



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring XO SIP Trunking with Avaya Aura® Communication Manager Evolution Server, Avaya Aura® Session Manager, and Avaya Aura® Session Border Controller – Issue 1.0

Abstract

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

XO 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 was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

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

Customers using this Avaya SIP-enabled enterprise solution with XO SIP Trunking are able to place and receive PSTN calls via a broadband WAN connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

1.1. Interoperability Compliance Testing

A simulated enterprise site using Communication Manager, Session Manager and the SBC was connected to the public Internet using a broadband connection. The enterprise site was configured to connect to XO SIP Trunking.

To verify SIP trunking interoperability, the following features and functionality were covered during the interoperability compliance test:

- Incoming PSTN calls to various phone types
Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types
Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya one-X Communicator (soft client)
Avaya one-X Communicator supports two modes (Road Warrior and Telecommuter). Each supported mode was tested. Avaya one-X Communicator also supports two Voice Over IP (VoIP) protocols: H.323 and SIP. Only the H.323 version of one-X Communicator was tested.
- Various call types including: local, long distance, international, outbound toll-free, inbound toll-free, operator assisted calls, local directory assistance (411) and emergency calls (911).
- Codec G.711MU and G.729A.
- DTMF transmission using RFC 2833
- Caller ID presentation and Caller ID restriction
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and mobility (extension to cellular)
- T.38 Fax

Items not supported or not tested included the following:

- Network Call Redirection using the SIP REFER method or a 302 response was not tested.
- G.711 pass-through fax was not tested as it is not recommended by Avaya for use over SIP trunks

Interoperability testing of XO SIP Trunking was completed with successful results for all test cases with the exception of the observations/limitations described below.

- **Max-Forwards:** On incoming PSTN calls to an enterprise SIP phone, the Max-Forwards value in the incoming SIP INVITE was too small to allow the message to traverse all the SIP hops internal to the enterprise to reach the SIP phone. Thus, the SBC was used to increase this value when the INVITE arrived at the SBC from the network. (See **Section 6.2.3**)
- **DTMF digits detection:** By default, XO sends DTMF digits as both out-of-band RTP events as per RFC 2833 and as in-band tones. For interoperability, XO must disable the sending of in-band digits and only send DTMF digits as out-of-band RTP events. The XO National Activations Center (NAC) technician will disable this at the time of service activation. Otherwise, the detection of incoming DTMF digits from the network is unreliable. In-band tones must be disabled when using either the G.711 or G.729A codec. In rare cases if problems persist, the workaround described in **Appendix B** can also be applied. However, this is not recommended unless absolutely necessary since it burdens the media resources of the Communication Manager with additional processing.
- **Calling Party Number (PSTN transfers):** The calling party number displayed on the PSTN phone is not updated to reflect the true connected party on calls that are transferred to the PSTN. After the call transfer is complete, the calling party number displays the number of the transferring party and not the actual connected party. Communication Manager provides the new connected party information by updating the Contact header in an UPDATE message. XO does not use the UPDATE message for this purpose but instead uses the contents of the INVITE From header for the calling party display information.
- **EC500 Extend:** EC500 is the Communication Manager mobility feature which allows a user to have incoming calls ring his desk phone as well as a remote number such as a cell phone. The Extend feature allows this same user to "extend" to the remote number an active call that was answered at the desk phone. The user can then hang up the desk phone and continue the call on the cell phone. In the case of the compliance test using XO SIP Trunking, when the extended call was hung up at the desk phone, the active call to the remote number was also disconnected.
- **EC500 Confirmed Answer:** The EC500 confirmed answer feature requires the user to enter a digit when the call is answered on the remote number. This prevents the call from being answered inadvertently by voicemail on the cell phone account, since the call is not recognized as answered until a digit is entered. In the case of the compliance test using XO SIP Trunking, the user is still not connected after the digit is entered and the call to the remote number is dropped while the desk phone continues to ring. It is not recommended to use confirmed answer with this solution.

1.2. Support

For technical support on XO SIP Trunking, contact XO using the Customer Care links at www.xo.com.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. Selecting the **Support Contact Options** link followed by **Maintenance Support** provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

2. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to XO SIP Trunking. This is the configuration used for compliance testing.

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

- Avaya S8300D Server running Communication Manager
- Avaya G450 Media Gateway
- Avaya S8800 Server running Session Manager
- Avaya S8800 Server running System Manager
- Avaya 9600-Series IP telephones (H.323 and SIP)
- Avaya 4600-Series IP telephones (H.323)
- Avaya 1600-Series IP telephones (H.323)
- Avaya one-X Communicator (H.323)
- Avaya digital and analog telephones

Located at the edge of the enterprise is the SBC. It has a public side that connects to the external network and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flows through the SBC. In this way, the SBC can protect the enterprise against any SIP-based attacks. The SBC provides network address translation at both the IP and SIP layers. For security reasons, any actual public IP addresses used in the configuration have been replaced with private IP addresses. Similarly, any references to real routable PSTN numbers have also been changed to numbers that can not be routed by the PSTN.

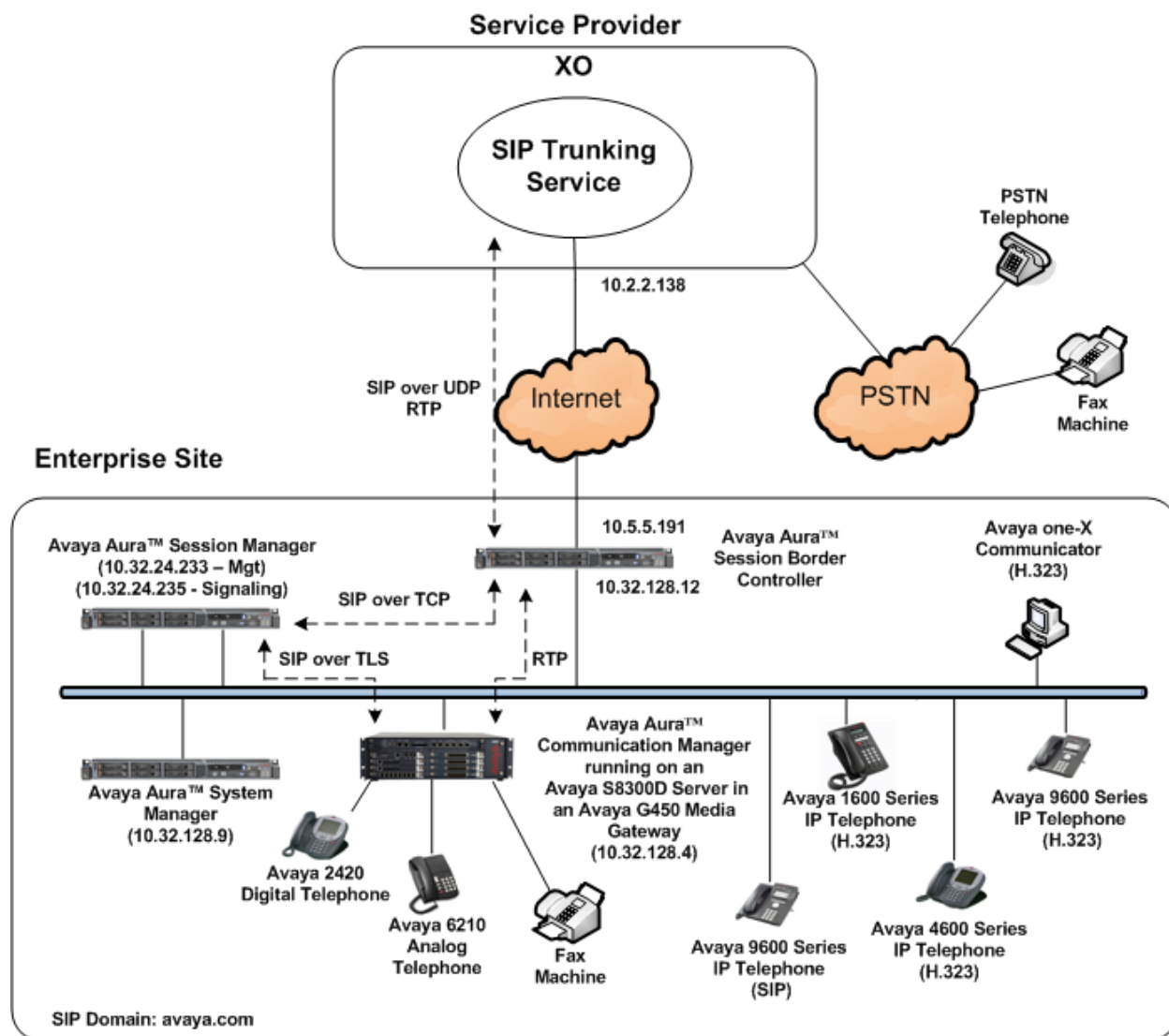


Figure 1: Avaya IP Telephony Network using XO SIP Trunking

A separate trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec setting required by the service provider could be applied only to this trunk and not affect other enterprise SIP traffic. In addition, this trunk carried both inbound and outbound traffic.

For inbound calls, the calls flow from the service provider to the SBC then to Session Manager. Session Manager uses the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case the Communication Manager) and on which link to send the call. Once the call arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed.

Outbound calls to the PSTN are first processed by Communication Manager and may be subject to outbound features such as automatic route selection, digit manipulation and class of service

restrictions. Once Communication Manager selects the proper SIP trunk, the call is routed to Session Manager. The Session Manager once again uses the configured dial patterns (or regular expressions) to determine the route to the SBC. From the SBC, the call is sent to XO SIP Trunking.

3. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Aura® Communication Manager running on an Avaya S8300D Server	6.0 SP0 (R016x.00.0.345.0-18246)
Avaya G450 Media Gateway	30.14.0
Avaya Aura® Session Manager running on an Avaya S8800 Server	6.0 (Build asm-6.0.0.0.600020)
Avaya Aura® System Manager running on an Avaya S8800 Server	6.0 (Build 6.0.0.0.556-3.0.6.1)
Avaya 1608 IP Telephone (H.323)	Avaya one-X Deskphone Value Edition 1.2.2
Avaya 4621SW IP Telephone (H.323)	2.9.1
Avaya 9640 IP Telephone (H.323)	Avaya one-X Deskphone Edition 3.1.1
Avaya 9630 IP Telephone (H.323)	Avaya one-X Deskphone SIP Edition 2.6
Avaya one-X Communicator (H.323)	6.0
Avaya 2420 Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Avaya Aura® Session Border Controller	6.0 (Build SBCT_6.0.0.1.4)
XO SIP Trunking Solution Components	
Component	Release
Sonus Networks Network Border Switch (NBS)	07.03.01 R006 07.03.01 R005
Sonus Networks PSX Routing Server	
Sonus Networks GSX Gateway	06.04.12 F001
Sonus Networks PSX Routing Server	06.04.17 R001
Broadsoft BroadWorks VoIP Applications Platform including:	Release 14
<ul style="list-style-type: none"> • Broadsoft Application Server (AS) • Broadsoft Network Server (NS) 	Rel_14.sp9_1.123 Rel_14.sp4_1.165

Table 1: Equipment and Software Tested

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Communication Manager and Session Manager.

4. Configure Communication Manager

This section describes the procedure for configuring Communication Manager for XO SIP Trunking. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from XO. It is assumed the general installation of Communication Manager, Avaya G450 Media Gateway and Session Manager has been previously completed and is not discussed here.

In configuring the Communication Manager, various components such as ip-network-regions, signaling groups, trunk groups, etc. need to be selected or created for use with the SIP connection to the service provider. Unless specifically stated otherwise, any unused ip-network-region, signaling group, trunk group, etc. can be used for this purpose.

The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. Note that the public IP addresses and phone numbers shown throughout these Application Notes have been edited so that the actual IP addresses of the network elements and public PSTN numbers are not revealed.

4.1. 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 the service provider. The example shows that 4000 licenses are available and 20 are in use. 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 sales representative to add additional capacity.

```

display system-parameters customer-options                               Page 2 of 11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
      Maximum Administered H.323 Trunks: 4000 36
    Maximum Concurrently Registered IP Stations: 2400 3
      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 0
      Maximum Administered SIP Trunks: 4000 20
    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 0
      Maximum TN2602 Boards with 80 VoIP Channels: 128 0
  
```

4.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow incoming calls 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 19
                    FEATURE-RELATED SYSTEM PARAMETERS
                    Self Station Display Enabled? n
                    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. This text string is entered in the two fields highlighted below. The compliance test used the value of **anonymous** for both.

```
change system-parameters features                               Page 9 of 19
                    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:
    International Access Code:

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
```

4.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the Avaya S8300D Server running Communication Manager (*procr*) and for Session Manager (*sessionMgr*). These node names will be needed for defining the service provider signaling group in **Section 4.6**.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
  Name          IP Address
  cmm           10.32.128.4
  default       0.0.0.0
  procr        10.32.128.4
  procr6        ::
  sessionMgr  10.32.24.235
```

4.4. Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, ip-codec-set 2 was used for this purpose. XO SIP Trunking supports G.711MU and G.729A. Thus, these codecs were included in this set. Enter **G.711MU** and **G.729A** in the **Audio Codec** column of the table. Default values can be used for all other fields.

```
change ip-codec-set 2                                     Page 1 of 2
                                                    IP Codec Set
  Codec Set: 2
  Audio Codec      Silence Suppression  Frames Per Pkt  Packet Size (ms)
  1: G.711MU      n                2              20
  2: G.729A      n                2              20
  3:
```

On **Page 2**, set the **Fax Mode** to **t.38-standard**.

```
change ip-codec-set 2                                     Page 2 of 2
                                                    IP Codec Set
  Allow Direct-IP Multimedia? n
  FAX Mode          Redundancy
  t.38-standard    0
  Modem             off          0
  TDD/TTY           US          3
```

4.5. IP Network Region

Create a separate IP network region for the service provider trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and the service provider versus calls within the enterprise or elsewhere. For the compliance test, IP-network-region 2 was chosen for the service provider trunk. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is *avaya.com*. This name appears in the “From” header of SIP messages 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 Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to *yes*. This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 4.4**.
- Default values can be used for all other fields.

```
change ip-network-region 2                                     Page 1 of 20
                                                           IP NETWORK REGION
Region: 2
Location: 1          Authoritative Domain: avaya.com
      Name: SP Region
MEDIA PARAMETERS          Intra-region IP-IP Direct Audio: yes
      Codec Set: 2          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
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
      Audio 802.1p Priority: 6
      Video 802.1p Priority: 5          AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS          RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
      Keep-Alive Count: 5
```

On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 2 will be used for calls between region 2 (the service provider region) and region 1 (the rest of the enterprise).

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

4.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and the Session Manager 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 3 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*. This specifies the Communication Manager will serve as an Evolution Server for the Session Manager.
- Set the **Transport Method** to the recommended default value of *tls* (Transport Layer Security). For ease of troubleshooting during testing, the compliance test was conducted with the **Transport Method** set to *tcp*. The transport method specified here is used between the Communication Manager and Session Manager. The transport method used between the Session Manager and the SBC is specified as TCP in **Sections 5.6** and **6.1.3**. Lastly, the transport method between the SBC and XO is UDP. This is defined in **Section 6.1.3** when the service provider name is selected.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so the SM can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to *5062*. (For TCP, the well-known port value is 5060).
- Set the **Peer Detection Enabled** field to *y*. The **Peer-Server** field will initially be set to *Others* and can not be changed via administration. Later, the **Peer-Server** field will automatically change to *SM* once Communication Manager detects its peer as a Session Manager.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Avaya S8300D Server running Communication Manager as defined in **Section 4.3**.

- Set the **Far-end Node Name** to *sessionMgr*. This node name maps to the IP address of Session Manager as defined in **Section 4.3**.
- Set the **Far-end Network Region** to the IP network region defined for the service provider in **Section 4.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- 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 Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Default values may be used for all other fields.

```

add signaling-group 3                                     Page 1 of 1
                                                    SIGNALING GROUP

Group Number: 3                Group Type: sip
IMS Enabled? n                Transport Method: tcp
    Q-SIP? n
    IP Video? n                SIP Enabled LSP? n
Peer Detection Enabled? y    Peer Server: Others
                                Enforce SIPS URI for SRTP? y

Near-end Node Name: procr                Far-end Node Name: sessionMgr
Near-end Listen Port: 5062                Far-end Listen Port: 5062
                                           Far-end Network Region: 2

Far-end Domain: avaya.com

Incoming Dialog Loopbacks: eliminate                Bypass If IP Threshold Exceeded? n
                                                    RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload                Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                IP Audio Hairpinning? n
    Enable Layer 3 Test? n                Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n                Alternate Route Timer(sec): 6

```

4.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 4.6**. For the compliance test, trunk group 3 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 **Service Type** field to *public-ntwrk*.
- Set **Member Assignment Method** to *auto*.
- Set the **Signaling Group** to the signaling group shown in the previous step.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Default values were used for all other fields.

```
add trunk-group 3                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 3                                     Group Type: sip           CDR Reports: y
  Group Name: SP Trunk                             COR: 1                   TN: 1           TAC: 1003
  Direction: two-way                               Outgoing Display? n
  Dial Access? n                                   Night Service:
  Queue Length: 0
  Service Type: public-ntwrk                       Auth Code? n
                                               Member Assignment Method: auto
                                               Signaling Group: 3
                                               Number of Members: 10
```

On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to the service provider. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **600** seconds was used.

```
add trunk-group 3                                     Page 2 of 21
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name: auto
                                               Redirect On OPTIM Failure: 5000
  SCCAN? n                                         Digital Loss Group: 18
  Preferred Minimum Session Refresh Interval(sec): 600
                                               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 calling party number (CPN) sent to the far-end. Beginning with Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP From, Contact and P-Asserted Identity headers. The addition of the + sign impacted interoperability with XO. Thus, the **Numbering Format** was set to *private* and the **Numbering Format** in the route pattern was set to *unk-unk* (see **Section 4.9**).

Set the **Replace Restricted Numbers** and **Replace Unavailable Numbers** fields to *y*. This will allow the CPN displayed on local endpoints to be replaced with the value set in **Section 4.2**, if the inbound call enabled CPN block. For outbound calls, these same settings request that CPN block be activated on the far-end destination if a local user requests CPN block on a particular call routed out this trunk. Default values were used for all other fields.

```
add trunk-group 3                                     Page 3 of 21
TRUNK FEATURES
    ACA Assignment? n                               Measured: none
                                                    Maintenance Tests? y
    Numbering Format: private
                                                    UII 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**, set the **Network Call Redirection** field to *n*. 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 calls back to the PSTN and some Extension to Cellular (EC500) call scenarios.

Set the **Telephone Event Payload Type** to *101*, the value preferred by XO.

```
add trunk-group 3                                     Page 4 of 21
                                                    PROTOCOL VARIATIONS
    Mark Users as Phone? n
    Prepend '+' to Calling Number? n
    Send Transferring Party Information? n
    Network Call Redirection? n
    Send Diversion Header? y
    Support Request History? y
    Telephone Event Payload Type: 101
    Convert 180 to 183 for Early Media? n
    Always Use re-INVITE for Display Updates? n
    Enable Q-SIP? n
```

4.8. Calling Party Information

The calling party number is sent in the SIP “From”, “Contact” and “PAI” headers. Since private numbering was selected to define the format of this number (**Section 4.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. The DID number will be one assigned by the SIP service provider. It is used to authenticate the caller.

In the sample configuration, three DID numbers were assigned for testing. These three numbers were assigned to the three extensions 40003, 40005 and 40010. Thus, these same 10-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these three extensions.

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

Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4			5	Total Administered: 4
5	40003	3	2145551234	10	Maximum Entries: 240
5	40005	3	2145551235	10	
5	40010	3	2145551236	10	

In a real customer environment, normally the DID number is comprised of the local extension plus a prefix. If this is true, then a single private numbering entry can be applied for all extensions. In the example below, all stations with a 5-digit extension beginning with 4 will send the calling party number as the **Private Prefix** plus the extension number.

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

Ext Len	Ext Code	Trk Grp (s)	Private Prefix	Total Len	
5	4	3	21455	10	Total Administered: 1
					Maximum Entries: 240

Even though private numbering was 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	
5	4			5	Total Administered: 4
5	40003	3	2145551234	10	Maximum Entries: 240
5	40005	3	2145551235	10	Note: If an entry applies to
5	40010	3	2145551236	10	a SIP connection to Avaya
					Aura(tm) Session Manager,
					the resulting number must
					be a complete E.164 number.

4.9. Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the service provider. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an “outside line”. This common configuration is illustrated below with little elaboration. Use the **change dialplan analysis** command to define a dialed string beginning with 9 of length 1 as a feature access code (**fac**).

```
change dialplan analysis                                     Page 1 of 12
                                                           DIAL PLAN ANALYSIS TABLE
                                                           Location: all                                     Percent Full: 2
```

Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
1	4	dac						
4	5	ext						
8	1	fac						
9	1	fac						
*	3	fac						
#	3	fac						

Use the **change feature-access-codes** command to configure **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:
Answer Back Access Code:
Attendant Access Code:
Auto Alternate Routing (AAR) Access Code: 8
Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
Automatic Callback Activation:                      Deactivation:
Call Forwarding Activation Busy/DA: *01 All: *02    Deactivation: *03
```

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 test. See **Section 1.1** for the complete list of call types tested. All dialed strings are mapped to route pattern 2 which contains the SIP trunk to the service provider (as defined next).

change ars analysis 0		ARS DIGIT ANALYSIS TABLE						Page 1 of 2
		Location: all				Percent Full: 2		
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd		
0	1	1	2	op		n		
0	11	11	2	op		n		
00	2	2	2	op		n		
011	10	18	2	intl		n		
1800	11	11	2	fpna		n		
1877	11	11	2	fpna		n		
1908	11	11	2	fpna		n		
411	3	3	2	svcl		n		

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route-pattern** command to configure the parameters for the service provider trunk route pattern in the following manner. The example below shows the values used for route pattern 2 during the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider. For the compliance test, trunk group 3 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.
- **Pfx Mrk:** **1** The prefix mark (Pfx Mrk) of one will prefix any FNPA 10-digit number with a 1 and leave numbers of any other length unchanged. This will ensure 1 + 10 digits are sent to the service provider for long distance North American Numbering Plan (NANP) numbers. All HNPA 10 digit numbers are left unchanged.
- **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 4.7**.
- **LAR:** **next**

change route-pattern 2													Page	1 of	3						
Pattern Number: 2													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	Intw							
1:	3	0	1										n	user							
2:													n	user							
3:													n	user							
4:													n	user							
5:													n	user							
6:													n	user							
BCC VALUE													TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
0 1 2 M 4 W														Request					Dgts	Format	
															Subaddress						
1:	y	y	y	y	y	n	n								rest		unk-unk	next			
2:	y	y	y	y	y	n	n								rest			none			
3:	y	y	y	y	y	n	n								rest			none			
4:	y	y	y	y	y	n	n								rest			none			

5. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation
- SIP Entities corresponding to Communication Manager, the SBC and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Regular Expressions, which also can be used to route calls
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager.

It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

5.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Login** (not shown). The screen shown below is then displayed. The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items will be located under the **Routing** link shown below.

The screenshot shows the Avaya Aura System Manager 6.0 Home Screen. At the top left is the Avaya logo. To its right is the text "Avaya Aura™ System Manager 6.0". Further right, it says "Welcome, admin Last Logged on at August 13, 2010 4:53 PM" and "Help | About | Change Password | Log off".

On the left is a navigation tree with the following items: Elements, Events, Groups & Roles, Licenses, Routing (expanded), Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, Defaults, Security, System Manager Data, and Users.

The main content area is titled "Home Screen" and contains a "Sub Pages" table:

Action	Description	Help
Elements	Interface to manage the application instances and contains the element managers for the different managed elements in the deployment.	Help for managing elements
Events	Interface to view and administer logs and alarms.	Help for managing logs and alarms
Groups & Roles	Interface to manage groups, resources and roles.	Help for managing groups and roles
Licenses	Interface to manage licenses for individual applications of Avaya Aura (TM) Unified Communication Solution.	Help for managing licenses
Routing	Interface to manage routing policies, adaptations, dial patterns, SIP elements.	Help for managing routing policies
Security	Interface to manage certificates .Certificates help enable setting up secure communication between different elements in the Avaya Aura (TM) Unified Communication Solution.	Help for managing certificates
System Manager Data	Interface to backup and restore System Manager data, manage data retention rules, list extension pack information, manage replication nodes, manage scheduled jobs and System Manager configuration.	Help for managing System Manager data and configuration
Users	Interface to administer users, contact lists, shared addresses and Access Control Lists (ACLs).	Help for managing users

5.2. Specify SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain (*avaya.com*).

Navigate to **Routing → Domains** in the left-hand navigation pane (**Section 5.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain.

Domain Management Commit Cancel

1 Item | Refresh Filter: Enable

Name	Type	Default	Notes
* avaya.com	sip	<input type="checkbox"/>	Enterprise Domain

* Input Required Commit Cancel

5.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing → Locations** in the left-hand navigation pane (**Section 5.1**) and click the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

In the **Location Pattern** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

The screen below shows the addition of the **Location 1**, which includes all equipment on the **10.32.128.x** subnet including Communication Manager, and the SBC. Click **Commit** to save.

Location Details

General

* **Name:**

Notes:

Managed Bandwidth: ▾

* **Average Bandwidth per Call:** ▾

Location Pattern

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.32.128.*	<input type="text"/>

Select : All, None

Repeat the preceding procedure to create **Location 2** which includes all equipment on the **10.32.24.x** subnet which includes the Session Manager.

Location Details Commit Cancel

General

*** Name:**

Notes:

Managed Bandwidth: Kbit/sec ▾

*** Average Bandwidth per Call:** Kbit/sec ▾

Location Pattern

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.32.24.*	<input type="text"/>

Select : All, None

5.4. Add Adaptation Module

Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products.

For XO interoperability, one adaptation is needed and maps inbound DID numbers from XO to local Communication Manager extensions. The adaptation is applied to the Communication Manager SIP entity.

To create the adaptation, navigate to **Routing** → **Adaptations** in the left-hand navigation pane and click on the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Adaptation Name:** Enter a descriptive name for the adaptation.
- **Module Name:** Enter *DigitConversionAdapter*.

Adaptation Details Commit Cancel

General

* **Adaptation name:**

Module name:

Module parameter:

Egress URI Parameters:

Notes:

To map inbound DID numbers from XO to Communication Manager extensions, scroll down to the **Digit Conversion for Outgoing Calls from SM** section. Create an entry for each DID to be mapped. Click **Add** and enter the following values for each mapping. Use default values for all remaining fields:

- **Matching Pattern:** Enter a digit string used to match the inbound DID number.
- **Min:** Enter a minimum dialed number length used in the match criteria.
- **Max:** Enter a maximum dialed number length used in the match criteria.
- **Delete Digits** Enter the number of digits to delete from the beginning of the received number.
- **Insert Digits:** Enter the number of digits to insert at the beginning of the received number.
- **Address to modify:** Select **both**.

Click **Commit** to save.

Digit Conversion for Incoming Calls to SM

0 Items | Refresh Filter: Enable

	Matching Pattern	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>							

Digit Conversion for Outgoing Calls from SM

3 Items | Refresh Filter: Enable

	Matching Pattern ▲	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	* 2145551234	* 10	* 10	* 10	40003	both ▼	
<input type="checkbox"/>	* 2145551235	* 10	* 10	* 10	40005	both ▼	
<input type="checkbox"/>	* 2145551236	* 10	* 10	* 10	40010	both ▼	

Select : All, None

*** Input Required**

5.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes Communication Manager and the SBC. Navigate to **Routing → SIP Entities** in the left-hand navigation pane (**Section 5.1**) and click on the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Enter *Session Manager* for Session Manager, *CM* for Communication Manager and *SIP Trunk* for the SBC.
- **Adaptation:** This field is only present if **Type** is not set to *Session Manager*. If applicable, select the **Adaptation Name** created in **Section 5.4** that will be applied to this entity.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.

The following screen shows the addition of Session Manager. The IP address of the virtual SM-100 Security Module is entered for **FQDN or IP Address**.

The screenshot shows a web-based configuration form titled "SIP Entity Details". At the top right, there are "Commit" and "Cancel" buttons. The form is divided into two sections: "General" and "SIP Link Monitoring".

General Section:

- Name:** Text input field containing "devcon-asm".
- * FQDN or IP Address:** Text input field containing "10.32.24.235".
- Type:** Dropdown menu with "Session Manager" selected.
- Notes:** Empty text input field.
- Location:** Dropdown menu with "Location 2" selected.
- Outbound Proxy:** Empty dropdown menu.
- Time Zone:** Dropdown menu with "America/New_York" selected.
- Credential name:** Empty text input field.

SIP Link Monitoring Section:

- SIP Link Monitoring:** Dropdown menu with "Use Session Manager Configuration" selected.

To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise.

Defaults can be used for the remaining fields. Click **Commit** to save.

For the compliance test, three **Port** entries were added.

The screenshot shows the 'Port' configuration section of a SIP Entity. It features an 'Add' button and a 'Remove' button. Below these is a table with 3 items, a 'Refresh' button, and a 'Filter: Enable' option. The table has columns for 'Port', 'Protocol', 'Default Domain', and 'Notes'. Each row contains a checkbox, a text input field for the port number, a dropdown menu for the protocol, a dropdown menu for the default domain, and a text input field for notes. The 'Select: All, None' option is visible below the table. At the bottom, there is a '* Input Required' warning and 'Commit' and 'Cancel' buttons.

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	
<input type="checkbox"/>	5060	UDP	avaya.com	
<input type="checkbox"/>	5061	TLS	avaya.com	

The following screen shows the addition of Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, this requires the creation of a separate SIP entity for Communication Manager than the one created at Session Manager installation for use with all other SIP traffic. The **FQDN or IP Address** field is set to the IP address of the Avaya S8300D Server running Communication Manager. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 5.4**.

SIP Entity Details Commit Cancel

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV:

* **SIP Timer B/F (in seconds):**

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

The following screen shows the addition of the SBC. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**).

SIP Entity Details Commit Cancel

General

* **Name:**

* **FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV:

* **SIP Timer B/F (in seconds):**

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

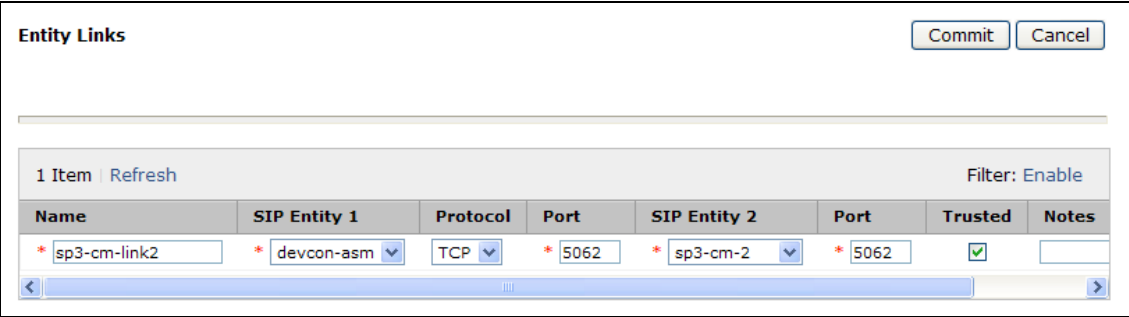
5.6. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created; one to the Communication Manager for use only by service provider traffic and one to the SBC. To add an Entity Link, navigate to **Routing** → **Entity Links** in the left-hand navigation pane (**Section 5.1**) and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 4.6**.
- **SIP Entity 2:** Select the name of the other system. For the Communication Manager, select the Communication Manager SIP Entity defined in **Section 5.5**.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. For the Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 4.6**.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 5.5** will be denied.*

Click **Commit** to save. The following screens illustrate the Entity Links to Communication Manager and the SBC. It should be noted that in a customer environment the Entity Link to Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic would not be encrypted. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 4.6**.

Entity Link to Communication Manager:



The screenshot shows the 'Entity Links' configuration page. At the top right are 'Commit' and 'Cancel' buttons. Below is a table with one row of data. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The data row shows: Name: sp3-cm-link2, SIP Entity 1: devcon-asm, Protocol: TCP, Port: 5062, SIP Entity 2: sp3-cm-2, Port: 5062, Trusted: checked, Notes: empty.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* sp3-cm-link2	* devcon-asm	TCP	* 5062	* sp3-cm-2	* 5062	<input checked="" type="checkbox"/>	

Entity Link to the SBC:

Entity Links Commit Cancel

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* toAuraSBC	* devcon-asm	TCP	* 5060	* sp-sbc1	* 5060	<input checked="" type="checkbox"/>	

< >

5.7. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 5.5**. Two routing policies must be added: one for Communication Manager and one for the SBC. To add a routing policy, navigate to **Routing → Routing Policies** in the left-hand navigation pane (**Section 5.1**) and click on the **New** button in the right pane (not shown). The following screen is displayed. Fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP Entity displays on the Routing Policy Details page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screens show the Routing Policies for Communication Manager and the SBC.

Routing Policy Details Commit Cancel

General

* **Name:**

Disabled:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
sp3-cm-2	10.32.128.4	CM	

Routing Policy Details Commit Cancel

General

* **Name:**

Disabled:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
sp-sbc1	10.32.128.12	SIP Trunk	

5.8. Add Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to XO and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane (**Section 5.1**) and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

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

Two examples of the dial patterns used for the compliance test are shown below. The first example shows that 11 digit numbers that begin with a 1 and have a destination domain of *avaya.com* from *Location 1* or *Location 2* uses route policy *SP AuraSBC route*.

Dial Pattern Details
Commit Cancel

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Location 1	SP Subnet(s)	SP Aura SBC route	0	<input type="checkbox"/>	sp-sbc1	
<input type="checkbox"/>	Location 2	Juan's Subnet(s)	SP Aura SBC route	0	<input type="checkbox"/>	sp-sbc1	

Select : All, None

The second example shows that 10 digit numbers that start with **214555** to domain **avaya.com** and originating from **Location 1** uses route policy **sp3-cm Route**. These are the DID numbers assigned to the enterprise from XO. Location 1 is selected because these calls come from the SBC which resides in location 1.

Dial Pattern Details Commit Cancel

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

SIP Domain: ▼

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Location 1	SP Subnet(s)	sp3-cm Route 2	0	<input type="checkbox"/>	sp3-cm-2	

Select : All, None

The complete list of dial patterns defined for the compliance test is shown below.

Dial Patterns

Edit New Duplicate Delete More Actions ▼ Commit

8 Items Refresh Filter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Notes
<input type="checkbox"/>	<u>0</u>	1	11	<input type="checkbox"/>	avaya.com	Dest: sp-sbc1
<input type="checkbox"/>	<u>011</u>	10	18	<input type="checkbox"/>	avaya.com	Dest: sp-sbc1
<input type="checkbox"/>	<u>1</u>	11	11	<input type="checkbox"/>	avaya.com	Dest: sp-sbc1
<input type="checkbox"/>	<u>214555</u>	10	10	<input type="checkbox"/>	avaya.com	Dest: sp3-cm-2
<input type="checkbox"/>	<u>411</u>	3	3	<input type="checkbox"/>	avaya.com	Dest: sp-sbc1

Select : All, None

5.9. Add/View Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add a Session Manager, navigate to **Elements** → **Session Manager** → **Session Manager Administration** in the left-hand navigation pane (**Section 5.1**) and click on the **New** button in the right pane (not shown). If the Session Manager already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

In the **General** section, enter the following values:

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description:** Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

The screen below shows the Session Manager values used for the compliance test.

View Session Manager Return

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

General ▾

SIP Entity Name

Description

Management Access Point Host Name/IP

Direct Routing to Endpoints

In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface.
- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the remaining Session Manager values used for the compliance test.

Security Module ▾

SIP Entity IP Address

Network Mask

Default Gateway

Call Control PHB

QOS Priority

Speed & Duplex

VLAN ID

6. Configure Avaya Aura® Session Border Controller

This section describes the configuration of the Avaya Aura® SBC. This configuration is done in two parts. The first part is done during the SBC installation via the installation wizard. These Application Notes will not cover the SBC installation in its entirety but will include the use of the installation wizard. For information on installing the Avaya Aura® System Platform and the loading of the Avaya Aura® SBC template see [1].

The second part of the configuration is done after the installation is complete using the SBC web interface. The resulting SBC configuration file is shown in **Appendix A**.

6.1. Installation Wizard

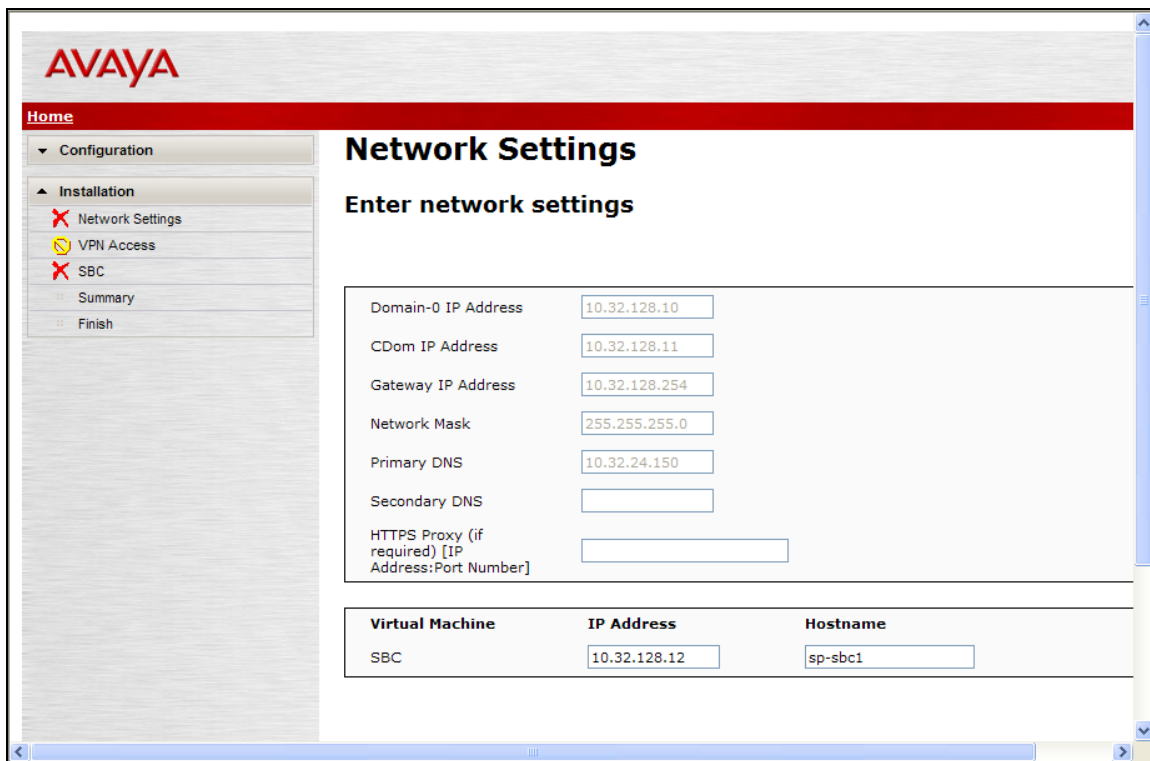
During the installation of the Avaya Aura® SBC template, the installation wizard will prompt the installer for information that will be used to create the initial configuration of the SBC.

6.1.1. Network Settings

The first screen of the installation wizard is the **Network Settings** screen. Fill in the fields as described below and shown in the following screen:

- **IP Address:** Enter the IP address of the private side of the SBC.
- **Hostname:** Enter a host name for the SBC.

Click **Next Step** (not shown) to continue.



The screenshot shows the Avaya Aura Network Settings installation wizard. The interface includes a navigation menu on the left with options like Configuration, Installation, Network Settings, VPN Access, SBC, Summary, and Finish. The main content area is titled "Network Settings" and "Enter network settings". It contains several input fields for network configuration: Domain-0 IP Address (10.32.128.10), CDom IP Address (10.32.128.11), Gateway IP Address (10.32.128.254), Network Mask (255.255.255.0), Primary DNS (10.32.24.150), Secondary DNS, and HTTPS Proxy (if required) [IP Address:Port Number]. Below these fields is a table for Virtual Machine settings.

Virtual Machine	IP Address	Hostname
SBC	10.32.128.12	sp-sbc1

6.1.2. VPN Access

VPN remote access to the SBC was not part of the compliance test. Thus, on the VPN Access screen, select **No** to the question, **Would you like to configure the VPN remote access parameters for System Platform?**

Click **Next Step** to continue.

AVAYA

Home

Configuration

Installation

- Network Settings
- VPN Access
- SBC
- Summary
- Finish

VPN Access

Configure VPN Access

Would you like to configure the VPN remote access parameters for System Platform?

Yes No

VPN Access Configuration

VPN Router IP Address

Remote Access Network

Remote Access Network Subnet Mask

The data on this page is used to configure static routes on System Platform to enable remote VPN access to the component applications and the Avaya Aura™ System Platform Web Console.

Once the template has been installed, the user must access the Avaya Aura™ System Platform Web Console and check the "Server Management -> Static Route Configuration" page to verify that the static routes configured by the Wizard are suitable for the intended remote access application.

If in doubt, please refer to the documentation.

[Previous Step](#) [Next Step](#)

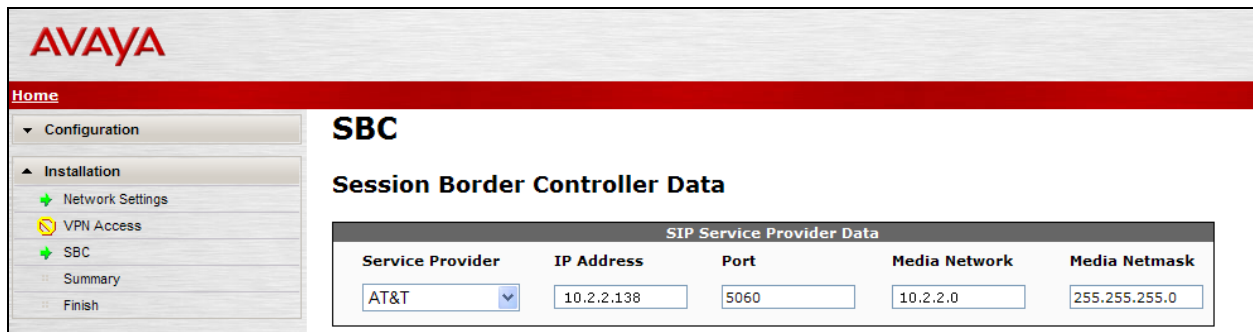
6.1.3. SBC

On the **SBC** screen, fill in the fields as described below and shown in the following screen:

In the **SIP Service Provider Data** section:

- **Service Provider:** From the pull-down menu, select the name of the service provider to which the SBC will connect. This will allow the wizard to select a configuration file customized for this service provider. At the time of the compliance test, a customized configuration file did not exist for XO. Thus, **AT&T** was chosen instead and further customization was done manually after the wizard was complete. A customized configuration file for XO should be available in a future SBC release.
- **IP Address:** Enter the XO provided IP address of the XO Sonus NBS. If the service provider has multiple proxies, enter the primary proxy on this screen and additional proxies can be added after installation.
- **Port:** Enter the port number that the service provider uses to listen for SIP traffic.
- **Media Network:** Enter the XO provided subnet where media traffic will originate. If media can originate from multiple networks, enter one network address on this screen and additional networks can be added after installation.
- **Media Netmask:** Enter the netmask corresponding to the **Media Network**.

Scroll down to continue.



The screenshot shows the Avaya SBC configuration interface. The top left corner features the Avaya logo. Below it is a navigation menu with options: Configuration, Installation, Network Settings, VPN Access, SBC, Summary, and Finish. The main content area is titled 'SBC' and 'Session Border Controller Data'. A section titled 'SIP Service Provider Data' contains a table with five columns: Service Provider, IP Address, Port, Media Network, and Media Netmask. The values entered are: AT&T (selected from a dropdown), 10.2.2.138, 5060, 10.2.2.0, and 255.255.255.0.

Service Provider	IP Address	Port	Media Network	Media Netmask
AT&T	10.2.2.138	5060	10.2.2.0	255.255.255.0

Further down on the same **SBC** screen, fill in the fields as described below:

In the **SBC Network Data** section:

- **Public IP Address:** Enter the IP address of the public side of the SBC.
- **Public Net Mask:** Enter the netmask associated with the public network to which the SBC connects.
- **Public Gateway:** Enter the default gateway of the public network.

In the **Enterprise SIP Server** section:

- **IP Address:** Enter the IP address of the Enterprise SIP Server to which the SBC will connect. In the case of the compliance test, this is the IP address of the Session Manager SIP signaling interface.
- **Transport:** From the pull-down menu, select the transport protocol to be used for SIP traffic between the SBC and Session Manager.
- **SIP Domain** Enter the enterprise SIP domain.

Click **Next Step** to continue. A summary screen will be displayed (not shown). Check the displayed values and click **Next Step** again to continue to the final step.

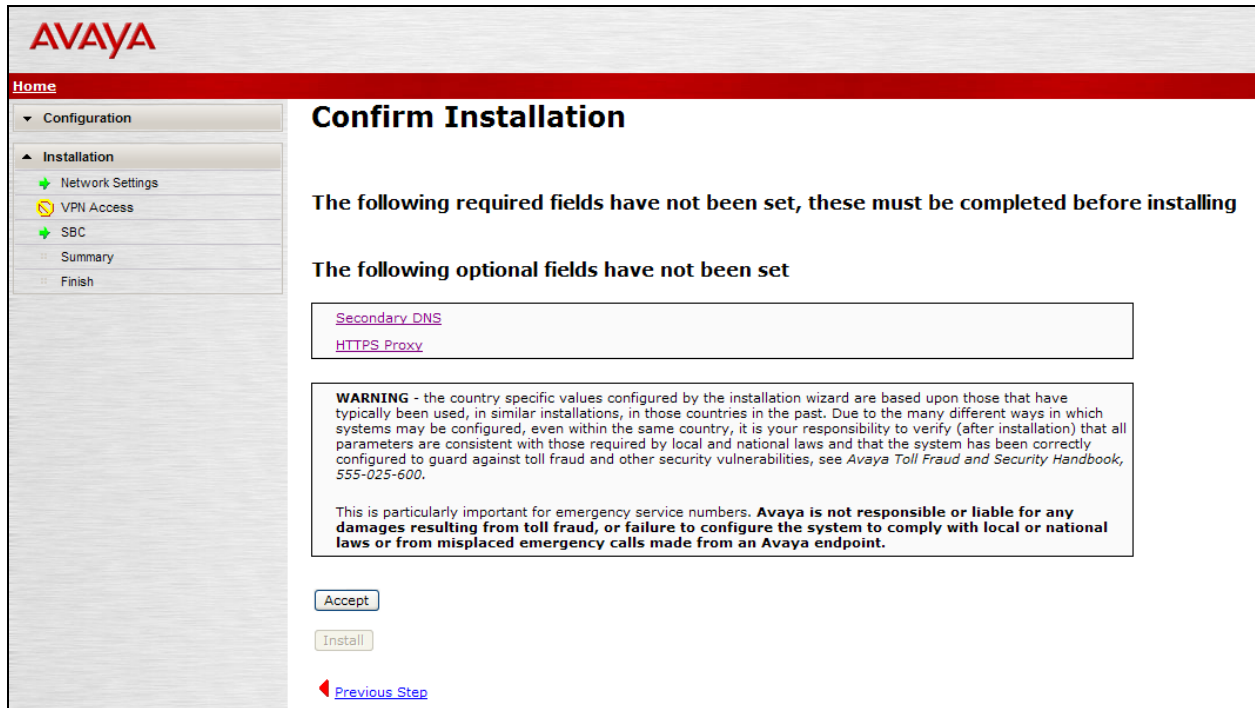
SBC Network Data			
Interface	IP Address	Net Mask	Gateway
Private (Management)	10.32.128.12	255.255.255.0	10.32.128.254
Public	<input type="text" value="10.5.5.191"/>	<input type="text" value="255.255.255.0"/>	<input type="text" value="10.5.5.254"/>

Enterprise SIP Server		
IP Address	Transport	SIP Domain
<input type="text" value="10.32.24.235"/>	<input type="text" value="TCP"/> ▼	<input type="text" value="avaya.com"/>

[◀ Previous Step](#) [Next Step ▶](#)

6.1.4. Confirm Installation

The **Confirm Installation** screen will indicate if any required or optional fields have not been set. The list of required fields that have not been set should be empty. If not, click **Previous Step** to navigate to the necessary screen to set the required field. Otherwise, click **Accept** to finish the wizard and to continue the overall template installation.



6.2. Post Installation Configuration

The installation wizard configures the Session Border Controller for use with the service provider chosen in **Section 6.1**. Since a different service provider other than XO had to be selected in the installation wizard then additional manual changes must also be performed. These changes are performed by accessing the browser-based GUI of the Session Border Controller, using the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured in **Section 6.1**. Log in with proper credentials.



6.2.1. Options Frequency

To set the frequency of the OPTIONS messages sent from the SBC to the service provider, first navigate to **vsp** → **enterprise** → **server** → **sig-gateway Telco**. Click **Show Advanced**.

The screenshot shows the Avaya Aura Configuration interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The main content area is titled 'Configure vspenterprise\servers\sig-gateway Telco' and includes a 'Show advanced' button and 'Help' and 'Index' links. Below this are 'Set', 'Reset', 'Back', 'Copy', and 'Delete' buttons. A list of links includes 'Manage connections', 'Log instant messages', 'Record media', 'Record files', 'Set up accounting', 'Change "from:" URI', and 'Change "to:" URI'. The 'general:' section contains the following fields:

* name	Telco
admin	enabled (Resource is active)
domain	
failover-detection	ping (Use OPTIONS to detect failures)

Scroll down to the **routing** section of the form. Enter the desired interval in the **ping-interval** field. Click **Set** at the top of the form (shown in previous figure).

The screenshot shows the 'routing:' section of the configuration page. It includes a 'routing-setting' section with a list of options: 'normalization', 'auto-tag-match', 'auto-domain-match', and 'pstn-backup'. Below this list are 'Select All' and 'Unselect All' buttons. Other fields include:

domain-alias	Edit domain-alias
domain-subnet	Edit domain-subnet
loop-detection	tight (Compare source and destination address/port/transport)
service-type	provider (Provider peer)
ping-interval	120 seconds

6.2.2. Blocked Headers

The P-Site header is sent in SIP messages from the Session Manager to the XO network. This header contains private IP addresses from the enterprise. These private IP addresses should not be exposed external to the enterprise. For simplicity, this header was simply removed (blocked) from both requests and responses for both inbound and outbound calls. To create a rule for blocking a header on an outbound call, first navigate to **vsp** → **session-config-pool** → **entry ToTelco** → **header-settings**. Click **Edit blocked-header**.

The screenshot shows the Avaya Aura Configuration interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The main content area is titled 'Configure vspsession-config-poolentry ToTelcoheader-settings'. On the left, a tree view shows the configuration hierarchy: cluster > box:sp-sbc1 > vsp > session-config-pool > entry ToTelco > header-settings. The main configuration area contains a table of header settings:

Header Type	Action
allowed-header	Edit allowed-header
blocked-header	Edit blocked-header
altered-header	Add altered-header
reg-ex-header	Add req-ex-header
header-normalization	Add header-normalization
altered-body	Add altered-body
reg-ex-collector	Add req-ex-collector
apply-allow-block-to	requests-and-responses (apply to requests and responses)
apply-to-allow-block-to-dialog	both (Apply to both inbound and outbound dialogs.)

In the right pane that appears, click **Add**. In the blank field that appears, enter the name of the header to be blocked. Click **OK**. The screen below shows the **P-Site** header blocked for the compliance test.

Configure vsp\session-config-poolentry ToTelco\header-settings blocked-header

X

The list of blocked headers for outbound calls will appear in the right pane as shown below. Click **Set** to complete the configuration.

The screenshot shows the Avaya Aura Configuration interface. The main title is "Configuration". The breadcrumb trail is "Home > Configuration > Status > Call Logs > Event Logs > Actions > Services > Keys > Access > Tools". The current page is "Configure vsp\session-config-poolentry ToTelco\header-settings".

On the left, there is a navigation tree under "Configuration: all":

- cluster
 - box:sp-sbc1
- vsp
 - default-session-config
 - tls
 - session-config-pool
 - entry ToTelco
 - sip-settings
 - to-uri-specification
 - from-uri-specification
 - request-uri-specification
 - p-asserted-identity-uri-s
 - contact-uri-settings-in-l
 - contact-uri-settings-out
 - header-settings
 - entry ToPBX
 - entry Discard
 - dial-plan
 - enterprise
 - dns
 - settings

The main content area shows the configuration for "blocked-header" with the value "P-Site". Other configuration options include:

- allowed-header: Edit allowed-header
- altered-header: Add altered-header
- reg-ex-header: Add reg-ex-header
- header-normalization: Add header-normalization
- altered-body: Add altered-body
- reg-ex-collector: Add reg-ex-collector
- apply-allow-block-to: requests-and-responses (apply to requests and responses)
- apply-to-allow-block-to-dialog: both (Apply to both inbound and outbound dialogs.)

To create a rule for blocking a header on an inbound call, first navigate to **vsp** → **session-config-pool** → **entry ToPBX** → **header-settings**, then repeat the procedure described earlier in this section. The list of blocked headers for inbound calls is shown below.

The screenshot shows the Avaya Aura Configuration web interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The main content area is titled 'Configure vsp|session-config-pool|entry ToPBX|header-settings'. On the left, a tree view shows the configuration hierarchy: cluster > vsp > session-config-pool > entry ToPBX > header-settings. The main configuration area contains a table of header settings:

Header Type	Value	Action
allowed-header		Edit allowed-header
blocked-header	P-Site	Edit blocked-header
altered-header		Add altered-header
reg-ex-header		Add reg-ex-header
header-normalization		Add header-normalization
altered-body		Add altered-body
reg-ex-collector		Add reg-ex-collector
apply-allow-block-to	requests-and-responses	(apply to requests and responses)
apply-to-allow-block-to-dialog	both	(Apply to both inbound and outbound dialogs.)

6.2.3. Max-Forwards Value

On incoming PSTN calls to a enterprise SIP phone, the Max-Forwards value in the incoming SIP INVITE is too small to allow the message to traverse all the SIP hops internal to the enterprise to reach the SIP phone. Thus, the SBC was used to increase this value when the INVITE arrived at the SBC from the network. To do this, navigate to **vsp** → **session-config-pool** → **entry ToPBX** → **header-settings** and click **Add altered-header**.

The screenshot displays the Avaya Aura Configuration web interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The main content area is titled 'Configure vspsession-config-poolentry ToPBXheader-settings' and features a 'Show advanced' button and links for 'Help' and 'Index'. Below the title are 'Set', 'Reset', 'Back', and 'Delete' buttons. The interface is divided into two main sections: a left-hand navigation tree and a right-hand configuration table.

Configuration: all

- Configuration
- Setup
- View

Configuration Tree:

- cluster
 - box:sp-sbc1
- vsp
 - default-session-config
 - tls
 - session-config-pool
 - entry ToTelco
 - entry ToPBX
 - to-uri-specification
 - request-uri-specification
 - contact-uri-settings-in-leg
 - contact-uri-settings-out-leg
 - header-settings
 - entry Discard
 - dial-plan
 - enterprise
 - dns
 - settings

Configuration Table:

allowed-header	Edit allowed-header
blocked-header	<input type="text" value="P-Site"/> Edit blocked-header
altered-header	Add altered-header
reg-ex-header	Add req-ex-header
header-normalization	Add header-normalization
altered-body	Add altered-body

In the right pane that appears, enter the following in the fields specified below.

- **number:** Enter an unique number for this altered header.
- **source-header:** Specify the header from which the system initially derives the data that is to be written to the destination header. In this case, enter *Max-Forwards*.
- **source-field type:** Enter *selection*. If *selection* is chosen, then the user may enter a value to match on and a replacement value.
- **source-field value:** Enter *.** as the value. This is a regular expression that allows the system to match on any value.
- **source-field replacement:** Enter the replacement value. In this case, the value of *70* was used.
- **destination:** Specify the destination header. In this case, enter *Max-Forwards*.
- **destination-field:** Enter *full*. This specifies that the full destination header will be over-written with the new one that was derived from the source header.

Click the **Create** button.

The screenshot shows the Avaya Aura Configuration interface. The main title is "Configuration". The breadcrumb trail is "Home > Configuration > Status > Call Logs > Event Logs > Actions > Services > Keys > Access > Tools". The page title is "Create vsp\session-config-pool\entry ToPBX\header-settings\altered-header 0 - Step 1 of 1: Edit altered-header 0". The page content includes a form with the following fields:

- * number: 1
- * source-header: enter Max-Forwards or select from <Not configured>
- * source-field:
 - * type: selection (Regular expression based selection of portion of the URI.)
 - * value: .* (regular expression)
 - * replacement: 70
- * destination: enter Max-Forwards or select from <Not configured>
- * destination-field:
 - * type: full (Entire value of the URI.)

Buttons: Create, Reset, Cancel

The right pane then displays the newly created altered header with default values for all other fields. Click the **Set** button on this page to complete the configuration.

The screenshot shows the Avaya Aura Configuration web interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The main content area is titled 'Configure vspsession-config-poolentry ToPBX\header-settings\altered-header 1'. On the left, a tree view shows the configuration hierarchy: cluster > box:sp-sbc1 > vsp > default-session-config > tls > session-config-pool > entry ToPBX > header-settings > altered-header 1. The right pane contains the configuration form with the following fields:

- admin**: enabled (Resource is active)
- * number**: 1
- * source-header**: enter Max-Forwards or select from Max-Forwards
- * source-field**:
 - * type: selection (Regular expression based selection of portion of the URI.)
 - * value: .* (regular expression)
 - * replacement: 70
- * destination**: enter Max-Forwards or select from Max-Forwards
- * destination-field**:
 - * type: full (Entire value of the URI.)
- apply-to-methods**: INVITE, REFER, MESSAGE, INFO (with Select All and Unselect All buttons)
- apply-to-responses**: * type: no (Do not apply to responses (requests only))
- apply-to-dialog**: both (Apply to both inbound and outbound dialogs.)
- session-persistent**: disabled (Resource is inactive)

6.2.4. Third Party Call Control

Disable third party call control. Navigate to **vsp** → **default-session-config** → **third-party-call-control**. Set the **admin** field to *disabled*. Disabling **third-party-call-control** will impact customers enabling this parameter to provide protocol conversion between the Microsoft usCSTA interface and the Broadworks Open Client Interface.

The screenshot shows the Avaya Aura Configuration interface. The breadcrumb path is **vsp** → **default-session-config** → **third-party-call-control**. The configuration parameters are as follows:

Parameter	Value	Notes
admin	disabled	(Resource is inactive)
status-events	both	(both call-legs)
handle-refer-locally	enabled	(Resource is active)
refer-maintain-identity	false	
ringback-file		Browse System Files
busy-file		Browse System Files
pre-call-announcement		Browse System Files

6.2.5. Contact Header

Using the settings chosen in the installation wizard, the SBC does not automatically pass to the service provider the updated Contact header that results from a redirected call. In order to have the updated Contact header passed to the service provider, first navigate to **vsp** → **session-config-pool** → **entry ToPBX**. Scroll down to the **uri** section and click **Configure** next to **contact-uri-settings-in-leg**.

The screenshot shows the Avaya Aura Configuration interface. The breadcrumb path is **vsp** → **session-config-pool** → **entry ToPBX**. The configuration parameters are as follows:

Parameter	Action
to-uri-specification	[Delete]
from-uri-specification	Configure
request-uri-specification	[Delete]
p-asserted-identity-uri-specification	Configure
contact-uri-settings-in-leg	Configure
contact-uri-settings-out-leg	Configure

In the right pane that appears, set the **add-maddr** field to *disabled* and the **use-incoming-contact** field to *enabled*.

The screenshot shows the Avaya Aura Configuration web interface. The main title is "Configuration". The navigation menu includes Home, Configuration, Status, Call Logs, Event Logs, Actions, Services, Keys, Access, and Tools. The left pane shows a tree view under "Configuration: all" with the following structure:

- cluster
 - box:sp-sbc1
- vsp
 - default-session-config
 - tls
 - session-config-pool
 - entry ToTelco
 - entry ToPBX
 - entry Discard
 - dial-plan
 - enterprise
 - dns
 - settings

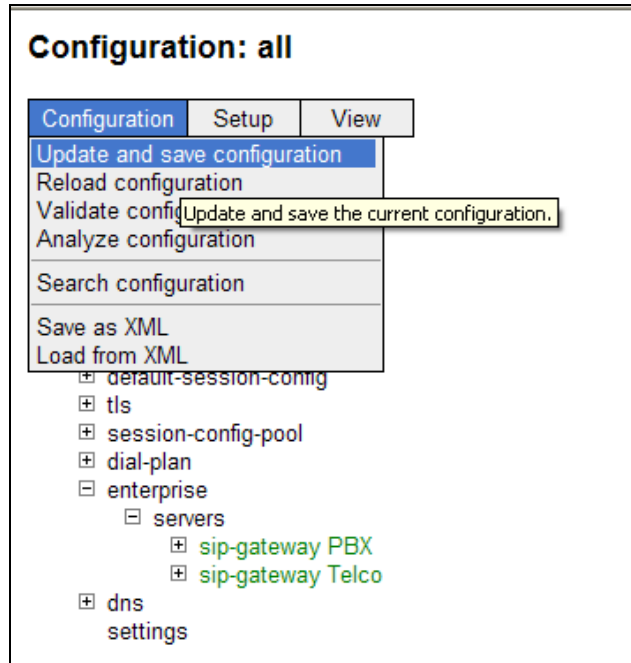
The right pane is titled "Configure vspsession-config-poolentry ToPBX\contact-uri-settings-in-leg". It contains a table of configuration fields:

Field	Value	Notes
user	enter <input type="text" value="contact-uri"/> or select from <input type="text" value="contact-uri"/> (Net-Net OS-E uses the value from the incoming CONTACT URI.)	
host	enter <input type="text" value="CXC-address"/> or select from <input type="text" value="CXC-address"/> (Net-Net OS-E uses the IP address of the Net-Net OS-E's local interface.)	
port	enter <input type="text" value="CXC-local-port"/> or select from <input type="text" value="CXC-local-port"/> (Net-Net OS-E uses the port number of the Net-Net OS-E's local interface.)	
transport	<input type="text" value="next-hop-transport"/> (Net-Net OS-E uses the transport type of the next-hop server.)	
add-maddr	<input type="text" value="disabled"/> (Resource is inactive)	
use-incoming-contact	<input type="text" value="enabled"/> (Resource is active)	
from-user-contact-uri	<input type="text" value="disabled"/> (Resource is inactive)	

Use the same procedure described in this section to set these same values for the **contact-uri-settings-out-leg**. Repeat again for the **contact-uri-settings-in-leg** and **contact-uri-settings-out-leg** of the ToTelco session-config-pool by navigating to **vsp** → **session-config-pool** → **entry ToTelco**.

6.2.6. Save the Configuration

To save the configuration, begin by clicking on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



7. XO SIP Trunking Configuration

To use XO SIP Trunking, a customer must request the service from XO using their sales processes. The process can be started by contacting XO via the corporate web site at www.xo.com and requesting information via the online sales links or telephone numbers.

During the signup process, XO will require that the customer provide the public IP address used to reach the SBC at the edge of the enterprise. XO will provide the IP address of the XO SIP proxy/SBC (XO Sonus NBS), IP addresses of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete the Communication Manager, Session Manager, and the SBC configuration discussed in the previous sections.

The configuration between XO and the enterprise is a static configuration. There is no registration of the SIP trunk or enterprise users to the XO network.

8. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Communication Manager, Session Manager and the SBC to connect to XO SIP Trunking. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 1.1**.

XO SIP Trunking passed compliance testing.

9. 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 that can be used to troubleshoot the solution.

Verification Steps:

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
3. Verify that the user on the PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

Troubleshooting:

1. Communication Manager:
 - **list trace station** <extension number> - Traces calls to and from a specific station.
 - **list trace tac** <trunk access code number> - Trace calls 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 access code number> - Displays trunk group information.
 - **status trunk** <trunk access code number/channel number> - Displays signaling and media information for an active trunk channel.
2. Session Manager:
 - **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, navigate to **Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.
 - **traceSM -x** – Session Manager command line tool for traffic analysis. Login to the Session Manager management interface to run this command.
3. Session Border Controller:
 - **Call Logs** - On the element manager user interface of the SBC, the **Call Logs** tab can provide useful diagnostic or troubleshooting information.

10. Conclusion

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

11. 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, June 2010.
- [2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.
- [3] *Administering Avaya Aura® Communication Manager*, May 2009, Document Number 03-300509.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, May 2009, Document Number 555-245-205.
- [5] *Installing and Upgrading Avaya Aura® System Manager 5.2 GA Version*, January 2010.
- [6] *Installing Avaya Aura® Session Manager*, January 2010.
- [7] *Administering Avaya Aura® Session Manager*, March 2010, Document Number 03-603324.
- [8] *Avaya 1600 Series IP Deskphones Administrator Guide Release 1.2.x*, February 2010, Document Number 16-601443.
- [9] *4600 Series IP Telephone LAN Administrator Guide*, October 2007, Document Number 555-233-507.
- [10] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide*, November 2009, Document Number 16-300698.
- [11] *Avaya one-X Communicator Getting Started*, November 2009.
- [12] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [13] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>
- [14] RFC 4244, *An Extension to the Session Initiation Protocol (SIP) for Request History Information*, <http://www.ietf.org/>

12. Appendix A: Avaya Aura® SBC Configuration File

```
#
# Copyright (c) 2004-2010 Acme Packet Inc.
# All Rights Reserved.
#
# File: /cxc/cxc.cfg
# Date: 12:16:35 Thu 2010-09-16
#
config cluster
config box 1
  set hostname sp-sbc1
  set timezone America/New_York
  set name sp-sbc1
  set identifier 00:ca:fe:09:42:38
config interface eth0
  config ip inside
    set ip-address static 10.32.128.12/24
  config ssh
  return
  config snmp
    set trap-target 10.32.128.11 162
    set trap-filter generic
    set trap-filter dos
    set trap-filter sip
    set trap-filter system
  return
  config web
  return
  config web-service
    set protocol https 8443
    set authentication certificate "vsp\tls\certificate ws-cert"
  return
  config sip
    set udp-port 5060 "" "" any 0
    set tcp-port 5060 "" "" any 0
    set tls-port 5061 "" "" any 0
  return
  config icmp
  return
  config media-ports
  return
  config routing
    config route Default
      set gateway 10.32.128.254
    return
    config route Static0
      set destination network 192.11.13.4/30
      set gateway 10.32.128.10
    return
    config route Static1
      set admin disabled
    return
    config route Static2
      set admin disabled
```

```

return
config route Static3
  set admin disabled
return
config route Static4
  set admin disabled
return
config route Static5
  set admin disabled
return
config route Static6
  set admin disabled
return
config route Static7
  set admin disabled
return
config route internal-sip-media
  set destination host 10.32.24.235
  set gateway 10.32.128.254
return
return
return
return
config interface eth2
config ip outside
  set ip-address static 10.5.5.191/24
config sip
  set udp-port 5060 "" "" any 0
  set tcp-port 5060 "" "" any 0
  set tls-port 5061 "" "" any 0
return
config media-ports
return
config routing
  config route Default
    set admin disabled
  return
  config route external-sip-media
    set destination network 10.2.2.0/24
    set gateway 10.5.5.254
  return
return
return
return
config cli
  set prompt sp-sbc1
return
config os
return
return
return
return

config services
config event-log
  config file access
  set filter access info

```

```
return
config file system
  set filter general info
  set filter system info
return
config file errorlog
  set filter all error
return
config file db
  set filter db debug
  set filter dosDatabase info
return
config file management
  set filter management info
return
config file peer
  set filter sipSvr info
return
config file cac
  set filter sipCAC warning
return
config file dos
  set filter dos alert
  set filter dosSip alert
  set filter dosTransport alert
  set filter dosUrl alert
return
config file krnlsys
  set filter krnlsys debug
return
config file acct
  set filter acct debug
return
return
return

config master-services
config accounting
return
config database
  set media enabled
return
return

config vsp
set admin enabled
config default-session-config
config media
  set anchor enabled
  set rtp-stats enabled
return
config sip-directive
  set directive allow
return
config log-alert
  set apply-to-methods-for-filtered-logs
```

```

return
config header-settings
return
config third-party-call-control
return
return
config tls
  config certificate ws-cert
    set certificate-file /cxc/certs/ws.cert
  return
return
config session-config-pool
  config entry ToTelco
    config sip-settings
    return
    config to-uri-specification
      set host next-hop
    return
    config from-uri-specification
      set host local-ip
    return
    config request-uri-specification
      set host next-hop
    return
    config p-asserted-identity-uri-specification
      set host local-ip
    return
    config contact-uri-settings-in-leg
      set add-maddr disabled
      set use-incoming-contact enabled
    return
    config contact-uri-settings-out-leg
      set add-maddr disabled
      set use-incoming-contact enabled
    return
    config header-settings
      set blocked-header P-Site
    return
  return
  config entry ToPBX
    config to-uri-specification
      set host next-hop-domain
    return
    config request-uri-specification
      set host next-hop-domain
    return
    config contact-uri-settings-in-leg
      set add-maddr disabled
      set use-incoming-contact enabled
    return
    config contact-uri-settings-out-leg
      set add-maddr disabled
      set use-incoming-contact enabled
    return
    config header-settings
      set blocked-header P-Site

```

```

config altered-header 1
  set source-header Max-Forwards
  set source-field selection .* 70
  set destination Max-Forwards
  set destination-field full
return
config reg-ex-header 1
  set destination Refer-To
  set create Refer-To "<sip:(.*)@avaya\.com(.*)>" "<sip:\1@\r\2>"
  set apply-to-methods REFER
return
return
config entry Discard
  config sip-directive
return
return
return
config dial-plan
config route Default
  set priority 500
  set location-match-preferred exclusive
  set session-config vsp\session-config-pool\entry Discard
return
config source-route FromTelco
  set peer server "vsp\enterprise\servers\sip-gateway PBX"
  set source-match server "vsp\enterprise\servers\sip-gateway Telco"
return
config source-route FromPBX
  set peer server "vsp\enterprise\servers\sip-gateway Telco"
  set source-match server "vsp\enterprise\servers\sip-gateway PBX"
return
return
config enterprise
config servers
  config sip-gateway PBX
    set domain avaya.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToPBX
  config server-pool
    config server PBX1
      set host 10.32.24.235
      set transport TCP
    return
  return
return
  config sip-gateway Telco
    set failover-detection ping
    set ping-interval 60
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
  config server-pool
    config server Telco1
      set host 10.2.2.138
    return

```

```
    return
  return
  return
return
config dns
  config resolver
  config server 10.32.24.150
  return
  return
return
config settings
  set stack-socket-threads-max 2
return
return

config external-services
return

config preferences
  config gui-preferences
  set enum-strings SIPSourceHeader Refer-To
  set enum-strings SIPSourceHeader Max-Forwards
  return
return

config access
  config permissions superuser
  set cli advanced
  return
  config permissions read-only
  set config view
  set actions disabled
  return
  config users
  config user admin
  set password 0x002bdd5d9fea2fefeb97b0115854a47db2c8b27a2fe0187e0274977f4b
  set permissions access\permissions superuser
  return
  config user cust
  set password 0x004803cd9fae4ee1b2462598359d6c5e179008f9083caa7b30b9b19b43
  set permissions access\permissions read-only
  return
  return
return

config features
return
```

13. Appendix B: Workaround for Double DTMF Digit Detection

This section describes the steps to enable a firmware workaround to address the condition when DTMF tones are received both in-band and out-of-band but are not properly aligned with each other. **Steps 1 and 2** describe the procedure if a TN2602 MedPro circuit pack is used. **Step 3** describes the procedure for the G450 and G430 gateways. This firmware workaround will only be applied for trunks that have enabled use of RFC2833 for DTMF transmission (i.e., the **DTMF over IP** field is set to *rtp-payload* on the Avaya Communication Manager signaling form). This procedure requires logging into the circuit pack directly via SSH or Telnet. The ability to access the circuit pack directly must first be enabled through the Avaya Communication Manager SAT interface.

Step	Description
1.	<p>Enable Session for TN2602 In order to log into the TN2602 Circuit Pack, first enable this capability by using the enable session command on the Avaya Communication Manager SAT interface. Set the parameters as described below.</p> <ul style="list-style-type: none"> ▪ Login: Create a login name for use when logging into the circuit pack. ▪ Password: Create a password for this login name. ▪ Reenter Password: Enter the password again. ▪ Secure?: Select <i>n</i> for Telnet or <i>y</i> for SSH. ▪ Time to login: Enter the number of minutes this login will be valid. The maximum value is 255. ▪ Board address: Enter the cabinet/carrier/slot location for the circuit pack that will be accessed. This value can be found from the list configuration all command. <div style="border: 1px solid black; padding: 10px; margin-top: 10px;"> <pre style="font-family: monospace;"> enable session Page 1 of 1 ENABLE SESSION Login: user Password: Reenter Password: Secure? n Time to login: 200 Board address: 1a03 </pre> </div>

Step	Description
2.	<p>Login and Set Parameters</p> <p>Log in to the TN2602 IP address using either Telnet or SSH as defined in the previous step. The IP address can be found using the list ip-interface all command. When prompted, enter the Login and Password as defined in the previous step. At the command prompt, enter the following three commands.</p> <ul style="list-style-type: none"> ▪ setVoipParam 60,1 - Sets up a temporary buffer with VOIP parameter 60 set to value 1. This parameter enables the firmware workaround. ▪ sendVoipParams - Sends any VOIP parameters to the DSPs. ▪ saveVoipParams - Save parameters to flash memory so the configuration will survive a reset. <div style="border: 1px solid black; padding: 10px; margin: 10px 0;"> <pre> Enter Login ID: user Password: SIMPLEX-> setVoipParam 60,1 value = 0 = 0x0 SIMPLEX-> sendVoipParams value = 0 = 0x0 SIMPLEX-> saveVoipParams value = 0 = 0x0 SIMPLEX-> </pre> </div>

Step	Description
3.	<p>G450/430 Gateway</p> <p>If a G450/G430 is used in the configuration, then log in to the gateway using proper credentials and issue the following command shown in bold below.</p> <ul style="list-style-type: none"> ▪ voip-parameters – Enter the VoIP parameters configuration mode. ▪ set id 60 value 1 - Sets up a temporary buffer with VOIP parameter 60 set to value 1. This parameter enables the firmware workaround. ▪ dsp-downlink - Sends any VOIP parameters to the DSPs. ▪ Exit – Exit the VoIP parameter mode. ▪ copy run start - Save parameters to flash memory so the configuration will survive a reset. <pre> sp3-g450-001(super)# voip-parameters Warning: The values chosen for non-default voip parameters can significantly affect the quality of service that users experience. Avaya recommends seeking technical assistance from Avaya before making any modifications to the voip parameter defaults. sp3-g450-001(super-voip-parameters)# set id 60 value 1 Done! sp3-g450-001(super-voip-parameters)# dsp-downlink Done! sp3-g450-001(super-voip-parameters)# exit sp3-g450-001(super)# copy run start Warning! It is a recommended policy to override default configuration master key with user defined secret - for details see user reference. Otherwise device saves configuration secrets using Avaya default secret. Beginning copy operation Done! sp3-g450-001(super)# </pre>

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