Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Spectralink 84-Series SIP Telephone Version 4.7.0 with Avaya Communication Server 1000 Release 7.6 - Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.6 and Spectralink 84-Series SIP telephone. During the compliance testing, the Spectralink 8400 was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Spectralink 8400 telephone was able to place and receive calls from the Communication Server 1000 Release 7.6 non-SIP and SIP Line clients. The compliance tests focused on basic telephony features.

Readers should pay attention to section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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 7.6 (hereafter referred to as CS 1000) and the model of Spectralink 84-Series was used for the compliance test was 8440 (hereafter referred to as 8440). The Spectralink 84-Series was tested with non-SIP and SIP clients using the CS 1000 SIP line release 7.6. All the applicable telephony feature test cases of release 7.6 SIP line were executed on the 8440, where applicable, to verify the interoperability with CS 1000.

These Application Notes assume that Avaya CS 1000 is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult the documentation library mentioned in **Section 9**.

2. General Test Approach and Test Results

The general test approach was to have the 8440 telephone register to the CS 1000 SIP line gateway successfully. From the CS 1000 telephone clients/users, place a call to and from the 8440 telephone and to exercise other telephony features such as busy, hold, Dual Tone Multi Frequency (DTMF), Message Waiting Indication (MWI) and codec negotiation.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the 8440 wireless telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the 8440 wireless telephone to the CS 1000 SIP Line Gateway.
- Telephony features: Basic calls, conference, transfer, DTMF RFC2833 transmission, voicemail with Message Waiting Indication notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer, multiple appearances Directory Number.
- PSTN calls over ISDN/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:

- Avaya has not performed audio performance testing or reviewed the 8440 compliance to required industry standards.
- The 8440 telephones are treated by the CS 1000 as 3rd party SIP endpoints and use CS 1000 3rd party SIP licenses.
- The Spectralink 8400 local forward busy feature which is set on the phone locally can be enabled but it is not used. The server call forward busy feature of CS 1000 SIP Line will take place before the local forward busy can be executed by the phone. For the call forward busy, use the forward busy on the CS 1000 switch instead.
- Transfer call performed on the 8440 failed if codec G.722 is used. It is recommended that G.722 be disabled on the 8440 when using to with Avaya CS 1000.
- With 3-way conference call on the 8440 by default set to **1** (Default) which is the conference call will be not terminated when the host conference 8400 disconnects. This will cause no audio for remaining parties in the conference call. To avoid this issue, setting **call.transferOnConferenceEnd** to **0** in **.cfg** file, and the conference call will be terminated when the host conference 8400 phone disconnects.
- In an event of losing the wireless signal, an active call between 8400 phones or between the 8400 and a desk phone causes the SIP Line trunk kept in busy until it is either manually released in call server, or the handshake timer expires (the timer default setting in the 8400 phones is about 8 minutes).

2.3. Support

Technical support on the Spectralink 84-Series telephone can be obtained through the following:

North America:
Phone: 1-800-775-5330
Email: nolarma@spectralink.com
Web: http://support.spectralink.com

EMEA:
Phone: +33 176774541
Email: emeaom@spectralink.com
Web: http://support.spectralink.com

Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya Communication Server 1000 and the Spectralink 8440. The 8440 phone registers to the CS 1000 SIP Line server by going through the Wi-Fi access point that connects to the lab network. Avaya Aura® Session Manager was used for routing SIP calls between the CS 1000 A and CS 1000 B for test cases off-net via SIP trunk. The PRI T1 trunk was configured to connect to PSTN for test cases off-net via PRI T1 trunk.

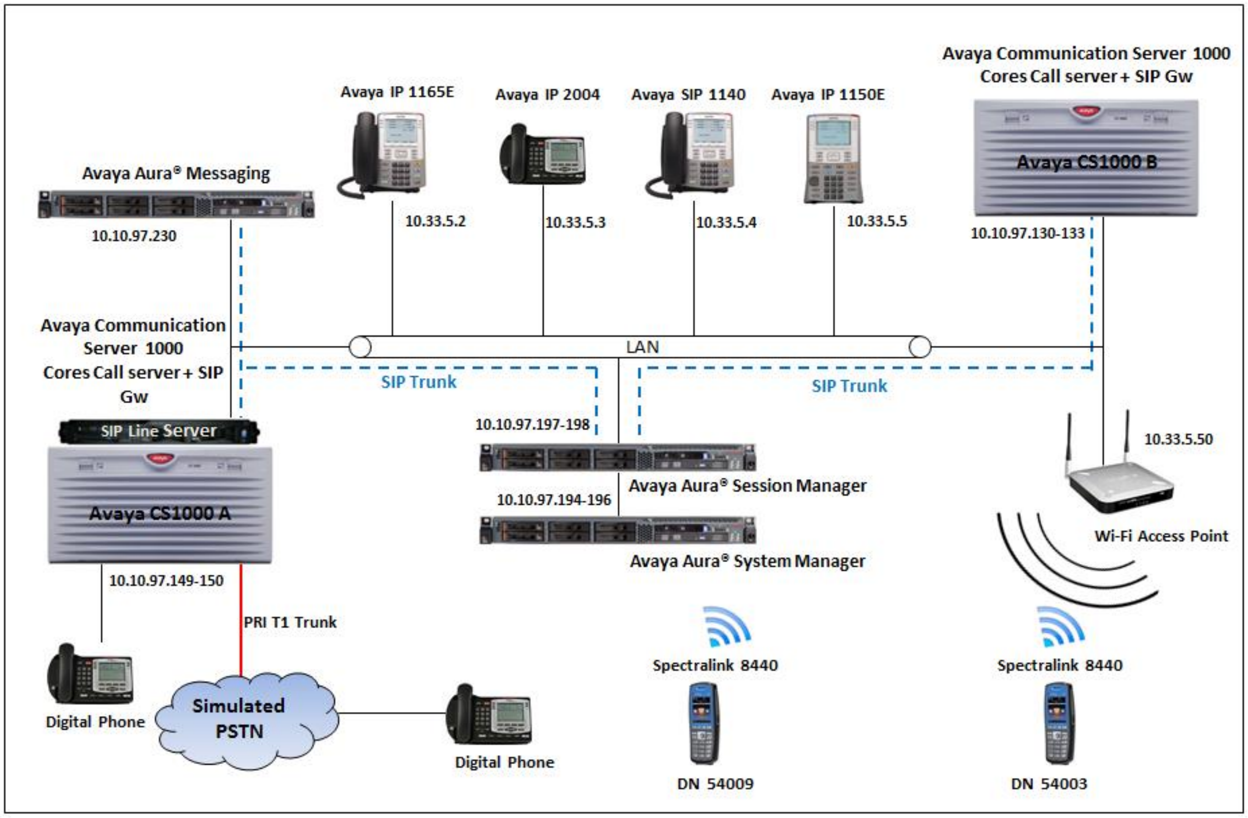


Figure 1: Test Configuration Diagram

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya S8800 server running Avaya Aura® Session Manager Server	6.3.7.0.637008
Avaya S8800 server running Avaya Aura® System Manager Server	6.3.7 (Build No 6.3.0.8.5682-6.3.8.3204 Software Update Revision No: 6.3.7.7.2275)
Avaya S8800 server running Avaya Aura® Messaging Server	6.3
Avaya Communication Server 1000E/CPPM	Avaya Communication Server Release 7.6 Q+ Deplist 1 (created: 2014-07-23) and Service Pack 5 (Created: 2014-Jul-10)
Avaya IP SIP Phone 1140E	4.3
Avaya IP Unistim Phone 1165E	0x25C8J
Avaya IP Unistim Phone 2004	0604DCN
Spectralink 8440	4.7.0.2327

4. Configure Avaya Communication Server 1000

This section describes the steps to configure the Avaya CS 1000 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 a summary of tasks required for configuring the CS 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service and configure local SIP 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 SIP Line Route Data Block (RDB).
- Create SIP Line Virtual Trunks.
- Create a SIP Line Phone.

4.1. Prerequisite

This document assumes that the CS 1000 SIP Line server has been:

- Installed with CS 1000 Release 7.6 Linux Base.
- Joined CS 1000 Release 7.6 Security Domain.
- Deployed with SIP Line Application.

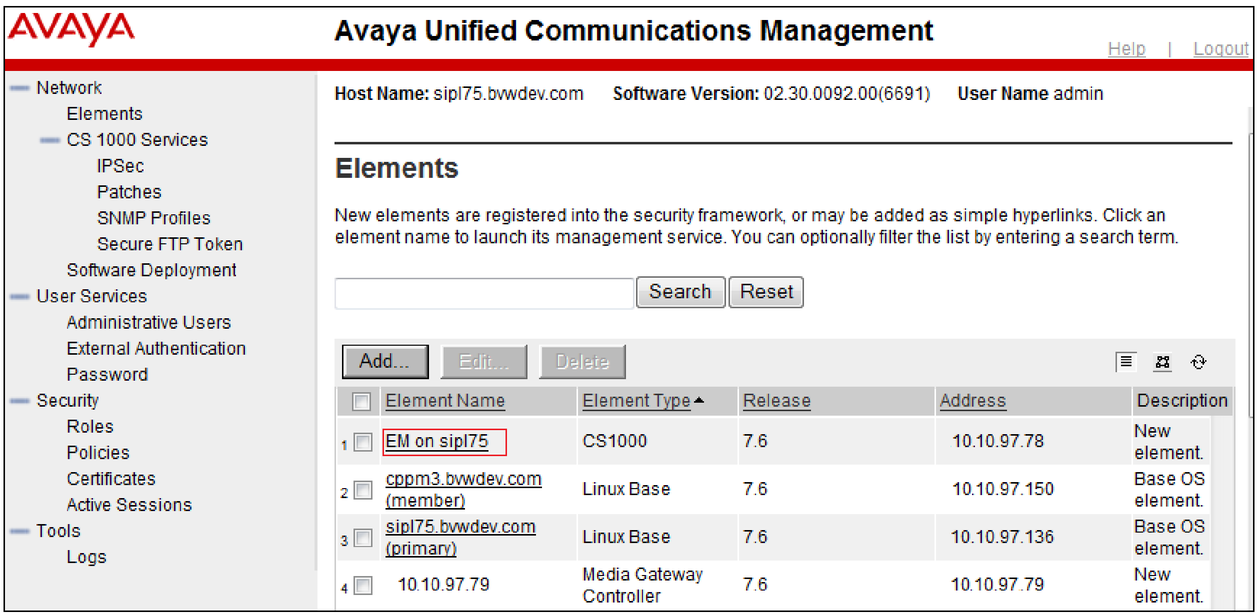
The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

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

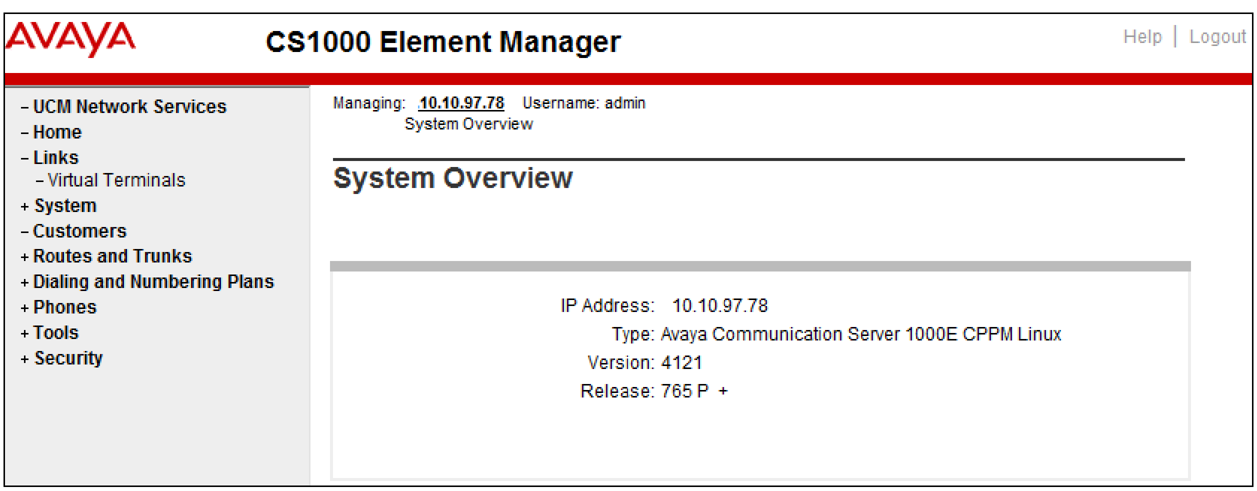
4.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

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 screen below. On the UCM home page, under the **Element Name** column, click on the Element Manager name of CS 1000 system that needs to be configured, in this sample that is **EM on sipl75**.



The CS 1000 Element Manager page appears as shown below.

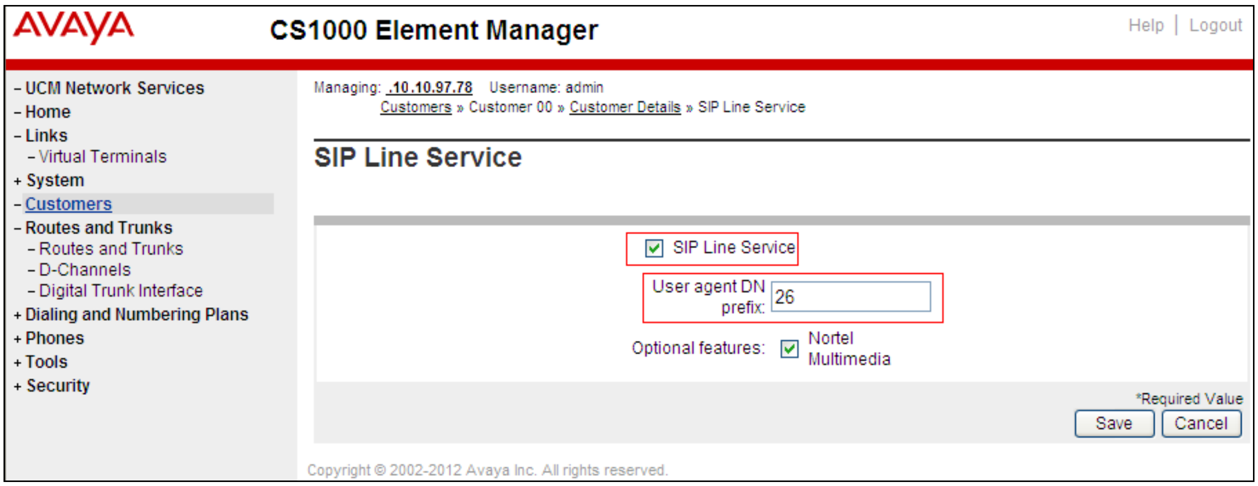


4.3. Enable SIP Line Service and configure local SIP domain

On the Element Manager page, navigate to **Customers** on the left menu. The list of Customer ID displays on the right, select the customer number (Customer 0) to be enabled with SIP Line Service (screen not shown). The screen below shows the SIP Line Service page.

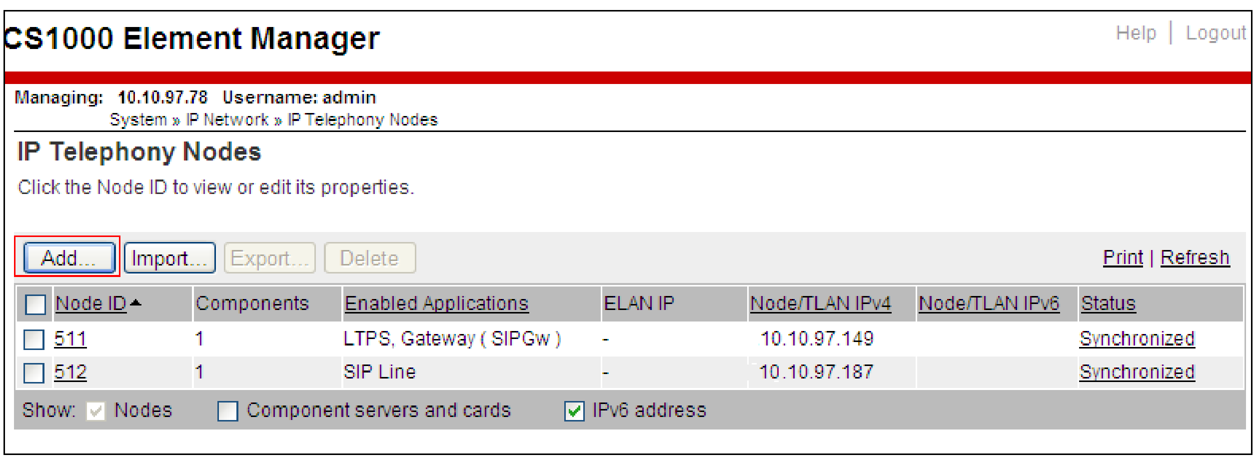
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box, e.g., **26** as shown below.

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



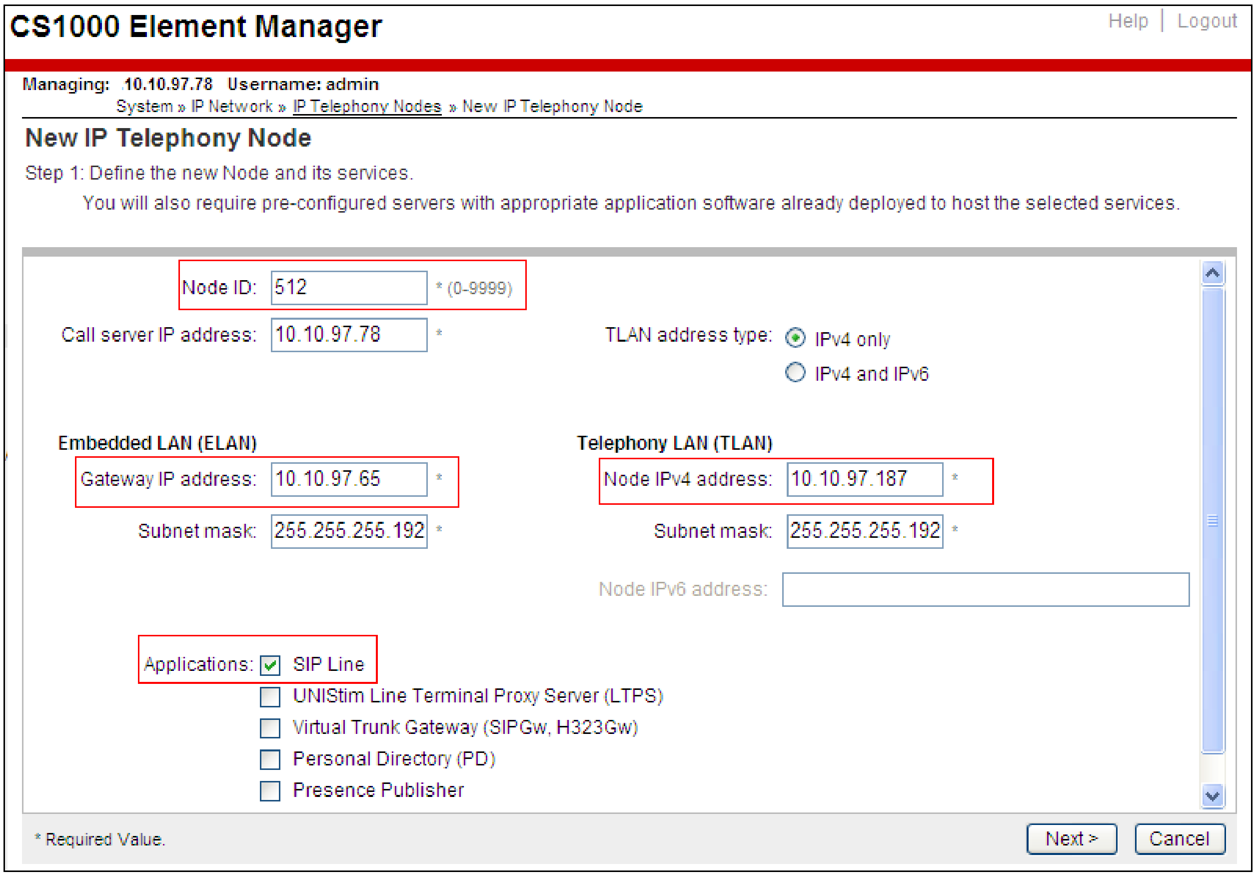
4.4. Create SIP Line Telephony Node

On the Element Manager page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as the screen below. Click **Add** button to add a new SIP Line Node to the IP Telephony Nodes.



The **new IP Telephony Node** page is displayed. Enter the information for each field shown below.

- **Node ID**: enter **512** which is the node ID of SIP Line server.
- **Telephony LAN (TLAN) Node IP Address**: Enter **10.10.97.187** which is the Node IP address of SIP Line.
- **Embedded LAN (ELAN)**: Enter **10.10.97.65** which is the gateway IP of Call server subnet.
- **Applications: SIP Line**: Check the check box to enable SIP Line service for this Node.

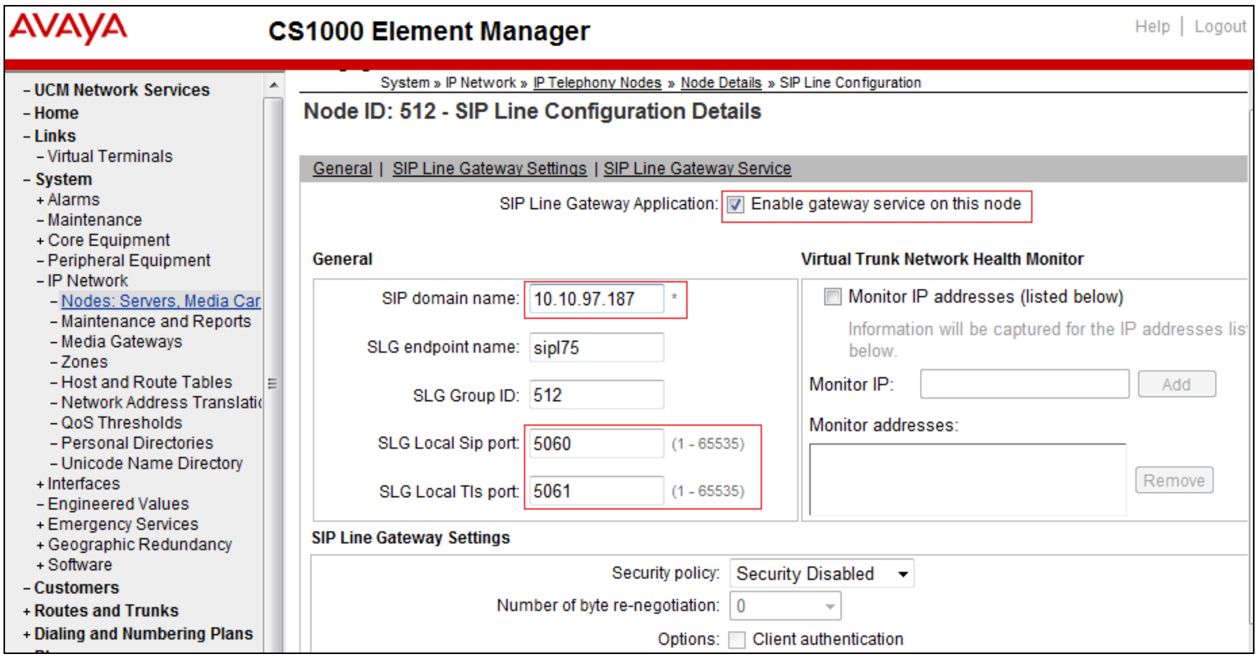


Click on the **Next** button to go to next page**.** The page, **New IP Telephony Node with Node ID** is displayed. On this page, in the **Select to Add** drop down menu list, select the desired server to add to the node. Click the **Add** button and select the check box next to the newly added server,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

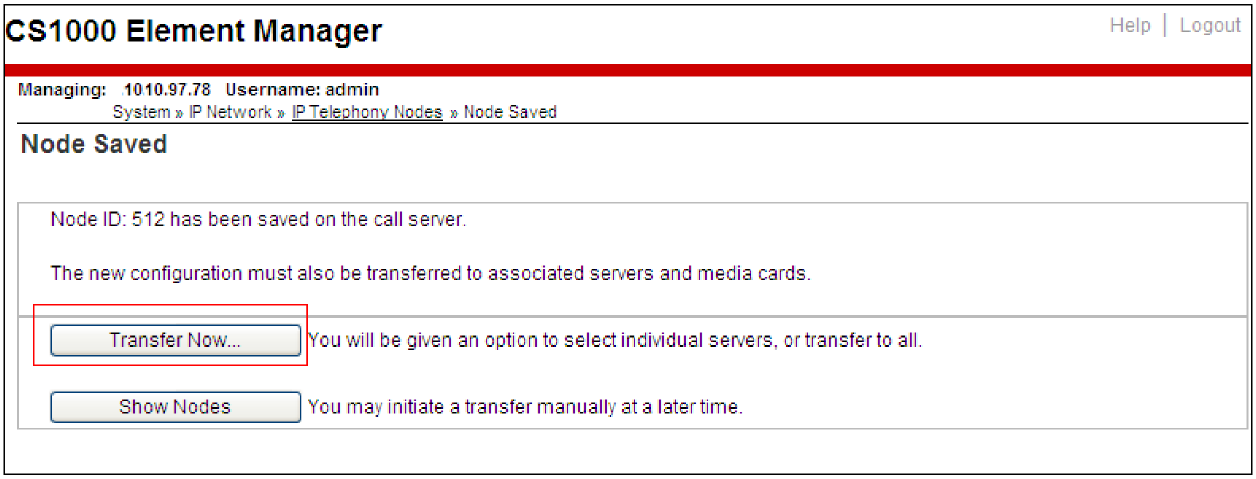
Click on the **Next** button to go to next page**.** The **SIP Line Configuration Details** page is displayed as the screen below.

- **SIP Line Gateway Application**: Check on the check box **Enable gateway service on this node**.
- In the **General** section:
  - **SIP domain name**: enter the SIP domain as "**10.10.97.187**".
  - **SLG Local Sip Port**: Enter port "**5060**".
  - **SLG Local Tls port**: Enter the port "**5061**".
  - Keep other sections as default.

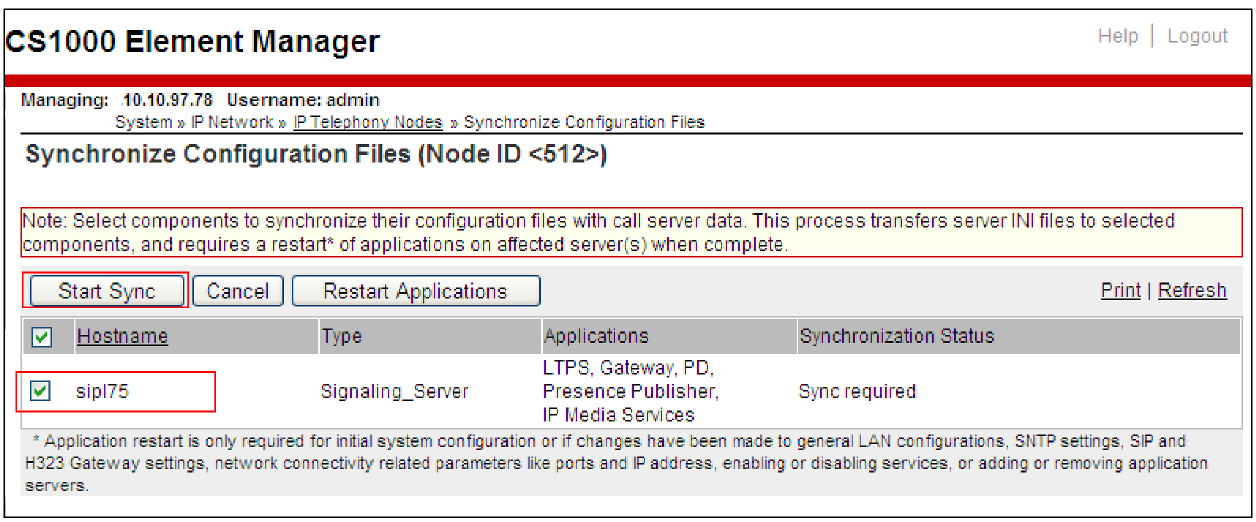
Click on the **Save** button to save the changes



Click **Next**. The **Confirm new Node details** page appears (screen not shown). Next click on the **Transfer Now** button in the **Node Saved** page as displayed in the screen below.



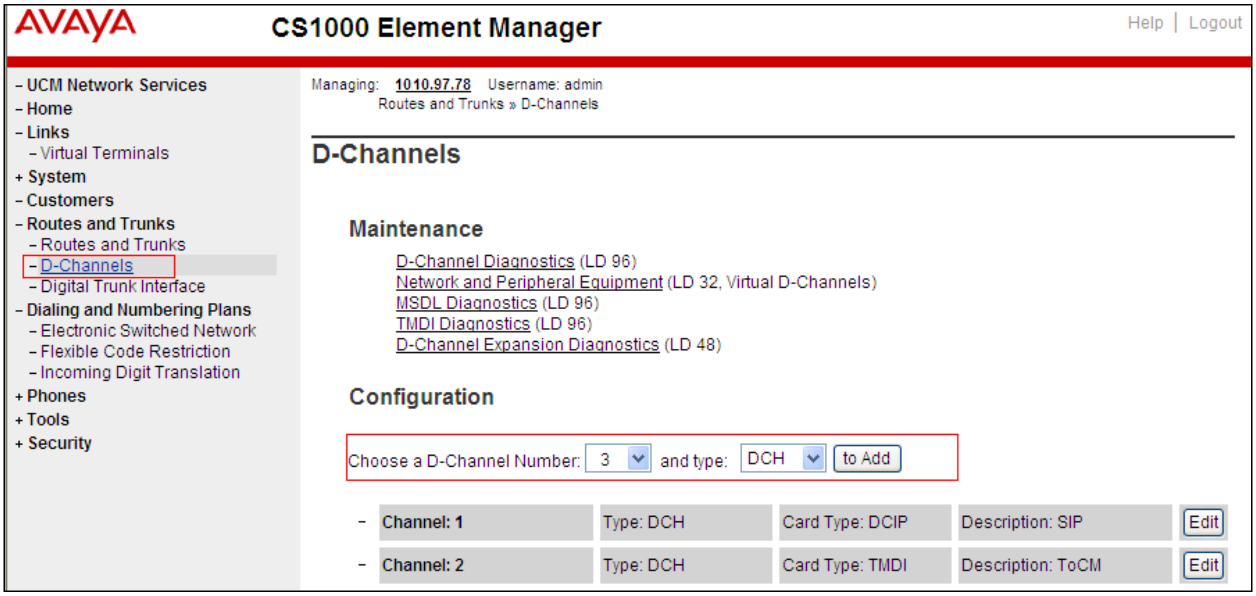
Click on the **Transfer Now** button, the **Synchronize Configuration Files (Node ID 512)** page is displayed. Select the SIP Line server that is associated with the changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers as shown below.



<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, check on the SIPLine server as shown above and click **Restart Application** button.

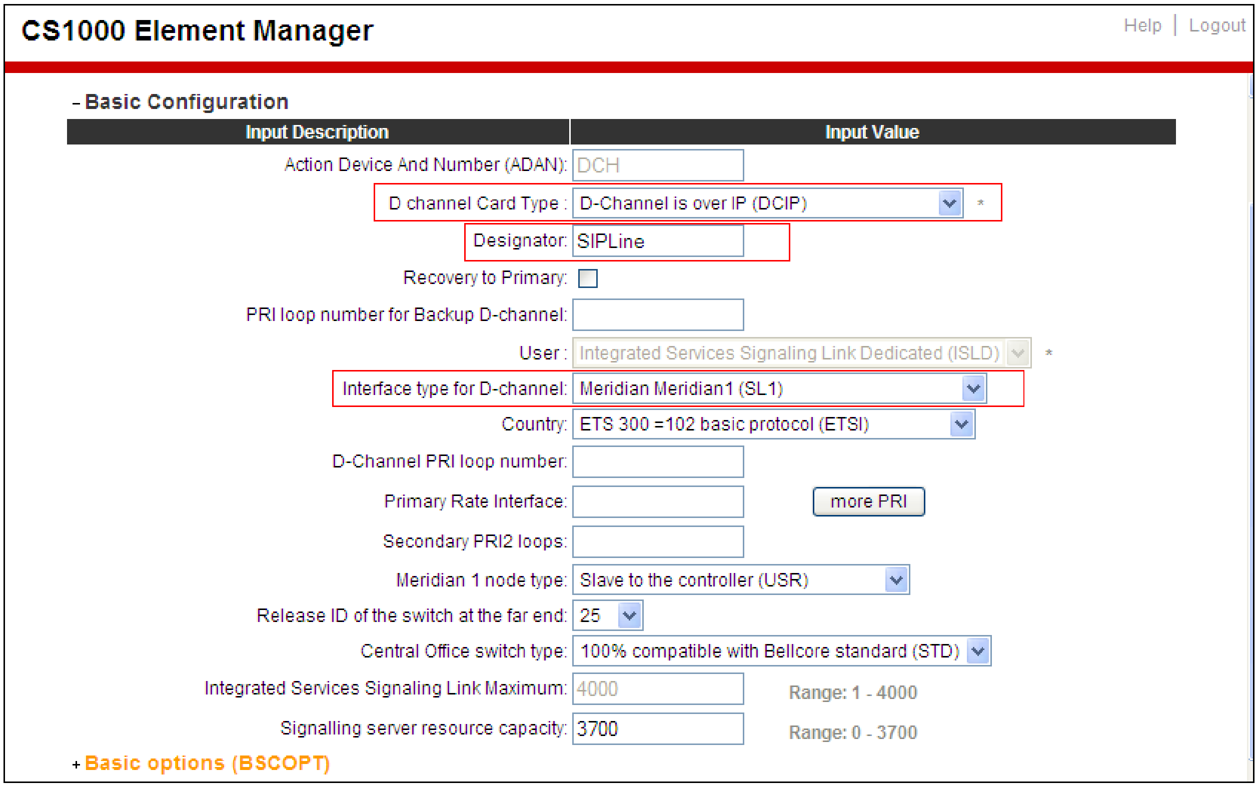
4.5. Create a D-Channel for SIP Line

On the Element Manager page, navigate to **Routes and Trunks → D-Channels**. The **D-Channels** page is displayed on the right, under the **Configuration** section as shown below, enter an available number in the **Choose a D-Channel Number** drop down menu, e.g., **3** and click on the "**to Add**" button.

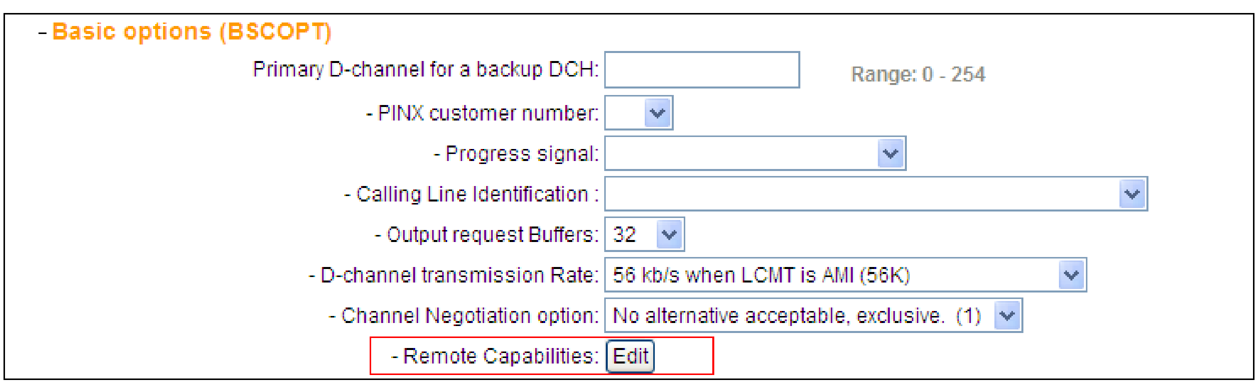


The **D-Channels 3 Property Configuration** page is displayed. In the **Basic Configuration** section:

- **D channel Card Type**: Select **D-Channel is over IP (DCIP)**.
- **Designator**: Enter a descriptive name, e.g., "**SIPLine**".
- **Interface type for D-channel:** Select **Meridian Meridian1 (SL1)**.
- Leave the other fields in the section at default values.

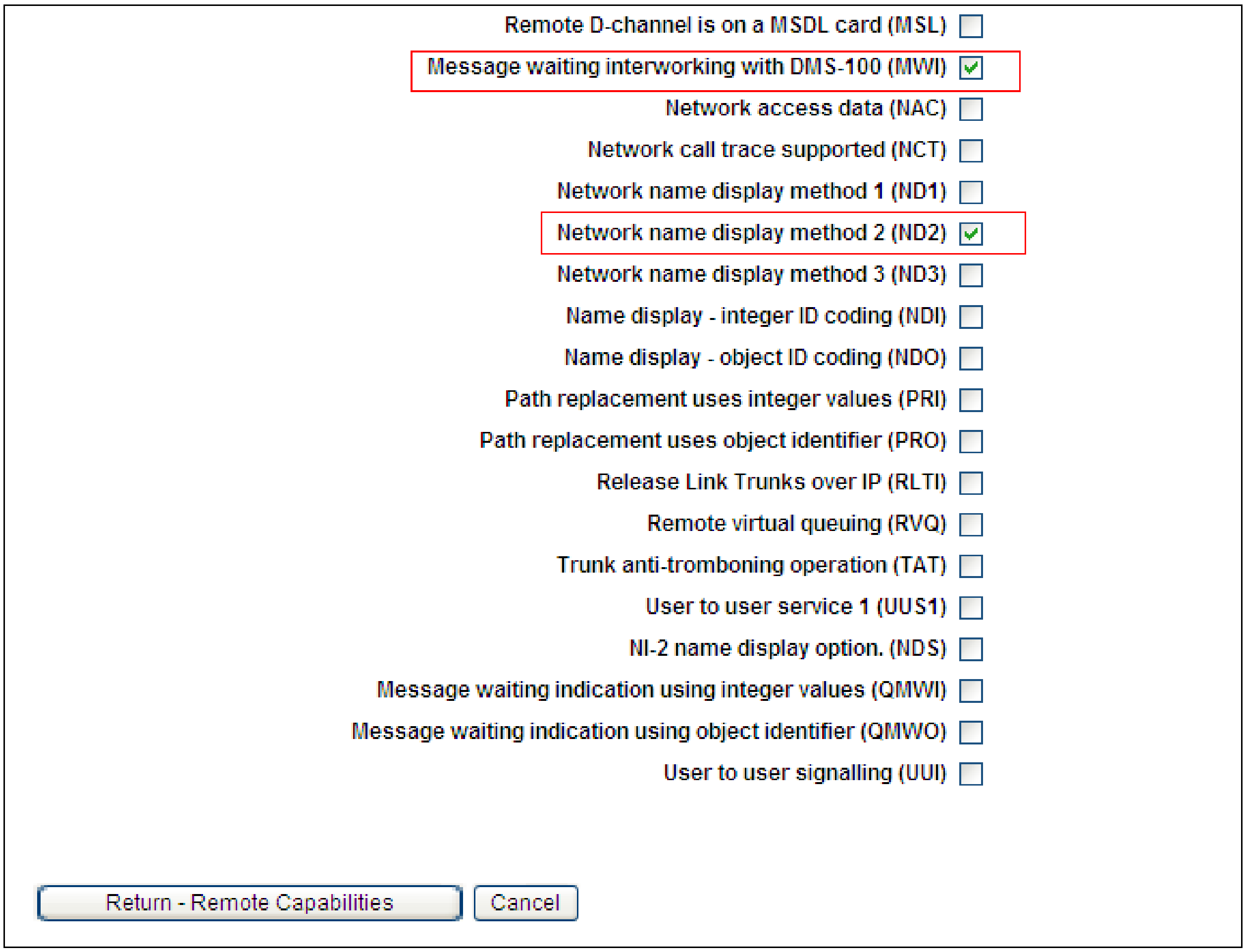


Click on the **Basic options (BSCOPT)** link to expand this section. The **Basic options (BSCOPT)** section is displayed as shown below. Click on **Edit** button to configure **Remote Capabilities (RCAP)**.



The **Remote Capabilities Configuration** page is displayed. Select the **Message waiting interworking with DMS-100 (MWI***)* and **Network name display method 2 (ND2)** check boxes. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** button to return the **D-Channel 3 Property Configuration** page.

Note that the **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints and **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.



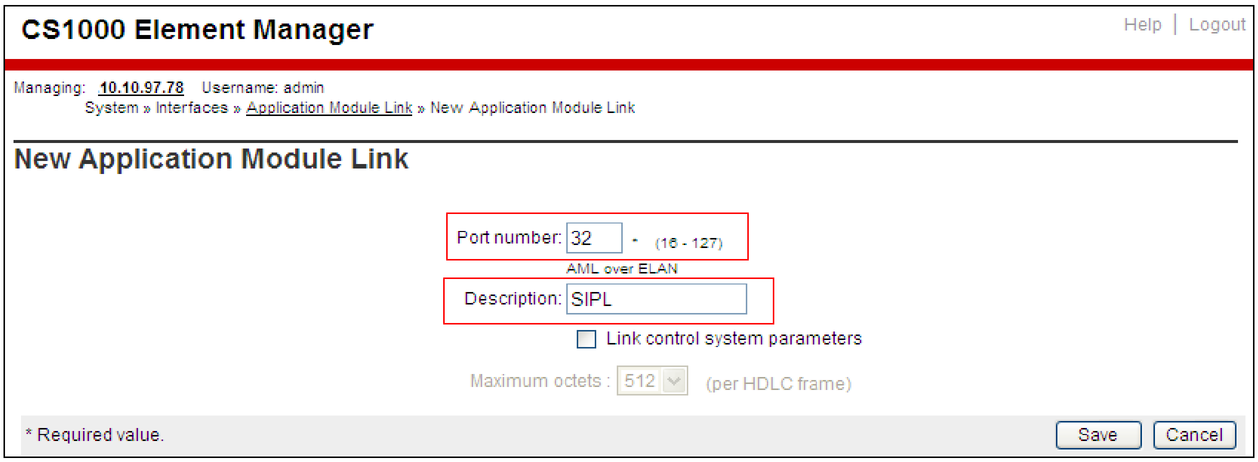
Leave the **Advance options (ADVOPT)** section at default.

Click on the **Submit** button at the bottom of the **D-Channel 3 Property Configuration** page to save changes and complete the creation of new D channel.

4.6. Create an Application Module Link (AML)

On the Element Manager page, navigate to **System → Interfaces → Application Module Link**. The **Application Module Link** page is displayed on the right (screen not shown), click on the **Add** button to add a new Application Module Link. The **New Application Module Link** page is displayed as below.

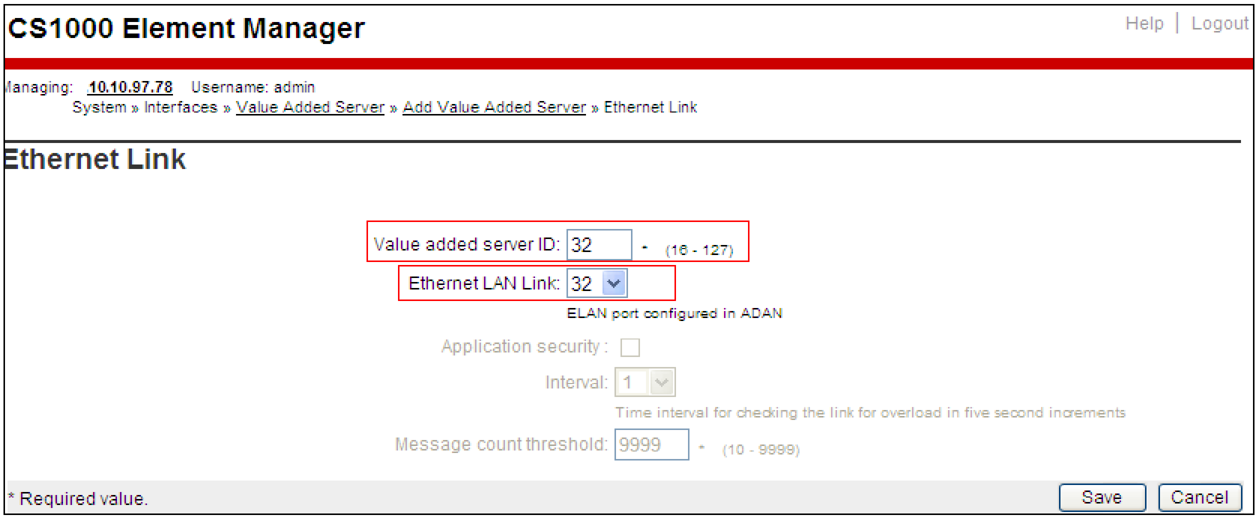
Enter an AML port number in the **Port number** text box, e.g., **32** and a descriptive name, e.g., "**SIPL**" in the **Description b**ox. Note that The AML of SIP Line Service can use any port from 32 to 127. In this case, SIP Line Service is configured to use port **32**. Click on the **Save** button to complete the addition of the new AML link.



4.7. Create a Value Added Server (VAS)

On the Element Manager home page, navigate to **System → Interfaces → Value Added Server**. The **Value Added Server** page is displayed on the right, click on the **Add** button. The **Add Value Added Server** page is displayed; select the link **Ethernet LAN Link**.

The **Ethernet Link** page is displayed as shown below. Enter a number in the **Value added server ID** field, e.g., **32** and in the **Ethernet LAN Link** drop down list, select the AML number of ELAN that was created in **Section 5.6**. Leave the other fields as default values and click on the **Save** button to complete the addition of the new **VAS**.

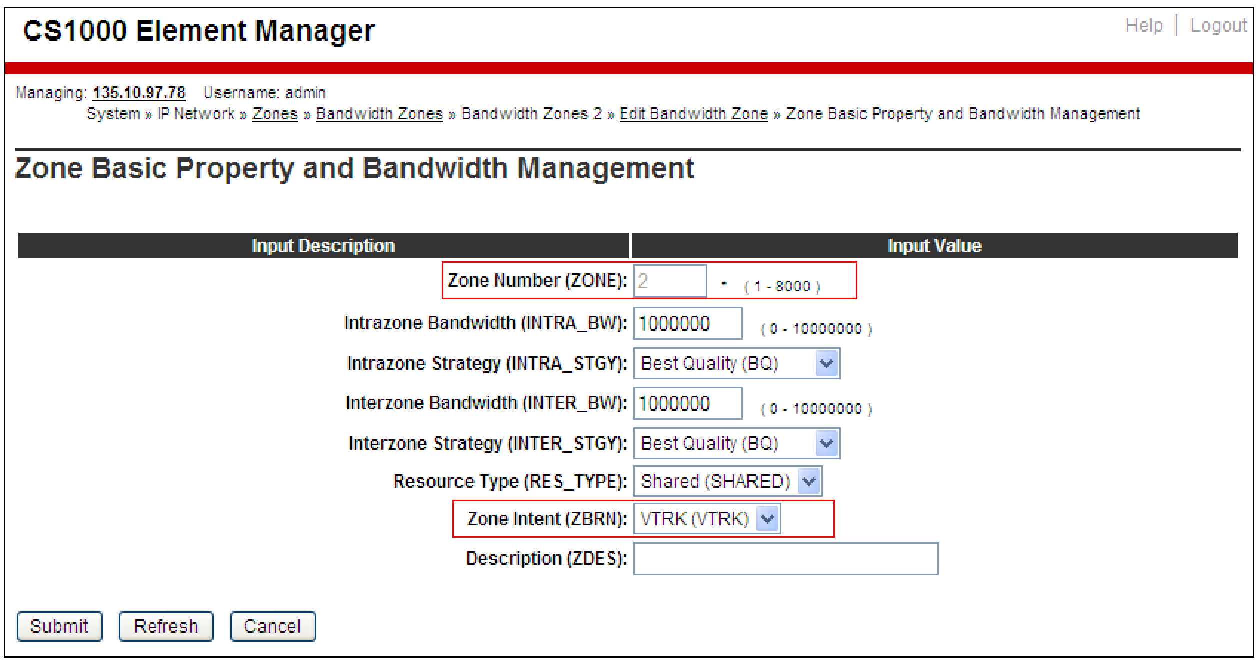


4.8. Create a Virtual Trunk Zone

On the Element Manager home page, navigate to menu **System → IP Network → Zones**. The **Zones** page is displayed 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 is displayed as shown the screen below.

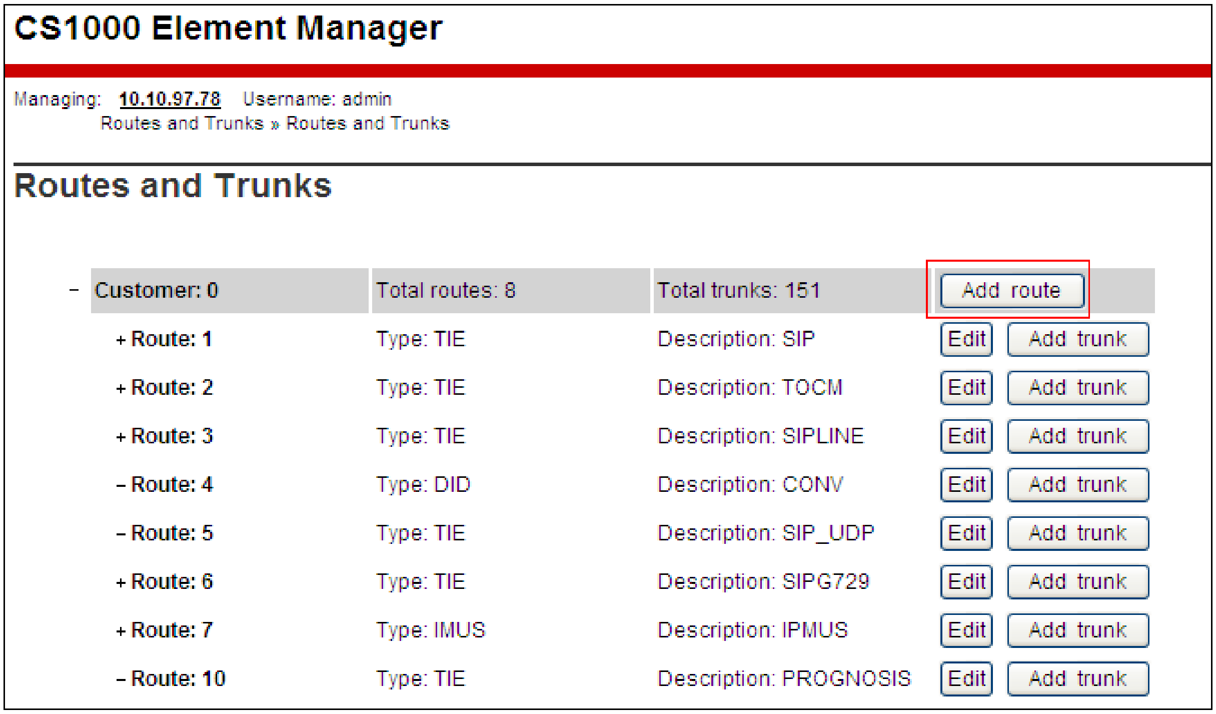
Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**. Leave other fields as default values and click on the **Submit** button to complete adding the Zone.

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

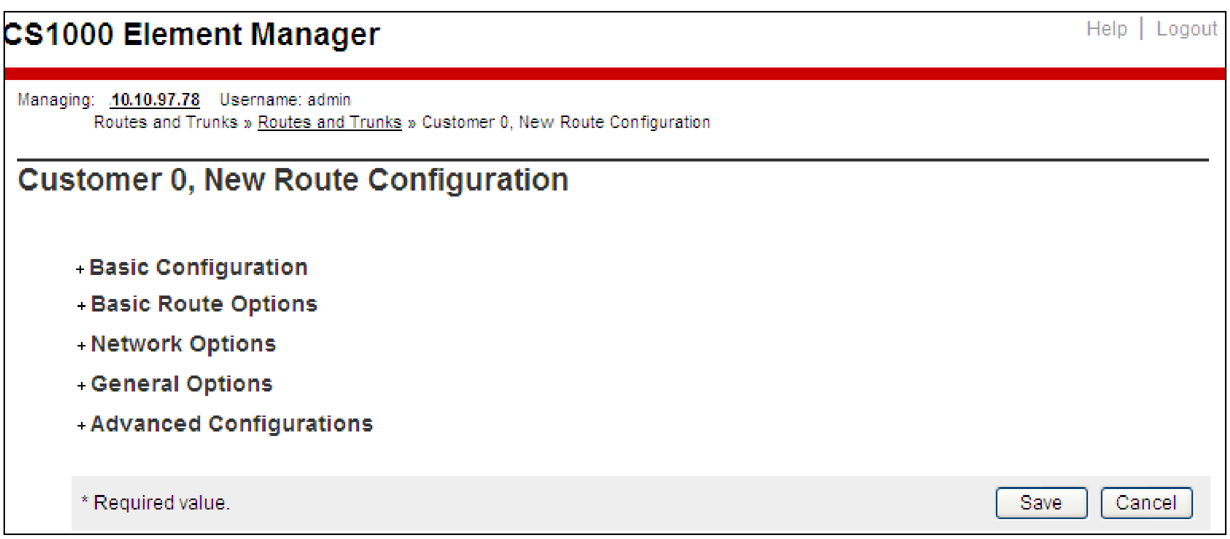


4.9. Create a SIP Line Route Data Block (RDB)

On the Element Manager home page, navigate to the menu **Routes and Trunks → Routes and Trunks**. The **Routes and Trunks** page is displayed on the right. In this page, click on the **Add route** button next to the customer number that the route will belong to.

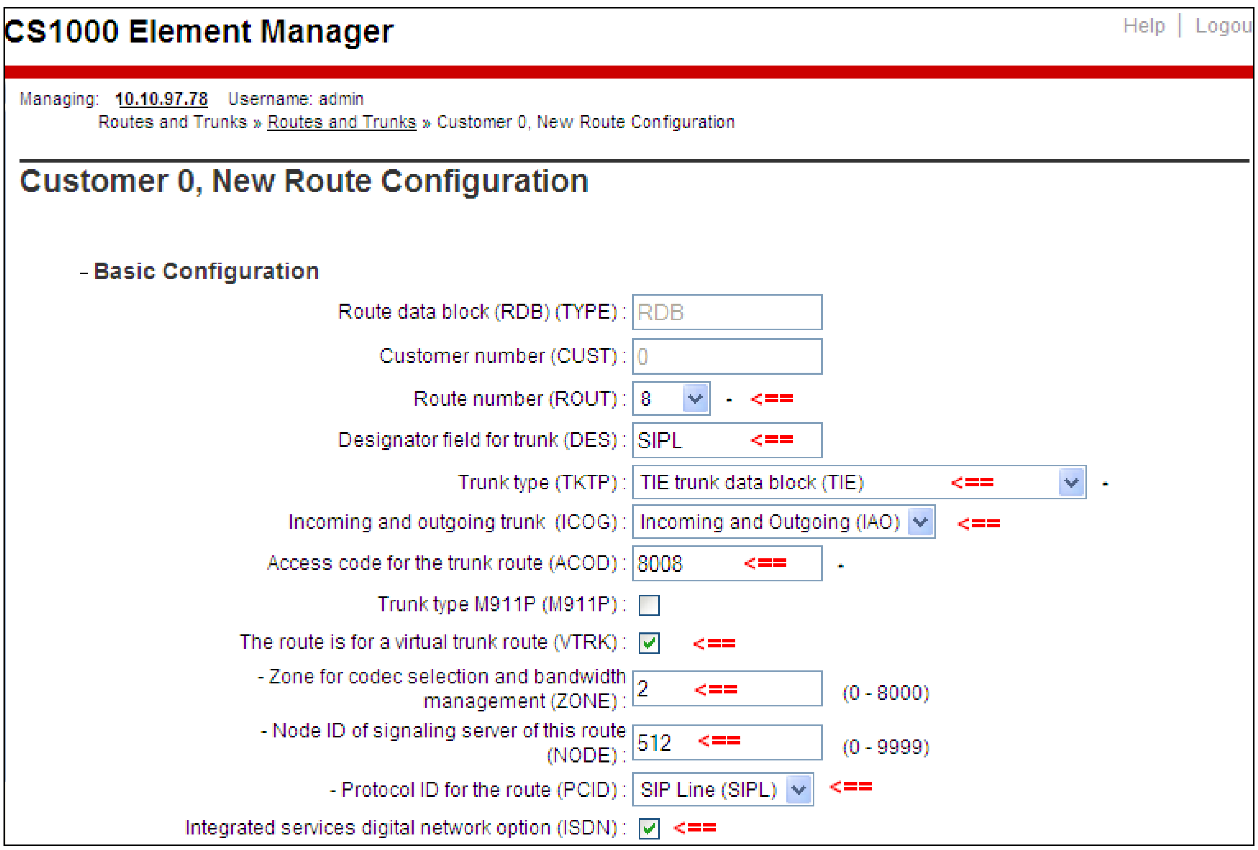


The **Customer ID, New Route Configuration** page is displayed. There are 5 sections in the new route configuration page.



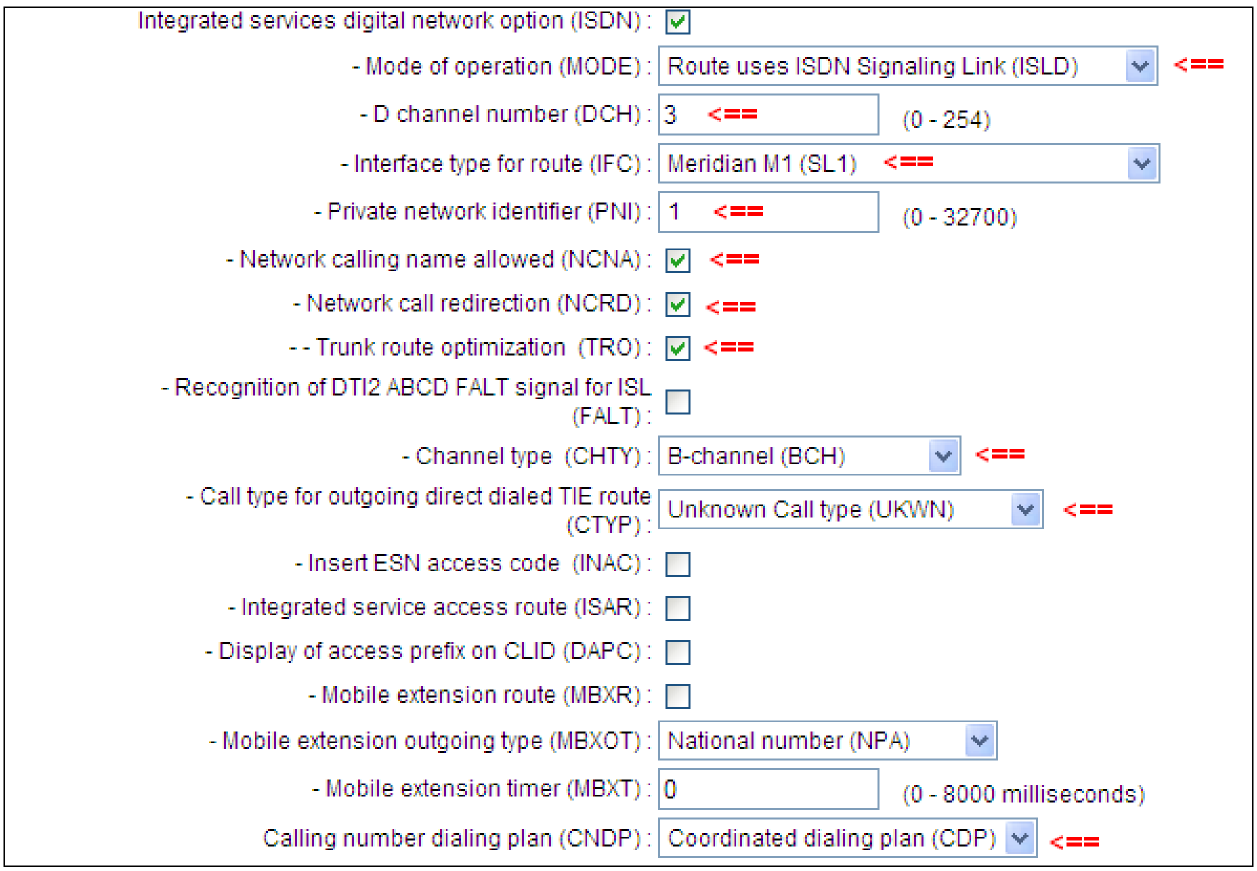
Expand the **Basic Configuration** section, and enter values as shown in the two screens below.

- **Route Number (ROUT)**: Select an available number in the list, e.g., **8**.
- **Designator field for trunk (DES)**: Enter a descriptive name, e.g., **SIPL**.
- **Trunk type (TKTP)**: Select **TIE trunk data block (TIE)**.
- **Incoming and outgoing trunk (ICOG)**: Select **Incoming and Outgoing (IAO)**.
- **Access code for trunk route (ACOD)**: Enter a number for ACOD, for example **8008**. Note that this number has to follow the dialing plan rule.
- **The route is for a virtual trunk route (VTRK)**: Check the checkbox.
- **Zone for codec selection and bandwidth management (ZONE*)***: Enter **2** which is the Virtual trunk zone number created in **Section 5.8**.
- **Node ID of signaling server of this route (NODE)**: Enter **512** which is the node ID of the SIP Line configured in **Section 5.4**.
- **Protocol ID for the route (PCID)**: Select **SIP Line (SIPL)** in the list.
- **Integrated services digital network option (ISDN)**: Check the check box.



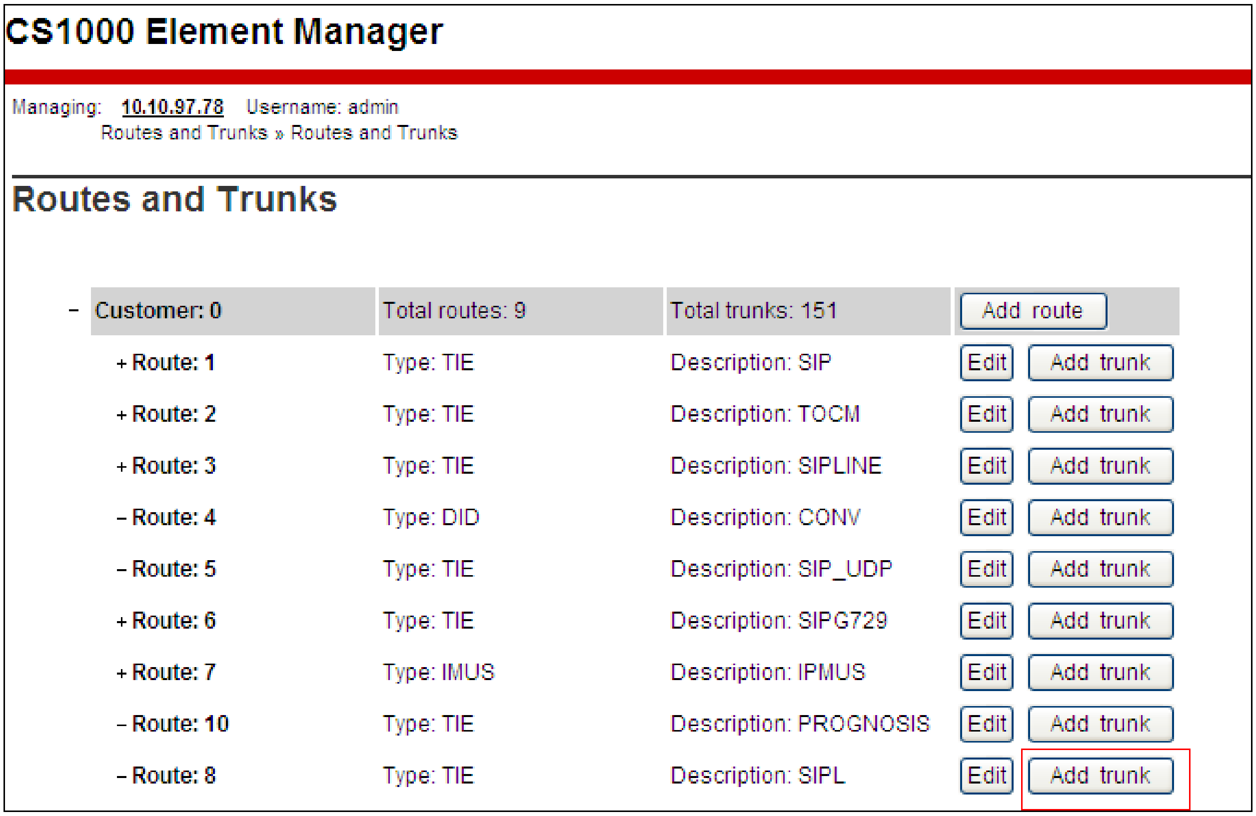
- **Mode of operation (MODE)**: Select **Route uses ISDN Signaling Link (ISLD)**.
- **D channel number (DCH)**: Enter **3** which is the D-channel number created in the **Section 5.5**.
- **Interface type for route (IFC)**: Select **Meridian M1 (SL1)**.
- **Network calling name allowed (NCNA):** Check the check box.
- **Network call redirection (NCRD):** Check the check box.
- **Trunk route optimization (TRO):** Check the check box.
- **Channel type (CHTY)**: B-channel (BCH).
- **Trunk route optimization (TRO)**: Check the check box.
- **Call type for outgoing direct dialed TIE route (CTYP)**: Select **Unknown Call type (UKWN)**.
- **Calling Number dialing plan (CNDP)**: Select **Coordinated 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 the addition of new route and save configuration.



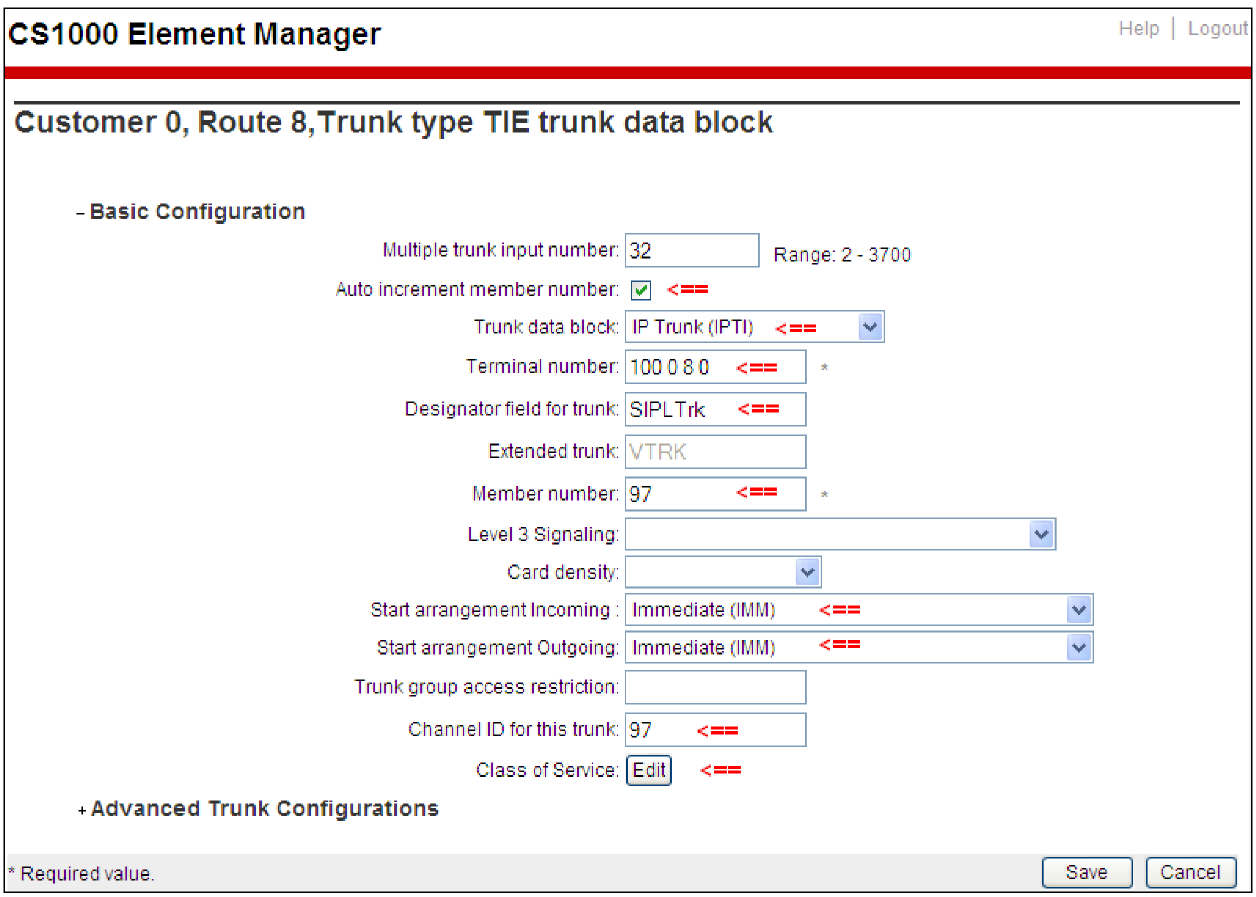
4.10. Create SIP Line Virtual Trunks

On the Element Manager home page, navigate to **Routes and Trunks → Routes and Trunks**. The **Routes and Trunks** page is displayed on the right, select the **Add trunk** button beside the route **8** that was created in the **Section 5.9** above to create new trunks.



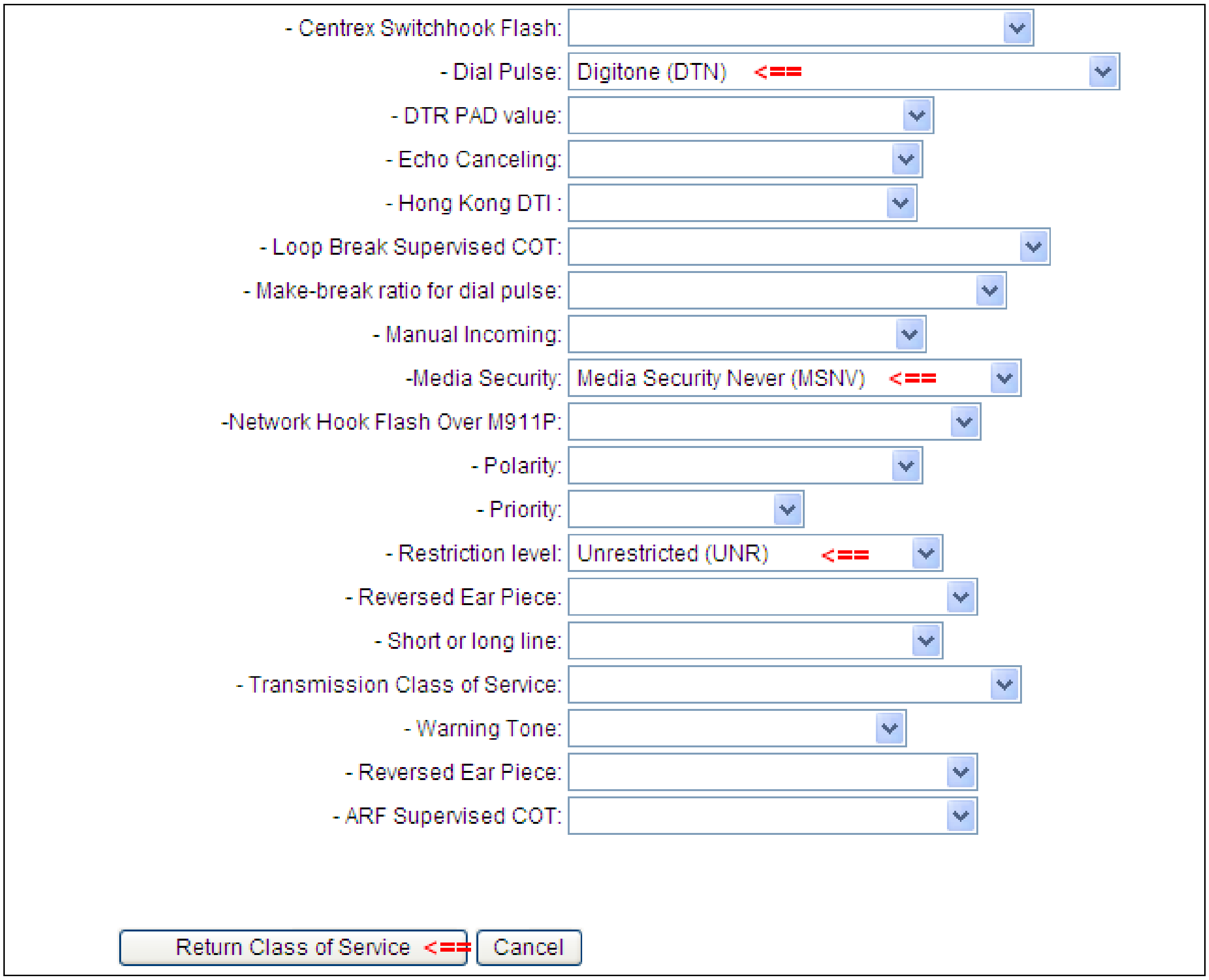
The **Customer 0, Route 8, Trunk type TIE trunk data block** page is displayed. Enter values for fields as shown below:

- **Multiple trunk input number**Enter **32** to create 32 trunks.
- **Auto increment member number**: Checked. The trunks are created incrementally.
- **Trunk data block**: Select **IP Trunk (IPTI)**.
- **Terminal Number**: **100 0 8 0**. Enter the first Terminal Number in a range of Terminal number.
- **Designator field for trunk**: Enter a descriptive name, e.g., "**SIPL Trk**".
- **Member number**: enter **97**. This is the ID of the trunk, just enter the first ID for the first trunk, next ID will be automatically created and incremented.
- **Start arrangement Incoming**: Select **Immediate (IMM)**.
- **Start arrangement Outgoing**: Select **Immediate (IMM)**.
- **Channel ID for this trunk**: **97**, this channel ID should be the same as the ID of Member Number and it has to be a unique number in the same type of trunk.



Click on the **Class of Service** button and assign following class of services as shown the screen below:

- **Dial Pulse** Select **Digitone (DTN)**.
- **Media security**: Select **Media Security Never (MSNV)**.
- **Restriction level**: Select **Unrestricted (UNR)**.
- Leave other class of services at default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page**.**



Leave the **Advance Trunk Configurations** section at default values and click on the **Save** button to complete the addition of new virtual trunks for SIP Line.

4.11. 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.

```
LD 20
Req prt
TYPE: uext
TN   104 0 0 2
DES  SL8741
TN   104 0 00 02 → Terminal number of Universal Extension of SIP Line phone
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL → Type of UXTY is SIP Line
MCCL YES
SIPN 0
SIP3 1  → 3rd SIP endpoint is enabled
FMCL 0
TLSV 0
SIPU 54009 → SIP user which is used in the SIP endpoint for registration
NDID 512  → The node ID of SIP Line.
SUPR NO
UXID
NUID
CFG_ZONE 00001 → Zone for SIP endpoint configured as MO
MRT
ERL
ECL  0
VSIT NO
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
XLST
SCPW 1234 → The password used to register to SIP Line server
SFLT NO
CAC_MFC 0
CLS  CTD FBA WTA LPR MTD FNA HTD TDD HFD CRPD → Depend on feature cls enabled
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXD ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```

```
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND 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
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 54009 0      MARP → The main directory number of SIP endpoint
        CPND
          CPND_LANG ROMAN
            NAME Poly1 54502
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 HOT U 2654009 MARP 0 → The Hot U with the prefix 26 configured in
adding SIP Line server.
     02 MSB → MSB key is used for Make Set busy feature on SIP endpoint
     03 CWT → CWT key is used for Call Waiting feature on SIP endpoint
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16
     20 RGA
     21 PRK
     22 RNP
     23
     24 PRS
     25 CHG
     26 CPN
```

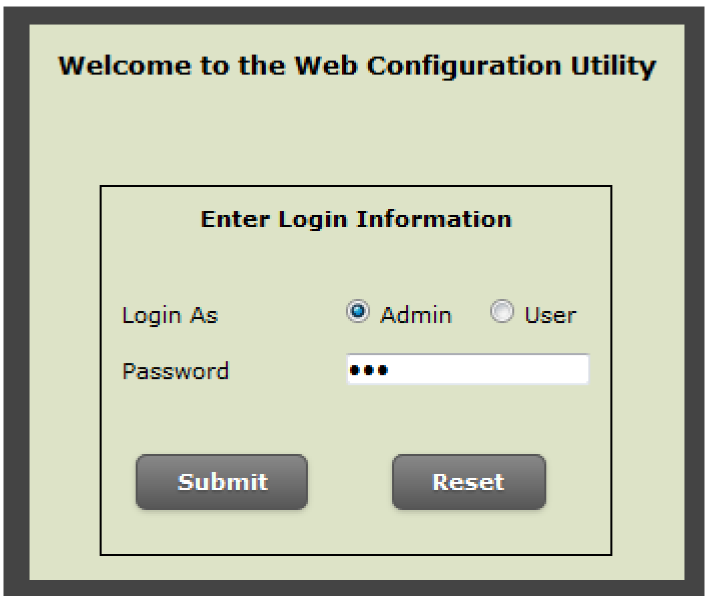
5. Configure Spectralink 84-Series

This section only describes the reference configuration of Spectralink 8440 SIP endpoint to work with the CS 1000. The installation and initial provisioning of the 8440 is described in the **Appendix B** section and more information on how to configure the 8440 please refer to the references in **Section 9**.

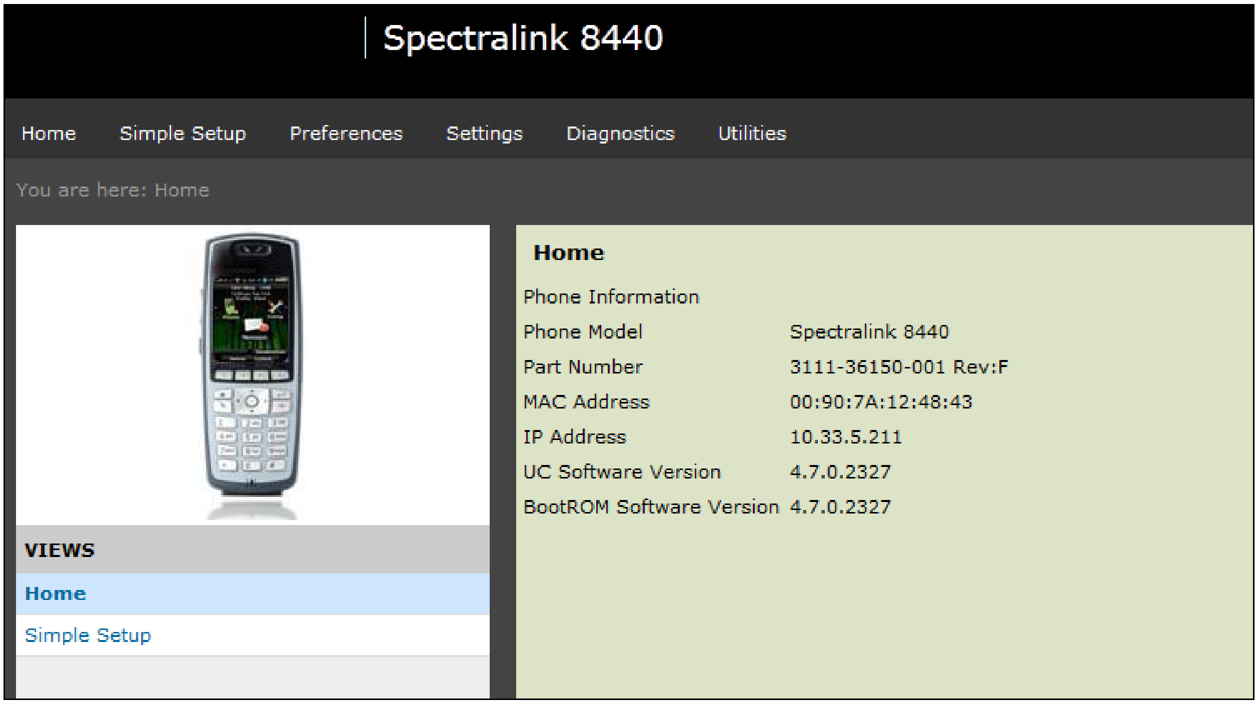
5.1. Login Spectralink 8440

This section shows how to log in to the home page of Spectralink 8440 to manage and configure the 8440 phone.

Open the web browser, and in the address box enter the 8440 IP address: http://ipaddress and the 8440 login page will appear as shown the screen below. Select the username, **Admin,** and enter its default password, **456** in the Password box. Click the **Submit** button to enter to the 8400 management page.



The screen below shows the home page of Spectralink 8440 phone which is one of the models in the 8400 series phone.



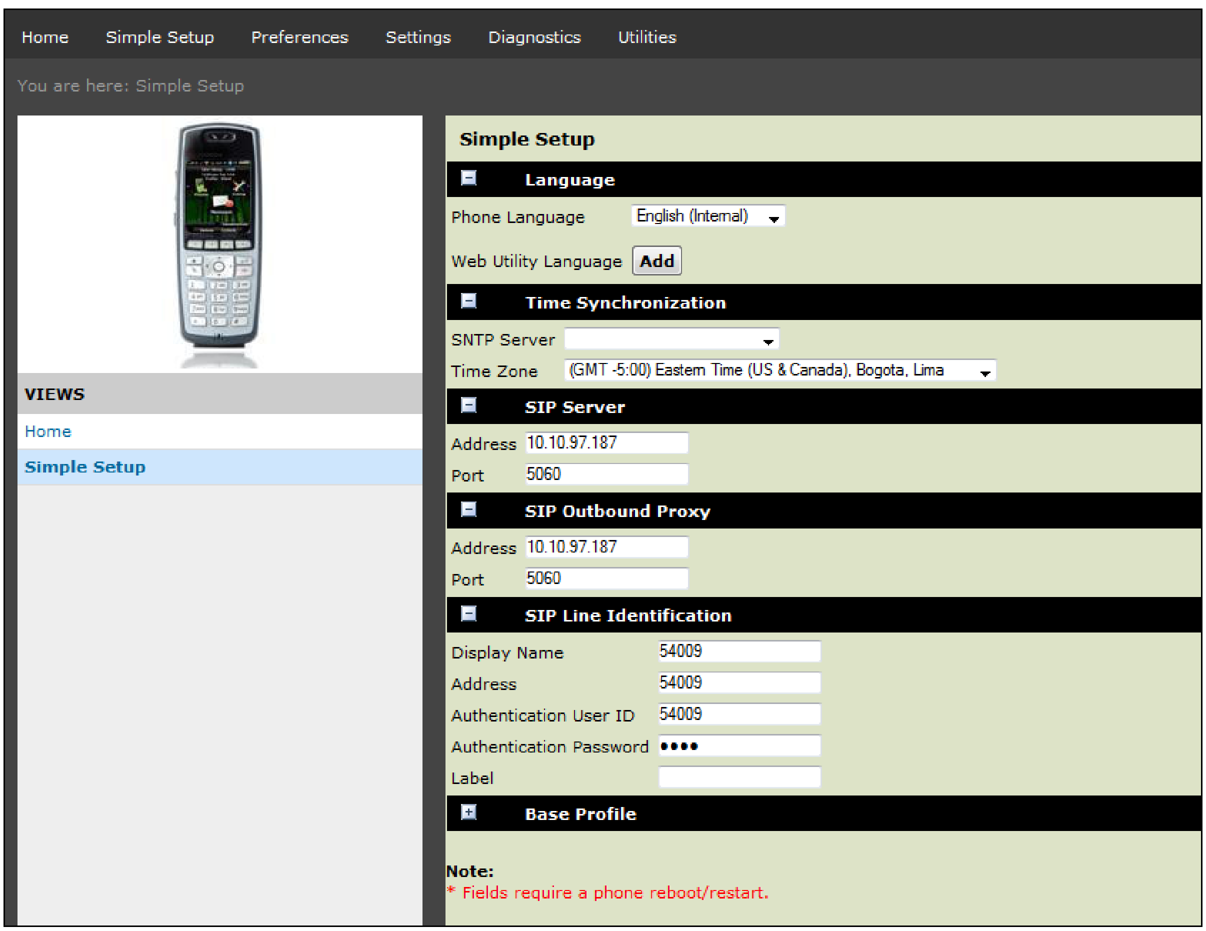
5.2. Register Spectralink 8440 to CS 1000 SIP Line

This section shows how to configure the 8440 telephone to register with the CS 1000 SIP Line gateway. On the homepage of the configuration screen, navigate to menu **Simple Setup**.

The **Simple Setup** page is displayed as shown in the screen below. Enter the values as shown below:

- **SIP Server:**
  - **Address**: enter **10.10.97.187** → which is node IP address of CS 1000 SIP Line server.
  - **Port:** enter **5060** → which is local sip port of CS 1000 SIP Line.
- **SIP Outbound proxy:**
  - **Address**: enter **10.10.97.187** →Use the same Node IP address of SIP Line server.
  - **Port**: 5060
- **SIP Line Identification:**
  - **Display Name**: Enter a descriptive name, e.g., **54009**
  - **Address**: Enter 54009
  - **Authentication User ID**: enter **54009** → This user ID that is configured in the field **SIPU** of Terminal Number of SIP Line phone in the **Section 5.11**
  - **Authentication Password**: enter **1234** → This password that is configured in the field **SCPW** of UEXT Terminal Number for SIP Line phone in the **Section 5.11**

Click on the **Save** button to save changes. Note that the phone needs to be rebooted for the changes to take effect.

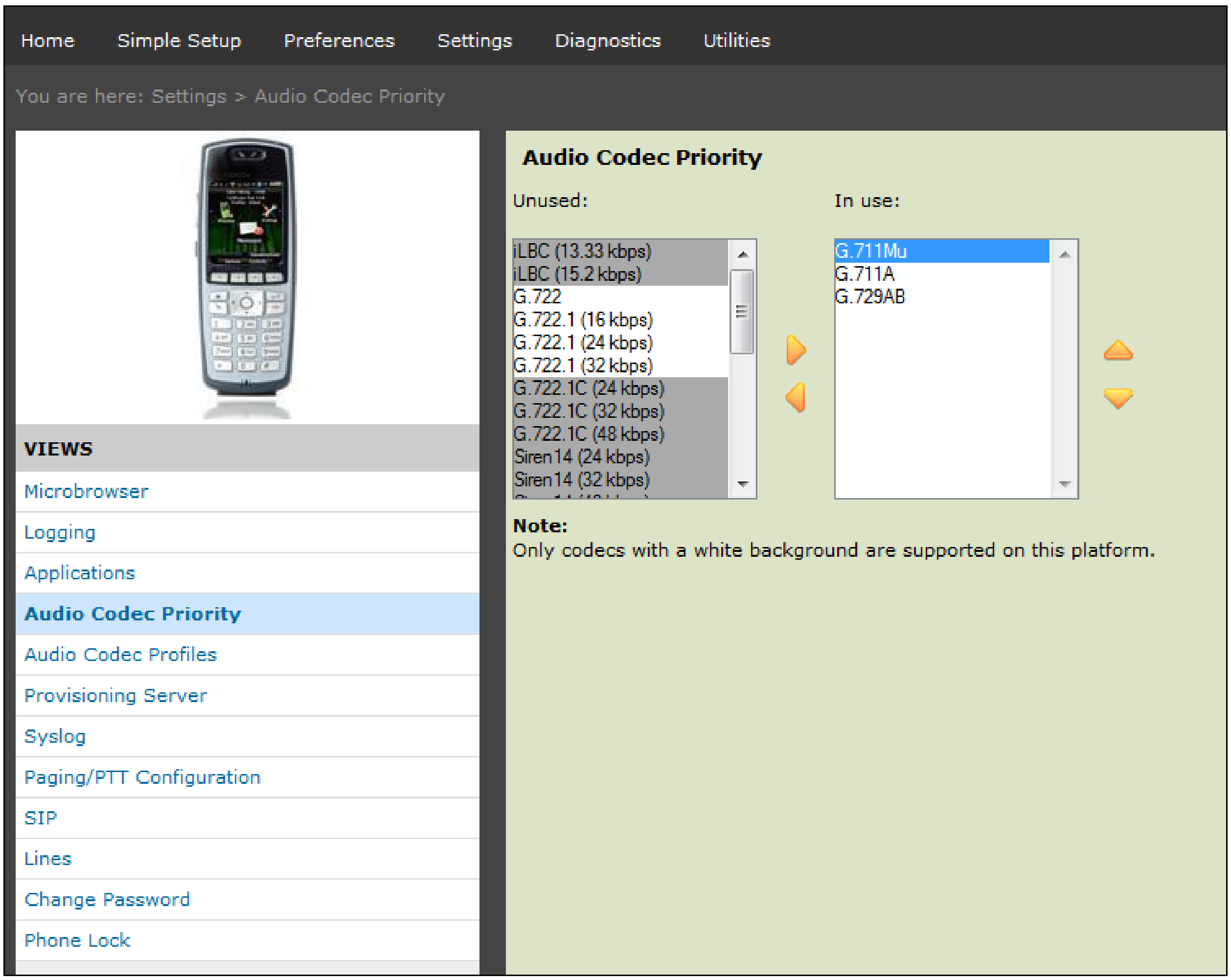


5.3. Configure Codec settings

This section shows how to set the codec on the Spectralink 8440 phone. The compliance testing has been done on three codecs: G.711 Mu, G711A law and G729.

On the homepage of the 8440, navigate to menu **Settings → Audio Codec Priority**, the **Audio Codec Priority** page is displayed as shown below. The list of audio codecs being used appear under the **In use** column. To use the codec G711Mu as the first choice, move it up to the top of the **In Use** list, repeat the same procedure for other codecs. Click on the **Save** button to save changes.

Note that the codec G.722 should not be used to avoid the transfer issue as mentioned in **Section 2.2 Test Result**.



6. Verification Steps

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

- From the 8440 phone, verify the Spectralink 8440 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.
- Verify that the 8440 phone registers successfully with the CS 1000 SIP Line Gateway server by using the CS 1000 Linux command.
- Log in to the SIP Line server as an administrator by using the Avaya account. Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked.

```
[admin@sipl75 ~]$ slgSetShowByUID 54009

=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------   ---------
          54009       54009     104-00-00-02        1      0  0x95545d0         SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = -1

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.33.5.211:5060:udp
         Type          = Unknown
         UserAgent     = Spectralink-SL_8440-UA/4.7.0.2327
         x-nt-guid     = 8bcd28934e1e3a341d5a0af58bcd1d89
         RegDescrip    =
         RegStatus     = 1
         PbxReason     = OK
         SipCode       = 200
         hTransc       = (nil)
         Expire        = 3600
         Nonce         = 04434b9bba35517ebb52e30d4d7963a2
         NonceCount    = 2
         hTimer        = 0x94e6eb8
         TimeRemain    = 2755
         Stale         = 0
         Outbound      = 0
         ClientGUID    = 0
         MSec CLS      = MSNV (MSEC-Never)
         Contact       = sip:54009@10.33.5.211
         KeyNum        = 255
         AutoAnswer    = NO

        Key  Func  Lamp  Label
        0    2     0     54009
        1    126   0     2654009
        2    29    0
```

- Place a call from and to the 8440 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.

7. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Spectralink 84-Series SIP telephone version 4.7.0 is considered to be in compliance with Avaya Communication Server 1000 Release 7.6.

8. Additional References

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

[1] Avaya CS 1000 Documents:
Avaya Communication Server 1000E Installation and Commissioning
Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6
Avaya Communication Server 1000 Element Manager System Reference – Administration
Avaya Communication Server 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

Product documentation for the Spectralink 84-Series products may be found at:

http://www.spectralink.com/product-information/wi-fi/spectralink-84-series-wireless-telephones

Appendix B

The Spectralink Software that you download contains configuration file templates, valid XML files that you can edit using an XML editor. These files contain all the parameters explained in this document that provision the handsets with features and settings. The configuration files are very flexible: you can rearrange the parameters within the file, move parameters to new files, or create your own configuration files with only those parameters you want. This flexibility is useful when you want to apply the same features and settings to a large number of handsets. Use of the configuration files to provision the handsets with features and settings is called the centralized provision method – the configuration files enable you to store a single set of configuration files on a central provisioning server and configure all of your handsets to read the same set of files. You can also configure a subset of handsets to use only specific files, thereby deploying different handsets with different sets of features.

Spectralink recommends that you configure handsets using the centralized provisioning method. By default, Spectralink sets FTP as the provisioning protocol on all Spectralink handsets. However, you can also configure individual handsets using the handset's menu system, accessible through the local user keypad interface, or you can configure select parameters by using the Web Configuration Utility.

You will need to keep in mind that there is a hierarchy among the configuration methods and settings. Using a higher-priority method will override settings you make using a lower-priority method. The following lists all of the available ways to set features and settings for the handsets. Spectralink strongly recommends becoming familiar with each of the configuration methods.

To download the Spectralink Software:

1 Access Spectralink Software from the Spectralink Support Website:
http://support.spectralink.com/products/wi-fi/spectralink-84-series-wireless-telephone

2 Acknowledge that you read the notices, accept the agreement, and choose **Submit**.

3 Save the Spectralink Software ZIP file download.

4 Extract (uncompress) the ZIP file.

Copy all files from the distribution ZIP file to working directory on the provisioning server, maintaining the same folder hierarchy. To simplify provisioning, Spectralink recommends, as a best practice, to start creating new configuration files from unedited template files containing the default values. Rename the template file to your specific file name as you configure and add specific parameter values for your site.

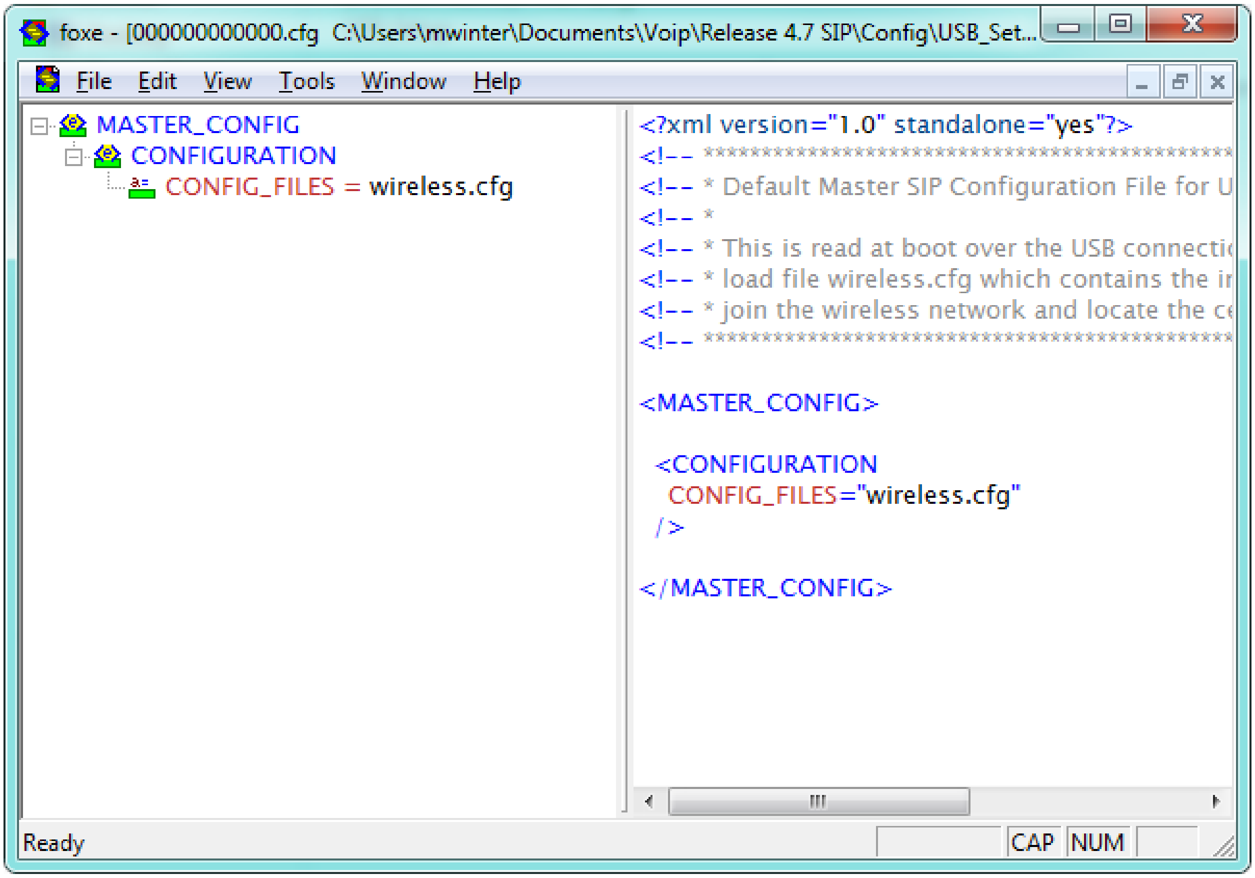
You can create as many configuration files as you want and your configuration files can contain any combination of parameters you put in them. You can put all parameters into one file or, for example, you can put SIP server parameters in one file and handset features parameters in another file. Configuration file variances are explained in the *Spectralink 84-Series Wireless Telephone Deployment Guide*: http://support.spectralink.com/resources/spectralink-84-series-wireless-telephone-deployment-guide.

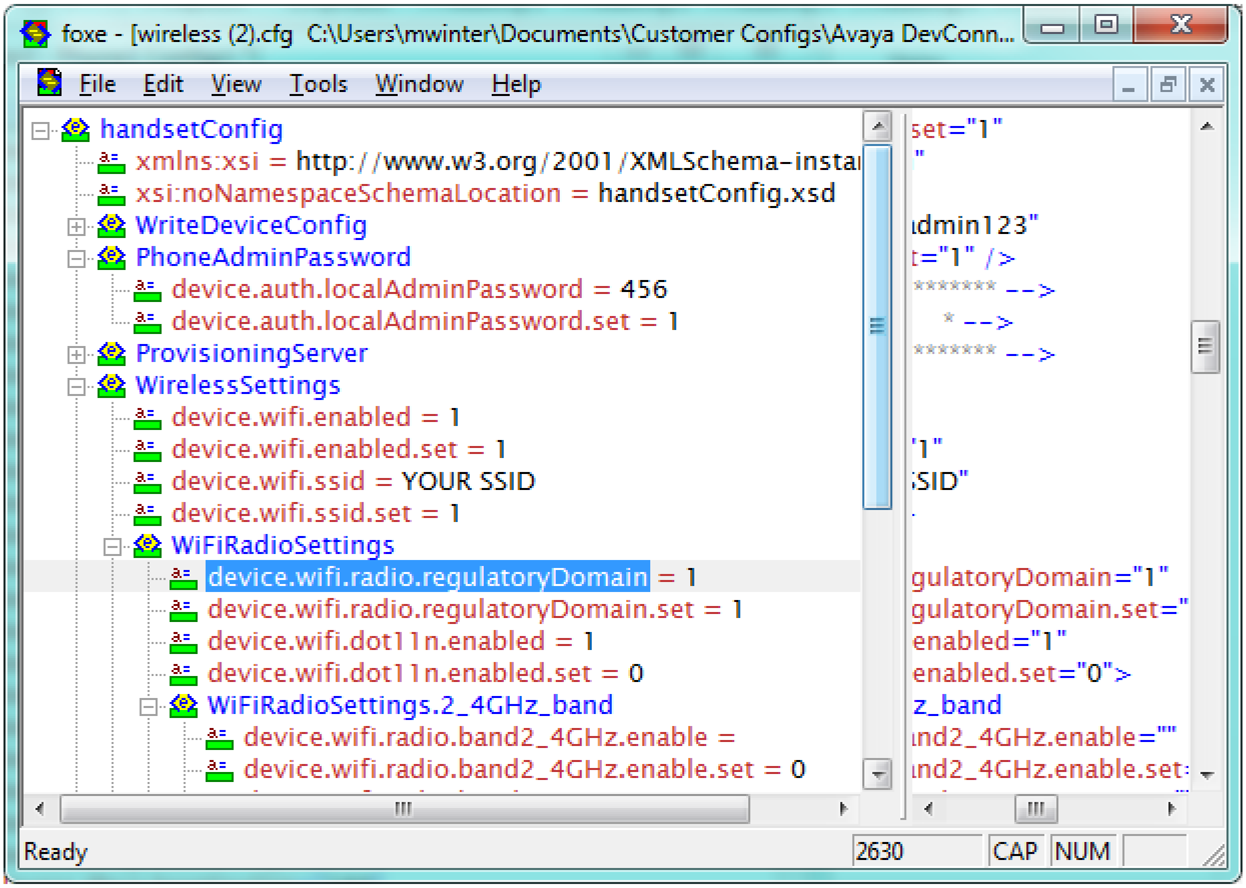
For large-scale deployments, the centralized provisioning method using configuration files is strongly recommended. For smaller scale deployments, the Web Configuration Utility or local interface may be used, but administrators need to be aware that settings made using these methods will override settings made using configuration files.

Authenticate the 8400 telephone to the wireless network

Because the handsets cannot access the wireless LAN before their wireless settings are configured, you will need to establish a wired connection between a computer and each handset and load the wireless settings via a USB Micro B connection. We will call this computer the initial provisioning computer. It is a temporary setup and does not require exceptional resources in the computer. Only one 84-Series handset is loaded at a time. Requirements are:

- USB port
- USB Micro B cable (available from Spectralink)
- Enough memory for the operation (minimal)
- FTP installed
- The original 000000000000.cfg file from the USB_Setup folder.
- The wireless.cfg file


foxe - [000000000000.cfg C:\Users\mwinter\Documents\Voip\Release 4.7 SIP\Config\USB_Set...
File Edit View Tools Window Help
MASTER_CONFIG
CONFIGURATION
CONFIG_FILES = wireless.cfg
<?xml version="1.0" standalone="yes"?>
<!-- * Default Master SIP Configuration File for U
<!-- * This is read at boot over the USB connecti
<!-- * load file wireless.cfg which contains the i
<!-- * join the wireless network and locate the c
<MASTER_CONFIG>
<CONFIGURATION
CONFIG_FILES="wireless.cfg"
/>
</MASTER_CONFIG>
Ready
CAP NUM



If the initial provisioning computer is not running Microsoft Windows 7, you will need to load a USB driver so that the computer can detect your Spectralink 84-Series handsets as a USB network device. Copy the correct 84xx.inf to it, using the steps itemized below, You will add the handset as a network device with Windows Add New Hardware wizard.

The 84xx.inf file applies to 32-bit computers running Microsoft Windows® XP SP3 and Microsoft Vista® SP1. If you are using a 64-bit computer running Microsoft Windows Vista operating system, you must use the 84xx-64.inf file.

Computers running Windows 7 or Linux do not require 84xx.inf or 84xx-64.inf.

To enable the handset's networking capabilities:

1 Log into the computer as the administrator.

2 Download and copy 84xx.inf onto your 32-bit computer or copy 84xx-64.inf onto your 64-bit computer from the Spectralink support site to an accessible location.

When the 84-Series handset is plugged in (using the USB cable) and the computer asks for a driver, specify the location where you saved the .inf file.

Download the wireless configuration to the handset

1 Ensure that the initial provisioning computer is functioning as an FTP server.

2 On the initial provisioning computer load the wireless.cfg file into the FTP root directory.

3 Apply power to 84xx handset.

4 Connect micro-USB cable between the 84xx handset and initial provisioning computer.

5 (Conditional) The Found New Hardware wizard opens. Connecting the handset to the initial provisioning computer launches the Found New Hardware wizard automatically. The Found New Hardware wizard only displays the first time you use each USB slot on your computer.

a Select No, not this time, and click Next.

b Select Install from a list or specific location (Advanced) and click Next.

c Select Search for the best driver in these locations.

d Select the check box for Include this location in the search:

e Browse to your 84xx.inf or 84xx-64.inf and click Next.

f The Linux USB Ethernet/RNDIS Gadget is installed.

g A warning will be displayed indicating this driver has not passed Windows Logo testing. Select Continue Anyway.

h Click Finish.

6 The handset will download the wireless configuration and then reboot making a tweedle noise when finished.

7 (Conditional) If handsets do not immediately (within 10 seconds) download and reboot after plugging the USB into them, you can manually force the configuration download by navigating to the Settings menu on the handset: **Settings**> **(1)Basic Settings**> **(6)Update Configuration> Yes**. If you use this option, **Updating…** remains on the display until it is finished.

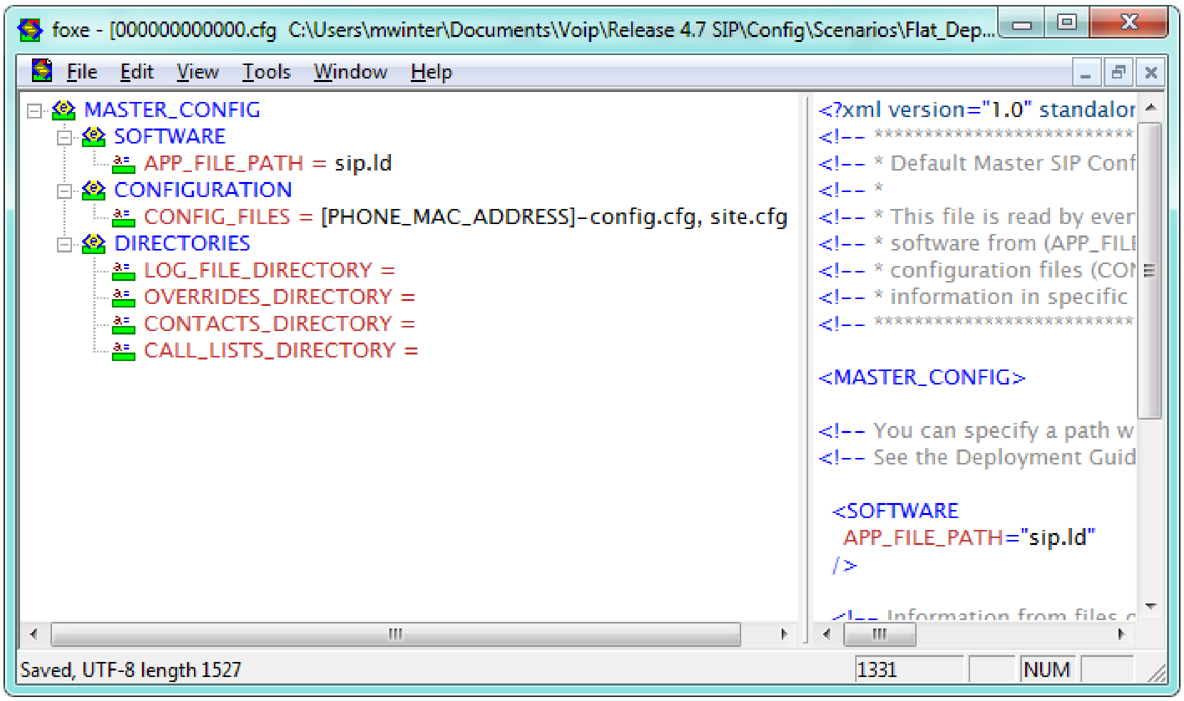
8 Once the handset reboots, disconnect the USB cable from 84xx handset and allow it to download the rest of its configuration files from the provisioning server.

9 Test the first few handsets to be sure your configuration is working as desired.

This section shows how to configure the 8400 telephone to register with the CS 1000 SIP Line gateway. Each phone is deployed to a specific extension and all phones have similar parameters. In this deployment, phones are typically linked to extensions which are then assigned to users. You will need to create one .cfg file for each extension/user.

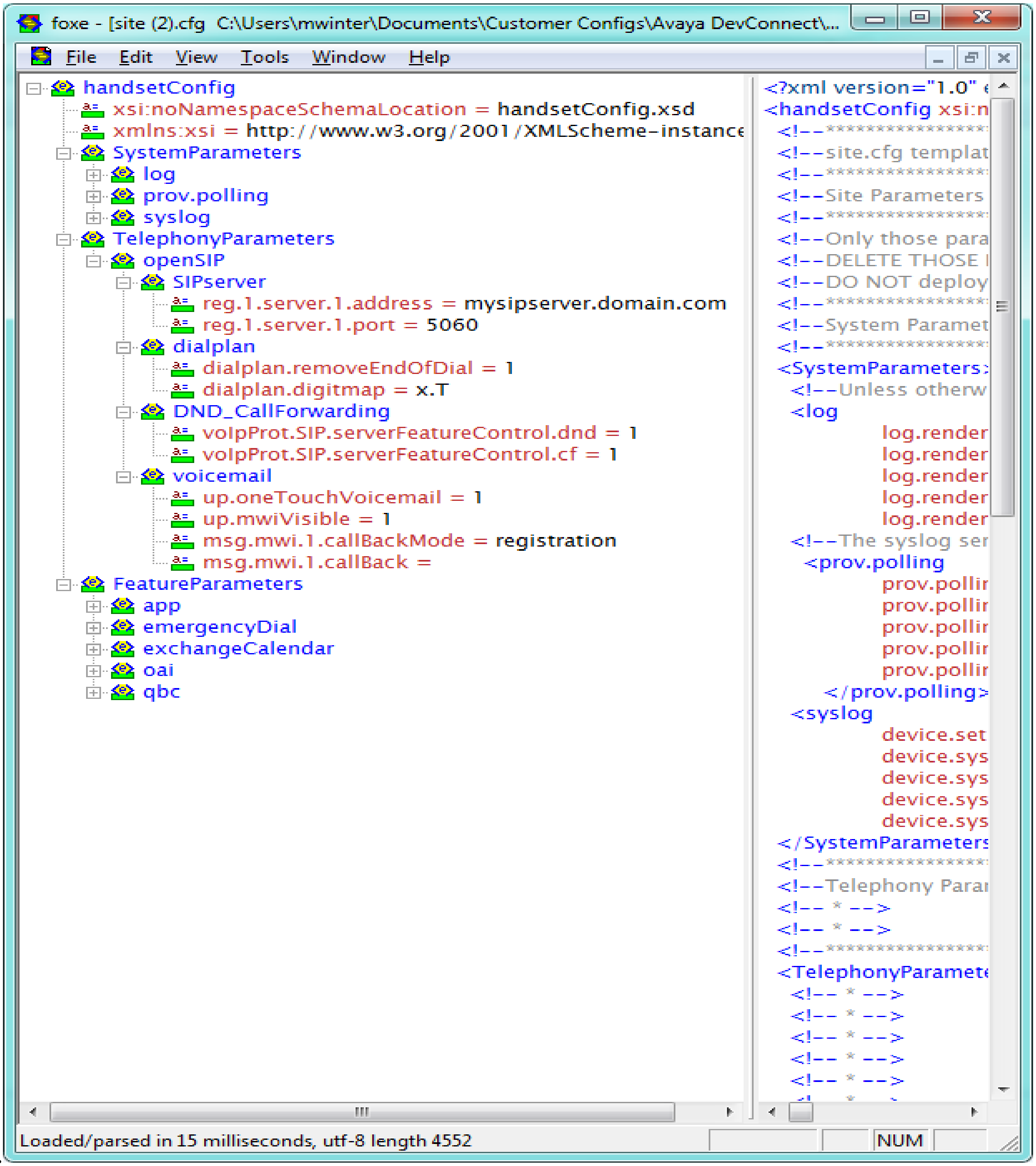
In our example, these three files are provisioned:

- 000000000000.cfg



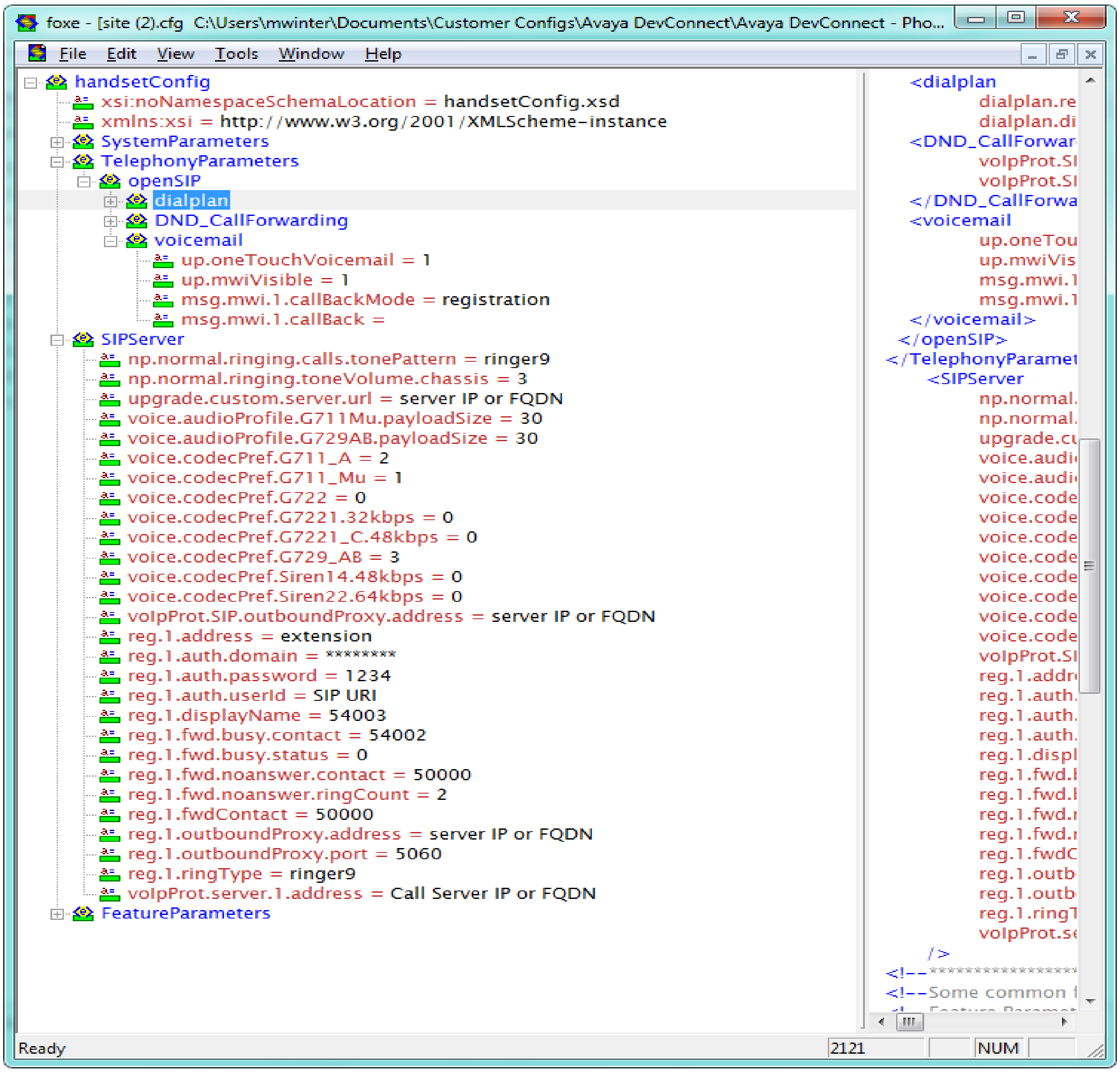
- Site.cfg

The site.cfg template contains most common parameters including network and telephony information that pertains to all of the handsets, such as SIP servers, dial plan, etc.



□ <MACaddress>-ext.cfg (one file for each extension/handset)

You must create a specific <MACaddress>-ext.cfg file for each phone/extension you deploy. The User spreadsheet you completed that lists each extension/user and the MACaddress of the phone assigned to that extension will help you create these files.
These files must be named with the identical structure as the variable used by the phone to find it. Therefore when the variable [PHONE_MAC_ADDRESS]-ext.cfg is used, the phone-specific files must be named <MACaddress>-ext.cfg.You will use the MACaddress-ext.cfg template to create the files for each extension/user. It contains the most common parameters, including network and telephony information, that pertain to all of the handsets, such as SIP servers, dial plan, etc.



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.