Biamp Vocia VOIP-1 and Avaya Communication Server 1000 SIP Line Release 7.5

1. Configure Avaya CS 1000 - SIP LINE

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 appropriate Avaya documentation.

The following is the summary of tasks for configuring the CS 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service.in Customer Data Block.
- 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.

1.1. 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://support.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

1.2. 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.

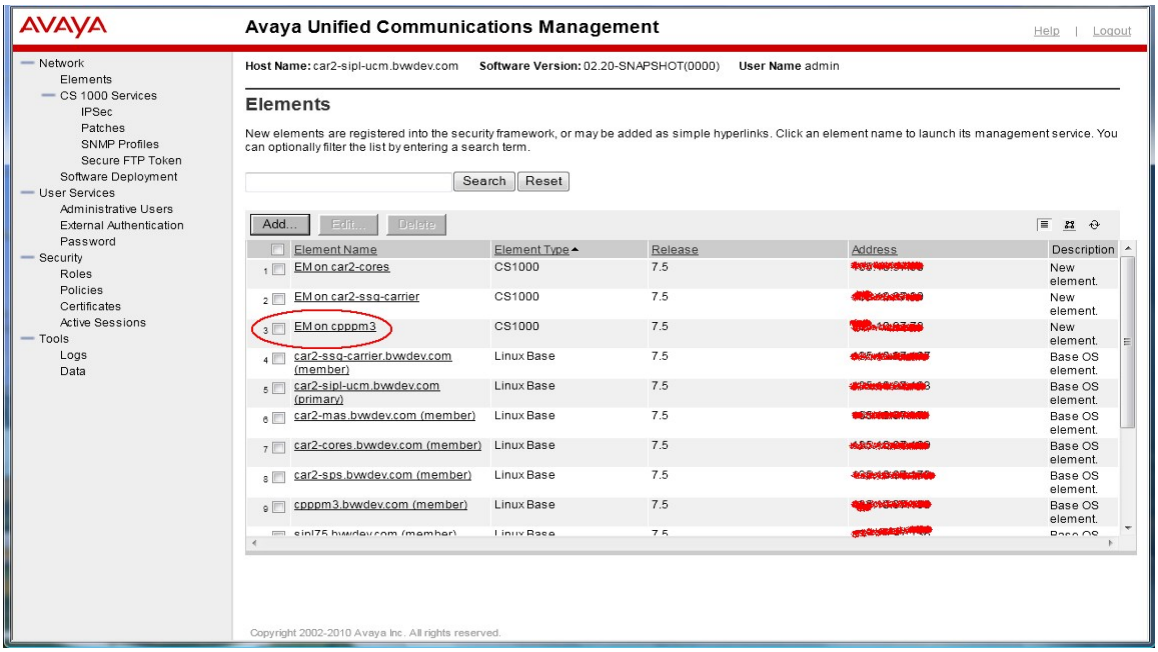

Figure 2: The UCM Home Page of CS 1000 Release 7.5

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** (see **Figure 2**). The CS 1000 Element Manager page appears as shown in **Figure 3** below.

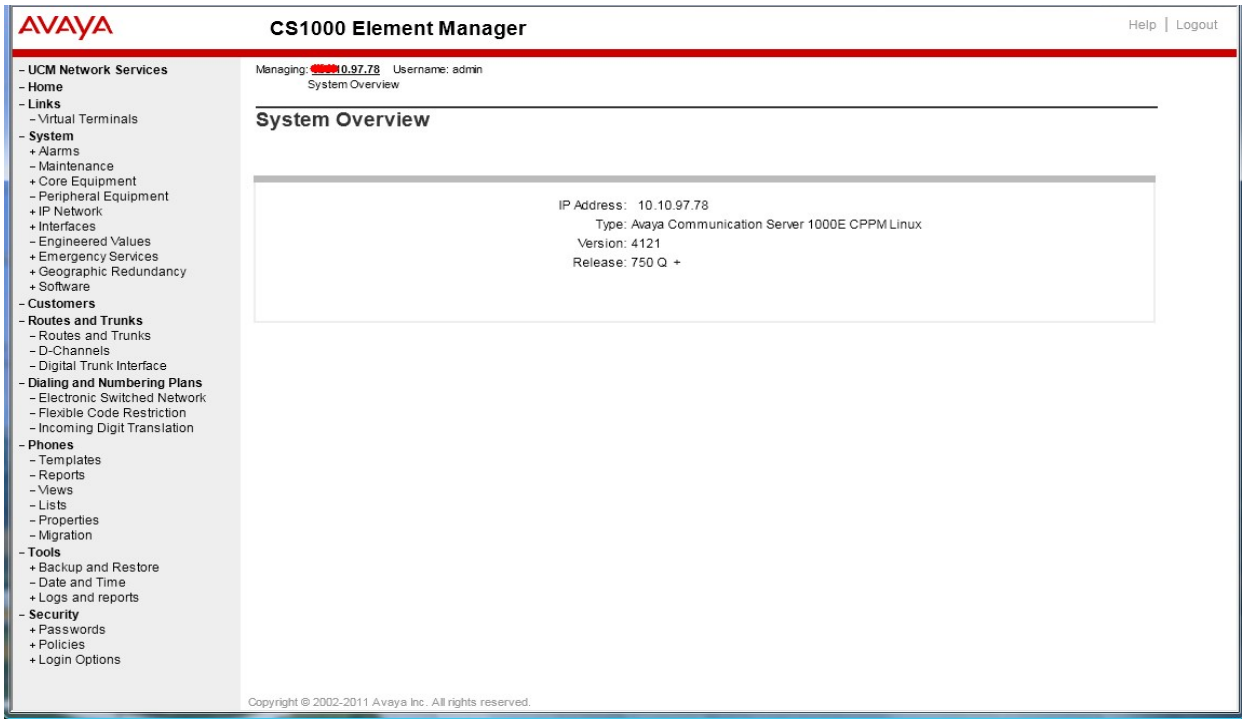

Figure 3: CS 1000 Release 7.5 EM Home Page

1.3. Enable SIP Line Service in 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**.

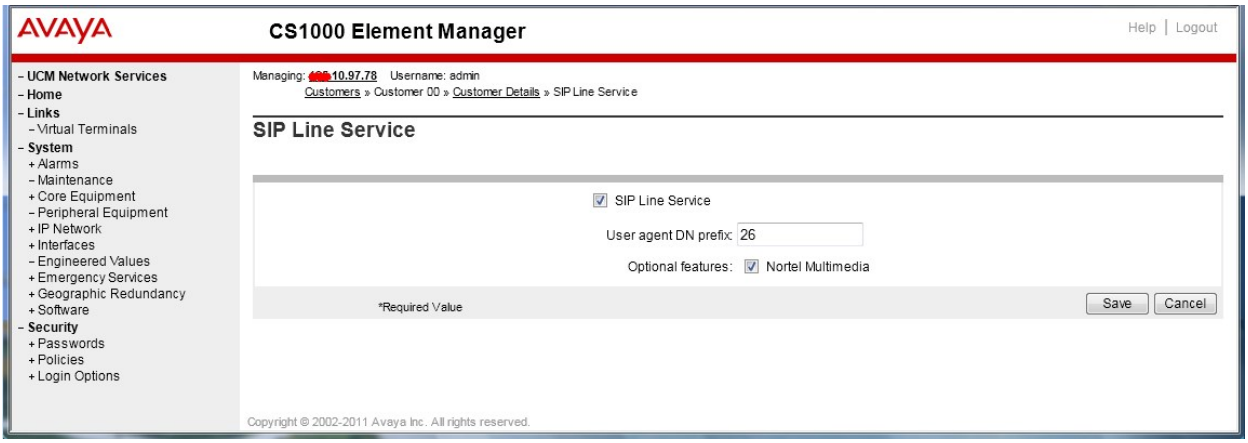

Figure 4: SIP Line Service in Customers Data Block

1.4. Create 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**: e.g., 512. This is the node ID of SIP Line server.
- **Call Server IP Address**: e.g. 10.10.97.78.
- **Node IP Address**: e.g. 10.10.97.187. This is the SIP Proxy IP address that SIP endpoint uses to register to.
- **Subnet Mask**: 255.255.255.192.
- **Embedded LAN (ELAN) Gateway IP Address**: e.g. 10.10.97.65.
- **Embedded LAN (ELAN) Subnet Mask**: 255.255.255.192.
- Check **SIP Line** check box to enable SIP Line for this Node.

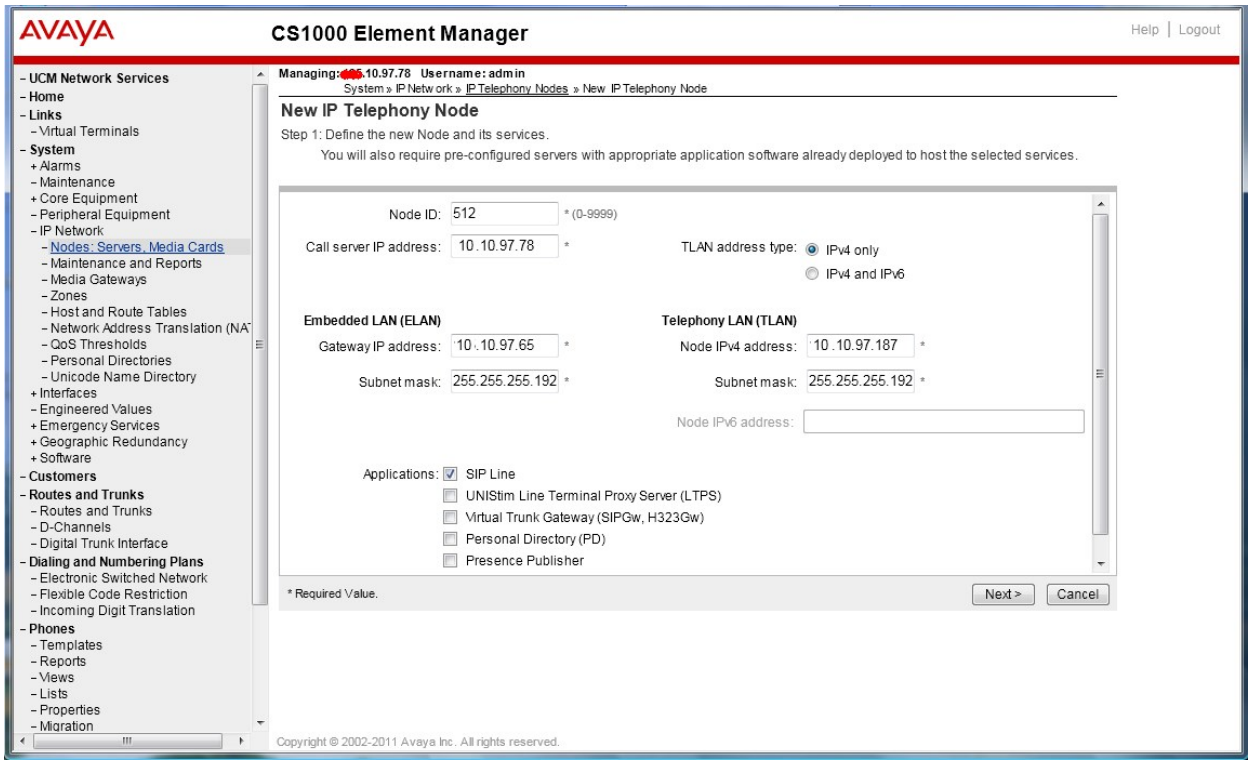

Figure 5: Adding a New IP Telephony Node

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**:

- Click **Select to Add** drop-down menu list to select the desired server to add to the node.
- Click **Add** button
- Select the check box next to the newly added server, and click **Make Leader** (not shown).

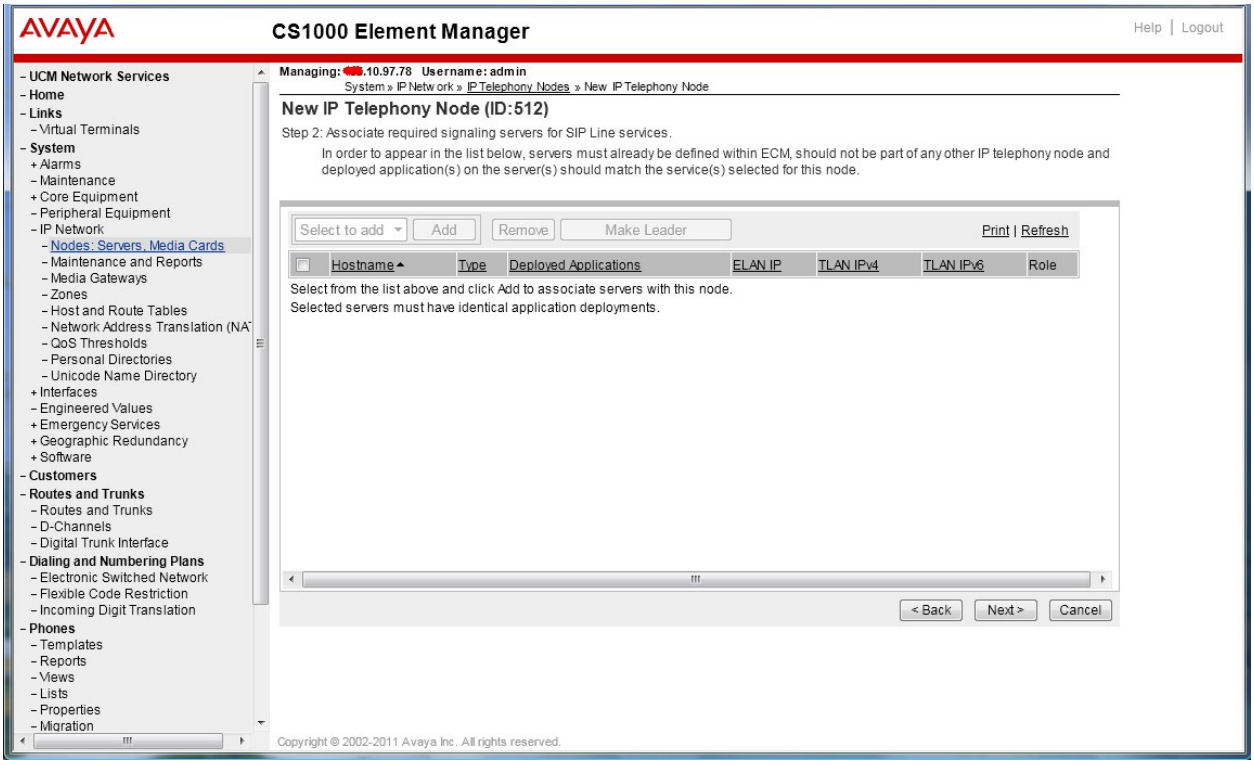

Figure 6: Adding a New IP Telephony Node (cont)

Click on the **Next** button to go to next page**.SIP Line Configuration Detail** page appears as shown in **Figure 7**:

- Enter SIP Line domain name in **SIP Domain name** text box, e.g., **sipl75.com**. **Note**: SIP domain name can be either a domain name or IP address, in this sample the SIP domain name is presented.
- Check the **Enable gateway service on this node** option for **SIP Line Gateway Application**.

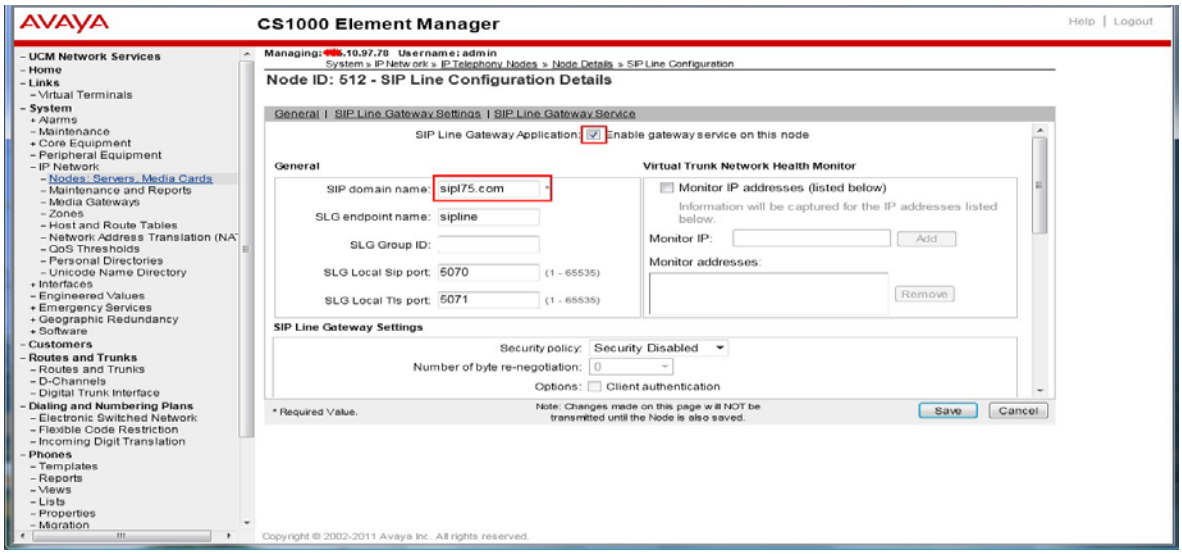

Figure 7: Adding a new IP Telephony Node (cont)

Scroll down to the **Branch / GR Office Settings** section under **SIP Line Gateway Service** as shown in Figure 8:

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

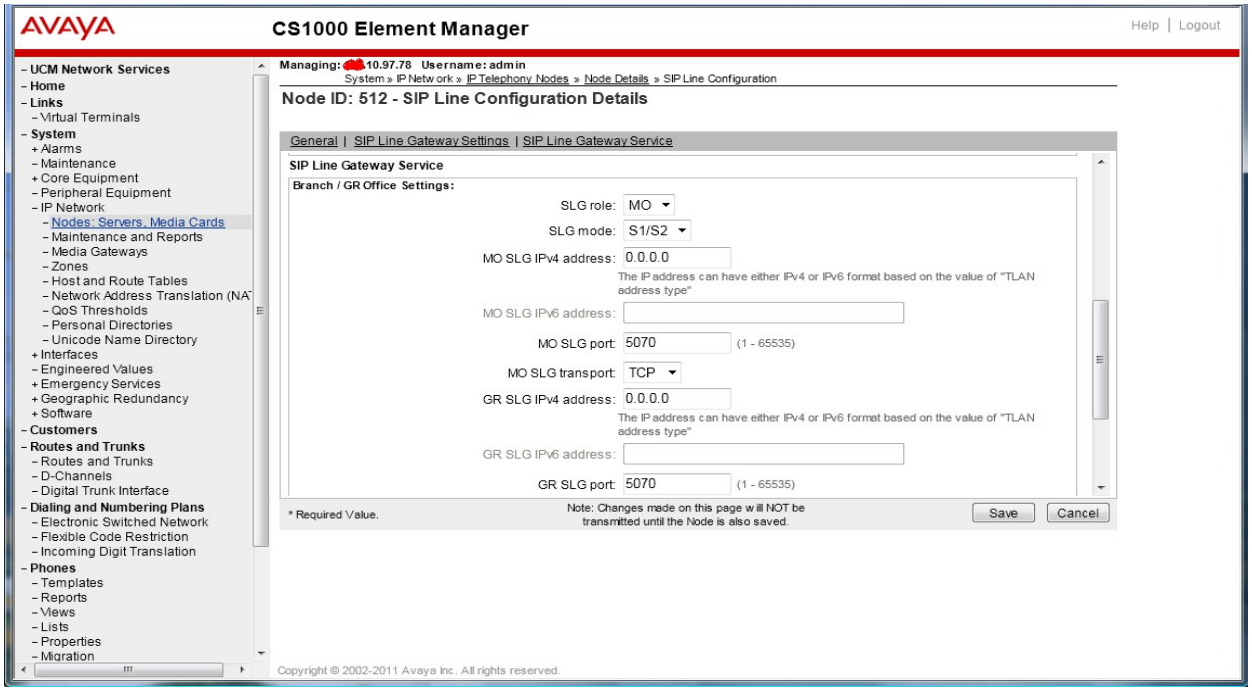

Figure 8: Adding a new IP Telephony Node (cont)

Click **Next**. The **Confirm new Node details** page appears (not shown).

Click on the **Transfer Now** button, the **Synchronize Configuration Files (Node ID 512)** page appears (not shown). Click **Finish** and wait for the configuration to be saved.

Node Saved page appears, as shown in **Figure 9**. Click the **Transfer Now** button.

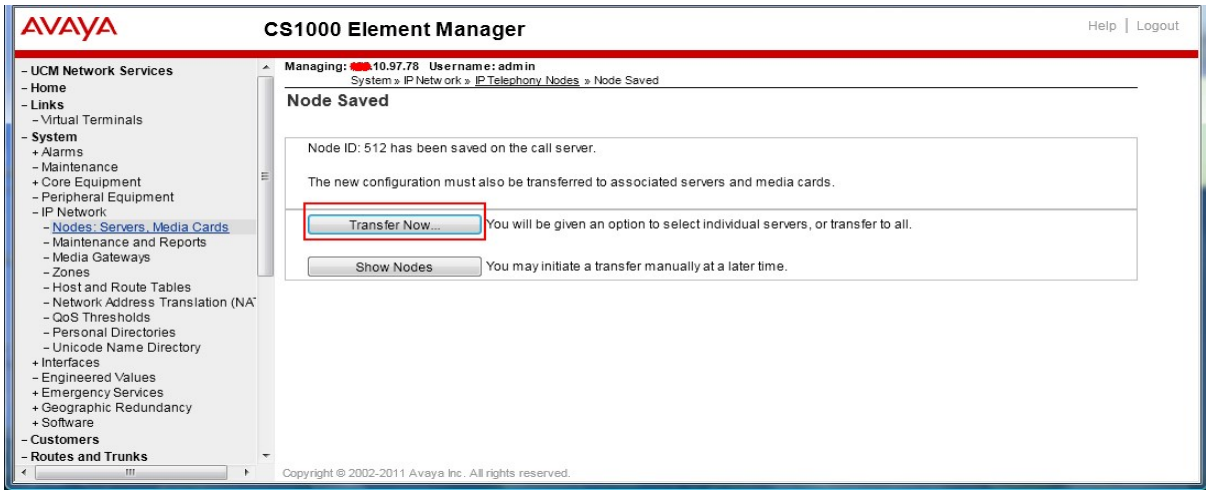

Figure 9: Node Saved with Transfer Configuration

Select the SIP Line server associated with changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers as shown in **Figure 10**.

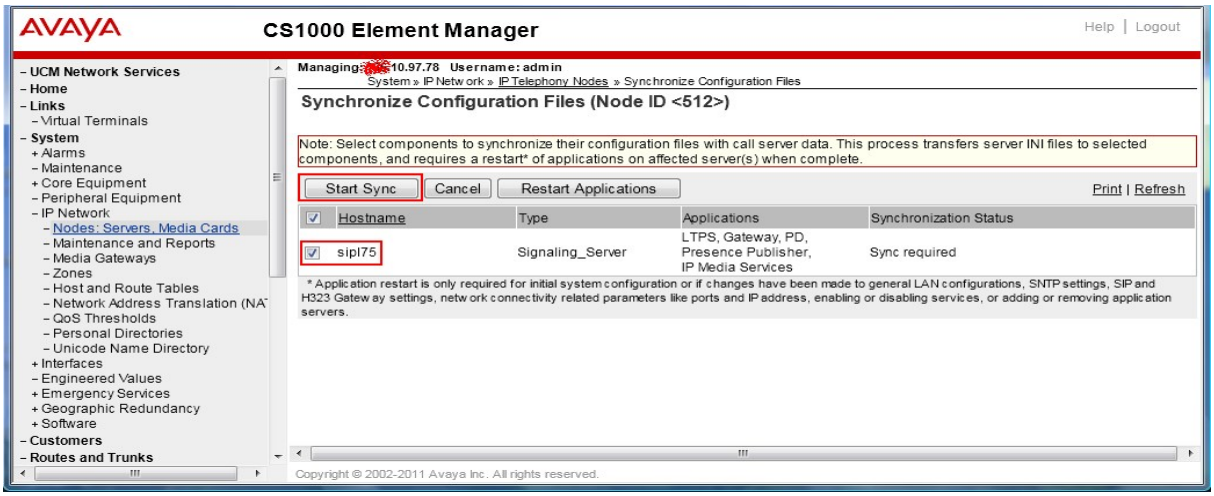

Figure 10: Synchronize Configuration Files

Note: 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 command: **appstart restart**.

1.5. Create D-Channel for SIP Line

On the EM page, navigate to **Routes and Trunks** → **D-Channels**.

Under the **Configuration** section as shown in **Figure 11**, enter a number in the **Choose a DChannel Number** field, and click on the **to Add** button.

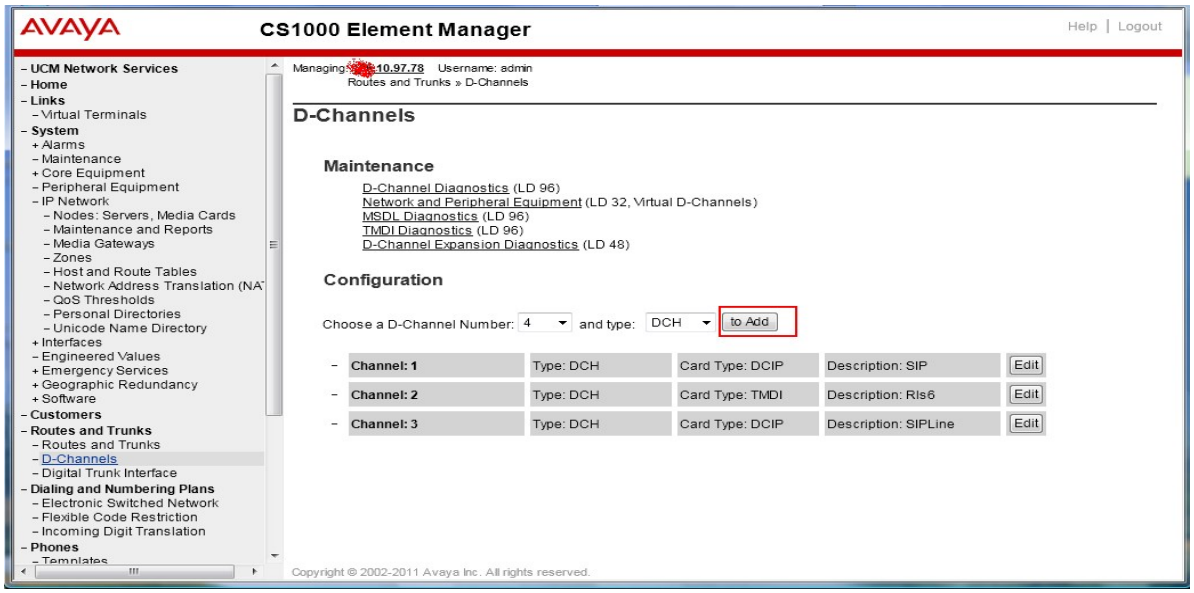

Figure 11: D-Channels configuration page

D-Channels xx Property Configuration page appears as shown in **Figure 12** where **xx** is the D-Channel number to add.

- From the **Interface type for D-channel (IFC)** list, select **Meridian Meridian1 (SL1)**. - Leave the other fields at default values.

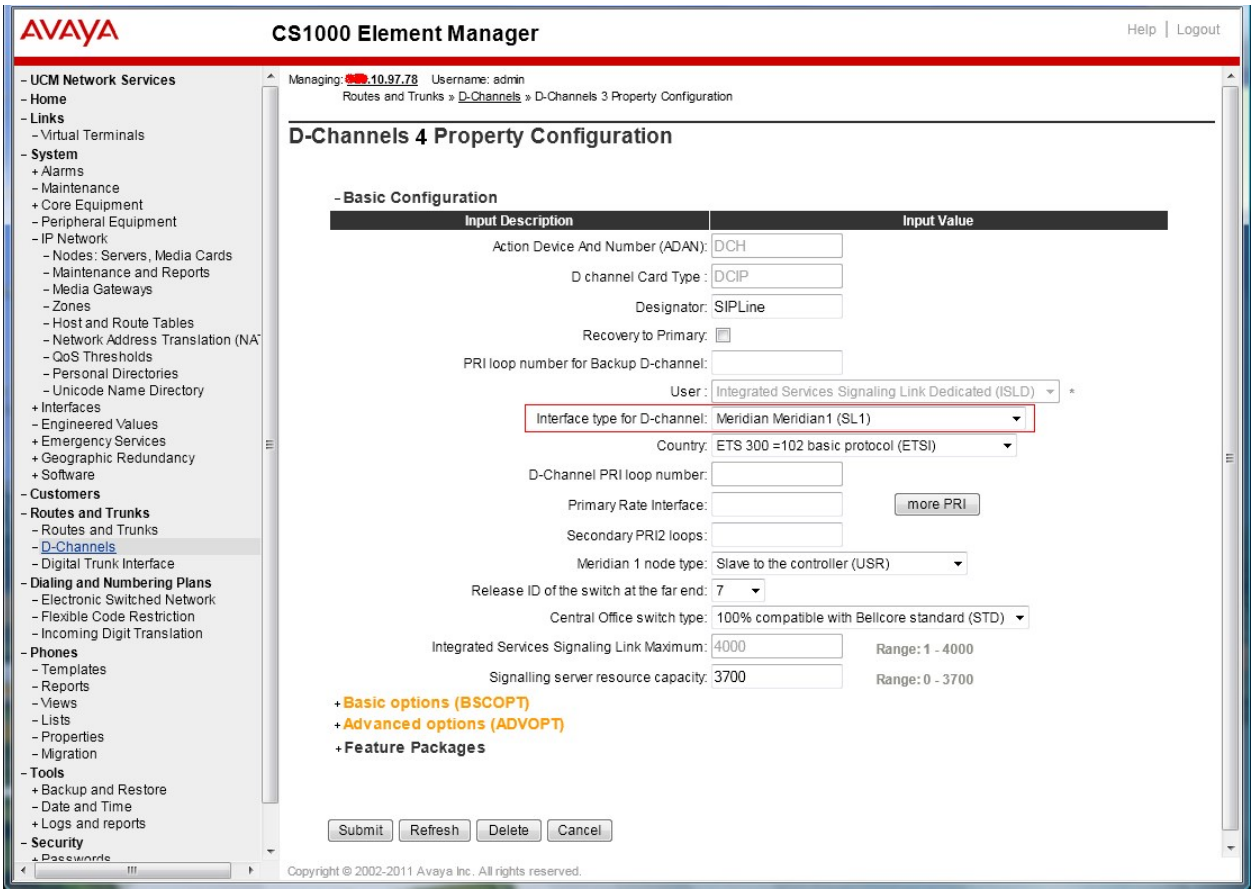

Figure 12: SIP Line D-Channel Property Configuration

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 Details** page will appear as shown in **Figure 13**.

- Select the **Message waiting interworking with DMS-100 (MWI*)*** check box. This option must be enabled to support voice mail notification on SIP Line endpoints.
- Select the **Network name display method 2 (ND2)** check box. This option must be enabled to support name display between SIP Line endpoints.
- Other check boxes are left unchecked.
- At the bottom of the **Remote Capabilities Configuration Details** page, click **Return - Remote Capabilities** to return to the **D-Channel xx Property Configuration** page.

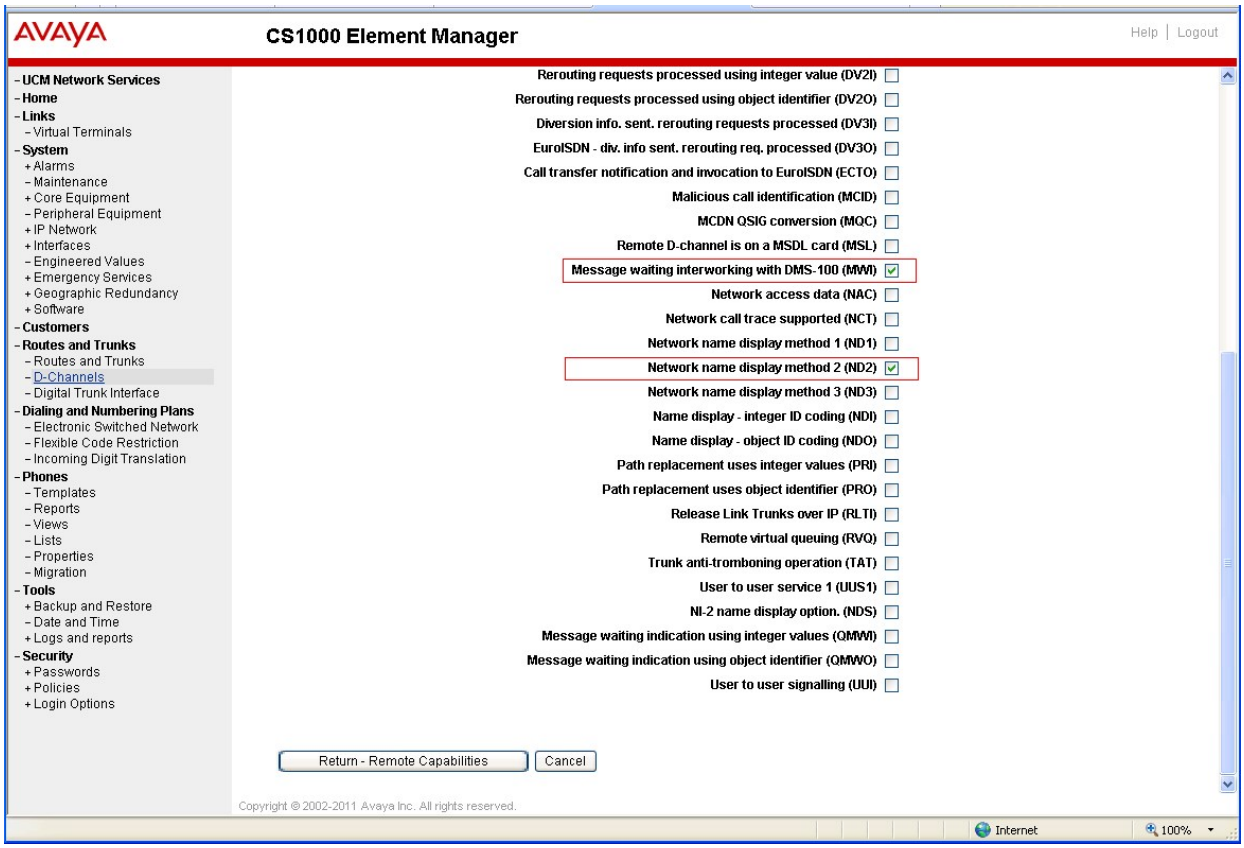

Figure 13: SIP Line D-Channel RCAP Configuration Details

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

1.6. Create an Application Module Link (AML)

On the EM page, navigate to **System** ☐**nterfaces** ☐**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.

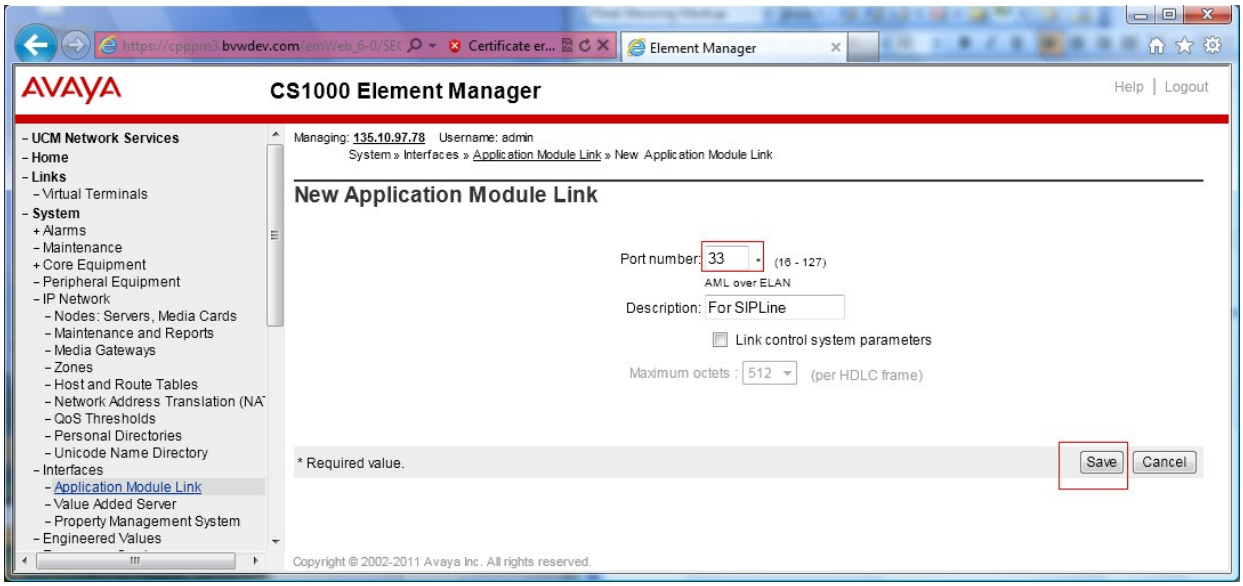

Figure 14: Adding a new Application Module Link

1.7. Create a Value Added Server (VAS)

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

The **Value Added Server** page appears (not shown). On this page, select the **Ethernet 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 **Section 1.6**.

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

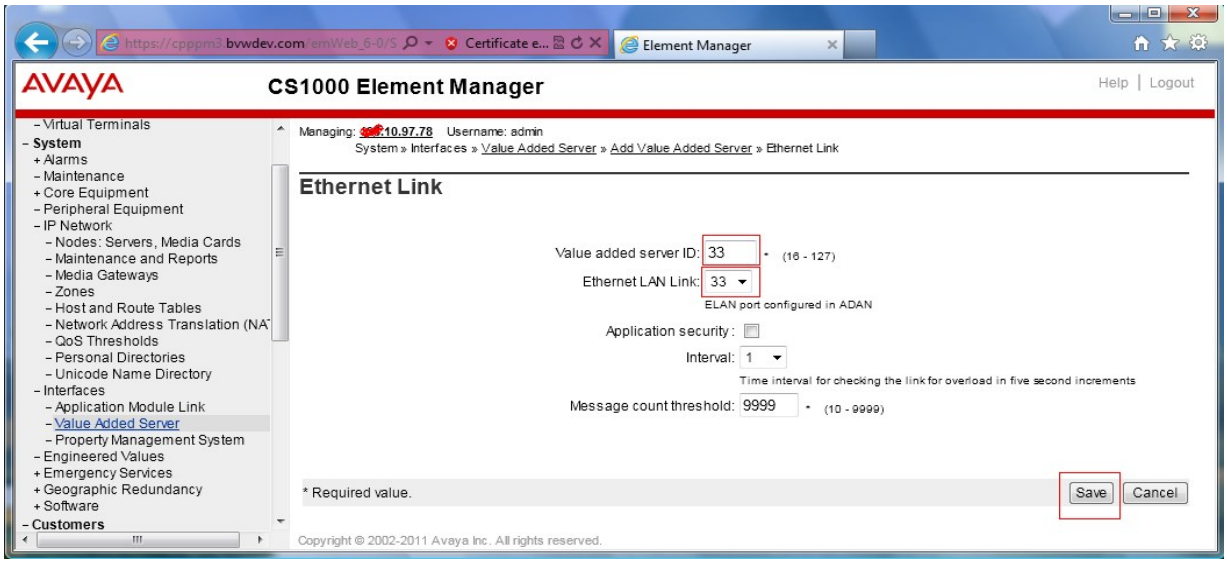

Figure 15: Adding a new Value Added Service for the AML

1.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System** ☐ **IP Network** ☐ **Zones**. The **Zones** page appears on the right (not shown). On this page select **Bandwidth Zones** link.

On the **Bandwidth Zones** page (not shown), 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. In the **Zone Intent (ZBRN)** drop-down menu select **VTRK (VTRK)**.Leave other fields at 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 (MO)**, instead of **VTRK** in the field **Zone Intent**.

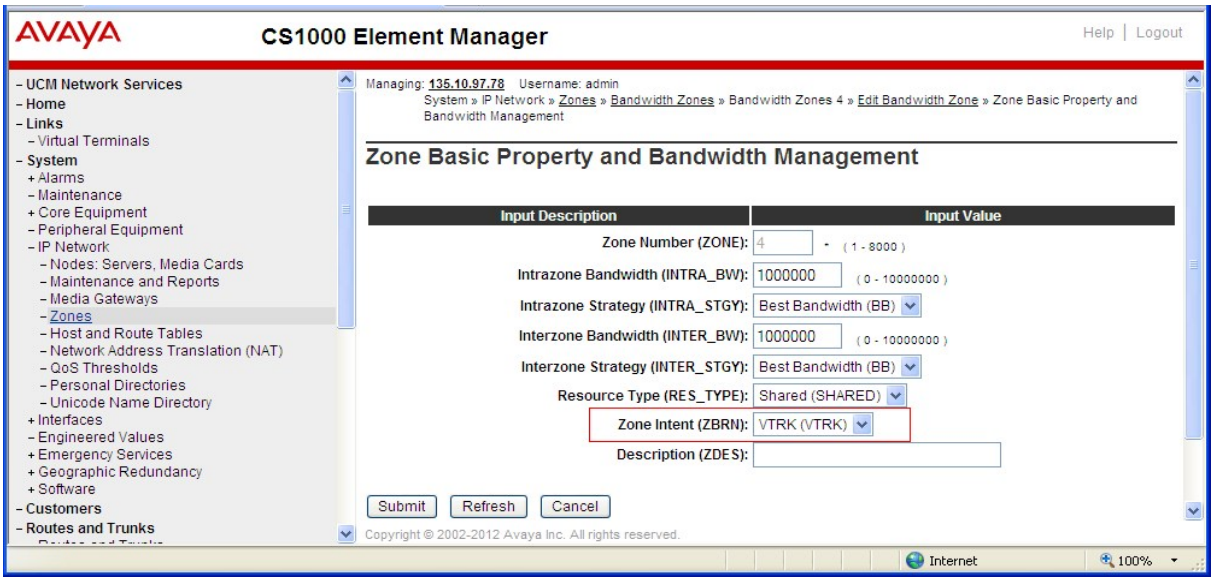

Figure 16: Adding a new Zone for Virtual Trunk

1.9. Create a 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). On 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 as shown in **Figure 17** and **18**.

- **Route Number (ROUT)**: 3
- **Trunk type(TKTP)**: TIE
- **Incoming and Outgoing trunk (ICOG)**: Incoming and Outgoing (IAO)
- **Access code for the trunk route (ACOD)**: a number for ACOD, for example 8003.
- **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 created in the **Section 1.8**.
- **Node ID of signaling server of this route (NODE)**: 512, this is the node ID of the SIP Line Telephony Node created in **Section 1.4**.
- **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 5.5**.
- **Interface type for route (IFC)**: Meridian M1 (SL1).
- **Network calling name allowed (NCNA)**: checked.
- **Channel type (CHTY)**: B-channel (BCH).
- **Call type for outgoing direct dialed TIE route (CTYP)**: Coordinate dialing plan (CDP).
- **Calling Number dialing plan (CNDP)**: Coordinate dialing plan (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.

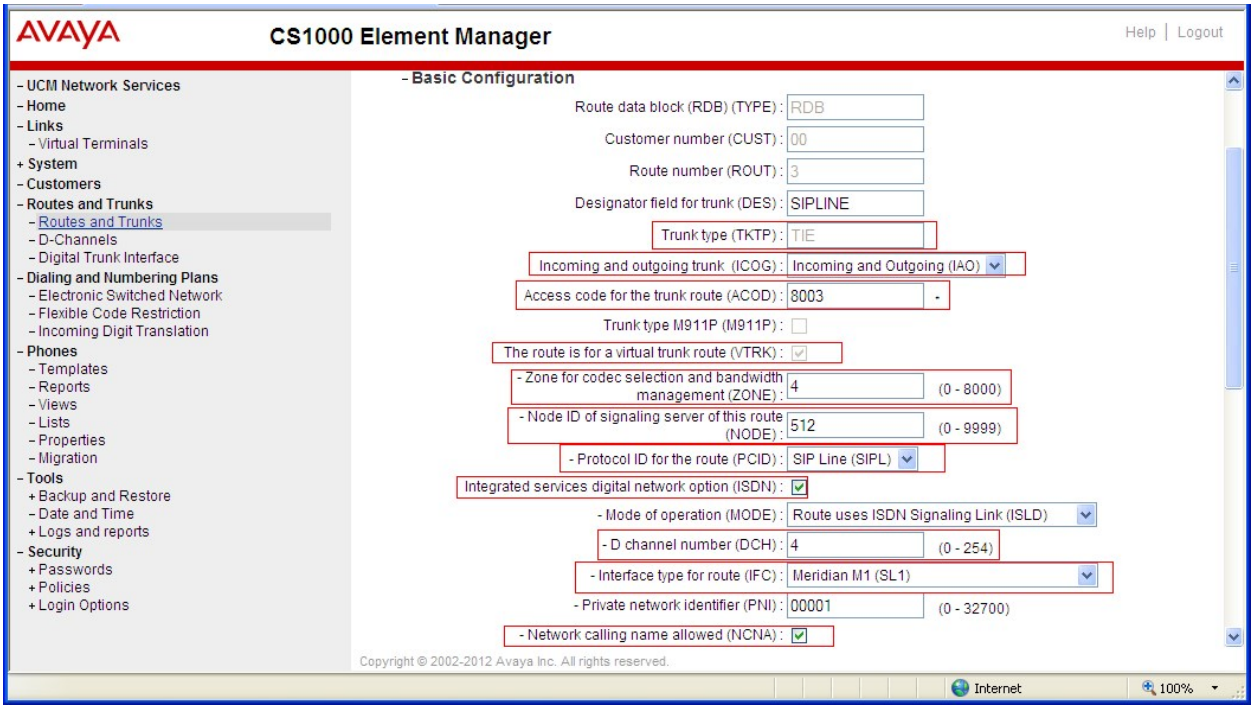

Figure 17: SIP Line Route Basic Configuration

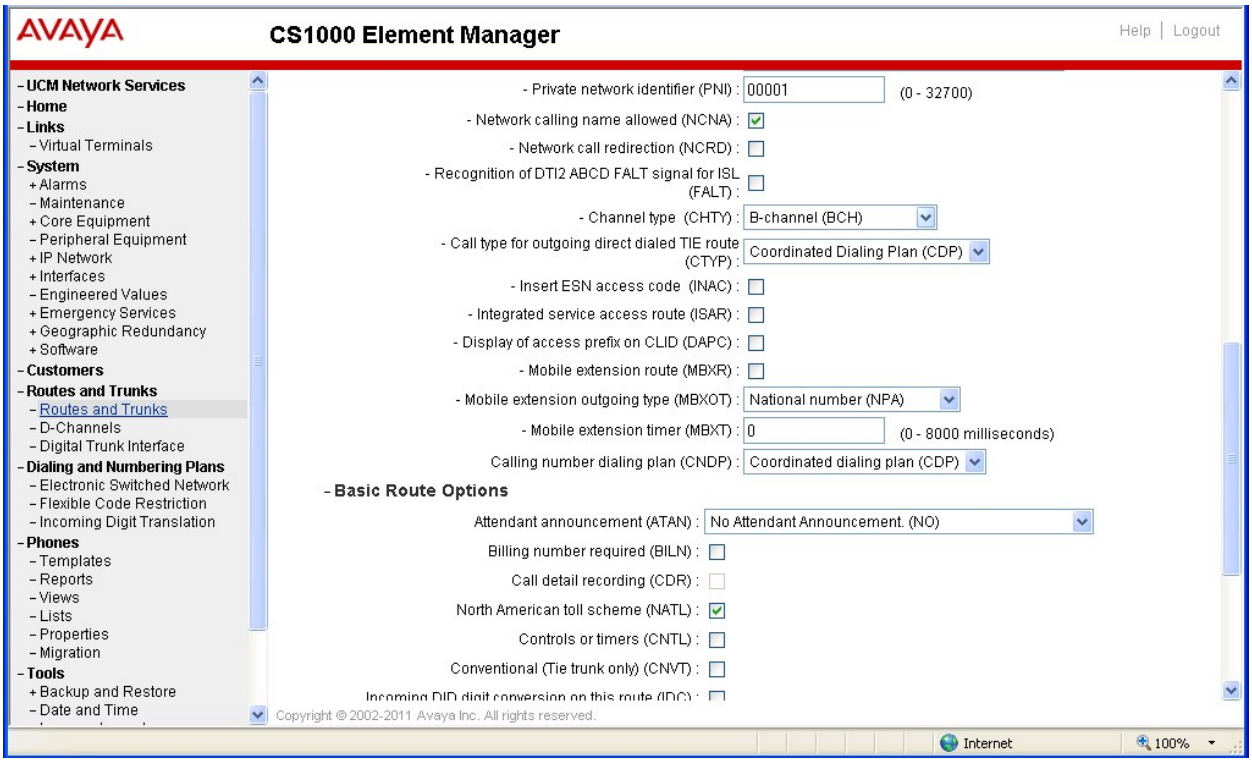

Figure 18: SIP Line Route Basic Configuration (cont)

1.10. Create SIP Line Virtual Trunks

On the EM page, navigate to **Routes and Trunks** ☐**Routes and Trunks** and select the **Add trunk** button beside the route created in the **Section 1.9** to create new trunks.

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

- **Multiple trunk input number**: 32 for creating 32 trunks.
- **Auto increment member number**: checked.
- **Trunk data block**: IP Trunk (IPTI).
- **Terminal Number (TN)**: 100 0 2 0, the first TN of a TN range.
- **Member number**: 33, ID for first trunk; next ID's will be automatically created.
- **Start arrangement Incoming**: Immediate (IMM). ☐**Start arrangement Outgoing**: Immediate (IMM).
- **Trunk Group Access Restriction**: 1.
- **Channel ID for this trunk**: 33, this ID should be the same as ID of **Member number**.

Click on the **Edit** button for **Class of Service** 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 **Customer ID, Route ID, Trunk type TIE trunk data block** page, then click on the **Save** button to complete adding virtual trunks for SIP Line.

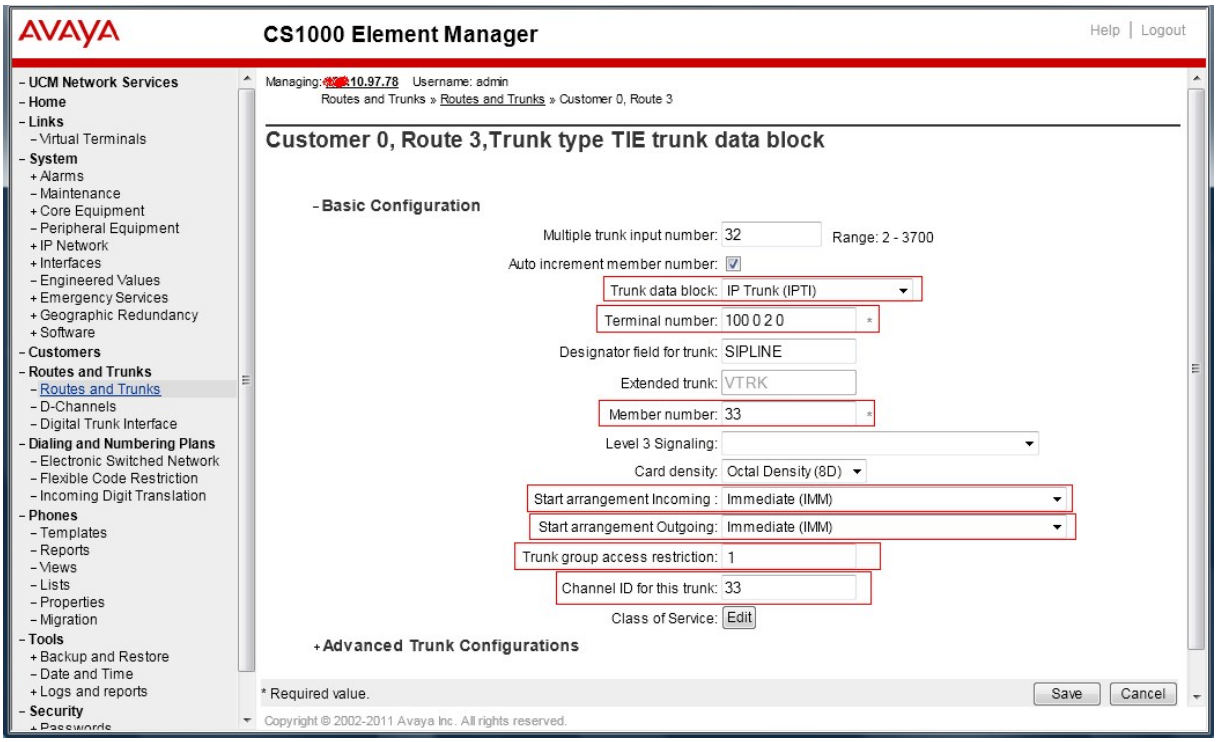

Figure 19: Adding virtual trunks for SIP Line Trunk

1.11. Create SIP Line Phones

To create a SIP Line phone on the Call Server, log in as administrator and use overlay command **LD 20** as shown below.

The bold fields must be properly inputted as they are configured on the Call server, for other fields press <Enter> to leave it at default values.

```
>LD 20

PT0000
REQ: NEW
TYPE: UEXT
TN   104 0 0 20 ==> Terminal Number of SIP Endpoint
DES  BIAMP
CUST 0
UXTY SIPL
MCCL YES
SIPN 0
SIP3 1       ==> enter 1 to enable for 3rd SIP endpoint
SIPU 54350 ==> SIP username, it can be either name or number
NDID 512    ==> SIPLine Node ID that configured in Section 5.4
ZONE 1       ==> Zone ID for SIPLine phone that configured in Section 5.8
SCPW 1234 ==> The password that is used to log in from SIP endpoint
CLS  dnda cnda swa ==> Class of services, SWA is for Call Waiting
HUNT
KEY 0 scr 54356 => Directory number (DN) of SIP endpoint.
 NOTE::UADN FORMED FROM UAPR IS:2654356
  MARP
  CPND new
    CPND_LANG
    NAME BIAMP SIP 54356
    XPLN
    DISPLAY_FMT
  VMB
KEY 1 hot u 2654356 ==> HOT U has to be followed format like: Prefix + DN with the
                                                    prefix configured in Section 5.3
  MARP
KEY 2 CWT ==> Key 2 for Call Waiting
```

Figure 20: Sample of creating a new SIP Line Phone

2. Configure Biamp Vocia VOIP-1

Biamp installs, configures, and customizes the Vocia VOIP-1 application for their end customers. How to configure a Vocia system is beyond the scope of these application notes. This section only provides steps to configure Biamp Vocia VOIP-1 to interface with Avaya CS 1000 SIP Line server. For more information on how to administer Biamp Vocia VOIP-1, please refer to the Vocia VOIP-1 documents from Biamp .

Select the Vocia icon from Workstation where Vocia software was installed to start Vocia software.

- Double-click on the **VoIP-1** entry in the configuration, as shown below.

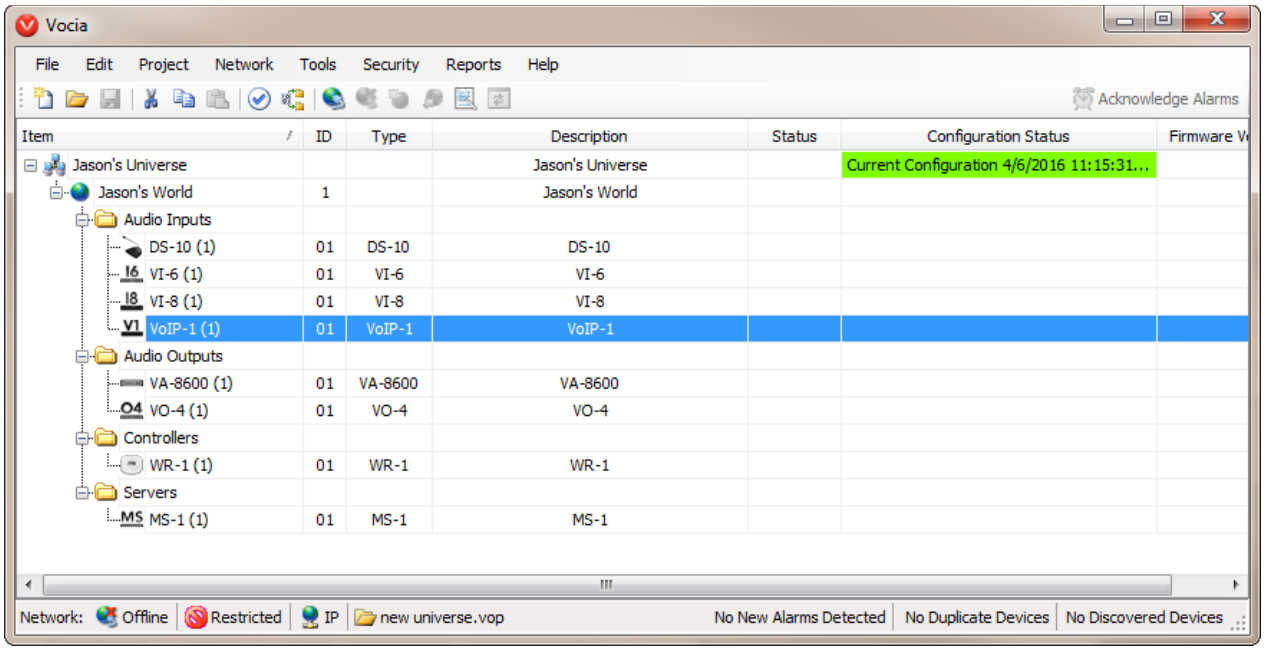

Figure 21: The main window of Vocia Software

Select the **VoIP Network** icon followed by the **Advanced VoIP Card Options** button to open the **Advanced VoIP Card Options** dialog.

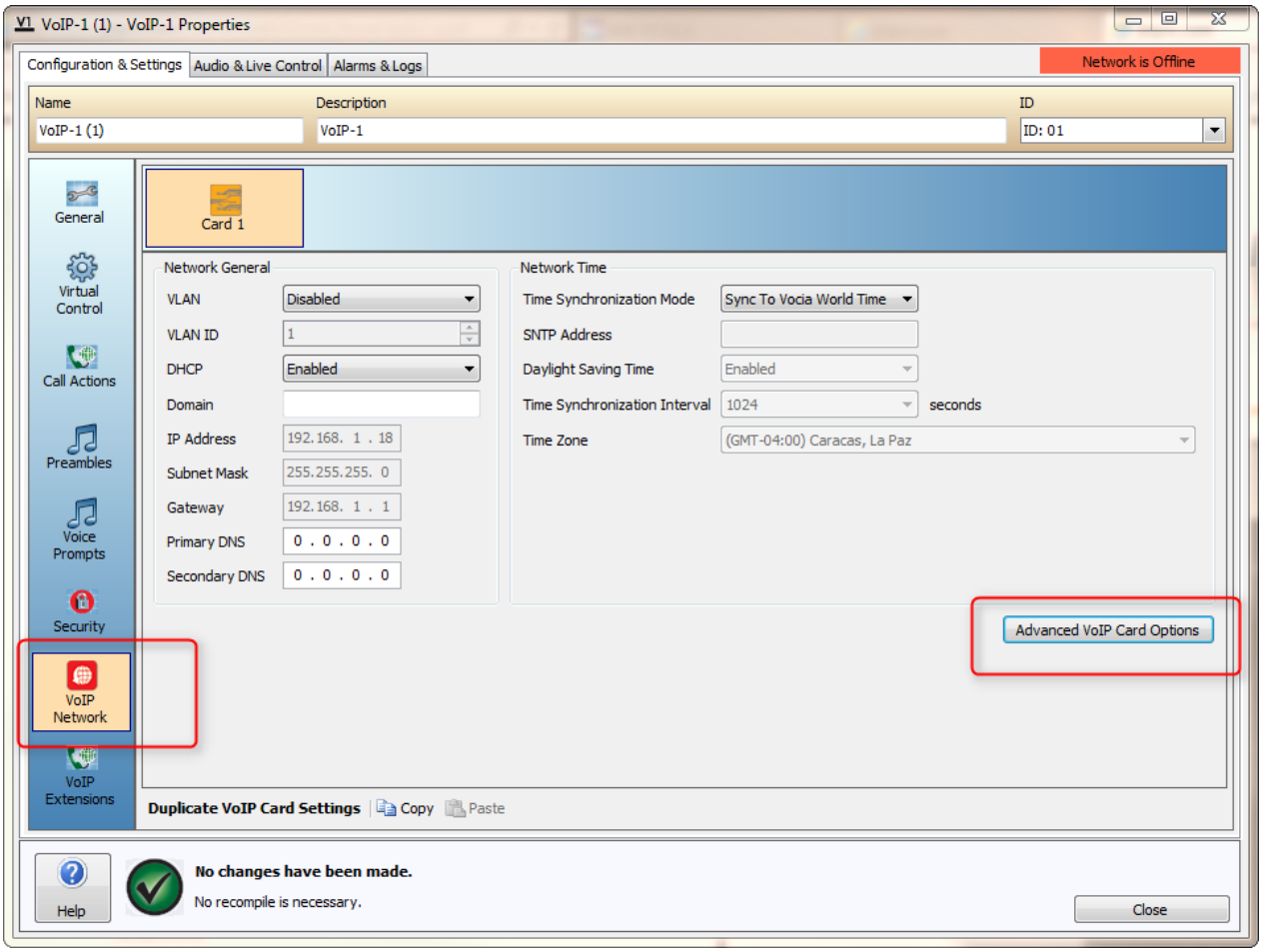

Figure 22: Properties window of VoIP Control

- Select the **Transport** pull-down selection box to configure the transport to be used. Select UDP from the drop-down selection box.
-

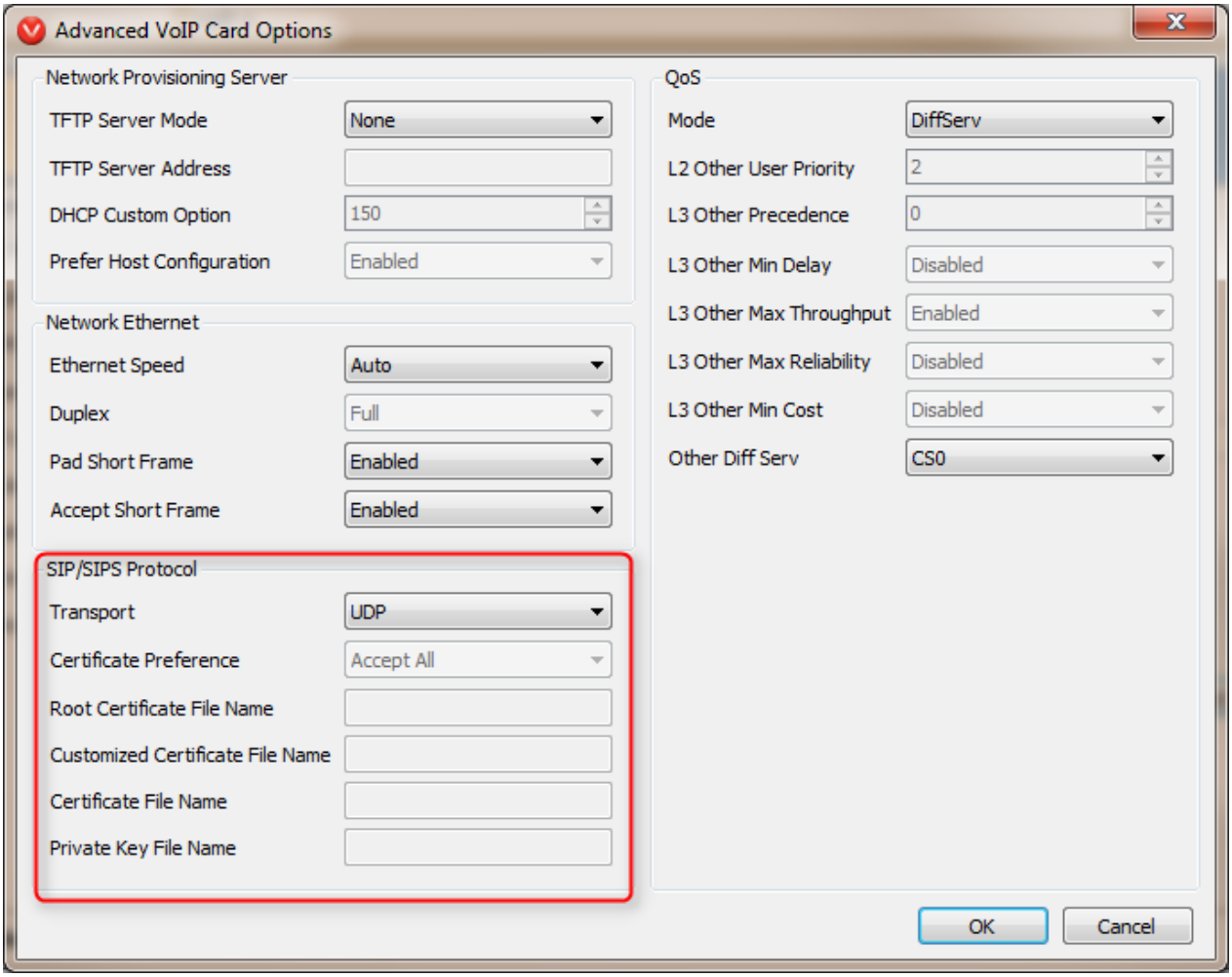

Figure 23: Properties window with UDP Transport

Click **OK** to return to the VoIP-1 Properties page.

Select the **VoIP Extensions** icon as shown below.

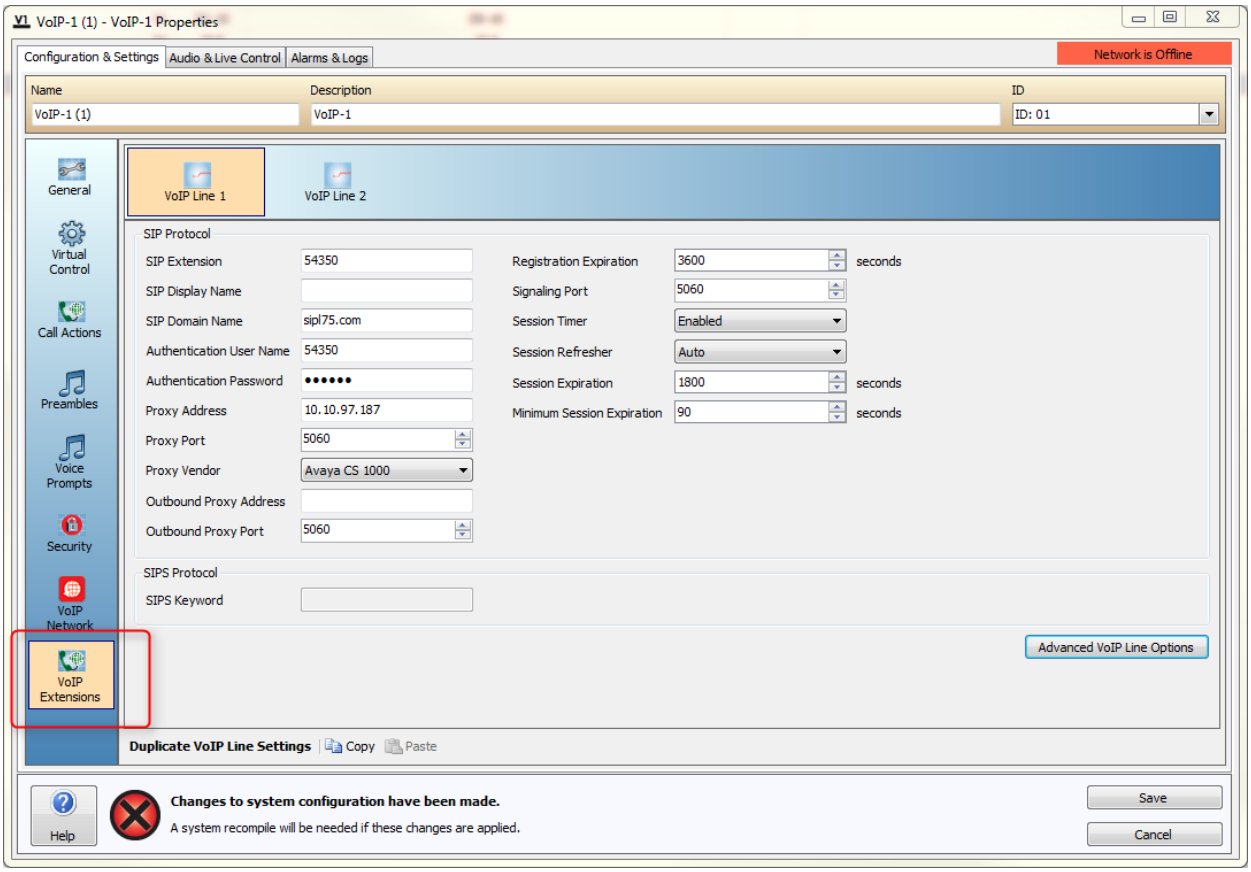

Figure 24: VoIP Extension Properties

Select either **VoIP Line 1** or **VoIP line 2** at the top of the window to configure each individual line.

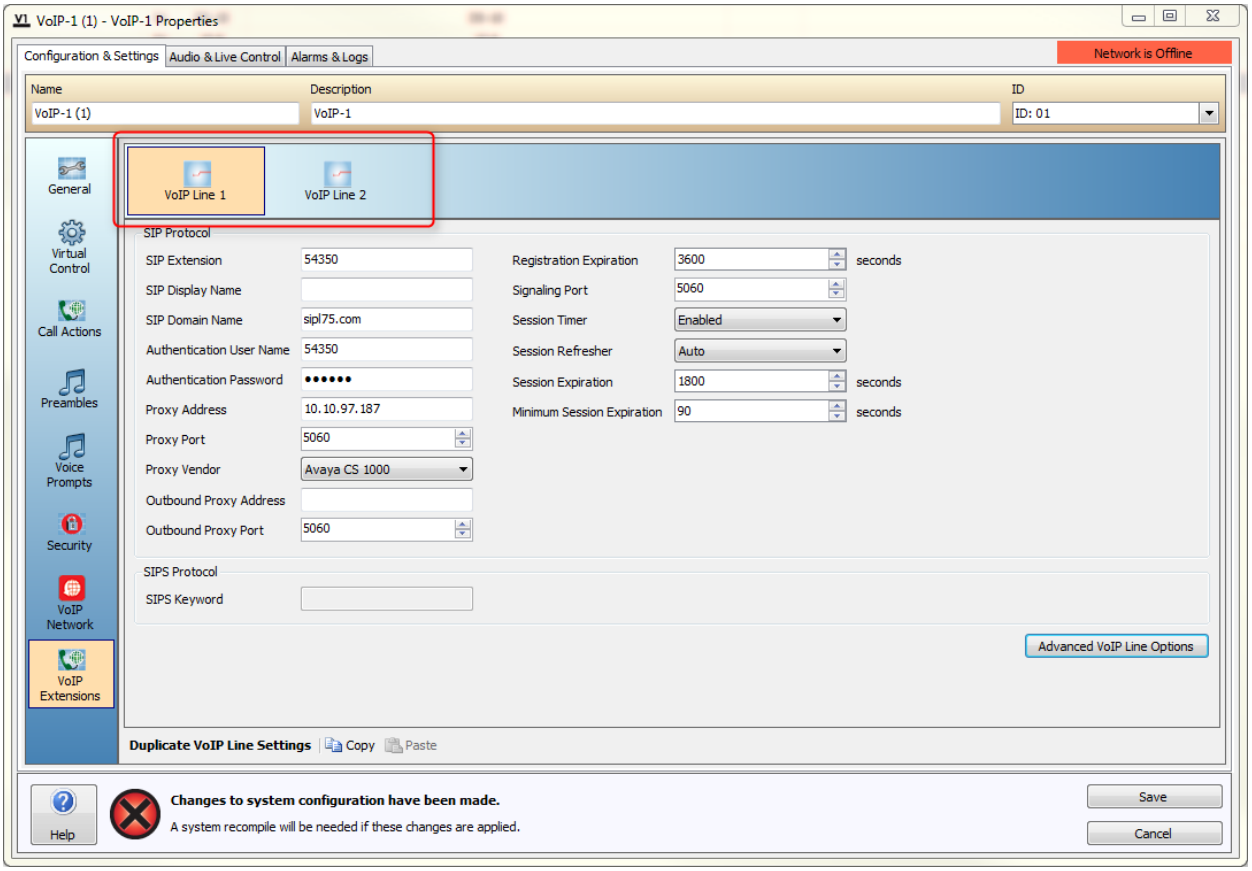

Figure 25: VoIP Line Properties Window

Note: *Biamp Vocia VOIP-1 can provide two inbound extensions (L1 and L2). Each line must be configured separately.*

In the **Protocol** tab, provide the following information:

- **SIP Extension** – Enter a user created in **Section 1.11**.
- **SIP Domain Name** – Enter the SIP domain name configured in **Section 1.4**
- **Authentication User Name** – Enter a user created in **Section 1.11**.
- **Authentication Password** – Enter the password for the user in **Section 1.11**
- **Proxy Address** – Enter the IP address of CS 1000 SIP Line server.
- **Proxy Port** – Enter either 5060 or 5061.
    - TLS – 5061
    - UDP – 5060
- **Proxy Vendor** – Select Avaya CS 1000.
- Click on the **OK** button. Default values may be used for all other fields.

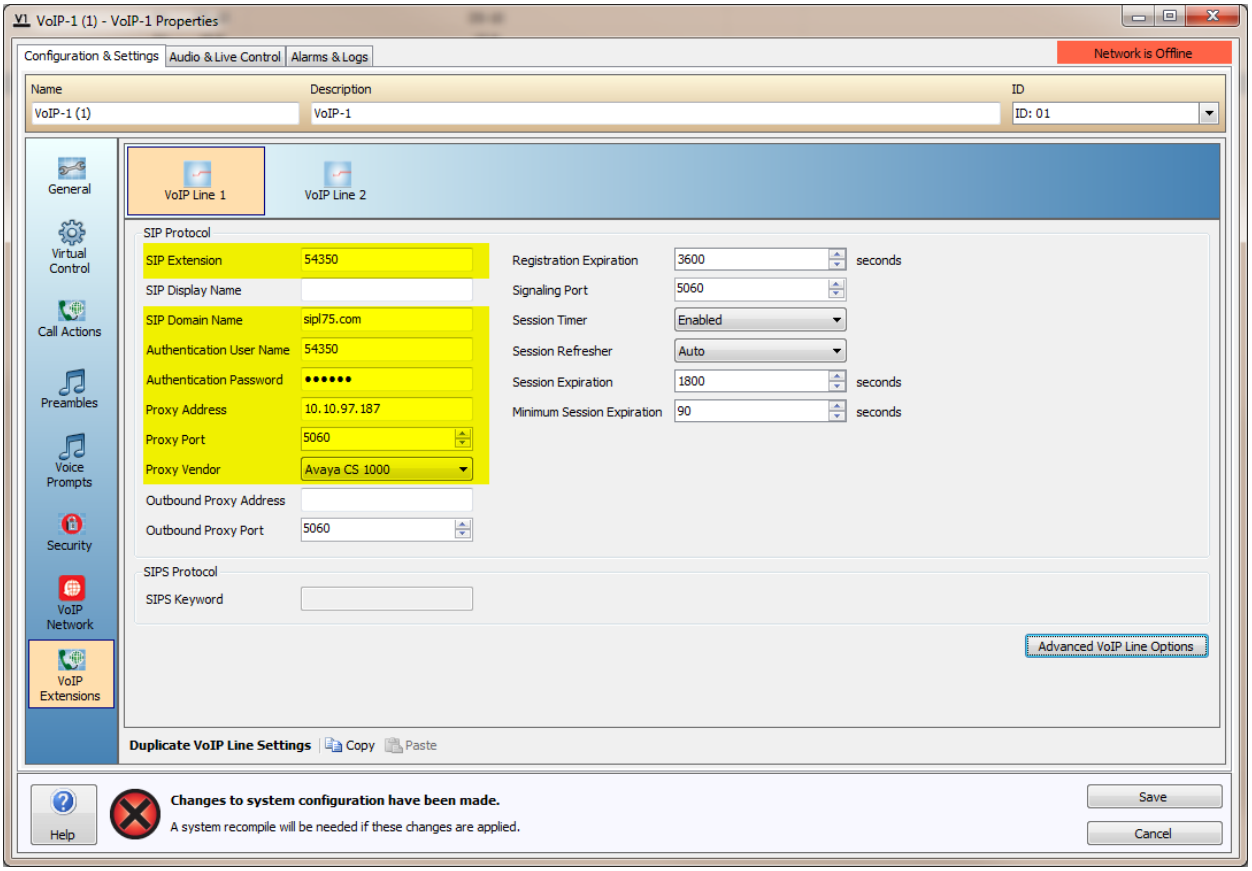

Figure 26: Protocol tab of VoIP Line Properties

***Note**: in case the SIP Line domain name in **Section 1.4** is configured to use IP address instead of SIP domain name, the registration of Biamp Vocia SIP user is configured as in **Figure 28** below. The **SIP Domain Name** field is left as blank.*

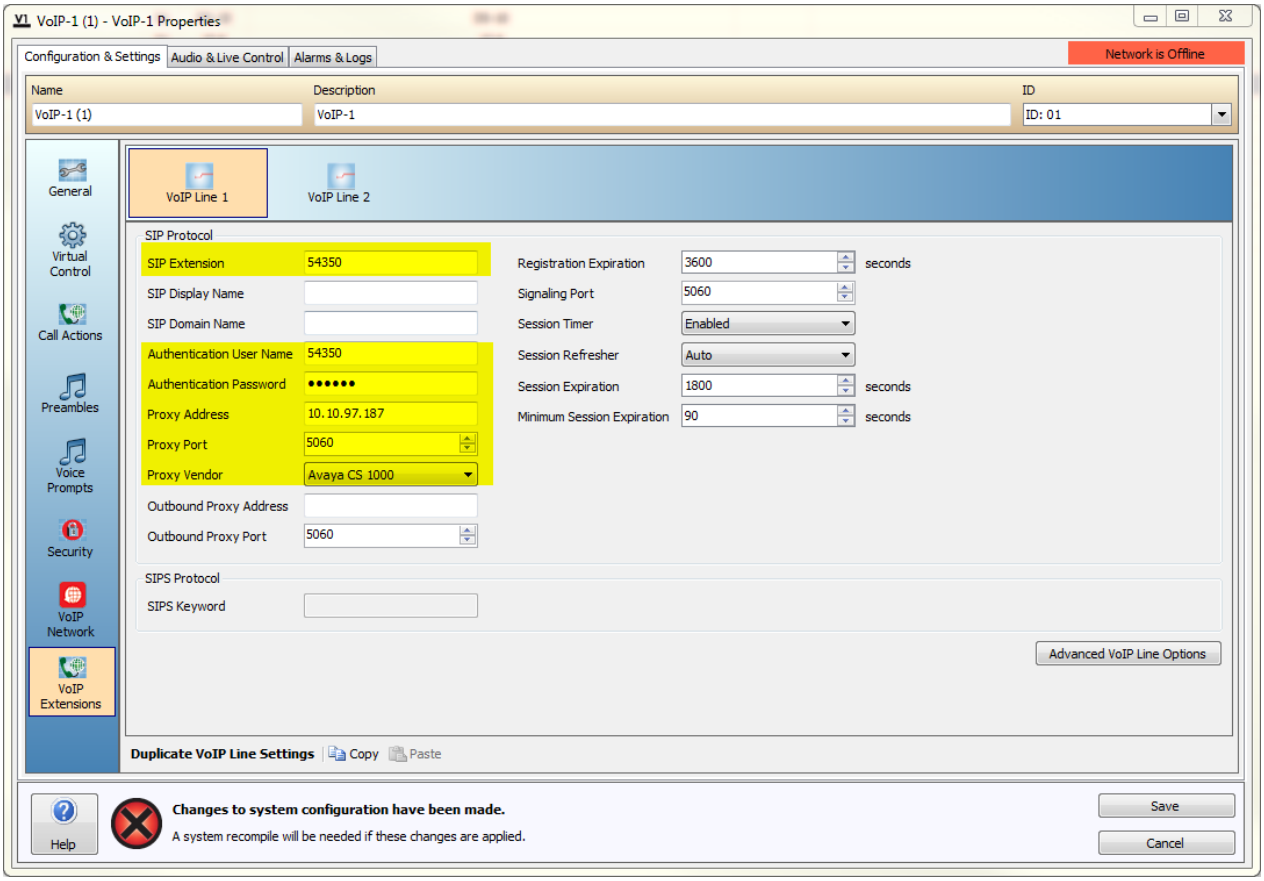

Figure 27: Protocol tab of VoIP Line Properties

3. Verification Steps

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

- Verify that the Biamp Vocia VOIP-1 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 by using Avaya account.
  - Issue command “slgSetShowByUID [userID]” where [userID] is SIP Line user’s ID being checked.

```
[admin@sipl ~]$ slgSetShowByUID 54350
=== VTRK ===
UserID           AuthId      TN               Clients  Calls
SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---
------  -------   ---------
          54350       54350     104-00-00-10        1      0
0x8fc4cf8         SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
```

```
            FeatureMask =
            CallProcStatus = 0

            Current Client = 0, Total Clients = 1

             == Client 0 ==
             IPv4:Port:Trans = 10.10.98.42:5060:udp
             Type          = SIP3
             UserAgent     = Vocia/1.0.0.37
             x-nt-guid     = 267d228547c1562399f1f743a2971fb5
             RegDescrip    =
             RegStatus     = 1
             PbxReason     = OK
             SipCode       = 200
             hTransc       = (nil)
             Expire        = 3600
             Nonce         =
f56a9946ba497bde7eb445efb518f4f1
             NonceCount    = 2
             hTimer        = 0x8f64e60
             TimeRemain    = 1338
             Stale         = 0
             Outbound      = 0
             ClientGUID    = 0
             MSec CLS      = MSNV (MSEC-Never)
             Contact       = sip:54530@10.10.98.42:5060
             KeyNum        = 255
             AutoAnswer    = NO

            Key  Func  Lamp  Label
            0    3     0     54009
            1    126   0     2654009
            2    9     0
            3    29    0
            4    22    0
            5    2     0     54334
            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


             == Subscription Info ==
             Subscription Event  = None
             Subscription Handle = (nil)
             SubscribeFlag = 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 104 0 0 10
IDLE REGISTERED 00
```

- Place a call to the Biamp Vocia VOIP-1 SIP user and verify that the call is established with 2-way speech path.
- During the call, use a sniffer tool (ethereal/wireshark) at the SIP Line Gateway and clients to make sure that all SIP request/response messages are correct.