Avaya Solution & Interoperability Test Lab

Application Notes for VTech Hospitality SIP Corded 2-Line S1220 Telephone Version 02.3.31.02 with Avaya Communication Server 1000 Release 7.5 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and VTech Hospitality SIP Corded 2-Line S1220 telephone.

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 detailed configurations of Avaya Communication Server 1000 SIP Line Release 7.5 and the VTech Hospitality SIP Corded 2-line S1220 telephone version 02.3.31.02. During the compliance test, the VTech S1220 telephone was tested with non-SIP, digital, and SIP endpoints using the CS1000 release 7.5. All the applicable telephony feature test cases of release 7.5 SIP Line were executed on the VTech S1220 telephone , where applicable, to ensure that they interoperate with CS 1000.

2. General Test Approach and Test Results

The general test approach was to have the VTech S1220 telephone register to the CS1000 SIP Line gateway successfully. From the CS1000 telephone clients/users, calls were placed to and from the VTech S1220 telephone. Other telephony features such as busy, hold, DTMF, MWI and codec negotiation were also exercised.

2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the VTech S1220 SIP telephone was able to interoperate with the CS 1000 SIP Line Server. The following areas were tested:

- Registration of the VTech S1220 SIP telephone to the CS1000 SIP Line Gateway.
- Call establishment of VTech S1220 SIP telephone with CS1000 SIP and non-SIP telephones.
- Telephony features: Basic calls, conference, transfer, DTMF (dual tone multi frequency) RFC2833, SIP Info and INBAND transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, ring again, make set busy, DND, Call Waiting and busy/no answer.
- PSTN calls over PRI trunk.
- Codec negotiation – G.711 and G.729.

2.2. Test Results

The objectives outlined in the **Section 2.1** were verified. The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the VTech S1220 telephone compliance to required industry standards.
- VTech SIP S1220 telephone is basically the SIP 3rd phone so it needs to be set as SIP 3rd and also requires the SIP 3rd license.
- The VTech SIP S1220 Local Forward Busy feature which is set on the phone locally can be enabled. However it will be not used for the busy call test since when the phone is in busy status the Server Call Forward Busy feature of CS1000 SIP Line will take place before it can be executed by the phone. It is recommended to set the call forward busy in the CS 1000 SIP Line server.
- It is highly recommended to disable class of service for the media security when provisioning SIP user account for the VTech phone on the Call Server to avoid some unexpected behaviors.

- The VTech SIP Corded 2-Line S1220 telephone is only able to register to the CS1000 SIP Line server with port 5060. It cannot use any other ports.
- The VTech 2-Line S1220 phone supports Call Waiting in case there is only one SIP account (or one real Line) configured. If there are two SIP accounts (or two real Lines) configured, the Call Waiting can be configured on Line 1 by setting forward call on busy to the Line 2 and the Line 2 has to be in idle

2.3. Support

For technical support for the VTech SIP S1220 telephone, please contact VTech Communication Inc technical support as shown below:

Telephone: 1-800-595-9511
Website: www.vtechphones.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the VTech SIP S1220 telephone.

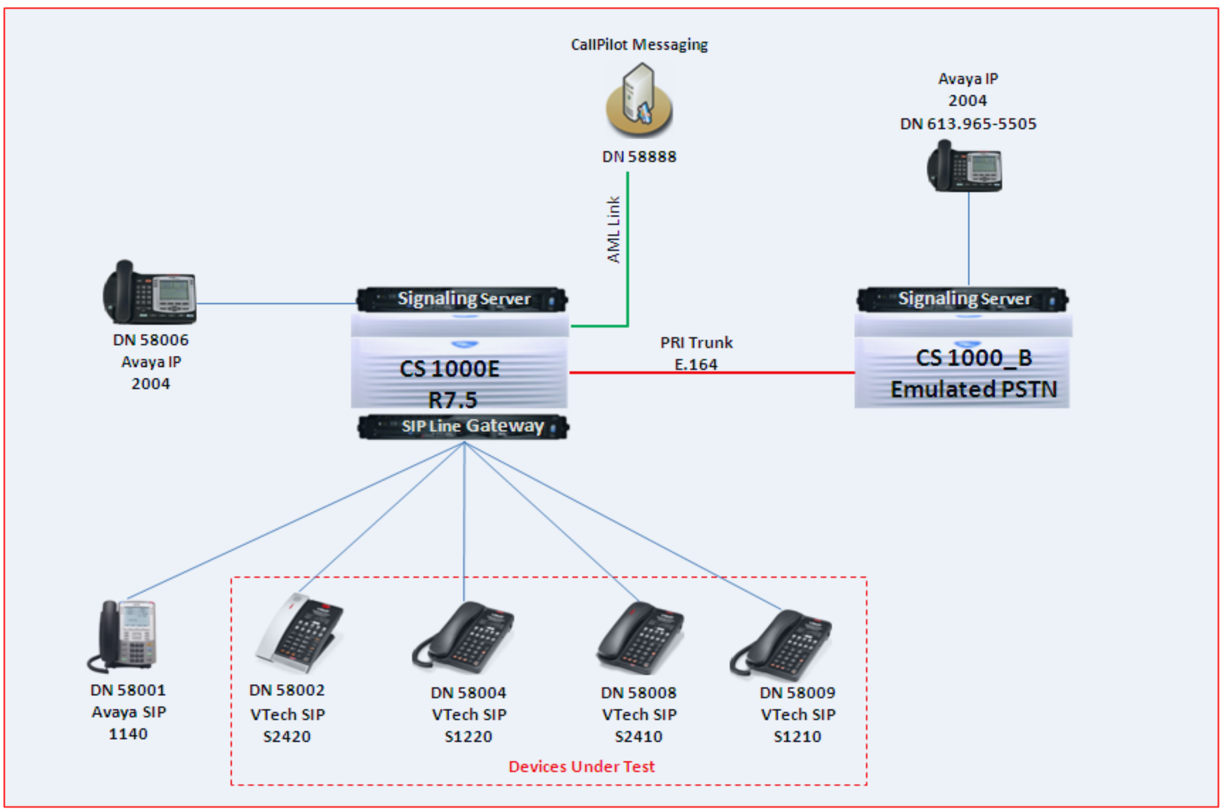


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment	Software Version
Avaya CS1000E	Call Server (CPPM): 7.50Q Signaling Server (CPPM): 7.50.17
Avaya CallPilot® Messaging System	5.0.1
Avaya IP Soft Phone 2050 Avaya IP Phone 1140 Avaya IP Phone 2004P2 Avaya IP Phone 2002P2	3.04.0003 0625C6O 0692D93 0604DC5
Avaya SIP 1140	02.02.21.00
VTech SIP Hospitality 2-Line Cordless S2420	SIP_02.3.31.02
VTech SIP Hospitality 1-Line Cordless S1410	SIP_02.3.31.02
VTech SIP Hospitality 2-Line Corded S1220	SIP_02.3.31.02
VTech SIP Hospitality 1-Line Corded S1210	SIP_02.3.31.02

5. Configure Communication Server 1000 SIP Line Gateway

This section describes the steps to configure the Avaya CS1000 SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS 1000 system. For detailed information on how to configure and administer the CS 1000 SIP Line, please refer to the **Section 9 [1]**.

The following is the summary of tasks needs to be done for configuring the CS 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service and Configure the Root Domain.
- Create SIP Line Telephony Node.
- Create D-Channel for SIP Line.
- Create an Application Module Link (AML).
- Create a Value Added Server (VAS).
- Create a Virtual Trunk Zone.
- Create a Route Data Block (RDB).
- Create SIP Line Virtual Trunks.
- Create SIP Line phones.

5.1. Prerequisite

This document assumes that the CS1000 SIP Line server has been:

- Installed with CS 1000 Release 7.5 Linux Base.
- Joined CS 1000 Release 7.5 Security Domain.
- Deployed with SIP Line Application.

The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_AVAYA	415	Avaya SIP Line package	Existing package	Global
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	Global

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

Use the Microsoft Internet Explorer browser to launch CS 1000 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.

Log in with the username/password which was defined during the primary security server configuration, the UCM home page appears as shown in the **Figure 2** below.

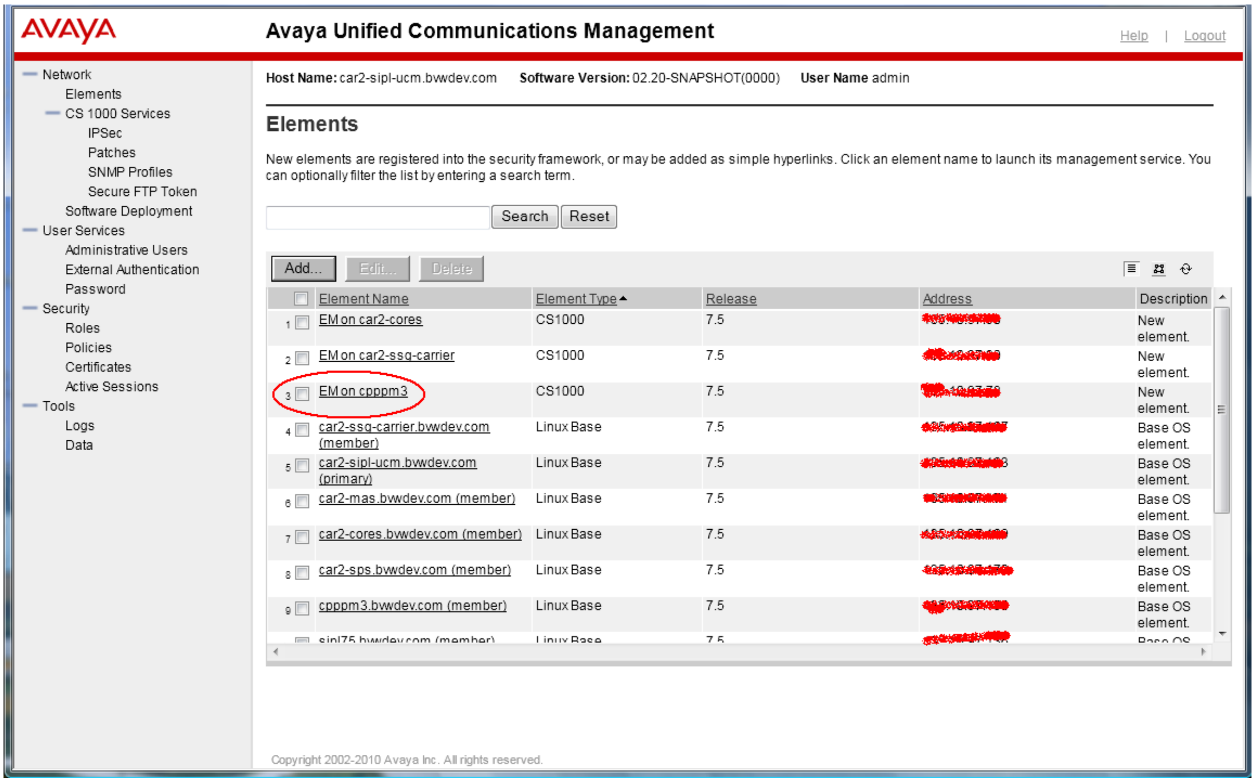


Figure 2: The UCM Home Page of CS 1000 Release 7.5

On the UCM home page, under the **Element Name** column, click on the EM name of CS 1000 system that needs to be configured, in this sample that is **cpppm3**. The CS 1000 Element Manager page appears as shown in **Figure 3** below.

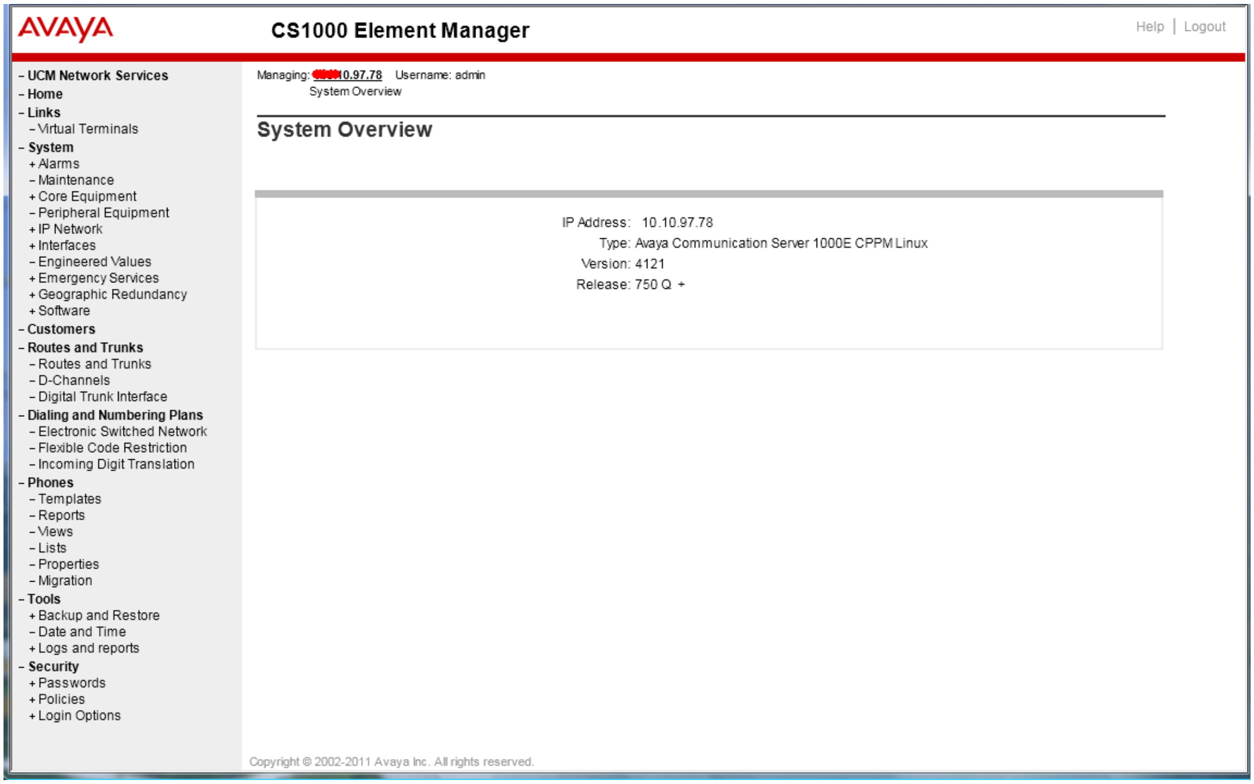


Figure 3: CS 1000 Release 7.5 EM Home Page

5.3. Enable SIP Line Service in the Customer Data Block

On the EM page, navigate to **Customers** on the left column menu; select the customer number to be enabled with SIP Line Service (not shown).

- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box as shown in **Figure 4**.

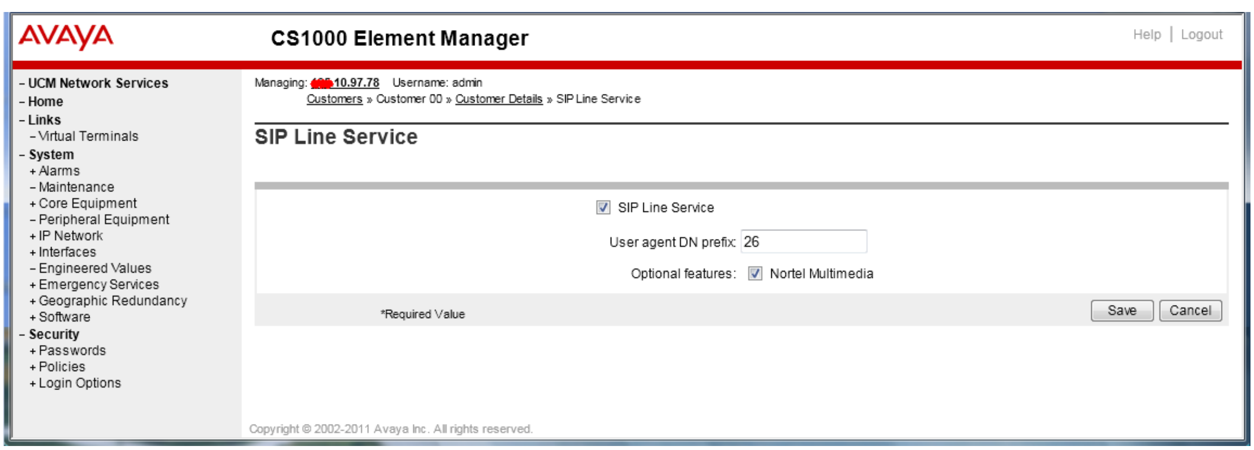


Figure 4: SIP Line Service in Customers Data Block

5.4. Add a new SIP Line Telephony Node

On the EM page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. Click **Add** to add a new SIP Line Node to the IP Telephony Nodes. The new IP Telephony Node page appears as shown in **Figure 5**.

Enter the information as shown below:

- **Node ID** text box: 512 -> this is the node ID of SIP Line server.
- **Call Server IP Address** text box: 10.10.97.78.
- **Node IP Address** text box: 10.10.97.187 -> this is the IP address that SIP endpoint uses to register to.
- **Subnet Mask** text box: 255.255.255.192.
- **Embedded LAN (ELAN) Gateway IP Address** text box: 10.10.97.66.
- **Embedded LAN (ELAN) Subnet Mask** text box: 255.255.255.192.
- Check **SIP Line** check box to enable SIP Line for this Node.

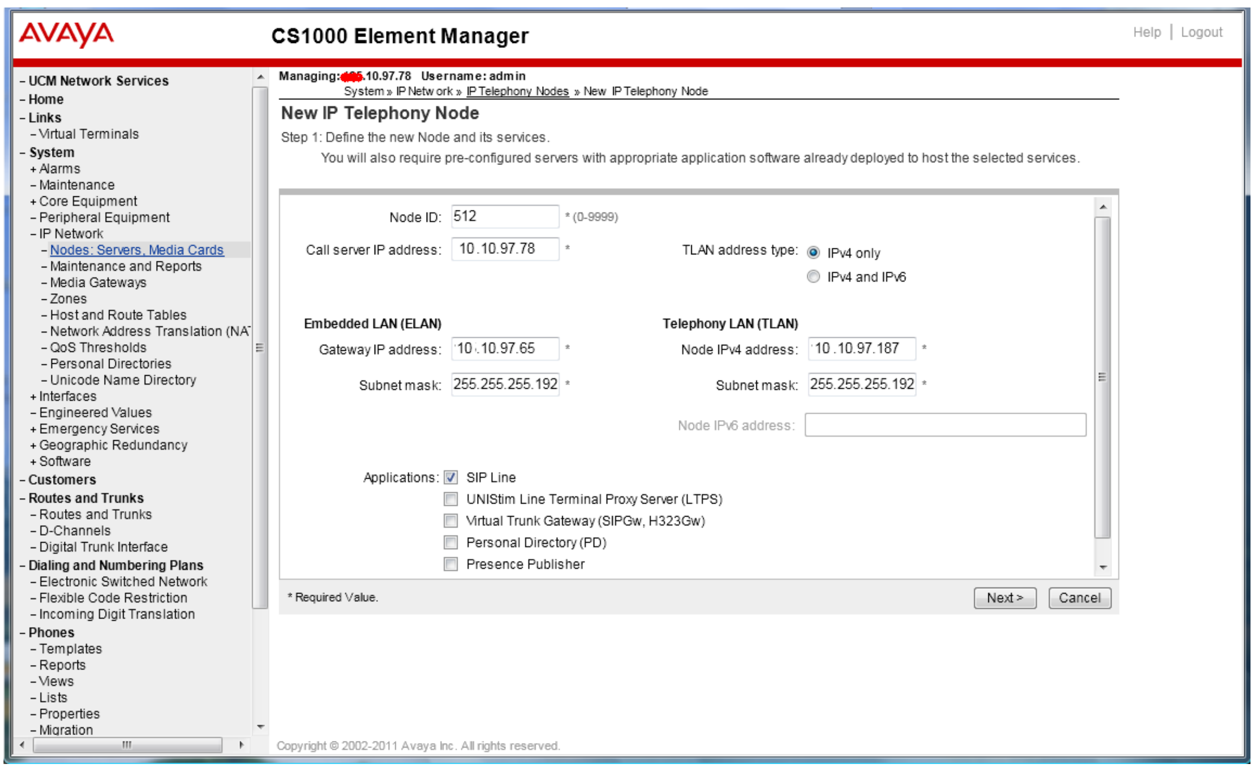


Figure 5: Adding a New IP Telephony Node

- Click on the **Next** button to go to next page**.** The page, New IP Telephony Node with Node ID, will appear as shown in **Figure 6**.
- On the **Select to Add** drop down menu list, select the desired server to add to the node.
- Click the **Add** button
- Select the check box next to the newly added server, and click **Make Leader** (not shown).

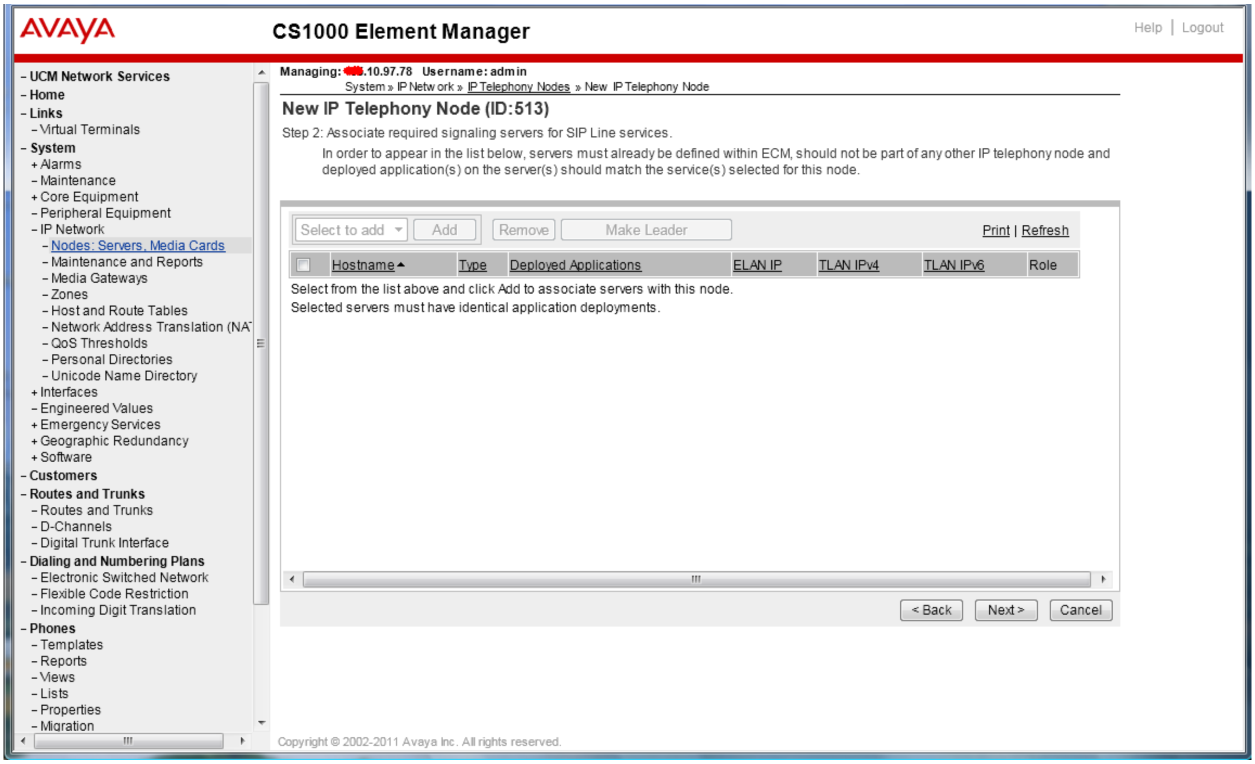


Figure 6: Adding a New IP Telephony Node (cont)

- Click on the **Next** button to go to next page**.** The **SIP Line Configuration Detail** page appears as shown in **Figure 7**.
- Enter SIP Line domain name in **SIP Domain name** text box, for example **sipl75.com**.

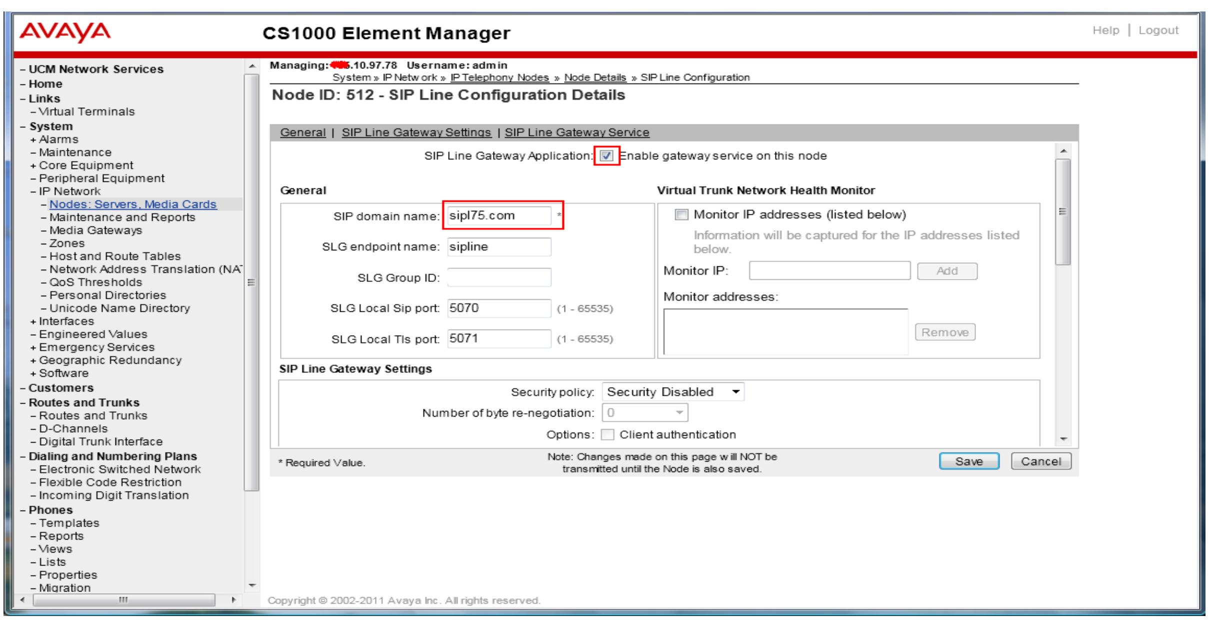


Figure 7: Adding a new IP Telephony Node (cont)

- 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), see **Figure 8**.

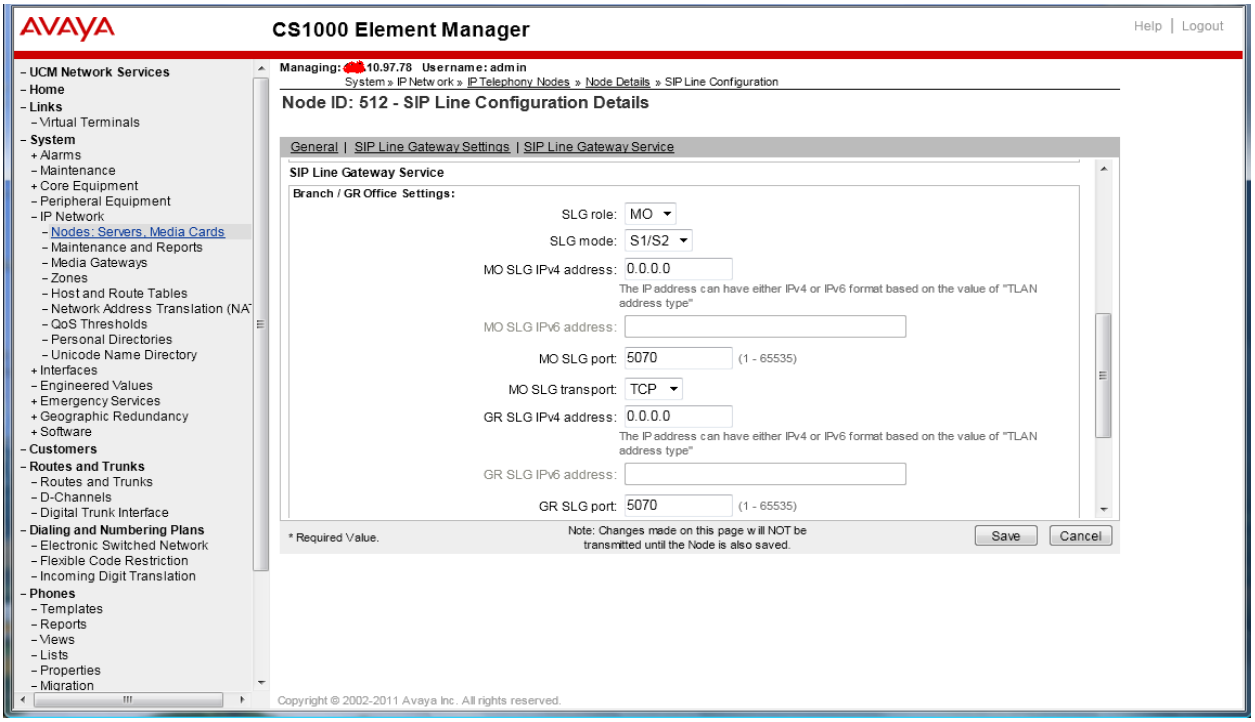


Figure 8: Adding a new IP Telephony Node (cont)

- Click **Next**. The **Confirm new Node details** page appears (not shown).
- Click on the **Transfer Now** button and then The **Synchronize Configuration Files (Node ID 512)** page appears.
- Click **Finish** and wait for the configuration to be saved. The **Node Saved** page appears, see **Figure 9**.

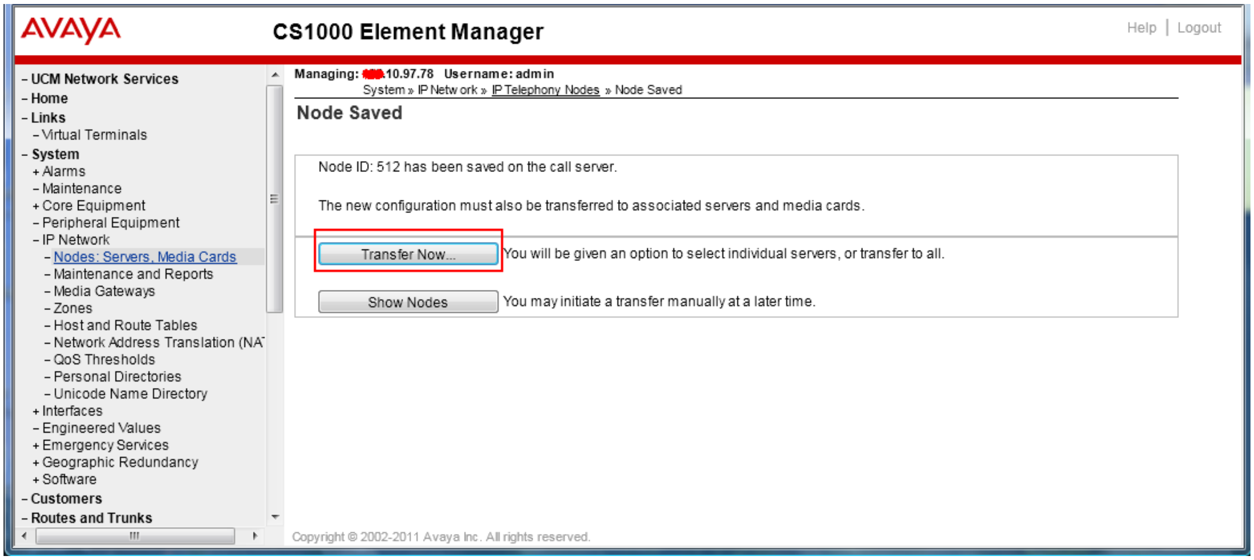


Figure 9: Node Saved with Transfer Configuration

- Select the SIP Line server that associated with changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers, see **Figure 10**.

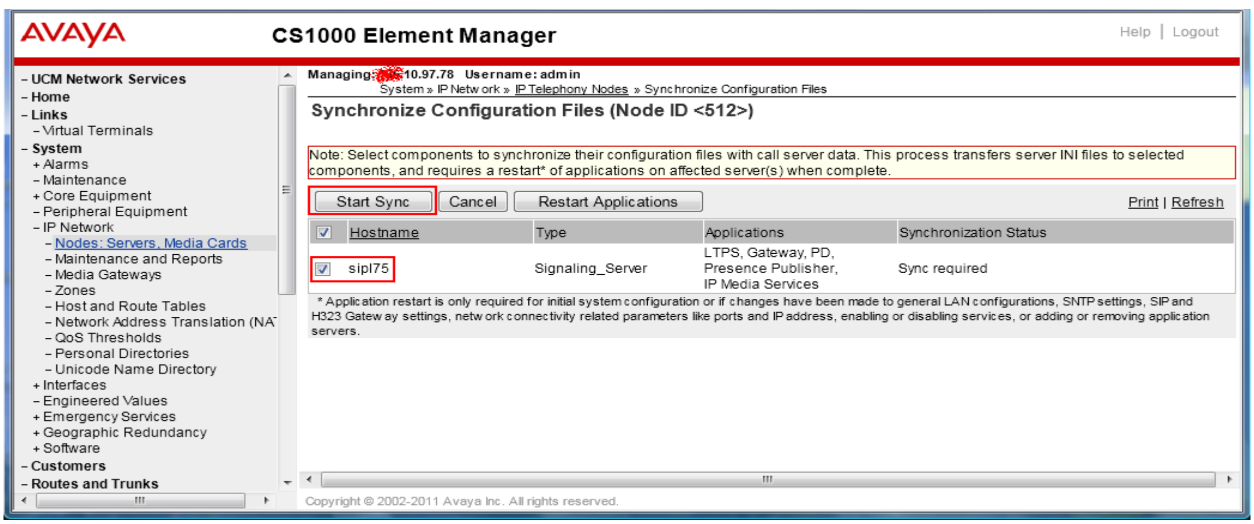


Figure 10: Synchronize Configuration Files

<u>**Note**</u>: The first time a new Telephony Node is added and transferred to the call server, the SIP Line services need to be restarted. To restart the SIP Line services, log in as administrator to the command line interface of the SIP Line server and issue the command: **appstart restart**.

5.5. Create a D-Channel for SIP Line

On the EM page, on the left column menu navigate to **Routes and Trunks -> D-Channels**. Under the **Configuration** section as shown in **Figure 11**, enter a number in the **Choose a D-Channel Number** field, and click on the **to Add** button.

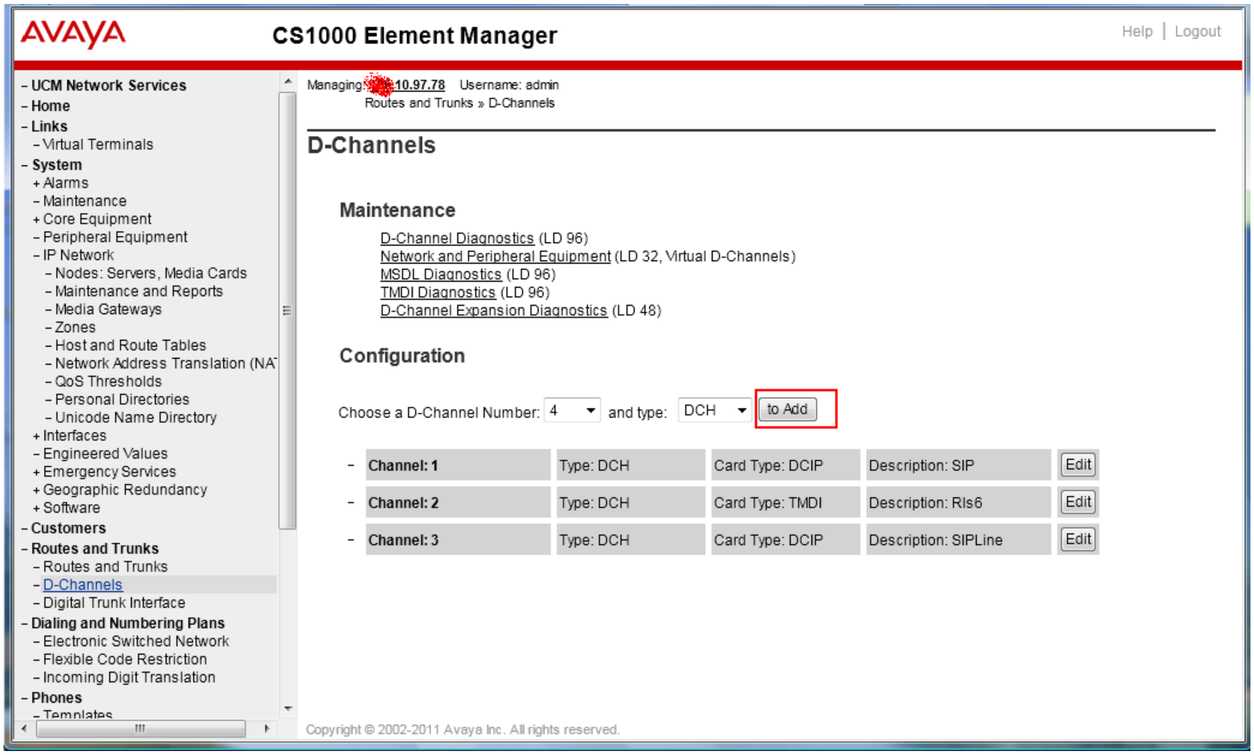


Figure 11: D-Channels configuration page

- The **D-Channels xx Property Configuration** page appears as shown in **Figure 12**.
- From the **Interface type for D-channel (IFC)** list, select **Meridian Meridian1 (SL1)**.
- Leave the other fields at default values.

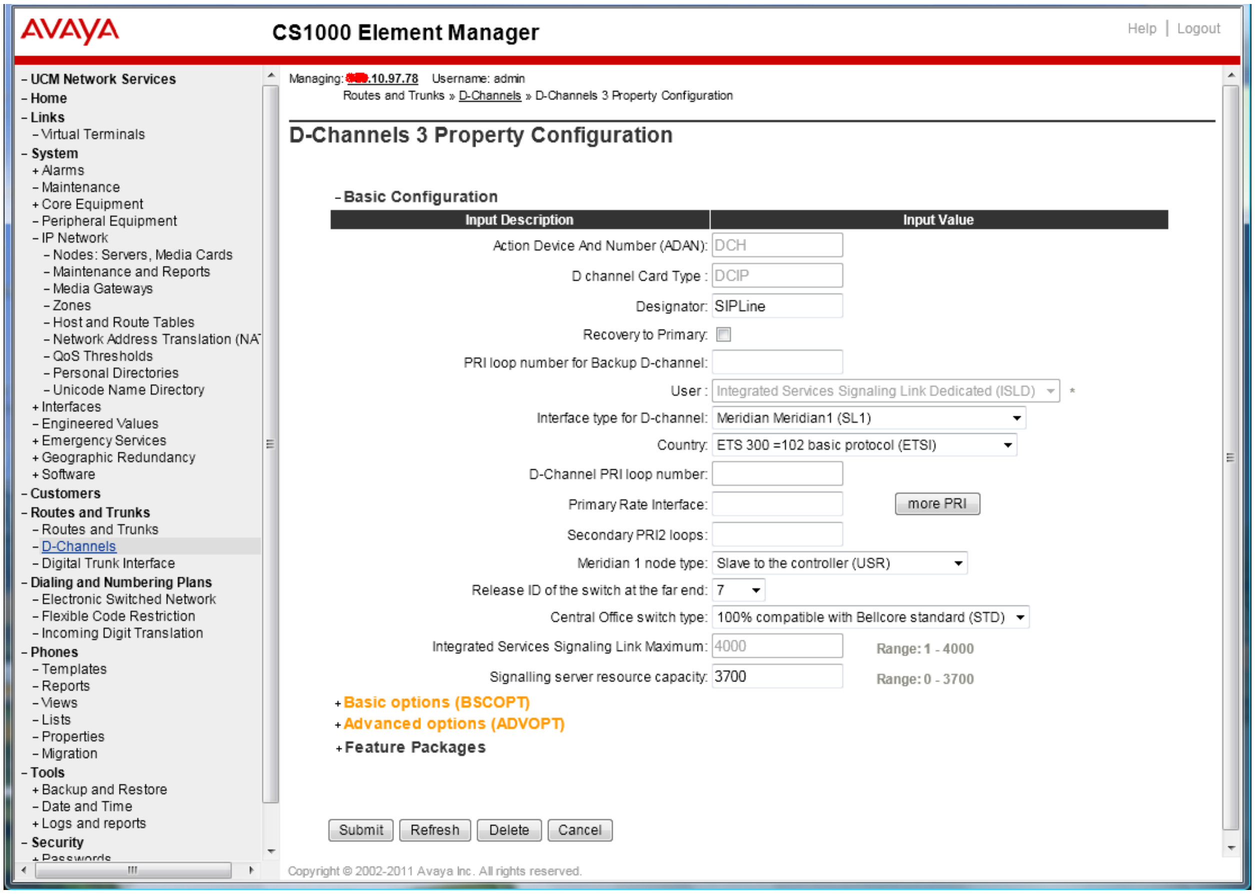


Figure 12: SIP Line D-Channel Property Configuration

- Click on the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands (not shown).
- Click on **Edit** to configure **Remote Capabilities (RCAP)**. The **Remote Capabilities Configuration detail page** will appear as shown in **Figure 13**.
- Select the **Message waiting interworking with DMS-100 (MWI***)* check box.
- Select the **Network name display method 2 (ND2)** check box.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** to return the **D-Channel xx Property Configuration** page.

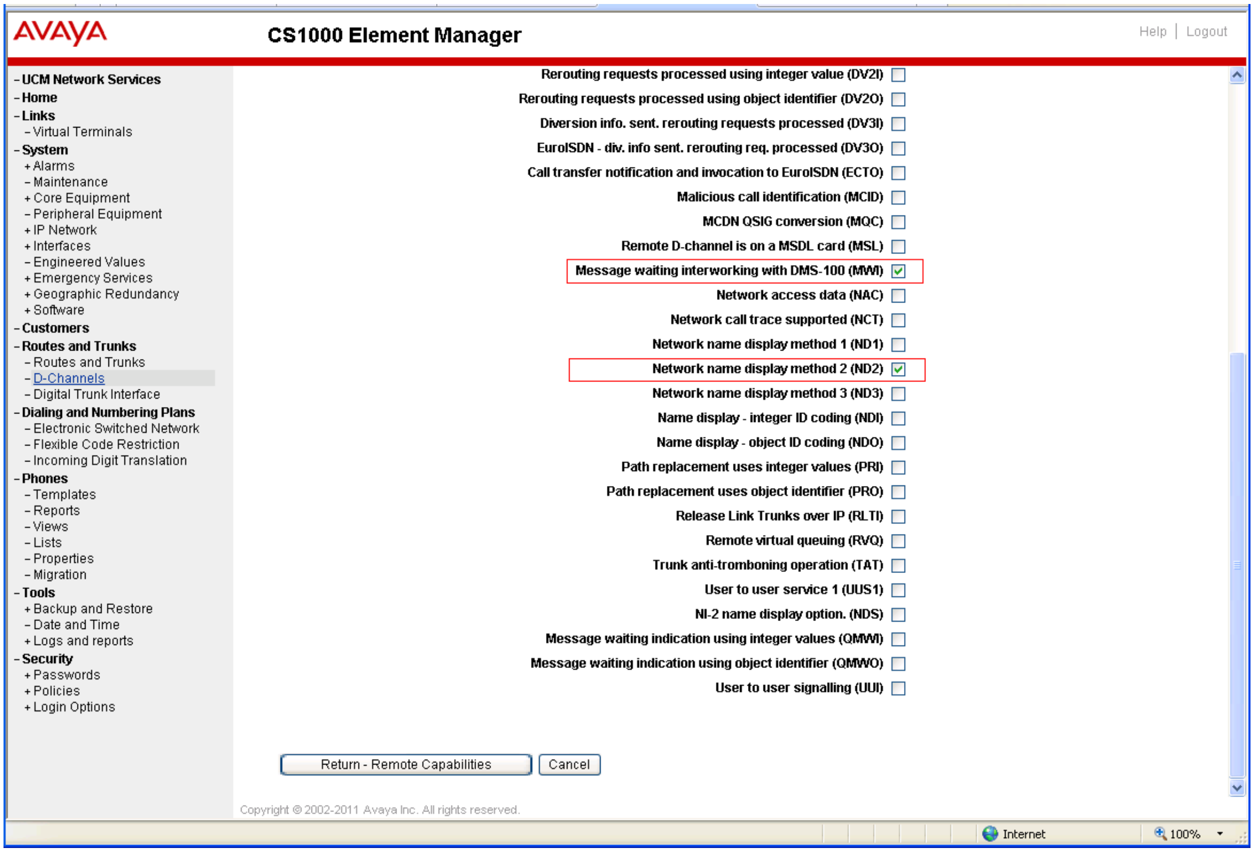


Figure 13: SIP Line D-Channel RCAP Configuration Details

- **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints.
- **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.
- Other check boxes are left unchecked.

Click on the **Submit** button of the D-Channel Property Configuration page to save changes.

5.6. Create an Application Module Link (AML)

On the EM page, navigate to **System -> Interfaces -> Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). The **New Application Module Link** page appears as shown in **Figure 14**.

Enter an AML port number in the **Port number** text box. The AML of SIP Line Service can use a port from 32 to 127. In this case, SIP Line Service is configured to use port 33.

Click on the **Save** button to complete adding the AML link, and to save the configuration.

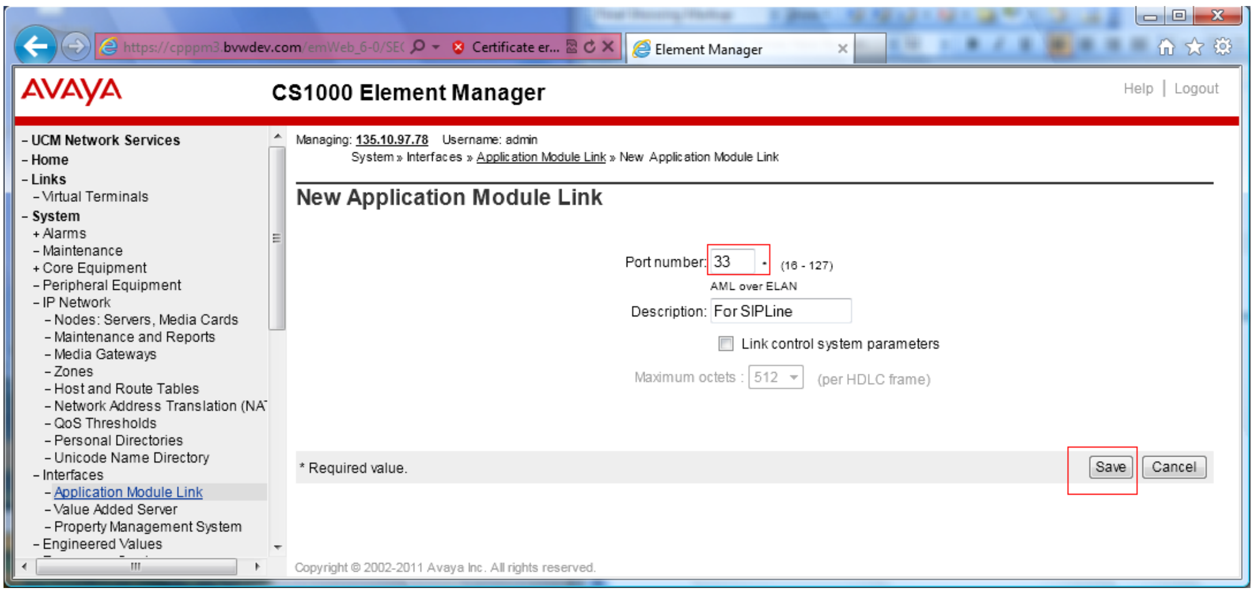


Figure 14: Adding a new AML

5.7. Create a Value Added Server (VAS)

On the EM page, navigate to **System -> Interfaces -> Value Added Server** and click on the **Add** button to add a new VAS.

The **Value Added Server** page appears (not shown), in this page, select the **Ethernet Link** link and the **Ethernet Link** page appears as shown in **Figure 15**.

Enter a number in the **Value added server ID** field, in this example **33** was used. In the **Ethernet LAN Link** drop down list, select the AML number of ELAN that was created in the **Section 5.6**.

Leave other fields as default values and click on the **Save** button to complete adding the **VAS** and save the configuration.

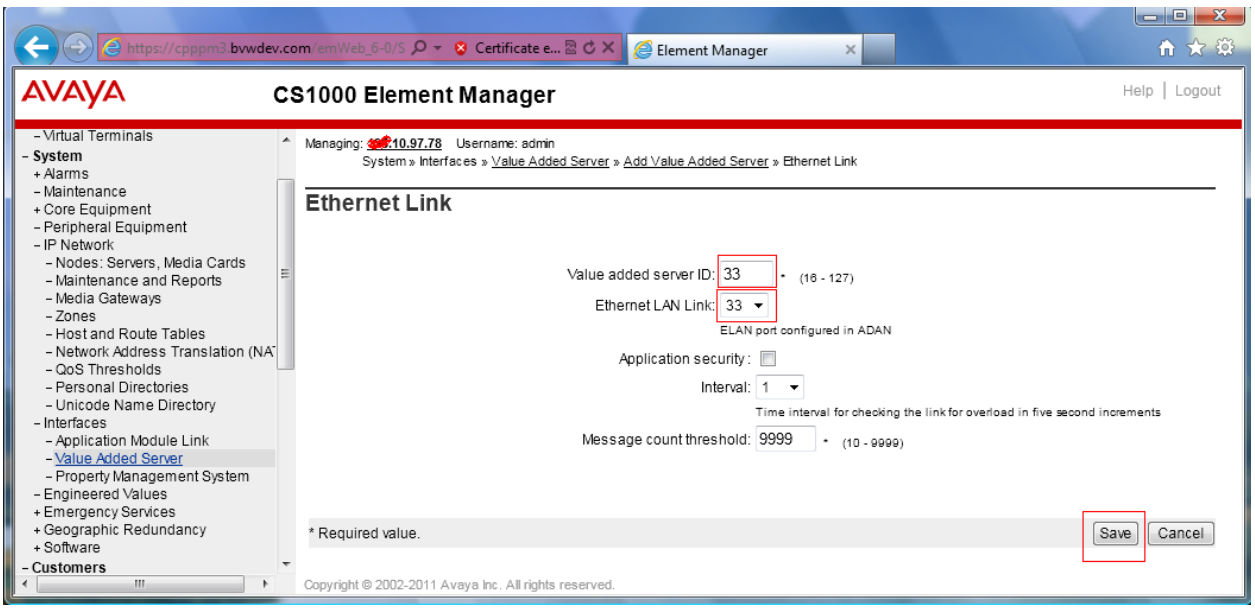


Figure 15: Adding a new Value Added Service for the AML

5.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System -> IP Network -> Zones**. The **Zones** page appears on the right, in this page select **Bandwidth Zones** link.

On the **Bandwidth Zones** page, click on the **Add** button, the **Zone Basic Property and Bandwidth Management** page appears as shown in **Figure 16**.

Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**.

Leave other fields as default values and click on the **Save** button to complete adding the Zone.

Note: Repeat the step above to create another zone for the SIP Line phone; however remember to select **MO**, instead of VTRK in the field **Zone Intent**.

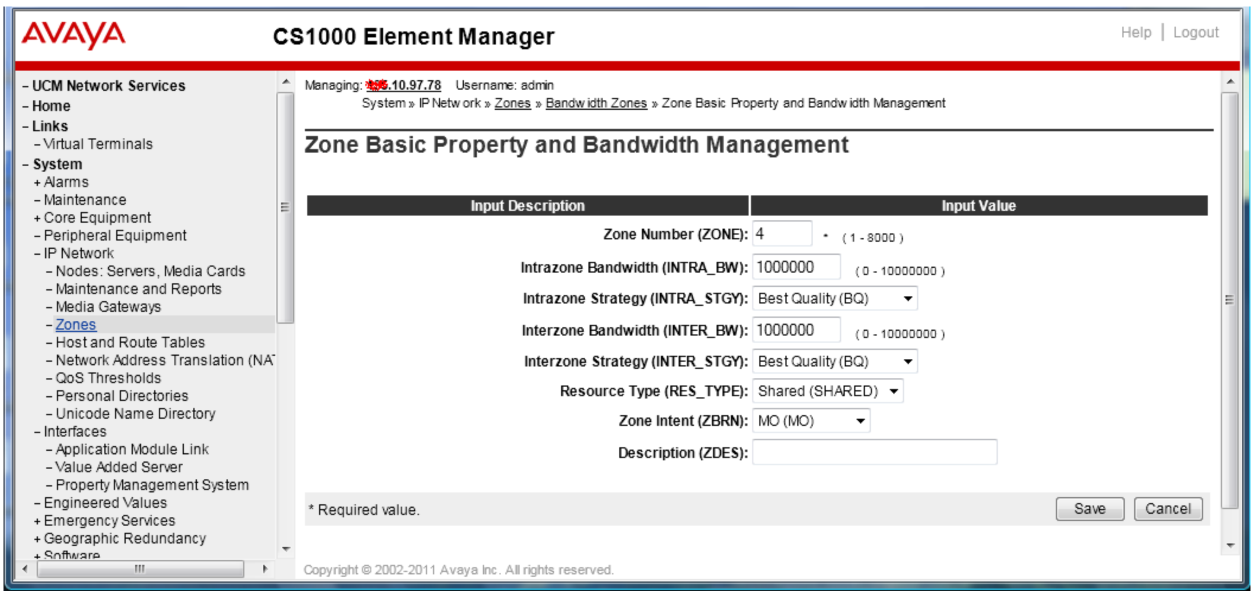


Figure 16: Adding a new Zone for Virtual Trunk

5.9. Create a SIP Line Route Data Block (RDB)

On the EM page, navigate to the menu **Routes and Trunks -> Routes and Trunks**; the **Routes and Trunks** page appears (not shown). In this page, click on the **Add route** button next to the customer number that the route will belong to.

The **Customer ID, New Route Configuration** page appears, expand the **Basic Configuration** tab, and enter values below and as shown in **Figure 17** and **18**.

- **Route Number (ROUT)**: 3
- **Trunk type(TKTP)**: TIE
- **Incoming and Outgoing trunk (ICOG)**: IAO
- **Access Code for Trunk group (ACOD)**: enter a number for ACOD, for example 757.
- **The route is for a virtual trunk route (VTRK)**: Checked.
- **Zone for codec selection and bandwidth management (ZONE*)***: 4, this is the Virtual trunk zone number that created in the **Section 4.8**.
- **Node ID of signaling server of this route (NODE)**: 512, this is the node ID of the SIP Line.
- **Protocol ID for the route (PCID)**: SIP Line (SIPL).
- **Integrated services digital network option (ISDN)**: checked.
- **Mode of operation (MODE):** Route uses ISDN Signaling Link (ISLD).
- **D channel number (DCH)**: 4, the D-channel number that was created in the **Section 4.5**.
- **Interface type for route (IFC)**: Meridian M1 (SL1).
- **Network calling name allowed (NCNA)**: checked.
- **Channel type (CHTP)**: B-channel (BCH).
- **Call type for outgoing direct dialed TIE route (CTYP)**: CDP.
- **Calling Number dialing plan (CNDP)**: CDP.

Leave default values for The **Basic Route Options, Network Options, General Options, and Advanced Configurations** sections.

Click the **Submit** button to complete adding the route and save configuration.

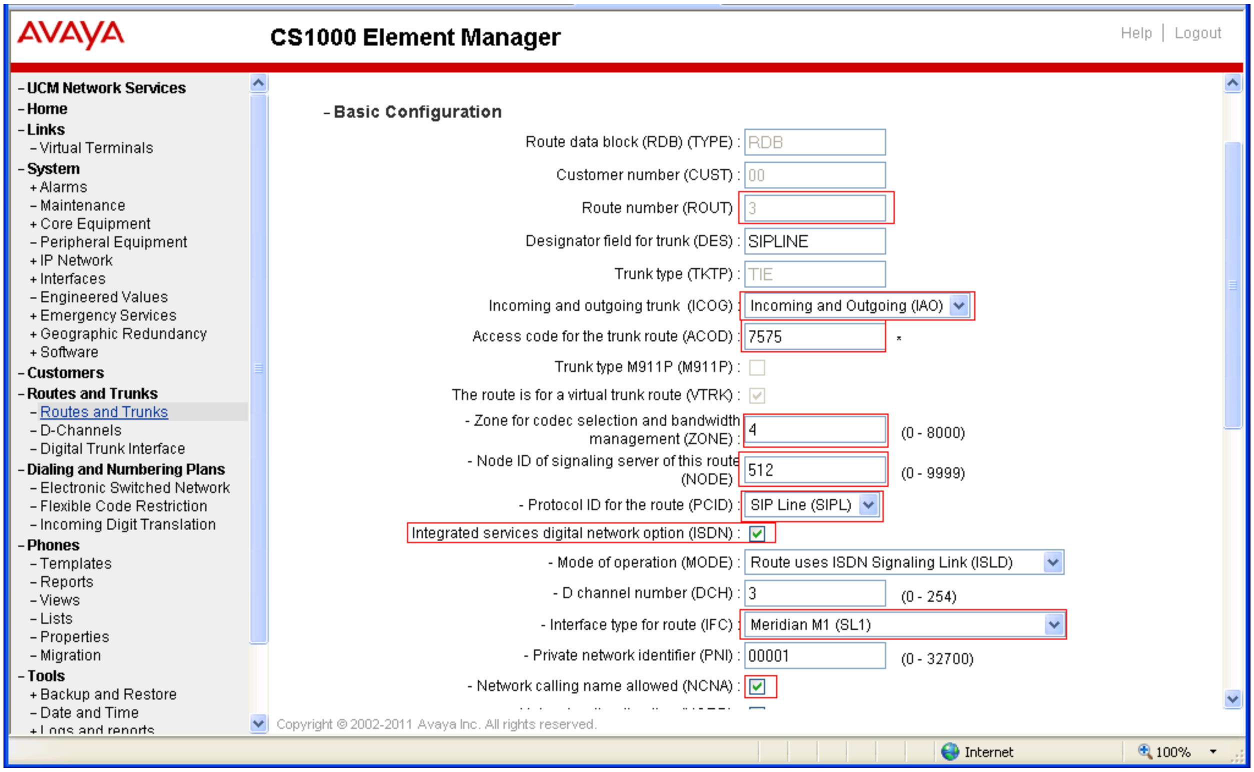


Figure 117: SIP Line Route Configuration

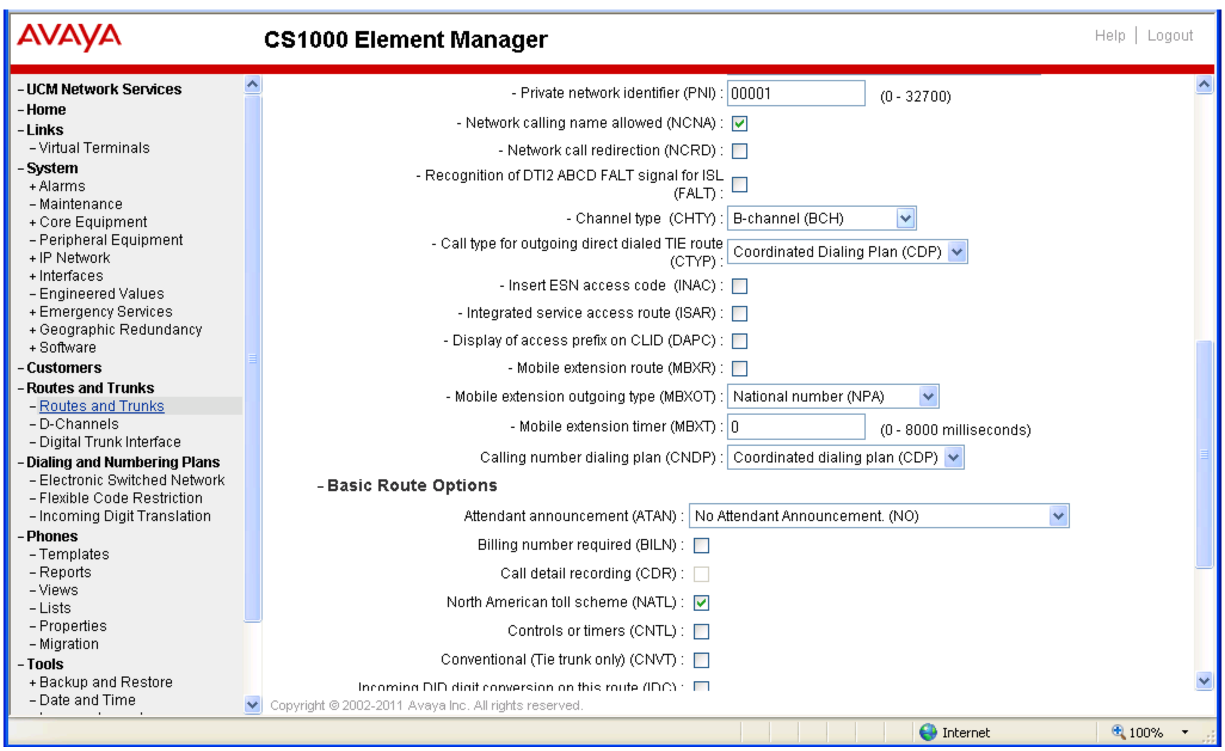


Figure 18: SIP Line Route Configuration (cont)

5.10. Create SIP Line Virtual Trunks

On the EM page, navigate to **Routes and Trunks -> Routes and Trunks** and select the **Add route** button beside to the route was created in the **Section 5.9** above to create new trunks.

The **Customer ID, Route ID, and Trunk type TIE trunk data block** page appears as shown in **Figure 19**, enter values for fields as shown below:

- **Multiple trunk input number (MTINPUT)**: 32 **->** create 32 trunks.
- **Auto increment member number**: checked.
- **Trunk data block (TYPE)**: IP Trunk (IPTI).
- **Terminal Number (TN)**: 100 0 2 0 **->** enter the first TN of a range TN.
- **Member number**: 33, this is ID of trunk, just enter the first ID for first trunk, next ID will be automatically created and incremented.
- **Start arrangement Incoming**: Immediate (IMM).
- **Start arrangement Outgoing**: Immediate (IMM).
- **Trunk Group Access Restriction (TGAR)**: 1.
- **Channel ID for this trunk**: 33, this ID should be the same with the ID of Member Number.

Click on the **Class of Service** button and assign following class of services (not shown):

- **Media security**: Media Security Never (MSNV).
- **Restriction level**: Unrestricted.

Leave other fields at default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page**.**

Click on the **Save** button to complete adding virtual trunks for SIP Line.

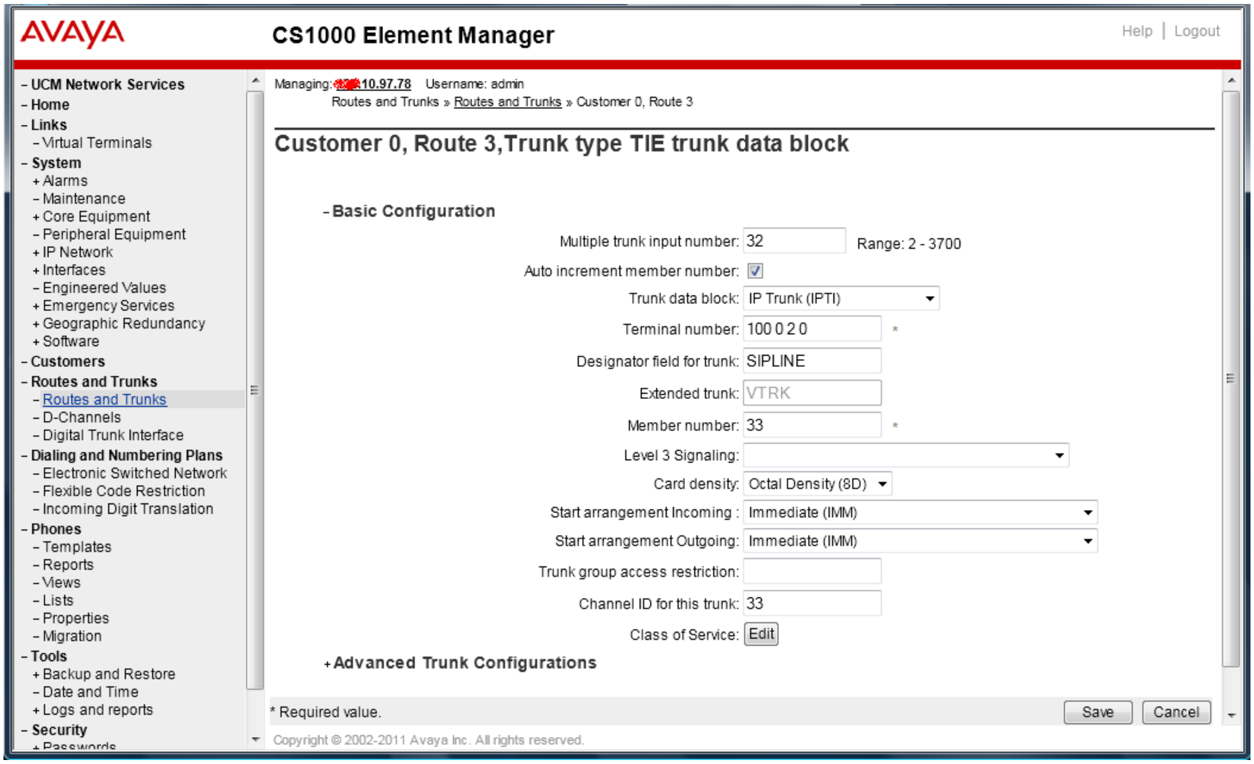


Figure 19: Adding virtual trunks for SIP Line Trunk

5.11. Create a SIP Line Phone

To create a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 20, the example values with some important values explained as shown in Figure 20 and 21 below.

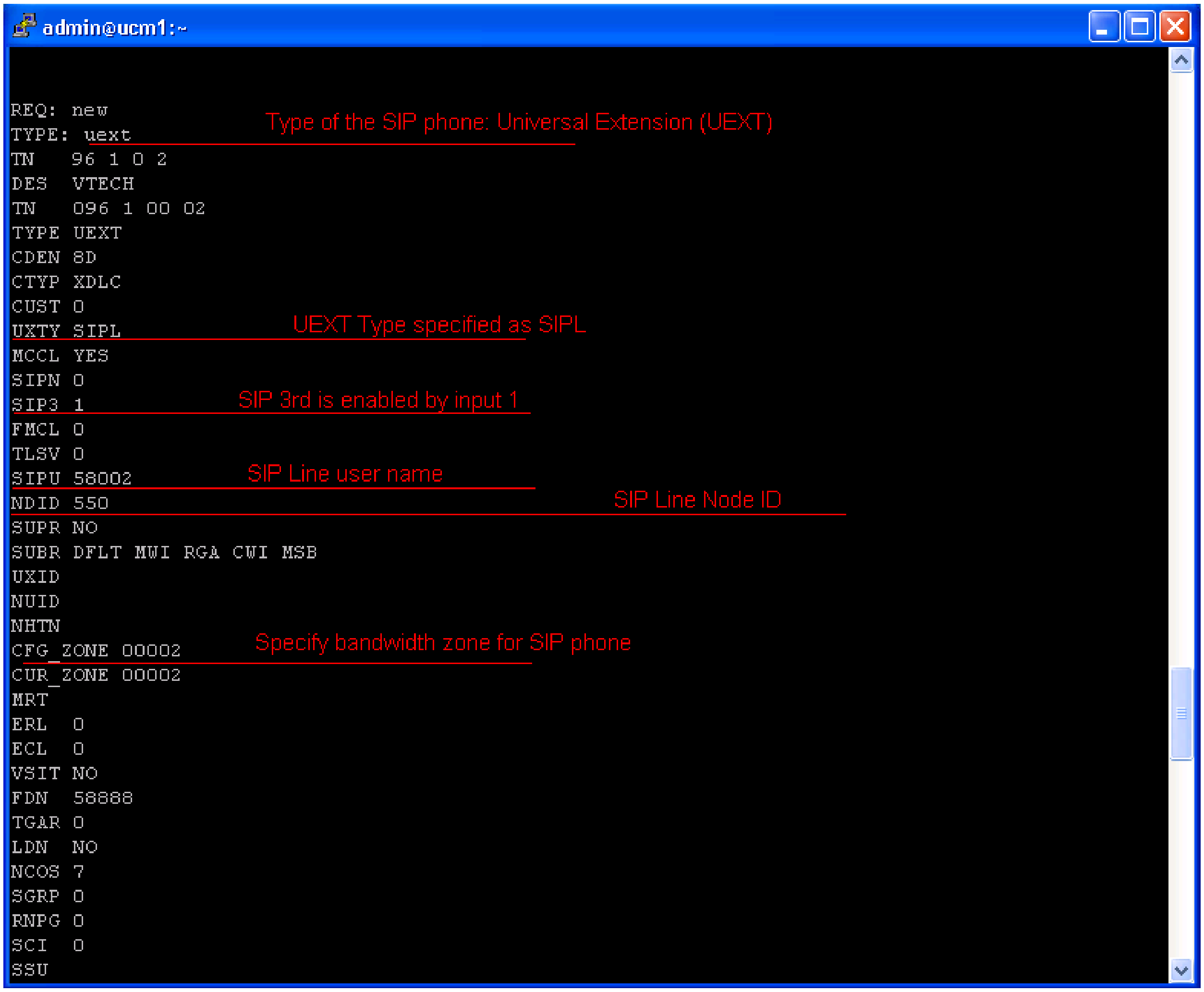
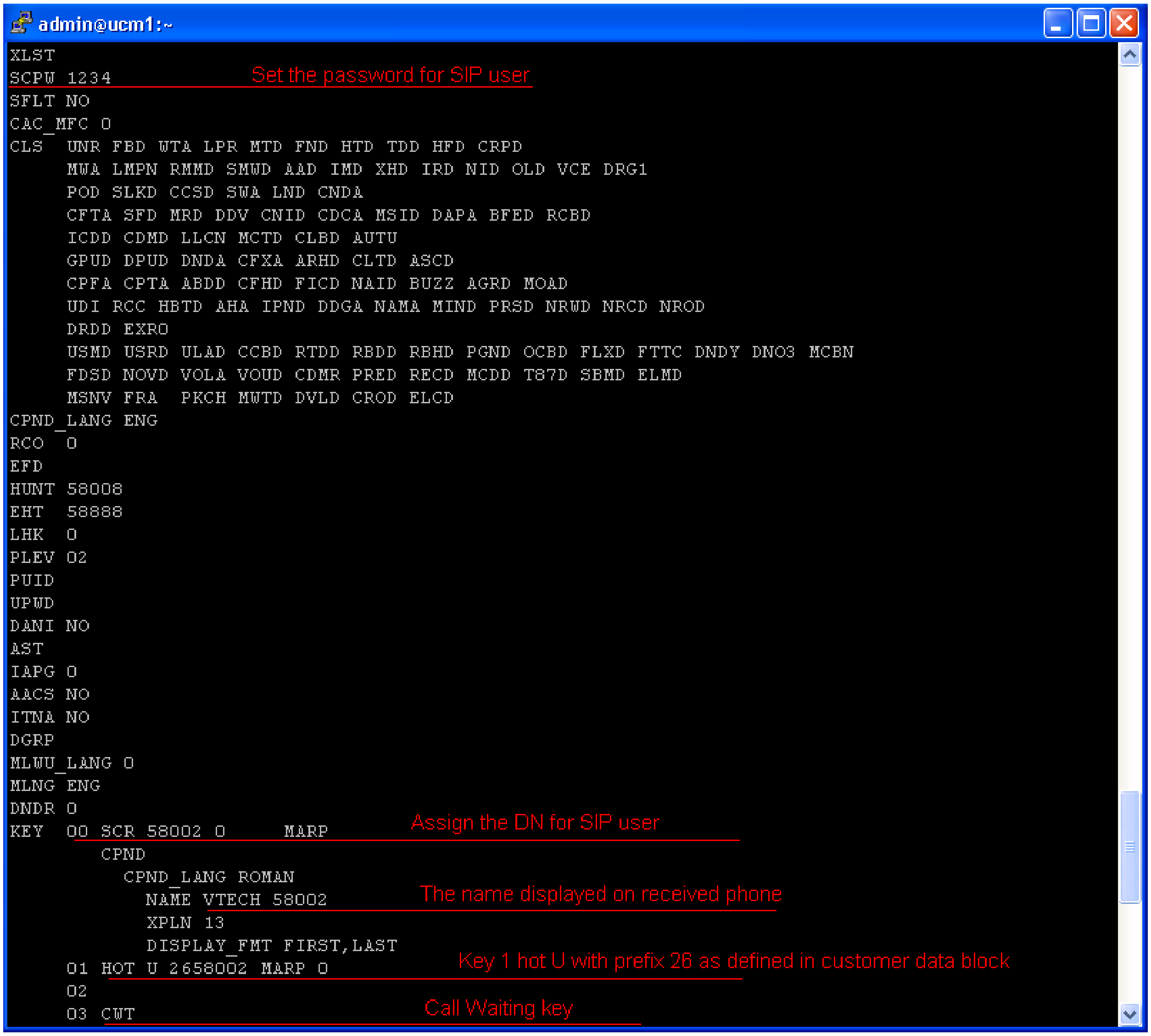


Figure 20: Creating a new sip user in Call Server



```
admin@ucm1:~
XLST
SCPW 1234                    Set the password for SIP user
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR MTD FND HTD TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWA LND CNDA
     CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD 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 OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
     MSNV FRA  PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO  0
EFD
HUNT 58008
EHT  58888
LHK  0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 58002 0     MARP         Assign the DN for SIP user
        CPND
          CPND_LANG ROMAN
            NAME VTECH 58002          The name displayed on received phone
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 HOT U 2658002 MARP 0            Key 1 hot U with prefix 26 as defined in customer data block
     02
     03 CWT                           Call Waiting key
```

Figure: Creating a new sip user in Call Server (cont)

6. Configure VTech Corded 2-Line S1220 Phone

This section describes how to access the VTech S1220 SIP endpoint web interface and configure the S1220 for testing. For more information on how to configure the VTech S1220, please refer to the document in the **Section 9[2]**.

6.1. Login VTech S1220 phone

This section shows how to log in to the home page of VTech S1220 to manage and configure the phone.

Open the web browser, in the address field enter the IP address of VTech S1220 phone: http://ipaddress and the VTech S1220 login page will appear as shown in **Figure 20**. Enter the username and its default password.

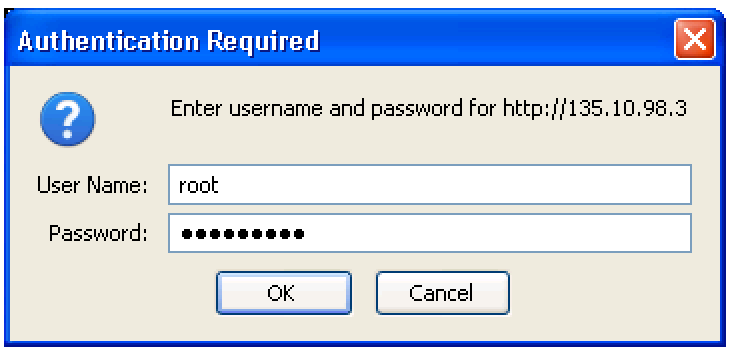


Figure 20: VTech S1220 Login Screen

Click the **OK** button, the homepage of VTech S1220 appears as in **Figure 21** below.

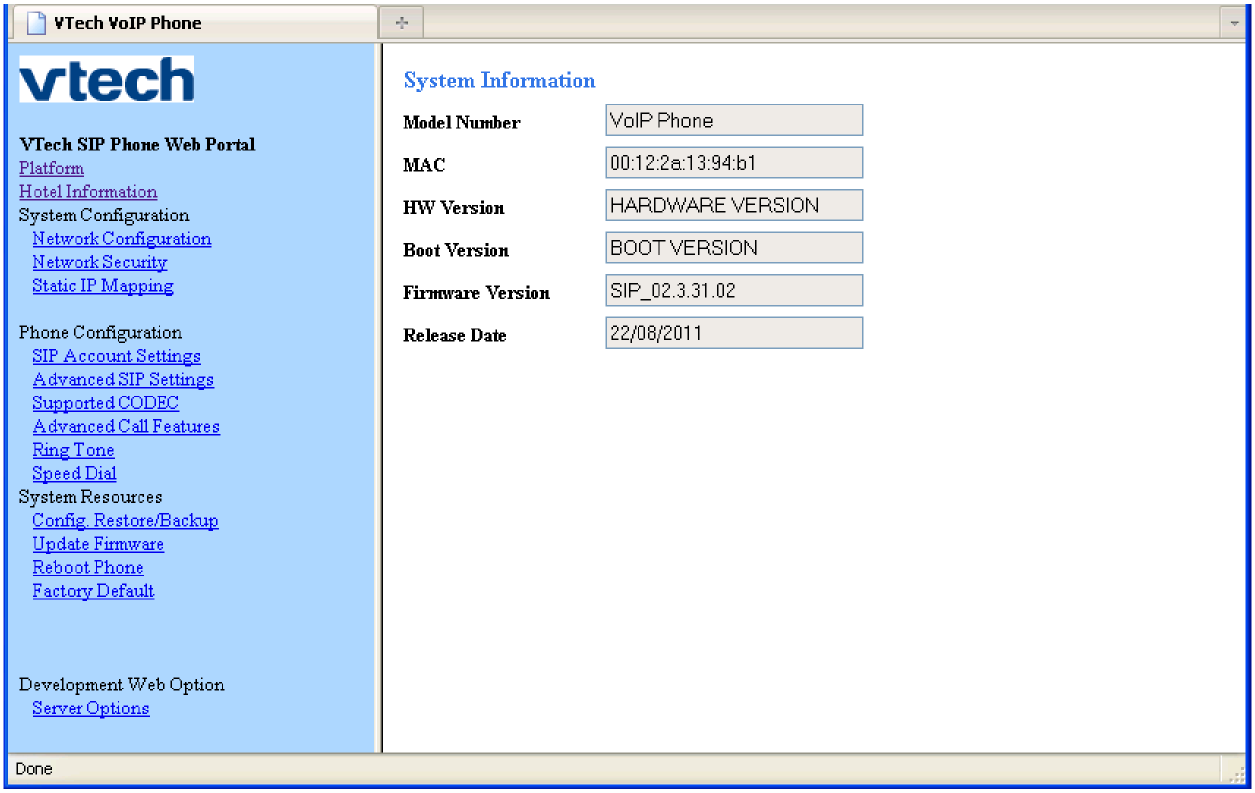


Figure 21: Home page of VTech S1220 phone

6.2. Configure registration for VTech S1220

This section shows how to configure the VTech S1220 telephone to register with the CS1000 SIP Line gateway.

On left-hand side of the homepage (see **Figure 21**), click on the **SIP Account Settings** link, the **Line Selection** appears in the middle of the page, click on the **Line 1** link, the Line 1 Account appears as shown in **Figure 22**. Enter extension number in the **Extension** field, user name in the **Authentication Name**, password in the **Password** field, and select DTMF Method in the dropdown list. Keep the External Call Prefix as default value since the CS1000 SIP Line doesn't use this field to route the call.

Click on the **Save** button to save changes.

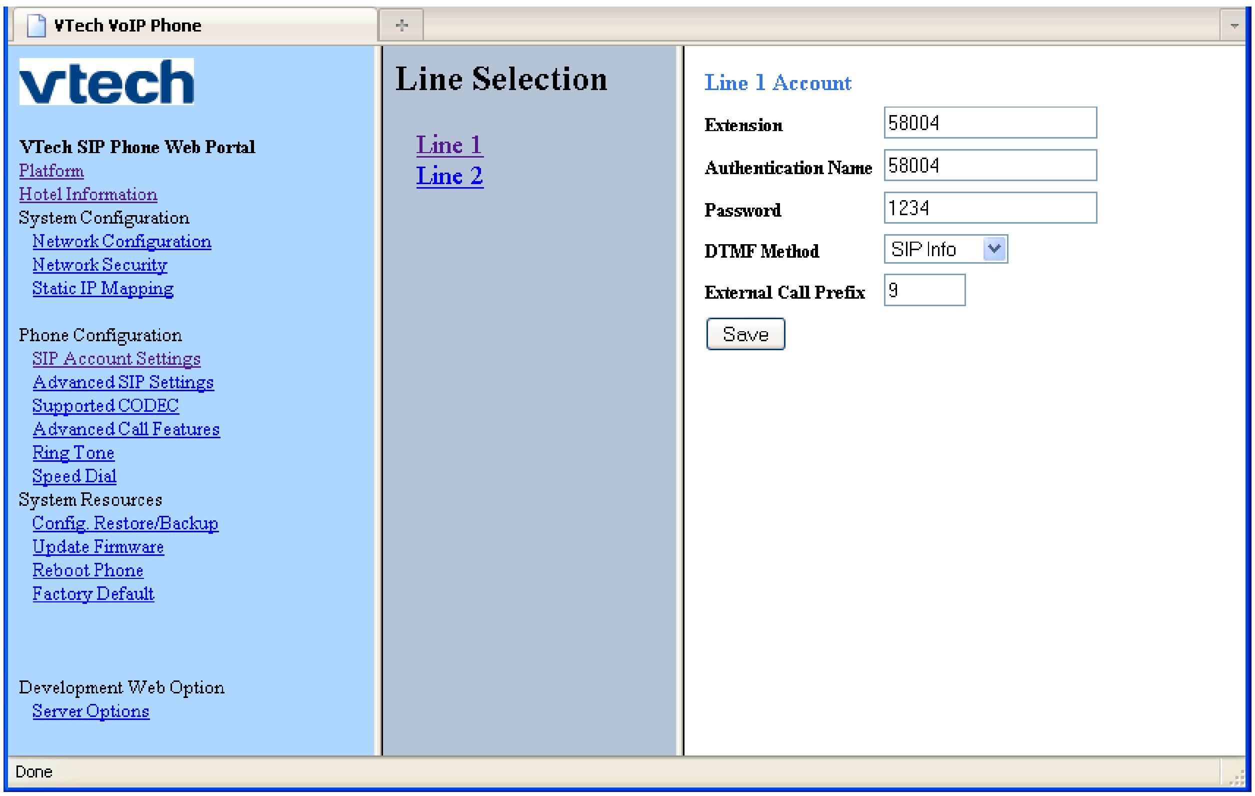


Figure 22: Line Configuration of VTech S1220

Click on the **Advanced SIP Settings** link, the Advanced SIP Settings page appears in the right-hand side of the page as shown in **Figure 23**. Enter the SIP Line domain sipl70.com and port 5060 in the **Registrar Server Address: Port** field, the Node IP address of SIP Line server 135.10.97.133 in the **Proxy Server Address: Port**, UDP in the **SIP Transport** field, ENABLE in the **Prack** field and keep other fields as the default.

Click on the **Save** button to save the changes.

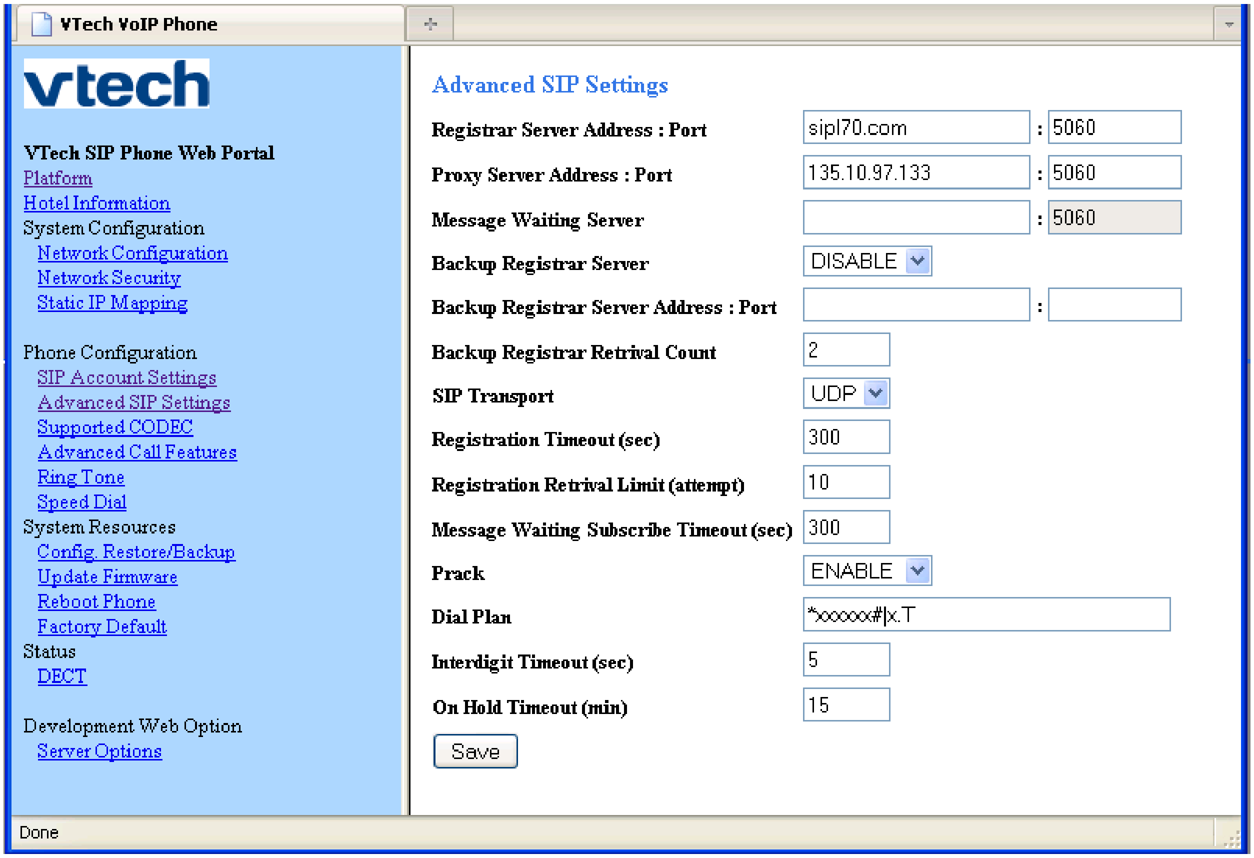


Figure 23: The Advanced SIP Settings of VTech S1220 phone

For every change on the VTech phone, the phone needs to be rebooted to take effect. To reboot the phone, click on the **Reboot Phone** link, the Reboot button appears on the right-hand side of the page as shown in **Figure 24**, click on the **Reboot** button and wait for 60 seconds until the page of the phone is refreshed and displayed again. The process of rebooting phone has been completed and the phone is able to use.

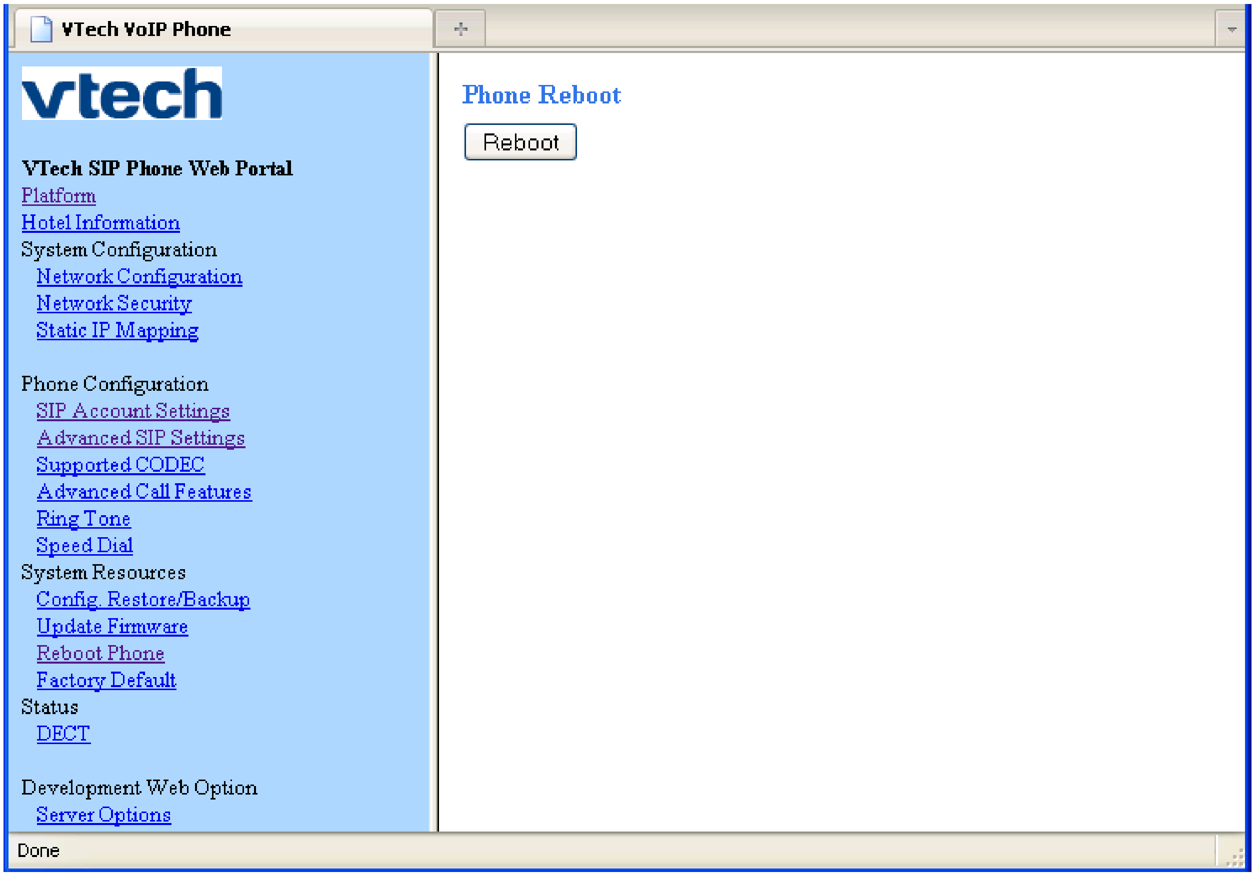


Figure 24: VTech S1220 Phone Reboot page

6.3. Local Call Forward Settings

This section shows how to configure "Local Call Forward" such as Call Forward All calls, Call forward busy and Call Forward No Answer on the VTech S1220 telephone.

On the homepage of VTech S1220 (see **Figure 21**), click on the **Advanced Call Features** link select the line in **Line Selection** page and the **Call Setting** page appears as shown in **Figure 25**. Select the call forward type in the **Call Forward Mode** field and enter the forward number in the **Call Forward Number** field.

Click the **Save** button to save the changes.

Note:

- The "Server Call Forward Always" is set for the VTech phone on the CS 1000 Call Server must be disabled, so that "Local Call Forward Always" on the VTech S1220 can be affected.
- The VTech S1220 telephone supports 3 types of call forward: Always, On Busy, and On No Answer.

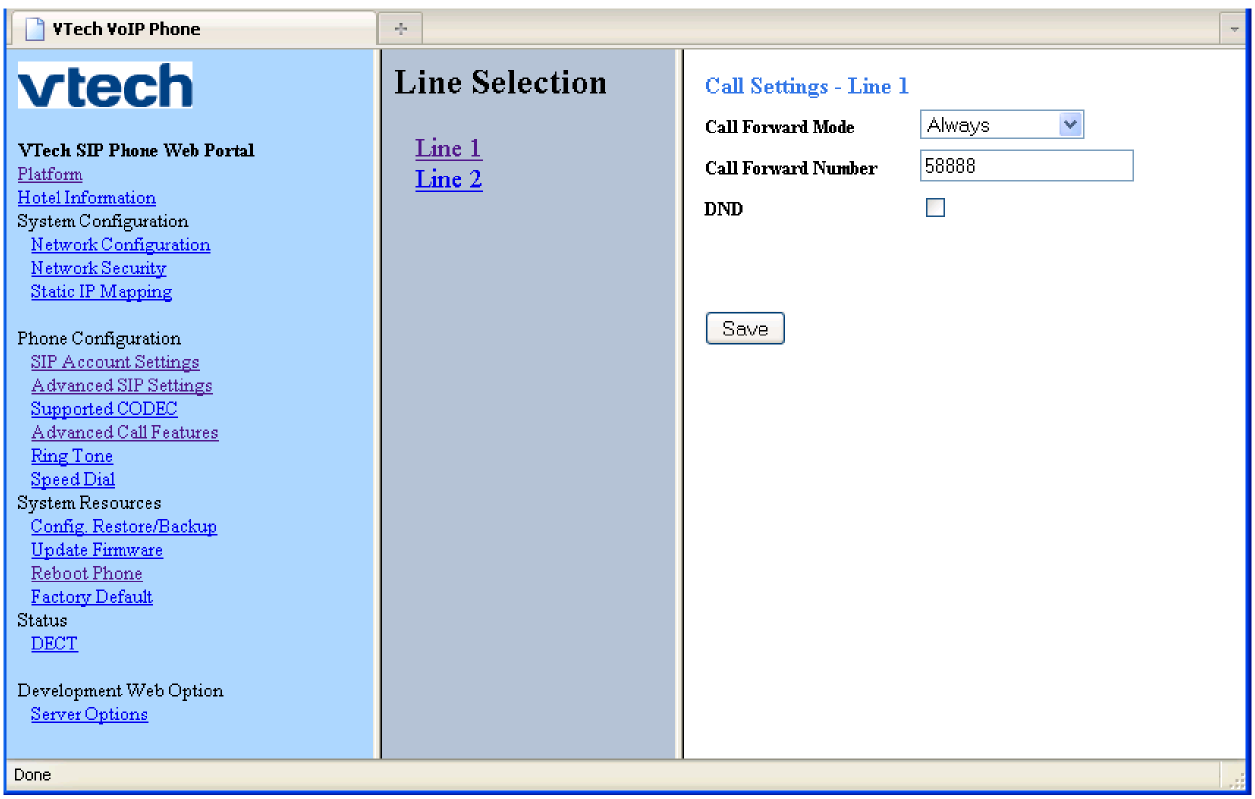


Figure 25: Call Settings section of VTech S1220 telephone

6.4. Codec settings

This section shows how to configure the Codec on the VTech S1220 phone.

On the homepage of VTech S1220 (see **Figure 21**), click on the **Supported CODEC** link and select the line in **Line Selection** page and the **Audio Setting - Line** page appears as shown in **Figure 26**. Click on the list supported audio codec in the **Audio Codec 1** field and select the desired codec for the first choice codec, repeat the same procedure for the Audio Codec 2, 3, and 4. Click on the **Save** button to save changes.

Note: It is recommended to have the audio codec G711u or G711a presented in one of 4 Audio Codec choices. For example, in case the audio codec G729 is selected as first choice in the Audio Codec 1, maintain the audio codec G711u and G711a in the second and third choice.

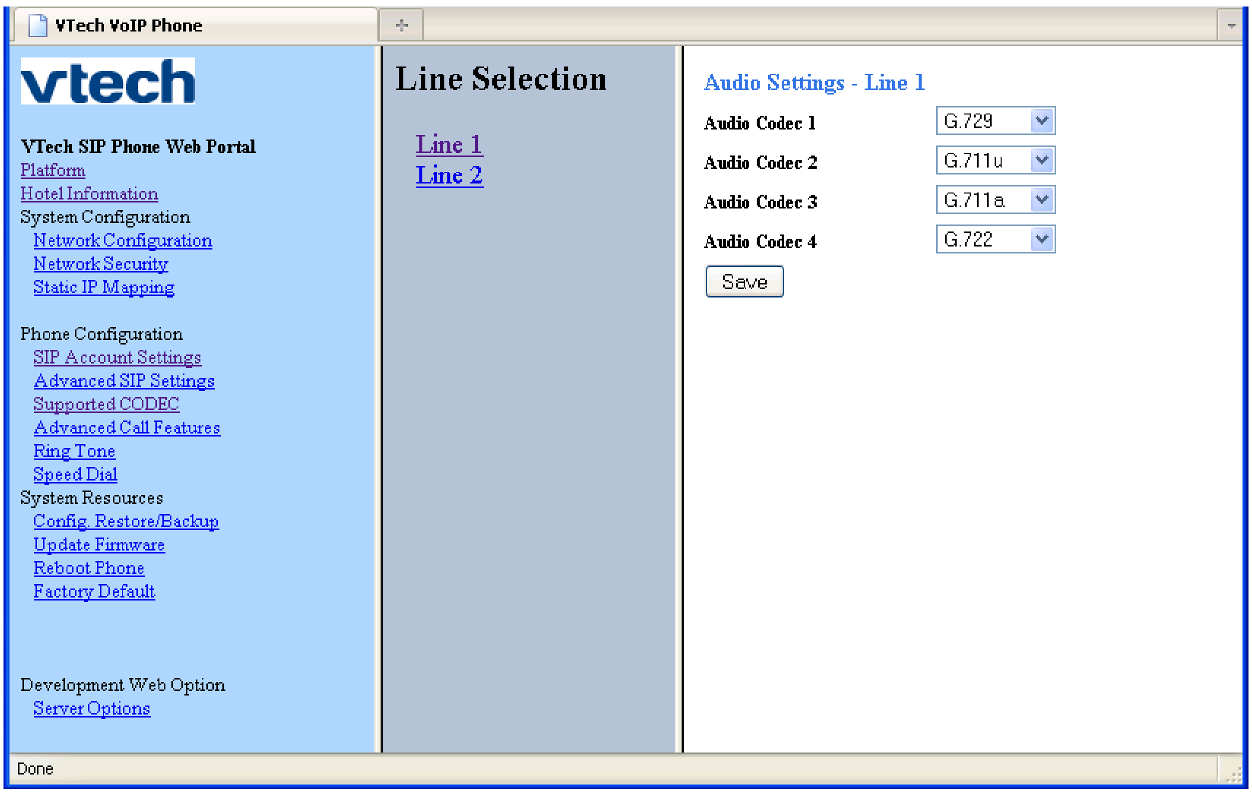


Figure 25: Audio Setting – Line of VTech S1220 phone

7. Verification Steps

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

- Verify that the VTech SIP Corded S1220 telephone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using the CS 1000 Linux command line and CS 1000 Call Server overlay LD 32.
    - Log in to the SIP Line server as an administrator.
    - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked. Figure below shows the detail of user 58004 as registered to the CS1000 SIP Line server.

```
admin@ucm1:~
[admin@ucm1 ~]$ slgSetShowByUID 58004

=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------   ---------
          58004       58004     096-01-00-04        1      0  0x94cb698           SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 135.10.98.24:5060:udp
         Type            = SIP3
         UserAgent       = Vtech/SIP_02.3.31.02
         x-nt-guid       = fbf86bd3ad4f41156b84f6e317935db6
         RegDescrip      =
         RegStatus       = 1
         PbxReason       = OK
         SipCode         = 200
         hTransc         = (nil)
         Expire          = 300
         Nonce           = 0e16d4efcf5ae0210dd71def410c3282
         NonceCount      = 5
         hTimer          = 0x94382a0
         TimeRemain      = 163
         Stale           = 0
         Outbound        = 0
         ClientGUID      = 0
         MSec CLS        = MSNV (MSEC-Never)
         Contact         = sip:58004@135.10.98.24:5060;line=e5a3a2d06dbaf0b
         KeyNum          = 255
         AutoAnswer      = NO

        Key  Func  Lamp  Label
        0    3     0     58004
        1    126   0     2758004
        4    22    0
        5    2     0     58010
        17   16    0
        18   18    0
```

    - Log in 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 1 0 4
IDLE REGISTERED 00
```

- Place a call from and to the VTech SIP S1220 telephone and verify that the call is established with 2-way speech path.
- During the call, use capture tool (ethereal/wireshark) to capture SIP packets at the SIP Line server and SIP phones 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 the **Section 2.1**, with some exceptions outlined in **Section 2.2**. The VTech Hospitality SIP Corded 2-Line S1220 version 02.3.31.02 is considered to be in compliance with Avaya Communication Server 1000 SIP Line System Release 7.5.

9. Additional References

Product documentation for the Avaya Communication Server 1000 products may be found at: https://support.avaya.com/css/Products/

Product documentation for the VTech Hospitality SIP Corded S1220 products may be found at: http://www.vtechhotelphones.com

[1] Avaya CS1000 Documents:
Avaya Communication Installation and Commissioning, Doc# NN43041-310, Issue 05.04, Date May 2011
Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals, Doc # NN43001-116, Issue 05.11, Date June 2011.
Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals, Doc # NN43001-509, Issue 03.02, Date June 2011
Avaya Communication Server 1000 Element Manager System Reference - Administration, Doc# NN43001-632, Issue 05.09, Date July 2011.
Avaya Communication Server 1000 SIP Line Fundamental, Doc# NN43001-508, Issue 03.03, Date November 2010

[2] VTech Hospitality SIP Corded Documents:
VTech SIP Cordless Series Master User Guide
VTech SIP Phone Configuration Guide

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.