Avaya Solution & Interoperability Test Lab

Application notes for Paetec (Broadsoft platform) Communication System with Avaya™ Communication Server 1000 release 6.0

Abstract

These Application Notes describe a solution comprised of Avaya™ Communication Server 1000E Release 6.0 and Paetec Communication SIP Trunk Product. The Primary focus of testing is the system verification of SIP trunk interoperability which includes the call scenarios such as basic call, call forward (all calls, busy, no answer), call transfer (blind and consult) and conference. Calls should be placed in both directions and should involve various set types

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

This document provides a typical network deployment of Communication Server 1000 (CS1000) utilizing the Paetec Communication SIP Trunking product offering. This document should serve as general guideline only, since it is not possible to document every possible variation of configuration. Further information may be obtained from your Nortel support representative. The CS1000E system is configured as a SIP gateway endpoint on the Paetec Communication network. The enterprise customer will require an additional signaling server for each SIP gateway that will be deployed as SIP trunking to the carrier. In the diagram shown below, the signaling server is shown as the onboard CPPM Cores option, but it can also be the outboard, rack-mounted 1U server. The CS1000, in this configuration, does not use SIP Redirect or Proxy for Carrier SIP trunking, the SIP Virtual Gateway is simply provisioned with the SBC as the static SIP endpoint of the SIP Trunk.

1.1. Interoperability Compliance Testing

System verification testing of SIP Trunking between CS1000 Rel. 6.0 and Carrier switch

- General call processing between systems including:
  - Codec/ptime negotiation and transcoding (G.711 u-law and G.729 verification / 20ms)
  - Hold/Retrieve on both ends
  - CLID displayed
  - Ringback tone
  - Speech path
  - Dialing plan support
  - Advanced features (Call on Mute, Call Park, Call Waiting, use Feature Access Code)
  - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- Fax G711 Pass Through (Fax T38 does not support)
- MedSec is not supported.
- DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server (hosted on Nortel system)
- Early Media Transmission
- Inter-office tandem Call

1.2. Caveats

• The Fax/Modem pass through feature provides a modem pass through allowed (MPTA) class of service (CLS) for an analog phone TN. MPTA CLS dedicates an analog phone TN to a modem or a Fax machine terminal. A connection that initiates from the dedicated TN, and/or calls that terminate at the dedicated TN through a Digital Signal Processor (DSP), use a G711 NO VAD codec on the Call Server. To ensure proper functioning of the MPTA CLS, the Enable Modem/Fax pass through mode check box must be selected in the Gateways section of Element Manager. This check box is selected by default in Element Manager.

• The packet interval for G.711 codec is set to 20 ms in MPT. The maximum speed supported for modem and fax is 33.6 Kb/s. This limit is imposed by the analogue line card. When MPTA CLS is configured on a TN, the T.38 protocol is no longer supported for that particular TN.

1.3. Dependencies

. CS1000 R6.0 software and implementation of latest patches

. Paetec Communications provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

1.4. Support

For technical support on Paetec Communication system, please contact Paetec technical support at:

- Toll Free: 800.967.2233
- E-mail: datatac@paetec.com

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the Communication Server 1000E and Paetec Communication System. This configuration is for a single Communication Server1000E deployment

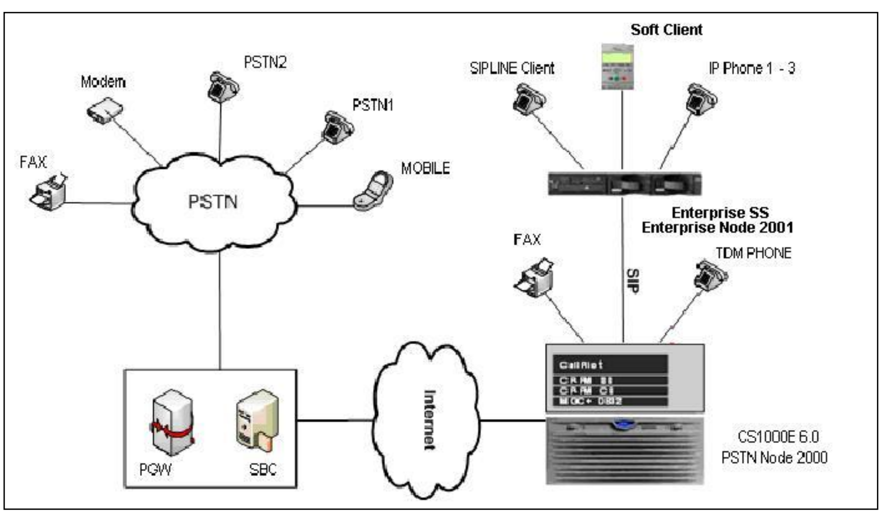

Figure 1- Net**twork diagram for Nortel-Paetec LAB setup**

Figure 2 is the deployment option for 2 or more Communication Server of 1000E with the Paetec communication system.

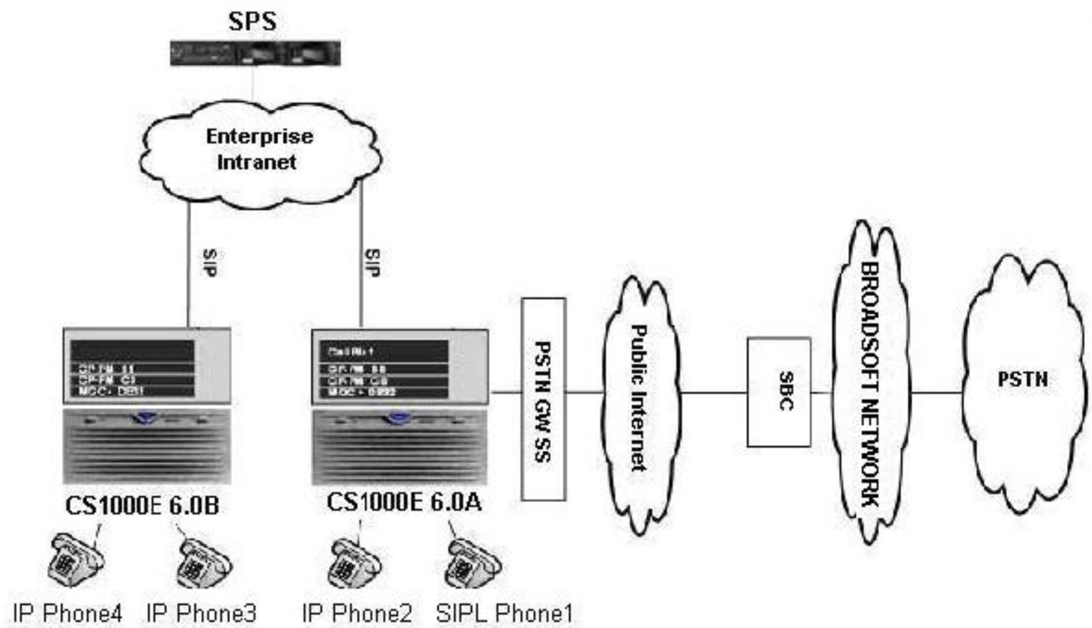

Figure 2 - Network topology for Multi-System configuration for Tandem Calls

The following assumptions were made for this lab test configuration:

1. CS1000 R6.0 software and implementation of latest patches
2. Paetec Communications provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

All test scenarios involving the establishment of calls will assume the following activities:

1. Calls will be checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing will be checked.
3. Calls will be checked in both hands-free and handset mode due to internal Nortel requirement.
4. Calls will be checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved will be checked for consistent and expected CLID, name and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system will be observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window will be open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.

8. Speech path and display checked before and after calls are put on/off hold from each end.
9. Applicable of files will be screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Nortel PBX files.
10. Calls will be checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends

3. Equipment and Software Validated

Additional software and patch lineup for the configuration is as follows:

Call Server: 6.00R plus latest DEPLIST
Signaling Server: SSE 6.00.18 plus latest DEPLIST

Patch ID	Issue	Title	Notes
MPLR28415	1	Ringback tone and speech path support in slow start CFNA scenarios	
MPLR28774	1	Delete element removes all elements-services mapping of associateroles	
MPLR28797	1	Unable to access overlays on inactive core when in split mode with UCM	
MPLR27408	1	SIP: Disable SIP Session Timer on CS1K.	
MPLR25946	1	SIP GW patch to remove outbound MCDN from SIP messaging	
MPLR22968	1	Replace domain population in the FROM field	
MPLR25529	1	PI: SIP: Partial support of DIVERSION	

Hardware system requirement and theirs soft/loadware version

System	Software/Loadware Version
Nortel CS1000E 6.0 (CPPM)	• Call Server: 6.00R • Signaling Server: 6.00.18
Nortel phones	• 2002 p2: 0604DCJ (Unistim) • 2004 p2: 0604DCJ (Unistim) • 1140: 0625C6O (Unistim) • 1120: 0624C6O (Unistim) • 2007: 0621C6M (Unistim) • 1220: 062AC6O (Unistim) • SIP 1140 i00v142 • SIP 1120 • SMC3456: Version 2.6 - RC14 build 53715
Paetec Broadsoft platform	• Release 14 SP9
Gateway	• Lucent LCS 3.14.1.5

4. Configure the Avaya Communication Server 1000E

4.1. Element Manager Configuration

4.1.1 Configure IP in CS1000 network

This section describes the steps for creating Node ID (1000) in CS 1000 network. Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Save...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 1000 was added to be configured as the SIP gateway to the carrier services.

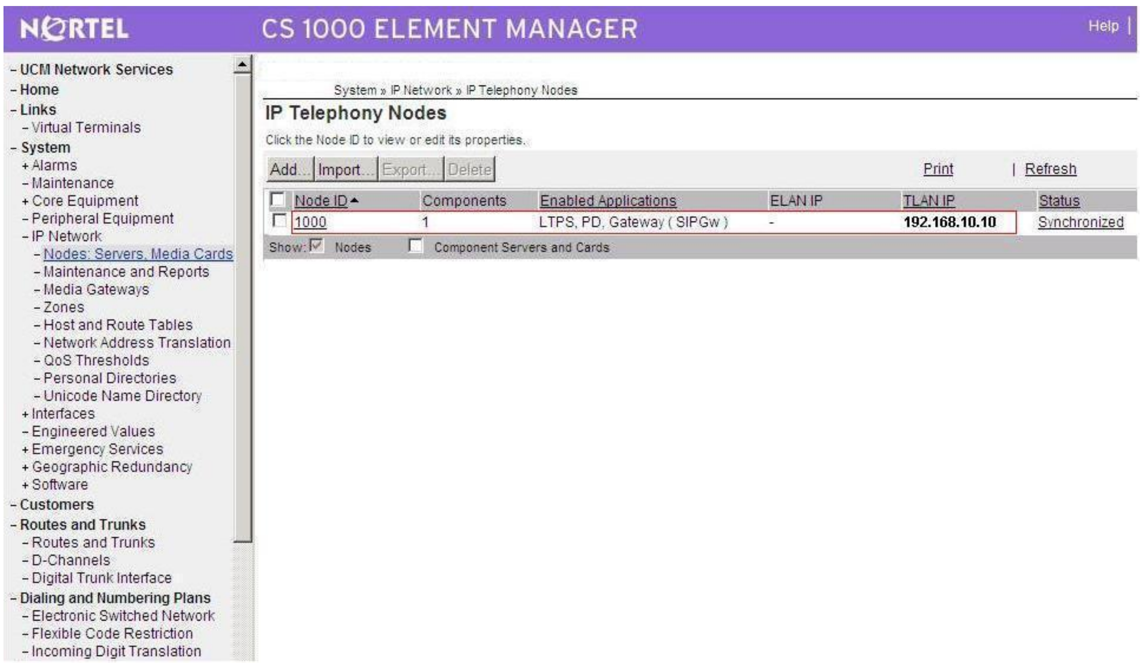

Figure 3 – Adding a node

Figure 4 describes the Call server IP configuration:

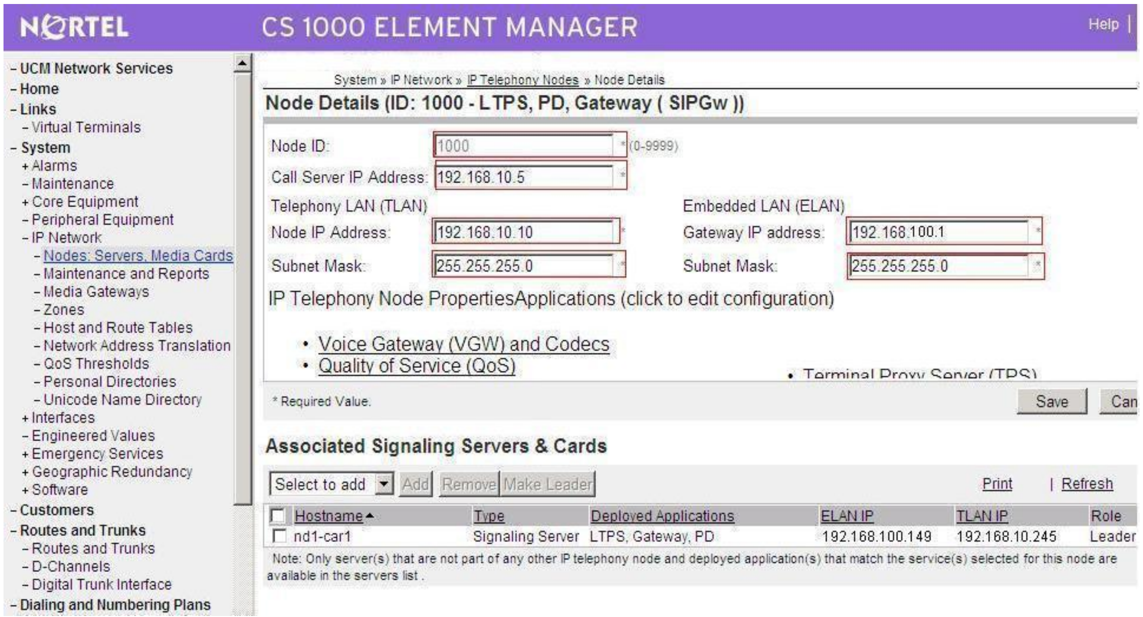

Figure 4 - Call Server IP Configuration

Since the carrier does not support TLS or sRTP, we have them disable in our CS1K configuration. For the primary proxy enter the IP address of the Session Border Controller (SBC). Use UDP SIP transport, port 5060 for SIP communication. The NRS is not enabled as all calls are routed by the SBC.

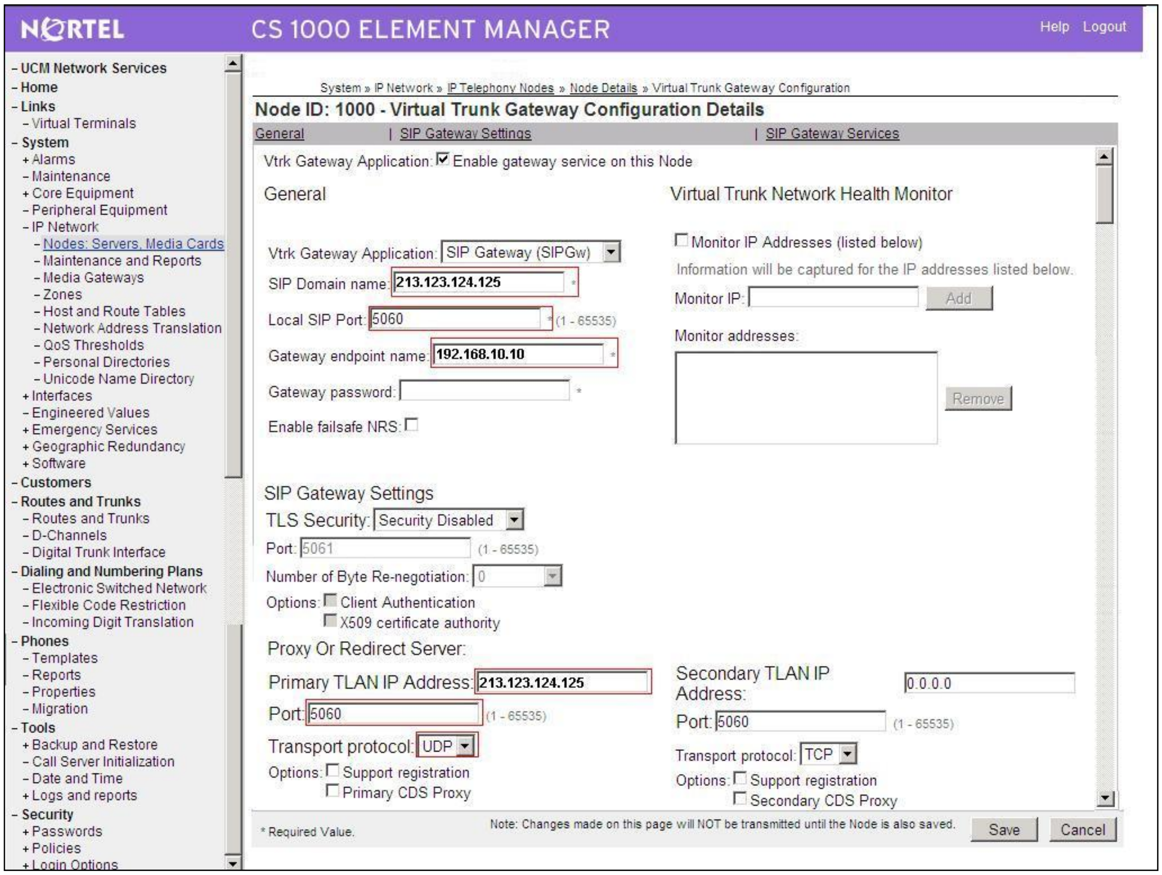

Figure 5 – Virtual Trunk Gateway configuration

4.1.2 Configure Voice Codec for Nortel IP Phone

This section describes the steps for administering a set of codecs in CS1000. This set of codecs is used in IP network for communication between Nortel IP Phones.

- Access EM by IE browser.
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".

Figure 6 and 7 are showing how to change Codec profile for IP Phone, select "VGW and IP phone codec profile".

Enable Modem FAX pass through mode for G711 and check V.21 Fax tone Detection (enabled as default) for MPT.

TN of sets with class of service = MPTD (Modem Pass Through Denied)

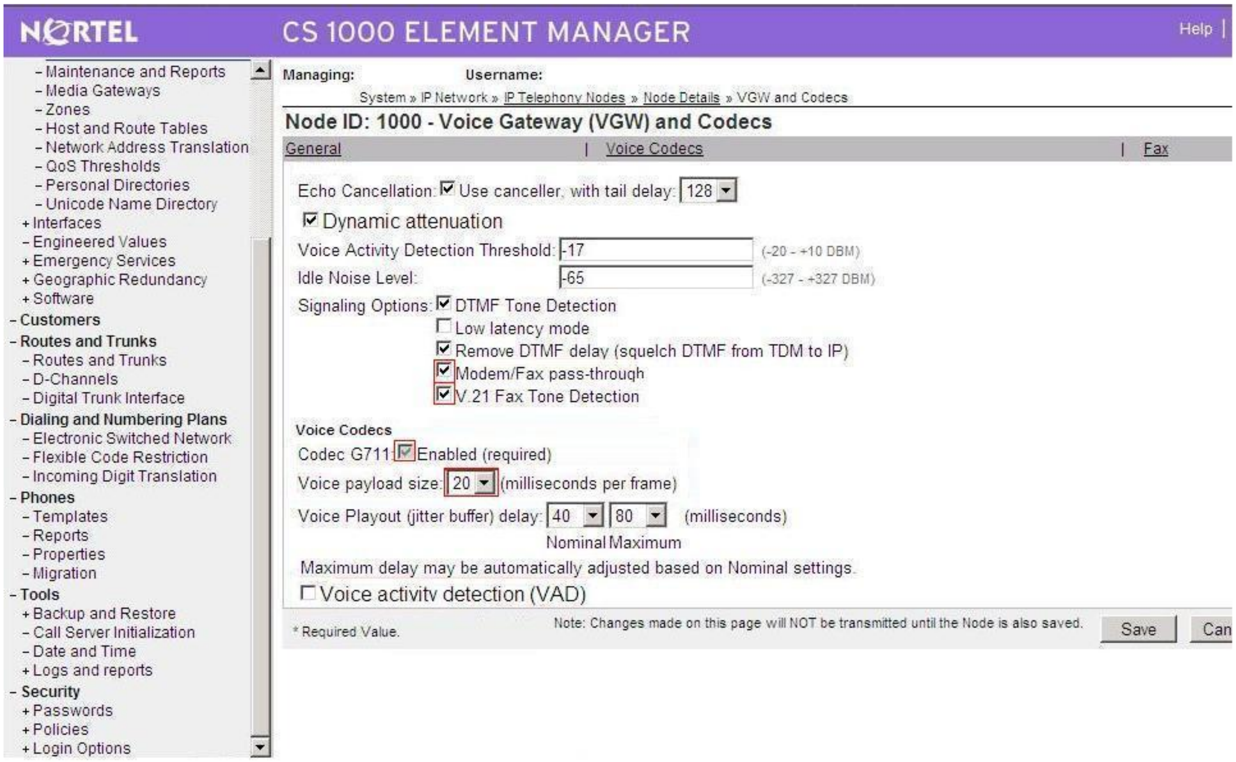

Figure 6 – Voice Gateway and Codec settings

Figure 7 shows how to configure the Voice gateway and IP phone codec settings. The Paetec Communication network supports both G.711 and G.729. The packet size is set to 20 to match the network also. FAX codec with T.38 FAX is as default in system.

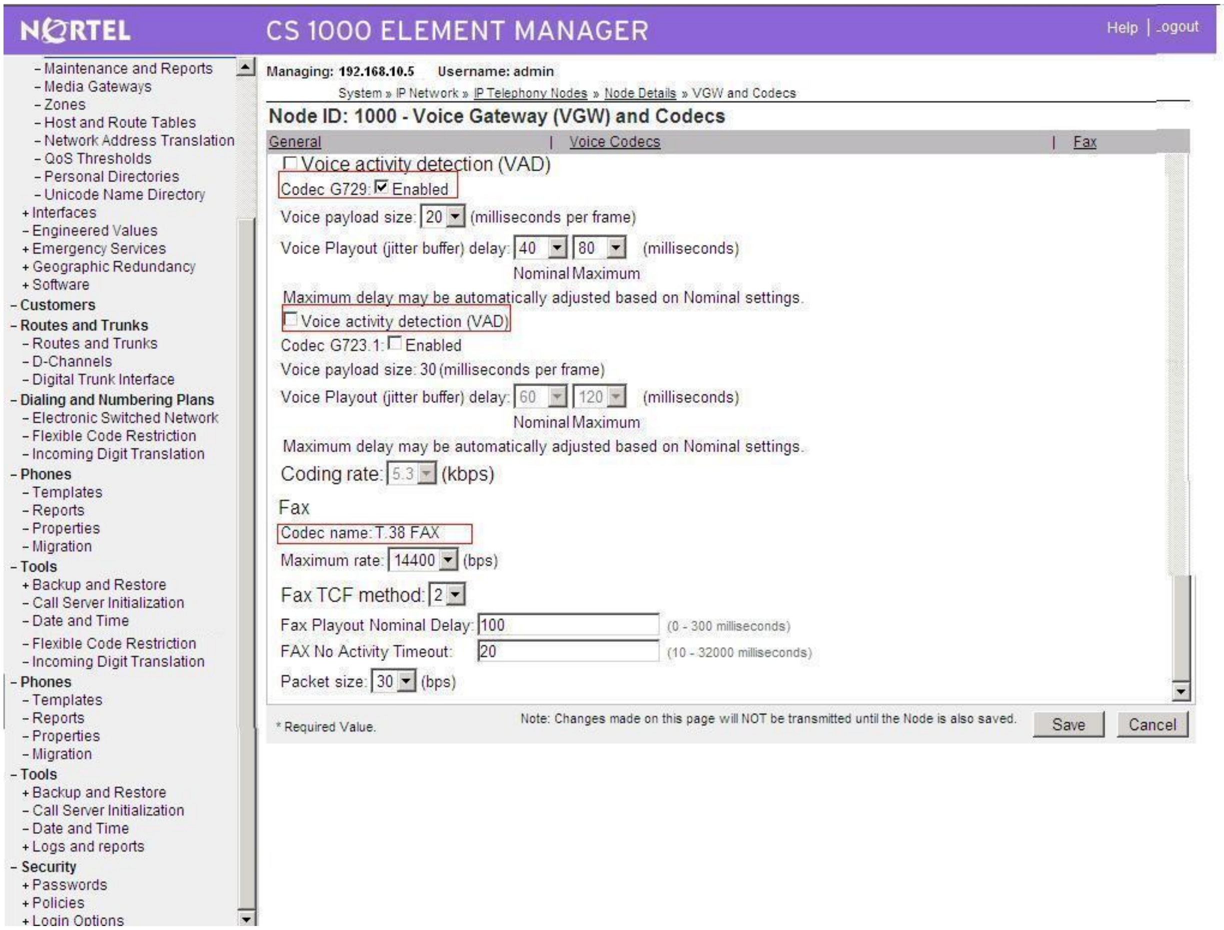

Figure 7 – Voice Gateway and codec settings

4.1.3 Configure Voice Codec for Media Gateways

This section describes the steps for administering a set of codecs in CS1000. This set of codec is used in IP network for communication through Media gateways.

- Access EM by IE browser.
- Choose "IP Network", then choose "Media gateways", select proper voice gateways
- To change Codec profile for IP Phone, select "VGW and IP phone codec profile".

Figure 8 shows how to configure the Voice Gateway and IP phone codec profile
Check Modem FAX pass through mode
TN of sets with class of service = MPTD (Modem Pass Through Denied)
Voice gateway and IP phone codec settings.

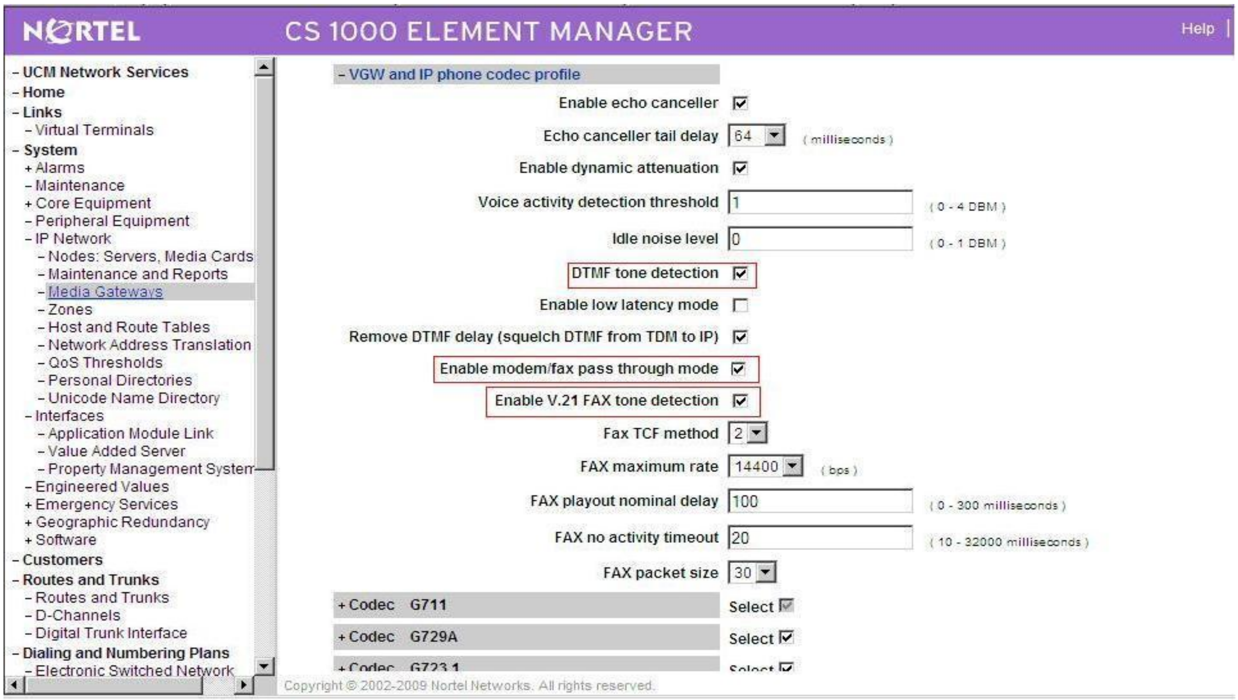

Figure 8 – Voice Gateway and IP phone codec profile settings

4.1.4 Configure Quality of Service

This section describes the steps for administering QoS in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".
- To change Quality of Service, select "QoS".

The default Diffserv values are correct in figure 9.

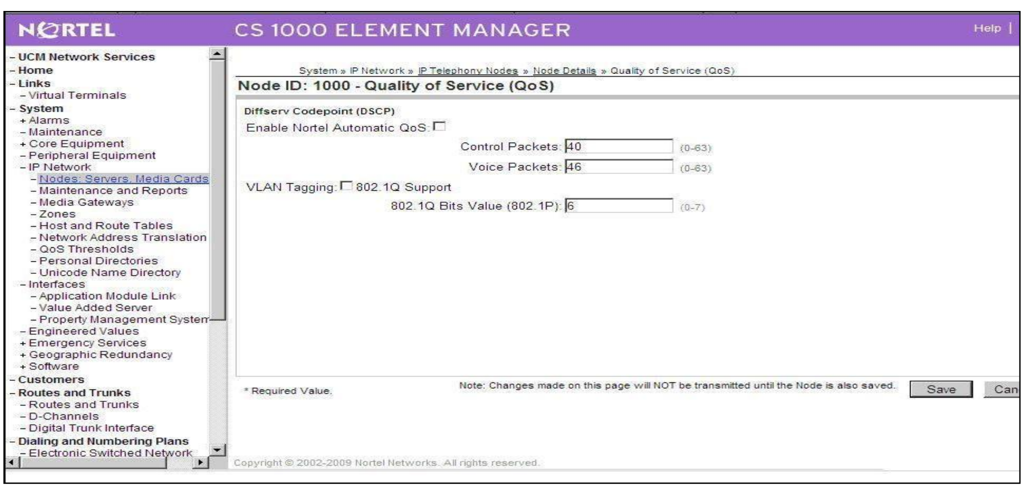

Figure 9 – Quality of Service settings.

4.1.5 Configure SIP URI

This section describes the steps for administering SIP URI configuration in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Nodes: Servers, Media Cards", select proper Node and press "Edit".
- To change SIP URI, select "SIP URI Map".

In figure 10, leave the SIP URI fields blank for E.164.

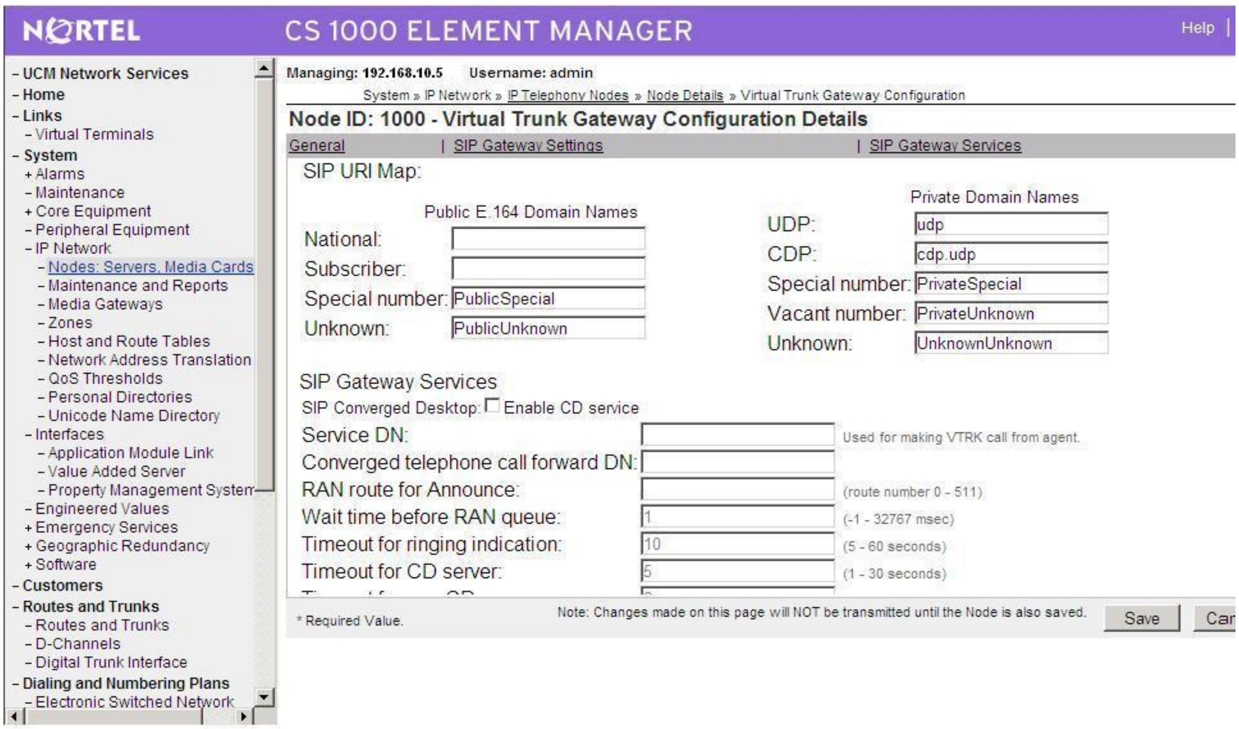

Figure 10 – SIP Gateway Services Settings

4.1.6 Configure Zones and Bandwidth Management

This section describes the steps for administering Zone configuration in CS1000.

- Access EM by IE browser
- Choose "IP Network", then choose "Zones", select proper “Zone Basic Property and Bandwidth Management”

Figure 11 shows how to configure a zone for IP sets and bandwidth management. If it does not already exist, create a zone for IP sets. The bandwidth strategy can be adjusted to preference.

Figure 12 shows how to configure a zone for new created SIP trunks.

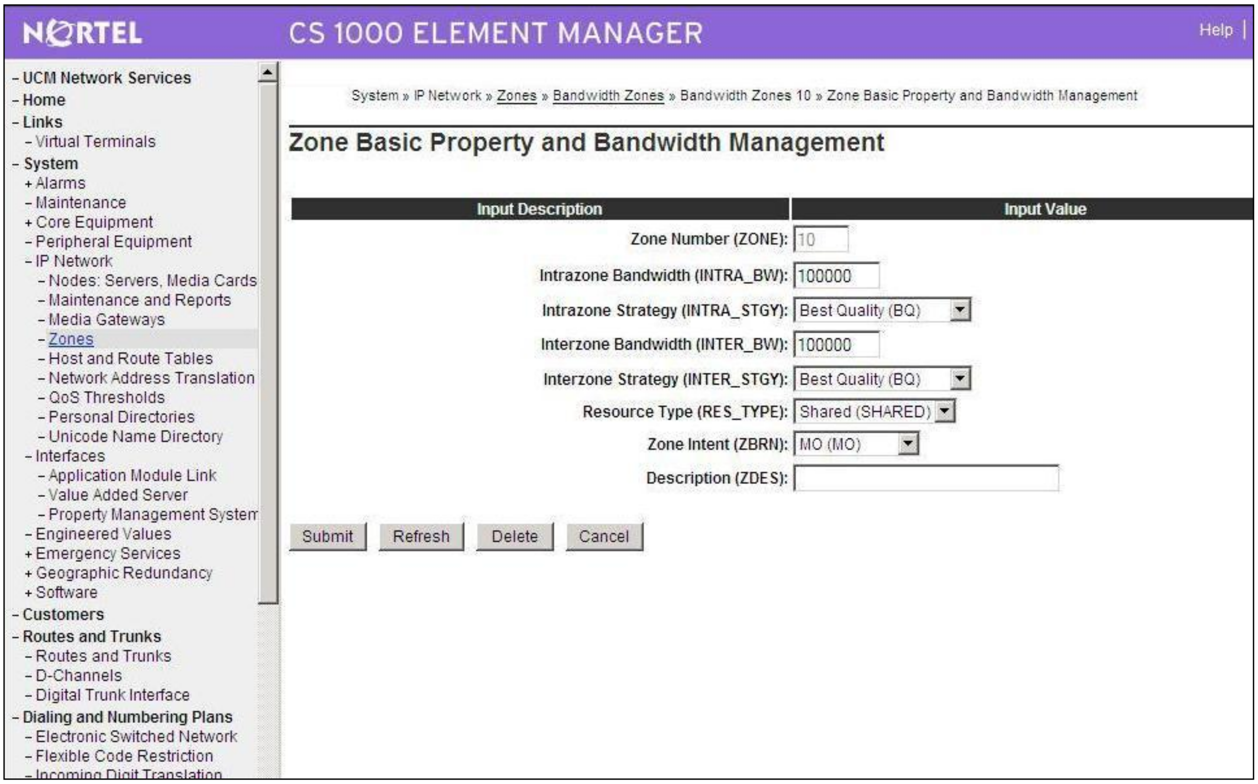

Figure 11 – Zone Basic Property Setting for IP phones

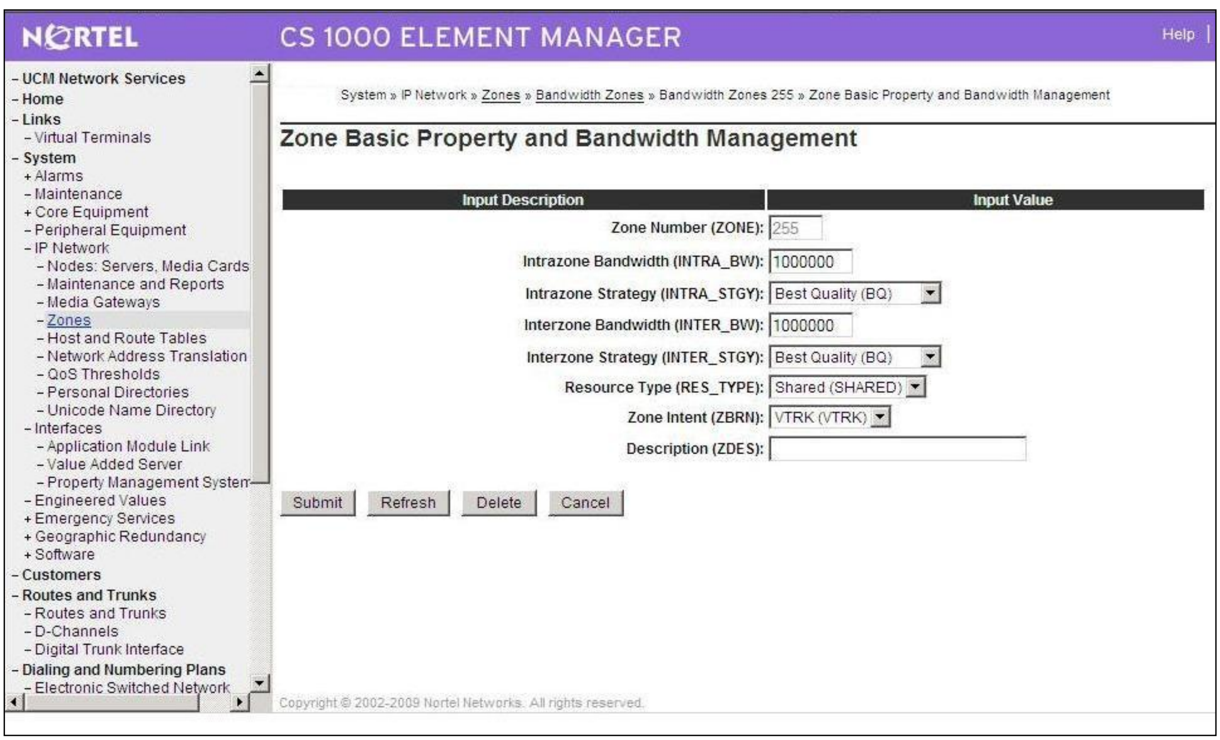

Figure 12 – Zone Basic Property Settings for (virtual) SIP trunk

4.1.7 Configure SIP trunk

This section describes the steps for establishing a SIP connection between CS 1000 switch and Carrier system.

1. Create D-channel (DCH)

- Launch Element Manager of CS 1000 6.0
- Choose D-Channels, enter D-channel number (i.e.: 100), select DCH for type

Click Add to create DCH 100

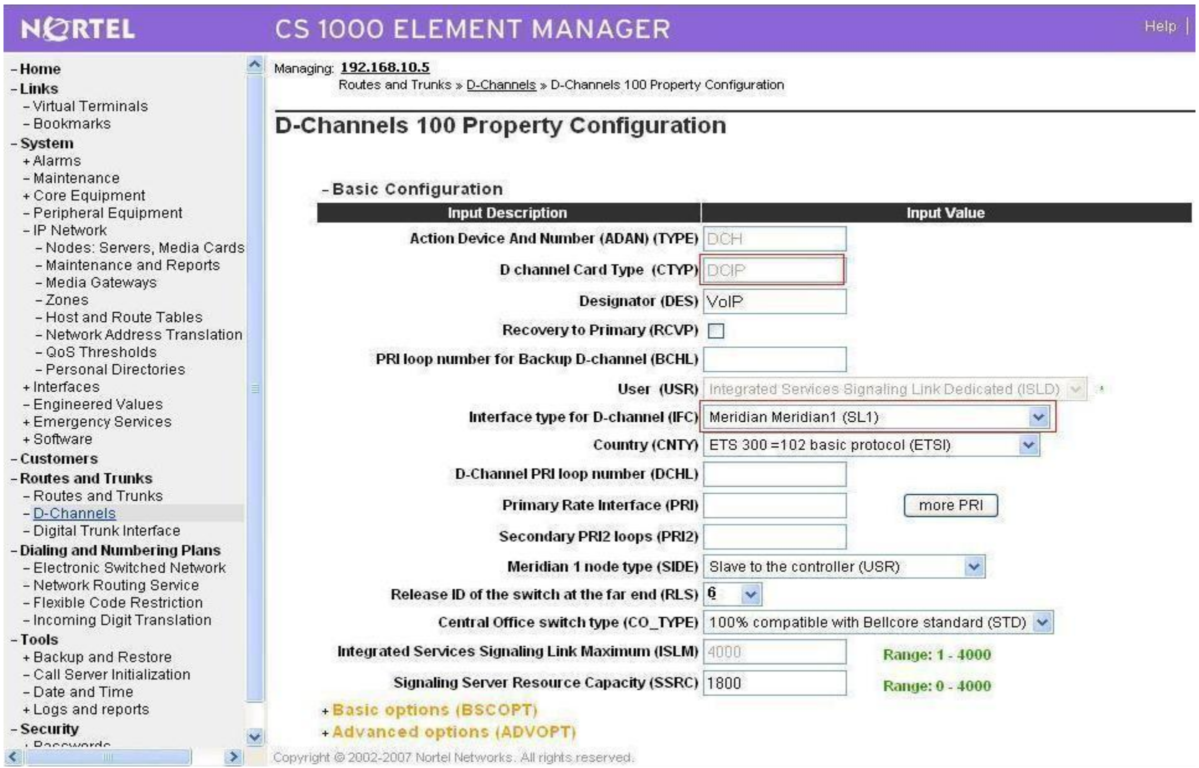

Figure 13 – D-Chanel Configurations

Also click on Basic Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

2. Create route: Create route 100 using DCH 100 for SIP trunks with figures 14 and 15

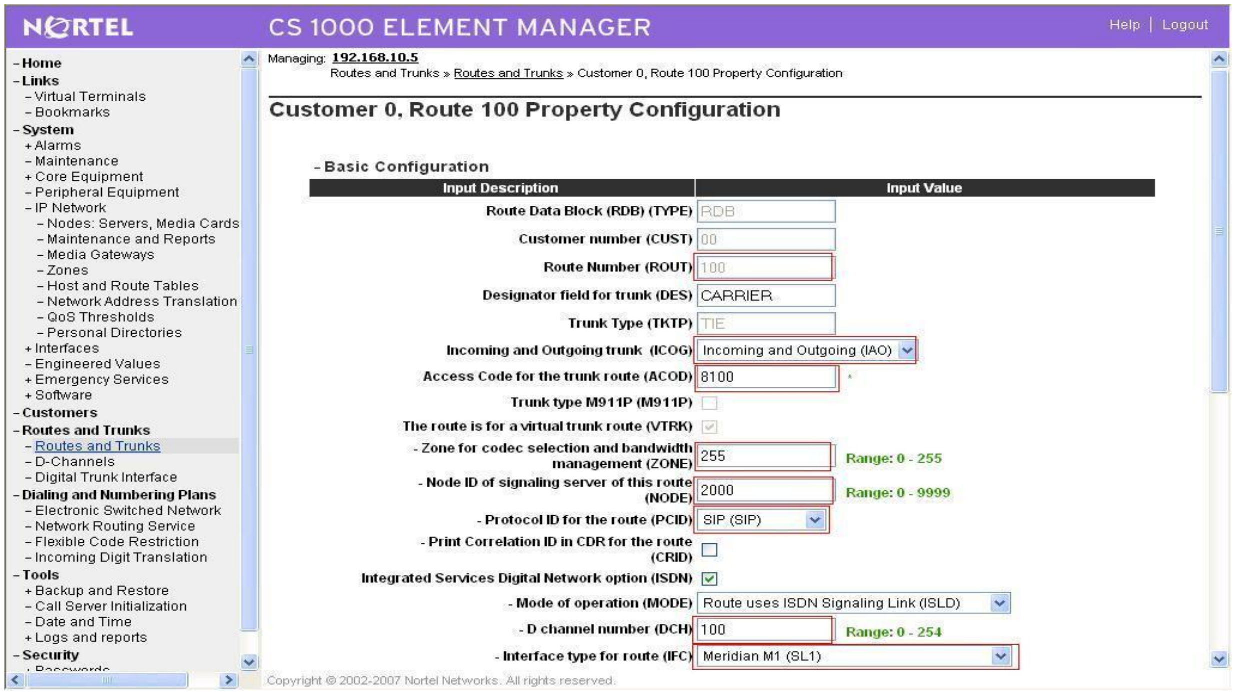

Figure 14 – Route Property Configuration

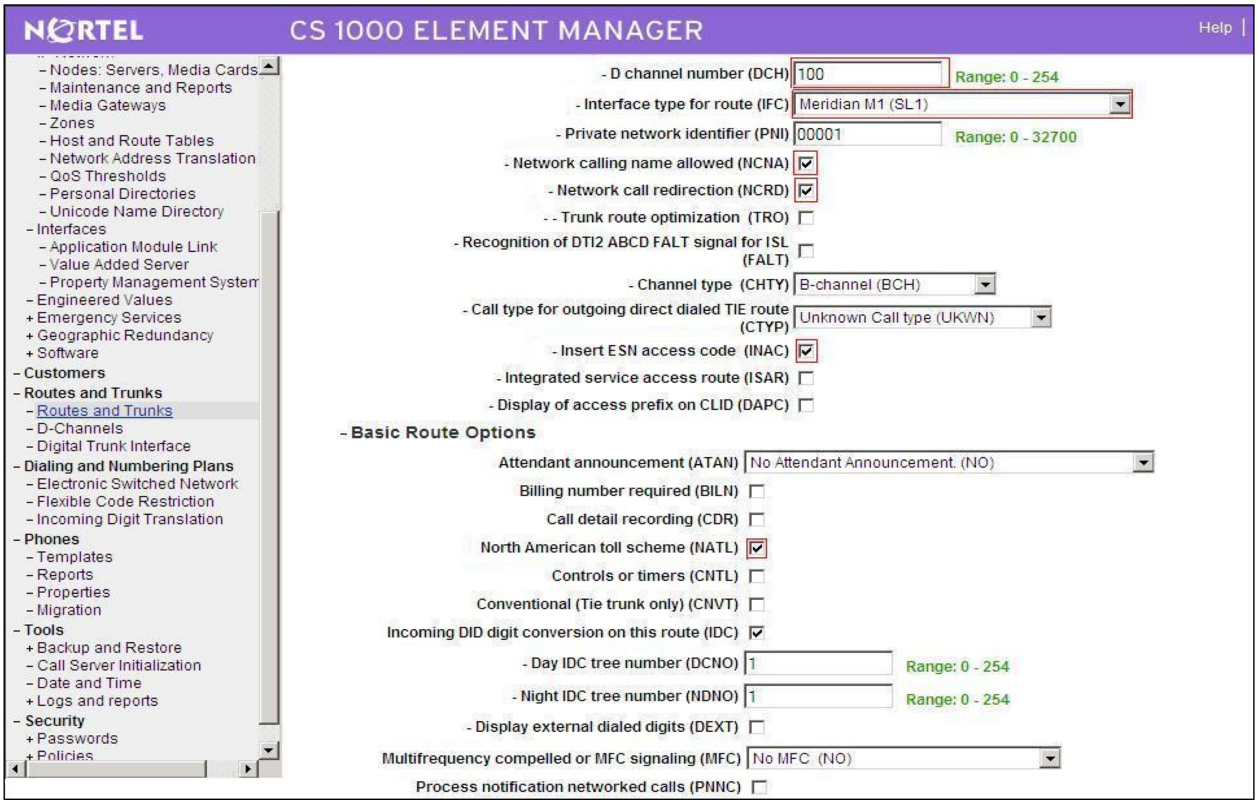

Figure 15 – Route Property Configuration (Cont..)

3. Create trunk: To create trunk using basic configuration in figure 16

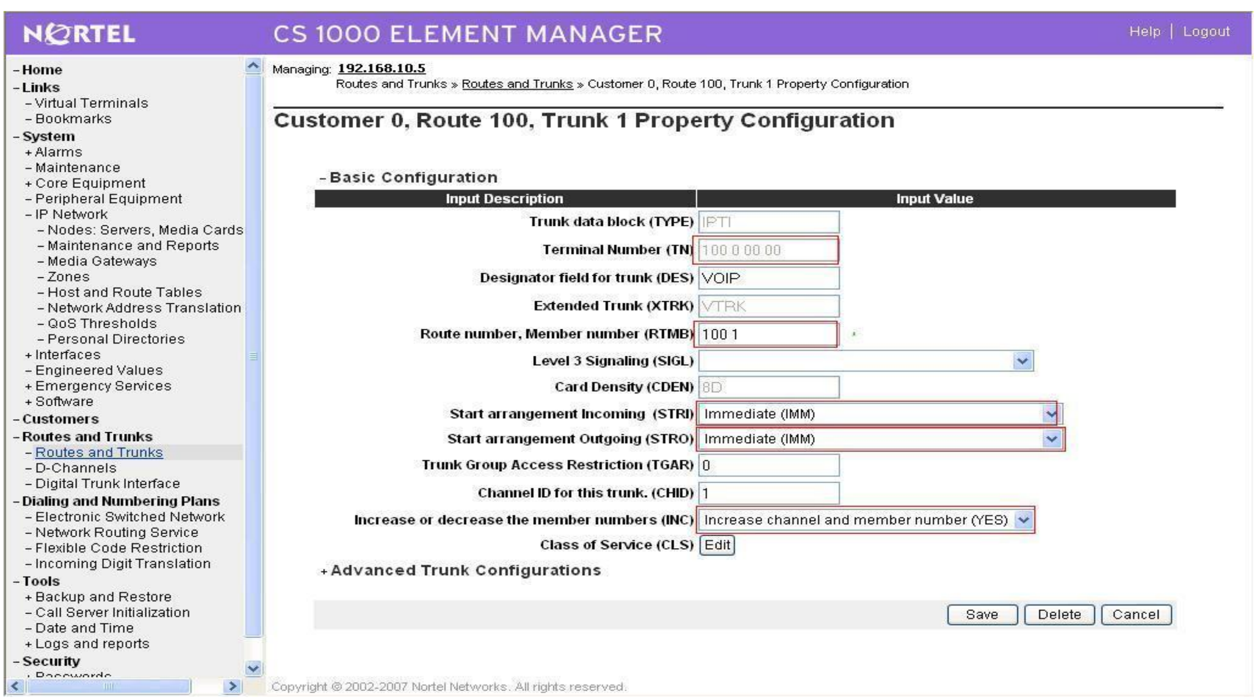

Figure 16 – Basic Trunk Configuration

Disable Media Security (sRTP) at the trunk level using figure 17 by editing the Class of Service (CLS) at the bottom basic trunk configuration page show in figure 17

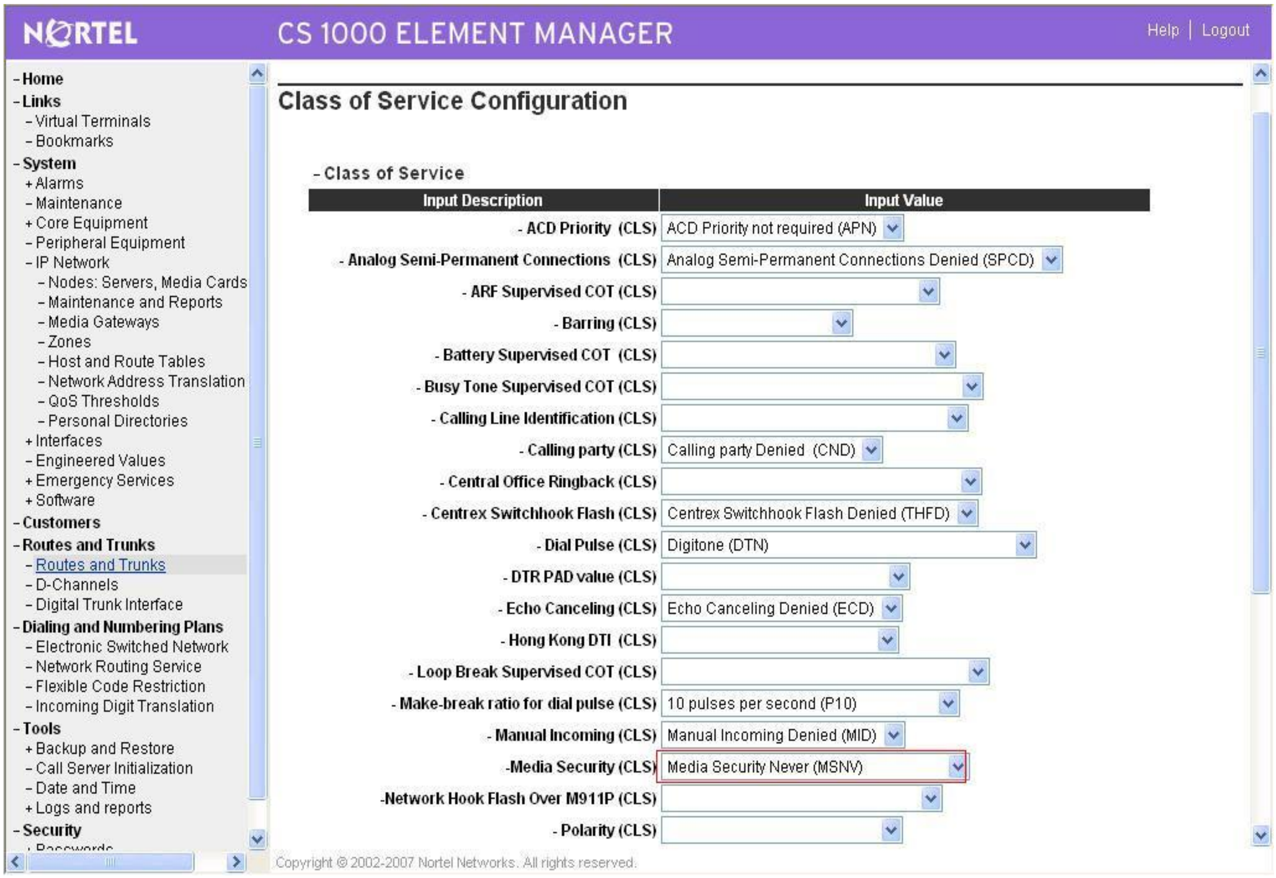

Figure 17 – Class of Service

Since Media security is not supported under Broadsoft system, Disable Media Security (MSNV) at the Trunk level as show in figure 17.

4. Create Dialing Plan:

Create Special number list:

Create special number list for outgoing dialing plan using figure 18

Launch Element Manager of CS 1000 6.0

Select "Dialing and Numbering plans → Electronic Switched Network → Number Plan (Net) → Access Code 1 (2) → Special Number (SPN).

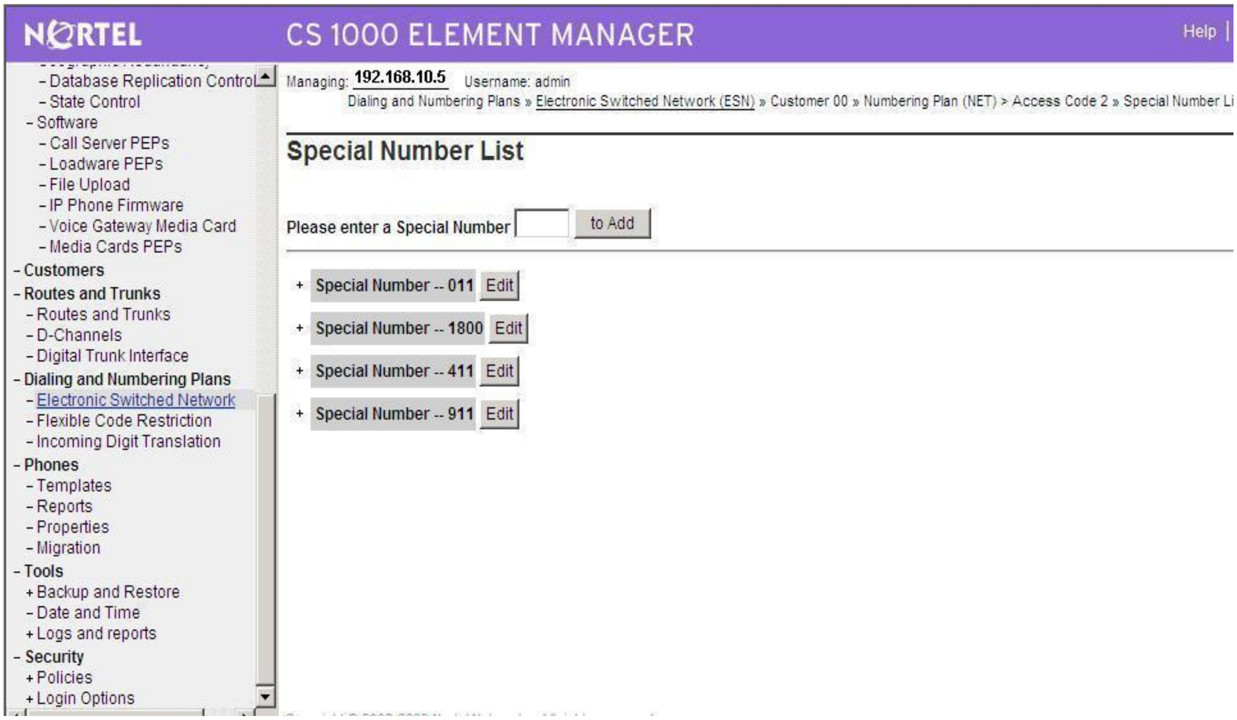

Figure 18 – Special Number List

Create special number SPN 011 (Use RLI_10) for outgoing dialing plan to International calls

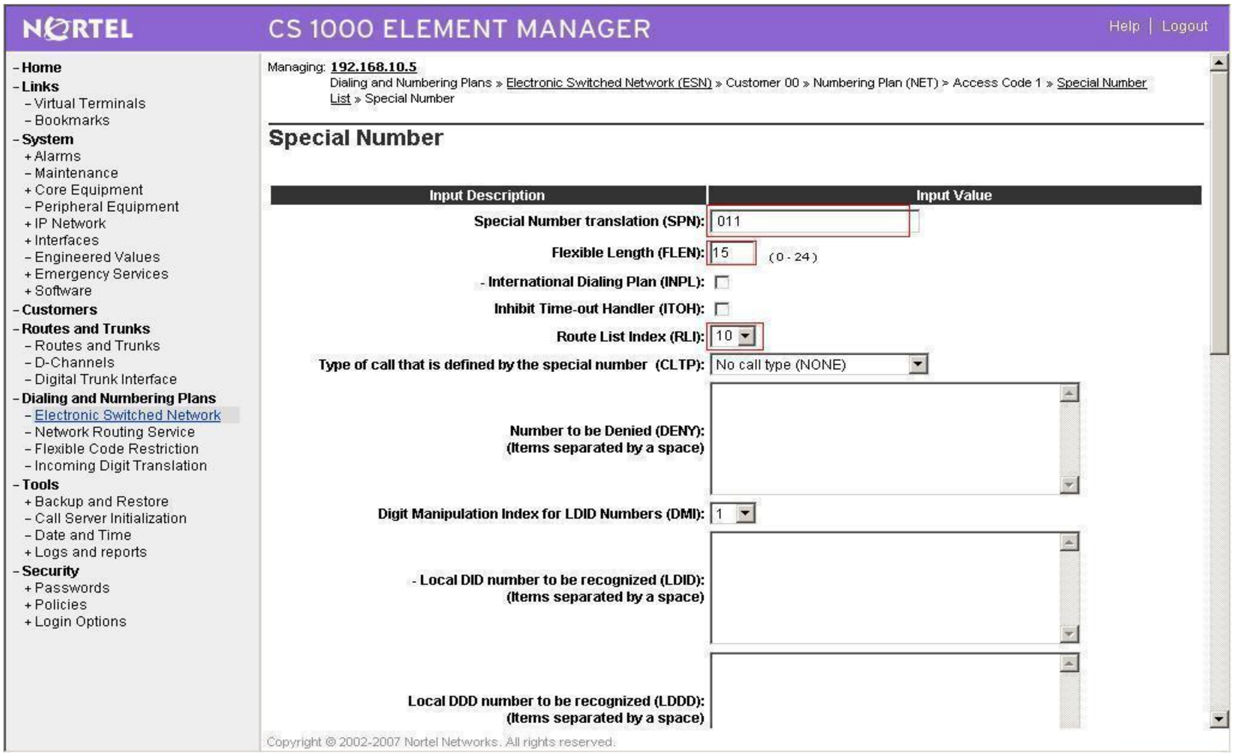

Figure 19 – Special Number for International Calls

Create special number SPN 1800 (Use RLI_10) for outgoing dialing plan to toll free calls

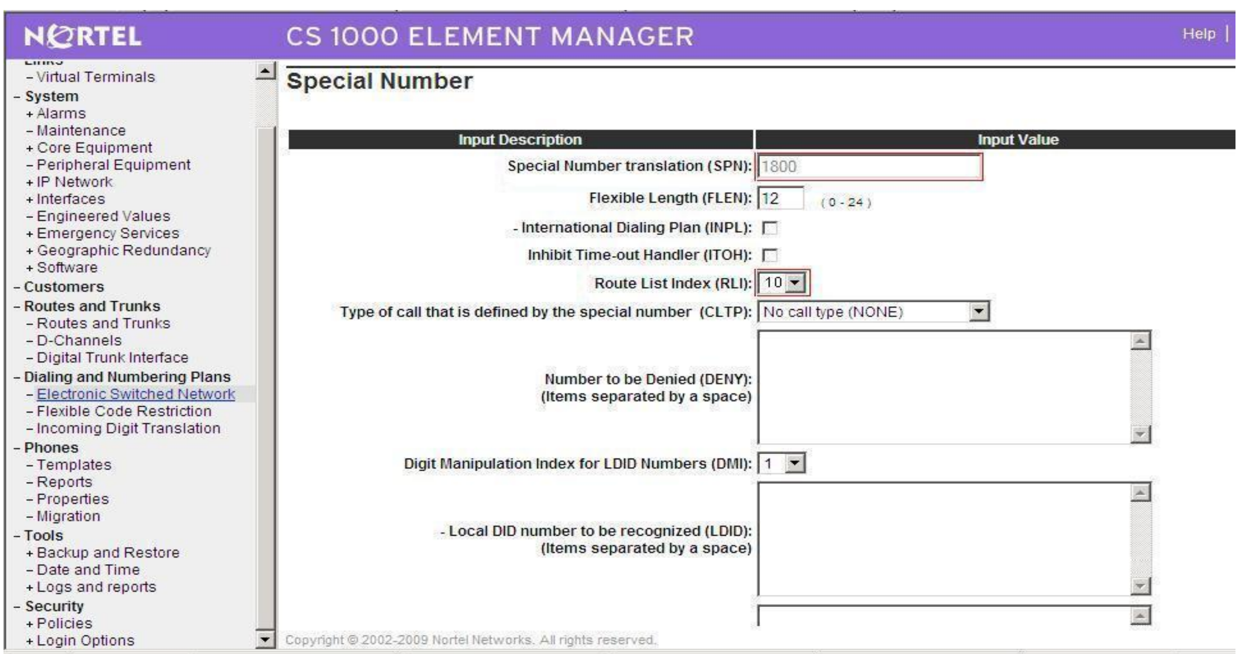

Figure 20 – Special Number for Tool Free Call

Create special number SPN 411 (Use RLI_10) for outgoing dialing plan to 411 service calls in figure 21

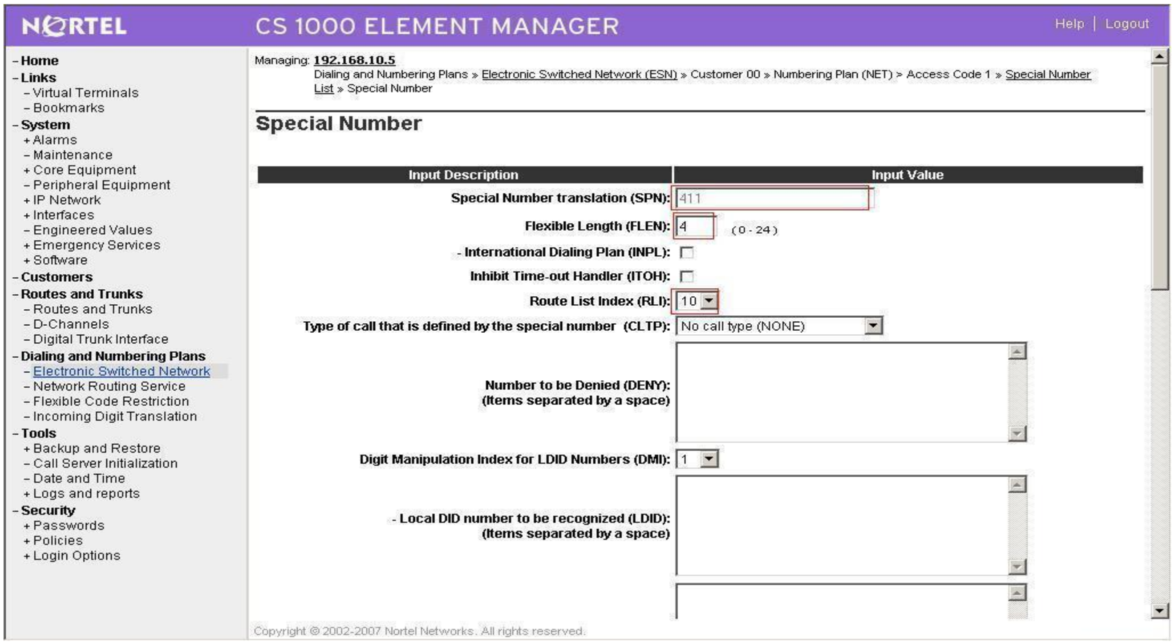

Figure 21 – Special Number for 411 Service Call

Create special number 911 (use RLI_10) to dial to Emergency service in figure 22

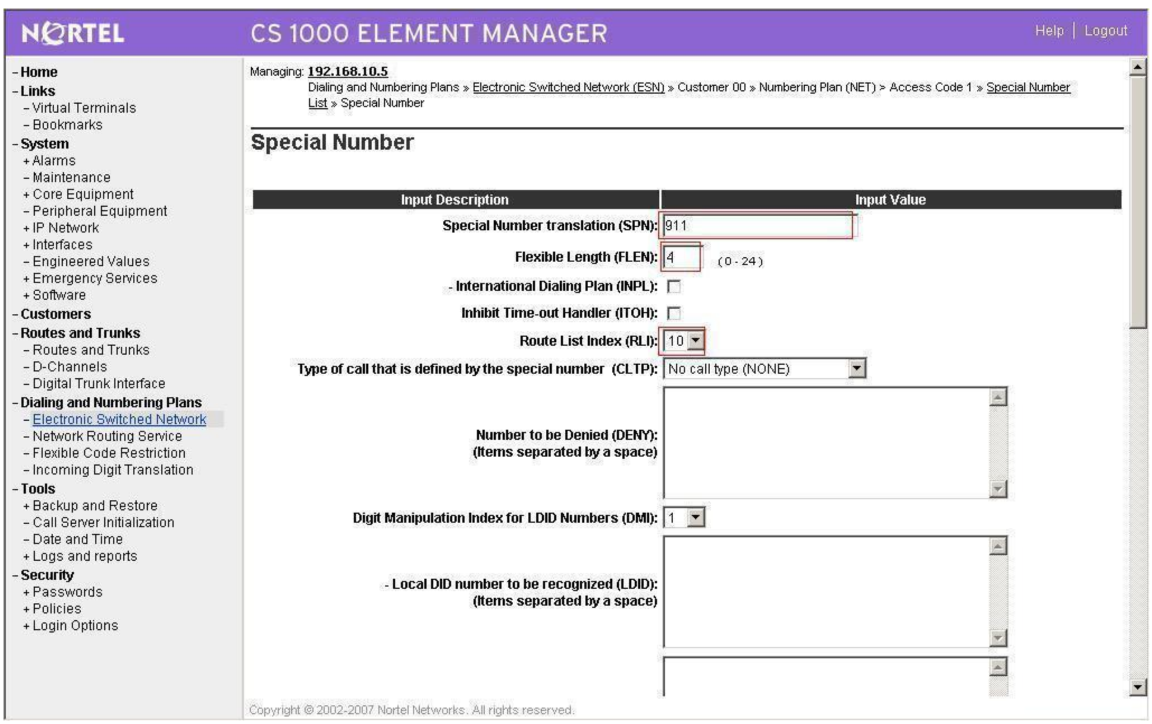

Figure 22 – Special Number for Emergency 911 dialing

Create Numbering Plan Area Code:

Create NPA numbers for outgoing.

NPA_1713: Create NPA_1713 for outgoing calls to numbers beginning with; Figure 23

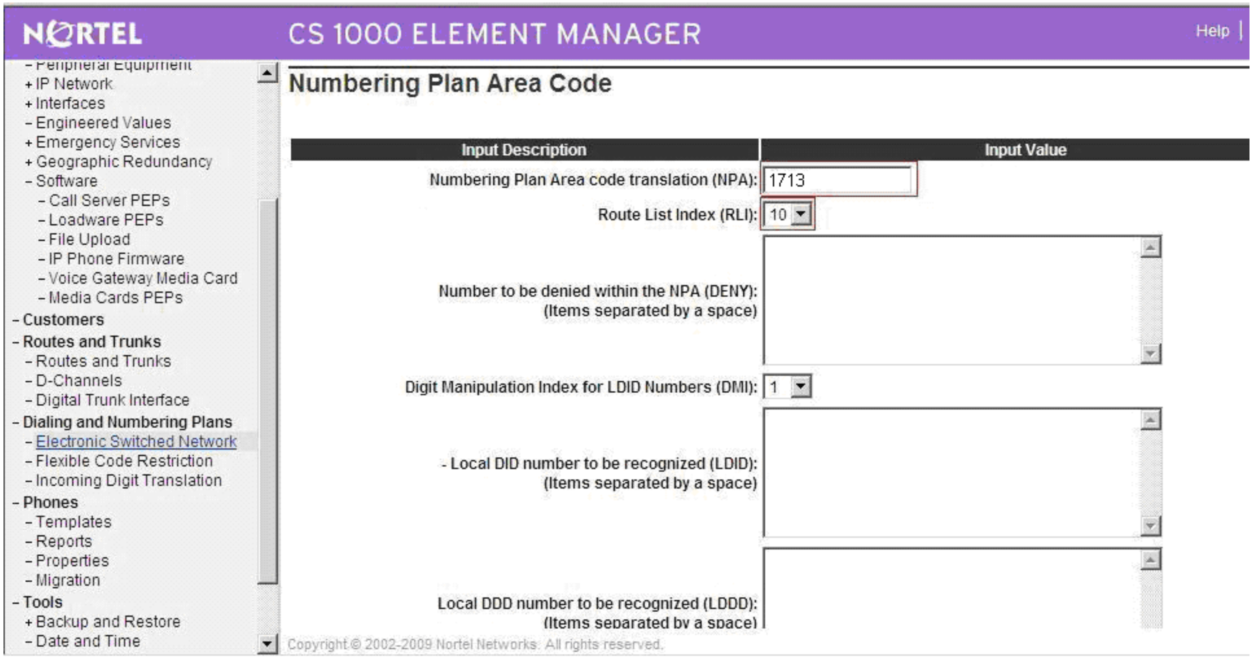

Figure 23 – Create NPA_1713 for outgoing calls

NPA_1613: Create NPA 1613 to dial to national DID numbers beginning with 613; Figure 24

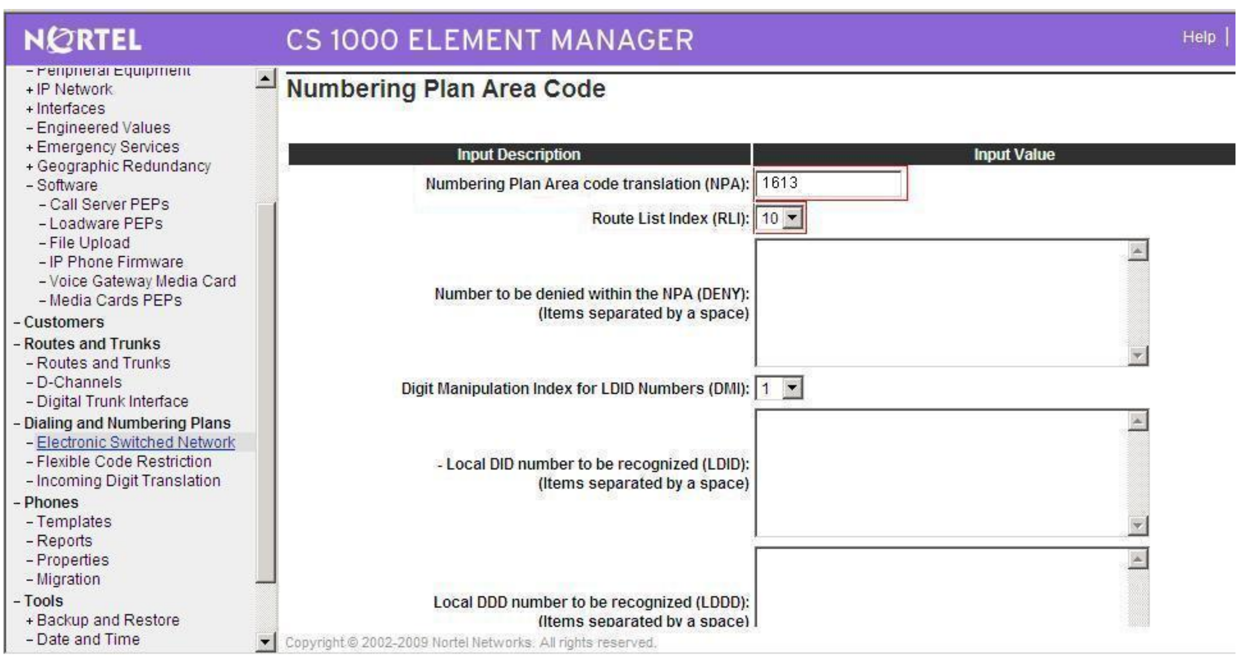

Figure 24 – Create NPA_1613 for outgoing calls to national numbers

Create Route List Block

Create RLI_10 for outgoing calls (Use route_100 and DMI_10), figure 25

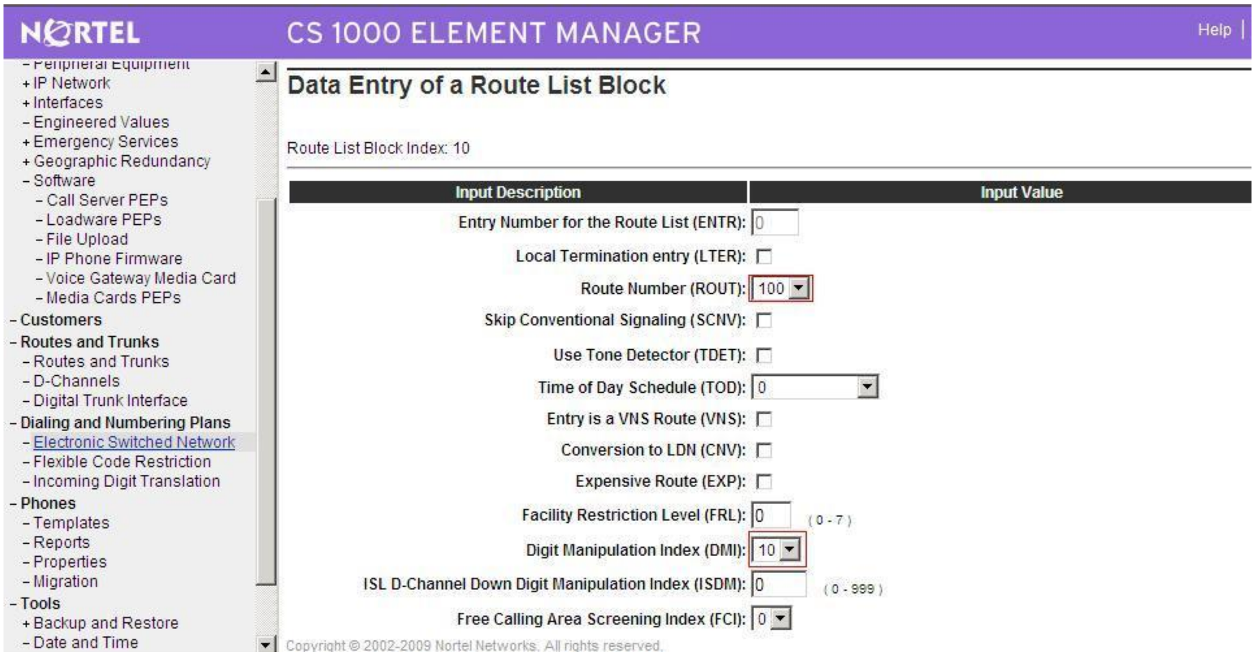

Figure 25- RLB for Outgoing calls

Create Local Steering Code

Create LSC_713 to terminate the incoming calls (Use DMI_3); Figure 26

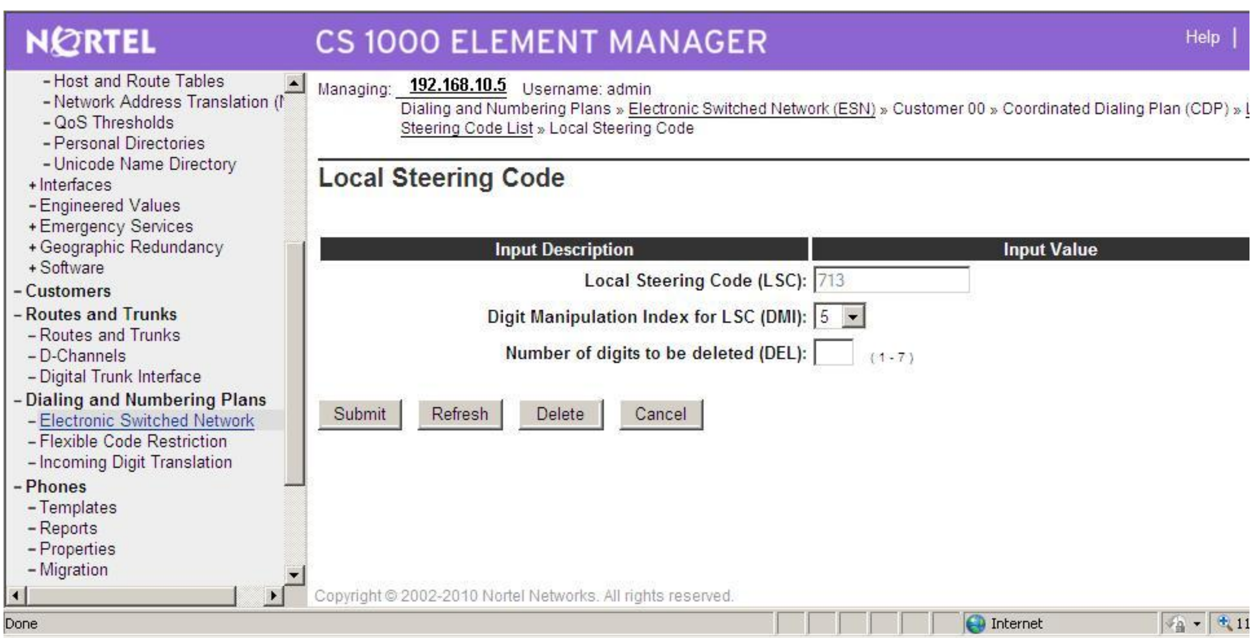

Figure 26 – Create LCS_713 for incoming call

Create Digit Manipulation Block

DMI_10: Digit Manipulation Block configuration for Outgoing calls; figure 27

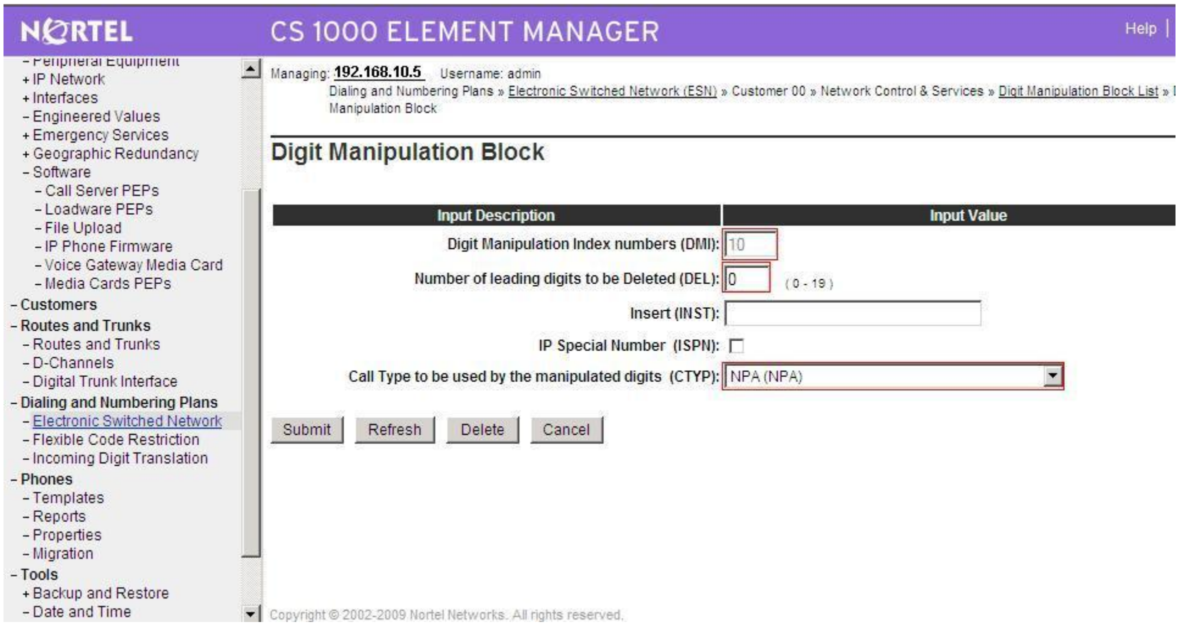

Figure 27 – Digit Manipulation for Outgoing calls

DMI_3: Digit Manipulation Block configuration for incoming calls; figure 28

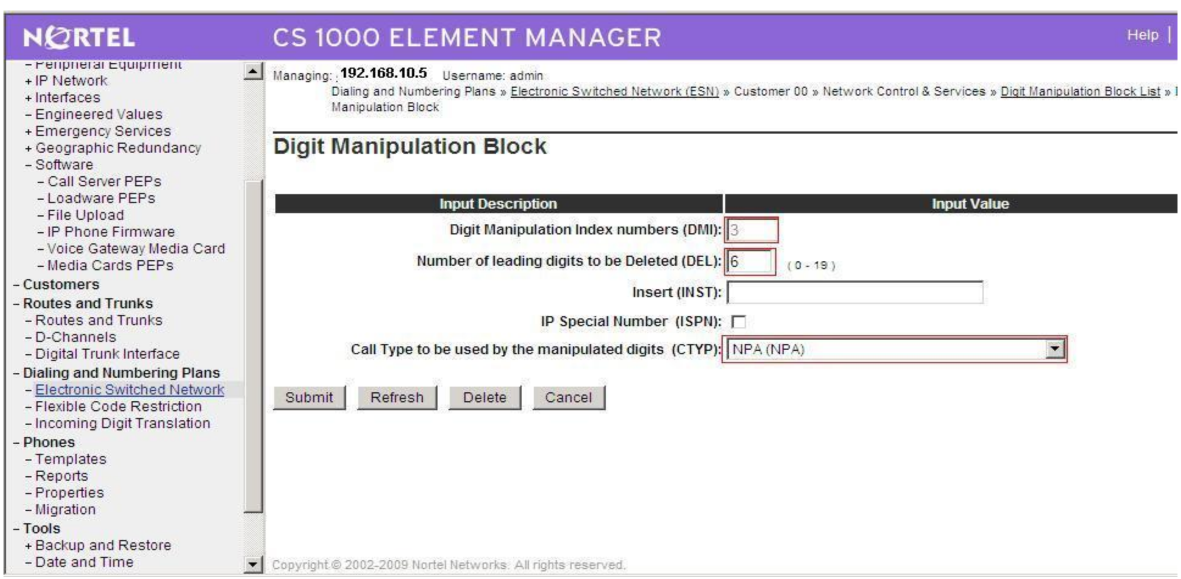

Figure 28 – Digit Manipulation for Incoming calls

4.2. Configure on CS1000 Voicemail System (Call Pilot)

4.2.1 Configuration Details on CallPilot Manager

Configure CS1000E switch on Call Pilot configuration by entering:

- CS1000 Call Server IP address
- Create Multimedia Chanel for communication between CS1000 and Callpilot system

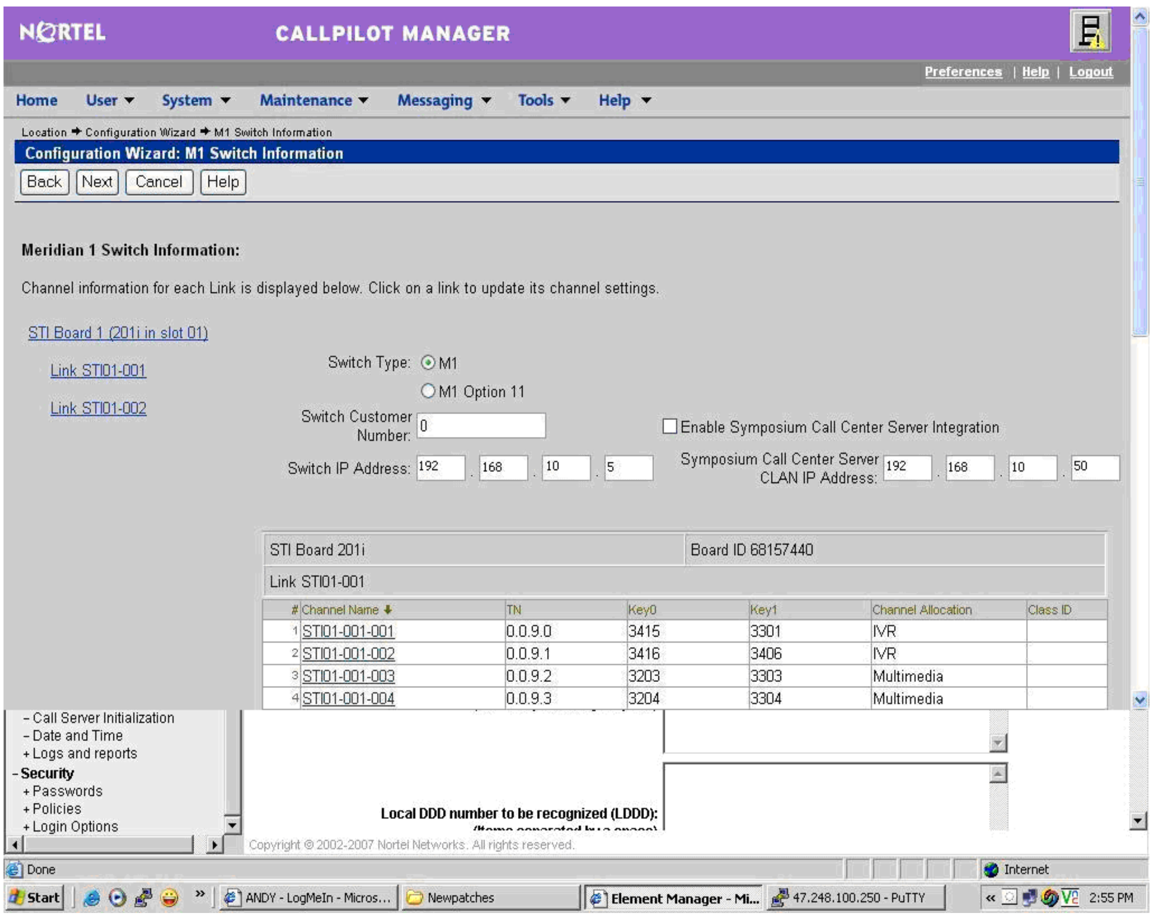

Figure 29 – CS1000 switch configuration on CallPilot Manager

Go to Maintenance pull down menu, select Channel Monitor to check status of the newly created multimedia channels on Call Pilot to see if the communication between Callpilot and CS1000 has been established, Figure 30.

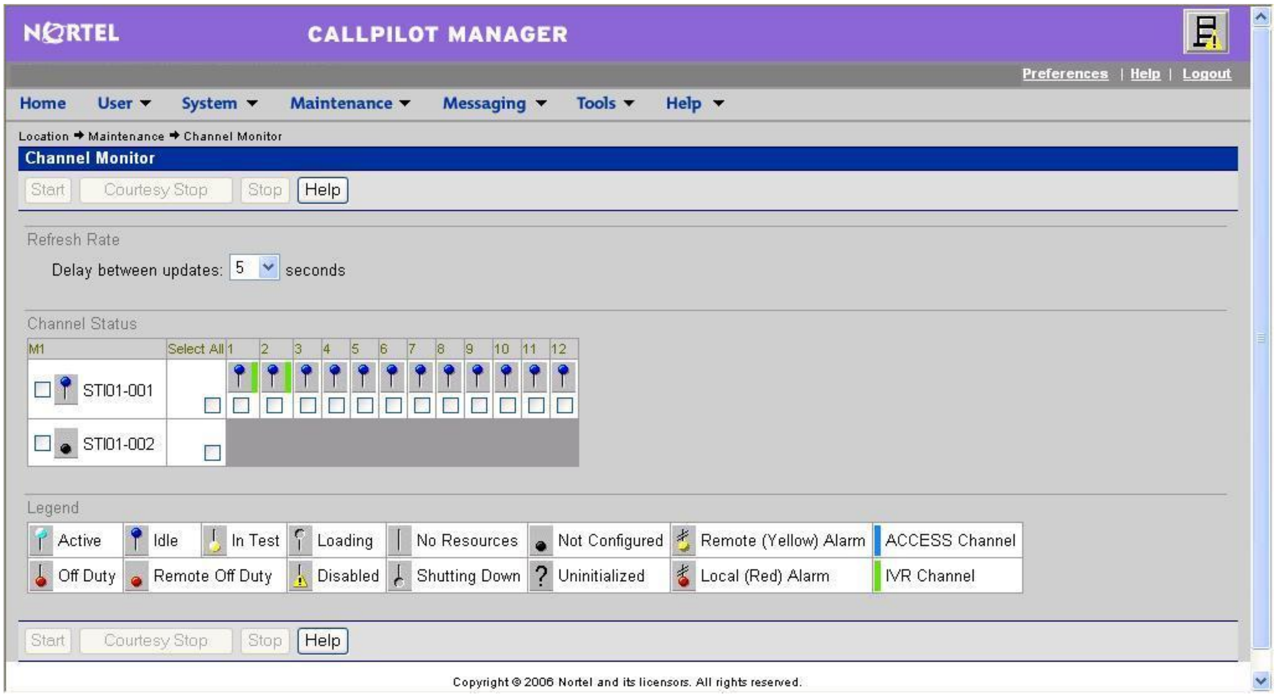

Figure 30 – Channel Monitor

Create Service DN for Voice Messaging system, Figure 31

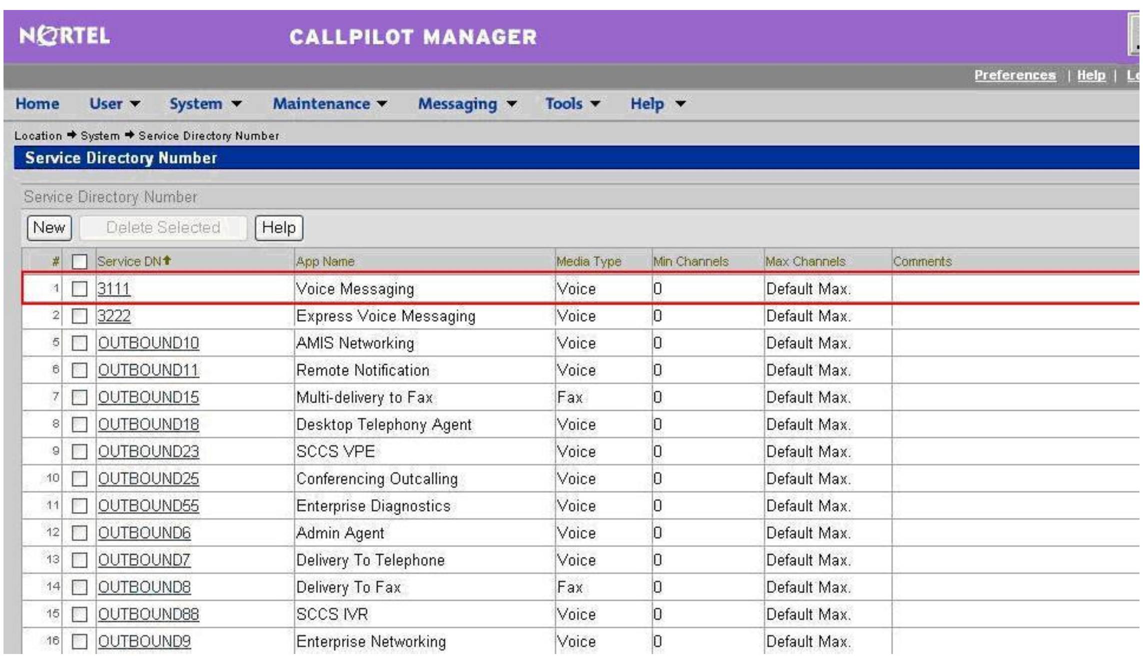

Figure 311 – Service Directory Number Page

4.2.2 Voicemail System (CallPilot) configuration detail on CS1000E Call Server

Configure CS1000E for voicemail system Call Pilot

Configure Voice messaging service DN 3111 on CS1000E

>ld 23

ACD DNS

REQ prt

TYPE CDN

CUST 0

CDN 3111

FRRT

SRRT

FROA NO

UUI NO

MURT

CDSQ NO

DFDN 3109

NAME NO

CMB NO

CEIL 2047

OVFL NO

TDNS NO

AACQ NO

CNTL NO

VSID

HSID

Configure ACD Agent #1 3110:

>ld 23

ACD DNS

REQ prt

TYPE ACD

CUST 0

ACDN 3110

MWC YES

MAXP 12

SDNB NO

BSCW NO

ISAP NO

AACQ YES

ASID 16

SFNB

USFB 1 3 4 5 6

CALB 1 3 4 5 6 8 11

RGAI NO

ACAA NO
FRRT
SRRT
NRRT
FROA NO
CALP POS
ICDD NO
NCFW
FNCF NO
CWTT NONE
HMSB YES
ACPQ NO
FORC NO
RTQT 0
SPCP NO
OBTN NO
RAO NO
CWTH 1
NCWL NO
BYTH 0
OVTH 2047
TOFT NONE
HPQ NO
OCN NO
OVDN
IFDN
OVBU LNK LNK LNK LNK
EMRT
MURT
RTPC NO
NRAC NO
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
CCBA NO
IVR YES
TRDN NONE
ABR NO
OBSC NO
OBPT 5
CWNT NONE

Configure ACD Agent#2

>ld 23

REQ prt

TYPE acd

CUST 0

ACDN 3109

TYPE ACD

CUST 0

ACDN 3109

MWC NO

DSAC NO

MAXP 1

SDNB NO

BSCW NO

ISAP NO

AACQ NO

RGAI NO

ACAA NO

FRRT

SRRT

NRRT

FROA NO

CALP POS

ICDD NO

NCFW

FNCF NO

CWTT NONE

HMSB YES

ACPQ NO

FORC NO

RTQT 0

SPCP NO

OBTN NO

RAO NO

CWTH 1

NCWL NO

BYTH 0

OVTH 2047

TOFT NONE

HPQ NO

OCN NO

OVDN

IFDN

OVBU LNK LNK LNK LNK

EMRT

MURT

RTPC NO

NRAC NO

RAGT 4

DURT 30

RSND 4

FCTH 20

CRQS 100

CCBA NO

IVR NO

ABR NO

OBSC NO

OBPT 5

CWNT NONE

4.3. Output configuration details from CS1000 and Callpilot

Using the command line interface to output some of configured Customer Data Block and configuration record details, which have been created in section 4.2, for your reference

4.3.1 Overlay 15 - Customer Data Block

```
REQ: PRT
TYPE CDB
CUST 00
AML_DATA
  OPT DNX
  VSID
  GP02
  GP03
  GP04
  GP05
  GP06
  GP07
  GP08
  GP09
  GP10
  GP11
  GP12
  GP13
  GP14
  GP15
ANI_DATA
  ANAT 4227
  ANLD 123
  M911_PANI NO
ATT_DATA
  OPT ABDD AHD BIND BIXA BLA BOHD DNCA DRE
```

```
    DNX DRE FACD IC1 XTG XDP XLF XBL
    FKA MCTD NCD CUI MWUD LOD PSD RECA
    REA SYD SLD SIAD THPD ATDA
  ATDN 7
  NCOS 0
  CWUP NO
  CWCL 0 0
  CWTM 0 0
  CWBZ NO NO
  EFLL 0
  MATT NO
  RTIM 30 30 30
  ATIM 0
  AQTT 30
  AODN
  SPVC 00
   SBLF NO
  RTSA RSAD
  SACP NO
  ABDN NO
  IRFR NO
  XRFR NO
  ADHT 0
  AFNT 0
  AFBT 0
  IDBZ NO
  PBUZ 02  10
  ICI 00
  ICI 01
  ICI 02
  ICI 03
  ICI 04
  ICI 05
  ICI 06
  ICI 07
  ICI 08
  ICI 09
  RICI
PAGE 002
AWU_DATA
  AWU NO
CAS_DATA
  CAS NO
CCS_DATA
  CCRS UNR
```

```
 ECC1 UNR
 ECC2 UNR
 CNCS 0
 PELK NO
CDR_DATA
 CDR YES
  IMPH NO
  OMPH YES
  AXID YES
  TRCR NO
  CDPR NO
  ECDR NO
  BDI  YES
  OTCR NO
  PORT
  CNI  DGTS
  BCAP NO
 CHLN 1
 FCAF NO
FCR_DATA
 NFCR YES
  MAXT 100
  OCB1 255
  OCB2 255
  OCB3 255
 IDCA YES
  DCMX 100
FFC_DATA
 CCRS UNR
 SCPL 0
 FFCS NO
  STRL 0
  STRG
 ADLD 0
 MFAC *
FTR_DATA
**DAPC**PREFIX TABLE NO: 00 **
     UNKN**INTL**NATL**ESPN**LOCL**ELOC**ECDP**
UNKN*
E164*      00    0
PRIV*
E163*      00    0
TELX*
X121*
NATL*
```

```
OPT ABDD AHD BIND BIXA BLA BOHD CFO CFRD
  COX CPA CTD DBD DNCA DNX DSX DRE
  DSTD FACD HTU HVD XBL IC1 XDP XLF
  IHD XTG FKA LOD LRA MCI MCTD CUI
  MWUD NCD PCMD PSD PVCA RECA REA RND
  RTR RTD ROX SBD SDDE SIAD SLD SYD
  THPD TTAD VOBD CCBD CWRD HLPD HRLD
  CXOD BWTD
DGRP 0
IRNG NO
PKND 1
DNDL NO
SPRE
PREO 0
BPSS NO
SRCD 0000
EEST NO
EESD NO
TTBL 0
MUS YES

PAGE 003
 MUSR 50
HCC NO
ALDN
RECD NO
PORT 0
STCB NO
NSCP NO
TFDR NO
RPA NO
MCDC NO
NAUT NO
IDEF NO
MTAR NO
LEND NO
MSCD NO
CPCI NO
ARDL_ATTEMPT 30
CONF_DSP
 CNFFIELD NO
 CNF_NAME CONF
 INTFIELD NO
  INT_NAME I
```

```
   EXTFIELD NO
     EXT_NAME E
  BSFE NO
  ASPCT 000
  FXS NO
  DFLT_LANG ENG
  STS_MSG
    MSG01  Please leave message
    MSG02  Back to work
    MSG03  In a meeting
    MSG04  On a conference call
    MSG05  At lunch
    MSG06  Busy call
    MSG07  Out of the office today
    MSG08  On a business trip
    MSG09  Project deadline today
    MSG10  Will reply after
  VO_ALO NO
  PCA ON
  TPDN
  BFS_CFW YES
  VO_CUR_ZONE_ZDM NO
  VO_CUR_ZONE_TD NO
ICP_DATA
  ICP NO
IMS_DATA
  IMS NO
INT_DATA
  ACCD OVF OVF OVF ATN
  CTVN OVF OVF OVF ATN
  MBNR OVF OVF OVF ATN
  CTRC OVF NAP OVF NAP
  CLDN NAP OVF NAP NAP
  NINV OVF OVF OVF ATN
  NITR OVF OVF OVF ATN
  NRES OVF OVF OVF ATN
  NBLK OVF OVF OVF ATN
  MFVOOVF OVF OVF ATN
  MFVN OVF OVF OVF ATN
  MFCG OVF OVF OVF ATN
PAGE 004
  LCKT BSY BSY BSY BSY
  RCLE ATN OVF ATN ATN
  CONG OVF
  DLT OVF
```

```
  LLT OVF
  DNDT BSY
  ESAM OVF
LDN_DATA
  OPT XLDN
  DLDN YES
  LDN0 2000
  LDA0
  LDN1
  LDA1
  LDN2
  LDA2
  LDN3
  LDA3
  LDN4
  LDA4
  LDN5
  LDA5
  LDBZ
  ICI 00
  ICI 01
  ICI 02
  ICI 03
  ICI 04
  ICI 05
  ICI 06
  ICI 07
  ICI 08
  ICI 09
MON_DATA
  USBM NO
MPO_DATA
FMOP
  RGNA STD STD
  AOCS DIS ATN
  RCY1 06
  RCY2 04
  RALL NO
  CDTO 14
IFLS NO
MHLD NO
PCDS
  CNFD 1
  TGLD 2
  DISD 3
```

CCDO NO

AFCO NO

ACNS NO

NET_DATA

OPT RTD

AC1 NPA SPN LOC

AC2 INTL NXX

FNP YES

ISDN YES

VPNI 1

PNI 1

PINX_DN

MBG 0

PAGE 005

BSGC 65535

PFX1

PFX2

HLOC 521

LSC

RCNT 5

PSTN NO

TNDM 15

PCMC 15

SATD 1

OCLI NO

TIDM NO

DASC

ROPT NRO

DITI YES

TRNX NO

EXTT NO

FTOP FRES

APAD 0 0

VNR NO

NIT 8

NAS_ATCL YES

NAS_ACTV NO

FOPT 6

CNDN

CNAT

PCAT

CNIP YES

DMWM NO

MWNS NO

CNTC
NATC
INTC
NIT_DATA
NIT1
TIM1
NIT2
TIM2
NIT3
TIM3
NIT4
TIM4
RPNS NO
ENS NO
OAS_DATA
ODN0
ODN1
ODN2
ODN3
ODN4
ODN5
ODN6
ODN7
ODN8
ODN9
ASTM 30
HDOPT 0
HDTM 30
RDR_DATA
OPT CFO CFRD DSTD PVCA CWRD MCI
FNAD HNT
FNAT HNT
PAGE 006
FNAL HNT
CFTA NO
CCFWDN
CFN0 3
CFN1 3
CFN2 3
DFN0 3
DFN1 3
DFN2 3
DNDH NO
MDID NO
NDID NO

MWFB NO

TRCL 0

DFNR 0

CRT0 00 00 00 00

CRT1 00 00 00 00

CRT2 00 00 00 00

CRT3 00 00 00 00

DAY0

DAY1

DAY2

DAY3

HOLIDAY0

HOLIDAY1

HOLIDAY2

HOLIDAY3

ROA_DATA

OPT ROX

RICI

TIM_DATA

FLSH 45 896

PHDT 30

DIND 30 32 30

DIDT 14 16 14

LDTT 6

DLAT 0

BOTO 14

DBRC 60

RTIM 30 30 30

ATIM 0

AQTT 30

ADLD 0

AFNT 0

NFNA 0

ADHT 0

HWTT 300

NIT 8

FOPT 6

ARDL_ACCEPT 20

ARDL_RETRY 30

TST_DATA

4.3.2 Overlay 17 – Configuration Record

REQ PRT

```
TYPE CFN
ADAN   HIST
  SIZE   25000
  USER MTC BUG
ADAN     TTY 0
  CTYP PTY
  DNUM 0
  PORT 0
  DES  PTY0
  FLOW NO
  USER MTC TRF SCH BUG OSN
  XSM  NO
  TTYLOG       0
  BANR NO
ADAN     TTY 1
  CTYP PTY
  DNUM 1
  PORT 1
  DES  PTY1
  FLOW NO
  USER MTC TRF SCH BUG OSN
  XSM  NO
  TTYLOG       0
  BANR NO
ADAN     TTY 2
  CTYP PTY
  DNUM 2
  PORT 2
  DES  PTY2
  FLOW NO
  USER MTC TRF SCH BUG OSN
  XSM  NO
  TTYLOG       0
  BANR NO
ADAN     TTY 3
  CTYP PTY
  DNUM 3
  PORT 3
  DES  PTY3
  FLOW NO
  USER MTC TRF SCH BUG OSN
  XSM  NO
  TTYLOG       0
  BANR NO
ADAN     TTY 4
```

CTYP CPSI
DNUM 4
PORT 0
DES
BPS 9600
BITL 8
STOP 1
PARY NONE
FLOW NO
USER MTC TRF SCH BUG OSN
XSM NO
TTYLOG 0
BANR NO
ADAN TTY 5
CTYP CPSI
DNUM 5
PAGE 001
PORT 1
DES
BPS 9600
BITL 8
STOP 1
PARY NONE
FLOW NO
USER MTC TRF SCH BUG OSN
XSM NO
TTYLOG 0
BANR YES
ADAN ELAN 16 (Configuration for CallPilot)
CTYP ELAN
DES CPilot
N1 512
ADAN DCH 100
CTYP DCIP
DES VoIP
USR ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA YES
IFC SL1
CNEG 1
RLS ID 5
RCAP ND2 **MWI (Configuration for CallPilot)**
MBGA NO

```
  H323
    OVLR NO
    OVLS NO
ADAN     DCH 101
  CTYP DCIP
  DES  Enterprise
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA NO
  IFC  SL1
  CNEG 1
  RLS  ID  25
  RCAP ND2 MWI
  MBGA NO
  H323
PAGE 002
    OVLR NO
    OVLS NO
PARM
  LPIB 3500
  HPIB 3500
  500B 2000
  SL1B 255
  DTIB 35
  DTOB 4
  NCR  20000
  MGCR 25
  CSQI 255
  CSQO 255
  TUBO NO
  NCPU 2
  CFWS NO
  PCML A
  ALRM YES
  ERRM ERR BUG AUD
  DTRB 100
  ABCD NO
  TMRK 128
  FCDR OLD
  PCDR NO
  TPO  NO
  TSO  NO
  CLID NO
```

```
 DUR5 NO
 MLDN NO
 MARP YES
 IPIE NO
 FRPT NEFR
 DCUS NULL
 DTDT NO
 MSCL 0
 PMSI
  MANU PMS1
  PMCR 0
  PORT NONE
 NDIS 20
 OCAC NO
 MTRO MR
 SBA_ADM_INS 000
 SBA_USER 512
 BCAP SPEECH
 IDLE_SET_DISPLAY
 ICON NO
 MSEC ON
  MSSD MSBT
  NKEY 31
  TKEY 24
CEQU
 MPED 8D
 TERM
 REMO
 TERD
 REMD
 TERQ
 REMQ
 SUPL V000 V096 V100 V200
 SUPC
PAGE 003
 SUPF
 DDCS       MG_CARD
 DTCS
 XCT
 CONF
 MGTDS      IPMG  IPMG_TYPE
      126  000  0   MGC
 MGCONF     IPMG  PORTS  IPMG_TYPE
      127  000  0   30    MGC
 MFSD * 126
```

```
  APVL
  MISP        MG_CARD
  SYNM 0
  EXT0  3PE
  EXT1  3PE
  MCFN  011 MB
 OVLY
  SID  0
  BKGD 044
  PBXH X
  TODR 00
  DROL 030 032 045 135 137
  MID_SCPU NO
  CY45 00
  MULTI_USER OFF
 VAS
  VSID   016
   DLOP
   ELAN 016
    SECU NO
    INTL 0001
    MCNT 9999
  VSID   022
   DLOP
   ELAN 022
    SECU YES
    INTL 0001
    MCNT 9999
  VSID   034
   DLOP
   ELAN 034
    SECU YES
    INTL 0001
    MCNT 9999

  VSID   035
   DLOP
   ELAN 035
    SECU NO
    INTL 0001
    MCNT 9999

  VSID   038
   DLOP
   ELAN 038
```

```
    SECU YES
    INTL 0001
    MCNT 9999
PAGE 004
ATRN
  CODE 0
  SOLR 12
  ROLR +45.00
  AOLR +45.00
  TOLR -45.00
  AGCD NO
  VOLR NO
  HRLR +42.00
  HTLR -44.00
ESA
  LIS EXT/DM
  DYNAMIC_ELIN_TIMEOUT 180
  DYNAMIC_ELIN_REUSE YES
  EXT_DM_UPDT_TIMEOUT 15
```

4.4 CS1K Tandem Configuration

This configuration is for the deployment model of 2 or more CS1000 with Paetec communication system. Represent here is for 2 CS1000s configuration.

4.4.1 Configure CS1000E A

1. **Create IP on CS1000E**

This section describes the steps for creating Node ID (1001) in CS 1000 network.
Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Add...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 1001 was added to be configured as the SIP gateway to the Enterprise services.

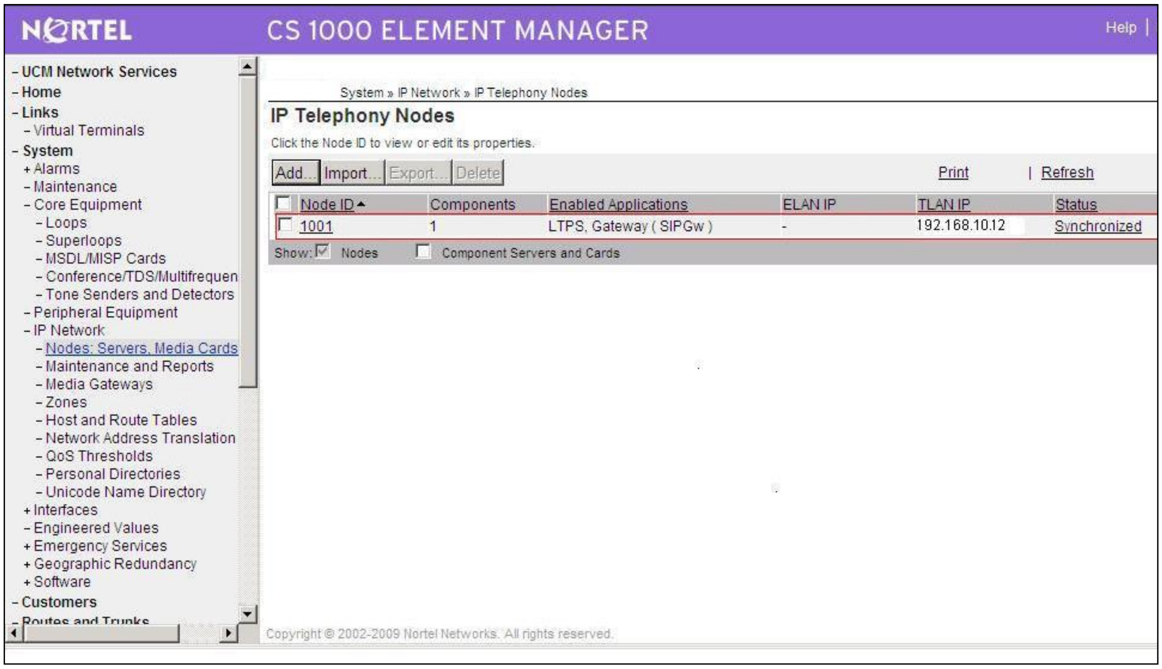

Figure 32 – Creating a node on CS1000 A

The node IP information is added. For the primary proxy enter the IP address of the SIP Proxy Server (SPS). Use UDP port 5060 for SIP communication, figure 33

Support registration

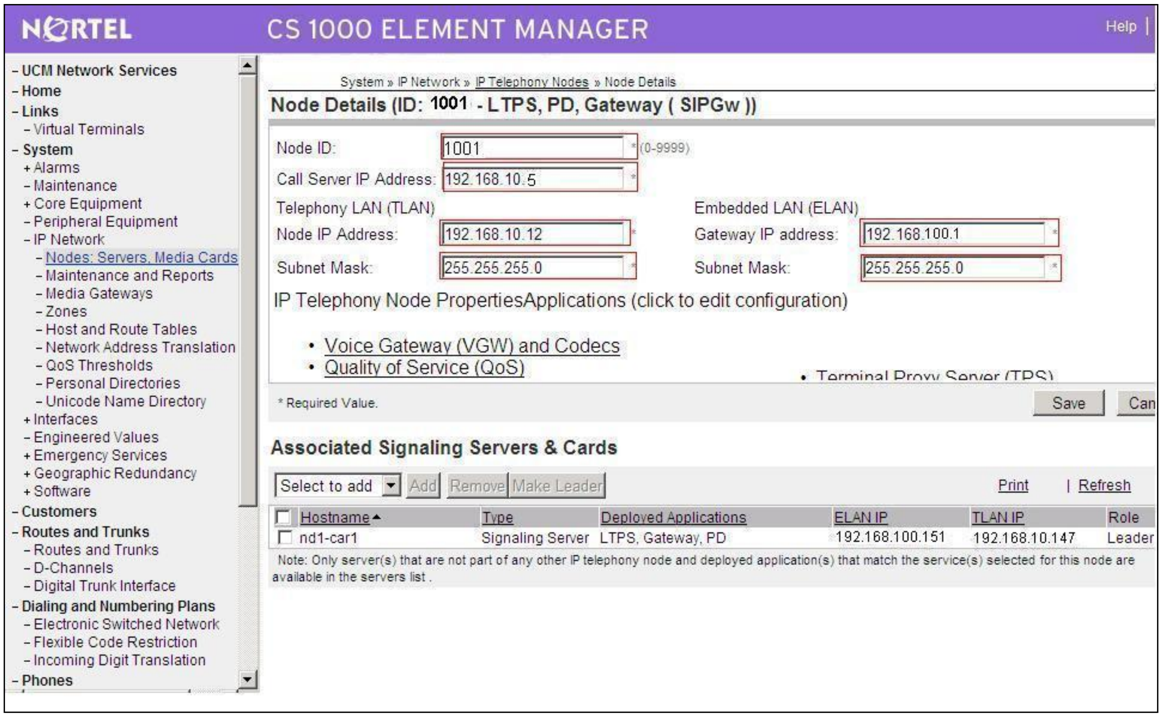

Figure 33 – CS1000 Node Detail Settings

In the signaling server properties, the Line TPS will be enabled if this signalling server will be used for IP set registration. If the role of this server is SIP gateway only then this can be left unchecked, figure 34.

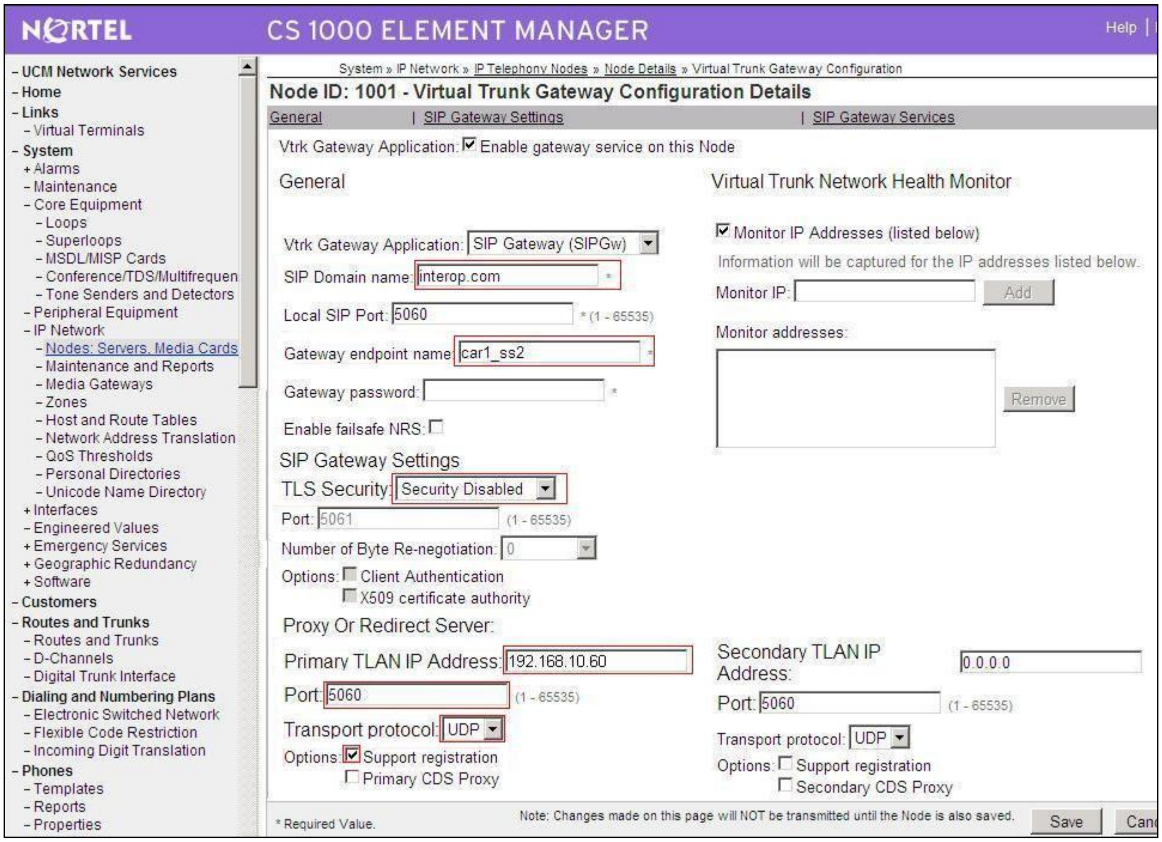

Figure 34 – SIP Gateway Settings

2. **Create D-channel (DCH)**
   - Launch Element Manager of CS 1000 6.0
   - Choose D-Channels, enter D-channel number (i.e.: 101), select DCH for type
   - Click Add to create DCH 101 in figure 35; Also click on Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

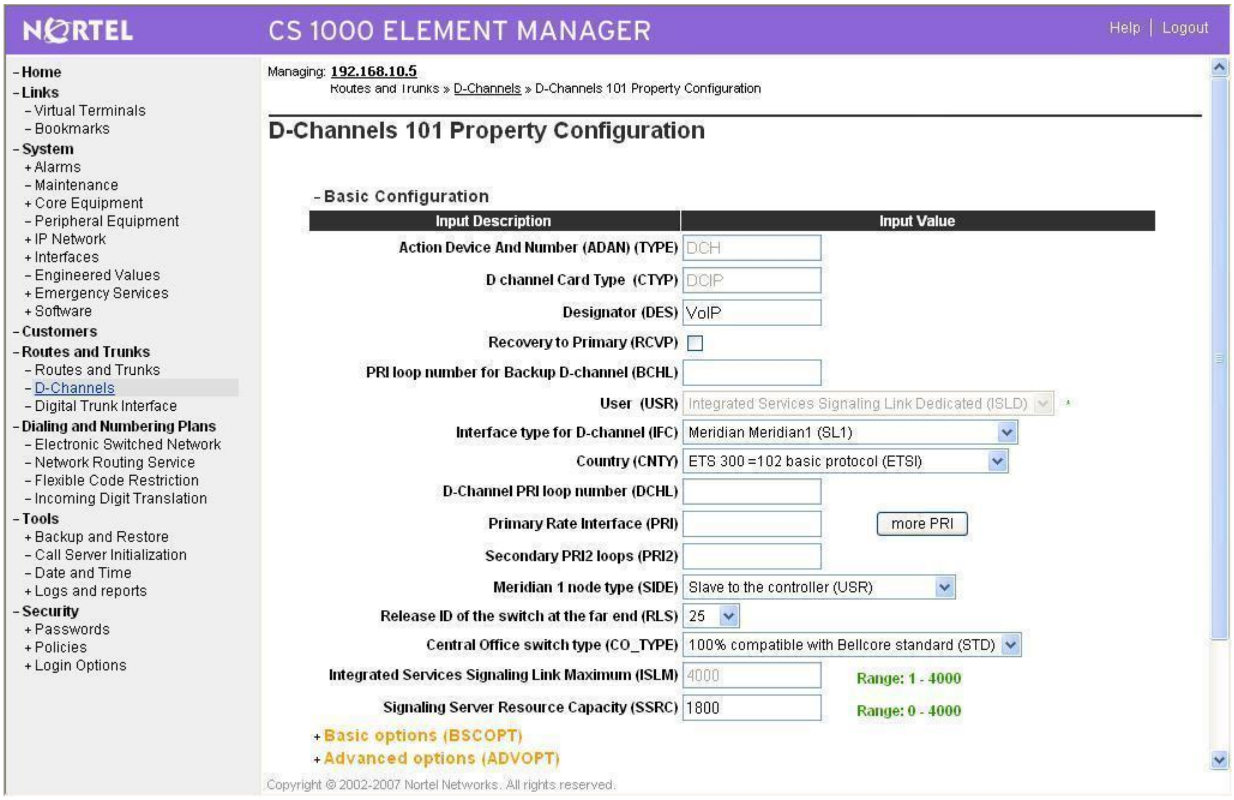

Figure 35 – D-Channels Property Configuration

3. Create Route

Create route 101 using DCH 101 for SIP trunks figure 36

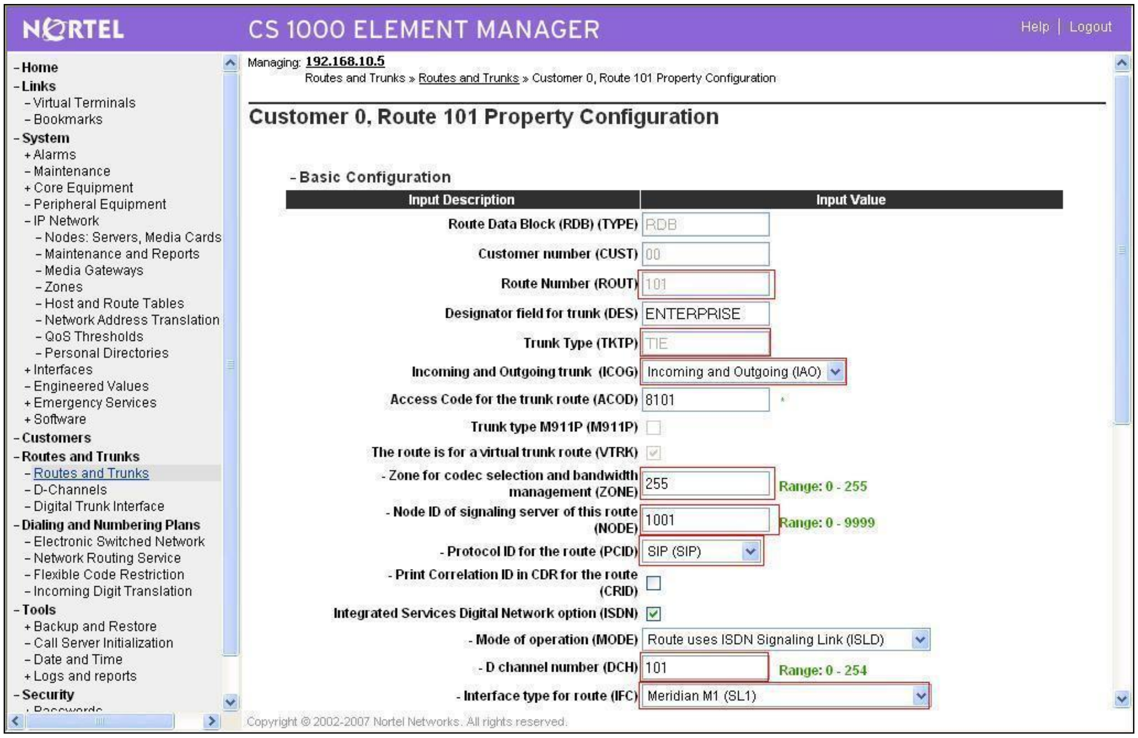

Figure 36 – Route Property Configuration

Configure Route 101 for SIP trunks continue, figure 37

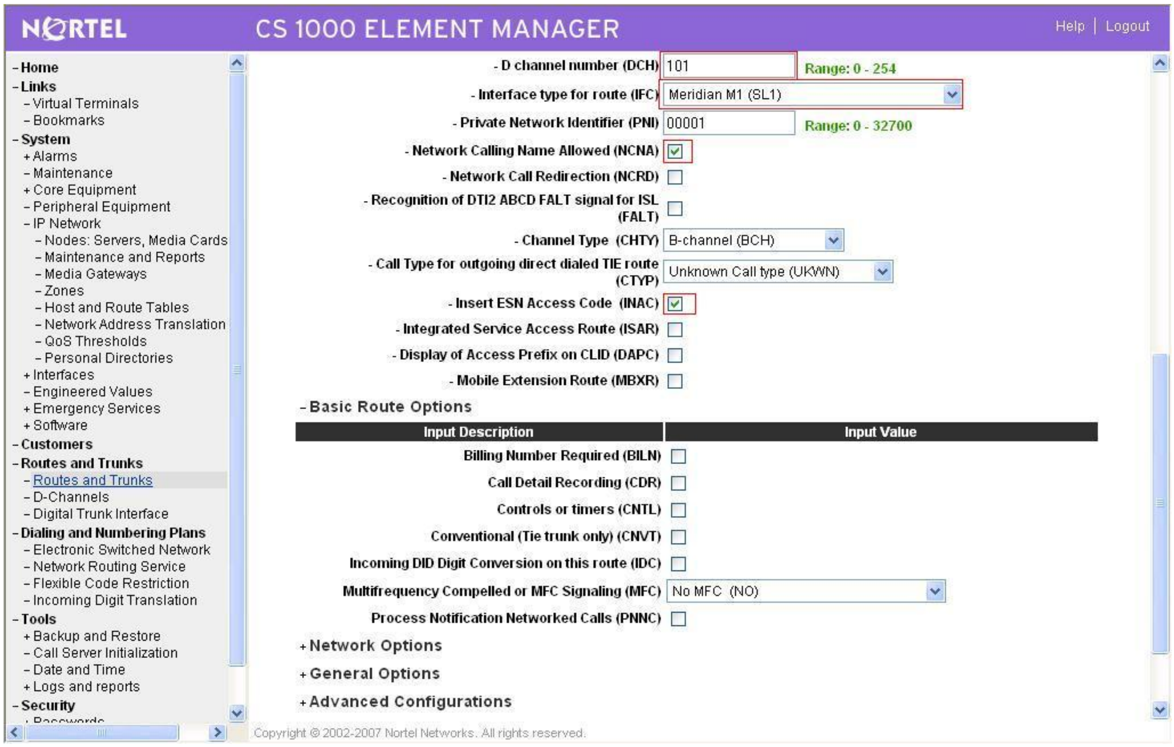

Figure 37 – Route Property Configuration Details (cont.)

4. **Create Trunk (figure 38)**

Since Media security is not support under Paetec system, Disable Media Security (SRTP) at the Trunk level as show in figure 38

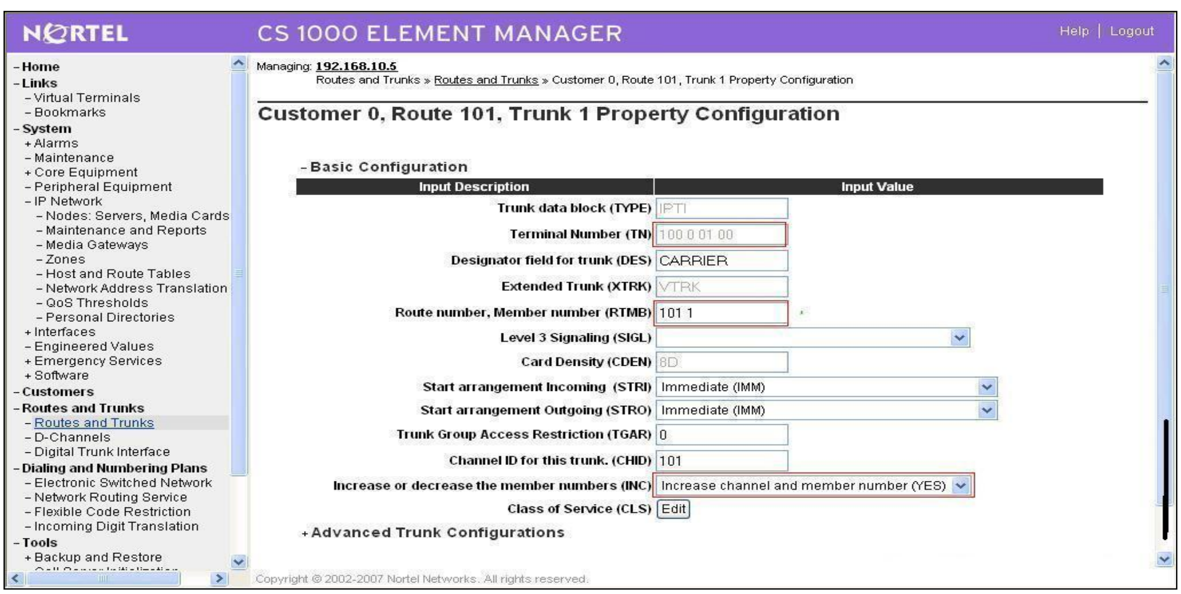

Figure 38 – Trunk Configuration Details

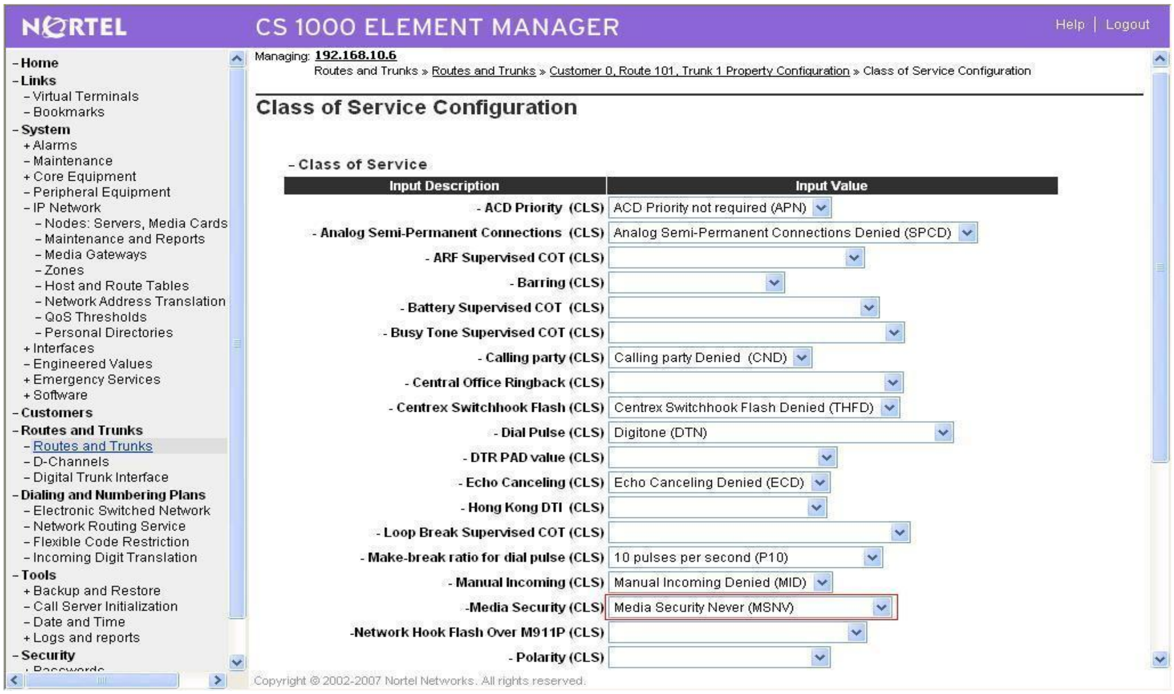

Figure 39 – Class of Services configuration Details (cont.)

- **Configure Dialing Plan for CS1000E_A**

  Create Location Code:

  Create LOC 521 for basic outgoing calls to CS1000E_B (Use RLI_5; DMI_0); Figure 40

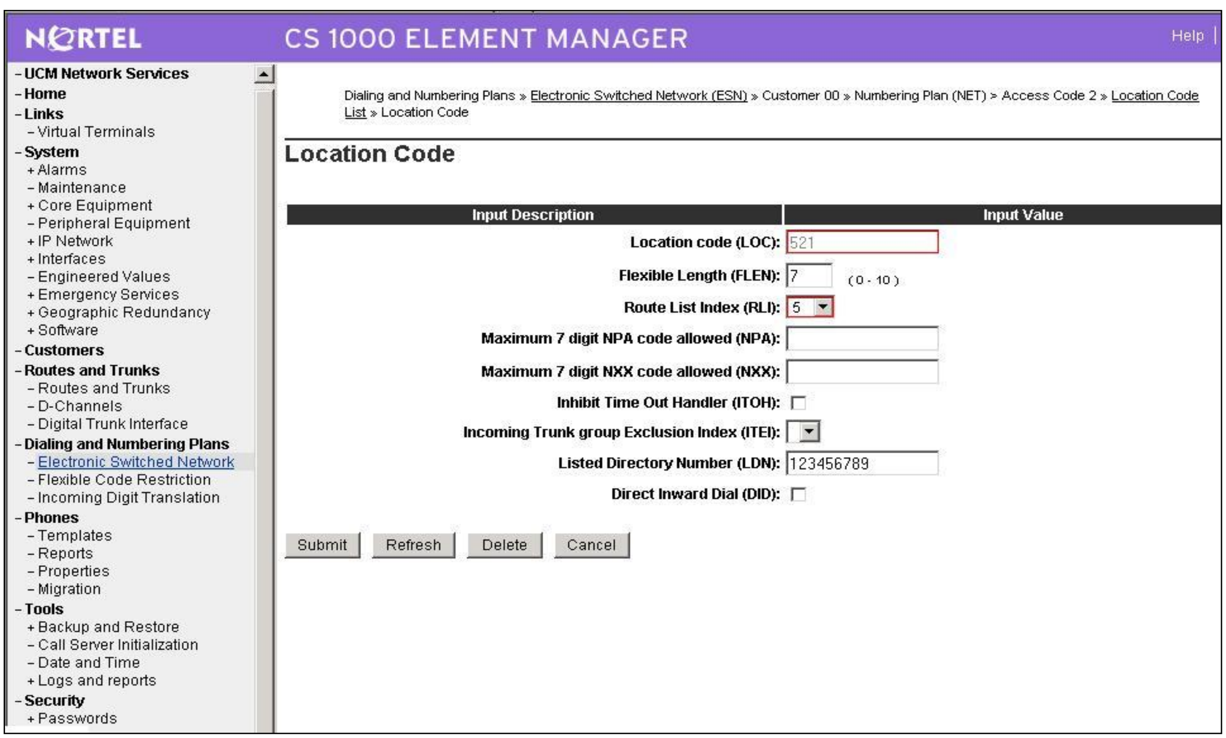

Figure 40 – LOC_521 for basic outgoing calls to CS1000EB

Create Home Location Code

Create HLOC_613 for incoming calls from CS1000E_B and outgoing calls to PSTN; Figure 41

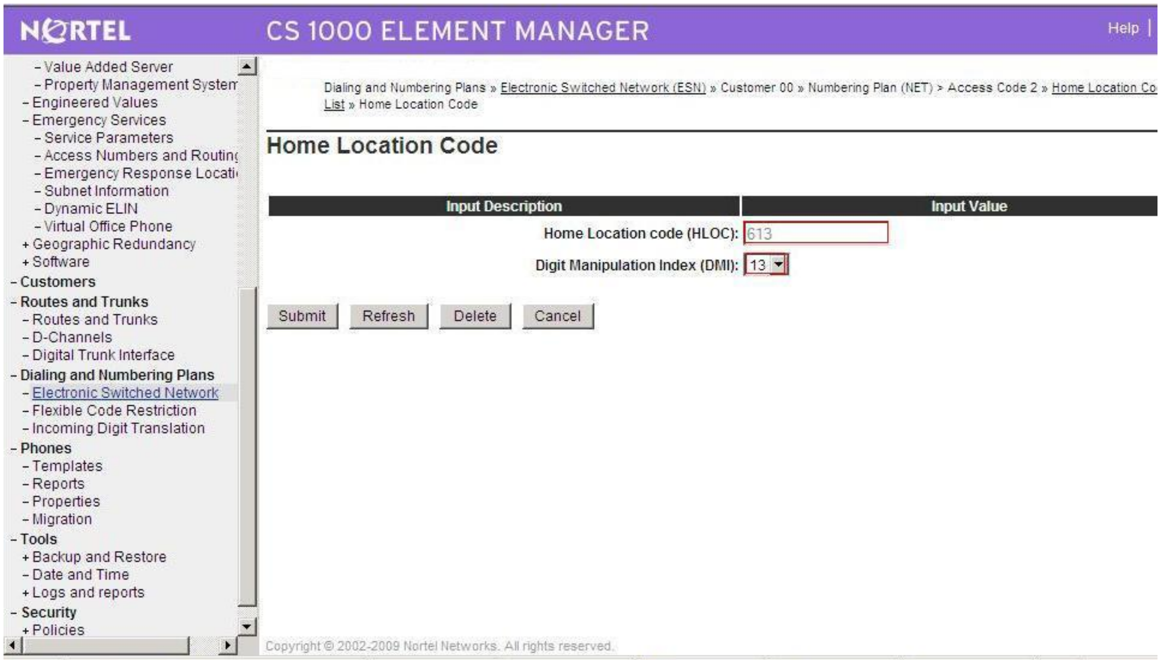

Figure 41 – HLOC: 613 to tandem calls from CS1000B to PSTN

Create HLOC_511 for basic incoming call from CS1000E_B (DMI_4); Figure 42

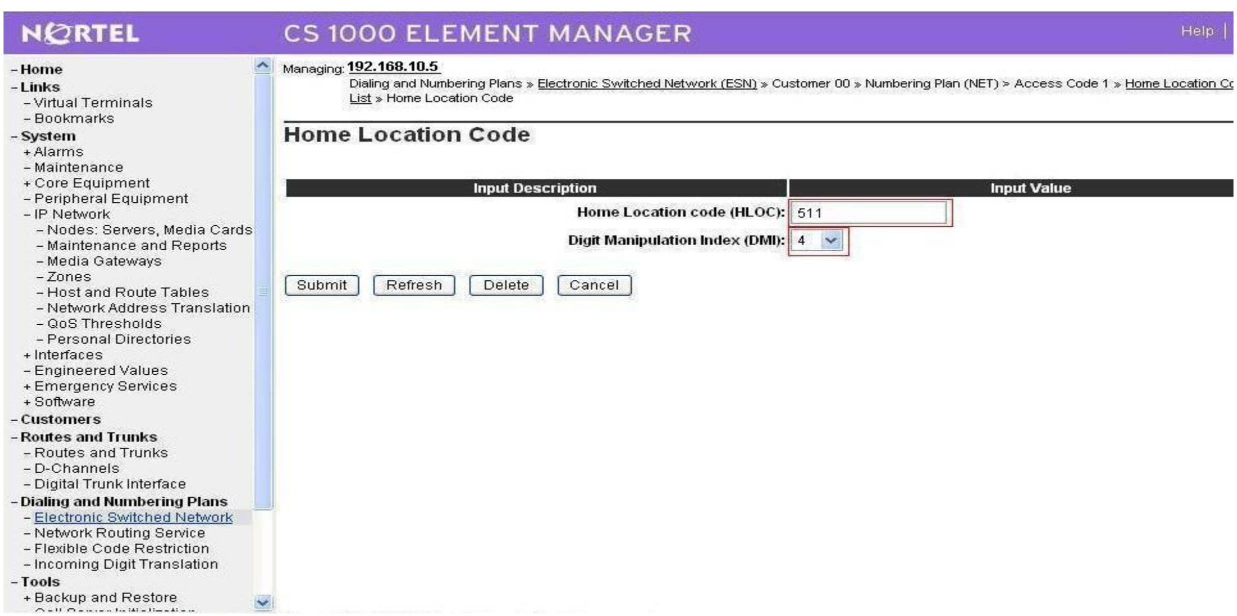

Figure 42 – HLOC_511 to terminate calls from CS1000E_B

Create Distant Steering Code

Create DSC_713 (RLI_6) to receive Calls from PSTN and tandem to CS1000E_B; Figure 43

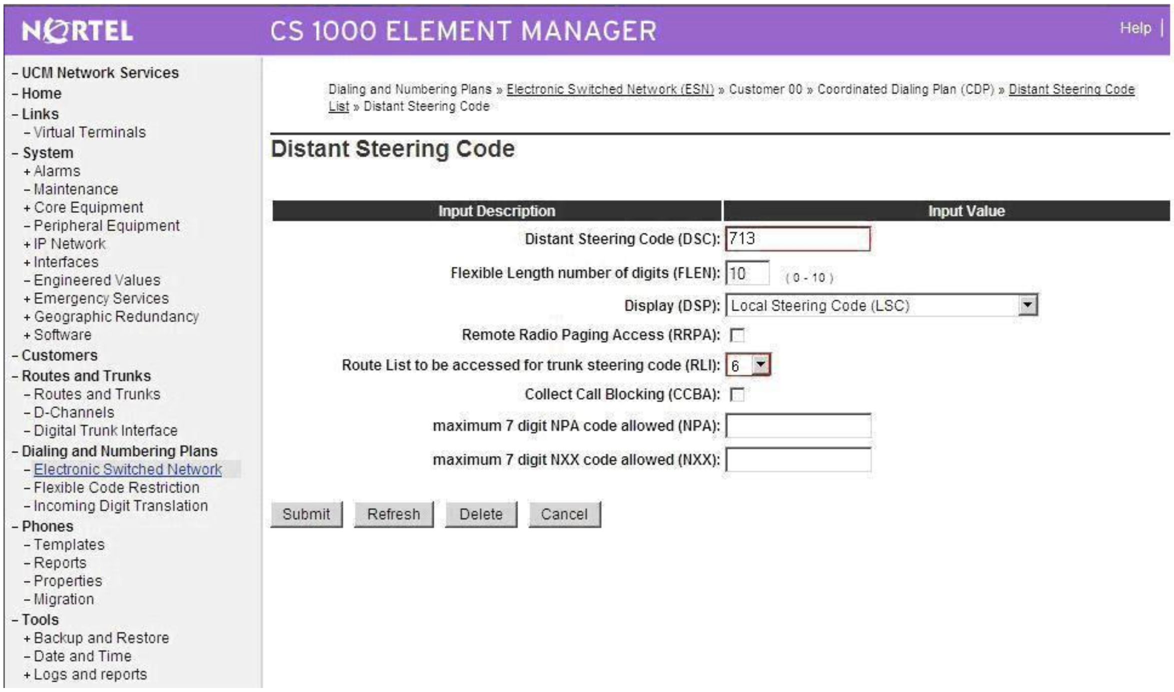

Figure 43 – DSC_713 to receive calls from PSTN and tendem to CS1000E_B

Create DSC_711 to tandem calls from CS1000E_A to CS1000E_B; Figure 44

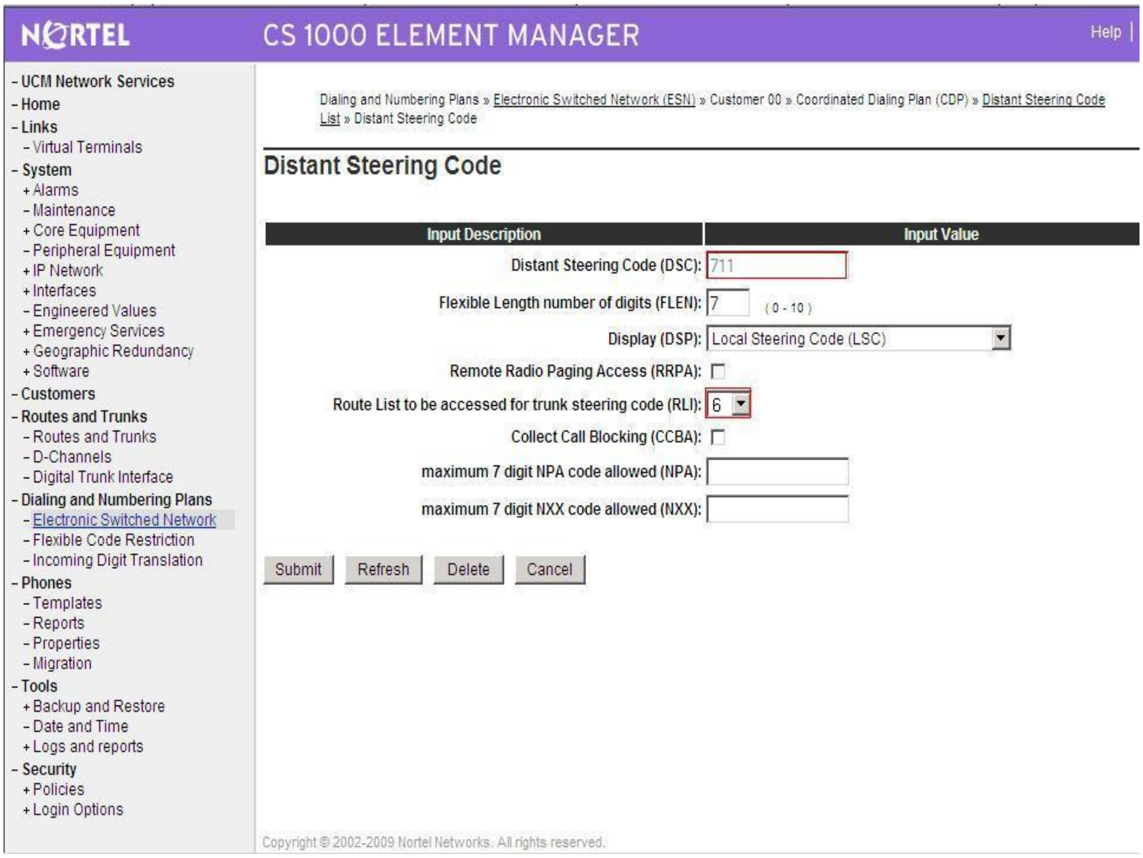

Figure 44 – DSC_711 to tandem calls to CS1000E_B.

Create Route List Block

Create RLI_5 for basic outgoing call to CS1000E_B (Use route 101, DMI_0); Figure 45

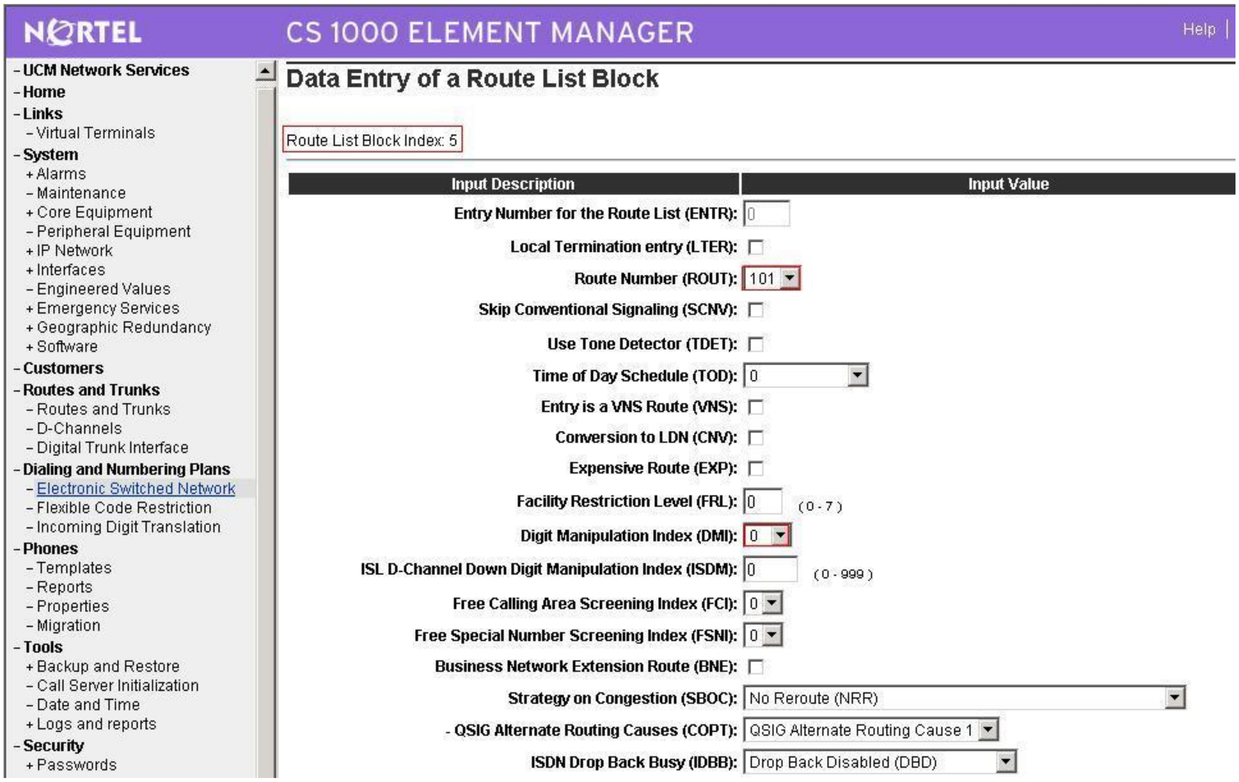

Figure 45 – Create RLI_5 for basic outgoing calls to CS1000E_B

Create RLI_6 to for incoming calls from PSTN and outgoing calls to CS1000E_B (Use route 101, DMI_6); Figure 46

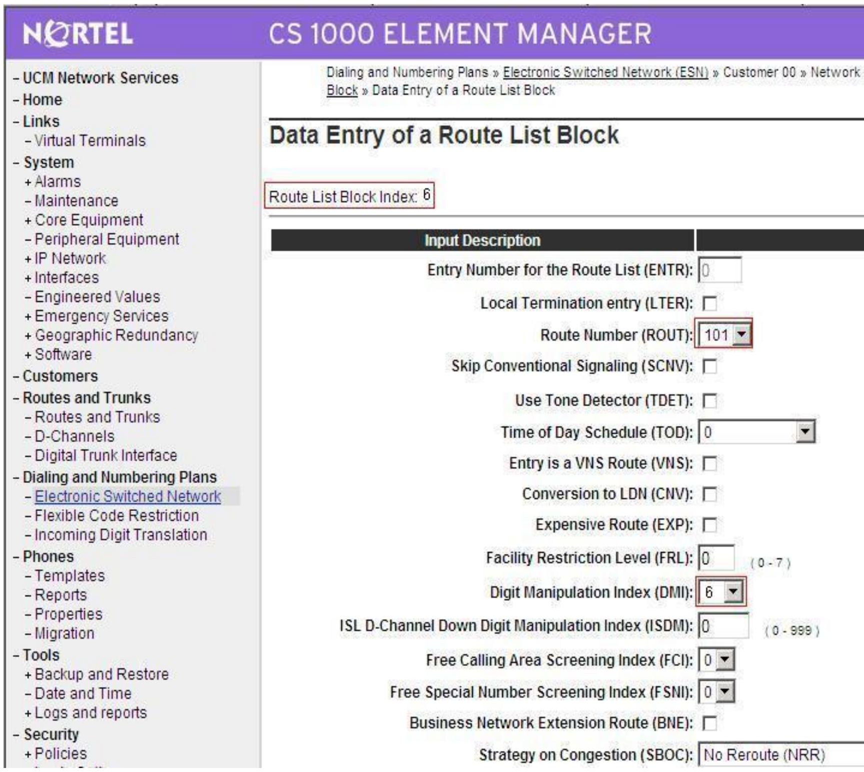

Figure 46 – **RLI_6 to tandem calls from PSTN to CS1000E_B**

Create Digit Manipulation Block

DMI_13 for Incoming calls from CS1000EB and Outgoing to PSTN; Figure 47

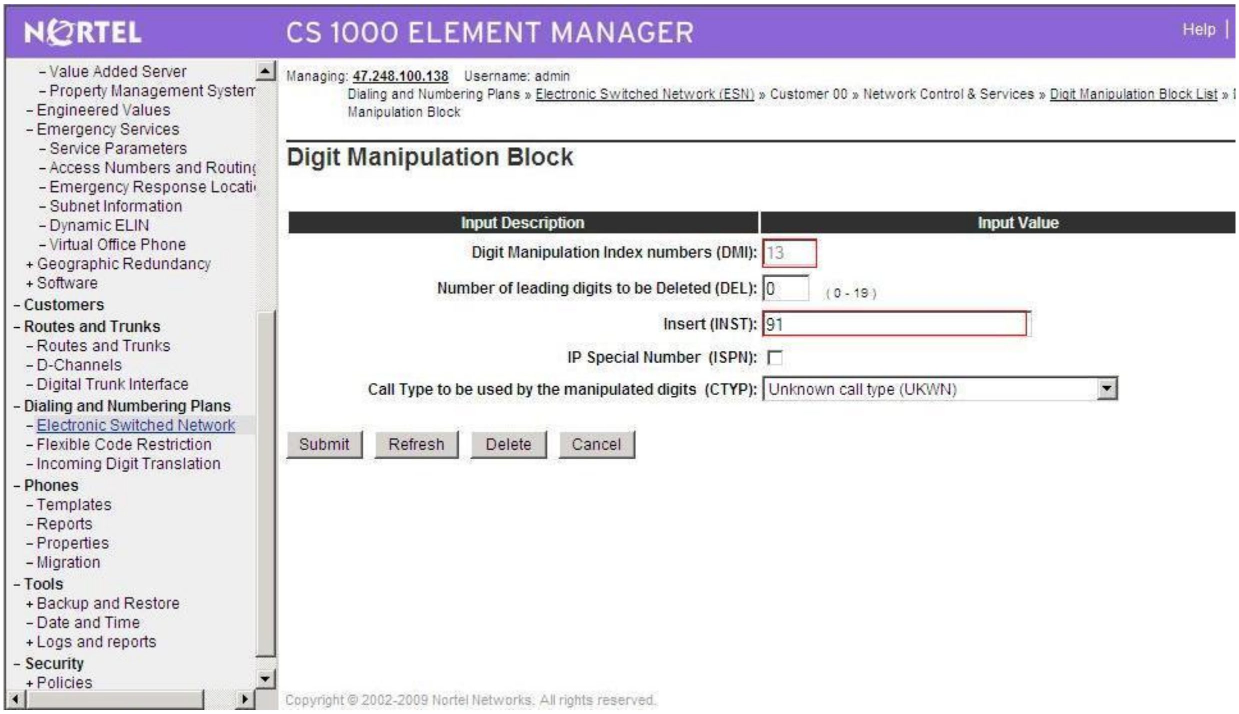

Figure 47 – DMI_13 for incoming from CS1000E_B and outgoing calls to PSTN

Create DMI_4 for incoming calls from CS1000E_B; Figure 48

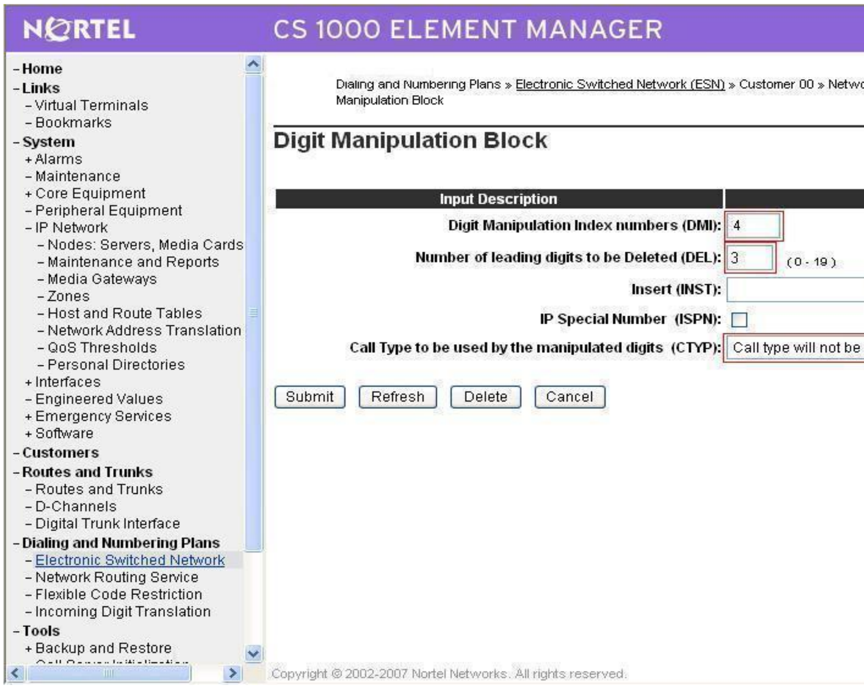

Figure 48 – DMI_4 to terminate calls from CS1000E_B

Create DMI_6: (Delete: 6) for incoming calls from PSTN and tandem calls to CS1000E_B; Figure 49

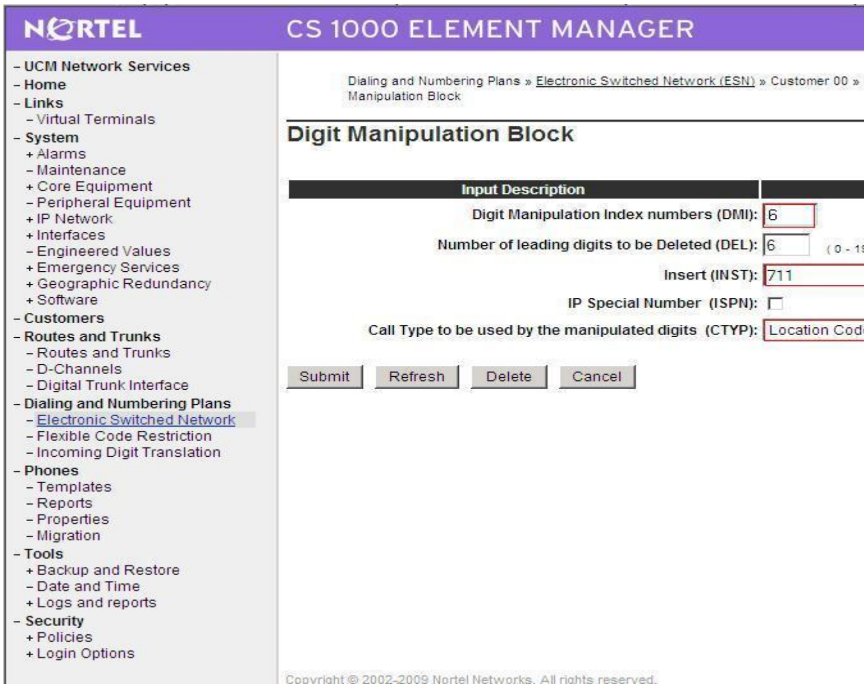

Figure 49 – DMI_6 for incoming calls from PSTN and tandem to CS1000E_B

4.4.2 Configure CS1000E 6.0 B:

1. **Create IP on CS1000E**

This section describes the steps for creating Node ID (2001) in CS 1000 network. Enter Element Manager through the IE browser (in IE address bar, type IP address of the Node IP or TLAN of Signalling Server).

- Input Node ID and press Add...
- Enter TLAN, ELAN IP address of Signalling Server.

Node 2001 was added to be configured as the SIP gateway to the Enterprise services, figure 50

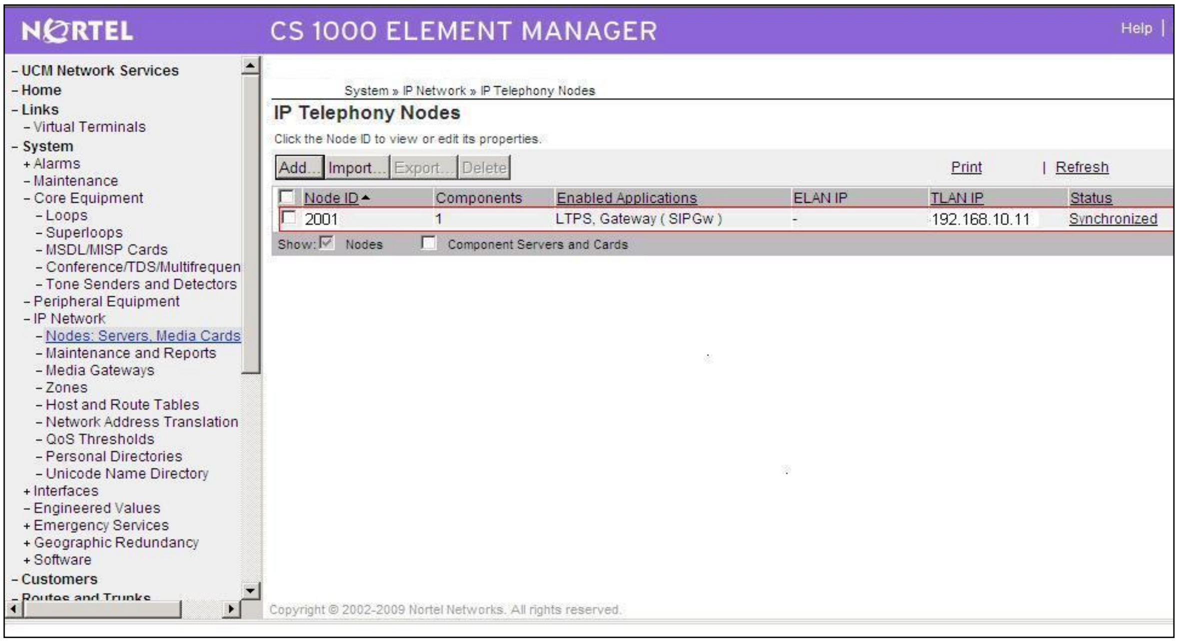

Figure 50 – Node Configured as Enterprise Service SIP Gateway

The node IP information is added. For the primary proxy enter the IP address of the SIP Proxy Server (SPS). Use UDP port 5060 for SIP communication. Support registration, figure 51

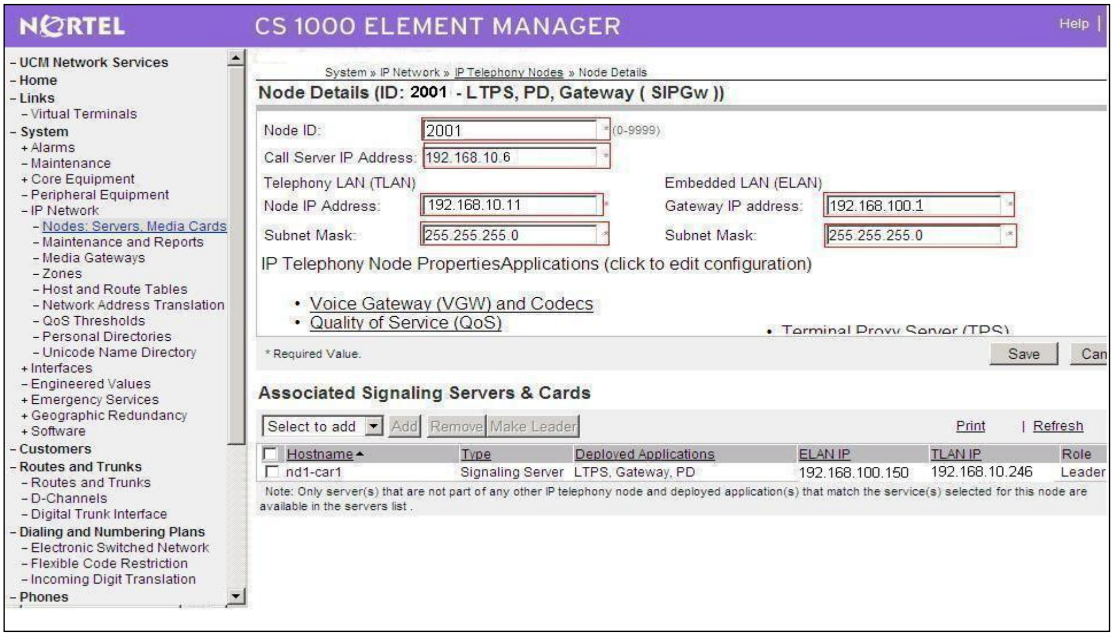

Figure 51 – Node Details Configuration

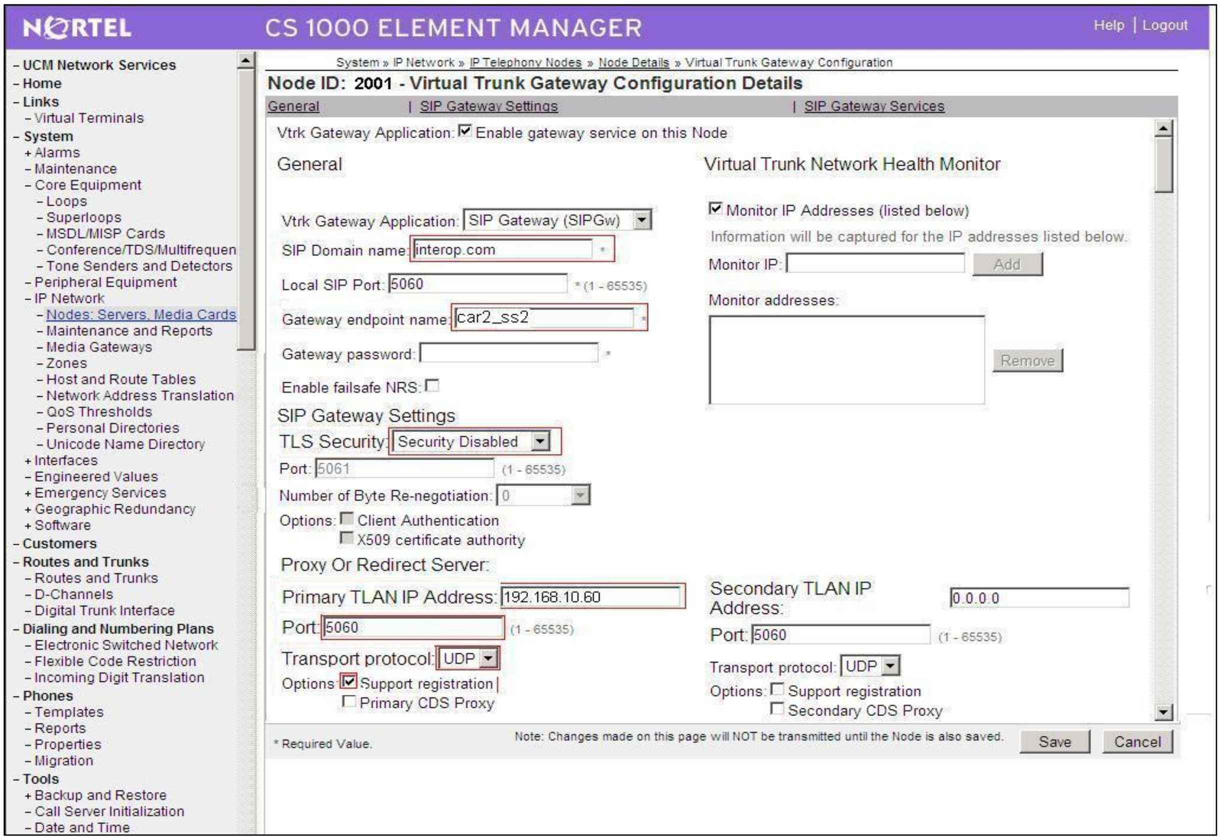

Figure 52 – Trunk Gateway Configuration Details

In the signalling server properties, the Line TPS will be enabled if this signalling server will be used for IP set registration. If the role of this server is SIP gateway only then this can be left unchecked.

2. **Create D-channel (DCH)**

- Launch Element Manager of CS 1000 6.0
- Choose D-Channels, enter D-channel number (i.e.: 101), select DCH for type

Click Add to create DCH 101, figure 53

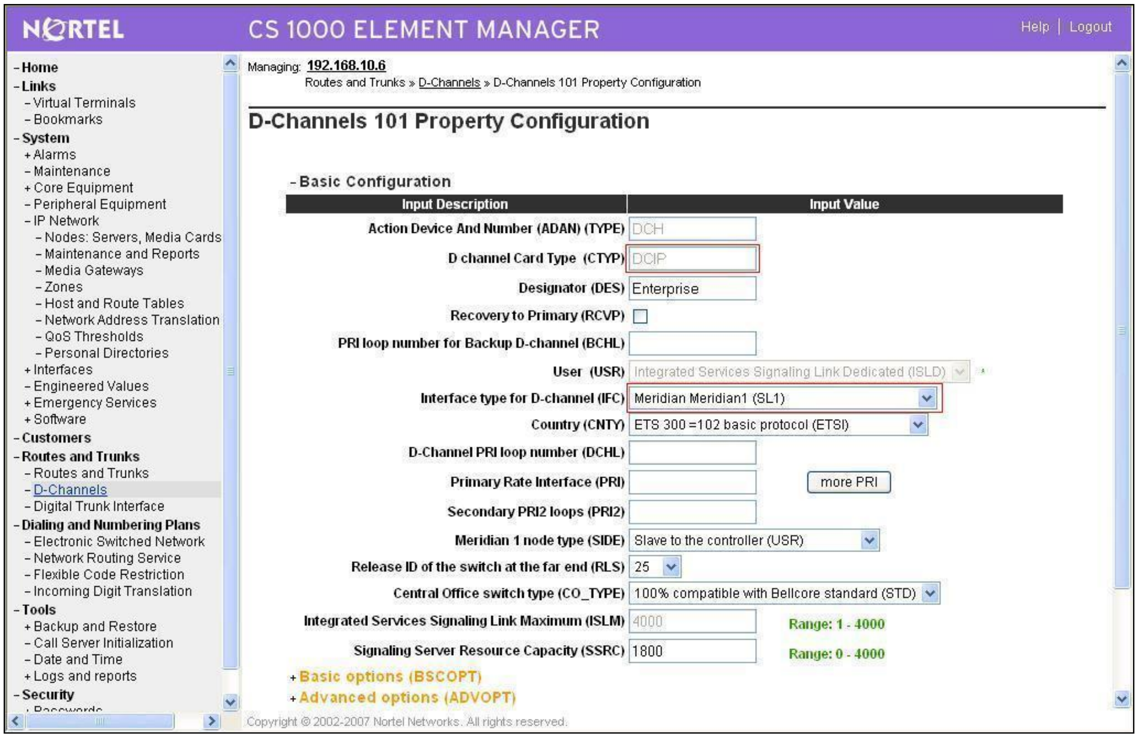

Figure 53 – D-Channel Property Configuration

Also click on Basic Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

3. Create Route

Create route 101 using DCH 101 for SIP trunks, figure 54

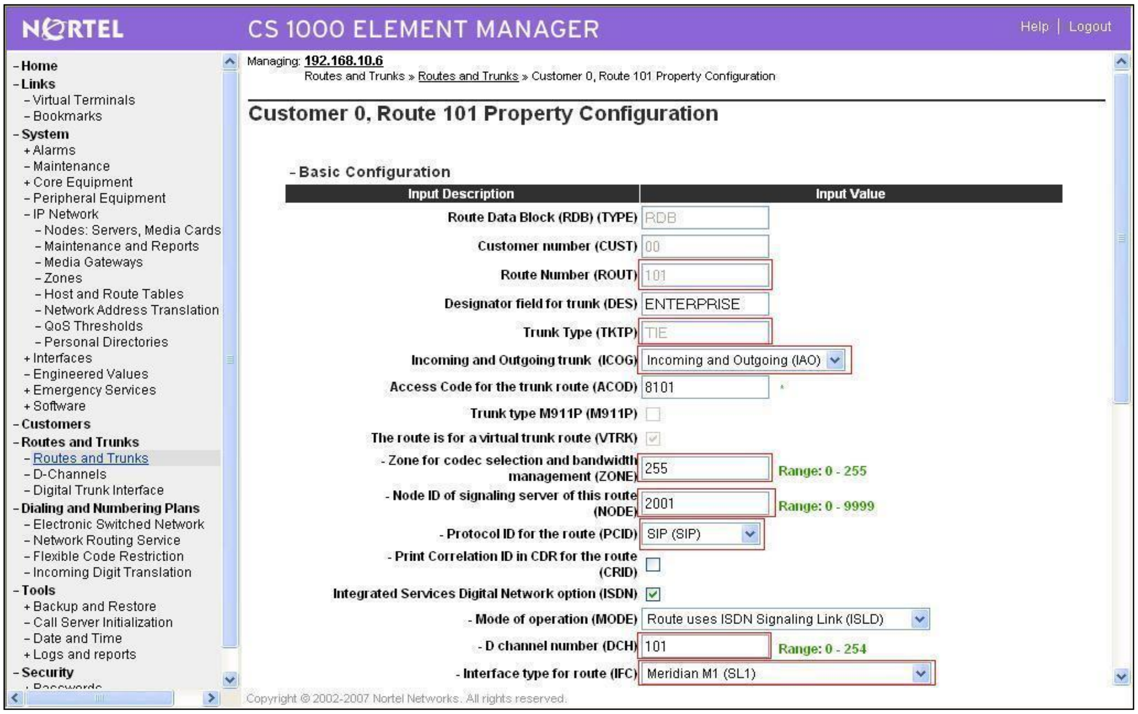

Figure 54 – Route Property Configuration

Configure Route 101 for SIP trunks, figure 55

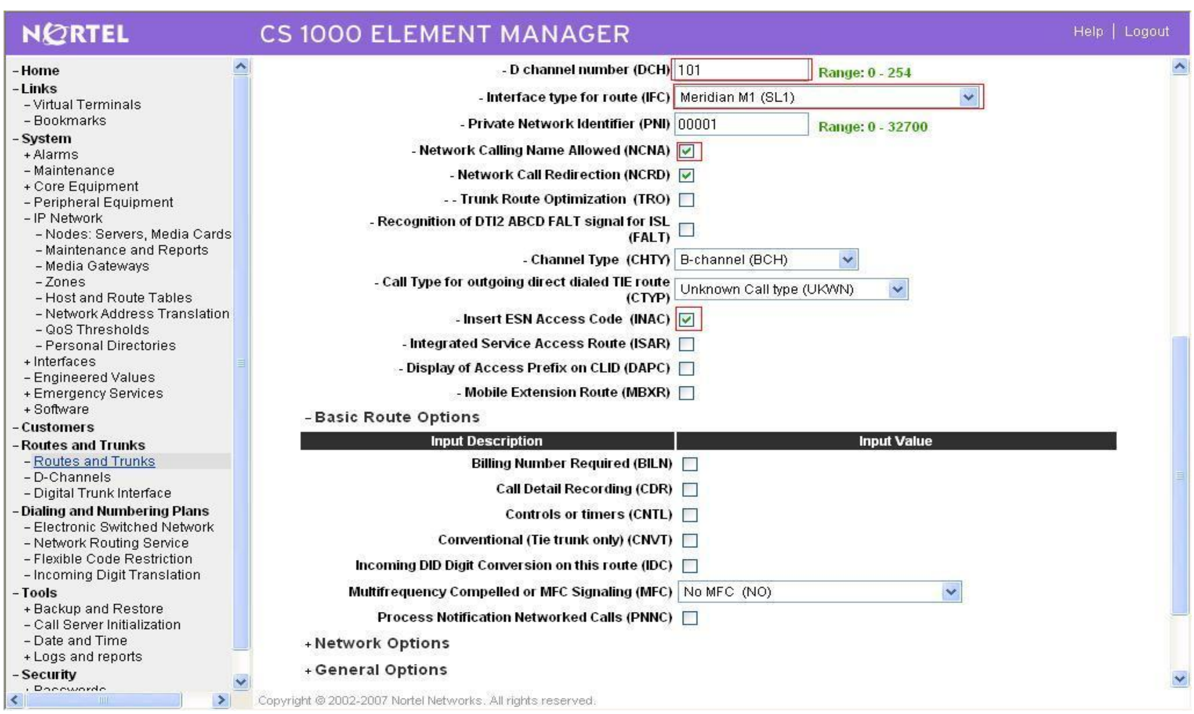

Figure 55 – Route Configuration

4. **Create Trunk (figure 56)**

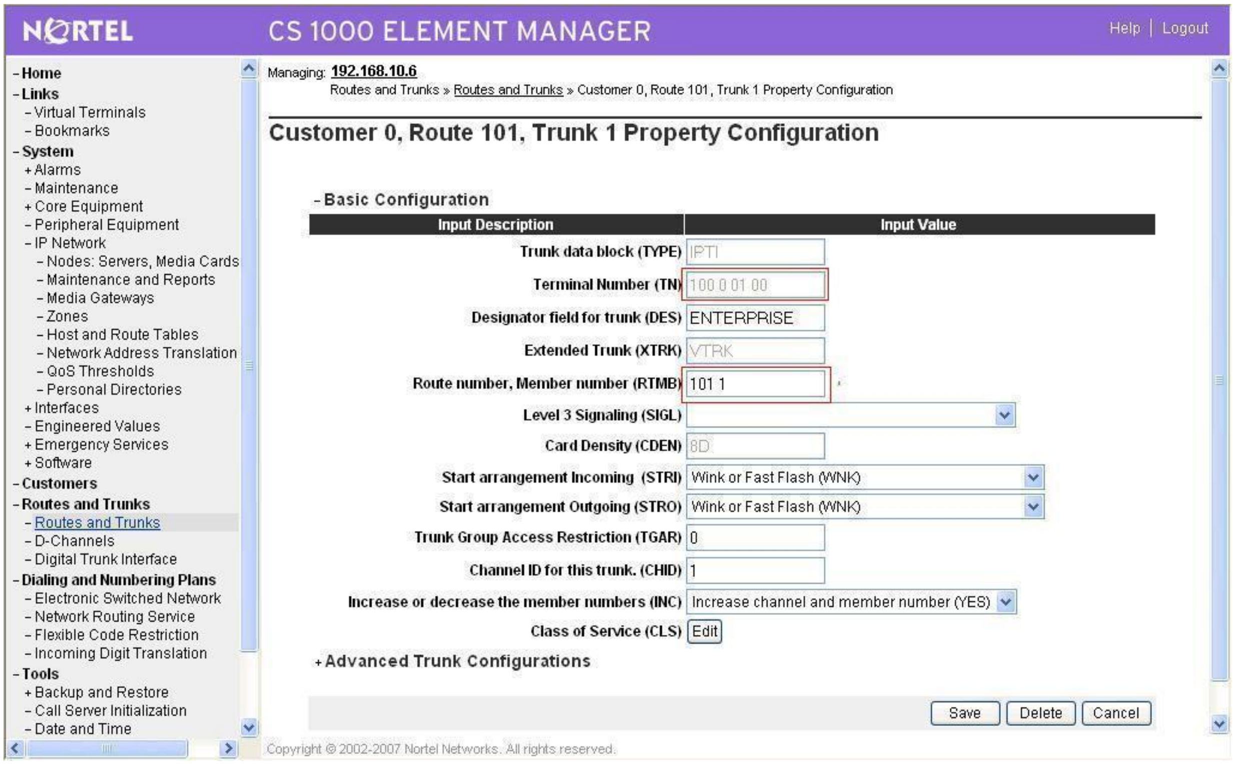

Figure 56 – Trunk Property Configuration

Disable Media Security (sRTP) at the trunk level, figure 57

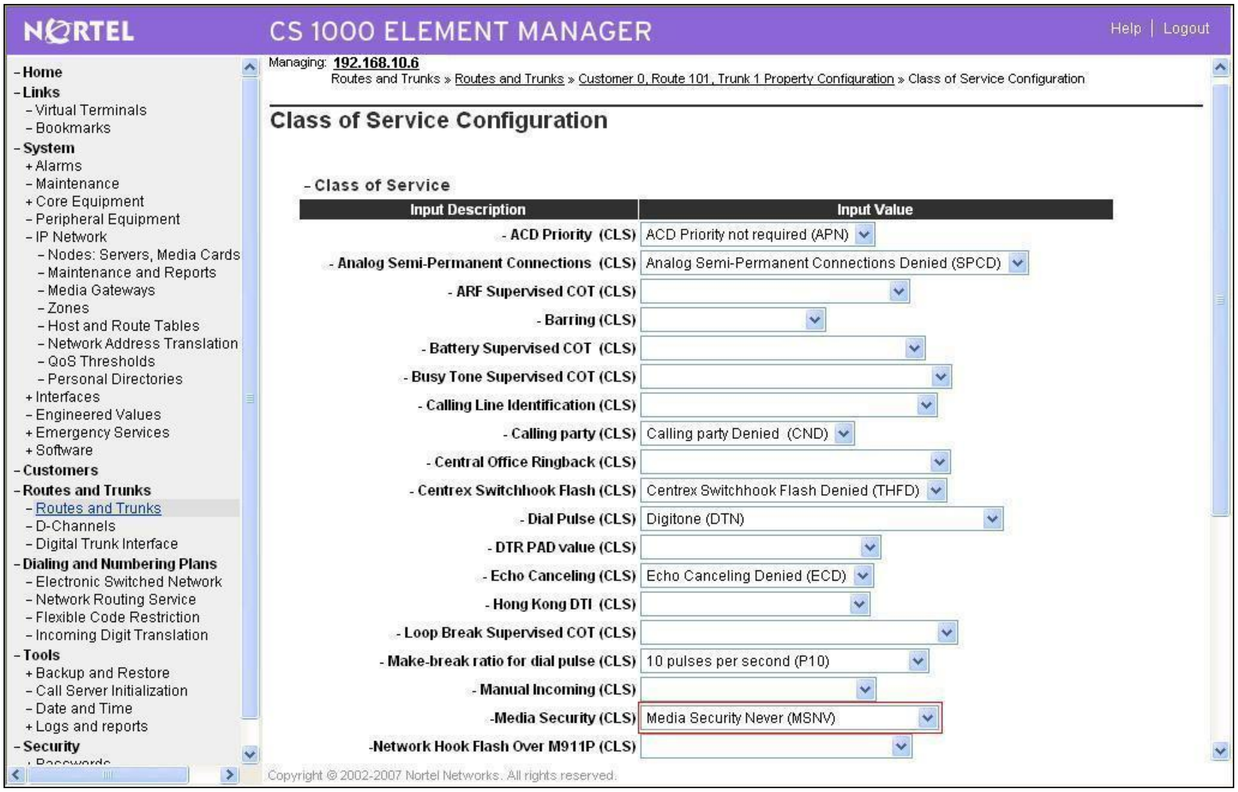

Figure 57 – Class of Service Configuration

5. **Create Dialing Plan**

Create Location Code

Create LOC 511 (Use RLI_5) for outgoing calls to CS1000E_A; Figure 58

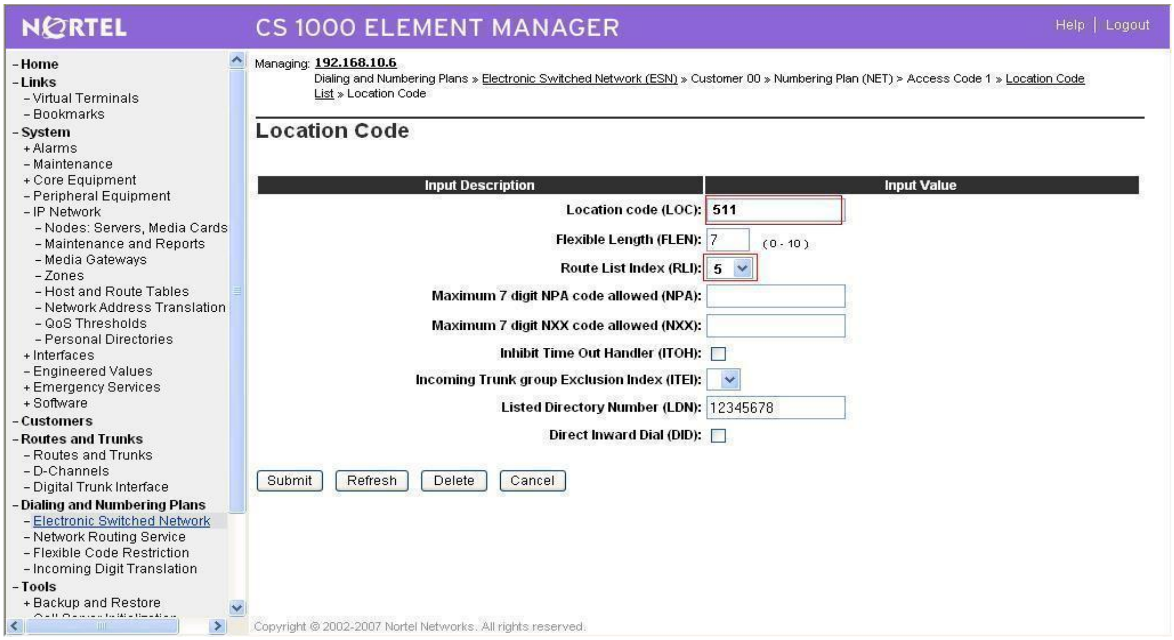

Figure 58 – LOC_511 for out going calls to CS1000E_A

Create Numbering Plan Area Code

Create NPA_613 (RLI_70) for Outgoing calls to PSTN through CS1000E_A; Figure 59

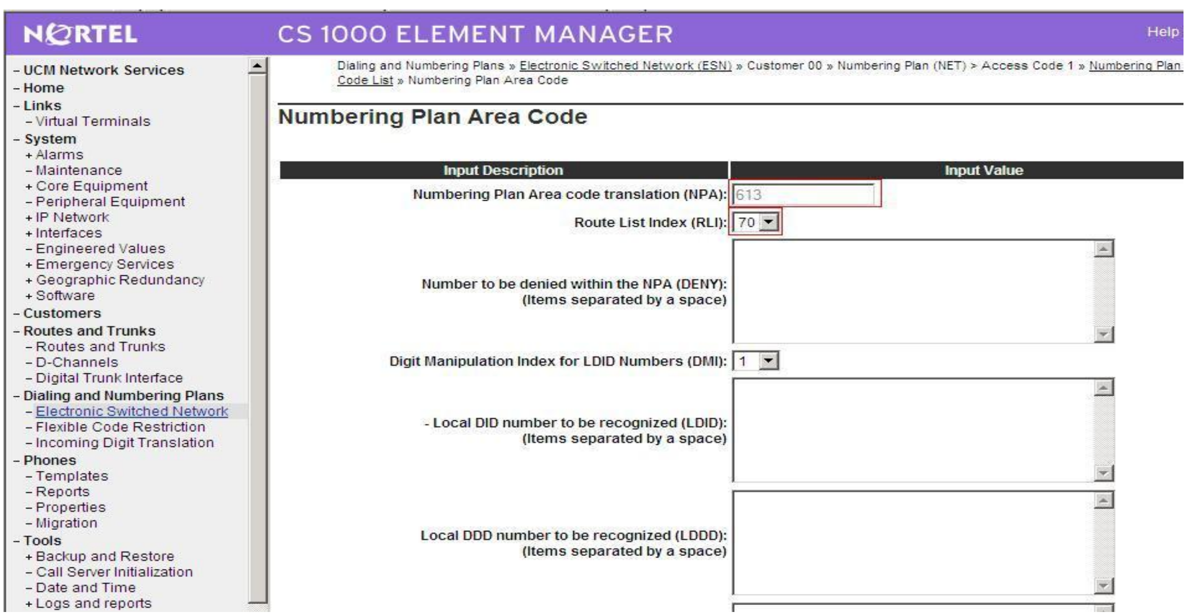

Figure 59 – NPA_613 for Outgoing calls to PSTN

Create Home Location Code

Create HLOC_521 (Use DMI_4) for incoming calls from CS1000E_A; Figure 60

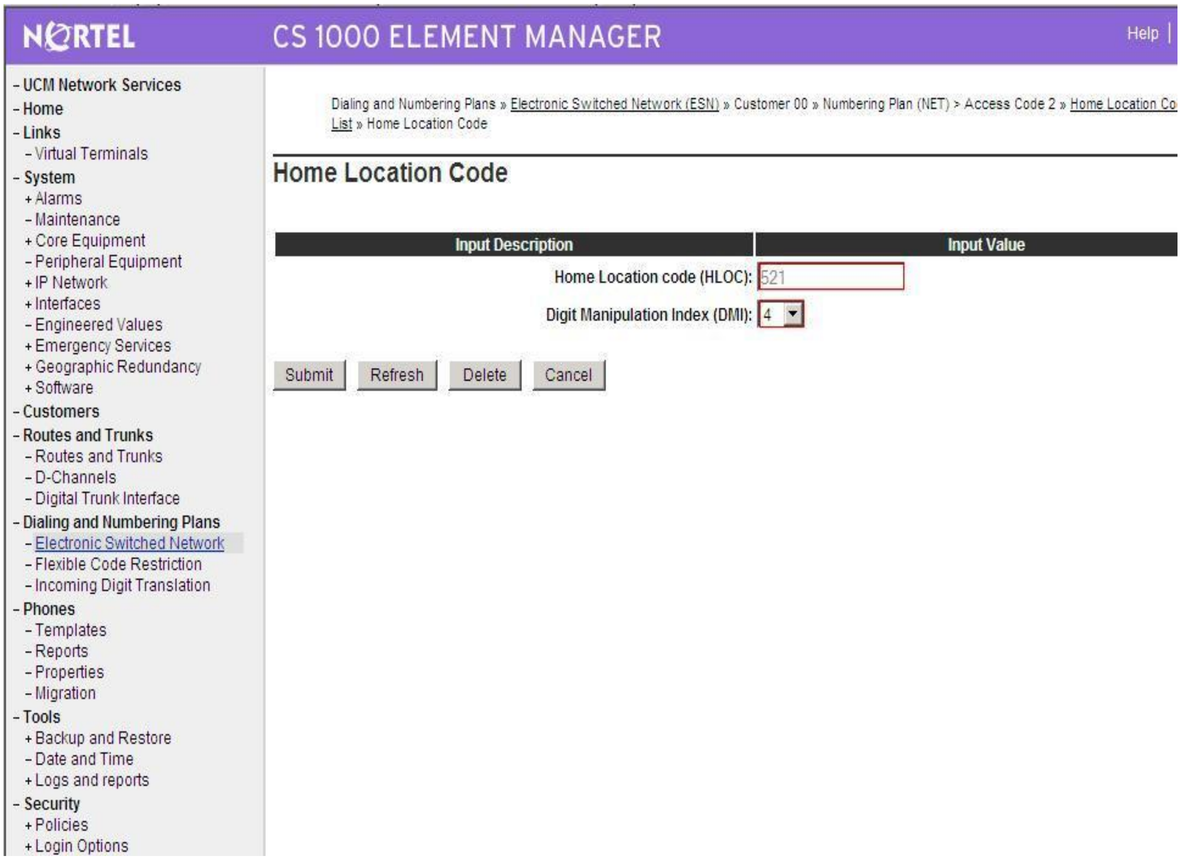

Figure 60 – Create HLOC 521 for incoming calls from CS1000E_A

Create Route List Block

Create RLI_5 for outgoing calls to CS1000E_A (Use DMI_0)

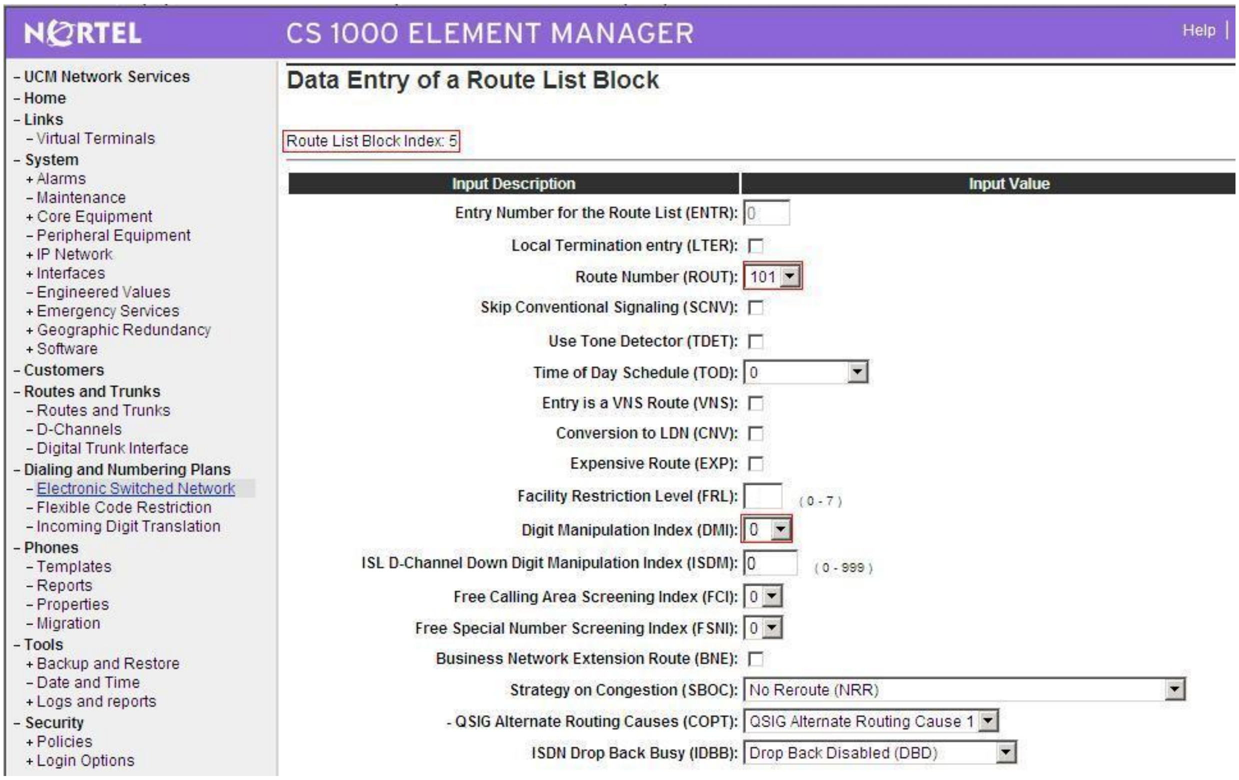

Figure 61 – RLI_5 (use DMI_0) for Outgoing calls to CS1000E_A

4.4.2 Configure SIP Proxy Server (SPS)

Create gateway endpoints on SPS

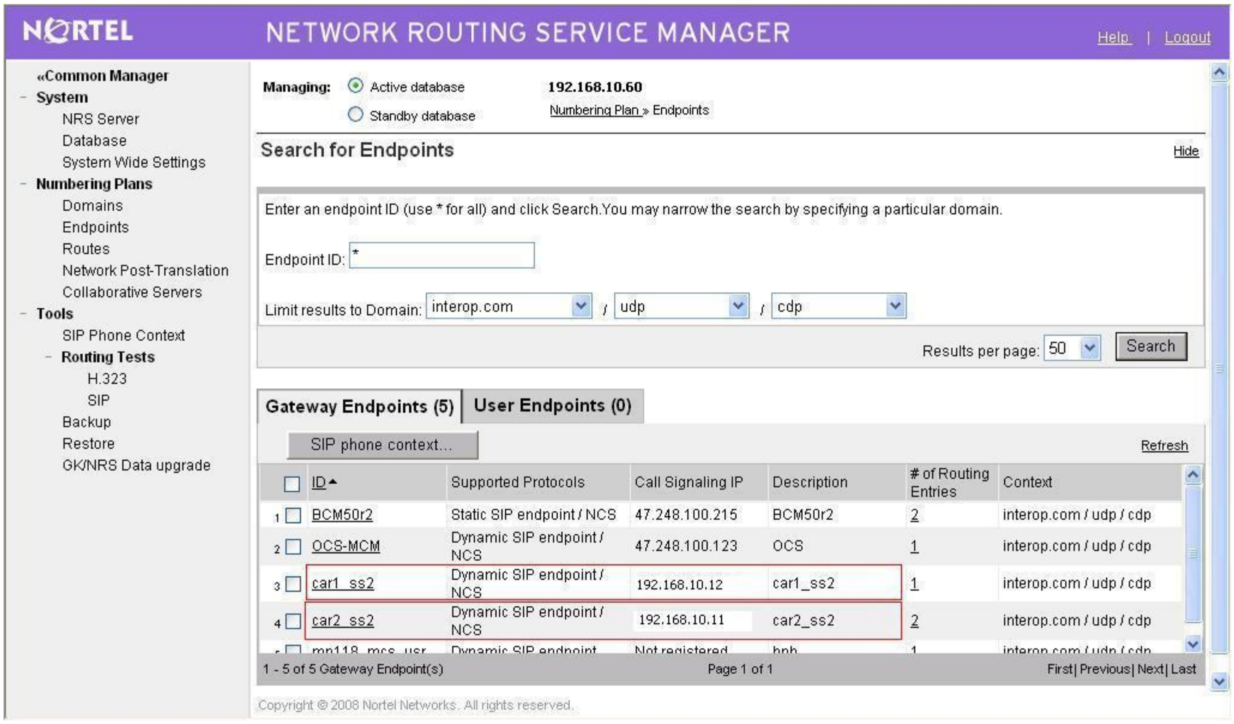

Figure 62 – SIP Gateway Endpoint Creation

Create routing entries for each of gateway endpoints on SPS, figure 63

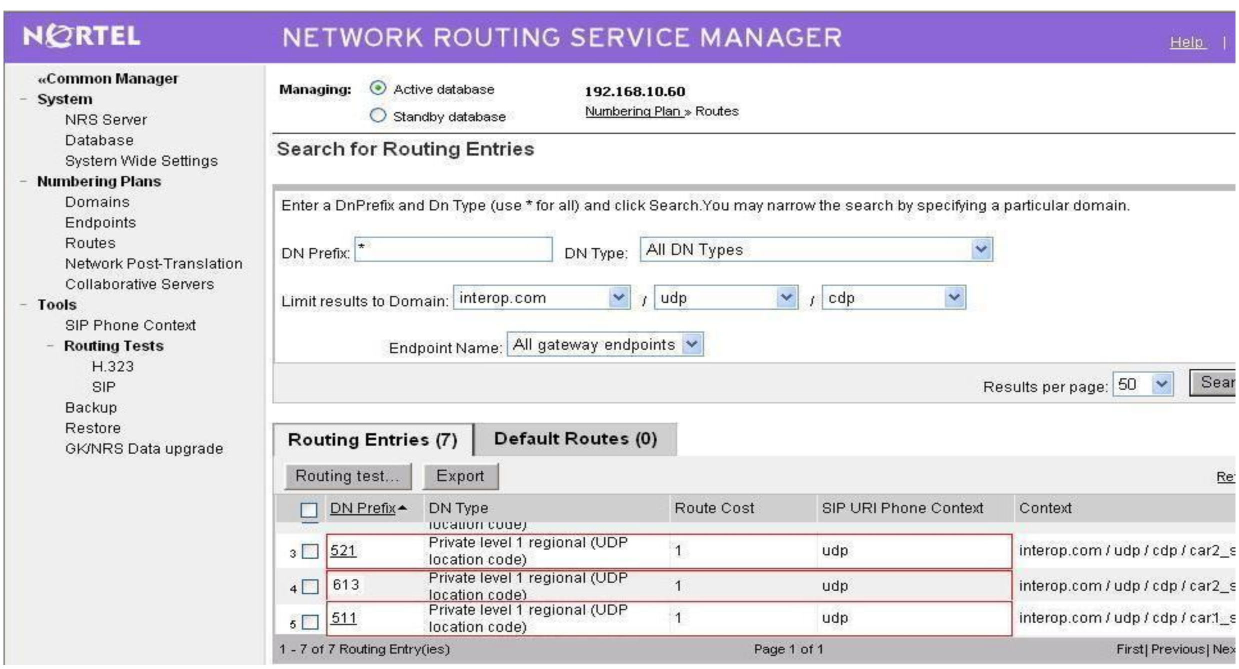

Figure 63 – Routing Entries for Gateway Endpoints

4.4.3 CS1000E SIPLINE CONFIGURATION

In this section, it shows how to configure a SIP LINE system on CS1000E. Follow the bellow steps to set up the SIP LINE server.

4.4.3.1 Configure SIP LINE CS1000E in Element Manager

Figure 64 show hot to add SIP LINE Node 1002 under System -> IP Network -> IP Telephony Nodes

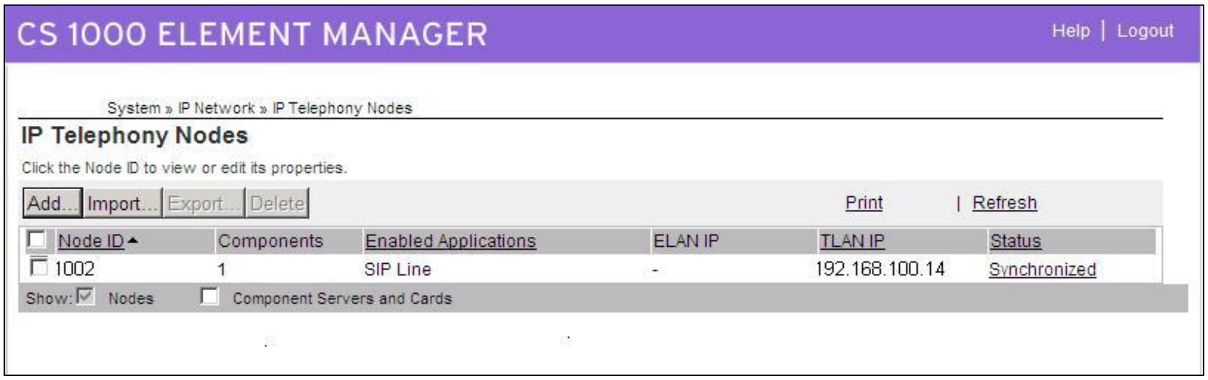

Figure 64 – IP Telephony Nodes

Figure 65, 66 and 67 show how to set up the SIP LINE Node 1002 configuration details

SAVE and SYNC are required – And then APPSTART RESTART on SLG server.

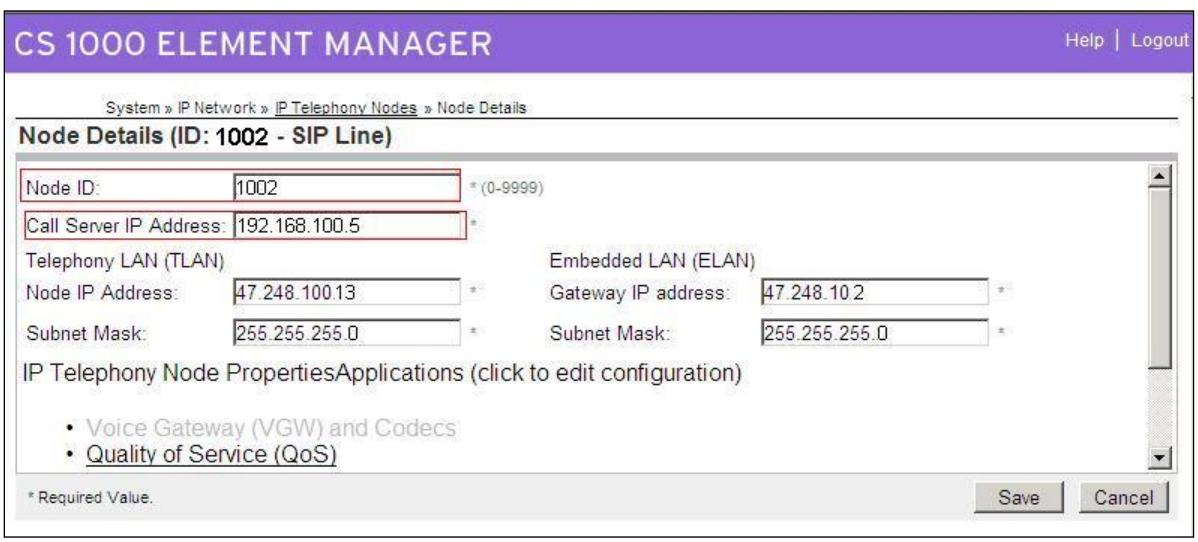

Figure 65 – Node Configuration Details

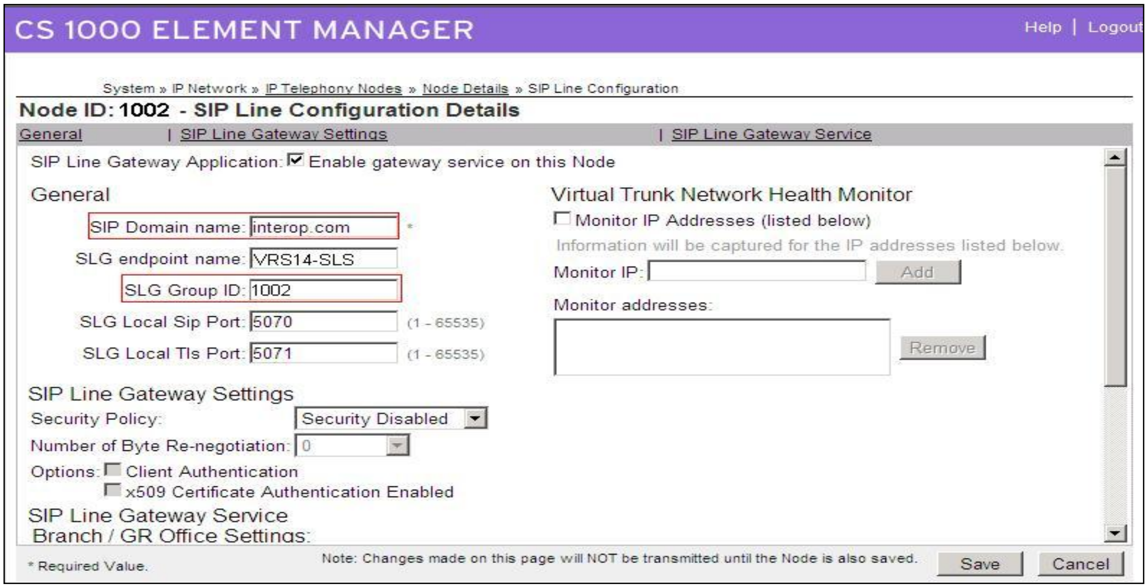

Figure 66 – Node Configuration Details (Cont…)

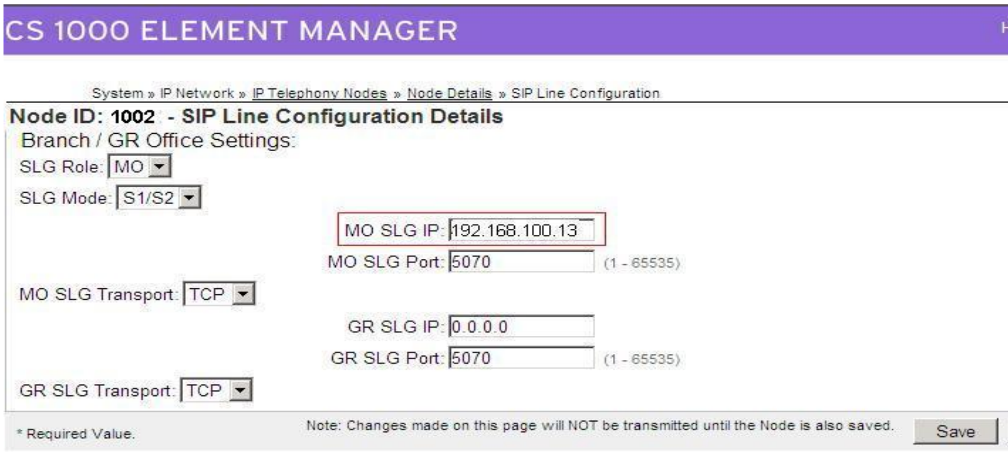

Figure 67 – Node Configuration Details (Cont…)

4.4.3.2 Configure CS1000E Call Server

For the configuration of SIP Line on Call Server, one needs to use command line to set it up. Follow the bellow steps to accomplish that.

Packages Required for SIP line on Call Server of CS1000E, these are keycode enablement

1. SLS_Package – 417 - SIP Line Service
2. FFC- 139 - Flexible Feature Codes
3. SIP_LINE_NT_PKG – 415 - Nortel SIP Line Package
4. SIP_LINE_3P_PKG – 416 - 3rdParty SIP Line Package

4.4.3.2.1 Configure SIPL service in LD15

LD 15
REQ CHG
TYPE SLS
CUST 0
SIPL_ON **YES**
SIPD **INTEROP.COM**
UAPR **222** - DN prefix used to auto-generate UADN for all SIPL clients of this customer
NMME NO

4.4.3.2.2 Configure DCH for SIPL in LD 17

```
LD 17
REQ CHG
TYPE ADAN
ADAN new dch 11
ADAN     DCH 11
 CTYP DCIP
 DES  SIPL
 USR  ISLD
 ISLM 4000
 SSRC 1800
 OTBF 32
 NASA NO
 IFC  SL1
 CNEG 1
 RLS  ID  25
 RCAP
 MBGA NO
 H323
  OVLR NO
  OVLS NO
```

4.4.3.2.3 Configure ELAN AML link in LD 17

LD 17
REQ CHG
TYPE ADAN
ADAN new elan 32
ADAN ELAN **32** – new AML ELAN link, link number should be bigger or equal to 32
CTYP **ELAN**
DES **SIPL**
N1 512

4.4.3.2.4 Configure VAS ID for AML link in LD 17

LD 17
REQ CHG

TYPE VAS
VAS new
VSID **32** – VAS ID number
ELAN **32** – Defined in step 3

4.4.3.2.5 Configure SIPL route

LD 16
REQ new
TYPE rdb
CUST 0
ROUTE 11
DES **SIPL**
TKTP **TIE**
…
VTRK **YES**
ZONE **10** – virtual trunk zone defined in LD117
PCID **SIPL**
…
NODE **1002** – node ID of SIPL node
DTRK NO
ISDN YES
   MODE **ISLD**
   DCH **11** – DCH defined in step 2
   IFC **SL1**
   PNI **00001**
   NCNA **YES**
   NCRD **YES**
   TRO NO
   FALT NO
   CTYP UKWN
   INAC **YES**
   ISAR NO
   DAPC NO
…
ICOG **IAO**
…
ACOD **8011** – route access code

4.4.3.2.6 Configure SIPL trunks

LD 14
REQ **NEW 256** – e.g. create 256 trunks
TYPE **IPTI**
TN   **124 0 0 0** - starting TN for virtual trunks
DES **SIPL**
CUST 0
RTMB **11 1** – route number and member
CHID 1
TGAR **0**
STRI **IMM**
STRO **IMM**
CLS **UNR**

4.4.3.2.7 Check status of the details configuration SIPL link is up on Call Server and SIP line Gateway

On Call Server

>*ld 96
DCH 011 : OPER EST ACTV AUTO DES : SIPL_N1402

On SLG

[nortel@vrf14-sls ~]$ slgShow
=== VTRK ===

===== General ======
SLG State = AppReady
Total User Registered = 1

===== AML Info ======

hAppBlk	TaskName	Tid	LinkState	NumRetry	LinkNum	Trace
0x1226c80	SLG	0xfb00	**Up**	0	32	0

4.4.3.2.8 Configure SIP Line Client

Setting password length for SIP line client using LD15

LD 15
REQ CHG
TYPE: **FFC**
TYPE FFC_DATA
CUST 0

SCPL **4** – password length is 4

4.4.3.2.9 Configure UEXT for SIPL client

LD 11
REQ NEW
TYPE UEXT

TN **104 0 00 11** - Virtual TN for SIPL client
CUST **0**
UXTY **SIPL** – UEXT type must be SIPL
MCCL **YES**
SIPN 1
SIP3 1
FMCL 0
TLSV 0

** Begin Note:
Sigma phone: SIPN-SIP3-FMCL-TLSV = 1-0-0-0
SMC3456: SIPN-SIP3-FMCL-TLSV = 1-0-0-0

SipToneV: SIPN-SIP3-FMCL-TLSV = 0-1-0-0

***End Note

SIPU **4861** – SIPL userID, often set equal to DN of the phone
NDID **1002** – NodeID of the SIPL node
ZONE 001 – MO zone configured in LD 117
TGAR **0** – should be 0, if not we can dial to SipToneV
…
SCPW **1234** – password for SIPL client to login
…
CLS **UNR**
…
KEY 00 SCR **4861** – DN of the phone
CPND **NEW** – in case you want to set CLID for phone
NAME **set4861**
XPLN 20
DISPLAY_FMT FIRST,LAST
01 HOT U **2224861** – autogenerate when you enter information for KEY 0

4.4.3.2.10 Check current status set registration on SLG

```
[nortel@vrf14-sls ~]$ slgSetShowAll

=== VTRK ===
UserID          TN              Clients  Calls  SetHandle
--------------- --------------- -------  -----  ---------
           4861     104-00-00-11       1      0  0xb7d8a0c8
```

4.4.4 SMC3456 softphone

Link to download:
http://livelink-ott.ca.nortel.com/livelink/livelink.exe?func=ll&objId=34471954&objAction=browse&sort=name&viewType=1.

After installation on the PC and apply the Licence key which is required for activate the SMC to be used. Run the SMC3456, you will see figure

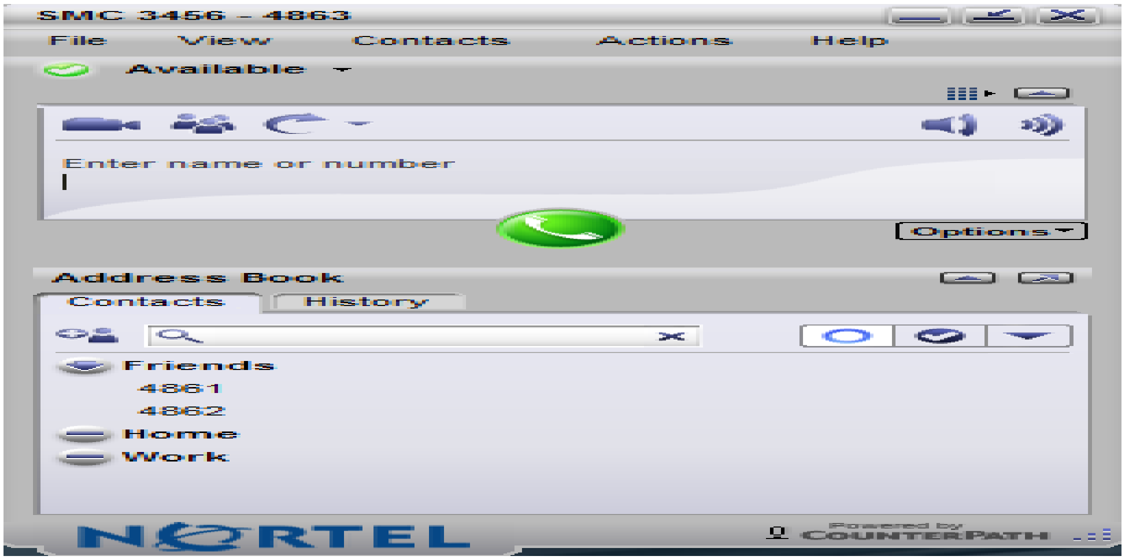

Figure 68 SMC Client

On the top menu bar, go to FILE -> PREFERENCES -> ADVANCED -> LOGIN SERVER

➔ No login server available

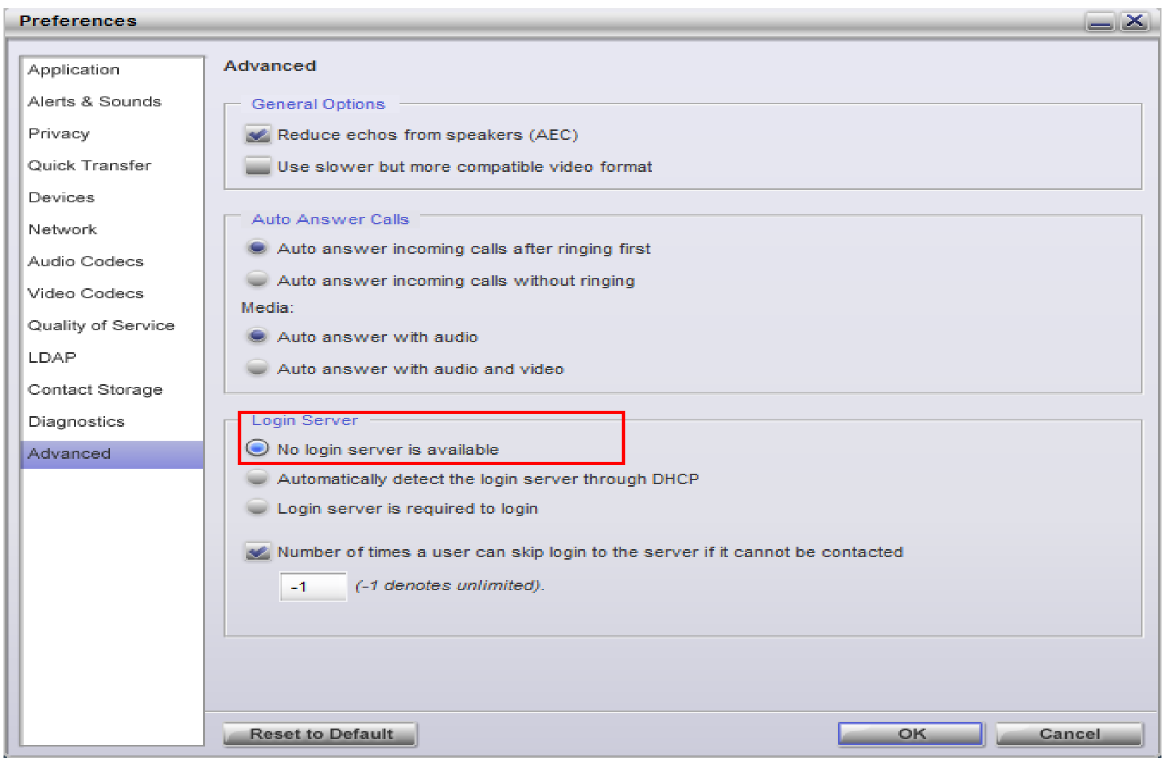

Figure 69 – Advanced Options Menu

4.4.4.1 Add a SIP Account on SMC3456

In order to create a SIP account for SMC3456 to be able to register to CS1000E SIP line server, From the top menu bar go to FILE -> ACCOUNT SETTINGS -> Add New SIP Account, see figure 70.

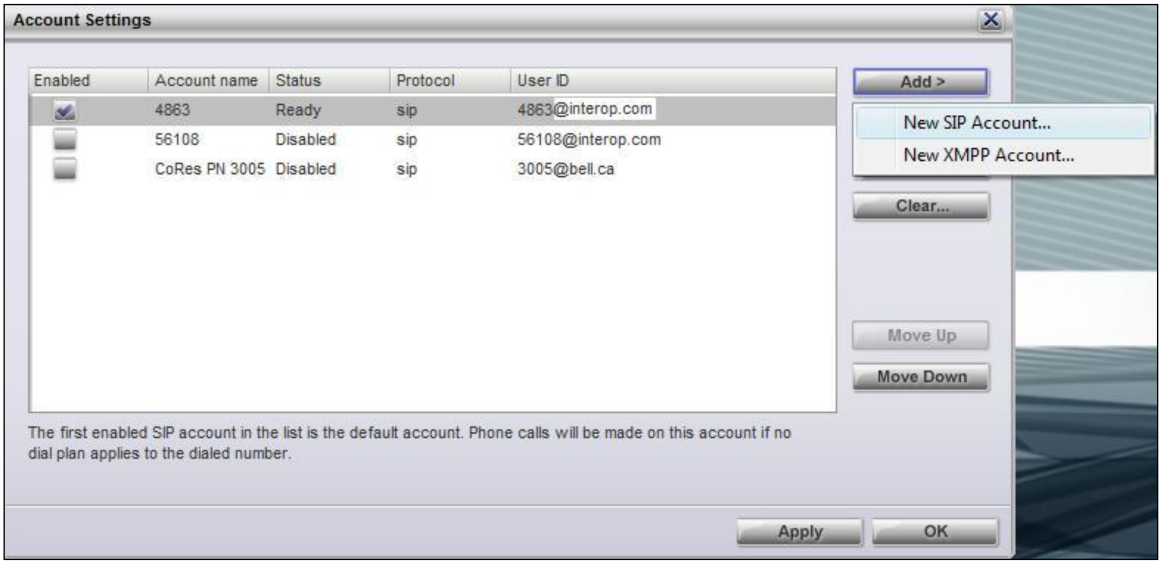

Figure 70 – Accounting Settings

The created account is appeared as figure 71.

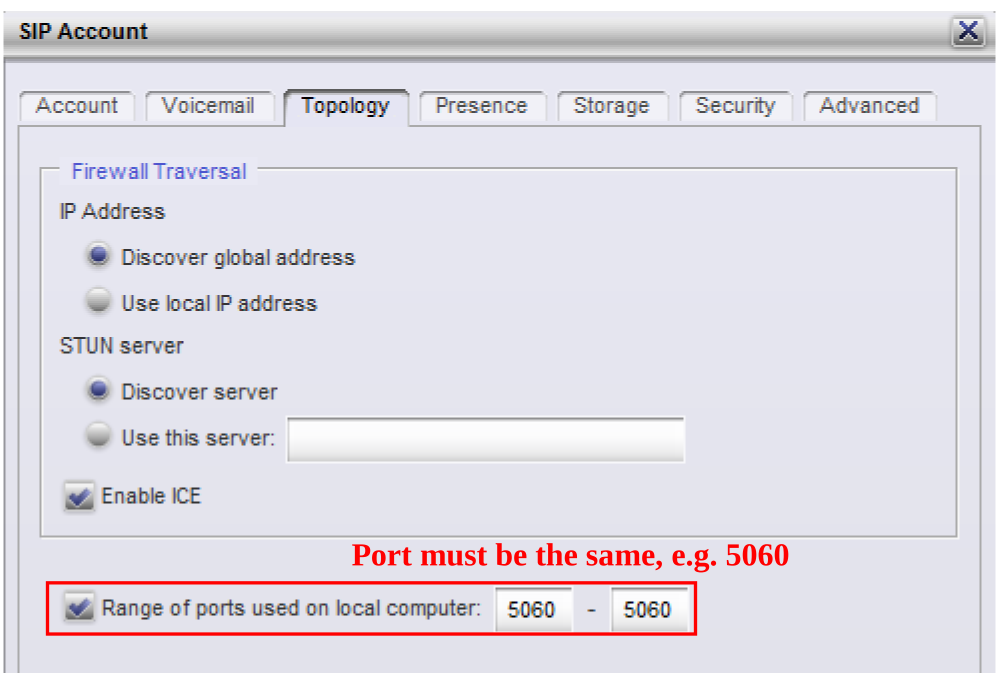

Figure 71 – Topology SIP Account Settings

Figure 72 shows how to set SIP account details by clicking on the Account menu tap.

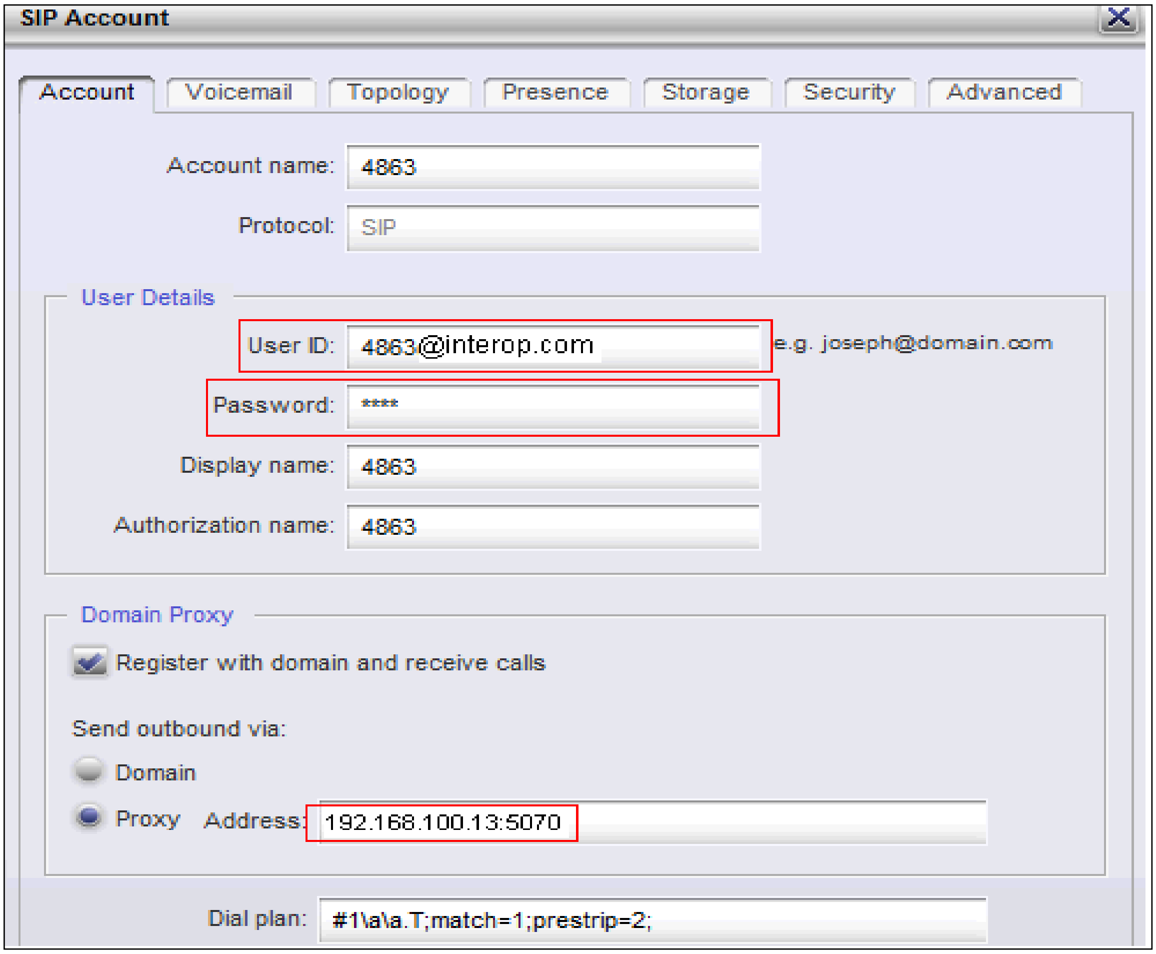

Figure 72 – SIP Account Details Setting

Figure 73 shows the newly created SIP account

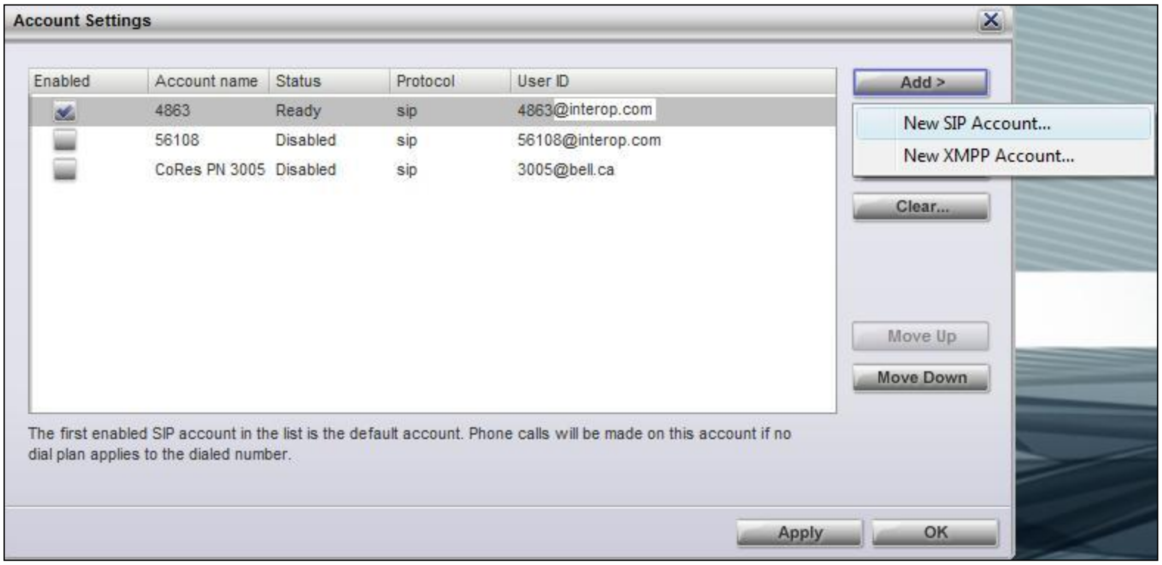

Figure 73 – New Created SIP Account

4.4.5 Provisioning SIP Phone Sets 1140 and 1120 on CS1000E Call Server

On CS1000E Call Server, use Command Line interface to configure the 1120 and 1140 phone sets.

```
TN   104 0 0 0
UXTY
DATE
PAGE
DES

DES  SIPL
TN   104 0 00 00  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
SIPU 4861
NDID 1002
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
```

NHTN
CFG_ZONE 010
CUR_ZONE 010
ERL 0
ECL 0
FDN
TGAR 0
LDN NO
NCOS 7
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR MTD FND HTD TDD HFD CRPD
    MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD DSX VMD SLKD CCSD SWD LND CNDD
    CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDD CFXD ARHD CLTD ASCD
    CPFA CPTA ABDD 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 VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH
CPND_LANG ENG
HUNT
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 496855 0   MARP
     01 HOT U 2224861 MARP 0

5 Paetec Communication System configuration

Paetec will have to provide this configuration notes.

6 General Test Approach and Test Results

The focus of this interoperability compliant testing was to verify the SIP trunk connectivity between the Paetec Communication systems and Avaya Communication Server 1000E release 6.0. The testing verified SIP signaling and media of the basic telephony features are communicating correctly. The following features were covered; basic calls, busy, music on hold, blind and consultative transfers, DTMF, MWI, codec negotiations, conference.

6.4 General test approach

The general test approach was to have Paetec Sonus system connected to CS1000E SIP Gateway using Sonus IP address. The SIP trunk communication should be established between CS1000E and Paetec Sonus system. Calls can be made from end to end, i.e. PSTN phone can call through created route from Paetec Communication system to CS1000Es' analog, IP, SIP phones via SIP trunk. The main objectives were to verify the basic SIP trunk features:

- Basic call from PSTN phone to CS1000E phones
- Perform basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold.
- Redirect call between users/clients/endpoints: blind/consultative transfers, call forward all call, busy and no answer.
- Perform codec negotiation
- Perform conferencing

6.5 Test Results

The objectives outlined in section 6.1 were verified and met. The following observations were made during the compliance testing:

- Dial to telephone number which begins with "*", i.e. *xxxxx does not match required format on Avaya CS1000E.
- CPND, Call Party Name Display, does not support on test set up. Telephone number is displayed instead.
- Music on hold is not enabled on CS1000E. i.e. User won't hear music when call is put on hold
- Media Security is not enabled on this test configuration.
- CS1000E_PHONE1 calls CS1000E_PHONE2. Call is established when CS1000E_PHONE2 answers. The CS1000E_PHONE1 does blind transfer to PSTN1. User expects that PSTN1 rings; CS1000E_PHONE2 hears ringback tone. Call is established two ways speech path between CS1000E_PHONE2 and PSTN1. However, Paetec does not support UPDATE, this will cause lack of ring back tone or early media. Following are some scenarios where the lack of ring back tone or early media will apply
  Scenario 1: PSTN 1 calls CS1K set 2. CS1K set 2 does a blind transfer to PSTN 3.
  - Issue: when CS1K set 2 completes the blind transfer and PSTN 3 is still ringing, due to lack of UPDATE support; user at PSTN 1 won't be able to hear ring back tone or early

media because CS1K cannot send SDP of PSTN 1 to PSTN 3 via UPDATE (call not yet connected on 2nd call leg between CS1K set 2 and PSTN 3; so we must send the SDP via UPDATE).

Scenario 2: CS1K set A calls CS1K set B. Set B does a blind transfer to PSTN

- Issue: when CS1K set completes blind transfer; and PSTN is still ringing; user on set A won't be able to hear ring back tone - same technical issue as Scenario 1 above.

Scenario 3: PSTN calls CS1k set A. Set A does a blind transfer to CS1K set C

- Issue: PSTN won't hear ring back tone when CS1k set A completes the transfer (and set C is still ringing).

Scenario 4: PSTN 1 calls 1k set A...set A does a local blind transfer to set B..B does CFNA to PSTN 2

- Issue: PSTN won't hear ring back tone when CS1K set A completes the transfer; as well as when PSTN 2 is ringing.

We have observed Paetec send <u>183+sdp</u> when calls are made to PSTN...The workaround patch on the CS1000 SipGW has the CS1K treats an incoming 183+sdp as 180 Ringing and generates a local ring back tone.

- **NOTE**: With this workaround, **<u>early media support is not possible</u>**...ie. CS1K will treat an **incoming** 183+sdp for ANY call as a 180 Ringing without sdp. Therefore, even for a simple call such as: CS1K set A calls PSTN...PSTN sends 183+sdp providing an announcement. The CS1K user will just hear the ring back tone and <u>not</u> the announcement.

Results with the workaround patch applied:

In all the above 4 scenarios; when blind transfer is completed; the originating user will hear ring back tone....and once the call has been picked up on far end; there will be 2 way speech. Please refer to CR Q02128833 which blocks test case

- PSTN1 calls CS1000E_PHONE which is set up to call forward all call to PSTN2. User expects that call will be forwarded to PSTN2 right after PSTN1 calls CS1000E_PHONE; PSTN2 will ring and PSTN1 will ringback tone but PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up to call forward no answer to PSTN2. User expects that call will be forwarded to PSTN2 after CS1000E_PHONE rings three times and then PSTN2 will ring and PSTN1 will ring back tone. Actually, PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up to call forward busy to PSTN2. User expects that call will be forwarded to PSTN2 if PSTN1 calls to CS1000E_PHONE which is busy. PSTN2 will ring and PSTN1 will ring back tone. Actually, PSTN2 does not ring, PSTN1 shows “Release and Try again”. Issue is resolved after changing configuration on Paetec’s SBC.

- PSTN1 calls CS1000E_PHONE which is set up call waiting. Call is established successfully when CS1000E_PHONE answers PSTN1. PSTN2 calls the same CS1000E_PHONE which is busy with PSTN1. CS1000E_PHONE will receive a call waiting tone/indication but there is no waiting tone/indication on CS1000E_PHONE. Please ensure DID numbers also are enabled the call waiting feature on Paetec. Issue is resolved after call waiting is enabled on Paetec.

- At this moment, SIPLINE clients testing result based on SU nortel-cs1000-vtrk-6.00.18.065-016.i386.001.ntl is installed on SS_Carrier and nortel-cs1000-vtrk-6.00.18.63-06.i386.001.ntl is installed on SLG, as CS1000E designer's suggestion. Please refer to Appendix B for details of patches installation.
  If SU "nortel-cs1000-vtrk-6.00.18.065-016.i386.001.ntl" is installed on both SS_carrier and SLG, all simple calls relate to SIPLINE will be failed between Unistim calls SIPLINE, PSTN calls SIPLINE, SIPLINE1 calls SIPLINE2 as one hangs up call. Another is NOT released. Please refer to CR Q02129692. 80% SIPLINE test cases are also failed with this patch as basic SIPLINE features do not work such as conference, blind transfer, consult transfer, call forward no answer.

7 Verification Steps

This section includes some steps that can be followed to verify the solution is working.

7.4 Verify that calls are established with two-way voice path when making calls from one CS1000E phone to another on the local CS1000E.

Verify that IP phones, digital, analog (Fax) register successfully show as below:

Verify status of IP phone registered

[nortel@nd1-car1 ~]$ isetShow
=== TPS ===
1. Set Information

IP Address	NAT	Model Name	Type	RegType	State	Regd-TN	FWVsn
47.248.101.117		IP Phone 1120E	1120	Regular	online	096-00-01-24	C60
47.248.101.120		IP Phone 2002 Phase 2	2002P2	Regular	online	096-00-01-06	DCJ
47.248.101.116		IP Phone 1140E	1140	Regular	online	096-00-01-26	C60
47.248.101.115		IP Phone 1220	1220	Regular	online	096-00-01-05	C6O

Verify status of digital phone registred:

LD 32
Stat 4 0 7
>ld 32

.stat 4 0 7
00 = UNIT 00 = IDLE (3904)
01 = UNIT 01 = IDLE (3902)

……….

Verify status of Analog (Fax machine registered):
LD 32
.stat 4 0 8
00 = UNIT 00 = IDLE (L500)
01 = UNIT 01 = IDLE (L500)

Verify the following basic calls in local CS1000E:

IP phone------------------------call--------------------IP phone
IP phone -----------------------call--------------------SIP Line Client
IP Phone -----------------------call--------------------Analog/Fax phone
IP Phone -----------------------call--------------------Digital phone
SIP Line Client----------------call--------------------Analog/Fax phone
SIP Line Client----------------call--------------------Digtal Phone
Analog/Fax phone---------------call--------------------Digital Phone
User can verify the same for calls from oposite direction.

Verify that calls are established with two-way voice path and busy status under CS1000E call server as below:

Verify status of IP phones which are busy
[nortel@nd1-car1 ~]$ isetShow
=== TPS ===

Set Information

IP Address	NAT	Model Name	Type	RegType	State	Regd-TN	UNIStimVsn
47.248.101.117		IP Phone 1120E	1120	Regular	busy	096-00-01-24	C6O
47.248.101.120		IP Phone 2002 Phase 2	2002P2	Regular	busy	096-00-01-06	DCJ
47.248.101.116		IP Phone 1140E	1140	Regular	busy	096-00-01-26	C6O
47.248.101.115		IP Phone 1220	1220	Regular	busy	096-00-01-05	C6O

Verfify status of digital phone is busy

LD 32 .stat 4 0 7 000 = UNIT 00 = BUSY (3904)
01 = UNIT 01 = BUSY (3902)
……

Verify status analog phone is busy

LD 32
.stat 4 0 8
00 = UNIT 00 = BUSY (L500)
01 = UNIT 01 = BUSY (L500)

Verify status of voice gateway if calls are established between IP phone/SIP line Clients to Analog/Digital phones or call to voice message

>>ld 32
NPR000
.stat 4 0 11
00 = UNIT 00 = BUSY (TRK)(IPTN REG)
01 = UNIT 01 = BUSY (TRK)(IPTN REG)
02 = UNIT 02 = BUSY (TRK)(IPTN REG)
03 = UNIT 03 = BUSY (TRK)(IPTN REG)

During the call, use pcap tool (ethereal/wireshark) at the TLAN media gateway card, RTP streams are going for call relate to analog, digital or voice message.

7.5 Verify that calls are established with two-way voice path when making calls from PSTN phone to Avaya phones on the CS1000 through Paetec Communication system via configured SIP trunk.

- Verify basic call between PSTN phones and Avaya phones. At the CS1000E SIP Gateway during the call, use pcap tool (ethereal/wireshark) to make sure that all SIP request/response messages
- Verify Codec, SIP trunk status when call is established under CS1000E call server by tracing DID number

LD 80
.trac 0 496856

ACTIVE VTN 096 0 01 06
ORIG VTN 096 0 01 06 KEY 0 SCR MARP CUST 0 DN 496856 TYPE 2002P2
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 47.248.101.120 PORT: 5200
TERM VTN 100 0 00 31 VTRK IPTI RMBR 100 32 OUTGOING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 217.110.230.98
FAR-END MEDIA ENDPOINT IP: 217.110.230.97 PORT: 6478
FAR-END VendorID: Not available

MEDIA PROFILE: **CODEC G.711 A-LAW** PAYLOAD 20 ms VAD OFF
RFC2833: RXPT 101 TXPT 101 DIAL DN 916139675258
MAIN_PM ESTD
TALKSLOT ORIG 21 TERM 53
QUEU NONE
CALL ID 511 941

---- ISDN ISL CALL (TERM) ----
CALL REF # = 416
BEARER CAP = VOICE
HLC =
CALL STATE = 10 ACTIVE
CALLING NO = 442033496856 NUM_PLAN:E164 TON:INTERNATIONAL
ESN:UNKNOWN
CALLED NO = 16139675258 NUM_PLAN:E164 TON:INTERNATIONAL
ESN:UNKNOWN

- Verify SIP Trunk is released when DID number is released the call by tracing that DID number under CS1000E call server

LD 80
.trac 0 496856 (DID number)

- **IDLE** VTN 096 0 01 06 MARP

8 Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated by Paetec and Avaya design teams. Some of these issues are considered as exceptions. The Paetec Communication System is considered compliant with Communication Server 1000E release 6.0.

9 Additional References

Product documentation for Avaya products may be found at:
http://support.nortel.com/go/main.jsp

[1] *Communication Server 1000E Overview Release 6.0, Revision 03.04, October 2009, Document Number NN43041-110*

[2] *Product Compatibility Matrix release 5.0/5.5/6.0, Revision 01.07, February 2010, Document Number NN43001-140*

[3] *Communication Server 1000 Network Routing Service Fundamentals, Release 6.0, Revision 01.04, Jun 2009, Document Number NN43001-130*

[4] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision 03.05, February 2010, Document Number NN43001-116*

[5] *Communication Server 1000 SIP Line Fundamentals, Release 6.0, Revision 01.08, February 10, Document Number NN43001-508*

[6] *Communication Server 1000 Dialing Plans Reference, Release 6.0, Revision 03.09, June 2009, Document Number NN43001-283*

10 Appendixes

Appendix A: CS1000E CPPM Call Server Rls 6.00R Patches Installed

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.mdp issp

VERSION 4121

RELEASE 6

ISSUE 00 R +

DepList 1: core Issue: 02 (created: 2010-02-02 13:33:25 (est)) ALTERED

IN-SERVICE PEPS

PAT#	CR #	PATCH REF #	NAME	DATE	FILENAME	SPECINS
000	Q01976701-01	ISS1:1OF1	p28211_1	29/03/2010	p28211_1.cpl	NO
001	Q02029209	ISS1:1OF1	p28469_1	29/03/2010	p28469_1.cpl	NO
002	Q02023636	ISS1:1OF1	p28475_1	29/03/2010	p28475_1.cpl	NO
003	Q02041702	ISS1:1OF1	p28698_1	29/03/2010	p28698_1.cpl	NO
004	Q02027777	ISS1:1OF1	p28471_1	29/03/2010	p28471_1.cpl	NO
005	Q02038440	ISS1:1OF1	p28674_1	29/03/2010	p28674_1.cpl	NO
006	Q02034835	ISS1:1OF1	p28569_1	29/03/2010	p28569_1.cpl	YES
007	Q02040015	ISS1:1OF1	p28657_1	29/03/2010	p28657_1.cpl	NO
008	Q02094012	ISS1:1OF1	p29370_1	29/03/2010	p29370_1.cpl	YES
009	Q02039217-01	ISS1:1OF1	p28760_1	29/03/2010	p28760_1.cpl	NO
010	Q02031118	ISS1:1OF1	p28680_1	29/03/2010	p28680_1.cpl	NO
011	Q02096711	ISS1:1OF1	p29394_1	29/03/2010	p29394_1.cpl	NO
012	Q02024135-04	ISS1:1OF1	p28381_1	29/03/2010	p28381_1.cpl	NO
013	Q02021470-02	ISS1:1OF1	p28776_1	29/03/2010	p28776_1.cpl	NO

014	Q02043231	ISS1:1OF1	p28712_1	29/03/2010	p28712_1.cpl	NO
015	Q02083027	ISS1:1OF1	p29233_1	29/03/2010	p29233_1.cpl	NO
016	Q02033321	ISS1:1OF1	p28801_1	29/03/2010	p28801_1.cpl	NO
017	Q02033951	ISS1:1OF1	p28579_1	29/03/2010	p28579_1.cpl	NO
018	Q01782930-01	ISS1:1OF1	p24964_1	29/03/2010	p24964_1.cpl	NO
019	Q02028560-04	ISS1:1OF1	p28564_1	29/03/2010	p28564_1.cpl	NO
020	Q02041981	p28695_1	p28719_1	29/03/2010	p28719_1.cpl	NO
021	Q02033139	ISS1:1OF1	p28582_1	29/03/2010	p28582_1.cpl	NO
022	Q02039181	ISS1:1OF1	p28644_1	29/03/2010	p28644_1.cpl	NO
023	Q02030977	ISS1:1OF1	p28507_1	29/03/2010	p28507_1.cpl	NO
024	Q02076740	ISS1:1OF1	p29154_1	29/03/2010	p29154_1.cpl	NO
025	Q02012100-06	ISS1:1OF1	p29368_1	29/03/2010	p29368_1.cpl	NO
026	Q02035555	ISS1:1OF1	p28814_1	29/03/2010	p28814_1.cpl	NO
027	Q02021384-01	ISS1:1OF1	p28615_1	29/03/2010	p28615_1.cpl	NO
028	Q02032955-02	ISS1:1OF1	p28529_1	29/03/2010	p28529_1.cpl	NO
029	Q02055997	ISS1:1OF1	p28895_1	29/03/2010	p28895_1.cpl	NO
030	Q02022264	ISS1:1OF1	p28486_1	29/03/2010	p28486_1.cpl	NO
031	Q02031323-01	ISS1:1of1	p28546_1	29/03/2010	p28546_1.cpl	NO
032	Q01987279-02	ISS1:1OF1	p28416_1	29/03/2010	p28416_1.cpl	NO
033	Q00349046-03	ISS1:1OF1	p17588_1	29/03/2010	p17588_1.cpl	NO
034	Q02049121-01	ISS1:1OF1	p28819_1	29/03/2010	p28819_1.cpl	NO
035	Q02029228-01	ISS1:1OF1	p28681_1	29/03/2010	p28681_1.cpl	YES
036	Q02071626	ISS1:1OF1	p29163_1	29/03/2010	p29163_1.cpl	NO
037	Q02020526	ISS1:1OF1	p28537_1	29/03/2010	p28537_1.cpl	NO
038	Q02034783-01	p28596	p28594_1	29/03/2010	p28594_1.cpl	YES
039	Q02039994	ISS1:1OF1	p28690_1	29/03/2010	p28690_1.cpl	NO
040	Q01986974-05	ISS1:1OF1	p28821_1	29/03/2010	p28821_1.cpl	YES
041	Q02035396	ISS1:1OF1	p28675_1	29/03/2010	p28675_1.cpl	NO
042	Q02035822-01	ISS1:1OF1	p29212_1	29/03/2010	p29212_1.cpl	NO
043	Q02097631	ISS1:1OF1	p28328_1	29/03/2010	p28328_1.cpl	NO
044	Q02093188	ISS1:1OF1	p29352_1	29/03/2010	p29352_1.cpl	NO
045	Q02071451	ISS1:1OF1	p29164_1	29/03/2010	p29164_1.cpl	NO
046	Q01983521-04	ISS1:1OF1	p27616_1	29/03/2010	p27616_1.cpl	NO
047	Q02077909	ISS1:1of1	p29272_1	29/03/2010	p29272_1.cpl	NO
048	Q02073690	ISS1:1OF1	p29208_1	29/03/2010	p29208_1.cpl	NO
049	Q02092594	ISS1:1OF1	p27830_1	29/03/2010	p27830_1.cpl	NO

050	Q02031359	p28679	p28725_1	29/03/2010	p28725_1.cpl	YES
051	Q02079849	ISS1:1OF1	p29238_1	29/03/2010	p29238_1.cpl	NO
052	Q02031959	ISS1:1OF1	p28728_1	29/03/2010	p28728_1.cpl	NO
053	Q02038675	ISS1:1OF1	p28665_1	29/03/2010	p28665_1.cpl	YES
054	Q02100914	ISS1:1OF1	p28597_1	29/03/2010	p28597_1.cpl	NO
055	Q02024455-01	ISS1:1OF1	p28717_1	29/03/2010	p28717_1.cpl	NO
056	Q02020734-02	ISS1:1OF1	p28668_1	29/03/2010	p28668_1.cpl	NO
057	Q02044341	ISS1:1OF1	p28957_1	29/03/2010	p28957_1.cpl	NO
058	Q02064503	ISS1:1OF1	p29196_1	29/03/2010	p29196_1.cpl	NO
059	Q02095838	1SS1:1OF1	p28852_1	29/03/2010	p28852_1.cpl	NO
060	Q02043398	ISS1:1OF1	p28869_1	29/03/2010	p28869_1.cpl	NO
061	Q02033000	ISS1:1of1	p28736_1	29/03/2010	p28736_1.cpl	NO
062	Q02089407	ISS1:1OF1	p29311_1	29/03/2010	p29311_1.cpl	NO
063	Q01981776-01	ISS1:1OF1	p29065_1	29/03/2010	p29065_1.cpl	NO
064	Q02017013-01	ISS1:1OF1	p28313_1	29/03/2010	p28313_1.cpl	NO
065	Q02077171	ISS1:1OF1	p29169_1	29/03/2010	p29169_1.cpl	NO
066	Q02031502	ISS1:1OF1	p28832_1	29/03/2010	p28832_1.cpl	YES
067	Q02086333	ISS1:1OF1	p29262_1	29/03/2010	p29262_1.cpl	YES
068	Q02102219-01	ISS1:1OF1	p29464_1	29/03/2010	p29464_1.cpl	NO
069	Q02092223	ISS1:1OF1	p29343_1	29/03/2010	p29343_1.cpl	NO
070	Q02077848-01	ISS1:1OF1	p29320_1	29/03/2010	p29320_1.cpl	NO

Appendix B: CS1000E CPPM Signaling Server Rls 6.00.18 Patches Installed

Product Release: 6.00.18.00

In system patches: 7

PATCH#	NAME	IN_SERVICE	DATE	SPECINS	TYPE	RPM
21	p28774_1	Yes	02/03/10	NO	FRU	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
22	p28797_1	Yes	02/03/10	NO	FRU	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
23	p29407_1	Yes	23/03/10	NO	FRU	nortel-cs1000-cs-6.00.R.100-00.i386
24	p27408_1	Yes	19/03/10	NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch
25	p25946_1	Yes	23/03/10	NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch
26	p22968_1	Yes	23/03/10	NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch
27	p25529_1	Yes	26/03/10	NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch
28	p28415_1	Yes	25/03/10	NO	FRU	nortel-cs1000-pi-control-1.00.00.00-00.noarch

```
In System service updates: 15
PATCH#  IN_SERVICE   DATE      SPECINS   REMOVABLE   NAME
0       Yes          22/03/10   YES       YES         nortel-cs1000-linuxbase-6.00.18.63-02.i386.000
1       Yes          02/03/10   YES       YES         nortel-cs1000-patchWeb-6.00.18.63-01.i386.000
2       Yes          02/03/10   NO        YES         submgr-2.00.02.00-01.i386.000
3       Yes          02/03/10   NO        YES         nortel-cs1000-gk-6.00.18.63-00.i386.000
4       Yes          02/03/10   NO        YES         nortel-cs1000-sps-6.00.18.63-00.i386.000
5       Yes          02/03/10   NO        YES         nortel-cs1000-tps-6.00.18.63-00.i386.000
6       Yes          02/03/10   NO        YES         nortel-cs1000-bcc_6-0-6.00.18.63-01.i386.000
7       Yes          02/03/10   NO        YES         nortel-cs1000-cs1000WebService_6-0-6.00.18.63-
01.i386.000
8       Yes          24/03/10   NO        YES         nortel-cs1000-shared-general-6.00.18.62-00.i386.000
9       Yes          25/03/10   NO        YES         nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
10      Yes          02/03/10   NO        YES         nortel-cs1000-emWeb_6-0-06.00.18.63-01.i386.001
11      Yes          02/03/10   NO        YES         nortel-cs1000-pd-6.00.18.62-00.i386.000
12      Yes          02/03/10   NO        YES         nortel-cs1000-nrsm-6.00.18.62-00.i386.000
13      Yes          02/03/10   NO        YES         nortel-cs1000-ftrpkg-6.00.18.62-00.i386.000
14      Yes          02/03/10   NO        YES         nortel-cs1000-dmWeb-6.00.18.62-00.i386.001
15      Yes          02/03/10   NO        YES         nortel-cs1000-csv-6.00.18.62-00.i386.000
16      Yes          02/03/10   NO        YES         nortel-cs1000-csmWeb-6.00.18.62-00.i386.001
17      Yes          02/03/10   NO        YES         nortel-cs1000-auth-6.00.18.62-00.i386.000
18      Yes          02/03/10   NO        YES         nortel-cs1000-ISECSH-6.00.18.62-00.i386.000
20      Yes          02/03/10   NO        YES         nortel-cs1000-dbcom-6.00.18.65-01.i386.001
30      Yes          26/03/10   NO        YES         nortel-cs1000-vtrk-6.00.18.065-016.i386.001
SIPLINE Gateway patches:

[admin@sl-node1 ~]$ pstat
Product Release: 6.00.18.00
In system patches: 2
PATCH#  NAME      IN_SERVICE   DATE     SPECINS  TYPE   RPM
9       p28774_1   Yes         25/03/10  NO      FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
10      p28797_1   Yes         25/03/10  NO      FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386

In System service updates: 9
PATCH#  IN_SERVICE   DATE      SPECINS   REMOVABLE   NAME
0       Yes          25/03/10   YES       YES         nortel-cs1000-linuxbase-6.00.18.63-02.i386.000
1       Yes          25/03/10   YES       YES         nortel-cs1000-patchWeb-6.00.18.63-01.i386.000
```

2	Yes	25/03/10	NO	YES	nortel-cs1000-shared-general-6.00.18.62-00.i386.000
3	Yes	25/03/10	NO	YES	nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
4	Yes	25/03/10	NO	YES	nortel-cs1000-dmWeb-6.00.18.62-00.i386.001
5	Yes	25/03/10	NO	YES	nortel-cs1000-csv-6.00.18.62-00.i386.000
6	Yes	25/03/10	NO	YES	nortel-cs1000-auth-6.00.18.62-00.i386.000
7	Yes	25/03/10	NO	YES	nortel-cs1000-ISECSH-6.00.18.62-00.i386.000
11	Yes	07/04/10	YES	yes	nortel-cs1000-vtrk-6.00.18.63-06.i386.001
30	Yes	26/03/10	NO	YES	nortel-cs1000-vtrk-6.00.18.065-016.i386.001

Appendix C: Configure SIP trunk in CS1000 using overlays

Procedure summary
This information is provided as a simple summary of tasks to complete when configuring IP Peer Networking, but it does not replace the full details provided in the IP Peer Networking Guide.

No.	Overlay	Element Management	Action
1	LD 97		Define a virtual super loop
2	LD 17	Select Configuration/D-Channel link	Create a virtual D-channel
3	LD 15	Select Configuration/Customer Explorer link	Define the customer to support ISDN
4	LD 16	Select Configuration/Customer Explorer /Add Route	Create a virtual service route
5	LD 14	Select Configuration/Customer Explorer /Add Trunk	Create virtual trunks

Define a virtual superloop

Use Overlay 97

Prompt	Response	Description
REQ	CHG	
TYPE	SUPL	Configuration data block
SUPL	V100	Virtual superloop number (96 - 112 and multiple of 4 for 11C systems.)//CS 1000E not vloop100

Create a virtual D-channel

Use Overlay 17

Prompt	Response	Description
REQ	CHG	
TYPE	ADAN	Configuration data block
ADAN	NEW DCH 100	Add a primary D-Channel port 100
CTYP	DCIP	D-channel is over IP
DES	VIRTUAL_TRK	Description
USR	ISLD	Integrated services signaling link dedicated
IFC	SL1	Interface type is Meridian 1 – Meridian 1
ISLM	4000	Integrated services signaling link maximum
SIDE	USR	Slave to the controller (USR).
RLS	25	X11 software release of far-end.//not need
RCAP	ND2	Name display format 2//not need

Define a customer with ISDN support

Use Overlay 15

Prompt	Response	Description
REQ	NEW	
TYPE	CDB	Customer data block
CUST	0	Customer number
ANAT	1111	ANI Attendant billing number for making ANI calls
ANLD	111	ANI listed directory number
ISDN	YES	Customer is equipped with ISDN.
VPNI	1	Virtual private network identifier//important
PNI	1	Private network identifier.//important

Define a virtual service route

Use Overlay 16

Prompt	Response	Description
REQ	NEW	
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	100	Route number
DES	VTRK	Designator field for trunk
TKTP	TIE	TIE trunk only, allowed between SL-1
ICOG	IAO	Incoming and outgoing
VTRK	YES	Virtual trunk route
ZONE	0	Zone for codec selection and bandwidth management
NODE	2000	Node ID of signaling server of this route.
PCID	SIP	Protocol ID for this route
ISDN	YES	ISDN option
MODE	ISLD	Route uses ISDN signaling link
DCH	100	D-channel number for this route
PNI	1	Customer private network identifier.
IFC	SL 1	Interface type : Meridian 1 to Meridian 1
NCNA	YES	Network calling name allowed.
NCRD	YES	Network call redirection.
CHTY	BCH	B-channel type.
CTYP	CDP	Coordinated dialing plan

Define virtual trunks

Use Overlay 14

Prompt	**Response**	**Description**
REQ	NEW 32	
TYPE	IPTI	IP trunk
TN	100 0 0 0	Virtual card and channel number
DES	VTRK	Designator field for trunk
CUST	0	Customer number
RTMB	100 1	Route number and member number.
STRI	IMM	Start arrangement incoming
STRO	IMM	Start arrangement outgoing
TGAR	1	Trunk group access restriction.
CHID	1	Channel ID for trunk

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.