



Configuring Avaya Communication Manager and Avaya SIP Enablement Services for SIP Trunks with Asterisk Business Edition PBX – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network comprised of Avaya Communication Manager and Avaya SIP Enablement Services at the Main site and Asterisk Business Edition PBX at the Remote site. SIP trunks are used to connect Avaya Communication Manager and Asterisk Business Edition PBX via Avaya SIP Enablement Services. All calls between the Main and Remote sites are carried over these SIP trunks.

For the sample configuration, Avaya Communication Manager is running on an Avaya S8500 Server with an Avaya G650 Media Gateway. Testing was conducted at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes present a sample configuration for a network comprised of Avaya Communication Manager and Avaya SIP Enablement Services (SES) at the Main site and Asterisk Business Edition PBX at the Remote site. At the Main site, there is an Avaya 6408D+ Digital Telephone that is connected directly to Avaya Communication Manager. Also at the Main site, there are Avaya SIP and H.323 telephones that register locally to Avaya SES and Avaya Communication Manager, respectively. The Avaya SIP telephone is the Avaya 9620 IP Telephone running the Avaya one-X™ Deskphone SIP software. The H.323 telephones include an Avaya 9630 IP Telephone, running the Avaya one-X Deskphone Edition software, and an Avaya 4621SW IP Telephone. At the Remote site, the Cisco 7960 IP Phone registers locally to the Asterisk Business Edition PBX. SIP trunks are used to connect Avaya Communication Manager and Asterisk Business Edition PBX via Avaya SES. All calls between the Main and Remote sites are carried over these SIP trunks.

NOTE: “Asterisk Business Edition PBX” is also referenced as “Asterisk” in these Application Notes.

For the sample configuration show in **Figure 1**, Avaya Communication Manager is running on an Avaya S8500 Server with an Avaya G650 Media Gateway. An Avaya Modular Messaging system, consisting of the Messaging Application Server and the Messaging Store Server, is available at the Main site to provide voice mail services and to verify delivery of DTMF. With the exception of the Avaya Digital Telephone, the components at each site are physically connected to separate Avaya C364T-PWR Converged Stackable Switches. A single PC, located at the Main site, provides HTTP and TFTP server support for both sites and is also used for sniffing the network. The HTTP server is needed for delivery of the firmware and configuration files for the Avaya IP telephones and the TFTP server is needed for delivery of firmware and configuration files for the Cisco IP telephone.

A five-digit Uniform Dial Plan (UDP) is used to facilitate dialing between the Main and Remote sites. Unique extension ranges are associated with Avaya Communication Manager at the Main site (2xxxx) and Asterisk at the Remote site (60xxx).

The configuration of the endpoint telephones and of Avaya Modular Messaging is not the focus of these Application Notes and will not be described. For administration of endpoint telephones and Avaya Modular Messaging, refer to the appropriate documentation listed in **Section 8**. These Application Notes will focus on the configuration of the SIP trunks.

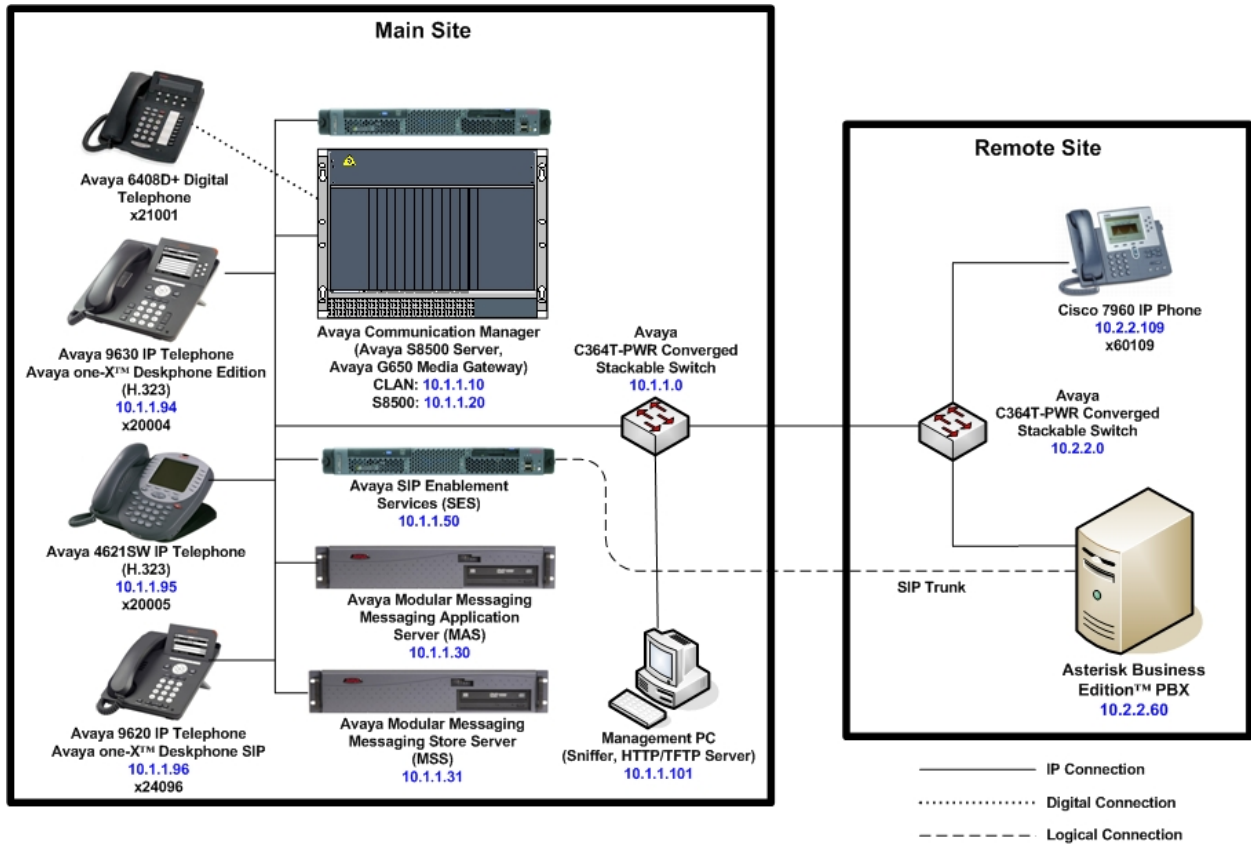


Figure 1: Network Configuration Diagram

2. Equipment and Software Validated

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

Equipment	Software
Avaya S8500 Server	Avaya Communication Manager 4.0.1 Load 731.2
Avaya G650 Media Gateway TN799DP C-LAN Circuit Pack TN2602AP IP Media Processor	HW01 FW012 HW11 FW024
Avaya SIP Enablement Services	4.0, Load 33.6
Avaya Modular Messaging	Messaging Application Server – 3.1 Messaging Store Server – 3.1
Avaya 9620 IP Telephone	Avaya one-X Deskphone SIP Version 1.0.13.1 (5) (SIP)
Avaya 9630 IP Telephone	Avaya one-X Deskphone Edition Version S1.5 (H.323)
Avaya 4621SW IP Telephone	2.8 (H.323)
Avaya 6408D+ Digital Telephone	N/A
Avaya C364T-PWR Converged Stackable Switch	4.5.14
Asterisk Business Edition PBX	B.1-3 running on Red Hat Enterprise Linux 4ES
Cisco 7960 Series IP Phone	POS3-08-7-00 (SIP)
Management PC	Windows 2000 Professional with Service Pack 2 Microsoft Internet Information Services 5.1

Table 1: Configuration Equipment and Version

3. Configure Avaya Communication Manager

The standard configuration for connecting Avaya Communication Manager and Avaya SES is covered in References [1-3]. For these Application Notes, the configuration of Avaya Communication Manager is limited to those steps that are pertinent to the configuration of the SIP trunk between Avaya SES and Asterisk and the configuration of call routing between the Main site and the Remote site.

The Avaya Communication Manager configuration presented in this section for this test configuration allows calls between Avaya Communication Manager endpoints to use the G.711 μ -law codec and calls between Avaya and Asterisk endpoints to use the G.729 codec. Because calls to Avaya SIP and Asterisk endpoints both require SIP trunks, separate SIP trunk groups along with separate signaling groups, network regions, and codec sets were created to allow for the use of different codecs. The actual codecs may vary.

In the test configuration, all Avaya endpoints (including the Avaya SIP endpoints) were configured to be in network region “1”. Codec set “1” with G.711 μ -law was used for connections within network region “1”. Incoming SIP trunk calls from Avaya SIP endpoints will use network region “1” based on the IP network map configuration. Calls to Avaya SIP endpoints will use network region “1” based on the Avaya SIP endpoints station mapping to a specific trunk group and signaling group.

In the test configuration, network region “6” was configured for Asterisk endpoints. Codec set “6” with G.729 was used for connections between network regions “1” and “6”. Incoming SIP trunk calls from Asterisk endpoints will use network region “6” based on the signaling group configuration. Calls to Asterisk endpoints will use network region “6” based on the Automatic Alternate Routing (AAR) configuration, which selects a specific trunk group and signaling group.

In the test configuration, the signaling group used for Avaya SIP endpoints and the signaling group used for Asterisk endpoints have the same value for the near-end node name, far-end node-name, and far-end domain (see **Section 3.4**). These signaling groups have different values for the far-end network region. To allow the G.711 μ -law codec to be used among Avaya endpoints while using the G.729 codec between Avaya and Asterisk endpoints, the signaling group whose far-end network region refers to the Asterisk region must be a lower number than other signaling groups using the same near-end and far-end node names. When two SIP signaling groups use the same near-end and far-end node names, the lower numbered signaling group will be selected first by Avaya Communication Manager for incoming calls. If it is impractical to ensure that these conditions are met, the same signaling group and trunk group can be used to reach both Avaya SIP and Asterisk endpoints, with the signaling group mapping to the same codec set used for Avaya endpoints. While not presented in these Application Notes, this simpler configuration where only one codec set is used for all endpoints was also verified.

The table below displays the configured network region, codec set, signaling group, trunk group associations and their purpose:

Purpose	Region	Codec	Signaling	Trunk
For Avaya SIP Endpoints	1	1	21	21
For Asterisk Endpoints	6	6	6	6

This section focuses on configuring the SIP trunks on Avaya Communication Manager to Avaya SES, which are used to reach the Avaya SIP endpoints and Asterisk endpoints. In addition, this section highlights selected features that are required for the interoperability and this section provides a sample routing using AAR. The configuration procedures include the following areas:

- Administer trunk-to-trunk transfer
- Display IP node names
- Administer IP codec sets and IP network regions
- Administer SIP trunk groups and signaling groups
- Administer route patterns
- Administer location and public unknown numbering
- Administer uniform dial plan and AAR analysis
- Administer IP network map and station mapping

The following configuration of Avaya Communication Manager was performed using the System Access Terminal (SAT). After completion of the configuration in this section, use the “save translation” command to make the changes permanent.

3.1. Administer Trunk-to-Trunk Transfer

Use the “change system-parameters features” command to allow trunk-to-trunk transfers. Submit the change.

This feature is required to transfer an incoming call from the Remote site back out to the Remote site (incoming trunk to outgoing trunk), and to transfer an outgoing call to the Remote site to another outgoing call to the Remote site (outgoing trunk to outgoing trunk). For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk and must be used with caution. For alternatives, the trunk-to-trunk transfer feature can be implemented based on the Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 8** for more details.

```
change system-parameters features                               Page 1 of 17
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? y
      Trunk-to-Trunk Transfer: all
Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y
      Music/Tone on Hold: music Type: port 01A0501
      Music (or Silence) on Transferred Trunk Calls? no
      DID/Tie/ISDN/SIP Intercept Treatment: attd
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n

      Abbreviated Dial Programming by Assigned Lists? n
      Auto Abbreviated/Delayed Transition Interval (rings): 2
      Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n
```

3.2. Display IP Node Names

Use the “display node-names ip” command to view the entries for the C-LAN and for Avaya SES. These names will be used for the creation of the signaling groups (see **Section 3.4**). In this case, “C-LAN1” is the name for the C-LAN and “SES_home1” is the name for the Avaya SES. The actual node names and IP addresses may vary.

```
display node-names ip
      IP NODE NAMES
      Name          IP Address
C-LAN1          10.1.1.10
SES_home1      10.1.1.50
default          0.0.0.0
medpro-1a03     10.1.1.11
procr           10.1.1.20
```

3.3. Administer IP Codec Sets and IP Network Regions

Administer two IP codec sets. One will be used for calls within the Main site (Avaya-Avaya) and the other will be used for calls between the Main and Remote sites (Avaya-Asterisk). Use the “change ip-codec-set n” command, where “n” is an existing codec set number to be used for the interoperability. Enter the desired audio codec type in the **Audio Codec** field. Retain the default values for the remaining fields and submit these changes.

In the codec sets displayed below, codec set “1” was used for Avaya-Avaya calls and codec set “6” was used for Avaya-Asterisk calls. The actual codec set number and codec type may vary. The same codec set number could have been used in the test configuration.

```
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:
```

For brevity, the fields at the bottom of the screen were removed.

```
change ip-codec-set 6                                     Page 1 of 2
                                                         IP Codec Set
Codec Set: 6
Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size(ms)
1: G.729AB      n           2          20
2: G.729        n           2          20
```

For brevity, the fields at the bottom of the screen were removed.

3.3.1. IP Network Region for Avaya Endpoints

Administer an IP network region to use for Avaya endpoints. Use the “change ip-network-region n” command, where “n” is an existing network region number to be used for the interoperability.

In the test configuration, network region “1” was used for Avaya endpoints. For the **Location** field, enter the location corresponding to the Main site from **Section 3.6**. For the **Authoritative Domain** field, enter the SIP domain name of Avaya SES from **Section 4.1**. Enter a descriptive name in the **Name** field. For the **Codec Set** field, enter the corresponding audio codec set number from the **IP Codec Set** screens for calls within the Main site. Enable the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio** fields. These settings will enable direct media shuffling for Avaya-Avaya calls. Retain the default values for the remaining fields and submit these changes.

```
change ip-network-region 1                                     Page 1 of 19
                                                           IP NETWORK REGION
Region: 1
Location: 1          Authoritative Domain: avaya.com
Name: Region 1
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
```

For brevity, the fields at the bottom of the screen were removed.

Navigate to page 3 and verify that the appropriate codec set is administered for calls between the Avaya network region “1” and the Asterisk network region “6”.

```
change ip-network-region 1                                     Page 3 of 19
                                                           Inter Network Region Connection Management
src dst codec direct WAN-BW-limits Video Dyn
rgn rgn set WAN Units Total Norm Prio Shr Intervening-regions CAC IGAR
1 1 1
1 2 2 y NoLimit n
1 3
1 4
1 5
1 6 6 y NoLimit n
1 7
```

For brevity, the fields at the bottom of the screen were removed.

3.3.2. IP Network Region for Asterisk Endpoints

Administer an IP network region to use for Asterisk endpoints. Use the “change ip-network-region n” command, where “n” is an existing network region number to be used for the interoperability.

In the test configuration, network region “6” was used for Asterisk endpoints. For the **Authoritative Domain** field, enter the SIP domain name of Avaya SES from **Section 4.1**. Enter a descriptive name in the **Name** field. For the **Codec Set** field, enter the corresponding audio codec set number from the **IP Codec Set** screens for calls with Asterisk. Enable the **Intra-region IP-IP Direct Audio**, **Inter-region IP-IP Direct Audio**, and **IP Audio Hairpinning** fields. These settings will enable direct media shuffling for Avaya-Asterisk calls. Retain the default values for the remaining fields, and submit these changes.

```
change ip-network-region 6                                     Page 1 of 19
                                                           IP NETWORK REGION
Region: 6
Location: 1          Authoritative Domain: avaya.com
Name: Asterisk
MEDIA PARAMETERS          Intra-region IP-IP Direct Audio: yes
Codec Set: 6             Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048      IP Audio Hairpinning? y
UDP Port Max: 3029
```

For brevity, the fields at the bottom of the screen were removed.

Navigate to page 3, and specify the appropriate codec set to use for calls between the Asterisk network region “6” and the Avaya network region “1”.

```
change ip-network-region 6                                     Page 3 of 19
                                                           Inter Network Region Connection Management
src dst codec direct WAN-BW-limits Video Dyn
rgn rgn set WAN Units Total Norm Prio Shr Intervening-regions CAC IGAR
6 1 6 y NoLimit
6 2
6 3
6 4
6 5
6 6 6
```

For brevity, the fields at the bottom of the screen were removed.

3.4. Administer SIP Trunk Groups and Signaling Groups

Administer two sets of SIP trunk groups and signaling groups. One set will be used to reach the Avaya SIP endpoints and the other set will be used to reach the Asterisk endpoints.

3.4.1. SIP Trunk and Signaling Groups for Avaya SIP Endpoints

In the test configuration, trunk group “21” and signaling group “21” were used to reach the Avaya SIP endpoints. Use the “add signaling-group n” command, where “n” is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** C-LAN node name from **Section 3.2**
- **Far-end Node Name:** Avaya SES node name from **Section 3.2**
- **Near-end Listen Port:** “5061”
- **Far-end Listen Port:** “5061”
- **Far-end Network Region:** Avaya network region number from **Section 3.3.1**
- **Far-end Domain:** SIP domain name of Avaya SES from **Section 4.1**
- **DTMF over IP:** “rtp-payload”; corresponds to using RFC 2833 for DTMF

```
add signaling-group 21                                     Page 1 of 1
                                                           SIGNALING GROUP
Group Number: 21                                         Group Type: sip
                                                           Transport Method: tls
Near-end Node Name: C-LAN1                               Far-end Node Name: SES_home1
Near-end Listen Port: 5061                               Far-end Listen Port: 5061
                                                           Far-end Network Region: 1
Far-end Domain: avaya.com
                                                           Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                               Direct IP-IP Audio Connections? y
                                                           IP Audio Hairpinning? n
Enable Layer 3 Test? n
Session Establishment Timer(min): 120
```

Use the “add trunk-group n” command, where “n” is an available trunk group number. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name
- **TAC:** An available trunk access code that is consistent with the dial plan
- **Service Type:** “tie”
- **Signaling Group:** Corresponding signaling group number from above.
- **Number of Members:** Desired number of trunk group members.

```
change trunk-group 21                                     Page 1 of 21
                                                         TRUNK GROUP
Group Number: 21                                         Group Type: sip           CDR Reports: y
  Group Name: SES Home 1                                COR: 1                   TN: 1           TAC: 101
  Direction: two-way                                    Outgoing Display? n
  Dial Access? n                                        Night Service:
  Queue Length: 0
  Service Type: tie                                     Auth Code? n
                                                         Signaling Group: 21
                                                         Number of Members: 40
```

3.4.2. SIP Trunk and Signaling Groups for Asterisk Endpoints

In the test configuration, trunk group “6” and signaling group “6” were used to reach the Asterisk endpoints. Use the “add signaling-group n” command, where “n” is an available signaling group number. Enter the following values for the specified fields and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** C-LAN node name from **Section 3.2**
- **Far-end Node Name:** Avaya SES node name from **Section 3.2**
- **Near-end Listen Port:** “5061”
- **Far-end Listen Port:** “5061”
- **Far-end Network Region:** Asterisk network region number from **Section 3.3.2**
- **Far-end Domain:** SIP domain name of Avaya SES from **Section 4.1**
- **DTMF over IP:** “rtp-payload”; corresponds to using RFC 2833 for DTMF

```
add signaling-group 6                                     Page 1 of 1
                SIGNALING GROUP
Group Number: 6                Group Type: sip
                Transport Method: tls
Near-end Node Name: C-LAN1      Far-end Node Name: SES_home1
Near-end Listen Port: 5061      Far-end Listen Port: 5061
                Far-end Network Region: 6
                Far-end Domain: avaya.com
                Bypass If IP Threshold Exceeded? n
                DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
                IP Audio Hairpinning? n
Enable Layer 3 Test? n
Session Establishment Timer(min): 3
```

Use the “add trunk-group n” command, where “n” is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name
- **TAC:** An available trunk access code that is consistent with the dial plan
- **Service Type:** “tie”
- **Signaling Group:** Corresponding signaling group number from above
- **Number of Members:** Desired number of trunk group members

```

add trunk-group 6                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 6          Group Type: sip          CDR Reports: y
  Group Name: To-Asterisk      COR: 1          TN: 1          TAC: 106
  Direction: two-way          Outgoing Display? n
  Dial Access? n              Night Service:
  Queue Length: 0
  Service Type: tie          Auth Code? n

                                     Signaling Group: 6
                                     Number of Members: 20
  
```

3.5. Administer Route Patterns

Administer two route patterns to correspond to the two newly added SIP trunk groups. Use the “change route-pattern n” command, where “n” is an available route pattern. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name
- **Grp No:** Trunk group number from **Section 3.4**
- **FRL:** Enter a level that allows access to this trunk, with “0” being least restrictive.

```

change route-pattern 21                             Page 1 of 3
                                     Pattern Number: 1   Pattern Name: Avaya
                                     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: 21    0
2:
3:
4:
5:
6:
                                     n user
                                     n user
                                     n user
                                     n user
                                     n user
                                     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
  
```

```

change route-pattern 6                                     Page 1 of 3
                Pattern Number: 6   Pattern Name: Asterisk
                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: 6      0
2:
3:
4:
5:
6:
                n   user
                n   user
                n   user
                n   user
                n   user
                n   user

  BCC VALUE  TSC CA-TSC   ITC BCIE Service/Feature PARM No. Numbering LAR
  0 1 2 M 4 W      Request          Dgts Format
                Subaddress
1: y y y y y n n          rest          none

```

3.6. Administer Location and Public Unknown Numbering

Use the “change locations” command to assign the route pattern for Avaya SIP endpoints to a location corresponding to the Main site. Add an entry for the Main site if does not exist already. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Name:** A descriptive name to denote the Main site
- **Timezone Offset:** An appropriate time zone offset for the Main site
- **Rule:** An appropriate daylight savings rule for the Main site
- **Proxy Sel Rte Pat:** The Avaya route pattern number from **Section 3.5**

```

change locations                                     Page 1 of 1
                LOCATIONS
                ARS Prefix 1 Required For 10-Digit NANP Calls? y
  Loc Name          Timezone Rule  NPA          Proxy Sel
  No              Offset
1: Main          + 00:00  0          21

```

Use the “change public-unknown-numbering 0” command to define the calling party number to be sent to Asterisk. Add a new entry for the trunk group defined in **Section 3.4.2** to reach Asterisk endpoints. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 6 will be sent as a 5-digit calling number. The calling party number will be sent to the far-end in the SIP “From” header. Submit these changes.

```
change public-unknown-numbering 0                                     Page 1 of 2
                                NUMBERING - PUBLIC/UNKNOWN FORMAT
```

Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	
5	6	6		5	Total Administered: 1 Maximum Entries: 9999

3.7. Administer Uniform Dial Plan and AAR Analysis

This section provides a sample AAR routing used for routing calls with dialed digits 6xxxx to Asterisk. Note that other methods of routing, such as Auto Route Selection (ARS), may be used (see Reference [4] for more information). Use the “change uniform-dialplan 0” command and add an entry to specify use of AAR for routing of digits 6xxxx. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Matching Pattern:** Dialed prefix digits to match on, in this case “6”.
- **Len:** Length of the full dialed number, in this case “5”.
- **Del:** Number of digits to delete, in this case “0”.
- **Net:** “aar”

```
change uniform-dialplan 0                                         Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                                                Percent Full: 0
```

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
6	5	0		aar	n	

Use the “change aar analysis 0” command and add an entry to specify how to route the calls to 6xxxx. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case “6”.
- **Total Min:** Minimum number of digits, in this case “5”.
- **Total Max:** Maximum number of digits, in this case “0”.
- **Route Pattern:** The Asterisk route pattern number from **Section 3.5**.
- **Call Type:** “aar”

```
change aar analysis 0
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE

Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
6	5	5	6	aar		n

3.8. Administer IP Network Map and Station Mapping

Use the “change ip-network-map” command to map the IP addresses of the Avaya SIP endpoints to the network region for Avaya endpoints from **Section 3.3.1**. This will enable the Avaya network region and codec set to be used for the Avaya SIP endpoints for incoming calls instead of the Asterisk network region associated with the lowered numbered SIP signaling group. A range of IP addresses may be configured when there is more than one Avaya SIP endpoint. The screen below shows one IP address corresponding to one of the Avaya SIP endpoints. Submit these changes.

```
change ip-network-map
```

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IP ADDRESS MAPPING

From IP Address	(To IP Address	Subnet or Mask)	Region	VLAN	Emergency Location Extension
10 .1 .1 .96	10 .1 .1 .96		1	n	

To associate an Avaya Communication Manager station with a SIP user on Avaya SES, use the “change off-pbx-telephone station-mapping n” command, where “n” is the extension number of an Avaya SIP endpoint. In the **Application** field, enter “OPS”. In the **Phone Number** field, enter the same number shown for **Station Extension**. In the **Trunk Selection** field, enter the trunk group defined in **Section 3.4.1** to reach Avaya SIP endpoints. In the **Config Set** field, enter the configuration set number that defines the desired call treatment options. This will enable the Avaya network region to be used for this Avaya SIP endpoint for outgoing calls. Submit these changes. Repeat this procedure for every Avaya SIP endpoint.

```
change off-pbx-telephone station-mapping 24096
```

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STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set
24096	OPS	-		24096	21	1

4. Configure Avaya SIP Enablement Services

This section provides the procedures for configuring SIP trunking between Avaya SES and Asterisk. The steps to install and configure Avaya SES to work with Avaya Communication Manager are covered in References [1-3]. These Application Notes assume the Avaya SES has already been configured with the proper domain, host, and media server information. The procedures to create the SIP trunk include the following:

- Obtain SIP domain
- Administer host address map
- Administer host contact
- Administer trusted host

These procedures assume that the user has already launched the Avaya SES **Administration Web Interface** (not shown). This interface is accessible by using the URL “http://<ip-address>/admin” in an Internet browser window, where “<ip-address>” is the IP address of Avaya SES. In this scenario, the URL is “http://10.1.1.50/admin”.

4.1. Obtain SIP Domain

From the **SIP Server Management** web page, select **Server Configuration** → **System Properties** from the left pane to display the **View System Properties** screen. Use the value in the **SIP Domain** field (in this case “avaya.com”) for configuring the **Authoritative Domain** and **Far-end Domain** fields in **Section 3.4**.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top navigation bar includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server IP address "Server: 10.1.1.50". A left-hand navigation pane lists various system components, with "Server Configuration" and "System Properties" highlighted. The main content area, titled "View System Properties", displays the following configuration details:

SES_Version	SES-4.0.0.0-033.6
System Configuration	simplex
Host Type	home/edge
SIP Domain*	<input type="text" value="avaya.com"/>

Note that the DNS domain is: avaya.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

License Host*	<input type="text" value="10.1.1.50"/>
Management System Access Login	<input type="text"/>
Management System Access Password	<input type="text"/>

DiffServ/TOS Parameters

Call Control PHR Value*	<input type="text" value="145"/>
-------------------------	----------------------------------

4.2. Administer Host Address Map

From the **SIP Server Management** web page, select **Hosts** → **List** from the left pane to display the **List Hosts** screen. Click **Map**.



AVAYA Integrated Management SIP Server Management Server: 10.1.1.50

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
- Emergency Contacts
- Hosts**
 - List**
 - Migrate Home/Edge

List Hosts

Status	Commands	Host	Type
up to date	Edit Map Go-To Test-Link Delete	10.1.1.50	home/edge

Force All
Migrate Home/Edge

In the **List Host Address Map** screen below, click **Add Map In New Group** in the right pane.



AVAYA Integrated Management SIP Server Management Server: 10.1.1.50

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
- Emergency Contacts
- Hosts**
 - List**
 - Migrate Home/Edge

List Host Address Map

Host 10.1.1.50

No address map entries.

Add Map In New Group

The **Add Host Address Map** screen is displayed. This screen is used to specify which calls are to be routed to Asterisk. For the **Name** field, enter a descriptive name to denote the routing. For the **Pattern** field, enter an appropriate syntax to match the Asterisk endpoints extensions. For the interoperability testing, a pattern of “`^sip:6[0-9]{4}`” is used to match to any extensions in the range of 60000-69999 at the Remote site. Retain the check in **Replace URI** and click **Add**.

4.3. Administer Host Contact

After a confirmation screen that the host address map has been added, the **List Host Address Map** screen is displayed. The screen is updated with the new address map. Click **Add Another Contact**.

Commands	Name	Commands	Contact
Edit Delete	ToAsterisk-6xxxx		

In the **Add Host Contact** screen, enter the following in the **Contact** field:

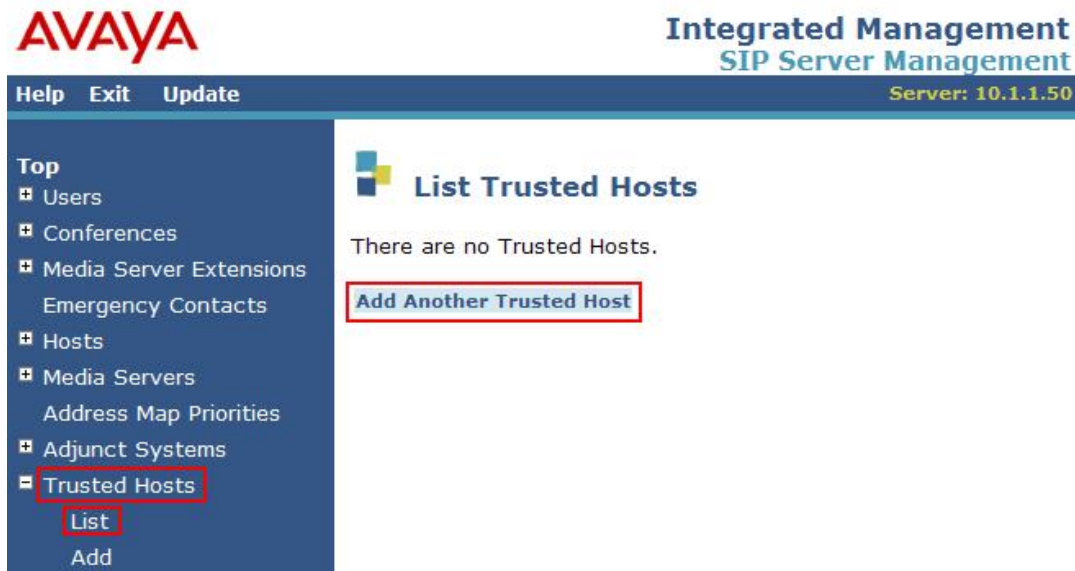
“sip:\$(user)<destination-IP-address>:5060;transport=udp”

In this case, the “<destination-IP-address>” is the IP address of the Asterisk. As Asterisk only supports UDP, “udp” was used for the transport protocol and “5060” for the port. Avaya SES will substitute “\$(user)” with the user portion of the request URI before sending the message. Click **Add**.



4.4. Administer Trusted Host

From the **SIP Server Management** web page, select **Trusted Hosts** → **List** from the left pane to display the **List Trusted Hosts** screen. Click **Add Another Trusted Host**.



The **Add Trusted Host** screen is displayed. Administer the Asterisk as a trusted host so that the SIP messages from the Asterisk will not be challenged by Avaya SES. For the **IP Address** field, enter the IP address of Asterisk. For the **Comment** field, enter a descriptive name to identify the trusted host entry. Click **Add**.

After a confirmation screen, the **List Trusted Hosts** screen is displayed once again. Click **Update** in the bottom left pane for all changes in this section to take effect.

Commands	IP Address	Trusted by Host	Comment
Edit Delete	10.2.2.60	10.1.1.50	Asterisk BE PBX

5. Configure Asterisk Business Edition PBX

This section focuses on configuring Asterisk to support a SIP trunk between Avaya SES and Asterisk. In addition, this section highlights selected features that are required for the interoperability and this section provides a sample routing scheme using the Asterisk dial plan.

The installation of the Asterisk Business Edition PBX is covered in References [10-11]. For additional information and examples on configuring Asterisk, see Reference [12]. For the configuration described in these Application Notes, the Asterisk software was installed on the Red Hat Linux Enterprise 4 operating system. These Application Notes do not cover the installation of the software, the configuration of the Asterisk endpoints, nor the detailed configuration of the voice mail system. These Application Notes assume that the following have been configured on Asterisk:

- Support for the SIP service
- SIP domain
- SIP endpoints
- Support for voice mail

The Asterisk is configured by editing configuration files on the Linux system. The configuration files are stored in the `/etc/asterisk` directory. These files can be edited with any available text editor on the system. Appropriate permissions are required to edit these files.

The configuration files are organized into sections called “contexts”. A context is created by placing the context name in brackets (e.g., [`general`]) followed by configuration parameters. Configuration parameters are grouped together under each context. A context is defined in the following format:

```
[<context1>]
<parameter1>=<value>
<parameter2>=<value>
```

The configuration procedures covered in this section include the following areas:

- Administer settings for internal calls
- Administer settings for calls to Avaya endpoints
- Administer settings for calls from Avaya endpoints
- Administer dial plan for internal calls and calls from Avaya endpoints
- Administer dial plan for calls and inbound calls
- Administer dial plan to deliver Caller ID to the voice mail system
- Restart Asterisk

5.1. Administer Settings for Internal Calls

For calls between Asterisk endpoints, the following features were configured.

- Shuffling is enabled. This is known as *re-invite* within the Asterisk.
- G.711 μ -law and G.729 codecs are used and in the preference order listed.
 - **NOTE:** A license was not installed for the G.729 codec in this configuration. When unlicensed, Asterisk supports the G.729 codec in a “pass-through” mode. This allows SIP endpoints that support the same G.729 codec variant to talk each other.
- RFC 2833 is used for DTMF transmission.

For this test configuration, the following parameters were configured in the **general** context in the `sip.conf` configuration file. The **general** context is the default context used by Asterisk for internal calls.

- **canreinvite:** “yes” Enable shuffling.
- **disallow:** “all” Turn off support for all codecs.
- **allow:** “ulaw”, “g729” Turn on support for G.711 μ -law and G.729 codecs.
- **dtmfmode:** “rfc2833” Use RFC2833 for sending DTMF.

```
[general]

canreinvite=yes           ; Enable direct IP-IP media (shuffling)

disallow=all             ; First disallow all codecs
allow=ulaw               ; Allow G.711 ulaw as 1st codec for internal SIP calls
allow=g729               ; Allow G.729 as 2nd codec for internal SIP calls

dtmfmode=rfc2833        ; Set dtmfmode to use RFC 2833 for sending DTMF.
```

5.2. Administer Settings for Calls to Avaya Endpoints

For calls from Asterisk endpoints to Avaya endpoints, the following features were configured.

- Shuffling is enabled.
- G.729 and G.711 μ -law codecs are used and in the preference order listed.
 - **NOTE:** A license was not installed for the G.729 codec in this configuration. When unlicensed, Asterisk supports the G.729 codec in a “pass-through” mode. This allows SIP endpoints that support the same G.729 codec variant to talk each other.
- RFC 2833 is used for DTMF transmission. This is the same as configured for Avaya Communication Manager in **Section 3.4**.

For this test configuration, the following parameters were configured in the **avaya-out** context in the `sip.conf` configuration file. The **avaya-out** context is the context used in this test configuration for calls to Avaya endpoints.

- **type:** “peer” Default for calls outbound to a SIP server.
- **fromdomain:** “avaya.com” SIP domain of Avaya SES from **Section 4.1**.
- **host:** “10.1.1.50” IP address of Avaya SES.
- **disallow:** “all” First, turn off support for all codecs.
- **allow:** “g729”, “ulaw” Turn on support for G.729 and G.711 μ -law codecs.
- **dtmfmode:** “rfc2833” Use RFC2833 for sending DTMF.
- **canreinvite:** “yes” Enable shuffling.

```
[avaya-out]
type=peer ; Default value for calls outbound to a SIP server
fromdomain=avaya.com ; Domain of Avaya SES
host=10.1.1.50 ; IP address of Avaya SES
disallow=all ; First disallow all codecs
allow=g729 ; Allow G.729 as 1st codec for calls to Avaya
allow=ulaw ; Allow G.711 ulaw as 2nd codec for calls to Avaya
dtmfmode=rfc2833 ; Set dtmfmode to use RFC 2833 for sending DTMF.
canreinvite=yes ; Enable direct IP-IP media (shuffling)
```

5.3. Administer Settings for Calls from Avaya Endpoints

For calls from Avaya endpoints to Asterisk endpoints, the following features were configured.

- Shuffling is enabled.
- G.729 and G.711 μ -law codecs are used and in the preference order listed.
 - **NOTE:** A license was not installed for the G.729 codec in this configuration. When unlicensed, Asterisk supports the G.729 codec in a “pass-through” mode. This allows SIP endpoints that support the same G.729 codec variant to talk each other.
- RFC 2833 is used for DTMF transmission. This is the same as configured for Avaya Communication Manager in **Section 3.4**.

For this test configuration, the following parameters were configured in the **avaya** context in the **sip.conf** configuration file. The **avaya** context is the context used in this test configuration for calls to Avaya endpoints.

- **type:** “user” Default for calls inbound from a SIP server.
- **fromdomain:** “avaya.com” SIP domain of Avaya SES from **Section 4.1**.
- **host:** “10.1.1.50” IP address of Avaya SES.
- **disallow:** “all” First, turn off support for all codecs.
- **allow:** “g729”, “ulaw” Turn on support for G.729 and G.711 μ -law codecs.
- **dtmfmode:** “rfc2833” Use RFC2833 for sending DTMF.
- **canreinvite:** “yes” Enable shuffling.

```
[avaya]
type=user ; Default value for calls inbound from a SIP server
fromdomain=avaya.com ; Domain of Avaya SES
host=10.1.1.50 ; IP address of Avaya SES
disallow=all ; First disallow all codecs
allow=g729 ; Allow G.729 as 1st codec for calls to Avaya
allow=ulaw ; Allow G.711 ulaw as 2nd codec for calls to Avaya
dtmfmode=rfc2833 ; Set dtmfmode to use RFC 2833 for sending DTMF.
canreinvite=yes ; Enable direct IP-IP media (shuffling)
```

5.4. Administer Dial Plan for Internal Calls and Calls from Avaya Endpoints

Unlike the configuration of the dial plan in Avaya Communication Manager, the dial plan in Asterisk is defined as a list of commands to execute for the extensions. The commands are generally executed in the order defined by a “priority” tag. The dial plan is defined in the following format in the `extensions.conf` configuration file. For more information on the dial plan, see Reference [15].

```
[<context>]
exten => <extension1>,<priority1>,<command1(parameters)>
exten => <extension2>,<priority2>,<command2(parameters)>
```

When an extension is dialed, the command tagged with a priority of 1 is executed, followed by the command tagged a with priority 2, and so on.

For this test configuration, the dial plan was configured in the `macro-stdexten` context. The command shown below defines the dial plan for internal calls and for calls from Avaya endpoints. This command indicates that internal extension calls will call the `Dial()` application passing along the number that was dialed, as the argument `ARG2`, and that the called party will be dialed for 20 seconds maximum. Typically, a call to a SIP station will use “Dial(SIP/\${EXTEN},20)” for `ARG2` where “\${EXTEN}” will be replaced by the called number. Note that the command shown below is the default command used for the dial plan for internal calls. For details on the command and to configure the dial plan differently, see References [14] and [15].

```
[macro-stdexten]
; exten => extension,priority,command(parameters) Standard definition for dial plan
exten => s,1,Dial(${ARG2},20) ; Ring the interface, 20 seconds maximum
```

5.5. Administer Dial Plan for Calls to Avaya Endpoints

Keeping with the format defined in **Section 5.4**, a command was added to cover numbers that are not covered by internal extensions. For this test configuration, the dial plan for calls to Avaya endpoints was configured in the `default` context in the `extensions.conf` configuration file. In this command, `extension` was set to “_24xxx” to cover calls to 5-digit numbers starting with 20 (20000-29999). `command` was set to “Dial(SIP/\${EXTEN}@avaya-out,20,)” where “\${EXTEN}” will be replaced by the called number. The inclusion of “@avaya-out” defines that the parameters defined for the `avaya-out` context (see **Section 5.2**) will be used to deliver the external call.

```
[default]
; exten => extension,priority,command(parameters) Standard definition for dial plan
exten => _20XXX,1,Dial(SIP/${EXTEN}@avaya-out,20,)
```

5.6. Administer Dial Plan to Deliver Caller ID to the Asterisk Voice Mail System

To ensure that the caller ID is passed along to the internal Asterisk voice mail system, the following must be administered in the **default** context in the **voicemail.conf** configuration file. The format defined in **Section 5.4** is used. For this test configuration, the voice mail pilot number was 60000. The “**\${CALLERIDNUM}**” parameter was added to provide the calling number to the voice mail system.

```
[default]

; exten => extension,priority,command(parameters) Standard definition for dial plan
; exten => 60000,1,VoiceMailMain ; Default configuration; commented out
exten => 60000,1,VoiceMailMain(${CALLERIDNUM}) ; Caller ID information added
exten => 60000,n,Hangup
```

5.7. Restart Asterisk

After all of the configuration changes are complete, the Asterisk server must be restarted. This can be done via the Asterisk console.

To access the Asterisk console, log into the Linux system using an account with appropriate permissions, such as the **root** account. From the command line interface (shown below), enter “**asterisk -r**” to access the console. Once in the console, enter “**restart gracefully**” to restart the Asterisk.

```
[root@sil-asterisk ~]# asterisk -r
Asterisk Business Edition ABE-B.1-3, Copyright (C) 1999 - 2006 Digium, Inc. and
others.
Created by Mark Spencer
Thank you for using Business Edition. This Software is provided by Digium Inc
under license. Please refer to the license agreement provided with the Software.
=====
Connected to Asterisk ABE-B.1-3 currently running on sil-asterisk (pid = 3594)
Verbosity is at least 3
sil-*CLI>
sil-*CLI> restart gracefully
sil-*CLI>

Disconnected from Asterisk server
```

6. Verification Steps

This section provides the tests that can be performed on Avaya Communication Manager and Avaya SES to verify proper configuration of Avaya Communication Manager and Asterisk.

6.1. Verify Avaya Communication Manager

6.1.1. Verify Status of Idle Trunk and Signaling Groups

Verify the status of the SIP trunk groups by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 3.4**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 6 Page 1
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0006/001	T00125	in-service/idle	no
0006/002	T00126	in-service/idle	no
0006/003	T00127	in-service/idle	no
0006/004	T00128	in-service/idle	no
0006/005	T00129	in-service/idle	no
0006/006	T00130	in-service/idle	no
0006/007	T00131	in-service/idle	no
0006/008	T00132	in-service/idle	no
0006/009	T00133	in-service/idle	no
0006/010	T00134	in-service/idle	no
0006/011	T00155	in-service/idle	no
0006/012	T00156	in-service/idle	no
0006/013	T00157	in-service/idle	no
0006/014	T00158	in-service/idle	no

```
status trunk 21 Page 1
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0021/001	T00165	in-service/idle	no
0021/002	T00166	in-service/idle	no
0021/003	T00167	in-service/idle	no
0021/004	T00168	in-service/idle	no
0021/005	T00169	in-service/idle	no
0021/006	T00170	in-service/idle	no
0021/007	T00171	in-service/idle	no
0021/008	T00172	in-service/idle	no
0021/009	T00173	in-service/idle	no
0021/010	T00174	in-service/idle	no
0021/011	T00175	in-service/idle	no
0021/012	T00176	in-service/idle	no
0021/013	T00177	in-service/idle	no
0021/014	T00178	in-service/idle	no

Verify the status of the SIP signaling groups by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 3.4**. Verify the signaling group is “in-service” as indicated in the **Group State** field shown below.

```
status signaling-group 6
                        STATUS SIGNALING GROUP

      Group ID: 6                Active NCA-TSC Count: 0
      Group Type: sip            Active CA-TSC Count: 0
      Signaling Type: facility associated signaling
      Group State: in-service
```

```
status signaling-group 21
                        STATUS SIGNALING GROUP

      Group ID: 21                Active NCA-TSC Count: 0
      Group Type: sip            Active CA-TSC Count: 0
      Signaling Type: facility associated signaling
      Group State: in-service
```

6.1.2. Verify Status of Connected Trunk Group Members

Make a call from the Avaya H.323 endpoint to the Avaya SIP endpoint. Verify the status of the SIP trunk group for Avaya SIP endpoints administered in **Section 3.4.1** by using the “status trunk n” command, where “n” is the trunk group number. Find the active trunk group member as shown below. In this example, trunk group member “15” on port “T00179” is connected to port “S00003”.

```
status trunk 21
                                                    Page 2

                        TRUNK GROUP STATUS

Member   Port      Service State      Mtce Connected Ports
          Port      State              Busy

0021/015 T00179 in-service/active no S00003
0021/016 T00180 in-service/idle   no
0021/017 T00181 in-service/idle   no
0021/018 T00182 in-service/idle   no
0021/019 T00183 in-service/idle   no
0021/020 T00184 in-service/idle   no
0021/021 T00185 in-service/idle   no
0021/022 T00186 in-service/idle   no
0021/023 T00187 in-service/idle   no
0021/024 T00188 in-service/idle   no
0021/025 T00189 in-service/idle   no
0021/026 T00190 in-service/idle   no
0021/027 T00191 in-service/idle   no
0021/028 T00192 in-service/idle   no
```

Verify the status of the connected SIP trunk group member by using the “status trunk x/y”, where “x” is the number of the SIP trunk group from **Section 3.4.1** to reach Avaya SIP endpoints, and “y” is the member number of a connected trunk, in this case trunk group “21” and member “15”. Verify that the **Service State** is “in-service/active”, and that the IP addresses of the C-LAN and SES server are shown in the **Signaling** section. In addition, the **Audio** section shows the G.711 codec and the IP addresses of the two Avaya endpoints. The **Audio Connection Type** displays “ip-direct”, indicating media shuffling.

```

status trunk 21/15                                     Page 1 of 2
                TRUNK STATUS

Trunk Group/Member: 0021/015                          Service State: in-service/active
                  Port: T00179                        Maintenance Busy? no
Signaling Group ID:

IGAR Connection? no

Connected Ports: S00003

                Port      Near-end IP Addr : Port      Far-end IP Addr : Port
Signaling: 01A0117  10. 1. 1. 10 : 5061      10. 1. 1. 50 : 5061
G.711MU   Audio:      10. 1. 1. 94 : 2258      10. 1. 1. 96 : 5004
                Video:
Video Codec:

                Authentication Type: None
Audio Connection Type: ip-direct

```

Make a call between the Avaya H.323 and Asterisk endpoints. Similar to the above steps shown for a call between an Avaya H.323 and an Avaya SIP endpoint, find the active trunk group member. In this example, trunk group member “6” on port “T00133” is connected to port “S00003”. Verify the status of the connected SIP trunk groups member by using the “status trunk x/y”, where “x” is the number of the SIP trunk group from **Section 3.4.2** to reach Asterisk endpoints, and “y” is the member number of a connected trunk, in this case trunk group “6” and member “9”. Verify that the **Service State** is “in-service/active”, and that the IP addresses of the C-LAN and SES server are shown in the **Signaling** section. In addition, the **Audio** section shows the G.729 codec and the IP addresses of the Avaya H.323 and Asterisk SIP endpoints. The **Audio Connection Type** displays “ip-direct”, indicating direct media between the two endpoints.

```

status trunk 6/9
                                     Page 1 of 2
                                     TRUNK STATUS

Trunk Group/Member: 0006/009          Service State: in-service/active
Port: T00133                          Maintenance Busy? no
Signaling Group ID:

IGAR Connection? no

Connected Ports: S00003

          Port      Near-end IP Addr : Port      Far-end IP Addr : Port
Signaling: 01A0117  10. 1. 1. 10 : 5061      10. 1. 1. 50 : 5061
G.729      Audio:      10. 1. 1. 94 : 2258      10. 2. 2.109 : 5004
          Video:
          Video Codec:

                                     Authentication Type: None
Audio Connection Type: ip-direct

```

6.2. Verify Asterisk Business Edition PBX

Verify that internal calls are using the proper dial plan and that the calls are shuffled. Make a call from one Asterisk endpoint to another, in this example from 60109 to 60111, and monitor the Asterisk console. Verify that the `Dial()` command is called using the SIP application between the calling party endpoint to the called party endpoint. Verify that the call is being shuffled as shown in the “native bridge” statement.

```

-- Executing Macro("SIP/60109-09e08390", "stdexten|60111|SIP/60111") in new stack
-- Executing Dial("SIP/60109-09e08390", "SIP/60111|20") in new stack
-- Called 60111
-- SIP/60111-09e10a18 is ringing
-- SIP/60111-09e10a18 answered SIP/60109-09e08390
-- Attempting native bridge of SIP/60109-09e08390 and SIP/60111-09e10a18

```


Verify that outgoing calls to Avaya endpoints are using the proper dial plan and that the calls are shuffled. Make a call from one Asterisk endpoint to an Avaya endpoint, in this case from 60109 to 24096, and monitor the Asterisk console. Verify that the `Dial()` command is called using the SIP application between the Asterisk endpoint and the Avaya endpoint. Verify that the “avaya-out” context is used. Verify that the call is being shuffled as shown in the “native bridge” statement.

```
sil-*CLI>
-- Executing Dial("SIP/60111-09e08390", "SIP/24096@avaya-out|20|") in new stack
-- Called 24096@avaya-out
-- SIP/avaya-out-09e10a18 is ringing
-- SIP/avaya-out-09e10a18 is making progress passing it to SIP/60111-09e08390
-- SIP/avaya-out-09e10a18 answered SIP/60111-09e08390
-- Attempting native bridge of SIP/60111-09e08390 and SIP/avaya-out-09e10a18
sil-*CLI>
```

6.3. Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Basic calls between various endpoints at the Main and Remote sites were made successfully in both directions via SIP trunks using G.711 μ -law and G.729 codecs. As mentioned previously, the Asterisk used in this test configuration was not licensed for the G.729 codec which limits G.729 codec interoperability. This requires that the two endpoints in a call support the same variant of the G.729 codec when negotiating the G.729 codec.
- Proper display of the calling party name and number information were verified for all endpoints with the basic call scenario. The Avaya SIP endpoints displayed the calling party name and all other endpoints displayed the calling party name and number.
- DTMF was verified across the SIP trunks where Avaya endpoints at the Main site successfully accessed the Asterisk voice mail mailboxes at the Remote site and endpoints at the Remote site successfully accessed Avaya Modular Messaging mailboxes at the Main site.
- Supplementary calling features were verified between various endpoints on the Main and Remote sites connected via SIP trunks. The feature scenarios involved additional endpoints on both the local and remote sites, such as performing an unattended transfer of the SIP trunk call to a local endpoint on the same site, and then repeating the scenario to transfer the SIP trunk call to a remote endpoint on the other site. The list of verified supplementary calling features includes:
 - Unattended transfer
 - Attended transfer
 - Hold/Unhold
 - Music on Hold/Tone on Hold
 - Consultation hold
 - Call forwarding
 - Conference

7. Conclusion

As illustrated in these Application Notes, Avaya Communication Manager can interoperate with Asterisk Business Edition PBX Version B.1-3 using SIP trunks via Avaya SES. The following is a list of interoperability items to note:

- One of the unattended transfer scenarios fails. In the scenario where an endpoint at the Remote site calls a SIP endpoint at the Main site and the Remote endpoint then transfers (unattended) the call to another endpoint at the Main site, the call does not always succeed (no audio). The workaround is to use an attended transfer in this scenario instead.
- For full G.729 codec interoperability, Asterisk must be licensed to support the G.729 codec. In an un-licensed mode, calls using the G.729 codec may not succeed as different endpoints support different variants of the G.729 codec.
- The Avaya 4600 Series IP Telephones running the SIP software were not used in the Main site for the test configuration. Due to the issue documented in Reference [16], calls to Asterisk endpoints from the Avaya 4600 Series IP Telephones running the SIP software did not succeed. Also, when a Cisco 7960 IP Phone was used in the Main site in this test configuration, calls to Asterisk endpoints from the Cisco SIP telephone did not succeed. Depending on the overall configuration, calls to Asterisk endpoints from other SIP endpoints that are connected to Avaya SES may also not succeed.

8. Additional References

The following are additional references. The Avaya documents are available at <http://support.avaya.com>.

- [1] *Installing and Administering SIP Enablement Services R4.0*, Issue 1.4, Doc ID 03-600768, May 2007.
- [2] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500, and S8700 series Media Server*, Issue 7, Doc ID 555-245-206, May, 2007.
- [3] *Avaya Extension to Cellular and OPS Installation and Administration Guide*, Version 6.0 Issue 9, Doc ID 210-100-500, June 2005.
- [4] *Administrator Guide for Avaya Communication Manager*, Issue 3.1, Doc ID 03-300509, February 2007.
- [5] *Messaging Application Server Administration Guide*, Release 3.1.
- [6] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide*, Release 1.0, Doc ID 16-601944, May 2007.
- [7] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, Release 1.5, Doc ID 16-300698, May 2007.
- [8] *4600 Series IP Telephone Release 2.8 LAN Administrator Guide*, Doc ID 555-233-507, February 2007.
- [9] *Asterisk™: The Future of Telephony*, First Edition, August 31, 2005, ISBN 0-596-00962-3, available at <http://www.oreilly.com/>.
- [10] *Asterisk Business Edition Technical Reference, Version B*. July 26, 2006. Available at <https://be.digium.com/documentation>.
- [11] *Asterisk Business Edition QuickStart Guide*. August 3, 2006. Available at <http://www.digium.com/en/products/software/abe.php>
- [12] Asterisk WiKi page on voip-info.org. <http://www.voip-info.org/wiki-Asterisk>.
- [13] *Asterisk cmd Dial*. Available at <http://www.voip-info.org/wiki/index.php?page=Asterisk+cmd+Dial>.
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