Avaya Solution & Interoperability Test Lab

Application notes for Frontier 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 Frontier 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 consultative) 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.

Introduction

This document provides a typical network deployment of Communication Server 1000 (CS1000) utilizing the Frontier 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 Avaya support representative.
The CS1000E system is configured as a SIP gateway endpoint on the Frontier 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.

Interoperability Compliance Testing

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

- 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 on Frontier)
- DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server (hosted on Avaya system)
- Early Media Transmission
- Inter-office tandem Call

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.

Dependencies

. CS1000 R6.0 software and implementation of latest patches

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

Support

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

- Toll Free: (800) 239 4430
- http://www.frontierhelp.com/techsupport.cfm

Reference Configuration

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

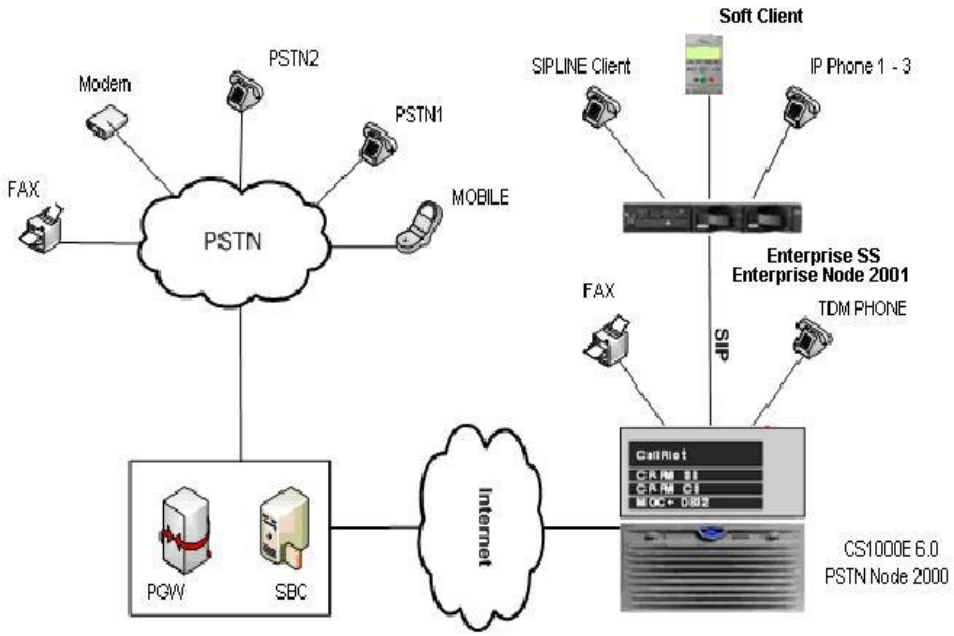

Figure 1- **Network diagram for Avaya-Frontier LAB setup**

Figure 2 depicts the deployment of two or more Communication Server of 1000E with the Frontier communication system.

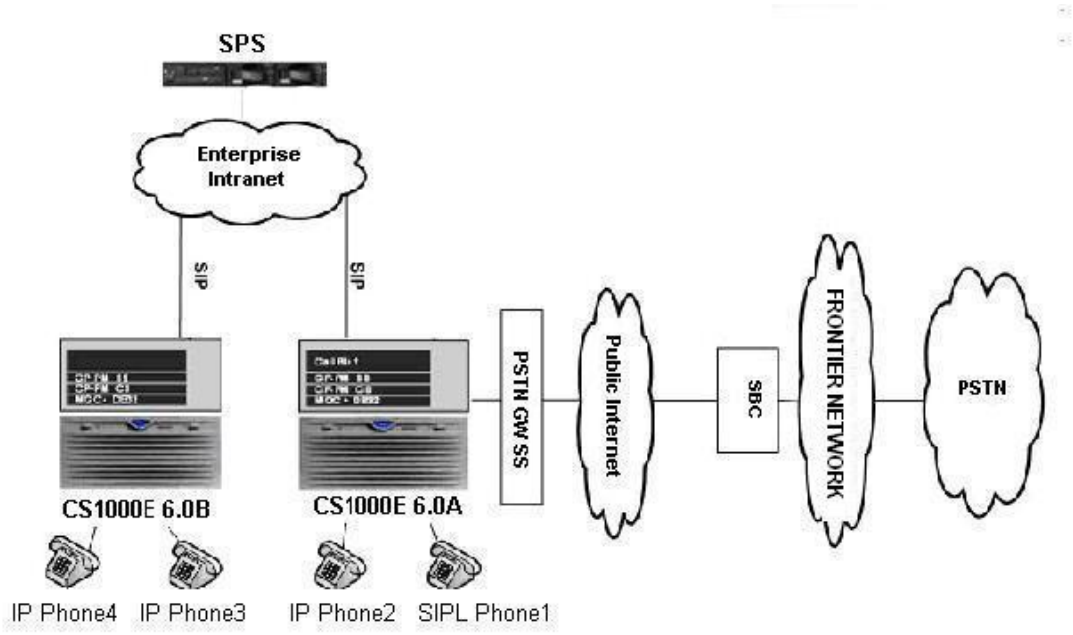

Figure 2 - Network topology for Multi-System configuration for Tandem Calls

The following assumptions were made for this lab test configuration:

1. CS1000 R6.0 software and implementation of latest patches
2. Frontier 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 Avaya 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 Avaya 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

Equipment and Software Validated

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

Call Server: 6.00 with latest deplists loaded
Signaling Server: SSE 6.00.18 with 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	
MPLR27159	1	Mandatory parameter "T38FaxRateManagement" isn't present in T38 SDP	

Hardware system requirement and theirs software/loadware version

System	Software/Loadware Version
Avaya CS1000E 6.0 (CPPM)	• Call Server: 6.00R • Signaling Server: 6.00.18
Avaya 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
Genband C3	• 7.2.40.40
Genband C6	• 10.4.7

Configure the Avaya Communication Server 1000E

Element Manager Configuration

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 Signaling Server).

- Input Node ID and press Save
- Enter TLAN, ELAN IP addresses of Signaling Server.

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

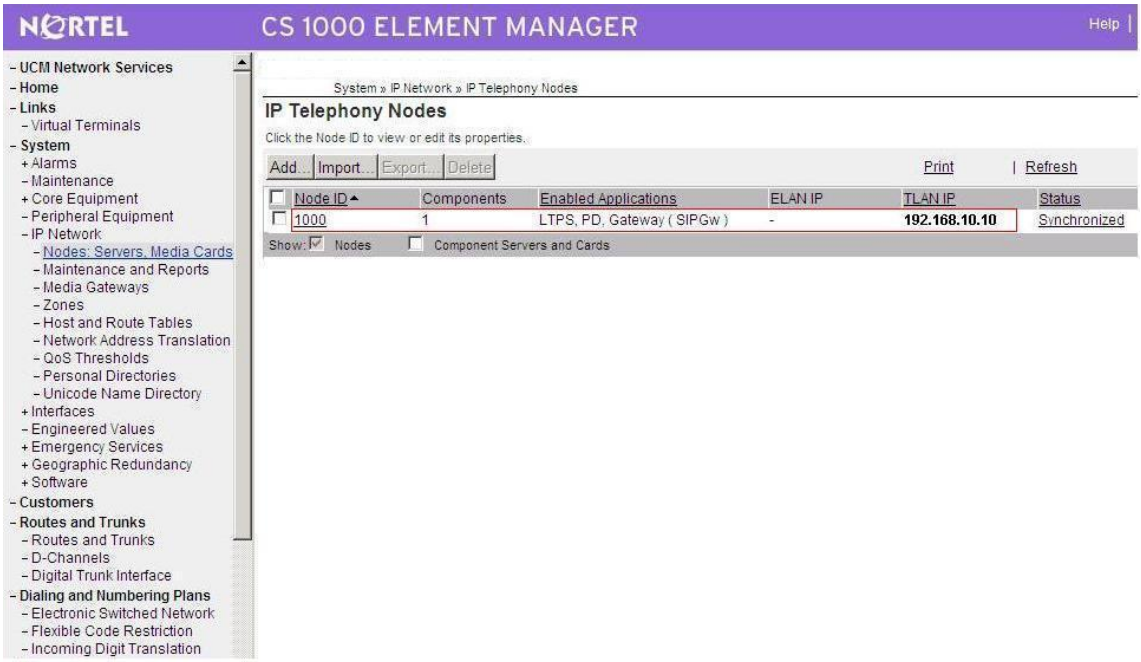

Figure 3 – Adding a node

Figure 4 describes the Call server IP configuration:

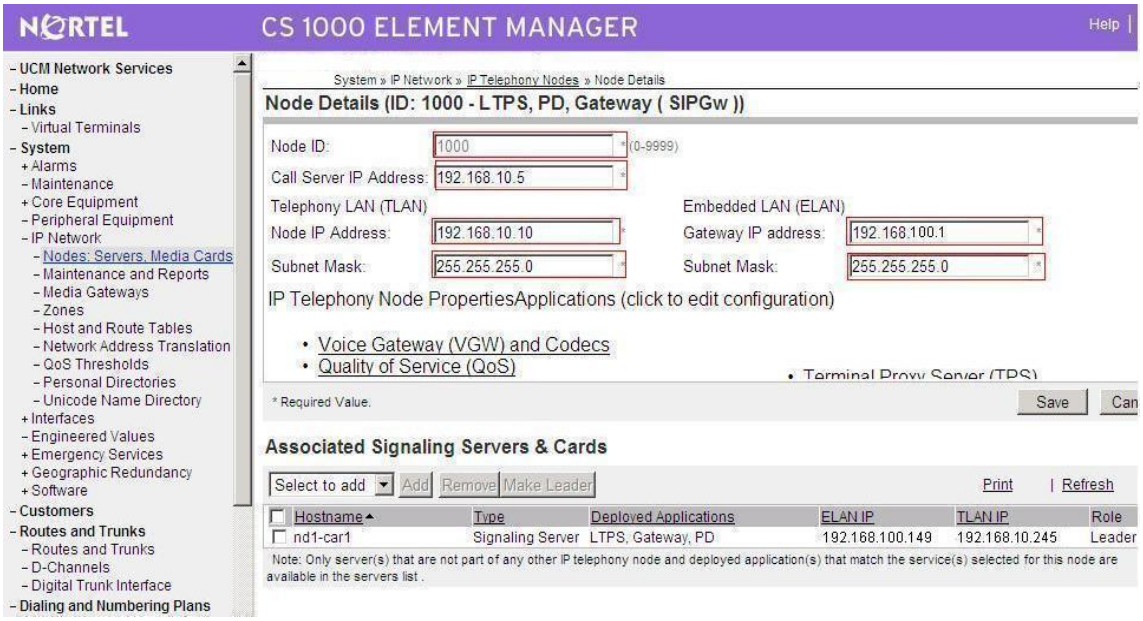

Figure 4 - Call Server IP Configuration

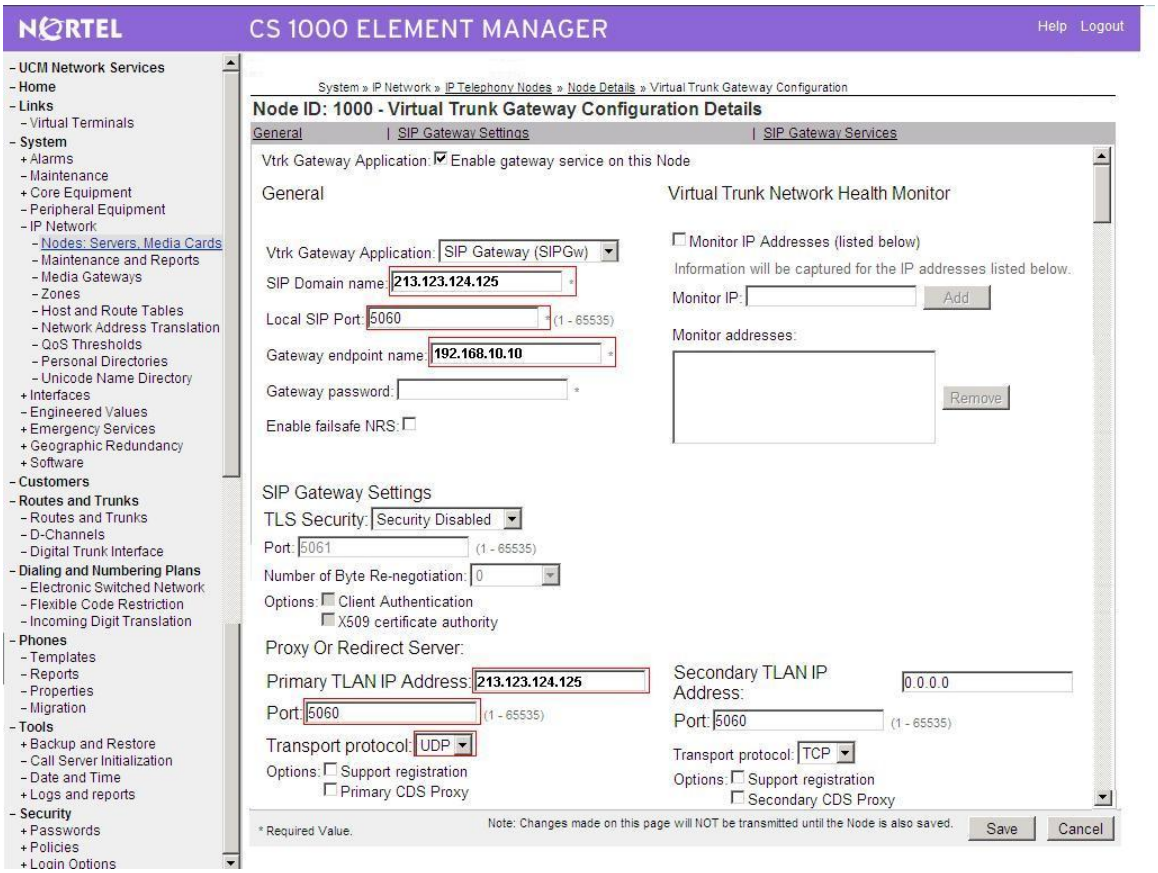

Figure 5 – Virtual Trunk Gateway configuration

Configure Voice Codec for Avaya 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 Avaya 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 by selecting "VGW and IP phone codec profile".

Disable Modem FAX pass through mode for G711 and check V.21 Fax tone Detection for tone detection by default. TN of sets with class of service =MPTD (Modem Pass Through Denied)

Figure 6 – Voice Gateway and Codec settings

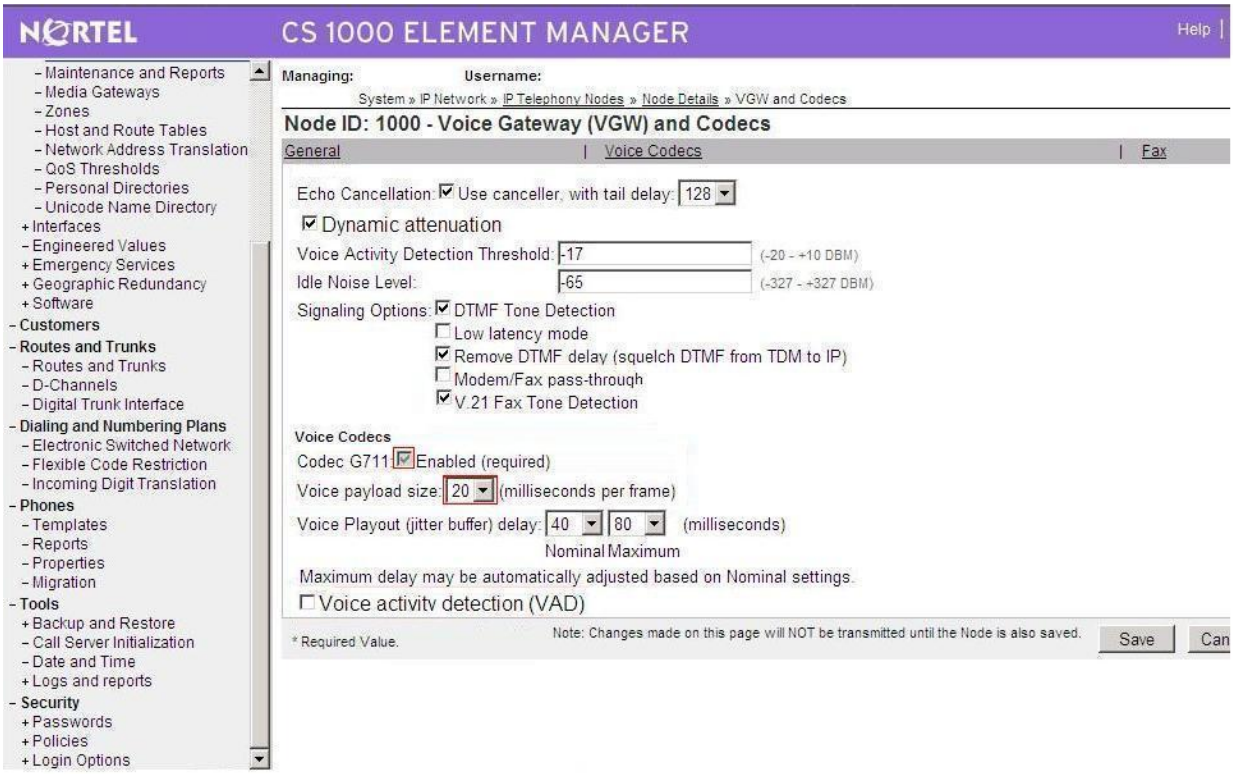

Figure 7 shows how to configure the Voice gateway and IP phone codec settings. The Frontier Communication network supports both G.711 and G.729. The packet size is set to 20 to match the network also.

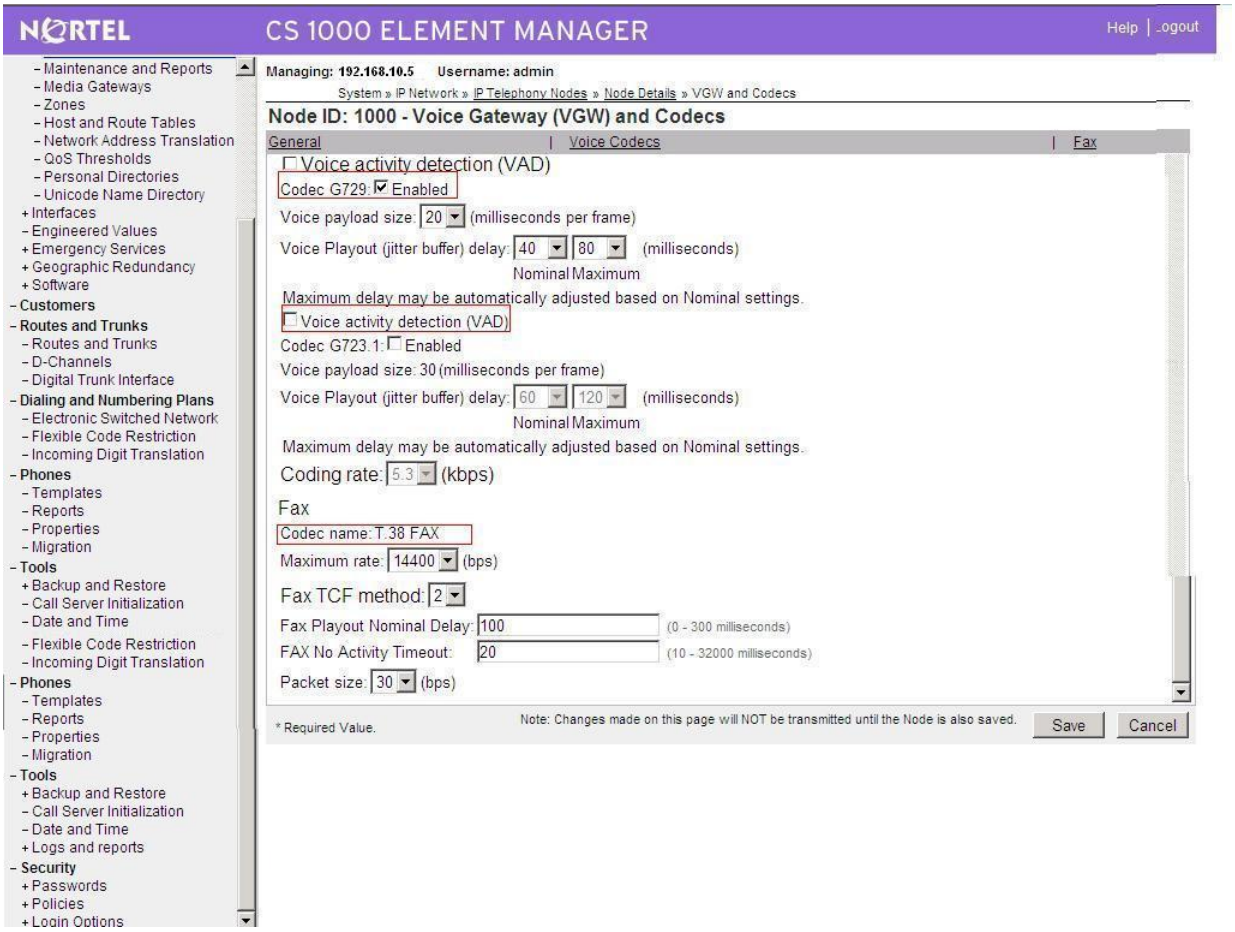

Figure 7 – Voice Gateway and codec settings

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
Disable Modem FAX pass through mode
TN of sets with class of service = MPTD (Modem Pass Through Denied)
Voice gateway and IP phone codec settings.

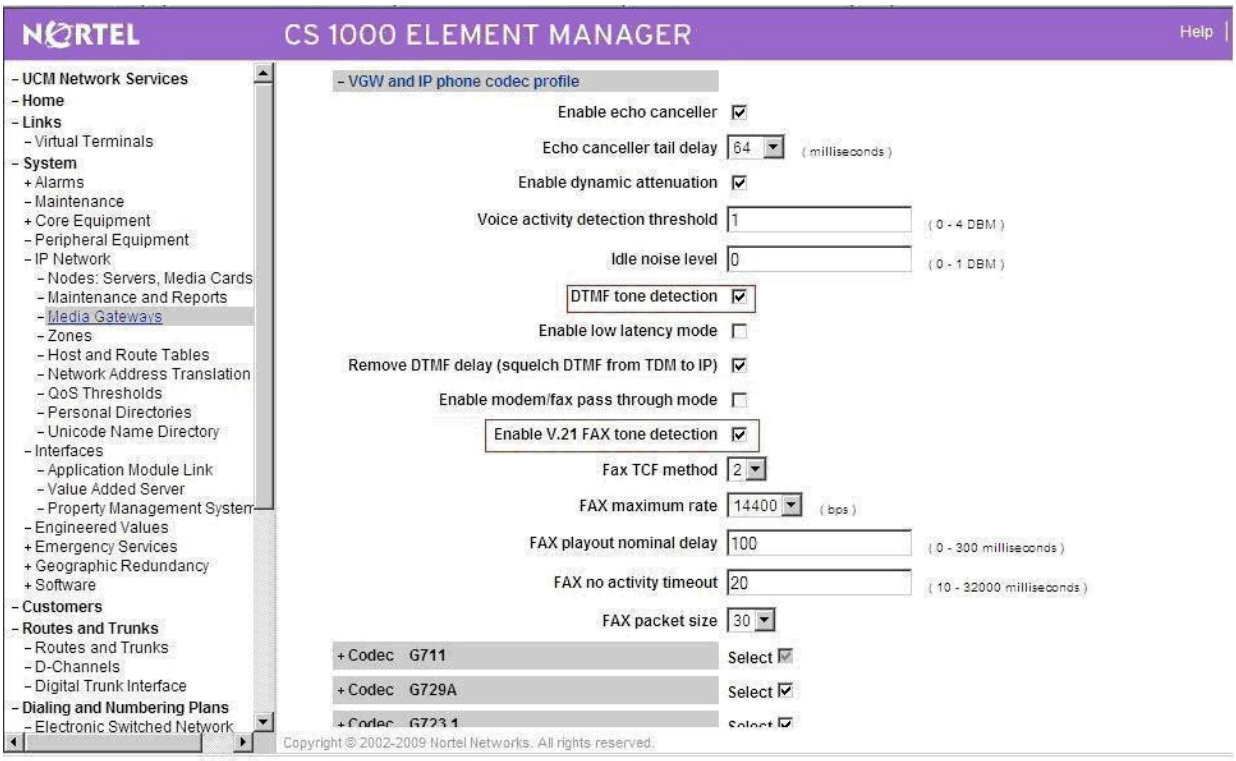

Figure 8 – Voice Gateway and IP phone codec profile settings

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.

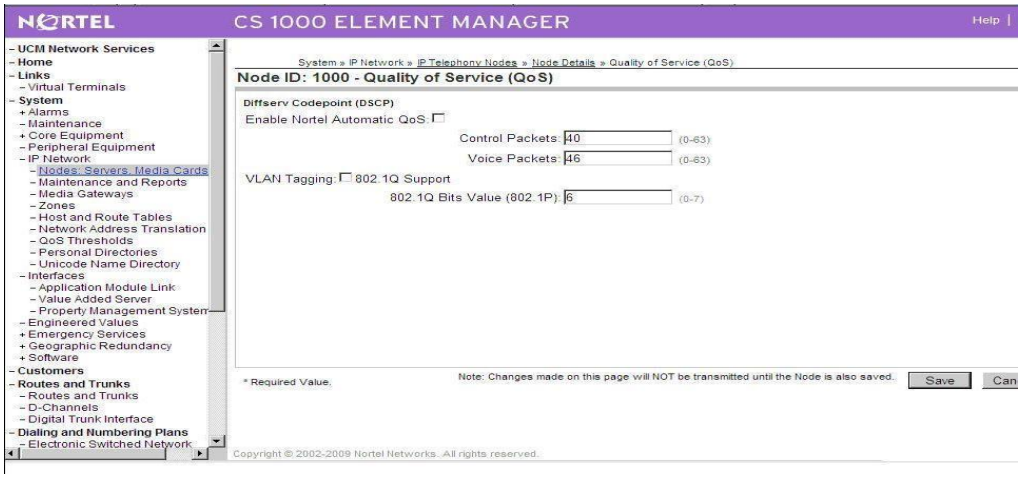

Figure 9 – Quality of Service settings.

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.

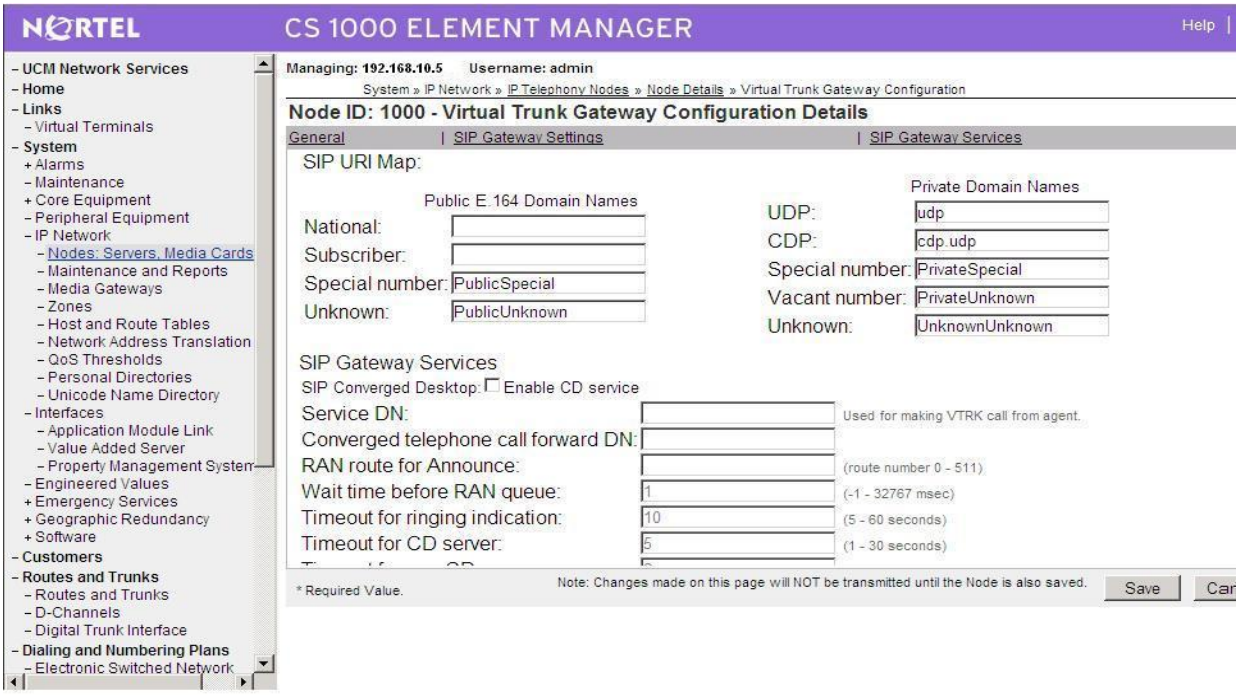

Figure 10 – SIP Gateway Services Settings

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 exist already, create a zone for IP sets. The bandwidth strategy can be adjusted to preference.

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

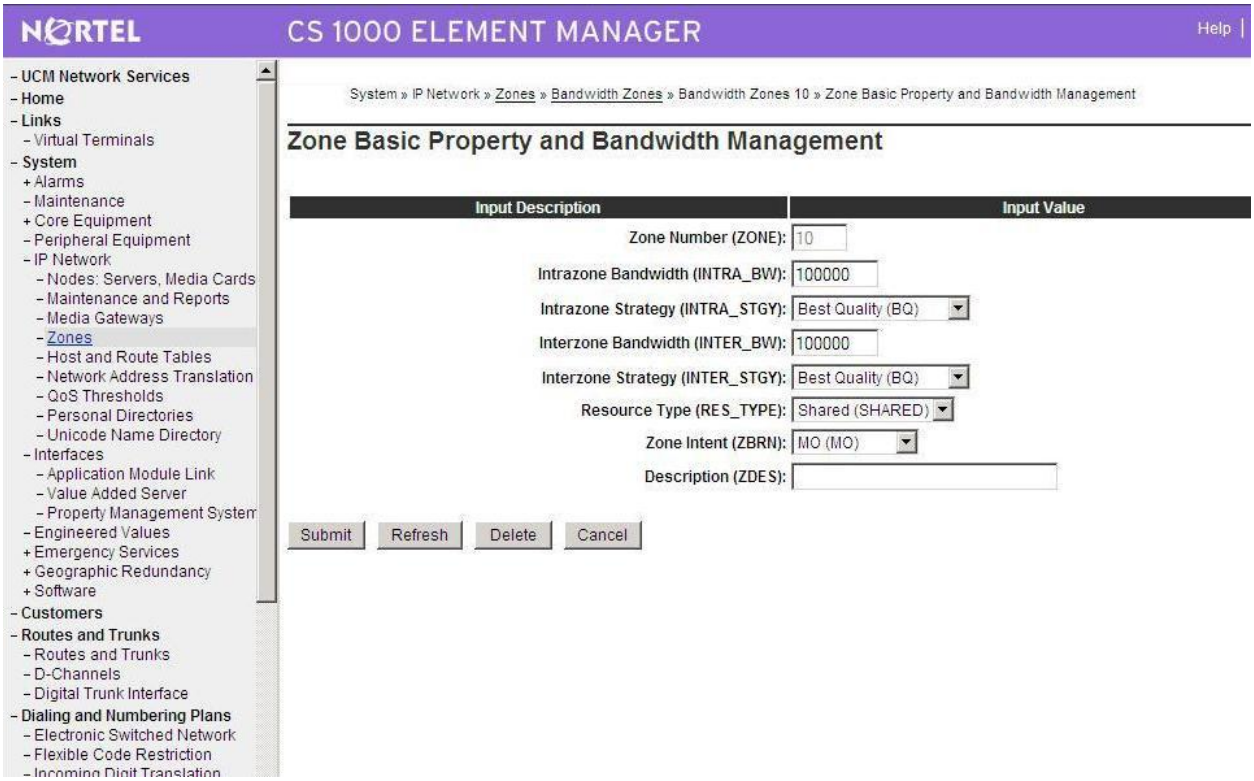

Figure 11 – Zone Basic Property Setting for IP phones

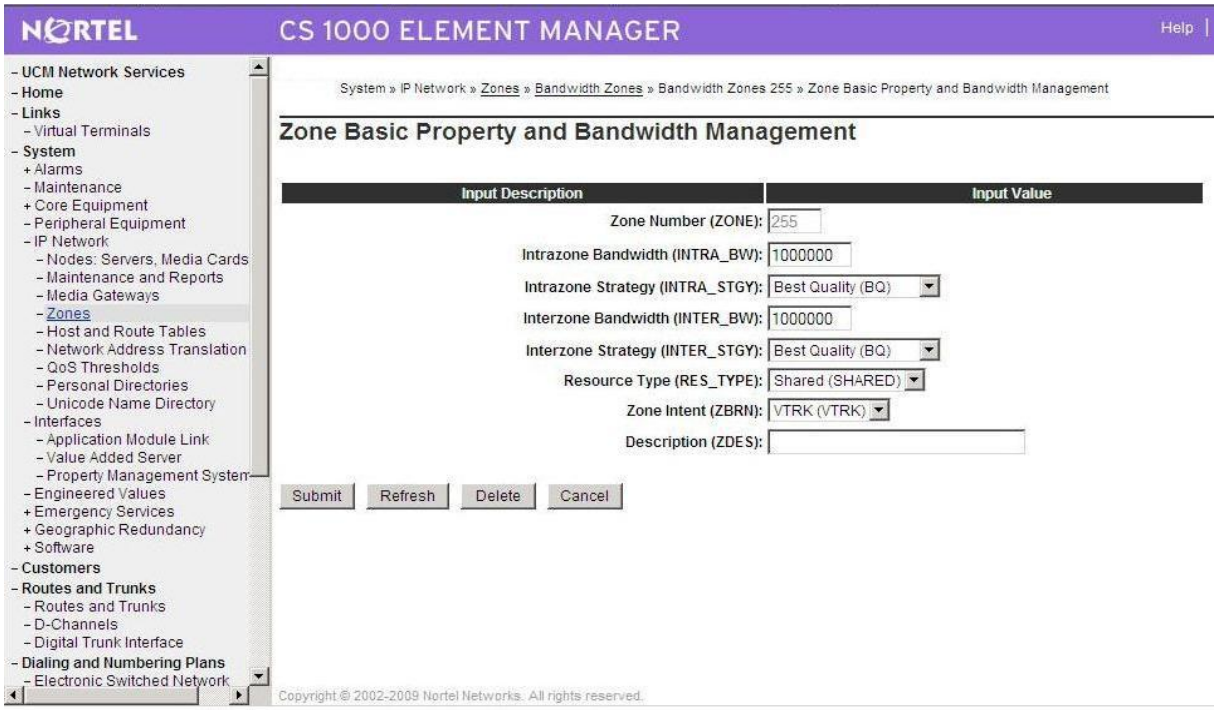

Figure 12 – Zone Basic Property Settings for (virtual) SIP trunk

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

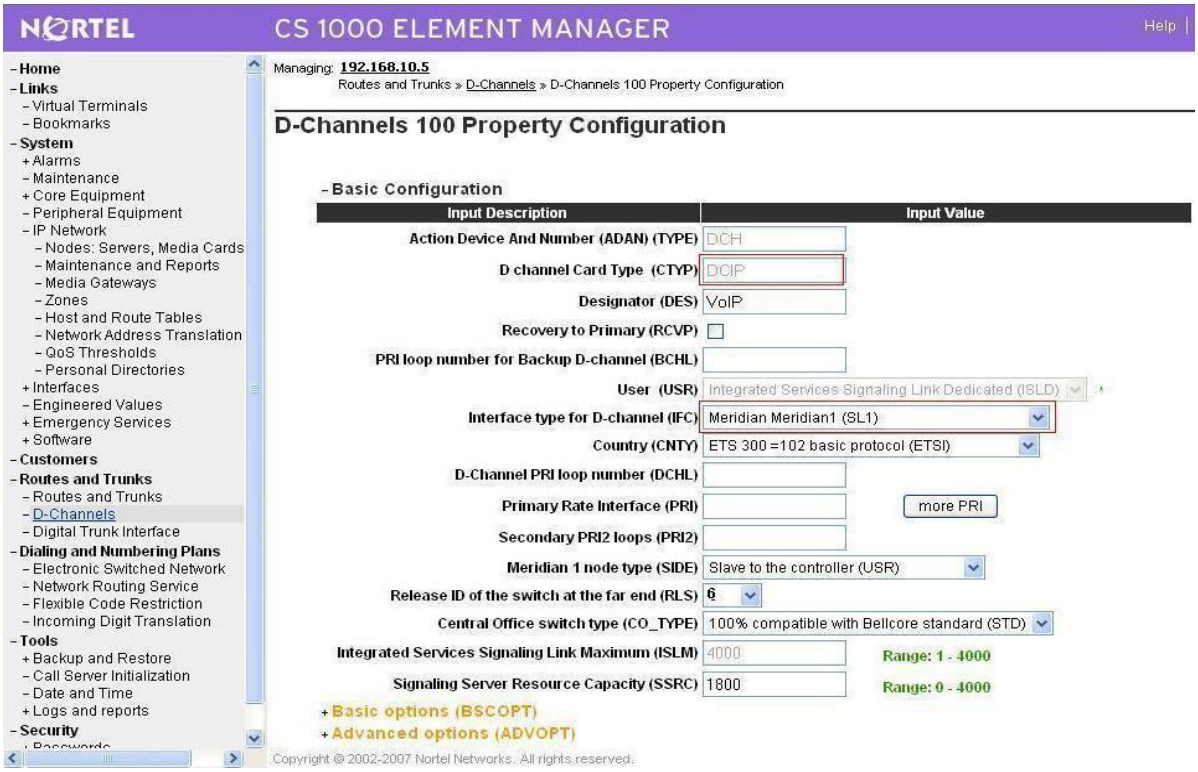

Figure 13 – D-Chanel Configurations

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

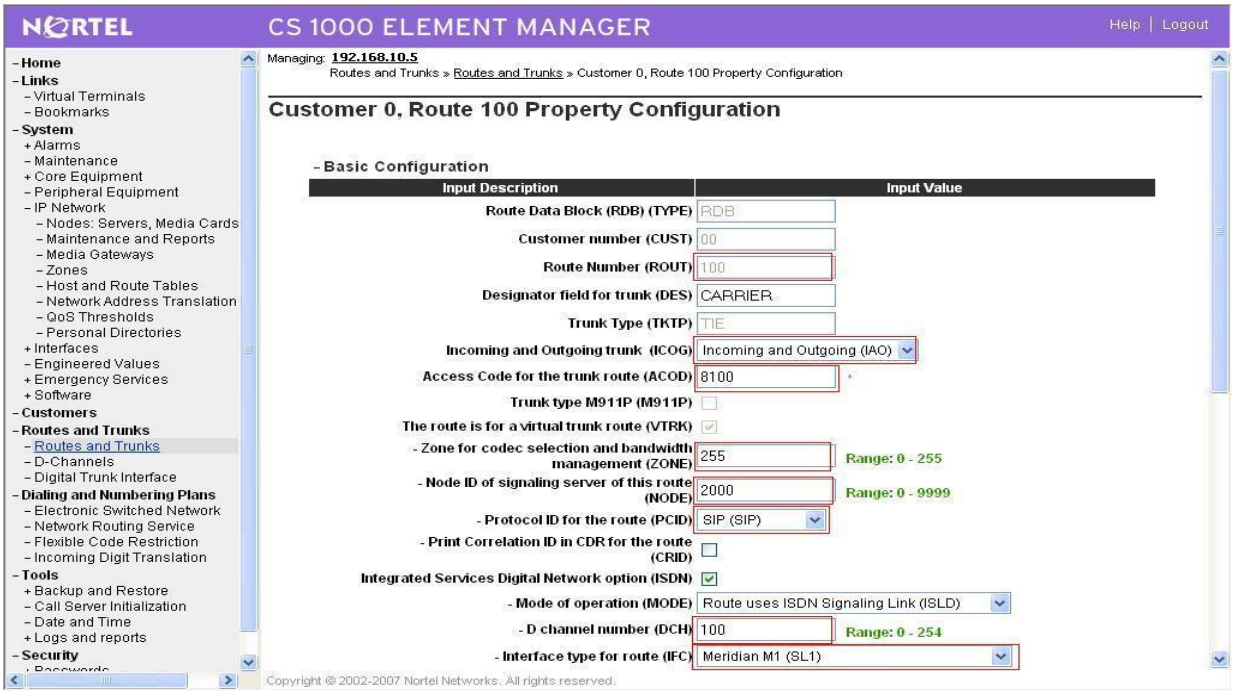

Figure 14 – Route Property Configuration

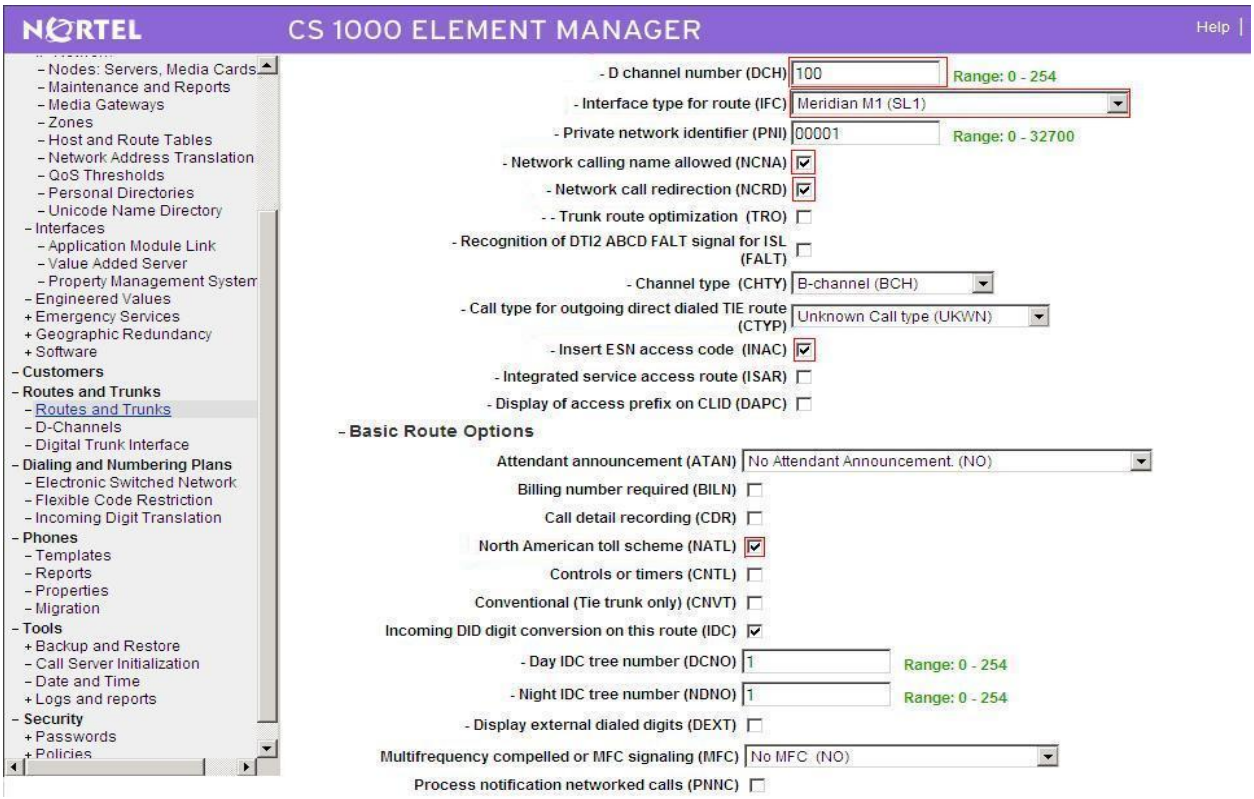

Figure 15 – Route Property Configuration (Cont..)

3. Create trunk: **C**reate trunk using basic configuration in figure 16

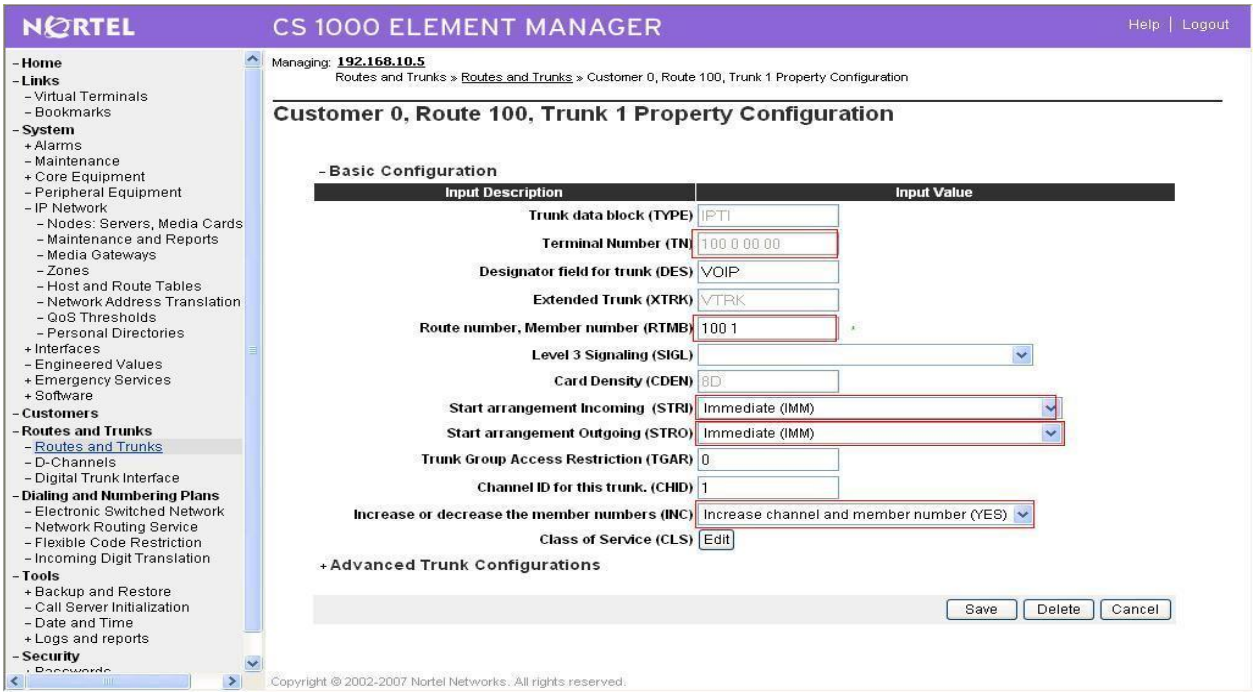

Figure 16 – Basic Trunk Configuration

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 shown in figure 17

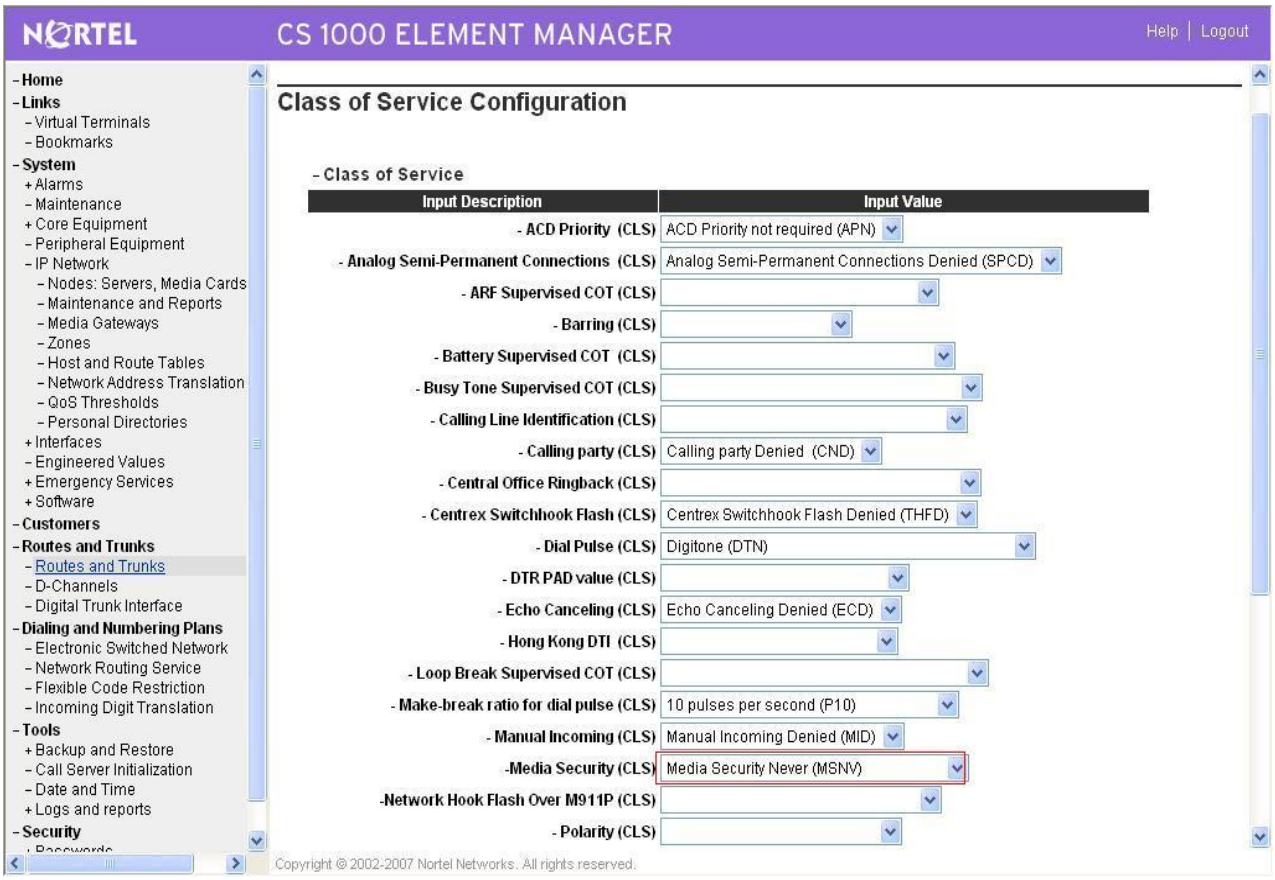

Figure 17 – Class of Service

Since the carrier does not support TLS or sRTP, we have them disabled 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.

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).

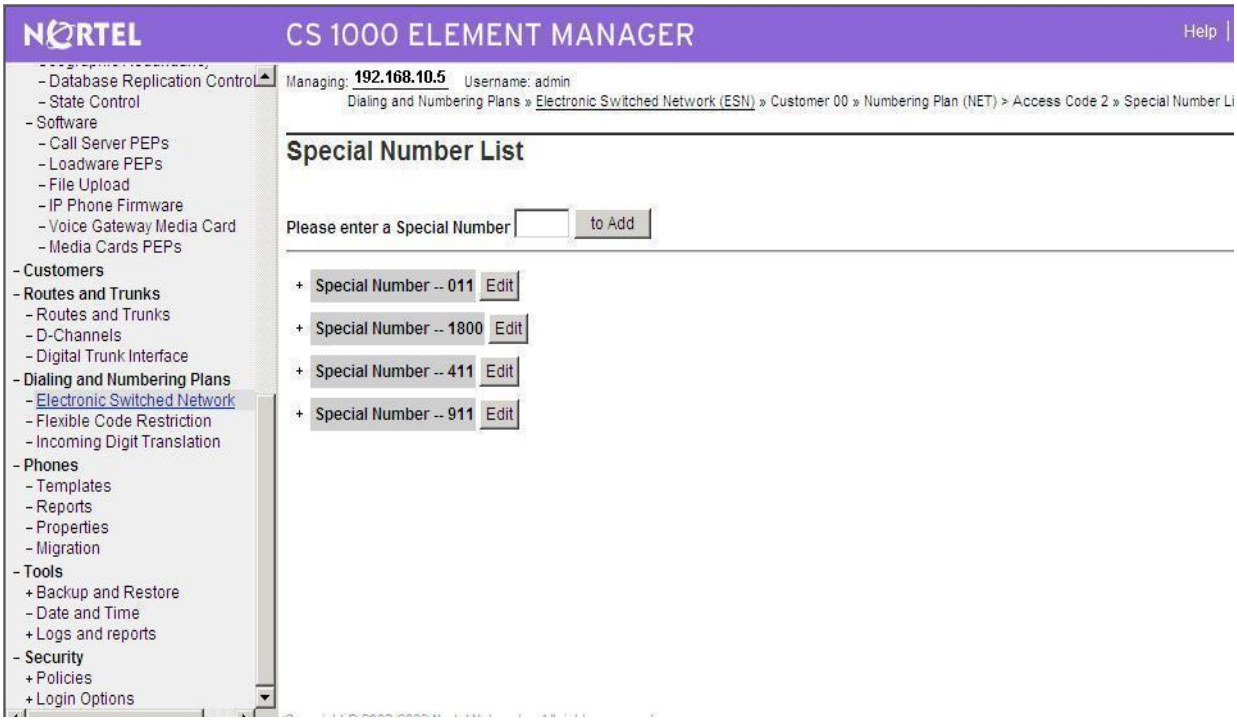

Figure 18 – Special Number List

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

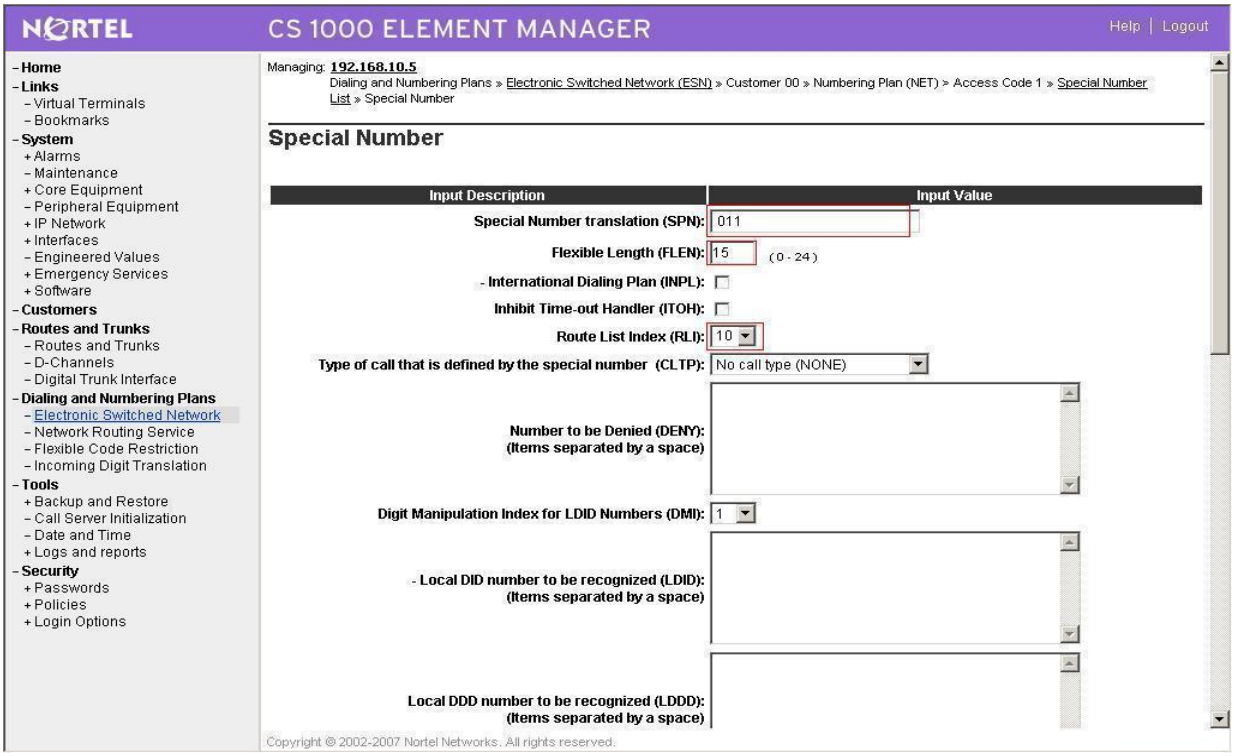

Figure 19 – Special Number for International Calls

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

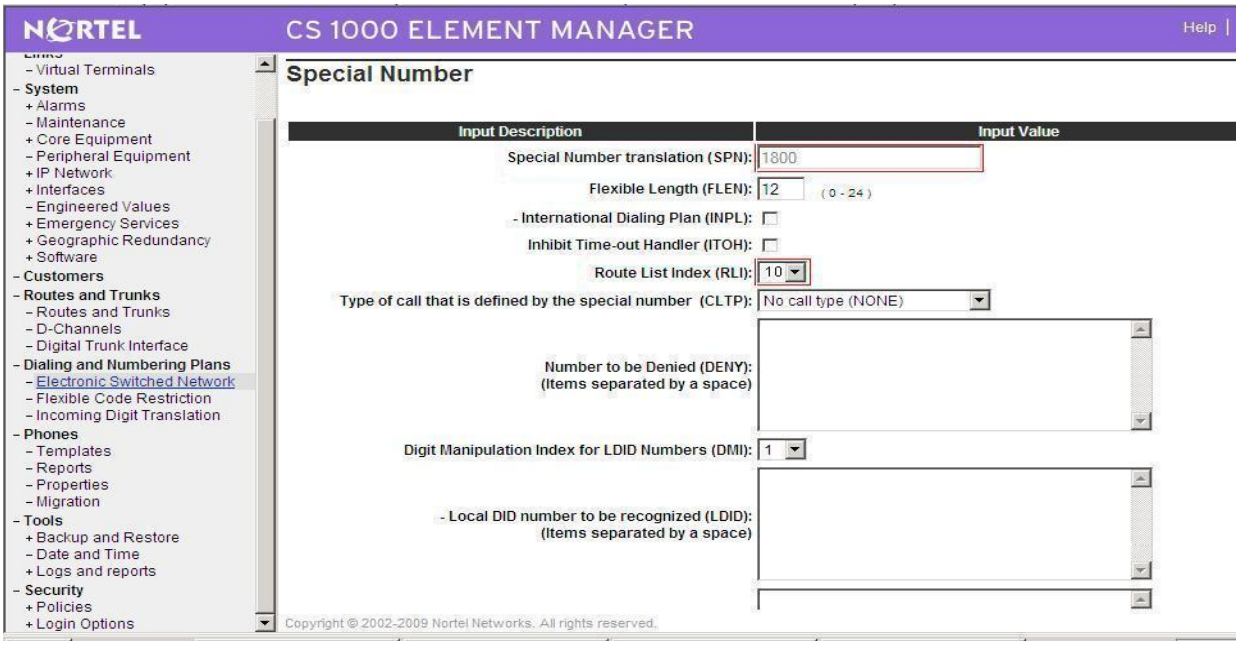

Figure 20 – Special Number for Tool Free Call

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

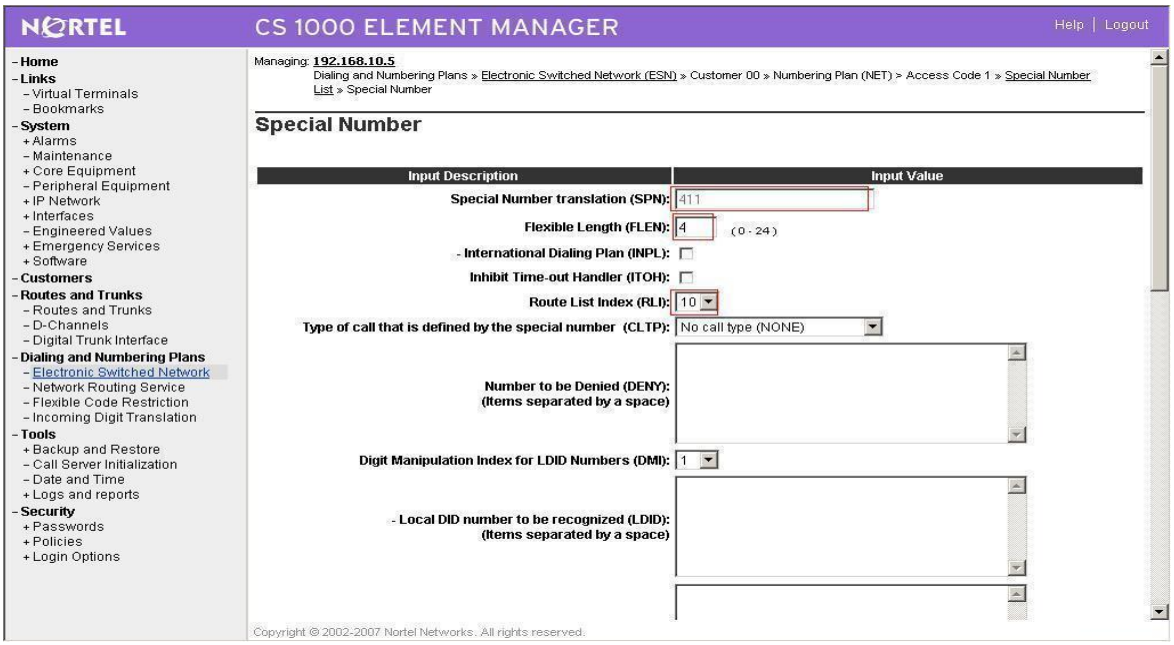

Figure 21 – Special Number for 411 Service Call

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

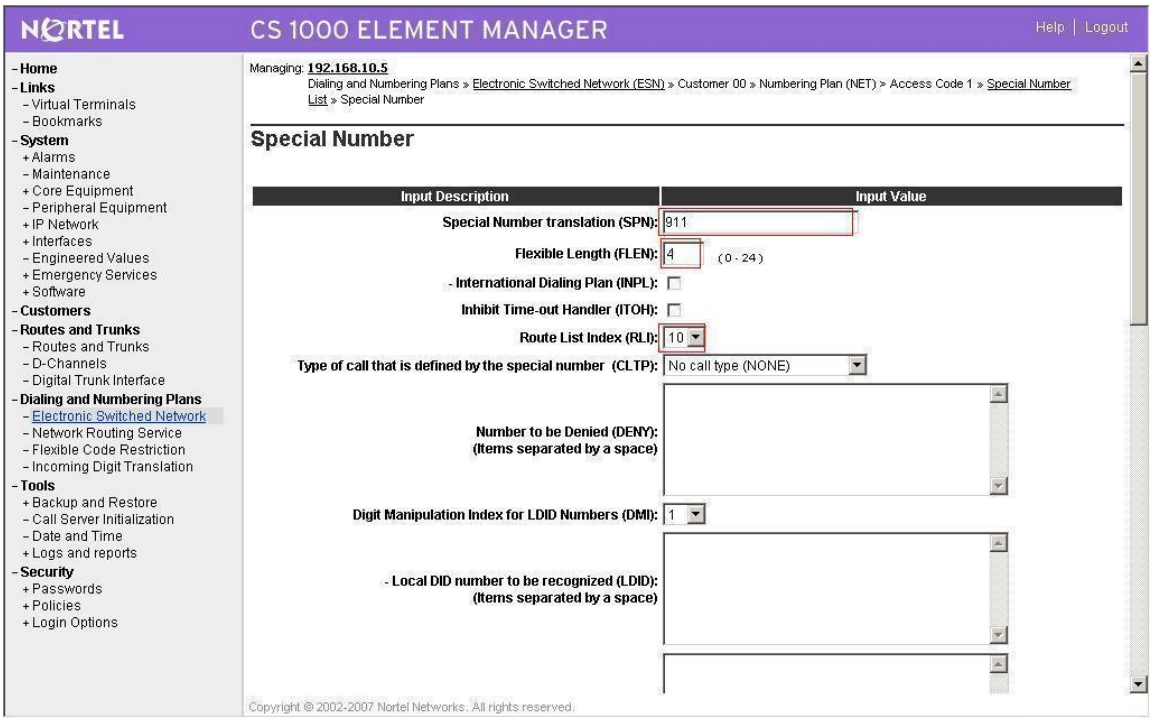

Figure 22 – Special Number for Emergency 911 dialing

Create Numbering Plan Area Code:

Create NPA numbers for outgoing.

NPA_1585: Create NPA_1585 for outgoing calls to numbers beginning with 585; Figure 23

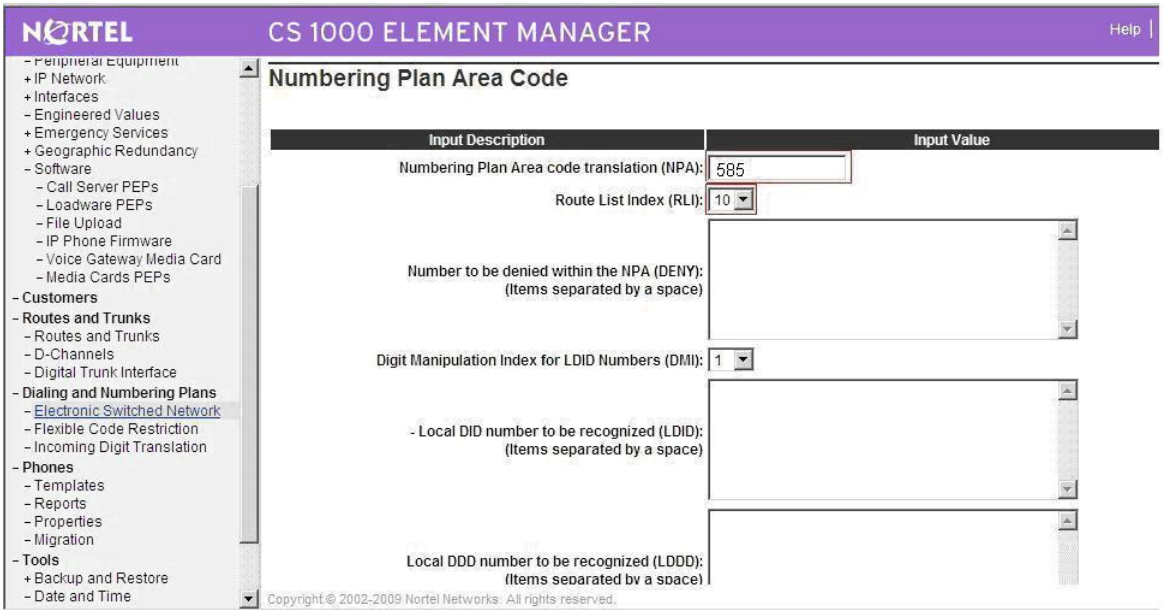

Figure 23 – Create NPA_1585 for outgoing calls

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

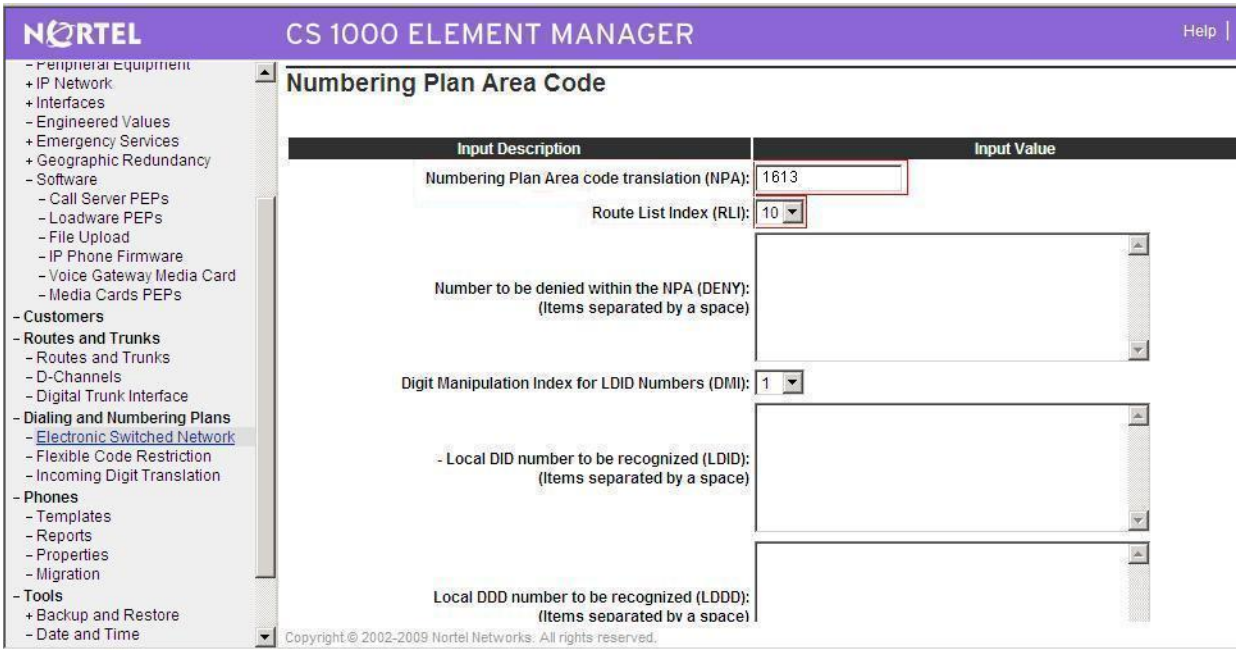

Figure 24 – Create NPA_1613 for outgoing calls to national numbers

Create Route List Block

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

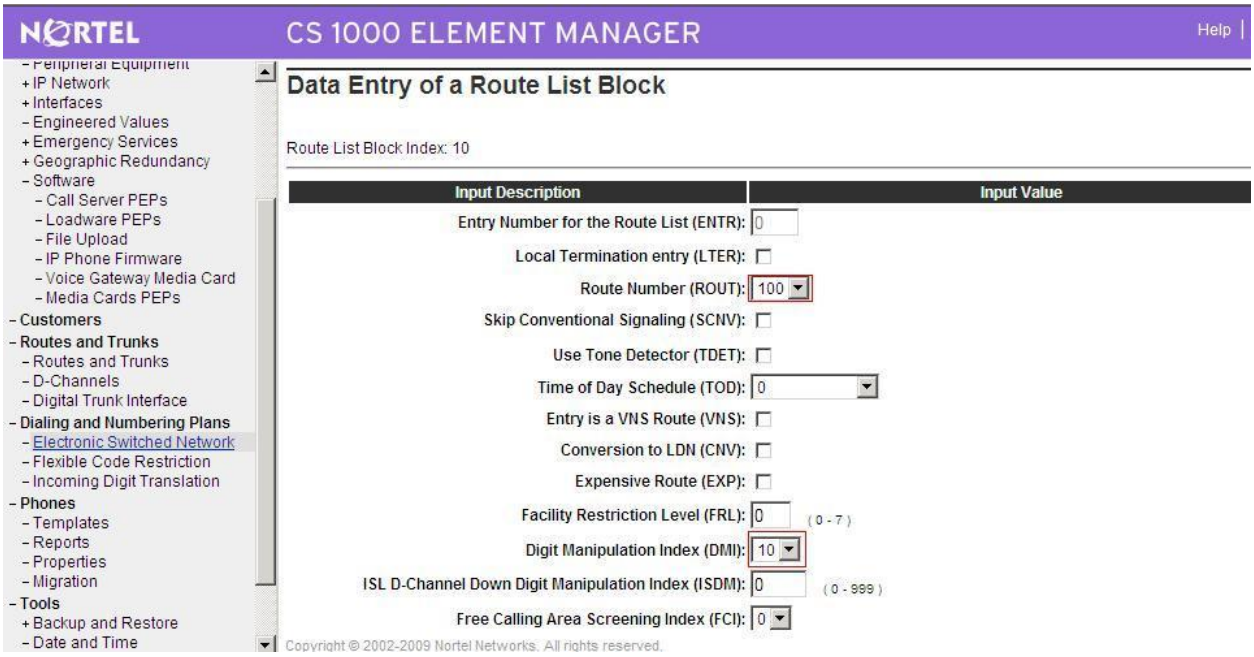

Figure 25- RLB for Outgoing calls

Create Local Steering Code

Create LSC_585 to terminate the incoming calls (Use DMI_3); Figure 26

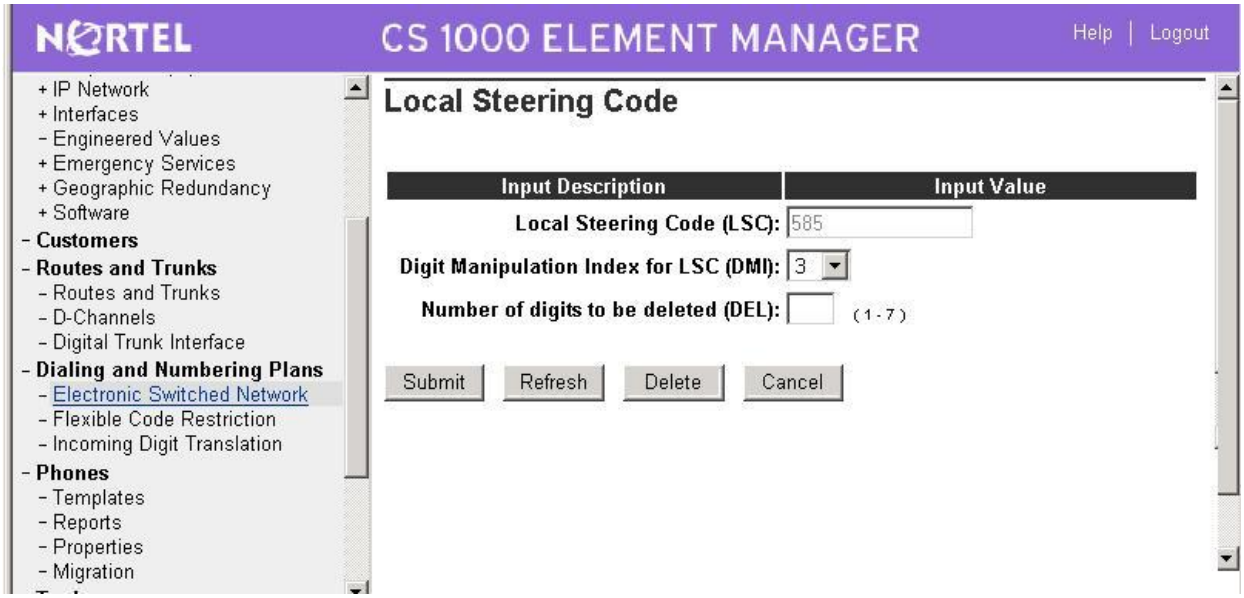

Figure 26 – Create LCS_585 for incoming call

Create Digit Manipulation Block

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

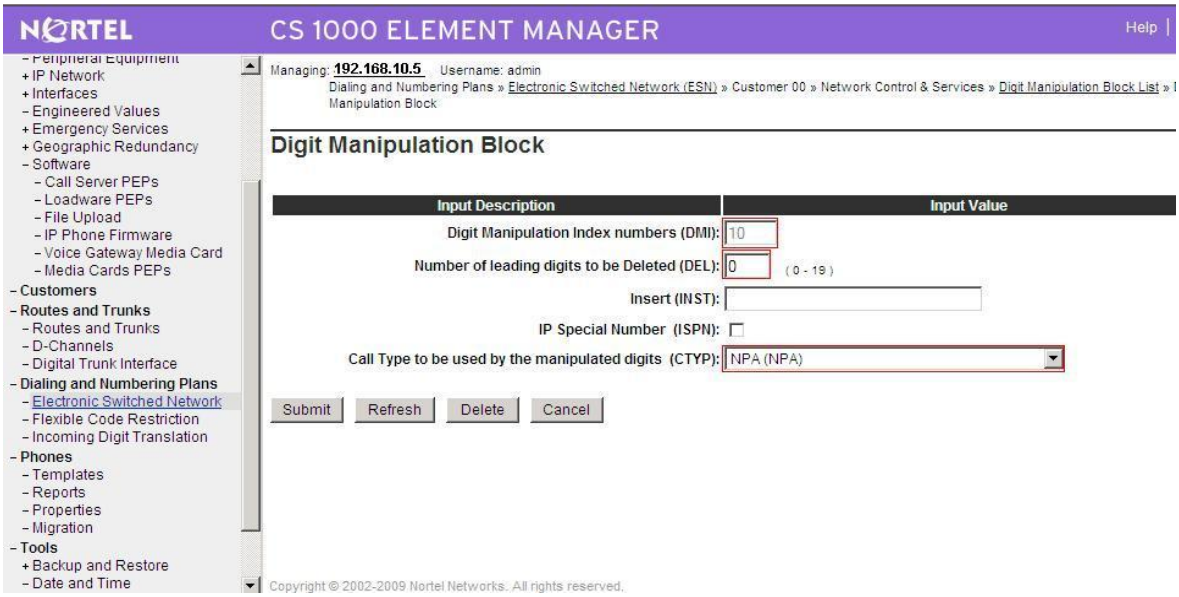

Figure 27 – Digit Manipulation for Outgoing calls

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

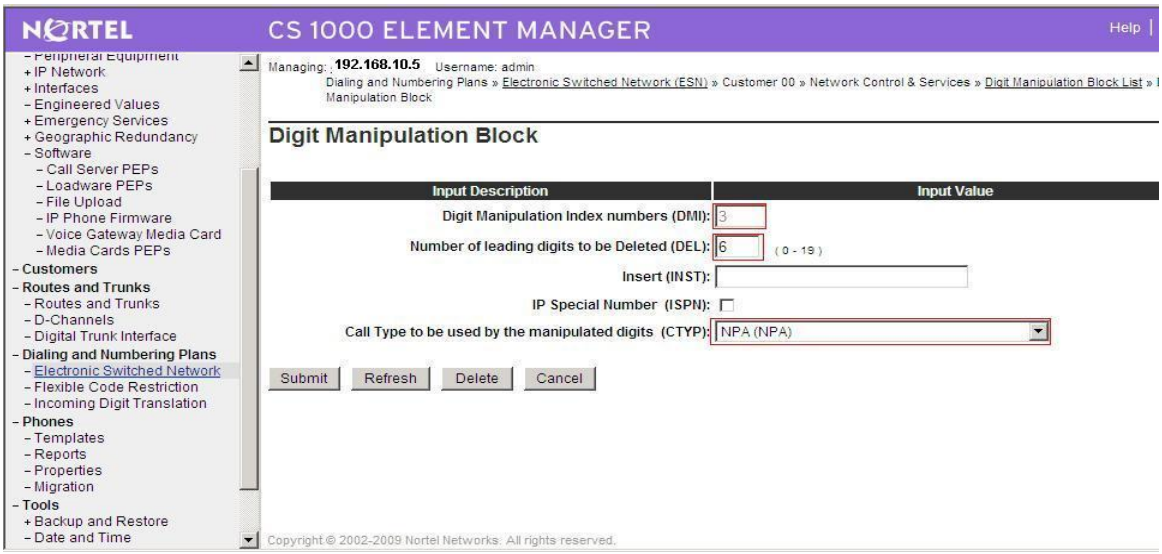

Figure 28 – Digit Manipulation for Incoming calls

Configure on CS1000 Voicemail System (Call Pilot)

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

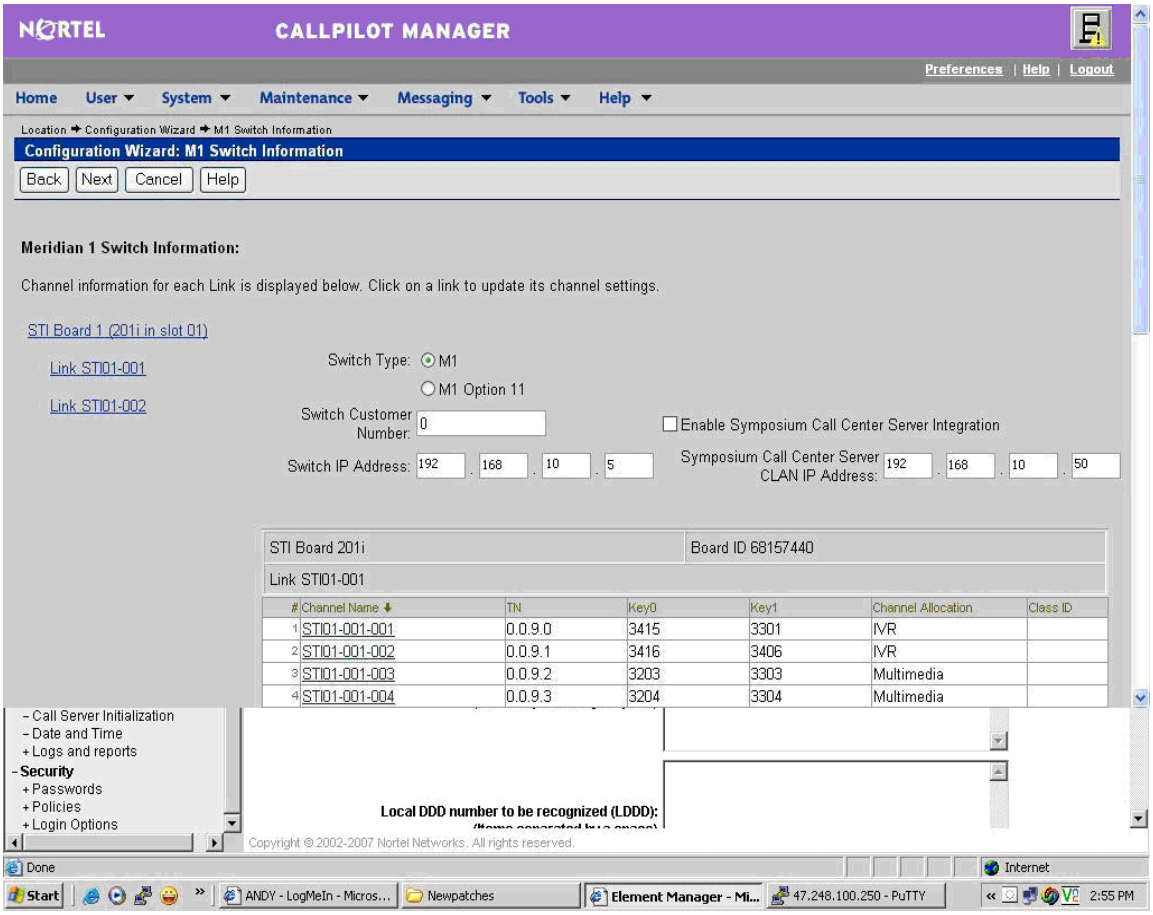

Figure 29 – CS1000 switch configuration on CallPilot Manager

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.

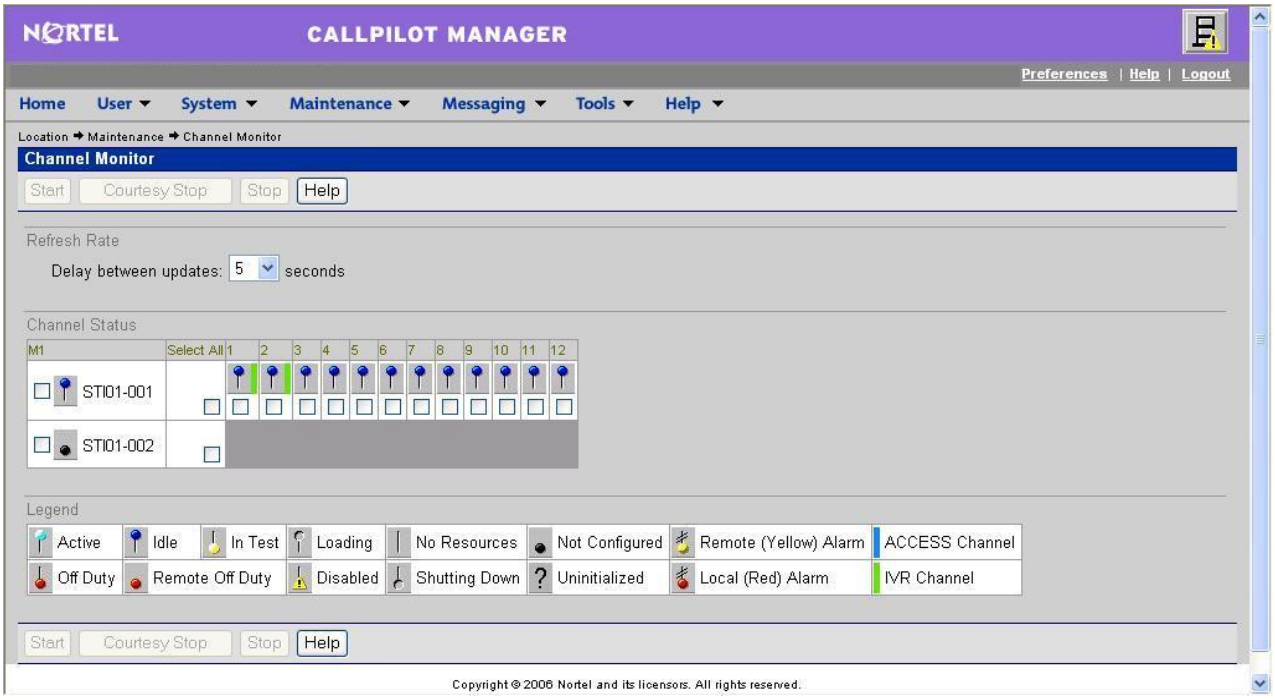

Figure 30 – Channel Monitor

Create Service DN for Voice Messaging system, Figure 31

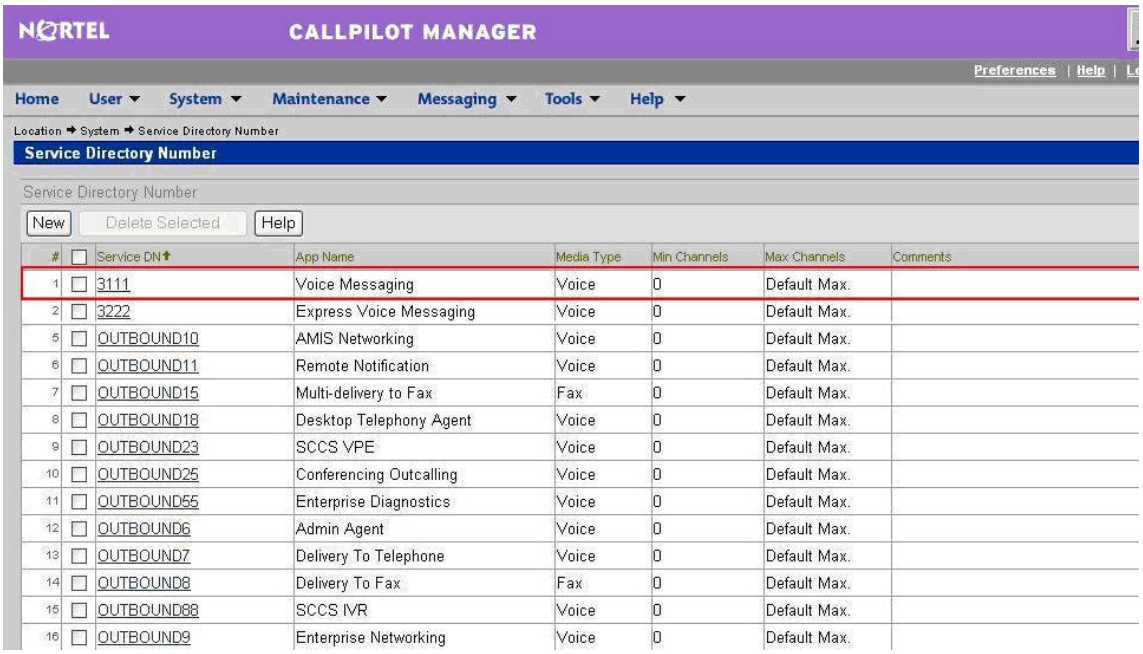

Figure 311 – Service Directory Number Page

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

Output configuration details from CS1000 and Callpilot

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

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

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
```

CS1K Tandem Configuration

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

Configure CS1000E A

1. Create IP Telephony Node on CS1000E

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

- Input Node ID and Click on **Add** button.
- Enter TLAN, ELAN IP address of Signaling Server.

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

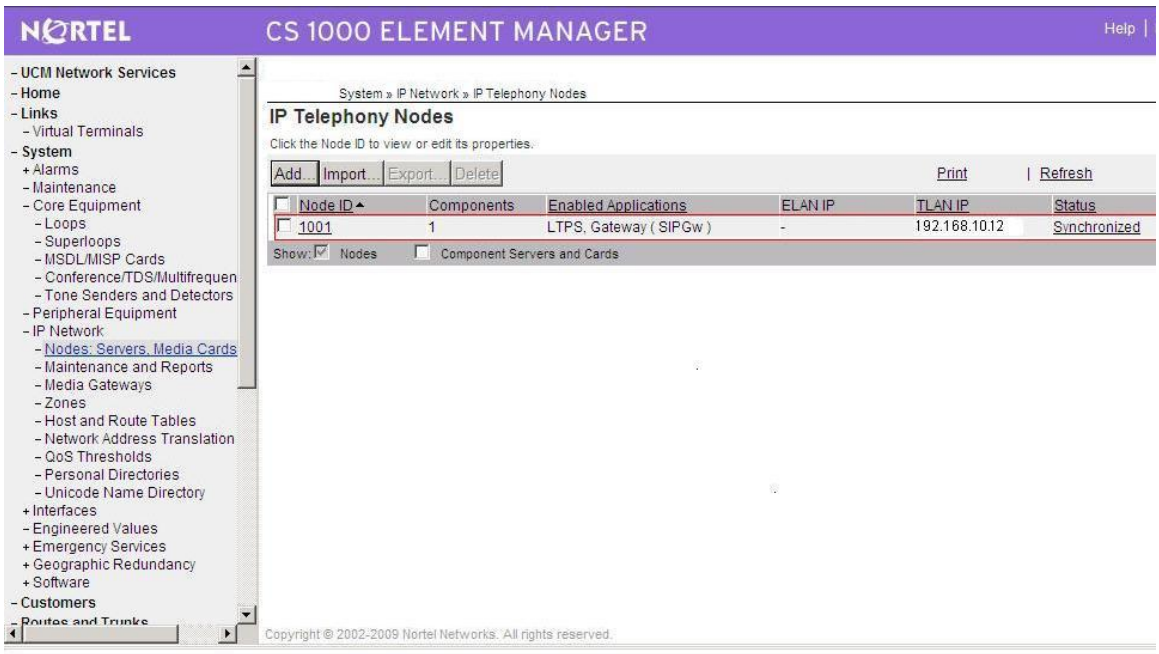

Figure 32 – Creating a node on CS1000 A

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

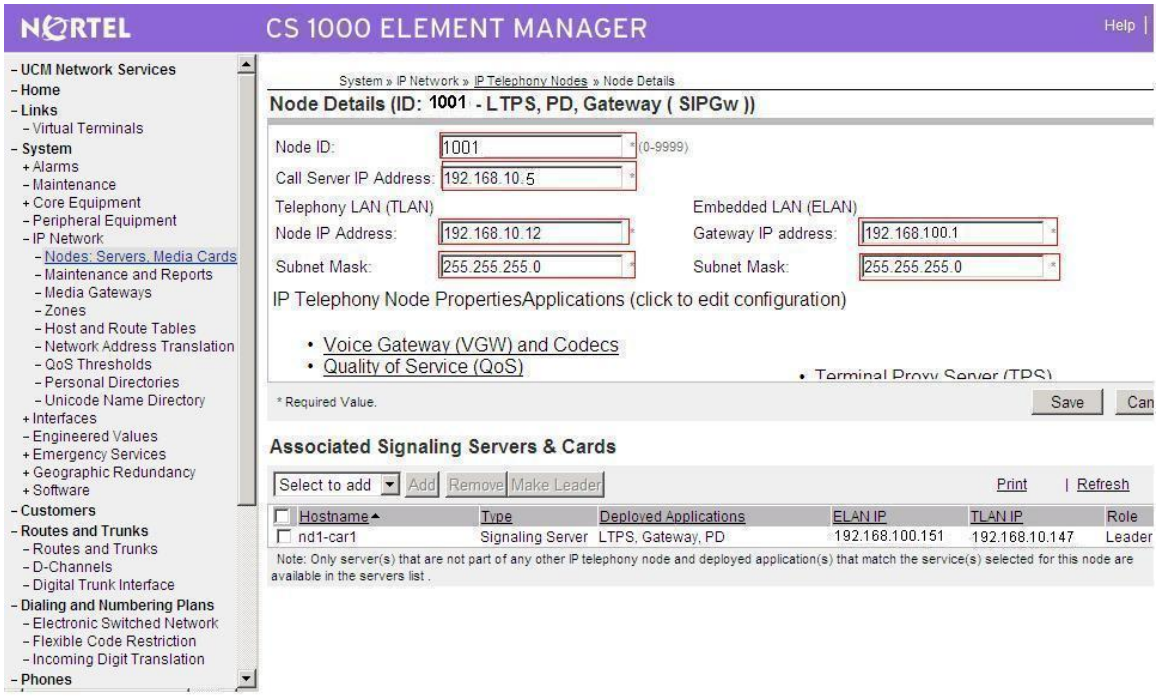

Figure 33 – CS1000 Node Detail Settings

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

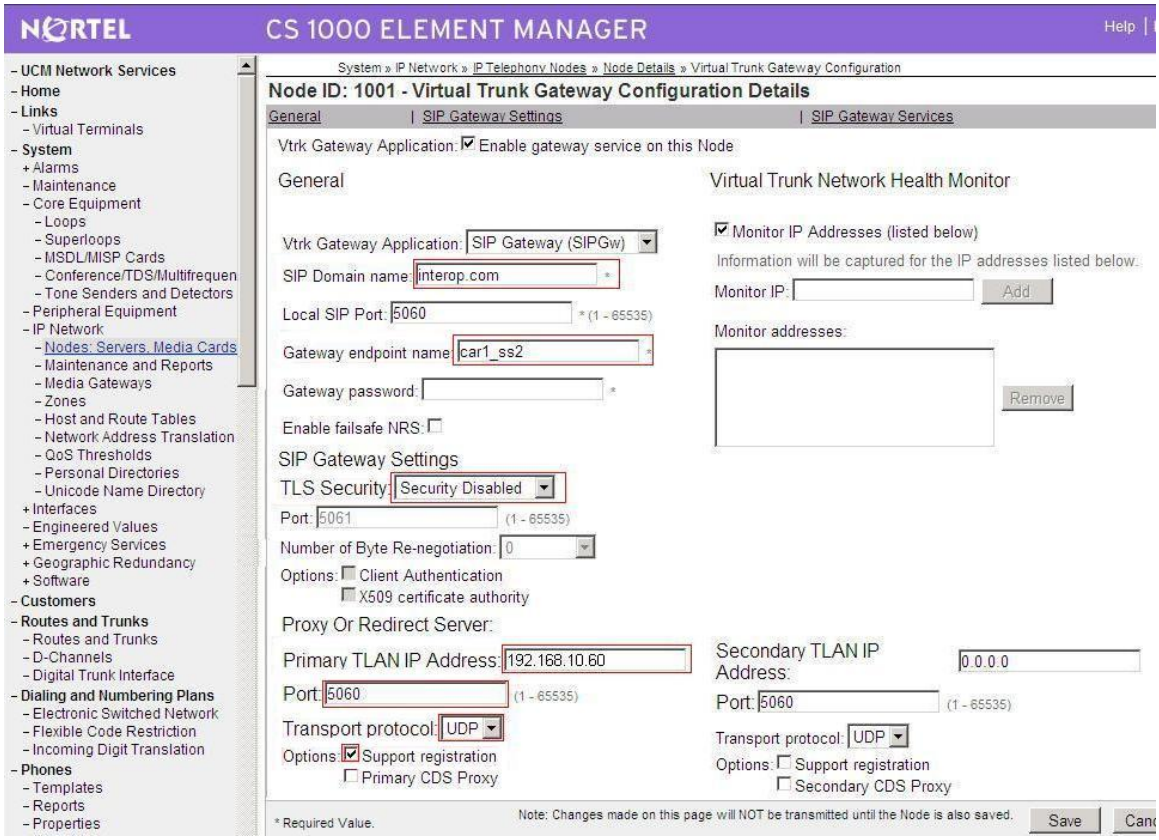

Figure 34 – SIP Gateway Settings

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 as shown in figure 35; Also click on Options and edit the Remote Capabilities (RCAP). Enable MWI if CS1K hosted voice mail will be used.

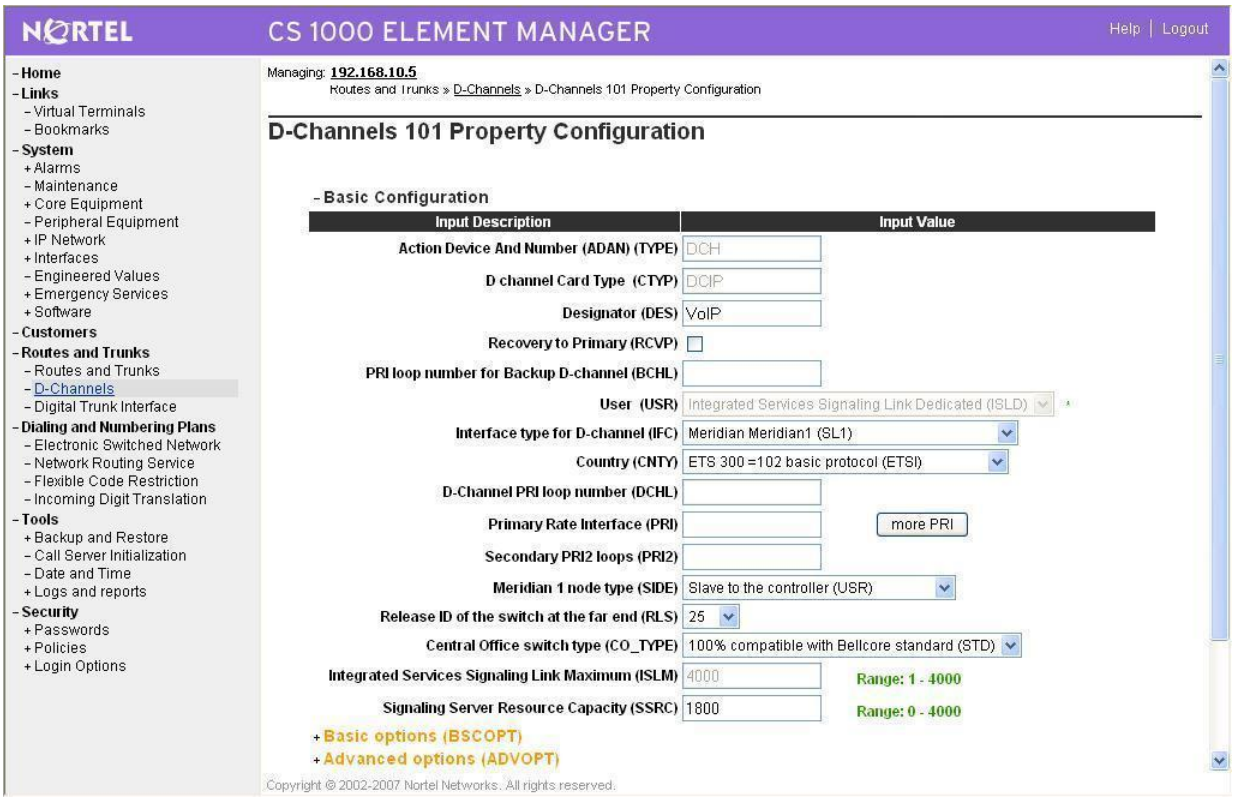

Figure 35 – D-Channels Property Configuration

3. Create Route

Create route 101 using DCH 101 for SIP trunks as shown in Figure 36

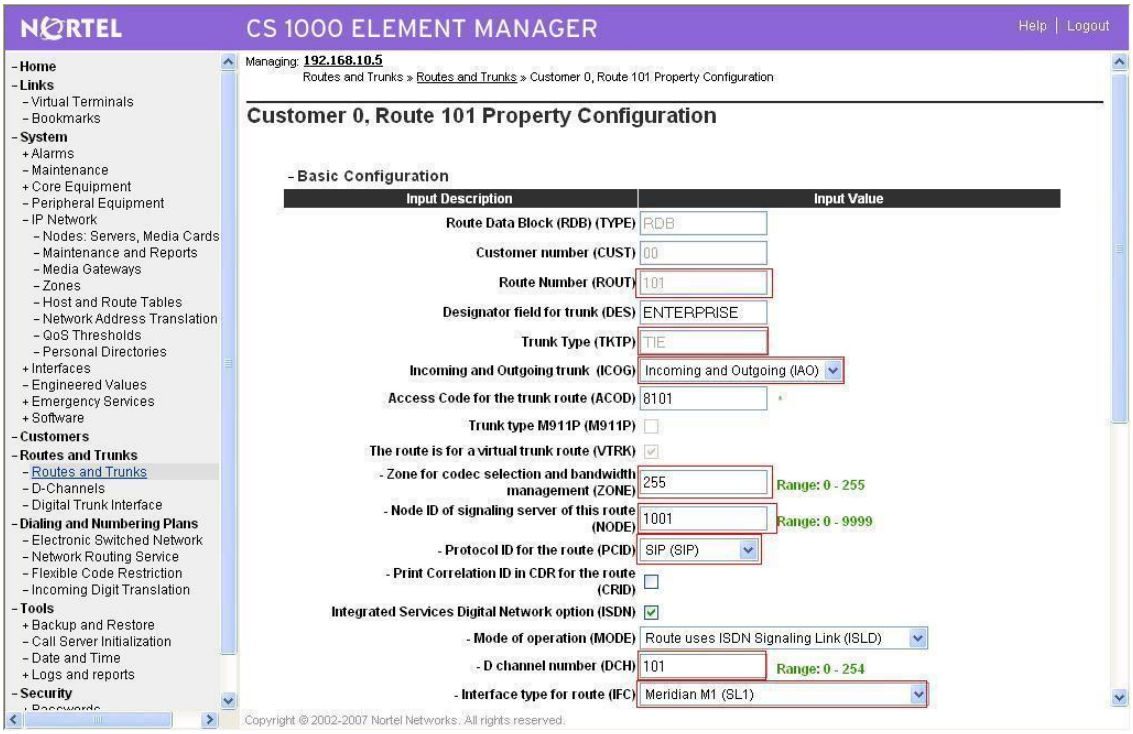

Figure 36 – Route Property Configuration

Configure Route 101 for SIP trunks continue as shown in figure 40

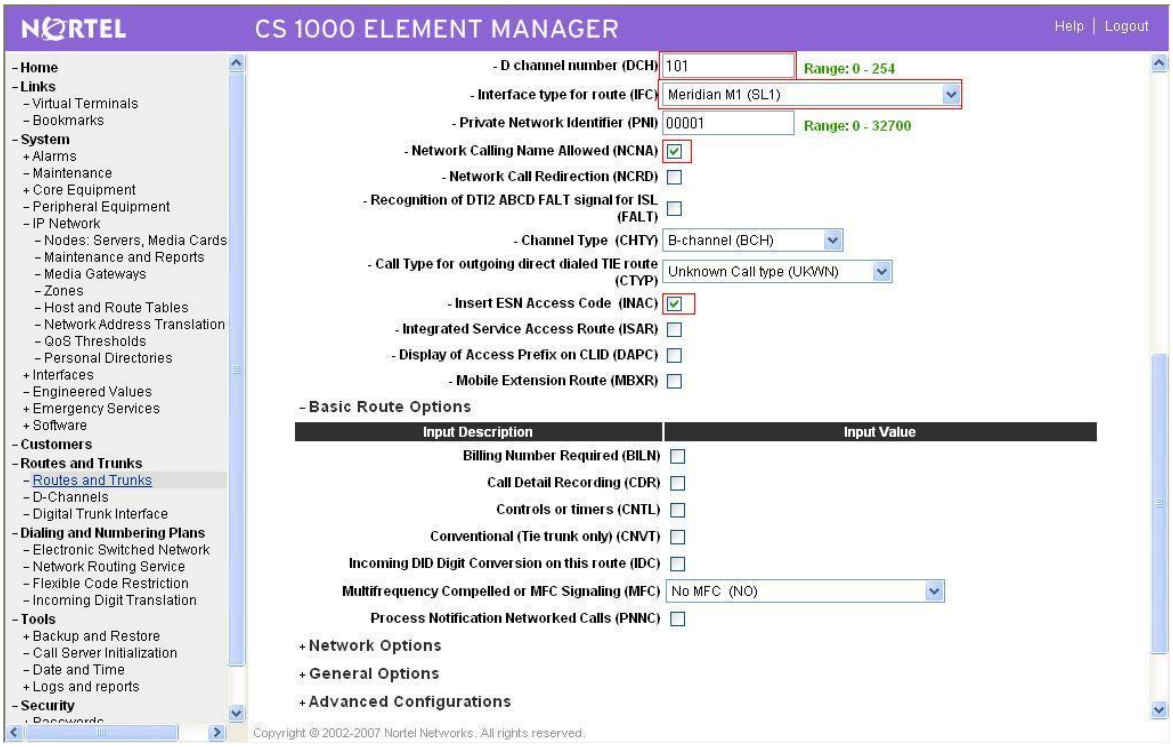

Figure 37 – Route Property Configuration Details (cont.)

4. Create Trunk (figure 38)

Since Media security is not supported under Frontier system, Disable Media Security (MSNV) at the Trunk level as show in figure 39

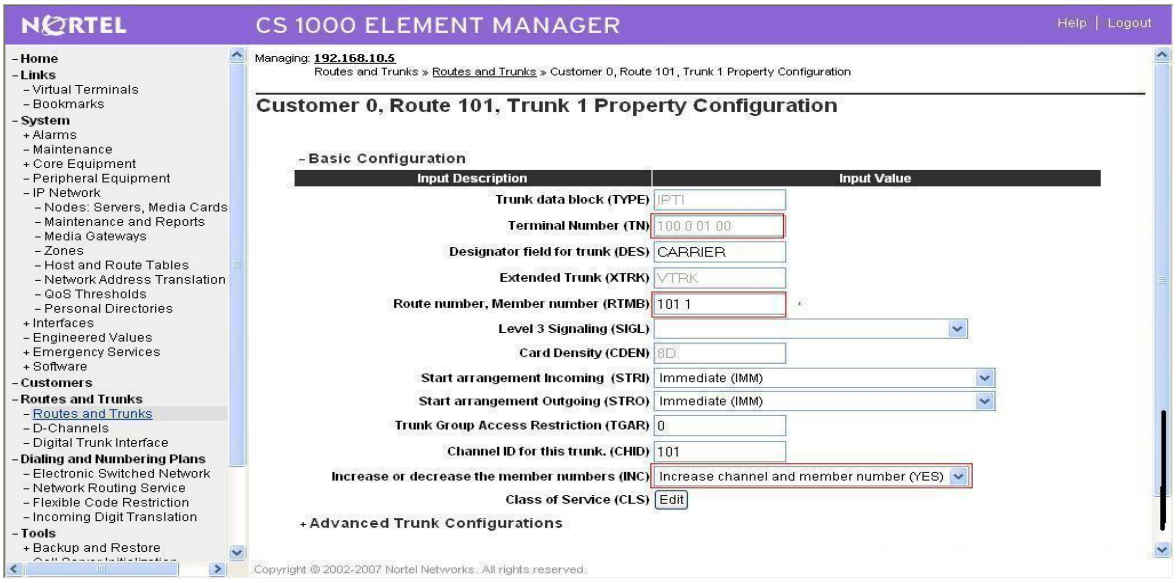

Figure 38 – Trunk Configuration Details

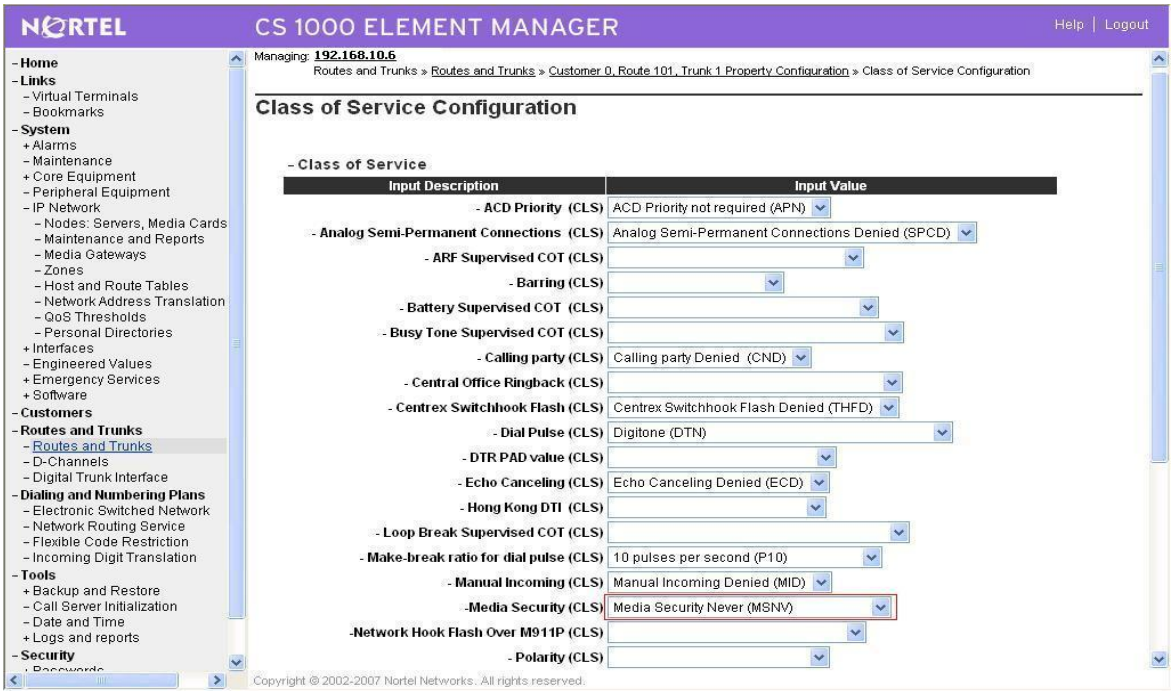

Figure 39 – Class of Services configuration Details (cont.)

5. 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

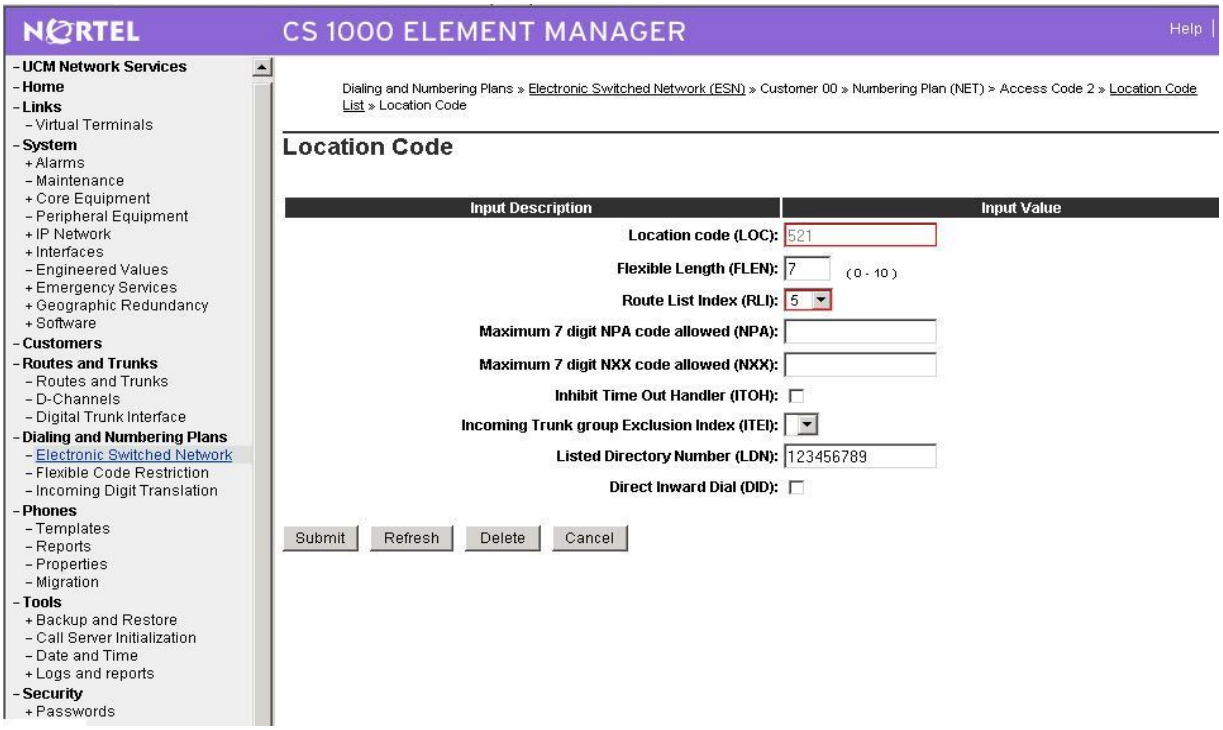

Figure 40 – LOC_521 for basic outgoing calls to CS1000EB

Create Home Location Code

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

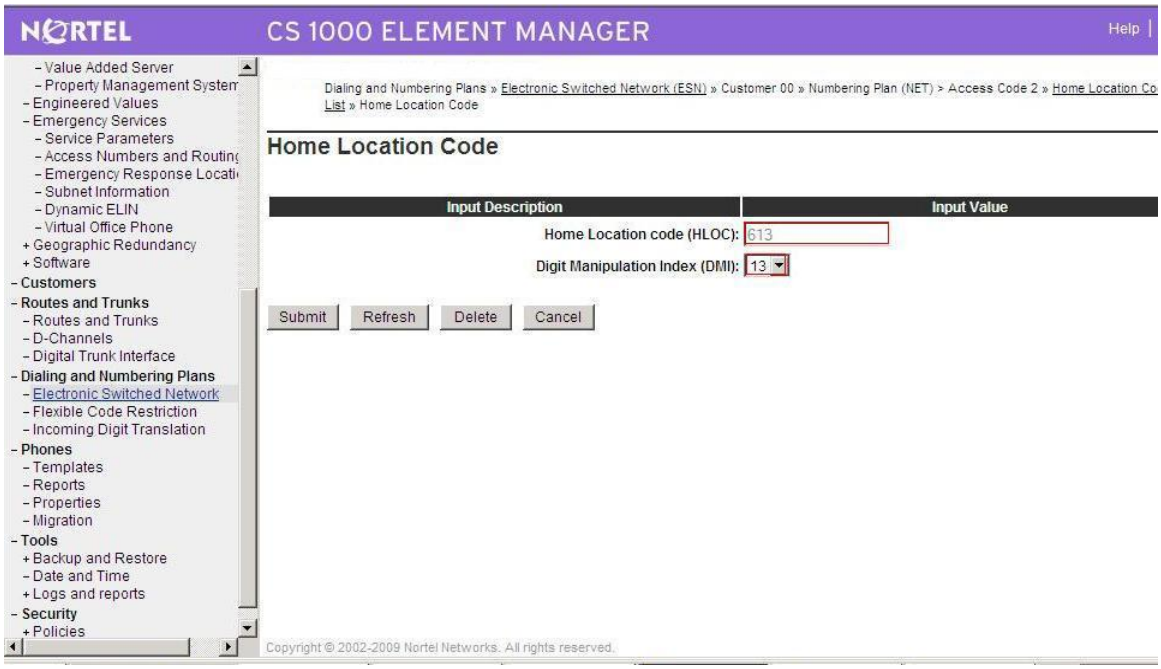

Figure 41 – HLOC: 613 to tandem calls from CS1000B to PSTN

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

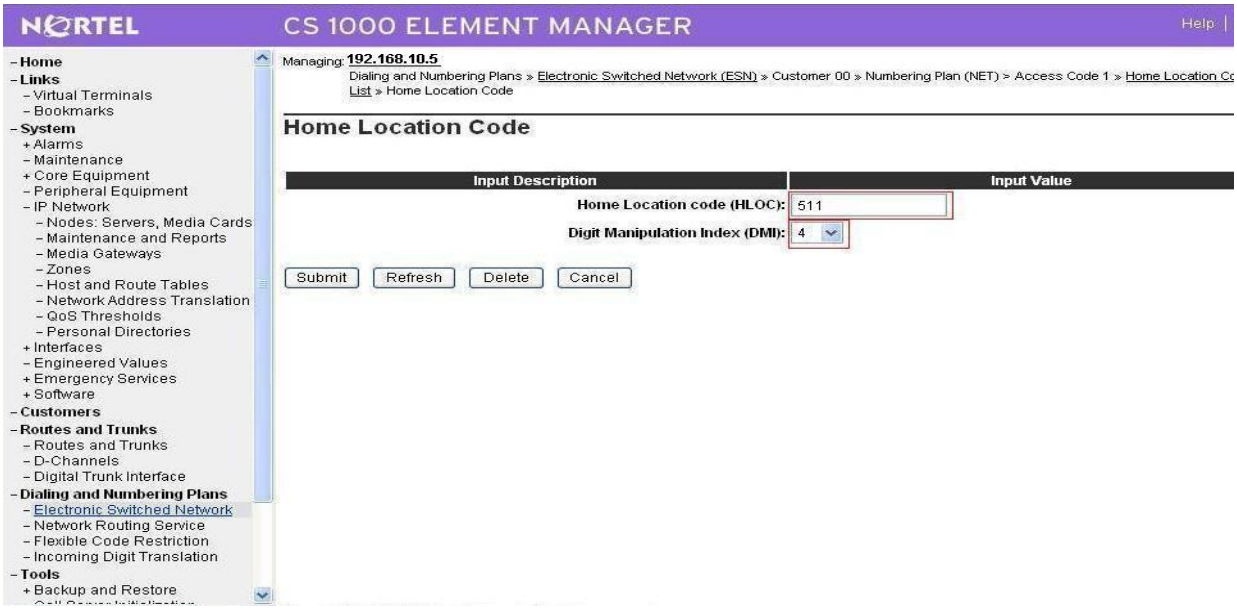

Figure 42 – HLOC_511 to terminate calls from CS1000E_B

Create Distant Steering Code

Create DSC_585 (RLI_6) to receive Calls from PSTN and tandem to CS1000E_B; Figure 43

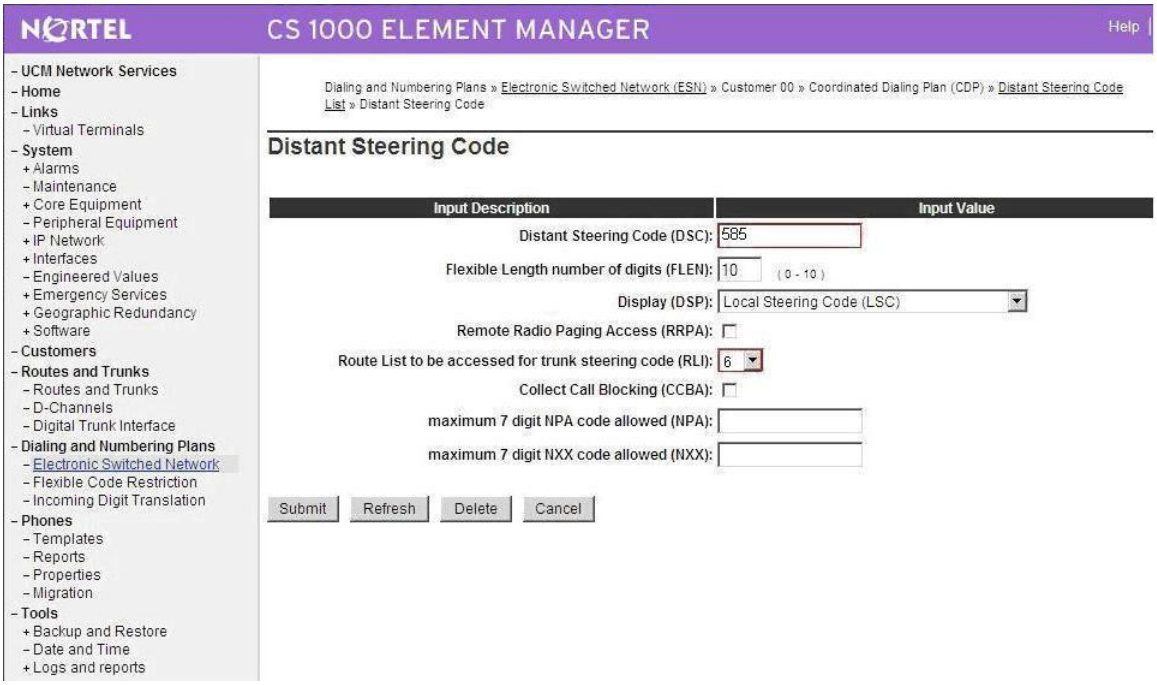

Figure 43 – DSC_585 to receive calls from PSTN and tendem to CS1000E_B

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

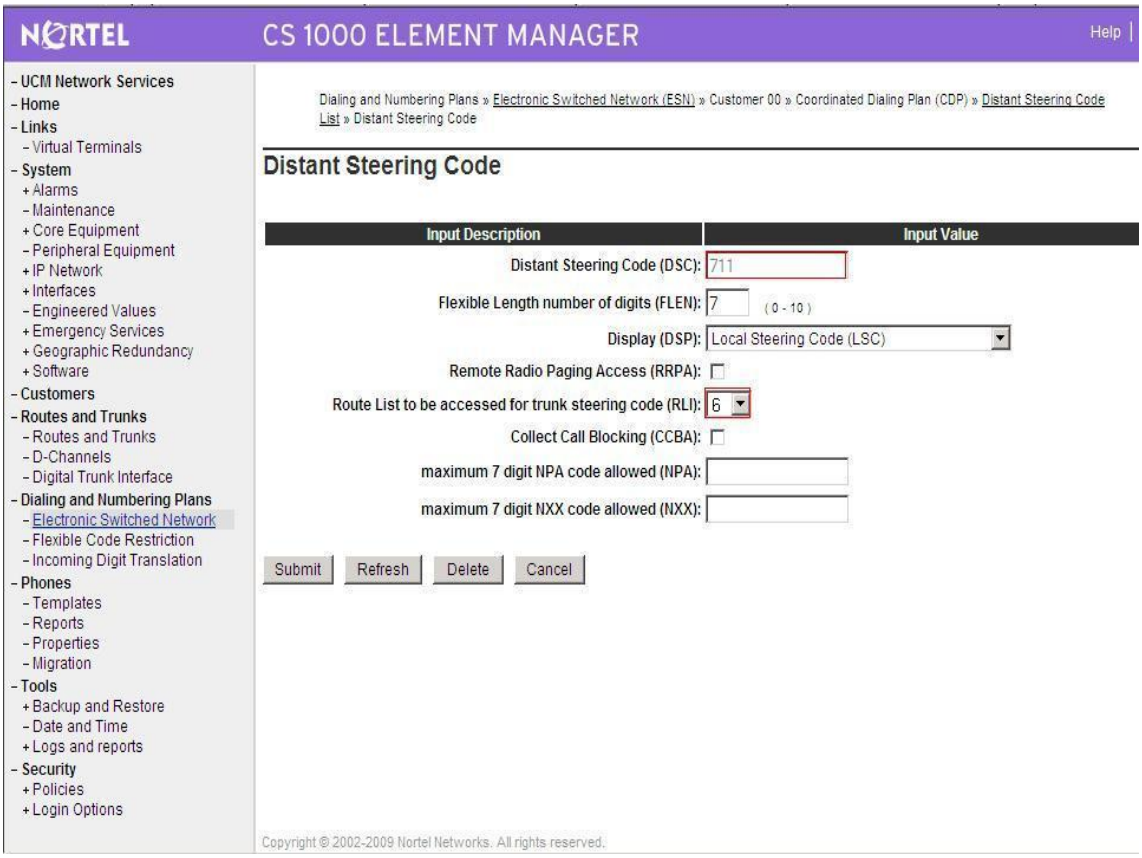

Figure 44 – DSC_711 to tandem calls to CS1000E_B.

Create Route List Block

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

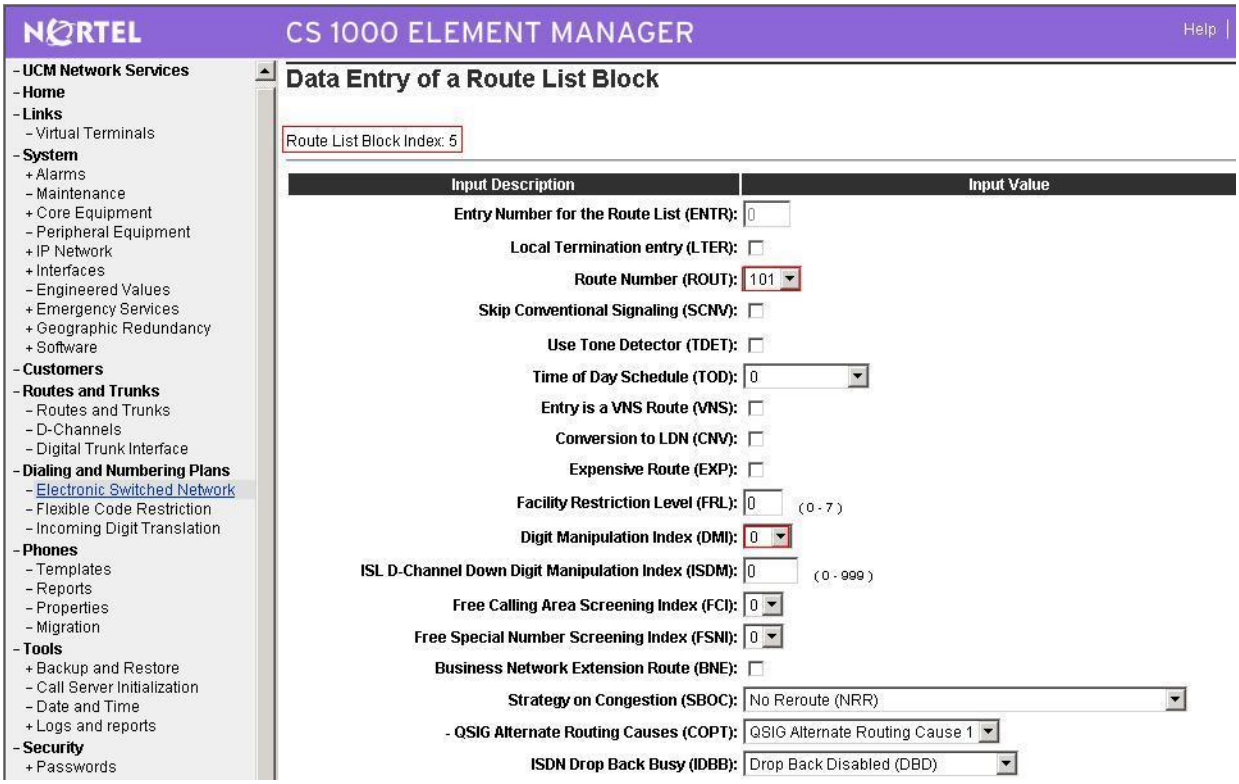

Figure 45 – Create RLI_5 for simple outgoing calls to CS1000E_B

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

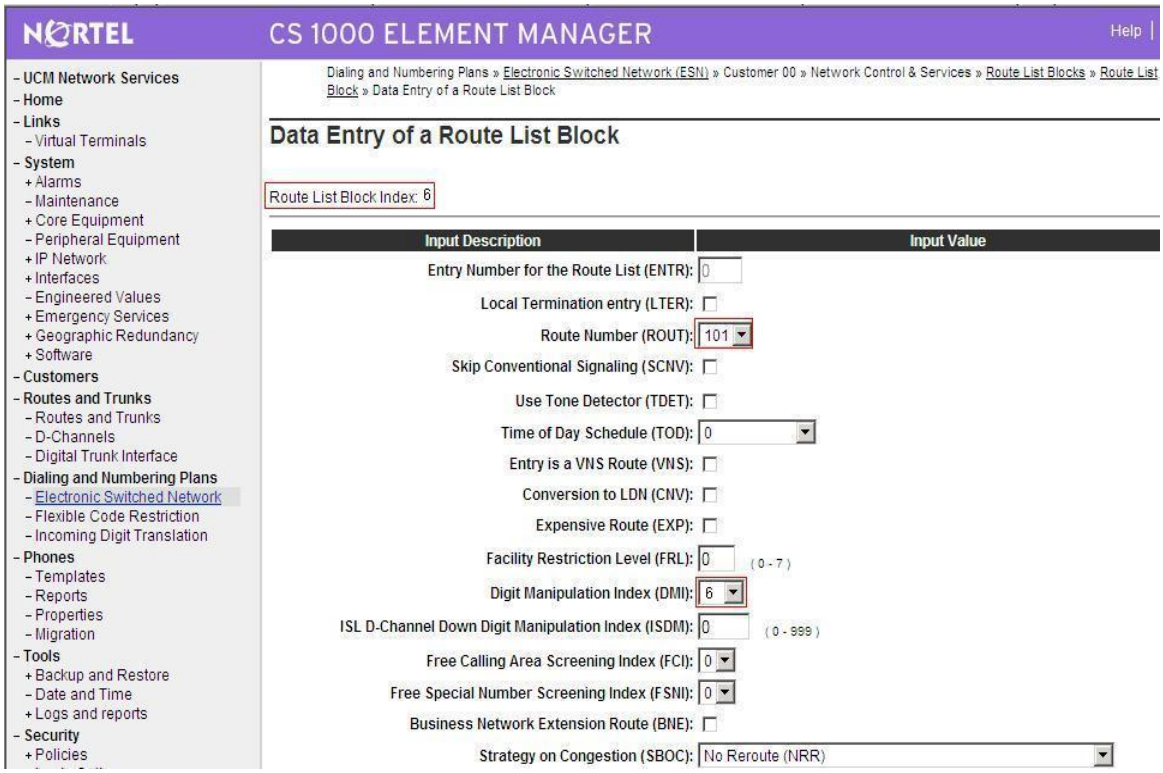

Figure 46 – RLI_6 to tandem calls from PSTN to CS1000E_B

Create Digit Manipulation Block

Create DMI_13 for Incoming calls from CS1000E_B and Outgoing to PSTN; Figure 47

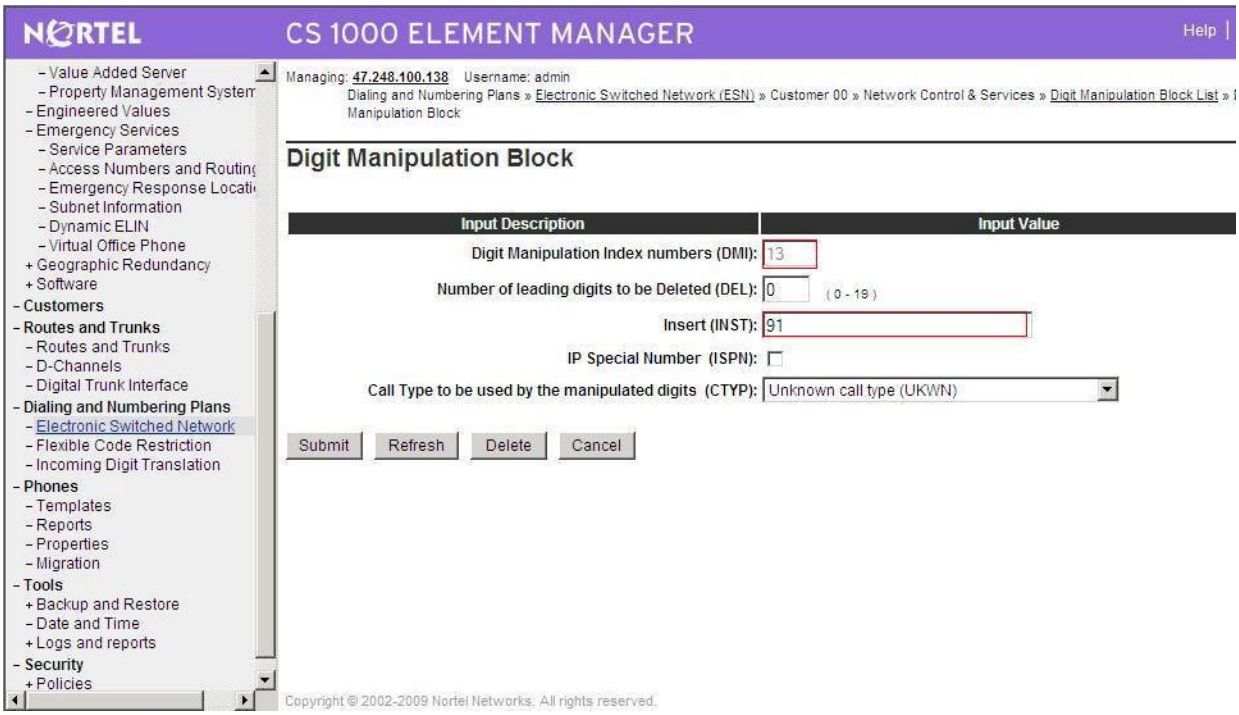

Figure 47 – DMI_13 for incoming from CS1000E_B and outgoing calls to PSTN

Create DMI_4 for incoming calls from CS1000E_B; Figure 48

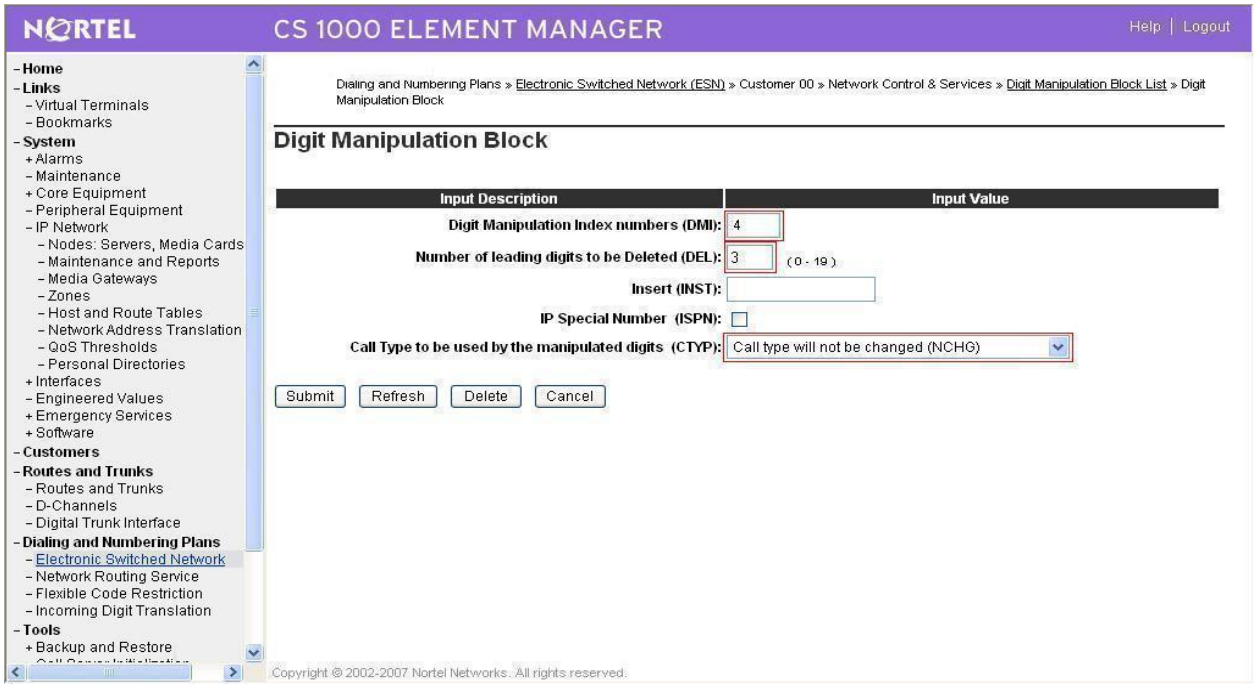

Figure 48 – DMI_4 to terminate calls from CS1000E_B

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

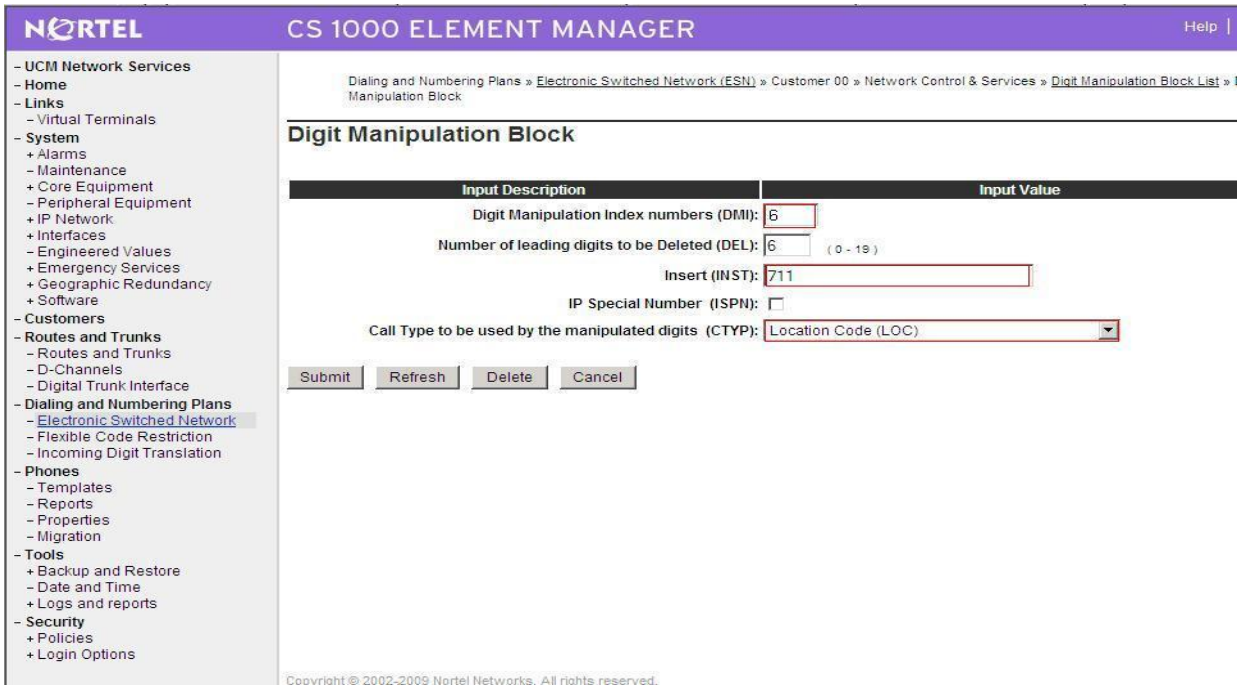

Figure 49 – DMI_6 for incoming calls from PSTN and tandem to CS1000E_B

Configure CS1000E 6.0 B:

1. Create IP Telephony Node 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 Signaling Server).

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

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

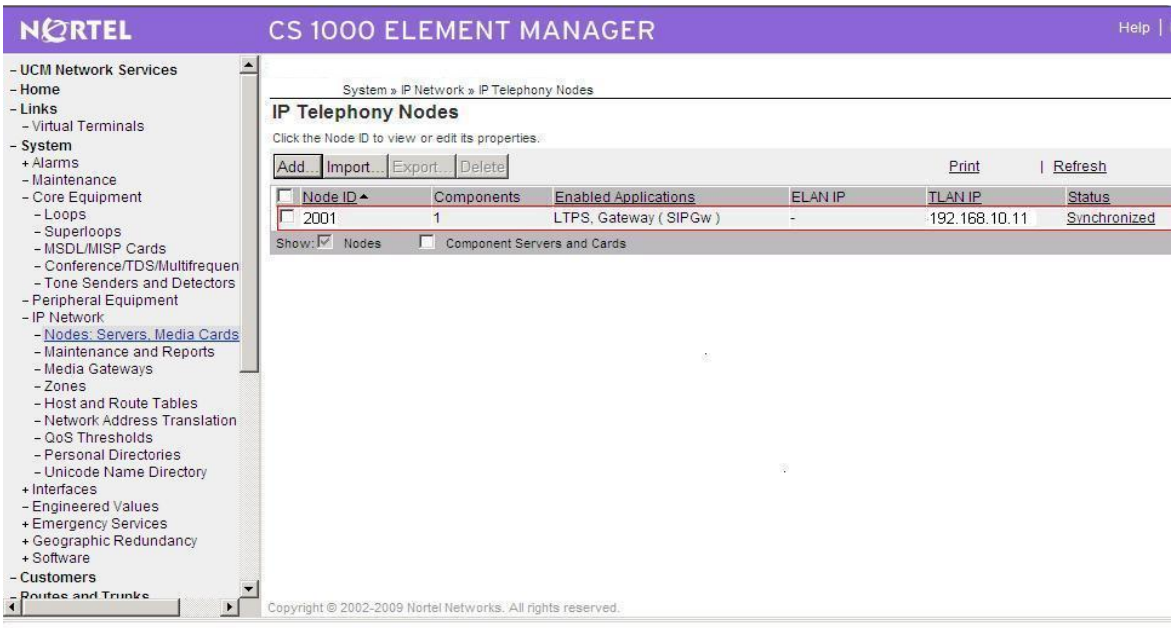

Figure 50 – Node Configured as Enterprise Service SIP Gateway

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 as shown in figure 51

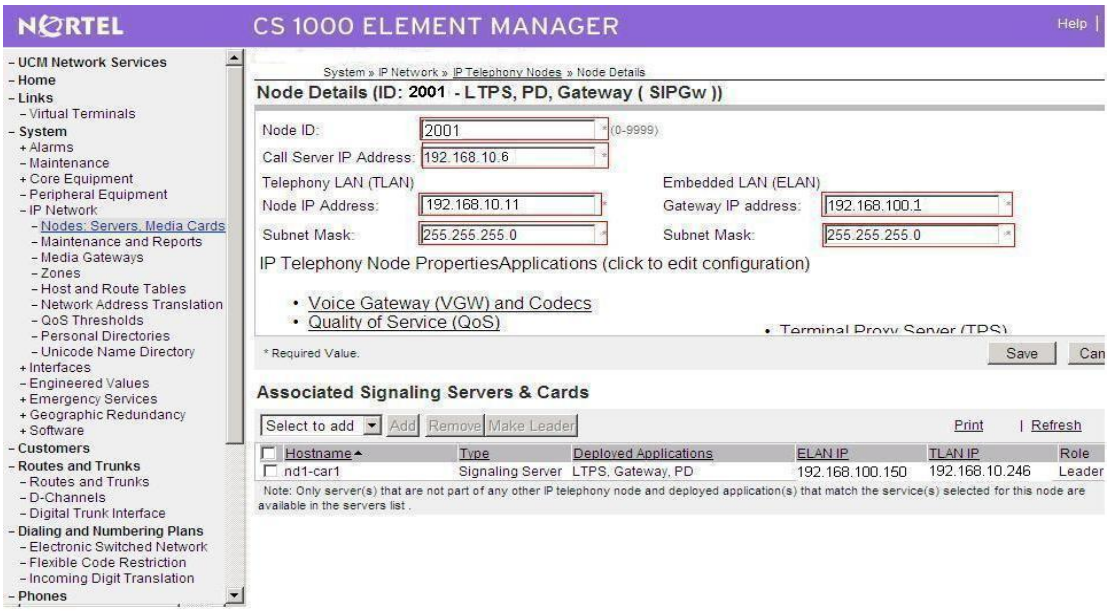

Figure 51 – Node Details Configuration

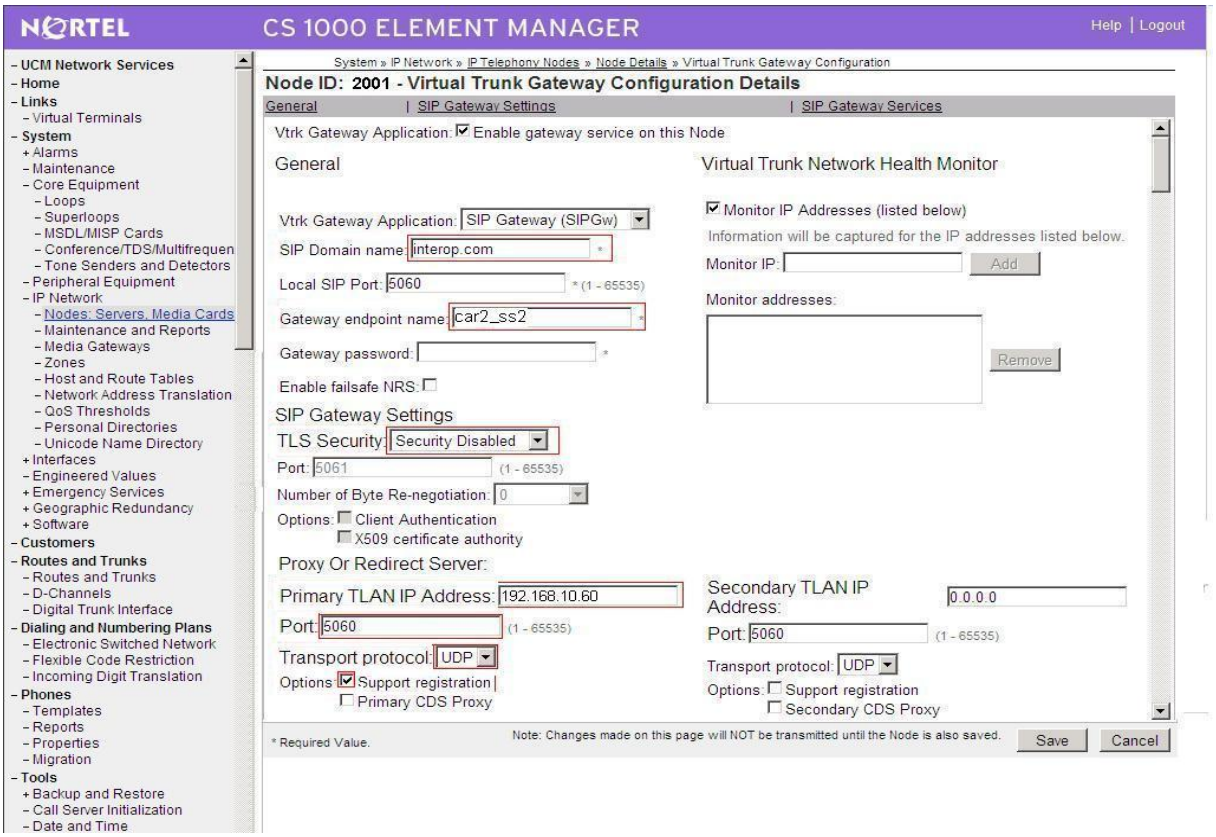

Figure 52 – Trunk Gateway Configuration Details

In the signaling server properties, the Line TPS will be enabled if this signaling 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

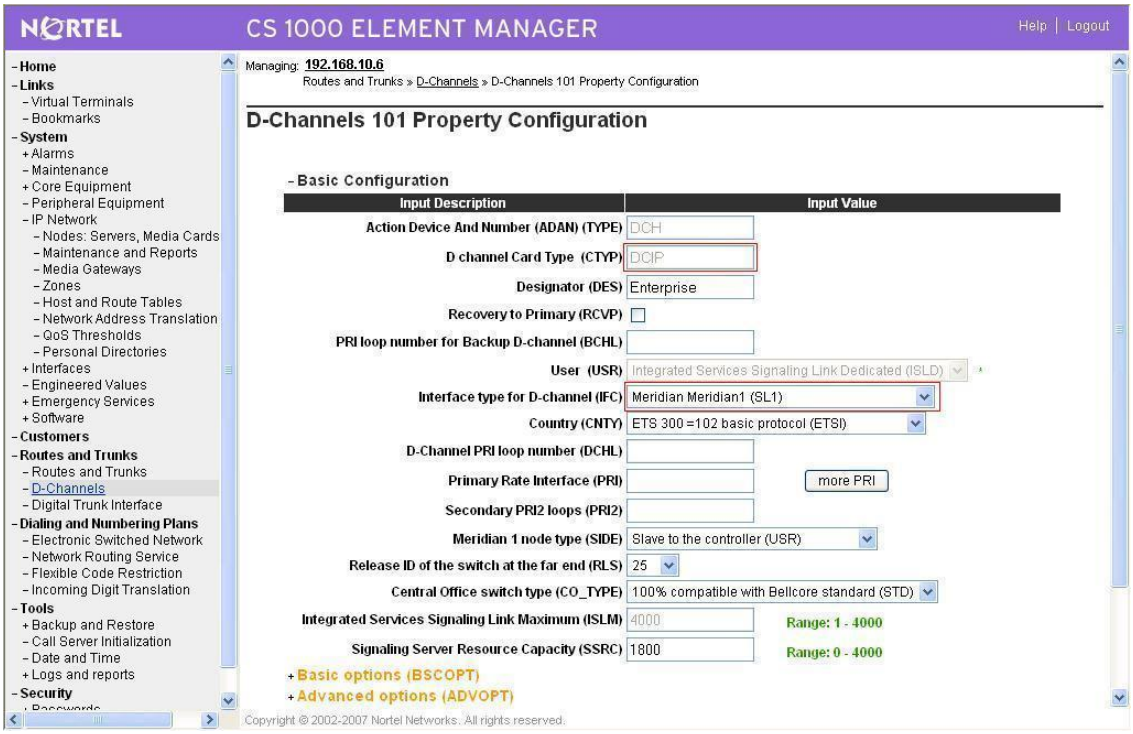

Figure 53 – D-Channel Property Configuration

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

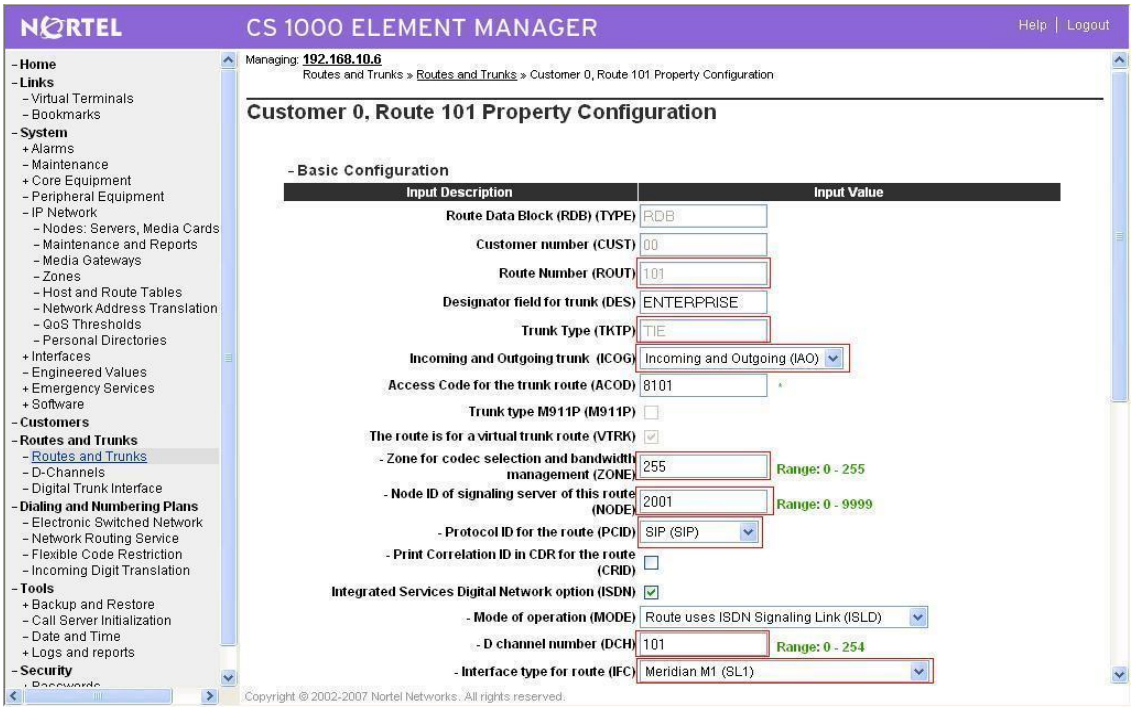

Figure 54 – Route Property Configuration

Configure Route 101 for SIP trunks, figure 55

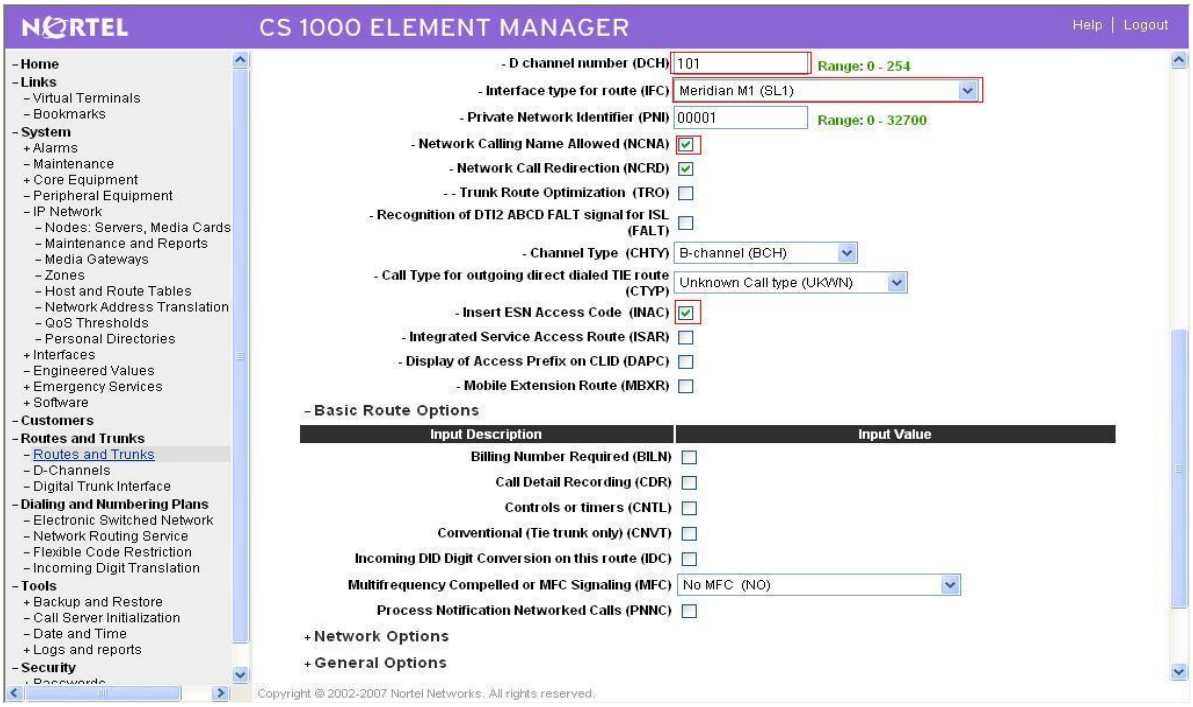

Figure 55 – Route Configuration

4. Create Trunk (figure 56)

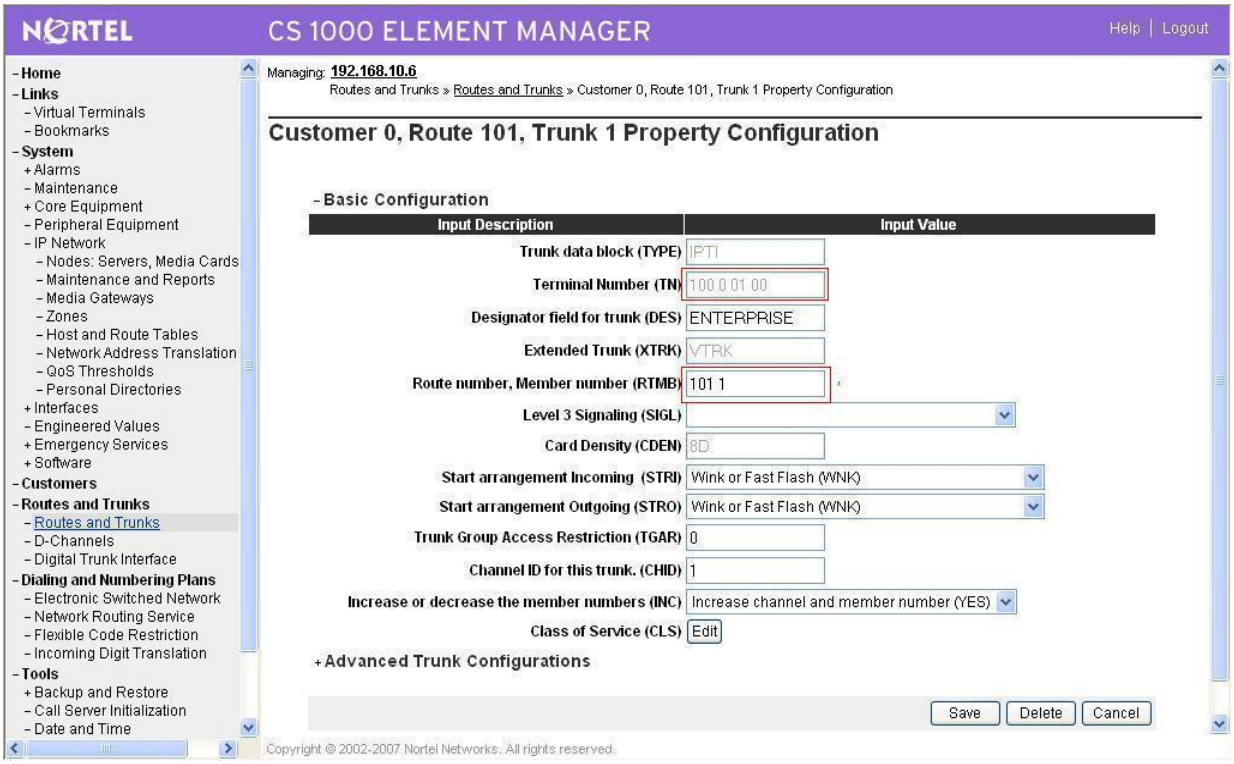

Figure 56 – Trunk Property Configuration

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

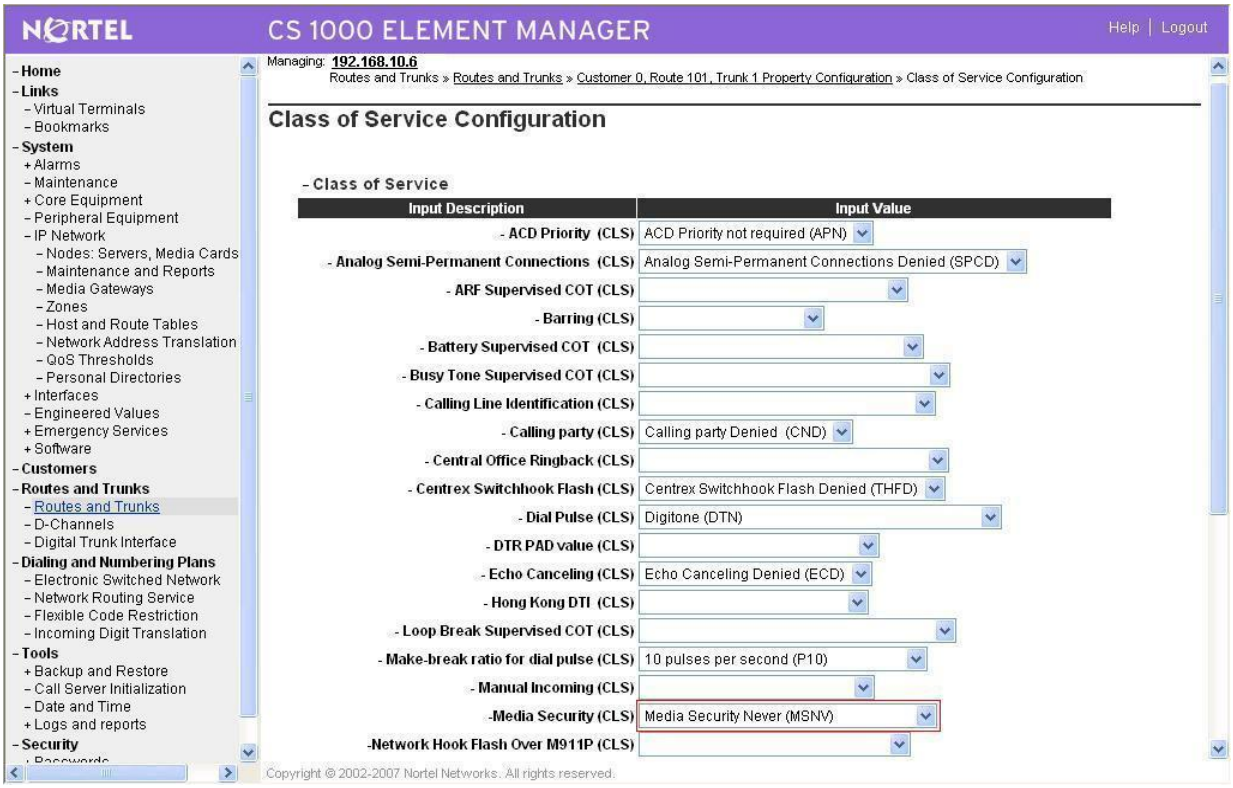

Figure 57 – Class of Service Configuration

5. Create Dialing Plan

Create Location Code

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

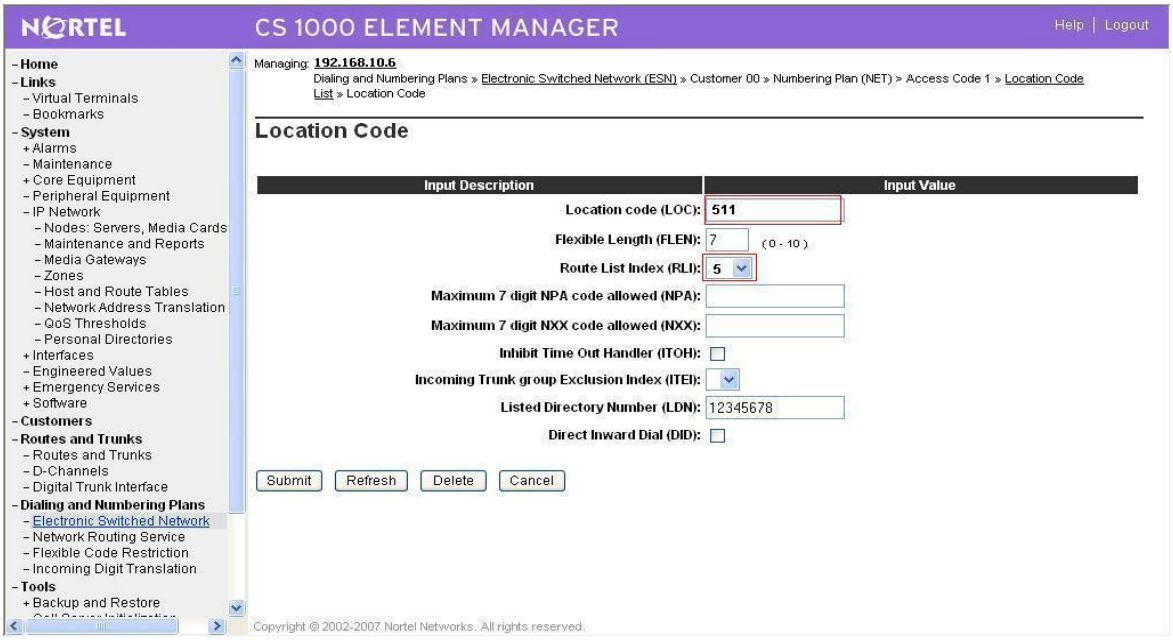

Figure 58 – LOC_511 for out going calls to CS1000E_A

Create Numbering Plan Area Code

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

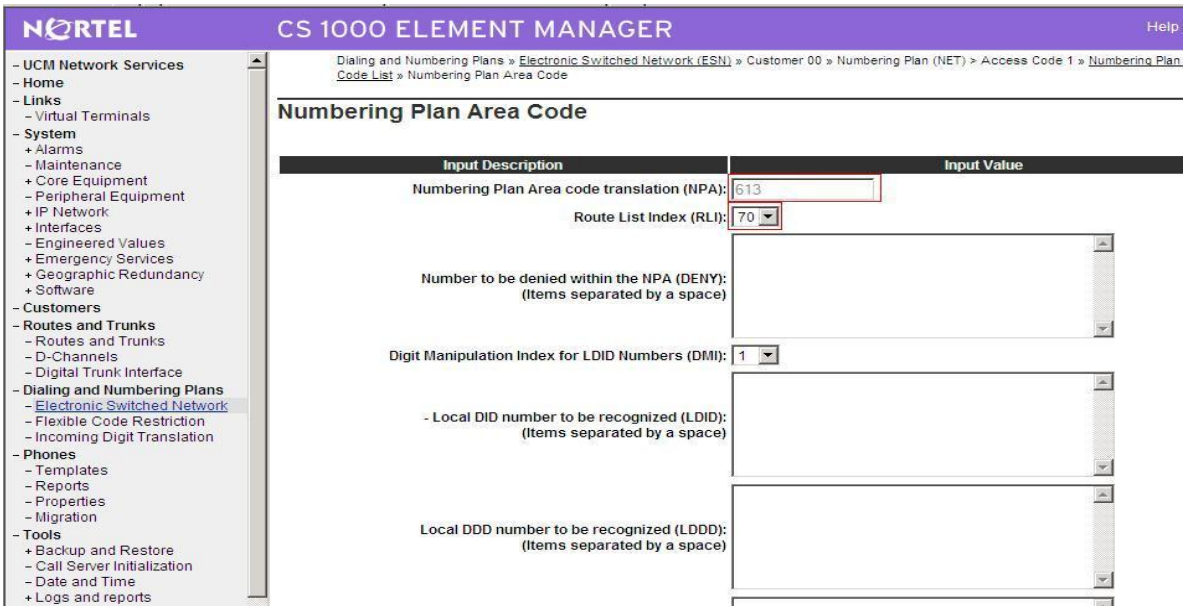

Figure 59 – NPA_613 for Outgoing calls to PSTN

Create Home Location Code

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

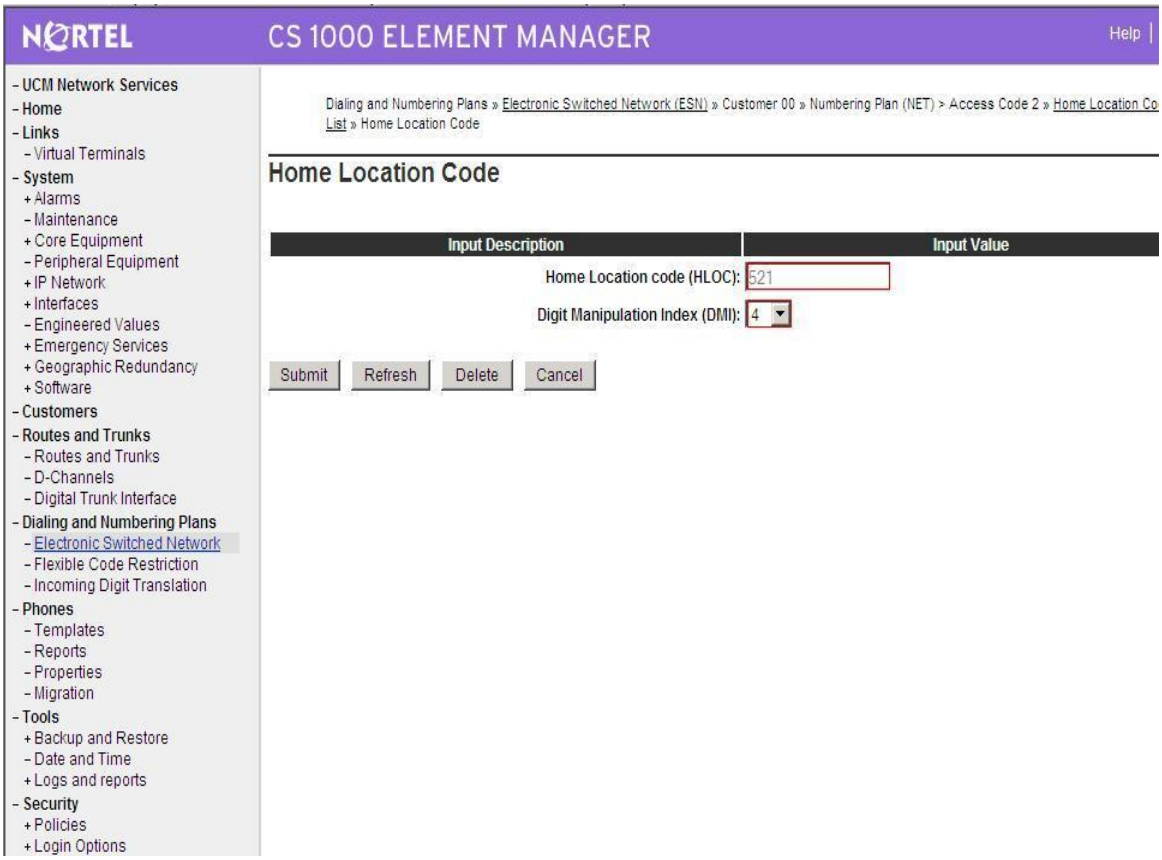

Figure 60 – Create HLOC 521 for incoming calls from CS1000E_A

Create Route List Block

Create RLI_5 for outgoing calls to CS1000E_A (Use DMI_0)

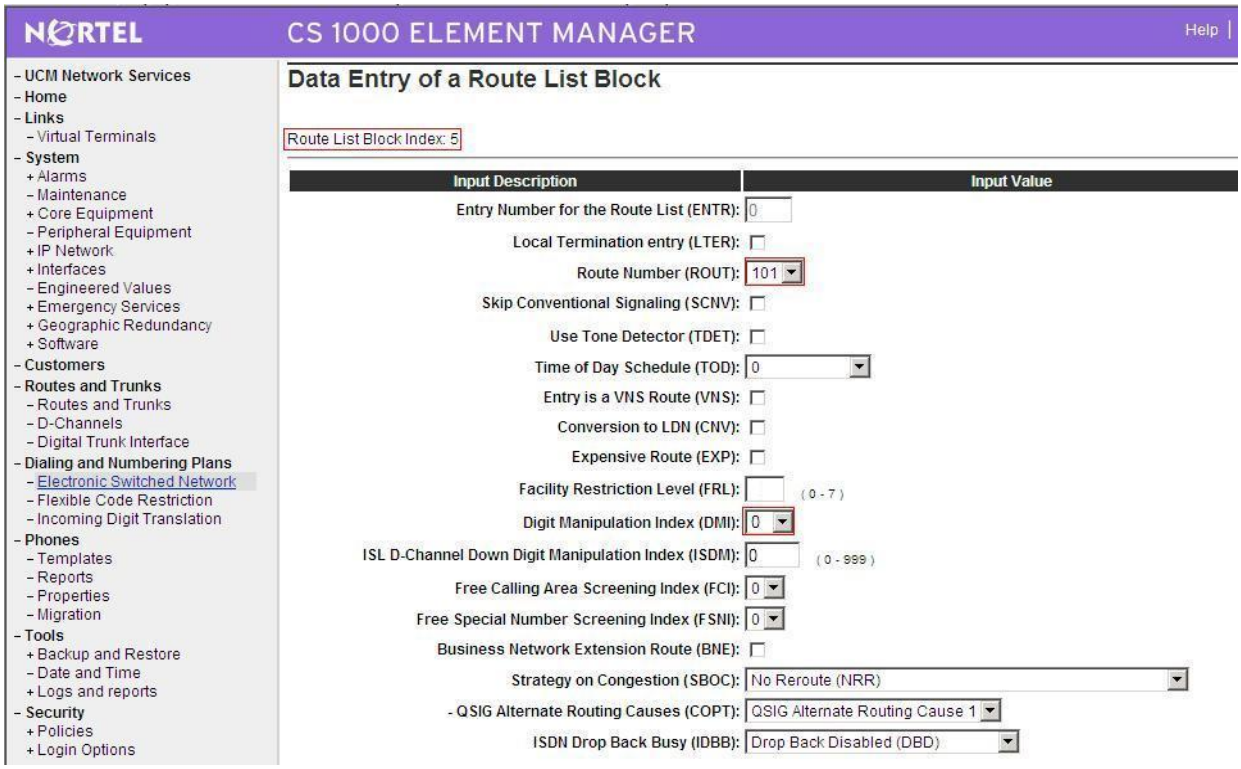

Figure 61 – RLI_5 (use DMI_0) for Outgoing calls to CS1000E_A

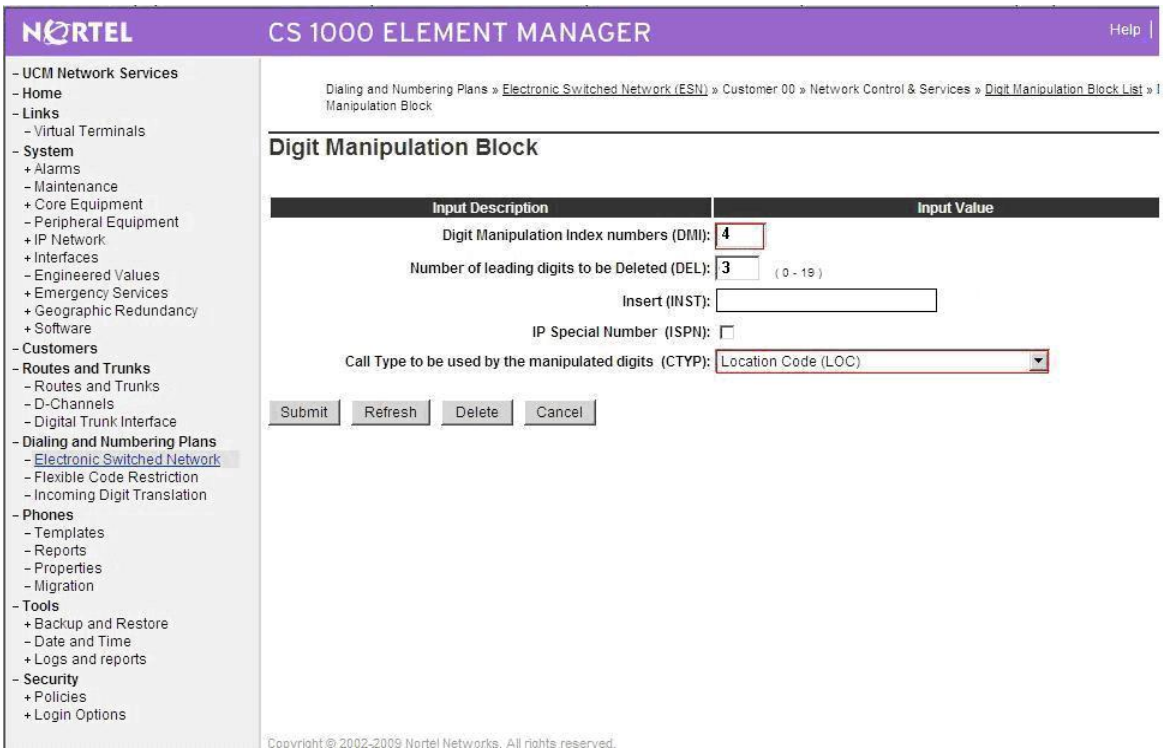

Figure 62 – DMI_4 for incoming call to this CS1000E_B

Configure SIP Proxy Server (SPS)

Create gateway endpoints on SPS

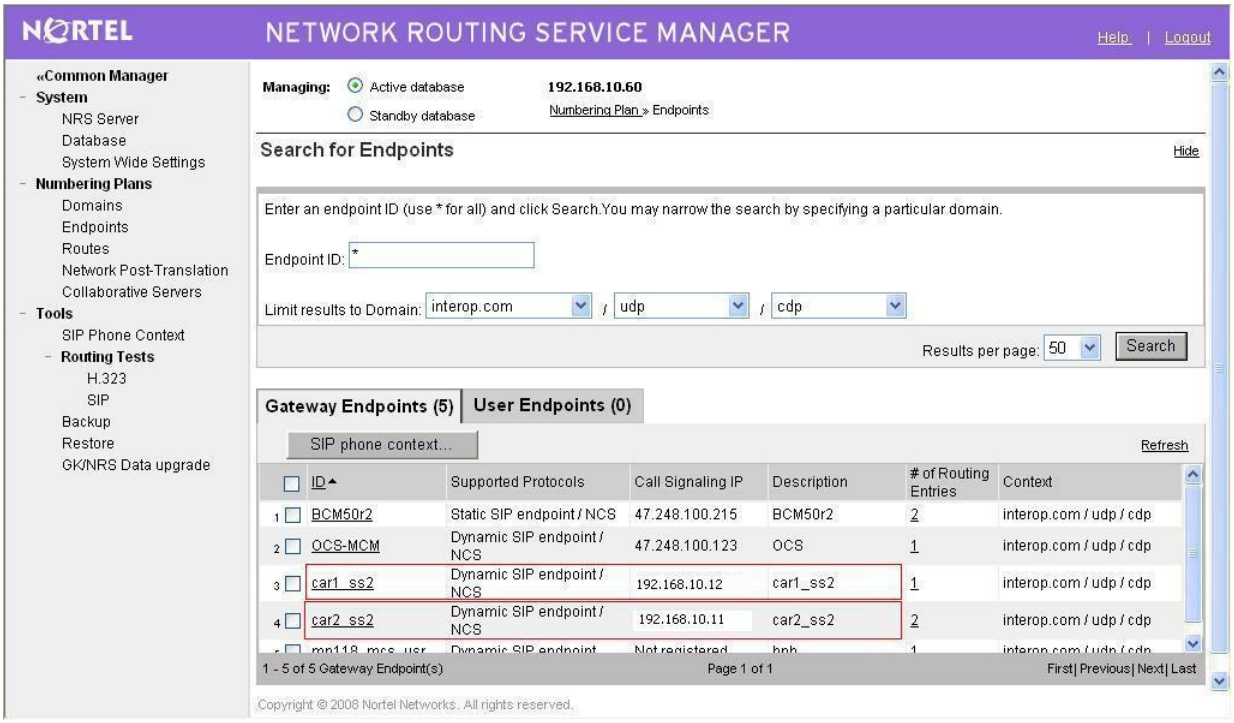

Figure 623 – SIP Gateway Endpoint Creation

Create routing entries for each of gateway endpoints on SPS, figure 64

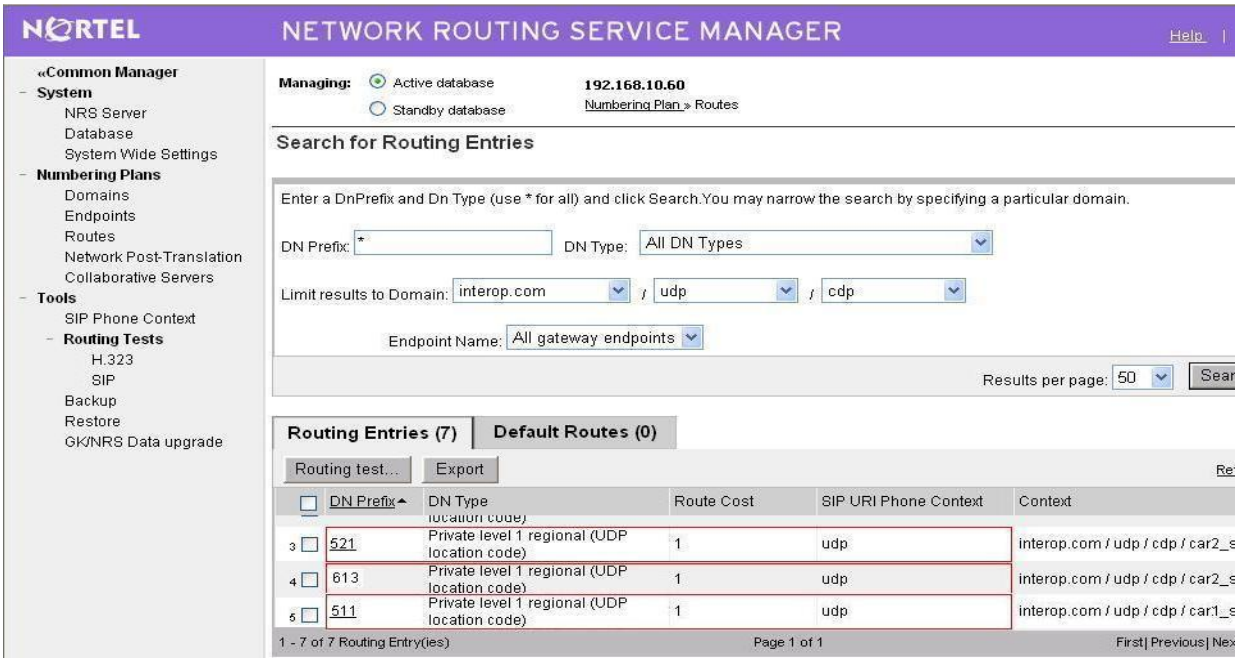

Figure 634 – Routing Entries for Gateway Endpoints

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.

Configure SIP LINE CS1000E in Element Manager

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

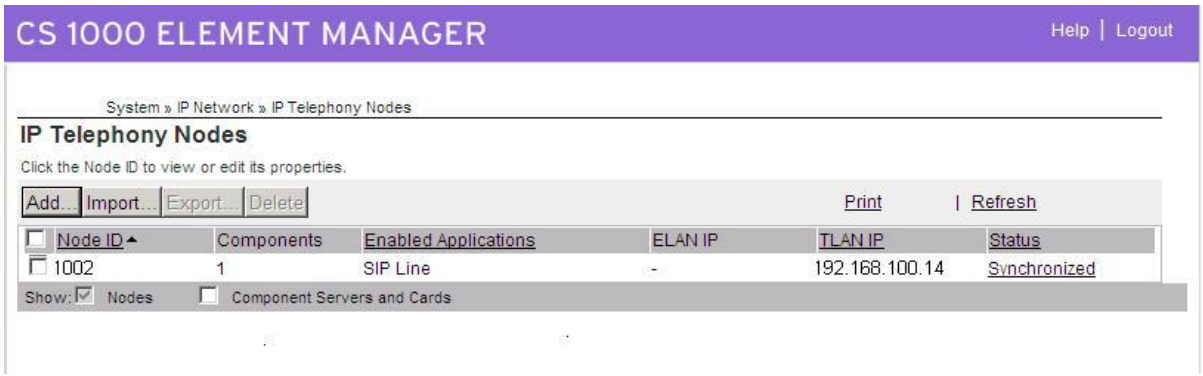

Figure 645 – IP Telephony Nodes

Figure 66, 67 and 68 show how to set up the SIP LINE Node 1002 configuration details

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

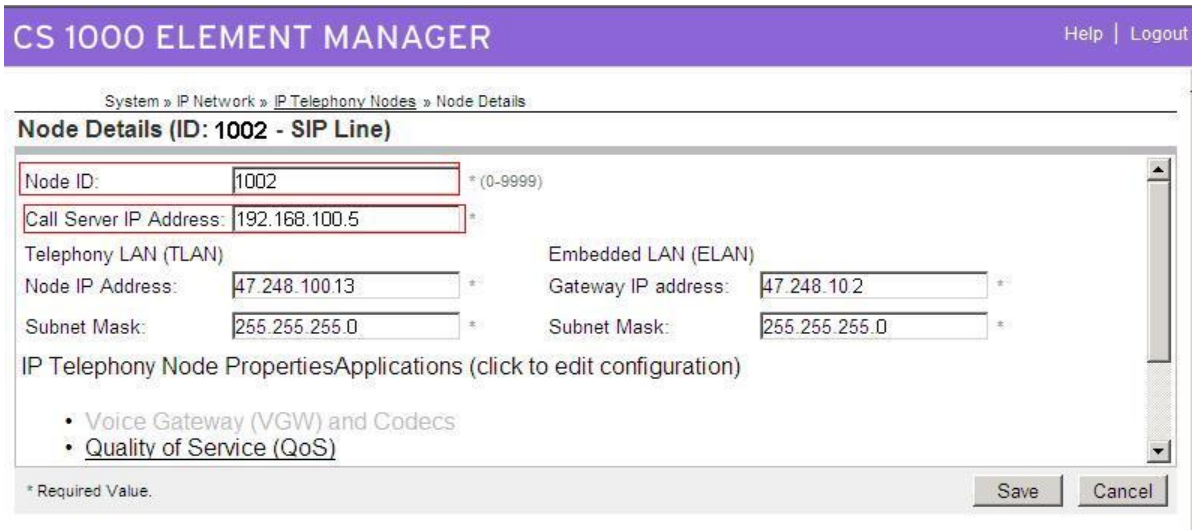

Figure 656 – Node Configuration Details

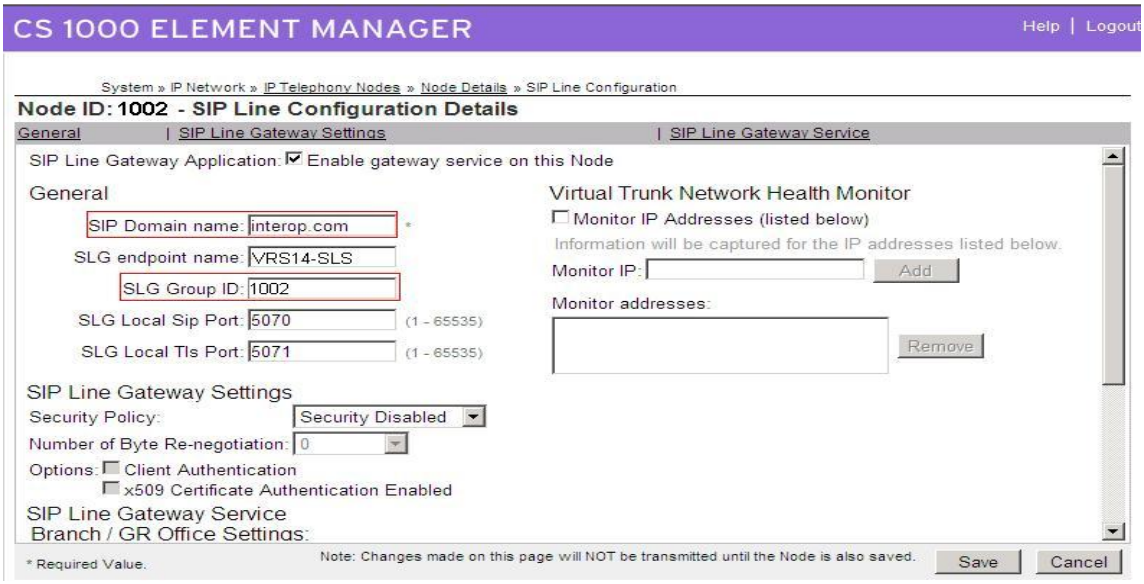

Figure 667 – Node Configuration Details (Cont…)

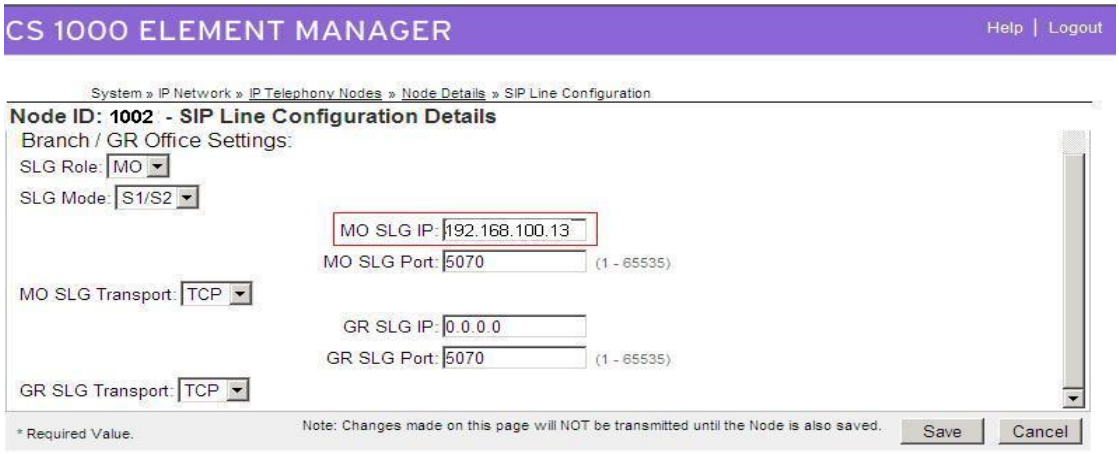

Figure 678 – Node Configuration Details (Cont…)

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 - Avaya SIP Line Package
4. SIP_LINE_3P_PKG – 416 - 3rdParty SIP Line Package

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

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

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

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

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

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**

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

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

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

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

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

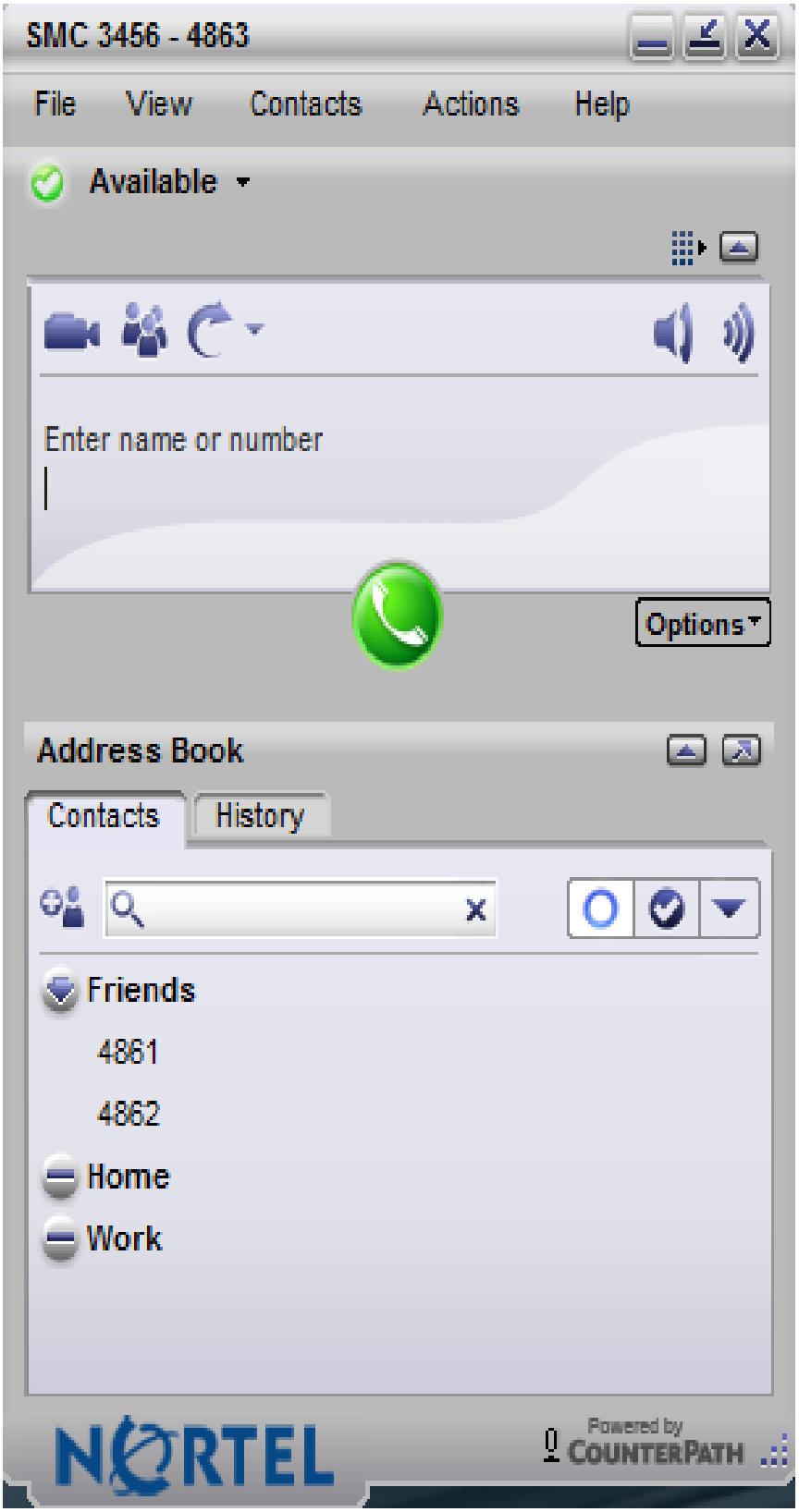

Figure 69 SMC Client

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

➔ No login server available

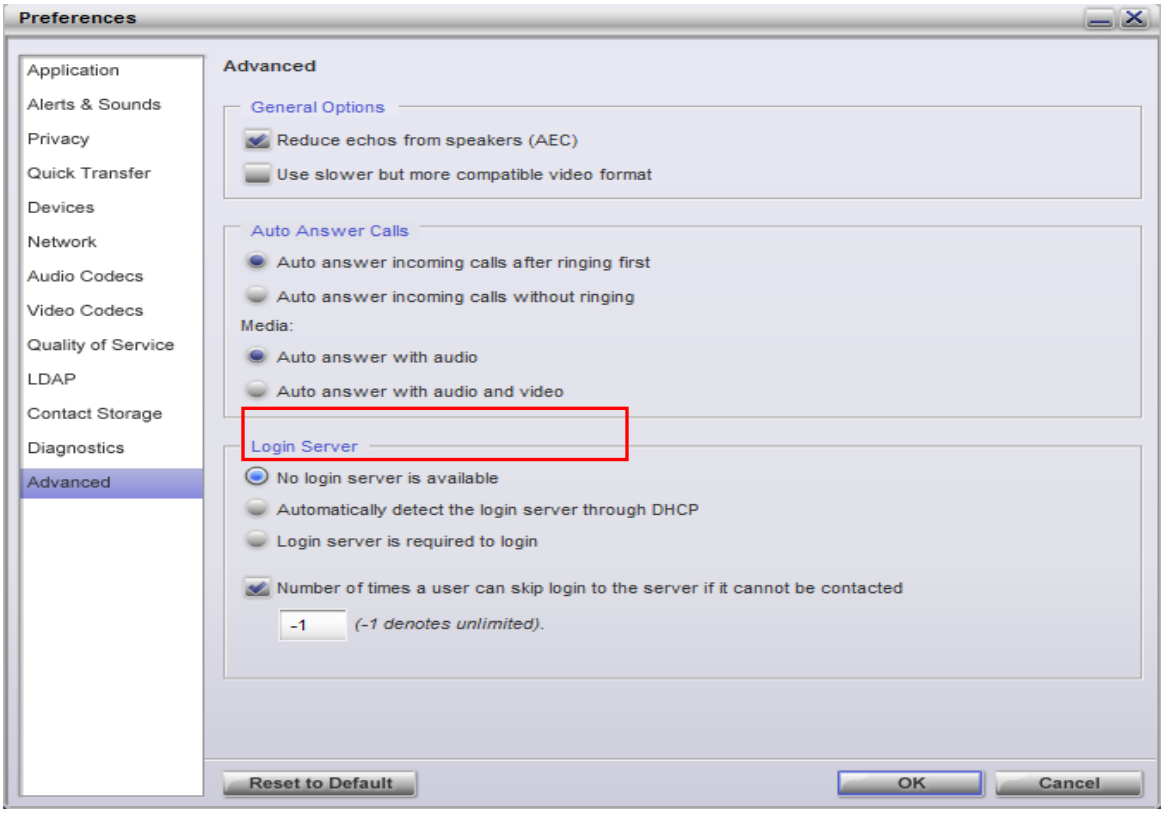

Figure 70 – Advanced Options Menu

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.

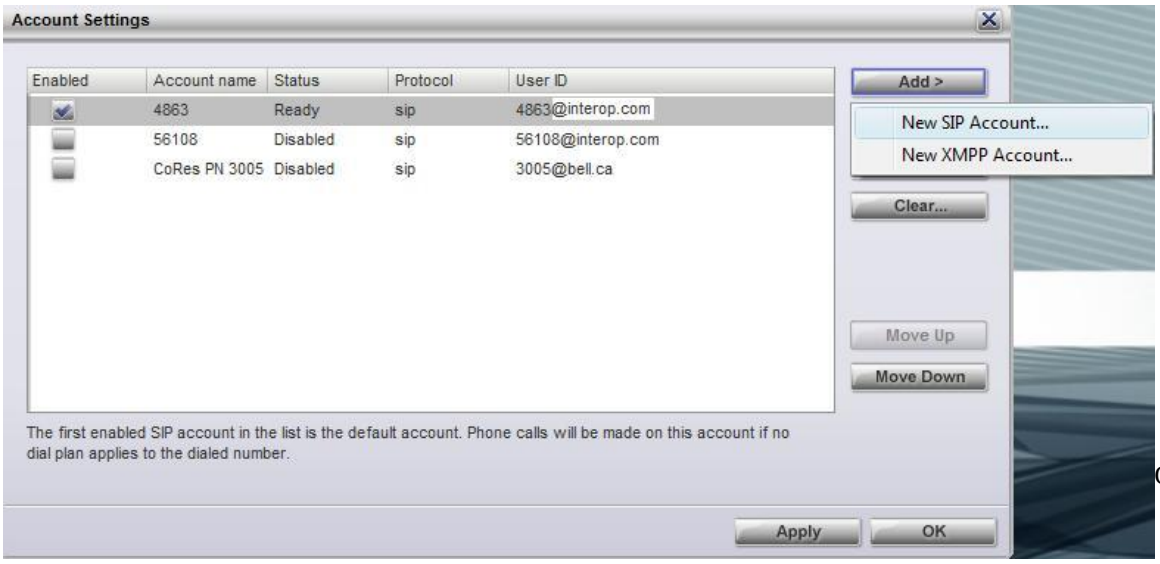

Figure 681 – Accounting Settings

The created account is appeared as figure 72.

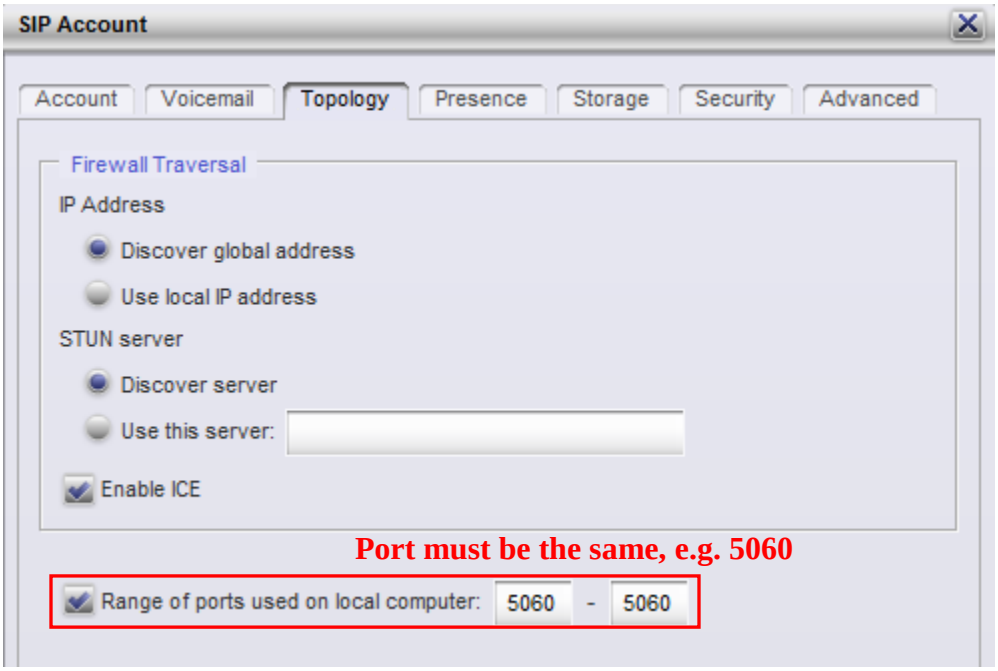

Figure 692 – Topology SIP Account Settings

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

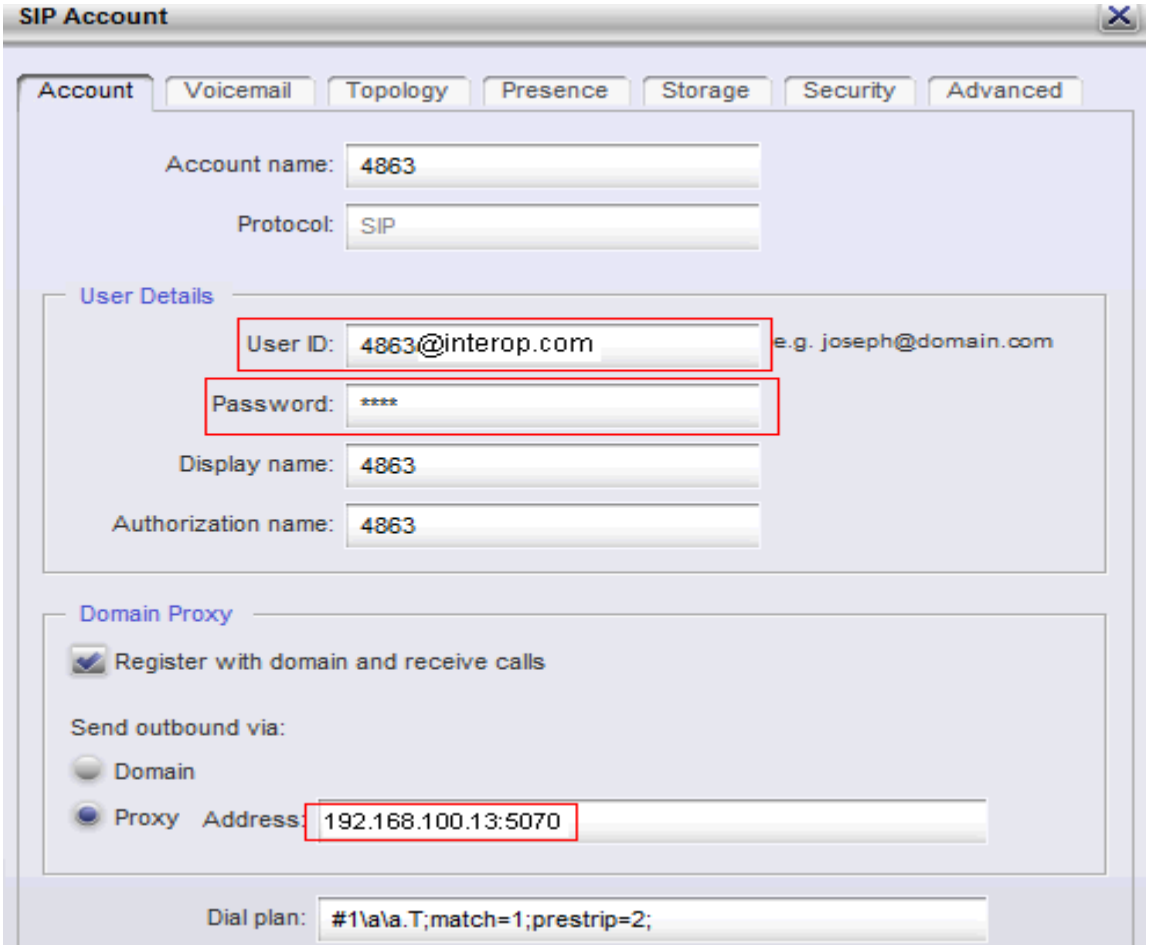

Figure 703 – SIP Account Details Setting

Figure 73 shows the newly created SIP account

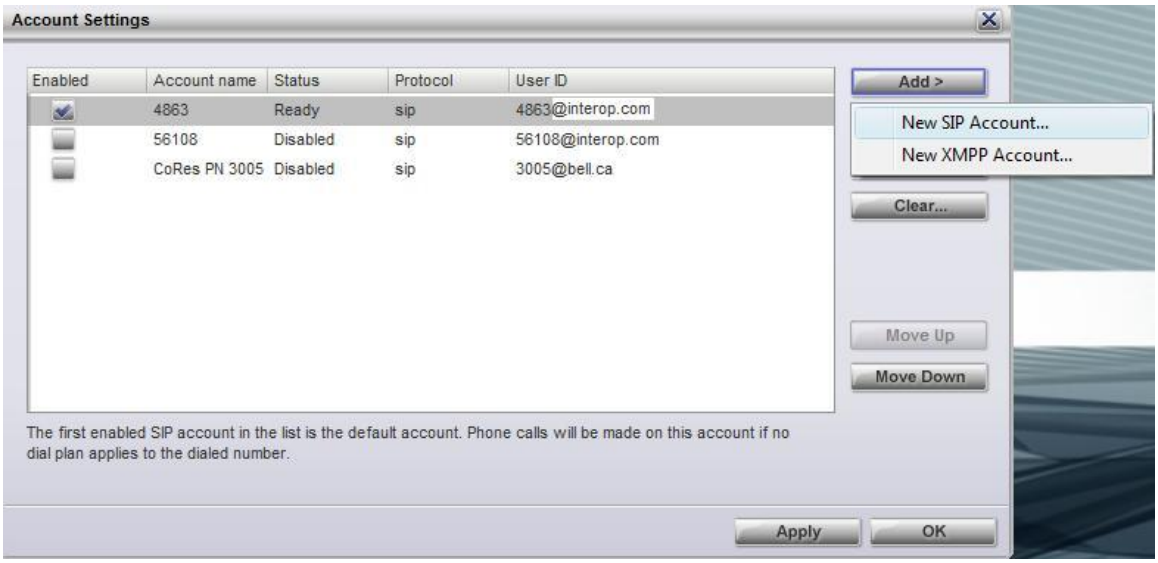

Figure 714 – Newly Created SIP Account

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
```

Frontier Communication System configuration

Frontier will have to provide this configuration notes.

General Test Approach and Test Results

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

General Test Approach

The general test approach was to have Frontier Genband system connected to CS1000E SIP Gateway using Genband IP address. The SIP trunk communication should be established between CS1000E and Frontier Genband system. Calls can be made from end to end, i.e. PSTN phone can call through created route from Frontier 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

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 (MSEC = off in LD 17) on both SIP trunk and phones on CS1K locally
- PSTN1 calls CS1000E_PHONE. Call is established between PSTN1 and CS10000E_PHONE. CS1000E_PHONE does blind transfer or consult transfer to PSTN2 but PSTN2 does not ring. CS1000E_PHONE hears greeting "Sorry your long distant service is not established. Please contact…." instead of ringback tone. In order to resolve this, the work around can be used by setting of DORG (Display Originator) parameter in CS1K call server to "No". However, this will cause a side effect on the CLID/CPND of PSTN phones. None of the CLID/CPND PSTN phones are displayed correct. This issue is due to the usage of proprietary MCDN feature of the CS1K to pass

the CLID information over the trunk, which does not understand by the Carrier Service Provider. Please refer to CR Q02141122.

- PSTN1 calls CS1000E_PHONE. Call is established between PSTN1 and CS1000E_PHONE. CS1000E_PHONE initiates a conference with PSTN2 but PSTN2 does not provide ringing tone. CS1000E hears greeting "Sorry your long distant service is not established. Please contact…." instead of ringback tone. Please refer to CR Q02135507. This issue can be resolved by configuration change in CS1K where DORG is set up to "No" in LD 86 as above.
- PSTN1 calls CS1000E_PHONE1. Call is established between PSTN1 and CS1000E_PHONE1 successfully. CS1000E_PHONE1 does blind transfer to CS1000E_PHONE2 which is set up to call forward no answer to PSTN2. The expectation is as follow: CS1000E_PHONE1 call is released, PSTN1 hears ring-back tone, CS1000E_PHONE2 rings for three times and then the call is forwarded to PSTN2. The actual result is as follow: CS1000E_PHONE2 is released after call is forwarded no answer to PSTN2. PSTN2 rings but PSTN1 does not hear ring back tone from PSTN2, and when PSTN2 answers the call, the call is dropped right away. This issue has been resolved by changing CS1K configuration where DORG is set up to "No" in LD 86. However, the CLID information was not displayed correctly on PSTN1.
- CS1000E_TDM calls CS1000E_IP_PHONE which is set up to call forward no answer to PSTN. User expects to hear PSTN phone ringing after CS1000E_IP_PHONE rings three times. Then the call should be forwarded to PSNT phone and the CS1000E_IP_PHONE is released so call is established between CS1000E_TDM and PSTN. However, CS1000E_TDM did hear the 3 ring-back tones, a silence period, then ring back tones and so on…. Please refer to CR Q02138017. Call will work if Media security (MSEC) is disabled for both SIP trunk and phones on CS1K locally. This CR is closed "No fix planned" with recommend MSEC needs to be set to "Off" in LD 17.

- PSTN1 calls to CS1000E_PHONE. Call is established successfully when CS1000E_PHONE answers. CS1000E_PHONE does blind transfer or consult transfer to PSTN2. CLID/CPND of PSTN1 and PSTN2 display CS1000E_PHONE instead of PSTN2 number is displayed on PSTN1 and PSTN1 number is displayed on PSTN2. Please refer CR Q02141122.

- CLID/CPND of PSTN2 displays incorrectly with the following scenarios:
  PSTN1-----call----CS1000_TDM----CFNA----PSTN2 displays "1771640-3" instead of PSTN1 number
  PSTN1 -----call-----CS1000_IP_PHONE2-----CFAC-----PSTN2 displays "1771640-1" instead of PSTN1 number
  PSTN1----call-----CS1000_IP_PHONE2------CFB-----PSTN2 displays "1771640-3" instead of PSTN1 number

  Please refer to CR Q02138314 as same root cause as Q02138314.

- FAX G711 Modem Pass Through Mode is not supported on Frontier network. Test cases from 5.4.3.1 to 5.4.3.4 were not tested.

- SIPLINE call failed on SU17. Issue on CS1K SIP line 6.0.Please refer to CR Q02129692

PSTN calls CS1000E_SIPLINE. There are two actual results when SIPLINE answers call as below:

1) SIPLINE rings and PSTN phone gets ringback tone. If SIPLINE answers the call, SIPLINE will display "Remote hold" without speech path between PSTN and SIPLINE.
2) Call is established with two ways speech path between SIPLINE and PSTN. However, SIPLINE is not released when PSTN ends call.
Addition, Simple call is failed between SIPLINE1 and SIPLINE2. Call is established between SIPLINE1 and SIPLINE2 successfully. However, SIPLINE2 is not released when SIPLINE1 ends call.
Simple call also failed between CS1000E_Unistim calls CS1000E_SIPLINE. CS1000E_Unistim calls CS1000E_SIPLINE. Call is dropped right after CS1000E_SIPLINE answers call.

Verification Steps

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

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.

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

- Verify basic call between PSTN phones and Avaya phones. During the call,at the CS1000E SIP Gateway, use pcap tool (ethereal/wireshark) to make sure that all SIP request/response messages are received properly.
- Verify Codec and 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
SIGNALING 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 SIGNALING 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

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 Frontier and Avaya design teams. Some of these issues are considered as exceptions. The Frontier Communication System is considered compliant with Communication Server 1000E release 6.0.

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*

Appendixes

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

ld 143

CCBR000

.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

6 p28774_1 Yes 17/03/10 NO FRU nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386

7 p28797_1 Yes 17/03/10 NO FRU nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
20 p22968_1 Yes 24/03/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch
22 p28415_1 Yes 24/03/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch
23 p25946_1 Yes 24/03/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch
24 p25529_1 Yes 26/03/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch
25 p27408_1 Yes 26/03/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 18

PATCH# IN_SERVICE DATE SPECINS REMOVABLE NAME
0 Yes 17/03/10 YES YES nortel-cs1000-linuxbase-6.00.18.63-02.i386.000
2 Yes 17/03/10 NO YES nortel-cs1000-dmWeb-6.00.18.62-00.i386.001
3 Yes 17/03/10 NO YES nortel-cs1000-auth-6.00.18.62-00.i386.000
4 Yes 17/03/10 NO YES nortel-cs1000-ISECSH-6.00.18.62-00.i386.000
5 Yes 17/03/10 YES YES nortel-cs1000-patchWeb-6.00.18.63-01.i386.000
8 Yes 17/03/10 NO YES nortel-cs1000-gk-6.00.18.63-00.i386.000
9 Yes 17/03/10 NO YES nortel-cs1000-sps-6.00.18. 63-00.i386.000
10 Yes 17/03/10 NO YES nortel-cs1000-tps-6.00.18. 63-00.i386.000
11 No 17/03/10 NO YES nortel-cs1000-bcc_6-0-6.00.18.63-01.i386.000
12 No 17/03/10 NO YES nortel-cs1000-cs1000WebService_6-0-6.00.18.63-01.i386.000
13 Yes 17/03/10 NO YES nortel-cs1000-shared-general-6.00.18.62-00.i386.000
14 Yes 17/03/10 NO YES nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
15 No 17/03/10 NO YES nortel-cs1000-emWeb_6-0-06.00.18.63-01.i386.001
16 Yes 17/03/10 NO YES nortel-cs1000-pd-6.00.18.6 2-00.i386.000
17 Yes 16/04/10 NO YES nortel-cs1000-vtrk-6.00.18.65-017.i386.000
18 Yes 17/03/10 NO YES nortel-cs1000-csv-6.00.18. 62-00.i386.000
19 No 17/03/10 NO YES nortel-cs1000-csmWeb-6.00. 18.62-00.i386.001
21 Yes 17/03/10 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

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.