for routing over PSTN trunk from Avaya G650 gateway in Headquarter to PRI interface on SR2330 in the Branch in survivable mode. In Normal mode, for numbers other than the matched VoIP dial peers, the SR2330 routes the call to central SIP server assuming that it has the necessary information for routing. Whether they are intra-site calls, inter-site calls reachable through extension dialing. Calls are routed over central PSTN facilities on Avaya Aura® Communication Manager

2.3. Network Modes
PSTN call routing is further determined within each of the trunking configurations based on the network status of each branch, either WAN connectivity up or down, as detailed below:

**Normal Mode:** 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, registration and routing decisions.

**Survivable Mode:** A Branch has lost communication with the Headquarters/Datacenter location, and the local branch SR2330 SIP call control is being used for all calls at that branch. Note: if Avaya Aura® Session Manager loses connectivity to the WAN; phones in all branches will go into survivable mode simultaneously.

2.4. Test Description and Coverage
This section presents the primary call flows for the Survivable SIP Gateway Solution in a Centralized 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.
The interoperability testing focused on dynamic switching from the Normal Mode (where the network connectivity between the main site and the branch site is intact) to the Survivable Mode (where the network connectivity between the main site and the branch site is broken) and vice versa. The following categories of tests were executed:

1. Centralized SIP call flows as defined in the current section.
2. TCP transport tested with SR2330
3. Survivable Mode Testing: Basic CM features, voice mail for calls between Branch and Headquarters / PSTN
4. Survivable Mode Testing: Calls between Branch and PSTN in Survivable Mode

Overview:
1. SIP Call Control: All SIP call control and call routing is provided by the centralized Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

2. Branch PSTN Outbound Non-Local: PSTN outbound calls from the branch to non-local numbers are routed to a centralized Avaya G650 Media Gateway controlled by Avaya Aura® Communication Manager. Note: In the sample configuration the same CM was used to terminate these PSTN calls and simulate the PSTN cloud. As well as to register stations in the Headquarter location.

3. Branch PSTN Inbound: Calls from the PSTN to a branch Listed Directory Number (LDN) enter the network at the local branch SR2330 PRI interface, then route to Avaya Aura® Session Manager/Avaya Aura® Communication Manager for call treatment.

4. Headquarters PSTN Inbound: Calls to Headquarters endpoints enter the network at the Headquarters Avaya G650 Media Gateway controlled by Avaya Aura® Communication Manager.

Call Flows - Centralized Trunking – Normal Mode:
1. 96xx/96x1/FXS at branch to/from 96xx/96x1 stations at HQ
   96xx/96x1/FXS ↔ SM ↔ CM ↔ HQ 96xx/96x1 station

2. 96xx/96x1/FXS at branch to/from PSTN Trunk – local calls.
   96xx/96x1/FXS ↔ SM ↔ CM ↔ PRI ↔ PSTN Trunk

3. 96xx/96x1/FXS at branch to/from PSTN Trunk – long distance toll calls
   96xx/96x1/FXS ↔ SM ↔ CM ↔ Avaya G650 Media Gateway ↔ PSTN Trunk

4. 96xx/96x1/FXS at branch to/from 96xx/96x1/FXS at same branch
   96xx/96x1/FXS ↔ SM ↔ CM ↔ SM ↔ 96xx/96x1/FXS

5. 96xx/96x1/FXS at branch/HQ to/from Centralized Application Servers
   96xx/96x1/FXS ↔ SM ↔ Aura® Messaging, Conferencing
Test Scenarios - Centralized Trunking – Normal Mode:
The test cases focused on Centralized Trunking endpoint to endpoint call flows and feature
invocation for branch connectivity in Normal Mode.

4. CM Features tested include:
   ● Hold/Resume, Conference Add/Drop, Call Transfer – Attended/Un-attended, Call
     Waiting and Voice Mail Dialing.

5. Verification of following Avaya Aura® Messaging scenario was done:
   ● Voicemail recording/playback (Using various means to retrieve/leave message)
   ● MWI status on recording/playback of voicemail

6. Conferencing features tested with Branch and Headquarter phones include:
   ● Users joining conference with Talk path
   ● Conference Mute/Un-mute, Hold/Un-hold, Lecture mode, Entry/Exit Tones,
     conference lock/Un-lock, Auto increase duration/ports, user dial-out/blast dial (FXS
     phones at branch and SIP/H.323 phones at HQ), Code duration, Transfers and
     conference to local branch phones from branch phones on the bridge.
   ● Lecture mode, Mute/Un-mute and Moderator Hang-up using DTMF

Calls Flows - Centralized Trunking – Survivable Mode:
1. PSTN Trunk to 96xx/96x1/FXS at branch
   PSTN Trunk → PRI - SR2330 Branch Gateway → 96xx/96x1/FXS

2. 96xx/96x1/FXS at branch to PSTN Trunk
   96xx/96x1/FXS → SR2330 Branch Gateway - PRI → PSTN Trunk

3. 96xx/96x1/FXS at branch to/from 96xx/96x1/FXS at same branch
   96xx/96x1/FXS ↔ SR2330 Branch Gateway ↔ 96xx/96x1/FXS

4. 96xx/96x1/FXS at branch to/from Centralized Applications at Headquarters
   96xx/96x1/FXS ↔ SR2330-PSTN Trunk ↔ Avaya G650 HQ Gateway ↔ Centralized
   Apps

5. 96xx/96x1/FXS at branch to/from 96xx/96x1 at Headquarters
   96xx/96x1/FXS ↔ SR2330-PSTN Trunk ↔ Avaya G650 HQ Gateway ↔ 96xx/96x1

Test Scenarios – Centralized Trunking – Survivable Mode:
1. Basic Telephony features Testing included:
   Hold/Resume, Conference Add/Drop, Call Transfer – Attended/Un-attended, Call Waiting.

2. Call Flows for failover in active/In-progress calls:
   ● Active call status between Branch phones after failover
   ● Active call status between HQ phones after failover
   ● New call status, intra-branch, HQ to HQ, Branch to HQ and HQ to Branch after
     failover and failback in active calls.
   ● In progress call status for calls between Branch phones, HQ to branch phones after
     failover
3. Verification of following Avaya Aura® Messaging scenarios was done:
   • Voicemail recording/playback
   • MWI status on recording/playback of voicemail in survivable mode
   • MWI status on recording/playback of voicemail after recovery to normal mode
   • Call Sender feature in failover mode.
   • MWI status for logoff login after failover and fallback

4. Conferencing features tested in survivable mode include the following. Branch phone interactions with HQ phones using PSTN connectivity:
   • Users joining conference with Talk path and Talk path for long conference duration in survivable mode
   • Conference Mute/Un-mute, Hold/Un-hold, Lecture mode, Entry/Exit Tones, conference lock/Un-lock, Auto increase duration/ports, user dial-out/blast dial (FXS phones at branch and SIP/H.323 phones at HQ), Code duration.
   • Status of active conference during failover to survivable mode and fallback
   • Transfer and conference to local branch phones in survivable mode from branch user who has joined conferencing bridge
   • Lecture mode, Mute/Un-mute and Moderator hang-up scenarios using DTMF from Branch phones in survivable mode

5. Negative Scenarios:
   • Reboot of servers like CM, SM, SR2330 and end points during active call in Normal Mode
   • Reboot of SR2330 and phones during active call in Survivable Mode
   • Verification of Avaya call features, active calls etc during and after WAN outage
   • Verification of Avaya call features, active calls etc during and after SM outage in Normal Mode
   • MWI status after logoff and login in normal mode when the MWI is lighted.

2.5. Test Results and Observations

Supported Features:
   • In survivable mode SR2330 supports the following features
     o 3-way conferences, call transfer, hold/resume.
   • HQ to branch call in survivable mode does not work, unless call routing is configured via PSTN
   • Phone Registration in Simultaneous mode recommended, Will also support Alternate
     o Max 300 SIP user registrations supported
     o Max 64 TDM channels supported
     o Max concurrent call count supported is 100

Defects/Findings:
   • ADVD is not supported to interoperate with SR2330 10.3.3
   • SR2330 R10.3.3 will have to be configured with small memory allocation when it is configured as both an Edge Router and SIP survivable gateway, default configuration is Large memory.
   • Hold/Un-hold operations are not supported over PSTN trunk on SR2330 R10.3.3
Limitations:
- TLS support not available in SR2330
- SR 2330 does not support Transcoding, therefore calls between phones with codec mismatch or transport mismatch (RTP Vs SRTP) will not work in survivable mode
- MWI does not glow in survivable mode
- FNE access does not work in survivable mode
- One-X Communicator works only in TLS mode. It was therefore registered only at HQ in this scope.
- Presence Services support only TLS protocol as opposed to SR2330, which supports TCP. Presence was therefore not tested with SR2330 in this scope.

3. Reference Configuration

The lab test environment used for the Survivable Intelligent Edge solution testing is shown in Figure 1. This test bed included the following components:

- Branch
  - Avaya SR2330 Advanced Gateway with analog FXS stations and PRI/PSTN trunks
  - Avaya 96xx and 96x1 SIP phones, Avaya 6210 analog phones
- Headquarters/Datacenter
  - Avaya S8800 Server running Communication Manager with Avaya G650 Media Gateway
  - Avaya Aura® Session Manager with companion Avaya Aura® System Manager
  - Avaya Aura® Messaging
  - Avaya Aura® Conferencing
  - HTTP Phone Configuration Server
- WAN
  - Avaya Layer 2 and Extreme Layer 2/3 switches in the core
- LAN
  - Avaya Cajun router/switch
- PSTN
  - Simulated PSTN using E1/T1 interface on Avaya G650 Media Gateway

Notes:
- In the test configuration phone Protocols include:
  - Headquarters: SIP, H.323 and analog telephones were used
  - Branch site: SIP phones, branch analog sets connected to the FXS ports on the Avaya SR2330 gateway, which registers them with SM as SIP users.

  - The following network diagram represents the configuration used for the Centralized Trunking arrangement where in Normal Mode all PSTN calls from the branch, both local and long-distance toll calls, were routed through the central T1/E1 facilities on the Avaya G650 Media Gateway located at the Headquarters location.
Figure 1: Test Configuration
3.1. Detailed Call Flow: Outbound to PSTN – Normal Mode

Many of Avaya Aura® Session Manager and Avaya Aura® Communication Manager configuration steps presented in Section 5, Section 6 and Section 7 are to support the location based routing requirements of the outbound calls to PSTN – centralized Mode call flows. The details of this call flow, specific to the sample configuration, are included here to illustrate the linkage of the various configuration steps. As mentioned earlier, the term “Communication Manager” refers to Avaya Aura® Communication Manager.

![Figure 2: Centralized Trunking - Call Flow](image)

**Outbound Call Flow over PSTN in Normal Mode:**

**Figure 2** details the steps involved in this call flow

Call Scenario: Headquarters SIP phone (seen as call leg 1 above) Branch SIP Phone or Branch Analog phone (seen as call leg 0 above) registered to Avaya Aura® Session Manager in Normal mode dials an outgoing call to PSTN Phone in HQ over HQ PSTN trunk.

**STEP 1:** Avaya Analog or 96x1/96xx SIP Phone (Branch) or a 96xx/96x1 SIP phone (HQ) sends SIP INVITE to Avaya Aura® Session Manager with dialed digit string.

**STEP 2:** Session Manager forwards the SIP INVITE to Avaya Aura® Communication Manager over SIP trunk configured in **Section 6.5**.

**STEP 3:** Avaya Aura® Communication Manager forwards the call with dialed digits string over the PSTN trunk.

**STEP 4:** PSTN Phone answers the call.
NOTE: Call is terminated to PSTN using routing functionality on CM to PSTN trunk in centralized deployment.

3.2. Network Topology

The network implemented for the sample configuration shown in Figure 1 was modeled after an enterprise consisting of a main Headquarters/Datacenter location and a branch location interconnected over a corporate WAN. The Headquarters location hosts Avaya Aura® Session Manager and a Avaya Aura® Communication Manager running on an Avaya S8800 server, providing enterprise wide SIP call control and advanced feature capabilities. The Headquarters network is mapped to IP Network Region 1 within Avaya Aura® Communication Manager. The Centralized Trunking capabilities of the solution utilize the location based call routing features of Avaya Aura® Session Manager and IP codec set selection features of Avaya Aura® Communication Manager.

The Headquarters location also hosts the following centralized components: an Avaya Aura® Messaging voice mail platform, Avaya Aura® Conferencing 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. These Application Notes assume that Avaya Aura® Messaging and Avaya Aura® Conferencing have been installed and configured with SIP trunks to Avaya Aura® Session Manager. The information for configuring Avaya Aura® Messaging can be found in Reference [13] and information for installing and configuring Avaya Aura® Conferencing can be found in Reference [14] in Section 11.

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 8 includes the parameters of the 46xxsettings.txt file used by the Avaya 96xx and 96x1 SIP Phones for survivability. The Avaya Aura® 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 pilot number via PSTN. The internal or private extension is configured in the 46xxsettings.txt file as the default voice mail access number or pilot number, or by dialing pilot number via PSTN. 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 96xx and 96x1 SIP Phones 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 96xx/96x1 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 Aura® Messaging, may enable other methods of notification that a message has been delivered.

The branch location consists of several Avaya 96xx and 96x1 SIP Phones and an SR2330 Secure Router with two analog phones 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 (Example: 172.16.4.10 in the sample configuration) for SIP signaling. Its SIP Media Gateway Module listens for SIP requests on port 5070. Requests can come from either Avaya Aura®
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 Avaya one-X® Deskphone SIP 96xx/96x1 Series IP Telephones, and proxies those requests to the Media Gateway module as necessary (e.g., for calls to the FXS interface). In normal mode, the IP telephones signal through the SR2330 directly to Avaya Aura® Session Manager. In normal mode the SR2330 acts more like a data router and simply forwards SIP messages to/from SIP endpoints and Avaya Aura® Session Manager.

![Network Topology Diagram]

**Figure 3: Network Topology**
4. Equipment and Software Validated

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

<table>
<thead>
<tr>
<th>Equipment/Software</th>
<th>Release/Version</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya Secure Router 2330</td>
<td>10.3.3</td>
</tr>
<tr>
<td>S8800 Duplex Servers with G650 Media Gateway</td>
<td>Avaya Aura® Communication Manager R 6.2.3 SP3</td>
</tr>
<tr>
<td>S8800 Server</td>
<td>Avaya Aura® Session Manager 6.2 FP1</td>
</tr>
<tr>
<td>S8800 Server</td>
<td>Avaya Aura® System Manager 6.2 FP1</td>
</tr>
<tr>
<td>Avaya One-X Communicator</td>
<td>R6.1.6</td>
</tr>
<tr>
<td>Avaya 96xx- H.323</td>
<td>R3.1.4</td>
</tr>
<tr>
<td>Avaya 96xx- SIP</td>
<td>R2.6.7</td>
</tr>
<tr>
<td>Avaya 96x1 – H.323</td>
<td>R6.2.1</td>
</tr>
<tr>
<td>Avaya 96x1 – SIP</td>
<td>R6.2</td>
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<td>Avaya 11xx SIP</td>
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</tr>
<tr>
<td>Avaya 12xx SIP</td>
<td>R4.03.09.00</td>
</tr>
<tr>
<td>Avaya 6219 Analog Endpoints</td>
<td>-</td>
</tr>
<tr>
<td>Avaya 2410,14xx Digital Endpoints</td>
<td>-</td>
</tr>
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<td>ADVD: Avaya Desktop Video Device</td>
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<tr>
<td>S8800 Server</td>
<td>Avaya Aura® Conferencing R6.0.1</td>
</tr>
<tr>
<td>Dell R610</td>
<td>Avaya Aura® Messaging R6.2</td>
</tr>
</tbody>
</table>

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager that implement the Survivable SIP Gateway Solution. The following areas are covered:

- Specify SIP Domains
- Add Locations
- Add SR2300 as a SIP Entity
- Define Entity Links for SR2330 (which defines the SIP trunk parameters used by Avaya Aura® Session Manager when routing calls to/from the SR2330 Secure Router in Centralized configuration).
- Define Time Ranges
- Define Routing Policies for Avaya Aura® Communication Manager
- Define Dial Pattern for Avaya Aura® Communication Manager
- User configuration for branch SIP and analog telephones

It is assumed that configuration of SIP trunks between Avaya Aura® Session Manager and Avaya Aura® Communication Manager has already been completed.
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of Avaya Aura® System Manager. Enter the appropriate credentials and click on Log On button.

The main menu is displayed. Click the Routing link on the home page of System Manager (not shown). The sub-menus displayed in the left column below will be used to configure Avaya Aura® Session Manager.

5.1. Specify SIP Domain
Configure the SIP Domain that will be used for SIP communication across Avaya Aura® setup in headquarters and branch setup in normal and survivable mode.

Click New and in the General section, under Name add a descriptive name. Under Notes add a brief description. Click Commit to save. Screen below shows all the list of SIP domains added.

Highlighted is the SIP domain used in the sample configuration.
5.2. Add Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. You may define a separate Location for bandwidth management to the Branch location.

Select Locations and click New and add the following in the General section:

- Name: Add a descriptive name
- Notes: Add a brief description, if desired.

Click Commit to save.
5.3. Add SR2330 as a SIP Entity

Configure a SIP Entity for Avaya SR2330. In the **Routing** menu and select **SIP Entities**, Click **New**. In the **General** section, add the following:

- **Name**: Add an identifier for SR2330 Gateway
- **FQDN or IP Address**: Enter the IP Address of SR2330
- **Type**: Select the appropriate menu option for the gateway
- **Notes**: Add a brief description
- **Location**: From the drop down select the Location added in **Section 5.2** for Branch phones
- **Time Zone**: From the drop down select the appropriate time zone
5.4. Define Entity Links for SR2330
To configure Entity Link for the SR2330, in the Routing menu select Entity Links. Click New. In the Entity Links section, add the following:

- **Name**
  Enter an identifier for the entity link.

- **SIP Entity 1**
  From dropdown, select IBSM-A the SIP Entity for Avaya Aura® Session Manager

- **Protocol**
  From dropdown select the required protocol. For the sample configuration TCP was selected.

- **Port**
  Enter the correct port for Avaya Aura® Session Manager. For the sample configuration port 5060 was selected.

- **SIP Entity 2**
  From dropdown, select the SIP Entity added in Section 5.3 for SR2330

- **Port**
  Enter the correct port for SR2330.

- **Trusted**
  Ensure the ticked box is clicked

- **Notes**
  Add a brief description

Click Commit to save.
5.5. Define Time Ranges

Configure the Time Ranges. In the Routing menu select Time Ranges. Click New. Under Name enter an identifier. Select the days of the week, enter time values for Start Time and End Time. Under Notes add a brief description. When completed, click Commit to save. Screenshot below shows the updated information.

5.6. Define Routing Policies for Communication Manager

This section describes how to define the routing policy to route calls over the E1/T1 interface on G650 gateway in the Headquarters to SR2330 situated in the Branch over PSTN trunk. This is used in Survivable mode in the Centralized Trunking configuration, the sample configuration used for these Application Notes supports the same.

To configure the routing policies for Avaya Aura® Communication Manager, select Routing Policies in the Routing menu and click New. In the General section, under Name add an identifier to define the routing policy for Avaya Aura® Communication Manager. Under Notes add a brief description. In the SIP Entity as Destination section, click Select.
The SIP Entities list page opens. Select the entry of Avaya Aura® Communication Manager added in Avaya Aura® System Manager and click Select.

![Image of Avaya Aura® System Manager 6.2 interface showing the SIP Entity as Destination with Name: 2409, Disabled: off, and Retries: 0.](image)

The CLAN entry seen below is added in this sample configuration.

![Image of Avaya Aura® System Manager 6.2 interface showing the SIP Entity List with three entries: ICDP-10.0.5045, ICDC-2209-5045, and ICDC-14A99.](image)

The selected SIP Entity is displayed on the Routing Policy Details page. In the Time of Day section, click on Add, and from the Time Ranges List page (not shown) select the desired Time Range and click on Select.
\textbf{5.7. Define Dial Pattern for Communication Manager}

In a Centralized trunk configuration, a route is configured for routing calls via Avaya Aura® Communication Manager PRI interface for routing over PSTN trunk in Failover mode. In Normal mode all phones are registered to Avaya Aura® Session Manager in the Headquarters.

To configure a route to Communication Manager expand \textit{Routing} and select \textit{Dial Patterns}. Click New. In the \textit{General} section add the following:

- **Pattern:** Add the pattern that the customer station will dial-out to reach an extension on SR2330.
- **Min:** Enter the minimum number digits that must be dialed.
- **Max:** Enter the maximum number digits that may be dialed.
- **SIP Domain:** In the dropdown, select the SIP Domain added in Section 5.1. In this sample configuration the default domain –\texttt{ALL}– is used for the dial plan.
- **Notes:** Add a brief description.

Click on \textit{Commit} to save.
In the sample configuration, the digits 20 are used as the pre-fix to dial PSTN line in order to reach users at branch location. This is used in Failover mode to route calls to Branch location over the PRI interface. The calls were routed to SR2330 via Avaya Aura® Communication Manager over PSTN trunk using the AAR route defined in Section 6.5 on Avaya Aura® Communication Manager.

Under Notes add a brief description.

In the Originating Locations and Routing Policies section click Add (as highlighted in the above screenshot). In the Originating Location section, select the appropriate location. In the sample configuration the routing policy was applied for all originating locations by selecting Apply The Selected Routing Policies to All Originating Locations checkbox.
In the **Routing Policies** section, select the appropriate routing policy for routing (defined in **Section 5.6**). Click the **Select** button seen in the screenshot below.
The **Dial Pattern Details** page (seen in the screenshot below) opens on clicking **Select**. Click on **Commit** to save on the **Dial Pattern Details** page seen below to save the new dial pattern for routing calls to SR2330.
In the sample configuration, 5 digit extensions with the pattern 6xxx were created and logged in to Avaya Aura® Session Manager in Normal Mode and Avaya Aura® Session Manager routes the requests for these extensions.

5.8. SIP Users

This section describes the administration of SIP telephones in Avaya Aura® Session Manager, and applies to the 96xx and 96x1 series SIP telephones as well as the analog telephones connected to the FXS ports of the SR2330, these register with Avaya Aura® Session Manager in Normal Mode. It is assumed that the SIP trunk between Avaya Aura® Communication Manager and Avaya Aura® Session Manager has already been provisioned. References [1] to [8] in Section 11 contain information on configuring SIP trunks between Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The following screens show a sample configuration for an Avaya 9630 SIP phone whose extension is 60003. The same procedure should be followed for all branch IP and analog telephones. On the main configuration page, select User Management on Home Page. Click Manage Users and click New to administer a new telephone user.
This will create a new User Profile. In the Identity section, enter a Last Name, First Name, Login Name and Authentication Type. The following screen shows what was entered for extension 60003. Note that fields marked with * are required to be filled in.

In the Communication Profile section, enter a Communication Profile Password. Note that the Communication Profile Password is the one the telephone is required to use when registering to Avaya Aura® Session Manager. The value Primary is populated by default and it is possible to create multiple profiles for the user. The profile that is made default is the active profile. There can be only one active profile at a time. Enter a communication address for the user, which comprises of its Fully Qualified Address.
In the *Communication Profile* section, there are two sub-sections that need to be filled in for this configuration: *Session Manager*, and *CM Endpoint Profile*. Clicking on the arrow next to Communication Profile reveals the other sections.

Expand the *Session Manager Profile* section. Click on the box next to *Session Manager*, and select the appropriate *Primary Session Manager* Instance from the list. Select the appropriate
Origination and Termination Application Sequence. Next select a Home Location for the user as defined in Section 5.2.

Select the appropriate Avaya Aura® Communication Manager in System box, Enter the extension number in the Extension box, it is also possible to click the arrow on the drop down and select from the available extensions on the selected Avaya Aura® Communication manager system. Select the appropriate CM template in the Template box, enter the password to be used to register the station to Avaya Aura® Session Manager and select IP port from the available list. Enter the voicemail number configured on Avaya Aura® Messaging in the Voice Mail Number field, refer to References[9] & [10] in Section 11 for voicemail configuration.
Click **Endpoint Editor** (as seen above) to complete mailbox configuration for the user. In the general Options page that opens, add the Coverage path in the field Coverage Path 1. This is the coverage path configured for Avaya Aura® Messaging.

![Endpoint Editor Screenshot](image)

On the next screen click **Feature Options** tab. Enter **sip-adjunct** in the MWI Served User Type field and click **Done** to return to the previous screen.

![Feature Options Screenshot](image)
Click **Commit** when done.

![Configuration Panel](image)

### 6. Configure Avaya Aura® Communication Manager

This section shows the necessary steps to configure Avaya Aura® Communication Manager to support the Survivable SIP Gateway Solution in a Centralized Trunking scenario. It is assumed that the basic configuration on Avaya Aura® Communication Manager and the required licensing have already been administered, as well as the SIP trunk to Avaya Aura® Session Manager. See **References** [1] to [8] in **Section 11** for additional information. All commands discussed in this section are executed on Avaya Aura® Communication Manager using the System Access Terminal (SAT).
6.1. 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.

```
change ip-codec-set 1

Codec Set: 1

Audio Silence Frames Packet
Codec Suppression Per Pkt Size(ms)
1: G.711MU   n   2   20
2: G.729A   n   2   20
3:        -   -   -
4:        -   -   -
5:        -   -   -
6:        -   -   -
7:        -   -   -

Media Encryption
1: none
2: ____________________________
3: ____________________________
```

6.2. IP Network Region

IP Network Regions are defined for each branch location as well as the Headquarters location 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.

- **Name:** Enter a descriptive name for the IP Network Region.
- **Codec Set:** Enter the codec set to be used for entities in this IP network region.
- **Authoritative Domain:** This is the SIP domain name defined in Avaya Aura® System Manager (see Section 5.1) and used throughout the enterprise for SIP communications.
The values used in the sample configuration for Branch IP Network Region 7 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 (if any).
The following screen illustrates a portion of Page 4 for network region 7. The connectivity between network regions is specified under the Inter Network Region Connection Management heading, beginning on Page 4. For example, codec set 1 is specified for connections between network region 7 and network region 1. If bandwidth usage is a concern, a different codec set could be defined that uses a compressed codec (e.g., G.729) between the Headquarters and Branch locations, and would be specified here.

### 6.3. Configure PSTN Trunk to SR2330

Configure trunk to SR2330 from CM of type ISDN. This trunk is used to route calls from branch phones on SR2330 to Avaya Aura® Communication Manager and vice versa in survivable mode.

### 6.4. Route Pattern to Avaya Aura® Session Manager

Use the change route-pattern command to modify the route pattern for calls routed to Avaya Aura® Session Manager. The changes made to Route Pattern 80 in the sample configuration are highlighted below. Route Pattern 80 uses SIP Trunk Group 80. In the case of the sample configuration, this causes all digits to be sent to Avaya Aura® Session Manager. SIP Trunk Group configuration is not covered in the scope of these Application Notes.
6.5. Add AAR for Routing for PSTN routing

In survivable mode the Headquarter users will dial a 20 prefix to reach the branch phones using PSTN trunk. The screenshot shows the change data screen as the data is already added in the sample configuration.

6.6. Configure Route Pattern to SR2330

The route pattern is configured to route calls to SR2330 (Over PSTN trunk) after deleting the preceding digits (20 in the sample configuration), the digits for the SIP/Analog branch extensions are then routed to SR2330 using PSTN trunk configured in Section 6.3,(200 in this case).

6.7. Configure Incoming Call Treatment

Configure incoming call treatment on the PSTN trunk between SR2330 and Avaya Aura® Communication Manager. This ensures that calls routed from the Branch phones over PSTN in Centralized Configuration in Normal Mode will be routed correctly by Avaya Aura® Communication Manager after pre-ceding 4 digits are deleted. The pre-ceding 9120 digits are deleted in the incoming call treatment for the trunk being used, 200 in the sample configuration.
7. Configure Avaya 2330 Secure Router

Presented below is an annotated version of the Avaya SR2330 Secure Router configuration used in Branch for Centralized Trunking. In the Centralized Trunking configuration the call is received over the PSTN trunk using the routing defined in Section 6.5. 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 ports, as well as SIP to PSTN gateway capabilities. It registers on behalf of its configured FXS ports to Avaya Aura® Session Manager in normal mode. The SIP Survivability module is implemented as a B2BUA and not the SIP Media Gateway.

The SIP Media Gateway does the standard SIP to TDM protocol conversion. The SIP telephones register through the secure router to Avaya Aura® Session Manager as their primary SIP registrar, and simultaneously to the SSM as the secondary registrar. See Section 3.2 and Figure 3 for more details on the SIP signaling configuration implemented below.

This technical configuration guide assumes that the reader is familiar with the operation and configuration of the S2330 and that the appropriate software version is loaded and operational. The following are the configuration commands used to configure SR2330 as a SIP survivable gateway. You configure the 2330 by entering commands in “configuration mode” via the CLI. To enter this mode type, “configure terminal” from the operational mode prompt (#).

The commands for basic configuration required are as below:

- Configure Ethernet interfaces: Configure one Ethernet interface (0/1 in this case) for communication with the headquarter network. Configure a second Ethernet interface (0/2 in this case) to communicate with the branch network.

```plaintext
interface ethernet 0/1
  ip address 172.16.4.10 255.255.255.0
  qos
    module
      exit module
    chassis
      exit chassis
      exit qos
      exit ethernet
interface ethernet 0/2
  switchport switchport pvid 945 qos
  module
    exit module
```
- Configure the isdn pri interface, 1/1 interface is used in the sample configuration.

```plaintext
interface bundle isdn_pr1
  exit bundle
interface bundle isdnpr1
  link pri_t1 1/1 voice
  isdn
    switch-type primary-5ess
  activate
  exit isdn
exit bundle
```

- For call routing, the Media Gateway points to the primary server as the active server, assuming the primary server is up and reachable. Otherwise, the Media Gateway points to the secondary server as the active server.
- The SIP server IP address is specified on the SR2330 using the sip-ua sipserver command for both primary and secondary sip servers. This configuration is used for call routing in both Centralized and Distributed configurations on SR2330.

```plaintext
sip-ua
  sip-domain dns:silpunelab.com
  sip-server ipv4:172.16.1.14:5060
  sip-server ipv4:192.168.1.1:5060 secondary
  transport tcp
  registrar ipv4:172.16.1.14 expires 3600
  keepalive target sip-server
  keepalive timer 40
exit sip-ua
```

- To establish calls between the two analog phones, you must configure a POTS dial peer on each SR2330. The POTS dial peer specifies the parameters and dialed digits for the analog phone connection

```plaintext
voice-port 1/1:23
  no shutdown
  exit voice-port
voice-port 2/1
  signal loop-start
  station name fxs1
  station number 60000
  no shutdown
  timeouts interdigit 3
exit voice-port
```
The dial peer destination-pattern, (9120.% in this sample configuration) is used to ensure that all outgoing calls (with specified prefix 9120) from the branch are routed over the PRI interface, 1/1 in this case.

dial-peer voice pots 11
  destination-pattern 9120.%
  port 1/1
  no digit-strip
  forward-digits all
  no shutdown
  exit pots

7.1. Call Routing Behavior on SR2330

Call Routing Behavior on SR2330 comprises of the following steps:

The SR2330 performs one of the following actions on receiving a call:

- Matching VoIP dial peer
  If the SR2330 can match the called number to a VoIP dial peer, it routes the call to the SIP server address or target network address specified by the VoIP dial peer. It applies the appropriate POTS dial peer connection attributes for the incoming call leg and the matched VoIP dial peer attributes for the outbound leg.

- No matching VoIP dial peer
  If the SR2330 finds no VoIP dial-peer match, it forwards the call to the central SIP server (specified using the sip-ua sip-server command) for further routing. For numbers other than the matched VoIP dial peers, the SR2330 routes the call to central SIP server assuming that it has the necessary information for routing.

Please find below the system.cfg used in the sample configuration for reference.

```
# SR2330 system configuration file (.CFG).
#
# AVAYA assumes no responsibility for product reliability, performance, or both if the user modifies the .CFG file. Full responsibility for any modification made to the .CFG file, by the user, is assumed by the user.
#
# Version: 10.3.3.50
# File Created: 03/02/2012-13:21:29

system logging
  console
    priority crit
    exit console
  syslog
    host_ipaddr 172.16.14.10
    module system sys9 debug
    module alarms local0 none
    module dos local0 none
```
module forwarding local0 none
module voip-ssm-cdr local0 none
module voip-cdr local0 none
module voip-gwy local0 none
enable
exit syslog
exit logging
hostname SR2330Branch01
log utc
module t1 1/1
clock_source network
exit t1
module t1 1/2
exit t1
vlan database   vlan 945   exit database
vlan classification exit classification
bridge mstp exit mstp exit bridge
lacp exit lacp
interface ethernet 0/1
  ip address 172.16.4.10 255.255.255.0
qos
  module
  exit module
chassis
  exit chassis
exit qos
exit ethernet
interface ethernet 0/2
  switchport   switchport pvid 945   qos
  module
  exit module
chassis
  exit chassis
exit qos
exit ethernet
interface bundle isdnpr1
  link pri_t1 1/1 voice
isdn
  switch-type primary-5ess
  activate
  exit isdn
exit bundle
interface vlan vlan945
  ip address 192.168.1.1 255.255.255.0
exit vlan
gvrp  exit gvrp
snmp-server
  engine-id
    local 0000000c000000007f000001
    exit engine-id
chassis-id SR2330Branch01
  exit snmp-server
oam
cfm
  enable
ethtype 88e6
exit cfm
exit oam
ftp_server
icmp_timestamp
telnet_server
ssh_server
    enable
    exit ssh_server
ip host_add silpunelab.com 192.168.1.1
ip load-balancing per-flow
ip route 0.0.0.0/0 172.16.4.1 ipv6 unicast-routing
ipv6 load-balancing per-flow
mpls tunnel-mode uniform
firewall global
    algs
dns
    exit dns
    exit algs
    max-connection-limit self 2048
    exit firewall
firewall internet
    exit firewall
firewall corp
    policy 1024 out permit
    exit policy
    exit firewall
voice class
    exit class
voice service voip
    sip
        bind all ipv4:192.168.1.1:5070
        rel1xx disable
        exit sip
    fax rate-management transferredTCF
codec 1 g711alaw 160
codec 2 g711ulaw 160
codec 3 g729r8 20
codec 4 g723r53 20
codec 5 g726r16 40
codec 6 g726r32 80
dtmf-relay rtp-nte
ssm
        bind ip ipv4:192.168.1.1
        enable
        registrar
            expires max 3600
            expires default 3600
            expires min 3600
            exit registrar
        sip-server
            domain dns:silpunelab.com
            exit sip-server
        protocol-header
            retry-after-interval 0
            server-header Avaya
            exit protocol-header
default-gateway ipv4:192.168.1.1:5070 transport tcp
exit ssm
exit voip
voice call
exit call
voice dsp
exit dsp
sip-ua
  sip-domain dns:silpunelab.com
  sip-server ipv4:172.16.1.14:5060
  sip-server ipv4:192.168.1.1:5060 secondary
  transport tcp
  registrar ipv4:172.16.1.14 expires 3600
  keepalive target sip-server
  keepalive timer 40
  exit sip-ua
voice-port 1/1:23
  no shutdown
  exit voice-port
voice-port 2/1
  signal loop-start
  station name fxs1
  station number 60000
  no shutdown
  timeouts interdigit 3
  exit voice-port
voice-port 2/2
  signal loop-start
  station name fxs2
  station number 60001
  no shutdown
  timeouts interdigit 3
  exit voice-port
dial-peer voice-pots 2
  destination-pattern 60000
  port 2/1
  forward-digits all
  no shutdown
  authentication 60000 123456
  register e164
  exit pots
dial-peer voice-pots 60001
  destination-pattern 60001
  port 2/2
  forward-digits all
  no shutdown
  authentication 60001 123456
  register e164
  exit pots
dial-peer voice-pots 11
  destination-pattern 9120.%
  port 1/1
  no digit-strip
  forward-digits all
  no shutdown
  exit pots
dst
  no enable
exit dst
### 8. Configure Avaya 96xx/96x1 Series IP Telephone (SIP)

Add a user in Avaya Aura® Messaging using References [9], [10] and [13] in Section 11. The following 46xxsettings.txt file parameters/values were used for the testing:

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Value Used in Sample Configuration</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP_CONTROLLER_LIST</td>
<td>172.16.1.14:5060;transport=tcp, 192.168.1.1:5060;transport=tcp</td>
<td>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. 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.</td>
</tr>
<tr>
<td>FAILBACK_POLICY</td>
<td>Auto</td>
<td>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.</td>
</tr>
<tr>
<td>FAST_RESPONSE_TIME_TIMEOUT</td>
<td>2</td>
<td>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.</td>
</tr>
<tr>
<td>MSGNUM</td>
<td>44888</td>
<td>The number dialed when the Message button is pressed while phone is in Normal Mode.</td>
</tr>
<tr>
<td>PSTN_VM_NUM</td>
<td>912044888</td>
<td>The number dialed when the Message button is pressed while phone is in Survivable Mode.</td>
</tr>
<tr>
<td>RECOVERYREGISTERWAIT</td>
<td>60</td>
<td>A Reactive Monitoring Interval. If no response to a &quot;maintenance check&quot; 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.</td>
</tr>
<tr>
<td>Parameter Name</td>
<td>Value Used in Sample Configuration</td>
<td>Description</td>
</tr>
<tr>
<td>--------------------------------</td>
<td>-------------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>PHNEMERGNUM</td>
<td>911</td>
<td>Number dialed when emergency soft key is pressed.</td>
</tr>
<tr>
<td>DISCOVER_AVAYA_ENVIRONMENT</td>
<td>1</td>
<td>Automatically determines if the active SIP Server is an Avaya server or not.</td>
</tr>
<tr>
<td>SIPREGPROXYPOLICY</td>
<td>simultaneous</td>
<td>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 (The sample configuration included phones in both registration modes)</td>
</tr>
<tr>
<td>SIPDOMAIN</td>
<td>silpunelab.com</td>
<td>The enterprise SIP domain. Must be the same for all SIP controllers in the configuration. SIPDOMAIN is set to “silpunelab.com” in the sample configuration.</td>
</tr>
</tbody>
</table>

### 9. Verification Steps

The steps in this section can be used to verify that the configuration steps have been done correctly. Verify the trunk status for connectivity between Avaya Aura® Session Manager and Avaya Secure Router.

#### 9.1. Off-PBX Telephone Mapping

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

```plaintext
<table>
<thead>
<tr>
<th>Station Extension</th>
<th>Application Dial Prefix</th>
<th>CC</th>
<th>Phone Number</th>
<th>Trunk Selection</th>
<th>Config Set</th>
<th>Dual Mode</th>
</tr>
</thead>
<tbody>
<tr>
<td>60003</td>
<td>OPS</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
The following list command output summarizes the configuration relevant to the sample configuration. Each Avaya SIP Telephone at the branch (e.g., 60003 and 60004), each analog device connected to an FXS port on the SR2330 (e.g., 60000 and 60001) can be observed. The corresponding registration of these users to Avaya Aura® Session Manager is shown in Section 9.3.

<table>
<thead>
<tr>
<th>Station Extension</th>
<th>Appl</th>
<th>CC</th>
<th>Phone Number</th>
<th>Config Trunk Set</th>
<th>Mapping Mode</th>
<th>Calls Allowed</th>
</tr>
</thead>
<tbody>
<tr>
<td>60003</td>
<td>OPS</td>
<td></td>
<td>60003</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60004</td>
<td>OPS</td>
<td></td>
<td>60004</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60005</td>
<td>OPS</td>
<td></td>
<td>60005</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60006</td>
<td>OPS</td>
<td></td>
<td>60006</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60007</td>
<td>OPS</td>
<td></td>
<td>60007</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60008</td>
<td>OPS</td>
<td></td>
<td>60008</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60009</td>
<td>OPS</td>
<td></td>
<td>60009</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
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<td>OPS</td>
<td></td>
<td>60010</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60011</td>
<td>OPS</td>
<td></td>
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<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
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<td>EC500</td>
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<td>9890074903</td>
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<td>both</td>
<td>all</td>
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<tr>
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<td></td>
<td>60012</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60013</td>
<td>OPS</td>
<td></td>
<td>60013</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60014</td>
<td>OPS</td>
<td></td>
<td>60014</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
<tr>
<td>60015</td>
<td>OPS</td>
<td></td>
<td>60015</td>
<td>1 / aar</td>
<td>both</td>
<td>all</td>
</tr>
</tbody>
</table>
### 9.2. Session Manager/Secure Router 2330 Link Status

In the browser-based GUI of Avaya Aura® System Manager, click **Session Manager** in the Home page. Expand **System Status** and click on **SIP Entity Monitoring**. The SIP entities monitored by Avaya Aura® Session Manager are listed on the lower portion of the page as shown below. Select the entity corresponding to the SR2330, in this case “SR2330”.

<table>
<thead>
<tr>
<th>Entity Name</th>
<th>Status</th>
<th>Select</th>
<th>SIP Entity Monitoring</th>
<th>Usage</th>
<th>Status</th>
<th>Registration</th>
<th>Summary</th>
<th>User Registrations</th>
<th>System Tools</th>
<th>Performance</th>
</tr>
</thead>
<tbody>
<tr>
<td>IBSM-Z</td>
<td>6/40</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Select: All, None</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### All Monitored SIP Entities

- **manish-Cln-Temp**
- **Mit_SIPp Tool**
- **MPP-POM**
- **multi-loc Manish**
- **MultiLocationSite-FQDN-1**
- **MultiLocationSite-FQDN-2**
- **MultiLocationSite-FQDN-3**
- **mx-bridge**
- **MX62-Bridge**
- **New-CM-MST-5.2**
- **PresenceServer**
- **SBC2**
- **solution-cm601**
- **SR2330**

Select the **SR2330** entity.
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 Avaya Aura® 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”. An indication of “Down” would imply that the SR 2330 is most likely in survivable mode, since the SR2330 uses the same OPTIONS method to determine if Session Manager is accessible, and the interval is the same as that configured for Avaya Aura® Session Manager (30 sec).

9.3. 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 is shown the command and the output, which indicates that the two FXS interfaces configured with extensions 60000 and 60001 have successfully registered with the available SIP server. The command is ‘show sip-ua register status’ in the ‘configure terminal’ menu.

```
SR2330Branch01# show sip-ua register status

<table>
<thead>
<tr>
<th>Line</th>
<th>peer</th>
<th>expires(sec)</th>
<th>registered</th>
</tr>
</thead>
<tbody>
<tr>
<td>3</td>
<td>60000</td>
<td>3600</td>
<td>yes</td>
</tr>
<tr>
<td>4</td>
<td>60001</td>
<td>3600</td>
<td>yes</td>
</tr>
</tbody>
</table>
```
The following command shows the Avaya 96xx/96x1 SIP phones registered in Backup Mode, the SSM module is always in backup mode, as explained in Section 7. The command is ‘show registered users all’ in the ‘configure terminal’ menu. This command also shows the expiry, mode of registration and contact header details as seen below.

```
SR2330Branch01# conf term
SR2330Branch01/configure# show ssm registered-users all

+-----------------+-----------------+-----------------+-----------------------+--------------------------+
| Subscriber      | Destination     | Expires (sec)  | Mode                  |
|-----------------|-----------------|----------------|-----------------------|--------------------------|
| 60006           | silpunealab.com | 2714           | backup (tcp)          | 1                       |
| 92.168.1.61:5060|                 |                |                       |                          |
| 60014           | silpunealab.com | 1742           | backup (tcp)          | 1                       |
| 92.168.1.54:5060|                 |                |                       |                          |
```

9.4. Session Manager Registered Users (Normal Mode)

To verify registration of the SR2330 supported analog stations (connected to the FXS interfaces) and the Avaya 96xx and 96x1 series IP phones, in the browser-based GUI of System Manager, click Session Manager in the Home page. Expand System Status and click on User Registrations. The following abridged screens show the last four users listed that correspond to those users, 60004 Headquarter user and 60009 & 60010 Branch users.

9.5. Timing Expectations for Fail-over to Survivable Mode

This section is intended to set approximate expectations for the length of time before Avaya 96xx/96x1 SIP Telephones in the branch will acquire service from the Secure Router 2330 when a failure occurs such that the branch is unable to communicate with the central Avaya Aura® Session Manager. 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 96xx/96x1 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 96xx/96x1 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 Avaya Aura® 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 Avaya Aura® Session Manager, idle Avaya 96xx/96x1 SIP phones in the branch will typically be registered with Avaya Aura® 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 Avaya Aura® 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. These Application Notes present the configuration steps to implement the Survivable SIP Gateway Solution based on the Avaya Secure Router to minimize service disruptions to these remote branch SIP endpoints. Hold/Unhold over PSTN trunk on SR2330 10.3.3 is not supported. SR2330 R10.3.3 will have to be configured with small memory allocation when it is configured as both an Edge Router and SIP survivable gateway, default configuration is Large memory.
11. Additional References

Avaya references, available at http://support.avaya.com

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

2) Installing and Configuring Avaya Aura® Session Manager, Doc ID xxx
3) Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325
4) Administering Avaya Aura™ Session Manager, Doc ID 03-603324
6) Installing and Upgrading Avaya Aura® System Manager http://support.avaya.com/css/P8/documents/100089250
10) Implementing Avaya Aura Messaging. R6.1, October 2011
15) Please add following SR reference docs.
   Configuration — SIP Media Gateway Avaya Secure Router 2330/4134 - Document number NN47263-508, 03.02 https://downloads.avaya.com/css/P8/documents/100120422