Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.5, Avaya Network Routing Server, Avaya Session Border Controller Advanced for Enterprise R4.0.5 to support Phonect SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describes the steps to configure Session Initiation Protocol (SIP) Trunking between Phonect SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Network Routing Server, Avaya Session Border Controller Advanced for Enterprise and Avaya Communication Server 1000E.

Phonect is a member of the DevConnect SIP 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.

NOTE: This Application Note focused on the SIP Trunking aspect of the Avaya Session Border Controller Advanced for Enterprise. Advanced enterprise capabilities such as Remote Worker "a.k.a. Remote SIP Endpoints", dual forking, and TLS/SRTP were not tested. As a result, the Avaya Session Border Controller for Enterprise is also considered Compliance Tested for this solution.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Phonect SIP Trunk Service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Network Routing Server (NRS), Avaya Communication Server 1000E (CS1000E) connected to Phonect SIP Trunk Service via an Avaya Session Border Controller Advanced for Enterprise (ASBCAE). Customers using this Avaya SIP-enabled Enterprise Solution with Phonect 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 NRS, ASBCAE and CS1000E. The enterprise site was configured to use the SIP Trunk to Phonect SIP Trunk Service.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming PSTN calls were made to Unistim, SIP, Digital and Analog telephones at the enterprise
- Incoming calls to the enterprise site from the PSTN routed to the DDI numbers assigned by Phonect
- Outgoing calls from the enterprise to the PSTN were made from Unistim, SIP, Digital and Analog telephones
- Outgoing calls from the enterprise site completed via Phonect to PSTN destinations
- Calls using the G.711A, G.711U and G.729 codec supported by Phonect
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- Call coverage and call forwarding for endpoints at the enterprise site
- Mobile-X call features were not tested

2.2. Test Results

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

- When making an outbound call, the ASBCAE has to send 5 INVITES before the Service Providers SIP Trunk Network replies with 100 Trying and 183 Session Progress messaging. This may result in a 7-8 second delay on outbound calls. All unwanted MIME was stripped on outbound calls. Service Provider has now passed this issue to their developers to be resolved

- No inbound toll free numbers were tested as none were available from the Service Provider
- No Emergency Services numbers tested as test calls to these numbers should be pre-arranged with the Operator

2.3. Support

For initial setup for your Avaya solution please contact leveranse@phonect.no. For technical support send email to bedriftsupport@phonect.no.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Phomect SIP Trunks Service. Located at the enterprise site are NRS, ASBCAE and a Communication Server 1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya IP Softphones (SMC3456, 2050 and one-X Communicator), Avaya Digital telephone, Analog telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

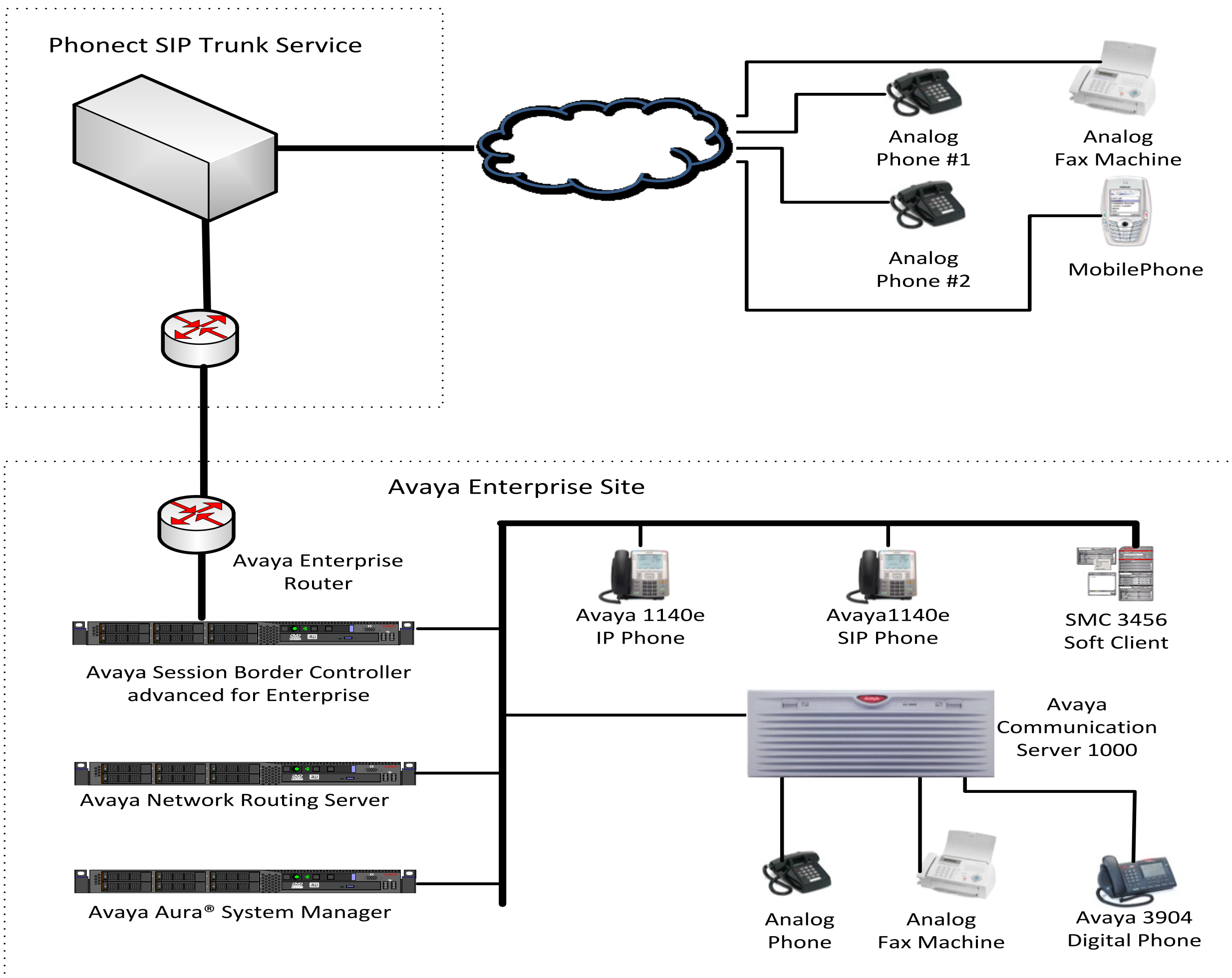


Figure 1: Phonect SIP Trunk Topology

4. Equipment and Software Validated

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

Equipment	Software
Avaya Communication Server 1000E running on CP+PM server as co-resident configuration	Avaya Communication Server 1000E R7.5, Version 7.50.17 Service Update: 7.50_17Jan11 Deplist: X21 07.50Q
Avaya Network Routing Server	Avaya Network Routing Server R7.50
Avaya S8800 server	Avaya Session Border Controller Advanced for Enterprise Build: 4.0.5.Q02
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC CD01 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA07 DSP1 Version: DSP1 AB03
Avaya 1140e and 1230 Unistim Telephones	FW: 0625C8A
Avaya 1140e and 1230 SIP Telephones	FW: 04.01.13.00.bin
Avaya SMC 3456	Version 2.6 build 53715
Avaya one-X® Communicator	one-X® Communicator -Version cs6.1.0.10
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
Phonect SIP Trunk Service	MC22

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. These SIP trunks carry SIP Signaling associated with Phonect SIP Trunk Service. For incoming calls, the NRS receives SIP messages from the ASBCAE; through which Phonect SIP Service directs incoming SIP messages to Communication Server 1000E (see **Figure 1**). Once a SIP message arrives at Communication Server 1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. Once Communication Server 1000E selects a SIP trunk, the SIP signaling is routed to the NRS. The NRS directs the outbound SIP messages to the ASBCAE and on to Phonect'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 NRS is presumed to have been previously completed and is not discussed here.

5.1. Log in to 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,** this will take the user into the vxworks shell of the call server. Next type **logi**, the user will then be asked to login with correct credentials. Once logged in the user can then progress to load any overlay.

5.2. Confirm System Features

The keycode installed on the Call Server controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya Sales representative to add additional capacity. Use the Communication Server 1000E system terminal and manually load overlay 22 to print the System Limits (the required command is SLT), and verify that the number of SIP Access Ports reported by the system is sufficient for the combination of trunks to Phonect'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.

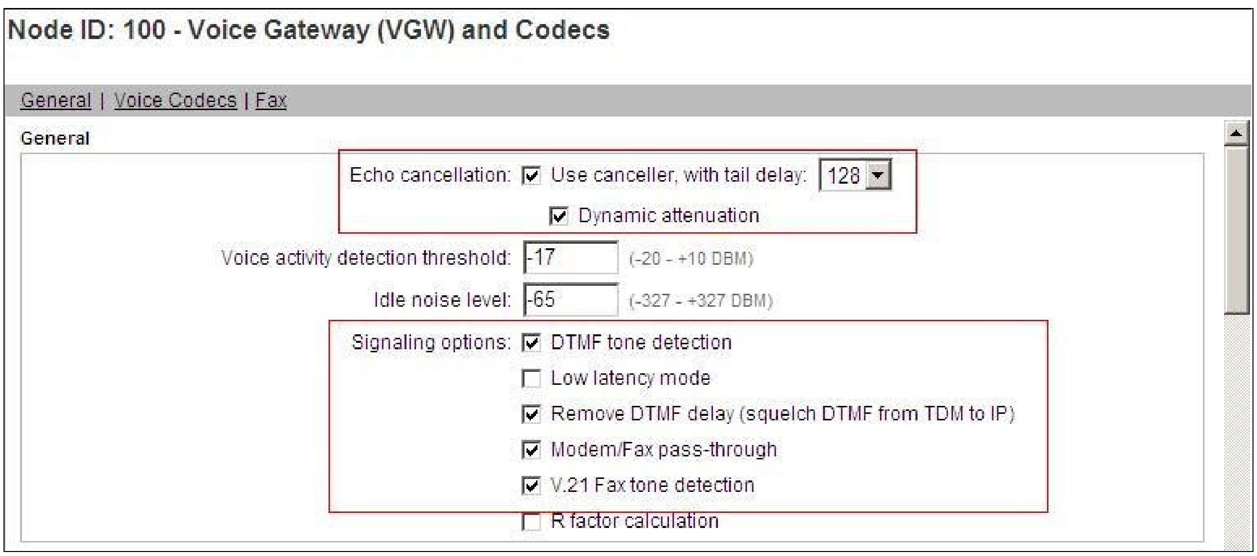
```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

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        2000    LEFT 1970     USED    30
```

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

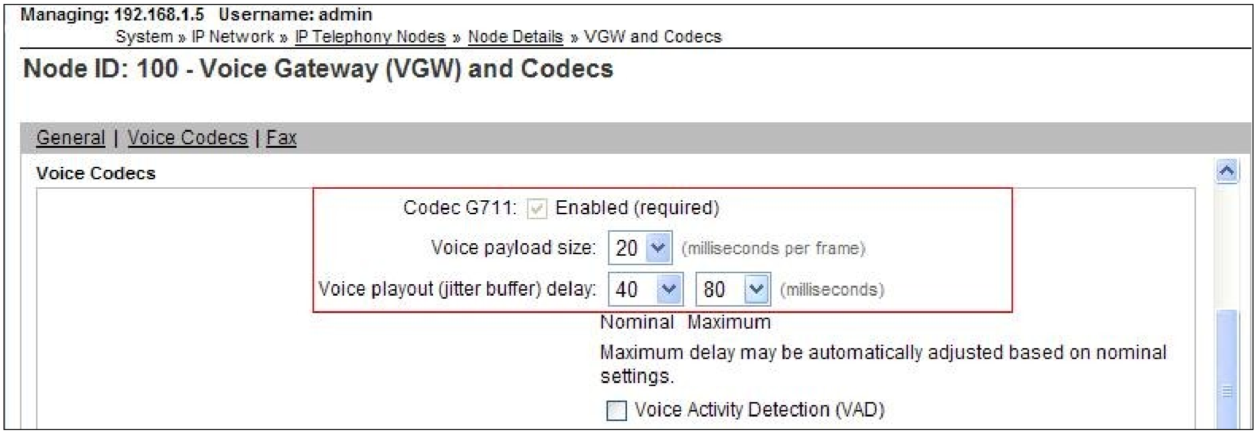
```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

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

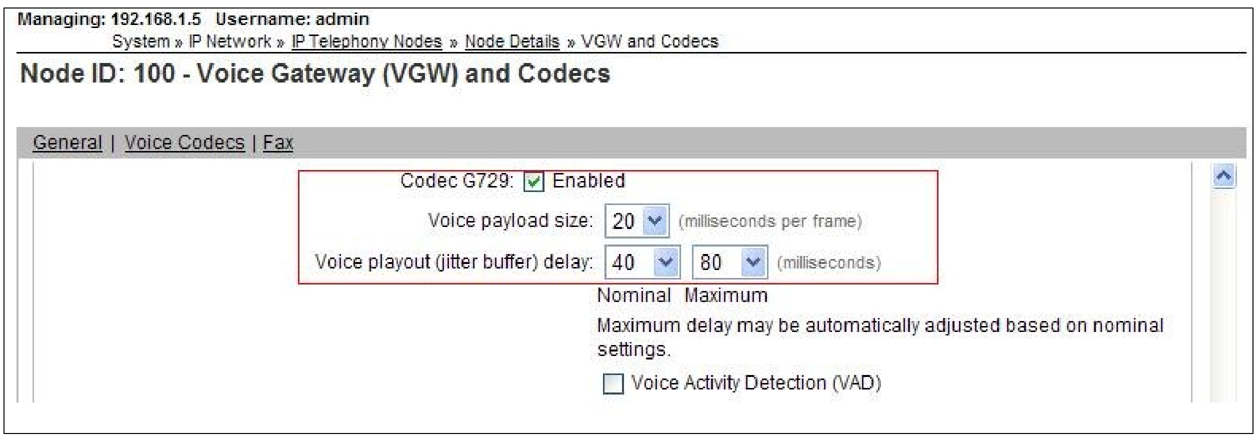
Phonect SIP Trunk service supports G.711A, G.711U and G.729 voice codec's and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW Gateway (VGW) and Codecs** property page and configure the Communication Server 1000E General codec settings as in the next screenshot.



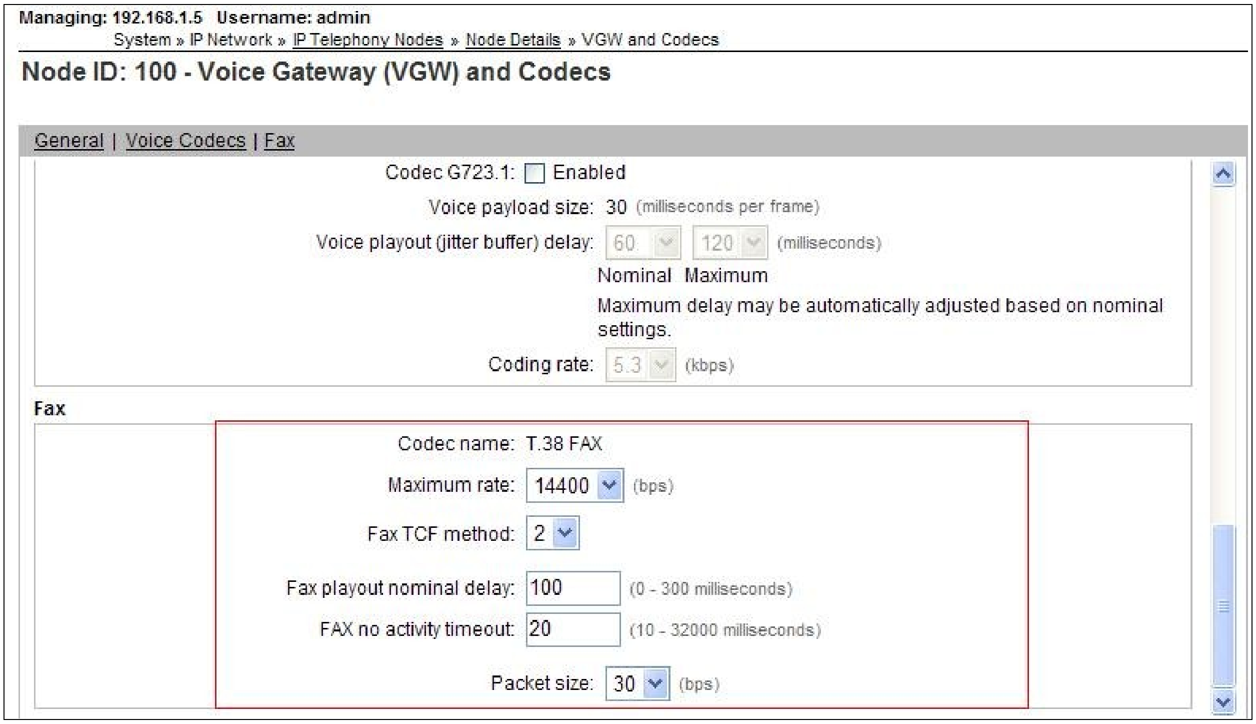
Next, scroll down and configure the **Codec G.711**. The relevant settings are highlighted in the following screenshot.



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

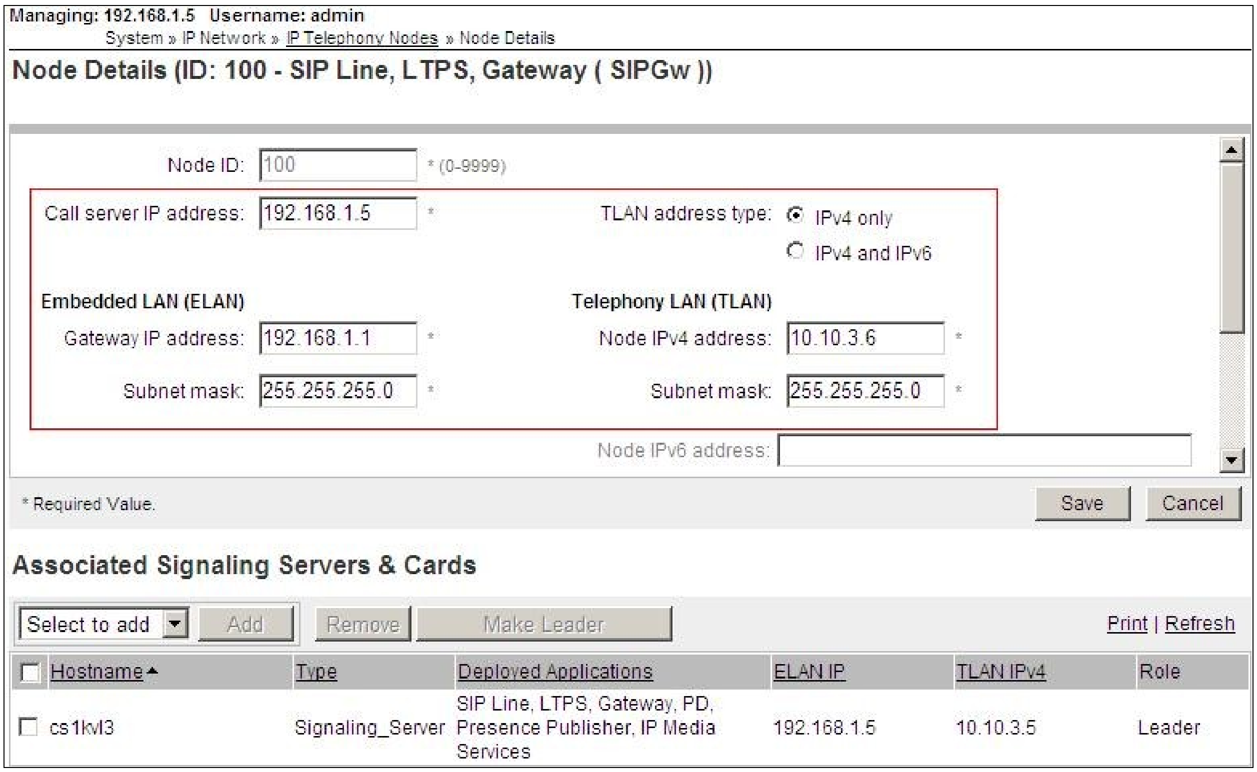


Finally, configure the **Fax** settings as in the highlighted section of the next screenshot.



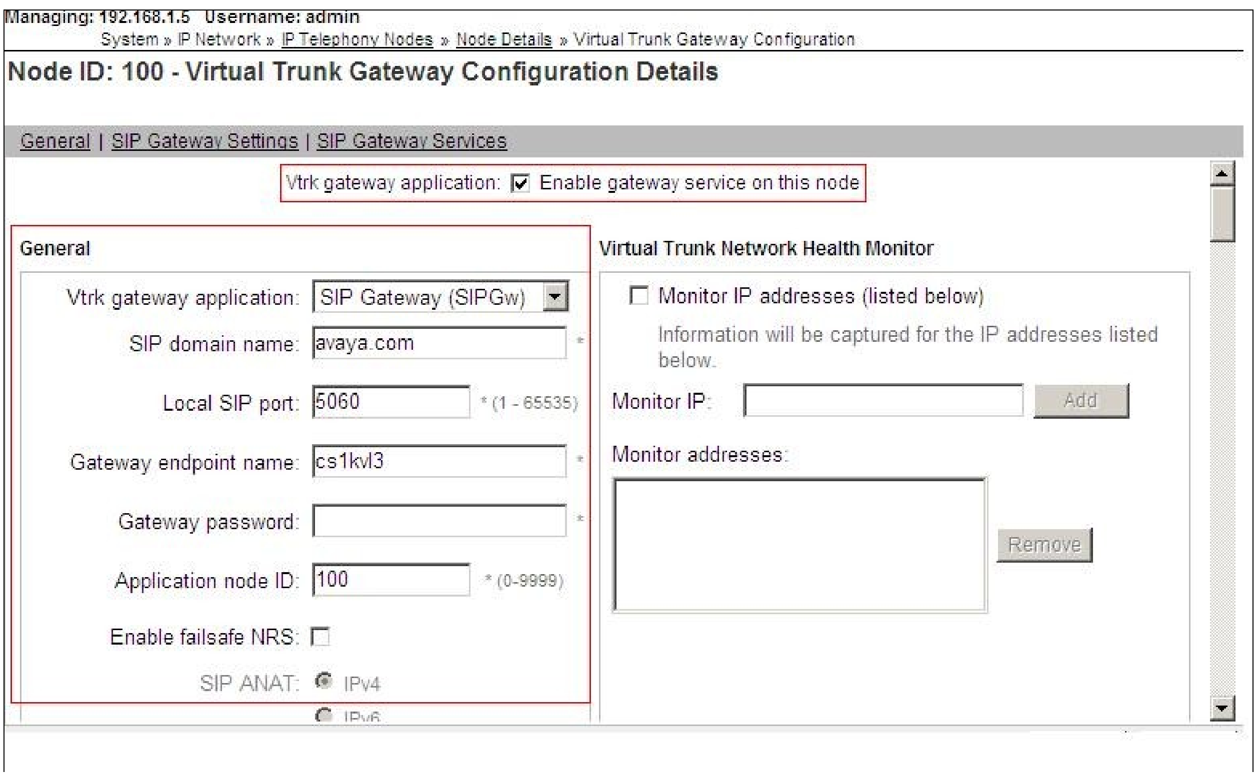
5.4. Virtual Trunk Gateway Configuration

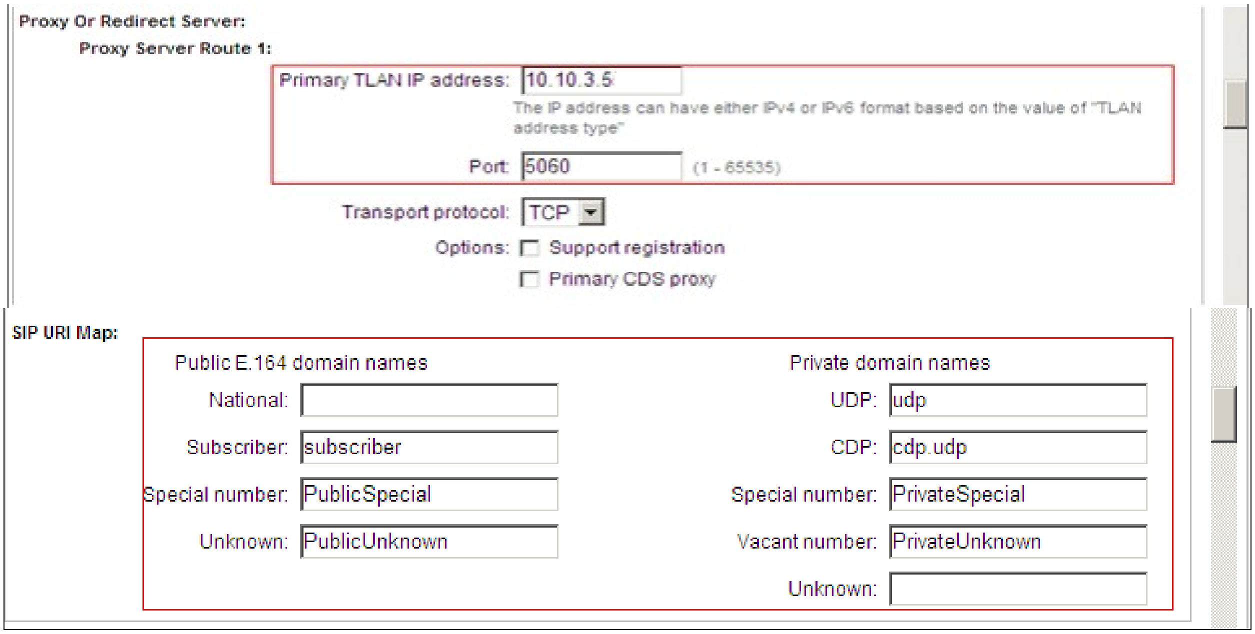
Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks of the Node. At this stage the call server has an ip address and so too does the signalling server. The Node ip is the ip address that the IP phones use to register. This is also where the SIP trunk connection is made to the 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.**
- **SIP domain name:** The SIP Domain Name is the SIP Service Domain. The SIP Domain Name configured in the Signaling Server properties must match the Service Domain name configured in the 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 is used when a Network Routing Server is used for registration of the endpoint.
- **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 **100**
- **Proxy or Redirect Server:** Primary TLAN ip address is the Security Module ip address of the NRS. The **Transport protocol** used for **SIP**, in this case is TCP
- **SIP URI Map: Public National** and **Private Unknown** are left blank. All other fields in the SIP URI Map are left with default values.

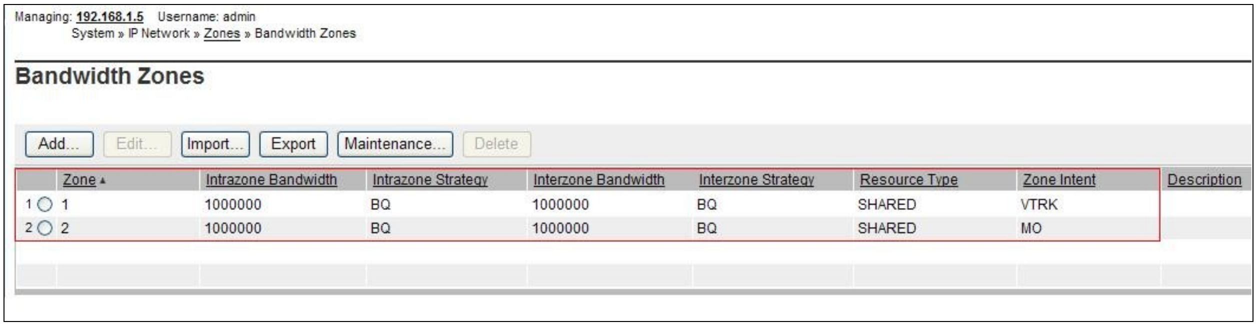




Proxy Or Redirect Server:
Proxy Server Route 1:
Primary TLAN IP address: 10.10.3.5
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)
Transport protocol: TCP
Options: Support registration
Primary CDS proxy
SIP URI Map:
Public E.164 domain names
National:
Subscriber: subscriber
Special number: PublicSpecial
Unknown: PublicUnknown
Private domain names
UDP: udp
CDP: cdp.udp
Special number: PrivateSpecial
Vacant number: PrivateUnknown
Unknown:

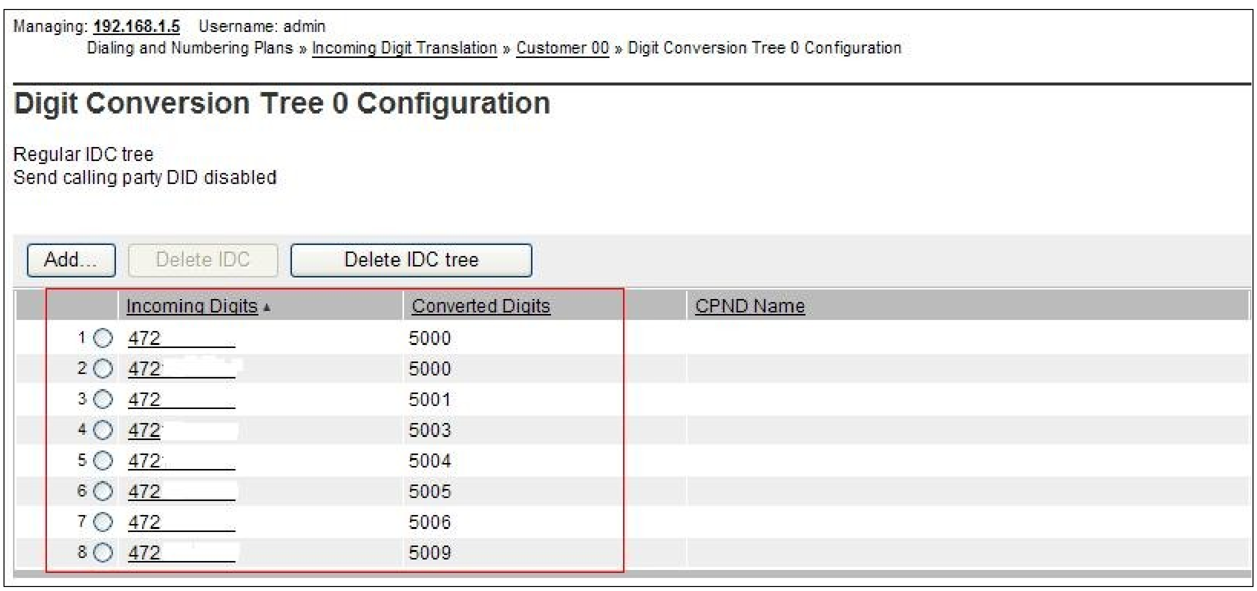
5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for Bandwidth Management. SIP trunks require a unique zone, not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in separate zones. Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **System → IP Network → Zones → Bandwidth Zones** and add new zones as required.



5.6. Configure Incoming Digit Conversion Table

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



5.7. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to Phonect'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 Digit Manipulation Data Block (**DGT**), configure using the Communication Server 1000E system terminal and overlay 86
- Configure a Route List Block (**RLB**); configure using the Communication Server 1000E system terminal and overlay 86
- Configure Co-ordinated Dialling Plan(s) (**CDP**); configure using the Communication Server 1000E system terminal and overlay 87

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

Next, configure the SIP trunk Route Data Block (RDB) using the 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**. The remaining highlighted values are important for correct SIP trunk operation.

```
Overlay 16
TYPE: RDB
CUST 00
ROUT 1
TYPE RDB
CUST 00
ROUT 1
DES  VIR_TRK
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
RPA  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 00001
PCID SIP
CRID NO
NODE 100
DTRK NO
ISDN YES
    MODE ISLD
    DCH  1
    IFC  SL1
    PNI  00000
    NCNA YES
    NCRD YES
    TRO  NO
    FALT NO
    CTYP UKWN
    INAC NO
    ISAR NO
    DAPC NO
MBXR NO
MBXOT NPA
MBXT 0
PTYP ATT
CNDP UKWN
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
```

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

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

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

Next, configure a Digit Manipulation data block (DGT) in overlay 86. Load **Overlay 86** at the system terminal and type **new**. The following example shows the values used. The value for **DMI** is the same used when inputting the **DMI** value during configuration of the Route List Block.

```
Overlay 86
CUST 0
FEAT dgt
DMI  10
DEL  0
ISPN NO
CTYP NPA
```

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

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

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

ISET 0
NALT 5
MFRL 0
OVLL 0
```

Next, configure Co-ordinated Dialling Plan(s) (CDP) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and **Overlay 87**. The following are some example CDP entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (**RLI**), this is the default PSTN route to the SIP Trunk service.

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

5.8. 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 5.5** for **VIRTUALSETS**.

```
Overlay 20 IP Telephone configuration
DES  1140
TN   100 0 01 0  VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00002
CUR_ZONE 00002
ERL  0
ECL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMD LLCN MCTD CLBD AUTR
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA  PKCH MUTA MWTD

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```

```
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DVLD CROD CROD
CPND_LANG ENG
RCO  0
HUNT 0
LHK  0
PLEV 02
PUID
DANI NO
AST  00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 5000 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME IP1140
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 5000 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   04 0 02 00  VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL  0
FDN  0
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 1
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMA LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
     CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD
CPND_LANG ENG
RCO  0
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 1
AACS
ACQ
ASID
SFNB
SFRB
USFB
CALB
FCTB
ITNA NO
DGRP
PRI  01
MLWU_LANG 0

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```

```
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MLNG ENG
DNDR 0
KEY  00 MCR 5008 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     01 MCR 5008 0
        CPND
          CPND_LANG ROMAN
            NAME Digital Set
            XPLN 10
            DISPLAY_FMT FIRST,LAST
     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 CLT
     28 RLT
     29
     30
     31
```

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

Overlay 20 – Analog Telephone Configuration

```
DES  500
TN   04 0 03 00
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN   5015
AST  NO
IAPG 0
HUNT
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI  0
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR DTN FBD XFD WTA THFD FND HTD ONS
     LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
     CFTD SFD MRD C6D CNID CLBD AUTU
     ICDD CDMD LLCN EHTD MCTD
     GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
     MBXD CPFA CPTA UDI RCC HBTD IRGD  DDGA NAMA MIND
     NRWD NRCD NROD SPKD CRD PRSD MCRD
     EXR0 SHL SMSD ABDD CFHD DNDY DNO3
     CWND USMD USRD CCBD BNRD OCBD RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
     FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD
PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR  DCFW 4
```

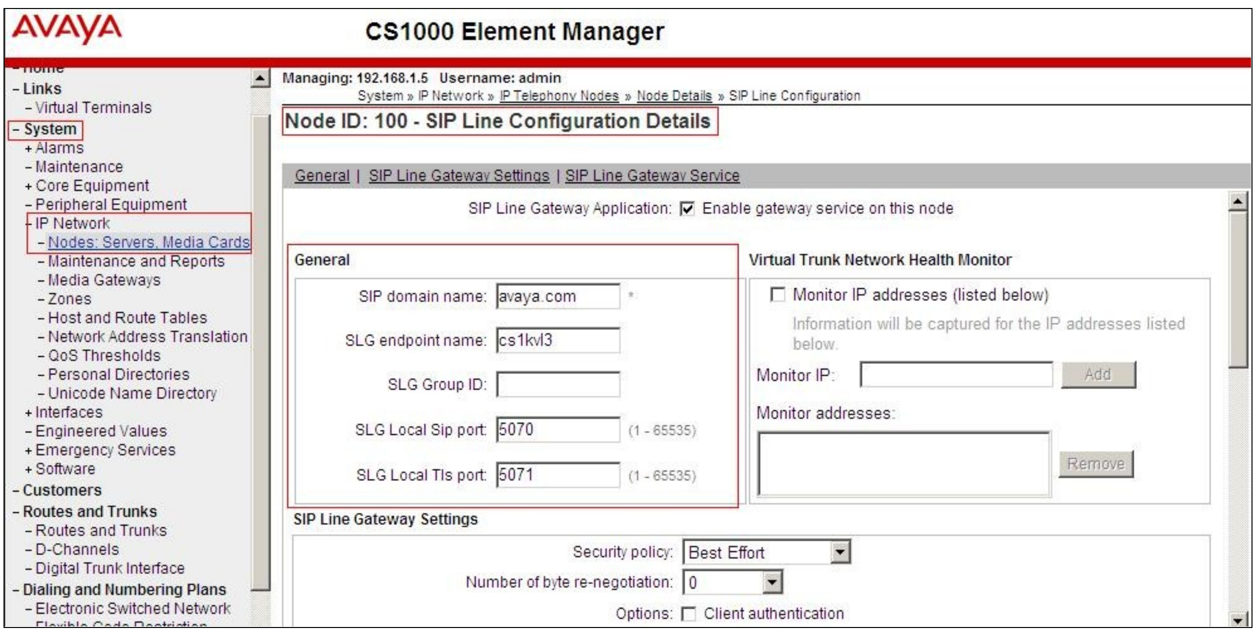
5.9. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the Communication Server 1000E system terminal and **Overlay 15** to activate SIP Line services, as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
  SIPL_ON YES
  UAPR 78
  NMME NO
```

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

- **SIP Line Gateway Application: Enable the SIP line service on the node**, check the box to enable
- **SIP Domain Name: The value must match that configured in Section 6.1**
- **SLG endpoint name:** The endpoint name is the same endpoint name as the SIP Line Gateway and will be used for SIP gateway registration
- **SLG Local Sip port:** Default value is **5070**
- **SLG Local TLS port:** Default value is **5071**



5.10. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the Communication Server 1000E system terminal and **Overlay 20** to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value for **SIPN** is 1. The **SIPU** value is the username, **SCPW** is the logon password and these values are required to register the SIP telephone to the SLG. The value for **CFG_ZONE** is the value set for **SIPLINEZONE** in **Section 5.5**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value and the telephone number used in **KEY 00**.

```
Overlay 20 – SIP Telephone Configuration
DES  SIPD
TN   100 0 01 10  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 5003
NDID 100
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID 100
NHTN 100 0 01 10
CFG_ZONE 00002
CUR_ZONE 00002
ERL  0
ECL  0
VSIT NO
FDN
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
     MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD

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

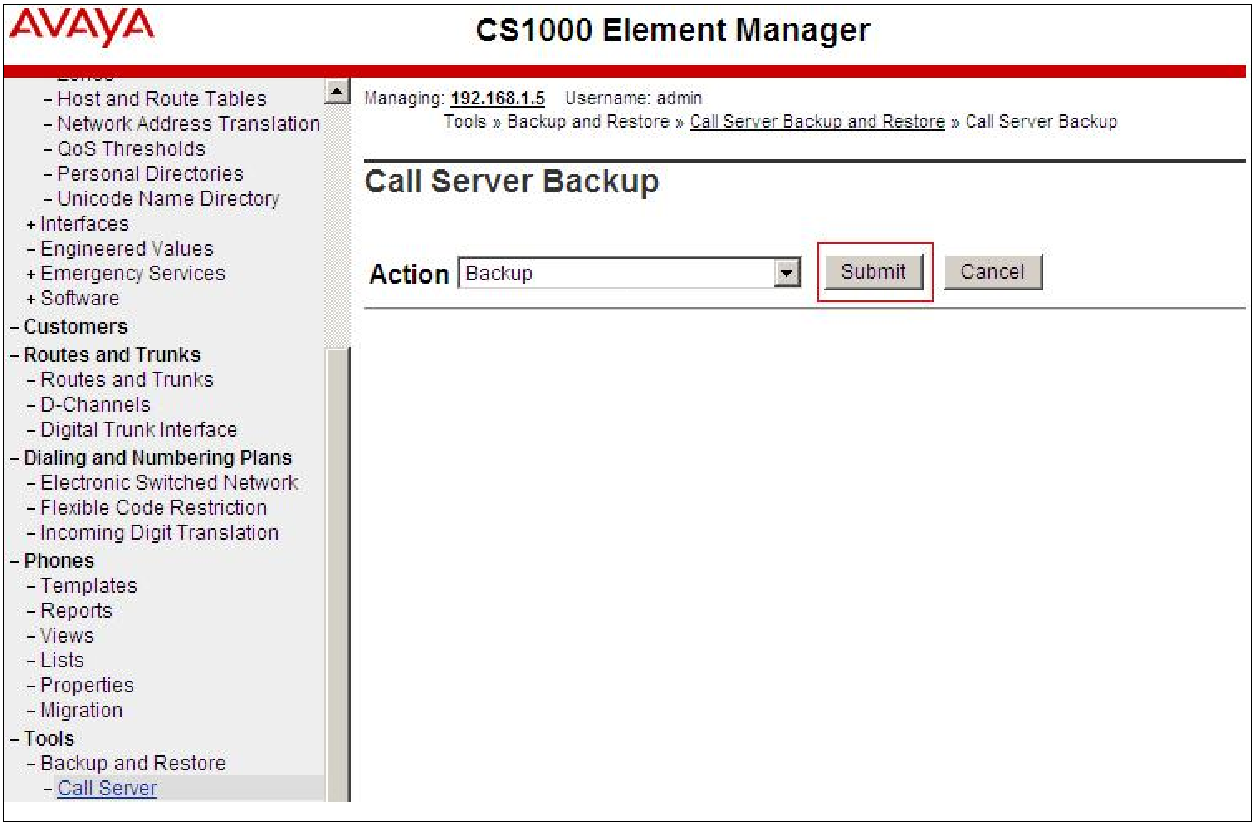
```
---continued from previous page---

     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 SCR 5003 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Sigma 1140
            XPLN 11
            DISPLAY_FMT FIRST,LAST*
     01 HOT U 115003 MARP 0
     02
     03
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23     *
     24 PRS
     25 CHG
     26 CPN
     27
     28
     29
     30
     31
```

5.11. Save Configuration

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



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

Configuration of Communication Server 1000E is complete.

6. Configure Avaya Network Routing Server

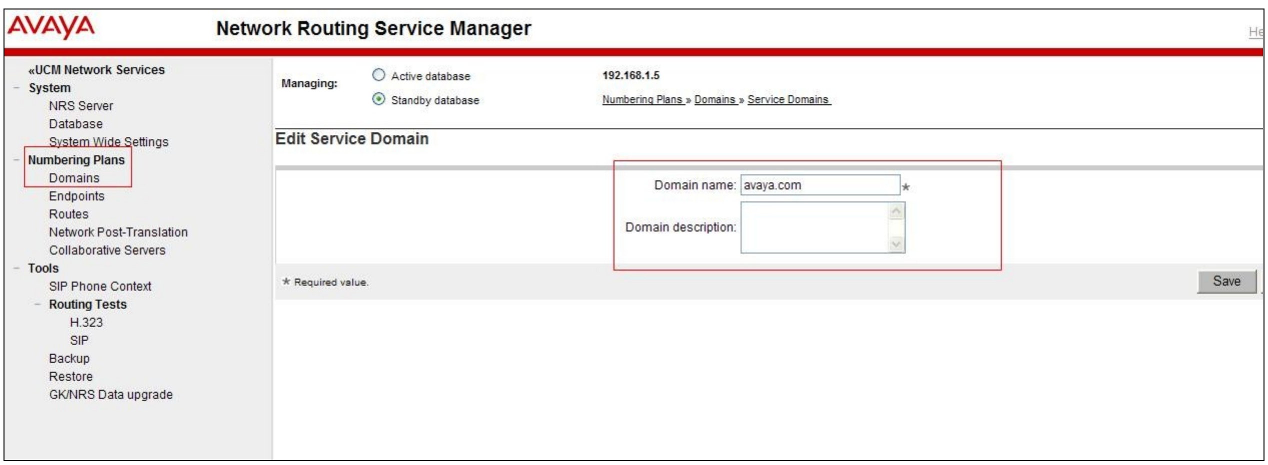
This section provides the procedure for configuring the NRS to receive and route calls over the SIP trunk between the ASBCAE and the CS1000E. These instructions assume other administration activities have already been completed such as defining system wide settings. The following administration activities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

- Define SIP Domain
- Define Endpoints
- Define Routes

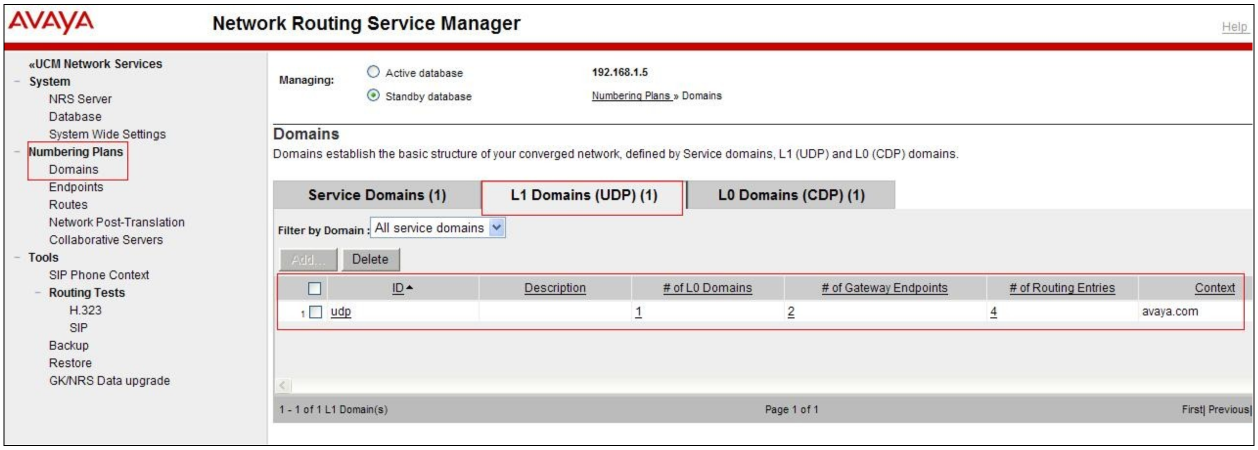
Configuration is accomplished by accessing the browser-based GUI of the Unified Communications Manager (UCM), using the URL **https://<ip-address>/network-login**, where **<ip-address>** is the IP address of UCM. Once logged on click on the NRS Manager link on the UCM front page and log in with the appropriate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

6.1. Define SIP Domain

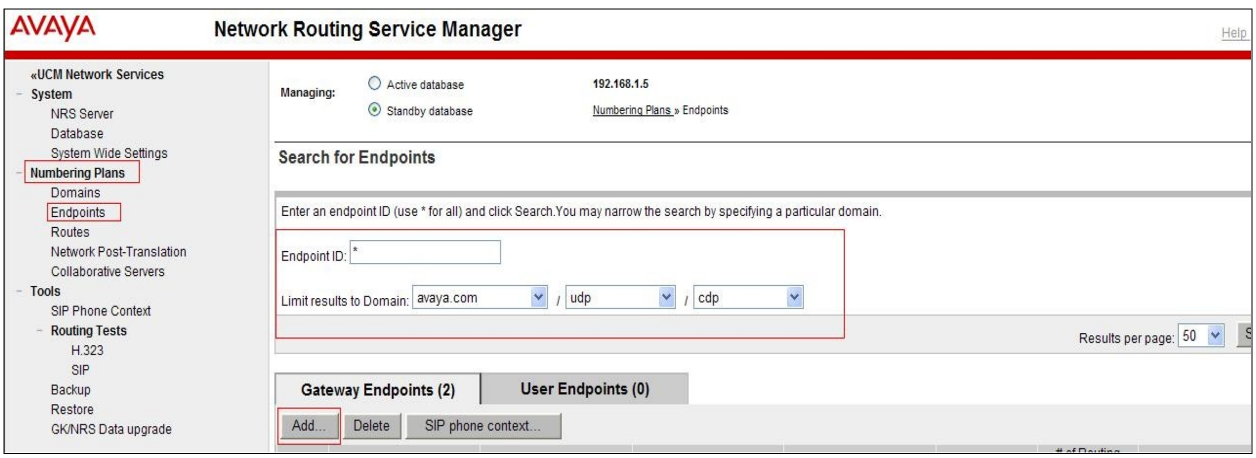
Create a SIP Domain by clicking on **Numbering Plans → Domains**, click **Add** (not shown). Enter a name for your Domain name and click on **Save**. In the test configuration **avaya.com** was used and this is the same domain that is configured on the CS1000E as per **Section 5.4.**



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

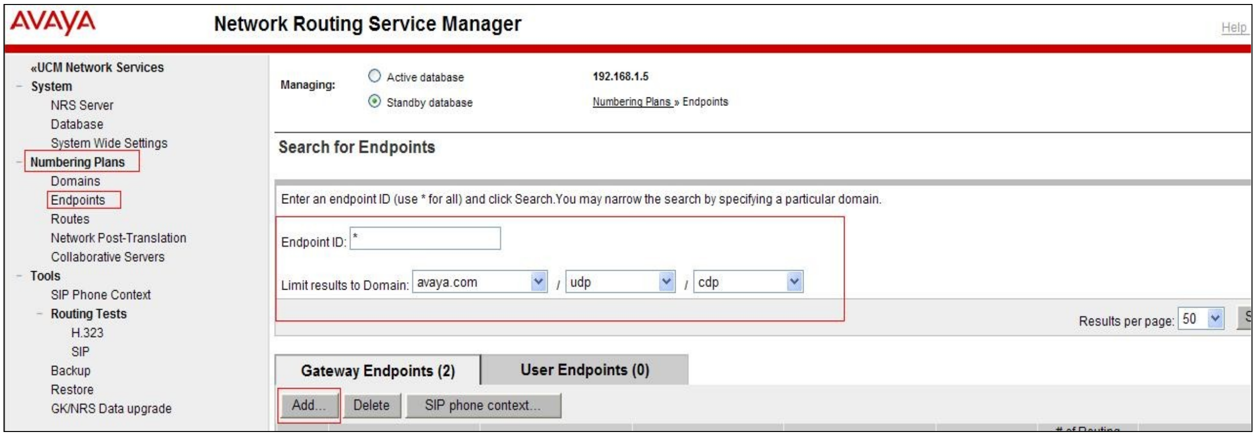


This is the CDP domain created for avaya.com/udp domain.



6.2. Define Endpoints

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

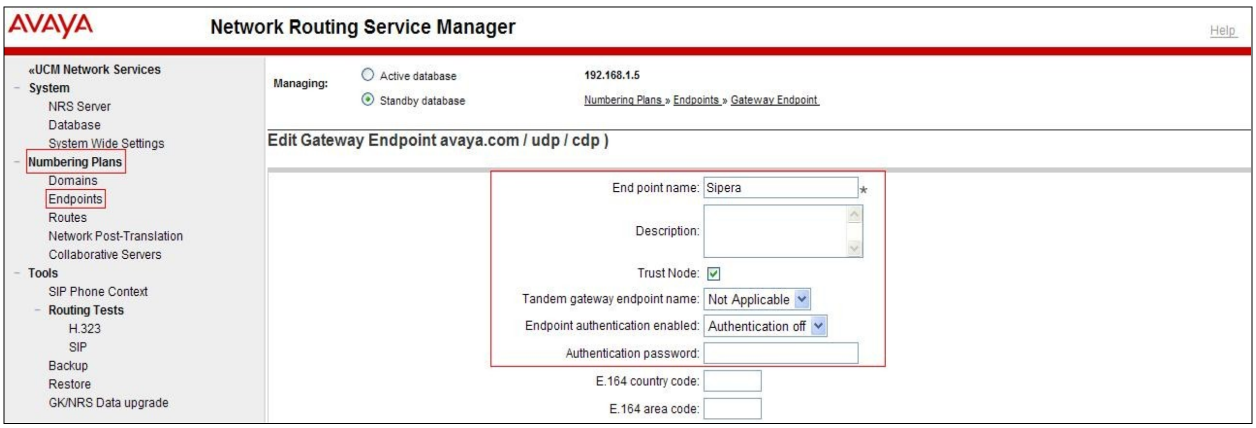


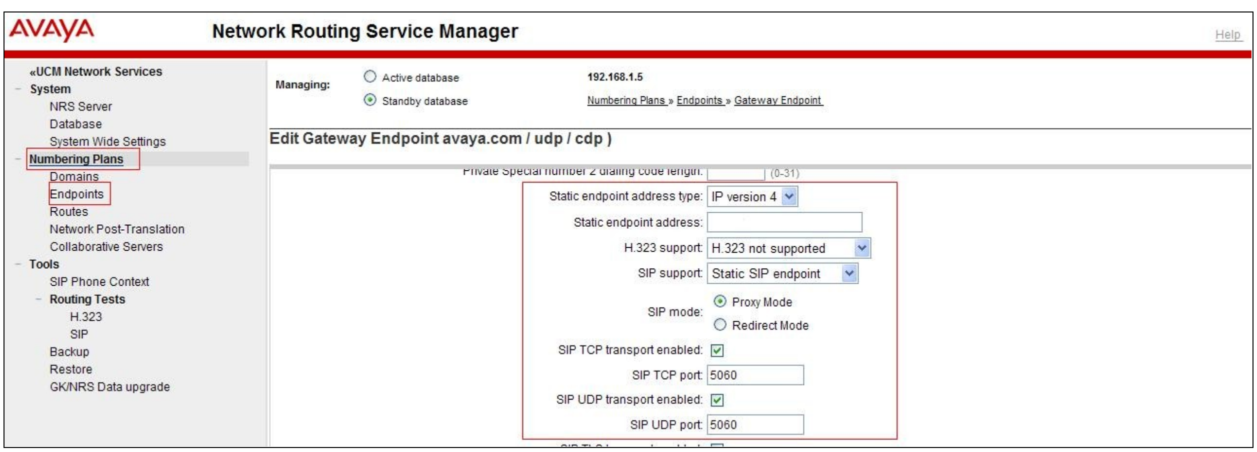
6.2.1. Configure Static Endpoint for Avaya Session Border Controller Advanced for Enterprise

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

- **End point name:** **Sipera** was used in this configuration
- **Description:** Enter a descriptive name
- **Static endpoint address:** This is the ip address of ASBCAE. 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 ASBCAE Endpoint configuration used for the testing.



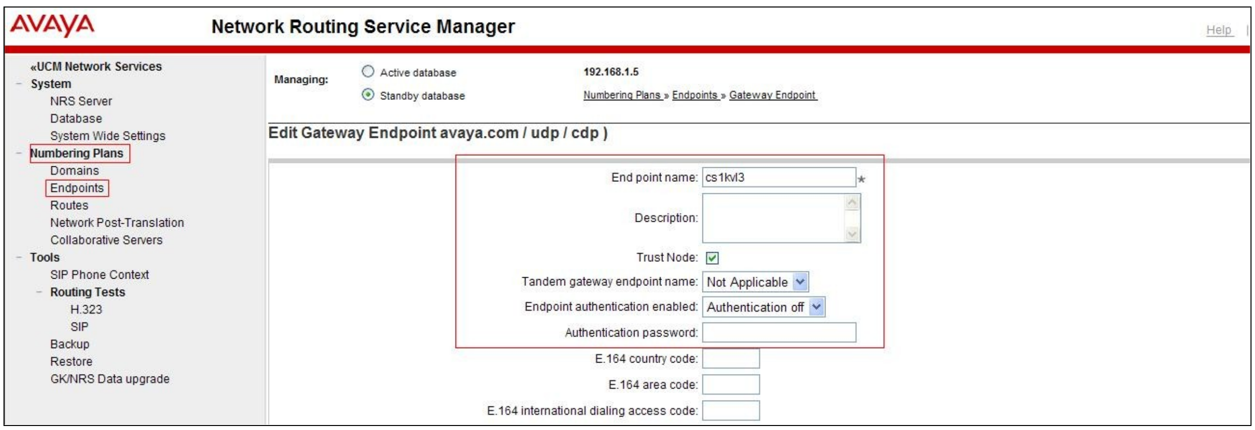


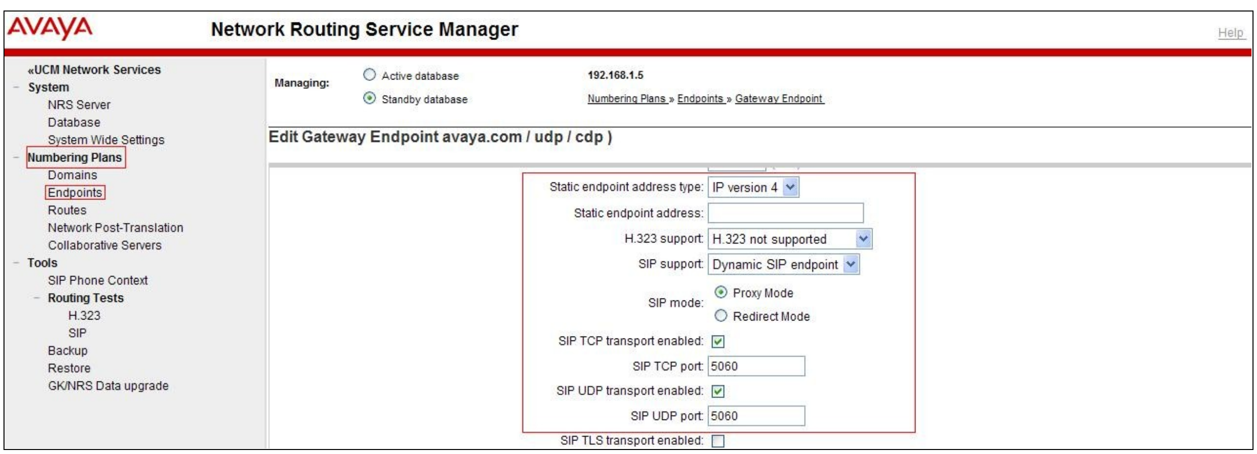
6.2.2. Configure Dynamic Endpoint for Avaya Communication Server 1000E

This section shows how to add a dynamic endpoint for the CS1000E. Enter the following values and use default values for remaining fields.

- **End point name:** **cs1kvl3** was used in this configuration
- **Description:** Enter a descriptive name
- **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.



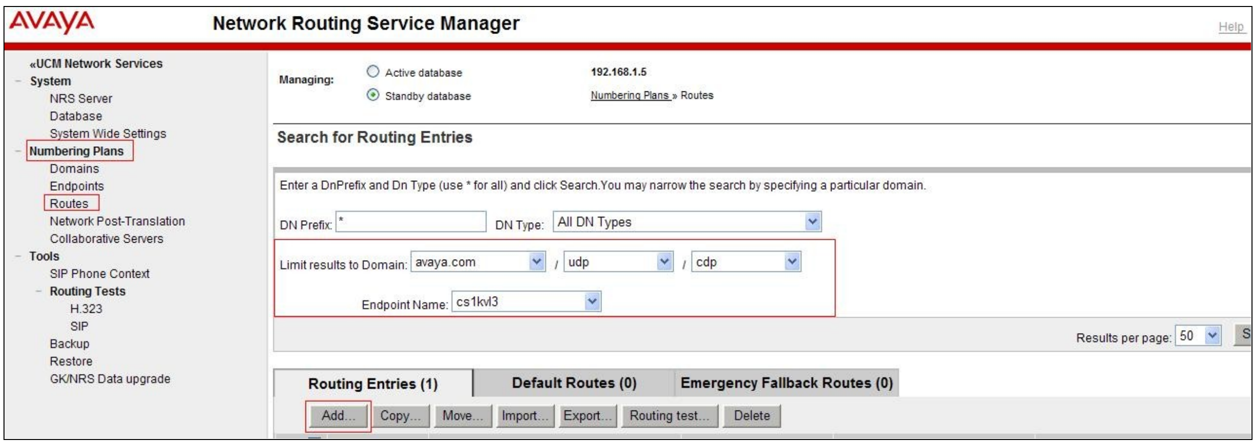


6.3. Define Routes

Routes need to be defined for each endpoint. Routes are 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

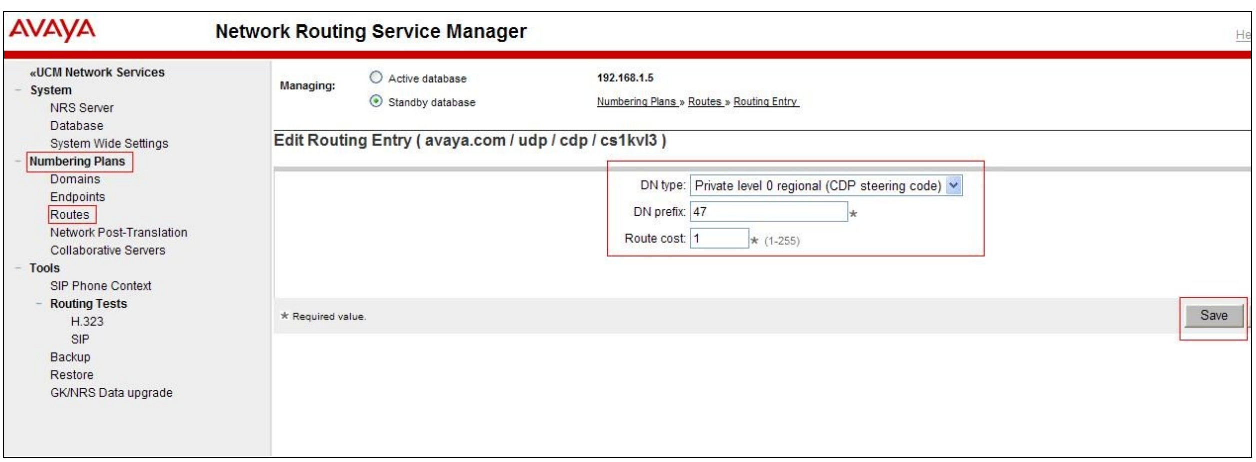
This section shows how to add routes to the CS1000E. This is incoming calls from the PSTN to the CS1000E. Create a Route by clicking on **Numbering Plans → Routes**. Select the domain and sub-domains (e.g. **avaya.com/udp/cdp**) and endpoint where you want to create the route (e.g. **cs1kvl3**) and click **Add**.



Enter the following values

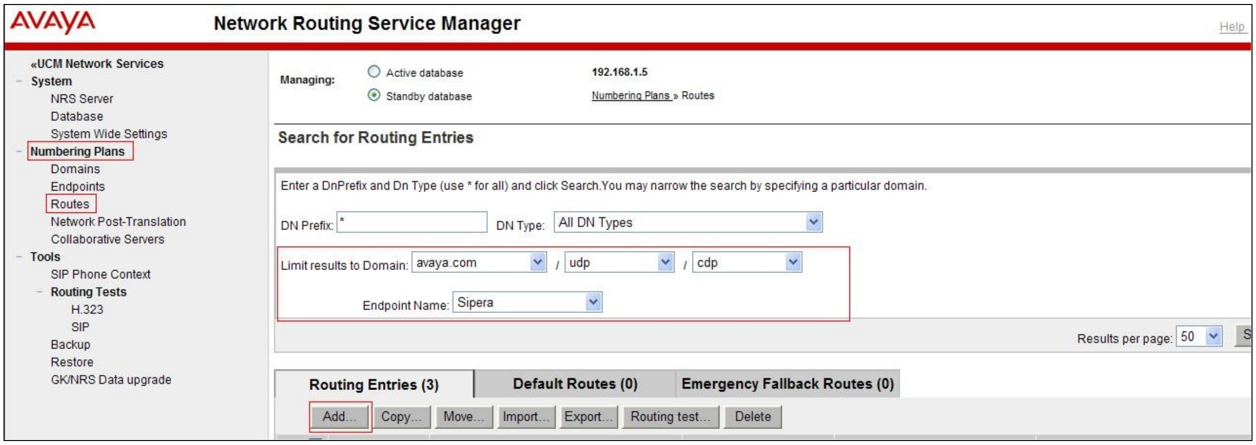
- **DN type:** Select **Private level 0 regional (CDP steering code)**
- **DN prefix:** **47** prefix matches the DN extensions on the test sets on the CS1000E. This also matches the first 2 digits of the DDI range given for the test
- **Route cost:** **1** is used as this is the only route available

Click on **Save.**



6.3.2. Configure Route to the Avaya Session Border Controller Advanced for Enterprise

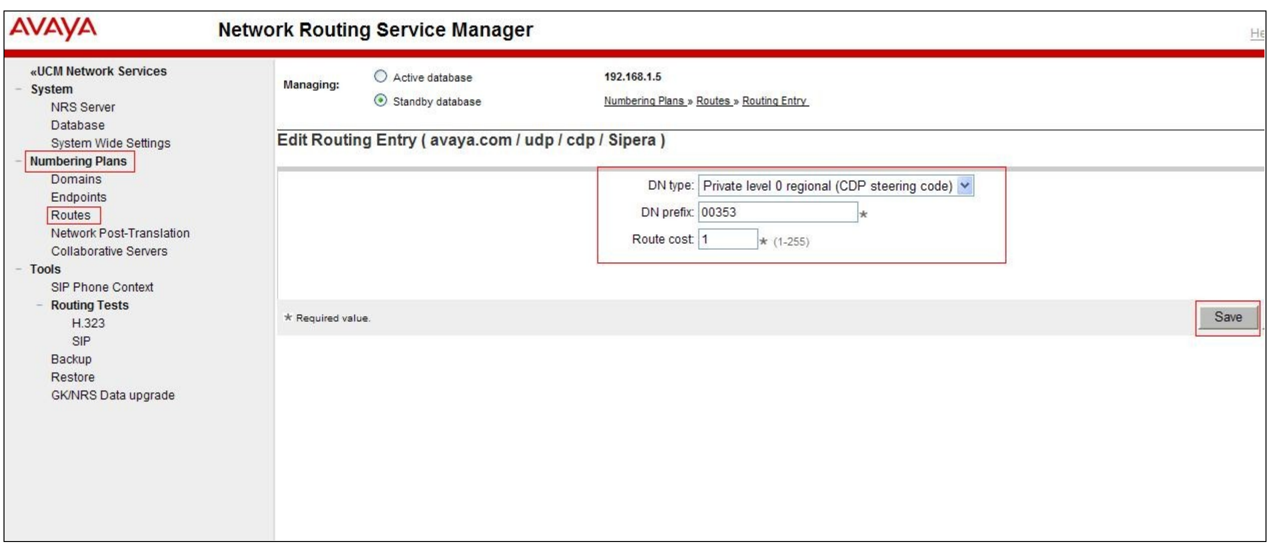
This section shows how to add routes to the ASBCAE. This is outgoing calls from the CS1000E to the PSTN. Create a Route by clicking on **Numbering Plans → Routes**. Select the domain and sub-domains (e.g. **avaya.com/udp/cdp**) and endpoint where you want to create the route (e.g. **Sipera**) and click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



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

- **DN type:** Select **Private level 0 regional (CDP steering code)**
- **DN prefix:** **00353** prefix matches the digits going to be dialed for an international call
- **Route cost:** **1** is used as this is the only route available

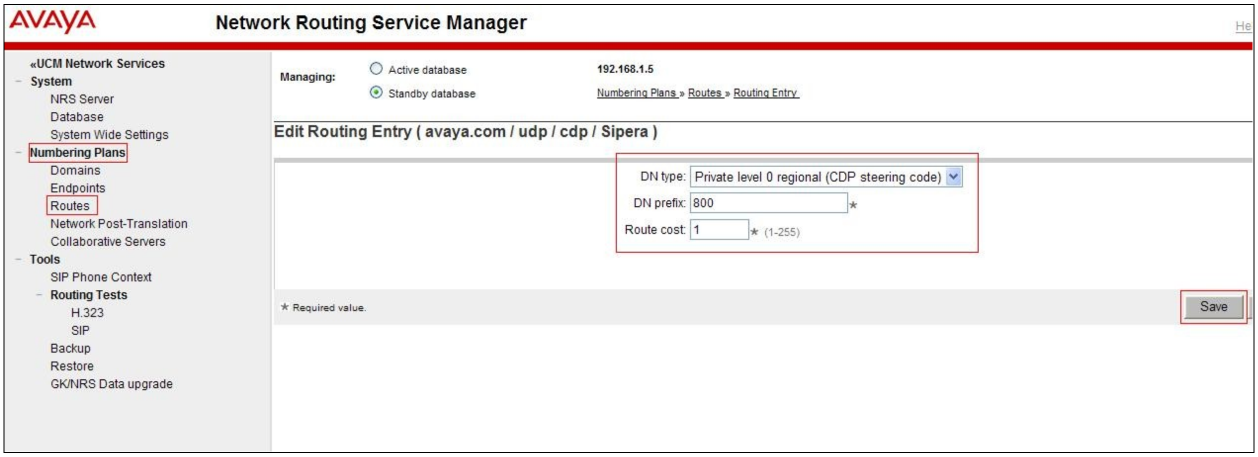
Click on **Save.**



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

- **DN type:** Select **Private level 0 regional (CDP steering code)**
- **DN prefix:** **800** 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 Session Border Controller Advanced for Enterprise

This section describes the configuration of the Session Border Controller. The ASBCAE is administered using the E-SBC Control Center.

7.1. Access Avaya Session Border Controller Advanced for Enterprise

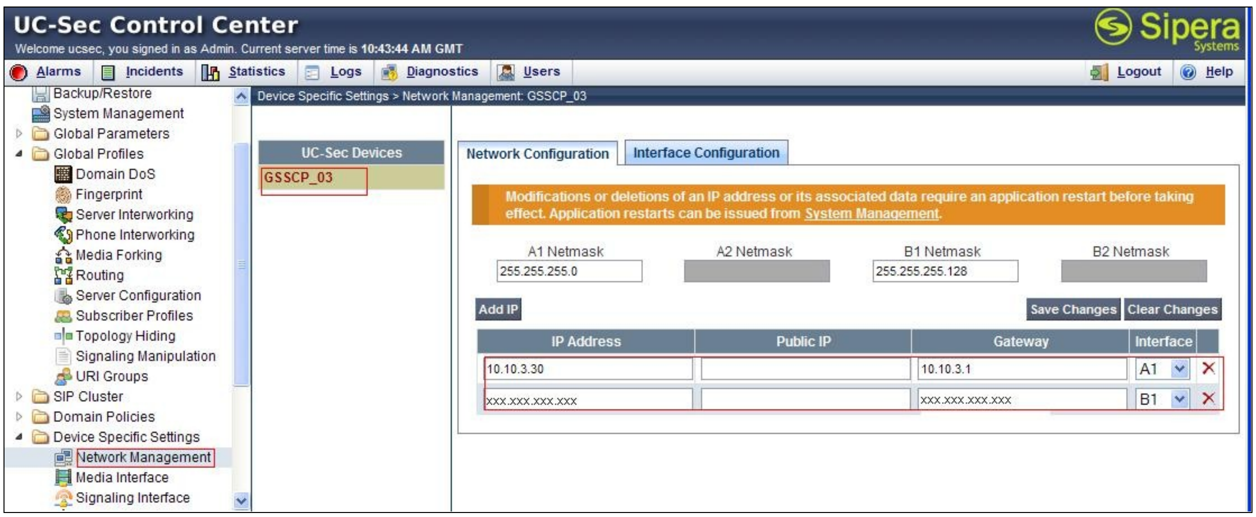
Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. Select the **UC-Sec Control Center.**

Select **UC-Sec Control Center** and enter the **Login ID** and **Password.**

7.2. Define Network Information

To define the network information for the ASBCAE, click on the **Device Specific Settings** to expand the options, then select **Network Management**.

- Click on **Add IP**
- Define the internal IP address with screening mask and assign to interface **A1**
- Select Save (not shown) to save the information
- Click on **Add IP**
- Define the external IP address (not shown) with screening mask and assign to interface **B1**
- Select Save (not shown) to save the information
- Select the **Network Configuration** tab and change the state of interfaces A1 and B1 to **Enabled**
- Click on **System Management** in the main menu
- Select **Restart Application** indicated by an icon in the status bar



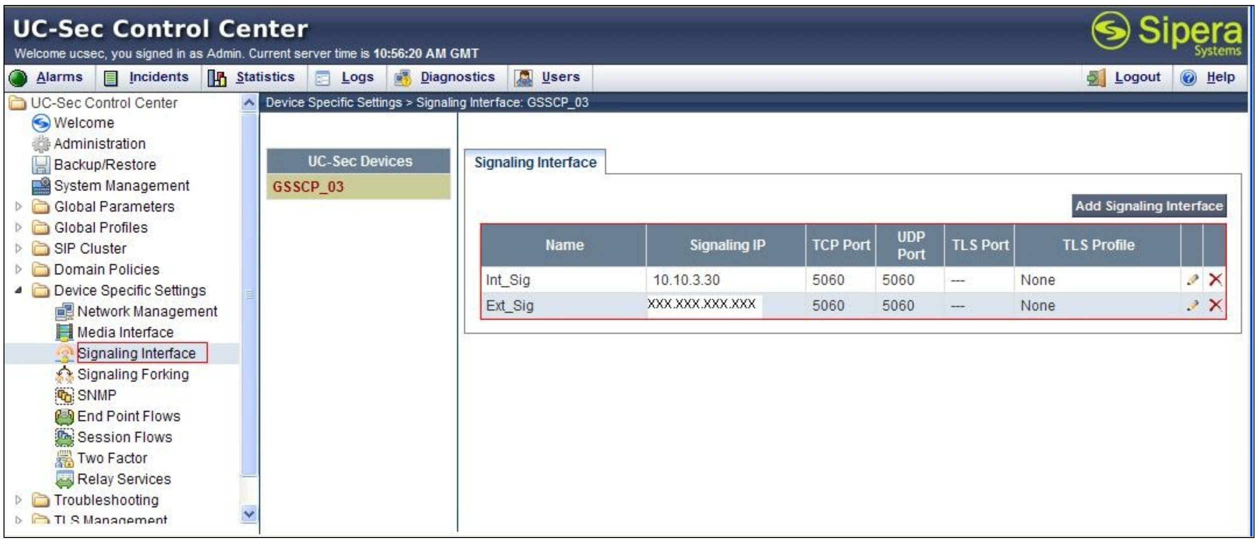
7.3. Define Interfaces

To define the signaling and media interfaces for the ASBCAE, click on the **Device Specific Settings** to expand the options.

7.3.1. Signalling Interfaces

Select **Signalling Interface** from the menu options.

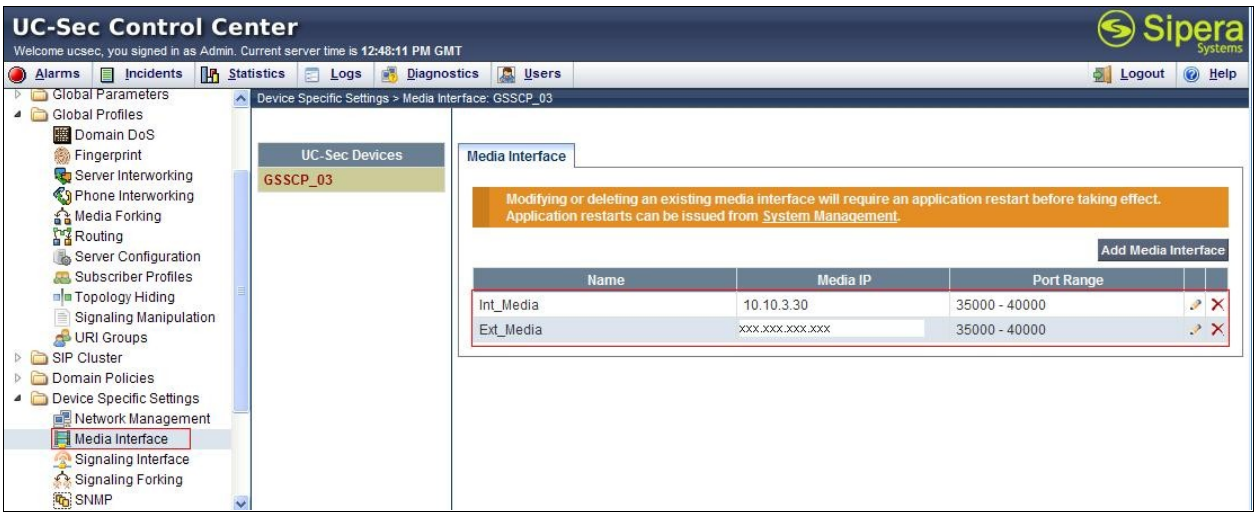
- Select **Add Signalling Interface**
- In the **Name** field enter a descriptive name for the internal signalling interface
- Select an **internal** interface IP address defined in **Section 7.2**
- Select **UDP** and **TCP** port numbers, **5060** is used internally in the lab
- Select **Add Signalling Interface**
- In the **Name** field enter a descriptive name for the external signalling interface
- Select an **external** interface IP address (not shown) defined in **Section 7.2**
- Select **UDP** and **TCP** port numbers



7.3.2. Media Interfaces

Select **Media Interface** from the menu options. The IP addresses for media can be the same as those used for signalling.

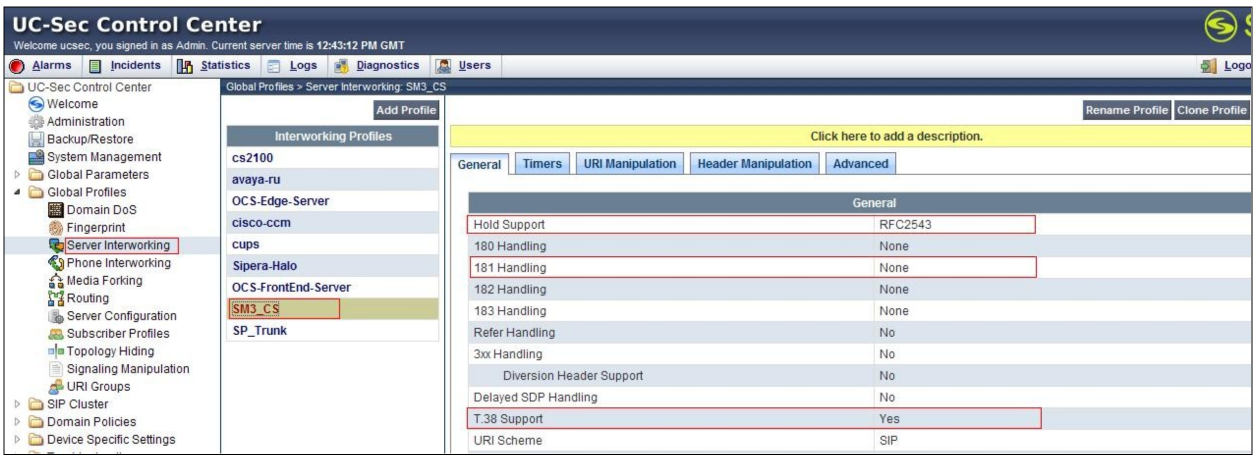
- Select **Add Media Interface**
- In the **Name** field enter a descriptive name for the internal media interface
- Select an **internal** interface IP address defined in **Section 7.2**
- Select RTP port ranges for the media path with the enterprise end-points
- Select **Add Media Interface**
- In the **Name** field enter a descriptive name for the external media interface
- Select an **external** interface IP address (not shown) defined in **Section 7.2**
- Select RTP port ranges for the media path with the Phonect SIP Trunk Service



7.4. Define Server Interworking

Server interworking is defined for the Phonect SIP Trunk Service and the NRS. To define server interworking, first click on **Global Profiles** to expand the menu options.

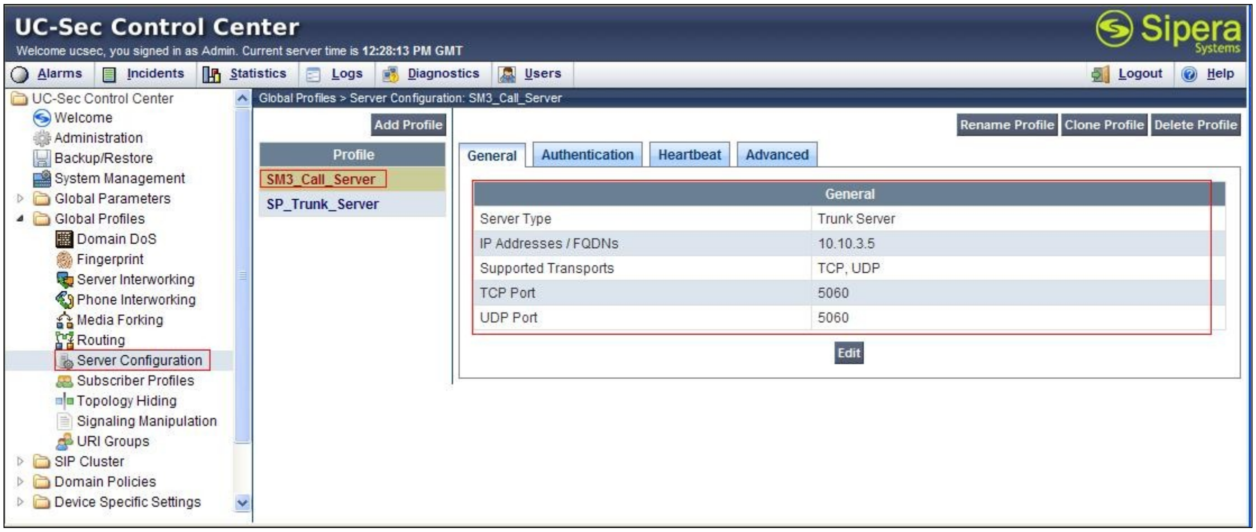
- Highlight the avaya-ru profile and select **Clone Profile**
- In the **Name** field enter a descriptive name for server interworking profile from the NRS to the Phonect SIP Trunk Service
- Click on **Finish**
- Select **Edit** and check the T.38 box, then **Next** and **Finish**
- Select **Add Profile**
- In the **Name** field enter a descriptive name for server interworking profile from the Phonect SIP Trunk Service to the NRS
- Select **Edit** and check the T.38 box
- Change the **Hold Support** RFC to **RFC2543**
- Select **Next** three times and **Finish**



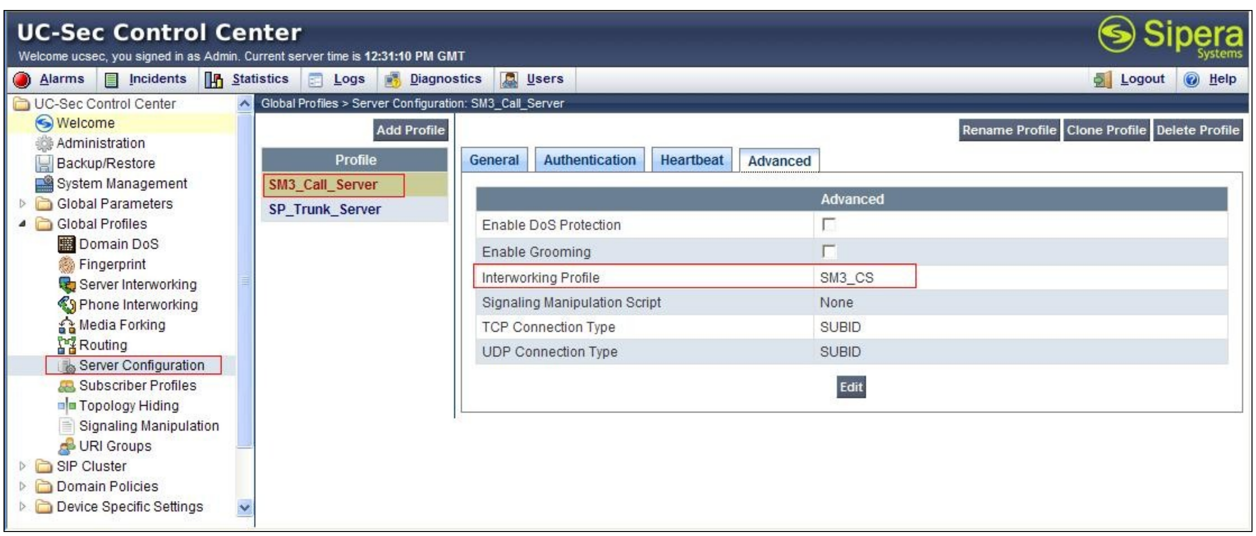
7.5. Define Servers

To define the servers and add the additional IP address for the Phonect SIP Trunk Service, click on **Global Profiles** to expand the menu. Select **Server Configuration** to add the Call Server which is the NRS**.**

- Select **Add Profile**
- In the **Name** field enter a descriptive name for the NRS
- Enter the NRS TLAN IP address in the IP address field
- Select **TCP** and **UDP** ports for SIP signaling

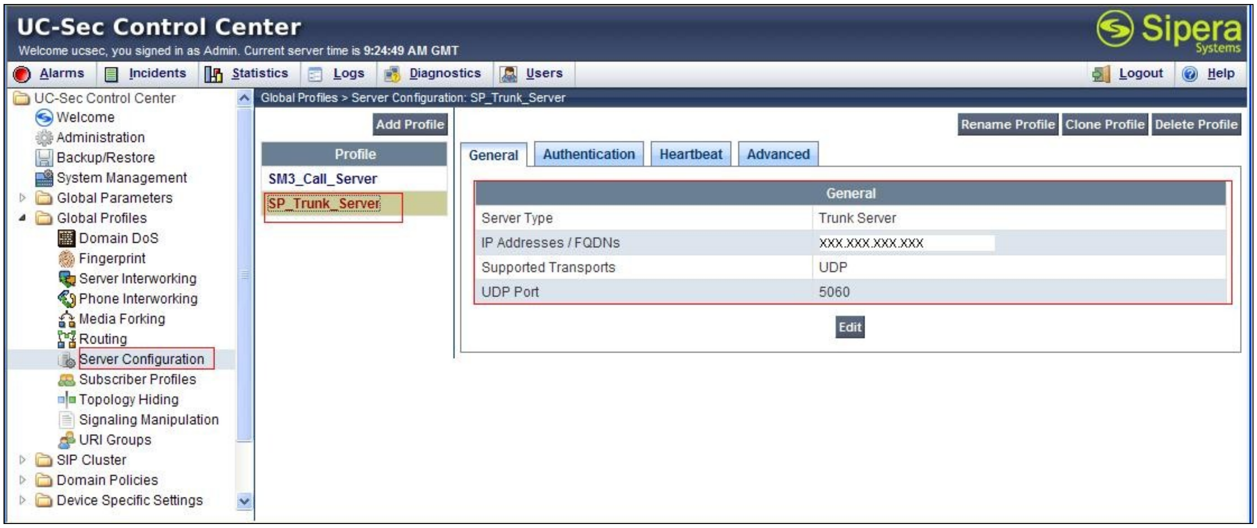


Select the server **Interworking Profile** for the call server defined in **Section 7.4.**



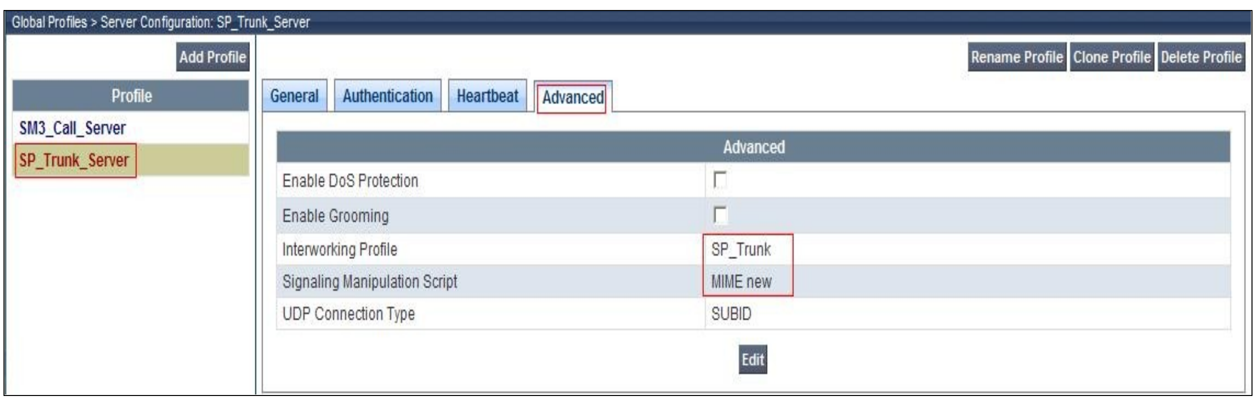
Select **Server Configuration** to add the Trunk Server which is the Phonect SIP Trunk Service**.**

- Select **Add Profile**
- In the **Name** field enter a descriptive name for the Phonect SIP Trunk Service
- Select the Phonect SIP Trunk Service IP address in the IP address field
- Select a **UDP** port for SIP signaling



Select the server **Interworking Profile** for the trunk server defined in **Section 7.4.**

- Select **MIME new** for **Signaling Manipulation Script**, this will be discussed in **Section 7.8**



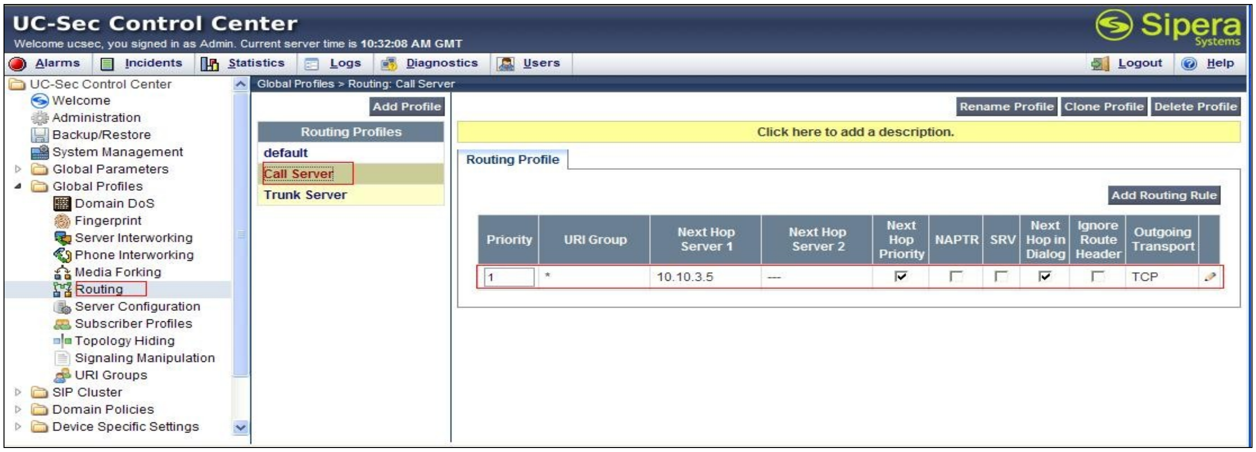
7.6. Define Routing

To define routing to the NRS, click on **Global Profiles** to expand the menu. Select **Routing.**

- Select **Add Profile**
- In the **Name** field enter a descriptive name for the NRS
- Enter the NRS TLAN IP address and port in the **Next Hop Server 1** field
- Check the **Next Hop in Dialog** box
- Select TCP for the **Outgoing Transport**

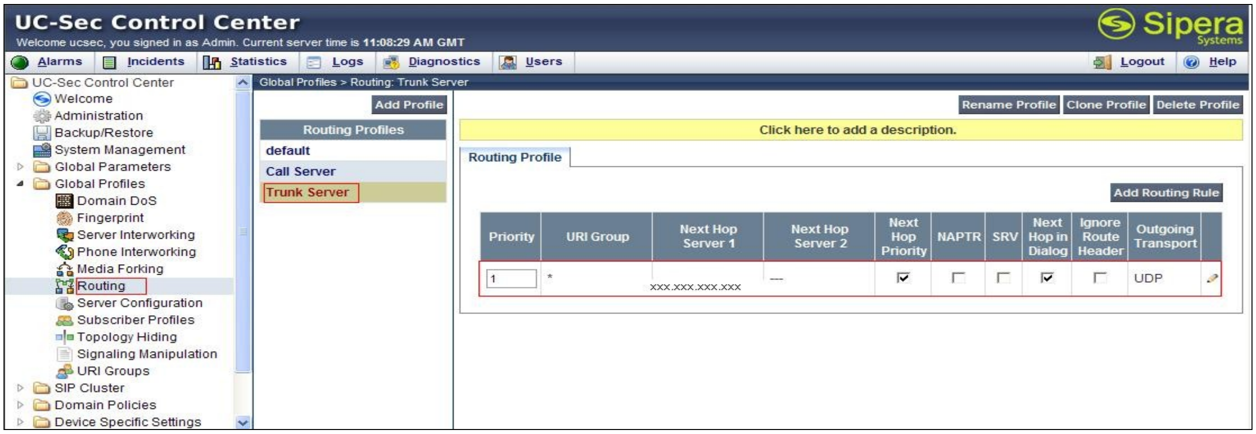
Note: Unless default port 5060 is used, this must be included in the next hop IP address.

Note: The **Next Hop in Dialog** is required to ensure that messages are sent to the next hop address regardless of the original destination. This is necessary where the Trunk Server sends messages to the address specified in the Contact header in the original INVITE message.



To define routing to the Phonect Trunk Server, create an additional profile

- Select **Add Profile**
- In the **Name** field enter a descriptive name for the Phonect SIP Trunk Service
- Enter the Phonect IP address (not shown) in the **Next Hop Server 1** field
- Select **UDP** for the **Outgoing Transport**

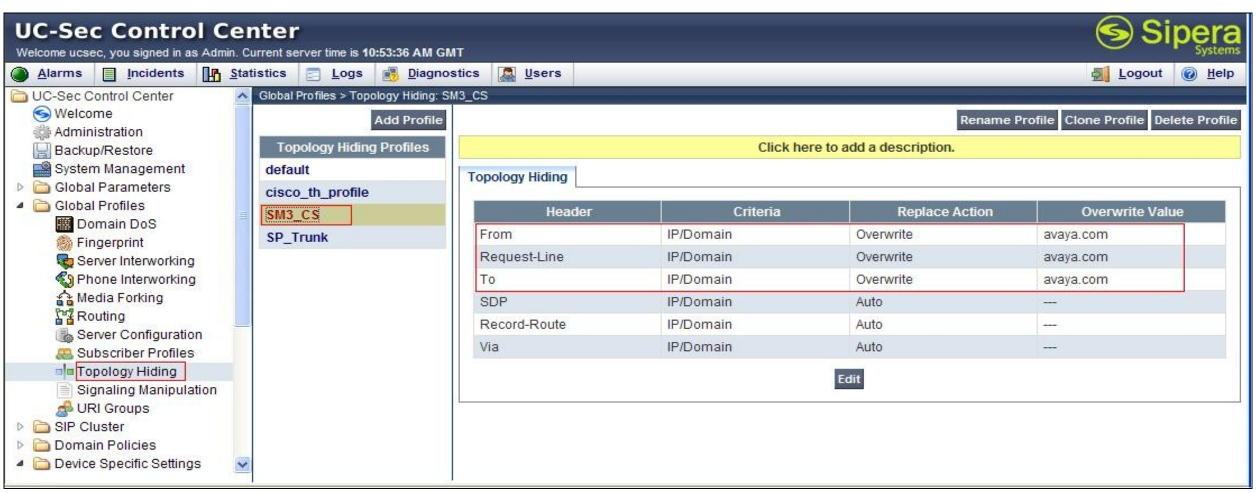


7.7. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. To define Topology Hiding for the NRS, click on **Global Profiles** to expand the menu and select **Topology Hiding**

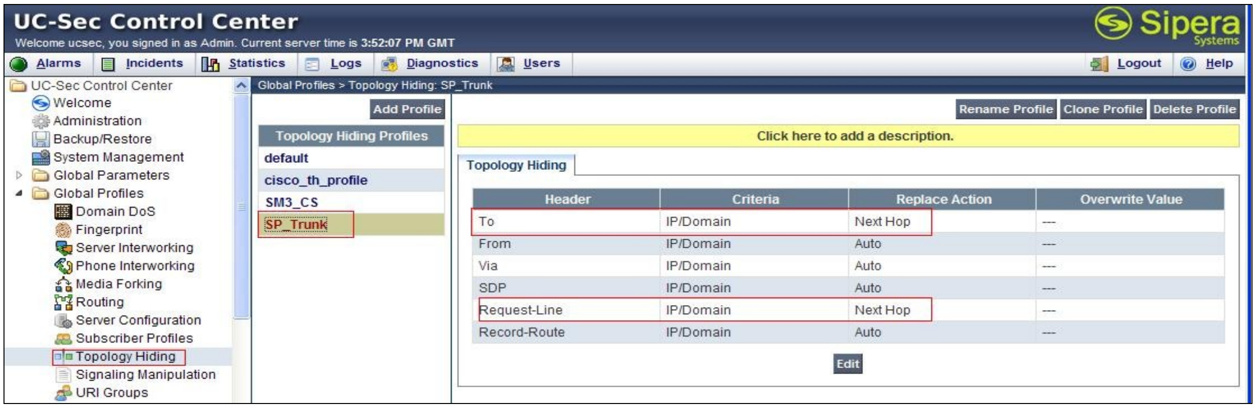
- Select **Add Profile**
- In the **Name** field enter a descriptive name for the NRS
- **Overwrite** the **From** field with a domain name for the Trunk Server, in test **avaya.com** was used
- **Overwrite** the **Request-Line** field and **To** field with a local domain name, in test **avaya.com** was used

Note: The different domain names could be used for the enterprise and the Phonect network.



To define Topology Hiding for the Phonect SIP Trunk Service, create an additional profile

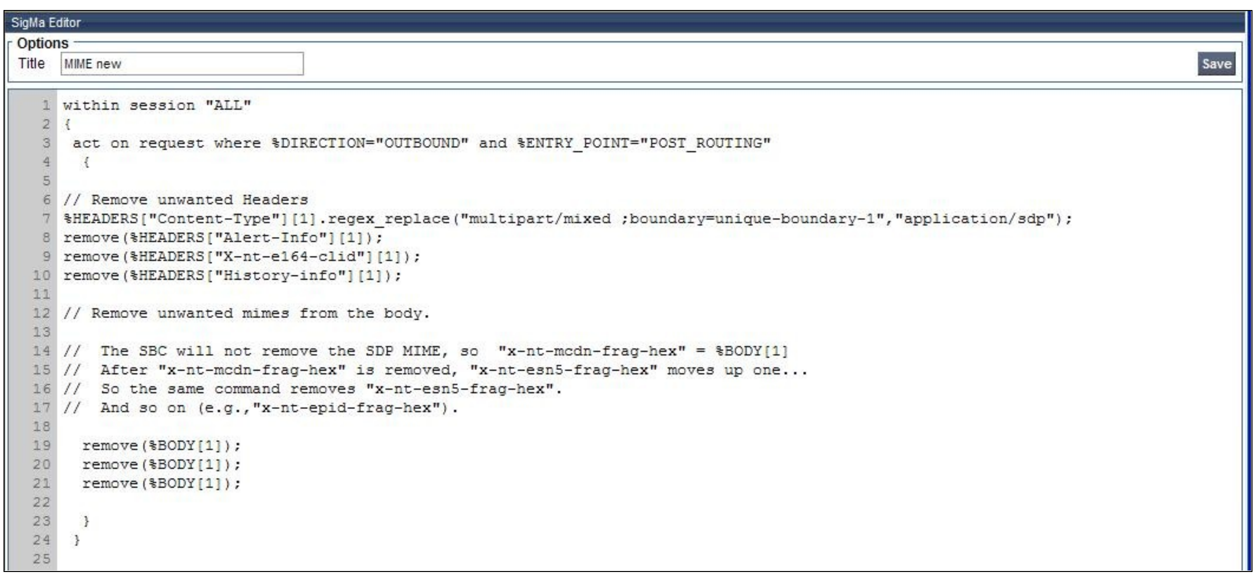
- Select **Add Profile**
- In the **Name** field enter a descriptive name for the Phonect SIP Trunk Service
- **Overwrite** the **From** field with a Replace Action Next Hop selection
- **Overwrite** the **Request-Line** field and **To** field with Replace Action Next Hop selection



7.8. Signaling Manipulation Scripts

This feature adds the ability to add, change and delete any of the headers and other information in a SIP message. During test, a script was written to remove unwanted MIME from INVITES on outbound calls.

From the lefthand menu select **Global Profiles → Signaling Manipulation** and click on **Add Script**. The script shown below was used for this test:



This script was then associated with **Server Configuration – Phonect side** defined in **Section 7.5** under Signaling Manipulation Script.

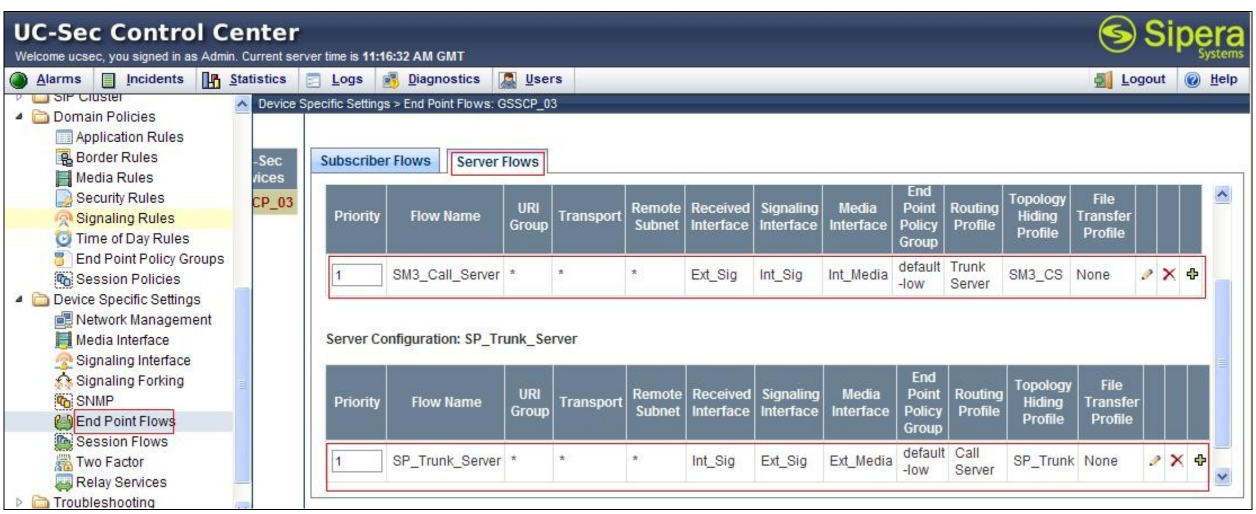
7.9. Server Flows

Server Flows combine the previously defined profiles into an outgoing flow from the NRS to the Phonect SIP Trunk Service and an incoming flow from the Phonect SIP Trunk Service to the a NRS. To define an outgoing Server Flow, click on **Device Specific Settings** to expand the menu and select **End Point Flows.**

- Click on the **Server Flows** tab
- Select **Add Flow**
- In the **Name** field enter a descriptive name for the outgoing server flow
- In the **Received Interface** field, select the SIP signalling interface for the Phonect SIP Trunk Service
- In the **Signalling Interface** field, select the SIP signalling interface for the NRS
- In the **Media Interface** field, select the media interface for the NRS
- In the **End Point Policy Group** field, select the default-low End Point Policy Group
- In the **Routing Profile** field, select the routing profile of the Phonect SIP Trunk Service
- In the **Topology Hiding Profile** field, select the topology hiding profile of the NRS

An incoming Server Flow is defined as a reversal of the outgoing Server Flow

- Select **Add Flow**
- In the **Name** field enter a descriptive name for the incoming server flow
- In the **Received Interface** field, select the SIP signalling interface for the NRS
- In the **Signalling Interface** field, select the SIP signalling interface for the Phonect SIP Trunk Service
- In the **Media Interface** field, select the media interface for the Phonect SIP Trunk Service
- In the **End Point Policy Group** field, select the default-low End Point Policy Group
- In the **Routing Profile** field, select the routing profile of the NRS
- In the **Topology Hiding Profile** field, select the topology hiding profile of the Phonect SIP Trunk Service



8. Phonect SIP Service Provider Configuration

The setup for the use of Phonect is by using the SIP trunk with an authenticated service. The configuration of Phonect's authentification service 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 Phonect's equipment and system configuration please contact an authorised Phonect representative.

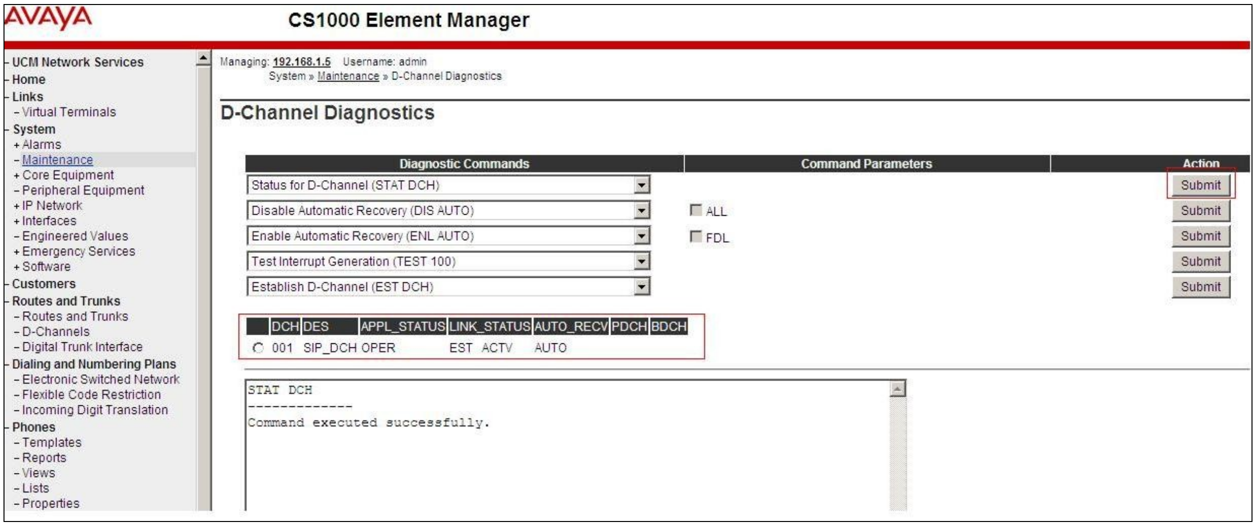
9. Verification

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**



10. Conclusion

These Application Notes describe the configuration necessary to connect the Avaya Communication Server 1000E, Avaya Network Routing Server and Avaya Session Border Controller Advanced for Enterprise to Phonect SIP Service. Interoperability testing of the sample configuration was completed with successful results for the Phonect SIP Trunk with observations which are detailed in **Section 2.2**.

11. Additional References

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

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

[2] Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02, available at http://support.avaya.com

[3] 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

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

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

[6] E-SBC (Avaya Session Border Controller Advanced for Enterprise) Administration Guide, November 2011

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

Appendix A – Avaya Communication Server 1000E Software

Avaya Communication Server 1000E call server patches and plug ins

```
TID: 46379

VERSION 4121

System type is - Communication Server 1000E/CP PM Linux

CP PM - Pentium M 1.4 GHz

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

RELEASE 7
ISSUE 50 Q  +
IDLE_SET_DISPLAY Avaya 7.5
DepList 1: core Issue: 01(created: 2012-01-10 16:47:54 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2012-01-24 11:17:37(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-01-11 11:07:13(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

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

Avaya Communication Server 1000E call server deplists

```
VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2012-01-10 16:47:54 (est)) ALTERED

IN-SERVICE PEPS
PAT# CR #           PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi00891626     ISS1:1OF1       p31051_1  01/02/2012  p31051_1.cpl   YES
001  wi00951837     ISS1:1OF1       p31485_1  01/02/2012  p31485_1.cpl   NO
002  wi00946477     ISS1:1OF1       p31426_1  01/02/2012  p31426_1.cpl   NO
003  wi00906163     ISS1:1OF1       p31205_1  01/02/2012  p31205_1.cpl   NO
004  wi00962211     ISS1:1OF1       p31580_1  01/02/2012  p31580_1.cpl   NO
005  wi00877592     ISS1:1OF1       p30880_1  01/02/2012  p30880_1.cpl   NO
006  wi00839134     ISS1:1OF1       p30698_1  01/02/2012  p30698_1.cpl   YES
007  wi00958682     ISS1:1OF1       p31540_1  01/02/2012  p31540_1.cpl   NO
008  wi00868729     ISS1:1OF1       p31163_1  01/02/2012  p31163_1.cpl   NO
009  wi00886321     ISS1:1OF1       p31009_1  01/02/2012  p31009_1.cpl   NO
010  wi00946282     ISS1:1OF1       p31204_1  01/02/2012  p31204_1.cpl   NO
011  wi00841980     ISS1:1OF1       p30618_1  01/02/2012  p30618_1.cpl   NO
012  wi00946681     ISS1:1OF1       p31428_1  01/02/2012  p31428_1.cpl   NO
013  wi00945533     ISS1:1OF1       p31421_1  01/02/2012  p31421_1.cpl   YES
014  wi00843623     ISS1:1OF1       p30731_1  01/02/2012  p30731_1.cpl   YES
015  wi00958776     ISS1:1OF1       p31542_1  01/02/2012  p31542_1.cpl   YES
016  wi00857362     ISS1:1OF1       p30782_1  01/02/2012  p30782_1.cpl   NO
017  wi00865477     ISS1:1OF1       p30893_1  01/02/2012  p30893_1.cpl   YES
018  wi00879526     ISS1:1OF1       p31007_1  01/02/2012  p31007_1.cpl   NO
019  wi00894243     ISS1:1OF1       p31087_1  01/02/2012  p31087_1.cpl   NO
020  wi00890475     p30952          p31048_1  01/02/2012  p31048_1.cpl   NO
021  WI00927300     ISS1:1OF1       p30999_1  01/02/2012  p30999_1.cpl   NO
022  wi00856991     ISS1:1OF1       p17588_1  01/02/2012  p17588_1.cpl   NO
023  wi00688381     ISS1:1OF1       p30104_1  01/02/2012  p30104_1.cpl   NO
024  wi00881777     ISS1:1OF1       p25747_1  01/02/2012  p25747_1.cpl   NO
025  WI00853473     ISS1:1OF1       p30625_1  01/02/2012  p30625_1.cpl   NO
026  wi00855423     ISS1:1OF1       p31328_1  01/02/2012  p31328_1.cpl   YES
027  wi00943172     ISS1:1OF1       p31402_1  01/02/2012  p31402_1.cpl   NO
```

028	wi00865477	ISS1:1OF1	p30898_1	01/02/2012	p30898_1.cpl	YES
029	wi00850521	ISS1:1OF1	p30709_1	01/02/2012	p30709_1.cpl	YES
030	wi00898327	ISS1:1OF1	p31136_1	01/02/2012	p31136_1.cpl	NO
031	wi00871739	ISS1:1OF1	p30856_1	01/02/2012	p30856_1.cpl	NO
032	wi00853031	ISS1:1OF1	p30531_1	01/02/2012	p30531_1.cpl	NO
033	wi00839821	ISS1:1OF1	p30619_1	01/02/2012	p30619_1.cpl	NO
034	wi00854130	ISS1:1OF1	p30443_1	01/02/2012	p30443_1.cpl	NO
035	wi00871969	ISS1:1OF1	p30768_1	01/02/2012	p30768_1.cpl	NO
036	wi00952381	ISS1:1OF1	p31410_1	01/02/2012	p31410_1.cpl	NO
037	wi00946876	ISS1:1OF1	p31430_1	01/02/2012	p31430_1.cpl	NO
038	wi00962557	ISS1:1OF1	p31581_1	01/02/2012	p31581_1.cpl	NO
039	wi00833910	ISS2:1OF1	p30492_2	01/02/2012	p30492_2.cpl	NO
040	wi00903085	ISS1:1OF1	p31164_1	01/02/2012	p31164_1.cpl	NO
041	wi00875425	ISS1:1OF1	p30943_1	01/02/2012	p30943_1.cpl	NO
042	wi00862574	iss1:1of1	p30870_1	01/02/2012	p30870_1.cpl	NO
043	wi00859499	ISS1:1OF1	p30694_1	01/02/2012	p30694_1.cpl	NO
044	wi00925208	ISS1:1OF1	p30986_1	01/02/2012	p30986_1.cpl	NO
045	wi00877442	ISS1:1OF1	p30844_1	01/02/2012	p30844_1.cpl	NO
046	wi00900668	ISS1:1OF1	p30456_1	01/02/2012	p30456_1.cpl	NO
047	wi00867905	ISS1:1OF1	p30640_1	01/02/2012	p30640_1.cpl	NO
048	wi00879322	ISS1:1OF1	p30954_1	01/02/2012	p30954_1.cpl	NO
049	wi00865477	ISS1:1OF1	p30895_1	01/02/2012	p30895_1.cpl	YES
050	wi00951925	ISS1:1OF1	p31486_1	01/02/2012	p31486_1.cpl	NO
051	wi00865477	ISS1:1OF1	p30894_1	01/02/2012	p30894_1.cpl	YES
052	wi00865477	ISS1:1OF1	p30897_1	01/02/2012	p30897_1.cpl	YES
053	wi00865477	ISS1:1OF1	p30892_1	01/02/2012	p30892_1.cpl	YES
054	wi00908933	ISS1:1OF1	p31239_1	01/02/2012	p31239_1.cpl	NO
055	wi00931028	ISS1:1OF1	p31354_1	01/02/2012	p31354_1.cpl	YES
056	wi00932948	ISS1:1OF1	p31077_1	01/02/2012	p31077_1.cpl	NO
057	wi00869695	ISS1:1OF1	p30654_1	01/02/2012	p30654_1.cpl	NO
058	wi00838073	ISS1:1OF1	p30588_1	01/02/2012	p30588_1.cpl	NO
059	wi00852365	ISS1:1OF1	p30707_1	01/02/2012	p30707_1.cpl	NO
060	wi00927321	ISS1:1OF1	p31286_1	01/02/2012	p31286_1.cpl	YES
061	wi00937114	ISS1:1OF1	p31310_1	01/02/2012	p31310_1.cpl	NO
062	wi00877367	ISS1:1OF1	p30534_1	01/02/2012	p30534_1.cpl	NO
063	wi00900096	ISS1:1OF1	p31006_1	01/02/2012	p31006_1.cpl	NO
064	wi00905660	ISS1:1OF1	p27968_1	01/02/2012	p27968_1.cpl	NO
065	wi00925141	ISS1:1OF1	p30802_1	01/02/2012	p30802_1.cpl	NO
066	wi00943748	ISS1:1OF1	p31516_1	01/02/2012	p31516_1.cpl	NO
067	wi00827950	ISS2:1OF1	p30471_2	01/02/2012	p30471_2.cpl	NO
068	wi00937119	ISS1:1OF1	p28005_1	01/02/2012	p28005_1.cpl	NO
069	wi00836981	ISS1:1OF1	p30613_1	01/02/2012	p30613_1.cpl	NO
070	wi00961267	ISS1:1OF1	p30288_1	01/02/2012	p30288_1.cpl	NO
071	wi00936714	ISS1:1OF1	p31379_1	01/02/2012	p31379_1.cpl	NO
072	wi00906022	ISS1:1OF1	p31202_1	01/02/2012	p31202_1.cpl	NO
073	wi00852389	ISS1:1OF1	p30641_1	01/02/2012	p30641_1.cpl	NO
074	wi00857566	ISS1:1OF1	p30766_1	01/02/2012	p30766_1.cpl	NO
075	wi00932204	ISS2:1OF1	p31305_2	01/02/2012	p31305_2.cpl	NO
077	wi00865477	ISS1:1OF1	p30890_1	01/02/2012	p30890_1.cpl	YES
078	wi00873382	ISS1:1OF1	p30832_1	01/02/2012	p30832_1.cpl	NO
079	wi00948274	ISS1:1OF1	p31365_1	01/02/2012	p31365_1.cpl	NO
080	wi00923899	ISS1:1OF1	p31270_1	01/02/2012	p31270_1.cpl	NO
081	wi00856410	ISS1:1OF1	p30749_1	01/02/2012	p30749_1.cpl	NO
082	wi00854415	ISS1:1OF1	p30593_1	01/02/2012	p30593_1.cpl	NO
083	wi00896394	ISS1:1OF1	p30807_1	01/02/2012	p30807_1.cpl	NO
084	wi00826075	ISS1:1OF1	p30452_1	01/02/2012	p30452_1.cpl	NO
085	wi00863876	ISS1:1OF1	p30787_1	01/02/2012	p30787_1.cpl	NO
086	wi00880386	ISS1:1OF1	p30977_1	01/02/2012	p30977_1.cpl	NO
087	wi00840590	ISS1:1OF1	p30767_1	01/02/2012	p30767_1.cpl	NO
088	wi00949627	ISS1:1OF1	p31462_1	01/02/2012	p31462_1.cpl	NO
089	wi00842409	ISS1:1OF1	p30621_1	01/02/2012	p30621_1.cpl	NO
090	wi00865477	ISS1:1OF1	p30896_1	01/02/2012	p30896_1.cpl	YES
091	wi00897096	ISS1:1OF1	p30676_1	01/02/2012	p30676_1.cpl	NO
092	wi00899584	ISS1:1OF1	p30809_1	01/02/2012	p30809_1.cpl	NO
093	wi00907707	ISS1:1OF1	p31228_1	01/02/2012	p31228_1.cpl	NO
094	wi00949273	ISS1:1OF1	p31411_1	01/02/2012	p31411_1.cpl	NO
095	wi00839255	ISS1:1OF1	p30591_1	01/02/2012	p30591_1.cpl	NO
096	wi00921340	ISS1:1OF1	p31266_1	01/02/2012	p31266_1.cpl	NO
097	wi00903369	ISS1:1OF1	p31165_1	01/02/2012	p31165_1.cpl	NO
098	wi00875701	ISS1:1OF1	p30942_1	01/02/2012	p30942_1.cpl	NO

```
099  wi00884699      ISS1:1OF1       p31000_1  01/02/2012  p31000_1.cpl   YES
100  wi00834382      ISS1:1OF1       p30548_1  01/02/2012  p30548_1.cpl   NO
101  wi00960133      ISS2:1OF1       p31557_2  01/02/2012  p31557_2.cpl   NO
102  wi00929140      ISS1:1OF1       p31284_1  01/02/2012  p31284_1.cpl   NO
103  wi00948931      ISS1:1OF1       p31407_1  01/02/2012  p31407_1.cpl   NO
104  wi00887744      ISS2:1OF1       p31026_2  01/02/2012  p31026_2.cpl   NO
105  wi00905600      ISS1:1OF1       p31201_1  01/02/2012  p31201_1.cpl   NO
106  wi00869243      ISS1:1OF1       p30848_1  01/02/2012  p30848_1.cpl   NO
107  WI00854150      ISS1:1OF1       p30468_1  01/02/2012  p30468_1.cpl   NO
108  wi00897176      ISS1:1OF1       p30418_1  01/02/2012  p30418_1.cpl   NO
109  wi00903381      ISS1:1OF1       p30421_1  01/02/2012  p30421_1.cpl   NO
110  wi00959854      ISS1:1OF1       p31556_1  01/02/2012  p31556_1.cpl   NO
111  wi00908598      ISS1:1OF1       p31235_1  01/02/2012  p31235_1.cpl   NO
112  wi00903437      ISS1:1OF1       p31167_1  01/02/2012  p31167_1.cpl   NO
113  wi00900766      ISS1:1OF1       p31159_1  01/02/2012  p31159_1.cpl   NO
114  wi00946558      ISS1:1OF1       p31358_1  01/02/2012  p31358_1.cpl   NO
115  wi00932958      ISS1:1OF1       p31115_1  01/02/2012  p31115_1.cpl   NO
116  wi00895090      ISS1:1OF1       p31105_1  01/02/2012  p31105_1.cpl   NO
117  wi00824257      ISS1:1OF1       p30447_1  01/02/2012  p30447_1.cpl   NO
118  wi00895181      ISS1:1OF1       p31106_1  01/02/2012  p31106_1.cpl   NO
119  WI00928455      ISS1:1OF1       p31297_1  01/02/2012  p31297_1.cpl   NO
120  wi00832106      ISS1:1OF1       p30550_1  01/02/2012  p30550_1.cpl   NO
121  wi00953900      ISS1:1OF1       p31494_1  01/02/2012  p31494_1.cpl   NO
122  wi00942734      ISS1:1OF1       p31409_1  01/02/2012  p31409_1.cpl   NO
123  wi00898200      ISS1:1of1       p31274_1  01/02/2012  p31274_1.cpl   NO
124  wi00882293      ISS1:1OF1       p31010_1  01/02/2012  p31010_1.cpl   NO
125  WI00843571      ISS1:1OF1       p30627_1  01/02/2012  p30627_1.cpl   NO
126  wi00835294      ISS1:1OF1       p30565_1  01/02/2012  p30565_1.cpl   NO
127  WI00836292      ISS1:1OF1       p30554_1  01/02/2012  p30554_1.cpl   NO
128  WI00900213      ISS1:1OF1       p30656_1  01/02/2012  p30656_1.cpl   NO
129  wi00921295      ISS1:1OF1       p31265_1  01/02/2012  p31265_1.cpl   NO
130  wi00957141      ISS1:1OF1       p31579_1  01/02/2012  p31579_1.cpl   NO
131  WI00836334      ISS1:1OF1       p30481_1  01/02/2012  p30481_1.cpl   NO
132  wi00858335      ISS1:1OF1       p30819_1  01/02/2012  p30819_1.cpl   NO
133  wi00859123      ISS1:1OF1       p30648_1  01/02/2012  p30648_1.cpl   NO
134  wi00959820      ISS1:1OF1       p31562_1  01/02/2012  p31562_1.cpl   NO
135  wi00905297      ISS1:1OF1       p31195_1  01/02/2012  p31195_1.cpl   NO
136  wi00907697      ISS1:1OF1       p31227_1  01/02/2012  p31227_1.cpl   NO
137  wi00951427      ISS1:1OF1       p31478_1  01/02/2012  p31478_1.cpl   NO
138  wi00883604      ISS1:1OF1       p30973_1  01/02/2012  p30973_1.cpl   NO
139  wi00962955      ISS1:1OF1       p31585_1  01/02/2012  p31585_1.cpl   NO
140  wi00860279      ISS1:1OF1       p30789_1  01/02/2012  p30789_1.cpl   NO
141  wi00909476      ISS1:1OF1       p31340_1  01/02/2012  p31340_1.cpl   NO
142  wi00925218      ISS1:1OF1       p30675_1  01/02/2012  p30675_1.cpl   NO
143  wi00836182      ISS1:1OF1       p30450_1  01/02/2012  p30450_1.cpl   NO
144  wi00841273      ISS1:1OF1       p30713_1  01/02/2012  p30713_1.cpl   NO
145  WI00889786      ISS1:1OF1       p30750_1  01/02/2012  p30750_1.cpl   NO
146  wi00894443      ISS1:1OF1       p31093_1  01/02/2012  p31093_1.cpl   NO
147  wi00896420      ISS1:1OF1       p30867_1  01/02/2012  p30867_1.cpl   NO
148  wi00941500      ISS1:1OF1       p31394_1  01/02/2012  p31394_1.cpl   NO
149  wi00950592      ISS1:1OF1       p31499_1  01/02/2012  p31499_1.cpl   NO
150  wi00927678      ISS1:1OF1       p31399_1  01/02/2012  p31399_1.cpl   NO
151  wi00930864      ISS1:1OF1       p31325_1  01/02/2012  p31325_1.cpl   NO
152  wi00957252      ISS1:1OF1       p31530_1  01/02/2012  p31530_1.cpl   NO
153  wi00880836      ISS1:1OF1       p30976_1  01/02/2012  p30976_1.cpl   NO
154  wi00865477      ISS1:1OF1       p30891_1  01/02/2012  p30891_1.cpl   YES
155  wi00896680      ISS1:1OF1       p30357_1  01/02/2012  p30357_1.cpl   NO
156  wi00856702      ISS1:1OF1       p30573_1  01/02/2012  p30573_1.cpl   NO
157  wi00897082      ISS1:1OF1       p31124_1  01/02/2012  p31124_1.cpl   NO
158  wi00853178      ISS1:1OF1       p30719_1  01/02/2012  p30719_1.cpl   NO
159  wi00938555      ISS1:1OF1       p30881_1  01/02/2012  p30881_1.cpl   YES
160  WI00839794      ISS1:1OF1       p28647_1  01/02/2012  p28647_1.cpl   NO
MDP>LAST SUCCESSFUL MDP REFRESH :2012-01-24 11:17:37(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-01-11 11:07:13(est)
```

Avaya Communication Server 1000E signaling server service updates

```
Product Release: 7.50.17.00
In system patches: 1
PATCH#  NAME       IN_SERVICE   DATE      SPECINS  TYPE   RPM
20      p30260_1   Yes          31/01/12  NO       FRU    cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 21
PATCH#  IN_SERVICE   DATE       SPECINS    REMOVABLE   NAME
0       Yes          20/01/12   NO         YES         cs1000-linuxbase-7.50.17.16-5.i386.000
1       Yes          20/01/12   NO         YES         cs1000-baseWeb-7.50.17.16-1.i386.001
2       Yes          20/01/12   NO         YES         cs1000-patchWeb-7.50.17.16-2.i386.000
3       Yes          20/01/12   NO         YES         cs1000-dbcom-7.50.17-02.i386.000
4       Yes          20/01/12   NO         yes         cs1000-sps-7.50.17.16-01.i386.000
5       Yes          20/01/12   NO         YES         cs1000-shared-pbx-7.50.17.16-1.i386.000
6       Yes          20/01/12   NO         YES         cs1000-kcv-7.50.17.16-1.i386.000
7       Yes          20/01/12   NO         YES         cs1000-nrsmWebService-7.50.17.16-1.i386.000
8       Yes          20/01/12   NO         YES         cs1000-dmWeb-7.50.17.16-1.i386.000
9       Yes          20/01/12   NO         YES         cs1000-nrsm-7.50.17.16-2.i386.000
10      Yes          20/01/12   NO         YES         cs1000-ipsec-7.50.17.16-1.i386.000
11      Yes          20/01/12   NO         YES         cs1000-ftrpkg-7.50.17.16-5.i386.000
12      Yes          20/01/12   NO         YES         cs1000-tps-7.50.17.16-8.i386.000
13      Yes          20/01/12   NO         YES         cs1000-csmWeb-7.50.17.16-2.i386.000
14      Yes          20/01/12   NO         YES         ipsec-tools-0.6.5-14.el5.3_avaya_1.i386.000
15      Yes          20/01/12   NO         YES         spiritAgent-6.1-1.0.0.108.208.i386.000
16      Yes          20/01/12   NO         YES         cs1000-EmCentralLogic-7.50.17.16-1.i386.000
17      Yes          20/01/12   NO         YES         cs1000-Jboss-Quantum-7.50.17.16-8.i386.000
18      Yes          20/01/12   NO         YES         cs1000-bcc-7.50.17.16-31.i386.000
19      Yes          20/01/12   NO         YES         cs1000-emWeb_6-0-7.50.17.16-9.i386.000
21      Yes          31/01/12   NO         YES         cs1000-vtrk-7.50.17.16-36TMP.i386.000
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Avaya Communication Server 1000E system software

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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     [patched]
   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               n/a         [patched]
Application configuration: CS+SS+NRS+EM
Packages:
CS+SS+NRS+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
   nrsm                         7.50.17     [patched]
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nrsmWebService                 7.50.17     [patched]
managedElementWebService       7.50.17
EmConfig                       7.50.17
emWeb_6-0                      7.50.17     [patched]
emWebLocal_6-0                 7.50.17
csmWeb                         7.50.17     [patched]
bcc                            7.50.17     [patched]
ftrpkg                         7.50.17     [patched]
cs1000WebService_6-0           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.