Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to support Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate SIP Trunk Services - Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Vodafone Netherlands Office Voice, Vodafone Netherlands OneVoice Corporate SIP Trunk Services and an Avaya SIP enabled enterprise solution. The Vodafone Netherlands Office Voice trunk is used for calls to and from fixed line PSTN locations, Vodafone Netherlands OneVoice Corporate trunk is used for calls to and from mobile telephone numbers as well as providing the ability for enterprise users to reach Vodafone mobile telephone numbers assigned to their account by dialing a four digit short code. The Avaya solution consists of Avaya Aura® Session Border Controller, Avaya Aura® Session Manager and Avaya Communication Server 1000E Vodafone Netherlands are 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 Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Vodafone Netherlands and an Avaya SIP enabled enterprise solution using Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate SIP Trunk Services. These services are offerred in conjunction with each other as a total solution, for clarity theses services will be collectivly refrered to in this document as Vodafone Netherlands SIP Trunk Solution. The Avaya solution consists of Avaya Aura® Session Border Controller (AASBC), Avaya Aura® Session Manager and Avaya Communication Server 1000E (CS1000E). Customers using this Avaya SIP enabled Enterprise solution with Vodafone Netherlands SIP Trunk Solution are able to place and receive calls via standards-based SIP trunks as an alternative to legacy Analogue or digital trunks.

The Vodafone Netherlands SIP Trunk Solution referenced within these Application Notes is designed for business customers. The solution provides two connections to the enterprise, Vodafone Netherlands Office Voice is a fixed line SIP trunk and Vodafone Netherlands OneVoice Corporate is a mobile SIP trunk. The Vodafone Netherlands Office Voice trunk is used for calls to and from fixed line PSTN locations, Vodafone Netherlands OneVoice Corporate trunk is used for calls to and from mobile telephone numbers as well as providing the ability for enterprise users to reach Vodafone mobile telephone numbers assigned to their account by dialing a four digit short code.

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 CS1000E, Session Manager and AASBC. The enterprise site was configured to use the SIP Trunk Solution provided by Vodafone Netherlands.

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 PSTN calls to various phone types. Phone types included SIP, Unistim and digital telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing calls from the enterprise site were completed via Vodafone Netherlands to PSTN destinations.
- Outgoing calls from the enterprise to the PSTN were made from SIP, Unistim and Digital telephones.
- Inbound and outbound PSTN calls to/from the Avaya one-X® Communicator soft phone.
- Calls to Emergency Services (112).

- Calls using G.729, and G.711A codec's.
- Fax calls to/from a fax machine at the enterprise to a PSTN connected fax machine.
- 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 Vodafone Netherlands requiring Avaya response and sent by Avaya requiring Vodafone Netherlands response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone Netherlands SIP Trunk Service with the following observations:

- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Mobile-X handoff works from twinned desk phone with patch p30260_1.ntl loaded on the CS1000E. INVITE sent to PSTN mobile contains no SDP information without the patch loaded, Vodafone do not support an INVITE with no SDP.
- Fax calls using T.38 for inbound and outbound with G.711 or G.729 do not work. Still under investigation.

2.3. Support

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

For technical support on Vodafone Netherlands SIP trunk services, contact Vodafone Netherlands support at http://www.vodafone.nl/zakelijk/totaal_oplossingen/vast_en_mobiel/.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Vodafone Netherlands SIP Trunk Solution. The Vodafone Netherlands Office Voice connection is represented in **Figure 1** as (Fixed) and the Vodafone Netherlands OneVoice Corporate connection is represented in **Figure 1** as (Mobile). Located at the Enterprise site is a Session Border Controller, Session Manager and CS1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya one-X® Communicator, Avaya Digital telephone, Analogue telephone (not shown) and fax machine.

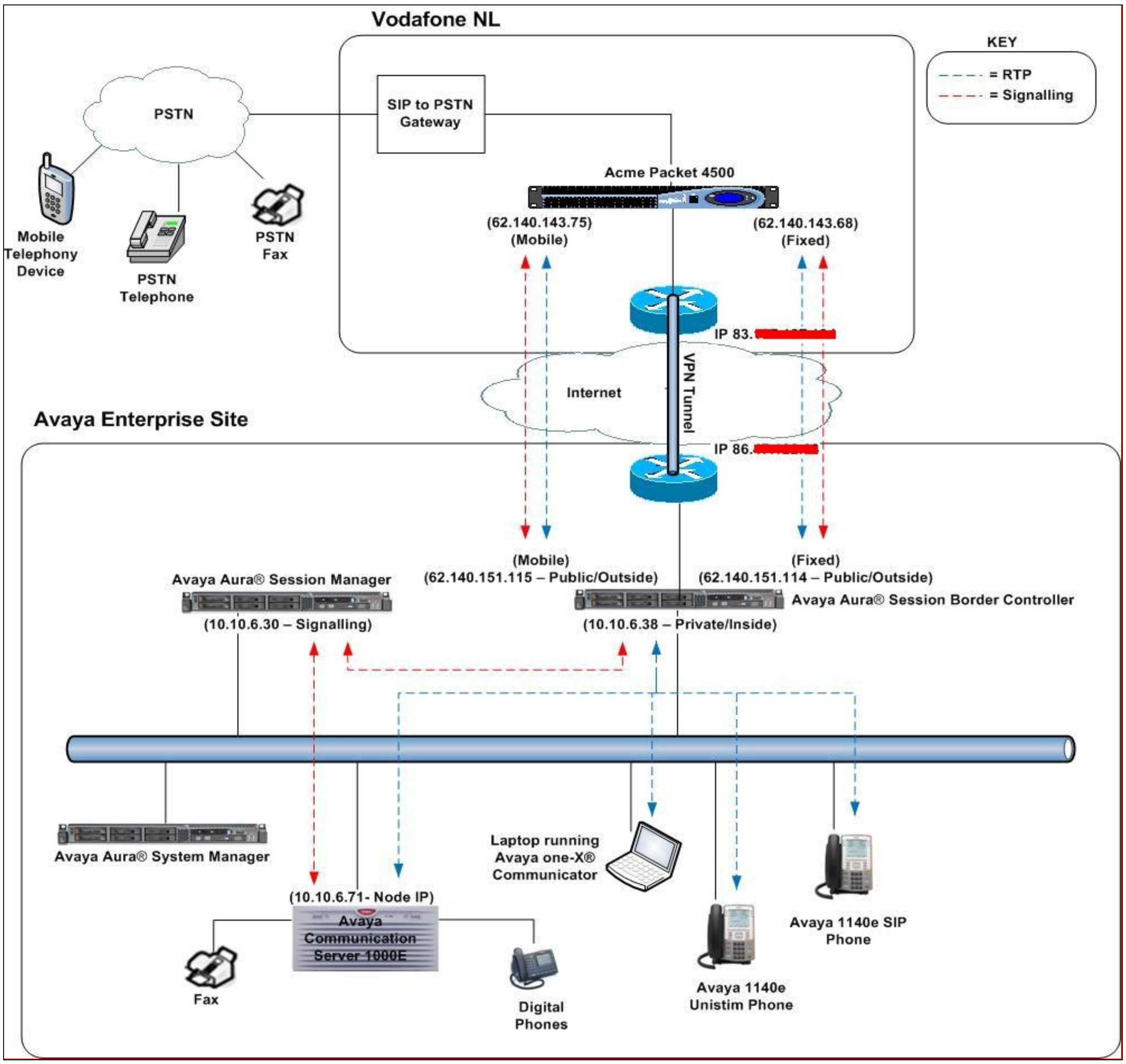


Figure 1: Avaya SIP Telephony Solution using Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate services

4. Equipment and Software Validated

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

Avaya Equipment	Software
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 Service Pack 4 (6.1.4.0.614005)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 Service Pack 4 (6.1.8.1.1551)
Avaya S8800 Server	Avaya Aura® Session Border Controller R6.1 (System Platform 6.0.3.3.3, Template E362M1)
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R7.5 Version 7.50.17 Deplist: X21 07.50Q All CS1000E patches listed in **Appendix A**
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: DSP2 AB06
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya One-X ® Communicator	Version cs6.1.0.10-263
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
Vodafone Netherlands	
Vodafone Office Voice	1.0
Vodafone OneVoice Corporate	1.0
ACME Packet Net-Net 4500	SCX6.2.0 MR-6 Patch 2 (Build 876)

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 (Analogue, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. These SIP trunks carry SIP signaling associated with Vodafone Netherlands SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from the AASBC, through which 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

Vodafone's network. Specific Communication Server 1000E configuration as 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. **Appendix A** has a list of all CS1000E patches, deplist and service packs loaded on the system.

5.1. Logging into the Avaya Communication Server 1000E

Log in using SSH to the ELAN ip address of the Call Server using a user with correct privileges. Once logged in type **csconsole** (not shown), this will take the user into the vxworks shell of the call server. Next type **logi** (not shown), 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 Vodafone Germany'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
```

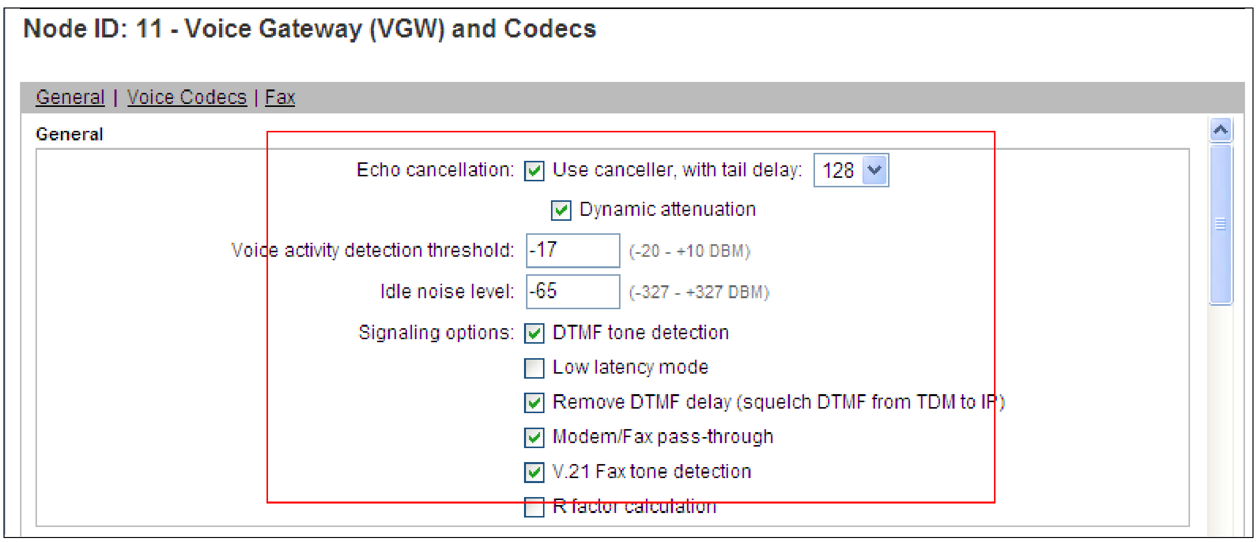
Load overlay 21, and confirm the customer is setup to use **ISDN** trunks (see below).

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

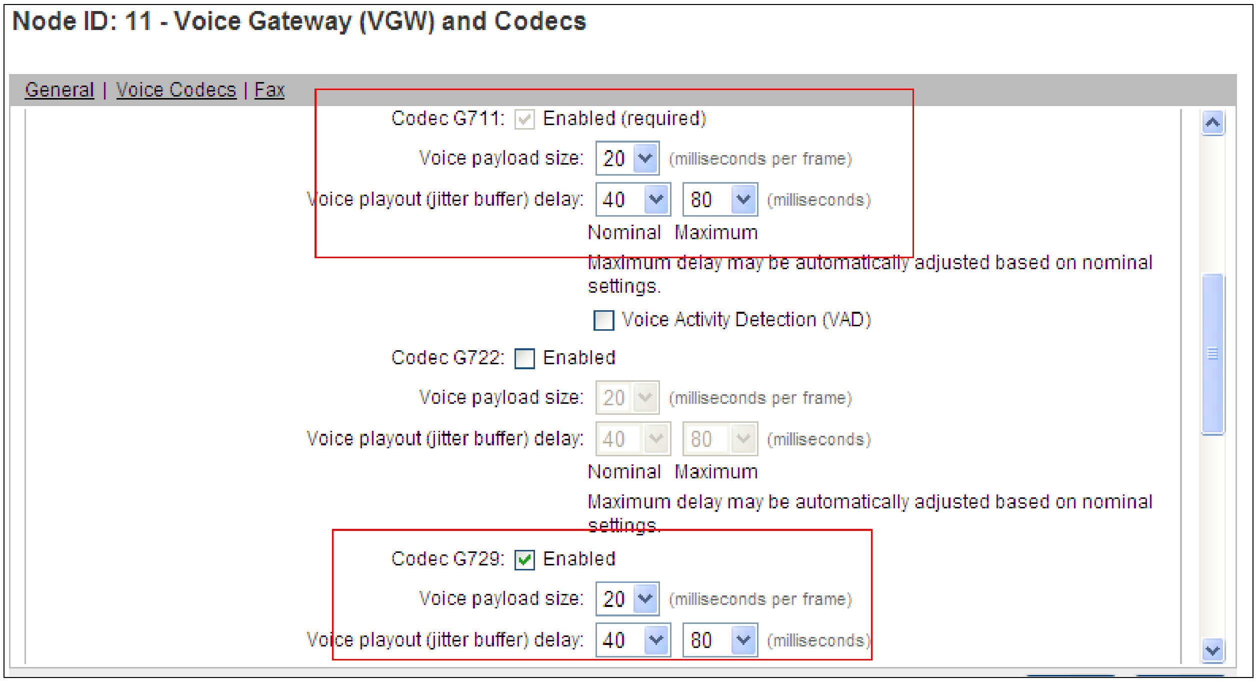
TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

5.3. Configure Codec's for Voice and FAX operation

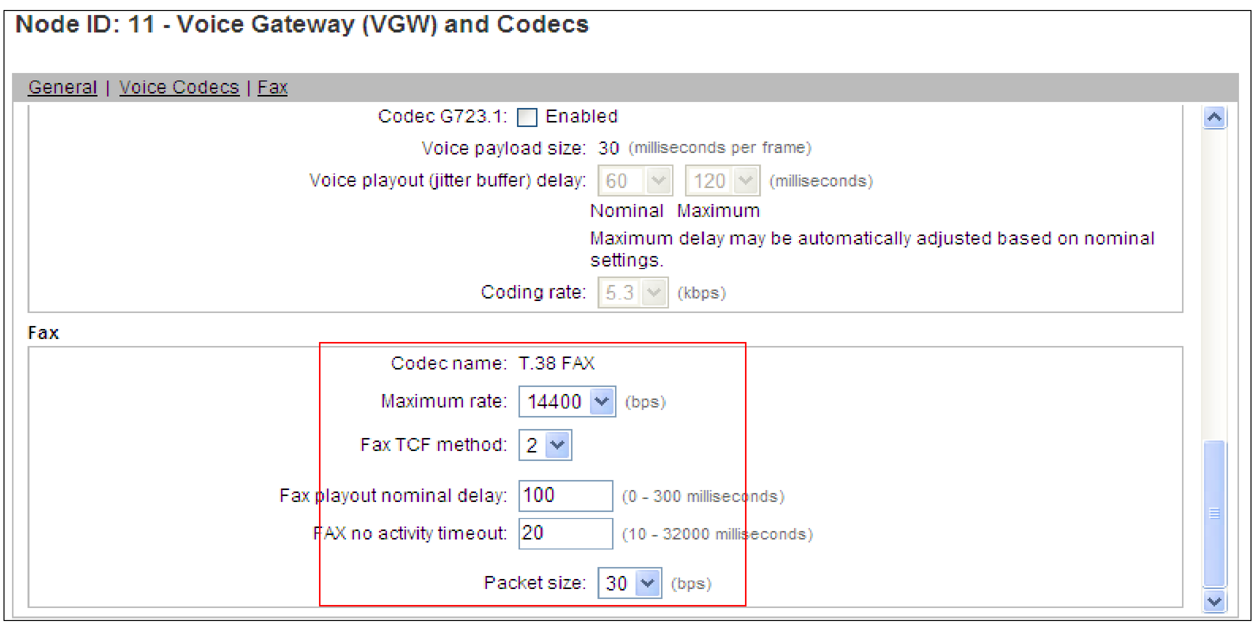
Vodafone Netherland SIP Trunk service supports G.711A/G.729A voice codec's transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → Voice Gateway VGW and Codecs** property page and configure the Communication Server 1000E General codec settings as in the next screenshot. The values highlighted below are system defaults but are required for correct operation.



Scrolling down the page, configure **G.711** and **G.729** codec settings. G.711 is enabled as default and cannot be disabled or enabled on the CS1000E.However, G.729 can be enabled or disabled, in this test G.729 was enabled and system defaults were used for payload size, jitter and delay. The relevant settings are highlighted in the following screenshot.

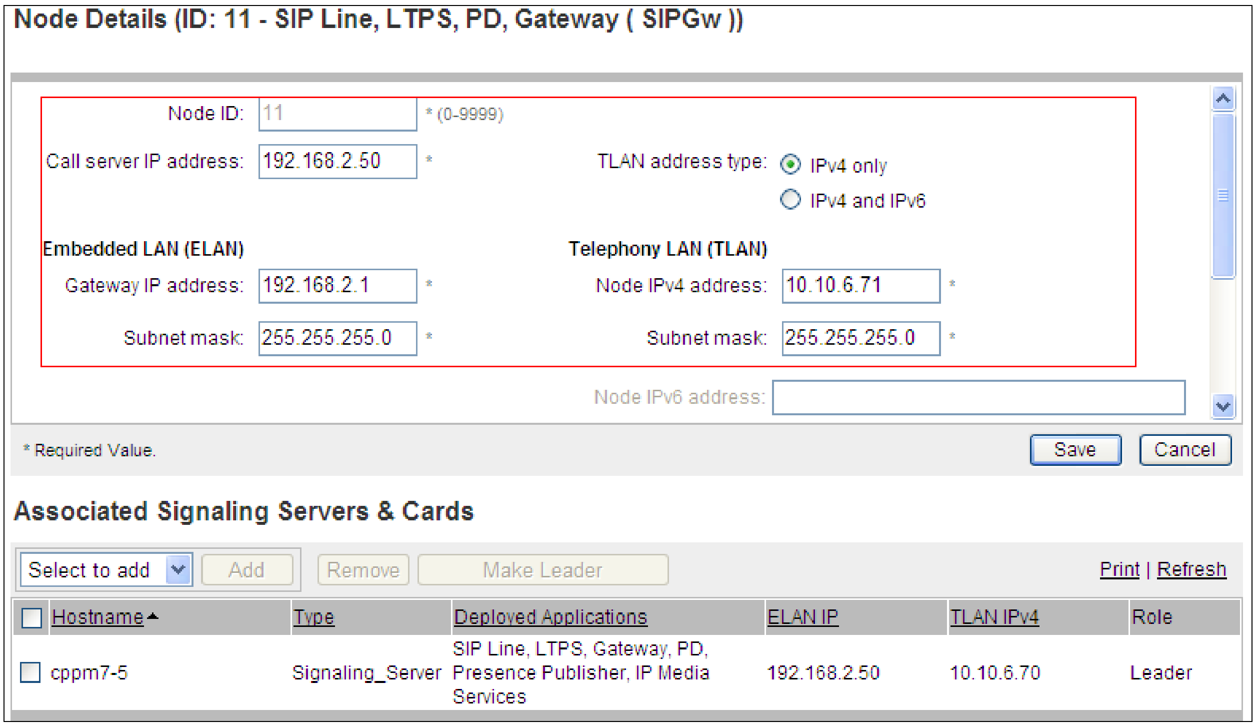


Finally configure **Fax** settings as highlighted in the screenshot below. System defaults were used. Please note T.38 cannot be disabled or enabled at the Node level and by default is enabled. Turning T.38 on or off is done at the endpoint level, by using different class of service as shown in **Section 5.7 Configure Analogue, Digital and IP Telephones**.



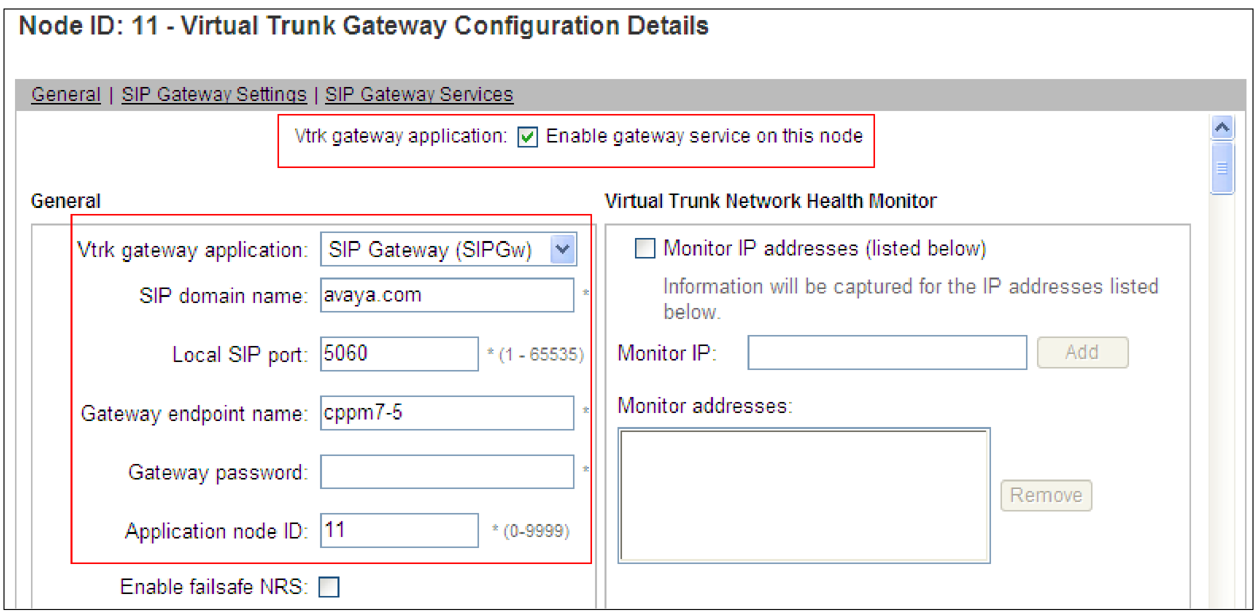
5.4. Virtual Trunk Gateway Configuration

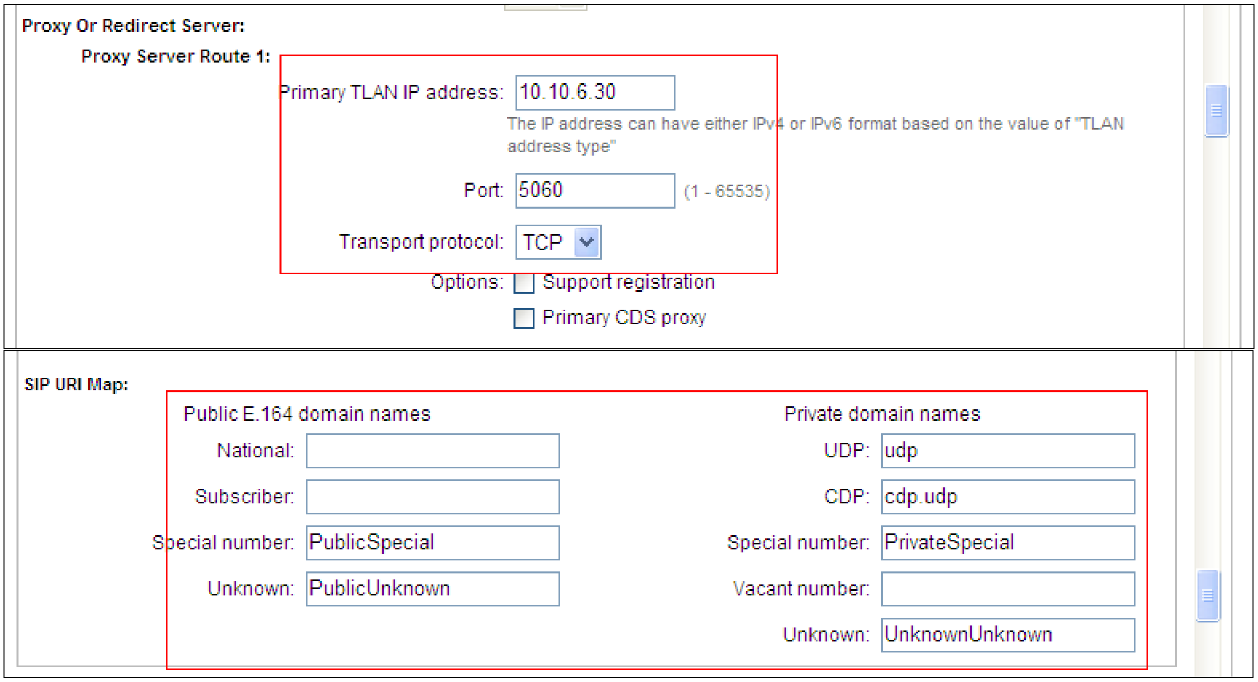
Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server 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. When an entity link is added in Session Manager for the CS1000E 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, in this case **avaya.com**. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the Session Manager, see **Section 6.2**
- **Local SIP port:** The Local SIP Port is the port to which the gateway listens. **The defaul**t 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 S**erver is used for registrat**ion 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 **11**
- **Proxy or Redirect Server: Primary TLAN IP** address is the SIP signalling interface ip address of the Session Manager. The **Transport protocol** used for SIP, in this case is **TCP**
- **SIP URI Map: Public E.164 – National, Subscriber** and **Private – Vacant Number** are left blank. All other fields in the SIP URI Map are left with default values

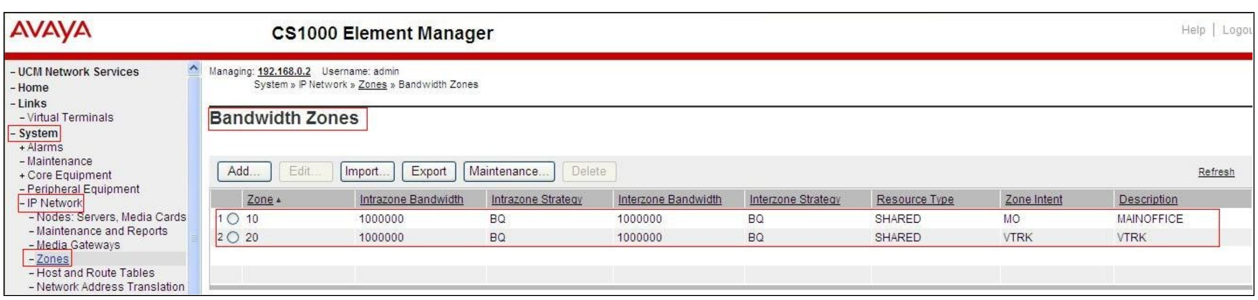




5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for Bandwidth Management. SIP trunks require a unique zone that are not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in a separate zone than SIP trunks. In the sample configuration SIP trunks use zone 20 and IP Telephones use zone 10, system defaults were used for each zone other than the parameter configured for **Zone Intent**. For SIP Trunks (zone 20), **VTRK** is configured for **Zone Intent**. For IP Telephones (zone 10), **MO** is configured for **Zone Intent**.

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 SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to Vodafone Netherland'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 5.4** The value for **ZONE** should match that used in **Section 5.5** for **SIP_VTRK**, which is zone 20. 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 0020
PCID SIP
CRID NO
NODE 11
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  NO
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
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 0020
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 Digit Manipulation Index (DMI) in overlay 87. Load **Overlay 87** at the system terminal and type **new**, at the **FEAT** prompt type **dgt** and at the **DMI** prompt set this to a unique **DMI** value. **DMI 1** is used for all traffic outgoing to the PSTN. No digits were deleted as the **DEL** prompt is set to **0**. Call type (**CTYP**) set to **UKWN**.

```
Overlay 87
REQ  new
FEAT dgt
DMI  1
DEL  0
ISPN NO
CTYP UKWN
```

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. This RLB was defined for international traffic and uses the **DMI 1**as previously entered in overlay 87.

```
Overlay 86
new
CUST 0
FEAT rlb
RLI  66
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  1
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 Trunk Steering Codes(s) (TSC) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 87. The following are some example TSC 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 00 FLEN 14 ITOH NO **RLI 66**	TSC 06 FLEN 10 ITOH NO **RLI 66**

5.7. Configure Analogue, 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 5.5** for **VIRTUALSETS**, which is zone 10.

```
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
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 9074 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 9074 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----

```
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MLNG ENG
DNDR 0
KEY  00 MCR 9072 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 9072 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
```

Analogue telephones are also configured using **Overlay 20**, the following example shows an Analogue 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. In the class of service (**CLS**) field **DTN** is required if the telephone uses DTMF dialing. Values **FAXA** and **MPTD** configure the port for T.38 Fax transmissions.

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

ERL 00000
WRLS NO
DN   9071
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
```

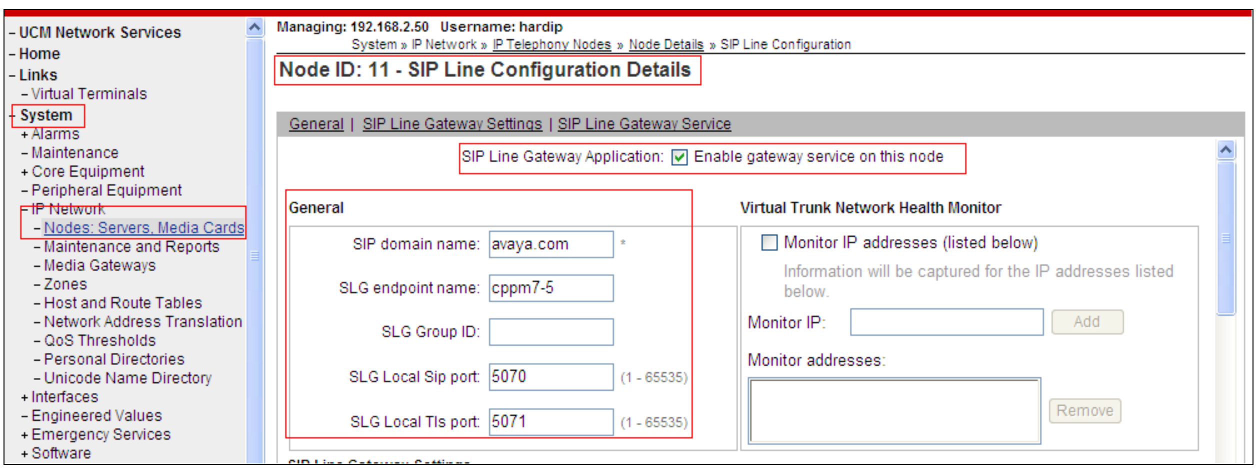
5.8. Configure the SIP Line Gateway Service

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 7. 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**.



5.9. 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 **MAINOFFICE** in **Section 5.5** 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 9079
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

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```

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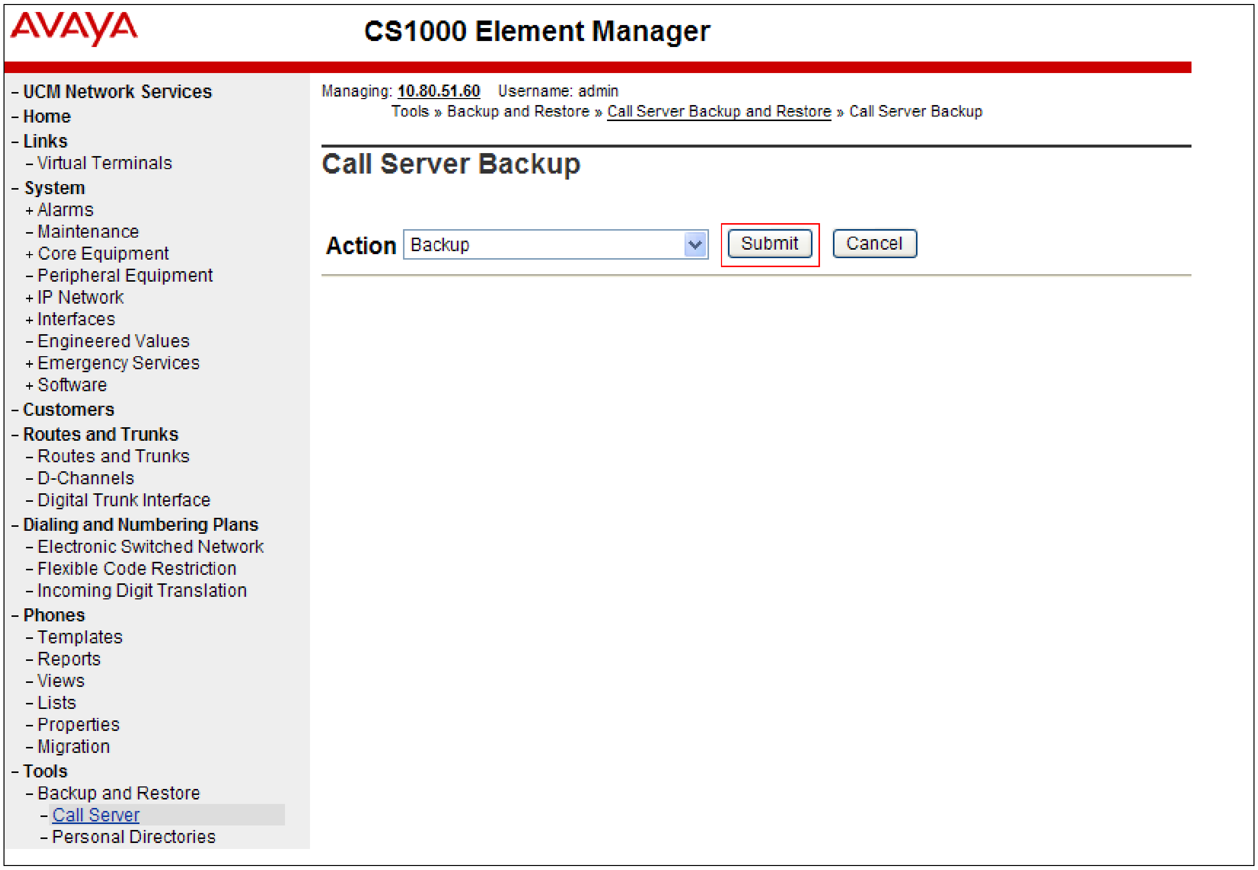
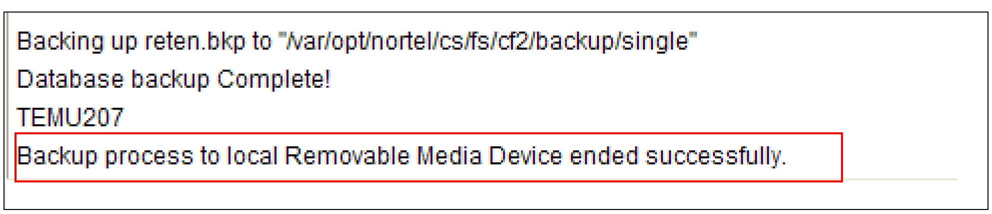
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA  PKCH MWTD DVLD
CROD CROD
CPND_LANG ENG
RCO  0
HUNT
LHK  0
PLEV 02
PUID
DANI NO
AST
IAPG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

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 9079 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 789079 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
```

5.10. 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.

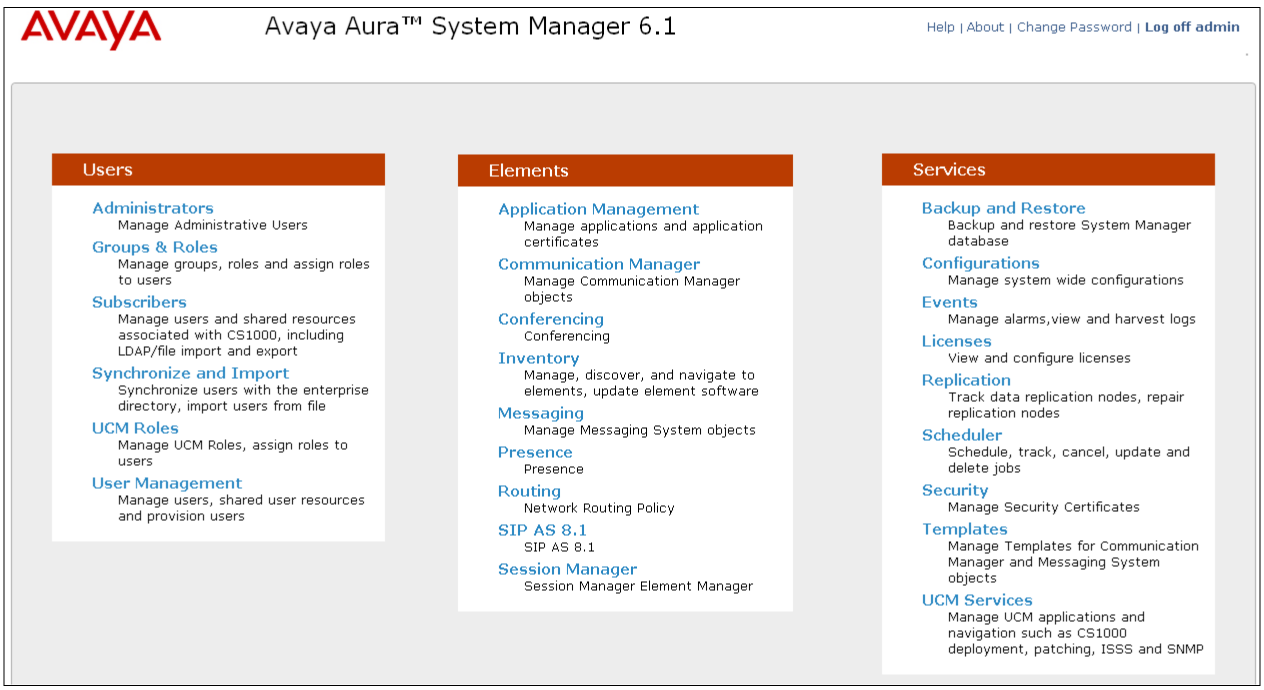
6. Configuring Avaya Aura® Session Manager

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

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

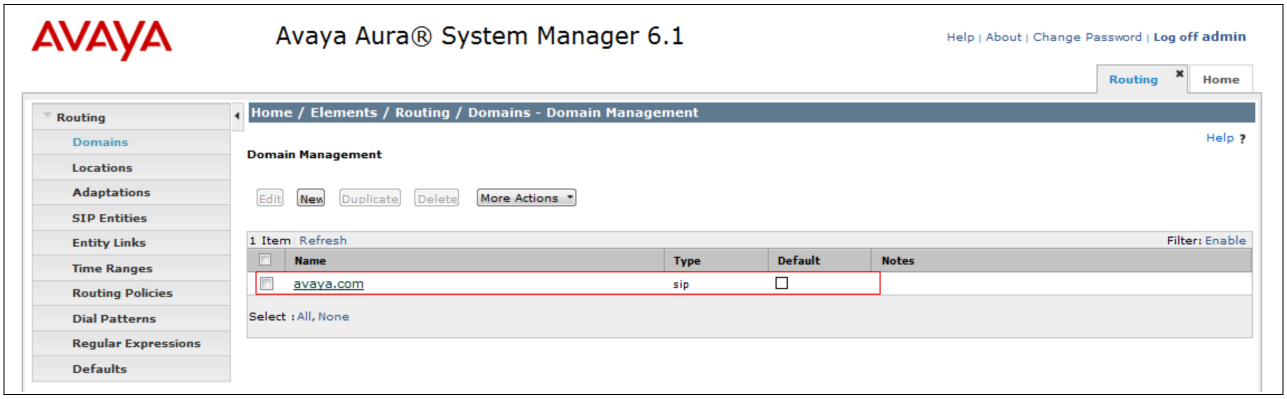
6.1. Log in to Avaya Aura® System Manager

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.



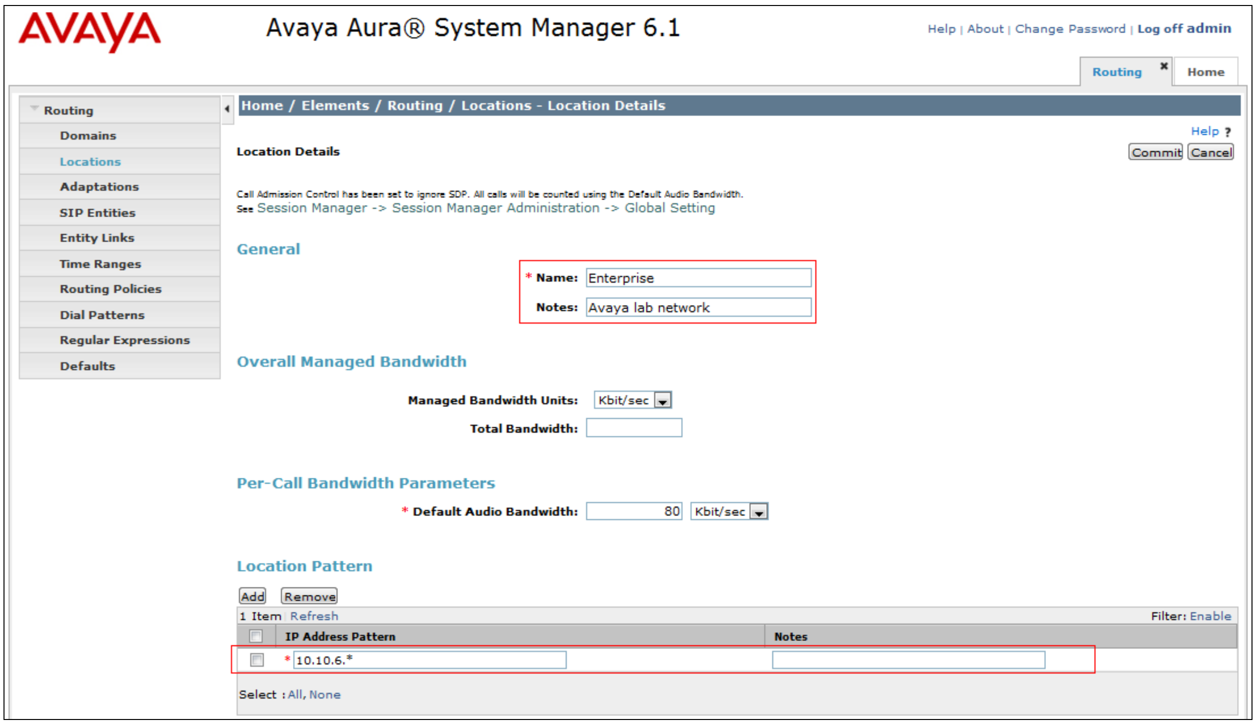
6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu (not shown) and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **avaya.com**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes (Not shown). The screen below shows the SIP domain that was previously configured.



6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field enter an informative name for the location and optionally a description for the location in the **Notes** field. Scroll to the bottom of the page and under **Location Pattern**, click **Add,** then enter an **IP Address Pattern** in the resulting new row, '*' is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.

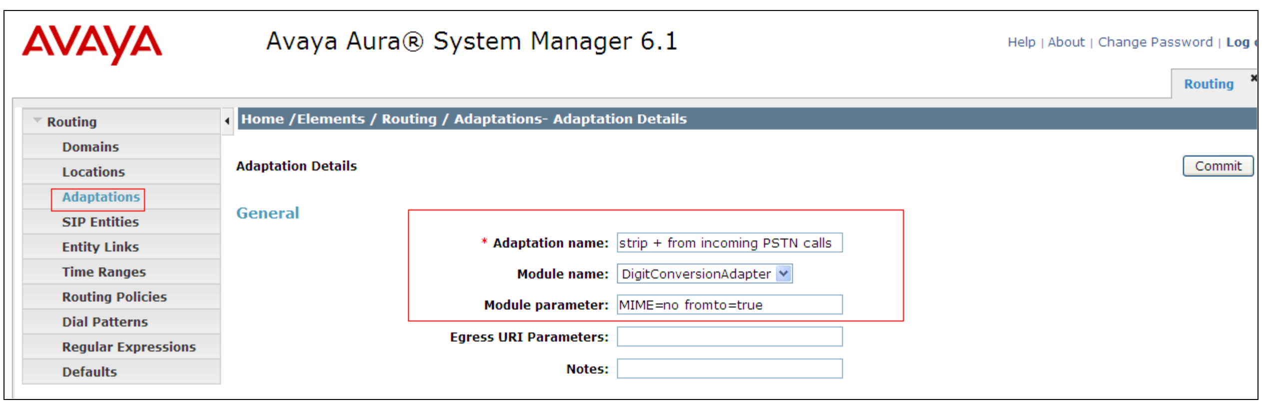


6.4. Administer Adaptation Module

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 Session Manager to ensure ingress messages have the hostname **avaya.com** when they are sent to the CS1000E. Also the adaptation was used to strip MIME messages before being sent on to Vodafone. Vodafone does not support MIME. 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, in the sample configuration **strip + from incoming PSTN calls** was used
- **Module Name:** **<click to add module>** from the drop down list and enter "DigitConversionAdapter" in the resulting **New Module Name** field
- **Module Parameter:** Enter **fromto=true** to allow the From and To headers to be modified by Session Manager (i.e., in addition to other headers such as the P-Asserted-Identity and Request-URI headers)
  Enter **MIME=no** to have Session Manager strip MIME message bodies on egress to Vodafone SBC, such that only SDP is present in the message body sent to Vodafone's SBC

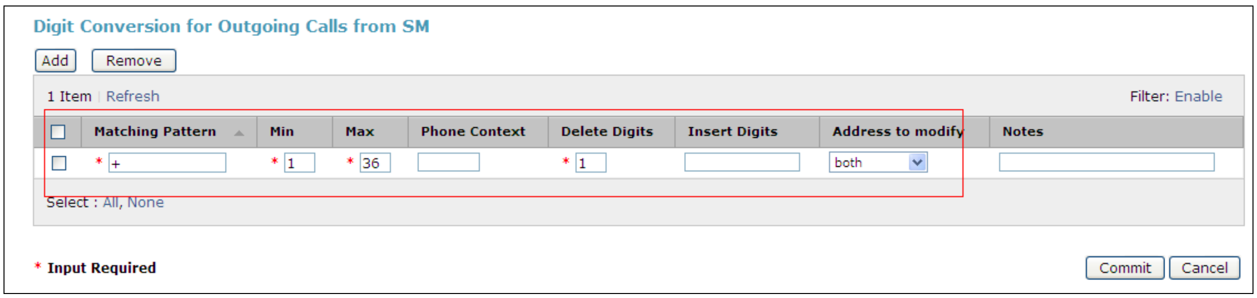
The whole string in module parameter is **MIME=no fromto=true**



Scroll down and make corresponding changes in the **Digit Conversion for Outgoing Calls from SM** section for calls from Vodafone to CS1000E users.

- **Matching Pattern:** In the sample configuration, + was used
- **Min:** Enter minimum number of digits (e.g., **1**)
- **Max:** Enter maximum number of digits (e.g., **36**)
- **Delete Digits:** Enter **1** to strip off +
- **Insert Digits:** Enter digits that need to be inserted
- **Address to modify:** Select **both**

Click **Commit** to save .



6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the 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 fields will need to be populated for each SIP Entity.

Under **General:**

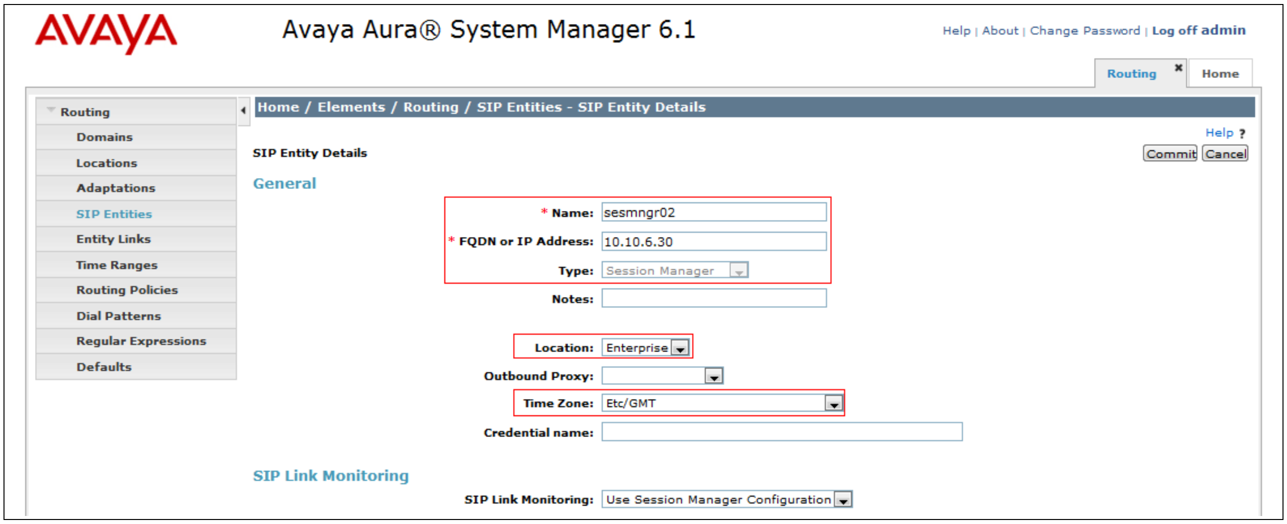
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **SIP TRUNK** for CS1000E SIP entity and **Gateway** for the AASBC SIP entity.
- In the adaptation field select the created adapatation in **Section 6.4** for the CS1000E and AASBC SIP entities.
- 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 1000E SIP Entity
- Session Border Controller SIP Entity

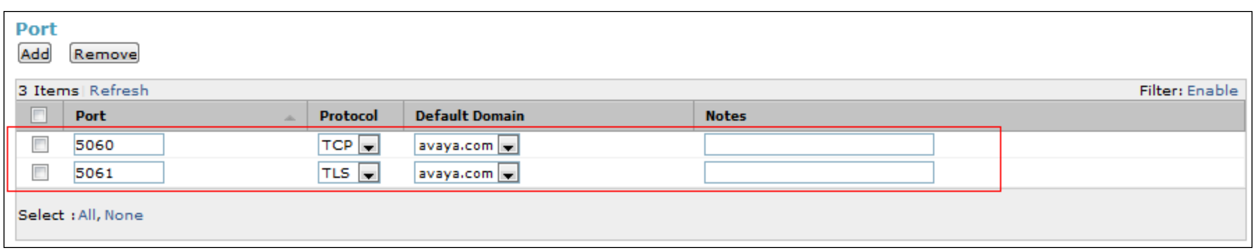
6.5.1. Avaya Aura® Session Manager SIP Entity

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 signaling interface.



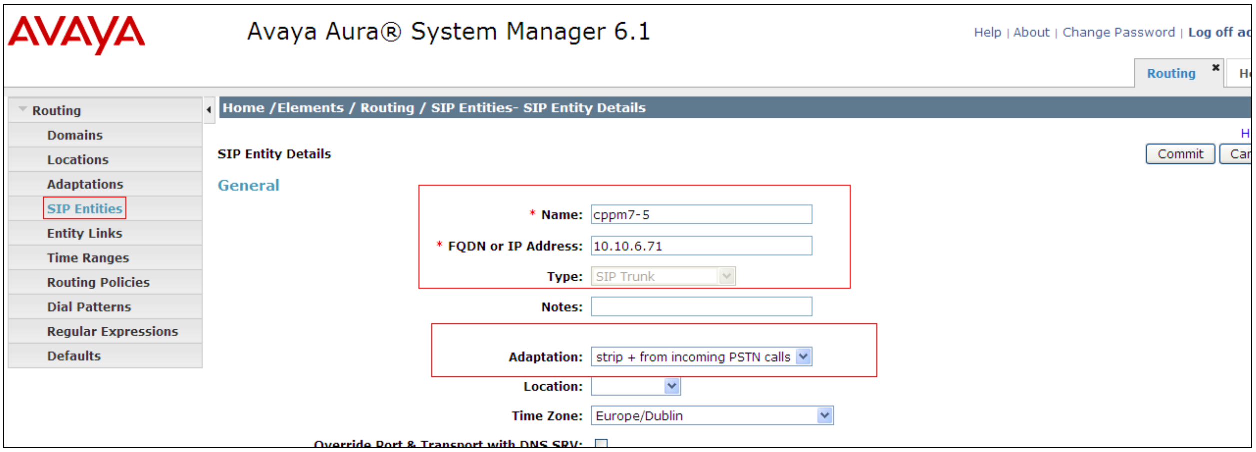
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

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



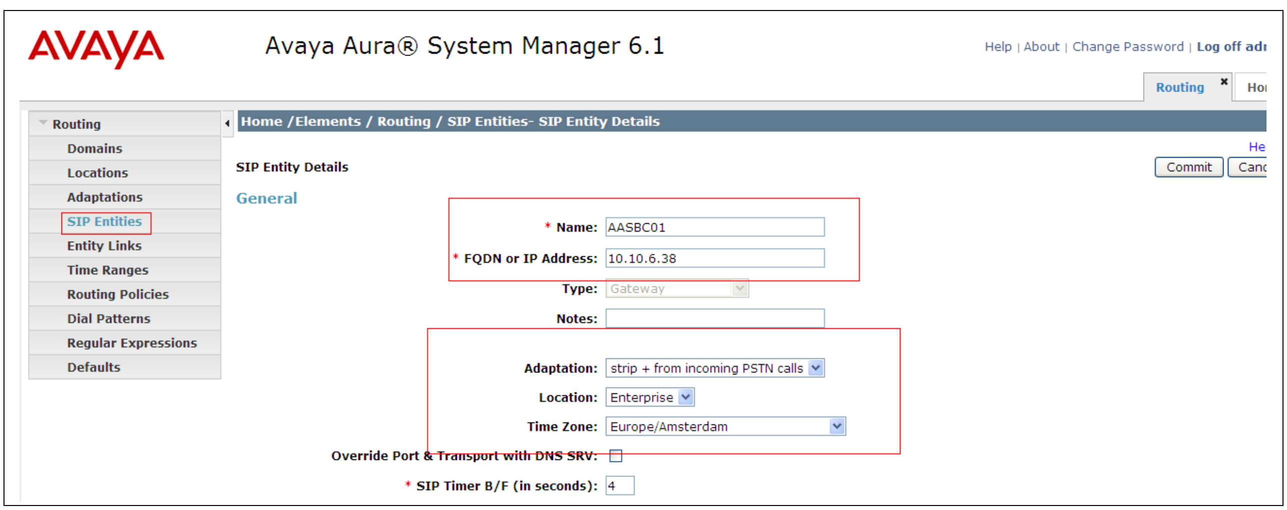
6.5.2. Avaya Communication Server 1000E SIP Entity

The following screens show the SIP entity for Communication Server 1000E. The **FQDN or IP Address** field is set to the IP address of the Node IP configured in **Section 5.4**. Note the adapation created in **Section 6.4** is applied to this entity link.



6.5.3. Avaya Aura® Session Border Controller SIP Entity

The following screen shows the SIP Entity for the AASBC. The **FQDN or IP Address** field is set to the IP address of the AASBC private network interface. Note the adapation created in **Section 6.4** is applied to this entity link.

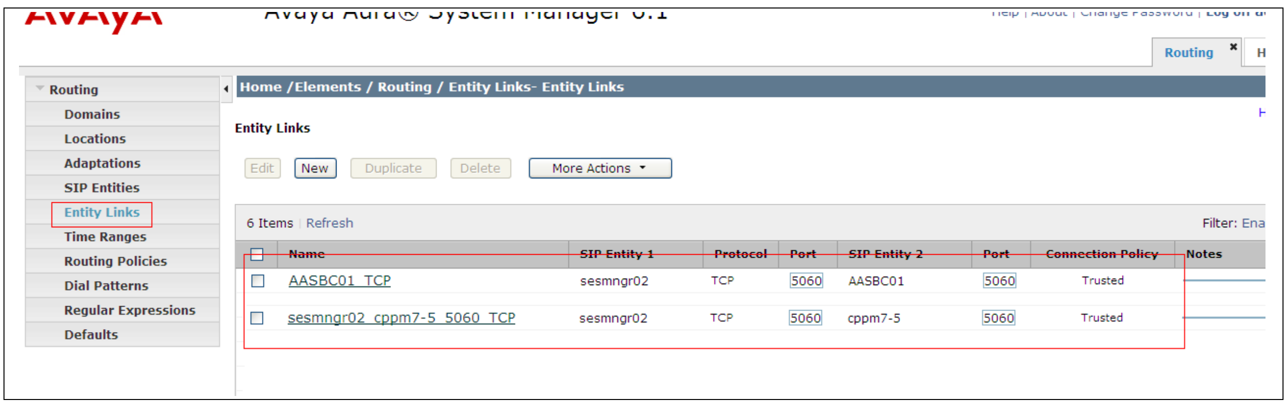


6.6. Administer Entity Links

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 and in the resulting screen 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 the SIP Entity for SessionManager i.e. **sesmngr02**.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.

Click **Commit** to save changes (not shown). The following screen shows the Entity Links used in this configuration.

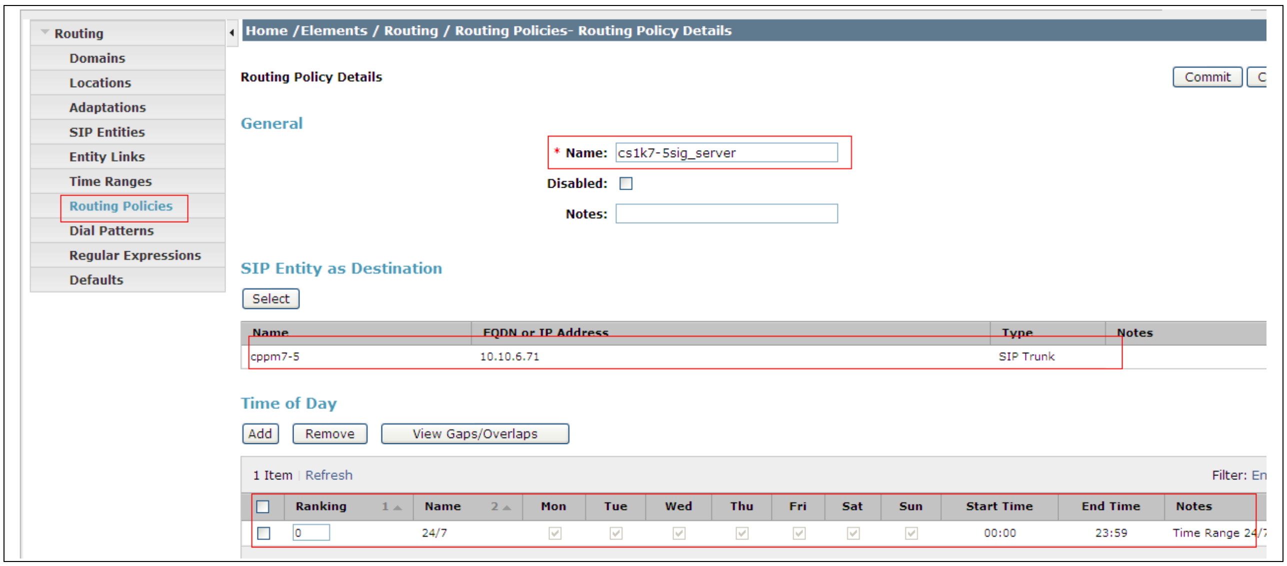


6.7. Administer Routing Policies

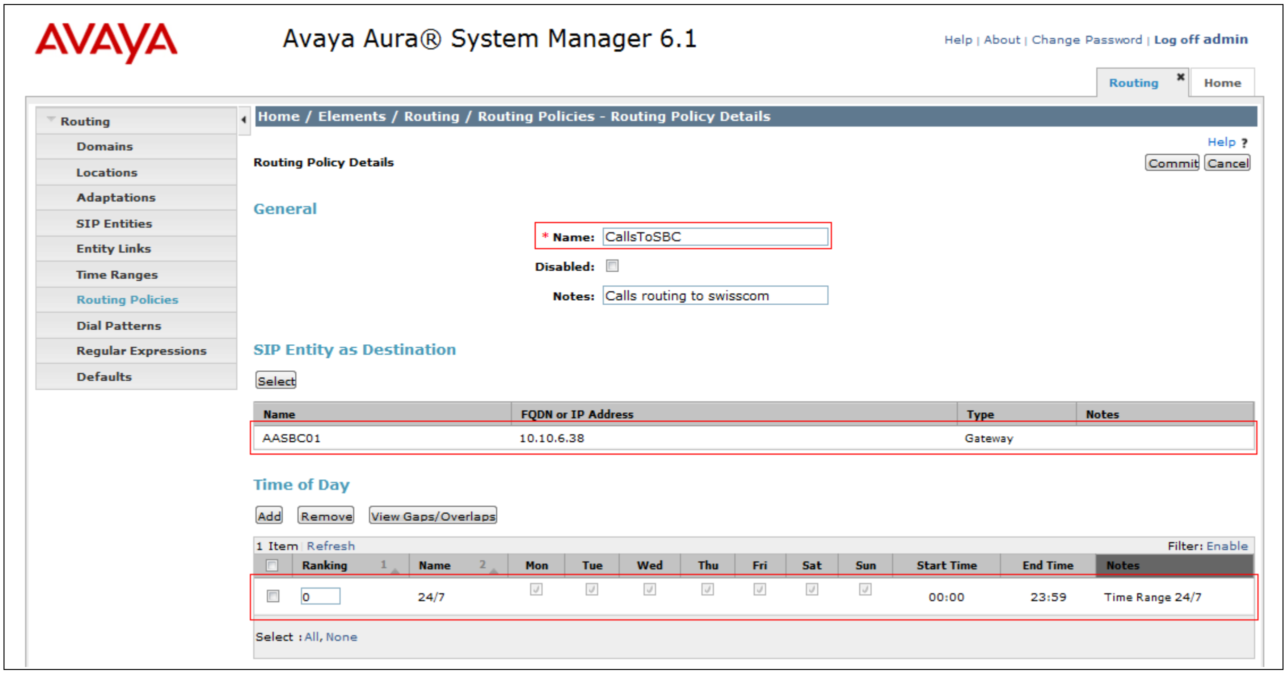
Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

- 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**, select the default **24/7** time range.

The following screen shows the routing policy for Communication Server 1000E



The following screen shows the routing policy for the AASBC



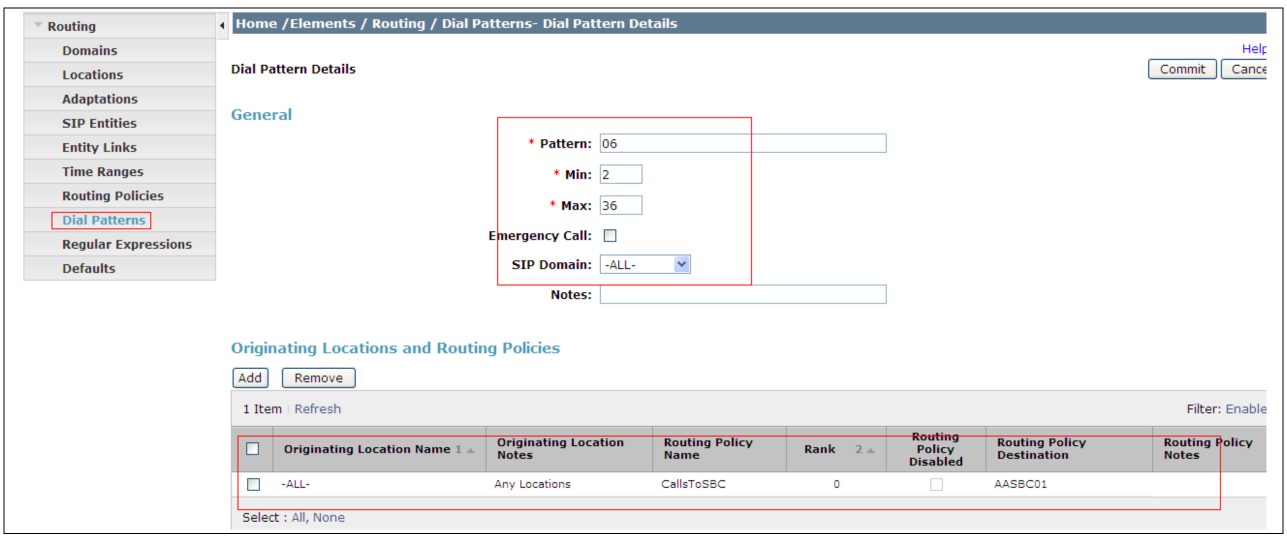
6.8. Administer Dial Patterns

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).

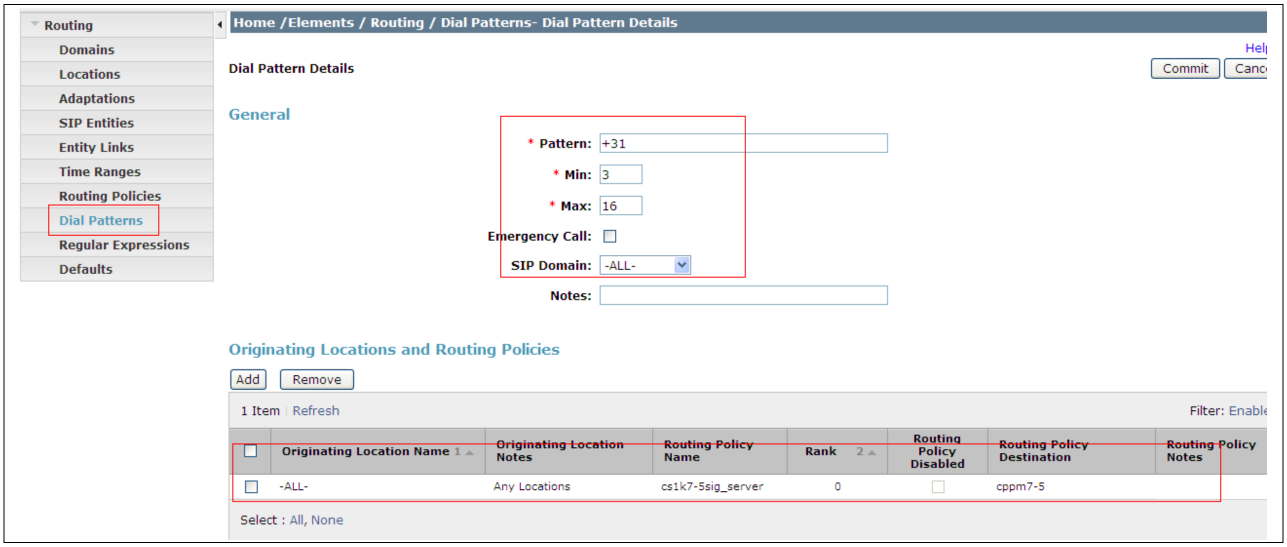
Under **General:**

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialed number
- In the **Max** field enter the maximum length of the dialed number
- In the **SIP Domain** field select **–ALL-** to allow calls from any domain to match the dial pattern

Under **Originating Locations and Routing Policies.** Click **Add**, in the resulting screen (not shown), under **Originating Location** select **ALL** and under **Routing Policies** select the appropriate routing policy defined in **Section 6.7.** Click the **Commit** button to save. The following screen shows an example dial pattern configured for AASBC which will route the calls out to the Vodafone Netherlands SIP Trunk Solution.



The following screen shows an example dial pattern configured for Communication Server 1000E.



7. Configure Avaya Aura® Session Border Controller

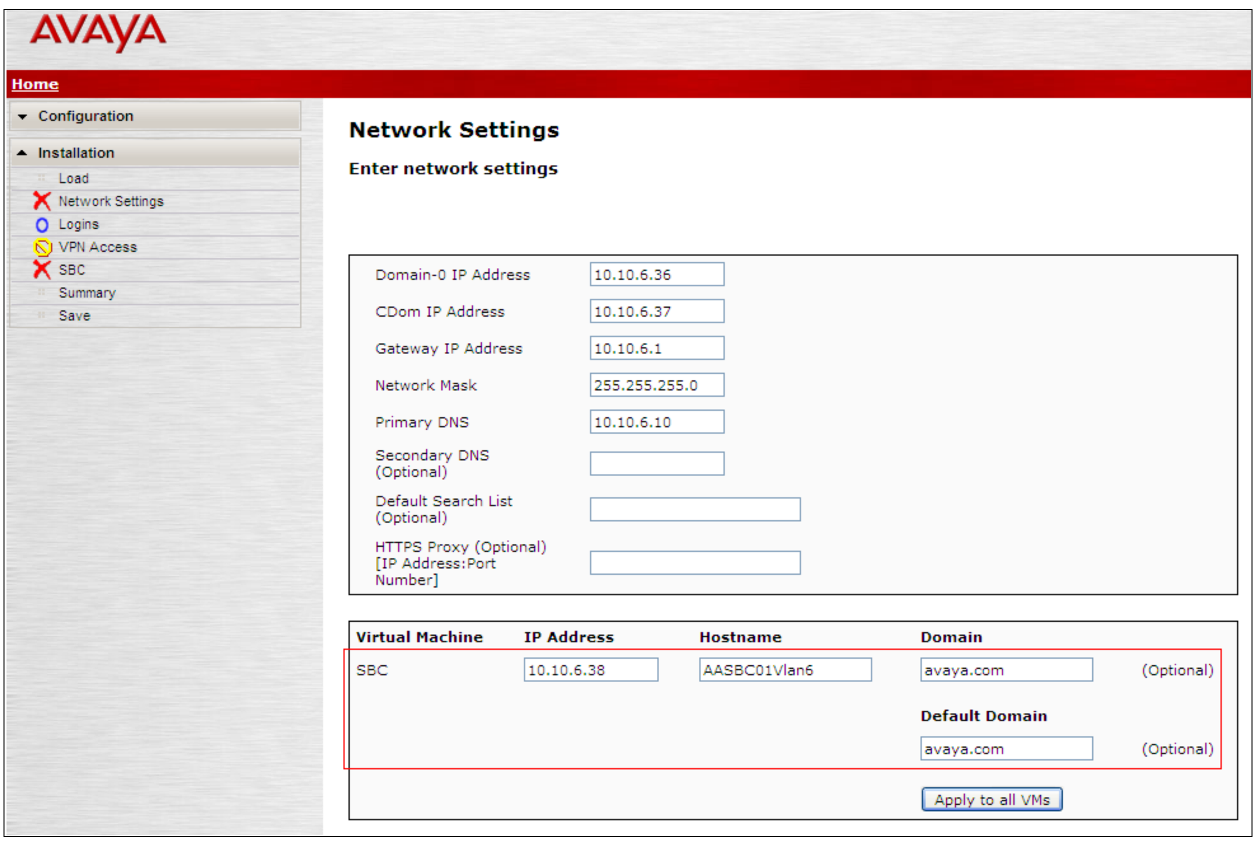
This section describes the configuration of the AASBC. This configuration is done in two parts. The first part is done during the AASBC installation via the installation wizard. These Application Notes will not cover the AASBC installation in its entirety but will include the use of the installation wizard. For information on installing the System Platform and the loading of the AASBC template see **[1]** & **[2]**.The second part of the configuration is done after the installation is complete using the AASBC web interface.

7.1. Installation Wizard

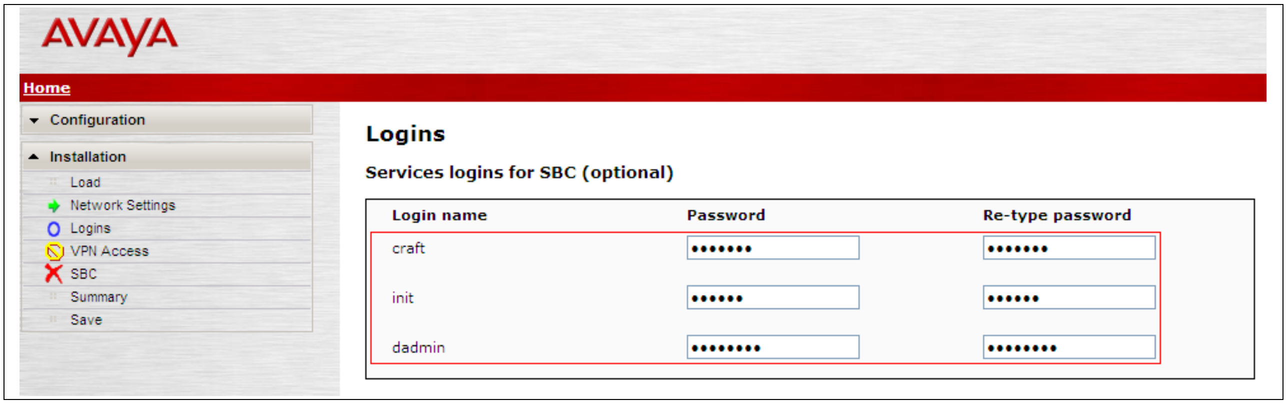
During the installation of the AASBC template, the installation wizard will prompt the installer for information that will be used to create the initial configuration of the AASBC. The first screen of the installation wizard is the Network Settings screen. Fill in the fields as described below and shown in the following screen:

- In the **IP Address** field enter the IP address of the private side of the AASBC
- In the **Hostname** field enter a host name for the AASBC
- Specify a domain in the **Domain** and **Default Domain** fields

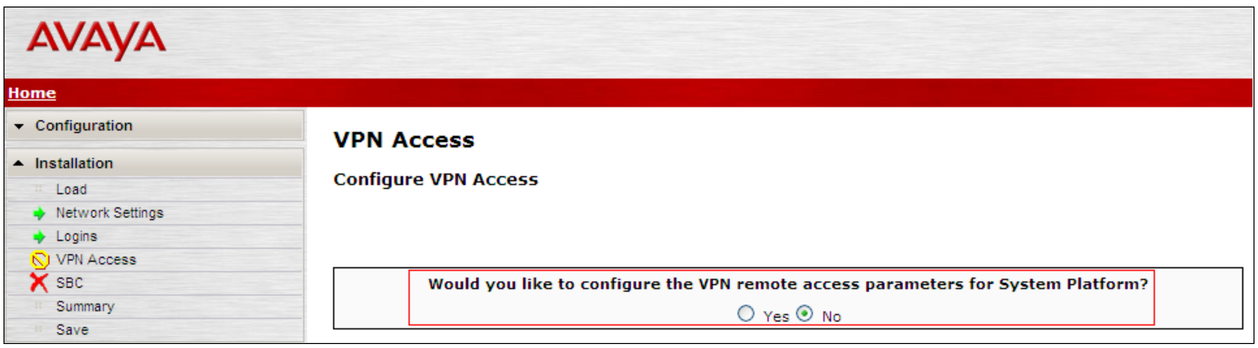
Click **Next Step** (not shown) to continue



From the **Logins** screen specify passwords for the services logins to the AASBC.

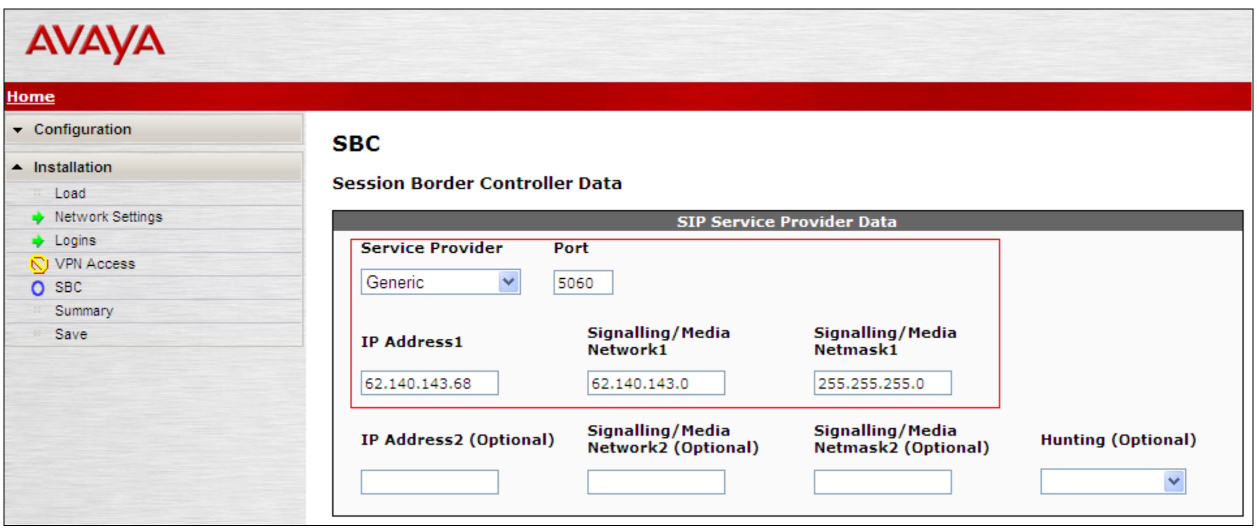


VPN remote access to the AASBC was not part of the compliance test. Thus, on the VPN Access screen, select **No** to the question, **Would you like to configure the VPN remote access parameters for System Platform?**



On the **SBC** screen, in the **SIP Service Provider Data** section fill in the fields as described below and shown in the following screen:

- In the **Service Provider** select the name of the Service Provider to which the AASBC will connect. This will allow the wizard to select a configuration file customized for this Service Provider. At the time of the compliance test, a customized configuration file did not exist for Vodafone Netherlands. Thus, **Generic** was chosen
- In the **Port** field enter the port number that Vodafone Netherlands uses to listen for SIP traffic
- In the **IP Address1** field enter the IP addresses provided by Vodafone Netherlands for the Vodafone Office Voice SIP Trunk Service (fixed). The IP address for the Vodafone OneVoice Corperate SIP Trunk Service (mobile) used during testing will be added after the AASBC template is installed **(Section 7.3)**
- In the **Signaling/Media Network1** field enter the Vodafone Netherlands provided subnet where media traffic will originate. An additional subnet can be provided for **Signaling/Media Network2**
- In the **Media Netmask** field enter the netmask corresponding to the Media Network
- Scroll down to continue



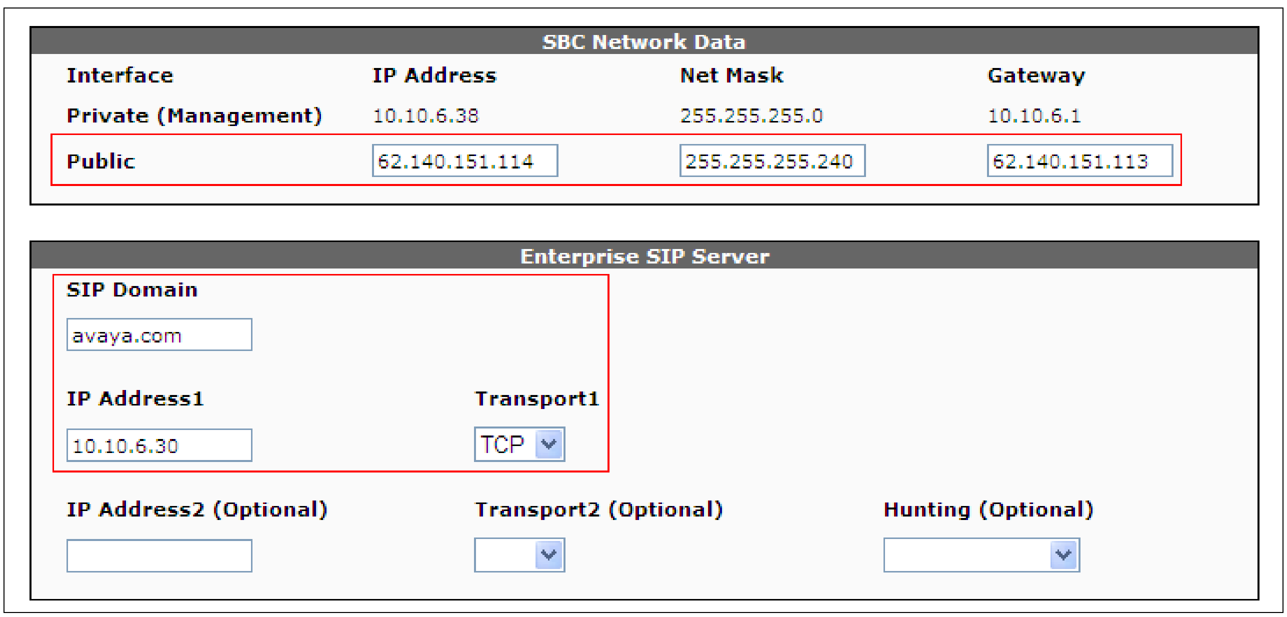
Further down on the same **SBC** screen, in the **SBC Network Data** section fill in the fields as described below:

- In the **Public IP Address** field enter the enterprsieIP address that will be used for the Vodafone Netherlands Office Voice SIP Trunk Service on the public side of the AASBC
- In the **Public Net Mask** field enter the netmask associated with the public network to which the AASBC connects
- In the **Public Gateway** field enter the default gateway of the public network

In the **Enterprise SIP Server** section fills in the fields as described below:

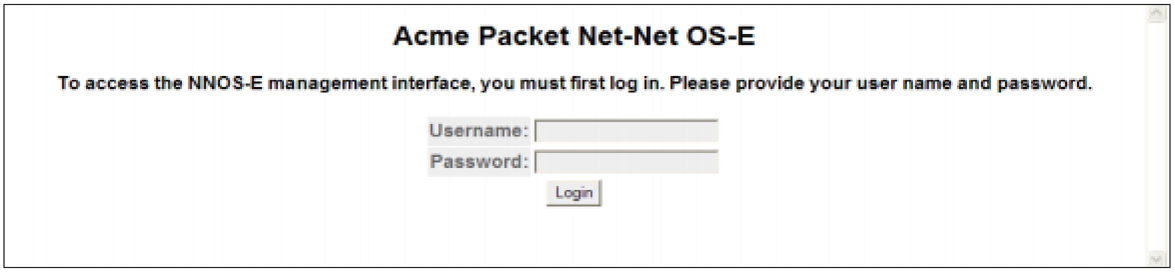
- In the **SIP Domain** field enter the enterprise SIP domain
- In the **IP Address** field enter the IP address of the Enterprise SIP Server to which the AASBC will connect. In the case of the compliance test, this is the IP address of the Session Manager SIP signaling interface
- In the **Transport1** field select the transport protocol to be used for SIP traffic between the AASBC and Session Manager

Click **Next Step** to continue. A summary screen will be displayed (not shown). Check the displayed values and click **Next Step** again to install the template with the values entered.



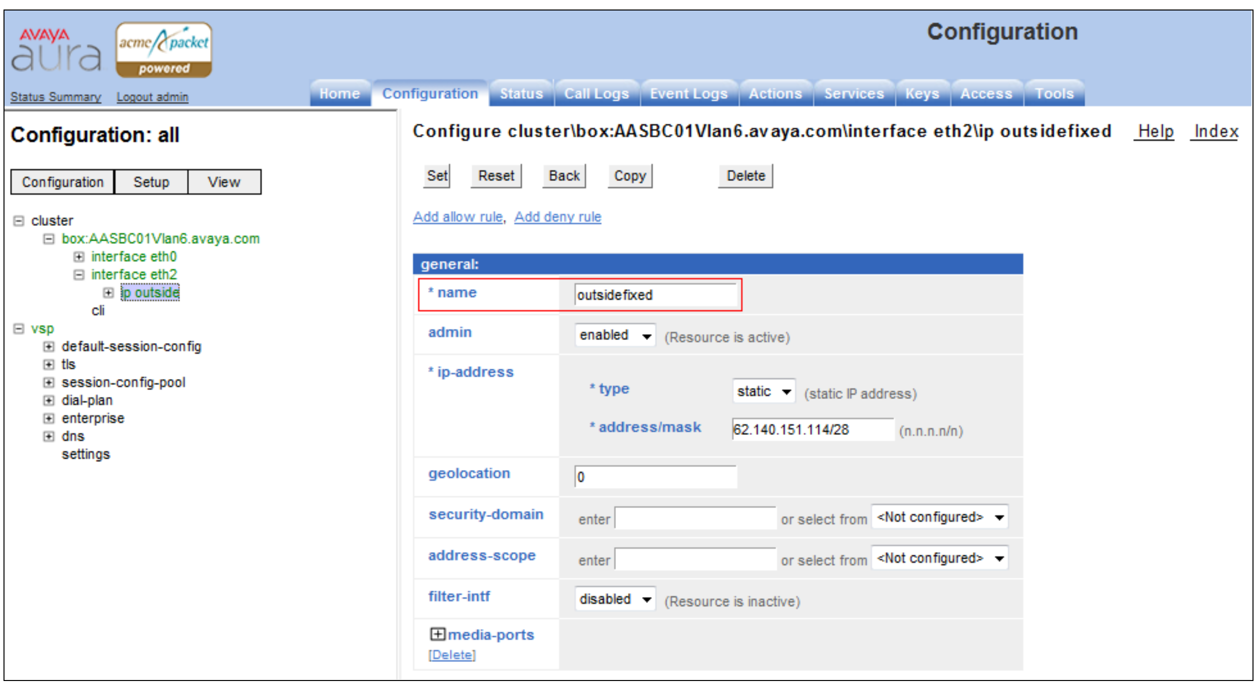
7.2. Access Avaya Aura® Session Border Controller

Access the AASBC using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured in **Section 7.1**. Log in with the appropriate credentials.

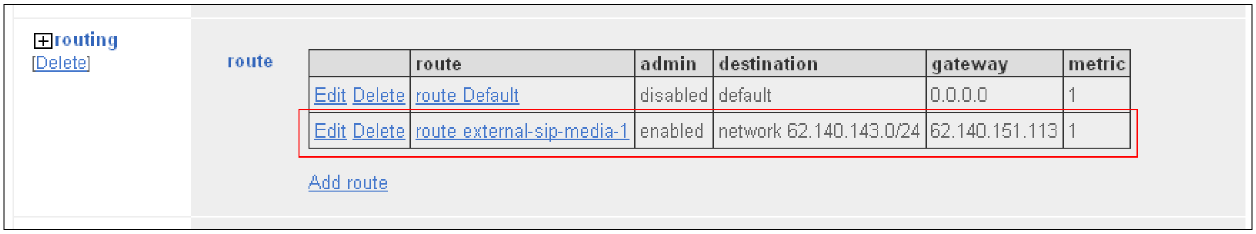


7.3. Configure Outside Interfaces

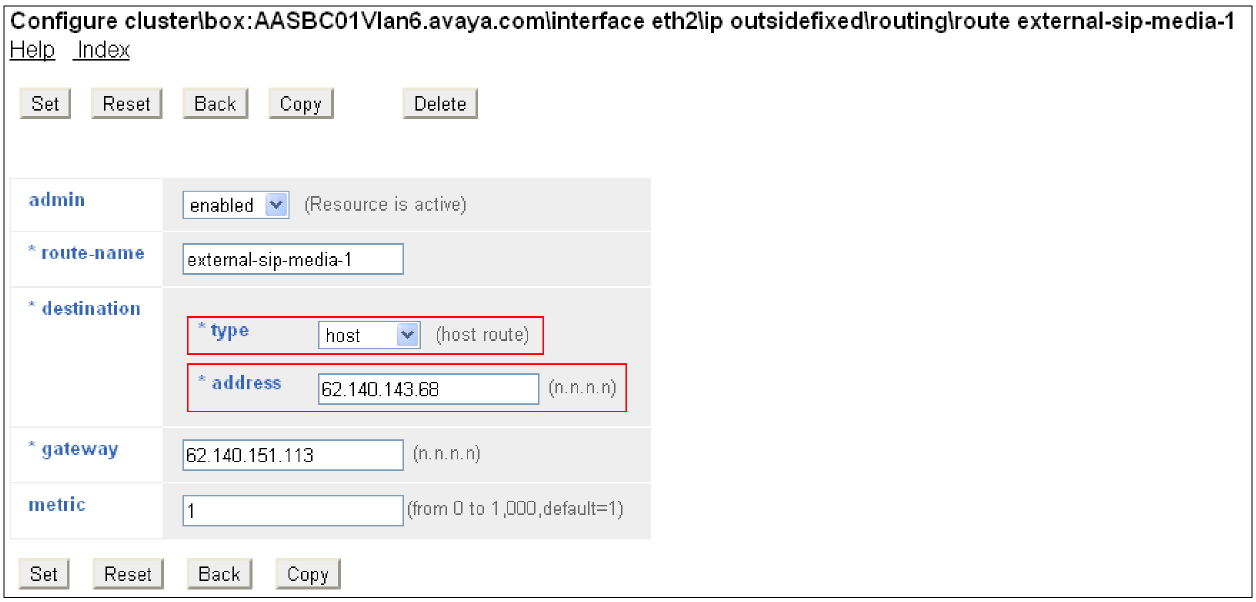
To allow two logical connections to be created between the enterprise and Vodafone Netherlands an additional IP address is created on the outside interface of the AASBC. Rename the IP address configuration created in **Section 7.1** by expanding **cluster → box:AASBC01Vlan6.avaya.com → interface eth2 → ip outside** and enter a descriptive name in the **name** field. The name **outsidefixed** is used as this is the IP address that will be used for the Vodafone Office Voice SIP Trunk Service. Scroll down to continue.



Further down on the same screen in the **routing** section click the edit link relating to the **route external-sip-media-1** route.



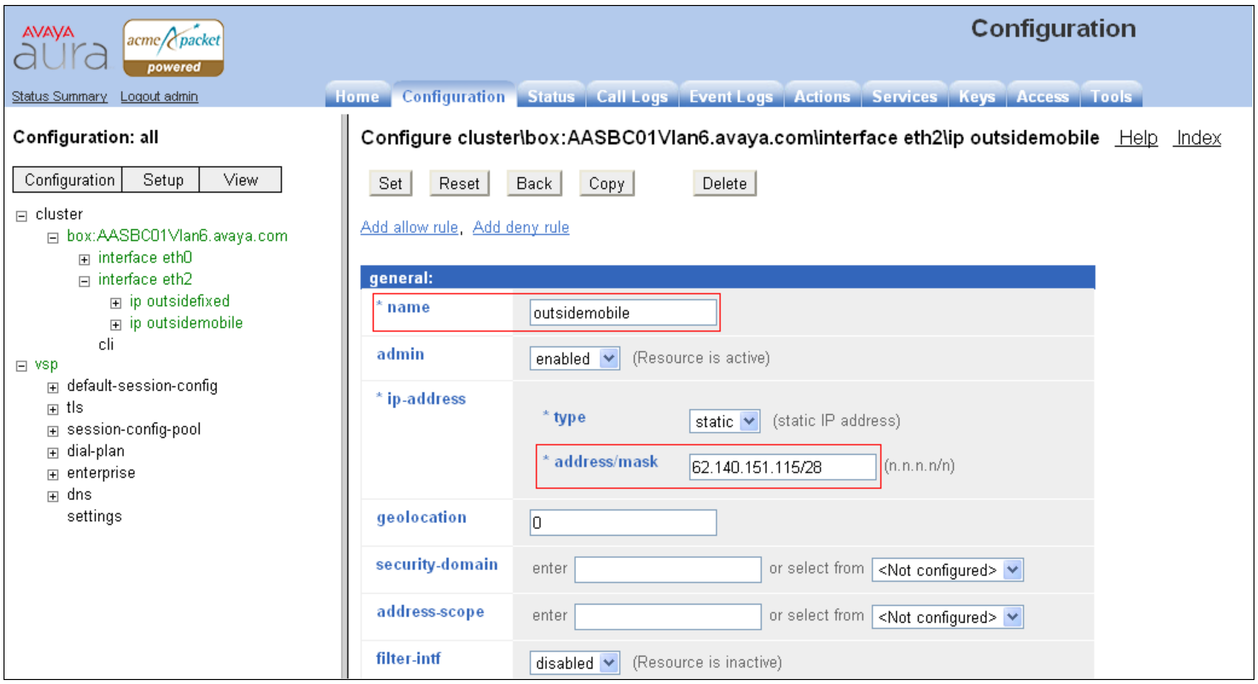
In the resulting screen in the **destination** section, select **host** from the **type** drop down menu. In the **address** field enter the IP address of the Vodafone Netherlands Office Voice SIP trunk service.



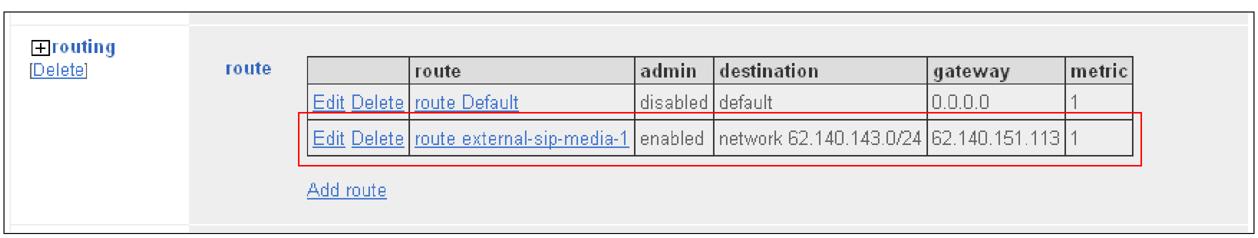
To create another IP address configuration navigate to **box:AASBC01Vlan6.avaya.com → interface eth2 → ip outsidefixed** and click **copy** (Not shown). In the resulting screen update the fields as shown below:

- In the **name** field enter a descriptive name. The name **outsidemobile** is used as this is the IP address that will be used for the Vodafone OneVoice Corporate SIP Trunk Service. Scroll down to continue.
- In the **address/mask** field enter the IP address that will be used on the public side of the AASBC for the Vodafone OneVoice Corporate SIP Trunk Service.

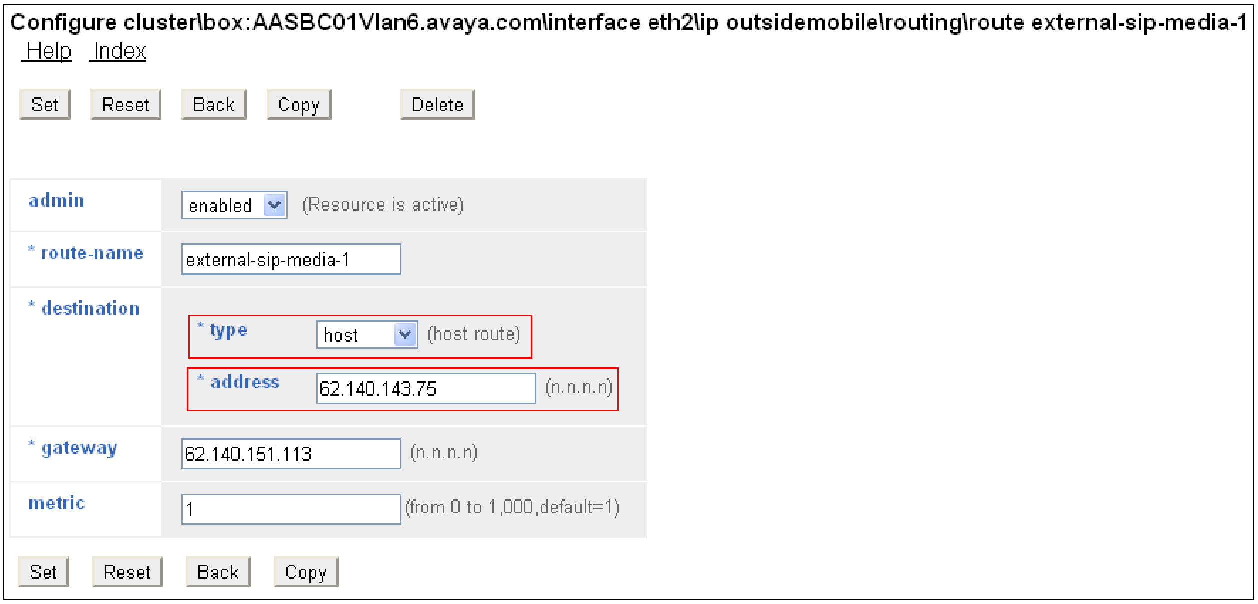
Scroll down to continue.



Further down on the same screen in the **routing** section click the edit link relating to the **route external-sip-media-1** route.

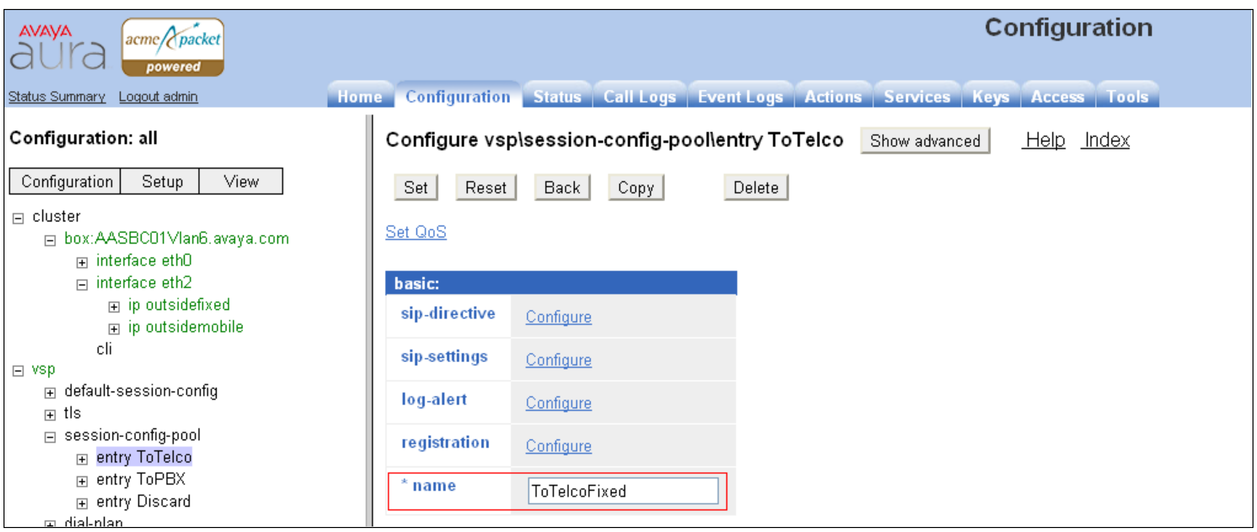


In the resulting screen in the **destination** section, select **host** from the drop down menu for **type**. In the **address** field enter the IP address of the Vodafone Netherlands OneVoice Corperate SIP trunk service.



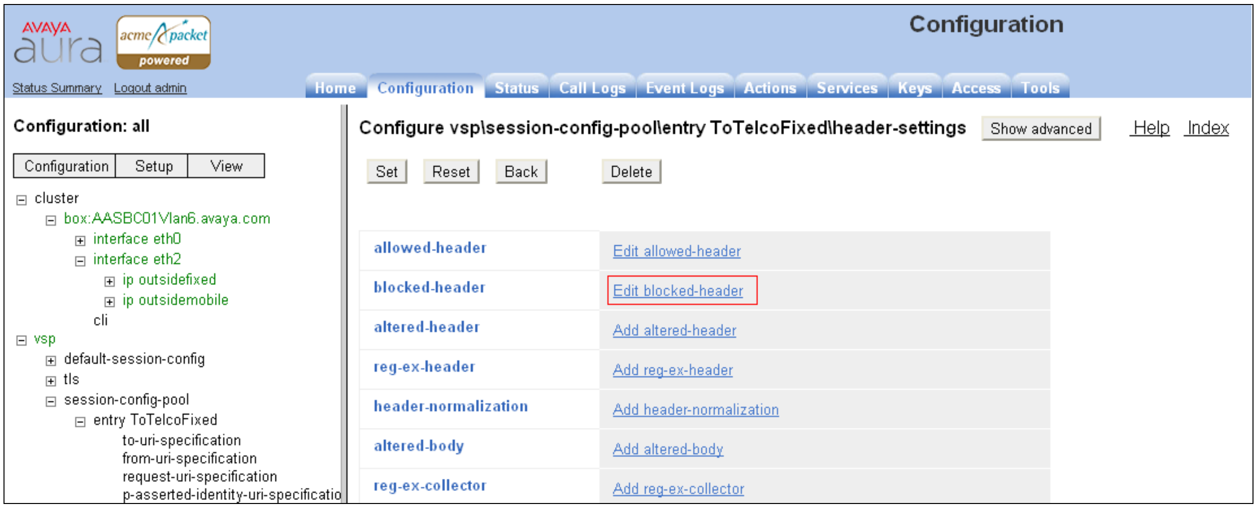
7.4. Session Config Pool

Navigate to **vsp → session-config-pool → entry ToTelco** and extend the entry in the **name** field to **ToTelcofixed**.

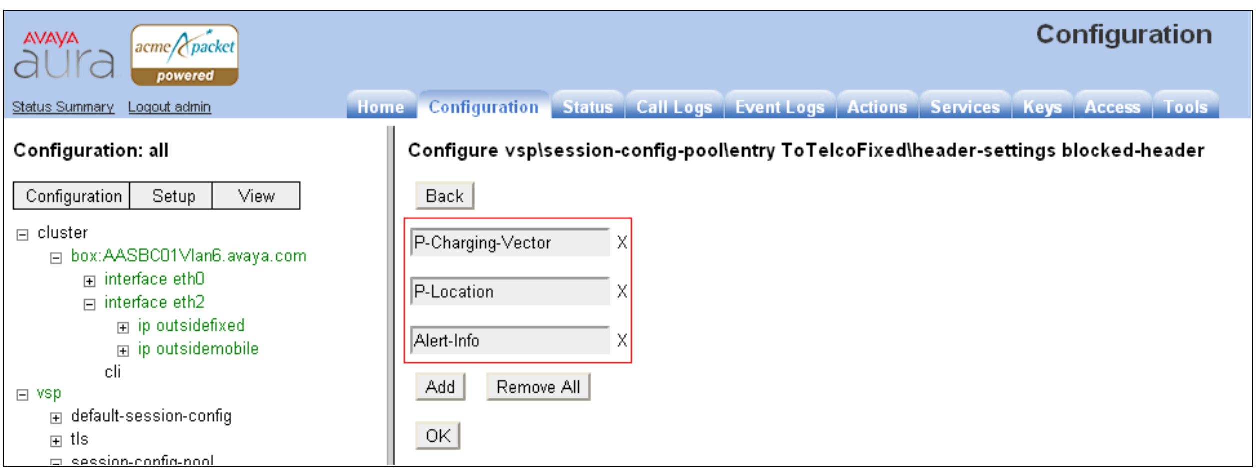


7.4.1. Stripping SIP Headers

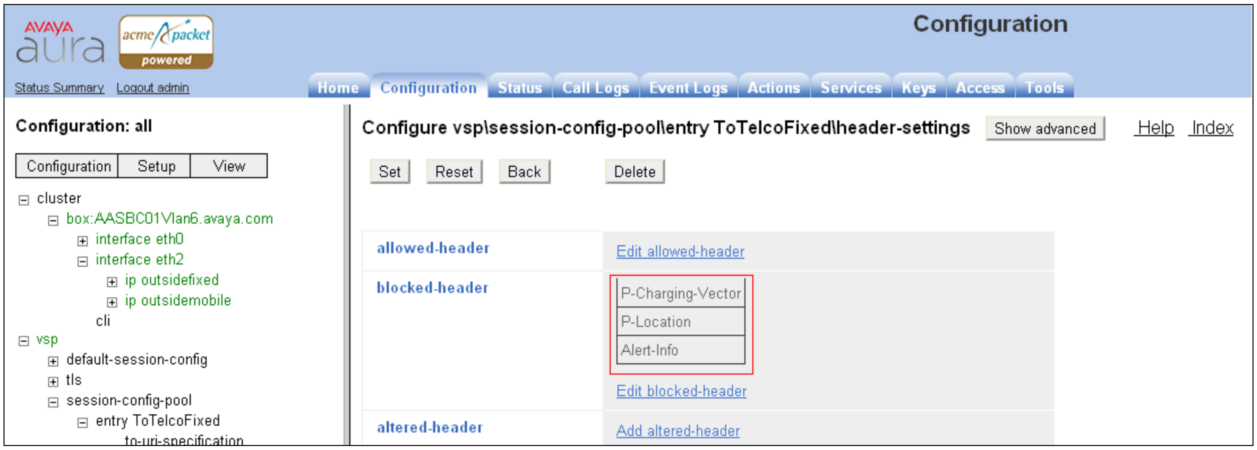
The AASBC can be used to strip SIP headers to prevent the header from being sent to the public SIP Service Provider. To strip a SIP header navigate to **vsp → session-config-pool → entry ToTelco → header-settings** and click on the **Edit blocked-header** link.



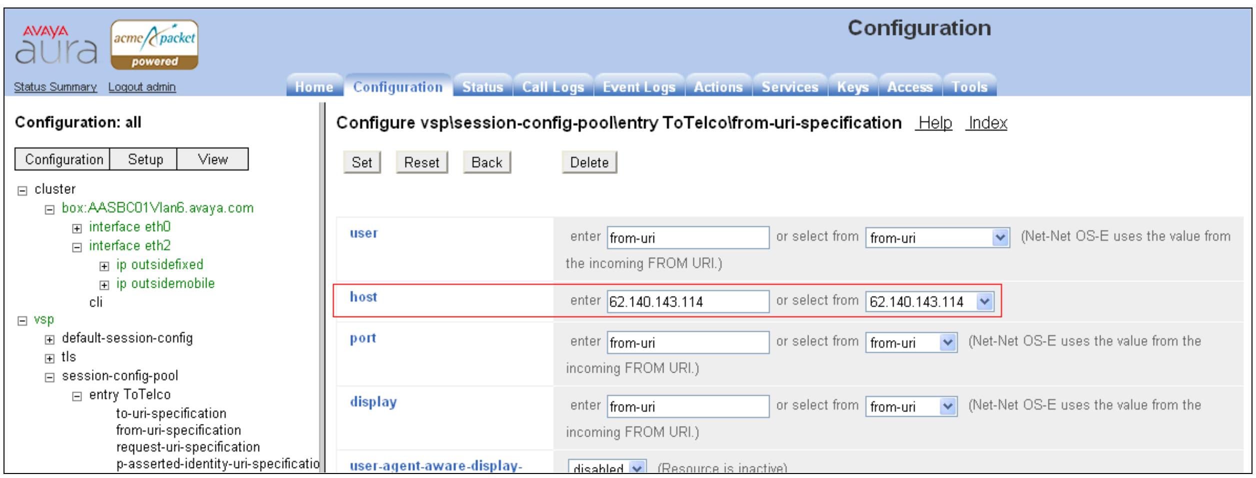
In the resulting page click the **Add** button to open a new entry field and enter the name of the header to be removed, repeat this action for all the headers to be removed. Click the **OK** button when finished.



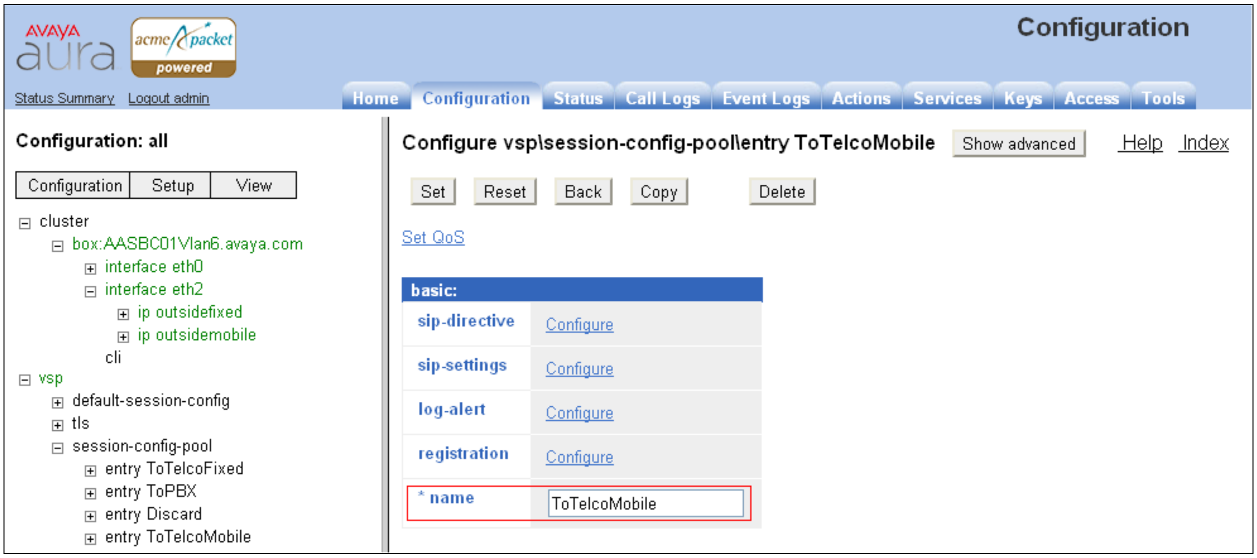
The following screen shows the headers being stripped during testing.



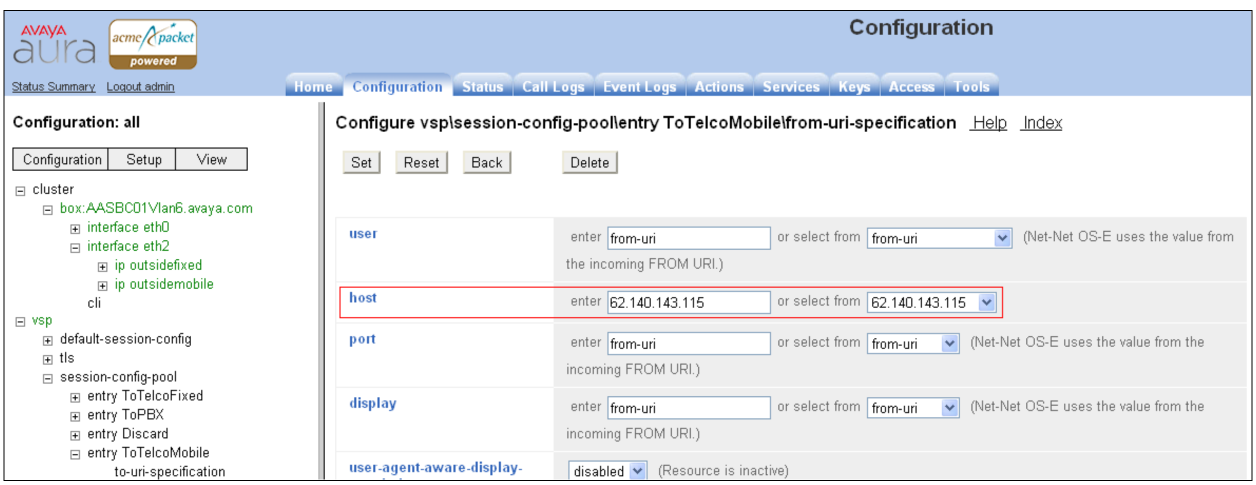
Navigate to **vsp → session-config-pool → entry ToTelco → from-uri-specification** and enter the IP address used on the public side of the AASBC for the Vodafone Netherlands Office Voice SIP trunk into the first host field. This will ensure that the host part of the From header is always set as the entered IP address. Click **Set** to save changes.



Navigate to **vsp → session-config-pool → entry ToTelco** and click **Copy** (not shown). This will produce an exact copy of the session config including the stripped SIP headers. In the resulting screen alter the entry in the **name** field to **ToTelcomobile**.

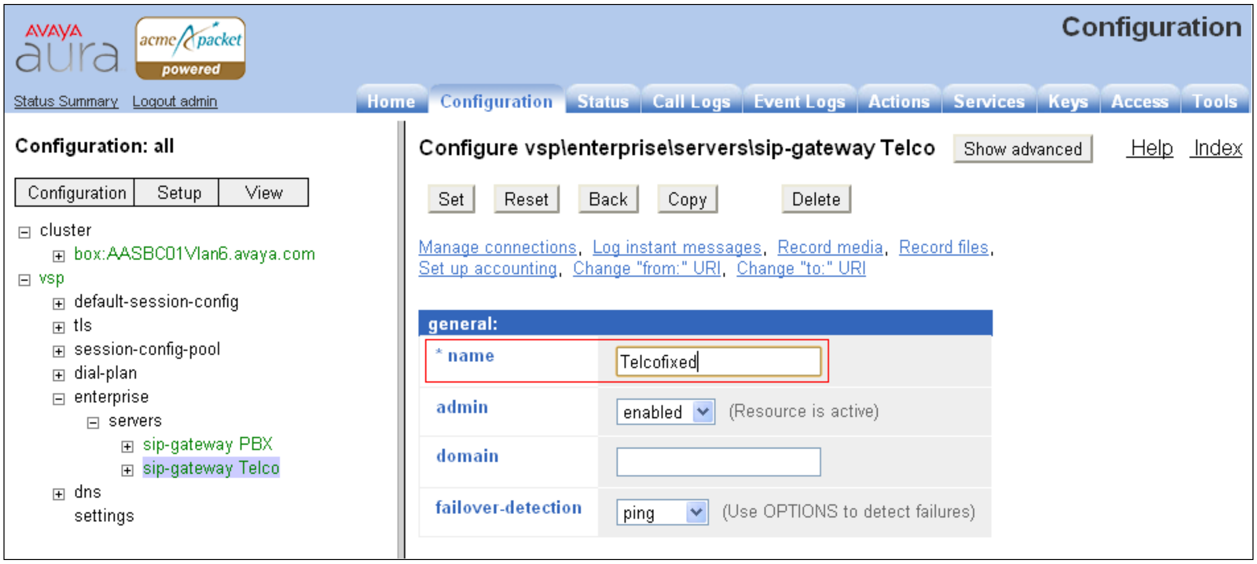


Navigate to **vsp → session-config-pool → entry ToTelcomobile → from-uri-specification** and enter the IP address used on the public side of the AASBC for the Vodafone Netherlands OneVoice Coporate SIP trunk into the first **host** field. This will ensure that the host part of the From header is always set as the entered IP address. Click **Set** to save changes.

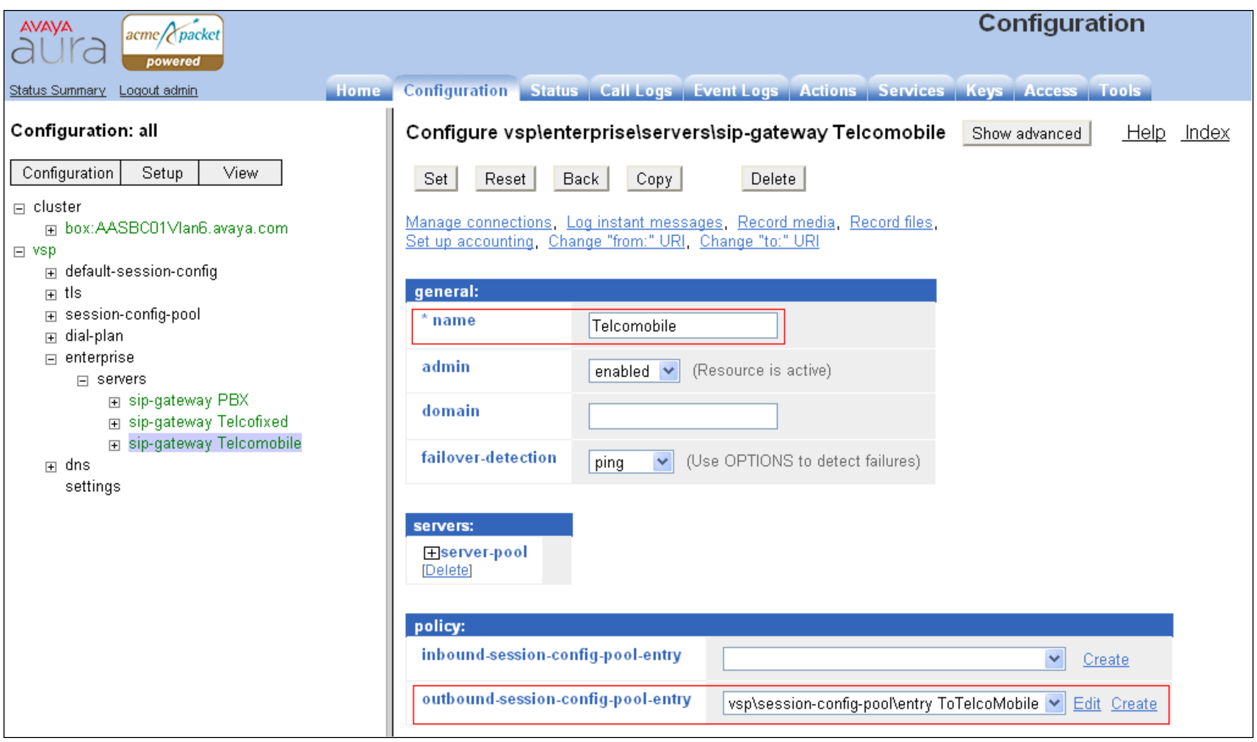


7.5. SIP Servers

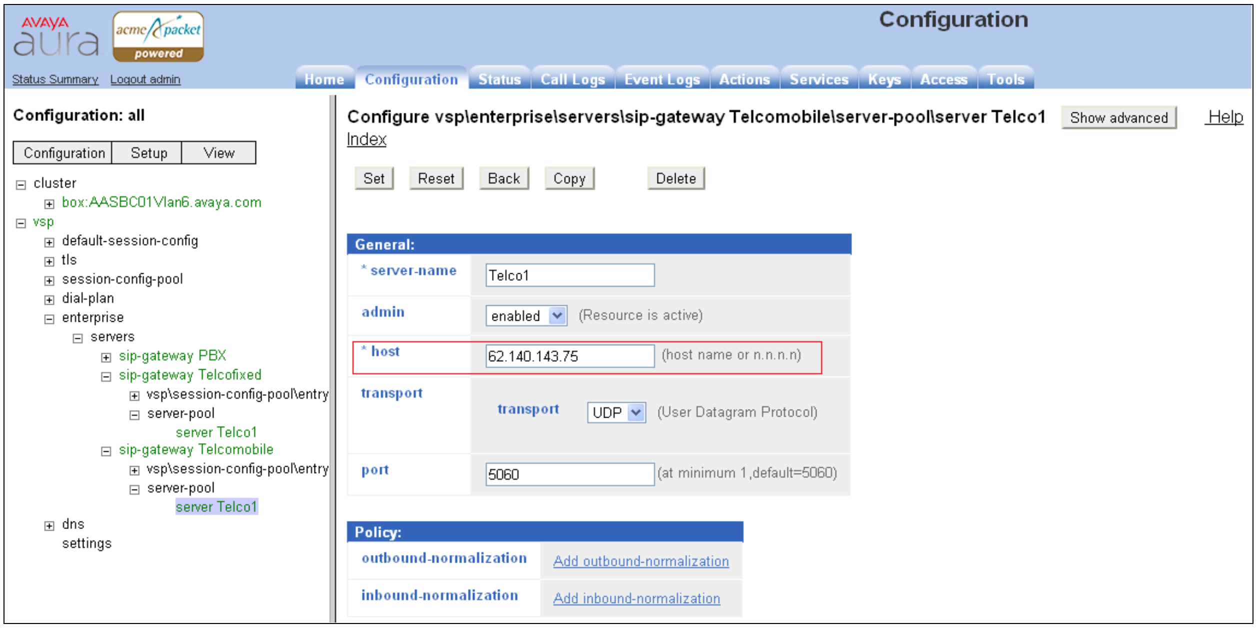
Navigate to **vsp → enterprise → servers → sip-gateway Telco** and alter the entry in the **name** field to **Telcofixed.** Click **Set** to save changes.



Navigate to **vsp → enterprise → servers → sip-gateway Telcofixed** and click **Copy** (Not shown). In the resulting screen alter the entry in the **name** field to **Telcomobile**. In the **outbound-session-config-pool-entry** field select the **ToTelcomobile** session config created in **Section 7.4** from the drop down menu. Click **Set** to save changes.



Navigate to **vsp → enterprise → servers → sip-gateway Telcomobile → server-pool → server Telco1** and enter the IP address provided by Vodafone Netherlands for the Vodafone Netherlands OneVoice Corporate SIP trunk connection in to the **host** field. Click **Set** to save changes.

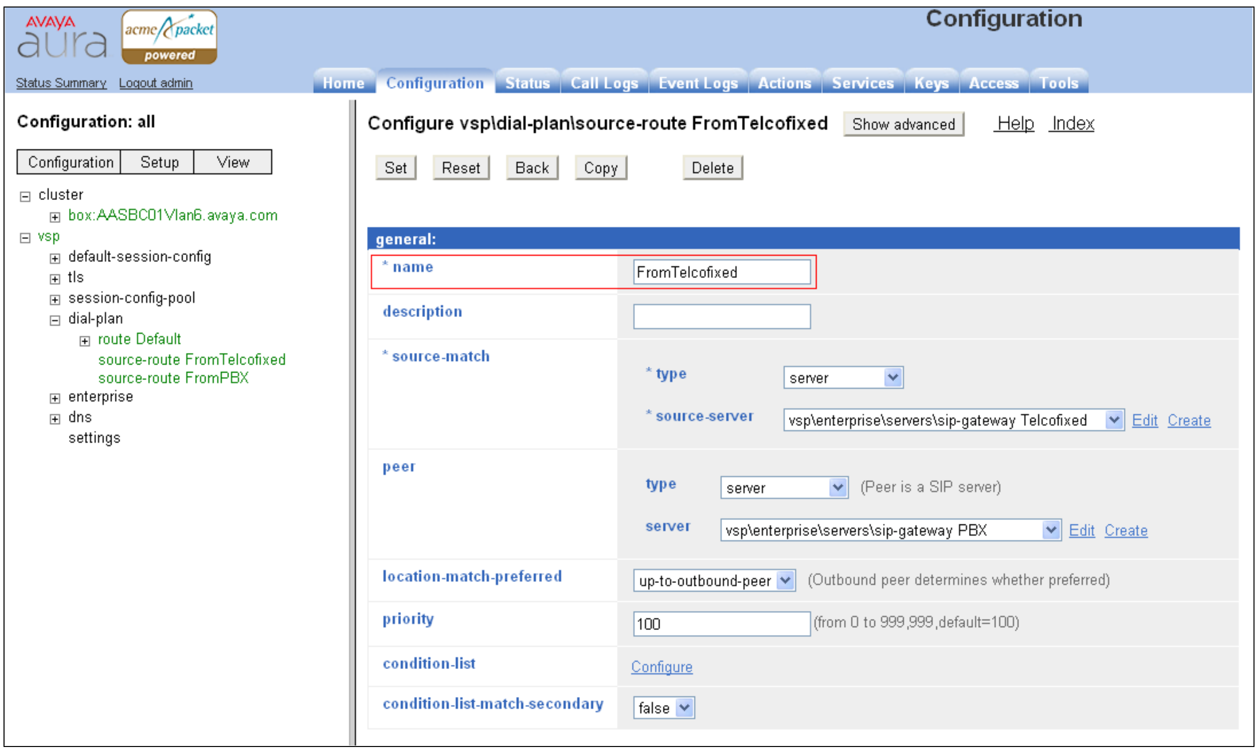


7.6. Dial Plan Configuration

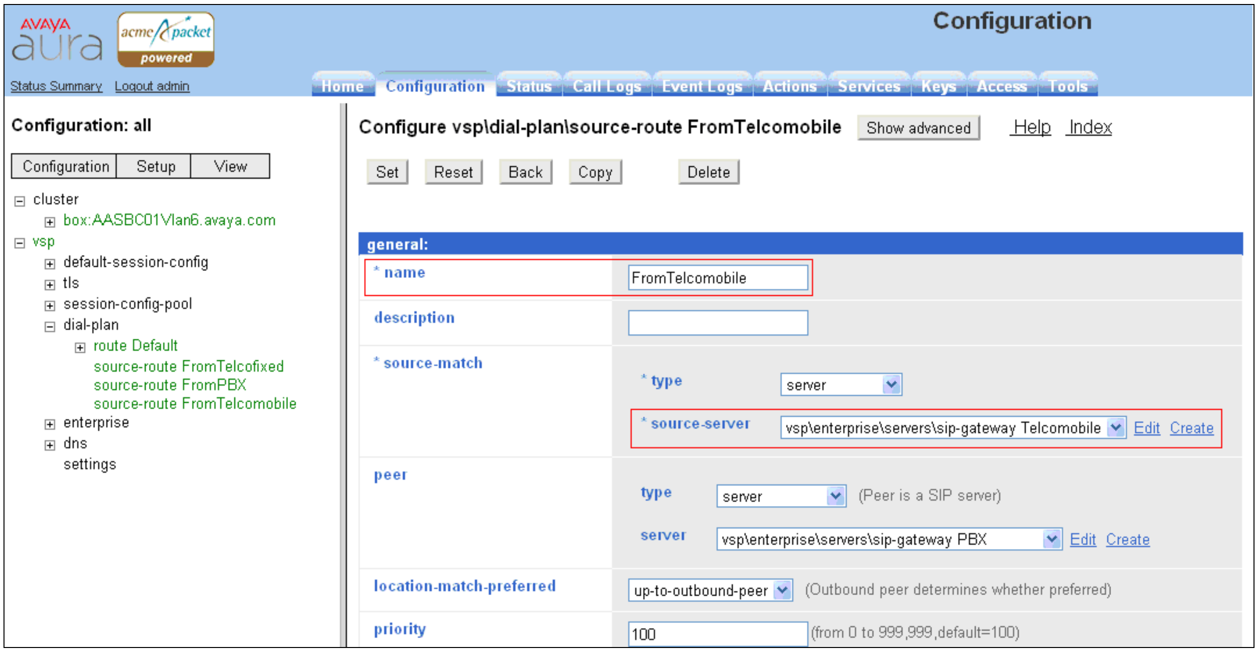
The dial plan is used to define how calls route between SIP entities. For the compliance test four routes are required.

- The route **FromTelcofixed** will be used to route fixed calls from Vodafone Netherlands to the Session Manager.
- The route **FromPBXfixed** will be used to route fixed calls from the Session Manager to Vodafone Netherlands.
- The route **FromTelcomobile** will be used to route mobile calls from Vodafone Netherlands to the Session Manager.
- The route **FromPBXmobile** will be used to route mobile calls from the Session Manager to Vodafone Netherlands.

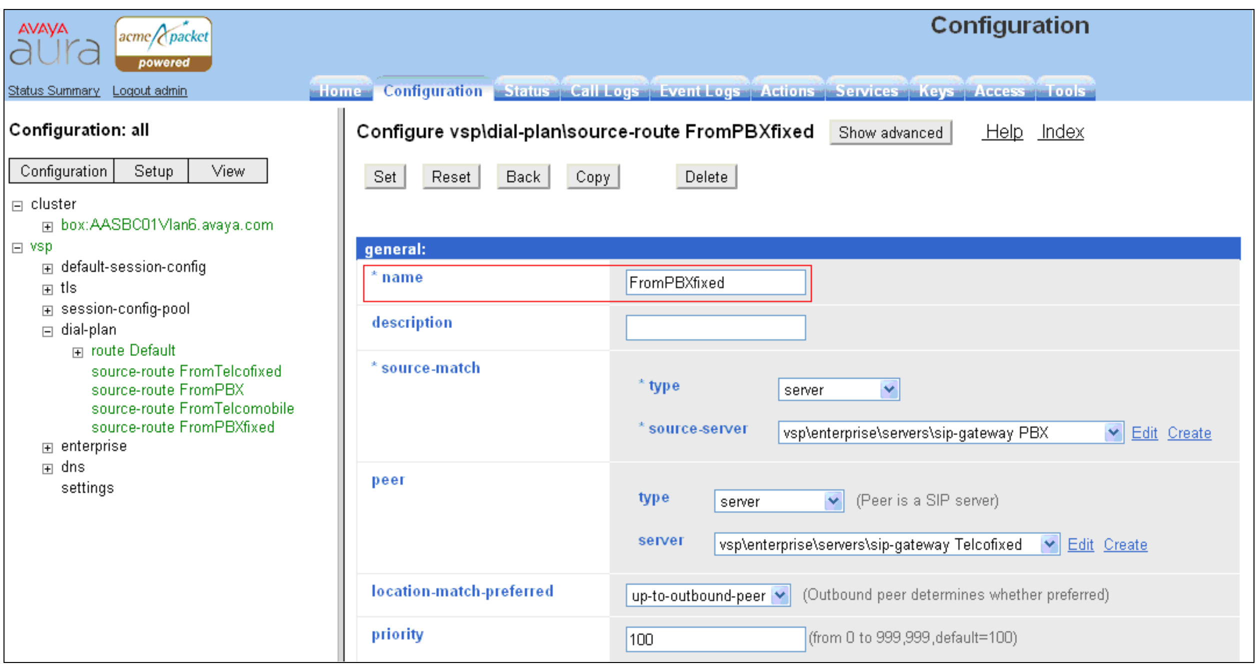
Navigate to **vsp → dial-plan → source-route FromTelco** (not shown) and alter the entry in the **name** field to **FromTelcofixed.** Click **Set** to save changes.



Navigate to **vsp → dial-plan → source-route FromTelcofixed** and click **Copy** (not shown). In the resulting screen alter the entry in the **name** field to **FromTelcomobile.** In the **source-server** field select the **Telcomobile** SIP server created in **Section 7.5.** Click **Set** to save changes.



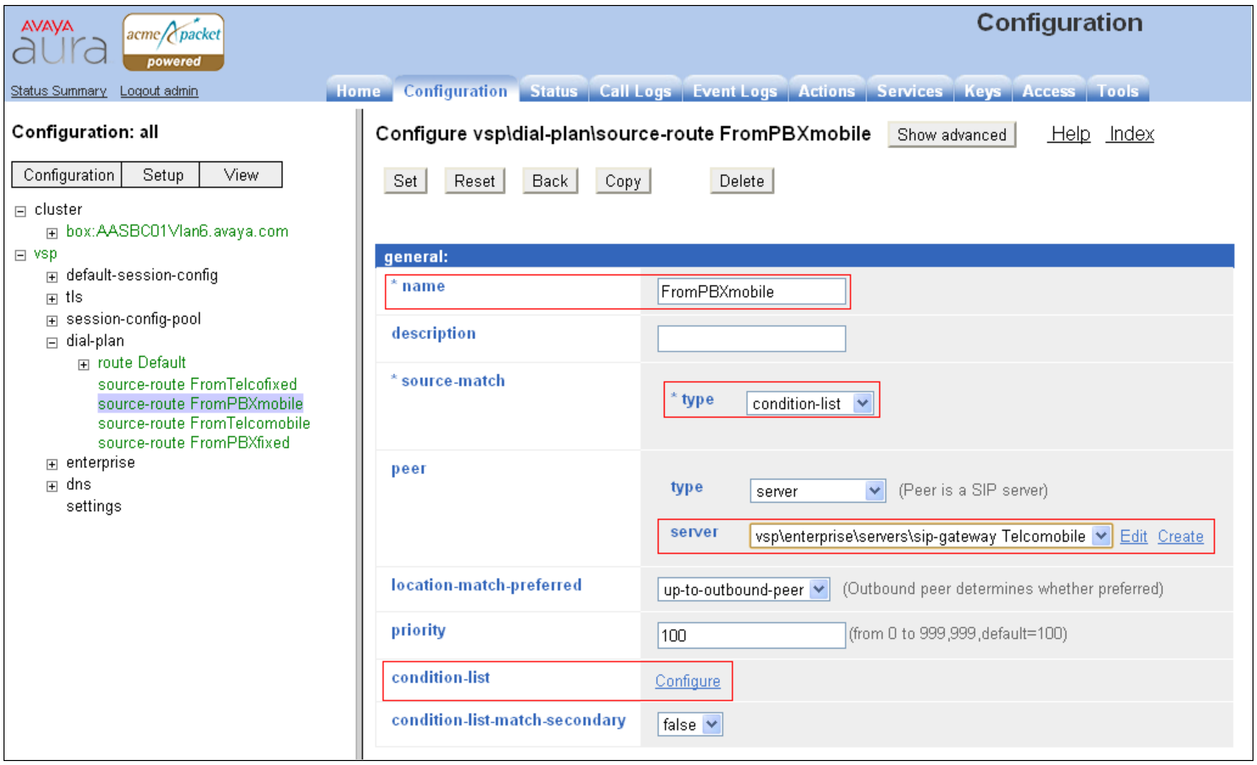
Navigate to **vsp → dial-plan → source-route FromTelcofixed** and alter the entry in the **name** field to **FromPBXfixed.** Click **Set** to save changes.



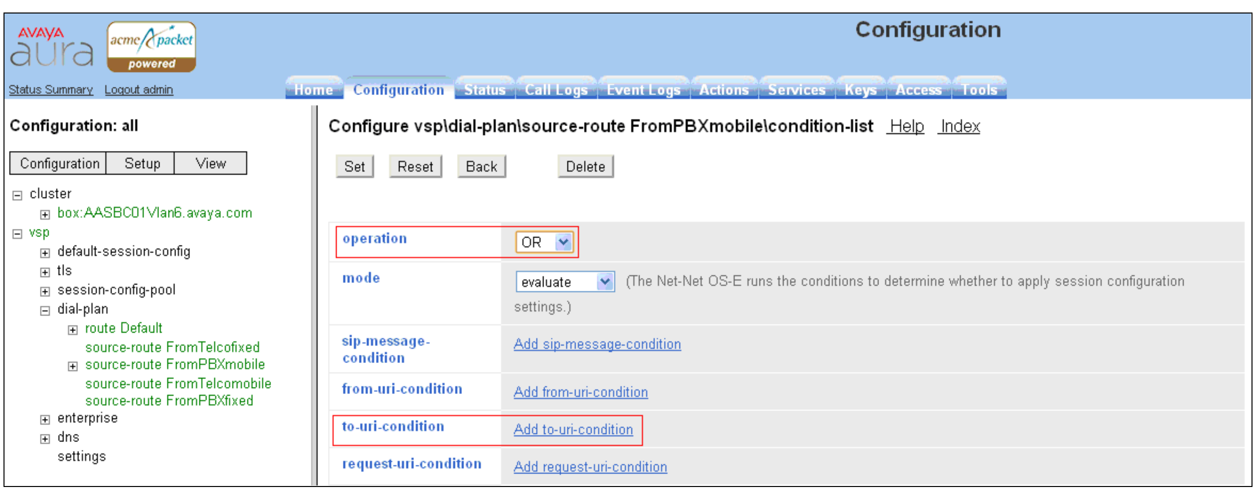
Navigate to **vsp → dial-plan → source-route FromPBXfixed** and click **Copy** (not shown). In the resulting screen update the fields as shown below:

- Alter the entry in the **name** field to **FromPBXmobile.**
- Under the **source-match** section, select **condition-list** from the drop down box in the **type** field.
- Under the peer section, in the **server** field select the **Telcomobile** SIP server created in **Section 7.5**

Click **Set** to save changes and then click the **configure** link under the **condition-list** section.



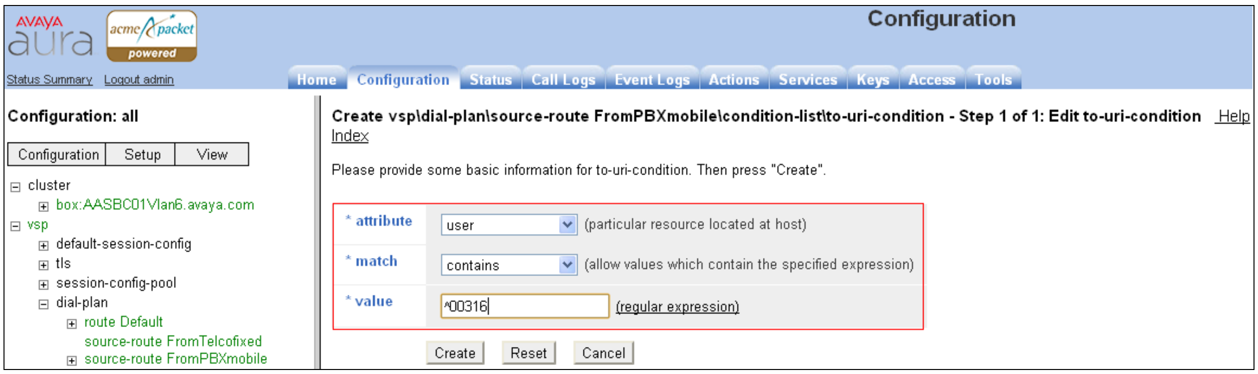
In the resulting screen select the **operation OR** from the drop down menu and click the **Add-to-uri-condition** link under the **to-uri–condition** section.



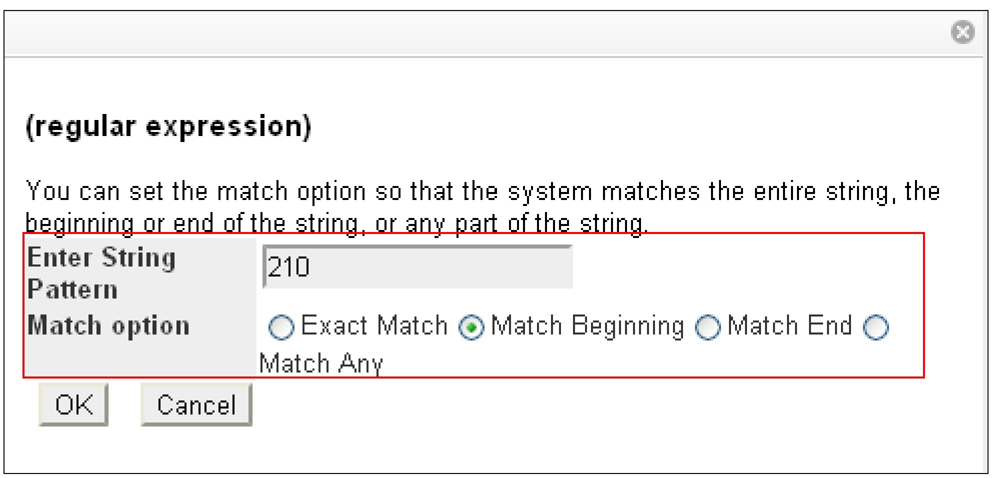
In the resulting screen define the dial patterns that the condition list should match by updating the fields as shown below:

- For **attribute** select **user** from the drop down menu. This means that the condition will try to match the user part of the uri.
- For **match** select **contains** from the drop down menu. This means that the condition list will match anything that contains the entry in the value field.
- In the **value** field enter the digits to match using regular expression.

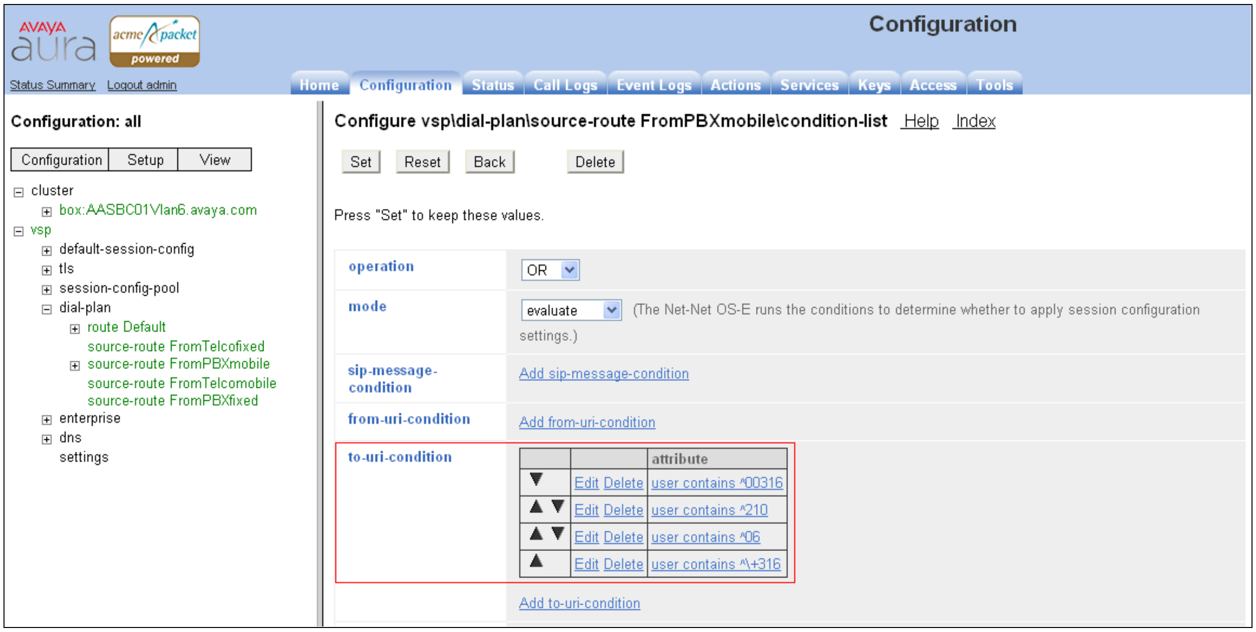
Click **Create** to save the condition.



The AASBC can be used to create the regular expression for the **value** field. Click the **(regular expression)** link next to the **value** field as seen in the previous screen. The following pop up box is displayed. Enter the digits to be matched and select the appropriate radio button for the type of match. The example below will match any digits beginning with 210, this will produce a regular expression of **^210**.

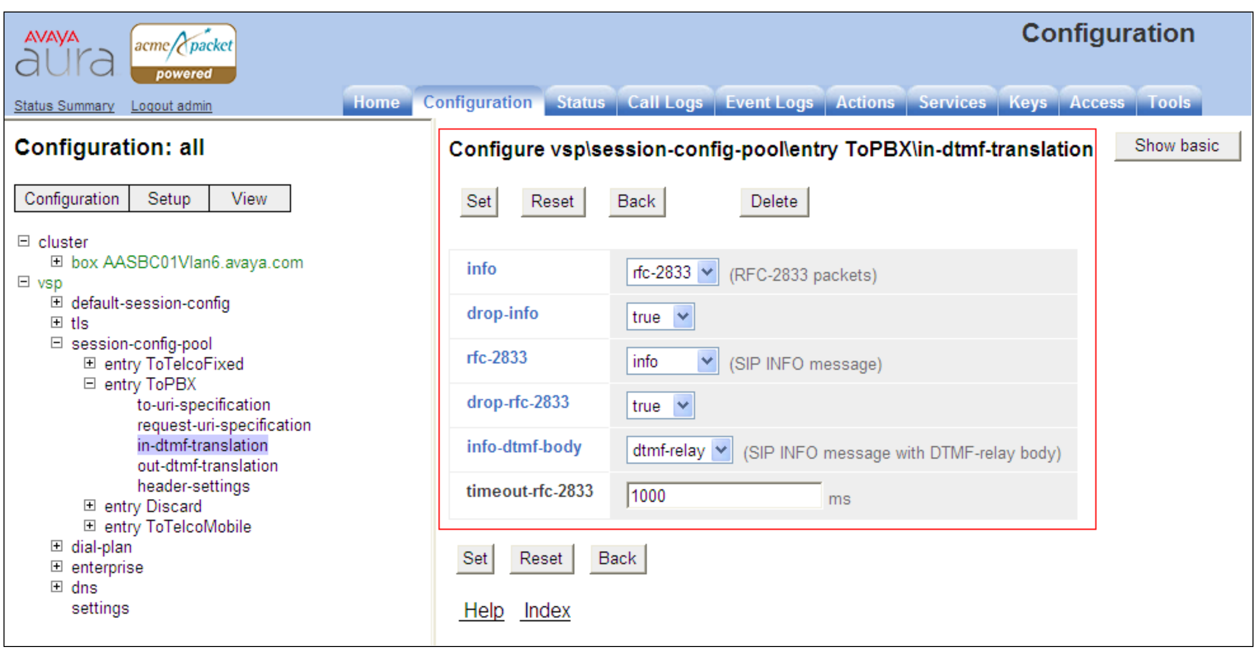


The following screen shows the to-uri-conditions used during the compliance test.

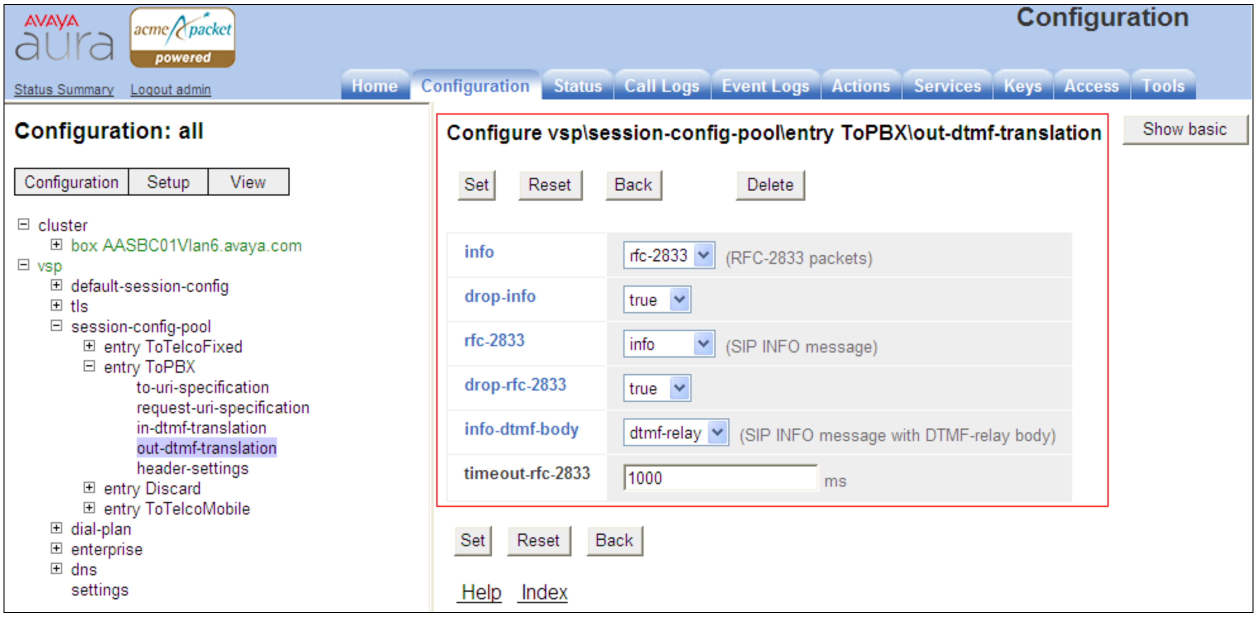


7.7. Mobile-X Mid call features

To allow Mobile-X mid-call features to work using DTMF tones from the mobile device, the tones need to be passed to the CS1000E as SIP INFO messages. In order to do this RFC2833 tones are converted into INFO messages using the AASBC. Navigate to **vsp → session-config-pool → entryToPBX → in-dtmf-translation** and click **Configure** (not shown). In the resulting screen update the fields as shown below and then click on **Set** to save:

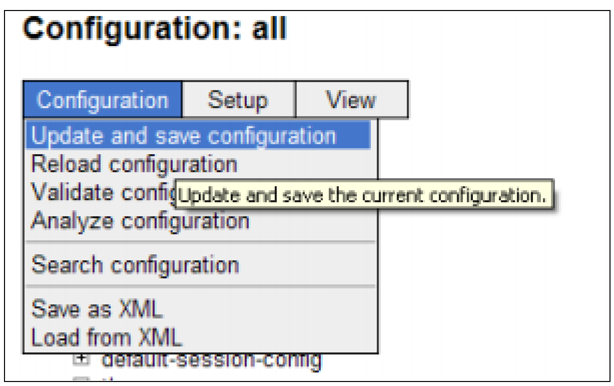


Navigate to **vsp → session-config-pool → entryToPBX → out-dtmf-translation** and click **Configure** (not shown). In the resulting screen update the fields as shown below and then click on **Set** to save:



7.8. Save the Configuration

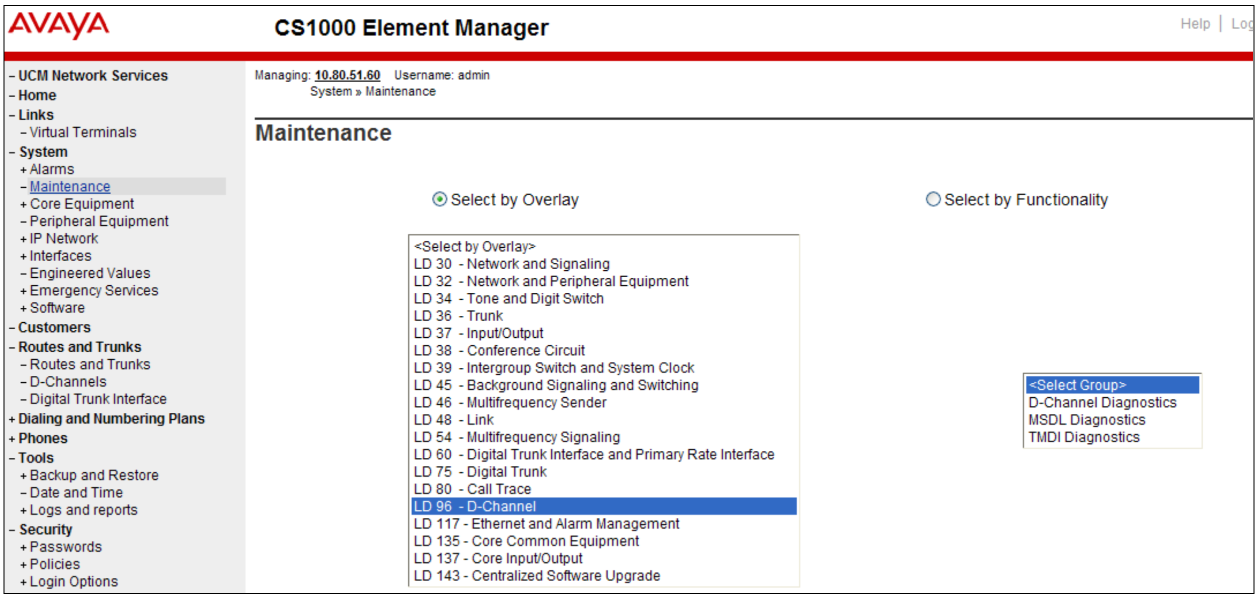
To save the configuration, click on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



8. Verification Steps

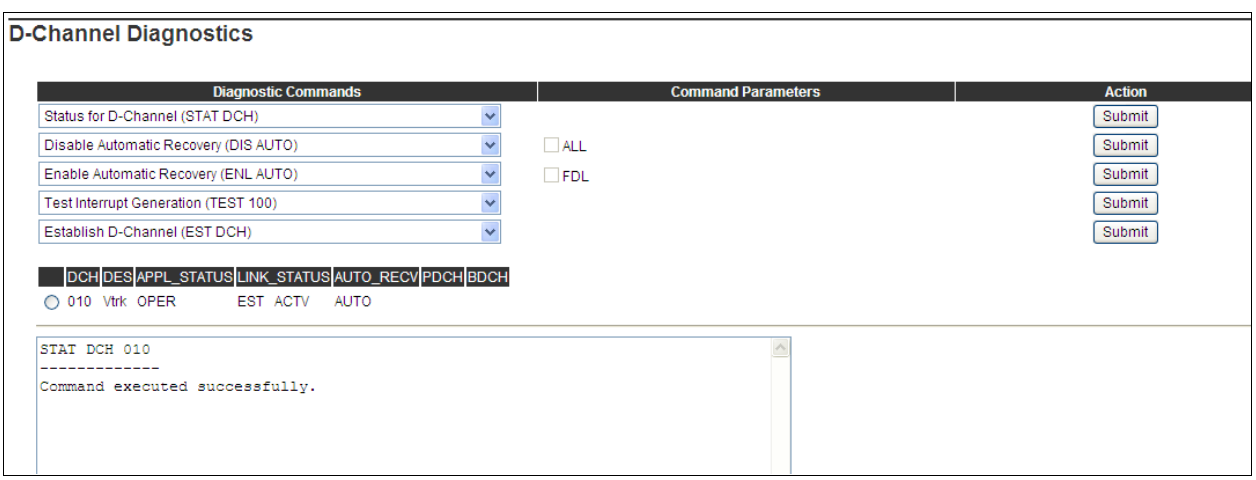
8.1. Verify Avaya Communication Server 1000E Operational Status

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:

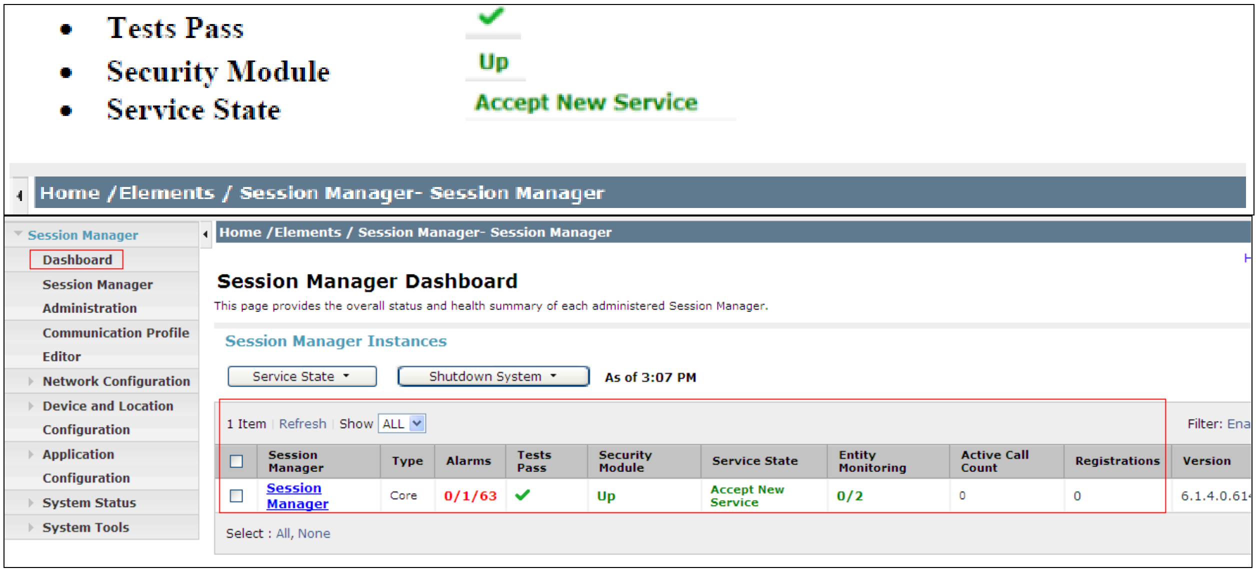
- **Appl_Status** Verify status is **OPER**
- **Link_Status** Verify status is **EST ACTV**



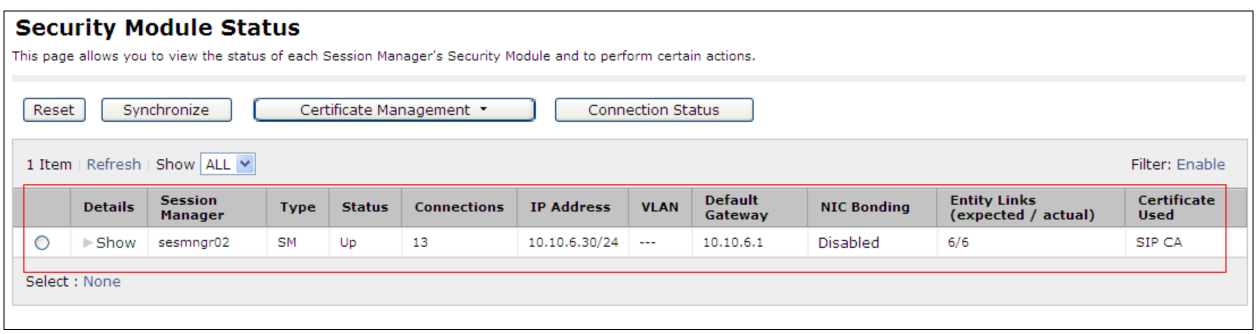
8.2. Verify Avaya Aura® Session Manager Operational Status

8.2.1. 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.

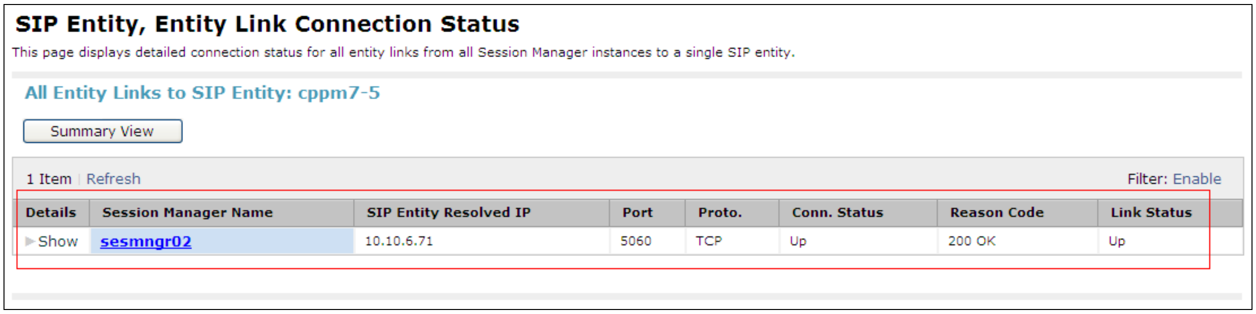


Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.



8.2.2. 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 Network Routing Server from 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: cppm7-5** table, verify the **Conn. Status** for the link is **Up** as shown below.


SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

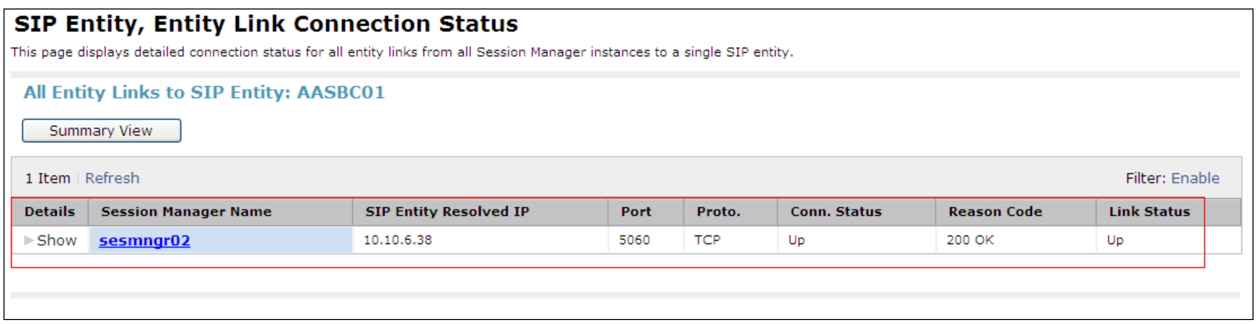
All Entity Links to SIP Entity: cppm7-5

Summary View

1 Item | Refresh   Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	sesmngr02	10.10.6.71	5060	TCP	Up	200 OK	Up

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


SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: AASBC01

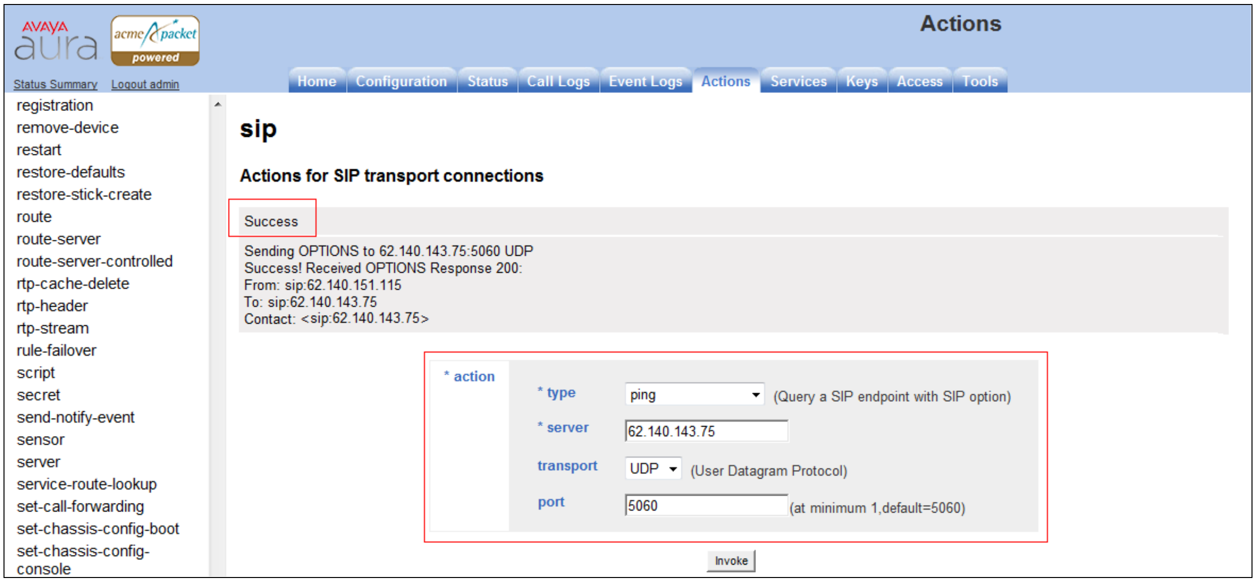
Summary View

1 Item | Refresh   Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	sesmngr02	10.10.6.38	5060	TCP	Up	200 OK	Up

8.3. Verify Avaya Aura ® Session Border Controller

From the AASBC **Actions** tab it is possible to send a SIP OPTIONS message to a specified IP address to confirm the correct response. Select **sip** from the left hand menu and select **ping** from the drop down menu in the **type** field. Enter the required IP address in the **server** field and specify the appropriate **transport** type and **port**. Click **Invoke** and the result of the test are shown towards the top of the page.



1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
3. Verify that the user on the PSTN can end an active call by hanging up.
4. Verify that an endpoint at the enterprise site can end an active call by hanging up.

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to Vodafone Netherlands SIP Trunk Solution comprising of Vodafone Office Voice and Vodafone OneVoice Corporate. Vodafone Netherlands SIP Trunk Solution is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. Vodafone Vodafone Netherlands SIP Trunk Solution comprising of Vodafone Office Voice and Vodafone OneVoice Corporate passed compliance testing. Please refer to **Section 2.2** for any observations or workarounds relating the testing covered by these Application Notes.

10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6, June 2010.

[2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.

[3] *Installing and Upgrading Avaya Aura® System ManagerRelease6.1*, November 2010.

[4] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473

[5] *Administering Avaya Aura® Session Manager,* March 2011, Document Number 03-603324.

[6] *Avaya Aura® Session Border Controller System Administration*, September 2010

[7] *Installing and Configuring Avaya Aura Session Border Controller*, May 2011

[8] *IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313, available at http://support.avaya.com*

[9] *Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116, available at http://support.avaya.com*

[10] *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*

[11] *Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125, available at http://support.avaya.com*

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

Additional Vodafone product documentation is available at http://www.vodafone.nl/zakelijk/totaal_oplossingen/vast_en_mobiel/

Appendix A – Avaya Communication Server 1000E Software

Communication Server 1000E call server patches and plug ins

```
TID: 46379

VERSION 4121

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

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

RELEASE 7
ISSUE 50 Q  +
IDLE_SET_DISPLAY NORTEL
DepList 1: core Issue: 01(created: 2011-09-13 15:12:45 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2011-10-11 13:28:54(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-09-21 10:45:48(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    wi00839337      ISS1:1OF1       DSP2AB06        16/04/2012  DSP2AB06.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 end
```

Communication Server 1000E call server deplists

```
VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2011-09-13 15:12:45 (est))

IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi00869243      ISS1:1OF1       p30848_1  07/11/2011  p30848_1.cpl   NO
001  WI00843571      ISS1:1OF1       p30627_1  07/11/2011  p30627_1.cpl   NO
002  wi00856702      ISS1:1OF1       p30573_1  07/11/2011  p30573_1.cpl   NO
003  WI00836292      ISS1:1OF1       p30554_1  07/11/2011  p30554_1.cpl   NO
004  wi00897176      ISS1:1OF1       p30418_1  07/11/2011  p30418_1.cpl   NO
005  wi00853178      ISS1:1OF1       p30719_1  07/11/2011  p30719_1.cpl   NO
006  wi00866570      ISS1:1OF1       p30477_1  07/11/2011  p30477_1.cpl   NO
007  wi00905600      ISS1:1OF1       p31201_1  07/11/2011  p31201_1.cpl   NO
008  wi00900766      ISS1:1OF1       p31159_1  07/11/2011  p31159_1.cpl   NO
009  wi00834382      ISS1:1OF1       p30548_1  07/11/2011  p30548_1.cpl   NO
010  wi00865477      ISS1:1OF1       p30897_1  07/11/2011  p30897_1.cpl   YES
011  wi00835294      ISS1:1OF1       p30565_1  07/11/2011  p30565_1.cpl   NO
012  wi00875701      ISS1:1OF1       p30942_1  07/11/2011  p30942_1.cpl   NO
013  wi00903085      ISS1:1OF1       p31164_1  07/11/2011  p31164_1.cpl   NO
014  wi00839134      ISS1:1OF1       p30698_1  07/11/2011  p30698_1.cpl   YES
015  wi00865477      ISS1:1OF1       p30890_1  07/11/2011  p30890_1.cpl   YES
016  wi00877367      ISS1:1OF1       p30534_1  07/11/2011  p30534_1.cpl   NO
017  wi00852365      ISS1:1OF1       p30707_1  07/11/2011  p30707_1.cpl   NO
018  WI00854150      ISS1:1OF1       p30468_1  07/11/2011  p30468_1.cpl   NO
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019	WI00853473	ISS1:1OF1	p30625_1	07/11/2011	p30625_1.cpl	NO
020	wi00865477	ISS1:1OF1	p30895_1	07/11/2011	p30895_1.cpl	YES
021	wi00841273	ISS1:1OF1	p30713_1	07/11/2011	p30713_1.cpl	NO
022	wi00905097	ISS1:1OF1	p31194_1	07/11/2011	p31194_1.cpl	NO
023	wi00865477	ISS1:1OF1	p30893_1	07/11/2011	p30893_1.cpl	YES
024	wi00838073	ISS1:1OF1	p30588_1	07/11/2011	p30588_1.cpl	NO
025	wi00879508	ISS1:1OF1	p30956_1	07/11/2011	p30956_1.cpl	NO
026	wi00871969	ISS1:1OF1	p30768_1	07/11/2011	p30768_1.cpl	NO
027	wi00853658	ISS1:1OF1	p30990_1	07/11/2011	p30990_1.cpl	NO
028	wi00856410	ISS1:1OF1	p30749_1	07/11/2011	p30749_1.cpl	NO
029	wi00906163	ISS1:1OF1	p31205_1	07/11/2011	p31205_1.cpl	NO
030	wi00907697	ISS1:1OF1	p31227_1	07/11/2011	p31227_1.cpl	NO
031	wi00896420	ISS1:1OF1	p30867_1	07/11/2011	p30867_1.cpl	NO
032	wi00839821	ISS1:1OF1	p30619_1	07/11/2011	p30619_1.cpl	NO
033	wi00860279	ISS1:1OF1	p30789_1	07/11/2011	p30789_1.cpl	NO
034	wi00865477	ISS1:1OF1	p30891_1	07/11/2011	p30891_1.cpl	YES
035	WI00889786	ISS1:1OF1	p30750_1	07/11/2011	p30750_1.cpl	NO
036	wi00863876	ISS1:1OF1	p30787_1	07/11/2011	p30787_1.cpl	NO
037	wi00921340	ISS1:1OF1	p31266_1	07/11/2011	p31266_1.cpl	NO
038	wi00900668	ISS1:1OF1	p30456_1	07/11/2011	p30456_1.cpl	NO
039	wi00908598	ISS1:1OF1	p31235_1	07/11/2011	p31235_1.cpl	NO
040	wi00896680	ISS1:1OF1	p30357_1	07/11/2011	p30357_1.cpl	NO
041	wi00869695	ISS1:1OF1	p30654_1	07/11/2011	p30654_1.cpl	NO
042	wi00854130	ISS1:1OF1	p30443_1	07/11/2011	p30443_1.cpl	NO
043	wi00883604	ISS1:1OF1	p30973_1	07/11/2011	p30973_1.cpl	NO
044	wi00905297	ISS1:1OF1	p31195_1	07/11/2011	p31195_1.cpl	NO
045	wi00854415	ISS1:1OF1	p30593_1	07/11/2011	p30593_1.cpl	NO
046	wi00827950	ISS2:1OF1	p30471_2	07/11/2011	p30471_2.cpl	NO
047	wi00859123	ISS1:1OF1	p30648_1	07/11/2011	p30648_1.cpl	NO
048	wi00897082	ISS1:1OF1	p31124_1	07/11/2011	p31124_1.cpl	NO
049	wi00906022	ISS1:1OF1	p31202_1	07/11/2011	p31202_1.cpl	NO
050	wi00859499	ISS1:1OF1	p30694_1	07/11/2011	p30694_1.cpl	NO
051	wi00871739	ISS1:1OF1	p30856_1	07/11/2011	p30856_1.cpl	NO
052	wi00894443	ISS1:1OF1	p31093_1	07/11/2011	p31093_1.cpl	NO
053	wi00850521	ISS1:1OF1	p30709_1	07/11/2011	p30709_1.cpl	YES
054	wi00839255	ISS1:1OF1	p30591_1	07/11/2011	p30591_1.cpl	NO
055	wi00908933	ISS1:1OF1	p31239_1	07/11/2011	p31239_1.cpl	NO
056	wi00865477	ISS1:1OF1	p30892_1	07/11/2011	p30892_1.cpl	YES
057	wi00905660	ISS1:1OF1	p27968_1	07/11/2011	p27968_1.cpl	NO
058	wi00841980	ISS1:1OF1	p30618_1	07/11/2011	p30618_1.cpl	NO
059	wi00879526	ISS1:1OF1	p31007_1	07/11/2011	p31007_1.cpl	NO
060	wi00895090	ISS1:1OF1	p31105_1	07/11/2011	p31105_1.cpl	NO
061	wi00865477	ISS1:1OF1	p30894_1	07/11/2011	p30894_1.cpl	YES
062	wi00898168	ISS1:1OF1	p31131_1	07/11/2011	p31131_1.cpl	NO
063	wi00895181	ISS1:1OF1	p31106_1	07/11/2011	p31106_1.cpl	NO
064	wi00832106	ISS1:1OF1	p30550_1	07/11/2011	p30550_1.cpl	NO
065	wi00881777	ISS1:1OF1	p25747_1	07/11/2011	p25747_1.cpl	NO
066	wi00836182	ISS1:1OF1	p30450_1	07/11/2011	p30450_1.cpl	NO
067	wi00686981	ISS1:1OF1	p30706_1	07/11/2011	p30706_1.cpl	NO
068	wi00852389	ISS1:1OF1	p30641_1	07/11/2011	p30641_1.cpl	NO
069	wi00858335	ISS1:1OF1	p30819_1	07/11/2011	p30819_1.cpl	NO
070	WI00836334	ISS1:1OF1	p30481_1	07/11/2011	p30481_1.cpl	NO
071	wi00887744	ISS2:1OF1	p31026_2	07/11/2011	p31026_2.cpl	NO
072	wi00865477	ISS1:1OF1	p30896_1	07/11/2011	p30896_1.cpl	YES
073	wi00865477	ISS1:1OF1	p30898_1	07/11/2011	p30898_1.cpl	YES
074	wi00903437	ISS1:1OF1	p31167_1	07/11/2011	p31167_1.cpl	NO
075	wi00888680	ISS1:1OF1	p30399_1	07/11/2011	p30399_1.cpl	NO
076	wi00842409	ISS1:1OF1	p30621_1	07/11/2011	p30621_1.cpl	NO
077	wi00857362	ISS1:1OF1	p30782_1	07/11/2011	p30782_1.cpl	NO
078	wi00882293	ISS1:1OF1	p31010_1	07/11/2011	p31010_1.cpl	NO
079	wi00894243	ISS1:1OF1	p31087_1	07/11/2011	p31087_1.cpl	NO
080	WI00900213	ISS1:1OF1	p30656_1	07/11/2011	p30656_1.cpl	NO
081	wi00897096	ISS1:1OF1	p30676_1	07/11/2011	p30676_1.cpl	NO
082	wi00899584	ISS1:1OF1	p30809_1	07/11/2011	p30809_1.cpl	NO
083	WI00839794	ISS1:1OF1	p28647_1	07/11/2011	p28647_1.cpl	NO
084	wi00857566	ISS1:1OF1	p30766_1	07/11/2011	p30766_1.cpl	NO
085	wi00903381	ISS1:1OF1	p30421_1	07/11/2011	p30421_1.cpl	NO
086	wi00873382	ISS1:1OF1	p30832_1	07/11/2011	p30832_1.cpl	NO
087	wi00876855	ISS1:1OF1	p30952_1	07/11/2011	p30952_1.cpl	NO
088	wi00886321	ISS1:1OF1	p31009_1	07/11/2011	p31009_1.cpl	NO
089	wi00826075	ISS1:1OF1	p30452_1	07/11/2011	p30452_1.cpl	NO

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090  wi00875425      ISS1:1OF1       p30943_1  07/11/2011  p30943_1.cpl   NO
091  wi00833910      ISS2:1OF1       p30492_2  07/11/2011  p30492_2.cpl   NO
092  wi00921295      ISS1:1OF1       p31265_1  07/11/2011  p31265_1.cpl   NO
093  wi00824257      ISS1:1OF1       p30447_1  07/11/2011  p30447_1.cpl   NO
094  wi00898652      ISS1:1OF1       p31158_1  07/11/2011  p31158_1.cpl   NO
095  wi00836981      ISS1:1OF1       p30613_1  07/11/2011  p30613_1.cpl   NO
096  wi00877442      ISS1:1OF1       p30844_1  07/11/2011  p30844_1.cpl   NO
097  wi00884699      ISS1:1OF1       p31000_1  07/11/2011  p31000_1.cpl   YES
098  wi00840590      ISS1:1OF1       p30767_1  07/11/2011  p30767_1.cpl   NO
099  wi00854409      ISS1:1OF1       p30479_1  07/11/2011  p30479_1.cpl   NO
100  wi00832626      ISS2:1OF1       p30560_2  07/11/2011  p30560_2.cpl   NO
101  wi00837461      ISS1:1OF1       p30597_1  07/11/2011  p30597_1.cpl   NO
102  wi00905550      ISS1:1OF1       p31220_1  07/11/2011  p31220_1.cpl   NO
103  wi00868729      ISS1:1OF1       p31163_1  07/11/2011  p31163_1.cpl   NO
104  wi00843623      ISS1:1OF1       p30731_1  07/11/2011  p30731_1.cpl   YES
105  wi00888396      ISS1:1OF1       p31027_1  07/11/2011  p31027_1.cpl   NO
106  wi00853031      ISS1:1OF1       p30531_1  07/11/2011  p30531_1.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2011-10-11 13:28:54(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-09-21 10:45:48(est)
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Communication Server 1000E signaling server service updates

```
Product Release: 7.50.17.00
In system patches: 2
PATCH#  NAME       IN_SERVICE   DATE      SPECINS  TYPE   RPM
14      p30260_1   Yes          13/12/11  NO       FRU    cs1000-pi-control-1.00.00.00-00.noarch
15      p30253_1   Yes          02/03/12  NO       FRU    cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 15
PATCH#  IN_SERVICE   DATE       SPECINS   REMOVABLE   NAME
0       Yes          22/09/11   NO        YES         cs1000-linuxbase-7.50.17.16-3.i386.000
1       Yes          22/09/11   NO        YES         cs1000-baseWeb-7.50.17.16-1.i386.001
2       Yes          22/09/11   NO        YES         cs1000-patchWeb-7.50.17.16-2.i386.000
3       Yes          22/09/11   NO        YES         cs1000-kcv-7.50.17.16-1.i386.000
4       Yes          26/09/11   NO        YES         cs1000-dbcom-7.50.17-02.i386.000
5       Yes          26/09/11   NO        yes         cs1000-sps-7.50.17.16-01.i386.000
6       Yes          26/09/11   NO        YES         cs1000-shared-pbx-7.50.17.16-1.i386.000
7       Yes          26/09/11   NO        YES         cs1000-dmWeb-7.50.17.16-1.i386.000
8       Yes          26/09/11   NO        YES         cs1000-tps-7.50.17.16-5.i386.000
9       Yes          26/09/11   NO        YES         cs1000-ipsec-7.50.17.16-1.i386.000
10      Yes          26/09/11   NO        YES         cs1000-bcc-7.50.17.16-19.i386.000
11      Yes          26/09/11   NO        YES         cs1000-Jboss-Quantum-7.50.17.16-5.i386.000
12      Yes          26/09/11   NO        YES         cs1000-ftrpkg-7.50.17.16-5.i386.000
13      Yes          26/09/11   NO        YES         cs1000-emWeb_6-0-7.50.17.16-7.i386.000
16      Yes          02/03/12   NO        YES         cs1000-vtrk-7.50.17.16-46.i386.000
```

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     [patched]
   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                       n/a         [patched]
   Snmp-Daemon-TrapLib         7.50.17
   ISECSH                      7.50.17
   patchWeb                    n/a         [patched]
   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                          7.50.17.16  [patched]
   vtrk                         7.50.17.16  [patched]
   pd                           7.50.17
   sps                          7.50.17.16  [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     [patched]
   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
```

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.