



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R7.0, Avaya Aura® Session Manager R7.0 and Avaya Session Border Controller for Enterprise R7.0 to support SFR TELEPHONIE SIP - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the SFR TELEPHONIE SIP service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. SFR is a member of the DevConnect Service Provider program.

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 Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the SFR TELEPHONIE SIP service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Aura® Communication Manager R7.0 (Communication Manager); Avaya Aura® Session Manager R7.0 (Session Manager); Avaya Session Border Controller for Enterprise R7.0 (Avaya SBCE). Note that the shortened names shown in brackets will be used throughout the remainder of the document. Customers using this Avaya SIP-enabled enterprise solution with the TELEPHONIE SIP service are able to place and receive Public Switched Telephone Network (PSTN) calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and Avaya SBCE. The enterprise site was configured to connect to the TELEPHONIE SIP service.

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

The interoperability test included the following:

- Incoming calls to the enterprise site from PSTN phones using TELEPHONIE SIP, calls made to SIP and H.323 telephones at the enterprise.
- Outgoing calls from the enterprise site completed via TELEPHONIE SIP to PSTN destinations, calls made from SIP and H.323 telephones.
- Inbound and outbound PSTN calls to/from an Avaya one-X® Communicator soft phone and Avaya Communicator for Windows client.
- Calls using the G.729A and G.711 A-Law codecs.
- Fax calls to/from a Group 3 fax machine to a PSTN connected fax machine using T.38.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media between the Avaya SBCE and the SIP and H.323 telephones.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by the TELEPHONIE SIP trunk requiring Avaya response, and sent by Avaya requiring SFR response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the TELEPHONIE SIP service with the following observations:

- Initially, ringback was not heard on some outbound calls. This was resolved by setting Initial IP-IP Direct Media to “n” on Communication Manager allowing early media and ringback from the far end.
- Communication Manager clears unanswered calls with a “480 Temporarily Unavailable” message after 3 minutes. The SFR network releases the call after the same period of time with a CANCEL message. During testing, the CANCEL from the network was not received at Communication Manager, and the 480 Temporarily Unavailable was not received at the network. This resulted in unnecessary re-transmission of messages, but the call clearing itself was acceptable.
- No inbound Toll-Free access was available for testing.
- When calling from the PSTN to a Communication Manager extension with EC500, ringback was heard from both Communication Manager and the mobile phone network.
- When making a an outbound call from one-X Communicator configured to connect via SIP, no ringback was heard when in “Other Phone” mode,
- When making an inbound call to Communication Manager where there are no free signalling resources available, Communication manager sends “503 Service Unavailable”. Instead of failing the call at that point, the network re-attempts call set-up for approximately 10 seconds resulting in a period of silence before the call fails.
- When making an inbound call to Communication Manager where the signalling link has failed, Session Manager sends “503 Service Unavailable”. Instead of failing the call at that point, the network re-attempts call set-up for approximately 20 seconds resulting in a period of silence before the call fails.

2.3. Support

Le Service Technique SFR Business Team est joignable 24H/24, 7J/7 par un numéro gratuit pour signalisation des incidents techniques sur le service Collecte SIP.

CENTRE SERVICE CLIENT SFR Business Team

0 800 950 920

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the TELEPHONIE SIP service. Located at the Enterprise site is an Avaya SBCE, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 16xx series IP telephones (with H.323 firmware), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator softphone and Avaya Communicator for Windows running on laptop PCs.

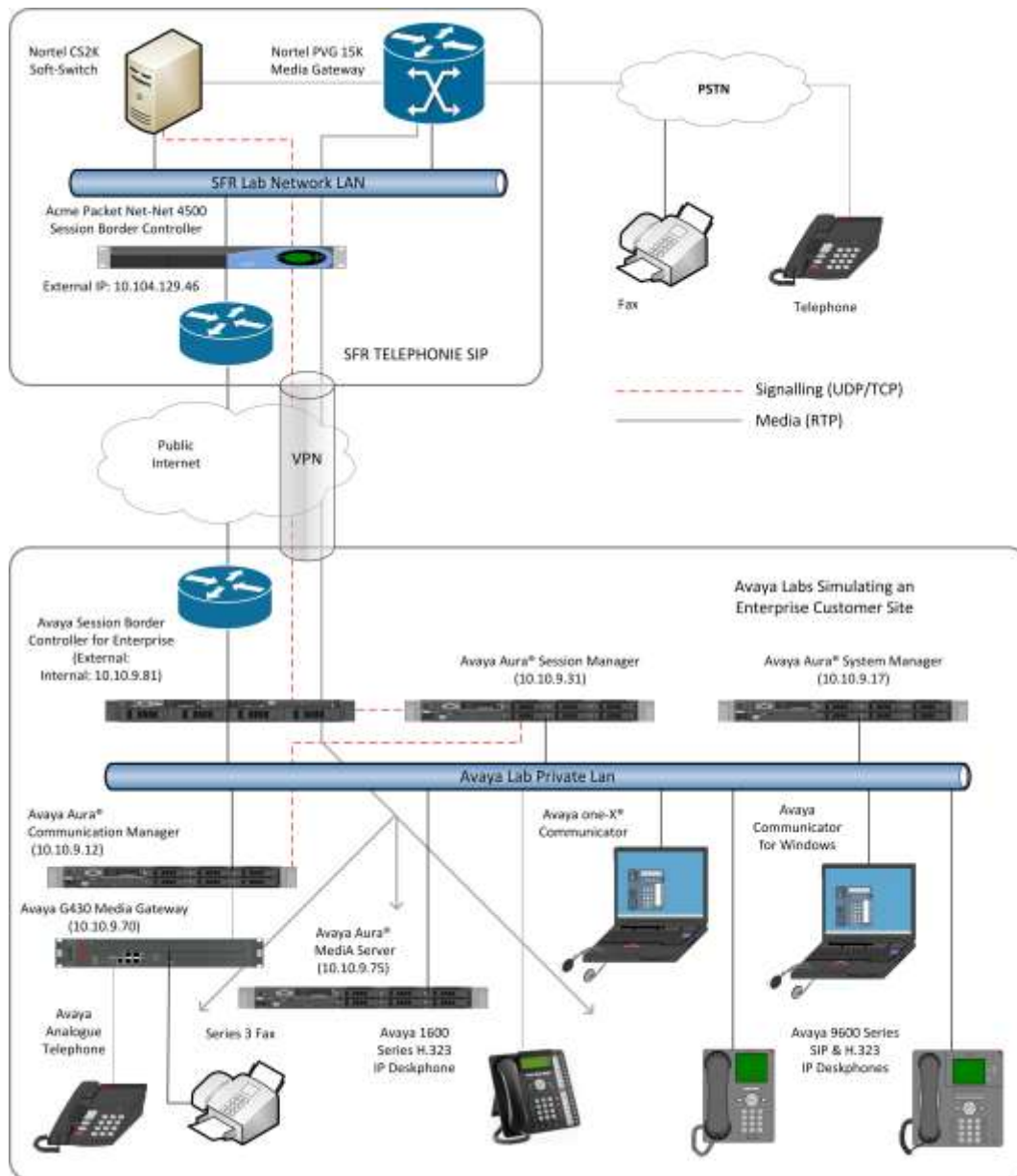


Figure 1: Test Setup SFR TELEPHONIE SIP to Avaya Enterprise

4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|--|--|
| Avaya | |
| Avaya Aura® Session Manager | 7.0.0.2.700201 |
| Avaya Aura® System Manager | 7.0.0.2.4416 |
| Avaya Aura® Communication Manager | 7.0-441 0-22856 |
| Avaya Session Border Controller for Enterprise | 7.0.1-03-8739 |
| Avaya G430 Media Gateway -MM711AP Analog MM | 37.21.0 HW30 FW100 |
| Avaya 9600 series Handsets SIP 96x0 SIP 9608 H.323 96x0 H.323 9608 H.323 1616 | 2_6_15_0 7.0.0 R39 3.2.6A 6.6.1.15 V474 1.380B |
| Avaya one-X® Communicator | 6.2.10.03-FP10 |
| Avaya Communicator for Windows | 2.1.3.80 |
| Avaya 2400 Series Digital Handsets | N/A |
| Analogue Handset | N/A |
| SFR | |
| SBC Net-Net 4500 | SCZ7.2.0 MR-5 P4 |
| SSW CS2K | CVM16 |
| Voice GW: PVG 15K | PCR 8.2 |

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP signalling associated with the TELEPHONIE SIP service. For incoming calls, Session Manager receives SIP messages from the Avaya SBCE and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signalling is routed to Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE at the enterprise site that then sends the SIP messages to the SFR network. 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. The general installation of the Servers and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the TELEPHONIE SIP service and any other SIP trunks used.

| 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 | 3 |
| Maximum Administered Remote Office Trunks: | | 4000 | 0 |
| Maximum Concurrently Registered Remote Office Stations: | | 2400 | 0 |
| Maximum Concurrently Registered IP eCons: | | 68 | 0 |
| Max Concur Registered Unauthenticated H.323 Stations: | | 100 | 0 |
| Maximum Video Capable Stations: | | 2400 | 0 |
| Maximum Video Capable IP Softphones: | | 2400 | 0 |
| Maximum Administered SIP Trunks: | | 4000 | 20 |
| Maximum Administered Ad-hoc Video Conferencing Ports: | | 4000 | 0 |
| Maximum Number of DS1 Boards with Echo Cancellation: | | 80 | 0 |

On **Page 5**, verify that **IP Trunks** field is set to **y**.

| | | |
|--|---|----------------------|
| display system-parameters customer-options | | Page 5 of 12 |
| OPTIONAL FEATURES | | |
| Emergency Access to Attendant? y | | IP Stations? y |
| Enable 'dadmin' Login? y | | |
| Enhanced Conferencing? y | | ISDN Feature Plus? n |
| Enhanced EC500? y | ISDN/SIP Network Call Redirection? y | |
| Enterprise Survivable Server? n | | ISDN-BRI Trunks? y |
| Enterprise Wide Licensing? n | | ISDN-PRI? y |
| ESS Administration? y | Local Survivable Processor? n | |
| Extended Cvg/Fwd Admin? y | Malicious Call Trace? y | |
| External Device Alarm Admin? y | Media Encryption Over IP? n | |
| Five Port Networks Max Per MCC? n | Mode Code for Centralized Voice Mail? n | |
| Flexible Billing? n | | |
| Forced Entry of Account Codes? y | Multifrequency Signaling? y | |
| Global Call Classification? y | Multimedia Call Handling (Basic)? y | |
| Hospitality (Basic)? y | Multimedia Call Handling (Enhanced)? y | |
| Hospitality (G3V3 Enhancements)? y | Multimedia IP SIP Trunking? y | |
| IP Trunks? y | | |
| IP Attendant Consoles? y | | |

Note: ISDN-PRI must also be enabled for IP Trunks.

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signalling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Session Manager. In this case, **Session_Manager** and **10.10.9.31** are the **Name** and **IP Address** for Session Manager SIP interface. Also note the **procr** IP address as this is the processor interface that Communication Manager will use as the SIP signalling interface to Session Manager.

| | | |
|------------------------|-------------------|---------------|
| display node-names ip | | IP NODE NAMES |
| Name | IP Address | |
| AMS | 10.10.9.75 | |
| Session_Manager | 10.10.9.31 | |
| default | 0.0.0.0 | |
| procr | 10.10.9.12 | |
| procr6 | :: | |

5.3. Administer IP Network Region

Use the **change ip-network-region n** command where **n** is the chosen value of the configuration for the SIP Trunk. Set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When direct media is used on a PSTN call, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **2** is used.
- The rest of the fields can be left at default values.

```
change ip-network-region 2                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 2
Location:      Authoritative Domain: avaya.com
Name: Trunk    Stub Network Region: n
MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes
Codec Set: 2   Inter-region IP-IP Direct Audio: yes
                IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5    AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS          RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

Note: In the test configuration, ip-network-region 1 was used within the enterprise and ip-network-region 2 was used for the SIP Trunk. In the configuration of the G430 and Avaya Media Server (not shown) ip-network-region 1 was used in such a way that either one could be selected at call set-up.

5.4. Administer IP Codec Set

Open the IP Codec Set form for the codec set specified in the IP Network Region form in **Section 5.3** by typing **change ip-codec set n** where **n** is the chosen value of the configuration for the SIP Trunk. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codecs supported by SFR were configured, namely **G.729A** and **G.711A**.

| | | | | |
|-----------------------|-------------|---------|-----------|-------------|
| change ip-codec-set 2 | | | | Page 1 of 2 |
| IP CODEC SET | | | | |
| Codec Set: 2 | | | | |
| Audio | Silence | Frames | Packet | |
| Codec | Suppression | Per Pkt | Size (ms) | |
| 1: G.729A | n | 2 | 20 | |
| 2: G.711A | n | 2 | 20 | |
| 3: | | | | |
| 4: | | | | |

TELEPHONIE SIP supports T.38 for transmission of fax. Navigate to **Page 2** and define T.38 fax as follows:

- Set the **FAX - Mode** to **t.38-standard**
- Leave **ECM** at default value of **y**. Note that during testing, this was set to **n** to avoid unnecessary retransmission.

| | | | | |
|-------------------------------|----------------------|------------|----------|------------------|
| change ip-codec-set 2 | | | | Page 2 of 2 |
| IP CODEC SET | | | | |
| Allow Direct-IP Multimedia? n | | | | |
| | Mode | Redundancy | ECM: | Packet Size (ms) |
| FAX | t.38-standard | 0 | n | |
| Modem | off | 0 | | |
| TDD/TTY | US | 3 | | |
| H.323 Clear-channel | n | 0 | | |
| SIP 64K Data | n | 0 | | 20 |

Note: Redundancy can be used to send multiple copies of T.38 packets which can help the successful transmission of fax over networks where packets are being dropped. This was not experienced in the test environment and **Redundancy** was left at the default value of **0**.

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the TELEPHONIE SIP service. During test, this was configured to use TCP and port 5062 though it's recommended to use TLS and port 5061 in the live environment to enhance security.

Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tcp**.
- Set **Peer Detection Enabled** to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Far-end Node Name** to Session Manager interface (node name **Session_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TCP is **5060**, though **5062** was used in test to separate the SIP Trunk from the SIP endpoints on Session Manager (See **Section 6.5**).
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3** (logically establishes the far-end for calls using this signalling group as region **2**).
- Leave **Far-end Domain** blank (allows Communication Manager to accept calls from any SIP domain on the associated trunk).
- Set **Direct IP-IP Audio Connections** and **H.323 Station Outgoing Direct Media** to **y**
- Set **Initial IP-IP Direct Media** to **n** to facilitate the use of Early Media to avoid ringback issues.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).

The default values for the other fields may be used.

| change signaling-group 2 | | Page 1 of 2 |
|---|------------------------------|------------------------------------|
| SIGNALING GROUP | | |
| Group Number: 2 | Group Type: sip | |
| IMS Enabled? n | Transport Method: tcp | |
| Q-SIP? n | | |
| IP Video? n | Enforce SIPS URI for SRTP? y | |
| 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 | | Far-end Node Name: Session_Manager |
| Near-end Listen Port: 5062 | | Far-end Listen Port: 5062 |
| Far-end Network Region: 2 | | |
| Far-end Domain: | | |
| Incoming Dialog Loopbacks: eliminate | | Bypass If IP Threshold Exceeded? n |
| DTMF over IP: rtp-payload | | RFC 3389 Comfort Noise? n |
| Session Establishment Timer(min): 3 | | Direct IP-IP Audio Connections? y |
| Enable Layer 3 Test? y | | IP Audio Hairpinning? n |
| H.323 Station Outgoing Direct Media? y | | Initial IP-IP Direct Media? n |
| | | Alternate Route Timer(sec): 6 |

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group n** command, where **n** is an available trunk group for the SIP Trunk. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **public-ntwrk** if the Diversion header is to be supported.
- Specify the signalling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**.
- Specify the **Number of Members** supported by this SIP trunk group.

| | | | |
|----------------------------|---------------------|--------------------------------|----------|
| add trunk-group 2 | | Page 1 of 21 | |
| TRUNK GROUP | | | |
| Group Number: 2 | Group Type: sip | CDR Reports: y | |
| Group Name: SIP_Trunk | COR: 1 | TN: 1 | TAC: 102 |
| 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: 2 | |
| | | Number of Members: 10 | |

On **Page 2** of the trunk-group form, the Preferred **Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with SFR to prevent unnecessary SIP messages during call setup. During testing, a value of **900** was used that sets Min-SE to 1800 in the SIP signalling.

| | | | |
|---------------------------------------|------------------------|--|--|
| add trunk-group 2 | | 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 | | | |

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in formats other than E.164 with leading “+”. During testing, CLIs were sent as national number with leading “0”. This format was successfully verified in the network.

| | |
|----------------------------------|---------------------------------|
| add trunk-group 2 | Page 3 of 21 |
| TRUNK FEATURES | |
| ACA Assignment? n | Measured: none |
| | Maintenance Tests? y |
| | |
| Numbering Format: private | |
| | UUI Treatment: service-provider |
| | Replace Restricted Numbers? n |
| | Replace Unavailable Numbers? n |

On **Page 4** of this form:

- Set **Network Call Redirection** to **n** as this function using either SIP 302 Moved Temporarily or REFER messages is not supported by SFR.
- Set **Support Request History** to **n** as this header is not supported by SFR.
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by SFR (this Payload Type is not applied to calls from SIP end-points).
- Set **Identity for Calling Party Display** to **From** to ensure that where CLI for incoming calls is withheld, it is not displayed on Communication Manager extension.
- Set **Convert 180 to 183 for Early Media** to **y** to ensure that Early Media is established correctly to the network to allow a backwards transmission path for ringback.

| | |
|---|--------------|
| add trunk-group 2 | 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? n | |
| Support Request History? n | |
| Telephone Event Payload Type: 101 | |
| | |
| Convert 180 to 183 for Early Media? y | |
| Always Use re-INVITE for Display Updates? n | |
| Identity for Calling Party Display: From | |
| Block Sending Calling Party Location in INVITE? n | |
| Accept Redirect to Blank User Destination? n | |
| Enable Q-SIP? n | |

5.7. Administer Calling Party Number Information

Use the **change private-numbering** command to configure Communication Manager to send the calling party number in the format required. During testing, calling party numbers were sent as national numbers prefixed with “0”. These calling party numbers are sent in the SIP From, Contact and PAI headers. The numbers are displayed on display-equipped PSTN telephones with any reformatting performed in the network.

| change private-numbering 0 | | | | | Page 1 of 2 |
|----------------------------|----------|------------|----------------|-----------|-----------------------|
| NUMBERING - PRIVATE FORMAT | | | | | |
| Ext Len | Ext Code | Trk Grp(s) | Private Prefix | Total Len | |
| 4 | 2 | 1 | | 4 | Total Administered: 9 |
| 4 | 2000 | 2 | 04274nnnn0 | 10 | Maximum Entries: 540 |
| 4 | 2290 | 2 | 04274nnnn8 | 10 | |
| 4 | 2291 | 2 | 04274nnnn2 | 10 | |
| 4 | 2316 | 2 | 04274nnnn3 | 10 | |
| 4 | 2391 | 2 | 04274nnnn1 | 10 | |
| 4 | 2396 | 2 | 04274nnnn9 | 10 | |
| 4 | 2400 | 2 | 04274nnnn4 | 10 | |
| 4 | 2401 | 2 | 04274nnnn5 | 10 | |

Note: During testing the extension numbers were reformatted to national numbers for Trunk Group 2 only. The numbers were analysed for Trunk Group 1 but not reformatted.

When the private numbering table is used, the extension numbers should also be analysed in the public numbering table using command **change public-numbering n**. This is useful for messaging where the Trunk Group details are not accessed directly, for example the Contact header in responses.

5.8. Administer Route Selection for Outbound Calls

In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to the SFR network. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit as the **Auto Route Selection (ARS) - Access Code 1**.

| change feature-access-codes | | Page 1 of 10 |
|---|--|----------------|
| FEATURE ACCESS CODE (FAC) | | |
| Abbreviated Dialing List1 Access Code: | | |
| Abbreviated Dialing List2 Access Code: | | |
| Abbreviated Dialing List3 Access Code: | | |
| Abbreviated Dial - Prgm Group List Access Code: | | |
| Announcement Access Code: *69 | | |
| Answer Back Access Code: | | |
| Attendant Access Code: | | |
| Auto Alternate Routing (AAR) Access Code: 8 | | |
| Auto Route Selection (ARS) - Access Code 1: 9 | | Access Code 2: |

Use the **change ars analysis** command to configure the routing of dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning 0. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 2**.

| | | | | | | | |
|--------------------------|---------------|-----------|-----------|---------------|-----------|----------|-----------------|
| change ars analysis 0 | | | | | | | Page 1 of 2 |
| ARS DIGIT ANALYSIS TABLE | | | | | | | |
| Location: all | | | | | | | Percent Full: 0 |
| | Dialed String | Total Min | Total Max | Route Pattern | Call Type | Node Num | ANI Req'd |
| | 0 | 8 | 12 | 2 | pubu | | n |
| | 00 | 13 | 15 | 2 | pubu | | n |
| | 0035391 | 13 | 13 | 2 | pubu | | n |
| | 1 | 3 | 3 | 2 | pubu | | n |
| | 118 | 5 | 6 | 2 | pubu | | n |
| | 3 | 4 | 4 | 2 | pubu | | n |
| | 7000 | 4 | 4 | 1 | pubu | | n |

Use the **change route-pattern n** command, where **n** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **2** is used to route calls to trunk group **2**. **Numbering Format** is applied to CLI and is used to set TDM signalling parameters such as type of number and numbering plan indicator. This doesn't have the same significance in SIP calls and during testing it was set to **lev0-pvt** to ensure that calling party number was not prefixed with a leading "+".

| | | | | | | | | | | | | | | | | | |
|---------------------------------------|--|--|--|--|--|--|--|--|--|--|--|--|--|-----------------------------------|--|--------------------------|--|
| change route-pattern 2 | | | | | | | | | | | | | | Page 1 of 3 | | | |
| Pattern Number: 2 | | | | | | | | | | | | | | Pattern Name: SIP_Endpoints | | | |
| 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 | | | |
| Dgts | | | | | | | | | | | | | | Intw | | | |
| 1: 2 0 | | | | | | | | | | | | | | n user | | | |
| 2: | | | | | | | | | | | | | | n user | | | |
| 3: | | | | | | | | | | | | | | n user | | | |
| 4: | | | | | | | | | | | | | | n user | | | |
| 5: | | | | | | | | | | | | | | n user | | | |
| 6: | | | | | | | | | | | | | | n user | | | |
| BCC VALUE TSC CA-TSC | | | | | | | | | | | | | | ITC BCIE Service/Feature PARM Sub | | Numbering LAR | |
| 0 1 2 M 4 W Request | | | | | | | | | | | | | | Dgts | | Format | |
| 1: y y y y y n n | | | | | | | | | | | | | | rest | | lev0-pvt none | |
| 2: y y y y y n n | | | | | | | | | | | | | | rest | | none | |
| 3: y y y y y n n | | | | | | | | | | | | | | rest | | none | |
| 4: y y y y y n n | | | | | | | | | | | | | | rest | | none | |
| 5: y y y y y n n | | | | | | | | | | | | | | rest | | none | |
| 6: v v v v v n n | | | | | | | | | | | | | | rest | | none | |

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to Communication Manager extensions. The incoming digits sent in the INVITE message from SFR can be manipulated as necessary to route calls to the desired extension. Use the **change inc-call-handling-trmt trunk-group x** command where **x** is the Trunk Group defined in **Section 5.6**. In the example shown, 10 digits are received. All 10 digits are deleted and the extension number is inserted. Note that some of the DDI digits have been obscured.

| change inc-call-handling-trmt trunk-group 2 | | | | | Page 1 of 3 | |
|---|---------------|------------------|-----|--------|-------------|--|
| INCOMING CALL HANDLING TREATMENT | | | | | | |
| Service/ Feature | Number Len | Number Digits | Del | Insert | | |
| public-ntwrk | 10 | 04274nnnn0 | 10 | 2000 | | |
| public-ntwrk | 10 | 04274nnnn1 | 10 | 2391 | | |
| public-ntwrk | 10 | 04274nnnn2 | 10 | 2291 | | |
| public-ntwrk | 10 | 04274nnnn3 | 10 | 2316 | | |
| public-ntwrk | 10 | 04274nnnn4 | 10 | 2400 | | |
| public-ntwrk | 10 | 04274nnnn5 | 10 | 2401 | | |
| public-ntwrk | 10 | 04274nnnn7 | 10 | 6002 | | |
| public-ntwrk | 10 | 04274nnnn8 | 10 | 2290 | | |
| public-ntwrk | 10 | 04274nnnn9 | 10 | 2396 | | |

5.10. EC500 Configuration

When EC500 is enabled on a Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 2291. Use the command **change off-pbx-telephone station-mapping x** where **x** is Communication Manager station.

- The **Station Extension** field will automatically populate with station extension.
- For **Application** enter **EC500**.
- Enter a **Dial Prefix** if required by the routing configuration, none was required during testing.
- For the **Phone Number** enter the phone that will also be called (e.g., **003538941nnnn7**).
- Set the **Trunk Selection** to **ars** so that the ARS table will be used for routing.
- Set the **Config Set** to **1**.

| change off-pbx-telephone station-mapping 2291 | | | | | | | Page 1 of 3 | |
|---|--------------|----------------|----|-----------------------|--------------------|---------------|--------------|--|
| STATIONS WITH OFF-PBX TELEPHONE INTEGRATION | | | | | | | | |
| Station Extension | Application | Dial Prefix | CC | Phone Number | Trunk Selection | Config Set | Dual Mode | |
| 2291 | OPS | - | | 2291 | aar | 1 | | |
| 2291 | EC500 | - | | 003538941nnnn7 | ars | 1 | | |

Note: The phone number shown is for a mobile phone in the Avaya Lab. To use facilities such as Feature Name Extension (FNE) for calls coming in from EC500 mobile phones, the calling party number received in Communication Manager must exactly match the number specified in the above table. The additional line in the previous screenshot with **Application** of **OPS** is standard

on SIP endpoints where the phone is registered to the Session Manager and is essentially “Off PBX”.

Save Communication Manager configuration by entering **save translation**.

5.11. Emergency Number handling

Emergency Number Handling was **not** tested during compliance testing and the following example is provided as a rough guide only. The Emergency Number is defined as a **dialed string** and the **Call Type** set to **emer** (emergency)

| | | | | | | | |
|--------------------------|---------------|---------------|---------------|-----------|----------|-----------|-----------------|
| change ars analysis 0 | | | | | | | Page 1 of 2 |
| ARS DIGIT ANALYSIS TABLE | | | | | | | |
| Location: all | | | | | | | Percent Full: 0 |
| | Dialed String | Total Min Max | Route Pattern | Call Type | Node Num | ANI Req'd | |
| 0 | | 9 12 | 3 | pubu | | n | |
| 00 | | 9 14 | 3 | pubu | | n | |
| 01 | | 10 10 | 3 | pubu | | n | |
| 118 | | 3 6 | 3 | pubu | | n | |
| 1xx | | 3 3 | 3 | pubu | | n | |
| 15 | | 2 2 | 3 | emer | | n | |
| 17 | | 2 2 | 3 | emer | | n | |
| 18 | | 2 2 | 3 | emer | | n | |
| 99 | | 12 12 | 99 | pubu | | n | |

On each site, define the number to be sent as the calling party when an emergency number is dialled. Example:

Site 1 Extension 2391 and 2291 are defined with DDI numbers 0427418051 and 0427418052 To define 2291 as the emergency location number for 2391, type **change station 2391** and on page 2, change the **Emergency Location Ext** to **2291**

| | | |
|--------------------------------|--|-------------|
| change station 2391 | | Page 2 of 5 |
| STATION | | |
| FEATURE OPTIONS | | |
| LWC Reception: spe | Auto Select Any Idle Appearance? n | |
| LWC Activation? y | Coverage Msg Retrieval? y | |
| LWC Log External Calls? n | Auto Answer: none | |
| CDR Privacy? n | Data Restriction? n | |
| Redirect Notification? y | Idle Appearance Preference? n | |
| Per Button Ring Control? n | Bridged Idle Line Preference? n | |
| Bridged Call Alerting? n | Restrict Last Appearance? y | |
| Active Station Ringing: single | | |
| | EMU Login Allowed? n | |
| H.320 Conversion? n | Per Station CPN - Send Calling Number? | |
| Service Link Mode: as-needed | EC500 State: enabled | |
| Multimedia Mode: enhanced | Audible Message Waiting? n | |
| MWI Served User Type: | Display Client Redirection? n | |
| AUDIX Name: | Select Last Used Appearance? n | |
| | Coverage After Forwarding? s | |
| | Multimedia Early Answer? n | |
| | Direct IP-IP Audio Connections? y | |
| Emergency Location Ext: 2291 | Always Use? n IP Audio Hairpinning? n | |

Configure the IP network mapping using the **change ip-network-map** command and define an **Emergency Location Extension**. This can be done using a static IP address for a single phone or one or more subnets for an IP network region. The example shows Emergency Location Extensions for two sites, both in the IP network region defined in **Section 5.3**

change ip-network-map

Page 1 of 63

IP ADDRESS MAPPING

| IP Address | Subnet Bits | Network Region | VLAN | Emergency Location Ext |
|-------------------|-------------|----------------|------|------------------------|
| FROM: 10.10.5.1 | /24 | 1 | n | 2391 |
| TO: 10.10.5.199 | | | | |
| FROM: 10.10.5.200 | /24 | 1 | n | 2291 |
| TO: 10.10.5.254 | | | | |
| FROM: | / | | n | |
| TO: | | | | |
| FROM: | / | | n | |

When an emergency call is made, Communication Manager identifies the mapping from the IP address of the phone. It then compares the Emergency Location Extension defined in the IP network map with the one defined for the station. If the two are the same, the CM sends the station extension. If the two are different, it sends the IP Address Mapping extension. Refer to Avaya Aura ® Communication Manager Feature Description and Implementation document for more detailed information.

The calling party number sent will be the Numéro de Désignation d'Installation (NDI) which is the main DDI number defined for Communication Manager, defined using the **change private-numbering** command as described in **Section 5.7**. The number for extension 2291 is 04274nnnn2 sent in the P-Asserted-Identity header.

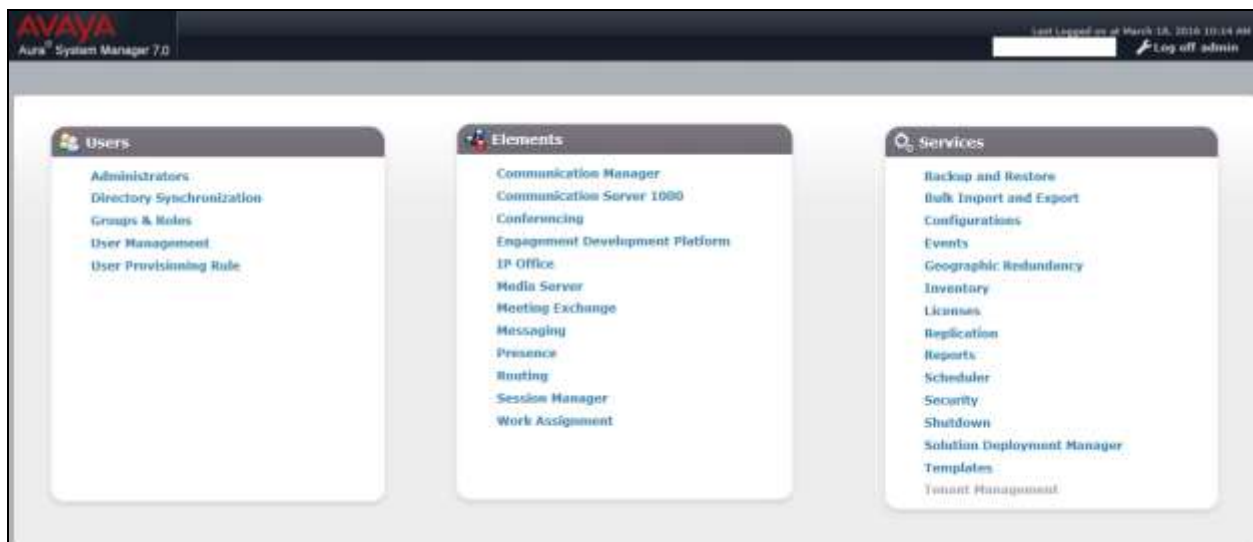
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured by opening a web browser to the System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a web browser and entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the **Home** tab will be presented with menu options shown below.



6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name of the enterprise site or a name agreed with SFR; this will be the same as specified in the Authoritative Domain specified in the IP Network Region on Communication Manager. Refer to **Section 5.3** for details. In test, **avaya.com** was used. Optionally, a description for the domain can be entered in the Notes field (not shown). Click **Commit** to save changes.



The screenshot shows the 'Domain Management' interface. On the left is a navigation menu with 'Routing' expanded, showing sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, and Dial Patterns. The main area has a breadcrumb 'Home / Elements / Routing / Domains' and a title 'Domain Management'. Below the title are buttons: 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. A table below shows '1 Item' with a refresh icon. The table has columns: Name, Type, and Notes. One row is visible with 'avaya.com' in the Name column and 'sip' in the Type column. At the bottom, there is a 'Select' dropdown with options 'All' and 'None'.

| Name | Type | Notes |
|-----------|------|-------|
| avaya.com | sip | |

Note: If the existing domain name used in the enterprise equipment does not match that used in the network, a Session Manager Adaptation can be used to change it (see **Section 6.4**).

6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu (not shown). Under **General**, in the **Name** field, enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the test enterprise.

The screenshot shows the 'Location Details' configuration page. At the top, there is a breadcrumb trail 'Home / Elements / Routing / Locations' and a 'Help ?' link. The page title is 'Location Details' with 'Commit' and 'Cancel' buttons. The 'General' section contains a 'Name' field with 'Galway' and a 'Notes' field. The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox, a 'Listed Directory Number' field, and an 'Associated CM SIP Entity' field. The 'Overall Managed Bandwidth' section includes 'Managed Bandwidth Units' (set to 'Kbit/sec'), 'Total Bandwidth', 'Multimedia Bandwidth', and an 'Audio Calls Can Take Multimedia Bandwidth' checkbox. The 'Per-Call Bandwidth Parameters' section has fields for 'Maximum Multimedia Bandwidth (Intra-Location)', 'Maximum Multimedia Bandwidth (Inter-Location)', '* Minimum Multimedia Bandwidth', and '* Default Audio Bandwidth'. The 'Alarm Threshold' section includes 'Overall Alarm Threshold', 'Multimedia Alarm Threshold', '* Latency before Overall Alarm Trigger', and '* Latency before Multimedia Alarm Trigger'. The 'Location Pattern' section at the bottom has 'Add' and 'Remove' buttons, a table with one row containing '*10.10.9.x' under 'IP Address Pattern', and a 'Select : All, None' option.

Home / Elements / Routing / Locations

Help ?

Location Details

Commit Cancel

General

* Name: Galway

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☐

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec

* Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

1 Item Filter: Enable

| IP Address Pattern | Notes |
|--------------------|-------|
| *10.10.9.x | |

Select : All, None

6.4. Administer Adaptations

Communication Manager and Session Manager make use of Avaya proprietary SIP headers to facilitate the full suite of Avaya functionality within the enterprise. These are not required on the SIP trunk however, and are not recognized by the SFR network. A Session Manager Adaptation is used to remove proprietary headers.

On the **Routing** tab select **Adaptations** from the left-hand menu. Click on **New** (not shown).

- In the **Adaptation Name** field, enter a descriptive title for the adaptation.
- In the **Module Name** drop down menu, select **DigitConversionAdapter**. This is used for simple digit conversion adaptations.
- In the **Module Parameter Type** drop down menu, select **Name-Value Parameter**.
- In the **Name** box, type **eRHdrs**
- In the **Value** box, type the list of headers to be deleted. During testing, the following list was used: "**P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, Recv-Info, P-Conference, Alert-Info, Reason**".

Home / Elements / Routing / Adaptations

Adaptation Details

Commit Cancel Help ?

General

* Adaptation Name: Header_Removal

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

| Name | Value |
|---------------------------------|--|
| <input type="checkbox"/> eRHdrs | "P-AV-Message-Id, P-Charging-Vector, Av-Global-Session-ID, P-Location, Endpoint-View, Recv-Info, P-Conference, Alert-Info, Reason" |

Select : All, None

Egress URI Parameters:

Notes:

Note: SFR provided the following list of headers to be removed to avoid the possibility of fragmentation: P-AV-Message-Id; P-Charging-Vector; AV-Global-Session-ID; P-rental; Accept-Language; Alert-Info; History-Info. Not all of these are included in the above adaptation for removal and can be added to the list if necessary. History-Info is removed using Communication Manager Trunk settings as described in **Section 5.6**.

Number analysis is used to apply the above Module Parameter rule. To apply the rule to all messages on the SIP Trunk, the called party number is analysed such that all called party numbers meet the criteria. Scroll down and in the section **Digit Conversion for Incoming Calls to SM**, click on **Add**. An additional row will appear (not shown). This allows information to be entered for the manipulation of numbers coming from the network.

The screenshot below shows analysis of called party numbers for incoming calls. The called party number is the DDI number associated with the Communication manager extensions.

- Under **Matching Pattern** enter the DDI number as received from the network.
- Under **Min** and **Max** enter the Minimum and Maximum digits of the incoming DDI number, in this case the DDI number length is fixed at **10**.
- Under **Delete Digits** enter 0 as the number is not to be modified.
- Leave the **Insert Digits** field blank as the number is not to be modified.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the called party number.

| Matching Pattern | Min | Max | Phone Context | Delete Digits | Insert Digits | Address to modify | Adaptation Data | Notes |
|------------------|-----|-----|---------------|---------------|---------------|-------------------|-----------------|-------|
| *0427nnnn5 | *10 | *10 | | *0 | | destination | | |

Note: In the above screenshot the DDI numbers are partially obscured.

Scroll down and in the section **Digit Conversion for Outgoing Calls from SM**, click on **Add**. An additional row will appear (not shown). This allows information to be entered for the manipulation of numbers going out to the network

The screenshot below shows analysis of called party numbers for outgoing calls. The called party number is the dialled public number.

- Under **Matching Pattern** enter the first dialled digit, in most cases this will be **0**.
- Under **Min** and **Max** enter the Minimum and Maximum digits of dialled number, in this case a minimum of **1** and a maximum of **36** was specified to cover all possible number lengths.
- Under **Delete Digits** enter 0 as the number is not to be modified.
- Leave the **Insert Digits** field blank as the number is not to be modified.
- Under **Address to Modify** choose **destination** from the drop down box to apply this rule to the called party number.
- Click **Commit** to save changes.

| Matching Pattern | Min | Max | Phone Context | Delete Digits | Insert Digits | Address to modify | Adaptation Data | Notes |
|------------------|-----|-----|---------------|---------------|---------------|-------------------|-----------------|-------|
| *0 | *1 | *36 | | *0 | | destination | | |

Note: It may be preferable to analyse the calling party number for the **Digit Conversion for Outgoing Calls from SM**. If so, set the parameters the same as for incoming calls, but set the **Address to Modify** to **origination**.

6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu, and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity. Under **General**:

- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **CM** for a Communication Manager SIP Entity and **SIP Trunk** for the Avaya SBCE SIP Entity.
- In the **Adaptation** field (not available for the Session Manager SIP Entity), select the appropriate Adaptation from the drop down menu.
- In the **Location** field select the appropriate location from the drop down menu.
- In the **Time Zone** field enter the time zone for the SIP Entity.

In this configuration there are four SIP Entities:

- Avaya Aura® Session Manager SIP Entity.
- Avaya Aura® Communication Manager SIP Entity for the SIP Endpoints
- Avaya Aura® Communication Manager SIP Entity for the SIP Trunk
- Avaya Session Border Controller for Enterprise (Avaya SBCE) SIP Entity for PSTN destinations.

6.5.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface.

The screenshot shows a web-based configuration interface for SIP Entities. The breadcrumb trail at the top is "Home / Elements / Routing / SIP Entities". The main heading is "SIP Entity Details" with "General" selected. There are "Commit" and "Cancel" buttons in the top right. The form fields are as follows:

- Name:** Session_Manager
- FQDN or IP Address:** 10.10.9.31
- Type:** Session Manager (dropdown menu)
- Notes:** (empty text area)
- Location:** Galway (dropdown menu)
- Outbound Proxy:** (empty dropdown menu)
- Time Zone:** Europe/Dublin (dropdown menu)
- Credential name:** (empty text area)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown menu)

The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop down menu select the domain added in **Section 6.2** as the default domain.

| Listen Ports | Protocol | Default Domain | Notes |
|-------------------------------|----------|----------------|-------|
| <input type="checkbox"/> 5060 | TCP | avaya.com | |
| <input type="checkbox"/> 5060 | UDP | avaya.com | |
| <input type="checkbox"/> 5061 | TLS | avaya.com | |
| <input type="checkbox"/> 5062 | TCP | avaya.com | |

6.5.2. Avaya Aura® Communication Manager SIP Entities

The following screen shows one of the SIP entities for Communication Manager which is configured as an Evolution Server. This SIP Entity is used for the SIP Trunk. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signalling. Set the **Location** to that defined in **Section 6.3**.

SIP Entity Details

General

* Name: CM Trunk

* FQDN or IP Address: 10.10.9.12

Type: CM

Notes:

Adaptation:

Location: Galway

Time Zone: Europe/Dublin

* SIP Timer B/F (in seconds): 4

Credential name:

Securable: ☐

Call Detail Recording: none

Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.

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

Supports Call Admission Control: ☐

Shared Bandwidth Manager: ☐

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Note: A second SIP Entity for Communication Manager is required for SIP Endpoints. In the test environment this is named “CM_SIP_Endpoints”.

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE used for PSTN destinations. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface used for PSTN fixed calls (see **Figure 1**). Set the **Adaptation** to that defined in **Section 6.4**, the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

Home / Elements / Routing / SIP Entities

SIP Entity Details

Commit Cancel

General

* Name: ASBCE

* FQDN or IP Address: 10.10.9.81

Type: SIP Trunk

Notes:

Adaptation: Header_Removal

Location: Galway

Time Zone: Europe/Dublin

* SIP Timer B/F (in seconds): 4

Credential name:

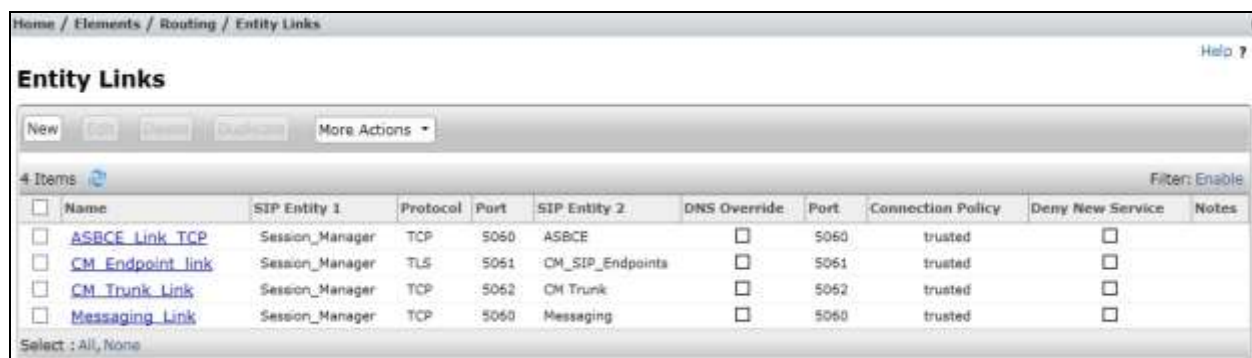
Securable: ☐

Call Detail Recording: egress

6.6. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select **Session Manager**.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.



| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | DNS Override | Port | Connection Policy | Deny New Service | Notes |
|--------------------------|----------------------------------|-----------------|----------|------|------------------|--------------------------|------|-------------------|--------------------------|-------|
| <input type="checkbox"/> | ASBCE_Link_TCP | Session_Manager | TCP | 5050 | ASBCE | <input type="checkbox"/> | 5050 | trusted | <input type="checkbox"/> | |
| <input type="checkbox"/> | CM_Endpoint_Link | Session_Manager | TLS | 5051 | CM_SIP_Endpoints | <input type="checkbox"/> | 5051 | trusted | <input type="checkbox"/> | |
| <input type="checkbox"/> | CM_Trunk_Link | Session_Manager | TCP | 5052 | CM_Trunk | <input type="checkbox"/> | 5052 | trusted | <input type="checkbox"/> | |
| <input type="checkbox"/> | Messaging_Link | Session_Manager | TCP | 5050 | Messaging | <input type="checkbox"/> | 5050 | trusted | <input type="checkbox"/> | |

Click **Commit** to save changes. The previous screen shows the Entity Links used in this configuration.

Note: There are two Entity Links for Communication Manager, one for the SIP Endpoints and the other for the SIP Trunk. These are differentiated by port number. The **Messaging_Link** Entity Link is used for the Avaya Aura ® Messaging system and is not described in this document.

6.7. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies
- Under **Time of Day**, click **Add**, and then select the time range

The following screen shows the routing policy for calls inbound from the SIP Trunk to Communication Manager.

Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

| Name | FQDN or IP Address | Type | Notes |
|----------|--------------------|------|-------|
| CM Trunk | 10.10.9.12 | CM | |

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

| Ranking | Name | Mon | Tue | Wed | Thu | Fri | Sat | Sun | Start Time | End Time | Notes |
|----------------------------|------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|------------|----------|-----------------|
| <input type="checkbox"/> 0 | 24/7 | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | 00:00 | 23:59 | Time Range 24/7 |

Select : All, None

The following screen shows the Routing Policy for the Avaya SBCE interface that will be routed to PSTN destinations via TELEPHONIE SIP.

Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

| Name | FQDN or IP Address | Type | Notes |
|-------|--------------------|-----------|-------|
| ASBCE | 10.10.9.81 | SIP Trunk | |

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

| Ranking | Name | Mon | Tue | Wed | Thu | Fri | Sat | Sun | Start Time | End Time | Notes |
|----------------------------|------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|------------|----------|-----------------|
| <input type="checkbox"/> 0 | 24/7 | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | 00:00 | 23:59 | Time Range 24/7 |

Select : All, None

6.8. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- In the **Pattern** field enter a dialled number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialled number.
- In the **Max** field enter the maximum length of the dialled number.
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**.

Under **Originating Locations and Routing Policies**:

- Click **Add**, in the resulting screen (not shown).
- Under **Originating Location**, select the location defined in **Section 6.3** or **ALL**.
- Under **Routing Policies** select one of the routing policies defined in **Section 6.7**.
- Click **Select** button to save.

The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls to PSTN destinations via TELEPHONE SIP.

Home / Elements / Routing / Dial Patterns

Dial Pattern Details Commit Cancel Help ?

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

| <input type="checkbox"/> | Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing Policy Notes |
|--------------------------|---------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> | -ALL- | PSTN_Outbound | 0 | | <input type="checkbox"/> | ASBCE | |

Select : All, None

The following screen shows the test dial pattern configured for Communication Manager.

Dial Pattern Details [Commit] [Cancel] Help ?

General

* Pattern: 0427nnnn5 x

* Min: 9

* Max: 10

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL- v

Notes:

Originating Locations and Routing Policies

[Add] [Remove]

1 Item Filter: Enable

| <input type="checkbox"/> | Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing Policy Notes |
|--------------------------|---------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> | -ALL- | | CM_Inbound | 0 | <input type="checkbox"/> | CM Trunk | |

Select : All, None

Note: The above configuration is used to analyse the DDI numbers assigned to the extensions on Communication Manager. Some of the digits of the pattern to be matched have been obscured.

6.9. Administer Application for Avaya Aura® Communication Manager

From the **Home** tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New** (not shown).

- In the **Name** field enter a name for the application.
- In the **SIP Entity** field select the SIP entity for Communication Manager.
- In the **CM System for SIP Entity** field select the SIP entity for Communication Manager and select **Commit** to save the configuration.

Application Editor [Commit] [Cancel]

Application

* Name: CM_App x

* SIP Entity: CM_SIP_Endpoints

* CM System for SIP Entity: CM1_Element [Refresh] [View/Add CM Systems](#)

Description:

Note: The Application described here and the Application Sequence described in the next section are likely to have been defined during installation. The configuration is shown here for reference. Note also that the Communication Manager SIP Entity selected is that set up specifically for SIP endpoints. In the test environment there is also a Communication Manager SIP Entity that is used specifically for the SIP Trunk and is not to be used in this case.

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New** (not shown).

- In the **Name** field enter a descriptive name.
- Under **Available Applications**, click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading. Select **Commit**.

Home / Elements / Session Manager / Application Configuration / Application Sequences Help ?

Application Sequence Editor

Commit Cancel

Application Sequence

*Name x

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

| <input type="checkbox"/> | Sequence Order (first to last) | Name | SIP Entity | Mandatory | Description |
|--------------------------|--------------------------------|------------------------|------------------|-------------------------------------|-------------|
| <input type="checkbox"/> | *** | CM_App | CM_SIP_Endpoints | <input checked="" type="checkbox"/> | |

Select : All, None

Available Applications

1 Item Filter: Enable

| | Name | SIP Entity | Description |
|---|------------------------|------------------|-------------|
| + | CM_App | CM_SIP_Endpoints | |

*Required Commit Cancel

6.11. Administer SIP Extensions

The SIP extensions are likely to have been defined during installation. The configuration shown in this section is for reference. SIP extensions are registered with Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g., 2291@avaya.com which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- In the **Password/Confirm Password** fields enter an alphanumeric password.
- Set the **Language Preference** and **Time Zone** as required.

The screenshot shows the 'New User Profile' form in the Avaya User Management interface. The form is divided into tabs: Identity, Communication Profile, Membership, and Contacts. The Identity tab is active, showing fields for User Provisioning Rule, Last Name, First Name, Login Name, Authentication Type, Password, Confirm Password, Localized Display Name, Endpoint Display Name, Title, Language Preference, Time Zone, Employee ID, Department, and Company. The form is pre-filled with example data: Last Name: SIP, First Name: 9608, Login Name: 2291@avaya.com, Authentication Type: Basic, Password: 9608, Confirm Password: 9608, Language Preference: English (United Kingdom), Time Zone: (0:0)GMT : Dublin, Edinburgh, L.

In the **Communication Profile** tab, enter a numeric **Communication Profile Password** and confirm it.

The screenshot shows the 'Communication Profile' tab in a configuration window. At the top, there are tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' section has two password fields: 'Communication Profile Password' and 'Confirm Password', both masked with dots. Below these is a 'Name' section with a 'New' button, a 'Delete' button, a 'Done' button, and a 'Cancel' button. The 'Name' field is set to 'Primary' and is marked as the 'Default' with a checked checkbox. Below the 'Name' section is a 'Communication Address' section with a 'New' button, an 'Edit' button, and a 'Delete' button. Below these buttons is a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'.

Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.

The screenshot shows the 'Communication Address' section expanded. It has buttons for 'New', 'Edit', and 'Delete'. Below these is a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'. Below the table, the 'Type' field is set to 'Avaya SIP'. The 'Fully Qualified Address' field is set to '2291' and the 'Domain' field is set to 'avaya.com'. At the bottom right, there are 'Add' and 'Cancel' buttons.

Expand the **Session Manager Profile** section.

- Make sure the **Session Manager Profile** check box is checked.
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field.
- Select the appropriate application sequence from the drop-down menu in the **Origination Sequence** field configured in **Section 6.10**.
- Select the appropriate application sequence from the drop-down menu in the **Termination Sequence** field configured in **Section 6.10**.
- Select the appropriate location from the drop-down menu in the **Home Location** field.

☒ **Session Manager Profile** ▼

SIP Registration

* Primary Session Manager

Session_Manager

| Primary | Secondary | Maximum |
|----------------------|-----------|---------|
| 4 | 0 | 4 |
| <div>< ></div> | | |

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

1 ▼

Block New Registration When Maximum Registrations Active?

☐

Application Sequences

Origination Sequence

CM_App_Seq ▼

Termination Sequence

CM_App_Seq ▼

Call Routing Settings

* Home Location

Galway ▼

Conference Factory Set

(None) ▼

Call History Settings

Enable Centralized Call History?

☐

Expand the **Endpoint Profile** section.

- Select Communication Manager Element from the **System** drop-down menu.
- Select **Endpoint** from the drop-down menu for **Profile Type**.
- Enter the extension in the **Extension** field.
- Select the desired template from the **Template** drop-down menu.
- In the **Port** field **IP** is automatically inserted.
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box.
- Select **Commit** (Not Shown) to save changes and the System Manager will add Communication Manager user configuration automatically.

☒ **CM Endpoint Profile** ▼

* System

CM1_Element ▼

* Profile Type

Endpoint ▼

Use Existing Endpoints ☐

* Extension

Q 2291

Endpoint Editor

* Template

9608SIP_DEFAULT_CM_7_0 ▼

Set Type 9608SIP

Security Code

Port IP

Voice Mail Number

Preferred Handle (None) ▼

Calculate Route Pattern ☐

Sip Trunk aar

Enhanced Callr-Info display for 1-line phones ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User ☒

Override Endpoint Name and Localized Name ☒

Allow H.323 and SIP Endpoint Dual Registration ☐

7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary.

7.1. Access Avaya Session Border Controller for Enterprise

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.



The login screen features the Avaya logo in red on the left. To the right, under the heading "Log In", is a "Username:" label followed by a text input field and a "Continue" button. Below the input field, there is a block of legal disclaimer text. At the bottom, it states "© 2011 - 2015 Avaya Inc. All rights reserved."

AVAYA

Session Border Controller for Enterprise

Log In

Username:


This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

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Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



The dashboard has a top navigation bar with links: Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the Avaya logo. A left-hand menu lists various configuration options. The main content area is divided into several sections: "Information" with system details, "Installed Devices" with a table of devices, and "Alarms (past 24 hours)" and "Incidents (past 24 hours)" both showing "None found".

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

Dashboard

Information

| | | |
|------------------------------|------------------------------|---------|
| System Time | 03:47:53 AM GMT | Refresh |
| Version | 7.0.1-03-8739 | |
| Build Date | Fri Jan 15 22:53:12 EST 2016 | |
| License State | OK | |
| Aggregate Licensing Overages | 0 | |
| Peak Licensing Overage Count | 0 | |
| Last Logged in at | 03/22/2016 03:29:15 GMT | |
| Failed Login Attempts | 0 | |

Alarms (past 24 hours)
None found.

Installed Devices

| |
|----------|
| EMS |
| GSSCP_V8 |

Incidents (past 24 hours)
None found.

7.2. Define Network Management

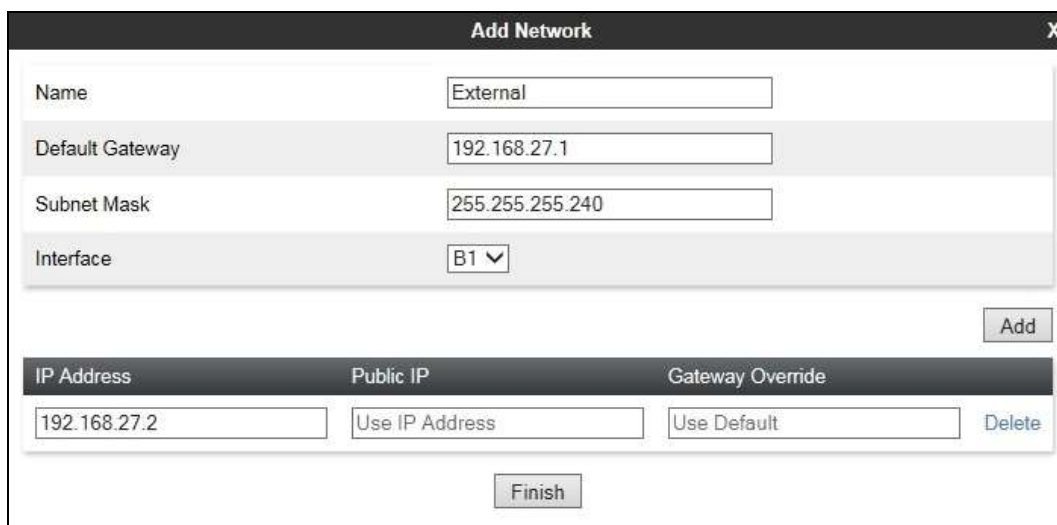
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one physical interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**.



Enter details for the external interfaces in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the test environment, this was **B1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the external IP address of the Avaya SBCE on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.



| Name | Default Gateway | Subnet Mask | Interface |
|----------|-----------------|-----------------|-----------|
| External | 192.168.27.1 | 255.255.255.240 | B1 |

Add

| IP Address | Public IP | Gateway Override |
|--------------|----------------|------------------|
| 192.168.27.2 | Use IP Address | Use Default |

Delete

Finish

Click on **Add** to define the internal interface. Enter details in the dialogue box (not shown):

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal IP address for the Avaya SBCE in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

The following screenshot shows the completed Network Management configuration:



Network Management: GSSCP_V9

Devices: GSSCP_V9

Interfaces Networks

Add

| Name | Gateway | Subnet Mask | Interface | IP Address | Edit | Delete |
|----------|--------------|-----------------|-----------|--------------|------|--------|
| Internal | 10.10.9.1 | 255.255.255.0 | A1 | 10.10.9.81 | Edit | Delete |
| External | 192.168.27.1 | 255.255.255.240 | B1 | 192.168.27.2 | Edit | Delete |

Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle the state. Change the state to **Enabled** where required.



Network Management: GSSCP_V9

Devices: GSSCP_V9

Interfaces Networks

Add VLAN

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

Note: to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

- Click on **System Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear that will indicate when the restart is complete.

7.3. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces. Testing was carried out with TCP used for transport of signalling between Session Manager and the Avaya SBCE, and UDP for transport of signalling between the Avaya SBCE and the TELEPHONIE SIP trunk. Two signalling and two media interfaces were required on both the internal and external sides of the Avaya SBCE to handle on-net and off-net traffic. This document shows the configuration for TCP and UDP, if additional security is required, it's recommended to use TLS and port 5061.

7.3.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Signaling Interface** (not shown) in the main menu on the left hand side. Details of transport protocol and ports for the external and internal SIP signalling are entered here.

- Select **Add** and enter details of the external signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external signalling interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was IP address **192.168.27.2** for the Avaya SBCE interface on the SIP Trunk.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the TELEPHONIE SIP service.

The screenshot displays the 'Session Border Controller' configuration interface. On the left is a sidebar menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, PPM Services, Domain Policies, TLS Management, Device Specific Settings (selected), Network Management, Media Interface, and Signaling Interface. The 'Signaling Interface' section is active, showing a 'GSSCP_VII' device. A 'Add Signaling Interface' dialog box is open on the right. It contains the following fields: 'Name' (set to 'External'), 'IP Address' (set to 'External (B1, VLAN 0)' with a dropdown showing '192.168.27.2'), 'TCP Port' (set to 'Leave blank to disable'), 'UDP Port' (set to '5060'), 'TLS Port' (set to 'Leave blank to disable'), 'TLS Profile' (set to 'None'), 'Enable Shared Control' (checkbox), and 'Shared Control Port' (text field). A 'Finish' button is at the bottom right of the dialog.

The internal signalling interface is defined in the same way; the dialogue box is not shown:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal signalling interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.
- Select **TCP** port number, **5060** is used for Session Manager.

The following screenshot shows details of the signalling interfaces:



Signaling Interface: GSSCP_V9

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

| Name | Signaling IP Network | TCP Port | UDP Port | TLS Port | TLS Profile | |
|----------|---------------------------------------|-------------|-------------|----------|-------------|-------------|
| Internal | 10.10.9.81 Internal (A1, VLAN 0) | 5060 | — | — | None | Edit Delete |
| External | 192.168.27.2 External (B1, VLAN 0) | — | 5060 | — | None | Edit Delete |

Note: In the test environment, the internal IP address was **10.10.9.81**.

7.3.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings** → **Media Interface** in the main menu on the left hand side. Details of the RTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

- Select **Add** and enter details of the external media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was IP address **192.168.27.2**.
- Define the RTP **Port Range** for the media path with the TELEPHONIE SIP service, during testing this was left at the default values.



Media Interface: GSSCP_V9

Add Media Interface

Name: External

IP Address: External (B1, VLAN 0)

Port Range: 35000 - 40000

Finish

The internal media interfaces are defined in the same way; the dialogue box is not shown:

- Select **Add** and enter details of the internal media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.

The following screenshot shows details of the media interfaces:



Note: In the test environment, the internal IP address was **10.10.9.81**.

7.4. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, the TELEPHONIE SIP trunk is connected as the Trunk Server and the Session Manager is connected as the Call Server. Configuration of interworking includes Hold support, T.38 fax support and SIP extensions.

To define server interworking on the Avaya SBCE, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for the TELEPHONIE SIP service, click on **Add** (not shown). A pop-up menu is generated. In the **Name** field enter a descriptive name for the SFR network and click **Next**.



Check the **T.38 Support** box and click on **Next**.

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: None

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

Click on **Next** and **Next** again to go through the next two dialogue boxes. During testing, these were left at default values.

Interworking Profile

All fields are optional

SIP Timers

Min-SE: [] seconds, [90 - 86400]

Init Timer: [] milliseconds, [50 - 1000]

Max Timer: [] milliseconds, [200 - 8000]

Trans Expire: [] seconds, [1 - 64]

Invite Expire: [] seconds, [180 - 300]

Back Next

Interworking Profile

Privacy

Privacy Enabled: ☐

User Name: []

P-Asserted-Identity: ☐

P-Preferred-Identity: ☐

Privacy Header: []

Back Next

In the final dialogue box, leave the **Record Routes** at the default setting of **None** and ensure that the **Has Remote SBC** box is checked. Note that Avaya extensions are not supported for the SIP Trunk. Click on **Finish**

The screenshot shows the 'Interworking Profile' configuration window. It contains several settings: 'Record Routes' is set to 'None' (selected with a radio button); 'Include End Point IP for Context Lookup' is unchecked; 'Extensions' is set to 'None' (dropdown); 'Diversion Manipulation' is unchecked; 'Diversion Condition' is set to 'None' (dropdown); 'Diversion Header URI' is an empty text field; 'Has Remote SBC' is checked (checkbox); 'Route Response on Via Port' is unchecked; and 'DTMF Support' is set to 'None' (selected with a radio button). At the bottom, there are 'Back' and 'Finish' buttons.

| | |
|---|---|
| Interworking Profile | |
| Record Routes | <input checked="" type="radio"/> None <input type="radio"/> Single Side <input type="radio"/> Both Sides <input type="radio"/> Dialog-Initiate Only (Single Side) <input type="radio"/> Dialog-Initiate Only (Both Sides) |
| Include End Point IP for Context Lookup | <input type="checkbox"/> |
| Extensions | None |
| Diversion Manipulation | <input type="checkbox"/> |
| Diversion Condition | None |
| Diversion Header URI | |
| Has Remote SBC | <input checked="" type="checkbox"/> |
| Route Response on Via Port | <input type="checkbox"/> |
| DTMF | |
| DTMF Support | <input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO |
| Back Finish | |

Repeat the process to define Server Interworking for Session Manager using the same parameter settings.

7.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. The TELEPHONIE SIP trunk is connected as a Trunk Server. Session Manager is connected as a Call Server.

To define the TELEPHONIE SIP Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.

The screenshot shows the 'Server Configuration: NTKW' page. On the left is a navigation menu with options: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles (selected), Domain DoS, Server Interworking, Media Forking, Routing, and Server Configuration (highlighted in red). The main content area has a title 'Server Configuration: NTKW' and an 'Add' button. A modal dialog titled 'Add Server Configuration Profile' is open, featuring a 'Profile Name' field with 'NTKW' entered and a 'Next' button. Below the dialog, a table shows configuration details:

| | |
|-------------------------------|-------|
| Signaling Manipulation Script | None |
| Connection Type | SUBID |

Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the SFR IP address.
- In the **Port** box, enter the port to be used for the SIP Trunk. This was left blank during testing which defaults to 5060 when UDP is used for transport.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Next**.

The screenshot shows the 'Edit Server Configuration Profile - General' dialog box. It contains a 'Server Type' dropdown menu set to 'Trunk Server'. Below this is an 'Add' button. A table lists the configuration details:

| IP Address / FQDN | Port | Transport |
|-------------------|------|-----------|
| 10.104.129.46 | 5060 | UDP |

At the bottom of the dialog are 'Back' and 'Next' buttons. A 'Delete' link is also present next to the configuration entry.

Click on **Next** and **Next** again. Leave the fields in the dialogue boxes at default values.

| Add Server Configuration Profile - Authentication | Add Server Configuration Profile - Heartbeat |
|---|---|
| Enable Authentication <input type="checkbox"/> | Enable Heartbeat <input type="checkbox"/> |
| User Name <input type="text"/> | Method <input type="text" value="OPTIONS"/> |
| Realm (Leave blank to detect from server challenge) <input type="text"/> | Frequency <input type="text" value="30"/> seconds |
| Password <input type="text"/> | From URI <input type="text"/> |
| Confirm Password <input type="text"/> | To URI <input type="text"/> |
| <input type="button" value="Back"/> <input type="button" value="Next"/> | <input type="button" value="Back"/> <input type="button" value="Next"/> |

Click on **Next** again to get to the final dialogue box. This contains the **Advanced** settings:

- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for TELEPHONIE SIP defined in **Section 7.4**.
- Leave the other fields at default settings.
- Click **Finish**.

| Add Server Configuration Profile - Advanced | |
|---|------------------------------------|
| Enable DoS Protection | <input type="checkbox"/> |
| Enable Grooming | <input type="checkbox"/> |
| Interworking Profile | <input type="text" value="SFR"/> |
| Signaling Manipulation Script | <input type="text" value="None"/> |
| Connection Type | <input type="text" value="SUBID"/> |
| Securable | <input type="checkbox"/> |
| <input type="button" value="Back"/> <input type="button" value="Finish"/> | |

Use the process above to define the Call Server configuration for Session Manager if not already defined.

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.
- Ensure that the Interworking Profile defined for Session Manager in **Section 7.4** is selected in the **Interworking Profile** drop down menu in the Advanced dialogue box

The following screenshot shows the **General** tab of the completed Server Configuration:

The screenshot shows the 'Server Configuration: CPE' window with the 'General' tab selected. The left sidebar shows 'Server Profiles' with 'CPE' and 'NTWK' listed. The main area has tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab contains a 'Server Type' dropdown set to 'Call Server'. Below it is a table with columns 'IP Address / FQDN', 'Port', and 'Transport'. The table has one row with values '10.10.9.31', '5060', and 'TCP'. An 'Edit' button is at the bottom right of the table. At the top right of the window are 'Rename', 'Clone', and 'Delete' buttons.

| IP Address / FQDN | Port | Transport |
|-------------------|------|-----------|
| 10.10.9.31 | 5060 | TCP |

The next screenshot shows the **Advanced** tab.

The screenshot shows the 'Server Configuration: CPE' window with the 'Advanced' tab selected. The left sidebar is the same. The main area has tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'Advanced' tab contains several settings: 'Enable DoS Protection' (checkbox), 'Enable Grooming' (checkbox), 'Interworking Profile' (dropdown set to 'ASM'), 'Signaling Manipulation Script' (dropdown set to 'None'), 'Connection Type' (dropdown set to 'SUBID'), and 'Securable' (checkbox). An 'Edit' button is at the bottom right.

7.6. Define Routing

Routing information is required for routing to the TELEPHONIE SIP trunk on the external side and Session Manager on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling.

To define routing to TELEPHONIE SIP, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box.

The screenshot shows the 'Routing Profiles: WAN' window. The left sidebar has a tree view with 'Global Profiles' expanded, showing 'Domain DoS', 'Server Interworking', 'Media Forking', and 'Routing' (highlighted). The main area has a title 'Routing Profiles: WAN' and an 'Add' button. Below it is a 'Routing Profiles' list with 'default' and a 'Routing Profile' button. A dialog box titled 'Routing Profile' is open, showing a 'Profile Name' field with 'WAN' entered and a 'Next' button.

Click on **Next** and enter details for the Routing Profile for the SIP Trunk:

- During testing, **Load Balancing** was not required and was left at the default value of **Priority**.
- Click on **Add** to specify an IP address for the SIP Trunk.
- Assign a priority in the **Priority / Weight** field, during testing **1** was used.
- Select the Server Configuration defined in **Section 7.5** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.

Repeat the process for the Routing Profile for Session Manager: The following screenshot shows the completed configuration:

7.7. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop or external interfaces.

To define Topology Hiding for TELEPHONIE SIP, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** to bring up a dialogue box, assign an appropriate name and click on **Next** to configure Topology Hiding for each header as required:

| Header | Criteria | Replace Action |
|--------------|-----------|----------------|
| Record-Route | IP/Domain | Auto |
| Request-Line | IP/Domain | Auto |
| Referred-By | IP/Domain | Auto |
| To | IP/Domain | Auto |

Enter details in the **Topology Hiding Profile** pop-up menu.

- Click on **Add Header** and select from the **Header** drop down menu.
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu depending on requirements. During testing the default **IP/Domain** was used for all headers that hides both domain names and IP addresses.
- Leave the **Replace Action** at the default value of **Auto** unless a specific domain name is required. In this case, select **Overwrite** and define a domain name in the **Overwrite Value** field.
- Topology hiding was defined for all headers where the function is available.

| Header | Criteria | Replace Action | Overwrite Value |
|--------------|-----------|----------------|-----------------|
| Request-Line | IP/Domain | Auto | |

The following screenshot shows the completed **Topology** Hiding configuration for TELEPHONIE SIP.

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

To define Topology hiding for Session Manager, follow the same process. This can be simplified by cloning the profile defined for TELEPHONIE SIP. Do this by highlighting the profile defined for SFR and clicking on **Clone**. Enter an appropriate name for Session Manager and click on **Next** (not shown). Make any changes where required, in the test environment the settings were left at the same values.

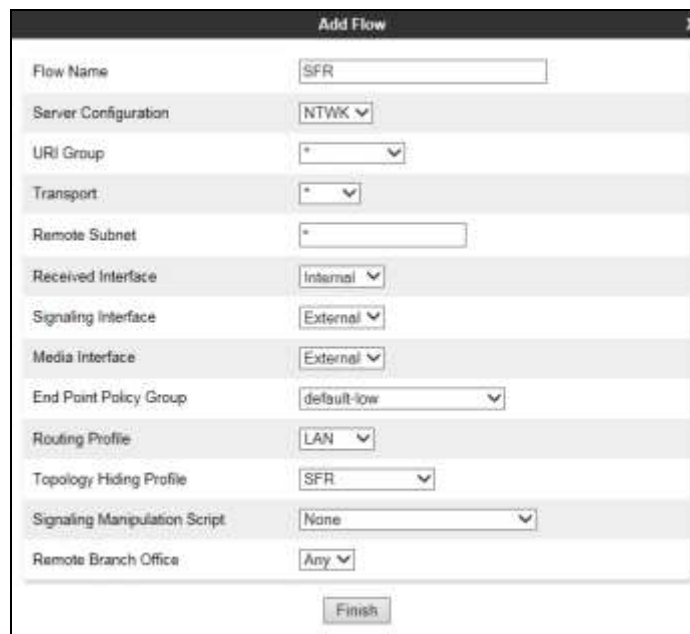
| Header | Criteria | Replace Action | Overwrite Value |
|--------------|-----------|----------------|-----------------|
| Via | IP/Domain | Auto | --- |
| Record-Route | IP/Domain | Auto | --- |
| Request-Line | IP/Domain | Auto | --- |
| Referred-By | IP/Domain | Auto | --- |
| To | IP/Domain | Auto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

7.8. Server Flows

Server Flows combine the previously defined profiles into two End Point Server Flows, one for the Session Manager and another for the TELEPHONIE SIP trunk. This configuration ties all the previously entered information together so that calls can be routed from the Session Manager to the TELEPHONIE SIP trunk and vice versa.

To define a Server Flow for the TELEPHONIE SIP trunk, navigate to **Device Specific Settings** → **End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for the TELEPHONIE SIP trunk, in the test environment **SFR** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the TELEPHONIE SIP trunk defined in **Section 7.5**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the SIP Trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for the SIP Trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 7.3**. This is the interface that media bound for the SIP Trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of Session Manager defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the TELEPHONIE SIP trunk defined in **Section 7.7** and click **Finish**.



The screenshot shows the 'Add Flow' dialog box with the following fields and values:

| Field | Value |
|-------------------------------|-------------|
| Flow Name | SFR |
| Server Configuration | NTWK |
| URI Group | * |
| Transport | * |
| Remote Subnet | * |
| Received Interface | Internal |
| Signaling Interface | External |
| Media Interface | External |
| End Point Policy Group | default-low |
| Routing Profile | LAN |
| Topology Hiding Profile | SFR |
| Signaling Manipulation Script | None |
| Remote Branch Office | Any |

At the bottom of the dialog is a 'Finish' button.

To define a Server Flow for Session Manager, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Server Configuration** drop-down menu, select the server configuration for Session Manager defined in **Section 7.5**.
- In the **Flow Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **CPE** was used.
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 7.3**. This is the interface that signalling bound for Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 7.3**. This is the interface that media bound for Session Manager is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the TELEPHONIE SIP trunk defined in **Section 7.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 7.7** and click **Finish**.

The screenshot shows a dialog box titled "Add Flow" with a close button (X) in the top right corner. The dialog contains the following fields and values:

| Field | Value |
|-------------------------------|-------------|
| Flow Name | CPE |
| Server Configuration | CPE |
| URI Group | * |
| Transport | * |
| Remote Subnet | * |
| Received Interface | External |
| Signaling Interface | Internal |
| Media Interface | Internal |
| End Point Policy Group | default-low |
| Routing Profile | WAN |
| Topology Hiding Profile | ASM |
| Signaling Manipulation Script | None |
| Remote Branch Office | Any |

At the bottom center of the dialog is a button labeled "Finish".

The information for all Server Flows is shown on a single screen on the Avaya SBCE.

End Point Flows: GSSCP_V9

Devices
GSSCP_V9

Subscriber Flows **Server Flows**

Add

Click here to add a row description

Server Configuration: CPE

| Priority | Flow Name | URI Group | Received Interface | Signaling Interface | End Point Policy Group | Routing Profile | |
|----------|-----------|-----------|--------------------|---------------------|------------------------|-----------------|------------------------|
| 1 | CPE | * | External | Internal | default-low | WAN | View Clone Edit Delete |

Server Configuration: NTWK

| Priority | Flow Name | URI Group | Received Interface | Signaling Interface | End Point Policy Group | Routing Profile | |
|----------|-----------|-----------|--------------------|---------------------|------------------------|-----------------|------------------------|
| 1 | SFR | * | Internal | External | default-low | LAN | View Clone Edit Delete |

8. Configure the SFR TELEPHONIE SIP service Equipment

The configuration of the SFR equipment used to support the SIP Trunk is outside the scope of these Application Notes and will not be covered. To obtain further information on SFR equipment and system configuration please contact an authorised SFR representative.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager **Home** tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entities from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.

| SIP Entity Name | SIP Entity Resolved IP | Port | Proto. | Deny | Conn. Status | Reason Code | Link Status |
|------------------|------------------------|------|--------|-------|--------------|-------------|-------------|
| CM_SIP_Endpoints | 10.10.9.12 | 5061 | TLS | FALSE | UP | 200 OK | UP |
| ASBCE | 10.10.9.81 | 5060 | TCP | FALSE | UP | 200 OK | UP |
| CM Trunk | 10.10.9.12 | 5062 | TCP | FALSE | UP | 200 OK | UP |
| Messaging | 10.10.2.82 | 5060 | TCP | FALSE | UP | 200 OK | UP |

2. From Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **in-service/idle**.

```
status trunk 2
```

| TRUNK GROUP STATUS | | | |
|--------------------|--------|-----------------|---------------------------|
| Member | Port | Service State | Mtce Connected Ports Busy |
| 0002/001 | T00011 | in-service/idle | no |
| 0002/002 | T00012 | in-service/idle | no |
| 0002/003 | T00013 | in-service/idle | no |
| 0002/004 | T00014 | in-service/idle | no |
| 0002/005 | T00015 | in-service/idle | no |
| 0002/006 | T00016 | in-service/idle | no |
| 0002/007 | T00017 | in-service/idle | no |

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.
7. Should issues arise with the SIP trunk, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBCE to the network SBCs are receiving a response.

To define the trace, navigate to **Device Specific Settings → Advanced Options → Troubleshooting → Trace** in the main menu on the left hand side and select the **Packet Capture** tab.

- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select the signalling interface IP address or **All** from the **Local Address** drop down menu.
- Enter the IP address of the network SBC in the **Remote Address** field or enter a * to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example.
- Specify the filename of the resultant pcap file in the **Capture Filename** field.
- Click on **Start Capture**.

The screenshot displays the Avaya SBCE configuration interface for packet capture. On the left, a navigation tree lists various system settings, with 'Trace' highlighted under the 'Troubleshooting' section. The main content area, titled 'Trace: GSSCP_V9', features a 'Packet Capture' tab. Below this tab is a 'Packet Capture Configuration' form. The form includes the following fields and values:

- Status:** Ready
- Interface:** S1 (selected from a dropdown)
- Local Address (IP Port):** All (selected from a dropdown)
- Remote Address:** * (entered in the text field)
- Protocol:** All (selected from a dropdown)
- Maximum Number of Packets to Capture:** 10000 (entered in the text field)
- Capture Filename:** SIP_Trunk_Test.pcap (entered in the text field, with a note: 'Using the name of an existing capture will overwrite it')

At the bottom of the configuration form are two buttons: 'Start Capture' and 'Clear'.

To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.



The trace is viewed as a standard pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response to OPTIONS in the form of a 200 OK will be seen from the SFR network.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura ® Communication Manager R7.0, Avaya Aura ® Session Manager 7.0 and Avaya Session Border Controller for Enterprise R7.0 to TELEPHONIE SIP. The SFR TELEPHONIE SIP service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. Additional References

This section references the documentation relevant to these Application Notes. SFR TELEPHONIE SIP is described by the SPECIFICATIONS TECHNIQUES D'ACCES AU SERVICE (STAS) document provided by SFR.

Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0, Nov 2015.
- [2] *Upgrading and Migrating Avaya Aura® applications to 7.0*, Release 7.0, Nov 2015.
- [3] *Deploying Avaya Aura® applications*, Release 7.0, Oct 2015
- [4] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, August 2015
- [5] *Administering Avaya Aura® Communication Manager* Release 7.0, August 2015.
- [6] *Deploying Avaya Aura® System Manager* Release 7.0, Nov 2015
- [7] *Upgrading Avaya Aura® Communication Manager to Release 7.0*, Release 7.0, August 2015
- [8] *Upgrading Avaya Aura® System Manager to Release 7.0*, Nov 2015.
- [9] *Administering Avaya Aura® System Manager for Release 7.0* Release 7.0, Nov 2015
- [10] *Deploying Avaya Aura® Session Manager on VMware* , Release 7.0, August 2015
- [11] *Upgrading Avaya Aura® Session Manager*, Release 7.0, August 2015
- [12] *Administering Avaya Aura® Session Manager* Release 7.0, August 2015,
- [13] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015
- [14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015
- [15] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Nov 2015
- [16] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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