



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.3 as an Evolution Server and Avaya Session Border Controller for Enterprise to support Eircom SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Eircom SIP Trunk service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager as an Evolution Server. Eircom 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 Eircom's SIP Trunk Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise (Avaya SBCE) and Avaya Aura® Communication Manager. Customers using this Avaya SIP-enabled enterprise solution with Eircom SIP Trunk are able to place and receive 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 and Avaya SBCE. The enterprise site was configured to use the SIP Trunking service provided by Eircom.

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 the SIP Trunk provided by Eircom, calls made to SIP, H.323, Digital and Analogue telephones at the enterprise
- All inbound PSTN calls were routed to the enterprise across the SIP trunk to Eircom
- Outgoing calls from the enterprise site completed via Eircom's SIP Trunk to PSTN destinations, calls made from SIP, H.323, Digital and Analogue telephones
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to Eircom
- Inbound and outbound PSTN calls to/from Avaya One-X Communicator and Avaya Flare Experience for Windows softphones
- Calls using the G.729 and G.711A codecs
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38
- Caller ID Presentation and Caller ID Restriction
- DTMF transmission using RFC 2833
- Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer and conference
- Call coverage and call forwarding for endpoints at the enterprise site
- Off-net call forwarding and twinning

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Eircom's SIP Trunk Service with the following observations:

- Inbound Toll-Free calls were not tested as no Toll-Free access was available for test.
- Emergency Services access was not tested as an Emergency Services test call was not booked with the Operator.

2.3. Support

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

For technical support on Eircom products please contact Eircom Customer Care at:

- Telephone: 1800 255 255
- Telephone: +353 1 4688530
- Email: servicedesk@eircom.ie

3. Reference Configuration

The following equipment in **Figure 1** illustrates the test configuration. The Avaya SBCE is located at the edge of the enterprise. It has a public side that connects to the external network and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise flows through the Avaya SBCE. In this way, the Avaya SBCE can protect the enterprise against any SIP-based attacks. The Avaya SBCE provides network address translation at both the IP and SIP layers. Also located at the enterprise site are Communication Manager and IP Desk Phone endpoints. These endpoints include Avaya 96xx series IP telephones, Avaya 46xx series IP telephones, Avaya 16xx series IP telephones, Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone running on a laptop PC.

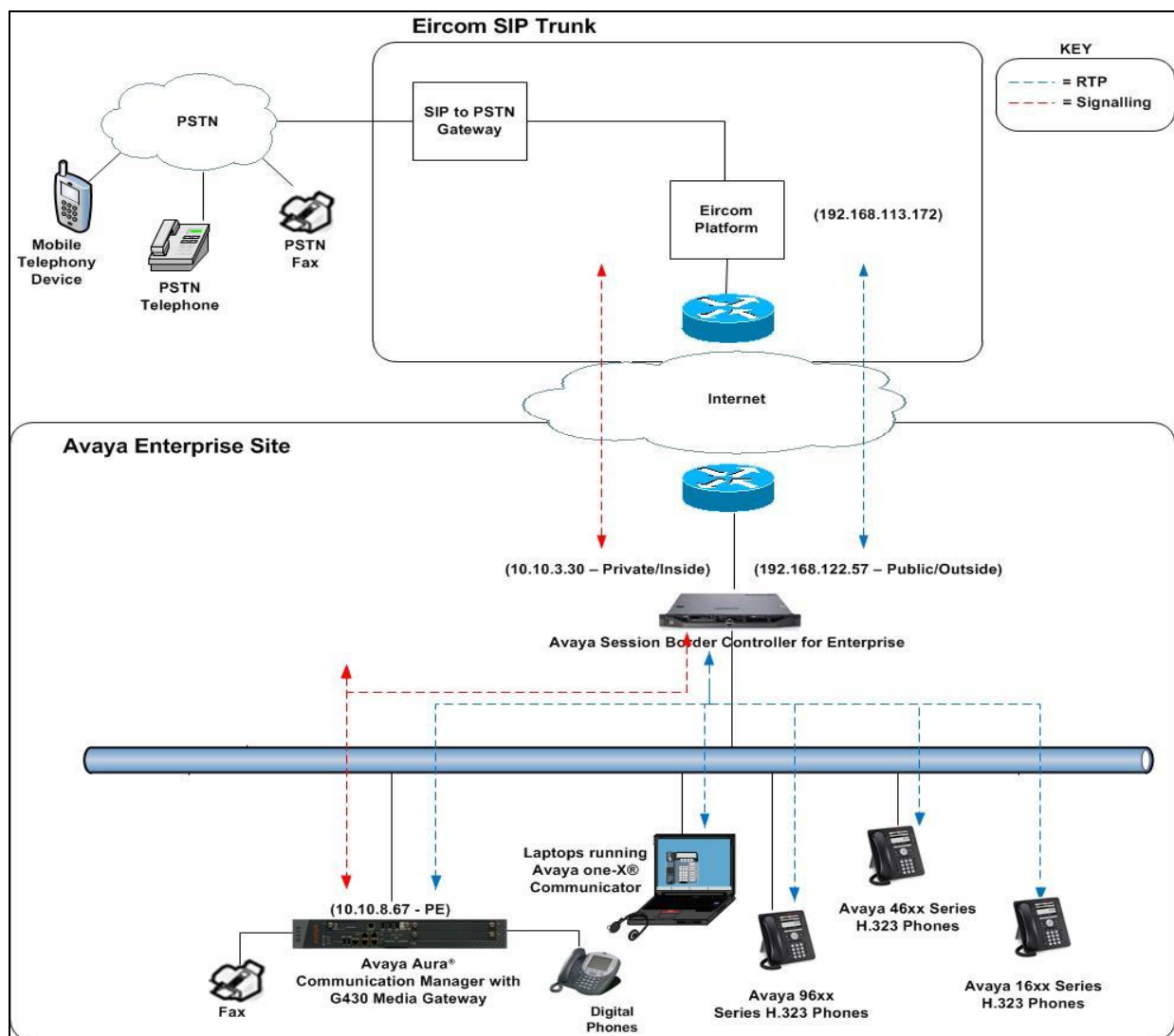


Figure 1: Test Setup Eircom SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|--|--|
| Avaya | |
| Avaya S8800 Server running Communication Manager | R016x.03.0.124.0 -21291 |
| Avaya Session Border Controller for Enterprise | 6.2.1.Q7 |
| Avaya 96xx Series IP DeskPhone (H.323) | 6.3 |
| Avaya 46xx Series IP DeskPhone (H.323) | 6.3 |
| Avaya 16xx Series IP DeskPhone (H.323) | 6.2.2 |
| Avaya one-X® Communicator (H.323) on Lenovo T510 Laptop PC | 6.1.8.06-SP8-40314 |
| Avaya 9408 Digital Handset | Rel 12.0 |
| Analogue Handset | N/A |
| Analogue Fax | N/A |
| Eircom | |
| Eircom SIP Trunk Service | Broadsoft Broadworks rel 19SP1 Ericsson IMS rel 13A AcmePacket SD running on 4500 platform, software release 6.4 |

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for the Eircom SIP Trunk service. SIP trunks are established between Communication Manager and Avaya SBCE. These SIP trunks will carry SIP signalling associated with the Eircom SIP Trunk. The transport protocol between Avaya SBCE and Eircom across the public IP network is UDP. The transport protocol between Avaya SBCE and Communication Manager across the enterprise network is TCP.

For incoming calls, the call flows from the service provider to the Avaya SBCE. After the Avaya SBCE performs the necessary security checks and interworking manipulation, the call is sent to Communication Manager where further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. Outbound calls to the PSTN are first processed by Communication Manager for outbound feature treatment such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects the correct SIP Trunk, the call is routed to the Avaya SBCE for additional interworking treatment before egress to the Eircom 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 Eircom SIP Trunk network, and any other SIP trunks used.

| display system-parameters customer-options | | Page | 2 of 11 |
|---|--------------|-----------|---------|
| OPTIONAL FEATURES | | | |
| IP PORT CAPACITIES | | USED | |
| Maximum Administered H.323 Trunks: | 12000 | 0 | |
| Maximum Concurrently Registered IP Stations: | 18000 | 3 | |
| Maximum Administered Remote Office Trunks: | 12000 | 0 | |
| Maximum Concurrently Registered Remote Office Stations: | 18000 | 0 | |
| Maximum Concurrently Registered IP eCons: | 414 | 0 | |
| Max Concur Registered Unauthenticated H.323 Stations: | 100 | 0 | |
| Maximum Video Capable Stations: | 41000 | 0 | |
| Maximum Video Capable IP Softphones: | 18000 | 0 | |
| Maximum Administered SIP Trunks: | 24000 | 10 | |
| Maximum Administered Ad-hoc Video Conferencing Ports: | 24000 | 0 | |
| Maximum Number of DS1 Boards with Echo Cancellation: | 522 | 0 | |
| Maximum TN2501 VAL Boards: | 128 | 0 | |
| Maximum Media Gateway VAL Sources: | 250 | 1 | |
| Maximum TN2602 Boards with 80 VoIP Channels: | 128 | 0 | |
| Maximum TN2602 Boards with 320 VoIP Channels: | 128 | 0 | |
| Maximum Number of Expanded Meet-me Conference Ports: | 300 | 0 | |

On **Page 4**, verify that **IP Trunks** field is set to **y**.

| | | |
|--|---|--------------|
| display system-parameters customer-options | | Page 4 of 11 |
| 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? y | |
| 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 | | |

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 Avaya SBCE. In the **IP Node Names** form, assign the node **Name** and **IP Address** for Avaya SBCE. In this case, **ASBCE** and **10.10.3.30** are the **Name** and **IP Address** for the Avaya SBCE SIP interface. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signalling interface to Avaya SBCE.

| | | |
|-----------------------|-------------------|---------------|
| display node-names ip | | IP NODE NAMES |
| Name | IP Address | |
| ASBCE | 10.10.3.30 | |
| default | 0.0.0.0 | |
| procr | 10.10.8.67 | |
| procr6 | :: | |

5.3. Administer IP Network Region

Use the **change ip-network-region x** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name of the enterprise. 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 a PSTN call is shuffled, 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 **1** is used.
- The rest of the fields can be left at default values.

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


Use the **change-ip-codec-set** command to define a list of codecs to use for calls between the enterprise and the service provider. For the compliance test, **ip-codec set 1** was used for this purpose. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec supported by Eircom was configured, namely **G.729** and **G.711A**.

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

- ```

change ip-codec-set 1

```
- Page 2 of 2
- ```

IP Codec Set

Allow Direct-IP Multimedia? n

```
- | | Mode | Redundancy | |
|---------------|----------------------|------------|--------------|
| FAX | t.38-standard | 0 | ECM y |
| Modem | off | 0 | |
| TDD/TTY | US | 3 | |
| Clear-channel | n | 0 | |

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Eircom SIP Trunk network. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of 5061 for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- Set **Peer Detection Enabled** to **y**. The **Peer Server** field defaults to **Others**
- 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 Avaya SBCE (node name **ASBCE** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Near-end Listen Port** and **Far-end Listen Port** to **5060** (Commonly used TCP port value)
- 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 network region 1)
- Leave **Far-end Domain** blank (allows the CM to accept calls from any SIP domain on the associated trunk)
- Set **Direct IP-IP Audio Connections** to **y**
- Set **Initial IP-IP Direct Media** to **n**
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from the Communication Manager)

The default values for the other fields may be used.

| | | |
|---|---|-------------|
| add signaling-group 1 | | Page 1 of 2 |
| SIGNALING GROUP | | |
| Group Number: 1 | 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: Others Prepend '+' to Outgoing | |
| Calling/Alerting/Diverting/Connected Public Numbers? y | | |
| Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n | | |
| Near-end Node Name: procr | Far-end Node Name: ASBCE | |
| Near-end Listen Port: 5060 | Far-end Listen Port: 5060 | |
| | Far-end Network Region: 1 | |
| Far-end Domain: | | |
| Incoming Dialog Loopbacks: eliminate | Bypass If IP Threshold Exceeded? n | |
| | RFC 3389 Comfort Noise? n | |
| DTMF over IP: rtp-payload | Direct IP-IP Audio Connections? y | |
| Session Establishment Timer(min): 3 | IP Audio Hairpinning? n | |
| Enable Layer 3 Test? y | Initial IP-IP Direct Media? n | |
| H.323 Station Outgoing 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 x** command, where **x** is an available trunk group. 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**
- 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 1 | | Page 1 of 21 | |
| TRUNK GROUP | | | |
| Group Number: 1 | Group Type: sip | CDR Reports: y | |
| Group Name: OUTSIDE CALL | COR: 1 | TN: 1 | TAC: 101 |
| 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: 1 | |
| | | 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 Eircom to prevent unnecessary SIP messages during call setup.

| | | | |
|--|------------------------|--|--|
| add trunk-group 1 | | Page 2 of 21 | |
| Group Type: sip | | | |
| TRUNK PARAMETERS | | | |
| Unicode Name: auto | | | |
| | | Redirect On OPTIM Failure: 10000 | |
| SCCAN? n | Digital Loss Group: 18 | | |
| Preferred Minimum Session Refresh Interval(sec): 900 | | | |
| Disconnect Supervision - In? y Out? Y | | | |
| XOIP Treatment: auto | | Delay Call Setup When Accessed Via IGAR? n | |

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in national formats.

| | |
|---------------------------|---------------------------------|
| add trunk-group 1 | 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 **Mark Users as Phone** to **y**
- Set **Send Transferring Party Information** to **n**
- Set **Network Call Direction** to **n**
- Set **Send Diversion Header** to **y**
- Set **Support Request History** to **n**
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Eircom
- Set **Always Use re-INVITE for Display Updates** to **y**
- Set the **Identity for Calling Party Display** to **P-Asserted-Identity**

| | |
|---|--------------|
| add trunk-group 1 | Page 4 of 21 |
| PROTOCOL VARIATIONS | |
| Mark Users as Phone? y | |
| Prepend '+' to Calling/Alerting/Diverting/Connected Number? n | |
| Send Transferring Party Information? n | |
| Network Call Redirection? n | |
| Send Diversion Header? y | |
| Support Request History? n | |
| Telephone Event Payload Type: 101 | |
| Convert 180 to 183 for Early Media? n | |
| Always Use re-INVITE for Display Updates? y | |
| Identity for Calling Party Display: P-Asserted-Identity | |
| Block Sending Calling Party Location in INVITE? n | |
| Accept Redirect to Blank User Destination? n | |
| Enable Q-SIP? n | |

5.7. Administer Calling Party Number Information

Use the **change private-numbering x** command to configure Communication Manager to send the calling party number in the format required. This calling party number is sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

| | | | | | |
|----------------------------|------|--------|------------|-------------|-----------------------|
| change private-numbering 0 | | | | Page 1 of 2 | |
| NUMBERING - PRIVATE FORMAT | | | | | |
| Ext | Ext | Trk | Private | Total | |
| Len | Code | Grp(s) | Prefix | Len | |
| 4 | 60 | 1 | 0768nnnn10 | 10 | Total Administered: 2 |
| 4 | 61 | 1 | 0768nnnn11 | 10 | Maximum Entries: 540 |

Note: The above configuration accepts all 4 digit numbers starting with 6, which includes all SIP and H.323 extension numbers, and passes them on with no prefix.

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 Eircom's SIP Trunk. 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: 7 | | |
| 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 1**.

| | | | | | | | |
|--------------------------|---------------|-----------|-----------|---------------|-----------|----------|-----------------|
| 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 |
| | 0 | 11 | 14 | 1 | pubu | | n |
| | 00 | 13 | 15 | 1 | pubu | | n |
| | 0035391 | 13 | 13 | 1 | pubu | | n |
| | 030 | 10 | 10 | 1 | pubu | | n |
| | 0800 | 8 | 10 | 1 | pubu | | n |
| | 0900 | 8 | 8 | 1 | pubu | | n |
| | 118 | 3 | 6 | 1 | pubu | | n |

Use the **change ars digit-conversion x** command to change a dialled number for more efficient routing. As Eircom require a prefix 0 to be inserted before all dialled numbers for calls to route correctly, the **change ars digit-conversion 0** replaces the need of an Avaya SBCE Sigma Script resulting in less header manipulation and SBC processing. The example entry shown will match outgoing calls to national and international numbers beginning with 0.

| | | | | | | | |
|-------------------------------|-----|-----|-----|--------------------|-----|------|-----------------|
| change ars digit-conversion 0 | | | | | | | Page 1 of 2 |
| ARS DIGIT CONVERSION TABLE | | | | | | | |
| Location: all | | | | | | | Percent Full: 0 |
| Matching Pattern | Min | Max | Del | Replacement String | Net | Conv | ANI Req |
| 0 | 1 | 16 | 0 | 0 | ars | y | n |

Use the **change route-pattern x** command, where **x** is an available route pattern, to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group **1**. **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 **unk-unk**.

| | | | | | | | | | | | | | | | |
|------------------------|-----|-------|-----|--------|------|-------------------|----------|-----------------|------|------|-----------|------------|------|----|---|
| change route-pattern 1 | | | | | | | | | | | | Page | 1 | of | 3 |
| | | | | | | Pattern Number: 1 | | Pattern Name: | | | | | | | |
| | | | | | | SCCAN? n | | Secure SIP? n | | | | | | | |
| Grp | FRL | NPA | Pfx | Hop | Toll | No. | Inserted | DCS/ IXC | | | | | | | |
| No | | | Mrk | Lmt | List | Del | Digits | QSIG | | | | | | | |
| | | | | | | | | Intw | | | | | | | |
| 1: 1 | 0 | | | | | | | | n | user | | | | | |
| 2: | | | | | | | | | n | user | | | | | |
| 3: | | | | | | | | | n | user | | | | | |
| BCC | | VALUE | TSC | CA-TSC | ITC | | BCIE | Service/Feature | PARM | No. | Numbering | LAR | | | |
| 0 | 1 | 2 | M | 4 | W | Request | | | | | Dgts | Format | | | |
| | | | | | | | | | | | | Subaddress | | | |
| 1: y | y | y | y | y | n | n | rest | | | | | unk-unk | none | | |
| 2: y | y | y | y | y | n | n | rest | | | | | | none | | |
| 3: y | y | y | y | y | n | n | rest | | | | | | none | | |

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Eircom can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by Eircom correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers **353768xxxxxxx** to a 4 digit extension by deleting all of the incoming digits and inserting an extension. Public DID numbers have been masked for security purposes.

| | | | | | | | | |
|---|--------|--------------|------------|--|------|---|----|---|
| change inc-call-handling-trmt trunk-group 1 | | | | | Page | 1 | of | 3 |
| INCOMING CALL HANDLING TREATMENT | | | | | | | | |
| Service/ | Number | Number | Del Insert | | | | | |
| Feature | Len | Digits | | | | | | |
| public-ntwrk | 12 | 353768nnnn10 | all 6010 | | | | | |
| public-ntwrk | 12 | 353768nnnn11 | all 6012 | | | | | |
| public-ntwrk | 12 | 353768nnnn12 | all 6102 | | | | | |

5.10. EC500 Configuration

When EC500 is enabled on the 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 6102. Use the command **change off-pbx-telephone station-mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** enter the phone that will also be called (e.g. **089434nnnn**)
- Set the **Trunk Selection** to **ARS**
- Set the **Config Set** to **1**

| | | | | | | | |
|---|-------------|-------------|----|--------------|-----------------|------------|-------------|
| change off-pbx-telephone station-mapping 2396 | | | | | | | Page 1 of 3 |
| STATIONS WITH OFF-PBX TELEPHONE INTEGRATION | | | | | | | |
| Station Extension | Application | Dial Prefix | CC | Phone Number | Trunk Selection | Config Set | Dual Mode |
| 6102 | EC500 | - | | 089434nnnn | ARS | 1 | |
| - | | | | | | | |

Note: The phone number shown is for a mobile phone used for testing at Avaya Labs and is in national format with national dialling prefix 0. To use facilities for calls coming in from EC500 mobile phones, the number received in Communication Manager must exactly match the number specified in the above table.

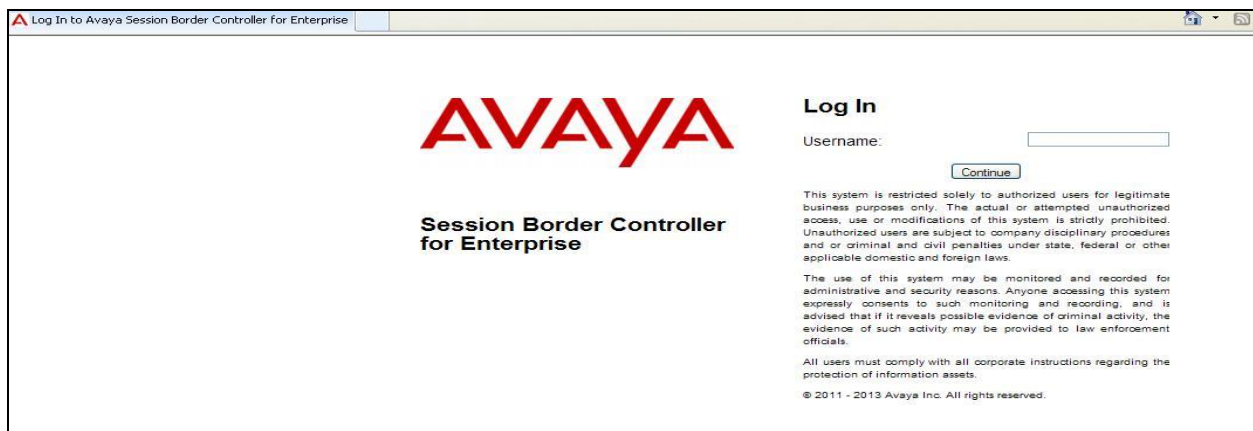
Save Communication Manager changes by entering **save translation** to make them permanent.

6. Configure Avaya Session Border Controller for Enterprise

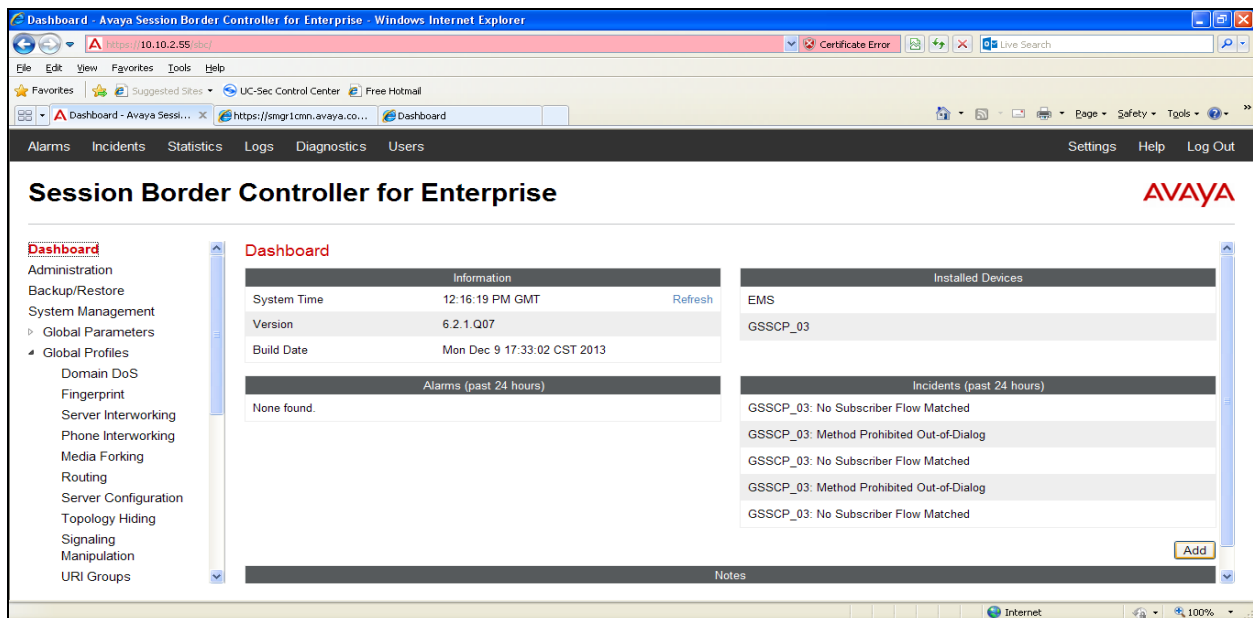
This section describes the configuration of Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation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

6.1. Access Avaya Session Border Controller for Enterprise

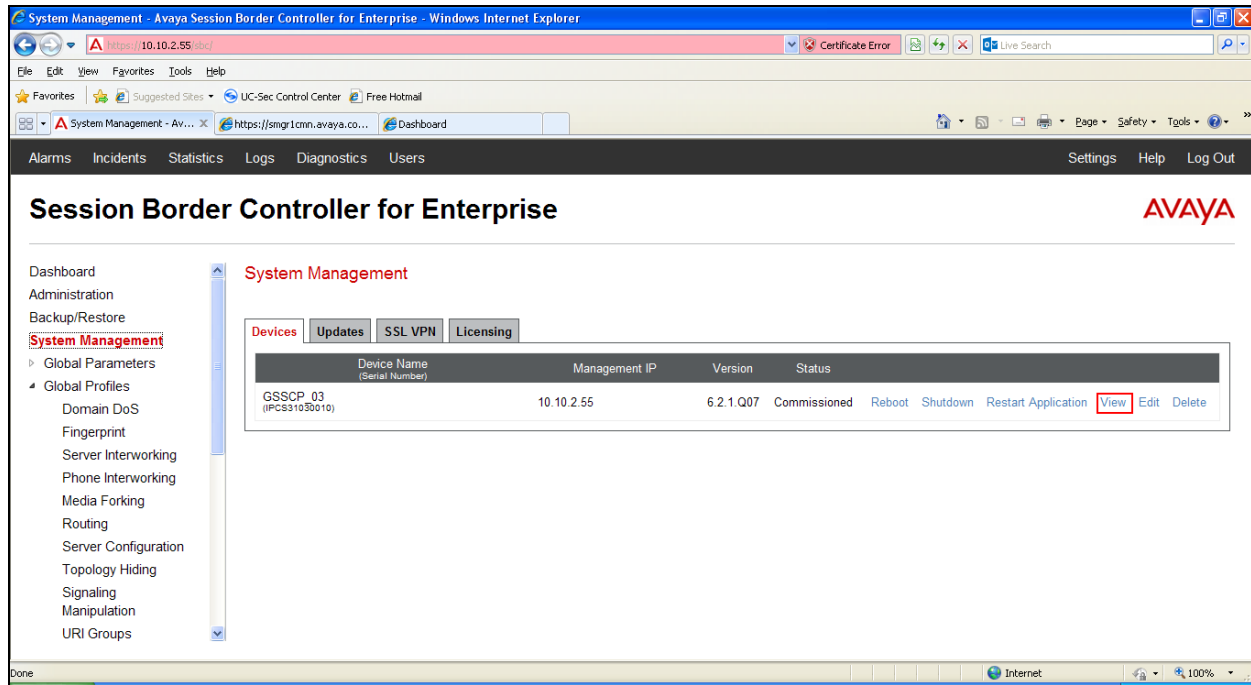
Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



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.



To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **GSSCP_03** is shown. To view the configuration of this device, click **View** (the third option from the right).



The System Information screen shows the **Appliance Name**, **Device Settings** and **DNS Configuration** information.

| System Information: GSSCP_03 | | | | |
|------------------------------|----------------|-----------------------------|---------------|-----------|
| General Configuration | | Device Configuration | | |
| Appliance Name | GSSCP_03 | HA Mode | No | |
| Box Type | SIP | Two Bypass Mode | No | |
| Deployment Mode | Proxy | | | |
| Network Configuration | | | | |
| IP | Public IP | Netmask | Gateway | Interface |
| 10.10.3.30 | 10.10.3.30 | 255.255.255.0 | 10.10.3.1 | A1 |
| 192.168.122.57 | 192.168.122.57 | 255.255.255.128 | 192.168.122.7 | B1 |
| DNS Configuration | | Management IP(s) | | |
| Primary DNS | 8.8.8.8 | IP | 10.10.2.55 | |
| Secondary DNS | 10.10.7.100 | | | |
| DNS Location | DMZ | | | |
| DNS Client IP | 192.168.122.57 | | | |

6.2. Global Profiles

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

6.2.1. Server Interworking - Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles** → **Server Interworking** and click on **Add**.

- Enter profile name such as **Avaya_CM** and click **Next** (Not Shown)
- Check **Hold Support=None**
- Check **T.38 Support**
- All other options on the **General** Tab can be left at default

The screenshot shows a configuration window titled "Profile: Avaya_CM" with a close button (X) in the top right corner. The window contains a "General" tab with various configuration options. The options are as follows:

| Option | Value |
|--------------------------|--|
| Hold Support | <input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly |
| 180 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 181 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 182 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 183 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| Refer Handling | <input type="checkbox"/> |
| URI Group | None (dropdown menu) |
| 3xx Handling | <input type="checkbox"/> |
| Diversion Header Support | <input type="checkbox"/> |
| Delayed SDP Handling | <input type="checkbox"/> |
| Re-Invite Handling | <input type="checkbox"/> |
| T.38 Support | <input checked="" type="checkbox"/> |
| URI Scheme | <input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY |
| Via Header Format | <input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543 |

At the bottom right of the window is a "Next" button.

Default values can be used for the **Advanced Settings** window. Click **Finish**

Profile: Avaya_CM X

| | |
|---|--|
| Record Routes | <input checked="" type="radio"/> None <input type="radio"/> Single Side <input type="radio"/> Both Sides |
| Topology Hiding: Change Call-ID | <input type="checkbox"/> |
| Call-Info NAT | <input type="checkbox"/> |
| Change Max Forwards | <input checked="" type="checkbox"/> |
| Include End Point IP for Context Lookup | <input type="checkbox"/> |
| OCS Extensions | <input type="checkbox"/> |
| AVAYA Extensions | <input type="checkbox"/> |
| NORTEL Extensions | <input type="checkbox"/> |
| Diversion Manipulation | <input type="checkbox"/> |
| Diversion Header URI | <input type="text"/> |
| Metaswitch Extensions | <input type="checkbox"/> |
| Reset on Talk Spurt | <input type="checkbox"/> |
| Reset SRTP Context on Session Refresh | <input type="checkbox"/> |
| Has Remote SBC | <input checked="" type="checkbox"/> |
| Route Response on Via Port | <input type="checkbox"/> |
| Cisco Extensions | <input type="checkbox"/> |

Finish

6.2.2. Server Interworking – Eircom

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles** → **Server Interworking** and click on **Add**.

- Enter profile name such as **Eircom** and click **Next** (Not Shown)
- Check **Hold Support** = **None**
- Check **T.38 Support**
- All other options on the **General** Tab can be left at default

Click on **Next** on the following screens and then **Finish**

The screenshot shows the 'Profile: Eircom' configuration window with the 'General' tab selected. The window contains the following settings:

| Setting | Value |
|--------------------------|--|
| Hold Support | <input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly |
| 180 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 181 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 182 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| 183 Handling | <input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP |
| Refer Handling | <input type="checkbox"/> |
| URI Group | None |
| 3xx Handling | <input type="checkbox"/> |
| Diversion Header Support | <input type="checkbox"/> |
| Delayed SDP Handling | <input type="checkbox"/> |
| Re-Invite Handling | <input type="checkbox"/> |
| T.38 Support | <input checked="" type="checkbox"/> |
| URI Scheme | <input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY |
| Via Header Format | <input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543 |

Next

Default values can be used for the **Advanced Settings** window. Click **Finish**.

Profile: Eircom X

| | |
|---|--|
| Record Routes | <input checked="" type="radio"/> None <input type="radio"/> Single Side <input type="radio"/> Both Sides |
| Topology Hiding: Change Call-ID | <input type="checkbox"/> |
| Call-Info NAT | <input type="checkbox"/> |
| Change Max Forwards | <input checked="" type="checkbox"/> |
| Include End Point IP for Context Lookup | <input type="checkbox"/> |
| OCS Extensions | <input type="checkbox"/> |
| AVAYA Extensions | <input type="checkbox"/> |
| NORTEL Extensions | <input type="checkbox"/> |
| Diversion Manipulation | <input type="checkbox"/> |
| Diversion Header URI | <input type="text"/> |
| Metaswitch Extensions | <input type="checkbox"/> |
| Reset on Talk Spurt | <input type="checkbox"/> |
| Reset SRTP Context on Session Refresh | <input type="checkbox"/> |
| Has Remote SBC | <input checked="" type="checkbox"/> |
| Route Response on Via Port | <input type="checkbox"/> |
| Cisco Extensions | <input type="checkbox"/> |

Finish

6.2.3. Routing

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

Routing information is required for routing to Communication Manager on the internal side and Eircom addresses on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used.

Create a Routing Profile for both Communication Manager and Eircom SIP trunk. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **URI Group:** Select “*” from the drop down box
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server
- **Routing Priority Based on Next Hop Server:** Checked(not shown)
- **Use Next Hop for In-Dialog Messages:** Select only if there is no secondary Next Hopserver (not shown)
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signaling packets (not shown)

Click **Finish** (not shown).

The following screen shows the Routing Profile to Communication Manager.

Routing Profiles: Avaya_CM

Add

Rename Clone Delete

Click here to add a description.

Routing Profile

Add

| Priority | URI Group | Next Hop Server 1 | Next Hop Server 2 |
|----------|-----------|-------------------|-------------------|
| 1 | * | 10.10.8.67 | --- |

View Edit

The following screen shows the Routing Profile to Eircom SIP Trunk.

Routing Profiles: Eircom

Add

Routing Profiles

default

Avaya_SM

Eircom

Rename

Clone

Delete

Click here to add a description.

Routing Profile

Add

| Priority | URI Group | Next Hop Server 1 | Next Hop Server 2 | |
|----------|-----------|-------------------|-------------------|----------------------|
| 1 | * | 192.168.113.172 | --- | <div>View Edit</div> |

6.2.4. Server Configuration – Avaya

Servers are defined for each server connected to the Avaya SBCE. In this case, Eircom is connected as the Trunk Server and Communication Manager is connected as the Call Server. The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options. From the left-hand menu select **Global Profiles** → **Server Configuration** and click on **Add** (not shown) and enter a descriptive name. On the **Add Server Configuration Profile** tab (not shown), set the following:

- Select **Server Type** to be **Call Server**
- Enter **IP Addresses / Supported FQDNs** to **10.10.8.67** (Communication Manager IP Address)
- For **Supported Transports**, check **TCP**
- **TCP Port:5060**
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs

The screenshot displays the 'Server Configuration Profile - General' window. It contains the following fields and options:

- Server Type:** A dropdown menu set to 'Call Server'.
- IP Addresses / Supported FQDNs:** A text area containing '10.10.8.67'.
- Supported Transports:** Three checkboxes: 'TCP' (checked), 'UDP' (unchecked), and 'TLS' (unchecked).
- TCP Port:** A text box containing '5060'.
- UDP Port:** An empty text box.
- TLS Port:** An empty text box.
- Finish:** A button at the bottom center.

On the **Advanced** tab:

- Select **Avaya_CM** for **Interworking Profile** defined in **Section 7.2.1**.
- Click **Finish**

The screenshot shows a window titled "Server Configuration Profile - Advanced" with a close button (X) in the top right corner. The window contains several configuration options:

- Enable DoS Protection**: A checkbox that is currently unchecked.
- Enable Grooming**: A checkbox that is currently unchecked.
- Interworking Profile**: A dropdown menu with "Avaya_CM" selected. This dropdown is highlighted with a red rectangular box.
- Signaling Manipulation Script**: A dropdown menu with "None" selected.
- TCP Connection Type**: Three radio buttons labeled "SUBID", "PORTID", and "MAPPING". The "SUBID" radio button is selected.

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

6.2.5. Server Configuration – Eircom

To define the Eircom SBC as a Trunk Server, navigate to select **Global Profiles → Server Configuration** and click on **Add** (not shown) and enter a descriptive name. On the **Add Server Configuration Profile** tab, click on **Edit** (not shown) and set the following:

- Select **Server Type** as **Trunk Server**
- Set **IP Address** to **192.168.113.172** (Eircom SIP Trunk)
- **Supported Transports**: Check **UDP**
- **UDP Port**: **5060**
- Hit **Next** (not shown)

The screenshot shows a window titled "Server Configuration Profile - General". It contains several fields and checkboxes. The "Server Type" dropdown menu is set to "Trunk Server". Below it, the "IP Addresses / Supported FQDNs" text box contains the IP address "192.168.113.172". In the "Supported Transports" section, the "UDP" checkbox is checked, while "TCP" and "TLS" are unchecked. The "UDP Port" text box is set to "5060". At the bottom of the window is a "Finish" button.

In the new window that appears, enter the following values, as Eircom require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

Click **Next** to continue.

Server Configuration Profile - Authentication

Enable Authentication ☒

User Name

Realm
(Leave blank to detect from server challenge)

Password
(Leave blank to keep existing password)

Confirm Password

Finish

In the new window that appears, enter the following values.

- **Enabled Heartbeat:** Checked
- **Method:** Select **REGISTER** from the drop-down box
- **Frequency:** Choose the desired frequency in seconds the Avaya SBCE will send SIP REGISTERS
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS
- **TO URI:** Enter an URI to be sent in the TO header for SIP REGISTERS

Click **Next** to continue.

Server Configuration Profile - Heartbeat

Enable Heartbeat ☒

Method REGISTER

Frequency 300 seconds

From URI xxxxxxxxx_TG1@ngv.eir

To URI xxxxxxxxx_TG1@ngv.eir

Finish

On the **Advanced** tab:

- Select **Eircom** for **Interworking Profile** as defined in **Section 7.2.2**.
- Click **Finish**

Server Configuration Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile Eircom

Signaling Manipulation Script None

UDP Connection Type ☒ SUBID ☐ PORTID ☐ MAPPING

Finish

6.2.6. 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. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

To define Topology Hiding for Communication Manager, navigate to **Global Profiles** → **Topology Hiding** from menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya_CM**
- If the required Header is not shown, click on **Add Header**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **avaya.com**.
- Click **Finish** (not shown)

The screenshot shows the 'Topology Hiding Profiles: Avaya_CM' configuration page. On the left, a sidebar lists 'Topology Hiding Profiles' with options: 'default', 'cisco_th_profile', 'Avaya_CM' (selected), and 'Eircom'. The main area has a blue header bar with 'Click here to add a description.' and buttons for 'Add', 'Rename', 'Clone', and 'Delete'. Below the header is a 'Topology Hiding' tab and a table with the following data:

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

The rows for 'Request-Line', 'To', and 'From' are highlighted with a red border.

To define Topology Hiding for Eircom, navigate to **Global Profiles** → **Topology Hiding** from the menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive **Profile Name** such as **Eircom** and click **Next**
- If the required Header is not shown, click on **Add Header**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **ngv.eircom.net**
- Click **Finish** (not shown)

Topology Hiding Profiles: Eircom

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

Topology Hiding Profiles

- default
- cisco_th_profile
- Avaya_SM
- Eircom**

Click here to add a description.

Topology Hiding

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

[Edit](#)

6.3. Define Network Information

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 interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** from the menu on the left-hand side and click on **Add IP**. Enter details in the blank box that appears at the end of the list

- Click on **Add IP**
- Define the internal IP address with screening mask and assign to interface **A1**
- Select **Save** to save the information
- Click on **Add IP**
- Define the external IP address with screening mask and assign to interface **B1**
- Select **Save** to save the information
- Click on **System Management** in the main menu
- Select **Restart Application** indicated by an icon in the status bar (not shown)

Network Management: GSSCP_03

Devices: GSSCP_03

Network Configuration | Interface Configuration

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

Changes will not take effect until the interface is updated.

A1 Netmask: 255.255.255.0 A2 Netmask: B1 Netmask: 255.255.255.128 B2 Netmask:

Add Save Clear

| IP Address | Public IP | Gateway | Interface | |
|----------------|-----------|---------------|-----------|--------|
| 10.10.3.30 | | 10.10.3.1 | A1 | Delete |
| 192.168.122.57 | | 192.168.122.7 | B1 | Delete |

Select the **Interface Configuration** tab and click on **Toggle State** to enable the interfaces.

Network Management: GSSCP_03

Devices: GSSCP_03

Network Configuration | Interface Configuration

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

6.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

6.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** from the menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

The Signalling Interface screen allows the IP address and ports to be set for transporting signaling messages over the SIP trunk. The Avaya SBCE listens for SIP requests on the defined ports. Create a Signaling Interface for both the inside and outside IP interfaces. To create a new Signaling Interface, navigate to **Device Specific Settings → Signaling Interface** and click **Add**.

- **Name: Int_Sig**
- **Signaling IP: 10.10.3.30** (Internal address for calls toward Communication Manager)
- **UDP Port: 5060**
- Click **Finish**
- Select **Add**
- **Name: Ext_Sig**
- **Signaling IP: 192.168.122.57** (External address for calls toward Eircom)
- **TCP Port: 5060**
- **UDP Port: 5060**
- Click **Finish**

The following screen shows the signaling interfaces created in the sample configuration for the inside and outside IP interfaces.

Signaling Interface: GSSCP_03

| Name | Signaling IP | TCP Port | UDP Port | TLS Port | TLS Profile | |
|---------|----------------|----------|----------|----------|-------------|-------------|
| Int_Sig | 10.10.3.30 | 5060 | 5060 | --- | None | Edit Delete |
| Ext_Sig | 192.168.122.57 | 5060 | 5060 | --- | None | Edit Delete |

6.4.2. Media Interfaces

The Media Interface screen allows the IP address and ports to be set for transporting Media over the SIP trunk. The Avaya SBCE listens for SIP media on the defined ports.

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

- Select **Add**
- **Name: Int_Media**
- **Media IP: 10.103.30** (Internal address for calls toward Communication Manager)
- **Port Range: 35000-51000**
- Click **Finish**
- Select **Add**
- **Name: Ext_Media**
- **Media IP: 192.168.122.57** (External address for calls toward Eircom)
- **Port Range: 35000-51000**
- Click **Finish**

The following screen shows the Media Interfaces created in the sample configuration for the inside and outside IP interfaces.

Media Interface: GSSCP_03

Devices

GSSCP_03

Media Interface

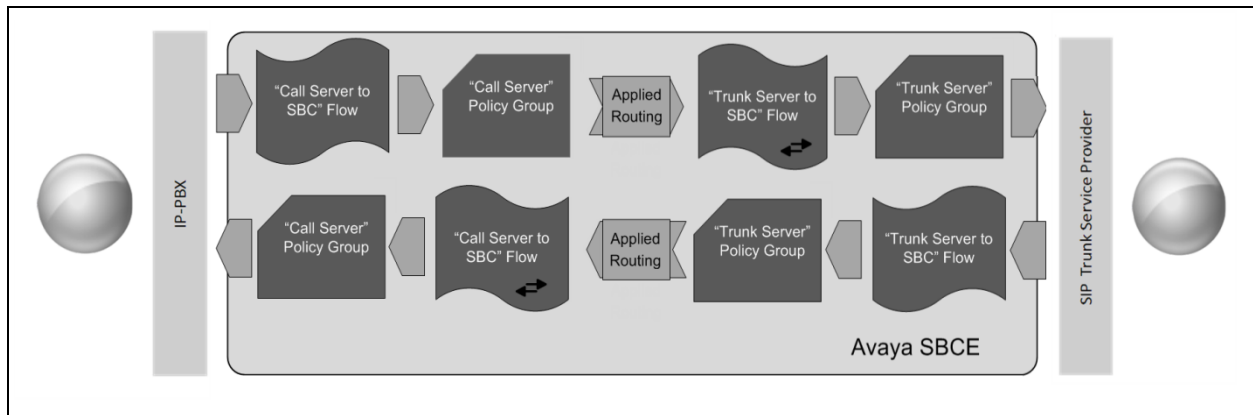
Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

| Name | Media IP | Port Range | Edit | Delete |
|-----------|----------------|---------------|------|--------|
| Int_Media | 10.10.3.30 | 35000 - 51000 | Edit | Delete |
| Ext_Media | 192.168.122.57 | 35000 - 51000 | Edit | Delete |

6.5. Server Flows

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



To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow**.

- **Flow Name:** Enter a descriptive name
- **Server Configuration:** Select a Server Configuration created in **Section 6.2.4** and **6.2.5** and assign to the Flow
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from
- **Signaling Interface:** Select the Signaling Interface used to communicate with the Server Configuration
- **Media Interface:** Select the Media Interface used to communicate with the Server Configuration
- **End Point Policy Group:** Select the policy assigned to the Server Configuration
- **Routing Profile:** Select the profile the Server Configuration will use to route SIP messages
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration

Click **Finish** to save and exit.

The following screen shows the Server Flow for Communication Manager.

The screenshot shows a configuration window titled "Flow: Call_Server". It contains the following fields and values:

| Field | Value |
|-------------------------|-------------|
| Flow Name | Call_Server |
| Server Configuration | Avaya_CM |
| URI Group | * |
| Transport | * |
| Remote Subnet | * |
| Received Interface | Ext_Sig |
| Signaling Interface | Int_Sig |
| Media Interface | Int_Media |
| End Point Policy Group | default-low |
| Routing Profile | Eircom |
| Topology Hiding Profile | Avaya_CM |
| File Transfer Profile | None |

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

The following screen shows the Server Flow for Eircom.

The screenshot shows a configuration window titled "Flow: Trunk_Server". It contains the following fields and values:

| Field | Value |
|-------------------------|--------------|
| Flow Name | Trunk_Server |
| Server Configuration | Eircom |
| URI Group | * |
| Transport | * |
| Remote Subnet | * |
| Received Interface | Int_Sig |
| Signaling Interface | Ext_Sig |
| Media Interface | Ext_Media |
| End Point Policy Group | default-low |
| Routing Profile | Avaya_SM |
| Topology Hiding Profile | Eircom |
| File Transfer Profile | None |

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

This configuration ties all the previously entered information together so that calls can be routed from Communication Manager to Eircom SIP Trunk service and vice versa. The following screenshot shows all configured flows.

End Point Flows: GSSCP_03

Devices

GSSCP_03

Subscriber Flows

Server Flows

Add

Hover over a row to see its description.

Server Configuration: Avaya_CM

| Priority | Flow Name | URI Group | Received Interface | Signaling Interface | End Point Policy Group | Routing Profile | |
|----------|-------------|-----------|--------------------|---------------------|------------------------|-----------------|--|
| 1 | Call_Server | * | Ext_Sig | Int_Sig | default-low | Eircom | View Clone Edit Delete |

Server Configuration: Eircom

| Priority | Flow Name | URI Group | Received Interface | Signaling Interface | End Point Policy Group | Routing Profile | |
|----------|--------------|-----------|--------------------|---------------------|------------------------|-----------------|--|
| 1 | Trunk_Server | * | Int_Sig | Ext_Sig | default-low | Avaya_CM | View Clone Edit Delete |

7. Configure Eircom SIP Trunk Equipment

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

8. Verification Steps

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

1. From the 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 1 | | | |
|--------------------|--------|-----------------|------------------------------|
| TRUNK GROUP STATUS | | | |
| Member | Port | Service State | Mtce Connected Ports Busy |
| 0001/001 | T00001 | in-service/idle | no |
| 0001/002 | T00002 | in-service/idle | no |
| 0001/003 | T00003 | in-service/idle | no |
| 0001/004 | T00004 | in-service/idle | no |
| 0001/005 | T00005 | in-service/idle | no |
| 0001/006 | T00006 | in-service/idle | no |
| 0001/007 | T00007 | in-service/idle | no |
| 0001/008 | T00008 | in-service/idle | no |
| 0001/009 | T00009 | in-service/idle | no |
| 0001/010 | T00010 | in-service/idle | no |

2. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
3. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
4. Verify that the user on the PSTN can end an active call by hanging up.
5. Verify that an endpoint at the enterprise site can end an active call by hanging up.
6. Should issues arise with the SIP trunk, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from Communication 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 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**

Trace: GSSCP_03

Devices
GSSCP_03

Call Trace | **Packet Capture** | Captures

Packet Capture Configuration

| | |
|---|---------------------|
| Status | Ready |
| Interface | B1 |
| Local Address <small>IP[:Port]</small> | 192.168.122.57 : |
| Remote Address <small>*, *:Port, IP, IP:Port</small> | * |
| Protocol | All |
| Maximum Number of Packets to Capture | 10000 |
| Capture Filename <small>Using the name of an existing capture will overwrite it.</small> | SIP_Trunk_Test.pcap |

Start Capture
Clear

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

Trace: GSSCP_03

Devices
GSSCP_03

Call Trace | Packet Capture | **Captures**

| File Name | File Size (bytes) | Last Modified | |
|------------------------------------|-------------------|------------------------------------|--------|
| SIP_Trunk_Test_20140916121852.pcap | 0 | September 16, 2014 12:18:52 PM GMT | Delete |

Refresh

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

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager R6.3 as an Evolution Server and Avaya Session Border Controller for Enterprise to Eircom's SIP Trunk Service. Eircom's SIP Trunk 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**.

10. Additional References

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

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3, May 2014
- [2] *Avaya Aura® Communication Manager using VMware® in the Virtualized Environment Deployment Guide*, April 2014
- [3] *Avaya Aura® Communication Manager 6.3 Documentation library*, August 2014
- [4] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2 June 2014
- [5] *Upgrading Avaya Session Border Controller for Enterprise* Release 6.2 July 2014
- [6] *Administering Avaya Session Border Controller for Enterprise* Release 6.2 March 2014
- [7] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

©2014 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.