Avaya Solution & Interoperability Test Lab

Application Notes for Syntec CardEasy Core Cloud solution with Avaya Session Border Controller for Enterprise, Avaya Aura® Communication Manager and using a SIP Trunk - Issue 1.0

Abstract

These Application Notes describe the configuration required to allow Syntec CardEasy Core Cloud solution to interoperate with Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager using a SIP Trunk. Syntec CardEasy Core allows customers to securely enter credit card details during a transaction with an agent and have the payment authorized and confirmed.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any 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

The Configuration used in these application notes was used to verify that Syntec CardEasy Core Cloud solution interoperates with Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager using a SIP Trunk. The CardEasy Core is a cloud based solution that interoperates with the Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager to allow Avaya Aura® Communication Manager agents to initiate a credit card payment and for a Customer to enter credit card details securely during the transaction. The Syntec CardEasy Core masks DTMF digits and Speech during the credit card verification process.

2. General Test Approach and Test Results

The general test approach was to configure the CardEasy Core to communicate with the Avaya Session Border Controller for Enterprise (Avaya SBCE) and Communication Manager (CM) via a SIP trunk. Syntec EPID collection utility was connected to AES and transmitted DTMF to the CardEasy Core to identify the called agent. Testing was performed by calling inbound to a VDN and using Vectors to allow the calling party to speak to an agent and enter credit card details and have a payment authorized during a transaction. The DTMF digits or spoken credit card details are masked and hidden from the agent and confirmation is sent to the Agents payment page.

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.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g. jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another, and may affect the reliability or performance of the overall solution. Different network elements (e.g. session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations, and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on receiving calls in different call scenarios and completing a credit card payment transaction. The tests included:

- Call Placed with Available Agents.
- Calls on Hold, Mute and Transferred.
- Credit Card Transaction with valid and invalid details.
- Failover/Service – Tests the behaviour of the CardEasy Core during certain failed conditions.

2.2. Test Results

All Tests were executed successfully with the following observation

- The SIP protocol must be the same for both the Service Provider and Communication Manager connections to the Session Border Controllers or certain call types will fail
- In Avaya Session Border Controller for Enterprise 7.1 SP1 an issue has been identified with DTMF transportation to the CardEasy Core. A Signaling Manipulation script has been applied as a workaround and is described in **Section 6.5**. A fix for this issue is planned for SP2 due in March 2017 and the fix can be obtained via Devconnect support ticket **24804**

2.3. Support

Technical Support can be obtained for Syntec products from the following:

Web: https://support.syntec.co.uk/portal/syntec
Email: support@syntec.co.uk
Telephone: +44 (0) 207 741 8000

3. Reference Configuration

Figure 1 below shows the system configuration for the interoperability between Syntec CardEasy Core, Avaya Session Border Controller for Enterprise and Communication Manager using a SIP trunk. Avaya 9611g H323 IP Deskphones were used with an Avaya Call Center Elite Agent logged in to receive incoming calls.

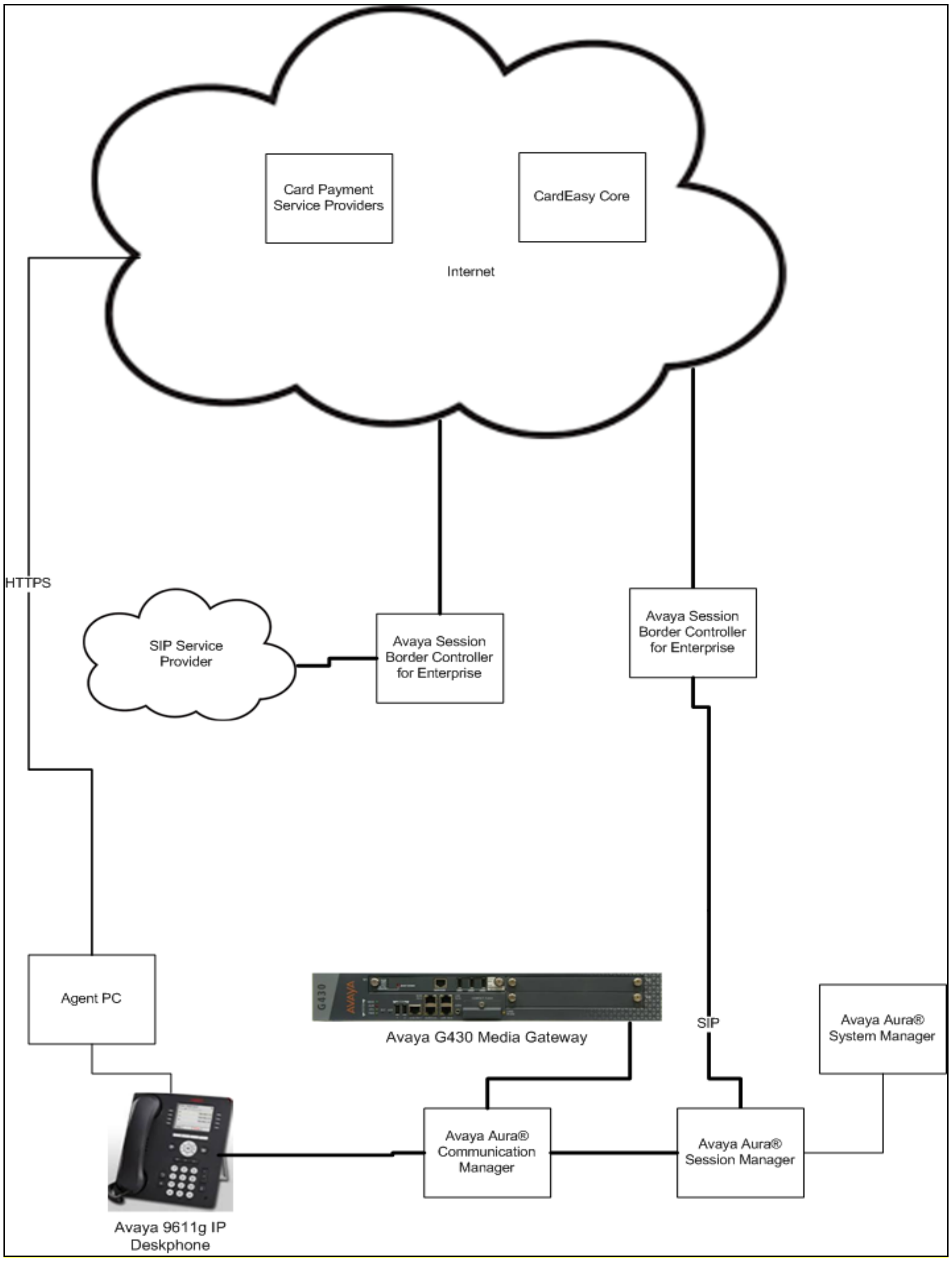

Card Payment Service Providers
CardEasy Core
Internet
HTTPS
SIP Service Provider
Avaya Session Border Controller for Enterprise
Avaya Session Border Controller for Enterprise
Agent PC
Avaya G430 Media Gateway
SIP
Avaya Aura® System Manager
Avaya Aura® Communication Manager
Avaya Aura® Session Manager
Avaya 9611g IP Deskphone

Figure 1: Syntec CardEasy Core with Avaya Session Border Controller for Enterprise and Communication Manager using a SIP Trunk

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on a VMware Virtual Server	R7.0.1.2 R017x.00.0.441.0 Version 7.0.1.2.0.441.23523 Patch: • Kernel-2.6.32.3.1.e16.AV4 • PLAT-rhel6.5-0050
Avaya G430 Media Gateway	37.41.0/1
Avaya Session Border Controller for Enterprise	7.1.0.0-04-11122
Avaya Aura® Session Manager	7.0.1.2.701230
Avaya Aura® System Manager	Version: 7.0.1.2 Build: 7.0.0.0.16266 Software Update Revision: 7.0.1.2.086007 Service Pack 2
Avaya 9611g IP Deskphone (H323)	6.6229
Syntec CardEasy Core	V2.3.21

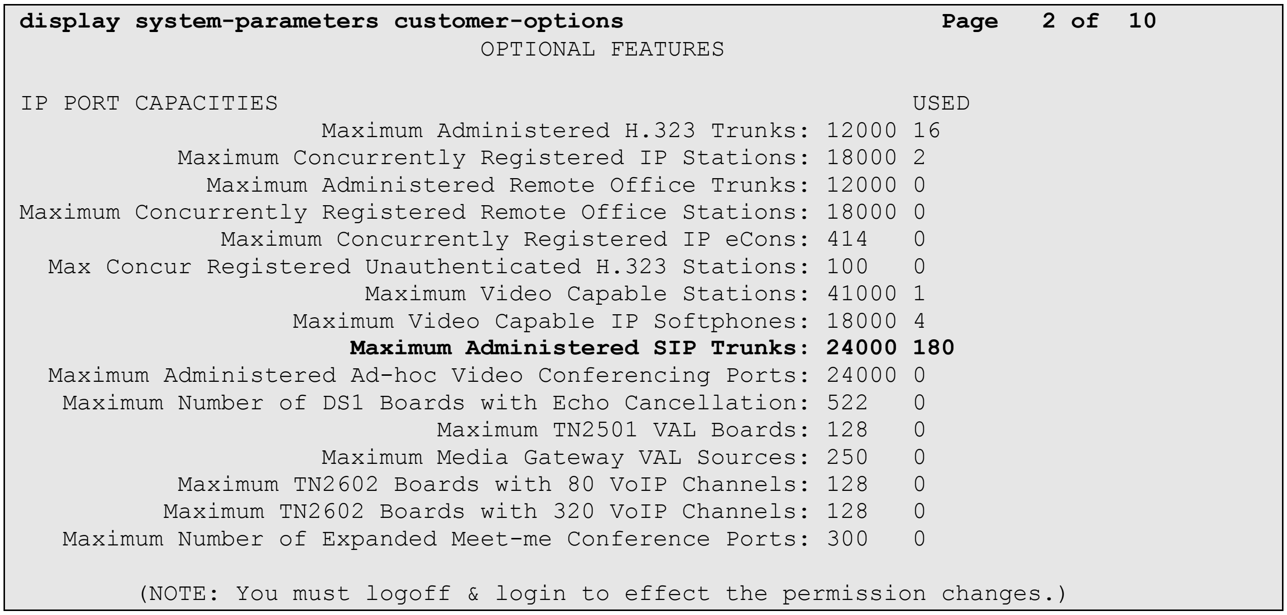
5. Configure Avaya Aura® Communication Manager

This section describes the steps required to connect the CardEasy Core using SIP. It is assumed that Communication Manager is installed and is in fully operational as this is out of the scope of this document. All configuration was administered using Communication Manager System Access Terminal (SAT). The steps documented are as follows.

- Check SIP trunk ports
- Configure Dial Access Code (DAC) in Dial plan
- Add Signaling group
- Add Trunk group

5.1. Check SIP Trunk Capacity

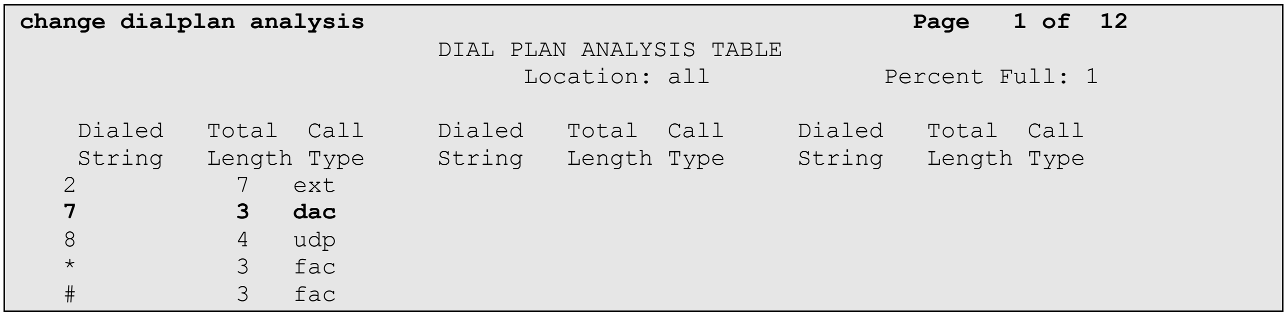
From the SAT use the command **display system-parameters customer-options**. On **Page 2** check that there are sufficient **Administered SIP Trunks** available.



```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 16
           Maximum Concurrently Registered IP Stations: 18000 2
             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 1
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 180
  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   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2. Add Dial Access Code in Dialplan

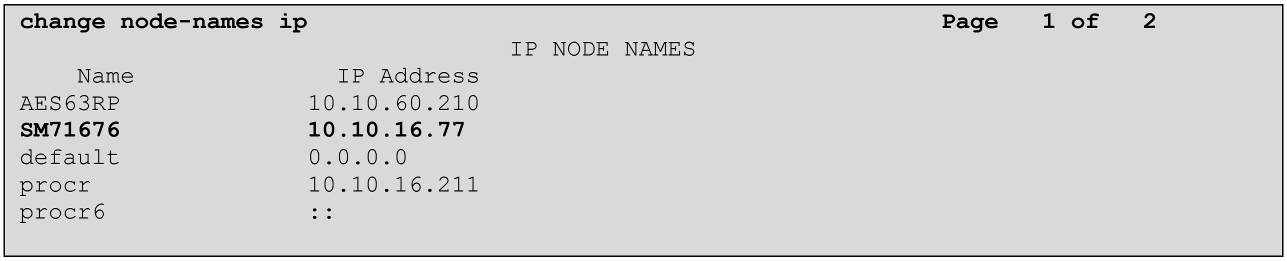
Use the **change dialplan analysis** command and enter under **Dialed String** the leading number of the Dial Access Code (DAC) (**7** in the example), a **Total Length** of **3** and **Call Type dac**



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

    Dialed   Total  Call     Dialed   Total  Call      Dialed   Total  Call
    String   Length Type     String   Length Type      String   Length Type
   2           7   ext
   7           3   dac
   8           4   udp
   *           3   fac
   #           3   fac
```

5.3. Configure Session Manager Node

For Communication Manager to communicate with Session Manager a node must be configured. The screen shot below shows **SM71676** with IP address **10.10.16.77** was used.
Note: 10.10.16.77 IP address of Session Manager SIP Signaling Interface.

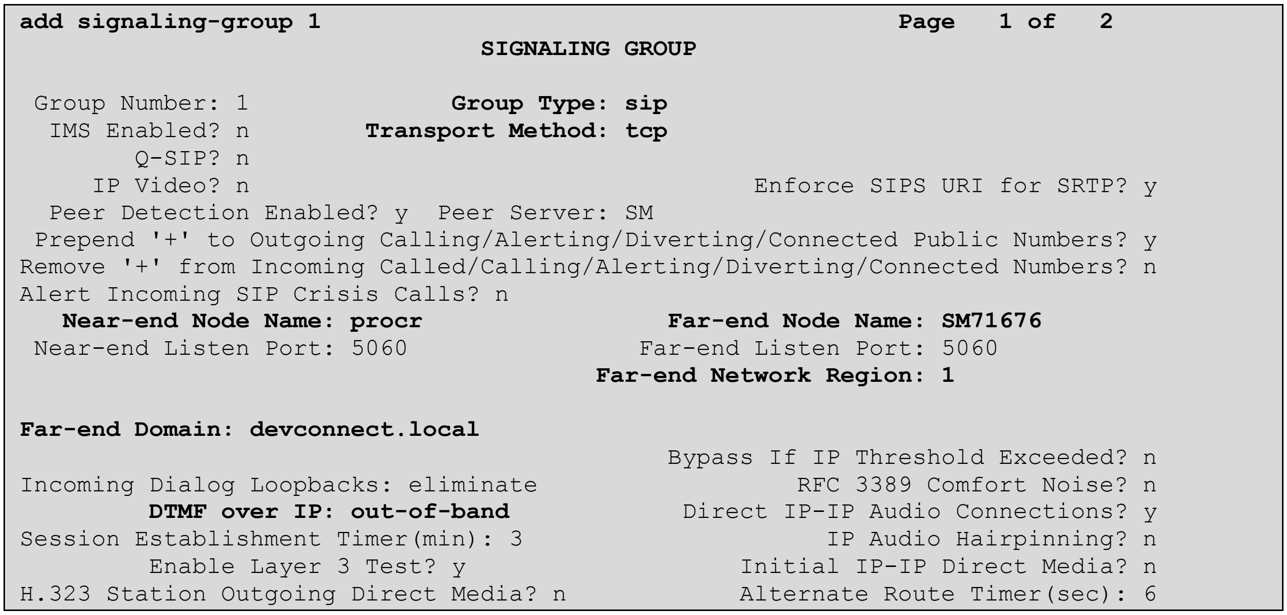


```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AES63RP             10.10.60.210
SM71676             10.10.16.77
default             0.0.0.0
procr               10.10.16.211
procr6              ::
```

5.4. Configure Signaling Group

A signaling group is required before a trunk-group can be configured. Use the **add signaling-group** command followed by next available signaling-group number to configure the following:

- **Group Type:** Enter **sip**
- **Transport Method** Enter **tcp**
- **Near-end Node Name:** Enter **procr**
- **Far-end Node Name:** Enter **SM71676** (Session Manager Node as configured in **Section 5.3**)
- **Far-end Network Region:** Enter the appropriate Network region (i.e. **1**)
- **Far End Domain:** Enter the appropriate Domain
- **DTMF over IP:** Enter **out-of-band** to send as SIP INFO

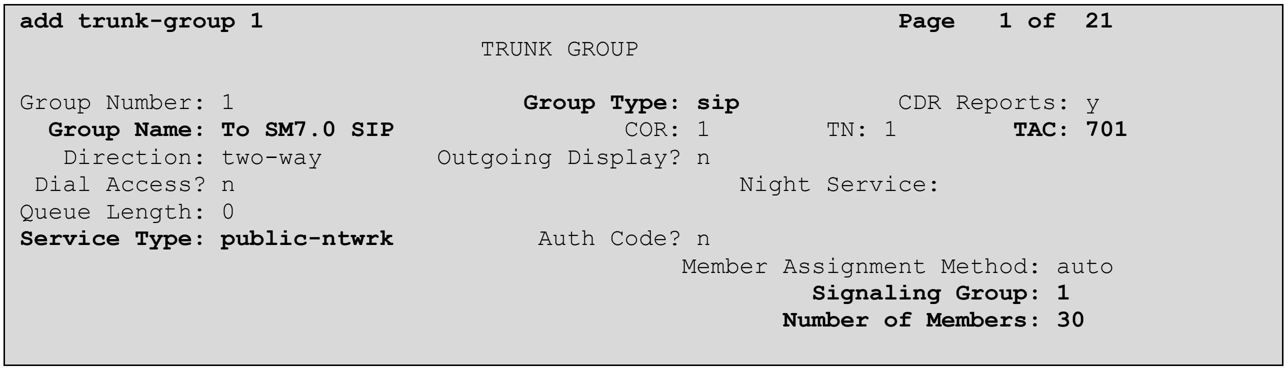


```
add signaling-group 1                                      Page   1 of   2
                                SIGNALING GROUP

 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: 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: SM71676
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain: devconnect.local
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: out-of-band             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
```

5.5. Configure Trunk Group

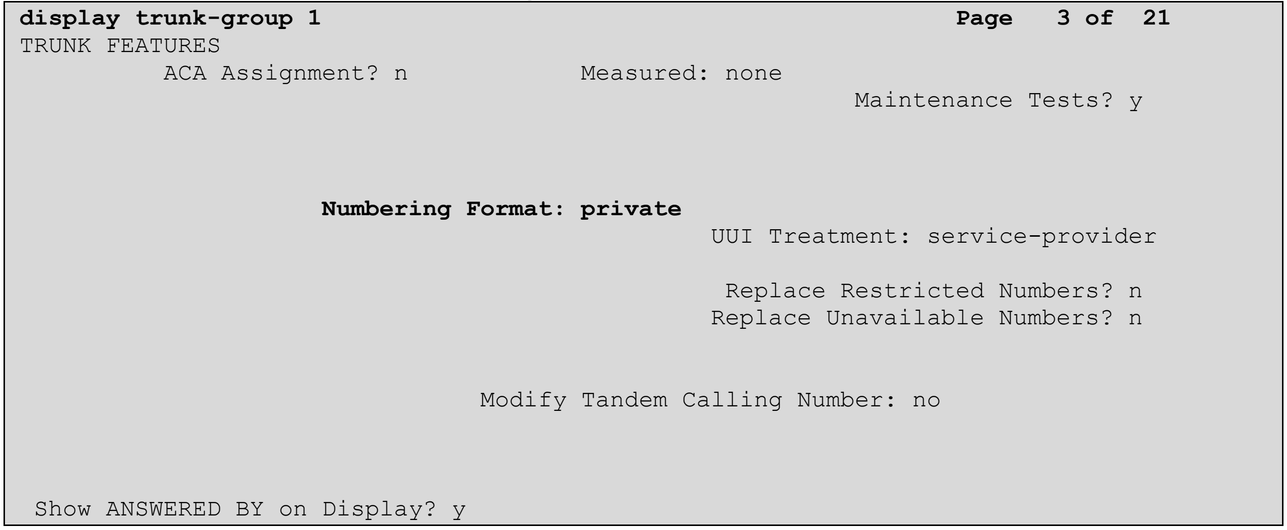
This section describes the Trunk Group configuration used during compliance testing. Use the **add trunk-group** command followed by next available trunk-group number and configure the following:

- **Group Type:** Enter **sip**
- **Group Name:** Enter an informative name for the trunk (i.e. **To SM7.0 SIP)**
- **TAC** Enter a TAC number (i.e. **701)**
- **Service Type:** Enter **public-ntwrk**
- **Signaling Group:** Enter the Signaling Group number as configured in **Section 5.4**
- **Number of Members:** Enter the number of channels required to connect to Session Manger (during compliance testing, 30 channels were used)



```
add trunk-group 1                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: To SM7.0 SIP                COR: 1        TN: 1        TAC: 701
   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: 30
```

On page 3 enter **private** for **Numbering Format**.



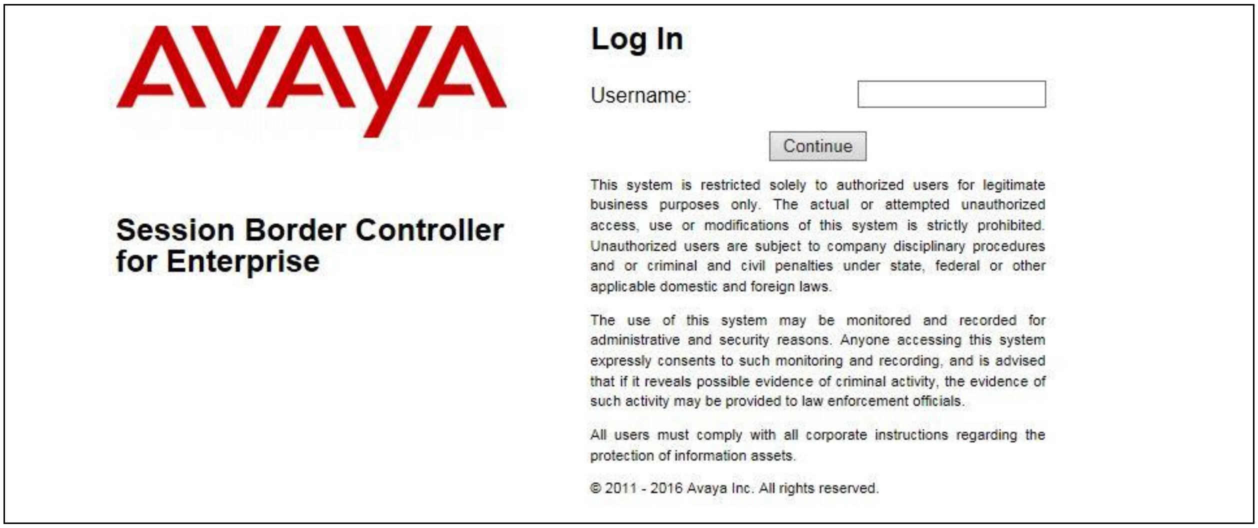
```
display trunk-group 1                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no



 Show ANSWERED BY on Display? y
```

6. 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 signaling to provide an interface to the CardEasy Core SIP Trunk.

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 private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.



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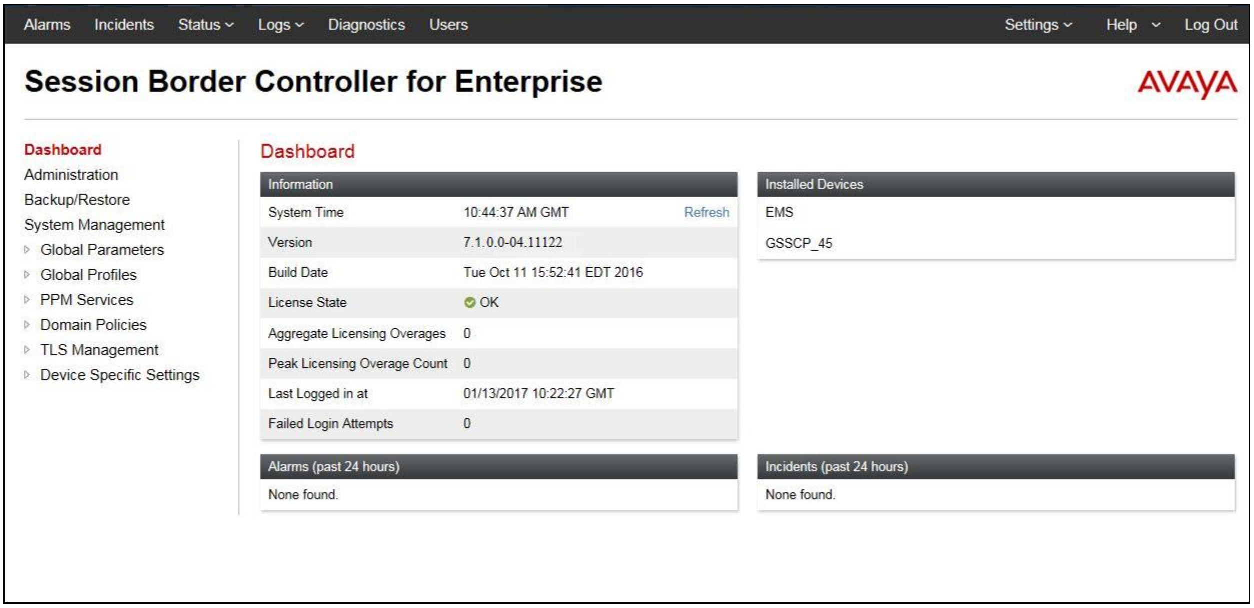
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



Information	
System Time	10:44:37 AM GMT
Version	7.1.0.0-04.11122
Build Date	Tue Oct 11 15:52:41 EDT 2016
License State	✓ OK
Aggregate Licensing Overages	0
Peak Licensing Overage Count	0
Last Logged in at	01/13/2017 10:22:27 GMT
Failed Login Attempts	0

Installed Devices
EMS
GSSCP_45

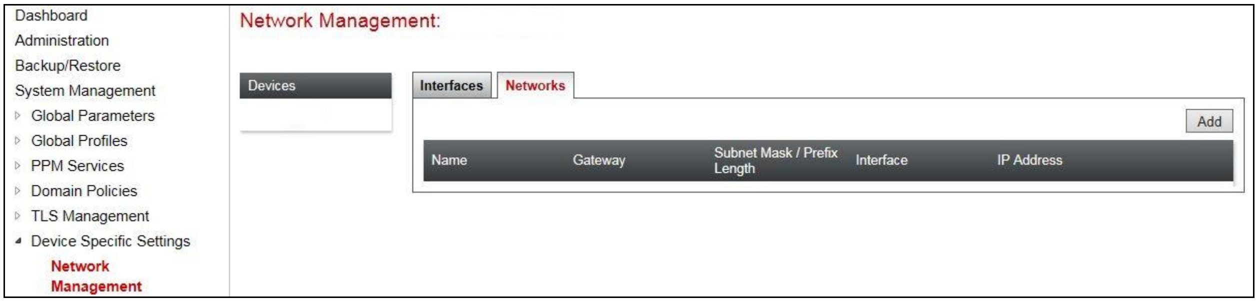
Alarms (past 24 hours)
None found.

Incidents (past 24 hours)
None found.

6.2. Define Network Management

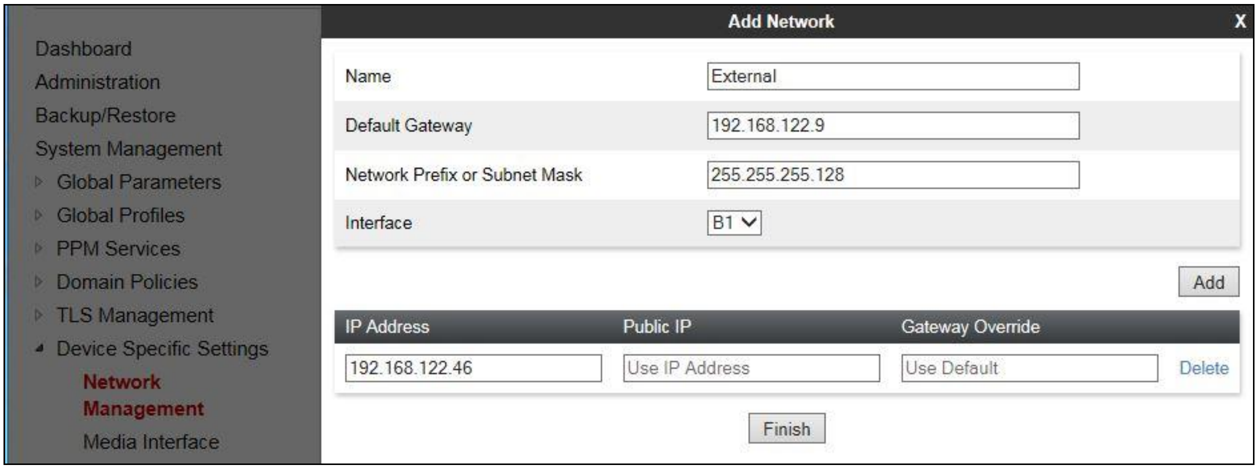
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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



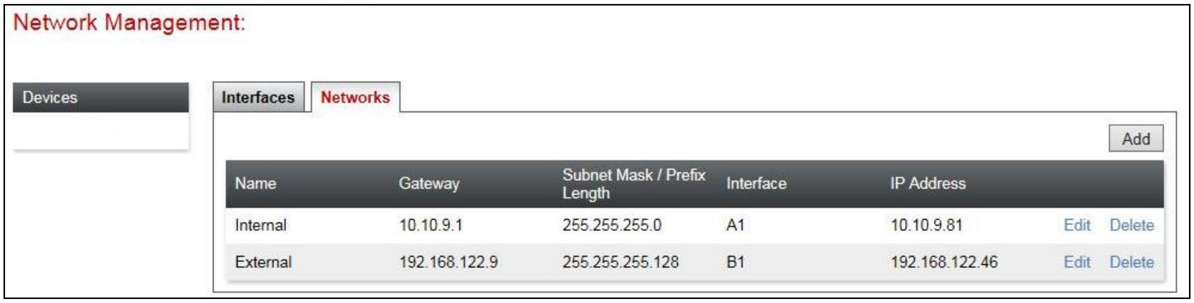
Name	External
Default Gateway	192.168.122.9
Network Prefix or Subnet Mask	255.255.255.128
Interface	B1

IP Address	Public IP	Gateway Override
192.168.122.46	Use IP Address	Use Default

Perform the same task to define the external interface. From **Device Specific Settings →** **Network Management** in the main menu on the left hand side and click on **Add**.
Enter details in the dialogue box (not shown):

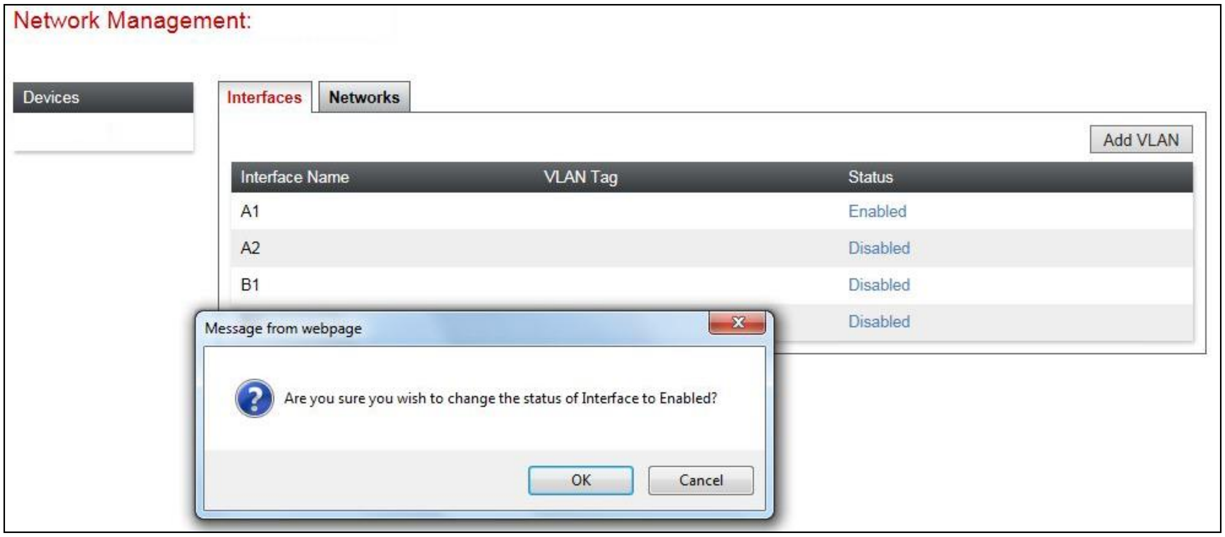
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the 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:



Name	Gateway	Subnet Mask / Prefix Length	Interface	IP Address		
Internal	10.10.9.1	255.255.255.0	A1	10.10.9.81	Edit	Delete
External	192.168.122.9	255.255.255.128	B1	192.168.122.46	Edit	Delete

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



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

Are you sure you wish to change the status of Interface to Enabled?

Note: to ensure that the Avaya SBCE uses the interfaces defined, the Avaya SBCE application must be restarted. Click on **System Management** in the main menu (not shown) and select **Restart Application** indicated by an icon in the status bar (not shown).

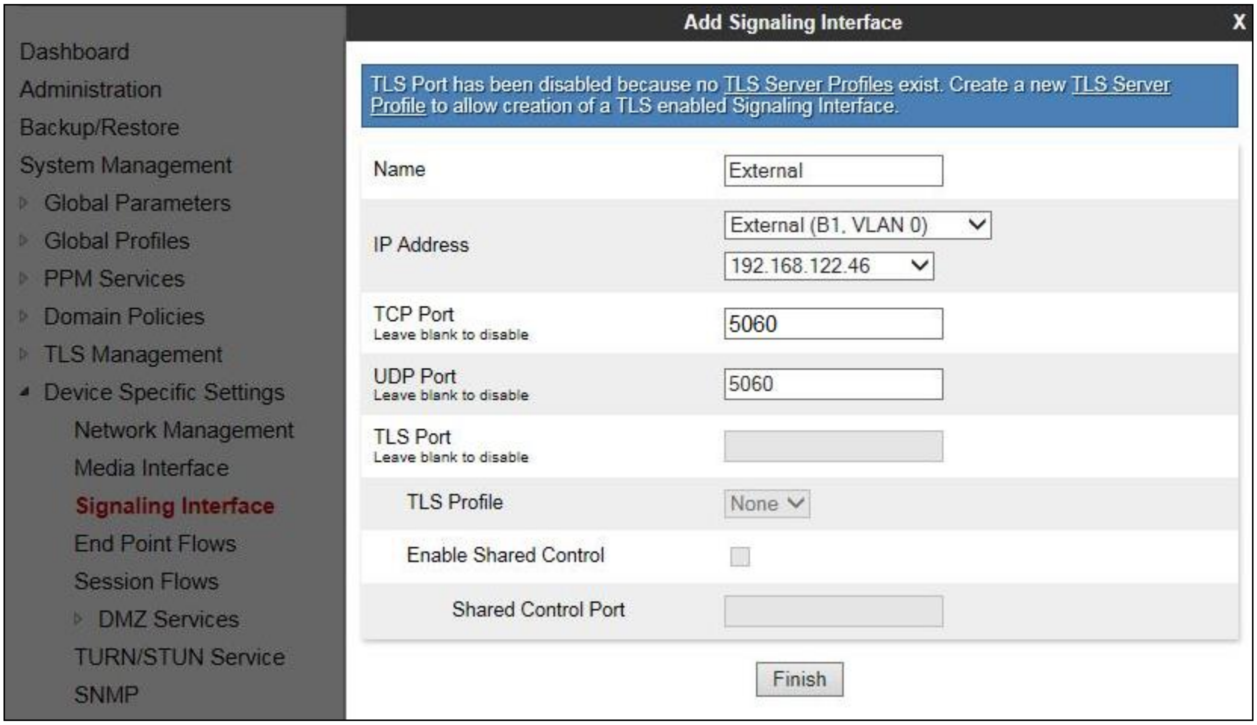
6.3. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signaling and media interfaces. Testing was carried out with TCP used for transport of signaling between Session Manager and the Avaya SBCE, and between the Avaya SBCE and the Cardeasy Core. A signaling and media interface was required on both the internal and external sides of the Avaya SBCE. This document shows the configuration for TCP and UDP, if additional security is required, it's recommended to use TLS and port 5061.

6.3.1. Signaling Interfaces

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

- Select **Add** (not shown) and enter details of the external signaling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external signaling 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 6.2**. In the test environment, this was IP address **192.168.122.46** for the Avaya SBCE interface on the SIP Trunk.
- Enter the TCP port number in the **TCP Port** field, **5060** is used for the CardEasy Core.
- Click on **Finish**



Add Signaling Interface

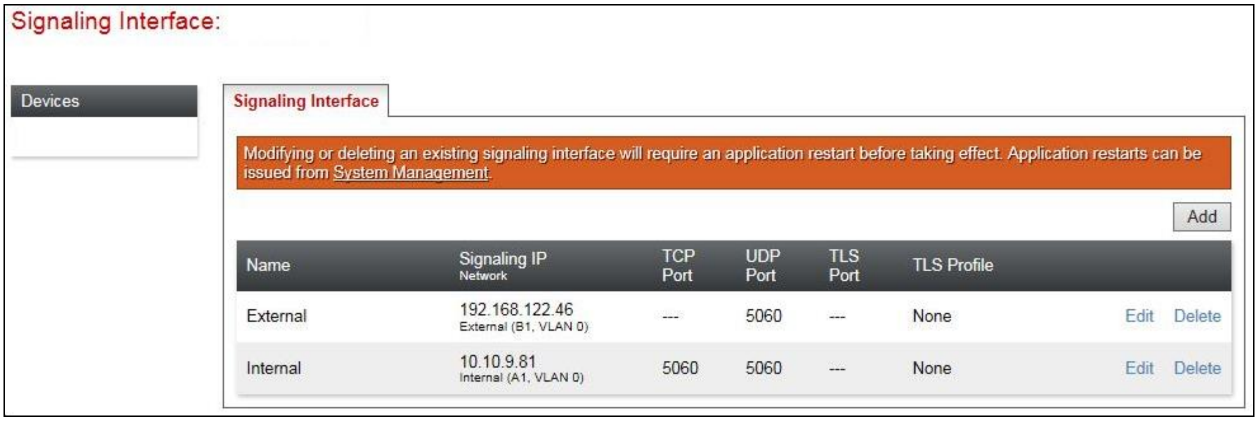
TLS Port has been disabled because no TLS Server Profiles exist. Create a new TLS Server Profile to allow creation of a TLS enabled Signaling Interface.

Name	External
IP Address	External (B1, VLAN 0) 192.168.122.46
TCP Port Leave blank to disable	5060
UDP Port Leave blank to disable	5060
TLS Port Leave blank to disable	
TLS Profile	None
Enable Shared Control	□
Shared Control Port	

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

- Select **Add** and enter details of the internal signaling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal signaling 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 signaling interfaces:



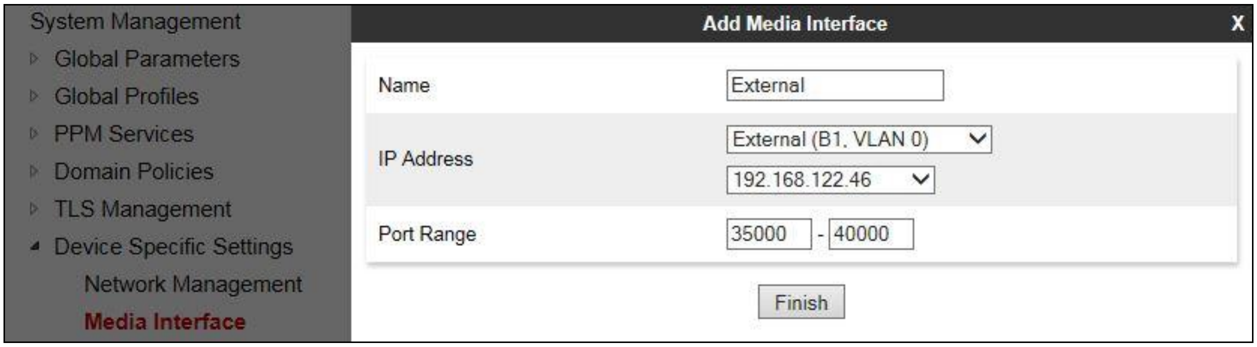
Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile		
External	192.168.122.46 External (B1, VLAN 0)	---	5060	---	None	Edit	Delete
Internal	10.10.9.81 Internal (A1, VLAN 0)	5060	5060	---	None	Edit	Delete

6.3.2. 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 signaling.

- Select **Add** (not shown) 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 6.2**. In the test environment, this was IP address **192.168.122.46**.
- Define the RTP **Port Range** for the media path with the CardEasy Core, during testing this was left at default values of **35000 - 40000**.

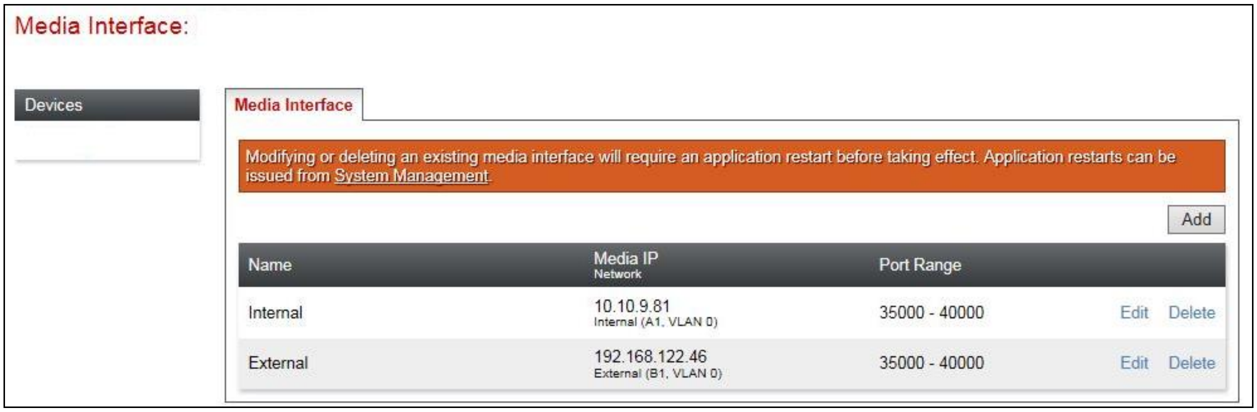


Name	External
IP Address	External (B1, VLAN 0) 192.168.122.46
Port Range	35000 - 40000

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:



Media Interface:

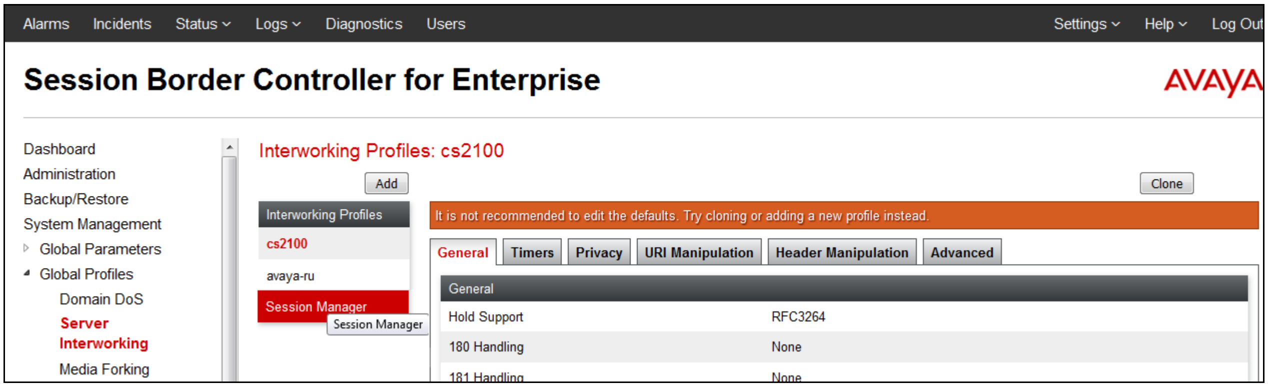
Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Name	Media IP Network	Port Range		
Internal	10.10.9.81 Internal (A1, VLAN 0)	35000 - 40000	Edit	Delete
External	192.168.122.46 External (B1, VLAN 0)	35000 - 40000	Edit	Delete

6.4. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, the CardEasy Core is connected as the Trunk Server and Session Manager is connected as the Call Server.

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 CardEasy Core, click on **Add**.



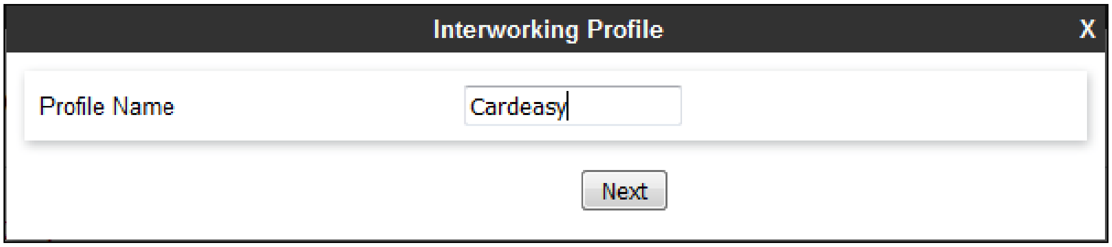
Session Border Controller for Enterprise

Interworking Profiles: cs2100

It is not recommended to edit the defaults. Try cloning or adding a new profile instead.

General	
Hold Support	RFC3264
180 Handling	None
181 Handling	None

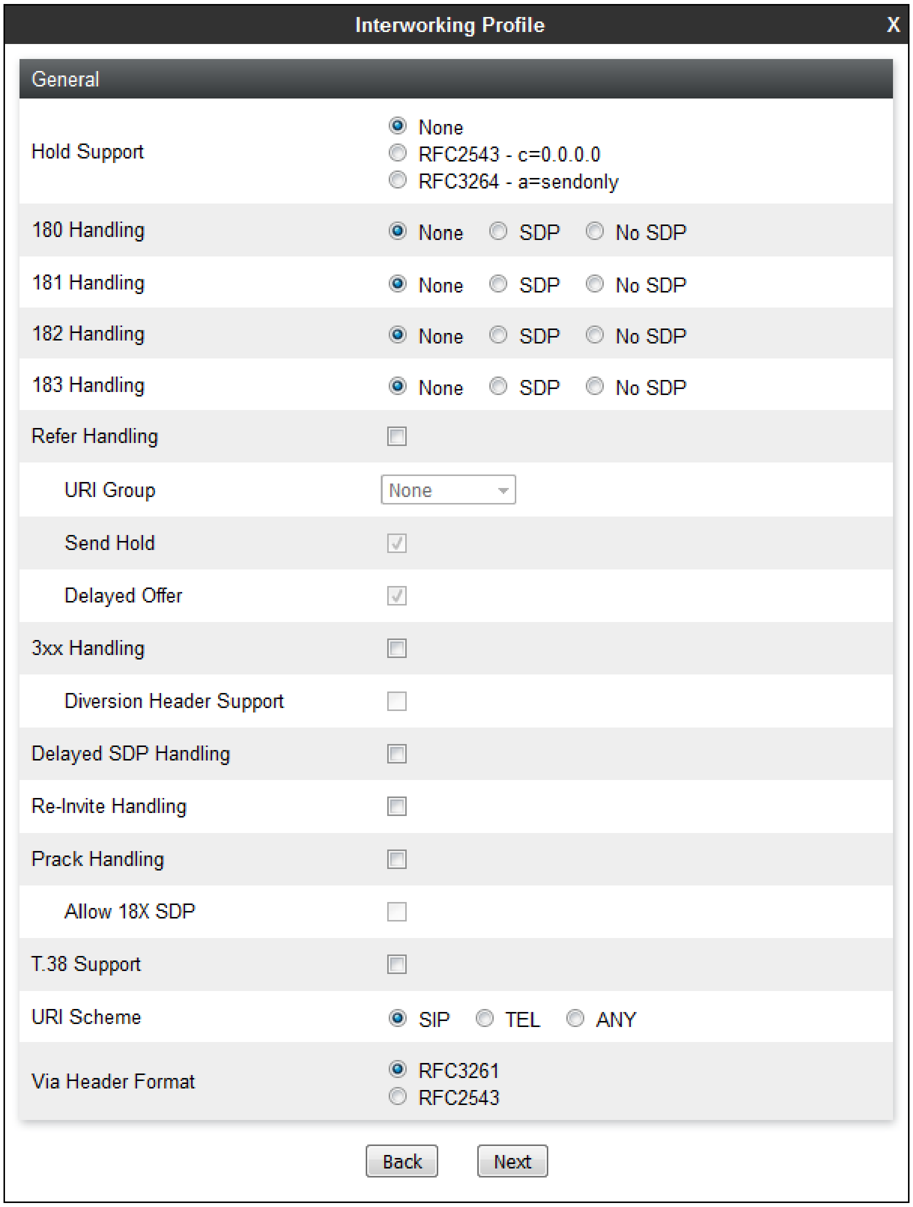
A pop-up menu is generated. In the **Profile Name** field enter a descriptive name for the CardEasy network and click **Next**.



Interworking Profile

Profile Name	Cardeasy

The general settings are default for Interworking Profile.

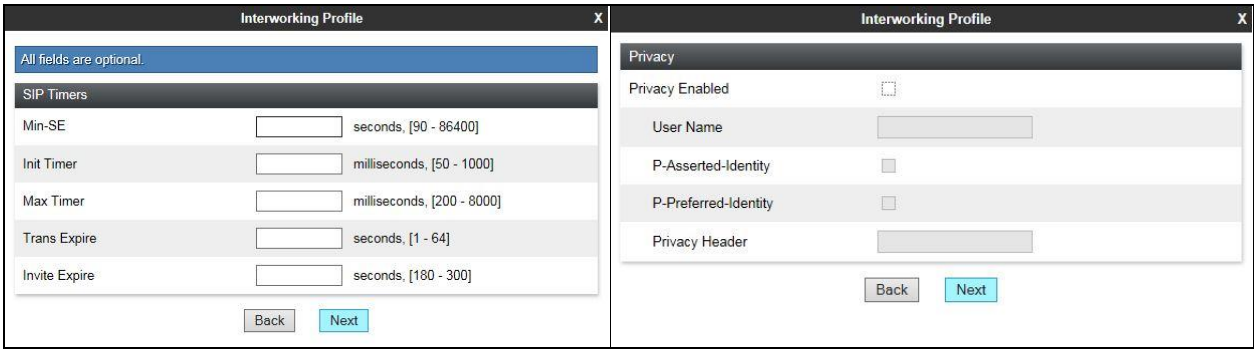


Interworking Profile

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

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

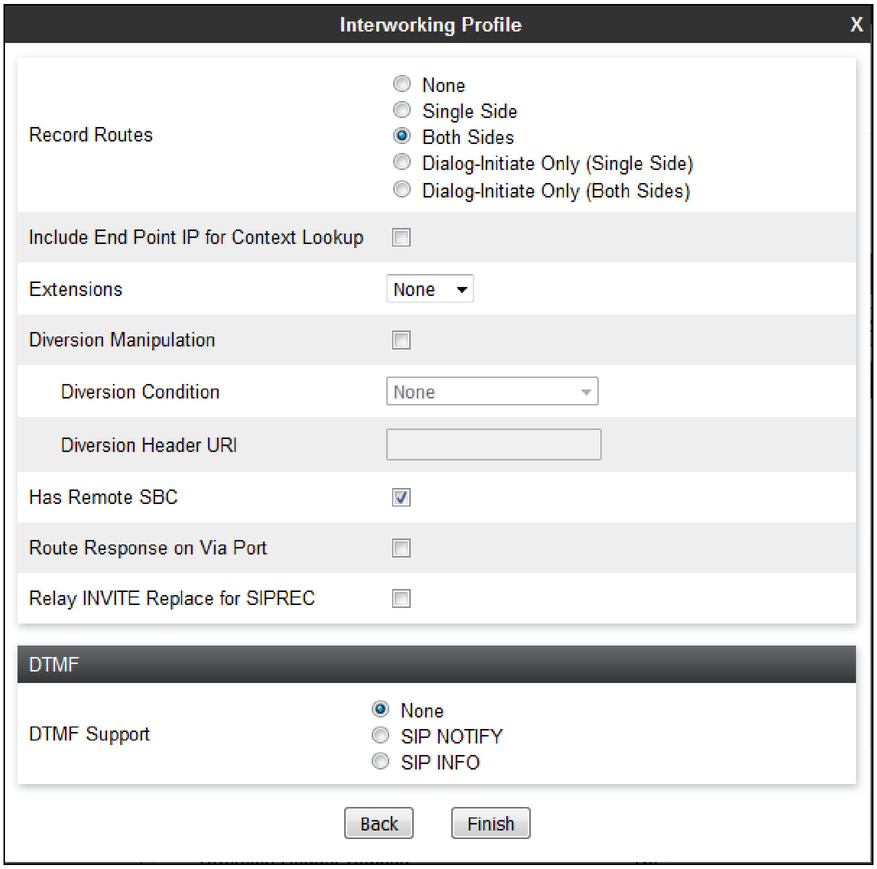
Min-SE		seconds, [90 - 86400]
Init Timer		milliseconds, [50 - 1000]
Max Timer		milliseconds, [200 - 8000]
Trans Expire		seconds, [1 - 64]
Invite Expire		seconds, [180 - 300]

Interworking Profile

Privacy

Privacy Enabled	□
User Name	
P-Asserted-Identity	□
P-Preferred-Identity	□
Privacy Header	

In the final dialogue box, leave the **Record Routes** at the default setting of **Both Sides** and ensure that the **Has Remote SBC** box is checked. Note that Avaya extensions are not supported for the SIP Trunk. Click on **Finish**



Interworking Profile

Record Routes	○ None ○ Single Side ● Both Sides ○ Dialog-Initiate Only (Single Side) ○ Dialog-Initiate Only (Both Sides)
Include End Point IP for Context Lookup	□
Extensions	None
Diversion Manipulation	□
Diversion Condition	None
Diversion Header URI	
Has Remote SBC	☑
Route Response on Via Port	□
Relay INVITE Replace for SIPREC	□

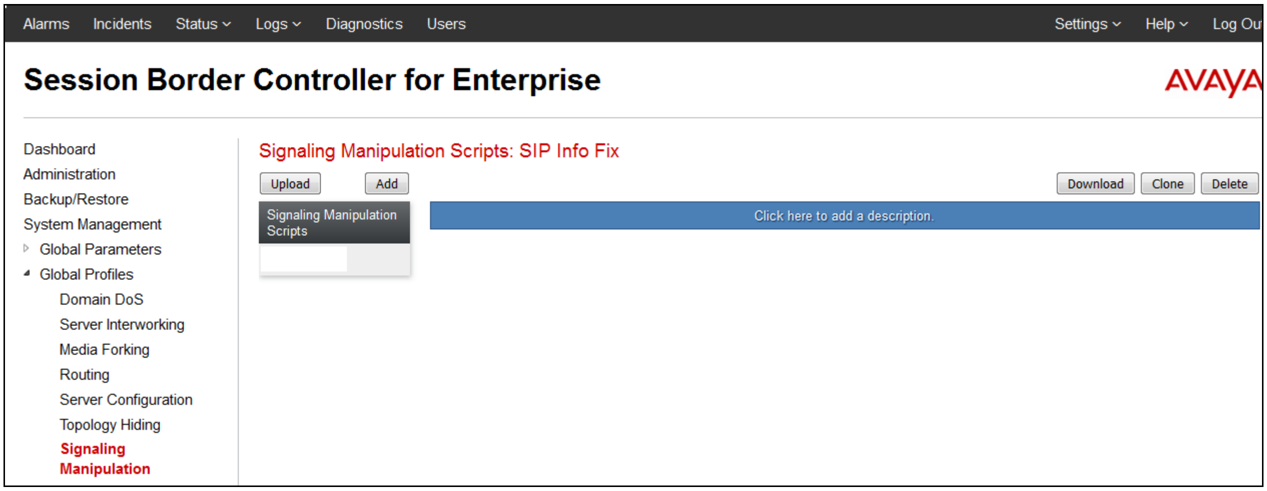
DTMF

DTMF Support	● None ○ SIP NOTIFY ○ SIP INFO

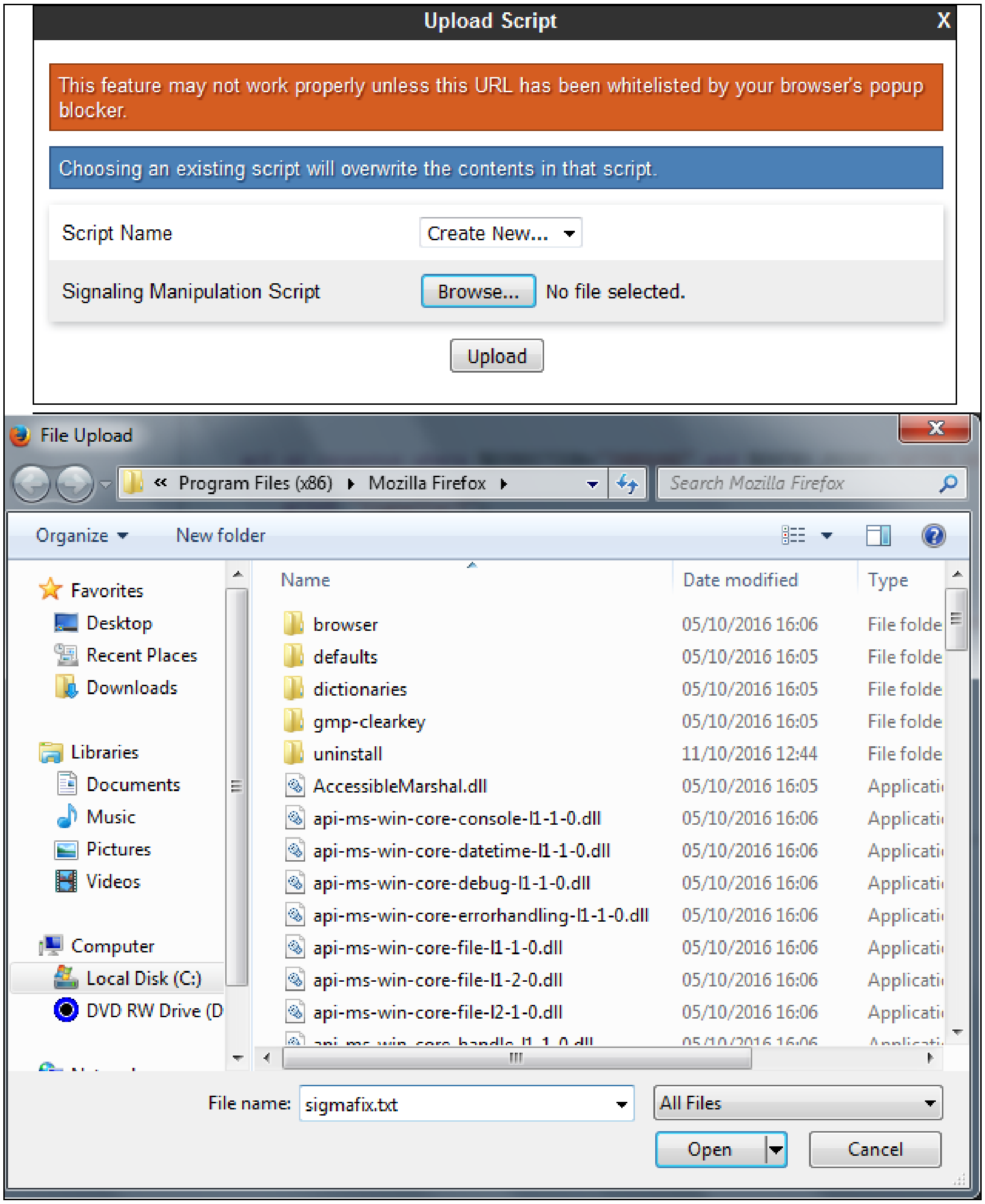
Repeat the process to define **Server Interworking** for Session Manager using the same parameter settings.

6.5. Upload Signaling Manipulation Script

A signaling manipulation script is required to allow SIP info to be sent out from the Avaya SBCE. The script must be obtained from Syntec or Devconnect. Save the script in a location that can be accessed from the browser running the Avaya SBCE web admin pages. From the Left hand menu select **Global Profiles → Signaling Manipulation**. Click on **Upload**.



From the resultant window click on **Browse** and then navigate to the scripts location. Click on **Open** to upload the script.



Upload Script

This feature may not work properly unless this URL has been whitelisted by your browser's popup blocker.

Choosing an existing script will overwrite the contents in that script.

Script Name	Create New...
Signaling Manipulation Script	Browse... No file selected.

File Upload

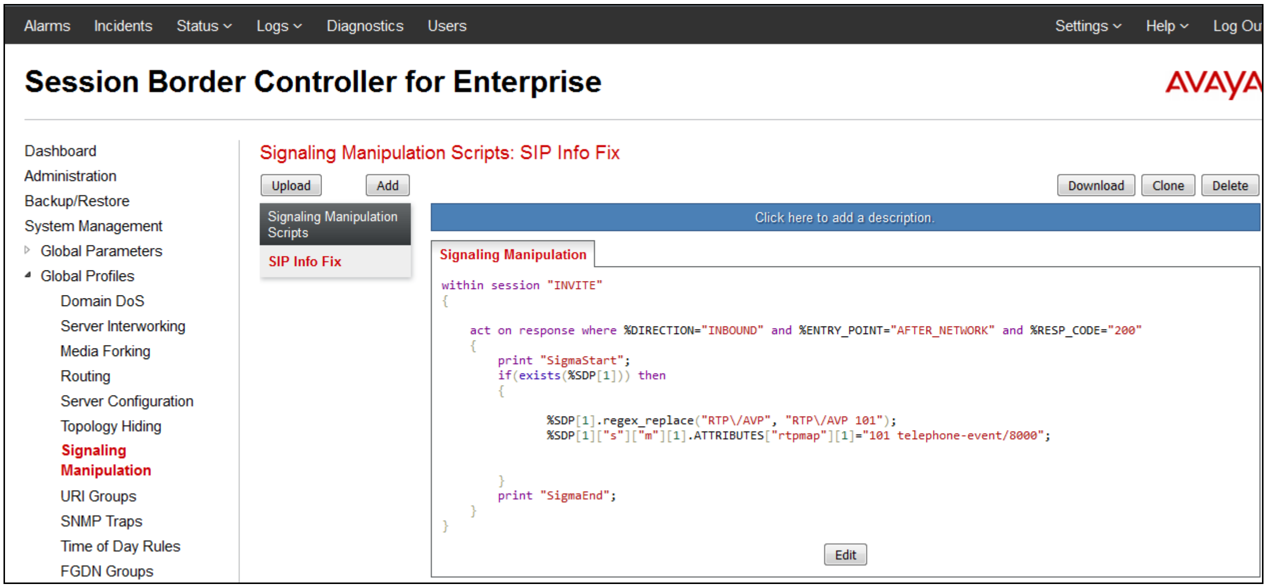
« Program Files (x86) ▸ Mozilla Firefox ▸

File name: sigmafix.txt

All Files

Name	Date modified	Type
browser	05/10/2016 16:06	File folde
defaults	05/10/2016 16:05	File folde
dictionaries	05/10/2016 16:05	File folde
gmp-clearkey	05/10/2016 16:05	File folde
uninstall	11/10/2016 12:44	File folde
AccessibleMarshal.dll	05/10/2016 16:05	Applicati
api-ms-win-core-console-l1-1-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-datetime-l1-1-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-debug-l1-1-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-errorhandling-l1-1-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-file-l1-1-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-file-l1-2-0.dll	05/10/2016 16:06	Applicati
api-ms-win-core-file-l2-1-0.dll	05/10/2016 16:06	Applicati

The Screen will go back and the script will be shown.



Signaling Manipulation Scripts: SIP Info Fix

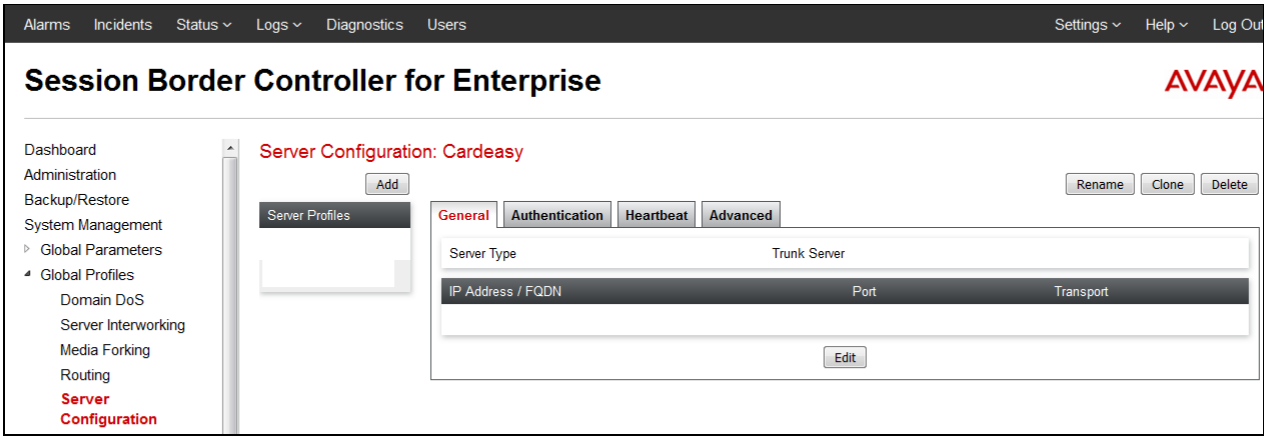
Signaling Manipulation

```
within session "INVITE"
{

    act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK" and %RESP_CODE="200"
    {
        print "SigmaStart";
        if(exists(%SDP[1])) then
        {

                %SDP[1].regex_replace("RTP\/AVP", "RTP\/AVP 101");
                %SDP[1]["s"]["m"][1].ATTRIBUTES["rtpmap"][1]="101 telephone-event/8000";


        }
        print "SigmaEnd";
    }
}
```

6.6. Define Servers

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

To define the CardEasy Core Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add**.

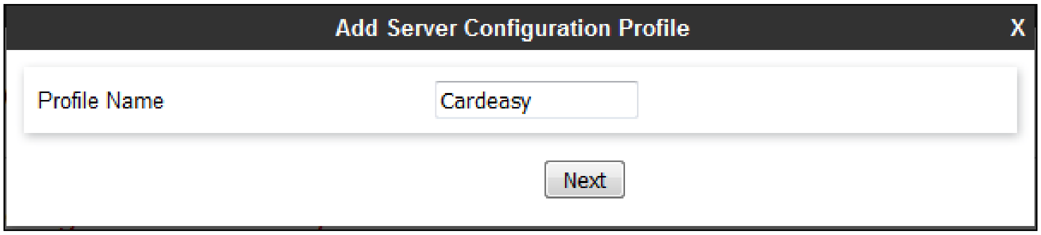


Server Configuration: Cardeasy

Server Type	Trunk Server

IP Address / FQDN	Port	Transport

Enter an appropriate name in the pop-up menu.

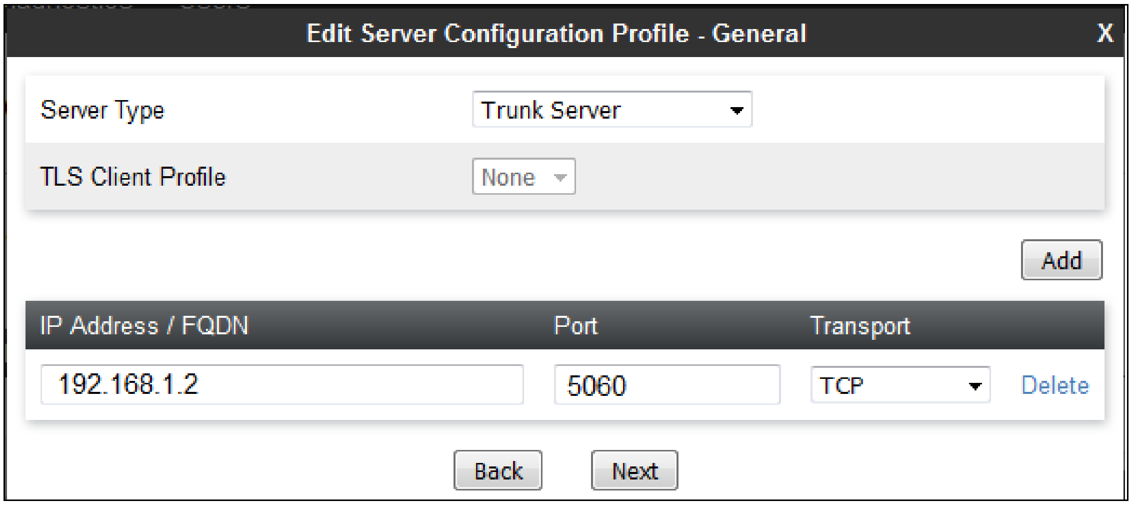


Add Server Configuration Profile

Profile Name	Cardeasy

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 CardEasy Core IP address.
- In the **Port** box, enter the port to be used for the SIP Trunk.
- In the **Transport** drop down menu, select **TCP**. Click on **Next**.



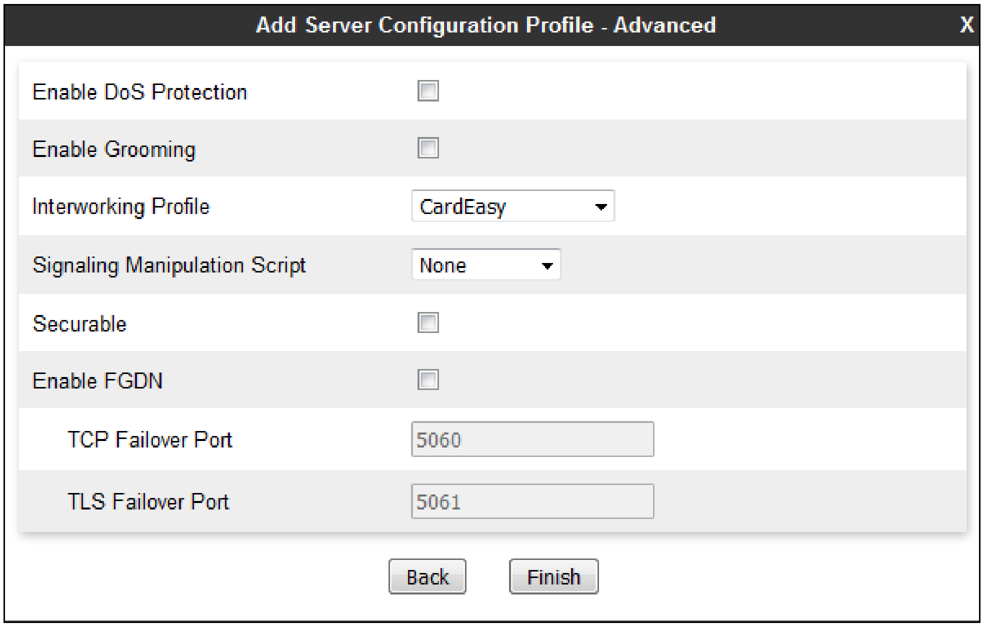
Edit Server Configuration Profile - General

Server Type	Trunk Server
TLS Client Profile	None

IP Address / FQDN	Port	Transport	
192.168.1.2	5060	TCP	Delete

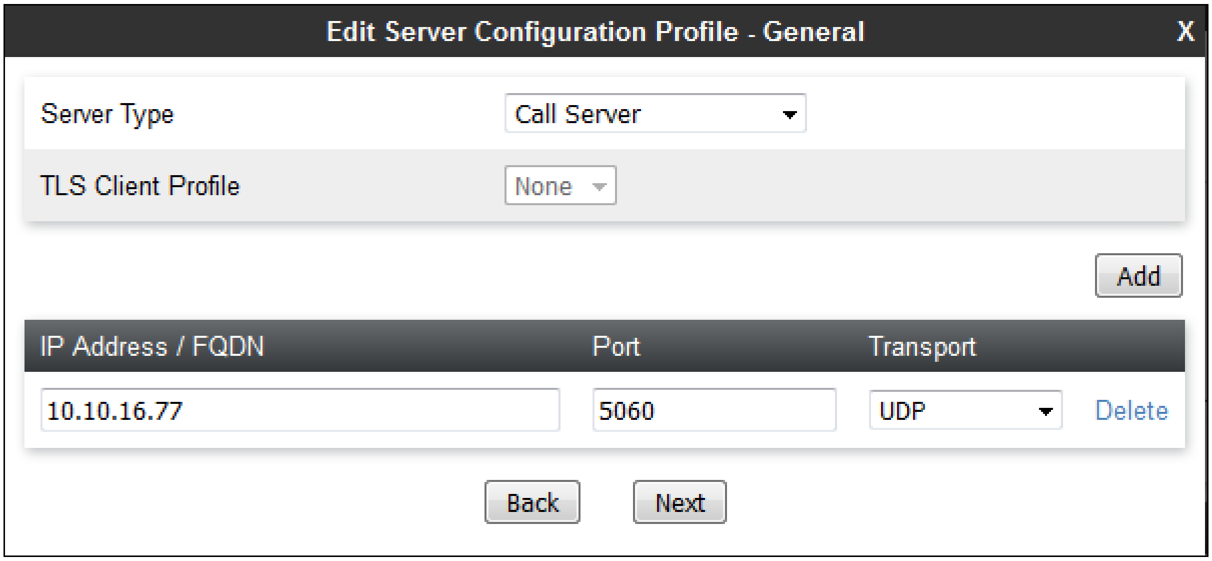
Click on **Next** until the final dialogue box is shown. This contains the **Advanced** settings:

- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for CardEasy Core defined in **Section 6.4**.
- Leave the other fields at default settings.
- Click **Finish**.



Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	CardEasy
Signaling Manipulation Script	None
Securable	□
Enable FGDN	□
TCP Failover Port	5060
TLS Failover Port	5061

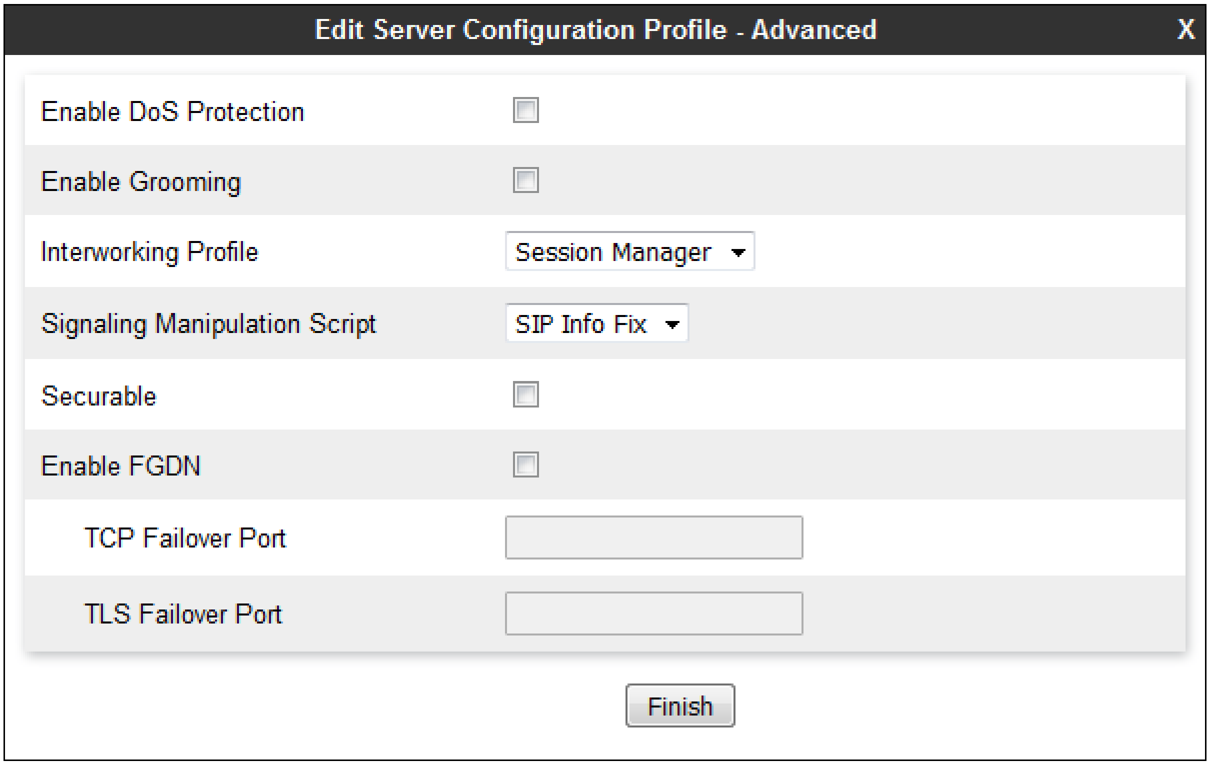
Use the following process described to define the Call Server configuration for Session Manager if not already defined. Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.



Server Type	Call Server
TLS Client Profile	None

IP Address / FQDN	Port	Transport
10.10.16.77	5060	UDP

In the Advanced dialogue box ensure that the Interworking Profile defined for Session Manager in **Section 6.4** is selected in the **Interworking Profile** drop down menu. Set the **Signaling Manipulation Script** to the script added in **Section 6.5**. Click on **Finish.**

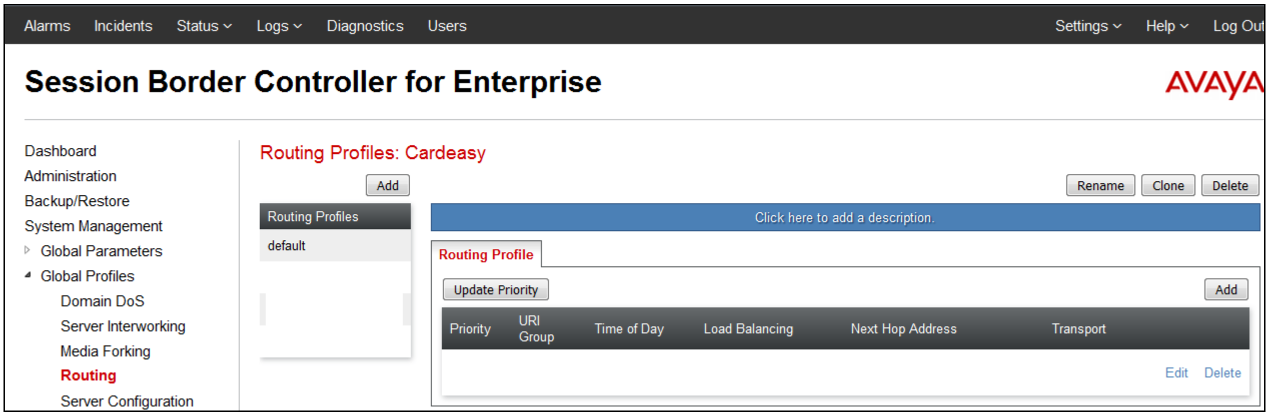


Edit Server Configuration Profile - Advanced

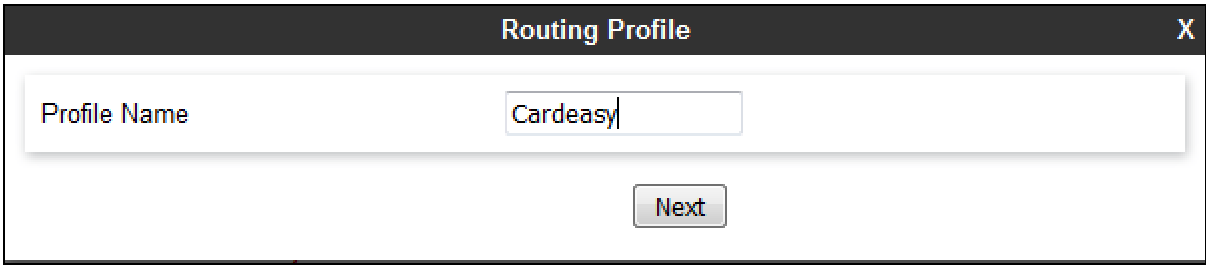
Enable DoS Protection	□
Enable Grooming	□
Interworking Profile	Session Manager
Signaling Manipulation Script	SIP Info Fix
Securable	□
Enable FGDN	□
TCP Failover Port	
TLS Failover Port	

6.7. Define Routing

Routing information is required for routing to the CardEasy Core 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 signaling. To define routing to CardEasy Core, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add**.



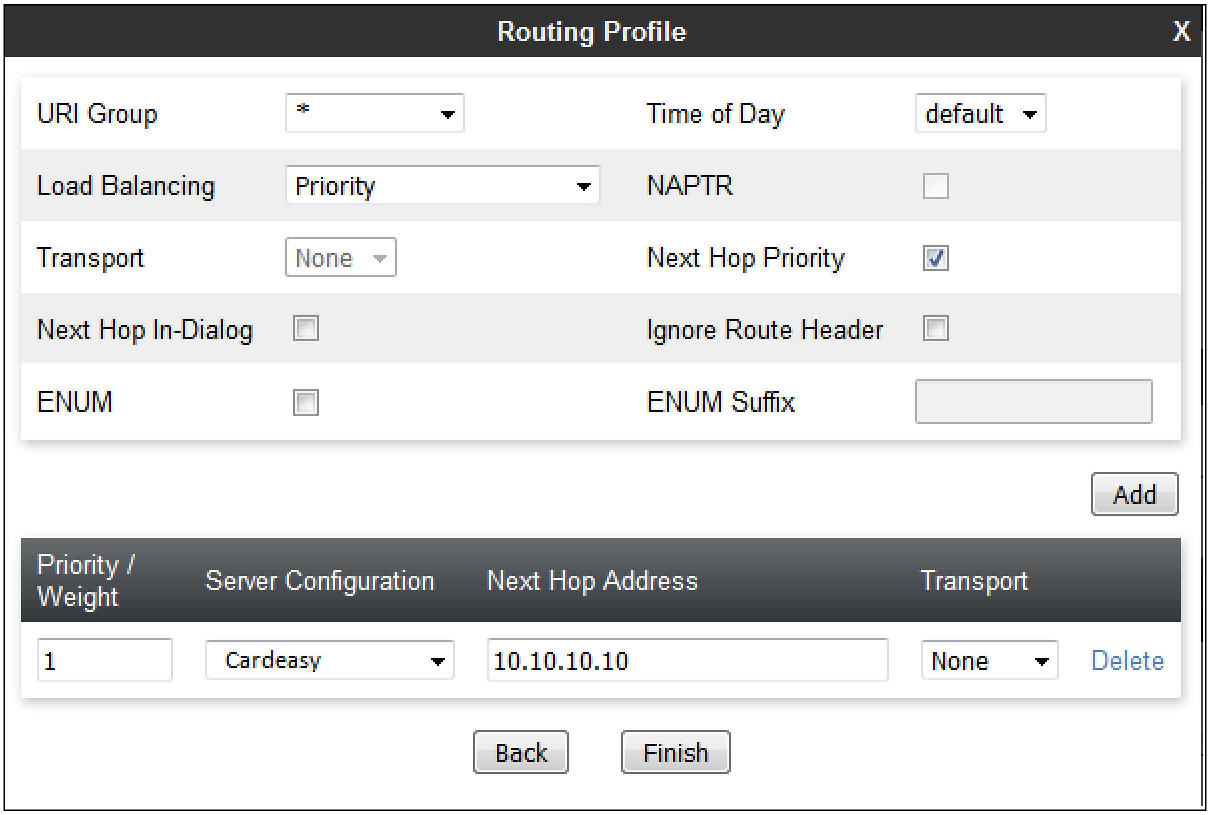
Enter an appropriate name in the dialogue box.



Profile Name: Cardeasy

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 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 6.6** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Click **Finish**.

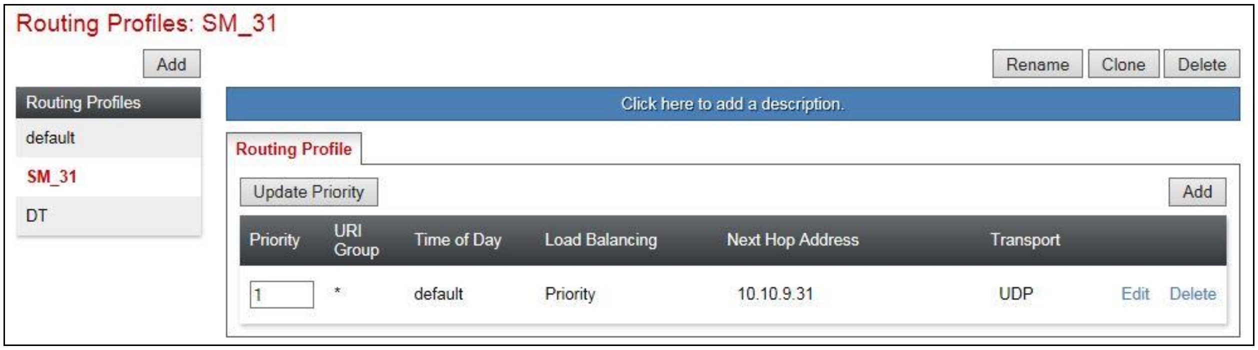


Routing Profile

URI Group	*	Time of Day	default
Load Balancing	Priority	NAPTR	□
Transport	None	Next Hop Priority	☑
Next Hop In-Dialog	□	Ignore Route Header	□
ENUM	□	ENUM Suffix	

Priority / Weight	Server Configuration	Next Hop Address	Transport	
1	Cardeasy	10.10.10.10	None	Delete

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

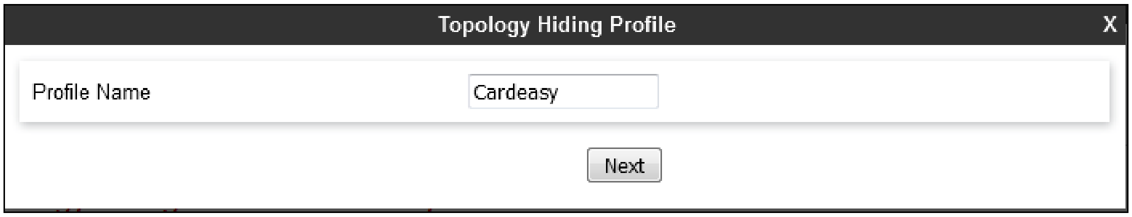


Priority	URI Group	Time of Day	Load Balancing	Next Hop Address	Transport		
1	*	default	Priority	10.10.9.31	UDP	Edit	Delete

6.8. 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 for termination information and the external interfaces for origination information.

To define Topology Hiding for CardEasy, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** (not shown) to bring up a dialogue box, assign an appropriate name and click on **Next** to configure Topology Hiding for each header as required:

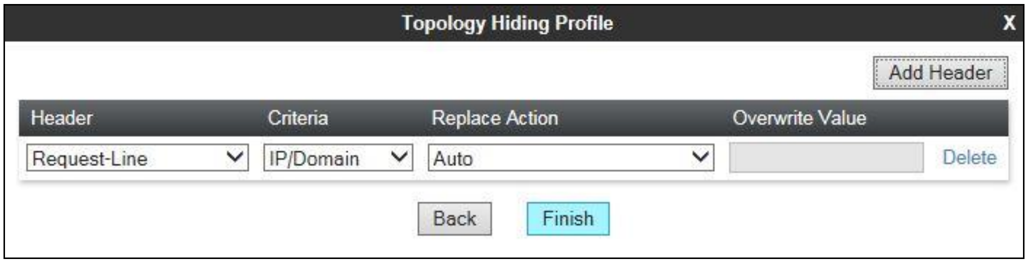


Topology Hiding Profile

Profile Name	Cardeasy

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

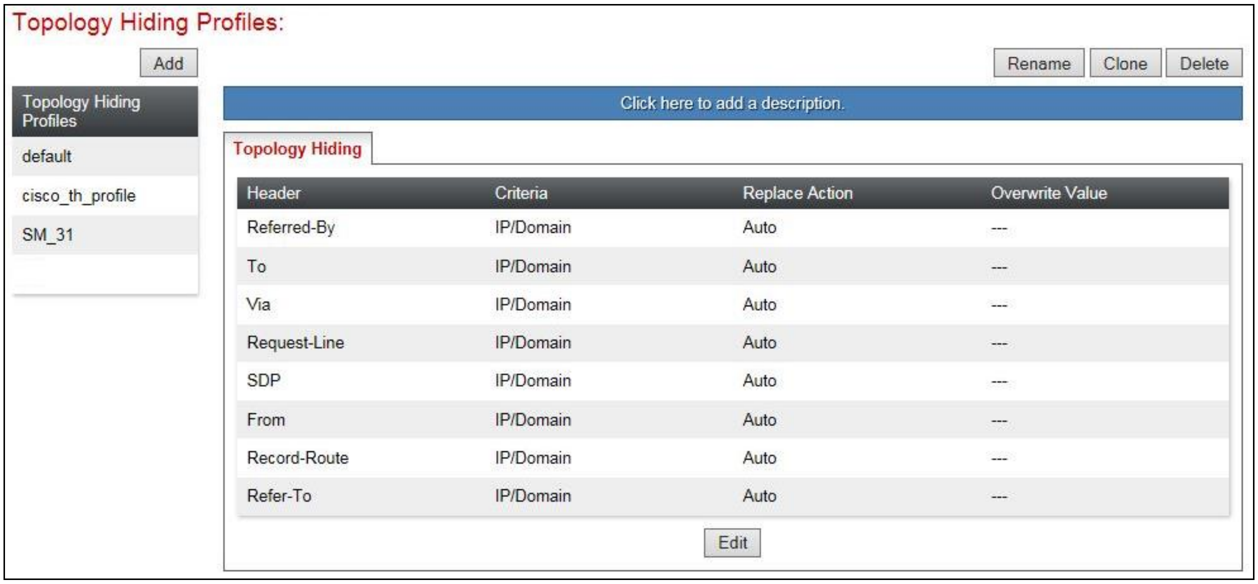
- Click on **Add Header** and **s**elect from the **Header** (not shown) drop down menu.
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu depending on requirements. During testing the default **IP/Domain** was used for all headers that hides both domain names and IP addresses.
- 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.
- Click **Finish**.
- Topology hiding was defined for all headers where the function is available.



Topology Hiding Profile

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

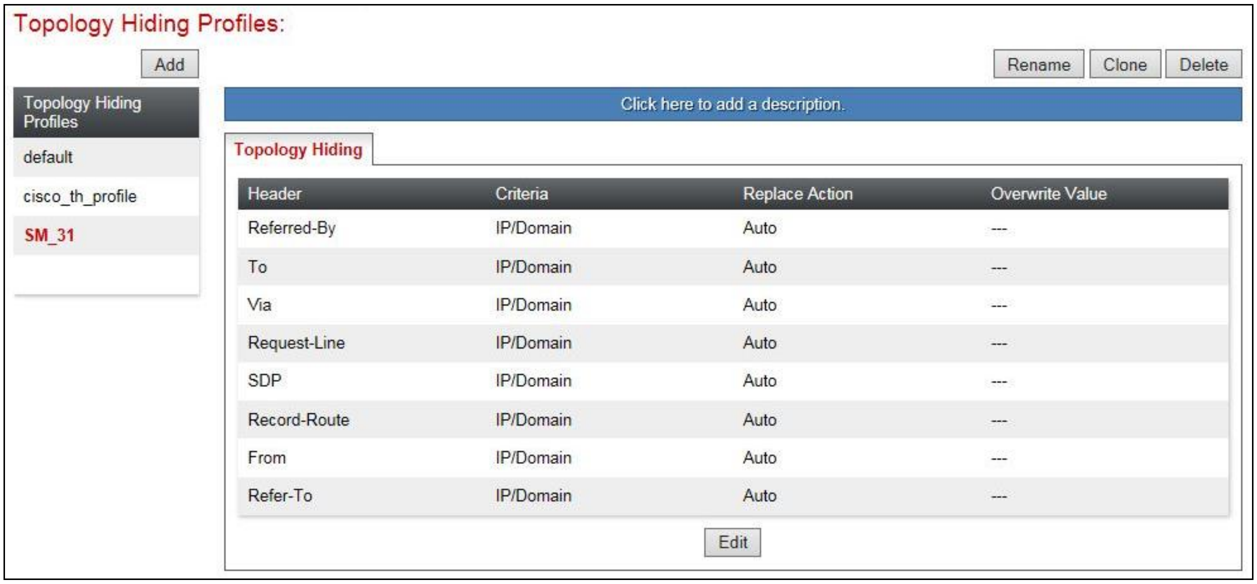
The following screenshot shows the completed **Topology Hiding** configuration for the CardEasy Core.



Topology Hiding Profiles:

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

To define Topology hiding for Session Manager, follow the same process. This can be simplified by cloning the profile defined for CardEasy Core. Do this by highlighting the profile defined for CardEasy 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.



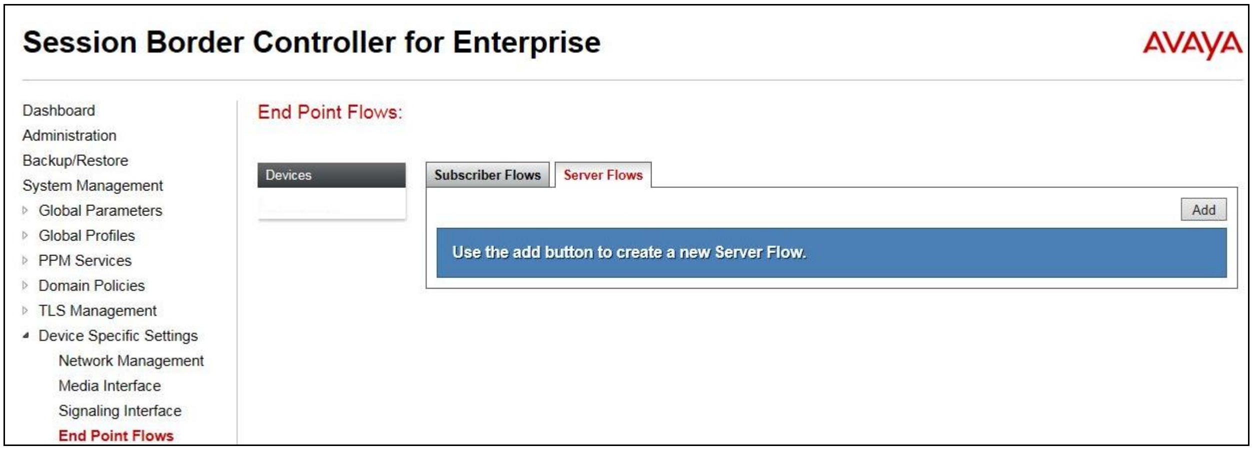
Topology Hiding Profiles:

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

6.9. Server Flows

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

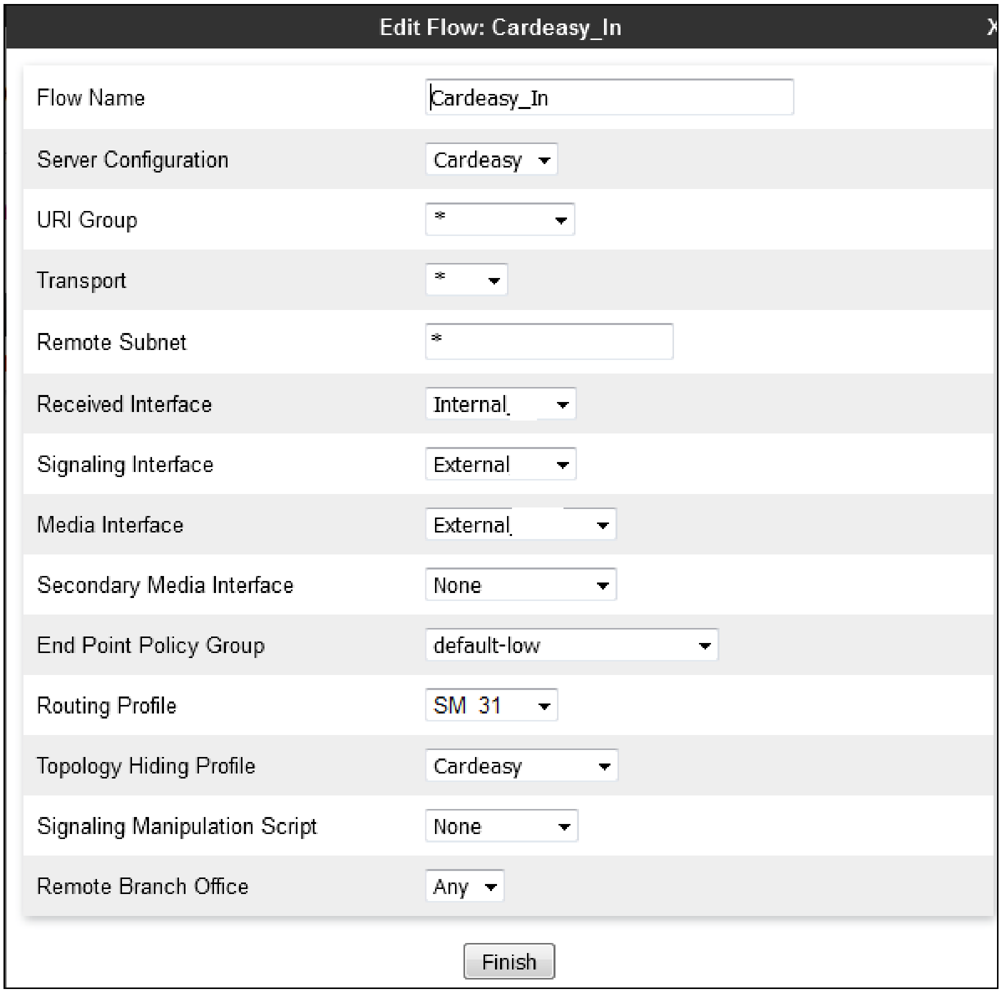
To define a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click on **Add**.



Use the add button to create a new Server Flow.

Define the Server flow for the CardEasy Core as follows:

- In the **Flow Name** field enter a descriptive name for the server flow for the CardEasy Core, in the test environment **CardEasy_In** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for the CardEasy defined in **Section 6.6**.
- In the **Received Interface** drop-down menu, select the internal SIP signaling interface defined in **Section 6.3**. This is the interface that signaling bound for the SIP Trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signaling interface defined in **Section 6.3**. This is the interface that signaling bound for the SIP Trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 6.3**. 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 6.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the CardEasy Core defined in **Section 6.8** and click **Finish**.

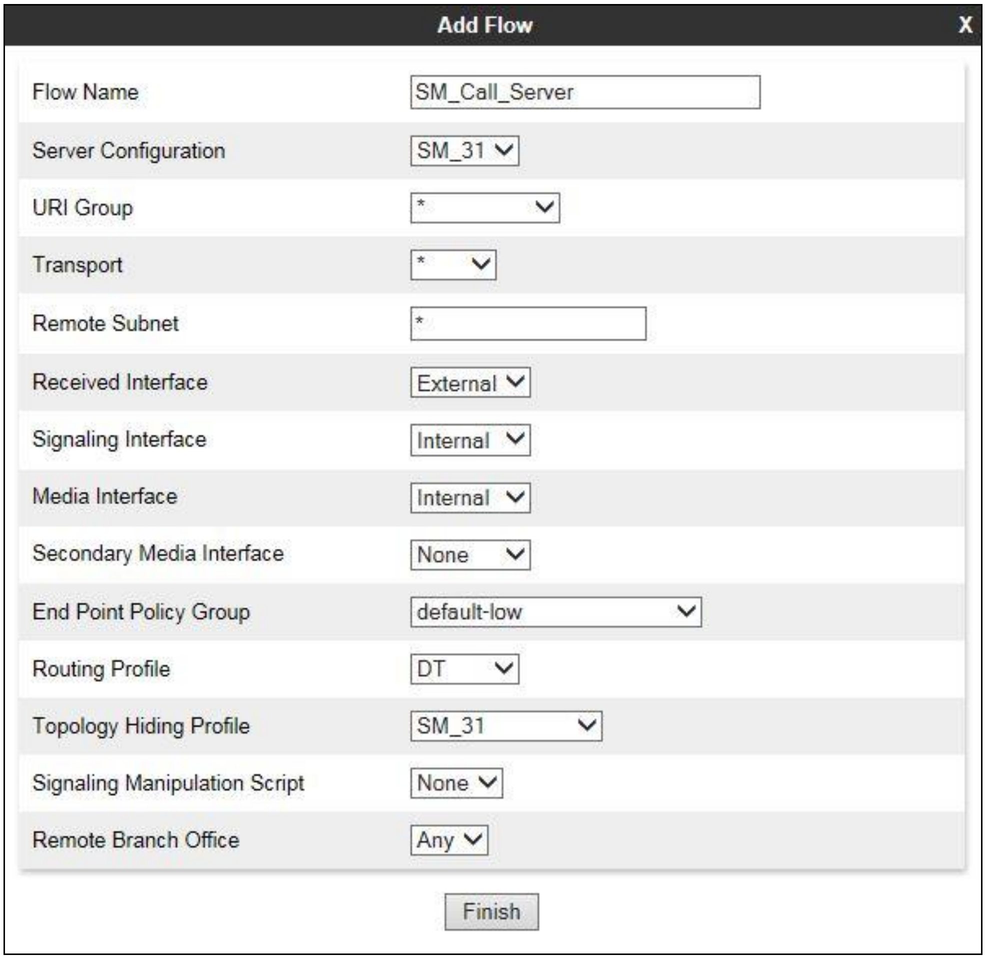


Edit Flow: Cardeasy_In

Flow Name	Cardeasy_In
Server Configuration	Cardeasy
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Internal
Signaling Interface	External
Media Interface	External
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	SM 31
Topology Hiding Profile	Cardeasy
Signaling Manipulation Script	None
Remote Branch Office	Any

Define a Server Flow for Session Manager as follows:

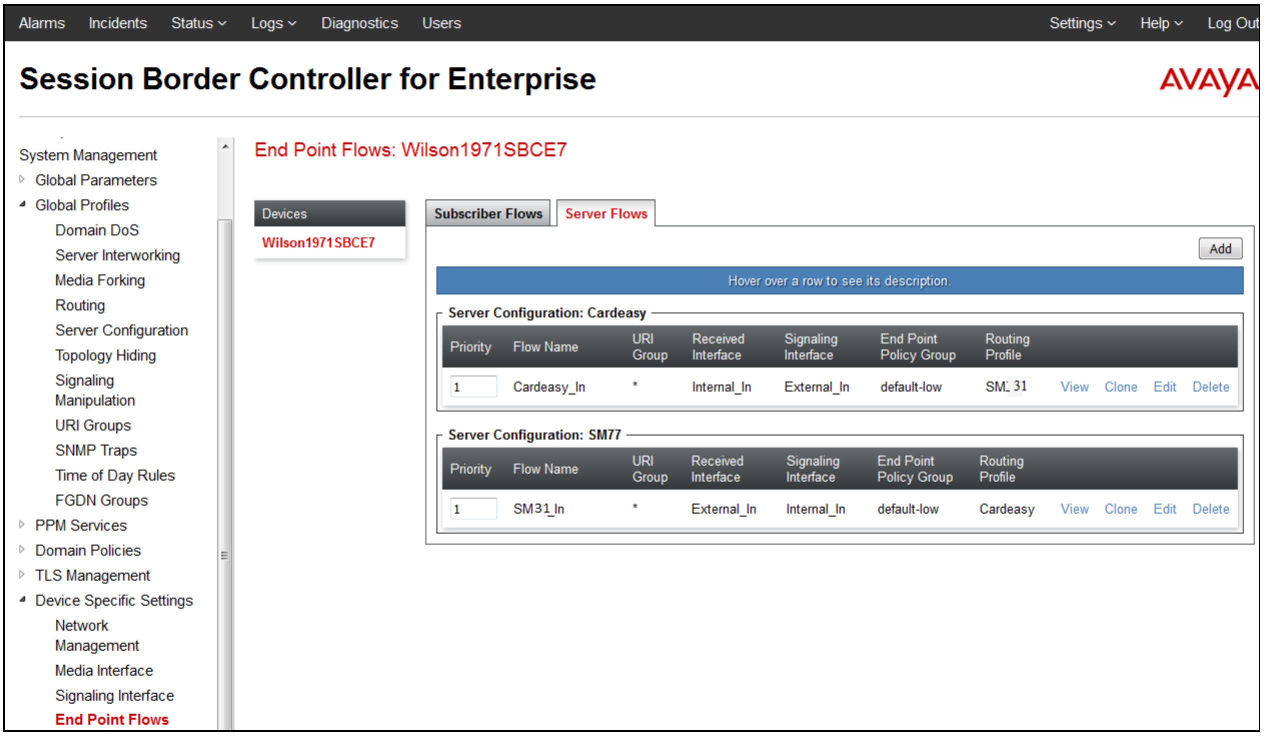
- In the **Flow Name** field enter a descriptive name for the server flow for Session Manager, in the test environment **SM_Call_Server** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for Session Manager defined in **Section 6.6**.
- In the **Received Interface** drop-down menu, select the external SIP signaling interface defined in **Section 6.3**. This is the interface that signaling bound for Session Manager is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signaling interface defined in **Section 6.3**. This is the interface that signaling bound for Session Manager is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 6.3**. 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 CardEasy Core defined in **Section 6.7**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of Session Manager defined in **Section 6.8** and click **Finish**.



Add Flow

Flow Name	SM_Call_Server
Server Configuration	SM_31
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	External
Signaling Interface	Internal
Media Interface	Internal
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	DT
Topology Hiding Profile	SM_31
Signaling Manipulation Script	None
Remote Branch Office	Any

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



End Point Flows: Wilson1971SBCE7

Server Configuration: Cardeasy

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Cardeasy_In	*	Internal_In	External_In	default-low	SM31	View	Clone	Edit	Delete

Server Configuration: SM77

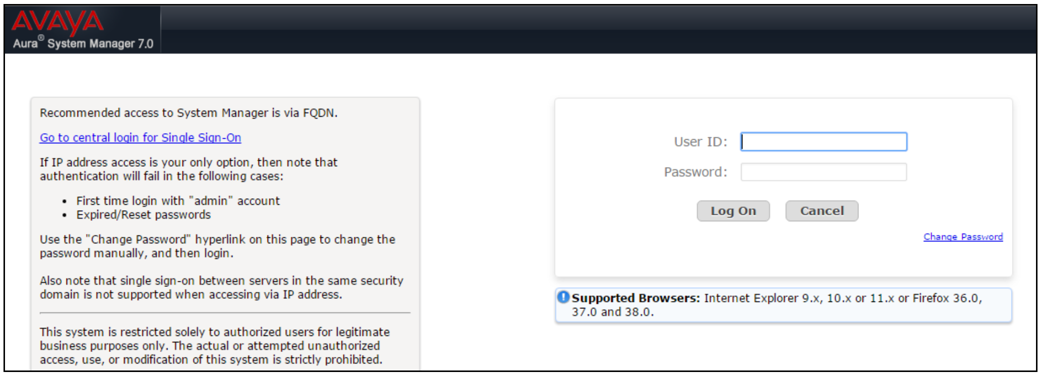
Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	SM31_In	*	External_In	Internal_In	default-low	Cardeasy	View	Clone	Edit	Delete

7. Configure Avaya Aura® Session Manager

This section describes the steps required to configure Session Manager to connect to Session Border Controller for Enterprise and forward calls to Communication Manager. It is assumed that Session Manager has been installed and configured and this configuration is therefore out with the scope of this document. All configuration was done via the Avaya Aura® System Manager web interface.

7.1. Log into System Manager

Using an internet browser go to **https://<system Manager IP>/SMGR**. Use valid credentials to log in.



Recommended access to System Manager is via FQDN.

Go to central login for Single Sign-On

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

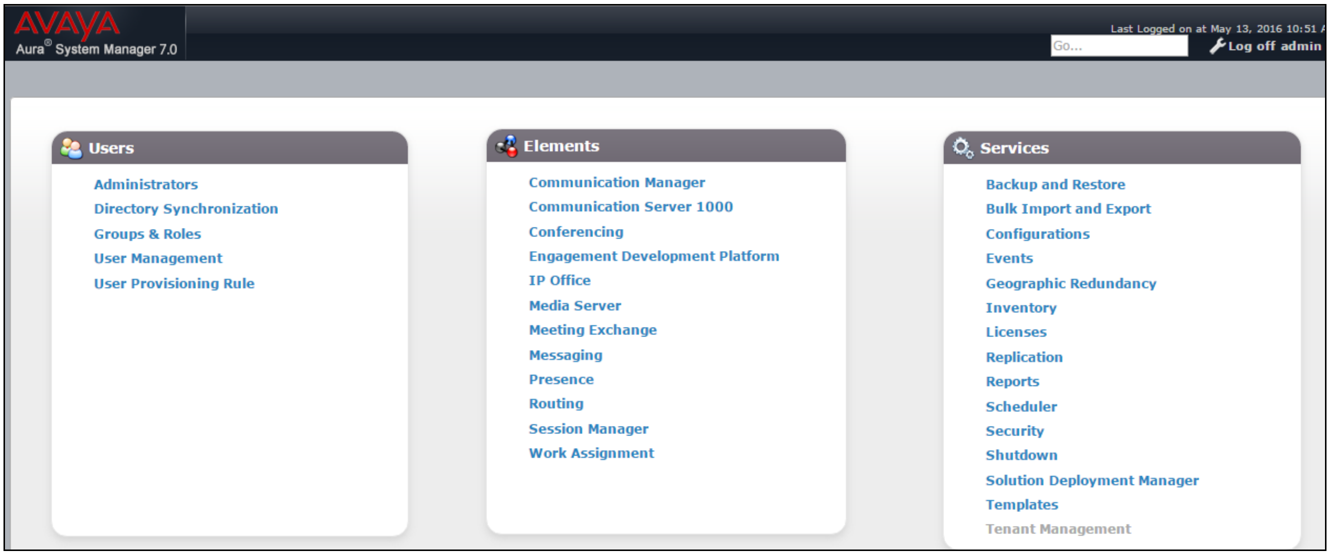
This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

User ID:

Password:

Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

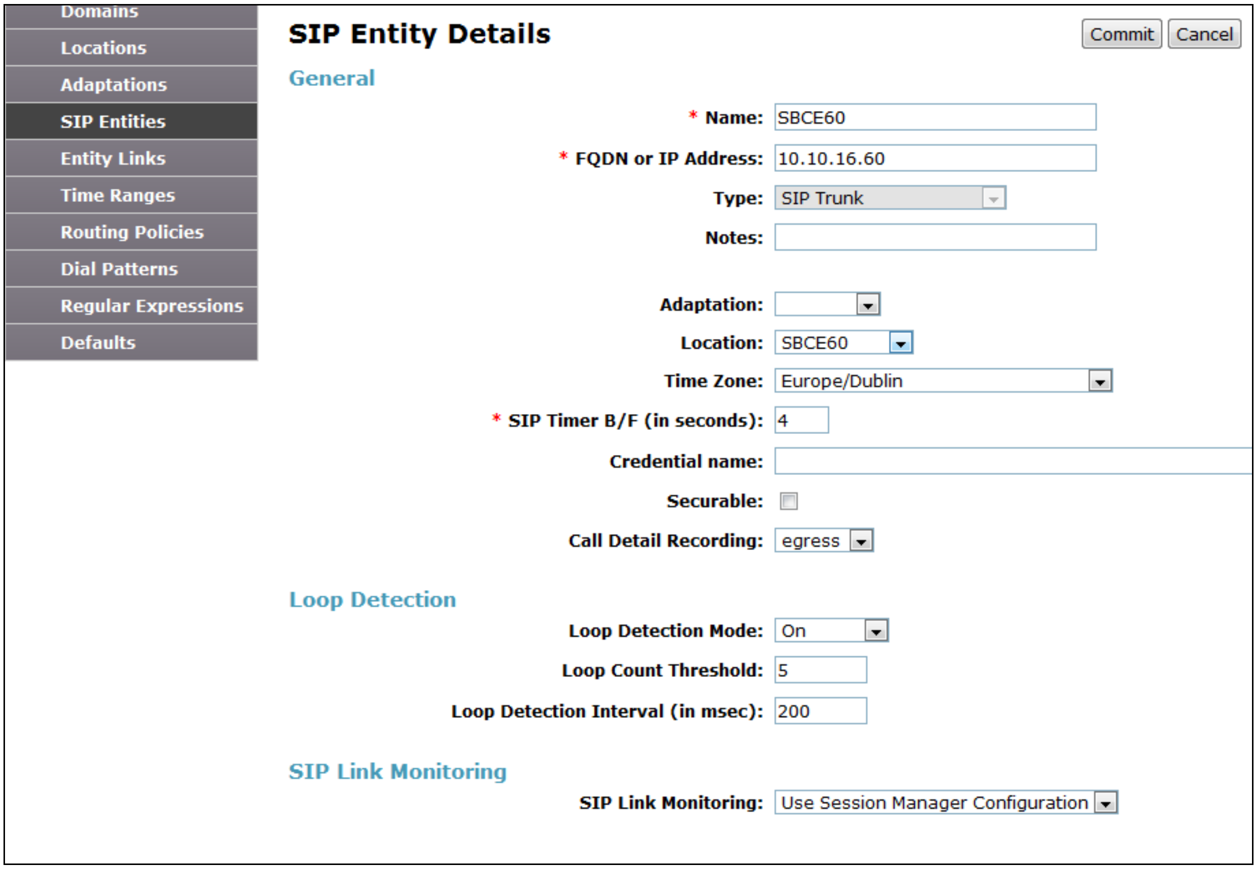
The Dashboard will be shown when logged in.



From the Dashboard select **Routing** from the **Elements** section (not shown). From the left hand menu select **SIP Entities** and click on **New** (not shown).

- Enter a Descriptive **Name**
- Enter the **FQDN or IP Address** of the Avaya SBCE
- Set Type as **SIP Trunk**

Other entries can be default.
Click on Commit.



SIP Entity Details

General

* Name: SBCE60

* FQDN or IP Address: 10.10.16.60

Type: SIP Trunk

Notes:

Adaptation:

Location: SBCE60

Time Zone: Europe/Dublin

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

Credential name:

Securable: □

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

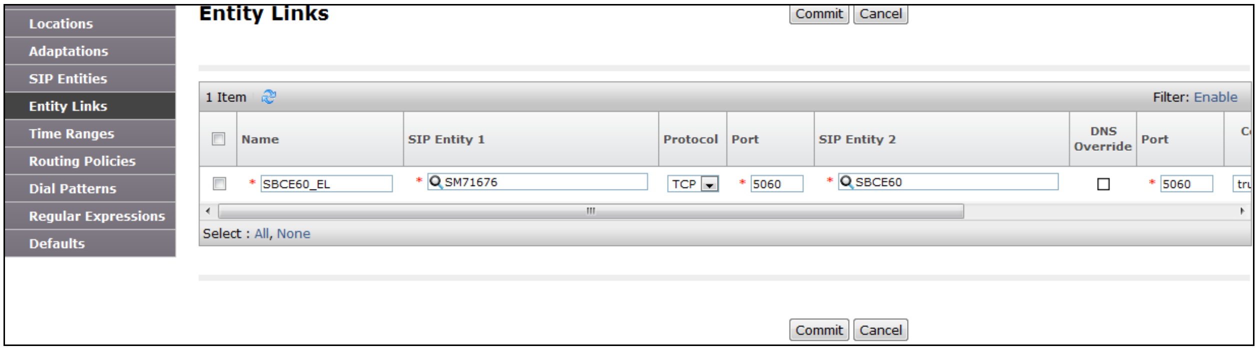
Loop Detection Interval (in msec): 200

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

From the left hand menu select Entity Links and click on New (not shown).

- Enter a descriptive Name
- Set **SIP Entity 1** as the Session Manager used to forward calls to Communication Manager.
- Set **SIP Entity 2** as the Session Border Controller for Enterprise added above.
- Set **Protocol** as **TCP** (port is set to **5060** automatically)
- Click Commit



Entity Links

1 Item

□	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	C
□	* SBCE60_EL	* SM71676	TCP	* 5060	* SBCE60	□	* 5060	tru

8. Configure CardEasy Core

All configuration of the CardEasy Core appliance and service is undertaken by Syntec as part of its managed service PCI offering.

9. Verification Steps

This section describes the steps to show that the SIP trunk is operational.

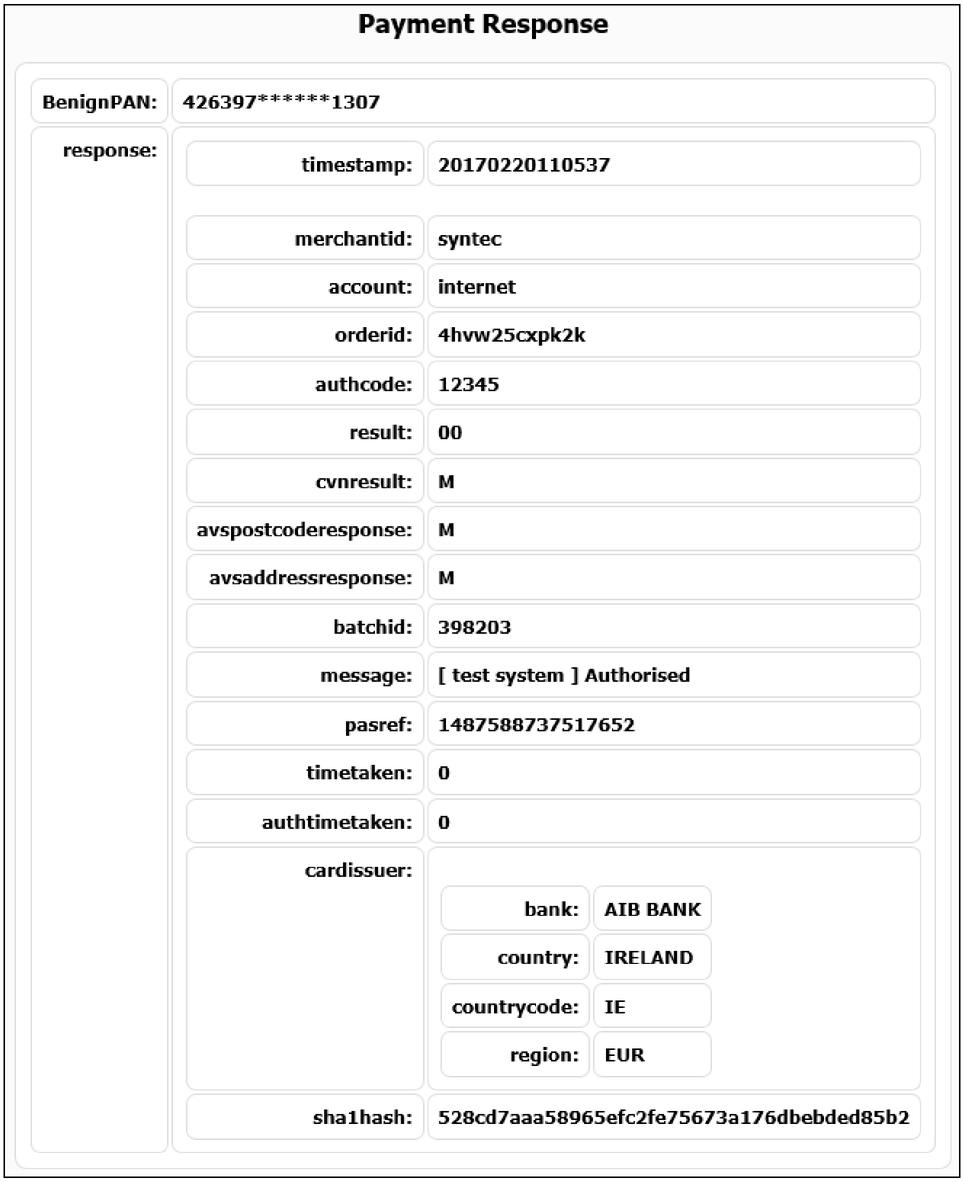
9.1. Verify SIP Trunk on Communication Manager

Use the **status trunk n** where **n** is the SIP trunk number. Make sure that all trunks are showing as **in-service/idle**. Make a call into Communication Manager and make sure that the call can be answered.

```
status trunk 11

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0011/001 T00266   in-service/idle    no
0011/002 T00267   in-service/idle    no
0011/003 T00268   in-service/idle    no
0011/004 T00269   in-service/idle    no
0011/005 T00270   in-service/idle    no
0011/006 T00271   in-service/idle    no
0011/007 T00272   in-service/idle    no
0011/008 T00273   in-service/idle    no
0011/009 T00274   in-service/idle    no
0011/010 T00275   in-service/idle    no
```

9.2. Verify CardEasy

During a call, process a credit card transaction and verify that an **Authorised** response is returned.



Payment Response

BenignPAN:	426397******1307		
response:	timestamp:	20170220110537	
	merchantid:	syntec	
	account:	internet	
	orderid:	4hvw25cxpk2k	
	authcode:	12345	
	result:	00	
	cvnresult:	M	
	avspostcoderesponse:	M	
	avsaddressresponse:	M	
	batchid:	398203	
	message:	[test system] Authorised	
	pasref:	1487588737517652	
	timetaken:	0	
	authtimetaken:	0	
	cardissuer:	bank:	AIB BANK
		country:	IRELAND
		countrycode:	IE
		region:	EUR
	sha1hash:	528cd7aaa58965efc2fe75673a176dbebded85b2	

10. Conclusion

These Application Notes describe the configuration required for Syntec CardEasy Core Cloud Solution to interoperate with Avaya Session Border Controller for Enterprise and Avaya Aura® Communication Manager using a SIP Trunk. All tests passed successfully with any observations noted in **Section 2.2**

11. Additional References

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

[1] Administering Avaya Aura® Communication Manager, Release 7.0, August 2015, *Document Number 03-300509*, Issue 1.

[2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.0, August 2015, *Document Number 555-245-205*, Issue 1.

Product Documentation for Syntec CardEasy can be requested from support@syntec.co.uk.

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.