



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Frontier SIP Trunking Service with Avaya Aura® Communication Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.0**

### **Abstract**

These Application Notes describe the steps to configure a Session Initiation Protocol (SIP) trunk between the Frontier SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager Evolution Server 6.3, Avaya Session Border Controller for Enterprise 6.2 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Session Border Controller for Enterprise.

Frontier is a member of the Avaya DevConnect Service Provider Program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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# 1. Introduction

These Application Notes describe the steps to configure a SIP trunk between Frontier SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 6.3 Communication Manager configured as an Evolution Server, Avaya Session Border Controller for Enterprise 6.2(Avaya SBCE) and various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with Frontier are able to place and receive PSTN calls via a broadband Internet connection. This converged network solution is an alternative to a traditional PSTN trunk such as analog and/or ISDN-PRI.

## 2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Frontier is a member of the Avaya DevConnect Service Provider Program. The general test approach is to connect a simulated enterprise to Frontier via the Internet and exercise the features and functionalities listed in **Section 2.1**.

### 2.1. Interoperability Compliance Testing

To verify Frontier SIP Trunking Service interoperability, the following features and functionalities are covered in the compliance testing:

- Inbound PSTN calls to various phone types including H.323, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outbound PSTN calls from various phone types including H.323, digital and analog telephone at the enterprise. All outbound calls to PSTN are routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator soft phone. Both the Computer Mode (where Avaya one-X® Communicator is used for call control as well as audio path) and the Telecommuter Mode (where Avaya one-X® Communicator is used for call control and a separate telephone is used for audio path) are tested.
- Dialing plans including local, long distance, international, outbound toll-free, operator assisted, local directory assistance (411) calls etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Codec G.711MU and G.729.
- Media and Early Media transmissions.
- Incoming and outgoing fax using G.711MU.
- DTMF tone transmissions as out-of-band RTP events as per RFC2833.

- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, forward and conference.
- Off-net call forward with Diversion method.
- EC500 mobility (extension to cellular) with Diversion method.
- Routing inbound vector call to call center agent queues.
- Response to OPTIONS heartbeat.
- Response to incomplete call attempts and trunk errors.
- Session Timers implementation.

Items that are supported and not tested:

- Inbound toll-free.
- Emergency calls (911).

Items that are not supported

- Fax T.38 is not supported.
- Incoming call redirection on VDN before answer using “302 Moved Temporarily” method is not supported.
- Incoming call redirection after answer of incoming VDN calls using REFER method is not supported.
- Avaya IP Desk-phones running SIP firmware are not supported.

## 2.2. Test Results

Interoperability testing of Frontier SIP Trunking Service with the Avaya SIP-enabled enterprise solution is completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Call Display on PSTN Phone** – Call display was not properly updated on PSTN phone involved in a call transfer. After the call transfer was completed, the PSTN phone did not display the actual connected party but instead showed the party that initiated the transfer. However, it does not affect the end user.
- **Call Display on PSTN Phone on PSTN Hold and Resume** – Call display was not properly updated on PSTN phone involved in PSTN Hold and Resume operation. After the inbound call from PSTN to Communication Manager was hold on PSTN phone, when the call was resumed on the PSTN phone, the PSTN phone displayed trunk number instead of the calling party ID. No user impact.

## 2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Frontier SIP Trunking Service, please contact Frontier technical support at <http://frontier.com/enterprise>.

### 3. Reference Configuration

**Figure 1** illustrates the sample Avaya SIP-enabled enterprise solution connected to the Frontier SIP Trunking Service (Vendor Validation circuit) through a public Internet connection.

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

The Avaya components used to create the simulated customer site included:

- Avaya S8800 Servers running Communication Manager
- Avaya G450 Media Gateway
- Avaya S8800 Server running Messaging
- Avaya Session Border Controller for Enterprise
- Avaya 9600Series IP Deskphones (H.323)
- Avaya one-X® Communicator soft phones (H.323)
- Avaya digital and analog telephones

Located at the edge of the enterprise network is the Avaya SBCE. It has a public side that connects to Frontier via Internet and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flows through the Avaya SBCE which can protect the enterprise against any outside SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. The transport protocol between the Avaya SBCE and Frontier across the public network is UDP. The transport protocol between the Avaya SBCE and Communication Manager is TCP.

[illegible]

QT; Reviewed:  
SPOC 10/15/2014

## 4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on an Avaya S8800 Server	6.3 (Avaya Communication Manager/ R016x.03.0.124.0 with Service Pack (03.0.124.0-21591))
Avaya G450 Media Gateway	33.13.0
Avaya Aura® Messaging running on an Avaya S8800 Server	6.1-11.0
Avaya Session Border Controller for Enterprise	6.2.1 Q16
Avaya 9611G IP Deskphone (H.323)	Avaya one-X® Deskphone Edition S6.0.0
Avaya 9630G IP Deskphone (H.323)	Avaya one-X® Deskphone Edition 3.1 SP5
Avaya one-X Communicator (H.323)	6.1.7.04-SP7-39506
Avaya 1408 Digital Telephone	1400R10
Avaya 6210 Analog Telephone	n/a
Frontier SIP Trunking Service Components	
Component	Release
Acme Packet Net-Net 4000 SBC	6.2
Metaswitch	8.1

**Table 1: Equipment and Software Tested**

**Note:** This solution will be compatible with other Avaya Server and Media Gateway platforms running similar version of Communication ManagerCommunication Manager.



## 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for Frontier SIP Trunking. It is assumed the general installation of Communication Manager and Avaya G450 Media Gateway has been previously completed and is not discussed here.

The configuration of Communication Manager was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

### 5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sale representative to add the additional capacity or feature.

<b>display system-parameters customer-options</b>		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	4000	50
Maximum Concurrently Registered IP Stations:	2400	1
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	3
<b>Maximum Administered SIP Trunks: 24000</b>		<b>289</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0
Maximum Number of DS1 Boards with Echo Cancellation:	80	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	50	1
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0
Maximum Number of Expanded Meet-me Conference Ports:	300	0
(NOTE: You must logoff & login to effect the permission changes.)		

## 5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow an incoming call from the PSTN to be transferred to another PSTN endpoint. If for security reasons, incoming calls should not be allowed to transfer back to the PSTN then leave the field set to **none**.

```
change system-parameters features                               Page 1 of 20
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? y
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      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
```

On **Page 9** verify that a text string has been defined to replace the Calling Party Number (CPN) for restricted or unavailable calls. The compliance test used the value of **anonymous** for restricted calls and unavailable calls.

```
change system-parameters features                               Page 9 of 20
      FEATURE-RELATED SYSTEM PARAMETERS

      CPN/ANI/ICLID PARAMETERS
      CPN/ANI/ICLID Replacement for Restricted Calls: anonymous
      CPN/ANI/ICLID Replacement for Unavailable Calls: anonymous

      DISPLAY TEXT
      Identity When Bridging: principal
      User Guidance Display? n
      Extension only label for Team button on 96xx H.323 terminals? n

      INTERNATIONAL CALL ROUTING PARAMETERS
      Local Country Code: 1
      International Access Code: 001

      ENBLOC DIALING PARAMETERS
      Enable Enbloc Dialing without ARS FAC? n

      CALLER ID ON CALL WAITING PARAMETERS
      Caller ID on Call Waiting Delay Timer (msec): 200
```

### 5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and Avaya Session Border Controller for Enterprise (**ASBCE62**). These node names will be needed for defining the service provider signaling groups in **Section 5.6**.

change node-names ip		Page 1 of 2
		IP NODE NAMES
Name	IP Address	
ASBCE62	10.10.98.13	
DevAAM	10.33.10.9	
default	0.0.0.0	
procr	10.10.97.246	
procr6	::	

### 5.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to be used for calls between the enterprise and the service provider. This compliance test used ip-codec-set 1. Frontier supports G.711MU and G.729. To use this codec, enter **G.711MU** and **G.729** in the **Audio Codec** column of the table in the order of preference.

The following screen shows the configuration for ip-codec-set 1. During testing, the codec set specifications are varied to test for individual codec support as well as codec negotiation between the enterprise and the network at call setup time.

change ip-codec-set 1

Page1 of 2

IP Codec Set

Codec Set: 1

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

On **Page 2**, set the **Fax Mode** to *pass-through* faxing which is supported by Frontier.

change ip-codec-set 1		Page 2 of 2
		IP Codec Set
		Allow Direct-IP Multimedia? n
	Mode	Redundancy
FAX	pass-through	1
Modem	off	0
TDD/TTY	US	3
Clear-channel	n	0

## 5.5. IP Network Region

A separate IP network region for the service provider trunk group is created. This allows separate codec or quality of service setting to be used (if necessary) for a call between the enterprise and the service provider versus a call within the enterprise or elsewhere. For the compliance testing, ip-network-region 1 was created by the **change ip-network-region 1** command with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In the compliance testing, the domain name is **bvwlabs.com**. This domain name appears in the “From” header of SIP message originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Media Gateway. By default, both **Intra-region** and **Inter-region IP-IP Direct Audio** are set to **yes**. Shuffling can be further restricted at the trunk level under the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values can be used for all other fields.

change ip-network-region 1		Page 1 of 20	
		IP NETWORK REGION	
Region: 1			
Location: 1	Authoritative Domain: bvwlabs.com		
Name: ToDevASM			
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: yes	
Codec Set: 1		Inter-region IP-IP Direct Audio: yes	
UDP Port Min: 2048		IP Audio Hairpinning? n	
UDP Port Max: 3329			
DIFFSERV/TOS PARAMETERS			
Call Control PHB Value: 46			
Audio PHB Value: 46			
Video PHB Value: 26			
...			

On **Page 4**, define the IP codec set to be used for traffic between region 1 and other regions. In the compliance testing, Communication Manager, the Avaya G450 Media Gateway, IP phones and the Avaya SBCE were assigned to the same region 1. To configure the IP codec set between regions, enter the desired IP codec set in the **codec set** column of the table with appropriate destination region (**dst rgn**). Default values may be used for all other fields. The example below shows codec set 1 will be used for a call between region 1 and other regions.

change ip-network-region 1										Page 4 of 20			
Source Region: 1		Inter Network Region Connection Management								I		M	
										G		A	
dst codec direct		WAN-BW-limits		Video		Intervening		Dyn		A		G	
rgn set		WAN Units		Total Norm		Prio Shr		Regions		CAC		R	
1 1												all	
2 1		y		NoLimit						n		t	
3										n		t	

Non-IP telephones (e.g., analog, digital) derive network region from IP interface of the Avaya G450 Media Gateway to which the device is connected. IP telephones can be assigned a network region based on an IP address mapping.

To define network region 1 for IP interface **procr**, use **change ip-interface procr** command as shown in the following screen.

<b>change ip-interface pr</b>		Page 1 of 2
IP INTERFACES		
<b>Type: PROCR</b>		
		Target socket load: 4800
Enable Interface? y	Allow H.323 Endpoints? y	
	Allow H.248 Gateways? y	
<b>Network Region: 1</b>	Gatekeeper Priority: 5	
...		

To define network region 1 for the Avaya G450 Media Gateway, use **change media-gateway** command as shown in the following screen.

<b>change media-gateway 1</b>		Page 1 of 2
MEDIA GATEWAY 1		
Type: g450		
Name: G450		
Serial No: 12TG18000244		
Encrypt Link? y	Enable CF? n	
<b>Network Region: 1</b>	Location: 1	
	Site Data:	
Recovery Rule: none		
...		

## 5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Avaya SBCE for use by the service provider trunk. This signaling group is used for inbound and outbound calls between the service provider and the enterprise. For the compliance test, signaling group **50** was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **y**.
- Set the **Transport Method** to **tcp**. The transport method specified here is used between Communication Manager and Avaya SBCE.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to **5060**.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP interface of **procr** defined in **Section 5.3**.

- Set the **Far-end Node Name** to *ASBCE62*. This node name maps to the IP address of Avaya SBCE as defined in **Section 5.3**.
- Set the **Far-end Network Region** to the IP network region *1* defined for the service provider in **Section 5.5**.
- Set the **Far-end Domain** to *bvwlab.com*.
- Set the **DTMF over IP** to *rtp-payload*. This setting enables Communication Manager to send or receive the DTMF transmissions using RFC2833.
- Set **Enable Layer 3 Test?** to *y*. This setting allows Communication Manager to send OPTIONS heartbeat to Avaya SBCE on the SIP trunk.
- Set **Direct IP-IP Audio Connections** to *y*. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint. If this value is set to *n*, then the Avaya G450 Media Gateway will remain in the media path between the SIP trunk and the endpoint for the duration of the call. Depending on the number of media resources available in the Avaya G450 Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **Direct IP-IP Early Media** is set to *n*.
- Set the **Alternate Route Timer** to *30*. This defines the number of seconds Communication Manager will wait for a response (other than 100 Trying) to an outbound INVITE before canceling the call.
- Default values may be used for all other fields.

<b>add signaling-group 50</b>		Page 1 of 1
SIGNALING GROUP		
Group Number: 50	Group Type: sip	
IMS Enabled? y	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y Peer Server: Others		
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y		
Near-end Node Name: procr	Far-end Node Name: ASBCE62	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: bvwlab.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early Media? n	
	Alternate Route Timer(sec): 30	

## 5.7. Trunk Group

Use the **add trunk-group** command to create trunk group for the signaling group created in **Section 5.6**. For the compliance testing, trunk group **50** was configured using the parameters highlighted below.

- Set the **Group Type** field to *sip*.
- Enter a descriptive name for the **Group Name**.
- Enter an available Trunk Access Code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Outgoing Display** to *y* to enable name display on the trunk.
- Set the **Service Type** field to *public-ntwrk*.
- Set the **Signaling Group** to the signaling group shown in **Section 5.6**.
- Set the **Number of Members** field to **32**. It is the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk group.
- Default values are used for all other fields.

```
add trunk-group 50                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 50                                     Group Type: sip          CDR Reports: y
  Group Name: SP Trunk                               COR: 1                 TN: 1             TAC: *003
  Direction: two-way                                Outgoing Display? y
  Dial Access? n                                     Night Service:
  Queue Length: 0
  Service Type: public-ntwrk                         Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 50
                                                    Number of Members: 32
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval (sec)** is set to a value acceptable to service provider. This value defines the interval a re-INVITES must be sent to refresh the Session Timer. For the compliance testing, a default value of **600** seconds was used.

```
add trunk-group 50                                     Page 2 of 21
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name: auto
                                                    Redirect On OPTIM Failure: 15000
  SCCAN? n                                           Digital Loss Group: 18
                                                    Preferred Minimum Session Refresh Interval(sec): 600
  Disconnect Supervision - In? y Out? y
  XOIP Treatment: auto                               Delay Call Setup When Accessed Via IGAR? n
```

On **Page 3**, set the **Numbering Format** field to *private*. This field specifies the format of the CPN sent to the far-end. The public numbers are automatically preceded with a + sign when passed in the “From”, “Contact” and “P-Asserted Identity” headers. The **Numbering Format** is set to *private* and the **Numbering Format** in the route pattern is set to *lev0-pvt* (see **Section 5.98**).

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to *y*. This will allow the CPN displayed on the local endpoint to be replaced with the value set in **Section 5.2**, if the inbound call enabled CPN block. Default values are used for all other fields.

add trunk-group 50	Page 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Numbering Format: private	UI Treatment: service-provider
	Replace Restricted Numbers? y
	Replace Unavailable Numbers? y
Modify Tandem Calling Number: no	
Show ANSWERED BY on Display? y	

On **Page 4**, the **Network Call Redirection** field should be set to *n*. The setting of **Network Call Redirection** flag to *y* enables use of the SIP REFER message to transfer an inbound call back to the PSTN.

- Set **Mark Users as Phone** to *y*.
- Set the **Send Diversion Header** field to *y*. This field provides additional information to the network if the call has been re-directed. This is needed to support call forwarding of inbound call back to PSTN and Extension to Cellular (EC500) call scenarios.
- Set the **Support Request History** field to *n*. This parameter determines if History-Info header will be excluded in the call-redirection INVITE from the enterprise.
- Set the **Telephone Event Payload Type** to *101*, the value is preferred by Frontier.

add trunk-group 50	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? y	
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? y	
Support Request History? n	
Telephone Event Payload Type: 101	
...	



## 5.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering is selected to define the format of this number (**Section 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID numbers are provided by the service provider. They are used to authenticate the caller.

The screen below shows a subset of the DID numbers assigned for testing. These 4 numbers were mapped to the 4 enterprise extensions 1130, 1131, 1132 and 1133. These same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these extensions.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
4	1130	50	5856664803	10	Total Administered: 11
4	1131	50	5856664804	10	Maximum Entries: 540
4	1132	50	5856664805	10	
4	1133	50	5856664806	10	

Even though private numbering is selected, currently the number used in the SIP Diversion header is derived from the public unknown numbering table and not the private numbering table. As a workaround for this, the entries in the private numbering table must be repeated in the public unknown numbering table.

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	
4	1130	50	5856664803	10	Total Administered: 4
4	1131	50	5856664804	10	Maximum Entries: 240
4	1132	50	5856664805	10	Note: If an entry applies to
4	1133	50	5856664806	10	a SIP connection to Avaya
					Aura(R) Session Manager,
					the resulting number must
					be a complete E.164 number.

## 5.9. Inbound Routing

In general, the incoming call handling treatment for a trunk group can be used to manipulate the digits received for an incoming call if necessary. DID number sent by Frontier can be mapped to an extension using the incoming call handling treatment of the receiving trunk-group. Use the **change inc-call-handling-trmt trunk-group** command to create an entry for each DID.

change inc-call-handling-trmt trunk-group 50					Page	1 of	3
INCOMING CALL HANDLING TREATMENT							
Service/ Feature	Number Len	Number Digits	Del	Insert			
public-ntwrk	10	5856664803	10	1130			
public-ntwrk	10	5856664804	10	1131			
public-ntwrk	10	5856664805	10	1132			
public-ntwrk	10	5856664806	10	1133			

## 5.10. Outbound Routing

In these Application Notes, the **Automatic Route Selection (ARS)** feature is used to route an outbound call via the SIP trunk to the service provider. In the compliance testing, a single digit 9 was used as the ARS access code. An enterprise caller will dial 9 to reach an outside line. To define feature access code (**fac**) 9, use the **change dialplan analysis** command as shown below.

change dialplan analysis										Page	1 of	12
DIAL PLAN ANALYSIS TABLE												
Location: all										Percent Full: 1		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type				
11	4	ext										
3	4	udp										
4	4	ext										
6	1	fac										
6	4	ext										
7	4	ext										
9	1	fac										

Use the **change feature-access-codes** command to define 9 as the **Auto Route Selection (ARS)** – Access Code 1.

change feature-access-codes										Page	1 of	10
FEATURE ACCESS CODE (FAC)												
Abbreviated Dialing List1 Access Code:												
Abbreviated Dialing List2 Access Code:												
Abbreviated Dialing List3 Access Code:												
Abbreviated Dial - Prgm Group List Access Code:												
Announcement Access Code: *007												
Answer Back Access Code:												
Attendant Access Code:												
Auto Alternate Routing (AAR) Access Code: *008												
Auto Route Selection (ARS) - Access Code 1: 9										Access Code 2:		

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. The example below shows a subset of the dialed strings tested as part of the compliance testing. See **Section 2.1** for the complete list of call types tested. All dialed strings are mapped to route pattern **50** for an outbound call which contains the SIP trunk to the service provider (as defined next).

change ars analysis 0							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
0	1	11	50	op		n	
011	13	13	50	intl		n	
1	11	11	50	pubu		n	
300	10	10	50	pubu		n	
411	3	3	50	svcl		n	
613	10	10	50	pubu		n	
866	10	10	50	pubu		n	
911	3	3	50	svcl		n	
585	10	10	50	pubu		n	

As being mentioned above, the route pattern defines which trunk group will be used for the outbound calls and performs necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for route pattern **50** in the following manner.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider. For the compliance testing, trunk group **50** was used.
- **FRL:** Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format:** **unk-unk** All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.8**.

change route-pattern 50															Page	1	of	3			
Pattern Number: 50															Pattern Name: SP Route						
SCCAN? n															Secure SIP? n						
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted								DCS/	IXC					
No			Mrk	Lmt	List	Del	Digits								QSIG						
																				Intw	
1:	50	0													n	user					
2:															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						unk-unk		none			
...																					

## 5.11. Saving Configuration Changes

The command “**save translation all**” can be used to save the configuration changes made on Communication Manager.

## 6. Configure Avaya Session Border Controller for Enterprise

In the sample configuration, an Avaya SBCE is used as the edge device between the Avaya CPE and Frontier SIP Trunking service.

These Application Notes assume that the installation of the Avaya SBCE and the assignment of a management IP Address have already been completed.

### 6.1. Avaya Session Border Controller for Enterprise Login

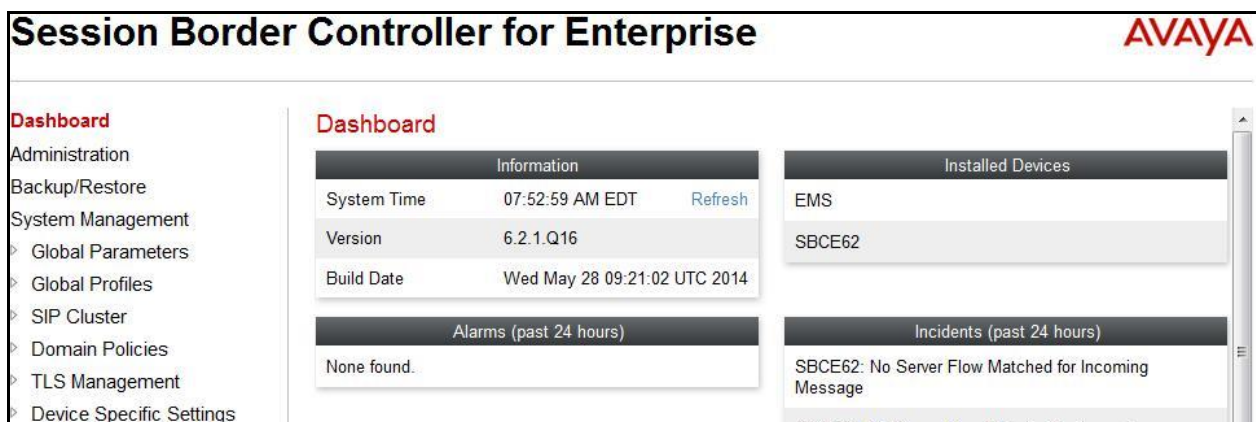
Use a Web browser to access the Avaya SBCE web interface, enter `https://<ip-addr>/ucsec` in the address field of the web browser (not shown), where `<ip-addr>` is the management LAN IP address of Avaya SBCE.

Enter appropriate credentials and click **Log In**.



The login page features the Avaya logo on the left. On the right, there is a 'Log In' section with fields for 'Username:' and 'Password:', followed by a 'Log In' button. Below the login fields, there is a disclaimer: 'This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws. The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials. All users must comply with all corporate instructions regarding the protection of information assets. © 2011 - 2013 Avaya Inc. All rights reserved.'

The main page of the Avaya SBCE will appear as shown below.



The dashboard displays system information and installed devices. On the left is a navigation menu. The main content area includes a 'Dashboard' header, an 'Information' table, an 'Installed Devices' table, and two 'Alarms' and 'Incidents' sections.

Information	
System Time	07:52:59 AM EDT <a href="#">Refresh</a>
Version	6.2.1.Q16
Build Date	Wed May 28 09:21:02 UTC 2014

Installed Devices
EMS
SBCE62

Alarms (past 24 hours): None found.

Incidents (past 24 hours): SBCE62: No Server Flow Matched for Incoming Message

## 6.2. Global Profiles

Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

### 6.2.1. Uniform Resource Identifier (URI) Groups

URI Group feature allows a user to create any number of logical URI Groups that are comprised of individual SIP subscribers located in that particular domain or group. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

For this configuration testing, “\*” is used for all incoming and outgoing traffic.

### 6.2.2. Routing Profiles

Routing Profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information and packet transport types.

To create a Routing Profile, select **Global Profiles → Routing**. Click on the **Add** button.

In the compliance testing, a Routing Profile **EN-to-SP** was created to use in conjunction with the server flow defined for Communication Manager. This entry is to route the outbound call from the enterprise to Frontier.

In the opposite direction, a Routing Profile named **SP-to-EN** was created to be used in conjunction with the server flow defined for Frontier. This entry is to route the inbound call from Frontier to the enterprise.

### 6.2.2.1 Routing Profile for Frontier

The screenshot below illustrates the routing profile from Avaya SBCE to the Frontier network, **Global Profiles → Routing: EN-to-SP**. As shown in **Figure 1**, the Frontier SIP trunk is connected with transportation protocol UDP. If there is a match in the “To” or “Request URI” headers with the URI Group **SP** defined in **Section 6.2.1**, the call will be routed to the **Next Hop Server 1** which is the IP address of Frontier SIP trunk on port 5060.



### 6.2.2.2 Routing Profile for Communication Manager

The Routing Profile for Frontier to Communication Manager, **SP-to-EN**, was defined to route call where the “To” header matches the URI Group **SP** defined in **Section 6.2.1** to **Next Hop Server 1** which is the IP address of Communication Manager, on port 5060 as a destination. As shown in **Figure 1**, the SIP trunk between Communication Manager and the Avaya SBCE is connected with transportation protocol TCP.



### 6.2.3. Topology Hiding

Topology Hiding is an Avaya SBCE security feature which allows changing certain key SIP message parameters to ‘hide’ or ‘mask’ how the enterprise network may appear to an unauthorized or malicious user.

To create a Topology Hiding profile, select **Global Profiles → Topology Hiding**. Click on the **Add** button.

In the compliance testing, two Topology Hiding profiles **EN-to-SP** and **SP-to-EN** were created.

### 6.2.3.1 Topology Hiding Profile for Frontier

Profile **EN-to-SP** was defined to mask the enterprise SIP domain **bvwlab.com** in “Request-URI” and “To” headers to Frontier IP address and “From” header to the Avaya SBC external interface IP address; mask the enterprise SIP domain **bvwlab.com** in the “From” and “PAI” headers to IP **10.10.98.111** (the Avaya SBCE public IP address). It is to secure the enterprise network topology and to meet the SIP requirement of the service provider.

#### Notes:

- The **Criteria** should be selected as **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in URI-Host.
- The masking applied on “From” header also applies to “Referred-By” and “P-Asserted-Identity” headers.
- The masking applied on “To” header also applies to “Refer-To” header.

The screenshots below illustrate the Topology Hiding profile **EN-to-SP**.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left sidebar shows a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, Server Configuration, Topology Hiding (highlighted), Signaling Manipulation, URI Groups, and SIP Cluster. The main content area is titled "Topology Hiding Profiles: EN-to-SP" and includes buttons for Add, Rename, Clone, and Delete. Below this, there's a section for "Topology Hiding" with a table of configurations. The table has columns for Header, Criteria, Replace Action, and Overwrite Value. The configurations are as follows:

Header	Criteria	Replace Action	Overwrite Value
Refer-To	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	10.10.98.111
To	IP/Domain	Overwrite	192.168.248.132
Record-Route	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	192.168.248.132
Referred-By	IP/Domain	Auto	---

An "Edit" button is located at the bottom right of the table.

### 6.2.3.2 Topology Hiding Profile for Communication Manager

Profile **SP-to-EN** was also created to mask Frontier URI-Host in “Request-URI”, “From”, “To” headers to the enterprise domain **bvwlab.com**, replace Record-Route, Via headers and SDP added by Frontier to internal IP address known to Communication Manager.

#### Notes:

- The **Criteria** should be **IP/Domain** to give the Avaya SBCE the capability to mask both domain name and IP address present in URI-Host.
- The masking applied on “From” header also applies to “Referred-By” and “P-Asserted-Identity” headers.
- The masking applied on “To” header also applies to “Refer-To” header.

The screenshots below illustrate the Topology Hiding profile **SP-to-EN**.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left sidebar shows the navigation menu with 'Global Profiles' expanded, and 'Topology Hiding' selected. The main content area is titled 'Topology Hiding Profiles: SP-to-EN'. It features a list of profiles on the left: 'default', 'SP-to-EN' (highlighted), and 'EN-to-SP'. An 'Add' button is above this list. On the right, the 'Topology Hiding' tab is active, showing a table of header configurations. Above the table is a blue bar with the text 'Click here to add a description.' and buttons for 'Rename', 'Clone', and 'Delete'. Below the table is an 'Edit' button.

Header	Criteria	Replace Action	Overwrite Value
Refer-To	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
From	IP/Domain	Overwrite	bwlab.com
To	IP/Domain	Overwrite	bwlab.com
Record-Route	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	bwlab.com
Referred-By	IP/Domain	Auto	---

#### 6.2.4. Server Interworking

Interworking Profile features are configured differently for Call Server and Trunk Server.

To create a Server Interworking profile, select **Global Profiles → Server Interworking**. Click on the **Add** button.

In the compliance testing, two Server Interworking profiles were created for Frontier and Communication Manager respectively.



### 6.2.4.1 Server Interworking profile for Frontier

Profile **SP=SI** was defined to match the specification of Frontier. The **General** and **Advanced** settings are configured with the following parameters while the other settings for **Timers**, **URI Manipulation** and **Header Manipulation** are kept as default.

General settings:

- **Hold Support** = *NONE*. The Avaya SBCE will not modify the hold/ resume signaling from Communication Manager to Frontier.
- **18X Handling** = *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from Communication Manager to Frontier.
- **Refer Handling** = *No*. The Avaya SBCE will not handle REFER. It will keep the REFER message unchanged from Communication Manager to Frontier.
- **T.38 Support** = *No*. Frontier does not support T.38 fax in the compliance testing.
- **Privacy Enabled** = *No*. The Avaya SBCE will not mask the “From” header with anonymous for the outbound call to Frontier. It depends on Communication Manager to enable/ disable privacy on an individual call basis.
- **DTMF Support** = *None*. The Avaya SBCE will send original DTMF method from Communication Manager to Frontier.

Advanced settings:

- **Record Routes** = *Both Sides*. The Avaya SBCE will send “Record-Route” header to both call and trunk servers.
- **Topology Hiding: Change Call-ID** = *Yes*. The Avaya SBCE will modify “Call-ID” header for the call toward Frontier.
- **Change Max Forwards** = *Yes*. The Avaya SBCE will adjust the original Max-Forwards value from Communication Manager to Frontier by reducing the intermediate hops involving in the call flow.
- **Has Remote SBC** = *Yes*. Frontier has a SBC which interfaces its Central Office (CO) to the enterprise SIP trunk. This setting allows the Avaya SBCE to always use the SDP received from Frontier for the media.

The screenshots below illustrate the Server Interworking profile **SP-SI**.

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
Global Parameters  
Global Profiles  
Domain DoS  
Fingerprint  
**Server Interworking**  
Phone Interworking  
Media Forking  
Routing  
Server Configuration  
Topology Hiding  
Signaling Manipulation  
URI Groups  
SIP Cluster  
Domain Policies  
TLS Management  
Device Specific Settings

**Interworking Profiles: SP-SI**

[Add](#) [Rename](#) [Clone](#) [Delete](#)

[Click here to add a description.](#)

**General** **Timers** **URI Manipulation** **Header Manipulation** **Advanced**

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

[Edit](#)

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
Global Parameters  
Global Profiles  
Domain DoS  
Fingerprint  
**Server Interworking**  
Phone Interworking  
Media Forking  
Routing  
Server Configuration  
Topology Hiding  
Signaling Manipulation  
URI Groups  
SIP Cluster  
Domain Policies  
TLS Management  
Device Specific Settings

**Interworking Profiles: SP-SI**

[Add](#) [Rename](#) [Clone](#) [Delete](#)

[Click here to add a description.](#)

**General** **Timers** **URI Manipulation** **Header Manipulation** **Advanced**

Record Routes	Both
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	No
NORTEL Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk Spurt	No
Reset SRTP Context on Session Refresh	No
Has Remote SBC	Yes
Route Response on Via Port	No
Cisco Extensions	No

[Edit](#)

#### 6.2.4.2 Server Interworking profile for Communication Manager

Profile **Communication Manager63** was defined to match the specification of Communication Manager. The **General** and **Advanced** settings are configured with the following parameters while the other settings for **Timers**, **URI Manipulation** and **Header Manipulation** are kept as default.

General settings:

- **Hold Support** = *None*.
- **18X Handling** = *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from Frontier to Communication Manager.
- **Refer Handling** = *No*. The Avaya SBCE will not handle REFER, it will keep the REFER messages unchanged from Frontier to Communication Manager.
- **T.38 Support** = *No*. Frontier does not support T.38 fax in the compliance testing.
- **Privacy Enabled** = *No*. The Avaya SBCE will not mask the “From” header with anonymous for an inbound call from Frontier. It depends on Frontier to enable/ disable privacy on an individual call basis.
- **DTMF Support** = *None*. The Avaya SBCE will send original DTMF method from Frontier to Communication Manager.

Advanced settings:

- **Record Routes** = *Both Sides*. The Avaya SBCE will send Record-Route header to both call and trunk servers.
- **Topology Hiding: Change Call-ID** = *No*. The Avaya SBCE will modify “Call-ID” header for the call toward Communication Manager.
- **Change Max Forwards** = *Yes*. The Avaya SBCE will adjust the original Max-Forwards value from Frontier to Communication Manager by reducing the intermediate hops involving in the call flow.
- **Has Remote SBC** = *Yes*. This setting allows the Avaya SBCE to always use the SDP received from Communication Manager for the media.

The screenshots below illustrate the Server Interworking profile **EN-SI**.

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
  Global Parameters  
  Global Profiles  
    Domain DoS  
    Fingerprint  
    **Server Interworking**  
    Phone Interworking  
    Media Forking  
    Routing  
    Server Configuration  
    Topology Hiding  
    Signaling Manipulation  
    URI Groups  
  SIP Cluster  
  Domain Policies  
  TLS Management  
  Device Specific Settings

**Interworking Profiles: EN-SI**

[Add](#) [Rename](#) [Clone](#) [Delete](#)

[Click here to add a description.](#)

**General** **Timers** **URI Manipulation** **Header Manipulation** **Advanced**

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

DTMF	
DTMF Support	None

[Edit](#)

**Session Border Controller for Enterprise** AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
  Global Parameters  
  Global Profiles  
    Domain DoS  
    Fingerprint  
    **Server Interworking**  
    Phone Interworking  
    Media Forking  
    Routing  
    Server Configuration  
    Topology Hiding  
    Signaling Manipulation  
    URI Groups  
  SIP Cluster  
  Domain Policies  
  TLS Management  
  Device Specific Settings

**Interworking Profiles: EN-SI**

[Add](#) [Rename](#) [Clone](#) [Delete](#)

[Click here to add a description.](#)

**General** **Timers** **URI Manipulation** **Header Manipulation** **Advanced**

Record Routes	Both
Topology Hiding: Change Call-ID	Yes
Call-Info NAT	No
Change Max Forwards	Yes
Include End Point IP for Context Lookup	No
OCS Extensions	No
AVAYA Extensions	Yes
NORTEL Extensions	No
Diversion Manipulation	No
Metaswitch Extensions	No
Reset on Talk Spurt	No
Reset SRTP Context on Session Refresh	No
Has Remote SBC	Yes
Route Response on Via Port	No
Cisco Extensions	No

[Edit](#)

### 6.2.5. Server Configuration

Server Configuration screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains.

To create a Server Configuration entry, select **Global Profiles → Server Configuration**. Click on the **Add** button.

In the compliance testing, two separate Server Configurations were created, server entry **SP-SC** for Frontier and server entry **EN-SC** for Communication Manager.

### 6.2.6. Server Configuration for Frontier

Server Configuration named **SP-SC** was created for Frontier. It will be discussed in detail below. **General** and **Advanced** tabs are provisioned for Frontier for outbound call from enterprise to PSTN. The **Heartbeat** tab is enabled to allow the Avaya SBCE to forward the OPTIONS heartbeat from Communication Manager to Frontier to query the status of the SIP trunk. The additional **DoS Whitelist** and **DoS Protection** tabs are displayed after **DoS Protection** is enabled under **Advanced** tab, the settings for these tabs are kept as default.

In the **General** tab, click on the **Edit** button then set **Server Type** for Frontier as **Trunk Server**. In the compliance testing, Frontier supported **UDP** and listened on port **5060**.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top header shows the title "Session Border Controller for Enterprise" and the Avaya logo. On the left is a navigation menu with categories like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, and Server Configuration (which is highlighted). The main content area is titled "Server Configuration: SP-SC" and includes an "Add" button. Below this is a list of server profiles: CM63, SM63, SP-SC (highlighted in red), and EN-SC. To the right of the list are buttons for "Rename", "Clone", and "Delete". The "General" tab is selected, showing a table with the following configuration:

Parameter	Value
Server Type	Trunk Server
IP Addresses / FQDNs	192.168.248.132
Supported Transports	UDP
UDP Port	5060

An "Edit" button is located at the bottom right of the configuration table.

Under **Advanced** tab, check on **Enable DoS Protection**. From the **Interworking Profile** drop down list, select **SP-SC** as defined in **Section 6.2.4**. For **Signaling Manipulation Script**, select **None**. This configuration applies the specific SIP profile to the Frontier traffic. The other settings are kept as default.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, and Routing. The main content area is titled "Server Configuration: SP-SC" and features a list of server profiles (CM63, SM63, SP-SC, EN-SC) on the left. The "Advanced" tab is selected, showing configuration options: "Enable DoS Protection" (checked), "Enable Grooming" (unchecked), "Interworking Profile" (SP-SI), "Signaling Manipulation Script" (None), and "UDP Connection Type" (SUBID). Buttons for "Add", "Rename", "Clone", "Delete", and "Edit" are visible.

### 6.2.7. Server Configuration for Communication Manager

Server Configuration named **EN-SC** created for Communication Manager is discussed in detail below. **General** and **Advanced** tabs are provisioned but no configuration is done for **Authentication** tab. The **Heartbeat** tab is kept as disabled as default to allow the Avaya SBCE to forward the OPTIONS heartbeat from Frontier to Communication Manager to query the status of the SIP trunk.

In the **General** tab, click on the **Edit** button then specify **Server Type** for Communication Manager as **Call Server**. In the compliance testing, the link between the Avaya SBCE and Communication Manager was **TCP** and listened on port **5060**.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, and Routing. The main content area is titled "Server Configuration: EN-SC" and features a list of server profiles (CM63, SM63, SP-SC, EN-SC) on the left. The "General" tab is selected, showing configuration options: "Server Type" (Call Server), "IP Addresses / FQDNs" (10.10.97.246), "Supported Transports" (TCP), and "TCP Port" (5060). Buttons for "Add", "Rename", "Clone", "Delete", and "Edit" are visible.



Under **Advanced** tab, click on the **Edit** button, from the **Interworking Profile** drop down list select **EN-SI** as defined in **Section 6.2.4** and from the **Signaling Manipulation Script** drop down list select **None**. The other settings are kept as default.

## 6.3. Domain Policies

Domain Policies configures various rule sets (policies) to control unified communications based upon criteria of communication sessions originating from or terminating at the enterprise. These criteria can be used to trigger policies which, in turn, activate various security features of the Avaya SBCE security device to aggregate, monitor, control and normalize call flow. There are default policies available for use, or a custom domain policy can be created.

### 6.3.1. Signaling Rules

Signaling Rules define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the Avaya SBCE, they are parsed and “pattern-matched” against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

To clone a Signaling Rule, navigate to **Domain Policies → Signaling Rules**. With the **default** rule chosen, click on the **Clone** button.

### 6.3.1.1 Signaling Rules for Frontier

In the compliance testing, created signaling rule **SP-SR** is discussed bellow. All the tabs are kept as default values except **Signaling QoS** tab.

In **Signaling QoS** tab, click on **Edit** button then check on **Enable** box. Then select **EF** value for **DSCP** option.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Domain Policies, Application Rules, Border Rules, Media Rules, Security Rules, and Signaling Rules (highlighted in red). The main content area is titled "Signaling Rules: SP-SR". It features a list of signaling rules on the left: default, No-Content-Ty..., EN-SR, and SP-SR (highlighted in red). The right pane shows the configuration for the selected rule. At the top, there are buttons for "Add", "Filter By Device...", "Rename", "Clone", and "Delete". Below this is a blue bar with the text "Click here to add a description.". The configuration is divided into tabs: General, Requests, Responses, Request Headers, Response Headers, and Signaling QoS (highlighted in red). The "Signaling QoS" tab is active, showing a table with the following data:

UCID	UCID
Signaling QoS	<input checked="" type="checkbox"/>
QoS Type	DSCP
DSCP	EF

At the bottom of the "Signaling QoS" tab is an "Edit" button.

### 6.3.1.2 Signaling Rules for Communication Manager

In the compliance testing, created signaling rule **EN-SR** is discussed bellow. All the tabs are kept as default values except **Signaling QoS** tab.

In **Signaling QoS** tab, click on **Edit** button then check on **Enable** box. Then select **EF** value for **DSCP** option.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain Policies, Application Rules, Border Rules, Media Rules, Security Rules, and Signaling Rules (highlighted in red). The main content area is titled "Signaling Rules: EN-SR". It features a list of signaling rules on the left: default, No-Content-Ty..., EN-SR (highlighted in red), and SP-SR. The right pane shows the configuration for the selected rule. At the top, there are buttons for "Add", "Filter By Device...", "Rename", "Clone", and "Delete". Below this is a blue bar with the text "Click here to add a description.". The configuration is divided into tabs: General, Requests, Responses, Request Headers, Response Headers, and Signaling QoS (highlighted in red). The "Signaling QoS" tab is active, showing a table with the following data:

UCID	UCID
Signaling QoS	<input checked="" type="checkbox"/>
QoS Type	DSCP
DSCP	EF

At the bottom of the "Signaling QoS" tab is an "Edit" button.

### 6.3.2. Endpoint Policy Groups

The rules created within the **Domain Policy** section are assigned to an **Endpoint Policy Group**. The **Endpoint Policy Group** is then applied to a **Server Flow** defined in the next section.

Endpoint Policy Groups were created for Frontier and Communication Manager.



To create a new policy group, navigate to **Domain Policies** → **Endpoint Policy Groups** and click on **Add**.

### 6.3.2.1 Endpoint Policy Group for Frontier

The following screen shows **SP-PG** created for Frontier:

- Set Application Rule to *default-trunk*.
- Set Border Rule to *default*.
- Set Media Rule to *default-low-med*.
- Set Security Rule to *default-high*
- Set Signaling Rule to *SP-SR* as created in **Section 6.3.1**.
- Set Time of Day Rule to *default*.



### 6.3.2.2 Endpoint Policy Group for Communication Manager

The following screen shows **EN-PG** created for Communication Manager:

- Set Application Rule to *default-trunk*.
- Set Border Rule to *default*.
- Set Media Rule to *default-low-med*.
- Set Security Rule to *default-low*.
- Set Signaling Rule to *EN-SR* as created in **Section 6.3.1**.
- Set Time of Day Rule to *default*.

# Session Border Controller for Enterprise

AVAYA

Dashboard

Administration

Backup/Restore

Domain Policies

- Application Rules
- Border Rules
- Media Rules
- Security Rules
- Signaling Rules
- Time of Day Rules
- End Point Policy Groups**

Policy Groups: EN-PG

Add

Filter By Device...

Rename

Clone

Delete

Showing page 1 of 2

> >>

Policy Groups

EN-PG

SP-PG

Click here to add a description.

Hover over a row to see its description.

Policy Group

Summary

Add

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	default-trunk	default	default-low-med	default-high	default	default	Edit Clone

## 6.4. Device Specific Settings

Device Specific Settings allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

### 6.4.1. Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information was defined such as device IP address(es), public IP address(es), netmask, gateway, etc. to interface the device to the network. This information populates the **Network Management** tab, which can be edited as needed to optimize device performance and network efficiency.

Navigate to **Device Specific Settings** → **Network Management** and under the **Network Configuration** tab verify the IP addresses assigned to the interfaces. The following screen shows the private interface is assigned to **A1** and the public interface is assigned to **B1**.

IP Address	Public IP	Gateway	Interface	
10.10.98.13		10.10.98.1	A1	Delete
10.10.98.111		10.10.98.97	B1	Delete

Enable the interfaces used to connect to the inside and outside networks on the **Interface Configuration** tab. The following screen shows interface **A1** and **B1** are **Enabled**. To enable an interface, click its **Toggle** button.

Name	Administrative Status	
A1	Enabled	Toggle
A2	Disabled	Toggle
B1	Enabled	Toggle
B2	Disabled	Toggle

### 6.4.2. Media Interface

The Media Interface screen is where the media ports are defined. The Avaya SBCE will open a connection for RTP on the defined ports.

To create a new Media Interface, navigate to **Device Specific Settings → Media Interface** and click **Add**.

Separate Media Interfaces were created for both inside and outside interfaces. The following screen shows the Media Interfaces created in the compliance testing.

**Note:** After the media interfaces are created, an application restart is necessary before the changes will take effect.

The screenshot shows the 'Media Interface: SBCE62' configuration page. On the left is a navigation menu with options like Dashboard, Administration, Backup/Restore, System Management, and Device Specific Settings. The 'Media Interface' option is highlighted. The main content area has a sub-header 'Media Interface' and a warning message: 'Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.' Below this is a table with two columns: 'Name' and 'Media IP', and a 'Port Range' column. The table lists two interfaces: 'InsideMedia1' with IP '10.10.98.13' and 'OutsideMedia1' with IP '10.10.98.111'. Both have a port range of '35000 - 40000'. Each interface has 'Edit' and 'Delete' links. An 'Add' button is located at the top right of the table.

Name	Media IP	Port Range	Edit	Delete
InsideMedia1	10.10.98.13	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>
OutsideMedia1	10.10.98.111	35000 - 40000	<a href="#">Edit</a>	<a href="#">Delete</a>

### 6.4.3. Signaling Interface

The Signaling Interface screen is where the SIP signaling port is defined. The Avaya SBCE will listen for SIP requests on the defined port.

To create a new Signaling Interface, navigate to **Device Specific → Settings → Signaling Interface** and click **Add**.

Separate Signaling Interfaces were created for both inside and outside interfaces. The following screen shows the Signaling Interfaces were created in the compliance testing with UDP/5060 for the outside interface to Frontier and TCP/5060 for the inside interface to Communication Manager.

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
TLS Management
Device Specific Settings
Network Management
Media Interface
**Signaling Interface**

Signaling Interface: SBCE62

Devices

SBCE62

Signaling Interface

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile	
InsideTCP	10.10.98.13	5060		---	None	Edit Delete
OutsideUDP	10.10.98.111	---	5060	---	None	Edit Delete

Add

#### 6.4.4. End Point Flows - Server Flow

When a packet is received by Avaya SBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

In the compliance testing, separate Server Flows were created for Frontier and Communication Manager. To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add** (not shown). In the new window that appears, enter the following values. The other fields are kept default.

- **Flow Name:** Enter a descriptive name.
- **Server Configuration:** Select a Server Configuration created in **Section 6.2.5** to assign to the Flow.
- **URI Group:** Select the URI Group created in **Section 6.2.1** to assign to the Flow.  
**Note:** URI Group can be set to “\*” to match all calls.
- **Received Interface:** Select the Signaling Interface created in **Section 6.4.3** that the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface:** Select the Signaling Interface created in **Section 6.4.3** used to communicate with the Server Configuration.
- **Media Interface:** Select the Media Interface created in **Section 6.4.2** used to communicate with the Server Configuration.
- **End Point Policy Group:** Select the End Point Policy Group created in **Section 6.3.2** to assign to the Server Configuration.
- **Routing Profile:** Select the Routing Profile created in **Section 6.2.2** that the Server Configuration will use to route SIP messages to.
- **Topology Hiding Profile:** Select the Topology-Hiding profile created in **Section 6.2.3** to apply to the Server Configuration.
- Click **Finish**.

The following screen shows the Server Flow **SP-SF** configured for Frontier.

Edit Flow: SP-SF	
Flow Name	SP-SF
Server Configuration	SP-SC
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	InsideUDP
Signaling Interface	OutsideUDP
Media Interface	InsideMedia1
End Point Policy Group	SP-PG
Routing Profile	SP-to-EN
Topology Hiding Profile	EN-to-SP
File Transfer Profile	None
<div>Finish</div>	

Similarly, the following screen shows the Server Flow **EN-SF** configured for Communication Manager.



Edit Flow: EN-SF	
Flow Name	EN-SF
Server Configuration	EN-SC
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	OutsideUDP
Signaling Interface	InsideTCP
Media Interface	InsideMedia1
End Point Policy Group	EN-PG
Routing Profile	EN-to-SP
Topology Hiding Profile	SP-to-EN
File Transfer Profile	None
Finish	

## 7. Frontier SIP Trunking Service Configuration

Frontier is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Avaya SBCE at the enterprise side. Frontier will provide the customer with the necessary information to configure the SIP connection from the enterprise to Frontier. The information provided by Frontier includes:

- IP address and port number used for signaling through security devices (if any).
- IP address and port number used for media through security devices (if any).
- Frontier SIP domain. In the compliance testing, Frontier preferred to use IP address as an URI-Host.
- CPE SIP domain. In the compliance testing, Frontier preferred to use IP address of the Avaya SBCE as an URI-Host.
- Supported codecs.
- DID numbers.

The sample configuration between Frontier and the enterprise for the compliance testing is a static configuration. There is no registration on the SIP trunk implemented on either Frontier or enterprise side.

## 8. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands.

## 8.1. Verification Steps

- Verify that endpoints at the enterprise site can place call to PSTN and that the call remains active for more than 35 seconds. This time period is included to satisfy SIP protocol timers.
- Verify that endpoints at the enterprise site can receive call from PSTN and that the call can remain active for more than 35 seconds. This time period is included satisfy SIP protocol timers.
- Verify that the user on PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

## 8.2. Protocol Traces

The following SIP headers are inspected using Wireshark trace analysis:

- Request-URI: verify the called party number and SIP domain.
- From: verify the calling party name and number.
- To: verify the called party name and number.
- P-Asserted-Identity: verify the calling party name and number.
- Privacy: verify the value “user” and/or “id” presents the private call scenario.

The following attributes in SIP message body are inspected using Wireshark trace analysis:

- Connection Information (c line): verify IP address of near end and far end endpoints.
- Time Description (t line): verify session timeout value of near end and far end endpoints.
- Media Description (m line): verify audio port, codec, DTMF event description.
- Media Attribute (a line): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

## 8.3. Troubleshooting:

### 8.3.1. The Avaya SBCE

Use a network sniffing tool (e.g., Wireshark) to monitor the SIP signaling messages between Frontier and the Avaya SBCE.

Following is an example inbound call from Frontier to the enterprise.

- Inbound INVITE request from Frontier:

```
INVITE sip:5856664804@135.10.98.111:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 173.84.248.132:5060;branch=z9hG4bK0h9hpul0eg8hhr0ve550.1
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info, as-
feature-event
Max-Forwards: 69
Call-ID: 4961CD0F@10.160.139.4
From:
<sip:01101116139675204@10.160.139.4:5060>;tag=10.160.139.4+1+34d40a+35430454;isup-
oli=00
To: <sip:5856664804@135.10.98.111>
CSeq: 1067064121 INVITE
Expires: 180
Organization: MetaSwitch
Supported: resource-priority, 100rel
```



```
Content-Length: 182
Content-Type: application/sdp
Contact: <sip:01101116139675204@173.84.248.132:5060;transport=udp>;isup-oli=00
P-Asserted-Identity: <sip:01101116139675204@10.160.139.4:5060>

v=0
o=- 4008647052 4008647052 IN IP4 173.84.248.132
s=-
c=IN IP4 173.84.248.132
t=0 0
m=audio 64024 RTP/AVP 18 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

- 200OK/SDP response by the enterprise:

```
SIP/2.0 200 OK
From:
<sip:01101116139675204@10.160.139.4:5060>;tag=10.160.139.4+1+34d40a+35430454;isup-
oli=00
To: <sip:5856664804@135.10.98.111>;tag=026a588b2ee4118553d958fc00
CSeq: 1067064121 INVITE
Call-ID: 4961CD0F@10.160.139.4
Contact: "H323 1131" <sip:5856664804@135.10.98.111:5060;transport=udp>
Record-Route: <sip:135.10.98.111:5060;ipcs-line=7429;lr;transport=udp>
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH,UPDATE
Supported: 100rel,join,replaces,sdp-anat,timer
Via: SIP/2.0/UDP 173.84.248.132:5060;branch=z9hG4bK0h9hpu10eg8hhr0ve550.1
Accept-Language: en
Server: Avaya CM/R016x.03.0.124.0
P-Asserted-Identity: "H323 1131" <sip:5856664804@135.10.98.111>
Session-Expires: 1200;refresher=uas
Content-Type: application/sdp
Content-Length: 173

v=0
o=- 1408481982 2 IN IP4 135.10.98.13
s=-
c=IN IP4 135.10.98.13
b=AS:64
t=0 0
m=audio 35010 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
```

Following is an example outbound call from the enterprise to Frontier.

- Outbound INVITE request from the enterprise:

```
INVITE sip:6139675204@173.84.248.132 SIP/2.0
From: "H323 1131" <sip:5856664804@135.10.98.111>;tag=04e1077c2ee41528553d958fc00
To: <sip:6139675204@173.84.248.132>
CSeq: 1 INVITE
Call-ID: d8b41d741b67d121065b2d565fb64634
Contact: "H323 1131" <sip:5856664804@135.10.98.111:5060>
Record-Route: <sip:135.10.98.111:5060;ipcs-line=7473;lr;transport=udp>
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH,UPDATE
Supported: 100rel,join,replaces,sdp-anat,timer
User-Agent: Avaya CM/R016x.03.0.124.0
Max-Forwards: 70
Via: SIP/2.0/UDP 135.10.98.111:5060;branch=z9hG4bK-s1632-000251807985-1--s1632-
Accept-Language: en
Alert-Info: <cid:internal@bvwlab.com>;avaya-cm-alert-type=internal
P-Asserted-Identity: "H323 1131" <sip:5856664804@135.10.98.111>
Session-Expires: 1200;refresher=uac
Min-SE: 1200
Content-Type: application/sdp
Content-Length: 255

v=0
o=- 1408482380 1 IN IP4 135.10.98.13
s=-
c=IN IP4 135.10.98.13
b=AS:64
t=0 0
a=avf:avc=n prio=n
a=csup:avf-v0
m=audio 35012 RTP/AVP 0 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
```

- 200OK/SDP response by Frontier:

```
SIP/2.0 200 OK
From: "H323 1131" <sip:5856664804@135.10.98.111>;tag=04e1077c2ee41528553d958fc00
To: <sip:6139675204@173.84.248.132>;tag=10.160.139.4+1+67981f+40cd4e98
CSeq: 1 INVITE
Call-ID: d8b41d741b67d121065b2d565fb64634
Via: SIP/2.0/UDP 135.10.98.111:5060;branch=z9hG4bK-s1632-000251807985-1--s1632-
Record-Route: <sip:135.10.98.111:5060;ipcs-line=7473;lr;transport=udp>
Server: DC-SIP/2.0
Organization: MetaSwitch
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info, as-
feature-event
Supported: resource-priority, 100rel
Allow: INVITE, ACK, CANCEL, BYE, REGISTER, OPTIONS, PRACK, UPDATE, SUBSCRIBE, NOTIFY,
REFER, INFO, PUBLISH
Accept-Encoding: identity
Accept: application/sdp, application/simple-message-summary, message/sipfrag,
application/isup, application/x-simple-call-service-info, multipart/mixed,
application/broadsoft, application/vq-rtpcpxr, application/media_control+xml,
application/dtmf-relay, text/plain, application/x-as-feature-event+xml
Contact: <sip:6139675204@173.84.248.132:5060;transport=udp>
Content-Length: 179
Content-Type: application/sdp

v=0
o=- 4009045995 4009045995 IN IP4 173.84.248.132
s=-
c=IN IP4 173.84.248.132
t=0 0
m=audio 63762 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

### 8.3.2. Avaya Aura® Communication Manager

- **list trace station** <extension number>. Traces call to and from a specific station.
- **list trace tac** <trunk access code number>. Trace call over a specific trunk group.
- **status station** <extension number>. Displays signaling and media information for an active call on a specific station.
- **status trunk** <trunk group number>. Displays trunk group information.
- **status trunk** <trunk group number/channel number>. Displays signaling and media information for an active trunk channel.

## 9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 to Frontier SIP Trunking Service. Frontier SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large the enterprises. Frontier SIP Trunking Service provides a flexible, cost-saving alternative to traditional analog and ISDN-PRI trunks.

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The Frontier SIP Trunking Service is considered **compliant** with Avaya Aura® Communication Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2.

## 10. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.2.2, December 2012.
- [2] *Administering Avaya Aura® System Platform*, Release 6.2.2, December 2012.
- [3] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509, Release 6.3, Issue 8, May 2013.
- [4] *Programming Call Vectoring Features in Avaya Aura® Call Center Elite*, Release 6.3, Issue 1, May 2013.
- [5] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, Release 3.1, November 2009, Document Number 16-300698.
- [6] *Avaya one-X® Deskphone SIP for 9600 Series IP Telephones Administrator Guide*, Release 2.6, June 2010, Document Number 16-601944.
- [7] *Administering Avaya one-X® Communicator*, April 2011.
- [8] *Using Avaya one-X® Communicator*, April 2011.
- [9] *Avaya SBCE Install Guide* (102-5224-400v1.01)
- [10] *Avaya SBCE Administration Guide* (010-5423-400v106)
- [11] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [12] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method*, <http://www.ietf.org/>
- [13] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>
- [14] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, March 2013
- [15] *Avaya Session Border Controller for Enterprise Overview and Specification*, Issue 2, March 2013

Product documentation for Frontier SIP Trunking Service is available from Frontier.

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