



Avaya Solution & Interoperability Test Lab

Application Notes for CenturyLink SIP Trunking Service (Sonus Platform) 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 CenturyLink 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.

CenturyLink 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 the CenturyLink (Sonus) SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager 6.3 configured as an Evolution Server, Avaya Session Border Controller for Enterprise (Avaya SBCE) 6.2 and various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with CenturyLink 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.

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

2.1. Interoperability Compliance Testing

To verify CenturyLink 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 (1XC) soft phone. Both the 1XC Computer Mode (where 1XC is used for call control as well as audio path) and the 1XC Telecommuter Mode (where 1XC 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.
- Proper codec negotiation with G.729 and G.711MU codecs.
- Early Media transmissions using G.729 and G.711MU codecs.
- 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 including the following:

- Inbound toll-free.
- Emergency calls (911 in US).
- G711Alaw and G729AB.

Items that are not support and therefore not tested including in the following:

- SIP REFER redirection is not fully support.

2.2. Test Results

Interoperability testing of CenturyLink 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.

- **OPTIONS** - CenturyLink will send OPTIONS if customer request/order it. Sonus Network Border Switch (NBS) will not send OPTIONS until 4 calls time out (address is not reachable). Then it will stop after 4 OPTIONS are successful. Both intervals are configurable.
- **Transfer Inbound Calls To PSTN Using REFER** – Some scenarios of transferring inbound calls to the PSTN failed. The failures occurred with the following transfer scenarios:
 - Blind transfer of inbound call to the PSTN (termed as "Attended/Consultative Transfer with Early Completion" by CenturyLink) where the enterprise transfer initiator completes the transfer before the outbound call to the PSTN transfer destination is answered.
 - Consultative transfer of inbound call to the PSTN by the enterprise SIP hard phone where the enterprise transfer initiator completes the call transfer after the outbound call to the PSTN destination is answered.
 - Blind or consultative transfer of inbound call to the PSTN by the H.323 one-X® Communicator softphone.

In all the above transfer failures, CenturyLink sent a NOTIFY to the enterprise, after accepting the REFER message, indicating "481 Call Leg/Transaction Does Not Exist".

The above problem was reported to CenturyLink for investigation/resolution. CenturyLink advised that Attended/Consultative Transfer with Early Completion (1st transfer scenario above) is not yet supported on the SIP Trunking service (Sonus Platform). Since transfer failures using REFER occurred with other transfer scenarios (2nd and 3rd scenarios above), it is recommended that use of REFER be turned off on Communication Manager (see **Section 5.7**) so that INVITE instead of REFER is used for call transfers until this problem is properly addressed (CenturyLink has opened a ticket with Sonus on this issue). In the compliance test, transfer of inbound calls to PSTN using INVITE was successfully verified.

2.3. Support

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

For technical support on CenturyLink SIP Trunking Service, please contact CenturyLink technical support at:

- Website: <http://www.CenturyLink.com>

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution connected to the CenturyLink 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 Avaya Aura® Communication Manager
- Avaya G450 Media Gateway
- Avaya S8800 Server running Avaya Aura® 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 CenturyLink 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 CenturyLink across the public network is UDP. The transport protocol between the Avaya SBCE and Communication Manager is TCP.

In the compliance testing, the Avaya Customer-Premises Equipment (CPE) environment was configured with SIP domain “bvwddev7.com” for the enterprise. The Avaya SBCE is used to adapt the enterprise SIP domain to the IP address based URI-Host known to CenturyLink.

Figure 1 below illustrates the network diagram for the enterprise. All voice application elements are connected to internal trusted LAN.

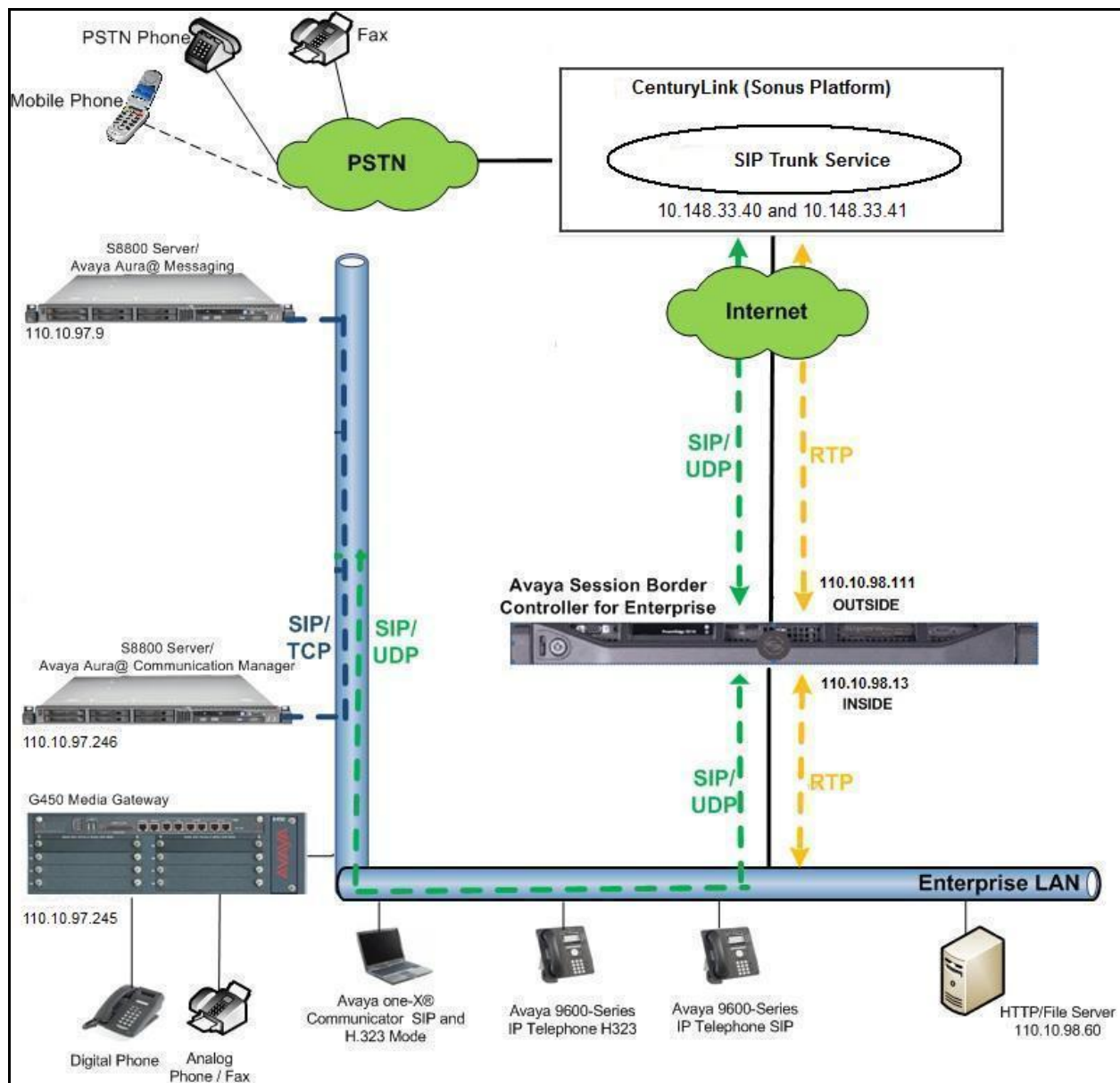


Figure 1: Avaya IP Telephony Network connecting to CenturyLink SIP Trunking Service

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 CM/ R016x.03.0.124.0 with Service Pack 1 (03.0.124.0-20850))
Avaya G450 Media Gateway	28.22.0
Avaya Aura® Messaging running on an Avaya S8800 Server	6.1-11.0
Avaya Session Border Controller for Enterprise	6.2.0 Q48
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	n/a
Avaya 6210 Analog Telephone	n/a
CenturyLink SIP Trunking Service Components	
Component	Release
CenturyLink iQ® SIP Trunk	7.3.7
Sonus NBS	7.3.7

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 Manager.

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager for CenturyLink 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.

display system-parameters customer-options		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	2
Maximum Administered SIP Trunks: 24000		138
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	110.10.98.13	
DevAAM	110.33.10.9	
default	0.0.0.0	
procr	110.10.97.246	
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. CenturyLink supports G.729A and G.711MU. To use these codecs, enter **G.729A** and **G.711MU** 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		Page 1 of 2
IP Codec Set		
Codec Set: 1		
Audio Codec	Silence Suppression	Frames Per Pkt
Packet Size(ms)		
1: G.711MU	n	2
2: G.729A	n	2
3:		

On **Page 2**, set the **Fax Mode** to **t.38-standard** faxing which is supported by CenturyLink.

change ip-codec-set 1		Page 2 of 2
IP Codec Set		
Allow Direct-IP Multimedia? n		
	Mode	Redundancy
FAX	t.38-standard	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 *bwvdev7.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: bvwdev7.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		t					
dst codec direct WAN-BW-limits Video Intervening										Dyn		A		G		c			
rgn set WAN Units Total Norm Prio Shr Regions										CAC		R		L		e			
1 1														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.

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

change media-gateway 1		Page 1 of 2
MEDIA GATEWAY 1		
Type: g450		
Name: G450		
Serial No: 12TG18000244		
Encrypt Link? y	Enable CF? n	
Network Region: 1	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 Session Border Controller (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 *n*.
- 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 **bvwddev7.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.

add signaling-group 50		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	
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: bvwddev7.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: *004
  Direction: two-way                                Outgoing Display? y
  Dial Access? n                                     Night Service:
  Queue Length: 0
  Service Type: public-ntwrk                         Auth Code? n
                                                    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 addition of the + sign impacted interoperability with the service provider. Thus, 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 *n* disables use of the SIP REFER message to transfer an inbound call back to the PSTN since CenturyLink does not fully support SIP REFER.

- 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 CenturyLink.

add trunk-group 50	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
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 5 numbers were mapped to the 5 enterprise extensions 1130, 1131, 1132, 1133 and 1134. These same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these 5 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	3006157104	10	Total Administered: 11 Maximum Entries: 540
4	1131	50	3006157105	10	
4	1132	50	3006157106	10	
4	1133	50	3006157107	10	
4	1134	50	3006157108	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	3006157104	10	Total Administered: 5 Maximum Entries: 240
4	1131	50	3006157105	10	
4	1132	50	3006157106	10	Note: If an entry applies to a SIP connection to Avaya Aura(R) Session Manager, the resulting number must be a complete E.164 number.
4	1133	50	3006157107	10	
4	1134	50	3006157108	10	

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 CenturyLink 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	3006157104	10	1130			
public-ntwrk	10	3006157105	10	1131			
public-ntwrk	10	3006157106	10	1132			
public-ntwrk	10	3006157107	10	1133			
public-ntwrk	10	3006157108	10	1134			

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: *111												
Answer Back Access Code:												
Attendant Access Code:												
Auto Alternate Routing (AAR) Access Code: *100												
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	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	

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											
								Dgts											
1:	50	0						Intw											
2:								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				unk-unk		none					
...																			

5.11. Saving Communication Manager 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 CenturyLink SIP Trunking service.

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

6.1. Avaya Session Border Controller for Enterprise Login

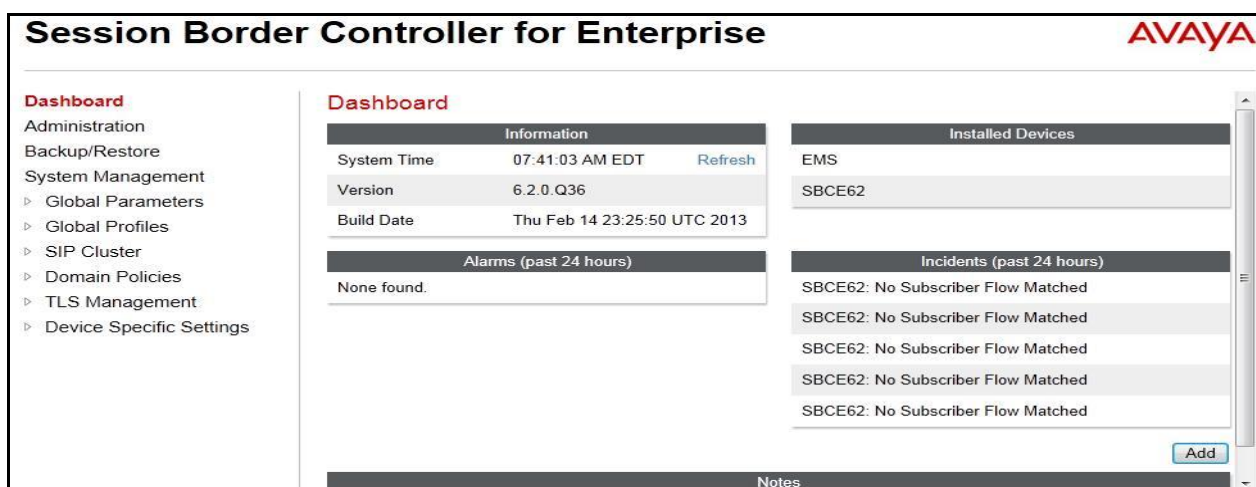
Use a Web browser to access the Avaya SBCE web interface, enter `https://<ip-addr>/ucsec` in the address field of the web browser (not shown), where `<ip-addr>` is the management LAN IP address of Avaya SBCE.

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



The login page features the Avaya logo on the left. The main heading is "Session Border Controller for Enterprise". 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." Another paragraph states: "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." A final line reads: "All users must comply with all corporate instructions regarding the protection of information assets." At the bottom right, it says "© 2011 - 2013 Avaya Inc. All rights reserved."

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



The main page has a header with "Session Border Controller for Enterprise" and the Avaya logo. On the left is a navigation menu with "Dashboard" (highlighted) and "Administration" (containing sub-items: Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies, TLS Management, and Device Specific Settings). The main content area is titled "Dashboard" and contains three sections: "Information" (a table with System Time, Version, and Build Date), "Alarms (past 24 hours)" (showing "None found."), and "Installed Devices" (showing EMS and SBCE62). Below these is a section for "Incidents (past 24 hours)" listing five incidents, all with the message "SBCE62: No Subscriber Flow Matched". At the bottom right of the incidents list is an "Add" button. A "Notes" section is at the very bottom.

Information	
System Time	07:41:03 AM EDT Refresh
Version	6.2.0 Q36
Build Date	Thu Feb 14 23:25:50 UTC 2013

Alarms (past 24 hours)	
None found.	

Installed Devices	
EMS	
SBCE62	

Incidents (past 24 hours)	
SBCE62:	No Subscriber Flow Matched
SBCE62:	No Subscriber Flow Matched
SBCE62:	No Subscriber Flow Matched
SBCE62:	No Subscriber Flow Matched
SBCE62:	No Subscriber Flow Matched

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.

To add an URI Group, select **Global Profiles → URI Groups**. Click on the **Add** button (not shown).

In this compliance testing, an URI group was used due to multiple service providers sharing the same Avaya SBCE. In the real customer deployment, this won't be needed at all.

URI Group named **WS** was added with URI type Regular Expression (not shown) and consists of:

- **.*bvwddev7\.com**: Enterprise domain used for calls across the enterprise networks. This domain matches the domain configured for Communication Manager (see **Section 5.5** and **Section 5.6**).
- **.*nonymous\.invalid**: enterprise domain, defined to support private call.
- **.*110\.10\.98\.111, .*10\.148\.33\.40 and .*10\.148\.33\.41**: IP address based URI-Host, used for public calls to/from the service provider. The Avaya SBCE public IP address, 110.10.98.111, is set as URI-Host of the "From", "PAI" and "Diversion" headers while the public IP address of CenturyLink, 10.148.33.41, is set as URI-Host of "Request-URI" and "To" headers.
- **.*110\.10\.97\.246 and .*110\.10\.98\.13**: IP address based URI-Host, defined to support routing for the outbound OPTIONS heartbeat originated by Communication Manager to the Avaya SBCE. The OPTIONS will be forwarded by the Avaya SBCE to the service provider for response to confirm the status of the SIP trunk.

This URI-Group is used to match the "From" and "To" headers in a SIP call dialog received from CenturyLink. If there is a match, the Avaya SBCE will apply the appropriate Routing Profile and Server Flow to route the inbound or outbound calls to the right destination. The Routing Profile and Server Flow are appropriately discussed in **Section 6.2.2** and **Section 6.4.4**.

The screenshot below illustrates the URI listing for the URI Group.

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

URI Groups: SP

Add Rename Delete

Click here to add a description.

URI Group

Add

URI Listing	
*10\148\33\36	Edit Delete
*10\148\33\37	Edit Delete
*10\148\33\40	Edit Delete
*10\148\33\41	Edit Delete
*110\10\97\246	Edit Delete
*110\10\98\111	Edit Delete
*110\10\98\13	Edit Delete
*bvwddev7\com	Edit Delete
*anonymous\invalid	Edit Delete

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 (not shown).

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 CenturyLink.

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

Routing Profile for CenturyLink

The screenshot below illustrates the routing profile from Avaya SBCE to the CenturyLink network, **Global Profiles → Routing: EN-to-SP**. As shown in **Figure 1**, the CenturyLink SIP trunk is connected with transportation protocol UDP (not shown). If there is a match in the “To” header 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 CenturyLink SIP trunk on port 5060.

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 (selected), Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, and Routing (highlighted in red). The main content area is titled "Routing Profiles: EN-to-SP" and includes an "Add" button, "Rename", "Clone", and "Delete" buttons. Below this is a blue bar with the text "Click here to add a description." The "Routing Profile" section contains a table with the following data:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2
1	SP	10.148.33.41:5060	---

There are "View" and "Edit" links next to the table row. An "Add" button is also present in the top right of the table area.

Routing Profile for Communication Manager

The Routing Profile for CenturyLink 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 (not shown).

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 (selected), Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, and Routing (highlighted in red). The main content area is titled "Routing Profiles: SP-to-EN" and includes an "Add" button, "Rename", "Clone", and "Delete" buttons. Below this is a blue bar with the text "Click here to add a description." The "Routing Profile" section contains a table with the following data:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2
1	SP	110.10.97.246:5060	---

There are "View" and "Edit" links next to the table row. An "Add" button is also present in the top right of the table area.

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 (not shown).

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

Topology Hiding Profile for CenturyLink

Profile **EN-to-SP** was defined to mask the enterprise SIP domain bwvdev7.com in “Request-URI” and “To” headers to IP **10.148.33.41** (the IP address CenturyLink uses as URI-Host portion for “Request-URI” and “To” headers to meet the SIP specification requirement of CenturyLink); mask the enterprise SIP domain bwvdev7.com in the “From” and “PAI” headers to IP **110.10.98.111** (the Avaya SBCE public IP address); and replace Record-Route, Via headers and SDP (originated from Communication Manager) by external IP address known to CenturyLink. 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 contains a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, and Server Configuration. The main content area is titled "Topology Hiding Profiles: EN-to-SP" and includes an "Add" button, a "Rename" button, a "Clone" button, and a "Delete" button. Below this, there is a "Topology Hiding" tab and a table with the following data:

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	10.148.33.41
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Overwrite	110.10.98.111
SDP	IP/Domain	Auto	---
To	IP/Domain	Overwrite	10.148.33.41
Via	IP/Domain	Auto	---

Topology Hiding Profile for Communication Manager

Profile **SP-to-EN** was also created to mask CenturyLink URI-Host in “Request-URI”, “From”, “To” headers to the enterprise domain bwvdev7.com, replace Record-Route, Via headers and SDP added by CenturyLink 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 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 (selected), Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, Server Configuration, and Topology Hiding (highlighted in red). The main content area is titled "Topology Hiding Profiles: SP-to-EN" and includes an "Add" button, "Rename", "Clone", and "Delete" buttons. Below this is a table with the following data:

Header	Criteria	Replace Action	Overwrite Value
Request-Line	IP/Domain	Overwrite	bwvdev7.com
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Overwrite	bwvdev7.com
SDP	IP/Domain	Auto	---
To	IP/Domain	Overwrite	bwvdev7.com
Via	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 (not shown).

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

Server Interworking profile for CenturyLink

Profile **SP_SI** was defined to match the specification of CenturyLink. 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 CenturyLink.
- **18X Handling** = *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from Communication Manager to CenturyLink.
- **Refer Handling** = *No*. The Avaya SBCE will not handle REFER. It will keep the REFER message unchanged from Communication Manager to CenturyLink.
- **T.38 Support** = *Yes*. CenturyLink 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 CenturyLink. 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 CenturyLink.

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 CenturyLink.
- **Change Max Forwards** = *Yes*. The Avaya SBCE will adjust the original Max-Forwards value from Communication Manager to CenturyLink by reducing the intermediate hops involving in the call flow.
- **Has Remote SBC** = *Yes*. CenturyLink 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 CenturyLink for the media.

The screenshots below illustrate the Server Interworking profile **SP_SI**.

Session Border Controller for Enterprise

- 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

Interworking Profiles

SP-SI

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
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
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

Session Border Controller for Enterprise

- 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

Interworking Profiles

SP-SI

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

Server Interworking profile for Communication Manager

Profile **EN-SI** 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** = *RFC3264*. Communication Manager supports hold/ resume as per RFC3264.
- **18X Handling** = *None*. The Avaya SBCE will not handle 18X, it will keep the 18X messages unchanged from CenturyLink to Communication Manager.
- **Refer Handling** = *No*. The Avaya SBCE will not handle REFER, it will keep the REFER messages unchanged from CenturyLink to Communication Manager.
- **T.38 Support** = *Yes*. CenturyLink 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 CenturyLink. It depends on CenturyLink to enable/disable privacy on an individual call basis.
- **DTMF Support** = *None*. The Avaya SBCE will send original DTMF method from CenturyLink 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 CenturyLink 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

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

Interworking Profiles

EN-SI

SP-SI

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
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
T.38 Support	Yes
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
--------------	------

Session Border Controller for Enterprise

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

Interworking Profiles

cs2100

avaya-ru

OCS-Edge-S...

cisco-ccm

cups

OCS-FrontEn...

SM63

Rogers

SM62

XO_Communi...

EN-SI

SP-SI

SM

BellCanada

CS1K76

Frontier

Rename Clone Delete

Click here to add a description.

General Timers URI Manipulation Header Manipulation Advanced

General

Record Routes	Both
Topology Hiding: Change Call-ID	No
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

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 (not shown).

In the compliance testing, two separate Server Configurations were created, server entry **WS_SC** for CenturyLink and server entry **CM52_SC** for Communication Manager.

Server Configuration for CenturyLink

Server Configuration named **SP-SC** was created for CenturyLink. It will be discussed in detail below. **General** and **Advanced** tabs are provisioned but no configuration is done for **Authentication** tab as CenturyLink does not implement authentication on the SIP trunk. The **Heartbeat** tab is kept as disabled as default to allow the Avaya SBCE to forward the OPTIONS heartbeat from Communication Manager to CenturyLink 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 to set **Server Type** for CenturyLink to **Trunk Server** **Server** (not shown). In the compliance testing, CenturyLink 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 options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, and Server Configuration (highlighted in red). The main content area is titled "Server Configuration: SP-SC" and includes an "Add" button, a list of "Server Profiles" with "SP-SC" selected, and action buttons "Rename", "Clone", and "Delete". Below this is a tabbed interface with tabs for "General", "Authentication", "Heartbeat", "Advanced", "DoS Whitelist", and "DoS Protection". The "General" tab is active, showing a table with the following configuration:

Server Type	Trunk Server
IP Addresses / FQDNs	10.148.33.40, 10.148.33.41
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** (not shown). From the **Interworking Profile** drop down list, select **SP-SI** as defined in Section 6.2.4. For **Signaling Manipulation Script**, select **None**. This configuration applies the specific SIP profile to the CenturyLink traffic. The other settings are kept as default.

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

Server Configuration: SP-SC

Add

Rename Clone Delete

Server Profiles

SP-SC

General
Authentication
Heartbeat
Advanced
DoS Whitelist
DoS Protection

Enable DoS Protection ☒
Enable Grooming ☐
Interworking Profile SP-SI
Signaling Manipulation Script None
UDP Connection Type SUBID

Edit

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 CenturyLink to Communication Manager to query the status of the SIP trunk.

In the **General** tab, click on the **Edit** button to specify **Server Type** for Communication Manager as **Call Server** (not shown). In the compliance testing, the link between the Avaya SBCE and Communication Manager was TCP and listened on port 5060.

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

Server Configuration: EN-SC

Add

Rename Clone Delete

Server Profiles

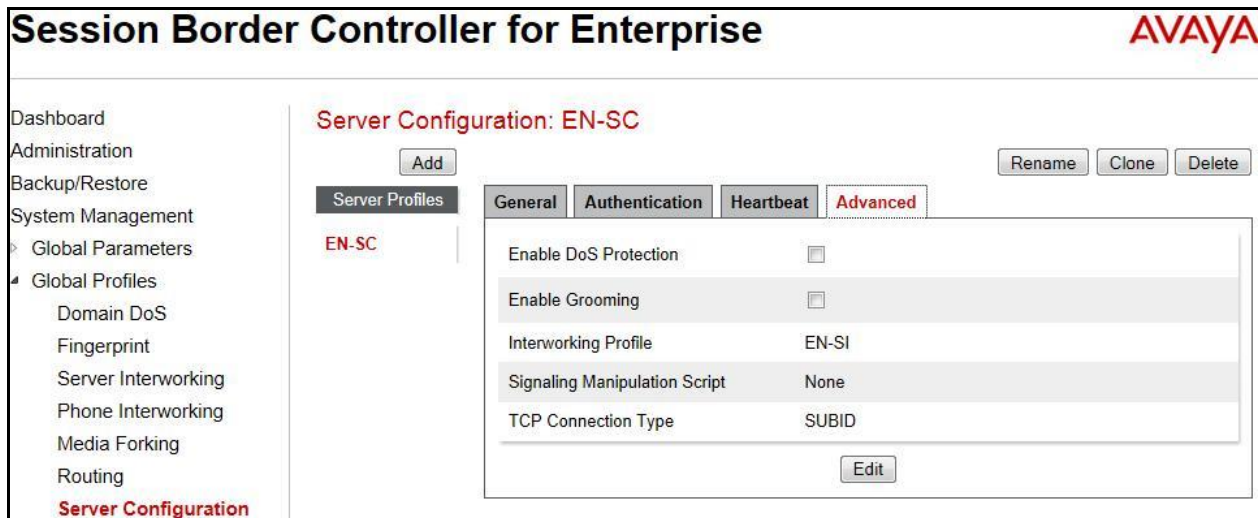
EN-SC

General
Authentication
Heartbeat
Advanced

Server Type Call Server
IP Addresses / FQDNs 110.10.97.246
Supported Transports TCP
TCP Port 5060

Edit

Under **Advanced** tab, click on the **Edit** button (not shown), 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. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, it is possible to configure the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

An Application Rule is created to set the number of concurrent voice traffic. The sample configuration is cloned and modified to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**.

To clone an Application Rule, navigate to **Domain Policies** → **Application Rules**. With the **default** rule chosen, click on the **Clone** button (not shown).

Enter a rule with a descriptive name **SP-AR** and click **Finish** (not shown).

Click the **Edit** button (not shown) to modify the rule. Set the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** for the **Voice** application to a value high enough for the amount of traffic the network is able to process. The following screen shows the modified **Application Rule** with the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** set to **1000**. In the compliance testing, Communication Manager is programmed to control the concurrent sessions by setting the number of members in the trunk group (**Section 5.7**) to the allotted number. Therefore, the values in the **Application Rule** named **SP-AR** are set high enough to be considered non-blocking.

Session Border Controller for Enterprise

AVAYA

Dashboard
Administration
Backup/Restore
System Management
Global Parameters
Global Profiles
SIP Cluster
Domain Policies
Application Rules
Border Rules
Media Rules
Security Rules
Signaling Rules
Time of Day Rules
End Point Policy

Application Rules: SP-AR

Add
Filter By Device...
Rename
Clone
Delete

Application Rules
default
SP-AR

Click here to add a description.

Application Rule

Application Type	In	Out	Maximum Concurrent Sessions	Maximum Sessions Per Endpoint
Voice	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	1000	1000
Video	<input type="checkbox"/>	<input type="checkbox"/>		
IM	<input type="checkbox"/>	<input type="checkbox"/>		
Miscellaneous				
CDR Support	None			
RTCP Keep-Alive	No			

6.3.2. Media Rules

Media Rules define RTP media packet parameters such as priority encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packet matching the criteria will be handled by the Avaya SBCE security product.

A cloned Media Rule is created to set the **Quality of Service** and **Media Anomaly Detection**. The sample configuration shows Media Rule **SP-MR** used for both the enterprise and CenturyLink.

In the compliance testing, Media Rule **SP-MR** is cloned from the predefined **default-low-med** Media Rule.

To create Media Rule, navigate to **Domain Policies → Media Rules**. With **default-low-med** selected, click the **Clone** button (not shown).

Enter a Media Rule with a descriptive name **SP-MR** and click **Finish** (not shown).

6.3.3. 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 (not shown).

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SPOC 11/15/2013

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In the compliance testing, two Signaling Rules; **SP-SR** and **EN-SR**, were created from the **default** Signaling Rule for CenturyLink and Communication Manager respectively (not shown).

6.3.4. 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 CenturyLink and Communication Manager.

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

Endpoint Policy Group for CenturyLink

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

- Set Application Rule to **SP-AR** as created in **Section 6.3.1**.
- Set Media Rule to **SP-MR** as created **Section 6.3.2**.
- Set Signaling Rule to **SP-SR** as created in **Section Error! Reference source not found.**
- Set Border Rule to **default**.
- Set Time of Day Rule to **default**.
- Set Security Rule to **default-high**.

Session Border Controller for Enterprise AVAYA

Dashboard
Administration
Backup/Restore
System Management
 > Global Parameters
 > Global Profiles
 > SIP Cluster
 ▣ Domain Policies
 Application Rules
 Border Rules
 Media Rules
 Security Rules

Policy Groups: SP-PG

Filter By Device...

Click here to add a description.

Hover over a row to see its description.

Policy Group

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	SP-AR	default	SP-MR	default-high	SP-SR	default	Edit Clone

Endpoint Policy Group for Communication Manager

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

- Set Application Rule to **SP-AR** as created in **Section 6.3.1**.
- Set Media Rule to **SP-MR** as created **Section 6.3.2**.
- Set Signaling Rule to **EN-SR** as created in **Section Error! Reference source not found..**
- Set Border Rule to **default**.
- Set Time of Day Rule to **default**.
- Set Security Rule to **default-low**.

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, SIP Cluster, and Domain Policies. Under Domain Policies, there are sub-options: Application Rules, Border Rules, Media Rules, and Security Rules. The main content area is titled "Policy Groups: EN-PG". It includes an "Add" button, a "Filter By Device..." dropdown, and "Rename" and "Delete" buttons. Below this is a table with a single row for "EN-PG" and a "Click here to add a description" link. A "Policy Group" section below the table shows a table with columns: Order, Application, Border, Media, Security, Signaling, Time of Day, and Edit/Clone. The table contains one row with values: 1, SP-AR, default, SP-MR, default-low, EN-SR, default, and Edit/Clone buttons.

6.3.5. Session Policy

Session Policy is applied based on the source and destination of a media session (i.e., which codec is to be applied to the media session between its source and destination). The source and destination are defined in the URI Group in **Section 6.2.1**.

In the compliance testing, a Session Policy named **SP-SP** was created to allow Avaya SBCE to anchor media in off-net call forward or off-net call transfer scenarios. It is applied to both Server Flows for Communication Manager and CenturyLink in **Section 6.4.5**.

To clone a Session Policy, navigate to **Domain Policies → Session Policies**. With the **default** rule chosen, click on the **Clone** button (not shown).

Enter a descriptive name **SP-SP** for the new policy and click **Finish** (not shown).

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, SIP Cluster, and Domain Policies. Under Domain Policies, there are sub-options: Application Rules, Border Rules, Media Rules, and Security Rules. The main content area is titled "Session Policies: SP-SP". It includes an "Add" button, a "Filter By Device..." dropdown, and "Rename", "Clone", and "Delete" buttons. Below this is a table with a single row for "SP-SP" and a "Click here to add a description" link. A "Session Policy" section below the table shows a table with columns: Codec Prioritization, Media Anchoring, and Media Forking Profile. The table contains one row with values: Media Anchoring (checked), and Media Forking Profile (None).

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**.

Session Border Controller for Enterprise AVAYA

Network Management: SBCE62

Devices: SBCE62

Network Configuration | Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from [System Management](#).

A1 Netmask	A2 Netmask	B1 Netmask	B2 Netmask
255.255.255.192		255.255.255.224	

IP Address	Public IP	Gateway	Interface	
110.10.98.13		110.10.98.1	A1	Delete
110.10.98.111		110.10.98.97	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.

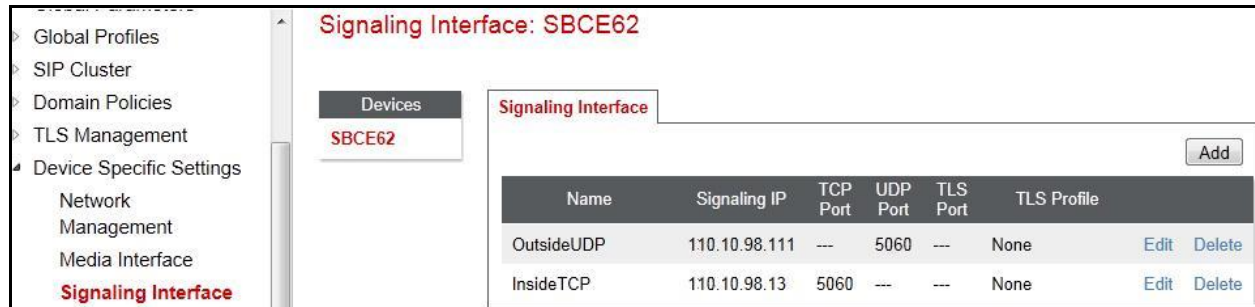
Name	Media IP	Port Range		
InsideMedia	110.10.98.13	35000 - 40000	Edit	Delete
OutsideMedia	110.10.98.111	35000 - 40000	Edit	Delete

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 CenturyLink and TCP/5060 for the inside interface to Communication Manager.



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 CenturyLink 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 6.3.4** 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 CenturyLink.

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

Finish

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	SP
Transport	*
Remote Subnet	*
Received Interface	OutsideUDP
Signaling Interface	InsideTCP
Media Interface	InsideMedia
End Point Policy Group	EN-PG
Routing Profile	EN-to-SP
Topology Hiding Profile	SP-to-EN
File Transfer Profile	None
Finish	

6.4.5. Session Flows

Session Flows allows defining certain parameters that pertain to the media portions of a call, whether it originates from the enterprise or outside the enterprise. This feature provides the complete and unparalleled flexibility to monitor, identify and control very specific types of calls based upon these user-definable parameters. Session Flows profiles SDP media parameters to completely identify and characterize a call placed through the network.

To create a session flow, navigate to **Device Specific Settings → Session Flows**. Click **Add** (not shown).

A common Session Flow was created for both CenturyLink and Communication Manager. In the new window that appears, enter the following values. Use default values for the remaining fields:

- **Flow Name:** Enter a descriptive name.
- **URI Group #1:** Select the URI Group created in **Section 6.2.1** to assign to the Session Flow as the source URI Group.
- **URI Group #2:** Select the URI Group created in **Section 6.2.1** to assign to the Session Flow as the destination URI Group.
- **Session Policy:** Select the session policy created in **Section 6.3.5** to assign to the Session Flow.
- Click **Finish**.

Note: A unique URI Group is used for source and destination, since it contains multiple URIs defined for the source as well as for the destination.

The following screen shows the Session Flow named **SP** was created.

Edit Flow: SP X

Flow Name	<input type="text" value="SP"/>
URI Group #1	<input type="text" value="SP"/> ▼
URI Group #2	<input type="text" value="SP"/> ▼
Subnet #1 Ex: 192.168.0.1/24	<input type="text" value="*"/>
Subnet #2 Ex: 192.168.0.1/24	<input type="text" value="*"/>
Session Policy	<input type="text" value="SP-SP"/> ▼

7. CenturyLink SIP Trunking Service Configuration

CenturyLink 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. CenturyLink will provide the customer with the necessary information to configure the SIP connection from the enterprise to CenturyLink. The information provided by CenturyLink 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).
- CenturyLink SIP domain. In the compliance testing, CenturyLink preferred to use IP address as an URI-Host.
- CPE SIP domain. In the compliance testing, CenturyLink preferred to use IP address of the Avaya SBCE as an URI-Host.
- Supported codecs.
- DID numbers.

The sample configuration between CenturyLink and the enterprise for the compliance testing is a static configuration. There is no registration on the SIP trunk implemented on either CenturyLink 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 CenturyLink and the Avaya SBCE.

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

- Inbound INVITE request from CenturyLink:

```
INVITE sip:3006157105@110.10.98.111:5060 SIP/2.0
Via: SIP/2.0/UDP 10.148.33.40:5060;branch=z9hG4bK04B53497c11b10b065b
From: "Unavailable" <sip:6139675203@10.148.33.40:5060>;tag=gK044b53d8
To: <sip:3006157105@110.10.98.111:5060>
Call-ID: 855901528_2868358@10.148.33.40
CSeq: 2212 INVITE
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,BYE,REFER,INFO,NOTIFY,PRACK,UPDATE,OPTIONS
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Contact: "Unavailable" <sip:6139675203@10.148.33.40:5060>
P-Asserted-Identity: "Unavailable" <sip:6139675203@10.148.33.40:5060>
Diversion: <sip:15134271991@10.148.33.40:5060>;privacy=off;screen=no; reason=unknown;
counter=1
Diversion: <sip:13006157105@10.148.33.40:5060>;privacy=off;screen=no; reason=unknown;
counter=1
Supported: 100rel
Content-Length: 280
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 31914 13869 IN IP4 10.148.33.40
s=SIP Media Capabilities
c=IN IP4 10.148.33.37
t=0 0
m=audio 28744 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

- 200OK/SDP response by the enterprise:

```
SIP/2.0 200 OK
From: "Unavailable" <sip:6139675203@10.148.33.40:5060>;tag=gK044b53d8
To: <sip:3006157105@110.10.98.111:5060>;tag=80be5342b426e31e005246c01700
CSeq: 2212 INVITE
Call-ID: 855901528_2868358@10.148.33.40
Contact: "H.323 9641" <sip:3006157105@110.10.98.111:5060;transport=udp>
Record-Route: <sip:110.10.98.111:5060;ipcs-line=86622;lr;transport=udp>
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,NOTIFY,REFER,INFO,PRACK,UPDATE
Supported: 100rel,join,replaces,sdp-anat,timer
Via: SIP/2.0/UDP 10.148.33.40:5060;branch=z9hG4bK04B53497c11b10b065b
Accept-Language: en
Server: Avaya CM/R016x.03.0.124.0
P-Asserted-Identity: "H.323 9641" <sip:3006157105@110.10.98.111:5060>
Session-Expires: 1200;refresher=uas
Content-Type: application/sdp
Content-Length: 175

v=0
o=- 1379354543 2 IN IP4 110.10.98.111
s=-
c=IN IP4 110.10.98.111
b=AS:64
t=0 0
m=audio 35402 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 CenturyLink.

- Outbound INVITE request from the enterprise:

```
INVITE sip:6139675203@10.148.33.41 SIP/2.0
From: "H.323 9611" <sip:3006157104@10.33.10.5>;tag=80dcd8a8b426e31eb05246c01700
To: <sip:6139675203@10.148.33.41>
CSeq: 1 INVITE
Call-ID: c97c974f395b023cd0d4916ebd97ee59
Contact: "H.323 9611" <sip:3006157104@110.10.98.111:5060>
Record-Route: <sip:110.10.98.111:5060;ipcs-line=86666;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 110.10.98.111:5060;branch=z9hG4bK-s1632-001895389282-1--s1632-
Accept-Language: en
Alert-Info: <cid:internal@bvwddev7.com>;avaya-cm-alert-type=internal
P-Asserted-Identity: "H.323 9611" <sip:3006157104@10.33.10.5>
Session-Expires: 1200;refresher=uac
Min-SE: 1200
Content-Type: application/sdp
Content-Length: 257

v=0
o=- 1379354713 1 IN IP4 110.10.98.111
s=-
c=IN IP4 110.10.98.111
b=AS:64
t=0 0
a=avf:avc=n prio=n
a=csup:avf-v0
m=audio 35408 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 CenturyLink:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 110.10.98.111:5060;branch=z9hG4bK-s1632-001895389282-1--s1632-
From: "H.323 9611" <sip:3006157104@10.33.10.5>;tag=80dcd8a8b426e31eb05246c01700
To: <sip:6139675203@10.148.33.41>;tag=gK04cece8d
Call-ID: c97c974f395b023cd0d4916ebd97ee59
CSeq: 1 INVITE
Record-Route: <sip:110.10.98.111:5060;ipcs-line=86666;lr;transport=udp>
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Contact: <sip:6139675203@10.148.33.41:5060>
Allow: INVITE,ACK,CANCEL,BYE,REFER,INFO,NOTIFY,PRACK,UPDATE,OPTIONS
Content-Length: 233
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 18911 26362 IN IP4 10.148.33.41
s=SIP Media Capabilities
c=IN IP4 10.148.33.37
t=0 0
m=audio 16698 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

8.3.2. 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 CenturyLink SIP Trunking Service. CenturyLink SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large the enterprises. CenturyLink 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 CenturyLink 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 6.2.1.0.9, December 2012.
- [2] *Administering Avaya Aura® System Platform*, Release 6.2.1, July 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] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3, Document Number 03-300509.
- [6] *Avaya one-X® Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, Release 3.1, November 2009, Document Number 16-300698.
- [7] *Avaya one-X® Deskphone SIP for 9600 Series IP Telephones Administrator Guide*, Release 2.6, June 2010, Document Number 16-601944.
- [8] *Administering Avaya one-X® Communicator*, April 2011.
- [9] *Using Avaya one-X® Communicator*, April 2011.
- [10] *Avaya SBCE Install Guide (102-5224-400v1.01)*
- [11] *Avaya SBCE Administration Guide (010-5423-400v106)*
- [12] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [13] *RFC 3515, The Session Initiation Protocol (SIP) Refer Method*, <http://www.ietf.org/>
- [14] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>
- [15] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, March 2013
- [16] *Avaya Session Border Controller for Enterprise Overview and Specification*, Issue 2, March 2013

Product documentation for CenturyLink SIP Trunking Service is available from CenturyLink Communication.

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