Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Aura® Session Manager R6.1 and Avaya Aura® Session Border Controller to support BT Wholesale/HIPCOM SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between BT Wholesale (BTW)/HIPCOM SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Communication Server 1000E and Avaya Aura® Session Border Controller.

BT is a member of the DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect lab.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between BT Wholesale/HIPCOM SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Communication Server 1000E (CS1K) connected to BT Wholesale/HIPCOM SIP Trunk Service via an Avaya Aura® Session Border Controller (SBC). Customers using this Avaya SIP-enabled enterprise solution with BT Wholesale/HIPCOM's SIP Trunk Service 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 normally results in lower cost for the enterprise.

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 CS1K, Session Manager and SBC. The enterprise site was configured to use the SIP Trunk Service provided by BTW/HIPCOM.

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 BTW/HIPCOM. Incoming PSTN calls were made to Unistim, SIP, Digital and analog telephones at the enterprise
- Outgoing calls from the enterprise to the PSTN were made from Unistim, SIP, Digital and analog telephones
- G.729 annex b (silence suppression) is not supported by BTW/HIPCOM's SIP Trunk Service and thus was not tested
- Calls using G.729 and G.711A codec's were tested
- Fax calls to/from a Group 3 fax machine to a PSTN connected fax machine using the T.38 mode
- 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

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for BTW/HIPCOM SIP Trunk Service with the following observations.

- Incoming call to busy trunks or SIP Trunk signaling failure the following was observed - PSTN receives NU Tone eventually and 500 Service Unavailable sip message. The global parameter set on BTW/HIPCOM's SBC is 4 hunts per call, so if the call doesn't set up on the first try BTW/HIPCOM's SBC will re-try a further 3 times.

2.3. Support

For technical support on BTW/HIPCOM products please contact the following website: http://www.hipcom.co.uk/support or http://ipvoicesupport.btwholesale.com.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to BTW/HIPCOM using SIP Trunks. Located at the enterprise site are Session Manager, SBC and a Communication Server 1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya IP Softphones (SMC3456, 2050 and one-X Communicator), 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.

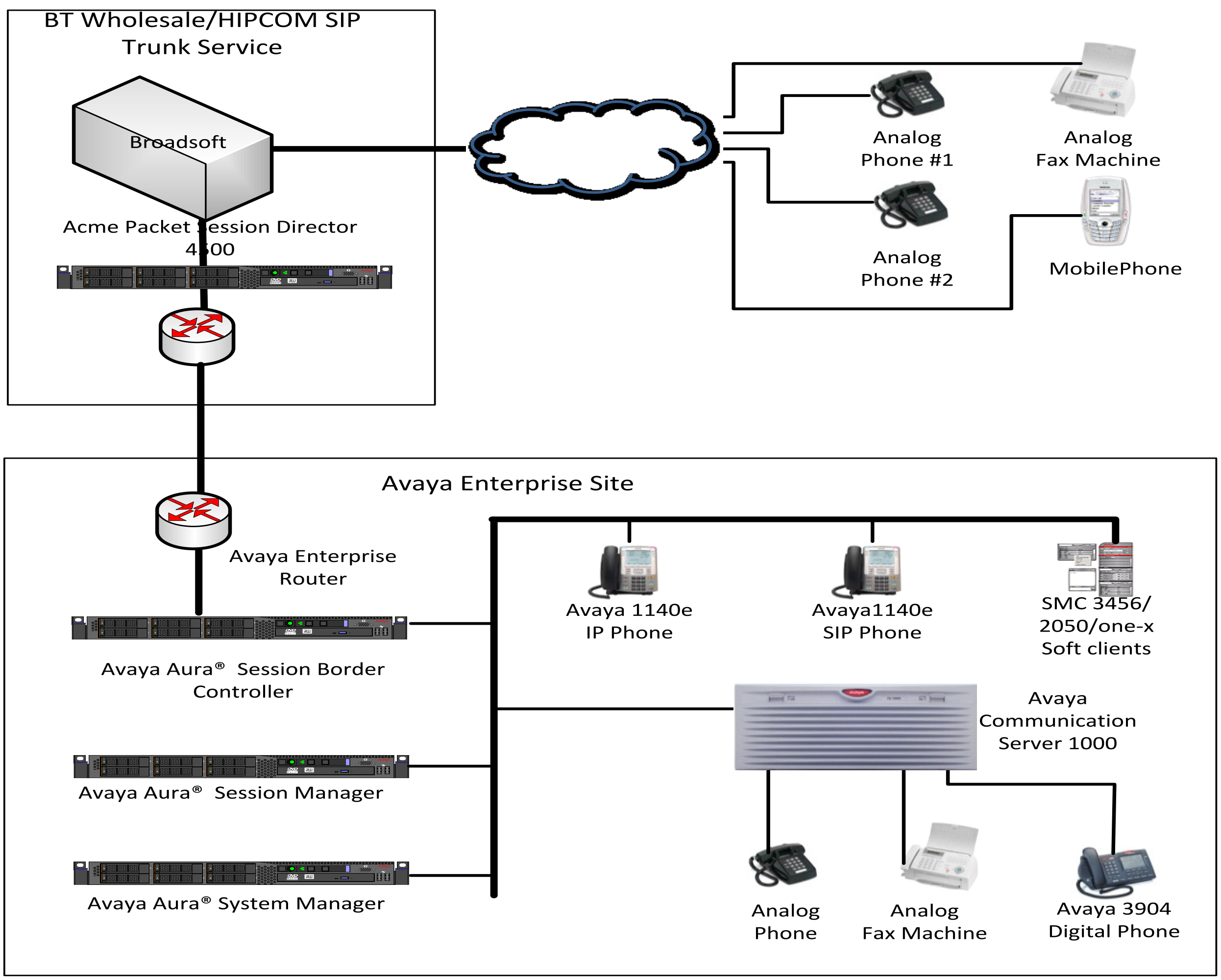


Figure 1: BT Wholesale/HIPCOM SIP Trunk topology with Avaya Communication Server 1000E

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 server	Avaya Aura® Session Manager R6.1 Build: 6.1.0.0.610023 Service Pack 3
Avaya S8800 server	Avaya Aura® System Manager R6.1 (6.1.0.0.7345 – 6.1.5.112) Service Pack 3
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R7.5, Version 7.50.17 Service Update: 7.50_17Nov23 Deplist: X21 07.50Q
Avaya Media S8800 server	Avaya Aura® Session Border Controller version 6.0.2.0.2 (E362P4)
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC CD01 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA07 DSP1 Version: DSP1 AB04
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 57666
Avaya one-X® Communicator	Avaya one-X® Communicator - cs6.1.0.10
Avaya 2050 IP Softphone	Release 4.0.2.0062
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
BTW/HIPCOM SIP Trunk Service	Acme Packet 4500 Net-Net SBC ver SCX6.1.0 Broadsoft - ver 14 Sevice Pack 9 Configuration version - HIPCOM v8.1

5. Configure Avaya Communication Server 1000E

This section describes the steps required to configure Communication Server 1000E for SIP Trunking and also the necessary configuration for terminals (analog, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP Signaling associated with BTW/HIPCOM's SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the SBC, through which the BTW/HIPCOM SIP Service directs incoming SIP messages to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. Once Communication Server 1000E selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the SBC and on to BTW/HIPCOM's network. Specific Communication Server 1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the Communication Server 1000E, System Manager and Session Manager is presumed to have been previously completed and is not discussed here.

5.1. Logging into the Avaya Communication Server 1000E

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

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 Communication Server 1000E 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 BTW/HIPCOM's 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 Communication Server 1000E.

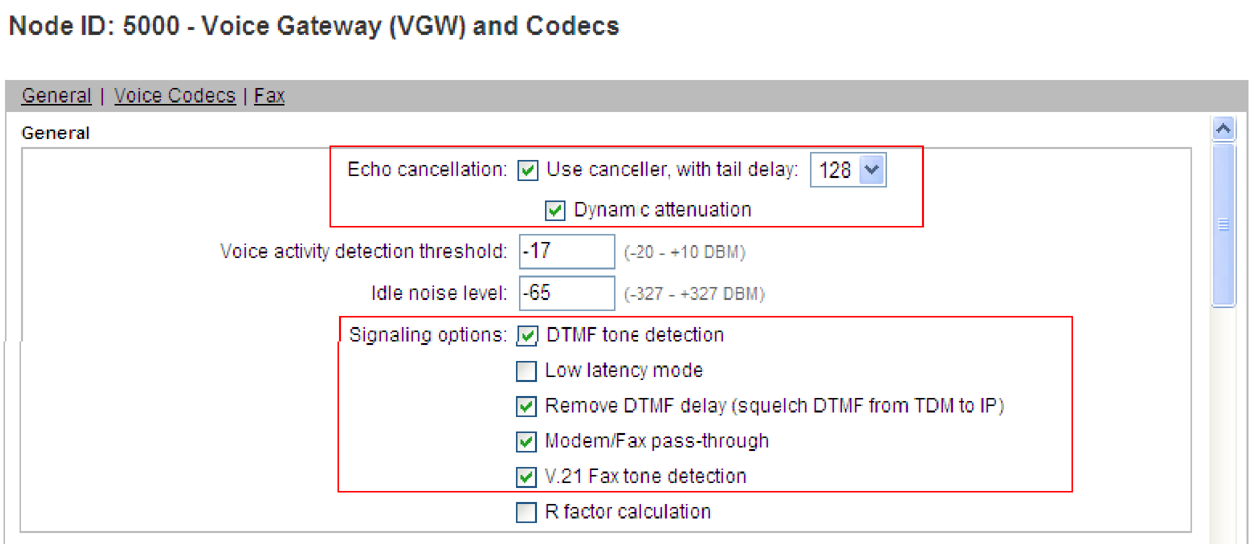
```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

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

TRADITIONAL TELEPHONES 32767    LEFT 32766    USED     1
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32744    USED    23
BASIC IP USERS         32767    LEFT 32766    USED     1
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32752    USED    15
MOBILE EXTENSIONS      32767    LEFT 32767    USED     0
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
NORTEL SIP LINES       32767    LEFT 32765    USED     2
THIRD PARTY SIP LINES  32767    LEFT 32761    USED     6
SIP CONVERGED DESKTOPS 32767    LEFT 32767    USED     0
SIP CTI TR87           32767    LEFT 32767    USED     0
SIP ACCESS PORTS       32767    LEFT 32752    USED    15
```

Load overlay 21 and confirm the customer is setup to use **ISDN** trunks (see below).

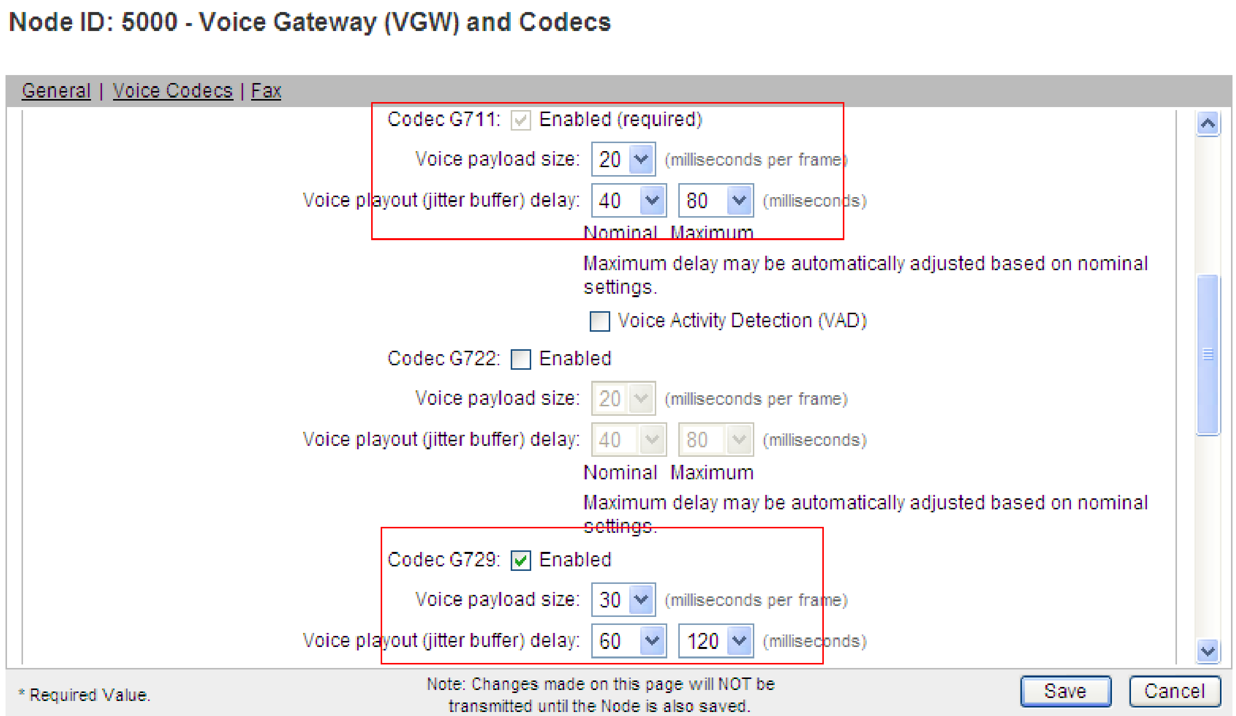
```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

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

5.3. Configure Codec's for Voice and FAX operation

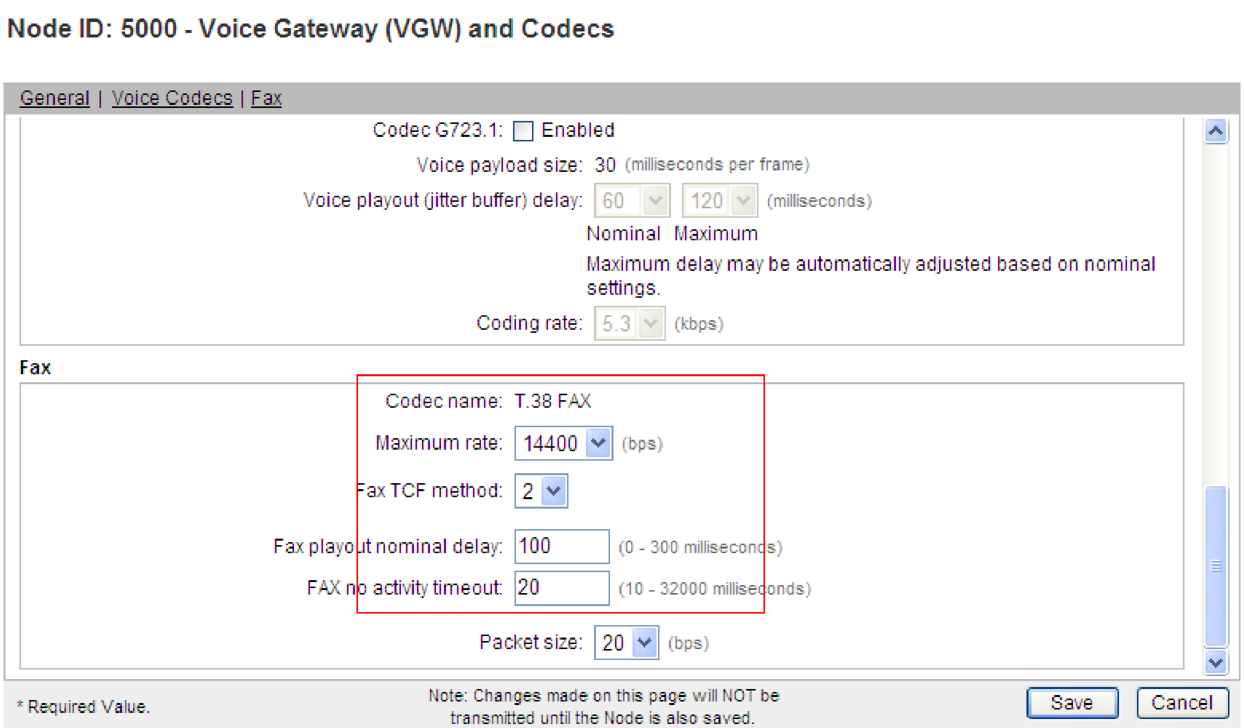
BTW/HIPCOM SIP Trunk service supports G.711A voice codec and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW and Codecs** property page and configure the Communication Server 1000E General codec settings as in the next screenshot. The values highlighted are required for correct operation.



Next, scroll down and configure the **G.711** and **G.729** codec settings. The relevant settings are highlighted in the following screenshot.

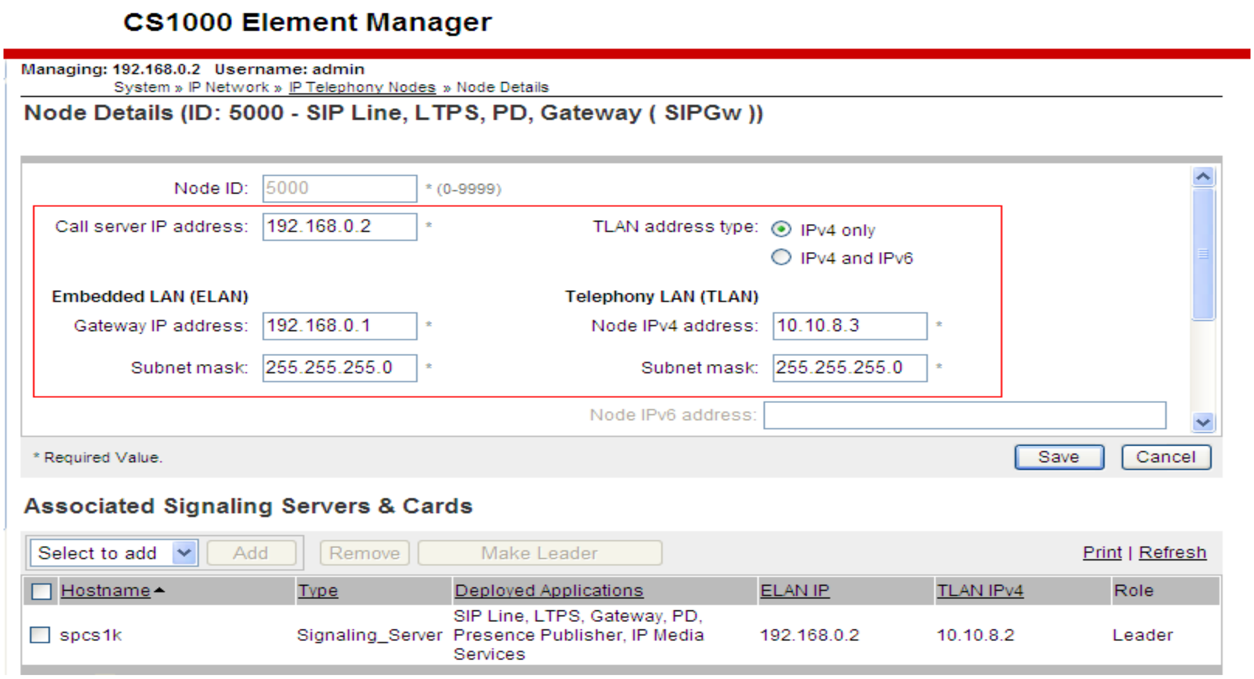


Finally, configure the **Fax** settings as in the highlighted section of the next screenshot.



5.4. Virtual Trunk Gateway Configuration

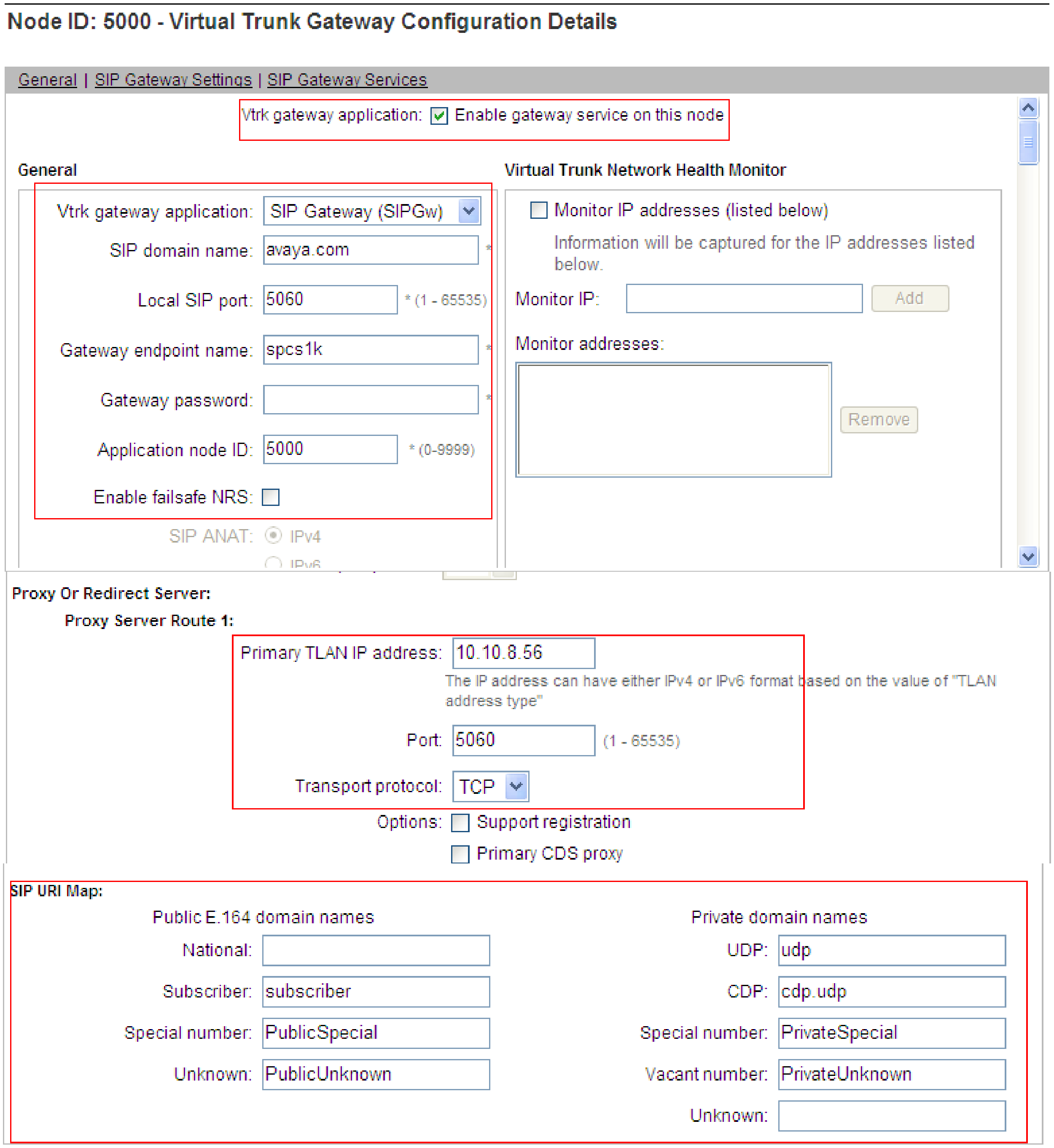
Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server has an ip address and so too does the signalling server. The Node ip is the ip address that the IP phones use to register. This is also where the SIP trunk connection is made to the Session Manager. When an entity link is added in Session Manager for the CS1K it is the Node ip that is used (see **Section 6.4** – 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.

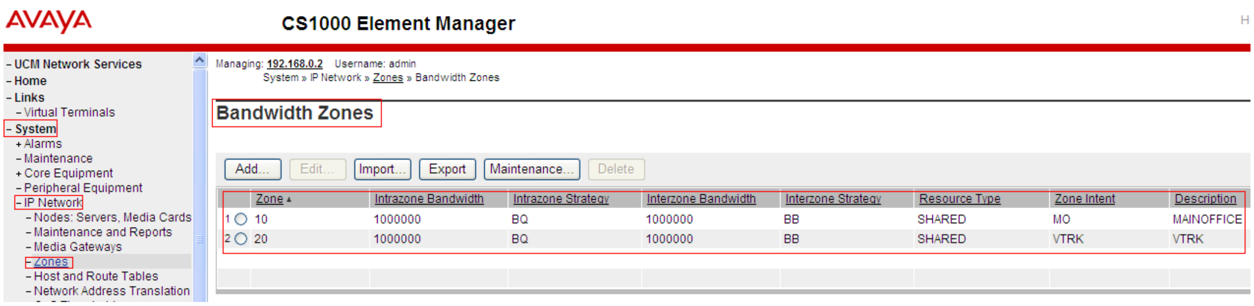
- **Vtrk gateway application:** Provides option to select Gateway applications. The three supported modes are **SIP Gateway (SIPGw)**, **H.323Gw**, and **SIPGw.**
- **SIP domain name:** The SIP Domain Name is the SIP Service Domain. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the Session Manager
- **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 5000

- **Proxy or Redirect Server:** Primary TLAN ip address is the SM100 ip address of the Session Manager. The **Transport protocol** used for **SIP**, in this case is TCP
- **SIP URI Map: Public National** and **Private Unknown** are left blank. All other fields in the SIP URI Map are left with default values.



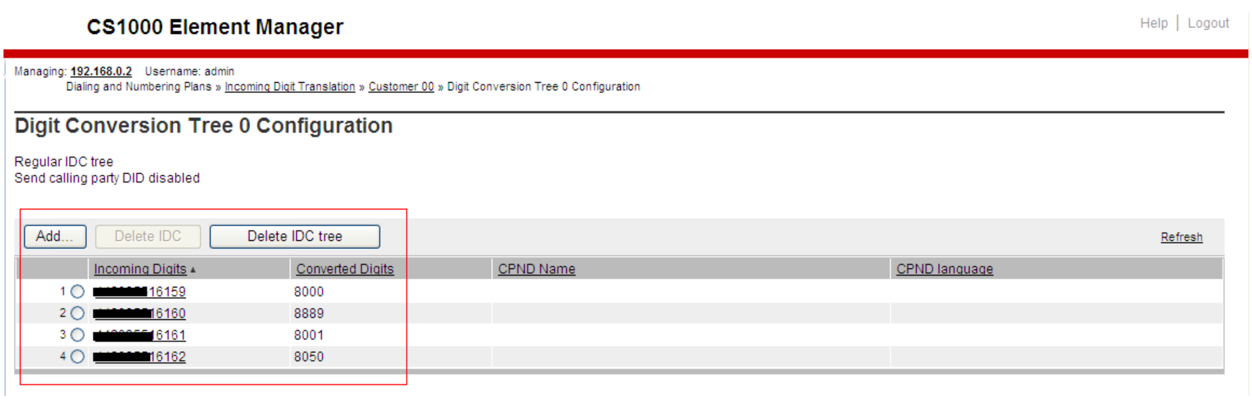
5.5. Configure Bandwidth Zones

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, IP telephones and Media Gateways are all placed in separate zones. 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.



5.6. Configure Incoming Digit Conversion Table

A limited number of Direct Dial Inwards (DDI) numbers were available; an IDC table was configured to translate incoming PSTN numbers to five digit local telephone extension numbers. The last four 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.



5.7. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to BTW/HIPCOM's SIP Trunk Service. Five separate steps are required to configure Communication Server 1000E virtual trunks:-

- Configure a D-Channel Handler (DCH); configure using the Communication Server 1000E system terminal and overlay 17
- Configure a SIP trunk Route Data Block (RDB); configure using the Communication Server 1000E system terminal and overlay 16
- Configure SIP trunk members; configure using the Communication Server 1000E system terminal and overlay 14
- Configure a Route List Block (RLB); configure using the Communication Server 1000E system terminal and overlay 86
- Configure Special Prefix Numbers (SPN's); configure using the Communication Server 1000E system terminal and overlay 90

The following is an example DCH configuration for SIP trunks. Load **Overlay 17** at the Communication Server 1000E 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 10
  CTYP DCIP
  DES  VIR_TRK
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA YES
  IFC  SL1
  CNEG 1
  RLS  ID  5
  RCAP ND2
  MBGA NO
  H323
    OVLR NO
    OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the Communication Server 1000E 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 5.3**. The value for **ZONE** should match that used in **Section 5.4** for **SIP_VTRK**. The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 16
TYPE: RDB
CUST 00
ROUT 100
TYPE RDB
CUST 00
ROUT 100
DES  VIR_TRK
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
RPA  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 00020
PCID SIP
CRID NO
NODE 5000
DTRK NO
ISDN YES
    MODE ISLD
    DCH  10
    IFC  SL1
    PNI  00001
    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 1600
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 Communication Server 1000E 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
new 30
TN   160 0 0 0
DATE
PAGE
DES  VIR_TRK
TN   160 0 00 00  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00020
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK  ANLG
NCOS 0
RTMB 100 1
CHID 1
TGAR 1
STRI/STRO WNK WNK
SUPN YES
AST  NO
IAPG 0
CLS  TLD DTN CND ECD WTA LPR APN THFD XREP SPCD MSBT
     P10 NTC
TKID
AACR NO
```

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 **new** CUST 0 FEAT rlb **RLI 24** ELC NO ENTR 0 LTER NO **ROUT 100** 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 0 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 Special Prefix Number(s) (SPN) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 90. The following are some example SPN 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.

SPN 999 FLEN 3 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 90 FLEN 7 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 2 FLEN 7 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE	SPN 15 FLEN 3 ITOH NO CLTP NONE **RLI 24** SDRR NONE ITEI NONE

5.8. 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 five digit number is entered for the **KEY 00** and **KEY 01** value. The value for **CFG_ZONE** is the same value used in **Section 5.4** for **VIRTUALSETS**.

```
Overlay 20 IP Telephone configuration
DES  1140
TN   096 0 01 16  VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
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

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```

```
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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 8000 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8000 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

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```

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```
MLNG ENG
DNDR 0
KEY  00 MCR 8866 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8866 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 T.38 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   8888
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
```

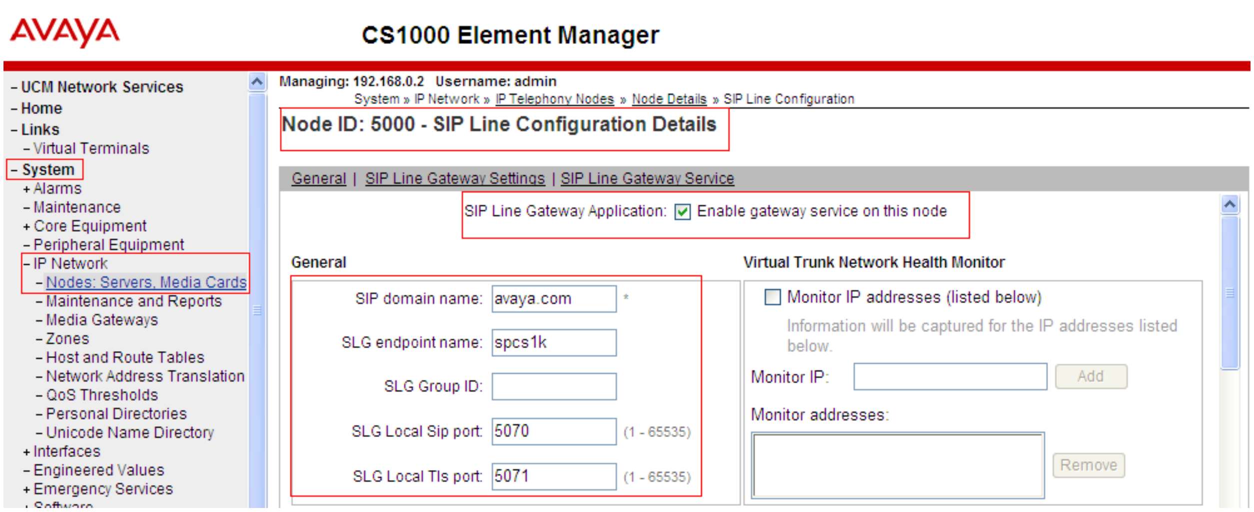
5.9. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server 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 Communication Server 1000E system terminal and overlay 15 to activate SIP Line services, as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
  SIPL_ON YES
  UAPR 78
  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. The value for **SIP Domain Name** must match that configured in **Section 6.1**.

- **SIP Line Gateway Application: □ Enable the SIP line service on the node**, check the box to enable
- **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**



5.10. Configure SIP Line Telephones

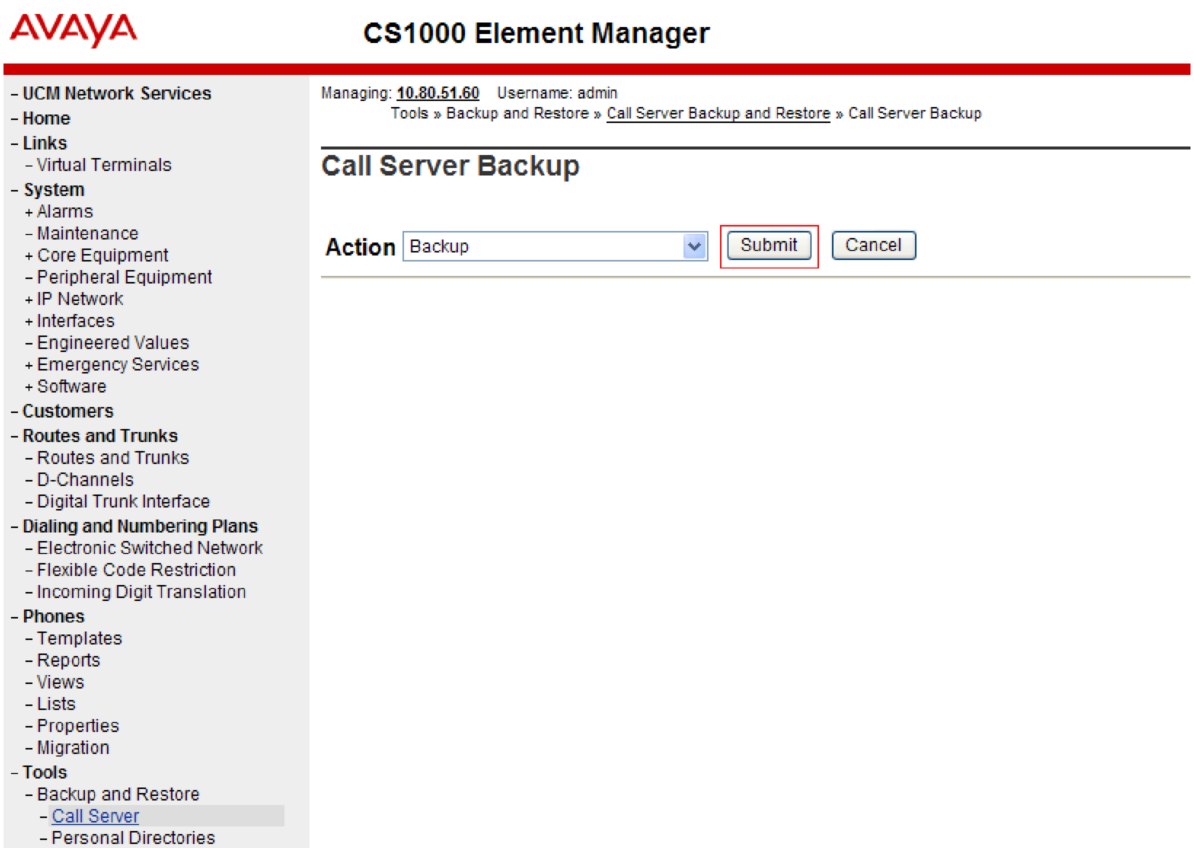
When SIP Line service configuration is completed, use the Communication Server 1000E system terminal and **Overlay 20** to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value 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 set for **SIPLINEZONE** in **Section 5.4**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value (set to 78 previously in this section) and the telephone number used in **KEY 00**.

```
Overlay 20 – SIP Telephone Configuration
DES  SIPD
TN   096 0 01 15  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 8889
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
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

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

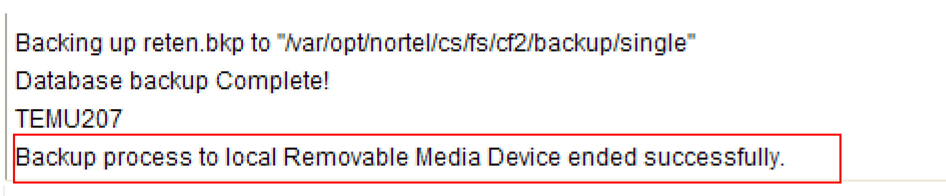
```
---continuedfrom previous page---

     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 8889 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 788889 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
```

5.11. 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.



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.



Configuration of Communication Server 1000E is complete.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager to receive and route calls over the SIP trunk between Communication Server 1000E and Session Manager. These instructions assume other administration activities have already been completed such as defining the SIP entity for Session Manager, defining the network connection between System Manager and Session Manager, and adding SIP endpoints. The following administration activities will be described.

- Define SIP Domain
- Define Location for Avaya Communication Server 1000E
- Configure the Adaptation Module.
- Define SIP Entities
- Define Entity Links
- Define Routing Policies
- Define Dial Patterns

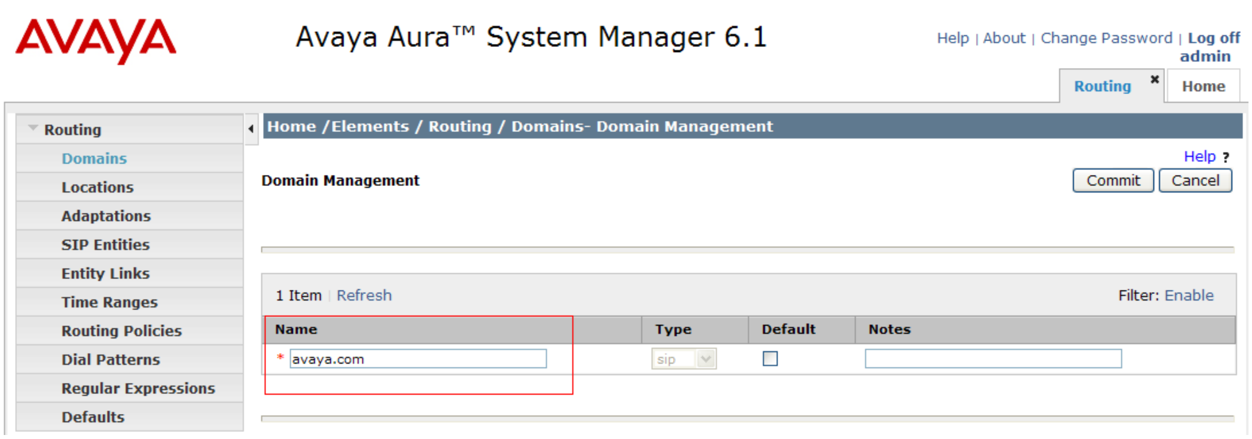
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Login with the appropriate credentials. Some administration screens have been abbreviated for clarity.

6.1. Define SIP domains

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 the Domain Name specified for the SIP Gateway in **Section 5.3.** 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.



6.2. Define Location for Avaya Communication Server 1000E

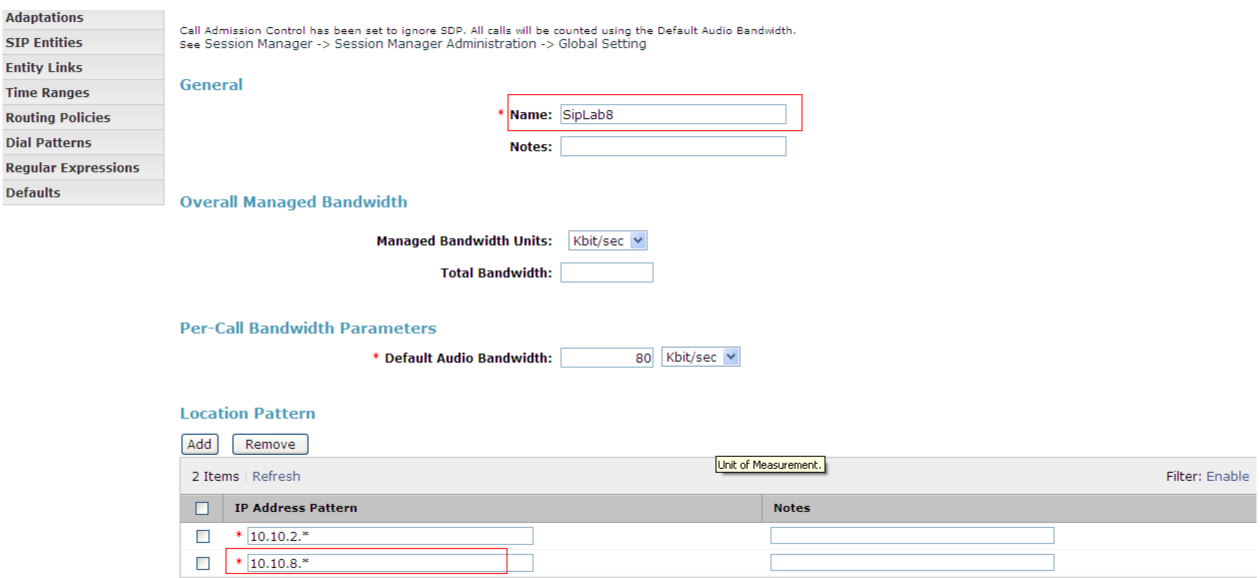
Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. Expand **Elements → Routing** and select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name** Enter a descriptive name for the location
- **Notes** Add a brief description [Optional]

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

- **IP Address Pattern** Enter the logical pattern used to identify the location. For the sample configuration, **10.10.8.*** was used
- **Notes** Add a brief description [Optional]

Click **Commit** to save. The screenshot below shows the Location defined for Communication Server 1000E in the sample configuration.



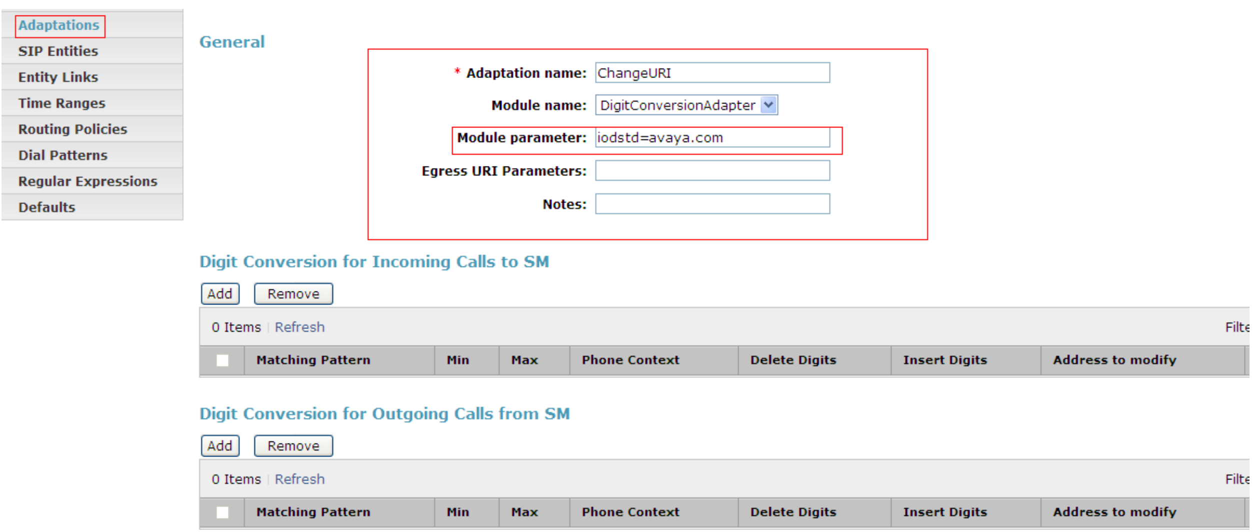
6.3. Configure Adaptation Module

Session Manager is installed with a module called **DigitConversionAdapter**, which can convert digit strings in various message headers as well as host names in the Request-URI (Uniform Resource Identifier). In this configuration the adaptation is used by the SBC to ensure ingress messages have the hostname **avaya.com** when they are sent to the Session Manager and to the CS1K. To add an adaptation, select **Adaptations** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- **Adaptation Name:** Enter an informative name
- **Module Name:** **<click to add module>** from the drop down list and enter **DigitConversionAdapter** in the resulting **New Module Name** field
- **Module Parameter:** Enter the modification parameters to be used. In this configuration the modification parameters used was **iodstd=avaya.com**

iodstd (or **ingressOverrideDestinationDomain**) replac**es the domain in** a Request-URI and Notify/message-summary body with the given value for ingress only. The reason why this was added was that incoming calls to the enterprise had BTW/HIPCOM's domain name in the SIP messages. The domain on the enterprise is avaya.com so this Adaption Module changed incoming SIP messages destined for the enterprise to a recognised domain.



6.4. Define SIP Entities

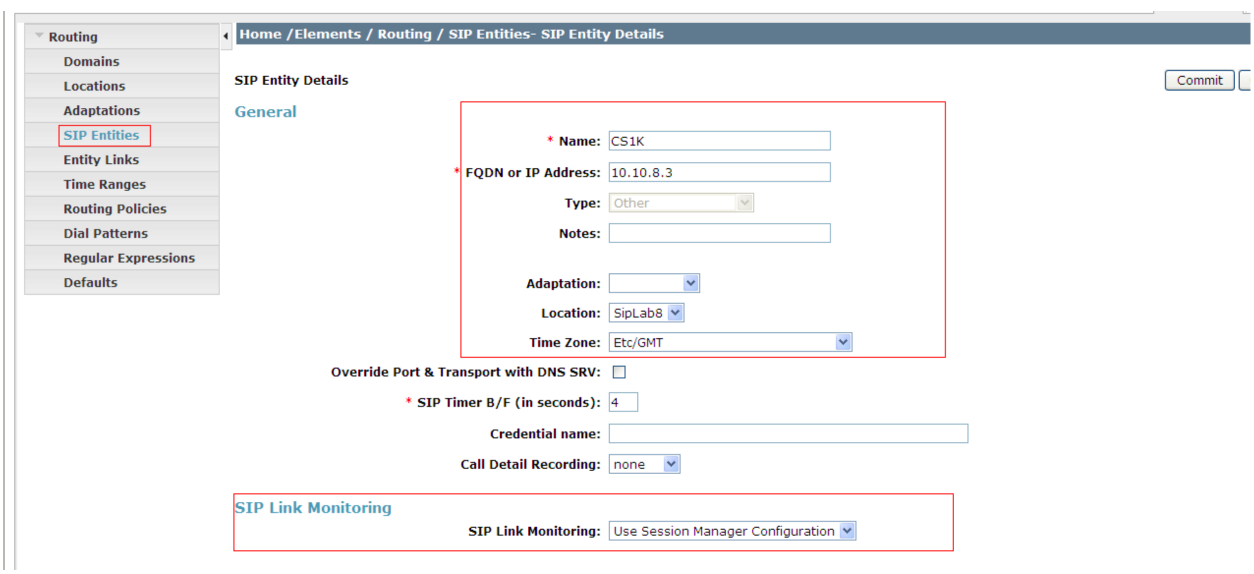
A SIP Entity must be added for Communication Server 1000E and also for the SBC. Expand **Elements → Routing** and select **SIP Entities** from the left navigation menu. 2 new SIP Entities will need to be added as noted above. Click **New** (not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Name** Enter an identifier for the SIP Entity
- **FQDN or IP Address** Enter TLAN IP address of Communication Server 1000E Node identified in **Section 5.3.** For the SBC enter the private interface IP address
- **Type** Select **Other** for the Communication Server 1000E and **gateway** for the SBC
- **Notes** Enter a brief description [Optional]
- **Adaptations** CS1000Adapter defined in **Section 6.3**
- **Location** Select the Location defined for Communication Server 1000E in **Section 6.2** and also apply this same location to the SBC

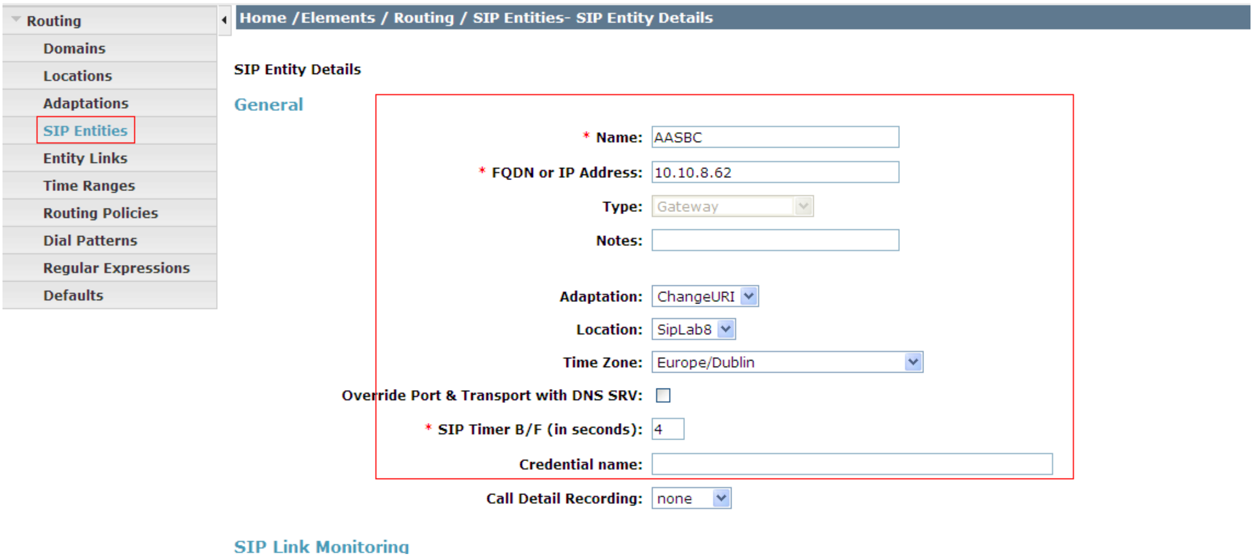
In the **SIP Link Monitoring** section.

- **SIP Link Monitoring** Select **Use Session Manager Configuration**

Click **Commit** to save the definition of the new SIP Entity. The following screenshot shows the SIP Entity defined for Communication Server 1000E in the sample configuration.



The following screenshot shows the SIP Entity defined for SBC in the sample configuration, note the adaption created in **Section 6.3** is associated with this entity link.



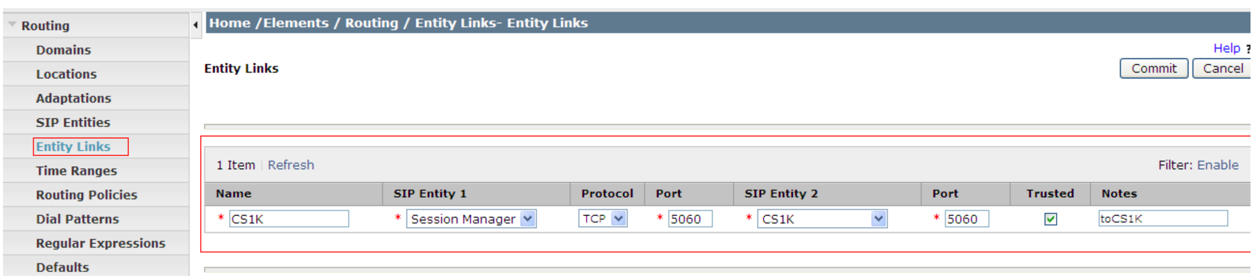
A SIP Entity link must also be defined for your Session Manager but that is not shown in this document.

6.5. Define Entity links

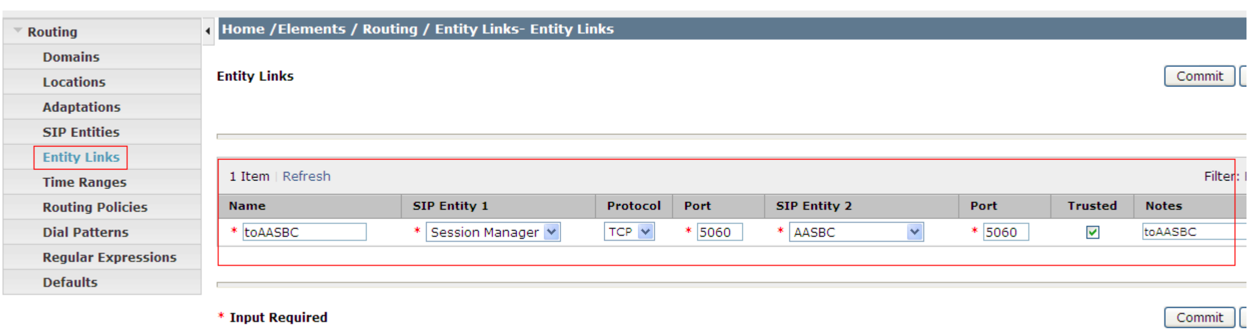
The SIP trunk between the Session Manager and the Communication Server 1000E is described by an Entity link. The same is needed between the Session Manager and SBC. Expand **Elements → Routing** and select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link to each telephony system
- **SIP Entity 1** Select SIP Entity defined for **Session Manager**
- **SIP Entity 2** Select the SIP Entity defined for Avaya Communication Server 1000E/SBC in **Section 6.3** i.e. **CS1K**
- **Protocol** After selecting both SIP Entities, select **TCP** as the required protocol
- **Port** Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is **5060**
- **Trusted** Enter a tick in the box
- **Notes** Enter a brief description [Optional]

Click **Commit** to save **Entity Link** definition. The following screen shows the entity link defined for the SIP trunk between Session Manager and Communication Server 1000E.



The following screen shows the entity link defined for the SIP trunk between Session Manager and SBC.



6.6. Define Routing Policy

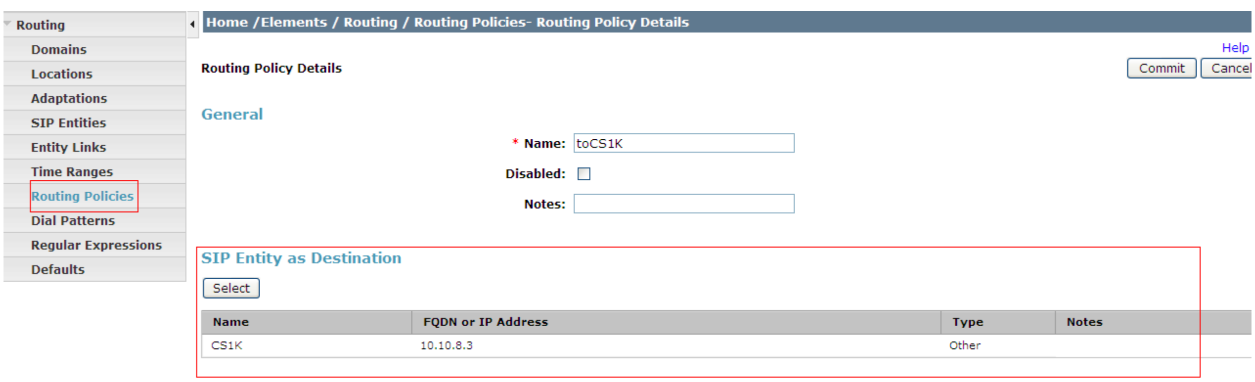
Routing policies describe the conditions under which calls will be routed to CS1K from either SIP endpoint registered to Session Manager or from other telephony system. It also describers the routing polices for which calls will be routed to the SBC and therefore to BTW/HIPCOM's SIP network. To add a routing policy, expand **Elements → Routing** and select **Routing Policies.** Click **New** (not shown). In the **General** section, enter the following values.

- **Name** Enter an identifier to define the routing policy
- **Disabled** Leave unchecked
- **Notes** Enter a brief description [Optional]

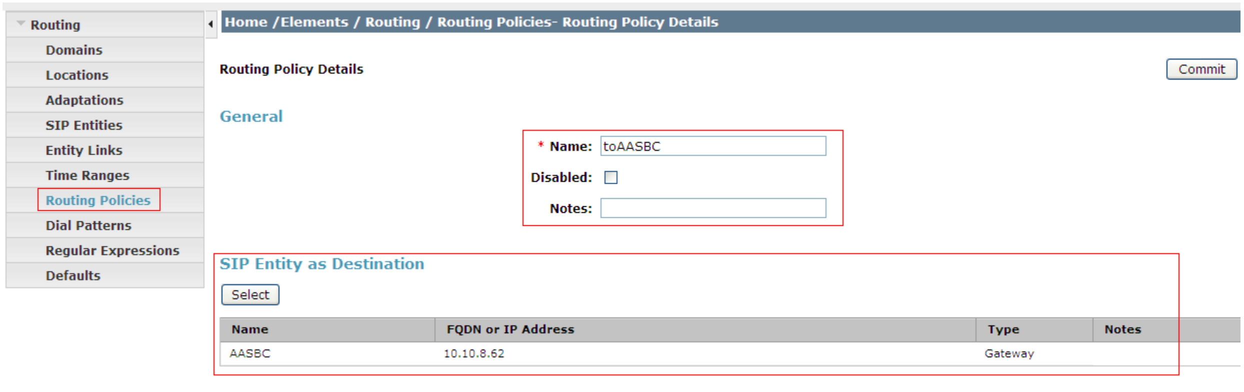
In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). For routing policy to the Communication Server 1000E, select the SIP Entity associated with Communication Server 1000E defined in **Section 6.4** and click **Select.** The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

Note: The routing policy defined in this section is an example and was used in the sample configuration. Other routing policies may be appropriate for different customer networks.

The following screenshot shows the Routing Policy for CS1K:



For routing policy to the SBC – BTW/HIPCOM SIP Trunk, select the SIP Entity associated with SBC defined in **Section 6.4** and click **Select.** The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition. The following screenshot shows the Routing Policy for SBC – BTW/HIPCOM SIP trunk.



6.7. Define Dial Pattern

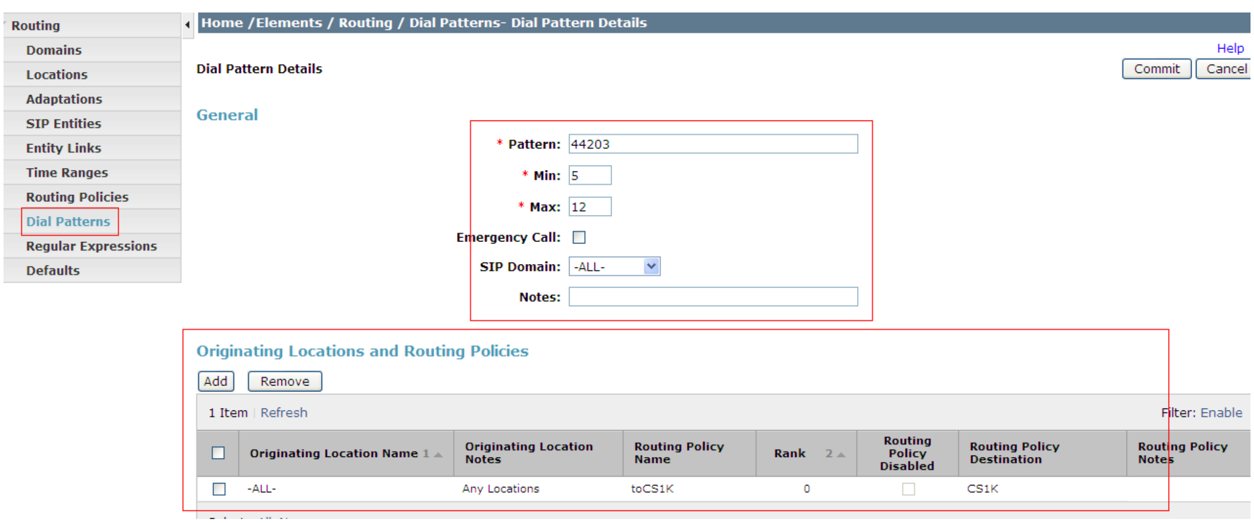
Dial patterns are used to route calls to appropriate SIP Entities. In the sample configuration, since the DDI range given for the testing all numbers that start with **44203** will be routed to the Communication Server 1000E for terminating to test sets. Alternately calls that are originated on the Communication Server 1000E that start with digits **00353** will be routed to the SBC and then on to BTW/HIPCOM's SIP network, there is a dialing pattern added for this as well. To define a dial pattern, expand **Elements → Routing** and select **Dial Patterns** (not shown). Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern** Enter dial pattern for calls to Avaya Communication Server 1000E
- **Min** Enter the minimum number digits that must to be dialed
- **Max** Enter the maximum number digits that may be dialed
- **SIP Domain** Select the SIP Domain from drop-down menu or select **All** if Session Manager should accept incoming calls from all SIP domains
- **Notes** Enter a brief description [Optional]

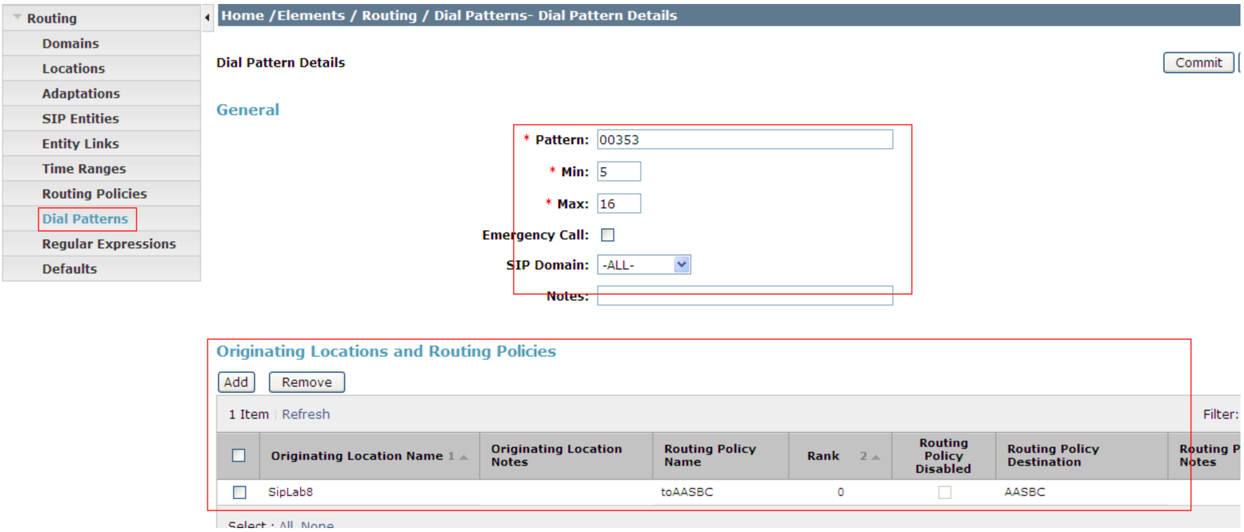
In the **Originating Locations and Routing Policies** section, click **Add.** The **Originating Locations and Routing Policy List** page opens (not shown).

- **Originating Locations** Select **ALL**
- **Routing Policies** Select the Routing Policy defined for Communication Server 1000E in **Section 6.6**

Click **Select** to save these changes and return to **Dial Pattern Details** page. Click **Commit** to save. The following screen shows the Dial Pattern defined for sample configuration. The following screenshot shows the Routing Policy for Communication Server 1000E.



Repeat the above steps to add the dial Pattern to the SBC, select the routing policy defined for the SBC in **Section 6.5**. The following screenshot shows the Routing Policy for SBC – BTW/HIPCOM's SIP network.

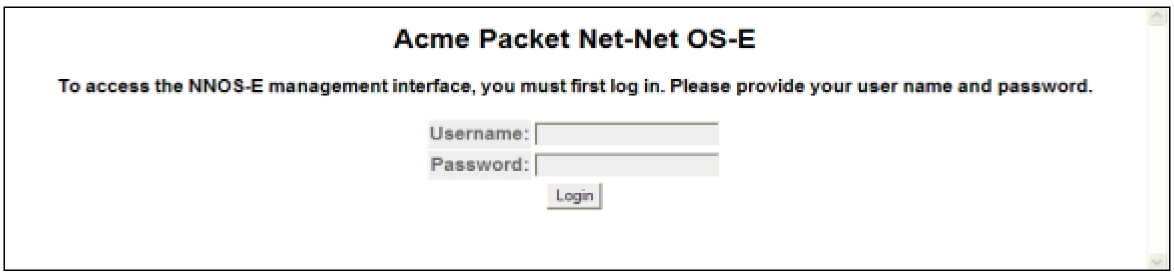


7. Configure Avaya Aura® Session Border Controller

This section provides the procedures for configuring SBC to receive and route calls over the SIP trunk between CS1K and BTW/HIPCOM SIP Trunks. These instructions assume other administration activities have already been completed such as the default configuration. This section will cover the configuration that was put in place specifically for BTW/HIPCOM. For more information regarding the configuration of the SBC, please refer to the sample SBC configuration file in **Appendix B**. In **Appendix B**, note that public IP addresses have been taken out for security purposes.

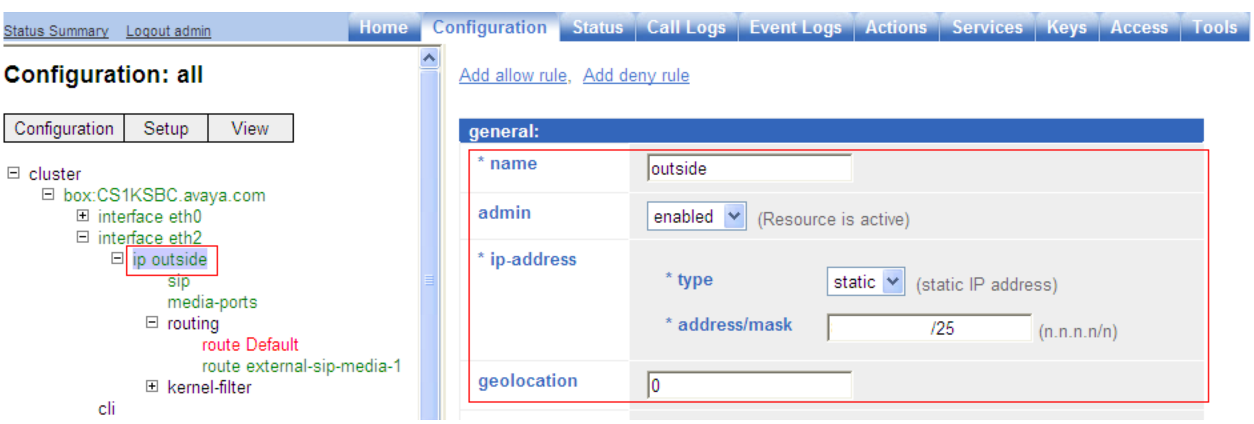
7.1. Access Avaya Aura® Session Border Controller

Access the SBC using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured.



7.2. Configuring Outside Interface

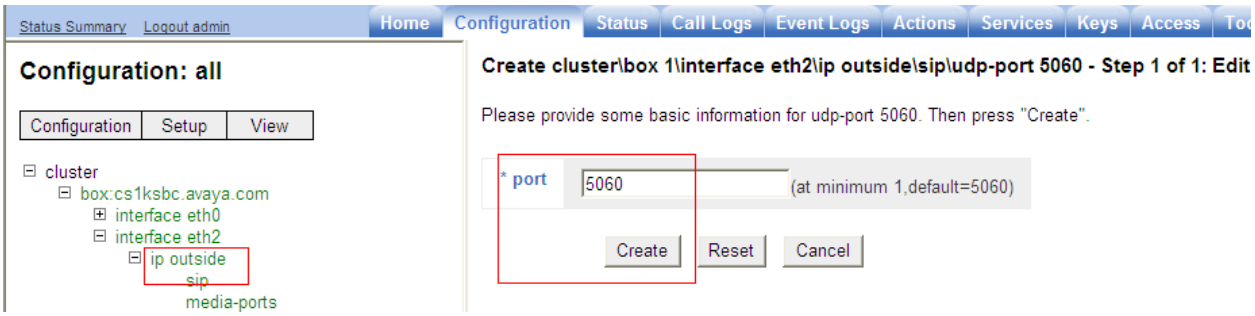
An ip address was given to the outside interface that is on the public internet. The ip address is blanked out in the screenshot below for security purposes. Click on the **Configuration** tab and browse to **cluster → interface eth2 → ip outside.**



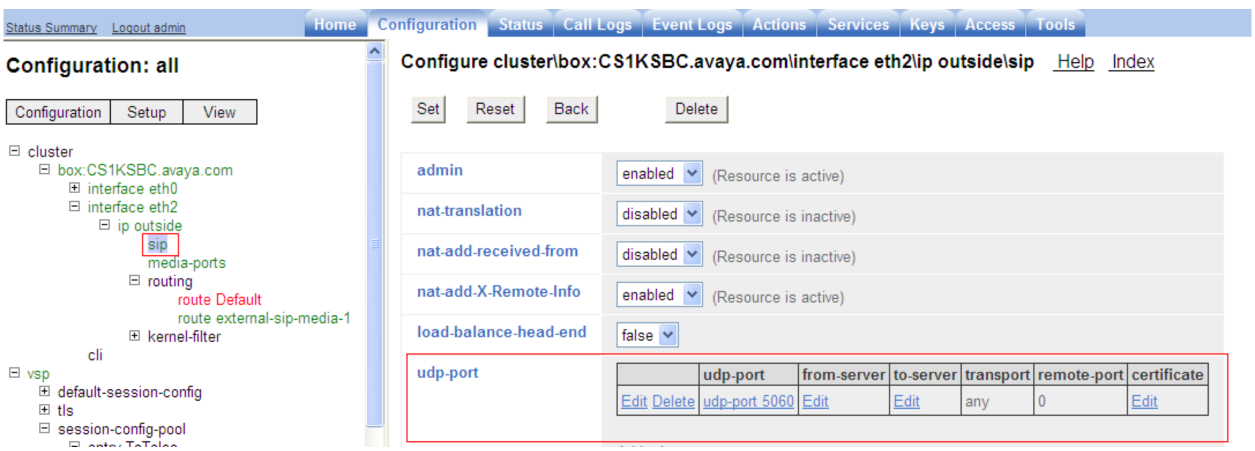
7.2.1. Configure SIP

For the outside interface a transport protocol needs to be configured. In the compliance testing UDP was used as the transport method for the SIP messaging. Click on the **Configuration** tab and browse to **cluster → interface eth2 → ip outside → sip → Add udp-port**

- **Port** Port number to be used for SIP messaging, default is **5060**

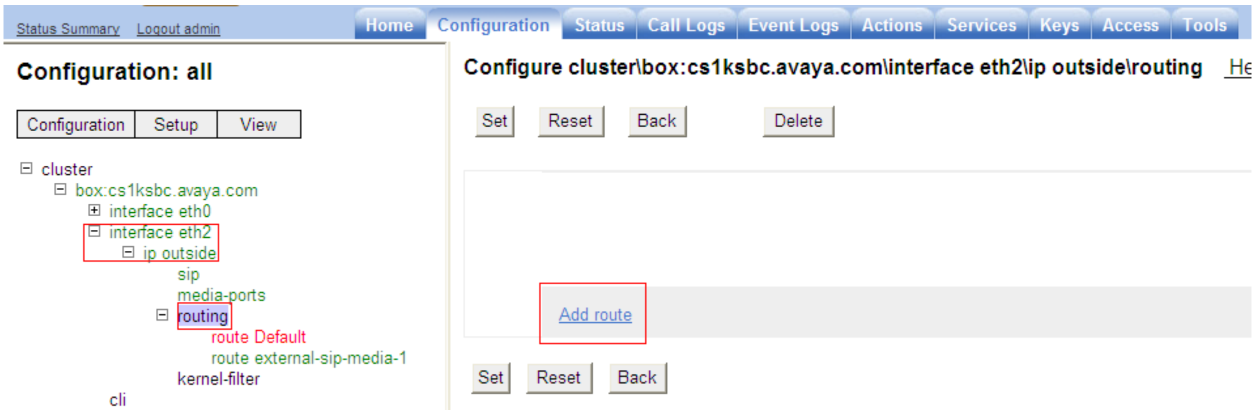


The newly created UDP port is shown below.



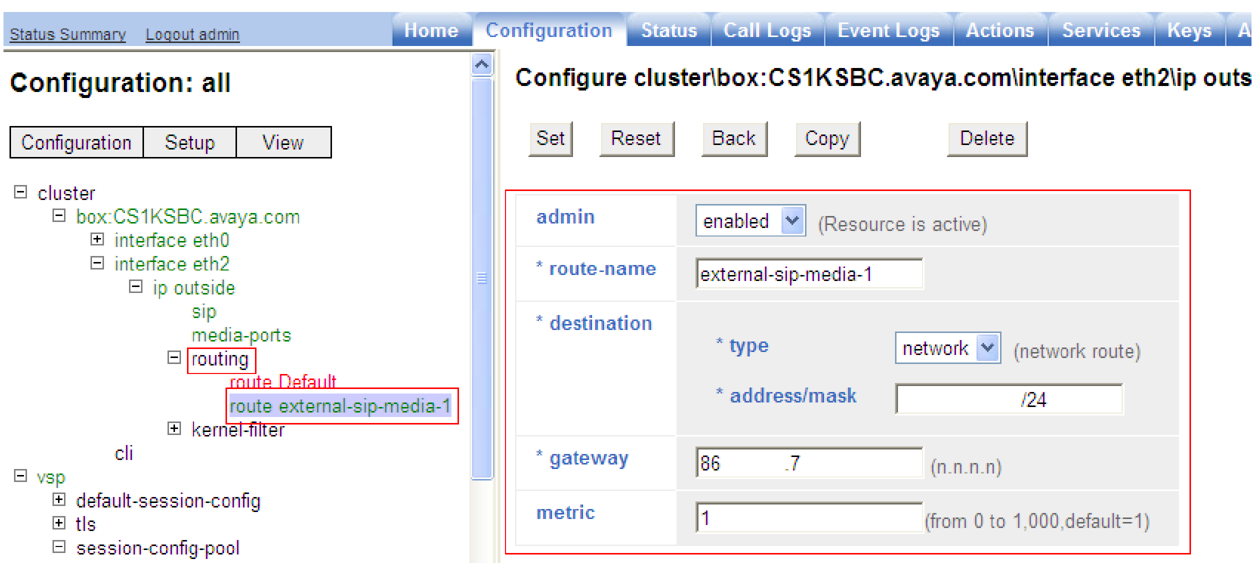
7.2.2. Configure Routing

For the outside interface routing needs to be configured to advise the SIP traffic how to route out to BTW/HIPCOM’s network from the outside interface of the SBC. The ip address is blanked out in the screenshot below for security purposes. Click on the **Configuration** tab and browse to **cluster → interface eth2 → ip outside → routing → add route.**



The following values need to be added for the new route that is being created:

- **admin** Enables or disables this route configuration
- **route name** Enter a name for the route
- **destination type** Use network as the network route
- **destination address/mask** The destination address is the subnet used by the service provider and mask
- **gateway** Sets the gateway or next hop IP address for the packet
- **metric** Associates a cost for the route, default is 1



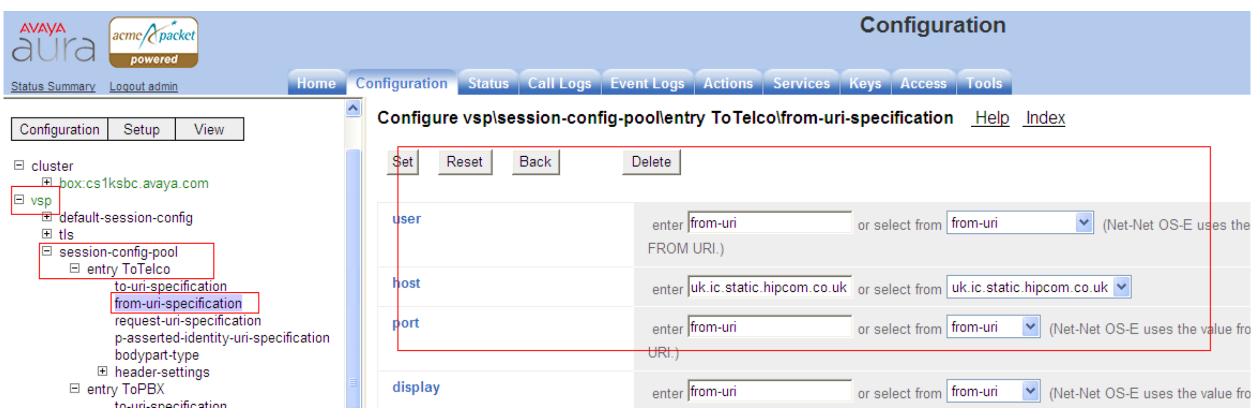
7.3. Configuring VSP

7.3.1. Configure Session-Config-Pool Entry ToTelco

In the **to-uri-specification** a valid host was added for BTW/HIPCOM. Expand **vsp → session-config pool → entry ToTelco → to-uri-specification**. For the testing domain **uk.ic.static.hipcom.co.uk** was used as shown below.

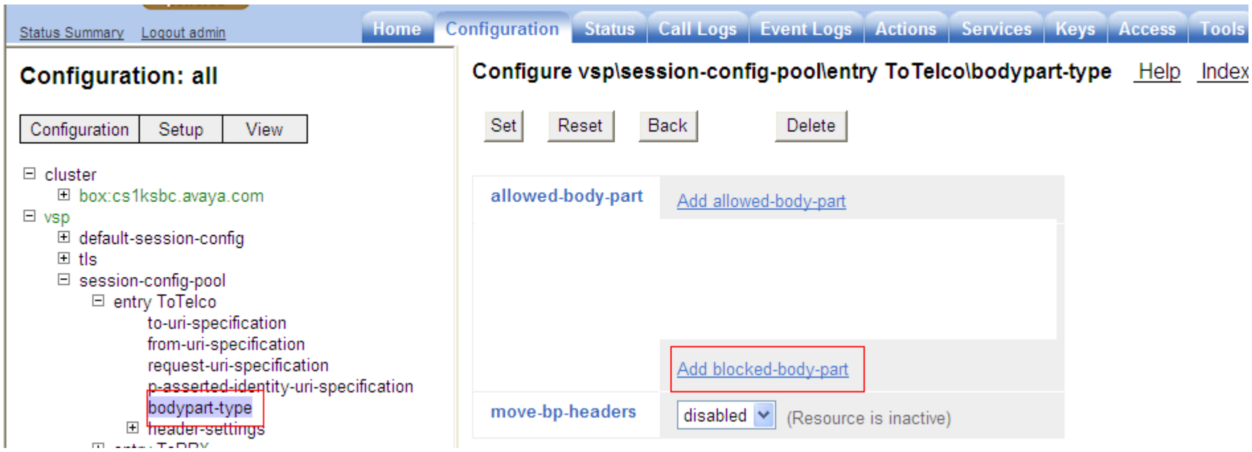
Notes: The domain name used by BTW/HIPCOM will change depending on access method, please consult BTW/HIPCOM to confirm what this will be.

In the **from-uri-specification** a valid host was added for BTW/HIPCOM. Expand **vsp → session-config pool → entry ToTelco → to-from-specification**. For the testing domain **uk.ic.static.hipcom.co.uk** as seen below.



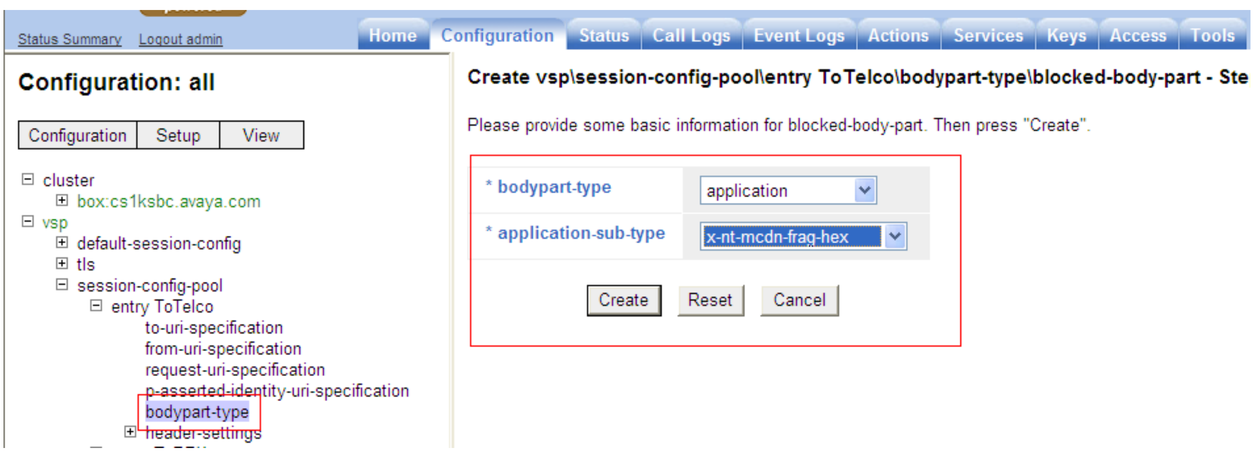
Repeat the same process to the change the host value in the request and p-asserted-identity headers to **uk.ic.static.hipcom.co.uk**, this is not shown**.**

Two new blocked body parts were also added. This sets the body part types to prohibit during the session. Any body sections that contain this type are removed from the message before forwarding. Expand **vsp → session-config pool → entry ToTelco → bodypart type → Add blocked-body-part.**

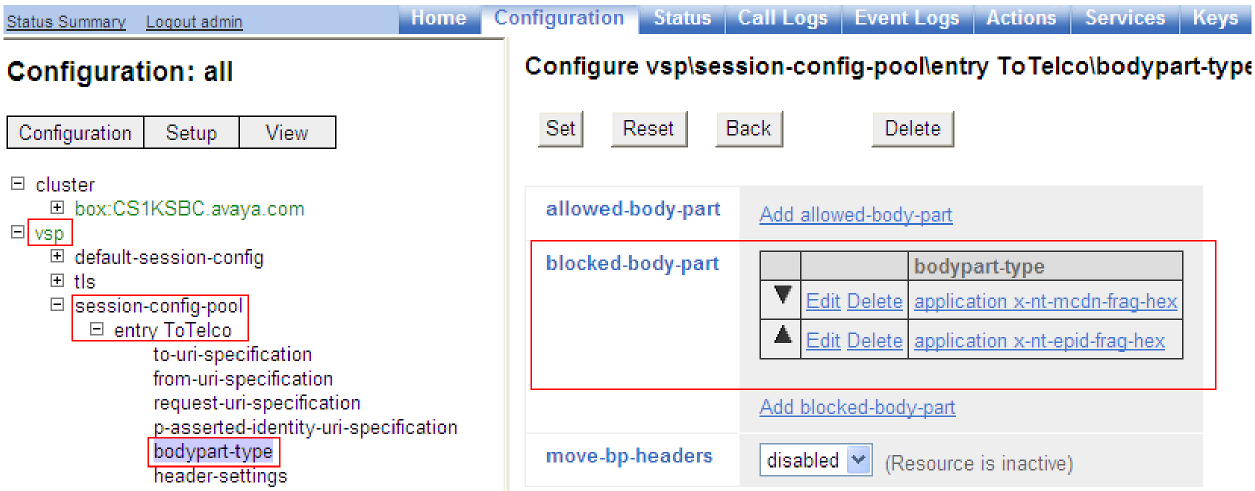


The following values need to be added for the blocked-body-part.

- **bodypart-type** set this to **application**
- **application-sub-type** set this to **x-nt-mcdn-frag-hex**



The same process outlined above needs to be added for blocked body part **x-nt-epid-frag-hex**. In the below screenshot both blocked body parts are shown.

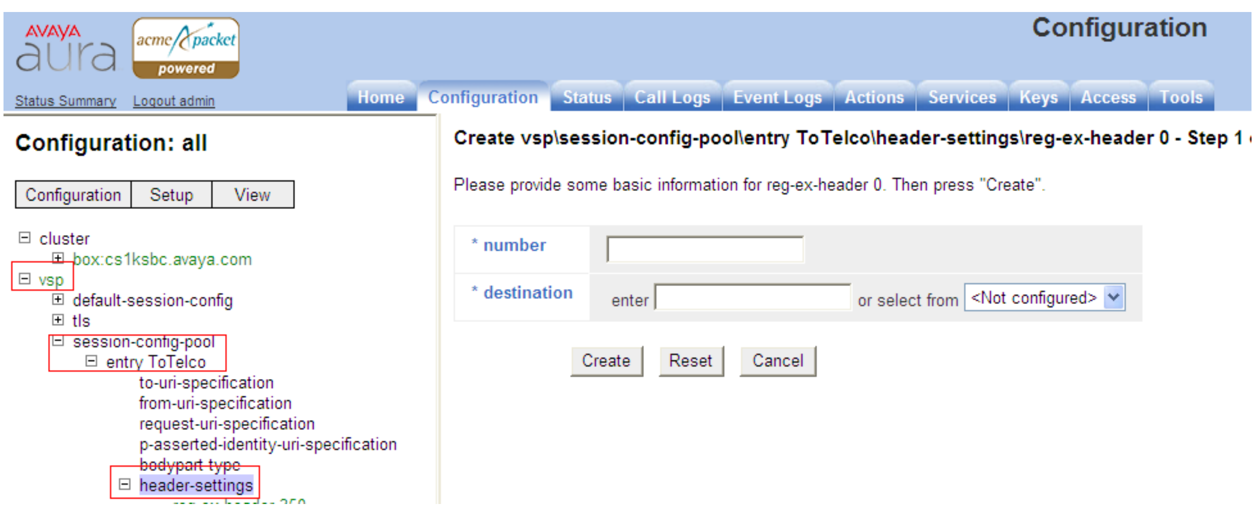


7.3.2. Creating a new reg-ex-header

A reg-ex header was also added to change the host portion in the History-Info header so that it had a value of **uk.ic.static.hipcom.co.uk**. To define a new reg-ex-header click on **header settings → add reg-ex-header** and a new screen will appear as shown below.

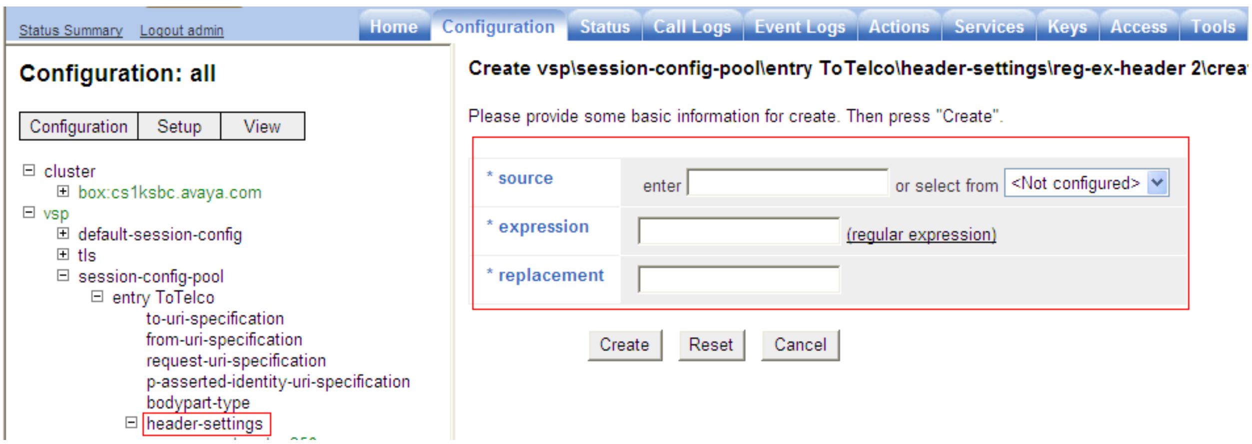
- **number:** Enter a unique number for the reg-ex-header
- **destination:** Enter the header that is going to be manipulated, in this case **History-Info**

Then click on **Create**.

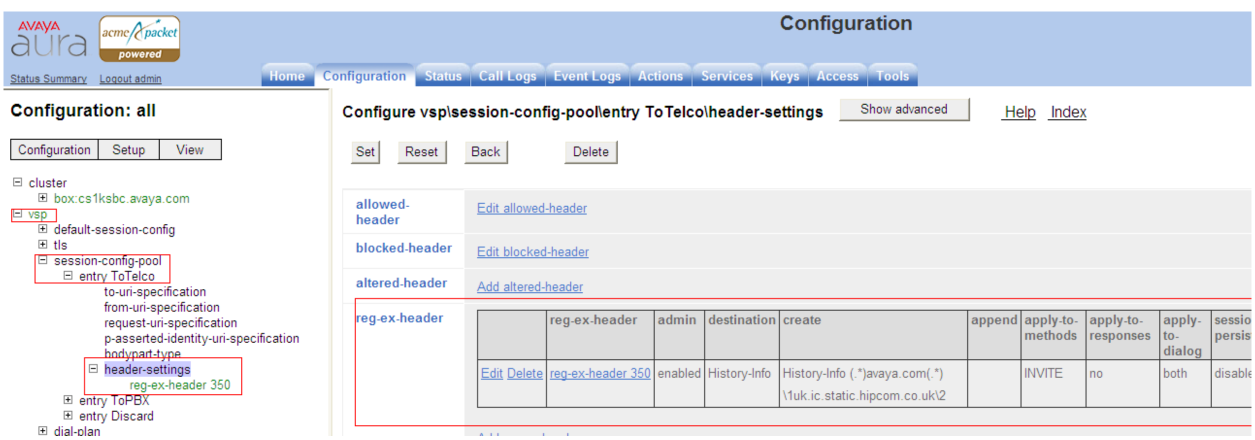


A new window will appear. In the create field click on **configure**. A new window appears as shown below with the following fields.

- **source** Enter the header that is going to be manipulated, in this case **History-Info**
- **expression** The expression to match in the manipulation, avaya.com - **(.*)avaya.com(.*)**
- **replacement** The expression to replace in the manipulation, uk.ic.static.hipcom.co.uk - **\1uk.ic.static.hipcom.co.uk\2**

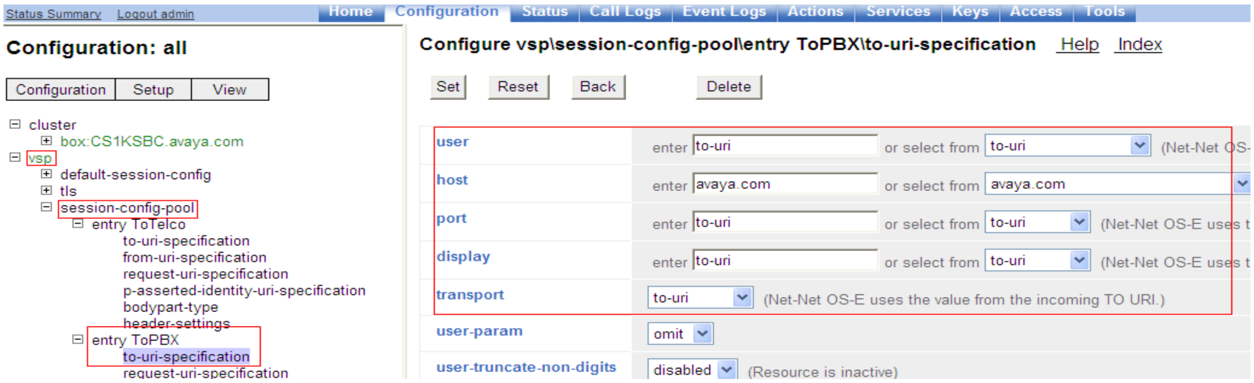


The following screen shows the actual reg-ex-header that was configured for BTW/HIPCOM.

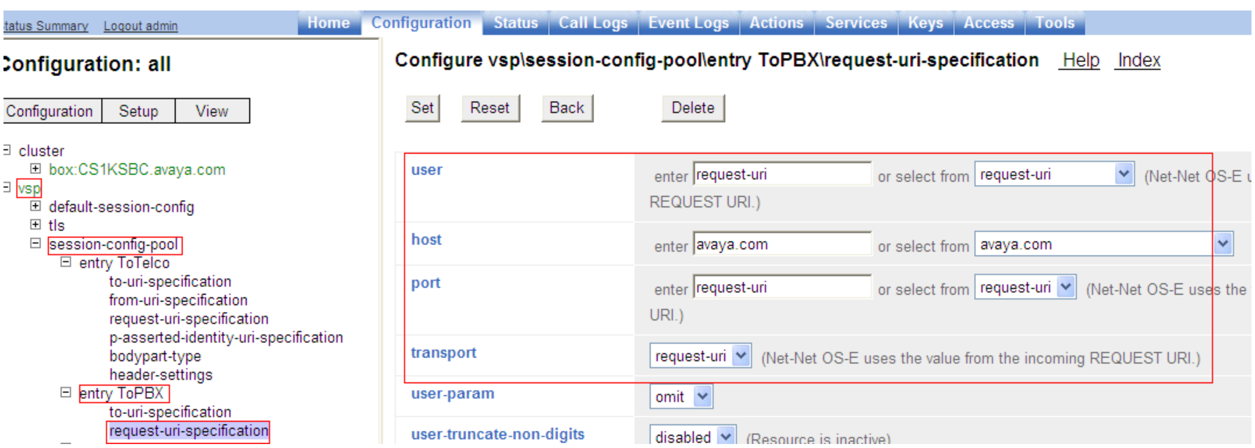


7.3.3. Configure Session-Config-Pool Entry ToPBX

In the **to-uri-specification** a new **host** was added **avaya.com.** This is the SIP domain used in the enterprise and is configured in **Section 6.1**. Expand **vsp → session-config pool → entry ToPBX → to-uri-specification**. Enter **avaya.com** in the **host** field and click on **Set**.



In the **request-uri-specification** a new **host** was added **avaya.com**. This is the SIP domain used in the enterprise and is configured in **Section 6.1**. Expand **vsp → session-config pool → entry ToPBX → request-uri-specification**. Enter **avaya.com** in the **host** field and click on **Set.**



7.3.4. Configuring Enterprise

In the **sip-gateway-Telco** the **domain** name used is **avaya.com**. A newly added server was created for BTW/HIPCOM's SBC; information needed here is the ip address, port and transport protocol. Click on the **Configuration** tab and browse to **vsp → enterprise → servers → sip-gateway Telco → server-pool.** Click the **Add server** link**.**

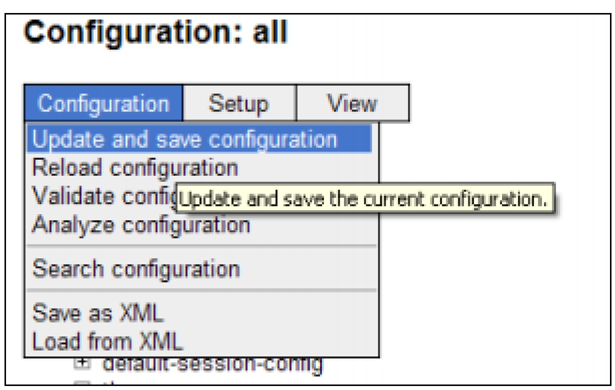
A new window will appear as shown below

- **server name** Enter the name of the server pool configuration instance that you want to create or modify
- **admin** Enable or disable this server, default is enabled
- **host** Specifies ip address of the service provider's SBC
- **transport** Specifies the protocol used by the server
- **port** Specifies the port used by the server for SIP traffic

All other values in other fields are set as default.

7.4. Save the Configuration

To save the configuration, click on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



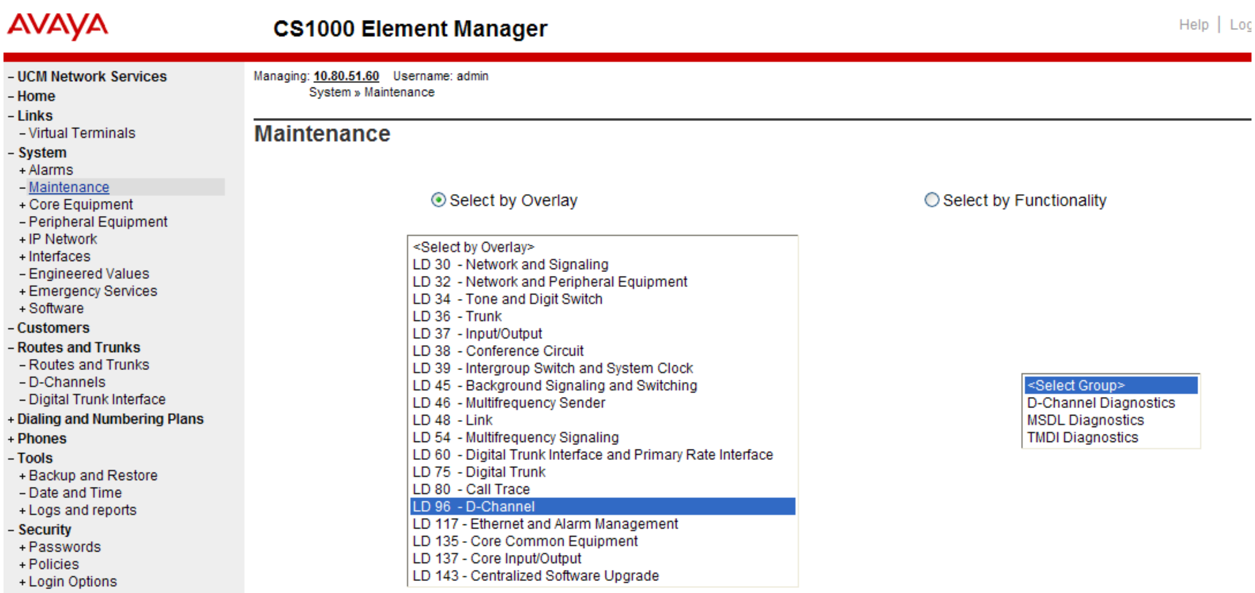
8. BT Wholesale/HIPCOM Service Provider Configuration

The configuration of BTW/HIPCOM's equipment used to support the SIP trunk service is outside of the scope for these application notes and will not be covered. To obtain further information on BTW/HIPCOM's equipment and system configuration please contact an authorised BTW/HIPCOM representative.

9. Verification

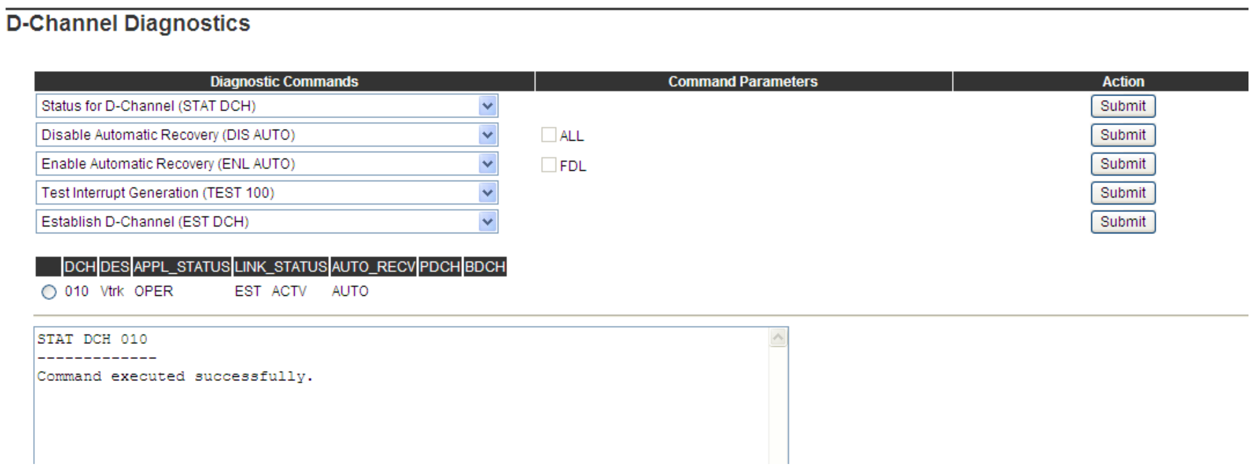
9.1. Verify Avaya Communication Server 1000E Operational Status

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 Group** 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**

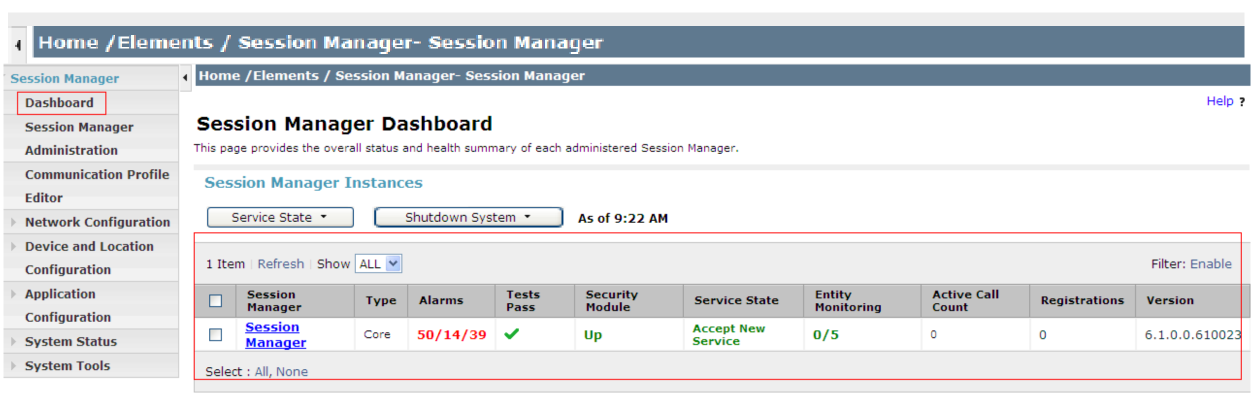


9.2. Verify Avaya Aura® Session Manager Operational Status

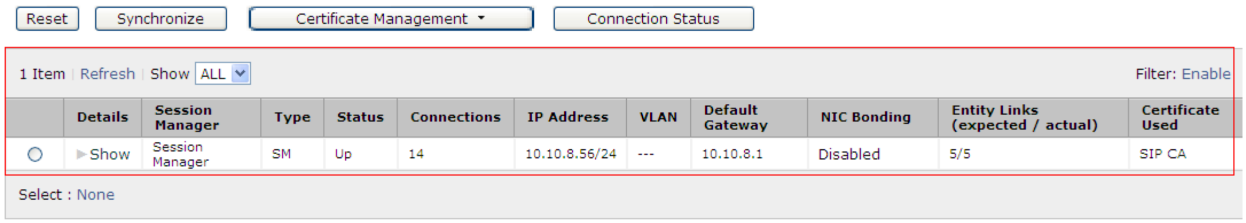
9.2.1. Verify Avaya Aura® Session Manager is Operational

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.

- **Tests Pass**
- **Security Module**
- **Service State**

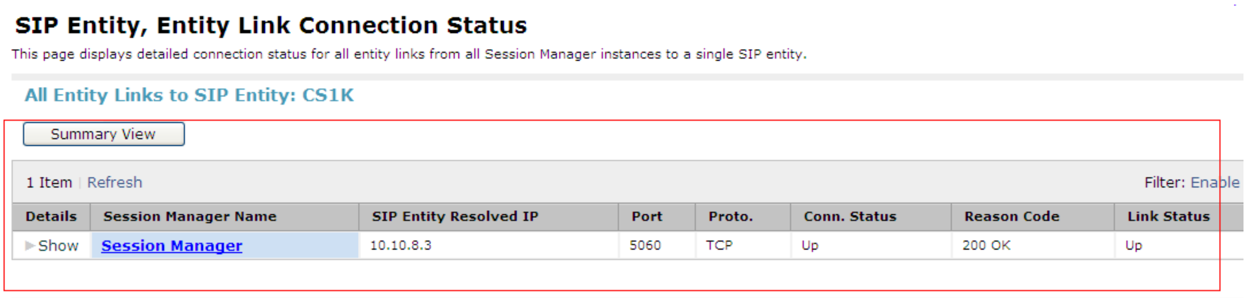


Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.



9.2.2. Verify SIP Entity Link Status

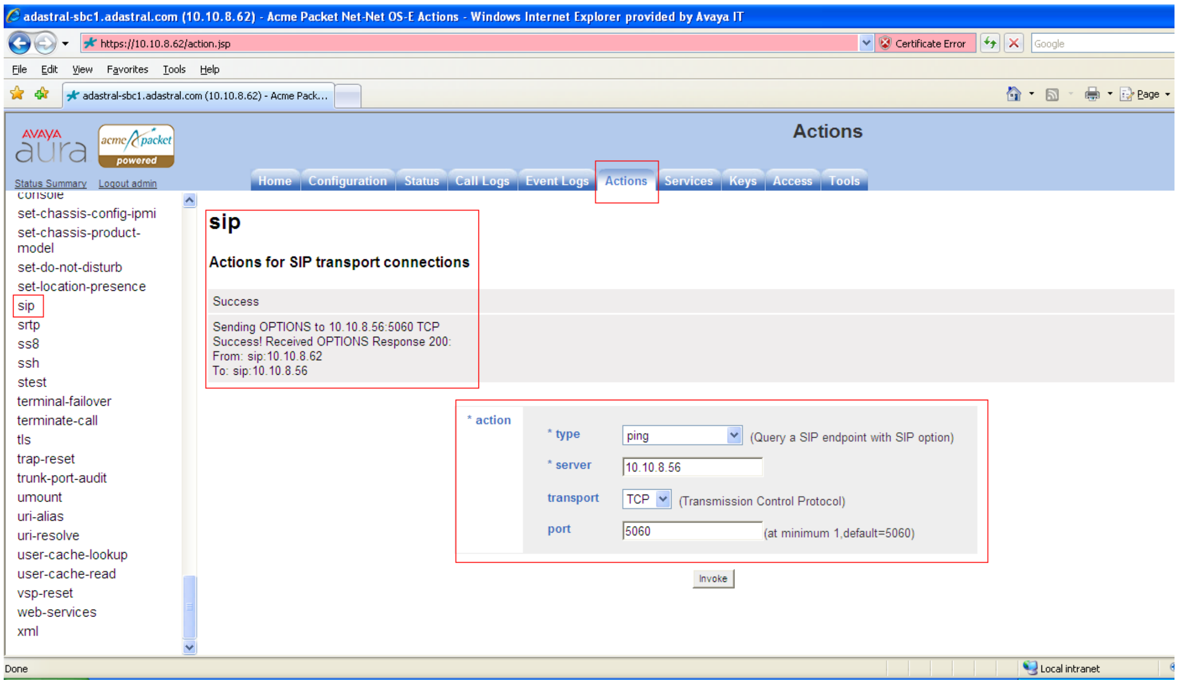
Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links. Select the SIP Entity for Communication Server 1000Efrom the **All Monitored SIP Entities** table (not shown) to open the **SIP Entity, Entity Link Connection Status** page. In the **All Entity Links to SIP Entity: CS1000 Rel7.5** table, verify the **Conn. Status** for the link is **Up** as shown below.



Verify the SIP link is up between the Session Manager and the SBC by going through the same process as outlined above but selecting the SIP Entity for the SBC in the **All Monitored SIP Entities** table (not shown).

9.3. Verify Avaya Aura® Session Border Controller Operational Status

Navigate to **Actions** → select **SIP (left hand menu)** → action type select **PING** and enter the server you want to verify, in the screenshot below it is with the Session Manager (SM100) interface. The SBC sends an option message and a 200 OK is sent back.



10. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to BTW/HIPCOM's SIP Service. The observation noted during this testing is detailed 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] Avaya Aura® Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.

[2] Installing and Configuring Avaya Aura® Session Manager, available at http://support.avaya.com.

[3] Avaya Aura® Session Manager Case Studies, available at http://support.avaya.com

[4] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

[5] Administering Avaya Aura® Session Manager, Doc ID 03-603324, available at http://support.avaya.com

[6] IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313, available at http://support.avaya.com
[7] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116, available at http://support.avaya.com
[8] Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02, available at http://support.avaya.com
[9] Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-509, available at http://support.avaya.com
[10] Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125, available at http://support.avaya.com

Appendix A
Avaya Communication Server 1000E Software

Communication Server 1000E call server patches and plug ins

```
08/04/11 10:25:28
TID: 008808096

VERSION 4021

System type is - Communication Server 1000E/CP PM

CP PM - Pentium M 1.4 GHz

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

RELEASE 7
ISSUE 50 Q  +
IDLE_SET_DISPLAY Avaya 7.5
DepList 1: core Issue: 02(created: 2010-11-30 15:12:45 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2010-12-06 15:33:54(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-12-01 08:31:36(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

LOADWARE VERSION: PSWV 100
INSTALLED LOADWARE PEPS : 0
ENABLED PLUGINS : 0
```

Communication Server 1000E call server deplists

```
VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2011-05-24 10:13:35 (est)) ALTERED

IN-SERVICE PEPS
PAT# CR #           PATCH REF #     NAME      DATE        FILENAME       SPECINS
012  wi00843623     ISS1:1OF1       p30731_1  16/06/2011  p30731_1.cpl   YES
013  WI00843571     ISS1:1OF1       p30627_1  16/06/2011  p30627_1.cpl   NO
014  wi00871739     ISS1:1OF1       p30856_1  16/06/2011  p30856_1.cpl   NO
015  wi00852365     ISS1:1OF1       p30707_1  16/06/2011  p30707_1.cpl   NO
016  wi00852389     ISS1:1OF1       p30641_1  16/06/2011  p30641_1.cpl   NO
017  wi00839134     ISS1:1OF1       p30698_1  16/06/2011  p30698_1.cpl   YES
018  wi00856702     ISS1:1OF1       p30573_1  16/06/2011  p30573_1.cpl   NO
019  wi00857566     ISS1:1OF1       p30766_1  16/06/2011  p30766_1.cpl   NO
020  wi00850521     ISS1:1OF1       p30709_1  16/06/2011  p30709_1.cpl   YES
021  wi00860722     ISS1:1OF1       p30784_1  16/06/2011  p30784_1.cpl   YES
022  wi00863876     ISS1:1OF1       p30787_1  16/06/2011  p30787_1.cpl   NO
023  WI00853473     ISS1:1OF1       p30625_1  16/06/2011  p30625_1.cpl   NO
024  wi00854130     ISS1:1OF1       p30443_1  16/06/2011  p30443_1.cpl   NO
025  wi00875425     ISS1:1OF1       p30943_1  16/06/2011  p30943_1.cpl   NO
026  wi00853658     ISS1:1OF1       p30990_1  16/06/2011  p30990_1.cpl   NO
027  wi00875701     ISS1:1OF1       p30942_1  16/06/2011  p30942_1.cpl   NO
028  wi00853031     ISS1:1OF1       p30531_1  16/06/2011  p30531_1.cpl   NO
029  wi00877367     ISS1:1OF1       p30534_1  16/06/2011  p30534_1.cpl   NO
030  wi00871969     ISS1:1OF1       p30768_1  16/06/2011  p30768_1.cpl   NO
031  wi00886321     ISS1:1OF1       p31009_1  16/06/2011  p31009_1.cpl   NO
032  WI00836334     ISS1:1OF1       p30481_1  16/06/2011  p30481_1.cpl   NO
033  wi00836182     ISS1:1OF1       p30450_1  16/06/2011  p30450_1.cpl   NO
034  wi00858335     ISS1:1OF1       p30819_1  16/06/2011  p30819_1.cpl   NO
035  wi00860279     ISS1:1OF1       p30789_1  16/06/2011  p30789_1.cpl   NO
036  wi00866570     ISS1:1OF1       p30477_1  16/06/2011  p30477_1.cpl   NO
```

```
037  wi00854415      ISS1:1OF1       p30593_1  16/06/2011  p30593_1.cpl   NO
038  WI00836292      ISS1:1OF1       p30554_1  16/06/2011  p30554_1.cpl   NO
039  WI00839794      ISS1:1OF1       p28647_1  16/06/2011  p28647_1.cpl   NO
040  wi00824257      ISS1:1OF1       p30447_1  16/06/2011  p30447_1.cpl   NO
041  wi00827950      ISS2:1OF1       p30471_2  16/06/2011  p30471_2.cpl   NO
042  wi00879814      ISS1:1OF1       p30970_1  16/06/2011  p30970_1.cpl   NO
043  WI00854150      ISS1:1OF1       p30468_1  16/06/2011  p30468_1.cpl   NO
044  wi00873382      ISS1:1OF1       p30832_1  16/06/2011  p30832_1.cpl   NO
045  wi00853178      ISS1:1OF1       p30719_1  16/06/2011  p30719_1.cpl   NO
046  wi00869695      ISS1:1OF1       p30654_1  16/06/2011  p30654_1.cpl   NO
047  wi00834382      ISS1:1OF1       p30548_1  16/06/2011  p30548_1.cpl   NO
048  wi00836472      ISS1:1OF1       p30626_1  16/06/2011  p30626_1.cpl   NO
049  wi00854409      ISS1:1OF1       p30479_1  16/06/2011  p30479_1.cpl   NO
050  WI00728461      ISS1:1OF1       p30346_1  16/06/2011  p30346_1.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2011-05-25 10:18:44(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-05-25 04:41:04(est)
```

Communication Server 1000E signaling server service updates

```
Product Release: 7.50.17.00
In system patches: 0
In System service updates: 8
PATCH#  IN_SERVICE   DATE       SPECINS   REMOVABLE   NAME
0       Yes          07/02/11   NO        YES         cs1000-baseWeb-7.50.17.01-1.i386.000
1       Yes          07/02/11   NO        YES         cs1000-linuxbase-7.50.17.04-00.i386.000
2       Yes          07/02/11   NO        YES         cs1000-sps-7.50.17-01.i386.000
3       Yes          07/02/11   NO        YES         cs1000-shared-pbx-7.50.17-01.i386.000
4       Yes          07/02/11   NO        YES         cs1000-bcc-7.50.17.03-00.i386.000
5       Yes          07/02/11   NO        YES         cs1000-Jboss-Quantum-7.50.17.01-1.i386.000
6       Yes          07/02/11   NO        YES         cs1000-vtrk-7.50.17-11.i386.000
7       Yes          07/02/11   NO        YES         cs1000-dmWeb-7.50.17.04-00.i386.001
There is no SP in loaded status.
The last applied SP: Service_Pack_Linux_7.50_17_20110118.ntl, It is a STANDARD SP.
Has been applied by user nortel on Mon Feb  7 14:59:01 2011
```

Communication Server 1000E system software

```
Product Release: 7.50.17.00
Base Applications
   base                         7.50.17     [patched]
   NTAFS                        7.50.17
   sm                           7.50.17
   cs1000-Auth                  7.50.17
   Jboss-Quantum                7.50.17     [patched]
   lhmonitor                    7.50.17
   baseAppUtils                 7.50.17
   dfoTools                     7.50.17
   nnnm                         7.50.17
   cppmUtil                     7.50.17
   oam-logging                  7.50.17
   dmWeb                        n/a         [patched]
   baseWeb                      n/a         [patched]
   ipsec                        7.50.17
   Snmp-Daemon-TrapLib          7.50.17
   ISECSH                       7.50.17
   patchWeb                     7.50.17
   EmCentralLogic               7.50.17
Application configuration: CS+SS+EM
Packages: CS+SS+EM
Configuration version:     7.50.17-00
   cs                           7.50.17
   dbcom                        7.50.17     [patched]
   cslogin                      7.50.17
   sigServerShare               7.50.17     [patched]
   csv                          7.50.17
   tps                          7.50.17
```

```
   vtrk                          7.50.17      [patched]
   pd                            7.50.17
   sps                           7.50.17      [patched]
   ncs                           7.50.17
   gk                            7.50.17
   EmConfig                      7.50.17
   emWeb_6-0                     7.50.17      [patched]
   emWebLocal_6-0                7.50.17
   csmWeb                        7.50.17
   bcc                           7.50.17      [patched]
   ftrpkg                        7.50.17
   cs1000WebService_6-0          7.50.17
   managedElementWebService      7.50.17
   mscAnnc                       7.50.17
   mscAttn                       7.50.17
   mscConf                       7.50.17
   mscMusc                       7.50.17
   mscTone                       7.50.17
```

Appendix B
Sample Avaya Aura® Session Border Controller Configuration File

As noted in **Section 7**, in the following sample SBC configuration file, public IP Addresses have been masked for security purposes.

```
#
#  Copyright (c) 2004-2011  Acme Packet Inc.
#  All Rights Reserved.
#
#  File: /cxc/cxc.cfg
#  Date: 09:23:04 Wed 2011-08-03
#
config cluster
 config box 1
  set hostname cs1ksbc.avaya.com
  set timezone Etc/GMT
  set name cs1ksbc.avaya.com
  set identifier 00:ca:fe:56:07:85
  config interface eth0
   config ip inside
    set ip-address static 10.10.8.62/24
    config ssh
    return
    config snmp
     set trap-target 10.10.8.61 162
     set trap-filter generic
     set trap-filter dos
     set trap-filter sip
     set trap-filter system
    return
    config web
    return
    config web-service
     set protocol https 8443
     set authentication certificate "vsp\tls\certificate ws-cert"
    return
    config sip
     set udp-port 5060 "" "" any 0
     set tcp-port 5060 "" "" any 0
     set tls-port 5061 "" "" TLS 0 "vsp\tls\certificate aasbc.p12"
    return
    config icmp
    return
    config media-ports
    return
    config routing
     config route Default
      set gateway 10.10.8.1
     return
     config route Static0
      set destination network 192.11.13.4/30
```

```
      set gateway 10.10.8.60
     return
     config route Static2
      set admin disabled
     return
     config route Static3
      set admin disabled
     return
     config route Static4
      set admin disabled
     return
     config route Static5
      set admin disabled
     return
     config route Static6
      set admin disabled
     return
     config route Static7
      set admin disabled
     return
    return
   return
  return
  config interface eth2
   config ip outside
    set ip-address static xx.xx.xx.xx/25
    config sip
     set udp-port 5060 "" "" any 0
     set tcp-port 5060 "" "" any 0
    return
    config media-ports
    return
    config routing
     config route Default
      set admin disabled
     return
     config route external-sip-media-1
      set destination network xx.xx.xx.0/24
      set gateway xx.xx.xx.xx
     return
    return
    config kernel-filter
    return
   return
  return
  config cli
   set prompt cs1ksbc.avaya.com
  return
 return
return

config services
 config event-log
  config file access
   set filter access info
   set count 3
```

```
  return
  config file system
   set filter system info
   set count 3
  return
  config file errorlog
   set filter all error
   set count 3
  return
  config file db
   set filter db debug
   set filter dosDatabase info
   set count 3
  return
  config file management
   set filter management info
   set count 3
  return
  config file peer
   set filter sipSvr info
   set count 3
  return
  config file dos
   set filter dos alert
   set filter dosSip alert
   set filter dosTransport alert
   set filter dosUrl alert
   set count 3
  return
  config file krnlsys
   set filter krnlsys debug
   set count 3
  return
 return
return

config master-services
 config database
  set media enabled
 return
return

config vsp
 set admin enabled
 config default-session-config
  config media
   set anchor enabled
   set rtp-stats enabled
  return
  config sip-directive
   set directive allow
  return
  config log-alert
   set apply-to-methods-for-filtered-logs
  return
  config header-settings
```

```
  return
  config third-party-call-control
   set handle-refer-locally disabled
  return
 return
 config tls
  config default-ca
   set ca-file /cxc/certs/sipca.pem
  return
  config certificate ws-cert
   set certificate-file /cxc/certs/ws.cert
  return
  config certificate aasbc.p12
   set certificate-file /cxc/certs/aasbc.p12
   set passphrase-tag aasbc-cert-tag
  return
 return
 config session-config-pool
  config entry ToTelco
   config to-uri-specification
    set host uk.ic.static.hipcom.co.uk
   return
   config from-uri-specification
    set host uk.ic.static.hipcom.co.uk
   return
   config request-uri-specification
    set host uk.ic.static.hipcom.co.uk
   return
   config p-asserted-identity-uri-specification
    set user from-uri
    set host uk.ic.static.hipcom.co.uk
   return
   config bodypart-type
    set blocked-body-part application x-nt-mcdn-frag-hex
    set blocked-body-part application x-nt-epid-frag-hex
   return
   config header-settings
    config reg-ex-header 350
     set destination History-Info
     set create History-Info (.*)avaya.com(.*)
"\1uk.ic.static.hipcom.co.uk\2"
    return
   return
  return
  config entry ToPBX
   config to-uri-specification
    set host avaya.com
   return
   config request-uri-specification
    set host avaya.com
   return
  return
  config entry Discard
   config sip-directive
   return
  return
```

```
 return
 config dial-plan
  config route Default
   set priority 500
   set location-match-preferred exclusive
   set session-config vsp\session-config-pool\entry Discard
  return
  config source-route FromTelco
   set peer server "vsp\enterprise\servers\sip-gateway PBX"
   set source-match server "vsp\enterprise\servers\sip-gateway Telco"
  return
  config source-route FromPBX
   set peer server "vsp\enterprise\servers\sip-gateway Telco"
   set source-match server "vsp\enterprise\servers\sip-gateway PBX"
  return
 return
 config enterprise
  config servers
   config sip-gateway PBX
    set domain avaya.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToPBX
    config server-pool
     config server PBX1
      set host 10.10.7.61
      set transport TCP
     return
    return
   return
   config sip-gateway Telco
    set domain avaya.com
    set failover-detection ping
    set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
    config server-pool
     config server Telco1
      set host xx.xx.xx.xx
     return
    return
   return
  return
 return
 config dns
  config resolver
   config server 10.10.7.100
   return
  return
 return
 config settings
  set read-header-max 8191
 return
return

config external-services
return
```

```
config preferences
 config gui-preferences
  set enum-strings RequestURIStringSource uk.ic.static.hipcom.co.uk
  set enum-strings GeneralURIStringSource uk.ic.static.hipcom.co.uk
  set enum-strings SIPSourceHeader History-Info
  set enum-strings RequestURIStringSource avaya.com
 return
return

config access
 config permissions superuser
  set cli advanced
 return
 config permissions read-only
  set config view
  set actions disabled
 return
 config users
  config user admin
   set password 0x00c3da084a51ad08836ccad49283a1babca7bb621d17ce3b5ff788fd94
   set permissions access\permissions superuser
  return
  config user cust
   set password 0x003eb1074c8d3d30b83e1a77de17d2ea682827c07690a9bab5af1823d5
   set permissions access\permissions read-only
  return
  config user init
   set password 0x00108cd8c2081746d3bc7b7a0a0600f43178a84eb6342452c47b8f6bbd
   set permissions access\permissions superuser
  return
  config user craft
   set password 0x006273d242a7d36dd742e5917d626593833ab915aede6611f4f383e324
   set permissions access\permissions superuser
  return
  config user dadmin
   set password 0x004f373649700feb0ae40cb9d23587eeb0675e6c063a7d4ce5ea20ba77
   set permissions access\permissions read-only
  return
 return
return

config features
return
```

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.