Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.0, Avaya Network Routing Server and Avaya Aura® Session Manager R6.1 to support Vodafone Germany SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Vodafone Germany SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Manager. Avaya Network Routing Server and Avaya Communication Server 1000E.

Vodafone 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 Vodafone Germany SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager, Avaya Network Routing Server (NRS) and Avaya Communication Server 1000E (CS1000E) connected to Vodafone Germany SIP Trunk Service. Customers using this Avaya SIP-enabled enterprise solution with Vodafone Germany 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 CS1000E, NRS and Session Manager. The enterprise site was configured to use the SIP Trunk Service provided by Vodafone Germany.

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 Vodafone Germany. 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 Vodafone Germany SIP Trunk Service and thus was not tested.
- Calls using G.729 and G.711A codec's tested.
- T.38 for fax is not supported by Vodafone and thus was not tested.
- 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.
- Mobile-X mid call features was not tested.

2.2. Test Results

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

- Inbound or outbound fax using T.38 was not tested. Vodafone does not support T.38 on their network.
- Incoming call to a SIP Client using DTMF RFC 2833 does not work. Vodafone uses payload type 106 for telephone event. The CS1000E agrees to use this but then sends out an INVITE to use payload type 101, Vodafone respond with 200OK but continues to use payload type 106 for DTMF. The DTMF tones are sent from Vodafone with an unrecognized payload type that the CS1000E does not recognize.

- Mobile-X handoff does not work from twinned desk phone. INVITE sent to PSTN mobile contains no SDP information so the call fails. Vodafone Germany does not support an INVITE that does not contain any SDP content. This behaviour of sending the INVITE with no SDP from the CS1000E is per design.
- DTMF tones incoming to a Call Pilot application do not work. Vodafone use payload type 106 for telephone event. The CS1000E agrees to use this but then sends out an INVITE to use payload type 101, Vodafone respond with 200OK but continue to use payload type 106 for DTMF. The DTMF tones are sent from Vodafone with an unrecognized payload type that the CS1000E/Call Pilot does not recognize.
- Blind transfers do not work between the CS1000E and Vodafone's SIP network.
- In the live environment it has been noticed that the Diversion to History-Info Header Adaptation is needed on the Session Manager (DiversionTypeAdapter). This was not configured during the Compliance testing. Vodafone require the use of the Diversion header, rather than the History-Info header to provide information related to how and why the call arrives to a specific application or user. This is relevant for Call Forward call scenarios.

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 DE products please visit the website at www.vodafone.de or contact an authorized Vodafone DE representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Vodafone Germany using SIP Trunks. Located at the enterprise site are Session Manager, NRS and a Communication Server 1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series IP telephones, Avaya IP Soft phone SMC3456, 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.

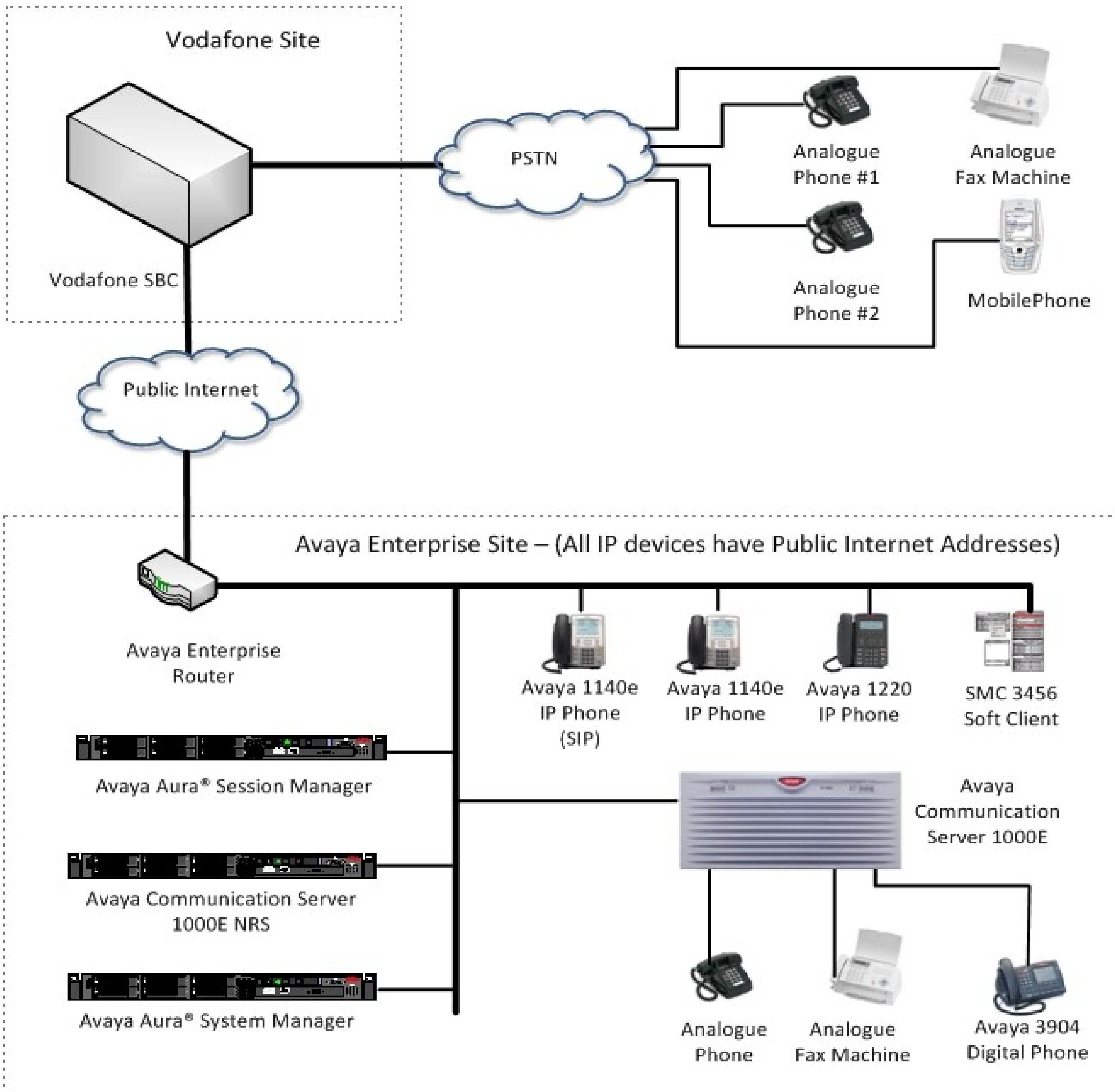


Figure 1: Network Topology of Vodafone Germany SIP Trunk with Avaya Communication Server 1000E

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: 6.1.4.0.614005
Avaya S8800 server	Avaya Aura® System Manager R6.1 Load: 6.1.8.1.1551Service Pack 4
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R 7.0 Service Pack 7.00.20 Deplist: x21 07.00Q + All CS1000E patches listed in **Appendix A**
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC BD01 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA07 DSP1 Version: DSP1 AB03
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C7J
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 57666
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
Vodafone Germany SIP Trunk Service	Softswitch: iSSW - 20.50.34-olxi SBC: Acme Packet 4250 - 6.1.0 MR5 MG: Cisco MGX 8880 - 5.5(10.204)P4

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 NRS and also between the NRS and the Session Manager. These SIP trunks carry SIP Signaling associated with Vodafone Germany's SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from Vodafone's SBC. Once a SIP message arrives at the Session Manager incoming digit translation is performed before the SIP messages are directed to the NRS, the NRS will then direct the SIP messages to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, further incoming call treatment such as 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 NRS. The NRS directs the outbound SIP messages to the Session Manager and this then directs the traffic on to Vodafone Germany'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. **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.

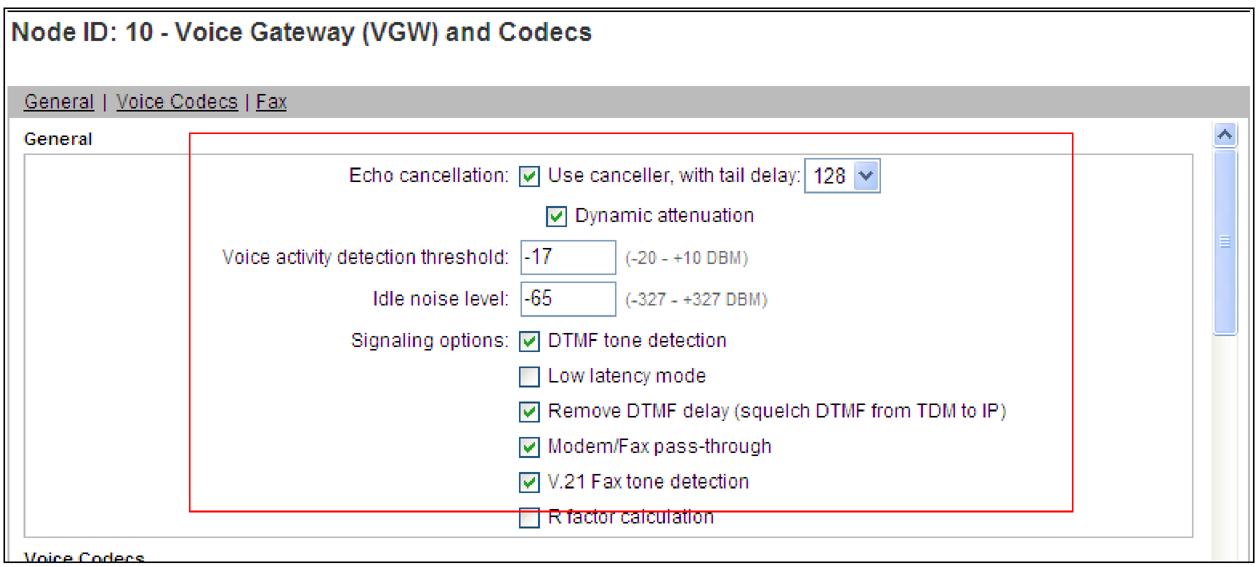
```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

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).

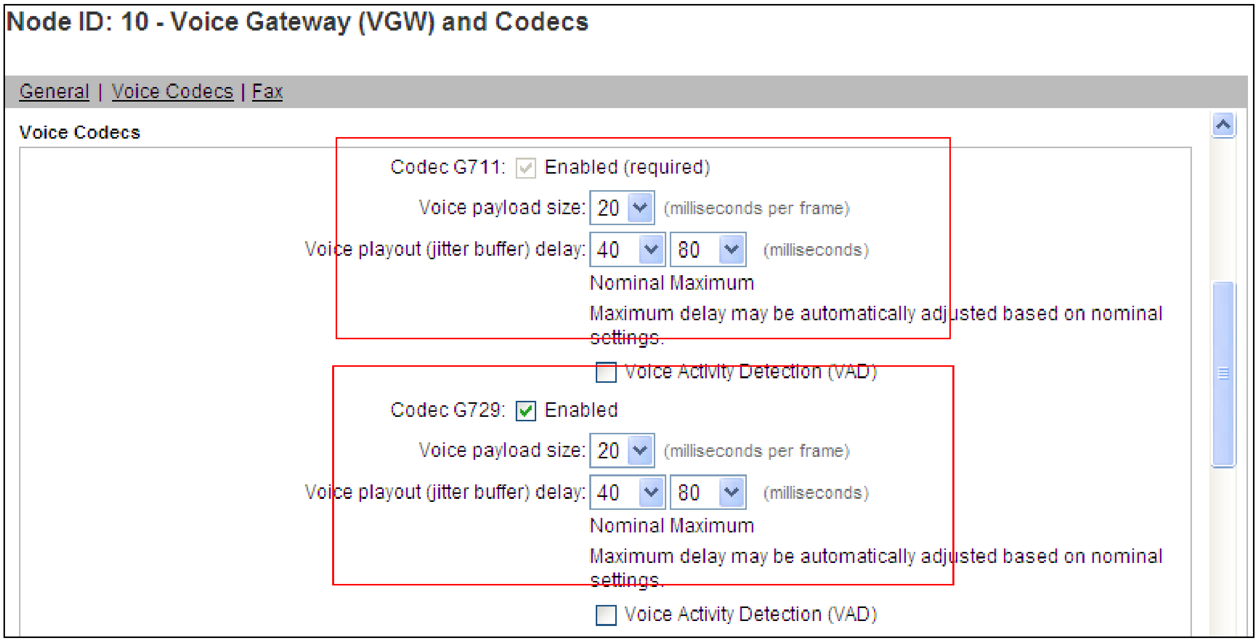
```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

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 Germany 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 → 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.

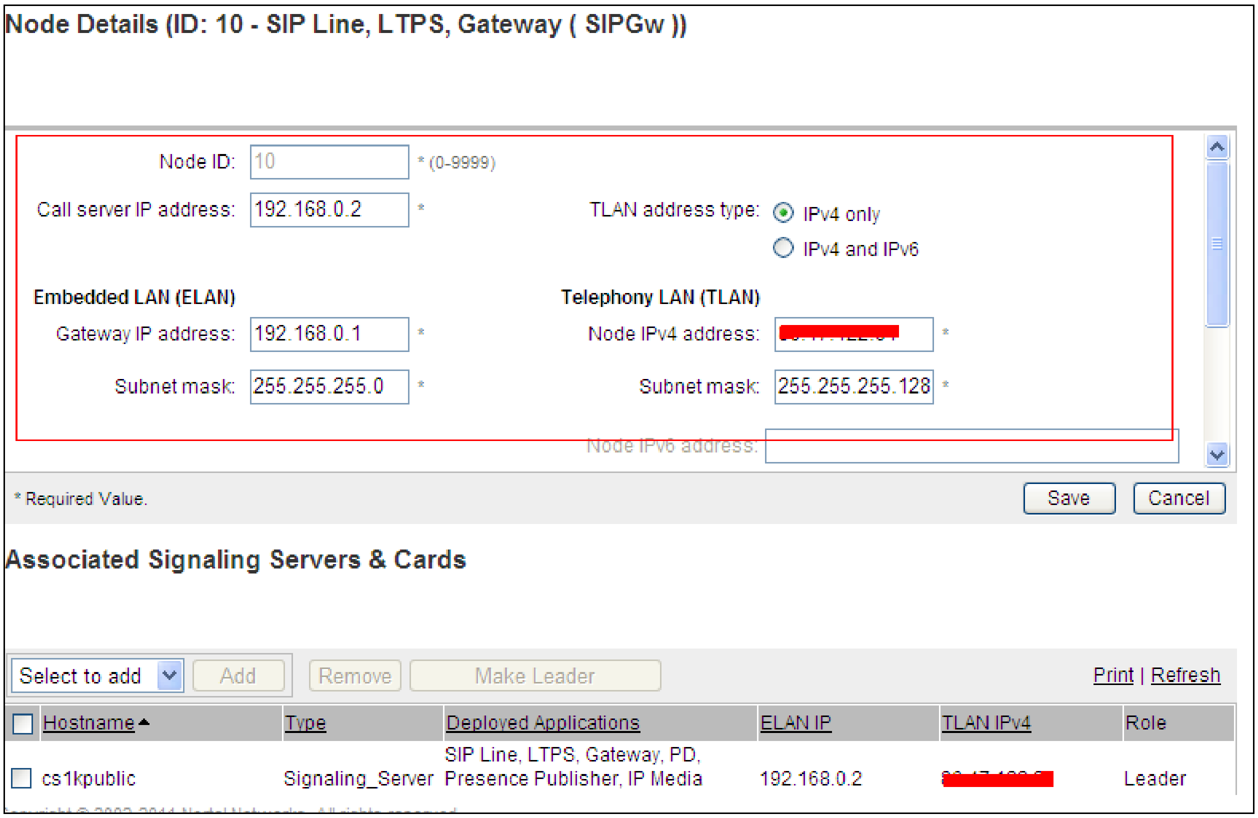


Next, scroll down and configure the **G.711** and **G.729** codec settings. The relevant settings are highlighted in the following screenshot.



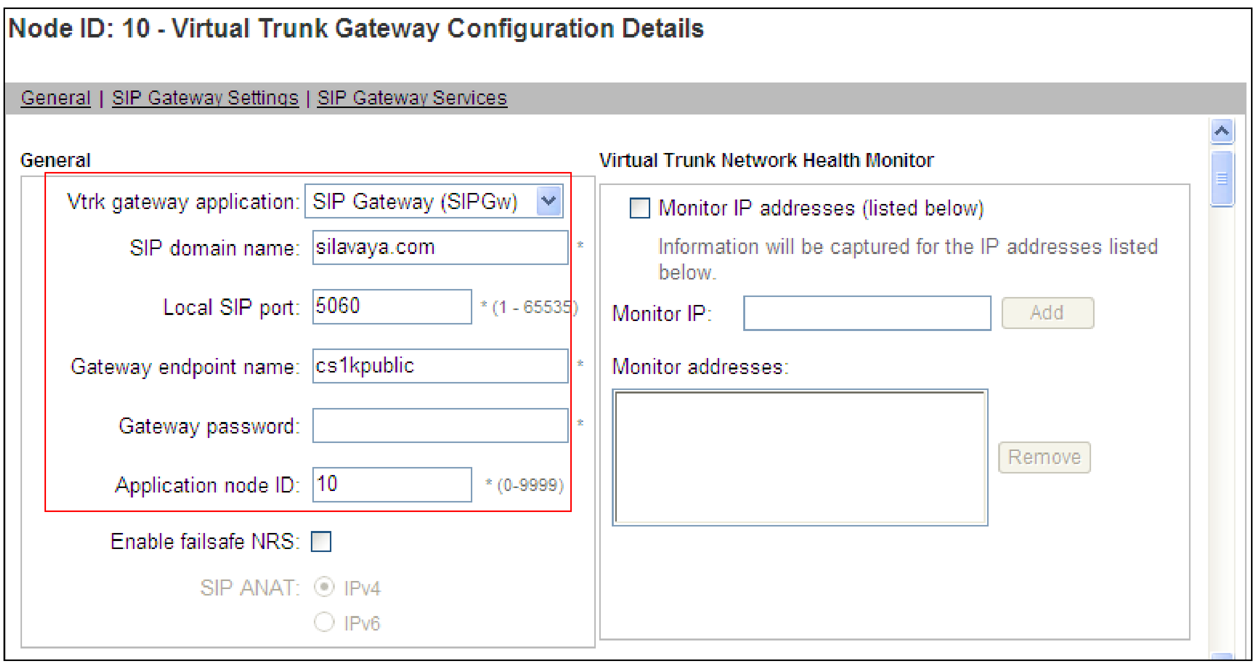
5.4. Virtual Trunk Gateway Configuration

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 NRS. The system in the lab is a Co-Resident Call server and Signalling Server system which also acts as the NRS.

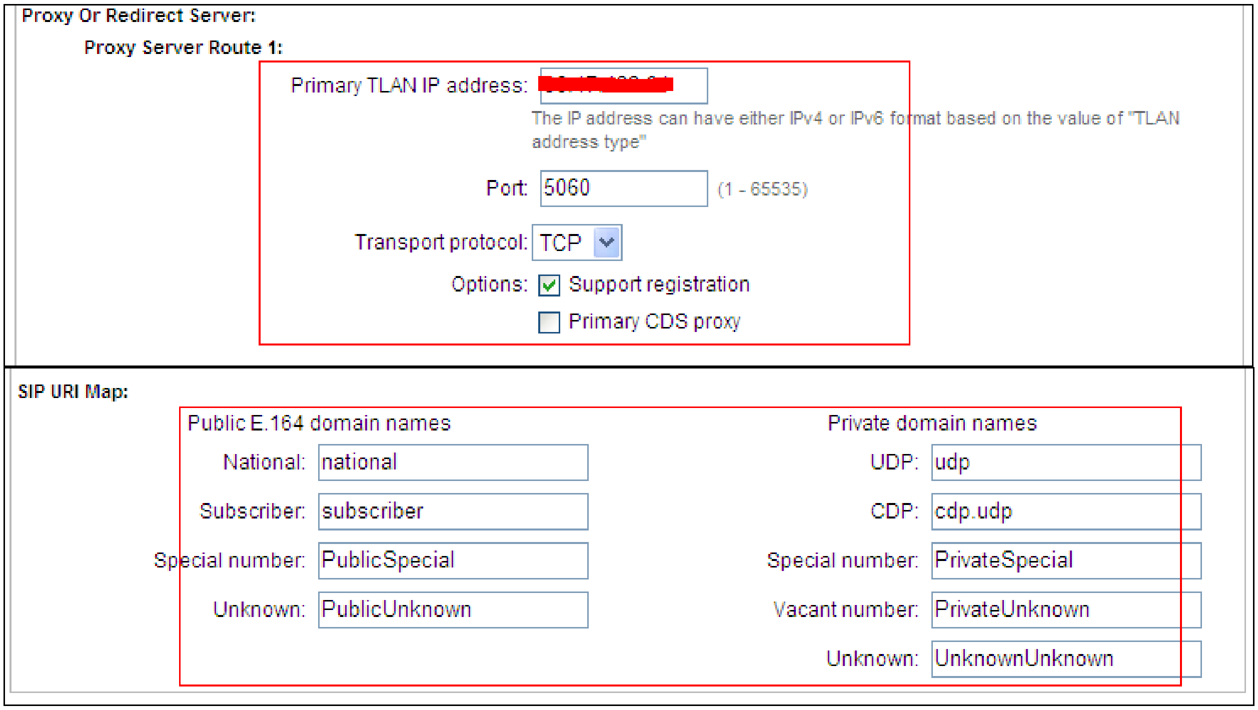


The next two screenshots show the SIP Virtual Trunk Gateway configuration, navigate to **System → IP Networks → IP Telephony Nodes → Node Details → Gateway (SIPGW) Virtual Trunk Configuration Details** and fill in the highlighted areas with the relevant settings.

- **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 **silavaya.com**. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the Session Manager and on the NRS.
- **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 10

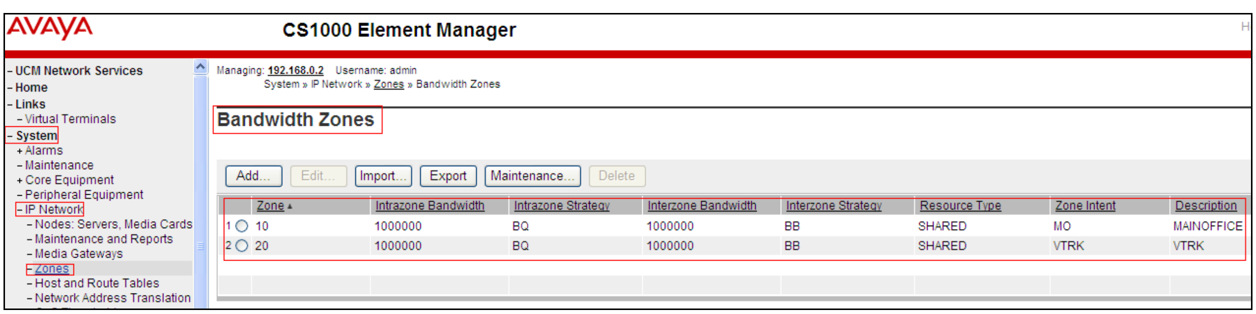


- **Proxy or Redirect Server:** Primary TLAN IP address is the NRS IP address. The **Transport protocol** used for **SIP**, in this case is **TCP**
- **SIP URI Map:** All values are left as default



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. 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 Germany'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 5.5** 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 10
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 e.g. 1. **DMI 1** is used for international traffic because **CYTP** (Call type) is **INTL**. **DMI 2** is used for local traffic because **CYTP** is **CDP**. In both cases **DEL** (delete digits) is **1**, this deletes the first digit dialed, in the same configuration that was access code 9 for an outside line.

Overlay 87	Overlay 87
REQ new	REQ new
FEAT dgt	FEAT dgt
DMI 1	DMI 2
DEL 1	DEL 1
CTYP INTL	CTYP CDP

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                                          FCI   0
new                                                 FSNI  0
CUST 0                                              BNE   NO
FEAT rlb                                            DORG  NO
RLI  66                                             SBOC  NRR
ELC  NO                                             PROU  1
ENTR 0                                              IDBB  DBD
LTER NO                                             IOHQ  NO
ROUT 100                                            OHQ   NO
TOD  0 ON  1 ON  2 ON  3 ON                         CBQ   NO
     4 ON  5 ON  6 ON  7 ON
VNS  NO                                             ISET  0
SCNV NO                                             NALT  5
CNV  NO                                             MFRL  0
EXP  NO                                             OVLL  0
FRL  0
DMI  1
CTBL 0
ISDM 0
```

This example shows a RDB defined for local traffic and uses DMI 2 as previously entered in overlay 87.

```
Overlay 86                                          FCI   0
new                                                 FSNI  0
CUST 0                                              BNE   NO
FEAT rlb                                            DORG  NO
RLI  67                                             SBOC  NRR
ELC  NO                                             PROU  1
ENTR 0                                              IDBB  DBD
LTER NO                                             IOHQ  NO
ROUT 100                                            OHQ   NO
TOD  0 ON  1 ON  2 ON  3 ON                         CBQ   NO
     4 ON  5 ON  6 ON  7 ON
VNS  NO                                             ISET  0
SCNV NO                                             NALT  5
CNV  NO                                             MFRL  0
EXP  NO                                             OVLL  0
FRL  0
DMI  2
CTBL 0
ISDM 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 900 FLEN 15 ITOH NO **RLI 66**	TSC 908 FLEN 11 ITOH NO **RLI 67**	TSC 9118 FLEN 15 ITOH NO **RLI 67**	

5.7. 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**, 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----
```

```
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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 8101 0      MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8101 0
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     02
     03 BSY
     04 DSP
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
```

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

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

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

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

MLNG ENG
DNDR 0
KEY  00 MCR 8102 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 8102 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 **FAXD** and **MPTA** configure the port for modem pass-through fax transmission. Vodafone do not support T.38 for fax.

Overlay 20 – Analog Telephone Configuration

```
DES  500
TN   100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN   8104
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 FAXD CNUD CNAD PGND FTTC
     FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTA
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR  DCFW 4
```

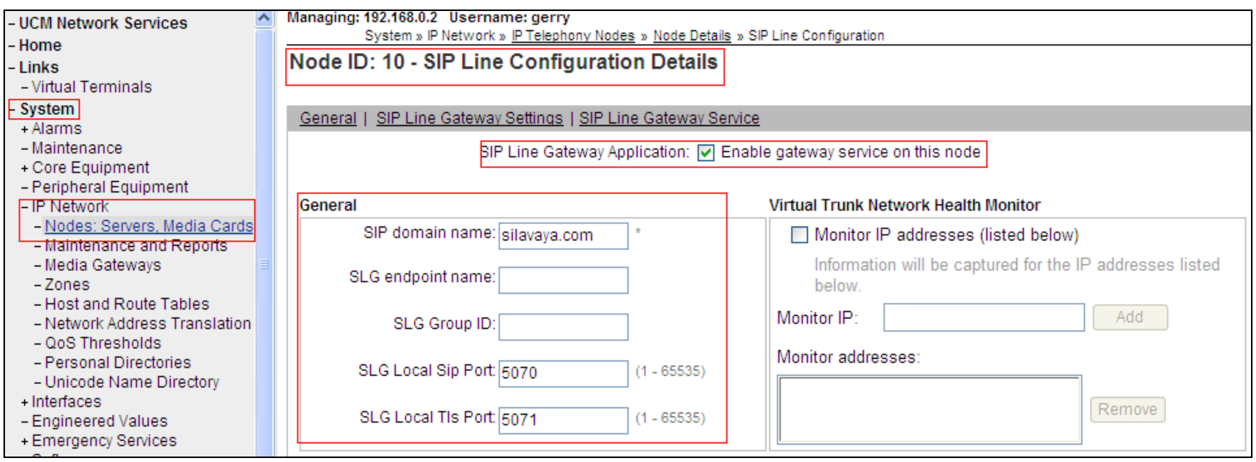
5.8. Configure the SIP Line Gateway Service

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

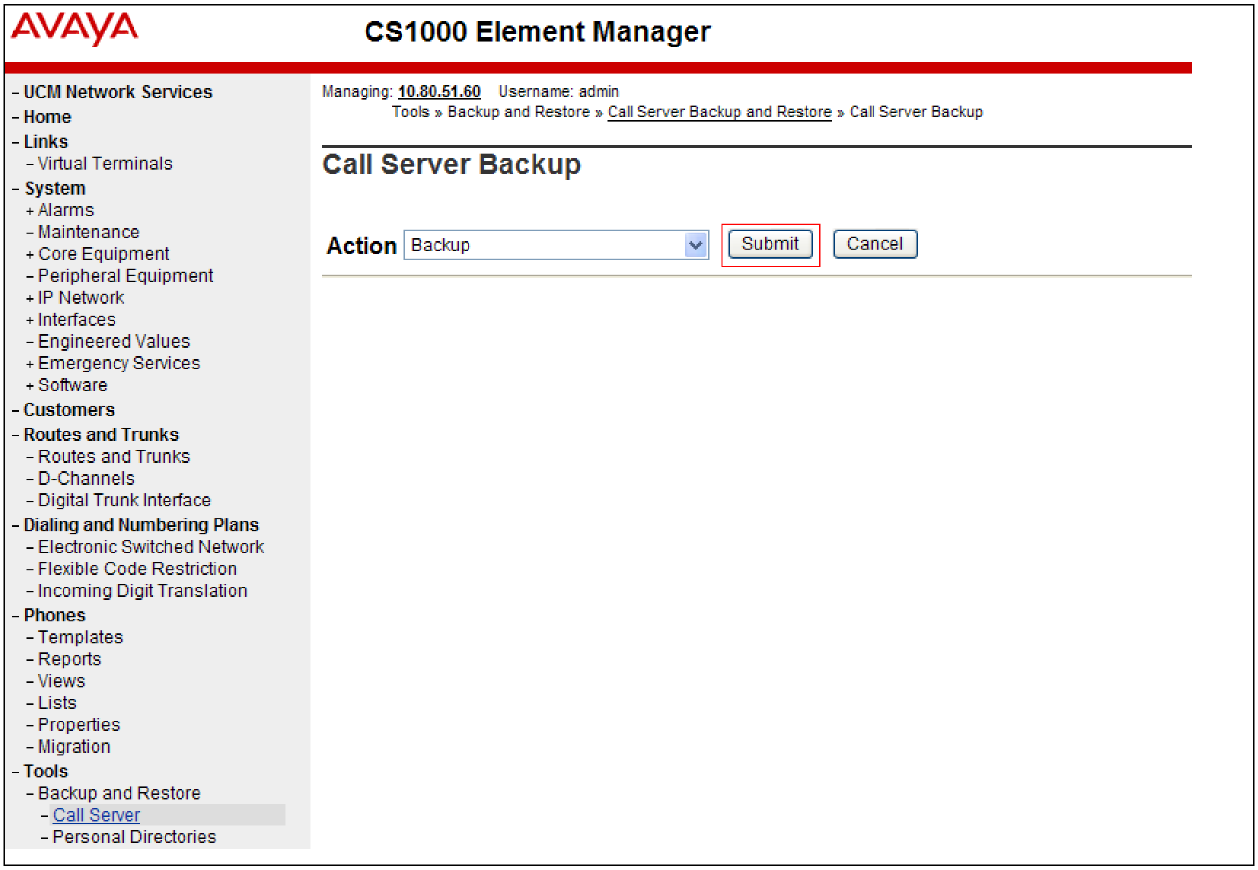
When SIP Line service configuration is completed, use the Communication Server 1000E system terminal and overlay 20 to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value 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 **SIPLINEZONE** 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**.

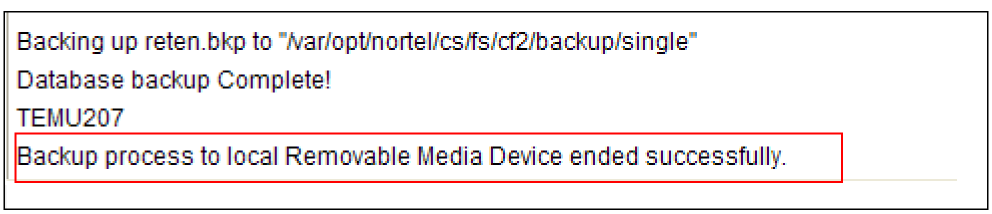
```
Overlay 20 – SIP Telephone Configuration
DES  SIPD
TN   096 0 01 15  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 8105
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
ERL  0
ECL  0
VSIT NO
FDN
TGAR 0
LDN  NO
NCOS 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---
```

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

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

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 8105 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 788105 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. Configure Avaya Network Routing Server

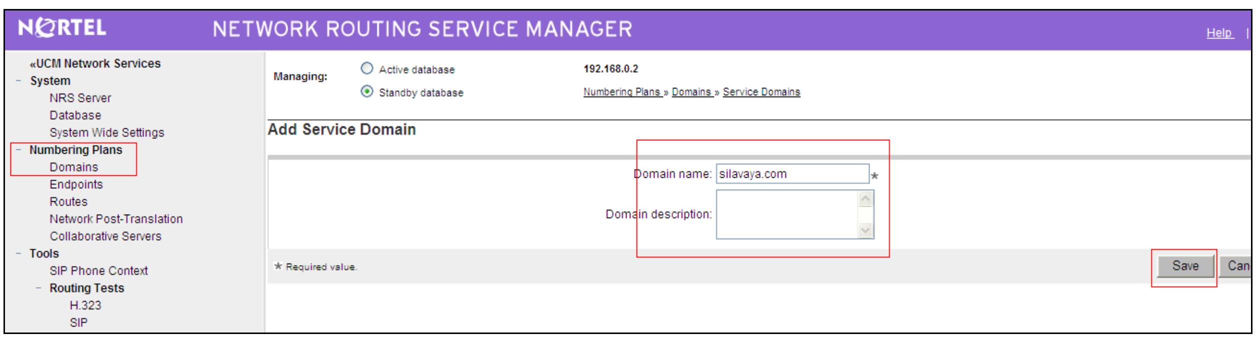
This section provides the procedure for configuring the NRS to receive and route calls over the SIP trunk between the Session Manager and the CS1000E. These instructions assume other administration activities have already been completed such as defining system wide settings. The following administration activities will be described.

- Define SIP Domain
- Define Endpoints
- Define Routes

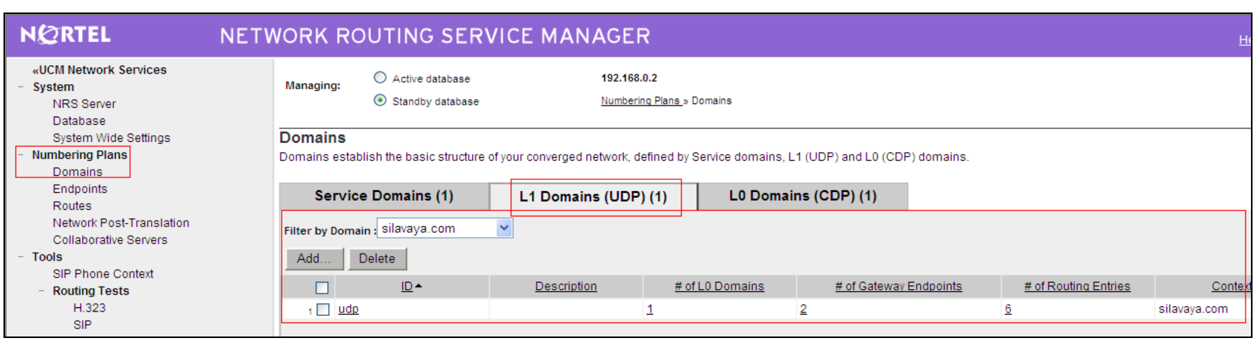
Configuration is accomplished by accessing the browser-based GUI of the Unified Communications Manager (UCM), using the URL **https://<ip-address>/network-login**, where **<ip-address>** is the IP address of UCM. Once logged on click on the NRS Manager link on the UCM front page and log in with the appropriate credentials.

6.1. Define SIP Domain

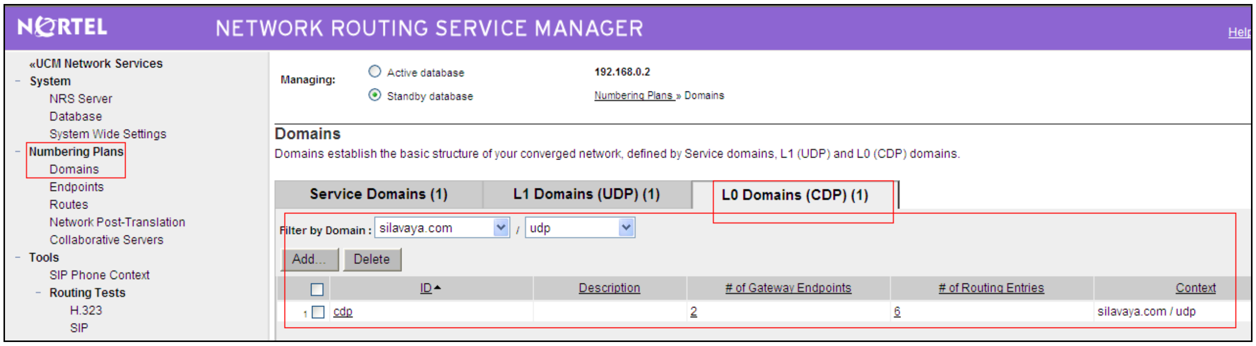
Create a SIP Domain by clicking on **Numbering Plans → Domains**, click **Add** (not shown). Enter a name for your Domain name and click on **Save**. In the test configuration **silavaya.com** was used and this is the same domain that is configured on the CS1000E as per **Section 5.4** and on the Session Manager per **Section 7.1**.



Use the same procedure as above to add a **UDP** and a **CDP** domain for **silavaya.com**. These domains are sub-domains to silavaya.com. This is the UDP domain created for silavaya.com.

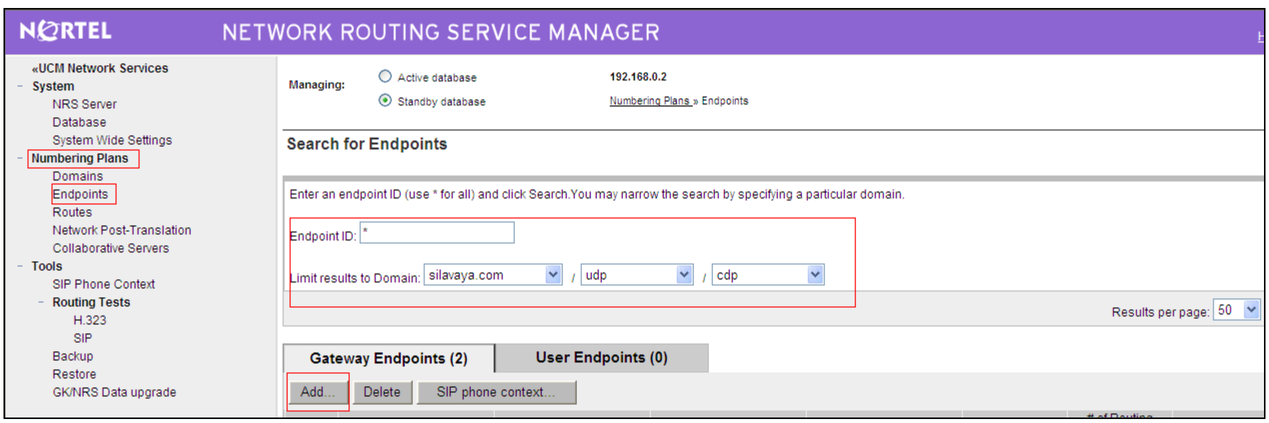


This is the CDP domain created for silavaya.com/udp domain.



6.2. Define Endpoints

For this test configuration two endpoints were configured on the NRS. A static endpoint was configured for the Session Manager and a dynamic endpoint for the CS1000E. Create an Endpoint by clicking on **Numbering Plans → Endpoints**. Select the domain and sub-domains (e.g. **silavaya.com/udp/cdp**) where you want to create the endpoint and click **Add**.

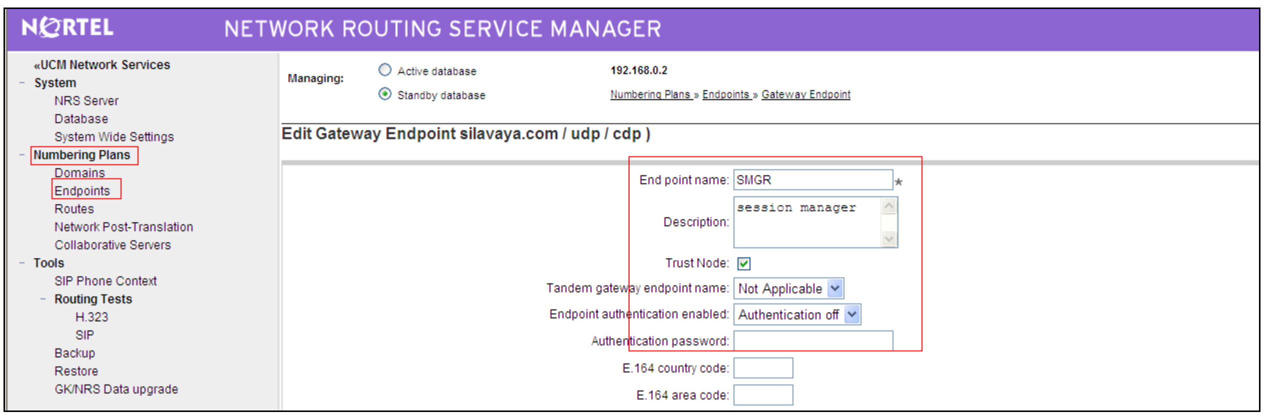


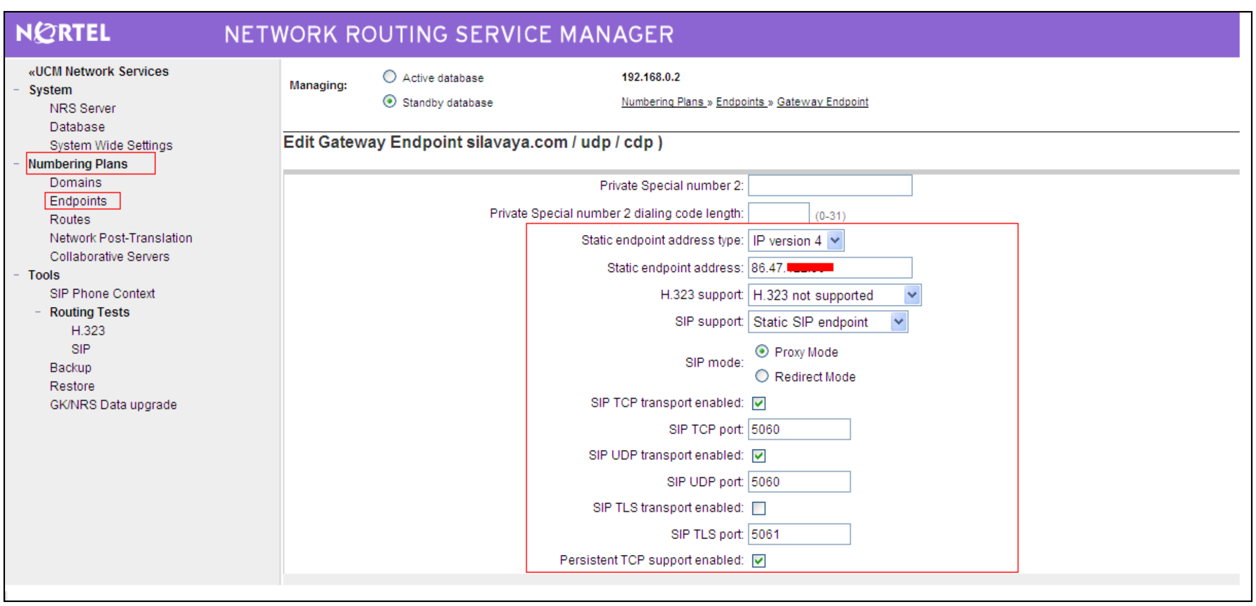
6.2.1. Configure Static Endpoint for Avaya Aura® Session Manager

This section shows how to add a static endpoint for the Session Manager. Enter the following values and use default values for remaining fields.

- **End point name:** **SMGR** was used in this configuration
- **Description:** **session manager** was used in this configuration
- **Static endpoint address:** This is the ip address of Session Manager SM100. This has been hidden for security purposes.
- **SIP support:** Set this to **Static SIP endpoint**
- **SIP mode:** Set this to **Proxy mode**
- **SIP TCP transport enabled:** Click the box to enable
- **SIP UDP transport enabled:** Click the box to enable

Click on **Save** (not shown). The two screenshots below show the Session Manager Endpoint configuration used for the testing.



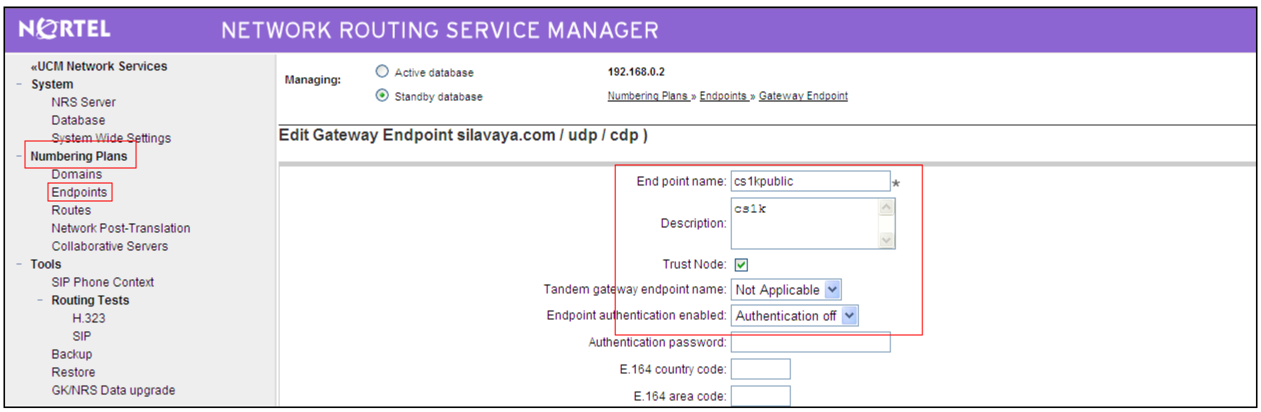


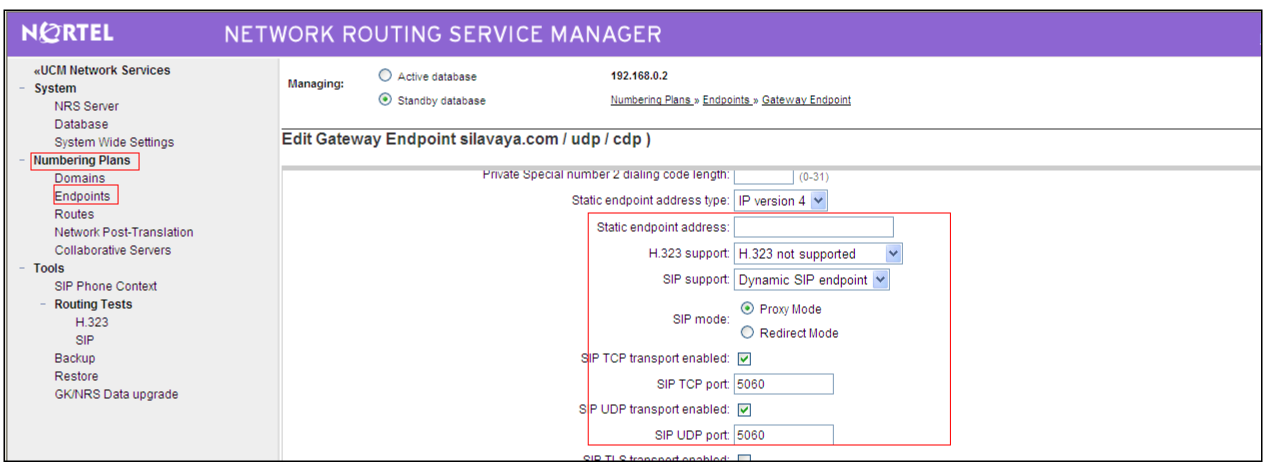
6.2.2. Configure Dynamic Endpoint for Avaya Communication Server 1000E

This section shows how to add a dynamic endpoint for the CS1000E, note that no IP address needs to be configured. Enter the following values and use default values for remaining fields.

- **End point name:** **cs1kpublic** was used in this configuration
- **Description:** **cs1k** was used in this configuration
- **SIP support:** Set this to **Dynamic SIP endpoint**
- **SIP mode:** Set this to **Proxy mode**
- **SIP TCP transport enabled:** Click the box to enable
- **SIP UDP transport enabled:** Click the box to enable

Click on **Save** (not shown).The two screenshots below show the CS1000E Endpoint configuration used for the testing.



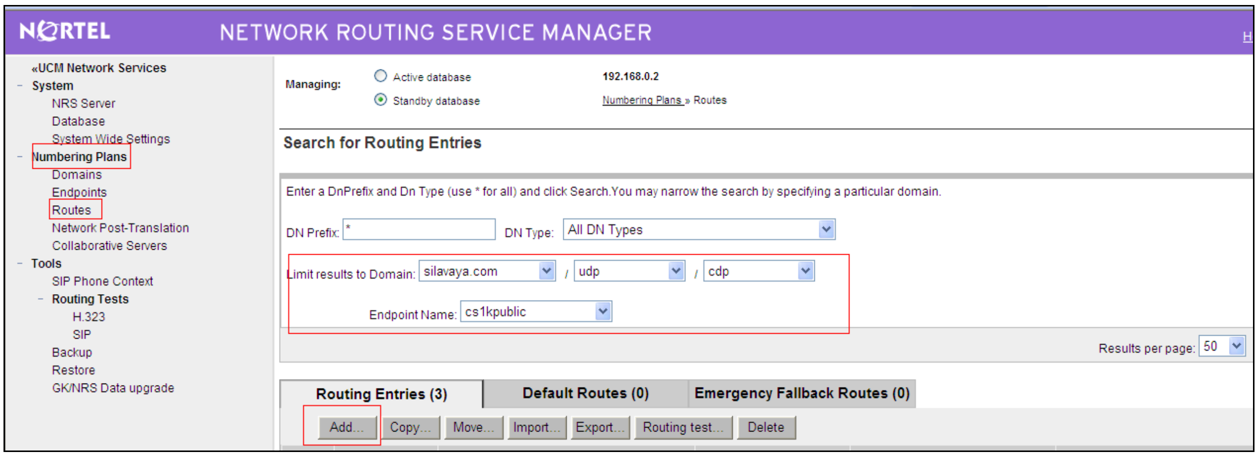


6.3. Define Routes

Routes need to be defined for each endpoint. Routes are the same as a dial pattern on the Session Manager and it is how the NRS routes out calls to an endpoint based on digits it receives.

6.3.1. Configure Route to the Avaya Communication Server 1000E

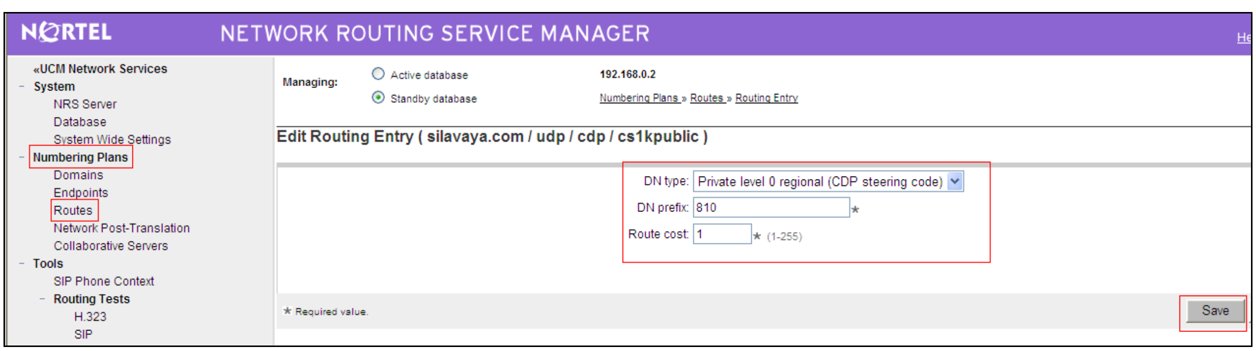
This section shows how to add routes to the CS1000E. This is for incoming calls from the PSTN to the CS1000E. Create a Route by clicking on **Numbering Plans → Routes**. Select the domain and sub-domains (e.g. **silavaya.com/udp/cdp**) and endpoint where you want to create the route (e.g. **cs1kpublic)** and click **Add**.



Enter the following values

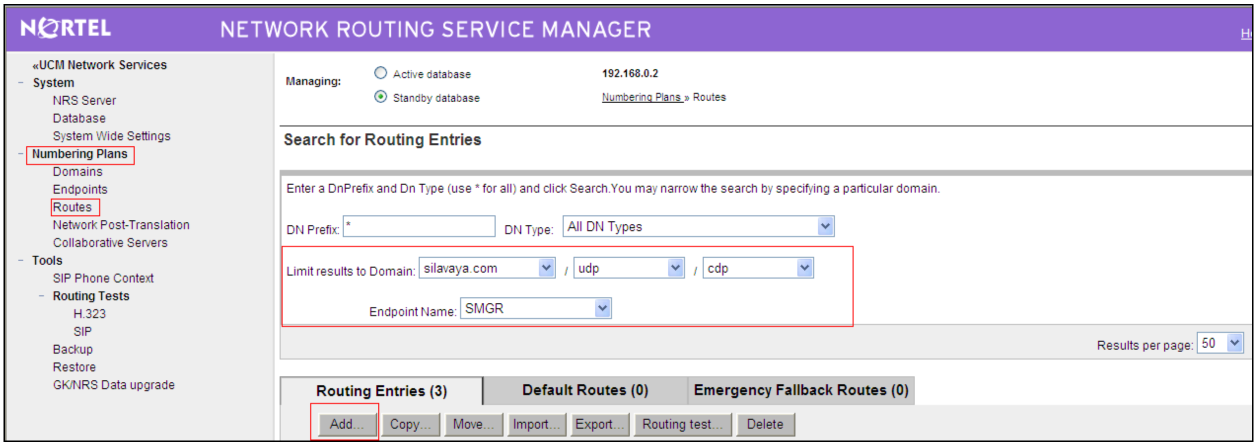
- **DN type:** Select **Private level 0 regional (CDP steering code)**. Only CDP dialing works for calls between the Session Manager and the NRS.
- **DN prefix:** **810** prefix matches the DN extensions on the test sets on the CS1000E. This also matches the last 4 digits of the DDI range given for the test
- **Route cost:** **1** is used as this is the only route available

Click on **Save.**



6.3.2. Configure Route to the Avaya Aura® Session Manager

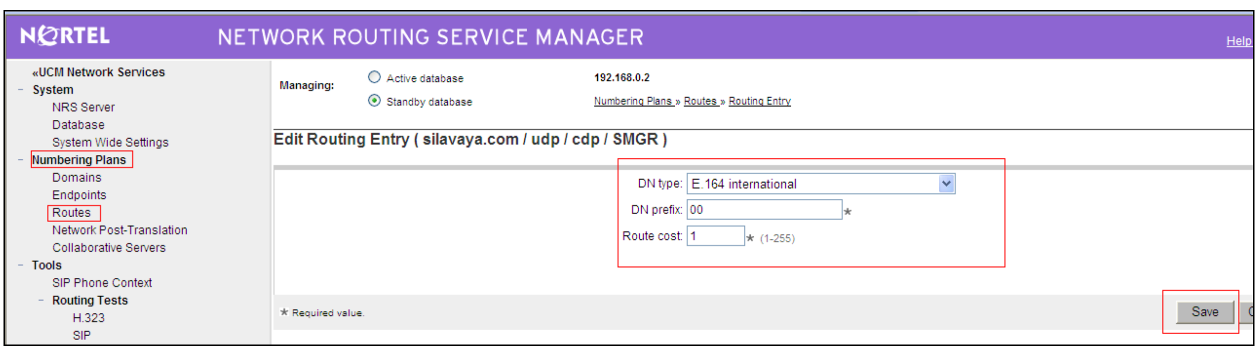
This section shows how to add routes to the Session Manager. This is outgoing calls from the CS1000E to the PSTN. Create a Route by clicking on **Numbering Plans → Routes**. Select the domain and sub-domains (e.g. **silavaya.com/udp/cdp**) and endpoint where you want to create the route (e.g. **SMGR)** and click **Add**.



Enter the following values; this is an example of E.164 International call.

- **DN type:** Select **E.164 international**
- **DN prefix:** **00** prefix matches the digits going to be dialed for an international call
- **Route cost:** **1** is used as this is the only route available

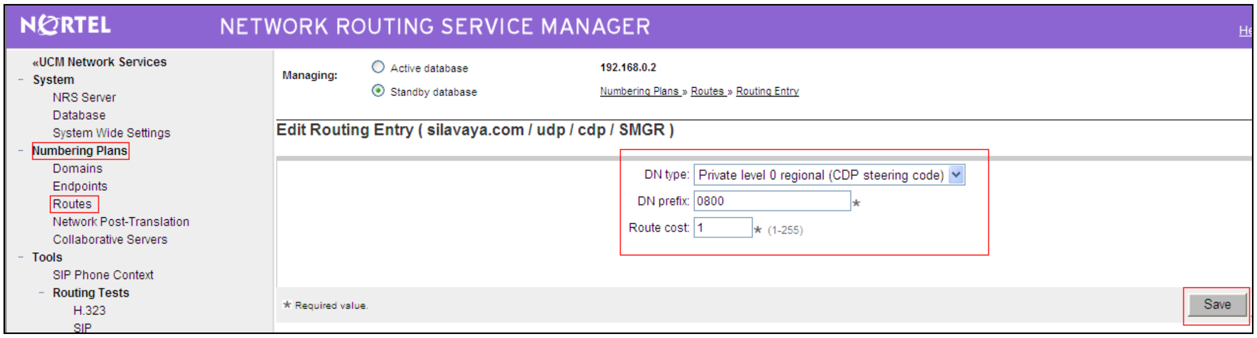
Click on **Save.**



This is an example of a local call using CDP dial plan, only CDP dialing worked for local calls.

- **DN type:** Select **Private level 0 regional (CDP steering code)**
- **DN prefix:** **0800** prefix matches the digits going to be dialed for a local call
- **Route cost:** **1** is used as this is the only route available

Click on **Save.**



7. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager to receive and route calls over the SIP trunk between Network Routing Server 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
- Configure the Adaptation Module
- Define SIP Entities
- Define Entity Links
- Define Routing Policies
- Define Dial Patterns

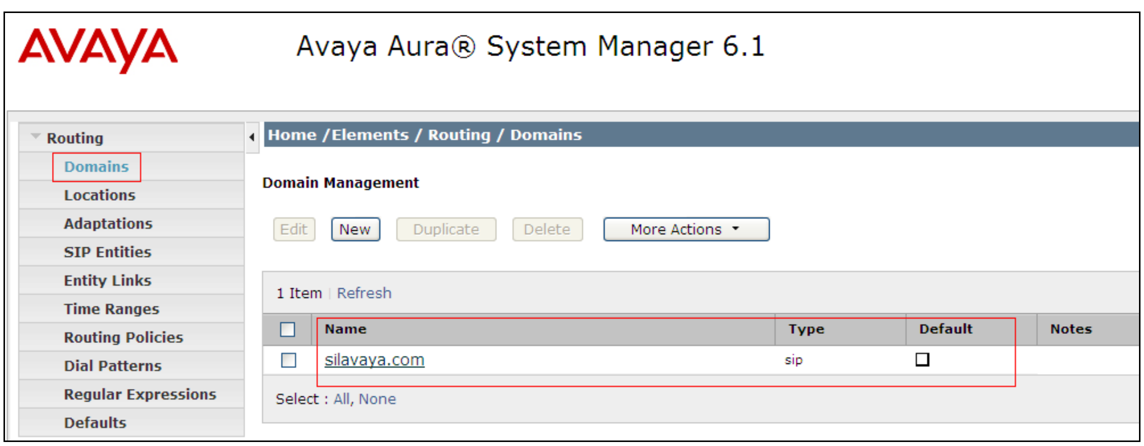
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials. Some administration screens have been abbreviated for clarity.

7.1. 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 5.4 and 6.1.** In the sample configuration, **silavaya.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.



7.2. Define Location

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. There were two locations defined in the sample configuration, one used by the NRS and one used by Vodafone's SBC.

7.2.1. Location for Avaya Network Routing Server

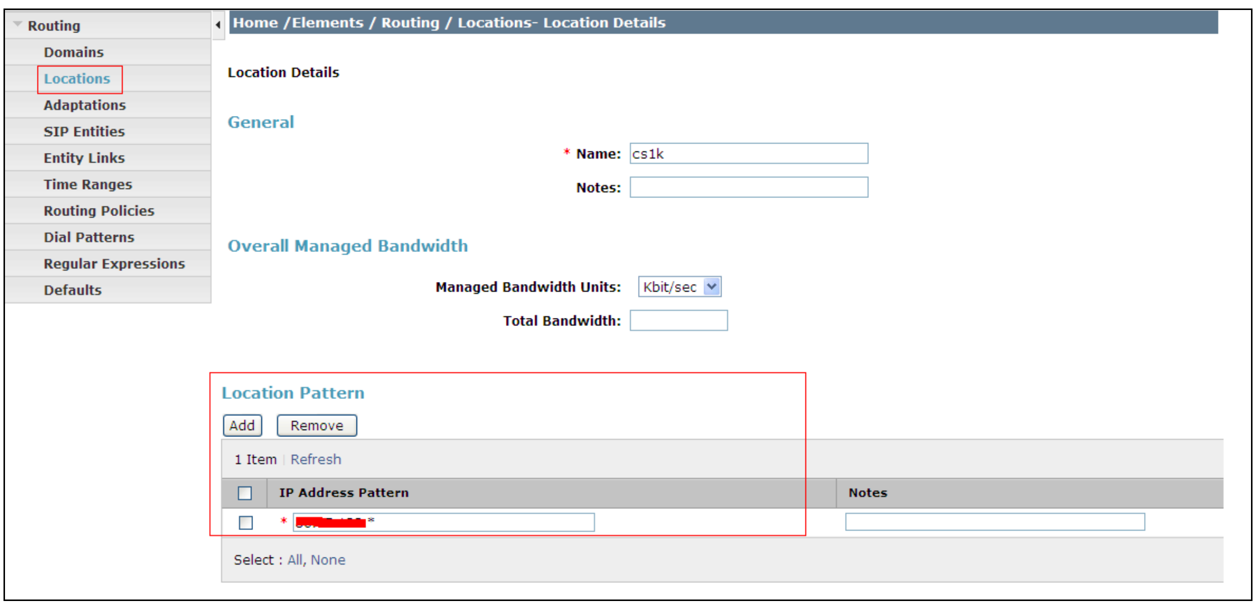
Expand **Elements → Routing** and select **Locations** from the left navigational menu. Click **New** (not shown)**.** In the **General** section**,** enter the following values and use default values for remaining fields**.**

- **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 for the CS1K Network Routing Server.

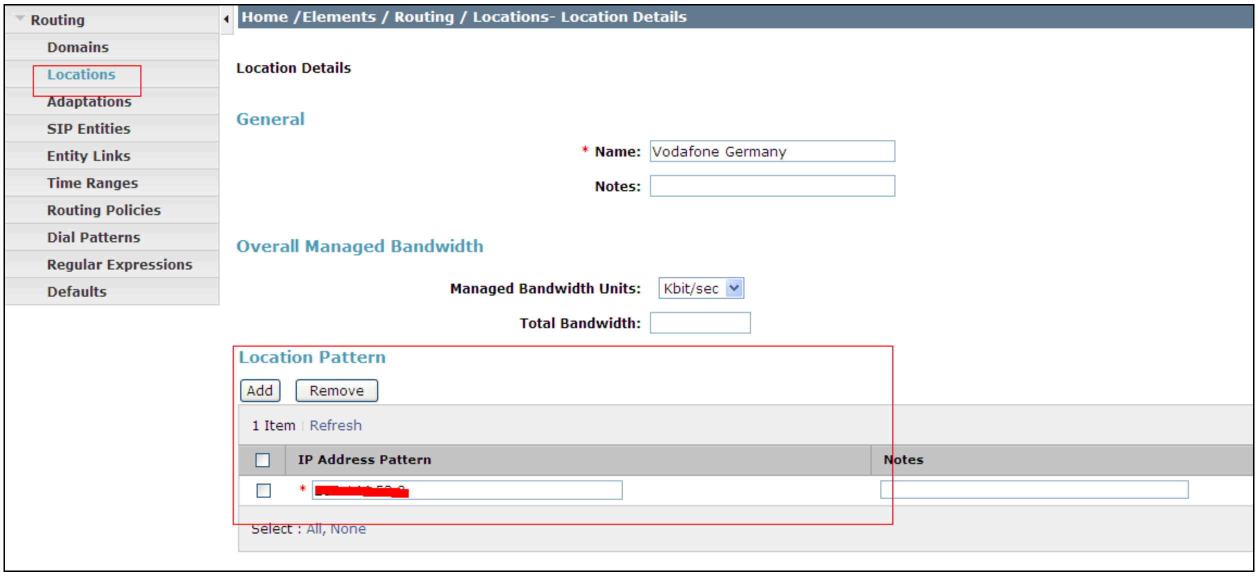
- **IP Address Pattern:** Enter the logical pattern used to identify the location for example **86.47.xx.***. For the sample configuration the pattern is hidden for security purposes because it is a public routable IP address
- **Notes:** Add a brief description [Optional]

Click **Commit** to save (not shown). The screenshot below shows the Location defined for Network Routing Server in the sample configuration.



7.2.2. Location for Vodafone SBC

Another location is defined for Vodafone SBC using the same process as outlined in **Section 7.2.1**. The screenshot below shows the Location defined in the sample configuration. The ip address pattern has been hidden for security purposes.



7.3. Configure Adaptation Module

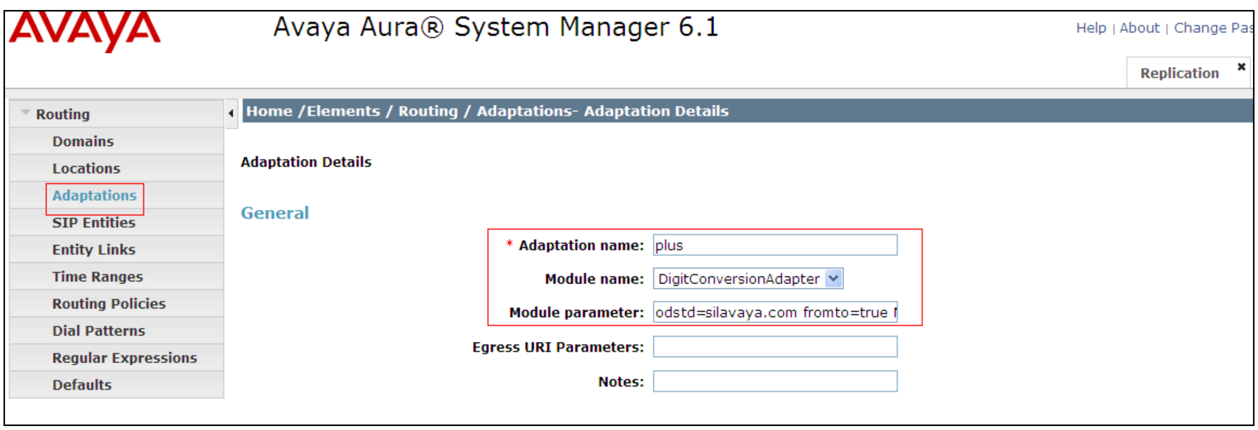
Session Manager is installed with a module called DigitConversionAdapter, which can convert digit strings in various message headers as well as host names in the Request-URI (Uniform Resource Identifier). In this configuration the adaptation is used by the Session Manager to ensure ingress messages have the hostname **silavaya.com** when they are sent to the NRS. Also the adaptation was used to strip MIME messages before being sent on to Vodafone. Vodafone does not support MIME. The adaptation is also used to manipulate the CLID number in the FROM and PAI headers.

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 **plus** 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 the modification parameters to be used. In this configuration the modification parameters used was **odstd=silavaya.com**
  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

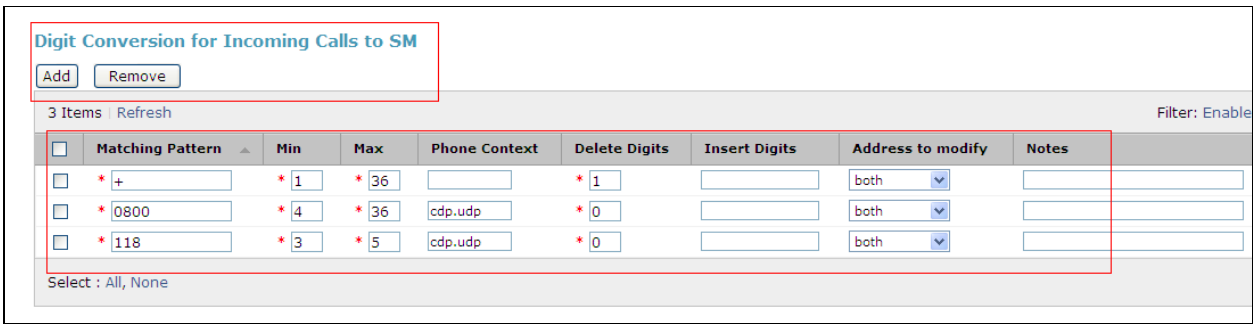
The whole string in module parameter is **odstd=silavaya.com fromto=true MIME=no**

odstd (or **OverrideDestinationDomain**) replaces 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 Vodafone Germany domain name in the SIP messages. The domain on the enterprise is silavaya.com so this Adaption Module changed incoming SIP messages destined for the enterprise to a recognised domain.



Scrolling down, in the **Digit Conversion for Incoming Calls to SM** section, click **Add** to configure entries for calls from CS1000E users to Vodafone.

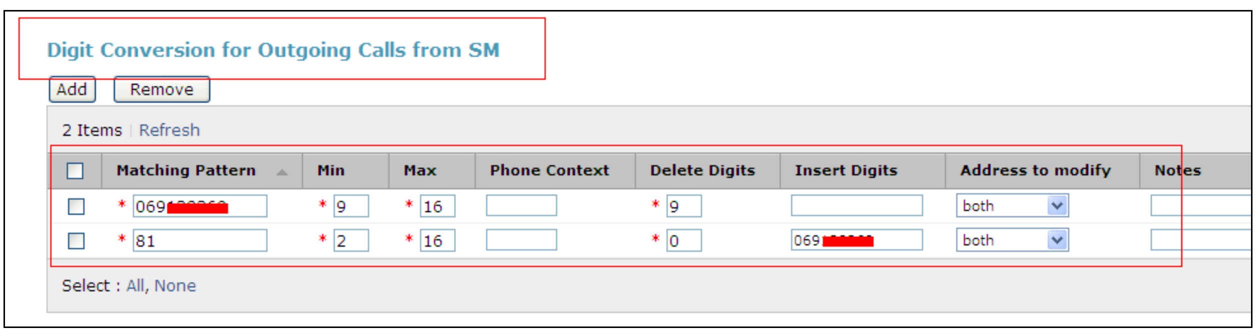
- **Matching Pattern:** Enter the digits for outbound calls to the PSTN. For international calls the NRS prefixes the dialled digits with a "+" before being sent on to the Session Manager as seen in row 1 of the screen shot below.
- **Min:** Enter minimum number of digits (e.g., **4**)
- **Max:** Enter maximum number of digits (e.g., **4**)
- **Phone Context:** Enter value if needed. For local calls CDP dialing is used on the CS1000E/NRS so **cdp.udp** is used, see **Section 6.3.2** for more information. This can be seen in rows 2 and 3 in the screen shot below.
- **Delete Digits:** Enter **0**, unless digits should be removed from dialled number before routing on by Session Manager. For international calls the "+" needs removing before passing on to the PSTN so **1** digit is deleted
- **Insert Digits:** Enter digits that need to be inserted
- **Address to modify:** Select **both**



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:** Enter the digits received from the PSTN to the CS1000E. This is the DDI range minus the last 4 digits. The last 4 digits are the DN number of the test sets used during the testing; this is seen in row 1 of the screen shot below, starting **069xxxxxx**. This functionality is used instead of overlay 49 on the CS1000E that does incoming digit conversion.
  Row 2 however is concerned with CLID manipulation for outbound calls to the PSTN. The test sets have a 4 digit DN starting with **81xx**. This adaptation manipulates the from and PAI headers in the SIP INVITE and inserts the correct DDI number using the **Insert** Digit field to the front of the 4 digit extension so a valid DDI number is presented to Vodafone
- **Min:** Enter minimum number of digits (e.g., **9**)
- **Max:** Enter maximum number of digits (e.g., **16**)
- **Phone Context:** Enter value if needed
- **Delete Digits:** Enter **9** for calls that are going to the CS1000E. This will delete the first 9 digits of the DDI number and leave the last 4 digits which are equal to the DN of the test sets on the CS1000E
- **Insert Digits:** Enter digits that need to be inserted – the example on row 2, in order to send a correct CLID these digits need to be inserted before the 4 digit DN of the test sets, in this case **069xxxxxx** plus **81xx**
- **Address to modify:** Select **both**

Click **Commit** to save (not shown).



7.4. Define SIP Entities

A SIP Entity must be added for Network Routing Server and also for Vodafone 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**).**

7.4.1. SIP Entity for Avaya Network Routing Server

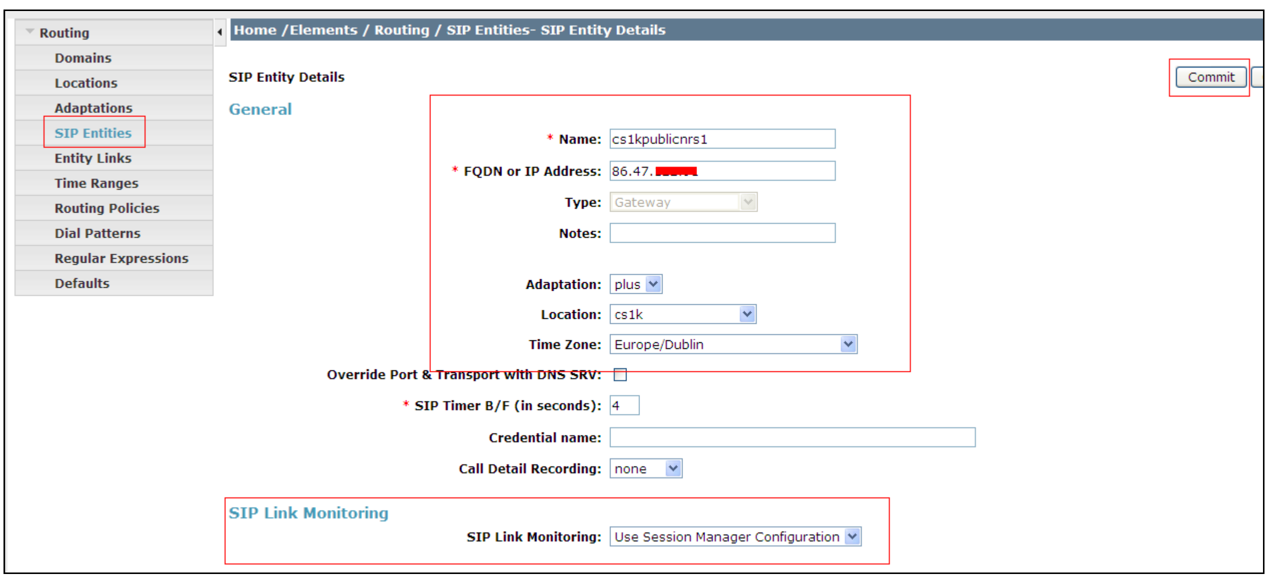
In the **General** section enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter TLAN IP address of Communication Server 1000E Node identified in **Section 5.4.** This has been partially hidden for security purposes
- **Type:** Select **gateway**
- **Notes:** Enter a brief description [Optional]
- **Adaptations:** DigitConversionAdapter defined in **Section 7.3**
- **Location:** Select the Location defined for Communication Server 1000E in **Section 7.2.1**

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 Network Routing Server in the sample configuration.



7.4.2. SIP Entity for Vodafone SBC

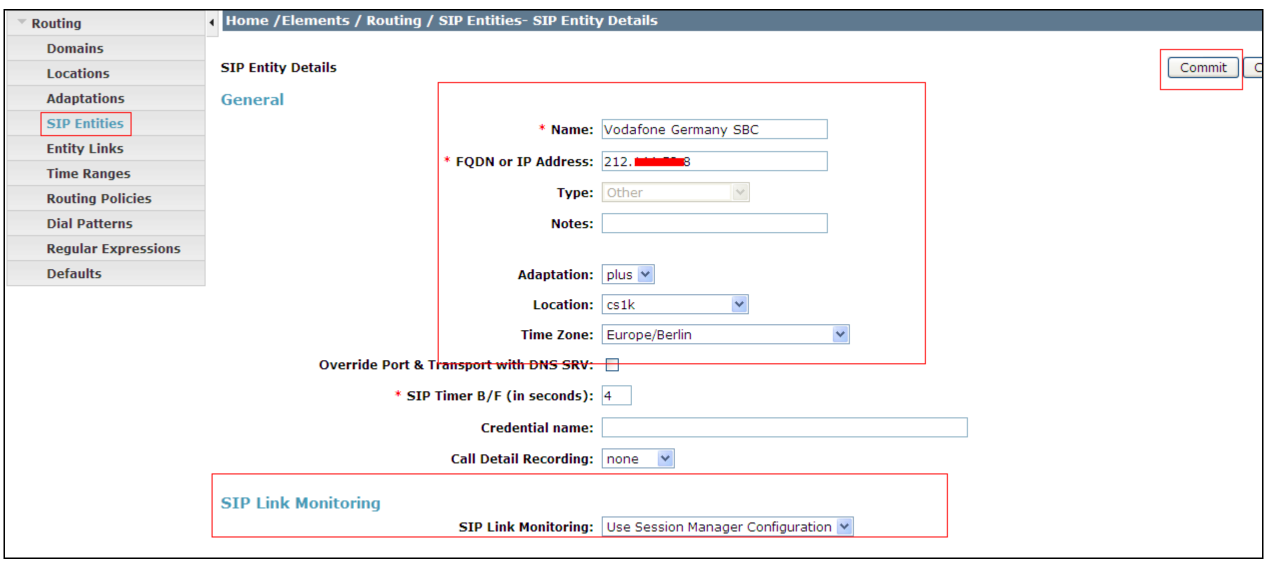
In the **General** section enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter the public interface IP address of Vodafone's SBC. This has been partially hidden for security purposes
- **Type:** Select **other**
- **Notes:** Enter a brief description [Optional]
- **Adaptations:** DigitConversionAdapter defined in **Section 7.3**
- **Location:** Select the Location defined for Communication Server 1000E in **Section 7.2.2**

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 Vodafone's SBC in the sample configuration.



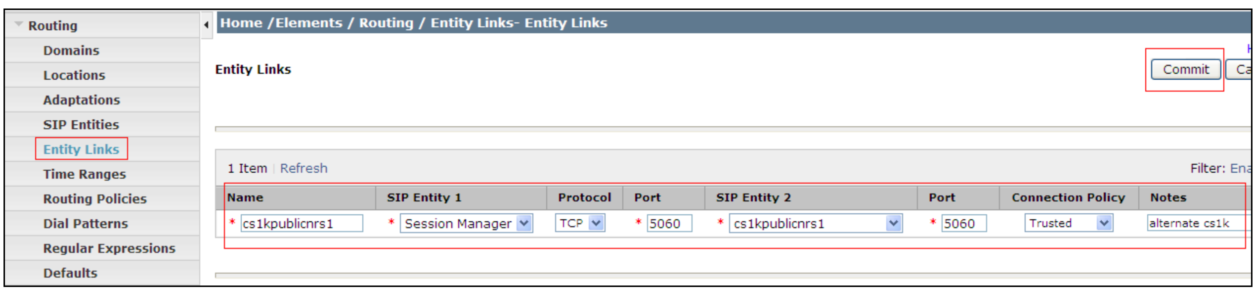
A SIP Entity link must also be defined for your Session Manager but that is not shown in this document.

7.5. Define Entity links

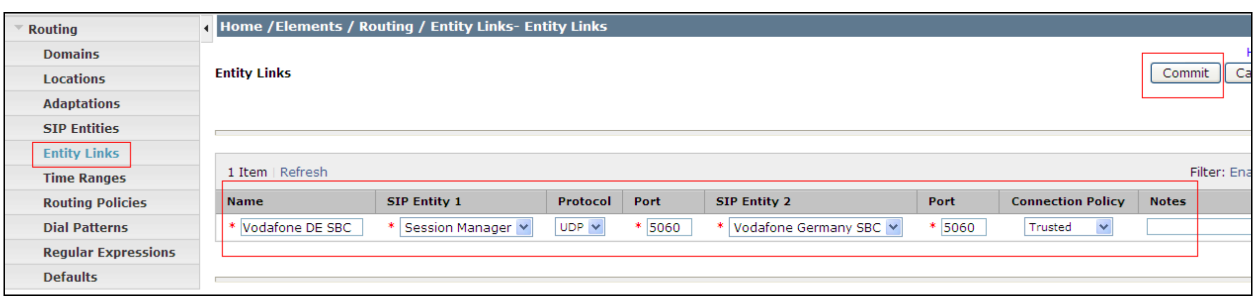
The SIP trunk between the Session Manager and the Avaya Network Routing Server is described by an Entity link. The same is needed between the Session Manager and Vodafone's SBC. Expand **Elements → Routing** and select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the 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 Network Routing Server/Vodafone SBC in **Section 7.4**
- **Protocol** After selecting both SIP Entities, select **TCP or UDP** as the required protocol. The NRS uses TCP and Vodafone SBC uses UDP
- **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 Network Routing Server.



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



7.6. Define Routing Policy

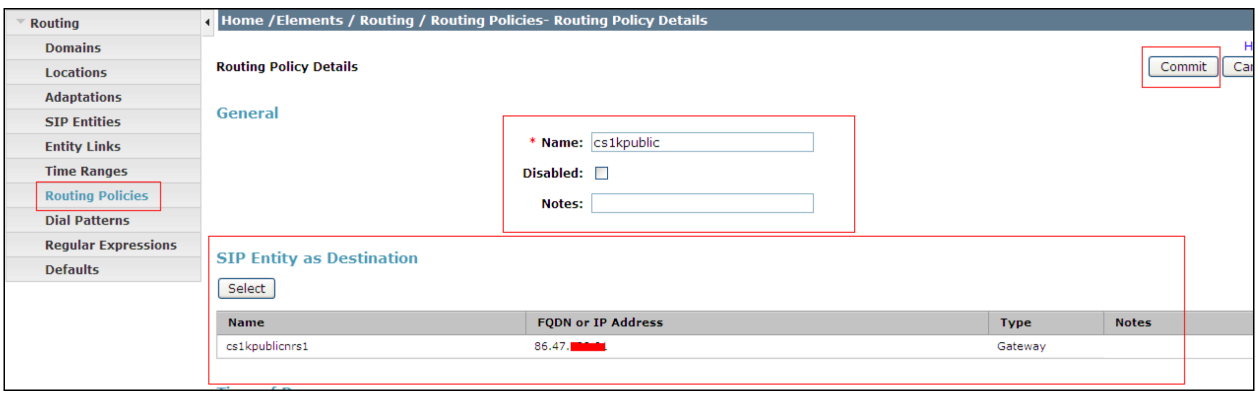
Routing policies describe the conditions under which calls will be routed to CS1000E from either SIP endpoint registered to Session Manager or from other telephony system. It also describers the routing polices for which calls will be routed to Vodafone's SIP network. To add a routing policy, expand **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]

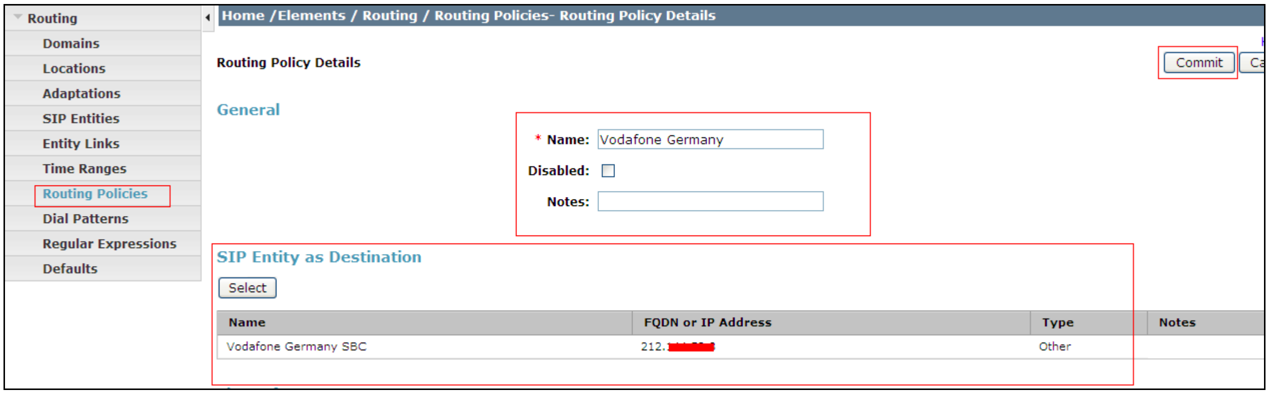
In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). For routing policy to the NRS, select the SIP Entity associated with the NRS defined in **Section 7.4** and click **Select.** The selected SIP Entity displays on the **Routing Policy Details** 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 the Network Routing Server:



For routing policy Vodafone’s SIP network, select the SIP Entity associated with Vodafone’s SBC defined in **Section 7.4** 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.



7.7. Define Dial Pattern

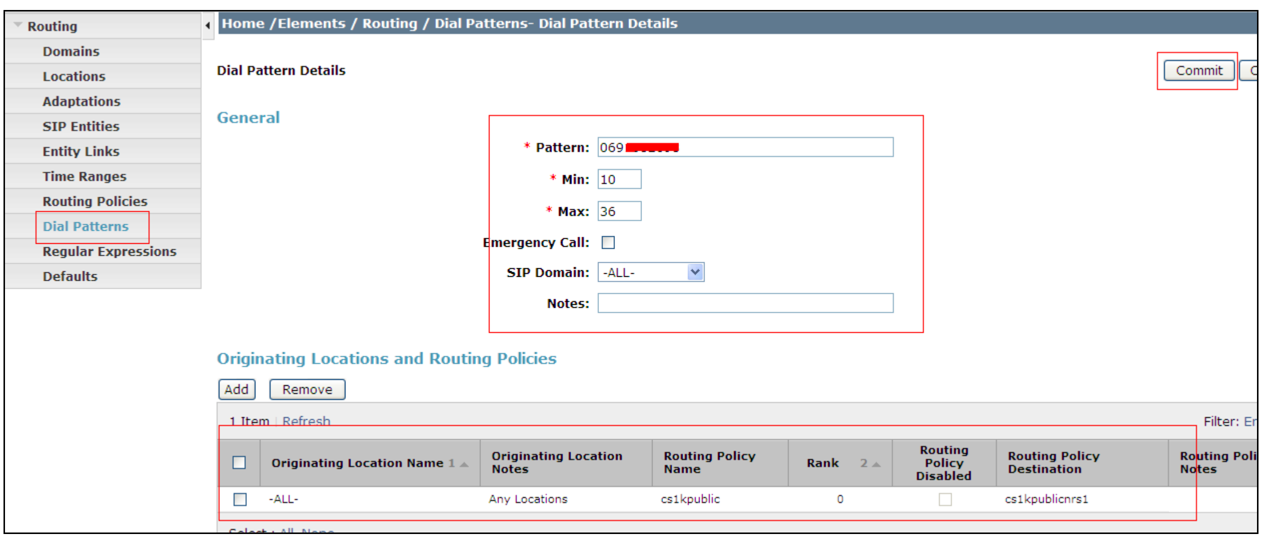
Dial patterns are used to route calls to appropriate SIP Entities. In the sample configuration, since the DDI range given for the testing had numbers that start with **069xxxxxx**, these will be routed to the Network Routing Server and then on to the Communication Server 1000E where they would be terminated on test sets. Alternately calls that are originated on the Communication Server 1000E that start with digits **00** will be routed to Vodafone's SBC, 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 Network Routing Server
- **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]

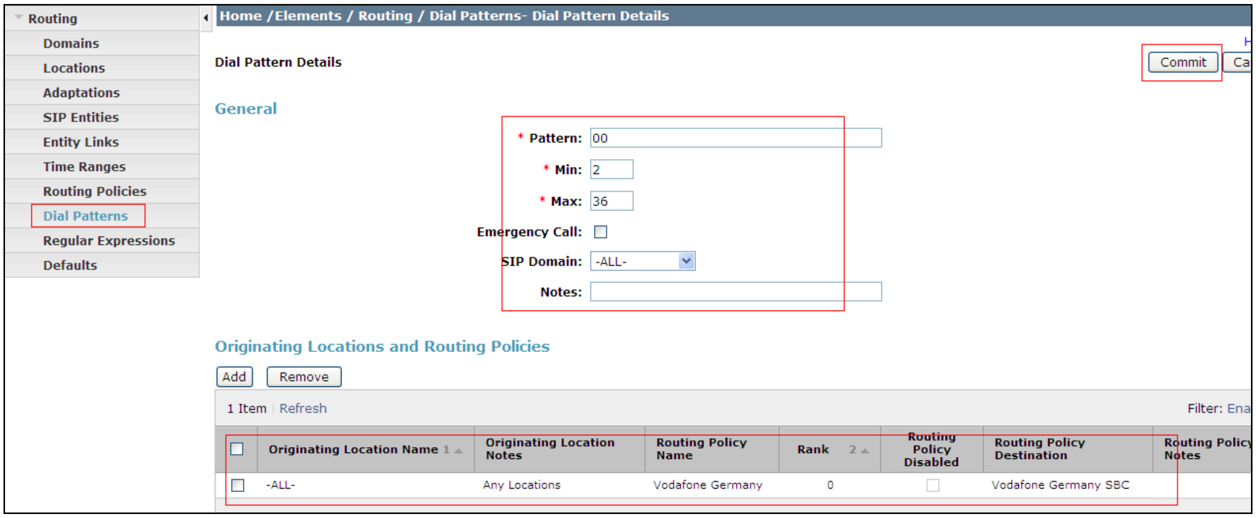
In the **Originating Locations and Routing Policies** section, click **Add.** The **Originating Locations and Routing Policy List** page opens (not shown).

- **Originating Locations** Select **All**
- **Routing Policies** Select the Routing Policy defined for Network Routing Server in **Section 7.6**

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 Network Routing Server.



Repeat the above steps to add the dial Pattern to Vodafone's SBC, select the routing policy defined for Vodafone's SBC in **Section 7.6**. The following screenshot shows the Routing Policy for Vodafone's SBC.



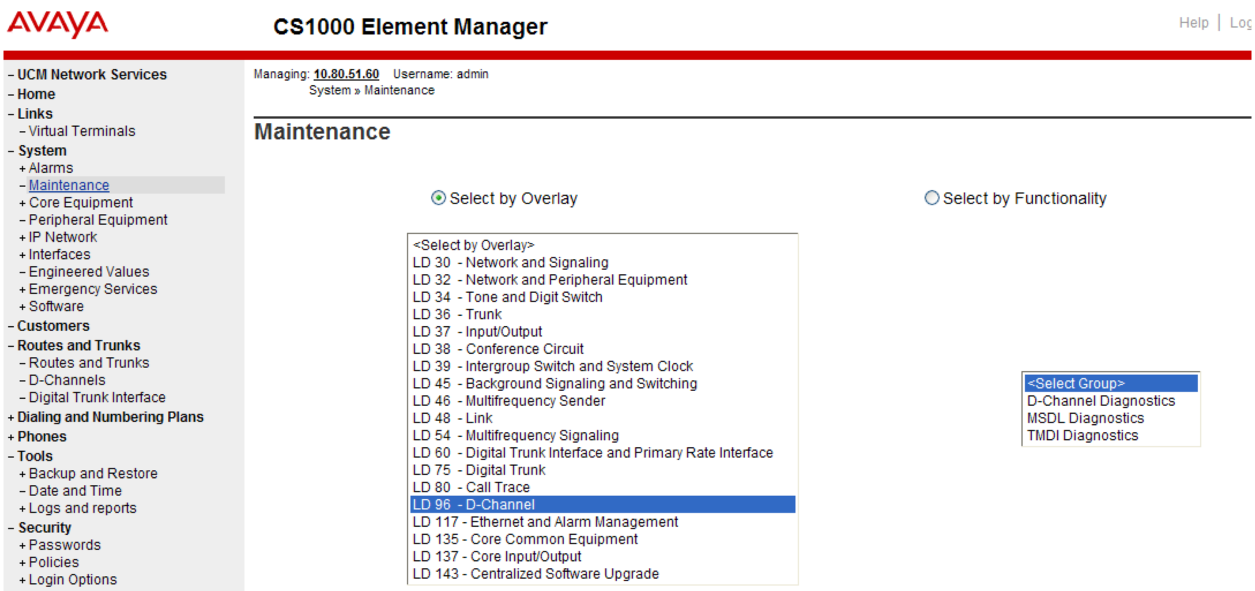
8. Service Provider Configuration

The configuration of Vodafone Germany'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 Vodafone Germany's equipment and system configuration please contact an authorised Vodafone Germany representative.

9. Verification

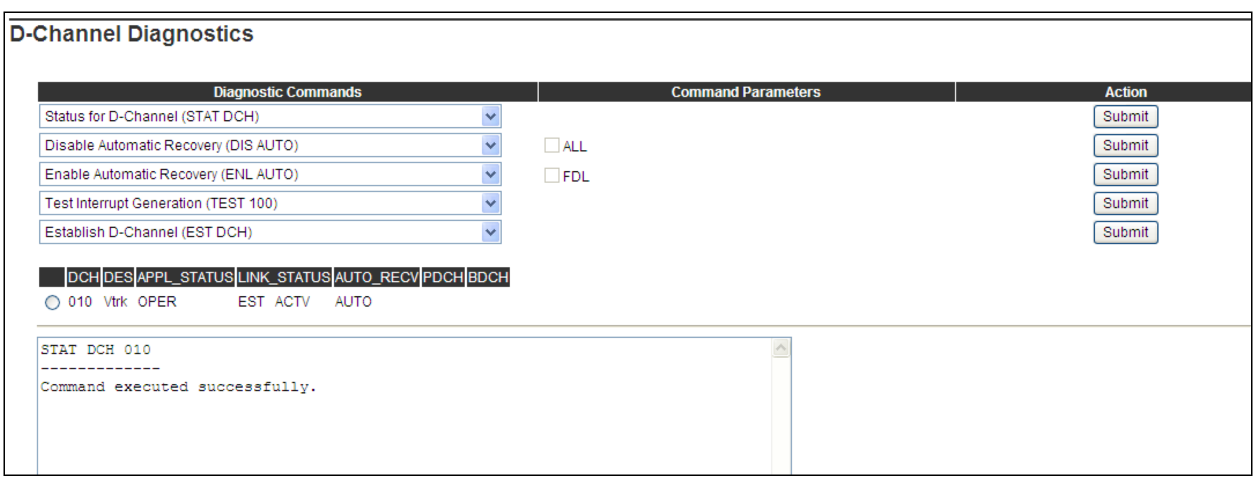
9.1. Verify Avaya Communication Server 1000E Operational Status

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:

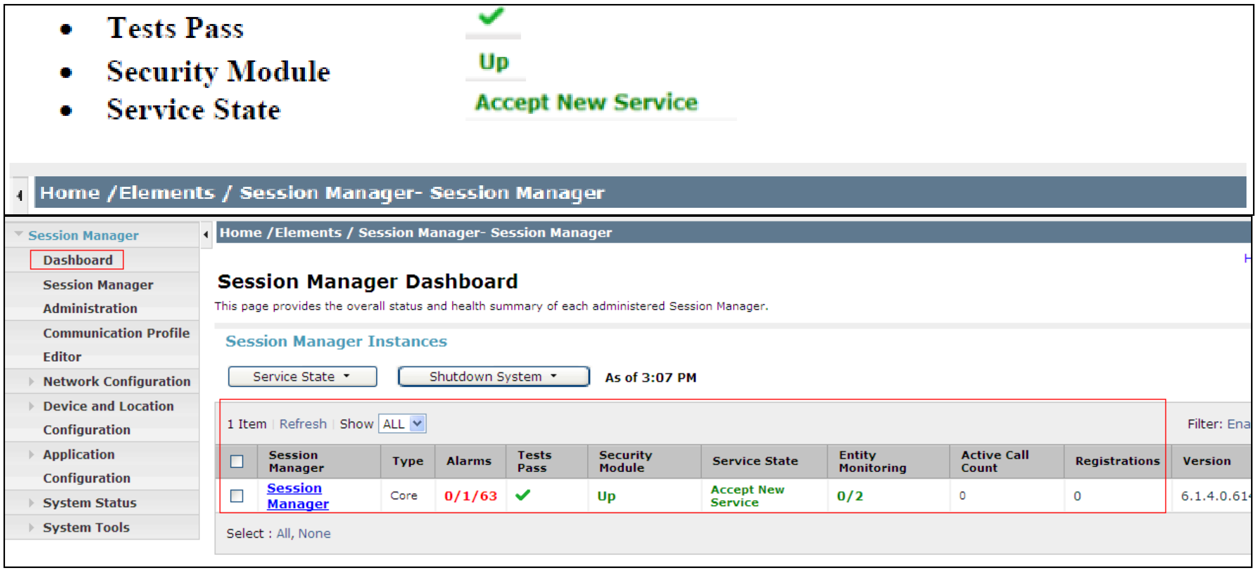
- **Appl_Status** Verify status is **OPER**
- **Link_Status** Verify status is **EST ACTV**



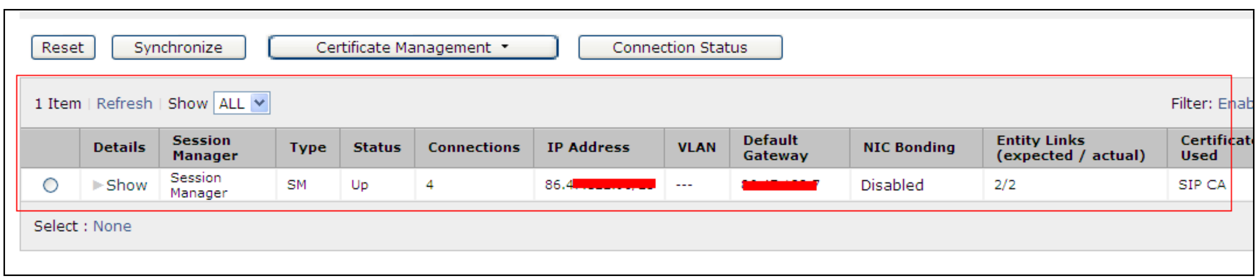
9.2. Verify Avaya Aura® Session Manager Operational Status

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.

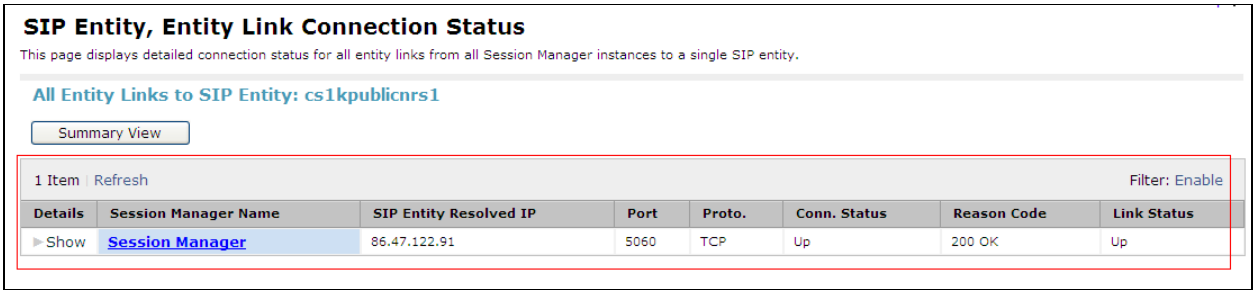


Navigate to **Elements → Session Manager → System Status → Security Module Status** (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays **Up** as shown below.



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 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: cs1kpublicnrs1** 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.

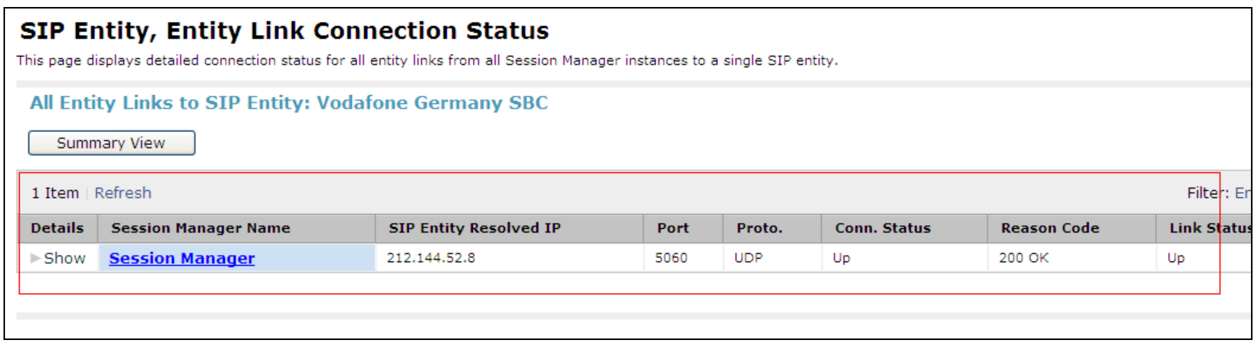
All Entity Links to SIP Entity: cs1kpublicnrs1

Summary View

1 Item | Refresh   Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶Show	Session Manager	86.47.122.91	5060	TCP	Up	200 OK	Up

Verify the SIP link is up between the Session Manager and Vodafone's SBC by going through the same process as outlined above but selecting the SIP Entity for Vodafone SBC 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: Vodafone Germany SBC

Summary View

1 Item | Refresh   Filter: En

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶Show	Session Manager	212.144.52.8	5060	UDP	Up	200 OK	Up

10. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya Communication Server 1000E, Avaya Network Routing Server and Avaya Aura® Session Manager to Vodafone Germany's SIP Service. The testing was successfully performed; refer to **Section 2.2** for more details.

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

```
11/11/11 15:44:22
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 00 Q  +
IDLE_SET_DISPLAY NORTEL
DepList 1: core Issue: 01(created: 2011-08-16 16:46:54 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2011-10-10 13:56:09(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-09-02 08:33:26(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE


LOADWARE VERSION: PSWV 100

INSTALLED LOADWARE PEPS : 0

ENABLED PLUGINS : 1

PLUGIN    STATUS      PRS/CR_NUM    MPLR_NUM      DESCRIPTION
-------------------------------------------------------------
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 00 Q +
DepList 1: core Issue: 01 (created: 2011-08-16 16:46:54 (est))

IN-SERVICE PEPS
PAT# CR #           PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi00855276     ISS1:1OF1       p29903_1  07/11/2011  p29903_1.cpl   NO
001  wi00687630     ISS1:1OF1       p31017_1  07/11/2011  p31017_1.cpl   NO
002  wi00855050     ISS1:1OF1       p30731_1  07/11/2011  p30731_1.cpl   YES
003  wi00848801     ISS1:1OF1       p30336_1  07/11/2011  p30336_1.cpl   YES
004  WI00853745     ISS1:1OF1       p29841_1  07/11/2011  p29841_1.cpl   YES
005  WI00844778     ISS1:1OF1       p30641_1  07/11/2011  p30641_1.cpl   NO
006  wi00890671     ISS1:1OF1       p31051_1  07/11/2011  p31051_1.cpl   YES
007  wi00853769     ISS1:1OF1       p30892_1  07/11/2011  p30892_1.cpl   YES
008  WI00853186     ISS1:1OF1       p30625_1  07/11/2011  p30625_1.cpl   NO
009  wi00854255     ISS1:1OF1       p30124_1  07/11/2011  p30124_1.cpl   NO
010  WI00853769     ISS1:1OF1       p30894_1  07/11/2011  p30894_1.cpl   YES
011  wi00883601     ISS1:1OF1       p30973_1  07/11/2011  p30973_1.cpl   NO
012  wi00848515     ISS1:1OF1       p30677_1  07/11/2011  p30677_1.cpl   NO
013  wi00852689     ISS1:1OF1       p29842_1  07/11/2011  p29842_1.cpl   NO
014  wi00641671     ISS1:1OF1       p29744_1  07/11/2011  p29744_1.cpl   NO
015  wi00869468     ISS1:1OF1       p30856_1  07/11/2011  p30856_1.cpl   NO
016  Q02161860      ISS2:1OF1       p30263_2  07/11/2011  p30263_2.cpl   NO
017  wi00868063     ISS1:1OF1       p30848_1  07/11/2011  p30848_1.cpl   NO
018  wi00825672     ISS1:1OF1       p30468_1  07/11/2011  p30468_1.cpl   NO
019  Q02159545      ISS1:1OF1       p30277_1  07/11/2011  p30277_1.cpl   YES
```

020	wi00891255	ISS1:1OF1	p30399_1	07/11/2011	p30399_1.cpl	NO
021	wi00840589	ISS2:1OF1	p30767_2	07/11/2011	p30767_2.cpl	NO
022	wi00686980	ISS1:1OF1	p30706_1	07/11/2011	p30706_1.cpl	NO
023	wi00853798	ISS1:1OF1	p30131_1	07/11/2011	p30131_1.cpl	NO
024	wi00862909	ISS1:1OF1	p30809_1	07/11/2011	p30809_1.cpl	NO
025	Q02165164	ISS1:1OF1	p30304_1	07/11/2011	p30304_1.cpl	NO
026	wi00874210	ISS1:1 OF 1	p30880_1	07/11/2011	p30880_1.cpl	NO
027	WI00836290	ISS1:1OF1	p30554_1	07/11/2011	p30554_1.cpl	NO
028	wi00688204	ISS1:1OF1	p30197_1	07/11/2011	p30197_1.cpl	NO
029	wi00857960	ISS1:1OF1	p30768_1	07/11/2011	p30768_1.cpl	NO
030	Q02156594	ISS1:1OF1	p30276_1	07/11/2011	p30276_1.cpl	YES
031	wi00853753	ISS1:1OF1	p30064_1	07/11/2011	p30064_1.cpl	NO
032	wi00847002	ISS1:1OF1	p30656_1	07/11/2011	p30656_1.cpl	NO
033	wi00891007	ISS1:1OF1	p31058_1	07/11/2011	p31058_1.cpl	NO
034	wi00816794	ISS1:1OF1	p30443_1	07/11/2011	p30443_1.cpl	NO
035	WI00853478	ISS1:1OF1	p30306_1	07/11/2011	p30306_1.cpl	NO
036	wi00853431	ISS1:1OF1	p29935_1	07/11/2011	p29935_1.cpl	NO
037	wi00861414	ISS1:1OF1	p30791_1	07/11/2011	p30791_1.cpl	NO
038	wi00853769	ISS1:1OF1	p30896_1	07/11/2011	p30896_1.cpl	YES
039	wi00856984	ISS1:1OF1	p17588_1	07/11/2011	p17588_1.cpl	NO
040	wi00641909	ISS1:1OF1	p30004_1	07/11/2011	p30004_1.cpl	NO
041	wi00895312	ISS1:1OF1	p31107_1	07/11/2011	p31107_1.cpl	NO
042	wi00852510	ISS1:1OF1	p30357_1	07/11/2011	p30357_1.cpl	NO
043	wi00862916	ISS1:1OF1	p30807_1	07/11/2011	p30807_1.cpl	NO
044	wi00884895	ISS1:1OF1	p30942_1	07/11/2011	p30942_1.cpl	NO
045	wi00877009	ISS1:1OF1	p31115_1	07/11/2011	p31115_1.cpl	NO
046	wi00688114	ISS1:1OF1	p30319_1	07/11/2011	p30319_1.cpl	NO
047	Q02149076-01	ISS1:1OF1	p30206_1	07/11/2011	p30206_1.cpl	NO
048	wi00893211	ISS1:1OF1	p30867_1	07/11/2011	p30867_1.cpl	NO
049	Q02145107-02	ISS1:1OF1	p30126_1	07/11/2011	p30126_1.cpl	NO
050	wi00869293	ISS1:1OF1	p30963_1	07/11/2011	p30963_1.cpl	NO
051	wi00894262	ISS1:1OF1	p31091_1	07/11/2011	p31091_1.cpl	NO
052	WI00853769	ISS1:1OF1	p30895_1	07/11/2011	p30895_1.cpl	YES
053	wi00647104	ISS2:1of1	p29747_2	07/11/2011	p29747_2.cpl	NO
054	wi00853750	ISS1:1OF1	p29938_1	07/11/2011	p29938_1.cpl	NO
055	wi00878934	ISS1:1OF1	p31077_1	07/11/2011	p31077_1.cpl	NO
056	wi00853388	ISS1:1OF1	p30065_1	07/11/2011	p30065_1.cpl	NO
057	wi00876852	ISS1:1OF1	p30952_1	07/11/2011	p30952_1.cpl	NO
058	wi00876798	iss2:1of1	p31039_2	07/11/2011	p31039_2.cpl	NO
059	Wi00687946	ISS1:1OF1	p30123_1	07/11/2011	p30123_1.cpl	NO
060	wi00839793	ISS1:1OF1	p28647_1	07/11/2011	p28647_1.cpl	NO
061	wi00892074	ISS1:1OF1	p29645_1	07/11/2011	p29645_1.cpl	NO
062	wi00853769	ISS1:1OF1	p30898_1	07/11/2011	p30898_1.cpl	YES
063	wi00839916	ISS1:1OF1	p30593_1	07/11/2011	p30593_1.cpl	NO
064	wi00688477	ISS1:1OF1	p29732_1	07/11/2011	p29732_1.cpl	NO
065	Wi00623072	ISS1:1OF1	p29992_1	07/11/2011	p29992_1.cpl	NO
066	wi00826342	ISS2:1OF1	p30471_2	07/11/2011	p30471_2.cpl	NO
067	wi00879820	ISS1:1OF1	p30719_1	07/11/2011	p30719_1.cpl	NO
068	wi00903367	ISS1:1OF1	p31165_1	07/11/2011	p31165_1.cpl	NO
069	Q02152936-01	ISS1:1OF1	p30249_1	07/11/2011	p30249_1.cpl	NO
070	wi00886633	ISS1:1OF1	p31012_1	07/11/2011	p31012_1.cpl	NO
071	wi00845667	ISS1:1OF1	p30676_1	07/11/2011	p30676_1.cpl	NO
072	wi00881957	ISS1:1OF1	p30982_1	07/11/2011	p30982_1.cpl	NO
073	wi00824288	ISS1:1OF1	p30461_1	07/11/2011	p30461_1.cpl	NO
074	Q02157114	ISS1:1OF1	p30251_1	07/11/2011	p30251_1.cpl	NO
075	WI00851975	ISS1:1OF1	p30312_1	07/11/2011	p30312_1.cpl	NO
076	wi00887742	ISS2:1OF1	p31026_2	07/11/2011	p31026_2.cpl	NO
077	wi00894239	ISS1:1OF1	p31087_1	07/11/2011	p31087_1.cpl	NO
078	wi00827512	ISS1:1OF1	p30479_1	07/11/2011	p30479_1.cpl	NO
079	WI00853769	ISS1:1OF1	p30890_1	07/11/2011	p30890_1.cpl	YES
080	wi00828961	ISS2:1OF1	p30492_2	07/11/2011	p30492_2.cpl	NO
081	wi00865152	ISS1:1OF1	p30826_1	07/11/2011	p30826_1.cpl	NO
082	wi00604003	ISS1:1OF1	p29726_1	07/11/2011	p29726_1.cpl	NO
083	wi00853769	ISS1:1OF1	p30893_1	07/11/2011	p30893_1.cpl	YES
084	Q02159250-01	ISS1:1OF1	p30280_1	07/11/2011	p30280_1.cpl	NO
085	wi00853778	ISS1:1OF1	p30210_1	07/11/2011	p30210_1.cpl	NO
086	wi00860278	ISS1:1OF1	p30789_1	07/11/2011	p30789_1.cpl	NO
087	Q02150073-01	ISS1:1OF1	p30160_1	07/11/2011	p30160_1.cpl	NO
088	wi00686889	ISS3:1OF1	p30074_3	07/11/2011	p30074_3.cpl	NO
089	wi00836181	ISS1:1OF1	p30450_1	07/11/2011	p30450_1.cpl	NO

```
090  Q02158718-01    ISS1:1OF1       p30311_1  07/11/2011  p30311_1.cpl   NO
091  wi00852325      ISS1:1OF1       p30167_1  07/11/2011  p30167_1.cpl   NO
092  wi00869693      ISS1:1OF1       p30654_1  07/11/2011  p30654_1.cpl   NO
093  wi00837793      ISS1:1OF1       p30573_1  07/11/2011  p30573_1.cpl   NO
094  wi00898326      ISS1:1OF1       p31136_1  07/11/2011  p31136_1.cpl   NO
095  Q02152254       ISS1:1OF1       p30271_1  07/11/2011  p30271_1.cpl   YES
096  wi00852317      ISS1:1OF1       p30176_1  07/11/2011  p30176_1.cpl   NO
097  wi00826074      ISS1:1OF1       p30452_1  07/11/2011  p30452_1.cpl   NO
098  Q02143641-01    ISS1:1OF1       p30159_1  07/11/2011  p30159_1.cpl   NO
099  Q02162391       ISS1:1OF1       p30272_1  07/11/2011  p30272_1.cpl   NO
100  wi00853837      ISS1:1OF1       p30172_1  07/11/2011  p30172_1.cpl   NO
101  wi00848697      ISS1:1OF1       p30621_1  07/11/2011  p30621_1.cpl   NO
102  WI00836333      ISS1:1OF1       p30481_1  07/11/2011  p30481_1.cpl   NO
103  wi00686977      ISS1:1OF1       p30223_1  07/11/2011  p30223_1.cpl   NO
104  wi00688037      ISS2:1OF1       p29376_2  07/11/2011  p29376_2.cpl   NO
105  wi00843569      ISS1:1OF1       p30627_1  07/11/2011  p30627_1.cpl   NO
106  wi00861072      ISS1:1OF1       p30787_1  07/11/2011  p30787_1.cpl   NO
107  wi00880384      ISS1:1OF1       p30977_1  07/11/2011  p30977_1.cpl   NO
108  wi00856261      ISS1:1OF1       p30751_1  07/11/2011  p30751_1.cpl   NO
109  wi00843647      ISS1:1OF1       p30186_1  07/11/2011  p30186_1.cpl   NO
110  wi00856244      ISS1:1OF1       p30418_1  07/11/2011  p30418_1.cpl   NO
111  wi00853769      ISS1:1OF1       p30891_1  07/11/2011  p30891_1.cpl   YES
112  wi00834114      ISS1:1OF1       p30456_1  07/11/2011  p30456_1.cpl   NO
113  wi00686928      ISS2:1of1       p29899_2  07/11/2011  p29899_2.cpl   NO
114  wi00883810      ISS1:1OF1       p30997_1  07/11/2011  p30997_1.cpl   NO
115  wi00869471      ISS1:1OF1       p30999_1  07/11/2011  p30999_1.cpl   NO
116  wi00853453      ISS1:1OF1       p30282_1  07/11/2011  p30282_1.cpl   NO
117  wi00833760      ISS1:1OF1       p30541_1  07/11/2011  p30541_1.cpl   NO
118  Q02152968-01    ISS1:1OF1       p30168_1  07/11/2011  p30168_1.cpl   NO
119  Q02154023       ISS1:1OF1       p30157_1  07/11/2011  p30157_1.cpl   NO
120  Q02143605-02    ISS1:1OF1       p30089_1  07/11/2011  p30089_1.cpl   NO
121  wi00688225      ISS1:1OF1       p30295_1  07/11/2011  p30295_1.cpl   NO
122  wi00701008      ISS2:1OF1       p30381_2  07/11/2011  p30381_2.cpl   NO
123  wi00839645      ISS1:1OF1       p30596_1  07/11/2011  p30596_1.cpl   NO
124  wi00856160      ISS1:1OF1       p30750_1  07/11/2011  p30750_1.cpl   NO
125  Q02150582-02    ISS2:1OF1       p30144_2  07/11/2011  p30144_2.cpl   NO
126  WI00865566      ISS1:1OF1       p30709_1  07/11/2011  p30709_1.cpl   YES
127  wi00687324      ISS1:1OF1       p16376_1  07/11/2011  p16376_1.cpl   NO
128  wi00862159      ISS1:1OF1       p30802_1  07/11/2011  p30802_1.cpl   NO
129  wi00688048      ISS1:1OF1       p25747_1  07/11/2011  p25747_1.cpl   NO
130  wi00824249      ISS1:1OF1       p30447_1  07/11/2011  p30447_1.cpl   NO
131  WI00877440      ISS1:1OF1       p30844_1  07/11/2011  p30844_1.cpl   NO
132  wi00857493      ISS1:1OF1       p30766_1  07/11/2011  p30766_1.cpl   NO
133  Q02154408       ISS1:1OF1       p30162_1  07/11/2011  p30162_1.cpl   NO
134  Q02162037       ISS1:1OF1       p30266_1  07/11/2011  p30266_1.cpl   YES
135  wi00688110      ISS1:1OF1       p30305_1  07/11/2011  p30305_1.cpl   NO
136  wi00853769      ISS1:1OF1       p30897_1  07/11/2011  p30897_1.cpl   YES
137  wi00853781      ISS1:1OF1       p30416_1  07/11/2011  p30416_1.cpl   NO
138  wi00834381      ISS1:1OF1       p30548_1  07/11/2011  p30548_1.cpl   NO
139  wi00856402      ISS1:1OF1       p30749_1  07/11/2011  p30749_1.cpl   NO
140  wi00865953      ISS1:1OF1       p30832_1  07/11/2011  p30832_1.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2011-10-10 13:56:09(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2011-09-02 08:33:26(est)
```

Communication Server 1000E signaling server service updates

```
Product Release: 7.00.20.00
In system patches: 2
PATCH#  NAME       IN_SERVICE   DATE      SPECINS  TYPE   RPM
29      p30179_1   No           04/10/11  NO       FRU    nortel-cs1000-OS-1.00.00.00-00.noarch
30      p30181_1   No           04/10/11  NO       FRU    nortel-cs1000-OS-1.00.00.00-00.noarch

In System service updates: 29
PATCH# IN_SERVICE   DATE     SPECINS  REMOVABLE    NAME
0       Yes         03/10/11   NO       YES        nortel-cs1000-linuxbase-7.00.20.10-02.i386.000
1       Yes         04/10/11   NO       YES        nortel-cs1000-baseWeb-7.00.20.10-1.i386.000
2       Yes         04/10/11   NO       YES        nortel-cs1000-patchWeb-7.00.20.10-3.i386.000
3       No          04/10/11   YES      YES        nortel-cs1000-csv-7.00.20.01-00.i386.000
```

```
4       No          04/10/11   YES       YES         nortel-cs1000-shared-tpselect-7.00.20.01-
00.i386.000
5       No          04/10/11   NO        YES         nortel-cs1000-mscConf-7.00.20-01.i386.000
6       No          04/10/11   NO        yes         nortel-cs1000-cppmUtil-7.00.20.01-00.i686.000
7       No          04/10/11   NO        YES         nortel-cs1000-cs1000WebService_6-0-7.00.20.03-
00.i386.000
8       No          04/10/11   NO        YES         nortel-cs1000-sm-7.00.20.00-01.i386.001
9       No          04/10/11   NO        YES         nortel-cs1000-dbcom-7.00.20.10-1.i386.000
10      No          04/10/11   NO        YES         nortel-cs1000-nrsm-7.00.20.10-1.i386.000
11      No          04/10/11   NO        YES         nortel-cs1000-shared-pbx-7.00.20.10-1.i386.000
12      No          04/10/11   NO        YES         nortel-cs1000-shared-xmsg-7.00.20.10-1.i386.000
13      No          04/10/11   NO        YES         nortel-cs1000-kcv-7.00.20.10-1.i386.000
14      No          04/10/11   NO        YES         ipsec-tools-0.6.5-14.el5.3.i386.000
15      No          04/10/11   NO        YES         nortel-cs1000-emWeb_6-0-7.00.20.10-3.i386.000
16      No          04/10/11   NO        YES         nortel-cs1000-sps-7.00.20.10-3.i386.000
17      No          04/10/11   NO        YES         nortel-cs1000-mscAttn-7.00.20.10-2.i386.000
18      No          04/10/11   NO        YES         nortel-cs1000-dmWeb-7.00.20.10-2.i386.000
19      No          04/10/11   NO        YES         nortel-cs1000-mscAnnc-7.00.20.10-2.i386.000
20      No          04/10/11   NO        YES         nortel-cs1000-mscMusc-7.00.20.10-2.i386.000
21      No          04/10/11   NO        YES         nortel-cs1000-mscTone-7.00.20.10-2.i386.000
22      No          04/10/11   NO        YES         nortel-cs1000-csmWeb-7.00.20.10-1.i386.000
23      No          04/10/11   NO        YES         nortel-cs1000-ftrpkg-7.00.20.10-3.i386.000
24      No          04/10/11   NO        YES         nortel-cs1000-Jboss-Quantum-7.00.20.10-8.i386.000
25      No          04/10/11   NO        no          nortel-cs1000-cnd-3.2.24-00.i386.000
26      No          04/10/11   NO        YES         nortel-cs1000-tps-7.00.20.10-11.i386.000
27      No          04/10/11   NO        YES         nortel-cs1000-bcc-7.00.20.10-20.i386.000
28      No          04/10/11   NO        YES         nortel-cs1000-vtrk-7.00.20.10-33.i386.000
```

Communication Server 1000E system software

```
Product Release: 7.00.20.00
Base Applications
   base                         7.00.20     [patched]
   NTAFS                        7.00.20
   sm                           7.00.20
   nortel-Auth                  7.00.20
   Jboss-Quantum                7.00.20
   lhmonitor                    7.00.20
   baseAppUtils                 7.00.20
   dfoTools                     7.00.20
   nnnm                         7.00.20
   cppmUtil                     7.00.20
   oam-logging                  7.00.20
   dmWeb                        7.00.20
   baseWeb                      n/a         [patched]
   ipsec                        7.00.20
   Snmp-Daemon-TrapLib          7.00.20
   ISECSH                       7.00.20
   patchWeb                     n/a         [patched]
   EmCentralLogic               7.00.20
Application configuration: CS+SS+NRS+EM
Packages:
CS+SS+NRS+EM
Configuration version:     7.00.20-00
   cs                           7.00.20
   dbcom                        7.00.20
   cslogin                      7.00.20
   sigServerShare               7.00.20
   csv                          7.00.20
   tps                          7.00.20
   vtrk                         7.00.20
   pd                           7.00.20
   sps                          7.00.20
   ncs                          7.00.20
   gk                           7.00.20
   nrsm                         7.00.20
   nrsmWebService               7.00.20
   managedElementWebService     7.00.20
   EmConfig                     7.00.20
```

```
   emWeb_6-0                     7.00.20
   emWebLocal_6-0                7.00.20
   csmWeb                        7.00.20
   bcc                           7.00.20
   ftrpkg                        7.00.20
   cs1000WebService_6-0          7.00.20
   mscAnnc                       7.00.20
   mscAttn                       7.00.20
   mscConf                       7.00.20
   mscMusc                       7.00.20
   mscTone                       7.00.20
```

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.