Avaya Solution & Interoperability Test Lab

Application notes for G-Tek SIP Telephone GE-202 Version 1820x.27.1.01 with Avaya™ Communication Server 1000 SIP Line Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya™ Communication Server 1000 SIP Line Release 6.0 and the G-Tek SIP phones GE-202 firmware version 1820x.27.1.01. During the compliant testing, the GE-202 was able to register, as a SIP Client endpoint, with the Communication Server 1000. The GE-202 was able to place and receive calls from the Communication Server 1000 Release 6.0 non-SIP and SIP Line clients. The tests of other telephony features were executed such as transfer, DTMF relay, bridge line appearance, dual call, etc.

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

1. Introduction

These application notes provide detail configurations of Avaya Communication Server 1000 SIP Line rel. 6.0 (hereafter referred to as CS1000) and SIP phone GE-202 rel. 1820X.27.1.01 (hereafter referred to as GE-202) during the compliance testing session. The GE-202 was tested against the non-SIP and SIP clients of the CS1000 SIP line 6.0. All the applicable telephony feature test cases of release 6.0 SIP line were executed on the GE-202, where applicable, to ensure the interoperability with CS1000.

1.1. Interoperability Compliance Testing

The focus of this compliance testing is to verify that the GE-202 is able to interoperate with our CS1000 SIP line system. The following interoperability areas were covered:

- Registration of GE-202 to the CS1000.
- Calls establishment of GE-202 with Avaya SIP and non-SIP phones on the CS1000.
- Calls establishment with emulated PSTN phones
- Telephony features: DTMF transmission, voicemail with MWI notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer.
- Specific hospitality feature requirements; bridge line appearance (BLA), dual call arrangement, multiple line appearance and multiple appearances Directory Number.
- Codec negotiation

1.2. Support

For technical support on G-Tek SIP telephones, please contact G-Tek technical support at:

- Telephone: +886-2-26962665 ext. 221
- E-mail: support@G-Tek.com.tw

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the Avaya CS1000 and the GE202.

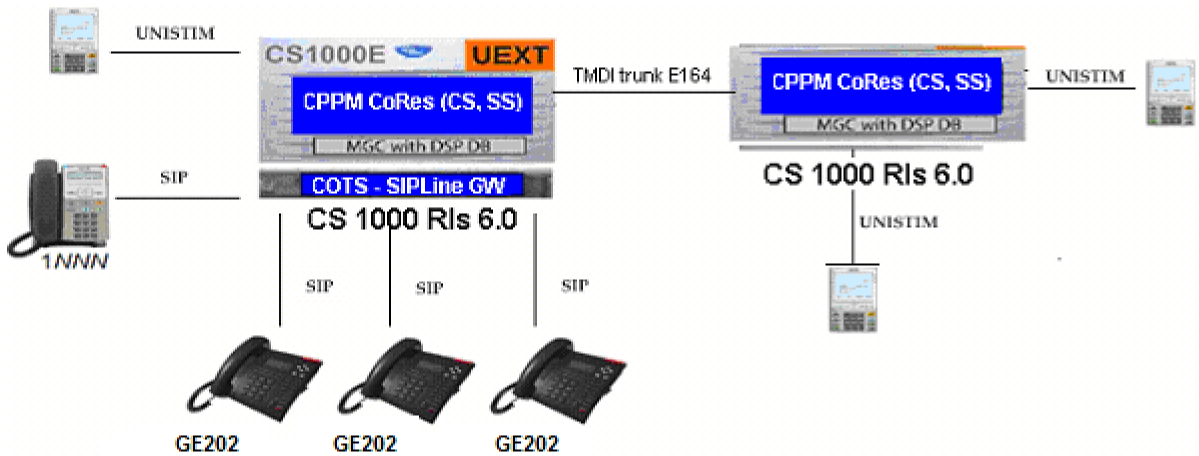

Figure 1: Lab Diagram

3. Equipment and Software Validated

System	Software/Loadware Version
CS1000	• Call Server (CPPM): 6.00RJ • Signaling Server (CPPM): 6.00.18 • SIP Line Gateway (HP DL320)
Call Pilot	• CallPilot (600r): 05.00.41.29
11xx SIP client (Sigma) SIP soft-phones IP phones	• 02.02.16.00 • SMC3456: v2.6 Build 53715 • 2050PC: 3.02.0045
GE202	• 1820X.27.1.01

4. Configure the Avaya CS1000 - SIP LINE

This section describes the steps to configure SIP Line using CS1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on CS1000 system. For detailed information, see **[1]**

4.1. Prerequisite

- CS1000 system has been upgraded to Release 6.0. For more information, see **[6]**
- A server which has been
    - Installed with CS1000 Release 6.0 Linux Base.
    - Joined CS1000 Release 6.0 Security Domain.
    - Deployed with SIP Line Application.

  For more information, see **[7].**
- The following packages are enabled in the keycode.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

4.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

- Using IE to launch CS1000 UCM web portal at http://<IP Address or FQDN> where <IP address or FQDN> is the UCM Framework IP address or FQDN for UCM server.
- Login with the username/password which was defined during the primary security server configuration. For more information, see **[8]**

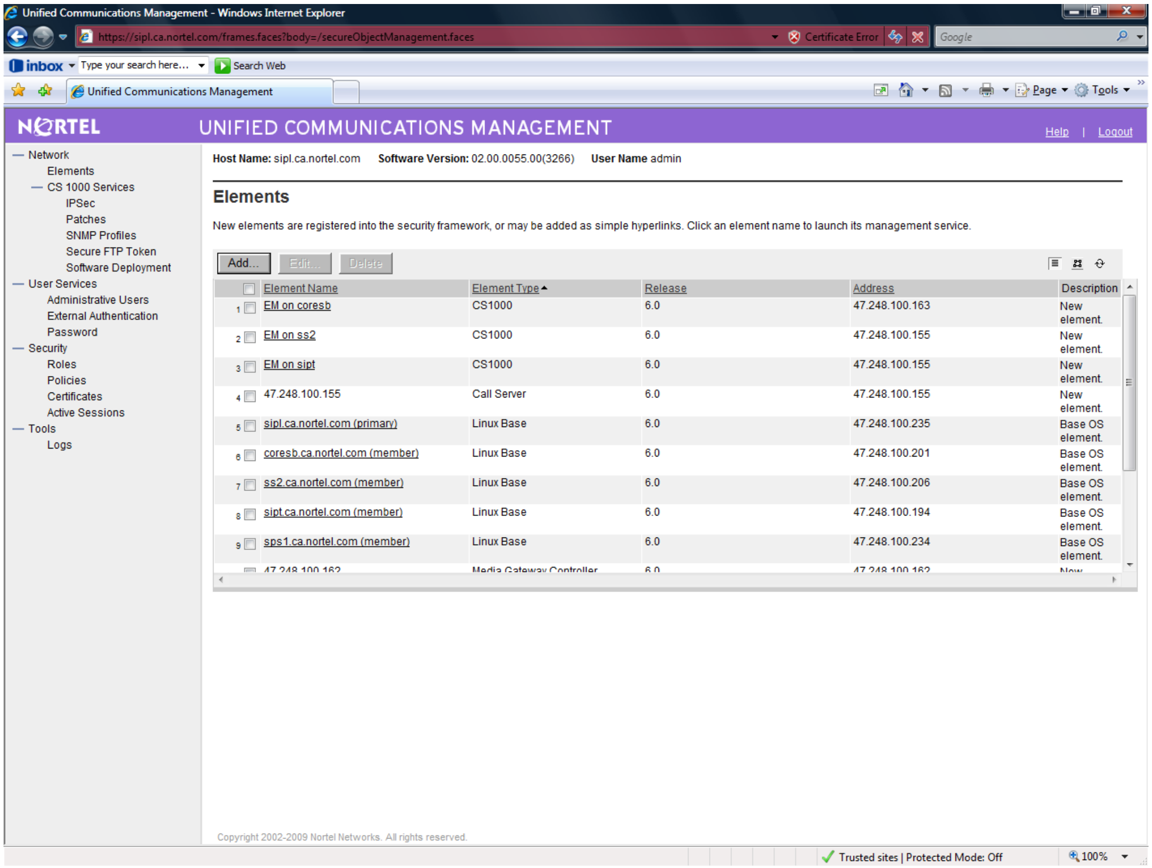

Figure 2: UCM Home Page

- On the Elements page of Unified Communications Management, under the Element Name column, click the server name to navigate to Element Manager for that server. The CS1000 Element Manager page appears as in figure 3 below.

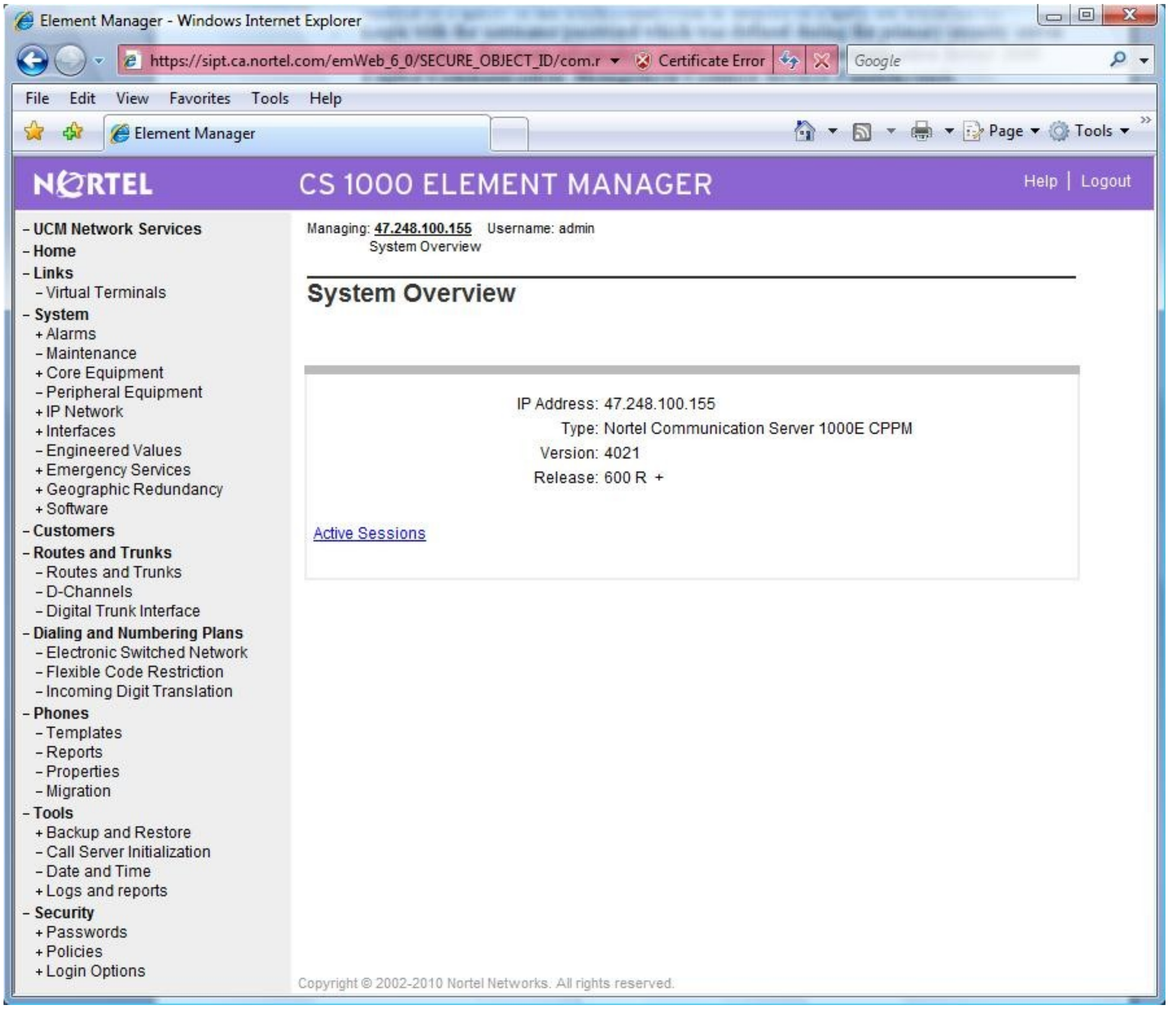

Figure 3: CS1000 EM Home Page

4.3. Enable SIP Line Service and Configure the Root Domain in Customer Data Block (CDB)

- On the EM page, navigate to ***Customers***, select the customer number to be enabled with SIP Line Service.
- Enable SIP Line Service by clicking on the ***SIP Line Service*** check box.
- Enter the SIP Line **Root Domain** name in the ***Root Domain*** text box.

Figure 4: SIP Line Service in Customers Data Block

4.4. SIP Line Telephony Node Configuration

- On the EM page, navigate to ***System* → *IP Network* → *Nodes: Servers, Media Cards.***
- Click ***Add*** to add a new SIP Line Node to IP Telephony Nodes. To see the SIP Line node details, click on the SIP Line Node ID.
- Enter Node ID in the ***Node ID*** text box.
- Enter Call Server IP Address in the ***Call Server IP Address*** text box.
- Enter Node IP Address in the ***Node IP Address*** text box.
- Enter TLAN Subnet Mask in the ***Subnet Mask*** text box.
- Enter ELAN Gateway IP Address in the ***Gateway IP Address*** text box.
- Enter ELAN Subnet Mask in the ***Subnet Mask*** text box.
- Check **SIP Line** check box to enable SIP Line for this Node.

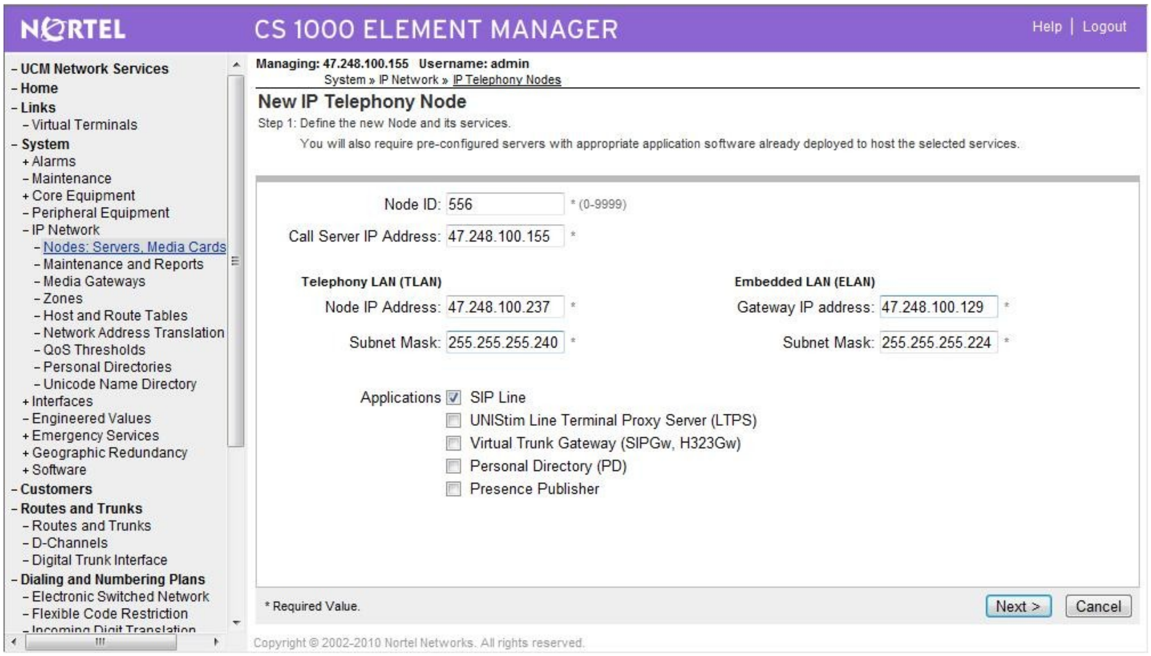

Figure 5 – IP Telephony Node

- Click ***Next.*** The page to add server to node appears (not shown).
- On Add Server page, from the ***Please Select Server*** list, select the server to add to the node.
- Click ***Add*** (Do not click the Next button).
- Select the check box next to the newly added server, and click ***Make Leader***.
- Figure 6 shows server “sipl” selected

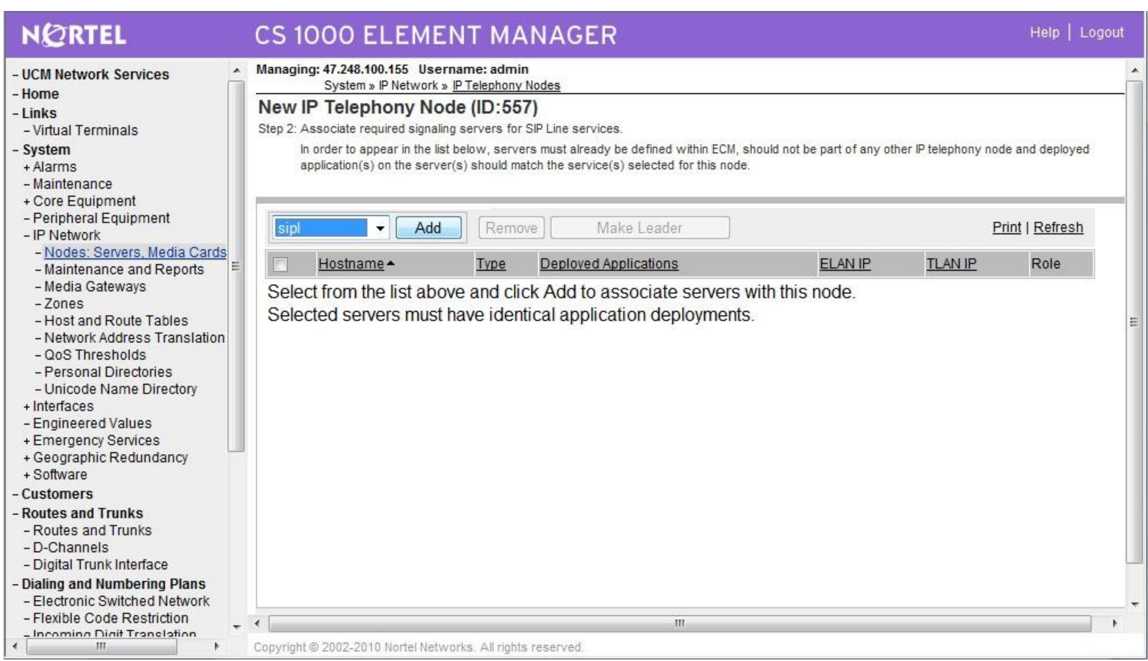

Figure 6 – IP Telephony Node – Add Server

- Click ***Next.*** The SIP Line Configuration Detail page appears.
- Enter SIP Line domain name in ***SIP Domain name*** text box. This must be the same as the domain name configured in ***Customers***, section 4.3.

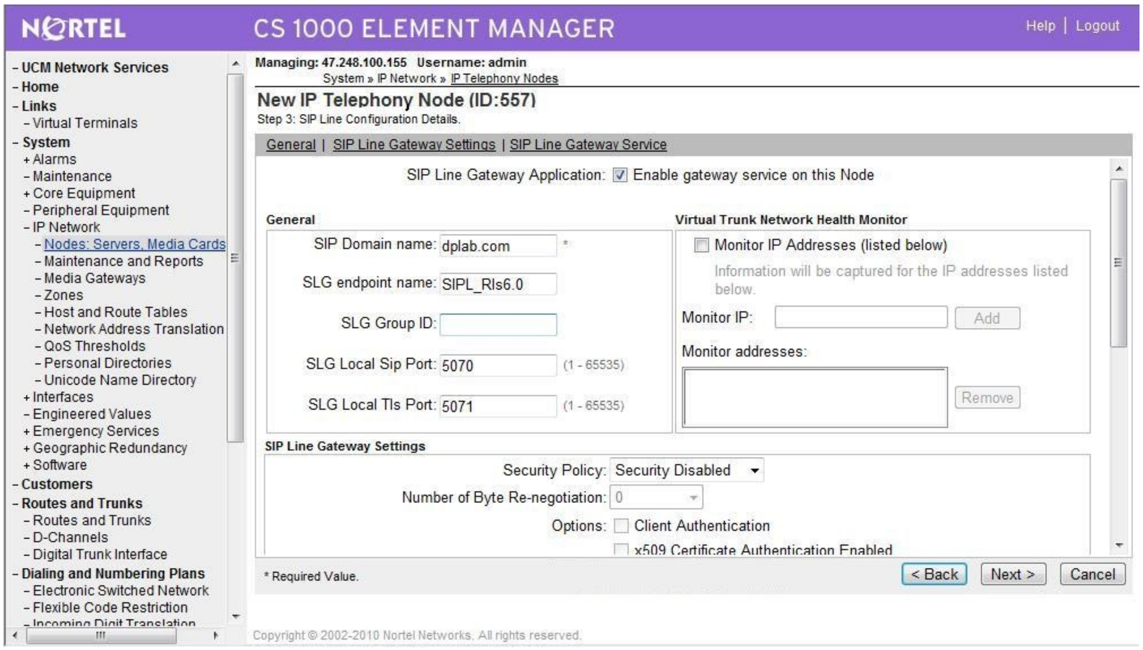

Figure 7 – SIP Line Node Details

- Under the ***SIP Line Gateway Services*** section, select ***MO*** from the ***SLG Role*** list.
- From the ***SLG Mode*** list, select ***S1/S2*** (SIP Proxy Server 1 and Server 2).

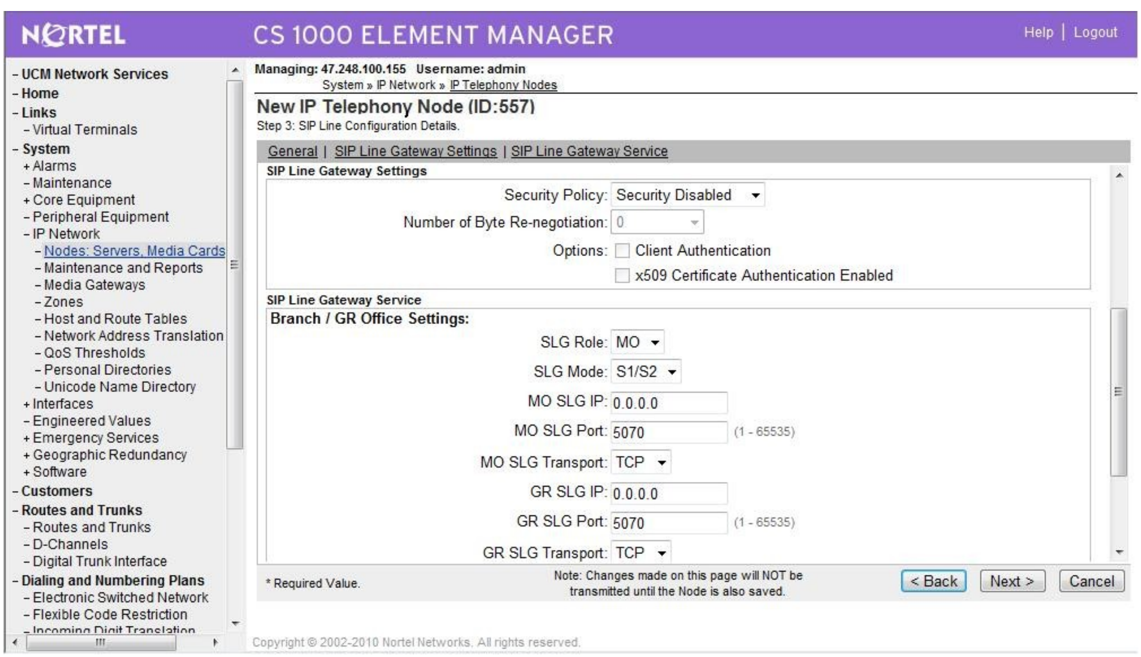

Figure 8 – SIP Line Node Details (cont.)

- Click ***Next***. The ***Confirm new Node details*** page appears.
- Click ***Finish*** and wait for the configuration to be saved. The ***Node Saved*** page then appears.

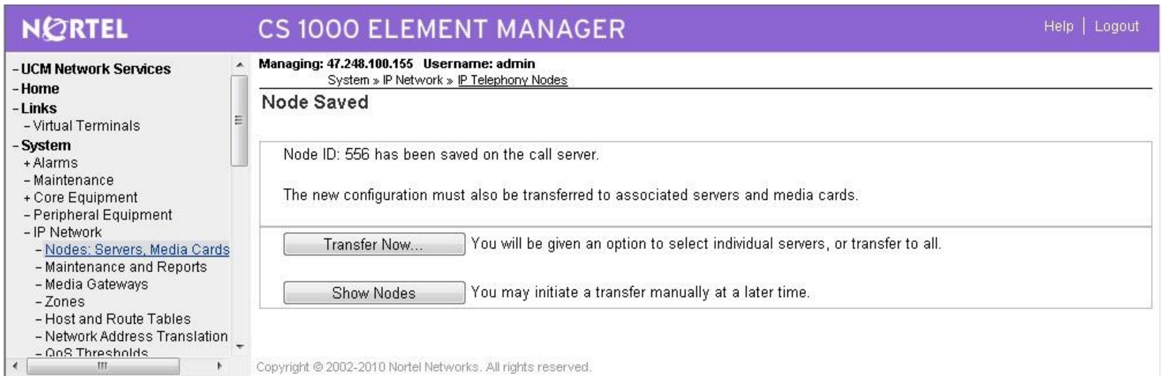

Figure 9 – Transfer Configuration

- Click ***Transfer Now.*** The ***Synchronize Configuration Files (Node ID 556) page*** appears.
- Select some or all of the node elements and then click ***Start Sync*** to transfer the configuration files to the selected servers.

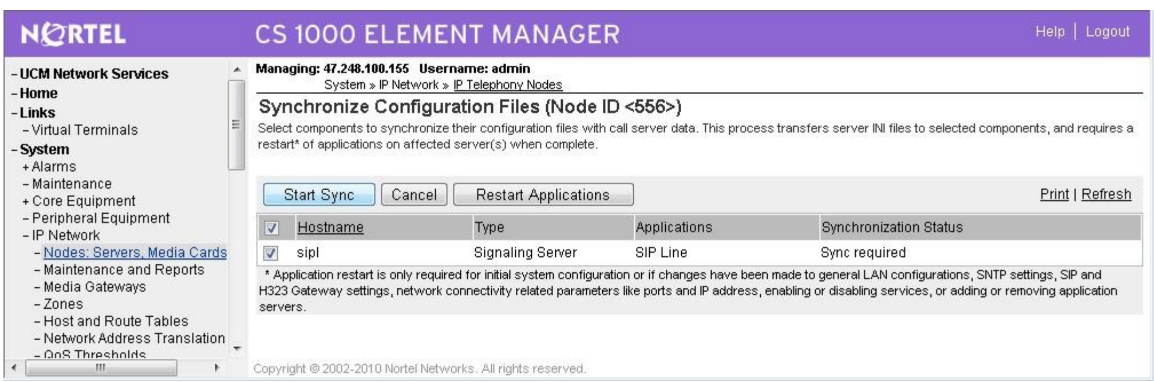

Figure 10 – Synchronize Configuration Files

4.5. D-Channel over IP Configuration

- On the EM page, navigate to ***Routes and Trunks → D-Channels.***
- Under the ***Configuration*** section, from the ***Choose a D-Channel Number*** list, select a D-Channel number (not shown).
- Under the Configuration section, from the ***Type*** list, select ***DCH.***
- Click to ***Add.***
- From the ***D channel Card Type (CTYP)*** list, select ***D-Channels is over IP (DCIP).***
- Click to-Add.
- The ***D-Channels xx Property Configuration*** page appears, as shown in Figure 11.
- From the ***Interface type for D-channel (IFC)*** list, select ***Meridian Meridian1 (SL1)***.

- Click the ***Basic options (BSCOPT)*** link. The ***Basic options (BSCOPT)*** list expands.
- Click ***Edit*** to configure ***Remote Capabilities (RCAP).***

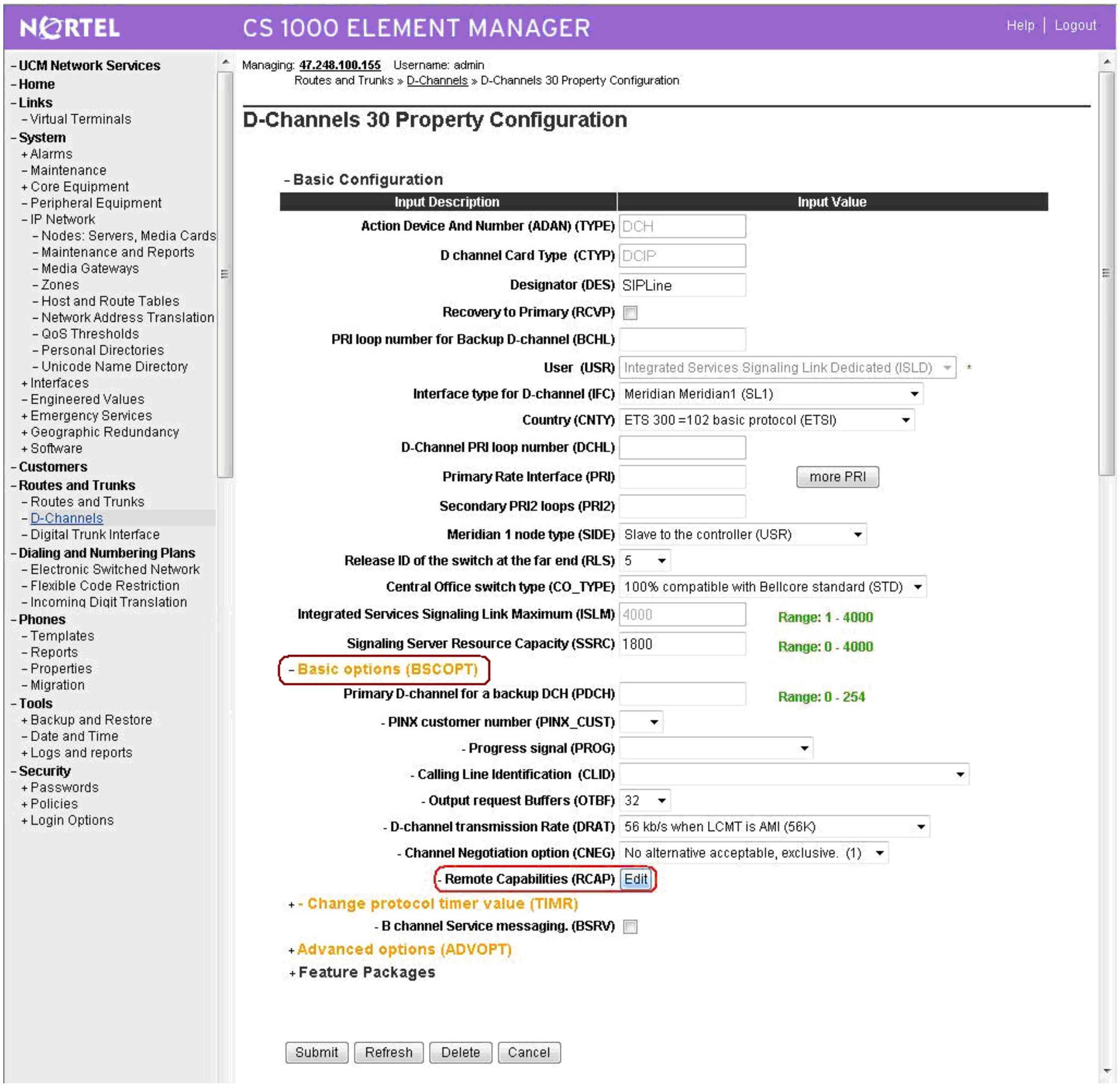

Figure 11 – SIP Line D-Channel Property Configuration

- Figure 12 now appears.
- Select the ***Message waiting interworking with DMS-100 (MWI)*** check box. This must be enabled to support voice mail notification on SIP Line endpoints.
- Select the ***Network name display method 2 (ND2)*** check box. This must be enabled to support name display between SIP Line endpoints.
- At the bottom of the ***Remote Capabilities Configuration*** page, click ***Return - Remote Capabilities***.
- The ***D-Channel xx Property Configuration*** page reappears. Click ***Submit*** button shown at the bottom of the page.

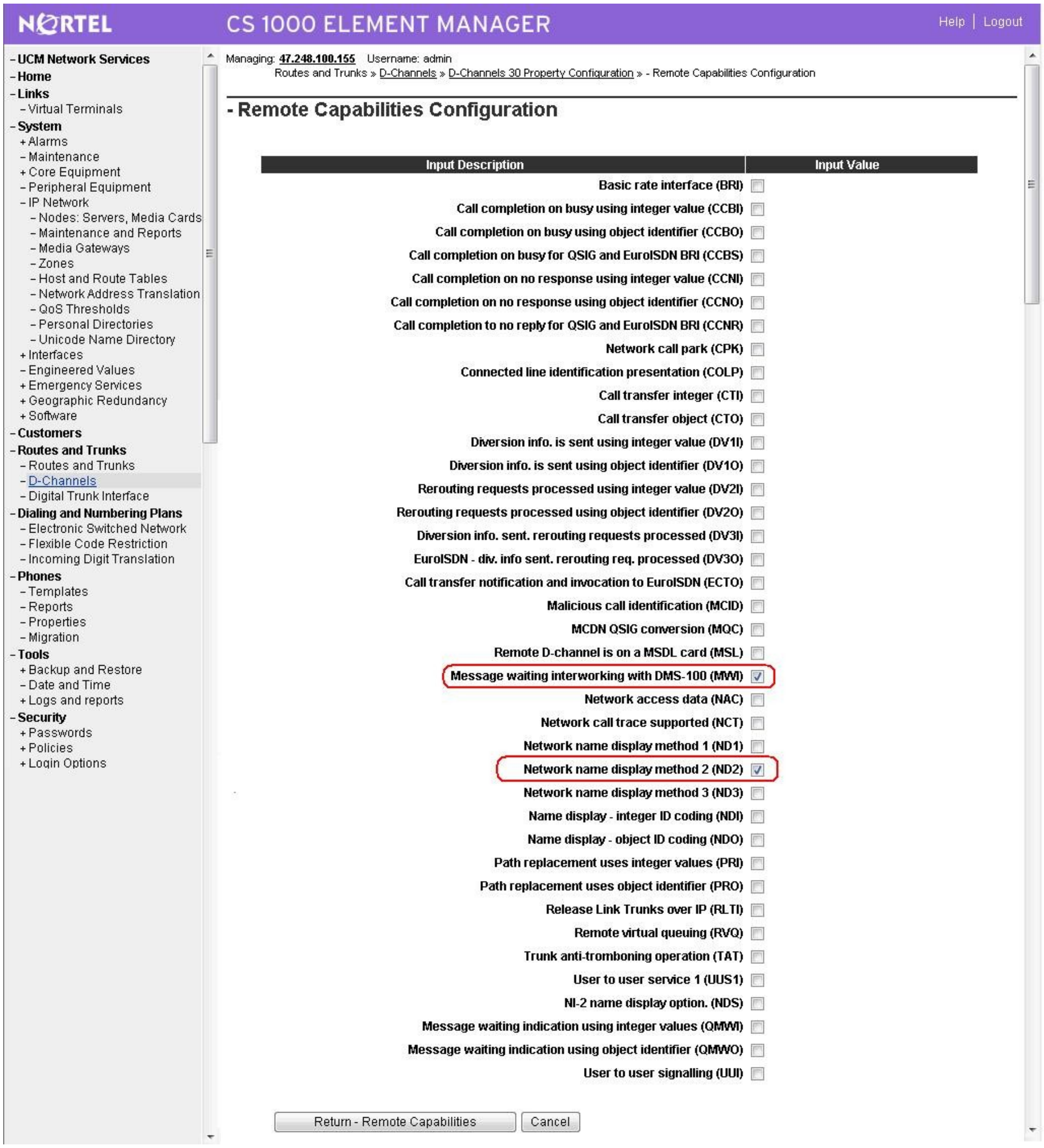

Figure 12 – SIP Line D-Channel RCAP Configuration Details

4.6. Application Module Link (AML) over Embedded LAN (ELAN) Configuration

- On the EM page, navigate to ***System → Interfaces → Application Module Link.***
- Click ***Add*** to add an Application Module Link. ***New Application Module Link*** page appears.
- Enter AML port in the ***Port number*** text box. The SIP Line Service can use ports 32 to port 127. In this case, SIP Line Service is configured to use port 32.
- Click ***Save*** to save the configuration.

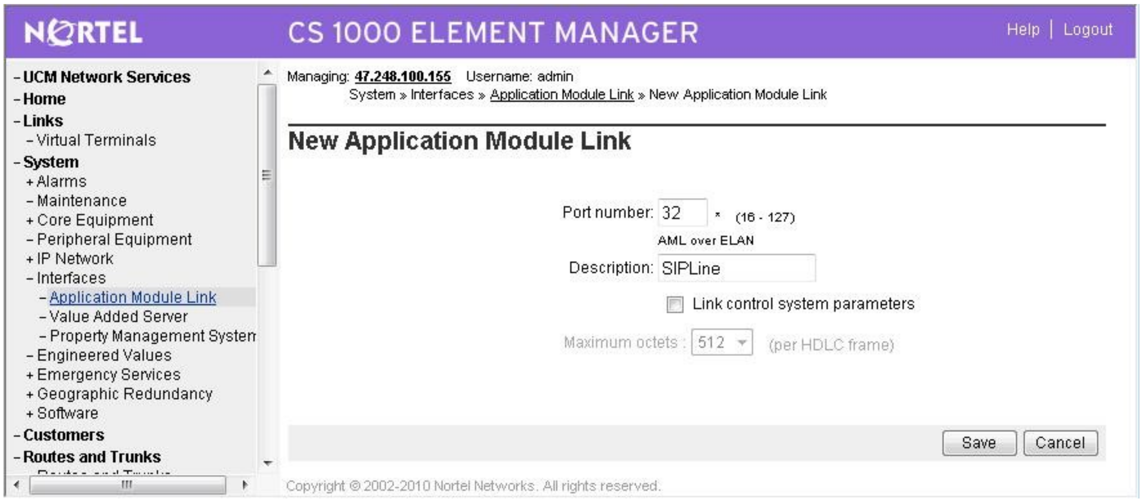

Figure 13 – Application Module Link Configuration

4.7. Value Added Server (VAS) Configuration

- On the EM page, navigate to ***System → Interfaces → Value Added Server.***
- Click ***Add*** to add new Value Added Server. The ***Add Value Added Server*** page appears.
- Click on the ***Ethernet LAN Link***.
- Enter the Ethernet LAN Link number in the ***Ethernet LAN Link*** text box.
- Ensure that the ***Application Security*** check box is unchecked.

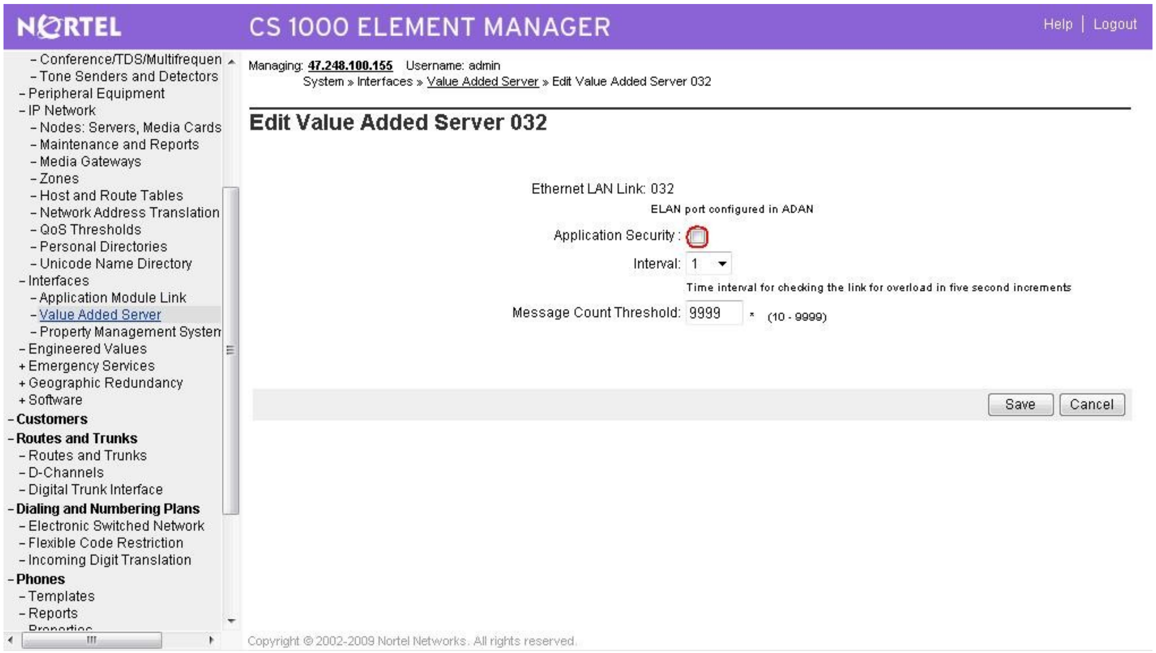

Figure 14 – Value Added Service for Application Module Link

4.8. Virtual Trunk Zone Configuration

- On the EM page, navigate to ***System → IP Network → Zones.***
- On the ***Zones*** page, select ***Bandwidth Zones***.
- On the ***Bandwidth Zones*** page, select a ***Bandwidth Zone number*** from the list, and click to ***Add***.
- On the ***Zone Basic Property and Bandwidth Management*** page, set the zone properties based on bandwidth availability. It is recommended to set the ***Zone Strategy*** to ***BestQuality (BQ).***
- From the ***Zone Intent (ZBRN)*** list, select ***VTRK (VTRK)***.
- Click ***Submit***.

Figure 15 – Virtual Trunk Zone Configuration

4.9. SIP Line Route Data Block (RDB) Configuration

- On the EM page, navigate to ***Routes and Trunks*** → ***Routes and Trunks.***
- Click ***Add*** to select a customer number.
- On the ***Customer xx, New Route Configuration*** page, from the ***Route number (ROUT)*** list, select a route number.
- From the ***Trunk type (TKTP)*** list, select ***TIE trunk data block (TIE).***
- When Trunk Type (TKTP) is selected, the following options appear:
    - Trunk type M911P (M911P)
    - The route is for a virtual trunk route (VTRK)
    - Digital trunk route (DTRK)
    - Integrated services digital network option (ISDN)
- From the ***Incoming and outgoing trunk (ICOG)*** field, select ***Incoming and Outgoing (IAO)***.
- In the ***Access code for the trunk route (ACOD)*** field, enter the access code.
- Select ***The route is for virtual trunk route (VTRK)*** check box.
- In the ***Zone for codec selection and bandwidth management (ZONE)*** field, enter the zone number. (Use the same zone as configured in **4.8** ***"Virtual Trunk Zone Configuration"***)
- In the ***Node ID of signaling server of this route (NODE)*** field, enter the node ID of the SIP Line Gateway.
- From the ***Protocol ID for the route (PCID)*** list, select ***SIP Line (SIPL).***
- Select the ***Integrated services digital network option (ISDN)*** check box.
- From the ***Mode of operation (MODE)*** list, select ***Route uses ISDN Signaling Link (ISLD)***.
- In the ***D channel number (DCH)*** field, enter the D-channel number.
- From the ***Interface type for route (IFC)*** list, select ***Meridian M1 (SL1)***.
- Ensure the ***Network calling name allowed (NCNA)*** and ***Insert ESN Access Code (INAC)*** check boxes are selected.
- For the Basic ***Route Options, Network Options, General Options, and Advanced Configurations*** sections. The default values were used.
- Click ***Save***.

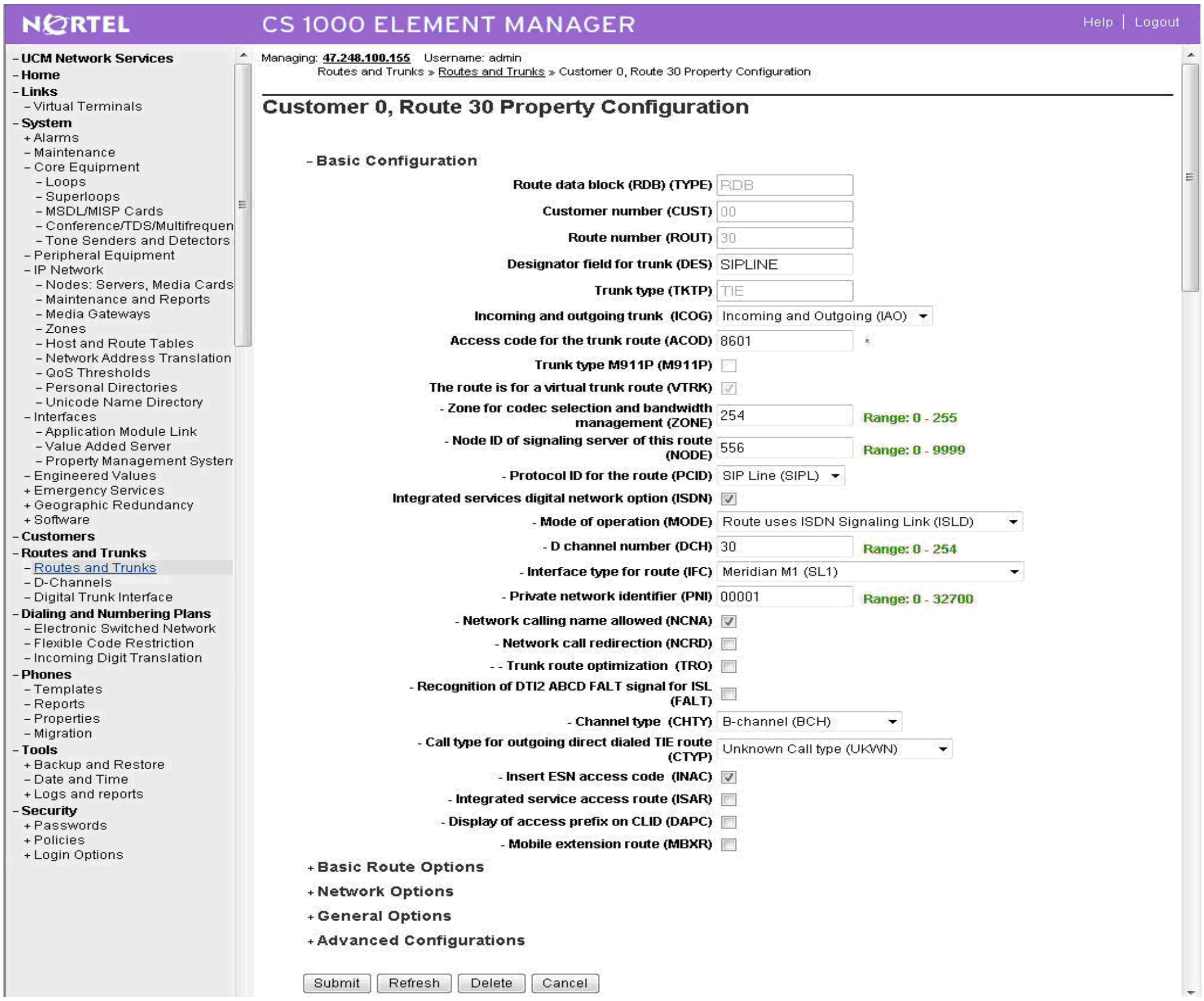

Figure 16 –SIP Line Route Configuration

Note: There is an outstanding issue (Q02073088) with the CS1000 Call Waiting feature which occurs when ***Network Call Redirection*** is enabled. If the Network Call Redirection feature is not required, uncheck the feature to make the Call Waiting work.

4.10. SIP Line Virtual Trunk Configuration

- On the EM page, navigate to ***Routes and Trunks → Routes and Trunks.***
- Select the customer for which you are configuring Virtual Trunks.
- Click ***Add trunk associated with the route listing*** to add new trunk members.
- The ***Customer xx, Route yy, New Trunk Configuration*** Web page appears.
- Choose ***Multiple trunk input number (MTINPUT)*** if you are using more than one trunk.
- From the ***Trunk data block (TYPE)*** list, select ***IP Trunk (IPTI)***.
- In the ***Terminal Number (TN)*** field, enter a TN.
- Enter a ***Route number***, ***Member number (RTMB)***.
- Enter a ***Trunk Group Access Restriction (TGAR)*** value.

- In the ***Channel ID for this trunk (CHID)*** field, enter a ***channel ID*** (where the range is 1 to 382).
- To specify a ***Class of Service (CLS)*** for the trunk, click ***Edit***. The ***Class of Service Configuration*** Web page appears (not shown).
- Select a ***Class of Service***.
- Click ***Return Class of Service*** to return to the ***New Trunk Configuration*** Web page.
- Select ***Basic Configuration***. The ***Basic Configuration*** list expands.
- From the ***Start arrangement Incoming (STRI)*** list, select a value for the start arrangement for incoming calls.
- From the ***Start arrangement Outgoing (STRO)*** list, select a value for the start arrangement for outgoing calls.
- Select ***Advanced Trunk Configurations***. The ***Advanced Trunk Configurations*** list expands (not shown).
- Configure ***Network Class of Service group (NCOS).***
- Click ***Save***.

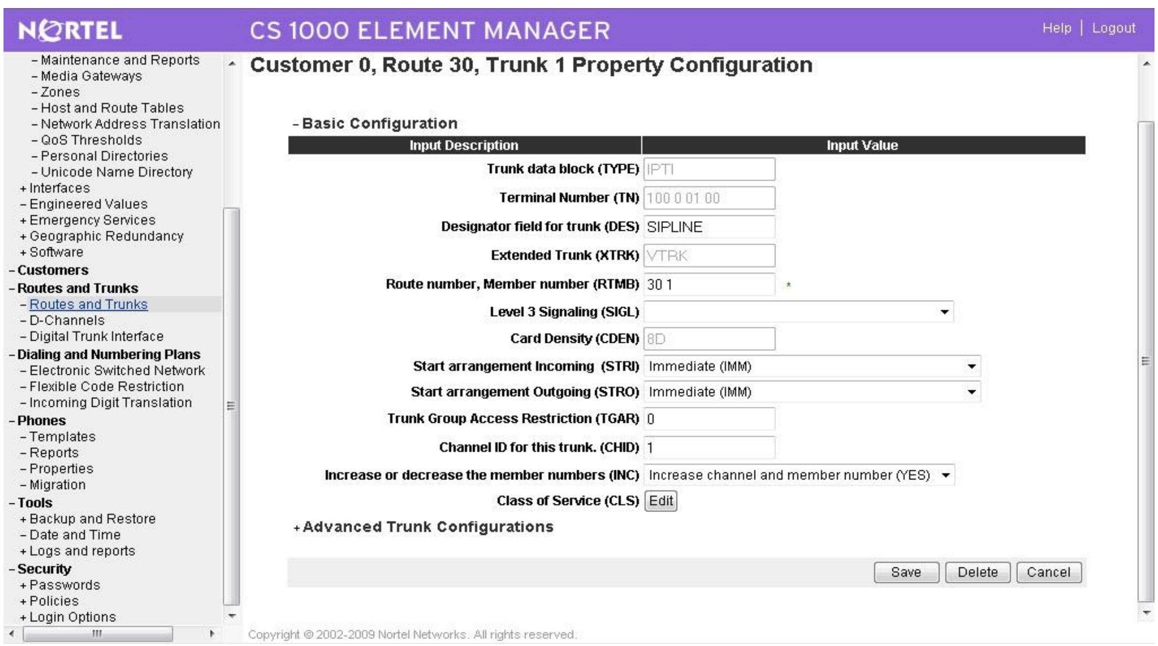

Figure 17 –SIP Line Trunk Configuration

4.11. SIP Line Phones Configuration

Following is a sample configuration for a Third Party SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly.

>LD 11

REQ: prt

TYPE: tnb
TN   96 0 1 27
DATE
PAGE
DES

DES POLYCO
TN   096 0 01 27 VIRTUAL
TYPE **UEXT**
CDEN 8D
CTYP XDLC
CUST 0
UXTY **SIPL**
MCCL **YES**
SIPN 0 ← Set this to 1 and set SIP3 to 0 if this TN is reserved for Nortel SIP Phones
SIP3 **1** ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones
FMCL 0
TLSV 0
SIPU **55573**
NDID **556**
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE **001**
CUR_ZONE 001
ERL
ECL 0
FDN **55576**   ← If CLS FNA is equipped, call will be forwarded no answer to this number
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG **2**   ← This field must be set first if call pickup is equipped (CLS PUA)
SCI 0
SSU
XLST
SCPW **1234**
SFLT NO
CAC_MFC 0
CLS UNR **FBA** WTA LPR **PUA** MTD **FNA HTA** TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD **SWD** LND **CNDA**
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU

GPUA DPUA **DNDA** CFXA 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 FLXD FTTC DNDY DNO3
MCBN
FDSD NOVD VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH
CPND_LANG ENG
RCO 0
HUNT **55576** ← If CLS HTA/FBA is equipped, call will be forwarded busy to this number
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 **SCR 55573** 0 MARP
CPND
CPND_LANG ROMAN
NAME **Polycom 55573**
XPLN 13
DISPLAY_FMT FIRST,LAST
01 HOT U **2655573** MARP 0
02 SCU **0004** ← Speed Call User
03
04 **MSB** ← This key can be different than key 04 to enable Make Set Busy (MBS) feature
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16 55574

20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31

4.12. PSTN Outside Trunk Configuration

Following is a sample configuration which was used during the compliance test. For more information about PRI Trunk Configuration, see **[3].**

Procedure summary

This procedure is applied for both CS1000 system under test and CS1000 PSTN simulator.

No.	Overlay	Action
1	LD 17	Adding a PRI card
2	LD 17	Adding a PRI D-Channel
3	LD 15	Defining a PRI customer
4	LD 16	Defining a PRI service route
5	LD 14	Defining service channels and PRI trunks
6	LD 73	Defining system timers and clock controller
7	LD 48	Enable TMDI or PRI MSDL card
8	LD 60	Enable Clock Controller
9	LD 60	Enable Digital trunk loop
10	LD 96	Enable D-channel

Adding a PRI card

The programming example below shows how to add a PRI card using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	**Response**	**Description**
REQ	CHG	Change data.
TYPE	CFN	Configuration data block.
CEQU	YES	Changes to common equipment.
DLOP	10	Digital Trunk Interface Loop
MG_CARD	4 0 1	MG card assigned to superloop
MODE	PRI	Mode of operation
TMDI	YES	Card is TMDI card
TRSH	0	Threshold

Adding a PRI D-channel

The programming example below shows how to add a PRI D-channel using LD 17. For all other fields not listed in the example press RETURN to use default values.

Prompt	**Response**	**Description**
REQ	CHG	Change existing data
TYPE	CFN	Configuration data block.
ADAN	NEW DCH 10	Add a primary D-channel (any unused SDI port.) xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. Xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.

CTYP	TMDI	Card type where: MSDL = The NTBK51BA Downloadable D-Channel Daughterboard. TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only. ***Note:*** 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	10	PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above. Where: xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
SIDE	NET	NET = network, the controlling switch (applied for CS1000 PSTN simulator USR = slave to the controller (applied for CS1000 system under test)
RLS	6	Software release of far-end. This is the current software release of the far-end. If the far-end has an incompatible release of software, it prevents the sending of application messages, for example, ’Network Ring Again.
RCAP	CCBI CCNI PRI DV3I CTI QMWI	Remote Capabilities.
PR_TRIGS PR_TRIGS PR_TRIGS PR_TRIGS	DIV 2 3 CNG 2 3 CON 2 3 CTR2 2 3	Path Replacement Triggers

Defining a PRI customer

The programming example below shows how to define a PRI customer using LD 15. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer data block.
CUST	0	Customer number.
ISDN	YES	Customer is equipped with ISDN.

Defining a PRI service route

The programming example below shows how to add a PRI service route using LD 16. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW	Create new data
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	10	Route number
DES	T1_QSIG	Designator field for trunk
TKTP	TIE	Trunk type
DTRK	YES	Digital trunk route
ISDN	YES	ISDN option
MODE	PRI	Route used for PRI only
PNI	1	Customer private network identifier. Is the same as the CDB PNI at far-end.
IFC	ISGF	Interface type.
ICOG	IAO	Incoming and outgoing
ACOD	8010	Trunk access code

Defining service channels and PRI trunks
The programming example below shows how to create service channels and PRI trunks using LD 14. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	NEW 23	Create 23 new trunks
TYPE	Tie	Trunk type
TN	10 1	Loop (card) and channel number for digital trunks
PCML	MU	System PCM law.
DES	T1_QSIG	Designator field for trunk
CUST	0	Customer number
RTMB	10 1	Service route number and trunk member number
CLS	UNR DTN	Trunk Class Of Service

Defining system timers and clock controller parameters
Note: This step is only applied for the CS1000 PSTN simulator system which keeps the clock controller.
The programming example below shows how to define system timers and clock controller parameters using LD 73. For all other fields not listed in the example press RETURN to use default values.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	DDB	Digital Data Block
MGCLK	4 0 1	Card slot number for Media Gateway 4 0
PREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.
SREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.

Enabling T1 QSIG Service

Enable TMDI card
The example below shows how to enable TMDI card using LD 48.

```
>ld 48
LNK000
.enl tmdi 4 0 1

OK
```

Enable Clock Controller

The example below shows how to enable clock controller using LD 60.

```
>ld 60
DTI000
.enl cc 4 0
.OK
```

Enable PRI loop

The example below shows how to enable PRI loop using LD 60.

```
>ld 60
DTI000
.enll 10

OK
.
```

Enable D-Channel

The D-Channel may not automatically come up. The example below shows how to enable PRI D-channel using LD 96.

```
>ld 96
DCH000
.enl dch 10
.
DCH: 10  EST CONFIRM  TIME: 19:38:44 30/09/2009

DCH 10 UIPE_OMSG CC_RESTART_REQ REF 00000000 CH 0 TOD  19:38:44 CK
E0DAF978
TYPE: ALL CHANNEL

DCH 10 UIPE_IMSG CC_RESTART_CONF  REF 00008000 TOD  19:38:44 CK E0DAF9C2
TYPE: ALL CHANNEL
```

5. Configure G-Tek GE202 SIP Phone

5.1. SIP Account Settings

Figure 18 shows the account settings used for the SIP phone.

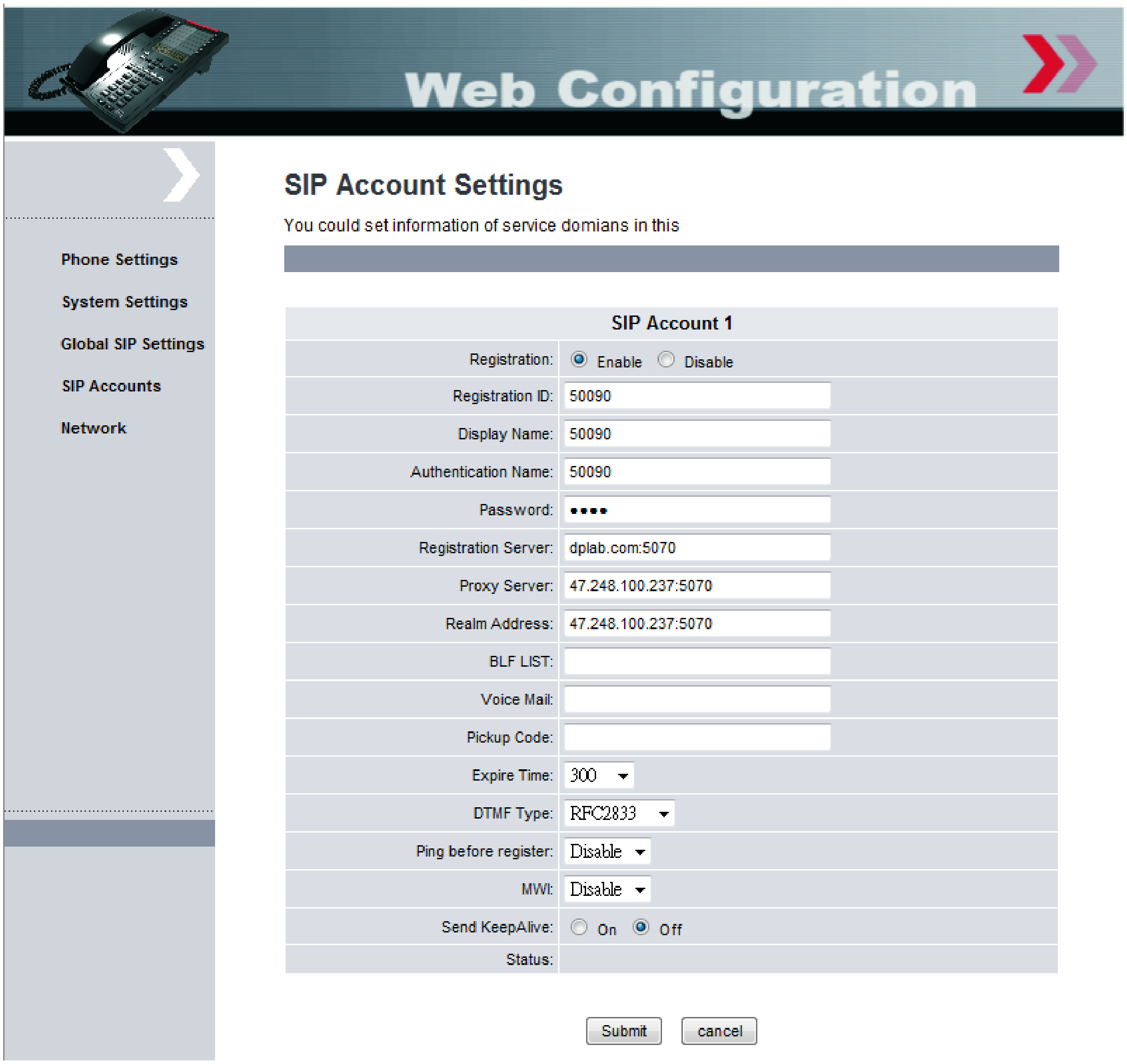

Figure 18 – G-Tek GE-202 SIP Account Settings

5.2. Codec settings

Figure 19 shows the codec settings for the SIP phone.

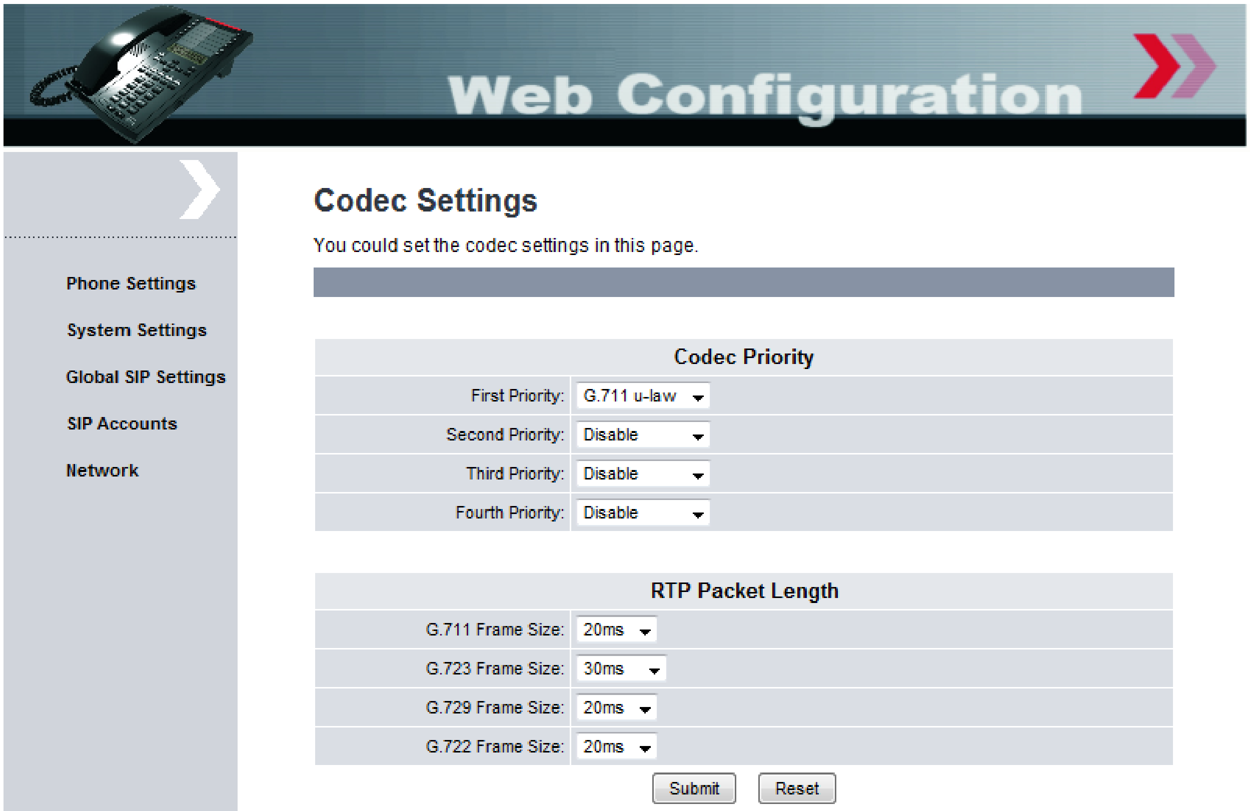

Figure 19 – G-Tek GE-202 Codec Settings

6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment on the GE202 phones and the feature operations such as: busy, hold, DTMF, MWI, codec negotiation, bridge line appearance and dual call.

6.1. General test approach

The general test approach was to have one of the CS1000 clients/users to place a call to and from the G-Tek GE-202 and exercise the telephony features. The main objectives were to verify the GE-202 successfully perform the following:

- Register to CS1000- SIP Line system domain.

- Call establishment with Avaya CS1000 SIP and non SIP phones/clients
- Call establishment with emulated PSTN phones.
- Basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold.
- Advance CS1000 Call Server features: speed dial, group call pickup, ring again busy/no answer, call park/retrieve (from Avaya phones), call forward (busy/all call/no answer), multiple line appearances and multiple appearances DN
- Call redirection and conference: Avaya phones as a transferor for blind/consultative transfers and as a moderator for the 3 way conference call.
- Specific hospitality feature requirement; bridge line appearance (BLA) and dual call arrangement
- Redirect call between users/clients/endpoints: bridge line appearance and dual call
- Codec negotiations.

6.2. Test Results

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

- Avaya has not performed audio performance testing or reviewed the G-Tek GE202 compliance to required industry standards
- Enable Network Call Redirection (NCRD) in CS1000 Call Server SIP Line Route will cause an issue with Call Waiting. CR Q02073088 has been raised against CS1000 SIP Line system. This will be fixed in a future release of CS1000.
- GE202 supports only G711 for this version of firmware.

7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Verify that the VVX1500 phone registers successfully with the CS1000 SIP Line Gateway server and Call Server by using CS1000 Linux command line and CS1000 Call Server overlay LD 32

  - Login to the sipline server using the nortel account.
  - Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked

```
[nortel@sipl ~]$ slgSetShowByUID 55524
=== VTRK ===
UserID           TN               Clients  Calls  SetHandle
---------------  ---------------  -------  -----  ---------
          55524     096-00-02-24        1      0  0xb7c25e10
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1
```

```
== Client 0 ==
IP:Port:Trans = 47.248.100.56:5060:udp
Type          = SIP3
UserAgent     = PolycomVVX-VVX_1500-UA/3.2.2.0481
x-nt-guid     = 9ad7a4871842718a35aeadf070608745
RegDescrip    =
RegStatus     = 1
PbxReason     = OK
SipCode       = 200
Expire        = 300
Contact       = sip:55524@47.248.100.56:5060
Nonce         = cad64489bbd9bca62aa9a1f833052da4
NonceCount    = 3
hTimer        = 0x9c659d0
TimeRemain    = 183
Stale         = 0
Outbound      = 0
ClientGUID    = 0

Key  Func  Lamp  Label
0    3     0     55524
1    126   0     2655524
2    3     0     55097
3    9     0
4    29    0
17   16    0
18   18    0
19   27    0
20   19    0
21   52    0
22   25    0
24   11    0
25   30    0
26   31    0
```

- Login to the call server using the admin account.
- Load overlay 32 and then issue command "stat [TN]" where TN is the SIP Line user's TN being checked

```
>ld 32
NPR000
.stat 96 0 2 24
IDLE REGISTERED 00
```

- Place a call from and to the GE202 phone and verify that the call is established with 2 way speech path.

- During the call, use pcap tool (ethereal/wireshark) at the SIPLine Gateway and clients to make sure that all SIP request/response messages are correct.

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 G-Tek and Avaya design teams. Some of these issues are considered as exceptions. The SIP phone GE202 version 1820X.27.1.01 is considered compliant with CS1000 SIP Line System 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 1000 SIP Line Fundamental, Release 6.0, Revision 01.08, February 2010, Document Number NN43001-508*

[2] *Communication Server 1000E Maintenance, Release 6.0, Revision 03.16, January 2010, Document Number NN43041-700*

[3] *Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Revision 01.03, August 2007, Document Number NN43001-301*

[4] *Troubleshooting Guide for Distributors, Release 6.0, Revision 02.02, December 2009, Document Number NN43001-730*

[5] *Communication Server 1000E Installation and Commissioning, Release 6.0, Revision 03.06, February 2010, Document Number NN43041-310*

[6] *Communication Server 1000E Software Upgrades, Revision 03.12, February 2010, Document Number NN43041-458*

[7] *Communication Server 1000E Linux Platform Base and Applications Installation and Commissioning, Revision 03.10, February 09, 2009 , Document Number NN43001-315*

[8] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision: 03.04, September 28, 2009, Document Number NN43001-116*

Product information for G-Tek products can be found at
http://www.G-Tek.com.tw/en/offering_products.php?mw=9

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.