Avaya Solution & Interoperability Test Lab

Application Notes for Ascom Wifi i62 SIP Telephone firmware version 2.2.22 with Avaya Communication Server 1000 Release 7.5 – Issue 1.1

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and Ascom Wifi i62 SIP telephone. During the compliance testing, the Ascom Wifi i62 was able to register as a SIP client endpoint with Communication Server 1000 SIP Line gateway. The Ascom Wifi i62 telephone was able to place and receive calls from Communication Server 1000 Release 7.5 non-SIP and SIP Line telephones. The compliance testing focused on basic telephone features.

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 Release 7.5 (hereafter referred to as CS 1000) and the Ascom Wifi i62 SIP telephone firmware version 2.2.22 used during the compliance testing. The Ascom Wifi i62 was tested with non-SIP and SIP telephones using the CS1000 SIP line Release 7.5. All the applicable telephony feature test cases of Release 7.5 SIP line were executed on the Ascom Wifi i62, where applicable, to ensure that the interoperability with CS 1000.

2. General Test Approach and Test Results

The general test approach was to have the Ascom Wifi i62 telephone to register to the CS1000 SIP line gateway. Calls were then placed from other CS1000 telephone clients/users to and from the Ascom Wifi i62 telephone. Other telephony features such as busy, hold, DTMF, MWI and codec negotiation were also verified.

2.1. Interoperability Compliance Testing

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/handsets for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

The focus of this testing was to verify that the Ascom Wifi i62 SIP telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the Ascom Wifi i62 SIP telephone to the CS1000 SIP Line Gateway.
- Call establishment of Ascom Wifi i62 with CS1000 SIP and non-SIP telephones.
- Telephony features: Basic calls, conference, blind and consultative transfer, DTMF (dual tone multi frequency) RFC2833 and SIP Info transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, group call pickup, call park, call waiting, ring again busy/no answer, multiple appearances Directory Number and Call forward on Busy, No answer and All Calls..
- PSTN calls over PRI trunk.
- Codec negotiation – G.711 and G.729.

2.2. Test Results

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

- It is highly recommended to disable the media security on the Call Server to avoid some unexpected behaviors such as one way audio from a call made from PSTN over a PRI trunk.
- Local Call Waiting and Call Forward Busy are not support due to CS1000 SIP Line Gateway will always response with 486 Busy Here.

2.3. Support

Technical support for the Ascom i62 product can be obtained through a local Ascom supplier. Ascom global technical support:

- Email: support@ascom.se or Help desk: +46 31 559450

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Ascom Wifi i62.

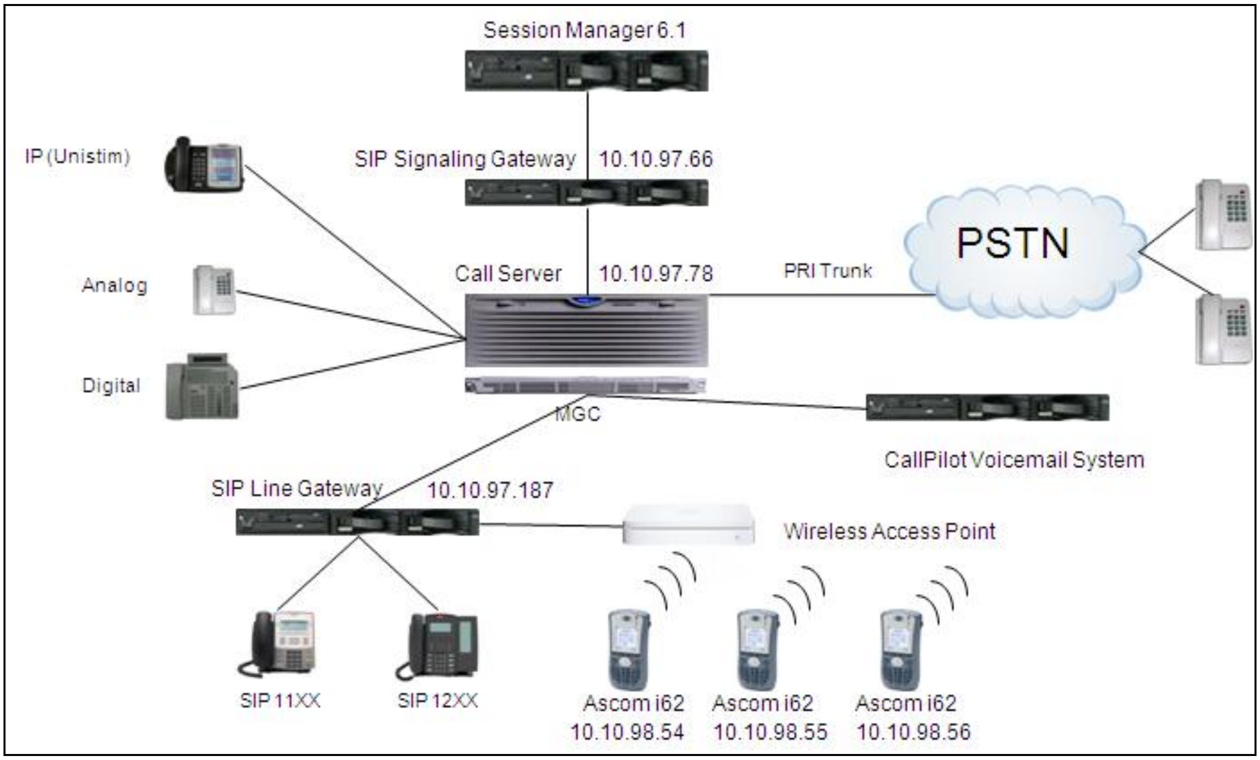

Figure 1: Network Configuration Diagram

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
Avaya Session Manager	6.1
Ascom Communication equipment	WIFI i62 sets firmware version 2.2.22 Wireless Access Point WinPDM 3.8.1 (Device Manger for Windows)

5. 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 **Section 9 Reference [1]**.

The following is the summary of tasks needed 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.

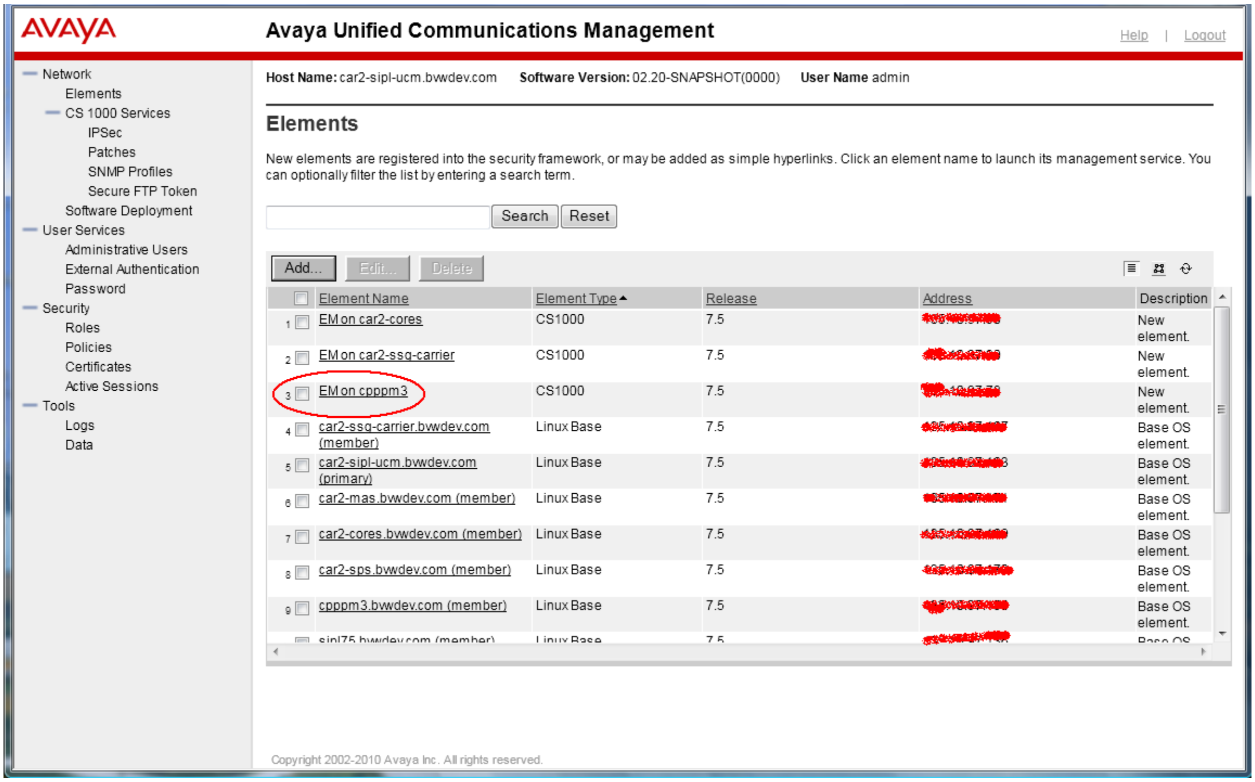

Figure 2: The UCM Home Page of CS 1000 Release 7.5

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.

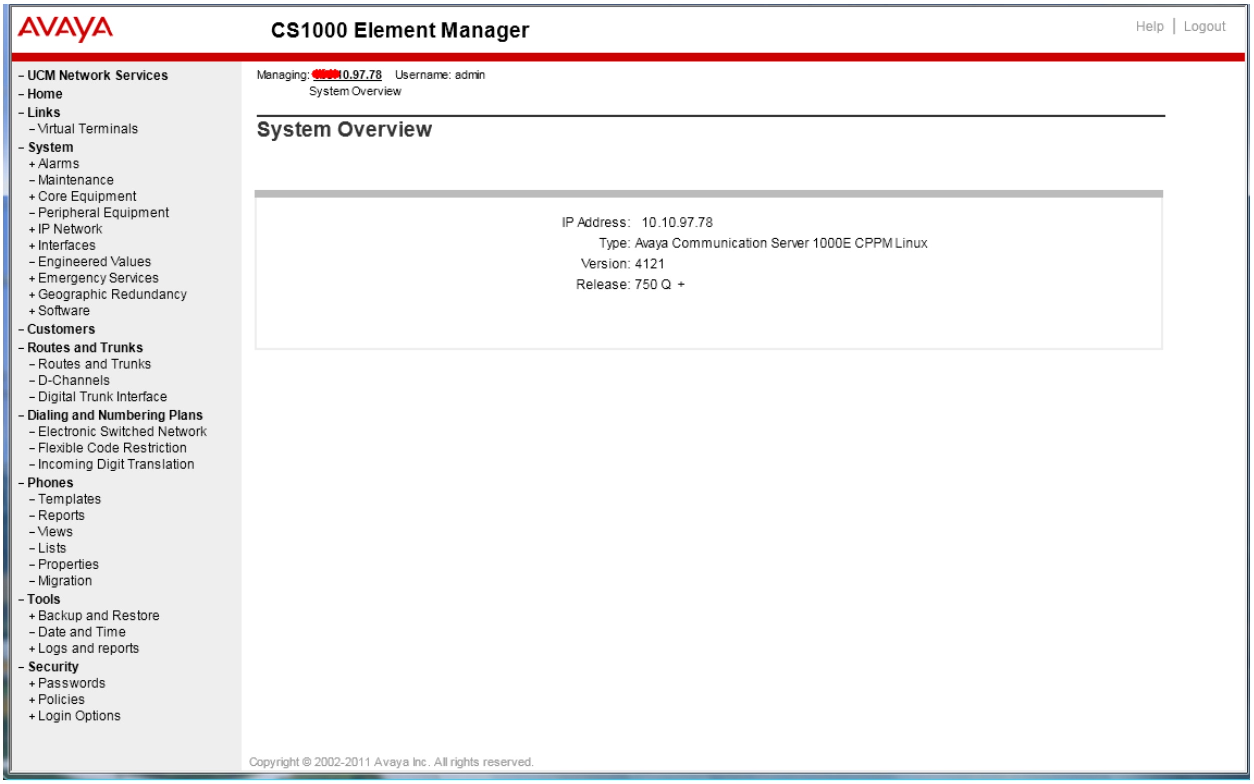

Figure 3: CS 1000 Release 7.5 EM Home Page

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**.

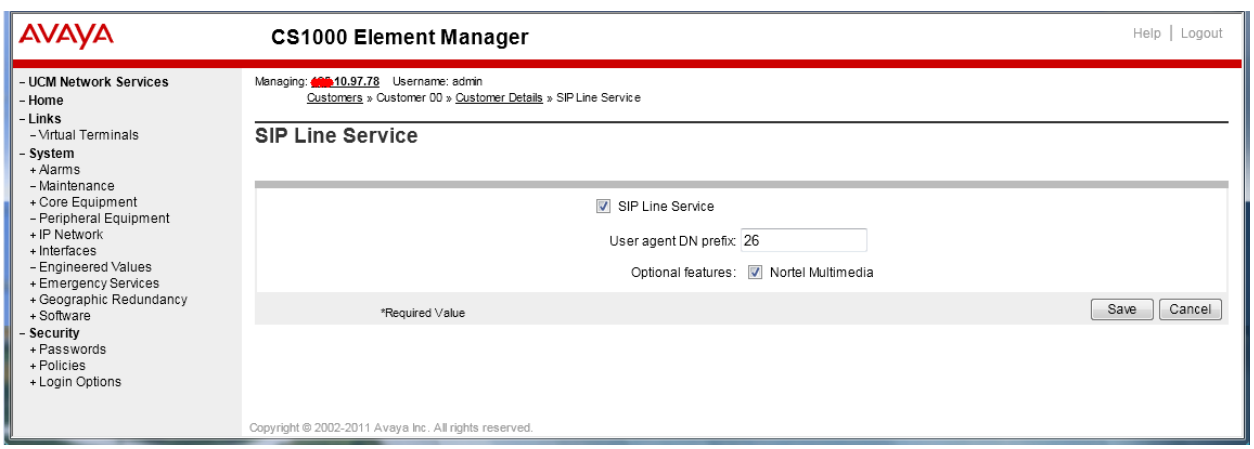

Figure 4: SIP Line Service in Customers Data Block

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 IPv4 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.

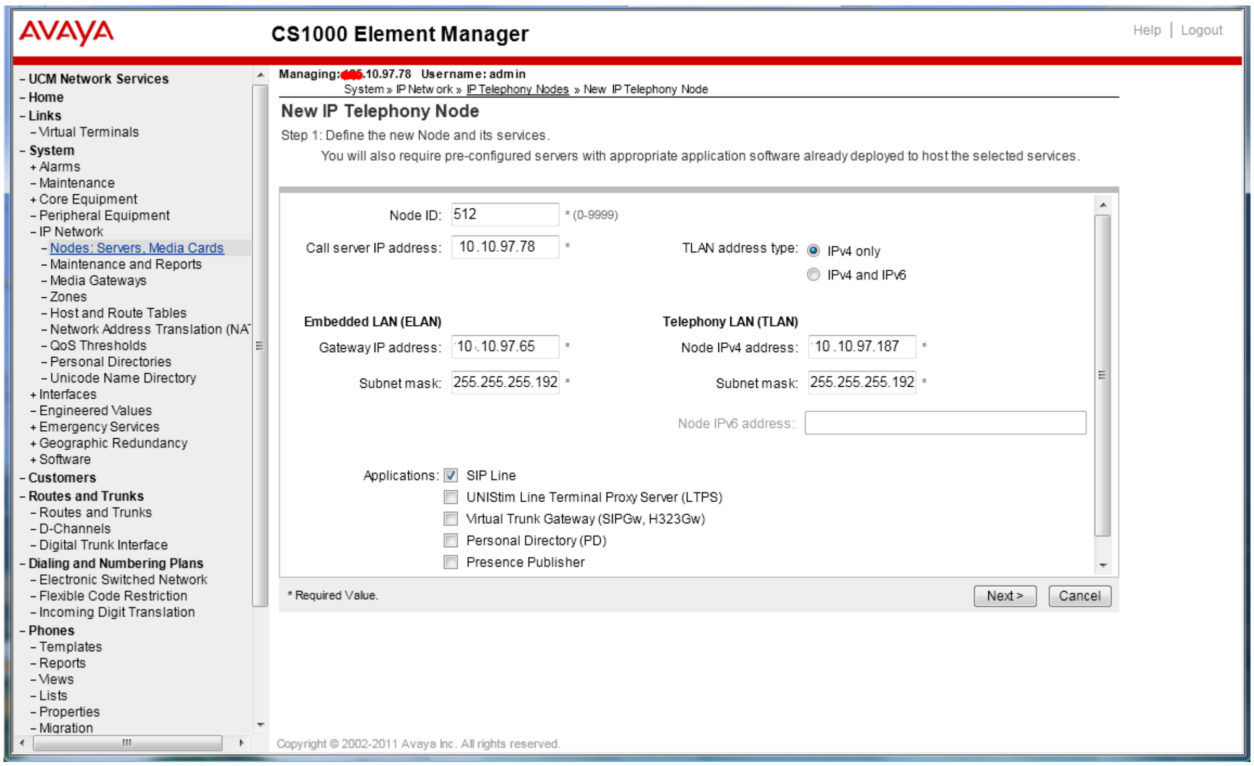

Figure 5: Adding a New IP Telephony Node

- 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).

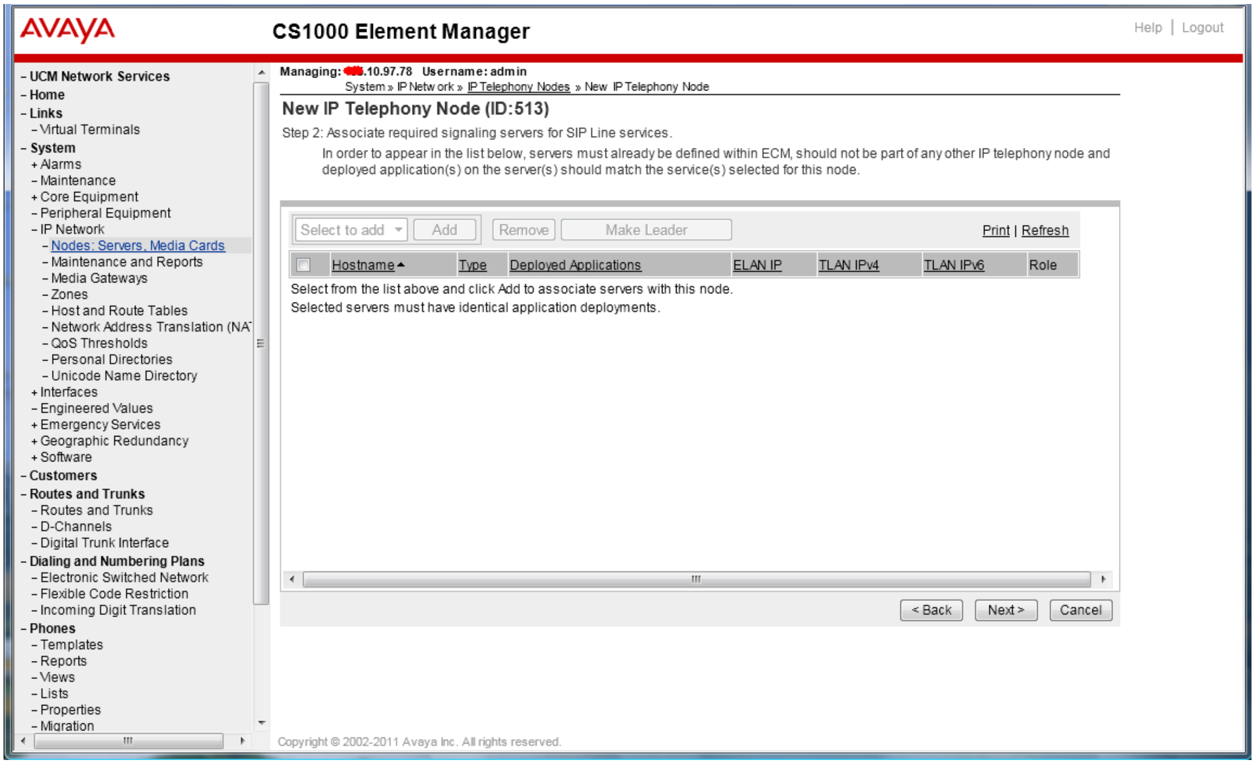

Figure 6: Adding a New IP Telephony Node (cont)

- 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**.

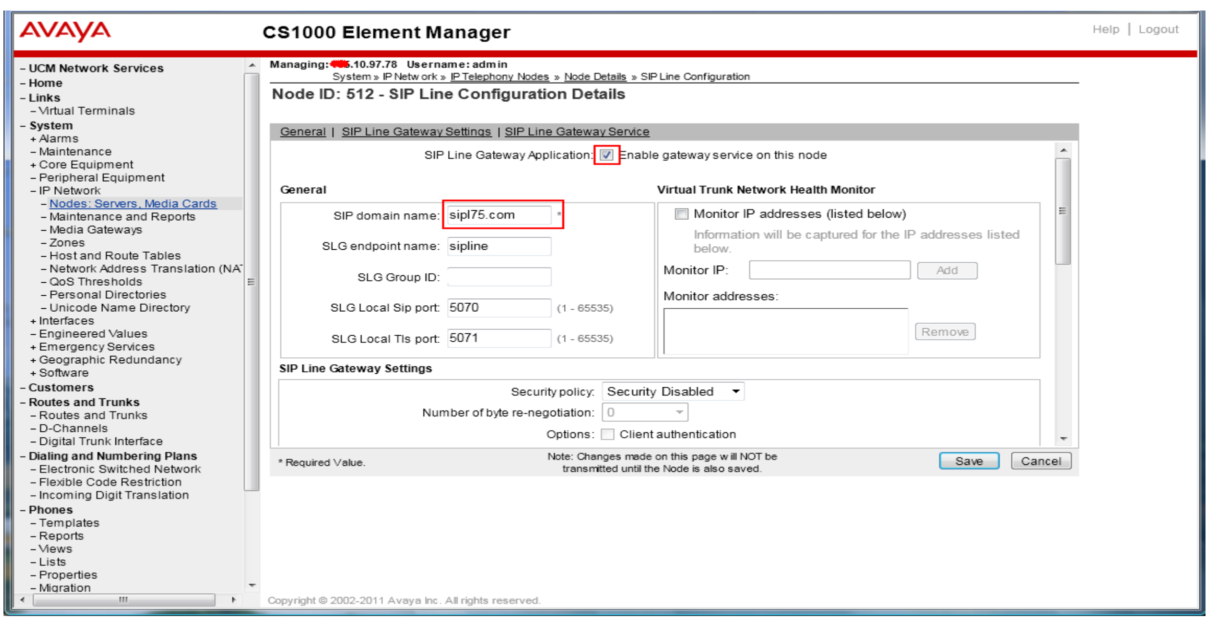

Figure 7: Adding a new IP Telephony Node (cont)

- Under the **SIP Line Gateway Service** 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**.

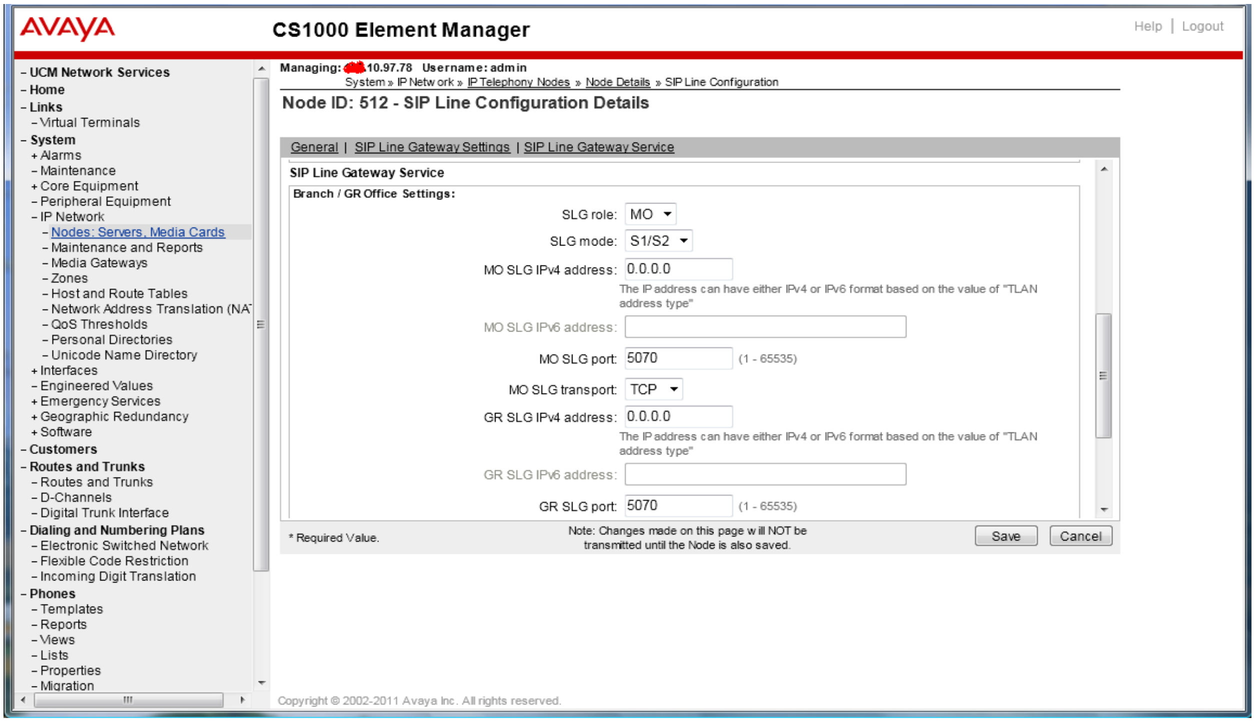

Figure 8: Adding a new IP Telephony Node (cont)

- 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**.

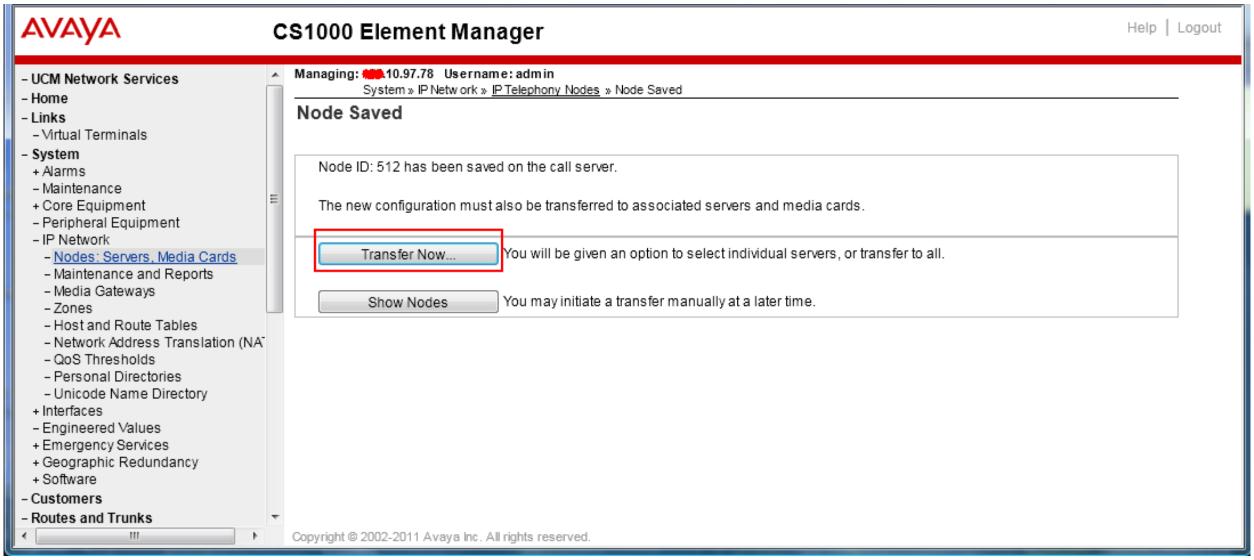

Figure 9: Node Saved with Transfer Configuration

- 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**.

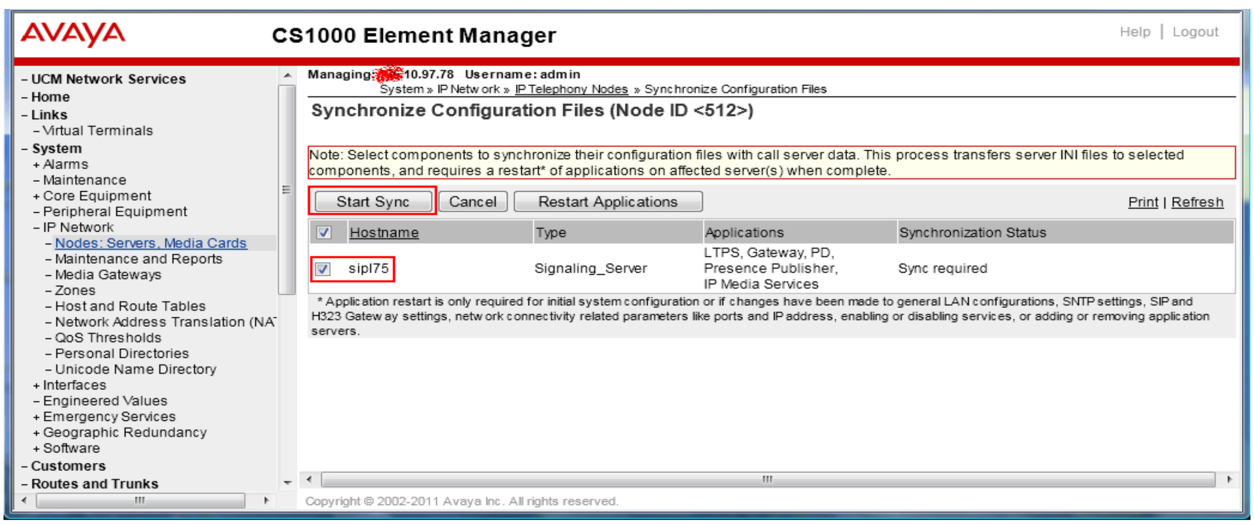

Figure 10: Synchronize Configuration Files

Note: The first time a new Telephony Node is added and transfered 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**.

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.

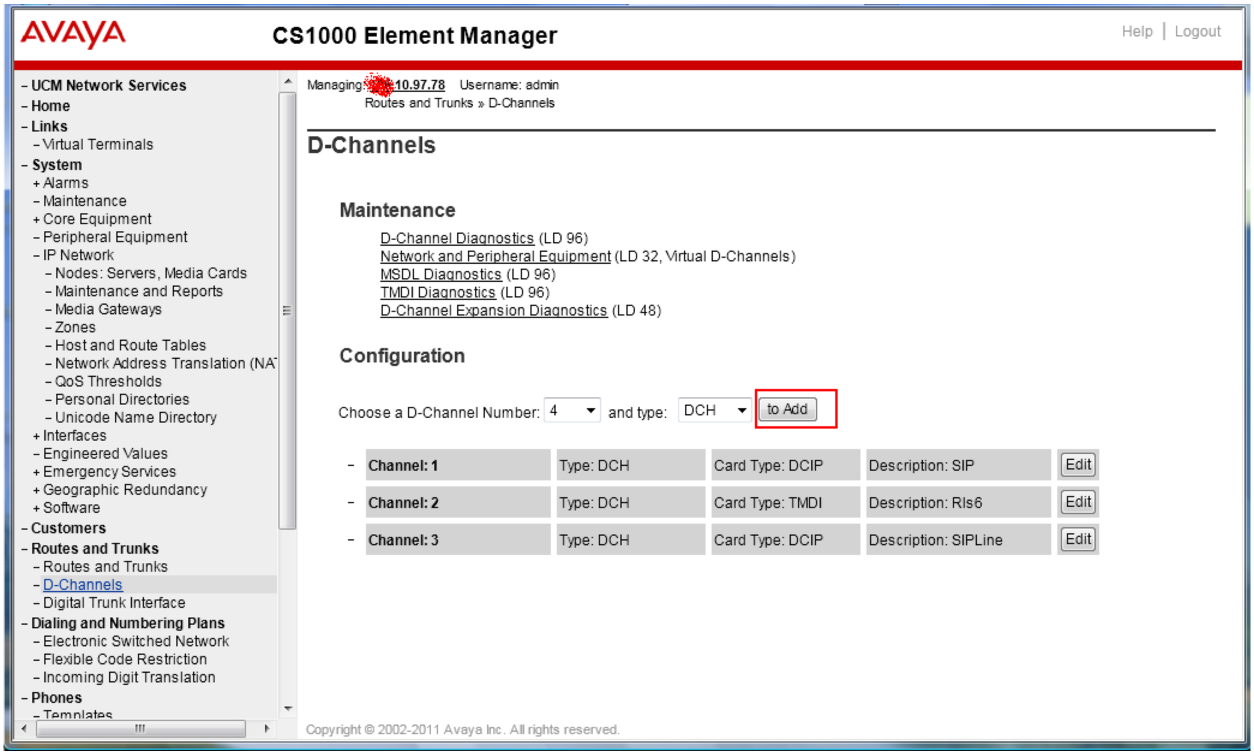

Figure 11: D-Channels configuration page

- 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.

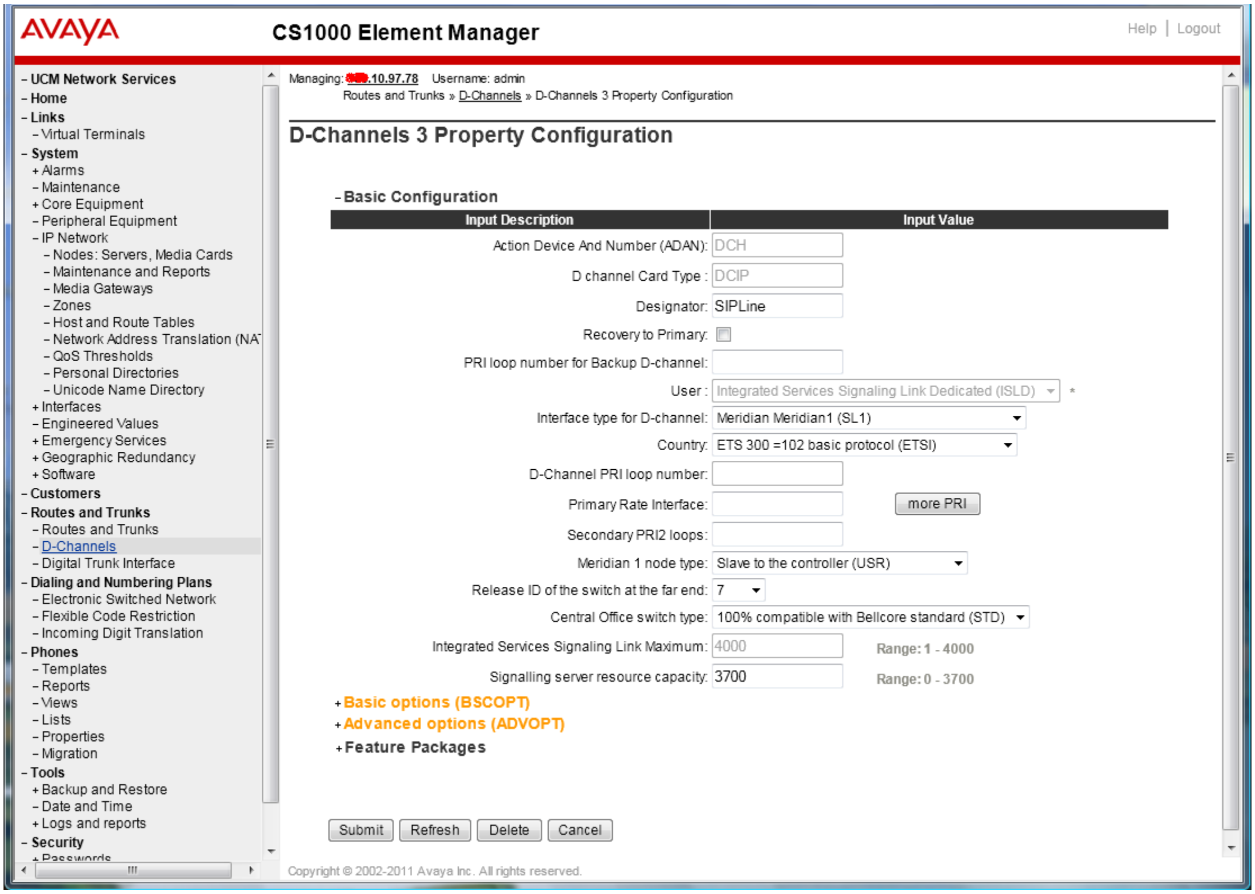

Figure 12: SIP Line D-Channel Property Configuration

- Click on the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands (not shown).
- Click on **Edit** to configure **Remote Capabilities (RCAP)** (not shown). 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.

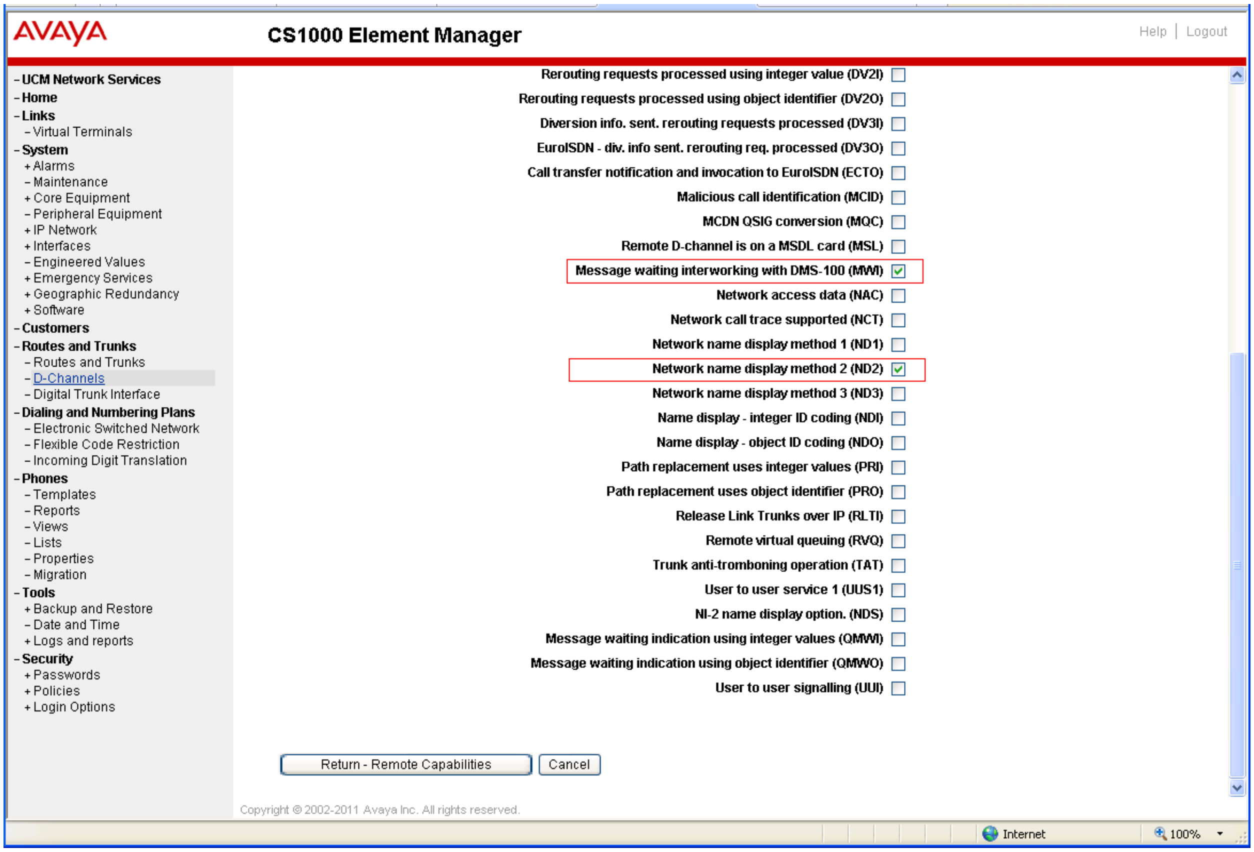

Figure 13: SIP Line D-Channel RCAP Configuration Details

- **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.

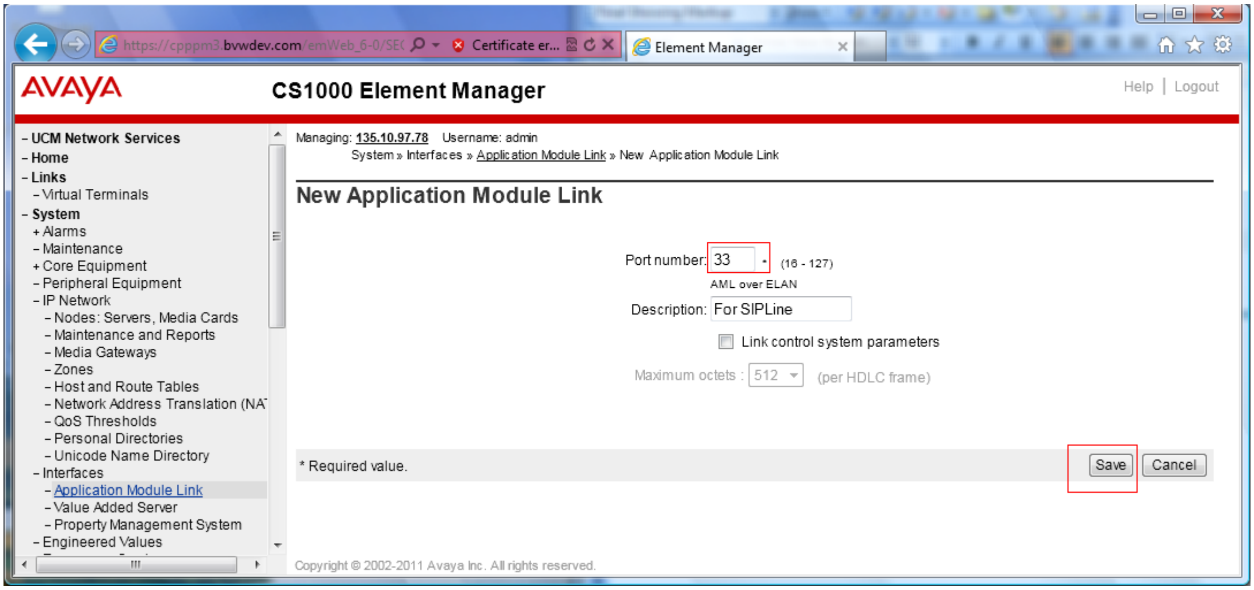

Figure 14: Adding a new AML

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.

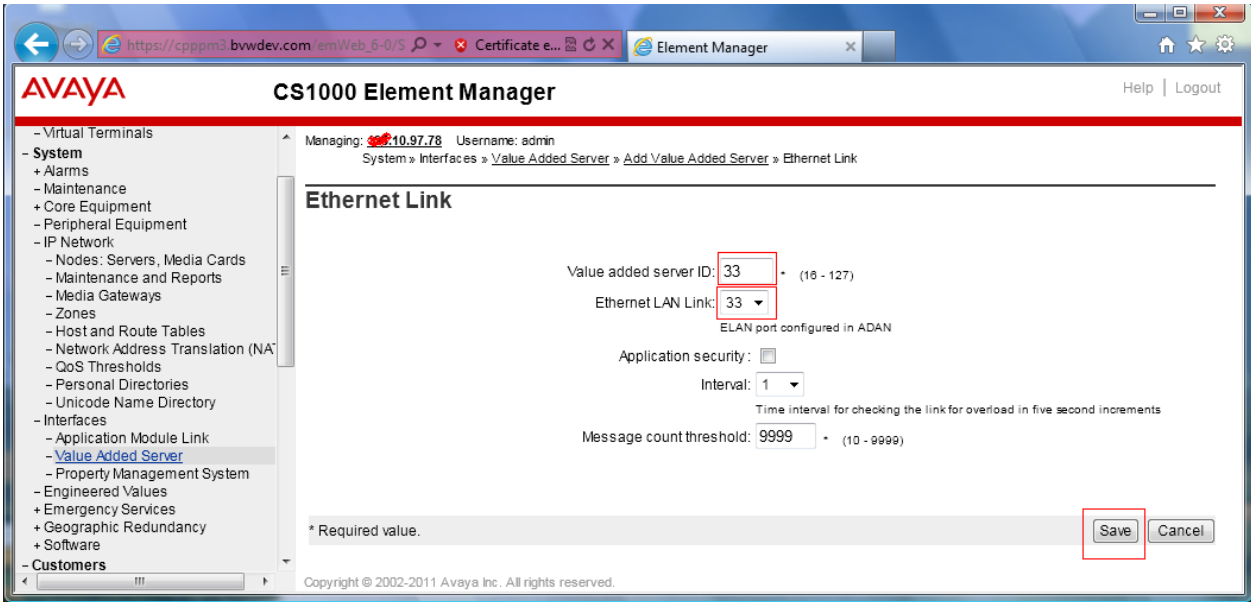

Figure 15: Adding a new Value Added Service for the AML

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 (not shown).

On the **Bandwidth Zones** page, click on the **Add** button (not shown), 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**.

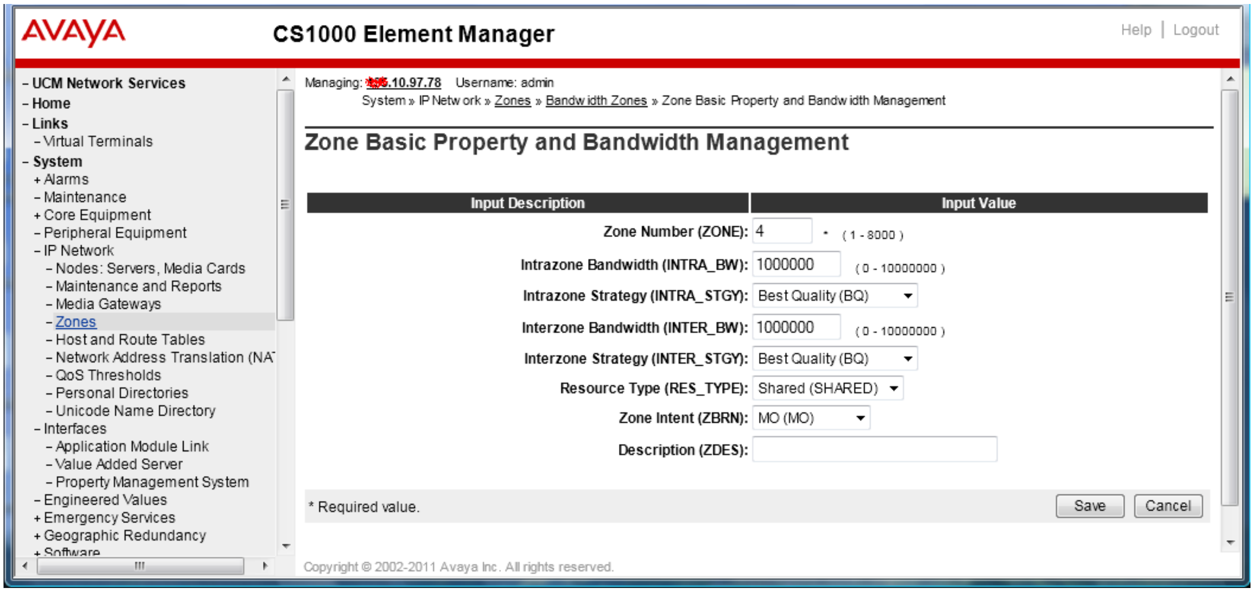

Figure 16: Adding a new Zone for Virtual Trunk

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 7575.
- **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 5.8**.
- **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 5.5**.
- **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.

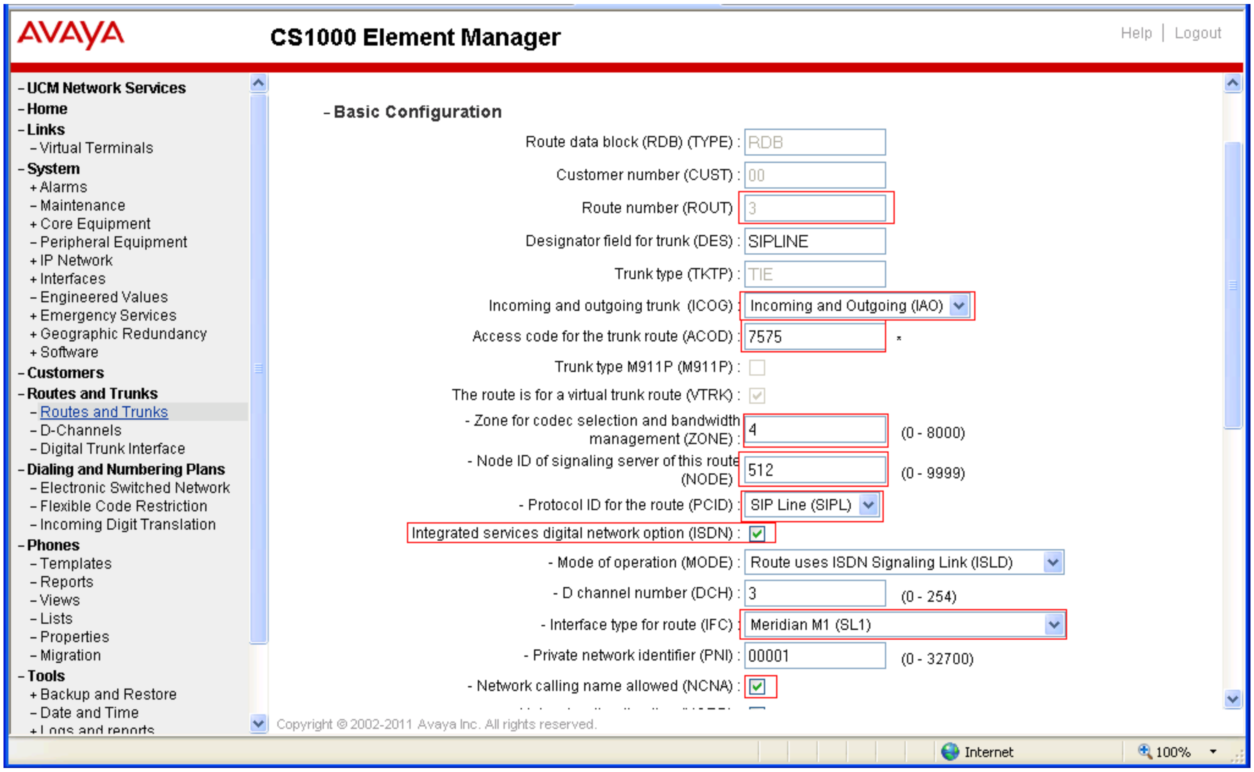

Figure 17: SIP Line Route Configuration

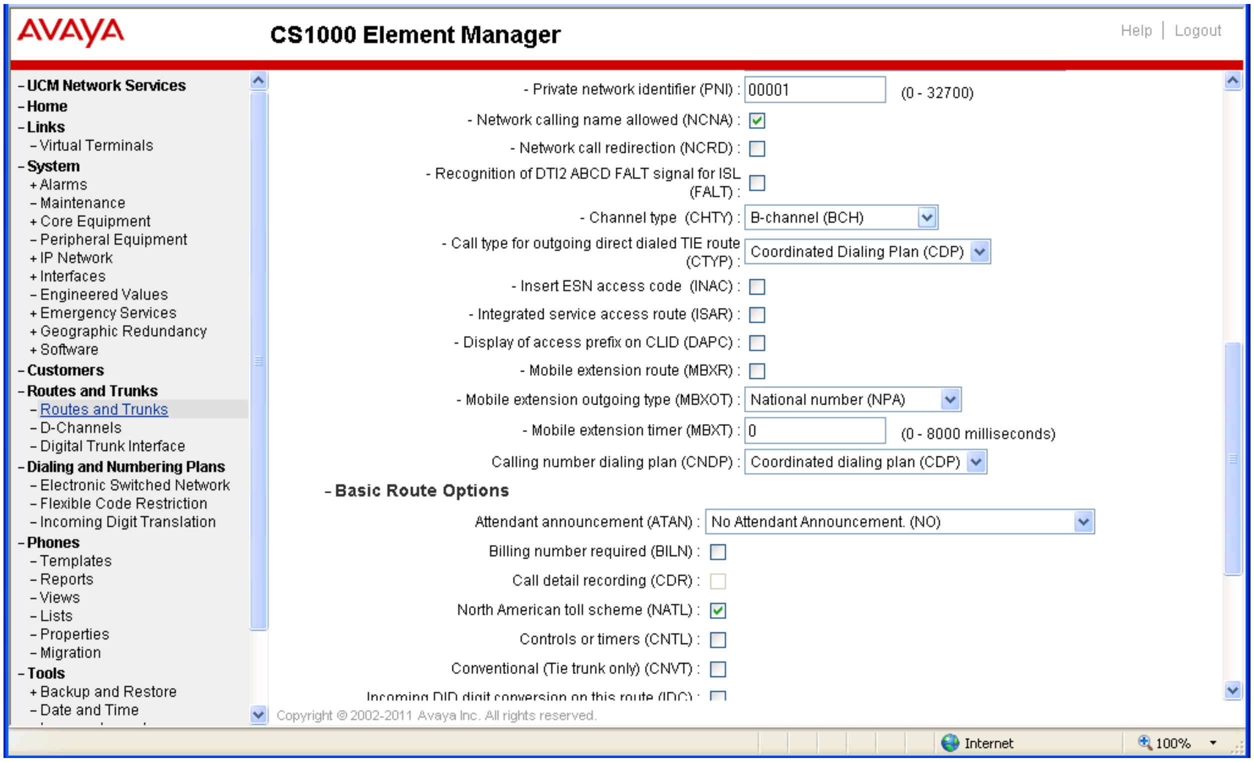

Figure 18: SIP Line Route Configuration (cont)

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 the route that 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.

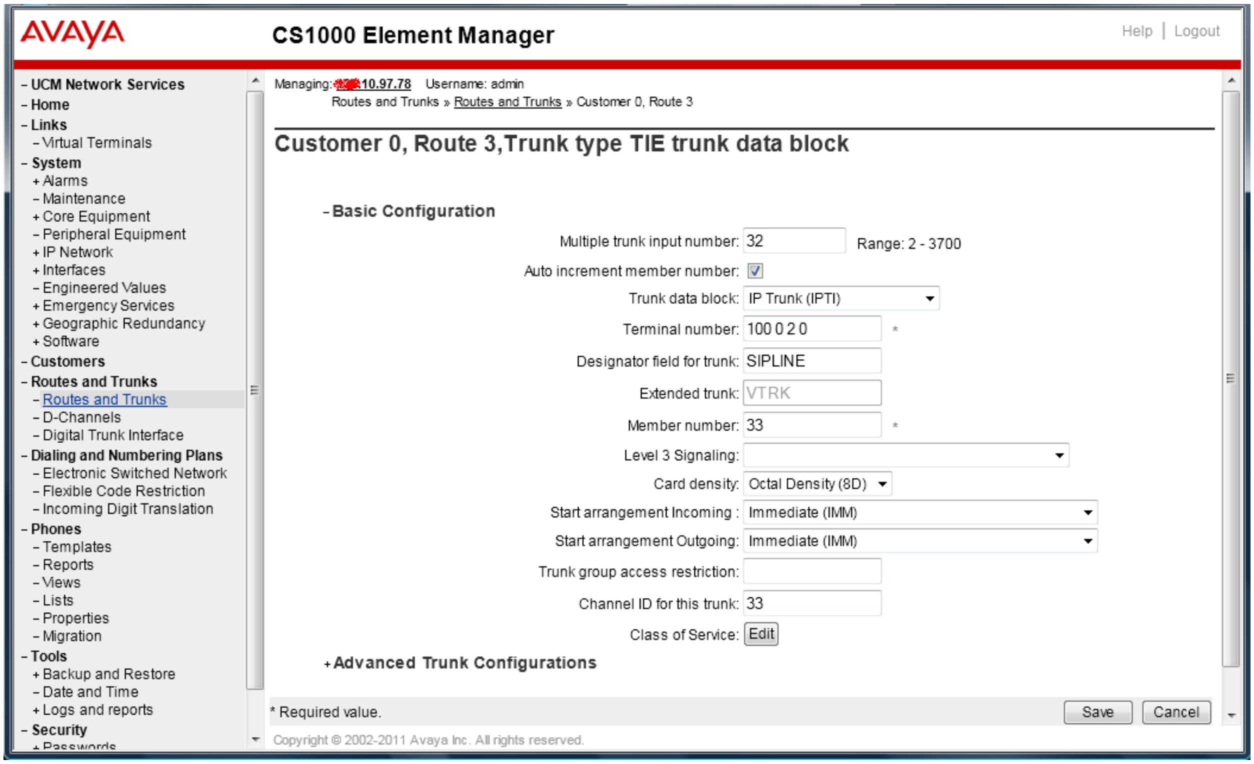

Figure 19: Adding virtual trunks for SIP Line Trunk

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) 11/20 as shown below.

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

LD20
PT0000
REQ: new
TYPE: **UEXT** -> Universal extension type for SIP Line phone
TN **104 0 0 1**

DES POLY1 -> Description of Phone.
CUST 0

UXTY SIPL -> Universal extension type is SIP Line
MCCL YES
SIPN 0
SIP3 1 -> For SIP phone third party, enter 1 in this field
FMCL
TLSV
SIPU 54008 -> SIP phone username
NDID 512 -> Node ID of SIP Line
SUPR
SUBR
UXID
NUID
NHTN
ZONE 3 -> Zone for SIP Line phone.
MRT
ERL
ECL
VSIT
FDN 54002 -> Forward No Answer to this DN, need to enable class of service FNA
TGAR 1
LDN
NCOS 7 -> Network Class of Service, 7 is highest level.
SGRP
RNPG
SCI
SSU
XLST
SCPW 1234 → Password to log in to SIP Line username 54008
SFLT
CAC_MFC
CLS FNA FBA HTA MWA DNDA CNDA CFXA -> class of service.
RCO
HUNT 54444 -> Forward busy to this DN, need to enable class of service FBA and HTA
PLEV
KEY 00 SCR 54008 0 MARP -> Key 0 is DN of SIP phone.
CPND new
CPND_LANG ROMAN
NAME Poly 8440 -> Display name of SIP Phone.
XPLN 13
DISPLAY_FMT FIRST,LAST
01 HOT U 2654008 MARP 0 -> Key 1 Hot U with prefix + DN
02 CWT -> Call Waiting key
03 MSB -> Make Set busy key
04 SCU 0000 -> Speech call dial key

6. Configure Ascom Wifi i62

This section describes how to access and configure the Ascom Wifi i62 SIP handset via the Windows Device Manager called WinPDM version 3.8.1, which can be downloaded via Ascom extranet and installed on a Windows PC. Remote device management “over the air” provides a similar graphical user interface. Insert the handset to be configured in the DP1 USB cradle, start the Ascom Device Manager, and select the “Devices” tab. The inserted i62 set is now being indicated with a check mark under the **Online** column as shown in **Figure 20**.

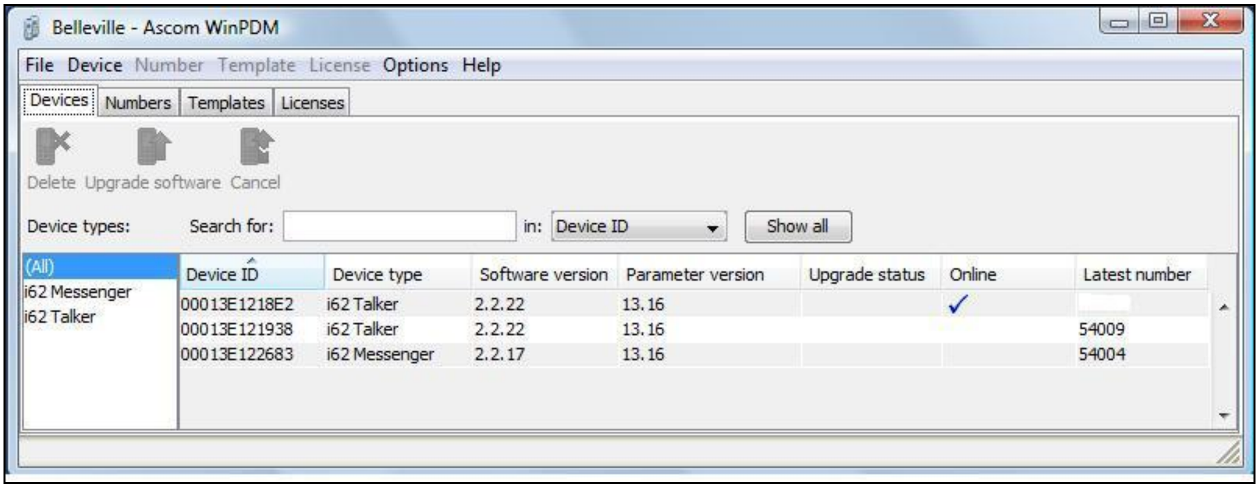

Figure 20: Ascom Device Manager Devices Tab

Select the **Numbers** tab as shown in **Figure 21**. Click on the **New** icon to add a new number **54008** in this example.

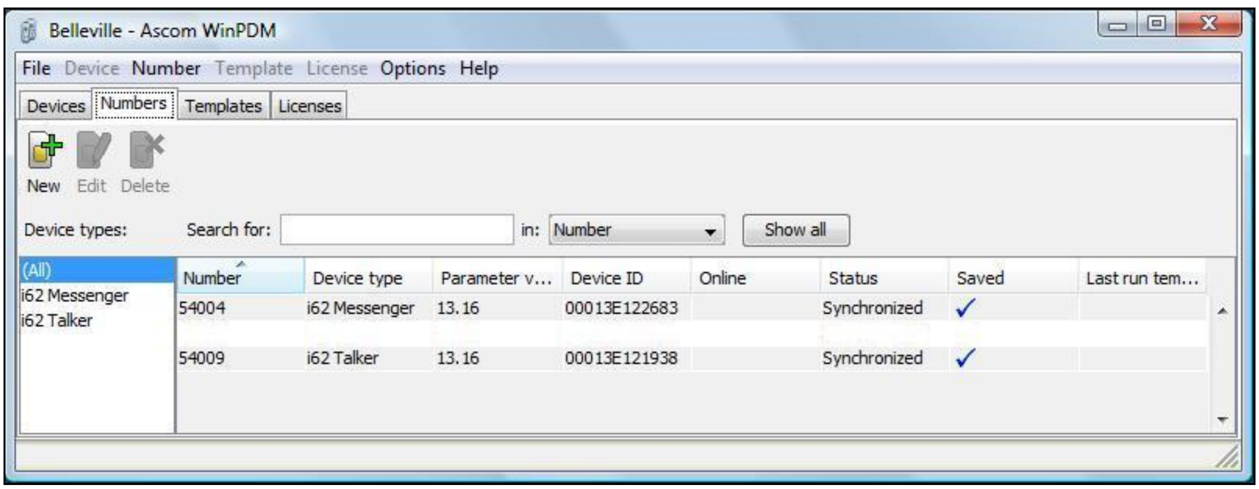

Figure 21: Ascom Device Manager Numbers Tab

There is a dialog box popping up as shown in **Figure 22.** Enter **54008** in the textbox of **Call number** parameter. Click **OK** to create the new number in the **Numbers** table.

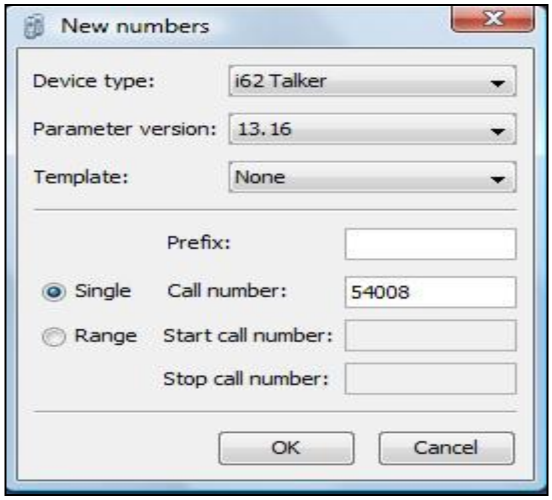

Figure 22: Device Manager Add New Numbers

On the **Numbers** tab, the number **54008** is now shown up on the Number list as shown in **Figure 23**.

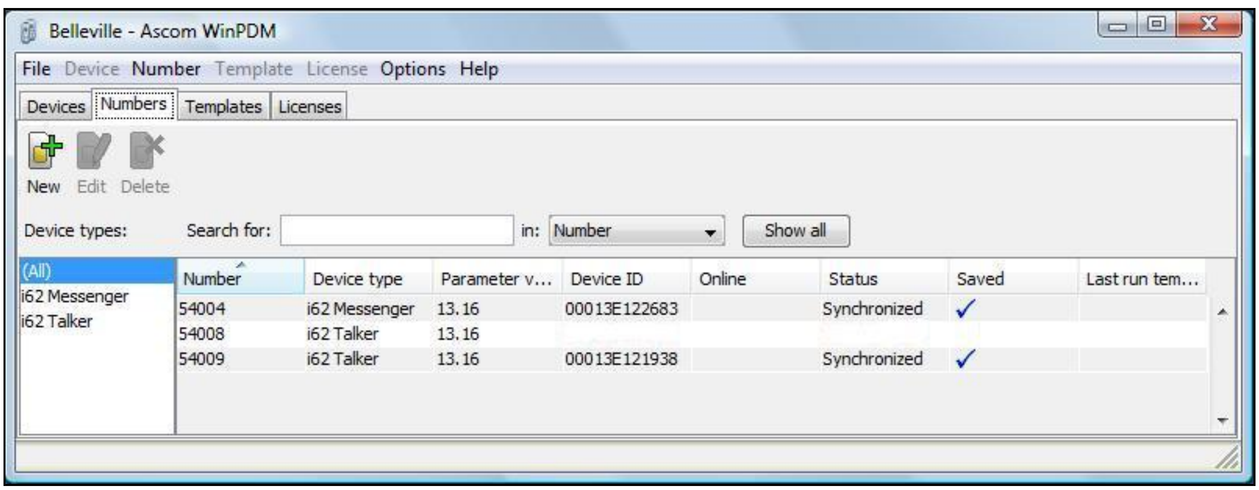

Figure 23: Device Manager with New Number Added

Right click on the newly created number **54008** and choose the **Associated Numbers** to associate the new number with the i62 physical device being inserted in **Figure 20.** Pop up **Associated Number** window will be as shown in **Figure 24.** Choose the i62 set to associate the number with and click **OK** to assign the number.

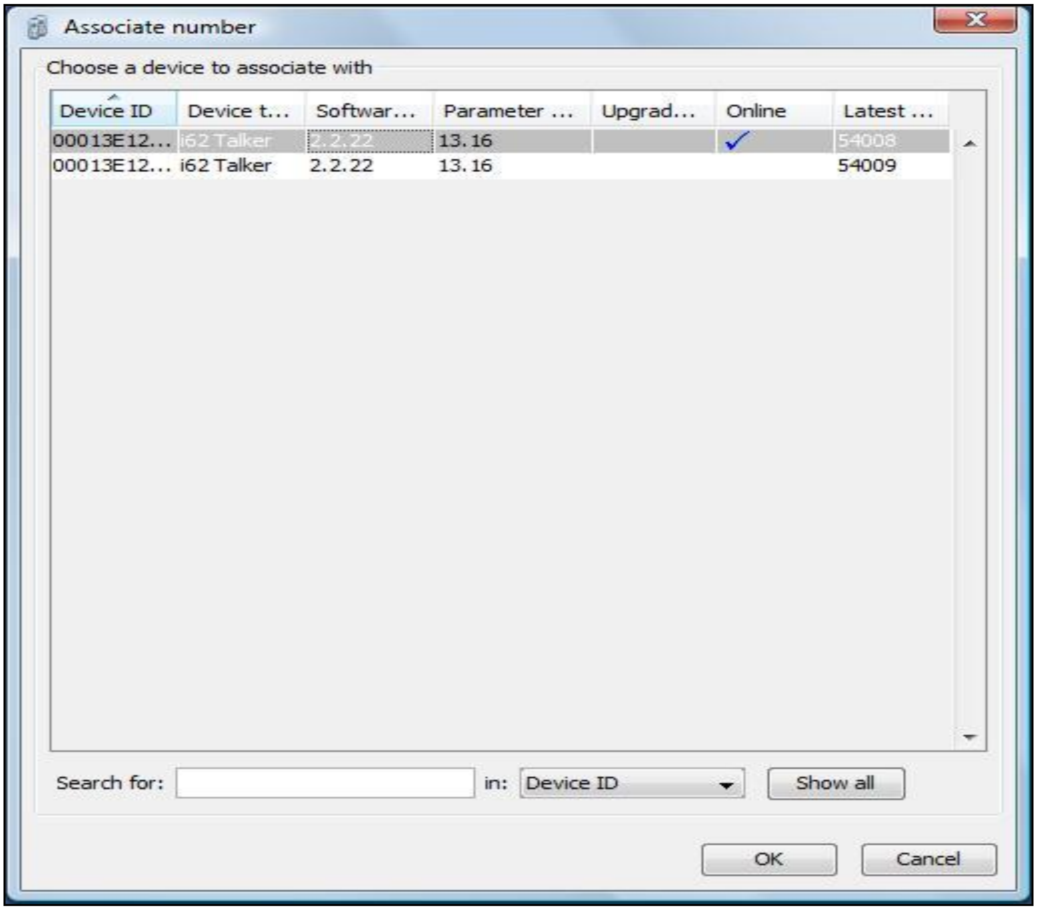

Figure 24: Associate a Number to Physical Set

Figure 25, below, shows the inserted i62 set with its assigned number **54008** in the **Numbers** table.

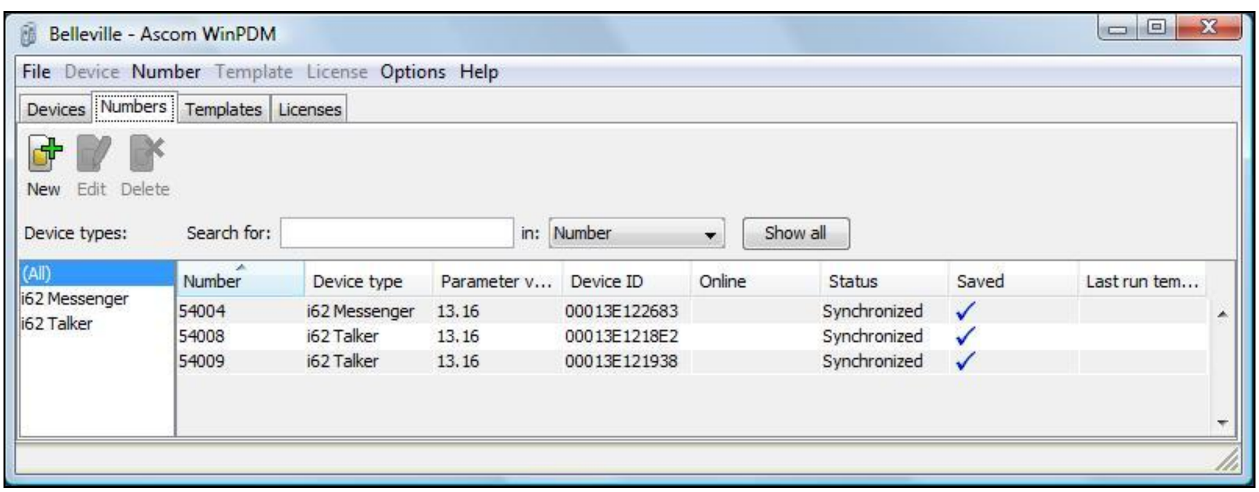

Figure 25: New Number with Associated i62 Set

Double click on the entry for the handset to be configured, select the **Network -> Network A,** an **Edit Parameters for 54008** Window will appear as shown in **Figure 26**. Fill in the parameters as highlighted in red.
Note: This setting is one of many ways to configure the network set up for the i62 handset. For more information how to configure this in a different way, refer to **Reference [2]**.

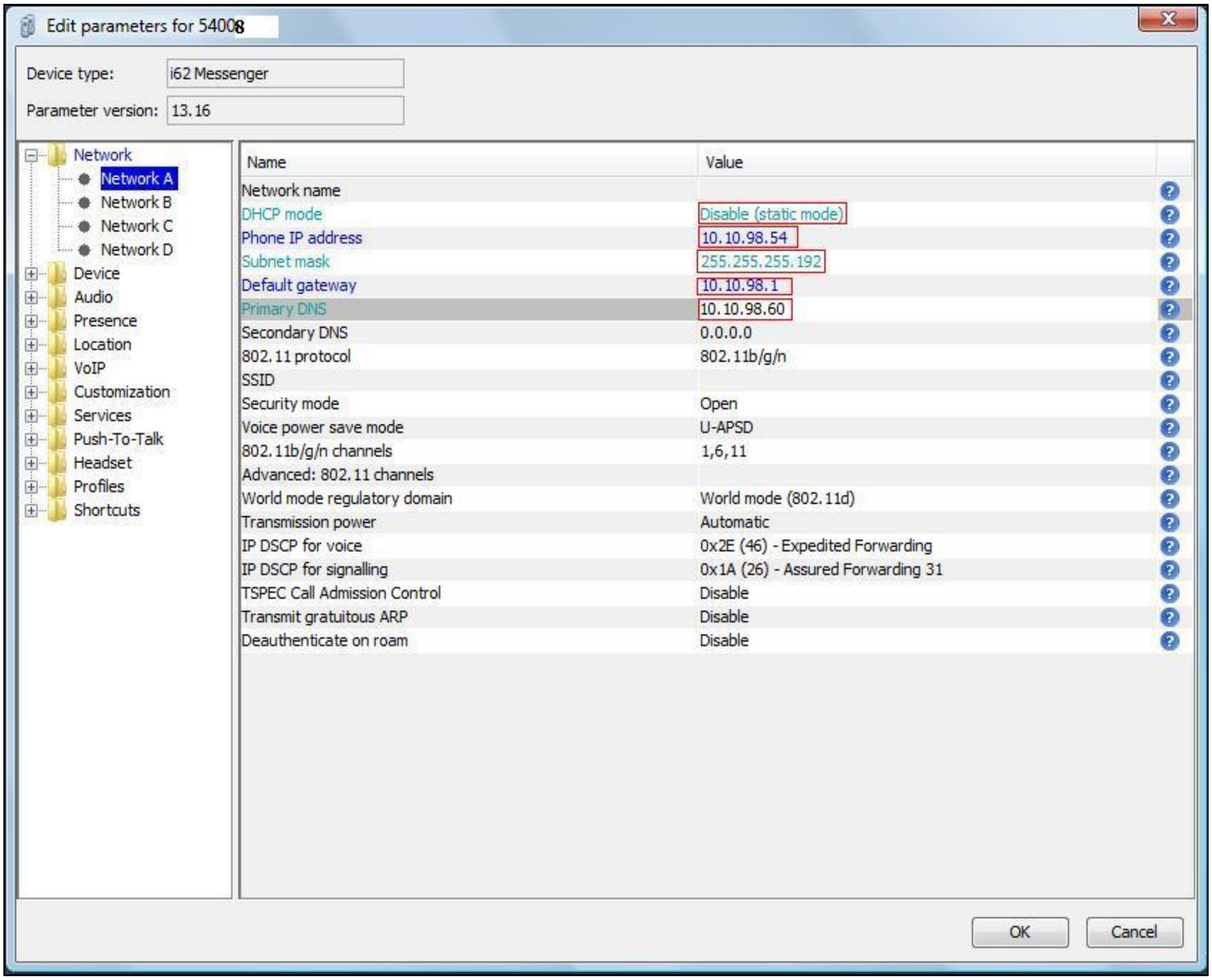

Figure 26: Network Parameters

Select the **VoIP**-> **General** menu, and enter the values highlighted in red as shown in the **Figure 27**. Click **OK** (not shown) to save the change.

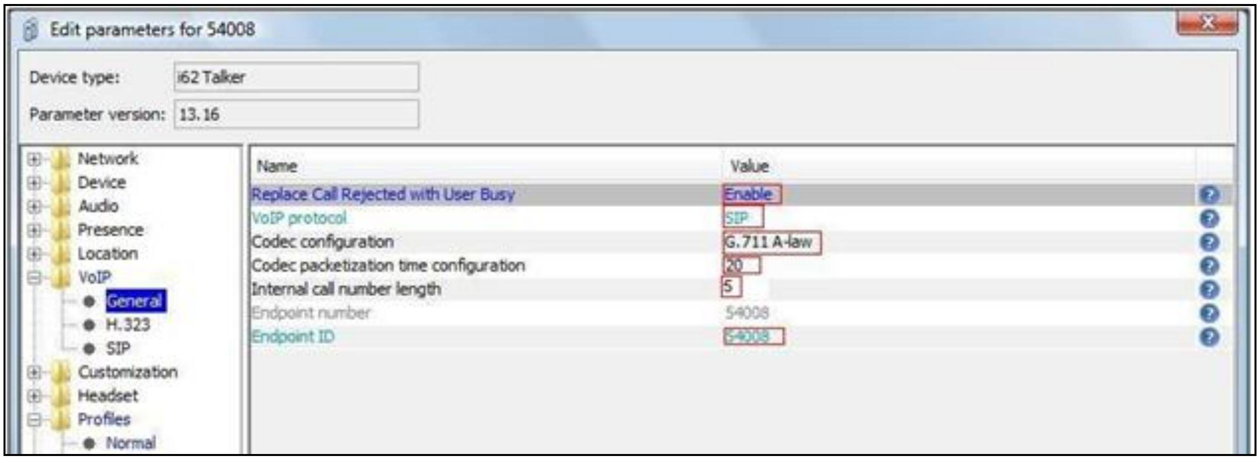

Figure 27: VoIP General Parameters

Select the **VoIP->SIP** menu point, and enter the values highlighted in red as shown in **Figure 28.** Click **OK** (not shown) to save the changes.

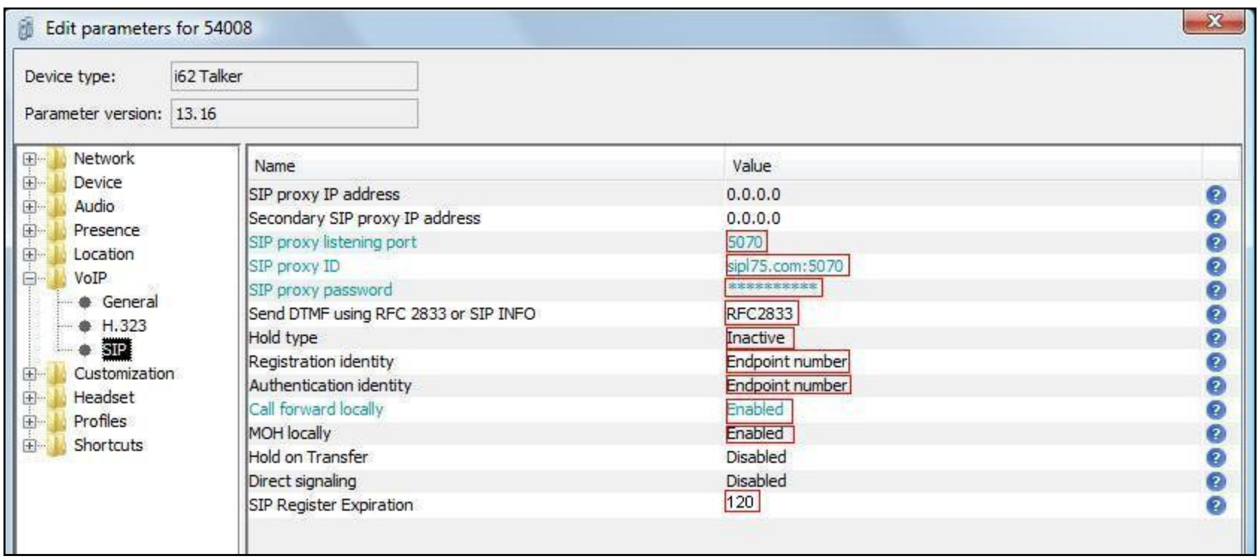

Figure 28: VoIP SIP Parameters

Note: For **SIP Register Expiration** parameter, it should be set at **120** as recommended by Ascom.

DNS entry is required to resolve the domain (sipl75.com) into IP address (10.10.97.187) as required. In this example configuration, the DNS entry is shown in figure bellow.

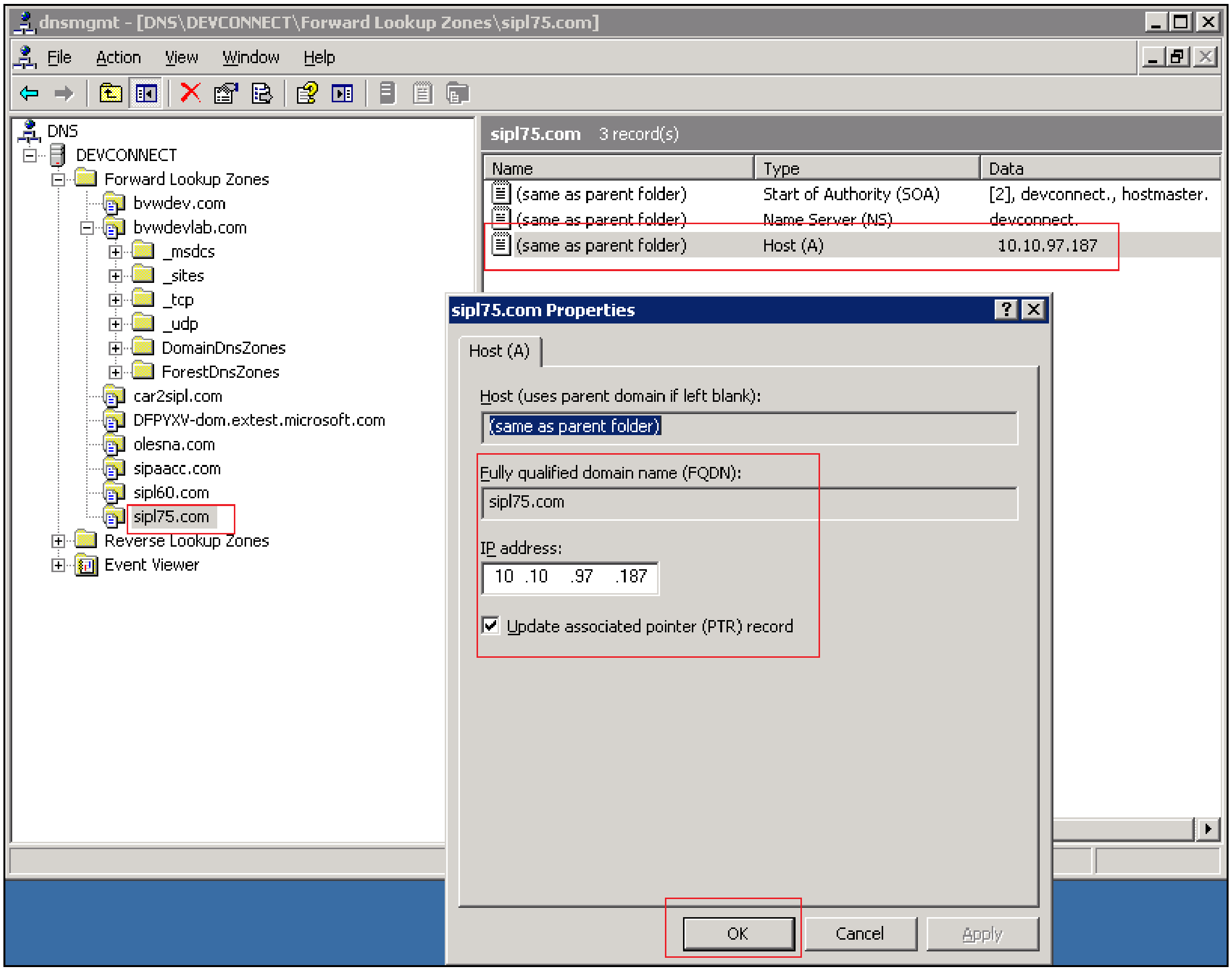

Figure 29: DNS resolution for sipl75.com domain

7. Verification Steps

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

- Verify that the Ascom Wifi i62 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 by using Avaya account.
  - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked.

```
[admin@sipl ~]$ slgSetShowByUID 54008
=== VTRK ===
UserID           AuthId      TN               Clients  Calls
SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---
------  -------   ---------
          54008       54008     104-00-00-01        1      0
0x8fc4cf8         SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.10.98.55:5060:udp
         Type           = SIP3
         UserAgent      = (Ascom i62/Ascom i62 2.2.22 \(2011-03-
30\) release )
         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:54008@10.10.98.55:5060
         KeyNum         = 255
         AutoAnswer     = NO

        Key  Func  Lamp  Label
        0    3     0     54008
        1    126   0     2654008
        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 1
IDLE REGISTERED 00
```

- Place a call from and to Ascom Wifi i62 telephone and verify that the call is established with 2-way speech path.
- During the call, use a pcap tool (ethereal/wireshark) at the SIP Line 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 the **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Ascom Wifi i62 firmware version 2.2.22 is considered to be in compliance with Avaya CS 1000 SIP Line System Release 7.5.

9. Additional References

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

Product documentation for the Ascom Wifi i62 products may be found at:
http://www.ascom.com

[1] Avaya CS1000 Documents:
Avaya Communication Server 1000E Installation and Commissioning

Avaya Communication Server 1000 SIP Line Fundamental, Release 7.5
Avaya Communication Server 1000 Element Manager System Reference – Administration
Avaya Communication Sever 1000 Co-resident Call Server and Signaling Server Fundamentals
Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals.
Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning

[2] Ascom Wireless i62 Wifi Phone Documents:
Ascom i62 VoWifi Handset Quick Reference Guide
Installation and Operation Manual, Portable Device Manager
Configuration Manual, Ascom i62 VoWifi Handset

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.