Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000 R7.65, Avaya Aura ® Session Manager 7.0 and Avaya Session Border Controller for Enterprise R7.0 to support BT Global Services SIP Trunk Platform (NOAS) - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between BT Global Services SIP Trunk and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Session Border Controller for Enterprise, Avaya Aura® Session Manager and Avaya Communication Server 1000. BT 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 BT Global Services SIP Trunk and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of the following: Avaya Communication Server 1000 R7.65 (CS1000); Avaya Aura ® Session Manager R7.0 (Session Manager) and Avaya Session Border Controller for Enterprise R7.0 (Avaya SBCE). Note that the shortened names shown in brackets will be used throughout the remainder of the document. Customers using this Avaya SIP-enabled enterprise solution with BT Global Services 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 Server 10000, Session Manager and Avaya SBCE. The enterprise site was configured to connect to the BT Global Services SIP Trunk Platform.

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 the PSTN were routed to the DID numbers assigned by BT. Incoming PSTN calls were terminated on Digital, Unistim, SIP and Analog telephones at the enterprise side.
- Outgoing calls from the enterprise site were completed via BT to PSTN telephones. Outgoing calls from the enterprise to the PSTN were made from Digital, Unistim, SIP and Analog telephones.
- Calls using the G.711A and G.729A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by BT Global Services SIP Trunk Platform requiring Avaya response and sent by Avaya requiring BT response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for BT Global Services SIP Trunk with the following observations:

- The CS1000 default configuration will not allow a blind transfer to be executed (incoming SIP Service Provider trunk to outgoing SIP Service Provider trunk) if the SIP Service Provider in question does not support the SIP UPDATE method. With the installation of plugin 501 on the CS1000, the blind transfer will be allowed and the call will be completed. The limitation of this plugin is that no ringback is provided to the originator of the call for the duration that the destination set is ringing. In addition to plugin 501, it is required that **VTRK SU version "cs1000-vtrk-7.65.16.22.-4.i386.000.ntl"** or higher be used on all SSG signalling servers to ensure proper operation of the blind transfer feature. The use of plugin 501 does not restrict the use of the SIP UPDATE method of blind transfer to other parties that do happen to support the UPDATE method, but rather extends support to those parties that do not. Note that plugin 501 is independent of and does not require the Global Plugin Package 409.
- When testing failover to alternative network SBC, outgoing calls took approximately 32 seconds to set up. A subsequent call did not attempt to set up via the non-operational SBC and was established within an acceptable time though there was no audio. An attempt was made to reduce the initial setup time by reducing SIP timer T1 on the Avaya SBCE but this did not function according to RFC 3261. Fault Report AURORA-7344 was raised to have this investigated by the Avaya SBCE support team.
- The SIP Trunk between the Avaya Galway Lab and the BT Sandbox was unstable and became non-operational several times during testing. This was deemed to be a network issue and not related to the functionality of the BT Global Services SIP Trunk Platform.
- The network responded to an outbound call to an invalid PSTN number with 404 "Service Unavailable-No ports available". This behaviour did not create an issue and a tone was heard on the calling phone. It is noted however, as the commonly used response is 404 "Not Found".
- The BT Sandbox did not have a voicemail system in operation at the time of test. Instead DTMF was successfully tested using IVR.
- There are no mobile phones available on the BT sandbox so Mobile X feature was tested with a fixed phone.
- All unwanted MIME was stripped on outbound calls using the Adaptation Module in Session Manager.
- No inbound toll free numbers were tested as none were available from the Service Provider.
- No Emergency Services numbers tested as test calls to these numbers should be pre-arranged with the Operator.

2.3. Support

For technical support on BT Global Services products please contact BT Global Services on 0800 028 5314 or visit their website at www.globalservices.bt.com

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to BT's SIP Trunk Service. Located at the Enterprise site is an Avaya SBCE, Session Manager and CS1000. Endpoints are Avaya 1140 series IP telephones (with Unistim and SIP firmware), Avaya 1200 series IP telephones (with Unistim and SIP firmware), Avaya IP 2050PC Softphone, Avaya Digital telephone, Analog telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

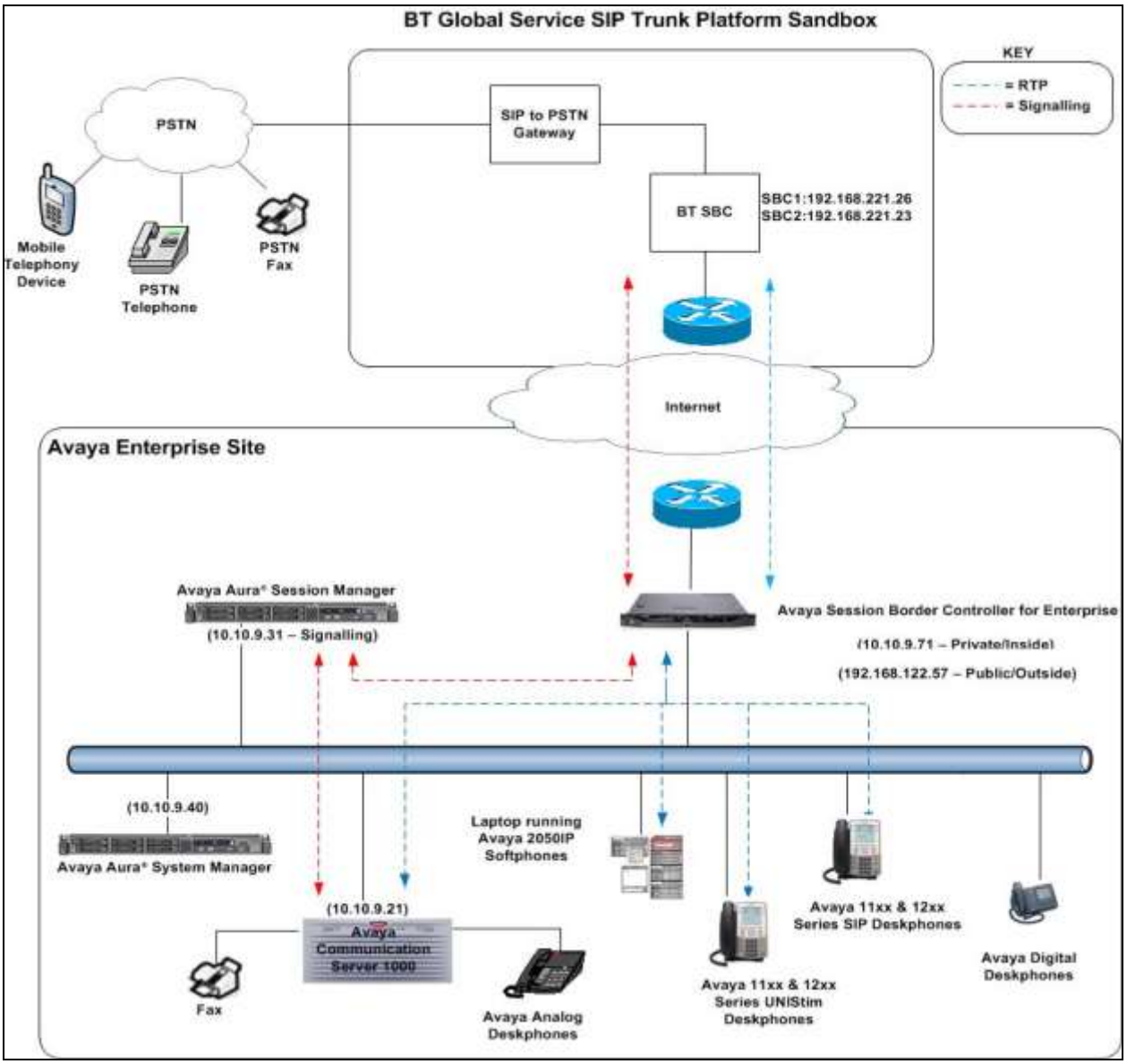


Figure 1: Test Setup BT SIP Trunk to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Aura® System Manager	7.0.0.0.16266
Avaya Communication Server 1000	Avaya Communication Server 1000E R7.6 Version 7.65.P Deplist: CPL_X21_07_65P All CS1000 patches listed in **Appendix A**
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC DC01 MSP Version: MGCM AB02 APP Version: MGCA BA18 FPGA Version: MGCF AA22 BOOT Version: MGCB BA18 DSP1 Version: DSP2 AB07
Avaya Session Border Controller for Enterprise	7.0.0-21-6602
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.10.18.00.bin
Avaya 2050PC	Release 4.3.0081
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
BT Global Services	
Genband S3 Session Border Controller	8.3.7.1
NOAS Call Server	4.38.0.1

5. Configure Avaya Communication Server 1000

This section describes the steps required to configure CS1000 for SIP Trunking and also the basic configuration for telephones (analog, SIP and IP phones). SIP trunks are established between CS1000 and Session Manager. SIP trunks are also established between Session Manager and the Avaya SBCE private interface. The Avaya SBCE public interface connects to the BT Global Services SIP trunks. Incoming PSTN calls from the BT Global Services SIP Trunk service traverse the Avaya SBCE and are directed to the Session Manager, which directs the calls to CS1000 (see **Figure 1**).

When a SIP message arrives at CS1000, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within CS1000 and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. When CS1000 selects a SIP trunk for outgoing PSTN calls, SIP signaling is directed to Session Manager. Session Manager directs the outbound SIP messages to the Avaya SBCE private interface. The Avaya SBCE public interface manages outgoing SIP sessions onwards to the BT Global Services SIP trunks.

Specific CS1000 configuration was performed using Element Manager and the system terminal interface. The general installation of the CS1000, System Manager, Session Manager and Avaya SBCE is presumed to have been previously completed and is not discussed here. Configuration details will be provided as required to draw attention to changes in default system configurations.

5.1. Logging into the Avaya Communication Server 1000E

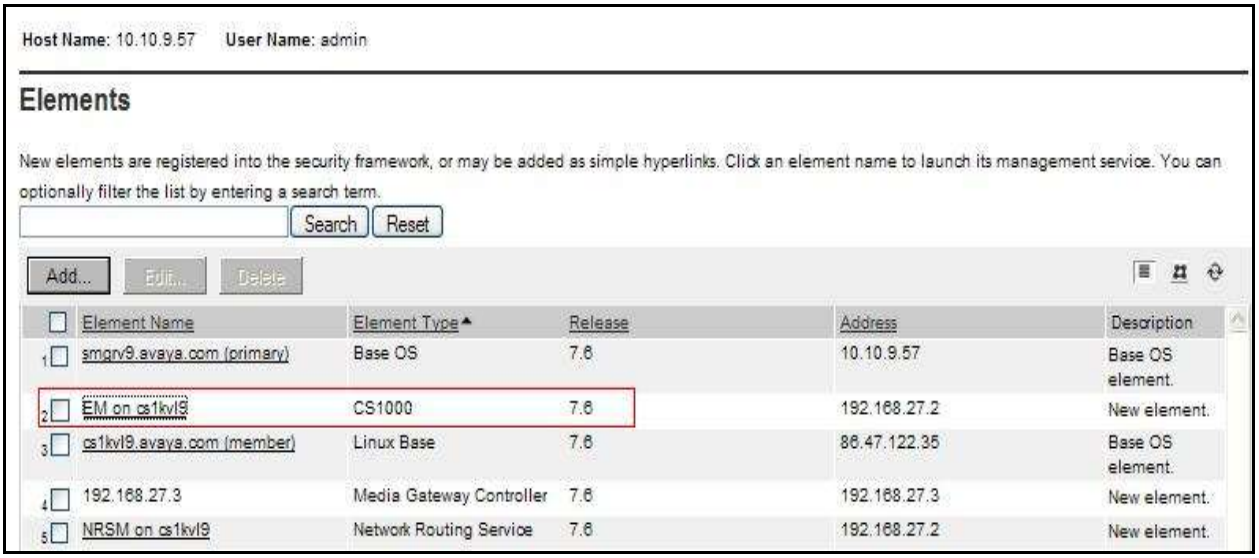
Configuration on the CS1000 will be performed by using both SSH Putty session and Avaya Unified Communications Management GUI.

Log in using SSH to the ELAN IP address of the Call Server with a username containing the correct privileges. Once logged in type **csconsole,** this will take the user into the vxworks shell of the call server. Next type **login**; the user will then be asked to login with correct credentials. Once logged-in the user can then progress to load any overlay.

Log in using the web based Avaya Unified Communications Management GUI. Avaya Unified Communications Management GUI may be launched directly via http://<ipaddress> where the relevant <ipaddress> is the TLAN IP address of the CS1000. Avaya Unified Communications Management can also be implemented on System Manager.

The following screen shows the login screen. Login with the appropriate credentials.

The Avaya Unified Communications Management **Elements** page will be used for configuration. Click on the Element Name corresponding to CS1000 in the Element Type column. In the abridged screen below, the user would click on the Element Name **EM on cs1kvl9**.



5.2. Confirm System Features

The keycode installed on the Call Server 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 CS1000 system terminal and manually load overlay 22 to print the System Limits (the required command is **slt**), and verify that the number of SIP Access Ports reported by the system is sufficient for the combination of trunks to the BT Global Services network, and any other SIP trunks needed. See the following screenshot for a typical System Limits printout. The value of **SIP ACCESS PORTS** defines the maximum number of SIP trunks for the CS1000.

```
System type is - Communication Server 1000/CP PM
CP PM - Pentium M 1.4 GHz

IPMGs Registered:               4
IPMGs Unregistered:             0
IPMGs Configured/unregistered:  2

TRADITIONAL TELEPHONES   120    LEFT   110    USED    10
DECT USERS                16    LEFT    16    USED     0
IP USERS               10000    LEFT  9954    USED    46
BASIC IP USERS            16    LEFT    13    USED     3
TEMPORARY IP USERS         8    LEFT     8    USED     0
DECT VISITOR USER         16    LEFT    16    USED     0
ACD AGENTS               192    LEFT   185    USED     7
MOBILE EXTENSIONS          8    LEFT     7    USED     1
TELEPHONY SERVICES        16    LEFT    13    USED     3
CONVERGED MOBILE USERS     8    LEFT     8    USED     0
AVAYA SIP LINES           16    LEFT    12    USED     4
THIRD PARTY SIP LINES     16    LEFT    16    USED     0
PCA                       20    LEFT    18    USED     2
ITG ISDN TRUNKS            0    LEFT     0    USED     0
H.323 ACCESS PORTS       524    LEFT   524    USED     0
AST                     6652    LEFT  6640    USED    12
SIP CONVERGED DESKTOPS    16    LEFT    16    USED     0
SIP CTI TR87              16    LEFT     8    USED     8
SIP ACCESS PORTS         524    LEFT   518    USED     6
RAN CON                   90    LEFT    90    USED     0
MUS CON                  120    LEFT   120    USED     0
```

Load Overlay 21 and confirm the customer is setup to use **ISDN** trunks by typing the **PRT** and **NET_DATA** commands as shown below.

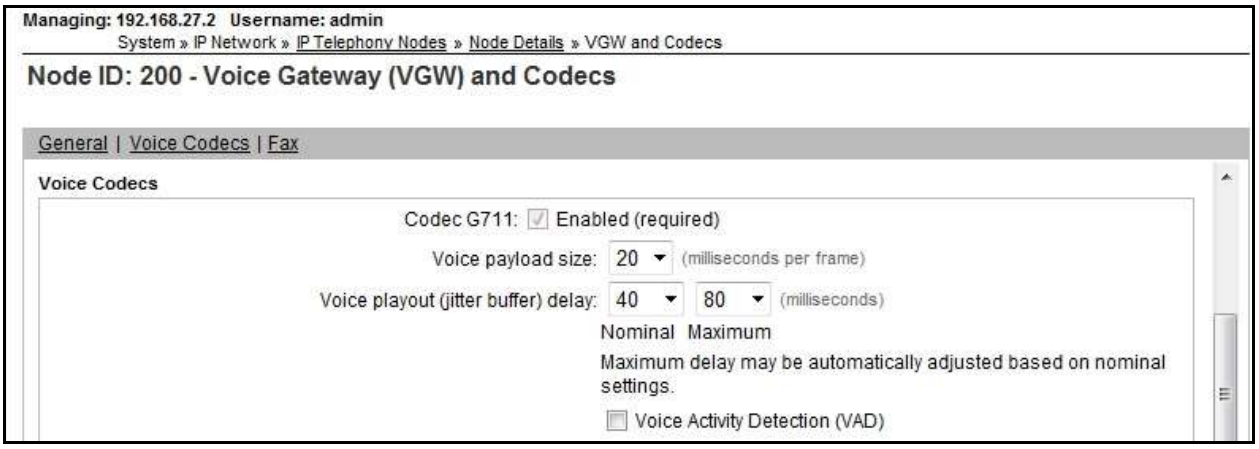
```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

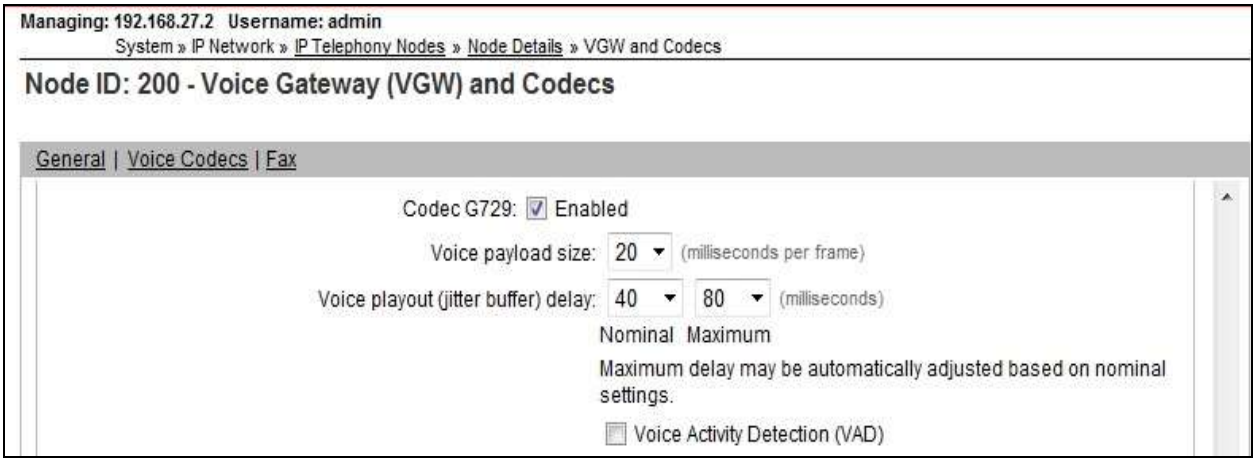
5.3. Configure Codecs for Voice and FAX operation

BT Global Service's SIP Trunk supports G.711A and G.729 voice codecs. Using the CS1000 Element Manager sidebar, select **Nodes, Servers, Media Cards**. Navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW and Codecs** property page and configure the CS1000 **General** codec settings as in the following screenshots. The values highlighted are required for correct operation. The following screenshot shows the necessary **General** settings.

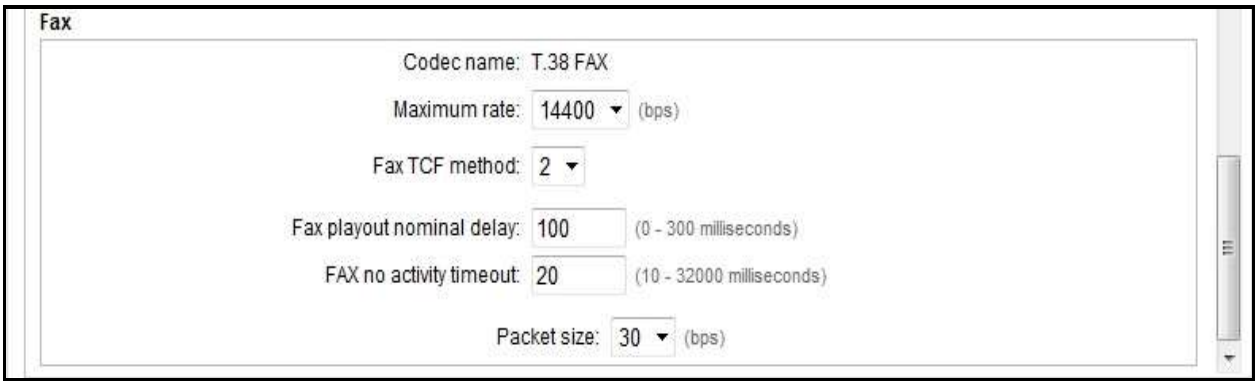
Move down to the Voice Codecs section and configure the G.711 codec settings. The following screenshot shows the G.711 codec settings.



Next, scroll down to the G.729 codec section and configure the settings.

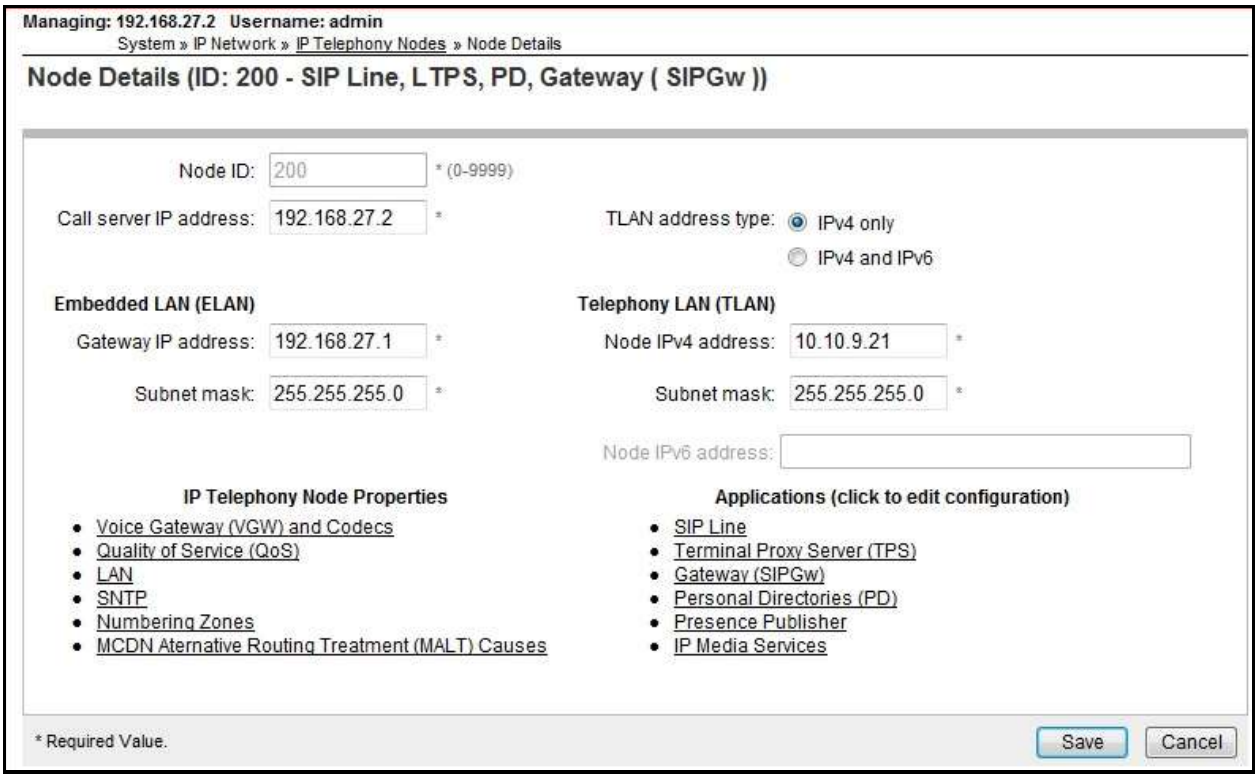


Finally, configure the Fax settings as in the highlighted section of the next screenshot. Click on the **Save** button when finished.



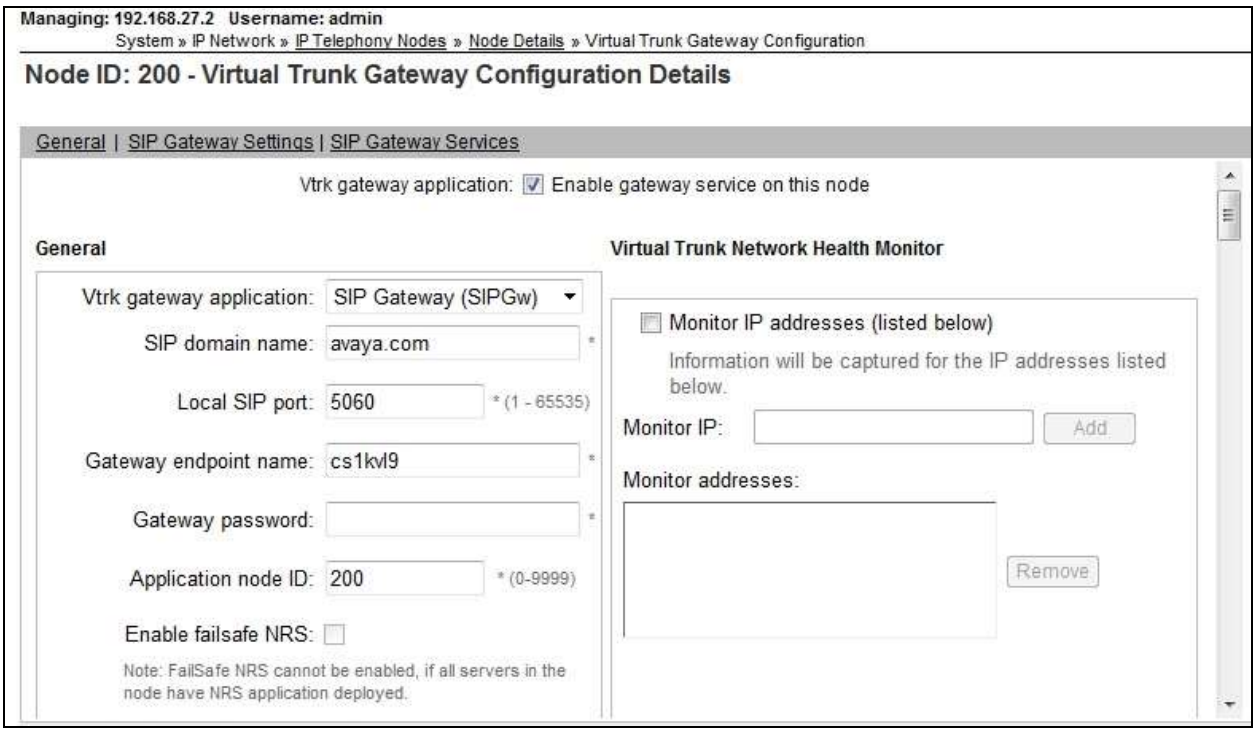
5.4. Virtual Trunk Gateway Configuration

Use CS1000 Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. The call server and signaling server have previously been configured with IP addresses. The Node IPv4 address is the IP address that the IP phones use to register. This is also where the SIP trunk connection is made to Session Manager. When an entity link is added in Session Manager for the CS1000, it is the Node IPv4 address that is used (see **Section 6.5** – Define SIP Entities for more details).



The next two screenshots show the SIP Virtual Trunk Gateway configuration, navigate to **System → IP Networks → IP Telephony Nodes → Node Details → Gateway (SIPGW) Virtual Trunk Configuration Details** and fill in the highlighted areas with the relevant settings.

- **Vtrk gateway application:** Provides option to select Gateway applications. The three supported modes are **SIP Gateway (SIPGw)**, **H.323Gw**, and **SIPGw and H.323Gw**
- **SIP domain name:** The SIP domain name is the SIP Service Domain. The SIP domain name configured in the Signaling Server properties must match the Service Domain name configured in Session Manager; in this case **avaya.com**
- **Local SIP port:** The Local SIP Port is the port to which the gateway listens. The default value is **5060**
- **Gateway endpoint name:** This field cannot be left blank so a value is needed here. This field is used when a Network Routing Server is used for registration of the endpoint. In this network a Session Manager is used so any value can be put in here and will not be used
- **Application node ID:** This is a unique value that can be alphanumeric and is for the new Node that is being created, in this case **200**
- **Proxy or Redirect Server:** Primary TLAN IP address is the Security Module IP address of Session Manager. The **Transport protocol** used for **SIP**, in this case is **TCP**
- **SIP URI Map: Public E.164 - National** and **Private - Unknown** are left blank. All other fields in the SIP URI Map are left with default values



Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.9.31

The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

Options: Support registration

Primary CDS proxy

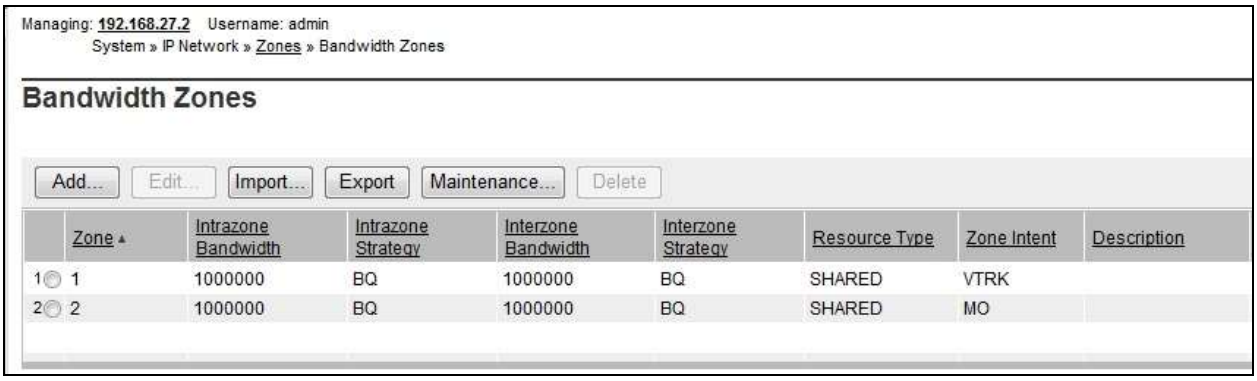
SIP URI Map:

Public E.164 domain names		Private domain names	
National:		UDP:	udp
Subscriber:	subscriber	CDP:	cdp.udp
Special number:	PublicSpecial	Special number:	PrivateSpecial
Unknown:	PublicUnknown	Vacant number:	PrivateUnknown
		Unknown:	

5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for bandwidth management. SIP trunks require a unique zone, not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in separate zones. In the sample configuration SIP trunks use zone 01 and IP and SIP Telephones use zone 02; system defaults were used for each zone other than the parameter configured for **Zone Intent**. For SIP Trunks (zone 01), **VTRK** is configured for **Zone Intent**. For IP, SIP Telephones (zone 02), **MO** is configured for **Main Office**.

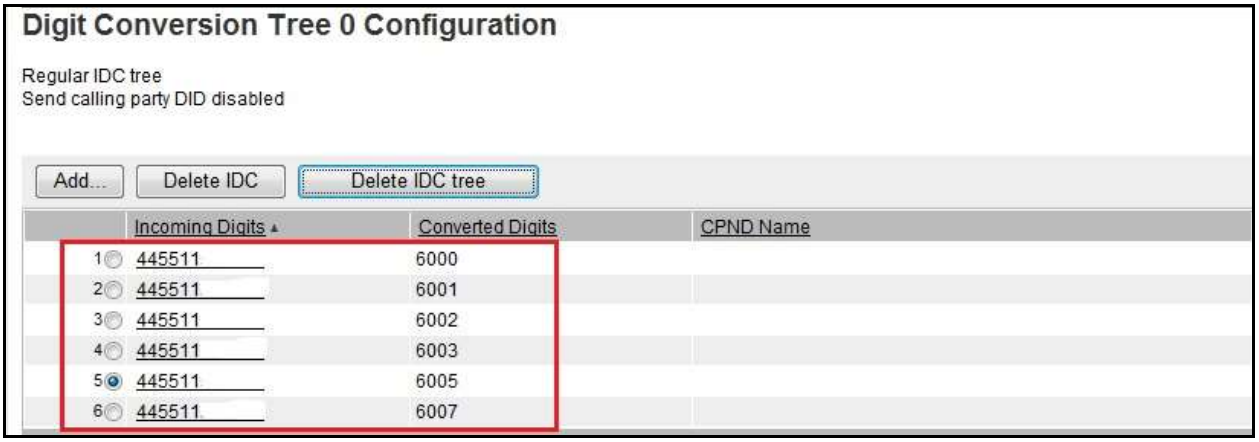
Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **System → IP Network → Zones → Bandwidth Zones** and add new zones as required.



	Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1	1	1000000	BQ	1000000	BQ	SHARED	VTRK	
2	2	1000000	BQ	1000000	BQ	SHARED	MO	

5.6. Configure Incoming Digit Conversion Table

A limited number of Direct Dial Inwards (DDI) numbers were available. The Incoming Digit Conversion (IDC) table was configured to translate incoming PSTN numbers to four digit local telephone extension numbers. The digits of the actual PSTN DDI number are obscured for security reasons. The following screenshot shows the incoming PSTN numbers converted to local extension numbers. These were altered during testing to map to various SIP, Analog, Digital or UNIStim telephones depending on the particular test case being executed.



	Incoming Digits	Converted Digits	CPND Name
1	445511	6000	
2	445511	6001	
3	445511	6002	
4	445511	6003	
5	445511	6005	
6	445511	6007	

5.7. Configure SIP Trunks

CS1000 virtual trunks will be used for all inbound and outbound PSTN calls to the BT Global Services SIP Trunk service. Six separate steps are required to configure CS1000 virtual trunks:

- Configure a D-Channel Handler (**DCH**); configure using the CS1000 system terminal and overlay 17
- Configure a SIP trunk Route Data Block (**RDB**); configure using the CS1000 system terminal and overlay 16
- Configure SIP trunk members; configure using the CS1000 system terminal and overlay 14
- Configure a Digit Manipulation Data Block (**DGT**), configure using the CS1000 system terminal and overlay 86
- Configure a Route List Block (**RLB**); configure using the CS1000 system terminal and overlay 86
- Configure Co-ordinated Dialling Plan(s) (**CDP**); configure using the CS1000 system terminal and overlay 87

The following is an example DCH configuration for SIP trunks. Load **Overlay 17** at the CS1000 system terminal and enter the following values. The highlighted entries are required for correct SIP trunk operation. Exit overlay 17 when completed.

```
Overlay 17
ADAN     DCH 1
  CTYP DCIP
  DES  VIR_TRK
  USR  ISLD
  ISLM 4000
  SSRC 3700
  OTBF 32
  NASA YES
  IFC  SL1
  CNEG 1
  RLS  ID  4
  RCAP ND2
  MBGA NO
  H323
    OVLR NO
    OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the CS1000 system terminal and overlay 16. Load **Overlay 16**, enter **RDB** at the prompt, press return and commence configuration. The value for **DCH** is the same as previously entered in overlay 17. The value for **NODE** should match the node value in **Section 5.4**. The value for **ZONE** should match that used in **Section 5.5** for **VTRK**. The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 16
TYPE: RDB
CUST 00
ROUT 1
TYPE RDB
CUST 00
ROUT 1
DES  VIR_TRK
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
RPA  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 00001
PCID SIP
CRID NO
NODE 200
DTRK NO
ISDN YES
    MODE ISLD
    DCH  1
    IFC  SL1
    PNI  00000
    NCNA YES
    NCRD YES
    TRO  NO
    FALT NO
    CTYP UKWN
    INAC NO
    ISAR NO
    DAPC NO
MBXR NO
MBXOT NPA
MBXT 0
PTYP ATT
CNDP UKWN
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
```

```
ACOD 1111
TCPP NO
PII NO
AUXP NO
TARG
CLEN 1
BILN NO
OABS
INST
IDC  YES
DCNO 0
NDNO 0 *
DEXT NO
DNAM NO
SIGO STD
STYP SDAT
MFC  NO
ICIS YES
OGIS YES
TIMR ICF  1920
     OGF  1920
     EOD  13952
     LCT  256
     DSI  34944
     NRD  10112
     DDL  70
     ODT  4096
     RGV  640
     GTO  896
     GTI  896
     SFB  3
     PRPS  800
     NBS  2048
     NBL  4096
     IENB  5
     TFD  0
     VSS  0
     VGD  6
     EESD  1024
SST  5 0
DTD  NO
SCDT NO
2 DT NO
NEDC ORG
FEDC ORG
```

```
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR  NO
NATL YES
SSL
CFWR NO
IDOP NO
VRAT NO
MUS  YES
MRT  21
PANS YES
RACD NO
MANO NO
FRL  0 0
FRL  1 0
FRL  2 0
FRL  3 0
FRL  4 0
FRL  5 0
FRL  6 0
FRL  7 0
OHQ  NO
OHQT 00
CBQ  NO
AUTH NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
OPR  NO
ALRM NO
ART  0
PECL NO
DCTI 0
TIDY 1600 100
ATRR NO
TRRL NO
SGRP 0
ARDN NO
CTBL 0
AACR NO
```

Next, configure virtual trunk members using the CS1000 system terminal and **Overlay 14**. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load **Overlay 14** at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 14
TN   100 0 0 0
DATE
PAGE
DES  VIR_TRK
TN   100 0 00 00  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00001
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK  ANLG
NCOS 0
RTMB 1 1
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST  NO
IAPG 0
CLS  UNR DIP CND ECD WTA LPR APN THFD XREP SPCD MSBT
     P10 NTC
TKID
AACR NO
```

Next, configure a Digit Manipulation data block (DGT) in overlay 86. Load **Overlay 86** at the system terminal and type **new**. The following example shows the values used. **Note: ISPN** is set to **0** as BT Global Services required a prefix of 0 to be inserted before the dialed number for outbound calls. The value for Digit Manipulation Index (**DMI)** is the same as when inputting the **DMI** value during configuration of the Route List Block.

```
Overlay 86
CUST 0
FEAT dgt
DMI  10
DEL  0
ISPN 0
CTYP NPA
```

Configure a Route List Block (RLB) in overlay 86. Load **Overlay 86** at the system terminal and type **new**. The following example shows the values used. The value for **ROUT** is the same as previously entered in overlay 16. The **RLI** value is unique to each RLB.

```
Overlay 86
CUST 0
FEAT rlb
RLI  10
ELC  NO
ENTR 0
LTER NO
ROUT 1
TOD  0 ON  1 ON  2 ON  3 ON
     4 ON  5 ON  6 ON  7 ON
VNS  NO
SCNV NO
CNV  NO
EXP  NO
FRL  0
DMI  10
CTBL 0
ISDM 0
```

```
FCI  0
FSNI 0
BNE  NO
DORG NO
SBOC NRR
PROU 1
IDBB DBD
IOHQ NO
OHQ  NO
CBQ  NO

ISET 0
NALT 5
MFRL 0
OVLL 0
```

Next, configure Co-ordinated Dialling Plan(s) (CDP) which users will dial to reach PSTN numbers. Use the CS1000 system terminal and **Overlay 87**. The following are some example CDP entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (**RLI**), this is the default PSTN route to the SIP Trunk service.

TSC 00353	TSC 18	TSC 800	TSC 08
FLEN 0	FLEN 0	FLEN 0	FLEN 0
RRPA NO	RRPA NO	RRPA NO	RRPA NO
RLI 10	**RLI 10**	**RLI 10**	**RLI 10**
CCBA NO	CCBA NO	CCBA NO	CCBA NO

5.8. Calling Line Identification

This section documents basic configuration relevant to the BT Global Services configuration. **Load Overlay 15** at system terminal and enter the required values in bold. As shown below, **CLID** is set to **YES** and **ENTRY** is set to **0**. **HNTN** and **HLCL** match the required digits assigned by BT Global Services and **DIDN** is set to **NO**.

```
Load Overlay 15
TYPE NET_DATA
CUST 0
OPT
AC2
FNP
CLID YES
  SIZE
  INTL
  ENTRY 0
HNTN 07689
  ESA_HLCL
  ESA_INHN NO
  ESA_APDN NO
  HLCL 11010
  DIDN NO
  DIDN_LEN 0
  HLOC
  LSC
  CLASS_FMT DN
```

5.9. Configure Analog, Digital and IP Telephones

A variety of telephone types were used during the testing, the following is the configuration for the Avaya 1140e UNIStim IP telephone. Load **Overlay 20** at the system terminal and enter the following values. A unique four digit number is entered for the **KEY 00**. The value for **CFG_ZONE** is the value used in **Section 5.5** for IP and SIP Telephones.

```
Load Overlay 20 IP Telephone configuration
DES  1140
TN   100 0 03 0  VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00002
CUR_ZONE 00002
ERL  0
ECL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMD LLCN MCTD CLBD AUTR
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA  PKCH MUTA MWTD
```
---continued on next page----

```
---continued from previous page----

DVLD CROD CROD
CPND_LANG ENG
RCO  0
HUNT 0
LHK  0
PLEV 02
PUID
DANI NO
AST  00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 6000 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 6000 0
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02
     03 BSY
     04 DSP
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
```

Digital telephones are configured using the overlay 20; the following is a sample 3904 digital set configuration. Again, a unique number is entered for the **KEY 00** and **KEY 01** value.

```
Overlay 20 – Digital Set configuration
TYPE: 3904
DES  3904
TN   000 0 09 08  VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMA LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
     CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD
CPND_LANG ENG
RCO  0
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 1
AACS
ACQ
ASID
SFNB
SFRB
USFB
CALB
FCTB
ITNA NO
DGRP
PRI  01
MLWU_LANG 0

---continued on next page----
```

```
---continued from previous page----

MLNG ENG
DNDR 0
KEY  00 MCR 6066 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 6066 0
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02 DSP
     03 MSB
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
     27 CLT
     28 RLT
     29
     30
     31
```

Analog telephones are also configured using overlay 20; the following example shows an analog port configured for Plain Ordinary Telephone Service (POTS) and also configured to allow fax transmission. A unique value is entered for **DN**, this is the extension number. **DTN** is required if the telephone uses DTMF dialing. Values **FAXA** and **MPTD** configure the port for T.38 Fax transmissions.

```
Overlay 20 - Analog Telephone Configuration
DES  500
TN   100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN   6004
AST  NO
IAPG 0
HUNT
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI  0
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR DTN FBD XFD WTA THFD FND HTD ONS
     LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
     CFTD SFD MRD C6D CNID CLBD AUTU
     ICDD CDMD LLCN EHTD MCTD
     GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
     MBXD CPFA CPTA UDI RCC HBTD IRGD  DDGA NAMA MIND
     NRWD NRCD NROD SPKD CRD PRSD MCRD
     EXR0 SHL SMSD ABDD CFHD DNDY DNO3
     CWND USMD USRD CCBD BNRD OCBD RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
     FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR  DCFW 4
```

5.10.Configure the SIP Line Gateway Service

SIP terminal operation requires the CS1000 node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the CS1000 system terminal and overlay 15 to activate SIP Line services (SLS_DATA), as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
  SIPL_ON YES
  UAPR 11
  NMME NO
```

If a numerical value is entered against the **UAPR** setting, this number will be pre appended to all SIP Line configurations, and is used internally in the SIP Line server to track SIP terminals. Use Element Manager and navigate to the **IP Network → IP Telephony Nodes → Node Details → SIP Line Gateway Configuration** page. See the following screenshot for highlighted critical parameters.

- **SIP Line Gateway Application:** Enable the SIP line service on the node, check the box to enable
- **SIP Domain Name:** The value must match that configured in **Section 6.2**
- **SLG endpoint name:** The endpoint name is the same endpoint name as the SIP Line Gateway and will be used for SIP gateway registration
- **SLG Local Sip port:** Default value is **5070**
- **SLG Local TLS port:** Default value is **5071**

5.1. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the CS1000 system terminal and **Overlay 20** to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value for **SIPN** is 1. The **SIPU** value is the username, **SCPW** is the logon password and these values are required to register the SIP telephone to the SLG. The value for **CFG_ZONE** is the value used in **Section 5.5** for IP and SIP Telephones. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** (set in **Section 5.8**) value and the telephone number used in **KEY 00**.

```
Load Overlay 20 – SIP Telephone Configuration
DES  SIPD
TN   100 0 03 3  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 6002
NDID 200
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00002
CUR_ZONE 00002
ERL  0
ECL  0
VSIT NO
FDN
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
     MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD

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```
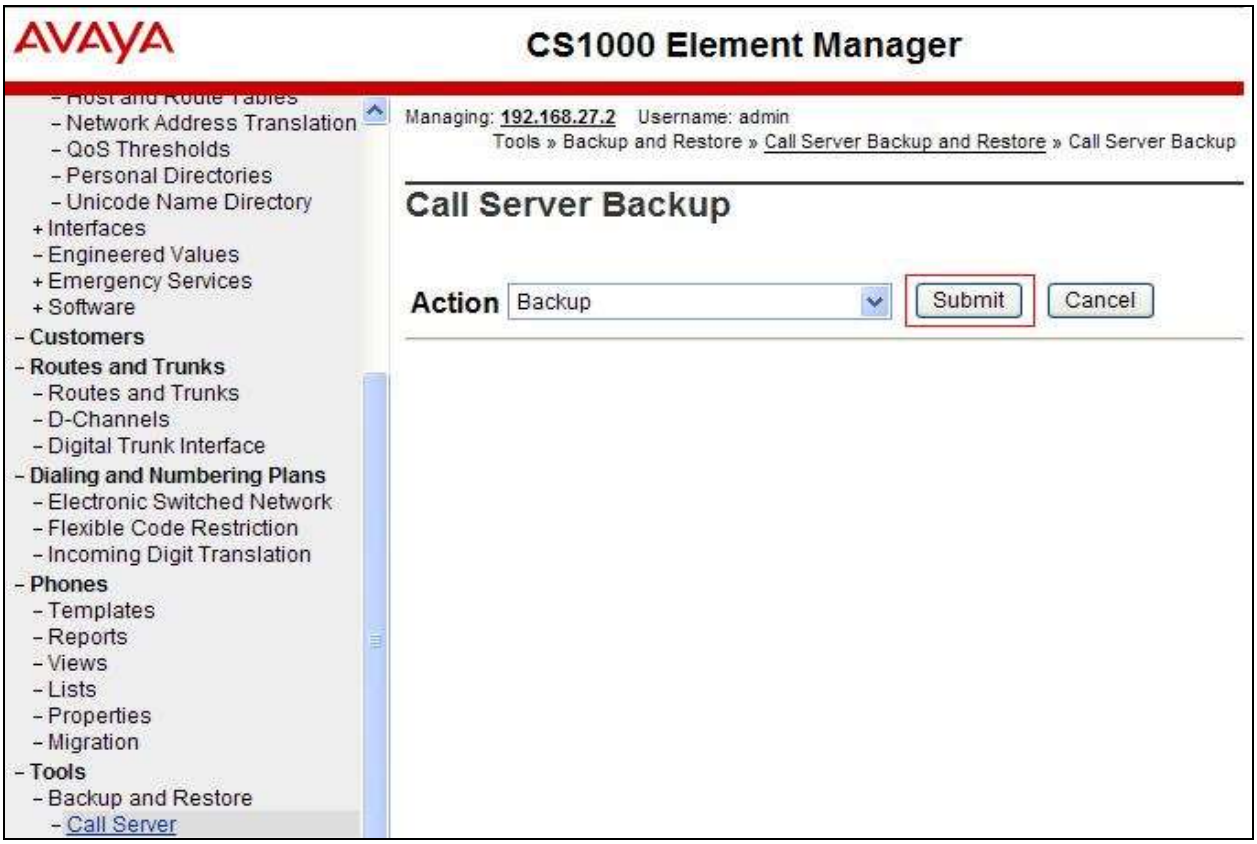
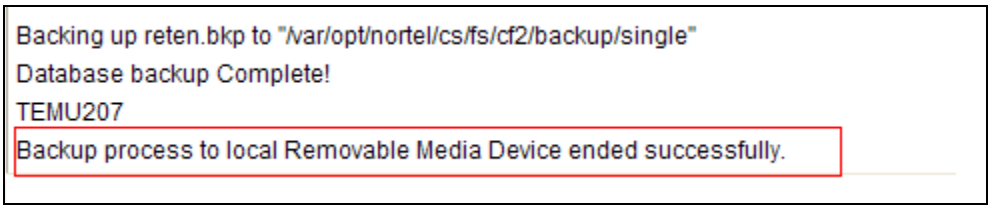

```
---continued from previous page---

     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA  PKCH MWTD DVLD
CROD CROD
CPND_LANG ENG
RCO  0
HUNT
LHK  0
PLEV 02
PUID
DANI NO
AST
IAPG 0 *

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 6002 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 116002 MARP 0
     02
     03
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23     *
     24 PRS
     25 CHG
     26 CPN
     27
     28
     29
     30
     31
```

5.2. Save Configuration

Expand **Tools → Backup and Restore** on the left navigation panel and select **Call Server**. Select **Backup** (not shown) and click **Submit** to save configuration changes as shown below.



The backup process will take several minutes to complete. Scroll to the bottom of the page to verify the backup process completed successfully as shown below.



6. Configuring Avaya Aura® Session Manager

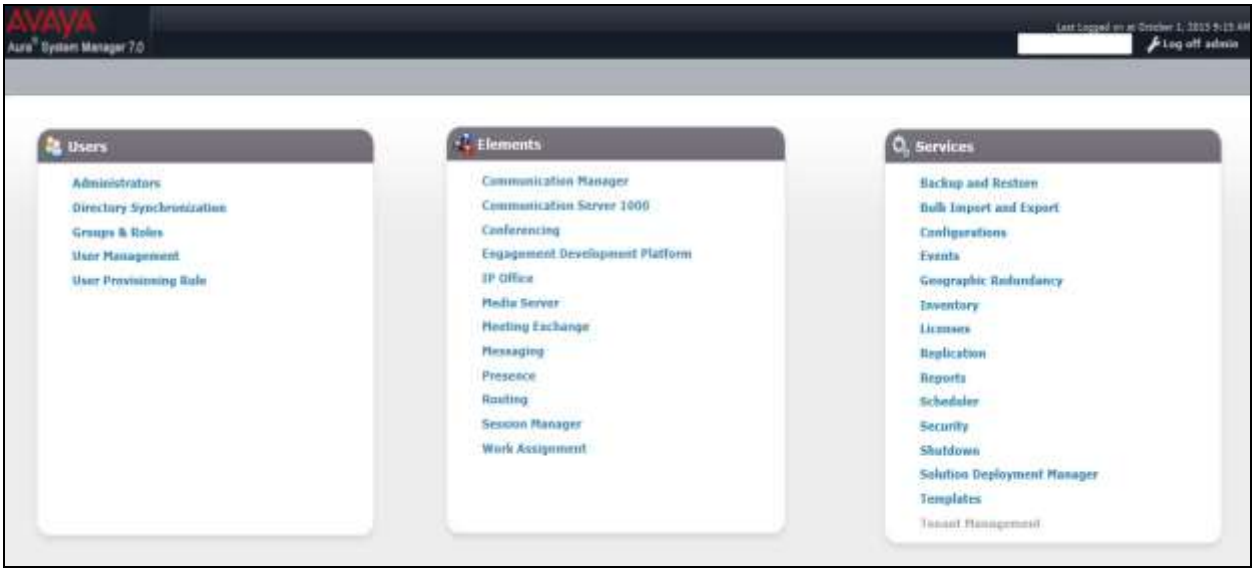
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager.
- Administer SIP Domain.
- Administer SIP Location.
- Administer Adaptations.
- Administer SIP Entities.
- Administer Entity Links.
- Administer Routing Policies.
- Administer Dial Patterns.

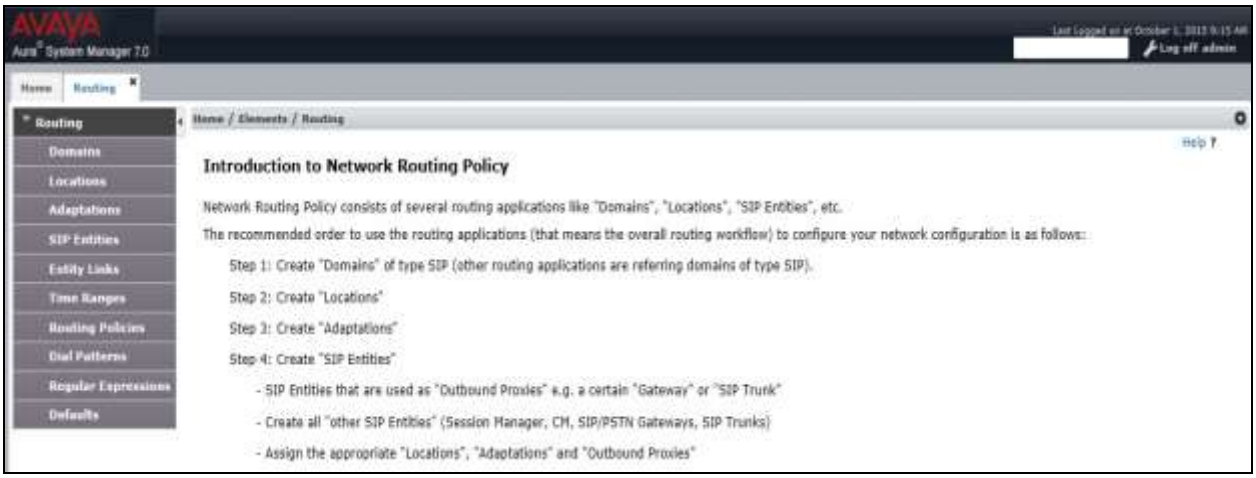
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session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

6.1. Log in to Avaya Aura® System Manager

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



Most of the configuration items are performed in the Routing Element. Click on **Routing** in the Elements column shown above to bring up the **Introduction to Network Routing Policy** screen.

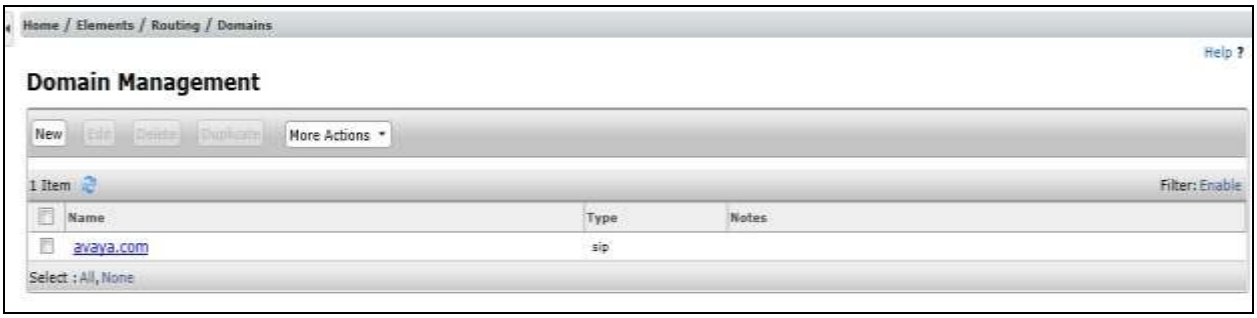


6.2. Administer SIP Domain

Create a SIP domain for each domain for which Session Manager will need to be aware in order to route calls. Expand **Elements → Routing** and select **Domains** from the left navigation menu, click **New** (not shown)**.** Enter the following values and use default values for remaining fields.

- **Name** Enter a Domain Name**.** In the sample configuration, **avaya.com** was used.
- **Type** Verify **SIP** is selected.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.



6.3. Administer Locations

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

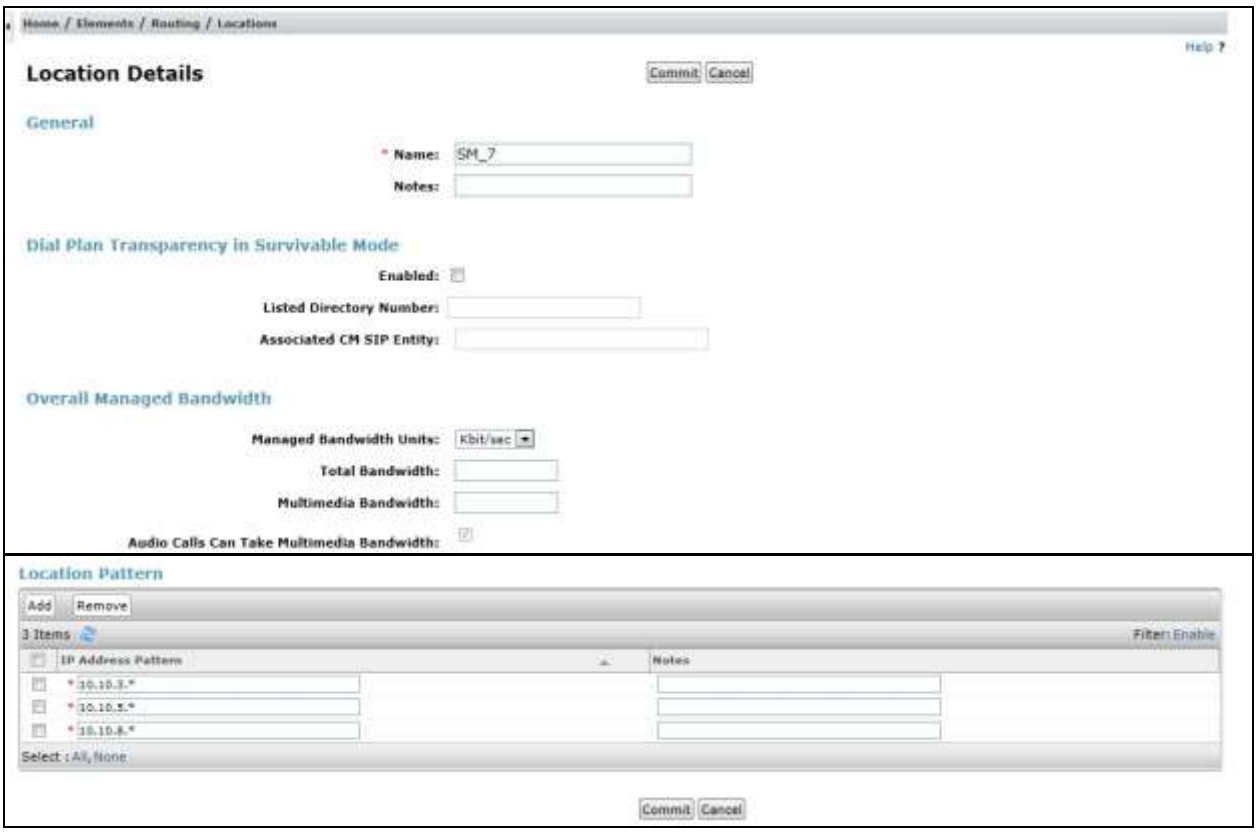
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

The Location Pattern is used to identify call routing based on IP address. Session Manager matches the IP address against the patterns defined in this section. If a call is from a SIP Entity that does not match the IP address pattern then Session Manager uses the location administered for the SIP Entity.

In the **Location Pattern** section, click **Add** and enter the following values.

- **IP Address Pattern** Enter the logical pattern used to identify the location.
- **Notes** Add a brief description [Optional].

Click **Commit** to save. The screenshot below shows the Location **SM_7** defined for the compliance testing.

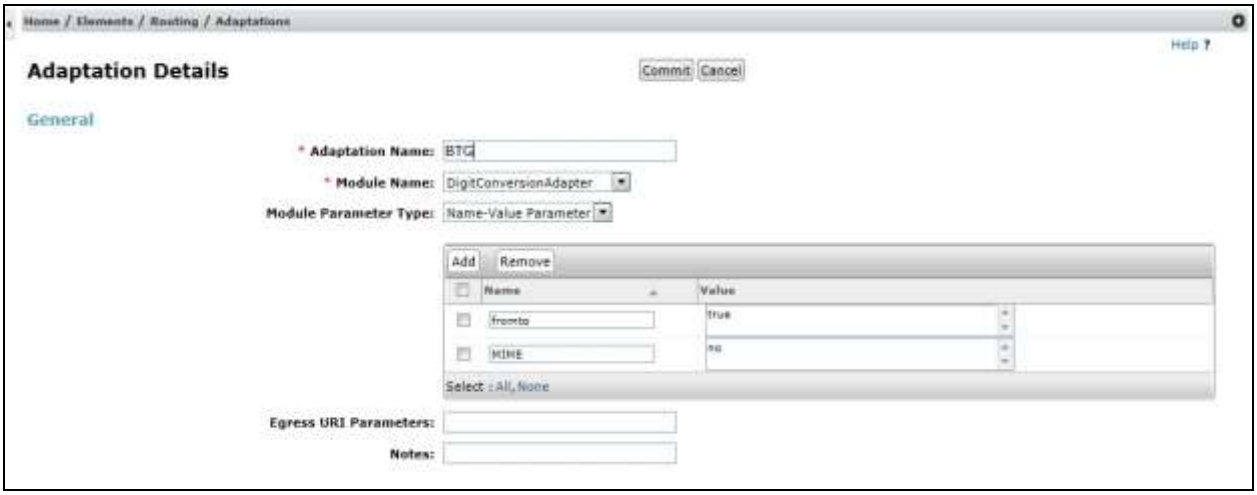


6.4. Administer Adaptations

Adaptations can be used to modify the called and calling party numbers to meet the requirements of the service. The called party number present in the SIP INVITE Request URI is modified by the **Digit Conversion** in the Adaptation. The example below was applied to the Avaya SBCE SIP Entity and was used in test to convert numbers being passed between the Avaya SBCE and Session Manager.

To add an adaptation, under the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown). Under **Adaptation Details →General**:

- In the **Adaptation Name** field enter an informative name.
- In the **Module Name** field click on the down arrow and then select the <**click to add module**> entry from the drop down list and type **DigitConversionAdapter** in the resulting **New Module Name** field.
- **Module parameter** **MIME =no** Strips MIME message bodies on egress from Session Manager
  fromto=true Modifies from and to headers of a message



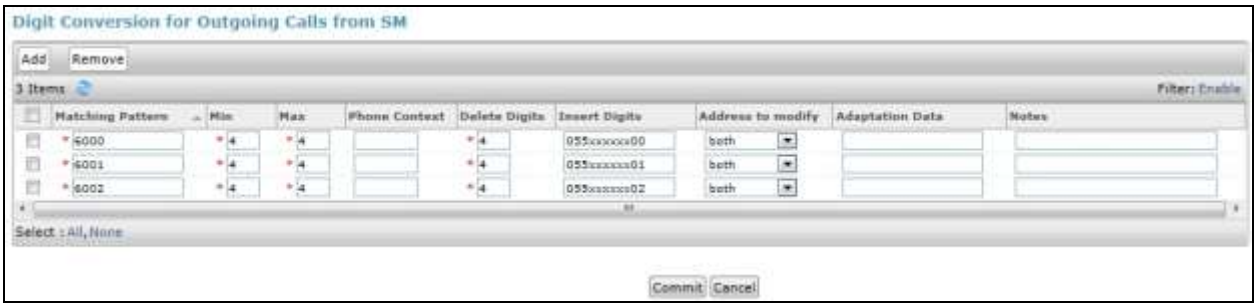
Scroll down the page and under **Digit Conversion for Incoming Calls to SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so **both** have been selected.

This will ensure any incoming numbers matching +44 will have the + digit removed before being presented to the Communication Server 1000.

In the **Digit Conversion for Outgoing Calls to SM** section, click **Add** and enter the following values.

- Enter the leading digits that will be matched in the Matching Pattern field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so **both** have been selected.



This will ensure any destination numbers beginning with 6 with have a specified CLID presented on outbound calls.

6.5. Administer SIP Entities

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

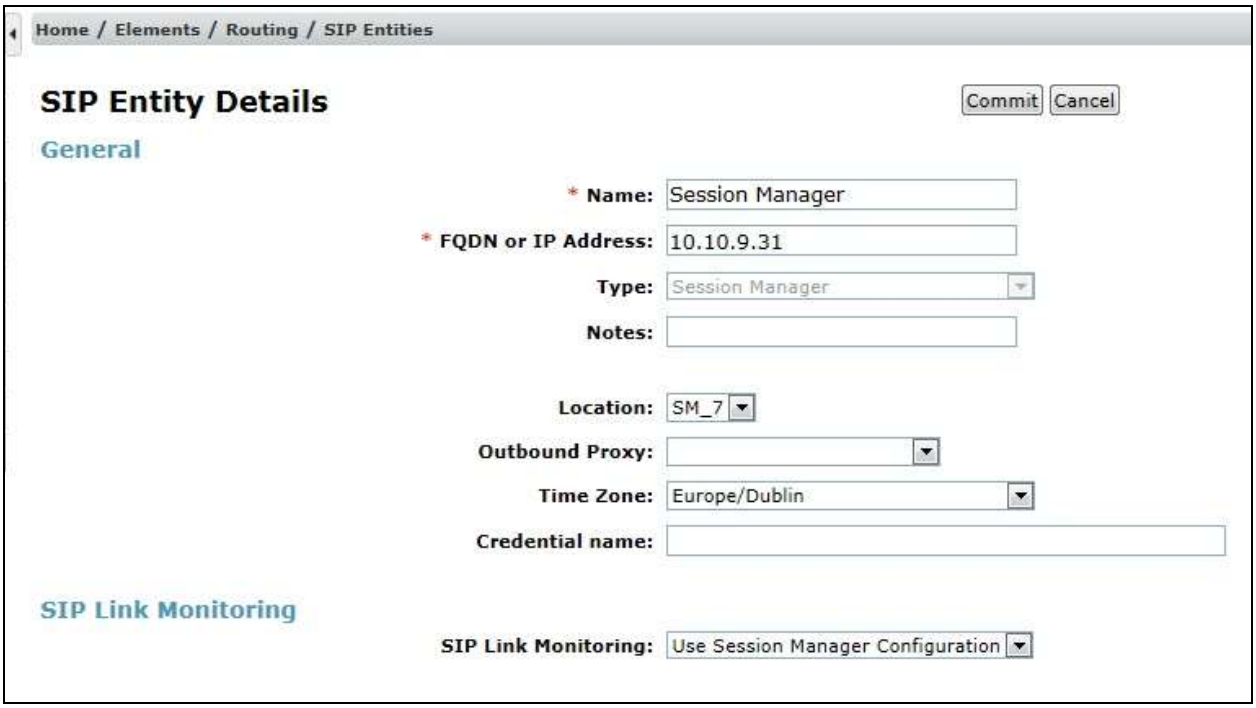
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP Entity, **Other** for a Communication Server 1000 SIP Entity and **SIP Trunk** for the Avaya SBCE SIP Entity
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are three SIP Entities.

- Session Manager SIP Entity
- Communication Server 1000 SIP Entity
- Avaya SBCE SIP Entity

6.5.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signalling interface and **Type** is **Session Manager**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time.

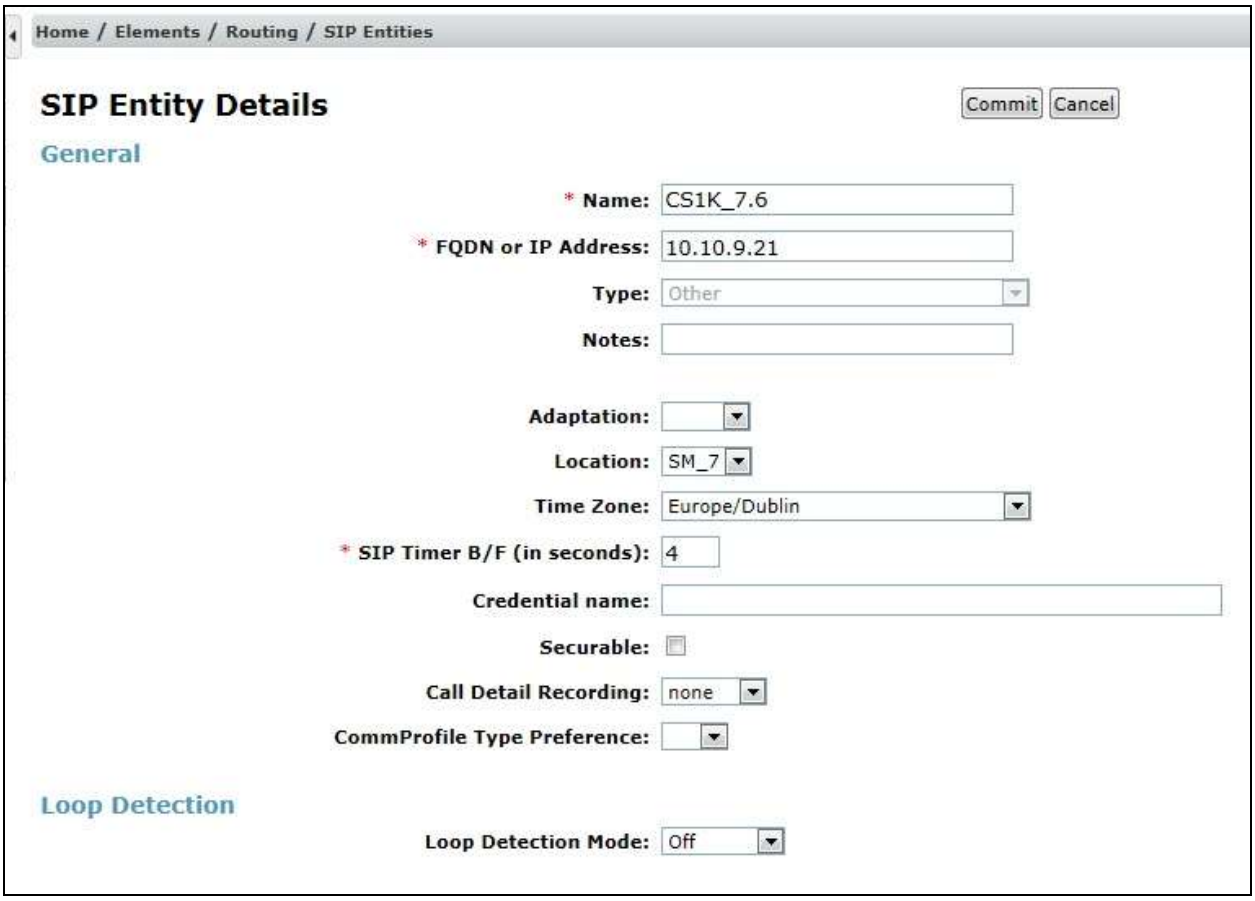


Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

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

6.5.2. Avaya Communication Server 1000 SIP Entity

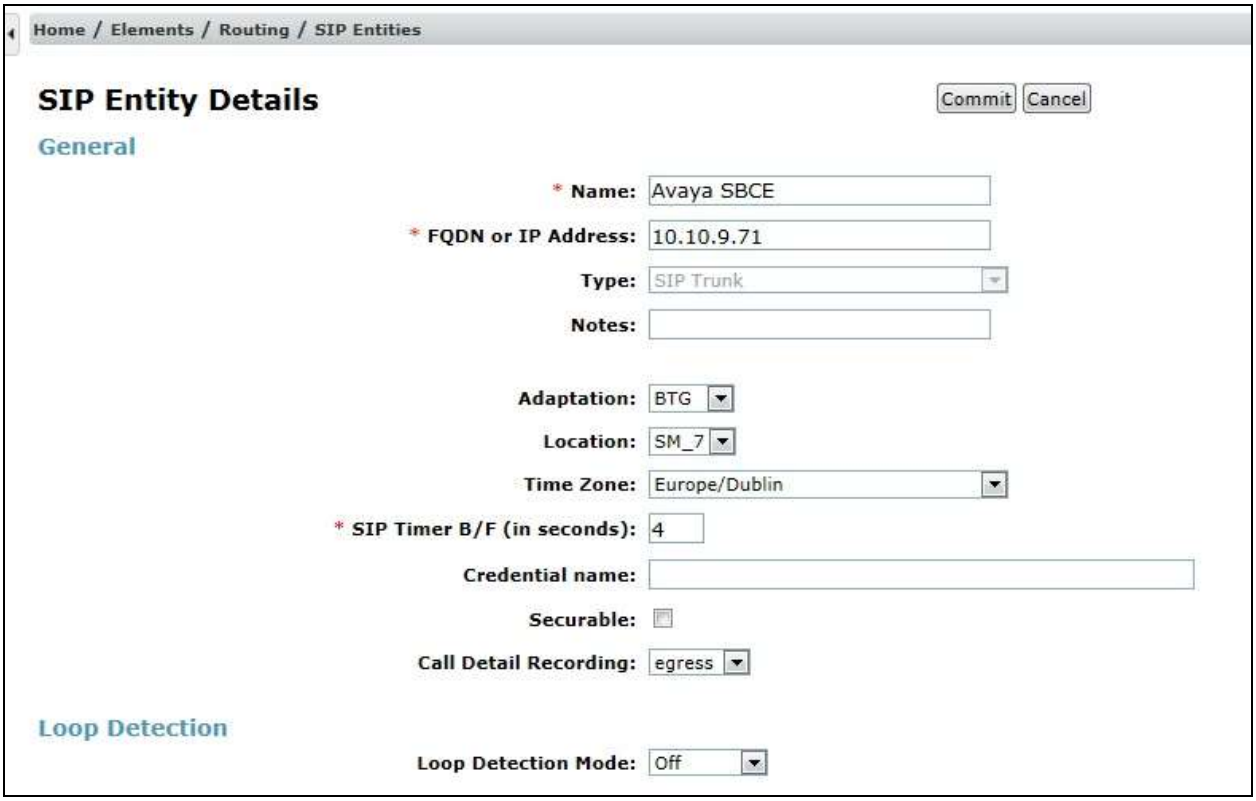
The following screen shows the SIP entity for CS1000. The **FQDN or IP Address** field is set to the IP address of the interface on CS1000 that will be providing SIP signalling and **Type** is **Other**. Set the **Location** to that defined in **Section 6.3** and the **Time Zone** to the appropriate time.



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

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP entity for the Avaya SBCE used for routing calls. The **FQDN or IP Address** field is set to the IP address of the private interfaces administered in **Section 7** of this document. Set the location to that defined in **Section 6.3**, set **Adaptation** to one created in **Section 6.4** and the **Time Zone** to the appropriate time zone.

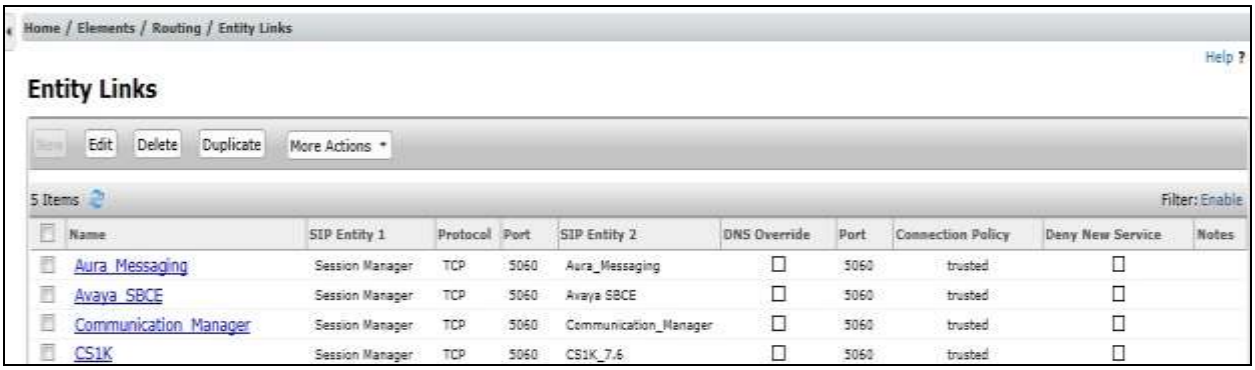


6.6. Administer Entity Links

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

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select **Session Manager**.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select **Trusted** from the drop-down menu to make the other system trusted.

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



Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
Aura_Messaging	Session Manager	TCP	5060	Aura_Messaging	☐	5060	trusted	☐	
Avaya_SBCE	Session Manager	TCP	5060	Avaya SBCE	☐	5060	trusted	☐	
Communication_Manager	Session Manager	TCP	5060	Communication_Manager	☐	5060	trusted	☐	
CS1K	Session Manager	TCP	5060	CS1K_7.6	☐	5060	trusted	☐	

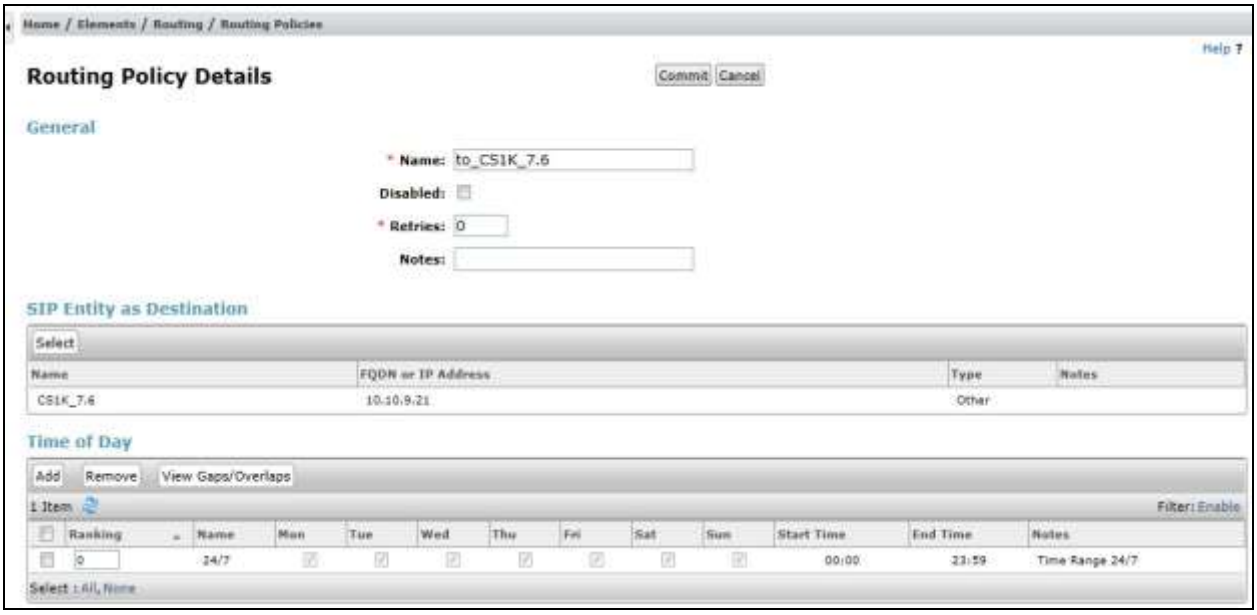
6.7. Administer Routing Policies

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

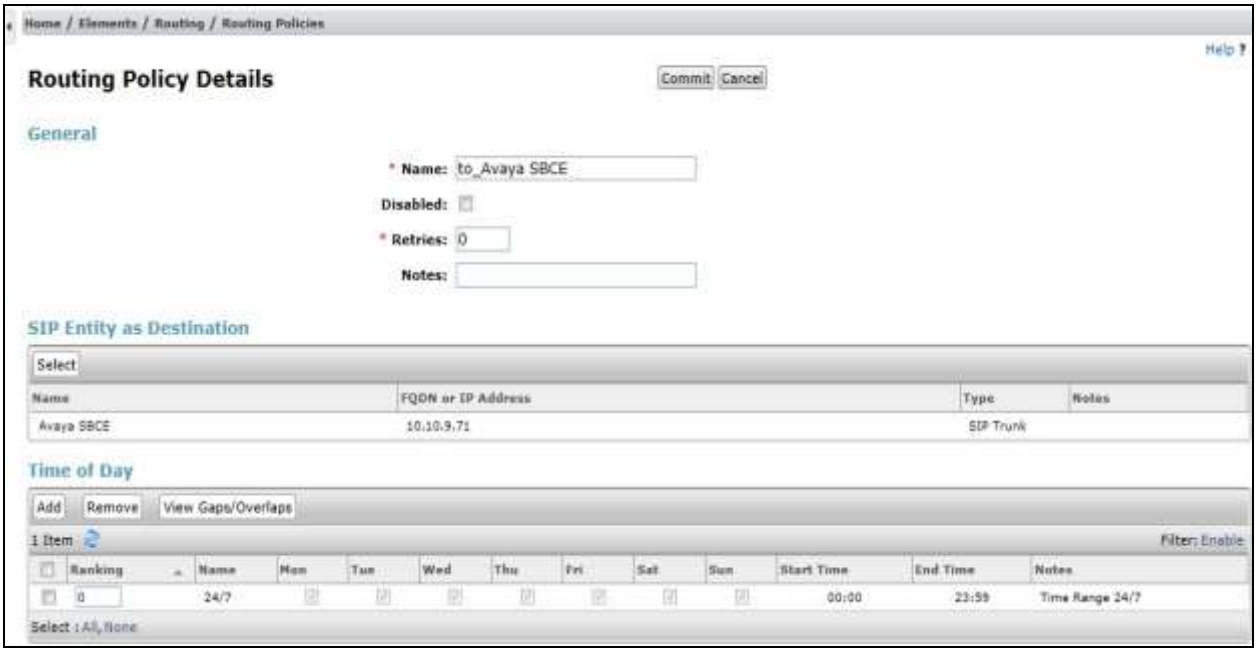
Under **General**:

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

The following screen shows the routing policy for CS1000.



The following screen shows the Routing Policy for the Avaya SBCE.



6.8. Administer Dial Patterns

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

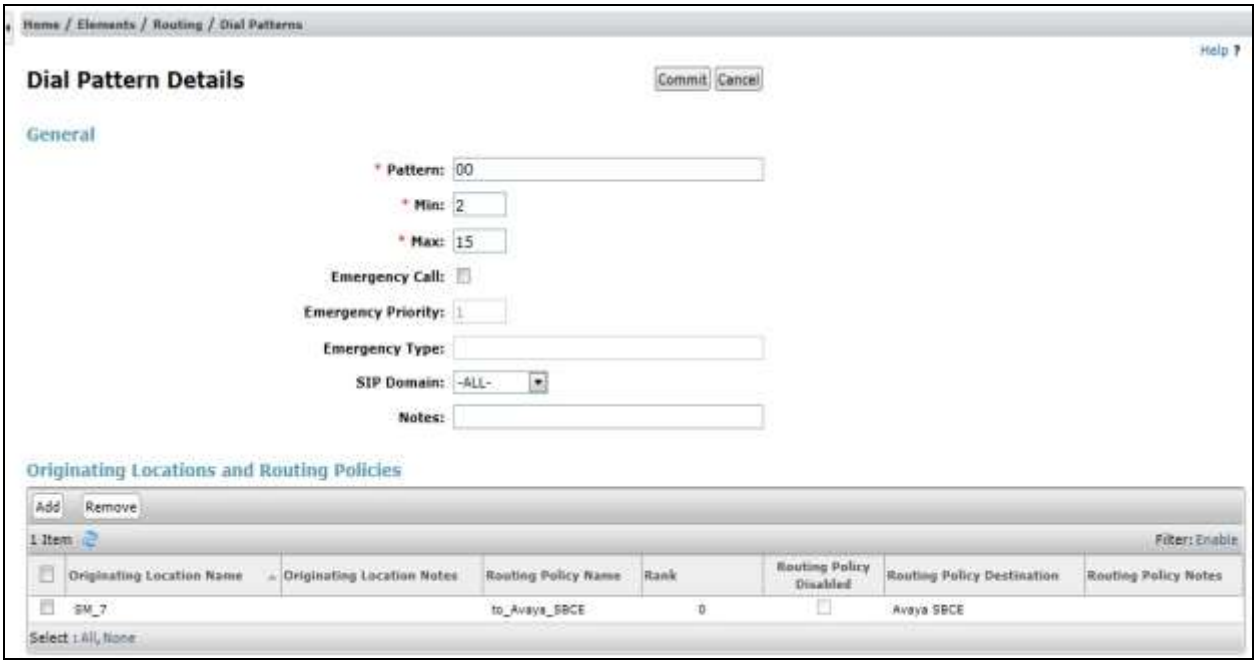
Under **General**:

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

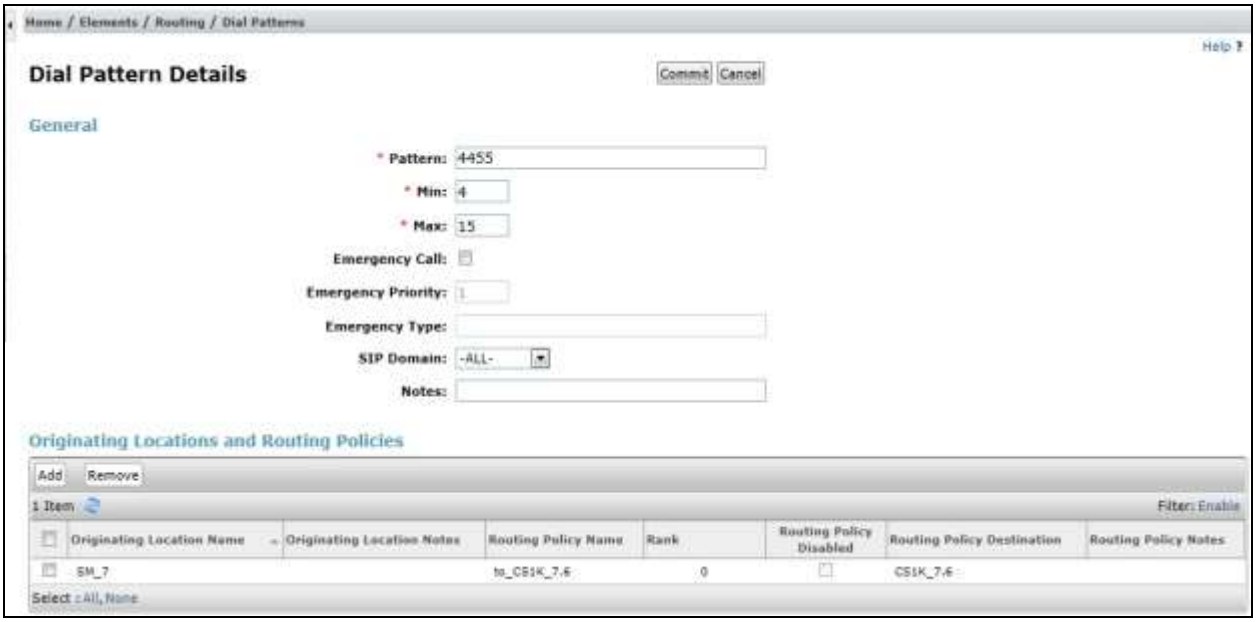
Under **Originating Locations and Routing Policies**:

- Click **Add**, in the resulting screen (not shown).
- Under **Originating Location**, select the location defined in **Section 6.3** or **ALL**.
- Under **Routing Policies** select one of the routing policies defined in **Section 6.6**.
- Click **Select** button to save.

The following screen shows an example dial pattern configured for the Avaya SBCE.



The following screen shows the test dial pattern configured for CS1000.

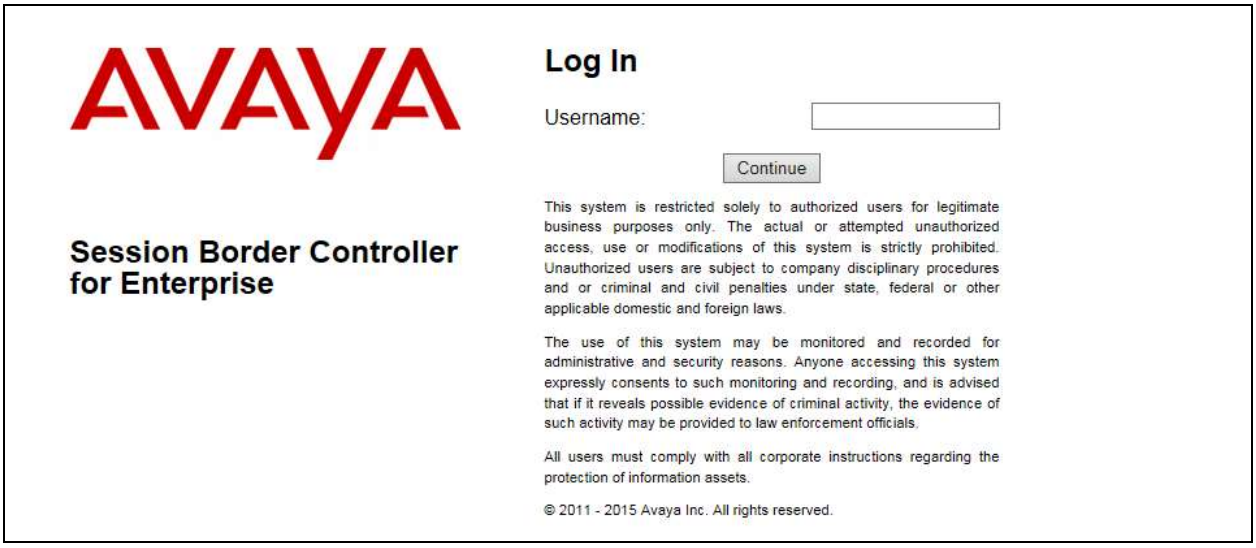


7. Configure Avaya Session Border Controller for Enterprise

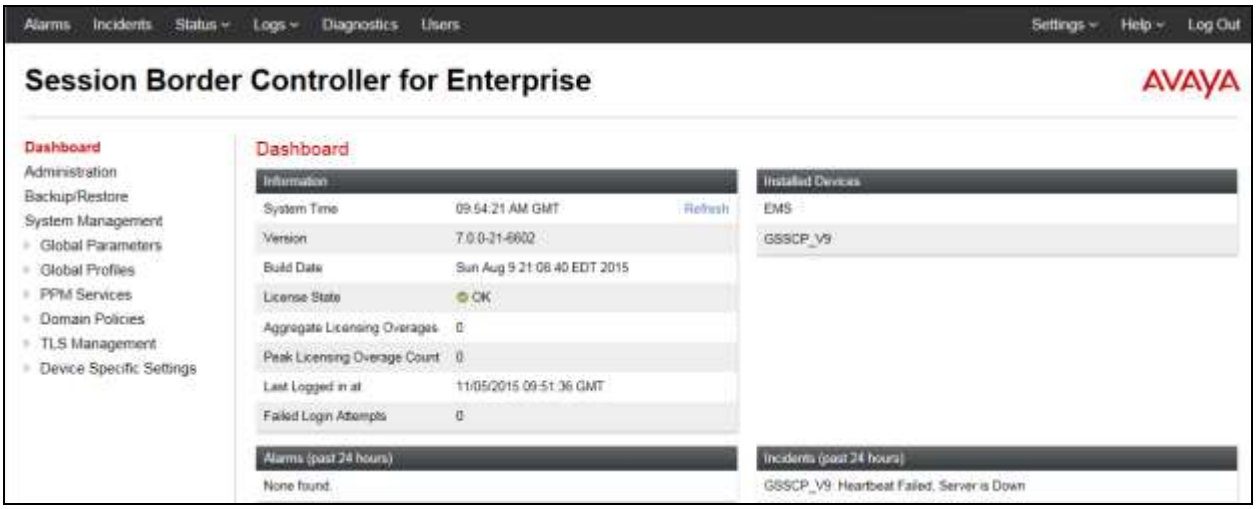
This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's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

7.1. Access Avaya Session Border Controller for Enterprise

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



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.



7.2. Define Network Management

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

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interface in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the external 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 in the IP Address field and leave the Public IP and Gateway Override fields blank.
- Click on **Finish** to complete the interface definition.

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

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interface in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal 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 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:

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

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

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

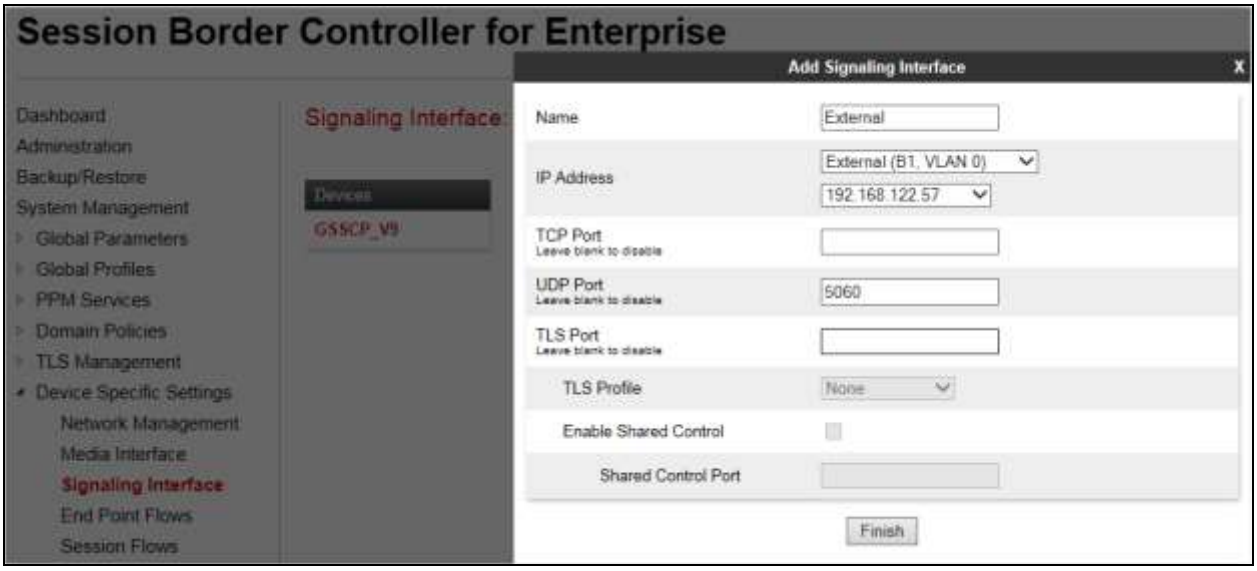
7.3. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces. Testing was carried out with TCP used for transport of signalling between the Session Manager and the Avaya SBCE, and UDP for transport of signalling between the Avaya SBCE and the BT Global Services SIP Trunk. This document shows the configuration for TCP and UDP, if additional security is required, it's recommended to use TLS and port 5061.

7.3.1. Signalling Interfaces

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

- Select **Add** and enter details of the external signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external signalling interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was a single IP address **192.168.122.57**.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the BT Global Services SIP Trunk.



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

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

The following screenshot shows details of the signalling interfaces:

Note. In the test environment, the internal IP address was **10.10.9.71**.

7.3.2. Media Interfaces

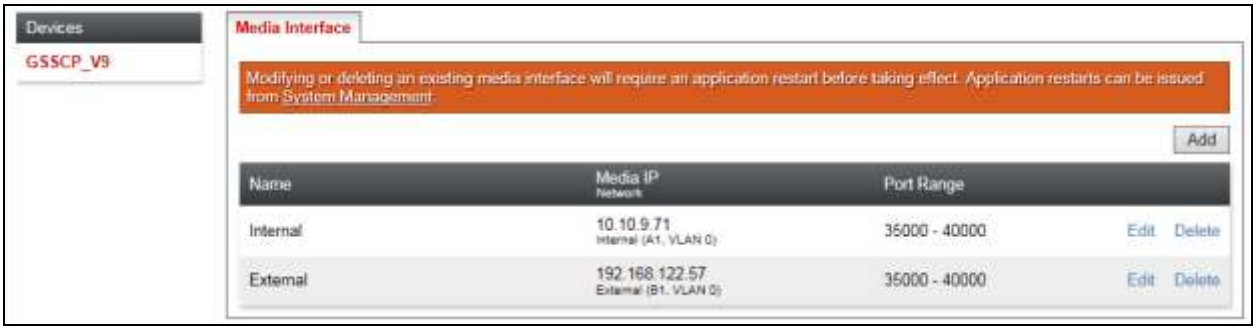
To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** in the main menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those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

- Select **Add** and enter details of the external media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 7.2**. In the test environment, this was a single IP address **192.168.122.57**.
- Define the RTP **Port Range** for the media path with BT Global Services SIP Trunk, during testing this was left at the default values.

The internal media interface is 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.

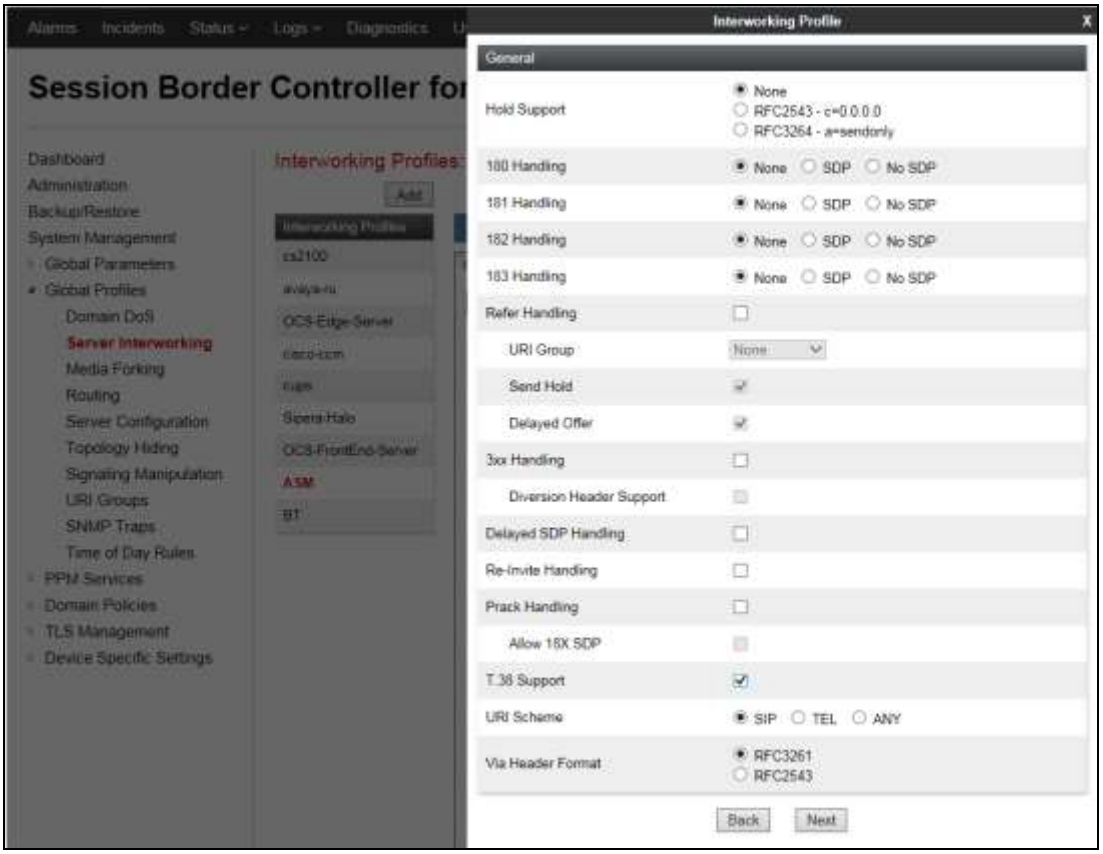
The following screenshot shows details of the media interfaces:



7.4. Define Server Interworking

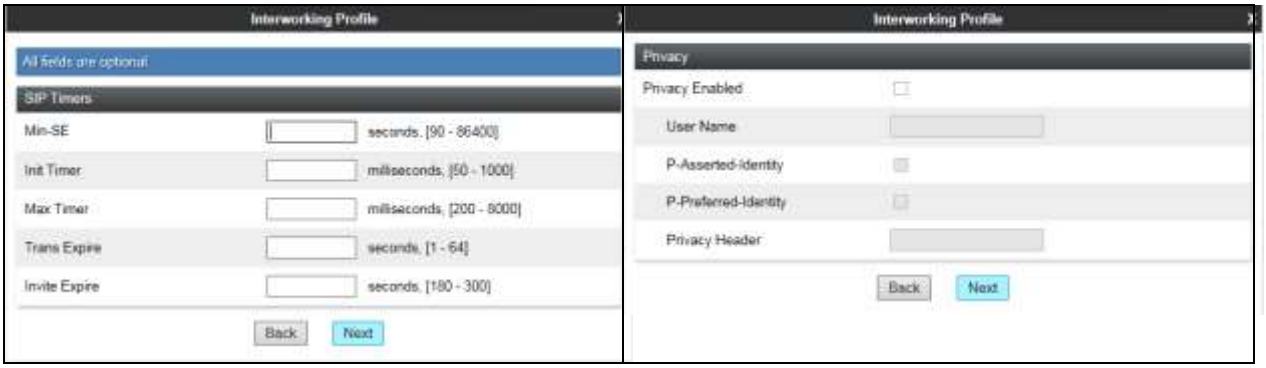
Server interworking is defined for each server connected to the Avaya SBCE. In this case, BT Global Services SIP Trunk is connected as the Trunk Server and the Session Manager is connected as the Call Server.

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 Session Manager, click on **Add** (not shown). A pop-up menu (not shown) is generated. In the **Name** field enter a descriptive name for the Session Manager and click **Next**.

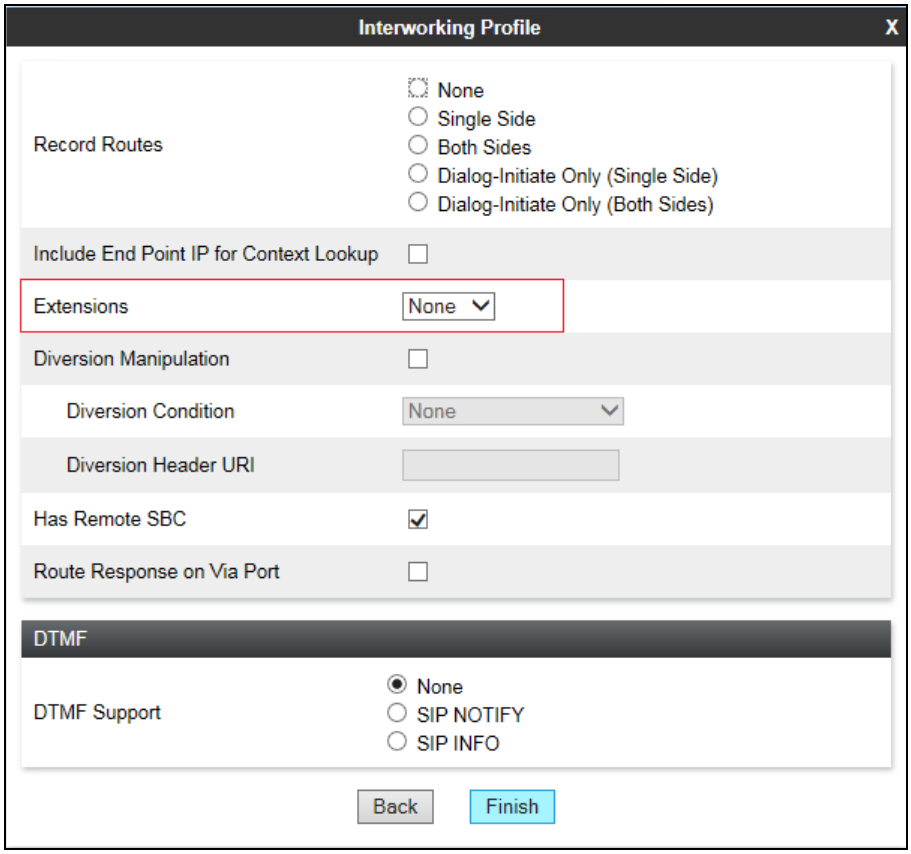


Configuration of interworking includes Hold support, T.38 fax support and SIP extensions.

- In the General dialogue box shown in the previous screenshot, check the **T.38 Support** box. During testing, the rest of the parameters were left at default values.
- Click on **Next** and **Next** again to go through the next two dialogue boxes. During testing, these were left at default values.



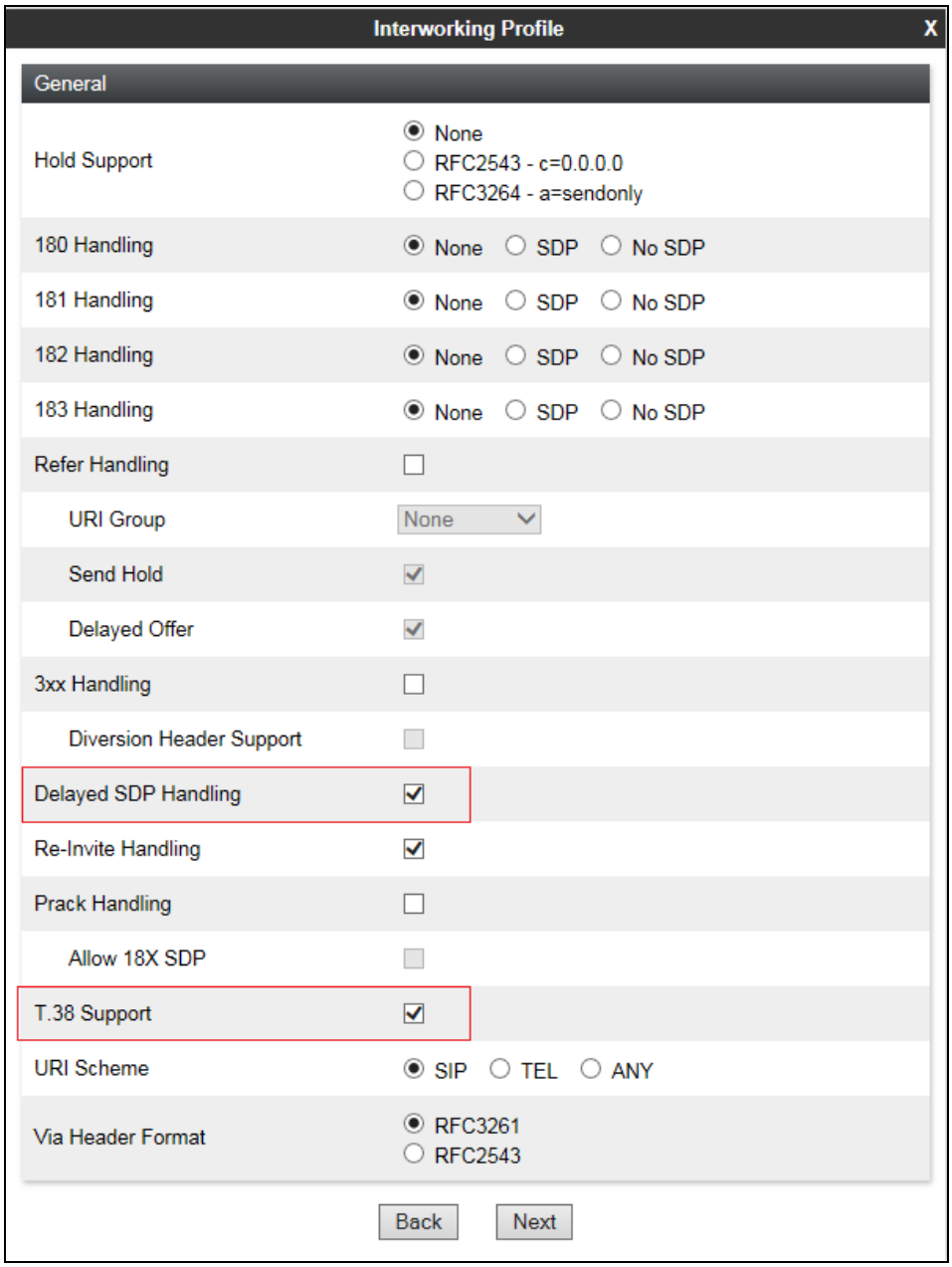
In the final dialogue box, select **None** from the **Extensions** box. And click on **Finish**



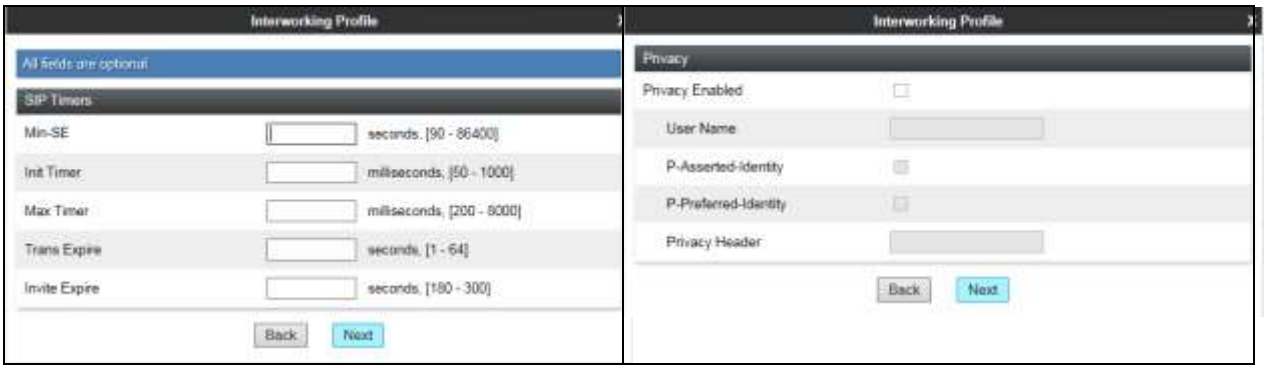
To define Server Interworking for BT Global Services SIP Trunk, click on **Add** (not shown). A pop-up menu (not shown) is generated. In the **Name** field enter a descriptive name for the BT Global Services SIP Trunk and click **Next**.

In the dialogue bow that appears, settings are as follows:

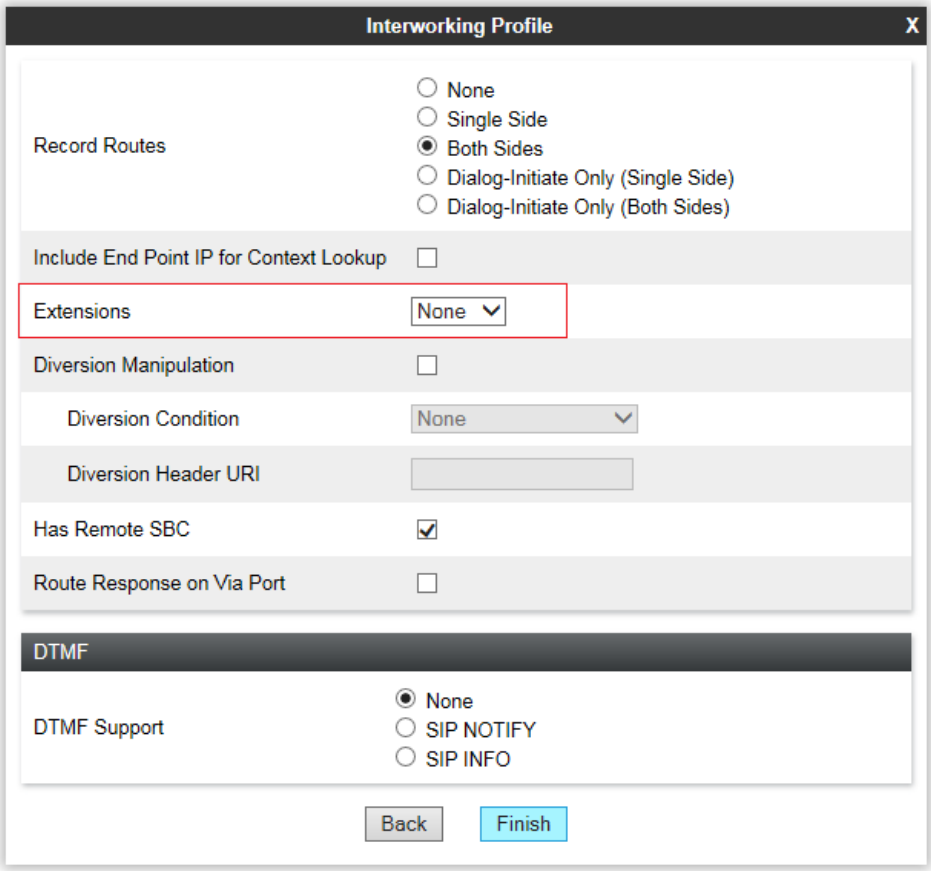
- Check the **Delayed SDP Handling** box. This inserts an SDP into the empty INVITE sent by the CS1000 when shuffling.
- Check the **T.38** box



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



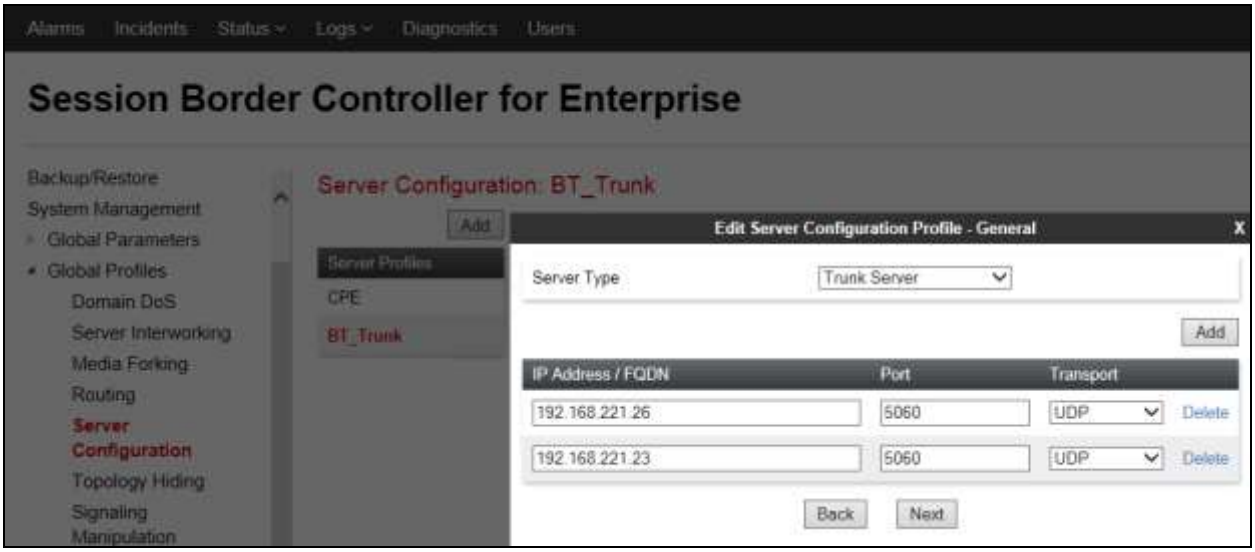
In the final dialogue box, select **None** from the **Extensions** box and click on **Finish**.



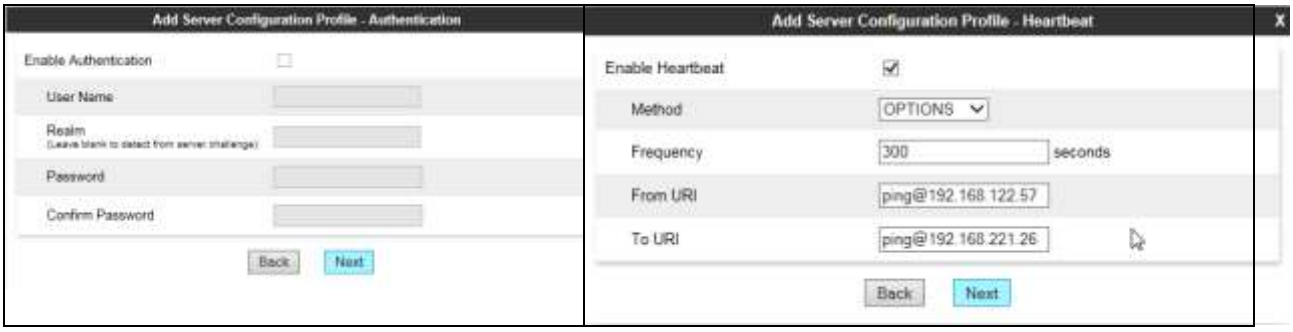
7.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. In this case, BT Global Services SIP Trunk is connected as the Trunk Server and the Session Manager is connected as the Call Server. To define the BT Global Services SIP Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu (not shown). 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 first BT Global Services network SBC interface address.
- In the **Port** box, enter the port to be used for the SIP Trunk. This was left blank during testing which defaults to 5060 when UDP is used for transport.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Add** and repeat the above for the alternative network SBC. Click on **Next**.



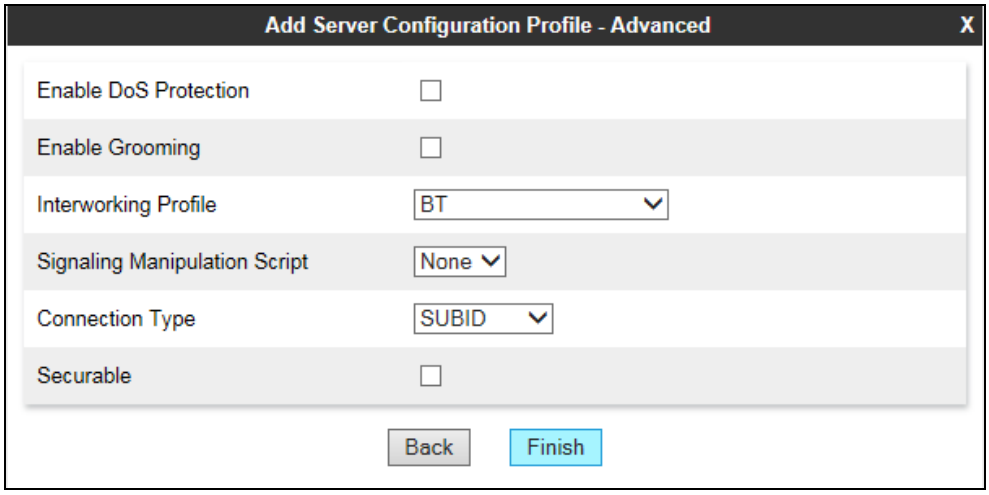
- Click on **Next** and **Next** again to go through the next two dialogue boxes. During testing, these were left at default values.



Note: Although the Heartbeat configuration was left at default values for most of the testing, the screenshot shows values used when verifying the SIP Trunk. For details, refer to **Section 9**.

The final dialogue box is the **Advanced** settings:

- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the BT Global Services SIP Trunk defined in **Section 7.4**.
- Click **Finish**.



BT Global Services use two network SBCs for resilience. A separate Trunk Server configuration is required for the alternative SBCs. Repeat the above process using the IP address of the alternative SBC, in the test environment this was 192.168.221.23.

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

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box (not shown).
- Ensure that the Interworking Profile defined for the Session Manager in **Section 7.4** is selected in the **Interworking Profile** drop down menu in the Advanced dialogue box (not shown).

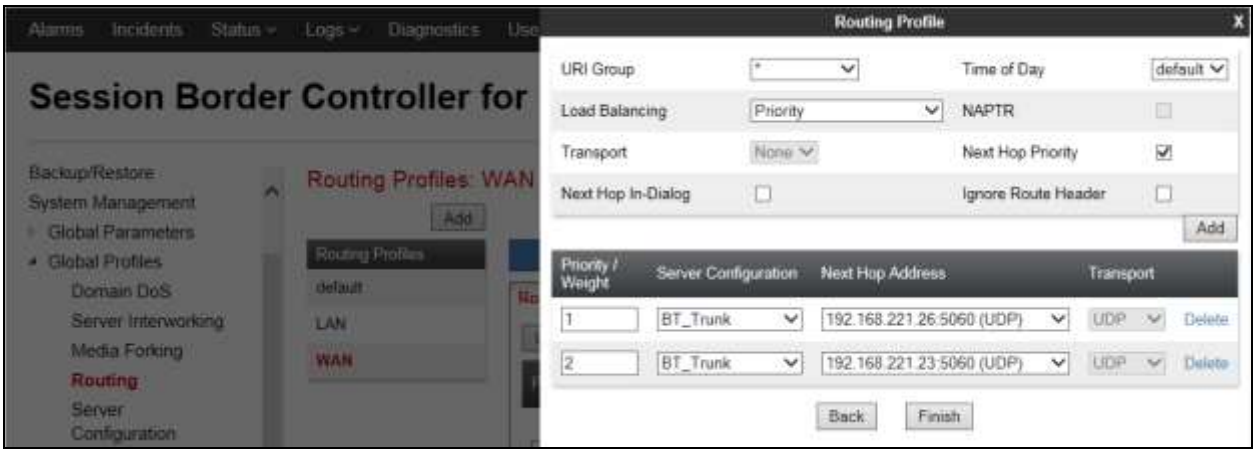
The following screenshot shows the completed entry for the Session Manager:

7.6. Define Routing

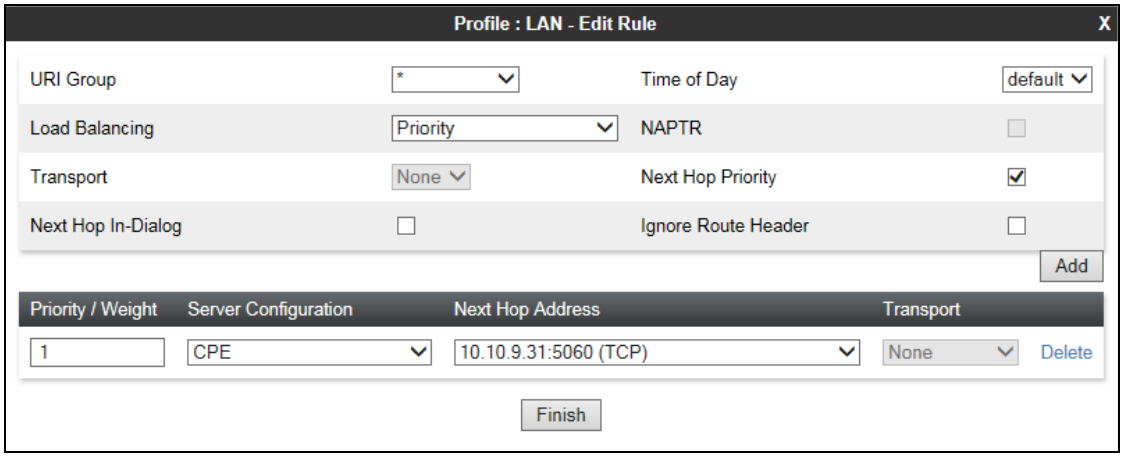
Routing information is required for routing to BT Global SIP Trunk on the external side and the Session Manager on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling.

To define routing to BT Global Service SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box (not shown), click on Next and enter details for the Routing Profile:

- In the **Load Balancing** drop down menu, select the method of load balancing required. During testing this was set to **Priority**. If an even distribution across the network SBCs is required, **Round Robin** could be used.
- Click on **Add** to specify an IP address for the first network SBC.
- Assign a priority in the **Priority / Weight** field
- Select the Server Configuration defined in **Section 7.5** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field
- Repeat for the alternative network SBC. Click **Finish**.



Repeat the above process for the Routing Profile for the Session Manager:



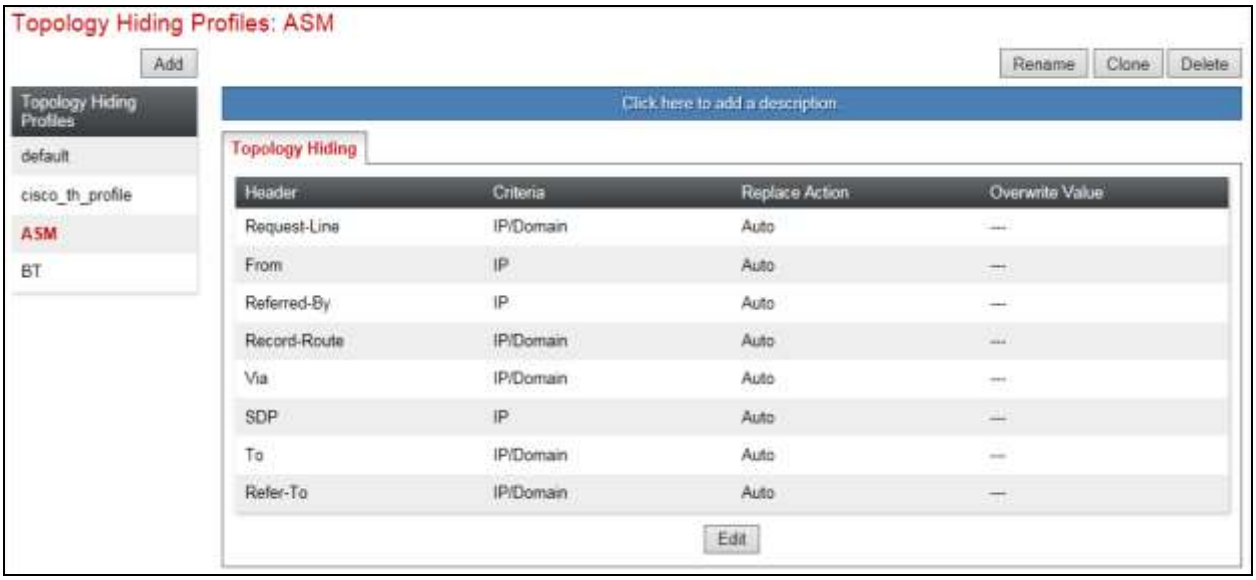
7.7. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop or external interfaces. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise, particularly from the Session Manager. 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 BT Global Service SIP Trunk, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for BT Global Service SIP Trunk and click **Next**.
- Click on **Add Header** and **s**elect from the **Header** drop down menu.
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu depending on requirements.
- Leave the **Replace Action** at the default value of **Auto** unless a specific domain name is required. In this case, select **Overwrite** and define a domain name in the **Overwrite Value** field.
- Topology hiding was defined for all headers where the function is available.

To define Topology hiding for the Session Manager, follow the same process. This can be simplified by cloning the profile defined for BT Global Service SIP Trunk. Do this by highlighting the profile defined for the Session Manager and clicking on **Clone**.
Enter an appropriate name for the Session Manager and click on Next. Make any changes where required, in the test environment the settings were left at the same values.

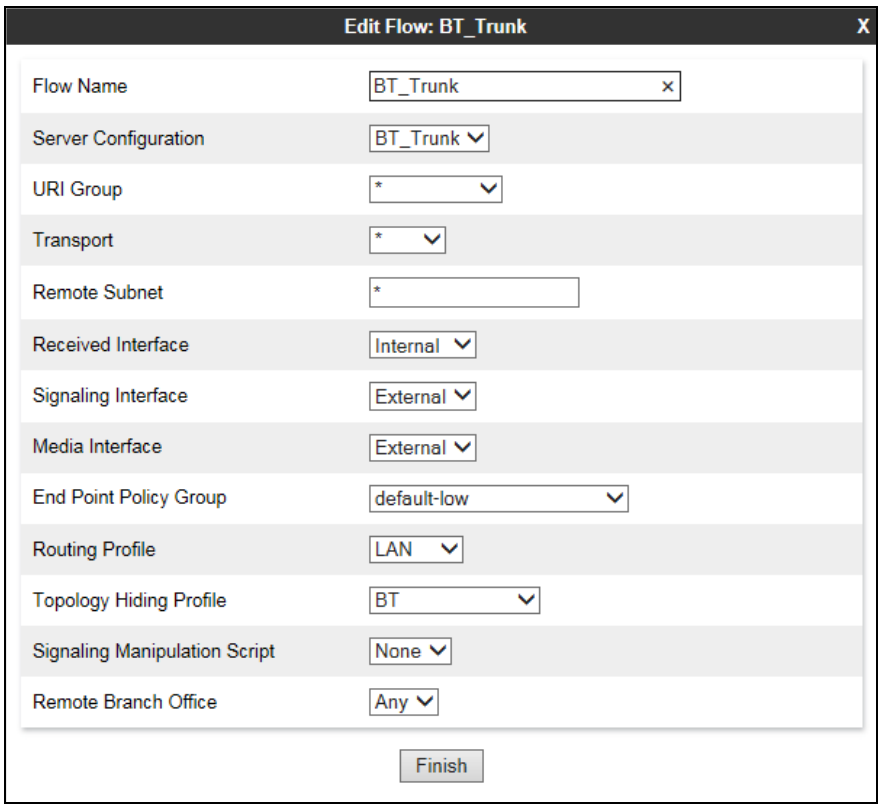


7.8. Server Flows

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

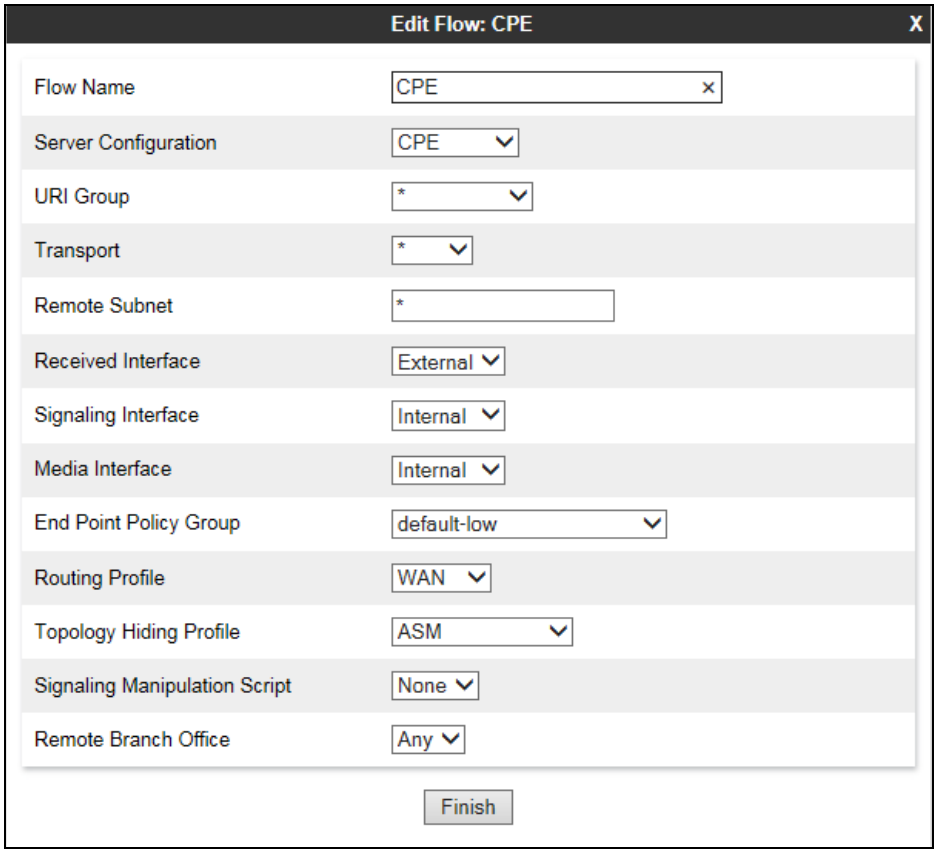
To define a Server Flow for the BT Global Services SIP Trunk, navigate to **Device Specific Settings → End Point Flows**.

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

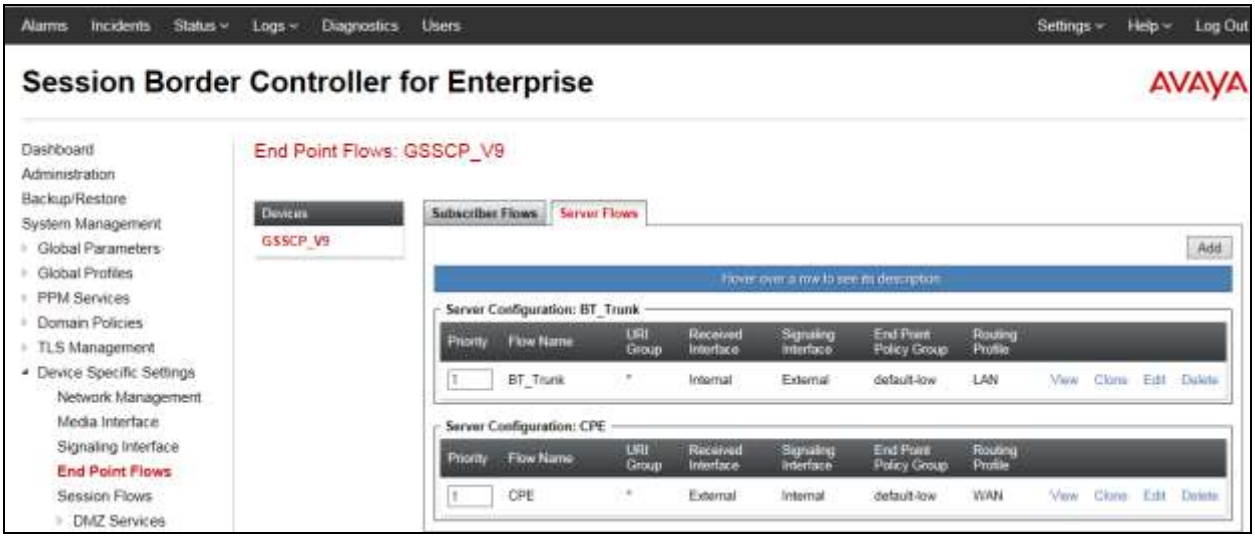


To define a Server Flow for the Session Manager, navigate to **Device Specific Settings → End Point Flows**.

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



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



8. Configure BT SIP Trunk Equipment

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

9. Verification Steps

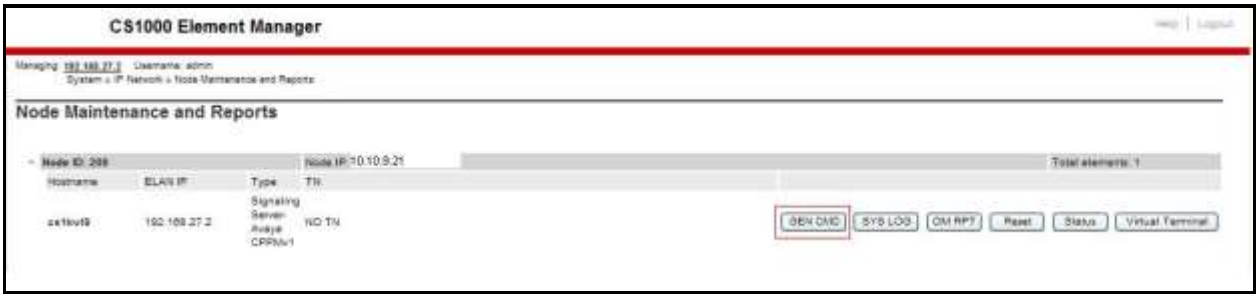
This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

9.1. Avaya Communication Server 1000 Verification

This section illustrates sample verifications that may be performed using the Avaya CS1000 Element Manager GUI.

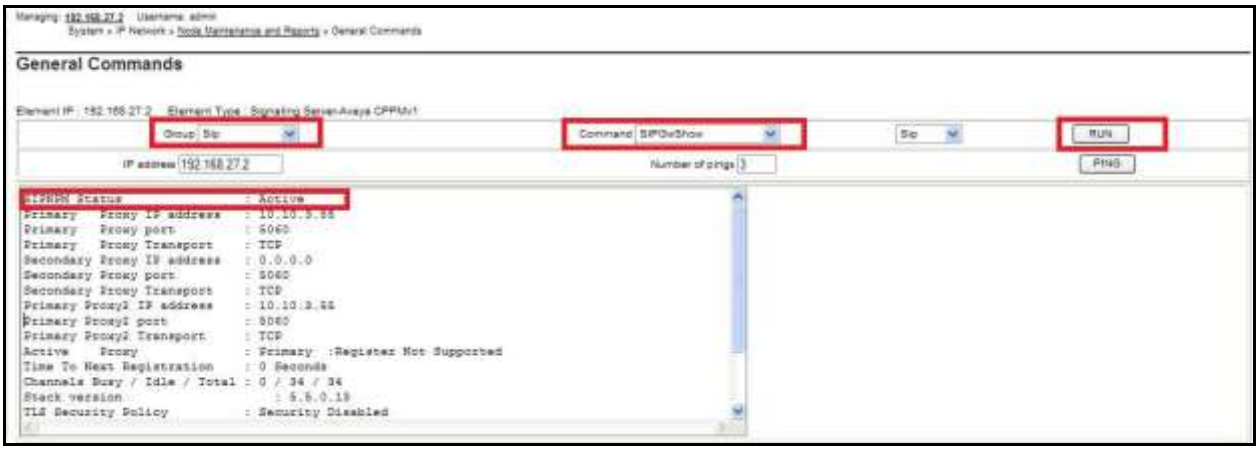
9.1.1. IP Network Maintenance and Reports Commands

From Element Manager, navigate to **System → IP Network → Maintenance and Reports** as shown below. In the resultant screen on the right, click the **Gen CMD** button.

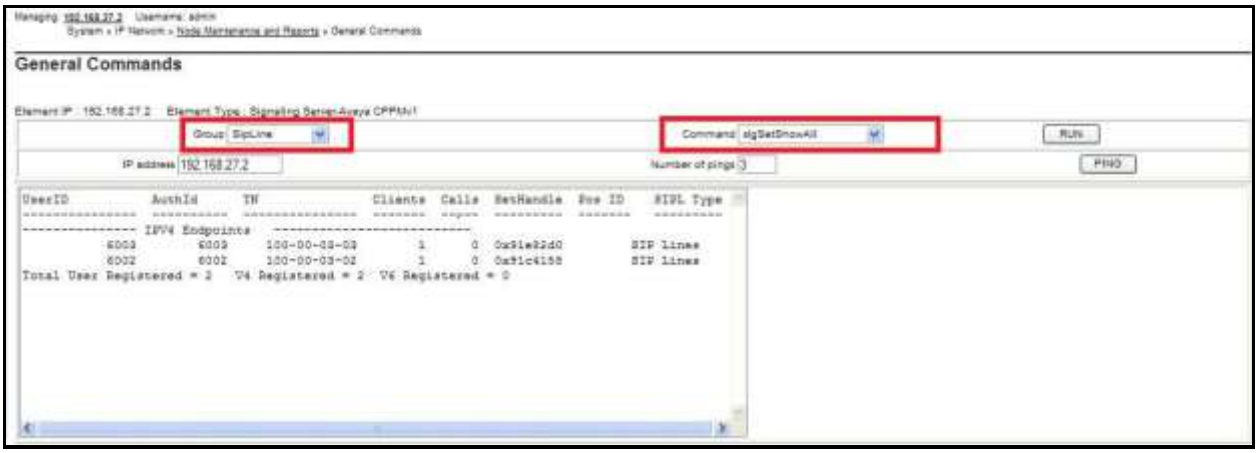


The **General Commands** page is displayed. A variety of commands are available by selecting an appropriate Group and Command from the drop-down menus, and selecting **Run**.

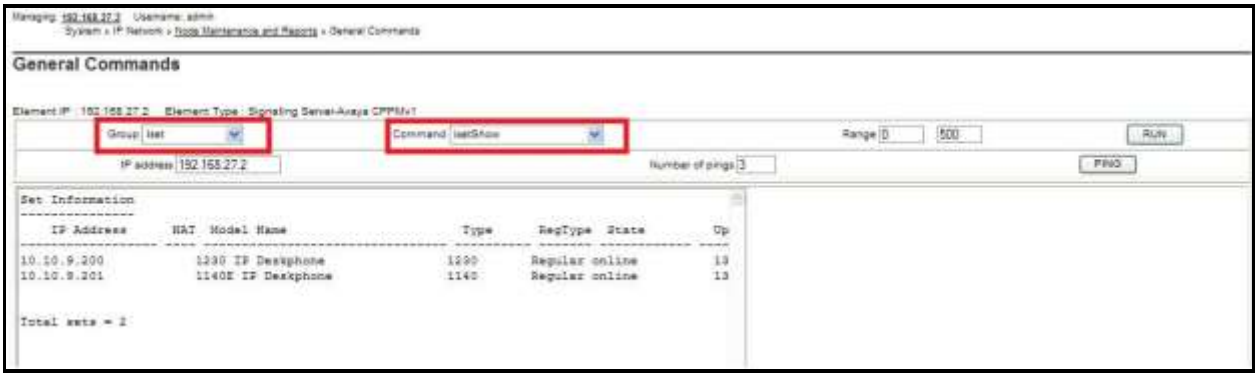
To check the status of the SIP Gateway to Session Manager in the sample configuration, select **Sip** from the Group menu and **SIPGwShow** from the **Command** menu. Click **Run**. The example output below shows that Session Manager has **SIPNPM Status** "**Active**".



The following screen shows a means to view registered SIP telephones. The screen shows the output of the **Command sigSetShowAll** in **Group SipLine**.



The following screen shows a means to view IP UNIStim telephones. The screen shows the output of the **Command isetShow** in **Group Iset**.

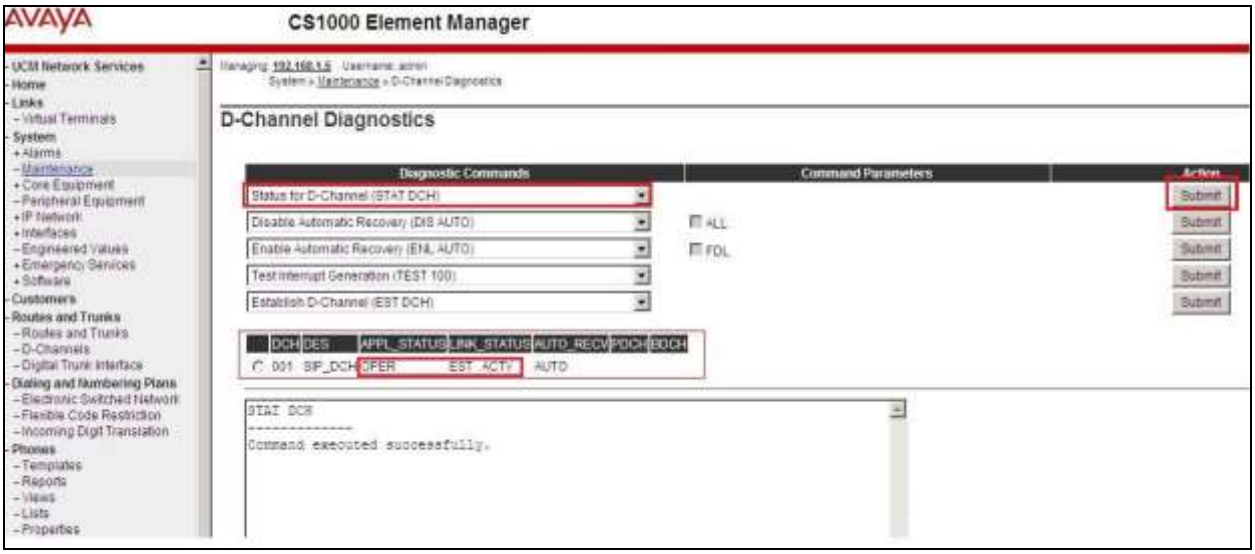


9.2. Verify Avaya Communication Server 1000 Operational Status

Expand **System** on the left navigation panel and select **Maintenance.** Select **LD 96 - D-Channel** from the **Select by Overlay** table and the **D-Channel Diagnostics** function from the **Select by Functionality** table as shown below.

Select **Status for D-Channel (STAT DCH)** command and click **Submit** to verify status of virtual D-Channel as shown below. Verify the status of the following fields.

- **APPL_STATUS** Verify status is **OPER**
- **LINK_STATUS** Verify status is **EST ACTV**

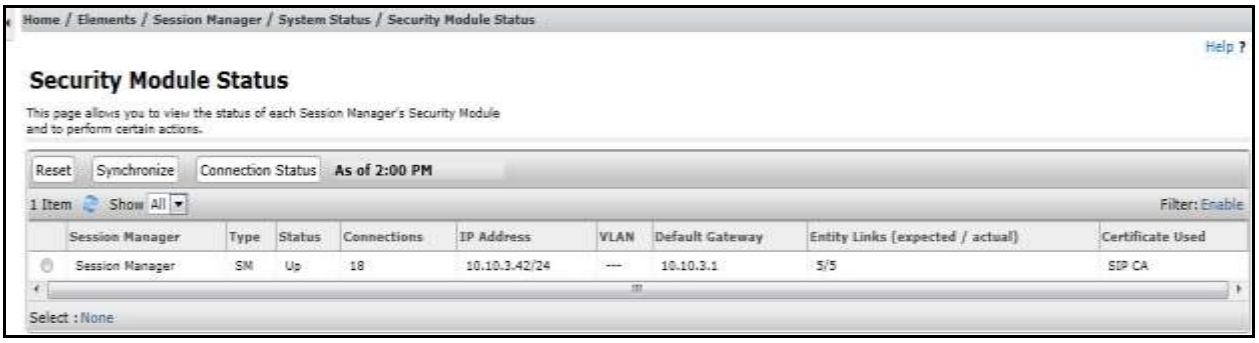


9.3. Verify Avaya Aura® Session Manager Operational Status

9.3.1. Verify Avaya Aura® Session Manager is Operational

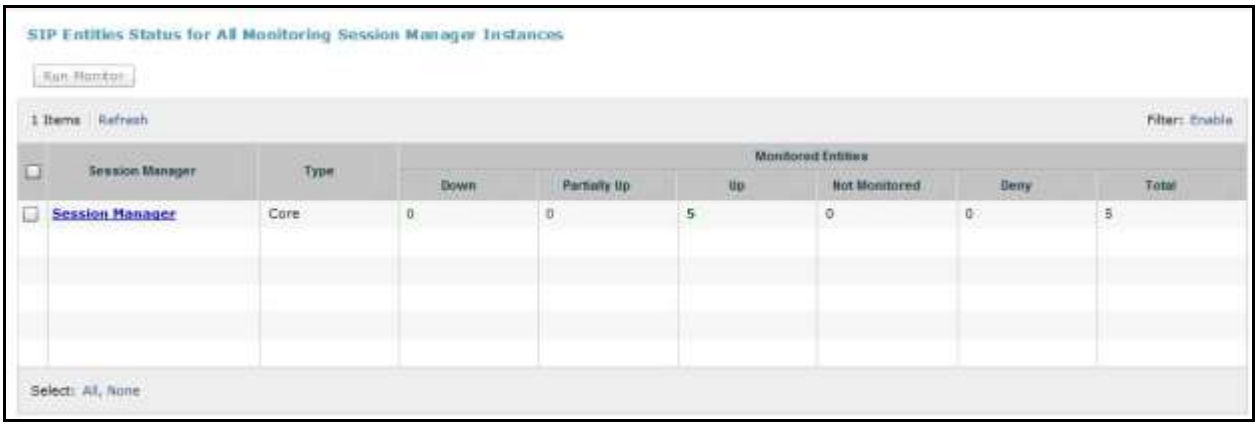
Navigate to **Elements → Session Manager → Dashboard** (not shown) to verify the overall system status for Session Manager. Specifically, verify the status of the following fields as shown below.

Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.



9.3.2. Verify SIP Entity Link Status

Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links. Select the SIP Entity for CS1000 from the **All Monitored SIP Entities** table (not shown) to open the **SIP Entity, Entity Link Connection Status** page.



Verify the status of the SIP link is up between Session Manager and CS1000 by going through the same process as outlined above but selecting the SIP Entity for the Avaya SBCE in the **All Monitored SIP Entities:** table.

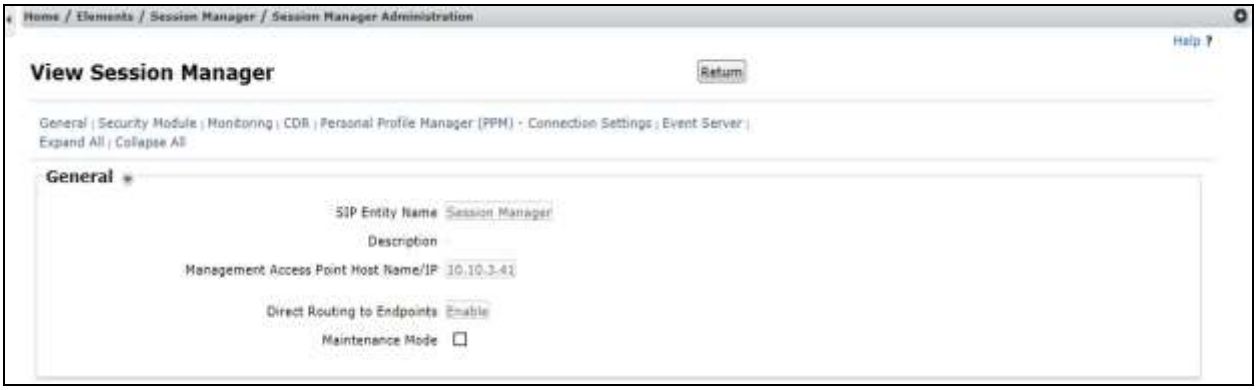
9.3.3. Verify Avaya Aura® Session Manager Instance

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

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

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

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



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

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

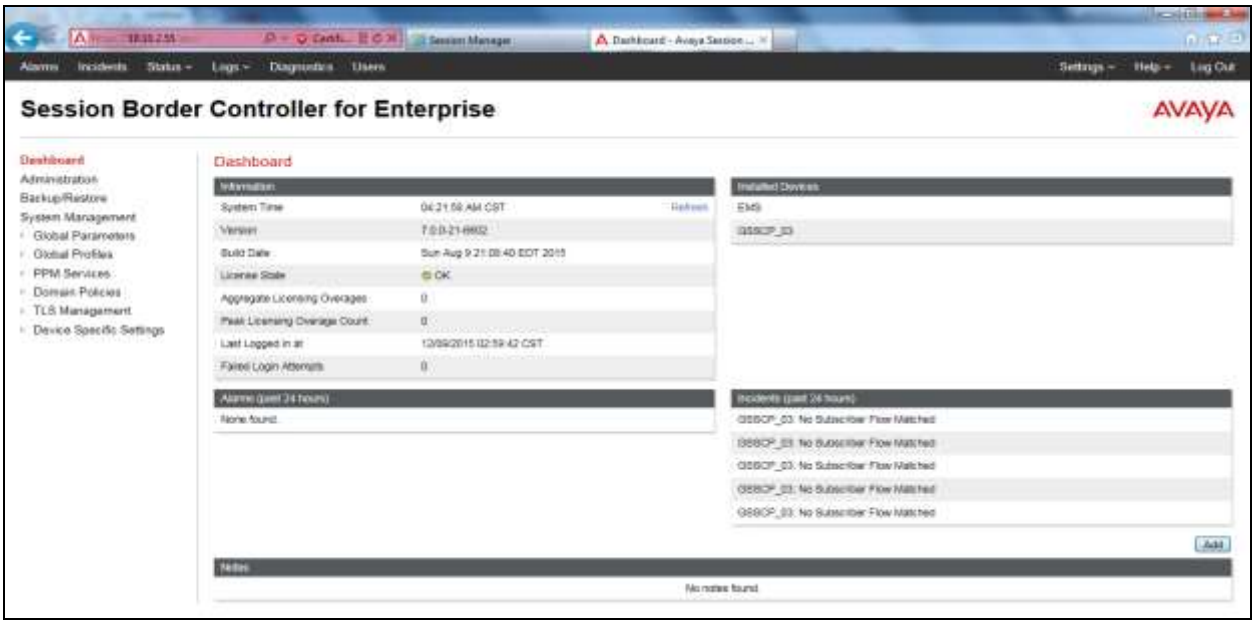
Use default values for the remaining fields. Click **Save** (not shown). The following screen shows the remaining Session Manager values used for the compliance test.

9.4. Avaya Session Boarder Controller for Enterprise Verification

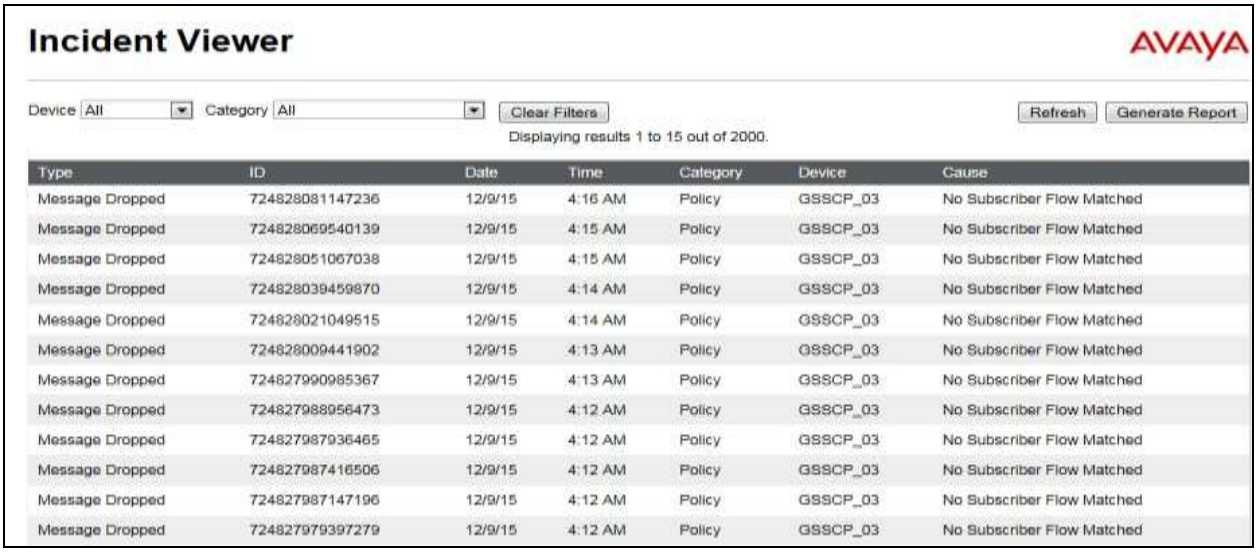
This section contains verification steps that may be performed using the Avaya Session Border Controller for Enterprise.

9.4.1. Incidents

The Incidents Log Viewer display alerts captured by the Avaya SBCE. Select the **Incidents** link along the top of the screen.



The following screen shows example SIP messages that do not match a Server Flow for an incoming message.



Type	ID	Date	Time	Category	Device	Cause
Message Dropped	724828081147236	12/9/15	4:16 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724828069540139	12/9/15	4:15 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724828051067038	12/9/15	4:15 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724828039459870	12/9/15	4:14 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724828021049515	12/9/15	4:14 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724828009441902	12/9/15	4:13 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827990985367	12/9/15	4:13 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827988956473	12/9/15	4:12 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827987936465	12/9/15	4:12 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827987416506	12/9/15	4:12 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827987147196	12/9/15	4:12 AM	Policy	GSSCP_03	No Subscriber Flow Matched
Message Dropped	724827979397279	12/9/15	4:12 AM	Policy	GSSCP_03	No Subscriber Flow Matched

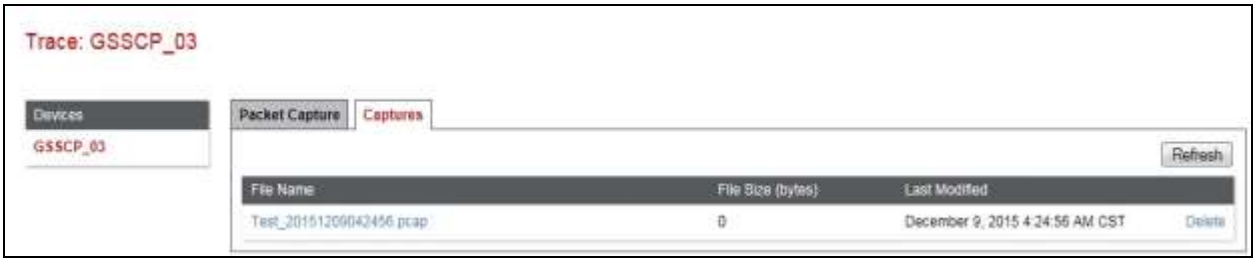
9.4.2. Trace Settings

The Trace Settings tool is for configuring and displaying call traces and packet captures for the Avaya SBCE.

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

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



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

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server R7.65, Avaya Aura® Session Manager R7.0 and Avaya Session Border Controller for Enterprise R7.0 to BT Global Services SIP Trunk. BT Global Services SIP Trunk is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. Additional References

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

[1] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0, Nov 2015.

[2] *Upgrading and Migrating Avaya Aura® applications to 7.0*, Release 7.0, Nov 2015.

[3] *Deploying Avaya Aura® applications*, Release 7.0, Oct 2015

[4] *Deploying Avaya Aura® System Manager* Release 7.0 Nov 2015

[5] *Upgrading Avaya Aura® System Manager to Release 7.0*, Nov 2015.

[6] *Administering Avaya Aura® System Manager for Release 7.0* Release 7.0, Nov 2015

[7] *Deploying Avaya Aura® Session Manager on VMware* , Release 7.0 August 2015

[8] *Upgrading Avaya Aura® Session Manager* Release 7.0, August 2015

[9] *Administering Avaya Aura® Session Manager* Release 7.0, August 2015

[10] *Avaya Communication Server 1000 Installation and Commissioning*, Document Number NN43041-310

[11] *Linux Platform Base and Applications Installation and Commissioning Avaya Communication Server 1000*, Document Number NN43001-315

[12] *Software Input Output Reference – Maintenance Avaya Communication Server 1000*, Document Number NN43001-711

[13] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[15] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Nov 2015

[16] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

Appendix A – Communication Server 1000 Software

Communication Server 1000 call server patches and plug ins

```
TID: 46379

VERSION 4121

System type is - Communication Server 1000/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:               1
IPMGs Unregistered:             0
IPMGs Configured/unregistered:  0

RELEASE 7
ISSUE 65 P  +
IDLE_SET_DISPLAY NORTEL
DepList 1: core Issue: 01(created: 2015-09-28 04:19:50 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2015-11-12 14:50:17(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2013-09-28 04:30:29(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE


LOADWARE VERSION: PSWV 100+

INSTALLED LOADWARE PEPS : 1
PAT#  CR #            PATCH REF #     NAME            DATE        FILENAME
00    wi01057886      ISS1:1OF1       DSP2AB07        13/09/2013  DSP2AB07.LW


ENABLED PLUGINS : 2

PLUGIN    STATUS      PRS/CR_NUM    MPLR_NUM      DESCRIPTION
-------------------------------------------------------------
201       ENABLED     Q00424053     MPLR08139     PI:Cant XFER OUTG TRK TO OUTG TRK
501       ENABLED     Q02138637     MPLR30070     Enables blind transfer to a SIP endpoint even
if SIP UPDATE is not supported by the far en
```

Communication Server 1000 call server deplists

```
VERSION 4121
RELEASE 7
ISSUE 65 P +
DepList 1: core Issue: 01 (created: 2013-05-28 04:19:50 (est))

IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi01058359      ISS1:1OF1       p32331 1  16/11/2015  p32331 1.cpl   NO
001  wi01064599      iss1:1of1       p32580_1  16/11/2015  p32580_1.cpl   NO
002  wi01056067      ISS1:1OF1       p32457 1  16/11/2015  p32457 1.cpl   NO
003  wi01063263      ISS1:1OF1       p32573_1  16/11/2015  p32573_1.cpl   NO
004  wi01065842      ISS1:1OF1       p32478_1  16/11/2015  p32478_1.cpl   NO
005  wi01062607      ISS1:1OF1       p32503_1  16/11/2015  p32503_1.cpl   NO
006  wi01070756      ISS1:1OF1       p32444 1  16/11/2015  p32444 1.cpl   NO
007  wi01039280      ISS1:1OF1       p32423_1  16/11/2015  p32423_1.cpl   NO
008  wi01087543      ISS1:1OF1       p32662 1  16/11/2015  p32662 1.cpl   NO
009  wi00933195      ISS1:1OF1       p32491_1  16/11/2015  p32491_1.cpl   NO
010  wi01071379      ISS1:1OF1       p32522_1  16/11/2015  p32522_1.cpl   NO
011  wi01068669      ISS1:1OF1       p32333_1  16/11/2015  p32333_1.cpl   NO
012  wi01066991      ISS1:1OF1       p32449 1  16/11/2015  p32449 1.cpl   NO
013  wi01070474      iss1:1of1       p32407_1  16/11/2015  p32407_1.cpl   NO
014  WI0110261       ISS1:1OF1       p32758 1  16/11/2015  p32758 1.cpl   NO
015  wi01094305      ISS1:1OF1       p32640_1  16/11/2015  p32640_1.cpl   NO
016  wi01047890      ISS1:1OF1       p32697_1  16/11/2015  p32697_1.cpl   NO
017  wi01055300      ISS1:1OF1       p32543 1  16/11/2015  p32543 1.cpl   NO
```

```
018  wi01082456      ISS1:1OF1       p32596_1  16/11/2015  p32596_1.cpl   NO
019  wi01058621      ISS1:1OF1       p32339_1  16/11/2015  p32339_1.cpl   NO
020  wi01061484      ISS1:1OF1       p32576_1  16/11/2015  p32576_1.cpl   NO
021  wi01078723      ISS1:1OF1       p32532_1  16/11/2015  p32532_1.cpl   NO
022  wi01048457      ISS1:1OF1       p32581_1  16/11/2015  p32581_1.cpl   NO
023  wi01075355      ISS1:1OF1       p32594_1  16/11/2015  p32594_1.cpl   NO
024  wi01053597      ISS1:1OF1       p32304_1  16/11/2015  p32304_1.cpl   NO
025  wi01045058      ISS1:1OF1       p32214_1  16/11/2015  p32214_1.cpl   NO
026  wi01075359      ISS1:1OF1       p32671_1  16/11/2015  p32671_1.cpl   NO
027  wi01025156      ISS1:1OF1       p32136_1  16/11/2015  p32136_1.cpl   NO
028  wi01061481      ISS1:1OF1       p32382_1  16/11/2015  p32382_1.cpl   NO
029  wi01035976      ISS1:1OF1       p32173_1  16/11/2015  p32173_1.cpl   NO
030  wi01088775      ISS1:1OF1       p32659_1  16/11/2015  p32659_1.cpl   NO
031  wi01070465      iss1:1of1       p32562_1  16/11/2015  p32562_1.cpl   NO
032  wi01088585      ISS1:1OF1       p32656_1  16/11/2015  p32656_1.cpl   NO
033  wi01063864      ISS1:1OF1       p32410_1  16/11/2015  p32410_1.cpl   YES
034  wi01034961      ISS1:1OF1       p32144_1  16/11/2015  p32144_1.cpl   NO
035  wi01055480      ISS1:1OF1       p32712_1  16/11/2015  p32712_1.cpl   NO
036  wi01034307      ISS1:1OF1       p32615_1  16/11/2015  p32615_1.cpl   NO
037  wi01065118      ISS1:1OF1       p32397_1  16/11/2015  p32397_1.cpl   NO
038  wi01075360      iss1:1of1       p32602_1  16/11/2015  p32602_1.cpl   NO
039  wi00884716      ISS1:1OF1       p32517_1  16/11/2015  p32517_1.cpl   NO
040  wi01068851      ISS1:1OF1       p32439_1  16/11/2015  p32439_1.cpl   NO
041  wi01053314      ISS1:1OF1       p32555_1  16/11/2015  p32555_1.cpl   NO
042  wi01059388      iss1:1of1       p32628_1  16/11/2015  p32628_1.cpl   NO
043  wi01087528      ISS1:1OF1       p32700_1  16/11/2015  p32700_1.cpl   NO
044  wi01072027      ISS1:1OF1       p32689_1  16/11/2015  p32689_1.cpl   NO
045  wi01052428      ISS1:1OF1       p32606_1  16/11/2015  p32606_1.cpl   NO
046  wi01053920      ISS1:1OF1       p32303_1  16/11/2015  p32303_1.cpl   NO
047  wi01070468      iss1:1of1       p32418_1  16/11/2015  p32418_1.cpl   NO
048  wi01067822      ISS1:1OF1       p32466_1  16/11/2015  p32466_1.cpl   YES
049  wi01060826      ISS1:1OF1       p32379_1  16/11/2015  p32379_1.cpl   NO
050  wi01075352      ISS1:1OF1       p32603_1  16/11/2015  p32603_1.cpl   NO
051  wi01043367      ISS1:1OF1       p32232_1  16/11/2015  p32232_1.cpl   NO
052  wi01083584      ISS1:1OF1       p32619_1  16/11/2015  p32619_1.cpl   NO
053  wi01060241      ISS1:1OF1       p32381_1  16/11/2015  p32381_1.cpl   NO
054  wi01053195      ISS1:1OF1       p32297_1  16/11/2015  p32297_1.cpl   NO
055  wi00897254      ISS1:1OF1       p31127_1  16/11/2015  p31127_1.cpl   NO
056  wi01061483      ISS1:1OF1       p32359_1  16/11/2015  p32359_1.cpl   NO
057  wi01085855      ISS1:1OF1       p32658_1  16/11/2015  p32658_1.cpl   NO
058  wi01075353      ISS1:1OF1       p32613_1  16/11/2015  p32613_1.cpl   NO
059  wi01070471      ISS1:1OF1       p32415_1  16/11/2015  p32415_1.cpl   NO
060  wi01074003      ISS1:1OF1       p32421_1  16/11/2015  p32421_1.cpl   NO
061  wi01060382      iss1:1of1       p32623_1  16/11/2015  p32623_1.cpl   YES
062  wi01068042      ISS1:1OF1       p32669_1  16/11/2015  p32669_1.cpl   NO
063  wi01072023      ISS1:1OF1       p32130_1  16/11/2015  p32130_1.cpl   YES
064  wi01065922      ISS1:1OF1       p32516_1  16/11/2015  p32516_1.cpl   NO
065  wi01057403      ISS1:1OF1       p32591_1  16/11/2015  p32591_1.cpl   NO
066  wi01069441      ISS1:1OF1       p32097_1  16/11/2015  p32097_1.cpl   NO
067  wi01070473      ISS1:1OF1       p32413_1  16/11/2015  p32413_1.cpl   NO
068  wi01056633      ISS1:1OF1       p32322_1  16/11/2015  p32322_1.cpl   NO
069  wi01052968      ISS1:1OF1       p32540_1  16/11/2015  p32540_1.cpl   NO
070  wi01072032      ISS1:1OF1       p32448_1  16/11/2015  p32448_1.cpl   NO
071  wi01073100      ISS1:1OF1       p32599_1  16/11/2015  p32599_1.cpl   NO
072  wi01035980      ISS1:1OF1       p32558_1  16/11/2015  p32558_1.cpl   NO
073  wi01041453      ISS1:1OF1       p32587_1  16/11/2015  p32587_1.cpl   NO
074  wi01032756      ISS1:1OF1       p32673_1  16/11/2015  p32673_1.cpl   NO
075  wi01092300      ISS1:1OF1       p32692_1  16/11/2015  p32692_1.cpl   NO
076  wi00996734      ISS1:1OF1       p32550_1  16/11/2015  p32550_1.cpl   NO
077  wi01022599      ISS1:1OF1       p32080_1  16/11/2015  p32080_1.cpl   NO
078  wi01060341      ISS1:1OF1       p32578_1  16/11/2015  p32578_1.cpl   NO
079  wi01091447      ISS1:1OF1       p32675_1  16/11/2015  p32675_1.cpl   NO
080  wi01070580      ISS1:1OF1       p32380_1  16/11/2015  p32380_1.cpl   NO
081  wi01089519      ISS1:1OF1       p32665_1  16/11/2015  p32665_1.cpl   NO
082  WI01077073      ISS1:1OF1       p32534_1  16/11/2015  p32534_1.cpl   NO
083  wi01080753      ISS1:1OF1       p32518_1  16/11/2015  p32518_1.cpl   NO
084  wi01065125      ISS1:1OF1       p32416_1  16/11/2015  p32416_1.cpl   NO
```

Communication Server 1000 signaling server service updates

```
In System service updates: 41
PATCH#  IN SERVICE   DATE       SPECINS   REMOVABLE   NAME
0       Yes          14/07/14   YES       YES         cs1000-csmWeb-7.65.16.22-2.i386.000
1       Yes          14/10/15   YES       YES         cs1000-dmWeb-7.65.16.23-4.i386.000
3       Yes          15/10/15   NO        YES         cs1000-sps-7.65.16.23-1.i386.000
4       Yes          14/07/14   YES       YES         cs1000-patchWeb-7.65.16.22-4.i386.000
5       Yes          14/10/15   YES       YES         cs1000-linuxbase-7.65.16.23-19.i386.000
7       Yes          14/07/14   YES       YES         cs1000-csoneksvrmgr-7.65.16.22-5.i386.000
8       Yes          27/09/13   NO        YES         cs1000-pd-7.65.16.21-00.i386.000
9       Yes          27/09/13   NO        YES         cs1000-shared-carrdtct-7.65.16.21-
01.i386.000
10      Yes          27/09/13   NO        YES         cs1000-shared-tpselect-7.65.16.21-
01.i386.000
11      Yes          14/07/14   YES       YES         cs1000-baseWeb-7.65.16.22-4.i386.000
12      Yes          27/09/13   NO        yes         cs1000-dbcom-7.65.16.21-00.i386.000
16      Yes          14/10/15   NO        YES         cs1000-Jboss-Quantum-7.65.16.23-5.i386.000
17      Yes          15/10/15   YES       YES         cs1000-cs-7.65.P.100-03.i386.000
18      Yes          15/10/15   NO        YES         bash-3.2-33.el5 11.4.i386.000
19      Yes          15/10/15   YES       YES         cs1000-shared-pbx-7.65.16.23-1.i386.000
20      Yes          15/10/15   YES       YES         cs1000-emWeb 6-0-7.65.16.23-3.i386.000
21      Yes          15/10/15   NO        YES         libxml2-2.6.26-2.1.25.el5_11.i386.000
22      Yes          15/10/15   NO        YES         libxml2-python-2.6.26-
2.1.25.el5_11.i386.000
23      Yes          02/04/14   NO        YES         cs1000-shared-omm-7.65.16.21-2.i386.000
24      Yes          15/10/15   NO        YES         freetype-2.2.1-32.el5 9.1.i386.000
26      Yes          15/10/15   NO        YES         cs1000-cs1000WebService_6-0-7.65.16.23-
1.i386.000
27      Yes          14/07/14   YES       YES         cs1000-oam-logging-7.65.16.22-4.i386.000
28      Yes          15/10/15   YES       YES         cs1000-ftrpkg-7.65.16.23-1.i386.000
29      Yes          15/10/15   NO        YES         cs1000-cppmUtil-7.65.16.23-4.i686.000
30      Yes          02/10/13   NO        YES         cs1000-snmp-7.65.16.21-00.i686.000
31      Yes          14/07/14   YES       YES         cs1000-csv-7.65.16.22-2.i386.000
33      Yes          14/07/14   YES       YES         cs1000-nrsm-7.65.16.22-3.i386.000
34      Yes          14/07/14   YES       YES         cs1000-mscTone-7.65.16.22-2.i386.000
35      Yes          14/07/14   YES       YES         cs1000-mscMusc-7.65.16.22-4.i386.000
36      Yes          14/07/14   YES       YES         cs1000-mscConf-7.65.16.22-2.i386.000
38      Yes          02/04/14   YES       YES         cs1000-emWebLocal 6-0-7.65.16.22-1.i386.000
39      Yes          15/10/15   NO        YES         tzdata-2015a-1.el5.i386.000
40      Yes          02/04/14   YES       YES         cs1000-ipsec-7.65.16.22-1.i386.000
41      Yes          15/10/15   YES       YES         cs1000-tps-7.65.16.23-15.i386.000
43      Yes          15/10/15   YES       YES         kernel-2.6.18-406.el5.i686.000
44      Yes          15/10/15   YES       YES         cs1000-vtrk-7.65.16.23-76.i386.000
45      Yes          15/10/15   YES       YES         cs1000-bcc-7.65.16.23-10.i386.000
47      Yes          14/07/14   YES       YES         cs1000-mscAnnc-7.65.16.22-2.i386.000
48      Yes          14/07/14   YES       YES         cs1000-mscAttn-7.65.16.22-2.i386.000
49      Yes          14/07/14   NO        YES         cs1000-gk-7.65.16.22-1.i386.000
53      Yes          14/07/14   YES       YES         cs1000-shared-xmsg-7.65.16.22-1.i386.000
```

Communication Server 1000 system software

```
Product Release: 7.65.16.00
Base Applications
   base                          7.65.16     [patched]
   NTAFS                         7.65.16
   sm                            7.65.16
   cs1000-Auth                   7.65.16
   Jboss-Quantum                 n/a         [patched]
   cnd                           7.65.16
   lhmonitor                     7.65.16
   baseAppUtils                  7.65.16
   dfoTools                      7.65.16
   cppmUtil                      n/a         [patched]
   oam-logging                   n/a         [patched]
   dmWeb                         n/a         [patched]
   baseWeb                       n/a         [patched]
   ipsec                         n/a         [patched]
   Snmp-Daemon-TrapLib           n/a         [patched]
   ISECSH                        7.65.16
   patchWeb                      n/a         [patched]
   EmCentralLogic                7.65.16
Application configuration: CS+SS+NRS+EM
Packages:
CS+SS+NRS+EM
Configuration version:     7.65.16-00
   cs                            7.65.16     [patched]
   dbcom                         7.65.16.21  [patched]
   cslogin                       7.65.16
   sigServerShare                7.65.16     [patched]
   csv                           7.65.16     [patched]
   tps                           7.65.16     [patched]
   vtrk                          7.65.16     [patched]
   pd                            7.65.16.21  [patched]
   sps                           7.65.16     [patched]
   ncs                           7.65.16
   gk                            7.65.16     [patched]
   nrsm                          7.65.16     [patched]
   nrsmWebService                7.65.16
   managedElementWebService      7.65.16
   EmConfig                      7.65.16
   emWeb_6-0                     7.65.16     [patched]
   emWebLocal 6-0                7.65.16     [patched]
   csmWeb                        7.65.16     [patched]
   bcc                           7.65.16     [patched]
   ftrpkg                        7.65.16     [patched]
   cs1000WebService_6-0          7.65.16     [patched]
   mscAnnc                       7.65.16     [patched]
   mscAttn                       7.65.16     [patched]
   mscConf                       7.65.16     [patched]
   mscMusc                       7.65.16     [patched]
   mscTone                       7.65.16     [patched]
```

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.