



## **Application Notes for Configuring Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0 and Avaya Session Border Controller for Enterprise Rel. 7.2 to support Telmex SIP Trunk Service – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunk Service on an enterprise solution consisting of Avaya Aura® Communication Manager Rel. 7.0, Avaya Aura® Session Manager Rel. 7.0 and Avaya Session Border Controller for Enterprise Rel. 7.2, to interoperate with the Telmex SIP Trunk service.

The Telmex SIP Trunk service provide customers with PSTN access via a SIP trunk between the enterprise and the Telmex network, as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 Telmex facility in Mexico.

## Table of Contents

1.	Introduction.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing.....	5
2.2.	Test Results .....	6
2.3.	Support .....	6
3.	Reference Configuration .....	7
4.	Equipment and Software Validated .....	10
5.	Configure Avaya Aura® Communication Manager .....	11
5.1.	Licensing and Capacity .....	11
5.2.	System Features.....	12
5.3.	IP Node Names.....	14
5.4.	Codecs .....	15
5.5.	IP Network Regions .....	16
5.6.	Signaling Group .....	17
5.7.	Trunk Group.....	19
5.8.	Calling Party Information.....	23
5.9.	Inbound Routing.....	24
5.10.	Outbound Routing .....	25
6.	Configure Avaya Aura® Session Manager .....	29
6.1.	System Manager Login and Navigation.....	30
6.2.	SIP Domain .....	31
6.3.	Locations .....	32
6.4.	Adaptations.....	33
6.5.	SIP Entities .....	34
6.6.	Entity Links .....	37
6.7.	Routing Policies .....	39
6.8.	Dial Patterns .....	40
7.	Configure Avaya Session Border Controller for Enterprise .....	43
7.1.	System Access.....	43
7.2.	System Management .....	44
7.3.	Network Management .....	45
7.4.	Media Interfaces .....	46
7.5.	Signaling Interfaces.....	48
7.6.	Server Interworking.....	50
7.6.1.	Server Interworking Profile – Enterprise.....	50
7.6.2.	Server Interworking Profile – Service Provider.....	53
7.7.	Signaling Manipulation .....	54
7.8.	Server Configuration .....	55
7.8.1.	Server Configuration Profile – Enterprise .....	55
7.8.2.	Server Configuration Profile – Service Provider .....	57
7.9.	Routing .....	60
7.9.1.	Routing Profile – Enterprise .....	60

7.9.2.	Routing Profile – Service Provider .....	61
7.10.	Topology Hiding.....	62
7.10.1.	Topology Hiding Profile – Enterprise .....	62
7.10.2.	Topology Hiding Profile – Service Provider.....	63
7.11.	Domain Policies.....	64
7.11.1.	Application Rules.....	64
7.11.2.	Media Rules.....	65
7.11.3.	Signaling Rules .....	67
7.12.	End Point Policy Groups .....	67
7.12.1.	End Point Policy Group – Enterprise .....	67
7.12.2.	End Point Policy Group – Service Provider.....	69
7.13.	End Point Flows.....	70
7.13.1.	End Point Flow – Enterprise .....	71
7.13.2.	End Point Flow – Service Provider .....	72
8.	Telmex SIP Trunk Service Configuration .....	73
9.	Verification and Troubleshooting .....	73
9.1.	General Verification Steps .....	73
9.2.	Communication Manager Verification.....	73
9.3.	Session Manager Verification .....	74
9.4.	Avaya SBCE Verification .....	75
10.	Conclusion .....	80
11.	References.....	80
12.	Appendix A: SigMa Script.....	81

# 1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) Trunk Service between the Telmex network and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Communication Manager Rel. 7.0 (Communication Manager), Avaya Aura® Session Manager Rel. 7.0 (Session Manager), Avaya Session Border Controller for Enterprise Rel. 7.2 (Avaya SBCE) and various Avaya endpoints, listed in **Section 4**.

The Telmex SIP Trunk service referenced within these Application Notes is designed for business customers. Customers using this service with this Avaya enterprise solution 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 analog and/or ISDN-PRI.

The terms “Service Provider” and “Telmex” will be used interchangeably throughout these Application Notes.

## 2. General Test Approach and Test Results

The general test approach was to simulate an enterprise site at the Telmex facility in Mexico by connecting the Avaya SIP-enabled enterprise solution to Telmex’s network, as depicted in **Figure 1**.

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

## 2.1. Interoperability Compliance Testing

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

- Static IP SIP Trunk authentication.
- Incoming PSTN calls to various phone types. Phone types included H.323, SIP, digital, and analog telephones at the enterprise. All inbound calls from the PSTN 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 calls to the PSTN were routed from the enterprise across the SIP trunk via the service provider network.
- Inbound and outbound PSTN calls to/from Avaya one-X® Communicator softphones using “This Computer” and “Other Phone” modes. (H.323, SIP).
- Inbound and outbound PSTN calls to/from Avaya Equinox softphones (SIP).
- Codec G.729, currently the only codec supported by Telmex.
- DTMF tones passed as out-of-band RTP events as per RFC 2833.
- Caller ID presentation and Caller ID restriction.
- Voicemail redirection and navigation.
- User features such as hold and resume, transfer and conference.
- Off-net call transferring, call forwarding and mobility (extension to cellular).
- Routing inbound PSTN calls to call center agent queues.
- Proper response/error treatment to different failure conditions.

The following items were not tested:

- Remote Worker was not tested.
- Inbound toll-free calls, 911 calls (emergency), “0” calls (Operator), 0+10 digits calls (Operator Assisted), were not tested.
- G.711 pass-through fax was not tested.
- T.38 fax is not currently supported by Telmex; therefore T.38 fax was not tested.
- SIP REFER is not currently supported by Telmex; therefore SIP REFER was not tested.

## 2.2. Test Results

Interoperability testing of the Telmex SIP Trunk Service with the Avaya SIP-enabled enterprise solution was completed with the observations/limitations noted below:

- **DTMF tones on H.323 end-points not recognized** – When attempting to login into PSTN voicemail systems from the enterprise using H.323 endpoints the DTMF tones were not recognized. This issue was solved by Telmex with the addition of “midcall-signaling block” to the Cisco CUBE configuration to allow DTMF pass-through. This issue was only seen on H.323 endpoints, this behavior was not seen on SIP endpoints.
- **Outbound Calling Party Number (CPN) Block:** To support outbound privacy calls (calling party number blocking), Communication Manager sends “anonymous” as the calling number in the SIP From header, uses the P-Asserted-Identity (PAI) header to pass the actual calling party number and includes “Privacy: id” in the INVITE. During testing this type of call resulted in call failure. For security reasons this feature may not be supported in Mexico.
- **SIP header optimization** – There are multiple SIP headers and parameters used by Communication Manager and Session Manager, some of them Avaya proprietary, that had no significance in the service provider’s network. These headers were removed with the purposes of blocking enterprise information from being propagated outside of the enterprise boundaries, to reduce the size of the packets entering the service provider’s network and to improve the solution interoperability in general. The following headers were removed from outbound messages using an Adaptation in Session Manager: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector and P-Location (**Section 6.4**).

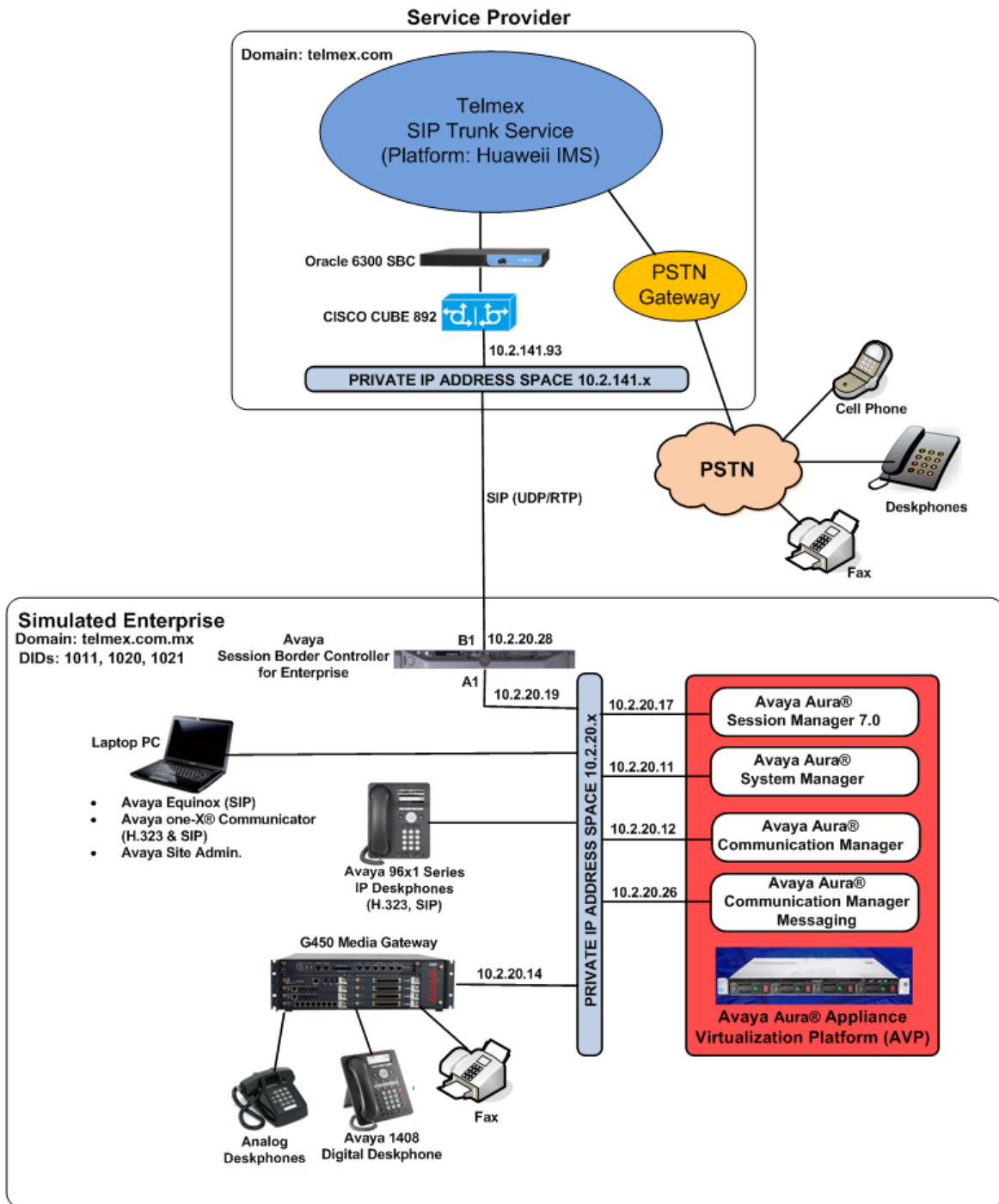
## 2.3. Support

For support on Telmex SIP Trunk Service visit the corporate Web page at:

<http://telmex.com/en/web/empresas>

### 3. Reference Configuration

**Figure 1** illustrates the sample Avaya SIP-enabled enterprise solution, connected to the Telmex SIP Trunk Service through a private connection.



**Figure 1: Avaya SIP Enterprise Solution connected to Telmex SIP Trunk Service**

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

- Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager.
- Avaya Session Border Controller for Enterprise.
- Avaya Aura® Communication Manager Messaging (CMM).
- Avaya G450 Media Gateway.
- Avaya 96x1 Series IP Deskphones (H.323 and SIP).
- Avaya one-X® Communicator softphones (H.323 and SIP).
- Avaya Equinox softphone (SIP).
- Avaya digital and analog telephones.

The Avaya SBCE was located at the edge of the enterprise. Its public side was connected to the Telmex private network, while its private side was connected to the enterprise infrastructure. All signaling and media traffic entering or leaving the enterprise flowed through the Avaya SBCE, protecting in this way the enterprise against any SIP-based attacks. The Avaya SBCE also performed network address translation at both the IP and SIP layers.

For inbound calls, the calls flowed from Telmex's network to the Avaya SBCE then to Session Manager. Session Manager used the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case Communication Manager) and on which link to send the call. Once the call arrived at Communication Manager, further incoming call treatment, such as incoming digit translation was performed.

Outbound calls to the PSTN were first processed by Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Communication Manager selected the proper SIP trunk, the call was routed to Session Manager. Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to the Avaya SBCE for egress to the Telmex's network.

A separate SIP trunk was created between Communication Manager and Session Manager to carry the service provider traffic. This was done so that any trunk or codec settings required by the service provider could be applied only to this trunk without affecting other enterprise SIP traffic. This trunk carried both inbound and outbound traffic.

As part of the Avaya Aura® version 7.0 release, Communication Manager incorporates the ability to use the Avaya Aura® Media Server (AAMS) as a media resource. The AAMS is a software-based, high density media server that provides DSP resources for IP-based sessions. Only Media resources from a G450 Media Gateway were utilized during the compliance test, the AAMS was not included as part of the compliance test, it's mentioned here simply as an option that could be used on customer deployments.

The Avaya Aura® Communication Manager Messaging (CMM) was used during the compliance test to verify voice mail redirection and navigation, as well as the delivery of Message Waiting



Indicator (MWI) messages to the enterprise telephones. Since the configuration tasks for Messaging are not directly related to the interoperability tests with the Telmex network SIP Trunk service, they are not included in these Application Notes.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products only (private network side). Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the compliance testing associated with this Application Note, the interface between the Avaya system and the Telmex network did not include the use of any specific encryption features, UDP Transport for signaling and RTP for media was used between the Avaya system and the Telmex network across the SIP Trunk. TLS transport for signaling and SRTP for media was used inside of the enterprise (private network side, in between Avaya components).

<p><b>Note</b> – The configuration tasks required to support security and encryption capabilities inside of the enterprise (private network side) are beyond the scope of these Application Notes; hence they are not discussed in this document.</p>
---

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
<b>Avaya</b>	
Avaya Aura® Appliance Virtualization Platform (AVP)	avaya-avp-7.0.1.0.0.5
Avaya Aura® Communication Manager	CM 7.0.1.2.0.441.23523
Avaya Aura® Session Manager	7.0.1.2.701230
Avaya Aura® System Manager	System Manager 7.0.1.2 Build No. - 7.0.0.0.16266 Software Update Revision No: 7.0.1.2.086007 Service Pack 2
Avaya Session Border Controller for Enterprise running on a DELL R210 V2 Server	ASBCE 7.2 7.2.0.0-18-13712
Avaya Aura® Communication Manager Messaging (CMM)	CMM 7.0.0.1.441.1
Avaya G450 Media Gateway	37.41.0
Avaya 96x1 Series IP Deskphones (SIP)	Version R7_1_0_1-061317
Avaya 96x1 Series IP Deskphones (H.323)	Version R6_6_4_01-102616
Avaya one-X® Communicator (H.323, SIP)	6.2.12.04-SP12
Avaya Equinox (SIP)	3.1.1.18
Avaya 1408 Series Digital Deskphones	46.0
Avaya 6210 Analog Deskphones	N/A
<b>Telmex</b>	
Cisco CUBE C892fsp	15.3(3)m7
Oracle SBC	Unknown
Huawei IMS	Unknown

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

**Note** – The Avaya Aura® servers used in the reference configuration, except for the Avaya Session Border Controller for Enterprise, as shown on the previous table, were deployed on an Avaya Aura® Appliance virtualization Platform (AVP). These Avaya components ran as virtual machines on an Avaya-supplied server, over a customized OEM version of VMware® (ESXi 5.5.). Consult the installation documentation on the **References** section for more information. The Avaya Session Border Controller for Enterprise was deployed on a DELL R210 V2 Server.

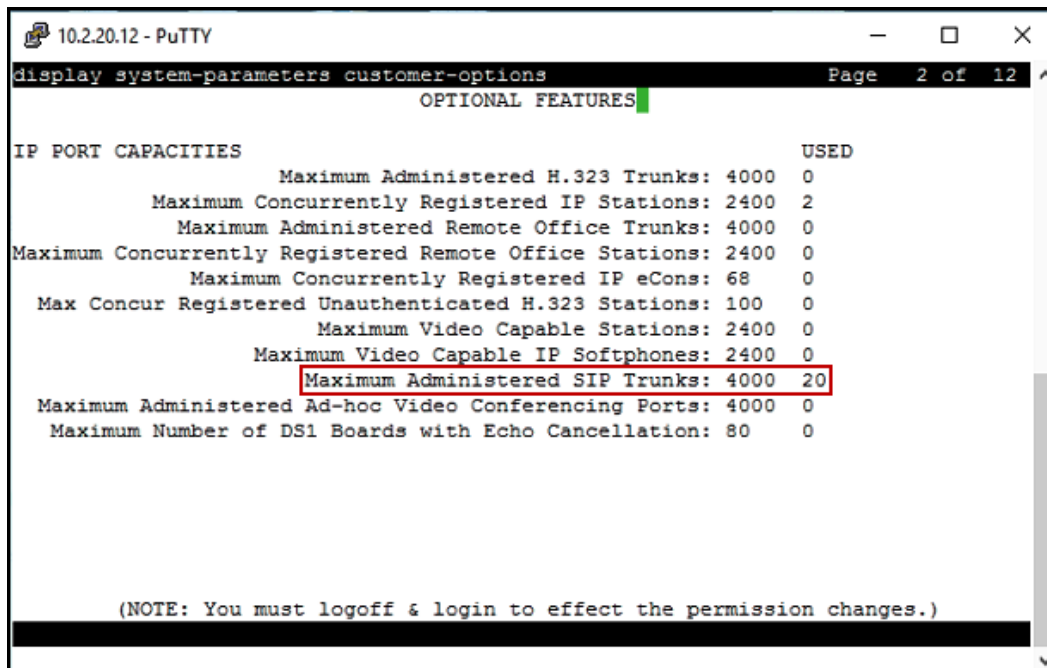
## 5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Communication Manager to work with the Telmex network SIP Trunk service. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from the service provider. It is assumed that the general installation of Communication Manager and the Avaya G450 Media Gateway has been previously completed and is not discussed here.

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. Some screens captures will show the use of the **change** command instead of the **add** command, since the configuration used for the testing was previously added.

### 5.1. Licensing and Capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to and from the service provider. The 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.



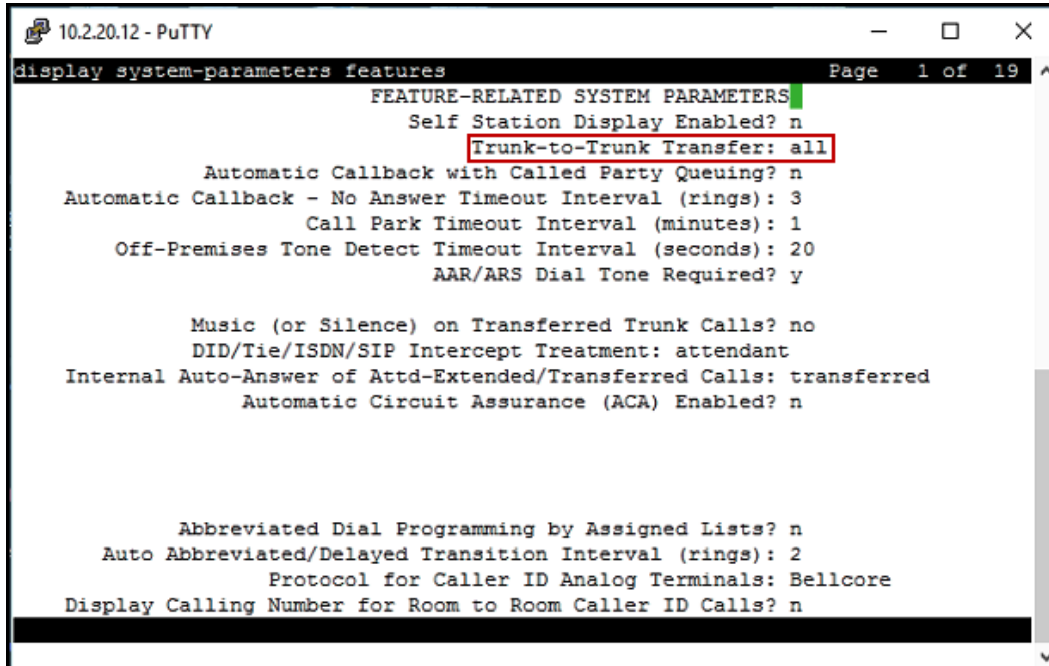
```
10.2.20.12 - PuTTY
display system-parameters customer-options Page 2 of 12
OPTIONAL FEATURES

IP PORT CAPACITIES USED
Maximum Administered H.323 Trunks: 4000 0
Maximum Concurrently Registered IP Stations: 2400 2
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

(NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. System Features

Use the **change system-parameters features** command to set the **Trunk-to-Trunk Transfer** field to **all** to allow 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**.

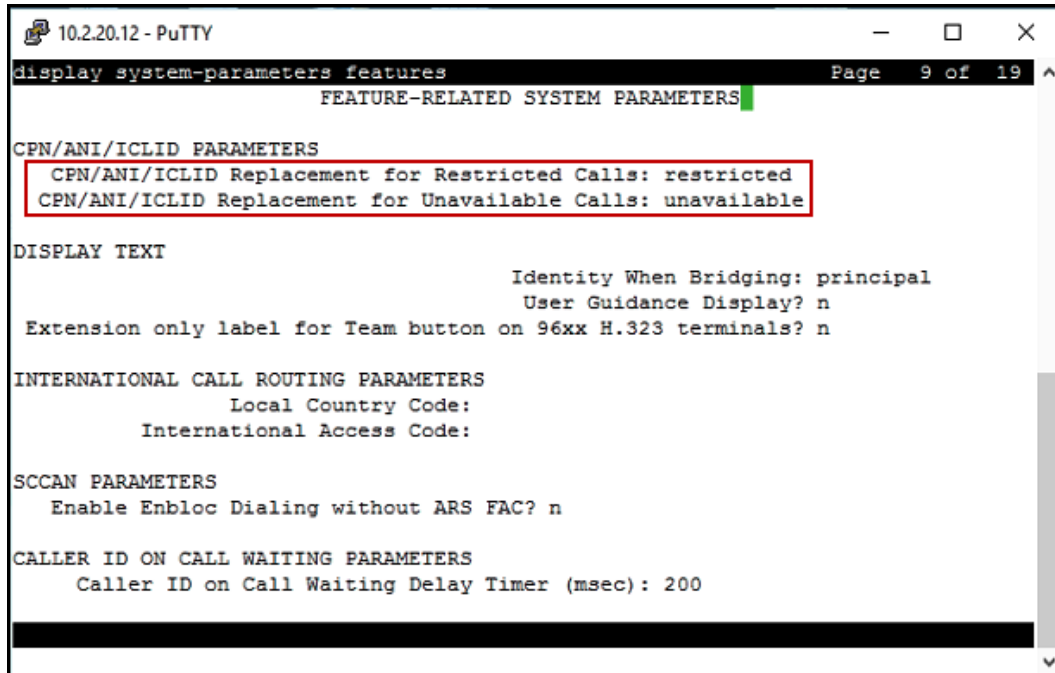


```
10.2.20.12 - PuTTY
display system-parameters features Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS
Self Station Display Enabled? n
Trunk-to-Trunk Transfer: all
Automatic Callback with Called Party Queuing? n
Automatic Callback - No Answer Timeout Interval (rings): 3
Call Park Timeout Interval (minutes): 1
Off-Premises Tone Detect Timeout Interval (seconds): 20
AAR/ARS Dial Tone Required? y

Music (or Silence) on Transferred Trunk Calls? no
DID/Tie/ISDN/SIP Intercept Treatment: attendant
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
Automatic Circuit Assurance (ACA) Enabled? n

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

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 *restricted* for restricted calls and *unavailable* for unavailable calls.



```
10.2.20.12 - PuTTY
display system-parameters features Page 9 of 19
FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
CPN/ANI/ICLID Replacement for Restricted Calls: restricted
CPN/ANI/ICLID Replacement for Unavailable Calls: unavailable

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

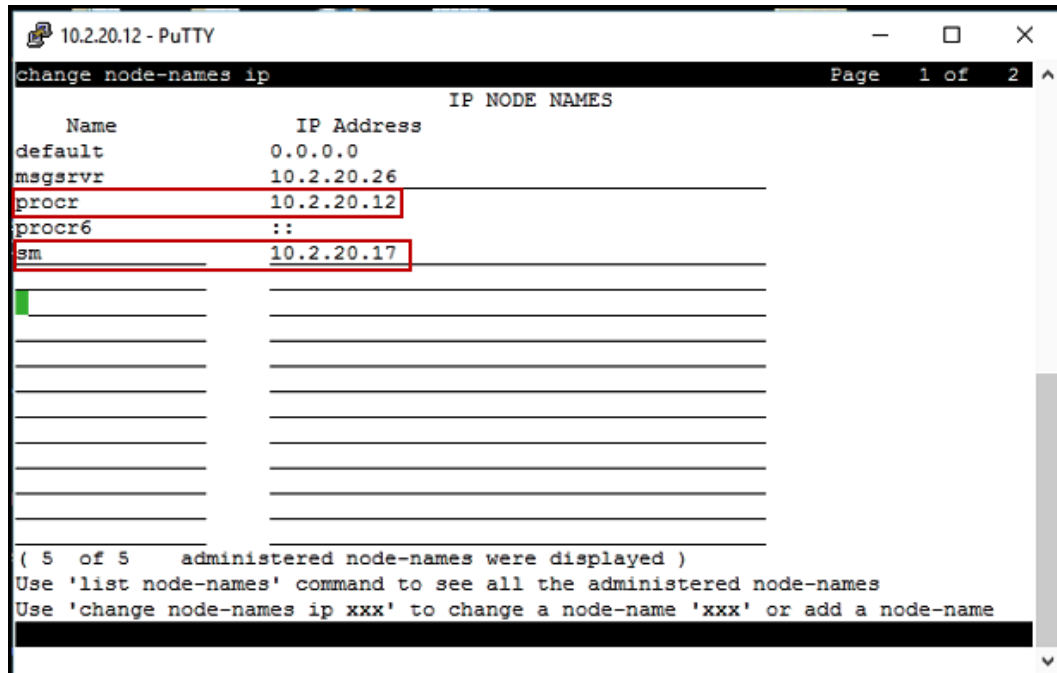
INTERNATIONAL CALL ROUTING PARAMETERS
Local Country Code:
International Access Code:

SCCAN PARAMETERS
Enable Enbloc Dialing without ARS FAC? n

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

### 5.3. IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of Communication Manager (**procr**) and the Session Manager security module (**SM**). These node names will be needed for defining the service provider signaling group in **Section 5.6**.



```
10.2.20.12 - PuTTY
change node-names ip                                     Page 1 of 2
IP NODE NAMES
Name      IP Address
default   0.0.0.0
msgsrvr    10.2.20.26
procr      10.2.20.12
procr6     ::
sm         10.2.20.17
( 5 of 5 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

## 5.4. 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 1 was used for this purpose. Enter the corresponding codec in the **Audio Codec** column of the table. During the compliance test codecs **G.729**, **G.711MU** and **G.711A** were used. Currently Telmex only supports codec G.729 for audio. Set **Media Encryption** to **1-srtp-aescm128-hmac80**, **2-srtp-aescm128-hmac32** and **none**, this value must match the Media Encryption value set under the Avaya SBCE Media Rules, Section 7.11.2. Set **Encrypted SRTCP** to **best-effort**.

10.2.20.12 - PuTTY  
change ip-codec-set 1 Page 1 of 2

IP CODEC SET

Codec Set: 1

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

Media Encryption

1:	1-srtp-aescm128-hmac80
2:	2-srtp-aescm128-hmac32
3:	none
4:	
5:	

Encrypted SRTCP: best-effort

On **Page 2**, set the **Fax Mode** to **off**.

10.2.20.12 - PuTTY  
change ip-codec-set 1 Page 2 of 2

IP CODEC SET

Allow Direct-IP Multimedia? n

	Mode	Redundancy	Packet Size (ms)
FAX	off	0	
Modem	off	0	
TDD/TTY	US	3	
H.323 Clear-channel	n	0	
SIP 64K Data	n	0	20

## 5.5. IP Network Regions

Avaya recommends creating a separate IP network region for the service provider trunk group. 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 1 was used for the service provider trunk and for calls between the enterprise. Use the **change ip-network-region 1** command to configure region 1 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **telmex.com.mx** as assigned to the test environment in the Avaya simulated enterprise. This domain name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Leave both **Intra-region** and **Inter-region IP-IP Direct Audio** set to **yes**, the default setting. This will 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 and Media Server. Shuffling can be further restricted at the trunk level on the Signaling Group form if needed.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.4**.
- Default values may be used for all other fields.

```
10.2.20.12 - PuTTY
change ip-network-region 1 Page 1 of 20
IP NETWORK REGION
Region: 1
Location: [ ] Authoritative Domain: telmex.com.mx
Name: Local Stub Network Region: n
MEDIA PARAMETERS
Codec Set: 1 Intra-region IP-IP Direct Audio: yes
Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048 IP Audio Hairpinning? n
UDP Port Max: 65535
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
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
RSVP Enabled? n
```



On **Page 4**, define the IP codec set to be used for traffic within 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 following example shows the settings used for the compliance test.

Source Region	Inter Network Region Connection Management	I	M	dst rgn	codec set	direct	WAN	Units	WAN-BW-limits	Video	Intervening	Dyn	A	G	C	R	L	e
1				1	1													
2				2	2	y		NoLimit										
3																		
4																		
5																		
6																		
7																		
8																		
9																		
10																		
11																		
12																		
13																		
14																		
15																		

## 5.6. Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and 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 and was configured using the parameters highlighted below, shown on the screen on the next page:

- 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 Session Manager.
- Set the **Transport Method** to the transport protocol to be used between Communication Manager and Session Manager. For the compliance test, *tls* was used.
- Set the **Peer Detection Enabled** field to *y*. The **Peer-Server** field will initially be set to *Others* and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to *SM* once Communication Manager detects its peer is a Session Manager.

**Note:** Once the **Peer-Server** field is updated to *SM*, the system changes the default values of the following fields, setting them to display-only:

- **Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?** is changed to *y*.

- Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? is changed to *n*.
- Set the **Near-end Node Name** to *procr*. This node name maps to the IP address of the Communication Manager as defined in **Section 5.3**.
- Set the **Far-end Node Name** to *sm*. This node name maps to the IP address of Session Manager, as defined in **Section 5.3**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid port. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to **5061**.
- Set the **Far-end Network Region** to the IP network region defined for the Service Provider in **Section 5.5**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set the **DTMF over IP** field to *rtp-payload*. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- 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 Avaya SBCE and the enterprise endpoint. If this value is set to *n*, then the Avaya Media Gateway or Media Server (if used) 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 Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Default values may be used for all other fields.

```

10.2.20.12 - PuTTY
change signaling-group 3
Page 1 of 2

SIGNALING GROUP

Group Number: 3
IMS Enabled? n
Q-SIP? n
IP Video? n
Peer Detection Enabled? y
Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr
Near-end Listen Port: 5061
Far-end Node Name: sm
Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain: telmex.com.mx
Incoming Dialog Loopbacks: eliminate
DTMF over IP: rtp-payload
Session Establishment Timer(min): 3
Enable Layer 3 Test? y
H.323 Station Outgoing Direct Media? n
Group Type: sip
Transport Method: tls
Enforce SIPS URI for SRTP? y
Bypass If IP Threshold Exceeded? n
RFC 3389 Comfort Noise? n
IP Audio Hairpinning? n
Initial IP-IP Direct Media? n
Alternate Route Timer(sec): 6
Direct IP-IP Audio Connections? y

```

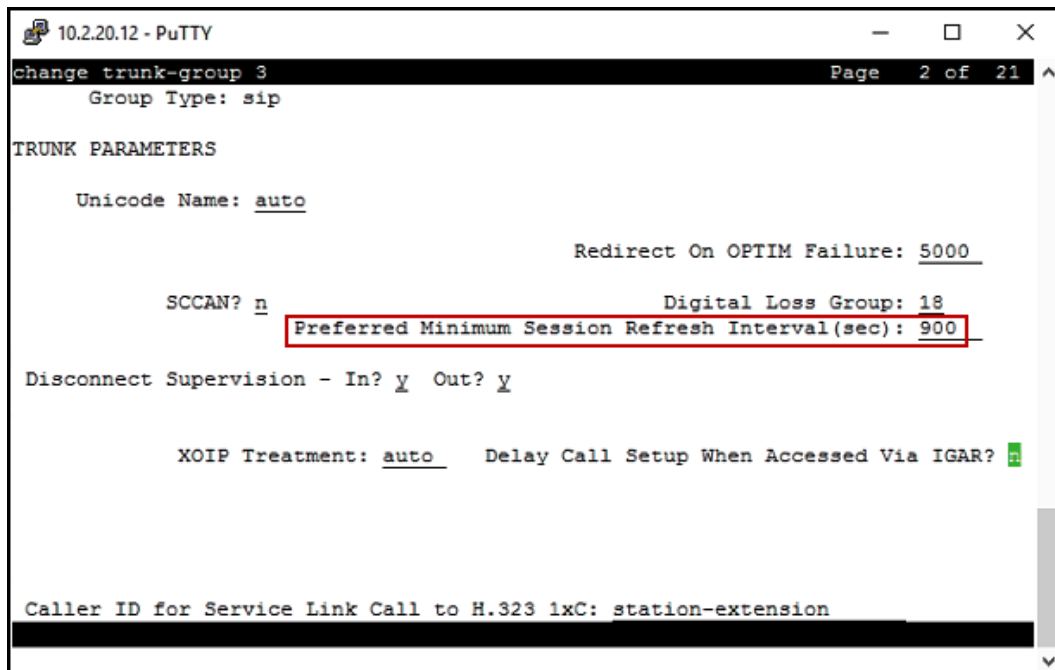
## 5.7. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.6**. 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 the **Signaling Group** to the signaling group shown in **Section 5.6**.
- 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.

```
10.2.20.12 - PuTTY
change trunk-group 3                                     Page 1 of 21
TRUNK GROUP
Group Number: 3      Group Type: sip      CDR Reports: y
Group Name: SIP      COR: 1      TN: 1      TAC: 803
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, **900** seconds was used.



```
10.2.20.12 - PuTTY
change 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): 900

Disconnect Supervision - In? y Out? y

  XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? off

Caller ID for Service Link Call to H.323 1xC: station-extension
```

**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. When *public* format is used, Communication Manager automatically inserts a “+” sign, preceding the numbers in the “From”, “Contact” and “P-Asserted Identity” (PAI) headers. To keep uniformity with the format used by Telmex, the **Numbering Format** was set to *private* and the **Numbering Format** in the route pattern was set to *unk-unk* (see **Section 5.10**).
- 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 5.2**, if the inbound call has enabled CPN block.

```
10.2.20.12 - PuTTY
change trunk-group 3 Page 3 of 21
TRUNK FEATURES
ACA Assignment? n Measured: none Maintenance Tests? y

Suppress # Outpulsing? n Numbering Format: private
UUI Treatment: service-provider

Replace Restricted Numbers? y
Replace Unavailable Numbers? y

Hold/Unhold Notifications? y
Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y
```

**On Page 4:**

- Set the **Network Call Redirection** field to *n*. With this setting, Communication Manager will not use the REFER method for calls being redirected back to the PSTN across the SIP trunk. The REFER method is not supported by Telmex.
- Set the **Send Diversion Header** field to *y* and **Support Request History** to *n*.
- Set the **Telephone Event Payload Type** to *101*.
- Set **Always Use re-INVITE for Display Updates** to *y*.
- Verify that **Identity for Calling Party Display** is set to *P-Asserted-Identity*.
- Default values were used for all other fields.

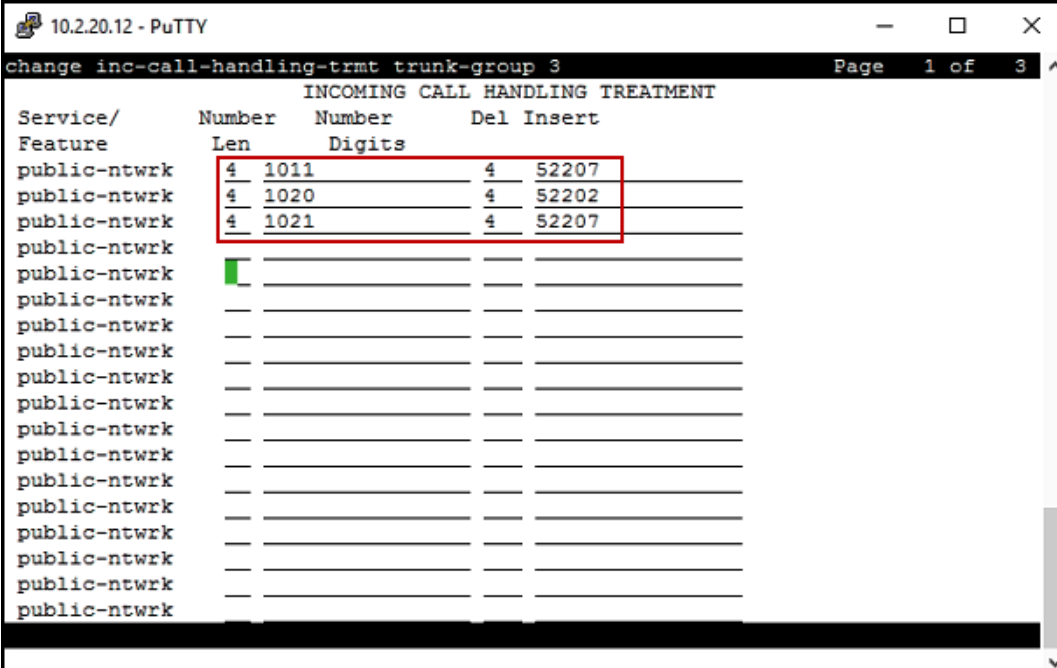
```
10.2.20.12 - PuTTY
change trunk-group 3                                     Page 4 of 21
PROTOCOL VARIATIONS
Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
Send Transferring Party Information? n
Network Call Redirection? n
Send Diversion Header? y
Support Request History? n
Telephone Event Payload Type: 101
Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? y
Identity for Calling Party Display: P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
Accept Redirect to Blank User Destination? n
Enable Q-SIP? n
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
Request URI Contents: may-have-extra-digits
```

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 5.7**), use the **change private-numbering** command to create an entry for each extension which has a DID assigned. DID numbers are provided by the SIP service provider. Each DID number is assigned in this table to one enterprise internal extension or Vector Directory Numbers (VDNs). In the example below, three DID numbers were assigned by the service provider for testing. The highlighted row shown below configures any four digit number beginning with 1 (i.e., 1xxx) that uses any trunk group to retain the original 4 digit number (i.e., no digit manipulation is specified), and the Total Len is 4. Thus, these same 4-digit numbers were used in the outbound calling party information on the service provider trunk when calls were originated from these 3 extensions.

23 of 82  
TelmCMSM70SBC72

## 5.9. Inbound Routing

In general, the “incoming call handling treatment” form for a trunk group can be used to manipulate the digits received for an incoming call if necessary. Since Session Manager is present, Session Manager can be used to perform digit conversion using an Adaptation, and digit manipulation via the Communication Manager incoming call handling table may not be necessary. If the DID number sent by Telmex is left unchanged by Session Manager, then the DID number can be mapped to an extension using the incoming call handling treatment of the receiving trunk group. Use the **change inc-call-handling-trmt** command to create an entry for each DID number needing to be converted to extensions numbers.



10.2.20.12 - PuTTY

change inc-call-handling-trmt trunk-group 3 Page 1 of 3

Service/ Feature	Number Len	Number Digits	Del	Insert
public-ntwrk	4	1011	4	52207
public-ntwrk	4	1020	4	52202
public-ntwrk	4	1021	4	52207
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				
public-ntwrk				



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*).

10.2.20.12 - PuTTY

change dialplan analysis Page 1 of 12

```

DIAL PLAN ANALYSIS TABLE
Location: all Percent Full: 1

Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
String   Length Type String Length Type String Length Type
1         4     ext
2         5     ext
3         3     fac
5         5     ext
8         3     dac
9         1     fac
*         2     fac
#         2     fac

```

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

```
10.2.20.12 - PuTTY
change feature-access-codes Page 1 of 10
FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code: 301
Abbreviated Dialing List2 Access Code: 302
Abbreviated Dialing List3 Access Code: 303
Abbreviated Dial - Prgm Group List Access Code:
Announcement Access Code: 399
Answer Back Access Code: 320
Attendant Access Code:
Auto Alternate Routing (AAR) Access Code: #8
Auto Route Selection (ARS) - Access Code 1: 9 Access Code 2:
Automatic Callback Activation: *5 Deactivation: #5
Call Forwarding Activation Busy/DA: *4 All: *2 Deactivation: #2
Call Forwarding Enhanced Status: Act: Deactivation:
Call Park Access Code: 315
Call Pickup Access Code: 317
CAS Remote Hold/Answer Hold-Unhold Access Code: #9
CDR Account Code Access Code:
Change COR Access Code:
Change Coverage Access Code:
Conditional Call Extend Activation: Deactivation:
Contact Closure Open Code: Close Code:
```

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 2.1** for the complete list of call types tested. All dialed strings are mapped to route patterns 11-16, which contains the SIP trunk group to the service provider.

10.2.20.12 - PuTTY

list ars analysis Page 1

ARS DIGIT ANALYSIS REPORT

Location: all

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Number	ANI Req
00	12	18	16	intl		n
001	13	13	15	intl		n
01	12	12	13	natl		n
020	3	3	11	locl		n
040	3	3	11	locl		n
044	13	13	12	locl		n
045	13	13	14	locl		n
0xx	3	3	2	locl		n
1	8	8	2	locl		n
2	8	8	2	locl		n
3	8	8	2	locl		n
4	8	8	2	locl		n

press CANCEL to quit -- press NEXT PAGE to continue

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

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 12 in the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP service provider.
- **FRL:** Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it.
- **Numbering Format:** Set to *unk-unk*. All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form for full details in **Section 5.7**.

10.2.20.12 - PuTTY

change route-pattern 12 Page 1 of 3

Pattern Number: 12 Pattern Name: Celular Local

SCCAN? n Secure SIP? n Used for SIP stations? n

Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
No	Mrk	Lmt	List	Del	Digits			QSIG	Intw
1:	<u>3</u>	<u>2</u>						<u>n</u>	<u>user</u>
2:								<u>n</u>	<u>user</u>
3:								<u>n</u>	<u>user</u>
4:								<u>n</u>	<u>user</u>
5:								<u>n</u>	<u>user</u>
6:								<u>n</u>	<u>user</u>

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	Sub	Numbering	LAR
0	1	2	M	4	W	Request		Dgts	Format	
1:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>		<u>unk-unk</u>	<u>none</u>
2:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>			<u>none</u>
3:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>			<u>none</u>
4:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>			<u>none</u>
5:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>			<u>none</u>
6:	<u>y</u>	<u>y</u>	<u>y</u>	<u>y</u>	<u>n</u>	<u>n</u>	<u>rest</u>			<u>none</u>

Repeat this procedure as needed to define additional route patterns. For the compliance test route patterns 11-16 were used. Calls to the PSTN, international, local/national, local/information and local/cellular are examples of call types that were assigned individual route patterns, only route pattern 12 (for cellular local calls) is shown on the screenshot shown above.

**Note** – Enter the **save translation** command (not shown) to save all the changes made to the Communication Manager configuration in the previous sections.

## 6. Configure Avaya Aura® Session Manager

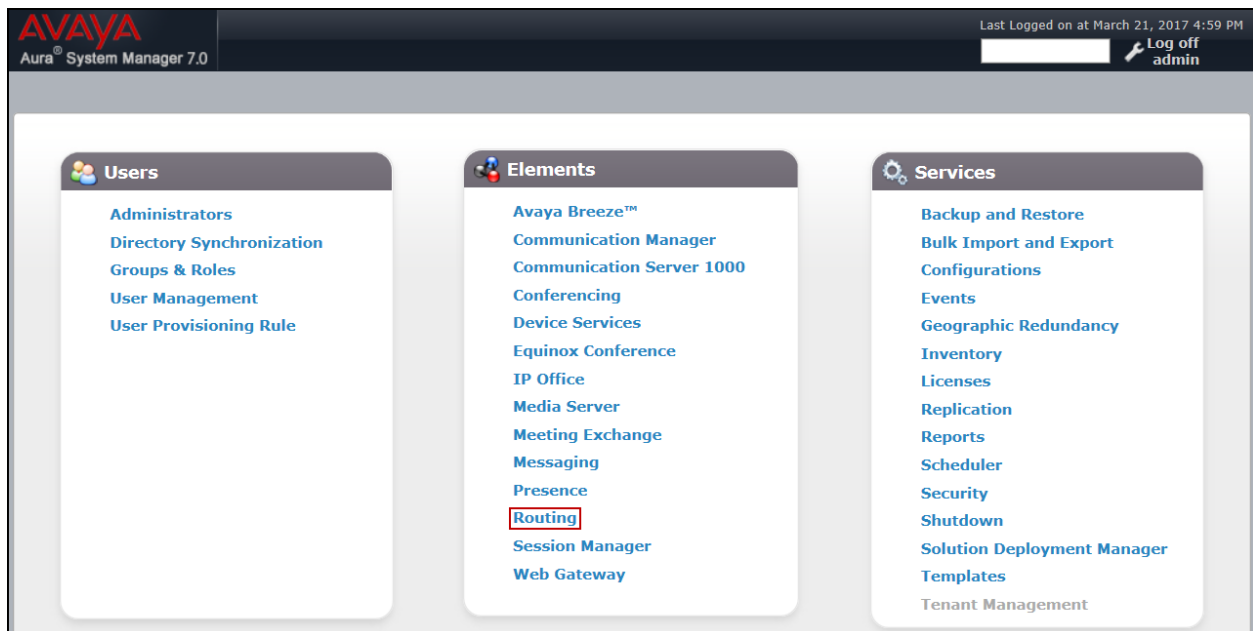
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- Adaptation module to perform header manipulations.
- SIP Entities corresponding to Communication Manager, Session Manager and the Avaya SBCE.
- 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.

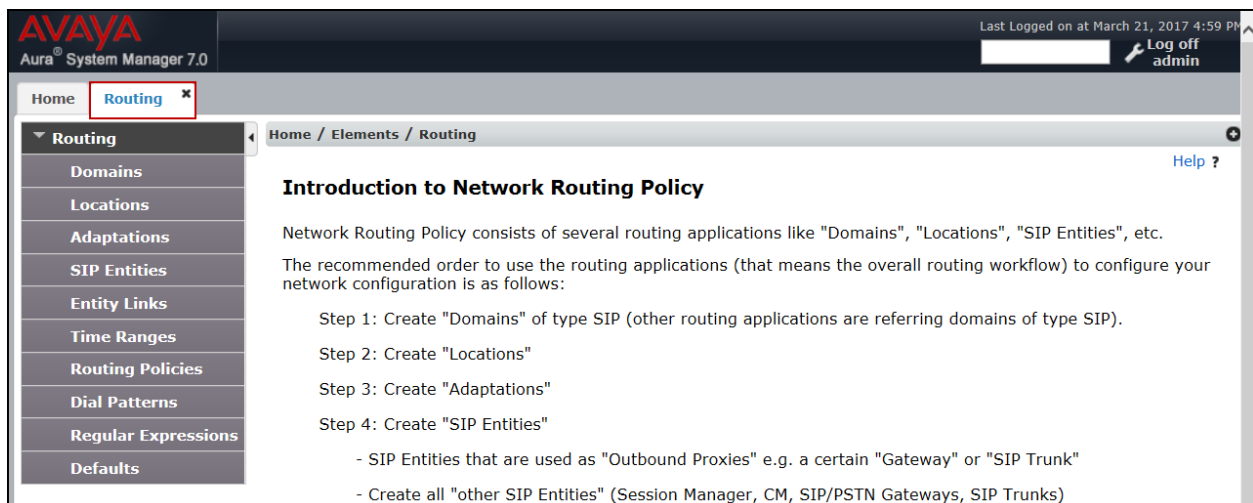
The following sections assume that the initial configuration of Session Manager and System Manager has already been completed, and that network connectivity exists between System Manager and Session Manager.

## 6.1. 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 **Log On** (not shown). The screen shown below is then displayed; click on **Routing**.



The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items discussed in this section will be located under the **Routing** link shown below.



## 6.2. SIP Domain

Create an entry for each SIP domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this was the enterprise domain, *telmex.com.mx*. Navigate to **Routing → Domains** in the left-hand navigation pane 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** to save.

The screen below shows the entry for the enterprise domain.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left-hand navigation pane is expanded, showing the 'Routing' menu with 'Domains' selected. The main content area is titled 'Domain Management' and shows a table with one item. The table has columns for 'Name', 'Type', and 'Notes'. The 'Name' column contains 'telmex.com.mx', and the 'Type' column contains 'sip'. The 'Notes' column is empty. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table area.

Name	Type	Notes
* telmex.com.mx	sip	

## 6.3. Locations

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

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

The following screen shows the location details for the location named *main*.

Later, this location will be assigned to the SIP Entities. Other parameters for this location (not shown) retained the default values.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The left-hand navigation pane shows a tree structure with 'Routing' selected, and 'Locations' highlighted. The main content area is titled 'Location Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' section contains a 'Name' field with the value 'main' and a 'Notes' field. Below this, the 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox. At the bottom, there are fields for 'Listed Directory Number' and 'Associated CM SIP Entity'.

AVAYA  
Aura® System Manager 7.0

Home Routing x

Home / Elements / Routing / Locations

**Location Details** Commit Cancel

**General**

\* Name: main

Notes:

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:



## 6.4. Adaptations

In order to improve interoperability with third party elements, Session Manager 7.0 incorporates the ability to use Adaptation modules to remove specific headers that are either Avaya proprietary or deemed excessive/unnecessary for non-Avaya elements.

For the compliance test, an Adaptation named **CM\_Outbound\_Header\_Removal** was created to block the following headers from outbound messages, before they were forwarded to the Avaya SBCE: AV-Global-Session-ID, AV-Correlation-ID, Alert-Info, Endpoint-View, P-AV-Message-Id, P-Charging-Vector and P-Location. These headers contain private information from the enterprise, which should not be propagated outside of the enterprise boundaries. They also add unnecessary size to outbound messages, while they have no significance to the service provider.

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

- **Adaptation Name:** Enter an appropriate name.
- **Module Name:** Select the *DigitConversionAdapter* option.
- **Module Parameter Type:** Select *Name-Value Parameter*.

Click **Add** to add the name and value parameters, as follows:

- **Name:** Enter *eRHdrs*. This parameter will remove the specified headers from messages in the egress direction.
- **Value:** Enter “*Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-Id, P-Location, Endpoint-View*”
- Click **Commit** to save.

The screen below shows the adaptation created for the compliance test. This adaptation will later be applied to the SIP Entity corresponding to the Avaya SBCE. All other fields were left at their default values.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left-hand navigation pane shows the 'Routing' menu expanded, with 'Adaptations' selected. The main content area is titled 'Adaptation Details' and includes a 'General' tab. The form contains the following fields:

- Adaptation Name:** CM\_Outbound\_Header\_Removal
- Module Name:** DigitConversionAdapter
- Module Parameter Type:** Name-Value Parameter

Below these fields is a table for adding parameters:

Name	Value
eRHdrs	Alert-Info, P-Charging-Vector, AV-Global-Session-ID, AV-Correlation-ID, P-AV-Message-Id, P-Location, Endpoint-View

At the bottom of the form, there are fields for 'Egress URI Parameters' and 'Notes'.

## 6.5. 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 Avaya SBCE. Navigate to **Routing** → **SIP Entities** in the left 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:

- **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 (see **Figure 1**).
- **Type:** Select *Session Manager* for Session Manager, *CM* for Communication Manager and *SIP Trunk* for the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**.  
If Adaptations were to be created, here is where they would be applied to the entity.
- **Location:** Select the location that applies to the SIP Entity being created, defined in **Section 6.3**.
- **Time Zone:** Select the time zone for the location above.
- Click **Commit** to save.

The following screen shows the addition of the *smS* SIP Entity for Session Manager. The IP address of the Session Manager Security Module is entered in the **FQDN or IP Address** field.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left navigation pane shows 'Routing' and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and 'General'. The form contains the following fields:

- Name:** smS
- FQDN or IP Address:** 10.2.20.17
- Type:** Session Manager
- Notes:**
- Location:** main
- Outbound Proxy:**
- Time Zone:** America/Mexico\_City
- Credential name:**
- SIP Link Monitoring:** Use Session Manager Configuration

The following screen shows the addition of the *cmS* SIP Entity for Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager, the creation of a separate SIP entity for Communication Manager is required. This SIP Entity should be different than the one created during the Session Manager installation, used by all other enterprise SIP traffic. The **FQDN or IP Address** field is set to the IP address of the “**procr**” interface in Communication Manager, as seen in **Section 5.3**. Select the **location** that applies to the SIP Entity being created, defined in **Section 6.3**. Select the **time zone** for the location above.

**AVAYA**  
Aura® System Manager 7.0

Home Routing \*  
Home / Elements / Routing / SIP Entities

**SIP Entity Details** [Commit] [Cancel]

**General**

\* Name: cmS  
\* FQDN or IP Address: 10.2.20.12  
Type: CM  
Notes: cmS

Adaptation: [v]  
Location: main [v]  
Time Zone: America/Mexico\_City [v]

\* SIP Timer B/F (in seconds): 4  
Credential name: [v]  
Securable: [checked]  
Call Detail Recording: none [v]

**Loop Detection**  
Loop Detection Mode: Off [v]

**SIP Link Monitoring**  
SIP Link Monitoring: Use Session Manager Configuration [v]

The following screen shows the addition of the *AvayaSBC* SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**). On the **Adaptation** field, the adaptation module *CM\_Outbound\_Header\_Removal* previously defined in **Section 6.4** was selected. Select the **time zone** for the location above.

**AVAYA**  
Aura® System Manager 7.0

Home Routing x

Home / Elements / Routing / SIP Entities

### SIP Entity Details

Commit Cancel

**General**

\* Name: AvayaSBC

\* FQDN or IP Address: 10.2.20.19

Type: SIP Trunk

Notes:

Adaptation: CM\_Outbound\_Header\_Removal

Location:

Time Zone: America/Mexico\_City

\* SIP Timer B/F (In seconds): 4

Credential name:

Securable: ☐

Call Detail Recording: egress

**Loop Detection**

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

## 6.6. 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 Avaya SBCE. To add an Entity Link, navigate to **Routing** → **Entity Links** in the left navigation pane 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 from the drop-down menu (**Section 6.5**).
- **Protocol:** Select the transport protocol used for this link (**Section 5.6**).
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end (**Section 5.6**).
- **SIP Entity 2:** Select the name of the other system from the drop-down menu (**Section 6.5**).
- **Port:** Port number on which the other system receives SIP requests from Session Manager (**Section 5.6**).
- **Connection Policy:** Select *Trusted* to allow calls from the associated SIP Entity.
- Click **Commit** to save.

The screen below shows the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group form in **Section 5.6**. *TLS* transport and port **5061** were used.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left navigation pane shows the 'Routing' menu with 'Entity Links' selected. The main content area is titled 'Entity Links' and contains a table with one item. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, Connection Policy, Deny New Service, and Notes. The item in the table is highlighted with a red border.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
*sm_to_cm	*Q.sM	TLS	*5061	*Q.cmS	No	*5061	trusted	No	

The Entity Link to the Avaya SBCE is shown below; *TLS* transport and port *5061* were used.

AVAYA  
Aura® System Manager 7.0

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
<input type="checkbox"/>	* SM_AvayaSBC	* Q smS	TLS	* 5061	* Q AvayaSBC		* 5061	trusted	<input type="checkbox"/>	

Select : All, None

## 6.7. Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies were added: an incoming policy with Communication Manager as the destination, and an outbound policy to the Avaya SBCE. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed:

- In the **General** section, enter a descriptive **Name** and add a brief description under **Notes** (optional).
- In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Choose the appropriate SIP entity to which this routing policy applies (**Section 6.5**) 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 Avaya SBCE.

AVAYA  
Aura® System Manager 7.0

Last Logged on at August 11, 2017 7:35 AM  
Go... Log off admin

Home Routing

Home / Elements / Routing / Routing Policies

### Routing Policy Details

Commit Cancel Help ?

**General**

\* Name: CM\_policy

Disabled: ☐

\* Retries: 0

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
cmS	10.2.20.12	CM	cmS

AVAYA  
Aura® System Manager 7.0

Last Logged on at August 11, 2017 7:35 AM  
Go... Log off admin

Home Routing

Home / Elements / Routing / Routing Policies

### Routing Policy Details

Commit Cancel Help ?

#### General

\* Name: Avaya SBC

Disabled: ☐

\* Retries: 0

Notes:

#### SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AvayaSBC	10.2.20.19	SIP Trunk	

## 6.8. Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns were needed to route calls from Communication Manager to the service provider 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 navigation pane 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:

- **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 (**Section 6.2**).
- **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 (**Section 6.3**).
- Lastly, select the routing policy from the list that will be used to route all calls that match the specified criteria (**Section 6.7**). Click **Select** (not shown).
- Click **Commit** to save.



The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. In the example, calls to 4 digit number **1011**, arriving from location **main**, used route policy **CM\_policy** to Communication Manager. The SIP Domain was set to **all**.

**AVAYA**  
Aura® System Manager 7.0

Home Routing

Home / Elements / Routing / Dial Patterns

**Dial Pattern Details** [Commit] [Cancel] [Help ?]

**General**

\* Pattern: 1011  
 \* Min: 4  
 \* Max: 4

Emergency Calls: ☐  
 Emergency Priority: 1  
 Emergency Type:   
 SIP Domain: -ALL-  
 Notes:

**Originating Locations and Routing Policies**

[Add] [Remove]

1 Item

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
main		CM_policy	0	<input type="checkbox"/>	cmS	

Select : All, None

Repeat this procedure as needed to define additional dial patterns for other range of DID numbers assigned by the service provider to the enterprise.

The example in this screen shows the 13 digit dialed numbers for outbound calls, beginning with **044**, for calls to local cellular numbers, arriving from the **main** location, will use route policy **AvayaSBC**, which sends the call out to the PSTN via Avaya SBCE and the service provider SIP Trunk. The SIP Domain was set to **all**.

**Avaya**  
Aure System Manager 7.0

Home / Elements / Routing / Dial Patterns

**Dial Pattern Details**

Commit Cancel

General

\* Pattern: 044

\* Min: 13

\* Max: 13

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
main		Avaya SBC	0	<input type="checkbox"/>	AvayaSBC	

Select : All, None

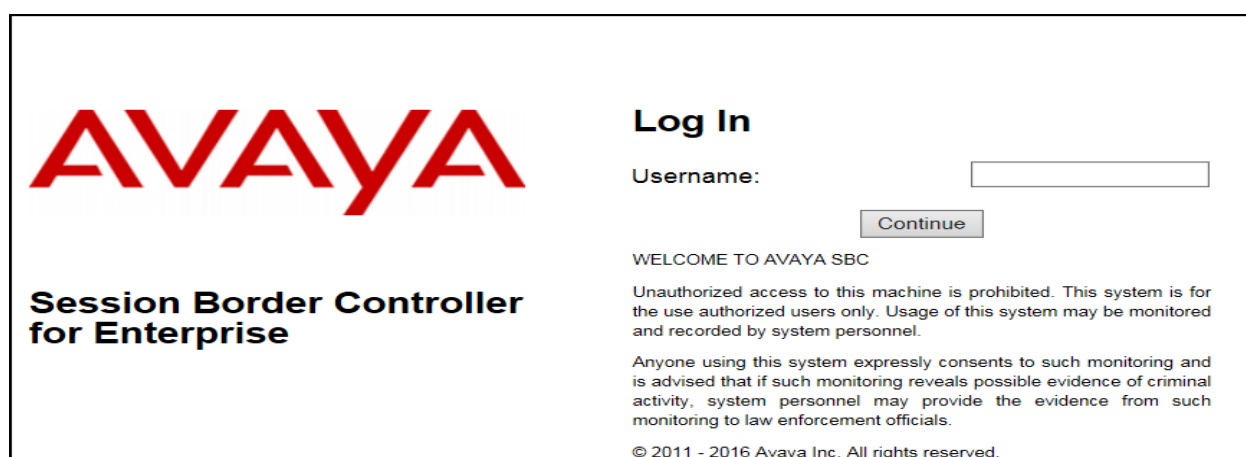
Repeat this procedure as needed, to define additional dial patterns for PSTN numbers to be routed to the service provider's network via the Avaya SBCE.

## 7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE. It is assumed that the initial installation of the Avaya SBCE, the assignment of the management interface IP Address and license installation have already been completed; hence these tasks are not covered in these Application Notes. For more information on the installation and initial provisioning of the Avaya SBCE consult the Avaya SBCE documentation in the **References** section.

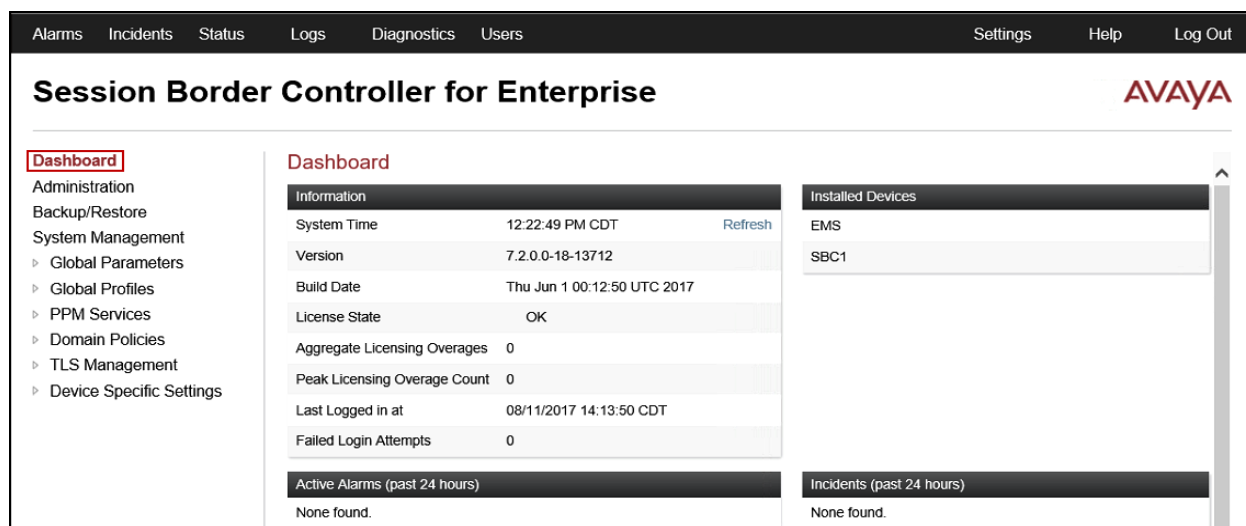
### 7.1. System Access

Access the Session Border Controller web management interface by using a web browser and entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation. Log in using the appropriate credentials.



The screenshot shows the login page for the Avaya Session Border Controller for Enterprise. On the left is the Avaya logo and the product name. On the right, under 'Log In', there is a 'Username:' label, a text input field, and a 'Continue' button. Below the login fields, there is a 'WELCOME TO AVAYA SBC' message, a disclaimer about unauthorized access, a consent statement, and a copyright notice for 2011-2016 Avaya Inc.

Once logged in, the Dashboard screen is presented. The left navigation pane contains the different available menu items used for the configuration of the Avaya SBCE. Verify that the status of the **License State** field is **OK**, indicating that a valid license is present. Contact an authorized Avaya sales representative if a license is needed.



The screenshot shows the dashboard of the Avaya Session Border Controller for Enterprise. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main content area is titled 'Session Border Controller for Enterprise' and features the Avaya logo. On the left is a navigation pane with 'Dashboard' highlighted. The dashboard itself is divided into several sections: 'Information' (System Time, Version, Build Date, License State, Aggregate Licensing Overages, Peak Licensing Overage Count, Last Logged in at, Failed Login Attempts), 'Installed Devices' (EMS, SBC1), 'Active Alarms (past 24 hours)', and 'Incidents (past 24 hours)'. The 'License State' is shown as 'OK'.

## 7.2. System Management

To view current system information, select **System Management** on the left navigation pane. A list of installed devices is shown in the **Devices** tab on the right pane. In the reference configuration, a single device named **SBC1** is shown. The management IP address that was configured during installation is blurred out for security reasons, the current software version is shown. The management IP address needs to be on a subnet separate from the ones used in all other interfaces of the Avaya SBCE, segmented from all VoIP traffic. Verify that the **Status** is **Commissioned**, indicating that the initial installation process of the device has been previously completed, as shown on the screen below.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the Avaya logo. On the left, a navigation pane lists various system management options, with "System Management" highlighted. The main content area is titled "System Management" and contains several tabs: Devices, Updates, SSL VPN, Licensing, and Key Bundles. The "Devices" tab is active, showing a table of installed devices. The table has columns for Device Name, Management IP, Version, and Status. A single device, SBC1, is listed with a blurred management IP, version 7.2.0.0-18-13712, and a status of "Commissioned". Action buttons for Reboot, Shutdown, Restart Application, View, Edit, and Uninstall are visible for the device.

Device Name	Management IP	Version	Status
SBC1	[Blurred]	7.2.0.0-18-13712	Commissioned

To view the network configuration assigned to the Avaya SBCE, click **View** on the screen above. The **System Information** window is displayed, containing the current device configuration and network settings.

The screenshot displays the "System Information: SBC1" window, which provides detailed configuration and network settings for the device. The window is organized into several sections:

- General Configuration:** Includes Appliance Name (SBC1), Box Type (SIP), and Deployment Mode (Proxy).
- Device Configuration:** Includes HA Mode (No) and Two Bypass Mode (No).
- License Allocation:** Lists various session types and their requested counts: Standard Sessions (10), Advanced Sessions (10), Scopia Video Sessions (10), CES Sessions (10), and Transcoding Sessions (10). It also shows an Encryption checkbox that is checked.
- Network Configuration:** A table showing the assigned IP addresses, public IPs, network prefixes, gateways, and interfaces.
- DNS Configuration:** Includes Primary DNS (8.8.8.8), Secondary DNS, DNS Location (DMZ), and DNS Client IP (10.2.20.19).
- Management IP(s):** Shows IP #1 (IPv4) as [Blurred] 127.245.

IP	Public IP	Network Prefix or Subnet Mask	Gateway	Interface
10.2.20.19	10.2.20.19	255.255.255.128	10.2.20.1	A1
10.2.20.28	10.2.20.28	255.255.255.128	10.2.20.1	B1

In the reference configuration, the private interface of the Avaya SBCE (10.2.20.19) was used to connect to the enterprise network, while its public interface (10.2.20.28) was used to connect to the Telmex private IP network for routing calls to and from the PSTN via Telmex's network. See **Figure 1**.

On the **License Allocation** area of the **System Information**, verify that the number of **Standard Sessions** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise. The number of sessions and encryption features are primarily controlled by the license file installed.

### 7.3. Network Management

The network configuration parameters should have been previously specified during installation of the Avaya SBCE. In the event that changes need to be made to the network configuration, they can be entered here.

Select **Network Management** from **Device Specific Settings** on the left-side menu. Under **Devices** in the center pane, select the device being managed, **SBC1** in the sample configuration. On the **Networks** tab, verify or enter the network information as needed.

Note that in the configuration used during the compliance test, the IP addresses assigned to the private (**10.2.20.19**) and public (**10.2.20.28**) sides of the Avaya SBCE are the ones relevant to these Application Notes.

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

## Session Border Controller for Enterprise

AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
‣ Global Parameters  
‣ Global Profiles  
‣ PPM Services  
‣ Domain Policies  
‣ TLS Management  
‣ Device Specific Settings  
‣ **Network Management**  
Media Interface  
Signaling Interface  
End Point Flows  
Session Flows  
‣ DMZ Services

Network Management: SBC1

Devices  
SBC1

Interfaces Networks

Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address	Edit	Delete
TelmexPrivate	10.2.20.1	255.255.255.128	A1	10.2.20.19	Edit	Delete
PublicTelmex	10.2.20.1	255.255.255.128	B1	10.2.20.28	Edit	Delete

Add

On the **Interfaces** tab, verify the **Administrative Status** is **Enabled** for the **A1** and **B1** interfaces. Click the buttons under the **Status** column if necessary to enable the interfaces.

**Session Border Controller for Enterprise** AVAYA

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

**Network Management: SBC1**

Devices **SBC1** Interfaces **Networks**

Add VLAN

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

Dashboard  
Administration  
Backup/Restore  
System Management  
‣ Global Parameters  
‣ Global Profiles  
‣ PPM Services  
‣ Domain Policies  
‣ TLS Management  
‣ Device Specific Settings  
  **Network Management**  
  Media Interface  
  Signaling Interface  
  End Point Flows  
  Session Flows  
  DMZ Services

## 7.4. Media Interfaces

Media Interfaces were created to specify the IP address and port range in which the Avaya SBCE will accept media streams on each interface. Packets leaving the interfaces of the Avaya SBCE will advertise this IP address, and one of the ports in this range as the listening IP address and port in which it will accept media from the Call Server or the Trunk Server.

To add the Media Interface in the enterprise direction, select **Media Interface** from the **Device Specific Settings** menu on the left-hand side, select the device being managed and click the **Add** button (not shown).

- On the **Add Media Interface** screen, enter an appropriate **Name** for the Media Interface.
- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- The **Port Range** was left at the default values of **35000-40000**.
- Select a **TLS Profile**.
- Click **Finish**.

**Add Media Interface** X

Name PrivateMed

IP Address TelmexPrivate (A1, VLAN 0) 10.2.20.19

Port Range 35000 - 40000

TLS Profile AvayaSBCServer

Finish

A Media Interface facing the public side was similarly created with the name ***PublicMed***, as shown below.

- Under **IP Address**, the network and IP address to be associated with this interface was selected.
- The **Port Range** was left at the default values.
- Click **Finish**.

The screenshot shows the 'Add Media Interface' configuration window. The fields are as follows:

Field	Value
Name	PublicMed
IP Address	PublicTelmex (B1, VLAN 0) / 10.2.20.28
Port Range	35000 - 40000
TLS Profile	None

Finish

## 7.5. Signaling Interfaces

Signaling Interfaces are created to specify the IP addresses and ports in which the Avaya SBCE will listen for signaling traffic in the connected networks.

To add the Signaling Interface in the enterprise direction, select **Signaling Interface** from the **Device Specific Settings** menu on the left-hand side, select the device being managed and click the **Add** button (not shown).

- On the **Add Signaling Interface** screen, enter an appropriate **Name** for the interface.
- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5061** for **TLS Port**, since TLS port 5061 is used to listen for signaling traffic from Session Manager in the sample configuration, as defined in **Section 6.6**.
- Select a **TLS Profile**.
- Click **Finish**.

**Add Signaling Interface** X

Name: PrivateSig

IP Address: TelmexPrivate (A1, VLAN 0) ▼  
10.2.20.19 ▼

TCP Port:   
Leave blank to disable

UDP Port:   
Leave blank to disable

TLS Port: 5061  
Leave blank to disable

TLS Profile: AvayaSBCServer ▼

Enable Shared Control: ☐

Shared Control Port:

Finish



A second Signaling Interface with the name **Public\_sig** was similarly created in the service provider's direction.

- Under **IP Address**, select from the drop-down menus the network and IP address to be associated with this interface.
- Enter **5060** for **UDP Port**, since this is the protocol and port used by the Avaya SBCE to listen to the service provider's SIP traffic.
- Click **Finish**.

The screenshot shows the 'Add Signaling Interface' dialog box. The fields are as follows:

Field	Value
Name	Public_sig
IP Address	PublicTelmex (B1, VLAN 0) / 10.2.20.28
TCP Port	
UDP Port	5060
TLS Port	
TLS Profile	None
Enable Shared Control	<input type="checkbox"/>
Shared Control Port	

Finish

## 7.6. Server Interworking

Interworking Profile features are configured to facilitate the interoperability between the enterprise SIP-enabled solution (Call Server) and the SIP trunk service provider (Trunk Server).

### 7.6.1. Server Interworking Profile – Enterprise

Interworking profiles can be created by cloning one of the pre-defined default profiles, or by adding a new profile. To configure the interworking profile in the enterprise direction, select **Global Profiles → Server Interworking** on the left navigation pane. Under **Interworking Profiles**, select *avaya-ru* from the list of pre-defined profiles. Click **Clone**.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms 1', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header is 'Session Border Controller for Enterprise' with the Avaya logo. The left navigation pane lists various configuration areas, with 'Global Profiles' expanded and 'Server Interworking' highlighted. The main content area is titled 'Interworking Profiles: avaya-ru' and features a list of profiles on the left, including 'cs2100', 'avaya-ru' (highlighted), 'OCS-Edge-Server', 'cisco-ccm', 'cups', 'OCS-FrontEnd...', 'Avaya-SM', 'SP-General', 'Avaya-IPO', 'Avaya-CS1000', and 'Avaya-CM'. A 'Clone' button is visible. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new profile instead.' Below this, the 'General' tab is selected, showing a table of configuration parameters.

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
Send Hold	No
Delayed Offer	No
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No

- Enter a descriptive name for the cloned profile.
- Click **Finish**.

The 'Clone Profile' dialog box is shown with a close button (X) in the top right corner. It contains two input fields: 'Profile Name' with the value 'avaya-ru' and 'Clone Name' with the value 'Avaya-SM'. A red rectangle highlights the 'Clone Name' field. A 'Finish' button is located at the bottom center.

Click **Edit** on the newly cloned *Avaya-SM* interworking profile:

- On the **General** tab, check *T.38 Support* (See note below).
- Leave remaining fields with default values.
- Click **Finish**.

The screenshot shows a window titled "Editing Profile: Avaya-SM" with a close button (X) in the top right corner. The "General" tab is selected. The following settings are visible:

- Hold Support:** Radio buttons for None (selected), RFC2543 - c=0.0.0.0, and RFC3264 - a=sendonly.
- 180 Handling:** Radio buttons for None (selected), SDP, and No SDP.
- 181 Handling:** Radio buttons for None (selected), SDP, and No SDP.
- 182 Handling:** Radio buttons for None (selected), SDP, and No SDP.
- 183 Handling:** Radio buttons for None (selected), SDP, and No SDP.
- Refer Handling:** A checkbox that is unchecked.
- URI Group:** A dropdown menu showing "None".
- Send Hold:** A checkbox that is unchecked.
- Delayed Offer:** A checkbox that is unchecked.
- 3xx Handling:** A checkbox that is unchecked.
- Diversion Header Support:** A checkbox that is unchecked.
- Delayed SDP Handling:** A checkbox that is unchecked.
- Re-Invite Handling:** A checkbox that is unchecked.
- Prack Handling:** A checkbox that is unchecked.
- Allow 18X SDP:** A checkbox that is unchecked.
- T.38 Support:** A checkbox that is checked and highlighted with a red box.
- URI Scheme:** Radio buttons for SIP (selected), TEL, and ANY.
- Via Header Format:** Radio buttons for RFC3261 (selected) and RFC2543.

A "Finish" button is located at the bottom right of the dialog.

**Note** – Currently Telmex does not support T.38 fax in their SIP Trunk service offering, T.38 fax support was enabled for future use by Telmex.

The **Timers**, **Privacy**, **URI Manipulation** and **Header Manipulation** tabs contain no entries. The **Advanced** tab settings are shown on the screen below:

The screenshot displays the Avaya Session Border Controller for Enterprise configuration interface. The top navigation bar includes links for Alarms (1), Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo.

The left sidebar contains a navigation menu with the following items: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server Interworking (selected), Media Forking, Routing, Server Configuration, Topology Hiding, Signaling Manipulation, URI Groups, SNMP Traps, Time of Day Rules, FGDN Groups, Reverse Proxy Policy, and PPM Services.

The main content area is titled "Interworking Profiles: Avaya-SM". It features a list of profiles on the left: cs2100, avaya-ru, OCS-Edge-Server, cisco-ccm, cups, OCS-FrontEnd..., Avaya-SM (selected), SP-General, Avaya-IPO, Avaya-CS1000, and Avaya-CM. An "Add" button is located above this list.

The right pane shows the configuration for the selected "Avaya-SM" profile. It includes tabs for General, Timers, Privacy, URI Manipulation, Header Manipulation, and Advanced (selected). The Advanced tab contains the following settings:

Click here to add a description.	
Record Routes	Both Sides
Include End Point IP for Context Lookup	Yes
Extensions	Avaya
Diversion Manipulation	No
Has Remote SBC	Yes
Route Response on Via Port	No
Relay INVITE Replace for SIPREC	No

Below these settings is a section for DTMF configuration:

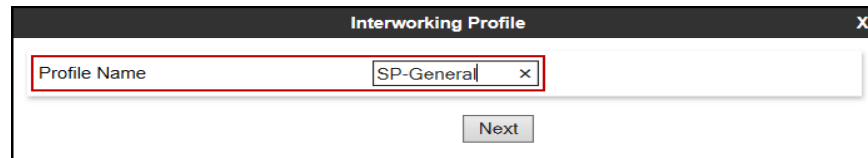
DTMF	
DTMF Support	None

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

## 7.6.2. Server Interworking Profile – Service Provider

A second interworking profile in the direction of the SIP trunk was created, by adding a new profile in this case. Select **Global Profiles → Server Interworking** on the left navigation pane and click **Add** (not shown).

- Enter a descriptive name for the new profile.
- Click **Next**.

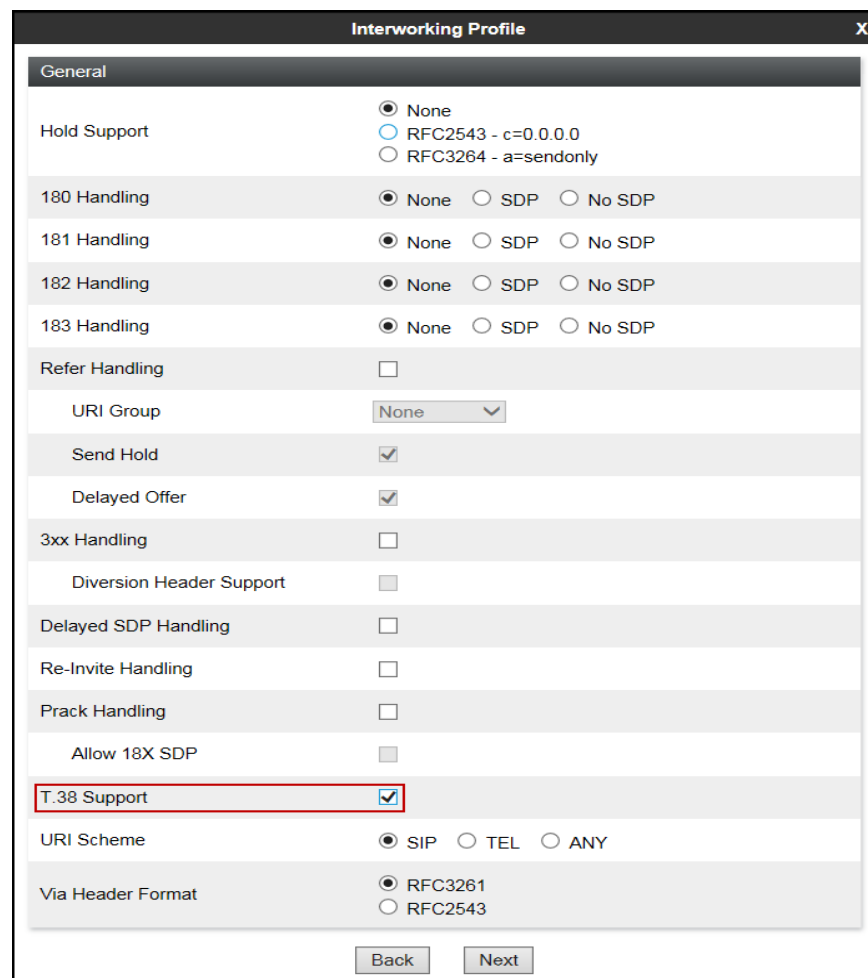


Interworking Profile

Profile Name  x

Next

- On the **General** tab, check **T.38 Support** (See note below). Click **Next**, then click **Finish** on the last tab leaving remaining fields with default values (not shown).



Interworking Profile

General

Hold Support ☒ None ☐ RFC2543 - c=0.0.0.0 ☐ RFC3264 - a=sendonly

180 Handling ☒ None ☐ SDP ☐ No SDP

181 Handling ☒ None ☐ SDP ☐ No SDP

182 Handling ☒ None ☐ SDP ☐ No SDP

183 Handling ☒ None ☐ SDP ☐ No SDP

Refer Handling ☐

URI Group

Send Hold ☒

Delayed Offer ☒

3xx Handling ☐

Diversion Header Support ☐

Delayed SDP Handling ☐

Re-Invite Handling ☐

Prack Handling ☐

Allow 18X SDP ☐

**T.38 Support ☒**

URI Scheme ☒ SIP ☐ TEL ☐ ANY

Via Header Format ☒ RFC3261 ☐ RFC2543

Back Next

**Note** – Currently Telmex does not support T.38 fax in their SIP Trunk service offering, T.38 fax support was enabled for future use by Telmex.

## 7.7. Signaling Manipulation

**Music on hold:** When calls from/to the PSTN were placed on-hold by the Communication Manager users, the PSTN users did not hear Music while on-hold, Telmex responded with “Invalid Media Type” to the “sendonly” message Communication Manager includes in the SDP of re-INVITEs when calls from/to the PSTN are placed on-hold. A SigMa script was created to replace the “sendonly” message with “sendrecv”; this allowed the PSTN users to hear Music while on-hold. The script was latter applied to the Service Provider side of the server configuration profile (refer to **Section 7.8.2**). The **change\_sendonly** script is shown below.

To add a Signaling Manipulation script, from the **Global Profiles** menu on the left panel, select **Signaling Manipulation**. Click **Add** to open the SigMa Editor screen, where the text of the script can be entered or copied.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The left-hand navigation menu is expanded to 'Global Profiles', and 'Signaling Manipulation' is selected. The main content area is titled 'Signaling Manipulation Scripts: change\_sendonly'. It features buttons for 'Upload', 'Add', 'Download', 'Clone', and 'Delete'. A description field contains the text 'Click here to add a description.' Below this, a 'Signaling Manipulation' script is shown in a text area, enclosed in a red box. The script is as follows:

```
within session "INVITE" {  
  act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"  
  {  
    //Removes the a=sendonly from the re-INVITE messages.  
    %BODY[1].regex_replace("a=sendonly","a=sendrecv");  
  }  
}
```

An 'Edit' button is located at the bottom right of the script text area.

The details of the script used can also be found in **Appendix A** in this document.

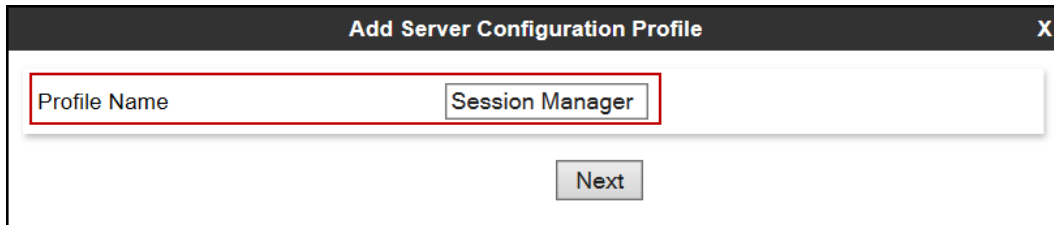
## 7.8. Server Configuration

Server Profiles are created to define the parameters for the Avaya SBCE peers; Session Manager (Call Server) at the enterprise and Telmex SIP Proxy (Trunk Server).

### 7.8.1. Server Configuration Profile – Enterprise

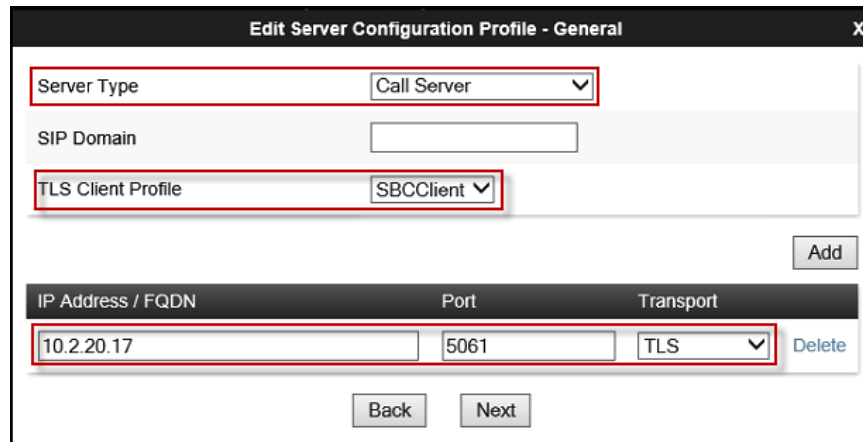
From the **Global Profiles** menu on the left-hand navigation pane, select **Server Configuration** and click the **Add** button (not shown) to add a new profile for the Call Server.

- Enter an appropriate **Profile Name** similar to the screen below.
- Click **Next**.



The screenshot shows a dialog box titled "Add Server Configuration Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" which contains the text "Session Manager". Below this field is a button labeled "Next".

- On the **Edit Server Configuration Profile – General** tab select **Call Server** from the drop down menu under the **Server Type**.
- Select a **TLS Client Profile**.
- On the **IP Addresses / FQDN** field, enter the IP address of the Session Manager Security Module (**Section 6.5**).
- Enter **5061** under **Port** and select **TLS** for **Transport**. The transport protocol and port selected here must match the values defined for the Entity Link to the Session Manager previously created in **Section 6.6**.
- Click **Next**.



The screenshot shows a dialog box titled "Edit Server Configuration Profile - General" with a close button (X) in the top right corner. The dialog contains several fields and buttons:

- Server Type**: A dropdown menu set to "Call Server".
- SIP Domain**: An empty text input field.
- TLS Client Profile**: A dropdown menu set to "SBCCClient".
- Add**: A button located to the right of the TLS Client Profile dropdown.
- IP Address / FQDN**: A text input field containing "10.2.20.17".
- Port**: A text input field containing "5061".
- Transport**: A dropdown menu set to "TLS".
- Delete**: A button located to the right of the Transport dropdown.
- Back** and **Next**: Buttons at the bottom of the dialog.

- Click **Next** on the **Authentication** and **Heartbeat** tabs (not shown).
- Select **Avaya-SM** from the **Interworking Profile** drop down menu.
- Click **Finish**.

**Add Server Configuration Profile - Advanced**

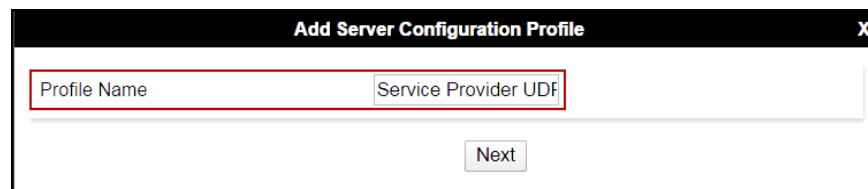
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya-SM ▼
Signaling Manipulation Script	None ▼
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None ▼



### 7.8.2. Server Configuration Profile – Service Provider

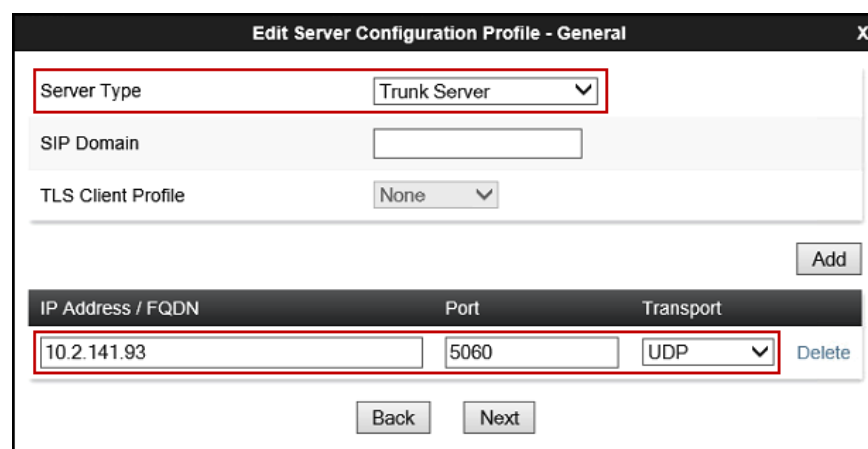
Similarly, to add the profile for the Trunk Server, click the **Add** button on the **Server Configuration** screen (not shown).

- Enter an appropriate **Profile Name** similar to the screen below.
- Click **Next**.



Dialog box titled "Add Server Configuration Profile" with a close button (X). It contains two input fields: "Profile Name" and "Service Provider UDF". Below the fields is a "Next" button.

- On the **Edit Server Configuration Profile - General** Tab select **Trunk Server** from the drop down menu for the **Server Type**.
- On the **IP Addresses / FQDN** field, enter Telmex's proxy IP address (See figure 1).
- Enter **5060** under **Port**, and select **UDP** for **Transport** for both entries.
- Click **Next**.

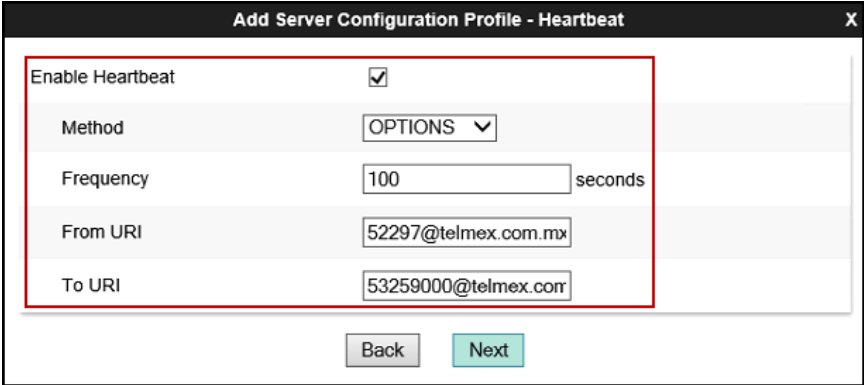


Dialog box titled "Edit Server Configuration Profile - General" with a close button (X). It contains several fields: "Server Type" (dropdown menu with "Trunk Server" selected), "SIP Domain" (text field), and "TLS Client Profile" (dropdown menu with "None" selected). Below these is an "Add" button. At the bottom, there is a table with three columns: "IP Address / FQDN", "Port", and "Transport". The first row contains "10.2.141.93", "5060", and "UDP" (dropdown menu). To the right of the table is a "Delete" button. At the very bottom are "Back" and "Next" buttons.

Click next on the **Add Server Configuration Profile - Authentication** window.

On the **Add Server Configuration Profile - Heartbeat** window:

- Check the **Enable Heartbeat** box.
- Under **Method**, select **OPTIONS** from the drop down menu.
- **Frequency**: Enter the amount of time (in seconds), **100** seconds was the value used during the compliance test.
- The **From URI** and **To URI** entries for the OPTION messages are built using the following:
  - **From URI**: An internal extension number was entered (52297) and the enterprise domain name (*telmex.com.mx*), as shown on the screen below.
  - **To URI**: An external extension number was entered (53259000) and the Service Provider's domain name (*telmex.com*), as shown on the screen below.
- Click **Next**.



Enable Heartbeat	<input checked="" type="checkbox"/>
Method	OPTIONS ▼
Frequency	100 seconds
From URI	52297@telmex.com.mx
To URI	53259000@telmex.com

Back Next

Click next on the **Add Server Configuration Profile - Ping** window

- On the **Add Server Configuration Profile - Advanced** tab, select *SP-General* from the **Interworking Profile** and *change\_sendonly* from the **Signaling Manipulation Script** windows created in **Section 7.7**.
- Click **Finish**.

**Add Server Configuration Profile - Advanced** X

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	SP-General ▼
Signaling Manipulation Script	change_sendonly ▼
Securable	<input type="checkbox"/>
Enable FGDN	<input type="checkbox"/>
TCP Failover Port	5060
TLS Failover Port	5061
Tolerant	<input type="checkbox"/>
URI Group	None ▼

Back Finish

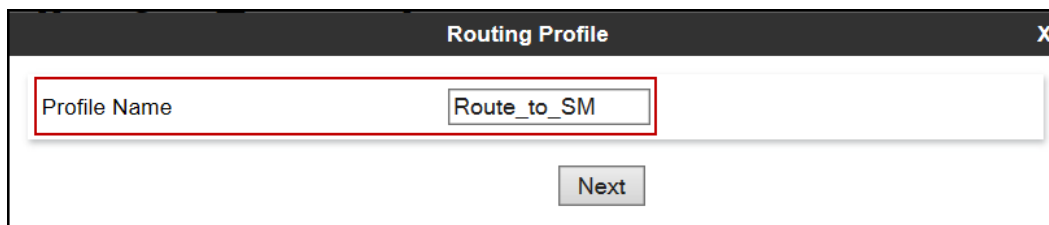
## 7.9. Routing

Routing profiles define a specific set of routing criteria that is used, in addition to other types of domain policies, to determine the path that the SIP traffic will follow as it flows through the Avaya SBCE interfaces. Two Routing Profiles were created in the test configuration, one for inbound calls, with Session Manager as the destination, and the second one for outbound calls, which are routed to the service provider SIP trunk.

### 7.9.1. Routing Profile – Enterprise

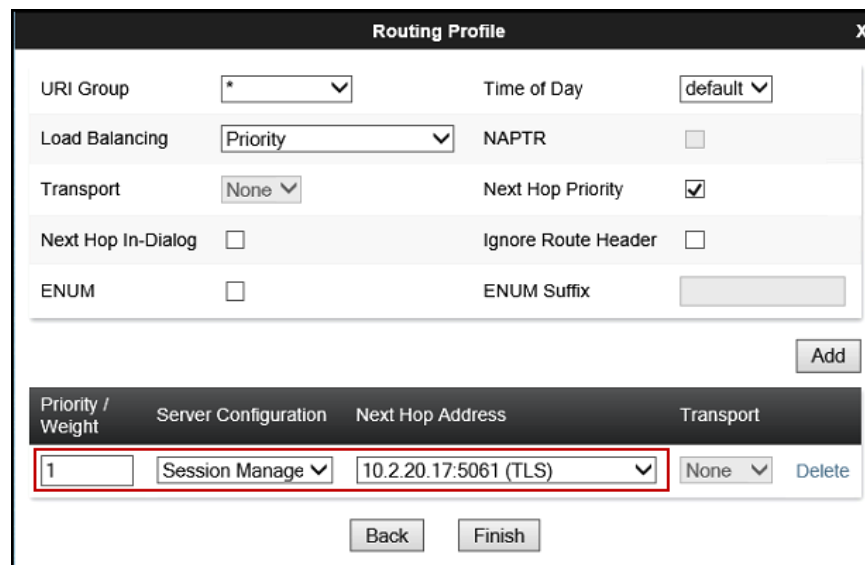
To create the inbound route, select the **Routing** tab from the **Global Profiles** menu on the left-hand side and select **Add** (not shown).

- Enter an appropriate **Profile Name** similar to the example below.
- Click **Next**.



The screenshot shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "Route\_to\_SM". Below the input field is a button labeled "Next".

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address.
- Under **Server Configuration**, select *Session Manager*. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Session Manager Server Configuration Profile in **Section 7.8.1**.
- Defaults were used for all other parameters.
- Click **Finish**.



The screenshot shows the "Routing Profile" dialog box with various configuration options. The "URI Group" is set to "\*", "Time of Day" is "default", "Load Balancing" is "Priority", "Transport" is "None", "Next Hop In-Dialog" is unchecked, "ENUM" is unchecked, "NAPTR" is unchecked, "Next Hop Priority" is checked, "Ignore Route Header" is unchecked, and "ENUM Suffix" is empty. The "Add" button is visible. Below the configuration options is a table with the following data:

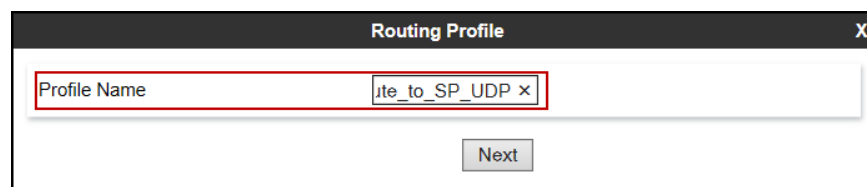
Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Session Manage	10.2.20.17:5061 (TLS)	None

The "Back" and "Finish" buttons are at the bottom of the dialog.

### 7.9.2. Routing Profile – Service Provider

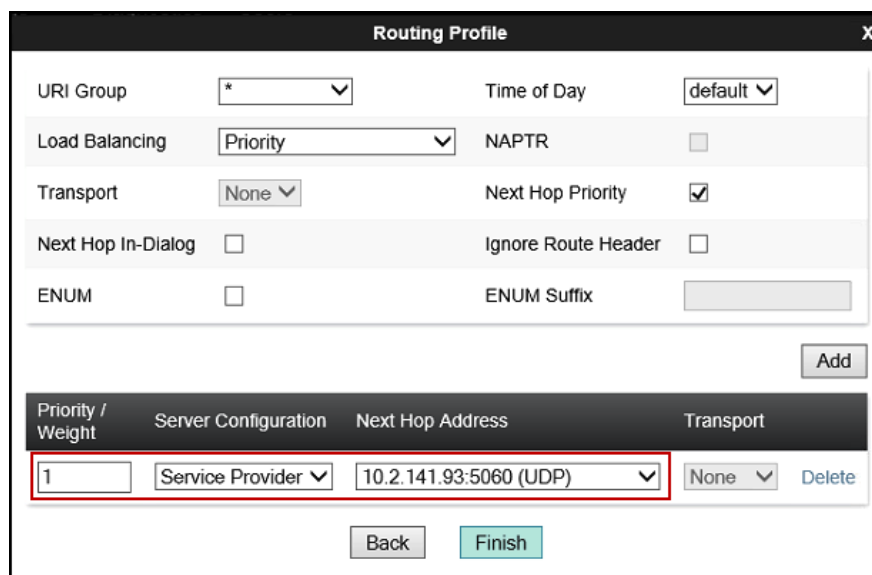
Back at the **Routing** tab, select **Add** (not shown) to repeat the process in order to create the outbound route.

- Enter an appropriate **Profile Name** similar to the example below.
- Click **Next**.



The screenshot shows a dialog box titled "Routing Profile" with a close button (X) in the top right corner. Inside the dialog, there is a text input field labeled "Profile Name" containing the text "ite\_to\_SP\_UDP". To the right of the input field is a small "X" icon. Below the input field is a "Next" button.

- On the **Routing Profile** tab, click the **Add** button to enter the next-hop address. Under **Server Configuration**, select the Server Configuration for the Service Provider created in **Section 7.8.2**. The **Next Hop Address** field will be populated with the IP address, port and protocol defined for the Service Provider Server Configuration Profile in **Section 7.8.2**.
- Defaults were used for all other parameters.
- Click **Finish**.



The screenshot shows the "Routing Profile" dialog box with various configuration options. The "URI Group" is set to "\*", "Time of Day" is "default", "Load Balancing" is "Priority", "NAPTR" is unchecked, "Transport" is "None", "Next Hop Priority" is checked, "Next Hop In-Dialog" is unchecked, "Ignore Route Header" is unchecked, "ENUM" is unchecked, and "ENUM Suffix" is empty. An "Add" button is visible. Below these options is a table with the following columns: "Priority / Weight", "Server Configuration", "Next Hop Address", and "Transport". The table contains one row with the following values: "1", "Service Provider", "10.2.141.93:5060 (UDP)", and "None". A "Delete" button is next to the "Transport" column. At the bottom of the dialog are "Back" and "Finish" buttons.

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	Service Provider	10.2.141.93:5060 (UDP)	None

## 7.10. Topology Hiding

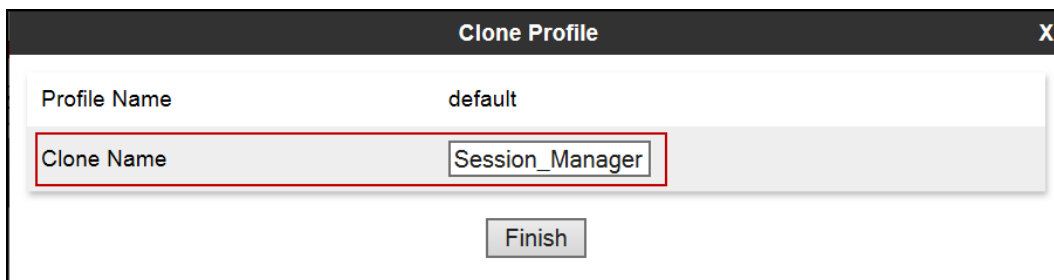
Topology Hiding is a security feature that allows the modification of several SIP headers, preventing private enterprise network information from being propagated to the untrusted public network.

Topology Hiding can also be used as an interoperability tool to adapt the host portion in the SIP headers to the IP addresses or domains expected on the service provider and the enterprise networks. For the compliance test, the default Topology Hiding Profile was cloned and modified accordingly. Only the minimum configuration required to achieve interoperability on the SIP trunk was performed. Additional steps can be taken in this section to further mask the information that is sent from the enterprise to the public network.

### 7.10.1. Topology Hiding Profile – Enterprise

To add the Topology Hiding Profile in the enterprise direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

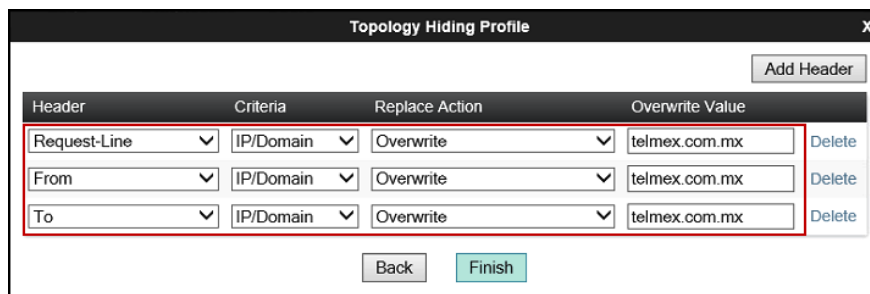
- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



The screenshot shows a 'Clone Profile' dialog box. It has a title bar with 'Clone Profile' and a close button 'X'. Inside, there are two input fields: 'Profile Name' with the value 'default' and 'Clone Name' with the value 'Session\_Manager'. The 'Clone Name' field is highlighted with a red rectangular box. Below these fields is a 'Finish' button.

On the newly cloned *Session\_Manager* profile screen, click the **Edit** button (not shown).

- For the, **From**, **To** and **Request-Line** headers, select *Override* in the **Replace Action** column and enter the enterprise SIP domain *telmex.com.mx*, in the **Override Value** column of these headers, as shown below. This is the domain known by Session Manager, defined in **Section 6.2**.
- Default values were used for all other fields.
- Click **Finish**.



The screenshot shows the 'Topology Hiding Profile' configuration screen. It has a title bar with 'Topology Hiding Profile' and a close button 'X'. There is an 'Add Header' button in the top right. Below it is a table with the following columns: Header, Criteria, Replace Action, and Override Value. The table contains three rows:

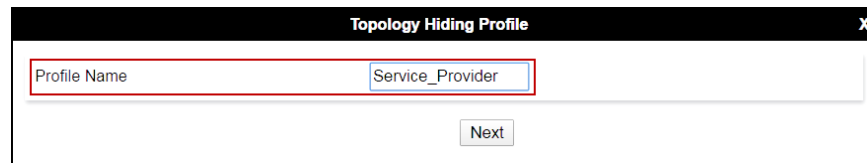
Header	Criteria	Replace Action	Override Value
Request-Line	IP/Domain	Override	telmex.com.mx
From	IP/Domain	Override	telmex.com.mx
To	IP/Domain	Override	telmex.com.mx

The first row is highlighted with a red rectangular box. To the right of each row is a 'Delete' button. At the bottom of the screen are 'Back' and 'Finish' buttons.

### 7.10.2. Topology Hiding Profile – Service Provider.

To add the Topology Hiding Profile in the Service Provider direction, select **Topology Hiding** from the **Global Profiles** menu on the left-hand side, select *default* from the list of pre-defined profiles and click the **Clone** button (not shown).

- Enter a **Clone Name** such as the one shown below.
- Click **Finish**.



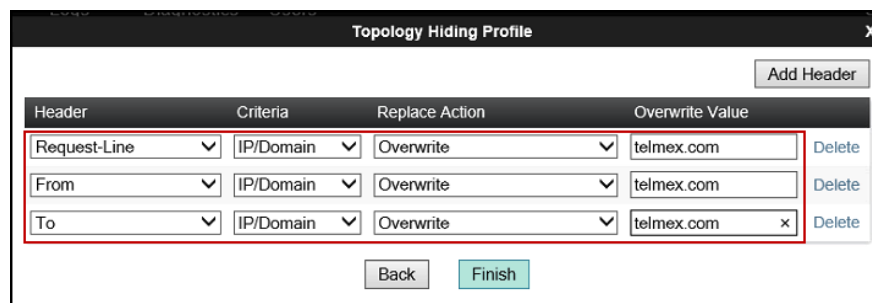
Topology Hiding Profile

Profile Name: Service\_Provider

Next

On the newly cloned *Service\_Provider* profile screen, click the **Edit** button (not shown).

- For the, **From**, **To** and **Request-Line** headers, select *Overwrite* in the **Replace Action** column and enter the Service Provider's SIP domain *telmex.com*, in the **Overwrite Value** column of these headers, as shown below.
- Default values were used for all other fields.
- Click **Finish**.



Topology Hiding Profile

Add Header

Header	Criteria	Replace Action	Overwrite Value	
Request-Line	IP/Domain	Overwrite	telmex.com	Delete
From	IP/Domain	Overwrite	telmex.com	Delete
To	IP/Domain	Overwrite	telmex.com	Delete

Back Finish

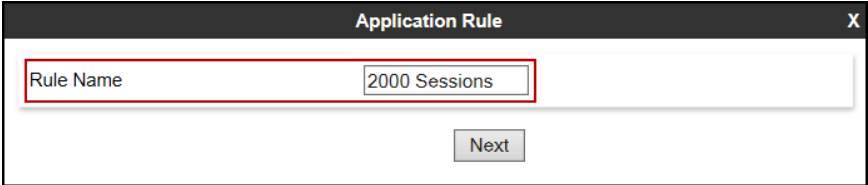
## 7.11. Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signaling, Security, etc.

### 7.11.1. Application Rules

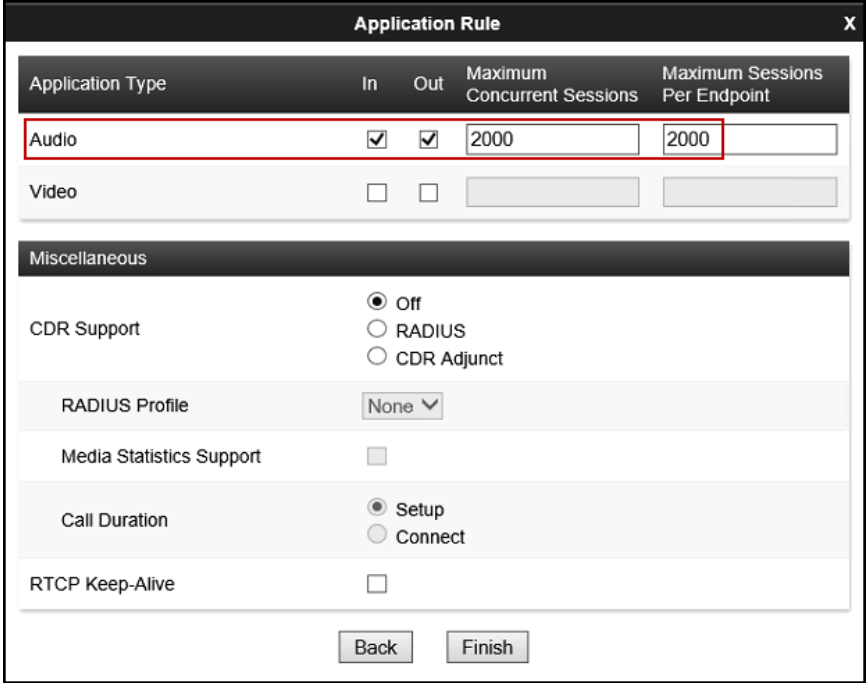
Application Rules define which types of SIP-based Unified Communications (UC) applications the UC-Sec security device will protect: voice, video, and/or Instant Messaging (IM). In addition, Application Rules define the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion. From the menu on the left-hand side, select **Domain Policies** → **Application Rules**, Click on the **Add** button to add a new rule.

- Under **Rule Name** enter the name of the profile, e.g., *2000 Sessions*.
- Click **Next**.



The screenshot shows a window titled "Application Rule" with a close button (X) in the top right corner. Inside the window, there is a text input field labeled "Rule Name" which contains the text "2000 Sessions". This field is highlighted with a red rectangular box. Below the input field, there is a "Next" button.

- Under **Audio** check *In* and *Out* and set the **Maximum Concurrent Sessions** and **Maximum Sessions Per Endpoint** to recommended values, the values of *2000* for Audio was used in the sample configuration.
- Click **Finish**.



The screenshot shows a window titled "Application Rule" with a close button (X) in the top right corner. The window contains a table with the following columns: "Application Type", "In", "Out", "Maximum Concurrent Sessions", and "Maximum Sessions Per Endpoint". The "Audio" row is highlighted with a red rectangular box. In this row, the "In" and "Out" checkboxes are checked, and the values for "Maximum Concurrent Sessions" and "Maximum Sessions Per Endpoint" are both set to "2000". Below the table, there is a "Miscellaneous" section with several options: "CDR Support" (radio buttons for Off, RADIUS, CDR Adjunct), "RADIUS Profile" (a dropdown menu set to "None"), "Media Statistics Support" (checkbox), "Call Duration" (radio buttons for Setup, Connect), and "RTCP Keep-Alive" (checkbox). At the bottom of the window, there are "Back" and "Finish" buttons.



## 7.11.2. Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product. For the compliance test, two media rules (shown below) were used; one toward Session Manager and one toward the Service Provider.

To add a media rule in the Session Manager direction, from the menu on the left-hand side, select **Domain Policies → Media Rules**.

- Click on the **Add** button to add a new media rule (not shown).
- Under **Rule Name** enter **SM\_SRTP** (not shown).
- Click **Next** (not shown).
- Under Audio Encryption, **Preferred Format #1**, select **SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80**.
- Under Audio Encryption, **Preferred Format #2**, select **RTP**.
- Under Audio Encryption, uncheck **Encrypted RTCP**.
- Under Audio Encryption, check **Interworking**.
- Repeat the above steps under Video Encryption.
- Under Miscellaneous verify that **Capability Negotiation** is checked.
- Click **Next**.

OSOCS Users

Media Encryption

Audio Encryption

Preferred Format #1 SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80

Preferred Format #2 RTP

Preferred Format #3 NONE

Encrypted RTCP ☐

MKI ☐

Lifetime  
Leave blank to match any value. 2^A

Interworking ☒

Video Encryption

Preferred Format #1 SRTP\_AES\_CM\_128\_HMAC\_SHA1\_80

Preferred Format #2 RTP

Preferred Format #3 NONE

Encrypted RTCP ☐

MKI ☐

Lifetime  
Leave blank to match any value. 2^A

Interworking ☒

Miscellaneous

Capability Negotiation ☒

Finish

- Accept default values in the remaining sections by clicking **Next** (not shown), and then click **Finish** (not shown).

For the compliance test, the **default-low-med** Media Rule was used in the Service Provider direction.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms (1), Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. On the left, a sidebar menu lists various configuration areas, with 'Domain Policies' and 'Media Rules' highlighted. The main content area is titled 'Media Rules: default-low-med' and features a list of media rules on the left and a configuration panel on the right. The configuration panel has tabs for 'Encryption', 'Codec Prioritization', 'Advanced', and 'QoS'. The 'Encryption' tab is active, showing settings for Audio and Video Encryption, including Preferred Formats (RTP) and Interworking (checked). A warning message at the top of the configuration panel states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' An 'Edit' button is located at the bottom right of the configuration panel.

### 7.11.3. Signaling Rules

For the compliance test, the **default** signaling rule was used.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left sidebar contains a navigation menu with 'Domain Policies' expanded, showing 'Signaling Rules' selected. The main content area is titled 'Signaling Rules: default'. It includes a 'Filter By Device...' dropdown and a 'Clone' button. A warning message states: 'It is not recommended to edit the defaults. Try cloning or adding a new rule instead.' Below this, there are tabs for 'General', 'Requests', 'Responses', 'Request Headers', 'Response Headers', 'Signaling QoS', and 'UCID'. The 'General' tab is active, showing 'Inbound' and 'Outbound' sections. Each section has a table with columns for 'Requests', 'Non-2XX Final Responses', 'Optional Request Headers', and 'Optional Response Headers', all set to 'Allow'. The 'Content-Type Policy' section has 'Enable Content-Type Checks' checked. The 'Exception List' is empty. An 'Edit' button is at the bottom.

### 7.12. End Point Policy Groups

End Point Policy Groups associate the different sets of rules under Domain Policies (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. Please note that changes should not be made to any of the default rules used in these End Point Policy Groups.

#### 7.12.1. End Point Policy Group – Enterprise

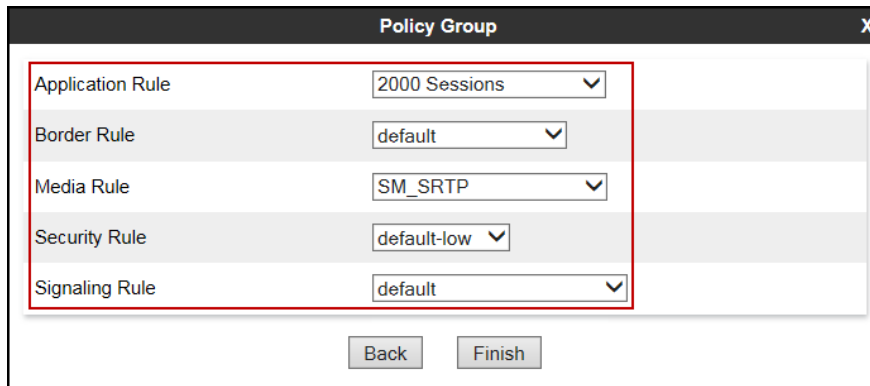
To create an End Point Policy Group for the enterprise, select **End Point Policy Groups** under the **Domain Policies** menu and select **Add** (not shown).

- Enter an appropriate name in the **Group Name** field.
- Click **Next**.

The screenshot shows a 'Policy Group' dialog box. It has a title bar with 'Policy Group' and a close button 'X'. Inside, there is a 'Group Name' label and a text input field containing the word 'Enterprise'. The input field is highlighted with a red rectangular box. Below the input field is a 'Next' button.

Under the **Policy Group** tab enter the following:

- **Application Rule:** *2000 Sessions* (Section 7.11.1).
- **Border Rule:** *default*.
- **Media Rule:** *SM\_SRTP* (Section 7.11.2).
- **Security Rule:** *default-low*.
- **Signaling Rule:** *default* (Section 7.11.3).
- Click **Finish**.



The screenshot shows a window titled "Policy Group" with a close button (X) in the top right corner. Inside the window, there is a list of five rules, each with a label and a dropdown menu. A red rectangular box highlights the first four rules: Application Rule, Border Rule, Media Rule, and Security Rule. The values selected in the dropdowns are "2000 Sessions", "default", "SM\_SRTP", and "default-low" respectively. The fifth rule, Signaling Rule, has a value of "default" and is not highlighted. At the bottom of the window, there are two buttons: "Back" and "Finish".

Rule Type	Selected Value
Application Rule	2000 Sessions
Border Rule	default
Media Rule	SM_SRTP
Security Rule	default-low
Signaling Rule	default

### 7.12.2. End Point Policy Group – Service Provider

A second End Point Policy Group was created for the service provider, repeating the steps previously described. In the Policy Group tab, all fields used one of the default sets already pre-defined in the configuration, except for the Media Rule, which was set to *default-low-med* (Section 7.11.2).

The screen below shows the End Point Policy Group named *Service Provider* after the configuration was completed.

**Session Border Controller for Enterprise** AVAYA

Alarms **1** Incidents Status ▾ Logs ▾ Diagnostics Users Settings ▾ Help ▾ Log Out

**Policy Groups: Service Provider**

[Add](#) [Filter By Device...](#) [Rename](#) [Clone](#) [Delete](#)

**Policy Groups**

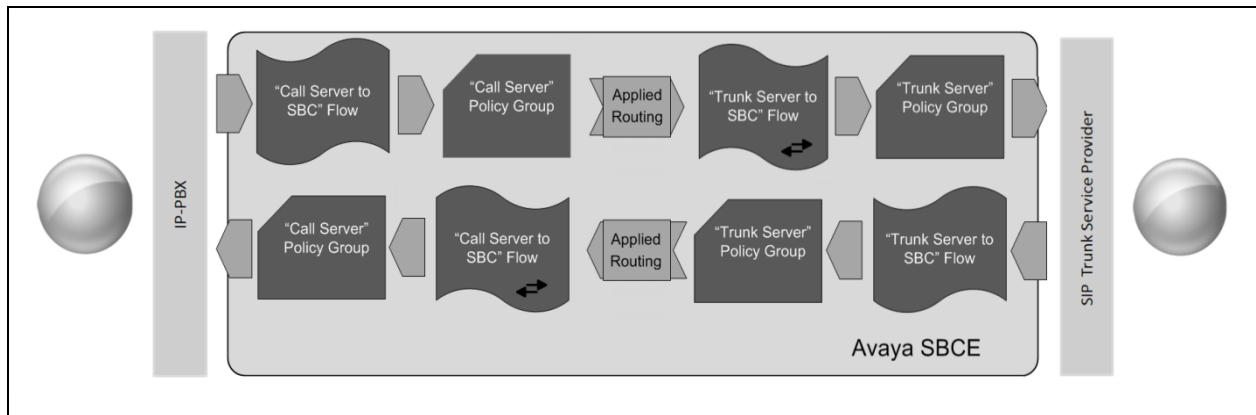
- default-low
- default-low-enc
- default-med
- default-med-enc
- default-high
- default-high-enc
- OCS-default-high
- avaya-def-low-enc
- avaya-def-high-s...
- avaya-def-high-s...
- Enterprise
- Service Provider**

**Policy Group** [Summary](#)

Order	Application	Border	Media	Security	Signaling	
1	2000 Sessions	default	default-low-med	default-low	default	<a href="#">Edit</a>

### 7.13. End Point Flows

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



The **End-Point Flows** defines certain parameters that pertain to the signaling and media portions of a call, whether it originates from within the enterprise or outside of the enterprise.

### 7.13.1. End Point Flow – Enterprise

To create the call flow toward the enterprise, from the **Device Specific** menu, select **End Point Flows**, then select the **Server Flows** tab. Click **Add** (not shown). The screen below shows the flow named *Session\_Manager\_Flow* created in the sample configuration. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for the Service Provider in **Section 7.9.2**, which is the reverse route of the flow. Click **Finish**.

Edit Flow: Session_Manager_Flow	
Flow Name	Session_Manager_Flow
Server Configuration	Session Manager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Public_sig
Signaling Interface	PrivateSig
Media Interface	PrivateMed
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	Route_to_SP_UDP
Topology Hiding Profile	Session_Manager
Signaling Manipulation Script	None
Remote Branch Office	Any

Finish

### 7.13.2. End Point Flow – Service Provider

A second Server Flow with the name *SIP\_Trunk\_Flow\_UDP* was similarly created in the Service Provider direction. The flow uses the interfaces, policies, and profiles defined in previous sections. Note that the **Routing Profile** selection is the profile created for Session Manager in **Section 7.9.1**, which is the reverse route of the flow. Also note that there is no selection under the **Signaling Manipulation Script** field. Click **Finish**.

Edit Flow: SIP_Trunk_Flow_UDP	
Flow Name	SIP_Trunk_Flow_UDP
Server Configuration	Service Provider UDP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	PrivateSig
Signaling Interface	Public_sig
Media Interface	PublicMed
Secondary Media Interface	None
End Point Policy Group	Service Provider
Routing Profile	Route_to_SM
Topology Hiding Profile	Service_Provider
Signaling Manipulation Script	None
Remote Branch Office	Any

Finish



## 8. Telmex SIP Trunk Service Configuration

To use Telmex SIP Trunk Service, a customer must request the service from Telmex using the established sales processes. The process can be started by contacting Telmex via the corporate web site at: <http://telmex.com/en/web/empresas>

During the signup process, Telmex and the customer will discuss details about the preferred method to be used to connect the customer's enterprise network to Telmex's network. Telmex will provide their SIP Proxy public IP address, Direct Inward Dialed (DID) numbers to be assigned to the enterprise, etc. This information is used to complete the Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise configuration discussed in the previous sections.

## 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 commands that can be used to troubleshoot the solution.

### 9.1. General Verification Steps

- 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.
- 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.
- Verify that the user on the PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

### 9.2. Communication Manager Verification

The following commands can be entered in the Communication Manager SAT terminal to verify the SIP trunk functionality:

- **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 signaling-group** <signaling group number>  
Displays signaling group service state.
- **status trunk** <trunk group number>  
Displays trunk group service state.
- **status station** <extension number>  
Displays signaling and media information for an active call on a specific station.

### 9.3. Session Manager Verification

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager** → **System Status** → **SIP Entity Monitoring**. Click the Session Manager instance (*smS* in the example below).

**SIP Entity Link Monitoring Status Summary**

This page provides a summary of Session Manager SIP entity link monitoring status.

**SIP Entities Status for All Monitoring Session Manager Instances**

Run Monitor

1 Items | Refresh Filter: Enable

Session Manager	Type	Monitored Entities						Total
		Down	Partially	Up	Not	Deny		
<input checked="" type="checkbox"/> smS	Core	0	0	4	0	0	4	

Select: All, None

Verify that the state of the Session Manager links to Communication Manager and the Avaya SBCE under the **Conn. Status** and **Link Status** columns is **UP**, like shown on the screen below.

**Session Manager Entity Link Connection Status**

This page displays detailed connection status for all entity links from a Session Manager.

**All Entity Links for Session Manager: smS**

Summary View

Status Details for the selected Session Manager:

4 Items | Refresh Filter: Enable

SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input checked="" type="radio"/> smS	10.2.20.12	5061	TLS	FALSE	UP	200 OK	UP
<input checked="" type="radio"/> SBC Telmex	10.2.141.93	5060	UDP	FALSE	UP	200 OK	UP
<input checked="" type="radio"/> cmn	10.2.20.26	5061	TLS	FALSE	UP	200 OK	UP
<input checked="" type="radio"/> AvayaSBC	10.2.20.19	5061	TLS	FALSE	UP	200 OK	UP

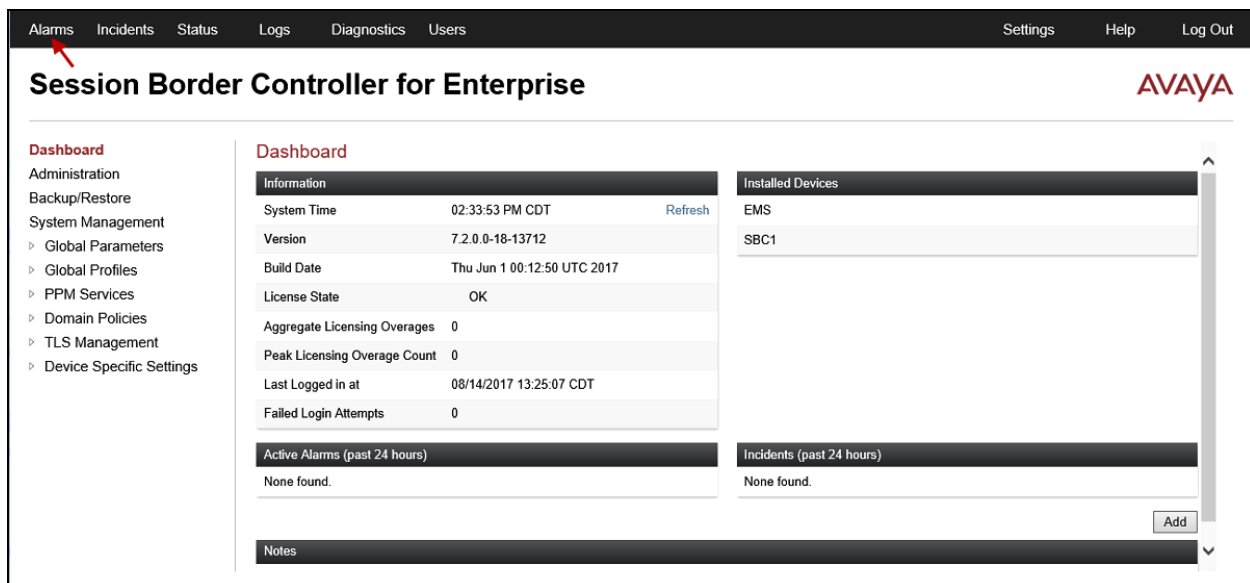
Other Session Manager useful verification and troubleshooting tools include:

- **traceSM** – Session Manager command line tool for traffic analysis. Login to the Session Manager command line management interface to run this command.
- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, from the System Manager Home screen navigate to **Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.

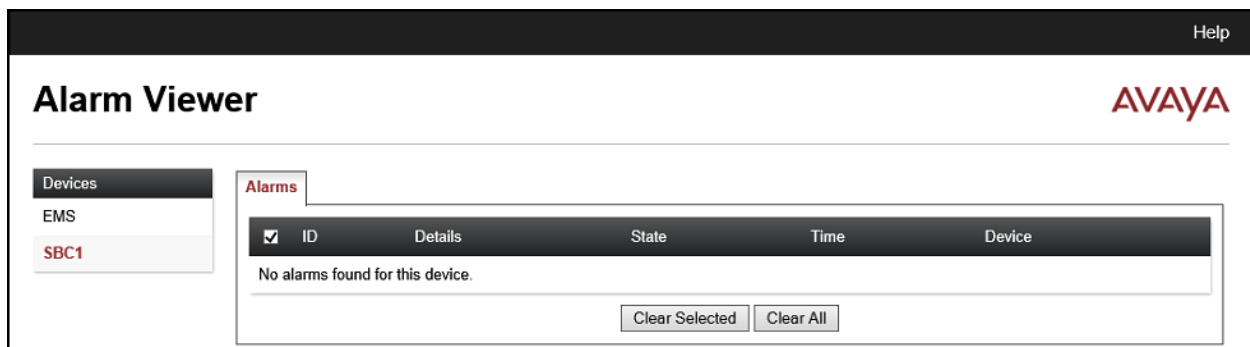
## 9.4. Avaya SBCE Verification

There are several links and menus located on the taskbar at the top of the screen of the web interface that can provide useful diagnostic or troubleshooting information.

**Alarms:** This screen provides information about the health of the SBC.



The following screen shows the **Alarm Viewer** page.



**Incidents** : Provides detailed reports of anomalies, errors, policies violations, etc.

**Session Border Controller for Enterprise**

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

**Dashboard**

- Administration
- Backup/Restore
- System Management
  - Global Parameters
  - Global Profiles
  - PPM Services
  - Domain Policies
  - TLS Management
  - Device Specific Settings

**Information**

System Time	02:33:53 PM CDT	Refresh
Version	7.2.0.0-18-13712	
Build Date	Thu Jun 1 00:12:50 UTC 2017	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	08/14/2017 13:25:07 CDT	
Failed Login Attempts	0	

**Installed Devices**

EMS
SBC1

**Active Alarms (past 24 hours)**

None found.

**Incidents (past 24 hours)**

None found.

Notes

Add

The following screen shows the Incident Viewer page.

**Incident Viewer**

Help

Device All Category Licensing Clear Filters Refresh Generate Report

Displaying results 0 to 0 out of 0.

Type	ID	Date	Time	Category	Device	Cause
No incidents found.						

<< < 1 > >>

**Diagnostics:** This screen provides a variety of tools to test and troubleshoot the Avaya SBCE network connectivity.

**Session Border Controller for Enterprise**

**Dashboard**

**Information**

System Time	02:33:53 PM CDT	<a href="#">Refresh</a>
Version	7.2.0.0-18-13712	
Build Date	Thu Jun 1 00:12:50 UTC 2017	
License State	OK	
Aggregate Licensing Overages	0	
Peak Licensing Overage Count	0	
Last Logged in at	08/14/2017 13:25:07 CDT	
Failed Login Attempts	0	

**Installed Devices**

EMS
SBC1

**Active Alarms (past 24 hours)**

None found.

**Incidents (past 24 hours)**

None found.

[Add](#)

**Notes**

The following screen shows the Diagnostics page with the results of a successful ping test.

**Diagnostics**

**Pinging 10.2.20.17**

Average ping from 10.2.20.19 [A1] to 10.2.20.17 is 0.211ms.

**Full Diagnostic** **Ping Test**

Outgoing pings from this device can only be sent via the primary IP (determined by the OS) of each respective interface or VLAN.

Source Device / IP: A1

Destination IP: 10.2.20.17

[Ping](#)

Additionally, the Avaya SBCE contains an internal packet capture tool that allows the capture of packets on any of its interfaces, saving them as *pcap* files. Navigate to **Device Specific Settings** → **Troubleshooting** → **Trace**. Select the **Packet Capture** tab, set the desired configuration for the trace and click **Start Capture**.

The screenshot displays the Avaya SBCE web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the AVAYA logo on the right.

On the left sidebar, the navigation menu is expanded to "Device Specific Settings", which includes sub-items like Network Management, Media Interface, Signaling Interface, End Point Flows, Session Flows, DMZ Services, TURN/STUN Service, SNMP, Syslog Management, Advanced Options, Troubleshooting, and DoS Learning. The "Troubleshooting" item is further expanded to show "Debugging" and "Trace". The "Trace" option is selected and highlighted.

The main content area is titled "Trace: : Avaya\_SBCE". It features two tabs: "Devices" and "Packet Capture". The "Packet Capture" tab is active. Below the tabs, a blue notification box states: "A packet capture is currently in progress. This page will automatically refresh until the capture completes."

The "Packet Capture Configuration" section contains the following fields and controls:

- Status:** In Progress
- Interface:** Any (dropdown menu)
- Local Address:** All (dropdown menu) : (text input field)
- Remote Address:** \* (text input field, with a note: "\*, \*.Port, IP, IP:Port")
- Protocol:** All (dropdown menu)
- Maximum Number of Packets to Capture:** 10000 (text input field)
- Capture Filename:** Test\_Capture.pcap (text input field, with a note: "Using the name of an existing capture will overwrite it.")
- Stop Capture:** (button)

Once the capture is stopped, click the **Captures** tab and select the proper *pcap* file. Note that the date and time is appended to the filename specified previously. The file can now be saved to the local PC, where it can be opened with an application such as Wireshark.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. On the left, a sidebar menu lists various system management options, with 'Device Specific Settings' and 'Troubleshooting' expanded. The 'Trace' option under Troubleshooting is highlighted. The main content area is titled 'Trace: : Avaya\_SBCE' and features two tabs: 'Packet Capture' and 'Captures'. The 'Captures' tab is active, showing a table of captured files. The table has columns for File Name, File Size (bytes), and Last Modified. A single entry is listed: 'Test\_Capture\_20170705164248.pcap' with a size of 172,032 bytes and a timestamp of July 5, 2017 4:43:08 PM EDT. A 'Delete' link is provided for this entry. A 'Refresh' button is located at the top right of the table.

File Name	File Size (bytes)	Last Modified
Test_Capture_20170705164248.pcap	172,032	July 5, 2017 4:43:08 PM EDT

## 10. Conclusion

These Application Notes describe the procedures required to configure Avaya Aura® Communication Manager 7.0, Avaya Aura® Session Manager 7.0 and Avaya Session Border Controller for Enterprise 7.2, to connect to the Telmex SIP Trunk service, as shown in **Figure 1**.

Interoperability testing of the sample configuration was completed with successful results for all test cases with the observations/limitations described in **Sections 2.1** and **2.2**.

## 11. References

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

- [1] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0.1, August 2016.
- [2] *Deploying Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2.1, October 2016.
- [3] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, August 2016, Document Number 03-300509.
- [4] *Administering Avaya Aura® System Manager* for Release 7.0.1, Issue 3, January 2017.
- [5] *Deploying Avaya Aura® System Manager*, Release 7.0.1, August 2016.
- [6] *Deploying Avaya Aura® Session Manager*, Release 7.0.1, Issue 3, November 2016.
- [7] *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016.
- [8] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.2, Issue 3, August 2017.
- [9] *Administering Avaya Session Border Controller for Enterprise*, Release 7.2, August 2017.
- [10] *Configuring Remote Workers with Avaya Session Border Controller for Enterprise Rel. 7.0, Avaya Aura® Communication Manager Rel. 7.0 and Avaya Aura® Session Managers Rel. 7.0 - Issue 1.0*.
- [11] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>
- [12] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>



## 12. Appendix A: SigMa Script

Following is the Signaling Manipulation scripts that was used in the configuration of the Avaya SBCE, **Section 7.8.2**. When adding these scripts as instructed in **Section 7.7** enter a name for the script in the Title (e.g., **change\_sendonly**) and copy/paste the scripts as shown below.

```
within session "INVITE" {  
  act on request where $DIRECTION="OUTBOUND" and $ENTRY_POINT="POST_ROUTING"  
  {  
  
    //Removes the a=sendonly from the re-INVITE messages.  
  
    $BODY[1].regex_replace("a=sendonly","a=sendrecv");  
  
  }  
}
```

---

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